A NEW, PSYCHOACOUSTICALLY MORE CORRECT WAY OF MEASURING LOUDSPEAKER FREQUENCY RESPONSES

PREPRINT NO 1963 (G4)

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**Presented at
the 73rd Convention
1983 March 15-18
Eindhoven, The Netherlands**

AN AUDIO ENGINEERING SOCIETY PREPRINT

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Abstract: The frequency response at the listener's ears is the most important property of the loudspeaker-listening room combination. Unfortunately conventional free-field and power response measurements do not give a correct picture of the time dependent frequency response a listener actually will hear in a normal listening room. A new method of loudspeaker frequency response measuring in the listening room is presented. The method is based on psychoacoustical experimental data in order to simulate the human hearing mechanism. Finally, measurement results are presented and compared to conventional measurements and listening results.

1. ABOUT LOUDSPEAKER MEASUREMENTS IN GENERAL

Loudspeaker measurements can roughly be divided into three different groups depending on in what environment the measurements are carried out:

- free-field measurements
- measurements in a reverberant room
- measurements at the listening position in a listening room

The results obtained in an anechoic chamber or in a reverberant room tell us something about the technical performance of the loudspeaker, but do not tell us what the speaker in question sounds like. This is true because of the following reasons:

- anechoic chambers and reverberant rooms do not acoustically resemble normal listening rooms
- the significance of the measured parameters is not exactly clear, because too little is known about the human hearing mechanism
- the measurements are carried out with a resolution, that differs from the resolution of the ear/brain system

"Laboratory" measurements are in no way meaningless or unnecessary. On the contrary they are very useful as loudspeaker design tools, as long as they are correctly interpreted. The outmost purpose of loudspeaker measurements should however be to tell the designer what the loudspeaker in question sounds like. In practice it has been shown, that good free-field and power responses are a must for a good sounding loudspeaker, but they do not quarantee a high performance.

It is well known, that the listening room acoustics and the loudspeaker positioning affect the perceived sound. Loudspeakers do sound different in different rooms and even in the same room, if they are positioned differently. Theremore it is clear, that loudspeakers should be measured under normal listening conditions at the listening position. The only problem is that the measurement method should resemble the hearing mechanism and to this day no method has done that. Before moving to new exotic measuring methods, it is probably sound to take a short look at, what and how humans hear.

2. BASIC FACTS ABOUT HUMAN HEARING /1/, /2/, althouh hifi-

The ear is very sensitive to errors in the frequency response. Most of the differencies between various hifi-components are due to differencies in the frequency response, although hifi-fanatics fanatics usually try to explain the differencies with some unknown form of distortion. The same phenomenon shows up when testing loudspeakers using A/B-comparisons; differencies are best heard, when the program material exhibits a broad spectrum, which clearly indicates that the greatest differencies are to be found in the frequency responses.

The ear analyses broad spectrum sounds with a kind of frequency analyser. The bandwidth of this biological bandpass filter, so called critical bandwidth Δf_G , is a constant 100 Hz for frequencies under 500 Hz and 15 - 20% of the center frequency for frequencies over 500 Hz (Fig. 1.). In practice this means that the ear performs a kind of integration of the sound pressure within one critical band /3/.

The natural frequency scale of the ear is linear for frequencies under 500 Hz and logarithmic for frequencies over 500 Hz.

The sensitivity of the ear to phase distortion using natural music signals has not been reliably proved.

The sensitivity to non-linear distortion depends significantly on the test signal and on the type of distortion involved. Distortion thresholds are not known well enough. In "good" loudspeakers of today the amount of non-linear distortion is so small, that a "bad" sound probably is caused by some other factor than distortion.

3. WHAT TO MEASURE

In view of what has been said it seems natural to pick the frequency response as the parameter that should be measured at the listening position. This should work out nicely, if the following aspects are taken into account:

- in addition to the direct sound, the room reflections should be taken into account to the degree they affect the perceived frequency response
- the resolution of the measurement method should be similar to that of the hearing mechanism
- the resulting document of the measurement should be easy to interprete, which means not more than two-dimensional

Lately it has become very popular to perform measurements, where the resulting document is three-dimensional. The frequency response of the speaker is shown in the amplitude-frequency and the amplitude-time domains at the same time. If a measurement like this is carried out at the listening position, the resulting graph contains a lot of useful information but the facts are not visible in their correct proportions - the important information disappears under the less important. In addition the three-dimensionality makes it difficult to read the graph with only a quick glance.

The requirements listed above will be met when proceeding in the following way:

- the amplitude-time domain of the three-dimensional "landscape" is shrinked into one dimension by summing the amplitudes using appropriate time and frequency dependent weighting
- the resolution is adjusted to correspond to the resolution of the ear

This operation is explained roughly in Fig. 2.

In order to achieve results like this, the developing of the WGT-method was started (WGT = Weighted Gating Technique).

4. THE WGT-METHOD

To put it short one could say that the WGT-method is a measuring technique, that uses gated sine-wave to measure the amplitude versus frequency response of a loudspeaker in a normal room. The sine-wave is taken from a sweep generator with a speed that corresponds to 1 mm/s on a B&K recorder. The WGT-method is unique, because it takes room reflections into account when deriving the actual amplitude at a certain frequency. The room reflections are weighted with respect to frequency and time.

How it functions: When a sine burst from the speaker reaches the measuring microphone, it trigs the gate to open. The gain of the measuring gate decreases as a function of time, which means that early reflections are weighted more strongly than late ones. The shape of the measuring gate is constant at all frequencies, but the time it stays open depends on the measuring frequency. See Fig. 3.

The same thing can be expressed mathematically:

(1) $$u_0 = \frac{k_1}{T_1}\int_0^{T_1} |p(t)|\,dt + \frac{k_2}{T_2 - T_1}\int_{T_1}^{T_2} |p(t)|\,dt + \frac{k_3}{T_3 - T_2}\int_{T_2}^{T_3} |p(t)|\,dt + \cdots + \frac{k_n}{T_n - T_{n-1}}\int_{T_{n-1}}^{T_n} |p(t)|\,dt$$

u_0 is result of measurement

$p(t)$ is sound pressure at the microphone

k is gain factor

$T = f(f)$ so, that when $f < 500$ Hz, T is constant
when $f > 500$ Hz, T is proportional to $1/f$

The resolution achieved by this method is $\Delta f \sim 1/\Delta t$, where Δt is the measuring time. By adjusting the measuring time according to the frequency, the measuring time can be kept at a correct value at all frequencies. From the ear's point of view the resolution is correct when Δt is constant for frequencies under 500 Hz (and of the right value) and reversely proportional to the frequency for frequencies above 500 Hz. The shape of the window is determined by k and T.

The weighting factor w can be calculated in the following way:

$$w_n = \frac{k_n(T_n - T_{n-1})}{T_1} \qquad (2)$$

It is completely useless to try to describe a gate window mathematical function, that would exactly correspond to the hearing mechanism, since there are no reliable and exact data available. The time window must therefore be shaped as well as possible relying on more or less empirical knowledge.

5. THE ACTUAL MEASURING DEVICE

The device was to be design so, that it could be used together with standard B&K equipment (i.e. measuring microphone, pre-amplifier, beat-frequency oscillator and recorder). The block diagram of the designed WGT-device is shown in Fig. 4.

The working principle is something like this: A sine burst made out of continuous sine-wave by a zero-point switch is fed into the loudspeaker. When the burst via the microphone reaches the input of the WGT-device and its amplitude is greater than the trig level, it starts the clock-oscillator of the logic which controls the measuring gate. The logic adjusts the gain of the gate to have one of four predetermined values. The received burst, which is shaped by the measuring gate, is rectified (full-wave) and integrated. At the end of a measuring cycle the integrator is read into a sample-and-hold circuit and reset. At the same time the logic stops its clock and the measuring cycle is over. In order for the resolution requirements to be fulfilled, the device has a frequency-to-voltage converter, which controls the following functions:

- the length of the burst fed to the speaker (always a little longer than the measuring cycle)
- the frequency of the clock-oscillator
- the time constant of the integrator

The output voltage of the f/V-converter is constant for frequencies under 500 Hz and proportional to the frequency above 500 Hz.

The device does not contain any oscillator controlling the repeat frequency of the sine bursts. A new burst is fed to the loudspeaker when the sound level is low enough for the measurements to be continued (adjustable level). This means that no bursts are fed to the speaker if the noise level in the room is too high. The repeat frequency of the bursts is therefore reversely proportional to the reverberation time at the frequency in question, granted also there is no non-correlating acoustic interference.

The determining of the geometry of the gate window was based on the earlier delay research work /2/ and updated psychoacoustical data. The measurement result follows equation (1), where $n = 4$. The time events T_1, T_2, T_3, T_4 and the effect of the weighting are seen in Fig. 5.

For example: At 1 kHz T = 3,16 ms. The resulting amplitude is the sum of four contributing time intervals in the following way:

- interval	0...3,16 ms	43%	of the total
- "	3,16...6,32 ms	22,8%	"
- "	6,32...15,8 ms	19,5%	"
- "	15,8...44,2 ms	14,7%	"

The burst reaches the microphone at T = 0

The resolution of this WGT-device is seen in Fig.6. As can be seen, the resolution is almost exactly the same as that of the ear, except at high frequencies where it is a bit better. The straight line in Fig. 6 shows the resolution of a third octave analyser. The resolution of the latter is unnecessary high at low frequencies and not high enough at high frequencies.

6. WGT-MEASUREMENT RESULTS

The loudspeakers chosen for the WGT-testing were all so called high-quality speakers, with reasonably flat free-field and power responses. From a sound quality point of view they could differ significantly from each other.

The speakers were as follows:

Speaker A : 3-way bass reflex, 200 mm woofer, 50 mm dome mid-range, 25 mm dome tweeter

Speaker B : 3-way closed box, 300 mm woofer, 100 mm cone mid-range, 25 mm dome tweeter

Speaker C : 2-way bass reflex, 200 mm woofer, 25 mm dome tweeter

Speaker D : 2-way bass reflex, 200 mm woofer, 25 mm dome tweeter

Speaker E : Open baffle, made directive on purpose, see /2/

Speaker F : Partly open baffle, equipped with a sub-woofer, made directive on purpose

It could be mentioned, that speakers B, C and D are very well-known British hifi-speakers.

The measurements and the listening tests were carried out in three different rooms, of which two exhibited quite similar acoustic conditions (heavily damped). These are called "soft room 1" and "soft room 2". The third room was acoustically harder than an average room and it is called "hard room". All speakers have not been investigated in all rooms, because this time the quality of the speakers was not essential. The interesting point was how well listening results and measurements correlate.

The measurements were taken at the listening position and the speakers were always positioned as optimally as possible. The distance between speaker and microphone varied between 2,5...3 m.

The block diagram of the measuring set-up is shown in Fig. 7.

The measurement results are shown in Fig. 8...13. Power and free-field responses, and at least one WGT-response is measured of all speakers. Some speakers are also measured with 1/3-octave noise with the help of a real-time analyser.

7. WHAT DO THE RESULTS TELL?

When comparing measurement and listening results, the following conclusions could be drawn:

1. If the WGT-curve is good (as flat as possible), the loudspeaker sounds good.

2. If the WGT-curve looks bad, the speaker might sound good with some particular program material but there are always signals (usually broadband) with which it sounds bad.

3. The overall frequency balance of the speaker is readily visible in the WGT-response.

4. If a loudspeaker sounds "muddy" or if it exhibits "lack of clarity", the WGT-curve shows a bad ripple in some frequency region. This kind of ripple is especially disturbing in the mid-frequency region.

5. In spite of the ripple in the WGT-response the overall frequency balance can be good. The only problem is the "muddyness" mentioned before.

6. A bad ripple in the mid-frequency region can partly be masked, if the loudspeaker reproduces low and/or high frequencies in excess.

When comparing measurement results obtained by different measurement methods, the following conclusions can be drawn:

7. Although the frequency response at the listening position measured with a 1/3-octave real-time analyser looks good, the loudspeaker still might sound "muddy". This particular method of measurement reveiles a bad overall frequency balance, but it does not tell whether the speaker suffers from a lack of clarity or not.

8. The first requirement for a good WGT-curve is a good free-field response. This fact may not be readily appearant from the responses in this paper, but that is because all the measured loudspeakers had a very nice-looking free-field response, and anyway this should be self-explanatory. The influence of the power response on the WGT-response depends on measuring distance, room acoustics and loudspeaker directivity.

9. The reason for the tight ripple in the WGT-response is of course the interference of the early reflections with the direct sound from the speaker. This matter has been discussed more extensivly in /2/. The only way to avoid a colouring ripple in the perceived frequency response is to make the loudspeaker directive. This can also be seen from the WGT-responses of speakers E and F. Of course the room interference can also be minimized by placing the speaker as far as possible from reflecting room boundaries.

9. FURTHER DEVELOPMENT POSSIBILITIES OF THE MEASUREMENT METHOD

As already been said, the frequency scale of the ear is linear for frequencies under 500 Hz. This means that measurement results also should be presented on a similar scale. In Fig. 14 we have a WGT-response presented on both the "old" scale and the "new" scale. The latter is easier to interprete correctly, because it shows the different frequency regions in the right proportions from the ear´s point of view.

One fact that makes this measurement method something less than ideal is, that the directivity pattern of a microphone does not resemble that of the human ear in the mid and high frequencies. A fully developed WGT-method should use a dummy head with acoustically correct earlobes. It would also be beneficial to extend the method to involve two-eared hearing, but that would create problems in deciding how to sum the right and left hand signals. This would without question require a lot of artificial brain from the signal processor, which in turn would require a lot of scientific research work. But at least it could be worth a try, because even a bad approximation of the right thing is better than nothing, as already proved by the WGT-measurement method.

10. CONCLUSIONS

A device which measures loudspeaker frequency responses with the same resolution as the human ear has been designed and constructed. In addition the device performs weighting of room reflections depending on their delay compared to the direct sound. Though the approximation is quite rough, the WGT-measurement results seem to correlate quite well with listening results. In fact, the correlation seems to be better than that of any other known measurement method.

The WGT-method can be used in the following situations:

- loudspeaker comparison measurements, loudspeaker tests
- in looking for the optimal speaker location in a certain room
- when improving room acoustics
- always, when a frequency response measured with the resolution of the human ear is wanted
- above all, when a loudspeaker is to be designed

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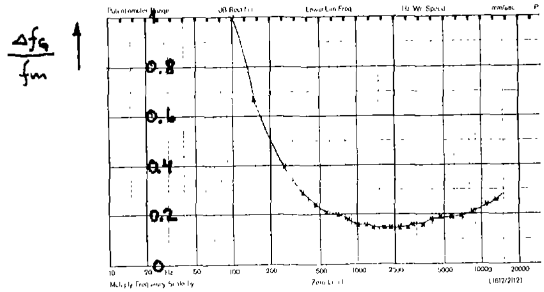


Fig. 1. The ratio of the critical bandwidth to the mid-frequency as a function of the frequency

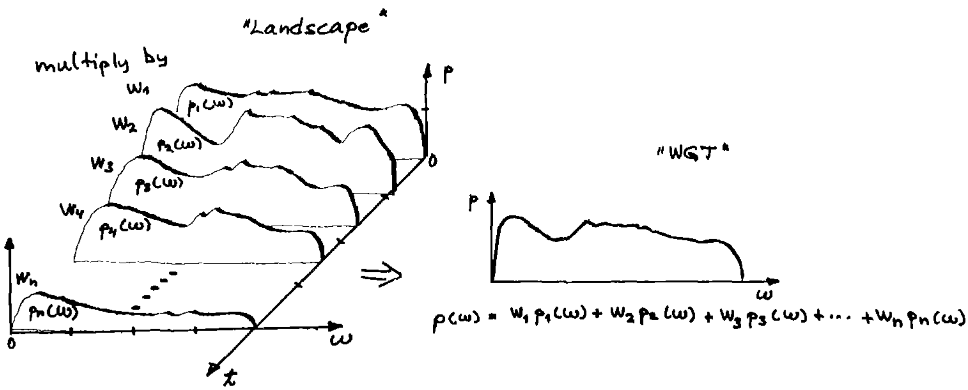


Fig. 2.

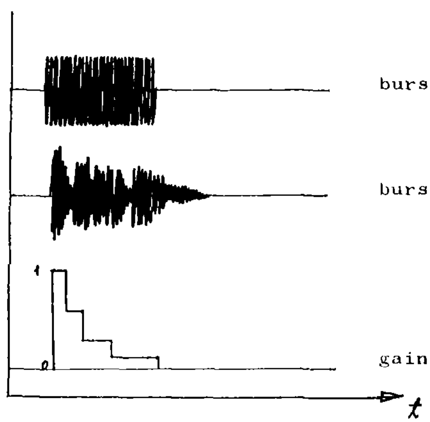


Fig. 3.

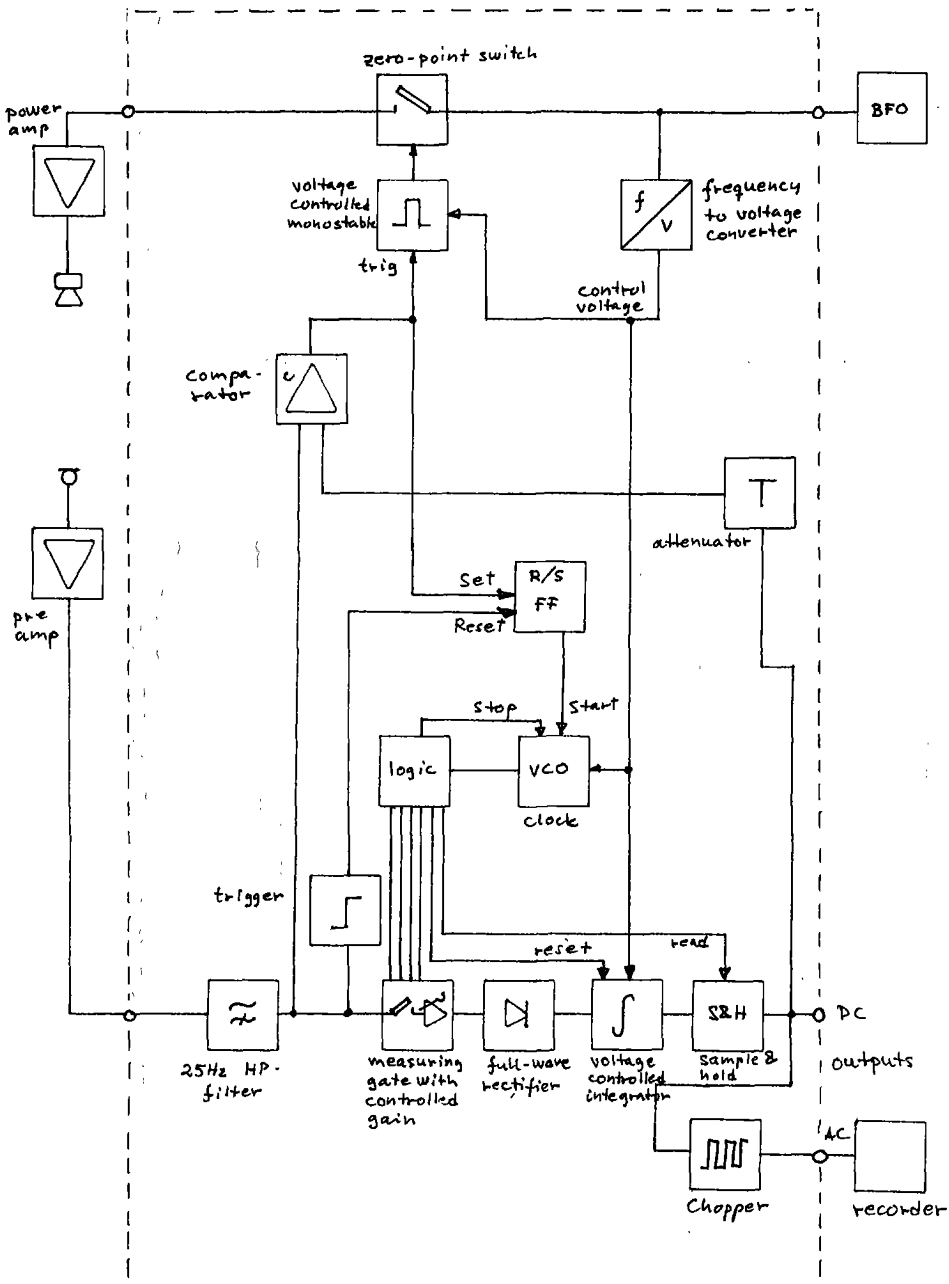
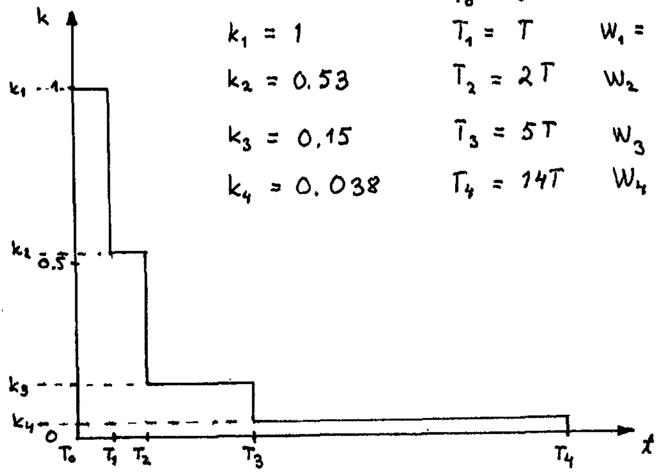


Fig. 4. Block diagram of the WGT-device



$k_1 = 1$ $\quad T_0 = 0$

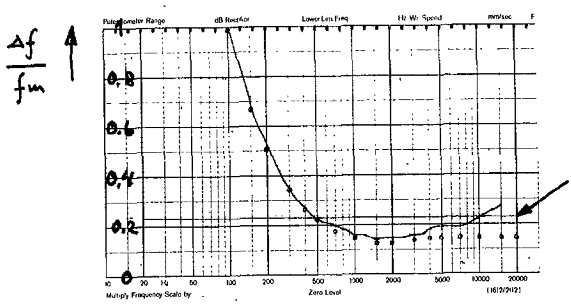
$T_1 = T \quad W_1 = \frac{1\cdot(T-0)}{T} = 1 \;\hat{=}\; 0\,\text{dB} \quad (2)$

$k_2 = 0.53 \quad T_2 = 2T \quad W_2 = 0.53 \;\hat{=}\; -5.5\,\text{dB}$

$k_3 = 0.15 \quad T_3 = 5T \quad W_3 = 0.45 \;\hat{=}\; -6.9\,\text{dB}$

$k_4 = 0.038 \quad T_4 = 14T \quad W_4 = 0.34 \;\hat{=}\; -9.3\,\text{dB}$

Fig. 5. The shape of the measuring gate



1/3-oct. $\Delta f/f_m = 0.23$

Fig. 6. Solid curve: $\Delta f_G / f_m$, same as in Fig. 1

∘∘∘ : $\Delta f/f_m$, the WGT-device

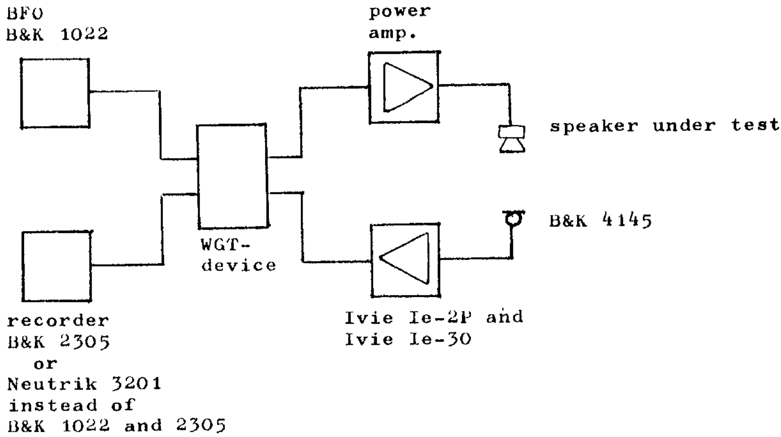


Fig. 7. The measuring set-up

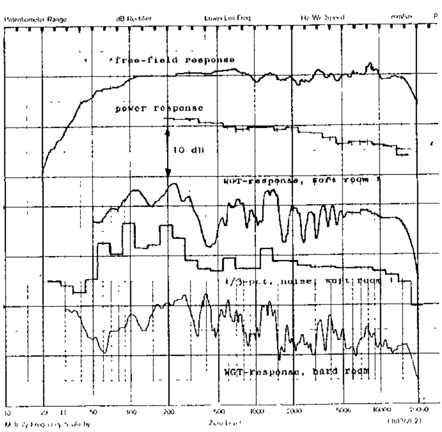


Fig. 8. Speaker A

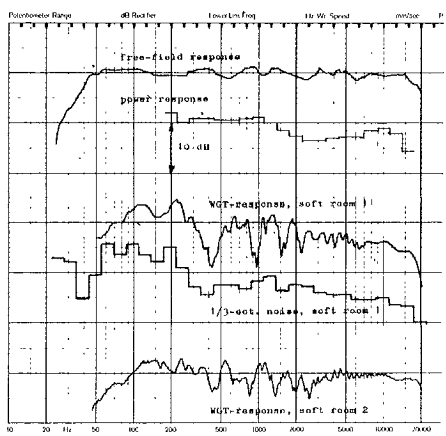


Fig. 9. Speaker B

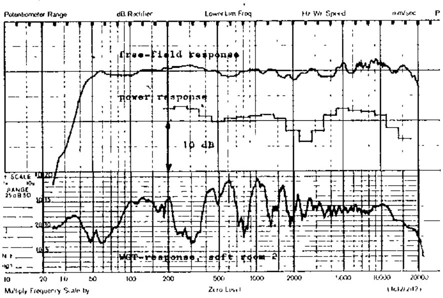


Fig. 10. Speaker C

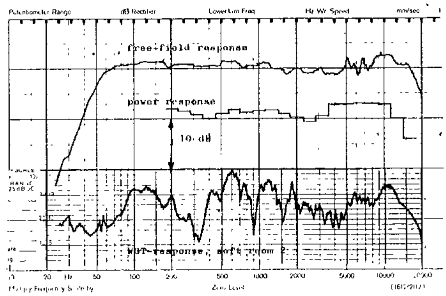


Fig. 11. Speaker D

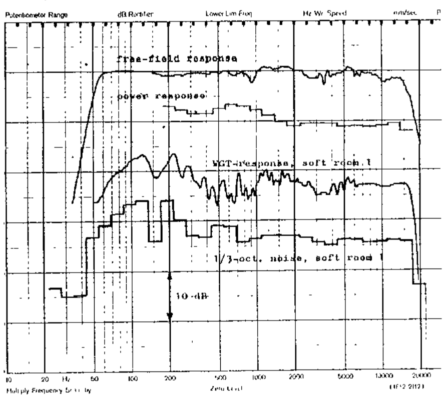


Fig. 12. Speaker E

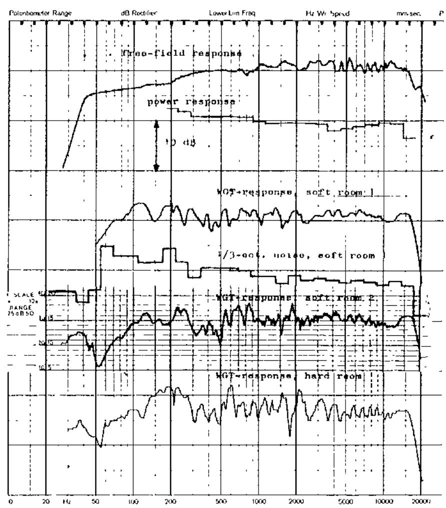


Fig. 13. Speaker F

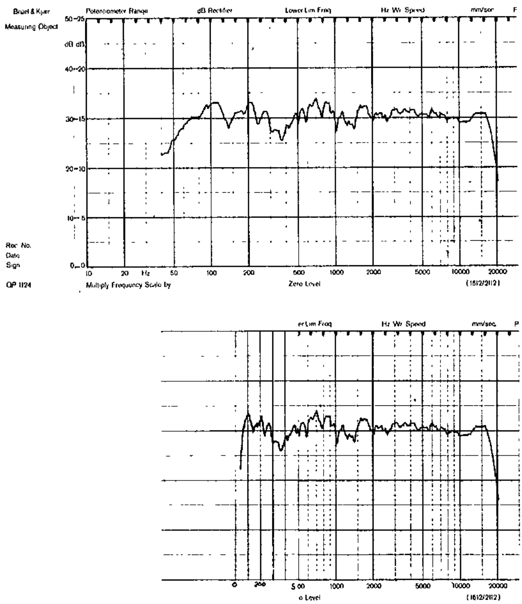


Fig. 14. The "old" and the "new" frequency scale