NOISE REMIVING FROM SPEECH SIGNAL USING LINEAR –PHASE TYPE II NOTCH FIR FILTER +

إزالة الضوضاء من إشارة الكلام باستعمال مرشح منع حزمة رقمي ذي استجابة نبضية محددة وخصائص طور خطية من النوع الثاني

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Abstract:

The current research aims to design Linear-Phase based Matlab6.5. Notch FIR filter for noise bands removing from the speech signal. FIR filter cost more than IIF filter, but it has only adjustable zeros and hence it is free of stability problems associated with IIR filter which have adjustable poles as well as zeros. The speech signal used in this project has two frequency noise components, thus the Type II linear phase digital notch filter has been designed because it has two nulls, one occurs at the design frequency and the other is automatic at the Nyquist frequency (fs/2) which is used for the other noise band removing. The design used the least square criteria, because arbitrary linear constrains can easily be included.

المستخلص:

يهدف البحث الحالي إلى تصميم فلتر رقمي إمرار حزمة بالاعتماد على برنامج الماتلاب 6.5 ويكون ذا استجابة طور خطية غير تكراري (FIR)، لإزالة حزمة الضوضاء من إشارة الكلام بالرغم من انه الأكثر كلفة مقارنة بالفلتر التكراري إلا أنه أكثر استقرارية لأنه يعتمد على الأصفار وليس عل الأقطاب كما في الفلتر التكراري. بما أن إشارة الكلام كانت تحتوي على ترددين من الضوضاء، لذا كان اختيار هذا النوع من الفلاتر من النوع الثاني الذي يرمز له (II) هو الأكثر مناسباً لأن منحني الاستجابة له يمتلك منطقتي قطع. المنطقة الأولى كانت بالتصميم والتي جعلت عند الحزمة الترددية الأولى للضوضاء، أما الثانية فهي تنتج أوتوماتيكياً عند تردد النيكويست Nyquist (fs/2) والتي استفيد منها لإزالة مركبة الضوضاء الثانية. وقد استخدمت طريقة مربع الخطأ الأدنى في التصميم. ولأن طريقة معيار المربع الأقل يمكن أن تتضمن المعوقات الخطية بسهولة.

Introduction:

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Noise may be defined as any unwanted signal that interferes with the communication, measurement or processing of an information-bearing signal. Noise is present in various degrees in almost all environments. For example, in a digital cellular mobile telephone system, there may be several variety of noise that could degrade the quality of communication, such as acoustic background noise, thermal noise, electromagnetic radio-frequency noise, co-channel interference, radio-channel distortion, echo and processing noise [1]. Noise can cause transmission errors and may even disrupt a communication process; hence noise processing is an important part of modern telecommunication and signal processing systems. The success of a noise processing method depends on its ability to characterize and model the noise process, and to use the noise characteristics advantageously to differentiate the signal from the noise.

Noise is the main limiting factors in communication and measurement systems. Therefore the modeling and removal of the effects of noise have been at the core of the theory and practice of communications and signal processing. Noise reduction removal is important problems in applications such as cellular mobile communication, speech recognition, image processing, medical signal processing, radar, sonar, and in any application where the signals cannot be isolated from noise.

Depending on its source, a noise can be classified into a number of categories, indicating the broad physical nature of the noise, as follows [2]:

(a) Acoustic noise: emanates from moving, vibrating, or colliding sources and is the most familiar type of noise present in various degrees in everyday environments. Acoustic noise is generated by such sources as moving cars, air-conditioners, computer fans, traffic, people talking in the background, wind, rain, etc.

(b) Electromagnetic noise: present at all frequencies and in particular at the radio frequencies. All electric devices, such as radio and television transmitters and receivers, generate electromagnetic noise.

(c) Electrostatic noise: generated by the presence of a voltage with or without current flow. Fluorescent lighting is one of the more common sources of electrostatic noise.

(d) Channel distortions, echo, and fading: due to non-ideal characteristics of communication channels. Radio channels, such as those at microwave frequencies used by cellular mobile phone operators, are particularly sensitive to the propagation characteristics of the channel environment.

(e) Processing noise: the noise that results from the digital/analog processing of signals, e.g. quantization noise in digital coding of speech or image signals, or lost data packets in digital data communication systems.

Depending on its frequency or time characteristics, a noise process can be classified into one of several categories as follows:

(a) Narrowband noise: a noise process with a narrow bandwidth such as a 50/60 Hz 'hum' from the electricity supply.

(b) White noise: purely random noise that has a flat power spectrum. White noise theoretically contains all frequencies in equal intensity.

(c) Band-limited white noise: a noise with a flat spectrum and a limited bandwidth that usually covers the limited spectrum of the device or the signal of interest.

(d) Colored noise: non-white noise or any wideband noise whose spectrum has a non-flat shape; examples are pink noise, brown noise and autoregressive noise.

(e) Impulsive noise: consists of short-duration pulses of random amplitude and random duration.

(f) Transient noise pulses: consists of relatively long duration noise pulses.

(g) Thermal noise: is produced by random motion of electrons in a conductor due to heat.

Linear Phase FIR Filters

In this section the special types of FIR filters are explained in which the coefficients h(n) are assumed to be symmetric or anti symmetric. Since the order of the polynomial in each of these two types can be either odd or even, there are four types of filters with different properties, which we describe below [3]:

Type I: The coefficients are symmetric [i.e., $h(n) = h(N - n)$], and the order N is even.

Type II: The coefficients are symmetric [i.e., $h(n) = h(N - n)$], and the order N is odd.

Type III: The coefficients are anti symmetric [i.e., $h(n) = -h(N - n)$], and the order N is even.

Type IV: The coefficients are anti symmetric [i.e., $h(n) = -h(N - n)$], and the order N is odd.

Properties of Linear Phase FIR Filters

The four types of FIR filters discussed above have shown us that FIR filters with symmetric or anti symmetric coefficients provide linear phase (or equivalently constant group delay); these coefficients are samples of the unit impulse response. Type I filters have a nonzero magnitude at $w = 0$ and also a nonzero value at the normalized frequency ($w/\pi =$

1)[4] (which corresponds to the Nyquist frequency), whereas type II filters have nonzero magnitude at w = 0 but a zero value at the Nyquist frequency. So it is obvious that these filters are not suitable for designing band pass and high pass filters, whereas both of them are suitable for low pass filters. The type III filters have zero magnitude at w = 0 and also at w/ π = 1, so they are suitable for designing band pass filters but not low pass and band stop filters. Type IV filters have zero magnitude at (w = 0) and a nonzero magnitude at (w/ π = 1). They are not suitable for designing low pass and band stop filters but are candidates for band pass and high pass filters.

Least Squares Criteria

Given the over determined linear set of equations, AX = **b**, where **A** is a known m × n matrix of rank n with m > n, b is a known m element vector, and **x** is an unknown n element vector, then the least squares solution is given by [4][5]:

$$X_{LS} = (A^T A)^{-1} A^T b \quad \ldots\ldots\ldots\ldots\ldots\ldots\ldots 1$$

(Note that if the problem is underdetermined, m < n, then equation (1) is not the solution, and in fact there is no unique solution; a good (i.e., close) solution can often be found however using the pseudo inverse obtained via singular value decomposition (SVD)). The least squares solution can be derived as follows:

Consider again the over determined linear set of equations [3][4]:

$$\underset{A}{\begin{bmatrix} a11\ a12 \ldots a1n \\ a21\ a22 \ldots a2n \\ a31\ a32 \ldots a3n \\ \vdots \\ am1\, am2 \ldots amn \end{bmatrix}} \underset{X}{\begin{bmatrix} x1 \\ x2 \\ \vdots \\ xn \end{bmatrix}} = \underset{b}{\begin{bmatrix} b1 \\ b2 \\ b3 \\ \vdots \\ bm \end{bmatrix}} \quad \ldots\ldots\ldots 2$$

If A is a nonsingular square matrix, i.e. m=n, then the solution can be calculated as:

$$X = A^{-1} b \quad \ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots 3$$

However if then A is a rectangular matrix and therefore not invertible, and the above equation cannot be solved to give exact solution for X. If m < n then the system is often referred to as underdetermined and an infinite number of solutions exist for X (as long as the m equations are consistent). If m > n then the system of equations is over determined and can look for a solution by striving to make AX be as close as possible to b, by minimizing (Ax – b) in some sense. The most mathematical tractable way to do this is by the method of least squares [6][7], performed by minimizing the 2-norm denoted by:

$$e = (Ax - b)^2 = (ax - b)^T (Ax - b) \quad \ldots\ldots\ldots 4$$

Depending on the above procedures, will develop a weighted least-square approach for design of the FIR filter, then linear equations are described that must be solved to minimize the following criteria (minimizing the error between the desired frequency response represented by a vector D(w) and the frequency response of the filter W(w)):

$$\varepsilon_2 = \int_0^{\pi} W(w)\,|H(w) - D(w)|^2\, dw \qquad \ldots\ldots\ldots 5$$

Where:

H(w): Frequency Response of the Filter.

W(w): Weighting Variable.

D(w): Desired Frequency Response.

And the derivative:

$$\frac{d\varepsilon_2}{dh(n)} = \int_0^{\pi} W(w).\frac{d|H(w) - D(w)|^2}{dh(h)}\, dw \qquad \ldots\ldots 6$$

where $|H(w) - D(w)|^2 = (H(w) - D(w)\overline{(H(w) - D(w))}$ and substituting in (6):

$$\frac{d\left\|(H(w) - D(w))\right|^2\Big|}{dh(h)} = (H(w) - D(w))\frac{dH(w)}{dh(n)} + (H(w) - D(w))\frac{\overline{dH(w)}}{dh(n)}$$

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where $H(w) = \sum_{n=0}^{N-1} h(n).e^{-jwn}$ and (N) is the filter length, substituting in (7):

$$\frac{d\left\|H(w) - D(w)\ \right|^2\Big|}{dh(n)} = \overline{(H(w) - D(w))}.\,e^{-jwn} + (H(w) - D(w)).e^{jwn} \qquad \ldots\ldots\ldots\ldots\ldots 8$$

<u>FIR filter design procedure</u>:

The steps of the design procedure for linear phase FIR filter in this research can be as follow:

1- Depending on the nature of the magnitude response, the value of the filter length (N) is chose for computing the filter coefficients values for (-N≤n≤N). Then a windows function must be chosen (Bartlett, Hamming, Hann, Kaiser, or other window) and computing its values (w(n)) for ($-N \le n \le N$).

2- Then multiplying the coefficients of the filter by (w(n)) to get the modified coefficients values ($h_w(n)$). The filter with these finite number of coefficients has a frequency response $H_w(e^{jw}) = h_w(-M) e^{jwM} + h_w(-M+1) e^{jw(-M+1)} + \ldots + h_w(1) e^{jw} + h_w(0) + h_w(1) e^{-jw} + \ldots + h_w(M) e^{-jwM}$.

3- The next step is to multiply $H_w(e^{jw})$ by e^{-jwM}, which is equivalent to delaying the coefficients by (M) samples to get h(n). By delaying the product of the filter coefficients and (w(n)) by (M) samples, a casual filter of finite length will be obtained.

Simulation Results:

Only difference between Type I and Type II FIR filters is periodicity and shape of amplitude response A(w) around ($w= \pi$). Type I is even ($A(\pi +w)=A(\pi -w)$) with respect to vertical axis at ($w= \pi$), but Type II is odd ($A(\pi +w)= -A(\pi -w)$) at that point, which means A(w) cross the horizontal axis at ($w= \pi$) in case of Type II [8][9]. Type I can be converted to Type II by including A(w) of the Type II filter with cos((N-1)/2) that makes Q1 and Q2 different than Type I.

Figure (1) shows the speech signal corrupted by noise which will be filtered for noise removing, figure (2) shows its spectrum appearing the noise bands locations.

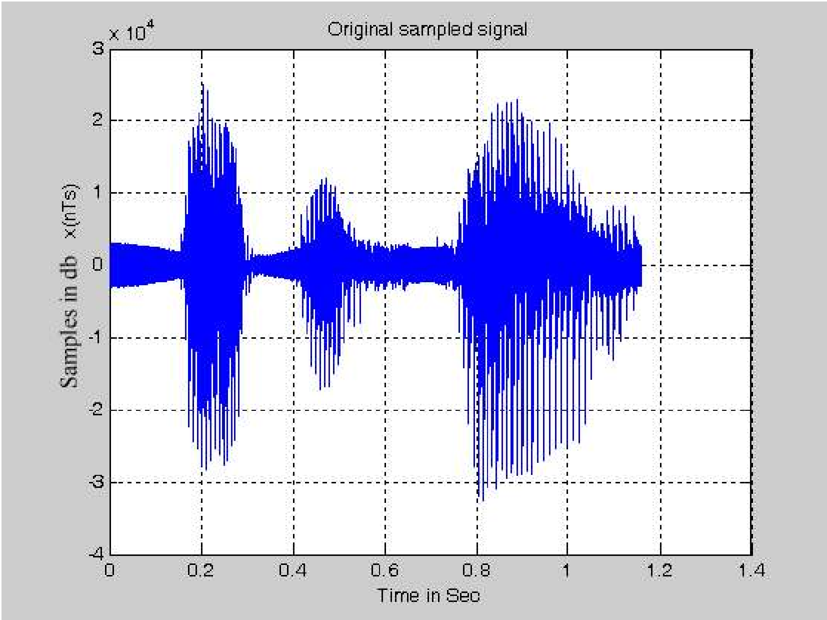

Figure 1: The original signal corrupted with noise

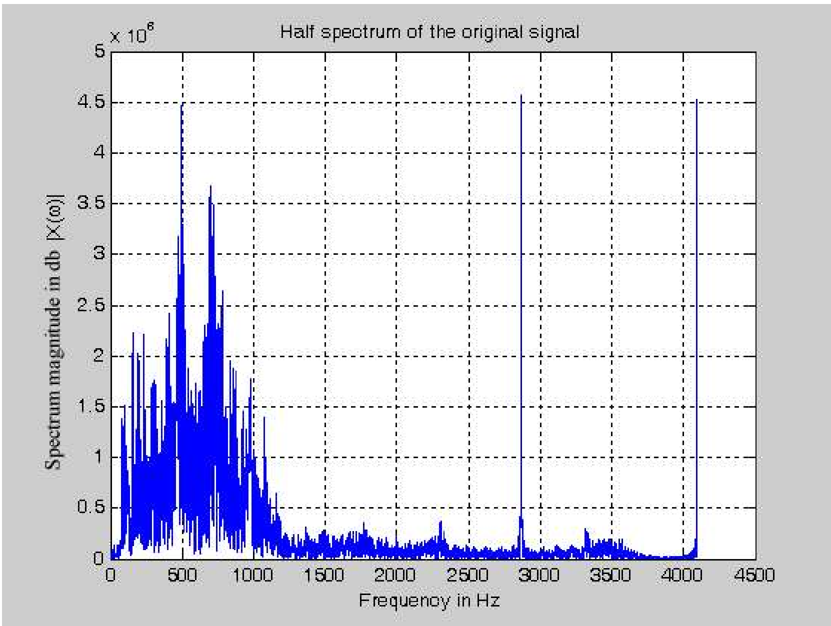

Figure 2: The spectrum of the signal corrupted with noise

After the designing, the impulse response, amplitude response and amplitude response in (db) of the filter will be as in the figures (3), (4) and (5) respectively.

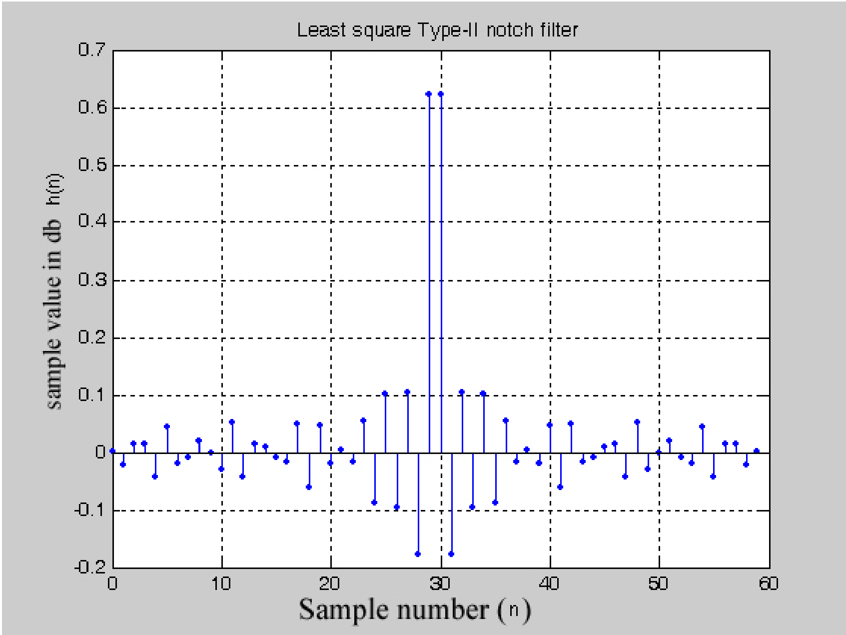

Figure 3: The impulse response of the designed filter

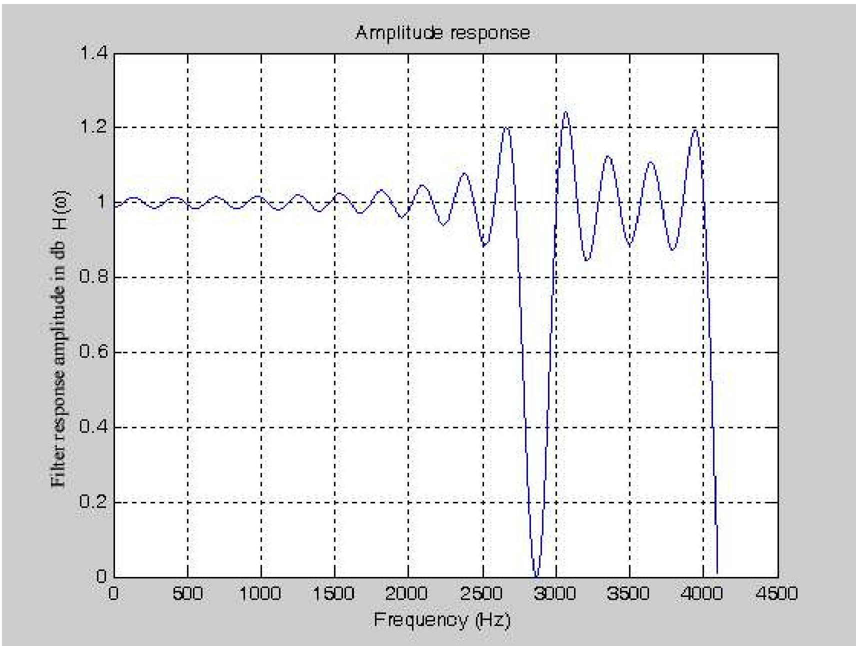

Figure 4: The amplitude response of the designed filter (N=60)

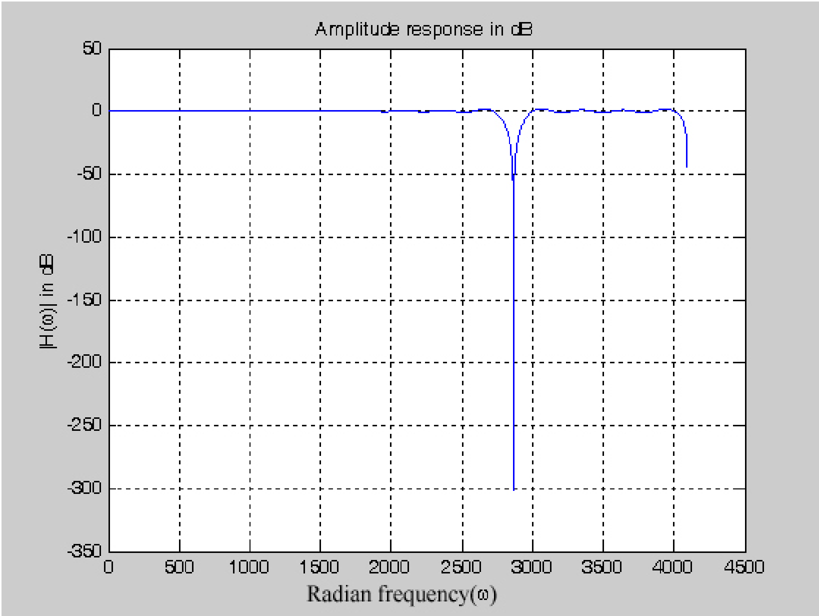

Figure 5: The amplitude response of the designed filter in db (N=60)

Now the corrupted signal will be passed through the designed filter and the output is as in the figure (6). Figure (7) shows the filtered signal spectrum but with no noise bands.

It is examined that the increasing in the filter length will results in narrower notch width, this is shown in the figure (8). It is noticed the decreasing in the notch width and making it limited to the noise band only will be best for removing it, because if the notch is wider than the noise band there will be removing frequencies components from the speech signal.

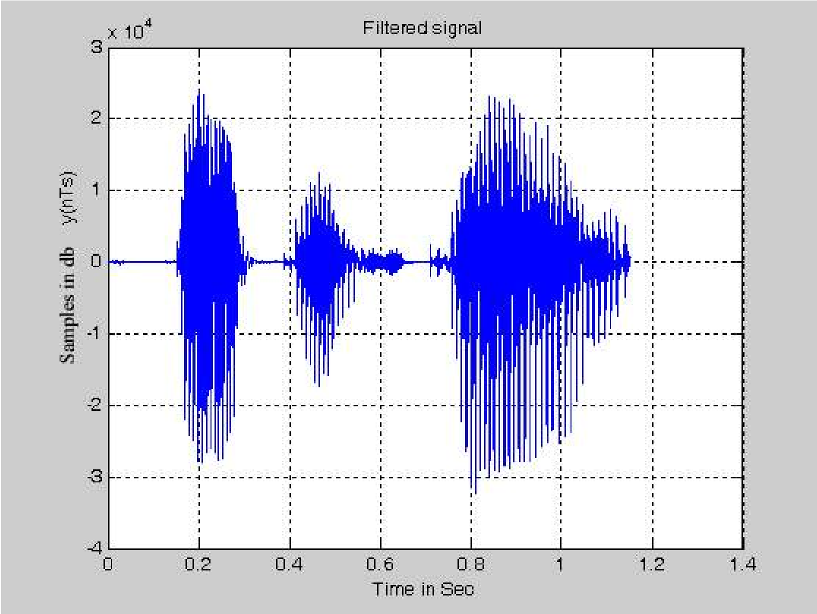

Figure 6: The filtered speech signal

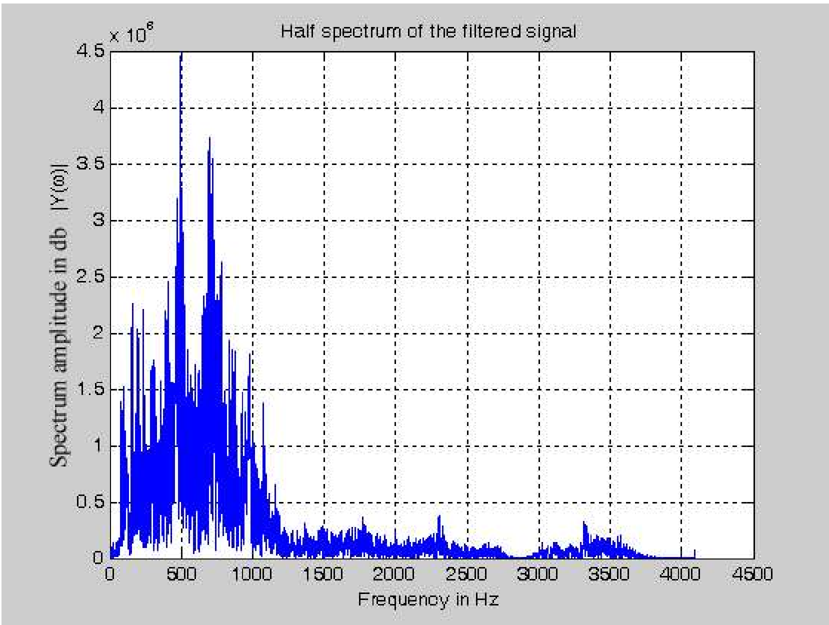

Figure 7: The spectrum of the filtered speech signal

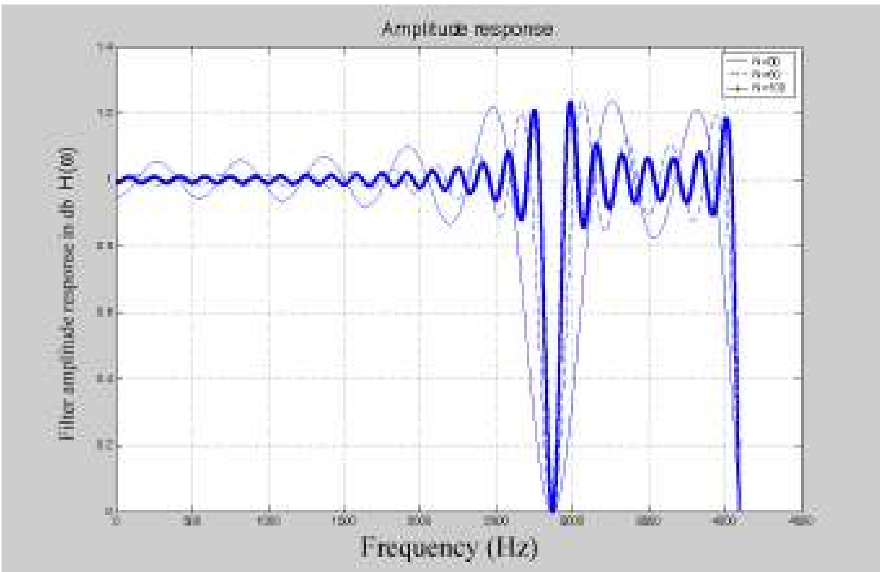

Figure 8: The FIR filter amplitude response at different lengths

Conclusions:

The art of designing a non recursive filter is to achieve an acceptable performance using a few coefficients as possible. Practical filters typically need between (say) 10 and 150 coefficients. In this research three values of coefficients number are chose to examine its effect on the filter operation. This will effect on the notch of the filter, making it narrower as the coefficients is increased as shown in the figure (8), where (N=30, N=60, N=100), and leads to get better voice at the output, where the narrower notch will remove the noise band with less losses in the voice components, and the notch width is desired to be equal to the noise bandwidth as possible as shown in the figure (7). However increasing the filter's coefficients will result in more complexity in the filter realization. In addition, the coefficients number will affect the ripple of the filter, thus a balance must be taken into account when specifying the number of the coefficients for the filter.

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