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Investigation of Cochlea Response Based Signal Processing

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Abstract An efficiency of cochlea has a significant contribution in a quality of human hearing and in a language development of newborn. The efficiency of cochlea clinically measured using Distortion Product Otoacoustic Emission (DPOAE). The measurement mainly restricted by acoustic interferences that disrupt response estimation. The disrupted estimation gives rise to repetition in measurement for many times or probably inaccurate efficiency assessment. In this study, investigation of cochlea response estimation was considered based on signal processing (SP), which was regarded as preliminary step toward interference reduction. An in-vivo measurement was performed on a left ear of 36 years old female with normal hearing, where cochlea stimulation and response recording was conducted using personal computer sound card in conjunction of sensitive probe system of ER-10C. The response signals were recorded and then analyzed off-line using SP of Fast Fourier Transform (FFT) and Band Pass-Finite Impulse Response (BP-FIR) filter. Results show that DPOAE frequency components can be extracted using proposed SP method at interference-free situation, where BP-FIR parameters of bandwidth 50Hz-200Hz and order 32-2048 have to be adjusted based on stimulation parameters. The findings dedicate useful investigated parameters for real-time implementation, and for further SP development at interference situation.

Keyword: Biomedical signal processing, Cochlea response estimation, Distortion product otoacoustic emission, Hearing test.

1. Introduction

Otoacoustic Emission (OAE) regarded as a noninvasive hearing screening modality which effectively provides information about cochlea activities [1,2]. OAE is a response of stimulated cochlea, which is

identified as a very low level of sound recorded at an external ear canal using sensitive microphone [3]. Healthy cochlea responses effectively, while weak response refers to dysfunction.
The OAE classified according to stimulation techniques as spontaneous and evoked OAEs. In clinic practice, evoked OAE of Transient Evoked OAE (TEOAE) and Distortion Product OAE (DPOAE) are routinely used. TEOAE produced by transient stimulus [4,5], while DPOAE elicited by two pure tones of f_1 and f_2 with ratio $\gamma>1$ ($f_2=\gamma f_1$)[6]. These two pure tones act as nonlinear interaction with the basilar membrane in the cochlea. As a result, an Outer Hear Cells (OHC) in the inner ear sends feedback to the basilar membrane that leads cochlea to amplify and produce series of combination of frequencies such of $2f_1$-f_2, $2f_2$-f_1, $3f_2$-f_1, $3f_1$-f_2. The most prominent of these series is the Cubic Difference Tone (CDT) of $2f_1$-f_2 [7-9].
In noisy background, the CDT occurred at low level that immersed under the frequency components of two f_1 and f_2. However, a robust signal processing was introduced to extract cochlea low level response of CDT[7–10].
In the literature, many methods were presented to estimate the DPOAE, Ziarani [10] proposed nonlinear adaptive method to extract the DPOAE with high degree of immunity and short measurement time. While, Choi [8], proposed another signal (SP) technique to extract DPOAE represented by heterodyne method after stimulus with swept frequency, his method showed high resolution of measurement, then Talmadage [11] used least mean- square-fit (LSF) filter to analyze DPOAE signal by continuously sweeping and given result with high resolution. However LSF filter requires long time recording. In addition, Jun Deng [3] investigated new method to evaluate the DPOAE by dynamic tracking filter (band pass filter and band stop filter), where dynamic tracking filter conducts an extraction of DPOAE at noisy situation.
In this study, the DPOAE technique is considered to investigate the effectiveness of frequency ratio of (γ=f2/f1) on conjunction of using FFT and BPF for cochlea response extraction. Results will be illustrated in a form of frequency spectrum for frequency range up to 12 kHz.

2. Methods

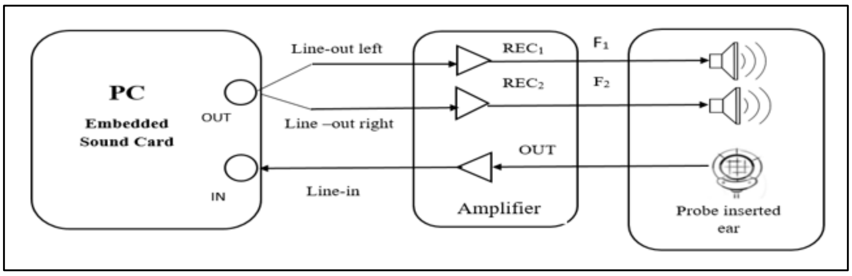
2.1. Subject

A left ear of normal hearing age of 36 years female volunteer (hearing threshold $\leq$ 20 dB HL) was examined by senior medical physicist at the center of hearing and communication of Baghdad medical city in Iraq, where examination included otoscope, tympanometry and audiometer as a preparation procedure for experiment conduction. Furthermore, the left ear canal cleaned before from wax and any blockage that probably prevents the conduction of the signals.

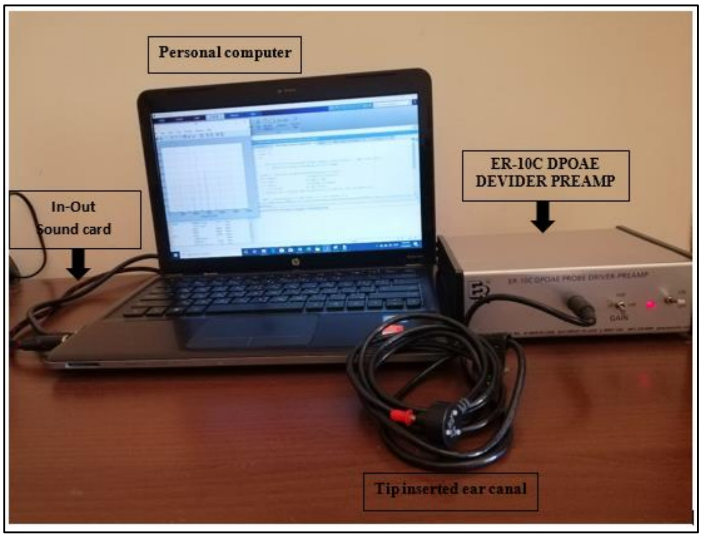
2.2. *Experiment procedure*

The DPOAE experiments were performed by generating the stimulation frequencies of two primaries for the range 500Hz-8kHz, for frequency ratios of 1.1, 1.22, 1.3, and 1.4, while the level of the primaries was considered to be L_1=L_2=65 dB SPL. A sound card (High definition audio device from Microsoft) embedded at personal computer was used as acquisition system for primaries generation and response recording. The *hp*-personal computer of Windows-10 operating system, pro 64-bit, Intel (R) core (TM) i3 processor CPU M380@ 2.53GHz, where sound generation and response recording at sampling frequency of 48 kHz. The DPOAE probe system of ER-10C (ER-10C Lo Noise™ DPOAE Probe system from E T Y M O T I C R E S E A R C H , I N C . 1) was used to deliver the primaries to the ear canal and also to record the cochlea response through a special fabricated probe.
The ER-10C consists of a low noise pre-amplifier that amplifies the response for 20dB and 40dB, where in the experiment 20dB was considered. The left and right terminals of PC sound card connected to the input-1 and input-2 of the ER-10C instrument.

The probe contains two speakers and sensitive microphone, inserted in patient's ear. The speakers used to deliver the combined frequencies to the inner ear for cochlea stimulation. In turn, cochlea responses to the ear canal in terms of intermodulation, which was recorded by the microphone, then converted to PC through ADC. The recorded signal stored for further analysis to estimate DPOAE. The analysis represented by processing and extraction as illustrated by system hardware shown in Figure 1.



(a)



(b)

Figure 1. System experiment represented by, (a) block diagram, (b) hardware connections.

2.3. Signal processing algorithm

The continuous-time response signal (cochlea response) converted to discrete-time signal using computer embedded sound card. The recorded signal then processed offline using Matlab 2014a.

The processing of recorded signal performed using Fourier analysis of FFT algorithm that's widely used in biomedical signal processing such as an extraction of heart signal of Electrocardiography (ECG), muscle signal Electromyography (EMG), and also DPOAE signals by differentiate the intermodulation distortion of cochlea activity in term of convert the signal from time domain to frequency domain [12,13] .

The Fourier analysis represented by Discrete Fourier Transform (DFT) can be represented as sequence of samples N [14] as follows:

$$X(k) = \sum_{n=0}^{N-1} x(n) W_n^{kN} \qquad 0 \le K \le N-1 \tag{1}$$

where, $x(n)$ is recorded discrete signal, K is frequency index, N is window length, $W_n = e^{-j2\pi/N}$.
The activity of healthy cochlea conducted on generation of distortion product frequencies such as CDT that demonstrated in Figure 2.

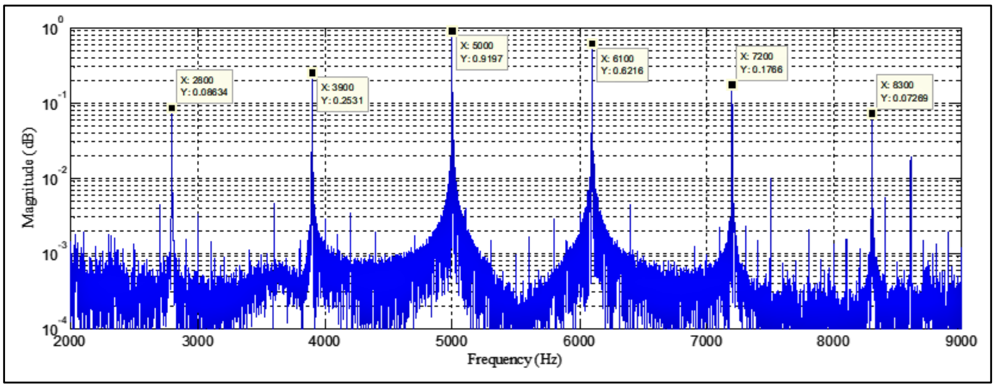


Figure 2. Spectrum of recoded signal.

Figure 2 shows the CDT of $2f_1$-f_2 in addition to other intermodulation distortion frequency components. The distortion product can be represented as follows [15],

$$x(n) = \sum_{i=0}^{3} k_i \cos(n\omega_i + \varphi_i) + l(n) \tag{2}$$

where x(n) is the recorded signal that contains frequency components of $k_1 cos(\omega_1 + \varphi_1), k_2 \cos(\omega_2 + \varphi_2)$ acts as stimulus frequencies, while $k_3 \cos(\omega_3 + \varphi_3)$ act as CDT component with $w_i = 2\pi f_i / f_s$, f_s represents sampling frequency of 48kHz, $l(n)$ represents additive noise, which can be removed using (SP) techniques[15]. As cochlea response contains multiple frequency components, the most common frequencies of ($2f_1$-f_2) and ($2f_2$-f_1) are considered for cochlea healthy sign. The desired tones analyzed by designing band pass digital filter that can be at narrow band, where center frequency at desired frequency component. As mentioned earlier, the digital filter used in this research was BP-FIR that has been considered as a stable system with linear phase response. The BP-FIR filter can be represented as differential equation with M order [16].

$$d(n) = \sum_{k=0}^{M-1} f_n w(n-k) = \sum_{k=0}^{M-1} b_k\, w(n-k) \tag{3}$$

The transfer function of BP-FIR filter

$$H(Z) = \sum_{k=0}^{M-1} b_k\, Z^{-k} \tag{4}$$

BP-FIR structure consists of three elements delay, multiplier and adder [16, 17] as shown in Figure 3.

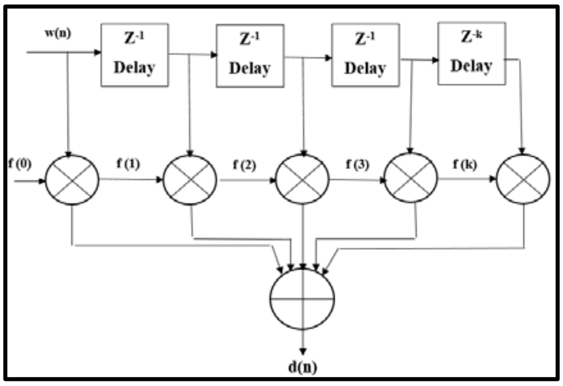


Figure 3. Basic structure of BP-FIR.

The output of digital filter $d(n)$ can be express mathematically by convolution of input *w(n)* with impulse response *f(n)* [18].

$$d(n) = w(n) * f(n) \tag{5}$$

and determined as:

$$d(n) = w(0)f(n) + w(1)f(n-1) + w(2)f(n-2) \ldots + w(n)f(0) \tag{6}$$

As mentioned earlier, the desired frequency components extracted from spectral component and reduced the residue noise by designing BP-FIR with the parameters of center frequency, band bandwidthm and order. In DPOAE, noise consists of both internal noise (swelling and coughing) and external noise (from running other instruments).

3. Results and Discussion

As mentioned earlier, the recorded signals were analyzed using FFT algorithm, an amplitude spectrum of cochlea response at stimulation parameters of f_1 (500Hz, 1 kHz, 1.5 kHz, and 8 kHz) and γ (1.1) shown in Figure (4), where the frequency components are separated by the factor of 1.1. Hence, as f_1 increased the separation band increased as shown in Figure (4a-d), where the separation at f_1 500 Hz was 50 Hz Figure (4a), at f_1 1 kHz was 100 Hz Figure (4b), at f_1 1.5 kHz was 150 Hz Figure (4c), at f_1 8 kHz was 800 Hz Figure (4d).

On the other hand, as γ increased (1.22, 1.3, and 1.4) the frequency components are separated further than at γ of 1.1 as shown in Figure (5). Figure (5a, c, and e) showed cochlea response at f_1 of 500Hz, while Figure (5b, d, f) showed cochlea response at f_1 of 8kHz. The higher separation between components was obvious at f_1 of 8 kHz and γ of 1.4. It has been noticed that at stimulation parameters of f_1=8 kHz and $\gamma \geq 1.3$, the cochlea response of frequency component of $2f_2$-f_1 was disappeared within the spectral band of view, Figure (5d and f).

To extract the cochlea response represented by frequency components of $2f_1$-f_2 and $2f_2$-f_1, BP-FIR filter was designed as Equiripple-FIR, where it has to be linear phase response. Figure 6 shows the BP-FIR magnitude and phase response for filter parameters of center frequency 450Hz, bandwidth of 200Hz, and order of 1024. The filter designed based on the DPOAE measurement, where f_1=500Hz, f_2=1.1f_1, $2f_1$-f_2=450Hz. Figure

6(a) shows the magnitude response as a flat pass band and low ripple in stop band, while Figure 6(b) shows a linear phase response within the filter pass band.

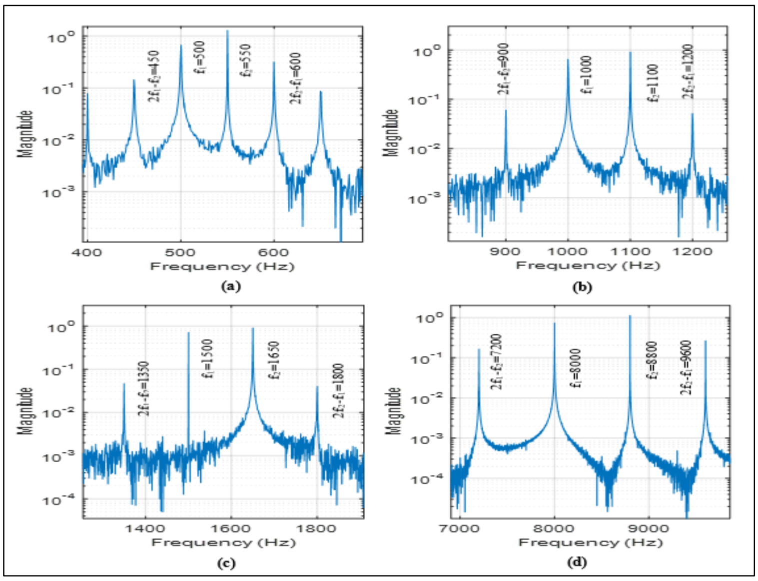


Figure 4. Magnitude spectra of cochlea response at frequency ratio of 1.1, at stimulation frequency of f_1 (a) 500Hz, (b) 1 kHz, (c) 1.5 kHz, and (d) 8 kHz.

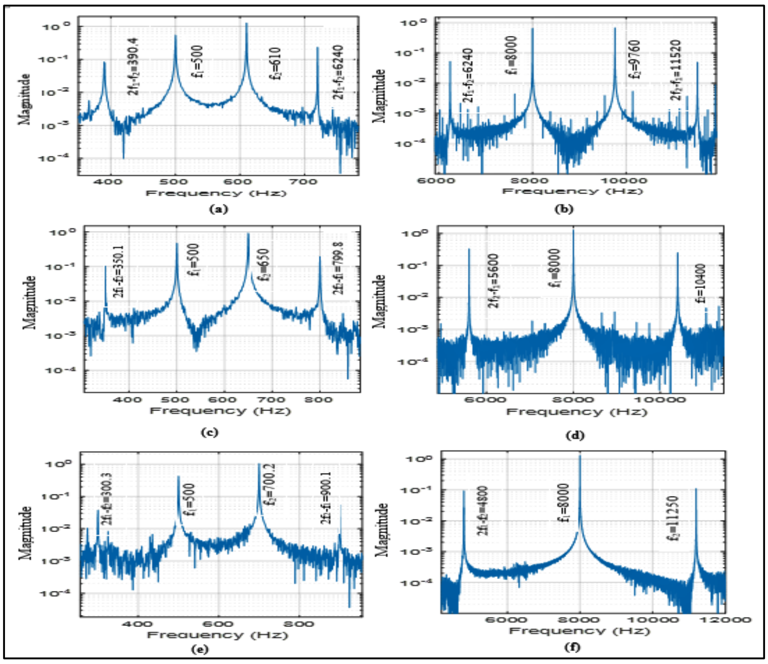


Figure 5. Magnitude spectra of cochlea response at frequency ratio of 1.22 (a and b),1.3 (c and d), 1.4 (e and f).

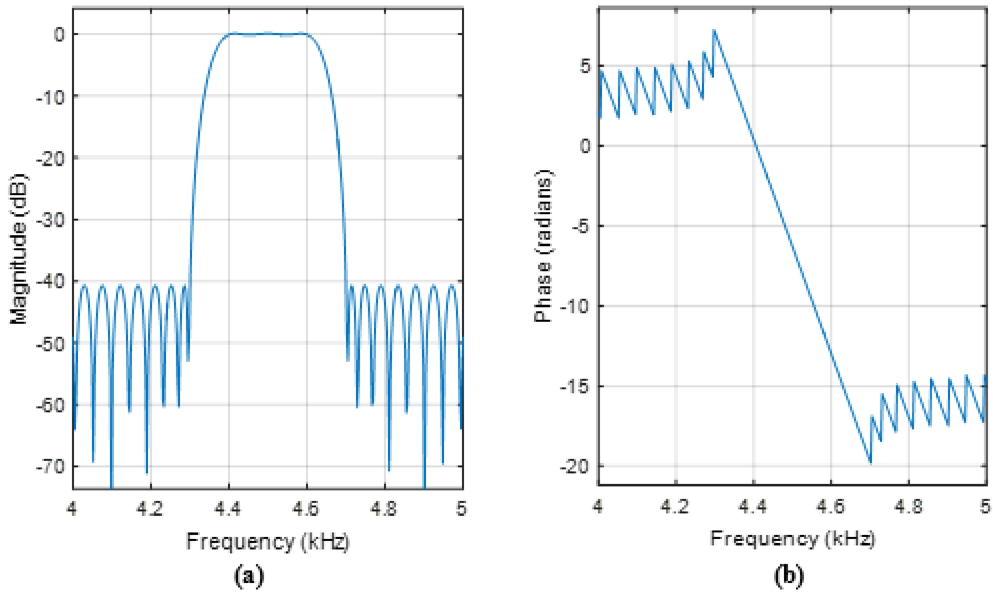


Figure 6. Equiripple BP-FIR filter, (a) magnitude response, (b) phase response.

Since the extraction of cochlea response frequency components of both $2f_1$-f_2 and $2f_2$-f_1 influenced by BP-FIR characteristics, filter design parameters are going to be examined at stimulation parameters of f1=500Hz and γ=1.1. Figure (7a and c) shows the performance of two BP-FIRs of 50Hz bandwidth, order of 512, center frequency of the first one is 450Hz ($2f_1$-f_2) and the second one is 600Hz ($2f_2$-f_1). It is obvious that spectrum of filter output includes multiple frequency components due to low robustness in filter characteristics. To improve filter performance in terms of extraction only frequency components of interest $2f_1$-f_2 and $2f_2$-f_1, Figure (7b and d) shows the improvement in the performance of filter for parameters of 20Hz bandwidth and order of 2048. As a result, when filter bandwidth decreases and order increases the filter characteristics improved by reduction in transition region located between pass band and stop band region. More investigation was carried out on the performance of the BP-FIR for cochlea stimulation by 1 kHz γ=1.1, where the response illustrated above in Figure 4(b).

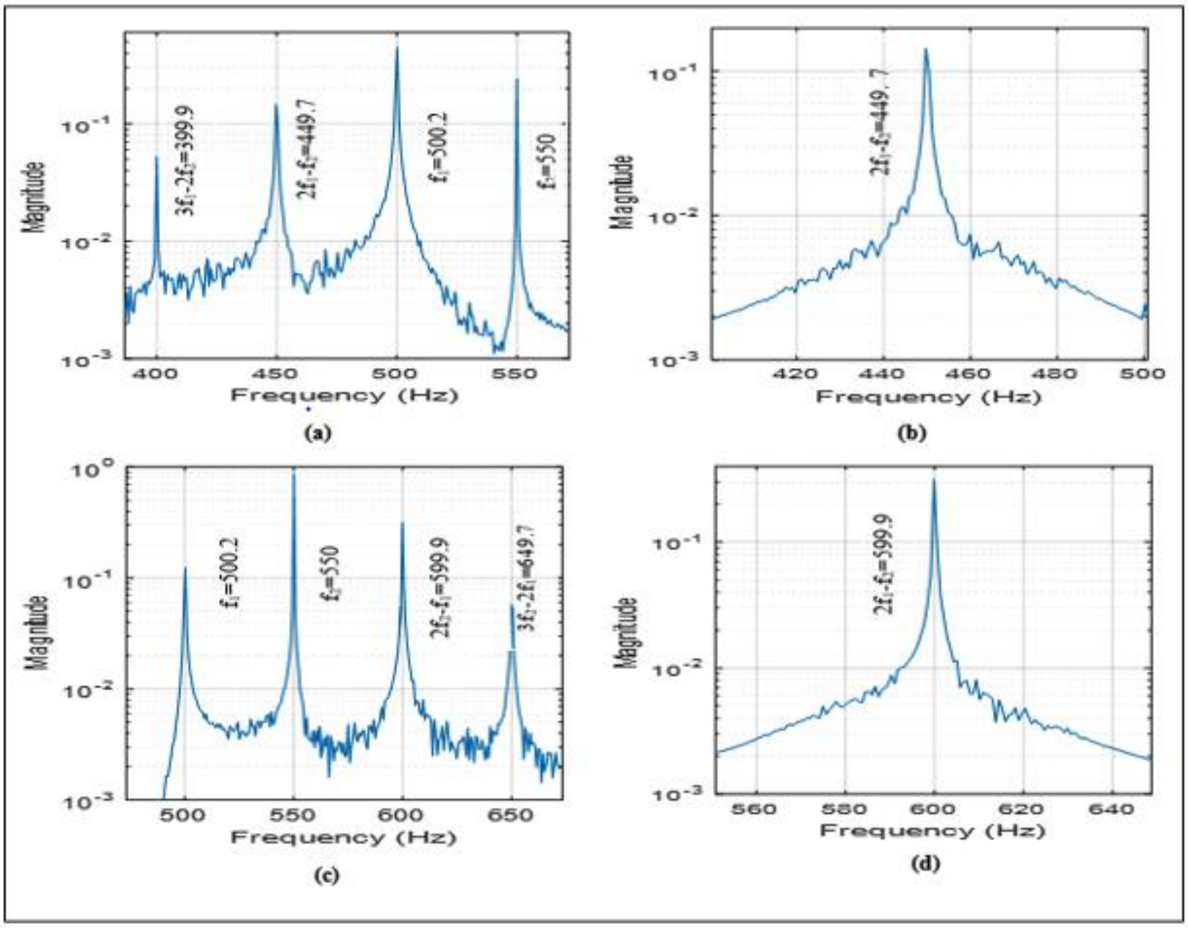


Figure 7. Magnitude spectra of cochlea response at the output of two BP-FIRs, (a and c) filter parameters of bandwidth 50Hz and order 512, (b and d) filter parameters of bandwidth 20Hz and order 2048.

Figure 8(a and c) shows the performance of two BP-FIRs of 50Hz bandwidth, order of 512, center frequency of the first one is 900Hz ($2f_1$-f_2) and the second one is 1.2kHz ($2f_2$-f_1). It is obvious that spectrum of filter output includes multiple frequency components due to low robustness in filter characteristics. Filter performance improved by reduces filter bandwidth and increases of filter order, which was conducted by extraction of only frequency components of interest $2f_1$-f_2 and $2f_2$-f_1, Figure 7(b and d) shows the improvement for parameters of 20Hz bandwidth and order of 2048. The result of Figure 8 agrees with that of Figure 7, which reinforces the robustness of filter performance by reduces bandwidth and increases order.

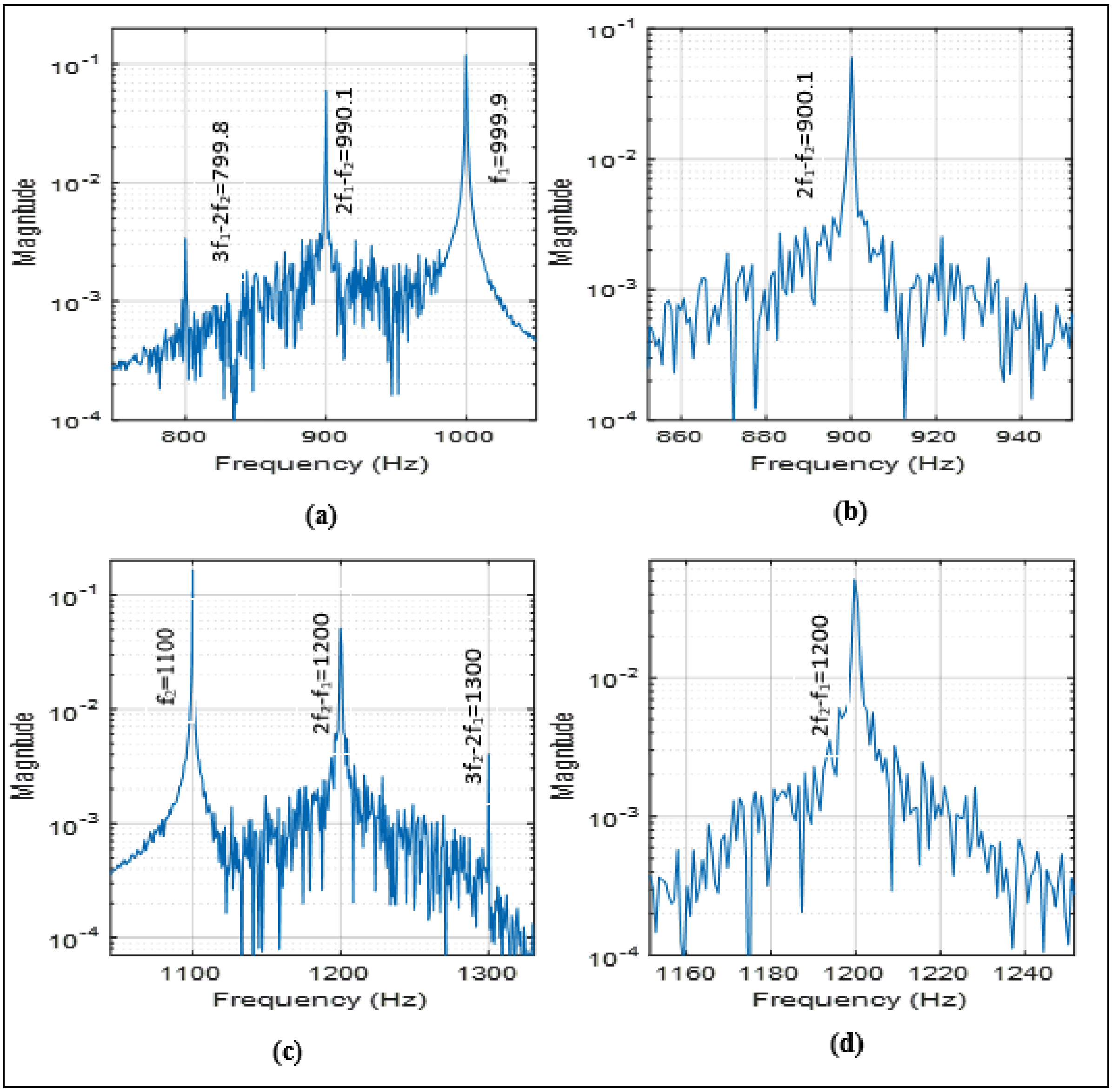


Figure 8. Magnitude spectra of cochlea response at the output of two BP-FIRs, (a and c) filter parameters of 50Hz bandwidth and order of 512, (b and d) filter parameters of 20Hz bandwidth and order of 1024.

Moreover, Figure 9 shows a perfect extraction of only interest frequency component of $2f_1$-f_2 due to improvement in filter robustness. As f_1 increased, separation of frequency components increases and as a result, filter parameters of bandwidth and order can be decreased to conduct perfect filter operation by extraction of single DPOAE component.

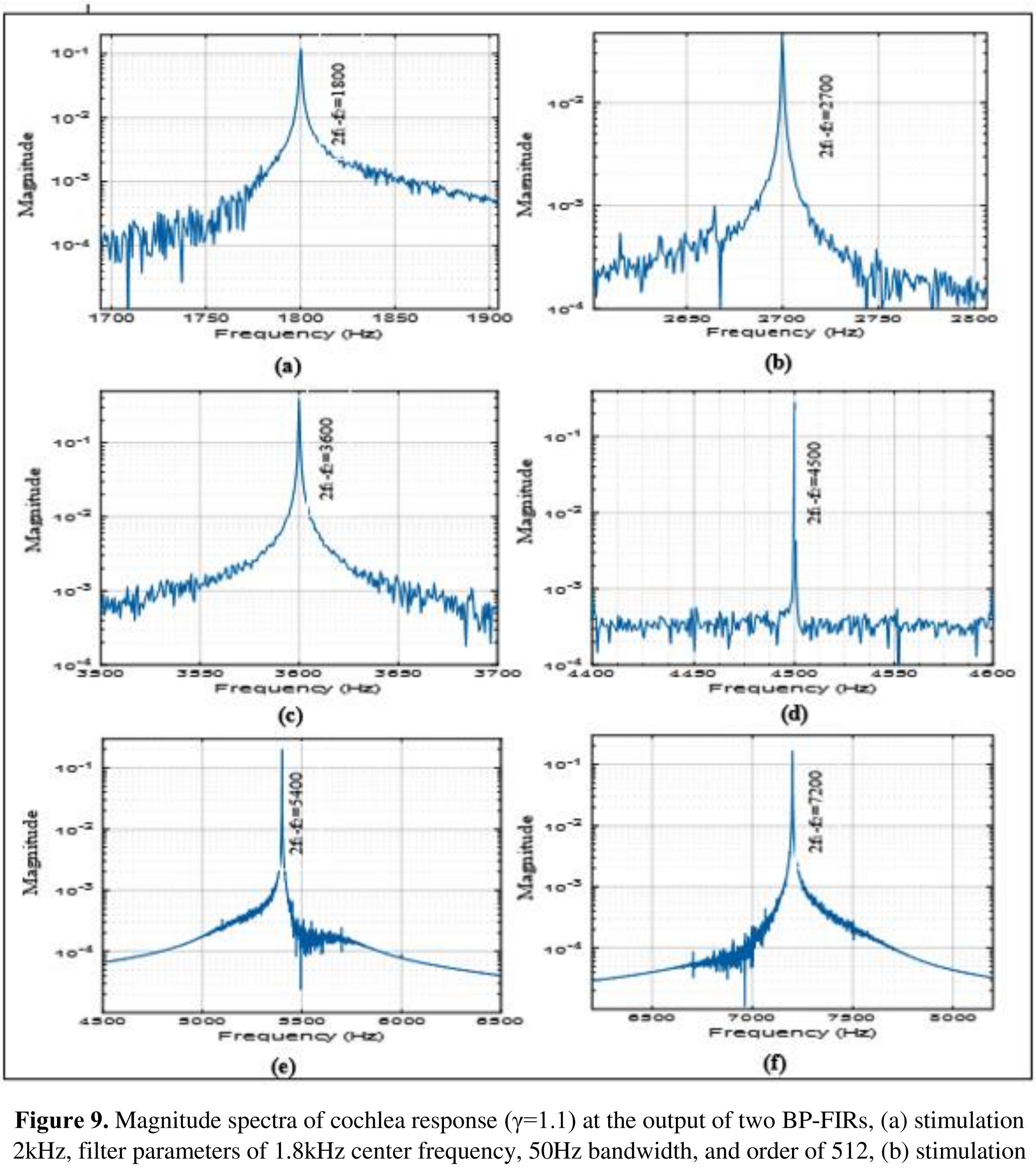


Figure 9. Magnitude spectra of cochlea response (γ=1.1) at the output of two BP-FIRs, (a) stimulation 2kHz, filter parameters of 1.8kHz center frequency, 50Hz bandwidth, and order of 512, (b) stimulation 3kHz, filter parameters of 2.7kHz center frequency, 100Hz bandwidth and order of 256, (c) stimulation 4kHz, filter parameters of 3.6kHz, 150Hz bandwidth and order of 256, (d) stimulation 5kHz, filter parameters of 4.5kHz center frequency, 200Hz bandwidth, and order of 256, (e) stimulation 6kHz, filter parameters of 5.4kHz center frequency, 300Hz bandwidth, and order of 128, (f) stimulation 8kHz, filter parameters of 7.2kHz center frequency, 400Hz bandwidth and order of 128.

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In addition, Table 1 introduced to illustrate BP-FIR filter order as a function of both stimulation parameters of f_1 and γ (f_2/f_1).

Table 1. Order of BP-FIR function of both f_1 and f_2/f_1

f_2/f_1 / f_1(Hz)	**1.1**	**1.22**	**1.3**	**1.4**
500Hz	2048	1024	512	512
1kHz	1024	512	256	256
2kHz	512	256	128	128
3kHz	256	256	128	64
4kHz	256	128	64	64
5kHz	256	64	64	64
8kHz	128	64	32	32

Finally, the presence of interferences represented by speech conference at clinic room was recorded, where cochlea response contaminated by high level of interferences at the same frequency band of response components. Figure (10), shows how proposed signal processing technique unable to extract cochlea response of $2f_1$-f_2 and $2f_2$-f_1.

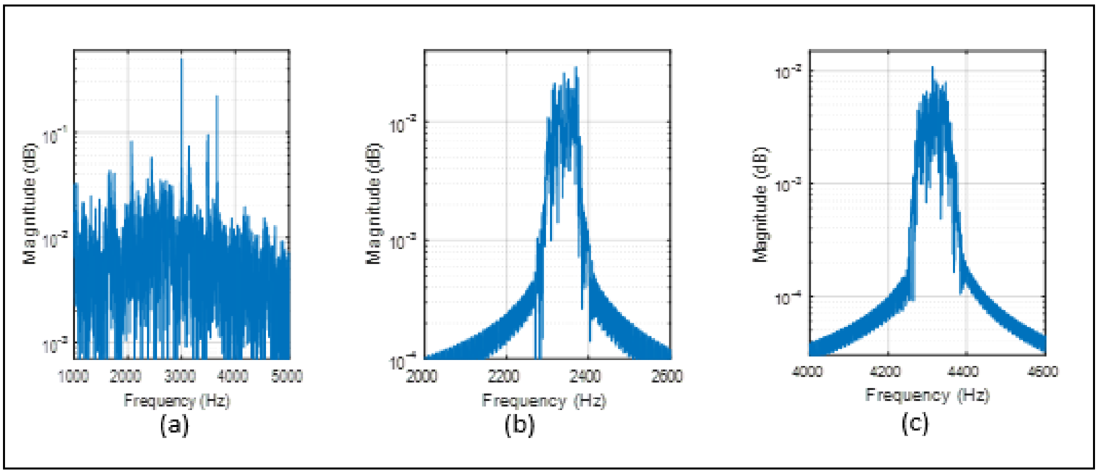

Figure 10. (a) Magnitude spectra of cochlea response at interference situation f_1=3kHz, γ=1.22, BP-FIR bandwidth 50Hz, order 2048, (b) filter output at center frequency of $2f_1$-f2=2.34kHz, (c) filter output at center frequency of $2f_2$-f_1=4.32kHz.

4. Conclusion

In this study, cochlea response was measured based on the analysis of recorded DPOAE signal. Experiments show that $2f_1$-f_2 and $2f_2$-f_1 can be extracted for the range of stimulation frequency (f_1) 500Hz-8 kHz and frequency ratio (f_2/f_1) of 1.1, 1.22, 1.3 and 1.4 using FFT and BP-FIR filter at interference free situation. Investigation recognized that at stimulation parameters of $f_1 \leq 1$kHz and $f_2/f_1 < 1.2$, BP-FIR parameters have to be of bandwidth≤50Hz and order of ≥1024. On the other hand, at stimulation parameters of $f_1 \geq 2$kHz and $f_2/f_1 \geq 1.2$, BP-FIR parameters have to be of bandwidth≥100Hz and order of <1024. For real time implementation, estimation parameters have to be reduced as much as possible, so that frequency ratio of 1.2-1.3 has been recommended, even though frequency ratio of 1.4 superior on using lower estimation parameters, but at the same time at $f_1 \geq 6$kHz DPOAE of $2f_1$-f_2 occurred out of frequency band of view. The investigation also showed that proposed SP method unable to extract the DPOAE frequency components at interference situation. From that end, robust SP algorithm has to be considered for interference situation based on the findings dedicated in this study.

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