The Subjective Importance of Uniform Group Delay at Low Frequencies*

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Analog recordings always have high group delay at low frequencies due to the combined effects of all the components in the record/replay chain, and in particular the analog recorder. Digital recorders now make it practical to remove much of this group delay. It is discussed whether it is worthwhile to produce a record/replay chain having uniform group delay down to the lowest audible frequencies.

0 INTRODUCTION

Distortionless transmission through a linear system requires that the amplitude/frequency characteristic $|H(\omega)|$ be flat and that the phase response $\phi(\omega)$ be proportional to frequency, that is, $\phi(\omega) = \phi T$. When the phase response is not linear, phase distortion occurs, and one measure of this is the group delay τ_g,

$$\tau_g = -\frac{d\phi(\omega)}{d\omega}$$

The deviation of the group delay from a constant value in the system passband has been called *group delay distortion*, $\Delta\tau_g(\omega)$ [1],

$$\Delta\tau_g(\omega) = \tau_g(\omega) - T$$

where T is the frequency-independent delay in the system.

Linear systems may be divided into two categories, minimum phase-shift and nonminimum phase-shift systems. Minimum phase-shift systems have the least amount of phase shift for their given amplitude response. Nonminimum phase-shift systems can be represented by a minimum phase-shift system in cascade with an all-pass system. Group delay distortion may be due to the minimum phase response and the frequency-dependent all-pass phase response of a system [1], [2].

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1 RECORDING AND REPRODUCING CHAIN

All parts of the recording and reproducing chain, from the microphone to the loudspeaker, do in general contribute to the overall amplitude and phase distortion. Most components exhibit minimum phase behavior—notable exceptions are analog tape recorders and most multiway dynamic loudspeaker systems.

A simulation of the by no means untypical amplitude and phase distortion in present-day recording and reproducing chains is shown in Fig. 1. Note in particular the rapid change in phase near the lower cutoff frequency at about 40–50 Hz. In an attempt to assess the subjective importance of phase distortion at low frequencies, it is obviously prudent not to use program material that has already been grossly phase distorted. The following procedures were therefore adopted in order to minimize phase distortion at low frequencies in both the recorded program and the playback system.

1.1 Recording Chain

For recording purposes a Bruel & Kjaer ½-in (12.7-mm) pressure microphone, type 4133, was used together with a home-made low-noise microphone amplifier and a Sony digital recording system consisting of a PCM F1 digital audio processor and an SL2000 video cassette recorder.

1.2 Reproducing Chain

The loudspeaker used for subjective evaluation was fed from a dc-coupled power amplifier—the low-frequency phase distortion in the loudspeaker, due to its high-pass characteristic, being corrected by means of a simple minimum phase-shift equalizer. The amplitude and phase characteristics of the closed-box loudspeaker after equalization corresponded to that of a second-order high-pass filter having a cutoff frequency of 5 Hz.

The transfer function $H(f)$ of the loudspeaker equalizer is given by

$$H(f) = \frac{f^2 + f \cdot f_0/Q_0 + f_0^2}{f^2 + f \cdot f_1/Q_1 + f_1^2} \quad (1)$$

where f_0 and Q_0 are the resonance frequency and the total Q of the closed-box system, and f_1 and Q_1 describe the low-frequency characteristics of the system after equalization.

Examples of suitable equalizers have been described by Linkwitz [3] and Greiner and Schoessow [4], but these were not used for these experiments due to their inflexibility. Instead, a special equalizer has been devised in which Q_1 and Q_0 can be varied in a continuous manner, and f_0 can be adjusted in increments of 1 Hz and f_1, the lower cutoff frequency, in third-octave intervals from 5 to 50 Hz. A bypass switch has been incorporated into the equalizer, thus allowing simple comparison to be made between the equalized and the unequalized loudspeakers. The frequency response of the complete recording/reproducing chain is shown in Fig. 2, from which it can be seen that both amplitude and phase are substantially flat to below 20Hz.

2 PROGRAM

Initial experiments, carried out using program material recorded using the equipment described earlier, suggested that *lowering* the cutoff frequency produced the subjective effect of *less* bass. This at first surprising result may be due to the fact that the ear is assessing the overall bass response more in terms of the frequency-dependent time delay at low frequencies, which imparts a "boomy" characteristic to the sound, rather than on spectral content. This effect was even noticeable on male speech, which contains very little energy below 100 Hz—the equalized system being noticeably free from the "chesty" sound so often present on recorded male speech.

To assess the effect of phase equalization of the loudspeaker upon orchestral music, a recording was made of a large symphony orchestra in a hall having a good acoustic with the B&K 4133 pressure microphone raised 12 ft (3.66 m) above the ground and situated some 12 ft (3.66 m) behind the conductor. A calibrated tone, corresponding to 94 dB sound pressure level at the microphone, was included at the beginning of the recording to enable the replay level to be set to that of the original program.

The phase-equalizing circuit effectively applied a bass boost of nearly 40 dB at 5 Hz. It might be thought, therefore, that the loudspeaker would be quite unable to reproduce extremely low frequencies at lifelike levels due to mechanical overload. Surprisingly this was not the case when replaying this particular item, even though it contained very loud tympani passages. The reason for this must be that the naturally occurring sounds, although containing spectral components down to very low frequencies, have these components progressively attenuated below 100 Hz. Fortuitously it appears that the natural rate of rolloff is quite close to the inverse of the low-frequency boost circuit, thus avoiding overload of the system.

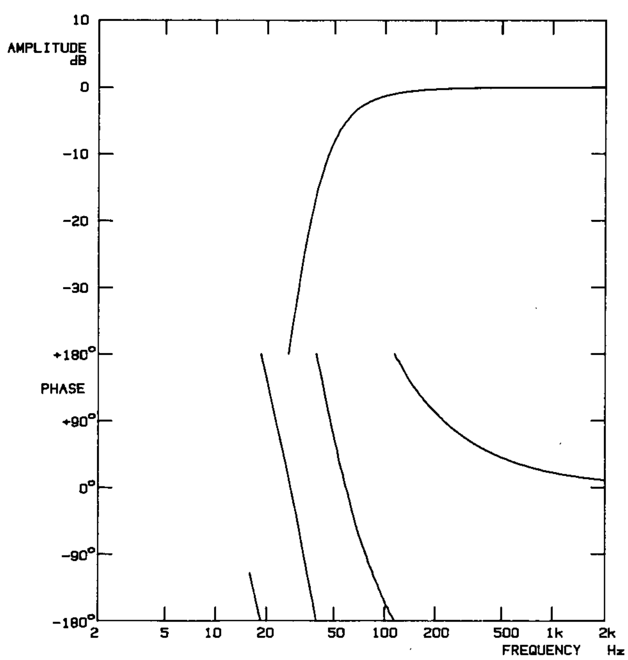

Fig. 1 Frequency response of typical analog recording/reproducing chain.

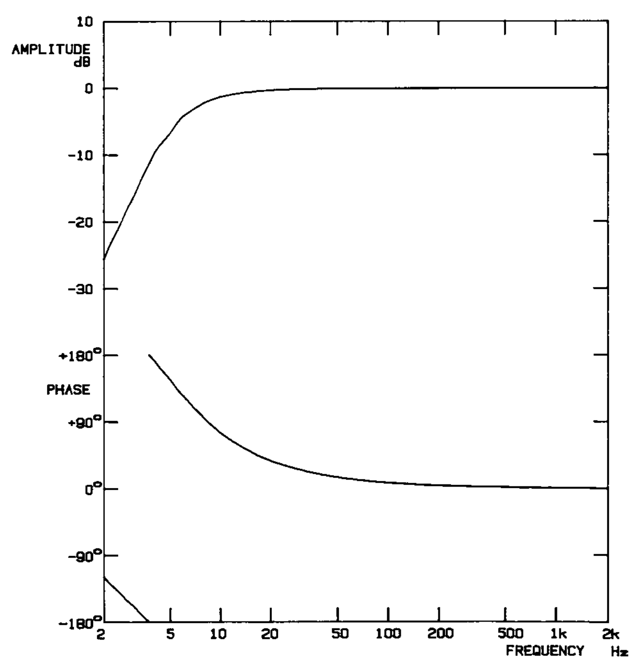

Fig. 2. Frequency response of experimental recording/reproducing chain.

It is worth noting in passing that attempts to replay normal analog recordings on the equalized loudspeaker system were a total failure, due primarily to the presence of very-high-level nonmusical signals below 20 Hz. These signal components were due mostly to record warp and rumble, both of which were exaggerated by the presence of the usual subsonic resonances in the pickup arm/stylus combination, which normally occurs between 8 and 12 Hz and can have a Q of up to 10.

Other low-frequency fault signals on the analog recordings may include those caused by air-conditioning noise, which often remain undetected during the recording process, possibly due to the restricted low-frequency response of most studio monitoring loudspeakers currently in use. This observation would tend to suggest that even if phase equalization at low frequencies were important, the minimum phase-shift equalizer approach would be impractical for the bulk of recorded material intended for replay on analog systems.

In the future, of course, there is no reason why phase equalization should not be achieved by means of a digital filter which would be capable of attenuating nonmusical signals below the audio band, while at the same time maintaining uniform group delay in the passband (Fig. 3). The results so far demonstrate that it is practical, using digital recording techniques, to record and replay live sounds that do not have any significant amplitude or phase distortion down to the lowest audible frequency.

Listening tests in carefully controlled conditions indicate that a reduction in group delay distortion at low frequencies in the replay chain is probably worthwhile only when the recorded material is itself also free from such distortion. The effects, however, are quite subtle, and in an attempt to exaggerate these for the purposes of public demonstration, a test signal consisting of eight cycles of a 40-Hz tone burst was used instead of musical program. The experiment consisted of inserting into the record/replay chain, which was flat down to very low frequencies, an all-pass filter whose phase characteristic was such that it added significant group delay at around 40–50 Hz, this region corresponding approximately to the likely frequency range of maximum group delay distortion in the typical chain shown in Fig. 1.

The circuit chosen for the all-pass network was that used by Lipshitz, Pocock, and Vanderkooy [2] and consisted of two second-order all-pass sections in cascade, each with a Q of $\sqrt{2}$. Switching this all-pass circuit in and out caused distinct audible differences in sound quality to be observed by most of the audience in a typical lecture theater when the excitation was the tone burst described earlier. This result suggests that the addition of excess phase distortion or group delay is audible even when it is not accompanied by a corresponding change in the amplitude response as would be the case with a minimum phase-shift system.

The complementary, but probably more important, experiment, namely, that of maintaining a flat phase characteristic and therefore a uniform group delay to below 20 Hz while allowing the amplitude response to vary, has not so far been carried out due to the instrumental complexity involved. The appropriate characteristics are most readily achieved through the use of digital filtering, and future experiments will utilize this technique.

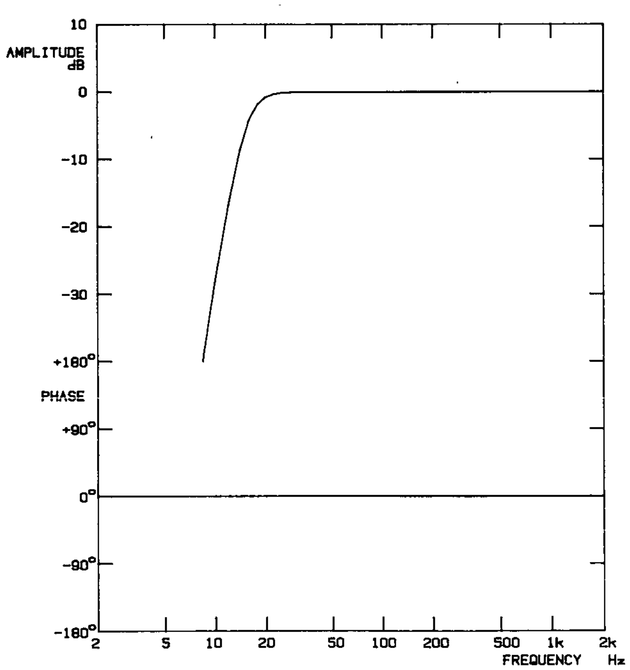

Fig. 3. Frequency response of proposed digital high-pass filter having uniform amplitude and uniform phase above 20 Hz.

3 CONCLUSIONS

Various simple techniques for producing a recording and reproducing chain having minimum group delay distortion at low frequencies have been described. Controlled listening tests indicate that group delay distortion at low frequencies can produce subtle but clearly audible changes in sound quality. More work is obviously needed so that the perceptual thresholds for group delay distortion can be established. It already seems likely, however, that if lifelike low-frequency reproduction is to be obtained, then the entire recording and reproducing chain may require flat amplitude and phase characteristics down to the lower limits of audibility.

Now that digital recorders make it practical for the first time to eliminate much of the low-frequency group delay distortion, it is perhaps time to implement changes in design to remove the group delay distortion that still remains in the rest of the record/replay chain. The current practice of inserting, for good, sound, practical reasons, admittedly, high-pass filters at various parts of the recording chain in order to remove problems caused by background noise in the studio, accidental thumps due to kicking of the microphone stand, slamming of doors, and so on, should be reviewed. Digital filtering provides an obvious means of eliminating all these types of fault signals without exceeding the perceptual thresholds of group delay distortion at low frequencies.

4 ACKNOWLEDGMENT

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Laurie Fincham's biography was published in the March issue.