Suppression Of NarrowBand Interference In WideBand Signal Using Adaptive Filters

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INTRODUCTION

The problem of NarrowBand Interference in WideBand signal occurs in Digital Communication and in Signal Detection where the desired signal is spread spectrum signal while NarrowBand Signal indicate signal from another user of frequency band or some intentional interference from a jammer who is trying to disrupt the communication or detection system.

From the filtering point of view , our objective is to design a filter that suppresses the narrowband interference. In effect, such a filter should place a notch in the Frequency band occupied by the interference. However the frequency band of the interference is unknown and Moreover the frequency band of interference may vary slowly in time.

In the given problem we are advised to design a Band Stop filter which stops the frequency of narrowband signal in the resultant signal thus returning the wideband signal. Frequency response of the suppression filter is such that it exhibit notch at the frequency of interference

When multiple narrowband signals(sinusoidal) are used, the multiple notch is observed at the frequency response of the Suppression filter.

THEORY

The narrowband characteristics of the interference allow us to estimate signal S[n] from past samples of the sequence X[n] = S[n] +W[n] and to subtract the estimate from the X[n]. Since the bandwidth of S[n] is narrow compared to the bandwidth of W[n] , the samples of S[n] are highly correlated. On the other hand , the wideband sequence W[n] has relatively narrow correlation.

The general configuration of interference suppression is shown in figure . The signal X[n] is delayed by D samples where the delay D is shown sufficiently large so that the wideband signal components W[n] and W[n-D] , which are contained in X[n] and X[n-D] respectively, are uncorrelated.
The output of the Adaptive FIR Filter is the estimate S[n] = ∑

H[k]X[n-k-D]

The error signal that is used in optimizing the FIR filter coefficients is E[n]=X[n]-S[n] . The minimization of sum of squared errors again leads to a set of linear equations for determining the optimum coefficients recursively becomes

$$H_n[k]=H_{n-1}[k]+\Delta E[n]X[n-k-D] , \qquad k=0,1,. \quad ,n-1$$

$$n=1,2,......$$

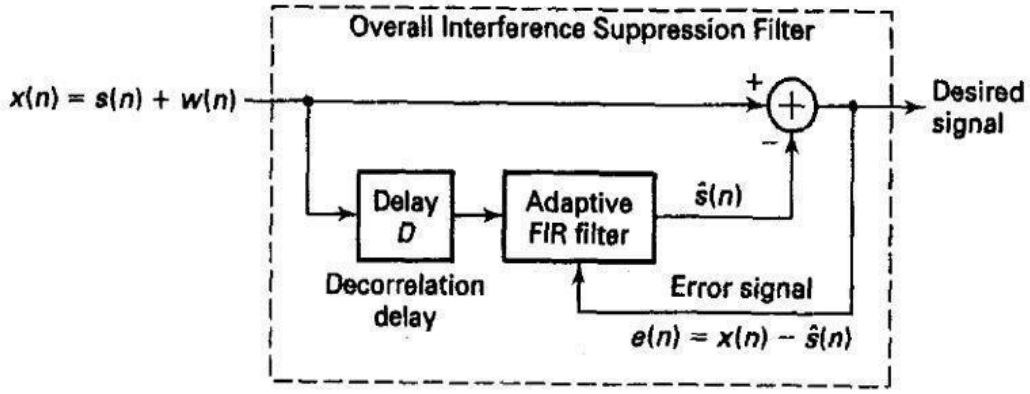
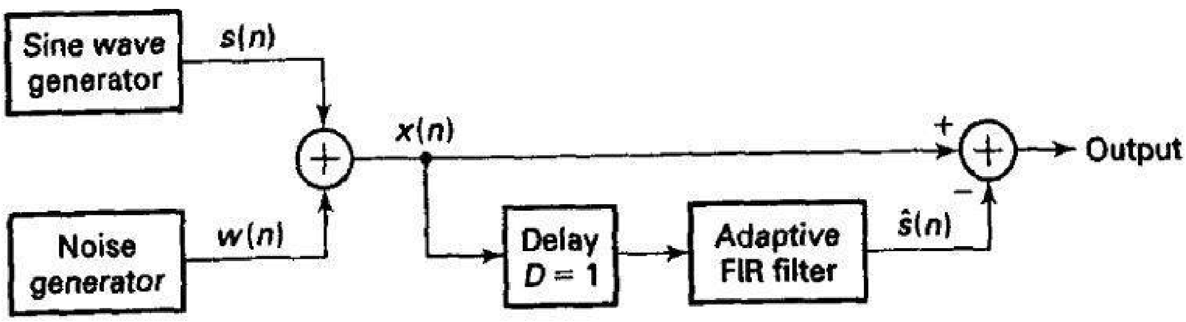


Figure-: <u>Adaptive Filter for estimating and Suppressing a narrowband interference</u>

For the given problem statement the suppression filter is given by -:



LMS/ GRADIENT DESCENT ALGORITHM

In both the above cases, we take an ‘initial guess’ as to what our filter coefficients should be, and keep on ‘learning’ them as time goes. In order to learn the coefficients, we feed back the error (i.e. the difference between the resultant signal and the output of the adaptive filter) to our adaptive filter, and the filter tries its best to minimize it. Since the algorithm tries to make the coefficients that make the mean squared error the least, it is called Least Mean Squared error (LMS algorithm).

The algorithm feeds back the rate of change of error,

i.e. the error ‘gradient’. In order to find the least error, we need to approach a minimum, i.e. where the gradient will become zero. Since the algorithm also tries to reduce the gradient all the time, it is also (popularly) known as ‘gradient descent’ algorithm.
Gradient Descent update rule:

$$\Delta\vec{w} = -\eta \nabla E[\vec{w}]$$

$$i.e.\ \Delta w_i = -\eta \frac{\partial E}{\partial w_i}$$

Here, w is a matrix of coefficients of the digital filter; E is the error (a function of weights w) and   is the learning rate.

Actual implementation: changes, difficulties and modifications

1. For the first scenario, we initially tried using narrowband signal and wideband signal using a sinusoidal generator and uniformly distributed random numbers with zero mean(statistically) respectively so that wideband has a range of frequencies and narrowband frequency lies within it. The power of these two signals are chosen such that there is strong interference of narrowband.
2. To overcome the problem, or rather to find a way around, we had to revert to a more
implementable method. In the simpler method, we took the length of filter such that the step size condition is satisfied using power i.e. Px =Ps+Pw

$$0 < \Delta < \frac{1}{10NP_x}$$

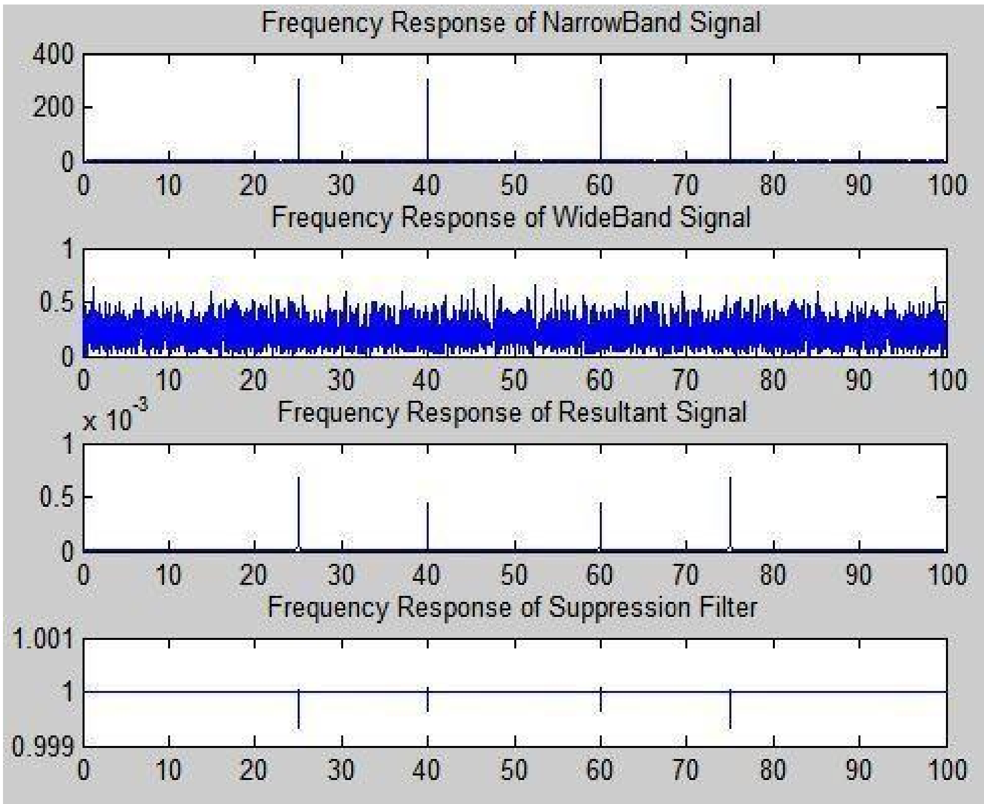
3. We kept the initial guess of w’s to be all zeros and all ones. Since the algorithm learns on its
now, the initial guess is not too important.

MATLAB Code

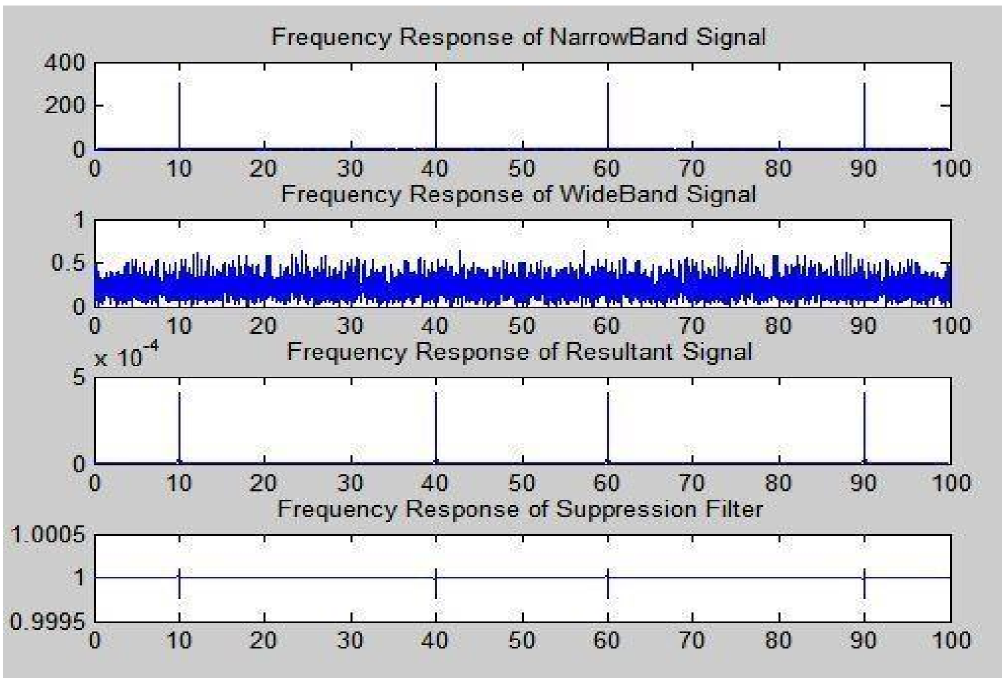
```
N=5000; %no of samples
n= -N/2:N/2-1;
w0=0.25; %frequency of narrowband
phi=0;
A=0.12; %amplitude considering power criteria Ps = 10Pw
sn=A*sin(2*pi*n*w0+phi)
a= -0.006;
b= 0.006;
                                           %band limited uniformly distributed
                                             signal considering power criteria
wn=a+(b-a).*rand(N,1); x=sn'+wn;
d=1;
len=length(x); x_d=zeros(len,1);

x_d(d:len)=x(1:len-d+1); %delayed signal
e=zeros(len,1);w=zeros(len,1);L=10;
w1=zeros(len,1);
mu=0.0000000126263; %step size considering power and length of filter

for i=L+1:len
e(i)=x(i)-transpose(x_d(i-L+1:i))*w(i-L+1:i); %Calculation of Error
w(i-L+2:i+1)=w(i-L+1:i)+2*mu*e(i)*x_d(i-L+1:i); %Calculation of the Weight
vector
w1(i)=transpose(x_d(i-L+1:i))*w(i-L+1:i); % Output signal of our code
end;


% subplot(411);
% plot(wn); title('wideband signal');
% subplot(412);
% plot(sn); title('narrowband signal');
% % subplot(211);
y=fft(sn,N);
m=abs(y);
y(m<1e-6) =0;
f=(0:length(y)-1)*100/length(y);
subplot(4,1,1);
plot(f,m); title('Frequency Response of NarrowBand Signal');
y1=fft(wn,N);
m1=abs(y1);
y1(m1<1e-6) =0;
f1=(0:length(y1)-1)*100/length(y1);
subplot(4,1,2);
plot(f1,m1); title('Frequency Response of WideBand Signal');
y2=fft(w1,N);
m2=abs(y2);
y2(m2<1e-6) =0;
f2=(0:length(y2)-1)*100/length(y2);
subplot(4,1,3);
plot(f2,m2); title('Frequency Response of Resultant Signal');
y3=1-y2;
m3=abs(y3);
y3(m3<1e-6) =0;
f3=(0:length(y3)-1)*100/length(y3);
subplot(4,1,4);
plot(f3,m3); title('Frequency Response of Suppression Filter');
```

Results

- Considering single narrowband signal-:



- Considering multiple narrowband signal-:



Comments and conclusion

- The rate of convergence depends upon the value of the learning rate or step size. As the value of Δ becomes smaller the rate of convergence becomes slower, but the tracking is more precise. If Δ is made bigger, the tracking is not as precise, but the convergence is faster.
- There is an optimal value for Δ. If we increase Δ beyond this limit, the algorithm diverges instead of converging
- As said above, the coefficients tracked by the adaptive filter are flipped as compared to the h[n] coefficients. This was because we used convolution in case of h[n], and linear combination in case of w's.
- When multiple signal input is given the suppression filter exhibits a notch at multiple points to stop narrowband.
- There is strong interference of narrowband signal as sinusoidal power is taken almost 10 times the wideband power, due to which its magnitude is much high in frequency spectrum
- The length of the reconstruction filter and the learning rate values are user programmable. In real world applications, the length of the reconstruction filter should be decided by the precision needed.

Application Areas-: Similar narrowband suppression is used in CDMA communication system technology.