MIR a New Technology For Crossover Filters

A White Paper From SEED Electronic

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***Abstract*— The employment of digital crossover filters in professional loudspeaker is a matter of fact. However, it is important to underline that this topic has been intensively studied in the last years, leading to the development of different crossover structures based on traditional digital filters such as IIR and FIR. On this basis, taking into consideration the main advantages of the traditional approaches a brand new crossover filter structure is presented in this paper. One of the main key-point of the proposed structure is the possibility of achieving a linear phase response with a lower latency with respect to FIR filter. This feature is extremely important allowing to improve the audio quality of the professional loudspeaker without introducing a significant latency that could limit the employment of the system in some context such as live performance.**

Keywords—audio, digital signal processing, crossover, digital filters, linear phase

I. CROSSOVER FILTER THEORY

Professional speaker systems require two or more different drivers in order to cover the entire audible frequency range correctly and avoiding distortions or drivers damages. More in detail, large surfaces speakers are required in order to reproduce low frequency (a large volume of air has to be moved), while smaller surfaces speakers are used for mid and high frequency reproduction (small drivers can be more easily accelerated at the required speed). On this basis, it is necessary to split the audio signal into appropriate sub-bands, one for each driver. This operation involves a filter bank structure based on a crossover network typically composed of a low-pass, band-pass and high-pass filters. Classic crossover structures are depicted in Figure 1.

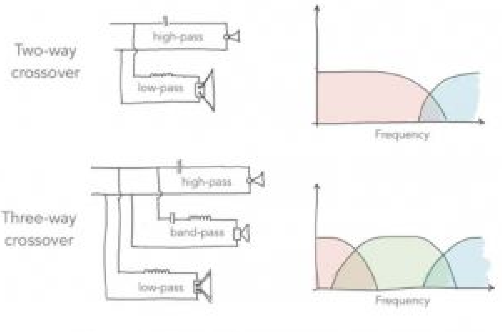

Figure 1 Classic Crossover structures.

The crossover filter is designed taking into consideration magnitude and phase response.

Magnitude Response:

It represents the gain of the system as a function of frequency. The overall system must maintain a level as more constant as possible with respect to the change of frequency (flat response). However, it is important to underline that some variation can be tolerated, especially if other desirable properties can be achieved.

Phase Response

It represents the phase shift of the system as a function of frequency. Group delay can be derived from phase response, it reports how the different frequencies contained in the crossover input signal are delayed. The overall system must maintain a linear phase (flat group delay). This means that all frequencies are delayed by the same value so keeping at the filter output the same phase alignment they had at the input, preserving therefore the signal «timbre». One of the most relevant and audible effect strictly connect to the phase response is the optimal reconstruction of the transients. This effect can be easily audible in impulsive sound such as percussive ones. In this kind of signal the time shifting of the frequency components introduces a loss of energy, dynamic and detail.

Different kinds of filter structures can be taken into account in the implementation of the crossover network: Infinite Impulse Response (IIR) and Finite Impulse Response (FIR).

IIR (Infinite Impulse Response)

The IIR filter output is a weighted sum of the most recent P inputs and Q outputs:

$$\begin{aligned} y[n] = \; & \frac{1}{a_0}(b_0 x[n] + b_1 x[n-1] + \cdots + b_P x[n-P] \\ & -a_1 y[n-1] + a_2 y[n-2] - \cdots - a_Q y[n-Q]) \end{aligned}$$

A more condensed form of the difference equation is:

$$y[n] = \frac{1}{a_0}\left(\sum_{i=0}^{P} b_i x[n-i] - \sum_{j=1}^{Q} a_i y[n-j]\right)$$

where:

- $x[n]$ is the input signal
- $y[n]$ is the output signal
- P is the feedforward filter order

- Q is the feedback filter order
- b_i are the feedforward filter coefficients
- a_i are the feedback filter coefficients

This filter class is typically employed in DSP applications since it is extremely efficient, it works faster and requires less memory space with respect to other types of filtering structures. Moreover, IIR filters provide a good matching with trusted analogue filters. The delay introduced (latency) by this kind of filter is negligible. On the other hand, no exact linear phase designs are possible, and the IIR is not guaranteed to be stable. Taking into account the IIR crossover filter implementation, Bessel and Linkwitz-Riley filters [1, 2] are currently used in today's applications due to the achievable high roll-off with a low computational complexity. However, it is important to underline that the phase response of such filters is not linear leading to a not flat group delay. On this basis various approaches have been presented in the literature [3] in order to achieve an approximately linear phase crossover using IIR filter structures.

FIR (Finite Impulse Response)

The FIR filter output is a weighted sum of the most recent inputs:

$$y[n] = b_0x[n] + b_1x[n-1] + \cdots + b_Nx[n-N]$$

A more condensed form of the difference equation is:

$$y[n] = \sum_{i=0}^{N} b_i x[n-i]$$

where:

- $x[n]$ is the input signal
- $y[n]$ is the output signal
- N is the filter order
- b_i is the filter coefficient (also known as tap).

With respect to IIR structure, the FIR filter class provides always stable filter (no feedback is present) with the possibility of achieving a linear phase. However, these advantages are counterbalanced by a higher computational cost and higher I/O audio latency. Moreover, it is important to underline that the performance of the FIR filter depends on the number of employed taps ($N+1$). More taps mean higher frequency resolution, opening to the possibility of achieving narrower filters and/or steeper roll-offs. Frequency resolution can be easily computed using the following equation:

$$Frequency\ Resolution = \frac{Sampling\ Frequency}{N}$$

Taking into example a low pass FIR filter created using a 48kHz sample rate and N = 1024 taps, the frequency resolution equals to 46.875 Hz. On this basis, it can be easily determined that the effective low frequency limit of this kind of filter has to be a couple of times higher than its frequency resolution (e.g. around 140Hz). So, short FIR filter, based on a small number of taps, is not very effective especially in low frequencies. However, long FIR filters, based on a large number of taps, become more powerful.

Finally, it is important to underline that although it is possible to design a linear phase FIR filter, this operation introduces a delay. A FIR filter is linear phase only if its coefficients are symmetrical around the center coefficients (i.e. the first coefficient is the same of the last, the second is the same of the next to last, etc… and the central one is the one having max value). So, the introduced delay can be easily computed using the following equation:

$$Delay = \frac{N}{2 * Sampling\ Frequency}$$

On this basis, taking into example a low pass FIR filter created using a 48kHz sample rate and N = 1024, the delay introduced is equal to 10 ms.

It is easy to notice that the use of linear phase FIR filter introduce a compromise between latency and resolution. An increase in resolution (which is good) leads to an increase in latency (which is bad). At the same time, a decrease in latency leads to a decrease in resolution. From a technical point of view, the value of latency and resolution as a function of sampling frequency are shown in Table 1 tacking into consideration the most common employed FIR filters.

TABLE I. RESOLUTION AND DELAY INTRODUCED BY A LINEAR PHASE FIR FILTER.

	Sampling Freq. 48kHz		*Sampling Freq. 96kHz*	
Taps Number	Resolution (Hz)	Delay (ms)	Resolution (Hz)	Delay (ms)
64	750	0.7	1500	0.35
128	375	1.3	750	0.7
256	188	2.7	375	1.3
512	94	5.3	188	2.7
1024	47	10.7	94	5.3
2048	23	21.3	47	10.7

Several applications have been proposed in the literature for the realisation of digital audio crossover using FIR filters [4, 5]. In most of these applications the high frequency channel is complementary of low frequency one and the filtering process can be performed using frequency domain methods such as overlap and save.

On the base of all the aforementioned considerations, it is important to reaffirm that the crossover frequency represent one of the main key point in the design of crossover networks. Indeed, especially for lower values, the performance of the audio crossover can be conditioned in terms of frequency response and audio latency.

II. CLASSIC APPROACH

In this section, the different types of crossover network typically employed in the professional audio equipment are analysed in detail. In particular the frequency responses of Butterworth, Bessel, and Linkwitz-Riley structures are analysed considering the magnitude and the phase behaviours.

Moreover, the square wave signal is employed in order to provide some information about the transient response. For each structure the magnitude frequency response is analysed for the different values of filter orders. Higher order value provides greater attenuation slope. Moreover a 4^{th} order crossover network with a crossover frequency of 1 kHz has been taken as case of study for a more detailed analysis of the aforementioned.

BUTTERWORTH

The Butterworth filter is a type of signal processing filter designed to have a frequency response as more flat as possible in the passband. The overall magnitude response of an audio crossover based on Butterworth structure with filter order from 1 to 8 is shown in Figure 2. While, a more detailed analysis of the behaviour of this circuit is depicted in figures from 3-6 referring to the 4^{th} order crossover filter we going to study. In particular, it is possible to notice a peak in the summed magnitude response at the crossover point (figure 3), and a nonlinear phase behaviour that lead to a non-constant group delay (figures 4 and 5).

Focusing on the transient response, it is possible to provide an insight of the system behaviour observing the square wave response. In terms of frequency response, a square wave is composed of the sums of a fundamental sine wave and a number of its odd harmonic at higher frequencies. So, if the harmonics are delayed with respect to the fundamental signal, the square wave reconstruction fails. However, in a linear phase system, all the frequency arrive at the same time, so the square wave reconstruction is guaranteed and a more defined transient is provided. In figure 6 the sum of a square wave input for the low-pass and high-pass branches is depicted.

It is important to underline that the change of order of the Butterworth filter not only produces a slope variation but also introduces other significant behaviours. Indeed, each filter introduce a phase shifting at the crossover point whose value depends on the order value. For example a first order low-pass filter introduces a phase shifting of -45 degrees, while for high-pass filter the phase shifting value is equals to +45 degrees. Increasing the order, the phase shift increases according to the law: *order * 45 degrees*.

On this basis, it can be easily demonstrated that a 4^{th} order crossover filter generate a -180 degrees shift on the low-pass output and a +180 degrees shift on the high-pass output. Differently, a 2^{nd} order crossover filter generate a -90 degree shift on the low-pass output and a +90 degrees shift on the high-pass output. In the first case an overall difference of 360 degrees is achieved keeping the driver in phase, differently, in the second case a phase difference of 180 degrees is produced between the drivers leading to a phase inversion problem. To cope with this issue it is necessary to apply a phase inversion to one of the driver.

A more complicated situation is presented during the employment of odd-order system. Indeed, it is not possible to apply a phase inversion to correct this problem. Finally it is important to underline that odd-order system are characterised by a not optimal polar response. In this system the main lobe does not results on-axis leading to a possible coloration of the perceived audio signal.

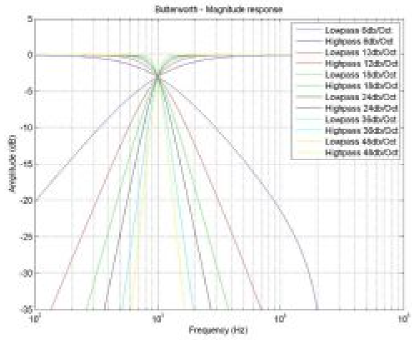

Figure 2 Butterworth Crossover - magnitude response

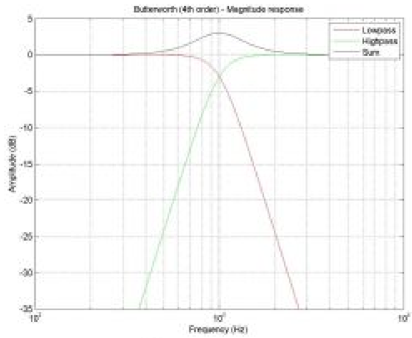

Figure 3 Butterworth 4^{th} Order Crossover - magnitude response

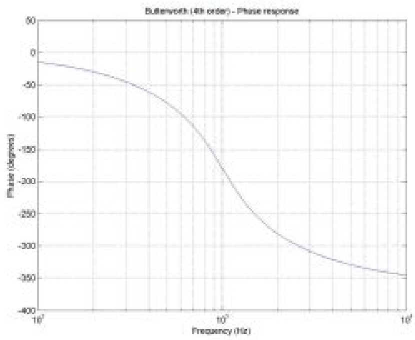

Figure 4 Butterworth 4^{th} Order Crossover - phase response

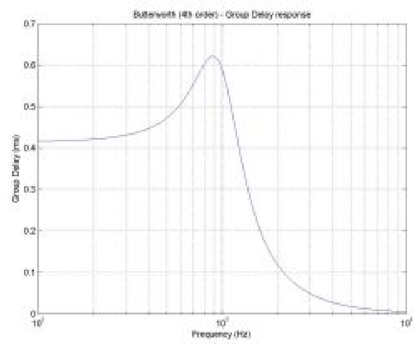

Figure 5 Butterworth 4^{th} Order Crossover – group delay

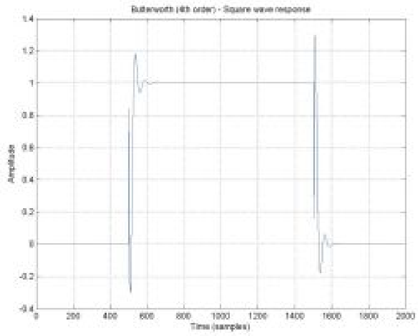

Figure 6 Butterworth 4th Order Crossover – Square wave response

LINKWITZ RILEY

The Linkwitz-Riley (L-R) filter is probably one the most used IIR structures employed in modern audio crossover. It is also known as “Butterworth double filter” due to its design technique. In greater detail, with the aim of providing a flat amplitude response, the Linkwitz-Riley structure it is based on the cascade of two Butterworth filters. Indeed, each Butterworth filter has -3dB gain at the cutoff frequency, so the resulting L-R filter is characterised by a -6dB. On this basis, summing the low-pass and high-pass outputs the gain at the crossover frequency will be 0dB. For this reason, this crossover acts like an all-pass filter having a flat magnitude response with a smoothly changing phase response.

The overall magnitude response of an audio crossover based on Linkwitz-Riley structure with filter order 2, 4, and 8 is shown in Figure 7. While, a more detailed analysis of the behaviour of this circuit is depicted in figures from 8-10 with reference to a 4th order crossover filter as a case of study. In particular, it is possible to notice a flat magnitude response (figure 8), and a nonlinear phase behaviour that lead to a non-constant group delay (figures 9 and 10).

Focusing on the transient response, the square wave response is reported in figure 11.

Further detail about this crossover structure are provided by Linkwitz (RANE Corporation) in [1].

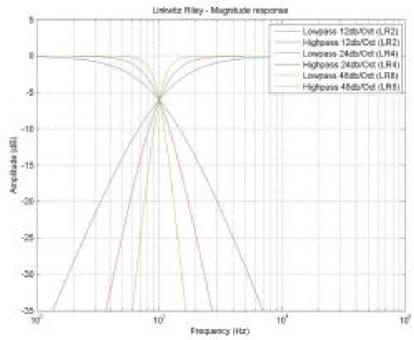

Figure 7 Linkwitz-Riley Crossover - magnitude response

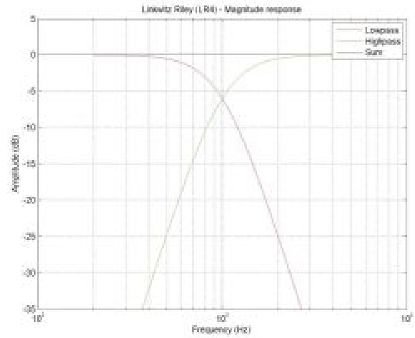

Figure 8 Linkwitz-Riley 4th Order Crossover - magnitude response

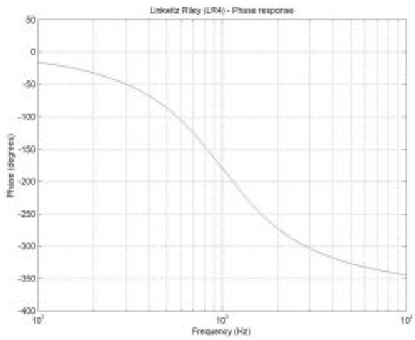

Figure 9 Linkwitz-Riley 4th Order Crossover - phase response

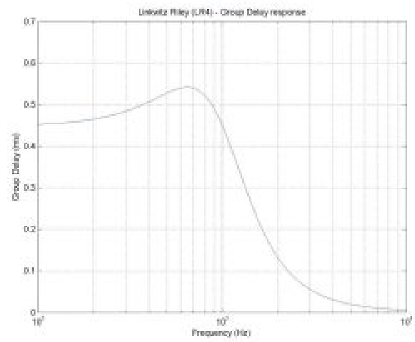

Figure 10 Linkwitz-Riley 4th Order Crossover – group delay

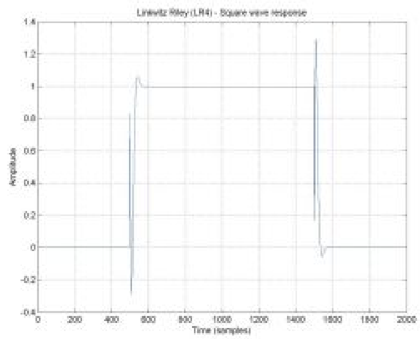

Figure 11 Linkwitz-Riley 4th Order Crossover – Square wave response

BESSEL

The Bessel is a type of signal processing filter designed to have a maximally flat/group delay for preserving the wave shape of filtered signals in the passband. For this reason Bessel filters are often used in audio crossover systems. On the other hand, one of the major drawbacks of the Bessel filter is the slow cutoff slope.

A detailed analysis of the behaviour of this circuit is depicted in figures from 12-15 with reference to a 4th order crossover filter as a case of study. In particular, it is possible to notice a dip in magnitude response (figure 12), and a nonlinear phase behaviour that leads to a non-constant group delay (figures 13 and 14).

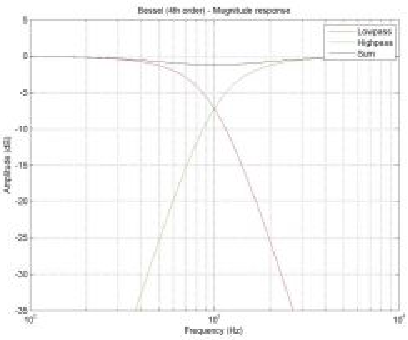

Figure 12 Bessel 4th Order Crossover - magnitude response

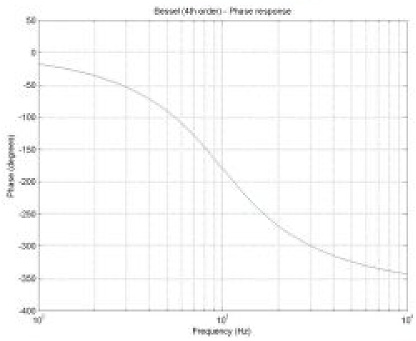

Figure 13 Bessel 4th Order Crossover - phase response

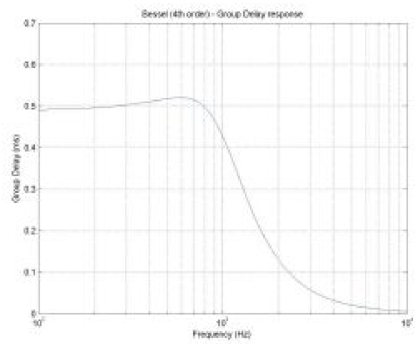

Figure 14 Bessel 4th Order Crossover – group delay

Focusing on the transient response, the square wave response is reported in figure 15.

Further detail about this crossover structure are provided by Miller (RANE Corporation) in [6].

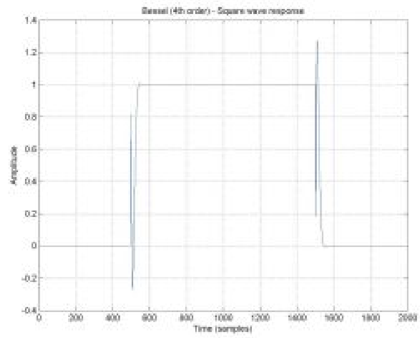

Figure 15 Bessel 4th Order Crossover – Square wave response

BRICK-WALL

The previously mentioned structures are characterised by a slightly smooth cutoff slope. With the aim of achieving a steeper cutoff slope, brick-wall filter can be employed.

For performing this task, FIR digital filters with a sufficient length can be employed. Different window types can be used: Rectangular, Kaiser, Hann, Hamming, Blackman.

Although, a steeper cutoff allows to reduce the crossover region, other problems could be arisen by the employment of Brick-Wall filter. For example filtering a signal with a low-pass Brick-Wall filter with cutoff frequency of 1kHz, means to remove the Fourier components above 1kHz. This truncation of the Fourier series gives rise to a ripple in the impulse response due to the Gibbs phenomenon [7].

III. MIR APPROACH

To cope with the general advantages and disadvantages of the previously mentioned structures: a brand new linear phase crossover approach has been developed by SEED Electronic. This new approach is called MIR.

The MIR technique joins the advantages of FIR and IIR filters. The founding element of this technology is an innovative algorithm that allows to determine the MIR parameters for modelling the shapes of the most common crossover introducing a linear phase response. With respect to standard FIR filter, this approach guarantees a lower computational cost and a lower I/O latency. On the other hand, unlike IIR filter, the MIR structure allows to generate linear phase filter (no group delay distortion is produced). By the way, it is important to underline that using this approach it is not possible to achieve a negligible latency especially for low value of the crossover frequencies.

A detailed analysis of the behaviour of this filter is depicted in figures from 16-19 where a MIR filter used for the modelling of a 4th order Linkwitz-Riley audio crossover has been considered as a case of study. In particular, in figure 16 it is possible to observe the same magnitude response of a LR24, while a linear phase behaviour with a constant group delay is depicted in figures 17 (with delay subtracted) and 18.

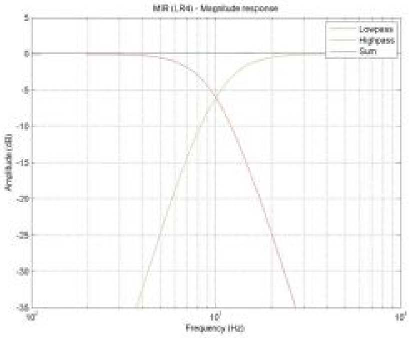

Figure 16 MIR 4th Order Crossover - magnitude response

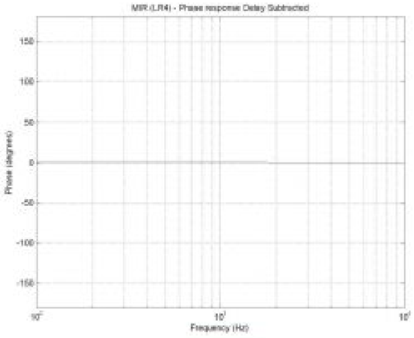

Figure 17 MIR 4th Order Crossover - phase response

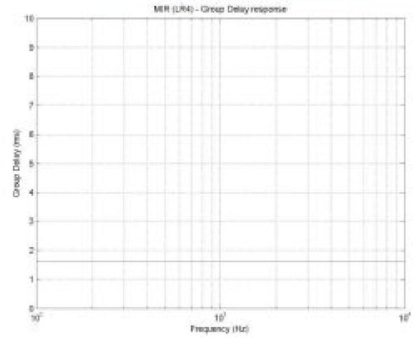

Figure 18 MIR 4th Order Crossover – group delay

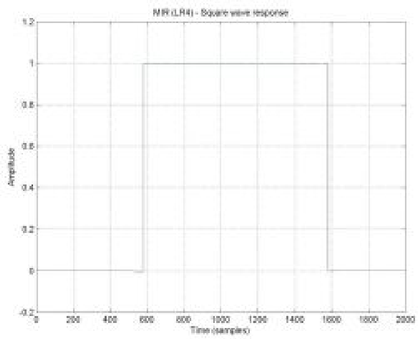

Figure 19 MIR 4th Order Crossover – Square wave response

Focusing on the transient response, the square wave response is reported in figure 19. It is possible to see that the linear phase allows to reproduce a perfect square wave guaranteeing an excellent transient response.

These results highlight the main advantages of the MIR structure with respect to IIR filters based crossover network. In the next section a deep analysis of the benefits of the MIR filter with respect to the FIR is reported.

Latency Analysis: a comparison between MIR and Linear Phase FIR

In order to provide an insight into the main advantages of the proposed structure, a comparison of the performance of MIR and linear phase FIR filters is reported in this section taking into consideration the audio latency delay problem.

Although similar results in terms of magnitude and phase response can be achieved using an opportunely tuned linear phase FIR filter, it is important to underline that the MIR approach allows to improve the performance in terms of audio latency reducing the delay introduced by the crossover structure.

More in detail, Figure 20 shows the plot of the latency generated by a FIR filter with respect to the MIR approach. First of all, it is easy to notice that for both the structures the delay tends to increase as the cutoff frequency decrease. This is a side effect linked to the compromise between frequency resolution and FIR filter length. The same effect can be easily noticed in the "step" behaviour of the FIR latency function. On the other hand, it is clearly visible the improvement in terms of audio delay achieved using a MIR structure. Indeed, the MIR filter always exhibits lower latency for all the cutoff frequency values and a smoother behaviour due to the employed structure.

Obtained the aforementioned results, the MIR approach has been further improved considering its employment in a live context. A value of audio latency greater than 5 milliseconds cannot be tolerated by musician during an audio musical performance. So, as reported in the data depicted in Figure 20, the MIR crossover could not be used in a live context for values of the cutoff frequency lower than 300Hz. To cope with this problem, the algorithm used to determine the MIR parameters has been tuned in order to always guarantee a delay of less than about 5 milliseconds as shown in Figure 21. This procedure, allows to maintain a low delay with a little compromise in terms of phase response. Indeed, although the phase response is kept linear for values higher than the cutoff frequency, this constraint is not respected for lower values where the delay group is not constant anymore. This effect can be observed in Figure 22, where the group delay of MIR low-pass filter with cutoff frequency of 100Hz is reported.

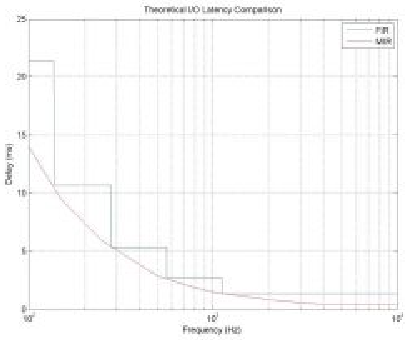

Figure 20 Theoretical IO latency comparison between MIR and Liner Phase FIR filter.

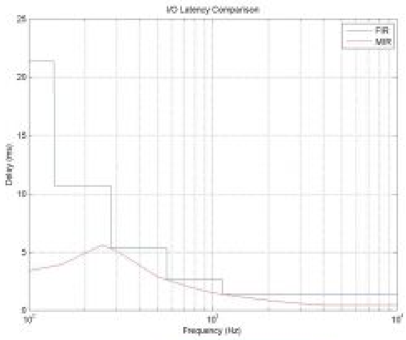

Figure 21 Real IO Latency comparison between MIR and Liner Phase FIR filter.

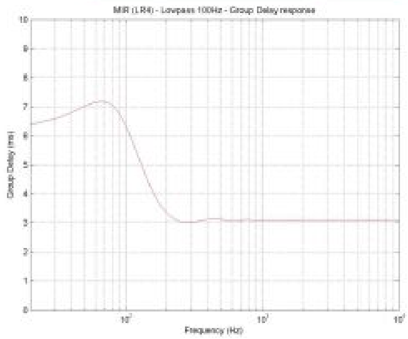

Figure 22 MIR 4th Order Low-pass Filter with 100Hz cutoff frequency – group delay.

CONCLUSION

A new approach for achieving linear phase crossover filter with a low audio latency has been presented in this work with the aim of improving the classic structure based on IIR and FIR filters. Crossover networks based on IIR filters are flexible in terms of type (slope shape) and order, are pretty easy to use, but linear phase is not guaranteed. On the other hand the employment of FIR filters allow to meet a linear phase constraint, but introduce a remarkable audio latency especially for low value of the cutoff frequency.

Taking into consideration the main advantages of FIR and IIR filters a brand new approach called MIR is presented in this paper. The founding element of this technology is an innovative algorithm that allows to determine the MIR parameters for modelling the shapes of the most common crossover introducing a linear phase response with a low audio latency.

Finally, several results have presented in order to show the effectiveness of the proposed approach, providing comparisons with the existing state of the art techniques in terms of objective measure related to the magnitude and phase responses.

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