

Crossovers, a step further !

What about the pulse response of your audio system?

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ETF2004

What is the goal of stereophony and multichannels reproduction?

- to mimic the live event
- to recreate the ambience of the recording session
- to reproduce the distance between instruments
- to reproduce the instruments with correct size and shape

Musical instruments

- They are not point sources
- Sound emission from different zones
- Different emitting zones for different frequency intervals

perception of stereophony

soundstaging:

characterized by

width

"wider than the space between the 2 loudspeakers"

depth

imaging:

according to Gordon Holt it refers to the ability of a given system to create phantom sources of the instrument at other places than the loudspeakers

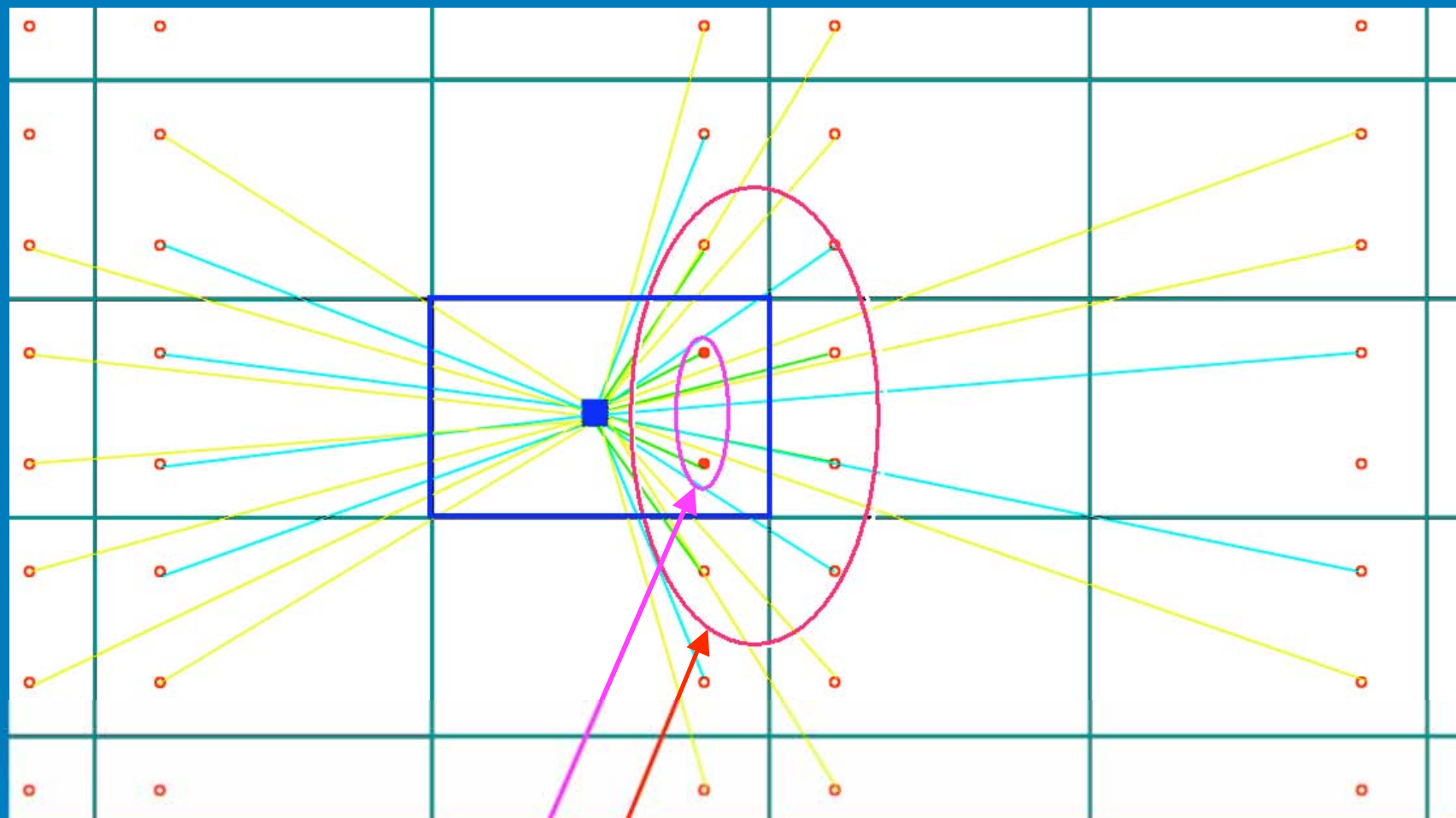
- illusion of distance between different instruments
- air between the instruments
- 3D image:

holographic image

reverberation and soundsaging

Results of different tests concluded that a fairly high ratio of reverberated energy over directly propagated energy was preferred by a majority of listeners.

It is commonly accepted that a large part of the soundstaging of a system is due to the reverberated field. Specially, the mirror images of a low directivity loudspeaker in a reverberating room is known to enlarge the width and the depth of the soundstage.



theoretical soundstage

perceived soundstage

widening and deepening of
the soundstage from low
directivity loudspeakers in a
reverberant room

Even if the widening and deepening of the soundstage due to reverberation is very pleasant in giving an illusion of the ambience of a live session we have to keep in mind that this detrimental to the fidelity of the recording.

A comparison performed with the same record on excellent headphones like Stax or Sony (MDR CD3000) is very enlightning for that purpose....

generally here most people think that they don't like to listen with headphones... ;-)

A different animal is the goal to obtain 3D imaging within the limits of stereophony....

Most records are recorded in amplitude stereophony.

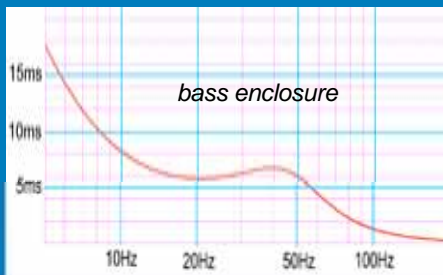
The position of a given instrument is defined by using a pan pot. (=> theoretically the position of the instrument is mostly at some point between the 2 control monitors). We rely on note amplitude, spectrum and decays to establish some distance perception.

Phase stereophony at recording requires

- Dummy head
- 2 microphones recording
- Matrixing

Phase/Time distortion origine

- **preamplifiers, amplifiers** (we will consider them as perfect)
- **crossovers** (mostly the subject of that lecture)
- **distorsion in loudspeakers and enclosures**



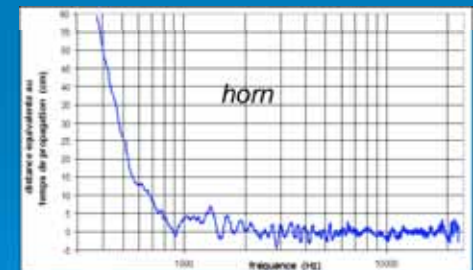
time delay curve of bass-reflex, closed enclosures and horns

time alignment of loudspeakers

- **listening room**

echos

reverberation



phase and time delay

2 aspects of a signal

time domain
 t

\Rightarrow

frequency domain
 f

$u(t)$

\Rightarrow

$\text{Re}(f), \text{Im}(f)$

\Rightarrow

$A(f), \varphi(f)$

real part

imaginary part

amplitude

phase

time delay: 3 definitions

- phase delay
- group delay
- differential delay (Marshall Leach)

we learn at college that a sine curve is defined like this:

$$V(t) = A \cdot \sin(\omega t + \phi) = V(t) = A \cdot \sin(\omega [t + \tau])$$

$$\phi = \omega \tau$$

$$\tau_p = \phi / \omega \quad \text{this is called "phase delay"}$$

$$\tau_g = d\phi / d\omega \quad \text{this is called "group delay"}$$

please notice that if phase varies linearly with frequency:

$$\phi = a \cdot f \quad \text{then} \quad \tau_p = \tau_g$$

and no phase distortion or delay time distortion exists

differential time delay

Leach argues that "what matters is the relative delay between the signal waveform and the envelope"

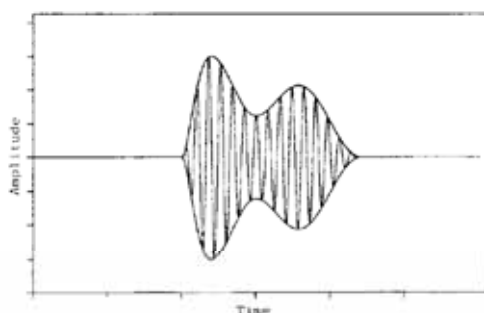


Fig. 1. Example narrow-band signal component modeled as sinusoidal signal with a amplitude envelope variation modulated on it.

J. Audio Eng. Soc., Vol. 37, No. 9, 1989 September

The transfer function of a first-order or second-order low-pass filter can be written in the form

$$F(s) = K \frac{1}{1 + a_1(s/\omega_0) + a_2(s/\omega_0)^2} \quad (7)$$

where K is the gain constant, s is the complex frequency, and $\omega_0 = 2\pi f_0$ is a normalization frequency. For a first-order transfer function, $a_1 = 1$ and $a_2 = 0$. For a second-order transfer function, $a_1 > 0$ and $a_2 = 1$.

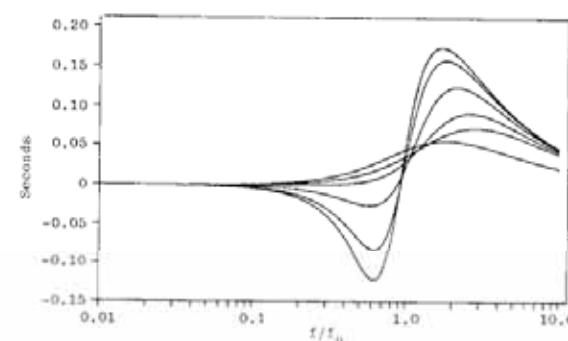


Fig. 6. Differential time-delay distortion $\Delta\tau$ in seconds versus normalized frequency f/f_0 for example low-pass filter transfer functions. Upper curve for $0.1 \leq f/f_0 \leq 1$ is first order, lower curves are second order in order of decreasing a_1 .

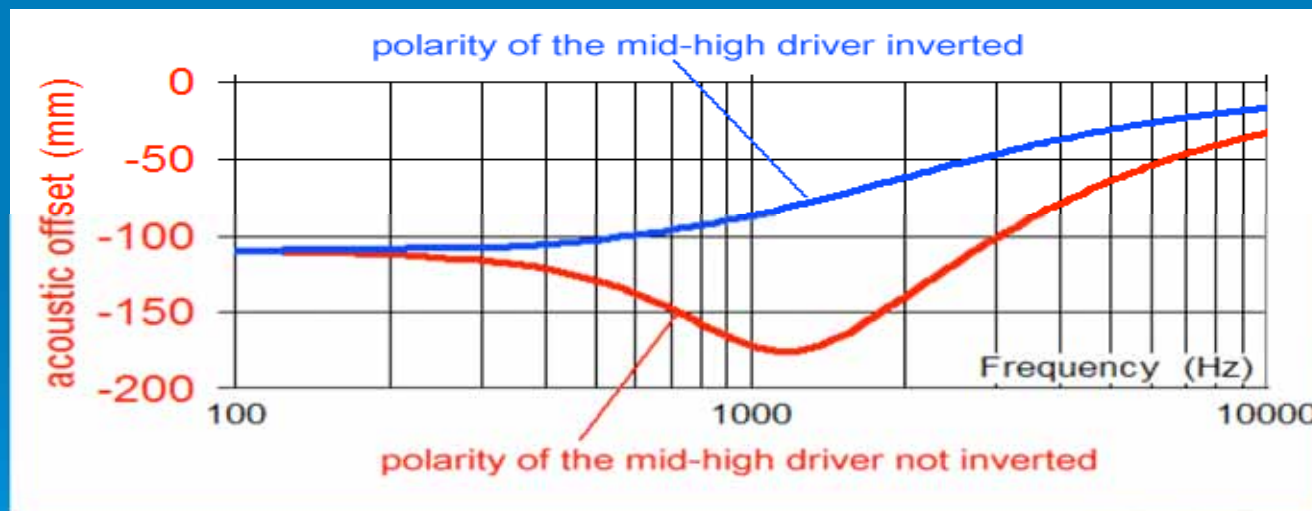
ref: "The Differential Time-Delay Distortion and Differential Phase-Shift Distortion as Measures of Phase Linearity" by Marshall Leach, JAES, Vol. 37, No.9, September 1989

➤ in a multiways system inverting the polarity of one way is equivalent to the addition of 180° to the phase of all the frequency components in the interval of frequency of the given way

➤ the equivalent time delay due to the phase inversion is $\tau = \frac{\Delta\phi}{\Delta\omega} = \frac{\pi}{2\pi f}$.

This means that inverting the polarity of one loudspeaker is equivalent to add a linearly frequency dependant delay to that loudspeaker.

- the inversion of the polarity of one loudspeakers versus the polarity of its lower and higher frequency loudspeaker is very interesting and an effective way to reduce some variation of the time delay curve of a multiways system, specially in the medium range.
- The commonest use of that method is with the 3rd order Butterworth crossover.

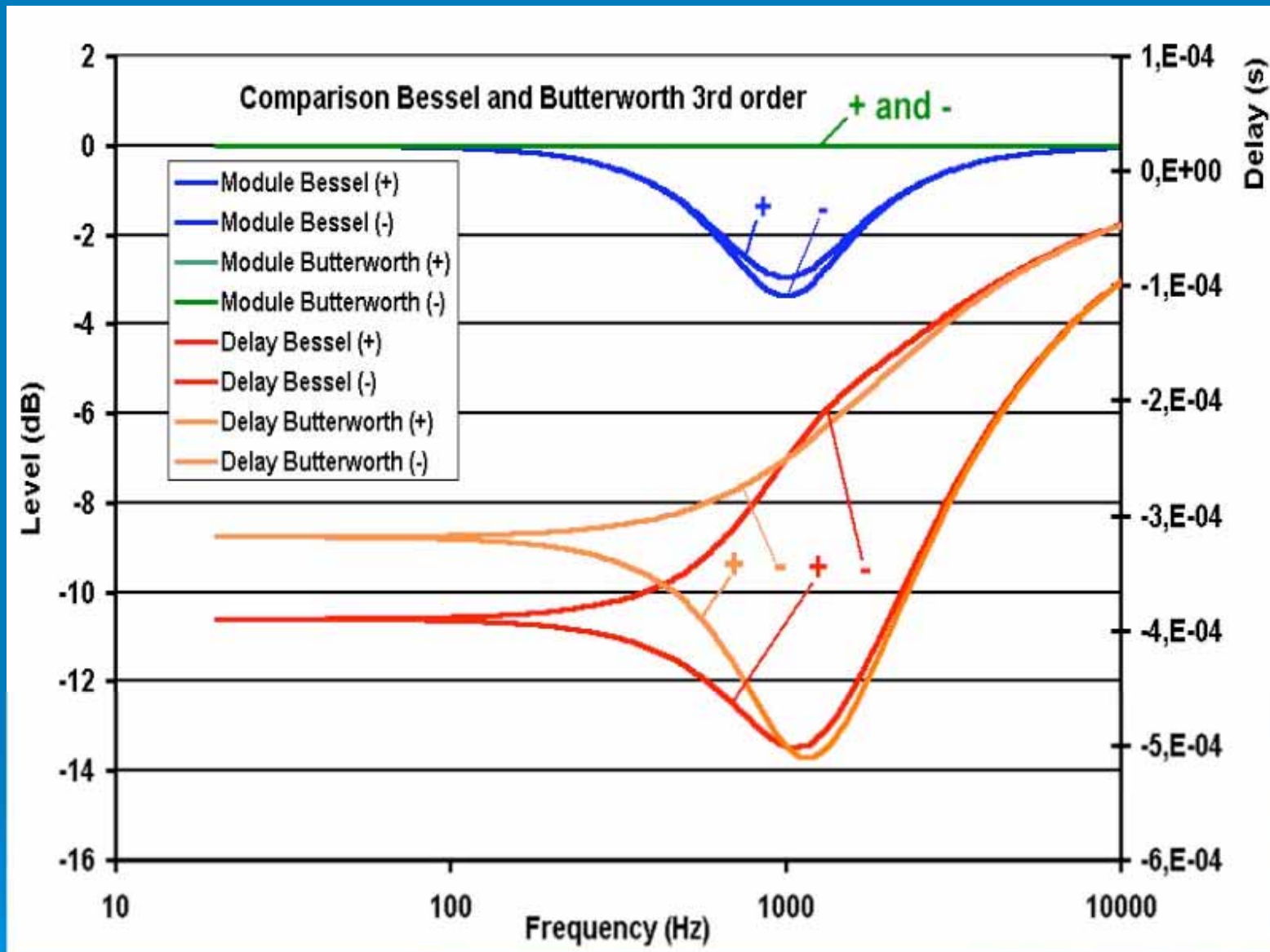


reduction of total time distortion under 4kHz by polarity inversion of the high-mid in a 2 ways system using the common 3rd order Butterworth crossover

- time delay can be expressed by its equivalent offset in centimeters according to the classical formula:

$$\text{offset} = 34400 \times \text{time delay}$$

on modern digital crossover an option is given to express the delay in ms or in mm



another example of reduction of the time-delay curve variation using an inversion of polarity

Phase distortion / time delay distortion are they audible

- B. B. Bauer, "Audibility of Phase Shift," *Wireless World*, (Apr. 1974).
- S. P. Lipshitz, M. Pocock, and J. Vanderkooy. "On the Audibility of Midrange Phase Distortion in Audio Systems," *J. Audio Eng. Soc.*, vol. 30, pp. 580-595 (Sep 1982).
- R. Lee. "Is Linear Phase Worthwhile," presented at the 68th Convention of the Audio Engineering Society, Hamburg, Mar 17-20, 1981, preprint no. 1732.
- H. Suzuki. S. Morita. and T. Shindo. "On the Perception of Phase Distortion," *J. Audio Eng. Soc.*, vol. 28, no. 9, pp. 570-574 (Sep 1980).
- many others

Blauert and Laws data on audibility thresholds for group delay. Below is a table that shows the data in terms of both delay time (in ms) and normalized delay in cycles.

Frequency (Hz)	Threshold (ms)	Threshold in T (periods or cycles)
8k Hz	2 ms	16 T
4k Hz	1.5 ms	6 T
2k Hz	1 ms	2 T
1k Hz	2 ms	2 T
500 Hz	3.2 ms	1.6 T

Some Experiments With Time.

On the audibility of phase shift.

article written by David L. Clark, manufacturer of the ABX Comparator, in 1981 for the Winter 1983 issue of the Syn-Aud-Con newsletter

- Three experiments were performed which confirm the audibility of time offset in loudspeaker drivers but indicate that this audibility is due only to the frequency response aberrations resulting from the time offset. Implications of these results are discussed.
- "Hearing a difference" means being able to identify 12 correct out of 16 tries.
- Given that the response alone *can* explain time offset audibility and time delay alone *cannot* explain this audibility it seems inescapable to conclude that arrival time compensation *by itself* has no audible value.

David L. Clark (1981)

Audibility and Musical Understanding of Phase Distortion

by Andrew Hon

Berkeley, fall 2002

Despite early beliefs (Ohm's Phase Law in the 1800s), studies have been conducted demonstrating that phase distortion is audible, however subtle and specific to certain circumstances. However, many people claim that previous research shows phase distortion is not audible. They simply not read the more current research.

Many loudspeaker designers are guilty in this regard. My assertion is: Depends - Yes, **phase distortion is audible under the right circumstances to certain people, but the consensus is it is rather subtle at best, especially in relation to other forms of distortion.** Phase is chaotic in reverberant environments, yes, but the direct sound is not affected by reverberation. In some situations such as choral music in a cathedral from the back of the audience, phase is totally messed up, but in most other cases it still matters!

Andrew Hon (2002)

The ABX Phase Distortion Challenge

Can I discern a 4th order Linkwitz-Riley filter that has 360 degrees of phase rotation between high and low pass?

Using the PCABX computer program:

http://www.pcabx.com/getting_started.htm

Results

10 correct out of 13 trials which is $p < 0.05$ (4.6 percent chance of guessing so correctly) significance level for "castanet" sample, discerning unaltered reference with a digitally processed 4th Linkwitz-Riley filter at 300 Hz and 3000 Hz (a common 3-way speaker configuration).

I CAN HEAR PHASE DISTORTION, at least in this "castanet" sample, on my system, in my familiar acoustical environment.

Difference is subtle but noticeable, I believe related to the phase distortion and not to any spurious in the sample presentation. Discerning requires a fair bit of concentration and rapid switching (repeated-music test not running-music).

Andrew Hon (2002)

Subjective Impression of Phase Distortion, discussion

4th order LR crossover have always sounded "disjointed" to me - transients sound blurred, and high frequencies don't match up with low frequencies.

Most noticeable with B&W audiophile speakers, which all use 4th order LR crossovers. The DM603 series is especially horrible sounding because the LR crossover is relatively low at 1-2kHz, whereas the DM303 series is not too bad because the LR crossover is at 4kHz, almost out of the midrange frequencies. 360 degrees of phase rotation is pretty horrible.

In the castanet sample, I listened for a subjective feel of the running notes. In LR filtered sample, the notes feel like they're stumbling over each other, while in the non-filtered sample, they are fast but liquid, flowing. 360 degrees of phase rotation at 10 kHz is 0.1 milliseconds, which seems inconsequential, but it means the source at 10 kHz would be positioned 1.356 inches closer to you, and smeared in physical location a couple inches over its full frequency spectrum. From the above diagrams, at 2 kHz, 180 degrees of phase rotation is 0.25 milliseconds.

Vijay Iyer describes how micro-timing in the single-digit milliseconds, controlled by musicians, can affect the emotional content of rhythmic music. Is it then surprising that sub-millisecond timing differences can be perceived? **Ill-defined audiophile terminology such as "PRAT", or "pace, rhythm, attack, timing" may be due to these sub-millisecond crossover delays.**

Steady-state timbre, congruent with previous research, does not sound much different between samples. The onset portion of the notes, however, is most important for timbre and one could say that **imperfect transient/step responses negatively effect the overall quality of timbre.**

Andrew Hon (2002)

one of the best work thesis about phase distortion audibility:

University of Miami

AURAL PHASE DISTORTION DETECTION

Presented by Daisuke Koya

University of Miami
May, 2000

Daisuke Koya (2000)

chapter 2. Audibility of Phase Distortion in Audio Signals

Although it was once believed that the human ear is "phase deaf," in accordance to Ohm's acoustical law, more **recent research has shown that relative phase has subtle effects on timbre,** in particular when changing phase relationships occur within a continuously sounding tone.

2.3 Previous Research

Although not in large numbers, previous research in investigation of the audibility of phase distortion has proven that it is an audible phenomenon.

Lipshitz *et al.* has shown that

on suitably chosen signals, even small midrange phase distortion can be clearly audible.

Mathes and Miller and Craig and Jeffress showed that

a simple two-component tone, consisting of a fundamental and second harmonic, changed in timbre as the phase of the second harmonic was varied relative to the fundamental.

The above experiment was replicated by Lipshitz *et al.*, with summed 200 and 400 Hz frequencies, presented double blind via loudspeakers resulting in a 100% accuracy score.

Daisuke Koya (2000)

An experiment involving polarity inversion of both loudspeaker channels resulted in an audibility confidence rating in excess of 99% with the two-component tone, although the effect was very subtle on music and speech.

Cabot *et al.* tested the audibility of phase shifts in two component octave complexes with fundamental and third-harmonic signals via headphones. The experiment demonstrated that **phase shifts of harmonic complexes were detectable.**

Daisuke Koya (2000)

Another very simple experiment conducted by Lipshitz *et al.* was to demonstrate that the inner ear responds asymmetrically.

Reversing the polarity of only one channel of a pair of headphones markedly produces an audible and oppressive effect on both monaural and stereophonic material. This effect predominantly affects frequency components below 1 kHz.

Because reversal of polarity does not introduce dispersive or time-delay effects into the signal, but merely reverses compressions into rarefactions and vice versa, these audible effects are due only to the constant 180° phase shift that polarity reversal brings about. Since interaural cross-correlations do not occur before the olivary complexes to which the acoustic nerve bundles connect, it must be concluded that what is changed is the acoustic nerve output from the cochlea due to polarity reversal. This change owes to two factors: cochlear response to the opposite polarity half of the waveform, and the waveform having a shifted time relationship relative to the signal heard by the other ear. **This reaffirms the half-wave rectifying nature of the inner ear.**

Daisuke Koya (2000)

"A frequent argument to justify why phase distortion is insignificant for material recorded and/or reproduced in a reverberant environment is that reflections cause gross, position sensitive phase distortion themselves. "

"Although this is true, it is also true that the first-arrival *direct* sound is not subject to these distortions, and directional and other analyses are determined during the first few milliseconds after its arrival, before the pre-dominant reverberation's arrival. Lipshitz *et al.* do not believe that the reverberation effects render phase linearity irrelevant, and there exists confirmatory evidence ."

Daisuke Koya (2000)

Lipshitz *et al.*'s research involved analog implementations of first- and second-order unity-gain all-pass networks ranging in frequency from roughly 100 Hz to 3 kHz, with frequency switchable in steps. The Q of the second-order networks was switchable in steps from $1/2$ to 2 .

Transducers used in the experiment were electrostatic Stax headphones and Quad loudspeakers, for their notable phase linearity.

Test material used and notable results include:

- low-frequency square waves of around 150 Hz - Displayed pitch or timbre changes as the all-pass networks were introduced into their chain for *all* Q and f_0 settings, with both first- and second-order networks. The effect was most audible at high levels, although it was detected down to about 60 dB SPL. **For the above test with Quad electrostatic loudspeakers, the effects were more audible for near-field auditioning.**

- very low repetition rate square waves (2 - 5 Hz) - Demonstrated audible phase effects which sound like the ringing of the all-pass network at f_0 . Low selected all-pass frequencies (113Hz - 529Hz) were the most audible, and detection became more difficult above 1kHz. The highest Q positions were the most audible.

Daisuke Koya (2000)

raised cosine modulated pulse - Although signal changes were audible on the second-order all-pass from 160 to 353 Hz at higher Q (> 1), it was not very audible at low Q values or with first-order networks.

assorted waveforms - The sawtooth waveform proved to be the most audible synthesized waveform. 440 Hz half-wave rectified sine wave with eight harmonics was audible on both first- and second-order networks from 113 to 1037 Hz.

pre-recorded music:

- male and female singing (with second-order all-pass with Q of 0.5 set to 160 and 240 Hz) - These were audible with a 95% confidence level.

- variety of live sounds (recorded anechoically) - male voices, handclaps, metal rods struck against each other, and blocks of plastic and wood clicked together, etc. The unpitched signals had a higher degree of audibility. Male voice changes were not detectable.

The conclusion drawn by Lipshitz was that midrange phase distortion can be heard not only on simple combinations of sinusoids, but also on many common acoustical signals. This audibility was far greater on headphones than on loudspeakers in a reverberant listening environment.

Daisuke Koya (2000)

Hansen and Madsen [13], [14] have conducted several experiments that have been completed regarding the audibility of phase distortion. A displaced sine wave (non-zero DC component) will have the time and frequency functions as shown in Fig. 2.11.

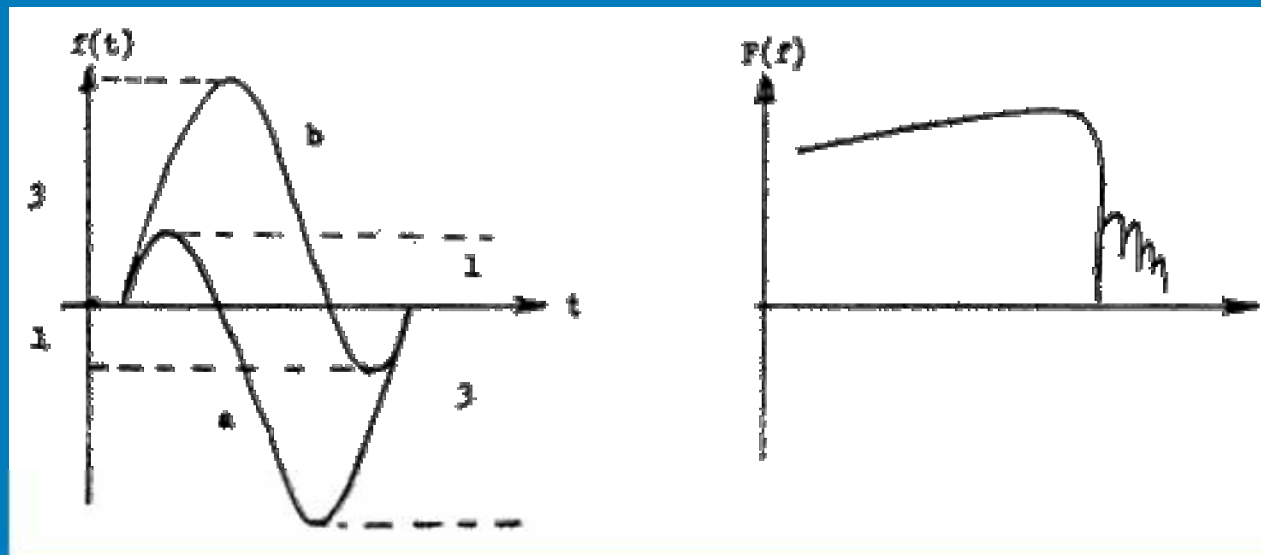


Fig. 2.11. Single-sine pulses with differing displacements (left) and their common spectral plot (right). [V. Hansen and E. R. Madsen, "On Aural Phase Detection," *J. Audio Eng. Soc.*, vol. 22, pp. 10-14 (1974 Jan./Feb.), pp. 12, Fig. 7.]

Frequency analysis demonstrated that there was no difference in the frequency spectrum. However, listening tests conducted with an electrostatic loudspeaker on both signals disclosed a clearly audible difference in timbre.

Daisuke Koya (2000)

Hansen and Madsen's second experiment [14] used a very narrow spectrum with three bars, as shown in Fig. 2.12 for a listening test.

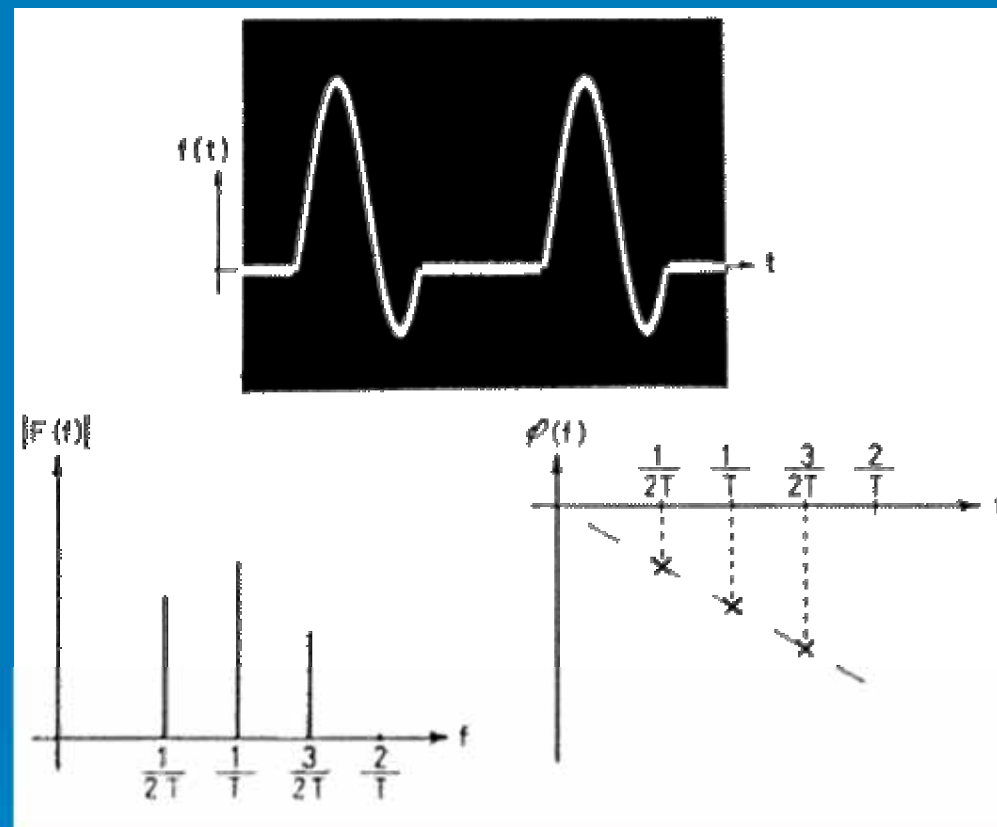


Fig. 2.12. Time function giving a three-bar spectrum. [V. Hansen and E. R. Madsen, "On Aural Phase Detection: Part II," *J. Audio Eng. Soc.*, vol. 22, pp. 783-788 (1974 Dec.), pp. 784, Fig. 3.]

A test was conducted with a Quad electrostatic loudspeaker in a standard living room. Average results for all listeners and the resulting plots of permissible phase distortion levels and phase deviations are shown in Figs. 2.15 and 2.16. The five curves on Fig. 2.15 represent the various quantities of phase difference ratio $h = A/B$ used as the plot parameter. The five curves on Fig. 2.16 represent the relative minimum sound pressure levels for just noticeable detection of phase change between the signals. As an interesting side note, it was found that **the tests revealed noticeably increased phase sensitivity with loudspeaker tests in reverberant environments as compared to headphone tests. This increased phase sensitivity may be due to reflections, or standing waves converted into amplitude shift present in the reverberant room.**

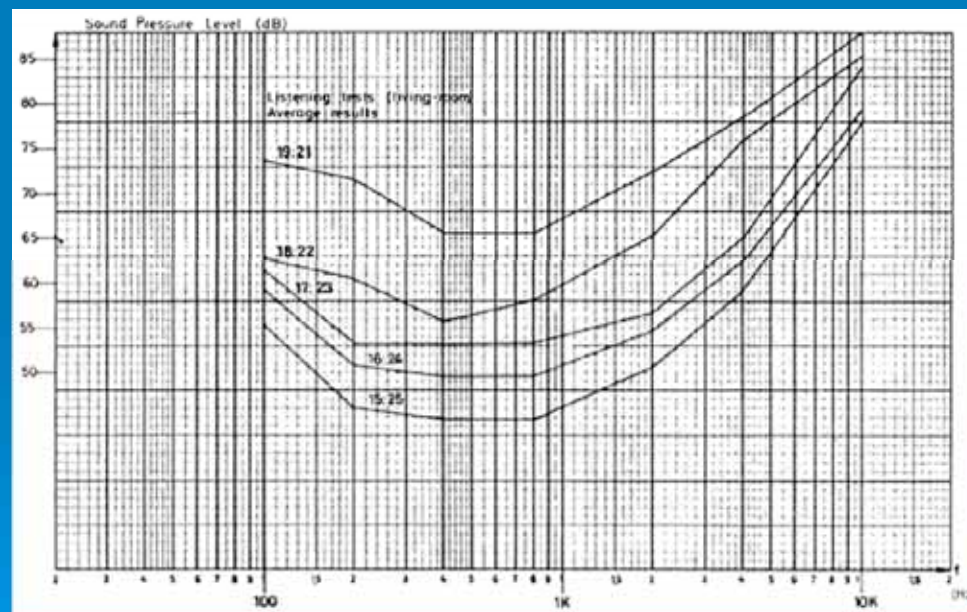


Fig. 2.15. Average results obtained from listening test in a reverberant environment with phase as a parameter. [V. Hansen and E. R. Madsen, "On Aural Phase Detection: Part II," *J. Audio Eng. Soc.*, vol. 22, pp. 783-788 (1974 Dec.), pp. 787, Fig. 7.]

Suzuki *et al.* conducted a phase distortion perception experiment with transient signals of short duration as shown in Fig. 2.17. The time interval T_0 for each signal was chosen so that $T_0 = 2/f_0$, where f_0 is the 90° phase shift frequency of an analog phase-lag type all-pass filter defined by
$$f_0 = \frac{1}{2\pi R_0 C_0}$$

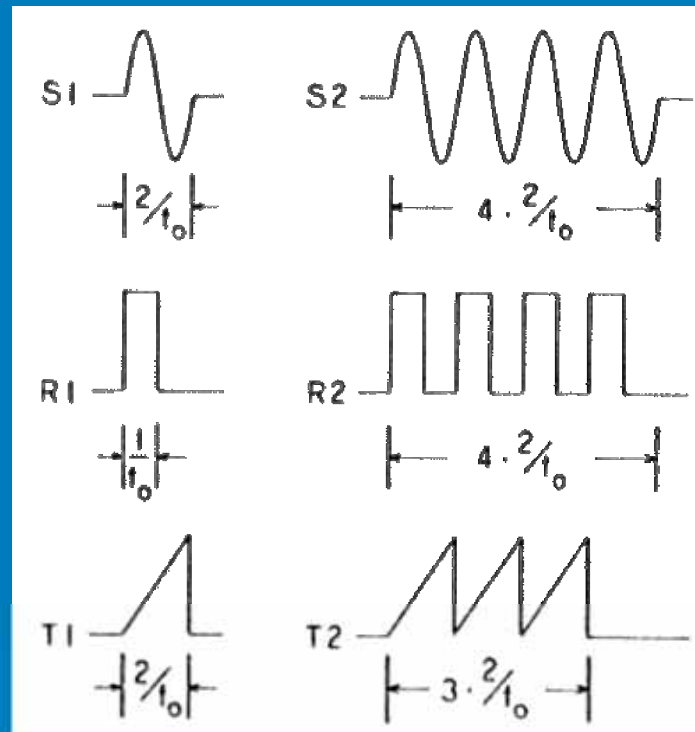


Fig. 2.17. Transient Signals used for hearing test. [H. Suzuki, S. Morita, and T. Shindo, "On the Perception of Phase Distortion," *J. Audio Eng. Soc.*, vol. 28, pp.570-574 (1980 Sep.), pp. 572, Fig. 4.]

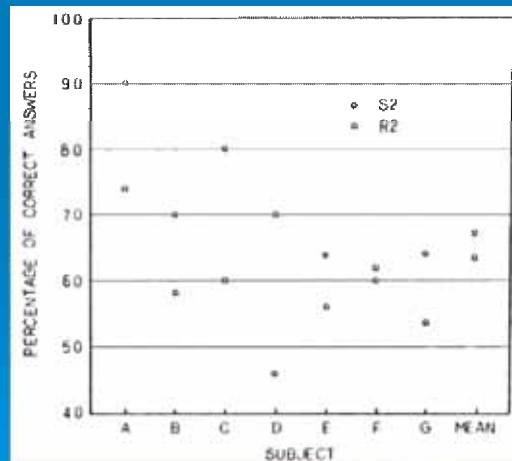
These transient signals were then phase shifted by a single-pole phase-lag type all-pass filter. Transient signal intervals were changed according to the frequencies of 300 Hz and 1 kHz, the value of f_0 explained previously.

Conclusions of Suzuki's work:

Certain people who participated in the test clearly heard the phase distortion present in the low frequencies of a single-pole all-pass filter when highly artificial signals were used.

In this sense, for high-fidelity reproduction, phase distortion is not permissible.

Another conclusion made by Suzuki *et al.* was that phase effects were highly individual and headphone listening showed much greater sensitivity than loudspeaker listening.



Percentages of correct answers of loudspeaker listening for S2 and R2 in an anechoic chamber, where $f_0 = 300\text{Hz}$. [H. Suzuki, S. Morita, and T. Shindo, "On the Perception of Phase Distortion," *J. Audio Eng. Soc.*, vol. 28, pp. 570-574 (1980 Sep.), pp. 573, Fig. 8.]

Fincham tested the effect of the reduction in group-delay distortion in the audio record/reproduction chain by means of a minimum phase-shift equalizer in carefully controlled conditions.

These effects were clearly heard but quite subtle. In another test, a 8 cycles of a 40-Hz tone burst was used which was cascaded with an all-pass filter with significant group delay around 40-50Hz. Loudspeakers were used.

Distinct audible differences in sound quality were observed by most of the lecture theater audience.

Preis *et al.* [17] conducted the audibility of phase distortion produced by minimum-phase 4-kHz and 15kHz anti-alias filters. In his experiment, group-delay distortion was doubled progressively until 67% mean correct discrimination was attained. Fig. 2.20 shows the mean correct discrimination between phase-distorted (minimum-phase) and undistorted (linear-phase) test signals for three low-pass systems (4-kHz elliptic and Butterworth, 15-kHz elliptic).

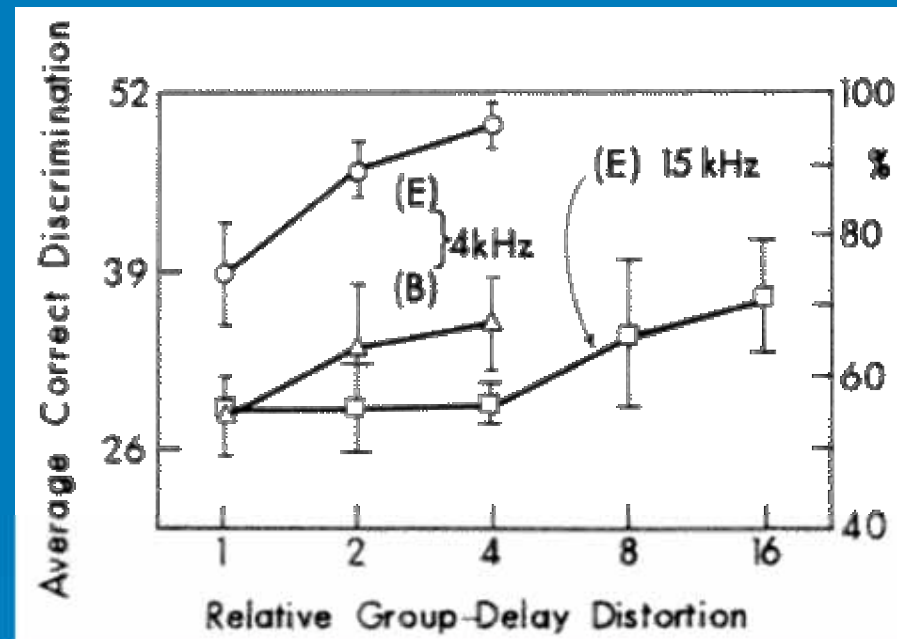


Fig. 2.20. Average correct discrimination between signals with no group-delay distortion and progressively doubled group-delay distortion. 52 presentations per subject of each of 11 test-signal pairs. 5 subjects; E - elliptic; B - Butterworth. [D. Preis and P. J. Bloom, "Perception of Phase Distortion in Anti-Alias Filters," *J. Audio Eng. Soc.*, vol. 32, pp. 842-848 (1984 Nov.), pp. 844, Fig. 2.]

It was concluded that for the impulsive test signals used and diotic (same signal in both ears) presentation via headphones, **the ear is significantly more sensitive in the middle of the audio band (4 kHz) than at the upper edge of the band (15kHz) to group-delay distortion.**

Phase and delay time distortion of crossovers

Crossovers

➤ The common view:

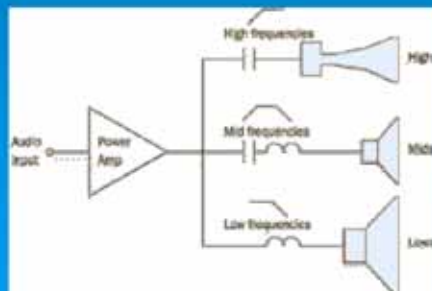
"Phase variation due to crossovers is not audible"

"Phase variation is not that important"

"All is known about crossovers + (crossovers is boring stuff)"

"We cannot ameliorate crossovers"

"That audioreview X said my loudspeaker has a good pulse response..."



the quest for a perfect crossover

the most used classical filters used today are:

Butterworth and Linkwitz-Riley

Stephen Butterworth described the response of the Butterworth filters in 1930.

Siegfried Linkwitz and Russ Riley were R&D engineers at Hewlett-Packard

S. H. Linkwitz "Active Crossover Networks for Non-coincident Drivers," J. Audio Eng. Soc., vol. 24, pp. 2-8 (Jan/Feb 1976).

we should add Bessel filter, Cauer,...

Crossovers in the audiopro world:

"Up until the mid `80s, the 3rd-order (18 dB/octave) Butterworth design dominated, but still had some problems."

"Since then, the development (pioneered by Rane and Sundholm) of the 4th-order (24 dB/octave) Linkwitz-Riley design solved these problems, and today is the norm."

from: <http://www.rane.com/pdf/linriley.pdf>

Effect of misalignment of a loudspeakers system

➤ geometrical alignment

- equivalent sources of the loudspeakers are aligned on a sphere at the same distance of the listener ears.

➤ time alignment

("time aligned" is registered)

- alignment of the arrival of pulses coming from the different loudspeakers (filtered by the crossover).
- few persons align the top of the first peak of the pulses, others align the inversion point of the left slope of the main peak.

Effects of a misalignment (1)

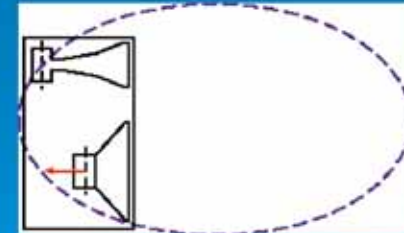
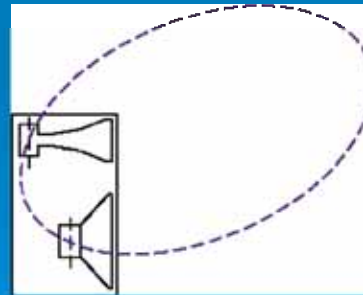
- the effects of a misalignment are predictable and add to the phase distortion (or delay time distortion) due to the crossovers.
- Misalignment analysis cannot be separate from crossover analysis. The 2 must be studied at the same time

Remark: that's why I design a software then a spreadsheet to predict those effects

Effects of a misalignment (2)

➤ effects on timbres

➤ effects on lobing (not so important in home listening excepted for the effects on the reverberated field)



➤ effects on imaging

analysis, measurement of the effects of delay time distortion

➤ my humble opinion:

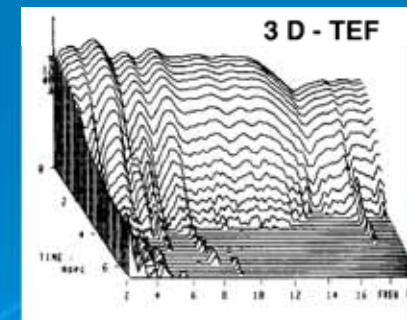
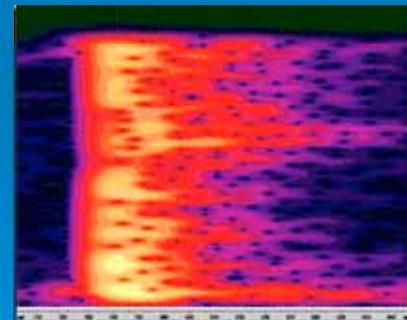
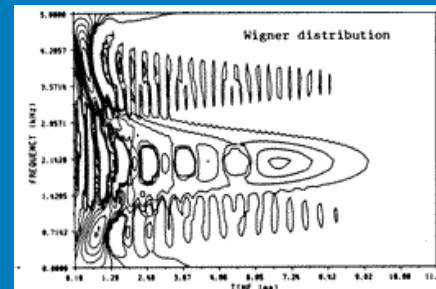
precise analysis of the effects of time distortion due to misalignment and crossovers cannot only be done by listening and by measurement of the frequency response curve

Tools to study time distortion

The best tools to study the time distortion are:

- Wigner-Deville transform
- sliding FFT which lead to the equivalent presentations called:

- spectrogram
- waterfall graph



The waterfall, due to the used perspective can mask some dips due to annulations, I prefer the spectrogram presentation

- wavelets

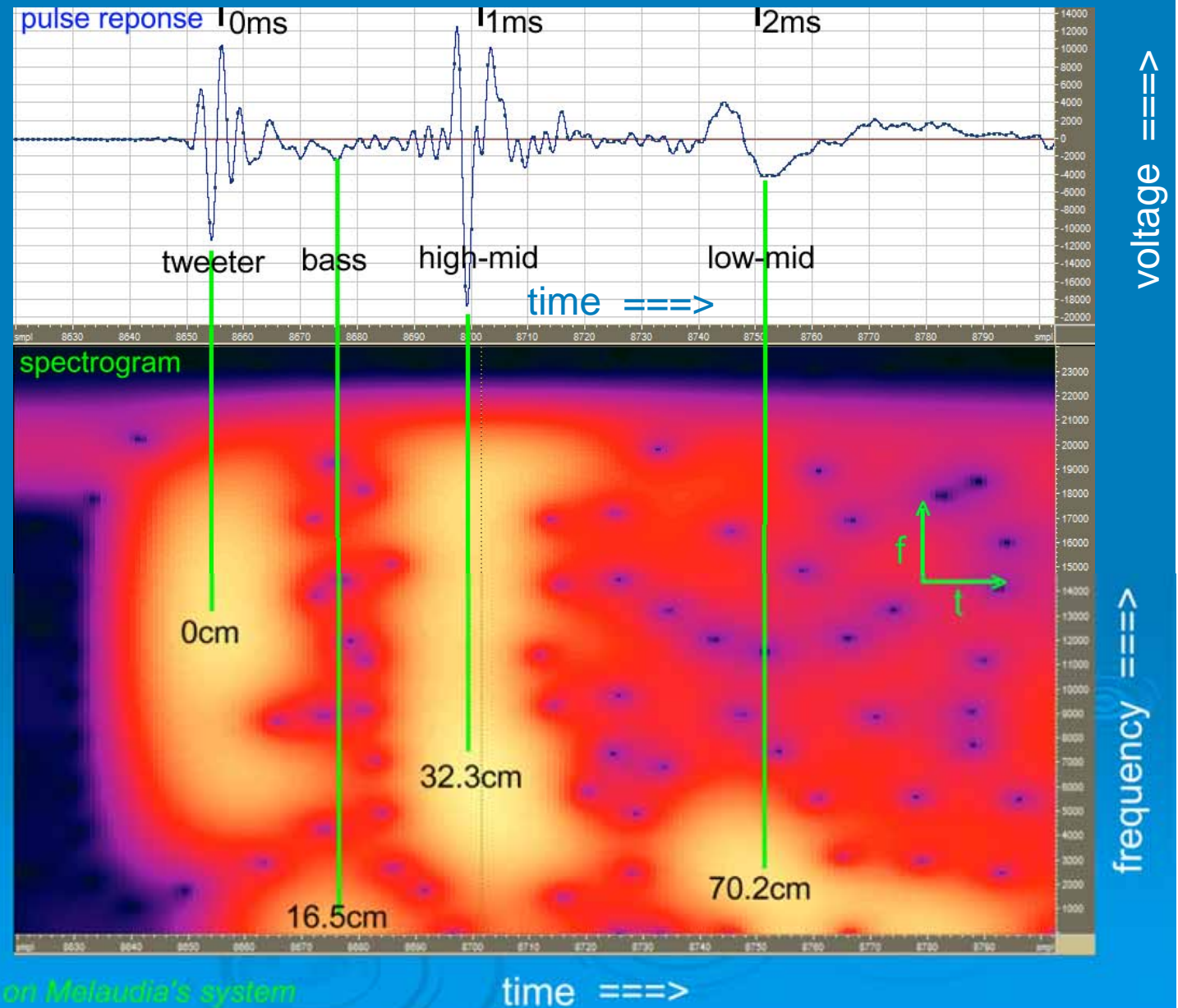
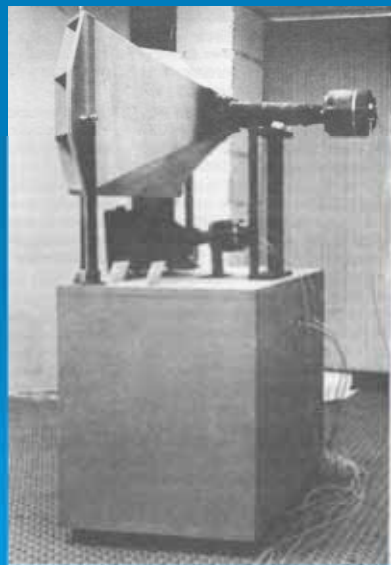
- the cheapest, easy-to-use and most efficient software for the DIYer is

Adobe Audition + Aurora plug-in

(Adobe Audition is an evolution of CoolEdit)

*The signals presented here were obtained using CoolEdit
2000 and the Aurora plug-in*

an example of an excellent sounding system not aligned: the Onken 4-ways system

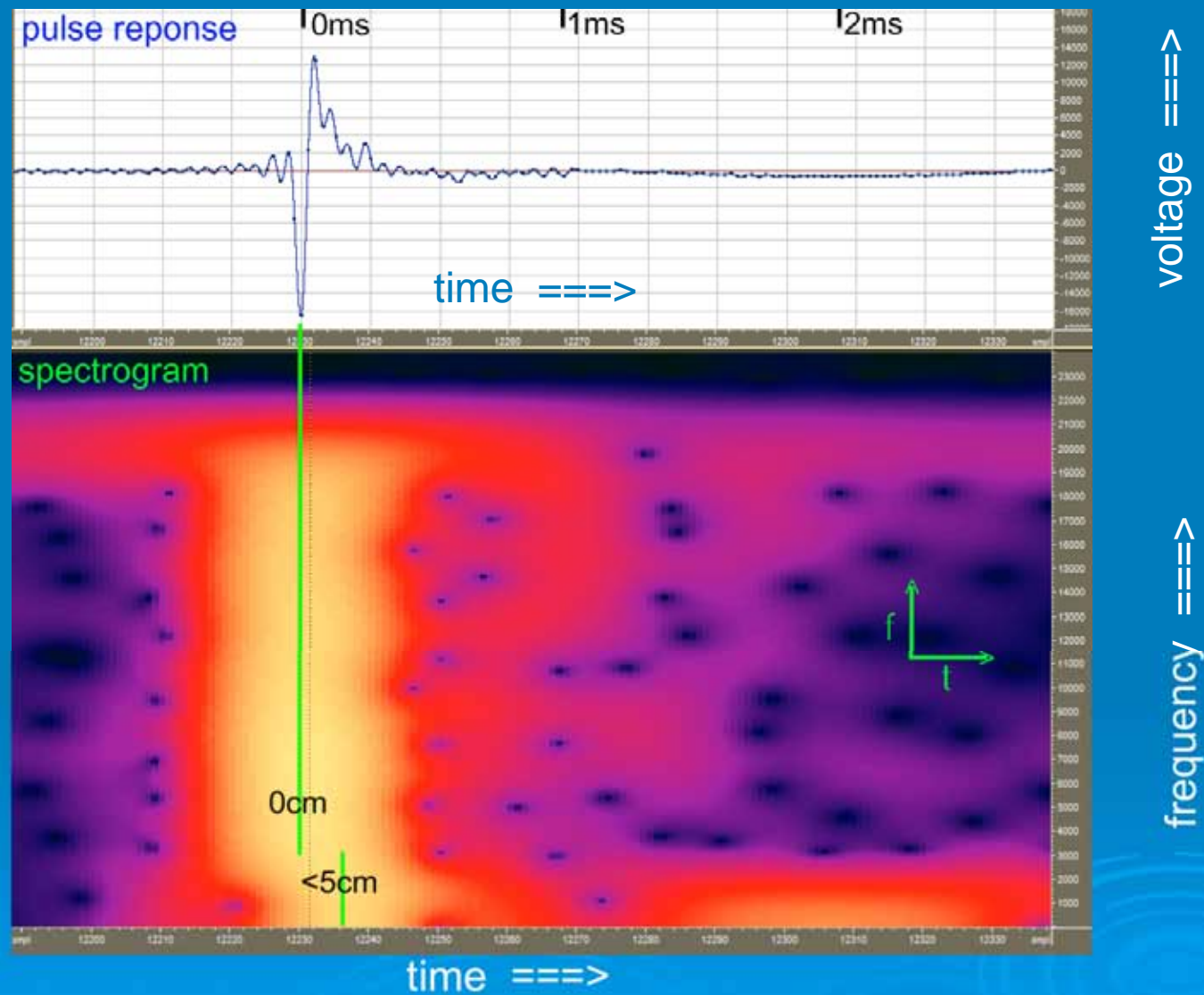


Measurements performed on Melaudia's system

- not aligned systems and specially horns system need to be listen from a quite long distance for the fusion effects to happen
 - from this quite general opinion, when people comes in a listening room with a horn system, they try to place themself at the largest distance possible from the horns.

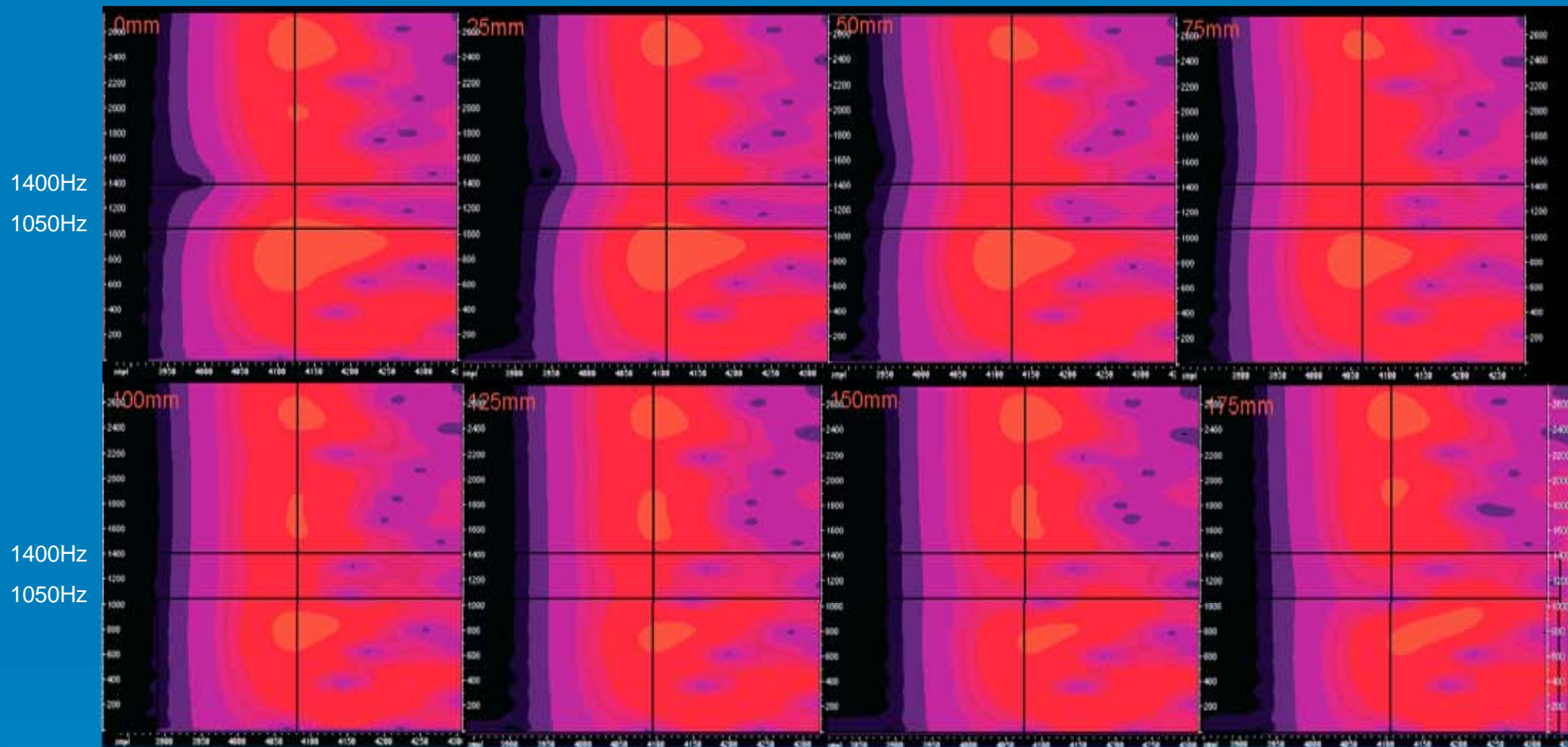
- aligned horns systems allow near-field listening with 3D imaging
 - but at the first time someone listen to them, they seem less involving.

an example of a well aligned horn system



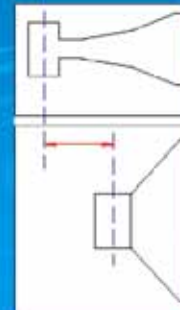
it is not necessary to compensate the remaining few centimeters misalignment due to an irremediable residual delay time due to the crossover (see spreadsheet use later)

time alignment of a 2 way system step by step



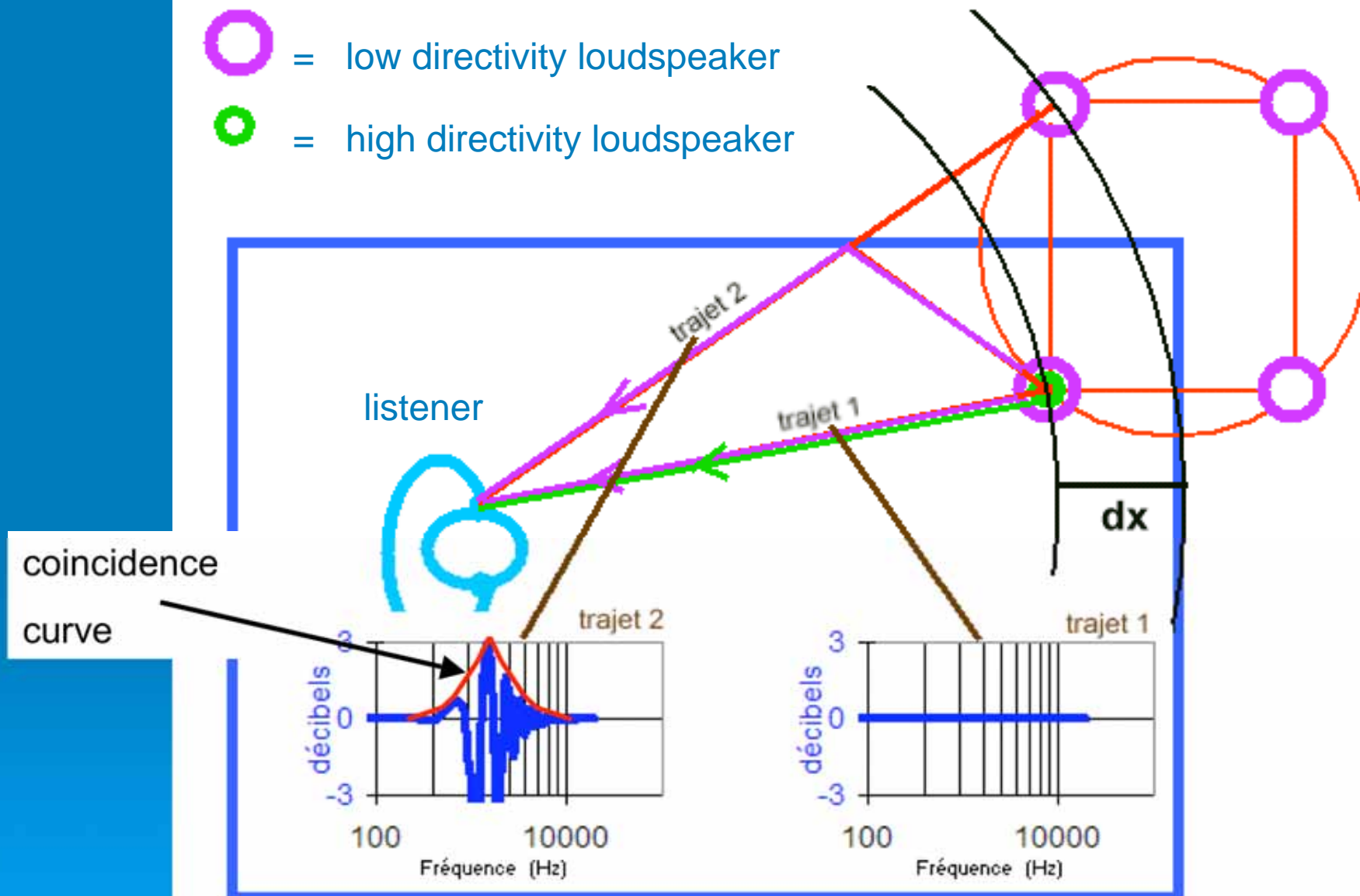
additional delay adjustment on the digital crossover by step of 25mms from 0 to 175mm

in that example the optimal additional delay happens to be 100mm



introducing the coincidence curve

-  = low directivity loudspeaker
-  = high directivity loudspeaker



- the coincidence curve is an important tool to understand the tonal signature of a given listening room
- in the "relay zone" different crossovers give different coincidence curves.

see use of the spreadsheet later

simulation of a multiways system must take in account:

for the different filters (low-pass and high-pass:

type

order (or slope)

cut-off frequency

for the different ways:

gain

polarity

additionnal delay or offset

additionnal phase correction

The spreadsheet I designed contains the main type of filters.

transfer functions of function classical polynoms (Butterworth, Bessel, etc.) are expressed as:

low-pass:

$$H(j\omega) = \frac{1}{a_d (j\omega)^0 + b_d (j\omega)^1 + c_d (j\omega)^2 + d_d (j\omega)^3 + \dots + d_n (j\omega)^n}$$

high-pass:

$$H(j\omega) = \frac{(j\omega)^n}{a_d (j\omega)^0 + b_d (j\omega)^1 + c_d (j\omega)^2 + d_d (j\omega)^3 + \dots + d_n (j\omega)^n}$$

notice that for a low-pass of order n the phase at f-3dB is given by the formula:

$$\varphi = n \cdot 45^\circ$$

Transfer function of a filter

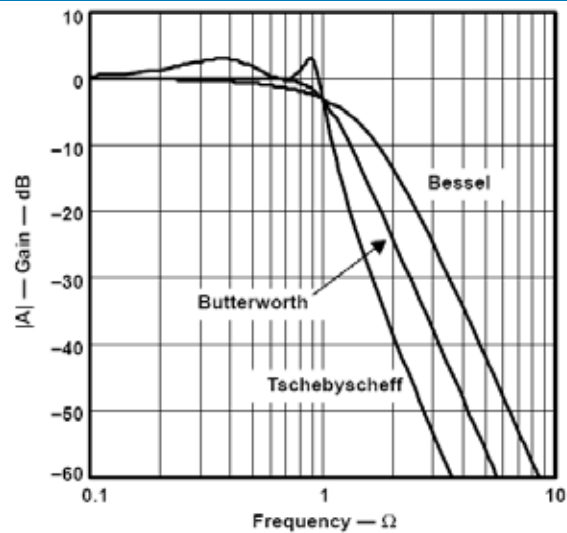
General expression = ratio of 2 complex polynoms

$$H(j_{-}) = \frac{a_n (j_{-})^0 + b_n (j_{-})^1 + c_n (j_{-})^2 + d_n (j_{-})^3 + \dots + d_n (j_{-})^n}{a_d (j_{-})^0 + b_d (j_{-})^1 + c_d (j_{-})^2 + d_d (j_{-})^3 + \dots + d_n (j_{-})^n}$$

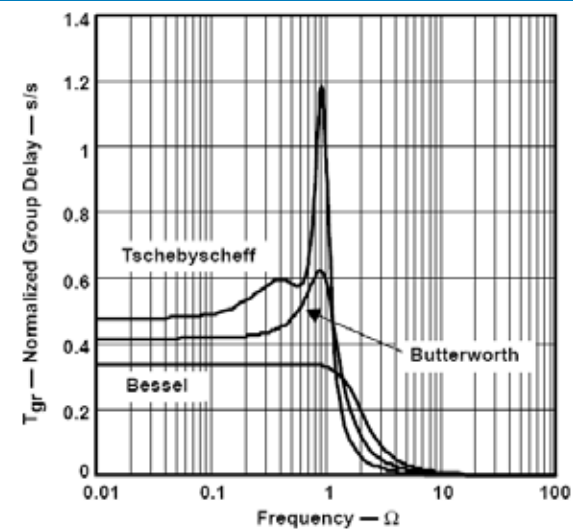
with :

$$j = \sqrt{-1}$$

$$j_{-} = -1$$

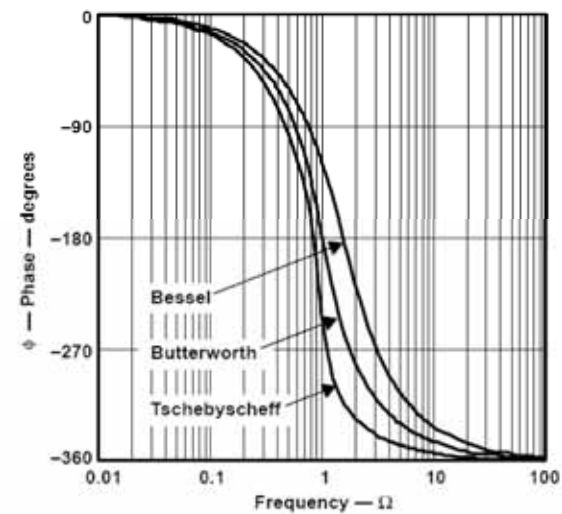


Comparison of Gain Responses of Fourth-Order Low-Pass Filters



Comparison of Normalized Group Delay (T_{gr}) of Fourth-Order Low-Pass Filters

Comparison of 3 low-pass filters



Comparison of Phase Responses of Fourth-Order Low-Pass Filters

from "Active filters design" by Texas Instrument

My goal was to try to find a method that leads for a multiways system to a better 3D imaging in near-field listening

1) excellent impulse response (directly propagated signal)

this means both:

- a minimal ondulation in the magnitude response curve
- the most constant delay time curve
(specially for frequency $\leq 4\text{kHz}$)

2) lesser room signature inside the relay frequency zone

this means

the lowest amplitude for the peaks on the coincidence curve

low frequency channel

low-pass 1

gain in
décibels =
0

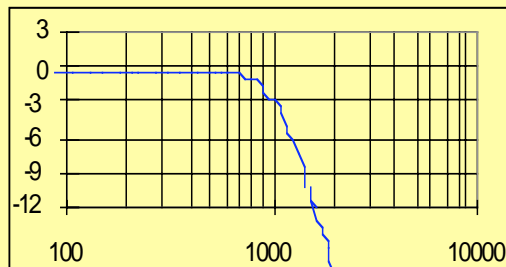
Type : Butt or Bess
or Casc or Link =
Butt

added phase
in degrees =
0

Order:
1, 2, 3 or 4
3

driver offset
in millimeters =
0

cut-off frequency
at -3dB, in Hz =
1000



mid frequency channel

high-pass 1

Type : Butt or Bess
or Casc or Link =
Butt

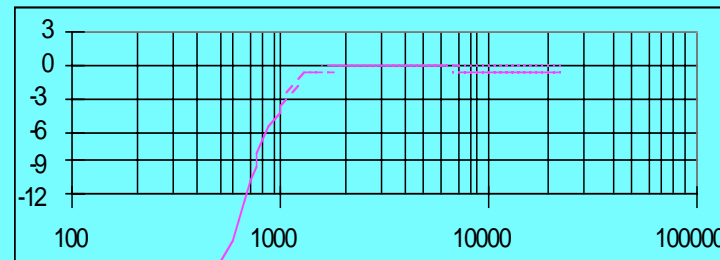
Order:
1, 2, 3 or 4
3

cut-off frequency
at -3dB, in Hz =
1000

gain in
décibels =
0

added phase
in degrees =
180

driver offset
in millimeters =
0



low-pass 2

Type : Butt or Bess
or Casc or Link =
Butt

Order:
1, 2, 3 or 4
1

cut-off frequency
at -3dB, in Hz =
1000000

high frequency channel

high-pass 2

Type : Butt or Bess
or Casc or Link =
Butt

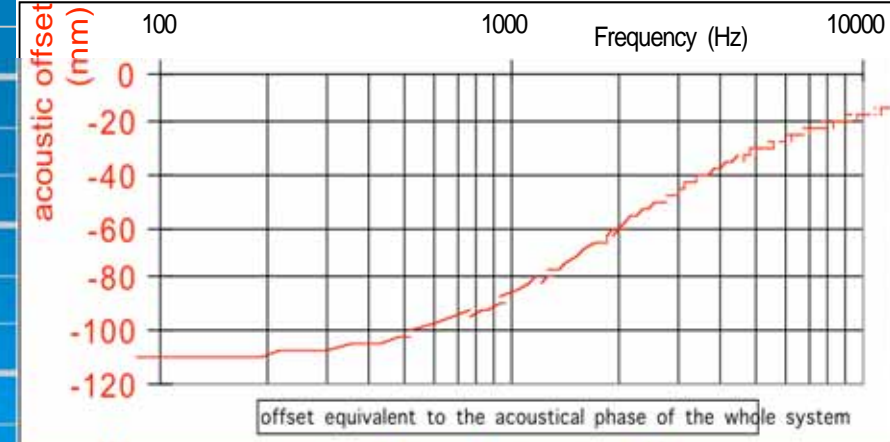
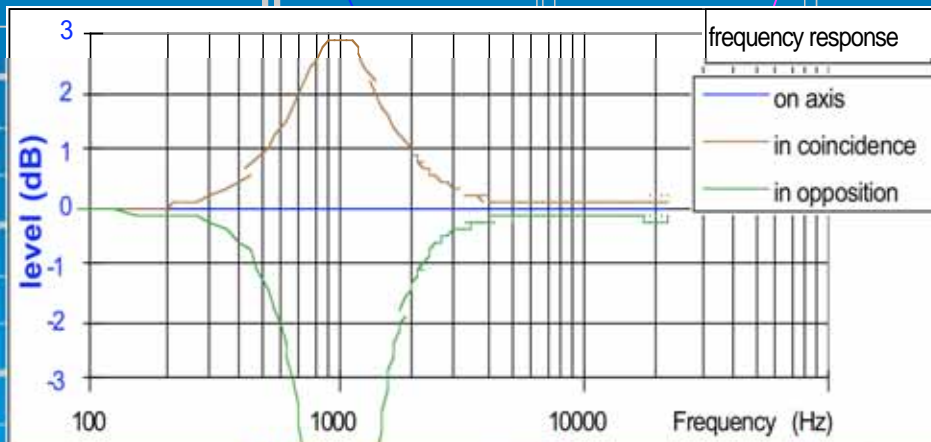
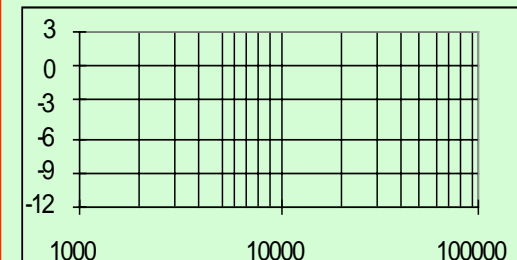
gain in
décibels =
-0.5

Order:
1, 2, 3 or 4
1

added phase
in degrees =
0

cut-off frequency
at -3dB, in Hz =
1000000

driver offset
in millimeters =
0

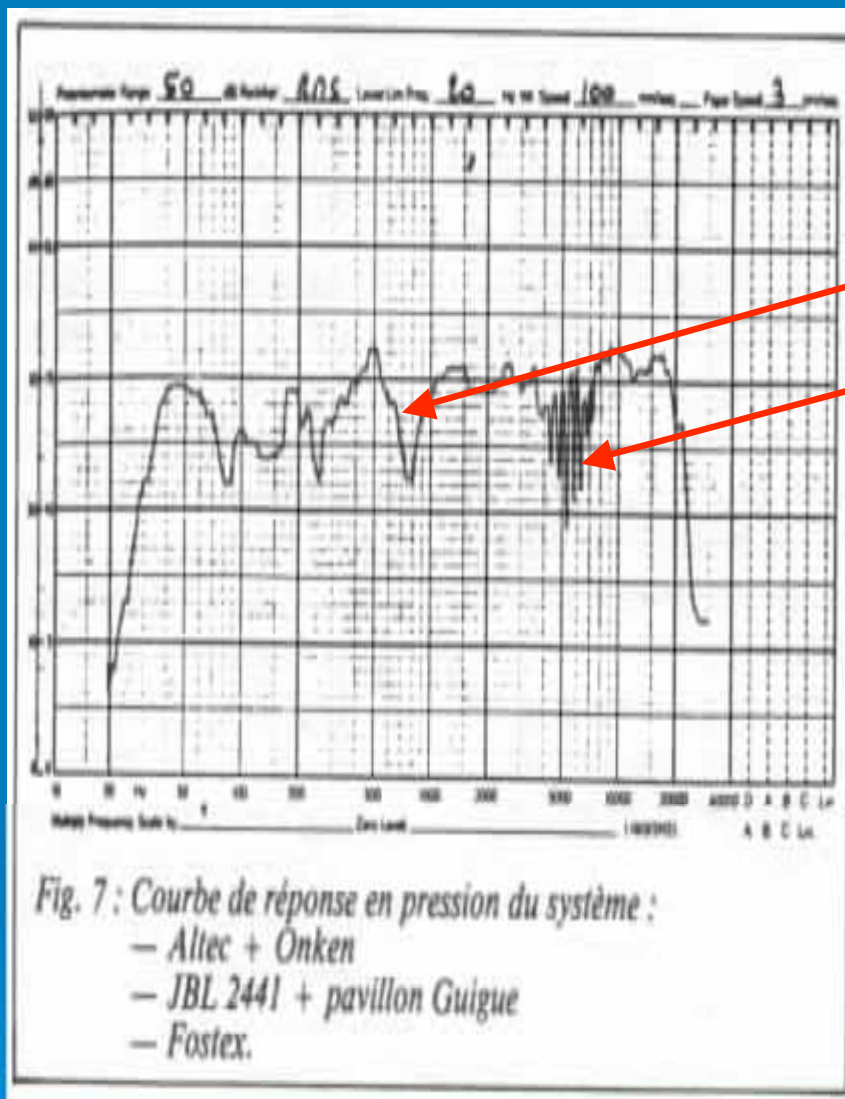


Ripple = 0,0 dB

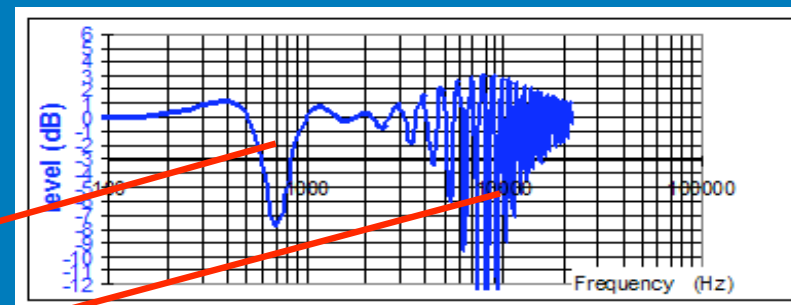
Enter your own parameters in the unprotected cells

offset variation = 72 mm
(f < 4kHz)





the response curve as published in
l'Audiophile



the result of the simulation using
the spreadsheet

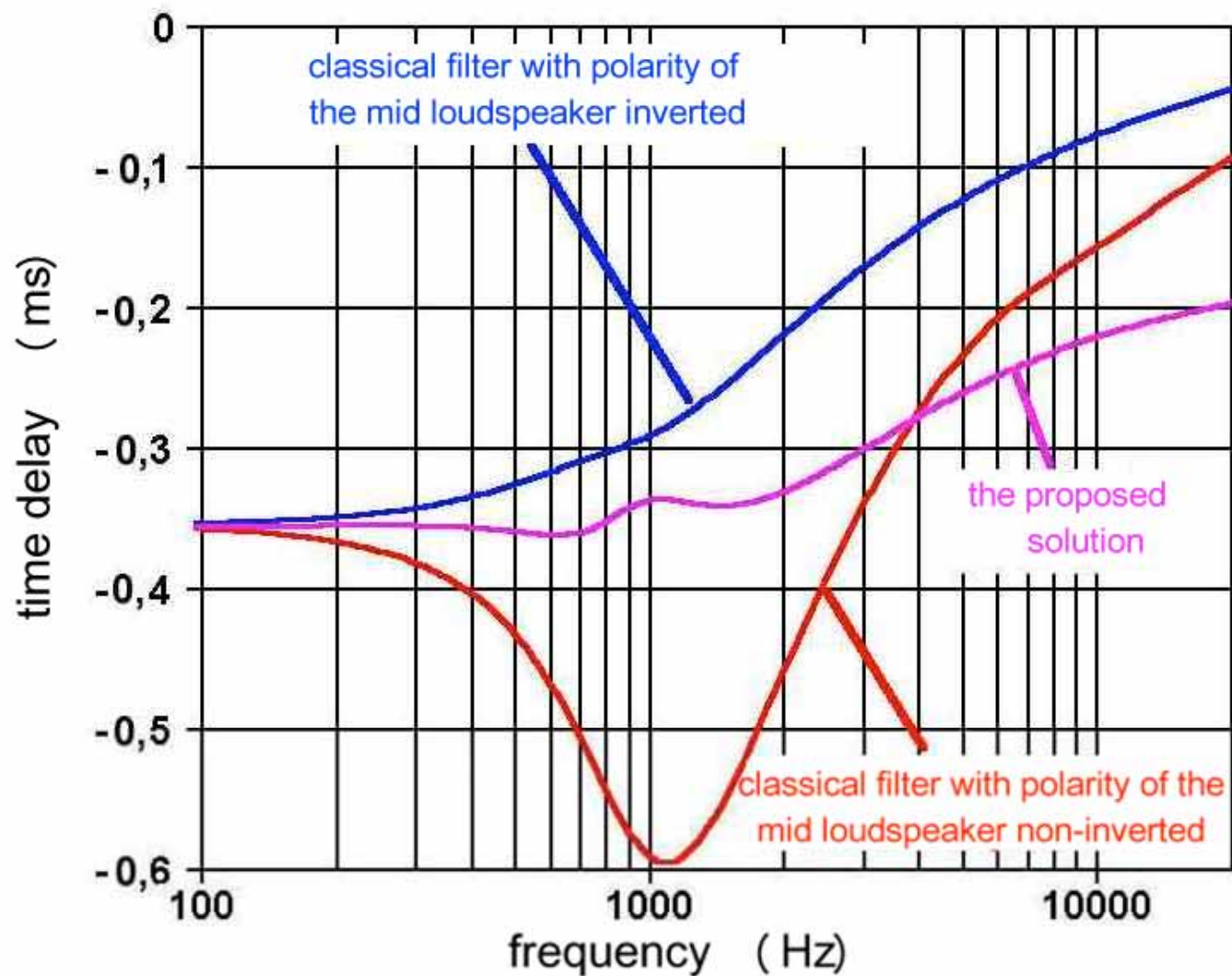
Simulation of the response curve of
a 3 ways system using the
proposed spreadsheet

Altec 416C in Grande Onken enclosure
 + JBL2441 on an lwata horn (delay due to
misalignment = 35 centimeters)
 + Fostex 925 tweeter

$F_{c1} = 600\text{Hz}$, $F_{c2} = 6000\text{Hz}$

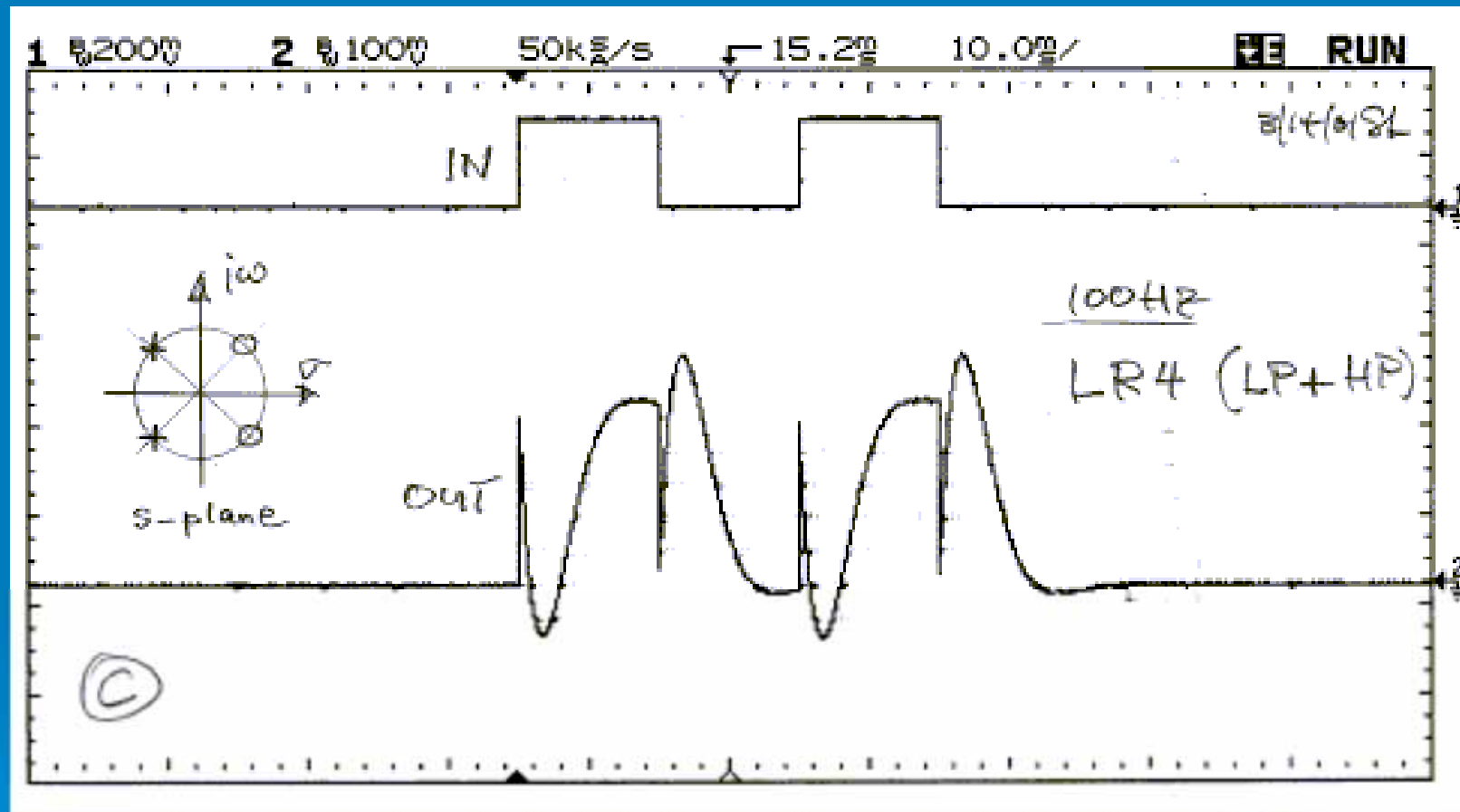
Butterworth 3rd orders

all speakers same polarity



**simulated time delay curves for 3 ways systems all using
3rd order Butterworth filters**

Simulation of response to square waves

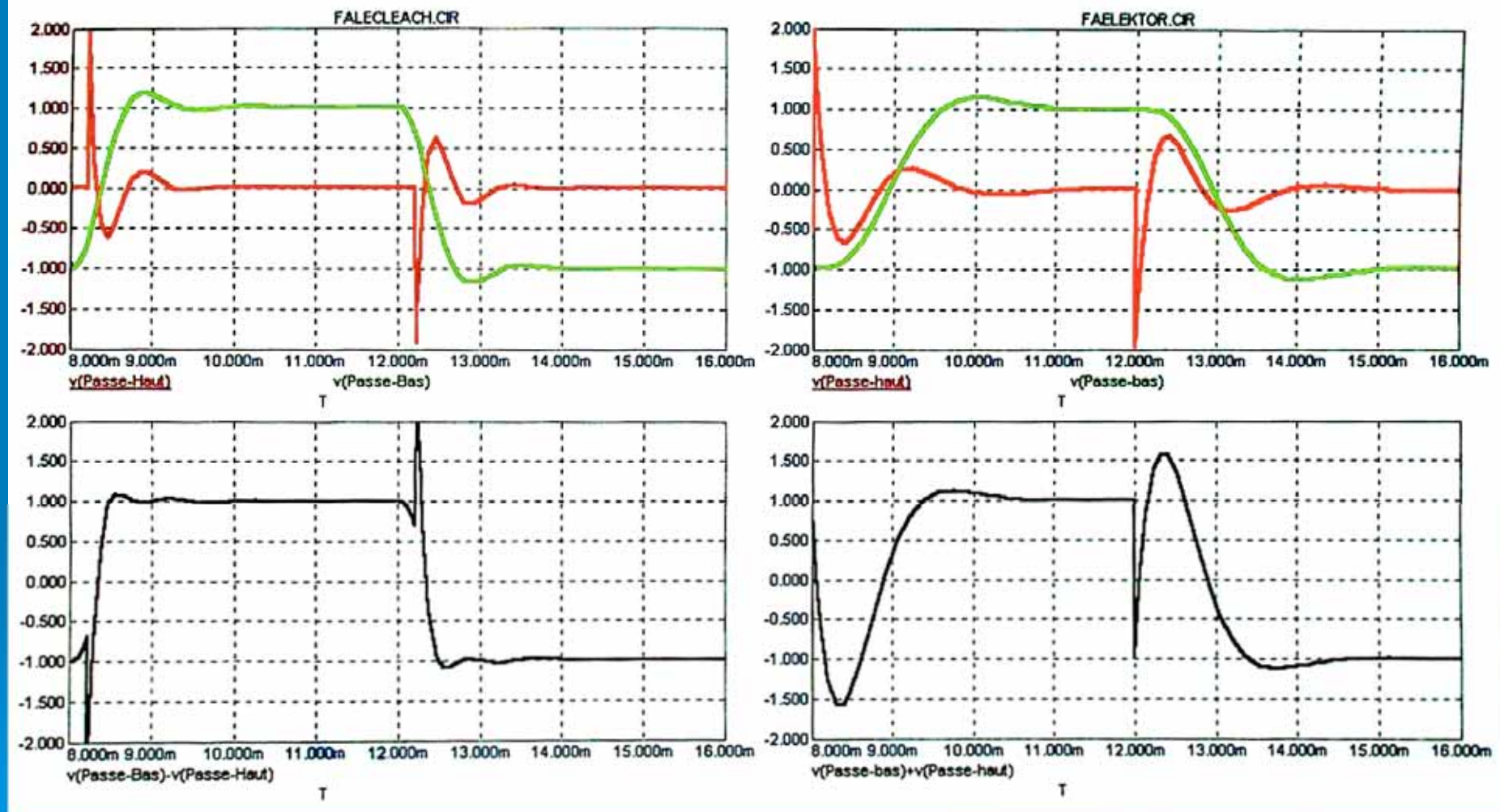


Linkwitz-Riley response to a square wave

by Siegfried Linkwitz

square wave response of 2 ways system using different crossovers and alignment procedures

by Francis Brooke



method Le Cleach

system using Linkwitz-Riley 4th order crossover

settings of a 2 ways system according the proposed method

1.....Only Butterworth 3rd order filters are used.

2.....The relay frequency F_r between the low-pass and the high-pass is defined at -5dB.

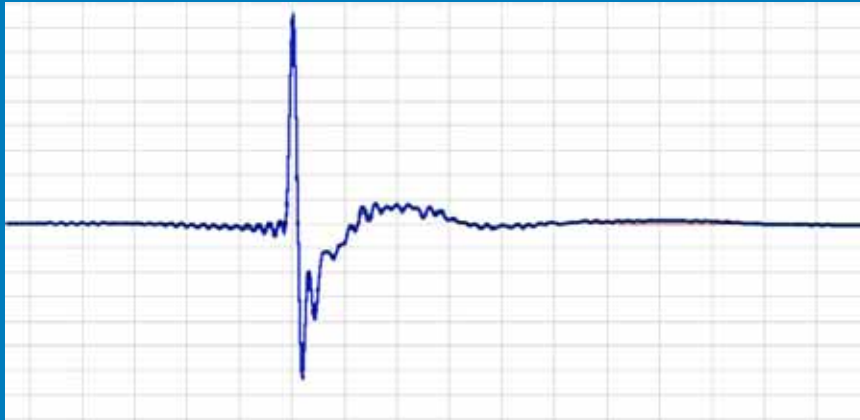
3.....The cut-off frequency F_l (at -3dB) of the low-pass filter is calculated by: $F_l = 0.87 \times F_r$

4.....The cut-off frequency F_h (at -3dB) of the high-pass filter is calculated by: $F_h = 1.14 \times F_r$

5.....If the 2 drivers are aligned at the same distance of the listener, then the low-frequency loudspeaker has to be moved toward the listener of a distance equal to $0.22 \times \text{wavelength at } F_r$.

6.....The polarity of the high frequency loudspeaker has to be reversed.

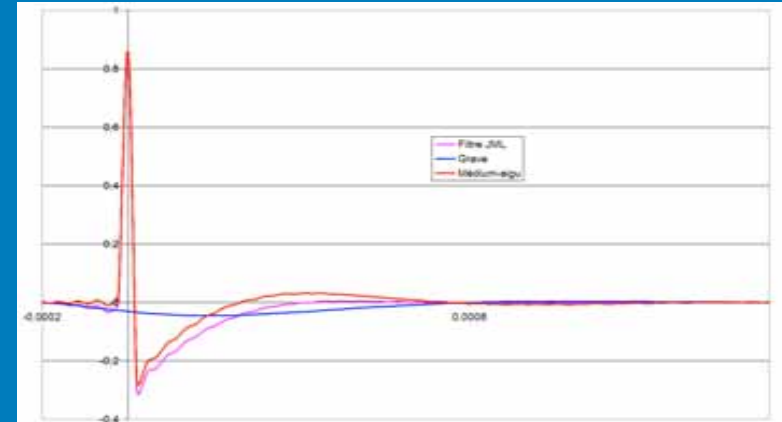
from theory to reality



measurement of the 2 ways system built by the "French Team" for ETF2004.

**measurement performed at
ETF2004 in Langenargen on
thursday December 2nd evening**

*pulse response measurement by Angelo
Farina's method (log-sweep +
convolution...)*



the simulation through
an inverse FFT of the
results given by the
spreadsheet

« Quasi-Linkwitz 3rd order »

by Francis Brooke:

Goal: to obtain a crossover low-pass + high-pass for which :

- the response in tension is quasi constant
- the delay time curve is quasi constant.

No classical filters like Butterworth 3rd order or Linkwitz-Riley 4th order can be used to reach that goals.

Theorical design: starting from the transfer function of a a low-pass of 3rd order

$$PB = 1 / (1 + a_1 \cdot p + a_2 \cdot p^2 + a_3 \cdot p^3) \quad \text{with} \quad p = j \cdot f / f_c$$

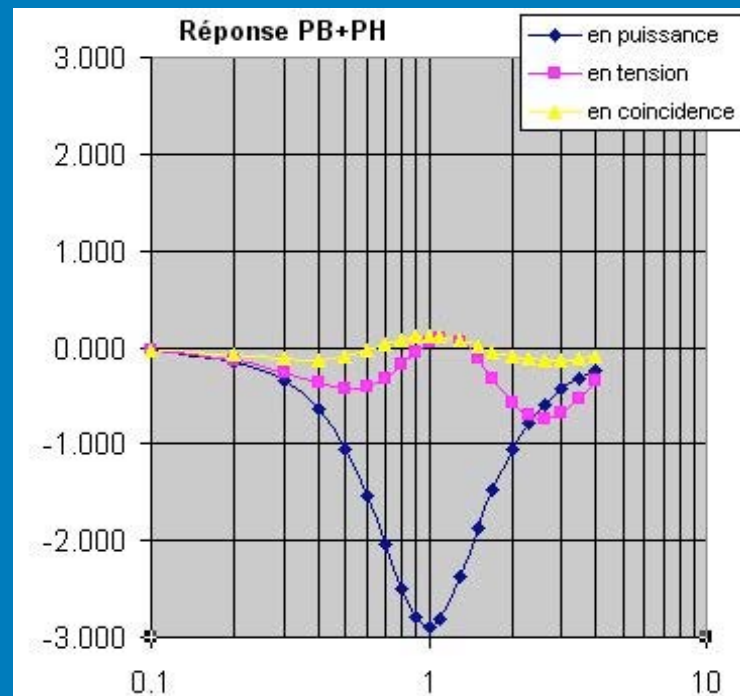
a global constant response for the low-pass + high-pass can be obtained if:

$$||PB|| = 1 / (1 + (f/f_c)^3) \quad \text{and} \quad ||PH|| = (f/f_c)^3 / (1 + (f/f_c)^3)$$

There is no exact solution to the problem but it is possible to propose an approximate solution choosing:

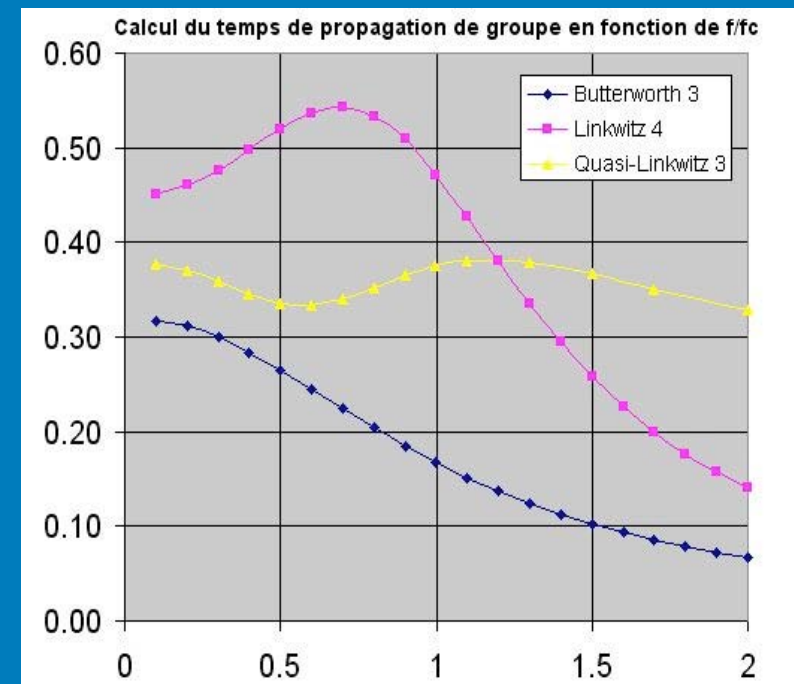
$$a_1 = 2.3732 \quad , \quad a_2 = 2.399 \quad \text{and} \quad a_3 = 0.9823$$

Then the high pass should be delayed of **0.21 wavelength** à the cut-off frequency.



The total response curve in tension for the low-pass + high-pass (yellow curve) is within an interval $-0.75\text{dB} + 0.09\text{ dB}$.

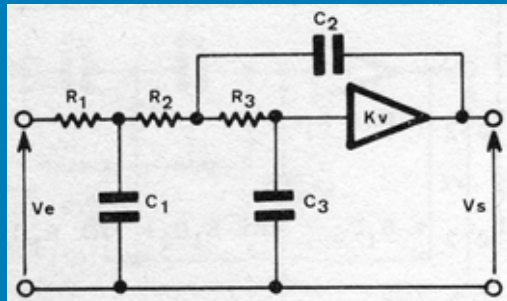
The coincidence curve (module of the low-pass + module of the high-pass) varies in a very small interval $-0.13\text{dB} + 0.12\text{ dB}$.



The resulting delay time curve (yellow) is nearly constant when compared to the delay time curve of the Butterworth 3rd order and Linkwitz-Riley 4th order.

Practical realization

Low-pass



The low-pass 3rd realization is classical
with $R_1 = R_4 = R_3 = R$ and $K_v=1$
we obtain:

$$a_1 = R \cdot (C_1 + 3 \cdot C_3) \cdot (2\pi f_c) \quad \text{with} \quad a_1 = 2.3732$$

$$a_2 = R \cdot (C_1 + C_2) \cdot 2C_3 \cdot (2\pi f_c) \quad \text{with} \quad a_2 = 2.399$$

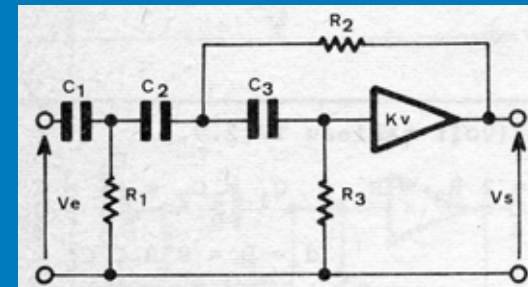
$$a_3 = R_3 \cdot C_1 \cdot C_2 \cdot C_3 \cdot (2\pi f_c)^3 \quad \text{with} \quad a_3 = 0.9823$$

$$\text{then with} \quad C = 1 / (R \cdot 2\pi f_c)$$

we obtain:

$$C_1/C = 1.332 \quad C_2/C = 2.125 \quad \text{and} \quad C_3/C = 0.347$$

High-Pass



In a similar way, the high-pass can be realized with:

$$C_1 = C_2 = C_3 = C \quad \text{and} \quad K_v = 1$$

we obtain :

$$a_2/a_3 = C \cdot (2R_1 + 2R_2) \cdot (2\pi f_c)$$

$$a_1/a_3 = C \cdot (3R_1 + R_3) \cdot R_2 \cdot (2\pi f_c)$$

$$1/a_3 = C^3 \cdot R_1 \cdot R_2 \cdot R_3 \cdot (2\pi f_c)^3$$

$$\text{then with} \quad R = 1 / (C \cdot 2\pi f_c)$$

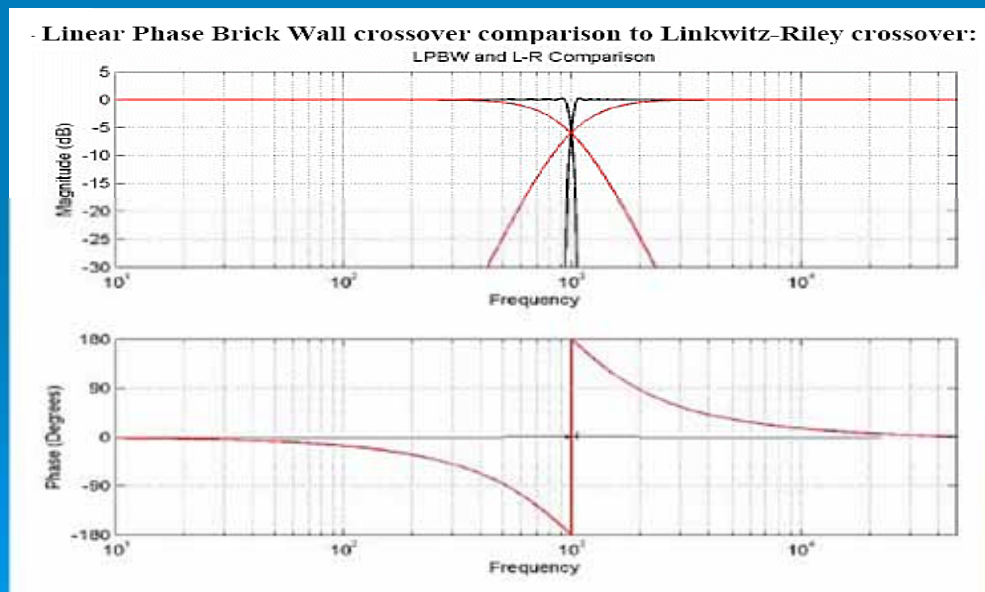
we obtain

$$R_1 / R = 0.751 \quad R_2/R = 0.471 \quad \text{and} \quad R_3/R = 2.882$$

a bit of prospective:

the future will be most probably and very soon different than the approach presented here that use only classical filters.

The introduction of Finite Impulse Response filters and the generalization of Digital Signal Processors (DSP) will allow more easy set-up of multiways systems and even more perfect impulse response.



from "Lake Contour brick Wall filters"