Time smear at unexpected places in the audio chain and the relation to the audibility of high-resolution recording improvements

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The criticisms on the CD format from audiophiles is mainly based on listening tests. For me too, it has never been a problem to demonstrate the superior sonic quality of analog records over the CD by playing "old" analog recorded music both on an old-fashioned record player and its CD version. But not only listening tests point at problems with the CD format, there is clear evidence that the steep anti-aliasing (during recording) and reconstruction (at playback) filtering introduce audible artifacts in the sound. One (but not the only one!) of the main problematic consequences is the "smearing in time", caused by these filters. This means that the duration of any sound during playback will be extended (in time domain) compared to the original, in other words, its energy is smeared out over a longer period of time. This also holds for *changes* in the signal itself (which take more time in the reproduced sound than the changes in the original), which can thus reduce transparency and induce loss of detail. The temporal effects of filtering have been the topic of a paper at the conference of the Audio Engineering Society in Berlin (Germany, 1993, ref. 1), which can also be found on our website. The problems with the CD format, some of which are mentioned above, have stimulated the development of "high-resolution" formats of which the Super Audio Compact Disk (SACD) is the most prominent. However, in several controlled listening tests, the differences between the CD and SACD formats was judged as too small, albeit perceptible, to change over to these high-resolution formats (ref. 2) or not even audible at all (ref. 3). This is in grave conflict with my own experiences, in which the SACD gives a clear, consistent and clearly audible improvement over the CD format. This is obvious when comparing the CD version of an analog master recording with both the analog record and the SACD version. But also comparing the sound, as produced by novel Direct Stream Digital (DSD) recordings with older digital recordings, gives a less aggressive, more open and more natural ("live") sound. This is most clear with small bells and the like (as e.g. can be heard in "Singing Winds, Crying Beasts" from the "Abraxas" album of Carlos Santana), from a group of violins as it performs in a symphony orchestra or a choir. The "explanation" as given in ref. 3 (which boils down to the argument that most SACD recordings are made by "audiophile" labels and thus pay more attention to the quality of the recordings) is in conflict with the comparison of old analog recordings (see above) and implicitly accuses the recording engineers of "ordinary" labels of doing a bad job, which is not in agreement with my own experiences, certainly when we talk about recordings of classical music.

A basic requirement for all these listening tests is that the equipment, used for playback, is at the upper limit of current technology in order to avoid the pitfall that the limitations of the playback equipment mask the improvements, brought by the high resolution formats[1]. The use of "better than mainstream" equipment (ref. 2) is on the one hand a rather imprecise statement and might on the other hand not be good enough to get consistent results. The playback systems should also have very limited "time smear" themselves (or a high temporal decay, as defined in ref. 1), as this one of the major drawbacks of the CD standard as has been discussed above and in ref. 1. The time smear of

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[1] Note that all kinds of lossy compression techniques were advocated as being "inaudible" when they were first presented. In reality, this proved to be incorrect and audiophiles, including myself, agree that e.g. the stereo image of MP-3 compressed music is "flat", basically caused by the loss of detail due to the lossy compression. The "inaudibility" was caused by the limitations of the playback equipment and **not** by the limitations of human hearing. In general, the current state-of-the-art audio systems still underperform compared to the abilities of human hearing, so any statement that certain effects are "inaudible" should be taken with at least a (and probably a lot more than one) grain of salt.

"mainstream" playback equipment is, however, substantial as this is an often ignored quality of this kind of equipment. This will be elucidated in this note. After a short recap of the basics of "time smear" by band-limited systems, we will try to identify the flaws of the tests as described in refs. 2 and 3 and see if we can find additional sources of "time smear" which could mask the improvements of the high-resolution formats.

Any band-limited transmission system has an impulse response which is wider than the width of the incoming impulse (which -in theory- should have a width of zero) and thus "smears out" the energy of the incoming signal in time domain. This is a fundamental property and a direct consequence of Fourier theory (refs. 4 - 6) and no sound reproduction system ("audio system" for short) is an exception to this rule. The discussion in the high-end audio world should therefore also focus on the question to what extent this phenomenon, here aptly referred to as "time smear", is audible. In other words, what are the requirements for an audio system such that this "time smear" is no longer audible. In a presentation at the Berlin conference of the AES (ref. 1), the author has introduced the concept of "temporal decay" to quantify this phenomenon for band-limited systems and a surprising result was that not only the bandwidth limit, but also the decay rate *outside* the transmission band was of prime importance for the resulting figure. The author showed that the long time required for the temporal decay of the CD anti-aliasing and reconstruction filtering will result in audible time smear and loss of detail. By now, many experts agree that the "high resolution" formats, like SACD, give a clear audible improvement over the CD quality (ref. 7). Surprisingly, however, in several listening tests the difference between the "low" and "high" resolution formats was often not clear (refs. 2 and 3). As the author has no problem to demonstrate the differences using the audio systems as produced by "Temporal Coherence", the question has come up whether more ordinary ("mainstream") audio systems introduce significant time smear which possibly masks the phenomena we are looking (or rather, listening) for. In a previous web-publication (ref. 8), the author has pointed at the problems of cross-over filters which often (to be fair in almost all systems) introduce incorrect temporal responses. Could these also introduce time smear? This question was touched upon in the Berlin presentation (ref. 1) but was not fully addressed.

To answer this question, a two-way 3rd order Butterworth cross-over filter has been analyzed. As cross-over frequency, 1 kHz has been chosen and the transfer characteristics have been calculated. The transfer function of the low-pass section is shown in fig. 1, of the high-pass section in fig. 2. The phase responses of both sections are shown in figs. 3 and 4. These characteristics are no surprise and completely as expected. The summed characteristics of the low and high pass sections is constant at 0 (zero) dB, this holds both for the situation in which the units are connected in-phase and out-of-phase. However, the phase characteristics are different for both cases. In fig. 5, the phase characteristic of the situation is shown when the (idealised) loudspeaker units[2] are connected in-phase, whereas in fig. 6 the same is presented when the loudspeaker units are connected out-of-phase. When looking at these characteristics, these might seem a bit surprising at first sight as the combined phase shift at the in-phase connection is not 270 + 270 = 540 degrees, but "only" 360 degrees. At second thought, this is not so surprising as the phases should be going to zero or a multiple of 360 degrees in order to fulfil the in-phase connection requirement. Similarly, at the out-of-phase connection, the phase is limited to 180 degrees (although in this case, 540 degrees would also fulfil the out-of-phase criterion). As can be seen from fig. 6, the phase characteristic is more gradual at the out-of-phase connection than at the in-phase connection. It is therefore interesting to study the responses in time domain of both connections.

The temporal impulse responses of both types of connections can be calculated using Fourier theory (refs. 4 - 6, a computer program to implement this calculation numerically can be found in ref. 8) and are shown in figs. 7 and 8. Both show a significant amount of time smear in the order of 0.5 - 1 msec. In agreement with the results, shown in figs. 5 and 6, the response at the out-of-phase connection is less dramatic than at the in-phase connections, albeit that the negative going impulse at the onset of the signal is obvious. But the time smear, introduced by both, is very likely to be clearly audible and will probably mask details. This is illustrated in fig. 9, in which the response of the system

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[2] In this analysis, the responses of the loudspeaker units in their enclosures is assumed to be perfect in order to determine the pure contribution of the cross-over filter. In reality, the loudspeaker units will also contribute to the time smear because of their non-perfect response.

to a tone burst of 1250 Hz of two cycles is shown when the units are connected in-phase. The distortion and extension of the signal in time is obvious. As expected, the result is better when the units are connected out-of-phase, but still the effect of time smear is clear.

N.B. Note that the results are "scalable": reducing the cross-over frequency to e.g. 333 Hz for the woofer-squawker filtering, the resulting time smear will be three times as wide as shown here.

The results above show that cross-over filters can introduce significant time-smear within the audible frequency range. In the view of the author it would be surprising if this would **not** be audible and this could (partly) explain why on many systems the improvements, brought by the high-resolution recordings are not or only marginally audible: the reduction in time-smear from the recording is masked by the time-smear, introduced by the cross-over filters. In A - B comparison tests, the audio systems used should therefore be selected for minimum time smear in the entire extended audio range (up to at least 40 kHz). As far as specified in refs. 2 and 3, this requirement is not fulfilled and therefore the tests cannot be used to conclude unambiguously that the high resolution formats give no audible improvements. The systems, as designed and build by "Temporal Coherence" do fulfil these requirements because these also use (a.o.) cross-over filters which show a correct behaviour in time domain (refs. 8 and 9) and therefore it is not surprising that these systems clearly present the major improvements by the high resolution recordings. Another important conclusion of the analysis, presented in this note, is that not only the decay rate *outside* the transmission band is of importance for the time smear of a system, but also the phase characteristic *inside* the transmission band. This puts even more restrictions on the properties of a sound reproduction system, but because of the emphasis, put on the designs by "Temporal Coherence", these systems also fulfil this additional requirement.

N.B. Note that the cross-over filters should not only have time-correct responses, but should also be "absolute causal", which means that they will not give any signal to the loudspeaker units before these are supposed to produce any sound. Cross-over filters can be designed (and build) which give out-of-phase (each other cancelling) signals to the loudspeaker units before the actual sound starts, but because of the non-perfect response of the loudspeaker units and their physical separation, it is impossible to get complete cancellation of the out-of-phase signals before the sound starts. More details can be found in ref. 8.

References

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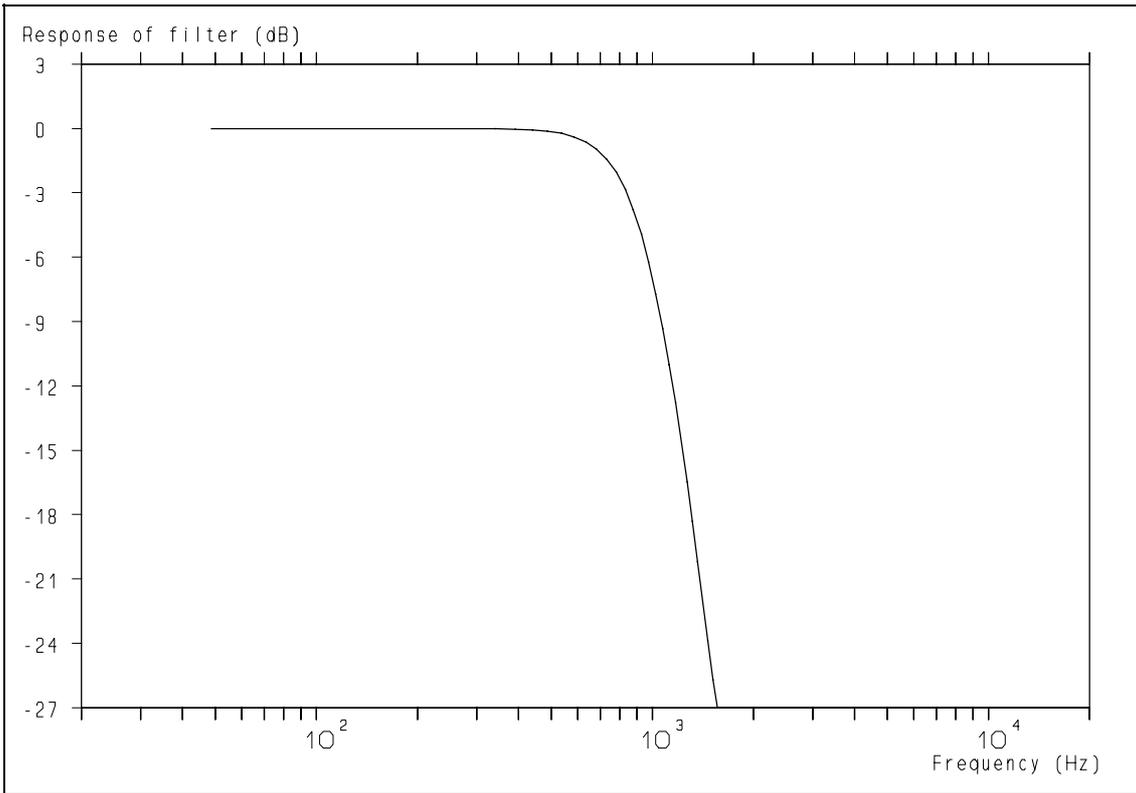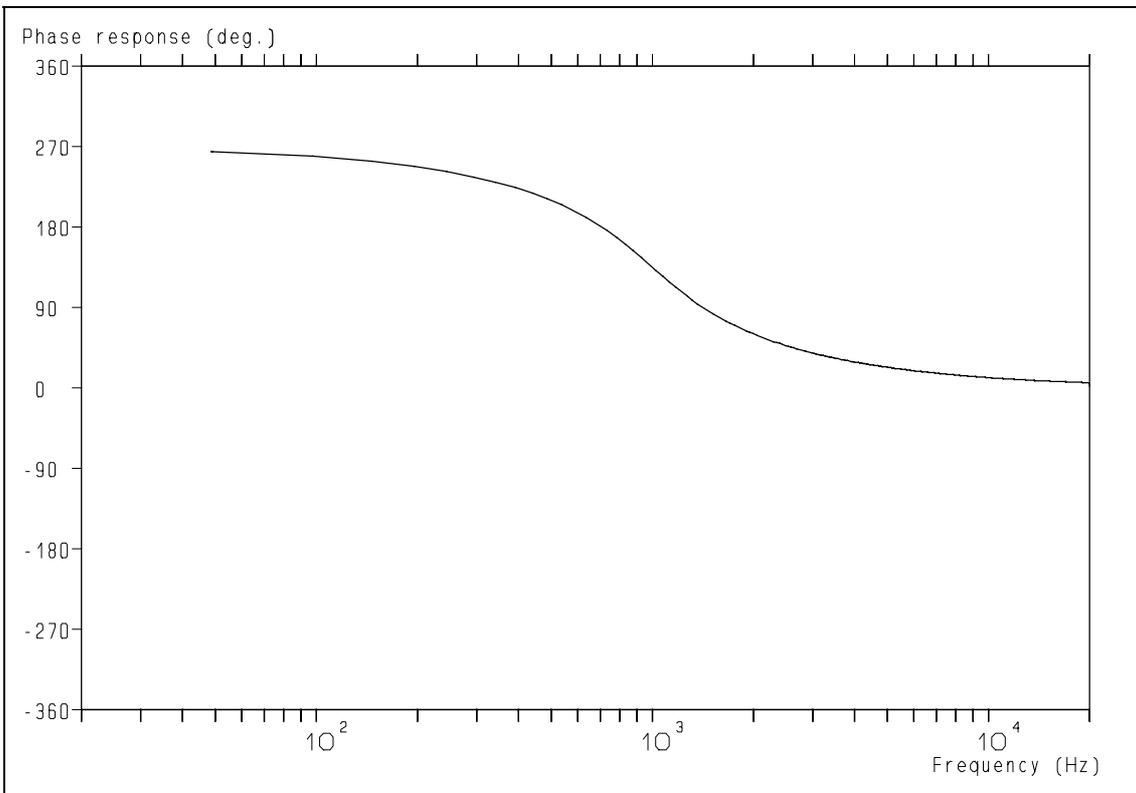
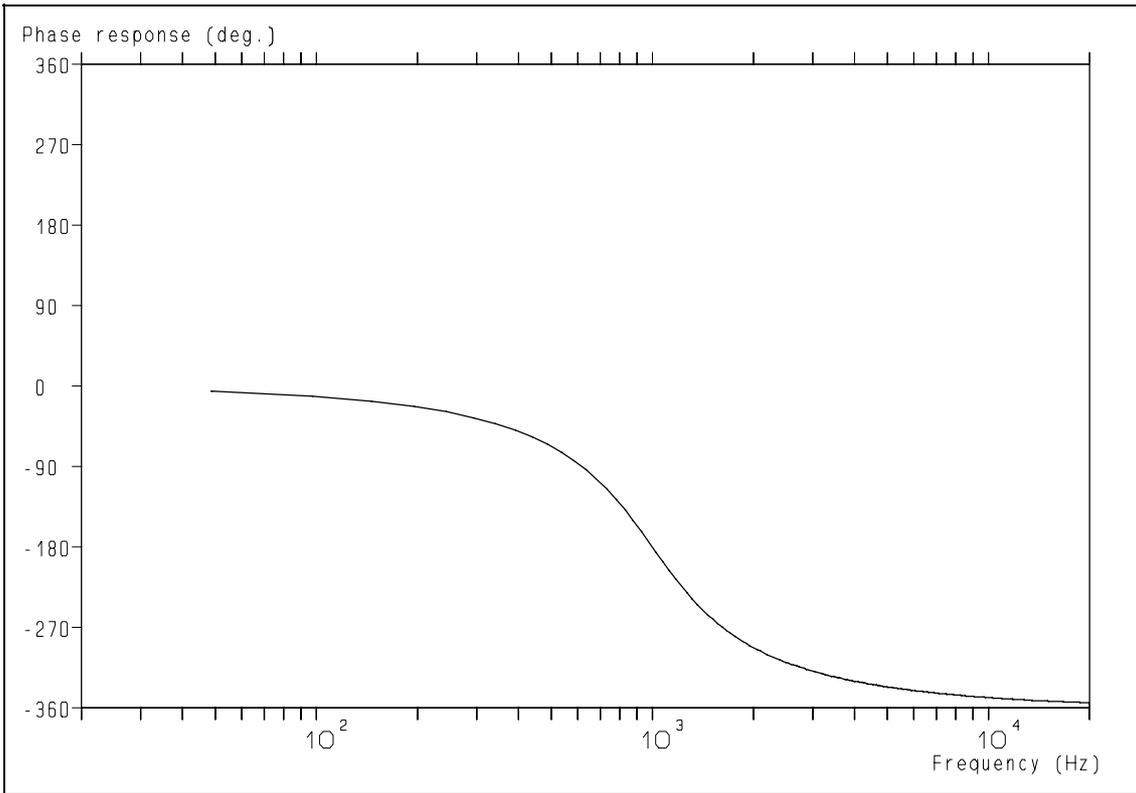**Figure 1**: Amplitude response of the low-pass section of the 3rd order Butterworth filter.
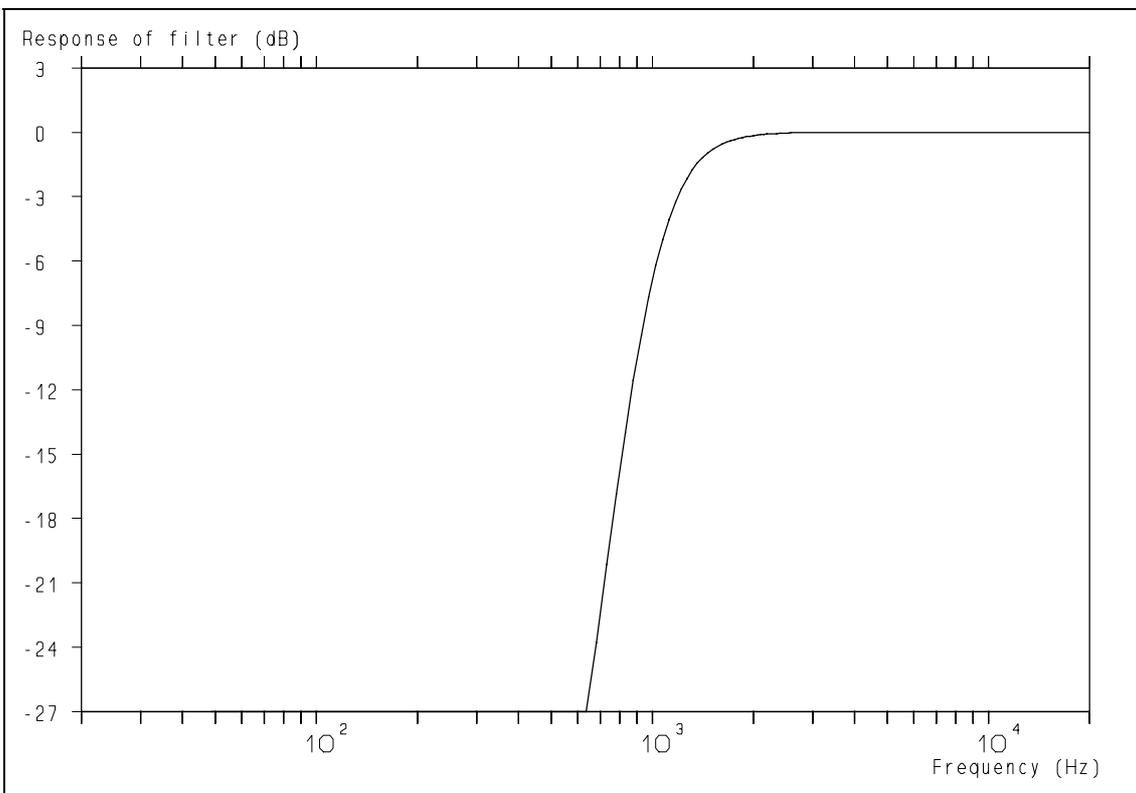
Figure 2: Amplitude response of the high-pass section of the 3rd order Butterworth filter.
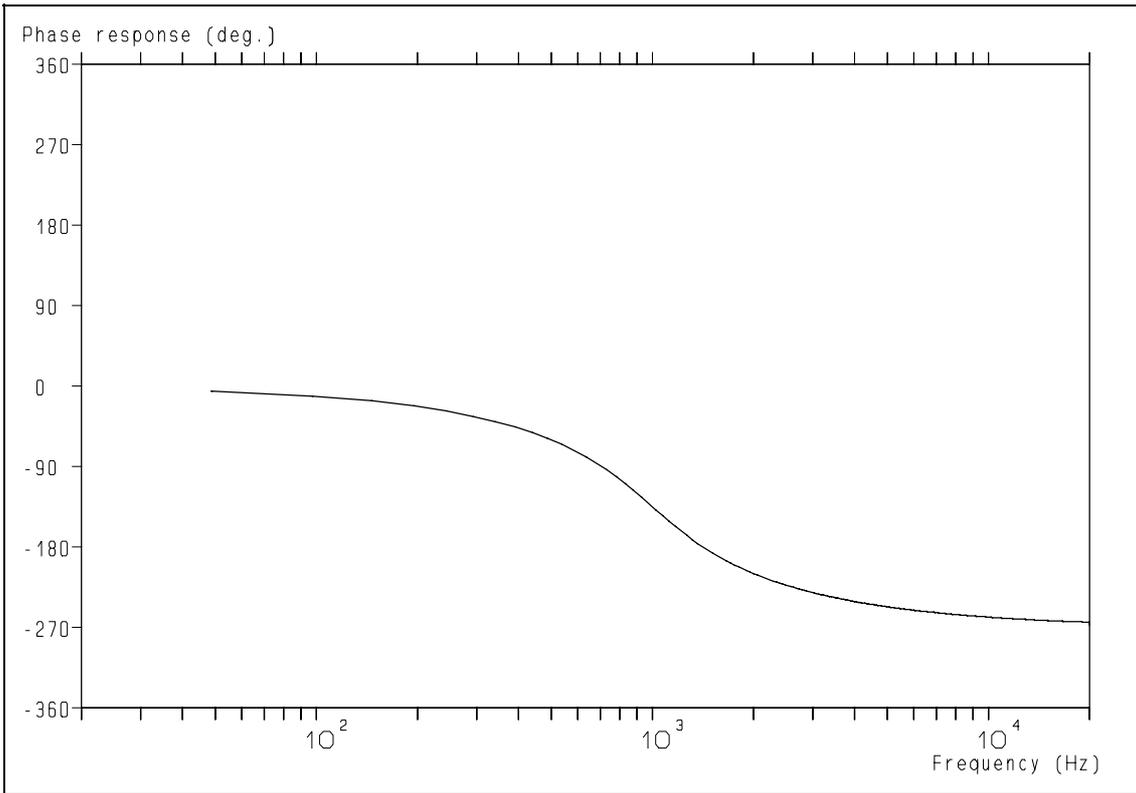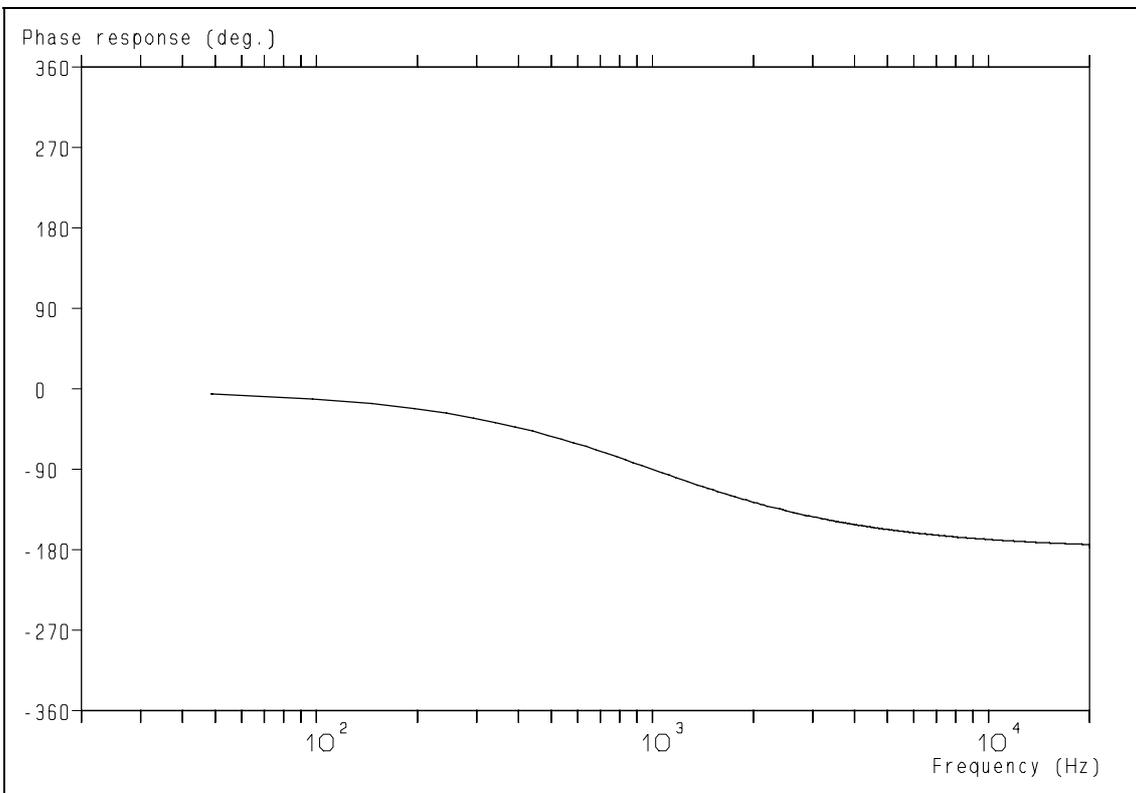
**Figure 3**: Phase response of the low-pass section of the 3rd order Butterworth filter.

**Figure 4**: Phase response of the high-pass section of the 3rd order Butterworth filter.

Figure 5: Phase response of the combined sections of the 3rd order Butterworth filter when the units are put in-phase.


Figure 6: Phase response of the combined sections of the 3rd order Butterworth filter when the units are mounted in anti-phase.
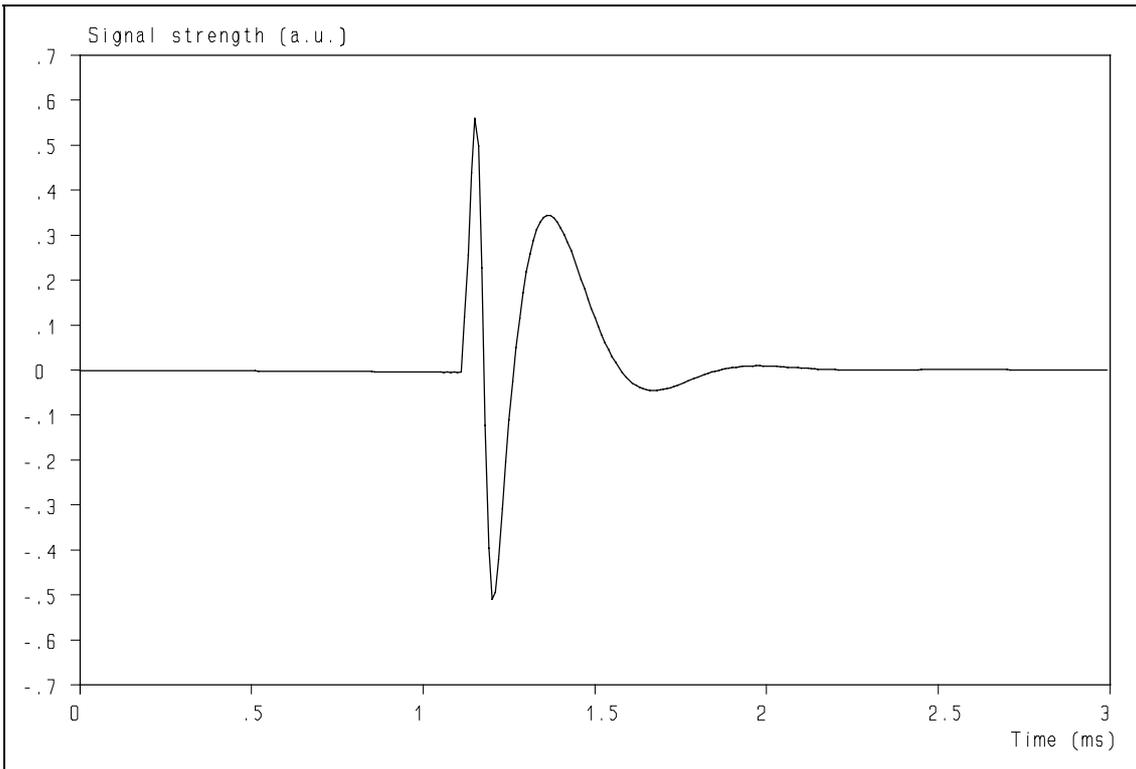
Figure 7: Dirac impulse response of idealised two-way system with 3rd order Butterworth filtering and the units operating in phase.
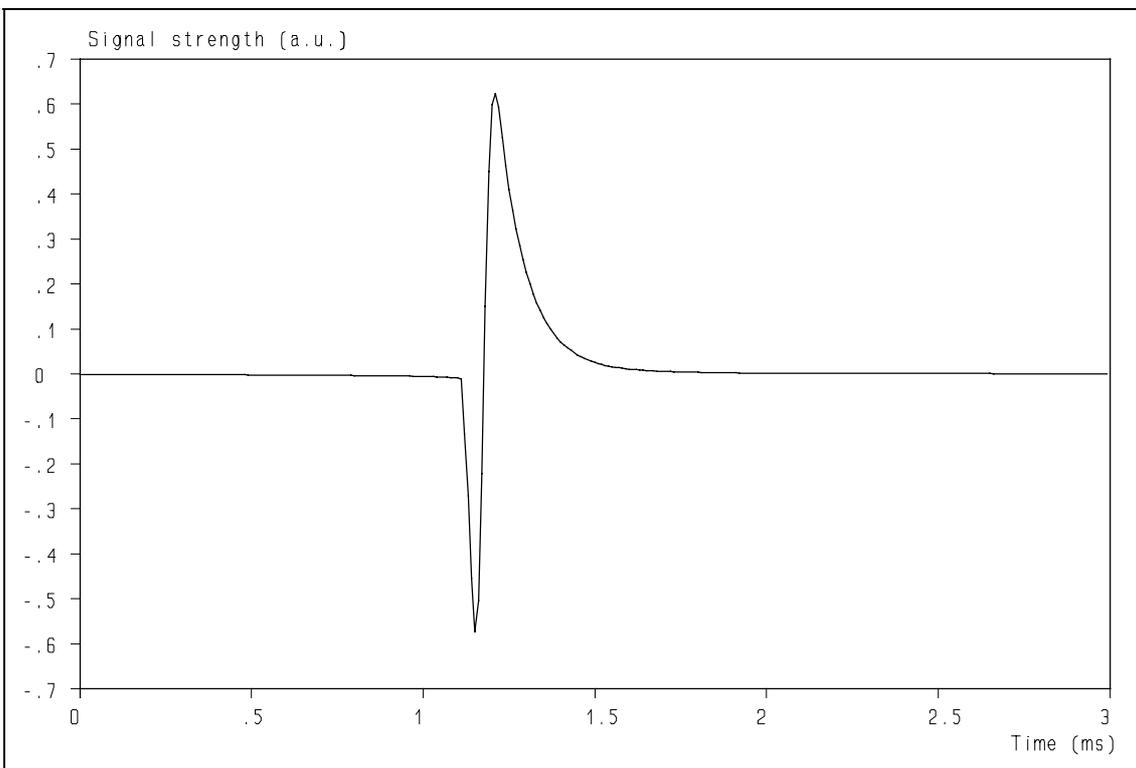
Figure 8: Dirac impulse response of idealised two-way system with 3rd order Butterworth filtering and the units operating out of phase.
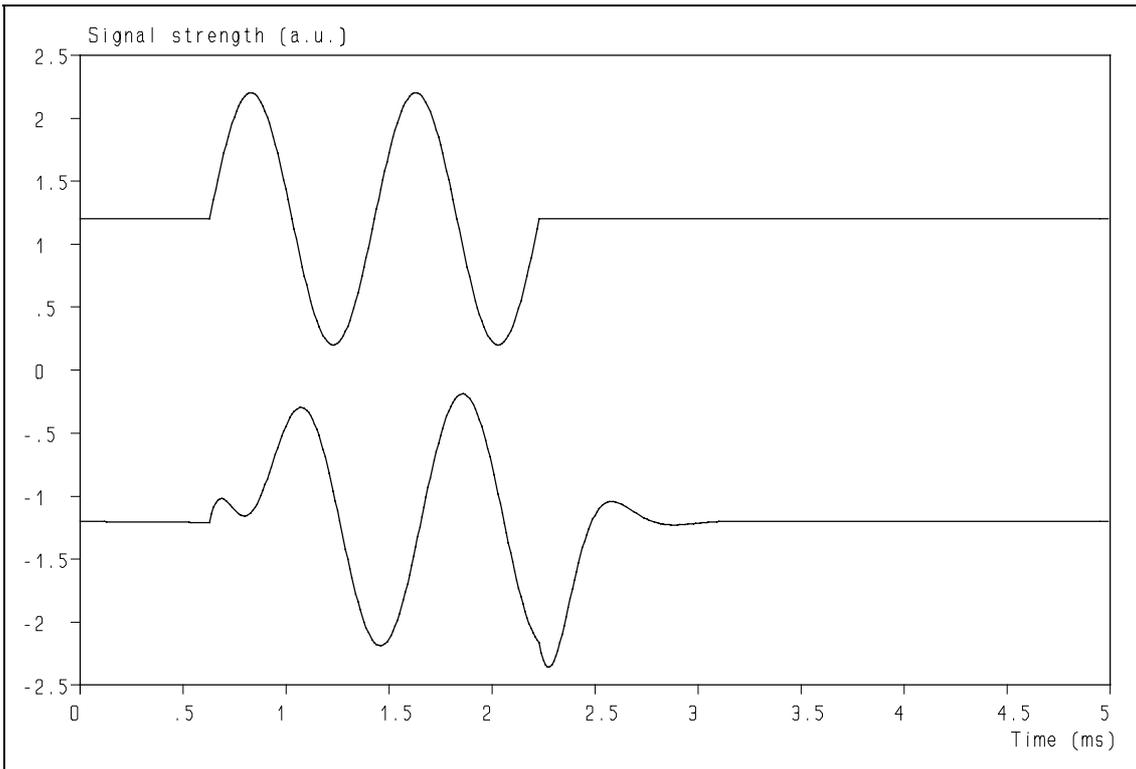
Figure 9: Response of the system to a two-cycle tone burst of 1250 Hz when the units are connected in-phase.
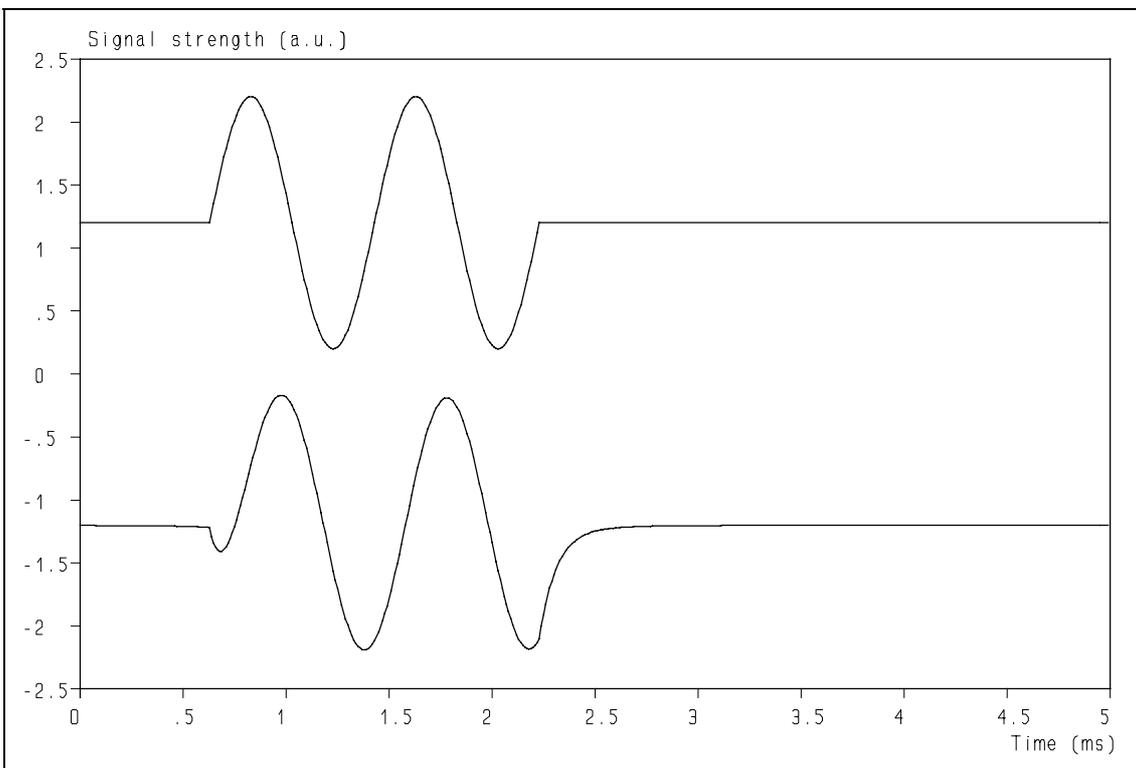
Figure 10: Response of the system to a two-cycle tone burst of 1250 Hz when the units are connected out-of-phase.