The audibility of Doppler distortion in loudspeakers

Edgar Villchur

Foundation for Hearing Aid Research, Woodstock, New York 12498

Roy. F. Allison

Allison Acoustics Inc., 7 Tech Circle, Natick, Massachusetts 01760

(Received 23 April 1980; accepted for publication 1 September 1980)

The existence of Doppler distortion in loudspeakers was demonstrated more than 30 years ago, but the question of whether or not such distortion is of significance in music reproduction has not been settled. In this study the audibility of Doppler distortion in simple direct radiators is investigated both theoretically and experimentally through (1) an analogy to tape-machine flutter, (2) an analysis of the effect of normal listening-room conditions on Doppler distortion (for a given low-frequency power the radiation-load and reverberant-field correction factors typically reduce this distortion to 0.36 of its value on the axis of a speaker radiating into anechoic half space), and (3) double-blind listening tests. An eight-member jury compared music reproduced in two modes, one of which was essentially free of Doppler distortion. In one experiment, two recordings were made of the output of a speaker, one with a sharp bass-attenuation filter in the speaker input circuit and the other with the same filter moved to the microphone circuit. In a second experiment, on-axis and off-axis speaker output were recorded in an anechoic chamber, the speaker input circuits being compensated to make the frequency responses in the two modes equal. The theoretical analysis indicates that for any practical cone velocity Doppler distortion will be less audible by an order of magnitude than the flutter of a 15-ips tape machine that meets the NAB Standard. The experimental results provided confirming evidence that it is inaudible.

PACS numbers: 43.85.Ry, 43.88.Ja

INTRODUCTION

When a loudspeaker radiates both high- and low-frequency signals simultaneously, the higher frequency is shifted upward by Doppler effect as the cone advances toward the observer during one-half of the low-frequency excursion, and shifted downward as the cone retreats during the other half. This frequency modulation creates inharmonic sidebands. A direct-radiator speaker unaided by a horn or by vent loading would have the worst of it, because it requires the greatest low-frequency cone velocity for a given acoustic output.

The existence of frequency-modulation distortion in speakers was demonstrated by Beers and Belar[1] more than thirty years ago, but the question of whether or not such distortion is of significance in sound reproduction has not been settled. Evidence relating to the significance of Doppler distortion in speakers has been derived from an analogy to tape-machine flutter, as discussed by Klipsch,[2,3] from the calculation and/or measurement of distortion percentage as the relative amplitude of frequency-modulation sidebands, as reported by Klipsch[4] and by Moir,[5] and from listening tests, as reported by Klipsch,[3,4] Moir,[5] Kurtin,[6] and Fryer.[7] Moir expresses the belief that FM distortion is the predominant residual distortion in simple direct-radiator speakers, while Klipsch writes that Doppler distortion from such direct radiators may be the major remaining defect in the entire reproducing system.

I. THE ANALOGY TO FLUTTER

Flutter in tape machines or record players, described in terms of percentage and rate of frequency deviation, has been studied sufficiently for standards of permissible flutter in professional recording and reproducing equipment to have been written. The audibility of specific percentages and frequencies of FM-distortion sidebands, however, has not been established for music reproduction. Thus the analogy between Doppler effect in speakers and flutter as it is measured in tape machines, although it must be qualified as discussed below, provides more meaningful evidence bearing on the audibility of Doppler than the calculation or measurement of FM sidebands.

The Doppler frequency shift of sound radiated on axis from a loudspeaker is approximately equal to the cone velocity at the modulating frequency divided by the velocity of sound. More rigorously, $\Delta f_2(\%) = [u_c/(c - u_c)](100)$ (see Appendix A), where Δf_2 = percent frequency shift of the modulated signal, u_c = cone velocity at the modulating frequency, in in./s, c = velocity of sound = 13527 in./s at 20°C. Since the cone velocity is very small compared to the velocity of sound, we will use the simpler expression

$$\Delta f_2(\%) = (u_c/c)(100). \quad (1)$$

The NAB standard for permissible flutter in tape machines[8] specifies that the flutter meter be calibrated to read the rms value of a sinusoidal frequency variation. We therefore use the rms value of cone velocity, calculated as the cone travel during one period divided by the period (which yields the average value), multiplied by 1.11 (which converts average to rms values for sinusoidal variations).[9]

$$u_c(\text{rms}) = (2x/T)(1.11) = 2.22xf_1, \quad (2)$$

where x = peak-to-peak cone travel at the modulating frequency (in inches), T = period (time in seconds of one cycle) of the modulating signal, f_1 = frequency of the modulating signal in Hz. Substituting $2.22xf_1$ for u_c in Eq. (1), the on-axis Doppler frequency shift may be

 0001-4966/80/121561-09$00.80

expressed as

$$\Delta f_2(\%) = \frac{2.22 x f_1}{13\,527}\,(100) = 0.016 x f_1 . \qquad (3)$$

Comerci[10] and others demonstrated that the perceptibility of flutter in music reproduction is strongly dependent on the flutter rate. In recognition of this phenomenon the NAB standard for tape machines, in common with other American and international standards, requires that flutter measurements be weighted with respect to the modulating frequency. Table I shows the amounts by which measured raw flutter percentages must be reduced, relative to flutter at a 4-Hz rate, for modulating frequencies up to 200 Hz. In the case of tape machines the weighting is built into the flutter-measuring network. For calculating weighted Doppler frequency shift the reduction factors must be applied to the calculated Δf. The on-axis weighted Doppler shift Δf_w thus becomes

$$\Delta f_w(\%) = 0.016 x f_1 W, \qquad (4)$$

where W=frequency weighting factor from Table I.

The most stringent NAB standard for tape machines allows 0.05% rms weighted flutter at 15 ips when the machine is reproducing an essentially flutter-free test tape, and 0.07% at $7\frac{1}{2}$ ips. The peak-to-peak cone travel allowable in a loudspeaker before the on-axis Doppler shift in frequency exceeds the 15-ips weighted flutter standard is readily calculable from Eq. (4) as slightly more than $\frac{1}{4}$ in. between 30 and 60 Hz, decreasing to 0.23 in. at 100 Hz. The linear peak-to-peak cone travel of a good loudspeaker is typically of the order of $\frac{1}{2}$ in. at frequencies below 40 Hz.

The excursion values calculated on the basis of Eq. (4) do not, however, represent allowable cone travel established by the constraints of Doppler distortion, because there are important limitations to the analogy between Doppler effect and flutter that favor the loudspeaker. In a tape machine or record player the entire frequency spectrum is subject to modulation at the flutter rate. In a speaker system only that part of the spectrum reproduced by the woofer is subject to significant FM, as Beers and Belar pointed out. Doppler effect in high-frequency speakers is small because of the small cone velocities. Beers and Belar also showed that the magnitude of FM sidebands varies directly with the higher, modulated frequency. A crossover frequency of 1 kHz in a two-unit speaker system thus limits the maximum FM sideband distortion to a small fraction of what it would be in a tape machine whose percentage of flutter was equal to the percentage of Doppler frequency shift. For recorded signals above 1 kHz, the fraction is equal to 1 kHz divided by the frequency of the signal being modulated. While the relative audibility of higher-frequency sidebands (compared to sidebands of lower frequency) has not been determined, it is reasonable to assume, as did Beers and Belar, that it is better to eliminate the higher-frequency sidebands than to leave them in the signal.

TABLE I. NAB weighting factors for flutter measurements.

Flutter frequency in Hz	Weighting factor (W) *re* 4 Hz
4	1.0
10	0.79
30	0.37
40	0.30
50	0.25
60	0.21
100	0.14
200	0.07

Another limitation to the analogy is that Doppler FM sidebands are accompanied by low-frequency tones generated by the same cone excursions that create the Doppler. These tones have a masking effect on the sidebands, through upward spread. The mechanical irregularities of a tape machine or record player, on the other hand, generate flutter without creating low-frequency tones.

A third limitation to the analogy is that tape-machine flutter is compounded: The amount of flutter allowed by the standard must be tolerated twice, once in the recording process and again, on top of the recorded flutter, in reproduction. The two modes of flutter add vectorially.

A fourth limitation to the analogy is that a woofer radiates most of its power off axis, where the Doppler effect is reduced or eliminated. In a normally reverberant room we hear the integrated acoustical power of a speaker more than the on-axis sound pressure, no matter where we sit in relation to the speaker. Frequency-modulation distortion in speakers is thus diluted by reflected off-axis sound, as discussed quantitatively in the next section, while tape-machine flutter modulates all of the recorded signal. However, a reverberant environment also has an opposite effect. Beers and Belar and Comerci have each pointed out that reverberation increases the audibility of a given amount of frequency modulation, because a shifting pattern of standing waves can convert frequency modulation to amplitude modulation. This last factor applies both to Doppler effect and to flutter, unless the tape machine is used with earphones.

When these limitations to the flutter analogy and the frequency-weighting factor are both taken into account, the analogy leads one to expect that for any practical cone excursions, Doppler effect in speakers, including totally enclosed direct radiators, will be less audible by an order of magnitude than the flutter of a 15-ips tape machine that meets the NAB standard.

II. CALCULATION OF FM SIDEBAND DISTORTION

The expression derived by Beers and Belar for frequency-modulation distortion as a function of cone excursion is

$$\mathrm{DF}_0(\%) = 0.033 A_1 f_2 , \qquad (5)$$

where A_1 = center-to-peak cone excursion at the modulating frequency (in inches), f_2 = modulated frequency, DF_0 = FM distortion factor (the square root of the ratio of power in the sidebands to the total power of the f_2 signal and sidebands, expressed as a percent) for radiation on the speaker axis in an anechoic environment.

This expression accurately represents the relative amplitude of the FM sidebands in a signal radiated on the speaker axis in an anechoic chamber. It does not, however, represent the relative amplitude of the sidebands in the reverberant field of a domestic living room, a limitation that Beers and Belar recognized.

At the modulated-frequency (f_2) signals of interest—those carried by the woofer in a multiunit loudspeaker system—the room is a reasonably good integrator of the power radiated in all directions. To calculate the total FM distortion content in the reverberant field it is necessary to evaluate how Doppler modulation varies with the angle to the speaker axis, and then to weight the results in accordance with the relative amount of power radiated at different angles from the axis.

Doppler sidebands can be generated in the signal radiated at any given angle only to the extent that there is a component of cone motion directed at that angle. At 90° from the axis the power radiated directly is not subject to Doppler effect. In Fig. 1, $\mathbf{x}_0$ is the on-axis excursion-amplitude vector of magnitude $\overline{AB}$, and $\mathbf{x}_\Theta$ is the effective-excursion vector for radiation at $\Theta°$, of magnitude $\overline{AC}$.[11] It is evident that $\mathbf{x}_\Theta = \mathbf{x}_0 \cos\Theta$. If DF_Θ is the FM-distortion factor for radiation at $\Theta°$ from the axis, then $\mathrm{DF}_\Theta = \mathrm{DF}_0 \cos\Theta$. The FM-distortion factor at angle Θ is reduced from the on-axis distortion by $\cos\Theta$, because the sideband amplitudes are directly proportional to the excursion-amplitude vector in the direction of radiation.

Very-low-frequency sound energy, for which the wave length is large compared with the woofer-cone diameter, will be propagated omnidirectionally; that is, the intensity is the same at any point on the surface S of the hemisphere shown in Fig. 2. It follows that the acoustic power radiated through any part of this surface is proportional to the area of the section through which it is radiated, regardless of the location or shape of that section.

Since an omnidirectional speaker radiates equal power per unit area of the hemisphere, and since the area projected per degree of angle to the speaker axis increases as the angle increases, the radiated power also increases at larger angles to the speaker axis. This increase is proportional to the sine of the angle,

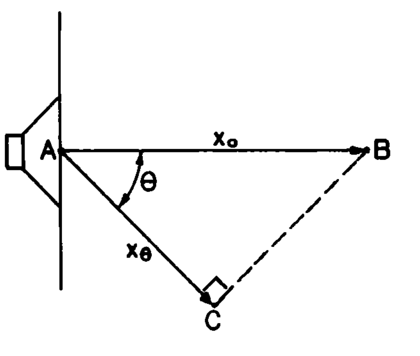

FIG. 1. $\mathbf{x}_0$ is the axial excursion amplitude of the loudspeaker diaphragm at the modulating frequency, and $\mathbf{x}_\theta$ is the effective excursion amplitude for radiation directed at $\theta°$. $\mathbf{x}_\theta = \mathbf{x}_0 \cos\theta$.

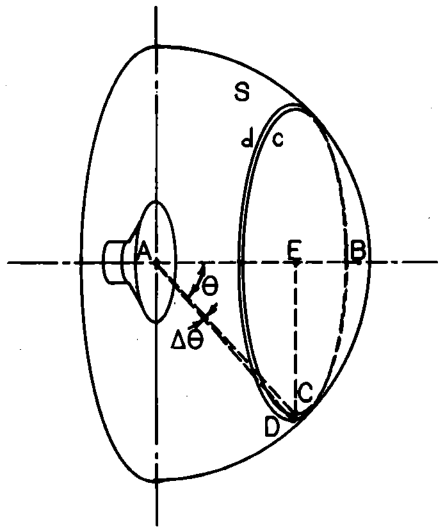

FIG. 2. S is the surface of a hemisphere drawn at radius $r = \overline{AB}$ in front of a loudspeaker system, with the woofer diaphragm at the center A. Power radiated at angle θ is proportional to the area of the ring-shaped segment bounded by circles c and d, which is in turn porportional to $\sin\theta$.

shown as follows: In Fig. 2, $\overline{AB} = \overline{AC} = r$; $CE = r\sin\Theta$ = radius of circle c; circumference of $c = 2\pi r \sin\Theta$. $2\pi r$ is a constant. The circumference of circle c is therefore proportional to $\sin\Theta$, and the area of a vanishingly narrow ring-shaped hemispherical segment adjacent to c (such as the ring bounded by circles c and d in Fig.2) is also proportional to $\sin\Theta$.

In summary, the FM distortion factor at progressively larger angles from the speaker axis decreases as the cosine of the angle, while the fundamental radiated power increases as the sine of the angle. In order to apply the Beers and Belar expression for the FM distortion factor to the total radiated power in a reverberant room, therefore, the sideband power radiated through the entire hemisphere must be calculated and compared with the total power radiated, using these trigonometric weighting factors. The distortion factor DF_T for the total radiated power is derived in Appendix B as 0.58 times the on-axis anechoic distortion factor. The value 0.58 may be called the reverberant-field correction factor. When it is desired to calculate the FM-distortion factor for a loudspeaker in a reverberant field, in terms of cone excursion, the Beers and Belar Eq. (5) must be changed to

$$\mathrm{DF}_T = 0.019 A_1 f_2 . \tag{6}$$

DF_T is not weighted by the modulating frequency, in order that it be comparable to the DF_0 of Beers and Belar. However, the audibility of a given amount of FM distortion is affected by the modulating frequency, as shown in Table I.

Beers and Belar also derived an expression for the on-axis FM-distortion factor in terms of the speaker's radiated power into half space (2π steradians):

$$\mathrm{DF}_0(\%) = 2900[f_2(P_1)^{1/2}/f_1^2 d^2] , \tag{7}$$

where f_2 = modulated frequency, P_1 = power output at modulating frequency (in acoustic watts), f_1 = modulating frequency, d = cone diameter in inches, DF_0 = on-axis FM-distortion factor for radiation into half space.

The cone excursion required to radiate a given amount of acoustic power at a given low frequency is proportional to the square root of the effective solid radiation angle at that frequency. For a given power output, therefore, the cone excursion for a speaker in a three-boundary corner will be half that required in the half-space environment assumed by Beers and Belar. The FM distortion, since it is directly proportional to the cone excursion, will also be halved.

Loudspeakers are not typically used in room corners. As Waterhouse[12] has shown, however, the effectiveness of a room boundary in reducing the radiation solid angle is determined not by its absolute distance from an acoustic source but by the distance in units of wave length. In domestic living rooms loudspeakers are never "far" from a corner when radiating signals of very low frequency. Assuming that a loudspeaker is located typically distant from a three-boundary intersection—with the woofer cone 1 ft. from the rear wall, 3 ft. from a side wall, and 4 ft. above the floor—for a given cone excursion the power radiated at 40 Hz will be 2.6 times that radiated into half space (see Appendix C), and the excursion requirement for a given radiated power will be $1/(2.6)^{1/2} = 0.62$ the value assumed by Beers and Belar when developing Eq. (7). The value 0.62 may be called the radiation-load correction factor for the conditions described above.

The value of the radiation-load correction factor will vary with the distance of the speaker from the three nearest room boundaries, and with the modulating frequency. The value of 0.62 in the example above is probably not far from an average value for speakers in domestic use radiating signals at frequencies close to or below their primary resonance, where maximum cone velocities are likely to occur. If this value is used together with the reverberant-field correction factor 0.58, the expression for FM distortion as a function of radiated power, for conditions more realistic than those assumed for Eq. (7), becomes

$$DF_r(\%) = 1043[f_2(P_1)^{1/2}/f_1^2 d^2], \qquad (8)$$

where DF_r = the square root of the ratio of power in the FM sidebands to the total power of the f_2 signal and sidebands in the reverberant field of a listening room, expressed as a percent.

When FM distortion is calculated in terms of low-frequency power output, Eq. (8) will yield sideband values approximately $\frac{1}{3}$—0.58×0.62—those correctly predicted by Beers and Belar for the on-axis output of a speaker operating in a half-space anechoic environment. However, when Eq. (6) is used to calculate the FM distortion factor in terms of a given cone excursion, only the reverberant-field correction factor applies directly. The effect of the radiation-load factor on Doppler is indirect, in that it decreases the excursion requirements for a given low-frequency power output.

Even when FM distortion is calculated as in Eqs. (6) or (8), its numerical value can become quite large. The relationship between the FM distortion factor DF_T and the unweighted frequency shift Δf_2, each expressed as a percentage, can be calculated by combining Eqs. (3) and (6):

$$DF_T = \Delta f_2 (f_2/f_1) 0.58 . \qquad (9)$$

Thus an unweighted frequency shift of 0.17% at a modulating frequency of 40 Hz (equivalent to 0.05% weighted shift), predicts 2.4% FM distortion if the woofer carries signals up to 1 kHz, and 12.3% if the speaker bandwidth is extended to 5 kHz.

In a tape machine, flutter modulates the entire spectrum [f_2 in Eq. (9) reaches higher values]; the modulation is not diluted by off-axis sound (the 0.58 multiplier is dropped); and the modulating frequency is likely to be much lower (f_1 is lower). A tape machine with 10-Hz, 0.063% unweighted flutter (0.05% weighted flutter) modulating a 5-kHz signal produces 31.5% FM sideband distortion. Such a value must be interpreted in the context of the high-quality performance of professional tape machines.

III. PREVIOUS LISTENING TESTS FOR DOPPLER DISTORTION

Klipsch[3] mounted an eccentric capstan on a tape machine to impose flutter, at rates of 10, 20, and 40 Hz, on a 1026-Hz signal. The shaft center of a 0.28-in.-diam. capstan was offset by 0.0014 in.: The capstan radius measured from the center of the shaft, and the length of driving surface per degree of turn, thus varied by ±1% with each revolution, producing 1% peak unweighted flutter or 0.7% rms flutter. (Klipsch counted the flutter as 0.35% rms, which we believe was a calculating error.) Klipsch reported that the flutter was unpleasant at 20 Hz and extremely irritating at 40 Hz. He extended his results to loudspeakers, stating that the flutter was analogous to 0.21-in. center-to-peak (0.42-in. peak-to-peak) excursions at 50 Hz. A recalculation of analogous speaker cone excursions on the basis of 0.7% flutter, using Eq. (3) and the qualifying data in Sec. II, yields much higher values. The excursions are an impractical 2.19-in. peak-to-peak at 20 Hz, 1.09 in. at 40 Hz, and 0.875 in. at 50 Hz for a listener on the speaker axis in a nonreverberant field, and they are approximately 1.7 times greater under real-life reverberant conditions, in which off-axis and on-axis sound are mixed.

In a second investigation, also by Klipsch,[4] a 20-member jury compared music reproduction through two speaker systems. One system was horn-loaded. The second system, made by the same manufacturer, used a simple direct-radiator bass speaker, with mid- and high-frequency speakers similar to those of the first system. All but one member of the jury considered the horn system to be superior. Klipsch found no significant differences between the two systems in harmonic distortion or power capability, and concluded that a difference in modulation distortion was responsible for the jury's choice. Many other differences are likely to exist, however, between the sound of a particular horn-loaded speaker and that of a particular direct radiator, whether or not the two systems are manufactured by the same company.

Moir[5] ingeniously separated Doppler sidebands in the 3-kHz region from the output of a single-cone, full-range loudspeaker, using an FM discriminator. A listening jury compared these sidebands with isolated AM sidebands in the same frequency region, produced from the same passage of music by a slightly overloaded amplifier. The FM sidebands were judged as more annoying, but whether or not these sidebands would have been audible when not separated from the body of the music was not determined.

Kurtin,[6] using a pure-tone simulator to represent Doppler effect, reported on the basis of listening sessions (no details were given) that in the range of modulated frequencies between 100 Hz and 4 kHz and of modulating frequencies between 10 Hz and 200 Hz, the critical factor affecting FM-distortion audibility was not the percentage but the absolute value of the frequency deviation. Deviations of 4 Hz or more became audible. Because of these findings Kurtin concluded that the performance of both woofers and tweeters would be adversely affected by FM distortion. Kurtin's results are inconsistent with the body of psychoacoustic test results on which current flutter Standards are based.

Fryer[7] introduced an electronic time-delay device into the amplifier circuit driving a loudspeaker to simulate Doppler distortion. He then reduced the amount of simulated Doppler until switching it in and out could not be detected by listeners. The speaker sound level was set low enough—80 dBA—to keep Doppler distortion generated by the speaker itself at a minimum. Fryer reported that Doppler became just audible at simulated peak-to-peak cone excursions of 10 mm (0.39 in.) at an estimated modulating frequency of 40 Hz. This excursion value is large enough so that the limit imposed is of little practical significance, but even this limit does not apply to the real case. Although Fryer used a four-unit speaker system his simulator acted on the entire frequency spectrum, imitating Doppler effect in a single-cone, full-range loudspeaker. Further, unlike real Doppler, Fryer's simulator acted equally on off-axis and on-axis signals radiated by the speaker, so that the reverberation-field correction factor was not taken into account. When these two factors are taken into account, Fryer's results are consistent with the conclusions in this paper.

IV. EXPERIMENT 1: TEST FOR AUDIBILITY OF DOPPLER DISTORTION, WITH LOW-FREQUENCY CUTOFF AS CONTROL

In this experiment a listening jury was asked to compare passages of music reproduced in two modes. One mode differed from the other primarily in the amount of frequency-and amplitude-modulation distortion.

A. Subjects and equipment

There were eight jurors, seven male and one female, aged 22 to 64. Seven had had considerable experience in judging music-reproducing devices, either professionally or as amateur enthusiasts. Two were musicians. The authors were not among the jurors, and six of the jurors had no institutional associations, past or present, with either of the authors.

The recording setup shown in Fig. 3 used the following equipment:

Acoustic Research turntable with Shure V-15 IV cartridge,
Acoustic Research 60-w amplifier,
Two Krohn-Hite 3750 R filters connected in series,
Acoustic Research AR-3 speaker system (12-in. woofer with a primary resonance frequency of 43 Hz, crossover at 1 kHz),
Bruel and Kjaer one-inch capacitor microphone 4132 mounted in a pinnaless dummy head,
General Radio 1560-P42 microphone preamplifier,
Revox A77 two-track tape recorder.

Sound-pressure levels were measured with a General Radio sound-level meter 1551-C in the "fast" position. The meter speed is in conformity with IEC R123-1961. The combined turntable and tape-machine flutter was measured with a Minicom 8100A flutter meter.

The listening tests, which were conducted at a different location from that of the recording, used a Dyna PAT-5 FET preamplifier and Dyna power amplifier, and Pioneer Monitor 10 earphones. Jurors switched from one mode of reproduction to the other with an SPDT toggle switch that was mechanically and electrically quiet, and whose break-before-make construction

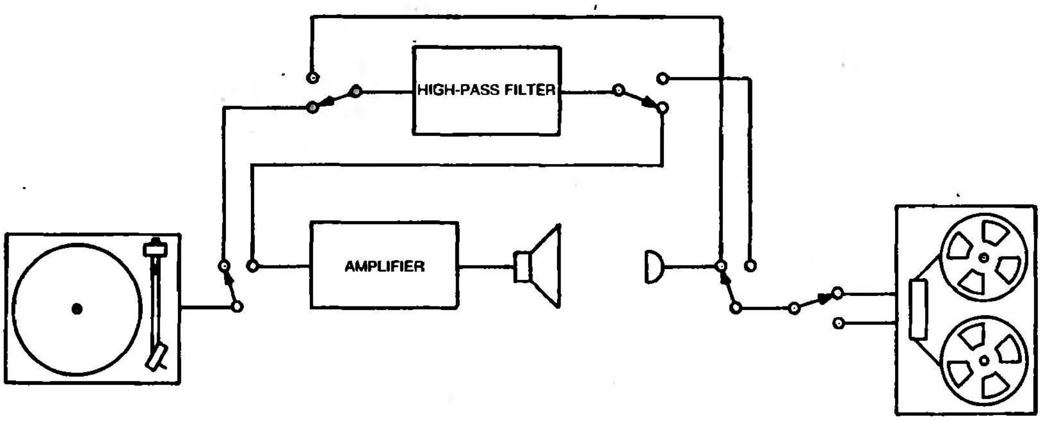

FIG. 3. Recording setup for expt. 1. The high-pass filter is switched between the input circuit to the speaker and the microphone circuit, so that when low frequencies are removed from the speaker the frequency response at the tape recorder remains the same.

prevented the two input lines from being connected in parallel momentarily during the switching operation.

The music for the Doppler tests was from these disk records:

Bach—Sinfonia from Cantata No. 29 (pipe organ), dB 1000; side 1, band 3,

Stravinsky—Firebird Suite, Telarc DG-10039; side 1, band 2.

B. Procedure

Orchestral and pipe-organ music was played through a loudspeaker system in a listening room of approximately 3500 ft^3 volume. The reproduced music created maximum sound-pressure levels of 97 dB *re* 0.0002 μbar, measured at a distance of 3 m from the speaker on its axis. Two speakers in stereo would have produced 100 dB, corresponding to maximum concert-hall SPL's for a full symphony orchestra.[13] Music was chosen whose heavy low-frequency content required high cone velocities—peak-to-peak excursions of 0.25 in. were observed frequently, and 0.375 in. occasionally, from each record.

Two types of tape recordings were made at 15 ips of the output of the speaker, with the microphone 2 m on axis. In one recording mode a 200-Hz, 48 dB/oct high-pass filter was inserted into the amplifier circuit driving the speaker, which sharply reduced low-frequency cone velocities and thereby largely eliminated both frequency- and amplitude-modulation distortion. In the second recording mode the same high-pass filter was moved to the recording circuit, allowing modulation distortion to reach its normal value for that speaker while matching the frequency balance of the second recording to that of the first.

The recording setup is shown in Fig. 3. Each type of recording was made on one channel of a two-track tape. The same musical passages were recorded on each channel, in synchronization with each other. The two recording and reproducing tape channels were matched in frequency response to within less than 1 dB.

As can be seen in Eq. (8), frequency-modulation distortion at a given modulating power varies inversely with the square of the modulating frequency. The maximum Doppler distortion remaining in the recording with bass-attenuated input to the speaker would thus be likely to consist of sidebands created by a 200-Hz modulating signal. This distortion, for a 200-Hz modulating signal equal in power to a 50-Hz modulating signal in the recording with full-bass input to the speaker, would be reduced by a factor of 16. The higher modulating frequency would also produce distortion of much lower audibility, as shown by the weightings listed in Table I.

AM sidebands whose amplitude is a function of cone excursion are also created. These are associated with nonlinearities in the mechanical, magnetic, and acoustical elements of the speaker system itself, and with nonlinearities in the radiation process. Since the reduction of low-frequency cone velocities reduces amplitude-modulation sidebands as well as frequency-modulation sidebands, audible differences between the two reproducing modes could be caused by AM sidebands. However, if it turned out that no differences were detectable, the contamination by amplitude modulation (or by harmonic distortion) would not invalidate the results.

A test tape was made up with one orchestral and one organ passage for the modulation-distortion comparisons. Two additional comparisons were included—one in which the two tape channels were identical and one in which there was a mild contrast in spectral balance (flat vs 6 dB/oct rolloff above 4 kHz). The passages were 1 to 2 min in length. The extra comparisons provided a partial test of the judging procedure, since a ballot was to be disqualified if the judge marked either the spectral-contrast band or the identical band incorrectly. These extra comparisons also gave the judges a set of bands that they could mark as different even if they heard no differences in the real test bands. It was thought possible that if a judge failed to hear differences in any of the bands there might be pressure to find differences where none existed.

Judges were given a level control that they were instructed to adjust according to their own musical preference, and a switch with which they could change from one tape channel to the other at will. The two channels had been balanced for equal level. Judges were asked to compare the two channels of each band for similarity, and to mark their judgments (*same* or *different*) on a form provided. If the channels were evaluated as different, the judges were to indicate which channel, if either, they thought was more accurate. The nature of the experiment was not known to the judges, who were told only that at least one set of channels was identical and at least one set different in reproducing quality. The test was administered by a person who did not know the nature of the recorded material, so that the experiment would be double blind. This person operated the tape machine, repeating bands as many times as requested. Each judge took the test alone, and no one other than the operator and one judge was present in the room.

Although the lack of bass in the recordings made it possible to play them back through a loudspeaker without adding significant FM distortion, earphones were used, with left and right phones driven in parallel for either channel. The use of earphones avoided the double reverberation that would have been produced by the inclusion of both recording and reproducing listening-room environments in the test sound.

The use of earphones also eliminates the effect of normal head diffraction, which would produce approximately 4 dB of emphasis at 1 kHz relative to 200 Hz. This 4-dB loss would reduce the relative amplitude of the sidebands being investigated, and so the recording microphone was mounted in a dummy head to restore the diffraction emphasis.

Over and above frequency modulation from Doppler effect, the test sound contained flutter from the turntable and from the tape machine in both the recording and the reproducing mode. The total NAB-weighted flutter introduced by these three operations was measured as 0.09%, a good part of which must be attributed to residual flutter in the test record.[14]

As White and Gust[15] have shown, FM sidebands are also generated by mechanical nonlinearities in the process of playing phonograph records. The major source of this FM is vertical tracking-angle error. We connected the left and right output channels of the pickup cartridge in parallel, in order to obtain monaural music signals for the test tapes from stereo records, and vertical signal components were thereby canceled. The equations of White and Gust indicate that the remaining lateral tracking-angle error, under the most extreme conditions, would produce less than 0.015% weighted flutter.

The lack of bass in these test tapes may be expected to make modulation-sideband distortion more audible than it would be in normal reproduction, since the amplitude of the sidebands becomes a larger percentage of the whole signal. Bass attenuation also removes a part of the signal that would have a masking effect on the sidebands. The low-frequency section could have been dubbed in later, but it was decided after listening to the tapes that this was unnecessary, and that use of the bass-light tapes would increase the sensitivity of the test.

C. Results

None of the judges marked the identical or the deliberately different bands incorrectly, and no judge was disqualified.

All eight of the judges evaluated the Doppler-contrast bands of organ music as the same.

Seven of the eight judges evaluated the Doppler-contrast bands of orchestral music as the same. One judge evaluated them as different, marking the band with Doppler as more accurate.

D. Conclusions and discussion

Although one response (out of a total of sixteen comparisons relative to Doppler distortion) indicated that the judge heard a difference between bands, this judge's preference for the band with Doppler makes it likely that the response had to do with some factor or factors other than Doppler distortion. FM sidebands have no inherent harmonic relationship to the frequency of the modulated signal, and can only contribute a raucous quality to the reproduced sound. In any case, Doppler distortion in this experiment, in the absence of masking by the normal low-frequency tones and with cone velocities that provided close to worst-case conditions, was inaudible in fifteen out of sixteen tests.

Klipsch[4] has written that a small amount of Doppler distortion, even when it is below the level detectable in an A–B comparison, may be objectionable because it contributes to "listening fatigue." It would be difficult to prove or disprove that there are subliminal frequency-modulation effects that create long-range annoyance without being immediately detectable. If such effects do exist, however, the analogy to flutter discussed earlier would lead one to expect much larger effects from tape machines and turntables than from loudspeakers.

V. EXPERIMENT 2: TEST FOR AUDIBILITY OF DOPPLER DISTORTION, WITH MICROPHONE POSITION AS CONTROL

As in experiment 1, a listening jury was asked to compare passages of music reproduced in two modes, one with and the other without Doppler distortion. In this experiment the contrast was achieved by a change of recording-microphone position from on-axis to off-axis, in an anechoic environment.

A. Subjects and equipment

The same subjects as in experiment 1 served as jurors. The equipment was also largely the same, with the following changes:

The two-mode recordings were made in the MIT anechoic chamber.

The speaker used for the recordings was a single-cone, 10-in. Allison woofer designed for a crossover frequency of 1 kHz. On- and off-axis frequency-response curves were made with a Bruel and Kjaer level recorder 2305, set for a paper speed of 3 mm/s and a writing speed of 50 mm/s.

The jury listened through two Allison: Five speakers connected in parallel. Since the judges did not listen with earphones, the recording microphone was used without the dummy head of experiment 1. The music for the Doppler tests was from the following disk records: Ives—Variations on "America" (pipe organ), Nonesuch H-71200; side 1, band 4, Stravinsky—Firebird Suite, Telarc DG 10039; side 1, band 2.

B. Procedure

Orchestral and pipe-organ music was played through a low-frequency loudspeaker placed in an anechoic chamber. As in experiment 1, peak-to-peak cone excursions of 0.25 in. were observed frequently, and 0.375 in. excursions occasionally, with each record.

Two types of 15-ips tape recordings were made of the output of the speaker, one on each tape channel. In one recording mode the microphone was on a line 90° to the axis of the speaker. Frequency modulation from Doppler effect does not exist at right angles to the cone's axis, except in reflected sound (as pointed out by Beers and Belar) or in diffracted sound (as shown by Queen[16]). For the second recording the microphone was positioned on the axis of the speaker, and the input circuit to the speaker was equalized with a specially constructed RC network to bring the effective on-axis speaker frequency response to within 1 dB of its 90° response. In each case the microphone was 2 m from the speaker. An additional set of bands was recorded with a deliberate difference between channels, in that one had been subjected to low-frequency rolloff below 125 Hz.

The effective frequency response of the speaker in each of the recording modes is shown in Fig. 4. Only a single-cone speaker system could have been used for this experiment, because interference effects in a multispeaker system would have made accurate frequency equalization of on-axis to off-axis response impossible. Cabinet diffraction also makes equalization difficult. Diffraction effects were substantially eliminated by rounding the edges of the speaker enclosure, and by applying sheets of glass fiber to the cabinet on each side of the speaker opening. The metal frame of the speaker itself was mounted so that the outer mounting flange was flush with the baffle surface, and any gaps or sharp edges were smoothed over with putty. Finally, the input signal to the speaker was rolled off above 1 kHz, where the most erratic response is likely to exist.

The absence of substantial reflections or diffraction prevented any significant FM sideband products from appearing at 90° to the speaker axis. The listening tests were conducted at the same time as those of expt. 1, and in the same way, except that the judges listened through loudspeakers. Since the recording was made in an anechoic chamber, a reverberant environment was used at the final stage. A 48-dB/oct rolloff below 200 Hz was introduced in the Doppler contrast bands to prevent the reproducing loudspeakers from generating Doppler distortion.

The reverberant-field correction factor that reduced the amount of Doppler distortion generated in experiment 1 is not operative in the anechoic environment, since the on-axis Doppler distortion that was recorded is not diluted by reflected off-axis sound. Thus the Doppler distortion in this experiment is approximately 1.7 (1/0.58) times as great as it would be if it had been generated by the same cone excursions in a typical living room.

C. Results

None of the judges marked the deliberately different bands incorrectly, and no judge was disqualified.

All eight of the judges evaluated the Doppler contrast bands of orchestral music as the same.

Five of the eight judges evaluated the Doppler contrast bands of organ music as the same, but one of the five added a question mark on the response form. Three judges evaluated the Doppler contrast bands as different; one of the three indicated that the band with Doppler was more accurate, one that the band without Doppler was more accurate, and one that neither was more accurate than the other.

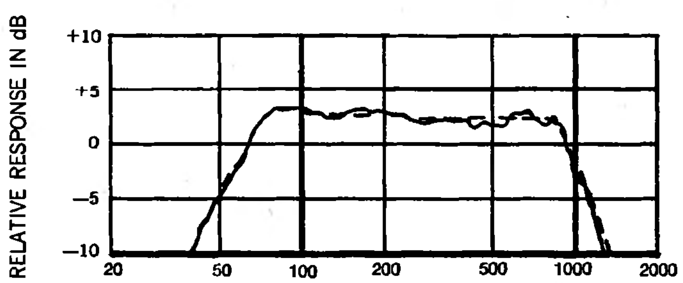

FIG. 4. Frequency response of speaker used in experiment 2: — measured 90° off axis, --- equalized and measured on axis.

D. Discussion and conclusions

There was no agreement among the three judges who detected a difference between Doppler contrast organ bands as to which band was more accurate, and this makes it appear likely to us that their responses were determined by a factor of factors other than Doppler distortion. One possibility is that the judges were influenced by the imperfect synchronization between the organ bands, whose time error was greater than in any other pair of bands in either experiment 1 or experiment 2.

We believe that the anechoic organ test was invalidated by an inadequate match between bands, not counting the Doppler contrast. Nevertheless, one conclusion that would be consistent with the results of this test would be that the Doppler distortion in the organ recording (which was 1.7 times the amount that would have been generated by the same speaker in a normally reverberant room, and which was not masked by the normal low-frequency tones) was marginally audible.

In the anechoic listening test with orchestra, Doppler distortion was inaudible in spite of the 1.7-fold increase and the absence of low-frequency masking.

ACKNOWLEDGMENT

We would like to thank the Auditory Perception Group of the Research Laboratory of Electronics at the Massachusetts Institute of Technology for allowing us to use the MIT anechoic chamber.

APPENDIX A: CALCULATION OF DOPPLER SHIFT

The frequency of a signal at the observation point, when there is motion between the source and the observer, is

$$f_0 = [(c - u_0)/(c - u_s)] f_s ,$$

where f_0 = frequency at the observation point, c = velocity of sound, u_0 = velocity of the observer, u_s = velocity of the source, f_s = frequency of the source. The frequency shift Δf_s is therefore

$$\Delta f_s(\%) = \left(\frac{f_0 - f_s}{f_s}\right)100 = \left(\frac{f_0}{f_s} - 1\right)100 = \left(\frac{c - u_0}{c - u_s} - 1\right)100.$$

In the case of a loudspeaker radiating low- and high-frequency signals simultaneously, u_0 is zero, u_s is the low-frequency cone velocity u_c, and f_s is the higher, modulated frequency f_2. The on-axis frequency shift Δf_2 of the modulated signal is

$$\Delta f_2(\%) = [(c/c - u_c) - 1]100 = [u_c/(c - u_c)]100 .$$

APPENDIX B: CALCULATION OF REVERBERANT-FIELD CORRECTION FACTOR

Beers and Belar define the FM distortion factor as $DF = (P_{SB0}/P_0)^{1/2}$, where DF = FM distortion factor for on-axis radiation, P_{SB0} = power in the FM sidebands for on-axis radiation, P_0 = power of the modulated signal plus sidebands directed on axis. Therefore $P_{SB0} = (DF_0)^2 P_0$.

For radiation at $\Theta°$, the distortion factor is

$$DF_\Theta = (P_{sB\Theta}/P_\Theta)^{1/2}, \text{ radi and } P_{sB\Theta} = (DF_\Theta)^2 P_\Theta,$$

where $P_{sB\Theta}$ = sideband power radiated through the ring-shaped hemispherical segment bounded by circles c and d in Fig. 2, P_Θ = modulated signal (f_2) power plus sideband power radiated through the segment bounded by circles c and d.

From Fig. 1, $DF_\Theta = DF_0 \cos\Theta$; therefore $P_{sB\Theta} = (DF_0)^2 \times \cos^2\Theta P_\Theta$.

Since f_2 power is equal per unit area of S in Fig. 2, and since the area of each ring-shaped segment in Fig. 2 is proportional to $\sin\Theta$,

$$P_\Theta = k \sin\Theta \, d\Theta,$$

where k is a constant. Therefore $P_{sB\Theta} = (DF_0)^2 k \cos^2\Theta \times \sin\Theta d\Theta$.

DF_T, the FM distortion factor for power in the reverberant field, is equal to the square root of the ratio of the total sideband power to the total f_2 power

$$DF_T = (P_{sBT}/P_T)^{1/2},$$

where P_{sBT} = total sideband power radiated through hemisphere S in Fig. 2, equal to

$$\sum_{\Theta=0'}^{90} P_{sB\Theta}.$$

P_T = total f_2 power radiated through S, equal to

$$\sum_{\Theta=0}^{90°} P_\Theta.$$

$$DF_T = \left((DF_0)^2 k \int_0^{90} \cos^2\Theta \sin\Theta \, d\Theta / k \int_0^{90} \sin\Theta \, d\Theta\right)^{1/2}$$

$$= [(DF_0)^2(-0.33)/-1]^{1/2} = 0.58 \, DF_0.$$

APPENDIX C: EFFECT OF ROOM ENVIRONMENT ON LOW-FREQUENCY POWER OUTPUT

Waterhouse[12] gives the following expression for the effect of nearby room boundaries on the power output of a small acoustic source:

$$\begin{aligned} W/W_f = 1 &+ j_0(4\pi x/\lambda) + j_0(4\pi y/\lambda) + j_0(4\pi z/\lambda) \\ &+ j_0[4\pi(x^2+y^2)^{1/2}/\lambda] + j_0[4\pi(x^2+z^2)^{1/2}/\lambda] \\ &+ j_0[4\pi(y^2+z^2)^{1/2}/\lambda] + j_0[4\pi(x^2+y^2+z^2)^{1/2}/\lambda], \end{aligned}$$

where W = the power radiated by a source located at x/λ, y/λ, and z/λ with respect to reflecting boundaries, W_f = the power that would be radiated by the source in full space (into 4π steradians), $j_0 = \sin a/a$, the spherical Bessel function. Each term has a maximum value of 1, which is approached as the distance (in units of wavelength) of the source to the boundaries becomes smaller. Therefore the expression has a maximum value of 8. With typical placement in a listening room (x = 1 ft, y = 3, z = 4 ft), and with λ equal to 28.25 ft (40 Hz), $W/W_f = 5.2$.

[1] G. L. Beers and H. Belar, "Frequency-Modulation Distortion in Loudspeakers," Proc. Inst. Radio Electron. Eng. Aust. **31**, 132–138 (1943).

[2] P. W. Klipsch, "Loudspeakers and Acoustic Fundamentals," Radio Electron. **28**(10), 44–45 (1957).

[3] P. W. Klipsch, "Subjective Effects of Frequency Modulation Distortion," J. Aud. Eng. Soc. **6**, 143 (1958).

[4] P. W. Klipsch, "Modulation Distortion in Loudspeakers," J. Aud. Eng. Soc. **17**, 194–206 (1969).

[5] J. Moir, "Doppler Distortion in Loudspeakers," Audio **60**(8), 42–52 (1976).

[6] S. Kurtin, "Frequency Modulation in Loudspeakers as a Loudspeaker Design Criterion," Audio **61** (12), 52–57 (1977).

[7] P. A. Fryer, "Simulation and Investigation of Doppler Distortion," Preprint 1197, Audio Engineering Society, 60 E. 42 St., New York, NY 10017.

[8] "Magnetic Tape Recording and Reproducing (Reel-to-Reel)," NAB Standard (April 1965), National Association of Broadcasters, 1771 N. Street N.W., Washington, DC.

[9] An alternate method of calculating cone velocity, used by Klipsch[4] and others, is to derive maximum velocity from the equation for the instantaneous amplitude of a sine wave ($x_i = x_{max} \sin\omega t$); x_i is the distance of the cone from its center position at any instant, and x_{max} is the center-to-peak cone excursion. The equation is differentiated to find the instantaneous velocity ($dx_i/dt = x_{max}\omega\cos\omega t$), and the differential is solved for maximum velocity by setting $\cos\omega t$ equal to one [$(dx_i/dt)\max = x_{max}\omega = 6.28\, x_{max} f$]. This maximum velocity may be converted to rms velocity by multiplying by 0.707, which yields $4.44 x_{max} f$ or $2.22 xf$ in terms of the peak-to-peak excursion, the same expression calculated by dividing cone travel by time and multiplying by 1.11.

[10] F. A. Comerci, "Perceptibility of Flutter in Speech and Music," J. SMPTE **64**, 117–122 (1955).

[11] An assumption must be made that the distance from the source of sound to the observer is large compared to the distance the source moves in one f_2 period.

[12] R. V. Waterhouse, "Output of a Sound Source in a Reverberation Chamber and Other Reflecting Environments," J. Acoust. Soc. Am. **30**, 4–13 (1958).

[13] Bell Lab. Rec. **12**, 315 (1934). Measurements of speaker output with a faster meter (50 μs rise time) yielded SPLs 10 dB higher.

[14] Julian Hirsch, equipment reviewer for Stereo Review, has informed us that the lowest weighted turntable flutter he has measured using a test record is in the range between 0.05% and 0.06%, while optical methods that do not require a test record yield values as low as 0.02%.

[15] J. V. White and A. J. Gust, "Measurement of FM Distortion in Phonographs," J. Aud. Eng. Soc. **27**, 121–133 (1979).

[16] D. Queen, "Doppler Distortion versus Loudspeaker Orientation in Real Rooms," J. Aud. Eng. Soc. **26**, 1002(A) (1978).