Design of Hearing Aid Using Variable Bandwidth Filter

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Design of Hearing Aid Using Variable Bandwidth Filter

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Abstract— The auditory system is a very sensitive and complex network. Diseases, drugs, noise, trauma and aging may have resulted in varying degrees of hearing loss, which makes hearing impairments one of the most common sensory disturbances in the world. A reconfigurable filter was designed based on the Acoustic Signal (AS) and the filter bank consists of Dyadic wavelet transform(DWT), Recursive Least Square filter (RLS),Inverse Short time Fourier transform (ISTFT) and Dimensional lifting Discrete Wavelet Transform filters. Different kinds of Acoustic Signal (AS) can be acquired under different environmental conditions (indoor, outdoor). The indoor and outdoor Acoustic Signal is processed in Frequency Range Reconfigurable (FRR) filter banks. The different type of Acoustic Signal is classified based on the frequency range and Bandwidth. The proposed method can be applied to 3m/5m/10m distance between source and hearing aid acquired signal with different environmental conditions. The proposed system is more efficient when compared with existing DSP based hearing aid. In this paper, a reconfigurable FRR filter bank is proposed with small complexity and acceptable delay.

Index Terms— Dyadic wavelet transform,Reconfigurable,Inverse Short time Fourier transform, RLS,FRR

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1 INTRODUCTION

The auditory system is a very sensitive and complex network. The most effective way to compensate hearing loss is to employ a hearing aid system which is an integration of voice amplification, noise reduction, feedback suppression, automatic program switching, environmental adaptation, and etc. The basic function of a hearing aid system is to amplify sounds selectively and then transfer the processed signal to the ear. A reconfigurable filter was designed based on Acoustic Signal. The filter bank consists of Dyadic wavelet transform(DWT), Recursive Least Square filter (RLS),Inverse Short time Fourier transform (ISTFT) and Dimensional lifting Discrete Wavelet Transform filters.

With the development of hearing aid technology, new demands to hearing aid systems appear. A reconfigurable filter bank which employed the frequency response masking technique was proposed. The complexity of the filter bank was low while the throughput is too long to be used in practice. Though the adjustable filter banks meet the new trends in digital hearing aids, the efficiency and effectiveness of such kind of algorithms needs to be further improved.

In existing system the filter banks are available only in DSP based hearing aid. In analog aid, Acoustic signal is amplified. Acoustic signal pass through all the filters in Filter bank, still selection of filter in filter bank is not based on the type of input acoustic signal. In existing hearing aid system , single filter bank is used.

Signal processing algorithms such as Discrete Fast Fourier Transform and Fast Fourier Transform to decompose the input frequency band into several small bands so that desired band of frequency can be filtered and reconstructed. These algorithms provide the flexibility in the hearing aid for the different level of hearing loss.In existing system,Irritation may be felt while hearing and Sound Quality is poor.

Syed ShabihHasan et.al [1] proposed the problem of creating predictive models for hearing aid outcomes that incorporate information about auditory abilities, hearing-aid features, and auditory contexts. These models are built on a collected dataset using a mobile phone application that measures auditory contexts and hearing aid outcomes using Ecological Momentary Assessments. The use of a mobile application allowed us to collect fine-grained hearing aid outcome measures in different auditory contexts.

John Dzarnoski et.al[2] proposed the use of embedded die packaging (or chip-in-flex) to drive significant further size reduction in custom and standard hearing instruments over what can be achieved using chip-on-flex or ceramic hybrid based technologies. The performance improvement, size reduction, changes in supply chain, impact on wafer test, impact on device test and challenges of working with wafers instead of die will be discussed. This paper descussed the results of extensive reliability testing including accelerated aging, thermal shock, pad integrity, drop tests, moisture sensitivity, ESD testing, light sensitivity and hearing aid assembly solder simulation testing.

S. Leeudomwonget.al[3] proposed the characteristics of hearing aids should be determined and regulated to ensure patient safety and to optimize the performance of the hearing aids. The objectives of this work were to design and develop a hearing aid measurement system to obtain the performance characteristics of the hearing aid. The implementation of this work is based on the method recommended in the International Electro technical Commission (IEC 60118-7). To verify the performance of the developed hearing aid measurement system, the hearing aid characteristics measured from developed measurement system (such as Maximum OSPL90, Frequency of Maximum OSPL90, HFAOSPL90,Maximum Full-on Gain, Frequency of FOG Maximum, HFA-FOG, Lowest Frequency, Highest Frequency, RTG @ RTS/60 dB, THD 500 Hz, THD 800 Hz and THD 1600 Hz) were compared with those measured from the commercial hearing aid analyzer and those provided from the hearing aid manufacturer's specifications.

AshutoshPandey et.al [4] proposed that digital hearing aids identify acoustic feedback signals and cancel them continuously in a closed loop with an adaptive filter. This scheme facilitates larger hearing aid gain and improves the output sound quality of hearing aids. However, the output sound quality deteriorates as the hearing aid gain is increased. They presented two methods to modify the forward path gain in digital hearing aids. The first approach employs a variable, frequency-dependent gain function that is lower at frequencies of the incoming signal where the information is perceptually insignificant. The second method automatically identifies and suppresses residual acoustical feedback components at frequencies that have the potential to drive the system to instability. The suppressed frequency components are monitored and the suppression is removed when such frequencies no longer pose a threat to drive the hearing aid system into instability.

RyotaShimokura et.al [5] proposed the cartilage conduction hearing aid has been developed for patients who cannot use conventional hearing aids owing to particular diseases of the external or middle ear. A user of such a hearing aid places the ring-shaped transducer gently at the entrance of the external auditory canal, so that it does not cause any feeling of discomfort. The cartilage conduction transducer vibrates the aural cartilage of the user and the transmitted vibration generates sound in the external auditory canal, particularly in the low and middle-frequency ranges. In this regard, the cartilage conduction hearing aid has a different principle of sound transmission from conventional hearing aids.

AshutoshPandey et.al [6] proposed that the acoustic feedback limits the gain provided by hearing aids. Digital hearing aids identify acoustic feedback signals and cancel them continuously in a closed loop with an adaptive filter in the digital domain. This scheme facilitates larger hearing aid gain and improves the output sound quality of hearing aids. However, the output sound quality of hearing aids deteriorates as the hearing aid gain is increased. This paper automatically identifies and suppresses residual acoustical feedback components at frequencies that have the potential to drive the system to instability. The suppressed frequency components are monitored and the suppression is removed when such frequencies no longer pose a threat to drive the hearing aid system into instability. Experimental results obtained with real world hearing aid gain profiles using speech and music signals indicate that the method provides less distortion in the output sound quality than classical feedback cancellers enabling the use of more comfortable style hearing aids for patients with moderate to profound hearing loss.

PengQiao et.al [7] proposed that the design and optimization of a digital hearing aid application. It aims to show that a suitably adapted ASIP can be constructed to create a highly optimized solution for the wide variety of complex algorithms that play a role in this domain. These algorithms are configurable to fit the various hearing impairments of different users. They proposed significant challenges to digital hearing aids, having strict area and power consumption constraints. Hideyuki Takagi et.al[8] proposed an interactive evolutionary computation (EC) fitting method that applies interactive EC to hearing aid fitting and the method is evaluated using a hearing aid simulator with human subjects. The advantages of this method is to optimize a hearing aid based on how a user hears and that it realizes whatever + whenever + wherever (W3) fitting.

II.PROPOSED SYSTEM

The different type of Acoustic Signal is classified based on the frequency range and Bandwidth. The data signal is acquired from 24 bit, 48000 Hz .The two wires in the microphone and speaker are the data and ground wire. A audio jack pin with 2 channel connected to the DAQ card. The ground wires from microphone and speaker is commonly grounded in the audio jack ground pin.

A database of these acquired signals under different environmental conditions such as male voice, female voice and music and also based on distance is created. According to deaf persons,there would be three levels of classifications(i.e.) good, bad and better. This classification results arrived after hearing the filter signal and raw signals separately and also varying in 3/5/10m distance. The classification for good, bad &better are in 0-5 point scaling, results are obtained from 20 deaf persons rating.

The same method is applied to 3m/5m/10m distance between source and hearing aid acquired signal with different environmental conditions. For the acquired signals, the frequency range and band values are calculated. Based on the values, various filters are applied and evaluated for their performance statistically and qualitatively. After the Hearing aid microphone signal(HAM) passes through the filter is statistically evaluated with LSSVM as shown in block diagram evaluated with the filtered signal from speaker(HAS) and raw signal are also analysed for their performance. These HAM, HAS, raw signal are also examined for their performance based on audio interpretation.The proposed system used for the avoidance of surgery and surgical complications.

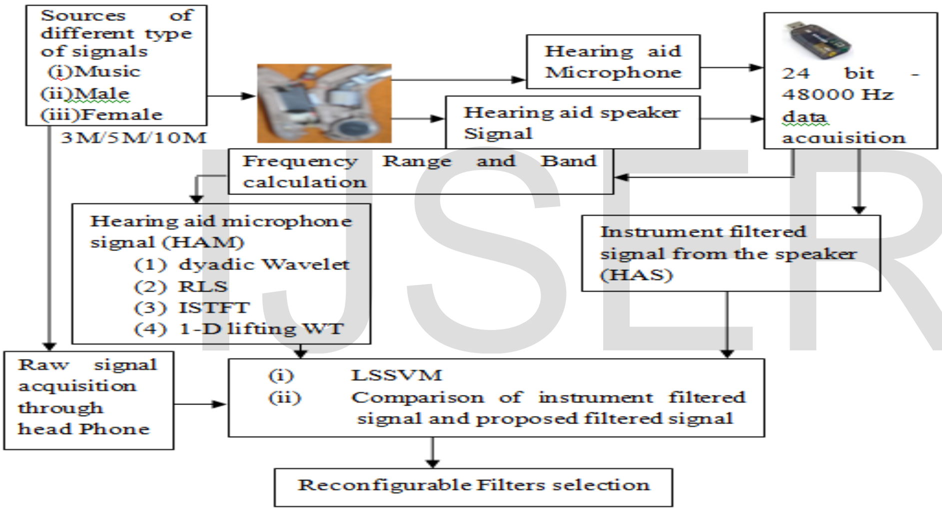

Fig.PROPOSED BLOCK DIAGRAM

DYADIC WAVELET TRANSFORM:

Dyadic wavelet transforms are scale samples of wavelet transforms following a geometric sequence of ratio 2. Time is not sampled. This transform uses dyadic wavelets.It is implemented by perfect reconstruction filter banks.

The family of dyadic wavelets is a frame of L2(R).To build dyadic wavelets, it is sufficient to satisfy the previous condition. To do so, it is possible to proceed as for the construction of orthogonal and biorthogonal wavelet bases, using conjugate mirror or perfect reconstruction filter banks. The wavelets satisfy then scaling equations and the fast dyadic wavelet transform is implemented using filter banks.

The fast dyadic wavelet transform uses the same filters for the computation of the fast wavelet transform of a discrete signal, except that no subsampling is performed.

RECURSIVE LEAST SQUARE TRANSFORM

The Recursive least squares (RLS) is an adaptive filter which recursively finds the coefficients that minimize a weighted linear least squares cost function relating to the input signals. This is in contrast to other algorithms such as the least mean squares (LMS) that aim to reduce the mean square error. In the derivation of the RLS, the input signals are considered deterministic, while for the LMS and similar algorithm they are considered stochastic. Compared to most of its competitors, the RLS exhibits extremely fast convergence. However, this benefit comes at the cost of high computational complexity.

RLS was discovered by Gauss but laid unused or ignored until 1950 when Placket rediscovered the original work of Gauss from 1821. In general, the RLS can be used to solve any problem that can be solved by adaptive filters. The benefit of the RLS algorithm is that there is no need to invert matrices, thereby saving computational power. Another advantage is that it provides intuition behind such results as the Kalman filter. compared to least mean squares (LMS) algorithms, recursive least squares (RLS) algorithms have a faster convergence speed and do not exhibit the eigen value spread problem. However, RLS algorithms involve more complicated mathematical operations and require more computational resources than LMS algorithms.

ONE DIMENSIONAL LIFTING WAVELET TRANSFORM:

This transform can be used to produce integer to integer wavelets transformation .This eliminates the need for scaling the coefficients for processing. The lifting scheme reduces the computational complexity of the discrete wavelet transform (DWT) by factoring the wavelet filters into cascades of simple lifting steps that process the input samples in pairs. We propose four compact and efficient hardware architectures for implementing lifting-based DWTs, namely, one-dimensional (1-D) and two-dimensional (2-D) versions of what we call recursive and dual scan architectures.

The 1-D recursive architecture exploits interdependencies among the wavelet coefficients by interleaving, on alternate clock cycles using the same data path hardware, the calculation of higher order coefficients along with that of the first-stage coefficients. The resulting hardware utilization exceeds 90% in the typical case of a five-stage 1-D DWT operating on 1024 samples. The 1-D dual scan architecture achieves 100% data path hardware utilization by processing two independent data streams together using shared functional blocks.

The main feature of the lifting scheme is that all constructions are derived in the spatial domain. It does not require complex mathematical calculations that are required in traditional methods. Lifting scheme is simplest and efficient algorithm to calculate wavelet transforms 81 .It does not depend on Fourier transforms. Lifting scheme is used to generate second-generation wavelets, which are not necessarily translation and dilation of one particular function. It was started as a method to improve a given discrete wavelet transforms to obtain specific properties. Later it became an efficient algorithm to calculate any wavelet transform as a sequence of simple lifting steps. Digital signals are usually a sequence of integer numbers, while wavelet transforms result in floating point numbers. For an efficient reversible implementation, it is of great importance to have a transform algorithm that converts integers to integers. Fortunately, a lifting step can be modified to operate on integers, while preserving the reversibility. Thus, the lifting scheme became a method to implement reversible integer wavelet transforms. Constructing wavelets using lifting scheme consists of three steps:

The first step is split phase that split data into odd and even sets.
The second step is predicting step, in which odd set is predicted from even set. Predict phase ensures polynomial cancellation in high pass.
The third step is update phase that will update even set using wavelet coefficient to calculate scaling function.

INVERSE SHORT TIME FOURIER TRANSFORM

The short-timeFourier transform (STFT), or alternatively short-term Fourier transform, is a Fourier-related transform used to determine the sinusoidal frequency and phase content of local sections of a signal as it changes over time. In practice, the procedure for computing STFTs is to divide a longer time signal into shorter segments of equal length.

Fourier transform separately on each shorter segment. This reveals the Fourier spectrum on each shorter segment. One then usually plots the changing spectra as a function of time. The STFT is invertible, that is, the original signal can be recovered from the transform by the Inverse STFT. The most widely accepted way of inverting the STFT is by using the overlap-add (OLA) method, which also allows for modifications to the STFT complex spectrum. This makes for a versatile signal processing method, referred to as the overlap and add with modifications method.

One of the pitfalls of the STFT is that it has a fixed resolution. The width of the windowing function relates to how the signal is represented—it determines whether there is good frequency resolution (frequency components close together can be separated) or good time resolution (the time at which frequencies change).

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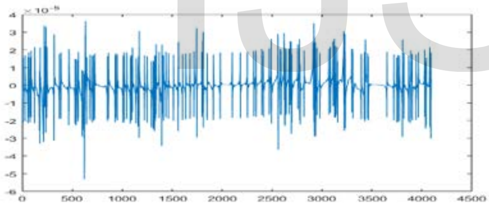
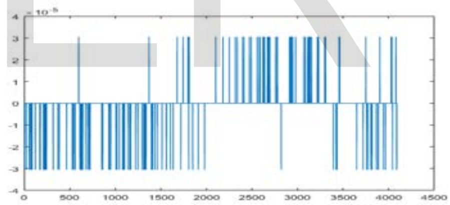
A wide window gives better frequency resolution but poor time resolution. A narrower window gives good time resolution but poor frequency resolution. These are called narrowband and wideband transforms, respectively. STFTs as well as standard Fourier transforms and other tools are frequently used to analyze music. The spectrogram can, for example, show frequency on the horizontal axis, with the lowest frequencies at left, and the highest at the right.

The height of each bar (augmented by color) represents the amplitude of the frequencies within that band. The depth dimension represents time, where each new bar was a separate distinct transform. Audio engineers use this kind of visual to gain information about an audio sample, for example, to locate the frequencies of specific noises (especially when used with greater frequency resolution) or to find frequencies which may be more or less resonant in the space where the signal was recorded. This information can be used for equalization or tuning other audio effects.

DYADIC WAVELET TRANSFORM RESULT:

INPUT IMAGEOUTPUT IMAGE





Using this transform, scaling and convolution are performed without subsampling process.So that the edges are clearly obtained.

LEAST SQUARE SUPPORT VECTOR MACHINE

Least squares support vector machines (LS-SVM) are least squares versions of support vector machines(SVM), which are a set of related supervised learning methods that analyze data and recognize patterns, and which are used for classification and regression analysis. In this version one finds the solution by solving a set of linear equations instead of a convex quadratic programming (QP) problem for classical SVMs. Least squares SVM classifiers, were proposed by Suykens and Vandewalle.

LS-SVMs are a class of kernel-based learning methods. LS-SVM capabilities are controlled by the choice of the kernel function. This allows the user to solve complex pattern recognition and regression problems by choosing the appropriate kernel (e.g. linear, gaussian, sigmoid). The LS-SVM is easy to analyze as the solution can be expressed analytically in terms of the parameters and sample first and second order statistics.

III.EXPERIMENTAL RESULTS

ONE DIMENSIONAL LIFTING WAVELET

TRANSFORM RESULT:

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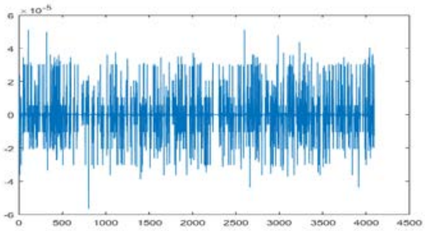
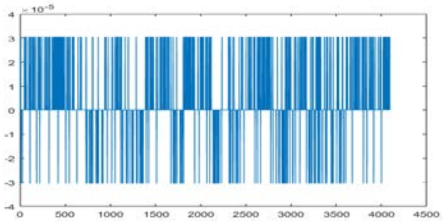
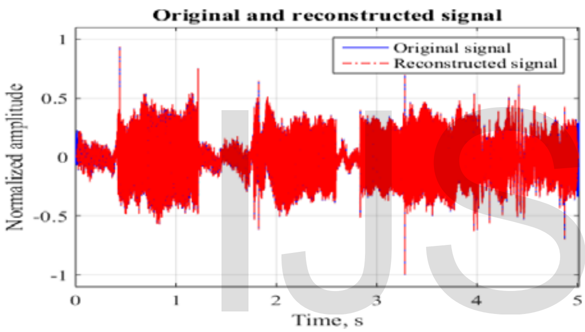


INPUT IMAGEOUTPUT IMAGE



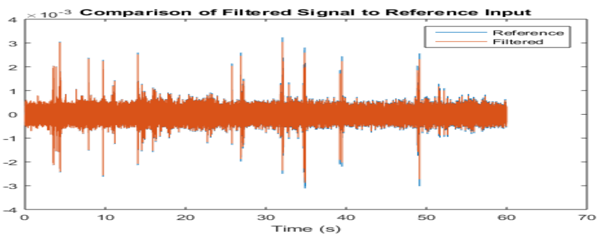
This transform is used to produce integer to integer wavelets transformation. It eliminates the need for scaling the coefficients for processing.

INVERSE SHORT TIME FOURIER TRANSFORM RESULT:



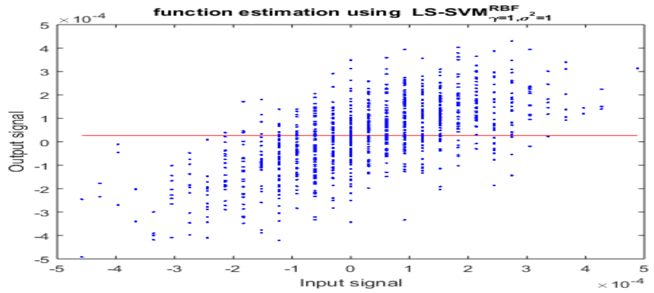
This transform is used to divide a longer time signal into shorter segments of equal length. It decomposes the incoming signals into their constituent frequencies.

RECURSIVE LEAST SQUARE TRANSFORM RESULT :



Using this transform, noise cancellation is achieved.It is mainly selected for audio signal processing.

LEAST SQUARE SUPPORT VECTOR MACHINE RESULT :



LSSVM is used for classification and regression analysis.When the incoming signal is close to the reference line,we can hear the signal clearly.

IV. CONCLUSION

The proposed system addresses the problem of processor for automatic selection of the filter in FRR filter bank to compensate the losses, due to various distances and environmental conditions between source and hearing aid and also for different type of signals. The proposed filter bank can meet different needs of hearing loss cases with acceptable frequency range.

The proposed FRR filter design selects the filter based on the frequency range and frequency bandwidth of the input signals. This process reduces the harmonic less than 3 %. The major problem of battery replacement can be avoided since the signals are not passed through all the filter in the filter bank. The proposed FRR gives a clear sound and loudly under any environmental conditions with respect to distance. From overall experimental results , we conclude that different type of signals under different environmental conditions are filtered through frequency range based filters improves the quality of hearing sound compared to the existing hearing aids.

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