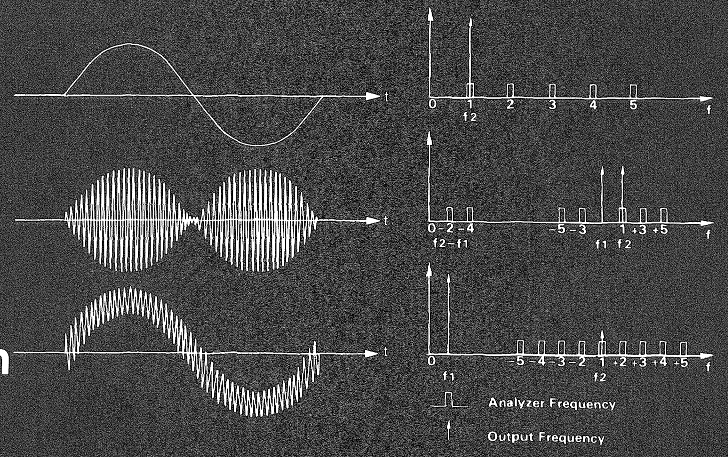
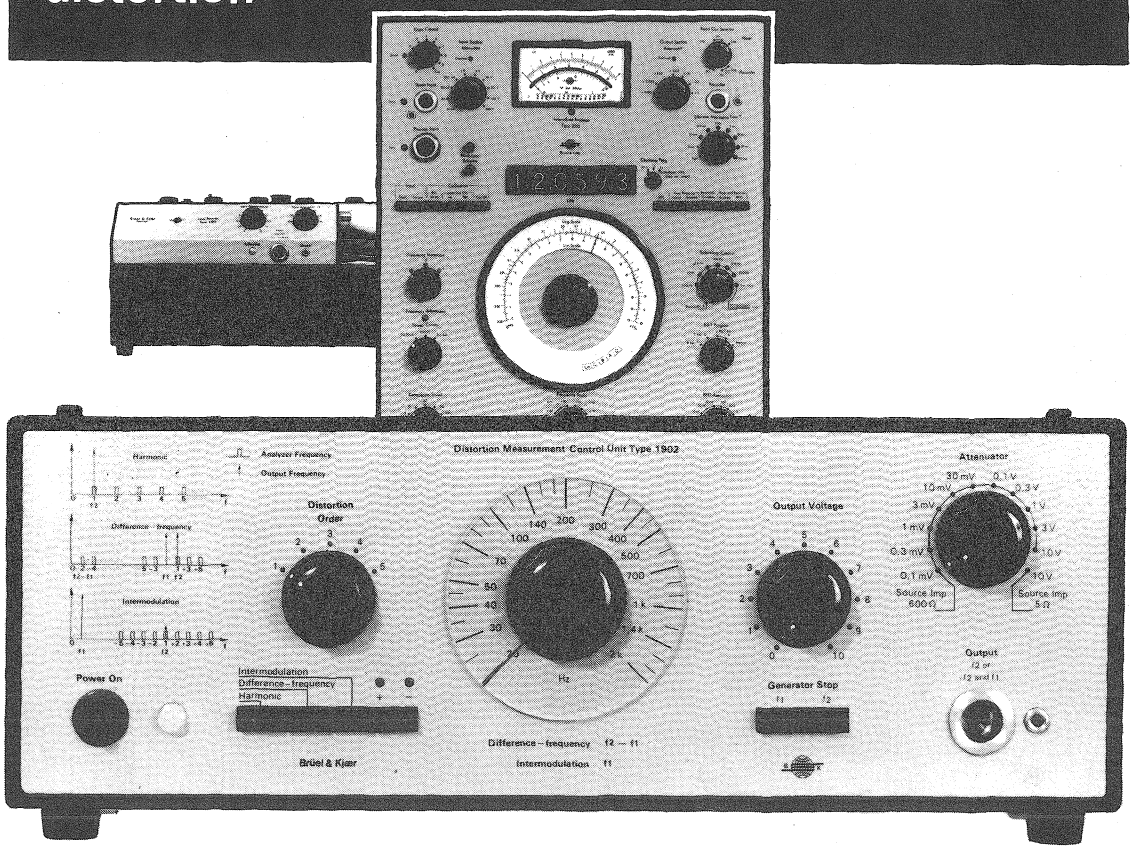
BRÜEL & KJÆR application notes

Swept measurements of harmonic difference frequency and intermodulation distortion





Swept measurements of harmonic, difference-frequency and intermodulation distortion

by *Carsten Thomsen* and *Henning Møller*, Brüel & Kjær

1. What Is Intermodulation Distortion?

Intermodulation Distortion is the interaction of the components of a complex signal to produce frequency components not found in the original signal. In practice, system unlinearities cause this intermodulation distortion (IM) to occur due to amplitude and frequency modulation of the higher frequency components by the lower frequency components.

This is illustrated in Fig.1 where 100 Hz and 800 Hz signals are introduced into an unlinear system. The resulting signal contains distortion components which are sidebands around 800 Hz. The frequencies of the sidebands are equal to the sum and difference of the upper frequency (800 Hz) and the integer multiples of the lower frequency: 800 Hz ± 100 Hz, 800 Hz ± 200 Hz, 800 Hz ± 300 Hz, and so on.

Difference-frequency distortion is a special case of intermodulation distortion which only considers the components which are the difference between the original frequency components and their harmonics, whereas IM considers both sum and difference components.

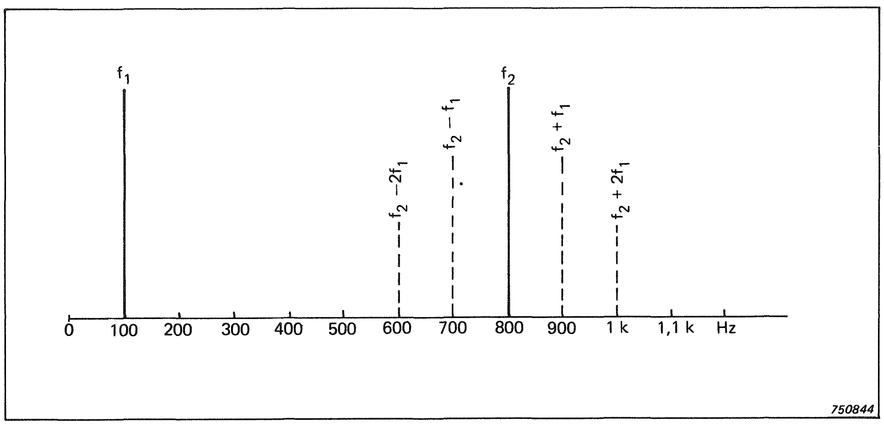


Fig.1. Illustration of IM distortion of 100 Hz and 800 Hz signals

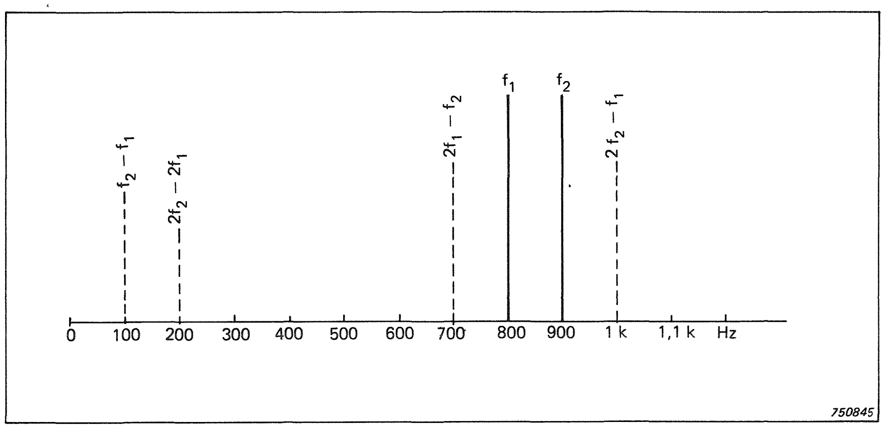


Fig.2. Difference-frequency distortion of 800 Hz and 900 Hz signals

Difference-frequency distortion is illustrated in Fig.2. Here two test tones of very close frequencies are introduced into the system. The resulting distortion components which are described mathematically as the difference between the two original frequencies, or multiples thereof, are shown. Thus, difference-frequency measurements ignore the sum products, which often lie outside the audible range.

The value of difference-frequency measurements is that they can be used for distortion measurements at the upper frequency limit of a system since the distortion components still fall inside the frequency range of interest.

Distortion Order is used to describe the frequency relationship of a given distortion component to the input signal(s). For harmonic distortion, distortion order is equal to the harmonic number. For intermodulation distortion and difference-frequency distortion the distortion components will always be of the form

$$A \sin (n\omega_1 \pm m\omega_2)$$

where A is the amplitude, n and m are integers, and ω_1 and ω_2 are the two test frequencies. The distortion order then is n + m and the + and — signs indicate whether it is a sum or difference component. Thus, for example, the signal $A \sin (3\omega_1 - 2\omega_2)$ is a fifth order difference-frequency distortion component.

In practice, most intermodulation distortion measurements are made using two tones, which seems to be a reasonable approximation of a complex signal. Measurements with more than two tones result in unmanageably complex instrumentation. However, one other class of measurements is possible — using random noise as the test signal. This, of course, contains an infinite number of frequencies.

2. Why Measure Intermodulation Distortion?

The measurement of IM distortion in an audio system is extremely important — perhaps even more so than harmonic distortion. Unfortunately this has been limited somewhat due to the lack of adequate instrumentation. We will list some of the important reasons why IM measurements are relevant.

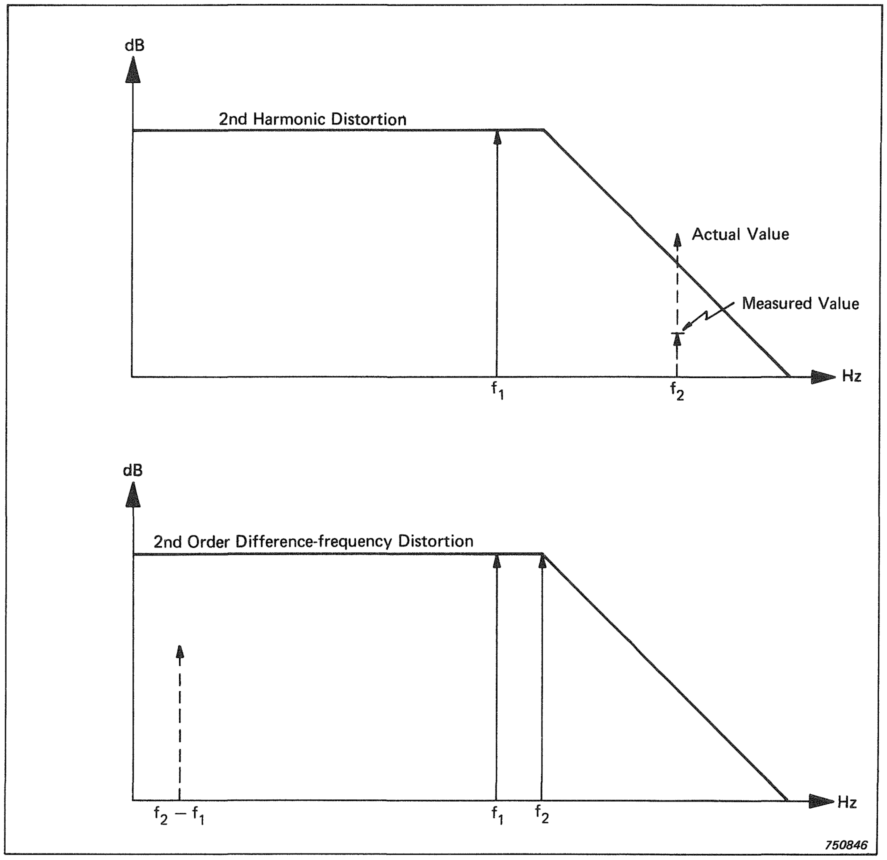


Fig.3. Harmonic distortion components are attenuated by the high frequency roll-off of the system, while difference-frequency distortion components remain inside the bandpass of the system

1. A two tone test signal is more realistic than a one tone signal. Since music and speech consist of many different frequencies occuring simultaneously the distortion test signal used should also contain more than just one frequency so there is an opportunity to see how the system causes interaction between the various frequency components. A single tone, of course, cannot be used to measure interaction phenomena, such as a string bass modulating a violin. Neither can single tone tests be used to describe Doppler distortion in loudspeakers.
2. IM distortion is audibly more annoying than harmonic distortion. Harmonic distortion components lie at frequency multiples of the fundamental, which often coincide with the harmonic components already found in the original music signal. For example, second and fourth harmonic distortion result in the generation of frequency components one and two octaves above the fundamental tone — which audibly can be quite acceptable. However, the sum and difference components arising in intermodulation distortion have no harmonic musical relationships and hence can be quite annoying.
3. IM distortion measurements can be used over the entire frequency range of the system, whereas harmonic distortion measurements become meaningless well below the high frequency cutoff.

For example, a harmonic distortion measurement of a tape recorder at 15 kHz will be unrealistically low due to the inherent steep high frequency roll-off in the system which will attenuate the distortion components at 30 kHz, 45 kHz etc. However, an intermodulation measurement with one tone at 15 kHz yields difference products below 15 kHz — still within the bandpass of the system. These difference products will also be of audible significance, whereas harmonic distortion components above 15 to 20 kHz can't be heard — and thus may not be considered important (Fig.3).

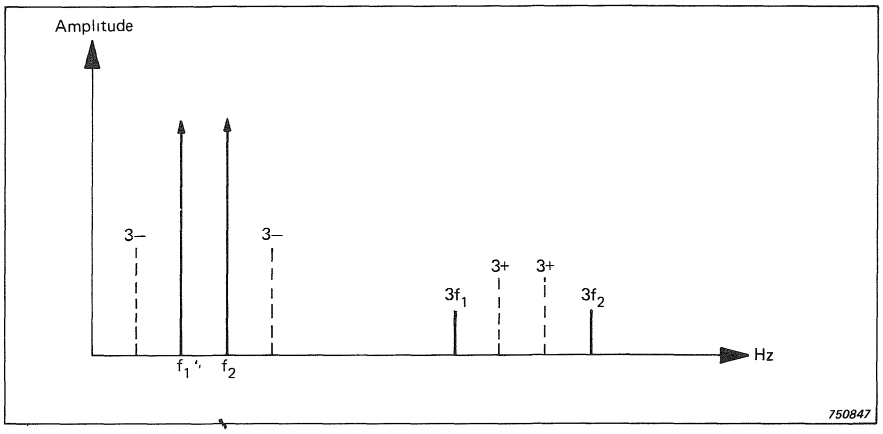


Fig.4. Relative amplitudes of IM (dashed lines) and harmonic (solid lines) components of a system with a third-order unlinearity

4. The amplitude of intermodulation components is often higher than the amplitude of the harmonic components. Thus IM distortion measurements are a more sensitive test of system unlinearities. This is due in part to the fact that two tones contain more energy than one tone.

   Fig.4 shows the calculated harmonic and intermodulation components of a non-linear third order system. Here it can be seen that any intermodulation component is higher than any harmonic distortion component. This is generally true for any order of distortion (see section 8).

5. A given unlinearity in a system will result in more intermodulation components than harmonic components. For example, for third order unlinearities, two harmonic components are generated but four intermodulation frequencies result. For higher orders of distortion, the ratio of the number of IM components to the harmonic distortion components increases.

6. Intermodulation distortion measurements using band-pass techniques are not as sensitive to noise and hum as harmonic distortion measurements using the band stop method. One reason is that the level of the IM components is higher than the harmonic components. A second reason is that IM measurements are usually made at the higher frequencies where the spectral noise density is not as high as at lower frequencies.

7. It is not usually possible to calculate the intermodulation distortion based on harmonic distortion measurements.

   Although there is a precise mathematical relationship between the two types of distortion, in practice it is nearly impossible to calculate without knowing the transfer characteristic (input amplitude vs. output amplitude) in precise detail. For example, a fourth order unlinearity will give rise both to second and fourth harmonic distortion. These second harmonic components will then add to those components due to a second order unlinearity. It then becomes very difficult to see which components are caused by what type of unlinearity. Therefore, the direct measurement of IM distortion is the only convenient way of determining this aspect of system performance, although in theory, it can be extracted from harmonic distortion measurements.

8. Intermodulation distortion at high frequencies seems to be related to transient intermodulation distortion (TIM).

   TIM describes the distortion arising due to the slowness of amplifier feedback loops. This same slowness can be measured using a high frequency test signal which has a very high slew rate, faster than the amplifier feedback system is able to properly respond to. (See section 7).

3. How Is Intermodulation Distortion Measured?

Traditionally, IM distortion has often been measured using two fixed tones, one at a low frequency and one at a high frequency. The high frequency tone was demodulated and the sum of the sideband components was then measured (Fig.5). Although this gives some useful information, the method is limited to

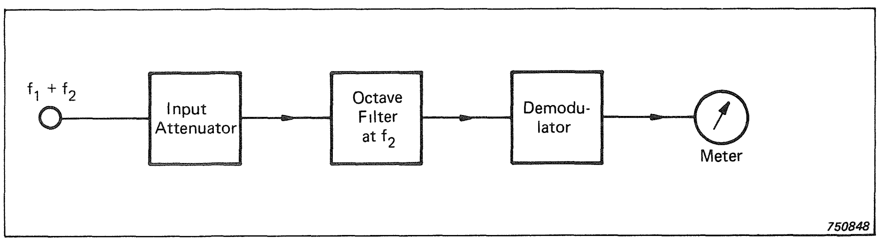


Fig.5. Traditional method of IM distortion measurement

only measuring at two fixed frequencies. It suffers from the same handicap as a distortion measurement at only one frequency — say 1 kHz. In addition, the method does not give much useful information to the design engineer about the cause of the distortion, because all the distortion components are summed in the measurement.

Another relatively simple technique has been used for measuring difference-frequency distortion. A two-tone generator with a fixed frequency difference between the tones has been used, and the distortion has been measured with a filter tuned to the frequency difference — this of course, being a fixed frequency. Although this method has the advantage of using a frequency sweep, it still only measures one distortion component, the second order difference component.

The ideal method of measuring intermodulation and difference-frequency distortion should permit measurements as a function of frequency, with provision for selecting any desired distortion component. This need is fulfilled by the new Distortion Measurement Control Unit Type 1902 developed by Brüel & Kjær. (Fig.6).

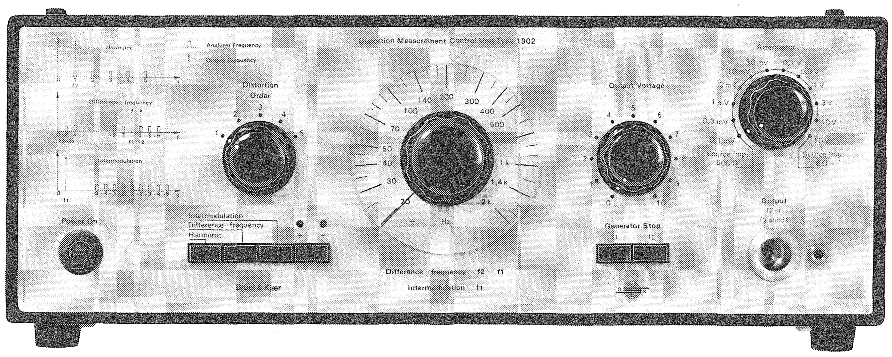


Fig.6. Distortion Measurement Control Unit Type 1902

4. A Practical Measurement System

The measurement system shown in Fig.7 permits the automatic measurement of intermodulation, difference-frequency, as well as harmonic distortion over wide frequency and dynamic ranges.

Although the system is very sophisticated and of a complex design, it is easy to operate, and distortion curves can be plotted automatically as easily and quickly as frequency response curves. (A technical description of the system is found in the Product Data sheet for Type 1902).

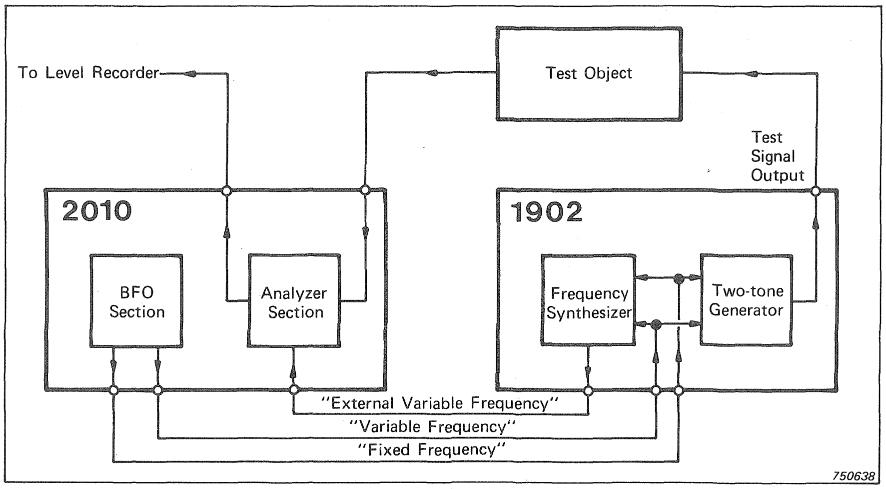


Fig.7. Principle of instruments used for distortion measurements

The system operates as follows: The Level Recorder is fitted with preprinted logarithmic paper on which the resulting distortion curves are *automatically plotted.* When the Level Recorder is running it controls the synchronous sweep of a high frequency oscillator in the Heterodyne Analyzer Type 2010 via a mechanical drive shaft (or via a voltage ramp). The frequency of this oscillator is in the 1,2 MHz to 1,0 MHz range and after mixing will be used to generate a low frequency sweep from 0 to 200 kHz. This high frequency signal is fed on to the Distortion Measurement Control Unit Type 1902 which converts the high frequency to the low frequency test tone output. In addition, the 1902 generates a second tone which is either a fixed frequency difference from the first tone (used in the difference-frequency mode) or, in the intermodulation mode, the second tone is a fixed frequency. These two tones are then applied to the object under test. These two tones are designated f_1 (lower tone) and f_2 (upper tone).

The output of the object under test is fed to the input of Heterodyne Analyzer Type 2010 for measurement of the desired distortion components. The 2010 is a tunable narrow band analyzer which gives an output voltage proportional to the distortion level at any given frequency, and feeds this to the Level Recorder to give a permanent record.

The tuning of the 2010 is achieved automatically by the Distortion Measurement Control Unit Type 1902 which generates the necessary control signal to tune the 2010 to the desired distortion component.

The 1902 is extremely easy to operate. Three push buttons on the left side of the front panel select either the harmonic, difference-frequency, or intermodulation distortion mode. The Distortion Order and + and — switches then select which distortion component should be measured. The chart printed on the front panel illustrates all the different possible combinations. The frequency control potentiometer is used to set the lower frequency f_1 in the IM mode anywhere from 20 Hz to 2 kHz. In the difference-frequency mode the potentiometer is used to set the frequency difference between the two test tones. Finally, a precision output attenuator and output voltage potentiometer permit adjustment of the output voltage over an extremely wide dynamic range from less than 100 μV to 10 V RMS.

The system has the following advantages:

1. Distortion is measured on a swept frequency basis, with automatic permanent documentation on the Level Recorder. Although spot frequency measurements can be useful, such measurements are especially in danger of being non-representative on devices with irregular response, such as loudspeakers. In addition, the system described above is totally automatic, requiring no manual tuning, nulling, or settling times.

2. Measurements are made over a wide frequency range —2 Hz to 200 kHz. This permits examination of the audio device one decade above and below the traditional audio range — which is very useful in finding problems that may result in audible degradation of the signal, or may indicate weakness in a given design. In addition, being able to make distortion measurements above 20 kHz is important because the mixing of high frequency components may result in products (difference-frequency components) which fold down into the audible range.

3. The frequency relationships between the two test tones are adjustable. As previously mentioned, the lower test tone in the IM mode is adjustable from 20 Hz to 2 kHz, instead of being fixed at the mains frequency, as is frequently the case in traditional systems. Also, the frequency difference between the two tones in the difference-frequency mode is adjustable over the same range.

4. Any amplitude relationship between the test tones is possible. In normal operation, the amplitude ratio between the two tones is 4 to 1 in the IM mode, and 1 to 1 in the difference-frequency mode. However, any arbitrary amplitude ratio can be obtained by only taking f_1 from the output of the 1902, and then taking f_2 from the BFO output of the 2010. Thus each tone has identical but separate attenuator controls over a 100 dB range. The two outputs can be combined directly without the need for any matching or isolation networks, although there will be a 6 dB drop in level.

5. The two tones are available separately and hence can be used for microphone intermodulation distortion measurements (Fig.8). For the measurement of IM distortion of a microphone, a single loudspeaker cannot be used since its distortion will probably be significantly higher than that of the microphone. But since each tone can be fed to a separate loudspeaker, intermodulation distortion can be avoided and although the harmonic distortion still may be relatively high, this will have no influence on the IM and difference-frequency measurements.

6. Any distortion component up to the fifth order is measured individually. This is especially of value to the design engineer, because it permits him to associate a given type of circuit or design technique with a given order of distortion. In addition,

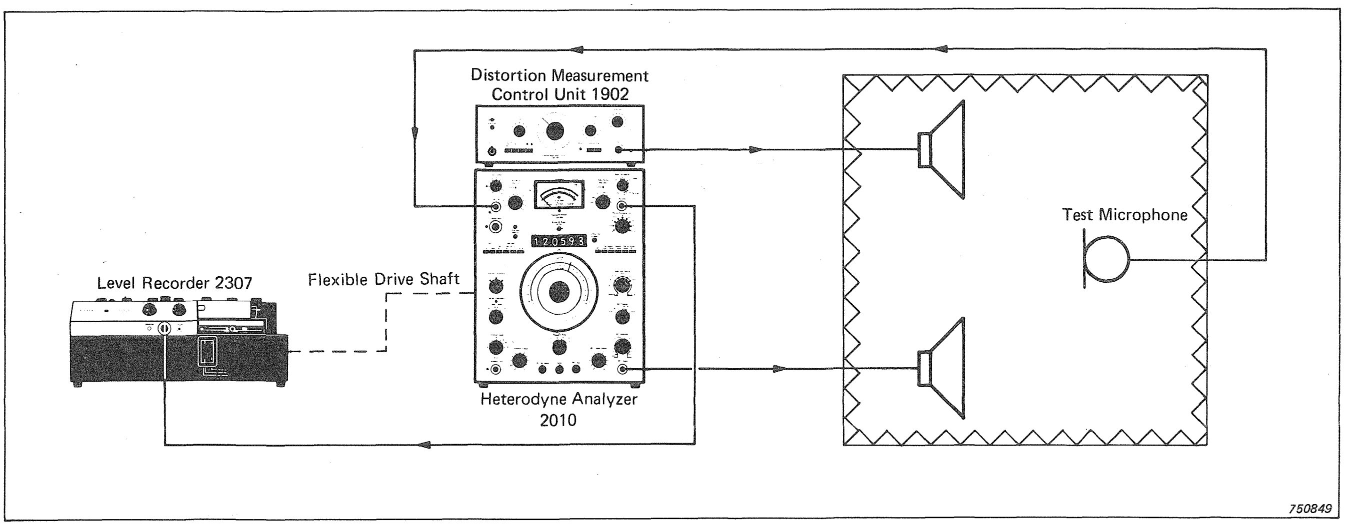


Fig.8. The two test tones are available separately permitting IM measurements on microphones

the use of very narrow bandpass filters in the analyzer (as narrow as 3,16 Hz) gives excellent noise immunity.

7. Relatively high sweep rates are permitted. For measurements on amplifiers, for example, a sweep rate of 3 mm/s on the Level Recorder may be used. This corresponds to sweeping one decade in 17 s, or the entire 20 Hz to 20 kHz range in only 50 s. Even three times as fast a sweep rate may be used without the filter dropping out or losing the distortion component it is tracking. Here the only limiting factor will be the filter response time. Figs.9 and 10 show sweeps made at 3 mm/s and at 10 mm/s over the 200 Hz to 200 kHz range. This represents a rather severe test of the tracking capability of the system, because with a logarithmic sweep, the average sweep rate at the top of the range from 100 kHz to 200 kHz is 67 kHz/s.

The only restrictions on sweep speed are those that occur due to extreme peakiness of the distortion characteristic such as found in loudspeakers and also those restrictions due to time delay in the system such as in tape recorders. These unavoidable restrictions are discussed in greater detail in section 5.

8. The system is self-checking — it can automatically document its own distortion residual. Although the system is very conservatively specified, the user can quickly document its distortion residual in any given mode by directly connecting its output to input and plotting the results on the level recorder. Thus measurements to levels much lower than those specified can often be made and the documentation is available showing what distortion is due to the test equipment, and what is due to the object under test. (Fig.11).

9. The dynamic range of the system may be extended in the intermodulation mode by adding a simple high pass filter after the object under test to attenuate the low frequency tone. A typical distortion residual curve is shown in Fig.12 which can be compared to Fig.11 which shows the residual without the high pass filter.

10. The system is a complete distortion measurement package, permitting not only IM and difference-frequency measurements but also harmonic distortion measurements. In addition, the same system may be used for the automatic measurement of frequency response, signal to noise ratio (Linear or A weighted), and noise spectral density for both electrical and acoustic systems.

11. The system permits measurements to fulfil the IEC 268-3 and DIN 45403 standards as well as the SMPTE method.

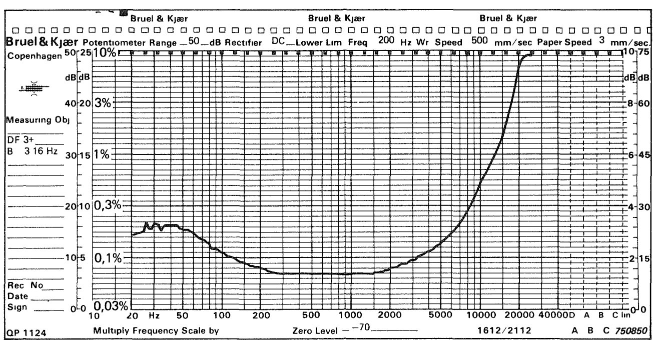


Fig.9. Distortion measurement of power amplifier using sweep rate of 3 mm/s

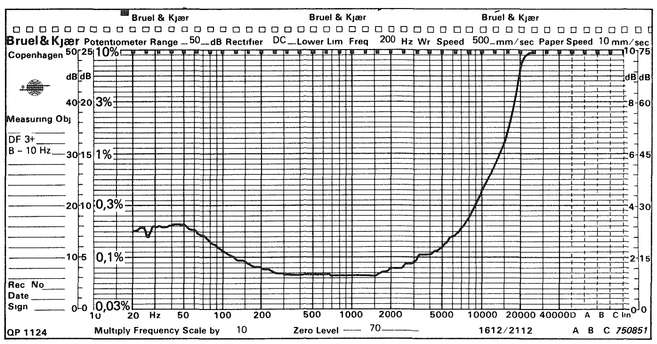


Fig.10. Distortion measurement of the same amplifier as in Fig.9 using a sweep rate of 10 mm/s

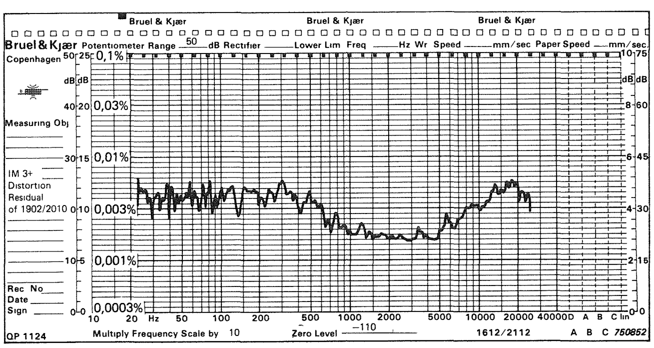


Fig.11. Distortion residual of the measuring system in the IM 3+ mode

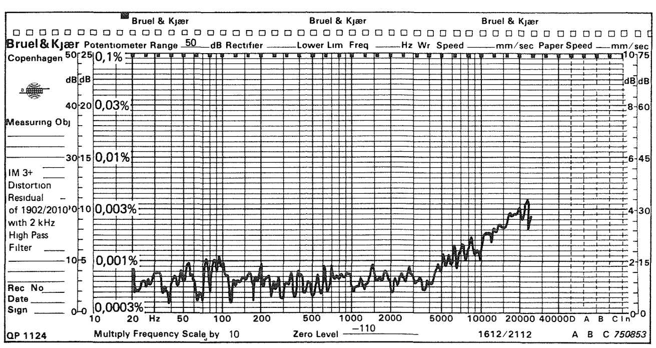


Fig.12. Distortion residual of the system is improved by adding a high pass filter

5. Practical Examples

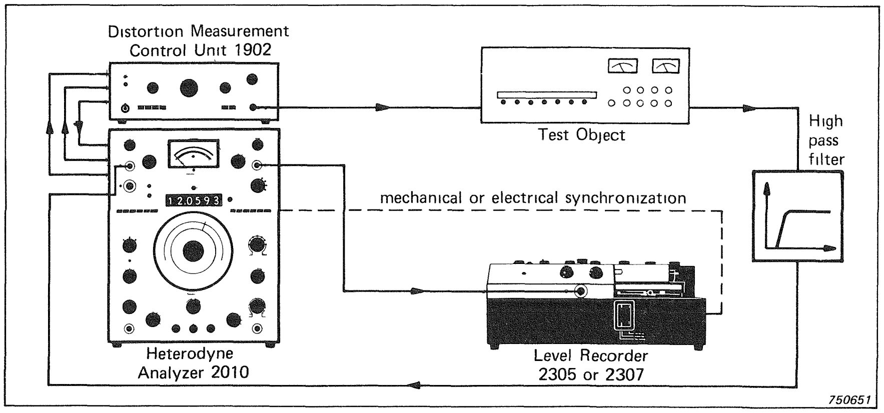


Fig.13. Instrument set-up for distortion measurements. The optional high-pass filter is used to extend the dynamic range in the IM mode

General Measurement Procedure

A typical instrument set-up for distortion measurements is shown in Fig.13. The only necessary adjustments are to select the desired distortion mode and distortion order being measured. Then set the output voltage fed to the amplifier, and measure the voltage coming out of the amplifier on the built-in meter of the 2010. The gain of the input section of the 2010 should be set so the meter gives precisely full-scale deflection with the analyzer set in the Linear mode. This guarantees optimum dynamic range of the analyzer. If this level is exceeded, the distortion specification of the analyzer is not guaranteed. This adjustment should be made with the output section attenuator of the 2010 set at x 1: The gain of the level recorder can then be calibrated to correspond to the meter deflection of the 2010. For optimum dynamic range, the output section attenuator of the 2010 should be used to increase the gain to measure lower distortion levels. This control gives as much as 60 dB extra gain and should be set such that at the maximum distortion point, the distortion gives close to full scale meter deflection. This adjustment is made with the 2010 in the "Selective" mode which of course is used for all measurements.

Traditionally, the Level Recorder is fitted with a 50 dB potentiometer. However, since the distortion level may vary over a very wide dynamic range, the use of a 75 dB potentiometer may at times be found useful.

Power Amplifiers

Power amplifiers are a popular object of intermodulation distortion measurements. Measurements are often made at several different power levels to determine the performance, both at just below clipping, at medium levels, and at low levels to test for crossover distortion. Fig.14 shows the distortion characteristics for a modern solid state receiver produced in 1975. Fig.15 shows the corresponding characteristics for another solid state power amplifier which shows higher distortion at the high frequencies — which in part is due to the use of an operational amplifier with very small open loop bandwidth in the input. Although amplifier A has higher distortion over much of the frequency range its distortion does not increase so much at the higher frequencies as amplifier B. This indicates that the design of amplifier A uses transistors with wider open loop bandwidth than amplifier B, and also uses less feedback. Therefore amplifier A would be judged to have less transient intermodulation distortion (see section 7).

Some care is required in measurement on power amplifiers. First of all, if the amplifier's output is not short circuit protected, care should be taken to prevent short circuits when connecting its output to the measuring instrumentation. This can occur, for example, when connecting the cable to the input of the 2010, and accidentally touching the center pin of the plug to the ground sleeve of the socket of the input.

A second consideration is of extreme importance in order to achieve correct measurements on power amplifiers. The input of the 2010 Heterodyne Analyzer must be connected directly at the output terminals of the power amplifier under test. It must not be connected at the load because the length of cable from the power amplifier output to the load (even with short leads) will cause a voltage drop due to the high currents produced by the amplifier. This voltage drop will be included in the measurement, since the 2010 has an unbalanced input. This is why it is so important to connect the 2010 directly to the output of power amp — and the higher the power, the more important this is.

It should also be pointed out that this instrument set-up forms a ground loop which may disturb the amplifier under test although this is usually not the case. This can readily be checked by simply disconnecting the ground lead of the input cable to the 2010 at the power amplifier end. This ground connection is not necessary for measurements since the 2010 also has a ground reference through the output cable from the 1902 to the input ground of the amplifier. However, if the input of the amplifier being tested is balanced, or if the internal ground connection between input and output of the power amplifier is not of a low enough ohmic value the ground connection to the 2010 may be necessary.

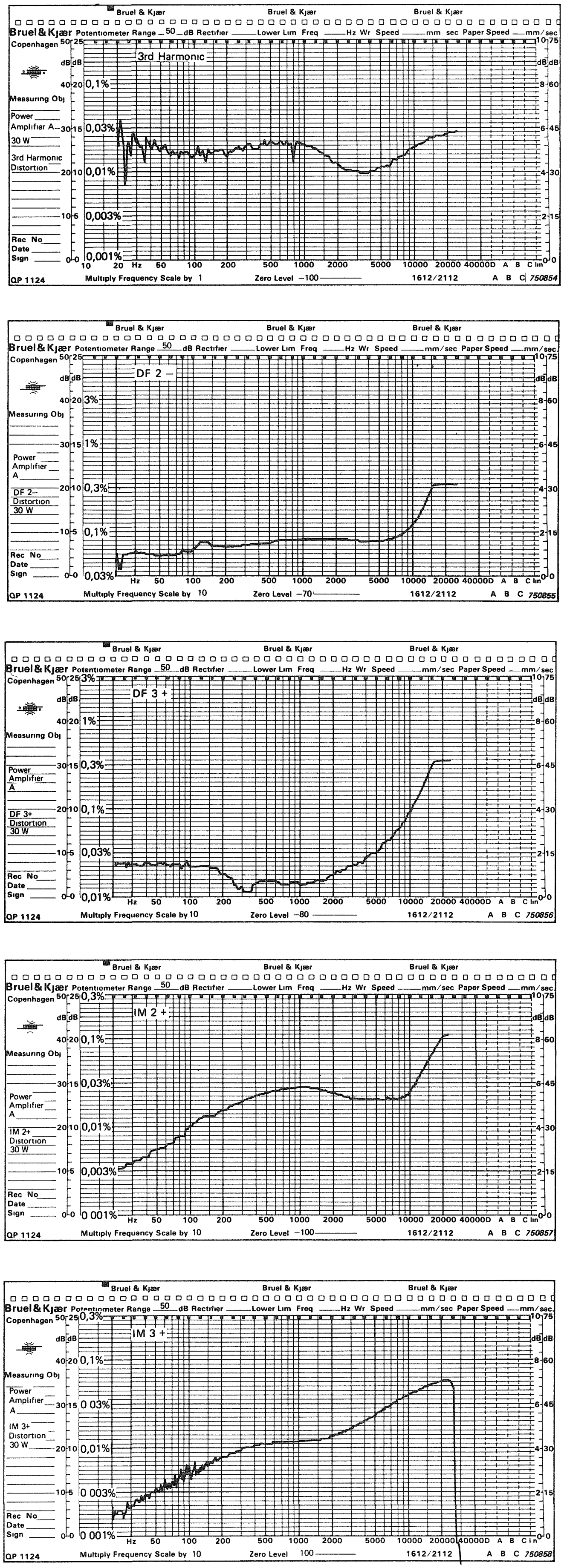


Fig.14. Distortion characteristics of power amplifier A

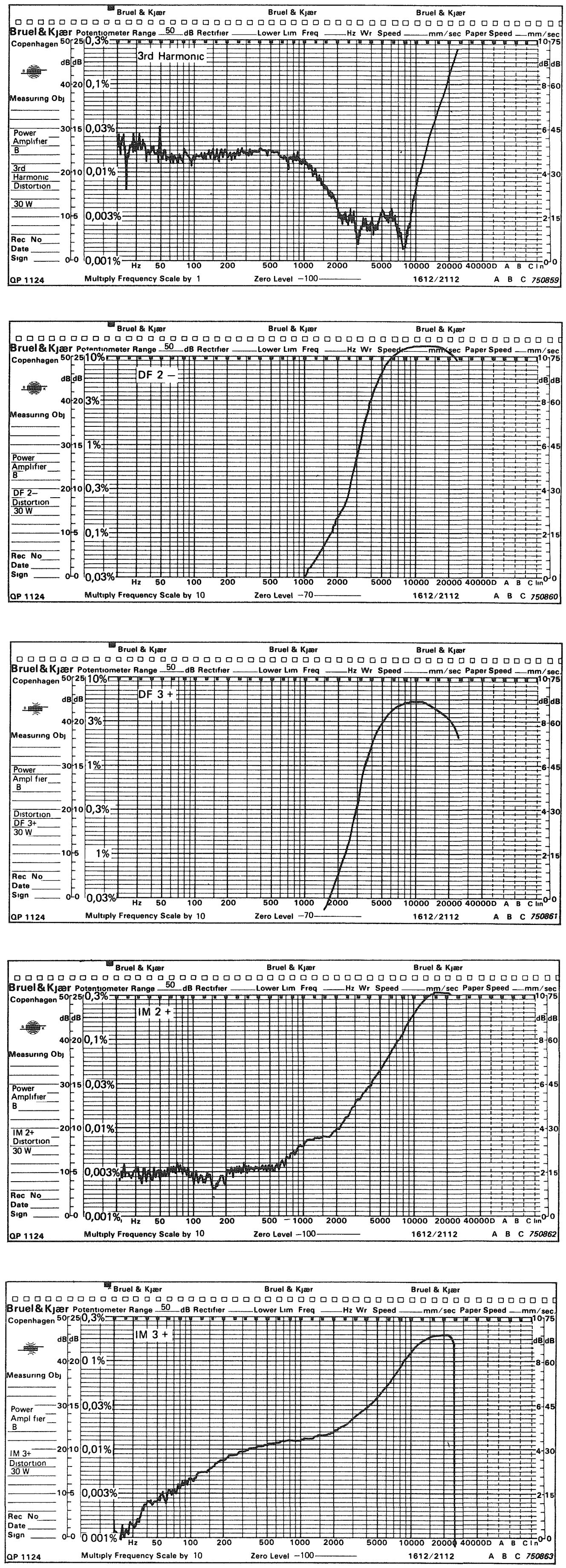


Fig.15. Distortion characteristics of power amplifier B

Tape Recorders

Although a similar technique is used to measure the record/reproduce characteristics of tape recorders, a restriction on the maximum sweep speed is introduced due to the time delay between record and reproduce heads. If the sweep is too fast, the analyzing filter will have moved on to a frequency higher than the frequency of the harmonic coming from the tape being played back. Therefore, the sweep must be slow enough so the desired distortion component is always inside the bandpass of the filter.

The nomograph in Fig.16 permits an easy determination of the optimum sweep speed for systems with a time delay. To use the nomograph follow these steps (examples given in parenthesis and indicated by the dashed line on the nomograph):

1. Draw a line from the maximum frequency (20 kHz) through the desired bandwidth (10 Hz) to the vertical line with a star. This is an intermediate stopping point.

2. From the intersection with the starred line, draw a straight line through the time delay (500 ms) of the system to the intersection with the Paper Speed line (0,01 mm/s). This gives the maximum Level Recorder Paper Speed in mm/s using a logarithmic sweep. The sweep time per decade (83,3 min) is given in parenthesis.

For harmonic distortion measurements, the Paper Speed obtained must be divided by the distortion order. For example, if a Paper Speed of 3 mm/s is obtained using the nomograph, and it is desired to measure third order harmonic distortion, the Paper Speed must be reduced to 1 mm/s.

The nomograph is based on the formula

$$P \leqslant \frac{10\,B}{f_{max.}\,t_d}$$

where P is the paper speed, B is the bandwidth of the analyzing filter, $f_{max.}$ is the maximum frequency, and t_d is the time delay of the system.

If extremely long measuring times are obtained, these can be shortened by using a wider bandwidth (as long as it does not compromise the dynamic range of the measurement) and by using a higher sweep rate at the lower frequencies.

Another restriction on tape recorder measurements is the wow and flutter which may cause the distortion components to sometimes swing outside the measurement bandwidth. This problem is most often seen using a 3,16 Hz bandwidth at the higher frequencies where the influence of wow and flutter is more pronounced. If the distortion value obtained is unstable, the bandwidth should be increased until a stable value is obtained.

When it is desired only to test the intermodulation distortion characteristics of the reproduce section of the tape recorder, a special test tape must be prepared and a Tracking Frequency Multiplier must be added to the instrument set-up as indicated in Fig.17. This is necessary when testing a tape deck that is a reproducer only, or when the recorder does not have separate record/reproduce heads.

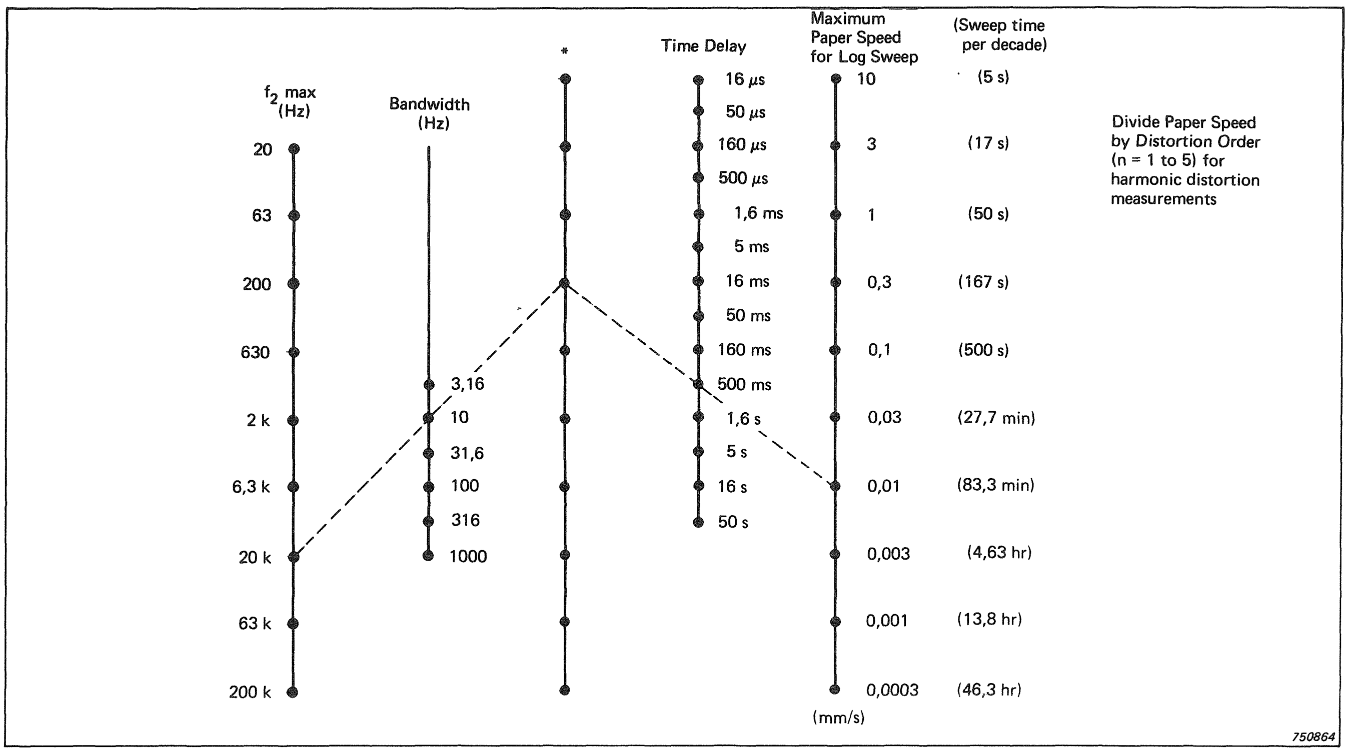


Fig.16. Nomograph for determination of the maximum sweep speeds for IM and difference-frequency measurements of systems with a time delay. For harmonic distortion measurements, the sweep rates obtained must be divided by the distortion order

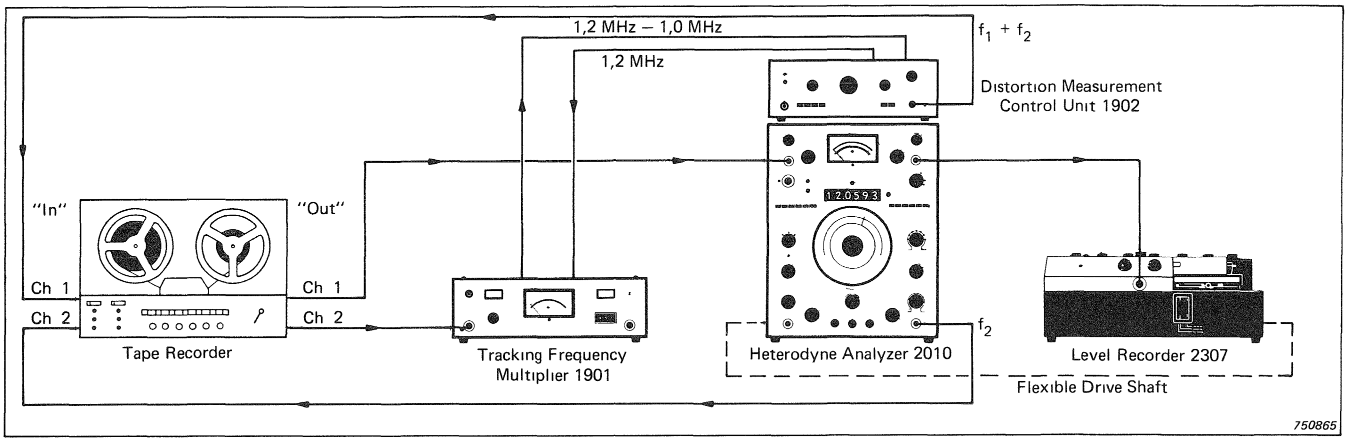


Fig.17. **Instrument set-up for distortion measurements on a pre-recorded IM test tape. Connections are also shown for recording such a test tape**

The test tape is prepared by recording f_2, the frequency of the BFO output of the 2010 on one channel to be used as a pilot control tone during playback, while simultaneously recording the two tone test signal from the 1902 output on the other channel. For playback, then, the pilot tone is fed to the input of the 1901 Tracking Frequency Multiplier, which then generates the necessary 1,2 MHz to 1,0 MHz signal to be connected to the rear of the 1902. The 1902 then tunes the 2010 to the desired distortion component. A DC output which is proportional to the input frequency of the 1901 is connected to the X-Input of the Level Recorder to control the paper throw and permit synchronization.

This test technique is also faster to use on a recorder with a large head spacing resulting in very slow sweep speeds. The test can then quickly be recorded, rewound and played back and analyzed with a sweep speed of 3 mm/s, the entire process taking about 2 min.

Typical distortion curves of a reel-to-reel recorder are shown in Fig.18 and for a cassette recorder in Fig.19.

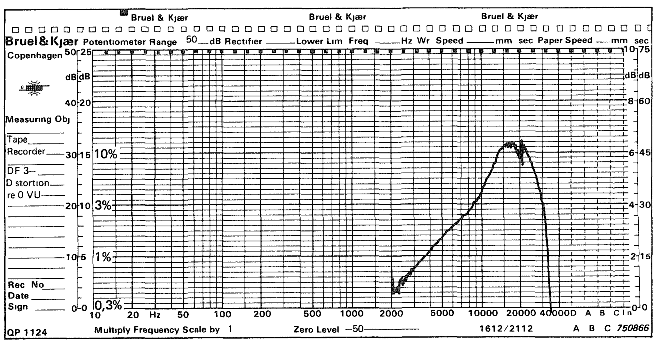


Fig.18. **Third-order difference-frequency distortion of a reel-to-reel tape recorder**

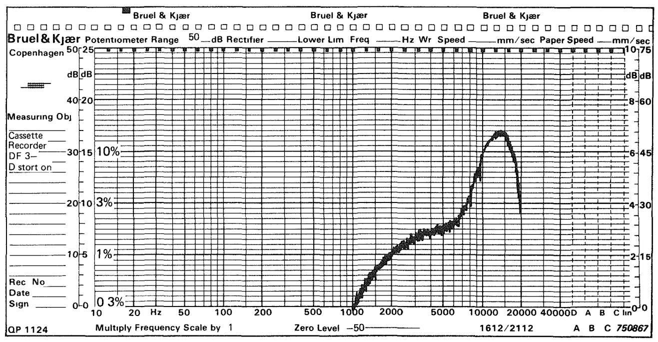


Fig.19. **Third-order difference-frequency distortion of a cassette recorder**

Loudspeakers

Measurements of the three different types of distortion may readily be made on loudspeakers using the instrument set-up indicated in Fig.20. Although there will be a time delay in the system, it will not be long enough to restrict the sweep rates in most cases. This can, of course, be determined in individual cases from the nomograph in Fig.16.

The typical results obtained are shown in Fig.21. From these figures it can be seen that the distortion characteristic is very peaky — and this can lead to some difficulty in interpretation. This peakiness corresponds to some extent to the peakiness of the frequency re-

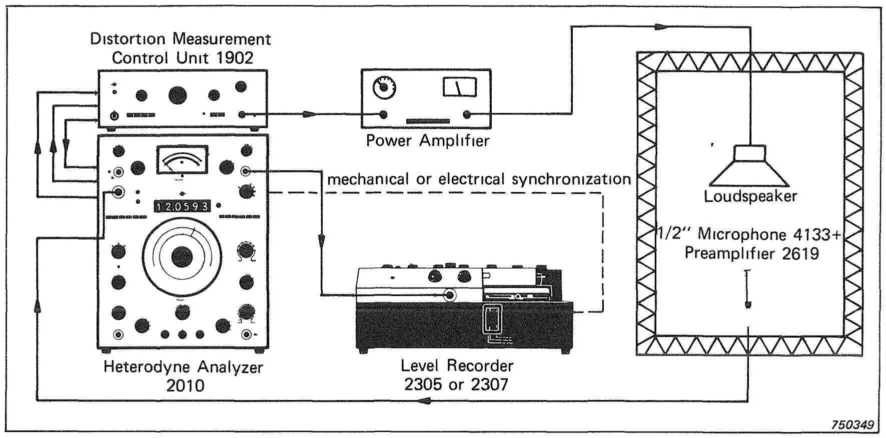


Fig.20. **Instrument set-up for loudspeaker distortion measurements**

sponse curve of the loudspeaker — and it is good practice to show both the frequency response and the distortion curves. Since the frequency response is so irregular, the reference level will be continually changing and the percent distortion must be calculated for each frequency by determining the difference between the two curves.

However, there is no economically practical method of running intermodulation distortion curves of loudspeakers while holding the level of both the tones constant. It could be done, but would require two additional tracking filters, and two separate compressor circuits. Therefore, it is practice to let the loudspeaker distortion be measured with the reference varying as the frequency response varies. This concept is termed "characteristic distortion" in the IEC 268-3 standard. In practice, such a measurement is probably a more realistic assessment of loudspeaker distortion than a measurement made with a constant output sound pressure level at all frequencies — a situation which does not occur in reality.

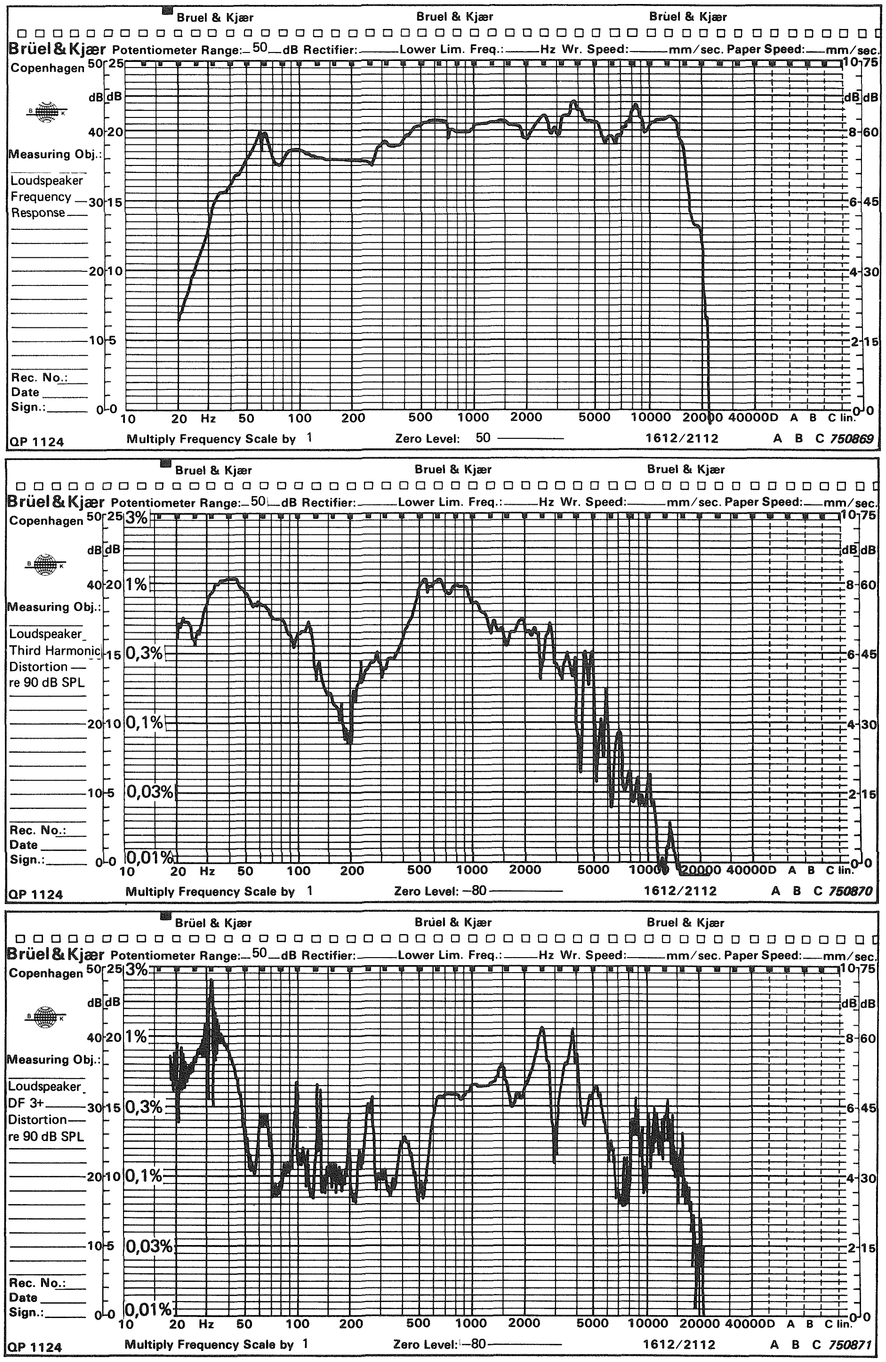


Fig.21. Frequency response and distortion characteristics of a high quality bookshelf loudspeaker

Components

Although resistors and capacitors are generally assumed to be linear devices, this is not always the case. An example of this is shown in Fig.22 which shows the distortion (IM) of a 10 nF ceramic capacitor as a function of voltage. When the capacitor was replaced by a high quality polystyrene capacitor, the distortion disappeared. This can be seen in Fig.12 where the high pass filter used contained such a capacitor.

Phonograph records and pick-ups

Some test records are available for measurement of intermodulation distortion using two fixed frequencies. The Heterodyne Analyzer may be used to measure this distortion by performing a sweep over the relevant frequency range to give a record of the frequencies and amplitudes of the various distortion components. Difference-frequency intermodulation distortion can also be measured using test records which have a test sweep of two tones separated by a fixed frequency. The Analyzer can then be tuned to the difference frequency, and the result recorded on the Level Recorder.

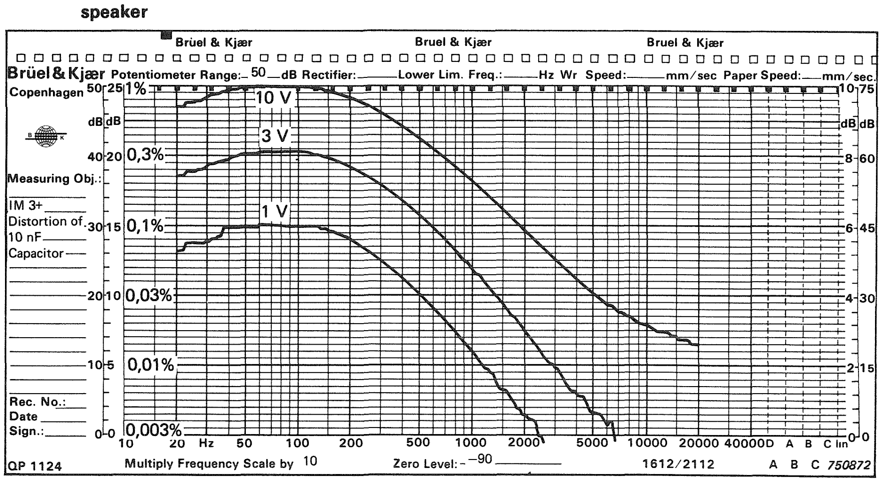


Fig.22. Intermodulation distortion of a 10 nF ceramic capacitor at various voltage levels

6. Measurement Considerations

Nature of Test Signals

It is important to understand the nature of the intermodulation and difference-frequency test signals used in order to make correct measurements. Fig.23 shows the pure sine wave used for harmonic distortion measurements. The signal shown has an RMS value of 1 V and a peak value of 1,41 V thus giving a crest factor of 1,41 (peak to RMS ratio). However, although the difference-frequency test signal shown in Fig.24 has the same RMS value as that of a pure sine, its peak value is 2 V thus giving a crest factor of 2. Finally, the IM test signal is shown in Fig.25. The amplitude ratio of the lower to the upper tone is 4 to 1. The signal shown also has an RMS value of 1 V, and the peak value is 1,71 V thus giving a crest factor of 1,71.

These various differences in the peak values of the test signals are pointed out because various standards require that the reference level used be the RMS value of the sine wave that has the same peak value as the two tone signal used. Thus, for example, for a two tone difference-frequency signal with an RMS voltage of 1 V, the reference level would be 1,41 V, since such a sine wave would have a peak value of 2 V, which is the same as the peak value of the two-tone signal.

One reason for using a reference level equivalent to the peak level is to prevent amplifier overload when measuring distortion characteristics at full power output.

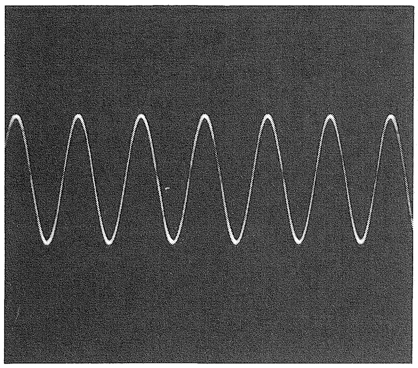


Fig.23. 1 V RMS sine signal

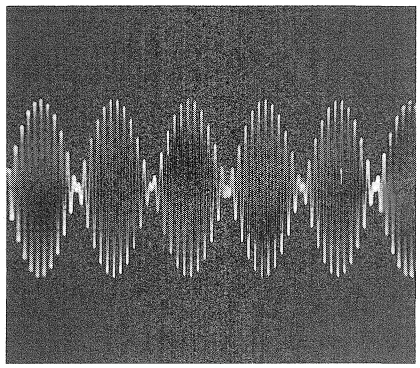


Fig.24. 1 V RMS difference-frequency signal, amplitude ratio of 1 to 1

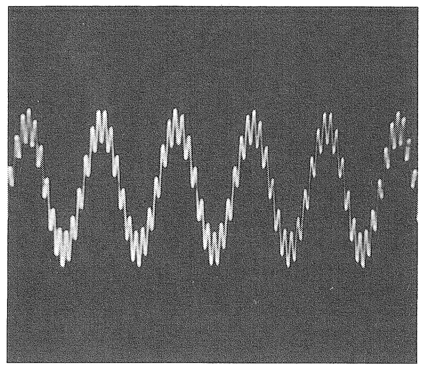


Fig.25. 1 V RMS intermodulation test signal, amplitude ratio 4 to 1

Filter Bandwidths

Six different bandwidths are available in the Heterodyne Analyzer Type 2010 ranging from 3,16 Hz to 1000 Hz. The proper bandwidth choice is a compromise between fast response time offered by the wide bandwidths, and the necessity to have a narrow enough filter to prevent measuring more than the desired distortion components.

The Fig.26 shows the filter skirt characteristics of the 2010. From this figure it can be seen for example, that the 100 Hz filter gives 55 dB attenuation 200 Hz away from its center frequency. Assume that it is desired to measure second order positive intermodulation distortion using a low frequency tone f_1 of 70 Hz. This sideband will then be removed 70 Hz from the upper tone f_2, and assuming that a measurement capability of at least 0,01% (—80 dB) is desired, we can see from the figure that the bandwidth chosen must be 10 Hz or less.

For measurements at frequencies below 20 Hz it can be seen that the dynamic range will be reduced due to the limitations of the sharpness of the 3,16 Hz filter. For example, if second harmonic distortion of a 5 Hz tone must be measured the measurement limitation is about —43 dB or slightly less than 1%.

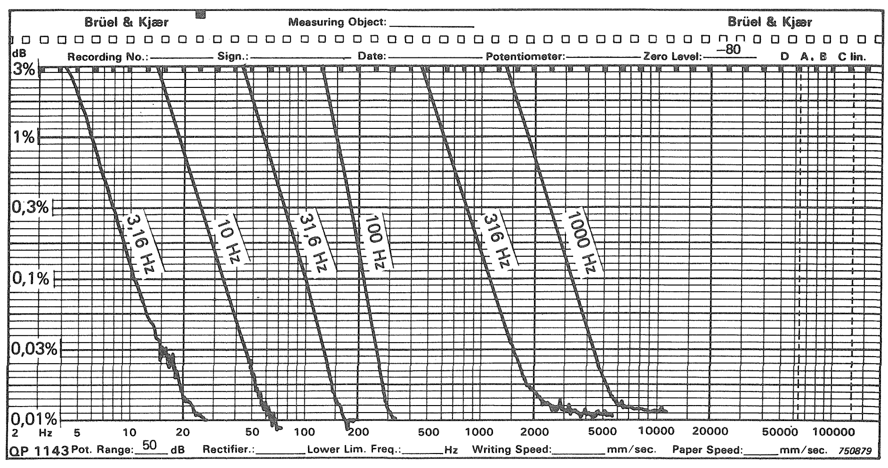


Fig.26. Filter characteristics of Heterodyne Analyzer Type 2010. The frequency scale shows the frequency above the center frequency of the filter which is normalized to be 0 Hz in this figure

Extending IM range

As already mentioned, the range of IM measurements may be extended by introducing a high pass filter after the measuring object so that the amplitude of the low frequency tone is reduced, thus permitting the input level to the 2010 to be increased giving greater dynamic range. Usually a simple CR filter such as the one shown in Fig.27 will be sufficient. Measurements can still be made below the cut-off frequency of 1,6 kHz provided that the resulting curves are corrected for the frequency response of the filter. It is important that the filter be of high enough impedance to not significantly load the object under test. In addition, the choice of components is important. As already seen in section 5, only high quality capacitors with low distortion can be used.

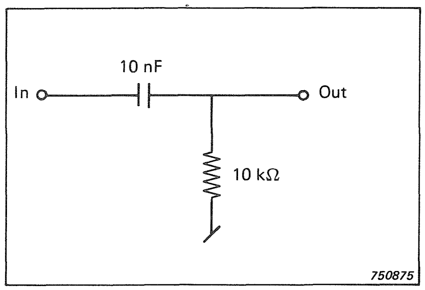


Fig.27. Simple high-pass filter to attenuate the lower test frequency for IM measurements

7. Transient Intermodulation Distortion

Transient Intermodulation Distortion (TIM) arises in amplifiers due to the time delay in the feedback loop, and the resultant clipping of the amplifier when a sudden change is applied to the input. This phenomenon can be seen by applying a square wave and observing the waveform at the feedback point of the amplifier where the overshoot is clearly seen. However, by observation of the output waveform, no degradation of the square wave quality is seen. However, if a high frequency sine wave is superiomposed on the input square wave, the sine wave will be seen to be distorted during the time that the amplifier is clipping due to the overshoot on the input transient. Once the amplifier has clipped the overshoot of the feedback signal, there will be a period of total amplifier blocking — or 100% distortion. The length of this period will be proportional to the amount of energy in the overload and the "slowness" of the amplifier.

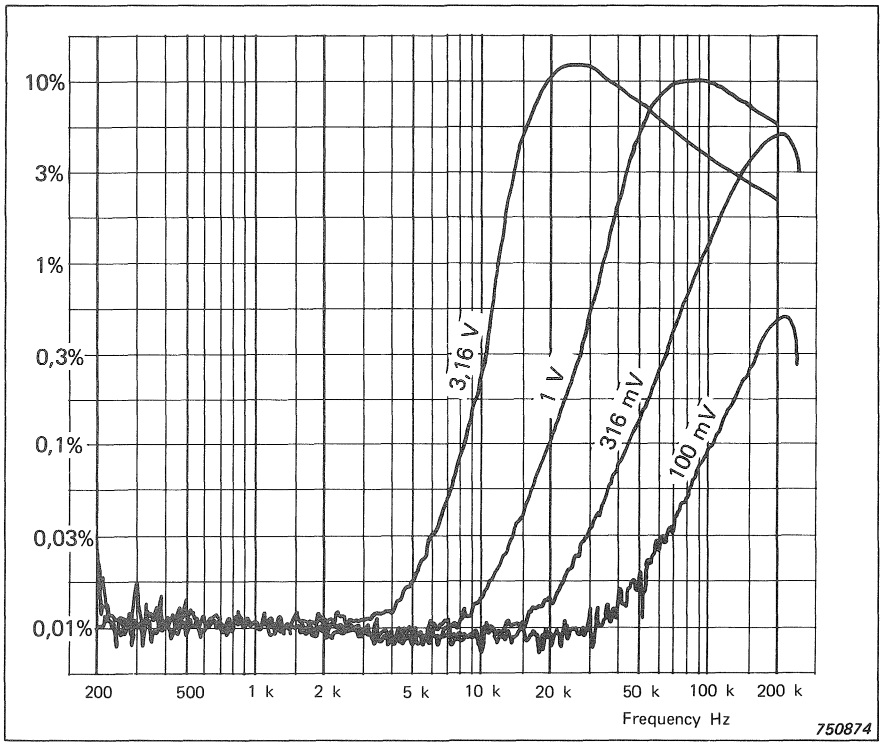


Fig.28. Third-order difference-frequency distortion of an operational amplifier for various voltage levels. The op amp operates at unity gain

Although TIM may be observed using these special techniques, it is also related to the power bandwidth and slew rate of the amplifier. At higher frequencies, restrictions in bandwidth and slew rate result in increased distortion, so that high frequency distortion measurements can be used as an indirect measure of TIM. This can be seen by comparing the two power amplifiers in Figs.14 and 15.

A particularly bad example of high frequency distortion is high gain operational amplifiers with narrow open loop bandwidths. The distortion characteristics of such an op amp are shown in Fig.28. Here it is seen that there is a point where the distortion begins to rise due to decreased feedback and the slowness of the feedback loop. The amplifier shown is operated at 0 dB gain, but has an open-loop gain of 105 dB and an open-loop bandwidth of about 5 Hz. Although this heavy feedback gives very low distortion figures at lower frequencies, it results in very poor performance with respect to transient intermodulation distortion, as can be seen by the high frequency distortion characteristics.

8. Mathematical Analysis of Intermodulation Distortion

Intermodulation distortion arises due to unlinearities in the amplitude transfer characteristic of the system. These unlinearities can be described by raising the input signal function to various integral powers. Thus a second-order unlinearity may be described by raising the two tone signal to the second power:

$(A \sin \omega_1 + B \sin \omega_2)^2$

And by raising it to the third power, the third order components are obtained, and so on. This has been done for up to the fifth power and the results are shown in Figs.29 and 30. In the figures the input signals, ω_1 and ω_2 are of unity amplitude and the resulting frequencies and amplitudes are shown when they are raised to the given power. Due to their length, the resulting equations are not given.

In practice it must be remembered that a system contains many different orders of non-linearities. Hence, the resulting spectrum will be a combination of the distortion products shown in the different figures. It should also be noted that the distortion components shown in the figures are for systems which contain 100% of the given squaring, cubing, etc. characteristic. Therefore, the amplitude of the distortion components is so high related to the fundamentals. However, in a practical system, a squaring function may only constitute 1% of the transfer characteristic and hence the amplitude of the distortion components shown in the figure will be reduced proportionately although their relative amplitudes remain the same.

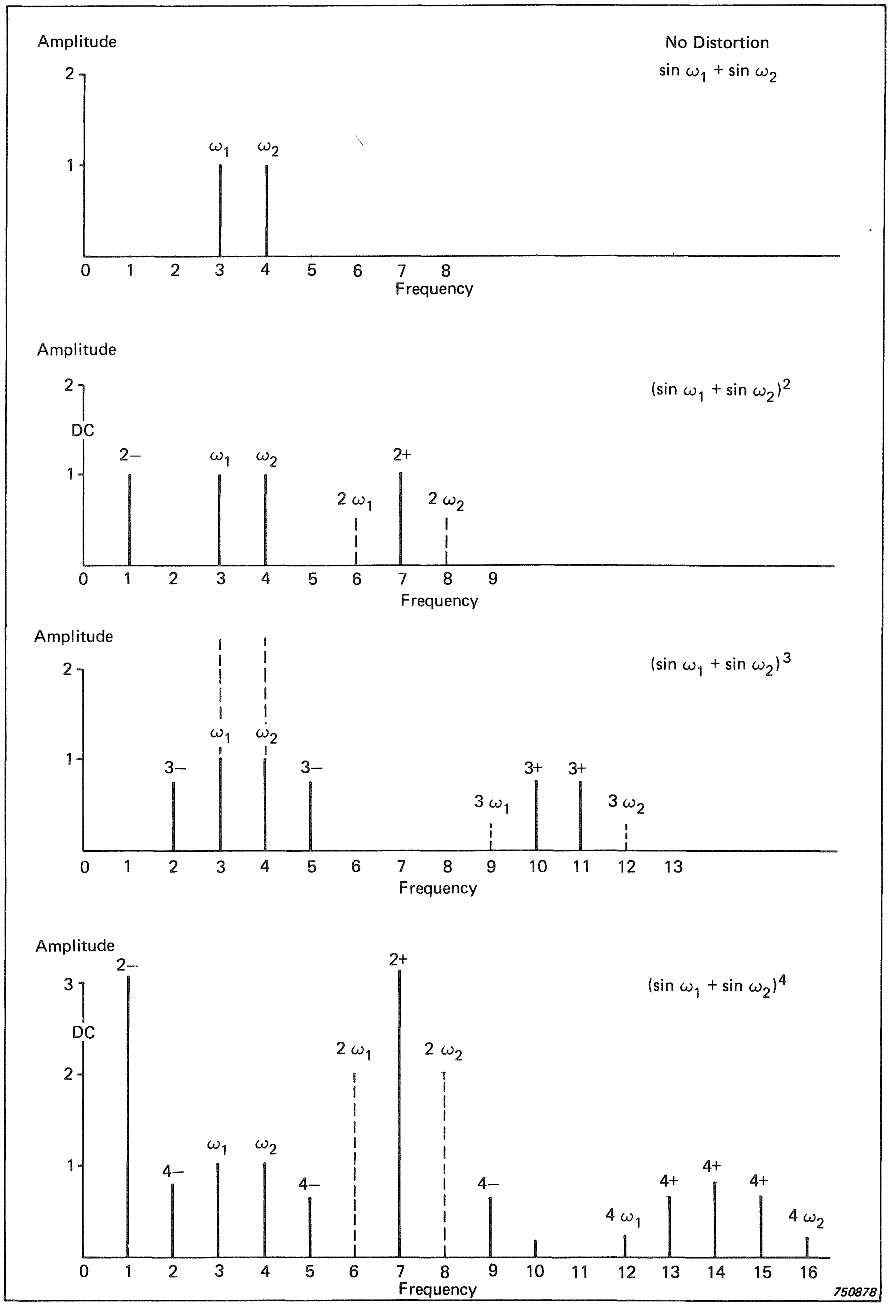


Fig.29. Frequency spectra resulting from raising a two-tone signal to the first, second, third and fourth powers

By examining the figures several facts become apparent:

1. The level of IM components is equal to or greater than the harmonic components.

2. There are more IM components for a given distortion order than harmonic components.

3. Even order distortion results in a DC component.

4. Odd order distortion changes the amplitude of the fundamentals.

5. Symmetrical sidebands are of equal amplitude.

6. Unlinearities of the n^{th} order result in distortion products not only of the n^{th} order, but also of the n-2, n-4, etc. order.

By examination of the equations, we see that even order distortion gives all cosine terms, while odd or-

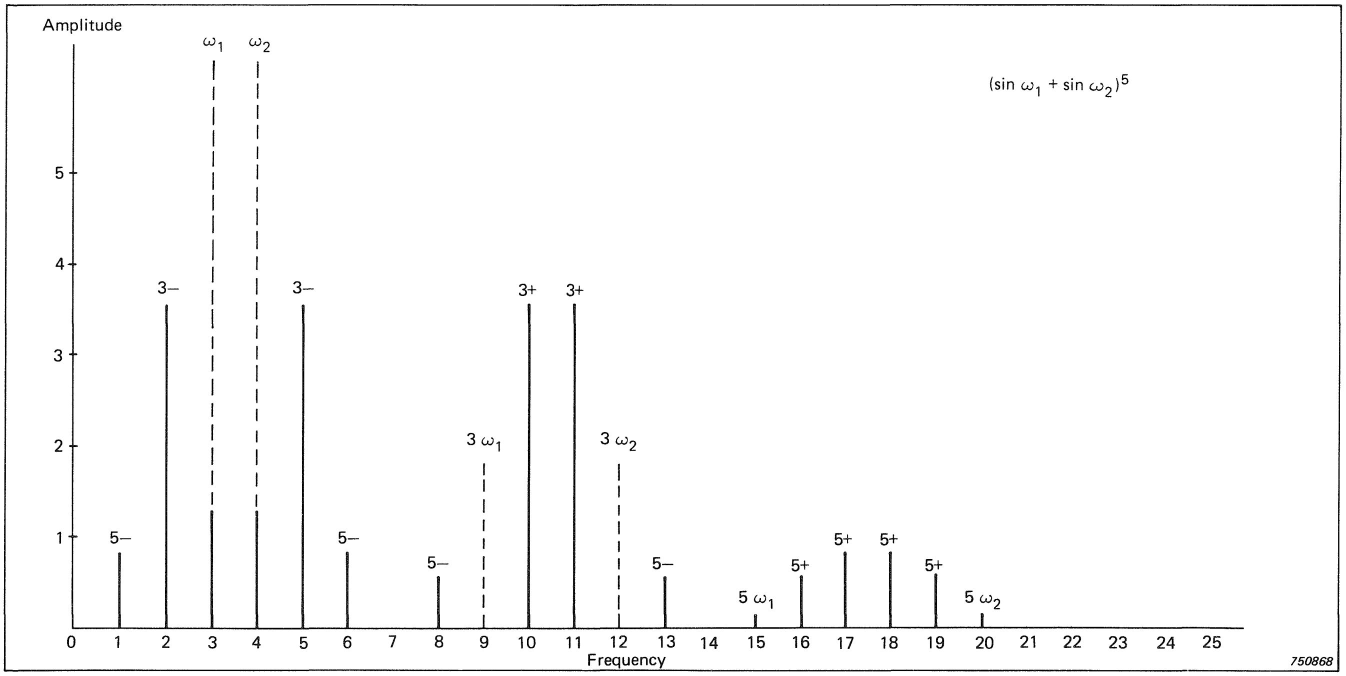


Fig.30. Frequency spectrum resulting from a fifth order unlinearity

der distortion gives sine terms. Hence even and odd order distortion components are 90° out of phase.

Because of these complex factors, it can be seen that even if the harmonic distortion characteristic is known in great detail, it will still be very difficult to calculate the intermodulation distortion, although there appears to be a relatively simple mathematical relationship.

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