# A Method for Measuring Transient Intermodulation Distortion (TIM)*

**EERO LEINONEN, MATTI OTALA, AND JOHN CURL**

*Technical Research Centre of Finland, Electronics Laboratory, SF-90101 Oulu, Finland*

A description of the square-sine intermodulation measuring method is given. The practical measuring procedure is described, and measurement results of commercial audio power amplifiers and integrated operational amplifiers are presented, in many cases showing appreciable transient intermodulation in units which have excellent total harmonic distortion and SMPTE intermodulation distortion specifications. The correlation of these measurements with results obtained using other methods, such as the CCIF-IM and the noise-transfer method, is treated, and the interdependence of these with the slew rate and the power bandwidth is discussed.

**INTRODUCTION:** The lack of correlation between conventional amplifier distortion measurements and listening tests has been noted by many designers of audio equipment. Modern amplifiers often measuring under 0.01% total harmonic distortion at 1 kHz or below 0.1% intermodulation distortion, as measured with the SMPTE method, may sound completely unacceptable. It appears therefore that the audible difference between various amplifiers is not due to these static distortion figures alone, but some other reason must be involved. One possible reason is dynamic intermodulation distortion, which is created by the frequency rather than the amplitude characteristics of the signal. One form of dynamic intermodulation distortion is the transient intermodulation (TIM) which has been described in detail elsewhere [1]–[3].

So far TIM has been measured by determining the onset of suppression of the internal drive signal in an amplifier [4], but this has only yielded a qualitative limit beyond which TIM can be expected. The aim of the present work has been to quantitatively measure this form of dynamic intermodulation distortion without the necessity of dealing with the internal circuitry of an amplifier. The method proposed here will measure both static and dynamic intermodulation distortions and can be used without any knowledge of the out-of-band behavior of the unit to be measured.

## MEASUREMENT METHOD

The measurement setup is shown in Fig. 1. The test signal consists of a low-pass filtered square wave and a sinusoid, having a peak-to-peak amplitude ratio of 4:1.

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The sinusoid should have as high a frequency as possible within the audio band, and 15.00 kHz is chosen as it is the highest specified frequency for broadcast use. Furthermore, it allows measurements of tuners, radio links, etc., as well as lower class audio equipment. For AM broadcasting and certain tape recorders, a lower frequency, for instance 6 kHz, can be chosen.

The selection of the square-wave frequency must be done so that the harmonics of it do not overlap with the sinusoid nor with the intermodulation products.

It has been shown elsewhere [5] that an optimum separation is obtained when the frequencies of the sinusoid and the square wave are related as

$$f_2/f_1 = \{(\gamma + 1)\,[\gamma(\gamma + 1)]^{\frac{1}{2}}\}^{\frac{1}{2}} \tag{1}$$

or

$$f_2/f_1 = \{\gamma[\gamma(\gamma + 1)]^{\frac{1}{2}}\}^{\frac{1}{2}} \tag{2}$$

where $f_1$ is the frequency of the square wave, $f_2$ is the frequency of the sinusoid, and $\gamma$ is a positive integer.

Within the bounds given by Eqs. (1) and (2), the square-wave frequency can be chosen freely. A high square-wave frequency increases the sensitivity of the method, but a low frequency is a more realistic drive signal for the amplifier. A very low frequency could also measure possible thermal adjustment distortion phenomena often encountered in power amplifiers and operational amplifiers [6], but would at the same time require very selective measuring equipment owing to the multiplicity of closely spaced intermodulation products falling to the audio band. As a compromise, a frequency of 3.18 kHz has been chosen for the square wave, corresponding to $\gamma = 4$. If a 6-kHz sinusoid is used, the preferred frequency of the square wave is 1.27 kHz.

The square wave is filtered with a single-pole low-pass filter to limit its rise time. The preferred cutoff frequency is 30 kHz (−3 dB) (1.6 dB down at 20 kHz), which is roughly equivalent to bandwidths found in many audio signal sources and lower class amplifiers. However, in many cases the modern sources furnish ultrasonic signals up to comparatively high frequencies. Quadraphonic pickups, for instance, usually have extended frequency ranges up to 50 kHz, and appreciable signal levels are produced at higher frequencies owing to record distortion. A 100-kHz low-pass filter is therefore recommended for the square wave in the case of measurements of all high quality equipment. It is to be noted that the fifth harmonic of the measurement signal is already more than −12 dB down relative to the fundamental of the 3.18-kHz square wave, without any low-pass filter. Therefore, the signal should not be too severe as far as amplifier stress is concerned.

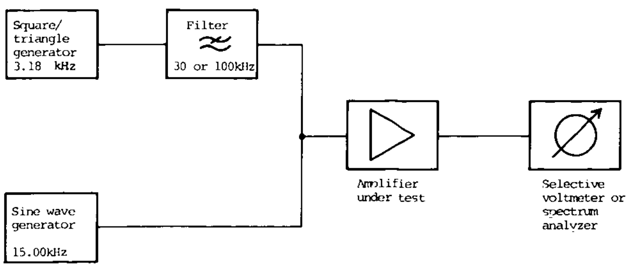


Fig. 1. Measurement setup.

To measure the intermodulation products, a selective voltmeter or a spectrum analyzer can be used. The distortion spectrum is outlined in Table I and shown graphically in Figs. 2 and 3. Fig. 2 shows the input frequency spectrum and Fig. 3 the measurement result of the popular operational amplifier $\mu$A 741, which represents a case of very strong dynamic intermodulations, where also second-order products such as $2f_2 \pm nf_1$, $3f_2 \pm nf_1$, etc. are generated. Traces of these can be found in Fig. 3.

The separation of the intermodulation products in frequency is about 1 kHz. Psychoacoustic investigations [7] seem to point out that less than 0.2% rms TIM is audible under no-masking conditions, indicating that the spectrum analyzer should have at least an 80-dB dynamic range and 500-Hz, −60-dB selectivity for reliable measurements. This may be obtained with most selective voltmeters and with some automatic graphic spectrum analyzers. Some oscilloscope-type spectrum analyzers, however, have too small a dynamic range and too poor selectivity for this measurement.

Total intermodulation distortion is given by

$$d_{\mathrm{DIM}}(\%) = 100\left[\sum_{n=1}^{9} V_{\mathrm{nt}}^{2}\right]^{\frac{1}{2}} / V_2 \tag{3}$$

where $V_{\mathrm{nt}}$ is the amplitude of each intermodulation component $f_2 - nf_1$, $n$ being a positive integer, and $V_2$ is the amplitude of the sinusoid.

Each component $V_{\mathrm{nt}}$ has two contributory parts,

1) the dynamic intermodulation component caused by the rise-time portion of the square wave driving the amplifier to frequency-dependent nonlinearity, for instance TIM;

Table 1. Signal and distortion components falling into the audio band.

| Input Signal | Distortion Component | Frequency (kHz) |
|---|---|---|
| | $f_2 - 5f_1$ | 0.90 |
| | $f_2 - 4f_1$ | 2.28 |
| $f_1$ | | 3.18 |
| | $f_2 - 6f_1$ | 4.08 |
| | $f_2 - 3f_1$ | 5.46 |
| $2f_1$ | | 6.36 |
| | $f_2 - 7f_1$ | 7.26 |
| | $f_2 - 2f_1$ | 8.64 |
| $3f_2$ | | 9.54 |
| | $f_2 - 8f_1$ | 10.44 |
| | $f_2 - f_1$ | 11.82 |
| $4f_1$ | | 12.72 |
| | $f_2 - 9f_1$ | 13.62 |
| $f_2$ | | 15.00 |
| $5f_1$ | | 15.90 |
| | $f_2 - 10f_1$ | 16.80 |
| | $f_2 + f_1$ | 18.18 |
| $6f_1$ | | 19.08 |
| | $f_2 - 11f_1$ | 19.98 |

2) the static intermodulation component caused by the amplitude-dependent nonlinearity of the amplifier.

The two components are in principle orthogonal and add vectorially, the results depending on the exact phase difference of the components. In order to separate the static intermodulation component, the square wave may be changed to a triangular wave of equal peak-to-peak amplitude. This decreases drastically the rise time, leaving only the intermodulation components caused by static nonlinearities. Fig. 4 shows the resulting static intermodulation spectrum for $\mu$A 741 on otherwise the same conditions as in Fig. 3. As such, this part of the test is roughly analogous to the SMPTE intermodulation test method.

## MEASUREMENT PROCEDURE

1) A source of sinusoidal voltage and a source of square-wave voltage are connected to the input of the amplifier under test by means of series resistors so that the sources do not load each other. The output of the square-wave source must be filtered with a single-pole low-pass filter having a cut-off frequency of 30 kHz (−3 dB) or 100 kHz (−3 dB), depending on quality requirements of the equipment being measured. When measuring frequency-dependent circuits, for instance, phono preamplifiers employing RIAA equalization, appropriate counter-equalization must be used to create a nominal-level test signal in the output.

2) The frequency of the sinusoidal source is adjusted to 15.00 kHz and the frequency of the square-wave source to 3.18 kHz. The amplitudes are adjusted so that the amplifier is operating under desired working conditions and the ratio between the peak-to-peak voltages of the sources is 4:1. This is equivalent to

a) the ratio of 5.66 :1 between the rms amplitudes
b) the ratio of 11.3 : 1 between the peak-to-peak value of the square wave and the rms value of the sinusoid.

The resulting peak-to-peak voltage (that is, 1.25 times the square-wave peak-to-peak voltage), measured at the output, is taken as reference output voltage. When documenting the test conditions, the corresponding output power is measured by replacing the test signal with a 3.18-kHz sinusoidal voltage of equal peak-to-peak amplitude.

3) The amplitude of the intermodulation products at the output of the equipment under test is measured, and the total distortion is calculated by root-mean-square (rms) summing all the products falling into the specified audio band. The distortion percentage is calculated by taking the rms sum of the intermodulation products, dividing it by the amplitude of the 15.00-kHz test signal component at the output, and multiplying the quotient by 100. This is the total intermodulation distortion, composed of the static and the dynamic components.

4) To measure the static intermodulation distortion only, the square-wave component is replaced by a triangular wave of the same frequency and equal peak-to-peak amplitude. The distortion is calculated as above.

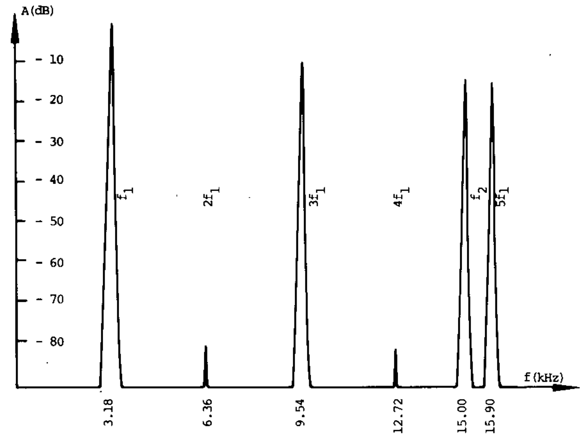


Fig. 2. Frequency spectrum of input signal.

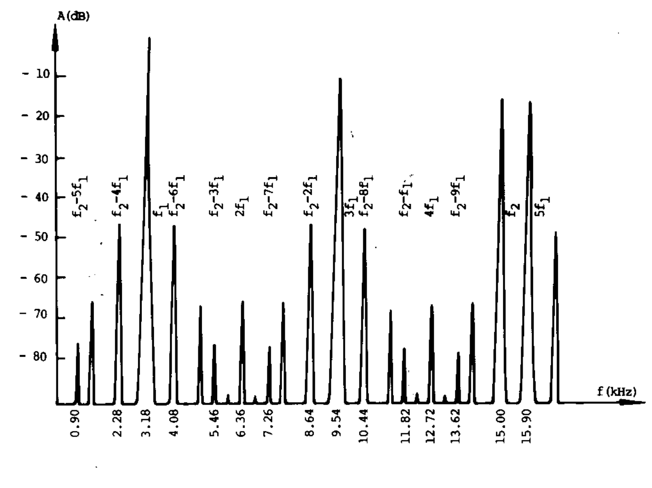


Fig. 3. Frequency spectrum of output signal of operational amplifier $\mu$A 741. Conditions—noninverting circuit; 20-dB gain; 5-k$\Omega$ load resistance; 5-V output voltage peak to peak; ± 5-V supply voltage.

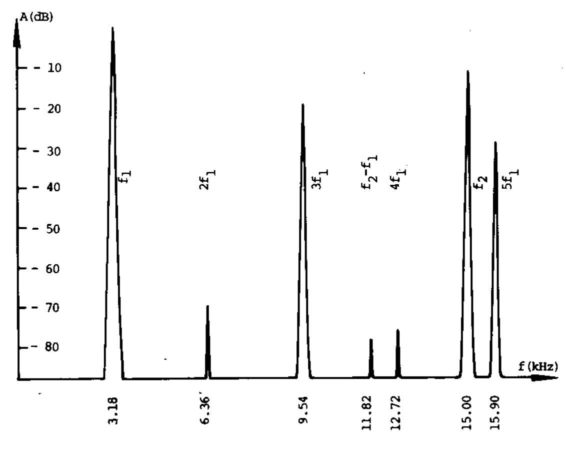


Fig. 4. Static intermodulation of operational amplifier $\mu$A 741 obtained by replacing the square wave with a triangle wave of equal frequency and peak-to-peak amplitude. Otherwise same conditions as in Fig. 3.

## INTERPRETATION OF THE RESULTS

The dynamic intermodulation distortion products are often generated by TIM, that is, the rapid rise of the square wave driving the amplifier near the onset of the slew rate limit [3], [5]. The type of the limiting mechanism can be deduced from the intermodulation spectrum by comparing the relative amplitude of the different products as follows.

1) If the amplitudes of the even products ($f_2 - 2f_1$, $f_2 - 4f_1$, . . . ) are dominant as, for instance, in Fig. 3, the limiting mechanism is symmetrical with respect to positive and negative slewing. In the case of perfect symmetry, the odd-order products ($f_2 - f_1$, $f_2 - 3f_1$, etc.) vanish. The higher the amplitudes of the odd products are, the more unsymmetrical the limiting mechanism is. If the even-order and odd-order products have equal amplitudes, the limiting is completely one sided.

2) If the amplitudes of the $f_2 - 2f_1$ product and the $f_2 - 8f_1$ product are about the same, the limiting mechanism is abrupt.

3) If the $f_2 - 8f_1$ product is small compared with the $f_2 - 2f_1$ product, the onset of the slew rate limit is gradual.

It is commonplace to find abrupt symmetrical limiting in well-designed operational amplifiers and gradual unsymmetrical limiting in power amplifiers and old operational amplifiers.

## TEST RESULTS

### Operational Amplifiers

The dynamic intermodulation was measured from nine operational amplifiers. These included the popular types $\mu$A 709, $\mu$A 739, $\mu$A 741, and LM 301, frequently used in audio circuitry, and some newer fast and ultrafast types such as LM 318, LF 356, LF 357, MC 1456, HA 2505, and CA 3140. Measurements were made using recommended operating voltages, a noninverting circuit, 20-dB gain setting, and 5-k$\Omega$ load resistance, unless otherwise specified. The recommended compensations for 20-dB gain and for unity gain were used.

The results are shown in Fig. 5. It shows that operational amplifiers $\mu$A 709, $\mu$A 739, $\mu$A 741, LM 301, and MC 1456 exhibit strong dynamic intermodulation even at low output voltages. This high distortion completely excludes the use of $\mu$A 741 in audio circuits, and restricts the use of $\mu$A 709, $\mu$A 739, and LM 301 to gains greater than 20 dB, and to output voltages less than a few volts peak to peak. Even then, extreme precaution must be taken to check possible distortion caused by improper compensation. Irrespective of the strong dynamic intermodulation, the static intermodulation values and the total harmonic distortion are extremely good for these units as will be shown later.

No dynamic intermodulation was found in LM 318, LF 356, LF 357, and HA 2505, although some very weak static intermodulation was found. Characteristic of these amplifiers was their high slew rate.

### Power amplifiers

Eight popular power amplifiers/receivers were tested. The amplifiers were Sansui 771, Marantz 2270, Harman Kardon 230 A, Sony TA 5650, Salora 2000, Dux TA 4000 (= Philips), Pioneer SX 535, and Tandberg TR 2075. The amplifiers were chosen to represent the newest medium-price generation in the European market. Each unit was tested using specified output load resistances and with tone controls carefully adjusted to flat frequency response. The volume control was set to maximum and the test signal was fed to the AUX-input of each amplifier. 30-kHz filtering of the square wave was used. For reference, the dynamic intermodulation was also measured without this filter, because in a good design, the preamplifier should limit the frequency response to that allowable for the power amplifier. The static intermodulation was also measured and all the results are shown in Figs. 6 to 9. The amplifiers appear in random order, with the numbering bearing no correlation to the order of the amplifiers in the list above.

The results show that some dynamic intermodulation is present in all the amplifiers tested. The contribution of dynamic distortion was small or medium in amplifiers 3 and 5 to 8, whereas amplifier 4 is an example of reasonable static intermodulation results combined with dramatic dynamic intermodulation increase for higher power levels.

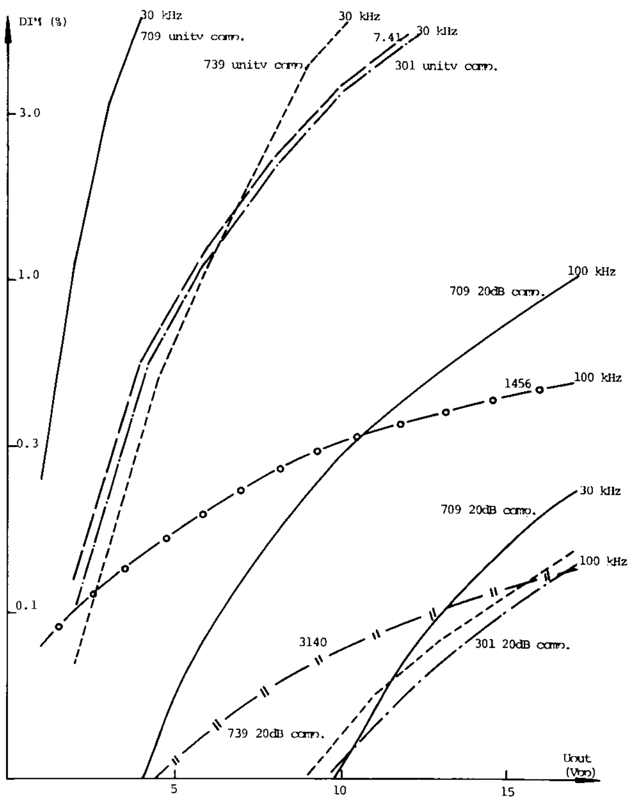


Fig. 5. Dynamic intermodulation distortion of different operational amplifiers as a function of output voltage, with square-wave filtering frequency as a parameter. Conditions—noninverting circuit; 20-dB gain; 5-k$\Omega$ load resistance; $\pm$15-V supply voltage. Types of amplifiers—$\mu$A 709, $\mu$A 741, MC 1456, LM 301, $\mu$A 739, LM 301, and CA 3140.

## CORRELATION WITH OTHER METHODS

Two standardized measurement methods are frequently used to specify audio amplifier distortion. These are the total harmonic distortion measurement method and the SMPTE intermodulation distortion measuring method, described in detail elsewhere [8]. A third method, the CCIF intermodulation method [8], has recently been used on certain high-quality products, and some reviewers have recently used a fourth method, the so-called noise-transfer method [9]. It is interesting to try to establish the correlation between the results obtained with all these methods.

The total harmonic distortion method uses a pure 1-kHz sinusoid as the input signal. The measured amplitudes of the harmonic frequencies at the output of the equipment under test are rms added and divided by the fundamental frequency to yield the distortion percentage. This test measures purely the static harmonic distortion. If higher test frequencies are used, several uncertainties of correctness are encountered, mostly caused by the fact that the harmonic frequencies to be measured may lie outside the passband of the equipment under test. Some of these uncertainties are outlined in the following sections.

The SMPTE-IM method uses two signals, a low-frequency signal, usually in the range of 81–400 Hz, and a high-frequency signal, usually 7 kHz. The amplitude modulation of the high-frequency signal is measured and given as a percentage. This method measures purely static intermodulation distortion at the frequency of the low-frequency signal.

The CCIF-IM method uses two closely spaced high-frequency sinusoidal test signals and measures their interference products. It measures both the static and the dynamic intermodulation, depending on the test frequencies.

The noise-transfer method uses as an input signal band-limited pink noise having a frequency range of 11–20 kHz. The intermodulation products of this noise at a frequency range of 0–10 kHz are measured. This test measures mostly the dynamic intermodulation.

Table II shows the measurement results for a number of popular operational amplifiers using the proposed method, as well as the four methods described above. In addition, the measured slew rate of the amplifier is shown.

The following conclusions may be deduced from the results.

1) The total harmonic distortion and the SMPTE

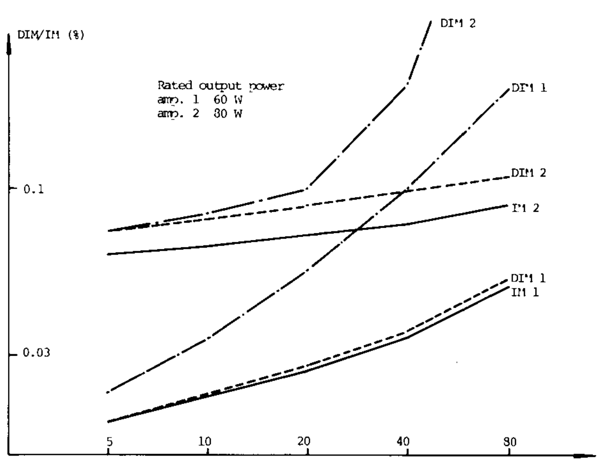


Fig. 6. Measured dynamic and static intermodulation distortion for commercial power amplifiers 1 and 2. Solid lines—static intermodulation; dashed lines—dynamic intermodulation with 30-kHz filtering of square wave; dot and dash lines—no filtering of square wave.

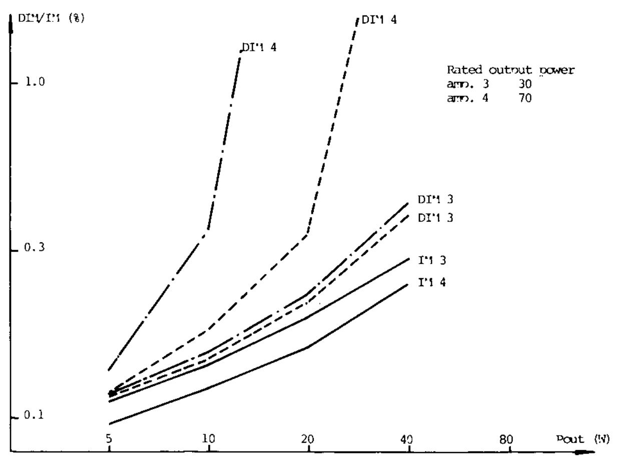


Fig. 7. Distortion characteristics of amplifiers 3 and 4. Conditions as in Fig. 6.

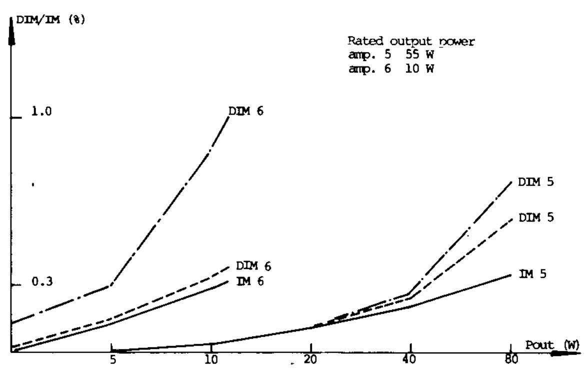


Fig. 8. Distortion characteristics of amplifiers 5 and 6. Conditions as in Fig. 6.

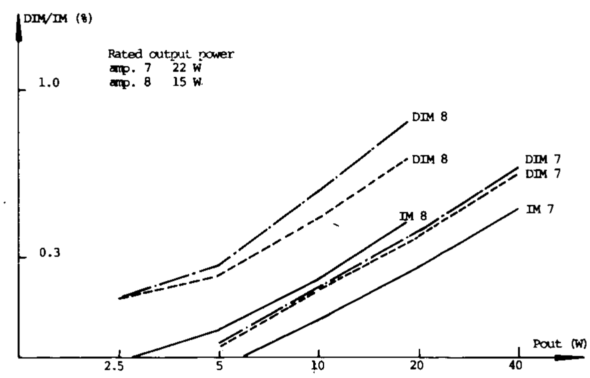


Fig. 9. Distortion characteristics of amplifiers 7 and 8. Conditions as in Fig. 6.

methods give very low distortion figures, even when the quality of the amplifier as judged with other tests is completely unacceptable. The use of the total harmonic distortion and the SMPTE-IM tests is questionable for audio equipment if dynamic distortion is suspected.

2) The noise-transfer method gives a clear indication of dynamic intermodulation distortion when it is relatively high. For the low distortion values its sensitivity seems to be rather poor, probably caused by two factors,

a) the input power density is rather small, resulting in low amplitude of the intermodulation products,

b) these intermodulation products are masked by the thermal noise.

3) The CCIF-IM method gives a reliable indication of dynamic intermodulation distortion. However, it seems to be less sensitive than the square-sine method.

4) The results of the noise-transfer method, the square-sine method, and the CCIF-IM method correlate roughly, individual departures probably depending on the slightly different sides of the more or less same phenomenon these tests are measuring.

The exact correlation between the different measuring methods is subject to a more detailed investigation [11].

## OTHER SPECIFICATIONS

Amplifiers are often characterized with the slew rate and the power bandwidth. These are basically related specifications, although usually only a first-order rough correlation exists between them, depending on their drastically different definitions.

### Slew Rate

The interdependence of the slew rate and the square-sine test results is shown in Table III. For each operational amplifier, signal levels required to produce dynamic intermodulation distortion values of 0.3%, 0.1%, and 0.05% were determined. The 0.1% value corresponds to the present knowledge of the threshold of audibility of TIM. From the signal levels obtained, the maximum rate of change of output voltage was calculated.

Depending on the compensation and the type of the operational amplifier, the distortion begins at output rates of change, measured as the rise time of the square-wave component, up to 8.7 times below the measured slew rate, thus confirming earlier predictions [2], [3]. Using this finding as a safety factor of 10 for usual signal levels of operational amplifiers and 30-kHz bandwidth, a safe minimum slew rate would be of the order of 10 V/μs, which only few operational amplifiers are able to attain. Applied to power amplifiers, the equivalent safe minimum slew rate should be around 100 V/μs, which only a few of the amplifiers on the market are able to handle. These results show that even the fastest present amplifiers must remain suspect as far as dynamic intermodulation is concerned. Correspondingly, slew rate specifications as low as 1 V/μs and 20 V/μs will probably lead into troubles in dynamic intermodulation performance.

Table II. Some operational amplifiers measured with different measuring methods.

| Type | Remarks | Square-Sine Distortion (%) 30-kHz Filter | Square-Sine Distortion (%) 100-kHz Filter | CCIF-IM, 14.00 kHz + 15.00 kHz (%) | Noise Test, Noise Level (dB) | SMPTE-IM 200 Hz + 8.0 kHz (%) | Total Harmonic Distortion, 1 kHz (%) | Slew Rate (V/μs) |
|---|---|---|---|---|---|---|---|---|
| μA 709 | 20-dB compensation | 0.039 | 0.28 | — | — | — | < 0.02 | 3.0 |
| μA 739 | 20-dB compensation | — | 0.04 | — | — | — | < 0.02 | 2.1 |
| MC 1456 | | 0.043 | 0.32 | — | — | — | < 0.02 | 1.8 |
| LM 301 | 20-dB compensation | — | 0.03 | 0.02 | — | — | < 0.02 | 1.3 |
| μA 739 | 0-dB compensation | 5.6 | 8.9 | 0.32 | −20 | 0.31 | < 0.02 | 0.64 |
| μA 741 | | 3.8 | 6.3 | 0.44 | −26 | 0.10 | < 0.02 | 0.61 |
| LM 301 | 0-dB compensation | 3.5 | 5.8 | 0.66 | −20 | 0.10 | < 0.02 | 0.58 |
| μA 709 | 0-dB compensation | 62 | 63 | 26 | −6 | 0.11 | < 0.02 | 0.20 |

General conditions: noninverting circuit; 20-dB gain; peak-to-peak output voltage 10 V; 5-kΩ load resistance; ±15-V supply voltage. Dash signifies unmeasurable distortion.

Table III. Maximum signal output rate of change (square-wave component) for different levels of total dynamic intermodulation distortion (DIM).

| Type | Remarks | Maximum Rate of Change in Output (V/μs) for 0.3% DIM | 0.1% DIM | 0.05% DIM | Measured Slew Rate (V/μs) |
|---|---|---|---|---|---|
| μA 741 | | 0.29 | 0.11 | 0.07 | 0.61 |
| LM 301 | 0-dB compensation | 0.25 | 0.11 | 0.07 | 0.58 |
| MC 1456 | | 1.8 | 1.6 | 1.2 | 1.8 |
| CA 3140 | | 13.5 | 13.5 | 10.4 | 13.5 |
| μA 709 | 20-dB compensation | — | 3.0 | 2.8 | 3.0 |
| μA 739 | 20-dB compensation | 2.1 | 2.1 | 1.7 | 2.1 |

General conditions: Noninverting circuit; 20-dB gain; ±15-V supply voltage. 2-kΩ load resistance is used in conformity with slew rate measurement practice of the amplifiers in question.

### Power Bandwidth

Power bandwidth is specified to be that frequency at which the amplifier is capable of delivering half of its rated output power with a specified maximum total harmonic distortion, usually 1% [10]. It is widely used as a criterion for the high-frequency capability of an amplifier.

Although this is true to a certain extent, there are a number of effects which make this criterion less consistent in the case of audio amplifiers. All these effects stem from the fact that in order to specify the power bandwith one must measure distortion components which are outside the passband of the amplifier.

As an illustration, consider an amplifier which has a measured power bandwidth of 20 kHz. If now an ideal 35-kHz low-pass filter is added to the output, the total harmonic distortion falls to zero per definition. The "resulting" power bandwidth in the range of 20–35 kHz then only depends on the amplifier topology and dimensioning.

The problem lies in the definition, because for real-life signals, such as music, the audible frequency band would be crowded with intermodulation products, and the sound would most probably be very bad, in spite of the fact that the total harmonic distortion would be very low and the power bandwidth would be high.

It is important to notice that it would also be impossible to use the SMPTE intermodulation measuring method to detect this intermodulation, because the SMPTE method only measures static intermodulation.

This illustration shows how a little trick could be used to fool the usual measuring methods. It is not implied that designers would use such tricks deliberately, but there exist a number of "built-in" mechanisms which perform the same operation.

1) It is usual to incorporate an RLC network in the output of power amplifier to ensure stability during capacitive output loading. This acts as a filter network above 50 kHz, and it usually decreases the total harmonic distortion some −2 to −6 dB at 20–30 kHz, depending on the harmonic spectra.

2) The amplifier feedback phase margin is usually not exactly −90 degrees. This departure may affect the amplitude of a given harmonic up to −10 to +20 dB.

3) Some of the harmonics may be outside the closed-loop bandwidth of the amplifier, above which an added attenuation of −6 to −18 dB per octave is generated. This may be rephrased by stating that most high-frequency distortion is generated in the driver circuits. The ultrasonic harmonic components may then be attenuated in the intrinsically slow power output stages.

This effect may result in some −2 to −20-dB reduction of the 20–30-kHz total harmonic distortion value, depending on the amplifier closed-loop bandwidth.

As can be seen, a number of effects decrease the reliability of the power bandwidth specification, and therefore no attempts to correlate power bandwidth with the CCIF-IM or square-sine method have been made in this project.

The definition of the power bandwidth would be much more reliable if it were not specified with 1% total harmonic distortion but, say, 0.1% CCIF-IM for any combination of amplifier passband signals. As is shown in Table II and by experience, a low value of total harmonic distortion combined with reasonable power bandwidth may in some cases indicate the presence of strong dynamic intermodulation.

## CONCLUSION

A new audio distortion test method has been proposed and applied. Its measurement results and correlations with known methods have been discussed showing that

1) many amplifiers having excellent total harmonic distortion and SMPTE-IM data show high values of distortion as measured with the proposed method,

2) the proposed method seems to yield qualitative correlation with other methods measuring the dynamic intermodulation distortion,

3) the proposed method seems to be more sensitive than these existing methods.

It is believed that the proposed test gives a stringent, but realistic, test signal to an amplifier with the option of convenient passband adjustment to simulate different signal sources and their capability to produce various levels of responses.

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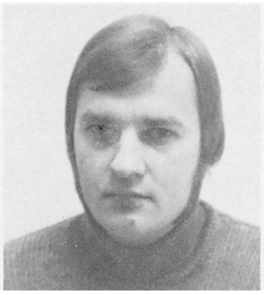
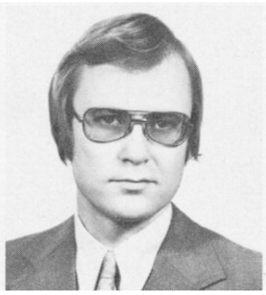
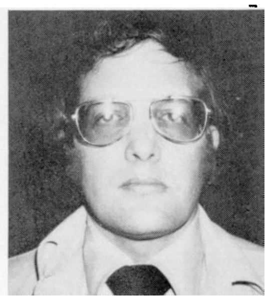


E. Leinonen



M. Otala



J. Curl

Eero Leinonen was born in Paltamo, Finland, on January 24, 1951. He received the M. Sc. degree from the University of Oulu, Oulu, Finland, in 1974.

He is presently employed by the Technical Research Centre of Finland, Oulu, where his work is concerned with audio amplifiers and microprocessors.

Mr. Leinonen is a member of the Audio Engineering Society and the Finnish Society of Electronics Engineering.

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Matti Otala was born in Oulu, Finland, in 1939. He received the M.Sc. and Lic. Techn. degrees from the Technical University, Helsinki, Finland, in 1963 and 1967, respectively, and the degree of Dr. Techn. from the University of Oulu, Oulu, Finland, in 1969.

From 1962 to 1966 he worked for Oy Helvar, Helsinki, Finland designing stereophonic equipment and later as technical manager. From 1966 to 1968 he worked for Oy Nokia Ab Electronics, microwave link division, as a project leader. In 1967 he joined the University of Oulu as Professor of Electronics, specializing in process instrumentation. From 1972 to 1973 he worked at the Philips Research Laboratories, Eindhoven, The Netherlands, on magnetic bubble technology, and in 1974 in the Centre National d'Etudes des Telecommunications, Paris, France, on time sharing telephone exchanges. In 1975 he was appointed Director of the Electronics Laboratory of the Technical Research Centre of Finland, Oulu. He has published some 60 scientific papers on audio, superconductivity, metal physics, instrumentation, magnetic bubbles, and computer memory organization. He holds a number of international patents.

Dr. Otala is a member of many Finnish and international societies.

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John Curl was born in San Francisco, California, in 1942. He graduated from San Francisco State University in 1966 with a B.A. in physics and was employed by Ampex Corporation from 1967 to 1969 in the professional audio and research departments. There, he worked primarily in low noise research and servo design.

Since 1971 he has been an independent consultant with a number of firms, including: Mark Levinson Audio Systems, Gale Electronics and Design (London), I.H.E.M. (Switzerland), and Alembic Inc. (San Francisco).

He is a member of the Audio Engineering Society and the Institute of Electrical and Electronics Engineers.