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The Audibility of Loudspeaker Phase Distortion

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The Audibility of Loudspeaker Phase Distortion

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Abstract

Digital equalisation of the loudspeaker transfer function enables separation of amplitude and phase response correction processes. Since dynamic loudspeakers exhibit in general a phase response having complicated form, the equalisation process allows for the independent audition of phase distortion that is dissimilar from the responses of simple all-pass filters. Preliminary results are presented, derived using real time DSP operating to full CD format.

List of symbols

a_j	feedback coefficients
b_i	feedforward coefficients
d_1, d_2	transfer function pole locations
$H(z)$	transfer function
$H_{ap}(z)$	all-pass function
$H_{min}(z)$	minimum phase function
M	number of feedback delays
N	number of feedforward delays
n_e	excess phase zero location
n_m	minimum phase zero location
z	z-transform variable

0 Introduction

Much work on the audibility of phase distortion has appeared in the audio press over recent years and with exception of a few papers, for example Klipsch's [1], most conclude that under controlled listening conditions and with restricted types of musical signals even mild phase distortions can be detected. However this is not the end of the story; as even those papers which recognised the audibility of phase distortion, in particular Lipshitz et al. [2], acknowledge that the simple tests so far performed do not provide sufficiently general results to form any firm conclusions apropos phase distortion in "real" situations. This is particularly the case for loudspeaker systems which, despite the fact they generate the most significant phase distortion, are typically placed in semi-reverberant environments thus masking out phase effects. Similarly loudspeakers exhibit notoriously poor amplitude responses (even the highest quality electrostatic speakers which display remarkable impulse responses can have significant amplitude deviations throughout the audio band) which may further mask phase effects, putting into question the reliability of phase audibility tests performed on standard loudspeaker systems. A further aspect which, to the authors' knowledge, seems to have been neglected is the effect of phase distortion on stereophonic reproduction, where both channels exhibit the same phase response. The question, should one go to great lengths to reduce phase distortion in loudspeakers ?, therefore remains unanswered.

Some of the experiments previously reported regarding loudspeaker phase distortion suffered serious limitations due mainly to the technology available at the time. To highlight the point consider the following scenario: in order to assess the phase distortion contributed by a loudspeaker it is necessary to compare the loudspeaker with and without phase distortion. This requires that there must be phase equalisation such that the loudspeaker system modifies the phase linearly with frequency, ie. a pure time delay. Achieving this requirement is not a particularly practical task when using traditional analogue techniques as most loudspeaker systems are non-minimum phase. Thus some of the previous approaches attempted to make headway into the problem by adding extra excess phase distortion using all-pass networks, see for example Lipshitz et al. or Fincham [3]. This approach tells us to what extent increased phase distortion becomes audible which, although a valuable measure, says little about the audibility of the loudspeaker's intrinsic phase response. Other methods, eg. Hansen et al. [4,5], involve the manipulation of electronic signals by alternative means, but similarly do not cater for the existing distortion within the transducer.

These limitations have now become largely redundant owing to digital signal processors which can perform powerful frequency response manipulations in real time. Processes, reported by Mourjopoulos [6] for example, permit the transfer function of loudspeaker systems to be equalised to the extent where the response approximates unity gain with a pure time delay (this response is usually based on an on-axis measurement). A strategy can be adopted where a reference phase linear system is compared to a similar, but non linear phase system. This solution usually involves cascading an all-pass filter to the equaliser, providing the non-linear phase system.

A previous paper by the authors' [7] discussed an equalisation method that tackles the problem from a different angle. In [7] the minimum and excess phase components of a loudspeaker's transfer function are identified and equalised individually. This approach has two beneficial aspects: firstly the minimum phase equalisation corrects any amplitude deviations and in so doing alleviates the earlier mentioned problem of amplitude masking; secondly with minimum phase equalisation, we are left with the loudspeaker's true excess phase response, not an artificial one contrived by mathematically generated all-pass networks, giving a more realistic insight to the audibility of a loudspeaker's inherent phase distortion. The last point seems to be of increasing importance to loudspeaker designers as it is now becoming possible to achieve very flat amplitude responses, either by good mechanical design or external equalisation, whilst leaving essentially an all-pass response (within much of the audible spectrum).

The main theme of the paper is an account of an experiment performed using the ideas introduced in the previous paragraph. A synopsis of the equaliser derivation process is given in section 1. which serves to give an insight into how the two equalisers will be utilised in the overall scheme. The complete equaliser used in the experiment is implemented by four independently operable TMS320C25 DSP devices, the minimum and excess phase equalisers each requiring one per channel. The equaliser is used in conjunction with a two box compact disc player system (CD player and outboard DAC) operating through the AES/EBU interfaces to full stereo CD format. The sections from 2. onwards describe the experiment giving testing methodology and an in depth discussion of the results.

1 Method of loudspeaker equalisation

This section is intended to give an outline of the processes involved in the formation of the minimum and excess phase equalisers. For a more detailed description the reader is referred to Greenfield et al. [7].

The first step in the process is to form an Infinite Impulse Response (IIR) model of the loudspeakers transfer characteristic. The modelling process uses the Least Mean Squares algorithm to find the feedback coefficients and from which, the feedforward ones are found deterministically. This approach can generate models of order up to about 60, permitting very precise equalisation. The model effectively derives a polynomial in the 'z' domain of the form

$$H(z) = \frac{\sum_{i=0}^{N} b_i . z^{-i}}{1 - \sum_{j=1}^{M} a_j . z^{-j}} \qquad 1.$$

After locating the pole and zero locations of 1. (equations 2. and 3 give an example of a second order non-minimum phase function) the excess phase zeros are found by inspection (excess phase zero locations are outside the unit circle), from which point the minimum phase and all-pass functions which make up the non-minimum phase transfer function, can be calculated.

$$H(z) = \frac{(z-n_m) \cdot (z-n_e)}{(z-d_1) \cdot (z-d_2)} \qquad 2.$$

$$H(z) = H_{min}(z) \qquad H_{ap}(z)$$

$$H(z) = \frac{(z-n_m) \cdot (z.n_e^*-1)}{(z-d_1) \cdot (z-d_2)} \cdot \frac{(z-n_e)}{(z.n_e^*-1)} \qquad 3.$$

The minimum phase equaliser is now derived from the inverse of the minimum phase component of 3. which is simply its reciprocal. This equaliser when used in conjunction with the specified loudspeaker will result in the loudspeakers response being all-pass in nature. The excess phase equaliser is a tapped delay line (FIR filter) derived from the time reversed impulse response of the all-pass component of 3.

2 Phase audibility listening tests.

The tests were carried out using the Celestion SL700 series loudspeakers. These were chosen as they have a good frequency response (both in amplitude and phase), where pairs are extremely well matched allowing the same equalisation on both channels. Further its seems likely that phase distortion is only going to be a serious consideration in the higher end of the Hi-Fi market in which case the SL700's are fairly representative. Response plots of these speakers are shown in fig 1. Fig 2. and fig 3 give the measured responses of the completely equalised system and the minimum phase equalised system respectively. Note the amplitude response remains the same in both cases whilst the phase and therefore the impulse response differ significantly. The frequency plots only go down to 100Hz owing to the measurement environment rendering poor accuracy below this frequency.

The auditions were carried out in the listening room at Celestion International, Ipswich, with five experienced listeners, three of whom are used to voice Celestion's loudspeakers. At the suggestion of Celestion the tests were carried out in the following manner. A piece of music was played with the phase equaliser set on or off (called setting A), the listener being unaware of the setting. The piece of music was replayed, now with the reverse setting (called setting B). After the two runs the listener could request another hearing stating which he would prefer next, A or B. At the end of the test the listener was asked to make comments on what he heard ie. was there an audible difference and if so what it was. The tests were carried out in this fashion because the listeners used are experienced at loudspeaker assessment where it is not practical to simply switch pairs half way through a piece of music and are therefore better acquainted with this form of listening comparison.

In the experiment two pieces of music were used, the first being "Julsang", track 9 from Cantate Domino (proprius PRCD

7762). This is a basically a vocal piece with both solo and choral interludes which contains good spatial clues giving both the perception of breadth and depth. The other piece was "There goes my baby", track 2 from Joe Cocker's Civilised man (EMI CDP 746038 2) which contains a wide variety of both musical and vocal sounds ranging from quite simple guitar chords to large ensembles of vocals and instruments.

3 Results of listening tests

At this point we will present only a summary of the reactions and comments made by the five listeners with discussion on these following in the next section. With each person the starting condition, which we will call A, of the excess phase equaliser is decided by the toss of a coin before the subject entered the room.

First person: (A represents excess phase equalised version)
Within 30 seconds of hearing the second playing the Cantate Domino track he exclaimed that there was no comparison between the two renditions. At the end of the hearing his comments were: A was more open, better stereo presentation, solid bass; B sounded more like loudspeakers. With the Joe Cocker track he said A was more open than B but the difference was less obvious. His overall preference was for A.

Second person: (B represents excess phase equalised version)
About the Cantate Domino track: A was more open on the choir/voices; B was more muddled, less impact, less power. With the Joe Cocker track he could detect no difference. His overall preference was for B.

Third person: (A represents excess phase equalised version)
About the Cantate Domino track: A seemed more harsh, sibilant; B

had less content at high frequency, more pure on vocals. With the Joe Cocker track he said the bass drums sounded more natural on B. His overall preference was for B.

Forth Person: (A represent excess phase equalised version)
About the Cantate Domino track: with B the acoustical size of objects seemed smaller, more defined in space, individual voices in the choir could be picked out; with A the image is less defined. With the Joe Cocker track the differences were less apparent B seemed more clinical ie. the image is centered at discrete points, left/right/center. A gives better stereo occupying a broader space either side of center. He did not express an overall preference.

Fifth person: (B is excess phase equalised version)
About the Cantate Domino track: A was more solid, more weight, more coherent picture; B was a more of a wash of sound, a bit smoother. He could not detect a difference with the Joe Cocker track. His overall preference was for A.

4 Discussion of results

It must be stated from the outset that although every body included in the experiment noted some differences between the equalisations, that these were very subtle effects indeed. Apart from person three the general consensus was that the changes were broadly of a spatial nature, although the actual interpretations offered by the listeners differed. This could be put down to use of language or it could be that the listeners did in fact have differing perceptions. To the authors' the general impression implied by the listeners regarding the excess phase equalised version seemed to be that the sound stage became broader, that is, the sound is no longer coming from two distinct sources but is distributed between them making images less focused or defined in space. This is almost the antipathies of what was expected, as one would have thought that by sharpening up the impulse response, the images would actually become better focused. A second interesting point to note was the effect the type of music had on the audibility. Again one may have thought that sounds with fast transients such as drums and symbols, would highlight the time dispersive property of phase distortion, yet it was the smoother choral track in which the differences were most apparent. This could be a consequence of the human brain not being capable locking on to, or concentrating on certain types of sound. Alternatively, the processes through which the sound has been generated could have under gone any amount of phase distortion, thus making this form of equalisation redundant as its simply modifies the distortion and not eliminates it.

A further point worth considering is the mechanism by which the brain perceived the changes in phase; is the human brain actually perceiving a phase change, or an amplitude change caused by loudspeaker non-linearities ?. Moving coil loudspeakers typically undergo non-linear distortions dependent on cone displacement. Mills [8] discusses the various mechanisms from which the distortions are born and gives plots of the effected parameters for the Celestion SL600 mid-range drive unit. Fig. 4 is a copy of these plots which should be fairly representative of the distortions encountered here as the SL700 and SL600 mid-range units are quite similar in construction. Consider now the effect of non-linearities on the loudspeakers response, the introduction of the excess phase equaliser to the signal path will alter the wave form appearing at the loudspeaker terminals thus exciting a different non-linear distortion, resulting in a slightly different output response. In a stereo system where the wave form going into each speaker contains different information, the modifications (which will be different owing to the non-linearities) to each signal caused by the equaliser could possibly result in a perceived change in image location, or indeed some other effect.

A secondary issue worth mentioning is the effective modification of the signal crest factor caused by the excess phase equaliser. The input signal is essentially convolved with the equaliser's impulse response causing a smearing of the signal, which will in most cases reduce the crest factor of the signal. This has beneficial aspects regarding amplifier power requirements (and possibly linearity as well), however it has exactly the opposite effect for loudspeakers where the output crest factor is now increased, thus driving the cone harder and into more non-linear regions of its operation. This too could have significant bearing on any perceived phase changes.

5 Conclusion

The aim of this paper was to describe a method of assessing the importance of phase distortion in loudspeakers systems by auditioning the actual phase distortion inherent in the loudspeaker system. This task was accomplished by an equalisation strategy which distinguished the excess phase response from the minimum phase response, thereby providing separate equalisation of the two. The minimum phase equaliser compensates for both magnitude and minimum phase irregularities whilst the excess phase equaliser is purely all-pass, resulting in a phase linear response when used with in conjunction with the former. Using a real time system working to full stereo CD format listening tests were performed with five experienced listeners. The results of these tests showed clearly that the addition of the excess phase equaliser had an effect on the perceived sound, albeit a very subtle one. However the nature of the distortion mechanisms and type of program material bring into question the basis of the perceived differences. It is the authors' (it is with much trepidation that they put it in writing) belief that, based on the results obtained here, in a stereo system with good quality equipment the apparent differences are perceived phase changes rather than amplitude deviations brought about by loudspeaker non-linearities. This view is arrived at from two different sources: firstly when measured by both white noise and impulse techniques, which are very different signals, the measured amplitude response remains the same regardless of the phase equaliser; secondly the nature of the apparent change seems inconsistent with non-linearity school of thought. If there was a tonal difference or some other ambiguous difference then this could be put down to non-linear distortion, however the general opinion was that it was the apparent sound stage that was modified, which the authors' cannot justify by this approach.

The question as to whether or not phase equalisation should be included in loudspeaker design remains to be answered. With most recordings which have undergone many phase modifying electronic processes, both analogue and digital, the audibility of phase differences is immaterial as one is simply detecting a change in phase distortion and not a correction of it and as such preferences would most likely be personal. However one can envisage the situation where the entire recording through to reproduction chain remains phase linear, in which case the results obtained here suggest the phase response of loudspeaker systems is of importance. With digital processors becoming widely available and ever more powerful a single equalisation scheme from microphone through to disc is not unrealisable, and indeed most desirable. With this scenario, a similar equalisation scheme for the complete reproduction system, from disc through to loudspeaker, must be greatly advocated.

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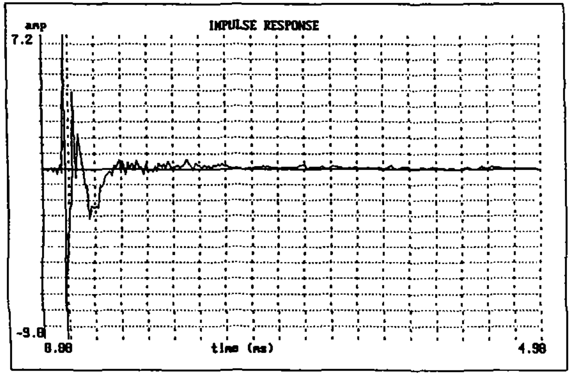
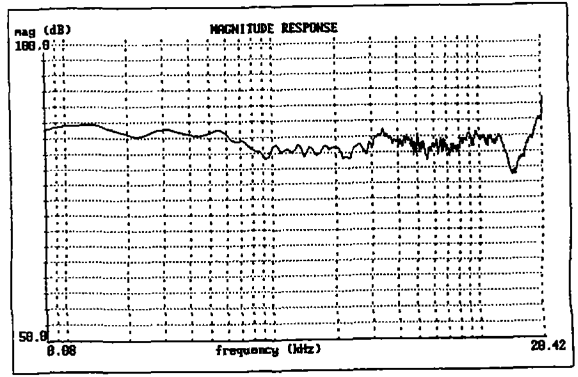




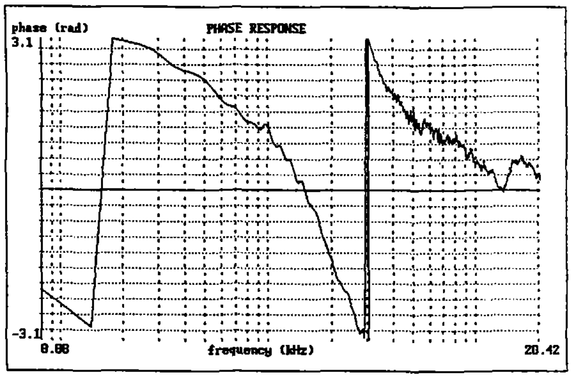
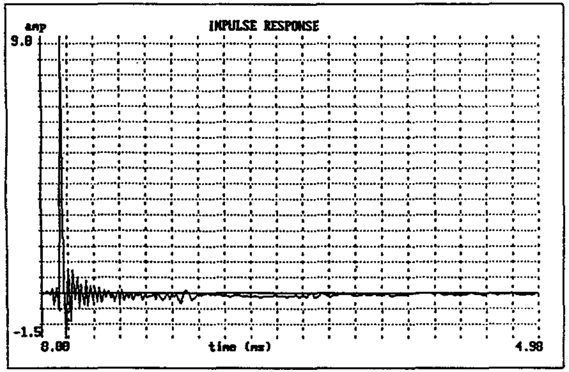
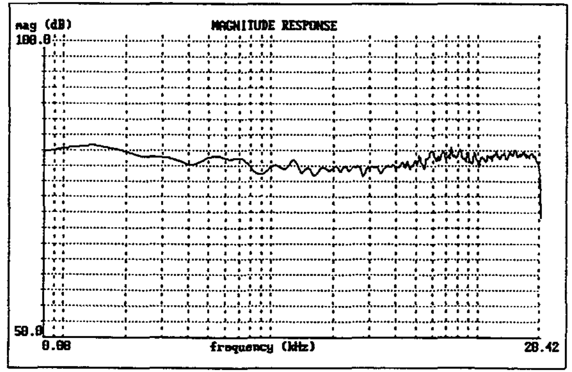


Figure 1: Measured responses of the SL700.





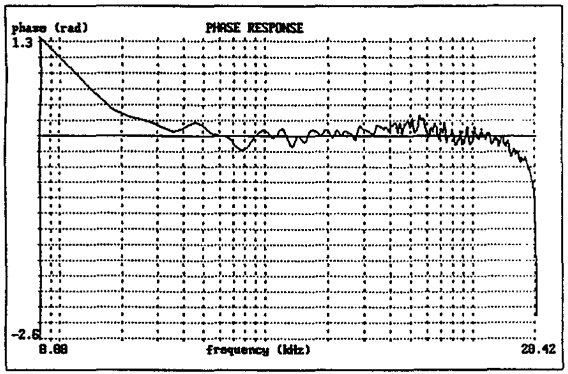
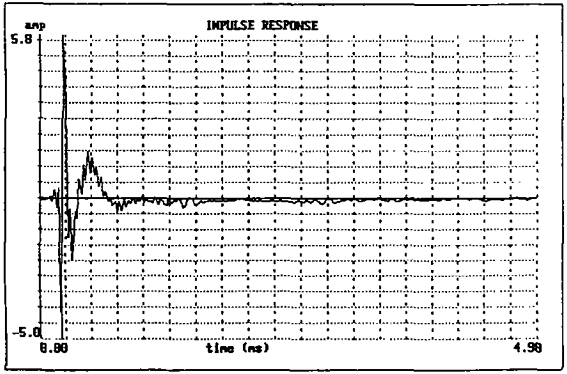
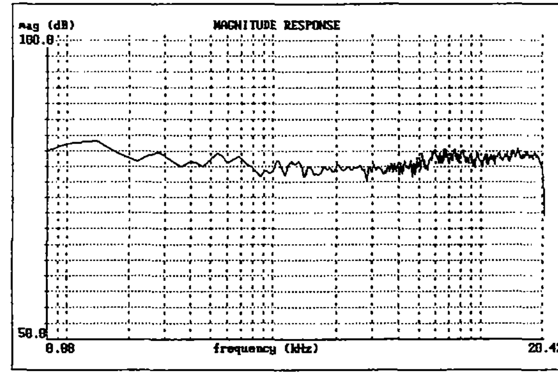


Figure 2: Measured responses of the SL700 with both magnitude and excess phase equalisation.





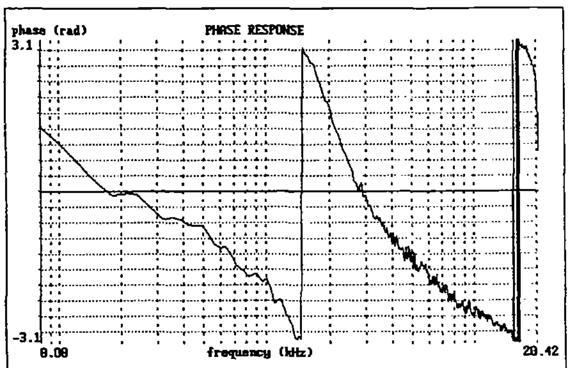
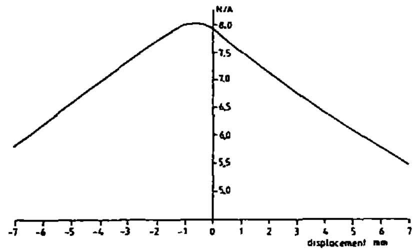
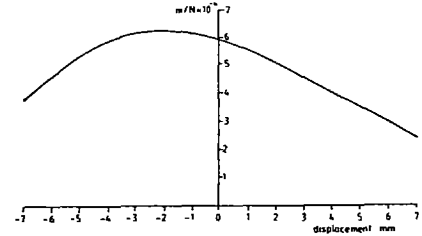


Figure 3: Measured responses of the SL700 with minimum phase (amplitude) equalisation.

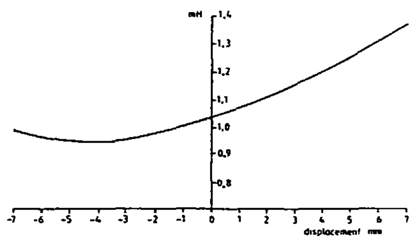


(a)



(b)

(b)



(c)

Figure 4: Variation of loudspeaker parameters with displacement. (a) Bl product. (b) Mechanical compliance. (c) Electric coil inductance.