Investigations in Self-Adaptive Filtering and Set-On Oscillators for EW Systems.

Steve Middleditch, Ian C. Hunter, Roger D. Pollard.
Institute of Microwaves and Photonics.
The University of Leeds
Woodhouse Lane,
Leeds. LS2 9JT

Abstract

In the time domain complex non-linear systems can be difficult and cumbersome to manipulate and understand; transforming them into the frequency domain using the signals Instantaneous Frequency (IF) allowing simplifications, that make analysis of these circuits more straightforward. Using these principles a software simulation program describing both a self-adaptive filter (SAF) and a set-on oscillator (SO) was developed in Matlab, results obtained show strong agreement with work previously published [1]. To corroborate the results further, a test rig of the system was built and tested at L band, results were positive. Alternative system architectures are outline; including the realization of a band stop version of the SAF. The main issues to be explored have been identified as; the development of a wideband, long analogue delay line, and the use of frequency dividers and multipliers to produce an accurate fast locking oscillator for use at X band (i.e.: close modal spacing).

Keywords: Excision jammers, linear frequency networks, Frequency divider, Self adaptive filter, Set On Oscillator, Matlab modelling of mm-wave signals.

Motivation

As feature sizes in silicon fabrication become smaller, ideas investigated some years ago become interesting again. A lack of appropriately wideband components and subsystems has not been available until recently. This is the case in many areas of research, new developments in manufacturing instigate reinvestigations into ideas previously passed over. In particular, wideband digital delay lines with delays in excess of several hundred nano-seconds were impractical to realize [2]. Hence interest has shifted toward placing signal processing blocks in the transmitter, where size, power consumption and computational resources are more abundant [3]. An attractive proposition within these systems would be the availability of an efficient, wideband, low power noise reduction system in the receiver, relaxing the constraints on the transmitted signal in terms of power, pre-distortion criterion and coding. There are many methods to achieve these goals [4-6], but many are limited by their use of digital techniques, which reduces bandwidth, due to inadequately fast clock speeds, this is constantly in flux as feature sizes reduce and clock speeds increase.

Programmable adaptive filters and oscillators are useful in many areas of communications systems where accurate, stable selection and generation of tones is required. Applications of these filters are widespread in all types of communications systems. Applications include mobile telephony transceiver front ends, hearing aids [7], digital magnetic storage [8], broadband adaptive filtering in wired communications systems, echo cancellation, voiceband modems, and frequency memory loops in Electronic Warfare (EW) systems. This paper investigates adaptive filtering techniques and their application in EW systems

For an EW system to be successful, it must complete several simple tasks;

-search for threat signals or objects,

-identify all signals and objects accurately,

-deceive other spectrum users, in any combination of range, angle or velocity.

To complete these tasks any system must be able to accurately estimate the incoming signals' Instantaneous frequency (IF). Once obtained, this data may be used to deceive the source of this unknown signal. For pulsed RADAR systems the speed at which a system can determine the IF is of paramount importance, particularly systems employing frequency hopping techniques. In these cases phase locked carrier recovery circuits are too slow [9]. In all of these applications systems that are robust, less expensive, easily implemented and incorporated into existing systems are favoured [10].

Introduction

A linear frequency network (LFN) is an arbitrary connection of system elements, which perform linear transformations on the IF of a signal. The measurement of IF's and hence research into LFN's has become increasingly more important with respect to applications in EW, wired and wireless communication systems. Non-adaptive methods have been shown to give poorer interference rejection when compared to adaptive methods (up to 21dB less)[5, 8, 11, 12]. This project concentrates on the development of a self-adaptive filter (SAF) and set-on oscillator (SO) originally investigated in 1988 [1, 13]. Some of the advantages of these techniques include; lower power consumption (which reduces heat dissipation issues), small size, greater reliability, lower cost, larger bandwidths, with lower complexity, increased stability and improved SNR. The revisiting of these concepts has been encouraged by new techniques employed in the manufacture of system on chip designs. Here Monolithic Microwave Integrated Circuits (MMIC) are populated with both analogue and digital components describing a complete system, for these reasons detailed models of systems and subsystems are required. The goal of this project is a detailed investigation of the modelling and physical realisation of a SAF and SO.

Examples of LFN's include; delay lines, frequency dividers, multipliers and mixers. The transfer functions relating the input to the output of these systems can be represented simply and directly, eg: for a delay, we have the output defined in terms of the input in the time domain; $\omega_0(t) = \omega_i(t-\tau)$, taking Laplace transforms gives the transfer function, $G(p) = z = e^{-p\tau}$.

A bandpass filter characteristic is formed by the connection of the components described in figure 1. Following the theory outlined here and in detail in [1] the transfer function of the system is;

$$\left|H(z, z_1)\right|^2_{p=j\omega} = \cos^2\left(\frac{\tau}{2}\omega\right).$$

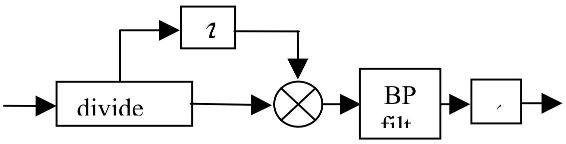

Figure 1: Self Adaptive
Filter Schematic.

When a feedback loop is added to the previous circuit a SO is formed, figure 2 shows a simplified schematic. Oscillation modes are set up in the system, these modes occur when integer 2π radian phase shifts exist in the loop. They will be at a frequency spacing of $\frac{1}{\tau}$ Hz. The output takes the form;

$$\omega_2(t) = \omega_0(t) + \psi_0 \sum_{i=1}^{\infty} \delta(t - i\tau),$$

where ψ_0 represents the phase discontinuities when the switch is closed every τ seconds and ω_0 is the fundamental frequency of the signal.

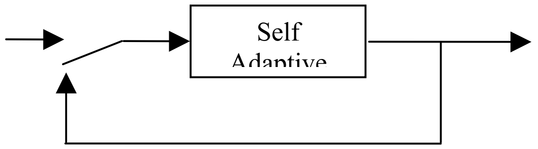

Figure 2: Frequency Set on Oscillator Schematic.

Approach

The focus of this project falls into several categories; modelling the system in Matlab, an experimental test set up and modelling in ADS using data from the rig. Using Matlab has allowed investigations in both the time and frequency domains.

The development of the model involved reducing the system into its individual components, each described by a simple function block. Each function was then developed and tested individually; the need for ancillary functions became clear as the model was developed. Figure 3 shows the model schematic and associated function calls, with its input and output variables: Note the addition of the loop in the main program block, this loop transforms the model of the SAF to one describing a SO, given that the loop were set to more than one. The scheme employed for this model was chosen so that function debugging, modification and maintenance could be carried out efficiently.

Having written the software to model the SAF it was simple to write the software describing the looping nature of an oscillator circuit. The inclusion of the main loop counter is equivalent to a feedback loop in the circuit (any number of 'rotations' is possible), thus forming a SO.

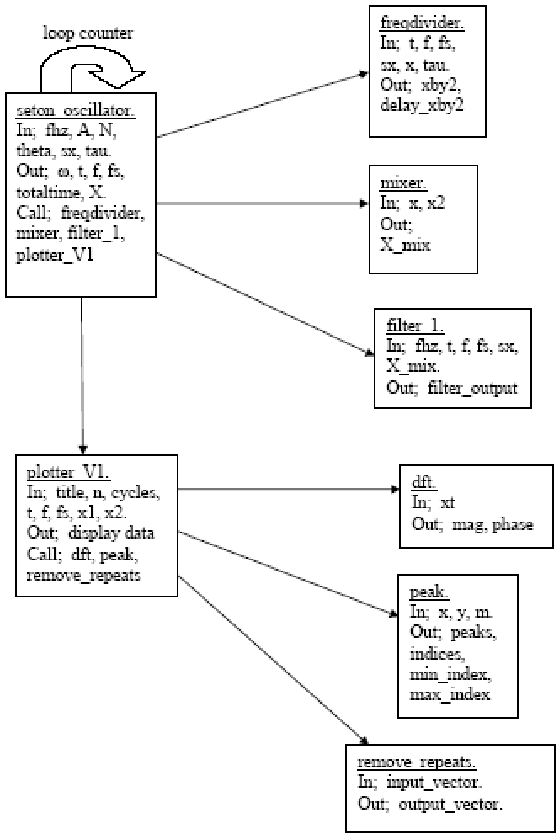

Figure 3: Schematic of Matlab system model, each block describes an individual function.

Review of Model Algorithm

The input signal vector consisting of a sinusoid (f_0 = 500MHz) with additive white noise (at -40dBc) was created in Matlab having the form;

$$v(t) = A\sin(\omega_0 t + \theta(t)),$$

where; ω_0 = carrier frequency,
t = time,
θ = phase angle
A = signal amplitude.

This signal was represented in Matlab as a sampled row vector, which is plotted against a similar vector representing the sample time intervals. The filter was designed with a 20% bandwidth centred at 500MHz as was intended for the realized system.

There are several points, which are of interest in the execution of the system described. The frequency divider employs a method of oversampling to form a new vector at half the instantaneous frequency of the original. This was done by averaging each pair of adjacent data points, which are then interleaved with the original signal vector. The time delay function removes a defined time period at the head of the signal vector, by removing the appropriate number of data points. The mixer simply multiplies the original and the delayed vectors element by element. Finally, a Butterworth filter was implemented in Matlab using one of its design tools, thus band limiting the signal at the output. Strictly speaking, the model does not require this function, there are no physical restrictions to the model, since every system block was programmed as ideal.

Preliminary Model Results

With one program loop, results for the SAF were obtained, a reduction of the noise floor of about 6 dB was gained, the noise floor also showed a periodic pattern, as expected, proportional to the delay used. Figures 4 shows the signals at the mixer inputs, showing how the divider has successfully divided the 500MHz instruction pulse by two and the delay has created the phase difference at the ‘head’ of the pulse. Figures 5 and 6 show clearly the way that the signal begins to oscillate at defined frequencies, note the way that these oscillations become more defined as the loop number increases and energy is translated into the only frequencies that are supported in the system. The separation of these modes was as expected from theory outlined in the introduction, [1 and 13].

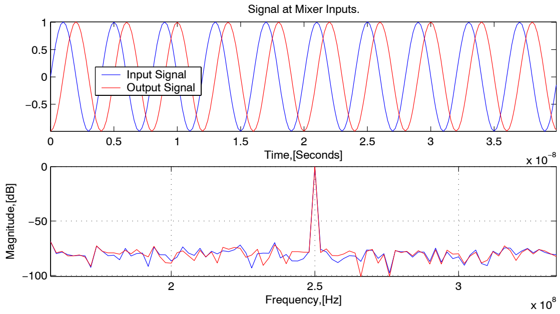

Figure 4: Signals at the Mixer Input. Note the phase difference at t=0 introduced by the time delay.

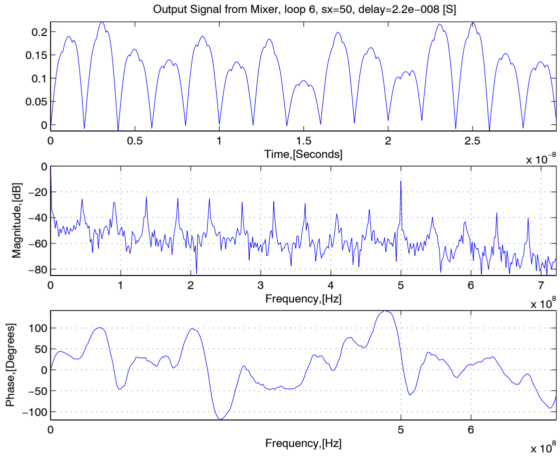

Figure 5: Signal at the mixer output after the sixth loop, modal oscillations are becoming clearer.

Notice in figure 5 how the signal shows a DC component, due to the DC term introduced when a squared sinusoidal signal is decomposed into a signal at twice the fundamental frequency and a DC term.

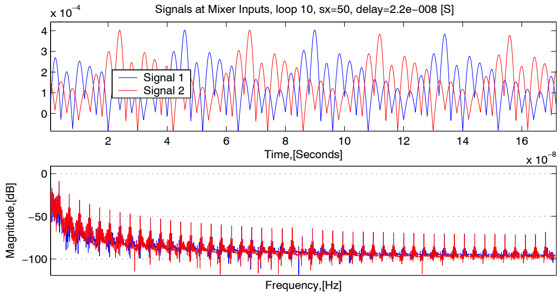

Figure 6: Full spectrum of signal at the mixer input after loop ten. The range of this graph is between 0Hz and the Nyquist

frequency (12.5GHz), where the spectrum will repeat itself as an artifact of the sampling of the signal.

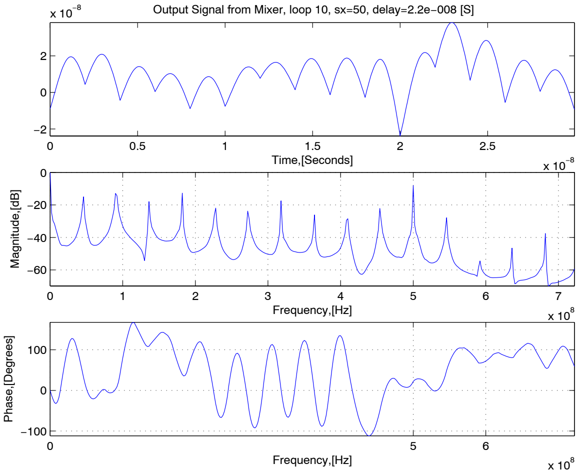

Figure 7: Signal at the Mixer Output in the tenth circulation around the loop. Modal frequencies of oscillation are very prominent.

After the program, or signal, has re-circulated ten times the power in the modal frequencies becomes prominent, see figure 7. The noise power has been 'transferred' into the only frequencies that can exist within the system, describing a typical oscillator. Worthy of note is the shape of the signal at the output of the band-limiting filter. A 'hump' can be seen to exist in the passband of the signal in the frequency domain. This phenomenon is due to the exponential envelope added at the head of the signal vector by the filter. This can be removed for clarity, by taking the Fourier transform of the signal after the envelope has decayed. Also of interest to Matlab users is the reaction of the software to using very high sample rates in conjunction with higher order filters. That is to say, the spectrum of signals tends to spread as the sample rate increases, due to illconditioned Eigenvectors within the filter structure, causing poor denominator coefficient calculations.

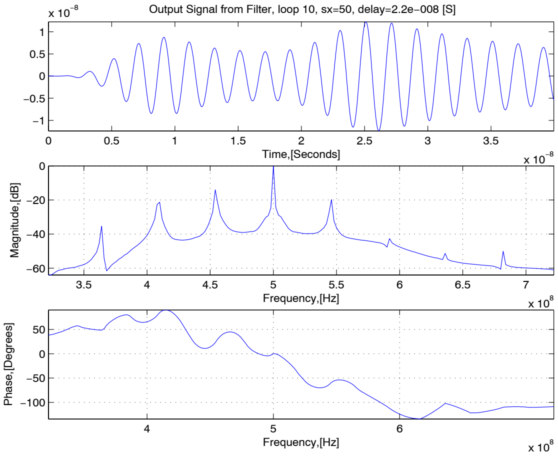

Figure 8: Output from band limiting filter after the tenth loop. Notice the 'hump' in the frequency domain plot, caused by the exponential envelope at the start of the signal. This is an artifact of the rise time of the filter.

Experimental Test Rig of the SAF

What follows is a description of the construction of an experimental test rig, used to confirm the concepts investigated. Data from the software model and rig was validated by direct comparison of results gained in [1 and 13].

All relevant system blocks were designed and fabricated, those components that would prove onerous to produce were bought in (i.e.: the frequency divider and amplifiers). The components were connected as shown in figure 9, omitting the feedback loop to form the SAF. What is clear from the results is the validity of the theory and the software model.

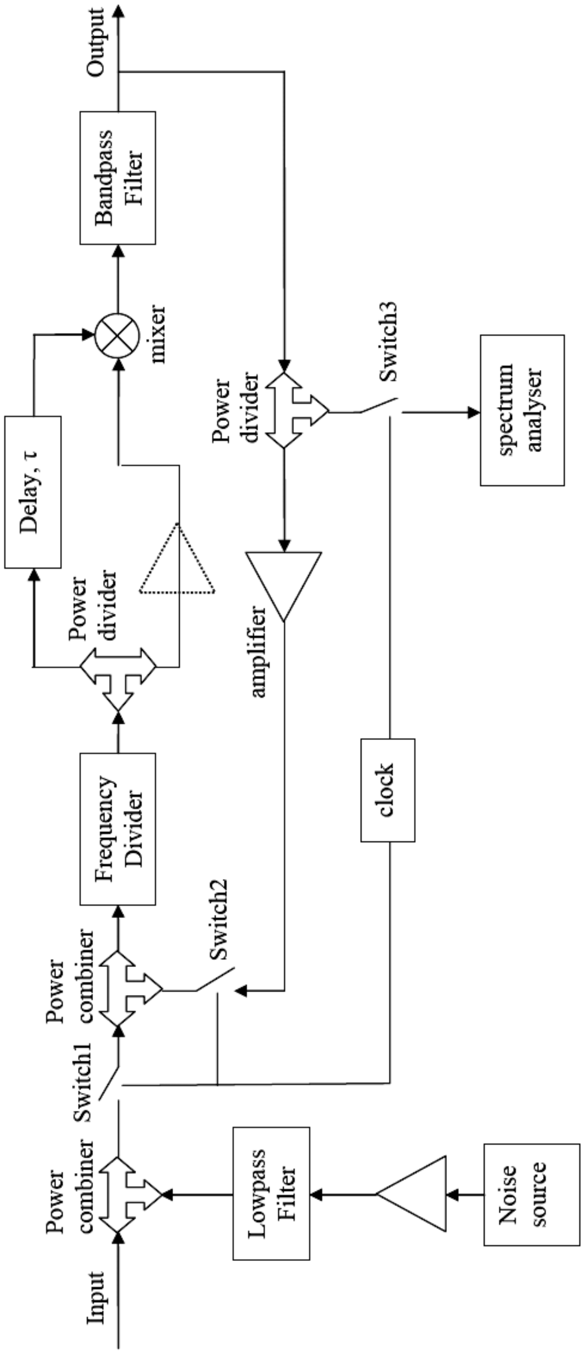

Figure 9: Schematic diagram of the experimental test rig, without the loop the system describes the SAF.

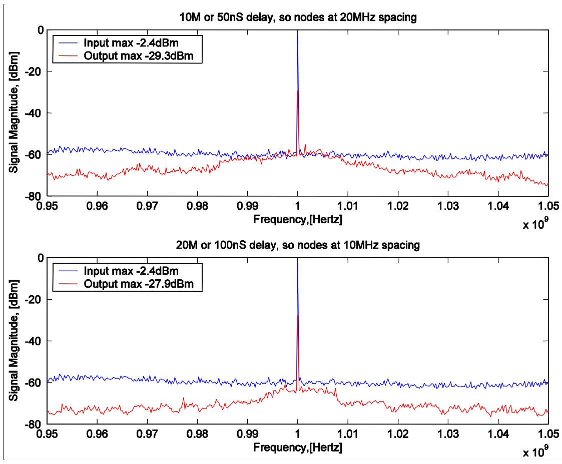

Figure 10: Input and output spectra from the experimental test rig, for delays of 50 and 100nS.

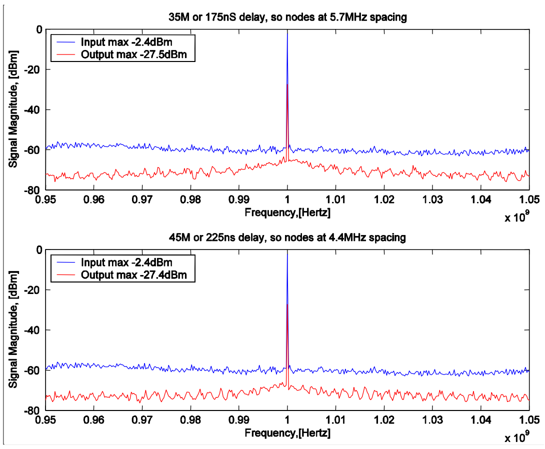

Figure 11: Input and output spectra from the experimental test rig, for delays of 175 and 225nS.

Figures 10 and 11 show the input signal directly related to the output signal. Closer inspection shows that although the signal amplitude has been attenuated, the noise floor is below its original level. Also noticeable are the modes set up in the system that will translate into oscillation frequencies once the feedback loop is included. Comparing any two graphs with different delay periods helps to highlight the periodicity of the ripples in the noise floor. The separation of these is as described earlier in this paper.

During the investigations outlined above, the idea of using a lower sideband mixer in the system, instead of the upper sideband mixer was explored. The transfer function of this 'new' device ,

$$\left|H(z,z_1)\right|^2_{p=jw} = \sin^2\left(\frac{\tau}{2}w\right),$$

shows the formation of a notch filter. Further research showed that this device had been developed and used some years ago and was known as a serradine receiver. This device may also be described as a baseband demodulator with no requirement for a local oscillator, the idea was abandoned.

Conclusions

The Matlab suite was used to successfully produce a model of both a SAF and SO. Results show direct correlation with those achieved previously at lower frequencies. In fact, the Matlab model can be easily set at any arbitrary frequency. Solutions for some of the errors encountered while developing the software model have been solved. The experimental test rig also corroborates the models validity, giving the project some legitimacy. The experimental test rig gives good results, yet power losses throughout the system are something that should be addressed. The ADS model that uses this data has proven to be quite onerous, in particular the frequency dividing of the IF, suitable solutions for this have not yet been found.

Future Plans

Having established a sound understanding of the problem and its related issues, a clear view of what is required has been established. Several interesting ideas will be investigated further. Having shown the principals to work at lower frequencies (L band), a test rig built at X band where applications in EW have been highlighted will be of use.

The development of a wideband delay of 100 nS is proving to be one of the more challenging parts of this project. Detailed examinations of differing types of circuit that may offer the appropriate delay will be undertaken. Studies of allpass networks, resonant circuits, SAW, Optical, high speed digital and YIG devices will be explored. The need for this long delay is propelled by the desire to increase the 'accuracy' of the locking oscillator (i.e.: narrow modal spacing, ≈ 10 MHz). The inclusion of a `divide by N' block, followed by a `multiply by N' block in the feedback path, will be investigated in an attempt to ratify these issues, see figure 12. While figure 13 shows a similar idea to reduce the modal spacing by employing a different architecture.

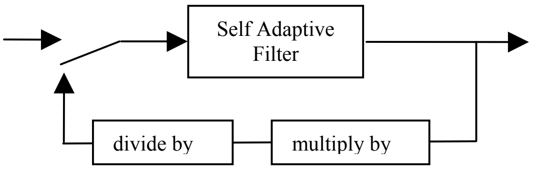

Figure 12: Schematic of system to reduce the modal spacing of the oscillator but retain the overall system bandwidth.

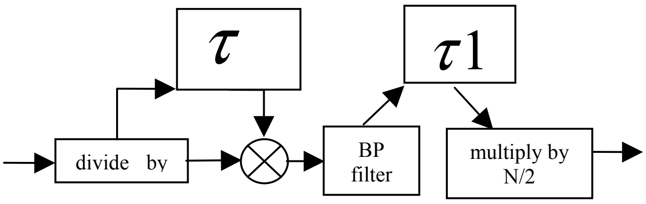

Figure 13: Schematic of system that needs further investigation, but may result in the creation of a highly selective filter and oscillator.

Acknowledgements

The author would like to thank the Electro-Magnetic Remote Sensing (EMRS) DTC for funding this research. Also, not forgetting the invaluable help from the University of Leeds staff and the two project supervisors; Professors Hunter and Pollard.

References

[1] J. D. Rhodes, "Linear frequency networks with applications to `frequency set-on' oscillators," presented at Circuits and Systems, 1988., IEEE International Symposium on 7-9 Jun 1988, 1988.

[2] F. Cornett and D. K. Reynolds, "Continuous-time adaptive signal processing," presented at Southeastcon '90. Proceedings., IEEE, 1990.

[3] M. Brandt-Pearce, "Transmitter-based multiuser interference rejection for the down-link of a wireless CDMA system in a multipath environment," *Selected Areas in Communications, IEEE Journal on*, vol. 18, pp. 407-417, 2000.

[4] S. Tanaka, F. Behbahani, and A. A. Abidi, "A Linearization Technique For CMOS RF Power Amplifiers," presented at VLSI Circuits, 1997. Digest of Technical Papers., 1997 Symposium on, 1997.

[5] A. N. D'Andrea, V. Lottici, and R. Reggiannini, "RF power amplifier linearization through amplitude and phase predistortion," *Communications, IEEE Transactions on*, vol. 44, pp. 1477-1484, 1996.

[6] A. R. Mansell and A. Bateman, "Transmitter linearisation using composite modulation feedback," *Electronics Letters*, vol. 32, pp. 2120-2121, 1996.

[7] D. Graupe, S. P. Basseas, and J. K. Grosspietsch, "Self adaptive filtering of noise from speech: a hearing aid application," presented at Circuits and Systems, 1988., IEEE International Symposium on, 1988.

[8] A. Carusone and D. A. Johns, "Analogue adaptive filters: past and present," *Circuits, Devices and Systems, IEE Proceedings [see also IEE Proceedings G-Circuits, Devices and Systems]*, vol. 147, pp. 82-90, 2000.

[9] W. A. Gardner and W. A. Brown, "Frequency-shift filtering theory for adaptive co-channel interference removal," presented at Signals, Systems and Computers, 1989. Twenty-Third Asilomar Conference on, 1989.

[10] J. D. Laster and J. H. Reed, "Interference rejection in digital wireless communications," *Signal Processing Magazine, IEEE*, vol. 14, pp. 37-62, 1997.

[11] D. C. Shin and C. L. Nikias, "Adaptive interference canceler for narrowband and wideband interferences using higher order statistics," *Signal Processing, IEEE Transactions on [see also Acoustics, Speech, and Signal Processing, IEEE Transactions on]*, vol. 42, pp. 2715-2728, 1994.

[12] S. Haykin, *Unsupervised Adaptive Filtering. Volume 1: Blind source Separation*, vol. 1, 1 ed: John Wiley and Sons, Inc., 2000.

[13] J. D. Rhodes, "Linear Frequency networks with applications to `Frequency Set-On Oscillators' and 'Self Adaptive Bandpass Filters'," *International Journal of Circuit Theory and Applications*, vol. 18, pp. 121-134, 1990.