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ENGLAND

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Distortion

MOST TRANSMISSION SYSTEMS are concerned with relaying speech or music from one location to another, and inevitably the final output soundwave is not a faithful reproduction of the input. Absolute fidelity is an ideal which can only be approached but never realized. This difference between input and output signal is the sum of the various forms of distortion introduced in all stages of the transmission. If excessive it detracts from the intelligence being conveyed or, in the entertainment field, the pleasure.

Lack of fidelity is due to many causes which may be broadly classified as noise, bandwidth limitations and frequency response, mains frequency hum, man-made interference, production of unwanted signals due to non-linearity and crosstalk, and finally the acoustical properties at the receiving and sending ends together with the general background noise. Reduction of these various sources to an acceptable level will depend upon the application and economics of the system, but unfortunately they are not all completely under the control of the designer. For example, interference from electrical equipment such as diathermy or r.f. heating apparatus, and also from vehicle ignition systems, is dependent upon the proximity to such sources, and legislation has been found necessary to contain it within reasonable bounds. On the other hand, distortion in the form of bandwidth limitations is purposely introduced on speech circuits for economic considerations in order to obtain the greatest number of channels and not squander the available frequency spectrum. Although it is generally considered that normal conversational speech occupies a bandwidth from 62 c/s to 8 kc/s, very little intelligence is lost when using the restricted telephone bandwidth of 300 c/s to 3400 c/s as the maximum voice energy for both men and women is concentrated within these frequency limits.

In the entertainment field it ceases to be a matter of just conveying information, and fidelity, especially for the transmission of music, is zealously pursued. Here the complete audio frequency range is transmitted, but except among the enthusiasts it is seldom that the receiver justifies the quality of transmission. The weakest link of all is the electro-acoustical transducer, which in addition to the normal forms of distortion is also capable of producing sub-harmonics. Latest advances in reliability and size of components, and the introduction of semiconductors to produce portable receivers that can be powered from small batteries, has brought no improvement in quality, but rather a deterioration as the loudspeaker is reduced in size to match the other components. It may be convenient to have in the pocket minute-by-minute access to the latest test match score, but the design of a reliable music stop filter is long overdue.

It is fortunate that one backward step in the art of sound reproduction—although painfully apparent—does not halt the forward march. Recently extensive tests have been conducted in the field of stereophonic reception. Basically this is achieved by having at the transmitting end two microphones paired with two loudspeakers at the receiving end. This can be easily accomplished with two transmitters as is done in the fortnightly broadcasts on alternate Saturday mornings by the B.B.C. The Third Network is used as one channel, and the Television sound as the other, thus enabling the public to savour stereophony on their existing receivers. For obvious reasons the use of two transmitters is undesirable, and trials have also taken place on systems

Museum telephone exchange radio tower in the course of construction in the Tottenham Court Road area of London. To minimize the possibility of intermodulation distortion between channels, due to imperfections in the waveguide runs, the radio receivers will be housed immediately beneath the aerial galleries

Photograph by courtesy of H.M. Postmaster General.

using only one transmitter with multiplexing. Last summer the B.B.C. conducted the second series of field trials on the GE-Zenith pilot tone stereophonic system using the Wrotham transmitter on 91·3 Mc/s, and in response to requests from manufacturers are continuing test transmissions weekly. This system is the one adopted from the eight submitted to the Federal Communications Commission. It is now in general use in the U.S.A. and has created a demand for special test gear. The multiple waveform to modulate the transmitter has a frequency spectrum, ignoring the storecasting requirements, which extends from 50 c/s to 53 kc/s. It has been found that the F.M./A.M. Signal Generator, type TF 995A/2, can be modified to accept this frequency range, and hence be a useful generator for this system. The modifications required are explained on page 95 of this issue in an article by the U.S.A. Office Assistant Sales Manager, who has had practical experience of these special requirements, which will also be applicable in this country if and when such a system is approved by the European Broadcasting Union.

Although Marconi Instruments have for many years marketed instruments for analysing waveforms and determining distortion using various techniques, very little information has appeared in this journal about the audio frequency instruments. The two instruments at present in our catalogue are the Wave Analyser type TF 455E and the Distortion Factor Meter type TF 142F which were first introduced in Volume 5, Numbers 5 and 3 (1955–56). Both these instruments have now been redesigned using solid state techniques and are described in this issue in two articles; one of which considers the method of measuring individual harmonics and the other total spurious signal content.

The redesign is not just a change from valve to solid state, although this would be welcome, from a size and weight consideration, but the measurement and frequency ranges have also been considerably increased, as well as additional features added to simplify measurements and to increase the possible applications. One such novel application is given on page 90, where the new feature on the Wave Analyser of a synchronously tuned b.f.o. output allows measurements to be made at levels which would normally be considered unpractical due to the noise content of the signal.

These instruments, as their respective photographs show, are part of the new Marconi Instruments 2000 series. Together with the new oscillators, attenuators and signal sources these will prove a powerful tool for those engaged in any form of audio frequency measurements.

Here the exceptionally low distortion version of the A.F. Oscillator, designated type TF 2100/1M1, will be particularly useful as the driving source for measurements on high fidelity systems. With a distortion content of less than 0·01% it will be possible to feed the oscillator directly into the apparatus under test knowing that the resultant distortion produced is a product of that apparatus alone. This eliminates the need to use low pass filters to ensure a pure tone.

P. M. R.

621.317.353

MARCONI INSTRUMENTS

Distortion Measurement in Audio Amplifiers

by D. E. O'N. WADDINGTON A.M.Brit.I.R.E.

The three main types of distortion which may occur in audio amplifiers—non-linear distortion, frequency distortion and phase distortion—are described together with their causes and effects. A discussion on wave analysers indicates the suitability of the heterodyne type for audio measurements. The principle of operation of the TF 2330 Wave Analyser which includes a synchronously tuned beat frequency oscillator is described. Methods of measurement using the TF 2330 are suggested for all three types of distortion.

DISTORTION occurs, to a certain extent, in all audio frequency amplifiers. The quality of reproduction depends very largely upon the degree of distortion and thus, to assess this, the distortion has to be measured. Several kinds of distortion are likely to occur but they may all be grouped under three main headings, non-linear distortion, frequency distortion and phase distortion.

Non-Linear Distortion

When a signal is amplified the transfer characteristic of the amplifier modifies the waveform. In the ideal amplifier the transfer characteristic would be a linear function and no distortion would occur, i.e.:

$$V_{out} = A\,V_{in}. \qquad (1)$$

In practice this state of affairs is very seldom obtained and the transfer function is non-linear, i.e.:

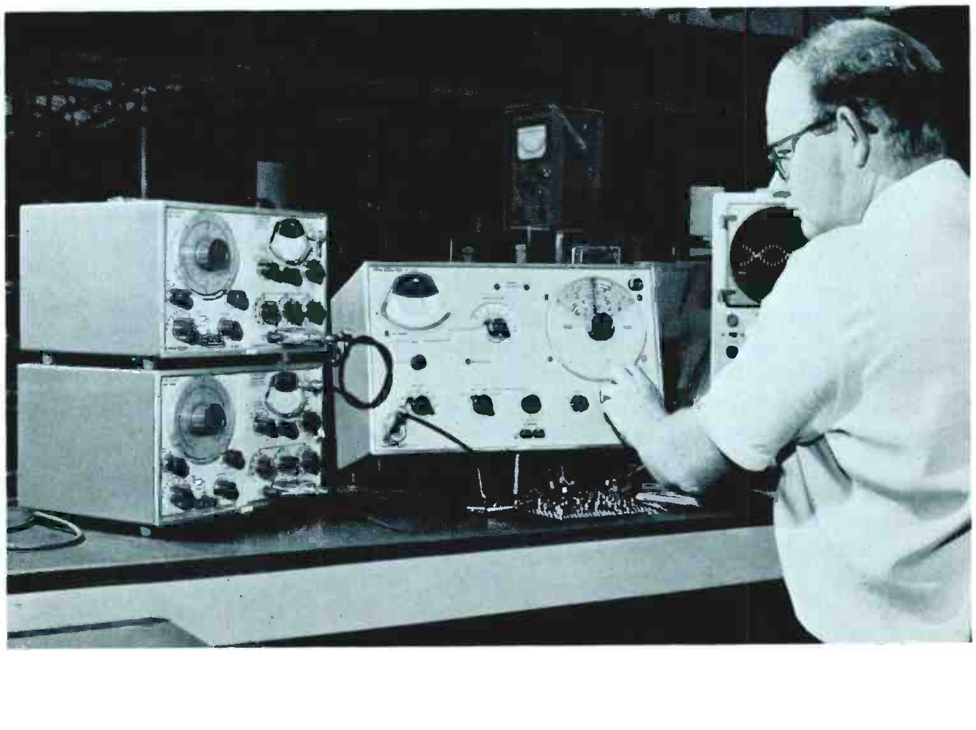
$$V_{out} = A_1 V_{in} + A_2 V_{in}^2 + A_3 V_{in}^3 \ldots A_n V_{in}^n. \qquad (2)$$

If a signal $V_{in} = P\sin pt$ is applied it will be seen that there will be components having frequencies of $2pt$, $3pt$ ... npt in the output as well as the fundamental component. These additional components constitute the harmonic distortion and by measuring them it is possible to determine how much distortion is introduced by the amplifier:

$$\%\ \text{distortion} = \sqrt{\frac{E_2^2 + E_3^2 + \ldots E_n^2}{E_1^2}} \times 100.$$

where E_1, E_2, E_3 ... E_n are the levels of the fundamental, 2nd harmonic, 3rd harmonic ... and nth harmonic.

These harmonics are not necessarily displeasing to the ear as it is well known that the harmonics present in musical instruments and in the human voice give them their quality and timbre. However, it is the relative amplitudes of these harmonics which determine the particular quality of the sound. If certain harmonics are either increased or decreased the quality of the reproduced sound will be changed. It is interesting to note that all



The author using a TF 2330 Wave Analyser to measure intermodulation distortion in an experimental amplifier. Two TF 2000 A.F. Signal Sources provide a two-tone test signal with very low interaction the harmonics up to and including the sixth 'harmonize' with the fundamental while the 7th, 9th, 11th, 13th, 14th, 15th, 17th, 18th and 19th are dissonant.[1] Higher orders may also be dissonant but they have been omitted from the list as 20th harmonic distortion and above is usually of negligible proportions. The sick sound of a violin bowed by an inexpert musician is due to it being played in such a way as to render the 7th harmonic audible.

The permissible harmonic distortion in an audio amplifier is very difficult to assess as it is a purely subjective effect and thus depends on the individual listener. A series of tests shows that the unpleasantness of distortion depends not only on the percentage of harmonic distortion but also on the distribution of power among the harmonics and the frequency range of the amplifier. Thus, in general, more distortion is permissible with triodes than with pentodes as the distortion in triodes is mainly 2nd harmonic, whereas in pentodes it consists mainly of higher order components. With transistors, from measurements of the distribution of distortion components it is reasonable to suppose that they will be similar in effect to pentodes. Generally speaking, distortion of the order of 0·8% in a 15 kc/s bandwidth will be just perceptible, 1·5% tolerable and 2·5% objectionable. If the bandwidth is reduced to 3·75 kc/s, 1·2% is perceptible, 6% tolerable and 12% objectionable.[2] These figures may all be increased slightly if the amplifier is only intended for speech reproduction.

The effects of non-linear distortion are not so readily apparent to the listener as harmonic distortion but rather as intermodulation distortion. This occurs when two signals are applied simultaneously to an amplifier. The signals intermodulate producing sidebands which are not harmonically related to either of the two signals and which therefore clash with them. If the frequencies of the two signals are p and q the frequencies of the intermodulation products will be:

$(p-q)$, $(p+q)$, $(2p-q)$, $(2p+q)$, $(p-2q)$, $(p+2q)$, $2(p-q)$, $2(p+q)$, etc.

Two types of music where intermodulation is very easily recognizable are organ music, where the low frequency notes tend to modulate the high frequency ones and coloratura soprano singing with flute accompaniment where the difference between the two notes shows up as a low frequency drone.

As the non-linearity of the transfer characteristic is not necessarily constant at all frequencies, the intermodulation distortion is most unlikely to be directly proportional to the harmonic distortion although it is reasonably safe to assume that if the harmonic distortion is low, the intermodulation distortion will also be low and vice versa. Insufficient tests have been done to state categorically what levels of intermodulation distortion are tolerable, perceptible or objectionable. The difficulty lies in standardizing the test frequencies and amplitudes of test signals. The S.M.P.T.E. method[3] has been used for some years in America and results using this system show that a high fidelity amplifier will have intermodulation distortion of 2–4% while the distortion in a typical radio receiver may be as high as 40%.

The causes of non-linearity distortion are many and the following are a few of the most common:

1. Inherent curvature of the characteristics of the valves or transistors.
2. Non-linearity in coupling transformers.
3. Mismatch between halves of push-pull stages. (In class A and AB stages this will cause excessive even harmonic distortion.)
4. Cross-over distortion in class B push-pull stages due to incorrect biasing. (This may well show up as an increase of percentage distortion as the amplitude of the test signal is decreased.)

Frequency Distortion

This may be defined as variation of amplification with applied frequency. As such it is almost inevitable but it is reasonably easy to control. The actual frequency range of an amplifier will depend very largely upon its intended application as it is quite usual to 'improve' the performance of an audio system by limiting the frequency range. This is not quite as paradoxical as it might appear. At the low frequency end of the band, hum may cause objectionable interference and for it to be reduced certainly improves the apparent performance. At high frequencies, noise such as record scratch and atmospherics may mar reproduction. Also with cheap loudspeakers the distortion increases rapidly with frequency. Thus reduction of the high frequency response may also result in improvement. Typical frequency responses for amplifiers are:

25 c/s–20 kc/s	High fidelity
100 c/s–5 kc/s	Medium quality domestic console
150 c/s–3·3 kc/s	Cheap domestic receiver.

Apart from these limitations it is usual to vary the frequency response of an amplifier:

(a) To compensate for recording or transmission characteristics.

(b) To regulate the tonal balance to suit the requirements of the listener and the acoustics of the room in which the equipment is operating.

Variations in frequency response frequently occur in amplifiers when they are operated at various levels. Thus the frequency response ought to be checked not only at maximum power output but also at levels of, say, 30 and 60 dB below maximum output.

Phase Distortion

This is the change, by the amplifier, of the phase angle between the fundamental and any one of its harmonics or between any two components of a complex wave. It usually occurs when reactive components are inserted into the circuit to limit or modify the frequency response. Although the ear cannot distinguish phase in monophonic systems, the effect of this form of distortion is to change the timbre of music, making it sound harsh. It also affects the transient response adversely so that 'attack' is lost. With the advent of stereophonic reproduction, phase distortion is assuming more importance.

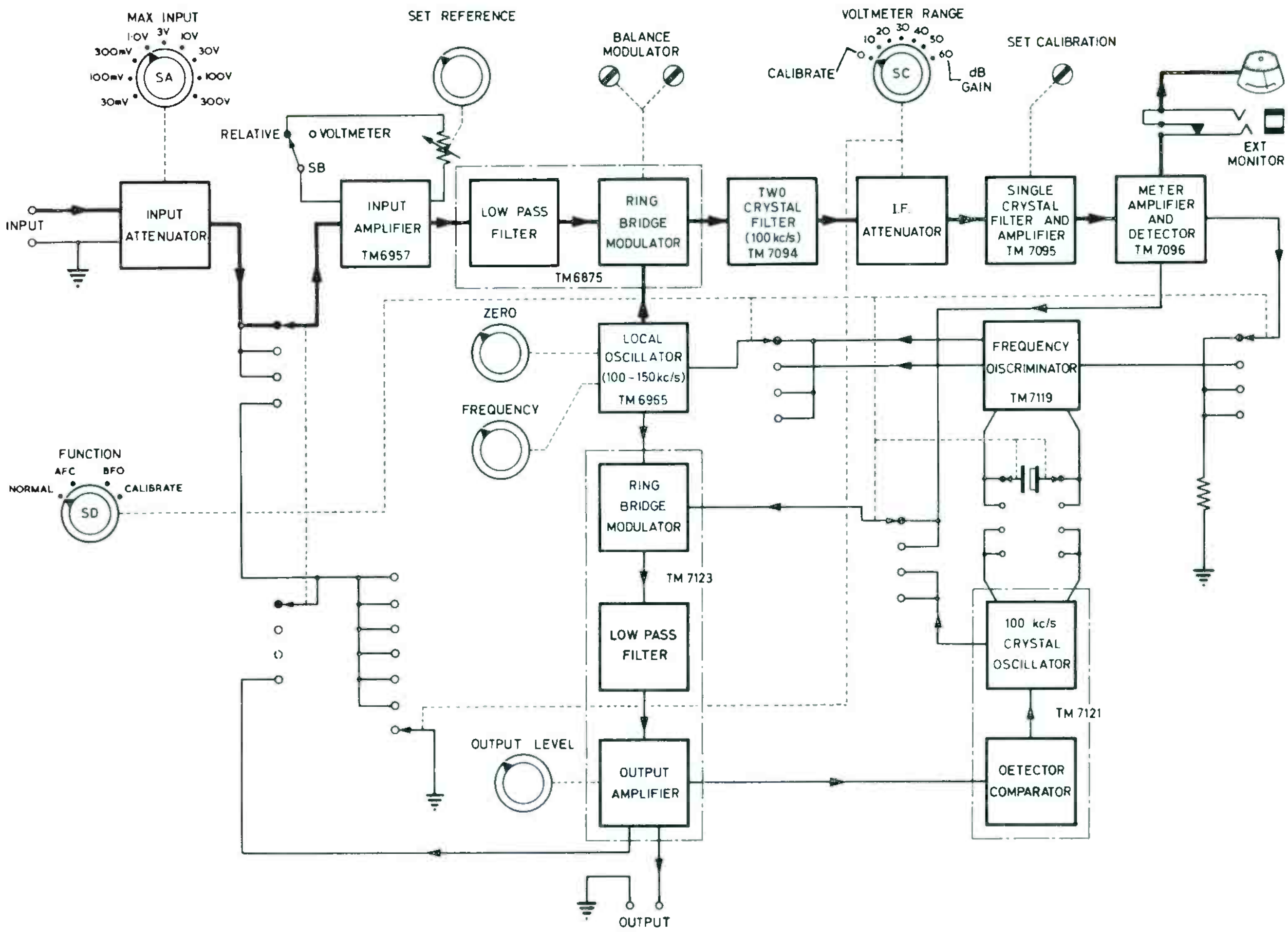


Fig. 1. Block diagram of Wave Analyser type TF 2330

Provided that the phase shift in both amplifier channels is the same, the remarks already made apply. However, if there is any difference between the phase shift in the two channels, the effect is to cause displacement of the apparent source. To date there are no criteria laid down for phase distortion and accepted practice is to keep it to a minimum.

Distortion Measurement using a Wave Analyser

Before describing the use of a wave analyser, a brief discussion of wave analysers in general and the new Marconi Instruments version, TF 2330, in particular will help to show how they may best be used. Essentially a wave analyser is a tunable voltmeter and will be of one of two types. The first is directly comparable to the t.r.f. radio receiver and consists of a tunable amplifier followed by an electronic voltmeter. Usually the tuned amplifier is of the resistance/capacitance type. This has the advantage that negative feedback may be employed to control the bandwidth and to ensure that the gain in the pass band is constant over the whole frequency range. With this type of circuit the bandwidth remains a constant proportion of frequency so that as the frequency is increased, the bandwidth becomes greater. The second type is comparable with the superheterodyne receiver having a constant bandwidth and it is into this category that the TF 2330 falls. The description of its operation may best be understood by referring to the block diagram, Fig. 1. The signal to be analysed is fed via an attenuator to the input amplifier which serves the dual purpose of providing a high input impedance and a certain amount of amplification while introducing a minimum of distortion. The gain of this amplifier is normally fixed but it may be varied for level setting purposes. Between this amplifier and the modulator is a low pass filter which restricts the flat bandwidth to 50 kc/s, thus preventing any 100 kc/s signal from riding straight through to the i.f. amplifier and also reduces the effects of second channel interference.

The modulator uses a conventional double balanced ring bridge which has been designed primarily to have a high degree of linearity (the distortion level under normal operating conditions is of the order of 0·001%) but it also has good carrier rejection. The carrier input to the modulator is derived from the local oscillator which uses permeability tuning. This gives a semi-logarithmic frequency scale shape with a scale length of 100 inches in $8\frac{1}{2}$ turns of the main tuning knob. A slow motion drive facilitates tuning in to the narrow bandwidth of the i.f. amplifier. The difference frequency output from the modulator is selected by the first crystal filter which consists of two 100 kc/s quartz crystals operated in their series mode and shunt capacitance coupled to produce an over-coupled double humped response. Up to this point in the instrument the operating level is set by the input attenuator and the gain control so that, provided that full-scale deflection on the meter is not exceeded when the instrument is tuned to the fundamental, the introduced distortion will be less than 0·01%.

The first crystal filter is followed by a step attenuator having six 10 dB steps which acts as the meter range switch giving ranges down to −60 dB or 0·1% full scale readings. This attenuator is followed by a single crystal filter which is tuned so that, with the double crystal filter, a frequency response such as is shown in Fig. 2 is obtained. The output from this filter feeds a high gain amplifier which in turn supplies the meter amplifier and rectifier circuit with its input. As frequency drift is a possible fault in signal sources which may be used in conjunction with the analyser, an automatic frequency control system has been included. To operate this, a signal at the intermediate frequency is taken from the meter amplifier, amplitude limited, and fed to a narrow band crystal discriminator whose centre frequency coincides with that of the i.f. amplifier pass band. An error signal derived from this discriminator is taken via a suitable low pass filter to a capacitor diode which applies suitable correction to the frequency of the local oscillator.

Under normal operating conditions an output from the meter amplifier is fed to a secondary modulator where it is mixed with a signal from the local oscillator. The difference frequency is selected by a low pass filter, amplified and fed via a level control to front panel terminals to provide a 'Restored' signal output. This signal will have the identical frequency to the component being investigated. As such it is useful in identifying positively, with the aid of a frequency counter, what component is being investigated. This is of particular use at high frequencies.

A feature in addition to the actual tuned voltmeter function is the inclusion of a b.f.o. system. The output of a 100 kc/s crystal oscillator is mixed with the local oscillator signal in the secondary modulator. The difference frequency is selected by a low pass filter after which it is amplified and fed via a level control to the output terminals on the front panel. An additional output from the amplifier is rectified and compared with a direct

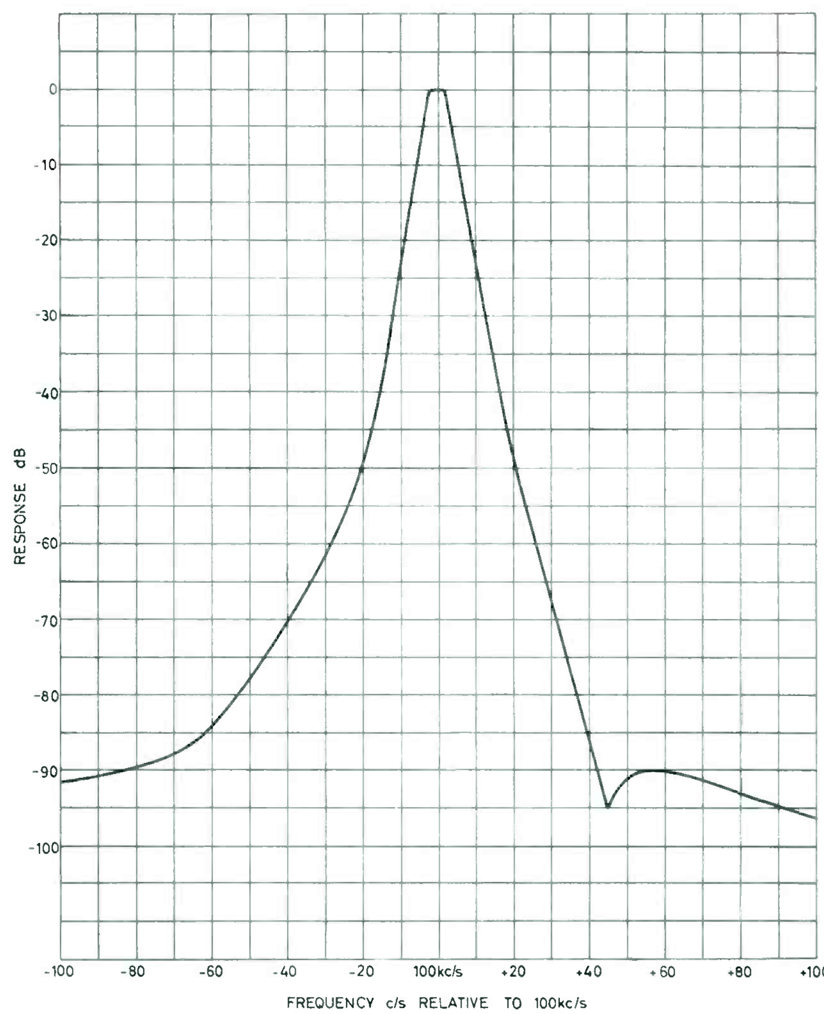
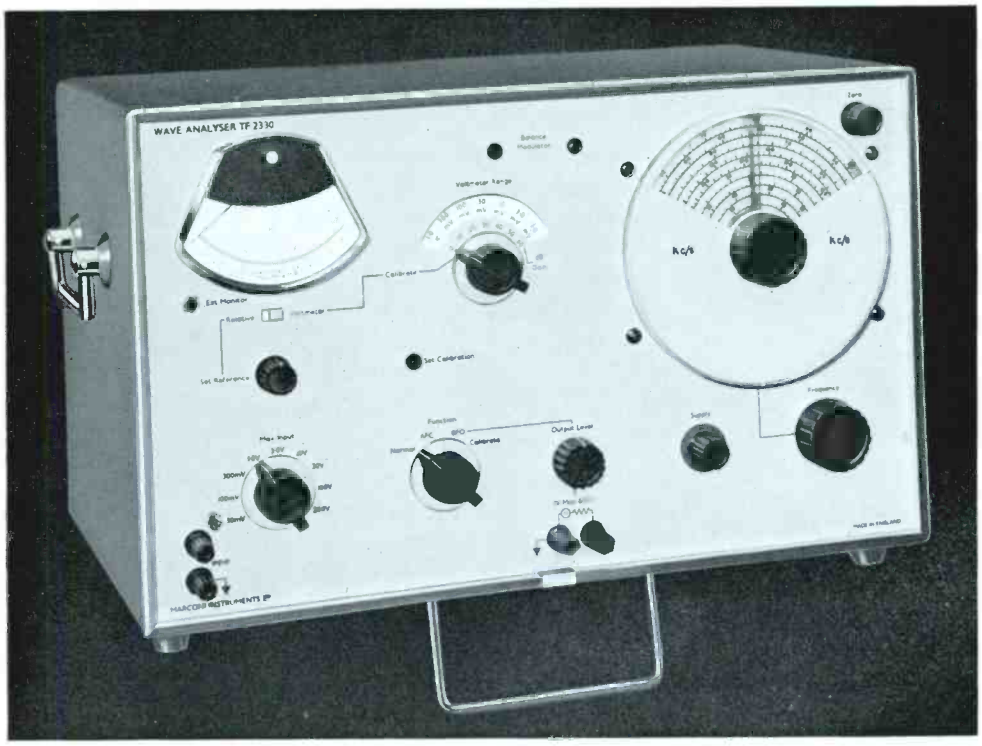


Fig. 2.
Excellent discrimination is shown by the response of the crystal filters. Automatic frequency control allows full advantage to be taken of the narrow pass band



Wave Analyser type TF 2330 in bench mounting form. A rack mounting version, TF 2330R, is also available

voltage derived from a low temperature coefficient zener diode. The resultant error signal is used to control the amplitude of the output from the crystal oscillator and hence the output from the b.f.o. system. This maintains the level to better than 1% over the frequency range. As the crystal oscillator frequency is tuned to the centre of the i.f. pass-band, the b.f.o. frequency will be the same as that to which the wave analyser is tuned and thus the instrument will constitute a synchronously tuned source and voltmeter.

Harmonic Analysis

The signal to be investigated is connected to the input terminals. The meter range switch is turned to give minimum sensitivity and, after tuning in the fundamental, the meter is set either to full-scale or 0 dB by adjusting the input attenuator and SET REFERENCE control. The instrument is then tuned to the 2nd harmonic of the signal, the sensitivity being increased *only* by using the meter range switch. The 3rd harmonic is measured in a similar fashion. For higher harmonics it is permissible to obtain an extra 10 dB of measuring range by use of the input attenuator as under these conditions the instrument introduces negligible high order distortion. Thus, used normally the instrument permits measurements down to 0·1% f.s.d. for 2nd and 3rd harmonics and 0·03% for higher order harmonics.

In checking distortion in an amplifier it is desirable to use a signal with as low distortion as possible for the test signal. The usual criterion is that the inherent percentage distortion in the test signal should be $\frac{1}{10}$ of the introduced distortion. Thus in order to measure distortion of the order of 0·1% it is necessary either to use a signal source with distortion of the order of 0·01% or to filter the signal before it is applied to the amplifier. If neither of these methods is available, another method is to measure the percentage of each harmonic component of the test signal before applying it to the amplifier and then to measure these components again at the output of the amplifier. The difference will then give a measure of the distortion introduced by the amplier. Unfortunately, this method is not very accurate, as true answers will only be obtained if the injected harmonic and the harmonic due to distortion in the amplifier are either in phase or in antiphase. Under any other phase conditions the results could be very misleading.

Intermodulation Distortion Measurement[4]

This overcomes the necessity for a pure signal for test purposes but only provided that the two test signals are added in such a way that there is no interaction in the combining network.

There are two accepted methods of measurement; the American or S.M.P.T.E. method and that recommended by the C.C.I.F.[5] These two methods each have their own fields of usefulness but there is no simple way of correlating the results obtained with them with each other or with the harmonic analysis method.

S.M.P.T.E.

In this method two signals, p and q, one having a low frequency (40–400 c/s) and the other a high frequency (greater than 1 kc/s) with amplitudes in the ratio of 4:1 are simultaneously applied to the input of the amplifier under test. The resultant spectrum, as shown in Fig. 3, is then analysed and the components a, a^1 (corresponding to 2nd harmonic distortion) and b, b^1 (corresponding to 3rd harmonic distortion), etc., are measured. The total

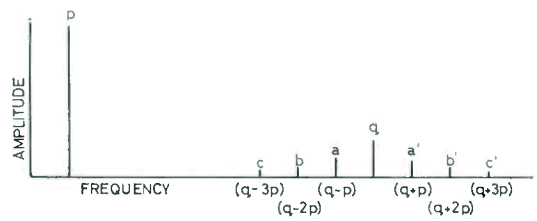


Fig. 3. Intermodulation spectrum-S.M.P.T.E. method

intermodulation distortion may then be calculated using the formula:

$$I.M.\ distortion = \frac{100\sqrt{(a+a^1)^2+(b+b^1)^2}}{q}\ \%.$$

The limitation of this method is that the wave analyser will be overloaded if the input attenuator is set so that the high frequency signal gives full-scale on the meter as the low frequency signal will be giving four times full-scale. This will introduce excessive distortion in the wave analyser. In order to overcome this the signal should be passed either through a high pass filter or a band stop filter tuned to the low frequency signal before applying it to the wave analyser input. If this is done, the high frequency signal may be set to full-scale with no fear of overloading and thus the full sensitivity of the system may be realized.

C.C.I.F.

In this method, two signals of equal amplitudes but with a relatively small frequency difference are applied simultaneously to the amplifier under test. The resultant

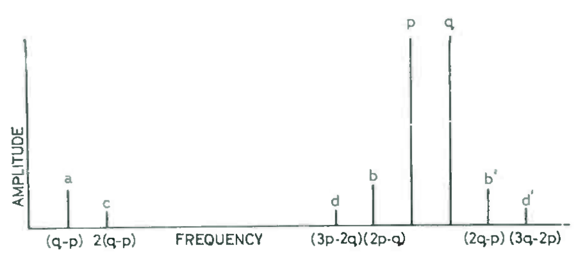


Fig. 4. Intermodulation spectrum-C.C.I.F. method

spectrum (see Fig. 4) is then analysed using the wave analyser, and the components a, b, c, d, etc., are measured. The only recommended formula for the calculation of percentage intermodulation distortion is:

$$I.M.\ distortion = \frac{100\ a}{p+q}\ \%.$$

This will give the same result as the use of an intermodulation test set. However, it only takes account of the component due to the second order curvature of the transfer characteristic. A more realistic formula could well be:

$$I.M.\ distortion = \frac{a+b+c+d\ ...}{p+q} \times 100\%.$$

This formula is based on the 'peak sum' method of distortion measurement.

Again the distortion introduced by the wave analyser could be a limiting factor. To obtain the best results it is advisable to set the signals p and q each to half-scale as by this means the distortion in the wave analyser is less than 0·01%.

General Considerations for Intermodulation Distortion Measurements

1. To obtain results as realistic as those obtained using simple harmonic analysis methods, ensure that the peak-to-peak signal applied to the amplifier is the same in all tests. This ensures that the portion of the transfer characteristic explored in each case is the same.
2. Use a hybrid network for combining the two signals. This is not necessary if Marconi Instruments type TF 2000 Signal Sources are used as there is negligible intermodulation when the outputs of two of these are connected in parallel.
3. Ensure that all the components to be measured are within the pass band of the amplifier under test.

Frequency Distortion

The TF 2330 is the ideal instrument for checking this as, when the function selector is set to B.F.O., the instrument operates as a synchronously tuned source and voltmeter. All that is necessary to do therefore, is to connect the B.F.O. output to the input of the amplifier, and the amplifier output to the wave analyser input terminals. The signal level is suitably adjusted and the instrument tuned through the frequency range while the relative response is noted. This should be repeated at various output levels.

Phase Distortion

This is not easily measured but its effects may be noted by using the set-up described for frequency distortion but also applying the input signal to the 'Y' input of an oscilloscope and the output to the 'X' input to produce

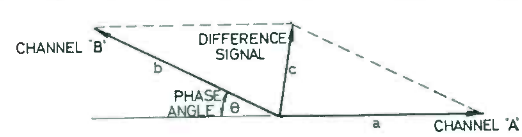


Fig. 5. Evaluation of stereophonic phase angle

$$a = b$$
$$c = \sqrt{2a^2\ 1-\cos\theta}$$

The differential phase shift, θ, can therefore be evaluated from the equation

$$\theta = \cos^{-1}\left(1-\frac{c^2}{2a^2}\right)$$

or from Fig. 6.

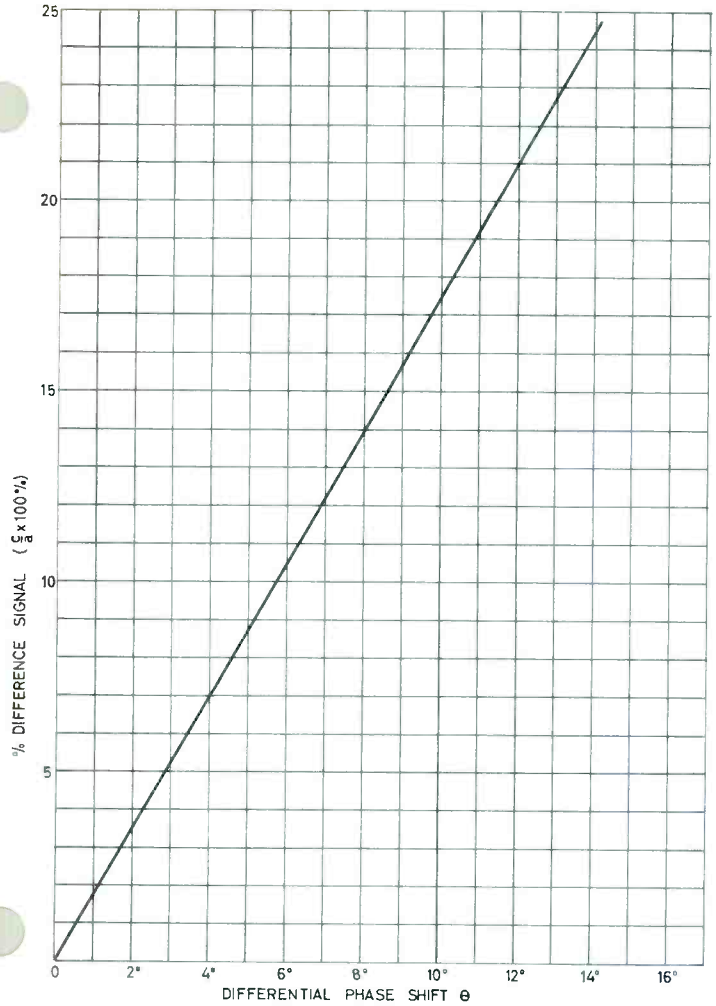


Fig. 6.
Stereophonic phase angle chart

a Lissajous figure. The phase distortion may then be observed easily although no precise measurements are possible.

In the case of a stereophonic system a test to check the difference in phase between the two channels may fairly easily be applied. The inputs to the two channels are connected in parallel to the B.F.O. output from the wave analyser. The outputs, a and b, from the amplifiers are connected in turn to the input of the wave analyser and adjusted to be equal to each other. They are then connected in series antiphase to the input of the wave analyser. The reading, c, on the wave analyser will now give a measure of the phase shift between the two channels. The actual phase shift may then be obtained from Fig. 6. It will be seen that phase shifts of the order of 1° are easily discernible.

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ABRIDGED SPECIFICATION

Input Frequency
WORKING RANGE: 20 c/s to 50 kc/s.

Input Voltage
WORKING RANGE: 3 µV to 300 V.
Lowest full scale 30 µV.

Residual Hum and Noise
At least 80 dB down.

Internal Distortion
At least 85 dB down.

Selectivity
3 dB down, bandwidth 6 c/s ±0·5 c/s.
45 dB down, less than ±20 c/s off tune.
65 dB down, less than ±40 c/s off tune.
85 dB down, less than ±70 c/s off tune.

Input Resistance
100 kΩ on 30 mV to 1 V ranges, 1 MΩ on 3 to 300 V ranges.

Restored Frequency Output
Variable up to 1 V r.m.s. at 600 Ω.

B.F.O. Output
Variable up to 1 V r.m.s. at 600 Ω.

Recorder Output
100 µA into 2·5 kΩ.

A.F.C.
Holds in over ±100 c/s.

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621.317.353

Distortion Factor Measurements

by E. C. CRAWFORD,
Graduate I.E.E.

A Distortion Factor Meter is a simple instrument for the rapid appraisal of total harmonic and noise content of audio signals. After considering and comparing various methods of calculating distortion factor, the article discusses the design of an instrument capable of measuring down to 0·05% distortion.

IF THE VOLTAGE of a distorted sinusoidal signal is multiplied by its 'Distortion Factor' the voltage of the total distortion content is obtained. The strictest definition requires that a true r.m.s. method is used for the voltage measurement, but it is found that the simpler average voltage measurement gives results adequate for practical purposes.

Distortion factor is closely related to the percentage distortion usually quoted for quality amplifiers, and to the percentage total harmonic content quoted in other contexts. The difference lies in that the distortion factor can include all noise and hum, not necessarily harmonically related to the fundamental, whereas percentage distortion usually relates to harmonic distortion only. All of them relate to the unwanted additions made by the imperfect equipment to a single pure sinewave input. It is important to note that the terms strictly refer only to a voltage ratio, and not a power ratio as the latter can seem to be very much better; 1% distortion on a power basis is equivalent to 10% by the usually quoted voltage method.

The distortion factor meter is an instrument arranged so that the distortion factor as explained above may be quickly and simply measured; these two qualities express the advantage of this instrument compared with the wave analyser, or intermodulation test set. It gives an answer in terms commonly quoted in specifications and which is obviously chosen to quickly summarize the performance of the amplifier or oscillator.

How it works

The predominant tone in the signal, that is the fundamental, must be eliminated before the distortion content alone can be measured. The usual method is to employ a sharply tuned rejection filter which attempts to block a single frequency only. This filter may be a bridged 'T' comprising capacitive and inductive elements; alternatively a twin 'T' or a Wien bridge using capacitive and resistive elements may be used.

Inductive filters can be the most effective, but are limited by their size and weight, particularly at low frequencies, and by a tendency to cause distortion if the core permeability is non-linear.

Capacitance-resistance filters may have improved skirt response, as shown in Fig. 1, if negative feedback is employed. It is important for instance that, when the fundamental is rejected, the rejection does not significantly spread to the harmonic content; otherwise the distortion factor will apparently be lower than it is. If

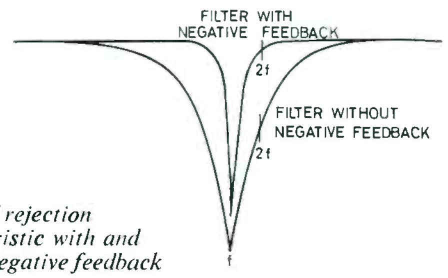


Fig. 1.
Shape of rejection characteristic with and without negative feedback

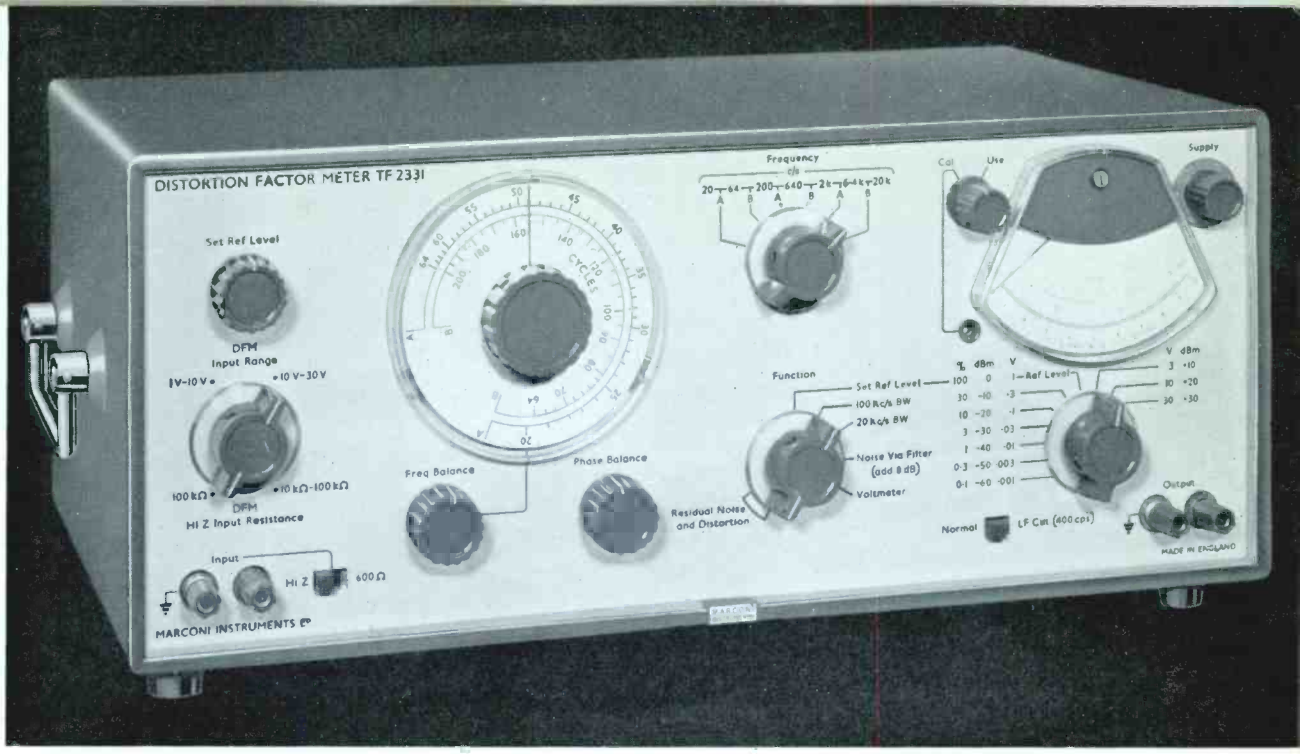
the fundamental rejection filter is in the forward signal path of an amplifier with amplification A without feedback, then the application of negative feedback will tend to maintain the overall gain constant except where A is reduced to almost zero by the rejection filter.

If we have a fundamental signal V with distortion D and noise N the total signal is $(V+D+N)$. If V is now eliminated $(D+N)$ remains.

There are two common methods of comparing $(D+N)$ with $(V+D+N)$. The first, as shown in Fig. 2, is to measure the two with a suitable voltmeter which inevitably is subject to range switching, frequency characteristic and scale law errors, but which on the other hand is simple and direct reading particularly if $(V+D+N)$ is somehow standardized to read 100%. $(D+N)$ is then read off as % distortion factor after V is rejected.

The second method, as shown in Fig. 3, is to read $(D+N)$ on a voltmeter scale, adjusting the sensitivity so that the reading falls on a suitable mark. The rejection is then by-passed allowing $(V+D+N)$ to show on the meter with an attenuator calibrated in percentage distortion inserted in the signal path. This attenuator is adjusted so that $(V+D+N)$ reads on the same mark as $(D+N)$. The accuracy of this second method is limited by the characteristics and calibration accuracy of the attenuator with many of the same sort of errors of the first method except that the accuracy of the meter does not enter into it. The scale law of the second method is limited by the accuracy of calibration of the fine variable control of the attenuator, whereas the first method is

Successor to the well-established TF 142F, this new solid state design has increased facilities for voltage, noise and distortion measurements. Also available as a rack mounted version



finally limited by the calibration accuracy of the meter.

If the final detector and meter is a true r.m.s. device the distortion factor is given by:

$$D.F._{\text{r.m.s.}} = \frac{\sqrt{D_1^2+D_2^2+\ldots+N^2}}{\sqrt{V^2+D_1^2+D_2^2+\ldots+N^2}}$$

This is the distortion factor as commonly defined. In many audio instruments, however, an average level

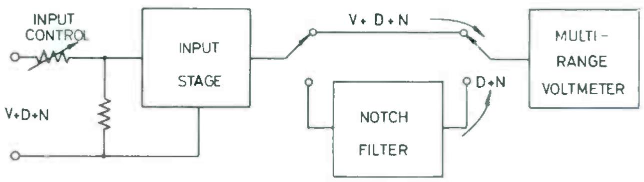


Fig. 2. Basic distortion factor meter: method 1

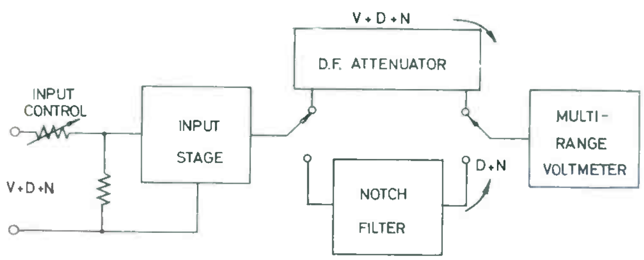


Fig. 3. Basic distortion factor meter: method 2

detector and indicating meter are used such that the distortion factor is given by

$$D.F._{\text{av}} = \frac{D_1+D_2+\ldots+N}{V+D_1+D_2+\ldots+N}$$

Both expressions for D.F. are commonly expressed as a percentage.

It can be shown that if the r.m.s. sum of $(D+N)$ is equal to V, then the D.F. cannot be greater than 70·7%. It is sometimes seen in specifications, however, that distortion of up to 100% may be measured. This must obviously refer to another definition of distortion which relates the noise and distortion $(D+N)$ not to $(V+D+N)$ but to V only. This definition is occasionally called the True D.F.

$$D.F._{\text{true}} = \frac{\sqrt{D_1^2+D_2^2+\ldots+N^2}}{\sqrt{V^2}}.$$

In normal audio equipment the discrepancy is negligible since, with a distortion as high as 14%, the difference between the $D.F._{\text{true}}$ and $D.F._{\text{r.m.s.}}$ is less than 1 part in 100.

The difference between $D.F._{\text{av}}$ and $D.F._{\text{r.m.s.}}$ largely depends upon the third harmonic content, its phase relationship to the fundamental, and whether it is about equal to the second or fourth harmonics in amplitude. The greatest difference will be about 1½ dB—which is hardly discernible by the human ear.

D.F.M. Input Circuits

The input resistance must either be a suitable load for the audio source or be sufficiently high to be negligible. It is thus unlikely that the fundamental rejection filter can be connected directly to the input terminals, so that there must be a buffer stage. If the signal level was always sufficient this buffer could obviously be a passive resistive attenuator, but many audio sources operate at outputs of 0 dBm which is 0·775 V on 600 Ω; therefore resistance isolation cannot be used without serious loss of signal/noise ratio, particularly if the distortion is approaching 0·1% or −60 dBm.

The alternative is a buffer amplifier which must transmit the signal with a minimum of added distortion and noise. The added noise may be reduced in significance by increasing the input level, but this can result in the distortion of the signal under test being increased. It is usual, therefore, for the input stage working conditions to be optimized for a predetermined level of signal, usually about 0 dBm. Stronger signals can then be attenuated to the reference level, but weaker signals cannot be tested to the same low level of distortion factor because the introduced noise and lack of sensitivity of the voltmeter section will eventually determine the lowest limit.

The noise and distortion introduced by the input stage are measured as if they were part of the signal distortion. In general a minimum figure, usually less than 0·1% D.F., is added to all readings. For example, if the instrument noise and distortion is quoted as 0·025% it is probably more accurate to deduct, say, 0·015% D.F. from all readings of D.F. of 0·2% and less, and also accept an error of about ±0·015% D.F. in addition to any other quoted source of error.

Noise Measurement

After the fundamental frequency has been rejected, the residual signal is a mixture of noise and distortion. If the input to the amplifier on test is now removed, the residual measured is the amplifier noise which is likely to be composed of two main components. One will be the mains hum, which will be predominantly $2f$ from the h.t. and f from the heaters, where f is the mains supply frequency. The other will be mainly white noise to the limit of the amplifier bandwidth. If this is heard on a loudspeaker, the predominant noise apart from any excessive hum will be of a s-s-s-sh nature. This is because the ear is most sensitive to frequencies in the band 3 to 6 kc/s. It is thus considered more realistic to assess the noise after it has been weighted according to the

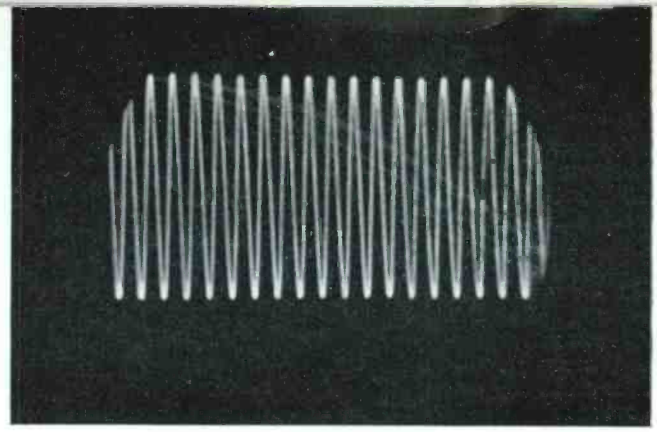


Fig. 5. Effect of l.f. cut filter in distortion measurements. (a) 400 c/s signal with about 1½% distortion

frequency sensitivity of the average ear and reproducing device.

The C.C.I.F. has decided upon standard responses for what are termed psophometric weighting filters. There are two of these, one of which is intended to represent

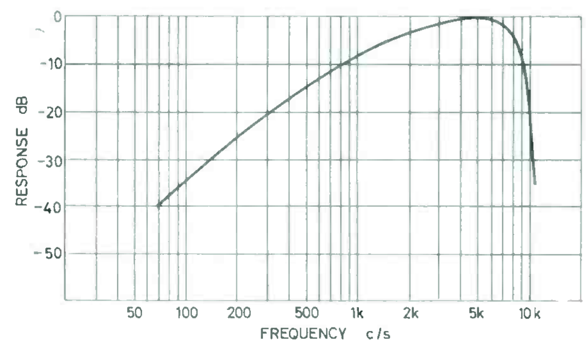


Fig. 4. C.C.I.F. broadcast weighting network response

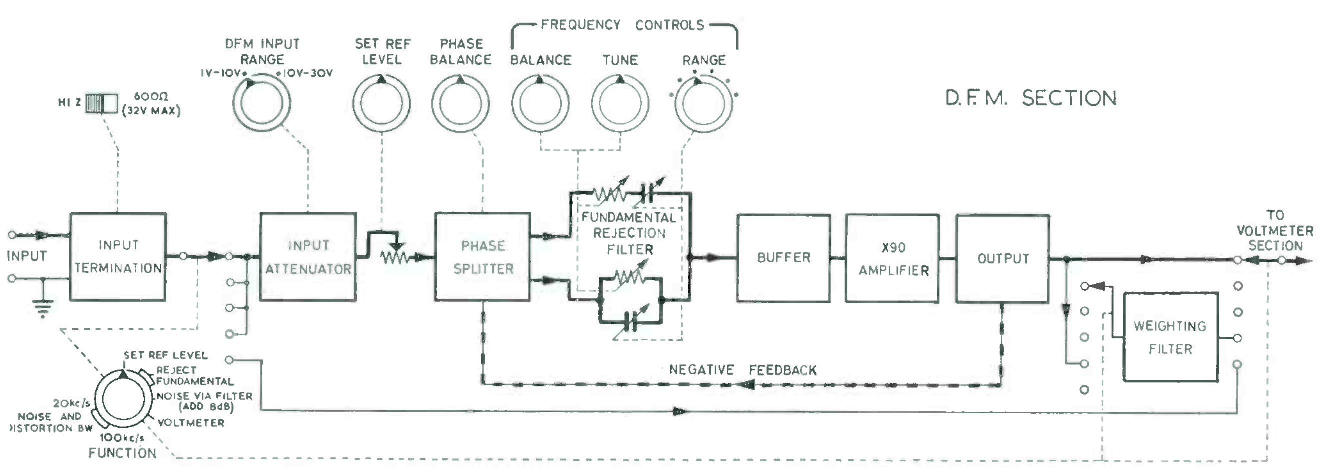


Fig. 6. (above and opposite). Block diagram of Distortion Factor Meter type TF 2331

ABRIDGED SPECIFICATION

Fundamental Frequency

RANGE: 20 c/s to 20 kc/s.

REJECTION: At least 80 dB.

Input (for noise or distortion measurement)

TERMINATED: 600 Ω. Minimum input 0 dBm.

HIGH IMPEDANCE:
10 kΩ/V up to 10 V input.
100 kΩ from 10 to 30 V input.

Distortion and Noise Measurement

RANGE:
Down to 0·05% from 200 c/s to 6 kc/s.
Down to 0·07% from 20 c/s to 200 c/s, and 6 kc/s to 20 kc/s.

BANDWIDTH: 100 kc/s or 20 kc/s.

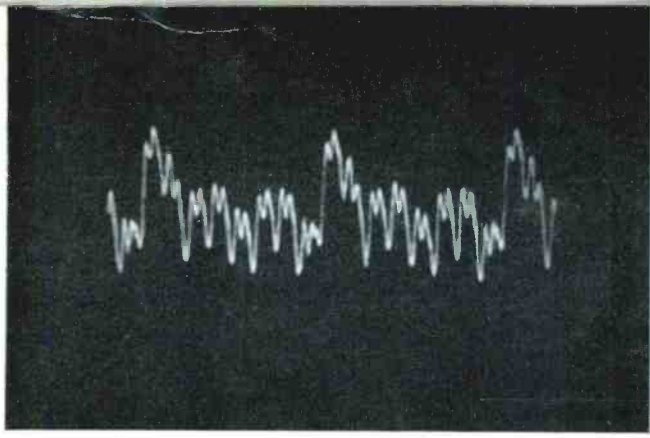
L.F. CUT: Can be introduced below 400 c/s to attenuate power supply frequencies.

NOISE WEIGHTING: Via C.C.I.F. broadcast weighting network.

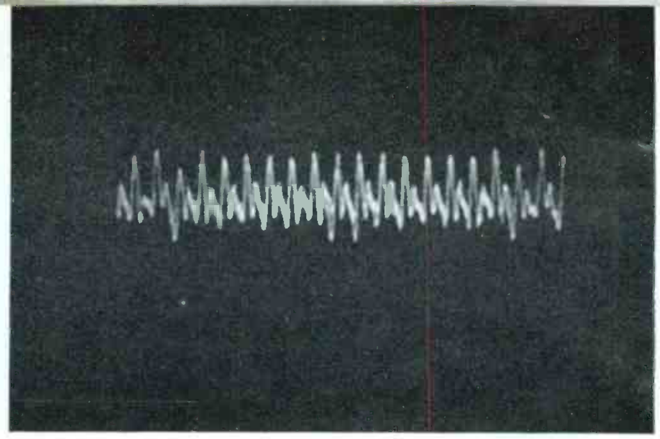
Voltage Measurement

INDICATION: 1 mV to 30 V full scale. Also decibel scale calibrated from −12 to +2 dB relative to 1 mW in 600 Ω.

INPUT RESISTANCE: 1 MΩ or 600 Ω.



(*b*) *Rejection filter eliminates fundamental. Large peaks are caused by 50 c/s hum*



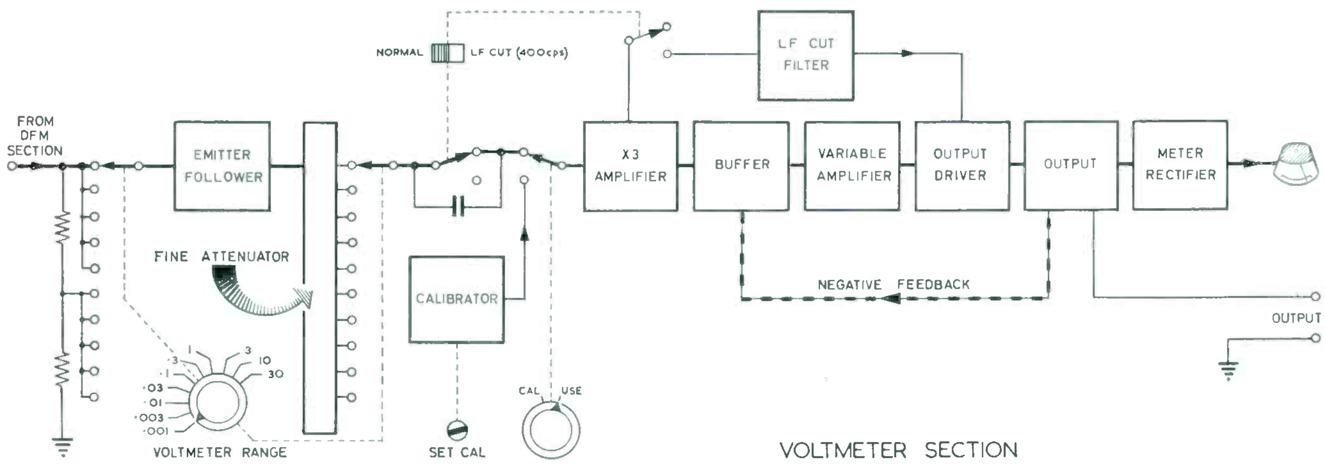
(*c*) *With l.f. cut filter switched in hum is removed and only distortion remains*

broadcast conditions, as shown in Fig. 4, and the other represents a telephone line and headset response. Both are rather peaky in their characteristic, and recognize the limitations of the sensitivity of the ear to low frequencies, and frequencies in excess of 10 kc/s.

A distortion factor meter can be fitted with such a weighting filter for noise measurement with tone off, but it is impracticable to include the filter when measuring distortion due to the very large insertion loss at low frequencies. To overcome this and yet maintain a suitable reference level for $100\% = V+D+N$ would result in gross overload of the input stage.

Alternative arrangements for noise measurement are a flat response to, say, 100 kc/s or to the limit of the audio band of 20 kc/s. In order to discriminate against mains hum, it is useful to have a low frequency elimination filter which cuts 100 c/s by over 20 dB. Such a filter gives negligible attenuation above, say, 400 c/s and the white noise content lost by its action is in general insignificant in a total noise bandwidth of at least 20 kc/s.

A 400 c/s high pass filter can also be very useful during distortion measurement when it is otherwise difficult to decide whether the residual signal is mainly due to distortion or whether there is a large proportion of mains hum. This is shown in Fig. 5.



Bandwidth for Distortion Measurement

It is debatable whether inaudible distortion is worth measuring. This includes distortion above a fundamental frequency of about 8 kc/s, as the usual upper limit of loudspeakers and the ear is limited at 16 kc/s. No doubt the presence of second or third harmonic components of a 16 kc/s tone are inaudible in themselves, but perhaps they may signify the possibility of audible intermodulation tones should two frequencies in this region be simultaneously amplified.

The distortion of test oscillators must be measured, however, and those for a.f. use usually tune up to 20 kc/s. For their benefit at least then, the bandwidth for distortion measurement must include the second and third harmonics of 20 kc/s. On the other hand, it is occasionally useful to limit the distortion measurement to a frequency range such as 20 kc/s, rejecting harmonics and noise when these must obviously be inaudible.

Distortion Factor Meter type TF 2331

This is a new distortion factor meter featuring complete solid state design. Although normally powered by a.c. mains, an external battery supply can be used. Input voltage range, for distortion measurement down to 0·05% D.F. on a direct reading meter of 0·1% full scale, is from 0·775 V up to 30 V r.m.s. The fundamental frequency rejection filter is tuned by a directly calibrated dial with fine controls so that virtually complete fundamental rejection can be obtained over a frequency range from 20 c/s to 20 kc/s.

Bandwidth for noise and distortion measurement is either 20 kc/s or 100 kc/s. Indication of distortion factor is presented on the internal voltmeter; this can also be used independently with full scale ranges of 1 mV to 30 V and a frequency range to 100 kc/s. An l.f. cut facility eliminates mains hum and a psophometric weighting filter enables effective noise assessment to be made. The input resistance is either a 600 Ω termination or high resistance from 10 kΩ to 100 kΩ depending upon the level. The voltmeter section has amplifier output terminals for oscilloscope examination of the residual noise and distortion or the original signal. Used as an independent voltmeter the input resistance is 1 MΩ.

621.396.822

WAVE ANALYSER IMPROVES SIGNAL TO NOISE RATIO

by J. M. PARKYN

A method of measurement is described whereby a general purpose receiver and signal generator can be used in conjunction with a wave analyser to make attenuation measurements where the signal is 20 or 30 dB below 1 μV, and would normally be masked by the presence of noise.

USING modern instrumentation techniques it is possible to design specialized instruments to make almost any required measurement. However, such instruments are generally very expensive and often limited in application to the one measurement for which they were designed. Hence most laboratories are equipped with general purpose test gear and specialized equipment is kept to a minimum. A system will be described here which makes use of general purpose laboratory test gear to make measurements of very low level signals which would

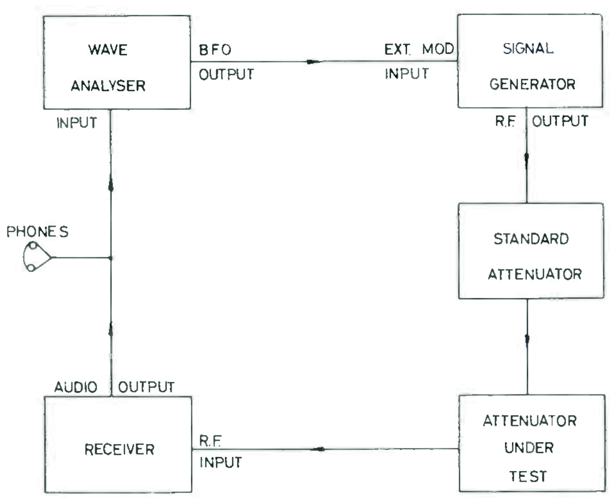


Fig. 1. Slideback method of comparing two attenuators

normally be masked by the noise produced in the receiver. A typical application is the comparison of two r.f. attenuators by the slideback method, and Fig. 1 shows the interconnections necessary. If the attenuator of the signal generator is sufficiently accurate the same system can be used without the external standard. Alternatively, omitting the external attenuator under test in the diagram, the system can be used to check the signal generator output against the standard attenuator.

Normal slideback measurements of this type are impracticable at the higher attenuation settings because of the low signal/noise ratio. This difficulty can be overcome by using a wave analyser of the type which delivers a b.f.o. output centred on the very narrow bandwidth of the selective amplifier within the instrument, such as the Marconi Instruments type TF 2330. This b.f.o. output is used to modulate a signal generator which is set to the carrier frequency required for the tests. The wave analyser is set to some convenient frequency well within the normal audio range to suit the signal generator and receiver, say 400 or 1000 c/s. A convenient modulation depth is about 30 to 50%, although the exact value is not important. In some cases, for example the TF 144H Standard Signal Generator, the b.f.o. output voltage from the wave analyser will not be sufficient to adjust the metering system of the signal generator to the 80% SET MOD deflection, but this is not important provided some modulation is produced. Also the type of modulation is not important provided a suitable signal generator and receiver are chosen, using either f.m. or a.m.

Monitoring with the phones, the receiver should be tuned, with the signal generator output level and the setting of the two external attenuators adjusted to give a small receiver input signal, which has a workable signal/noise ratio, say somewhere between 1 and 10 μV. The wave analyser sensitivity controls can now be adjusted to give a substantial meter deflection with the half volt or so which will be present at the receiver output.

The full r.f. attenuation can now be brought into circuit and the wave analyser sensitivity increased. Measurements can be made at and below 0·1 μV with a steady wave analyser meter deflection in spite of the signal at the phones being lost in the noise level.

With this system, the wave analyser and receiver have in fact become a very narrow band 6 c/s receiver, but the frequency stability normally required for such a receiver is not in this case necessary since the receiver output, being derived from the wave analyser b.f.o. output, will always match the very narrow bandwidth of the wave analyser. Receiver bandwidth can be large enough to allow for receiver and signal generator frequency drift and error of setting. The drift normally encountered in the backed off d.c. coupled circuit when connecting an S-meter to a receiver is also avoided.

621.317.75

MARCONI INSTRUMENTS

Special versions of Oscilloscope . . TYPE TF 2200

by M. W. G. HALL,
A.M.Brit.I.R.E.

Five modified versions of the 35 Mc/s measuring oscilloscope type TF 2200 have been produced to suit special applications. The additional facilities offered include sine² pulse and bar testing of television equipment, single shot triggering of the delaying time base, extra control of fine delay for use with the delaying time base in the gated mode, and versatile external triggering of the delaying time base independently of the main time base.

THE TF 2200 is a high grade wide band measuring oscilloscope[1] equipped with comprehensive facilities for the display and measurement of many types of waveform.

Various specially modified versions have been derived to extend the use of this instrument to the particular requirements of certain users.

These instruments have the following type numbers:

1. TF 2200/1M1. Main time base modified for sine² pulse and bar waveform testing of TV equipment.
2. TF 2200/2M1. Enables single shot triggering of delaying time base.
3. TF 2200/3M1. Has the facilities of TF 2200/1M1 but with additional variable fine delay for use with delaying time base in the gated mode.
4. TF 2200/4M1. As TF 2200/3M1 but less the modified time base.
5. TF 2200/5M1. The delaying time base may be externally triggered independently of the main time base by either positive or negative going edges.

Below is given a detailed account of the modifications and the particular applications for which they were developed.

TF 2200/1M1

The choice of the sine² pulse and bar waveform for the assessment of performance of television video equipment has been described in detail elsewhere.[2, 3, 4] A typical waveform is shown in Fig. 1a. Measurements of distortion are made using a k factor graticule as illustrated in Fig. 1b.

Briefly, the pulse characteristics are such that the envelope of the frequency spectrum falls with increasing frequency. The component harmonics of the 10 kc/s repetition frequency (fundamental line frequency) have zero amplitude at f_c, the system cut-off frequency, for the 2T pulse and are 6 dB down in the case of the T pulse. For the British 405 line television system these considerations dictate a pulse width at half amplitude of 333 nsec for the 2T pulse and 167 nsec for the T pulse.

Using the k factor graticule it is necessary to set the sweep speed to 333 nsec/cm. With oscilloscopes other than specialized instruments this involves careful adjustment of sweep speed, X gain, etc., and sometimes the imposition of a timing waveform.

To obviate tiresome adjustments and consequent checking during a protracted measurement period, it was decided to delete the fastest time base range, viz. 50 nsec/cm, and substitute one of 333 nsec/cm. This is simply achieved by changing the range resistor.

Fig. 1a. Sine-squared pulse and bar waveform

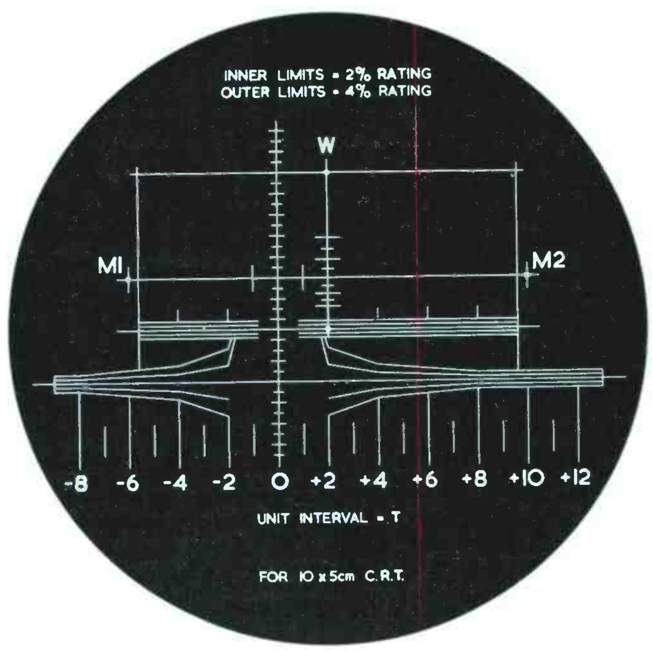


Fig. 1b. K-factor graticule

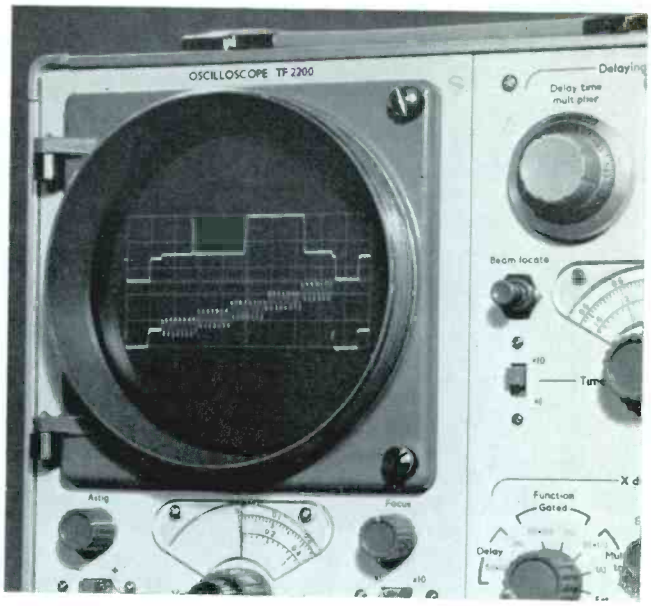
TF 2200/2M1

The standard instrument has a system for single shot operation which permits the main time base to be either fired manually or armed manually and then fired by internal or external electrical trigger pulses.

For a particular application, an aerodynamic shock tube experiment, it was required to permit single shot operation with the instrument arranged for use in the delayed sweep mode. That is, it was necessary to arrange for the B time base, or delaying sweep, to be initiated by either an external or internal trigger pulse and then, at the completion of the pre-set delay period, to fire the main time base, the resultant trace being recorded photographically. A previous *Instrumentation* article[1] describes the action of the function circuit. To provide single shot operation the twin triode forming this circuit is switched to operate as a bistable stage which turns off the diode gate at the completion of the main sweep. When the FUNCTION switch is set to DELAY this circuit is rearranged so that one half functions as a cathode follower.

Thus, to provide single shot operation in the DELAY mode it is necessary to re-wire the function stage so that it can again operate as a bistable pair, provide the necessary bypass circuit since the cathode follower no longer exists, and to provide an additional diode gate so that the delaying time base can itself be locked out at the completion of the first sweep. To assist in this switching operation an additional wafer is assembled on to the ARM/NORMAL/FIRE switch.

The action of the circuit may be described by referring to Fig. 2, which is a modified form of the simplified function circuit which appeared as Fig. 12 in the original article. Two main additions have been made, a resistor-diode combination formed by R_1, R_2, R_3 and D_7 and a similar circuit formed by R_4, R_5 and D_8. The anode side of D_7 is tied to earth via R_3, the other side being connected to the cathodes of the single shot multivibrator stage. When the bistable is in the 'ready' state, i.e. V_2 off, the cathodes are at zero potential and thus diode D_7 will permit a negative going trigger pulse to fire the delaying time base and hence, after the selected



Display of television test waveforms by Oscilloscope type TF 2200

delay period, initiate the main time base sweep. At the completion of the forward stroke the trailing edge of the tube unblanking pulse reverses the condition of the bistable pair, cutting off V_1; this causes the potential of the cathodes to rise to +12 V, so reverse biasing D_7 and preventing further trigger pulses from reaching the delaying time base.

Since V_2 no longer performs as a cathode follower, an alternative path must be provided for the delay time

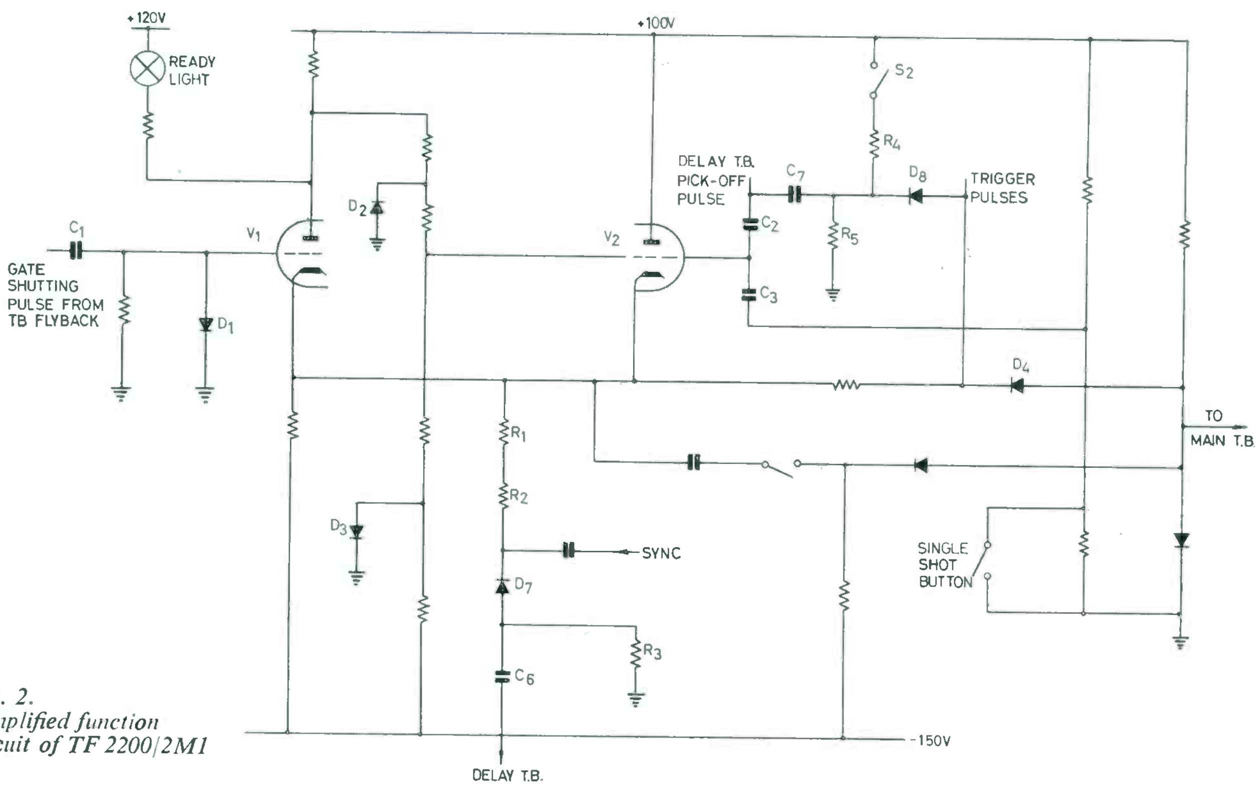


Fig. 2.
Simplified function circuit of TF 2200/2M1

base pick-off pulse to the main time base, via the gate diode D_4. This path is formed by C_7 and D_8, the bias condition of the latter being controlled by the switch S_2 and resistors R_4 and R_5. The diode D_4 is normally reverse biased via S_2 and R_4 by the 100 V supply line. However, when the ARM/NORMAL/FIRE switch is set to ARM the additional contacts are opened so that D_8 is forward biased, thus passing the delay time base pick-off pulse to the main time base.

TF 2200/3M1

The standard version was designed to provide the basic facilities for the great majority of oscilloscope measurements. The trigger mode selector switch may be set to a position labelled TV under which conditions the time base will be triggered by the frame synchronizing pulses of a television signal.

Televising the internationally renowned Black and White Minstrel Show at the studios of the B.B.C. Behind the scenes, Marconi Instruments Oscilloscopes TF 2200/1M1 and TF 2200/3M1 play their part in keeping up the highest standards of performance in the complex studio equipment

Photograph by courtesy of the B.B.C.

To carry out the sine² pulse and bar measurements referred to in the section describing the -/1M1 version it may be required to strobe a particular line from each alternate interlaced field. This demands that the oscilloscope shall trigger at frame frequency, provide a delaying sweep and then display the desired line. Thus the main sweep duration may be either 100 μsec, to display the complete line (405 line system) or 3·33 μsec to carry out k factor measurements. This occurs at a repetition rate of 25 per secoud with a delay between triggering and main sweep initiation varying up to 20 msec. It is therefore clear that the most stringent requirements are placed upon the oscilloscope performance. In the case of the TF 2200 series the high performance of the cathode-ray tube with its 10 kV p.d.a. potential overcomes the difficulty of high writing speeds at low p.r.f's.

This oscilloscope has an invaluable feature in that the FUNCTION switch may be set to a position labelled DELAY GATED. In this condition at the completion of the delaying sweep, a diode gate is opened which permits the main time base to be fired by the next excursion of the function under examination through the level selected by the TRIGGER LEVEL control. This procedure substantially eliminates jitter between the trigger pulse and the portion of waveform to be examined, and indeed eliminates any slight inherent jitter in the action of the delaying sweep. Using the variable delay control will then cause the display to 'jump' from one repetitive portion of the signal to the next. Thus it is not possible to explore the portion of waveform selected by using a fast main time base range.

This version was produced to enable extra variable delay to be introduced while operating in the gated mode. The requirement was to cover a variable range in excess of 100 μsec so that a sine² pulse appearing as video information on a particular line could be isolated and examined on the previously mentioned special time base range of 333 nsec/cm.

Fig. 3 gives a block diagram showing how the additional variable delay generator has been inserted between

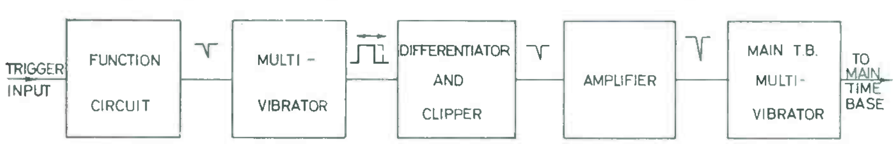


Fig. 3.
Block diagram of additional delay circuit in TF 2200/3M1

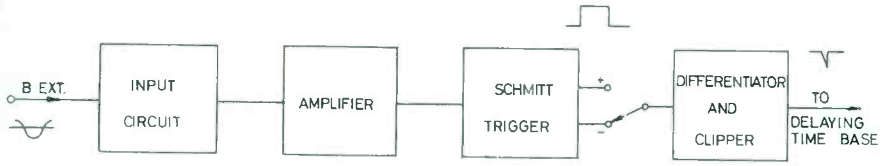


Fig. 4.
Block diagram of triggering circuit for delaying time base in TF 2200/5M1

the diode gate, which forms the output stage of the FUNCTION circuit, and the input to the multivibrator, which controls the main time base. The delay is generated by a cathode-coupled monostable multivibrator of variable period, the output of which is differentiated, and the positive going spike removed, the remaining negative going pulse is then amplified to a level sufficient to trigger the main time base.

TF 2200/4M1

This has the additional variable delay of the previous model but is intended for use on 625 line TV systems where the video bandwidth extends to 5 Mc/s. For such a system the 2T sine^2 pulse has a half amplitude duration of 200 nsec/cm and as a time base range of 200 nsec/cm is already provided as standard, no further modification is necessary.

TF 2200/5M1

In this development rather more versatile facilities for separately triggering the delaying time base are provided. On the standard version the TRIGGER MODE switch may be set to B EXT, when it becomes possible to trigger the delaying sweep with a negative going pulse in excess of 2 V applied to the adjacent socket. The particular requirement was to obtain this facility for both positive and negative going signals over an amplitude range of 0·5 to 500 V.

Fig. 4 shows a block diagram of the additional circuitry which, for ease of installation, is transistorized. Following the input protection circuit formed by the ratio between a fixed resistor and the forward resistance of a diode, the trigger pulse is passed to a two-stage amplifier, the first stage being a normal common emitter configuration, the second an emitter follower. The output from the latter operates a Schmitt trigger, the signal from either collector being selected to correspond to a positive or negative-going input signal. Since the delaying time-base input Schmitt circuit requires a negative going pulse, the output of the transistor Schmitt circuit is differentiated and the positive going spike removed.

An additional potentiometer is fitted to adjust the input level, so that for large signals spurious triggering will not occur on noise pulses.

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2. Macdiarmid, I. F., 'Waveform Distortion in Television Links'. *P.O.E.E.J.*, July and October 1959, 52, pp. 108, 188.
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4. Potts, K. H., 'The Performance Evaluation of Television Line and Microwave Links Using Sine-Squared Test Techniques'. *Radio and Electronic Engineer* (*J. Brit. I.R.E.*), April 1963, 25, p. 377.

NEW DIRECTOR FOR MARCONI INSTRUMENTS

PROFESSOR BARLOW, Ph.D., B.Sc., F.R.S., M.I.Mech.E., M.I.E.E., Pender Professor of Electrical Engineering at University College, London, has been elected to the Marconi Instruments Ltd. Board of Directors. He is well known in scientific circles for his work in the micro-wave field and in power measurement.

He was born in London in 1899 and educated at Wallington Grammar School, City and Guilds Engineering College, and University College, London. He gained his early practical engineering training with the East Surrey Ironworks and Barlow and Young Ltd.

In 1925 he joined the Faculty of Engineering at University College, London, and, apart from war service, he has been a member of the Academic Staff ever since. During the war he worked on the development of radar with the Telecommunications Research Establishment at the Air Ministry, and in 1943 he was appointed Superintendent of the Radio Department at the Royal Aircraft Establishment, Farnborough.

At the end of the war he rejoined University College as Professor of Electrical Engineering, and a year later he was awarded his Fellowship. In 1949 he was appointed Dean of the Engineering Faculty. Since 1961 he has been a member of the University College Committee.

Professor Barlow is a member of the B.B.C. Scientific Advisory Committee, the Radio Research Board of the D.S.I.R. and the Council of the I.E.E. He is the author of many scientific papers.

621.396.615.12

MARCONI INSTRUMENTS

An F.M. Stereo Version

OF F.M./A.M. SIGNAL GENERATOR . . . . . *TYPE TF 995A/2*

by W. OLIVER

A modified TF 995A/2 F.M./A.M. Signal Generator is described which in conjunction with a stereo modulation generator provides an f.m. stereo signal. This meets the requirements of the U.S. Federal Communications Commission (F.C.C.) for an f.m. transmitter using the GE-Zenith F.M. Stereo system.

MANY F.M. receiver manufacturers in the U.S.A. use the TF 995A/2 as a general purpose signal generator, and, with the advent of f.m. stereo broadcasting[1] in June 1961, it became necessary to extend the modulation frequency response of the TF 995A/2 to meet these new requirements.

Description of the GE-Zenith System

Standard f.m. broadcast signals have a modulation frequency range of 50 c/s to 15 kc/s, maximum deviation is ±75 kc/s and the f.m. broadcast band extends from 88 to 108 Mc/s (Band II). To produce a stereophonic effect for the listener, it is necessary to transmit two independent modulation frequency signals, one corresponding to the signal picked up by a microphone on the left-hand side of the broadcasting studio and the other by a microphone on the right.

The f.m. stereo system approved by the F.C.C. for use in the U.S.A. is the GE-Zenith type. In this, the left plus right channel signals (L + R) are transmitted in the modulation frequency range 50 c/s to 15 kc/s and the left minus right channel signals (L−R) are transmitted in a similar bandwidth on a 38 kc/s subcarrier; the spectrum of the resulting complex modulation signal is shown in Fig. 1.

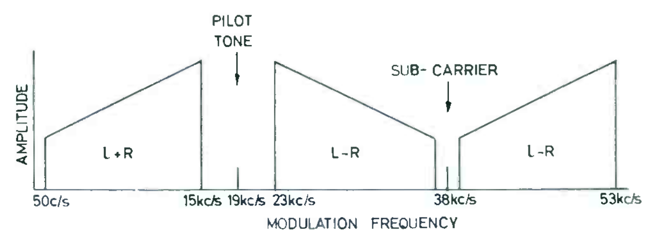
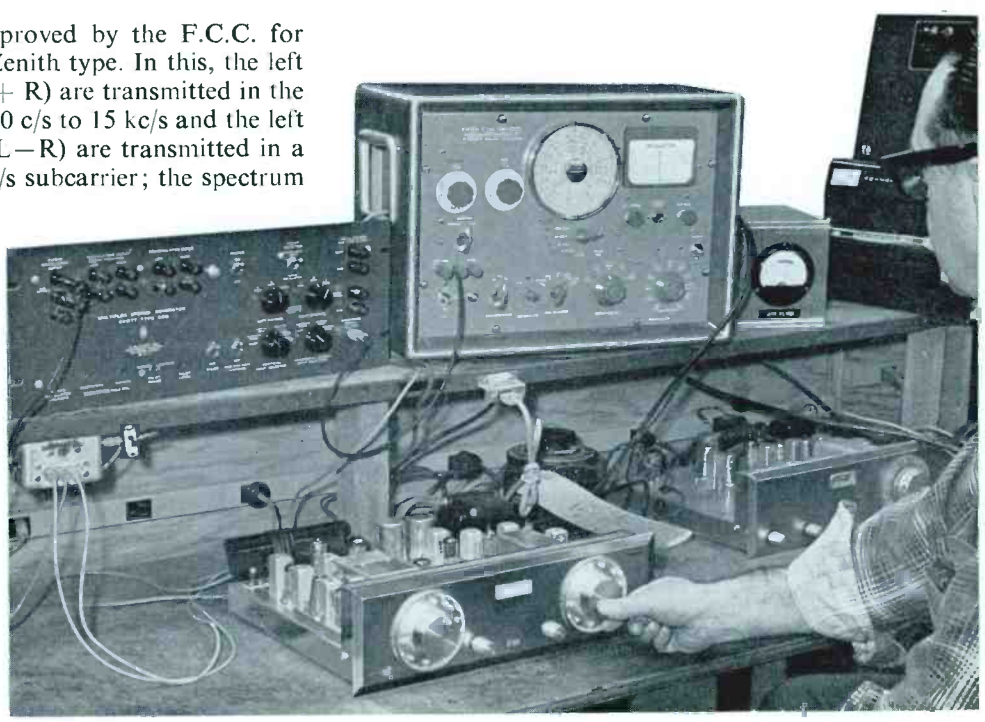


Fig. 1. Frequency spectrum of stereo modulation signal

Pre-emphasis is used on both the main channel and subcarrier so that corresponding de-emphasis may be used at the receiver to reduce the effective audio bandwidth and therefore to improve the signal/noise ratio.



Quality control technician testing an H. H. Scott A.M./F.M. stereo receiver with the TF 995A/8M1 Signal Generator

The left and right channels are added and form the modulating signal for the standard modulation frequency band of 50 c/s to 15 kc/s. This is the signal that a monophonic receiver accepts and a listener to such a receiver would not realize that a stereo signal was being picked up. In other words, the GE-Zenith Stereo system is compatible with existing monophonic f.m. receivers.

The L−R channel signal is used to amplitude modulate a 38 kc/s suppressed carrier. Since the modulation frequency components extend from 50 c/s to 15 kc/s, the 38 kc/s carrier sidebands extend from 23 to 53 kc/s. To demodulate a double sideband suppressed carrier signal, it is necessary to re-insert the carrier in frequency and phase; a pilot tone synchronizing signal is transmitted at 19 kc/s for this purpose. After f.m. demodulation, the 19 kc/s signal is separated from the other modulation components, applied to a frequency doubler and becomes the 38 kc/s re-inserted carrier. If conventional amplitude modulation were used to transmit the L−R signal instead of the double sideband suppressed carrier system, a large 38 kc/s carrier component would be present at all times. This would produce a standing deviation on the transmitter which would reduce the deviation available for the transmission of information of the other components, and give a reduction in the system signal/noise ratio and an increase in crosstalk.

The complex waveform corresponding to Fig. 1 is produced by a stereo modulation generator and it is this composite signal that is applied as modulation to the f.m. transmitter or signal generator.

One of the F.C.C. requirements for an f.m. stereo system is that it must be possible to add a further subcarrier at 67 kc/s which would be frequency modulated ±7 kc/s with 'background' music for restaurants, hotels, etc. This channel, known as the S.C.A. sub-channel (Subsidiary Communication Authority), is not shown in Fig. 1 because it does not relate directly to f.m. stereo.

Requirements for an F.M. Stereo Signal Generator

To meet F.C.C. requirements for an f.m. stereo transmitter (which would apply to a good f.m. stereo signal generator also), the external f.m. response should be flat within ±0·3 dB from 50 c/s to 53 kc/s and phase shift should be less than ±3° over this range. These two specifications are assumed to be met if the channel separation exceeds 29·7 dB over the frequency range 50 c/s to 15 kc/s. In addition to these particular requirements, the generator should tune over the i.f. (10·7 Mc/s) and r.f. (88–108 Mc/s) ranges of f.m. stereo receivers, should have adequate r.f. output and shielding, and have an attenuator suitable for use down to 1 μV or below. Also, since this is a 'high fidelity' application, harmonic distortion should be as low as possible (say <1%).

The TF 995A/2 F.M./A.M. Signal Generator was chosen for this application and, with modifications to improve the external modulation frequency response and sensitivity, it makes an excellent signal generator for f.m. stereo purposes. This modified version is designated TF 995A/8M1.

After modification, the external f.m. frequency response was measured as 1·5 dB down at 3 c/s and 150 kc/s. The flatness of modulation response and amount of phase shift over the 50 c/s to 53 kc/s range was inferred by measuring the channel separation from 50 c/s to 15 kc/s as suggested in the F.C.C. specification. Channel separation was measured with a stereo modulation generator and a Marconi Instruments type TF 928 F.M. Deviation Meter, which was used as a standard f.m. demodulator. The figures obtained include the performance of all three instruments from which only the modulation generator's result can be subtracted simply. Combined separation figures of the TF 995A/2 and TF 928 were calculated to be 30 dB at the band limits, rising to 36 dB at 1 kc/s, showing that the separation of the individual instruments exceeds the F.C.C. requirements.

TF 928 is a very useful instrument in f.m. stereo work because, with minor changes to the response of the l.f. amplifier, it can be used for channel separation, frequency response and distortion measurements. The overall distortion of a TF 928 as a demodulator is typically less than 0·2% total. The distortion of the modified TF 995A/2 was also 0·2% total measured with 1 kc/s modulation frequency and ±75 kc/s deviation at 100 Mc/s carrier frequency rising to 1% at 38 kc/s modulation frequency.

Use with Stereo Modulation Generators

The instrument used to test the TF 995A/2 was an H. H. Scott Model 830 Multiplex Stereo Generator. This unit provides a maximum of 5 V r.m.s. composite stereo signal which is used to externally modulate the f.m. signal generator. However, the TF 995A/2 requires 25 V r.m.s. to produce full-scale deviation (±25 or ±75 kc/s) over the r.f. range of the instrument when the DEVIATION switch is in the NORMAL position and 6·25 V r.m.s. on the 54 to 108 Mc/s range with the switch in the HIGH position. In the HIGH position the DEVIATION switch bypasses the network which maintains constant deviation as different frequency multipliers are brought into circuit. To increase the deviation sensitivity of the generator, the switch was re-wired to bypass more of the tracking network.

With this modification, the deviation sensitivity becomes 3·5 V r.m.s. for ±75 kc/s deviation at 100 Mc/s in the HIGH position. In the NORMAL position, the instrument performs as a standard TF 995A/2.

REFERENCE 1. F.C.C. Docket 13506, April 1961.

Acknowledgements

The co-operation of H. H. Scott Co. of Maynard, Massachusetts, U.S.A., particularly Mr. D. R. Von Recklinghausen, their Chief Research Engineer, is acknowledged with thanks.

MARCONI INSTRUMENTS

621.317.761

COUNTER/FREQUENCY METER

TYPE TF 1417/2

by R. I. OSTLER, B.Sc.(Eng.)

A NUMBER of improvements have been incorporated into Counter type TF 1417 since it was described in *Marconi Instrumentation*[1] and it now has the number TF 1417/2. The specification is unchanged except in the details listed below.

External Standard Frequency Input

An amplifier has been added which enables a standard frequency of either 100 kc/s or 1 Mc/s at a level between 100 mV and 5 V to be used. The signal is applied to a BNC socket at the rear of the instrument and the input impedance is approximately 1 kΩ. The amplifier includes a ×10 multiplier which produces a 1 Mc/s output from a 100 kc/s input or passes directly a 1 Mc/s standard input. An INTERNAL/EXTERNAL STANDARD switch on the rear panel of the instrument allows either the internal oscillator to function and drive the 10 Mc/s multiplier or switches off the internal oscillator and connects the amplifier output to the 10 Mc/s multiplier.

100 kc/s Time Base Output

With Counter TF 1417 the time base output selected by the FREQ-TIME switch is available at a socket on the rear panel of the instrument. It has been found useful, however, to have the 100 kc/s time base output permanently available and this facility has been added to TF 1417/2. The output, which is available at a rear panel socket, is of approximately 4 V amplitude and at a source impedance of approximately 5 kΩ.

Extended Gating Period

The gating period may be extended to any desired multiple of 10 seconds, allowing greater accuracy to be obtained in the measurement of low frequencies. To use this facility the counter should be set up for frequency measurement in the normal way, with the gating period set to 10 seconds. Having allowed the count to start, the GATE switch should be moved from AUTO to OPEN during the first 10 seconds. The count should then be allowed to proceed until 10 seconds before the end of the chosen gating period and during this last 10 seconds the GATE switch should be moved back to AUTO. At the end of this 10 second period, the gate will close in the normal way.

Trigger Level Monitor

When the input signal is either noisy or varying in amplitude it is important that the trigger level control be set correctly. The optimum setting, for a sine wave input, is when the Schmitt trigger in the input amplifier is operating on the 'zero' part of the waveform and this results in a square wave at the amplifier output. Varying the trigger level will correspondingly vary the output mark/space ratio. This property of the amplifier is used in the trigger level monitor incorporated in TF 1417/2. The mean level of the amplifier output is obtained via a simple low pass filter and is presented to two level comparators, one of which is set a little below half the supply potential to the amplifier and the other is set to a little above half the supply potential. An 'and' gate allows the monitor lamp, on the front panel, to be switched on if the mean level of the signal from the amplifier lies between the levels of the two comparators. The lamp therefore lights when the amplifier output is approximately square, corresponding to triggering at the zero crossing points of the sine wave.

An incidental feature of the circuit which is found to be very useful in practice is that the angular range of the TRIGGER LEVEL control over which the lamp is lit is a function of the incoming signal level. If the input signal is low the setting of the trigger level to light the lamp will be critical, and conversely a high input signal will light the lamp over practically the whole range of the TRIGGER LEVEL control.

The trigger level monitor is effective for inputs in the frequency range 20 c/s to 10·5 Mc/s.

In addition to the modifications listed above TF 1417/2 is fitted with a mains filter unit to attenuate mains-borne interference which may be present in such places as transmitter stations.

REFERENCE

1. Bisset, D. W. 'Counter/Frequency Meter Type TF 1417'. *Marconi Instrumentation*, December 1961, **8**, p.80.

MARCONI INSTRUMENTS

621.317.761

Frequency Converter

FOR THE RANGE 10 Mc/s to 510 Mc/s . . . *TYPE TF 2400 & TM 7164*

by G. PETERS, B.Sc.

A transistorized frequency converter is described which extends the frequency range of the Counter/Frequency Meter, TF 1417, from 10 Mc/s up to 510 Mc/s. The instrument is in two sections: the 10–110 Mc/s Converter, TF 2400, with a sensitivity of 10 mV; and the 100–500 Mc/s Converter, TM 7164, which is an optional accessory fitting into the TF 2400 case and giving a sensitivity of 100 mV up to 510 Mc/s.

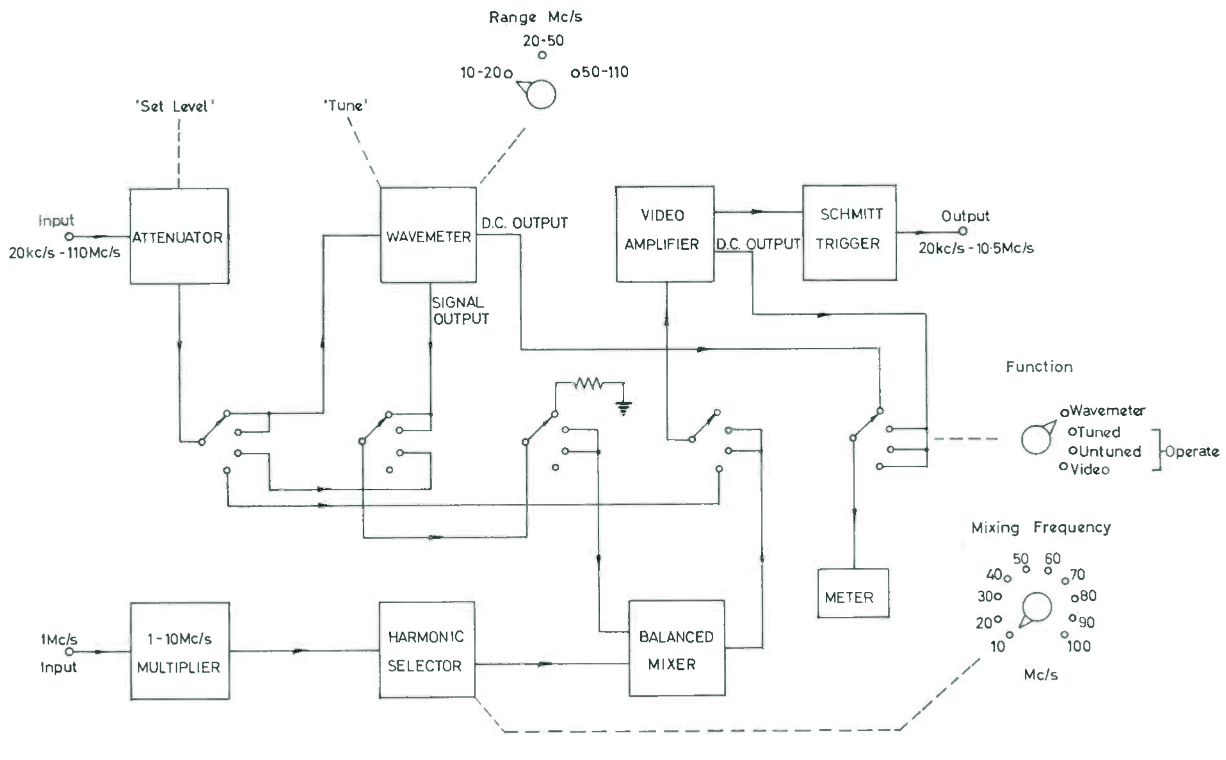
THIS 10–510 Mc/s Converter was designed primarily to extend the frequency range of the Counter/Frequency Meter, TF 1417, up to 510 Mc/s, but it may be used with any similar 10 Mc/s Counter/Frequency Meter which provides a suitable standard 1 Mc/s drive.

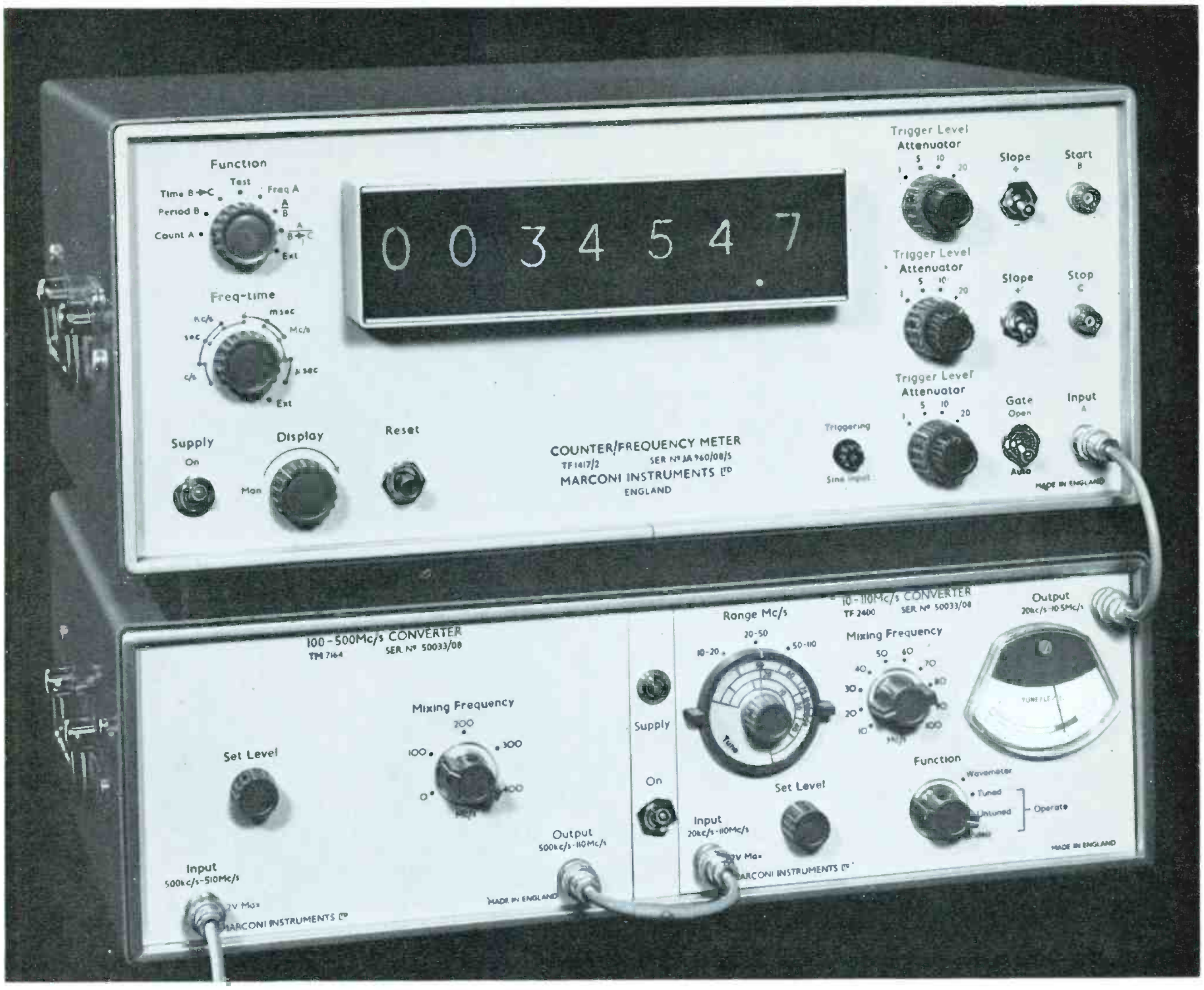
The instrument consists of one basic unit, TF 2400, and an optional unit, TM 7164, which mounts in the same case. TF 2400 extends the frequency range of the counter up to 110 Mc/s with a sensitivity better than 10 mV. It also acts as an amplifier for signals between 20 kc/s and 10·5 Mc/s to improve the counter sensitivity to 10 mV. For normal use the input is untuned, giving simple and rapid operation, but a tuned input stage is provided for use with input signals in the presence of unwanted frequencies. A wavemeter is also provided to give an approximate indication of a completely unknown frequency.

TM 7164 is an optional accessory, used in conjunction with the 10–110 Mc/s Converter to extend the frequency range up to 510 Mc/s with a sensitivity better than 100 mV. Both units can accommodate a wide range of input signal levels.

By virtue of its operating principle, the Converter retains the basic accuracy of the associated counter. The instrument is small, light and simple to use and has very low power consumption. It can be supplied either complete as a 10–510 Mc/s Converter, or the TF 2400 can be supplied alone giving frequency cover up to 110 Mc/s. In the latter case the TM 7164 can subsequently be simply fitted by the user if required.

Fig. 1. Block diagram of 10-110 Mc/s Converter type TF 2400





The Frequency Converter mounts conveniently beneath Counter/Frequency Meter type TF 1417/2 to form a compact assembly

Operating Principle

The converter makes use of the familiar heterodyne principle. In the 10–110 Mc/s Converter a 1 Mc/s standard signal from the Counter/Frequency Meter is multiplied to 10 Mc/s, which produces harmonics of 10 Mc/s up to 100 Mc/s. The signal to be measured is mixed with the nearest harmonic of 10 Mc/s below it to produce a difference-frequency signal in the range of the counter. The correct input frequency is given by the counter reading added to the harmonic of 10 Mc/s used, which is indicated by a switch on the TF 2400 front panel. An approximate indication of the frequency of an unknown signal is given by the wavemeter to facilitate selection of the correct harmonic for mixing.

In the 100–500 Mc/s Converter, TM 7164, the 1 Mc/s signal from the counter is multiplied to 100 Mc/s, and harmonics of 100 Mc/s up to 400 Mc/s are generated. The incoming signal is mixed with the appropriate harmonic below it to produce a difference-frequency signal in the range 500 kc/s to 110 Mc/s. This signal is in the range of the TF 2400 and is measured by using the 10–110 Mc/s Converter in conjunction with the counter. The input frequency is given by the counter reading added to a multiple of 10 Mc/s indicated on the TF 2400 and a multiple of 100 Mc/s indicated by a switch on the TM 7164 front panel.

Description of 10–110 Mc/s Converter TF 2400

Fig. 1 is a simplified block schematic diagram demonstrating the four operating modes, selected by the FUNCTION switch. In all cases the input signal is first taken to a 50 Ω continuously variable SET LEVEL attenuator.

The VIDEO setting is used for signals between 20 kc/s and 10·5 Mc/s. The signal from the attenuator is taken to the video amplifier, and the amplified signal drives a Schmitt trigger circuit. This provides a rectangular waveform of approximately 2 V amplitude to drive the counter and prevents it being operated by small spurious signals.

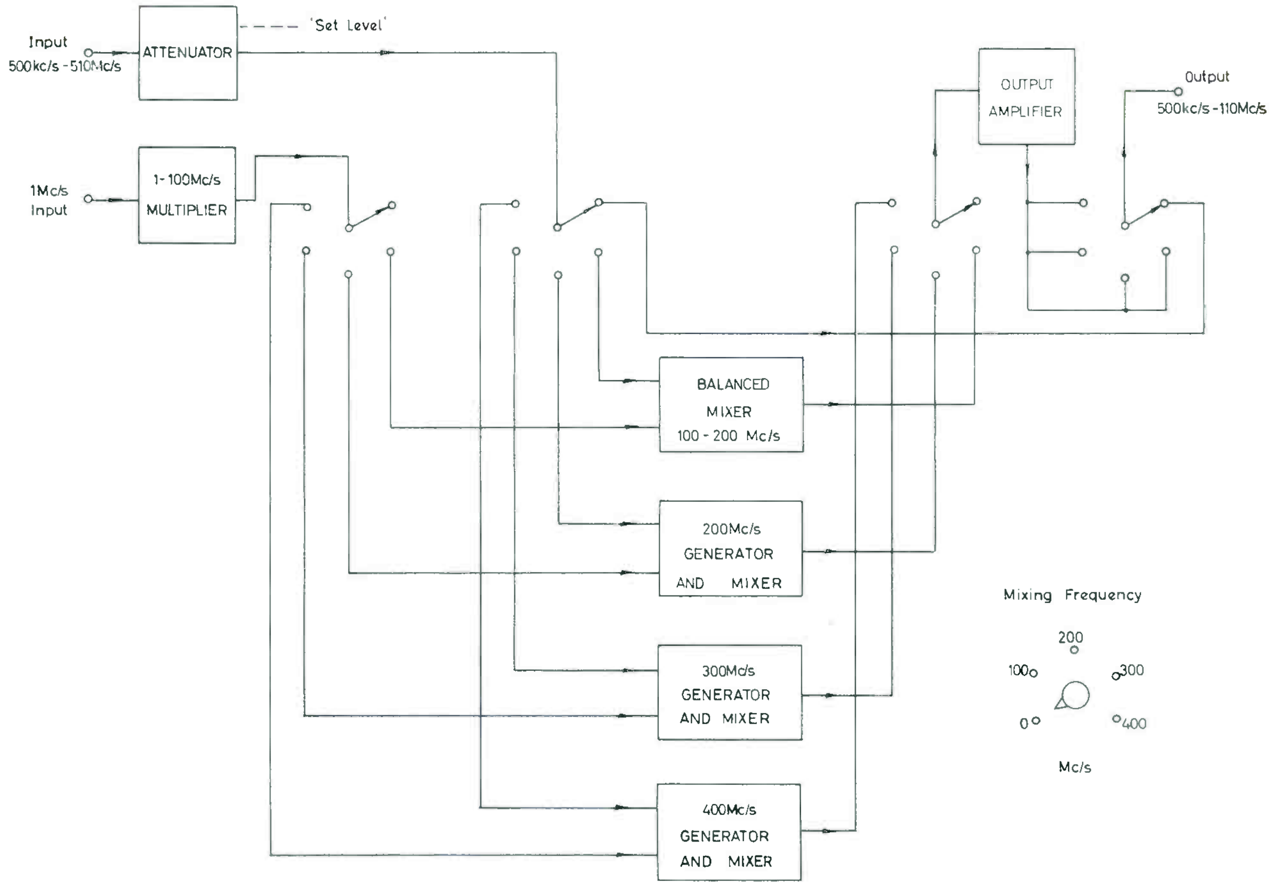


Fig. 2. Block diagram of 100-500 Mc/s Converter type TM 7164

A rectified output from the video amplifier is taken to the meter which, when indicating in the green arc, shows there is sufficient signal to correctly operate the trigger circuit and hence the counter. When using the instrument on any operating range the SET LEVEL control should be adjusted to give a meter deflection in the green arc. The RANGE MC/S switch, the MIXING FREQUENCY switch, and the TUNE dial are all inoperative on this setting of the FUNCTION switch and the input signal frequency is displayed directly on the counter.

The WAVEMETER setting is not an operating position in which accurate measurements of frequency can be made, but is used to give an approximate indication of the frequency of an unknown signal between 10 Mc/s and 110 Mc/s. The signal from the attenuator is taken to a tuned amplifier with three ranges of 10–20 Mc/s, 20–50 Mc/s and 50–110 Mc/s, selected by the RANGE MC/S switch. Concentric with this switch is the TUNE dial, calibrated at intervals of 10 Mc/s, which gives continuous tuning over each range. An output signal from the amplifier is rectified to provide drive to the meter. An approximate determination of an unknown frequency is made by switching to each of the three ranges in turn and rotating the dial until maximum meter deflection is obtained with the cursor in the calibrated section of the dial. The cursor now indicates the correct setting for the MIXING FREQUENCY switch.

The UNTUNED setting is the one which would normally be used with reasonably pure input signals in the frequency range 10·1 Mc/s to 110 Mc/s. After passing through the attenuator the input signal is switched to one input of the balanced mixer. The 1 Mc/s standard frequency from the counter is multiplied in the 1–10 Mc/s Multiplier to produce pulses with a 10 Mc/s repetition rate. In the harmonic selector one of the harmonics of 10 Mc/s between 10 Mc/s and 100 Mc/s is extracted from these pulses by means of tuned circuits selected by the MIXING FREQUENCY switch. The harmonic is taken to the second input of the mixer and is of such order that when it is mixed with the signal the difference frequency produced is between 100 kc/s and 10·5 Mc/s. This difference-frequency signal is amplified and fed to the counter as described for the VIDEO setting, the meter again monitoring the amplifier output. The method of operation on this setting is very simple as only the MIXING FREQUENCY switch has to be set to the correct frequency, the tuning controls being inoperative. The input frequency is given by the reading displayed on the counter added to the multiple of 10 Mc/s indicated by the MIXING FREQUENCY switch.

The TUNED setting is to be preferred when measuring input signals in the presence of unwanted frequencies. A signal in the range 10·1 to 110 Mc/s is taken to the Wavemeter tuned amplifier and thence to the balanced

ABRIDGED SPECIFICATION

TF 2400

Input Frequency Range
20 kc/s to 110 Mc/s.
Input Voltage Range
10 mV to 2 V r.m.s. into 50 Ω.
Output Voltage
2 V rectangular $\pm\frac{1}{2}$ V.
Output Impedance
100 Ω in series with 0·1 μF approx.

Standard Frequency Input
FREQUENCY: 1 Mc/s.
LEVEL: 250 mV to 5 V r.m.s.
INPUT IMPEDANCE: 4 kΩ min. shunted by 50 pF approx.

TM 7164 with TF 2400

Input Frequency Range
100·5 Mc/s to 510 Mc/s.
Input Voltage Range
100 mV to 2 V r.m.s. into 50 Ω.

Standard Frequency Input
FREQUENCY: 1 Mc/s.
LEVEL: 250 mV to 5 V r.m.s.
INPUT IMPEDANCE: 2 kΩ shunted by 100 pF approx.

mixer where it mixes with a harmonic of 10 Mc/s, as for the UNTUNED setting. The harmonic is again selected to give a difference-frequency output in the range 100 kc/s to 10·5 Mc/s which is amplified and fed to the counter. To use the instrument on this setting the input amplifier is tuned, using the TUNE dial in conjunction with the RANGE MC/S switch, for maximum deflection on the meter, which again indicates video amplifier output. The MIXING FREQUENCY switch is set to the appropriate harmonic. Again, the input-signal frequency is given by the counter reading added to the harmonic of 10 Mc/s used.

Description of 100–500 Mc/s Converter

Fig. 2 is a simplified block schematic diagram of the 100–500 Mc/s Converter. The TM 7164 has five frequency ranges, selected by the MIXING FREQUENCY switch. In all cases the signal is first taken to a 50 Ω continuously variable SET LEVEL attenuator.

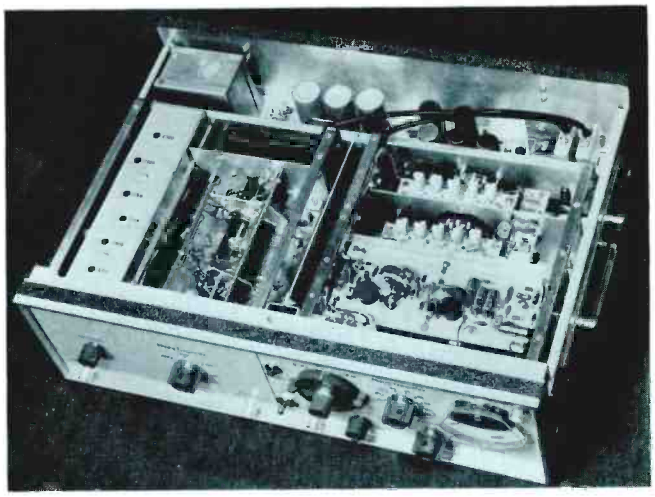
On the '0' setting of the switch the signal is taken from the attenuator direct to the output socket for connection to the TF 2400 input. This setting is provided to remove the need for changing the input lead from the TM 7164 to the TF 2400 input socket, particularly when measuring frequencies on either side of 100 Mc/s.

On the '100 Mc/s' setting, the input signal, in the frequency range 100·5 Mc/s to 210 Mc/s, is taken to one input of the 100–200 Mc/s balanced mixer. The 1 Mc/s standard frequency from the counter is multiplied in the 1–100 Mc/s Multiplier to produce a 100 Mc/s waveform which forms the second input to the mixer. These two inputs produce a difference-frequency output from the mixer in the range 500 kc/s to 110 Mc/s and this signal is amplified and delivered at low impedance to the output socket for measurement on the TF 2400. Thus the input frequency is obtained by adding 100 Mc/s to the frequency measured on the TF 2400.

On the '200 Mc/s' setting the input signal, in the range 200·5 Mc/s to 310 Mc/s, is taken to one input of the 200–300 Mc/s mixer. The 100 Mc/s signal from the multiplier is multiplied by two in a harmonic generator and the resultant 200 Mc/s signal is taken to the second input of the mixer to be mixed with the input signal. Again the difference-frequency output is in the range 500 kc/s to 110 Mc/s and is processed as on the 100 Mc/s range. The input frequency is given by adding 200 Mc/s to the frequency measured by the TF 2400.

The '300 Mc/s' and '400 Mc/s' ranges are exactly similar to the '200 Mc/s' range except that the harmonic inputs to the mixers are respectively 300 Mc/s and 400 Mc/s, produced by individual harmonic generators from the 100 Mc/s signal. The frequency range thus extends up to 510 Mc/s.

When measuring the output from TM 7164 on TF 2400 the latter would normally be switched to the VIDEO or UNTUNED positions depending on the difference-frequency output from TM 7164. If the input is not a clean signal the TUNED position may be used. The WAVEMETER may be used to indicate the approximate value of the difference-frequency output from TM 7164 to facilitate correct adjustment of TF 2400.



Interior view of the complete Converter showing the planned sectional layout

Construction

The instrument is styled to match the TF 1417 in a case similar in shape but only $5\frac{1}{4}$ inches high. Along the rear of the chassis lies the power supply, with the two converter units fitting side by side in front. The units are not plug-in as, once fitted, they would not normally need to be removed, but they are easily removable from the main case by undoing two screws at the rear of each unit and withdrawing the unit from the front panel. When the TF 2400 is supplied alone the left-hand section of the case is covered by a blank panel.

Connection from the power supply to each unit is by means of a flying lead attached to the unit. These leads are terminated by shielded plugs fitting into sockets in the power supply. The 1 Mc/s standard frequency signal is brought into the instrument at the back panel and is taken to the two units by means of coaxial leads. The use of flying leads for signal and power supplies permits easy removal of either unit.

All other signal inputs and outputs are taken to front-panel sockets, and connection between the two converter units is by means of a short jumper lead supplied with the TM 7164.

Most of the circuitry of the instrument is on printed boards, arranged so that all preset controls are adjustable with the units connected in the chassis. In the TM 7164, because of the frequencies involved, it is necessary to keep all wiring as short as possible. To achieve this, separate mixer printed boards for each range and the output amplifier board are all arranged radially around the MIXING FREQUENCY switch.

Summaries of Articles appearing in this issue

RESUME D'ARTICLES PUBLIES DANS LE PRESENT NUMERO

MESURE DES DISTORSIONS DANS LES AMPLIFICATEURS BF

L'exposé comprend:

L'analyse des trois sortes de distorsion qui surviennent dans les amplificateurs basse fréquence: distorsion non linéaire, distorsion linéaire et distorsion de fréquence. Leurs causes et leurs effets sont également analysés.

Une étude critique des analyseurs montre l'intérêt des analyseurs à hétérodyne pour les mesures basse fréquence.

Le principe de fonctionnement de l'analyseur type TF 2330 qui comporte un oscillateur de battements accordé et synchronisé.

Des suggestions en matière de méthodes de mesure relatives aux trois sortes de distorsion lorsqu'on utilise le TF 2330. Page 79

MESURES DU TAUX DE DISTORSION

Un distorsiomètre est un appareil simple qui permet d'évaluer rapidement le contenu harmonique total et le bruit que présentent les signaux basse-fréquence.

Après un exposé comparant les différentes méthodes de calcul du taux de distorsion, l'article comporte une étude critique relative à la conception d'un appareil capable de mesurer des taux de distorsion aussi faibles que 0,05%. Page 86

MESURE DU RAPPORT SIGNAL/BRUIT AU MOYEN D'UN ANALYSEUR

Description d'une méthode de mesure utilisant un récepteur et un générateur classiques, associés à un analyseur, permettant d'effectuer des mesures d'atténuation sur un signal dont le niveau est inférieur de 20 ou 30 dB à 1 µV, signal qui serait normalement masqué par la présence du bruit. Page 90

VERSIONS SPECIALES DE L'OSCILLOSCOPE, TYPE TF 2200

De manière à l'adapter à des applications particulières, cinq versions de l'oscilloscope 35 MHz, type TF 2200, ont été produites.

Les perfectionnements complémentaires proposés comportent: l'essai en impulsions 'sinus²' et en mire des équipements de télévision; le déclenchement coup par coup de la base de temps retardée; l'adjonction d'une commande supplémentaire permettant un réglage précis du retard, lorsqu'on utilise le mode 'porte' de la base de temps retardée; la possibilité d'utiliser les modes de déclenchement extérieurs les plus divers pour la base de temps retardée et de façon indépendante de la base de temps principale. Page 91

VERSION FM 'STEREO' DU GENERATEUR FM/AM, TYPE TF 995A/2

Description du générateur à modulation de fréquence ou d'amplitude TF 995A/2, modifié, qui, associé à un générateur de modulation 'stéréo', fournit un signal 'stéréo' en modulation de fréquence. Les caractéristiques correspondent aux spécifications de l' 'U.S. Federal communications commission' (F.C.C.) relatives à un émetteur à modulation de fréquence utilisant le procédé stéréo 'G.E.-Zenith'. Page 95

CONVERTISSEUR DE FREQUENCE POUR LA GAMME 10 MHz à 510 MHz, TF 2400/TM 7164

Description d'un convertisseur de fréquence transistorisé, qui permet d'étendre la gamme de mesure du compteur-fréquencemètre TF 1417 de 10 MHz jusqu'à 510 MHz. Cet appareil comprend deux parties: le convertisseur 10-110 MHz, type TF 2400, qui présente une sensibilité de 10 mV et le convertisseur 100-500 MHz, type TF 7164, constitue un complément livrable sur commande, qui se place dans le coffret du convertisseur TF 2400. Sa sensibilité est de 100 mV jusqu'à 510 MHz. Page 98

ZUSAMMENFASSUNG DER IN DIESER NUMMER ERSCHEINENDEN BEITRÄGE

VERZERRUNGSMESSUNGEN AN TONFREQUENZ-VERSTÄRKERN

Die drei Hauptarten von Verzerrungen (nichtlineare Verzerrungen, Dämpfungsverzerrung und Phasenverzerrung), die in Tonfrequenzverstärkern auftreten können, sowie deren Ursachen und Wirkungen werden in diesem Aufsatz beschrieben. Eine Betrachtung über Frequenzanalysatoren zeigt, daß das Überlagerungsprinzip bei Tonfrequenzmessungen ratsam ist. Die Arbeitsweise des Frequenzanalysators TF 2330, der einen im Gleichlauf abgestimmten Schwebungsoszillator besitzt, wird beschrieben. Mit dem Gerät TF 2330 durchführbare Meßmethoden für alle drei Arten von Verzerrungen werden vorgeschlagen.

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KLIRRFAKTORMESSUNGEN

Ein Klirrfaktormeßgerät ist ein einfaches Gerät zur schnellen Beurteilung des Gesamtanteils von Oberwellen und Rauschen in einem Tonfrequenzsignal. Im Anschluß an eine Betrachtung und einen Vergleich der verschiedenen Methoden zur Berechnung des Klirrfaktors wird in dem Aufsatz der Entwurf eines Gerätes beschrieben, welches Klirrfaktoren bis herunter auf 0,05% messen kann.

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FREQUENZANALYSATOR HILFT BEI DER VERBESSERUNG DES RAUSCHVERHÄLTNISSES

Es wird eine Meßmethode beschrieben, bei der ein Universalempfänger und Meßsender in Verbindung mit einem Frequenzanalysator zu Dämpfungsmessungen benutzt werden kann, bei denen der Signalpegel 20 oder 30 dB unter 1 µV liegt und normalerweise von vorhandenem Rauschen verdeckt wird.

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SPEZIALAUSFÜHRUNGEN DES OSZILLOGRAPHEN TF 2200

5 abgeänderte Ausführungen des 35-MHz-Meßoszillographen TF 2200 wurden für besondere Zwecke gebaut. Die zusätzlichen Einrichtungen umfassen die sin²-Impuls- und Balkenmusterprüfungen von Fernseheinrichtungen, die Einzelablenkauslösung der Vorlauf-Zeitachse, zusätzliche Feineinstellung für die Vorlauf-Zeitachse in torgeschalteter Betriebsart und eine vielseitige Fremdauslösung der Vorlauf-Zeitachse unabhängig von der Hauptzeitablenkung.

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EINE FM-STEREO-AUSFÜHRUNG DES FM/AM MEßSENDERS TF 995A/2

Es wird der abgeänderte FM/AM-Meßsender TF 995A/2 beschrieben, der in Verbindung mit einer Stereomodulationsquelle ein FM-Stereo-Signal liefert. Die Bedingungen der US-Federal Communications Commission (FCC) für FM-Sender mit GE-Zenith FM-Stereomodulation werden erfüllt.

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FREQUENZWANDLER TF 2400/TM 7164 FÜR DEN BEREICH 10 BIS 510 MHz

Es wird ein transistorisierter Frequenzwandler beschrieben, der den Frequenzbereich des Zähl- und Frequenzmeßgerätes TF 1417 von 10 MHz auf 510 MHz erweitert. Der Wandler besteht aus zwei Teilen, dem 10-110 MHz Frequenzwandler TF 2400 mit einer Empfindlichkeit von 10 mV und dem 100-500 MHz Frequenzwandler TM 7164, der ein in das Gehäuse TF 2400 einsetzbarer, getrennt lieferbarer Zusatz ist und eine Empfindlichkeit von 100 mV bei Frequenzen bis zu 510 MHz besitzt.

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SOMMARIO DEGLI ARTICOLI PUBBLICATI IN QUESTO NUMERO

MISURE DI DISTORSIONE IN AMPLIFICATORI AUDIO

Si descrivono, considerandone anche le cause e gli effetti, i tre tipi principali di distorsione che possono verificarsi in amplificatori audio—distorsione non-lineare, distorsione di frequenza e distorsione di fase. Una discussione sugli analizzatori d'onda indica come il tipo ad eterodina sia adatto per misure audio. Viene descritto il principio di funzionamento dell'analizzatore d'onda TF 2330, il quale comprende un oscillatore a battimenti con accordo sincrono. Si suggeriscono metodi di misura per tutti e tre i tipi di distorsione, impiegando il TF 2330.

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MISURE DEL FATTORE DI DISTORSIONE

Un misuratore del fattore di distorsione è un semplice strumento per la rapida valutazione del contenuto totale di armoniche e rumore di segnali audio. Dopo avere considerato e confrontato vari metodi per il calcolo del fattore di distorsione, l'articolo discute il progetto di uno strumento capace di misurare distorsione di anche soltanto 0,05%.

Pagina 86

MIGLIORAMENTO DEL RAPPORTO SEGNALE/DISTURBO MEDIANTE UN ANALIZZATORE D'ONDA

Si descrive un metodo di misura nel quale un ricevitore per usi generali ed un generatore di segnali possono essere impiegati in unione ad un analizzatore d'onda per effettuare misure di attenuazione ove il segnale abbia un livello di 20 o 30 dB al di sotto di 1 µV, così da essere normalmente mascherato dalla presenza di rumore.

Pagina 90

VERSIONI SPECIALI DELL'OSCILLOSCOPIO TIPO TF 2200

Cinque versioni modificate dell'oscilloscopio di misura da 35 MHz, tipo TF 2200, sono state prodotte per convenire ad applicazioni speciali. Le prestazioni addizionali offerte comprendono mezzi per il collaudo di apparecchiature televisive con impulsi a seno² e barra, sganciamento a scansione singola della base dei tempi di ritardo, comando aggiuntivo di ritardo fine per impiego con la base dei tempi di ritardo nel modo a sblocco (gated) e versatili possibilitá di sganciamento esterno della base dei tempi di ritardo indipendentemente dalla base dei tempi principale.

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UNA VERSIONE DEL GENERATORE DI SEGNALI A MF/MA TF 995A/2 PER STEREO A MOD. DI FREQ.

Si descrive un generatore di segnali a MF/MA TF 995A/2 modificato, il quale, in unione ad un generatore di modulazione stereo, fornisce un segnale stereo a mod. di freq. Ciò risponde alle esigenze della Commissione Federale Telecomunicazioni degli Stati Uniti (F.C.C.) per un trasmettitore a MF impiegante il sistema stereo a MF GE-Zenith.

Pagina 95

CONVERTITORE DI FREQUENZA TF 2400/TM 7164 PER IL CAMPO DA 10 MHz A 510 MHz

Si descrive un convertitore di frequenza transistorizzato che estende il campo di misura del contatore/frequenzimetro TF 1417 da 10 MHz fino a 510 MHz. Lo strumento si compone di due sezioni: il convertitore per la gamma 10-110 MHz, TF 2400, con una sensibilitá di 10 mV; ed il convertitore per la gamma 100-500 MHz, TM 7164, che è un accessorio facoltativo montabile entro la custodia contenente il TF 2400 e fornente una sensibilità di 100 mV fino a 510 MHz.

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RESUMENES DE ARTICULOS QUE APARECEN EN ESTE NUMERO

MEDIDA DE DISTORSION EN AMPLIFICADORES DE AUDIO

Se describen las tres clases principales de distorsión que pueden tener lugar en amplificadores de audio—no lineal, de frecuencia, y de fase—así como sus causas y efectos. Después de comparar los analizadores de onda, se llega a la conveniencia del tipo heterodino para medidas en audio. Se describe el principio de funcionamiento del analizador de onda TF 2330, que comprende un oscilador de frecuencia de batido, con sintonía sincronizada. Se recomiendan procedimientos de medida para las tres clases de distorsión utilizando el TF 2330.
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MEDIDA DEL FACTOR DE DISTORSION

El medidor del factor de distorsión es un sencillo instrumento para un rápido conocimiento del total de armónicos y del contenido de ruidos en las señales de audio. Después de considerar y comparar varios métodos de cálculo del factor de distorsión, el artículo trata del proyecto de un instrumento capaz de medir hasta un 0,05% de distorsión.
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EL ANALIZADOR DE ONDA MEJORA LA RELACION SEÑAL/RUIDO

Se describe un procedimiento de medida donde un receptor corriente y un generador de señal pueden utilizarse con un analizador de ondas para efectuar medidas de atenuación, cuando la señal es 20 o 30 dB por debajo de 1 μV, y está normalmente enmascarada por la presencia de ruido.
Página 90

VERSIONES ESPECIALES DEL OSCILOSCOPIO TIPO TF 2200

Se han fabricado cinco versiones modificadas del osciloscopio de 35 MHz, tipo TF 2200, para adaptarlo a aplicaciones especiales. Las nuevas características comprenden: impulsos seno^2 y prueba de impulsos en equipos de televisión; disparo único de base de tiempos retardada; control extra de retardo fino, para utilizarlo con la base de tiempo retardada con impulso de umbral, y disparo externo en varias modalidades adaptables a la base de tiempo retardada, independientemente de la principal.
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VERSION ESTEREO DE F.M. DEL GENERADOR DE SEÑAL DE F.M./A.M., TF 995A/2

Se describe el generador de señal, de F.M./A.M., TF 995A/2 modificado que, en unión de un generador de modulación de estéreo, produce una señal de F.M. estéreo. Esto responde a las exigencias de la Federal Communications Commission (F.C.C.), de Estados Unidos, respecto de transmisores de F.M. que utilizan el sistema estéreo (GE-Zenith).
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CONVERTIDOR DE FRECUENCIA DE 10 A 510 MHz, TF 2400/TM 7164

Se describe un convertidor de frecuencia, con transistores, que amplia el margen de frecuencia del medidor/contador de frecuencia TF 1417, desde 10 hasta 510 MHz. El instrumento consta de dos partes: el convertidor de 10 a 110 MHz, TF 2400, con sensibilidad de 10 mV, y el convertidor de 100 a 500 MHz, TM 7164, que es un accesorio opcional, y puede acoplarse a la caja del TF 2400. Proporciona una sensibilidad de 100 mV a 510 MHz.
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SIGNAL GENERATORS

NOISE GENERATORS

OSCILLATORS

OSCILLOSCOPES

PULSE GENERATORS

MODULATION METERS

DISTORTION ANALYSERS

SPECTRUM ANALYSERS

FREQUENCY COUNTERS AND METERS

POWER METERS

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