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Contents

Power Based Measurements of Sound Insulation

by

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ABSTRACT

When determining the transmission loss of walls and partitions it is necessary to measure the sound pressure level difference between the two rooms and the equivalent absorption area in the receiving room. In this article the latter is determined using the reverberation time method and secondly by using a reference sound power source. Measurement results are presented which indicate that the transmission loss values obtained using the first method are too high at low frequencies. The reasons for this error are given as well as the advantages of using the latter method in which the sound power output of a reference sound source is compared to the sound power emitted by the wall into the receiving room.

SOMMAIRE

Lorsque l'on détermine la perte de transmission de murs et de partitions, il est nécessaire de mesurer la différence de niveau de pression sonore entre les deux chambres, et la surface d'absorption équivalente de la chambre réceptrice. Dans cet article, cette dernière grandeur est déterminée, premièrement par la méthode de mesure du temps de réverbération, et deuxièmement en utilisant une source de puissance sonore de référence. Les résultats des mesures sont présentés, et montrent que les valeurs de perte de transmission par la première méthode, sont beaucoup trop élevées aux basses fréquences. Les causes de cette erreur sont données, de même que sont présentés les avantages offerts par la seconde méthode; méthode dans laquelle la sortie de puissance sonore d'une source sonore de référence, est comparée à la puissance sonore émise, vers la chambre réceptrice, par le mur.

ZUSAMMENFASSUNG

Zur Bestimmung der Übertragungsverluste von Wänden und Raumteilern ist es notwendig, die Schalldruckpegeldifferenz in den beiden Räumen sowie die äquivalente Absorptionsfläche im Empfangsraum zu messen. In diesem Artikel wird die Bestimmung der letztgenann-

ten mit der Nachhallzeitmethode und zweitens unter Verwendung einer Bezugs-Schalleistungsquelle beschrieben. Es liegen Ergebnisse vor, die zeigen, daß die mit der ersten Methode gemessenen Übertragungsverluste bei tiefen Frequenzen zu hoch liegen. Dieses wird begründet und es werden die Vorzüge für die zweite Methode gegeben, die die Schalleistung, abgegeben von einer Bezugsschallquelle, mit der Schalleistung, die von der Wand in den Empfangsraum abgestrahlt wird, vergleicht.

Introduction

According to the Standard ISO 140 [1], the sound reduction index, R, (transmission loss) for a wall between two rooms is defined by:

$$R = 10 \log (W_1/W_2) \quad \text{dB} \tag{1a}$$

where W_1 is the sound power incident on the wall in the transmitting room (source room) and W_2 is the sound power radiated from the wall into the receiving room.

Under the assumption of diffuse sound fields in the transmitting and receiving rooms it has been common use to evaluate the transmission loss from the following formula, given in the standard

$$R = L_1 - L_2 + 10 \log (S/A_2) \quad \text{dB} \tag{1b}$$

where L_1 is the mean sound pressure level in the transmitting room,
L_2 is the mean sound pressure level in the receiving room,
S is the area of the wall specimen,
and A_2 is the absorption area of the receiving room.

In the standard, two methods are suggested for the determination of the absorption area:

a) by measuring the reverberation time T_2 of the receiving room and calculating the absorption area from Sabine's formula:

$$A_2 = \frac{0{,}163\, V_2}{T_2}$$

where V_2 is the volume of the receiving room,

b) by using a reference sound source.

While the first method (in the following called the classical method) has been widely acepted and is in common use everywhere, less attention has been paid to the second method. Attempts made by some investigators to use this method have shown poor agreement with results obtained using the classical method.

In this article both methods are investigated theoretically and experimentally. As the ISO Standard does not describe the method using the reference sound source, a measurement procedure (called the alternative method) has been outlined here.

Classical Method

As mentioned in the Introduction, this method is based on the formula:

$$R = L_1 - L_2 + 10 \log (S/A_2)$$

where the absorption area A_2 of the receiving room is evaluated from Sabine's equation:

$$A_2 = 0{,}163 \times V_2/T_2 \qquad (2)$$

Substituting (2) in (1b) we obtain:

$$R = L_1 - L_2 + 10 \log \frac{S}{S_o} - 10 \log \frac{V_2}{V_o} + 10 \log \frac{T_2}{T_o} - 10 \log 0{,}163$$

where $S_o = 1 m^2$, $V_o = 1\ m^3$ and $T_o = 1 s$ are reference values.

By rearranging and setting $(10 \log 0{,}163) \triangleq -8$ we obtain:

$$R = L_1 + 10 \log \frac{S}{S_o} - 6 - \left(L_2 + 10 \log \frac{V_2}{V_o} - 10 \log \frac{T_2}{T_o} - 14\right) \qquad (3)$$

Comparing the term in parenthesis with the formula given in the Standard ISO 3741 [2] for sound power measurements in a reverberation room:

$$L_w = L_p + 10 \log \frac{V}{V_o} - 10 \log \frac{T}{T_o} + 10 \log \left(1 + \frac{\lambda S}{8V}\right) - 14 \qquad (4)$$

it can be seen that the formulae are identical except for the Waterhouse correction term $10 \log (1 + S\lambda/8V)$, where S is the surface area of the room, V the volume of the room and λ the wavelength of the centre frequency of the band. Thus the term in parenthesis in eq. (3) indicates *that measurement of transmission loss using the classical method in reality involves determination of sound power W_2 emitted in the receiving room making use of the reverberation time.*

The so-called Waterhouse correction term, 10 log $(1 + \lambda S/8V)$ has been included in the formula (4) to compensate for the increased sound pressure and energy density along the walls [3] relative to the central portion of the room where the sound pressure is generally measured. When averaging the sound pressure levels L_1 and L_2 for transmission loss measurements according to eq. (3), the Waterhouse correction term should ideally be used for both the transmission and receiving rooms. They would, however, be cancelled out as they have opposite signs (see eq. (3)) if the two rooms are of the same size and shape. In practice for normal rooms used for dwellings the difference would be small and the terms can be neglected.

Alternative Method

In the alternative method the receiving room's absorption area A_2 in the formula:

$$R = L_1 - L_2 + 10 \log (S/A_2) \tag{5}$$

is determined by exciting the receiving room by a reference sound source and measuring the sound pressure level resulting from it. From theory it is known that

$$A_2 = \frac{4\rho c \times W_R}{p_{2R}^2} \tag{6}$$

where ρ is the density of air
c is the velocity of sound in air
W_R is the sound power emitted by the reference sound source
and p_{2R}^2 is the mean sound pressure squared (averaged over the entire room).

By substituting (6) in (5), setting ρc = 400 Ns/m^3 (mks rayls) and introducing the reference values $W_o = 10^{-12}$ W, p_o = 20 μPa and S_o = 1 m^2 we obtain:

$$R = L_1 - L_2 + 10 \log \frac{S}{S_o} - 10 \log \frac{W_R}{W_o} + 10 \log \left(\frac{p_{2R}}{p_o}\right)^2 - 10 \log \frac{4\rho c \;\; W_o}{S_o \;\; p_o{}^2}$$

i.e. $$R = L_1 - L_2 + 10 \log \frac{S}{S_o} - L_{WR} + L_{2R} - 6 \qquad (7)$$

where $L_{WR} = 10 \log (W_R/W_o)$

and $L_{2R} = 10 \log (p_{2R}/p_o)^2$

As in the case of the classical method, the Waterhouse correction term $10 \log (1 + \lambda S/8V)$ should be included for each of the sound pressure level measurements, i.e. one for the source room and two for the receiving room. However, the two sound pressure level measurements in the receiving room have opposite signs and therefore their correction terms are cancelled out requiring only the correction term for the source room:

$$R = L_1 + 10 \log \left(1 + \frac{\lambda S_1}{8V_1}\right) - L_2 + 10 \log \frac{S}{S_o} - L_{WR} + L_{2R} - 6 \qquad (8)$$

Rearranging we obtain:

$$R = L_1 + 10 \log \left(1 + \frac{\lambda S_1}{8V_1}\right) + 10 \log \frac{S}{S_o} - 6 - \left(L_2 + L_{WR} - L_{2R}\right)$$

Since L_2 and L_{2R} are the sound pressure levels in the receiving room when the transmitting room and the receiving room are excited respectively, it can be seen that the term $(L_2 + L_{WR} - L_{2R})$ is equal to the sound power level, L_{W2}, emitted into the receiving room by the wall under test (see eq.1a). *Thus determination of sound transmission loss by the alternative method in reality involves measurement of sound power W_2 emitted by the wall into the receiving room by the comparison method using a reference sound source.*

Sound Power

As mentioned earlier, measurement of transmission loss using the classical method in reality involves determination of sound power emitted into the receiving room making use of the reverberation time. However, in recent years it has been shown in the literature that at low frequencies, determination of sound power by this method gives lower values than those determined in a free-field over a reflecting plane according to ISO 3745 [4]. The latter method is generally known to give the most accurate results.

P.V. Brüel [5] has shown that the main reason for the discrepancies is due to the reverberation time being determined from the slope of the curve between —5 to —35 dB as suggested by ISO R 354 and 3382 instead of determining it from the upper part of the decay curve. Fig.1 is reproduced from [5] showing the decay curves and reverberation times for the two different slopes for each curve. Fig.2 shows the difference between sound power levels of a sound source determined in a reverberation room and in a free-field above a reflecting plane (for both reverberation times). As can be seen, better agreement is achieved when reverberation times determined from the upper part of the decay curves are used. As the classical method of transmission loss measurements involves indirect sound power determination using reverberation times, one would expect the same discrepancies to arise here, i. e. the transmission loss determined would give too high values at low frequencies as the sound power determined in the receiving room would be too low (see eq. 3). The problem of reverberation time

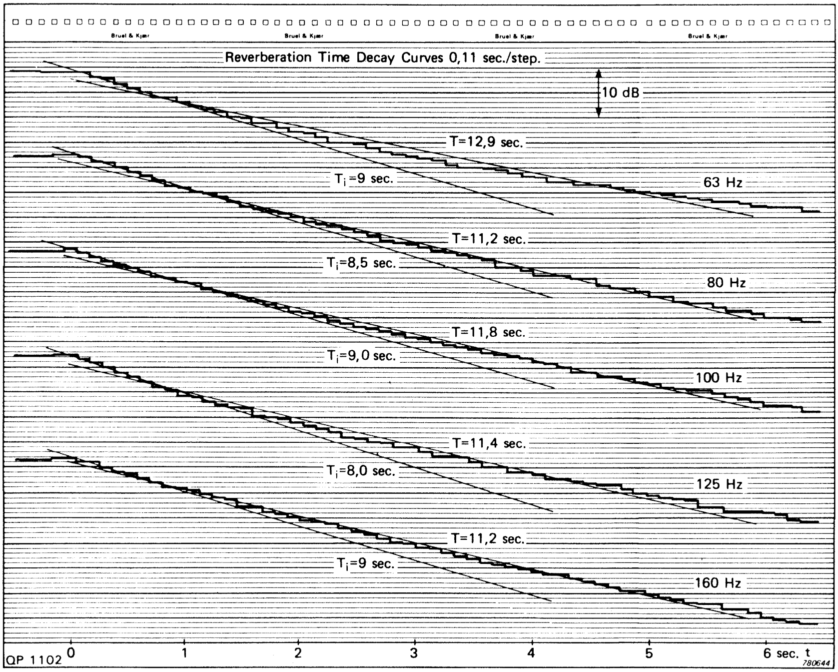

Fig.1. Averaged value of 1600 decay curves measured in a reverberation room (63 — 160 Hz)

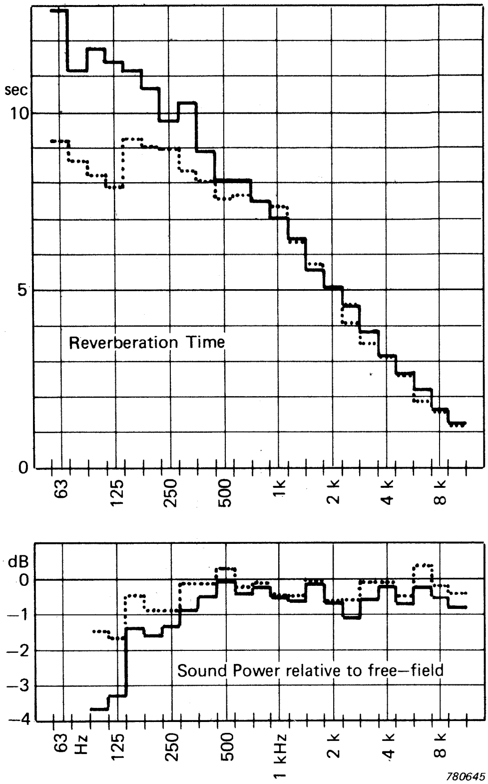

Fig.2. Difference between sound power determination in a reverberation room and in a free-field above a reflecting plane. Dotted curve obtained using early decay rate

measurement is avoided by making use of a reference sound source and determining the sound power emitted in the receiving room by the comparison method as shown for the alternative method.

Measurement Procedure for the Alternative Method

The instrumentation set-up for the determination of transmission loss by the alternative method is shown in Fig.3, where a 1/2" condenser microphone with flat diffuse field characteristics is used. As only sound pressure level measurements are required, the instrumentation is relatively simple. A Sound Power Calculator Type 7507 can be conveniently used as it operates as a 1/3 octave parallel analyzer in the relevant frequency range 100 — 10000 Hz and automatically corrects for

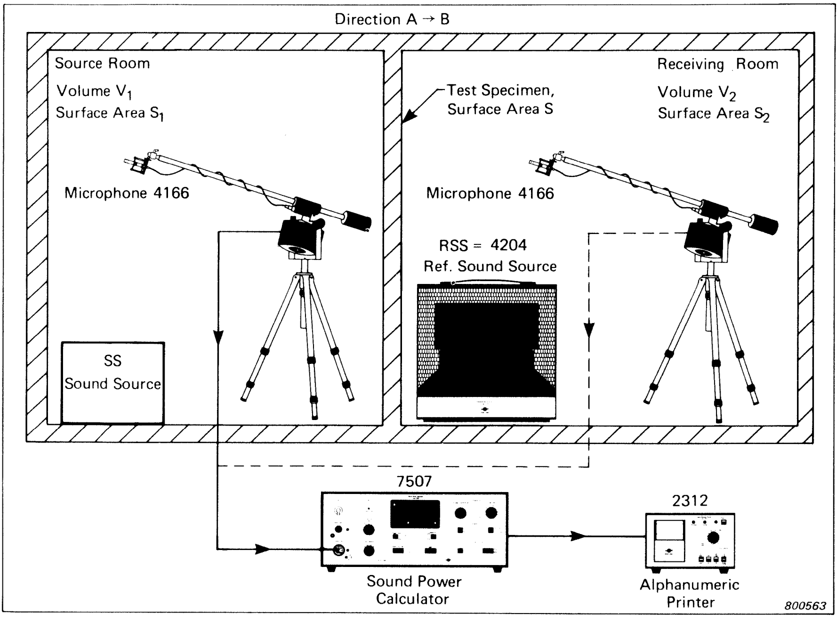

Fig.3. Instrumentation set-up for measurement of sound transmission loss using the alternative method

the background noise. The results are printed on the Alphanumeric Printer Type 2312.

The measurement procedure is as follows:

1. With the sound source (SS) switched on in the transmitting room (A), the averaged sound pressure levels L_1 and L_2 in the transmitting and receiving room (B) respectively are measured. If necessary, the levels are corrected for background noise.

2. The reference sound source (RSS) is turned on in the receiving room and the averaged sound pressure level L_{2R} in the receiving room is measured.

3. The transmission loss R in direction A → B is calculated from eq. (8).

If the transmission loss in direction B → A is desired, only one more measurement is required, namely, measurement of sound pressure level in room A while the reference sound source RSS is operating in room B (provided the reference sound source (RSS) was used as sound source (SS) in 1).

The reference sound source should be placed sufficiently far from the walls as the proximity of the walls influences the sound power output of the sound source. A distance of at least 1,5 m is recommended. In small rooms it may be difficult to satisfy this condition as the source should also be far enough from the microphone. In the appendix a method is described to compensate for the increased sound power emitted when the reference sound source is placed near a corner.

Measurement Results

a) Transmission Loss Measurements between two small office rooms

A cross-section of the two rooms is shown in Fig.4. The transmission loss was measured in one direction only using the classical and the alternative method. For both methods the same level difference $L_1 - L_2$ were used. The results are shown in Fig.5.

As expected, the values obtained at low frequencies using the alternative method are significantly lower than those obtained using the classical method. For the 100 and 125 Hz centre frequencies, however, the agreement is good. The reason for this is explained in the appendix.

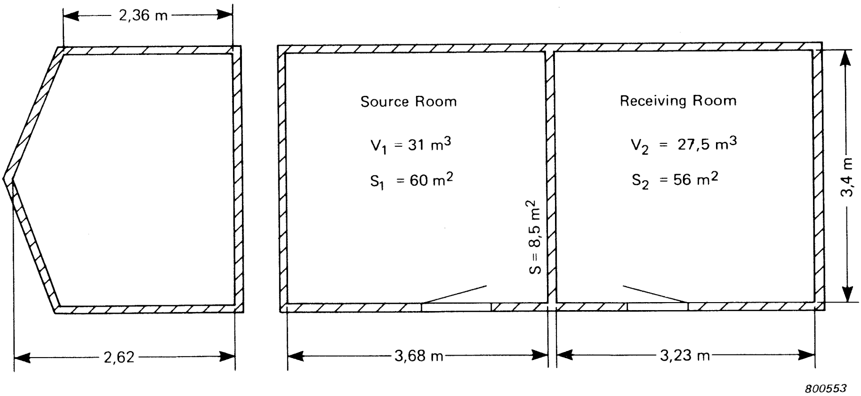

Fig.4. Cross section of the office rooms used for measurement of sound transmission loss

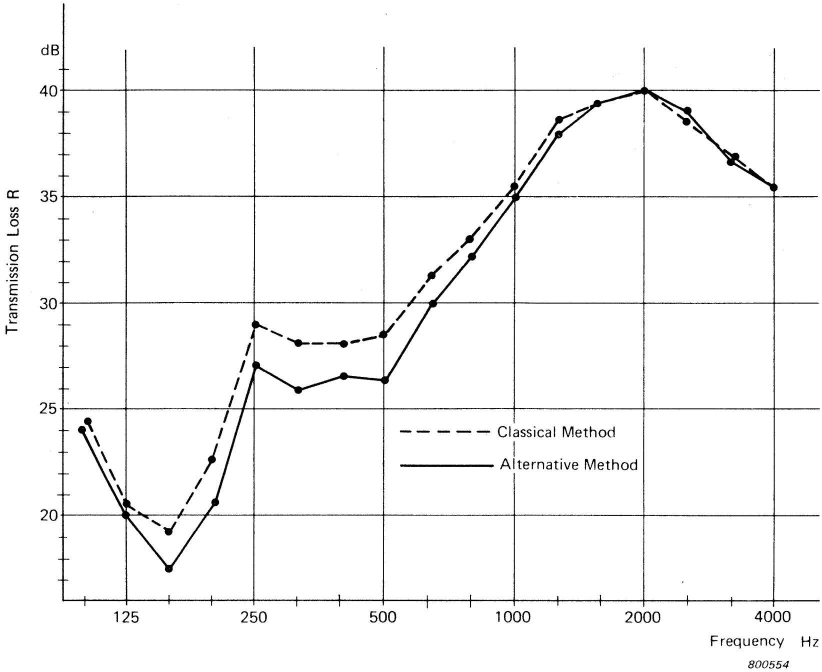

Fig.5. Sound transmission loss determined using classical and alternative methods

b) Transmission Loss Measurements between two reverberation rooms
To carry out measurements under well-defined conditions two reverberation rooms, A and B Fig.6, at the Acoustics Laboratory of the Technical University, Lyngby were used. A window (1,21 m × 1,21 m) was mounted in a highly insulated wall common to both rooms, the reverberation times of which were in the order of 5 s up to about 1600 Hz decreasing to about 2 s at 5 kHz.

Three independent measurements were made, two using the classical method and one using the alternative method, for both directions (A → B and B → A). The two classical methods used different instrumentation except for the loudspeakers and microphone paths. In each room one loudspeaker was mounted in the corner opposite the test wall: thus only one sound source position for each direction was used for the classical method. For averaging the sound pressure levels a microphone was rotated on a boom in the middle of the rooms.

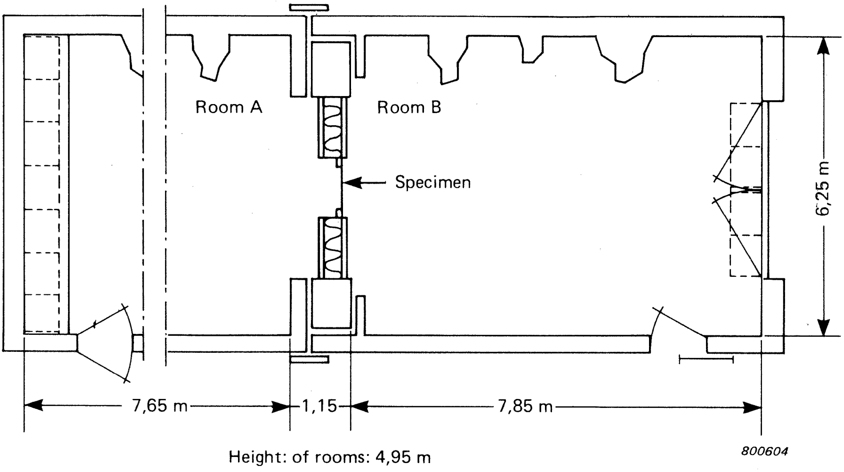

Fig.6. Cross section of the two reverberation rooms at the Danish Technical University (DTH) used for determination of sound transmission loss. A window was mounted in the test wall

For the alternative method the measurements were carried out using the instrumentation shown in Fig.3. The Reference Sound Source Type 4204 was used as the sound source in both rooms and the sound pressure levels in both rooms were averaged for three source positions for each direction A → B and B → A.

For each of the three methods the transmission loss values are compared for the two directions in Fig.7. It can be seen that better agreement between the two directions is obtained for the alternative method than for the two classical methods, the reasons for which are explained later.

To compare the three methods the mean values of the two directions are plotted in Fig.8. The spread in the results of the three methods can be clearly seen and can either be due to the errors in the measurement of the sound pressure level difference $L_1 - L_2$, or in the determination of the absorption in the room, since the formula for the transmission loss is made up of two components:

$$R = (L_1 - L_2) + K$$

where $K = 10 \log S/A_2$ for the classical method

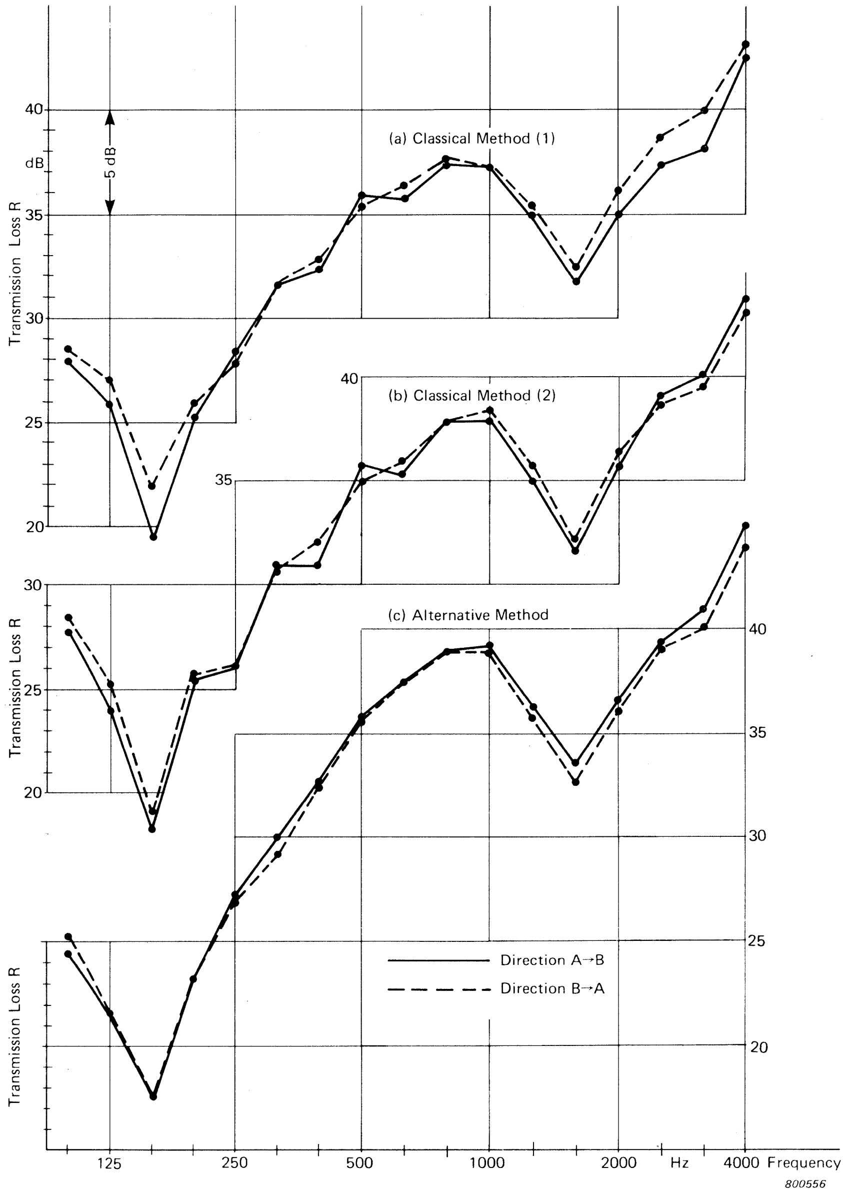

Fig. 7. Sound transmission loss determined in both directions using the classical and alternative methods

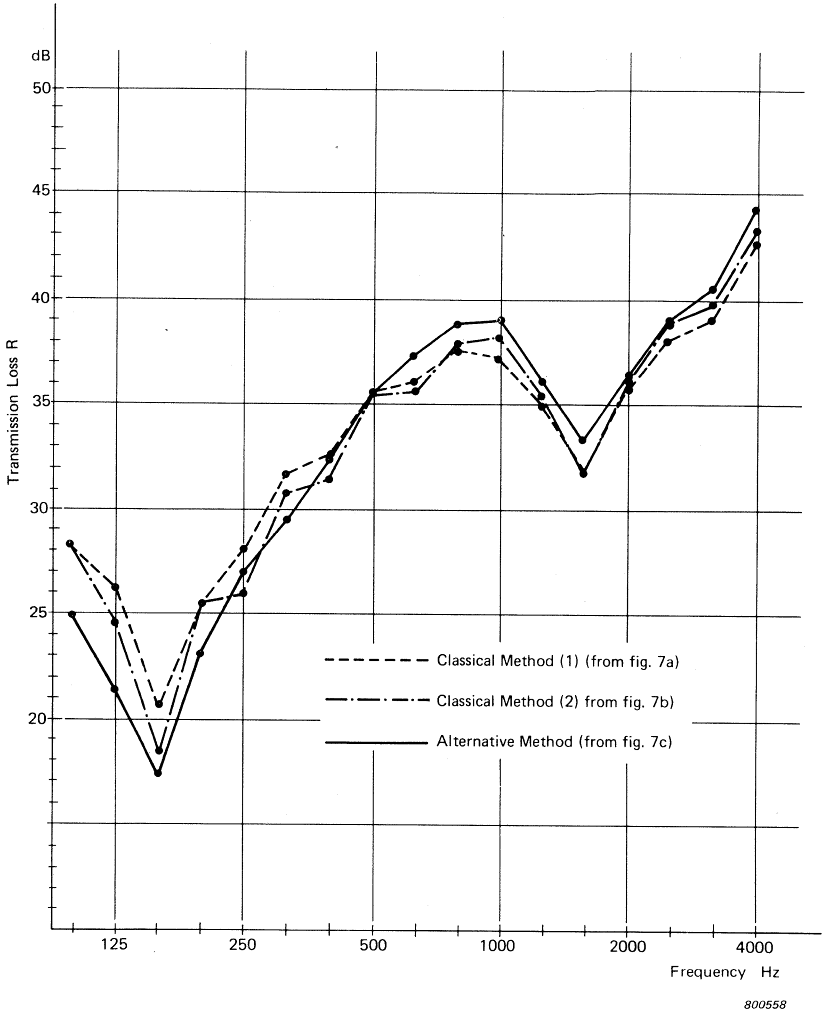

Fig. 8. Transmission loss mean values of both directions obtained from Fig. 7 for the three methods

and $$K = 10 \log \frac{S}{S_o} - L_{WR} + L_{2R} + 10 \log \left(1 + \frac{\lambda S_1}{8V_1}\right) - 6$$

for the alternative method (see eq. 8).

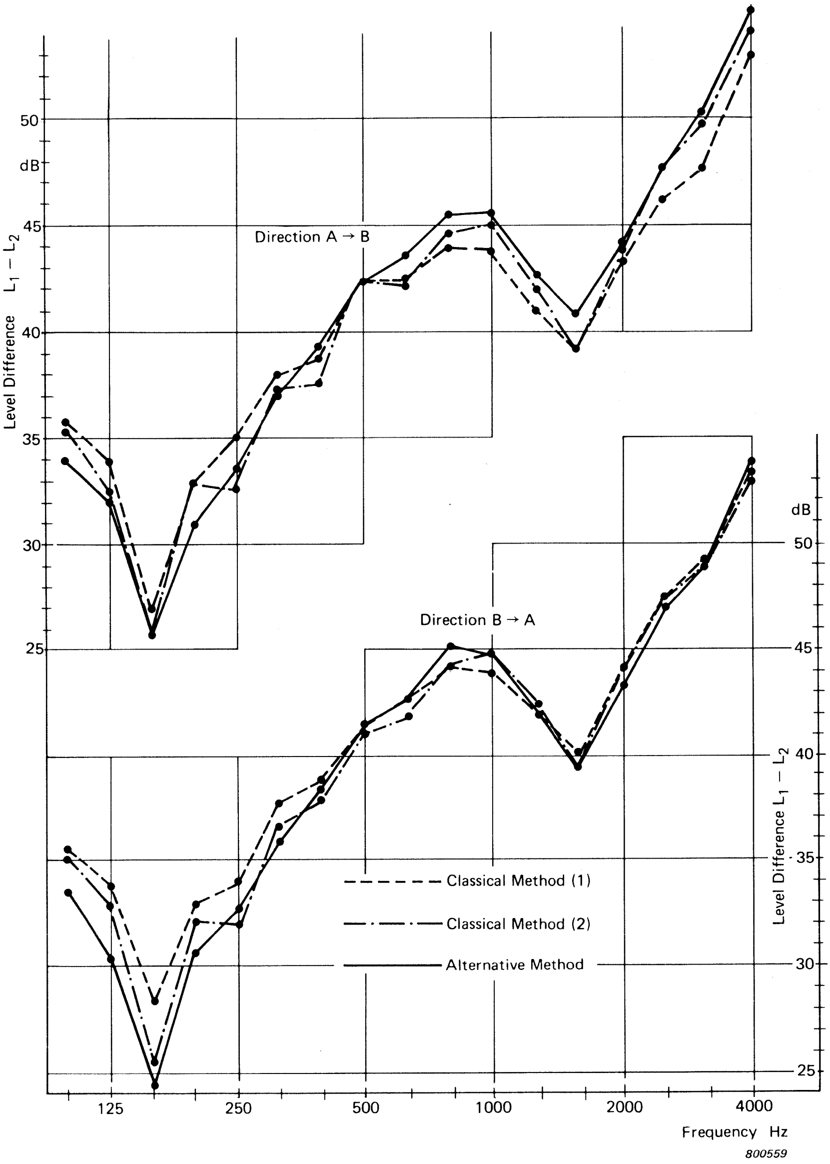

Fig.9. Level differences (L_1 — L_2) for the three measurements in both directions

In Fig.9 the sound pressure level difference (L_1 — L_2) for the three methods are compared for both directions, while the absorption terms, K, are compared in Fig.10. From the figures it is seen that the spread in (L_1 — L_2) measurements is greater than in the absorption term measurements. Furthermore, the spread in Fig.9 is practically over the whole frequency range whereas the spread in the absorption terms is mainly at the low frequencies. However, the problem of accurate sound pressure level difference measurements is common to all three methods and therefore does not reveal the merits of one method over the

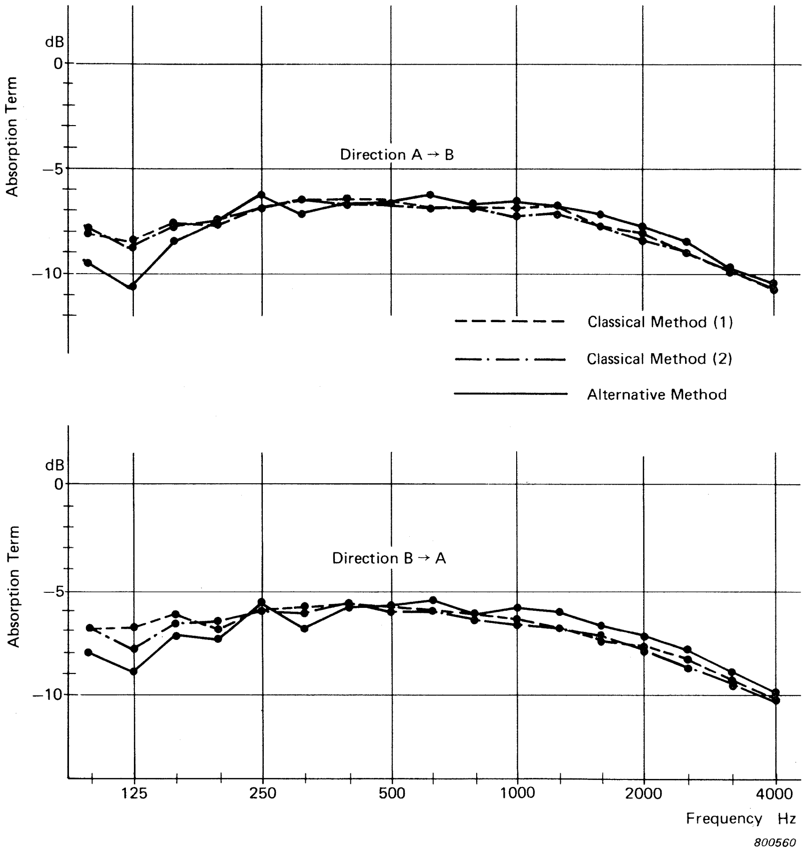

Fig.10. Absorption terms, K, for the three measurements in both directions

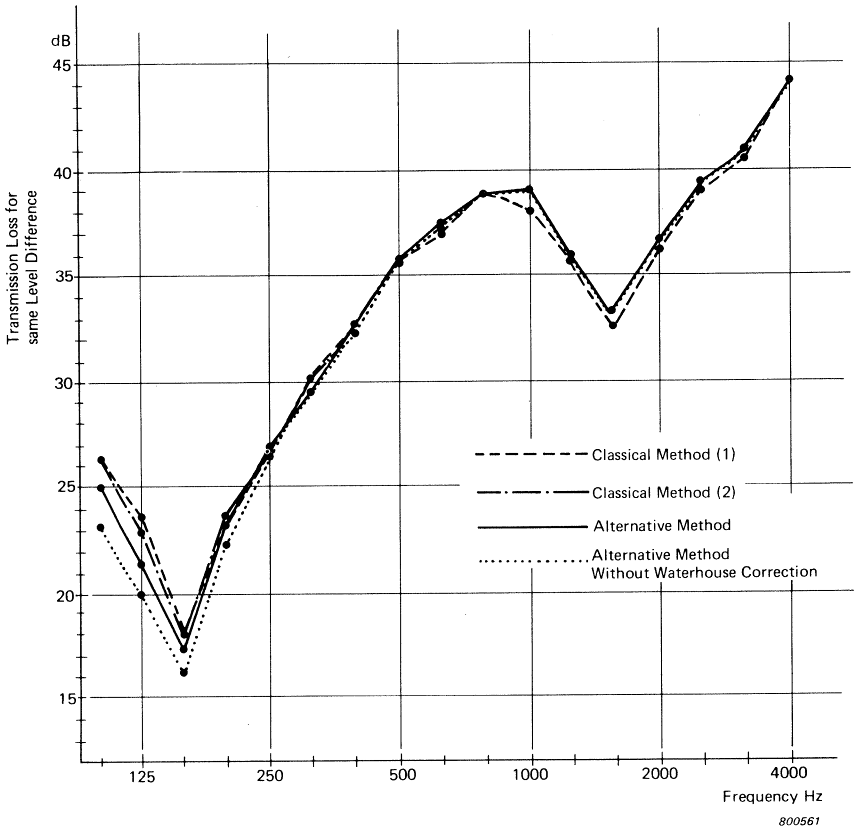

Fig.11. Transmission loss mean values of both directions using the same level difference from the alternative method

other. To effectively compare the methods, the transmission loss for the three methods should be plotted using the same sound pressure level difference. This is done in Fig.11, where the mean value of the transmission loss in both directions is plotted for the three methods using the same level difference from the alternative method.

As can be seen, the results are in good agreement except at the low frequencies, as expected. The dotted curve in Fig.11 shows the results of the alternative method *without the use of the Waterhouse correction term 10 log (1 + λS/8V) in the transmitting room, showing an increase in the discrepancy of the results.*

Discussion of Results

When a specimen is tested in a transmission suite, where the transmission and receiving rooms are approximately of the same size and properties, one would expect the transmission loss measurements to give the same values in both directions. Thus the use of the mean value of the two directions improves the validity of the results [9].

From Fig.7 it can be seen that the closest agreement in the transmission loss for the two directions is obtained for the alternative method. In view of the above criterion it would mean that the sound pressure level difference has been measured most accurately for this method since the spread in the absorption term values is seen to be less significant than in the (L_1 — L_2) measurements. This is probably because:

a) three sound source positions were used for the alternative method, while only one position was used for the two classical methods.

b) the averaging time was longer for the alternative method (3 × 64s) than for the classical methods (64s and 32s respectively).

c) an aerodynamic reference sound source (smooth frequency spectrum) was used for the alternative method instead of a conventional loudspeaker (which could have peaks in the frequency spectrum due to resonances in the cabinet).

Conclusion

It has been shown that measurement of transmission loss using the classical method in reality involves determination of the sound power emitted by the wall into the receiving room, making use of the reverberation time. When the alternative method is used the sound power emitted is determined by the comparison method, using a reference sound source.

Experimental results have revealed that the errors in the measurement of sound pressure level difference (which are present in both methods) are greater than in the determination of the absorption term in the room. The difference between the two methods is revealed at low frequencies where the classical method tends to give higher values of transmission loss than the alternative method. This is because the sound power emitted by the wall and determined by the reverberation time method for the classical method is lower than that determined in a free-field. This error is avoided in the alternative method since the sound power is determined using the comparison method. Here it is important that the Waterhouse correction term $10 \log(1 \times S\lambda/8V)$ for the transmission room is taken into account. Another advantage of the alternative method is that the instrumentation required is simple, since only sound pressure level measurements need to be carried out (the time consuming reverberation time measurements are avoided).

Acknowledgement

Thanks are due to Mr. Nic Michelsen of the Acoustical Laboratory, ATV, Lyngby for providing some of the results and for valuable discussions.

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Appendix

A reference sound source is normally calibrated in a free-field over a reflecting plane. However, the environment in which it is normally used in practice is different and therefore has some influence on the sound power output of the source (for example, when it is placed near a wall). Measurements have shown that if the distance to the walls, or other large objects, is greater than 1,5 m the sound power output is practically unaffected. Fig.A 1 shows the sound power output from a Reference Sound Source Type 4204 placed on the floor in a corner 0,35 m and 1 m from the two walls, relative to the sound power output when placed 1,5 m from the walls. From the figure the increase in the sound power emitted at low frequencies can be seen.

If an unknown sound source of approximately the same physical dimensions is placed on the floor near a wall, it would be similarly affected and would emit higher sound power levels at low frequencies. When the reference sound source is used to determine the sound power emit-

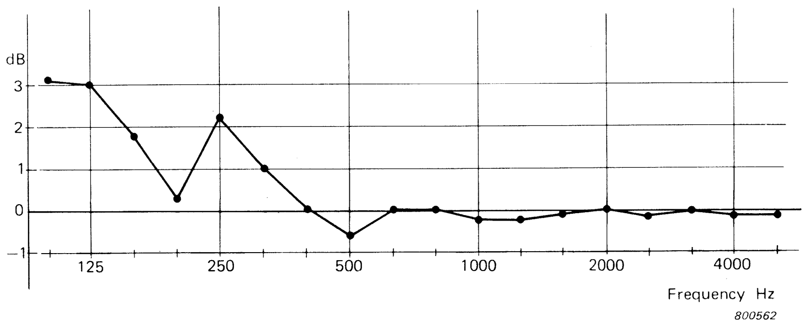

Fig.A1. Sound power level emitted by Reference Sound Source Type 4204 when placed on the floor in a corner 1 m and 0,35 m from the two walls, relative to the power emitted when placed 1,5 m from the two walls ted by the unknown source using the comparison method, two possibilities arise:

1) The reference sound source can be placed in the same position as the unknown source, in which case the sound power determined for the unknown source *would be that it would emit in a free-field over a reflecting plane.*

2) The reference sound source could be placed away from the walls (at 1,5 m) in which case the sound power determined for the unknown source *would be that it actually emits in situ.*

It is here assumed that the sound pressure level comparisons are made in the diffuse field.

In determining the sound power emitted by a wall, with the reference sound source placed away from the walls, the sound power determined would be that the wall actually emits — in agreement with the definition of transmission loss.

However, a particular problem arises when the receiving room is small such that the reference sound source cannot be placed at least 1,5 m away from the walls to allow for the microphone to be placed far enough from the sound source. In this case the reference sound source can be placed in a corner at a distance of 0,35 m and 1 m from the two walls. The increase in the sound power level emitted from Fig.A1 can be added to the calibrated values of the sound power source.

However, this technique was not used in the case of the transmission loss measurement between two small office rooms of Fig.4, where the sound source was placed 0,5 — 1 m from the walls. The increased sound power emitted by the reference sound source at low frequencies was not accounted for. Therefore the transmission loss values at 100 and 125 Hz in Fig.5 for the alternative method is higher than what it should have been.

Acoustical Measurement of Auditory Tubal Opening

by

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ABSTRACT

Several methods for the examination of the Eustachian (auditory) tube have been reported, however, most of them have not been generally accepted in everyday clinical use. This article discusses a simple and reliable method — sonotubometry — for accurate measurement of the tubal opening by using sound transmission through the Eustachian tube to the external auditory canal. It also analyses the acoustic events occurring during swallowing while a steady tone is delivered into the nose.

SOMMAIRE

Plusieurs méthodes d'examination des trompes d'Eustache ont été étudiées, cependant la plupart d'entre elles n'ont généralement pas été acceptées comme méthodes de soins courants dans les hôpitaux. Cet article traite d'une méthode simple et fiable, la sonotubométrie, pour la mesure précise de l'ouverture du canal auditif par transmission sonore vers le conduit auditif externe à travers la trompe d'Eustache. Elle analyse également ce qui se passe pendant la déglutition en injectant un ton stable dans le nez.

ZUSAMMENFASSUNG

Verschiedene Methoden zur Untersuchung der Eustachischen Röhre (Tuba auditiva) wurden veröffentlicht, jedoch haben sich die meisten im täglichen klinischen Gebrauch nicht durchgesetzt. In diesem Artikel wird eine einfache und zuverlässige Methode — die Sonotubometrie — zur genauen Messung der Röhrenöffnung diskutiert. Sie verwendet die Schallübertragung durch die Eustachische Röhre zum äußeren Gehörgang. Ebenso werden die akustischen Ereignisse analysiert, die während des Schluckens auftreten, wenn ein gleichmässiger Ton in die Nase gegeben wird.

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Introduction

The Eustachian or auditory tube forms a mucous-lined connection between the middle ear and the nasopharynx. The primary function of the Eustachian tube is the equalization of the air pressure between the middle-ear cavity and the atmosphere. The middle ear is normally aerated when the Eustachian tube is opened by muscular action during swallowing. The most common malfunction of the auditory tube is the inability of the tube to open, and less commonly, the failure of the tube to close.

There is general agreement that adequate Eustachian tubal function is necessary for successful middle-ear surgery. If the tube fails to open, air is slowly absorbed from the middle ear and the consequences will be permanent retraction of the tympanic membrane and hearing loss. Thus a reliable assessment of the Eustachian tubal function is of great importance for planning surgical procedures, since a correctly functioning Eustachian tube is an essential prerequisite for post-operative aerated middle ear.

Modern aviation, with its rapid changes in altitude, and rapid plunging by divers, often involve large variations in surrounding pressure. These changes may not be equalized in the middle ear because of a poor Eustachian tubal function in some people, and may cause hearing loss, vertigo and pain. It has been suggested that, before starting training for flying or diving, people should be tested with regard to the function of the auditory tube, in order to avoid choosing unsuitable employment or hobbies.

Various methods for the examination of the Eustachian tube function have been reported with the goal of testing the tubal function in an objective way. One of the methods used for this purpose is based on the transmission of sound through the momentarily opened Eustachian tube, caused by swallowing. In this method the test sound is introduced into the nostril and recorded at the side of the ear. A general discussion of the problems in the earlier studies of the sound-conduction method can be found in reference [1] and therefore will not be dealt with here. The following article will be concerned with a new method of this type — sonotubometry. For a more detailed discussion of sonotubometry the reader is referred to references [1, 2].

Measuring Equipment

The block diagram of the equipment designed for this sound conduction test is illustrated in Fig.1. The equipment consists of an insert ear-

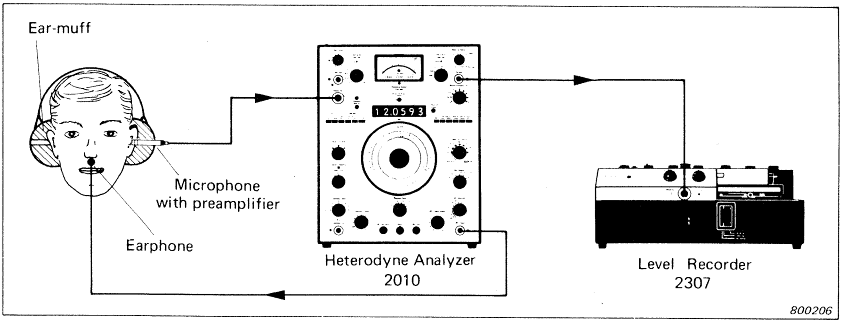


Fig. 1. Instrumentation set-up for the test

phone (Hearing Aid Earphone, Oticon A/S, Type AF M8) used as a sound source, a calibrated Condenser Microphone Type 4134 connected to a Preamplifier Type 2619 and embedded in a circumaural ear-muff (Exel OY, Silenta-Super), a Heterodyne Analyzer Type 2010 functioning as a signal source, an amplifier and as a filter, and a Graphic Level Recorder Type 2307.

The insert earphone was connected to an interchangeable nasal olive tip, held snugly by the patient in one of his nostrils. The microphone was coupled into the external auditory meatus with a probe of suitable size (Fig.2). A soft standard ear tip at the end of the microphone probe was slightly compressible enabling it to conform to the contours of the external auditory canal and ensuring a comfortable seal. A thick plastic window was mounted in the wall of the ear-muff and a small light was mounted inside the ear-muff enabling the tip of the microphone probe to be carefully inserted into the external auditory meatus. In the other ear-muff there was a hole for giving instructions to the patient when necessary during the examination. The amplified output of the microphone was fed through a 3,16 Hz band-pass filter in the frequency analyzer in order to suppress background noise, and the sound pressure level from the filter section was recorded by the level recorder. Following trials with various combinations of writing and paper speeds, a writing speed value of 1250 mm/s and a paper speed value of 10 mm/s were chosen, which gave a sufficiently clear record of the essential features of the response.

Fig.3 shows typical records of tubal openings obtained on an otologically normal subject with and without test tone delivered into one nostril. A low level signal, due to vibration of the soft tissues and bones of

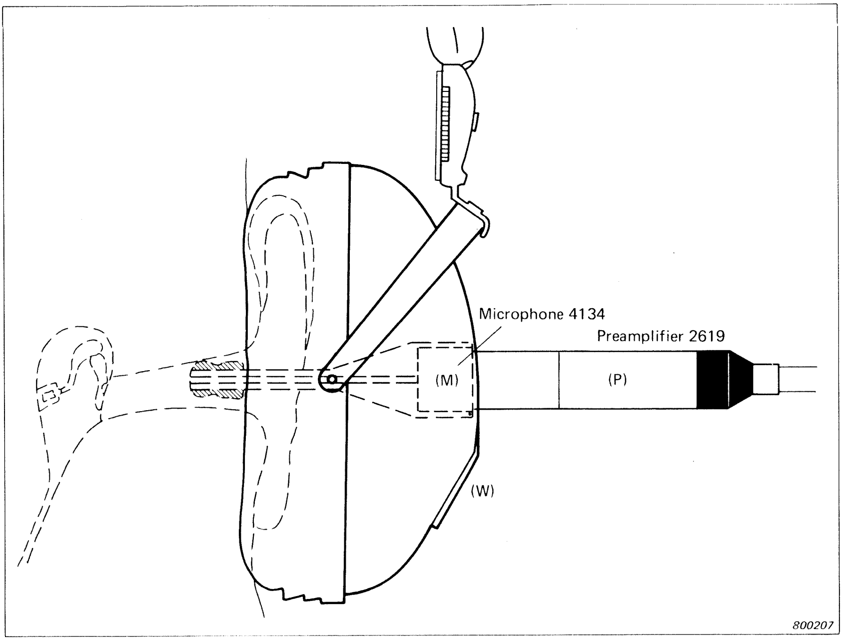


Fig.2. A sketch of the ear-muff with a microphone M, a preamplifier P and a plastic window W

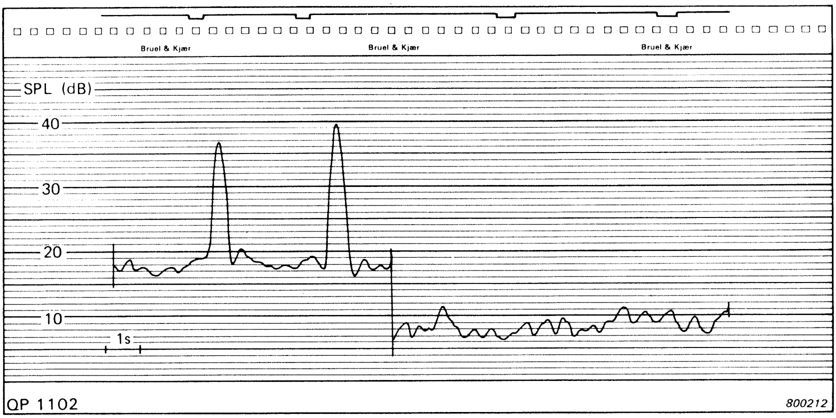


Fig.3. Typical records of tubal openings with a 7 kHz tone (left) and without tone (right) introduced into the nose and recorded at the external auditory meatus. At the top of the chart the markings of the swallowing moments

the head, is recorded as background level and becomes the base-line on the recording paper.

Preliminary experiments

In order to evaluate the mechanics of sound transmission through the Eustachian tube during swallowing and to interpret the acoustic phenomena related to tubal function as measured from the ear canal, the following experiments were carried out.

Sensitivity of earphone to pressure change

During ordinary swallowing the open state of the nose does not allow any pressure changes to develop in the nasopharynx, but under conditions of nasal obstruction pressure changes in the nasal cavity may be generated. It is thus possible that these pressure changes (+ 32 mm Hg to —34 mm Hg) in the nasal cavity may influence the earphone diaphragm. The effect of these rapid changes of static air pressure on the sensitivity of the earphone was determined as follows.

The earphone was made air-tight by means of a suitable adapter to the coupler of the Pistonphone Type 4220, and a pressure change in the coupler was produced manually by means of a syringe, and registered with a manometer. Here the earphone was used as a microphone, because the relative change in its sensitivity is the same as when it is used as a sound source, and the Pistonphone was used as a constant level sound source (124 dB SPL at 250 Hz frequency). The output voltage of the earphone was measured as a function of the static pressure generated in the coupler, and fed to the Level Recorder.

The change in the output voltage of the earphone was found to be less than 1 dB for an abrupt change of static pressure from + 40 cm H_2O to —30 cm H_2O. Thus possible pressure changes in the nasal cavity do not influence the sensitivity of the earphone during swallowing and can be disregarded.

Transfer function between nostril and ear canal in the frequency range 100 — 2000 Hz

A sinusoidal test sound was delivered to one nostril through the nasal olive and the frequency response curve was recorded in the range 100 — 2000 Hz from both ear canals and also from the other nostril on 15 subjects without them swallowing.

Fig.4 shows some frequency response curves (A) when recorded from the ear canal and (B) from the other nostril. As can be seen they are

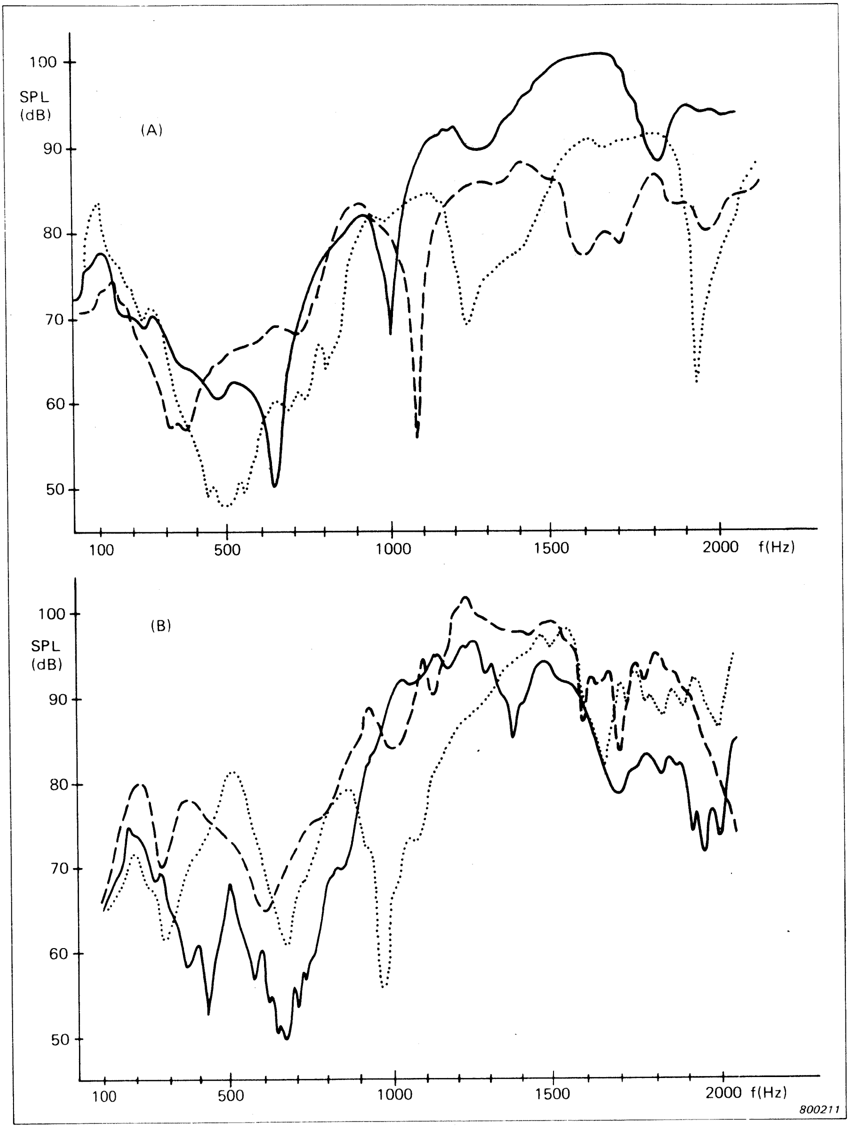


Fig. 4. *Individual response curves in the range 100 — 2000 Hz as measured (A) from the external auditory canal, and (B) from the other nostril. The curves drawn with a continuous line represent the same subject*

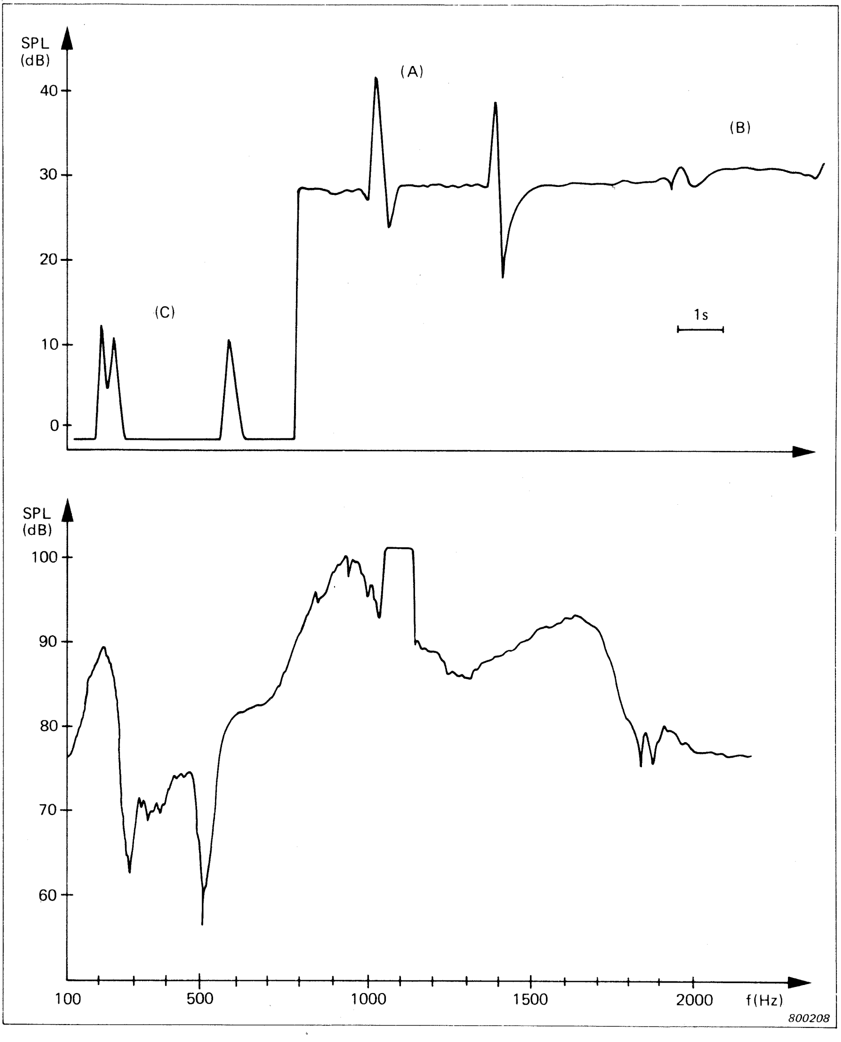


Fig.5. Tubal opening response with a test tone of 400 Hz (A) and 500 Hz (B) and without the test tone (C). Below, frequency response curve from 100 — 2000 Hz measured at the ear canal of the same person

similar in form but the levels of the base-line are different depending upon whether the test sound was picked up from the other nostril or from the external auditory meatus. There are resonance and antiresonance peaks in all of the recorded curves and vary significantly even for the same subject in successive measurements.

Fig.5 shows how the test tone of 500 Hz (B) is attenuated, while the one of 400 Hz (A) describes the tubal opening in successive swallowings very clearly. The tubal opening response without the test tone (C) exhibits a component of 400 Hz from the swallowing sound which was able to pass through the filter. On the frequency response curve of the same person a corresponding antiresonance peak can be seen at 500 Hz.

Spectrum of swallowing sound

The frequency analysis of the swallowing sound itself was measured from the external auditory meatus in 14 normal adults. The test procedure was as follows: each subject, while sitting in a relaxed position quietly, and with the mouth closed was asked to swallow either a sip of water or saliva several times. The sound of swallowing was picked up by the microphone inserted in the circumaural ear-muff, and a third-octave spectrum was obtained from both ears of each subject using a Real-Time 1/3 Octave Analyzer. The integration time of the analyzer was chosen such that the complete event was averaged and captured on the display.

The spectra of all the swallowing sounds showed to be similar on the display screen. 35 of these spectra were chosen at random, the mean value of which is shown in Fig.6 where the broadband character of swallowing sound can be seen. The variations in the spectra from different swallowings of the same subject were most significant in the frequency range of 100 — 2000 Hz, and above 7 kHz the sound pressure did not exceed the level of 30 dB.

The frequency response curves measured from the external ear canal and nostril were similar in form to the spectra of swallowing sound in the range of 100 — 2000 Hz. Thus, the swallowing sound is also influenced by the resonance and antiresonance effects of nasal and oral cavities, hypopharynx and pharynx. On the basis of these data the frequencies most suitable for the Eustachian tube opening measurements would be the high frequencies, above 6 kHz, where the intensity of swallowing sound is weakest.

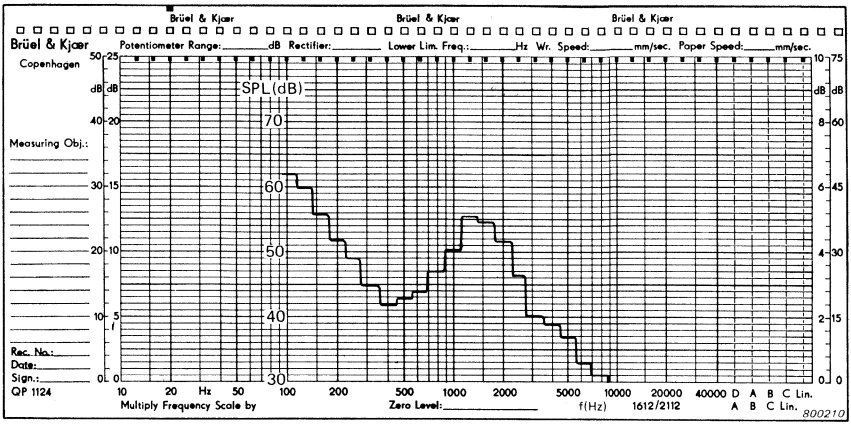


Fig.6. An average third-octave spectrum of swallowing sound recorded at the external auditory meatus

Tubal opening response in the frequency range 1 — 20 kHz

Measurements of the tubal function were made on 42 otologically normal persons. After delivering the test tone to the nose the subjects were asked to swallow first a sip of water, and then saliva. This was done in one kHz increments in the range of 1 — 20 kHz. For analysis of

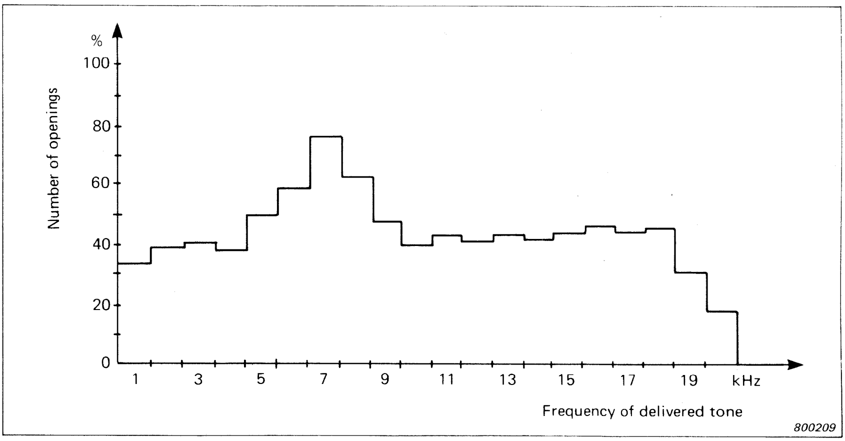


Fig.7. Tubal opening responses in 42 otologically normal subjects in the frequency range 1 — 20 kHz

the results an opening response was considered affirmative if the increase in the sound pressure level was $\geqslant 5$ dB at the moment of swallowing. It is probable that with this criterion some smaller openings are overlooked, but pen excursions are easily distinguished from the background activity recorded from the external auditory canal. It can be seen (Fig.7) that the response of the tubal opening was most evident at 6, 7 and 8 kHz.

Conclusions

The results of the experiments revealed that the useful frequency range for measurements by the sound conduction method is above 6 kHz. One may reasonably expect that various kinds of resonance effects of head cavities also influence the higher frequencies (> 6 kHz), but the wider dynamic range here makes it possible to record the acoustic phenomenon. Since the use of only one frequency did not always provide reliable information on account of the resonance effects, use of three frequencies (6, 7 and 8 kHz) is generally advisable. Ambient noise does not interfere with the test and even small changes in the acoustic energy can be recorded. The change in the sound pressure level, indicating tubal opening, was registered in 95% of normal subjects during swallowing. Repeated tests showed good reproducibility and the experiments performed in the presence of the upper respiratory tract infection testify to the sensitivity of this method. A thorough discussion of these experiments can be found in reference (1).

Sonotubometry is a physiological test and as such gives a reliable picture of the opening of the Eustachian tube during swallowing.

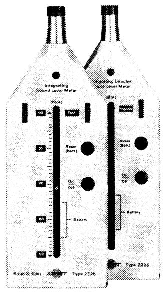
References

1. H. VIRTANEN 1977: Eustachian tube sound conduction. Sonotubometry, an acoustical method for objective measurement of auditory tubal opening. M. D. Thesis, Univ. Helsinki.

2. H. VIRTANEN 1978: Sonotubometry. Acta Otolaryngol. (Stockh.) 86: 93 — 103.

News from the Factory

Integrating Sound Level Meter Type 2225 and Integrating Impulse Sound Level Meter Type 2226



Both the Types 2225 and 2226 are slim (22 mm) lightweight (370 g), pocket-size instruments intended for noise and sound level measurements to Type 2 standards. They offer a wide range of measurement possibilities previously available only on larger, more comprehensive instruments. Special emphasis has been placed on ease of use, with a clear linear display, automatically ennumerated scale, and simple control layout.

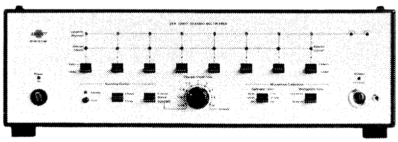
Both Types measure A-weighted sound levels with either "Fast" or "Slow" time constants and "60 s L_{eq}", from which SEL (Sound Exposure Level) can be easily found. The "60 s L_{eq}" mode eliminates the uncertainties of visually averaging a moving display when measuring a fluctuating sound level. In addition, the Type 2225 has a "Peak Hold" feature with a rise time of 30 μs for measuring absolute peak values of impulsive noise. Type 2226 has an "Impulse" time constant to international standards; and normal continuous readout or "Max. Hold" of the sound level may be selected. A DC output socket enables all functions, including "60 s L_{eq}" to be recorded continuously on a level recorder for long term monitoring purposes, from either instrument.

Fitted with a Prepolarized Condenser Microphone Type 4175 as standard, the total measuring range is from 25 dB(A) to 140 dB(A) peak in four 40 dB display ranges, 20 to 60, 50 to 90, 80 to 120, and 100 to 140 dB(A). Crest factor is 20 dB over the top of the display (except in the uppermost (100 to 140 dB(A)) range where the maximum measurable value of 140 dB(A) peak is overriding) increasing linearly with falling display value to a maximum of 40 dB (50 dB for "60 s L_{eq}"). Readout is via a linear "thermometer" display of light emitting diodes giving a resolution of 0,5 dB. The brightness of the display is controlled automatically with respect to the ambient light level so that it is readable under all lighting conditions, even in direct sunlight.

The microphone and adapter can be removed from the instrument, enabling the microphone to be mounted remotely using cables AO 0185 (3 m) and AO 0186 (10 m).

Power is supplied by three widely available IEC type R6, 1,5 volt cells QB 0013 (supplied) giving up to 30 hours of measurement. Both instruments are delivered with a Prepolarized Condenser Microphone Type 4175, a leather carrying case, foam windscreen, and 2,5 mm plug for the output socket.

8 Channel Multiplexer Type 2811

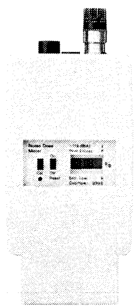


The Multiplexer is primarily intended for multi-microphone measurements with the Sound Power Calculator Type 7507. Scanning of up to eight microphone or direct input channels may be controlled manually, automatically from a built-in clock (9 dwell times, 1/16 to 16 seconds), or externally. Up to four 2811's can be used together to multiplex a total of 32 channels. An IEC 625-1 compatible interface permits external scanning by a controller, and independant scanning of a second multiplexer built into the 2811. Under IEC bus control, the 2811 can stop and start Types 1405 Noise Generator and 4205 Sound Power Source for reverberation-time measurements, and can start averaging in Type

7507. The 2811 has facilities for by-passing or selecting individual channels, and resetting the scan. It may also be used for scanning other signal sources, e.g., vibration measuring channels.

Channel input is by standard B & K 7-pin socket accepting Microphone Preamplifiers Types 2619 or 2627, or direct input via standard BNC sockets. There is a choice of 0 V, 28 V or 200 V polarization voltage. The BNC sockets double as channel outputs for recording or monitoring. Individual ± 3 dB channel sensitivity adjustments and a dual-LED tuning-type indicator are provided for calibration using Pistonphone Type 4220 or Sound Level Calibrator Type 4230. The frequency response is 2 Hz to 200 kHz ± 0,5 dB. Crosstalk is better than —80 dB up to 20 kHz, —60 dB up to 200 kHz.

Personal (OSHA) Noise Dose Meter Type 4431



The Personal Noise Dose Meter Type 4431 is a completely self-contained, pocket-size unit which measures the true accumulated noise exposure in accordance with OSHA. A digital display gives continuous reading of the percentage of the allowable noise exposure to which the wearer has been subjected.

As standard a half inch microphone Type 4125 is mounted directly on the Noise Dose Meter, but if desired it can be mounted on a Microphone Preamplifier Type ZE 0300 for clip fastening near the wearer's ear. A Low level detector is incorporated to inhibit measurements below 89 dB(A) as required by OSHA. A Peak Excess Detector which is set to trigger on sound levels in excess of 140 dB(A) warns of exposure to noise of dangerously high level. It responds to noise peaks as short as 100 μs and when activated flags a "P" beside the percentage noise dose indicated on the display.

The 4431 has a separate "Cal." mode which boosts the count rate giving more than a 150 times faster indication to facilitate quick accurate calibration with for example, the Sound Level Calibrator Type 4230. The "Cal." mode can also be used for accelerated or "short term" measurements over measurement periods substantially less than 8 hours. This is useful in extrapolating the allowable exposure time of personnel in fixed locations enabling work rotas to be planned. Using conversion tables supplied, the actual L_{eq} according to OSHA (q = 5) criterion may be derived from the displayed % noise dose for measurement periods of 5 min., 15 min., 1 h., 2 hrs., 4 hrs. and 8 hrs.

Power to the 4431 can be supplied by a single 9 V transistor radio battery, however, with a 9 V Alkaline Battery QB 0016 supplied, the overall life is approximately 60 hours for 8 hours of continuous use per day.