Applications of the Motorola DSP56200 Digital Filter Chip

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ABSTRACT

The Motorola DSP56200 chip is a unique peripheral dedicated to digital filtering. It is designed for use in low-cost, high performance systems, and performs one of two principal functions - Finite Impulse Response (FIR) filtering or adaptive filtering. Each function finds a wide variety of applications, including standard filtering (highpass, lowpass, etc.), echo cancelling, interference cancelling, telephone line equalization, Hilbert transforms, interpolation, and matched filtering. This paper discusses useful applications of high-speed FIR and adaptive filters.

Introduction

Digital Filtering is rapidly becoming a cost effective solution to many signal processing problems. Digital components do not drift with temperature, voltage, or age, giving digital filters a significant advantage over their analog counterparts. Improved quality is attainable, and new functions such as adaptive filtering are possible using digital techniques. Two common structures found in digital signal processing (DSP) are Finite Impulse Response (FIR) filters and adaptive filters.

The Motorola DSP56200 is an algorithm specific DSP peripheral, capable of high speed filtering either as a FIR filter or as an adaptive filter using the Least-Mean-Squares (LMS) algorithm. It uses wide coefficient and data words to minimize quantization error, and can run at speeds in excess of 200 KHz. The number of taps is programmable up to 256 taps per chip, and the DSP56200 can easily be cascaded for longer tap lengths.

Digital filtering can be applied to a wide set of problems in areas such as telecommunications, speech processing, and general signal processing. FIR filters can closely approximate any desired frequency response and adaptive filters provide a unique solution to difficult problems such as noise and interference cancellation.

Digital Filtering Basics

1. Finite Impulse Response Filters

Finite impulse response (FIR) filtering [1,2,3] is a common method of performing digital filtering. This filtering process can be viewed as a weighted moving average of past data values. The FIR filtering equation (1) describes how the filter's output is related to its input:

$$y(n) = \sum_{i=0}^{N-1} h(i)\, x(n-i) \qquad (1)$$

where:

n = the time index
i = the filter tap index
N = the number of taps
h(i) = the "ith" coefficient
x(n-i) = the "ith" most recent data sample

Thus, the FIR filtering process is an accumulation of N product terms. The coefficients are usually pre-calculated using software design tools, and are based on the user's desired frequency response. These coefficients represent the impulse response of the desired filter. Equation 1 describes the discrete convolution of a digitized input waveform with an impulse response function.

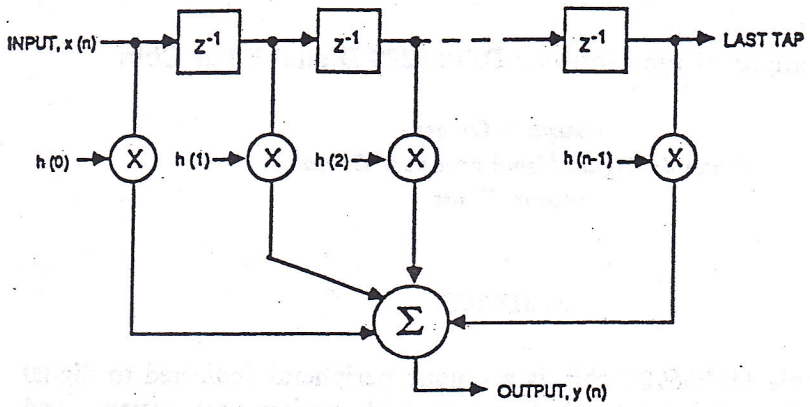

Figure 1. Structure of a Finite Impulse Response (FIR) Filter

FIR filtering can be readily implemented in hardware as shown in Figure 1. Each z^{-1} represents a memory element with a delay of one sample period. N of these memory elements are cascaded together to form a tapped shift register of length N, which is used to store the N most current data samples, each of which is called a "filter tap". The width of the shift register is the number of bits used to represent the data samples - 16 bits in the case of the DSP56200. At the start of a sample period, a new data sample is entered into the shift register and the oldest data sample is discarded.

A multiplier-accumulator (MAC) is also required to calculate and accumulate the product terms. The MAC multiplies a coefficient with its corresponding data sample and adds this product with the sum of the previous products already in the accumulator. Thus, N multiply-accumulates occur each time a new sample is shifted into the FIR filter.

FIR filters can be designed to have a linear phase characteristic, and are inherently stable since they have no poles. The linear phase characteristic is desirable since it ensures that an input data signal is always delayed by a constant length of time, independent of its frequency content.

2. Adaptive Filters

Adaptive filters [3,4] are a special class of filters used to solve a unique set of signal processing problems. These filters have two inputs, x(n) and d(n), which are usually correlated in some manner (Figure 2). One input, x(n), is passed through an internal time varying filter, which tries to form an estimate of the desired input, d(n). The parameters of this internal filter eventually converge to a point where the internal filter can accurately estimate the desired input, minimizing the adaptive filter's output. This output represents the difference or error e(n), between the desired signal and the estimate of the desired signal. There are two aspects to adaptive filters - the internal filter structure and the adaptation algorithm.

The most common adaptive filter implementation is based on a FIR filter structure whose

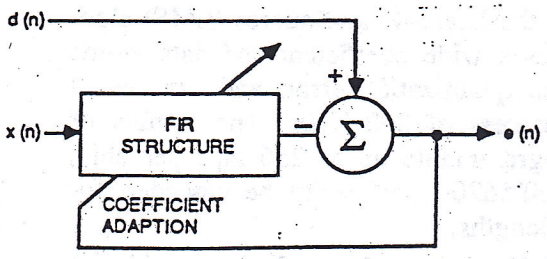

Figure 2. Block Diagram of an Adaptive Filter

coefficients are adapted using the Least-Mean-Squares (LMS) algorithm. First, the signal x(n) is FIR filtered using equation 1 to provide an estimate of the second input, d(n). The error in the estimate is calculated using the following formula:

$$e(n) = d(n) - \sum_{i=0}^{N-1} h(i)\, x(n-i) \qquad (2)$$

This error term is then used to modify every FIR filter coefficient using an equation derived from the LMS adaptation algorithm:

$$h(i)_{new} = h(i)_{old} + Kex(n-i) \qquad (3)$$

where:

h(i) new = the value of the updated "ith" coefficient to be used during the next sample period

h(i) old = the value of the "ith" coefficient used during the current sample period

K = the gain constant
e = the calculated error term
x(n-i) = the "ith" most recent data sample

At the beginning of the next sample period, two new samples, x(n) and d(n), are shifted into the system and the process repeats.

After several iterations, the FIR filter coefficients converge to values which consistently minimize the mean square error (the filter's output). At this point, the adaptive filter is able to estimate the d-input by passing the x-input through the FIR filtering hardware. It is interesting to note that once the filter has converged, the coefficients of the internal FIR filter resemble the impulse response of a filter whose input is x(n) and output is d(n). Equation 3 is derived from an approximation of an equation from the steepest descent algorithm, where the minimum error point is found by updating in the direction opposite the gradient [3,4].

DSP56200 Description

The DSP56200 is an algorithm specific peripheral designed to perform the computationally intensive tasks associated with digital filtering [11]. No DSP software development is required because the algorithms have already been implemented in hardware. The chip operates in one of three modes - Single FIR Filter, Dual FIR Filter (two independent FIR filters on one DSP56200), and Single Adaptive FIR Filter. The DSP56200 contains both a 256-tap FIR filter section and the update circuitry necessary to implement the LMS adaptation algorithm. The input data word width is 16 bits, the coefficient word width is 24 bits, and the output word width is 32 bits. In order to maximize throughput in applications not requiring a full 256-tap FIR section, the user may program the number of taps from 4 to 256 taps.

A block diagram of the DSP56200 is shown in Figure 3. All numerical operations are handled by a 16-bit by 24-bit Arithmetic Unit which performs MAC operations and coefficient update operations. The coefficients are stored in a 24-bit, 256 location static RAM, and the data samples are stored in a 16-bit, 256 location static RAM configured as a circular queue. Bus controllers control the data transfers between the two RAM arrays and the Arithmetic Unit. Additional features of the DSP56200 include a dc tap option, optional 16-bit rounding of the output, a programmable loop gain (Single Adaptive Filter mode), and an adaptation disable capability (Single Adaptive Filter mode).

The DSP56200 interfaces to a host processor through an 8-bit parallel port in a manner similar to a fast static RAM. The DSP56200's Parallel Interface is an independent unit which operates concurrently with its Computation Unit, allowing a host processor to access the DSP56200 registers anytime during a sample period (Figure 4).

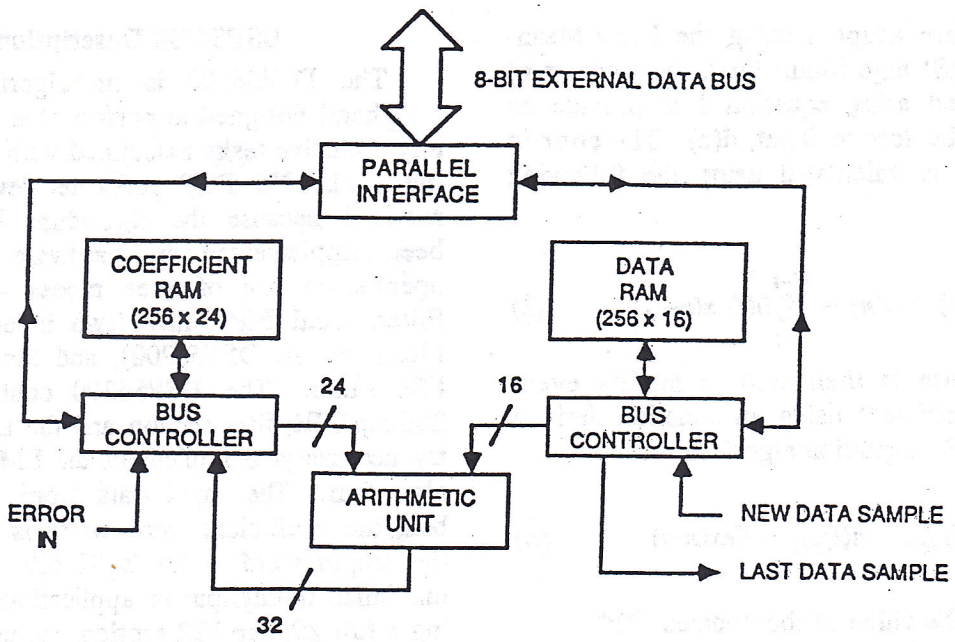

Figure 3. Computation Unit Block Diagram

Longer tap lengths and/or higher sampling rates can be realized by cascading several DSP56200s together. The Cascade Interface Unit controls all the interchip communication required in multi-chip filter systems, eliminating the need for extra "glue" chips or communication software in the host. In addition, higher sampling rates can be attained by cascading more DSP56200s together and decreasing the number of taps used on each chip. Performance figures for several example configurations are shown in Table 1.

Table 1. DSP56200 Performance

Single Chip

Maximum Sampling Frequency (kHz)				
Mode	Number of Taps			
	32	64	128	256
FIR Filter	227	132	71	37
Adaptive Filter	123	69	37	19
Dual FIR Filter	122	68	36	-

Eight Chips in Cascade

Maximum Sampling Frequency (kHz)				
Mode	Total Number of Taps			
	256	512	1024	2048
FIR Filter	204	132	71	37
Adaptive Filter	115	69	37	19

Four Chips in Cascade

Maximum Sampling Frequency (kHz)				
Mode	Total Number of Taps			
	128	256	512	1024
FIR Filter	222	132	71	37
Adaptive Filter	120	69	37	19

Sixteen Chips in Cascade

Maximum Sampling Frequency (kHz)				
Mode	Total Number of Taps			
	512	1024	2048	4096
FIR Filter	175	132	71	37
Adaptive Filter	105	69	37	19

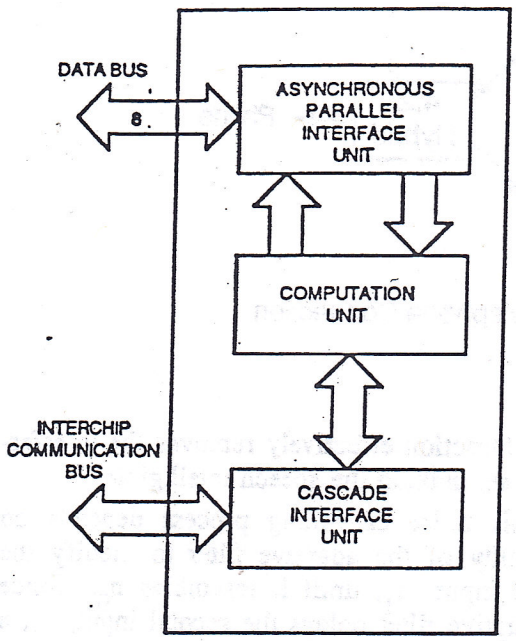

Figure 4. Parallel Units Within the DSP56200

Adaptive Filtering Applications

The adapting nature of adaptive filters makes them useful in systems with unknown or slowly varying characteristics. Applications fall into four general classes:

Prediction
System Identification
Inverse Filtering
Interference Cancelling

Prediction finds uses in signal encoding. System identification applications seek to find the impulse or frequency response of an unknown system. Undesired filtering effects such as dispersion can be compensated for with an inverse filtering operation, and interference cancelling is used to extract a desired signal from background interference or noise. Some specific applications are discussed in the following text. Other applications include adaptive control and extraction of periodic components in the presence of broadband noise [4].

1. Echo Cancellation

One popular application of adaptive filters is in the area of echo cancellation [3,4,7,8]. Consider the telephone system shown in Figure 5. A 4-wire link is used for long distance transmission and is converted to a local 2-wire link through a hybrid transformer. Ideally, all of the transmitted message should pass through the transformer to the 2-wire link, but due to impedance mismatches, some of the signal is actually reflected into the Rx path at the hybrid. This reflection results in an audible echo received by the speaker. As the distance of transmission increases, the delay of the echo also increases.

An adaptive filter provides an excellent solution to this problem (Figure 6). The filter synthesizes an impulse response modeling the hybrid connection and the transmission delays. The input is passed through the FIR structure, producing a synthetic echo which is then subtracted from the actual echo. As the adaptive filter converges, the error decreases, resulting in less echo returned to the speaker.

The Motorola DSP56200 is well suited to solving the above problem. Long delays can be cancelled by a set of chips in cascade. The adaptation process can be inhibited with a control bit, eliminating any incorrect adaptation when both parties talk simultaneously, a situation called doubletalk. A programmable leakage term is included on the chip to prevent the coefficient drift which results from narrowband input signals such as tones. The DSP56200 can also be used in a similar manner to provide acoustic echo cancellation for speaker phones.

2. Noise Cancellation - Two Inputs

Adaptive filters are very effective at noise cancelling. Consider an environment with a strong level of background interference such as a manufacturing area or the cockpit of a loud aircraft. A microphone in this environment will pick up both the desired signal, s, and the interfering noise, n_0, resulting in a signal with a poor signal-to-noise ratio.

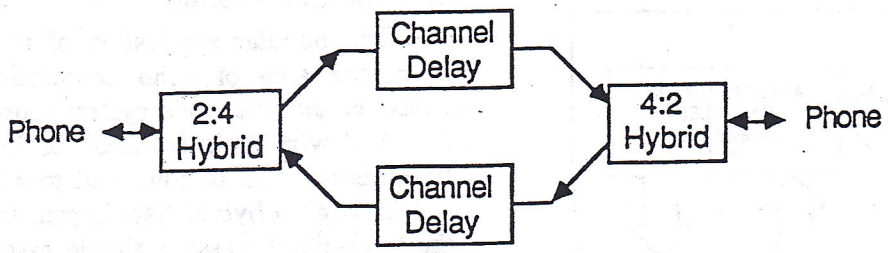

Figure 5. Model of a Long Distance Telephone Connection

The adaptive filtering system shown in Figure 7 can be used to remove the background noise. A second microphone is required to pick up only the background noise, n_1. Note that n_1 will be highly correlated with n_0, but may differ in amplitude and phase since it is not picked up from the same location. The adaptive filter will filter n_1 until it resembles n_0, and then subtract this result from the signal coming from the first microphone, $s+n_0$.

$$e(n) = s(n) + n_0(n) - \sum_{i=0}^{N-1} h(i)\, n_1(n-i)$$

$$\approx s(n)$$

The subtraction effectively removes the interfering noise, making the speech intelligible.

This noise cancelling process depends on the ability of the adaptive filter to modify the second input, n_1, until it resembles n_0. Since the adaptive filter delays the second input, n_1, a delay of about N/2 taps is also needed in the primary input, where N is the number of taps in the adaptive filter. This delay allows for a better matchup of the n_0 signal and the filtered n_1 signal, resulting in better cancellation.

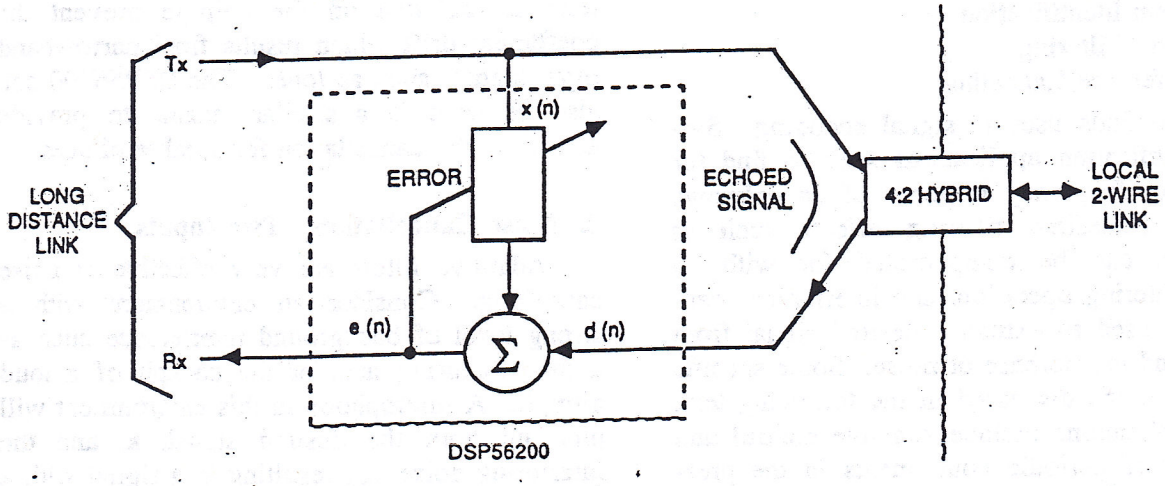

Figure 6. Echo Cancelling Application

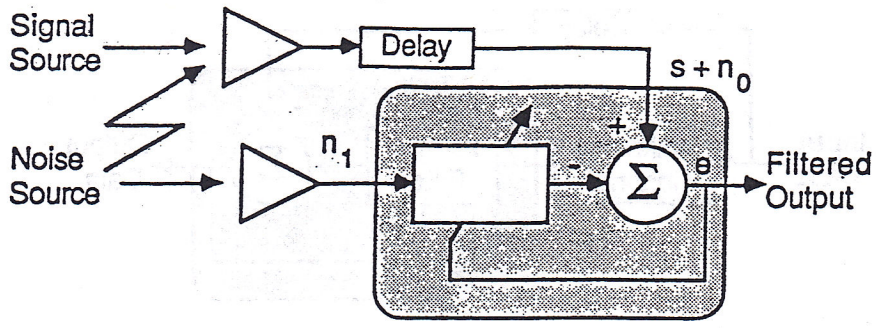

Figure 7. Noise Cancelling with a Two Input System

3. Noise Cancellation - Single Input

The previous system works well for interference cancelling, but often it is not possible to obtain the second adaptive filter input which is free of the desired input signal. An example of this situation is where the data has already been recorded with the noise present, such as tape hiss on a cassette deck. In this case, it is still possible to cancel the noise if the noise is periodic in nature.

The cancelling circuit is shown in Figure 8. The signal and additive interference, s+n, take two paths to the adaptive filter - one path directly in and one path which is delayed. By the time the delayed signal reaches the adaptive filter, the signal component present on the undelayed input is no longer correlated with the signal present on the delayed version, and the adaptive filter will not be able to cancel out the desired signal. Since the noise is periodic, however, the noise components on both inputs will always be correlated. As a result, the adaptive filter converges to a filter which allows the undelayed noise component to be predicted by the delayed noise component. This estimate is subtracted from s+n, leaving the desired signal as the error term of the adaptive filter.

This method is successful when the noise is periodic, such as 60 Hz powerline interference or engine noise. The delay must be long enough to decorrelate the desired signal components at both inputs of the adaptive filter.

4. Adaptive Line Equalization

Adaptive equalization is a technique for compensating for any undesired frequency response introduced through a communication channel. In high speed modem applications, for example, high data rates are hard to transmit due to the poor frequency response of telephone lines. Not only is there limited bandwidth, but

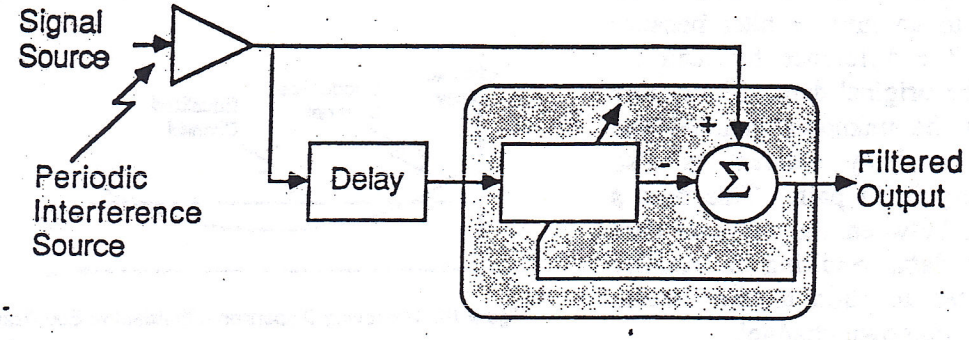

Figure 8. Cancelling Periodic Noise with a Single Input System

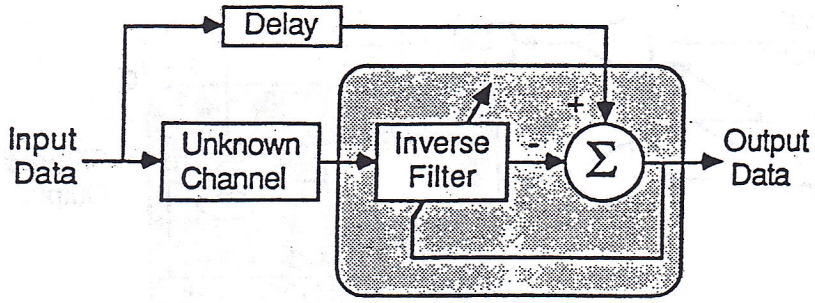

Figure 9. The Adaptive Equalization Concept

the phone line also introduces dispersion (phase distortion), delaying different frequencies by different amounts. This creates a situation called intersymbol interference, where neighboring data pulses interfere with each other. The purpose of an adaptive equalizer is to correct for the undesirable amplitude and phase response of the telephone line, thereby increasing the reliability of the transmission.

An adaptive filter can be used for the inverse filtering operation. The concept is demonstrated in Figure 9. A stream of data is transmitted through an unknown channel which causes intersymbol interference. The data is then passed through the adaptive filter which adapts to the characteristics of an inverse filter. Upon convergence, the combined frequency response of the unknown channel and the adaptive filter has a flat amplitude spectrum and a linear phase characteristic (all frequencies are delayed by the same amount) over the frequencies of interest. Figure 10 shows how the phase response is corrected. The adaptive filter correctly converges to an inverse filter because it tries to minimize the difference between the equalized data and the original data. Since there is delay inherent in the unknown channel and the adaptive filter, a delay element is also needed in the primary input path. This results in better correlation between the original data and the transmitted data, resulting in smaller error terms and better adaption to the inverse characteristics of the unknown channel.

In many applications, however, the original input data is not available at the receiver. In this case, the primary input to the adaptive filter is derived in an alternate fashion using a method called "decision-directed" learning. More information on this topic can be found in [5,9,10].

FIR Filtering Applications

FIR filters have several characteristics which make them desirable for use in DSP applications. Unusual frequency responses can be synthesized with FIR filters, using software tools to calculate the required coefficients. Stability is guaranteed since the filter has no poles. The linear phase characteristic means there is no dispersion or phase distortion of incoming data. This characteristic is particularly useful in telecommunication applications, where intersymbol interference is a common problem.

Two factors determine how well a FIR filter can approximate a desired frequency response - the number of bits in the coefficient word and the number of filter taps. The DSP56200 provides a high degree of accuracy. Its coefficients

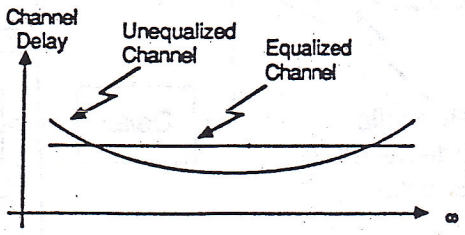

Figure 10. Removing Dispersion with Adaptive Equalization

are represented in 24 bits, and the number of filter taps can be arbitrarily increased through cascading. Steep "brickwall" frequency responses are attainable with the DSP56200.

Several popular applications of FIR filters include lowpass, highpass, bandpass, and band-reject filters. Bandsplitting filters find use in speech applications, and matched filtering with a FIR filter is an optimal way of detecting known signals in the presence of noise. Interpolation [6] is a technique for increasing the sampling rate of a sampled signal, and is used in compact disc players to reduce the order and complexity of the reconstruction filter. The use of FIR filtering for Hilbert transforms is discussed below.

Hilbert Transforms

Hilbert transforms [2,3] find use in communication systems. They are used in demodulating phase modulated waveforms transmitted from modems, and can also be used in single sideband modulation systems. The Hilbert transform operation is performed by a quadrature filter, which adds a -90 degree phase shift to all frequencies of interest. This is equivalent to multiplying the signal by the imaginary constant '-j'. The frequency response of such a filter is shown in Figure 11.

FIR filters can approximate any desired frequency response and are a natural solution for quadrature filtering. The impulse response of the desired filter is found by taking the inverse transform:

$$h(n) = \frac{1}{2\pi} \int H(e^{jw})\, e^{jwn}\, dw$$

$$h(n) = \begin{cases} \dfrac{2}{\pi n}\sin^2(\pi n/2) & n \neq 0 \\ 0 & n = 0 \end{cases}$$

Linear programming techniques [2] found in filter design software can be used to derive the FIR filter coefficients necessary to obtain the frequency response in Figure 11, and these coefficients will resemble the values of the filter's impulse response. As the number of filter taps is increased, the FIR filter gives a better approximation to the ideal response.

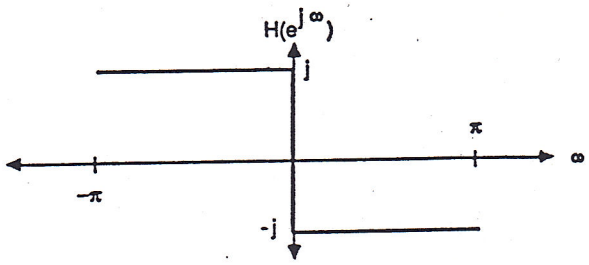

Figure 11. Frequency Response of a Quadrature Filter

Conclusions

Adaptive and FIR filters provide excellent solutions to many signal processing problems. FIR filters can approximate any desired frequency response, and the adaptation capability in adaptive filters makes them useful in sytems with unknown or slowly varying characteristics. Adaptive filters are very effective solutions for noise cancelling, echo cancelling, inverse modelling, and system identification. The Motorola DSP56200 provides both the speed and accuracy needed for high performance filtering, making it a cost-effective solution for digital filtering applications.

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