Project 2: Adaptive Noise Canceler or Adaptive Notch Filter

ANC Background

In the adaptive noise canceler, we assume the availability of both a primary signal and a reference signal. This reference signal contains only the sinusoidal interference and may be obtained by locating a sensor near the noise source so that the noise source dominates. In the configuration one might initially think that a simple subtraction of the reference from the primary would eliminate the noise. However, we cannot assume that the sinusoid in the primary has the exact amplitude and phase as in the reference. Therefore, simple subtraction will not work and we must construct an adaptive filter to adjust the sinusoid in the reference so that it matches that in the primary. The filter output is then subtracted from the primary leaving only the signal of interest.

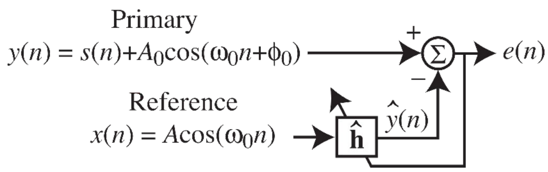

Figure: Adaptive Noise Canceller (ANC)

The ANC can be constructed with two simple one-coefficient adaptive filters. Here we note that the reference signal, $x(n)$ is split into two paths: the upper path is directed to $\hat{h}_0$ and the lower path is first shifted by 90-degree and then directed to $\hat{h}_1$. Because of the fact that the input to the adaptive filter actually consists of two separate input signals, we must adjust each filter coefficient separately using the proper input in either LMS or NLMS.

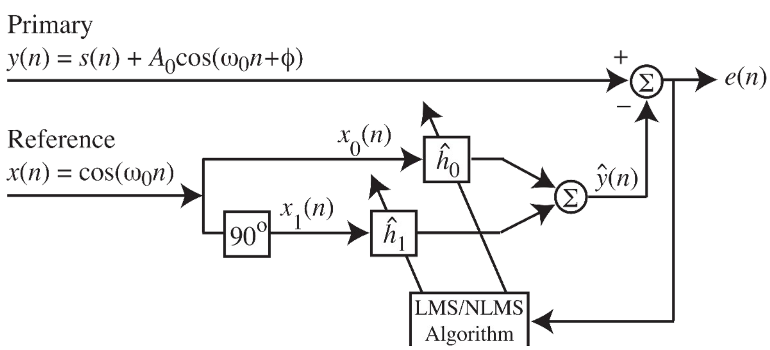

Figure: Adaptive noise canceler

The 90-degree phase shift can be digitally implemented with a *Hilbert filter*. In MATLAB such a filter can be designed with the command (in MATLAB, 'help remez' for more information)

```
hilbert_coefs = firpm(30,[.1 .9],[1 1],'Hilbert');
```


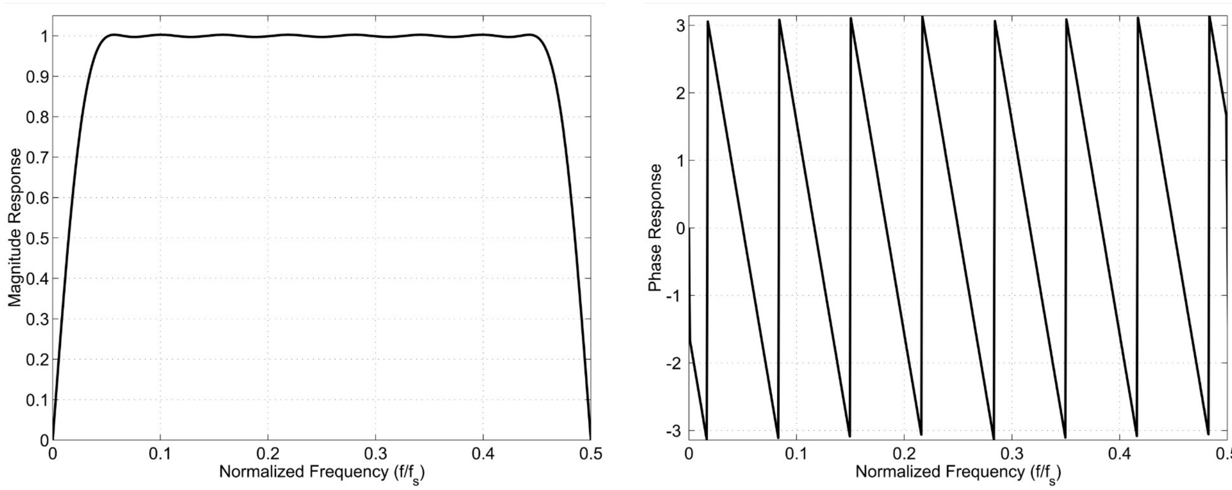

Figure: Sample Hilbert filter frequency response

Assuming the length of the Hilbert filter is *P* (odd) and that the filter has a linear phase, the filter delay will be (*P* – 1) / 2. In order to keep the two inputs to the adaptive filter time-aligned (coherent), we must add a delay of (*P* – 1) / 2 samples to the input which directly feeds $\hat{h}_0$.

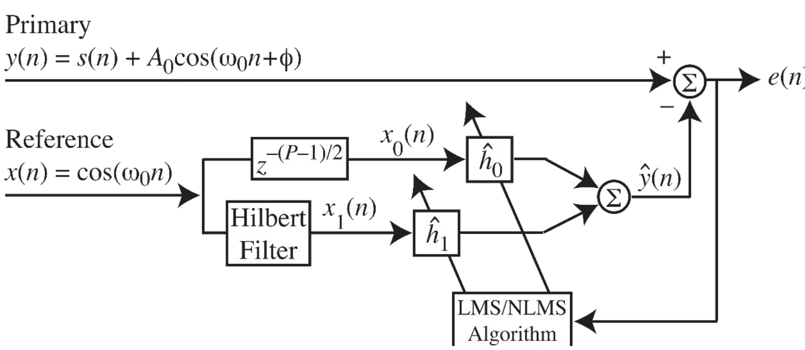

Figure: A fully digital version of the adaptive notch filter.

Simple Analysis of the ANC

In order to understand how ANC in the Figure above operates, we note that the objective is for $\hat{y}[n]$ to be close to $A_0\cos(\omega_0 n+\phi)$ so that $e[n]\approx s[n]$ resulting in cancelation of the sinusoid in the primary. Applying a simple trigonometry relation, we have that

$$A_0\cos(\omega_0 n+\phi)=A_0\cos(\phi)\cos(\omega_0 n)-A_0\sin(\phi)\sin(\omega_0 n)$$

Because of the 90º phase shift,

$$\hat{y}[n]=\hat{h}_0\cos(\omega_0 n)+\hat{h}_1\sin(\omega_0 n)$$

Thus if the filter coefficients adjust themselves so that

$$\hat{h}_0=A_0\cos(\phi)$$
$$\hat{h}_1=-A_0\sin(\phi)$$

the objective is obtained. The LMS/NLMS algorithm will in fact, adjust the coefficients in such a manner.

Algorithm

The algorithm can be found in Section 6.4 of the textbook.

Memory Layout

The anc.dat file is available on the EE492/EE592 web page.

