A Mel- Filter and Kepstrum based algorithm for noise suppression in cochlear implants

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Abstract- In this paper, a novel algorithm is proposed that uses the concepts of Kepstrum approach and Mel filtering for processing of speech signals for cochlear implants. Here the speech signals are first divided into various frames of approximately 10 mS duration. FFT and periodogram of each frame is calculated after which the signal is passed through the Mel filter bank. Then the kepstrum coefficients are calculated, which are used for estimation of the coefficients of the whitening filter. This filter is used for the noise suppression of the input speech signal. The algorithm is tested on various speech signals and the SNR for the speech signal before and after processing are tabulated. The results show that there is a good amount of increase in the SNR which indicates the efficiency of the algorithm in noise suppression.
Keywords: Kepstrum approach, Mel filters

I. INTRODUCTION

Cochlear implants are auditory prosthesis that is used by severe, profound and bilaterally deaf people. In certain disorders related to the vestibulocochlear nerve like vestibular schwannoma, the patients become bilaterally deaf. For such patients, cochlear implants are suggested [11]. These devices stimulate the nerve endings of the vestibulocochlear nerve according the frequency of the speech signal. There are several noise cancellation techniques used in the speech processor of a cochlear implant [6][8]. This paper introduces a new technique based on kepstrum approach for noise cancellation in speech processor of cochlear implants. Kepstrum is the acronym for "*Kolmogorov equation for power series*", [5] which are, to some extent, similar to cepstrum. The main difference between kepstrum and cepstrum is that the latter takes into consideration, only the real parts of the signal, while the former takes into account both the real and complex parts of the signal. Kepstrum method has been successfully implemented for generation of characteristic equations of Wiener and Kalman filters [5] [9]. It has also been proven that kepstrum method can be used to calculate the acoustic transfer functions of two different acoustic sources, say two microphones [2]. Kepstrum has also been implemented for noise cancellation and speech enhancement of two microphone systems [1]. So, an attempt was made to implement it for speech processing in cochlear implants and good results were obtained. In kepstrum analysis, the periodogram of the signal is calculated after the signal is converted into frequency domain with the help of fast fourier transform (FFT). After calculation of periodogram, the signal is converted in Logarithmic domain by taking natural log and a bias equal to Euler's constant (γ) is added to the signal. The kepstrum coefficients are obtained by taking the IFFT of the signal. Fig 1 shows the block diagram for calculation of kepstrum coefficients

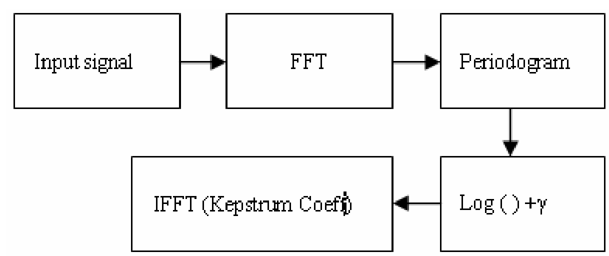

Fig 1: Block diagram for calculating Kepstrum Coefficients

Using the kepstrum coefficients calculated as above, the equation of whitening filter is calculated and applied to the signal. This suppresses the white Gaussian noise.

II. ALGORITHM

General block diagram of the speech processor of a cochlear implant is as shown in fig 2. In real time, a microphone is used to collect the input signal and it is sent to the pre-filters. These are usually low pass filters which filter out the high frequency noises. Then the signal is passed through filter banks which separate the signal into various frequency bands. These signals are log compressed and then the envelop is detected before modulation and finally transmitting it to the implanted section of the cochlear implants where the signal is received, again filtered and finally sent to the electrode array for stimulation.

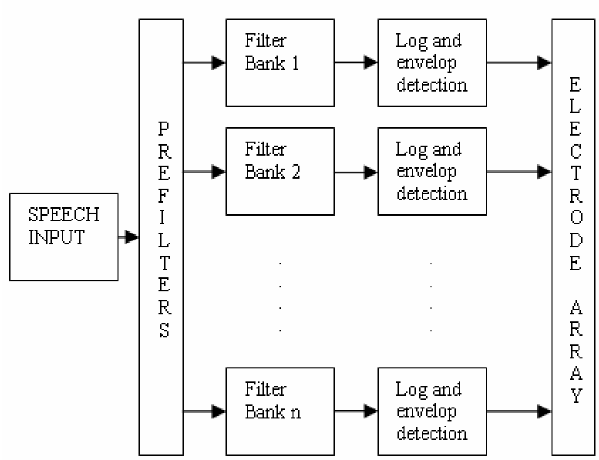

Fig 2: General block diagram of speech processor used in cochlear implants[6]

The proposed algorithm does slight modification to the general speech processor. Here first of all the signal is passed through low pass Chebyshev filter of order 15 for pre-filtering purpose. After pre-filtering, calculate the fast fourier transform of the signal. If x(n) is the speech signal after pre-filtering, then FFT is given by [12]

$$X(k)= \sum_{n=0}^{N-1} x(n)W_N^{ik}, \quad k= 1,2,3,\ldots,N \qquad \ldots\ldots\ldots\ldots\ldots(1)$$

The periodogram of the X(k) is calculated as

$$S= 1/N\,|X(k)|^2 \qquad \ldots\ldots\ldots\ldots\ldots(2)$$

The Mel-filter bank is defined as [4]

0 if k<f(m-1) and
if k>f(m+1)
h(k)={ k-f(m-1)/f(m)-f(m-1)
if f(m-1)<=k<=f(m) ……(3)
f(m+1)-k/f(m+1)-f(m) if f(m)<k<=f(m+1)

Where f(m) is the centre frequency. Mel filter bank is chosen for this purpose because it closely resembles the human cochlea in characteristics. In Mel filter bank, just like the human cochlea, the filter characteristics are linear till 1 kHz and logarithmic above 1 kHz. The commercially available electrode arrays have fixed distance between two electrodes. So it is wise to design the speech processor according to the distance of electrode from the apex of cochlea [3]. Hence with the distance known, the centre frequency is calculated using Greenwood function [10] which is given as

$$F_{centre}= B(10^{Ax}-1) \qquad \ldots\ldots\ldots\ldots\ldots\ldots(4)$$

Greenwood established that for human beings, B= 165.4 and A= 2.1. Using these details, the centre frequencies (in Hz) for Mel filter were calculated as

f = [250,375,500,625,750,100,1125,1250,1437.5,1687.5,1937.5, 2187.5, 2500, 2875, 3312.5, 3812.5, 4375, 5000, 5687.5, 6500, 7437.5]

Fig 3 shows the overlapping triangular Mel-filter bank

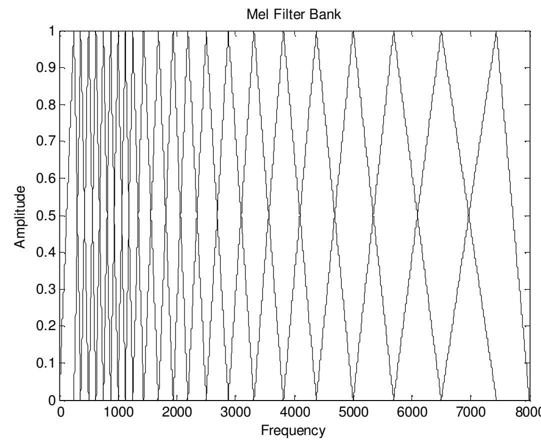

Fig 3: Overlapping Triangular Mel filter banks.

The signal is then passed through Mel-filter bank and the output is compressed by taking the logarithm of the signal and finally the discrete cosine transform. This is termed as Mel-frequency kepstral coefficients denoted by K.

The kepstrum coefficients are kept till a value L<= N/2-1, where N is the length of sequence and the first kepstrum coefficients is reduced to half of its original value. This halved first coefficient is used to calculate the innovations variance by the formula

$$\sigma_E = \exp(2K_0) \qquad \ldots\ldots\ldots\ldots\ldots\ldots(5)$$

Using the truncated kepstral coefficients, the Kolmogorov equation for power series is calculated as

$$nP(n)= - \sum_{r=1}^{n} rP(n-r)\,K(r), \quad n= 1,2,3,\ldots\ldots,L \quad \ldots\ldots..(6)$$

The innovations estimate of the whitening filter is calculated as

$$E(k) = \sum_{n=1}^{L} P(n)\,x(k-n) \qquad \ldots\ldots\ldots\ldots\ldots...(7)$$

Finally the output is reconstructed by the equation

$$Y(k) = x(k) - \sigma_V^2 / \sigma_E^2 \left(\sum_{n=1}^{d} P(n)\,E(k+n)\right) \qquad \ldots\ldots\ldots(8)$$

The summary of the steps involved in the proposed algorithm are as follows:

1) Pass the signal through a low pass Butterworth filter of the order 15

2) Sample the signal into 10 mS frames
3) Calculate the fourier transform of each frame
4) Calculate the periodogram and concatenate the signal
5) Design Mel-filter banks and separate the signal into 22 bands with length equal to the filter bank length and filter the signal by passing it through the filter bank
6) Compress the signal by taking Logarithm of the signal
7) Take the inverse fourier transform to get Mel-frequency kepstral coefficients
8) Calculate the coefficients of the power series polynomial
9) Calculate the innovations of the whitening filter using the coefficients of power series equation
10) Apply this whitening filter to each of the 22 bands

For stimulation of the electrode array, envelop of the output of 22 bands is detected. This envelop will hence be modulated and transmitted to the implanted part of the implant. The block diagram of the proposed algorithm is as shown in figure 4.

III. SIMULATION RESULTS

Proposed algorithm is developed in MATLAB. For testing purpose, white Gaussian noise was added to input signal and the SNR before and after processing has been tabulated.
Several speakers were asked to speak different words and their voices were recorded using Computerized Speech Laboratory (CSL). These recorded signals were then corrupted with white Gaussian noise and fed to the algorithm and the performance of the algorithm was estimated.
Table 1 shows the SNR values of few subjects for the word “Flower”. The subjects were both males and females. The algorithm shows effective reconstruction in the speech signal irrespective of the gender of the speaker.
From the values in table 1, it is clear that the algorithm shows good improvement in the signal to noise ratio after the processing it through kepstrum approach.

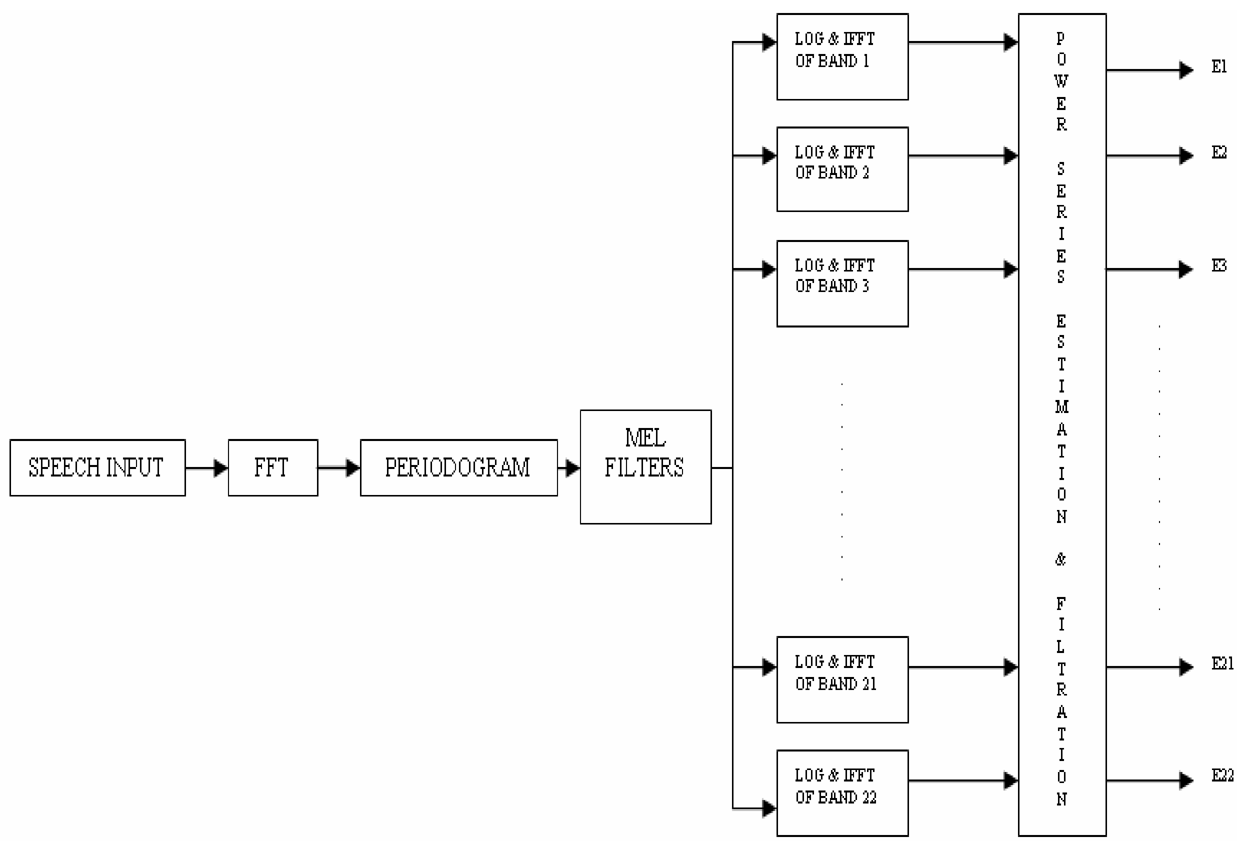

Fig 4: Flow diagram of the proposed algorithm

Table 1- SNR values for some of the male subjects

SUBJECT	SNR1* (dB)	SNR2** (dB)	ENHANCEMENT (dB)
1	10.53	40.26	29.73
2	11.29	38.63	27.34
3	8.53	39.23	30.70
4	10.20	42.10	31.90
5	9.56	39.06	29.50
6	9.72	41.19	31.47
7	9.01	36.09	27.08
8	11.45	35.43	23.98

** SNR of signal+noise, ** SNR of the processed signal*

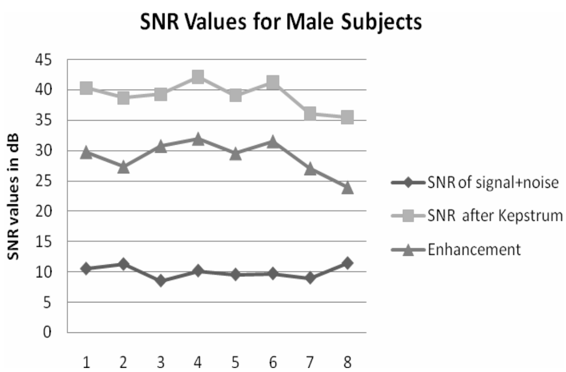

Fig 5: Plot showing the SNR values for male subjects

Table 2- SNR values of some of the female subjects

SUBJECT	SNR1* (dB)	SNR2** (dB)	ENHANCEMENT (dB)
1	9.53	36.60	27.07
2	10.86	32.63	21.77
3	9.58	43.23	33.65
4	10.58	45.80	35.22
5	9.05	39.73	30.68
6	8.56	40.09	31.53
7	9.87	39.43	29.56
8	10.89	34.85	23.96

*SNR of signal+noise ** SNR after Kepstrum

The following figures show the simulation results obtained. The first one is the original speech wave, the second is the one corrupted with Gaussian noise with 1 dB power.

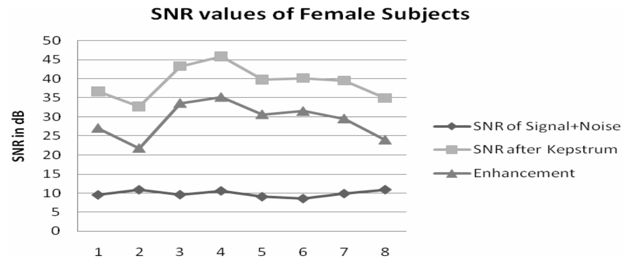

Fig 6: Plot showing the SNR values of female subjects

From the waveform itself, it is evident that there is very effective noise cancellation and the output nearly resembles the desired waveform with effective increase in the power.

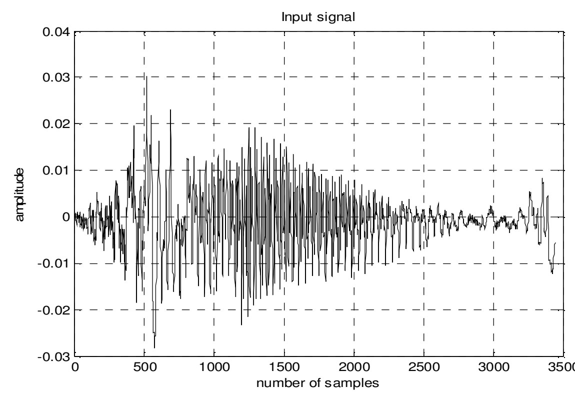

Fig 7: Original signal for the word

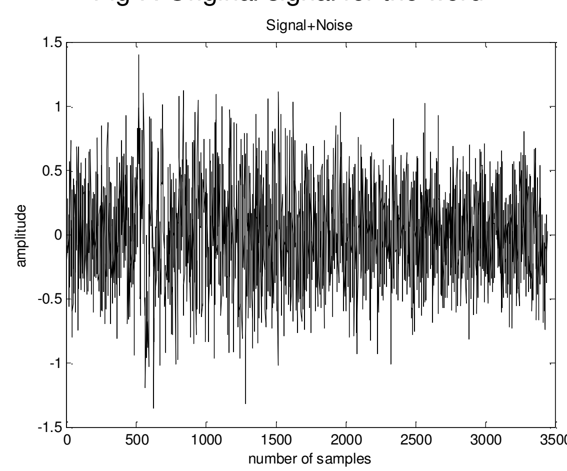

Fig 8: Signal corrupted with Gaussian noise

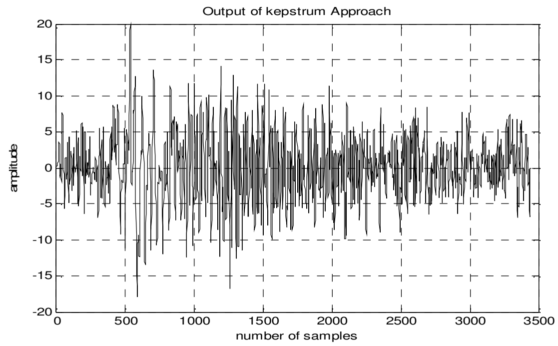

Fig 9: Output after kepstrum approach

The output for the waveform “Hello” is as shown in the following figures.

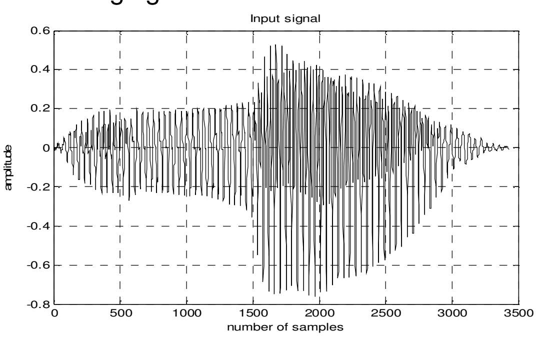

Fig 10: Input waveform for “Hello”

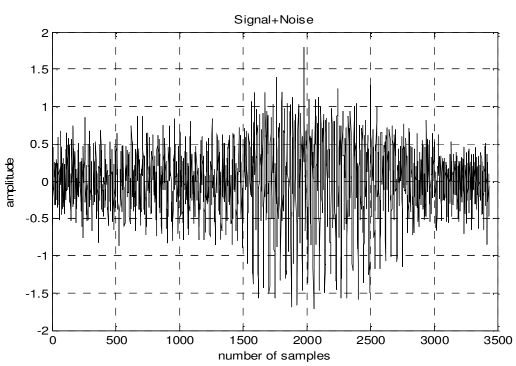

Fig 11: Signal corrupted with white Gaussian noise

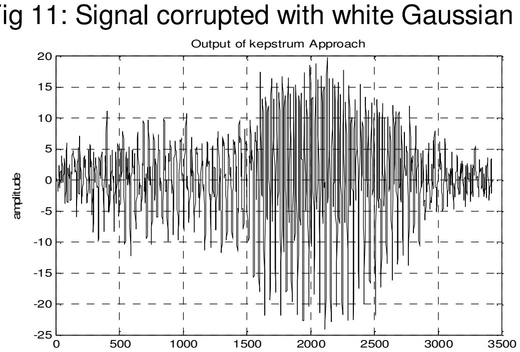

Fig 12: Output after kepstrum approach

CONCLUSION AND FUTURE WORK

By the simulation results, it can be inferred that the proposed algorithm provides good noise suppression and speech enhancement, thus making it suitable for noise suppression in speech processors used in cochlear implants. In future, a simulink model of the algorithm can be formulated and the performance of the algorithm can be analyzed. Also it can be combined with various speech processing algorithms that are in use for cochlear implants and check if it aids in the betterment of the performance of the existing algorithm.

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