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Introduction to Polyphase Filters and Its Applications

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ABSTRACT

Polyphase filters are becoming a very important component in the design of various filter structures due to the fact that it reduces the cost and complexity of the filter by doing the process of decimation prior to filtering which reduces the multiplications per input sample. In this paper, a review of the basic polyphase filter and its applications has been reviewed.

Keywords- **AFE, APPF, CMOS, DSP, FIR.**

I. INTRODUCTION

The world of science and engineering is filled with signals such as images from the remote space probes, voltages generated by the heart and brain and countess other applications. Digital signal processing is used in a wide variety of applications. Digital refers to operating by the use of discrete signals to represent data in the form of numbers. Signal refers to a variable parameter by which information is conveyed through an electronic circuit. Processing means to perform operations on data according to programmed instructions. So, DSP is defined as changing or analyzing information which is measured as discrete sequences of numbers. By its very nature DSP is a mathematically heavy topic and to fully understand it students need to understand the mathematical developments underlying DSP topics. However, relying solely on mathematical developments often clouds the true nature of the foundation of a result. It is likely that students who master the mathematics may still not truly grasp the key ideas of a topic. Teaching DSP necessarily requires heavy use of mathematics – the nature of the material requires mathematics to precisely specify the methods and firmly establish their characteristics and performance. In digital signal processing, many application areas require sampling rate alteration. The processes involved in the alteration of the sampling rate are interpolation and decimation. One of the most efficient structures to implement interpolation and decimation operations is the polyphase structure. Decimation is the process of reducing the sample rate Fs in a signal processing system, and interpolation is the opposite, increasing the sample rate Fs in a signal processing system. These processes are very common in signal processing systems and are nearly always performed using an FIR filter. First, why are sampling rates changed? The most common reason is to ease the interface of the digital signals to the outside environment .As we saw in previous chapters, signals have a frequency representation, and this frequency representation must be less than the Nyquist frequency, which is defined as Fs/2.This sets a lower bound on Fs. The amount of hardware or software processing resources is normally proportional to Fs, so we usually want to keep Fs as small as practical. So while there is no upper bound on Fs, it is usually less than 10Â the frequency representation of the signal. A minimum Fs is needed to ensure the highest frequency portion of the signal does not approach the F-Nyquist frequency.

II. DECIMATION AND INTERPOLATION

The idea of decimation polyphase filters is developed as follows. First, the idea of filter-then-decimate is introduced. Namely, the signal to be decimated, $x[n]$, is first filtered by a filter with impulse response $h[n]$ to give the intermediate signal $v[n]$ given by

$$v(n) = \sum_i x(i)h(n-i), \qquad (1)$$

and then that filtered signal is decimated by an integer factor M to give the lower rate signal $y[n]$ as

$$y(n) = v(nM) = \sum_i x(i)h(nM-i), \qquad (2)$$

which is simply (1) with n replaced by nM, to enact the decimation. In such developments it is then pointed out that although this structure accomplishes the desired goal it is computationally inefficient.

Up to here the development is intuitive and instructive. However, at this point the polyphase structure is then developed one of two ways, neither of which provides much insight or understanding – even when fully understood. The first way is a time-domain development that uses a non obvious re-indexing of the summation in (2) given by

$$i = i'M + m \ with\ i' \in Z, \quad \mathrm{m} = 0,1,2,\dots,\mathrm{M}-1 \qquad (3)$$

which, after some mathematical manipulation of double summations, leads to the desired polyphase filter structure.

The second way is a z-domain development that starts by first demonstrating that the filter transfer function can be reorganized into a sum of the polyphase component transfer functions, as given by

$$H(z) = \sum_{m=0}^{M-1} z^{-m} \ P_m(z^m) \qquad (4)$$

with

$$P_m(z) = \sum_n h[nM + m]z^{-n}$$

The development is then completed by multiplying this form of $H(z)$ by $X(z)$, applying the z-domain result for decimation, and finally exploiting the noble identities (also called the multirate identities) to move the decimation to the front of the resulting system[1].

III. ACTIVE AND PASSIVE POLYPHASE FILTERS

Typical modern radio transceiver architectures require circuits for generation of high frequency local oscillator quadrature signals. There are several well-known methods to implementing the quadrature generation for the local oscillator signal on chip. These include e.g. divider, quadrature VCO and RC-CR based circuits. Passive polyphase filter implementation is known to require good quality passives, which is no longer self-evident with deep-submicron CMOS processes. Additionally, the tuning of the passive polyphase filters is difficult. Therefore, with the high-speed transistors available, active polyphase filters (APPF) have been proposed to overcome these limitations [2].

The active polyphase filters provide important advantages over their passive counterparts. With APPFs signal amplification, filter response calibration and tuning are possible. Contrary to PPFs, the most critical design parameters of APPFs are determined by the active elements [3]. Compared to passive polyphase filter implementation, benefits of the APPF include the possibility for filter calibration, tuning and signal amplification. Although practical APPF implementations have been presented, this is the first time APFFs have been analyzed in terms of gain and stability. Stability analysis is crucial for APPFs because due to the nature of the APPFs, they have a tendency to oscillate easily. Quadrature signal generation is an essential part of modern telecommunication RF front-end signal processing. Nowadays, commonly used direct-conversion and low-IF receivers, see, e.g., [4]–[6], require two local oscillator (LO) signals in quadrature. Three commonly applied methods for in-phase (I) and quadrature-phase (Q) signal generation are the use of phase shifter, divide-by-two circuit, and coupled oscillator, see, e.g., [7]. It is a manifold matter to select among these. All are relevant and selection depends on the targeted system, selected radio architecture, and applied IC process. Amplitude and phase balance of the generated I and Q signals affect strongly on the image rejection of the receiver and thus on the quality of reception.

IV. APPLICATIONS OF POLYPHASE FILTERS

In many applications we need to compensate for nonlinear distortions. Consequently, we have to apply nonlinear signal processing because linear methods do not provide acceptable results. Examples of such applications include equalization of digital communication channels, pre- and post-linearization, e.g., for loudspeakers, microphones, or analog front-ends (AFE) of communication systems.

4.1 Predistortion for the Linearization of Weakly Nonlinear Systems:

In the literature, several predistortion methods are known. Here, we focus on a special method for weakly nonlinear systems proposed by Gao and Snelgrove [8]. The basic scheme is shown in Fig. 1. Unfortunately, the preprocessor must operate at the high sampling rate [9]. This means that under the assumption of a pth-order nonlinearity the necessary sampling rate is -times higher than the sampling rate of the baseband input signal.

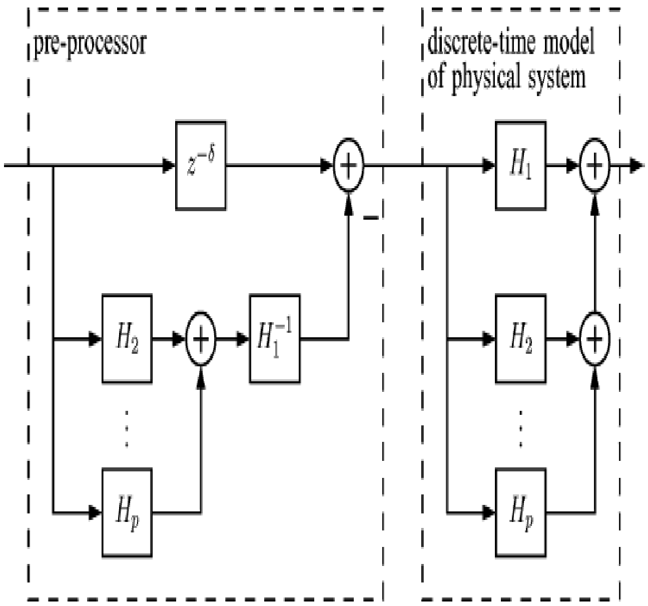

Fig-1 Linearization of a weakly nonlinear system modelled as a pth-order Volterra filter using a preprocessor

4.2 Nonlinear Echo Cancellation with Volterra Filters:

We consider the simplified block diagram of a full-duplex data transmission system as shown in Fig.2. Typically, if the sampling rate of the transmit/receive signal is not too high, oversampling converters are employed for the AD and DA conversion. Therefore, we have to upsample (downsample) the transmit (receive) signal to (from) the higher sampling rate of the converter. After the transmit filter, the line driver amplifies the signal and feeds it to the line via a hybrid circuit which performs the two-wire/four-wire conversion. The hybrid itself is by nature a leaky device. Some part of the transmit energy is reflected back into the receive path, creating the echo signal which should be attenuated by the echo canceller. Modelling the echo path, we have to consider that some analog circuitry can exhibit strong nonlinear behaviour. Consequently, we have to use nonlinear filters.

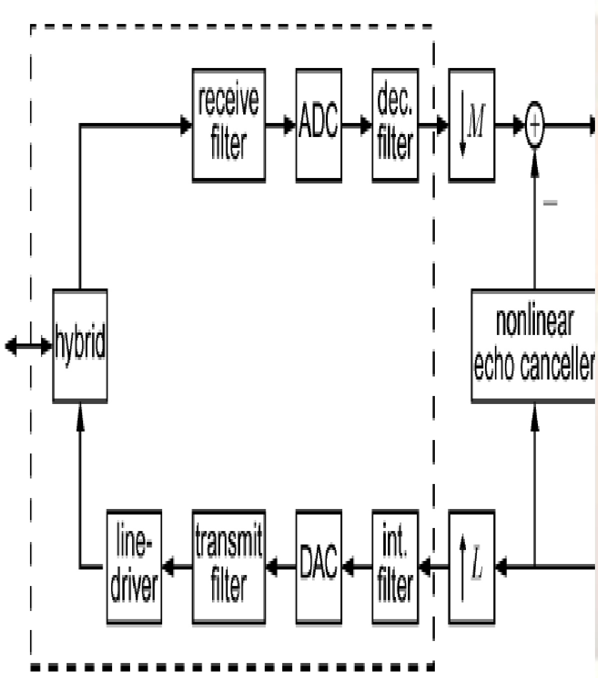

Fig-2 Echo path in a full-duplex communication system

REFERENCES

[1] Mark L. Fowler, "*POLYPHASE FILTERS – A MODEL FOR TEACHING THE ART OF DISCOVERY IN DSP"*, IEEE International Conference on Acoustics, Speech and Signal Processing (ICASSP), 2011, pp 2908-2911

[2] Tillman, F.; Sjoland, H., "*A polyphase filter based on CMOS inverters"*, NORCHIP Conference, 2005, Page(s):12 – 15.

[3] Kaltiokallio, M.; Ryynänen, J.; Lindfors, S. ,"*Active Plyphase Filter Analysis"*, IEEE International Symposium on Circuits and Systems (ISCAS), Proceedings of 2010 , Publication Year: 2010 , Page(s): 1125 - 1128.

[4] B. Razavi, RF Microelectronics. Englewood Cliffs, NJ: Prentice-Hall, 1998, p. 335.

[5] A. A. Abidi, "*Direct-conversion radio receivers for digital communications,"* IEEE J. Solid-State Circuits, vol. 30, no. 12, pp. 1399–1410, Dec. 1995.

[6] J. Crols and M. S. J. Steyaert, "*A single-chip 900 MHz CMOS receiver front-end with a high performance low-IF topology,"* IEEE J. Solid-State Circuits, vol. 30, no. 12, pp. 1483–1492, Dec. 1995.

[7] A. Y. Valero-Lopez, S. T. Moon, and E. Sánchez-Sinencio, *"Self-calibrated quadrature generator for WLAN multistandard frequency synthesizer,"* IEEE J. Solid-State Circuits, *vol. 41, no. 5*, pp. 1031–1041, May 2006.

[8] X. Y. Gao and W. M. Snelgrove, *"Adaptive linearization schemes for weakly nonlinear systems using adaptive linear and nonlinear FIR filters,"* in Proc. 33rd Midw. Symp. Circuits Syst. (MWSCAS), 1990, pp. 9–12.

[9] W. A. Frank, *"On the equalization of nonlinear systems,"* in Proc.Conf. Record 30th Asilomar Conf. Signals, Syst. Comput., 1997, pp. 1157–1160.

[10] David Schwingshackl, Gernot Kubin, *"Polyphase Representation of Multirate Nonlinear Filters and Its Applications"*, in IEEE TRANSACTIONS ON SIGNAL PROCESSING, *VOL. 55*, NO. 5, MAY 2007,pp 2145-2157