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Design and Implementation of Single/Multiple Frequency Notch Filter using Adaptive Noise Cancellor for Communication Applications

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Abstract

This paper deals with the design of single/multiple frequency notch filter using Adaptive Noise Canceller technique. The normalized LMS algorithm used for any adaptive filter design is employed in the paper for developing adaptive noise canceller, which is in-turn utilized for realizing a single/multiple notch filter. Notch filters are commonly applied in communication, control, instrumentation, bio-medical engineering and many other fields, to eliminate noise and power line interferences. The experimental results presented in the paper shows that the bandwidth and the mean square error of the designed notch filter output is minimum compared to regular method of notch filter design.

Keywords— *Adaptive filter, Notch filter, Adaptive Noise cancellation, Least Mean Square (LMS) Algorithm.*

I. INTRODUCTION

The major issue in communication system is to eliminate the unwanted noise and interference. If we take a situation where we have to cancel the 60Hz interference, the signal should be passed through a filter so that the interference will be removed. This paper will provide a filter that will give solution for this kind of problems. A filter is a circuit that will allow some band of frequencies by removing other frequencies. Band-stop filter is a filter that attenuates some band of frequencies and allows most of the frequencies[2]. It does the reverse action of the band pass filter. A notch filter is also a kind of band stop filter which will eliminate single or multiple frequencies. Adaptive Filter: It

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adjusts the transfer function itself corresponding to the suitable adaptive algorithm which is conveyed by a corrupted or error signal. Most of these are digital filters. Adaptive noise cancellation: It is a method that uses adaptive filter to remove the narrow band frequencies. Adaptive noise cancellation can be achieved by using adaptive filter. Nowadays, digital signal processing is one among the fastest growing field. They play an important role in many applications. Digital systems are mostly attractive for its accuracy, reliability, occupying small areas and its flexibility. Adaptive signal processing is one part of digital signal processing whose applications are increasing rapidly. Adaptive signal processing has gained its popularity in advanced digital technology. Its computing capacity has been increased and it has gained a huge scope in DSP. The main difference between the usual signal processing method and adaptive signal processing technique is that, it is concerned with the digital systems that are time varying. It is a system where it can automatically adjust the coefficient considering the changing environment or the input signal. Adaptive digital filters can be implemented with the DSP microprocessor.

Adaptive filters are used when there is an unpredictable information about the signal characters or when there is a continuous change in the signal characteristics where there will be no change in the filter characters. Their applications are also found in domains like suppression of jammers, equalization adaptive channels, inverse modeling, predicting the signal and enhancing it. Here single and multiple frequency notch filters with the help of adaptive noise cancellation are discussed. Interference of the noise is one of the major issues that makes the complete system inefficient leading to the inaccurate output. Usually, noise is present in the higher frequencies and has a wideband spectrum. So, to remove this we can use a low pass filter and pass the noise signal to it so that it eliminates all the higher frequency noise components. Doing this process in turn reduces the signal bandwidth. But if we are having the information signal at the higher frequencies this process would be ineffective and there is a chance of losing our information signal which is not advisable. However, by doing this, the bandwidth of the signal would be effectively reduced. So a notch filter would be best advised in this case. Notch filter just eliminates the frequency where the noise is present. Notch filter is a simple band stop filter which has a very sharp bandwidth and could easily remove the noise frequency[3]. To overcome this problem, a notch filter could be implemented. If there are any variations in the noise frequencies and if it tends to shift its frequencies up and down adaptiveness of the notch filter will be very much useful. So, an adaptive notch filter will serve the purpose better than the normal notch filter. Adaptiveness is nothing but the filter should be able to find out the noise and remove the same. Many a times noise signals would not have the constant frequency throughout the process, so this advantage of adaptiveness will help us. So, in this paper a solution is given for the interference cancelling problems[4]. So, a digital notch filter is designed and implemented to remove the single or multiple frequency noises with narrow bandwidth and a signal with wide bandwidth.

II. Adaptive Filters & Adaptive Noise Cancellor

A. *Adaptive Filters*

These are the filters that adjust the transfer function itself corresponding to the suitable adaptive algorithm which is conveyed by a corrupted or error signal. Most of these are digital filters. Adaptive filters can adjust their weight coefficients to remove the noise in the mixed signal on the input side. If the signal is narrowband and noise is broadband, or vice versa, no prior information is needed, but if this is not the case, they require a signal for reference. The reference should be in such a way that it should be correlated to the signal that is to be estimated and adaptive filters can adaptively track the signal under non-stationary conditions [1].

The block diagram for the adaptive noise cancellation can visualize from Fig.1 where, n denotes index of the iteration , *x(n)* is the input signal, *y(n)* denotes the output of the adaptive filter, and d(n) is the mixed (recorded voice + noise) signal. e(n) is the error signal obtained from subtraction of the d(n) and y(n) the, e(n) value changes on every iteration. Adaptive algorithm updates the weight vector in order to minimize the weight vector. The equation for the error signal can be seen in (1).

$$e(n) = d(n) - y(n) \quad (1)$$

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B. *Adaptive Noise Cancellor (ANC)*

It is a method that uses adaptive filter to remove the narrow band frequencies. The goal of the adaptive noise cancellation is to the minimize the noise by using adaptive filtering techniques with the help of reference signal. Using the correlation property between the noise in the desired signal and the reference noise given alone. There are two inputs for the adaptive noise canceller as shown in the Fig.2. The first one is the primary input d(n) = s + n where it contains the signal which has information signal ‘s’ that is corrupted by some noise ‘n’. The other input is the reference n’ which is uncorrelated with the signal ‘s’ but is correlated with the noise ‘n’ in some or the other way.

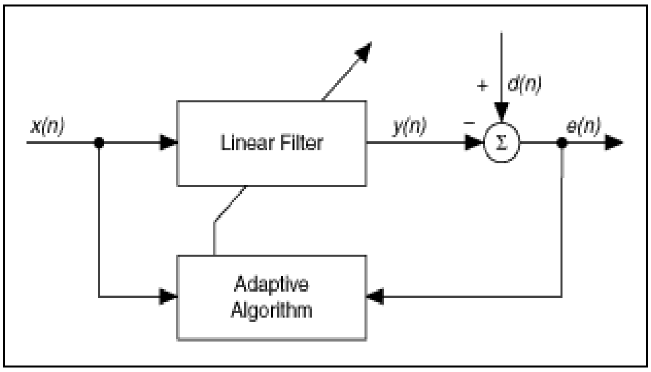

Fig.1. Basic structure of Adaptive Filters

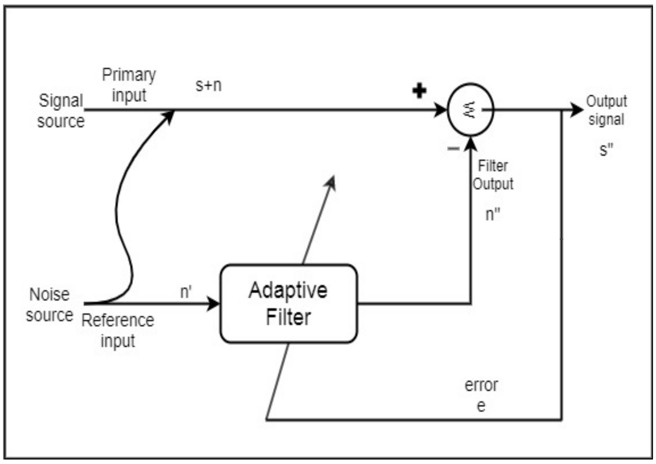

Fig.2. Adaptive Noise Cancellor

The noise n’ is passed through an adaptive filter, and it produces an output n”, which is approximately close to noise got mixed up with the input signal. This noise estimate is removed from mixed signal d(n) which results in producing the signal s’’, the ANC system output, which is approximately same as s.

In this system the main objective is to remove the noise mixed up with the input signal and results, with the output, s'' = s + n – n”. This objective is obtained by feeding the filter output for the subtraction with the mixed signal and feed it back to the LMS algorithm back to the adaptive filter and adjusting the filter coefficients with the help of LMS adaptive algorithm to minimize the system output power. In other words, the output of the system acts as a error signal for the adaptive noise cancellation

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process. Therefore, we are minimizing the error to get good SNR ratio. Here we are not directly minimizing the error but, we reduce the error by taking square of its mean.

$$s'' = s + n - n'' \quad (2)$$

squaring on both sides we get

$$s''^2 = s^2 + (n - n'')^2 + 2s(n - n'') \quad (3)$$

taking expectation on both sides we get

$$E\,[s''^2] = E[s^2] + E[(n - n'')^2] + 2E[s(n - n'')] \quad (4)$$

As the signal and noise are uncorrelated with each other, $E[s(n-n'')] = 0$ and $E[s^2]$ is a constant signal power, hence minimizing (4), we get

$$\min E\,[s''^2] \approx \min E[(n - n'')^2] \quad (5)$$

Thus, minimizing the mean square value of adaptive filter output s'' in-turn minimizes the mean square of difference between the signal mixed noise and the estimated noise, which increases the signal to noise ratio of the adaptive filtered output.

We have various adaptive algorithms like Least Mean Square Algorithm (LMS), Normalised LMS Algorithm (NLMS), Recursive Least Square Algorithm (RLS) and RLS-QR Algorithm (RLS-QR) many more. LMS algorithm is good at computation and in the storage requirements. This makes, LMS algorithm become the first choice in many applications.

C. LMS Algorithm

In this paper, Least Mean Square (LMS) algorithm is used its simplicity and ease of computation. Generally, the LMS algorithm is used in FIR filters Fig.3. The LMS algorithm updates the weight coefficients using the steepest

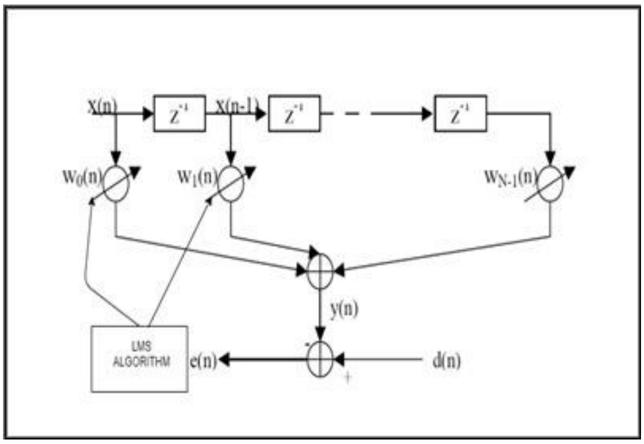

Fig.3. FIR structure of Adaptive filter with LMS algorithm

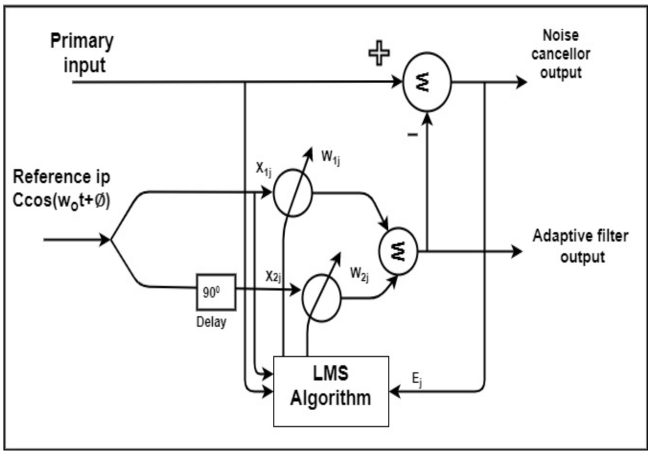

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Fig.4. Implementation of SFNF using ANC

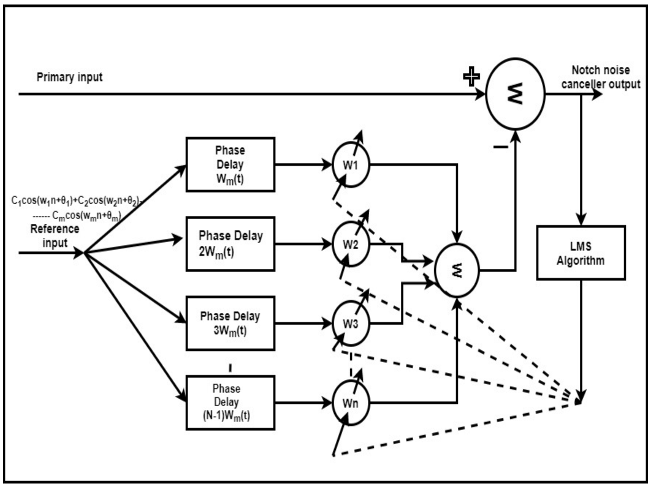

Fig.5. Implementation of MFNF using ANC

descent algorithm where the weight coefficient vector w(n) is updated in the way that error function e(n) is minimized. The adaptive filter output is given by (6).

$$y(n) = x^T(n)w(n) \qquad (6)$$

where x(n) is the adaptive filter input.

In LMS Algorithm differentiation of squared error signal is taken as gradient estimate:

$$\hat{\nabla}_n = \begin{bmatrix} \frac{\partial e^2(n)}{\partial w_0} \\ \frac{\partial e^2(n)}{\partial w_1} \\ \vdots \\ \frac{\partial e^2(n)}{\partial w_{N-1}} \end{bmatrix} = 2e(n) \begin{bmatrix} \frac{\partial e(n)}{\partial w_0} \\ \frac{\partial e(n)}{\partial w_1} \\ \vdots \\ \frac{\partial e(n)}{\partial w_{N-1}} \end{bmatrix} \qquad (7)$$

$$\hat{\nabla}_n = -2e(n)x(n) \qquad (8)$$

Where, $x(n) = \begin{bmatrix} x_0(n) \\ x_1(n) \\ \vdots \\ x_{N-1}(n) \end{bmatrix}$

From the steepest descent algorithm, the weights are updated using equation given in (9).

$$w(n+1) = w(n) - \mu\hat{\nabla}_n \qquad (9)$$

Substituting (8) in (9), thus gives the LMS Algorithm weight update equation as in (10).

$$w(n+1) = w(n) + 2\mu\, e(n)x(n) \qquad (10)$$

III. Noise Filter Using Adaptive Noise Cancellor

A. Single frequency Notch filter using ANC

Single frequency notch filter (SFNF) can reject a single interference frequency. A single frequency notch filter implemented using ANC is shown in Fig.4.

We have taken a 2-tap adaptive FIR filter structure to which a correlated narrow band interference is provided as input signal. The adaptive filter output generated is the estimate of the actual noise interference added at the primary input. Thus, the noise canceller output is the narrow band interference removed clean signal.

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B. Multiple frequency Notch filter using ANC

Multiple frequency notch filter (MFNF) in Fig.5, is the extension of the SFNF. Here, notches are created for several frequencies. This filter is applied where there is more than one narrow band interference in the signal of interest.

IV. RESULTS & DISCUSSIONS

For simulation and result analysis, we have used MATLAB 2013a version platform. The design is verified for a real time speech input signal of 3 to 5 seconds duration, which is corrupted with a single narrow band sinusoidal interference of 1KHz. The resultant notch filtered speech signal output for a step size of 0.006 and order of the filter as 5 is shown in Fig.6. The narrowband noise corrupted speech signal shown in second subplot of Fig.6 sounds like a complete beat sound without any clear presence of the speech signal. This signal, when filtered using the notch filter which is designed in this paper gives a clean speech signal as shown in third subplot of Fig.6.

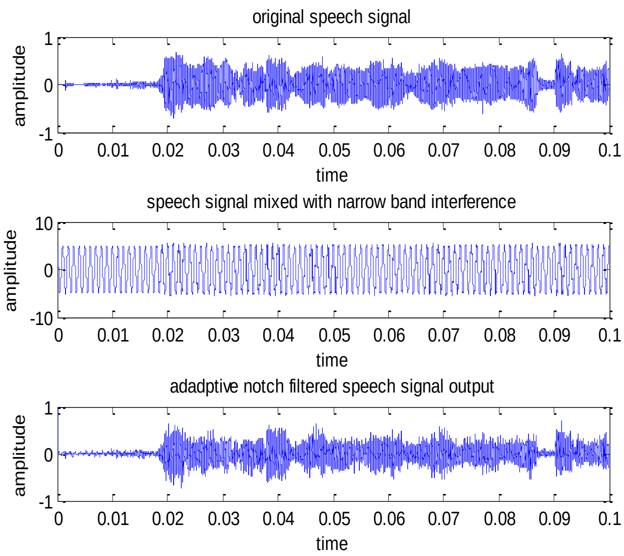

Fig.6. Speech signal output with μ=0.006 order=5

Fig.7, a practical speech signal is given as input with a sinusoidal interference and the LMS algorithm is provided with a step size of 0.1 and with an order of 5. It is observed that as the step size is increased the filtering performance decreases with an increased mean square error (MSE) of the output.

The notch filter is also tested for a random input signal containing of all continuous frequencies from 10Hz to 50Hz as shown in Fig.8.

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The notch filter is tested to produce a single notch at 35Hz for the input signal shown in Fig.8. The Fig.9 shows the sharp notch obtained at the 35Hz which is output of the adaptive notch filter. Fig.10 shows the output spectrum of adaptive notch filter, which is designed to produce two notches at 25Hz and 35Hz.

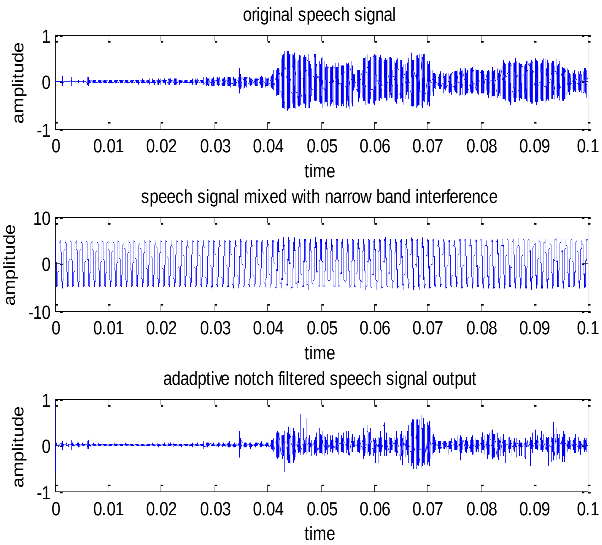

Fig.7. Speech signal output with μ =0.1 and order=5

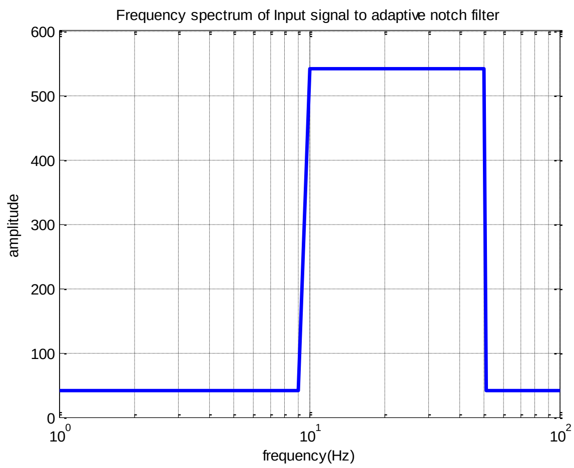

Fig.8. Frequency spectrum of input signal

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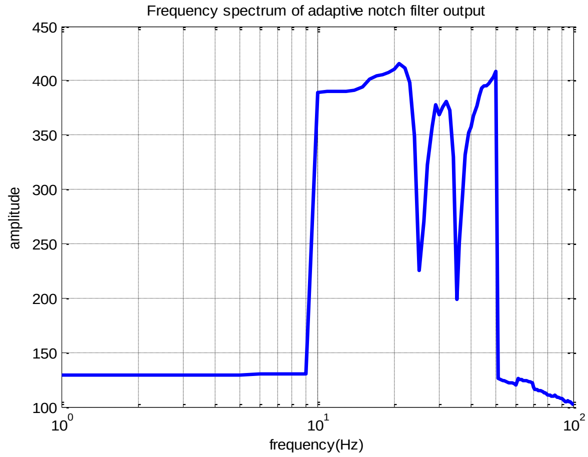

Fig.9. Spectrum of Notch filter output for a single notch at 35Hz

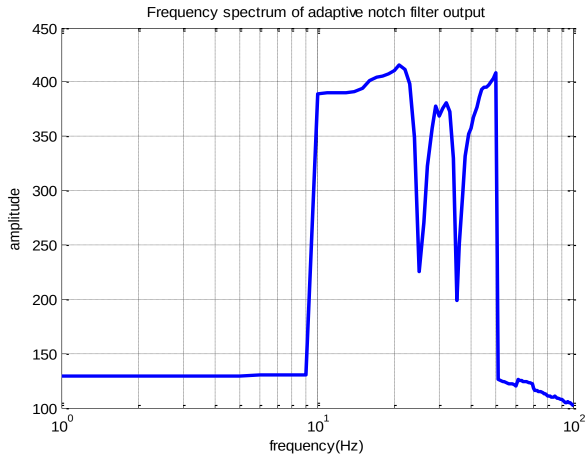

Fig.10. Spectrum of Notch filter output for multiple notches at 25Hz and 35Hz

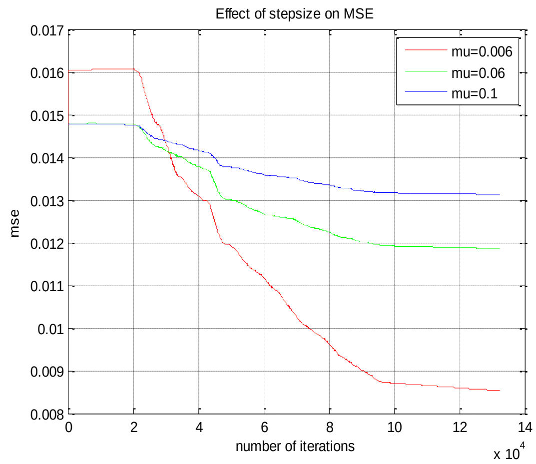

Fig.11. MSE v/s number of iterations for different step size

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and 35Hz

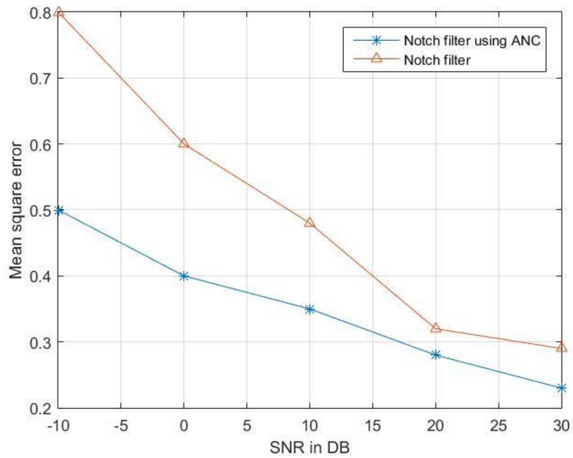

Fig.12. Performance comparison between normal notch filter and notch filter using ANC

Fig.11 shows the graph of mean square error v/s number of iterations for variations in the step size where the μ values are 0.006, 0.06 & 0.1. We can observe that, smaller the step size, lesser will be the MSE except for the initial iterations. Also, smaller step size takes more number of iterations to reach minimum MSE when compared to slightly higher step sizes.

Fig.12 shows the performance comparison of regular notch filter with the notch filter designed using adaptive noise canceller. For all the various SNR values, it is observed that the MSE of the adaptive notch filter output is lesser compared to any regular notch filter for the same input conditions, step size and order of the filter.

V. Conclusion

In this paper, an adaptive noise canceller based single/multiple notch filter design is introduced, implemented and tested. From the simulation results, it is observed that performance of adaptive notch filter is much better when compared with a regular notch filter. It is also noted that design of notch filter using ANC approach is simpler and less complex. The notch obtained using ANC approach is very steep and is nearly identical to an ideal notch filter frequency response. Hence this method of notch filter design is best suitable for all communication applications where a regular notch filter finds its place.

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