A new approach on linearity for RCF live loudspeakers

In this whitepaper, we will discuss the importance of excluding phase distortions in the sound reinforcement systems and how RCF made it possible with FiRPHASE processing, reaching near-sub frequency phase linearity without annoying time delays.

A brief history of phase perception

The Georg Ohm's acoustic law (1843) states that a musical sound is perceived by the ear as a set of a number of constituent pure harmonic tones. Later, Von Helmholtz agreed to the Ohm's law and deepened his future researches saying that aural perception depends only on the amplitude spectrum of a sound and is independent of the phase angles of the various complements contained in the spectrum.

In 1841, August Seebeck, a scientist from Dresden University, was arguing with his experiments that the phase differences are clearly audible. The missing fundamental effect explain how a perceived pitch of a sound can be altered by phase differences of harmonics. The debate continued for twenty years in the scientific journal Annalen der Physik und Chemie, Seebeck then died young and his knowledge was forgotten. Until 1959, when Schroeder, in his work entitled New results concerning monaural phase sensitivity (1959), demonstrated the phenomenon. Schroeder states that Ohm's conclusion is invalid and it's only true in some particular cases. He postulates then the Schroeder's phase masking effect: by just modifying the individual phase components of two signals of identical envelopes, it is possible to produce strong varying pitch perception, e.g., when playing melodies.

Today we have several demonstrations who oppose the earlier belief that the human ear is phase-deaf, as the work of Lipshitz et al. in the Journal of Audio Engineering Society in 1982: "We have found that midrange phase distortion can be heard not only on simple combinations of sinusoids, but also on many common acoustical

signals." He pointed out that those problems exist but can be subtle and transducer designers can make an intelligent decision on the significance (not the existence) of phase effects.

In another late AES conference in 1996, Johansen & Rubak stated that "the conclusion must be: we cannot allow the excess phase to be neglected, and we will have to get around the equalization task in another way."

The perception of the phase spectrum has also been studied in relation to many topics, such as concert hall acoustics, pitch perception, vowel identification, masking, speech processing, and binaural rendering.
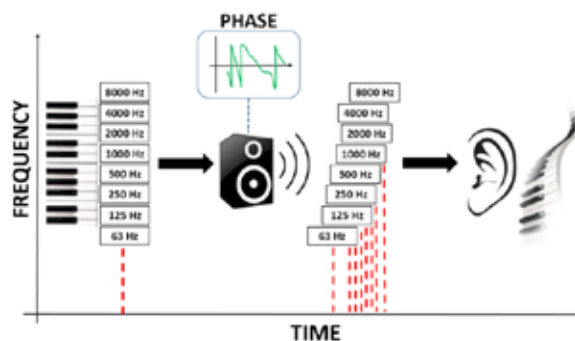

Fig.1 – Visual example of the monaural phase distortions generated by a sound system that can be made of transducers, EQ, crossovers, and amplifiers

0° Linear Phase

Linear-Phase or constant group delay describe a characteristic of linear systems where all the spectral components of a signal travel through the system at the same speed. In a particular case, a linear phase system can be called 0°-phase: all spectral components of a signal arrive at the output at the same time.

In a system with linear frequency response and 0° phase, the shape of the output signal is ideally an exact replica of the input signal, where

the magnitude depends only on the gain of the system.

Multi-way speaker cabinets and traditional IIR filter-based analog or digital crossovers are typical examples of non-linear phase systems with some amount of 'time smearing' due to the all pass nature of the summed electric or acoustic response.

The goal of a loudspeaker designer is to deliver a "transparent" sound, where the loudspeaker is able to reproduce a sound most as possible close to the original, an important characteristic for voice-based applications. Any sound characterization such as equalizations or distortions should be made by, e.g., the musician and sound engineer hands, giving them the freedom to present their own sound to the public. In classical music applications, the sound can be perfectly transduced without alterations.
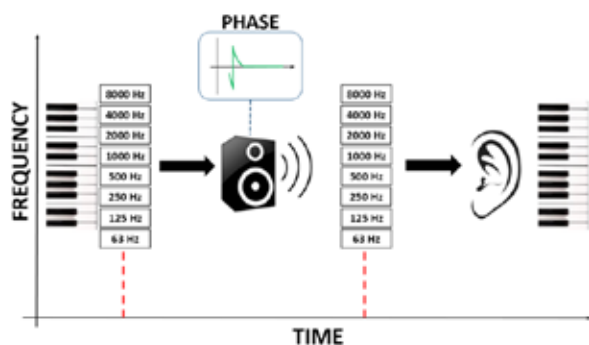

Fig.2 – Visual example of an ideal linear system

The square wave problem

A 0°-phase loudspeaker delivers to the listener all the frequencies at the same time, without relative delays, with the result of a true reconstruction of the original sound. One of the most relevant and audible effects in the passage between "not-0°-phase" and "0°-phase" is the optimal reconstruction of the transients. Let us think to a snare, or a picked guitar string: a lot of energy and frequencies in a very small amount of time. If the frequencies of the kick or the pick arrive at the ear not packed but a bit distributed in time, the impulse loses energy, dynamic, detail. This could be understood by using a squared wave that is the sum of a fundamental sin wave and a number of its odd harmonic at higher

frequencies. If the harmonics are delayed respect to the fundamental, the reconstruction fails.

	Original squared wave signal at the loudspeaker input.	
(a)	Squared wave signal reconstructed by the loudspeaker with harmonics out of phase.	
(b)	Squared wave signal reconstructed by the loudspeaker with harmonics in phase.	

Table.1 – Visual example of a square wave reconstruction from (a) a phase-distorted system and (b) a phase-coherent system

The loudspeaker is not only made of transducers but crossover and equalization filters act a fundamental role in the final result. Analog filters or digital IIR filters produce phase distortions around the frequency on which they act adding them to the ones already present in the transducers.

FIR filters for phase linearization

The modern DSPs permit a pre-compensation of these phase distortions in order to deliver a 0°-phase signal. The most useful and powerful way is the use of FIR filters (Finite Impulse Response filters). A FIR filter is nothing but a set of coefficients, representable as an impulse response (IR) in the time domain. The digital audio signal is filtered, hence modified, with the FIR by mean a mathematic operation called "convolution".

This kind of filters introduces a delay, the time necessary to the signal for passing all the length of the filter. Luckily, the time delay is equal for all the frequencies (no relatives delays between frequencies): in this particular case, they are named linear phase. A linear phase FIR filter can manipulate the amplitude equalization of a signal without distorting its phase, it can act as a bank

of IIR filters without their side effects on signal phase. For example, FIR filters can be used for a crossover filter instead of common Low Pass and High Pass IIR filters, reaching very high slopes without modifies on the phase.

Delay issues of FIR filters

Unfortunately, all that glitters is not gold: there is a cost to pay also for the use of FIR filters. The lowest frequency controlled by the filter (its resolution) is proportional to the length of the filter in terms of samples and hence to the latency that introduce in the DSP chain. As shown in the Table.2, the minimum length of a filter useful for managing all the audible frequencies introduces a delay of 21 ms (at 48 kHz of sampling frequency), delay not acceptable for live performances. The use of this kind of filters becomes a compromise between resolution and latency.

Considering the price in terms of latency, FIR filters can be hence used for correcting a large part of phase deviations from 0° creating a sort of Dirac delta (all pass filter): an impulse that doesn't affect the amplitude spectrum of the signal but modifies the phase in order to temporally align the frequency components of the sound.

	Sampling freq. 48 kHz		Sampling freq. 96 kHz	
Number of taps	Resolution (Hz)	Delay (ms)	Resolution (Hz)	Delay (ms)
32	1500	0.33	3000	0.17
256	188	2.7	375	1.3
1024	47	11	94	5.3
2048	23	21	47	11
4096	12	43	23	21

Table.2 – Delay introduced by FIR filters
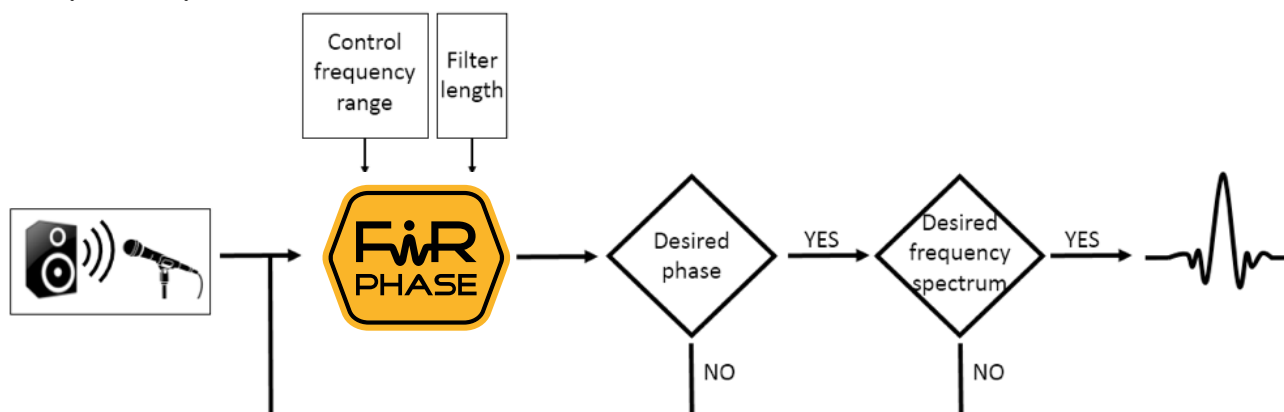
The temporal alignment of the frequency components is clearly visible in terms of Impulse Response measurements. The phase alignment increases the dynamic of the signal reproduced by the loudspeaker, because the energy is concentrated around the same time and not distributed as in the case of absence of FIR filter. The design of the FIR filter for this specific purpose should start from an accurate measurement of the loudspeaker phase.

Fig.3 – sum of the FIR filter phase with the loudspeaker phase

FiRPHASE

RCF FiRPHASE processing uses this measurement and try to invert the loudspeaker's phase without touching the amplitude equalization. The heart of the advanced technique used by FiRPHASE is a recursive method (Least Squares method) combined with a proprietary RCF algorithm that calculates the best FIR filter coefficients set in according to amplitude and phase constrains. The algorithm corrects phase and amplitude (if necessary) by taking into account the weak points of the transducers and the resonances or cancellations due to the cabinet of the loudspeaker. This technique permits to the designers a deep control of phase at mid-low frequency with relatively small filters, reaching a higher resolution than that one which theory suggests.
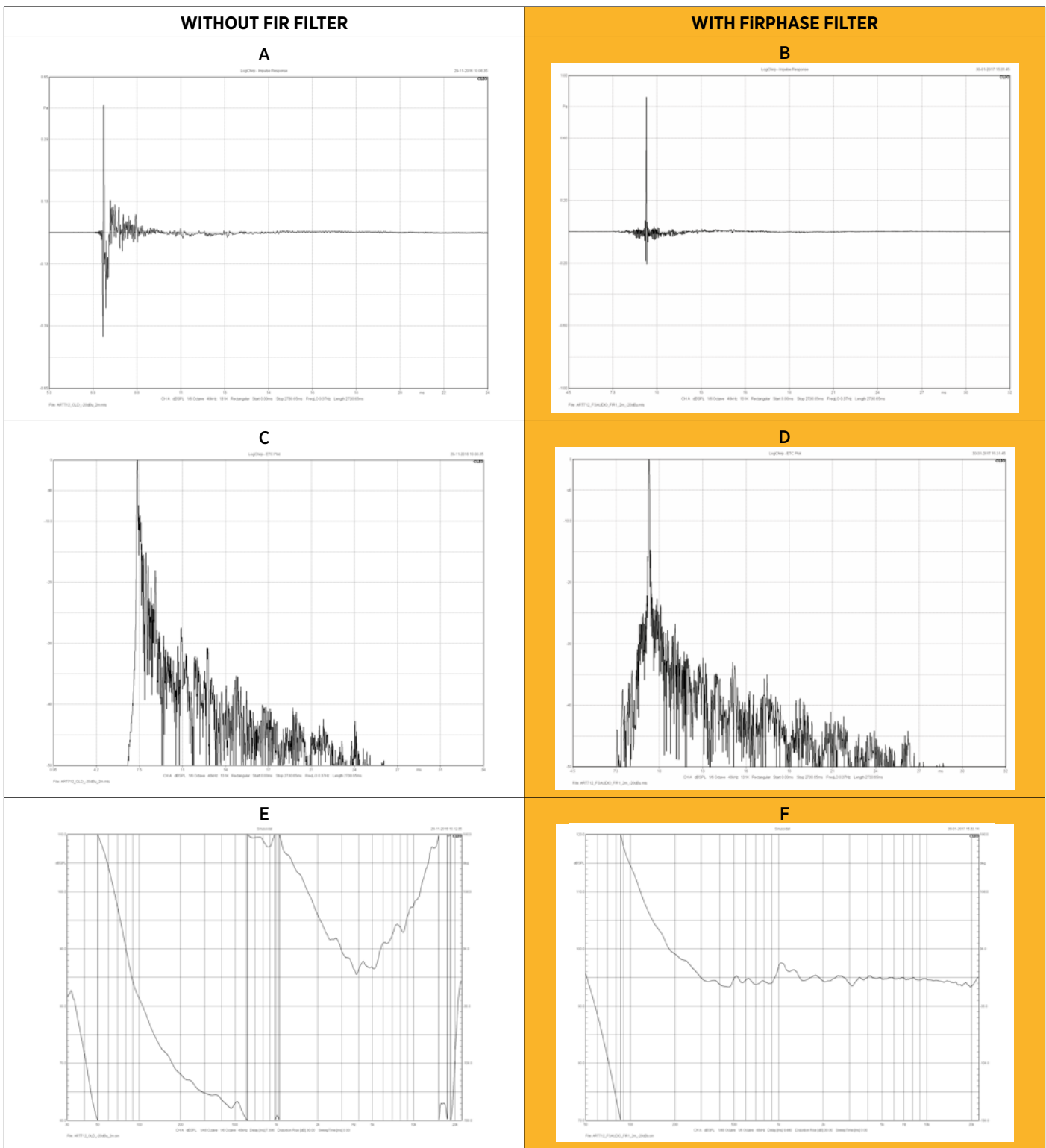
WITHOUT FIR FILTER	WITH FiRPHASE FILTER
A	B
C	D
E	F

Fig.4   A – Impulse Response, no FIR; B – IR after FIR filtering; C – Energy Time Curve plot, no FIR; D – ETC plot after FIR filtering; ,E – Phase plot, no FIR; F – Phase plot after FIR filtering.

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