

# Nonlinear Distortion Measurement Using Composite Pulse Waveform\*

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A new method for measuring nonlinear distortions in circuits such as audio amplifiers is presented. The method uses composite pulse waveforms as the test signal. The great advantage of this method is that it permits predicting the form of nonlinearity of a circuit from the figure of the nonlinear distortion given by the measurement.

## 0 INTRODUCTION

In the past a sine wave test signal has been used for testing the linearity of a circuit. Such an approach, generally known as the "total harmonic distortion measurement," is unsatisfactory to test linearity in circuits such as high-quality audio amplifiers in which a high degree of linearity is desired. Furthermore, it has been recognized that the total harmonic distortion method generally does not give good correlation with subjective assessment of sound quality. The gap between hearing tests and the total harmonic distortion measurement is explained first by the difference of signals used, namely, music and a pure tone.

To give an improved subjective agreement, several methods of measuring nonlinear distortions have been proposed [1]–[7]. The common drawback of these methods is their inability to provide a direct indication of the form of nonlinearity, which is the fundamental property

of characterizing the tone of audio amplifiers. The new method uses composite pulse waveforms as the test signals which permit the form of nonlinearities of amplifiers predicting directly from nonlinear distortion figures given by the measurement.

## 1 PRINCIPLE OF THE METHOD

The theoretical study on the nonlinear distortion of audio instruments made by one of the authors shows that the nonlinearity of an amplifier gives rise to a dc component coupled with an increase of a certain low-frequency component of a composite pulse waveform. On the other hand the nonlinearity of a loudspeaker does not give rise to such a dc compound, but changes a middle- or high-frequency component. It was also suggested that we can distinguish the tone of an amplifier from that of a loudspeaker in hearing tests. The principle of the new method for measuring nonlinear distortions in amplifiers is based on the theory mentioned above [8].

The waveforms of four types of composite pulse test

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signals are shown in Fig. 1. Test signal waveform 1, for example, can be obtained by switching two input voltages  $V_1$  and  $-V_2$  using a multiplexer. The block diagram and front panel of the measuring instrument is shown in Fig. 6. The repetition frequency  $1/T$  of the test signals is 220 Hz, which was chosen by experiments. A crystal oscillator is used to ensure that the lengths of time periods  $T_1$ ,  $T_2$ , and  $T$  are maintained at fixed values.

It is important to note that the area of the part of the waveform above the zero-voltage axis equals the area that is below the zero axis, and also that  $V_1$  is not equal to  $V_2$ , namely,

$$(V_1 + V_2)T_1 = V_2T_2 \quad \text{and} \quad V_1 \neq V_2 \text{ (or } T_1 \neq T_2\text{).} \quad (1)$$

This requirement is practically achieved by attenuating either  $V_1$  or  $V_2$  so that the needle indicates the proper value on the meter (theoretically 0% at dc or 0.2% at 220 Hz).

The frequency spectrum of test signal waveform 1 is

$$S_1(f) = \frac{(V_1 + V_2)\sin(\pi f T_1)}{\pi f} - \frac{V_2\sin(\pi f T_2)}{\pi f}. \quad (2)$$

For  $fT_1, fT_2 \ll 1$ , the  $S_1(f)$  can be approximated by using Eq. (1) as follows:

$$S_1(f) \approx \left( \frac{V_2 T_2}{6} \right) (\pi f)^2 (T_2^2 - T_1^2). \quad (3)$$

The frequency spectrum of the reference signal wave-

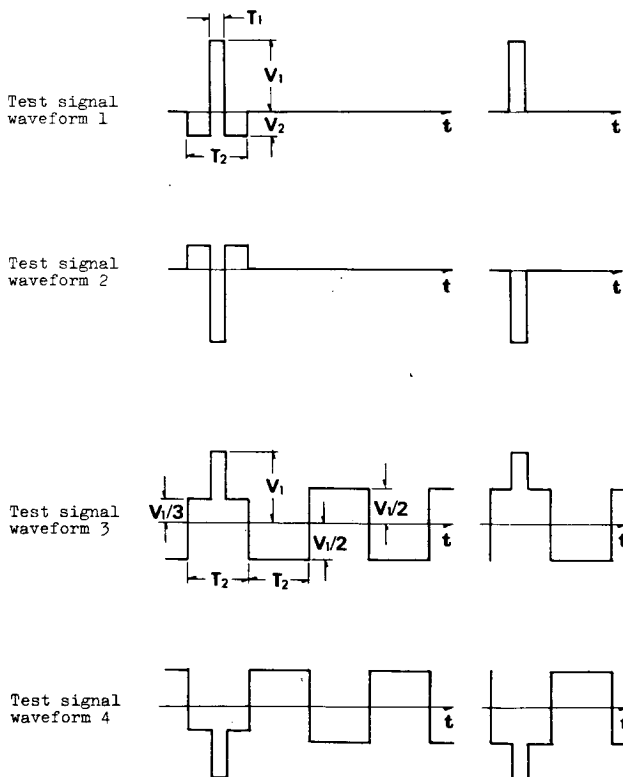


Fig. 1. Four types of test signal waveforms (left) and the reference signals (right).

form of test signal 1 is given by

$$S_0(f) = \frac{V_1 \sin(\pi f T_1)}{\pi f}. \quad (4)$$

For  $fT_1 \ll 1$ , the  $S_0(f)$  is approximated as follows:

$$S_0(f) \approx V_1 T_1. \quad (5)$$

Normalizing to make  $S_0(f) = 1$ , results in

$$\bar{S}_1(f) = \frac{S_1(f)}{S_0(f)} \approx \frac{1}{6} (\pi f T_1)^2 \times \left( 1 + \frac{V_2}{V_1} \right) \left[ \left( \frac{T_2}{T_1} \right)^2 - 1 \right]. \quad (6)$$

For the purpose of testing the linearity of audio amplifiers, the frequency  $f$ , the length of time period  $T_1$ , and the ratio of the amplitude of the positive and negative pulses  $V_1/V_2$  are, preferably,

$$\begin{aligned} f &= 220 \text{ Hz} \\ T_1 &= 35.5 \mu\text{s} \quad (T_2 = 4T_1) \\ V_1/V_2 &= 3. \end{aligned} \quad (7)$$

In this case the normalized frequency spectrum of test signal waveform 1 is

$$\bar{S}_1(f) = 0.0020 \dots \approx 0.2\%. \quad (8)$$

It will be understood that this value (0.2%) corresponds to a theoretical level indicative of the absence of non-linear distortions.

A plot of the frequency spectrum  $S_1(f)$  of test signal waveform 1 is shown in Fig. 2. The repetition period  $T$  is selected to equal 4.545 ms. Each frequency component is shown as a thick solid line, having a separation of  $1/T$ . Fig. 2 also shows the frequency spectrum  $S_0(f)$  of the reference signal waveform. When an amplifier under test alters the applied test signal waveform, the dc balance condition no longer holds, and the altered waveform shows a marked increase in the level of its low-frequency components, as indicated by dashed lines in Fig. 2.

Fig. 3 illustrates the relation of an input-output non-linearity of a circuit and test signal waveform 1, where  $\Delta$  indicates a deviation from a linear and  $\Delta(t)$  the distur-

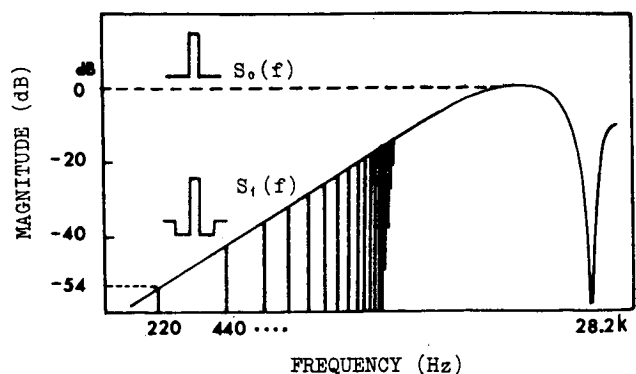


Fig. 2. Frequency spectrum of  $S_0(f)$  and  $S_1(f)$ .

tion waveform. The frequency spectrum of the distortion waveform normalized by  $S_0(f)$  is

$$\bar{D}(f) = \Delta/V_1. \tag{9}$$

Thus the normalized frequency spectrum of the altered test signal waveform is

$$\bar{S}_1'(f) = \bar{S}_1(f) + \bar{D}(f). \tag{10}$$

At the frequency of 220 Hz, the  $\bar{S}_1'(f)$  is approximated as follows:

$$\bar{S}_1'(f) \approx 0.002 + \Delta/V_1 \approx 0.002 + \Delta/V_1', \tag{11}$$

$(\Delta \ll V_1').$

Since a measurable quantity is the absolute value of the  $\bar{S}_1'(f)$ , we define the nonlinear distortion ND as follows:

$$ND = |\Delta/V_1' + 0.002| - 0.002. \tag{12}$$

Thus for  $\Delta/V_1' > -0.002$ , we have

$$ND = \Delta/V_1'. \tag{13}$$

For  $\Delta/V_1' < -0.002$ , we also have

$$ND' = -\Delta/V_1' - 0.004. \tag{14}$$

As shown in Eq. (13) or Eq. (14), the relation of the measured quantity ND and the deviation  $\Delta$  is very simple.

Fig. 4 illustrates typical forms of static nonlinearities and corresponding ND figures.

The above discussion covers the cases of test signal waveforms 2, 3, and 4. In the case of test signal waveform 3 (or 4), the rectangular waveform is superimposed on test signal waveform 1 (or 2). Fig. 5 illustrates the relation of an input-output nonlinearity of a circuit and test signal waveform 3. The repetition period of the rectangular waveform is selected to equal  $2T_2$ , so that the frequency component of 220 Hz of signal waveform 1 or 2 whose level is to be measured is isolated from the frequency components of the rectangular waveform.

The power of test signal waveforms 3 and 4 is increased by a factor of approximately 24 as compared with the power of test signal waveforms 1 and 2. This

may disclose distortion due to a weak power source of the amplifier under test.

2 MEASUREMENTS

The block diagram of the nonlinear distortion meter is shown in Fig. 6(a). The reference signal waveform is generated by the signal generator for calibration. During this time, the amplifier under test is disconnected from the apparatus with the output of attenuator 1 being connected directly to the input of attenuator 2. Attenuator 2 is adjusted until a reading of unity (100%) is indicated by the linearity indicating meter. Then the mode of the test signal generator is switched to produce the test signal waveform, and the generator is adjusted by manipulating the potentiometer of the dc voltage

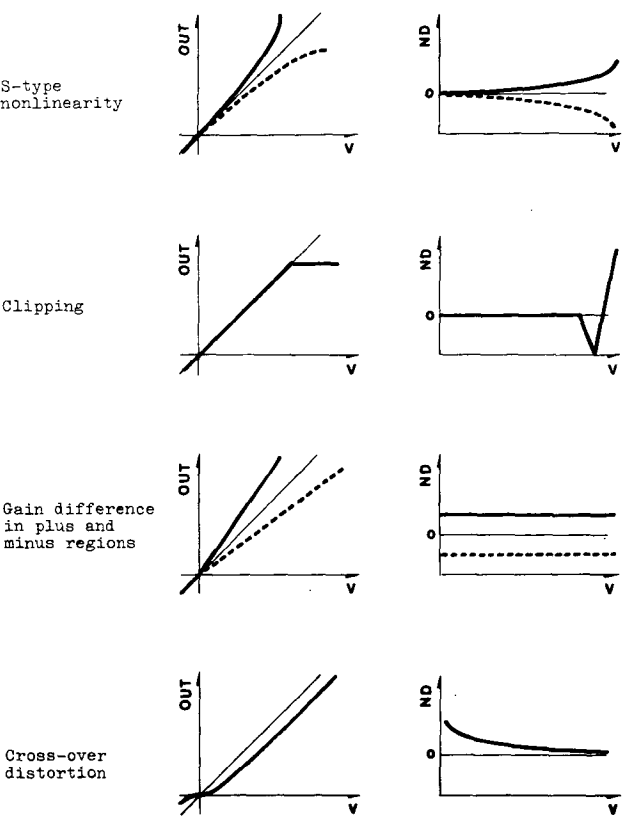


Fig. 4. Typical forms of static nonlinearities (left) and the ND figures (right).

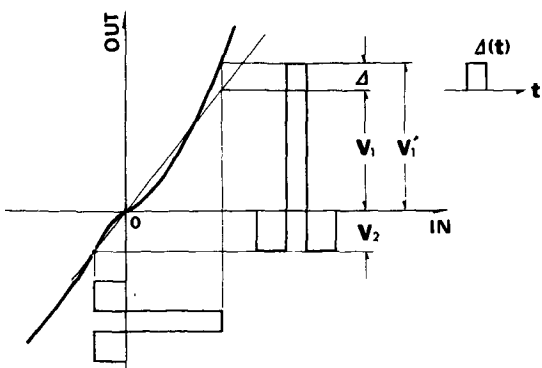


Fig. 3. Input-output nonlinearity and test signal waveform 1.

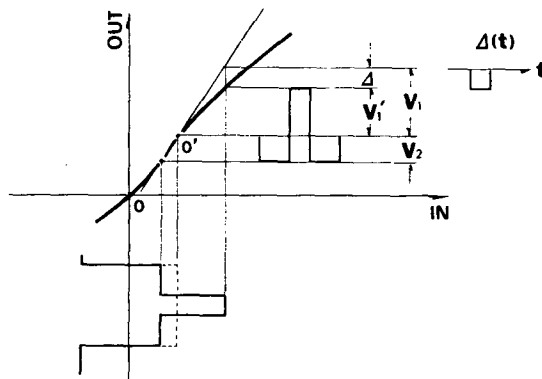


Fig. 5. Input-output nonlinearity and test signal waveform 3.

source such that the meter has a reading of 0.002 (0.2%), which is equal to the theoretical value as described above. The next step is to connect an audio amplifier with an 8- $\Omega$  resistance load. As an example, between attenuators 1 and 2, apply the reference signal waveform to the amplifier under test, and adjust attenuator 2 so that the meter has a reading of unity (100%). With the amplifier under test connected, the reading of the meter is taken by applying the test signal waveform to the amplifier under test. The difference between the reading of the meter and the theoretical value gives a measure of the linearity of the amplifier under test. In practice this difference is indicated directly on the meter.

Tests with the amplifier connected to the apparatus proceed with varying amplitudes of the reference signal waveform and test signal waveform by adjusting attenuator 1. By studying changes in nonlinear distortion figures for varying amplitudes of the reference and test signal waveforms, it is possible to determine the form of nonlinearity, that is, whether it is S-type nonlinearity, crossover distortion, or clipping.

Measurements of more than 40 amplifiers were made using the new method with conventional total harmonic distortion measurements. It was found that high-quality amplifiers with a total harmonic distortion of 0.01–0.001% show a nonlinear distortion of more than 0.04%. Figs. 7–9 show examples of the measured nonlinear distortion figures of audio amplifiers. The abscissa is given by the reference output power which is defined by  $V_1^2/16$ , where  $V_1$  is a peak value of the test signal voltage across the 8- $\Omega$  resistance load of an amplifier under test. The real power of test signal waveforms 1 and 2 is about

1/48 of the reference power, and the power of test signal waveforms 3 and 4 is about a half of the reference power  $V_1^2/16$ , which is identical to the power being generated by applying a sine wave voltage with an amplitude of  $V_1$  to an 8- $\Omega$  resistance.

Figs. 7–9 also show the forms of nonlinearity predicted from the nonlinear distortion figures measured by using test signal waveforms 1 and 2. Subjective impressions of these amplifiers are as follows. Amplifier A: soft, glossy; amplifier B: live, strong; amplifier C: definite, clear. These subjective impressions were given by three hearing testers by common consent.

Fig. 10 shows the nonlinear distortion figure of a high-quality audio amplifier where the measurement was made by the new method using sound reproduced from a loudspeaker which was connected at the output of the amplifier under test. The measuring arrangement is illustrated in Fig. 11. The microphone was set at the face of a woofer so that the received signal sound waveform was smooth enough so that the effect of the nonlinearities of both the microphone and the measuring amplifier on the data could be neglected. Fig. 10 also shows the nonlinear distortion figure given by measuring the voltage across the voice coil of the loudspeaker.

### 3 CONCLUSIONS

This paper has discussed a new method of measuring nonlinear distortions in circuits such as amplifiers. This method can also be applied to the measurement of nonlinear distortions introduced in a signal transmission system. The great advantage of this method is that it

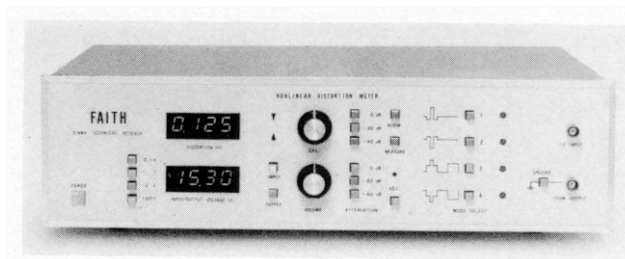
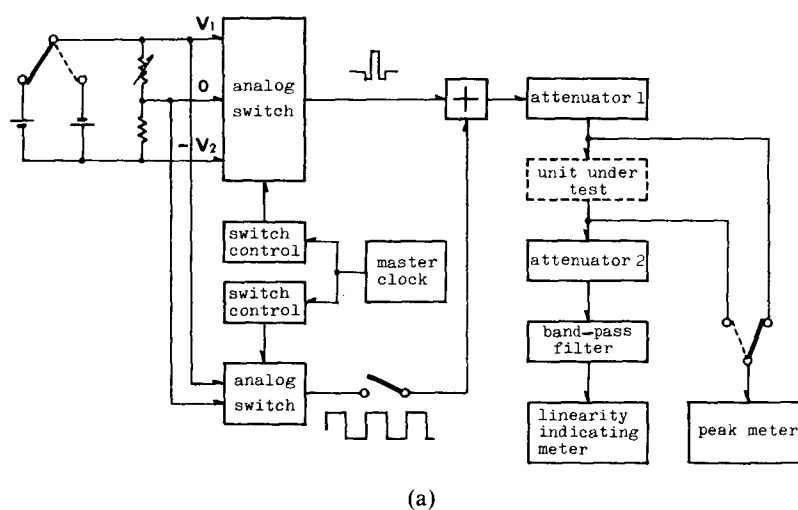


Fig. 6. Block diagram (a) and front panel (b) of nonlinear distortion meter.

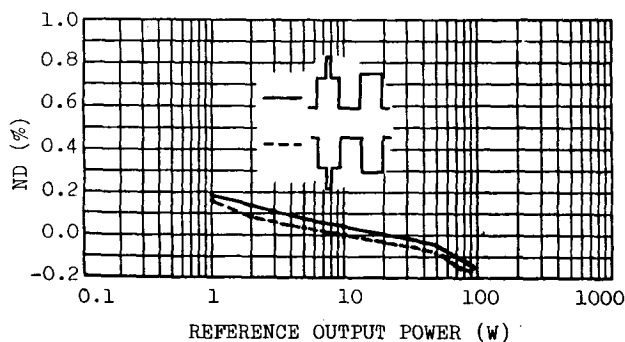
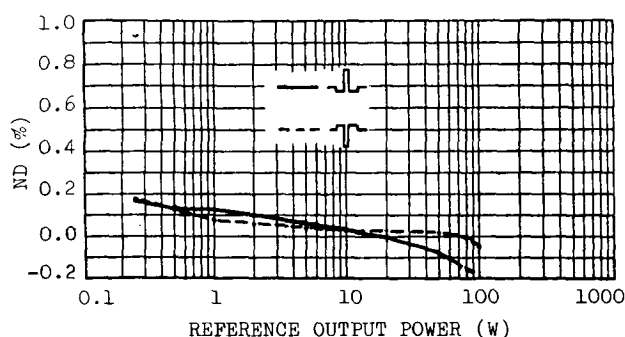
enables the form of nonlinearity of a circuit to be predicted from the figure of the nonlinear distortion. It is to be noted that the nonlinear distortion in an amplifier can be measured by this method using sound reproduced from a loudspeaker which is connected at the output of the amplifier under test, just as we do in hearing tests of amplifiers.

#### 4 ACKNOWLEDGMENT

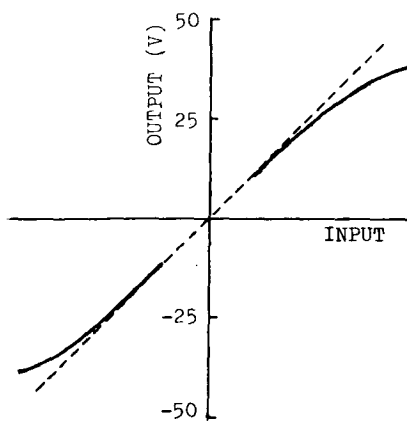
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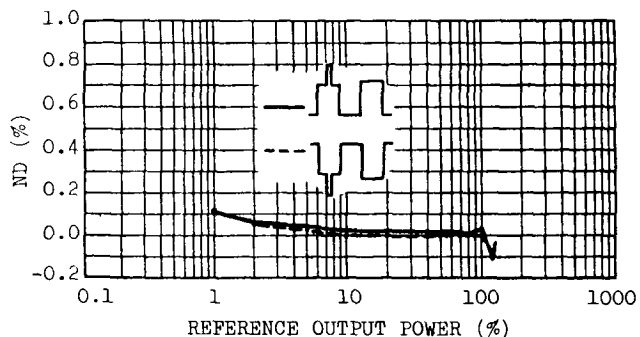
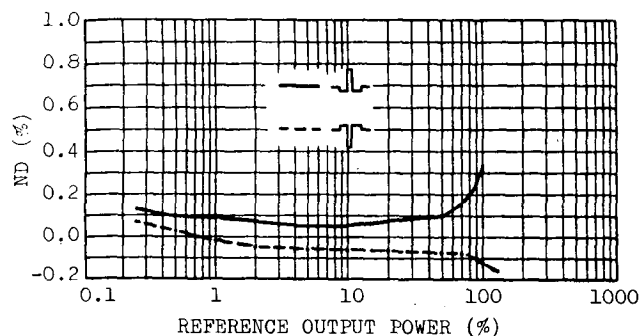


(a)

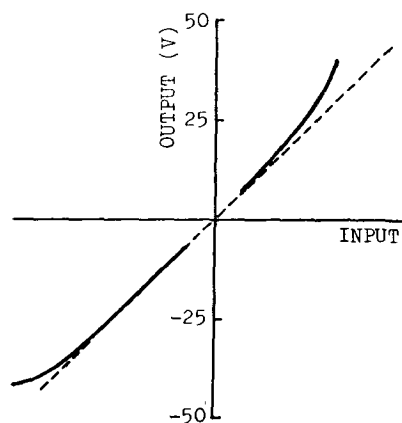


(b)

Fig. 7. Amplifier A (B750 MKI). (a) Nonlinear distortion figures. (b) Form of nonlinearity (solid line) where distortion is magnified.



(a)



(b)

Fig. 8. Amplifier B (CA-S1). (a) Nonlinear distortion figures. (b) Form of nonlinearity (solid line).

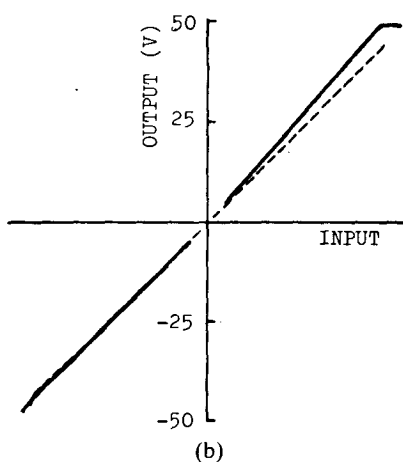
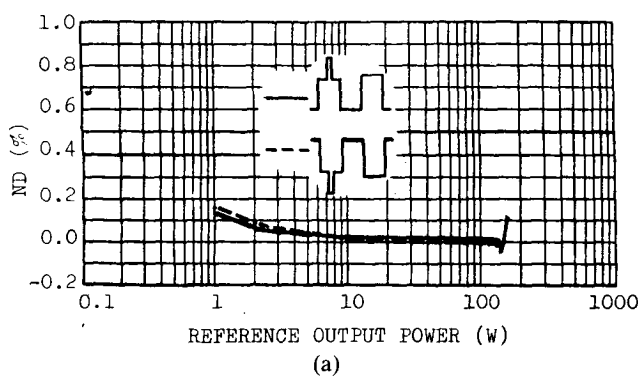
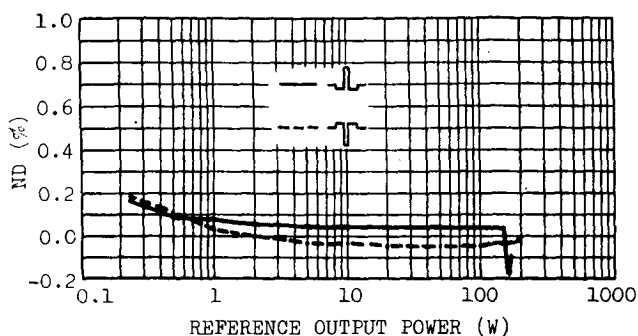


Fig. 9. Amplifier C (KA-9900). (a) Nonlinear distortion figures. (b) Form of nonlinearity (solid line).

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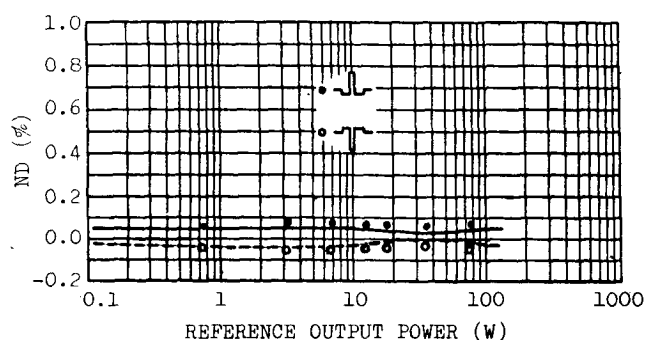


Fig. 10. Nonlinear distortion of an amplifier (L-07M11). •, ○—through a loudspeaker; —, — — — electric measurements.

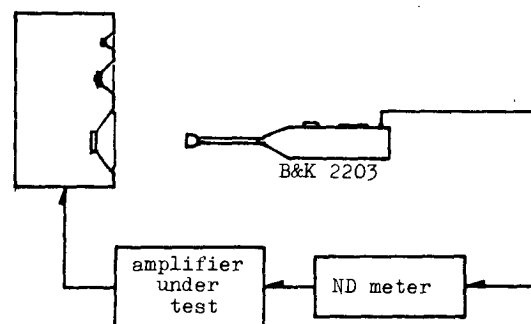


Fig. 11. Measuring arrangement of the nonlinear distortion in an amplifier using sound emitted from a loudspeaker.

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