FIR Processing White Paper

March 2012

1. Introduction

There are two types of digital filters; Infinite Impulse Response (IIR) filters and Finite Impulse Response (IIR) filters. Generally, IIR filters are suitable for relatively simple filter responses which can be realized with a small amount of computation, whereas FIR filters are suited to achieve flexible filter responses although it requires a larger amount of computation.

Recently, FIR filters have been utilized widely in the pro audio industry as well as other industries thanks to progress in digital processors. This progress helps to realize the required processing capacity with low cost.

This document explains the characteristics of each of the IIR and FIR filters, and then shows technologies and products such as the MY8-LAKE card, the DSR/DXR series powered loudspeakers which utilize FIR filters.

2. IIR Filters and FIR Filters

IIR (Infinite Impulse Response) Filters

IIR filter configuration is utilized by both analog EQ and digital EQ based on analog EQ, using a feedback process to achieve the filter efficiently with relatively little computation. With a 2nd order (biquad) IIR filter as shown in Fig.1, widely-used equalizer responses such as peaking, shelving, highpass/lowpass filters can be achieved.

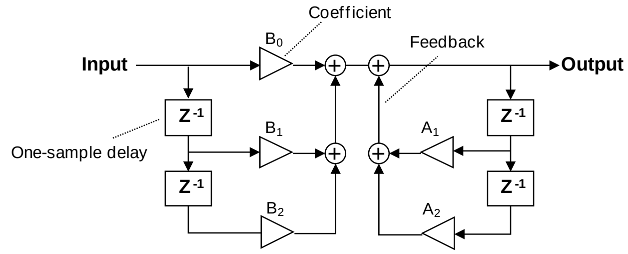

Fig.1 2nd order (biquad) IIR filter block diagram

FIR (Finite Impulse Response) Filters

In contrast to IIR, FIR filters obtain flexible and complex filter responses by utilizing a large amount of computation. The higher the filter order utilized, the more closely the desired result can be achieved.

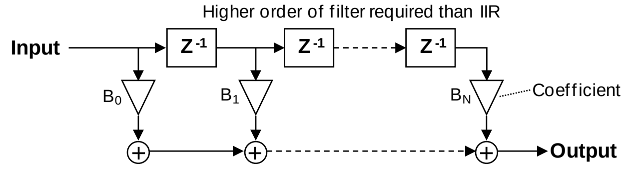

Fig.2 FIR filter block diagram

In short, IIR filters have feedback type processing for efficiency, whereas on the other hand, FIR filters have feed-forward type processing for flexibility.

An IIR filter multiplies multiple times the same coefficients in order to realize steep filter slopes by simple filter configuration. Although an IIR filter is suitable for simple responses with efficient computation, in order to obtain complex filter responses (i.e. an atypical response) a very complex logical block diagram is required which results in a huge amount of computation.

On the other hand, an FIR filter multiplies many coefficients to obtain the desired response. If it is required to multiply one hundred times, for example, to achieve a desired response, the FIR filter needs to allocate one hundred coefficients. However, each coefficient can be individually adjusted, which allows the FIR filter to achieve a complicated response.

To describe in a metaphor, IIR is cutting with a saw. FIR is engraving with a chisel. So if you want to cut a simple shape in wood, the saw is suitable. If you want to cut a complex shape the chisel is suitable.

3. Characteristics of FIR Filter

In addition to the high flexibility described above, FIR filters can be further categorized into two types in terms of phase response and latency.

Minimum Phase FIR

This is a type of FIR filter which is designed to minimize the group delay. This filter cannot control the phase but can realize the desired amplitude response which conventional IIR filters find difficult to achieve. In addition, this type of filter can be realized with minimal latency compared with other types of FIR filters.

This filter is effective for compensating for complex amplitude responses such as acoustic space (room EQ) and speaker tuning.

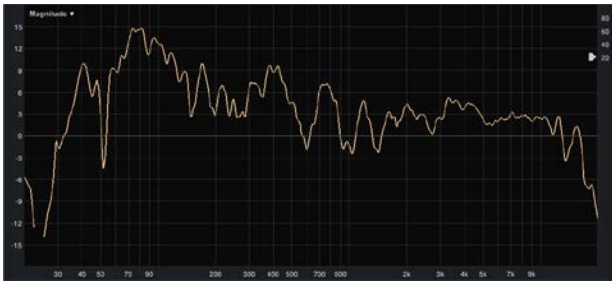

Fig.3 Example of complex amplitude response

Linear Phase FIR

In contrast to minimum phase, linear phase FIR is a type of FIR filter which is designed to fix the group delay; in other words this filter does not have any effect on phase response.

If a conventional IIR filter is applied to a speaker's crossover processing with a steep slope, the phase response is affected dramatically and the phase interference between LF and HF drivers causes cancellations in the amplitude response. However, a linear phase FIR filter can realize a steep amplitude response without affecting the phase response. This characteristic helps to minimize the crossover range between drivers and connect both of the amplitude and phase responses smoothly with linear phase.

Generally, linear phase FIR filter requires latency in order to compensate phase. The length of latency depends on the desired properties (curve). The lower the frequency controlled, the longer the latency required.

4. FIR processing in MY8-LAKE

MY8-LAKE is a compact DSP card for Yamaha digital mixers containing a Lake processor which has become a standard in the world of live sound. It has various features including the Lake Controller software that provides unique and intuitive operations. Here we focus on the characteristics derived from the FIR processing.

Mesa EQ & Ideal Graphic EQ

Thanks to Mesa EQ which has asymmetrical slope characteristics, flexible and quick EQ operation is possible. For example it can compensate for a high-frequency roll-off of a constant directivity horn. Another example is subwoofer boosting; you can draw a shallow slope property for a required frequency range which also has a very steep cut-off for out of band frequencies (Fig.4). Only one Mesa EQ is needed to achieve this.

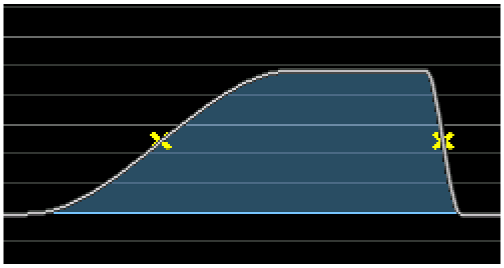

Fig.4 Mesa EQ

In addition, because the sides of the slope have a steep slope characteristic called Raised Cosine, interference with adjacent filters can be minimized.

This Raised Cosine filter is also applied to the graphic EQ so it allows intuitive operations unlike traditional graphic EQs whose real frequency response do not truly match the graphical representation of the faders. The "ideal" graphic EQ allows actual frequency response to match the response on-screen such as a flat amplitude response between adjacent bands and minimized cancellation with alternative boost and cut (Fig.5).

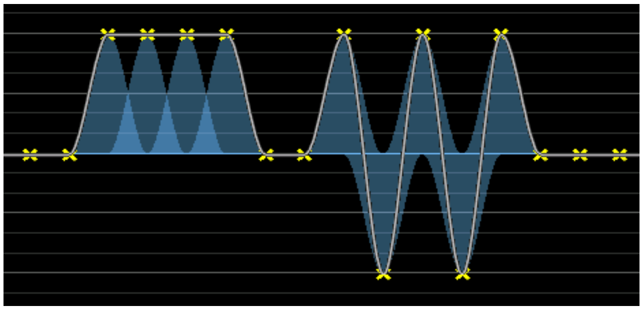

Fig.5 Ideal Graphic EQ

These Mesa EQs and graphic EQs consists of one minimum phase FIR filter(high order filter) on the processor; it is therefore possible to overlay up to 256 of EQ curves per module.

Linear Phase Crossover

When the Contour mode is used as a crossover processor, a linear phase FIR filter crossover can be chosen in addition to a classic IIR type crossover. Thanks to linear phase and from 24dB/oct to brick wall steep slopes, it is possible to achieve much smoother transitions with minimum phase interference between drivers.

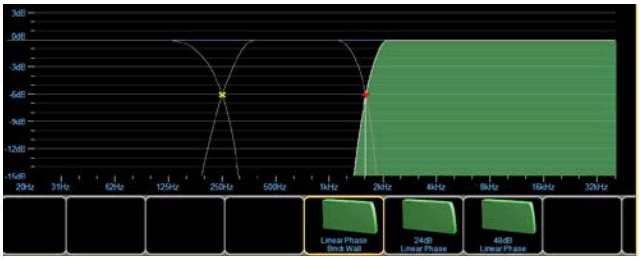

Fig.6 Linear Phase Crossover

However, in order to achieve linear phase crossover, 1.25ms or longer alignment delay in addition to normal processing latency is required. Lower crossover frequencies require longer alignment delay times. In live SR applications which seriously require low latencies, you should consider whether the latency is acceptable or not, and for example classic IIR based crossover for subwoofer might be another choice.

As a reference, Table.1 shows the relationship between alignment delays and the lowest crossover frequency for linear phase crossovers with 48dB/oct slope.

Table.1 48dB/oct 2/3-Way Alignment Delay

Alignment Delay	Lowest Frequency
1.25 ms	1.03 kHz
2.50 ms	515 Hz
5.00 ms	258 Hz
10.0 ms	129 Hz
20.0 ms	32.5 Hz
40.0 ms	31.6 Hz

5. FIR processing in DSR/DXR Series

FIR-X tuning

Yamaha's powered speaker DSR and DXR series adopts a linear phase FIR filter for built-in digital processing crossover. It achieves smooth amplitude and phase transition around the crossover frequency (Fig.7).

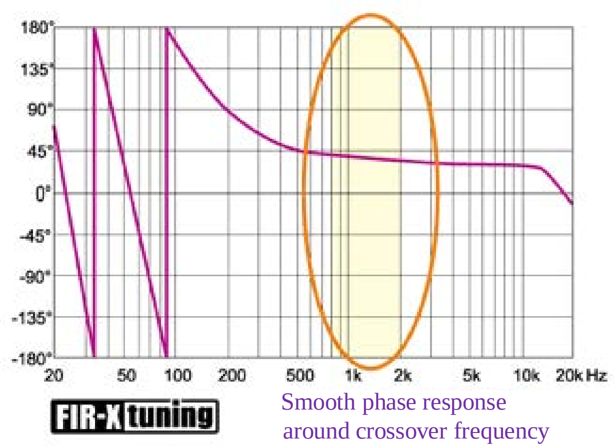

Fig.7 Phase response of DSR series FIR-X tuning

Each model has its own optimized tuning within the processor which helps to minimize latency.

6. References

"Digital Signal Processing," A. V. Oppenheim and R. W. Schafer, Prentice Hall

Yamaha Corporation

Pro Audio Division

10-1 Nakazawa-cho, Naka-ku, Hamamatsu City,

Shizuoka Prefecture, Japan

http://www.yamahaproaudio.com/