ACOUSTICS for CHOIR and ORCHESTRA

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ACOUSTICS FOR CHOIR AND ORCHESTRA

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PREFACE

"Acoustics for Choir and Orchestra", the tenth yearly Music Acoustics seminar, organized by the Music Acoustics Committee of the Royal Academy of Music, was held on January 28th, 1985, at the Royal Institute of Technology in Stockholm. The seminar was addressed to interested conductors, musicians and teachers, and aimed to present both classical and recent findings in the acoustics of rooms and performing ensembles. The papers given at the seminar are presented in this book, which is similar to earlier books in this series in that the authors were asked to make their presentations accessible to interested musicians with a modest scientific background.

In his opening speech, Eric Ericson, professor of Choral Conducting, entertainingly related several of the acoustic puzzlements that he has encountered in his work with choirs, and expressed his excitement and satisfaction that research in the acoustics of choirs and orchestral stages is being undertaken. This, together with the high attendance of qualified listeners at the seminar, was of course greatly encouraging to all the authors.

In "From Echo to Reverberation", Sven Lindblad, professor of Building Acoustics, gives an introductory overview of room acoustics, presenting the concepts and terminology of the subject while pointing out topics of current research interest. The paper "Acoustics of Choir Singing", by Johan Sundberg and myself, reviews previous research and then goes on to give examples of choir-related acoustics. Anders Christian Gade tackles the important problem of relating subjective impressions to objective measurements, and in his paper suggests some perceptually relevant measures of concert platform acoustics. Anders Askenfelt has studied the acoustic effects of using a riser upon which to play the double bass. The final paper is a joint presentation of two investigations which both sought to answer the question "Do Musicians of the Symphony Orchestra Become Deaf?"

At the end of the choir acoustics session, a panel consisting of Messrs. Ericson, Anders Colldén (teacher of conducting at the State Academy), Sundberg and Ternström answered questions from the audience and pondered matters of potential interest for future research.

KTH, November 1985
Sten Ternström, editor

FROM ECHO TO REVERBERATION

Professor Sven Lindblad

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Introduction

In any room, the sound emanating from a source reaches the listener as a combination of direct sound and myriads of indirect reflections, or "echoes". Our sense of hearing somehow resolves this combination into separate impressions of the source and of the room, and this is done so elegantly that we normally are unaware of the many echoes.

When we try to describe the acoustics of a room in technical terms, however, the great number of variables involved becomes a problem. Much of the research in room acoustics is concerned with finding simple but perceptually relevant measures of room characteristics. One such measure is the reverberation time (RT), as suggested by W.C. Sabine around 1900. By modern standards, of course, RT gives a very simplified picture, even if several measurements are made (e.g. one in each octave band).

Wavelengths and reflector dimensions

To some extent, sound waves can be likened to rays of light which have a distinct direction and are reflected in a specular manner (as in a mirror). This is the approach of ray acoustics, in which the concepts of optics are directly applicable. Whereas light waves are microscopically small, however, sound waves have much the same dimensions as the things around us. Sound and light therefore act rather differently when they strike objects or surfaces that are irregular or undulated.

An analogy is often made with waves on water. A pole standing in the water will reflect small waves, e.g. from a bird, while the swell of the sea remains largely unaffected.

For a tone with the pitch of C_4 (262 Hz, nearest to the piano's key hole), the wavelength (wl) of the fundamental is about 1.3 m. With every octave step upwards, the wl is halved, and with every octave down, the wl is doubled. A C_2 (bass tone), for example, has a wl of about 5 m. This means that reflectors used around a stage area must be quite large if they are to reflect most of the musical sound spectrum. It also gives us a way of achieving frequency dependent reflection. The reflected sound will usually be boosted in the treble as compared to the bass. We believe that relatively small reflectors could sometimes be sufficient for directing certain high frequency sounds of importance to ensemble play. It could be worthwhile to study the effect of e.g. a beam or a proscenium arch reflecting back to instrumentalists or a singer. Depending of the dimensions ("height") of the beam, we get a tunable correlation of the reflected, or, for a small beam, spread, (back scattered) sound, as shown in Fig. 1.

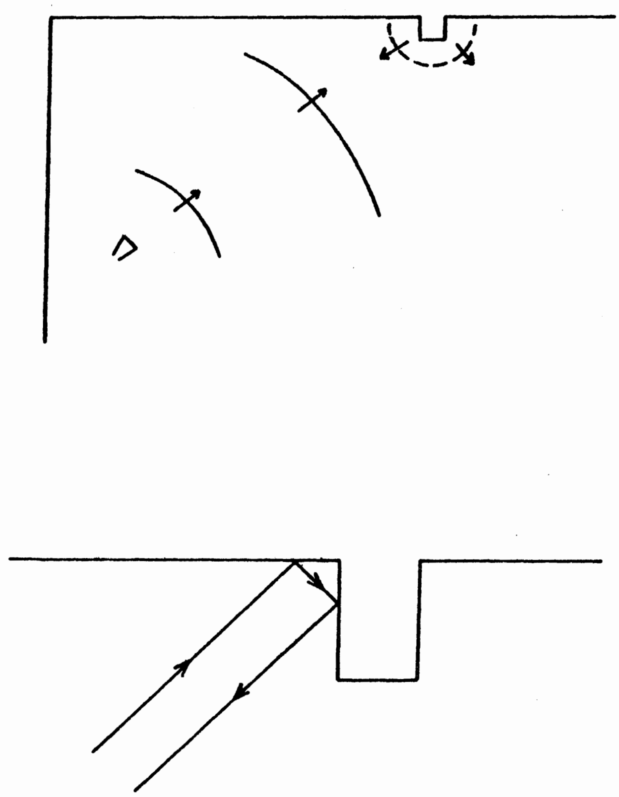

Figure 1: Large wl and/or small beam leads to weak reflection, spreading cylindrically with the beam as center, so called scattering (upper). Small wl and/or large beam leads to specular corner reflection (lower).

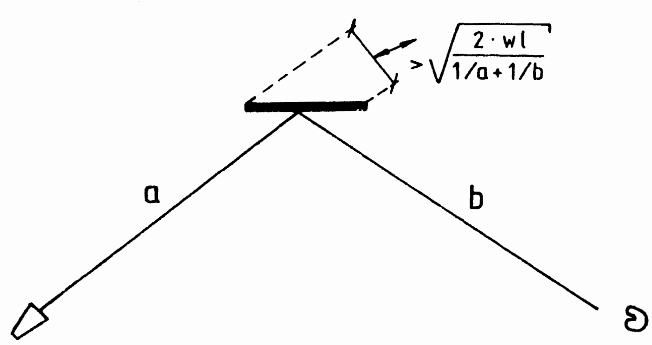

Figure 2: For full specular reflection the projected dimensions in the "a" or "b" direction must be at least as large as in the formula.

The Fresnel zone and "sound sausages"

According to one of the "V.I.P.'s of acoustics", professor Lothar Cremer, a reflector should not be smaller than one Fresnel zone, if full specular reflection of distant sources to distant listeners is to be achieved (Fig. 2).

In everyday words, the sound is more like a sausage than a ray. The thickness at a place with distance a from the source and b from the listener is just the value given by the root in the above figure. This means that in the midpoint a , this gives the the thickness

$$\sqrt{wl \cdot a}$$

For a = 10 m and wl = 0.9 m, this gives

$$\sqrt{10 \cdot 0.9} = 3 \text{ m}$$

So the thickness of the sound sausage is 3 m and the projected size of a reflector in the midpoint ought to be at least 3 m. If placed close to the source or the listener, it could be a little smaller, but not much. 5 + 15 m instead of 10 + 10 m gives 2.6 m (Fig. 3).

One question that is often brought up concerns the significance for the listener of the high note (high frequency) reflection from small irregularities on walls and ceiling and from small objects. Some acousticians, e.g. Dr. Ted Schultz, believe that these are very important. They are often very difficult to detect by instrumentation and also difficult to simulate in physical or computer models.

Weight per unit area of reflectors

The weight of sound reflectors is often of interest, especially when dealing with auxiliary reflectors that are to be moved around. What weight per m^2 is needed to get adequate reflections? Is a plastic cloth enough or is a board needed? Table 1 gives some examples of decibels lost for different values of weight per unit of area.

Weight kg/m^2	Lost dB at C_3 (c:a 125 Hz)
0.5	6
1	2
2	0.5

Table 1.

More generally, the number of decibels lost is determined by the formula

$$\text{Lost dB} = 10 \log (1 + (130/D)^2)$$

where

$$D = \text{frequency (Hz)} \times \text{weight } (kg/m^2) .$$

The "lost" part is usually transmitted through the light-weight reflector. The formula gives a limiting case of 3 dB for D=130 . This means that for good reflection at a given frequency we need a weight in kg/m^2 that is greater than 130 divided by the frequency in Hz.

Sound absorption at screens

When using screens around groups of performers, the reflected sound can sometimes be too strong. Some sound absorption is then in order; normally, glass fibre board is used.

When sound waves penetrate into fibrous material, the viscosity of the air acts as an acoustic resistance. Friction then transforms the sound into heat (only a few milliwatts per m^2, even for strong sounds). The absorption at lower frequencies (like C_3) depends strongly

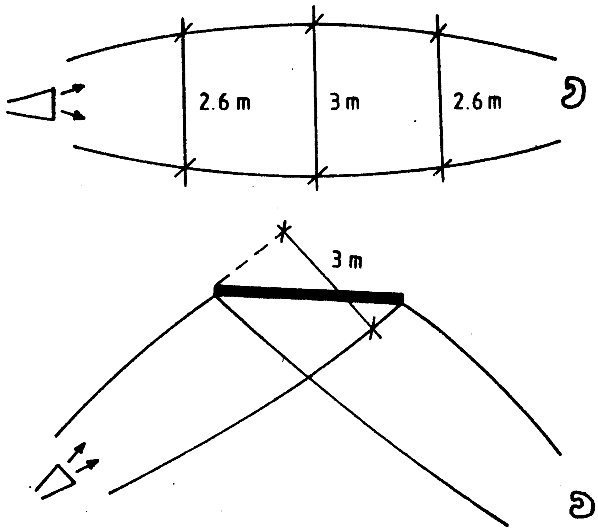

Figure 3: A direct and a reflected sound sausage with minimum reflector size.

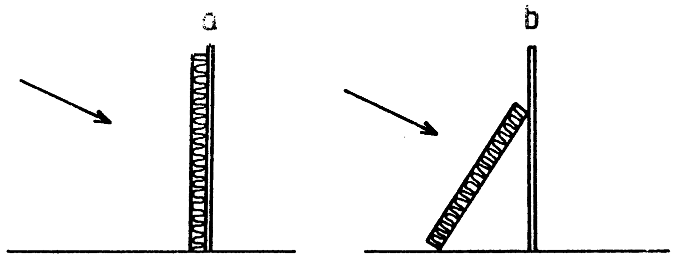

Figure 4: The oblique arrangement in "b" gives more air between the front of the absorbent and the hard screen than in "a" (a screen is always impervious). This leads to better low frequency absorption.

on the distance from surface to hard background. Absorption at a screen could therefore be arranged as shown in Figure 4.

New measures and measuring methods

The electrical signal (alternating voltage) from a microphone is proportional to the sound pressure at the microphone membrane. By connecting the microphone to an oscilloscope, we can study the time history of e.g. an impulsive sound from a small gun, a spark or a loudspeaker. Some oscilloscopes also have "memory", so that the time history is "frozen" on the screen. When using loudspeakers, the impulse energy is often made more concentrated by filtering to one octave band at a time. Filtering can also always be performed between microphone and oscilloscope. A pulse which has been filtered e.g. to the C_5 octave band (around 500 Hz) sounds like a "ping". It is a pulse with a tonal quality to it, with a pitch corresponding to the frequency in the middle of the octave band used.

A time record with direct sound plus reflection is sketched in Figure 5. Clearly, there is too much information in a picture like the one above. It is also difficult to see details in the weak signal at the end of the time record. The signal can be processed, however, so as to extract more interesting information. First, we can turn the negative part positive, by shifting sign in the signal system.

The usual way of doing this is to square the signal. This gives only positive values as

$1^2=1$, $2^2=4$ but also $(-1)^2=1$ and $(-2)^2=4$.

Then, this positive squared signal can be smoothed. This means that the fastest undulations are averaged over a very short time period. The resulting signal is then transformed to sound pressure level in dB. This corresponds better to our perception of sound levels, and lets us deal with both strong and weak parts of the time history, within the same representation (Fig. 6). When the level is expressed in dB, the downward slope tends to be piece-wise linear. From this slope, we can obtain the reverberation time (RT). In the sixties, however, Manfred Schroeder showed that it was even better to sum or integrate the squared sound pressure over time. This must be done backwards (from the right). What happens is demonstrated by some simple square pulses (Fig. 7). We see that the full line represents the integral from the right. The integral "eats" the pulses and rises continuously on its way leftwards. The integral curve is transformed to a level curve in dB, and gives a smooth graph that still shows alterations in slope during the reverberation.

The Schroeder reverberation curve in Figure 8 shows a typical situation, with steeper slope in the start. Schroeder and others have also shown that with this smooth curve available, only the first 10 dB fall ought to be used instead of the usual 30 dB. This early decay time, EDT, correlates better to subjective ratings than the old 30 dB RT. The reason for using so many dB of slope earlier was that the reverberation curve obtained with direct registration of impulse sounds or interrupted noise was too ragged over short intervals.

Schroeder has shown that his method gives the (ensemble) average curve to all interrupted noise curves with fixed positions. So the method can be seen as an improvement of the previously best method, which uses interrupted noise. The Schroeder curve gives more information than EDT and RT but not too much,

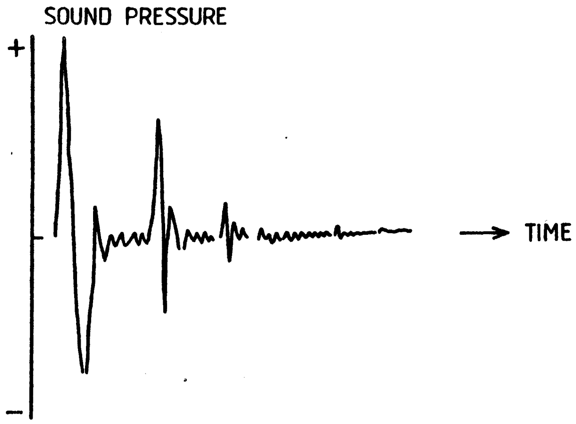

Figure 5: Sound pressure time history of an impulse with reflections.

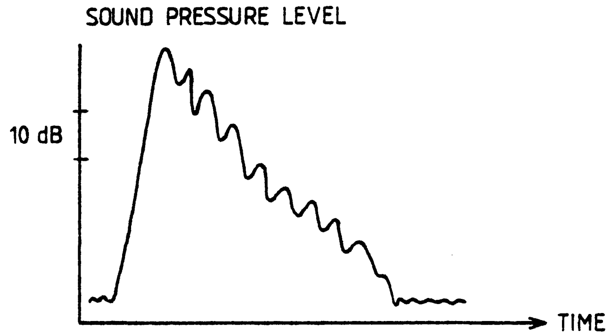

Figure 6: Level time history. Level produced by squaring the sound pressure signal, smoothing and taking 10 times the logarithm.

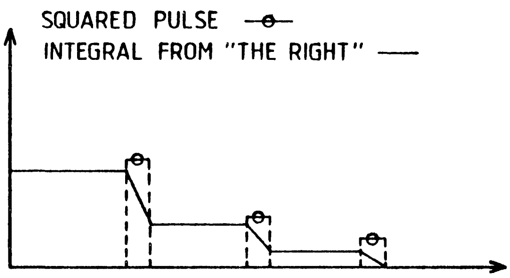

Figure 7: Principle of summation or integration from the end to early time, "from right to left."

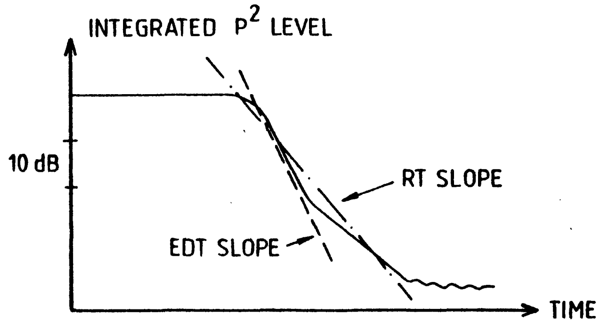

Figure 8: Integrated squared impulse response level, "Schroeder curve".

and it is a good tool for work with halls and especially stage acoustics.

From the squared sound pressure for an impulsive sound, many specific measures can be calculated, for example the definition, which is defined as the integral (the area under the squared curve) from 0 to 1/20 s, compared to the total integral. Several other measures have also been suggested and used to a smaller or greater extent.

ACOUSTICS OF CHOIR SINGING

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Introduction

Even though the choir is an important musical instrument, and although choir singing probably is the musical activity which musters the largest number of practitioners, research in the acoustics of choir singing has hitherto been scarce. This paper first reviews other choir acoustics research known to us, and then gives an account of our own experiments, which to date mostly have been concerned with intonation in choirs. Three sets of experiments are described: the first, on the effect of differences in loudness between your own voice and your companion singers' voices; the second, on the influence of various spectral properties of the sound that each singer hears; and the third, on pitch errors induced by changes in vowel articulation. The reader who is interested in learning more about the basic concepts, such as spectra and formants, is referred to e.g. Sundberg (1980 and forthcoming).

Review of work in choir acoustics

Lottermoser and Meyer (1960) studied the intonation of simultaneous intervals and the dispersion in fundamental frequency (defined in a special way by the authors), using commercial recordings of three unnamed but reputable choirs. They found that the choirs tended to make major thirds rather wide (average 416 cent) and minor thirds quite narrow (average 276 cent). The authors suggested that this serves to increase the contrast between the major and minor tonality of chords. Octaves and especially fifths were sung very close to pure intonation. The average of their dispersion measure varied greatly (2-60 cent), but was typically in the range 20-30 cent.

Hagerman and Sundberg (1980) studied the intonation of barbershop quartets. They found that the accuracy in fundamental frequency was very high, and practically independent of vowel. Intervals with many common partials were found to be easier to tune than those with few common partials. The exact size of most intervals deviated systematically from the values stipulated in both just and Pythagorean intonation. The deviations were found not to give rise to beats; the proposed explanation for this was the finite degree of periodicity of tones produced by the singers.

Meyer and Marshall (1985) had a quartet sing in a room with structural reflections from the floor only; the rest of the room acoustics was synthesized, systematically varied and played to the singers over loudspeakers. Especially the effect of the early reflections was studied, Marshall having found previously that these are paramount to instrumental ensemble playing (1978). The singers were asked to rate the difficulty of singing in the various reverberation fields. Rather than the early reflections, the loudness of the reverberation appeared to be most important to the choir singers. The reverberation time,

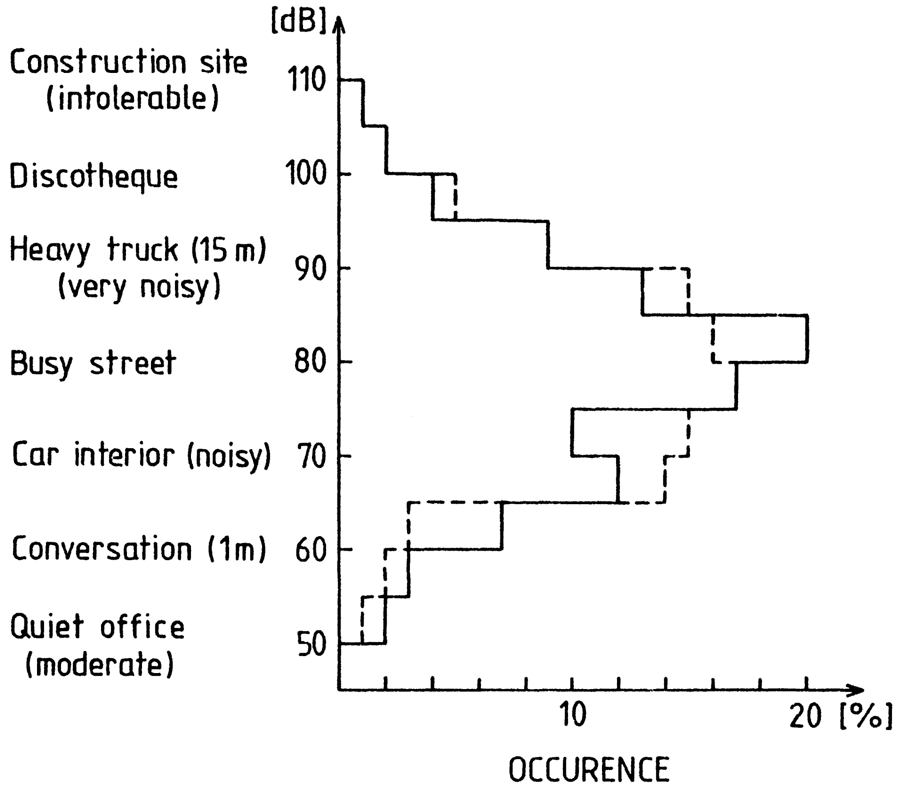

Figure 1. Histograms of the sound pressure level as measured during 30 minutes of rehearsal in two mixed choirs (dashed and solid lines). The length of the bars indicates the percentage of the total time spent at each sound pressure level. Some familiar sound levels are included for comparison.

however, was of little significance. Lateral early reflections were more appreciated than vertical ones, especially if the level of late reflections was high. Irrespective of the reverberation time, the singers preferred early reflections in the time range 15-35 ms (corresponding to reflecting walls at distances of 2.5 - 6 m). Early reflections arriving with 40 ms delay (6.5 m) were particularly disliked. The kind of music used was not described. Similar results were obtained with a larger ensemble.

Sound pressure levels and intonation

How loud are the sounds that the choir singers hear inside the choir? In acoustics, loudness is quantified using the "sound pressure level", or SPL, which is given in decibels (dB). This entity was measured by making calibrated recordings of the sound in the different sections of rehearsing choirs, and then making SPL histograms of these recordings. Fig. 1 shows the overall distributions for the two choirs measured. In both cases the most common SPL was 80 to 85 dB. The highest levels were observed on high-pitched, loud notes in the soprano sections, where the SPL occasionally would exceed 115 dB.

It is actually a noteworthy rule-of-thumb that the SPL of music sounds increase when the pitch increases. This has to do with the sensitivity of the ear and with acoustic laws of tone production and radiation, and it applies equally to singers and instruments. In fact, the variations in sound pressure level that are necessarily associated with large pitch changes are often larger than those made for expressive purposes As listeners, we tend to pay attention to the intended rather than the pitch-induced variations of SPL.

Hearing oneself

The sound of one's own voice, which will be referred to as the feedback in the following, reaches one's own ears in a rather different way from that of other people's voices (Lindqvist-Gauffin & Sundberg, 1974). Some of it is transmitted by the bones of the head and neck, and the rest is airborne. These two routes both tend to attenuate the high-frequency (treble) part of our own voice, resulting in a "fuller" sound. This timbral bias is strong enough to make one's own voice sound quite strange and "thin" when rendered the way everybody else hears it, in a recording. It also implies that a singer's own voice will selectively mask the low end of the spectrum for all other sounds, while he is singing.

Hearing the others

Each singer must of course be able to hear his feedback, but also the sound of the other singers' voices, which we will call the reference. Interestingly, the difference in loudness between the feedback and the reference depends on two room acoustic factors, which to some degree can be controlled. One is the spacing between singers: if the singers are placed further apart, they will hear less of the reference, thus favouring the feedback. Another factor is the reverberation of the room: since reverberation contributes to the total energy of the sound, a choir singing with a given effort will be louder in a reverberant room than in a damped room. Hence each singer will find it harder to hear his feedback in a reverberant room.

These effects can be computed. Fig. 2 shows an estimate of the difference in decibels between the feedback and the reference, under certain simplifying assumptions. The curves reflect how this difference would vary with the reverberation time of the room and with the average distance between singers.

Feedback/Reference experiment

Do singers perform well even when there is a large difference in SPL between the feedback and reference, or is it important that an exact balance of loudness be maintained?

Experienced choir singers were asked to sing in unison with synthetic vowel tones played to them over headphones. The loudness of these reference tones was varied considerably (from 60 to 100 dB), but the subjects had to sing at 90 dB every time, as indicated by a level meter in front of them. In this way, feedback-to-reference SPL differences of 30 to -10 dB were obtained. The reference tones had vibrato, to lessen the possibility of using beats to adjust the intonation. Each tone lasted for

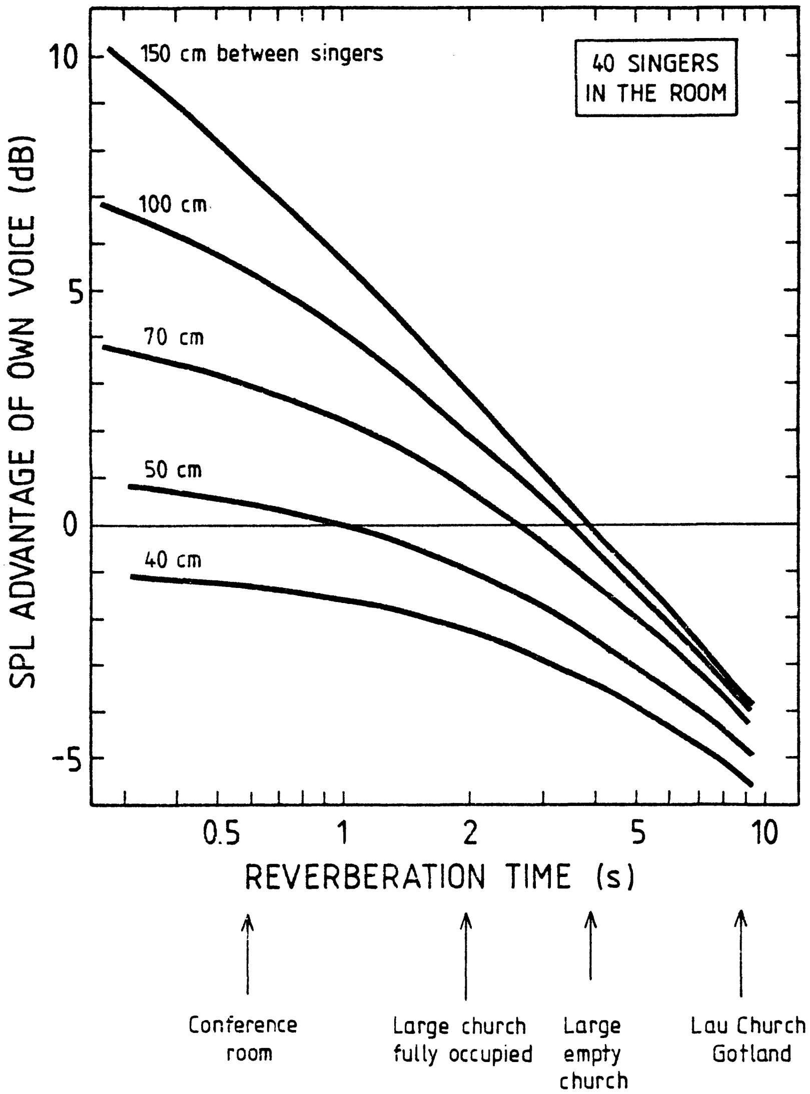

Figure 2. The estimated difference in sound pressure level between one's own voice and the rest of the choir, as a function of the reverberation time of the room. The different curves correspond to different distances between adjacent singers. The estimate is based on 40 singers standing in a cluster and all singing equally loud; how loud does not matter. Note that a difference of zero dB does not necessarily mean that the two sounds are perceived as being equally loud.

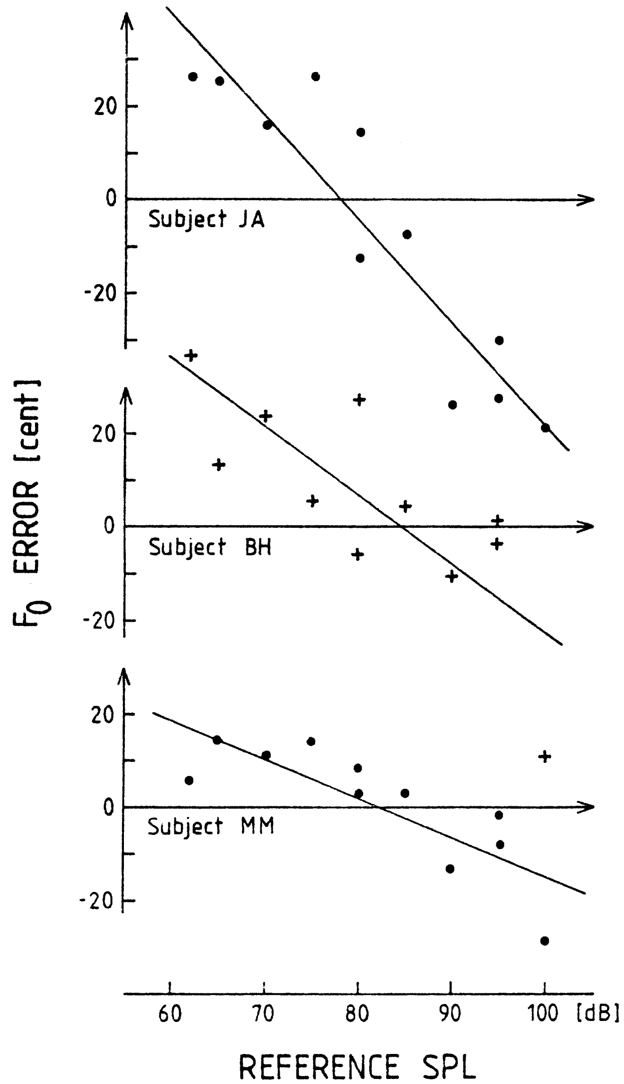

Figure 3. Some examples of pitch effects in the subjects' responses for the vowel /u/. The trend is that a loud reference tone would elicit too low a response, and vice versa. This did not occur for the vowel /a:/.

nine seconds, and the subjects were to start singing in unison with the reference tone as soon as possible. Two different vowels were used: /u:/, which is poor in high partials but rich in bone conduction, and /ɑ:/, which produces more radiated energy at higher frequencies of the spectrum, but less feedback via the body than /u:/.

The average fundamental frequency was measured for each sung tone, and compared to that of the reference vowel. Figure 3 shows one aspect of the results. For the vowel /u:/, eight of the nine subjects tended to respond with too high a pitch when the reference was soft, and too low a pitch when the reference was very loud. For the vowel /ɑ:/, only one

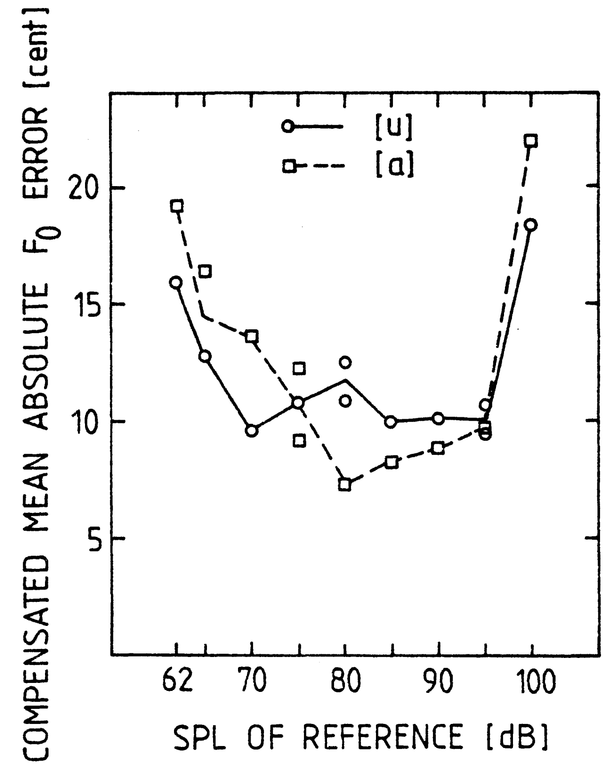

Figure 4. The absolute value of the Fo error (cent) averaged over nine subjects. Individual pitch perception effects (Fig. 4) have been compensated for.

subject behaved in this way. This effect of SPL on pitch is in fact well known from psycho-acoustic studies. Our ear perceives small pitch changes if a tone with constant frequency is varied in amplitude (Terhardt, 1975). The effect varies from person to person and is most marked for tones with few but strong low partials, such as in the vowel /u:/. Note that in some cases the effect was as large as 50 cent, or half a semitone. It seems possible that choral intonation of loud or soft chords on closed vowels could be affected by this auditory phenomenon. The effect might be more relevant in a solo-accompaniment situation, where large SPL differences are more common (Rossing & al, 1985).

In order to evaluate the influence of the Feedback/Reference SPL difference, the individual pitch effect of the SPL first had to be compensated for. This was done for each subject that exhibited the effect by mathematically fitting a straight line through the data points and then subtracting the pitch error predicted by this line. Fig. 4 shows the results, with the magnitude of the remaining average error at each reference SPL expressed in cent. We see that when the reference SPL was close to the subject's SPL, or about 90 dB, the average errors were at their smallest, or 8-10 cent. When the reference SPL was more than 5 dB louder, the errors increased abruptly, indicating that the subjects

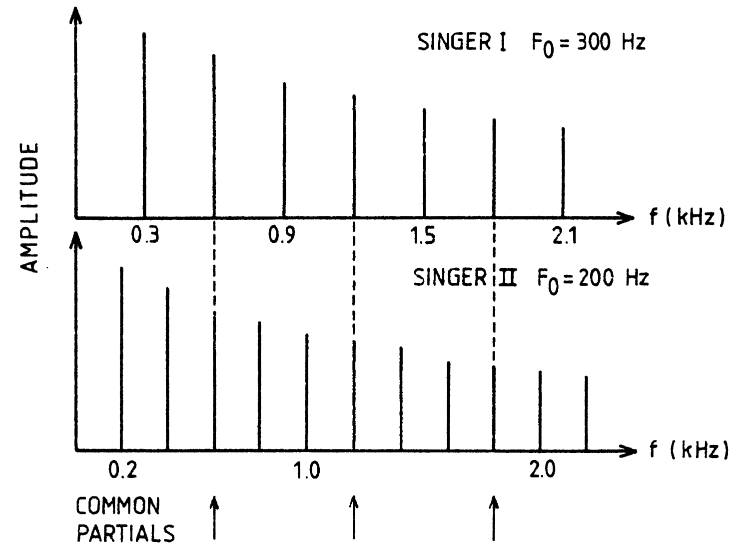

Figure 5. Illustration of common partials for the "fifth" interval.

had difficulty in hearing their own voice; often they would start to "hunt" for the correct pitch, thereby increasing the pitch errors. When, on the other hand, the reference SPL was made softer than the subject, the errors increased only gradually, and only beyond -15 dB. Here, the subjects could hear themselves perfectly well, and even though they might not hear the reference tone once they started to sing, they could still remember its pitch. This would account for the more gradual increase in errors for low reference tone SPL's. We see that the singers' precision in intonation did not deteriorate for SPL differences of -15 to +5 dB. Hence the SPL balance was not very critical, at least not as long as the feedback was unimpaired.

The vowel spectrum and intonation

Apart from sound levels in the choir, there are many questions regarding the spectral aspects of the choral sound, that is, those properties of the sound that relate to its timbre. Clearly, the reference sound (from the rest of the choir) contains information on which each singer must depend for his own performance. But which aspects of the reference sound are relevant to intonation? Three candidates for factors that might affect intonation are discussed in the following: "common" partials, high frequency partials, and vibrato.

The spectrum of the singing voice is harmonic, meaning that the frequencies of the partials are equidistant on a linear frequency scale. For example, if the fundamental, or first partial, is at 100 Hz, then the second partial is at 200 Hz, the third is at 300 Hz, and so on. But what kind of spectrum do we get when several voices are singing simultaneously at harmonic intervals from each other?

In so-called "just" intonation, the frequency ratio of the two tones in a consonant dyad can be expressed using small integers. Thus, the octave has a frequency ratio of 2:1, the fifth has 3:2, the fourth has 4:3, the major third has 5:4, and so on. Two singers singing a pure fifth would therefore produce the spectra depicted in Fig. 5. They are shown separately for clarity, but in reality they would be superimposed as a single spectrum. Notice that every third partial of the lower tone coincides with every other partial of the higher tone. These coinciding partials appear in all consonant dyads, and are called common partials. If the interval is not exactly tuned, the common partials will give rise to beats or "roughness". It is possible that such beats or roughness could serve as intonation clues. By experimenting with artificial vowel tones from which the common partials have been removed, their value to intonation could be tested.

Theoretically, all partials but the lowest, or fundamental, are redundant when determining the frequency of a complex, harmonic tone, because they are all at integer multiples of the fundamental frequency. However, low partials in the reference sound are more likely to be masked by the singer's feedback, as explained earlier. Also, in the loudness experiment, we saw that the perceived pitch follows fundamental frequency more closely when there are strong high partials in the sound, especially at extremes of loudness. This indicates that the presence of high partials could be of importance to intonation.

It has been shown that for a single complex tone, a sinusoidal vibrato does not affect the accuracy of the perceived pitch (Sundberg, 1978). If beats are used as an intonation clue, however, vibrato might still be undesirable, since regular beats arise only from the interference between straight tones; if the tones have vibrato, which involves a periodic variation of fundamental frequency, then the possibilty of beats will be eliminated.

Spectral factors experiment

Amateur male choir singers were asked to sing fifths and major thirds with artificial vowel tones presented over a loudspeaker in an anechoic room. The vowels had the characteristics of a single bass voice, which had been synthesized with all possible combinations of the properties discussed earlier:

- with or without vibrato
- with or without high partials
- with or without common partials

The experiment was run with one subject at a time. The disagreement in fundamental frequency between the subjects (expressed as the standard deviation over subjects) was taken as a measure of the difficulty of intonation. The results showed that straight tone (lack of vibrato), presence of high partials, and presence of common partials all had the effect of reducing the pitch disagreement between subjects. The standard deviation was about 15 cent for the easiest stimuli, and about 40 cent for the most difficult ones. The largest errors made by any one subject were on the order of 100 cent.

For comparison, the standard deviations under normal rehearsal conditions were measured in a separate experiment. Simultaneous recordings were made of six subjects in the bass section of an amateur choir. Here, the scatter ranged from 10 to 15 cent with a mean of 13 cent.

It is possible to selectively reinforce a given partial in a sung tone, by choosing fundamental frequency and vowel so that the partial coincides in frequency with a formant. An experiment was performed to find out whether such vowel-induced variations in the amplitude of the lowest common partial have any effect on the precision of intonation.

First, a male chorus of twenty-five singers was recorded singing sustained vowels in unison. The recording was made using binaural microphones, which gives a very realistic sound field when played back over headphones. The pitches and vowels were chosen beforehand so as to

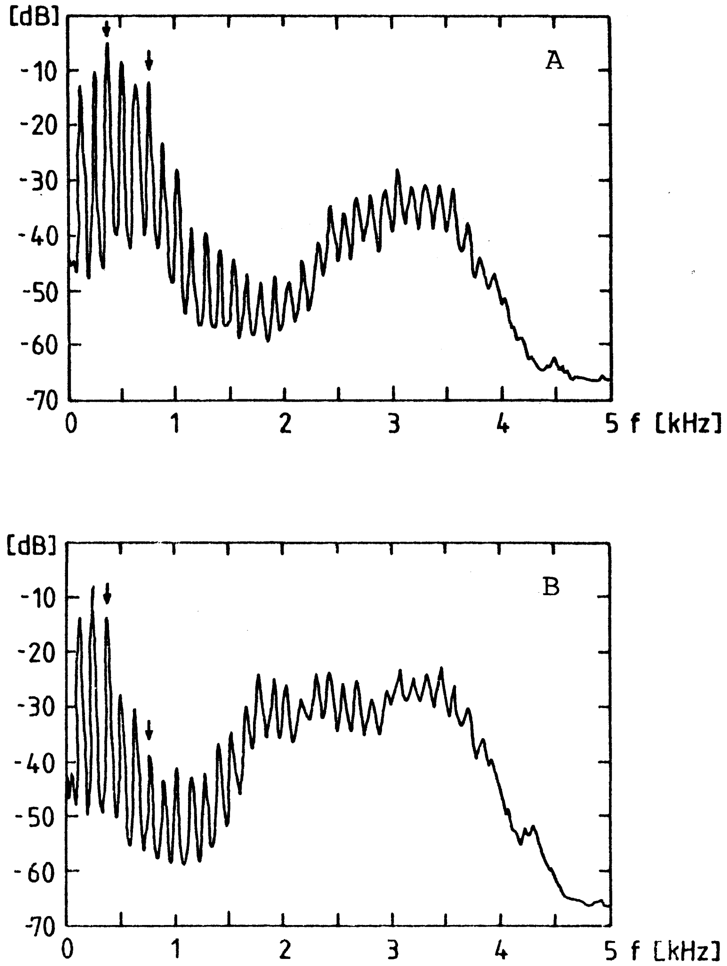

Figure 6. Spectra of two "fifths" stimulus tones with the lowest common partials reinforced (a) or suppressed (b) by the formant structure of the vowels. The tones are recordings of a male choir singing sustained tones in unison.

reinforce or attenuate the third or the fifth partial (Fig. 6). Then, individual subjects were asked to sing fifths and major thirds above the sound of the choir in the headphones. As before, the standard deviation in fundamental frequency over subjects was determined. It turned out that the subjects agreed more closely on the size of the intervals, when the lowest common partial was reinforced through choice of vowel, and more so, the more prominent this partial was. This suggests that choral intonation perhaps could be improved by rehearsing cadences specially written with a text that often would reinforce the common partials.

Vowel articulation and intonation

In speech research, several investigators have found that certain vowels

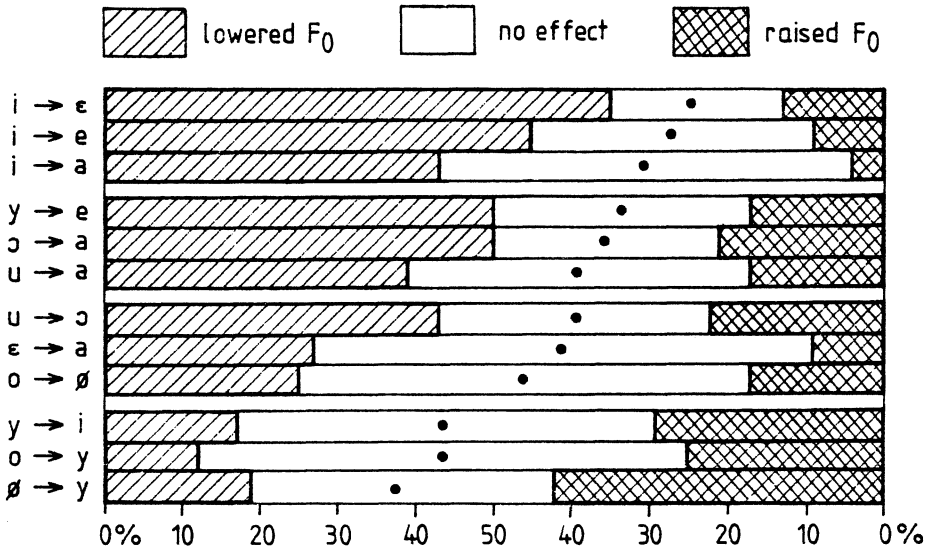

Figure 7. Ranking of vowel changes by their tendency to change the phonation frequency. The length of the bars denote the percentage of responses with a drop (left bars) or rise (right bars) in Fo, when changing from the first to the second vowel.

tend to be pronounced with a slightly higher or lower pitch than others; therefore, such vowels are said to possess an intrinsic pitch. A singer would presumably correct his pitch for such deviations, at least as long as he can hear his own voice. In a choir, however, feedback can at times be masked by the reference, so the intrinsic pitch of vowels might in fact affect intonation. This might explain the observation, frequently made by choir directors, that some vowels tend to be sung out uf tune. An experiment was performed to verify the existence of this effect.

Six experienced choir singers were asked to sustain various tones, and change from one vowel to another in the middle of the tone. They were to sing the vowel pair with noise (in headphones) completely masking their own feedback, and then immediately repeat the task with the masking noise removed. Twelve different vowel pairs were tested.

The fundamental frequency contours were plotted for all responses, and examined for frequency transitions that coincided with the change in vowel. As expected, it was found that the subjects made larger frequency shifts when they could not hear their own voice, but also that the proportion of responses with frequency shift remained the same, whether the masking noise was present or not. This indicates that the subjects did in fact try to compensate for the perturbed frequency, without entirely succeeding.

Changing to the vowels /i:/ and /y:/ from any of the other vowels tried was found to raise the fundamental frequency slightly, whereas changing to /ε:/ and /e:/ tended to lower it (fig. 7). These results agree with findings from speech research. The explanation generally proposed (Bush, 1981) is that the raised tongue positions of /i:/ and /y:/ involve an upward pull on the larynx which tends to affect the vocal folds so as to raise the phonation frequency. It might be mentioned that the vowel /ɑ:/, which was not included in the experiment, is known to have a marked lowering effect on pitch (Ewan, 1979).

Conclusion

The intonation accuracy of choir singers seems to depend on several properties of the sound that the singer hears: the loudness of the feedback in relation to the reference, the stability of the fundamental frequency of the reference tone, the prominence of the so-called common partials in conconant intervals, and the overall shape of the spectrum. All of these properties are affected both by the room acoustics and by the vocal behaviour of the choir singers. Relevant room acoustic factors include the reverberation time, the spectral bias of the reverberation, and the spacing between singers. Psycho-acoustic and physiological factors include auditory masking, dependence of pitch on sound pressure level, (lack of) perception of consonance from beats and/or roughness, and articulatory perturbation of phonation frequency.

It is our general impression that the effect of any one of these factors on intonation accuracy is rather small if taken in isolation. However, combinations of favourable or unfavourable circumstances could substantially affect a choir's performance, especially if the singers are amateurs.

Acknowledgments

The participation of the choir singers, particularly those of Teknologkören at KTH, is gratefully acknowledged. This research was supported by the Swedish Council for Research in the Humanities and Social Sciences.

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ACOUSTICS OF THE ORCHESTRA PLATFORM FROM THE MUSICIANS' POINT OF VIEW

A review of research on the room acoustic needs of musicians carried out at The Technical University of Denmark.

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What properties of a hall determine our judgement regarding its acoustics being good or bad? This question has occupied many researchers in the post war period, and today it can be answered quite satisfactorily. This is only true, however, if the person asking is a listener. Musicians' acoustic needs and perception of the sound of their own or their co-players' instruments have only been dealt with by very few acousticians. In 1979 this situation motivated work to be started on the subject at the Technical University of Denmark. In the following, our research methods and the main results obtained so far will be described. In order to aid the understanding of our approach, the basic concepts and goals for this work will be explained first.

Concepts and goals of subjective room acoustics

The concepts dealt with in this field of research are illustrated in Fig. 1. In the upper half of the figure are three boxes that represent phenomena in the real world. From left to right the boxes represent the physical auditoria, the acoustic properties of these spaces, and finally the subjective*, auditory impressions which acoustics impart to human beings.

The acoustic properties of a room for a particular set of sound source and receiver position are in principle fully described by the so-called impulse response recorded between these two points (see the previous paper by Sven Lindblad). However, the impulse response gives a picture with far too many details, in which it is difficult to distinguish between important and irrelevant properties. For instance, the impulse response reveals even slight changes in the delay, level and spectrum of each reflection; but the ear will only be able to detect more substantial changes in the picture. In the architectural and subjective domains, too, we find that the real world is too complex to be used directly for a relevant description of the acoustic phenomena. Details in the hall design do not always noticeably influence the acoustics, and in the subjective domain, people may express their judgements using different and rich vocabularies without necessarily having had different impressions.

In all aspects of the problem it is

*) In this context, "subjective" should not be understood as unreliable from a technical point of view, but rather as referring to judgement through human perception - as opposed to measurement by technical, so-called objective instruments.

therefore necessary to find and extract the important factors. These factors form the contents of the three lower boxes, which represent the abstract world of scientific description. The primary task of subjective room acoustics would be to fill these three lower boxes. This is done most logically as follows:

First we have to find out along which scales we are making judgements, i.e. we have to define a set of subjective (room acoustic) parameters, which cover the main elements of our room acoustic perception. For each subjective parameter we then want a corresponding objective (room acoustic) parameter, i.e. a formula which can evaluate the impulse response with respect to this particular aspect of the perceived acoustics. Finally, we wish to uncover what aspects of the hall design are responsible for changes in each objective parameter and its subjective counterpart. These design aspects could analogously be called the architectural (room acoustic) parameters.

After having established these three sets of parameters we will, in principle, have reached a full understanding of room acoustic phenomena. We will then be able to assist the architect in selecting optimal conditions for every use and taste and to check them with objective measures - even before the building is erected, if the impulse response can be simulated with sufficient accuracy by using computers or scale models.

Now, this ultimate goal may seem intimidating. The aim is not, however, to impose an acoustic dictatorship in hall design, but to allow the architect and his client to make qualified decisions based on acoustical as well as on financial, esthetical, and other considerations. In any case, the research has by no means reached such an omniscient state at present, in particular not with respect to the room acoustic conditions of musicians.

REVERBERANCE	'SOLOIST'-CONCERN
SUPPORT	
TIMBRE	
DYNAMICS	
HEARING EACH OTHER	'ENSEMBLE'-CONCERN
TIME DELAY	

Table 1: Subjective room acoustic parameters of relevance to musicians.

Subjective parameters covering musicians' room acoustic impression

Due to the scarcity of previous works dealing with the subject, it was necessary first to get an overview of musicians' room acoustic needs, i.e. to establish an a priori set of subjective parameters. This was done by carrying out an interview survey (Gade, 1981). 32 performers of classical music (conductors, pianists, singers and players of various orchestra instruments) were asked to describe the elements of their room acoustic concern and to rank the relative importance of these elements in different playing situations. From their answers, the set of subjective parameter candidates listed in Table 1 was formed. In addition to these parameters, which are further explained below, musicians will, of course, be sensitive to mere acoustical faults such as background noise and echoes.

REVERBERANCE is mainly perceived during breaks or shift of tone played, since it sustains the tones just played. It binds adjacent notes together, can blur details in the performance and may

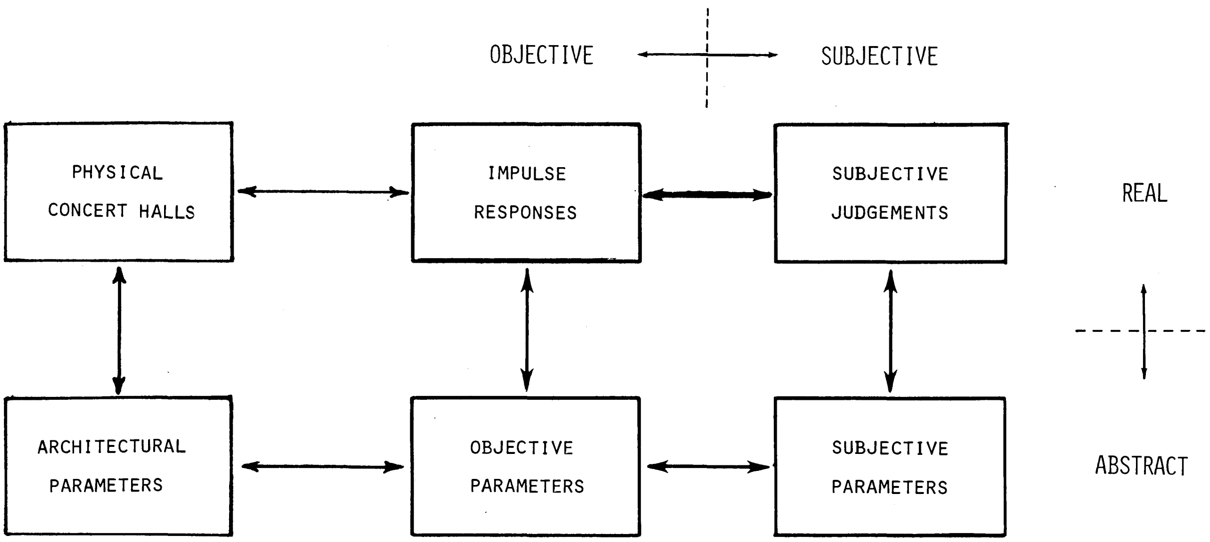


Figure 1: Illustration of the various aspects involved in subjective room acoustics.

give a sense of response from the hall.

SUPPORT is the property which makes the musician feel that he can hear himself and that it is not necessary to force the instrument. It is effective even while short tones are played and is therefore a property different from reverberance.

TIMBRE means the influence of the room on the tone colour of the instrument and on the egality in level in different registers. In ensembles, TIMBRE may also influence the musicians' impression of tonal balance between the instruments.

DYNAMICS describes the dynamic range obtainable in the room and the degree to which the room obeys the dynamic intention of the player.

HEARING EACH OTHER is the property required for a group of musicians to play in ensemble, i.e. with rhythmic precision, in tune and in balanced levels. In large ensembles it is important to have contact both among members within each group of instruments and among the different groups. The situation is satisfactory only when a delicate balance exists between hearing oneself and others. According to Jürgen Meyer (1978), if you hear your co-player(s) well, but not yourself, rhythmic precision is possible, but intonation suffers; whereas if one hears oneself well, but not the co-player(s), the intonation may be all right, but rhythmic precision will be hard to achieve.

TIME DELAY is a consequence of the speed of sound being limited; it can become quite disturbing when orchestra members have to sit far apart. It may cause rhythmic precision and tempo to deteriorate.

With regard to the relative importance of these parameters in different situations of playing, the first four may be called the "soloist concern" and the latter two the "ensemble concern".

Having defined this set of subjective parameters, the next step (according to the discussion of Fig. 1) is to look for

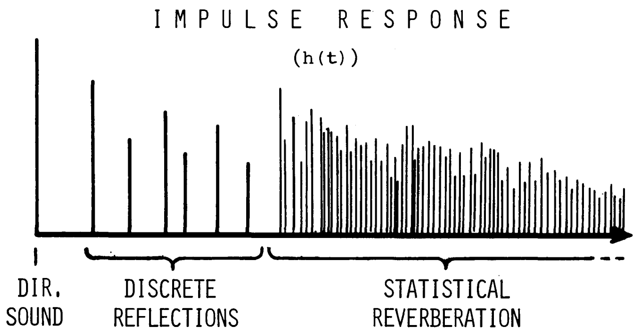


Figure 2: Impulse response in principle as regarded for electroacoustic simulations.

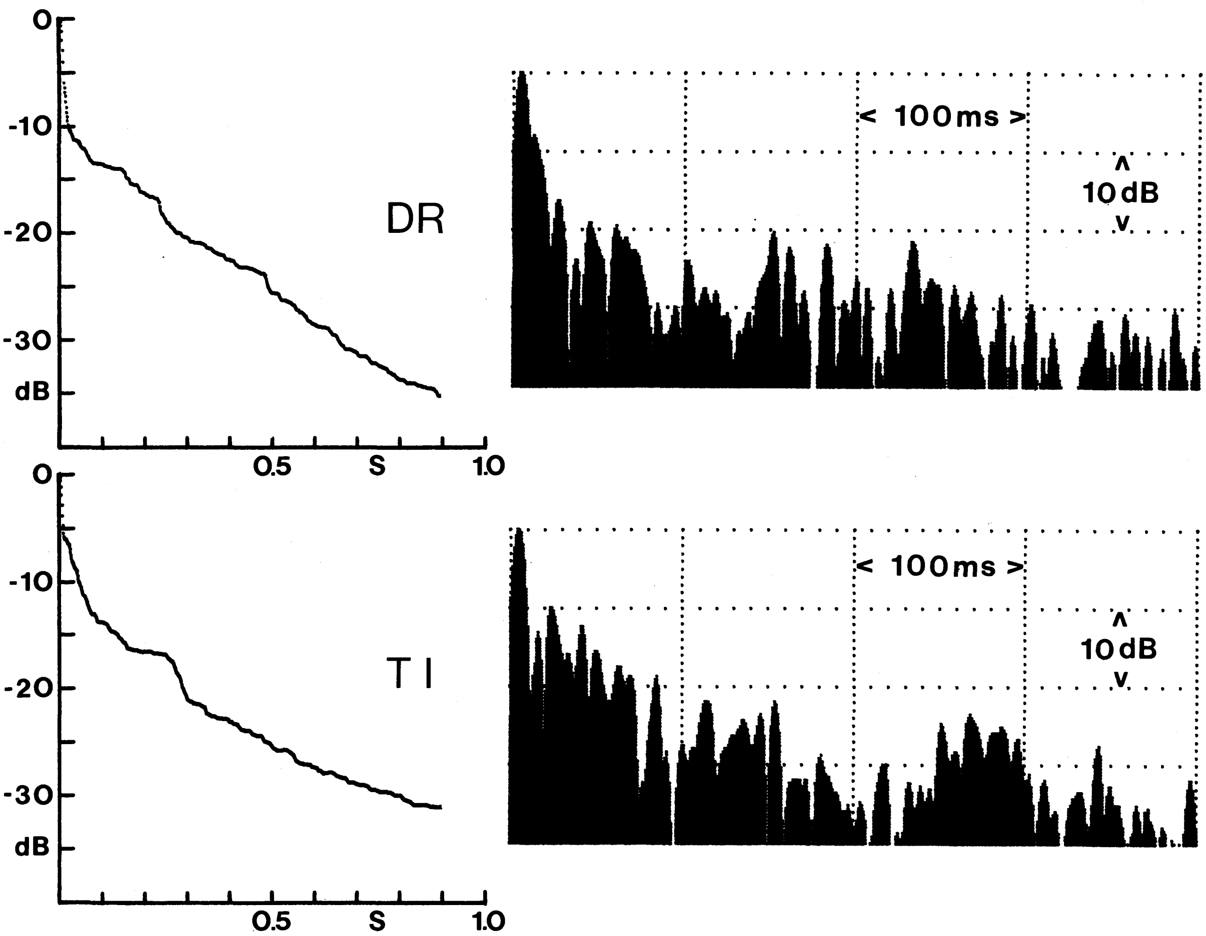


Figure 3: Reverberation decays and impulse responses recorded at the flautist leader's position in two different halls (source-microphone distance = 1 m).

those properties of the sound field that govern the changes in the subjective parameters. In our experimental work, we decided to focus on SUPPORT and the two ensemble parameters, which are aspects not already known from research on listener conditions.

Experimental method

In order to investigate the relationship between the subjective parameters and the objective sound field properties, the first requirement is to have variable sound fields available and to have these judged by performing musicians (Fig. 1). Fortunately, it is not necessary to move the subjects around to different concert halls, since the sound fields can be created in the laboratory by electroacoustic means. Use of artificial sound fields relies on the basic assumption that it is subjectively relevant and adequate to regard the impulse response as composed of three parts (Fig. 2):

1) the direct sound (if one or more musicians are placed in the same room, this component will propagate naturally and need not be considered in the simulation),

2) a number of discrete early reflections; each of these can vary in delay relative to the direct sound, in level, spectrum, and direction of incidence,

3) diffuse reverberation with delay, level, spectrum, and decay rate as variables.

An example of such a set-up is shown in Fig. 4. The musician is placed in an anechoic chamber where the sound from his instrument is picked up by a microphone. This signal is amplified, frequency-shaped and fed into electronic delay units (marked τ in the figure). From one of the delay outputs the signal is emitted into a reverberation room, where a number of microphones are placed. The diffuse nature of the reverberation is simulated by using several microphones and by emitting the sound through different loudspeakers in the anechoic chamber. The early reflections, which are taken directly from the delay outputs, can be given specific directions of incidence by being routed to selected loudspeakers via the mixer.

Of course, this technique has limitations as well as advantages. Regarding the drawbacks, it is clear that the simulation offers only a simplified description of real acoustic conditions (e.g. the number of early reflections will often be too small), the sound quality is limited, and the surroundings are unnatural. As for the advantages, however, they are very important in this scientific context: the sound field properties are fully controllable, they can be changed from one situation to the next very quickly (which makes subjective comparisons much easier and more reliable), and there are no visual indications to disturb the auditory judgement.

After the sound fields have been presented and judged, the next step is the analysis, consisting of:

- measurement of possible objective parameter candidates in the sound fields,

- extraction of the subjective parameters underlying the subjective judgements, and finally

- a correlation analysis to reveal if any of the objective parameters or variables show close relationship with the subjective parameters.

Here the second point may be a little surprising; but the fact is that the judgements need not be expressed directly in terms of the subjective parameters - the subject may even be unaware of their existence. This is possible when using modern so-called multidimensional scaling techniques, which reveal the underlying parameters or "dimensions" even from simple answers like "A was better than B" in a series of pair comparisons. The technique applied in this work is further described in Gade (1982), and more basic knowledge about these exciting statistical tools may be found in Schiffmann & al (1981).

Despite the attempts to make quick sound field alterations and the judgement procedure simple, the task of the subjects was, by no means, a simple one. They had to be able to play well while concentrating on listening; and it can be difficult to hear minor room acoustic details when the strongest sound impression is always the direct sound from one's own instrument. Therefore, it was found necessary to use experienced musicians as subjects (more than half of them were members of professional orchestras in Copenhagen) and to select fairly easy pieces to be played.

Experiments on soloist aspects

As mentioned in section 2, one of the topics selected for the experimental work was the "soloist" parameter: SUPPORT. During the interviews some of the musicians had mentioned two particular halls, the Tivoli Concert Hall (TI) and the Concert Hall, Studio 1 at the Danish Broadcasting House (DR), as being markedly different in this respect. Impulse response measurements were made on the platforms of both halls, revealing a big difference especially in the amount of early reflection energy (Fig. 3). In both halls the impulse responses shown were recorded at the flautist leader's position with a microphone placed one meter from the source (i.e. at a distance comparable to the instrument-to-ear distance for a musician). The interviewees stated that in TI it is easy to hear oneself whereas in DR it is difficult, especially for string players, who feel tempted to force the instrument, resulting in deterioration of the tone quality. Comparing these statements with the impulse responses in Fig. 3, it is likely that the feeling of support in TI is related to the high level of early reflection energy being sent back to the musician himself in this hall.

It is questionable, however, whether these early reflections are audible at all, in view of the dominating level of the direct sound (the leftmost peak in each of the impulse responses) at such a short distance from the sound source. Consequently, the aim of the first experiment was to determine the threshold of audibility for a single early reflection of the sound from the musician's own instrument (i.e. the level at which the reflection is just audible).

This threshold can be expected to depend on many factors such as delay, spectrum and direction of incidence of the reflection, the instrument, the motif played and the presence of other sounds. However, for practical reasons, this experiment was restricted to deal with the threshold of perception for a reflection from above, for six different delays and for three different instruments: flute, violin and cello. The subjects were placed in the set-up shown in Fig. 4, in which the ceiling reflection was emitted through a loudspeaker 3 meters above the subject's head, while a constant, diffuse reverberation was emitted through all five loudspeakers.

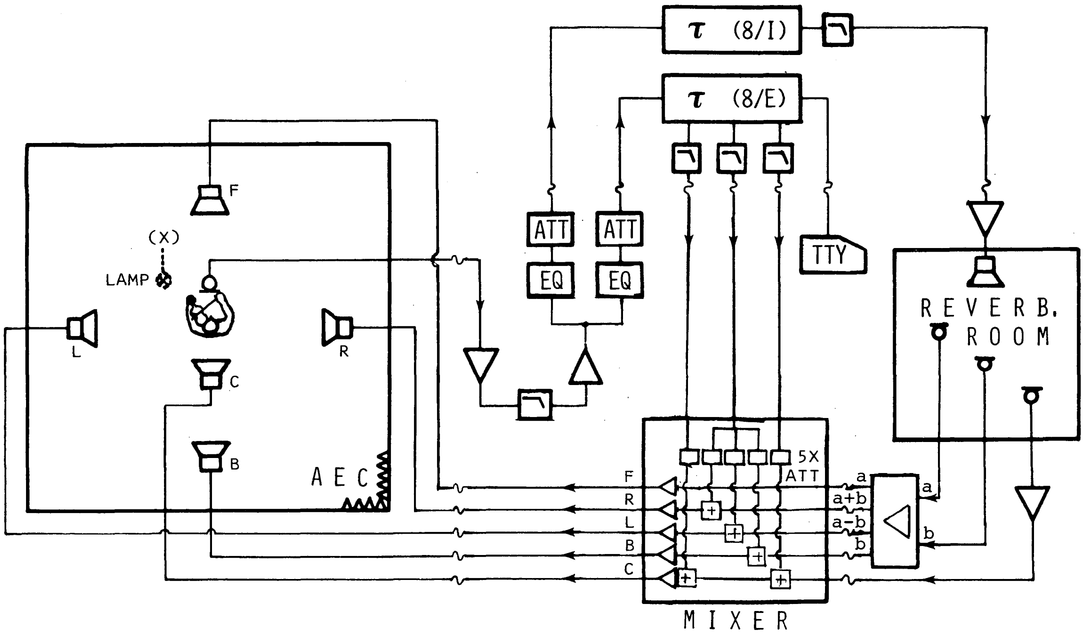


Figure 4: Diagram of simulation set-up used for experiments with soloists.

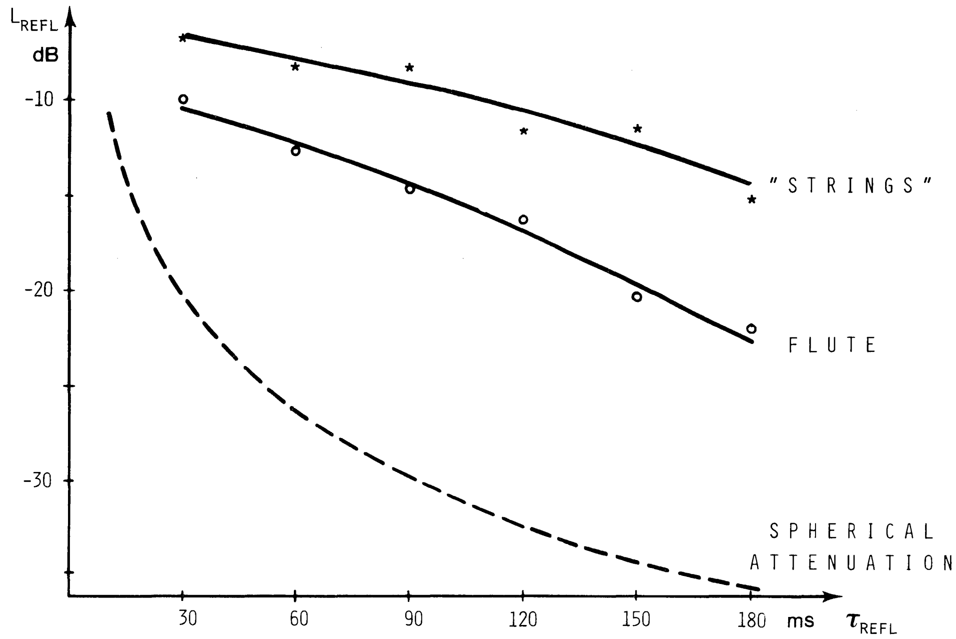


Figure 5: Threshold of perception of a single reflection for soloists. Mean values for strings (violins and celloes) and flutes. The dashed curve represents the level of a single spherically attenuated reflection from a plain, hard surface. The reflection levels are plotted in dB's relative to the level of the direct sound 1 m from the source.

At each delay the threshold of audibility was determined using the so-called Two Alternatives Forced Choice method. According to this method, a number of reflection levels around the supposed threshold are chosen. At each level a brief motif is played twice, with the reflection being added randomly the first or the second time. After playing this pair, the musician has to answer in which of the two presentations the reflection was present. After repeating this process a number of times (for each reflection delay and reflection level), a probability of detection can be calculated from the number of right and wrong answers. The threshold may then be defined as the level corresponding to 50% probability of detection.

The first Sound Example illustrates the situation: first we hear a flautist playing two pairs of the brief motif (one deep long note followed by two short higher ones) and afterwards two pairs played and judged by a cello player. On this and the following recordings it is easier to hear the "acoustics" when the subjects are talking; however, in the case of the cellist, the reflection is clearly audible in the last presentation.

In Fig. 5 the averaged results obtained for three violin players and three cello players (combined in one curve named "strings") and for three flute players are shown. At levels above a curve the ceiling reflection is audible to the player in question; at levels below, it is not. Thus, it appears that it is easier for the flute players than for the string players to perceive the influence of the reflection, and for all players it becomes easier as the reflection is further delayed relative to the strong direct sound.

The dotted curve in Fig. 5 represents the relationship between level and delay for a single reflection from a plain, hard surface. Consequently, such a reflection alone will not be audible, and the question now arises, whether enough early reflections are present in real halls for the energy in this part of the impulse response to have any influence at all. This point can be illuminated by Fig. 6, in which the total level of early reflection energy within the interval 20 to 100 ms for two positions on three platform has been compared to the lowest value of the threshold within the same time interval from Fig. 5. In cases where the total early reflection energy does not reach the threshold, there is no reason to believe in this energy having any audible effect. Thus, from this figure, the previously mentioned statement of especially string players lacking support in the DR hall is understandable.

Another experiment showed that early reflection energy of sound from the musician's instrument can be audible to the musician himself. Instead of the individual musicians, flute-violin-cello trios were placed in the set-up of Fig 4. In pair comparison tests, the subjects judged four sound fields with respect to preference regarding the sound of their own instrument. (Ease of ensemble judgements were also asked for; but the results were very vague). The four sound fields represented variations with respect to level of reverberation, as well as realistic variations of early reflection energy (see Fig. 7). The motif played was Joh. Seb. Bach, Trio Sonata No. 2, second movement, bar 1-33, which can be heard twice in the second sound example. The first version is played in sound field No. 1 (max. level of both early reflections and reverberation), whereas the latter is played with the whole set-up turned off, i.e. under anechoic conditions. I hope that the difference is apparent; if not, it can be judged from the subjects' reactions

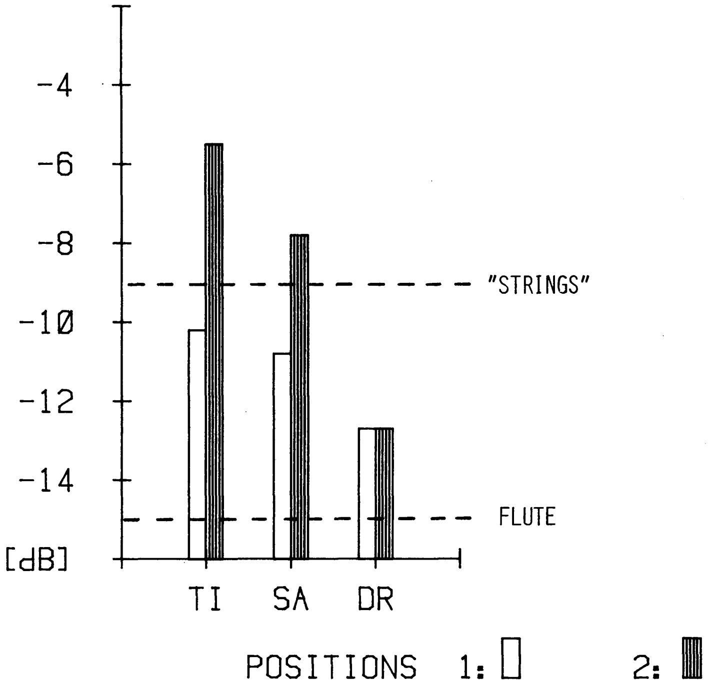


Figure 6: Comparison of threshold values for perception of single reflections (dashed line) and levels of early reflection energy (vertical bars) between 20 and 100 ms on three orchestra platforms.

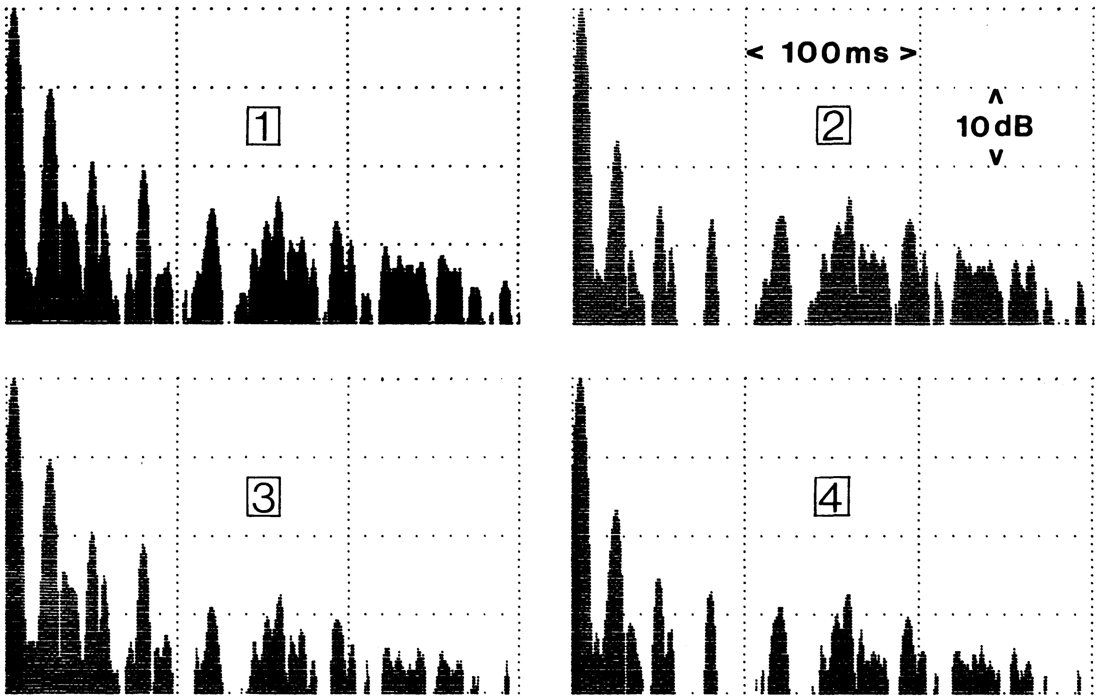


Figure 7: Impulse responses of four sound fields presented in experiment with trios. Variation from left to right: reflection level reduced by 6 dB. Variation from top to bottom: reverberation level reduced by 4 dB.

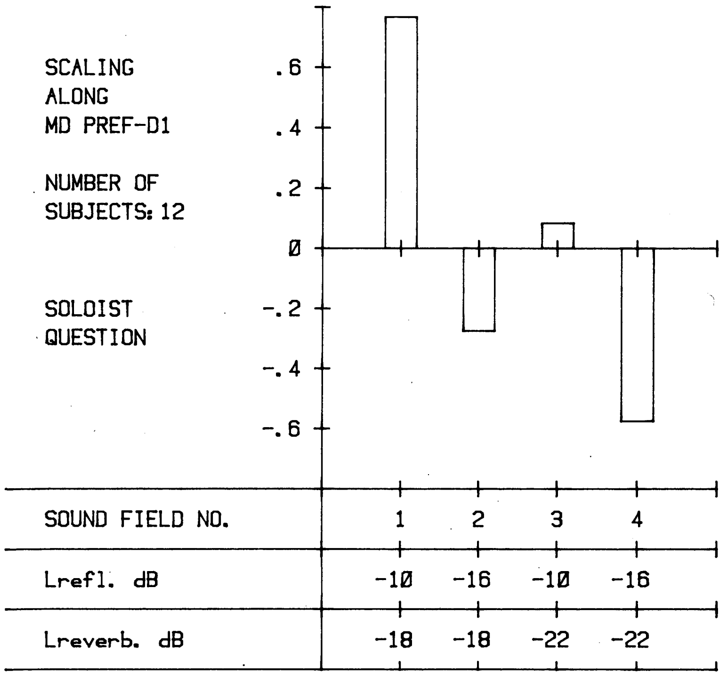


Figure 8: Subjective preference scores for the four sound fields in Fig. 7.

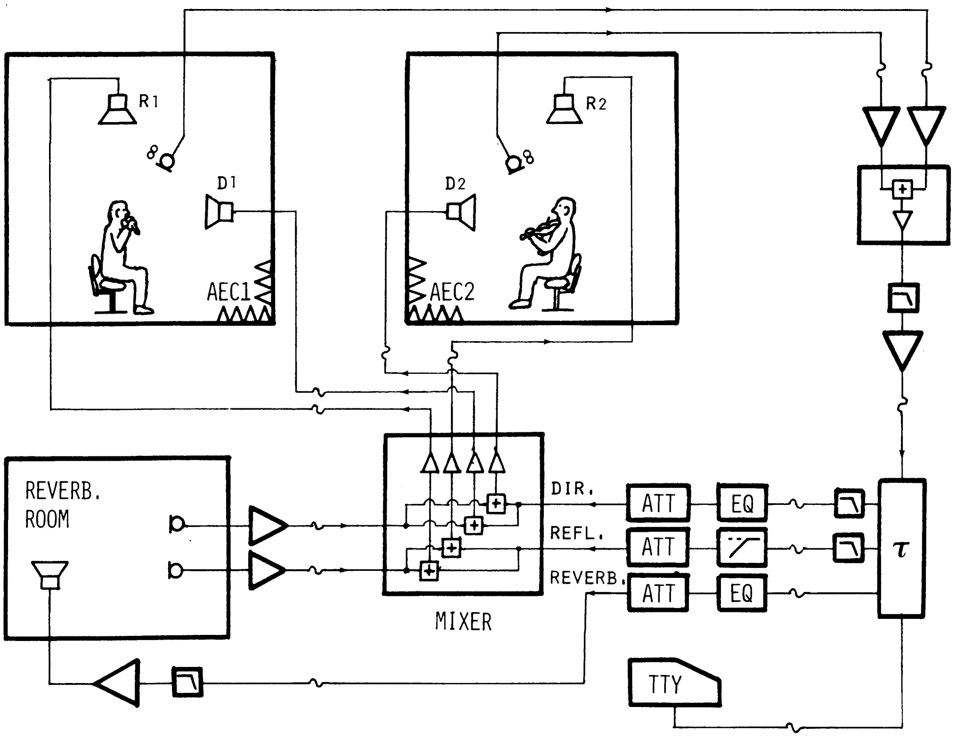


Figure 9: Set-up used for simulation of ensemble conditions for two musicians in a symphony orchestra.

after playing.

After having removed "irrelevant" information from the answers, using the multidimensional statistical method mentioned in section 3, the preference results appeared as shown by the bars in Fig. 8. The scores for the sound fields Nos. 1 and 3 with high early reflection levels are positive, whereas Nos. 2 and 4 with low reflection levels are disliked. (It also appears that a high reverberation level is liked better - or disliked less - than a low reverberation level). Thus, high, realistic, early reflection levels can be audible and are regarded as favourable. Although it has not been demonstrated explicitly, the hypothesis still prevails that the associated subjective effect is one of support.

Experiments on ensemble aspects

In the following, two experiments dealing with the ensemble aspects time delay and hearing each other will be described. These experiments were carried out in the set-up shown in Fig. 9. The separation of the two subjects in two different anechoic rooms made it possible to simulate delay and level of the direct sound corresponding to big distances between two players in an orchestra. A ceiling reflection and reverberation could also be simulated.

In each experiment, five violin/cello and five violin/flute duos participated. The subjects were asked to give preference judgements with respect to ease of ensemble playing after playing their respective parts in Mozart Symphony No. 40, third movement, bars 1-14. Sound example No. 3 demonstrates what this sounded like. First a cello/violin duo is heard and then a flute/violin duo. The recording is stereophonic in the sense that the sound from the cello/ /flute-anechoic room is on the left channel and the sound from the violin-ditto on the right.

The aim of the first experiment was to determine the limit beyond which the delay of the direct sound propagating between musicians influences the ensemble playing. Six sound fields were presented in which the direct sound had different delays relative to the time of emission. Delay and level of the reverberation were equal in all sound fields and the ceiling reflection was turned off. The delay of the direct sound covered the range 7-80 ms equivalent to a range in distance between musicians of 2 to 27 m (Fig. 10). In order to focus on the delay effect, the level of the direct sound was held constant. The fixed level corresponded to an 8 m distance between two musicians sitting in an orchestra.

The results (after having been treated as mentioned for the trio experiment in Fig. 8) appeared as shown in Fig. 11. As one would expect, there seems to be a region of high preference for delays below a certain critical limit (as indicated by the horizontal segment of the dotted line). Beyond that limit, the delay has a negative influence - the preference drops steadily with increasing delay. The critical delay is seen to be placed around 20 ms, corresponding to a 7 m distance. Of course, this limit may depend on the motif played. However, the Mozart piece played here was not particularly rhythmically demanding, and still a 7 m critical distance is well below the dimensions of a symphony orchestra. In accordance with this result, brass and percussion players sitting far back in the orchestra often state that they have to play before the conductor's baton to avoid his accusing them of being too late.

The aim of the other experiment was to determine which components of the impulse response promote musicians' possibility of hearing each other. Eight

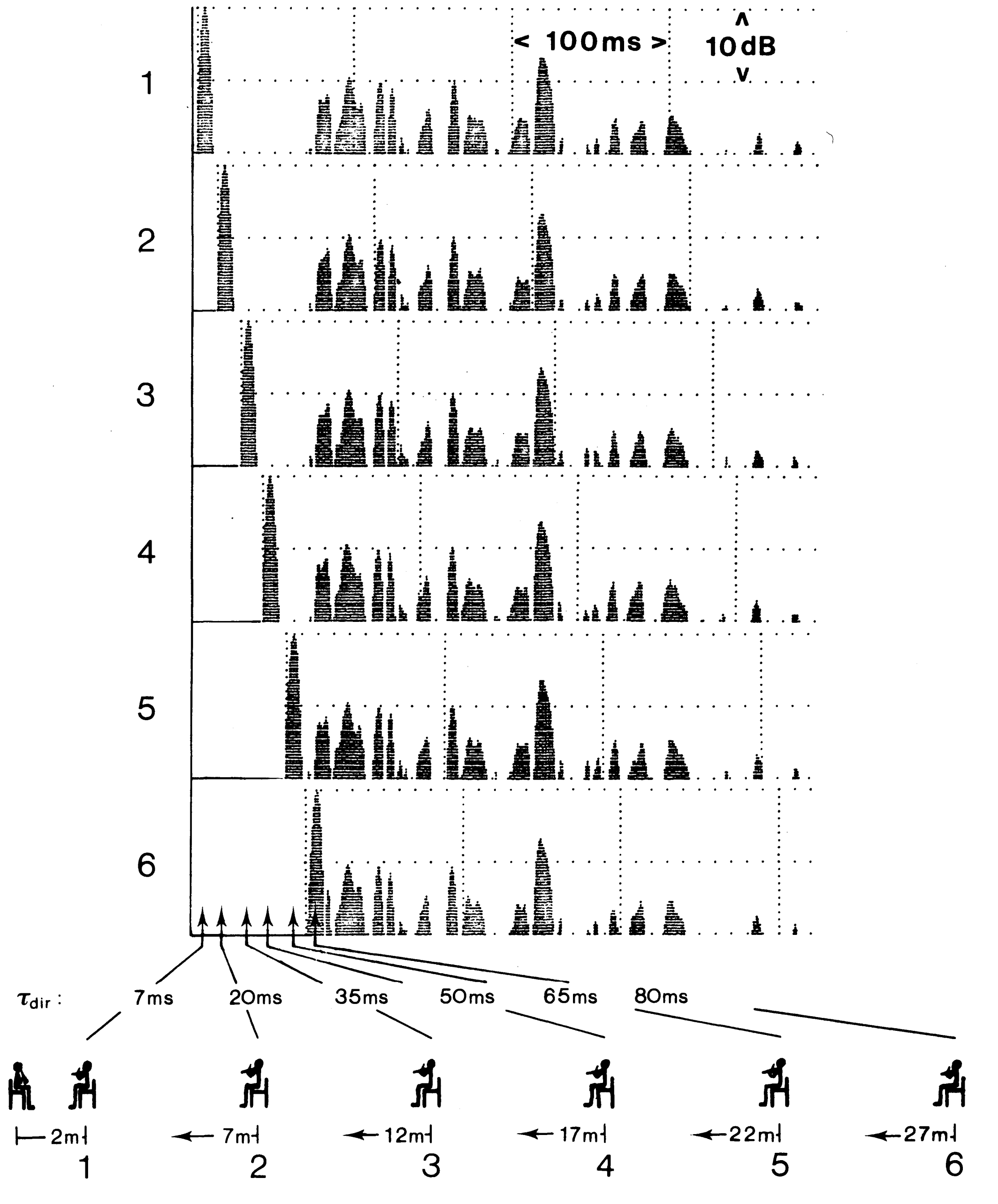


Figure 10: Impulse responses for six sound fields presented in ensemble experiment on influence of delayed direct sound.

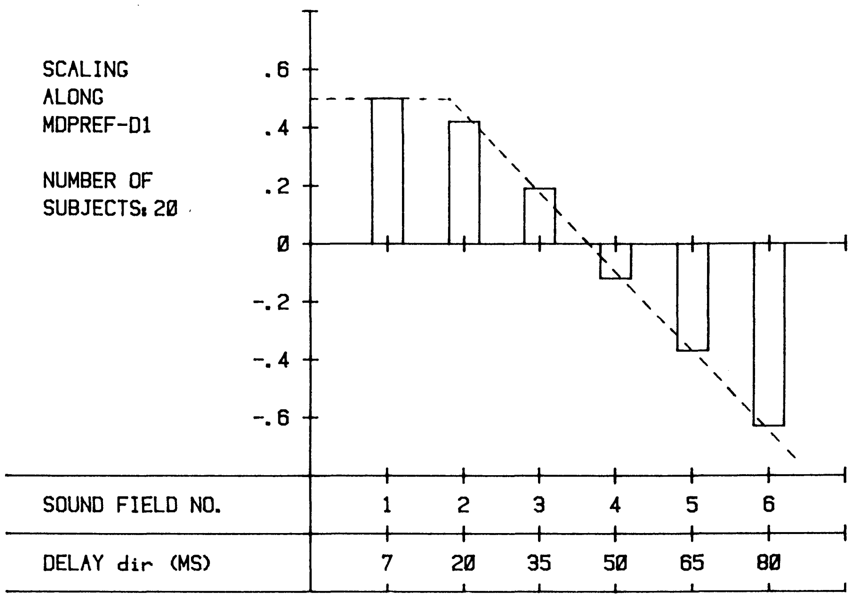


Figure 11: Subjective preference scores for the six sound fields in Fig. 10.

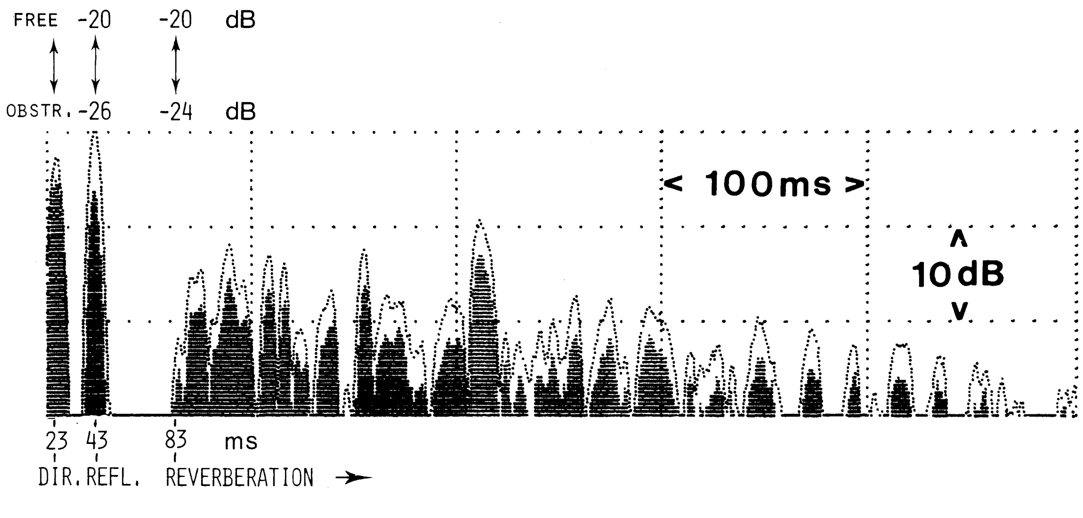


Figure 12: Impulse response indicating the ranges of level variation in ensemble experiment with three sound field components. (The differences in direct sound level were more pronounced at higher frequencies than at the 1 kHz octave shown here.)

different sound fields were created by varying the level of each of the three components: direct sound, ceiling reflection, and reverberation, in two steps as shown in Fig. 12. The direct sound variation is rather small in this 1 kHz impulse response. It was more pronounced at higher frequencies, since it represented the difference in spectrum of the direct sound with and without the sightline between source and receiver being obstructed by the other orchestra members.

The subjects' judgement in terms of MDS-treated preference scalings appears in Fig. 13. When trying to correlate this scaling with objective parameters, it turned out that the key factor was the joint level of direct sound and early reflection relative to the level emitted. The relationship was significant at a 1% level. Similarly, the two sound fields which have been given the lowest scores are those with low level of both direct sound and early reflection. In other words, the vital factor is seen to be the efficiency of early energy transmission between the players.

Another secondary tendency should also be mentioned. If the scalings are compared for those sound fields, between which the only difference is the reverberation level, that is, if sound field No. 1 is compared to No. 2, 3 to 4, 5 to 6, and 7 to 8, it is seen that in three out of the four cases the sound field with lowest reverberation level has been given the highest preference. This may be due to reverberation having a masking effect, but the tendency is not statistically significant.

Suggested objective parameters

Regarding the relationship between the subjective impressions and the properties of the impulse response, the following can be concluded from the experiments described above.

"Soloist concern":

- for players of certain instruments early reflections (between 10 and 100 ms) may be completely masked in some halls, the threshold of perception being 10 to 20 dB higher than the level of a single reflection from a flat, hard surface. Nevertheless,

- audible levels of early reflections are preferred, and the hypothesis still prevails that the audible effect is one of "support".

"Ensemble concern":

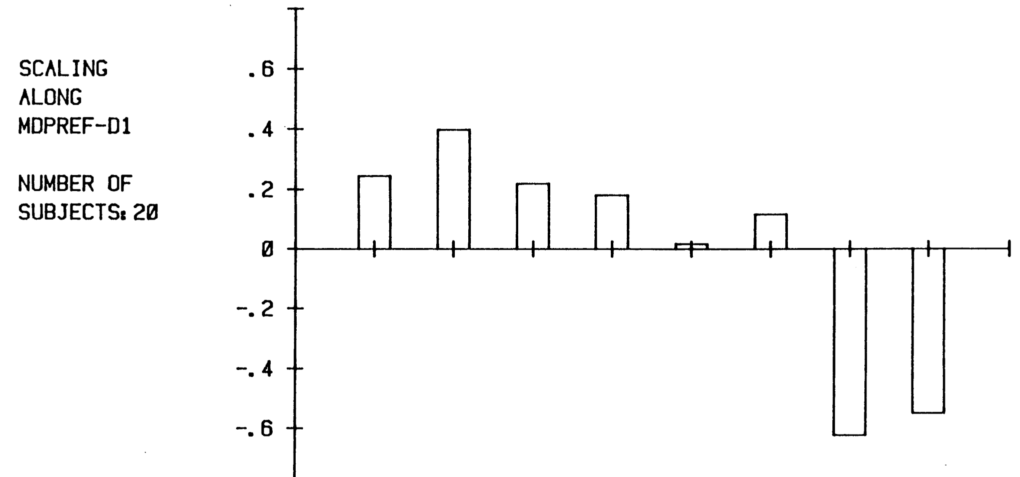
- the delay of the first component, relative to the time of emission, should be small. An unmasked direct sound is therefore very desirable. If masked, it cannot be fully compensated for by strong but more delayed early reflections.

- the level of received early energy, relative to the energy emitted, is important for musicians' possibility of hearing each other, and

- it is possible that reverberation has a negative influence in this respect.

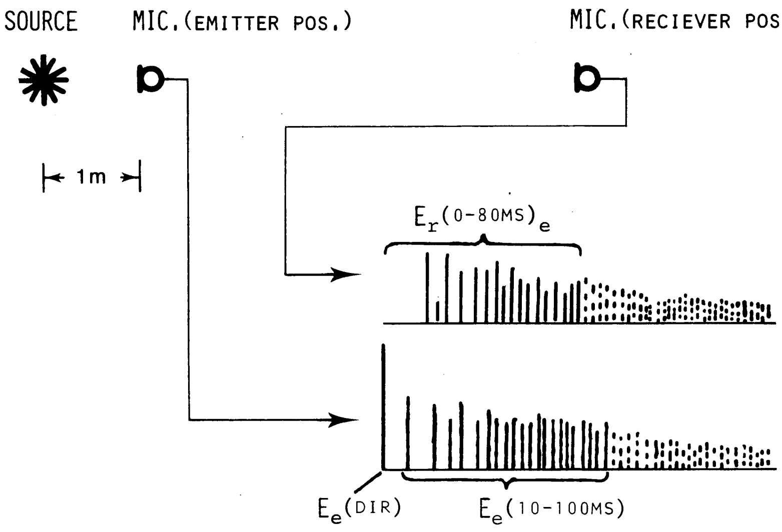
Based on these findings, two objective parameters, Support (ST), and Early Ensemble Level (EEL), have been defined (Fig. 14). The parameters are based on calculations of energy fractions from impulse responses recorded on the orchestra platform.

<u>ST</u> describes the ratio between the energy of the early reflections and the energy of the direct sound. This ratio is measured one meter from the source,



SOUND FIELD NO.	1	2	3	4	5	6	7	8
Ldir-HF (dB)	-19	-19	-19	-19	-26	-26	-26	-26
Lrefl (dB)	-20	-20	-26	-26	-20	-20	-26	-26
Lreverb (dB)	-20	-24	-20	-24	-20	-24	-20	-24

Figure 13: Subjective preference scores for the eight different sound fields which can be created from Fig. 12.



$$ST = 10 \cdot \text{LOG} \frac{E_e(10\text{-}100\text{MS})}{E_e(\text{DIR})} \qquad EEL = 10 \cdot \text{LOG} \frac{E_r(0\text{-}80\text{MS})_e}{E_e(\text{DIR})}$$

Figure 14: The definitions of the objective parameters "Support" (ST) and "Early Ensemble Level" (EEL).

which is comparable to the distance from the performer's ears to his own instrument. ST is thus intended to measure how much the early reflections assist the performer's own efforts - the direct sound - as heard by himself. High ST values correspond to a strong feeling of Support, and vice versa. Depending on the instrument played, the threshold for perceived Support may correspond to ST values around -15 to -10 dB, but this point is still very vague.

EEL is defined as the ratio between the received early energy and the energy emitted - described by the direct sound measured at 1 m distance. The higher the EEL value, the better the possibility for musicians to hear each other.

EEL has been made sensitive to the negative effect of the delay, by the integration interval for the early energy in the numerator being counted from the time of emission. Therefore, just increasing the delay will also result in lower EEL values. It is only natural to use the time of emission as a reference, keeping in mind that one important aspect of the ensemble playing is the synchronization. As a result, EEL measures the efficiency of the sound transmission within the orchestra, with respect to speed as well as to level.

From these experiments, with only two musicians participating, it is not possible to suggest optimal ranges for the EEL parameter. In the orchestra, optimal conditions for the musicians to hear each other will also require that the transmitted sounds from various instruments do not mask each other. In view of the physical arrangement of the different instruments on the platform and their different powers and directional patterns, it is likely that EEL values need to be different between different groups of instruments.*

Relationship between objective parameters and hall design

The research described above has only concerned the four rightmost boxes in Fig. 1. The actual design of halls and orchestra platforms has not been dealt with explicitly. When looking for architectural parameters, however, the experimental process is in principle the same as that described in section 3, except that the attention is now shifted to the four leftmost boxes of Fig. 1. First we need to collect architectural data from and make impulse response recordings in different hall designs. This material should then be analysed by

- derivation of possible architectural parameters from the hall data,
- measurement of the objective parameters from the impulse responses, and finally
- correlation of the two sets of parameters in order to see, if any of the architectural parameter candidates show a promising relationship with any of the objective parameters.

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*) Here a certain discrepancy between the results mentioned above and those of some other researchers should be mentioned. For instance, Marshall et al (1978) and Nakayama (1984) have stated that early reflections are only useful or preferred within a much narrower time interval than the 80 or 100 ms considered by ST and EEL. However, another experiment not mentioned in this paper indicated that such results are likely to be caused by the limitations of the simulated sound-field approach. Therefore, focusing on smaller time intervals is hardly relevant in practice.

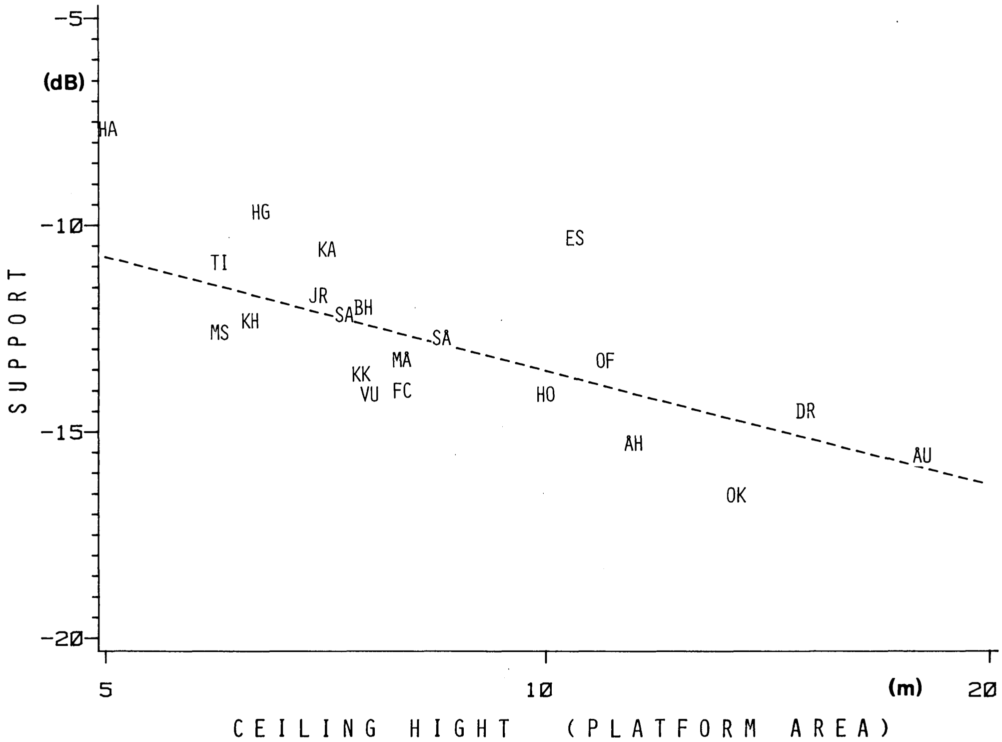

Figure 15: ST values versus average ceiling height over the platform for 21 Danish halls. (The dashed line is the best fitting linear model: the correlation coefficient is -0.74).

This procedure was tried in connection with a major survey of acoustic conditions in Danish concert halls (Gade & Rindel, 1984), where 21 halls of major importance for the performance of symphonic music were investigated. Among the objective parameters measured were ST and EEL as defined in the previous section. The averaged parameter values from three positions on the platforms were correlated with various geometrical data derived from drawings. Both ST and EEL showed high correlations with various measures of distance to reflecting surfaces around the platform. This was not an unexpected result, since both parameters are sensitive to the amount of early reflection energy returned to the platform. Due to the fact that this energy is governed by a complicated interaction between many factors (mutual distances, angles and absorption characteristics of the various surfaces), it may not be possible to create a single architectural "wonder" parameter including all relevant factors. As examples of the relationships found, however, the corresponding values of ST and ceiling height over the platform are shown i Fig. 15 and the values of EEL versus platform "volume" in Fig. 16. As expected, the values of both objective parameters are seen to be reduced as the distances increase.

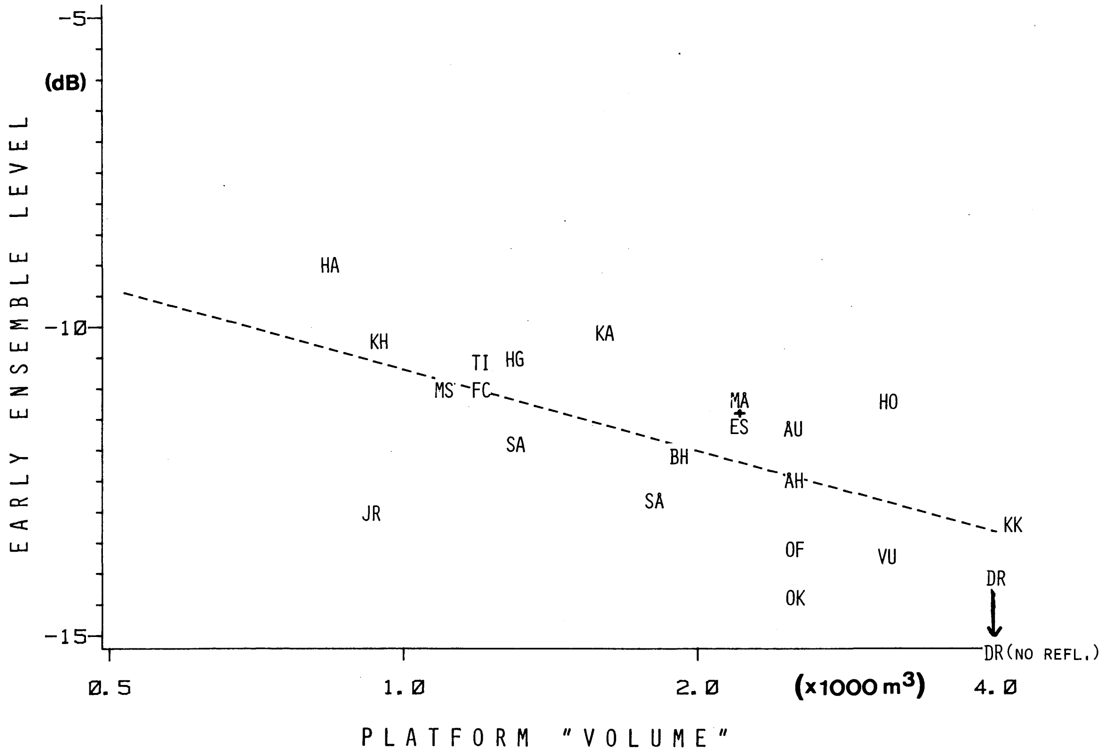

Figure 16: EEL values for 21 Danish halls versus platform "volume" (meaning average ceiling height x average distance between sidewalls x average distance from platform front to rear wall. The dashed line is the best fitting linear model; the correlation coefficient is -0.66).

The data points corresponding to each of the 21 halls have been marked with initials, among which TI and DR mentioned in section 4 can be found. A few years after the inauguration of DR (in 1945) an array of reflector "clouds" was installed over the platform (Fig. 17), because the musicians complained that it was difficult to hear each other. In order to investigate the effect of these reflectors, EEL was measured both with the reflectors in their normal position (about 7 m above the platform floor) and raised to immediately below the ceiling. From the corresponding EEL values in Fig. 16 it is seen that the reflectors have only a very limited influence. The EEL value with reflectors is still far from being comparable with the values from halls with a good reputation concerning ensemble conditions, such as TI. And - as could be expected - the musicians are still complaining.

Examples like this illustrate the usefulness of objective parameters in design and in attempts to make improvements. Had a relevant objective parameter been available at that time, one could have argued for a more effective solution.

In addition to the relationships verified through the concert hall survey

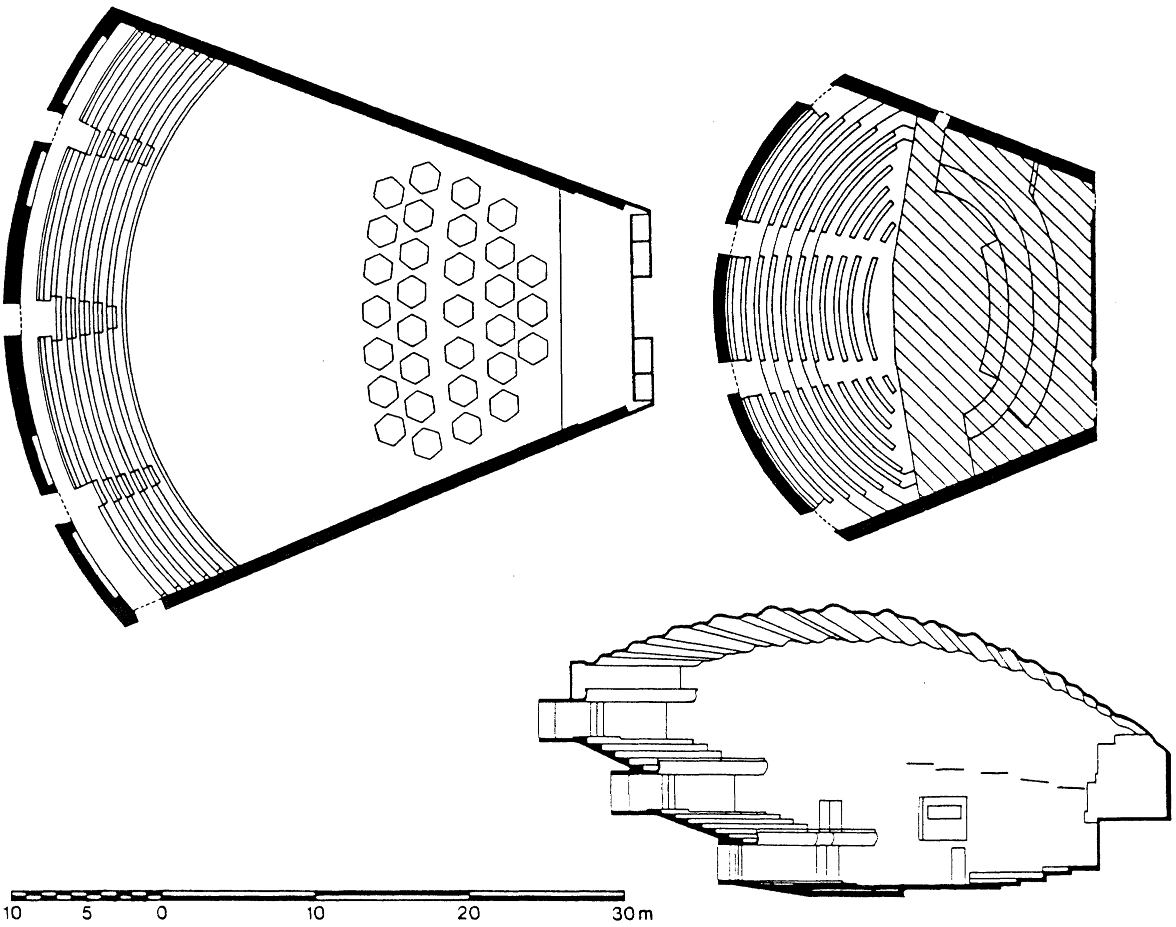

Figure 17: Plan and section of DR showing normal position of platform reflectors.

mentioned above, there are a number of other factors which obviously will influence EEL, i.e. the ease of ensemble playing. For instance, the propagation of the direct sound will be influenced by the use of risers for the outer sections of the orchestra and by the mutual distances between the players. Because of the directivity of the instruments and of musicians' ability to hear sounds from others while playing (Meyer & Biassoni de Serra, 1980), it is also possible that some problems can be solved just by trying new seating arrangements for the orchestra (Meyer, 1978), or by placing a few reflecting surfaces at strategic places (Meyer & Biassoni de Serra, 1980).

Design aspects of importance for ease of ensemble playing have also been discussed by a number of other authors, e.g. by A.H. Benade at an earlier seminar in this series (Benade, 1980).

Concluding remarks

In this paper we have discussed

- the various room acoustic requirements of musicians in terms of a set of subjective parameters,

- experimental results leading to a definition of two objective parameters ST and EEL, whereby the quality of a concert hall with respect to the requirements for "Support" and "ease of ensemble playing" can be measured objectively, and finally,

- some of the factors in architectural design which govern the variation in these objective and subjective parameters.

As can be seen, all aspects of the room acoustic question as outlined in section 1 have been considered. Still, one should not be tempted to think that all problems have been solved. Rather it should be remembered that this work is just one of the first serious attempts in entering this field, and many loose ends exist with respect to the proper definition of the objective parameters. For instance, one could well imagine reflections beyond 100 ms being useful for support, and with the present technique EEL cannot take the directional characteristics of instruments and of the listening ability into account.

I nonetheless hope that the presentation of this paper may contribute to architects' and consultants' understanding of musicians' room acoustic problems; and that the rather detailed description of the experimental techniques has demonstrated to perhaps impatient musicians (who have known these problems longer than any acoustician) that progress in this field is not made as easily as snapping one's fingers.

Acknowledgments

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STAGE FLOORS AND RISERS – SUPPORTING RESONANT BODIES OR SOUND TRAPS?

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Introduction

A tuning fork which is struck and held freely in the hand will only produce a faint sound. If instead the handle of the fork is held against a hard surface, such as a table or a guitar, the emitted sound becomes much stronger. This everyday experience illustrates the question addressed in this study.

The largest of the bowed string instruments, the cello and the double bass, are supported on the floor via an adjustable metal pin - the end pin, or the peg. This arrangement may have acoustic implications. As with the tuning fork, it is possible that part of the body vibrations of these instruments could be transmitted down through the end pin, setting the stage floor into vibration, see Fig. 1. The vibrating stage floor would then act as an enlargement of the instrument body, and contribute to the radiated sound.

The prime function of the end pin is to support the instrument, so any acoustic action on its part would be a kind of side effect. The end pin of the cello has developed in order to facilitate the playing. In earlier days the cello was held between the knees. As for the double bass, its size has always necessitated support, but in the baroque era, the instrument was often supported on a low, soft foot-stool instead of directly on the floor.

Vibrating tuning forks and radiation impedance

Why is the sound reinforced when the tuning fork is held against the table? In short, the answer is captured in the saying: "You can't fan a fire with a knitting-needle!" An acoustician would rephrase this to: "The radiation impedance increases when the tuning fork is held against the table."

Consider one of the vibrating rods of the struck tuning fork. When the rod is moving in one direction, the air particles on one side of the rod are temporarily forced away into the neighboring air layers. This abnormal concentration of air particles gives rise to a temporary increase in pressure in that region. Later, when the rod swings back in the opposite direction, this pressure increase

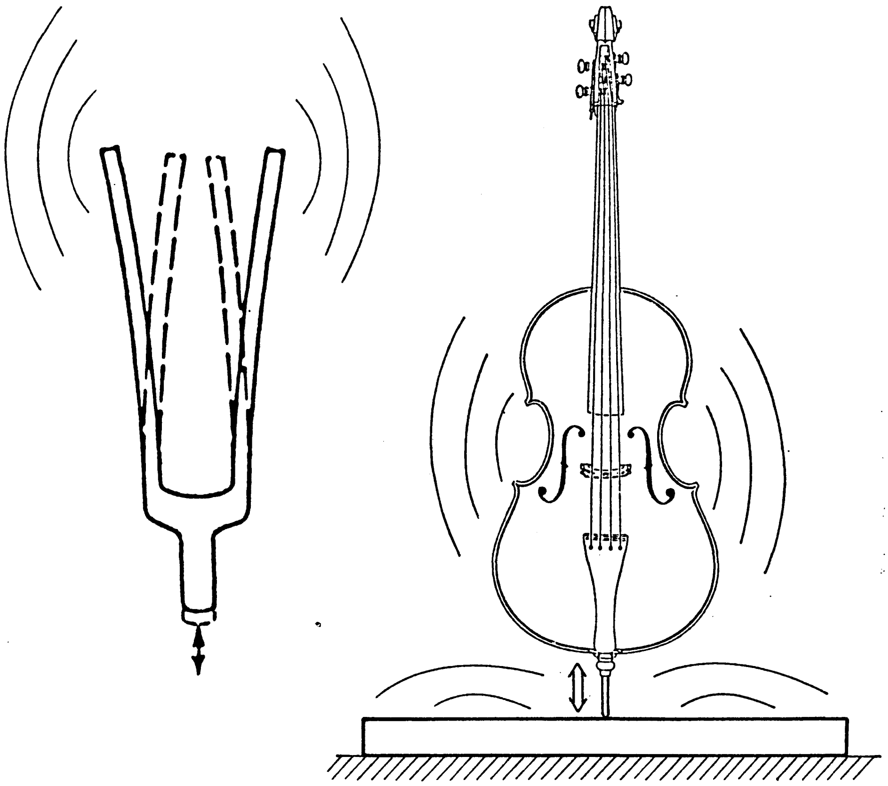

Figure 1: The vibrating rods of a struck tuning fork transmit vibrations down into the handle. When the fork is held against a table, the perceived loudness increases, as the large tabletop is also set into vibration and radiates sound. A similar phenomenon may occur when the cello and the double bass are played on a resonant support.

gradually changes into a rarefaction. A sound wave consisting of alternating compressions and rarefactions will result, propagating out from the tuning fork.

However, the rod is of small dimensions, and does not succeed in moving very much air back and forth. Further, before the rod swings back, much of the air which has been temporarily displaced on one side of the rod will manage to rush round to the opposite side, where, at the moment, air is "lacking". So, because of the tiny dimensions of the rod, the expected compressions and rarefactions in the air will be very much reduced. As a consequence, a listener at some distance will perceive only a faint sound, even though the tuning fork may be vibrating vigorously.

When the tuning fork is held against a tabletop, a large area is set into vibration. This large area is much better at moving air and at preventing compressed regions from being cancelled by rarefactions. The radiated sound is reinforced compared to the free fork and heard at a larger distance. Returning to the fire and the knitting-needle, the keen experimenter fanning the fire will soon discover the higher radiation efficiency of a larger object. The knitting needle does not impede the motion of his arm to any appreciable extent, but a large tray will.

In acoustics, the term radiation impedance is used to denote the efficiency of a vibrating object in radiating sound. The radiation impedance depends on both the size of the vibrating object and on the frequency of vibration. A large object meets a higher radiation impedance than does a small one, and as the frequency of vibration is increased, the radiation impedance also increases. When the tuning fork is applied to a tabletop, the vibrating system is considerably enlarged, and so its radiation impedance increases. In the "fire" example, the high radiation impedance seen by the large tray makes it possible to deliver more energy into the "sound wave" fanning the fire.

Radiation impedance and body size

How large must an object be in order to radiate sound efficiently? It depends on the frequency of the sound as hinted at above. As a rule of thumb, the vibrating body must have dimensions which are a considerable fraction of the wavelength of the radiated sound. The wavelength is calculated as the speed of sound in air (approximately 340 m/s) divided by the frequency of the sound. This means that the wavelength becomes shorter the higher the frequency. If the tuning fork is made to sound an A_4=440 Hz, the corresponding wavelength is roughly 0.8 m. In comparison, the rods of the tuning fork are certainly small, whereas a table is not. Consequently, the fork and the table together constitute a more efficient radiator of sound than the fork alone.

As for the string instruments, the fundamental of the lowest note on the cello (65 Hz) has a wavelength of approximately 5 m. On the double bass, the lowest note (41 Hz) corresponds to a wavelength of more than 8 m. Compared to these wavelengths, the instrument bodies are relatively small, about one-seventh, so the lowest partials can be assumed to be weakly radiated.

Even a table will occupy only a fraction of a wavelength at these low frequencies. This means that a hypothetical grand tuning fork, sounding a bass note, would gain but little in sound radiation when held against the table.

Making a short digression, we note that the gain in radiation impedance from the table, which facilitates the radiation of the tuning fork sound, will also make the note sound a shorter time. The energy quantum which is fed into the tuning fork at the initial blow is now rapidly drained because of the increased efficiency of the sound radiation. The free tuning fork consumes the same amount of energy by sounding a faint note for a longer time. This trade-off between loudness and duration of a note applies, for example, to the guitar and the piano. With the bowed instruments, which are excited continuously by the bow, an increased radiation efficiency means that more of the mechanical energy supplied by the player is converted to acoustic energy and reaches the listener as radiated sound.

Summarizing our reasoning on sound radiation, we have learned that it is quite possible to fan a fire efficiently if the fanning object is large enough. Similarly, it is possible for a sound source to convert its vibration energy efficiently into a sound wave, provided that the vibrating area of the source is large enough. Adapting to the acoustic terms introduced, we can be more specific and say that the source must be large enough to give a reasonably high radiation impedance at the vibration frequency. This requires that the dimensions of the sound source be comparable with the wavelength of the sound.

Vibrating floors and mechanical impedance

We are now clear as to how the sound emission from the cello and the double bass could benefit from a resonant support. But will the instruments manage to set the stage floor into vibration?

The answer depends on the vibrational properties of the floor. Intuitively, we expect a thin wooden floor, resting on open beams, to be easy to set into vibration. In contrast, a thick concrete floor would be resistant even to violent excitations. The scientific measure of the "reluctancy to vibrate" is called mechanical impedance. A high mechanical impedance means that the object is hard to set in vibration; the object impedes the motion. A wooden floor will exhibit a considerably lower mechanical impedance than will the concrete floor.

However, a given type of floor can not be characterized by just a single impedance value. If the instrument happens to be on top of a supporting beam of the wooden floor, the response from the floor is closer to that of the concrete floor than when the instrument is located somewhere between beams. So, depending on where on the floor the instrument happens to be located, the floor will respond with vibrations of different strength. This variation in floor impedance with position will be relatively large in a floor constructed from supporting elements covered with plates, such as the wooden floor. The mechanical impedance of a concrete floor is uniform and very high, so for our purposes it can be considered rigid.

A maximum delivery of vibration energy from instrument to floor is theoretically reached when the mechanical impedance of instrument and floor are of equal magnitude. This situation probably never occurs in practice, as floors are usually rather rigid. Accordingly, the reinforcement of the sound by the enlarged "instrument" body-floor should be more prominent when the instruments are placed on rickety floors or risers. This also tallies with the experience of musicians.

Vibrating floors and sound traps

Before continuing, we must note that there is a snag in this whole idea of "sound reinforcement" by the floor. The vibrations of the floor need not necessarily cooperate with the vibrations of the instrument! If at any one moment a flow of air is delivered out from the instrument ("push"), but the floor at the same time retreats ("pull"), this air flow will not contribute to a propagating sound wave. The instrument and the floor will exchange a flow of air, an effect which is detectable only in the immediate vicinity of the instrument. The floor then acts like a "sound trap" rather than a "sound reinforcer", and the sound reaching the audience becomes attenuated.

If, on the other hand, the floor moves upwards at the same moment as the instrument delivers an air flow, the "two" sound sources are cooperative and the radiated sound will be reinforced. So, depending on the relation between the directions of the vibrations of the instrument and the floor respectively ("push/push" or "push/pull"), the net result from the added floor vibrations may vary continuously from a reinforcement to a reduction of the radiated sound.

What's up so far?

In our discussion of vibrating instruments and floors we have arrived at the following conclusions.

A sound source which is small, compared to the wavelength of the emitted sound, is poor at radiating its vibration energy as a sound wave. This case applies to all bowed instruments when played in the lower range of their compasses. In order to increase the amount of radiated sound it would be desirable to improve the radiation efficiency, which amounts to raising the radiation impedance. The radiation impedance of an instrument in the lower range is limited by its size. With the cello and the double bass it may be possible to increase the radiation efficiency by transmitting some of the vibrations in the instrument body down into the floor. The floor has a larger area and therefore a higher radiation impedance.

However, in order to benefit from the properties of a vibrating floor, the instrument must manage to set the floor into vibration. The ratio between the vibration energy which leaves the instrument via the end pin and the energy which remains in the instrument is determined by the ratio between the mechanical impedance of the instrument and of the floor, respectively. If the floor is very rigid, which means that its mechanical impedance is high, any applied motion is effectively impeded and only a very small part of the vibrations of the instrument body will enter the floor.

Well, does it work?

So far, we have anticipated the existence of desirable vibrations in the stage floor, but what is the practical experience of musicians?

Informal questioning reveals that the acoustic support from stage floor and risers as perceived by the musicians can be considerable, provided the vibrational properties of the support are favorable. The literature on architectural acoustics also gives evidence of attention being paid to floor vibrations in concert halls. The Neues Gewandhaus in Leipzig (built in 1896, destroyed in the last war) was famous for the "powerful" sound of the cello and double bass sections (Bagenal & Wood 1931, Beranek 1962), an effect which was attributed to vibrations in the thin wooden parquet floor on which the entire stage rested. However, acoustical measurements in the hall at that time gave only weak support for this theory (Meyer & Cremer 1933, Cremer 1981).

The famous acoustician Beranek proposes the use of a wooden stage floor, "as thin as other considerations permit," with reference to well-liked older halls (Beranek, Johnson, Schultz & Watters, 1964). As today's building codes preclude lightweight floors, new concert halls are designed with a rather stiff stage floor, which exclude acoustical effects due to floor vibrations. This can be compensated for, however, by seating the celli and basses on risers of thin wood.

Apart from safety codes, there are other reasons for making the stage floors rather rigid. Nowadays, concert halls are also used for performances other than orchestral concerts, for example ballet and television shows. For these purposes, a rigid floor is desirable, in order to minimize disturbing noise from dancers and other performers. A further argument for a rigid stage floor is that it can be expected to benefit the acoustics of the hall in a certain respect (Cremer 1981). For the small instruments, for example the violin, which project sound waves to the floor but no mechanical vibrations, a rigid floor can be assumed to be advantageous, since a rigid surface reflects the sound more efficiently than does a vibrating one.

Summarizing, a construction with a thin, wooden stage floor seems to be desirable, as acquired by experience from old approved concert halls. This solution is not possible to use today, but there remains the possibility of using light-weight stage risers as a substitute.

Which experiments were made?

A series of acoustic measurements was carried out to determine the practical influence of stage risers. The measurements were conducted in the Berwald Hall, the concert hall of The Swedish Radio Company. This hall was particularly well suited for the experiments. The stage floor is constructed from muninga hardwood, bonded in asphalt to a concrete foundation resting on bedrock. This construction yields a very stiff stage floor, almost totally resistent to vibrations.

Two professional double bass players were asked to perform, first with their instruments placed directly on the stage floor, in a second session with the instruments supported on risers, and finally with the instruments placed on the stage floor again. Two types of risers were used in the experiments, one small (1 x 1 m), and one large (1 x 2 m) with a cross-bar at the middle of the long side. Both types were covered with 1/2" plywood. The players were placed stage right at their usual position in the orchestra. The recordings were made with several microphones in different positions. The results presented in the following refer to a near-field microphone very close to the players (about 1 m), and a far-field microphone above the audience, at a distance of about 12 m.

The players performed chromatic scales with a compass from the low E-string to the open G-string, covering a fundamental frequency range from 41 Hz to 98 Hz. The instruction to the players was to play all scales identically with regard to dynamic level, tempo, bow velocity, bow force and distance between bow and bridge. The players chose to play at a mezzoforte level.

The recordings were analyzed by computing the Long Time Average Spectra (LTAS). In a LTAS-analysis the audible frequency range (20 - 20 000 Hz) is split up into 30 sections, "frequency bands". The curve in the LTAS-diagram represents the average sound levels in these frequency bands during the analyzed sound excerpt. In contrast to the usual type of spectra, the strength of the individual partials are not discernable in a LTAS.

What were the results?

The acoustic influence of the risers was determined by comparing LTAS from performances made on the floor and on the two types of stage risers used, respectively, see Fig. 2. Each curve represents the average of several scales, corresponding to between two and four minutes of analyzed sound. The figure shows that playing on the risers gives a gain in the radiated sound in the frequency range around 50 Hz and also above 150 Hz. The gain is different for the two microphone positions and for the two types of risers used. The largest effects are observed with the large riser in the near-field position. Focus-

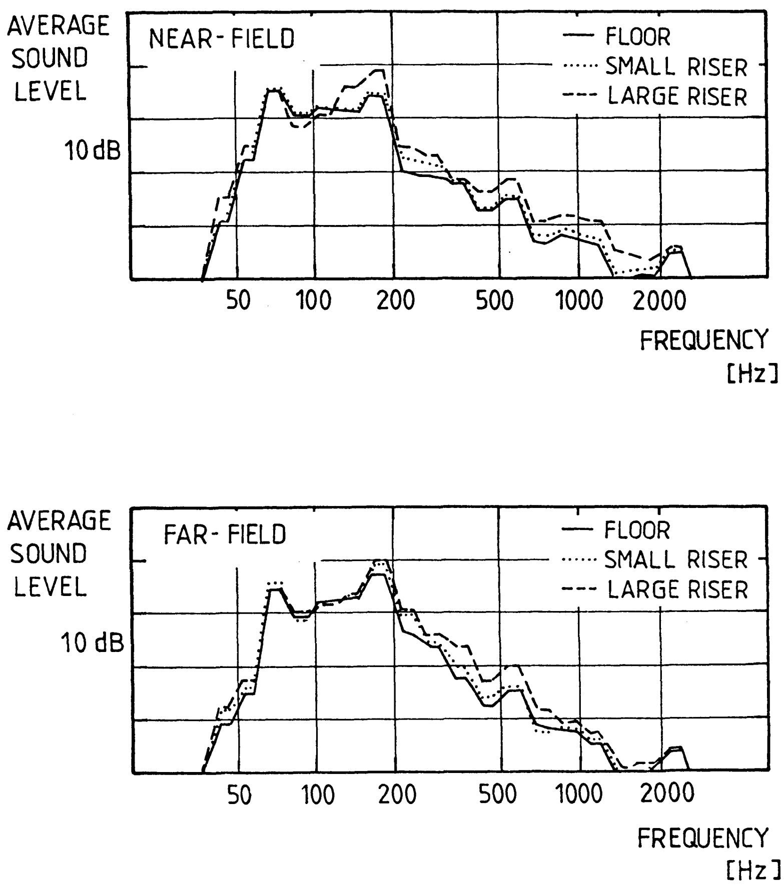

Figure 2: The double bass can gain in sound radiation when played on stage risers. The gain is different depending on the type of riser. The figure shows the Long Time Average Spectra (LTAS) of chromatic scales played on two double basses in a concert hall, as recorded by a microphone in the near-field, close to the instruments (top), and in the far-field, above the audience (bottom). The basses were positioned on the stiff stage floor (full lines), small risers (dotted lines) and large risers (dashed lines). Each curve represents an average of several repeats, amounting to a couple of minutes of analyzed sound.

ing on the far-field microphone, which is representative of the sound perceived by the listener, the large riser gives a maximum gain around 50 Hz on the order of 3 dB. In the range 150 -- 1000 Hz the maximum gain reaches about 5 dB. These effects of the large riser are not negligible, and can be expected to have a perceptible influence on the sound in the hall.

It should also be noted that there are regions in which the sound radiation

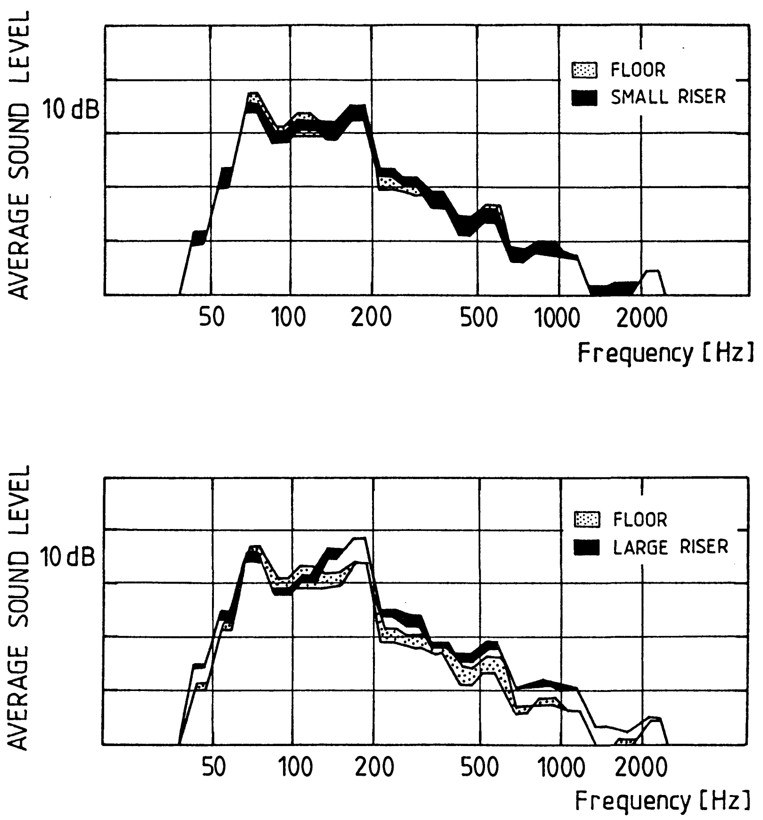

Figure 3: The players will not play the scales identically from one rendering to the next. The figures show the variation in LTAS between repeats (near-field microphone). The dotted area in both diagrams represents the variation in LTAS between seven repeats played on the floor. This condition is compared with the performances on the risers (filled areas); eight repeats in two positions on the small riser (top), and four repeats in two positions on the large riser (bottom).

is decreased when playing on risers, especially around 100 Hz in the near-field of the small riser.

LTAS and sound quality

Emphasizing the sound level in a certain frequency range will change the perceived sound quality, the change being different depending on frequency. An illustration of the perceptual contributions from different frequency ranges to the double bass sound is given in Sound Example 1. In this example, a recording of a double bass is alternately low-pass and high-pass filtered using a successively lower cut-off frequency of the filters. In this way selected parts of the spectrum can be eliminated, demonstrating the effects of a reinforcement in different frequency ranges on the perceived sound quality. The listener will notice that a reinforcement above 200 Hz will enhance the sonorous quality of the bass sound, while a reinforcement below 200 Hz contributes to a full bass sound.

Are the results reliable?

Now, are the observed differences in the radiated sound really due to the risers, or do they reflect differences between the performances on the floor and risers?

In fact, the musicians were skilled at making consistent repeats, both on the floor and and on the risers. The areas in Fig. 3 represent the variation in LTAS between several repeats on the floor and on the two types of risers used. The repeats on the risers were made using two different positions, and consequently include possible variations in reinforcement with position on the riser. The figure shows that the rather small variations between repeats do not make the LTAS-areas merge to the extent that they would render the differences between the three playing conditions meaningless. Consequently, the players could be considered capable of making repeats under the same conditions with satisfactory reproducability.

This does not necessarily imply that they played identically when seated on the floor and on risers respectively. Some of the observed reinforcement may have been caused by the players, who reported a stronger "response" when playing on the risers. This may have elicited a different way of playing as explained below, although the players were explicitly instructed to play as identically as possible in all repeats.

"Press" and "flow"

A string player can vary the harmonic content of the radiated spectrum by changing the bowing. By using a high bow force and bowing close to the bridge, the player can enhance the higher partials. By moving the bow towards the fingerboard, decreasing the bow force and increasing the bow velocity, the player may produce a note at the same dynamic level but with enhanced lower partials and suppressed higher partials. The perceptual dimension associated with this variation in spectrum content may be termed "press - flow" in analogy with the terminology used for the human voice (Sundberg, 1986). A demonstration of the perceptual differences between "press" and "flow" is given in Sound Example 2.

The LTAS of an exaggerated pressed and exaggerated flowed performance as registered in the near-field position are shown in Fig. 4. The figure clearly illustrates the typical changes in radiated spectra between the two extremes of playing. The differences in bowing between these performances give changes in the LTAS which approach the differences observed between playing on the floor and on risers respectively. However, it was necessary for the players to change their bowing drastically, in order to give differences as large as those shown in the figure. It seems most unlikely that the strictly instructed, professional players would have first used a certain bowing on the floor, changed to an essentially different bowing when playing on the risers, and then reverted to the first mode when returning to the floor; nor were any such changes observed by the experimenter. In order to verify this observation by measurements, it would be possible to repeat the experiment with accelerometers mounted on the bridges. In this way, any changes made by the players could for certain be separated from the contributions of the risers.

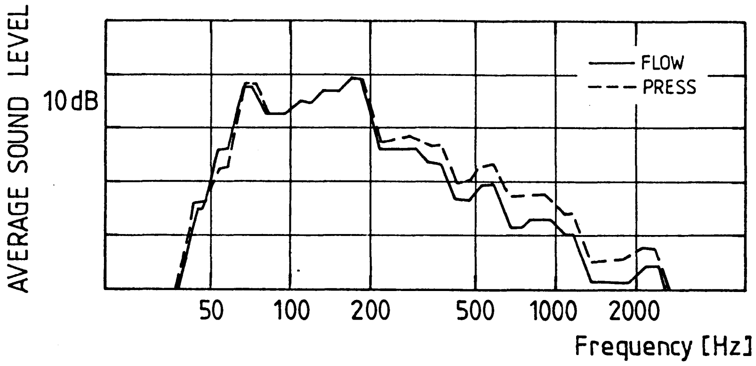

Figure 4: String players can vary the spectral content of the sound by changing the bowing. The figure illustrates the difference between LTAS of a "pressed" and "flowed" performance (exaggerated versions, near-field microphone).

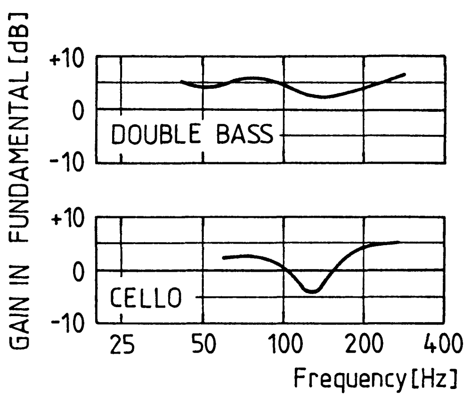

Figure 5: Gain in fundamental for the double bass section in the Philharmonic Hall, New York, when playing on risers compared with the concrete floor. Adapted from Beranek et al (1964).

However, already the above results and considerations indicate that the variability in the measurements on part of the players was small enough to make conclusions from the LTAS meaningful.

Are the results plausible?

The influence of risers on the sound of cello and double basses has to some extent been studied earlier. Beranek et al (1964) measured the differences in the radiated sound of celli and double basses when played on a concrete floor and on provisional risers, in connection with alterations to the Philharmonic Hall in New York. The results were presented as the gain in the level of the fundamental as measured in the reverberant sound field in the hall, see Fig. 5. The gain reported for the double basses was approximately 5 dB between 40 Hz and 100 Hz and above 200 Hz. In the range 100-200 Hz the gain was less, about 3 dB on the average.

For the cello, Beranek et al observed a weaker reinforcement than for the double bass. In the range 100 - 150 Hz the radiated sound was even weaker when playing on risers than playing on the concrete floor. This loss may be caused by cancellation between the sounds from the riser and the instrument respectively, as outlined above. Another possible explanation might be that the vibrating riser does not reflect that part of the radiated sound which is directed downwards from the instrument. In contrast to the floor, the riser then acts as an absorber rather than as a reflector.

In another study of the cello and risers (Fuchs 1964), a bowing machine was used to excite the instrument. The measurements were performed in an anechoic chamber using an authentic riser from a symphony orchestra. The results were given as the gain in overall sound pressure level in the nearfield, about 1.5 m from the instrument. A substantial gain in the radiated sound, attributable to the riser, was observed between 64 Hz and 80 Hz, reaching a maximum of about 5 dB. However, the general trend was a loss in sound radiation, especially between 130 Hz and 180 Hz. As pointed out, the observed gain in the low frequency range is probably desirable, since the instrument body lacks resonances to promote radiation below the Helmholtz resonance (about 95 Hz in the cello).*

A fundamental question dealing with the proportion between the sound energy directly radiated from the instrument and the energy transmitted through the end pin was studied by Cremer as early as the 1930's (Cremer 1981). By measuring the sound levels in the room where the instrument was played and in a room underneath the floor, he was able to estimate the amount of vibration energy transmitted through the end pin. For a double bass, the radiation from the floor was about 5% of the acoustic energy directly radiated from the instrument body. For the cello the corresponding value was only about 1 %. In retrospect, however, the experimenter points out that these values probably are underestimations due to lack of appropriate measurement techniques at the time of the experiments.

Cremer concluded that the vibrations transmitted via the end pin would make only a minor contribution to the sound radiation in the room where the instrument was played. On the other hand, the end pin could certainly be expected to give a substantial improvement in the sound transmission down to a space underneath the floor, such as a neighboring flat.

The gain in sound radiation by risers reported in the studies is surprisingly high, sometimes approaching 6 dB in certain frequency ranges. In these cases, the risers would contribute to the sound radiation almost as much as the instrument bodies themselves. In the present study, the observed maximum gain reached the same magnitude, although the overall trend was lower than in the double bass study by Beranek et al (1964). However, some caution in making the comparisons is necessary, due to the differences in the analyses.

In summary, the studies indicate that the radiated sound of celli and double basses can be reinforced by the use of risers. Although the overall effect usually is one of louder sound, it may well occur that the radiation in certain frequency ranges is attenuated.

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*) The Helmholtz resonance is the "bottle" resonance in the body of a bowed instrument, easy to hear when blowing across the f-holes of a violin.

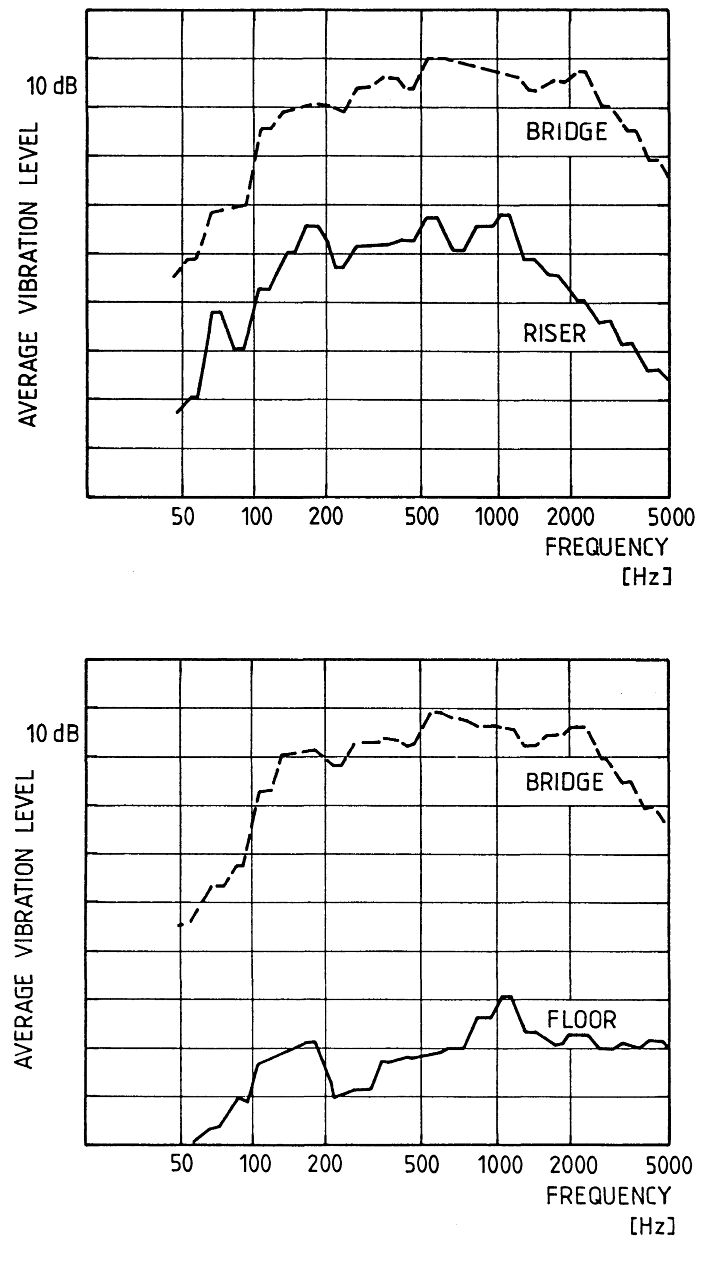

Figure 6: The vibrations of a played double bass are partly transmitted down into the support via the end pin. The figure shows LTAS of the vibration (acceleration) in the bridge and in the support of an played ("active") bass, when positioned on a riser (top), and on a stiff stage floor (bottom).

What about the colleagues in the orchestra?

Up to now we have been concerned with the transmission of vibrations from an instrument down into the stage floor, which is set in vibration and in turn radiates sound. However, not all of the vibration energy in the floor will be radiated as sound. Apart from losses during the transmission in the floor, adjacent basses and celli in the orchestra could be assumed to consume energy from the floor by picking up vibrations through their end pins. In addition to this indirect transmission path via the floor, the instruments will also exchange vibrations directly via the sound transmission in the air.

The efficiency of this exchange of energy between instruments was investigated by comparing the vibrations in the bridge and in the support close to the end pin for two conditions, both on a riser.

In the "active" condition, the bass was played. The vibration levels (acceleration) in the bridge and in the riser close to the end pin of this "active" bass are compared in Fig. 6 (top). The difference in vibration levels between bridge and riser is roughly 30 dB.

In passing, the corresponding vibration levels for the case where the bass is supported directly on the stage floor is shown in Fig. 6 (bottom). The difference between the vibration levels in the bridge and in the floor is more than 50 dB, which confirms the initial assumption that the stage floor itself is almost rigid.

Turning to the second condition, the same bass was placed on the same riser, but left "passive", and exposed to the excitation from another bass, which was played on the same riser at a distance of about 1.5 m. The "passive" bass is now set into vibration by the excitation from the played instrument. Figure 7 shows the vibration level in the bridge of the "passive" bass compared with the corresponding level when it was "active". The difference is large at higher frequencies, more than 30 dB above 200 Hz. At lower frequencies, however, the transmission between the basses grows more efficient, and at approximately 60 Hz the bridge vibrations are of equal amplitude whether the bass is "active" or "passive". This high vibration level in the "passive" bass at low frequencies is probably due to the Helmholtz resonance (about 65 Hz), which is easily excited externally.

The proportion between vibrations transmitted through the air and via the end pins was estimated by supporting the passive bass on a piece of foam rubber, with the aim of preventing the vibrations in the riser from reaching the "passive" bass. The insertion of the foam rubber support did not noticeably change the vibration level in the passive bass, apart from a loss between 100 Hz and 200 Hz, see Fig. 7. This result indicates that the air path is more important to the coupling between the instruments than are the riser vibrations.

The above result was corroborated when the experiment was repeated with both instruments on the stiff floor. The only discrepancy compared with the measurement on the riser was a substantial loss in transmission below 70 Hz. In this frequency range, the vibration transmission via the end pins seems to make a major contribution to the bridge vibrations in the "passive" bass. This effect was not discernible in the foam rubber experiment, probably because the foam rubber could not prevent the riser vibrations from entering the "passive" bass at the lowest frequencies.

In all, an exchange of vibrations was observed between adjacent basses, which at low frequencies was considerable. It

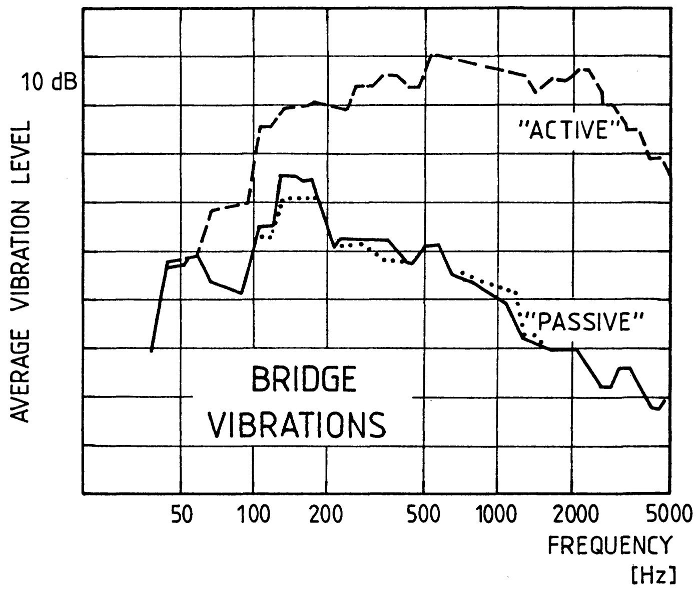

Figure 7: The vibrations in the support together with the near-field sound will set other basses into vibration. The figure illustrates the difference in the bridge vibrations when a bass is played on a riser ("active"), and when exposed to the excitation from another bass on the same riser ("passive"). The dotted line shows a case where the "passive" bass was isolated from the riser vibrations by means of a piece of foam rubber.

is an open question whether this mutual exchange of vibrations within a double bass section, be it placed on risers or not, is important to the ensemble playing, for instance as regards intonation.

The vibrations in the floor, riser and the bridge of the "passive" bass as picked up by the accelerometer are recorded in Sound Example 3. Interestingly, the riser as well as the instrument body act as "microphones" to all incident sound waves. The voice of the experimenter instructing the players is clearly audible via the riser or bridge vibrations, but colored by the uneven "frequency response" of the "microphones."

Body size and radiated partials

Returning to the sound radiation in the air, we recall the following statement from the introductory sections; "A sound source which is small compared

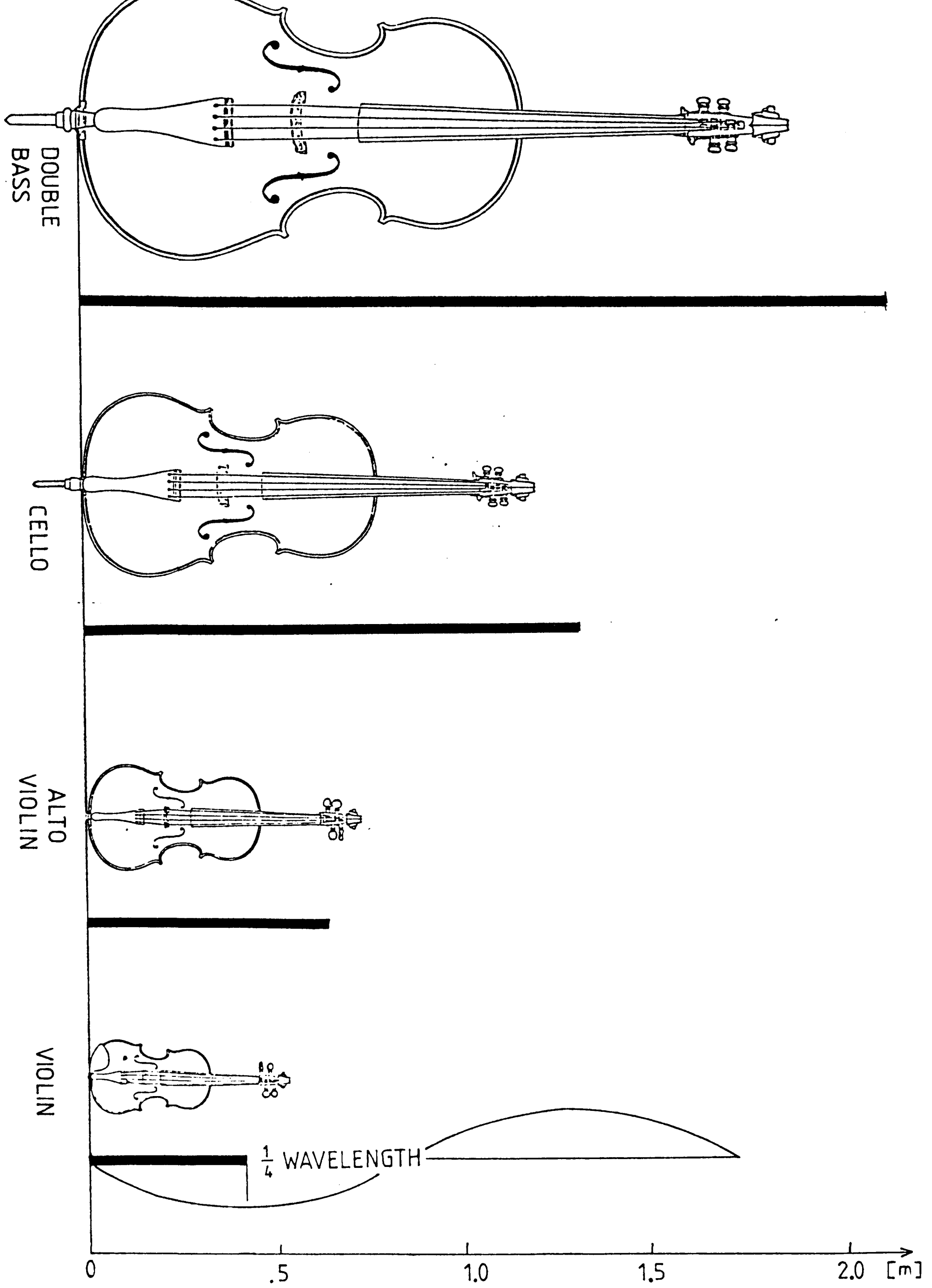

Figure 8: All bowed instruments are small compared with the wavelength of the sound in their lowest register. The figure illustrates the relation between body size and the longest wavelength the instruments can produce for the violin, alto violin, cello and double bass. The bars next to the instruments, correspond to a quarter of this wavelength, i.e. the wavelength of the fundamental of the lowest note on the instrument.

with the wavelength of the sound is poor at radiating the vibration energy as a sound wave." Applying the radiation impedance concept, we could rephrase this and say that the low radiation impedance of a small source makes an efficient sound radiation at low frequencies impossible. This fact sheds some light on a characteristic change in sound quality over the compass of the bowed instruments.

If we apply the above statement to a music instrument, the sound of which consists of many partials, it means that low partials with long wavelengths are less efficiently radiated than are the higher partials. As a consequence, when playing a descending scale on e.g. a violin, the amplitude of the low partials in the radiated sound decreases in comparison with the higher partials. In particular, the radiated fundamental rapidly gets weaker, the available vibration energy being trapped in the instrument body. This change in harmonic content of the radiated sound corresponds to a significant change in tone quality with pitch.

The awkward conditions for sound radiation in the low register of the bowed instruments are clearly illustrated in Fig. 8, which shows the relation between body size and the wavelength of the fundamental of the lowest note, for the violin, the viola, the cello and the double bass. The bodies of the instruments are much too small in comparison with the wavelength of the low fundamentals in order to be efficient sound radiators. This is especially true for the double bass and cello, which ought to have their body lengths increased by almost 50 % in order to compete with the violin.

As a consequence of the "small" body size, the radiated spectra for the lowest notes on all bowed instruments exhibit a very weak fundamental, typically 20 dB weaker than the second harmonic. This is illustrated in Fig. 9 which shows the spectrum for four notes in the lowest octave of the double bass. The figure also shows that the fundamental's amplitude rises with increasing fundamental frequency. Approximately one fifth above the lowest note, the fundamental reaches the same amplitude as the second partial, and an octave above the bottom note the fundamental is well above the other partials.

A strong fundamental is associated with a perceptually full tone quality. A wide open organ pipe, flute rank, is an example of a sound source in which the fundamental is always stronger than the other partials. When used in the low register, this rank gives a perceptually full bass sound. Accordingly, a reinforcement of the lowest harmonics in the cello and the double bass by stage vibrations would promote a full sound quality.

Stage risers can have such an effect on the radiated sound of the double bass, as illustrated by the spectra of some low notes in Fig. 10. The spectra show that the fundamental may gain as much as 10 dB on certain notes, and also that some dips in the spectra at higher frequencies are leveled, when the instrument is played on stage risers.

The perceptual implications of these spectral differences may be larger than expected, because of a peculiarity of our sense of hearing: a given increase in sound pressure at low frequencies raises the perceived loudness more than the same change would do at higher frequencies. This means that even small contributions to the amplitudes of the lowest harmonics are perceptually important.

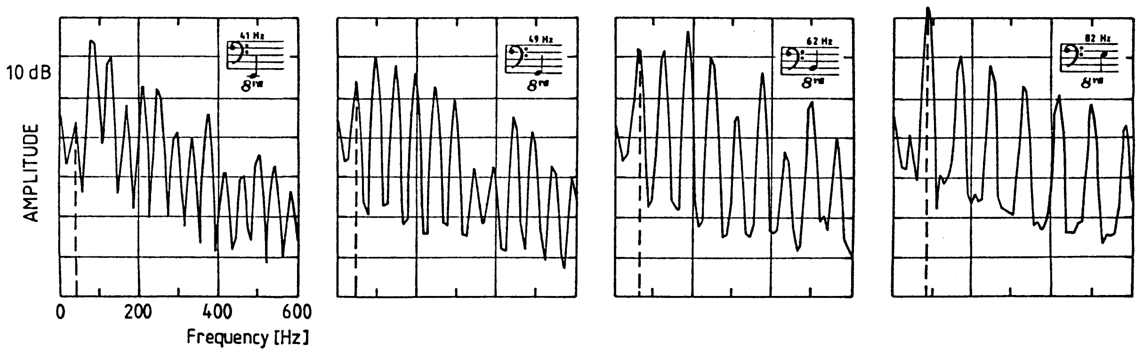

Figure 9: The bowed instruments produce a weak fundamental in their lowest register. In the figure, the spectra of four notes in the lowest octave of the double bass illustrate how the fundamental increases in strength with rising pitch (dashed lines).

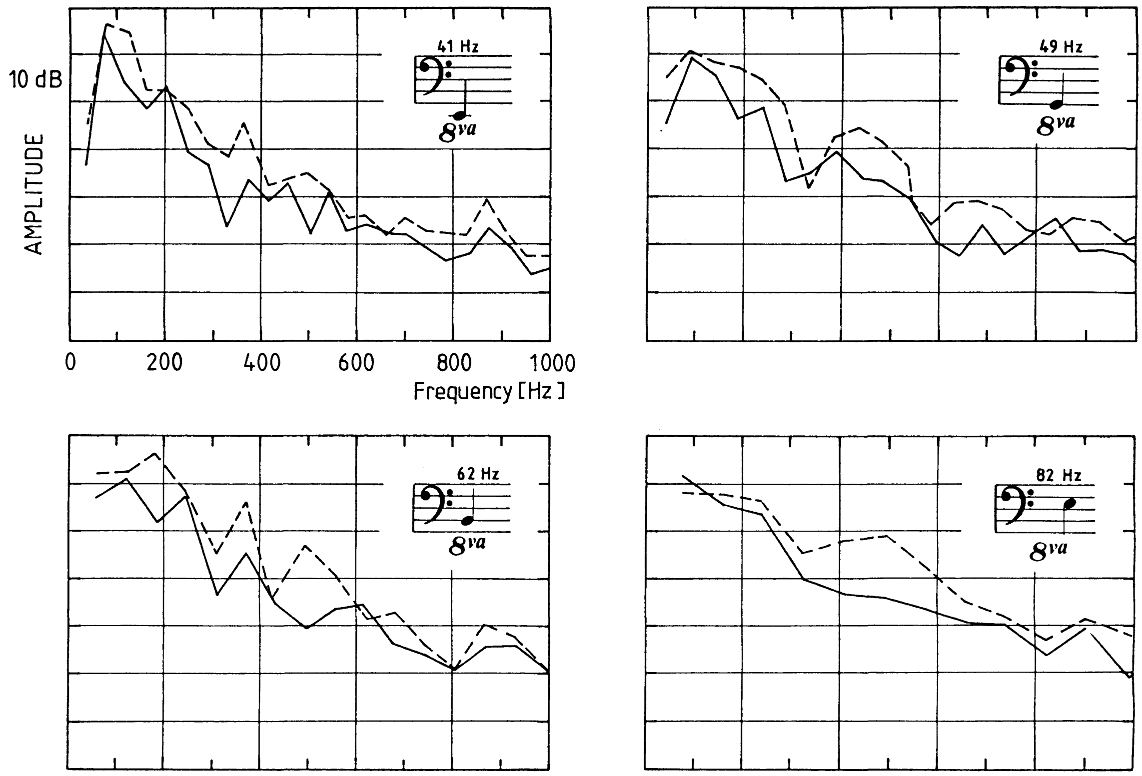

Figure 10: The amplitude of the fundamental and other low harmonics can be increased by supporting the double bass on a riser. The figure shows the spectrum contour of four notes in the lowest octave of the double bass, when played on a stiff stage floor (solid lines), compared with when played on a riser (dashed lines). The curves represent the median amplitude value of three repeats.

Sound radiation and position on stage

The sound radiation from the low instruments can be influenced by the position on the stage. Acoustic theory tells us that a sound source positioned close to a hard, reflecting surface will gain in sound radiation. Here, "close" means a distance that is a fraction of a wavelength of the sound. The primary reflecting surface is the floor, which is always close to the instrument. However, for the double bass section of the orchestra, the "close"-criterion is often also met by a wall, and sometimes even by two walls (a corner), as the wavelengths for the low harmonics in the low register are several meters.

The explicit acoustic explanation of this phenomenon tells that the radiation impedance as seen by the bass is increased due to a "mirroring" of the instrument in the reflecting walls (Waterhouse 1958, Weinreich 1985). This reinforcement of the low frequency range may be of the same magnitude as the observed acoustic support from risers.

What did we learn?

In summary, it seems clear that it is possible to increase the sound radiation from the double bass in perceptually important frequency ranges, by supporting the instruments on a suitable stage floor or riser. The maximum gain observed during measurements in a concert hall was about 3 dB around 50 Hz and 5 dB between 150-1000 Hz. A presumption is that the support is not too rigid, but is rickety enough to be set into vibration by the vibrating instrument.

The use of risers in music performance gives a combined effect due in part to the actual acoustic reinforcement from the risers and in part to a possible change in playing due to the responding support. The listener will not be able to distinguish between the contributions of the risers and of the players, and there is no need to do so. The point is that an arrangement with risers may improve the performance.

Apart from any acoustical benefit of risers, the importance of familiar stage floor vibrations for satisfactory performance is illustrated by some of today's prominent cello soloists (Cremer 1981). On tour, not only the instrument but also a small riser is taken round the world.

Acknowledgments

The author is indebted to double bass players Alf Petersén, Encho Radukanov and Robert Röjder for their kind participance in the experiments. The technical staff at the Berwald Hall is greatly acknowledged for their assistance. Valuable comments on the manuscript were given by Johan Sundberg, Erik Jansson and Johan Liljencrants.

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DO MUSICIANS OF THE SYMPHONY ORCHESTRA BECOME DEAF?

A review of the investigations of musicians in Gothenburg and Stockholm.

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Introduction

Do musicicans, and especially musicians of the symphony orchestra, sustain hearing impairments as a consequence of their work? Two extensive investigations of the hearing of musicians have been conducted in Sweden, one in Gothenburg (Axelsson & Lindgren, 1981) and one in Stockholm (Karlsson et al, 1983). In the first investigation it was found that more musicians than expected had a measurable hearing loss. In the second investigation it was found that the musicians as a group had normal hearing, i.e. their hearing thresholds were similar to those of people not exposed to excessive noise. The results seemed contradictive and it was therefore decided to review the data of the two investigations.

This review is intended to explain the results in terms understandable to the "non-specialist". A simple non-technical language has therefore been used. Note also that very little is known of how hearing deficiencies affect musical performance.

The review starts with short summaries of: (1) sound and hearing, (2) measuring the loss of hearing, (3) dangerous sounds, and (4) sound levels in the orchestra, before the measured hearing losses are presented (5) and discussed (6).

Sound and Hearing

An ear is built from several delicately assembled parts. Sound travels through the ear canal to the tympanic membrane, or ear drum. The vibrations of the tympanic membrane are coupled via the middle ear ossicles to the cochlea. There, the sound is transformed into electrical signals on the basilar membrane, which is arranged in a spiral inside the cochlea. The stapedius muscle,

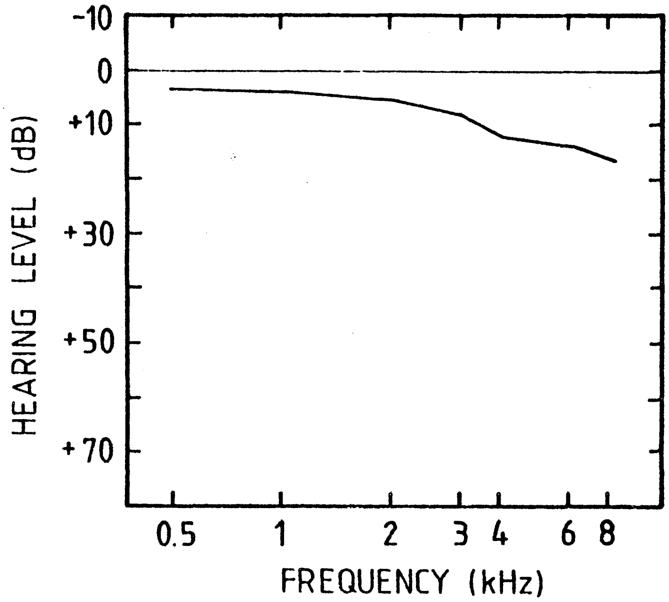


Figure 1. Pure-tone audiogram and hearing threshold level.

which is attached to the middle ear ossicles, protects the ear against too strong low-frequency sounds. Since it takes some time for this protective mechanism to react, it offers no protection against very sudden sounds, i.e. so called impulse sounds.

Different parts of the basilar membrane are sensitive to sounds of different frequencies. Hence strong sounds at one particular frequency may destroy the ear's function mostly at or near that frequency.

The ear is a very good microphone and amplifier. It has a working range of 120 dB at 1 kHz. The most quiet perceivable sound defines the so-called threshold of hearing. As the loudness increases, the sound becomes disturbing and eventually reaches an upper limit, which is called the threshold of pain.

Measuring the loss of hearing

Hearing is usually tested by recording pure-tone thresholds of hearing. The threshold levels are measured at specific frequencies (pitches), usually at 125, 250, 500, 1000, 3000, 4000, 6000 and 8000 Hz. Via headphones, a weak tone is presented to one ear. The strength, or level, of the tone is varied, and that level is noted at which the tone is just audible. This level is the threshold for this particular frequency and ear. The thresholds at the other frequencies are measured in the same way. The entire procedure is then repeated with the other ear. The measured hearing threshold levels, called hearing levels, are plotted in a so-called pure-tone audiogram.

In the pure-tone audiogram, the hearing threshold levels are plotted as measured at the different frequencies (Fig. 1). Along the horizontal axis, the frequency is given in Hz or kHz (kHz means thousands of Hz, i.e. 3 kHz equals 3 000 Hz). Along the vertical axis, we have the threshold level in dB (decibels). A threshold of zero dB means that the acuity of hearing equals the "statistically perfect" hearing level. If

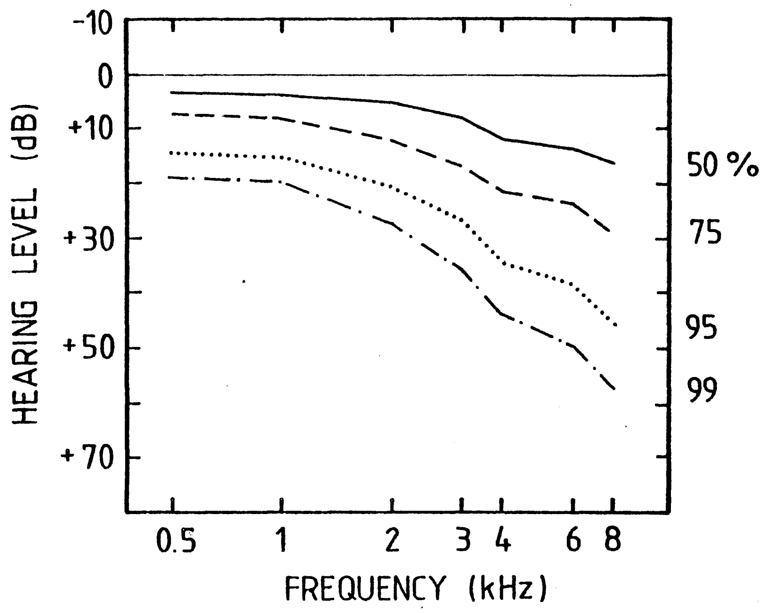


Figure 2. Hearing thresholds at different ages (median values from ISO/DISS 1999 Data Base A).

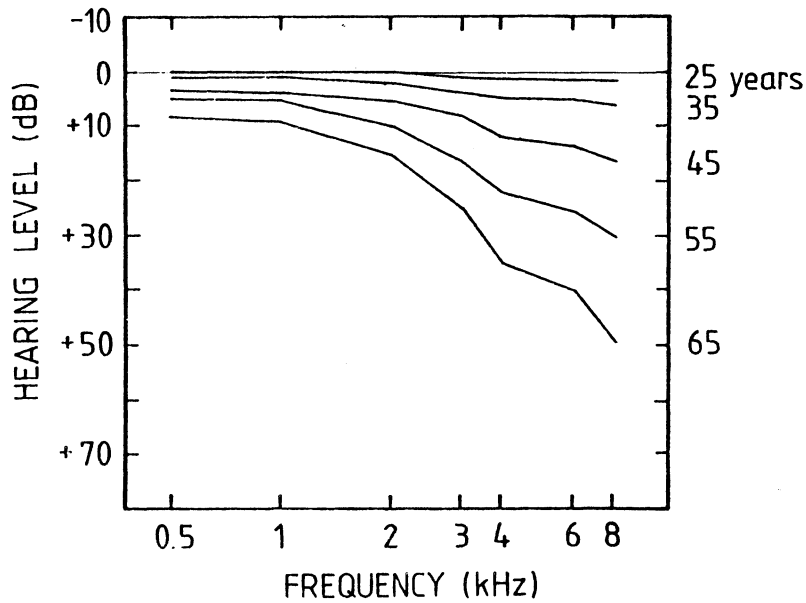


Figure 3. Distribution of hearing thresholds for 45 year old men (from ISO/DISS 1999 Data Base A).

the hearing threshold (the thick line) is below the zero level, there is a hearing loss at that ear. The magnitude of the threshold deterioration is given by the dB-value read from the vertical axis. The farther the threshold is below the zero level, the stronger a tone must be presented in order to be heard.

When determining whether a measured threshold is normal or not, we encounter two complications: 1) all humans have more or less different hearing thresholds, and 2) the hearing deteriorates with increasing age.

The differences in hearing between different people mean that a large group must be measured to obtain a reliable "average". Usually the 50 % level (the median value) is used as a measure of the "average", i.e. that threshold level which discriminates the "better" 50 % of the subjects from the "worse" 50 %.

Since every person is different, it is not sufficient to present the 50 % level (the median value) when a complete description of a group is desired. One must also know the distribution of thresholds above and below the median. An illustration of this is given in Fig. 2, which shows the distribution of hearing thresholds vary in a group of 45 year old men. The first line below the zero level is the 50 % level. Next line shows the 75 % level, i.e. the limit between 75 % better and 25 % worse hearing thresholds of the group. The lowest two lines similarly show the 95 % levels and the 99 % levels.

Hearing deteriorates with age. This is reflected in the audiogram by hearing thresholds that droop more and more below the zero level with rising age of the subjects (Fig. 3). The diagram shows the 50 % levels for a reference group not exposed to excessive noise. The hearing loss induced by advancing age is negligible (less than 2 dB) at age 25, and it is still small at 35 (less than 7 dB). From the age of 45 years, hearing deteriorates markedly; also, the hearing loss is greater at high frequencies, but remains moderate for frequencies below 2 kHz. Clearly, pure-tone audiograms can only be compared within the same age group, or after adjustments that compensate for age.

When comparing the pure-tone audiograms of two persons it is thus difficult to say what the cause is for the differences in hearing. The remaining differences after adjustments for age can derive from inherent physiological differences in hearing, from noise exposure, from disease, or from a combination of these factors.

The hearing threshold is affected in different ways by noise and by age (Fig. 4). It deteriorates slowly in youth and more rapidly at higher age. Excessive noise exposure results in rapidly worsening hearing thresholds in the early years of exposure, but more slowly at increased years of exposure. At high ages the impairment is predominantly due to age effects.

Dangerous sound

To evaluate the possible damaging effects of noise, a Swedish standard has been defined (SEN 590111, 1972). If the noise exceeds 85 dB(A) sound level, hearing protective action should be taken. If the noise level varies, an average, known as the equivalent sound level during a typical working week, should be measured and used. Occasional rare peaks are not considered dangerous, provided their levels are below 140 dB.

Sound levels in the symphony orchestra

The sound level in the symphony orchestra varies considerably (Fig. 5).

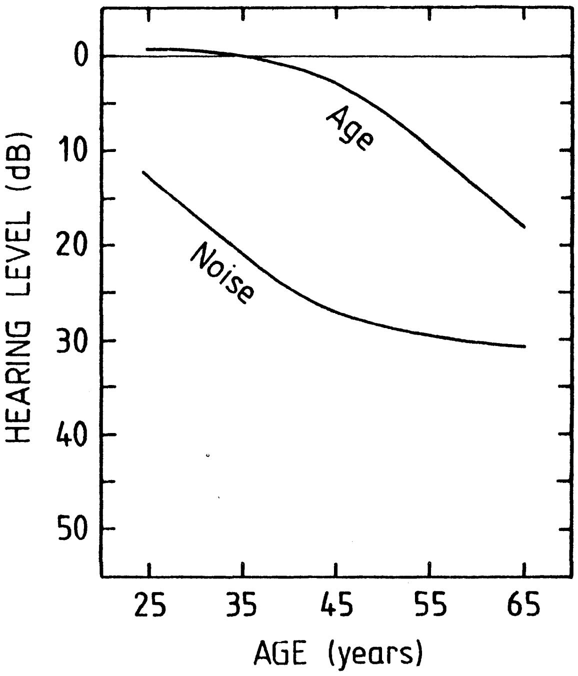


Figure 4. Example of the shift in hearing level from age and noise exposure (2000 Hz, adapted from Corso, 1980).

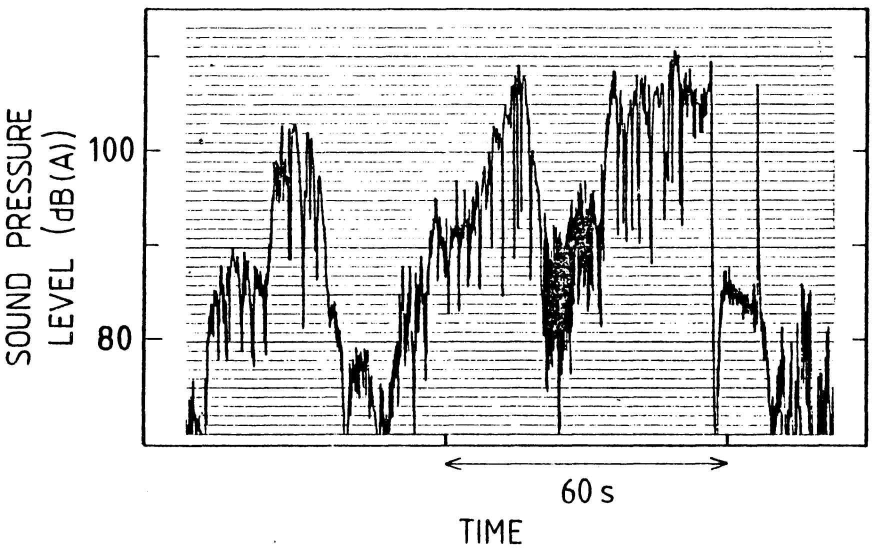
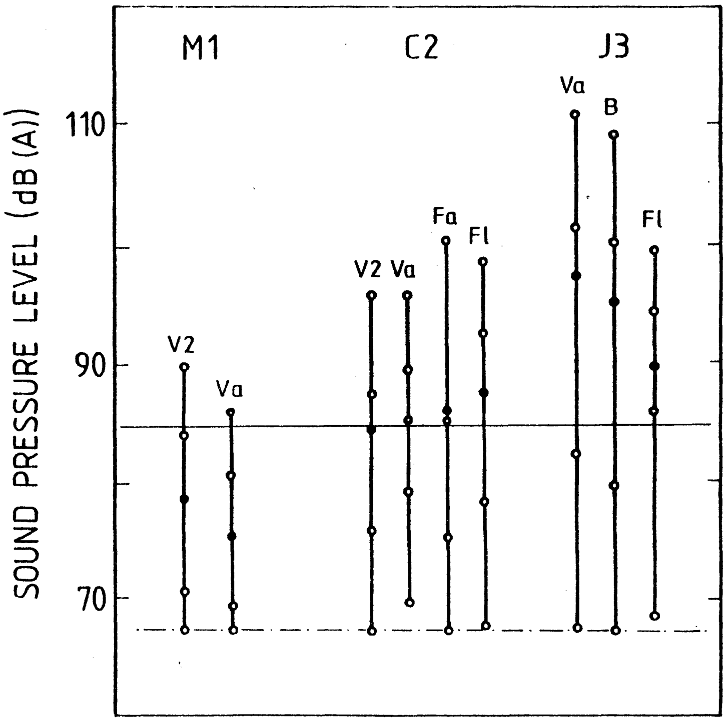


Figure 5. Example of the time course of the sound level in dB(A) for "heavy music".



ORCHESTRA SECTION AND MUSIC

Figure 6. Example of measured average sound levels during performance. Equivalent sound levels with filled circles.

The large variations mean that the abovementioned average, the so-called equivalent sound level, must be used. Results of such measurements are shown in Fig. 6. The sound level is represented on the vertical axis. The nine vertical bars mark nine different measurement conditions.

The two leftmost bars (M1) mark the levels from a performance of a Mozart violin concerto. The measuring microphones were placed in the second violin and the viola sections respectively. The filled circles mark the equivalent sound levels (the open circles represent, from top to bottom, the levels exceeded during 1, 10, 50 and 90 % of the time). In both the violin and the viola sections the equivalent sound levels are below the norm of 85 dB(A). The plotted levels are measured only during the actual performance of the piece. Silent work time such as pauses should be included, although it has little influence (if a performance is followed by a pause of equal duration the equivalent sound level decreases 3 dB).

The middle four bars (C2) show sound levels measured in the second violin, viola, bassoon and flute sections during the performance of a Chopin piano concerto. In these four cases the equivalent sound levels are at or above the 85 dB(A) norm.

The three rightmost bars (J3) represent sound levels measured in the viola, the double bass and flute sections during performance of Janaceks "Sinfonietta", i.e. "heavy" music. The equivalent sound levels are considerably above

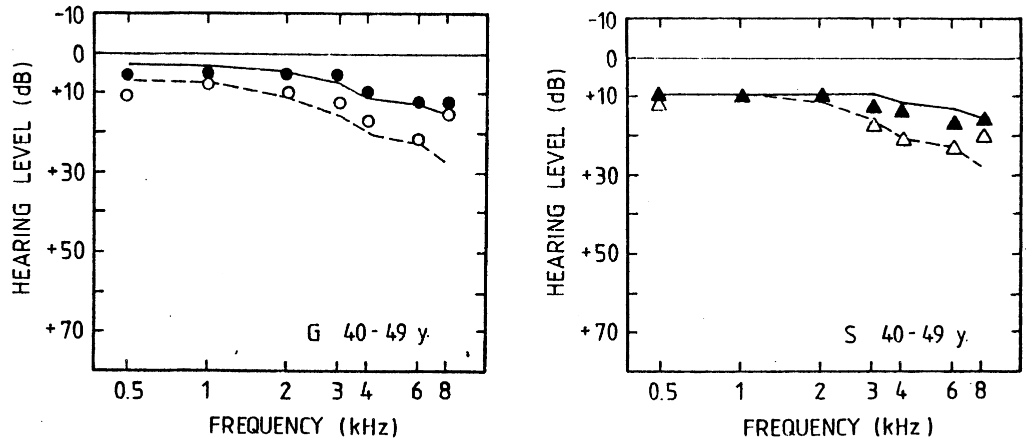


Figure 7. Example of measured hearing levels for the 40 - 49 year bracket. Filled circles and triagles mark meassured median levels and 75 % levels for the Gothenburg and Stockholm groups respectively. Unbroken and broken lines mark median and 75 % levels for 45 year old men not exposed to noise (from ISO/DISS 1999 Data Base A).

85 dB(A), and slightly more than 5 dB higher in the more exposed positions.

The results are typical. A moderately large ensemble performing a solo concerto produces equivalent sound levels of approximately 75 dB(A); the full symphony orchestra usually produces levels at 85dB(A). "Heavy" music gives levels of typically 90 - 95 dB(A) in "normal" and "exposed" positions respectively. To summarize, symphonic music gives levels around the 85 dB(A) level and sometimes louder. The sound levels within the orchestra show variations of 5 dB. In the Stockholm investigation, the peak levels did not exceed 125 dB.

Conclusion: Equivalent sound levels within a symphony orchestra are frequently close to or above 85 dB(A), a sound level which constitutes the upper limit allowed in industrial environments when ear protection is not used.

Hearing thresholds of the musicians

1) How do the hearing thresholds of musicians as a group compare to those of people not exposed to noise?

2) On an individual basis, is poor hearing more frequent among musicians than in a non-exposed population?

3) Do different sound levels in different sections of the orchestra result in different hearing thresholds?

or, somewhat rephrased: Is hearing endangered by work in the symphony orchestra, and if so, do those musicians which are exposed to the highest sound levels show the poorest hearing thresholds?

A. Hearing thresholds of the musicians as a group

In Fig. 7 the left panel shows the hearing thresholds of a group of 40-49 year old musicians in Gothenburg. The thresholds are displayed as median and 75 % levels. The filled circles are measured thresholds for the musicians,

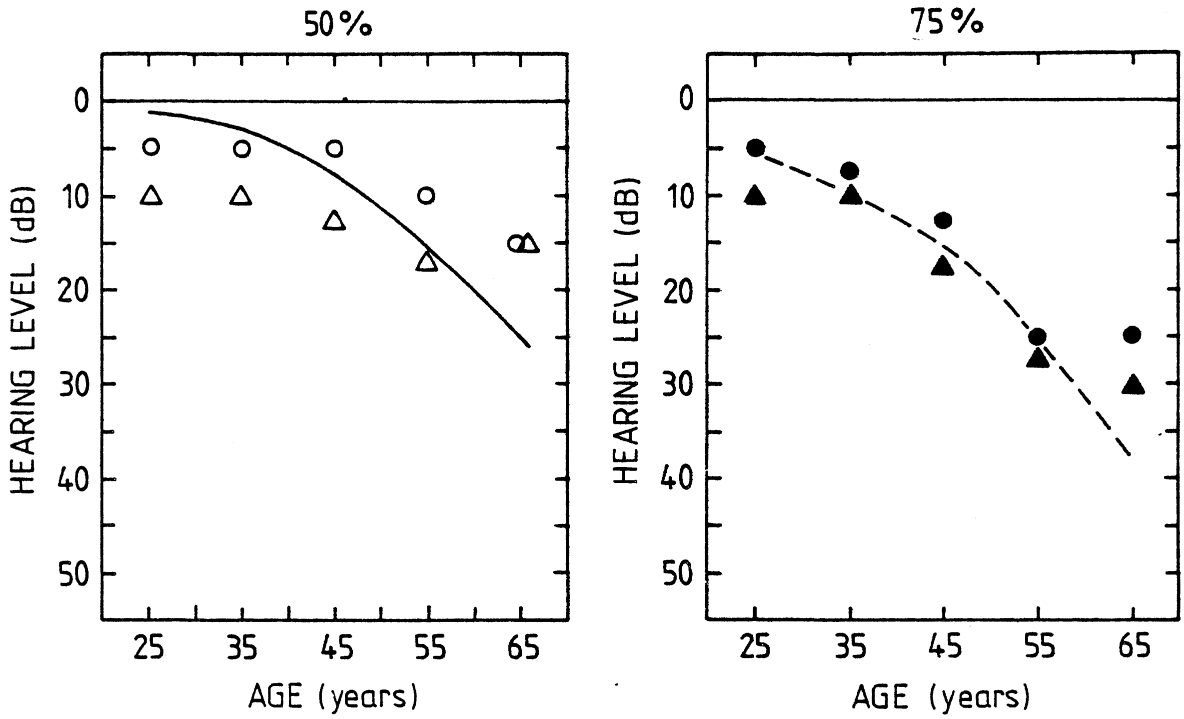


Figure 8. Measured hearing thresholds for all age groups displayed as in fig. 7.

and the lines show the median and 75% percentile in a corresponding reference group with "non-impaired" hearing (the reference levels are those of 45 year old men, a weighting of 40-49 year old men would have given slightly worse reference levels). The thresholds of the musicians are close to those of the reference group - somewhat worse at low frequencies but somewhat better at high frequencies. The open circles and the dashed lines show the 75 % levels. The hearing thresholds of the musicians are better than those of the reference group except at the lowest frequencies.

The right-hand panel of Fig. 7 shows the hearing thresholds of the 40-49 year old musicians in the Stockholm investigation. Filled triangles mark the median levels of the musicians, and the lines those of the reference group. Open triangles and dashed lines represent the 75 % levels. In Stockholm a screening level of 10 dB HL was used, i.e. hearing thresholds of 10 dB and better are measured and marked as 10 dB. The hearing thresholds of the reference group have been plotted at 10 dB in such cases. The hearing thresholds of the musicians is somewhat worse at 3, 4 and 6 kHz but otherwise similar to those of the reference group.

Conclusion: Musicians in the 40-49 year age bracket show normal hearing thresholds as a group.

Hearing at different ages

The results for each age group are shown in Fig. 8 as in Fig. 7, with the results for Gothenburg in the left panel and for Stockholm in the right panel. The diagrams for the age groups 20-29

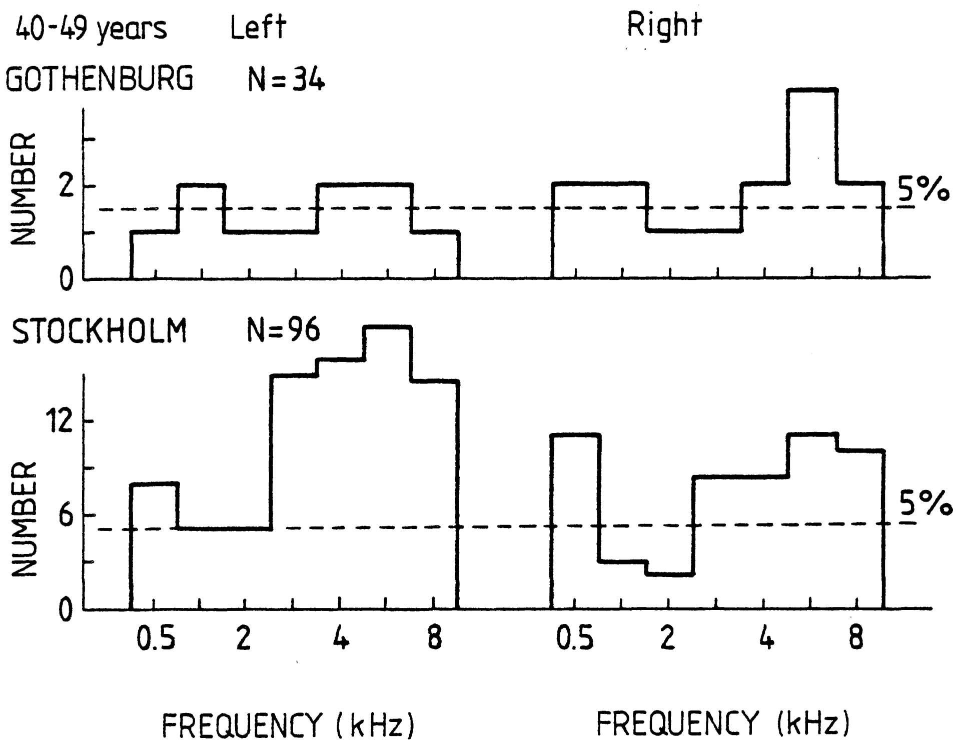


Figure 9. Example of measured deterioration of hearing level at 3 kHz with increasing age.

and 30-39 show hearing thresholds close to normal. The diagrams for the age groups 50-59 and 60-69 show hearing thresholds better than corresponding values for a non-exposed population. The hearing thresholds for the musicians considered as group must consequently be regarded as normal.

Age and noise both have an adverse effect on hearing, particularly at high frequencies. Measured thresholds at 3 kHz for different age groups are plotted in Fig. 9. To the left, values are plotted corresponding to the median and to the right corresponding to the 75 % levels. It can be seen that the measured values for the musicians agree fairly well with the reference values. The only exception is the median value for 65 years of age; those musicians have better hearing than the reference group. The results support the interpretation that hearing of musicians as a group has not been affected by the "musical noise".

Conclusion: The musicians as a group show normal hearing thresholds at all ages.

B. Hearing thresholds of individual musicians

On an individual basis, do more musicians show poor hearing thresholds than people not exposed to noise? One way of testing this is to count the number of musicians' ears that have worse hearing thresholds than a specified, "acceptable"

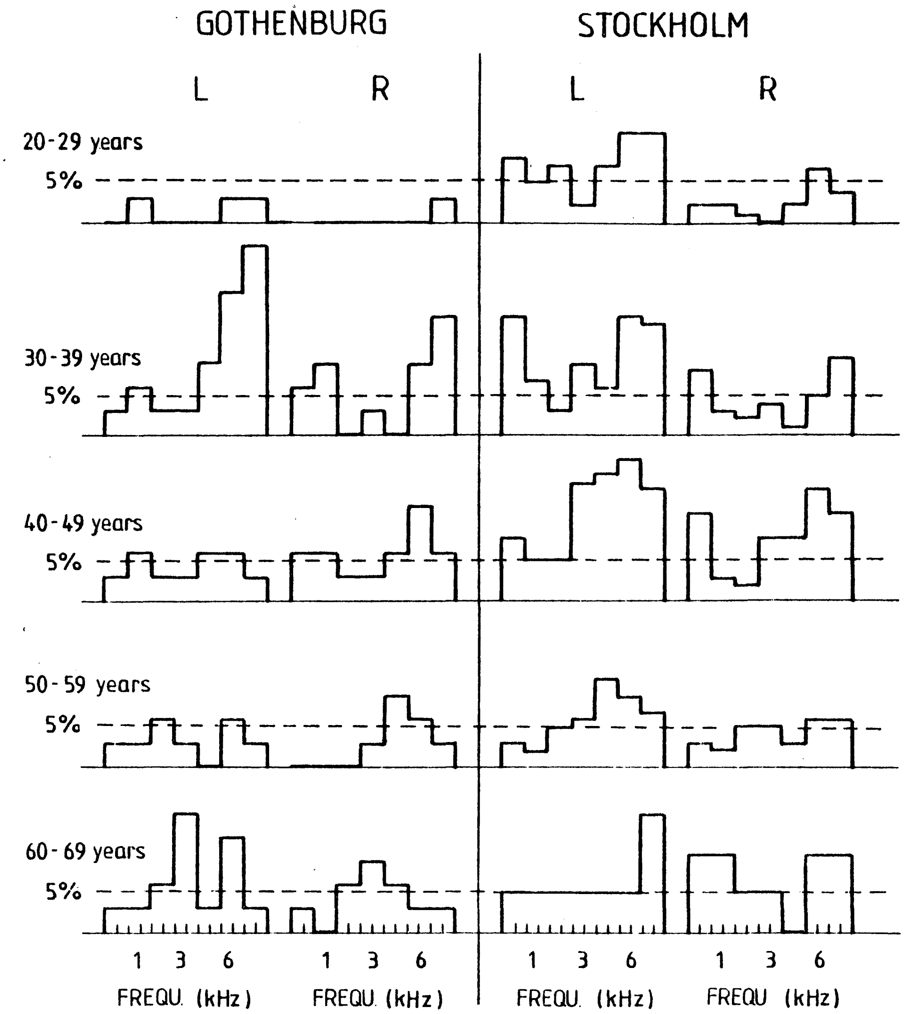


Figure 10. Example of measured number of ears with "nonacceptable" hearing thresholds for the 40 to 49 year bracket.

level. We arbitrarily defined "acceptable" as the hearing threshold level which is exceeded by only 5 % of a normal population (P95); this is like postulating that normally, one person in twenty has an "unacceptable" hearing impairment.

In Fig. 10 the results for 40-49 year old musicians have been summarized; the figures for Gothenburg are shown in the upper panel and for Stockholm in the lower panel. The two diagrams to the left show the left ears and the two diagrams to the right show the right ears. The figure should be interpreted as follows. The vertical bars denote the percentages of measured ears with thresholds worse than P95. The dashed line demarcates 5 % of the number of ears measured, i.e. the expected percentage of ears with threshold P95 in a normal population. If more than 5 % of the

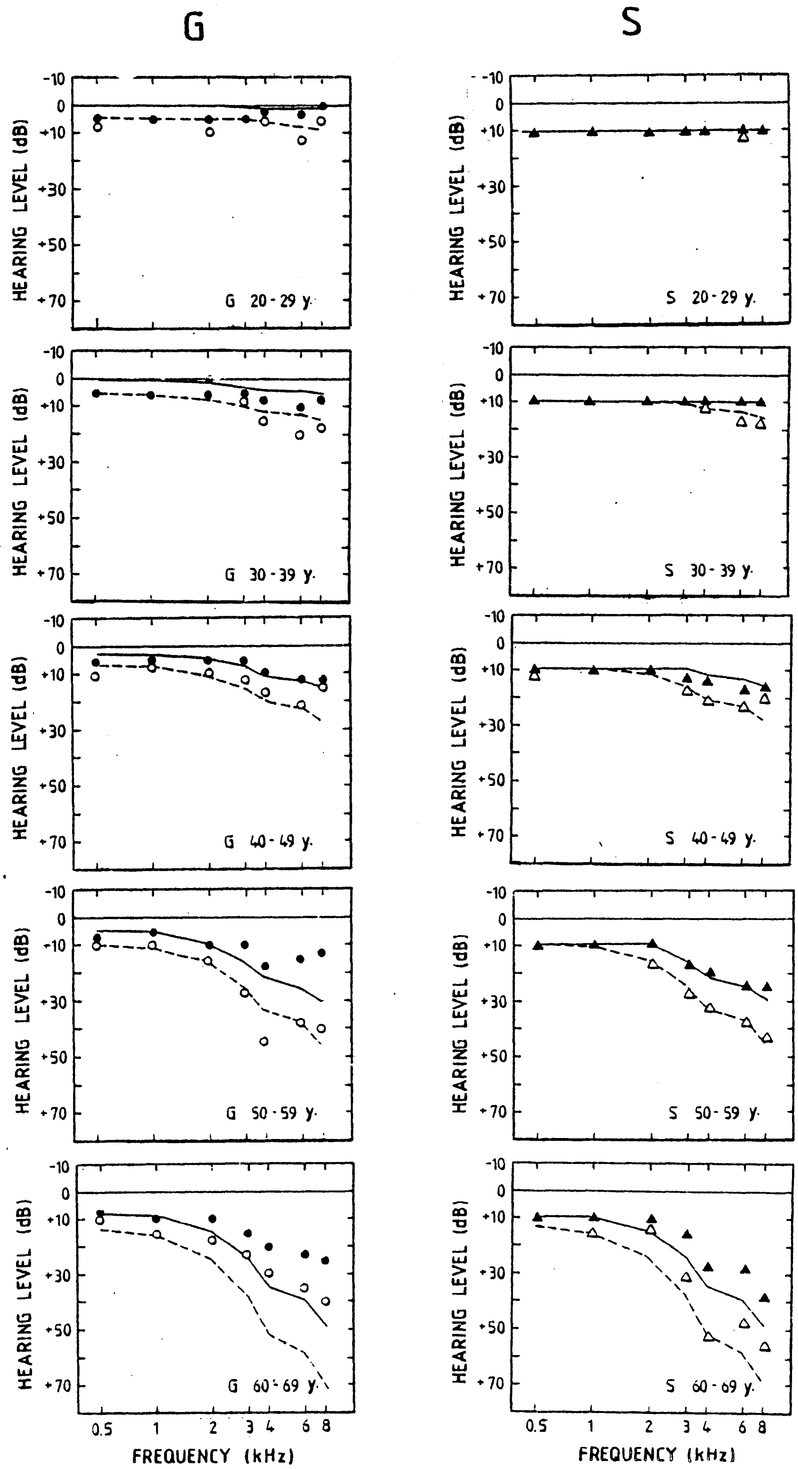


Figure 11. Measured number of ears with "nonacceptable" hearing thresholds for all age groups.

musicians' ears have hearing worse than P95, the bars rise above the broken line.

The number of ears in this age-group in the Gothenburg material is small, i.e. 34 left and right ears. This means that two ears constitute more than 5 % of the total, while one ear is less. Slightly more than 5 % of the musicians' ears have thresholds worse than P95.

The number of ears in the Stockholm material is somewhat larger, with about five ears corresponding to 5 %. More representative results can thus be expected from this material. P95 is exceeded by the musicians' ears at 0.5kHz and at 3-8 kHz.

The results for all age groups are summarized in Fig. 11. More than the expected normal of 5 % of the musicians' ears show poor hearing thresholds, particularly at high frequencies and for the left ear.

Conclusion: On an individual basis, more musicians have poor hearing thresholds than a population not exposed to noise.

C. Hearing thresholds for different sections of the orchestra

Do different sound levels in different sections of the orchestra result in different hearing thresholds? Sound level measurements within the orchestra showed 5 dB higher levels in "exposed" than in "normal" positions. It is difficult, however, to make a clear distinction between "exposed" and "normal" positions. Qualitatively, the violins can be considered to be in "normal" positions in relation to the surrounding instruments. Within and close to the wind and percussion sections, one would expect to find "exposed" musicians. A complication is that these sections often participate in only the louder portions of the music. In addition, individual musicians move within the orchestra and its sections. There are probably only a few musicians who are exposed to permanent or long-standing high sound levels.

In Gothenburg the different sections rank in order (from the best to the worst hearing thresholds), 1) percussion, 2) strings, 3) woodwind and 4) brass, i.e. as expected, except for the percussion.

If we look at the different string sections, we find that double bass players had the best hearing thresholds and the violins the worst in Gothenburg. In Stockholm the result is the opposite; bass players had the worst and violin players the best hearing thresholds (Karlsson et al, 1983, fig. 7).

Within the woodwind section in Gothenburg the worst hearing thresholds were found for the bassoon sections, followed by oboe, clarinet and flute sections. In Stockholm the bassoon and the flute sections have the worst hearing thresholds and the clarinet and the oboe sections the best. There is thus agreement in three out of four cases.

Within the brass section the hearing thresholds are worst within the french horn section and "fairly bad" for trumpets and trombones in Gothenburg. In Stockholm the hearing thresholds are the worst for the trumpets, normal for french horns and with no remarks for trombones. It is thus doubtful if there is any agreement between brass sections of the two investigations.

Conclusion: It is difficult to draw definite conclusions from the two investigations as to whether some instruments are more dangerous to play than others.

Summary

Hearing is jeopardized by long-standing exposure to high sound levels. High

sound levels over an extended time period but also short and very strong sounds may be dangerous. Sustained hearing loss is measured as shifts in hearing thresholds. The high frequency properties of the hearing are influenced earliest and the most. The two reviewed investigations show:

1) The average sound levels (equivalent sound levels) within the orchestra are close to the limits for noise allowed in industrial environments. For "heavy" music in "exposed" positions there is approximately a 5 dB higher level than in "normal" positions.

2) Hearing tests show that at least 75 % of the musicians have normal hearing thresholds. Among the remaining 25 % there are tendencies to worse hearing thresholds. The hearing thresholds of musicians in different sections do not indicate that it is more harmful to work in some sections than in others.

The hearing of the majority of the musicians had thus not been measurably impaired. On the other hand, musicians as a group have done less military service, may be more cautious with noisy sounds, and are well motivated test subjects. We may therefore expect them to have better than normal hearing thresholds.

It would be interesting to analyze further the hearing of a population as large as possible, i.e. to combine the results of Gothenburg and Stockholm; and in particular, to investigate in greater detail the 25 % of the musicians with the worst hearing thresholds, as well as the potential danger of "exposed" sections. Comparisons should also be made between a non-exposed population and the combined musician population. In Gothenburg it was found that 39% had a hearing loss, which seems a large number; however, the corresponding percentage must be known for a reference group of a non-exposed population, before conclusions can be drawn.

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SOUND EXAMPLES

Side A

Track:

A.C. Gade: **Acoustics of the Orchestra Platform**

1. One of the subjects' tasks was to discriminate between conditions possessing or lacking artificial reverberation. Here a flautist and a cellist both play a short motif twice, and are asked to judge in which case the reverberation was present. **1**

2. Two versions of a trio excerpt, with maximum or no reverberation (see text). The music is J.S. Bach's Trio Sonata No. 2, second movement, bars 1–33. **2**

3. The effect of delay due to sound propagation over some distance was studied by having two musicians in separate rooms play duos with each other, while the transmission between rooms was artificially delayed in both directions. Here a cello/violin duo and then a flute/violin duo play their parts from the orchestral score of Mozart's Symphony No. 40. **3**

S. Ternström and J. Sundberg: **Acoustics of Choir Singing**

Examples of tuning a formant to the lowest common partial. A male choir sings sustained vowels in unison. A sinusoidal tone is presented before each vowel, to direct the listener's attention to the partial in question.

1. The third partial lies an octave plus a fifth above the fundamental; it is the lowest common partial of the "fifth" interval. First a tone is sung with a vowel in which this partial is enhanced by a formant, then the tone is sung on a vowel which suppresses the third partial. The sequence is repeated once. The subjects sang in fifths above these tones and were more agreed on the size of the fifth with the first vowel. These sounds have the spectra depicted in Figure 6. Note that partial number six, which is the second common partial, also is stronger than its neighbors. **4**

2. The same demonstration as above, but for the major third interval. Here, partial number five, which sounds like a major third, is first enhanced by the sung vowel, and then suppressed. **5**

Side B

Track:

A. Askenfelt: **Stage floors and risers – supporting resonant bodies or sound traps?**

1. Demonstration of how different frequency ranges contribute to the perceived sound quality of a double bass. A short bass etude is low- and high-pass filtered and compared to the original sound (original - low-pass - high-pass - original). The example is repeated with successively lower cut-off frequencies of the filters: 1000 - 500 - 200 - 125 Hz. 1

2. Demonstration of the differences between "press" and "flow" in bowing. 2

3. Recordings of vibrations picked up by an accelerometer in different positions. The experimenter's voice is heard via the vibrations in the riser and instrument body respectively. 3

   a) Vibrations in a riser, close to the end pin of an "active" (played) bass.

   b) Vibrations in the stiff stage floor close to the end pin of an "active" bass.

   c) Vibrations in the bridge of a "passive" bass, excited by a played bass on the same riser.

E. Jansson, A. Axelsson, F. Lindgren, K. Karlsson and T. Olaussen:
Do Musicians of the Symphony Orchestra Become Deaf?

Examples of sounds recorded inside an orchestra, one meter in front of the trumpets. The high level of tape hiss in softer passages results from setting the recording level so as to cope with peaks in the orchestral sound. In commercial recordings, the dynamics are compressed somewhat, by electronic means, to reduce the very large changes in sound level. Note that typical home stereo equipment is not capable of reproducing the loudest of these sounds at a realistic level. The music is from the ballet "Manon" by Massenet.

1. Soft to average sounds (piano to mezzoforte, trumpets silent). 4

2. Mezzoforte, with trumpets. 5

3. Very strong trumpet sounds and percussion (forte fortissimo). 6