BRÜEL & KJÆR
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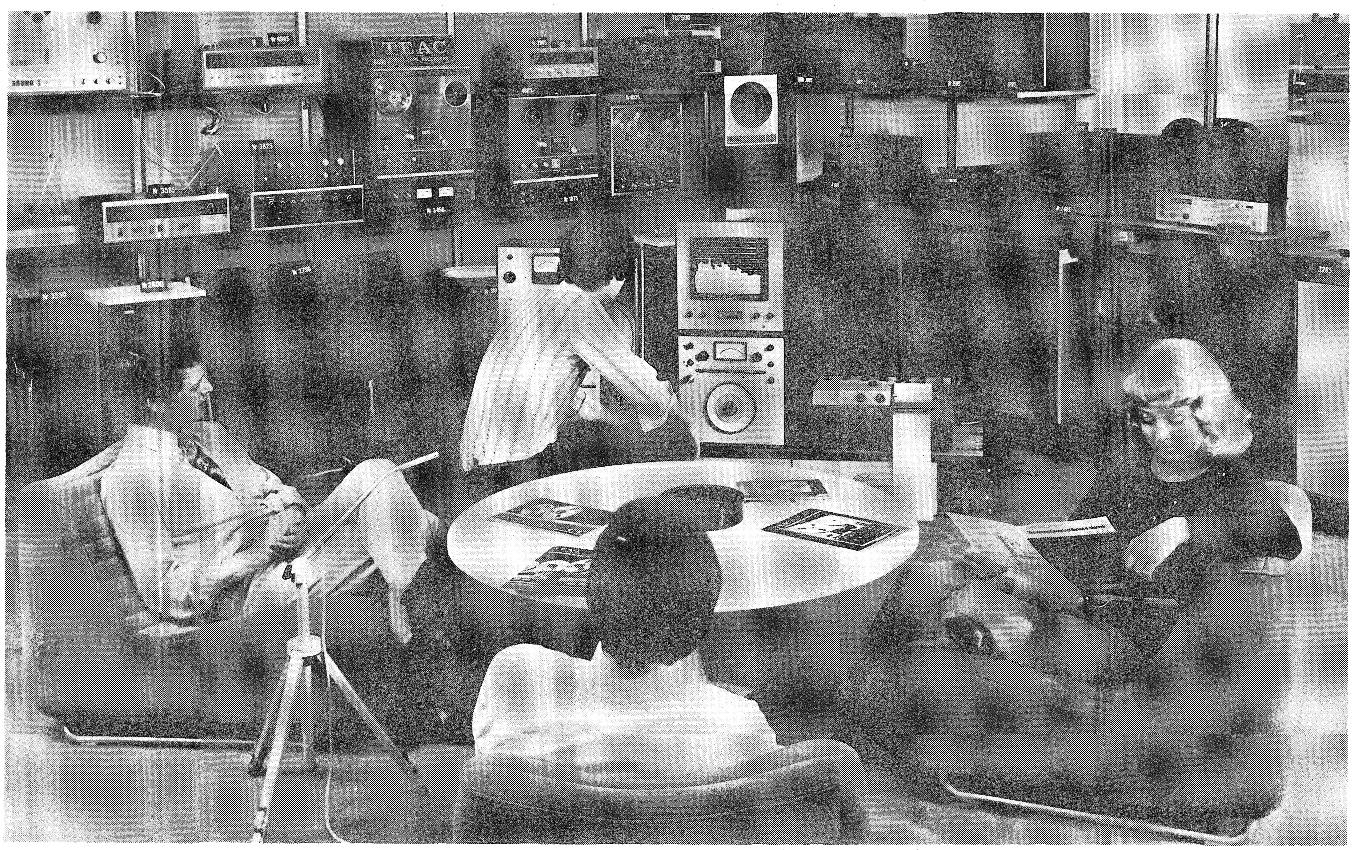
BRÜEL & KJÆR application notes

Hi-Fi Tests



— with 1/3 octave, pink weighted, random noise

Hi-Fi tests ought to be carried out at the exact location where the Hi-Fi set is to be used or under closely approximating conditions

By Henning Møller, Brüel & Kjær

Abstract

The "sound" of a Hi-Fi set is to a great extent room dependent. Too often, the final result is decided by the room rather than by the actual equipment.

The objective test method, which seems to correspond best with subjective judgement, is the "1/3 octave pink-weighted random noise method" carried out in the listening room. From measurements made according to this method, it is possible to say which set, that in the actual room, will give the best result. In practice, the method can be carried out in many ways. It can be made portable, simple and inexpensive. It can be made more accurate and fast, and if a computer is available, it can, in one operation, give the complete transfer function i.e. amplitude and phase response.

This application note will try to describe these possibilities, and it will show some results of the method and compare these with results made by listening tests. 5 loudspeakers, in 3 different rooms, and a test team consisting of 5 critical listeners were used.

Introduction

Too often, it happens that people, after listening to different Hi-Fi sets in the dealers showroom, and often after having spent a great deal of money, find that the Hi-Fi set which they have finally decided upon turns out to be the wrong one — when they hear it at home they are immediately dissatisfied. The music does not sound quite as good as it did at the dealers. The reason is often that the actual listening room has not been taken into account.

It is well known that the output voltage of an electric circuit, for instance, is very dependent upon the actual loading. In the same way, an acoustic system is dependent upon the acoustic loading — i.e. the listening room. Thus, a measurement of the complete system must be made under the normal acoustic working conditions, and of course measurements must not alter the listening conditions.

If a manufacturer produces loudspeakers to suit standard measurements in anechoic chambers, he can sell an ideal transducer and let the customer change his room accordingly. He can make it a little easier for the customer, if he builds in a correction network, but the optimal result can normally first be reached after a measurement of the actual listening room. In fact, it can be very difficult, just by ear, to decide exactly what is wrong. If, for instance, there is a resonance between 100 and 200 Hz, one could say, that there is something wrong in the bass, but not exactly where it is wrong. Normally neither loudspeakers nor rooms are ideal, and therefore the problem is often to find a reasonable combination. Usually, this can be achieved by selecting the best suited equipment rather than by paying a higher price.

Lately, a great deal of investigations have been made to find suitable measuring methods in listening rooms. In Denmark, the so-called "Højttalerundersøgelse" (loudspeaker investigation, Ref. 1) has been made in order to try to find connections between objective measurements and subjective judgements. The objective measurement that seems to correspond best with results from listening tests, was as mentioned the "1/3 octave pink weighted random noise method" carried out in the listening room. Also, phase measurements and power characteristics seem to correspond reasonably well, but less significantly.

Depending on how exact and how fast one wishes to measure, and of course on how much one needs to know, one can select from the following methods.

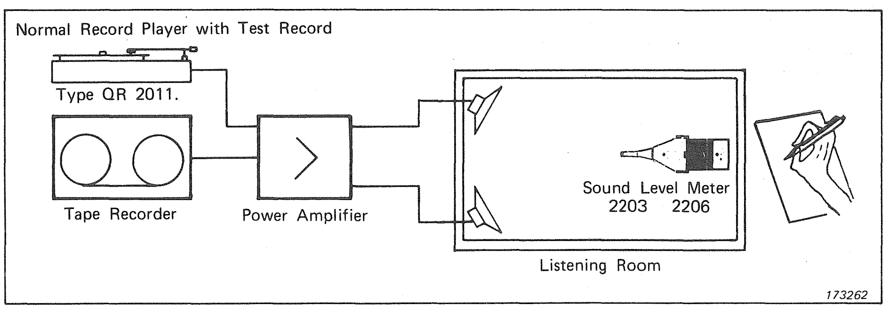


Fig.1a. The portable and inexpensive method. The recording is manual with one point for each 1/3 octave

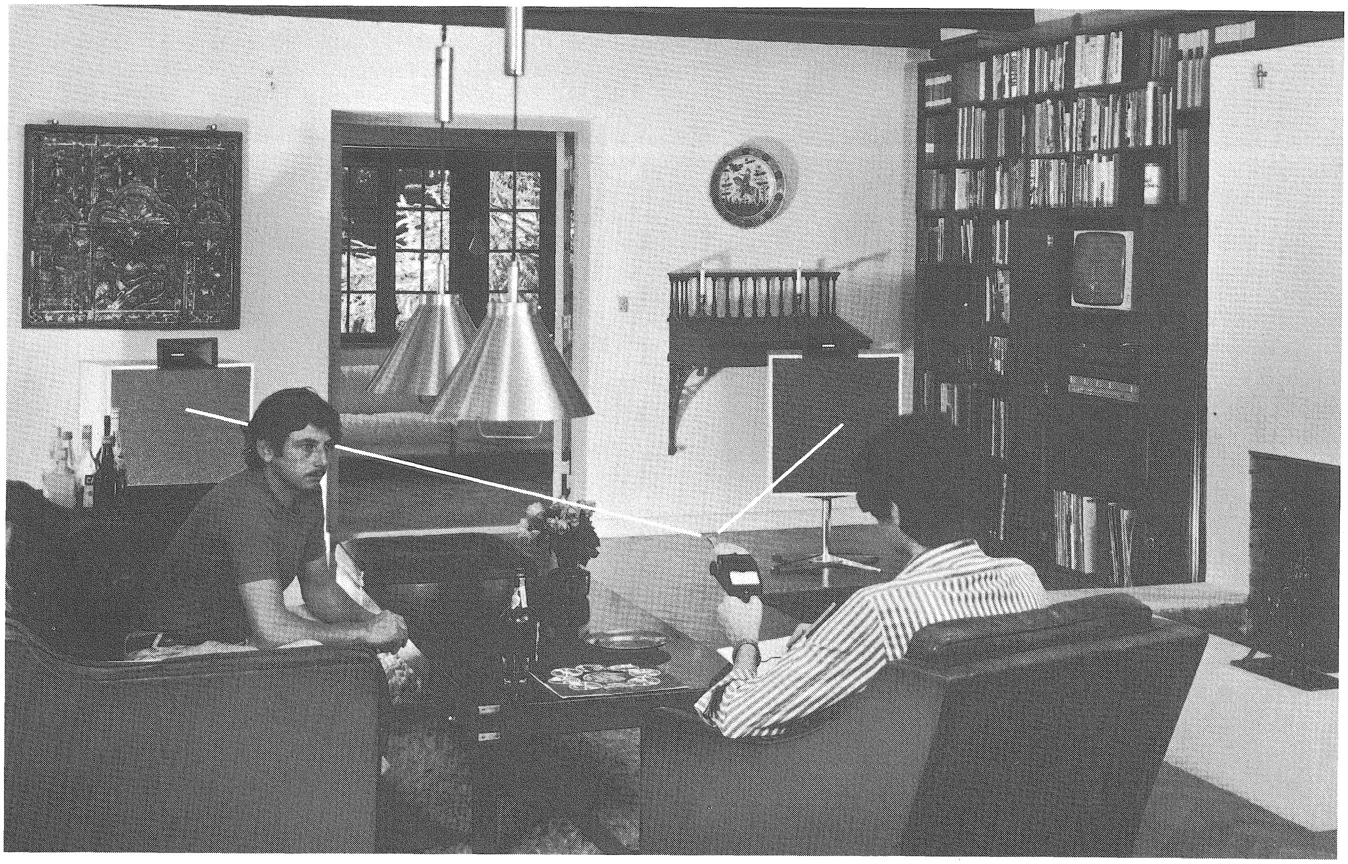


Fig. 1b. The portable and inexpensive method in practice

The portable and inexpensive method

The Hi-Fi enthusiast and the small dealer of course require a simple, but reasonably accurate method. It must be portable and cheap. Actually it is possible to fulfil these requirements, so that one can test the Hi-Fi set in ones own living room without too much trouble. The test method is shown schematically in Fig.1a, and in practice in Fig.1b.

A prerecorded tape on a normal tape recorder, or a Test Record (Type QR 2011) played on a normal record player is used. The test signal will then go through the Hi-Fi set and out into the listening room under exactly the same conditions as a normal commercial recording would have. Measurement is made with a Sound Level Meter (Type 2203/2206) placed exactly at the position where the listener would normally sit. The Test Record, or tape, delivers pink weighted bandpass noise in 1/3 octaves, one at a time. Each noise band (Bandwidth 1/3 octave) will cause a certain deflection on the Sound Level Meter according to the actual sound level at that frequency. The only requirement of the operator is to note this level and record it on preprinted paper.

Results from this method are shown in Fig.11.

The purpose of using bandpass noise instead of the traditional sine signal is that averaging of the received signal is automatically achieved. A sine sweep will excite many local resonances, which will have almost no connection with the normal music signals.

Some results of the method are shown at the end of this application note, where they are compared with results from listening tests.

In general, the best loudspeaker has the most flat response in the actual listening room, and the loudspeaker, that wins in the listening tests, is also the one that has the "flattest" curve. Of course in another room another loudspeaker might be the best.

This is valid when the music is recorded in a reverberant field. Often however, recordings are made with a combination of reverberant — and near-field information in a multichannel mixer, or they may only consist of mixed near-field information. The sound of the recorded signal is, in that case, slightly different from the sound received at the acoustically best position of the original concert hall, where the recording took place. The reason for this is that the reverberant field contains the information of the acoustics in the concert hall, where a rolloff at high frequency and a slight boost at low frequency is quite normal. The near-field does not have this information, and therefore compensation has to be introduced to reproduction equipment when the recording has been made in the near-field.

On an ideal Hi-Fi set, this means a set with a possible linear response in the listening room, the compensation is (Ref.2 p 458) on average + 5 dB at 60 Hz and —9 dB at 10 kHz with smooth slopes. As mentioned earlier most recordings are a combination of direct and reverberant recordings, therefore a compensation from the linear acoustic response of about + 3 dB at 60 Hz and —3 dB at 10 kHz seems to be a good compromise for most recordings.

In this connection, it seems reasonable to mention, that the response in the listening room does not necessarily disclose everything there is to know about the system. Alone it is not a pure scientific truth. The picture **might** be disturbed by the socalled non-minimum phase behaviour (Ref.3) and possibly by other unknown phenomena. This however, has normally less influence on the final result, and will therefore not be dealt with further right here. A supporting measurement might be phase (see later text).

The idea behind "The portable and inexpensive method" is to let the test tape, or the test record, playback generator signals which might be used professionally. The principle will become more clear when we now take a closer look at the more professional methods. Possible setups are shown Fig.2 and Fig.3.

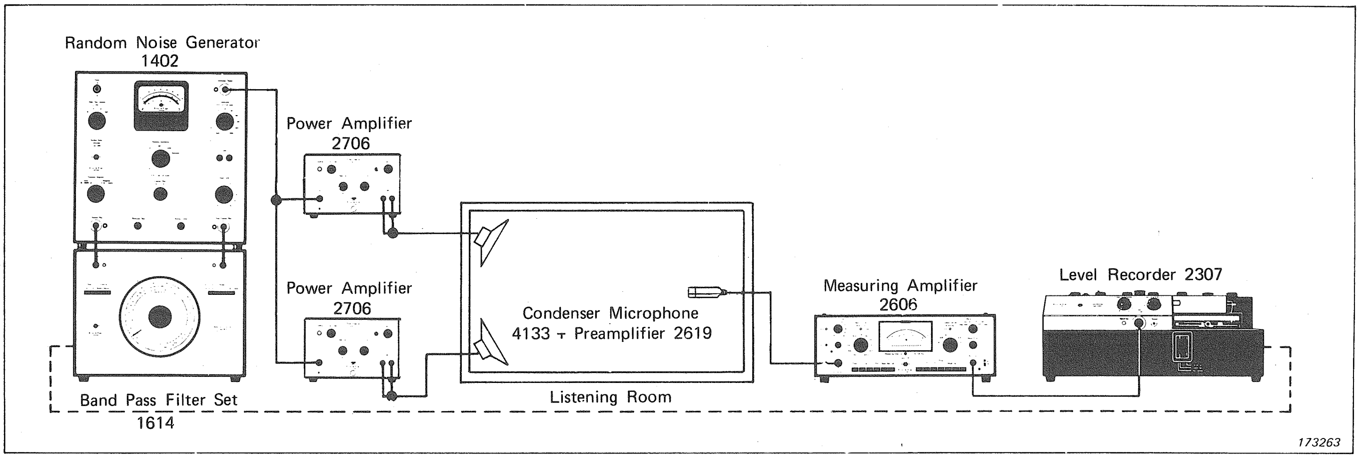


Fig.2. Set-up for "1/3 octave, pink weighted, random noise method"

More professional methods

If increased accuracy and speed are required one must utilize more sophisticated equipment. The professional methods seem relevant in cinemas, theaters, concert halls and especially in recording- and radio studios.

When a recording or a direct radiobroadcast is made, the sound picture in the control room is of great importance. The producer and the musicians usually mix the signals, until it sounds good in the control room — so this particular sound picture really ought to be correct.

The radio dealer who wishes to do things professionally can of course use these methods too.

In Fig.2, a Noise Generator (Type 1402) is used. It produces "pink" noise ("white" noise, weighted —3 dB/octave). The reason for using "pink" noise is that the following filter (Type 1614) has constant percentage bandwidth in 1/3 octaves and the center frequencies are changed in steps of 1/3 octave. the absolute bandwidth and therefore the received power will increase linearly with frequency. Now the voltage of "white" noise is proportional to the square root of the bandwidth, so the necessary correction will be —10 dB/decade or —3 dB/octave.

The generated noise band is fed to two Power Amplifiers (Type 2706) and further to the loudspeakers. The measurement is made linearly by a Condenser Microphone (Type 4133) at the position, which will normally be occupied by the listener and the received signal is finally read-out on the Level Recorder (Type 2307), which also delivers pulses to the

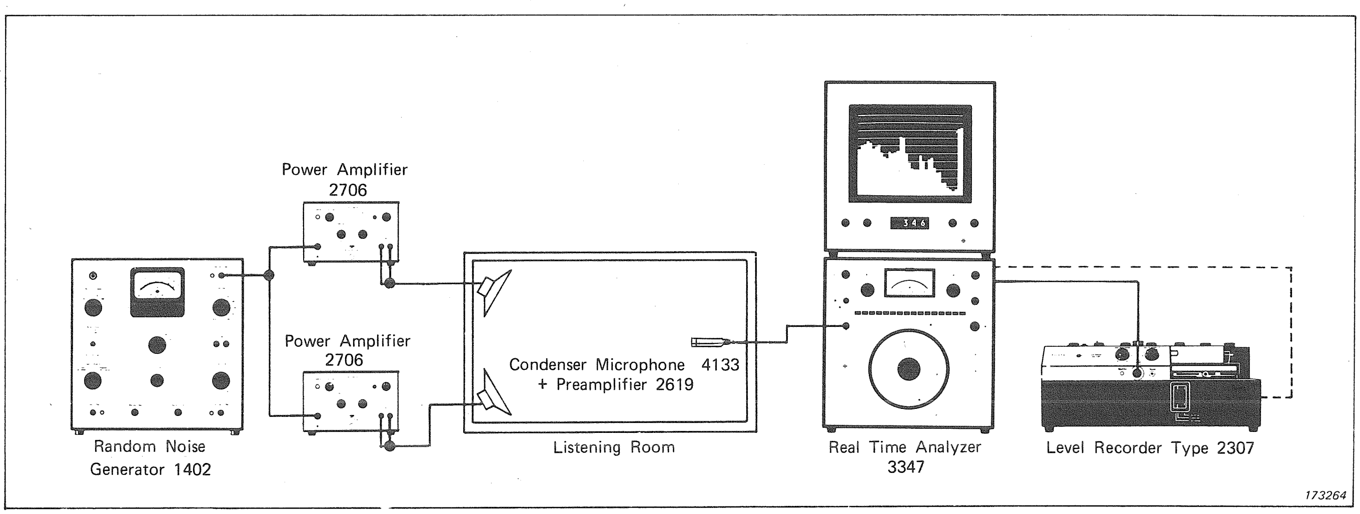


Fig.3. Measuring set-up with "pink" noise 20 Hz — 20 kHz. The result is immediately read out on the screen

1/3 octave filter set, ensuring synchronization between the pen of the recorder and the center frequency of the filter set.

This method is more accurate, but essentially much faster than the portable and inexpensive method shown in Fig.1.

The fastest method, and probably the most clear, is shown in Fig.3. Here, the entire range of "pink" noise, i.e. bandwidth 20 Hz — 20 kHz, is presented to the loudspeakers at the same time. The receiving side consists of the Real Time 1/3 Octave Analyzer (Type 3347), where all 30 1/3 octave filters are in parallel. The level in each 1/3 octave is at once displayed on the 12" screen in the form of a bright column. The total picture can be "stored", so that it remains on the screen after the signal has passed. If a hard copy is required, the "stored" signal can be read-out to the Level Recorder (Type 2305/07).

The last version of the method seems to be the one coming closest to normal music signals. The "pink" distribution is not so far removed from the distribution in music signals. In fact, there is no difference in results whether one generates 1/3 octave noisebands and measures linearly, or whether one generates the entire range of "pink" noise and measures selectively in 1/3 octaves. Of course the best signal-to-noise ratio will be achieved by both generating and measuring in 1/3 octaves.

Phase measurements

The methods discussed until now have only considered the amplitude response of the Hi-Fi set and the room, and it has been said that this method seems to compare much better with results from listening tests than the classical sine measurements in anechoic chambers.

If it is required to measure the phase response of the system without using a computer (see later text), it is necessary to use an anechoic chamber, because phase and sine are connected by nature, and sine is very inconvenient when used in reverberant rooms. Why then should phase measurements have anything to do with the sound picture in the actual listening room, when amplitude response measured with sine signals has almost no connection? The answer is, that a straight phase characteristic together with a straight amplitude characteristic implies a perfect impulse response, and that the ear mainly perceives the front edge of the pulse. Thus, considering impulses, it is mainly the first part of the signal which has influence. Now the part that arrives first to the ear in the actual room is the part which is transmitted directly, and it is exactly this part which is measured in the anechoic chamber. In this way, phase measurements made in anechoic chambers seem relevant, also concerning Hi-Fi sets used in normal listening rooms.

The problem with measuring phase on loudspeakers is, that the signal has to pass through the air, which takes time. This time delay means a phase shift which, moreover, is frequency dependent.

Brüel & Kjær has solved this problem by constructing a Delay Line (Type 5675) which, independent of frequency, delays the signal in a reference channel for a period corresponding to the time which the signal takes to travel through the air between the loudspeaker and the measuring microphone. The Delay Line is a digital instrument, and it is possible to select delay times corresponding to measuring distances of 0,5 — 1,0 — 1,5 — 2,0 — 2,5 and 3,0 m. The set-up is shown schematically in Fig.4. The Phasemeter (Type 2971) generates a DC voltage proportional to the phase difference between the output voltage of the microphone amplifier and a reference voltage, which is taken directly from the loudspeaker terminals and delayed for a period corresponding to the selected measuring distance. The phase difference is read-out digitally on the Phasemeter (3-digits) while at the same time, it is fed to the Level Recorder (Type 2307) in the form of an analogous DC voltage.

Using a Heterodyne Analyzer (Type 2010), as here, it is possible to obtain a linear sweep. The slope $\Delta\phi/\Delta\omega$, where ϕ is the phase and ω is the angular frequency, is then of interest. Normally the frequency

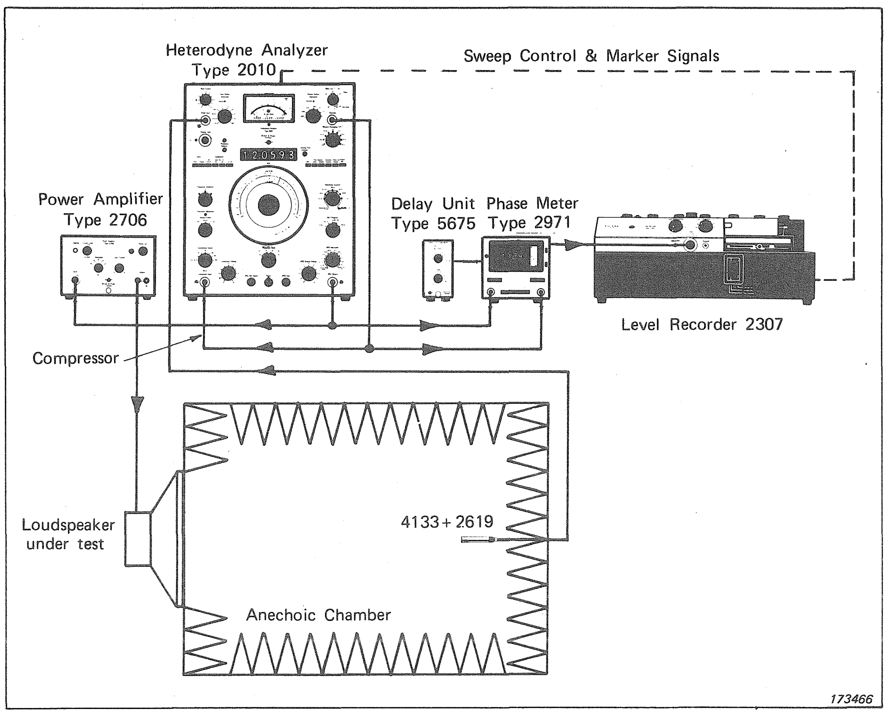


Fig.4. Set-up for phase measurement

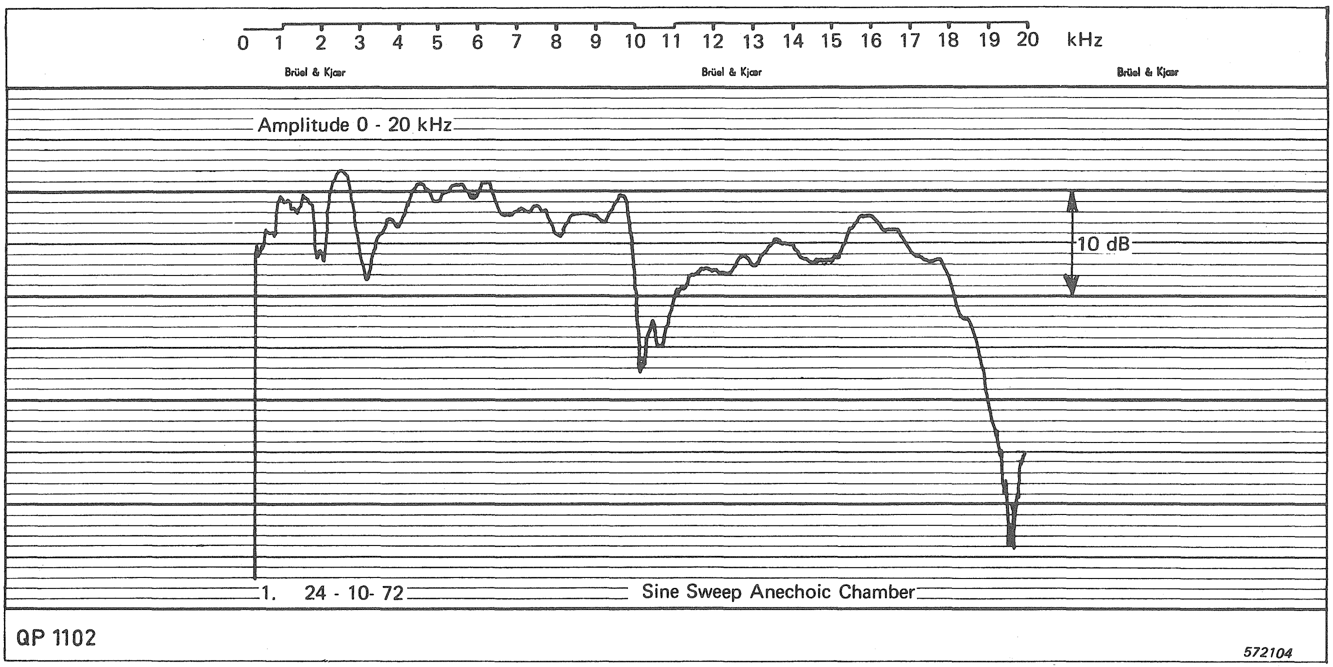
range can be divided into intervals with different constant average slopes modulated with some minimum phase-variation. We talk about minimum-phase variation, when there is synonymous connection between amplitude and phase, i.e. when maximum and minimum amplitudes correspond to the point of inflection in phase and vice versa.

When these intervals are caused by the different loudspeakers in a complex system, it is possible to find the relative time delay between, for instance, the midrange speaker and tweeter speaker. Analogous with the group delay $tg = -d\phi/d\omega$ we have the relative time delay $\Delta t = -\Delta\phi_1/\Delta\omega_1 + \Delta\phi_2/\Delta\omega_2$ — and now, knowing the velocity of sound $v = 340\,m/s$ we have the corresponding displacement $d = v \cdot \Delta t$. A Phase true loudspeaker can be made simply by mounting the midrange and tweeter with this physical displacement between them. The trouble with not having a straight phase characteristic is the introduced time delay distortion which means for instance, that the high frequencies in a complex signal reach the ear before the low frequencies.

Quite recently it was shown (Ref.4, p 31) that the human ear can detect a phaseshift of about 10° in the lower and medium frequency range. This is valid for normal listening rooms with more than 70 dB sound pressure level.

Using the Frequency Analyzer (Type 2010) the linear sweep possibilities are 0 — 2 kHz, 0 — 20 kHz or 0 — 200 kHz with extremely low distortion (< 0,015%, 20 Hz — 50 kHz).

Examples of corresponding amplitude and phase caracteristics are shown in Fig.5.



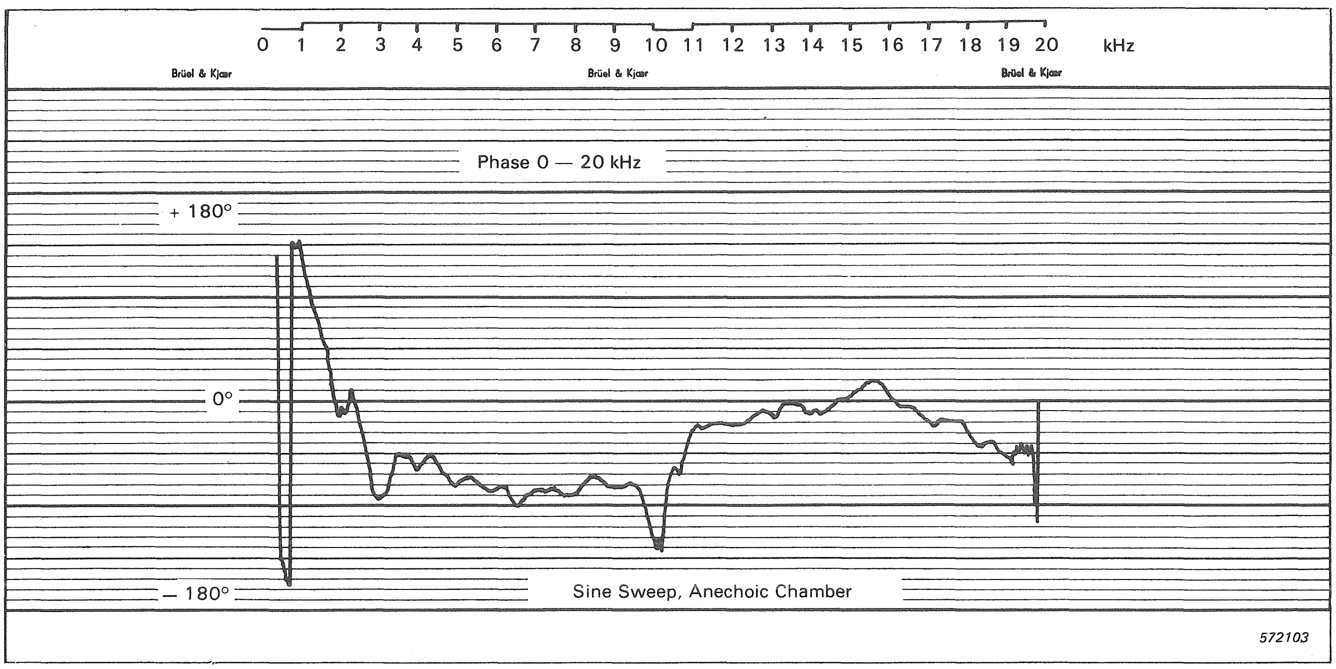


Fig.5. Amplitude and Phase Characteristics measured using 0 — 20 kHz Swept Sine Wave in Anechoic Chamber. This is loudspeaker H4

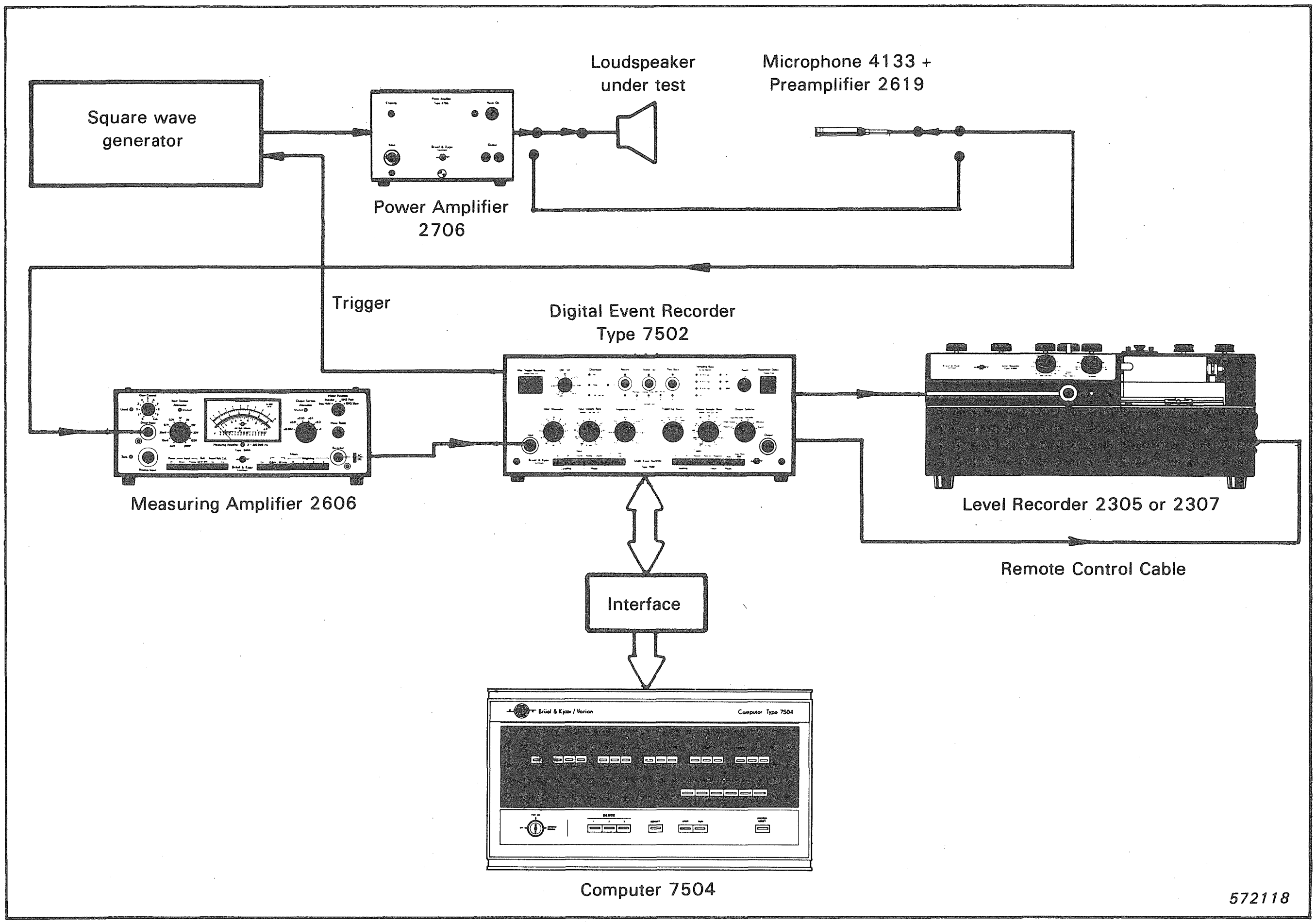


Fig.6. Experimental set-up for measuring Loudspeaker Transfer Functions

The computer method

The most advanced method requires that a mini Computer (Type 7504) and a Digital Event Recorder (Type 7502) are available. A set up is shown in Fig.6. In this case it is not necessary to use a special type of test signal. In principle any signal might be used, even music.

Of course, the most reasonable test signal to use is an impulse covering the audio frequency range, for instance, a $\sin^2 x$ pulse (Fig.7). This method gives the whole transfer-function $H(s) = A(s)\, e^{j\phi(s)}$ i.e. both amplitude characteristic $A(s)$ and phase characteristic $\phi(s)$. If appropriate one can find $H(s)$ for the system in the actual room, or one can remove that part of the information which corresponds to the reflected signal, which would then leave the same information one would have obtained in an anechoic chamber remaining. This method then gives the possibility of measuring the free-field characteristic without having to build an expensive anechoic chamber. The computer method and the theory of the Fast Fourier Transform are described in more detail in the B & K Application Note 12—244.

In order to show the application of the methods, the results from some of the tests made at Brüel & Kjær will now be discussed.

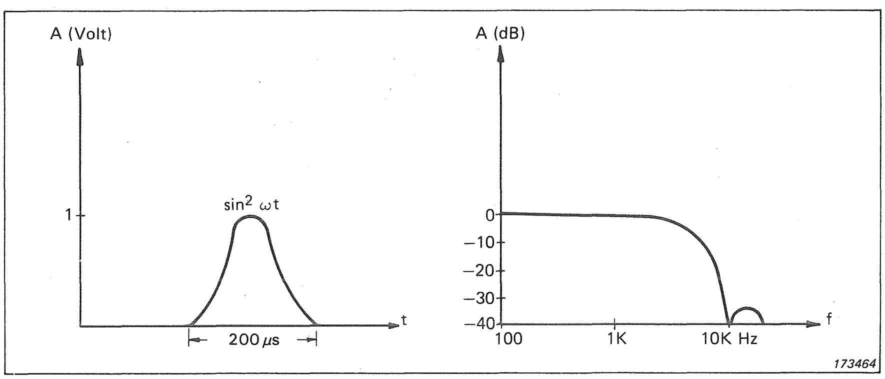


Fig.7. Sin^2 x pulse and its spectrum

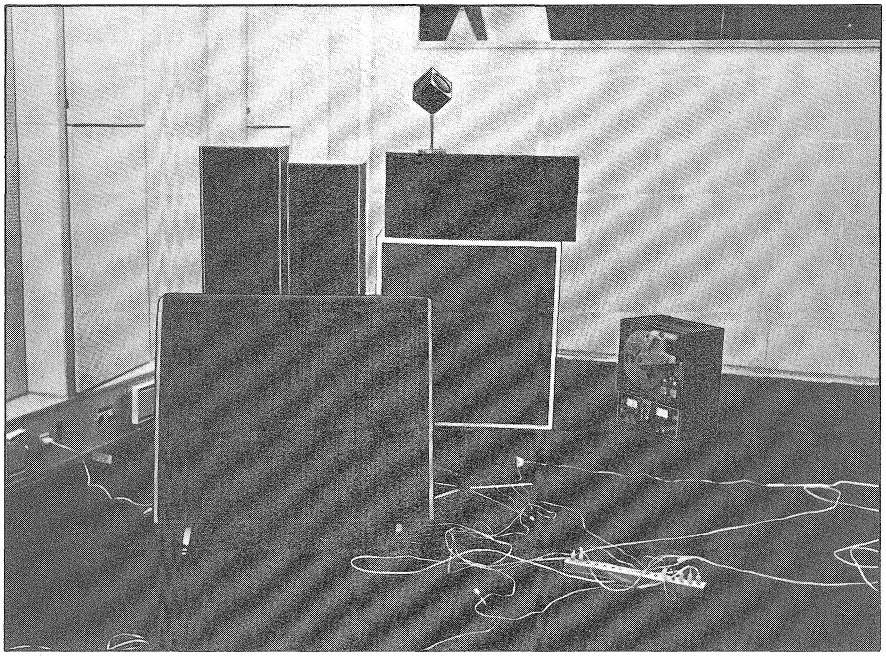


Fig.8. Room L1

It is obvious, that the characteristics for the loudspeakers differ quite considerably from room to room, so it seems a reasonable conclusion, as mentioned in the introduction, that the actual listening room ought to be measured.

The curves shown in Fig.11 were considered in one of two ways in order to decide which speaker performed the best in that particular room.

First, the curve should be as smooth as possible and it should, as mentioned on page 3, have a + 3 dB boost at 60 Hz and a —3 dB roll-off at 10 kHz.

Second, the normal frequency range of musical recordings was taken into account, the full level range being usually only from 60 Hz to 6 kHz — so this range was given more consideration than the rest.

From these criteria we arrived at the following preference sequence:

Results of the portable and inexpensive method

Analog to the "Loudspeaker investigation" (Ref.1), 5 loudspeakers and 5 listening persons were used. In the "Loudspeaker investigation" almost all kinds of measuring methods were tried, but only in one specially built average room. As mentioned earlier, the best measurement method was the "1/3 octave pink weighted random noise method" used in the listening room, therefore we concentrated on this method, and compared it with listening tests. Here, however, three different rooms were used. The rooms, and the method of positioning the 5 loudspeakers are shown in Fig.8, 9, 10 and the measured results are shown in Fig.11.

Room L 1: H1 - H2 - H4 - H3 - H5
Room L 2: H1 - H2 - H4 - H3 - H5
Room L 3: H2 - H4 - H5 - H1 - H3

Preference sequence from measurements

i.e. in room L1 H1 is the best, H2 second best and so on.

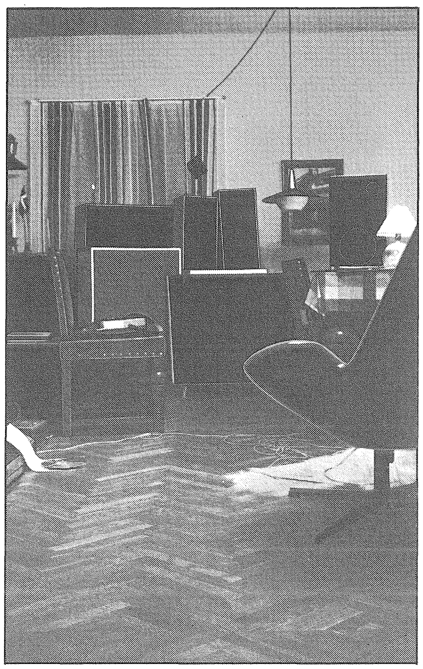


Fig.9. Room L2

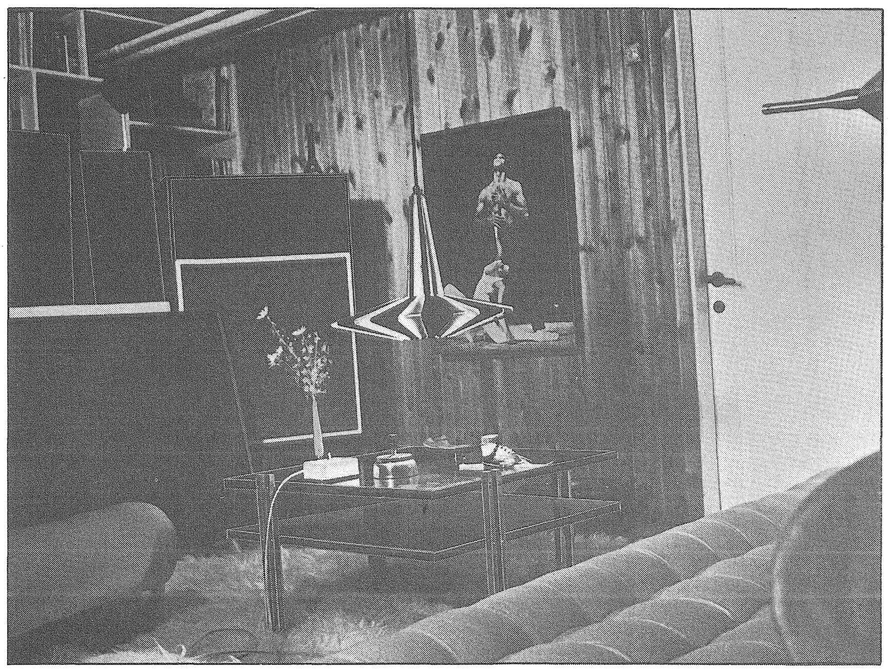


Fig.10. Room L3

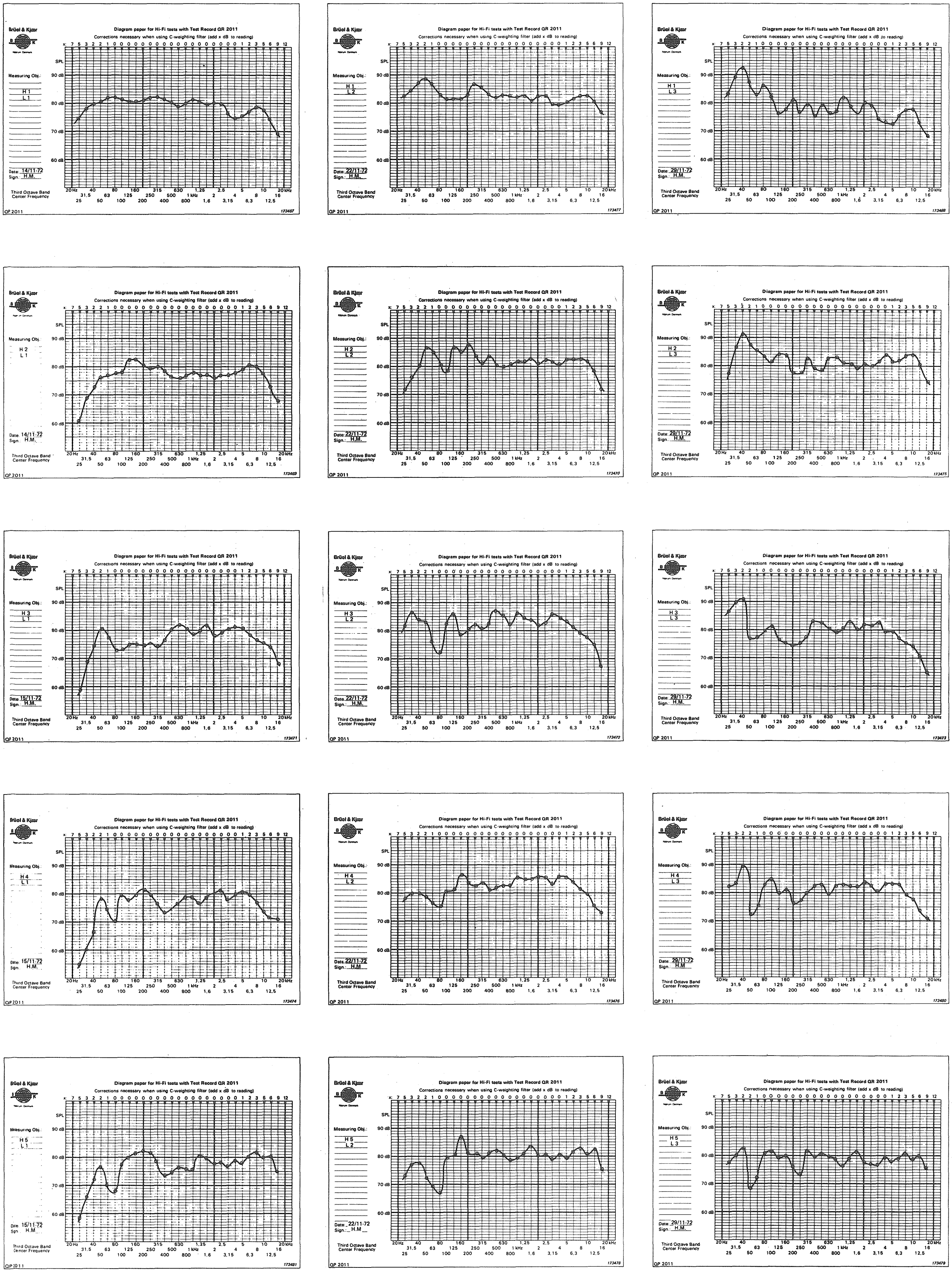


Fig.11. These curves show how the 5 loudspeaker responses differ in the 3 rooms. The measurements were made by the portable and inexpensive method Fig.1a. Note that the 3 in-line curves are for the same loudspeaker tested in different rooms

Results from listening tests in the three rooms

Throughout the listening tests, the loudspeakers were compared two by two. The person listening was given the choice which of the two loudspeakers he at a given time wanted to listen to. All the loudspeakers were equalised with "pink" noise to the same sound pressure level.

6 different, short, music pieces were used: opera, string quartet, church music, beat, jazz and popular music.

A diagram using 35 characteristics to describe the "sound" was used and for each characteristic, the listening person was to select the loudspeaker which in his opinion possessed most of that particular characteristic. Using 5 loudspeakers, 5 listeners, 3 rooms and 35 characteristics, a total of 5250 comparisons ((4 + 3 + 2 + 1) x 5 x 3 x 35 = 5250) were made.

It would not be reasonable to go into details of the statistical treatment of this material here, so let us simply examine the main result.

Fig.12 shows the number of times a given loudspeaker has been generally characterized as being the best one, as a function of the loudspeakers. It is seen, that the results for H1 differ 100% from room L3 to room L1, this would also seem to underline the room dependence. So from the listening test we arrive at the following preference sequence:

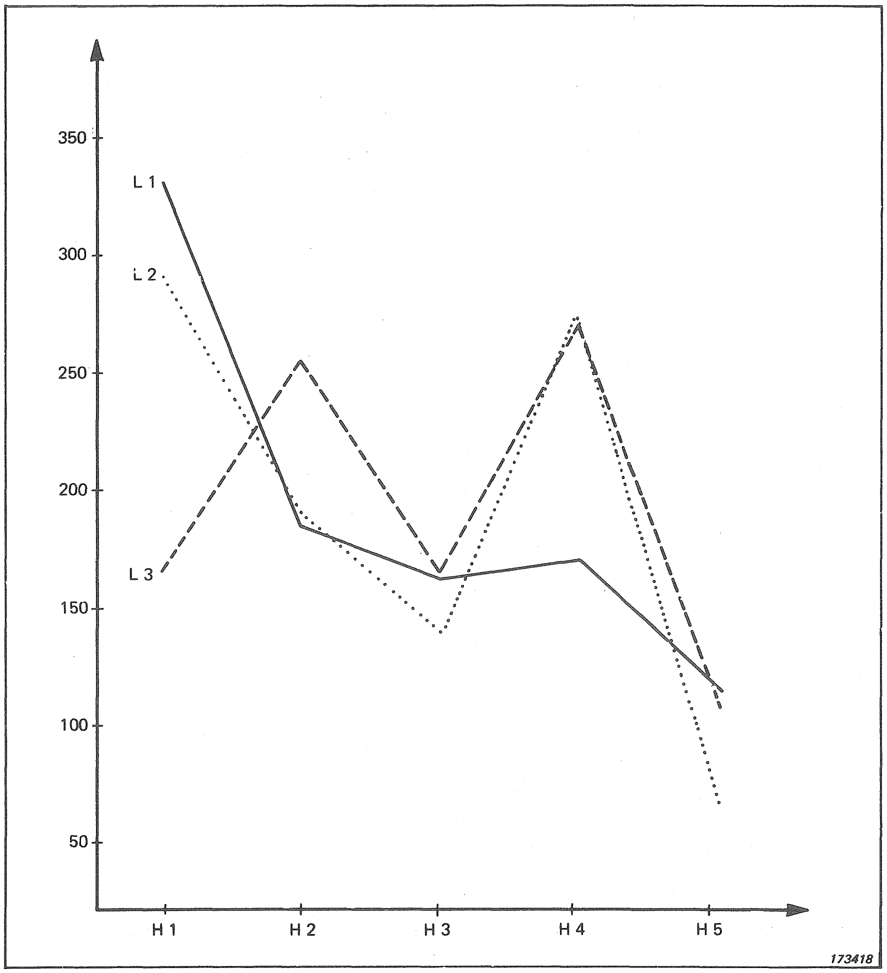


Fig.12. **Curves showing how the subjective quality evaluation of a particular loudspeaker strongly depends on the actual listening room. The y-axes shows the number of times a loudspeaker was preferred. These results are based on answers to positively orientated questions**

Room L1: H1 - H2 – H4 – H3 – H5
Room L2: H1 - H4 – H2 – H3 – H5
Room L3: H4 - H2 – H1 – H3 – H5

Preference sequence from listening tests

Now comparing this sequence with the measured results, it can be seen that the only essential difference in the results is that loudspeaker H5 in room L3 from measurements was placed as No.3 while from listening tests it was placed as No.5. The difference between loudspeakers H2 and H4 in room L2 is so small that it would seem almost impossible to state a preference.

Early Decay Time

Finally, the reverberation time (EDT) in the three rooms was measured by the Reverberation Processor (Type 4422). The results are shown in Fig.13.

Conclusion

It would appear then that the "1/3 octave, pink weighted, random noise method" used in the listening room is a reasonably valid method. Alone, it does not give complete information about the system but for normal use, it gives sufficient information, as the tests indicate. If the phase characteristic is measured also, then an almost complete picture regarding the system would be obtained. Of course there are other characteristics which could be measured, such as: Efficiency, Directional Characteristics, Harmonic Distortion, Intermodulation, Rumble, Wow and Flutter, Hum and Noise, Cross-talk etc. — but these, however, usually have low importance concerning the final result, and almost no influence on the results from listening tests.

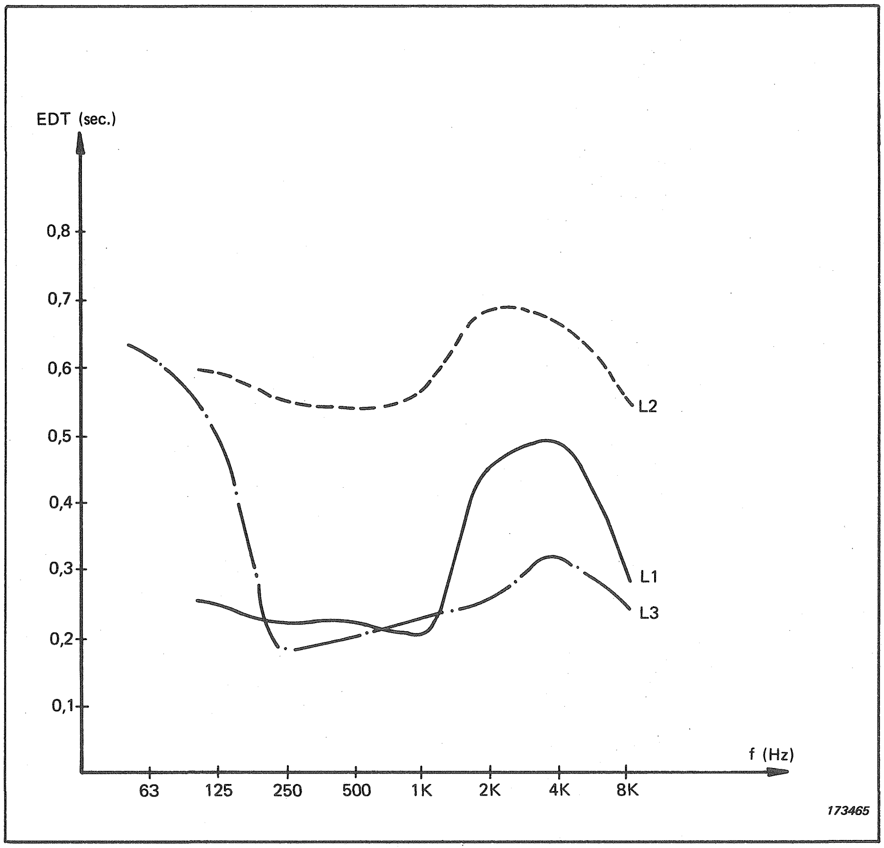


Fig.13. Reverberation Time (EDT) versus frequency in the three rooms

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Denmark 1972.

Ref.2. Roy F. Allison and Robert Berkovitz
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Journal of the Audio Engineering Society
July/August 1972, Vol. 20, No.6

Ref.3. Richard C. Heyser
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Journal of the Audio Engineering Society
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Threshold of Phase Detection by Hearing
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