The case of the missing 6 dB

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In 1933 data were reported which indicated that thresholds of hearing for low frequencies apparently depended upon whether the source was an earphone (MAP) or a loudspeaker (MAF). A decade or so later the same type of discrepancy appeared when loudness balances were made at 100 Hz between an earphone source and a loudspeaker source. In both cases approximately 6 dB more sound pressure level at the eardrum was required when the earphone was the source than when the loudspeaker was the source. Later research added credence to this paradox, namely, the ear should act as a pressure operated device, and there should be no difference between MAP and MAF; yet a difference seemed to exist. Research reported in abstract form and orally by the author in 1962 and 1963 showed that (a) the difference at threshold was due to physiological noise generated in the ear canal by the earphone–cushion–head combination (and could be eliminated with a special earphone-coupling system), and (b) the suprathreshold differences obtained with loudness balancing were due to a number of subtle procedural and experimental techniques (techniques which could be modified so as to avoid all of the problems of past experimenters). This research is reported here for the first time in full detail. A total of 15 different subjects participated in eight experimental comparisons using three to nine subjects each, with sufficient replications so that most subjects' MAF–MAP and/or loudness differences were determined within 1 or 2 dB at the 95% confidence level. It was often possible to replicate previous results using previous methods, but with the modified methods reported here the average difference across experiments was less than 0.2 dB, and no subject in any experiment exhibited more than a 1.8-dB difference averaged across trials. The case of the missing 6 dB should be considered closed.

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I. BACKGROUND

One part of the missing 6-dB problem started in 1933 with a publication by Sivian and White (1933) which showed that pressure thresholds at low frequencies using conventional earphones mounted in flat cushions were approximately 6 dB higher than thresholds on the same subjects when a loudspeaker was the sound source and the subject's ears were uncovered. There was no acceptable explanation given as to why the minimum audible pressure (MAP) differed significantly from the minimum audible field (MAF). This earlier problem was compounded during World War II when loudness balance techniques were used to measure the real ear response of earphones and the attenuation of earphone cushions. Beranek (1949) reported that, when equal loudness judgments at low frequencies were made for sounds generated by conventional earphones or generated by a loudspeaker, it was necessary to have approximately 6 dB more sound pressure level on the subject's eardrum when the earphone was the source than when the loudspeaker was the source.

The problem, whether relating to thresholds or loudness balances, has been referred to as "the missing 6 dB" throughout the subsequent literature. As late as 1952, Munson and Wiener (1952) found that for loudness balancing the reported difference still existed and they had no explanation for the difference. This was followed by a statement of Robinson and Dadson (1956) that they found there was still a difference for low-frequency thresholds and that the cause for this difference was not entirely clear but was probably of objective origin.

It has always been assumed that the ear is a pressure operated device, yet the "missing 6 dB" paradox has remained in the literature. The explanation of the first part of the problem was given in a very brief form by Rudmose (1962); however, the explanation of the second part was not given until 1963 by Rudmose (1963) as an oral presentation at the fall meeting of the Acoustical Society of America. The following will present the total explanation of the missing 6-dB problem by means of material not included in the Rudmose (1962) paper with the single exception of the results (presented in a different form), along with material which has been described only by an abstract. There has been no attempt to provide a review of recent literature as the philosophy has been to write the paper as it would have been written in 1963 following the oral presentation. For those who would like references to work since 1963, the paper by Killion (1978) is recommended. He verifies the fact that for low-frequency thresholds there is no missing 6 dB; however, no recent paper has explained the missing 6 dB for loudness balances.

There has apparently been the feeling that a single, generalized explanation should exist for both parts of the problem because the pressure level differences were essentially the same. Such is not the case. There are truly two problems, each with its own solution. Because of this, the problems will be presented as two separate issues with the usual format of procedure, results, and conclusions.

II. THE THRESHOLD PROBLEM

A. Introduction

The cause of the problem is that physiological noise [first reported by Brogden and Miller (1947) and later much more fully documented by Shaw and Percy (1962) as it relates to audiometric thresholds] is generated in the ear canal due to the excitation of the earphone cushion which is tightly coupled to the small volume (approximately 6 cm^3—the equivalent volume of the concha, ear canal, and eardrum), and this physiologi-

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cal noise elevates the earphone threshold as a consequence of the masking due to the noise. If the volume is made substantially larger, or if a modest acoustical leak exists, the problem goes away. The question is, therefore, how to maintain experimentally an acoustically tight seal and a 6-cm^3 volume, yet reduce the level of the physiological noise sufficiently so that it does not mask the true threshold level of the subject. The complete solution requires: (1) the experimental demonstration that the problem does exist when using an earphone mounted in a cushion as the pressure source; (2) a technique for reducing the physiological noise while keeping a 6-cm^3 volume and a tight seal; (3) the experimental agreement between MAP and MAF using the new type of pressure source; (4) the physical measurement of the masking noise under the earphone cushion; (5) the calculation of the masked threshold based on the data obtained in (4) and the agreement of the calculated masked threshold with the experimental threshold; and (6) measurement of the physiological noise obtained in (2) and showing that thresholds measured with the new pressure system are no longer masked by the physiological noise.

B. Method

Since the noise is generated in the ear canal by virtue of the physiological noise in the skull being coupled by the tight contact of the large area of the flat earphone cushion (MX41/AR) to the ear canal, the area of contact with the ear must be reduced. This was accomplished by using a conventional earmold used with hearing aids. A tight seal was ensured by putting grease on the earmold each time it was used. The 6-cm^3 total equivalent volume was obtained using a machine fitting, designed to mate with the ring in the earmold, which had a short metal tube as its termination. A piece of rubber (4 mm i.d., 8 mm o.d.) was fitted over this metal tube. The other end of the rubber tube was attached to a conventional hearing aid earphone by means of a metal adapter. The length of the rubber tube (19 cm) was such that the total volume enclosed between the earphone driver unit and the eardrum was approximately 6 cm^3. The sound pressure level on the eardrum was measured by a probe tube (the tip embedded in the earmold and terminating at the eardrum end of the earmold) connected to a condenser microphone. Finally, the earphone-tube assembly was suspended from the wedges of the anechoic chamber and not from the subject's chair (which was mechanically isolated from the floor). Figure 1 is a photograph of the assembly and Fig. 2 shows the method of suspension. Thresholds obtained with this system are referred to as MAP (earmold). The 6-cm^3 volume and effectiveness of ear seal were checked by applying a given voltage to the driver unit and measuring the sound pressure level in the ear canal, then transferring the driver unit and measurement probe to a 6-cm^3 volume (machined metal) and measuring the same (within 0.2 dB) sound pressure level with the voltage on the driver unit remaining constant.

For earphone measurements, a Permoflux dynamic earphone in a flat cushion was mounted in a conventional headset adjusted for approximately 750-g force against the pinna. This force is generally accepted as sufficient to provide an essentially tight seal for most subjects when using this cushion. The pressure-measuring probe was a small piece of plastic tubing connected to a condenser microphone (via a machined adapter) on one end. The open end of the probe was placed at the entrance of the ear canal. The tube was positioned in the small notch in front of the pinna and did not affect the cushion seal as verified by comparing the eardrum pressure to the pressure in a 6-cm^3 coupler with constant voltage on the earphone. The two pressures were found to be equal within 0.2 dB. For threshold measurements, the condenser microphone preamplifier was suspended from the wedges of the chamber. Thresholds obtained with this system are referred to as MAP (earphone).

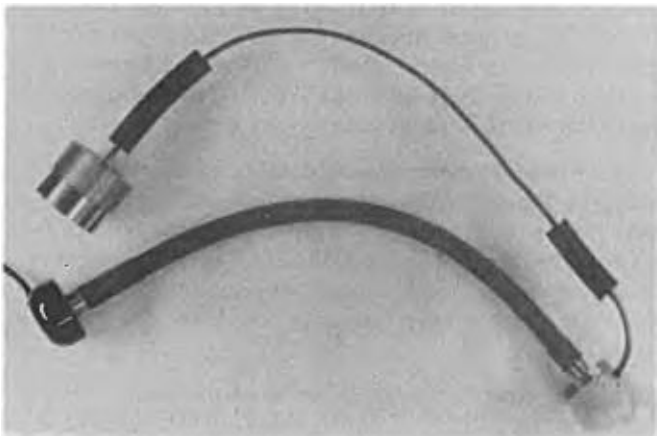

FIG. 1. Photograph showing earmold coupled to hearing aid driver unit via rubber tubing. Enclosed volume is 6 cm^3. Pressure in the ear canal is measured by the probe tube connected to a standard 2.5-cm condenser microphone.

FIG. 2. Photograph showing the subject, the earmold with driver, and the pressure measurement probe connected to condenser microphone on its preamplifier. Note that the driver system and the pressure measurement system are mounted from the anechoic wedges so as to be mechanically isolated from the subject. Subject's chair is mechanically isolated from the floor.

The loudspeaker used was a folded exponential horn type with a high quality driver unit. The subject's chair was mechanically isolated from the chamber floor, and the eardrum pressure level was measured with the same probe system used for the headphone system. The plastic probe was attached to the ear by adhesive tape. For monaural loudness balancing, the nonlistening ear was occluded to ensure monaural listening when the loudspeaker was the source. Thresholds obtained with this system are referred to as MAF.

Bekesy-type thresholds were obtained using a motor driven attenuator controlled by a handswitch. Pressures were measured using standard Bruel & Kjaer equipment; however, a General Radio 736-A wave analyzer (4-Hz bandwidth) was modified to operate between the microphone amplifier output and the graphic recorder. Thus with such a narrow bandwidth analyzer, the actual threshold pressures were recorded directly on the graphic level recorder. All measurements were made with this type of system. Physiological noise was analyzed by a motor drive attached to the GR analyzer. The response of the probe tube was uniform within ±1 dB over the frequency range 20–100 Hz. The noise analysis usually followed a series of threshold measurements before the earphone or earmold system was disturbed.

On any given day the subjects' free field and pressure thresholds were obtained along with physiological noise measurements. Several sets of data were obtained. This procedure was repeated on subsequent days; however, the order of taking thresholds was reversed. The difference between pressure and free-field thresholds was determined from the Bekesy tracings. Since the same measurement technique was always used, calibration errors are minimized and the standard deviation of the threshold differences was much smaller than for typical threshold measurements made over a number of days. Ten to 20 sets of data taken over several days were obtained for six subjects. (A larger number of subjects is not required for this type of experiment as a "population" is not being represented by the data.)

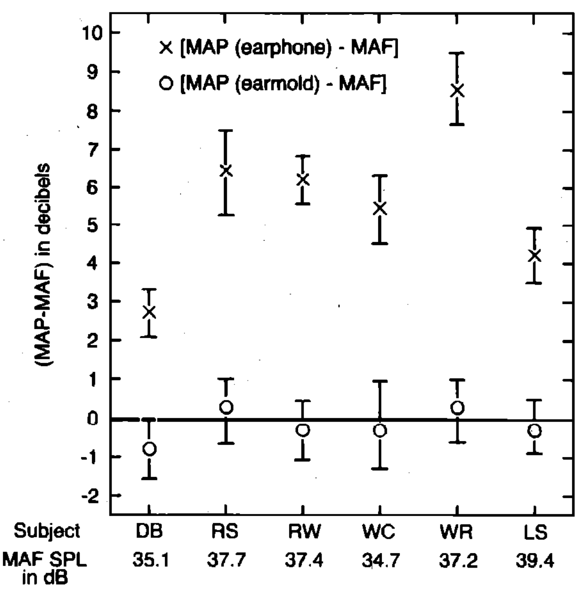

FIG. 3. Threshold differences between pressure thresholds (MAP: earphone or earmold source) and free-field thresholds (MAF: loudspeaker source). Monaural listening with opposite ear properly occluded. Frequency 100 Hz; bars are 95% confidence intervals.

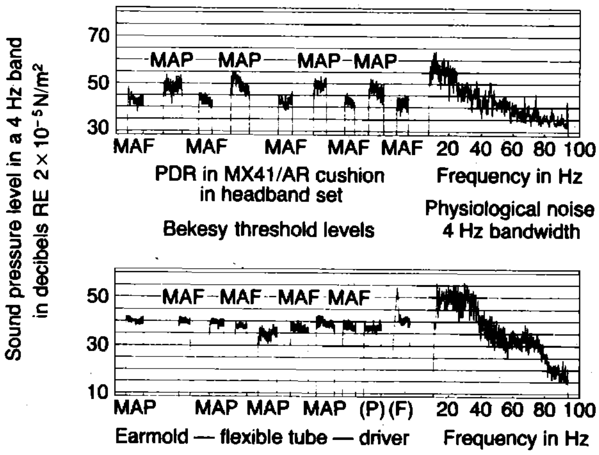

FIG. 4. Typical fixed frequency Bekesy pressure level thresholds using different sound sources for two different subjects. MAF is the threshold measured using the loudspeaker as the source. MAP is the SPL threshold using either a conventional earphone as the source (top record) or the special earmold source (bottom record). Frequency 100 Hz. The physiological noise analyses shown at the right of both records were obtained by sweeping a 4-Hz bandwidth analyzer from 20 to 100 Hz with the measurement probe under the earphone cushion and terminating at entrance to the ear canal. There was no electrical signal on the earphone while the noise was analyzed. The lower record is the same except the noise in the ear canal is measured with the earmold driver system in the ear, no electrical signal on the driver unit.

C. Results

The results of these experiments are shown graphically in Fig. 3. It is quite evident that the "missing 6 dB" is present with earphone listening and absent with earmold listening. The data are presented in the order in which the subjects participated with subject DB (not decibel but initials) first and LS last. After the series with subject DB was finished, the headband force was checked and found to be about 500 g. A stronger headband was used on the remaining subjects.

Two samples (different subjects) of data are illustrated in Fig. 4. The difference measurements (MAP−MAF) were measured as the difference between two successive thresholds (an extra MAF threshold occurs in the upper set of determinations—the tapes were unfortunately cut to have the same length and not the same data).

Physiological noise levels were tabulated at 10-Hz intervals over the 20–100-Hz range. Typical spectra are illustrated in Fig. 5. It was not necessary to record noise spectra above 100 Hz as the masking is controlled by the levels below 100 Hz due to the rapid decrease in spectral levels above 100 Hz. The ambient

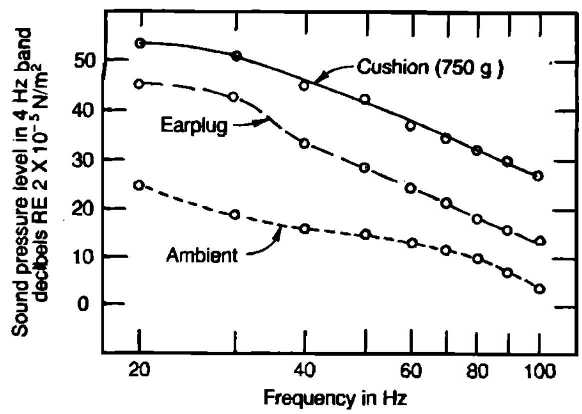

FIG. 5. Average physiological noise in 4-Hz band at the entrance of the ear canal when the ear was covered by a cushion and a conventional earphone (no electrical signal on earphone) with a force of 750 g holding the cushion against the pinna (top curve); when the earmold system was in the ear (middle curve); and the ambient noise spectrum measured at the ear with the ear canal open (lower curve).

levels clearly had no effect on the open ear measurements.

Once the physiological noise spectra were obtained, the output of a noise generator was equalized to approximate the average physiological noise spectrum, and the results are shown in Fig. 6. It was not necessary to have exact replication below 60 Hz as the critical band for monaural listening at 100 Hz has been determined as approximately 80 Hz (40 Hz below and 40 Hz above 100 Hz). Masking experiments using this simulated noise spectrum were conducted to verify the 80-Hz value of the critical bandwidth. The average of the masked thresholds was within 1 dB of the sound pressure level in an 80-Hz band centered at 100 Hz.

With the critical bandwidth thus verified, the average physiological noise spectrum was determined for each of the six subjects along with the average measured value of the subject's thresholds using the earphone MAP. Using the physical data, the masked threshold level due to the physiological noise was calculated.

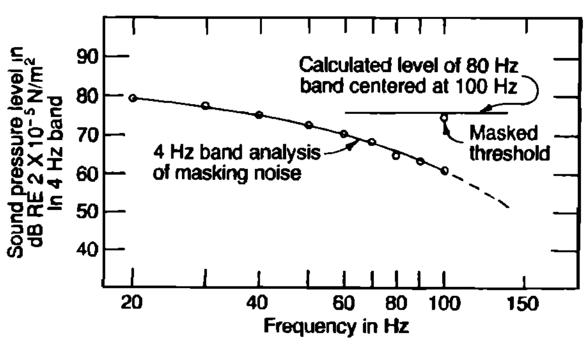

FIG. 6. Simulated physiological noise used in masking experiments to verify the critical bandwidth for monaural masking of 100 Hz due to this type of spectrum. With the noise adjusted as shown, subjects adjusted the level of a 100-Hz tone to obtain their masked thresholds. The average value of their masked thresholds was within 1 dB of the calculated level of the energy in an 80-Hz band centered at 100 Hz due to the masking noise.

The data are given in Table I and the agreement between measured threshold levels and calculated masked threshold levels is certainly satisfactory.

One final note pertains to the absolute level of the monaural free-field levels. Although six subjects represent too small a sample to be significant in terms of representing a population, it is interesting to note that the average of the monaural free-field thresholds for the six subjects is 36.9 dB SPL with a range of 34.7–39.4 dB SPL. This average level is in reasonable agreement with the level typically associated with 100 Hz.

D. Conclusion

It seems evident that thresholds for 100 Hz using earphones with flat cushions and tight seals are masked thresholds due to physiological noise transferred from the head to the ear canal via the large cushion. If the noise is reduced by reducing the area of the "cushion" in contact with the head, yet still keeping a 6-cm^3 volume with a tight seal, physiological noise in the ear canal can be reduced sufficiently so that pressure thresholds agree with free-field thresholds at 100 Hz for monaural listening. The agreement of measured and calculated masked thresholds based strictly on physical measurements further verifies that MAP (earphone) are masked thresholds.

One final question may still be raised. What if the hearing aid driver is attached directly to the earmold, thus reducing the enclosed volume to approximately 2 cm^3—is there now a masked MAP? This experiment was tried by one subject only (WR). An unmasked threshold was obtained if after adjusting a manual attenuator for just slightly above threshold (less than 1 dB) and then completely relaxing all muscles, the answer was the same as MAF. If, however, the slightest muscle activity occurs such as touching the roof of the mouth with the tongue, the threshold was masked. Certainly a Bekesy threshold (using a handswitch) or raising a finger to signify "I hear" would require enough muscle activity to produce a masked threshold.

III. THE LOUDNESS BALANCE PROBLEM

A. Introduction

If a loudness balance is obtained using two different sources, one an earphone in a flat cushion tightly

TABLE I. Comparison of measured and calculated masked thresholds. Values shown are SPL in dB *re* 2×10^{-5} N/m^2.

Subject	Earphone threshold	Calculated masked threshold
DB	37.8	35.0
RS	44.1	44.0
RW	43.6	42.7
WC	40.1	40.4
WR	45.8	43.6
LS	43.6	43.0

sealed against the pinna, the other a loudspeaker (no earphone on the ear), past experiments [Beranek (1949), Munson and Wiener (1952), and Robinson and Dadson (1956)] have indicated that approximately 6 dB more sound pressure level at the eardrum is required when the earphone is the source than when the loudspeaker is the source. This assumes that each source is driven by the same low frequency, and the sound pressure levels of the test tones are in the range of 65–80 dB. Furthermore, the phenomenon occurs whether loudness balancing is monaural or binaural.

At the range of sound pressures used in the loudness balancing experiments, physiological masking noise could play no role in explaining the problem. As described below, there are a number of different experimental factors which affect the subject performing the balancing, and not all of the factors affect all subjects. After three years of research a procedure was finally developed which satisfied all subjects, and if these procedures are followed there is no missing 6 dB for either monaural or binaural loudness balancing. The reasons which explain this part of the problem are thus procedural and, consequently, are not as scientifically satisfying as the solution just presented for the threshold problem.

Munson and Wiener (1952) almost had the solution but did not recognize it, or (as they stated) they did not have the time to pursue the problem further. The experimental factors which affect some or all subjects are (1) mechanical coupling of the subject's chair to the loudspeaker, (2) the "far" or "near" loudspeaker source problem, (3) the earphone and loudspeaker distortion problem, (4) the formal procedure for performing the balancing, and (5) for the monaural case, the problem of successfully occluding the nontest (or transfer) ear. These matters will be discussed and data will be presented to show the effect whenever it occurs.

B. Method

Most, but not all, subjects when listening to 100 Hz radiated by the loudspeaker are affected by the vibrational energy transmitted from the loudspeaker to the listener at the sound pressure levels used for loudness balancing. This energy transfer typically propagates via the floor to the chair of the listener, and the effect can be eliminated by proper mechanical isolation of the chair. An effective, but not elegant, method of isolation is to use the inner tube of an automobile tire between the floor and a piece of plywood on which the chair is mounted. By properly inflating the inner tube, the isolation for the 100-Hz vibrational signal can be made satisfactory. The effect of this vibrational problem is easily shown by obtaining loudness balances with and without the vibration isolator.

The "far" or "near" loudspeaker source problem is the name given to a measurable phenomenon that, when performing loudness balances between sounds generated by a loudspeaker located across the room with that generated by a loudspeaker near the ear (ear or ears open), some subjects require more sound pressure from the near source than from the distant source for equal loudness. To demonstrate this effect, a cast iron pressure chamber (volume approximately 0.03 m^3) was modified by attaching to the chamber a machined tube to make a Helmholtz resonator. The length of the tube was made variable so that the resonator could be tuned to 100 Hz; consequently, the radiated sound had very low distortion. This technique solved the problem of how to suspend a large loudspeaker (the large size is generally required to produce the higher sound pressure levels with low distortion) near the listener. Listeners who demonstrate this phenomenon evidently perceive the distant source as having a "large acoustic size" whereas the near source is perceived as much "smaller," consequently, the smaller source must be "stronger" (produce more sound pressure) to equal the loudness of the larger source. At the time the phenomenon was observed, it was discussed with von Bekesy, who stated he had observed this effect many years before when still in Europe but had not published the information. He also pointed out that, once a subject discovers this phenomenon, he can be trained to eliminate it. This result was verified by the author.

The distortion problem is subtle. One would expect, based on equal loudness contours that change between 100 and 300 Hz in terms of sound pressure, that relatively small amounts of 2nd and 3rd harmonic distortion would affect the loudness judgment of a 100-Hz signal. In a sense it does, but in only one of the subjects used was it demonstrable to a significant degree. The effect was demonstrated by using two different tone generators, one modified to produce harmonic distortion of the 100-Hz signal in the range of 5%. Most subjects, when matching the loudness of these two sources had difficulty with the "match" but eventually after reaching a decision that the two sources were equally loud, the sound pressures were equal. One subject did show a significant effect, but the distortion levels required to produce the effect were much higher than the distortion levels produced by the experimental equipment.

There is an effect that relates to distortion, however, and that is the "quality" of the balance as reported by the subjects. The electronic equipment was the same whether the source was the loudspeaker or the phones, but subjects would comment "the sources don't sound alike even though judged to be equally loud." This report was especially common when using conventional earphones. The resolution of the "quality" problem, which was successful for most subjects, was the development of sets of tapered cones made of the same material used for making semisoft earmolds for hearing aids. The material was hard enough to hold its shape but soft enough to effect a tight seal with the ear canal. A pressure probe tube was embedded in each cone and the usual mounting ring for hearing aid drivers embedded in the large end of the cone which was about 2.5 cm in diameter. The cone mold tapered to a diameter of about 0.5 cm at the small end, and the length of the cone was about 6 cm. A hole along the center line of the cone conducted the sound from the large end to the small end. The hearing aid driver unit was attached to the cone by a rubber tube much the same as to the earmold (Fig. 1). The listener held the cone in his hand and inserted the open tip end into his ear canal. To adjust for different size ear canals, the molded cones were cut off at various lengths to give a range of tip sizes for the subjects to choose the size that gave the best seal. The purpose of the tight seal arose from the fact that most subjects observed that, when using earphones, the sound was more like the sound from the loudspeaker if the subject pressed the earphone cushion very tightly against the pinna. Such increased pressure clearly reduces the acoustical leak via the cushion from the ear canal to the outside. Although all subjects used both types of phone systems (earphones in cushions and tapered cones with separate driver units) and the data are so reported, most, but not all, said the cone system was better, that balancing was easier and quicker, and that the sounds from the two sources were much more alike. The results, however, were the same using either driver system.

The experimental procedure problem is quite significant and is clearly the principal reason Munson and Wiener (1952) failed to solve the problem in 1952. Their experiment was automated using the "ABX" method whereby the loudspeaker served as the first source and the earphones served as the second source. They felt that the sudden placement of the earphones produced a slight excess static pressure in the ear canal and thus caused problems, but a water manometer showed this pressure rise to be small. The automated procedure gave very little time for listening to the earphone sound before having to make a judgment. When the ABX procedure was replicated in the present study, difficulties were observed especially if the earphone sound was the second sound. The time interval for earphone listening needs to be quite long compared to the time required for loudspeaker listening if the subjects are to feel that their results are meaningful. As Munson and Wiener pointed out, it is also necessary to measure the earphone sound pressure with a probe at all times, otherwise the cushion seal produces significant differences.

In making monaural balances between a loudspeaker source and an earphone source, two different procedures have been used in the past. One method uses just one ear and occludes the other. The open ear then balances the loudspeaker sound against the earphone sound by listening first to one source and then the other. The major pitfall with this procedure is to be certain the nonlistening ear is really not listening. At the sound pressures used, a loudness change of two to one requires approximately a 10-dB change in the pressure level. Thus to reduce the nonlistening ear so that its response to loudness is now one-fourth, the occlusion must provide at least 20-dB attenuation. It is really better to have more, otherwise the nonlistening ear will contribute to the loudness perceived by loudspeaker listening and will not contribute for earphone listening. It does not take much contribution from the nonlistening ear to upset the experiments. At 100 Hz it is not easy to obtain the required attenuation. What was successful in most cases was the combined use of a well fitted earplug (coated with grease) in the ear canal and a well designed earmuff over the ear.

The other method for monaural balancing is sometimes referred to as the transfer method. An earphone is on one ear (the transfer ear) and the other ear is uncovered. The loudness of the loudspeaker in the open ear is balanced by adjusting the earphone level in the other ear (the transfer ear). Once this equal loudness judgment is made, the signal to the transfer ear is left constant, the loudspeaker is turned off, and another earphone is placed over the open ear and its level adjusted to be equally loud as the sound from the transfer ear. This procedure has the same problem as the first method, namely two-ear listening to the loudspeaker source and one-ear listening with the transfer ear. A conventional earphone on the transfer ear does not provide sufficient attenuation. A well fitted earplug in the ear canal with a grease seal is again required with the earphone on top of the earplug.

The experimental equipment for the loudness balancing test was essentially the same as described earlier with a few exceptions. The Bekesy handswitch was replaced by an attenuator with 1-dB steps. This subject attenuator was in series with another attenuator located in the control room. Thus the experimenter could adjust the control room attenuator as sound source levels were changed so that the subject received no clues from the settings of his attenuator knob. Once the level of, say, the loudspeaker was set by the experimenter, the subject had complete control. He could switch the signal from one source to the other whenever he wished. This was done by a foot switch. His attenuator could vary the signal either to the loudspeaker or to the earphones depending upon which procedure was being used. The only variable that did not change was that the earphone or cone system was always the first signal the subject listened to, and he was instructed to listen to it long enough until there was no change in loudness or quality. Then simultaneously with removing the earphones, the subject switched the sound to the loudspeaker to compare the loudness of the two sounds. If they were different, he adjusted his attenuator and repeated the test. When he decided the two sources produced equally loud sounds, he left the attenuator set and repeated the on–off procedure several times to assure himself of his judgment. When finally convinced, he started (by another foot switch) the pressure recording system in the control room to record the left ear pressure and then the right ear pressure. After that he removed the earphones, turned on the loudspeaker (with his foot switch) and recorded the left ear pressure and the right ear pressure. While doing this he still listened to the two sounds to ensure that they were still equally loud. If they were, he said "OK"; if not the pressure measurements were voided and the procedure repeated. This was an easy procedure to follow. What made it easy was that, when the two sources were set for equal loudness, the sounds were so alike that, when the earphones were removed and the sources switched simultaneously, it was difficult for the subject to sense that the source had changed. If the seal was not good or distortion was causing an effect, the loudness was

FIG. 7. Photograph showing the experimental arrangement of the probes and microphones used to measure the pressure level in the ear canal when conventional earphones mounted in flat cushions served as the sources for binaural loudness balancing. The probes were attached to the ear by tape and fitted in the small notch at the forward part of the pinna. The acoustic seal of the cushion was not affected by this small plastic tube. When the earphones were removed for free-field listening, the probes measured the levels at the ear canal entrance.

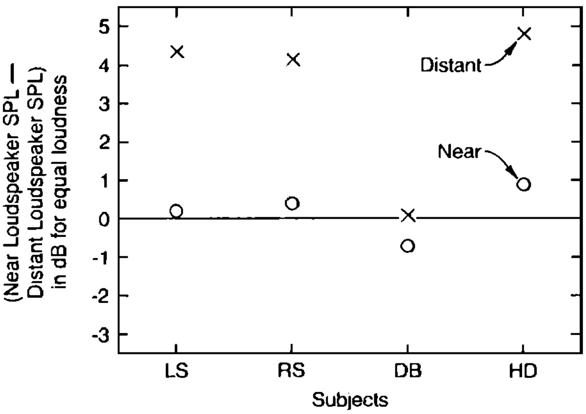

FIG. 9. The source location effect. The results shown are for four subjects performing monaural loudness balances between a loudspeaker near the head and a loudspeaker located across the anechoic chamber (about 4–5 m) from the subject. Differences in sound pressure level between the near source and the distant source are shown for a frequency of 100 Hz. Subject's chair was mechanically isolated from the floor. The subject's nonlistening ear was occluded with an earplug covered by earmuff.

the same but the quality differed so that it was obvious that the sources had changed.

Figures 7 and 8 show how the probe tubes, driver units, microphones, and preamplifiers were located and mounted with respect to the subject. The experimenter had no control except to set the level of the loudspeaker (or the earphone) and the attenuator in series with the subject's attenuator. Once this was done the subject switched from source to source as many times as he wished until he was finally satisfied that the two sources produced equally loud sounds. Then he recorded the pressure in each ear due to each source with the experimenter completely out of the "loop." The same probes measured the sound pressures for the loudspeaker source as for the earphone source. The same was true when using the cones as the subject kept the tip of the cones near the ear when out of the ear so that the loudspeaker pressure was measured with the same probe as the cone pressure. The wavelength of 100 Hz is so large compared to the distance the probe tip was from the ear canal that no corrections are involved for the free-field measurement. Since sound pressure differences were the measured quantity, calibration effects were not of consequence.

FIG. 8. Photograph showing the experiment arrangement of the probes and the cone type driver systems. When the subject removed the cones to listen to the loudspeaker source, the cones were held near the ear canal and the probe system measured the pressure level of the free-field stimulus.

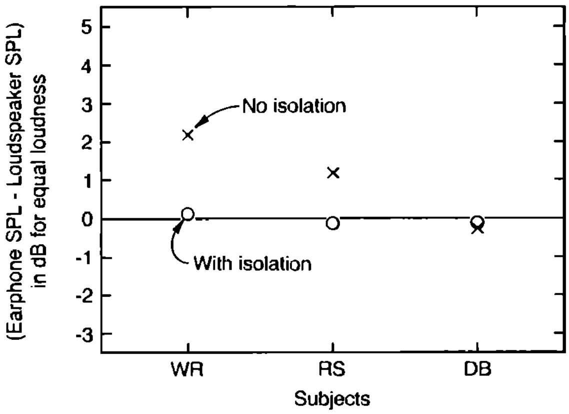

FIG. 10. Chair vibration effects. The differences in sound pressure level from an earphone source and from a distant loudspeaker source are shown for three subjects performing binaural loudness balances for 100-Hz signals from the two sources. One set of data was obtained with the subject's chair in direct contact with the woven wire floor of the anechoic chamber (no isolation). The other set was obtained with the subject's chair mechanically isolated from the chamber floor.

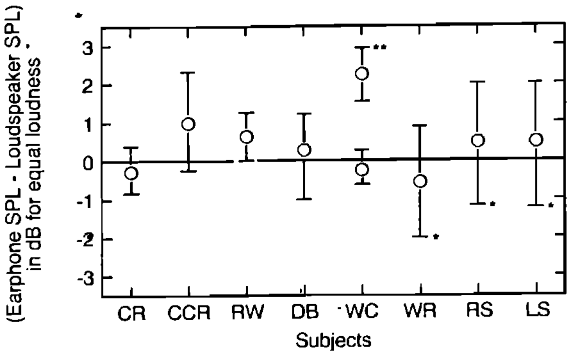

FIG. 11. The results of monaural loudness balancing at 100 Hz for various subjects using earphones in cushions as one source and the loudspeaker as the other source. The nonlistening ear was properly occluded; the subject's chair was mechanically isolated; the SPL range was 65–85 dB; and the bars are 95% confidence intervals. For subject WC two sets of data were obtained to show the effect of not occluding the nonlistening ear properly (shown as **). The data for WR, RS, and LS show larger confidence intervals due to the number of measurements being small (shown as *—where N ranged from 5 to 10).

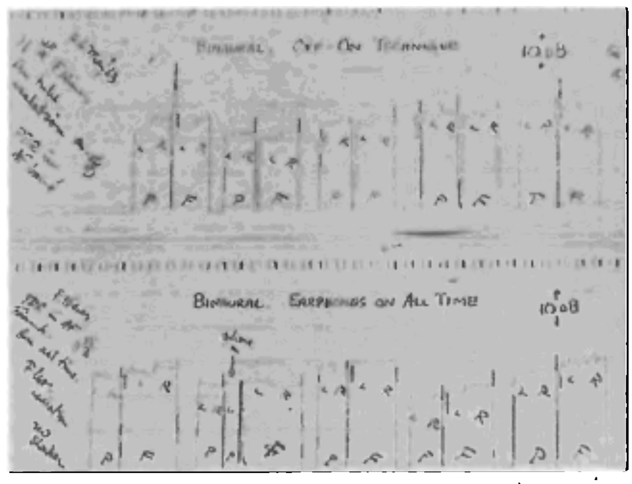

FIG. 13. Photograph of data for two separate sets of binaural loudness balancing using earphones as one source and a loudspeaker as the other source. The subject listened first to the earphone source and then the loudspeaker source. When satisfied that the two sources were equally loud, the subject actuated the pressure measurement system and recorded (upper record) the left (L) and right (R) pressure level under the earphone cushion (P); then recorded the pressure level with earphones removed and loudspeaker on (F) for the left ear canal (L) and the right ear canal (R). The process was repeated at least five times, and the subject was given a chance to rest. Note that the absolute sound pressure levels changed (the experimenter controlled this) between sets of pairs of balances. The lower recording is similar to the upper recording except that the earphones were left on the subject's head when the loudspeaker was the source. The subject was Francis Wiener.

C. Results

Figures 9 and 10 show the results of source location effects and chair vibration effects. The number of subjects is small in each case because once the effects were observed, steps were taken to correct the problem. The results for one subject (DB) are shown to illustrate the fact that not all subjects react the same way. However, all possible effects had to be evaluated even though they might be small.

Figures 11 and 12 show the results of a number of subjects (both male and female) performing monaural loudness balances using conventional earphone versus loudspeaker as sources and using the cone system versus loudspeaker. Notice in Fig. 11 the two different results for subject WC when the nonlistening ear was not properly occluded. A question might be raised as to why the transfer method was not used for monaural balancing. A few subjects verified that this method would work if the transfer ear were properly occluded, but the procedure was just too cumbersome and did not seem to add any additional information.

Turning to binaural balancing, Fig. 13 shows a typical series of pressure level measurements following five successive loudness balances with the top record showing the results when the earphones were removed from

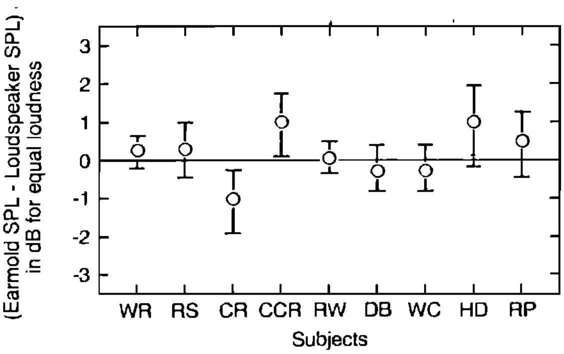

FIG. 12. The results of monaural loudness balancing at 100 Hz for various subjects using the conical earmold-driver system as one source and the loudspeaker as the other source. The nonlistening ear was properly occluded; the subject's chair was mechanically isolated; the SPL range was 65–85 dB; and the bars are 95% confidence intervals. (N typically ranged from 20 to 30 over four or five different days.)

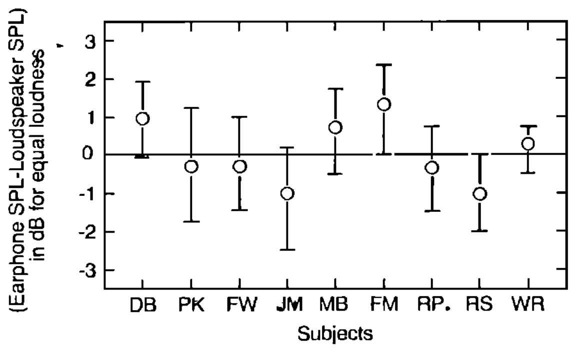

FIG. 14. The results of binaural loudness balancing at 100 Hz for various subjects using earphones in cushions as one source and a loudspeaker (with earphones *off* the ears) as the other source. Average differences for each subject in sound pressure levels are plotted. The subject's chair was mechanically isolated; the SPL range was 65–85 dB; and the bars are 95% confidence intervals.

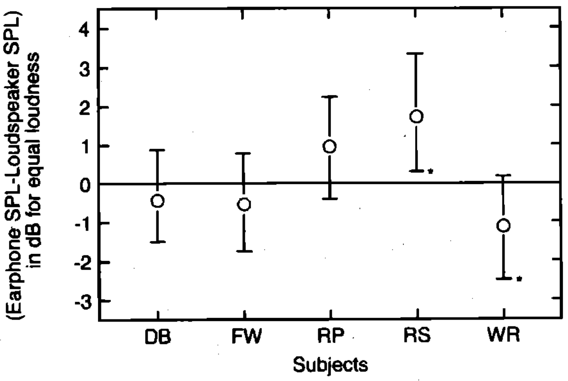

FIG. 15. The results of binaural loudness balancing at 100 Hz for various subjects using earphones in cushions as one source and a loudspeaker (with earphones left *on* the ears—no signal voltage on earphones) as the other source. Average differences for each subject in sound pressure levels are plotted. The subject's chair was mechanically isolated; the SPL range was 65–85 dB; and the bars are 95% confidence intervals. (*N for these data ranged from 6 to 10.)

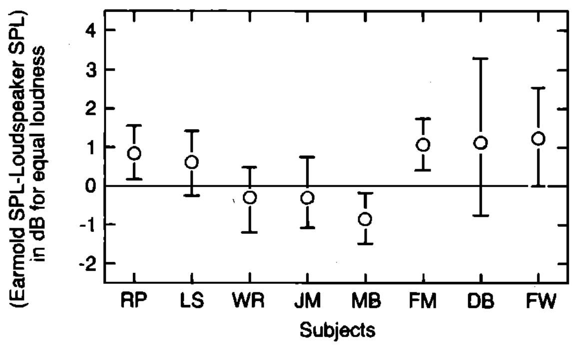

FIG. 16. The results of binaural loudness balancing at 100 Hz for various subjects using the conical earmold pressure system as one source and a loudspeaker (conical earmolds out of the ear canal) as the other source. Average differences for each subject in sound pressure levels are plotted. The subject's chair was mechanically isolated; the SPL range was 65–85 dB; and the bars are 95% confidence intervals.

the ears when listening to the loudspeaker, and the lower record showing the pressure level measurements where the earphones were left on the ears when the loudspeaker was the source and the 100-Hz signal from the loudspeaker reached the ear canal after a slight attenuation by the cushion. The value of this attenuation was unimportant since the probe tubes measured ear canal pressures. The reason both methods were used with the headphone data was that Wiener and Munson showed no missing 6 dB by this latter procedure but did show the missing 6 dB if the earphones were removed to listen to the loudspeaker. This result tends to verify the contention that their ABX procedure did not allow enough time for earphone source listening. It should also be stated that Francis Wiener was the subject for the data shown in Fig. 13. He was kind enough to participate in these experiments, and after many measurements and studying the pressure recordings agreed that when the experiment was performed this way there was no missing 6 dB.

The results for all of the subjects are shown in Figs. 14–16. Subject FW is Francis Wiener. It is interesting to note that he was one of the few subjects who preferred the quality of the earphone sound source to the cone sound source. The results show that the value of his standard deviation is smaller using earphones as the source than when using the cones as the source. This was also true for subject DB. The remaining subjects preferred the cone-type source.

D. Conclusions

All of the results certainly support the conclusion that, if the procedures used in these experiments are followed, there is no missing 6 dB for loudness balancing tests. Most of the data that have been previously reported were replicated and found correct for the procedures used in their experiments. It was difficult, however, to speculate whether the subjects of other authors exhibited the source location effect, as this determination can be made only by actually performing that particular test on each subject.

Finally, the problem stated by Beranek (1949) can be resolved for the frequencies below 500 Hz by the fact that the missing 6 dB should not exist if the proper precautions and procedures are observed, and above 500 Hz the curves disagree because of an unfortunate use of correction curves. Had the free-field data been corrected to ear canal entrance and compared to the earphone pressure measurements which were made at the entrance to the ear canal, there would have been rather good agreement [as confirmed recently by Killion (1978)]. Instead, both free-field data and earphone data were only partly corrected to eardrum pressure, resulting in the discrepancy. These findings were shown to Wiener when he served as a subject, Wiener having been the experimenter in Beranek's report.

It is hoped that the results presented in this paper will help put to rest, finally, the paradox of the missing 6 dB.

ACKNOWLEDGMENTS

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It was an interesting experience to return to notebooks written almost 20 years ago and find all of the data well documented and self-evident. My thanks go especially to Mead Killion who kept pressing me to publish these data.

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