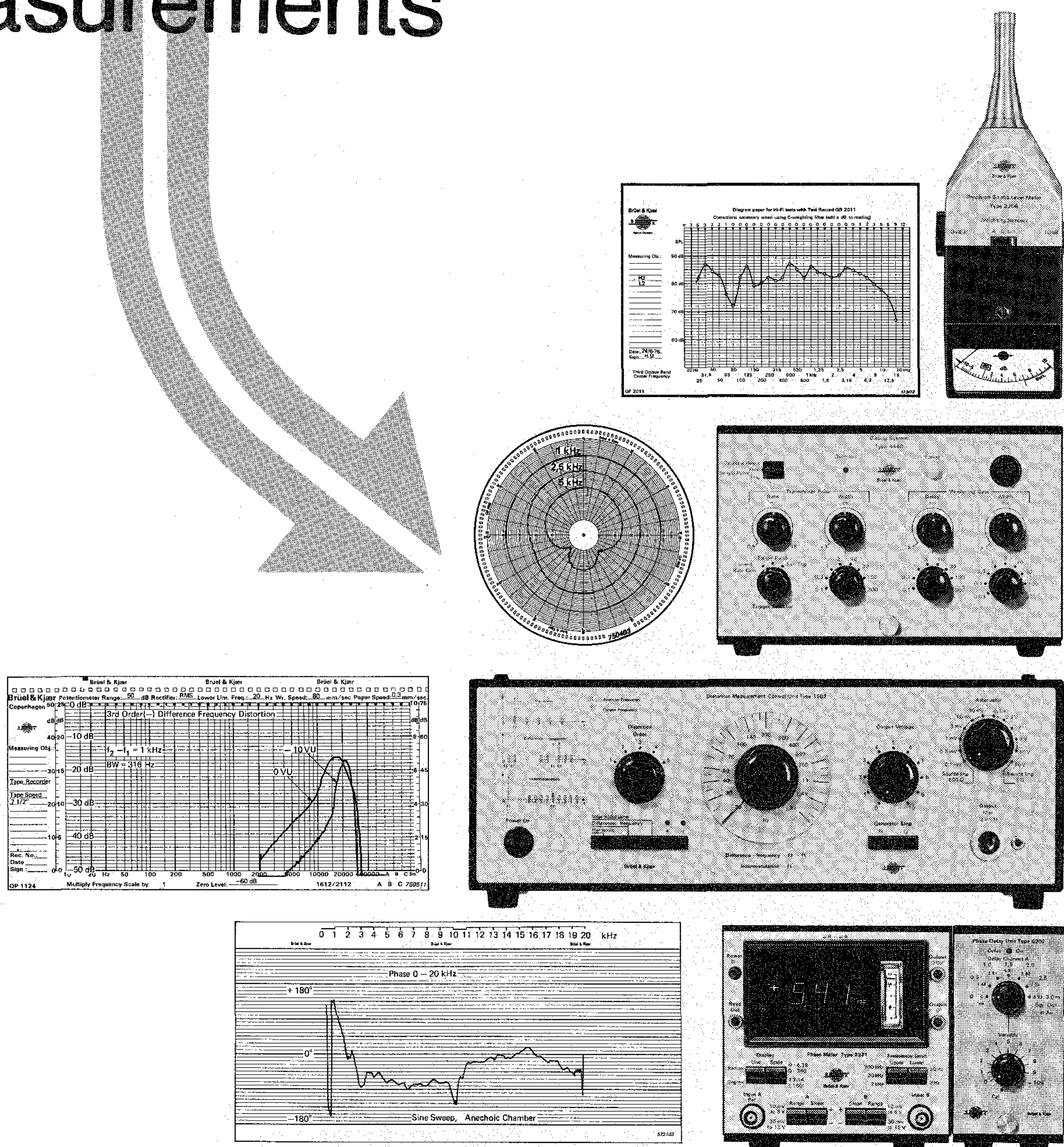
ELECTRO ACOUSTIC Measurements



Brüel & Kjær

16—035

CONTENTS

ELECTRO ACOUSTIC Measurements

By *Henning Møller,* Bruel & Kjær

1. Why Make Electro-acoustic Measurements?

Frequency response, phase response, harmonic distortion, intermodulation distortion, transient intermodulation distortion, directivity index, efficiency, sensitivity   the list of possible measurements is almost endless Too often the mass of data assembled leads to confusion, and not insight about the quality of the product being measured

Therefore, the fundamental problem for the electro-acoustic engineer is to select the appropriate measurements and evaluate the results by giving the proper weight to the various measurements He must then translate the objective results into the subjective domain using understandable and relevant terms However, it is an art to do this It requires a great deal of talent, insight, instrumentation, and time Only after a wide variety of measurements have been made is it possible to obtain "the complete picture"

In practice, the objective and subjective evaluations must work together to improve the final product Objective measurements alone are not enough, subjective judgments alone do not suffice They must be combined Then the objective measurements will lead to an improved product, and the subjective judgments will suggest improved, perhaps even new objective measure ment techniques

The purpose of all these measurements and subjective judgments is an improved product In practice, that means higher fidelity, a better loudspeaker, a superior microphone, an improved tape recorder And in the final analysis, whether or not the product has been improved will be judged subjectively But objective measurements will have played an important role in achieving the desired end result

These notes will concentrate on a broad range of measurements and will emphasize the dual application of most techniques, to acoustical and electrical measurements The notes are also structured along a line beginning with simple and then moving to more complex instrumentation The Bruel & Kjær system is a building block system, where the addition of one or more instruments to the basic combination permits new measurements This avoids duplication of equipment and permits advanced measurements to be obtained for a relatively small marginal cost

2. What Measurements are Relevant?

Only those measurements that in some way, directly or indirectly affect the subjective quality evaluation can be considered relevant Here we will deal with the primary ones in brief before discussing detailed instrumentation techniques

Frequency Response

The most common electrical and acoustical measurement is frequency response The pure sine frequency response is relevant for the electrical part of the system But when loudspeakers or other acoustical transducers are measured in an anechoic room, the pure sine excitation is not sufficient since the influence of the acoustical load, the listening room, is ignored Therefore, additional measurements such as sound power and measurements in a "typical" room should be made

Recent investigations have shown that frequency response measurements in the actual listening room using third octave, pink weighted, random noise, along with power response and phase response give the best correlation to subjective judgments

The main problem is defining a typical room, and it is doubtful whether this will ever be solved because of the wide variety in room dimensions depending on building type and also the country. However, if a typical room could be agreed upon, it would permit comparison of measurements from different manufacturers, and more important, would make it easier to predict the performance of a loudspeaker in a given consumer's room. Lacking this standard room, it would still be valuable for a manufacturer to make his own standard room with well-defined acoustical characteristics.

However, anechoic facilities are still necessary to evaluate the many different phases of loudspeaker design. Fortunately, it has recently become possible to synthesize anechoic conditions in a normal reflective room by the use of gating techniques (see section 5). Thus the combination of anechoic and normal room measurements should yield valuable results.

Phase Response

Frequency response alone tells us little about the transient response of the system. But when combined with the phase response, the transient response can be predicted. Traditionally, transient response has been measured — or rather evaluated — using tone bursts, but now a more complete picture can be obtained since phase can be measured as easily as frequency response. The more linear the phase response, the better the response of the system to transients such as found in percussion instruments, the attack of a trumpet, the pizzicato of violins, etc.

Distortion

These notes will also describe measuring techniques for various types of distortion, all of which have audible relevance. There is total harmonic distortion, individual harmonics measured separately, intermodulation and difference frequency distortion, and transient intermodulation distortion (TIM). Since distortion measurement at a single or at spot frequencies may give non-representative results, all of the measurement techniques described here will permit swept measurements of most types of distortion over the 2 Hz to 200 kHz range.

Frequency response, phase response, and distortion: these are probably the three most important categories of measurements. But in reality, all three categories are various types of distortion. If the frequency response is not flat, the amplitude will be distorted, and if the phase response is not linear, a time distortion will result. In fact, virtually all measurements seek to detect distortion of the original waveform in some manner or another. For example, wow and flutter results in frequency distortion, and noise adds to the waveform, thus distorting it.

As the various techniques for measuring these types of distortion are described in the following chapters it is hoped that insight will be gained, not only in better measurement technique, but in the art of combining the mass of objective measurements in a sensible manner and transforming them to the subjective domain. If this goal is attained, the ultimate purpose of obtaining better high fidelity can be approached.

3. Electrical Frequency Response, Signal to Noise

The basic system which will be the foundation of most measurements discussed in this course consists of three instruments, a sine generator, a voltmeter, and a level recorder. In the Brüel & Kjær system, these three instruments have a number of unique features that give them great flexibility and permit them to be the heart of a more complex measuring system. The generator used is the Sine Generator Type 1023 which provides a three decade linear or logarithmic sweep in synchronization with the Level Recorder Type 2307. The 2307 records the results of the measurements on pre-printed calibrated paper. The Measuring Amplifier Type 2606 is a true RMS voltmeter providing a calibrated AC or DC output for connection to a Level Recorder. With this basic system, frequency response, signal to noise ratio, and power can be measured. Later in the notes, other instruments will be added to this system to permit a wide variety of other measurements.

A typical instrument set-up for these measurements is shown in Fig 1 and a typical curve is shown in Fig.2.

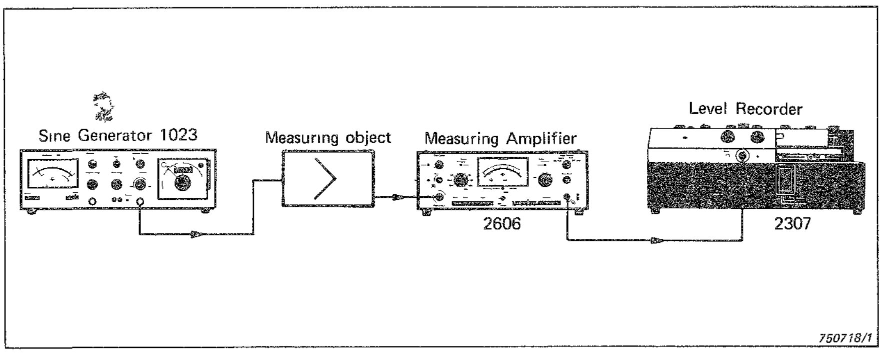


Fig.1. Instrument set-up for electrical frequency response

To give greater flexibility, three different generators are available

1023 10 Hz to 20 kHz, lin or log sweep, 0,1% distortion, warble tone
1027: 2 Hz to 200 kHz, lin and log sweep, 0,01% distortion, narrow band random noise, pink noise, white noise
2010: 2 Hz to 200 kHz, lin and log sweep, 0,01% distortion, plus a narrow band analyzer with the same frequency and dynamic range

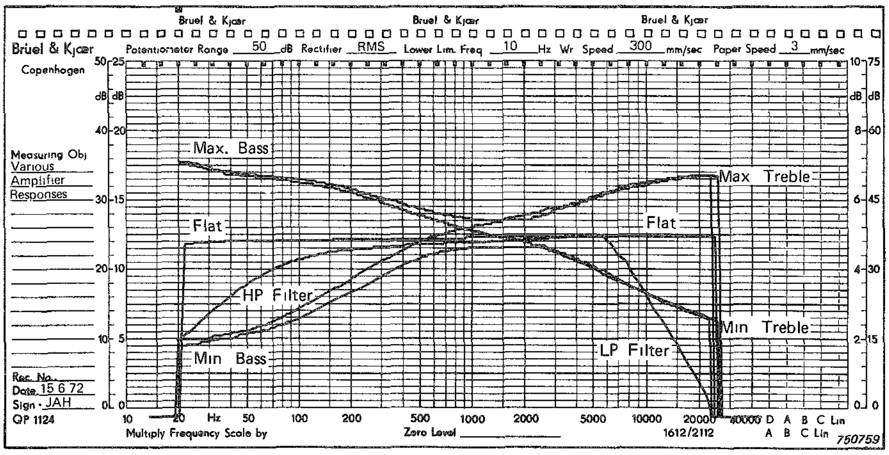


Fig.2. Frequency response curves measured with set-up from Fig.1 Influence of tone controls and filter of an audio amplifier is shown

Often, frequency response curves are made using a simple RC oscillator and a voltmeter. However, this only permits measurements of one frequency at a time.

The advantage of the Brüel & Kjær system is that the frequency of the oscillator is automatically swept in synchronization with the Level Recorder, giving permanent documentation on the preprinted paper. The generator covers three decades in an accurate logarithmic sweep, and unlike most RC oscillators, has a stable output amplitude while sweeping. And since the signal is generated using two high frequency signals, these may be used for easy synchronization of add-on slave filters for distortion measurements. The only disadvantage of the beat frequency oscillator is that the ultimate distortion figure currently attainable is about 0,01% while a few of the very best RC generators may be slightly better.

For overall record/playback response measurements of tape recorders, the 1023/2307 combination may be used provided that the sweep is slow enough so the delay between heads is not significant. Or the paper can be off-set to correspond to the delay, thus permitting fast sweeps. When creating a standard test tape to use for playback response measurements, the Response Test Unit Type 4416 is used to provide the proper automatic synchronization to the sweep of the Level Recorder (Fig.3).

Signal to Noise

The above system of Beat Frequency Oscillator and Measuring Amplifier is also used for signal to noise ratio measurements The maximum output voltage (S) of the system is measured when operating into a typical load. This is measured using a pure sine in the mid-frequency range of the system, and the maximum point is usually defined at the point where clipping begins, or the distortion reaches a given percent (for example 1 or 3%). The signal is then removed and the input is terminated with its characteristic impedance The gain of the measuring amplifier is then increased to read the noise voltage (N). The ratio of the two voltages (S/N) in dB is the signal to noise ratio

When measuring noise, the bandwidth or weighting network used must be specified. If the measurement is made with linear response, the upper and lower frequency limits must still be specified. The A weighting or a psophometer filter may also be used to give a closer approximation to audible levels Usually, measurements with a psophometer to DIN 45405 will give about a 7 dB higher noise level than with the A weighting network.

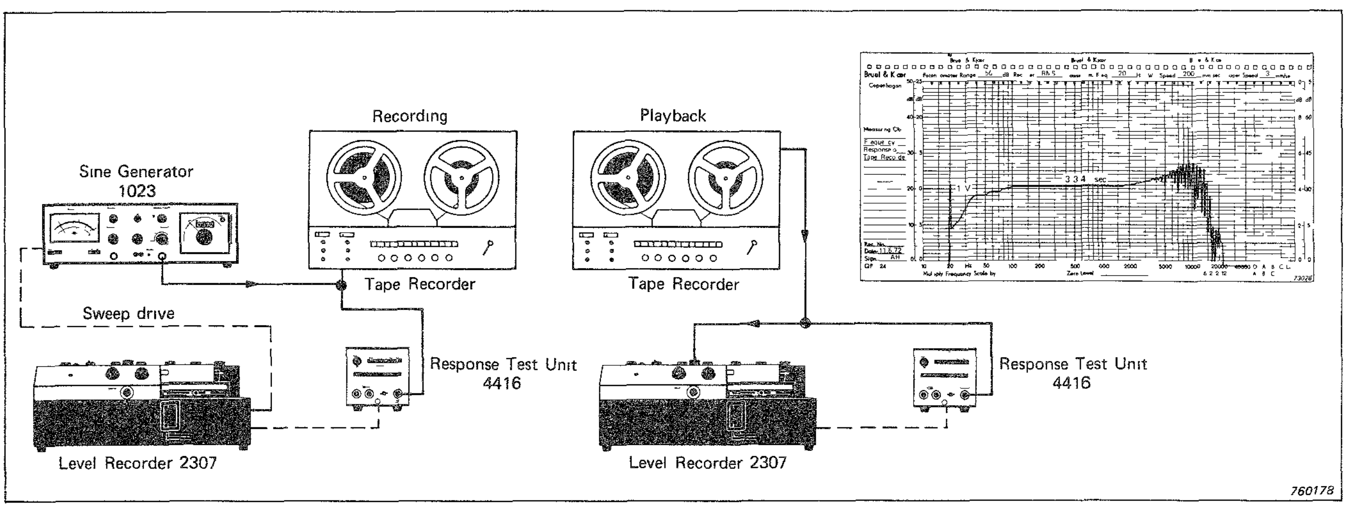


Fig.3. Frequency response measurements of a tape recorder

4. Acoustical Frequency Response in a Free Field

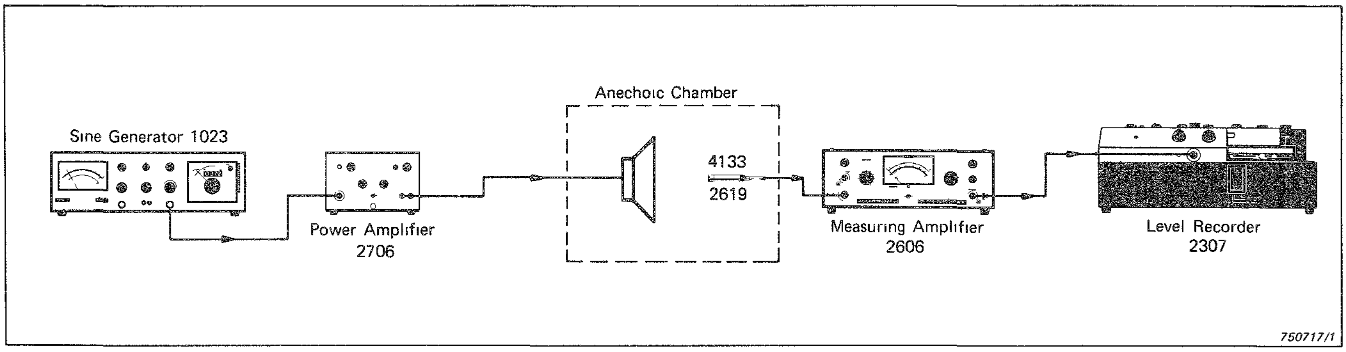


Fig 4 Set up for acoustical frequency response measurements

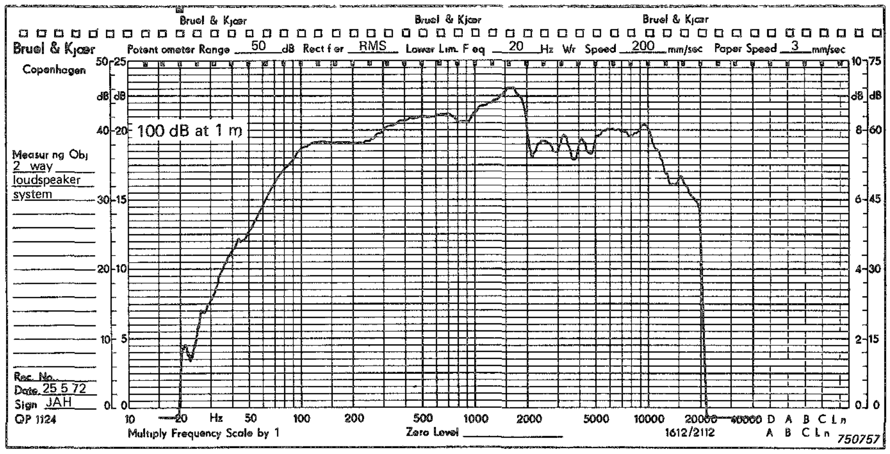


Fig 5 Sine response curve of a medium quality two way loudspeaker system

Loudspeakers

By adding a Power Amplifier Type 2706 to the output of the generator and a Condenser Microphone Type 4133 with Preamplifier Type 2619 to the basic system described above for electrical measurements acoustical response measurements may be made The Condenser Microphone has flat response within 2 dB over the entire 4 Hz to 40 kHz frequency range

This measurement requires a free sound field — either an anechoic room or an outdoor location (see Appendix) This makes this type of measurement somewhat inconvenient and expensive if free field facilities are not already available (In section 5 we will show how to simulate free field conditions using gating techniques) A typical instrument set-up and response curve is shown in Figs 4 and 5

This measurement may also be performed using bands of noise Two types of noise bandwidths are available With constant bandwidth noise, the bandwidth is independent of the center frequency Therefore, this is most often used with a linear frequency scale For constant percentage bandwidth noise, the bandwidth is always equal to a fixed percentage of the center frequency, such as 23% for third octave noise (Fig 6) Therefore the absolute bandwidth increases with increasing frequency, but when presented on a logarithmic scale the bandwidth will appear to be constant at all frequencies

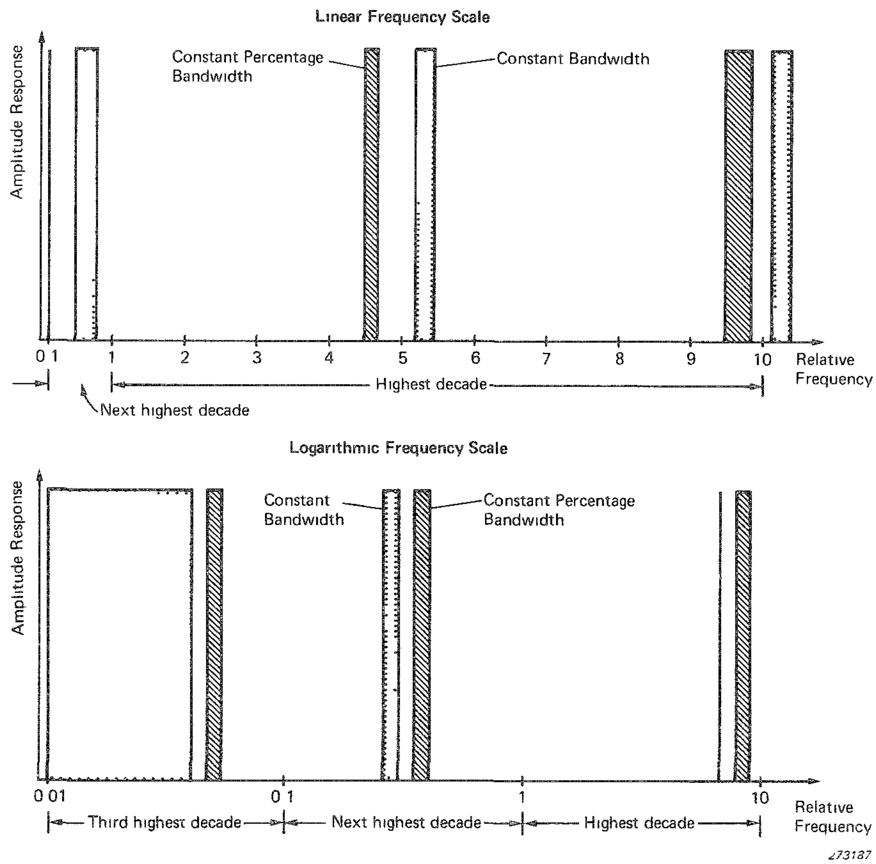


Fig 6 Constant bandwidth and constant percentage bandwidth filters

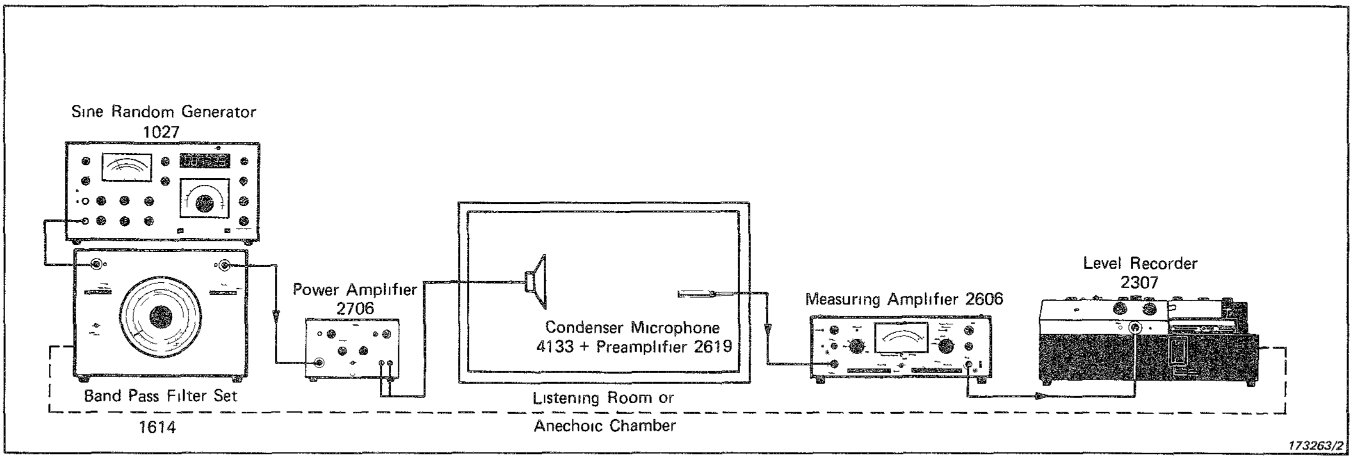


Fig.7. Set-up for response measurements using narrow band noise

Sine Random Generator Type 1027 may be used to generate both these types of noise. It provides constant bandwidth noise with bandwidths selectable from 3,16 Hz to 1000 Hz. It also provides pink noise which then may be filtered by an external third octave filter such as Type 1615. If only constant percentage bandwidth noise is required, Noise Generator Type 1405 may be used instead. A typical instrument set-up is shown in Fig.7 and the resulting curve is shown in Fig.8. In this set-up the switching of the filters of the 1615 from one center frequency to the next is automatically synchronized with the Level Recorder.

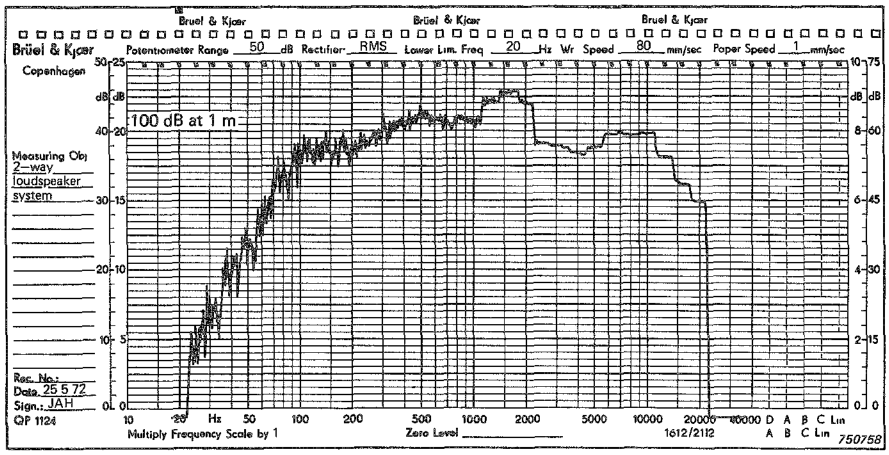


Fig.8. Response of loudspeaker to third octave noise

Microphones

Microphone measurements are slightly more difficult since they require the establishment of a sound field with a constant sound pressure independent of frequency. This, of course, implies a loudspeaker with flat frequency response, which is not attainable. However, this can be artificially achieved by using a reference microphone (Type 4133 is generally recommended) whose signal is fed back to a compressor circuit in the Sine Generator 1023 (Fig.9). The output level is then automatically adjusted to give flat frequency response at the point of the reference microphone. The microphone under test is then placed relatively close to the reference microphone and the frequency response at that location is assumed to be flat. If the two microphones are placed too close to each other, however, the reflections from the microphone under test will interfere with the sound field at the reference microphone, and vice versa.

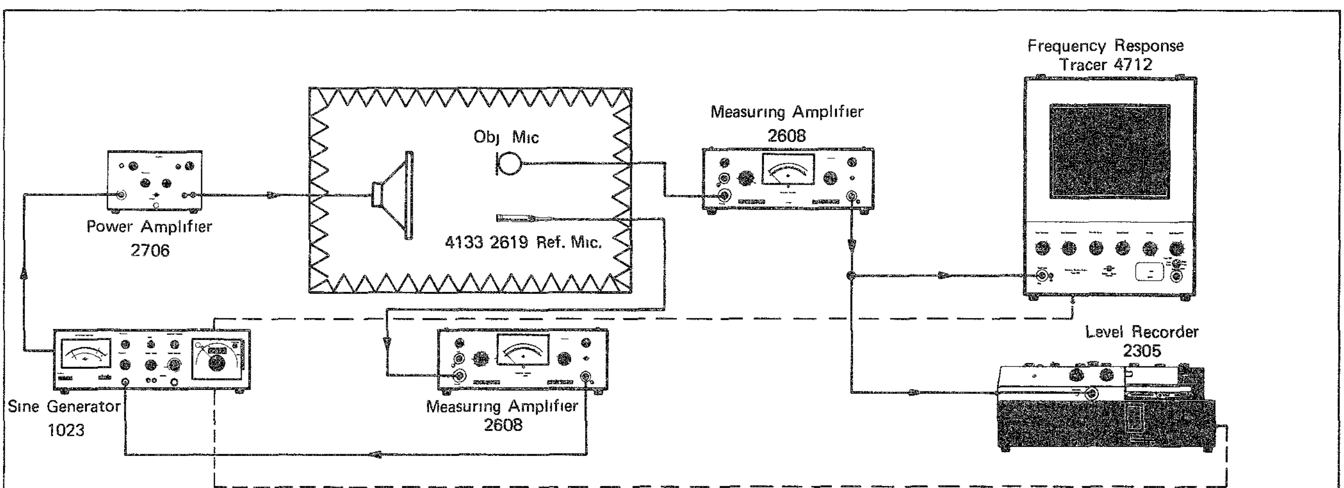
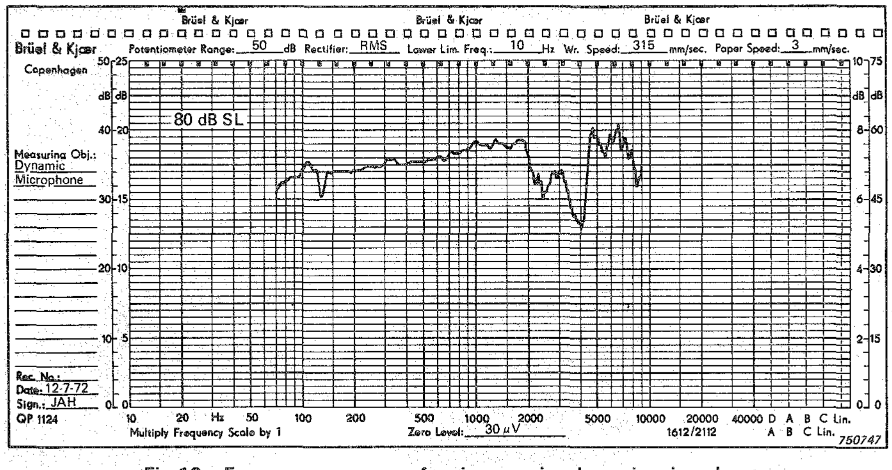


Fig.9. Compressor loop holds sound pressure constant for measurement on microphone



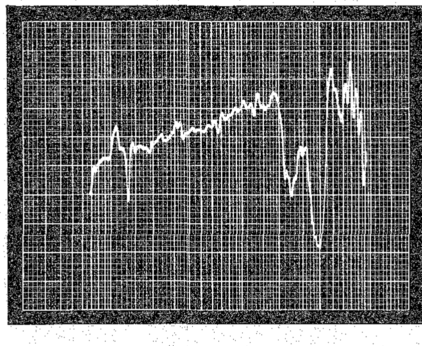


Fig.11. Response displayed on screen of Frequency Response Tracer

Fig.10. Frequency response of an inexpensive dynamic microphone

The resulting response curve (Fig.10) may be plotted on Level Recorder paper, or may be displayed on Frequency Response Tracer Type 4712 (Fig.11). The response tracer is especially of use in production testing since upper and lower limits may be drawn on the screen. However, the trace on the 4712 is not retained permanently.

5. "Free Field" Response in Ordinary Rooms using Gating Techniques

Basic Principles

By adding the Gating System Type 4440 to the above mentioned basic system of Oscillator, Measuring Amplifier, and Level Recorder, free field conditions may be simulated in ordinary reflective rooms. The restriction on lower limiting frequency is related to the size of the room, as is the case for anechoic rooms. With the gating system, frequency response, directional characteristics, harmonic distortion, and phase of the loudspeaker may be measured. In addition, the Gating System may be used as a tone burst generator for traditional tone burst evaluations of transient response. The system may also be used for measurement of early reflections, indicating the response of the loudspeaker a few milliseconds after the stop of the tone burst.

Since these measurements are described in detail in Application Note 15—107, "Electro-acoustic Free Field Measurements in Ordinary Rooms — using Gating Techniques" (Fig.12), only a short description will be given here.

The principle of the gated measurements is shown in Fig.13. A toneburst (A) is fed to a loudspeaker placed in an ordinary room. The signal received by the measuring microphone contains not only the original burst, but also several reflections from walls, floor, ceiling, etc. (B).

In addition, the shape of the initial burst is distorted. The first part of the burst contains the overshoot of the loudspeaker and the last part contains the overhang of the loud-

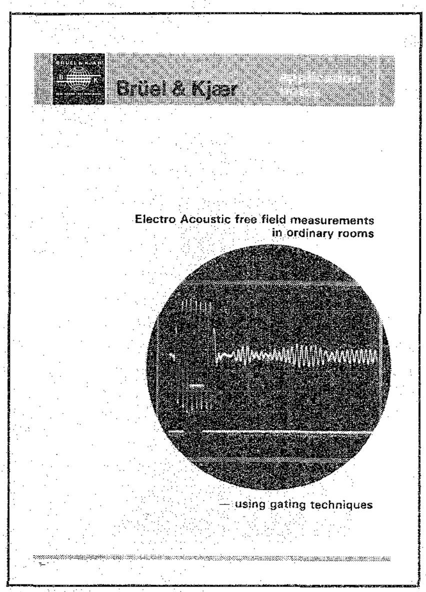


Fig.12. B & K Application Note 15—107

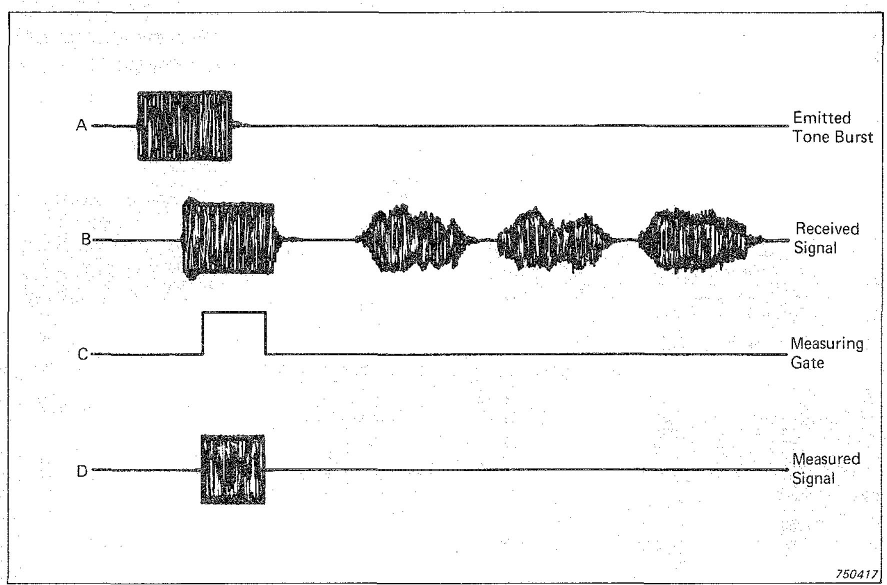


Fig.13. Principle of gated measurements

speaker But somewhere in the middle is a steady state portion whose amplitude is equal to the free field response This section is selected by a measuring gate (C) with appropriate width and time delay The level of the gated signal (D) can then be measured

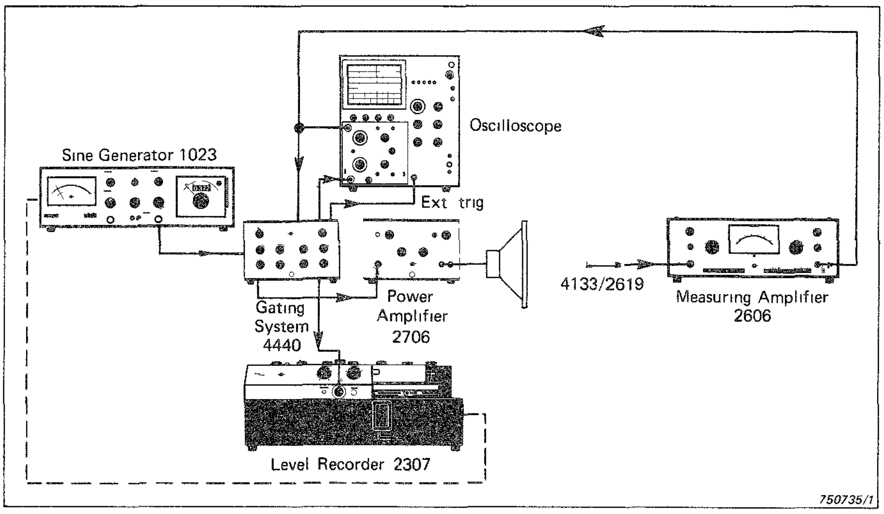


Fig 14 Set-up using Gating System

A key idea in the Gating System is that the received signal can be "edited" using the measuring gate Thus all parts of the signal may be examined individually and interesting phenomena such as early reflections may be studied

A typical instrument set up is shown in Fig 14 The output of the Sine Generator 1023 is passed through the transmitting section of the Gating System Type 4440 and transformed into a toneburst of adjustable length (0,3 ms to 1 s) which then is fed through the power amplifier to the loudspeaker The received signal from the Measuring Amplifier passes through the receiving section of the Gating System and is gated by the measuring gate, whose width and delay are also adjustable over a wide range A peak detector measures the amplitude of the desired signal and feeds a DC voltage proportional to this value to the Level Recorder for automatic recording of frequency response The peak detector contains a hold circuit which is reset for each new toneburst to permit capturing the new amplitude as the frequency changes To monitor the adjustment of the Gating System, a two channel oscilloscope is essential (Fig 15)

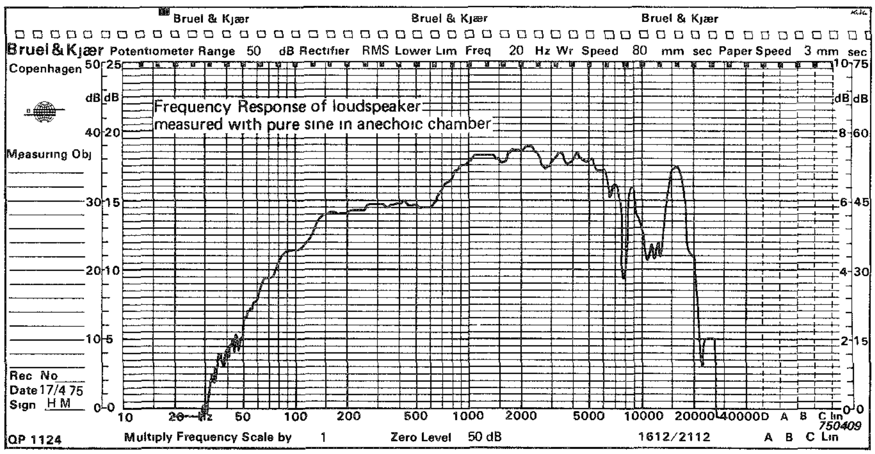


Fig 16 Frequency response of loudspeaker measured in anechoic chamber

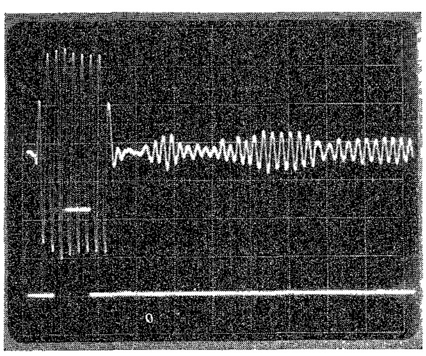


Fig 15 Proper adjustment of gate seen on oscilloscope

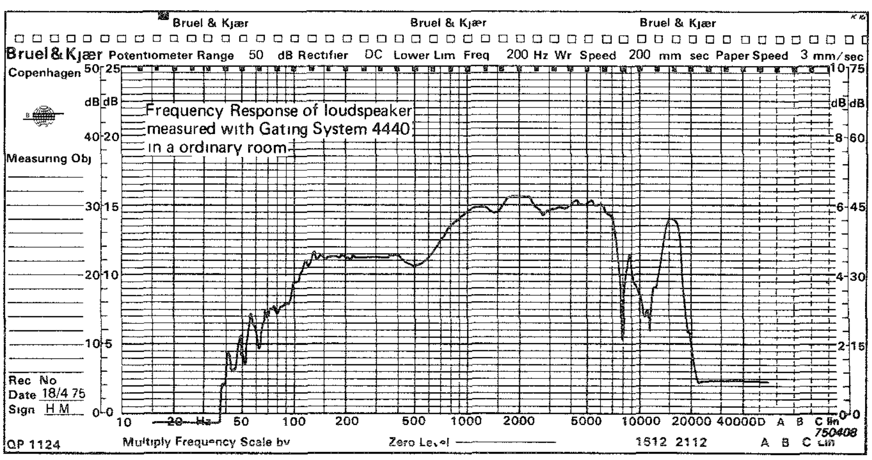


Fig 17 Frequency response of same loudspeaker measured with the Gating System

Typical results of gated measurements are shown in Figs 16 to 19 and show good correlation between anechoic measurements and Gating System measurements

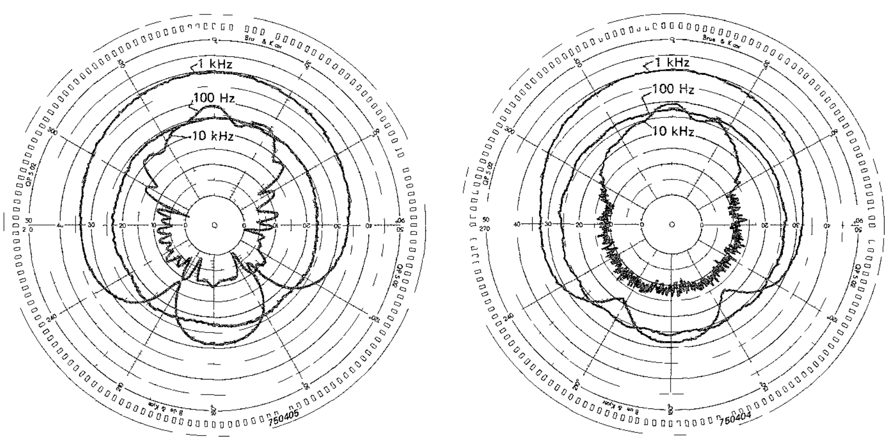


Fig 18

Directional characteristics of loudspeaker measured with pure sine in anechoic chamber

Directional characteristics of same loudspeaker measured with the Gating System in an ordinary room Note the 10 kHz curve shows the background noise level of the room

Early Reflections — or "Box Sound"

If the measuring gate of the Gating System 4440 is adjusted just after the end of the tone burst, early reflections inside the box which often are predominant at cabinet resonance frequencies, may be measured as a function of frequency. Such typical curves are shown in Fig.20 It is seen that the steady state curve contains only a slight peak at 3 kHz which is much more pronouced in the early reflection curves Thus the subjective sound coloration at 3 kHz might be greater than the steady state curve suggests

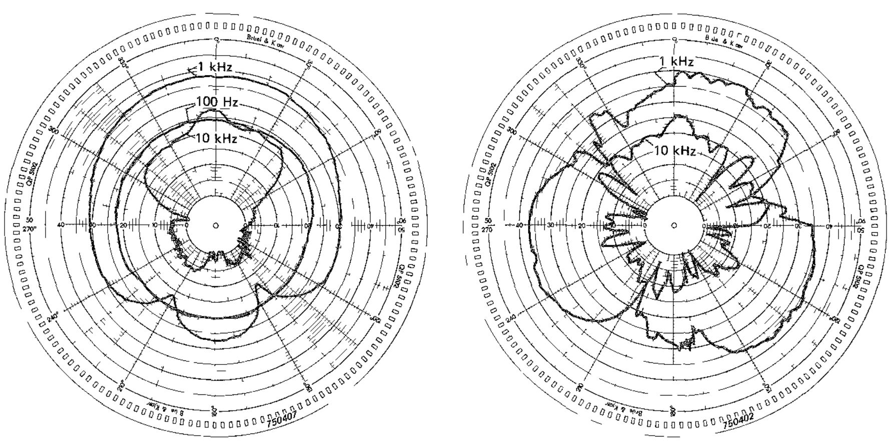


Fig 19

Directional characteristics of loudspeaker measured with Gating System in anechoic chamber. The sharp dips and peaks are not as pronounced as for pure sine

Directional characteristics of loudspeaker measured with pure sine in an ordinary room

A more sophisticated set-up for this type of measurement is shown in Fig.21 Here, white noise is gated in bursts and the transmitting section of a second gating system is used to process the received signal and pass it on to Real Time Narrow Band Analyzer Type 3348 which displays the entire frequency spectrum with 400 lines resolution

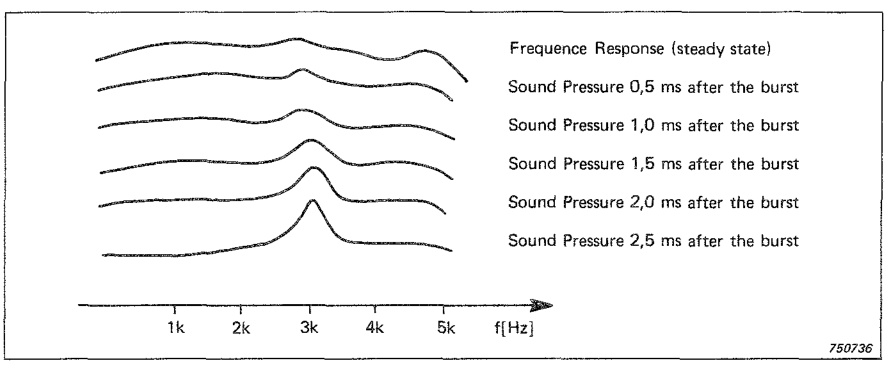


Fig.20. Frequency response curves as a function of time after the stop of the tone burst indicating the influence of early reflections

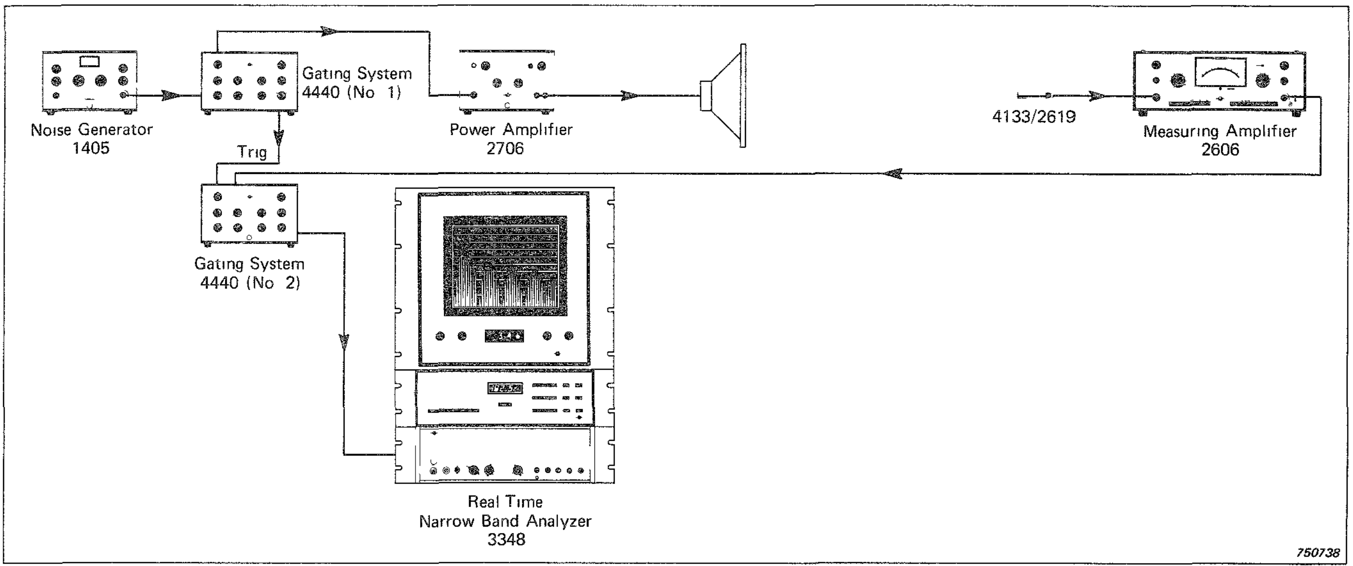


Fig 21 Set-up for detailed investigation of early reflections using two Gating Systems, and Real Time Narrow Band Analyzer Type 3348 to examine gated white noise

Mechanical Resonances of Cabinet

One possible reason for early reflections is mechanical resonances of the loudspeaker cabinet These can readily be measured by placing an accelerometer on the cabinet and sweeping the generator and plotting the acceleration level on the Level Recorder This will clearly point out the resonances (Fig 22) Any section of this curve may be expanded for a more detailed examination, and the frequency marking provided by the Heterodyne Analyzer Type 2010 still permits frequency calibration of the paper as also shown in the figure

Recent tests in Denmark have shown the value of these types of measurements By use of appropriate damping materials, the amplitude of resonance peaks was reduced approximately 10 dB and an audible improvement was noted

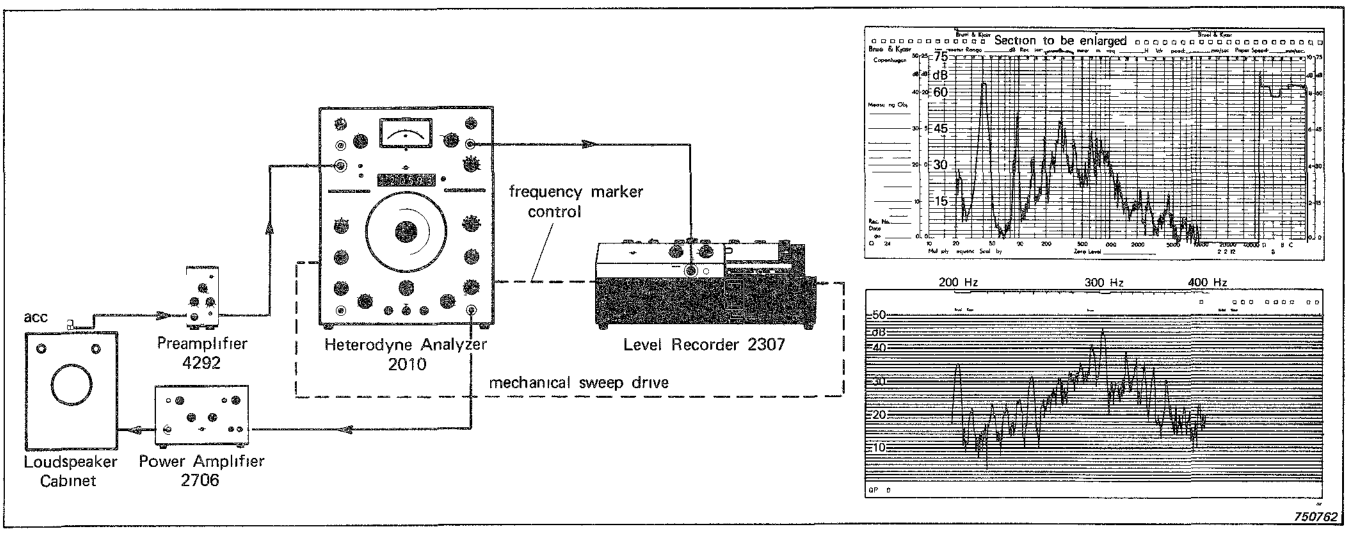


Fig 22 Measurement of loudspeaker cabinet resonances

6. Directional Characteristics

The basic measurement system consisting of the Sine Generator Type 1023, Level Recorder Type 2307, and Condenser Microphone Type 4133 (with preamp.) can easily be expanded to perform directional characteristic measurements by adding Turntable Type 3922. A practical set-up and typical result is shown in Fig 23 and Fig 24.

These measurements are especially of importance in determining high frequency dispersion and can also be used for calculating sound power. The loudspeaker (or microphone) under test is placed on the turntable which rotates in synchronization with the Level Recorder which is fitted with polar paper. Power to the loudspeaker is provided through slip rings to prevent cables from getting tangled. If complete directional characteristics are not desired, the system may be set to only sweep through a restricted angle, say 60° while still maintaining Level Recorder synchronization.

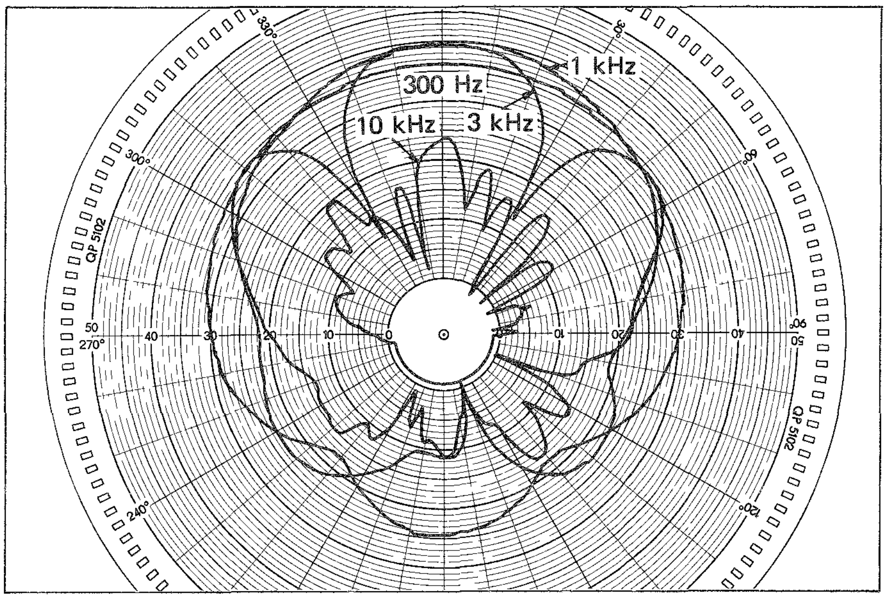


Fig 23. Typical directional characteristics of a loudspeaker

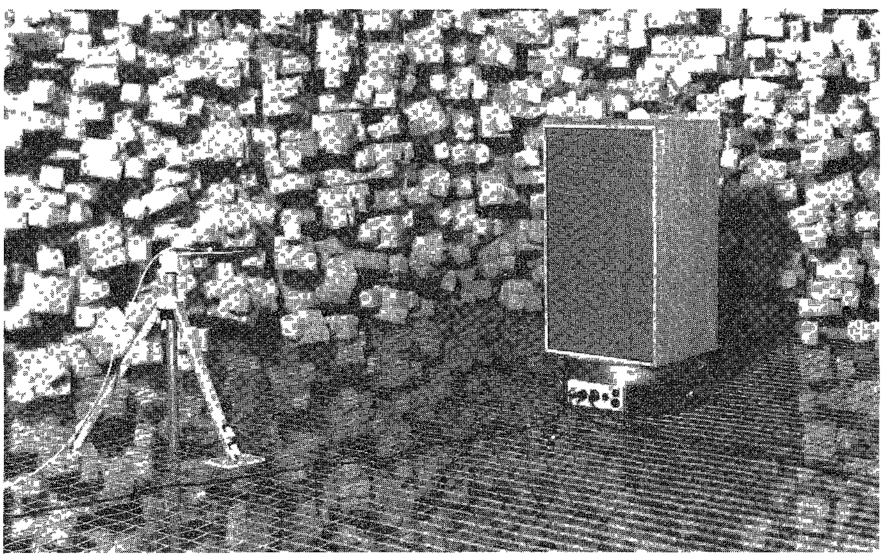


Fig.24. Turntable is used to measure directional characteristics

7. Harmonic Distortion

Total Harmonic

Unfortunately amplifier and loudspeaker systems introduce other frequencies into the output of the system than were present at the input. This is primarily due to unlinearities in, for example, transistors, or the magnetic field of the loudspeaker. For a single input frequency this results in harmonic distortion, for several input frequencies, intermodulation distortion.

Harmonic distortion is illustrated in Fig.25 where a 1 kHz tone is in-

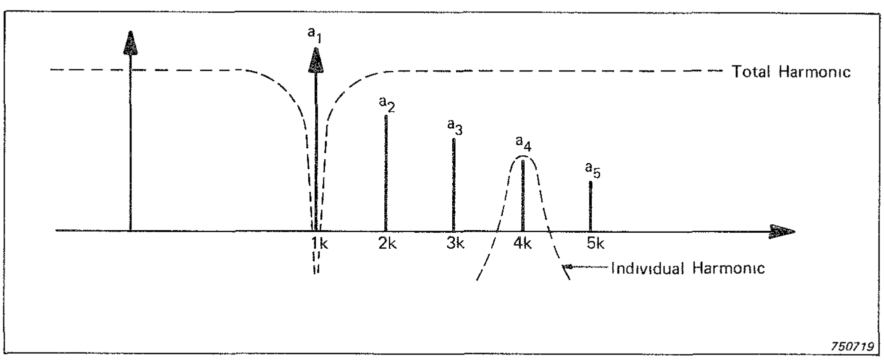


Fig.25. Illustration of harmonic distortion. Band stop filter rejects fundamental for measurement of total harmonic distortion. Band pass filter can measure distortion components individually

troduced in the system which then generates distortion components at 2 kHz (second harmonic), 3 kHz, 4 kHz and so on. The total harmonic distortion for this signal is the RMS sum of the distortion components related to the RMS value of the total output signal.

$$d = \sqrt{\frac{a_2^2 + a_3^2 \dots}{a_1^2 + a_2^2 \dots}} \times 100\%$$

where a_x is the amplitude of the x'th harmonic.

This measurement is readily made by a rejection filter which rejects the fundamental while passing all harmonics (including noise and hum). The output of the rejection filter is then the distortion level, which when plotted on Level Recorder paper will be in dB below the reference level (—20 dB = 10%, —40 dB = 1%, —60 dB = 0,1% etc.).

This measurement as a function of frequency (which is required for example, by DIN 45500) is made by simply adding Heterodyne Slave Filter Type 2020 to the basic instrumentation package mentioned. A typical set-up and result is shown in Fig.26. The filter is connected to the high frequency tuning signals provided by the Sine Generator Type 1023 thus ensuring automatic synchronization

Individual Harmonic Distortion

Although total harmonic distortion measurements are useful from a convenience standpoint, more detailed information, especially of interest to the designer, is available by measuring the amplitude of each of the harmonics separately as a function of frequency. This is simply done by expanding the system by adding Tracking Frequency Multiplier Type 1901 as shown in Fig.27. The 1901 locks onto the low frequency sine output of the 1023 and generates the tuning signals necessary to tune the 2020 to the required harmonic. This harmonic may be selected by a simple thumbwheel setting on the 1901. Even subharmonics can be measured since the 1901 has a resolution in steps of 0,1 — i.e. from the 0,1 to the 99,9th harmonic. This instrumentation set-up has a dynamic range of 70 dB (0,03% distortion) in the band-pass mode, but is restricted to 55 dB (0,2%) in the rejection mode. If greater dynamic range is required, the instrumentation shown in Fig.29 or Fig 32 is suggested

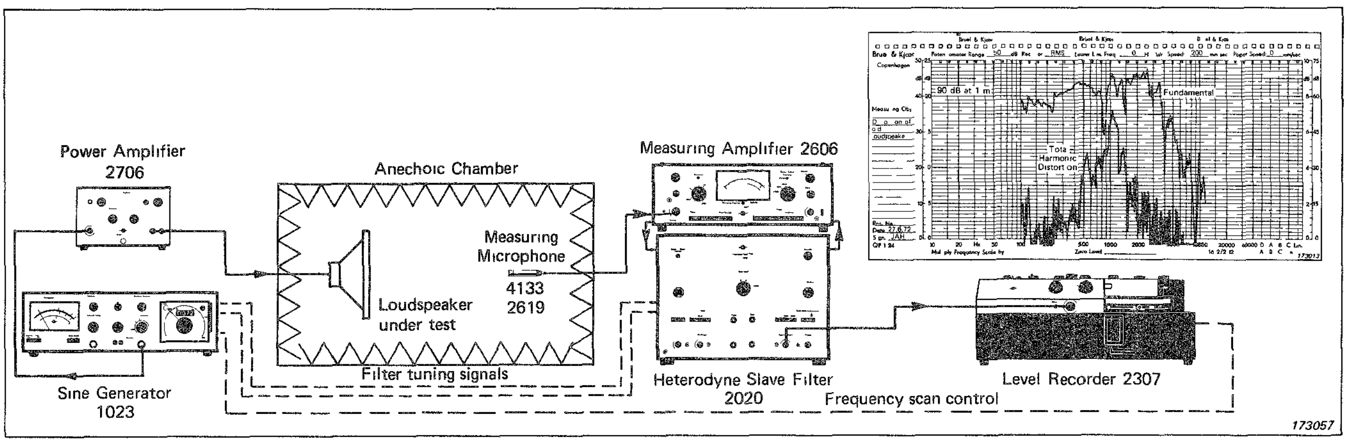


Fig.26. Total harmonic distortion measurement

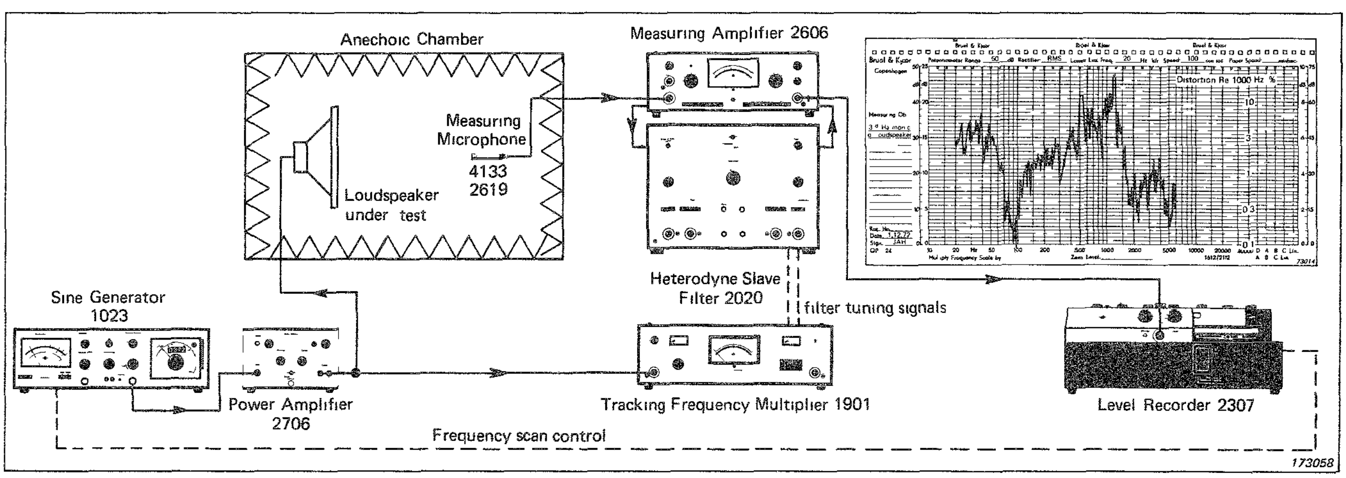


Fig.27. Individual harmonic distortion is measured by adding Tracking Frequency Multiplier 1901 to set-up in Fig 26.

Distortion of Phonograph Pick-ups

Distortion of pick-up cartridges may be measured using a similar set-up. However, instead of the generator, Brüel & Kjær Test Record QR 2009 or QR 2010 is used which contains a sine sweep that can be synchronized with the Level Recorder using Response Test Unit Type 4416. Such a set-up is shown in Fig.28.

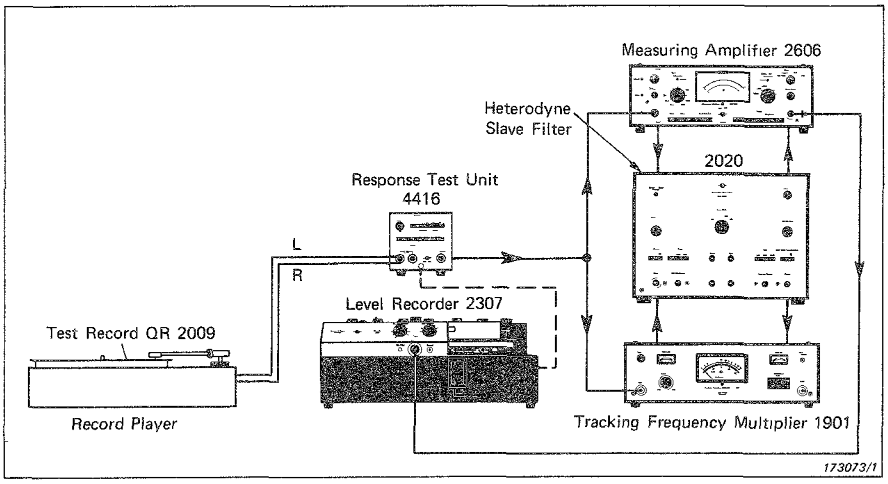


Fig.28. Distortion measurement of pick-ups

Alternate Methods

Harmonic distortion measurements may also be performed using Frequency Analyzers Type 2120 or Type 2121. These analyzers contain constant percentage bandwidth filters of bandwidths of 1%, 3%, 10% and 23% (1/3 octave) plus a band stop filter, high pass and low pass filter. All of these filters are continuously tunable over the 2 Hz to 20 kHz range, although they cannot be synchronized with the generator. Hence measurements can only be made at discrete frequencies. However, for measurements of distortion at very low frequencies (below 20 Hz), these filters are superior to constant bandwidth filters since they have a narrower absolute bandwidth at these frequencies.

Real Time Analyzers Types 3347 and 3348 can also be used to view distortion, and have the advantage that all distortion components are seen simultaneously. The nominal dynamic range of 50 dB (0,3% distortion) of these analyzers can be extended by using a high pass filter (such as that in Type 2120) to attenuate the fundamental.

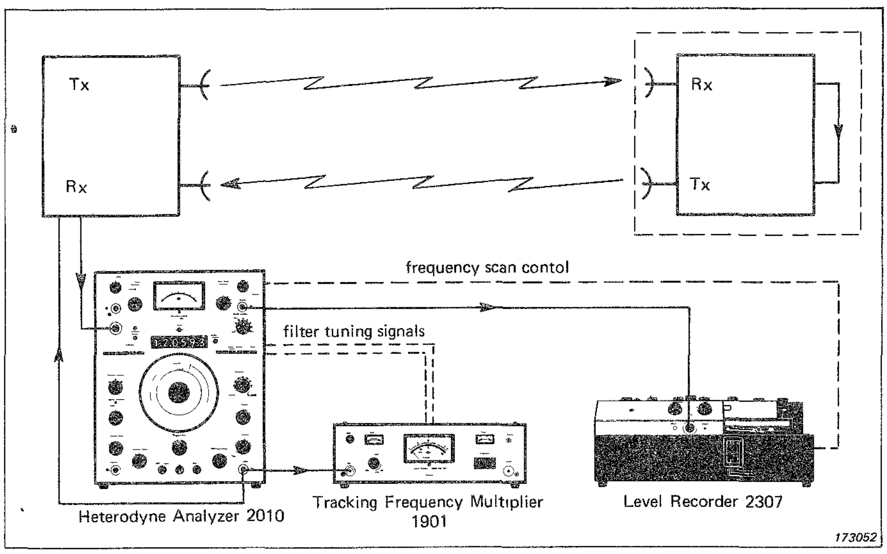


Fig.29. To obtain maximum dynamic range, Heterodyne Analyzer 2010 is used for distortion measurements

Optimum Distortion Measurements

When a wider dynamic and frequency range is required than possible with the previous instrument set-ups, Heterodyne Analyzer Type 2010 can be used to extend the frequency range from 2 Hz to 200 kHz and the dynamic range to 80 dB (0,01%) or better. The 2010 contains both a low distortion oscillator and a narrow band filter. The Filter can be tuned to the chosen distortion component using either Tracking Frequency Multiplier Type 1901 or Distortion Measurement Control Unit Type 1902. With the 1901, only harmonic distortion can be measured while the 1902 permits intermodulation and difference frequency distortion measurements. A typical set-up using the 2010 for distortion measurements on a complete tele-communications link is shown in Fig.29.

8. Difference Frequency and Intermodulation Distortion

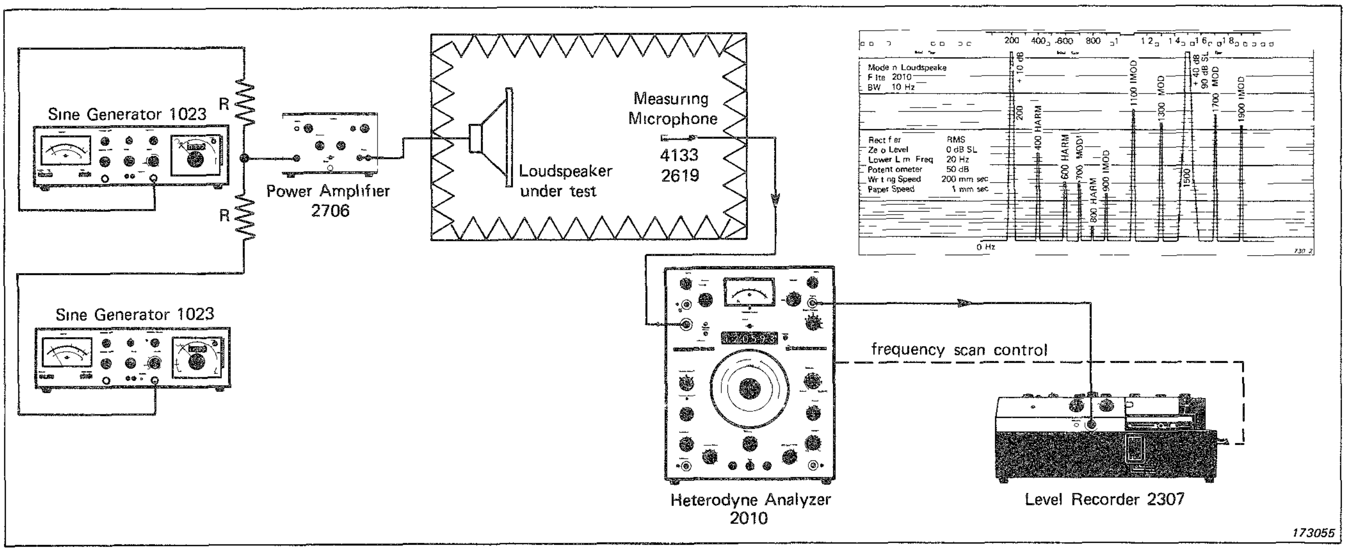


Fig.30. **Intermodulation distortion measurement using two fixed frequency tones**

Discrete Frequencies

When two or more single frequencies are present in a system, the unlinearities of the system will result not only in harmonic distortion, but will also generate sum and difference components called intermodulation distortion. This measurement can be made using two generators tuned to different frequencies and then analyzing the output of the object under test using Heterodyne Analyzer Type 2010. The plot (Fig.30) then shows the various distortion components and their amplitudes. In the figure, test tones of 200 Hz and 1,5 kHz are used. However, this method only yields information about the system at these two frequencies, so many curves of this type need to be run to get a more complete picture.

Swept Measurements

However, by using Distortion Measurement Control Unit Type 1902 with the 2010, a complete system is available that permits distortion measurements while the test frequencies are swept. The system serves as a two-tone generator, thus permitting intermodulation and difference frequency, as well as harmonic distortion measurements. The different capabilities of the 1902/2010 combination are shown in Fig.31 and a typical instrument set-up and distortion curve are shown in Figs. 32 and 33.

In the harmonic mode, a single tone output is provided by the 1902 which is in the 2 Hz to 200 kHz range. The analyzer may be tuned automatically to any harmonic up to the 5th.

In the difference frequency mode, the 1902 generates two frequencies which are swept together with

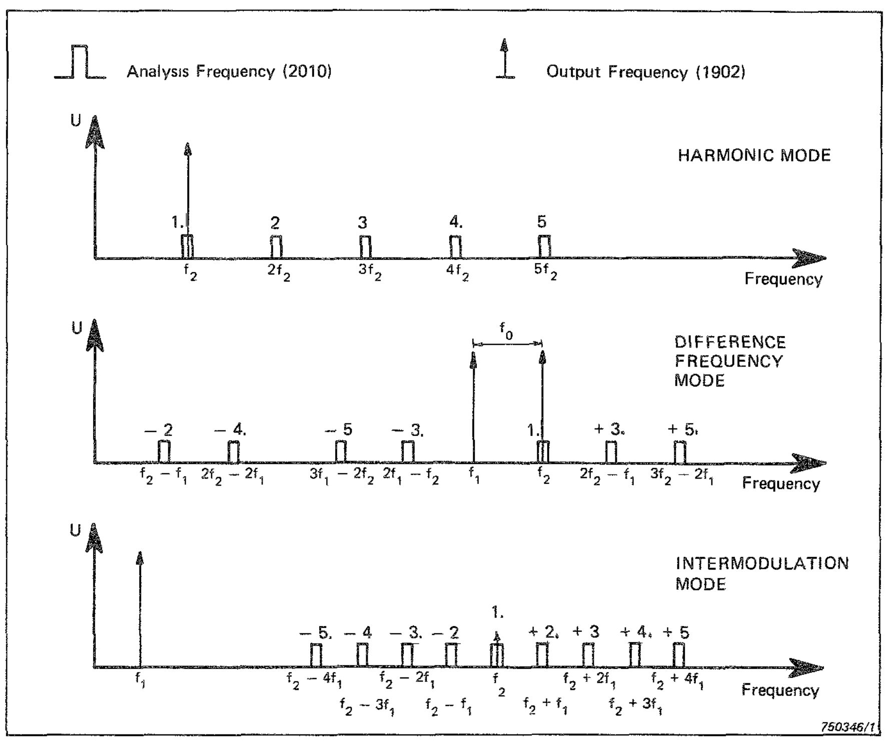


Fig.31. **Relative positions of test frequencies generated by the 1902 and the analyzer frequencies of the 2010 for three types of distortion measurements**

a constant frequency difference between the two. This frequency difference is adjustable from 20 Hz to 2 kHz. The analyzer is then set to track any distortion sidebands up to the fifth order and the result is automatically plotted on the Level Recorder. Fig.33 shows third order minus distortion measurements on a typical high quality tape recorder. It is seen that the distortion level reaches approximately 10% at the higher frequencies where tape saturation occurs. A harmonic distortion measurement would not be as sensitive to this because of the inherent high frequency roll off of the tape recorder which would reduce the level of the harmonics. This distortion is also of great importance audibly since the distortion components (the difference between the two frequencies) fall in the audible range, whereas harmonic distortion components at these frequencies might not be considered important since they fall outside the audible range.

In the intermodulation mode the 1902 also generates two frequencies. However, this time the lower

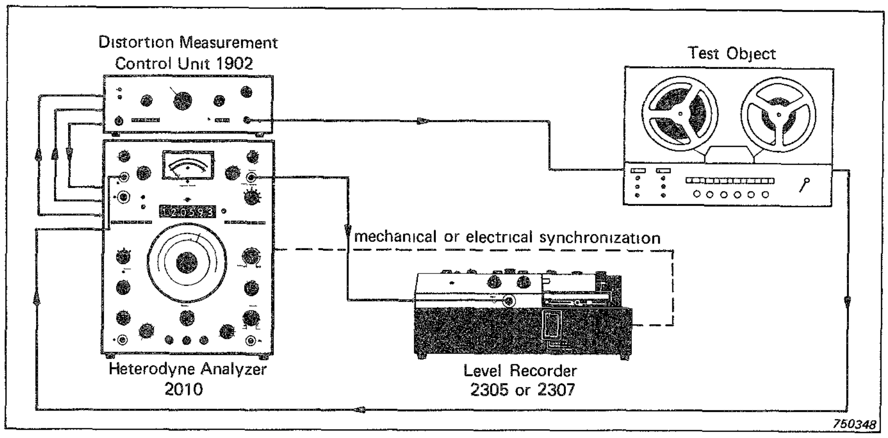


Fig.32. Set-up for distortion measurements using the 1902/2010

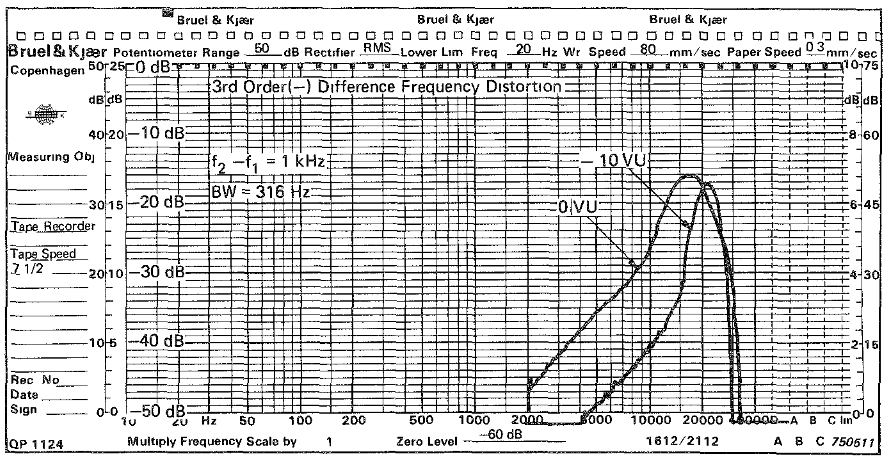
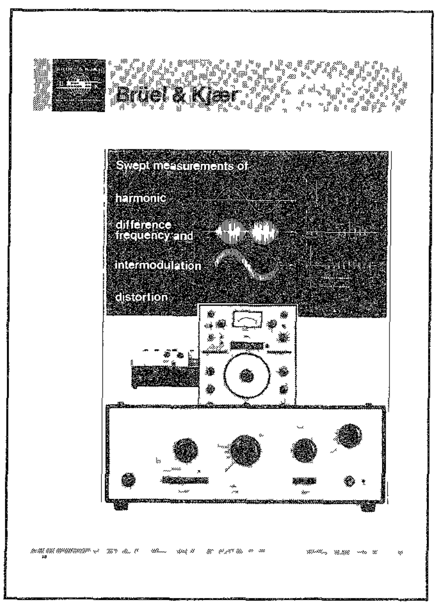


Fig.33. Difference frequency distortion of a tape recorder



frequency is fixed, but selectable from 20 Hz to 2 kHz while the upper is swept and the analyzer is tuned to the plus or minus sidebands of the upper tone. The amplitude ratio of the two tones is 4 to 1 (12 dB) as specified in various standards. However, any other amplitude ratio may also be obtained.

If the difference frequency and intermodulation distortion measurements must be performed on a tape recorder that does not have simultaneous record/reproduce facility, a test tape must first be prepared by recording the twin tone test signal on one channel of the tape while simultaneously recording the higher of the test tones (available at the BFO output of the 2010) on the second channel. This second channel is then used in playback to trigger Tracking Frequency Multiplier Type 1901 which then controls the 1902 and permits automatic tracking of the distortion components.

A similar test can also be devised for test records, although it requires the cutting of a special test record.

9. Transient Intermodulation Distortion (TIM)

Transient intermodulation distortion arises due to the time delay in the feedback loop of an amplifier, and the resulting clipping of the feedback signal. Measurement of the phenomenon is difficult and several different techniques have been suggested. One technique suggests using a broadband pink noise signal with one frequency band missing, for example a 1/3 octave. The receiving side (Fig.34) would then have a band pass filter which would measure the amount of energy in this band, which in a perfect system should be zero. This measurement should be sensitive to harmonic, intermodulation and TIM since the test signal is random noise which might be considered to have a transient-like nature.

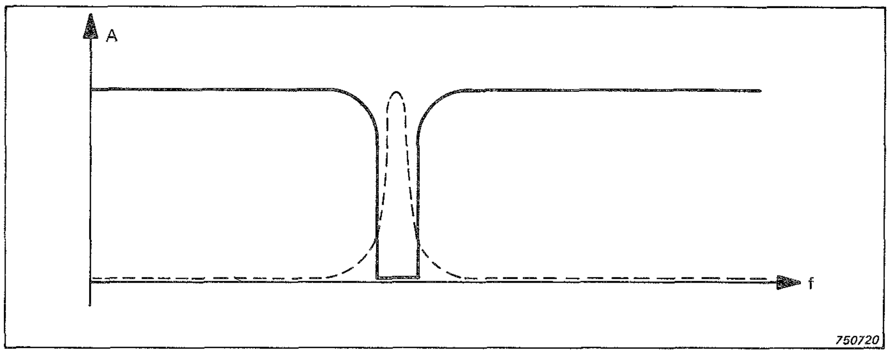


Fig.34. TIM measurement using broad band noise with a missing narrow band

Otala in Finland has indicated that there is a relationship between TIM and audible quality. He points out that TIM depends on the relation of bandwidth of the preamplifier to the bandwidth of the power amplifier, and also the amount of feedback.

Another approach to measuring TIM is to use a low frequency square wave with a high frequency sine wave superimposed. The influence of transient intermodulation distortion can then be seen by examining the frequency spectra. If desired, this measurement can also be performed using the 1902/2010 combination simply by adding a Schmitt trigger to the 1902 output to convert f_1 to a square wave.

The disadvantages of all these methods is that none of them yield results in numbers, or well-defined distortion characteristics. However, a twin tone test using the 1902 and 2010 gives a well defined characteristic and is also related to TIM. This is because TIM is primarily a high frequency phenomenon related to the slew rate of amplifiers and their bandwidths. Traditional intermodulation distortion measurements will not disclose these problems since they only cover the audible frequency range. However, with the 1902/2010 system, swept intermodulation measurements can be made up to 200 kHz and it is at the frequencies above 20 kHz that many amplifiers show a dramatic rise in distortion due to their restricted slew rates etc. Hence the use of high frequency swept intermodulation measurements is suggested as a standardized measure of TIM.

From Fourier analysis we know that if the performance of the system at all frequencies is defined, then its transient performance is defined. Unfortunately the Fourier theory is only applicable for linear systems, therefore not for distortion.

High frequency distortion measurements are not the complete answer to TIM but it seems in practice to be the best indication. Thus a problem with many current attempts at TIM measurements is that a wide enough frequency range has not been explored.

As indicated in Fig.35, the steady-state distortion of an amplifier with amplification A and feedback β is reduced by $1/(1-\beta A)$. But, what will happen if a transient or a step function is introduced? Then, the more feedback the greater the transient distortion. This is illustrated Fig.35 where amplifier (a) typically will have less feedback, hence higher steady state distortion but lower transient distortion than amplifier (b).

More details on this method of distortion measurement is given in the application note 15—098 "Swept Measurement of harmonic, difference frequency, and intermodulation distortion".

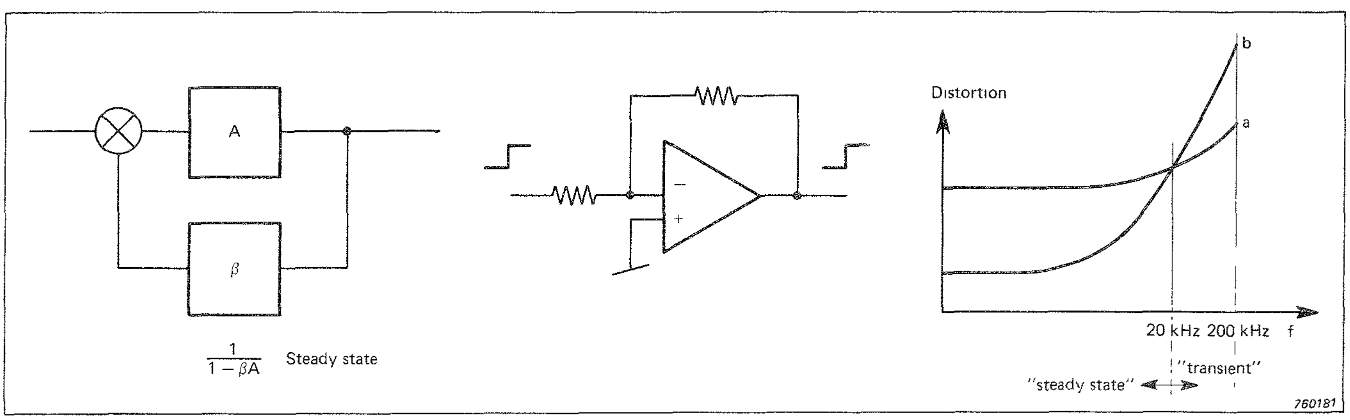


Fig.35. Increased feedback gives lower steady state distortion but higher transient distortion

10. Impulse Response

For investigations of impulse response, the Digital Event Recorder Type 7502 is an ideal instrument. The digital memory holds up to 10000 samples, each of 8 bits. A signal captured in this memory can be reproduced indefinitely and may be transformed up or down in time to fit the requirements of oscilloscopes, external analyzers, or computers. The set-up shown (Fig.36) is used to record the response of a loudspeaker when excited by a single, squared sine pulse.

The advantage of a squared sine pulse is that it contains the entire frequency spectrum, and if suitable processing facilities are available, the frequency and phase response of the loudspeaker can be calculated using the Fast Fourier Transform. The pulse may also be viewed on an oscilloscope for subjective evaluation.

As indicated in Fig.36, the Gating System 4440 might be used for this application since it, with a narrow pulse width, will produce one cycle. By using a diode in series with the output, a $\sin^2 x$ function is approximated. Since the burst width can be as ahort as 0,1 ms it will imply pulses with a frequency content up to 10 kHz.

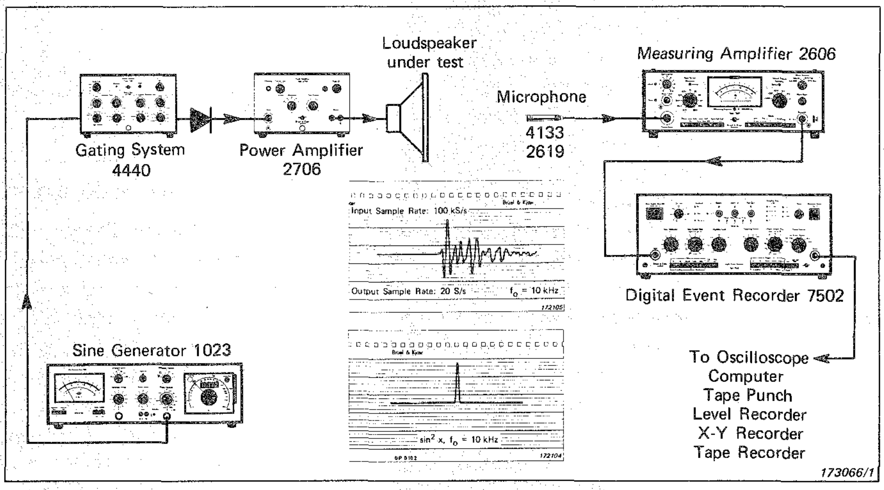


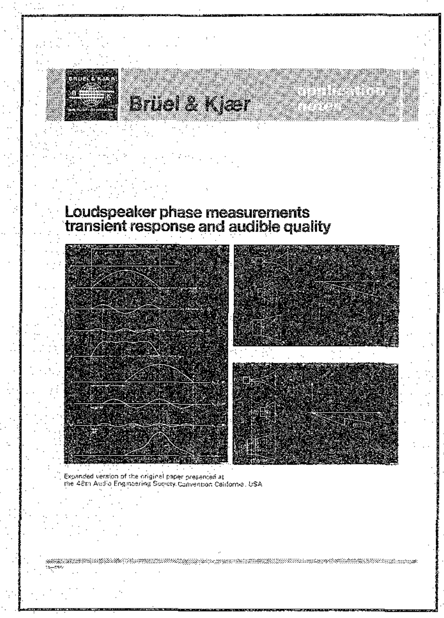
Fig.36. Set-up for impulse response measurements

11. Phase Response and Related Transient Response

Basic Principles

The most complete way of characterizing the transient response of a system is to measure its phase and amplitude response. Then its entire transfer function is known and the transient response is defined. The more linear the phase response, the better the transient reproduction, which especially is noticeable on musical transients such as the attacks of notes, the various percussion instrument and string pizzicato.

Phase response measurements are outlined in great detail in the Brüel & Kjær Application Note No. 15—090 "Loudspeaker Phase Measurements, transient response and audible quality". Therefore, only the main points will briefly be summarized here.



Loudspeaker phase measurements are diffucult to make because the time delay between loudspeaker and microphone results in a large continually increasing phase change. However, this problem is overcome by introducing a corresponding delay in the reference channel of the Phase Meter using Phase Delay Unit Type 6202. The 6202 corrects both for the time delay and for the phase response error of Heterodyne Slave Filter Type 2020, if this filter is used. A typical instrument set-up for phase measurements is shown in Fig.37. It is seen that this builds on the same instruments previously used with the only addition being the Phase Meter Type 2971 and the Phase Delay Unit Type 6202.

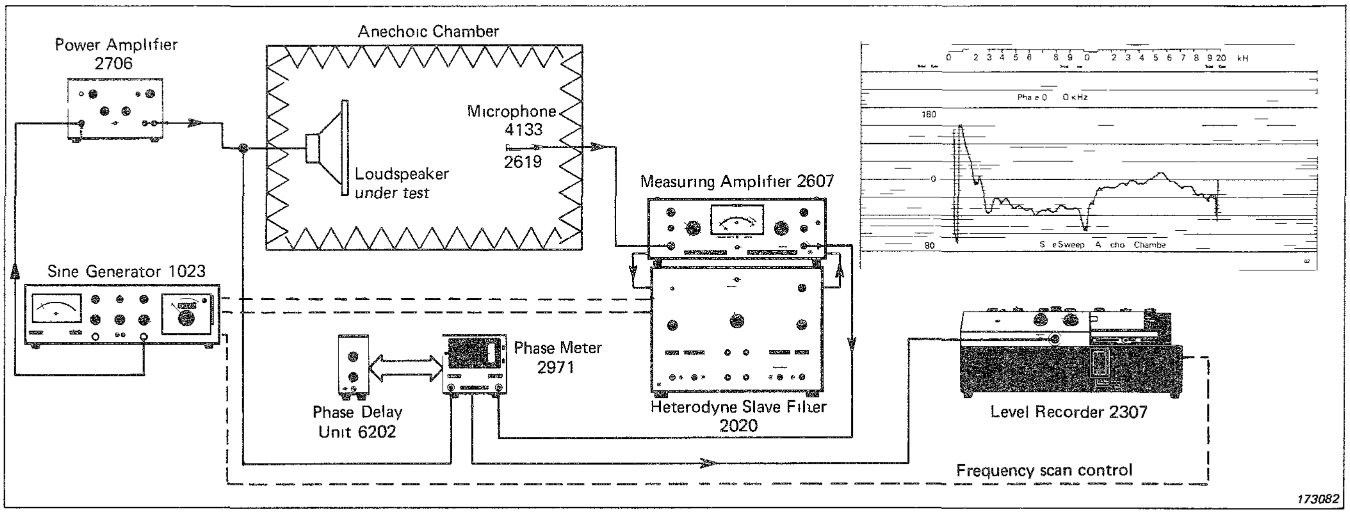


Fig 37 Set-up for loudspeaker phase measurements

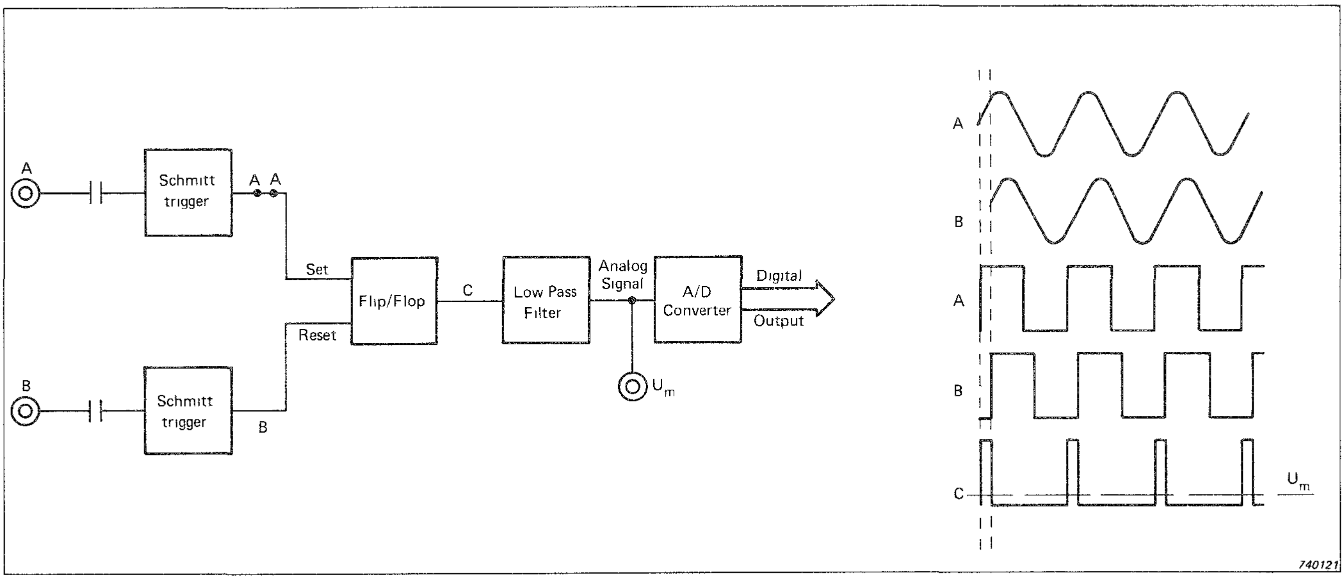


Fig 38 Simplified block diagram of Phase Meter

The principle of operation of the Phase Meter is shown in Fig 38 From the diagram it is seen that the length of time that the flip-flop is set is directly related to the phase angle When the output of the flip-flop is low pass filtered, the remaining DC component will be proportional to the duty cycle, and hence the phase angle The Phase Delay Unit Type 6202 (Fig 39) is seen to consists of two separate sections six shift registers which can be clocked internally or externally to determine the delay time, and a low pass filter to compensate for the phase response of Heterodyne Slave Filter Type 2020

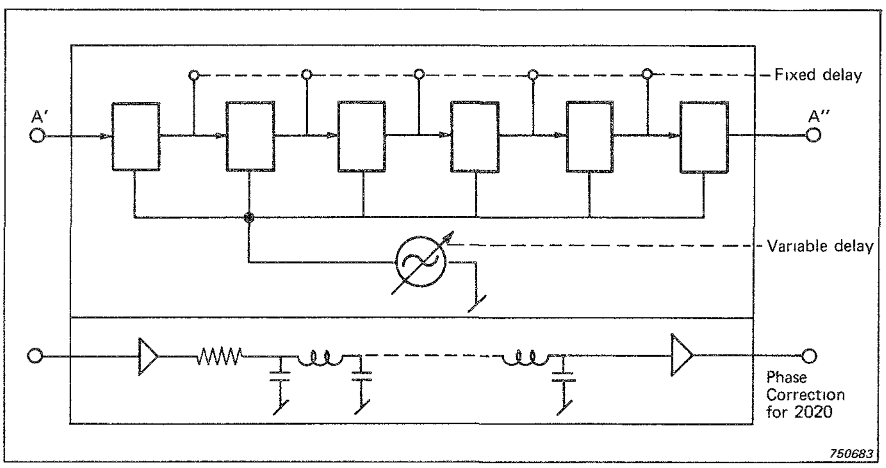
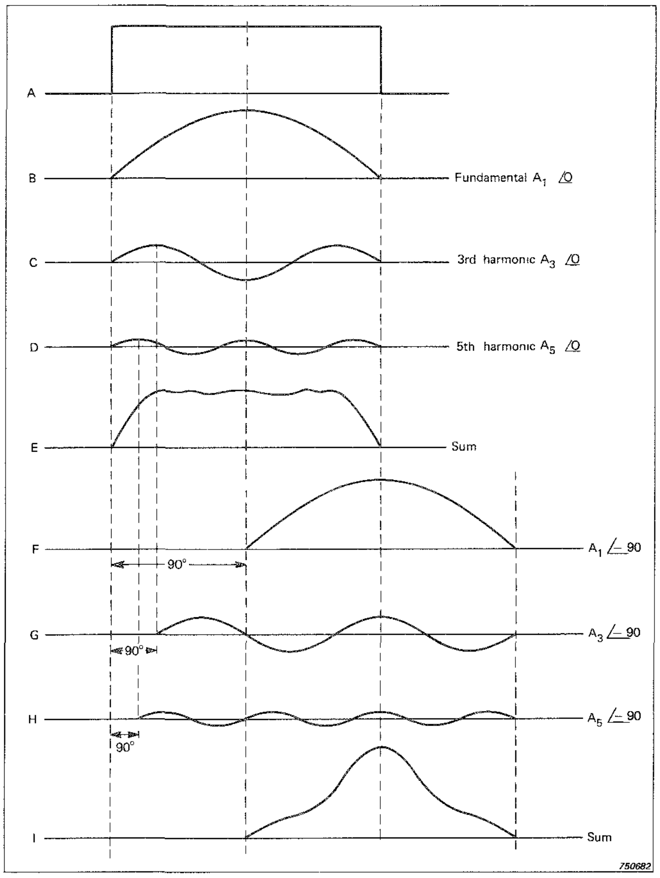


Fig 39 Principle of Phase Delay Unit Type 6202



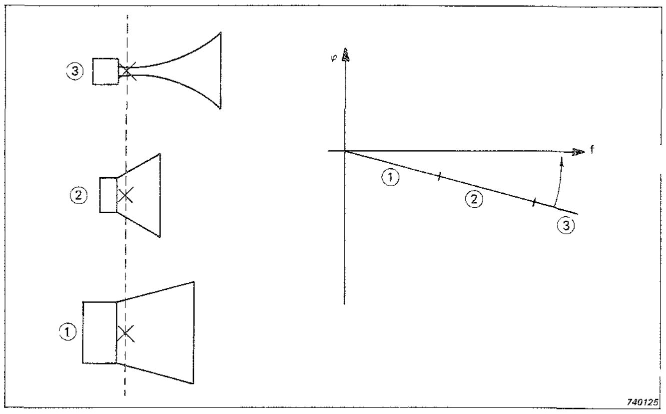


Fig.40. Repositioning individual units gives flat phase response

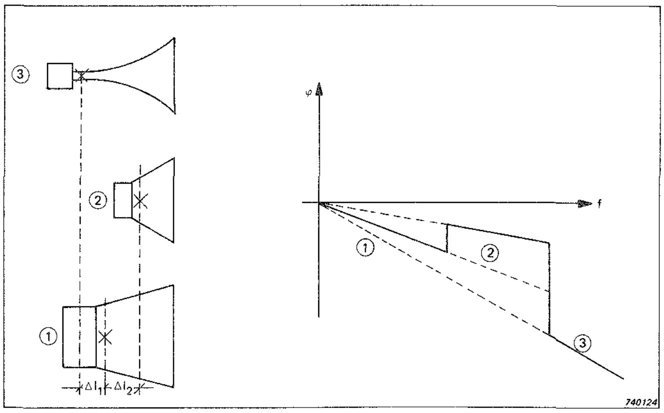


Fig.41 Non-ideal phase response for "normal" positions of speaker units

Practical Example

Fig.41 shows a typical three-way loudspeaker system in which each of the individual drivers is assumed to have linear phase response. The acoustic centers of the drivers are indicated by an x and are assumed to be independent of frequency. The resulting phase response curve clearly indicates the relative offset of the acoustic centers of the three drivers. From this curve, it can be calculated how much to reposition the drivers to obtain flat phase response as shown in Fig.40. With the acoustic centers of the speakers in line, the transient performance will be improved since the respective frequency components will now arrive at the listener with the proper time relationship.

Phase response seems especially relevant to audible quality at the higher frequencies, while third octave response seems more relevant at low to mid frequencies. Phase response at the very low frequencies near the speaker cut-off frequency will also reveal its low frequency transient response, which if good, results in a tight sound, or if poor, gives a muddy, and poorly defined bass.

Systems with Long Delay

Since the Phase Delay Unit Type 6202 may be used with an external clock it can be used for phase measurements of systems with delays up to approximately 200 ms, depending on the maximum frequency required. Thus it can be used for measurements of audio delay lines, be they of digital, acoustic, or analog bucket brigade design. The technique can also be applied to tape recorders (Fig.42). However, wow and flutter may make the phase reading unstable at higher frequencies.

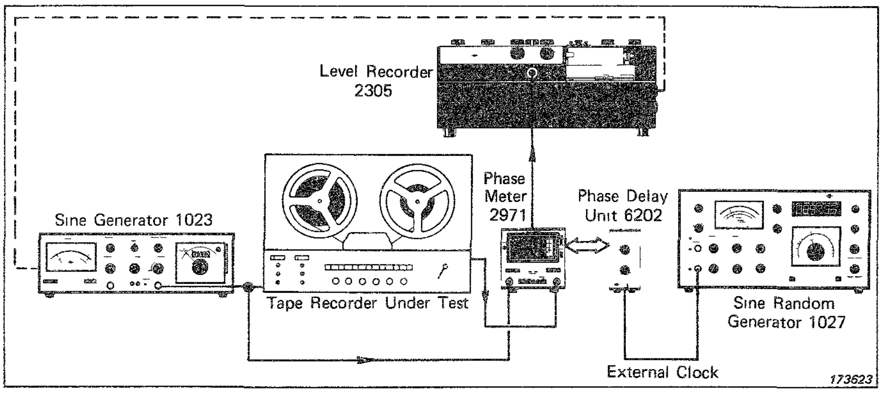


Fig.42. Phase response measurements on tape recorders

12. Mechanical Stability of Tape Recorders and Turntables

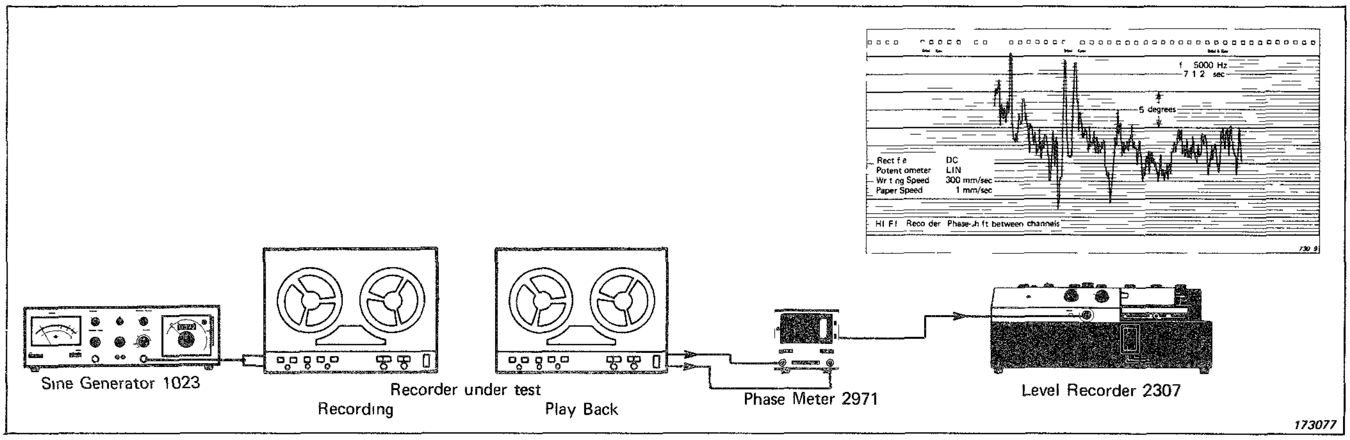


Fig.43. Phase measurements between channels of tape recorder

The Phase Meter Type 2971 may also be used as a very sensitive test of the mechanical stability of mechanical reproducers. If for example, the tape skews slightly, this will show up in the phase between two channels of a tape recorder. An instrument set-up for this measurement and a resulting curve is shown in Fig.43. The curve indicates the phase variations as a function of time.

The same measurment may also be made between the two channels of a stereo record. This is especially important for the socalled "matrix" four-channel systems (such as the SQ and QS systems) where the four-channel information is encoded both as amplitude and phase variations. In addition, the phase between channels is of great importance in all multi-channel recording, reproducing, and transmitting links to ensure monaural compatability and a precise stereo image.

The overall stability and wow and flutter of a tape recorder can also be evaluated by comparing the input and output phase of the tape recorder (Fig.44). Any speed variations or tape stretching will then be seen as a phase change. The Phase Delay unit is not necessary for this test, since only the relative phase is of interest. Therefore the delay can be compensated for by setting the frequency of the generator to give a pen deflection around mid-paper. It is important to note that the higher the test frequency, the more sensitive will be the test of mechanical stability.

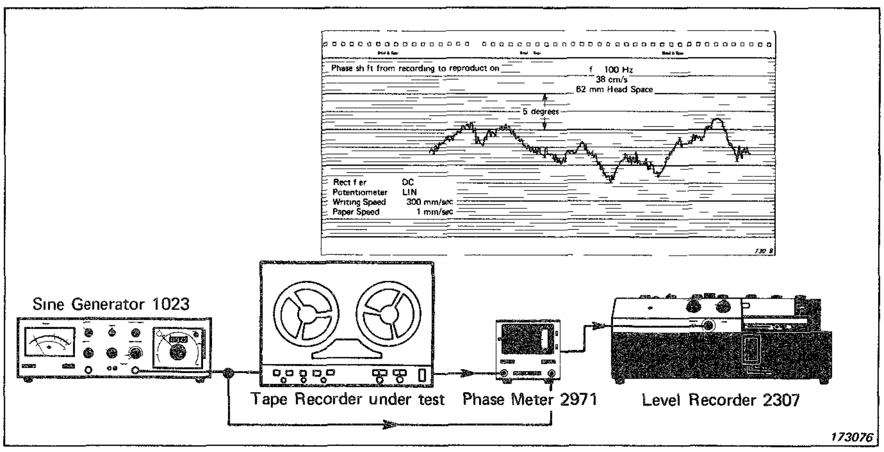


Fig.44. Phase measurement of tape recorder to evaluate mechanical stability

13. Complex Impedance

To measure impedance of a loudspeaker requires a generator which is constant current source This may readily be accomplished with the Beat Frequency Oscillator by measuring the voltage across a resistor placed in series with the loudspeaker and feeding this voltage back to the Compressor Input of the oscillator (Fig.45) This voltage across the resistor will then be held constant giving a constant current to the loudspeaker. The voltage across the loudspeaker will then be directly proportional to the impedance. The results of such a measurement are shown in Fig.46.

However, impedance is a complex quantity and hence also requires the measurement of the phase angle. This may be done using the Phase Meter 2971 in the set-up indicated in Fig.47.

The complex impedance can be described as the vector $|Z| \angle\phi$ from the origin (0,0) to a point on the circle (Fig.48). The point closest to the origin corresponds to a frequency of 0 Hz (DC) where the amplitude is minimum and the phase is zero. When passing clockwise around the circle it is seen that both the amplitude $|Z|$ and the phase angle $\angle\phi$ increase until the vector is tangent to the circle. The magnitude $|Z|$ then continues to increase until resonance is reached where the impedance is purely resistive since the phase angle is zero. Above resonance, the magnitude decreases and the vector swings around the bottom of the circle until it reaches a point close to the DC value.

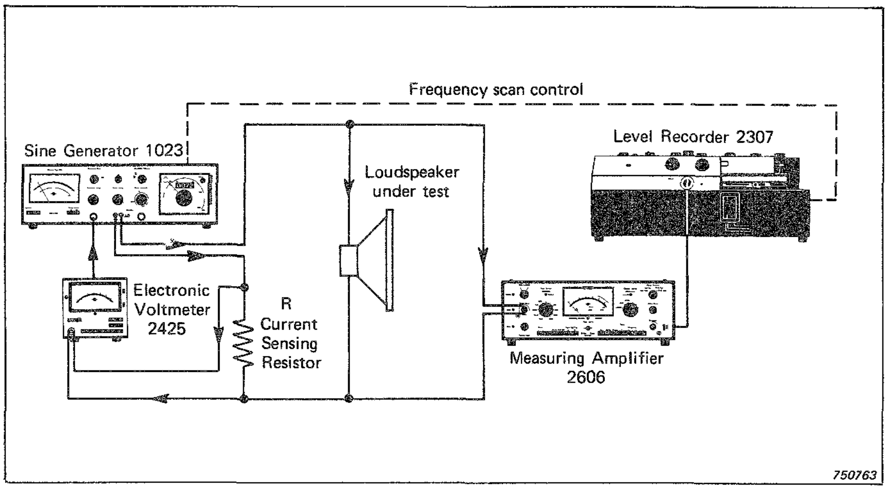


Fig.45. Use of Beat Frequency Oscillator as constant current generator

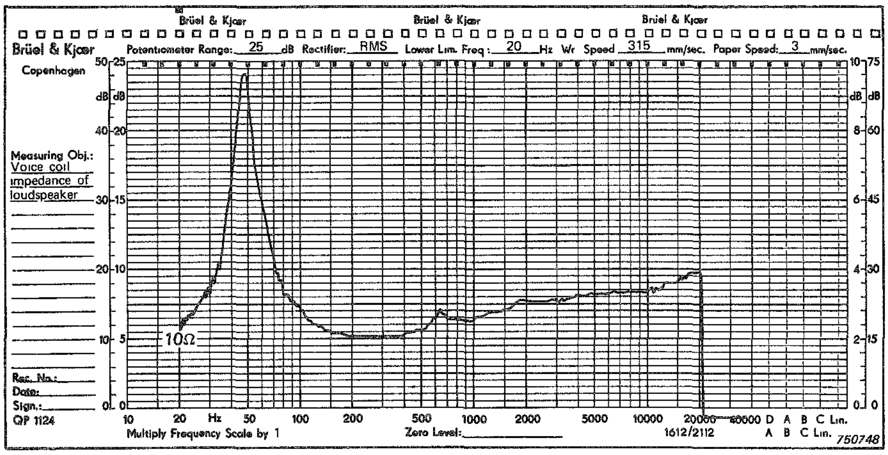


Fig.46. Typical loudspeaker impedance curve

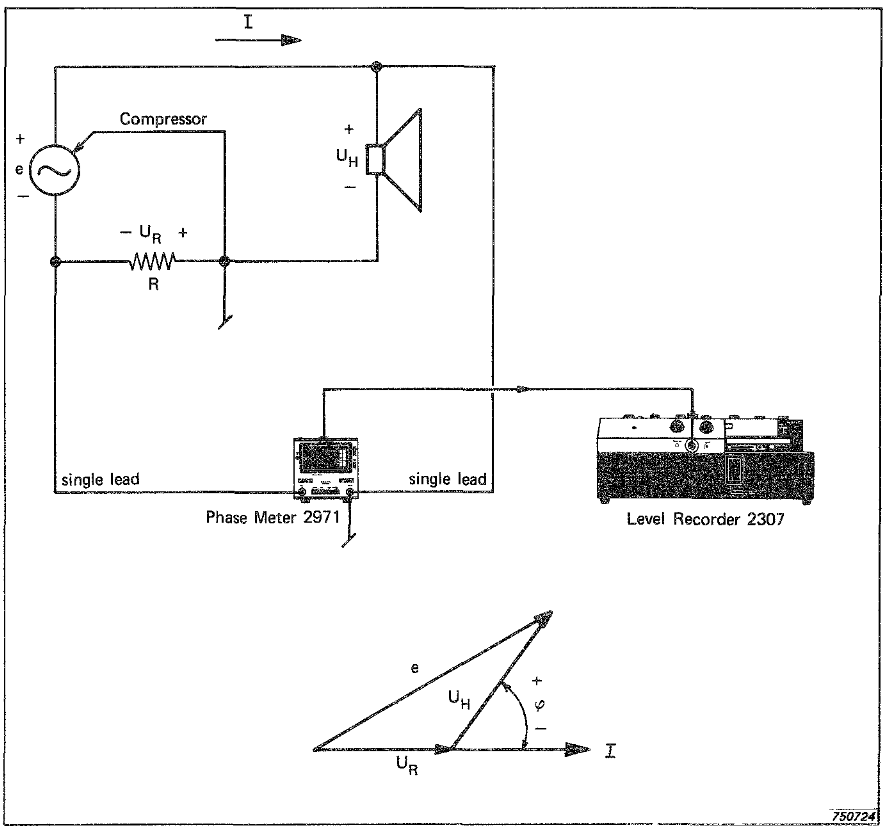


Fig.47. Set-up for measuring phase component of impedance

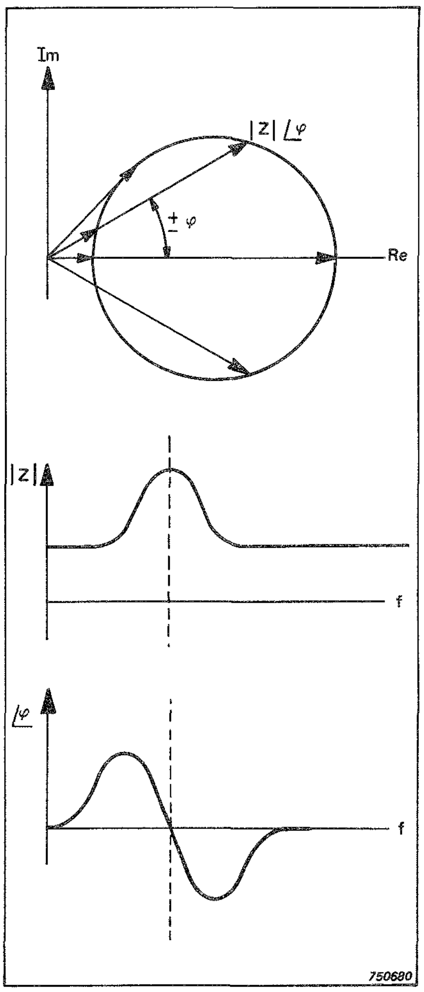
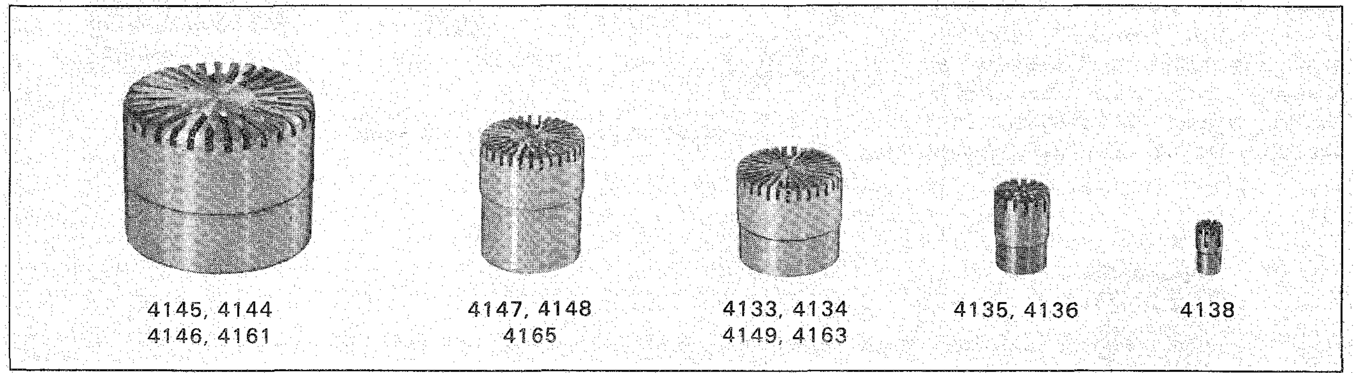


Fig.48. Vector representation of complex impedance

14. Instrumentation Microphones for Music Recording



A striking example of the importance of phase response is seen when comparing the audible quality of Brüel & Kjær 1" and 1/2" condenser microphones. The two microphones both have flat response in the audible region, but the only difference is the phase response, which is far superior for the half-inch microphone. The half-inch microphone (Type 4133) has a 90° phase shift (indicating resonance) at 25 kHz, well outside the audible region. However, the one-inch microphone (Type 4145) has a resonance frequency of only 10 kHz, still inside the audible range. When comparing the two microphones on transient filled material, the 4133 (half-inch) gives an audible superior transient reproduction — a more clean and transparent sound. This difference in audible quality is most likely due to the difference in phase response of the microphones.

In addition, the microphones are eminently suited for music recording in all respects, with performance unequalled by any other omnidirectional microphone commercially available. The one-inch microphones provide state-of-the-art low noise performance. This microphone has a noise floor of only 10 dB(A) while still being able to handle a maximum SPL of 148 dB. However, for the majority of applications, a higher noise level will make no difference since the recorded level is so high. In these cases, the half-inch microphone offers the best choice due to its superior transient reproduction. Still its noise floor is only about 25 dB(A) and the maximum SPL is 160 dB — thus ensuring that all transients and peaks will be reproduced distortion free. When studio microphones are placed near or inside musical instruments, peak sound levels begin to approach this figure, and overload of most microphones occurs on peaks. The ample overload margin of the 4133 assures that this will not occur.

The free-field response of various Brüel & Kjær microphones is shown in Fig.49. The sensitivity differences between the microphones is also seen.

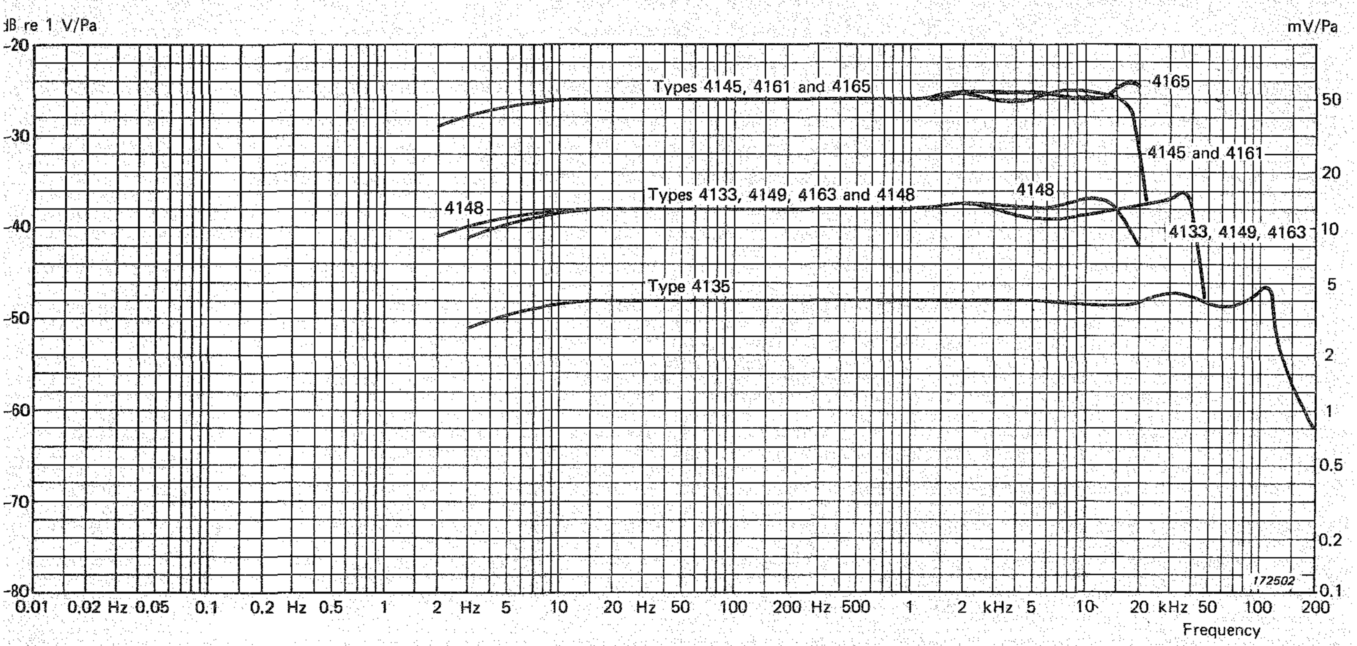


Fig.49. Frequency response of Brüel & Kjær free-field microphones

15. Test Records

Brüel & Kjær produces three different test records whose main features are indicated in Fig.50. With these records, frequency response, wow and flutter, rumble and other measurements may be made. Fig.51 shows the program contained on record QR 2009. To synchronize the test records with the Level Recorder for automatic plotting of frequency response, the Response Test Unit is required. A 1 kHz tone at the beginning of the test sweep on the record is detected by the Response Test Unit which then starts the Level Recorder (Fig.52).

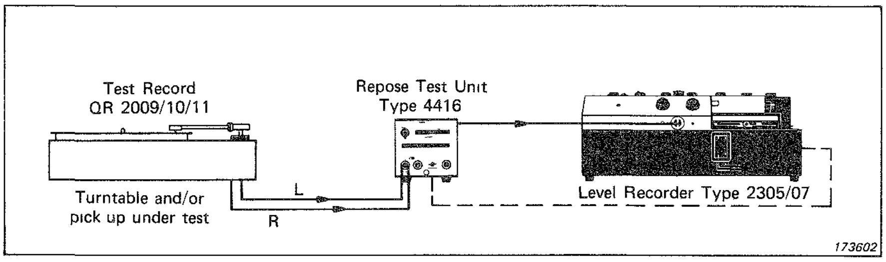


Fig.52. Frequency response measurements using test record

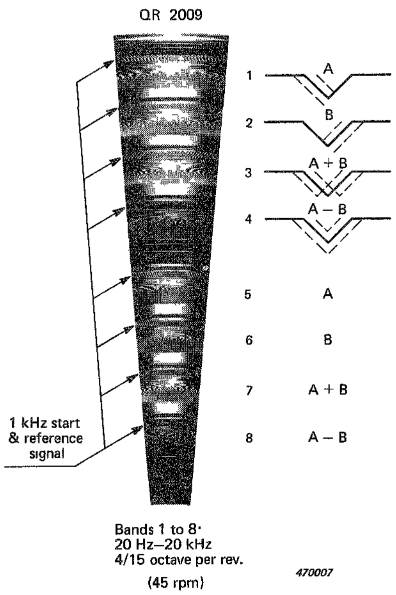


Fig.51. Test record QR 2009

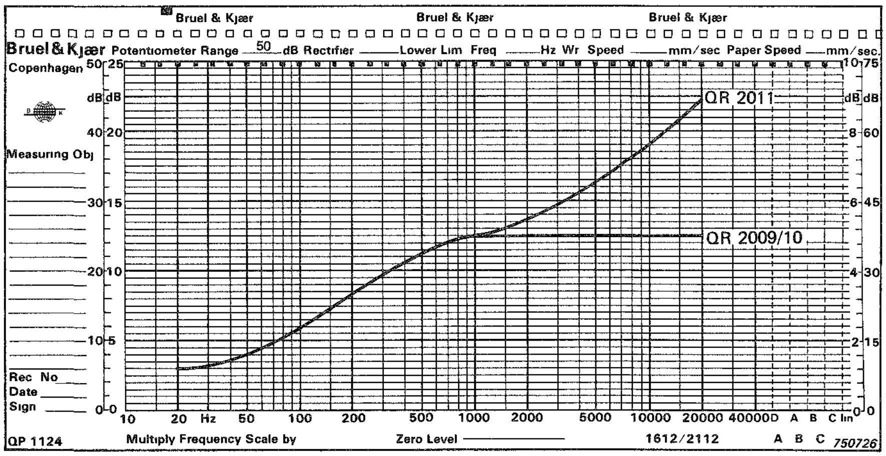


Fig.53. Recording characteristics of test records

All three records are recorded according to IEC Recommendation 98. However, QR 2009 and QR 2010 are recorded with constant velocity above 1 kHz to prevent too high accelerations at high frequencies and to avoid too low a reference level. This is compensated for in the Response Test Unit. However, QR 2011, is recorded with the full normal RIAA correction above 1 kHz because this record is designed for use on normal hi-fi equipment available in a typical listening room. This may result in some mistracking at the higher test frequencies. The various recording characteristics are shown in Fig.53.

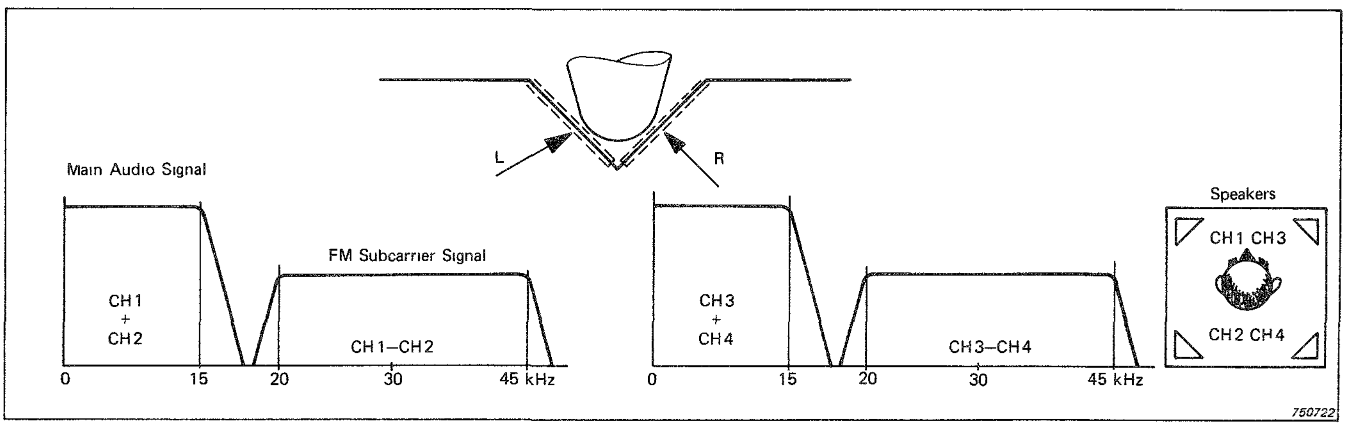


Fig.54. Principle of the CD-4 system

	Band No	Type of Modulation	Level*)	Accuracy	Frequency	Duration	Sync Burst	Remarks	Applications
QR 2009	1 & 5	Left							
	2 & 6	Right	−10 dB	± 1 dB	20 Hz to 20 kHz log sweep	16,7 s/dec, total 50 s	Yes		Frequency response, cross-talk and balance measurements
	3 & 7	L + R							
	4 & 8	L — R							
QR 2010	1	Left	−10 dB	± 1 dB	20 Hz to 45 kHz log sweep	5 s/dec total 16,7 s	Yes		Frequency response and cross-talk measurements
	2	Right							
	3	L + R	+ 8 dB					Distortion < 4%	
	4	L + R	+ 6 dB					Distortion < 3%	
	5	L + R	+ 4 dB	± 0,5 dB	1 kHz	15 s each	No	Distortion < 1%	Determination of maximum tracking ability
	6	L + R	+ 2 dB					Distortion < 1%	
	7	L + R	0 dB					Distortion < 1%	
	8	L+ R	—12 dB	± 1 dB	3150 Hz	60 s	No	Max Wow < ±0 06% (peak-weighted)	Wow measurements on turntables
	9	Left				3 s			
		Right				3 s			
		L + R	0 dB	± 0,5 dB	1 kHz	1 s	No		Quick polarity check
		L — R				1 s			
		L + R				1 s			
	10	Left	—20 dB						
		Right	—10 dB	± 2 dB	30 kHz	5 s each	No	Crosstalk < 20 dB	Crosstalk measurements at 30 kHz in 4-channel transducers
		Right	—20 dB						
		Left	—10 dB						
	11	L + R	—11,3 dB	± 1 dB	315 Hz	15 s	No	Rumble according to IEC-method A Weighting < 50 dB B-Weighting < 65 dB	Rumble measurements on turntables
		Empty				60 s	No		
	12	Left	—20 dB		1 kHz	3 s			
		Right	0 dB		1 kHz	2 s			
		Right	0 dB	± 1 dB	400 Hz to 10 kHz log sweep	5 s/dec total 7 s	No	Crosstalk < 20 dB < 30 dB at 1 kHz	Crosstalk measurement requiring only an AC-voltmeter
		Right	—20 dB		1 kHz	3 s			
		Left	0 dB		1 kHz	2 s			
		Left	0 dB		400 Hz to 10 kHz log sweep	5 s/dec total 7 s			
	13	Left	—10 dB	± 1 dB	20 Hz to 20 kHz log sweep	5 s/dec total 15 s	Yes		Measurement of response and crosstalk at small mechanical wavelengths
	14	Right	—10 dB						
	15	L + R	—20 dB	± 2 dB	5 Hz to 20 Hz log sweep	50 s/dec total 30 s	Yes	Constant velocity	Investigations of arm resonances
QR 2011A	1	Left			1 kHz, 1/3 oct Noise	60 s	No	At the beginning of each band a voice states centre frequency	Calibration
	2	Left	—22 dB	± 1 dB	20 Hz to 20 kHz, 1/3 oct Noise	500 s	Yes		Manual response measurement in listening room
	3	Right			1 kHz, 1/3 oct Noise	60 s	No	At the beginning of each band a voice states centre frequency	Calibration
	4	Right			20 Hz to 20 kHz, 1/3 oct Noise	500 s	Yes		Manual response measurement in listening room
QR 2011B	1	L + R	—22 dB	± 1 dB	1 kHz 1/3 oct Noise	60 s	No	At the beginning of each band a voice states centre frequency	Calibration
	2	L + R			20 Hz to 20 kHz, 1/3 oct Noise	500 s	Yes		Manual response measurement in listening room
	3	L + R	—22 dB	± 1 dB	20 Hz to 20 kHz 1/3 oct Noise	150 s	Yes	No voice comments	Automatic response measurement in listening room
	4	L + R	—24 dB	± 1 dB	20 Hz to 20 kHz, Wideband Noise	30 s each	No	Correct phase	Phase check of the entire system
		L — R						Reversed phase	
	5	L + R			20 Hz to 1 kHz, Noise				
		L + R	—22 dB	± 1 dB	1 kHz to 4 kHz, Noise	15 s each	No		Individual phase check
		L + R			4 kHz to 20 kHz, Noise				
	6	L + R	—10 dB	± 1 dB	20 Hz to 1 kHz, Sine	85 s	Yes		Tracing resonating parts
	7	L + R	—24 dB	± 1 dB	20 Hz to 20 kHz, Wideband Noise	240 s	No		Room distribution of wide range signal

*) Relative to 10 cm/s RMS, Lateral (L + R) at 1000 Hz

Fig.50. Specifications of B & K test records

Frequency Response of CD-4 Systems

Test record QR 2010 contains a sine sweep from 20 Hz to 45 kHz which makes it ideal for tests of systems for reproduction of four-channel sound encoded in a high frequency sub-carrier (Fig.54). Such a typical pick-up frequency response is shown in Fig.55.

Tracks on this test record are also provided where only one channel is swept at a time, thus permitting channel separation measurements.

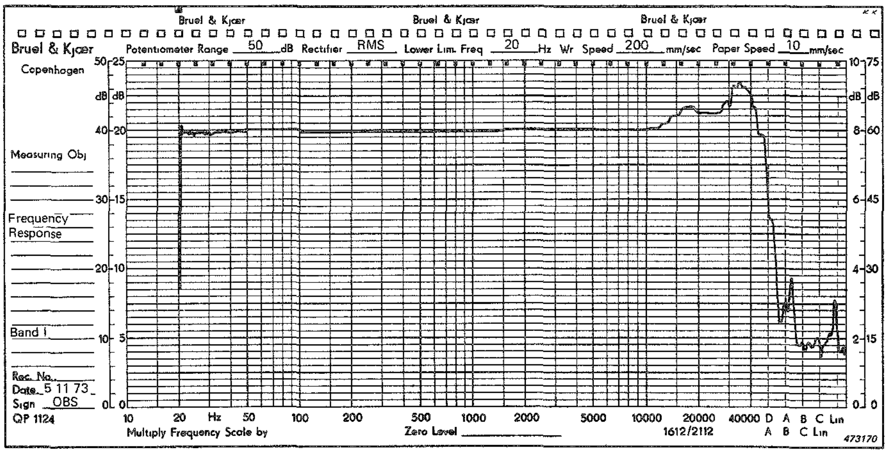


Fig.55. Frequency response of a pick-up suited for CD-4

Trackability

To determine the maximum tracking ability of a pick-up, test record QR 2010 contains a track with a 1 kHz signal recorded at very high levels from 0 dB up to + 8 dB (ref. 10 cm/s RMS) in steps of 2 dB. By monitoring the signal on an oscilloscope or by a frequency analyzer, the maximum highest level without mistracking can be determined (Fig.56).

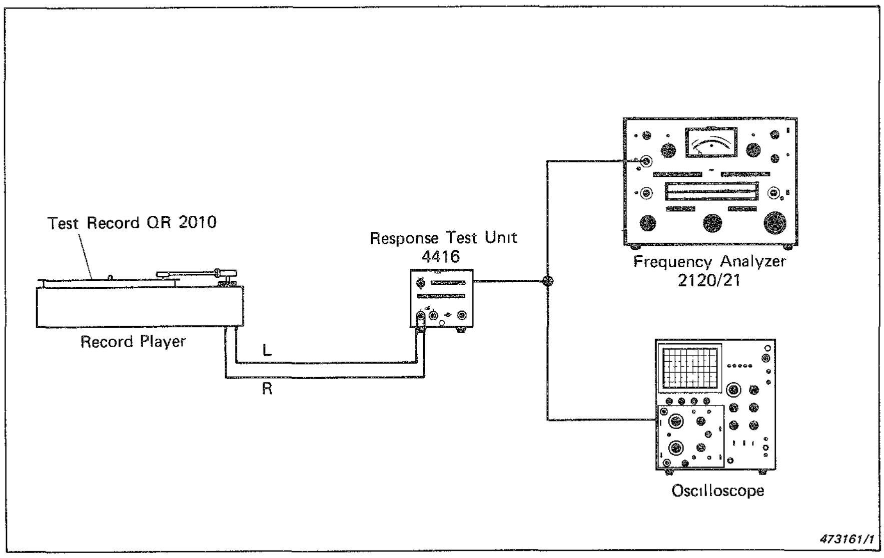


Fig.56. Monitoring distortion level to check trackability

Wow and Flutter

Small speed variations are termed wow and flutter. Variations below 10 Hz are called wow, and above 10 Hz, flutter. This can be measured as the frequency modulation of a 3,15 kHz signal recorded on test record QR 2010 by use of a flutter meter (not yet available from B & K) (Fig.57).

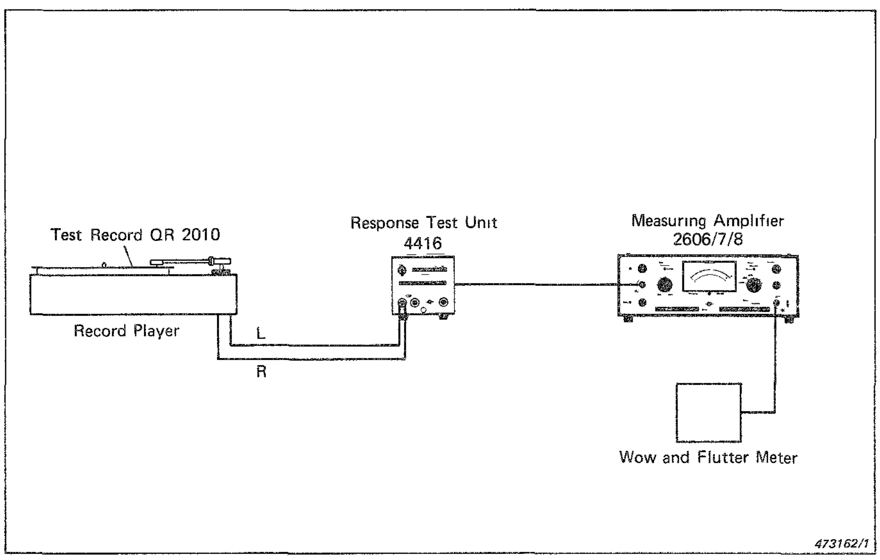


Fig.57. Wow and flutter measurements

Rumble

Rumble is a signal to noise measurement using either the A or B filters shown in Fig.58 Track 11 of test record QR 2010 contains a 15 second 315 Hz reference tone followed by a 60 second silent groove for rumble measurements. The A and B weighting curves are built into the Response Test Unit Type 4416. A typical instrument set-up is shown in Fig 58.

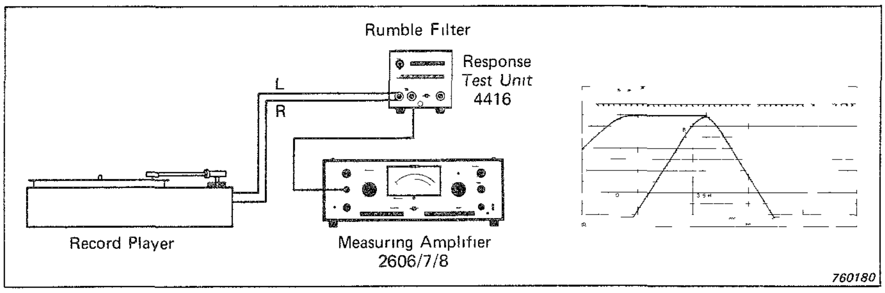


Fig.58. Set-up for rumble measurements

Tone Arm Resonance

Test Record QR 2010 can also be used to measure tone arm resonances. These resonances can be so large that the movement of the arm can sometimes be seen by the naked eye. The measurement is made using track 15 which contains a slow sweep from 5 Hz to 20 Hz. The output of the pick-up cartridge is recorded on the Level Recorder which clearly shows the resonance frequency (Fig.59) Note that the frequency scale is multiplied by 0,1

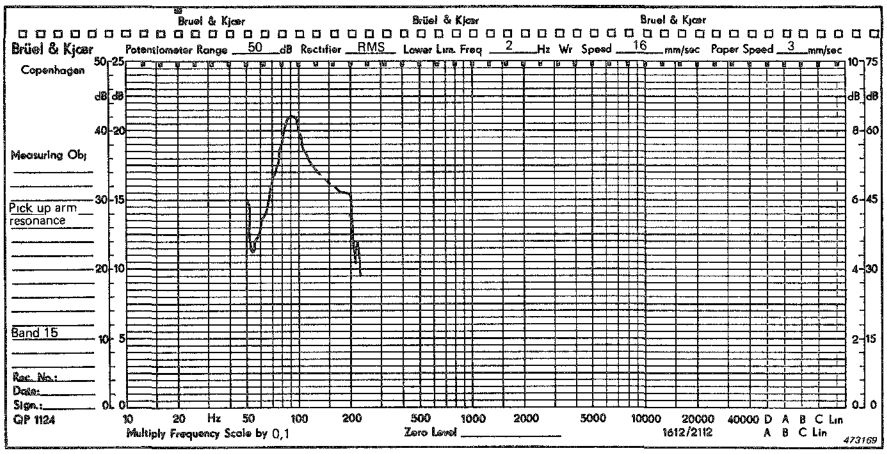


Fig.59. Response curve showing tone arm resonance

Distortion

A set-up for swept measurements of individual harmonic distortion components was shown in Fig 28. However, this can be expanded to Difference-Frequency and Intermodulation measurements as indicated in Fig.60. The BFO frequency from 2010 is used as a tracking signal in one channel while the two-tone test signal is recorded synchronously in the other channel.

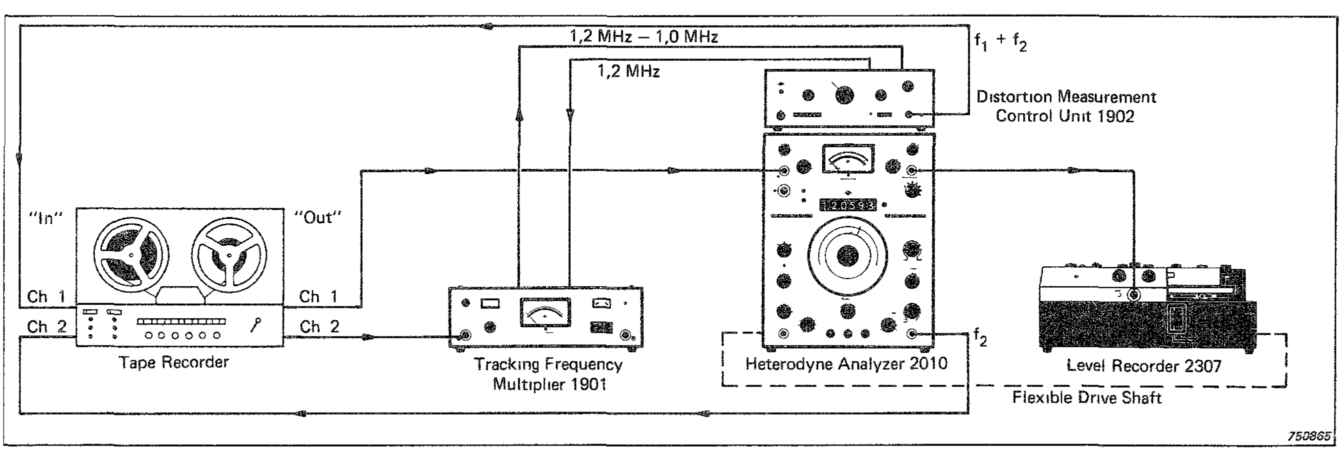


Fig.60. Set-up for Difference-Frequency and Intermodulation measurements

16. Frequency Response in the Actual Listening Room — Using 1/3 Octave Pink Weighted, Random Noise

One objective measuring method which gives good correlation to audible quality is the measurement of frequency response in the actual listening room using 1/3 octave pink weighted random noise. Since this subject has been treated in greater *detail in Application Note No.* 15—067 (Fig.61) only the high points will be discussed here.

Listening and measuring tests were performed using five different loudspeakers each in three different rooms (Fig.62). The results of the objective third octave measurements are shown in Fig.63. Here the vertical columns represent the three rooms and the horizontal rows represent the five speakers. From these, the influence of the rooms on the same loudspeaker shows some very significant differences. This underlines the importance of including the influence of the room in loudspeaker measurements.

The results of the listening evaluations by five experienced listeners showed that the subjective evaluations of the speakers in the various rooms corresponds very closely to the third octave measurements.

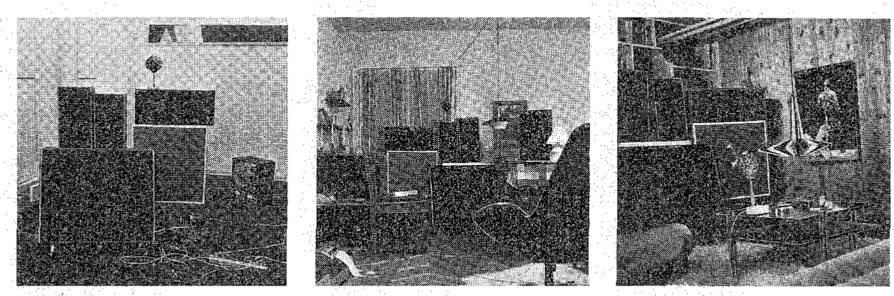
Brüel & Kjær application notes

Relevant loudspeaker tests
in studios
in Hi-Fi dealers' demo rooms
in the home etc.
– using 1/3 octave, pink-weighted, random noise

Paper presented at the 47th
Audio Engineering Society Convention
1974-02-26/29 Copenhagen Denmark

15–067

Fig.61. Application Note No 15—067



L1 L2 L3

Fig.62. The three rooms and five loudspeakers used in the third octave and listening tests

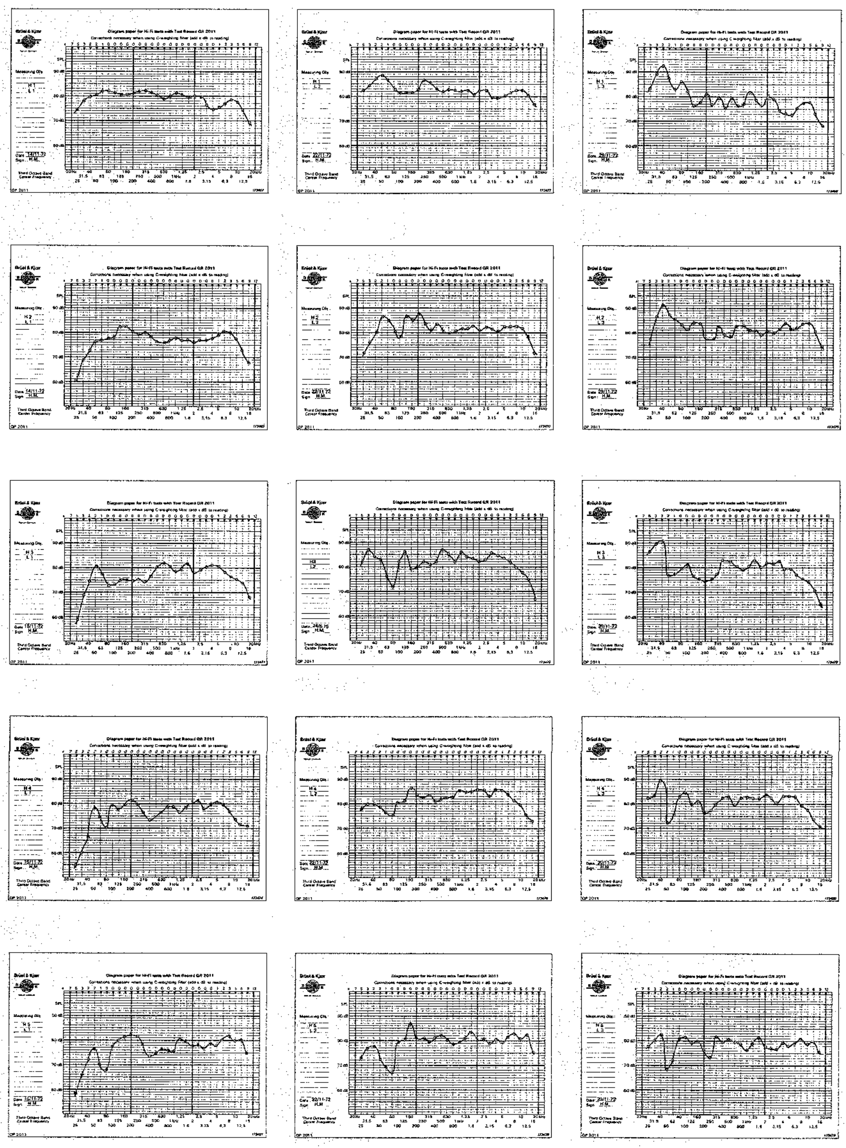


Fig.63. Third octave measurements of the five loudspeakers in three different rooms. Three horizontal charts are for the same loudspeaker test in the three different rooms

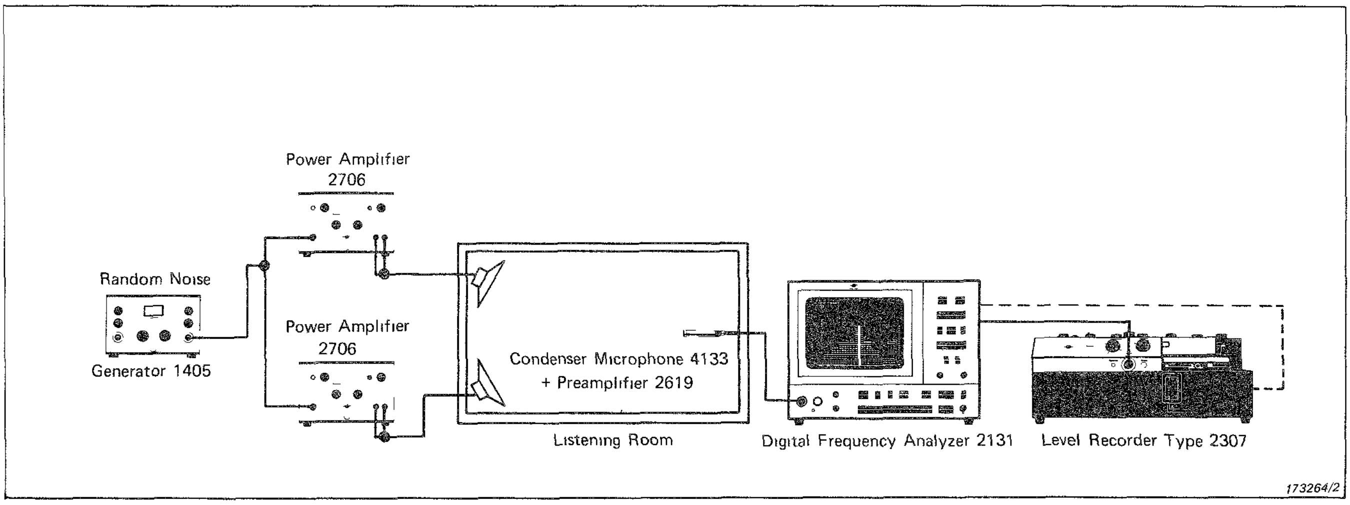


Fig.64. Use of Real Time Analyzer for third octave loudspeaker measurements

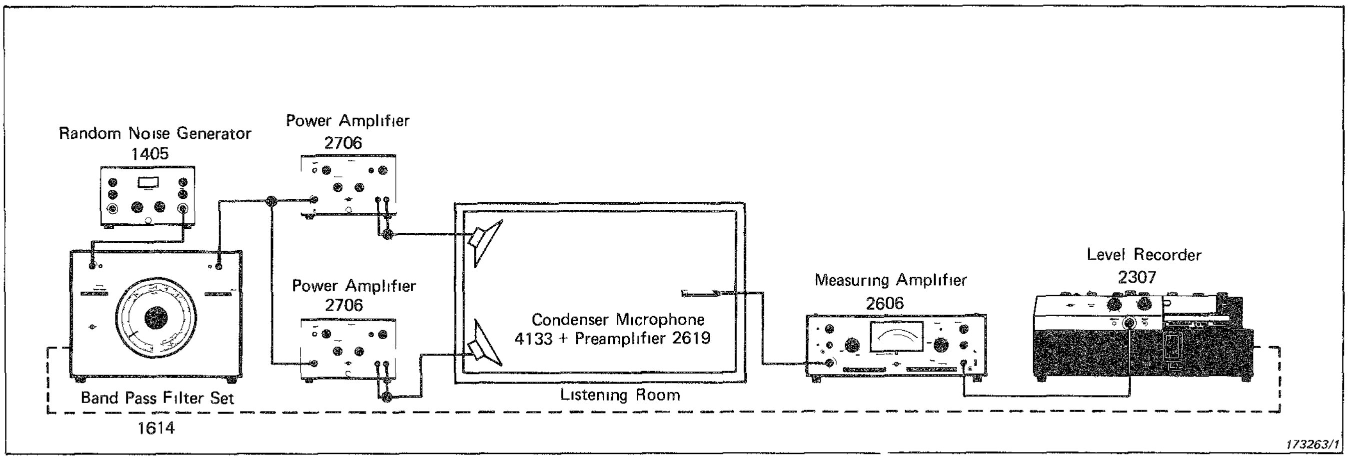


Fig.65. Sequential measurement of third octave response

The third octave measurements may be made in three different ways of various degrees of convenience and expense The most sophisticated technique uses a Digital Frequency Analyzer Type 2131 which displays the level of all third octave bands simultaneously (Fig.64). A simpler technique involves the transmission of one third octave band at a time and plotting the results on the Level Recorder (Fig 65). The simplest and least expensive technique uses Pink Noise Test Record QR 2011 which contains the third octave bands of noise The results are measured with Precision Sound Level Meter Type 2206 (Fig.66) and are manually noted on the graph paper provided with the test record. The more sophisticated techniques are especially relevant for sound system equalization of cinemas, theaters, concert halls, and recording studios. However, all three methods show excellent agreement with each other and with the results of subjective tests.

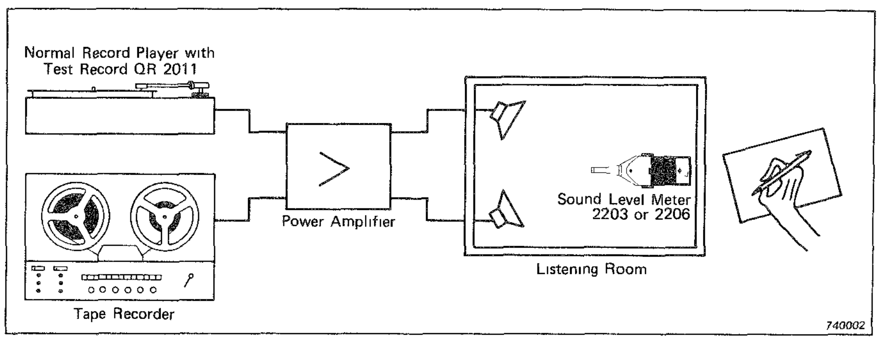


Fig.66. Portable third octave measurements using special test record

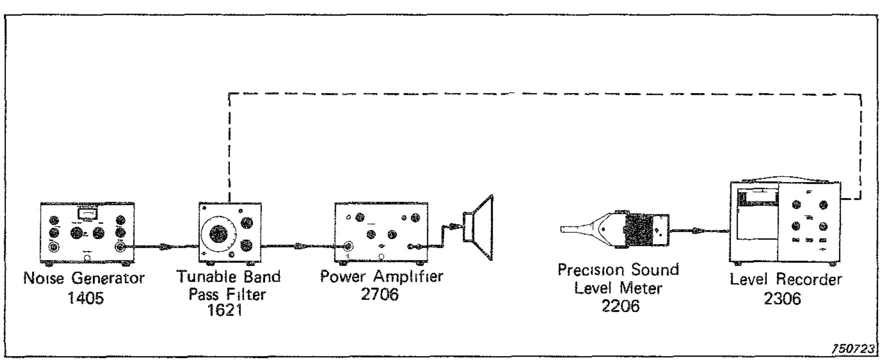
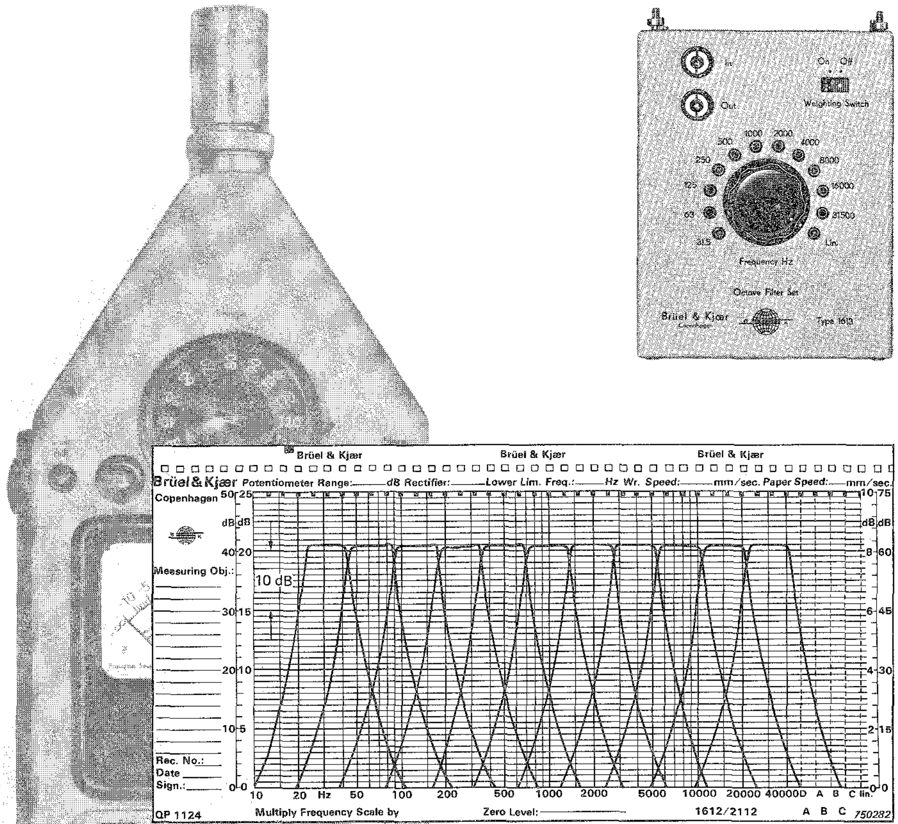


Fig.68. Portable equipment for third octave measurements with automatic recording of results on 2306

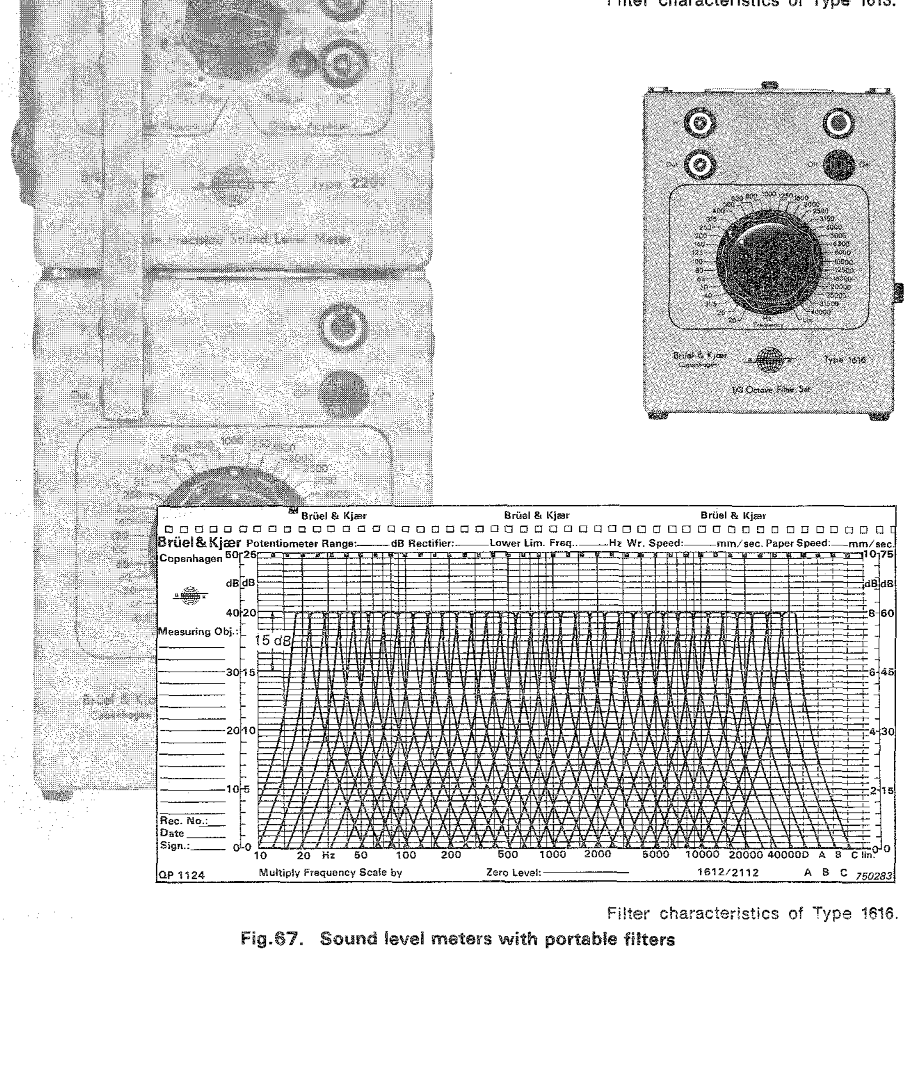
If more portable instrumentation is required on the receiving end, Precision Sound Level Meters Type 2203 or 2209 fitted with Third Octave Filter Type 1616 can also be used to measure the response of the system to the pink noise excitation (Fig.67).

The portable instrument set-up indicated in Fig.66 may be expanded by use of Portable Level Recorder Type 2306. This may be synchronized with band B-3 of the test record QR 2011 by manually starting the Level Recorder at the end of the 1 kHz tone and using a paper speed of 1 mm/s.

Another portable set-up is shown in Fig.68. Here the output of Noise Generator Type 1405 is filtered through Tunable Band Pass Filter Type 1621 set in the third octave mode. The filter is swept manually, but pulses from the filter sent to the Level Recorder 2306 ensure syncronization of the two instruments.



Filter characteristics of Type 1613.



Filter characteristics of Type 1616.

Fig.67. Sound level meters with portable filters

17. Reverberation Time

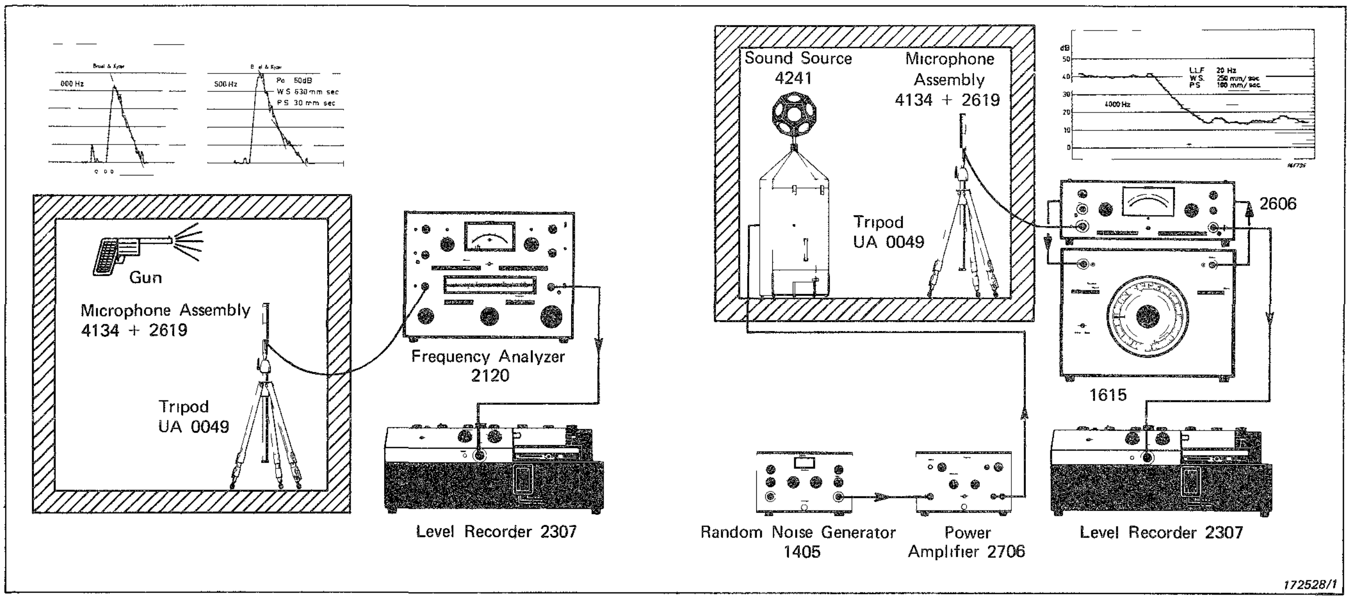


Fig.69. Set-ups for measurements of reverberation time

The reverberation time of a room is defined as the time it takes the sound level in a room to decrease 60 dB when a sound source is suddenly switched off The reverberation time of a room is one of the acoustic characteristics that has found most widespread use when a room's acoustic qualities must be determined.

To measure reverberation time, a sound source and a sound measuring set-up connected to the level recorder are used. When the sound source is switched off, a decay curve is recorded on the level recorder and the reverberation time is found from the slope of the curve. A special protractor, SC 2361, is available for this purpose. As the reverberation time usually changes with frequency, either the transmitting or receiving instrumentation can be made frequency selective. Selective reception increases the dynamic range by attenuating background noise, while selective transmission reduces the power requirement. Fig.69 shows two set-ups for determining reverberation time as a function of frequency and also shows typical curves obtained

Narrow bands of noise or a warbled sine wave signal may be used for selective excitation Pure sine is not recommended due to standing waves. The noise signal has the advantage that many room resonances are excited simultaneously. Constant bandwidth narrow band noise may be generated by Sine Random Generator Type 1027 Constant percentage bandwidth noise (1/3 octave, for example) may be generated using the pink noise from the 1027 or from Noise Generator Type 1405 and filtering it through Filter Set Type 1615. An instrument set-up using both selective transmission and reception is shown in Fig.70 All the curves are plotted on the same paper by forming the level recorder paper into a continuous loop. Filter switching is automatically synchronized with the Level Recorder.

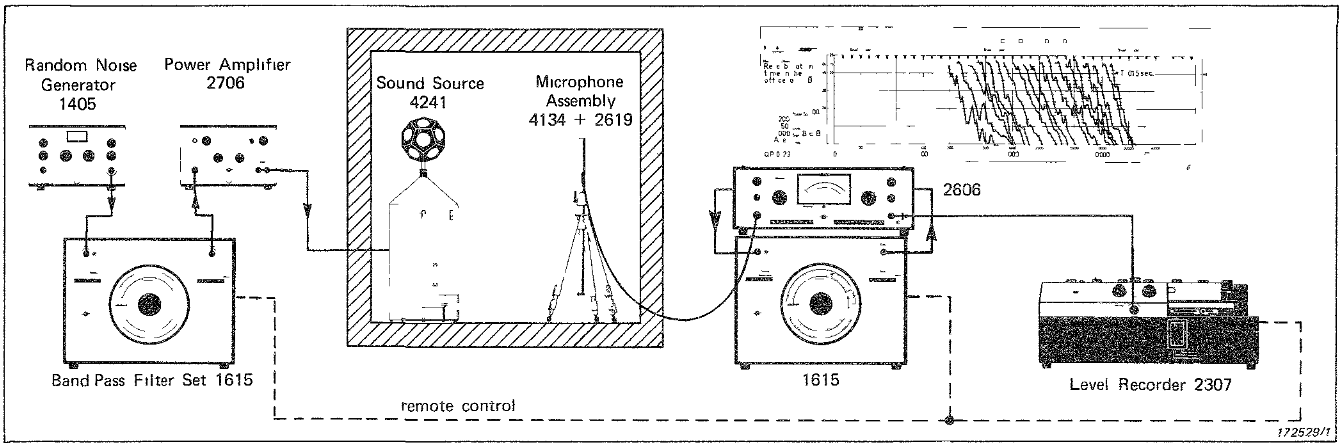


Fig.70. Automatic measurement of reverberation time

Reverberation Processor Type 4422 (Fig.71) allows measurements of reverberation time according to the integrated tone burst method. This results in very smooth and reproducible decay curves and gives a much more accurate determination of the initial slope than the methods using noise or warbled sine. Since the upper 10 to 15 dB of the decay curve are of special importance when evaluating acoustic quality of the room, the 4422 has been designed to give a direct reading of "Early Decay Time" (EDT) on its built-in meter.

Portable equipment is often required for reverberation time measurement. Two portable set-ups are therefore shown in Figs.72 and 73.

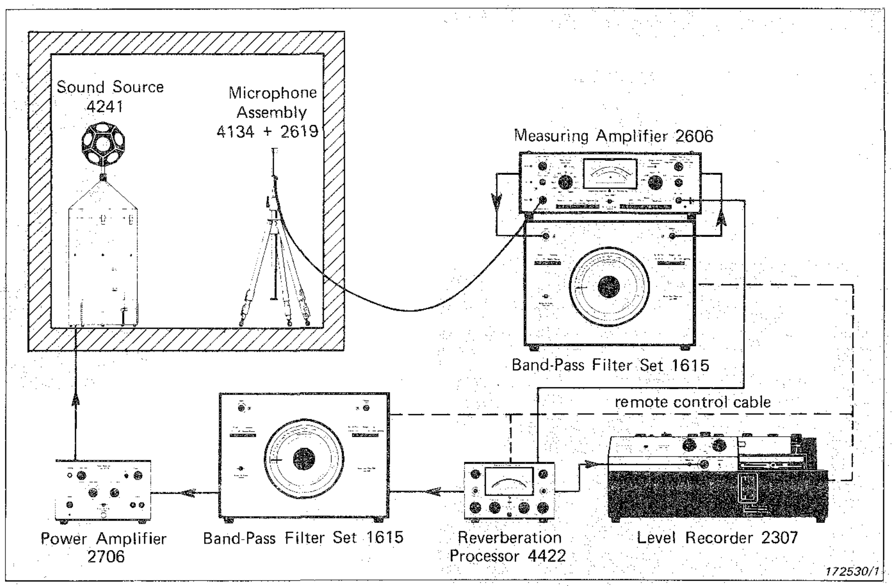


Fig.71. Measurement of Early Decay Time (EDT)

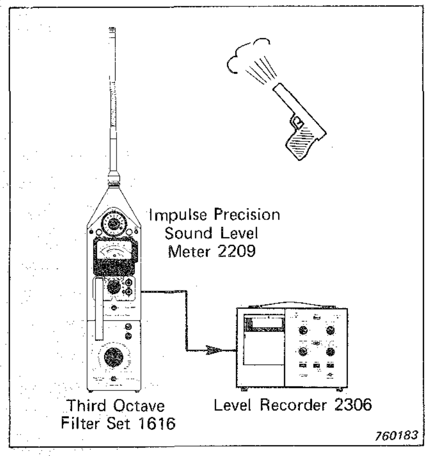


Fig.72. Portable set-up

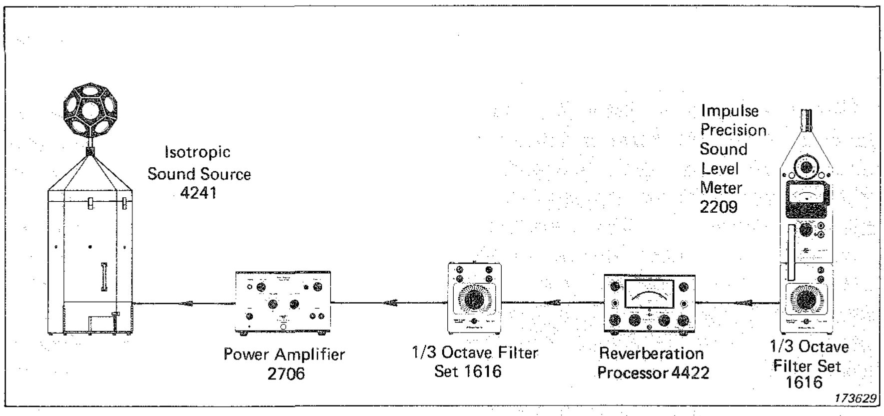


Fig.73. Optimum portable set-up

18. Sound Power

Sound power is a loudspeaker parameter that gives reasonably good correlation to listening tests. Sound power is defined as the total acoustical power radiated by the loudspeaker as a function of frequency.

If this measurement is performed in a free field (anechoic chamber) a reasonably large number of points on a sphere around the speaker should be measured to permit calculation of the power. In theory, all points in all directions should be measured and averaged.

In a reverberant room, the sound pressure at all points should be the same and hence only one measurement should be necessary at each frequency. However, this ideal is not always achieved, and averaging of several points may be necessary. This can be achieved by placing the microphone on Rotating Microphone Boom Type 3923 (Fig.74) or by using a multiplexed array of microphones and averaging the results by Computer (Fig.75). The sound power can then be calculated based on the mean sound pressure, the reverberation time, and the volume of the room according to the formula on page 33.

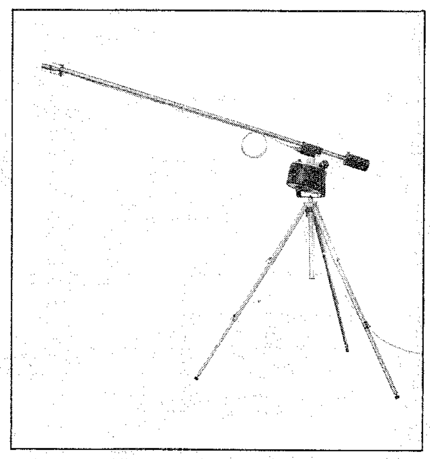


Fig.74. Rotating Microphone Boom Type 3923

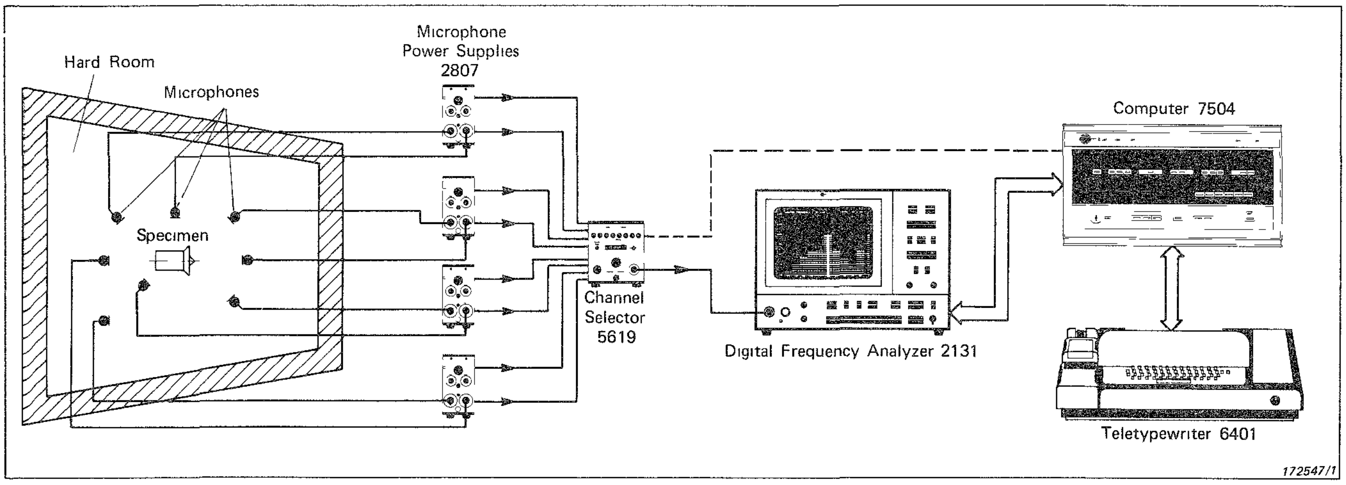


Fig.75. Use of Computer to calculate sound power from multiplexed microphones

$$10 \log_{10} \left(\frac{P}{P_o}\right) = 20 \log_{10} \left(\frac{p_m}{p_o}\right) - 10 \log_{10} \left(\frac{T}{T_o}\right) + 10 \log_{10} \left(\frac{V}{V_o}\right) - 14 \text{ dB}$$

where

- P is the power in watts
- P_o is the reference power of 10^{-12} W
- p_m is the mean sound pressure in Pa (N/m^2)
- p_o is the reference sound pressure of 20 μPa
- T is the reverberation time
- T_o is the reference reverberation time of 1 s
- V is the room volume
- V_o is the reference volume of 1 m^3

This computation may be made automatically by Computer Type 7504 using a software program available from Brüel & Kjær. A typical print-out is shown in Fig.76

```
CENTRE
FREQUENCY HZ    125   160   200   250   315   400   500   630   800   1000

REVERBERATION
TIME SEC        7.0   6.5   5.0   4 5   4.0   4.0   4.5   3.5   3.5   3.5

                1250  1600  2000  2500  3150  4000  5000  6300  8000  10,000

                3.2   3.2   2.8   2.8   2.8   2.6   2.2   2.4   2.5   2.5

READ-IN INTERVALS (SEC) = 1

12345678

SOUND POWER LEVELS IN ONE-THIRD OCTAVE BANDS

FREQUENCY   SOUND                                         SOUND POWER
CHANNEL     POWER                                         MEASURED BY
NUMBER      LEVEL                                         CONVENTIONAL
                                                          INSTRUMENTS
  REF.:     50 0 DB                                          DB
    21   xxxxxxxxxxxxxxxxxx  68.2 DB                         73.5
    22   xxxxxxxxxxxxxxxxxxxxxxxx  73.6 DB                   77
    23   xxxxxxxxxxxxxxxxxxxxxxxxxxxxx 79.0 DB               81
    24   xxxxxxxxxxxxxxxxxxxxxxxxx 75.4 DB                   77
    25   xxxxxxxxxxxxxxxxxxxxxxxxxxx 76.8 DB                 78
    26   xxxxxxxxxxxxxxxxxxxxxxxxxxxxxxx 81.2 DB             82
    27   xxxxxxxxxxxxxxxxxxxxxxxxxx  75.8 DB                 77
    28   xxxxxxxxxxxxxxxxxxxxxxxxxxxx  77.8 DB               78.5
    29   xxxxxxxxxxxxxxxxxxxxxxxxxx  76.0 DB                 77
    30   xxxxxxxxxxxxxxxxxxxxxxxxx 74.9 DB                   76
    31   xxxxxxxxxxxxxxxxxxxxxxxxx 75.1 DB                   76.5
    32   xxxxxxxxxxxxxxxxxxxxxxxx  73.8 DB                   74.5
    33   xxxxxxxxxxxxxxxxxxxxxxx 73.4 DB                     74
    34   xxxxxxxxxxxxxxxxxxxxxx  71.8 DB                     73
    35   xxxxxxxxxxxxxxxxxxxx  69.8 DB                       71.5
    36   xxxxxxxxxxxxxxxxxxxx  69.7 DB                       71
    37   xxxxxxxxxxxxxxxxxxx 68.7 DB                         70
    38   xxxxxxxxxxxxxxxx  65.5 DB                           66.5
    39   xxxxxxxxx 58.7 DB                                   60.5
    40   x 50.5 DB                                           52.5
```

172494

Fig.76. Typical computer print-out of sound power calculation

19. Visual Analysis of Speaker Motion

Motion Analyzer Type 4911 allows visual observation of mechanical movements by use of the stroboscopic principle. Special features of this motion analyzer are a slow motion position whereby a fast motion can be made to appear to move slowly, and a phase deviation feature whereby a motion can be frozen in any position. Provision for external triggering permits the flashing of the light to be synchronized with the movements as the speed or frequency changes. A typical set-up is shown in Fig.77 and a photograph showing wire movements of a loudspeaker is shown in Fig.78.

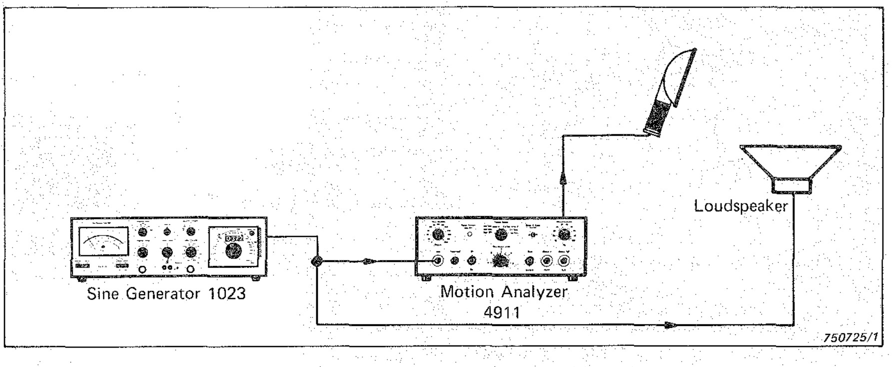


Fig.77. Use of Motion Analyzer

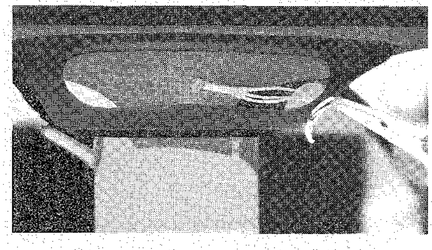


Fig.78. Motion Analyzer "freezes" motion of loudspeaker wires

When loudspeakers or other objects are placed on a vibration exciter, resonance search is a valuable aid for design and test purposes. The Motion Analyzer can be controlled directly from the sweep generator and all resonances can be visually observed without any further adjustments of the Motion Analyzer. For the study of particular resonances, the generator sweep can be stopped and the exact resonance frequency be found by manual tuning. The resonance can then be studied in greater detail using the phase and slow motion adjustments of the motion analyzer (Fig.79).

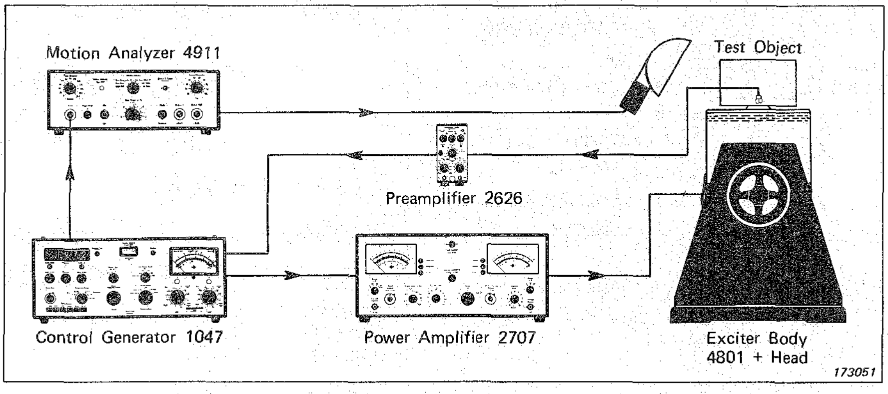


Fig.79. Examining resonances of object on vibration exciter

20. Frequency Analysis of Noise and Crosstalk

The various B & K Analyzers are well suited for frequency analysis of all kinds of noise Typical measurements relevant to the Hi-Fi industry are shown in Figs 80 to 82

Modulation noise

Modulation noise shows up as sidebands to a pure tone recorded on a tape due to the discrete nature of the magnetization of the individual particles of the tape In practice this phenomenon will be difficult to separate from mechanical frequency modulation due to tape speed variations A typical result measured with the Heterodyne Analyzer 2010 is shown in Fig 80

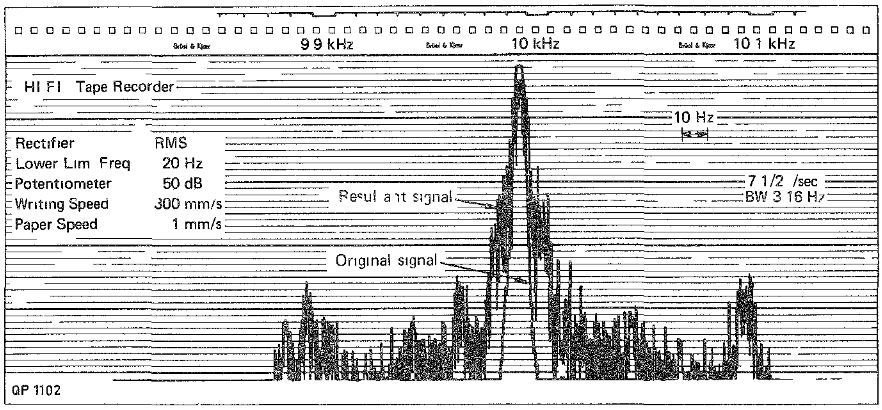


Fig 80 Modulation noise

Crosstalk

A frequency analysis of the other channel of a stereo system when the first channel is swept with a pure tone indicates the channel separation or the crosstalk A typical result for a tape recorder measured with the 1/3 Octave Filter Set 1615 is shown in Fig 81

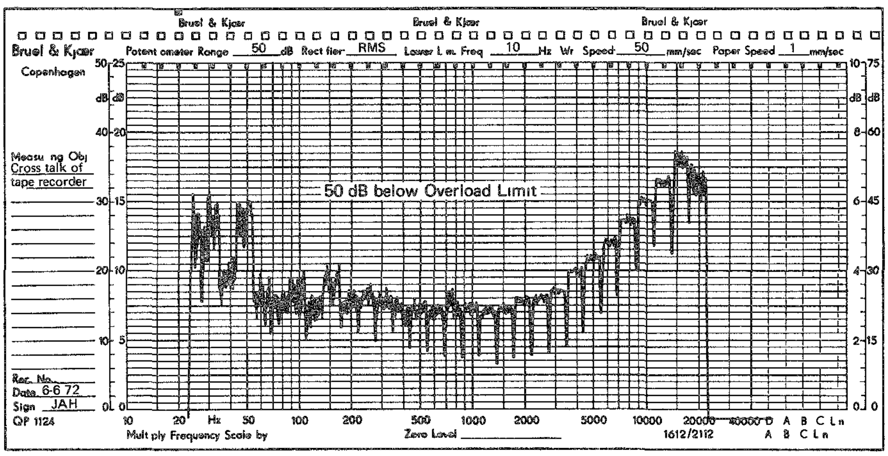


Fig 81 Crosstalk

Hum and Noise

The total noise of a system as a function of frequency can easily be measured using the various B & K Analyzers A typical result for a tape recorder measured with Frequency Analyzer 2120 is shown in Fig 82

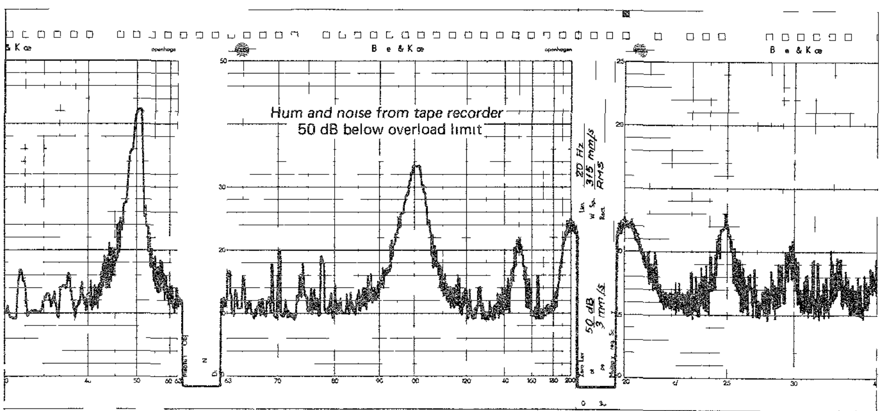


Fig 82 Hum and Noise

21. Production Control

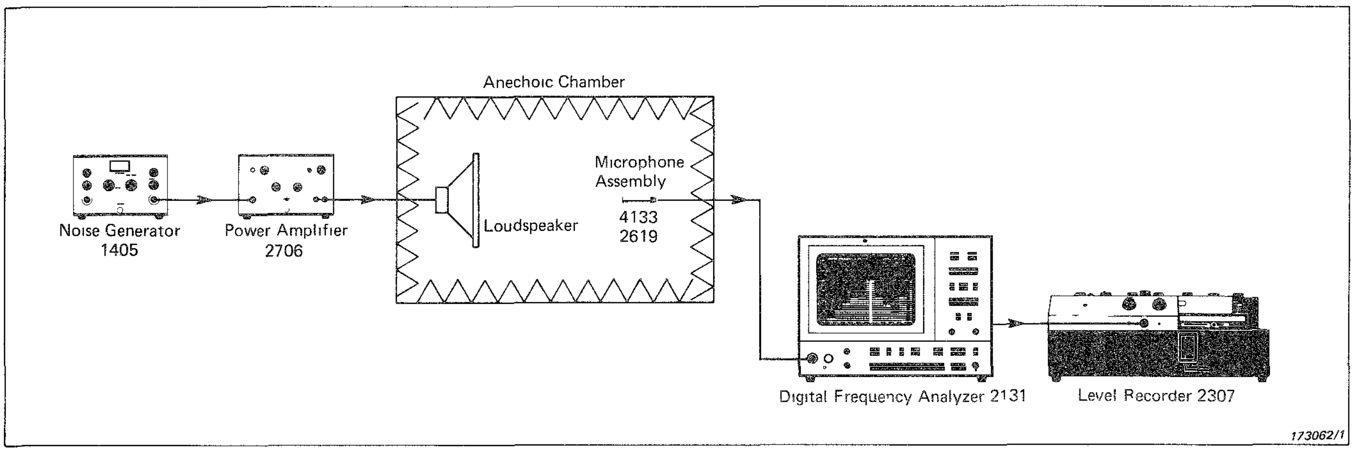


Fig 83. Production control using real time analyzer

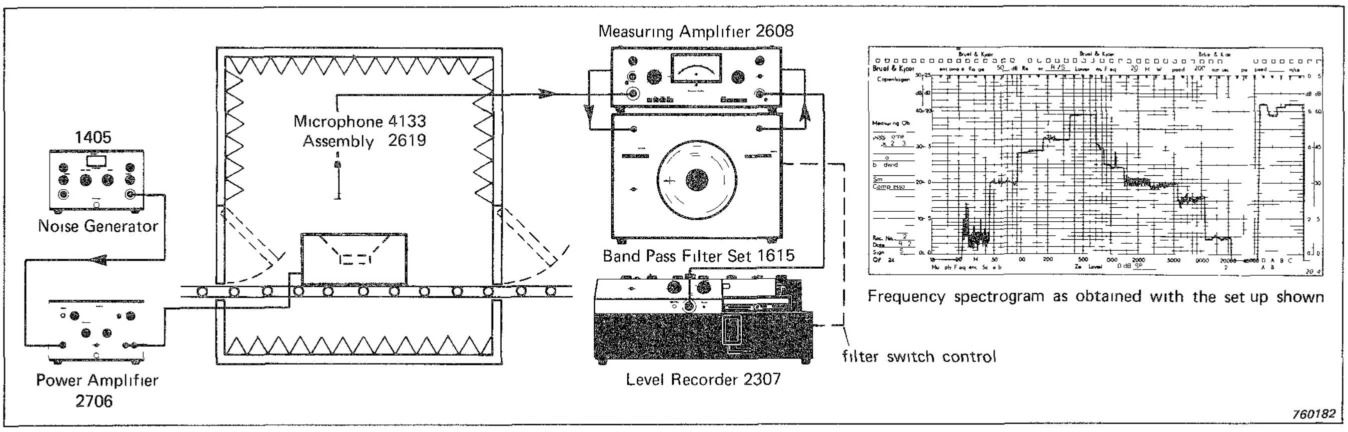


Fig 84. Production control using stepped filters

If a high test speed is required, Frequency Response Tracer Type 4712 may be used This instrument gives a fast and accurate picture of the response and thus is well suited for production control A set-up is shown in Fig.85. Another method is to excite the speaker with pink noise and measure the response with the Digital Frequency Analyzer Type 2131 (Fig.83) The nature of pink noise is more closely related to actual audio material than a sine excitation and hence is a realistic test. The Digital Frequency Analyzer Type 2131 is a new octave and 1/3 octave analyzer. It is almost entirely digital in operation since it uses digital filtering, detection and averaging techniques The results are displayed on an 11" screen. The frequency range is 1,6 Hz to 20 kHz and the dynamic range on the screen is 60 dB There are linear and logarithmic averaging possibilities and two memory capabilities so comparison with for instance a reference spectrum, which can be read-in digitally, can be made Digital read-out is possible and the store feature permits read-out to the Level Recorder while the loudspeaker under test is being replaced by the next speaker

Similar, but slower set-ups may be used incorporating combinations of the generators and analyzers already discussed. An example is shown in Fig.84

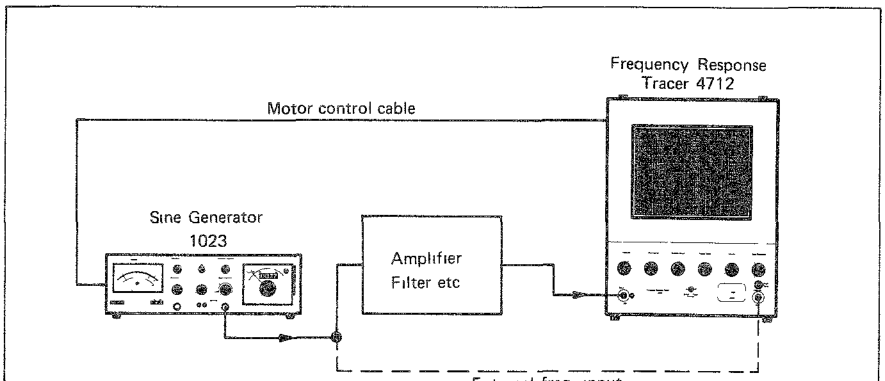


Fig.85. Set-up for finding frequency response of amplifiers, filters, etc

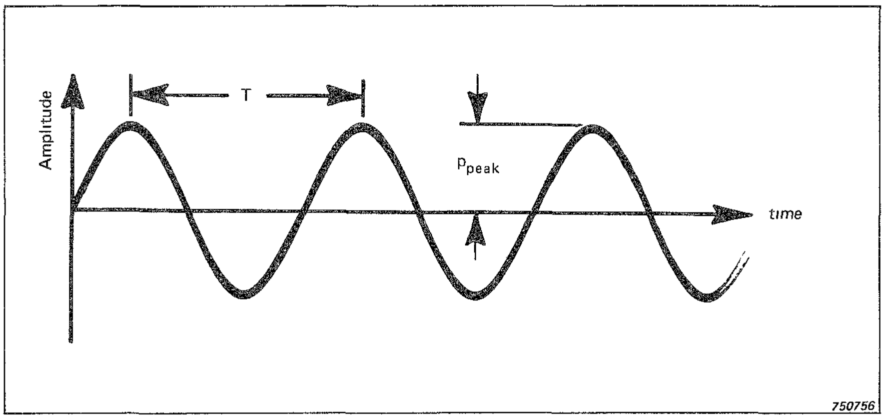
APPENDIX

1. Basic Acoustic Properties and Formulae

Properties of Sound

Sound is vibrations in gaseous, liquid or solid media. It manifests itself as pressure fluctuations in the medium and it is characterized by the instantaneous amplitudes of the pressure fluctuations, as well as by their frequencies and phase relationships.

The simplest sound signal is a pure tone, or sinusoid, which is fully determined by its peak pressure amplitude p_{peak} and its frequency f (equal to 1/T, T is the period of oscillation).

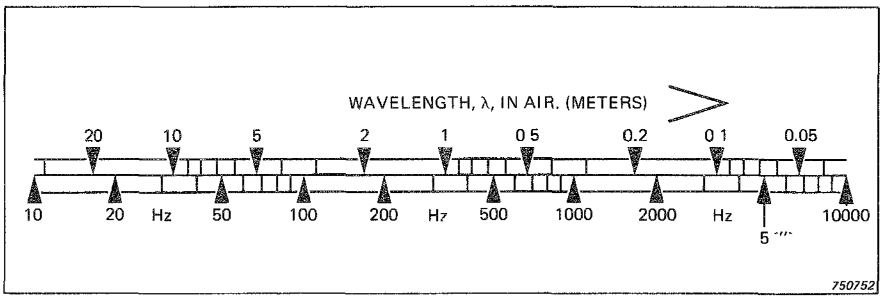


Sound Propagation

The sound is propagated at a speed c which is characteristic for the medium and which, thereby, determines the wavelength λ for a given frequency from the formula:

$$\lambda = cT = c/f$$

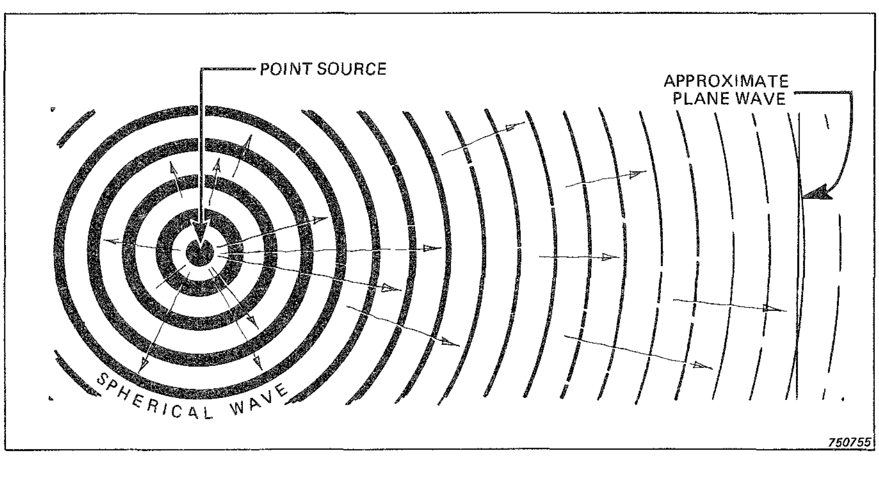
The velocity of sound in air at room temperature is app. 344 m/s and in fresh water app. 1450 m/s.



In a non-dissipative medium the intensity I of the sound at a distance r is given by the sound power W emitted by the source divided by the surface area of the sphere with radius r

$$I = \frac{W}{4\pi r^2}$$

At a sufficient distance r away from the sound source (in the far field) the intensity is also proportional to the square of the sound pressure p whereby p is proportional to 1/r. This relationship is called the "inverse distance law" or the "inverse square law".



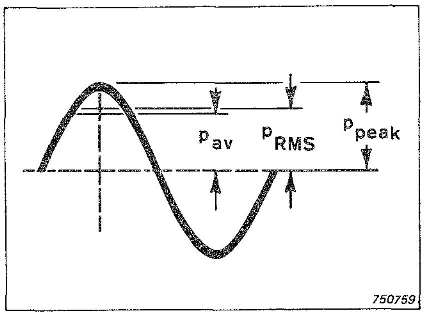
Sound Pressure Magnitude and Sound Pressure Level

The sound pressure magnitude is often characterized by other quantities than p_{peak}. The most commonly used value is the RMS value (root-mean-square value) because of its direct relationship to the sound energy. It is defined as

$$p_{RMS} = \sqrt{\frac{1}{T} \int_0^T p^2(t)\,dt}$$

Another important characteristic value is the average value

$$p_{av} = \frac{1}{T} \int_0^T |p(t)|\,dt$$



The Sound Pressure Level (SPL) in decibel (dB) is defined as 20 times the logarithm of the ratio of the sound magnitude p_{RMS} to the reference magnitude $p_{0\ RMS}$ = $20\mu Pa$.

The relationships between the above magnitude quanties are often used to characterize a signal

The Form Factor $F_1 = \frac{p_{RMS}}{p_{av}}$

(= 1,11 for pure sinusoid)

The Crest Factor $F_c = \frac{p_{peak}}{p_{RMS}}$

(= $\sqrt{2}$ for pure sinusoid)

1 Pa = 1 N/m^2 = 10 μbar = 10 dynes/cm^2 = 94 dB (Ref. 20 μPa)

1 Atmosphere = 1,013 bar = 760 mm Hg. = 1,013 x 10^5 Pa = 194 dB

$20 \log \frac{p}{p_0} = 94\ \text{dB} = (4 \times 20 + 14)\ \text{dB} \Rightarrow$

$p = 10^4 \times 5{,}012\, p_o = 10^4 \times 5{,}012 \times 20 \times 10^{-6}\ \text{Pa} \cong 1\ \text{Pa}$

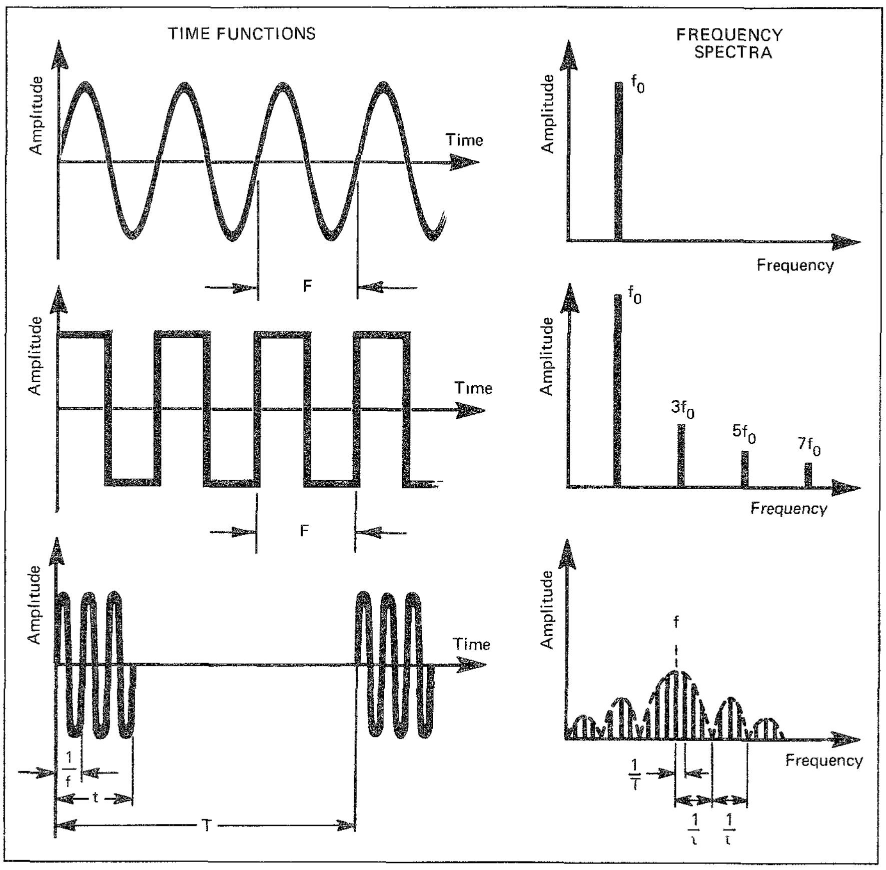
Frequency Distribution of Sound

All finite signals can be described as a combination of a number of sine waves. These signal components constitute the signal frequency spectrum. A single sine wave (a pure tone) is represented as a line in the frequency spectrum and any periodic or quasiperiodic signal is given by a number of discrete sine waves, or lines in the frequency spectrum.

For noise signals, however, the frequency spectrum consists of infinetely closely spaced frequency components and the spectrum in usually given as a spectral density. The most commonly used quantity is the Power Spectral Density (PSD) which is defined mathematically as

$$W(f) = \lim_{B \to 0} \lim_{T \to \infty} \frac{1}{BT} \int_0^T p_B{}^2(t,f)\,dt$$

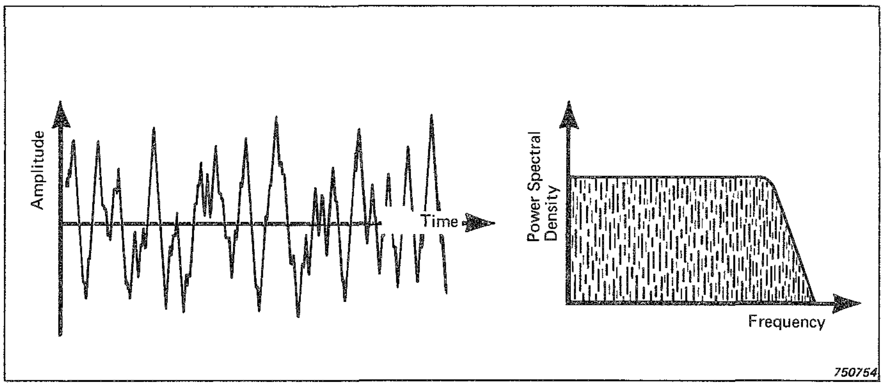
where B is the signal bandwidth and T the averaging time. One of



the advantages of power spectral density is the fact that from the frequency spectrum, the power (and thereby the p_{RMS} value) can be calculated for any given bandwidth.

For transients and impulses there is no steady state power and the frequency spectrum may, therefore, better be given as an energy spectrum

$$E(f) = \lim_{B \to \infty} \frac{1}{B} \int_0^T p_B^2(t,f)\, dt$$

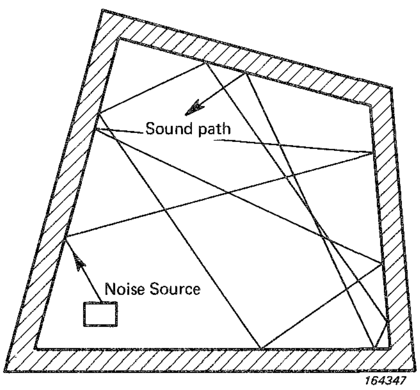


Free-field and Reverberation field

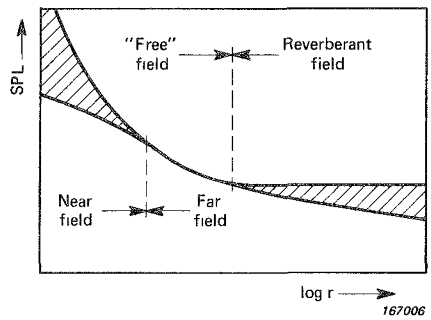
A free field is a defined as a space in which the boundaries (if any) exert a negligible influence on the sound field. Thus a free field lacks any physical object which might disturb the sound by reflecting it or by diffraction.

Free field conditions can be found outdoors at some distance above the ground, or inside in rooms where the walls are lined with highly absorbent materials therby preventing reflections. In a free-field, the sound pressure level will drop 6 dB for a doubling of distance from the source. This is call the inverse square law.

A reverberant, or diffuse field is a space in which reflection causes a uniform sound distribution over the entire space. This is achieved by the use of very hard, reflective surfaces which are placed non-parallel to prevent standing waves which would destroy the homogeneity of the diffuse field.



A normal listening room, of course, is somewhere between these two extremes, and hence is termed a semi-reverberant room.



2. Typical Instrument Combinations

A main point in these notes has been the "add-on" principle indicating that the Brüel & Kjær instruments are built to work together, so the addition of an instrument to the existing package enables new and more advanced measurements. The ideal situation is therefore to have all the instruments.

For various reasons, smaller instrument combinations may be required. Therefore the most typical combinations are listed below. The add-on possibilities are indicated by the vertical lines.

	Sine System. (pure tone)	Sine/ Noise. (pure tone + noise).	Complete Sophistic. System (pure tone + noise).	Portable 1/3 Octave System.	Real Time 1/3 Octave System.
Generator	1023	1027	2010	QR 2011	1405
Measuring Amplifier	2608	2606/07		2206	2606
Level Recorder	2305	2307	2307	2306	2307
Response Test Unit	4416	4416	4416	4416	
Microphone	4133	4133	4133	4148	4133
Preamplifier	2619	2619	2619		2619
Power Amplifier	2706	2706	2706		2706
Response Tracer	4712				
Gating System	4440	4440	4440		4440
Turntable	3922	3922	3922		3922
Slave Filter (Constant bandwidth Analyzer)	2020	2020	(2010)		
Tracking Frequency Multiplier	1901	1901			
Dist. Measurem. Control Unit			1902		
Phase Meter	2971	2971	2971		
Phase Delay Unit	6202	6202	6202		
Digital Event Recorder		7502	7502		
Voltmeter	2425	2426	2427		2425
Analyzer (Constant percentage bandwidth)	2121	2120	2120	1621	
1/3 Octave Filter				1616	1615
Real Time Analyzer		3348	3347/48		3347 or 2131
Motion Analyzer	4911	4911	4911		
Reverberation Processor		4422			4422
Rotating Microphone Boom	3923	3923	3923		3923
Test Records	QR 2009/10	QR 2009/10	QR 2009/10		
Computer	7504	7504	7504		
Tape Read/Punch	7102/6301	7102/6301	7102/6301		

3. Electroacoustic Standards

Organi-zation	Number	Date	Sta-tus*)	Short Description
IEC	94	1968	S	Tape recorders, recording and reproducing characteristics, speeds and dimensions of tape, position of tracks
IEC	200	1966	S	Loudspeakers, frequency response, polar plot, resonant frequency, power handling, test conditions, standard baffle
ANSI	S. 1.5	1963	S	Loudspeaker, impedance, frequency response, polar plot, distortion, effeciency, power handling, recommended
BS	1568	1960/66	S	Tape recording at 38, 19, 7½ and 3¾ in/sec. Originally revised and extended 1966
BS	1927	1953	S	Loudspeaker, physical dimensions, impedance and resonant frequency
BS	1928	1965	S	Disc records, characteristics of reproducing equipment
BS	1988	1953	S	Wow and flutter, general recommendation for measurements
BS	2498	1945	S	Loudspeakers, frequency response, distortion, efficiency polar plot, impedance, transient response, conditions
BS	3499	1969	S	School music equipment, tape recorders, wow and flutter, frequency response, noise, distortion
BS	3499	1966	S	Amplifiers for musical instruments, function of tone controls, hum, noise
BS	3860	1965	S	Amplifiers, distortion, rated power, frequency response, intermodulation, hum and noise, methods and data presentation
DIN	45 500	NOV 1970	D	Disc reproducing equipment, wow and flutter, rumble, pick up frequency response, distortion, crosstalk
DIN	45 500	MAY 1971	D	Tape recorders, wow and flutter, frequency response, distortion, noise, crosstalk, erase damping
DIN	45 500	APR 1966	S	Microphones, frequency response, polar characteristic, distortion
DIN	45 500	AUG 1971	D	Amplifiers frequency range, distortion, intermodulation, crosstalk, noise, output power, damping ratio
DIN	45 500 sheet 1–3	FEB 1971	S	Loudspeakers, frequency response, power handling, distortion, music power
DIN	45 500	MAY 1971	S	Combined equipment, frequency response, crosstalk, distortion, intermodulation
DIN	45 507	OCT 1966	S	Sound recorders, measurement of wow and flutter
DIN	45 511 sheet 1–3	MAY 1971	S	Tapes ¼″, ½″ and 1″ width. Position of tracks
DIN	45 512 sheet 2	FEB 1969	D	Tapes, determination of electroacoustic properties.
DIN	45 513 sheet 1–6	1962–71	S	Test tapes for ¼″ tape at 76, 38, 19, 9½ and 4¾ cm/sec.
DIN	45 519 sheet 2	MAY 1971	D	Tape, determination of print-through and nonuniformity of recorded flux
DIN	45 521	OCT 1963	S	Tape recorders, determination of crosstalk
DIN	45 542 to 45	FEB 1966	S	Test records, wow and flutter, rumble, crosstalk, tracking angle 33 and 45 rev/min.
DIN	45 573 sheet 1	JUL 1962	S	Loudspeakers, test conditions and methods for type test
DIN	45 573 sheet 2	JAN 1969	S	Loudspeakers, power handling and life test
DIN	45 575	MAY 1962	S	Loudspeakers, standard baffle for measurements

*) S – Standard
D – Draft

Above is given a list of the most relevant standards applied to electro acoustics. Many standards in the border area of this subject have been omitted and thus the list cannot be considered as being complete. Futhermore it is a known fact that several standard organizations have relevant subjects under consideration which may outdate the present survey in a relatively short period.

. . . first in Sound and Vibration

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