# ADAPTIVE HETERODYNE FILTERS (AHF) FOR DETECTION AND ATTENUATION OF NARROW BAND SIGNALS

*Karl E. Nelson and Michael A. Soderstrand*

Digital Signal Processing and Communications Laboratory (DSP/C)
Electrical and Computer Engineering Department
One Shields Avenue
University of California, Davis, CA 95616 USA

## ABSTRACT

A fixed filter may be converted into an adaptive filter with a single adaptation parameter through the use of a new Adaptive Heterodyne Filter (AHF) concept in which the frequency of the heterodyne signal is adjusted thereby translating the entire filter transfer function in frequency. If the fixed filter is selected to be a very narrow-band band-pass filter, the new AHF concept can be used very effectively in the elimination of narrow band interference in wide-band communications or control systems. A specific example of the removal of a slow-moving time-varying mechanical resonance from the control signal for a flight control system demonstrates the power of the new AHF concept.

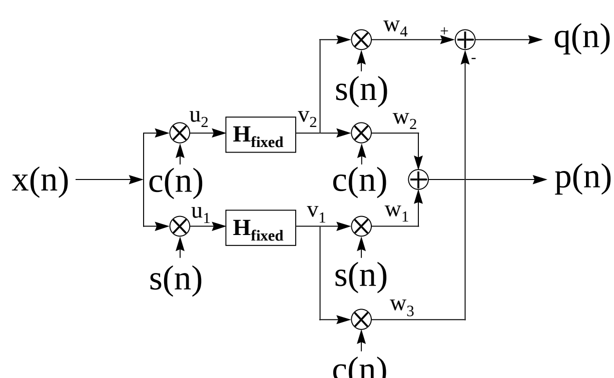
## 1. INTRODUCTION

Narrow-band interference is a common problem in modern communications and control applications. In Frequency-Hoping Spread-Spectrum (FHSS) and Direct-Sequence Spread-Spectrum (DSSS) communications systems, security and efficient channel usage are achieved by spreading the energy of the communications signal across a wide band of frequencies allowing for simultaneous use of the channel by multiple users [1, 2, 3, 4]. However, strong narrow-band signals from standard AM and FM radio transmissions within the communications channel can make it impossible to detect the spread-spectrum signal [5, 6, 7, 8]. Similarly, in many control applications a narrow-band signal from a mechanical resonance can interfere with the feedback control signals from various transducers making it impossible to detect and control the parameters of the plant [9, 10]. In many of these situations it is not possible to use a fixed filter to eliminate the narrow-band interference because either there are many interfering signals popping on and off at different times and frequencies or the frequency of the narrow-band interfering signals is time-varying such as the case of a mechanical resonance varying in frequency as temperature changes [9]. In these situations, an adaptive narrow-band filter for detection and attenuation of the narrow-band interference is required.

## 2. ADAPTIVE HETERODYNE FILTER

### 2.1 Basic Concept

The basic concept of the Adaptive Heterodyne Filter (AHF) is quite simple. As in a super-heterodyne radio receiver and Intermediate Frequency (IF) signal is mixed (heterodyned) with the incoming signal to translate the detection and demodulation problem to a fixed-frequency spectrum in which it is most convenient to implement the detector and demodulator. Similarly, the AHF system mixes an IF signal with the input signal to translate the filtering problem to a fixed frequency where it is convenient to do the filtering operation. However, in the case of the super-heterodyne radio, the incoming signal is at a fixed frequency whereas in the case of the AHF the interference is time varying in frequency. Thus in order to maintain a fixed frequency for the filter, we must constantly adjust the IF frequency in order to translate the input interference to the fixed filter that is used to attenuate the interference [10].




$$c(n) = \cos \sum_{i=0}^{n} \phi_i \qquad s(n) = \sin \sum_{i=0}^{n} \phi_i$$

Figure 1. Heterodyne Filter Block Diagram.

### 2.2 AHF Block Diagrams

Figure 1 shows the block diagram for the Adaptive Heterodyne Filter (AHF) [10]. The input signal is multiplied by the intermediate frequency (IF) function *c(n)* in the upper branch and by the IF function *s(n)* in the lower branch thus translating the input signal to the intermediate frequency (IF) where a fixed filter removes the interference. The frequency $\phi_i$ of the IF signal is adapted as a function of $i$ in such a way as to bring the narrow-band interference frequency to the frequency of the fixed filter. At the output of the fixed filter, IF functions *c(n)* and *s(n)* are used to translate the signal back to the original frequency band (baseband) in such away that we generate two signals *p(n)* and *q(n)* which are used in the standard LMS adaptive algorithm. Figure 2 shows the LMS adjustment scheme used to adjust the frequency $\phi_i$ of the IF signals *c(n)* and *s(n)*. The output *p(n)* from the circuit of Figure 1 is the filtered signal output which forms the error function $\varepsilon$ for the LMS algorithm of Figure 2 and the output *q(n)* from the circuit of Figure 1 forms $d\varepsilon / d\phi$ , the derivative of the Heterodyne Filter Output ( $p(n) = \varepsilon$ ) with respect to the adaptive parameter $\phi$ which is used in the LMS algorithm of Figure 2 [5, 6, 10].

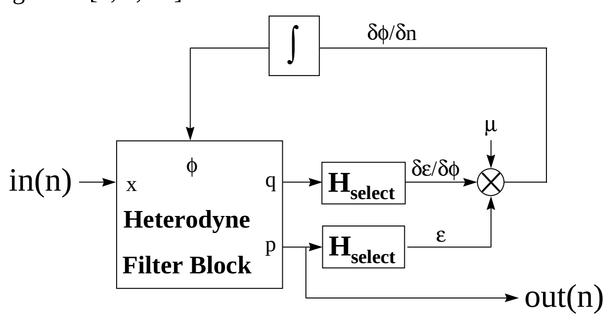



Figure 2. Adaptive LMS Algorithm Adapts IF frequency $\phi_i$

After passing through the Heterodyne Filter Block, the two signals pass through two identical selection filters that determine the area of sensitivity [9, 10]. The selection filters are made of 3 Gray-Markel lattice band-pass filters in series [11]. These selection filters limit the adaptation range for the notch filter so that the notch will not wander into an area of critical control signals. The selection filter also limits the range over which the IF frequency must be adjusted. The in-phase signal is the error of the system. The out-phase is the derivative of that error with respect to $\phi$. These are multiplied together with $\mu$ to form the derivative of $\phi$ with respect to time, (based on the a LMS algorithm) which is then integrated to form $\phi$ which is the change in phase per sample for the heterodyne.

## 2.3 Derivation of the Filter Response

First let us take a look at the Heterodyne Filter Block of Figure 1. Assuming for the moment that $\phi$ constant, we may write the expression for $u_1$ and $u_2$ in the frequency domain as:

$$U_1(e^{j\omega}) = X(e^{j\omega})*S(e^{j\omega}) = 1/2j\,[X(e^{j(\omega-\phi)}) - X(e^{j(\omega+\phi)})] \quad (1)$$

$$U_2(e^{j\omega}) = X(e^{j\omega})*C(e^{j\omega}) = 1/2\,[X(e^{j(\omega-\phi)}) + X(e^{j(\omega+\phi)})] \quad (2)$$

We see that the signal $X(\omega)$ has been translated up to the new IF frequency f where the fixed filter may operate on it. We may then translate this back to the baseband. Making use of equations (1) and (2) we find $v_1$ and $v_2$ in the frequency domain:

$$\begin{aligned} V_1(e^{j\omega}) &= U_1(e^{j\omega})H_f(e^{j\omega}) \\ &= 1/2j\, H_f(e^{j\omega})\,[X(e^{j(\omega-\phi)}) - X(e^{j(\omega+\phi)})] \end{aligned} \quad (3)$$

$$\begin{aligned} V_2(e^{j\omega}) &= U_2(e^{j\omega})H_f(e^{j\omega}) \\ &= 1/2\, H_f(e^{j\omega})\,[X(e^{j(\omega-\phi)}) + X(e^{j(\omega+\phi)})] \end{aligned} \quad (4)$$

We may then find the output signal *p(n)* in the frequency domain as follows:

$$\begin{aligned} W_1(e^{j\omega}) &= V_1(e^{j\omega})*S(e^{j\omega}) \\ &= -1/4\,[H_f(e^{j(\omega-\phi)})\,(X(e^{j(\omega-2\phi)}) - X(e^{j\omega}))] \\ &\quad +1/4[H_f(e^{j(\omega+\phi)})(X(e^{j\omega}) - X(e^{j(\omega+2\phi)})] \end{aligned} \quad (5)$$

$$\begin{aligned} W_2(e^{j\omega}) &= V_2(e^{j\omega})*C(e^{j\omega}) \\ &= 1/4\,[H_f(e^{j(\omega-\phi)})\,(X(e^{j(\omega-2\phi)}) + X(e^{j\omega}))] \\ &\quad +1/4[H_f(e^{j(\omega+\phi)})(X(e^{j\omega}) + X(e^{j(\omega+2\phi)})] \end{aligned} \quad (6)$$

adding the terms we form:

$$P(e^{j\omega}) = ½\,[H_f(e^{j(\omega-\phi)}) + H_f(e^{j(\omega+\phi)})]X(e^{j\omega}) \quad (7)$$

so the system transfer function is

$$\begin{aligned} P(e^{j\omega})/X(e^{j\omega}) &= H_{system}(e^{j\omega}) \\ &= ½\,(\,H_f(e^{j(\omega-\phi)}) + H_f(e^{j(\omega+\phi)})) \end{aligned} \quad (8)$$

This is a most remarkable result as it says that there will be no aliases created by the heterodyne in this filter arrangement. Both filters are of identical shape and the same order. They are centered about $\pi/2-\phi$ and $\pi/2+\phi$ (if the original was around $\pi/2$). If there is no signal of interest above $\pi/2$ the result is that we have moved the filter purely by heterodyne. No heterodyne-aliasing filters are required and no delay is introduced as was the case in previous AHF filters [10].

## 2.4 Derivation of LMS Gradient

Assuming that are gradient is dependent on $H(e^{j\omega})$, we will need its derivative with respect to the IF frequency $\phi$. For this we will start with $H(e^{j\omega})$ and transform it back to time space, then take a derivative, and then manipulate it back to frequency space so we can generate this from other terms available in the filter.

$$H(e^{j\omega}) = ½ [H_f(e^{j(\omega-\phi)}) + H_f(e^{j(\omega+\phi)})] \qquad (9)$$

In the time domain, we must express h(n) as a sequence of changing frequencies $\phi_i$ since $\phi$ is not a constant:

$$h(n) = \cos\{\Sigma^{n}_{i=0}\phi_i\}\ h_f(n) \qquad (10)$$

take derivative of equation (10) term by term:

$$\delta h(n)/\delta\phi = -\sin\{\ \Sigma^{n}_{i=0}\phi_i\}\ h_f(n) \qquad (11)$$

Now make the typical assumption that is made in most LMS algorithms that we can estimate the actual derivative by the current sample:

$$h(n) = -\sin(\phi n)\ h_f(n) \qquad (12)$$

$$\delta h(n)/\delta\phi = 1/(2j)\ \ [-e^{+jn\phi}\ h_f(n) + e^{-jn\phi}\ h_f(n)\ ] \qquad (13)$$

Now going back to the frequency domain:

$$\delta H(e^{j\omega})/\delta\phi\ = -1/(2j)\ [\ H_f(e^{j(\omega-\phi)}) - H_f(e^{j(\omega+\phi)})\ ] \qquad (14)$$

From Figure 1 we see that equation (14) can be obtained from the terms $w_3$ and $w_4$ :

$$\begin{aligned} W_3(e^{j\omega}) &= V_1(e^{j\omega})*C(e^{j\omega}) \\ &= 1/(4j)\ [H_f(e^{j(\omega-\phi)})\ (X(e^{j(\omega-2\phi)}) - X(e^{j\omega}))] \\ &\quad +1/(4j)[H_f(e^{j(\omega+\phi)})(X(e^{j\omega}) - X(e^{j(\omega+2\phi)})] \qquad (15) \end{aligned}$$

$$\begin{aligned} W_4(e^{j\omega}) &= V_2(e^{j\omega})*S(e^{j\omega}) \\ &= 1/(4j)\ [H_f(e^{j(\omega-\phi)})\ (X(e^{j(\omega-2\phi)}) + X(e^{j\omega}))] \\ &\quad -1/(4j)[H_f(e^{j(\omega+\phi)})(X(e^{j\omega}) + X(e^{j(\omega+2\phi)})] \qquad (16) \end{aligned}$$

Taking the difference of equation (15) and (16) yields the desired derivative of equation (14) Hence q(n) in the diagram of Figure 1 is the derivative needed for the LMS algorithm to adjust the IF frequency.

## 3. EXPERIMENTAL RESULTS

### 3.1 Experimental Setup

The Adaptive Heterodyne Filter (AHF) was applied to the problem of detecting and attenuating a narrow-band signal caused by a mechanical resonance in a flight control system designed for Rockwell Division of Boeing North America [9]. The mechanical resonance frequency varies slowly with time due to temperature changes and ranges in frequency between 100 Hz and 115 Hz. The control signals are all located below 100 Hz. The sampling rate of the system is fixed by other considerations at 6 kHz. Therefore our fixed Filter of Figure 1 is chosen to be the bandpass filter at $\pi/2$ or 1.5 kHz as shown in Figure 3. The notch is achieved by subtracting the resulting bandpass filter from one. Technically it is slightly more complex than this because the output p(n) in Figure 1 is characterized by the sum of two bandpass filters as seen in equation (9). In general, each of the passbands of these bandpass filters is corrupted slightly by the tail of the other bandpass filter requiring a correction factor in order to produce the desired notch filter. However, by choosing the bandpass filter as shown in Figure 3, the correction factor happens to turn out to be achieved by subtraction it from the input as shown in Figure 3. Details of this will be presented in a future paper. It is also possible to replace the bandpass filter with a notch filter and achieve the same results, but once again a correction factor is needed which can be implemented the same way.

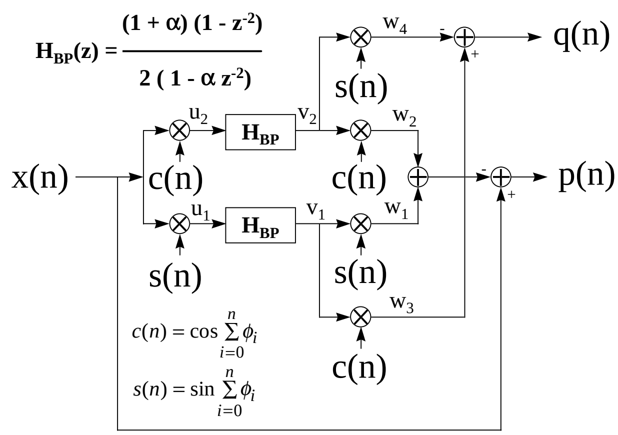



Figure 3. AHF for Experimental Results

### 3.2 Attenuation of a Stationary Sine in Noise

The first experiment was designed to demonstrate that the AHF can detect and attenuate a fixed sine wave without attenuating the desired control signals. The selection band was set from 100 Hz to 115 Hz and the filter setup of Figure 3 was used. Figures 4 shows the simulation results for an interfering sine wave at 107 Hz (amplitude 1.0) in noise (mean 0.5 uniformly distributed) with control signals at 20 DC (amplitude 0.5) and at 200 Hz (amplitude 1.0). The fixed filter is at 1.5 kHz as shown in Figure 3 and the adaptive circuit successfully adjusted the IF frequency to attenuate the sine wave. We can see in Figure 4 that the filter output (dotted line) has provided a 40db attenuation of the mechanical resonance dropping it to the noise floor. The bandpass filter uses α = 0.99 and the optimum LMS step size was μ = 0.00004. Convergence occurred within 3000 samples.

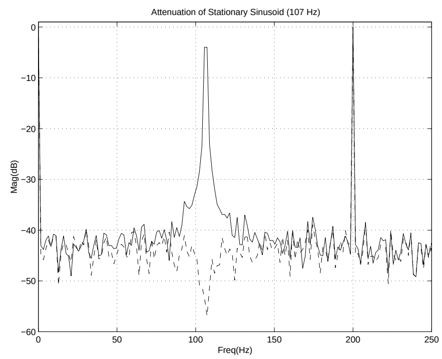



Figure 4. Attenuation of a Stationary Sine in Noise

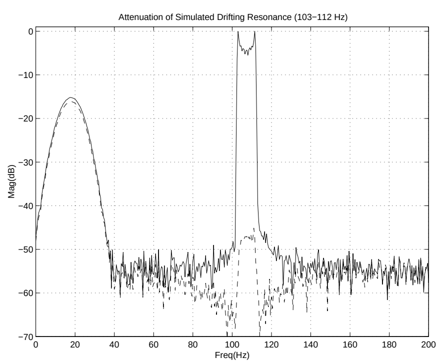



Figure 5. Attenuation of a Time Varying Sine in Noise

### 3.3 Attenuation of Time Varying Sine in Noise

The second experiment was designed to demonstrate that the AHF can detect and attenuate a time varying sine wave without attenuating the desired control signals. Once again we chose the selection band to be from 100 Hz to 115 Hz and the filter setup of Figure 3 was used. Figures 5 shows the simulation results for a time varying interfering sine wave (fm modulated with amplitude 1.0, center frequency 107.5 Hz and deviation 4.5 Hz) in noise (mean 0.5 uniformly distributed) with control signals represented more realistically by a Gausian distributed signal at 18 Hz with noise as before. The fixed filter is at 1.5 kHz as shown in Figure 3 and the adaptive circuit successfully adjusted the IF frequency to attenuate the time varying sine wave. We can see in Figure 5 that the filter output (dotted line) has provided a 45db attenuation of the mechanical resonance dropping it to the noise floor. The bandpass filter uses α = 0.95 and the optimum LMS step size was μ = 0.0006. Convergence occurred within 3000 samples.

## 4. SUMMARY

Using the AHF of Figure 1 with the LMS adaptive algorithm of Figure 2, we are able to accomplish adaptive heterodyne filtering without the need for anti-ailiasing filters. This is a major improvement over previous AHF designs.

## 5. REFERENCES


[1] K. Feher, *Advanced Digital Communications: Systems and Signal Processing Techniques,* Prentice-Hall, Inc., Englewood Cliffs, NJ, 1987.

[2] K. Feher, *Wireless Digital Communications: Modulation and Spread Spectrum Applications,* Prentice-Hall, Inc., Englewood Cliffs, NJ, 1995.

[3] W. Gao, M.A. Soderstrand and K. Feher, *Gaussian Filter Screens TDMA and Frequency-Hopping Spread-Spectrum Signals,* Microwave and RF, May 1995, pp. 17-19.

[4] H. Yan, M.A. Soderstrand, J. Borowski and K. Feher, *DSP Implementation of GFSK, GMSK and FQPSK Modulated Wireless Systems,* RF Digital Communications, June 1995, pp. 28-32, June 1995.

[5] M.A. Soderstrand, T.G. Johnson, R.H. Strandberg and H.H. Loomis, Jr., *Suppression of Multiple Narrow-Band Interference Using Real-Time Adaptive Notch Filters,* IEEE Transactions on Circuits and Systems, Vol. 44, No. 3, March 1997, pp. 217-225.

[6] R.H. Strandberg, M.A. Soderstrand and H.H. Loomis, *Elimination of Narrow-Band Interference Using Adaptive Sampling Rate Notch Filters,* Proceedings 28th IEEE Asilomar Conference on Signals Systems and Computers, Pacific Grove, CA, October 1992, pp. 861-865.

[7] R.H. Strandberg, J.C. Le Duc, L. G. Bustamante, V.G. Oklobdzija and M.A. Soderstrand, *Efficient Realizations of Squaring Circuit and Reciprocal used in Adaptive Sample Rate Notch Filters,* Journal of VLSI Signal Processing Systems for Signal, Image, and Video Technology, Vol 14, No. 3, December 1996, pp. 303-310.

[8] L.G. Bustamante and M.A. Soderstrand, *Switched-Capacitor Adpative Sample Rate Filter,* Proceedings 40th IEEE International IEEE Midwest Symposium on Circuits and Systems, Sacramento, CA, August 1997.

[9] K.E. Nelson, P.V.N. Dao and M.A. Soderstrand, *A Modified Fixed-Point Computational Gradient Descent Gray-Markel Notch Filter Method for Sinusoidal Detection and Attenuation,* IEEE International Symposium on Circuits and Systems, Hong Kong, China, June 1997.

[10] K.E. Nelson and M.A. Soderstrand, *AdaptiveFiltering Using Heterodyne Frequency Translation,* Proceedings 40th IEEE International Midwest Symposium on Circuits and Systems, Sacramento, CA, August 1997.

[11] A.H. Gray, Jr. And J.D. Markel, *Digital Lattice and Ladder Filter Synthesis,* IEEE Transactions on Audio, Vol AU-21, No. 6, December 1973, pp. 491-500.