

type 5842

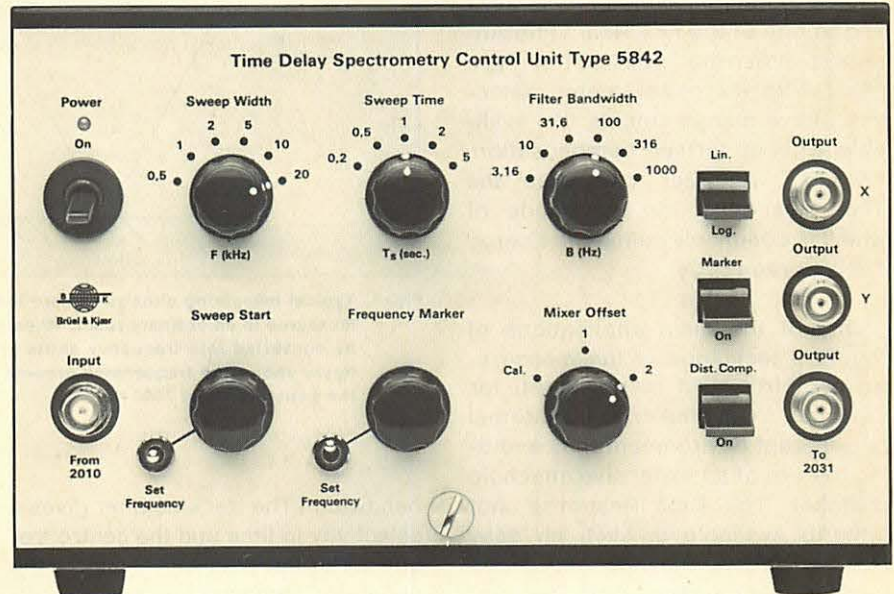
Time Delay Spectrometry Control Unit

USES:

- Free-field measurements in noisy and reflective environments
- Amplitude and phase response of loudspeakers, microphones, hydrophones, telephones, hearing aids etc. in typically 0,5s to 2s
- On site measurements of reflections and sound insulation
- Time response measurements of any electrical, acoustical or mechanical two-port
- Intermodulation and difference frequency distortion of systems incorporating a delay e.g. loudspeakers and taperecorders
- Leak and crack detection in mechanical structures
- Investigation of anechoic rooms

FEATURES:

- Sweeps Heterodyne Analyzer 2010 linearly through any frequency range between 0 Hz and 200 kHz
- Continuous sweep for use with display units
- Choice of lin. or log. frequency axis for display and recording
- Fast logarithmic detector provides DC output of magnitude response
- Reference signal for selective measurements of phase using Phase Meter 2971



When used in conjunction with the Heterodyne Analyzer Type 2010 and the Distortion Measurement Control Unit Type 1902, the **Time Delay Spectrometry Control Unit Type 5842** enables fast, time selective measurements of both magnitude and phase of the frequency response. The phase is available with the addition of the Phase Meter Type 2971 to the system. On addition of one of the FFT Analyzers Type 2031 or 2033 the magnitude of the time response (commonly known as the Energy Time Curve denoted by ETC) is calculated and displayed on the FFT analyser's screen. The frequency responses are monitored on a variable persistence screen. All the responses are measured in a few seconds and can easily

be traced out via an X-Y Recorder Type 2308. Time delay spectrometry is a general technique for measuring the performance of any two-port. The main virtue of this technique is its remarkable capability of rejecting noise and reflections with short measuring times. Time delay spectrometry has a vast number of applications in both development and quality control work. A typical application is the free-field measurement of loudspeakers in a reflective environment. Other applications are found in building acoustics and underwater acoustics. The instruments constituting the rest of the system retain their full measuring capabilities and can be used on their own for a host of other measurements.

- Output for display of time response on FFT Analyzer 2031 or 2033
- Provision for centring the time range on the FFT Analyzer's screen
- Remote control of Heterodyne Analyzer's filter bandwidth
- Frequency marker for accurate readout of any frequency within the sweep limits
- Automatic compensation for the attenuation due to the inverse square law in time response measurements

Introduction

The Time Delay Spectrometry Control Unit Type 5842 is designed to be used in conjunction with the Heterodyne Analyzer Type 2010 and the Distortion Measurement Control Unit Type 1902. When used with these instruments the 5842 is the control centre for time selective frequency response measurements. Furthermore the 5842 can be connected to the Phase Meter Type 2971 and to one of the FFT Real Time Analyzers either the Type 2031 or Type 2033. With the phase meter, selective phase measurements are available without further compensation. The FFT Analyzer calculates the Time Response, the magnitude of which is commonly called the Energy Time Curve (ETC).

One of the main applications of the TDS technique is the measurement of free-field response of, for example, loudspeakers in a normal reverberant environment thus avoiding the use of an expensive anechoic chamber. The Time Response, not hitherto available in such an easy manner, adds to the versatility of the measurement system. Both frequency and time response are typically available in "real time", that is, new data is produced faster than it can be assimilated by the human mind. Compared to other techniques TDS offers exceptional performance in situations where speed of operation and immunity to background noise is essential.

The TDS Principle

TDS is a general technique for measuring the performance of two-port systems (i.e. systems which have a pair of input terminals and a pair of output terminals).

The driving signal is a linear sine sweep i.e. the instantaneous frequency is linearly related to time. The constant sweep rate, S , links time and frequency together in such a way that a selectivity in time - and consequently in space - can be obtained by filtering in the frequency domain. Similarly a delay can be compensated by a frequency offset. This relationship between time and frequency is the essence of TDS.

The response signal is measured using a tracking filter with a constant

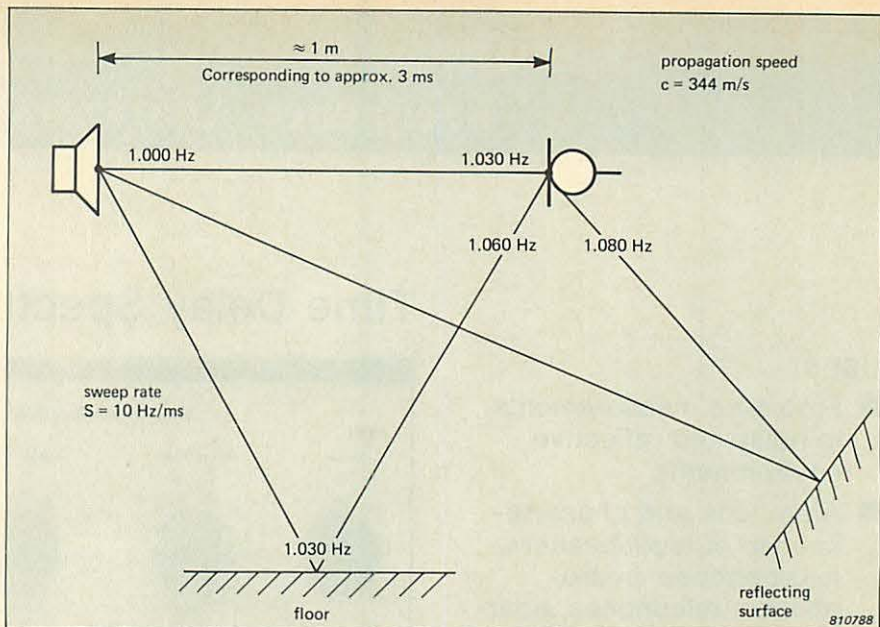


Fig. 1. Typical measuring situation where the free-field response of a loudspeaker has to be measured in an ordinary room. When employing a linear sweep, the various delays will be converted into frequency shifts proportional to the delay and sweep rate, S . The figure shows the frequencies present at the microphone at the moment in time when the generator is at 1000 Hz

bandwidth. The tracking filter gives a selectivity in time and the centre frequency of the filter either tracks the generator frequency or may be offset in frequency in order to compensate for a delay.

The linear sweep causes the sweep rate, S , to be time independent. The sweep rate is given by:

$$S = F/T_s \quad [\text{Hz/s}]$$

where F is the swept frequency range and T_s is the sweep time. As an example consider the typical measuring situation shown in Fig.1, where the free-field response of the loudspeaker has to be measured in an ordinary room (i.e. a reflective environment). In this example the frequency range, F , of 20 kHz is swept in 2 seconds giving a sweep rate, S , of 20 kHz / 2 s = 10 Hz/ms. At a specific moment in time the generator is at, say, 1000 Hz. The spectrum received at the microphone will contain different components corresponding to the various delays present in the set-up i.e. the reflections. For instance, the direct sound from the loudspeaker has travelled 1 m. With a speed of propagation of sound in air of 344 m/s this means that the sound received at the microphone had been trans-

mitted from the loudspeaker approximately 3 ms previously. The TDS Control Unit Type 5842 sweeps the generator from the high to low frequencies so that the direct sound received by the microphone at the moment of time in question is 1000 Hz + $S \cdot 3 \text{ ms} = 1030 \text{ Hz}$. The same simple calculation can be made for the floor reflection shown in Fig.1. In this case the path length of the sound is 2 m which means a frequency shift of 60 Hz yielding a frequency of 1060 Hz at the microphone.

Fig.2 shows the spectrum at the microphone corresponding to three distinct times during the sweep. Time t_1 corresponds to the situation in Fig.1. The relative positions of the frequency components corresponding to the different delays remain unchanged as a function of time although the whole picture is shifted towards the lower frequencies during the sweep. The levels of the components will, however change as a function of time corresponding to the frequency responses of the individual paths.

To recapitulate, in TDS a frequency sweep converts a delay into a frequency shift and, as a linear sweep is used, the conversion ratio is constant and equal to the sweep rate, S .

Terminology

Traditionally the terms frequency response and phase response have been used for the magnitude of the frequency response and the phase of the frequency response respectively. As the time response was not normally dealt with, these terms entailed no ambiguity. However, with TDS the magnitude and phase responses are available in both the time and the frequency domains and therefore the terminology must be more explicit. The terms used for the four responses are: *the magnitude of the frequency response* (traditionally known as the frequency response), *the phase of the frequency response* (traditionally known as the phase response), *the magnitude of the time response* (also known as the Energy Time Curve) and *the phase of the time response*. This terminology is not only unambiguous but also emphasizes the analogy between the time and the frequency domain.

Frequency Response

To measure the frequency response of the direct sound alone, i.e. the free-field response, the microphone signal is passed through a filter which tracks the generator frequency with an offset of 30 Hz corresponding to the frequency shift of the direct sound. Reflections can be excluded from the measurement by using a filter of sufficiently narrow bandwidth (see Fig.2).

The frequency response of any reflection can be measured by changing the offset of the tracking filter. For example by changing the offset to 60 Hz the frequency response of the floor reflection is obtained. In this case it may be necessary to reduce the filter bandwidth, B , even further as the floor reflection is closer to the second reflection than to the direct sound.

Frequency responses measured in this way include the phase responses. Except for the linear phase shift associated with a delay, the phase response is the phase difference between the filtered microphone signal and the generator. In the B & K TDS System, however, the linear phase shift is automatically compensated for when the filter is tuned to the desired component.

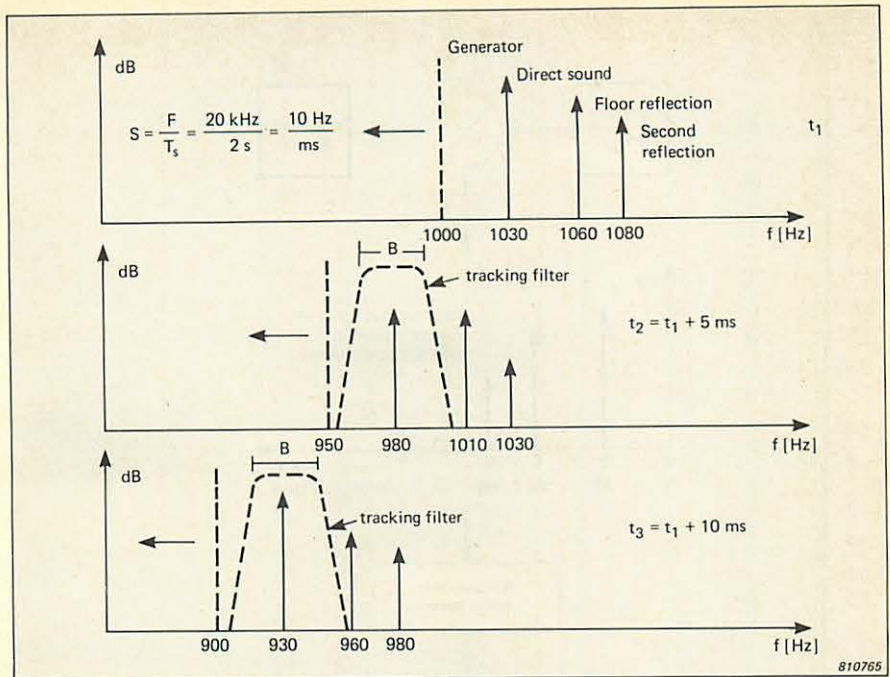


Fig. 2. Spectrum at the microphone corresponding to three distinct times during the sweep. t_1 corresponds to the instantaneous picture in Fig.1. During the sweep the whole picture will be shifted towards the lower frequencies but the relative position of the components remains unchanged. The levels of the individual components will change in accordance with the frequency responses. Here the tracking filter is tuned to measure the direct sound

Time Response

If the microphone signal is mixed with the generator frequency, the whole instantaneous spectrum will be shifted towards DC as shown in Fig.3.a & b. At this stage the processed signal no longer contains information about the original frequencies. The spectra in Fig.2 are aligned with the generator frequency at DC.

The magnitude and phase of the individual components will, as stated above, change in accordance with the frequency responses. This means that amplitude and phase modulations occur which produce sidebands in the spectrum. Therefore if the mixed down microphone spectrum is averaged over time, T_s , the spectrum will be smeared as shown in Fig.3.c. This is the time response corresponding to the swept frequency range, F . The magnitude of the time response is often referred to as the Energy Time Curve (ETC) as it shows the arrival of energy as a function of time.

Analogy between Frequency and Time Response

The magnitude of the time response is completely analogous to the magnitude of the frequency response.

Both responses are normally displayed on a logarithmic amplitude scale i.e. dB scale. The detector used for measuring the magnitude of the frequency response can be considered as an envelope detector. Similarly, in the time domain the magnitude of the time response (the Energy Time Curve) is the true envelope of the impulse response. In contrast to the impulse response itself, the time response is much easier to interpret as the picture is not confused by oscillations at the various peaks in the transfer function. Furthermore, the logarithmic amplitude scale, not applicable to the impulse response, improves the dynamic range visually and exponential decays appear as straight lines.

The phase of the frequency response also has its counterpart in the time domain. The complete time and frequency responses i.e. including the phase and magnitude of each, represent exactly the same information. Both are equally valid descriptions of the linear performance of a two-port. Both representations have their uses. In some cases deviations from "ideal" behaviour will appear much more clearly in one domain than in the other.

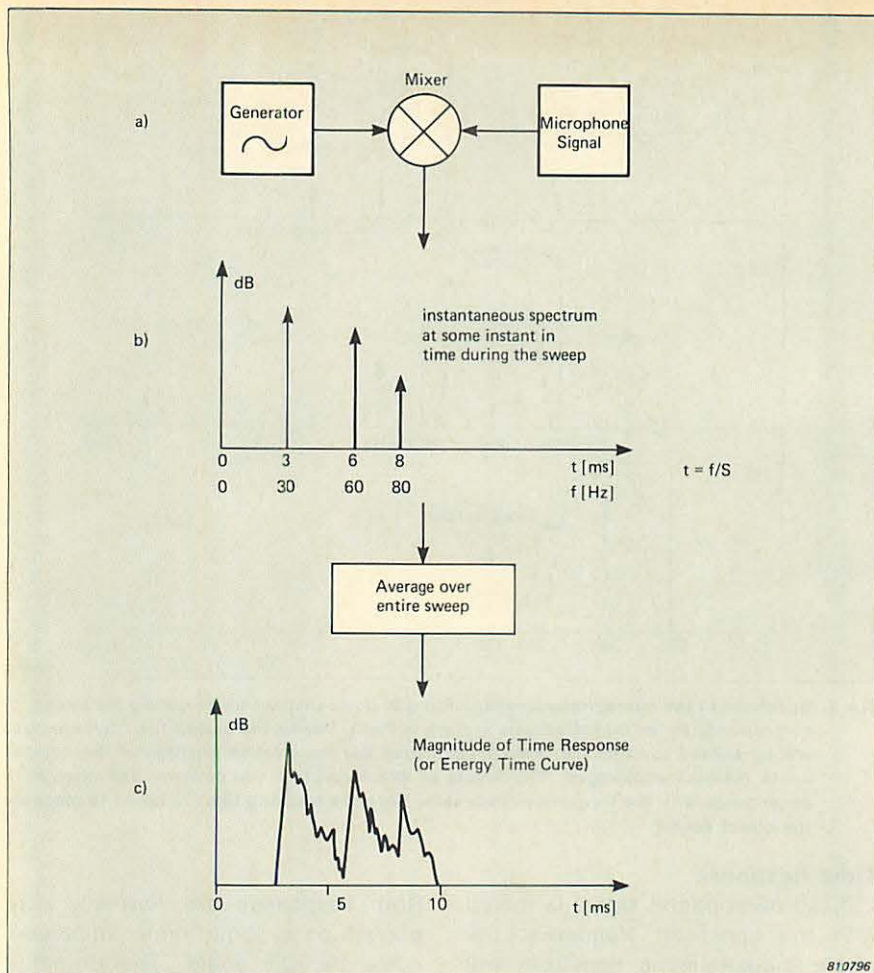


Fig. 3. Signal processing for obtaining the Time Response. If the microphone signal is mixed with the generator signal (a) the instantaneous spectra in Fig.2 will be aligned with the generator frequency at DC (b). As the individual components will be amplitude and phase modulated the spectrum will be smeared when averaged over the entire sweep (c)

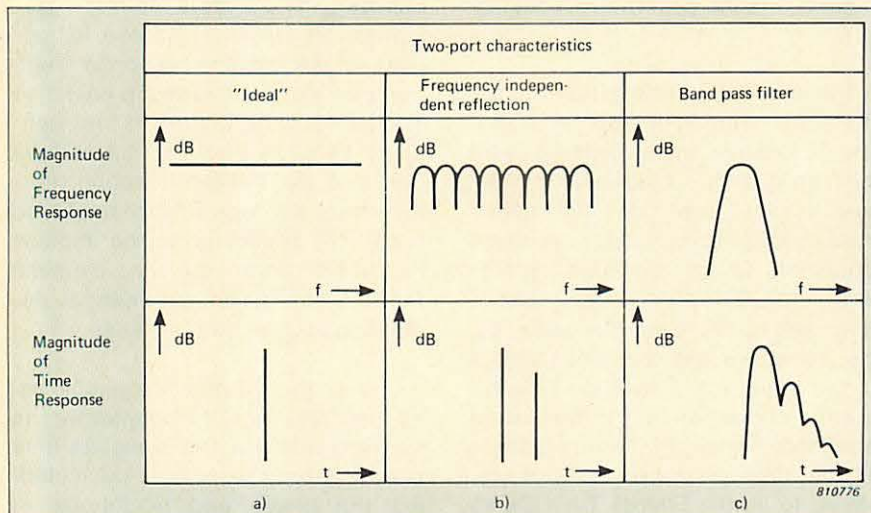


Fig. 4. Frequency and time domain representation of three typical two-ports. An ideal system appears as a straight line in both domains (a). A frequency independent reflexion is most easily recognised in the time domain (b) while a band pass filter is best studied in the frequency domain (c). When the two-port exhibits both frequency and time anomalies both representations will prove useful

Fig.4. shows the magnitude of both frequency and time response for three different two-ports. An "ideal"

two-port i.e. one with a flat frequency response is manifest in the time response as a vertical line (Fig.4.a).

If a frequency independent reflection is introduced into the system, the reflection will appear as another vertical line in the time response but as a comb-filter effect in the frequency response as shown in Fig.4.b. On the other hand, a band-pass filter may appear similar to a $(\sin x)/x$ function in the time domain (Fig.4.c). Clearly the differences in the linear performance should be studied in the domain where they appear more clearly.

Often it is not easy to distinguish between time and frequency domain effects. In the case of loudspeakers, for example, the response is due to the combined responses of the individual driver units, the cross-over network, diffraction and reflection within the enclosure etc. In such cases, where there is a mixture of both time and frequency phenomena, both representations will prove useful.

Relation between Frequency and Time Domains

As the two domains are so closely related it follows that the parameters describing the two representations must also be related. Fig.5 (bottom right) shows the representation in the two domains. The time response shows the magnitude while the frequency response shows both magnitude and phase. Both magnitudes are shown on a logarithmic amplitude scale and the phase is shown on a linear scale. Both frequency and time scales are linear.

Assume that the frequency response covers a range of F Hz with a resolution of Δf Hz and that the time response covers a range of T seconds with a resolution of Δt seconds. Then the frequency resolution, Δf , is given by the time range, T :

$$\Delta f = 1/T$$

where

$$T = B/S = B T_s/F$$

Similarly the time resolution, Δt , is given by:

$$\Delta t = 1/F$$

That is, the resolution in one domain is given by the range in the other domain.

FREQUENCY & TIME RESPONSE NOMOGRAM

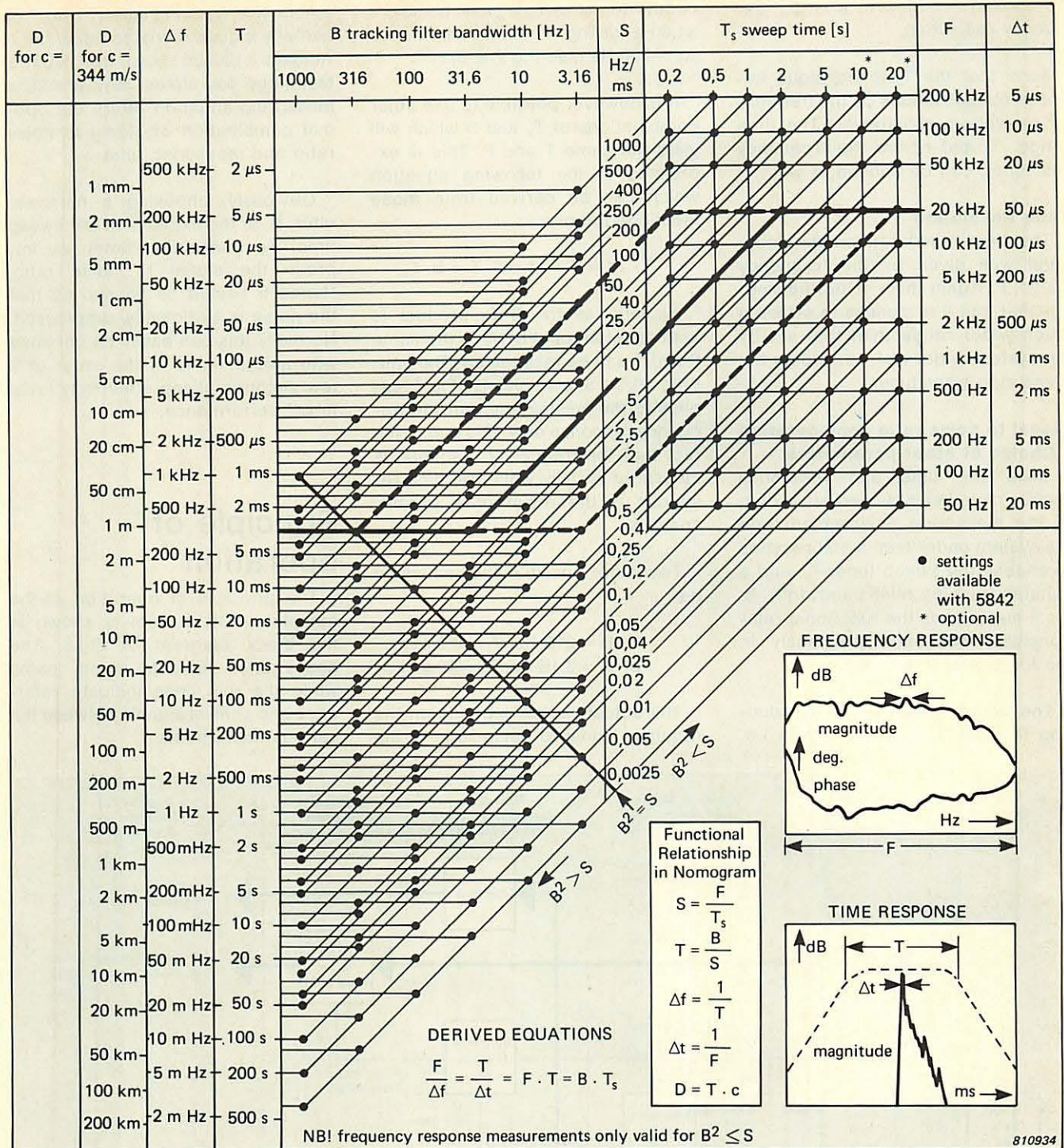


Fig. 5. Choice of measuring parameters. The example used in this introduction is indicated by the thick line. Two other combinations of sweep time, T_s , and tracking filter bandwidth, B , are indicated by dotted lines. All yield the same frequency and time resolution but those with the narrower filters give better signal to noise ratio at the expense of the measuring time

The spatial range, D , is calculated from the time range, T , when the speed of propagation, c , is known:

$$D = T \cdot c$$

A single point in one domain is an average of all the information contained in the other; that is frequency information is lost in the time do-

main and time information is lost in the frequency domain.

Frequency Resolution

As with all other time selective techniques the frequency resolution is restricted by the time range (or time window), T , as mentioned above. The time range can normally be selected within wide limits but

must, however, often be restricted in order to exclude unwanted reflections from the measurements. It is then the measurement environment that determines the resolution. If a large room is available the reflections can be moved away from the direct sound and a wider time range selected. On the other hand if the reflections are close to the direct

sound, a narrower time range must be chosen resulting in a larger frequency resolution.

Note that the TDS technique imposes no restrictions on the frequency resolution whatsoever. The time range, T , and hence the frequency resolution can be chosen at will.

Time Resolution

The time resolution on the other hand was given by the frequency range, F . Again there is no theoretical limit, but it is useless to sweep a much wider range than that set by the system under test - including the measuring transducer.

Signal to noise ratio consideration - Choice of other parameters

Once the time and frequency ranges have been chosen according to the measuring environment and the system under test, it still remains to choose the sweep time, T_s , and a suitable tracking filter bandwidth, B . Fig.5 also shows the functional relationships mentioned previously in the form of a nomogram.

The example used in this introduction is given in the nomogram i.e.

$F = 20 \text{ kHz}$, $T_s = 2 \text{ s}$. A suitable choice for B in this case could be $31,6 \text{ Hz}$ giving $T \sim 3 \text{ ms}$ and hence $\Delta f \sim 300 \text{ Hz}$ (see Fig.2 & 3).

It is however possible to use other combinations of T_s and B which will yield the same T and F . This is expressed in the following equation which can be derived from those mentioned above:

$$F / \Delta f = T / \Delta t = F \cdot T = B \cdot T_s$$

As long as the $B \cdot T_s$ product is kept constant (and $B^2 < S$) the same F and T can be maintained. The limitation $B^2 < S$, also given in Fig.5, applies to the measurement of the frequency response only. It is due to the fact that the frequency response is measured serially while the time response is the result of a parallel analysis.

Two other possible choices would be

$$B = 316 \text{ Hz}, T_s = 0,2 \text{ s}$$

$$\text{or } B = 3,16 \text{ Hz}, T_s = 20 \text{ s.}$$

The actual choice depends on the required signal to noise ratio in the

measurement. As the TDS technique employs a tracking filter, it has inherently a good signal to noise ratio. Actually it can be shown that the TDS technique for almost any practical measuring situation offers the optimal combination of signal to noise ratio and measuring time.

Obviously, choosing a narrower filter, B , at the expense of the sweep time, T_s , (measuring time) will improve the signal to noise ratio. Hence B should be chosen so that the noise is sufficiently suppressed. Normally this can easily be obtained with sweep times of the order of a few seconds giving effectively "real time" performance.

Principle of operation

The principle of operation of the complete TDS system is shown in the block diagram of Fig.6. The heavy lines indicate signal paths while the thin lines indicate reference and control signals between the various instruments.

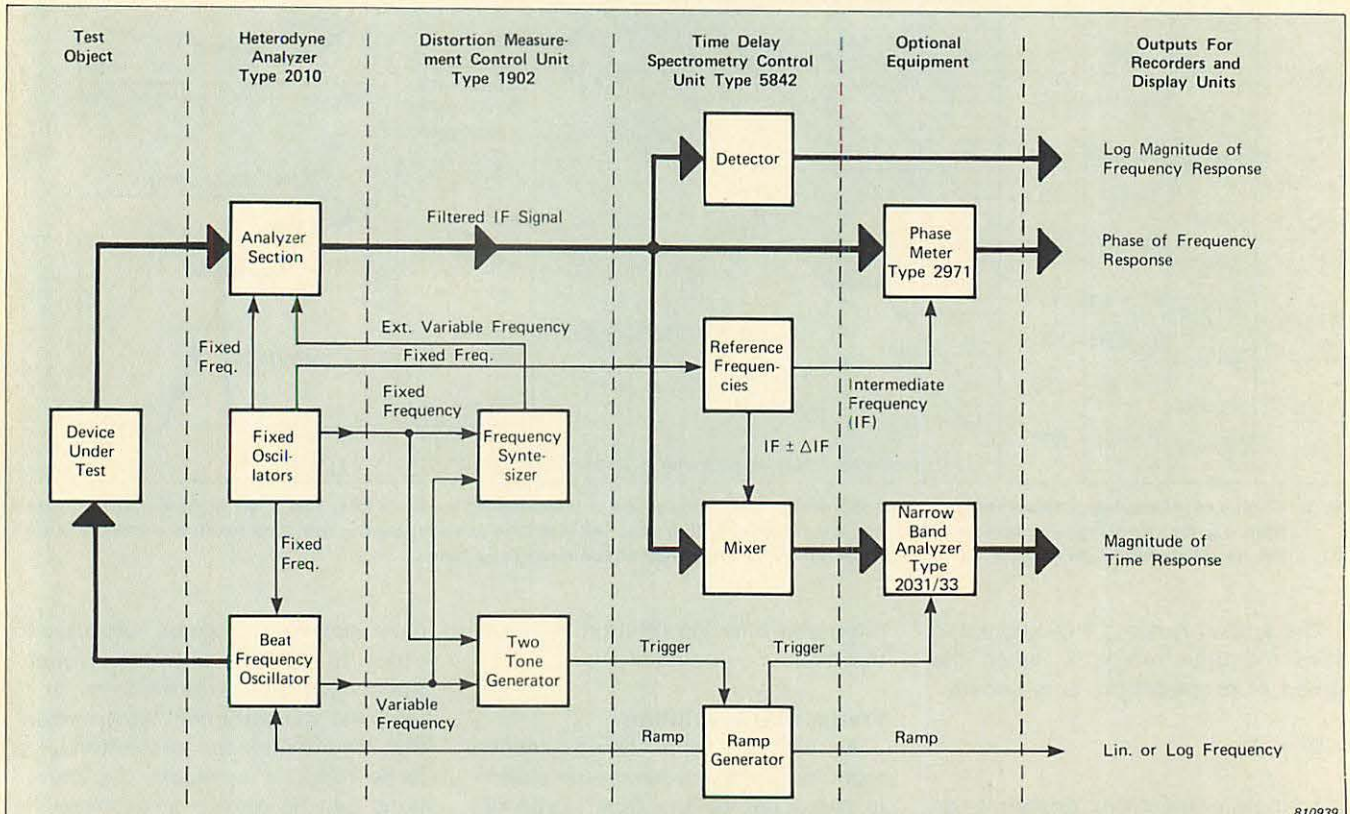


Fig. 6. Block diagram of the complete system. Signal paths are indicated by thick lines while reference and control signals are shown with thin lines. The signal processing in the system is based on the filtered intermediate frequency signal, IF, present at the output of the 2010 Analyzer Section when this is set to its selective mode. All response curves eventually appear as DC outputs for recorders and display units

Signal processing in the Heterodyne Analyzer Type 2010 and Distortion Measurement Control Unit Type 1902

The input for the device under test is taken from the BFO output of the Heterodyne Analyzer Type 2010. The output from the device under test is fed to the input of the Analyser Section of the 2010. Here the signal is mixed with the Ext. Variable Frequency coming from the Frequency Synthesizer section of the Distortion Measurement Control Unit Type 1902. The signal is then stepwise mixed down and filtered. Eventually it appears as a signal centred about an intermediate frequency of either

750 Hz or 30 kHz depending on the bandwidth, B. The filter then tracks the generator with some fixed frequency off-set.

The two instruments are locked together by the fixed frequency of 1,2 MHz while the variable frequency ranging from 1 to 1,2 MHz carries the information about the Generator signal. When the Heterodyne Analyzer is operated on its own the input is mixed with the variable frequency so that the filter tracks the generator. When the Distortion Measurement Control Unit is connected, however, it takes over control of the filter position. Normally in the TDS set-up

the Distortion Measurement Control Unit is operated in the Intermodulation mode so that the filter can be given some fixed offset relative to the generator frequency.

A more detailed description can be found in the data sheets for Heterodyne Analyzer Type 2010 and Distortion Measurement Control Unit Type 1902.

Signal processing in the TDS Unit

The filtered IF signal is then taken to the TDS Control Unit for further processing. Here the signal is split. One part is fed to a fast detector the output of which is a DC voltage pro-

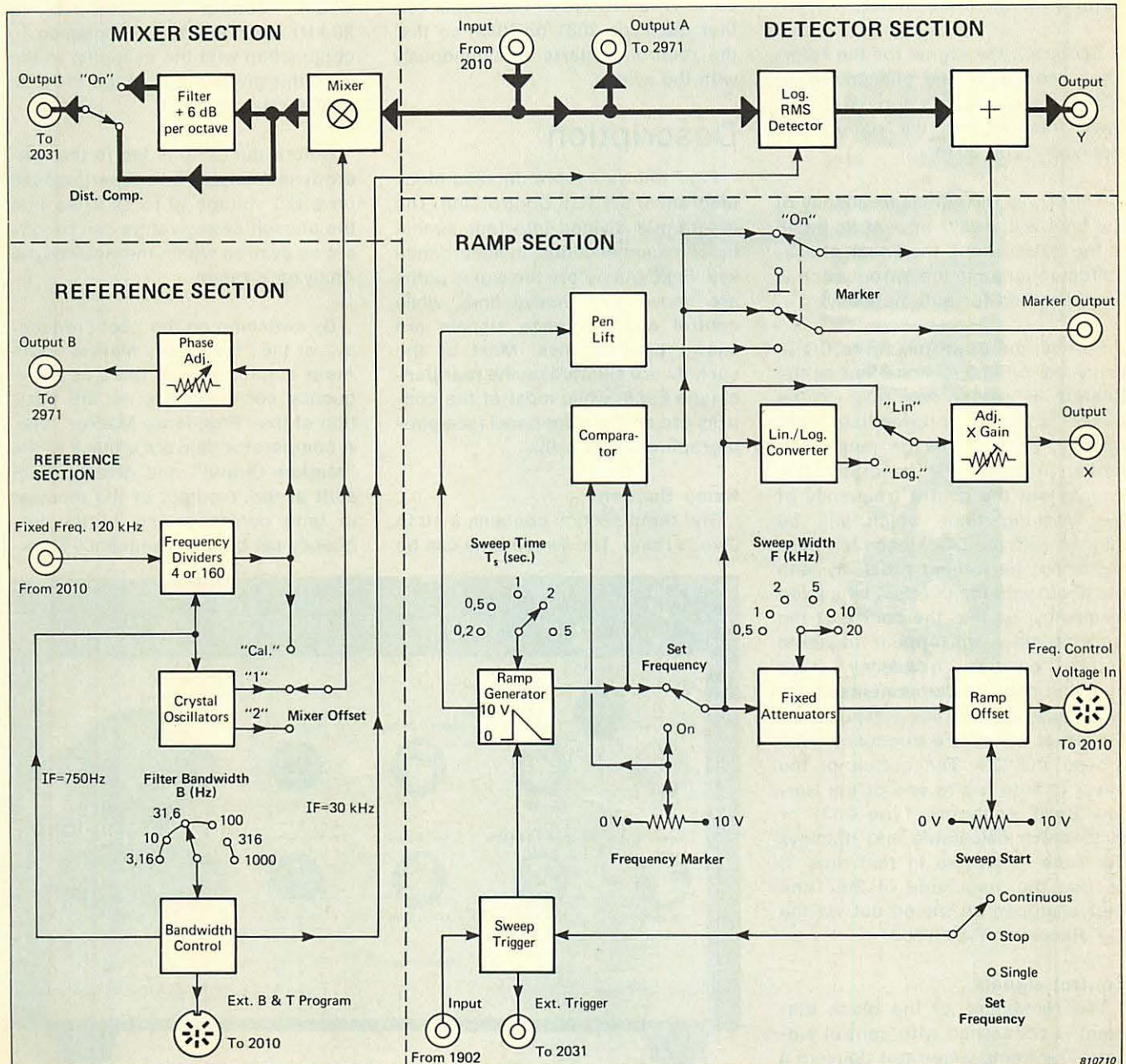


Fig. 7. Block diagram of TDS Control Unit. Signal paths are indicated by thick lines while reference and control signals are shown with thin lines. The diagram is divided into four almost independent sections (Mixer, Detector, Reference and Ramp Sections) in accordance with Fig.6.

portional to the log magnitude of the frequency response. Another part is fed directly to one channel of the phase-meter which in turn delivers a DC voltage proportional to the phase. Finally the signal is fed to the Mixer section in the TDS Control Unit.

In contrast to the explanation given in the introduction (Figs.2 & 3) the filtered signal is already mixed to the IF. This has several advantages.

Firstly, as the detector always operates at the same frequency, the time constant of the detector can be optimised to give the fastest possible ripple-free detection. The same applies for the phasemeter.

Secondly, the signal for the reference channel of the phasemeter is simply the IF which is derived from a fixed frequency in the Heterodyne Analyzer Type 2010.

Thirdly, as the centre frequency of the filter will always appear as the IF in the 2010 output, the linear phase-shift mentioned in the introduction is compensated for automatically.

Fourthly, the down-mixing to DC illustrated in Fig.3 is simplified as the filtered IF signal has only to be mixed with the Intermediate Frequency. For the reason mentioned earlier, it is not the generator frequency but the centre frequency of the tracking filter which will be aligned with the DC. Hence the mixing is not performed precisely with the IF but with the IF offset by a fixed frequency, so that the centre of the tracking filter will appear at some known positive frequency, thus avoiding that the components in the magnitude of the Time Response to the left of the centre frequency folds around the DC. The output of the mixer is then fed to one of the Narrow Band Analyzers Type 2031 or 2033 which calculates and displays the Time Response in real time. If desired the magnitude of the Time Response can be traced out via the X-Y Recorder Type 2308.

Control signals

The remainder of the block diagram is concerned with control signals. The Ramp Generator delivers a fixed ramp voltage for tuning the X-drive of the recorder and display

units. A variable ramp is used to tune the VCO in the Heterodyne Analyzer. In order to obtain a stable phase-curve the sweep must be started at a time where the phase between the generator and the tracking filter has a well-known relationship. The low note of the two tone generator in the Distortion Measurement Control Unit when operated in the IM mode contains the necessary information. This frequency not only corresponds to the filter offset frequency (or some fraction of it) but the phase will also be the phase difference between the tracking filter and the generator. It can therefore be used to trigger the sweep start of the ramp generator. When the ramp is started, a trigger pulse is fed to the FFT Analyzer (either the Type 2031 or 2033) so that the recording starts simultaneously with the sweep.

Description

Fig.7 shows a more detailed block diagram of the TDS Control Unit. The diagram is divided into four almost independent sections, in accordance with Fig.6. As before the signal paths are shown with heavy lines while control and reference signals are shown by thin lines. Most of the sockets are situated on the rear panel (see Fig.8) while most of the controls are on the front panel (see photograph on pp.8 & 9).

Ramp Section

The ramp section contains a 10 to 0 volts ramp. The sweep time can be

varied from 0,2 to 5 s in a 1, 2, 5 sequence. The ramp is either used directly to tune the X-deflection of the display of the oscilloscope or the X-Y Recorder Type 2308 or can be passed through a lin./log. converter. In the latter case the two uppermost decades will cover the whole frequency scale. In both cases the ramp is passed through the rear panel adjustment "Adj. X-Gain" to allow for different sensitivity of recorder and display units.

The ramp output is also fed to a set of fixed attenuators so that the Heterodyne Analyzer Type 2010 can be swept through various sweep ranges selected by the "Sweep Width" control from 0,5 kHz to 20 kHz again in a 1, 2, 5 sequence. In conjunction with the multiplier in the 2010 this gives a total range of 50 Hz to 200 kHz.

Before the ramp is fed to the Heterodyne Analyzer it is superimposed on a DC voltage (0 to 10 V) so that the chosen sweep range can be offset as desired within the Heterodyne Analyzer's range.

By switching on the "Set Frequency" of the "Frequency Marker" both ramp outputs will be fixed at a frequency corresponding to the position of the "Frequency Marker" dial. A comparator delivers a pulse at the "Marker Output" and gives a 5 dB shift at the Y-output at the moment in time corresponding to the frequency set by the "Frequency Mark-



Fig. 8. Rear panel of TDS Control Unit showing sockets, the "Phase Adj." control and the "Adj. X Gain" potentiometer

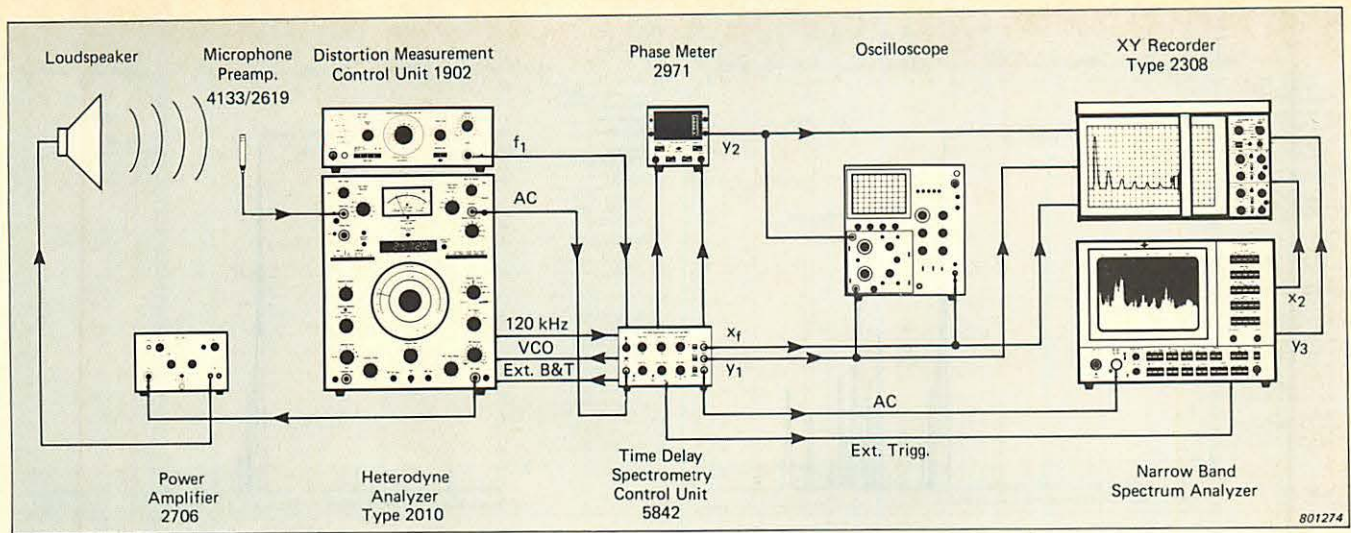


Fig. 9. TDS system used for loudspeaker development. The same arrangement is used for other types of measurements by exchanging the microphone and loudspeaker systems for the appropriate transducers and amplifiers

er" when the marker is on. With the marker off, the Pen Lift lifts the pen from the X-Y Recorder's recording paper during the return sweep.

Depending on the position of the "Set Frequency" of the "Sweep Start", the Ramp Generator will sweep the Heterodyne Analyzer Type 2010 as follows. In Continuous the Heterodyne Analyzer sweeps continuously, a new sweep commencing when at least 0.2 s have elapsed and the sweep being triggered from the Distortion Measurement Control Unit Type 1902. In Stop the ramp will be fixed at 10 V. If Single is pressed, one sweep is produced as soon as the trigger condition is fulfilled and the sweep will return to the upper limit of 10 V. The Sweep Trigger also delivers a trigger pulse to the "Ext. Trigger" input of either the FFT Analyzer Type 2031 or Type 2033 so that the time record can be started synchronous with the sweep.

Detector Section

To obtain the best possible dynamic range, the detector is of the log. RMS type. The time constant is selected according to the Intermediate Frequency of the Heterodyne Analyzer output to obtain the fastest ripple free detection. The IF depends on the bandwidth B of the Heterodyne Analyzer, so the time constant is determined by the position of the Bandwidth Control in the Reference Section.

The Detector Section also contains an adding network for superimposing the 5 dB change in the output

of the Log. Magnitude of the Frequency Response. The input from the Heterodyne Analyzer is fed directly to channel A of the Phase Meter.

Reference Section

Apart from setting the time constant of the detector, the Reference Section delivers the reference frequencies for the Phase Meter and the Mixer Section. The reference frequency for the Phase Meter is a square wave with a frequency corresponding to the IF. The IF is derived from the Heterodyne Analyzer's 120 kHz master clock by dividing by 4 or 160 according to the position of the Bandwidth Control. The rear panel control "Phase Adj." can change the duty-cycle of the square wave so that the phase can be continuously adjusted.

With Mixer Offset in "Cal." the reference frequency for the Mixer Sec-

tion is the same as for the Phase Meter i.e. the IF. With the Mixer offset in "1" or "2" the reference frequency is derived from the crystal controlled oscillators and will be the IF offset by 100 Hz and 250 Hz respectively. For convenience the Bandwidth control remotely sets the bandwidth, B, of the Heterodyne Analyzer.

Mixer Section

In the Mixer Section the filtered IF signal from the Heterodyne Analyzer is mixed with the reference frequency from the Reference Section to produce the time signal for obtaining the Time Response by a FFT transform in the FFT Analyzer Type 2031 or 2033. Zero time corresponds to 0 Hz, 100 Hz or 250 Hz in positions "Cal.", "1" or "2" respectively.

When "Dist. Comp" is "On" the Mixer output is passed through a 6 dB per octave filter. As a delay is

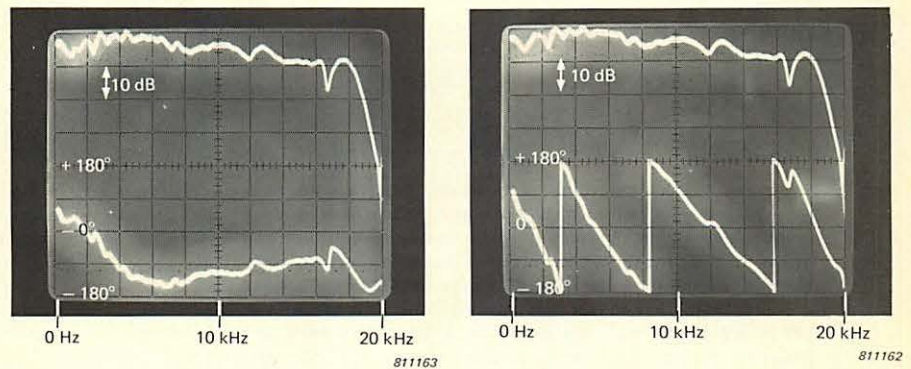


Fig. 10. Magnitude (top) and phase (bottom) of frequency response of a commercial loudspeaker as they appear on the oscilloscope screen. In the left-hand photograph the tracking filter has been tuned for the direct sound. In the right-hand photograph the filter is slightly off-tune resulting in a tilted phase trace. The jumps in right-hand phase trace are caused by the Phasemeter shifting 360°

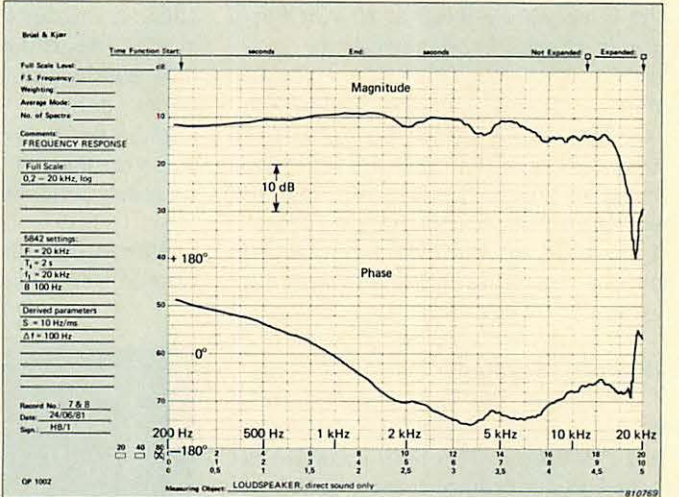
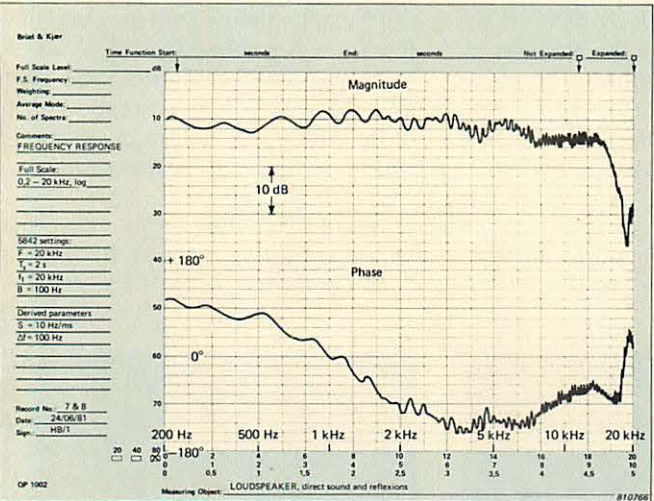
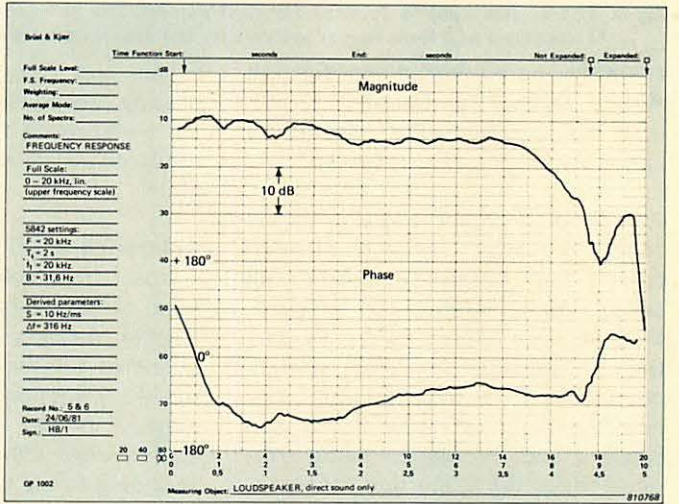
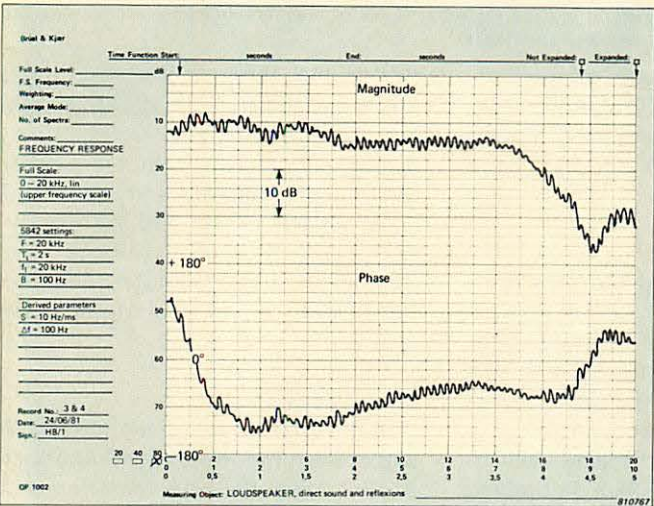
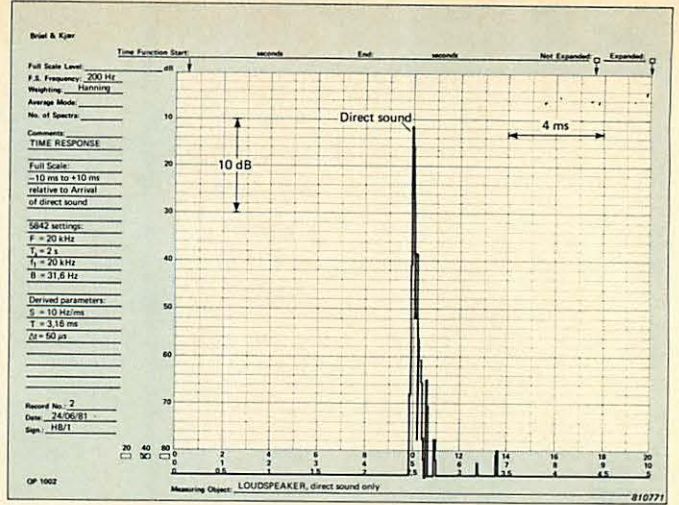
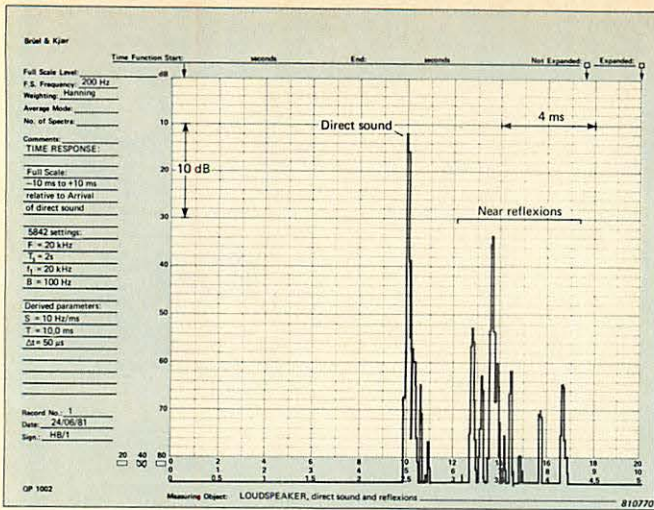


Fig. 11. Response of a commercial loudspeaker. Top: magnitude of time response. Centre: magnitude and phase of frequency response on a linear frequency scale. Bottom: magnitude and phase of frequency response on a logarithmic frequency scale. From the time response it is seen that near reflections are included in the measurements resulting in a comb-filter effect in the frequency response. This effect is best seen on a linear frequency scale. At high frequencies the comb-filter effect could easily be mistaken for noise when displayed on a logarithmic frequency scale

Fig. 12. Same as Fig.11 except that a narrower filter is used in order to exclude the near reflections thus measuring only the direct sound i.e. the free-field response of the loudspeaker itself

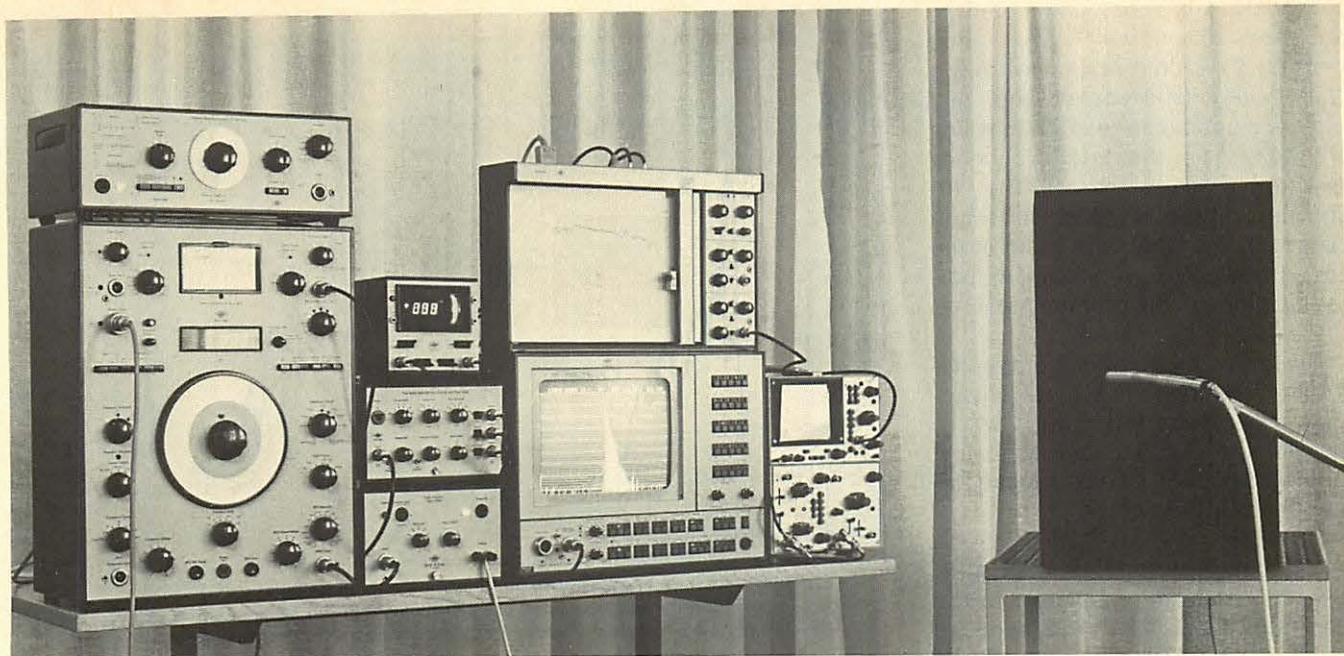


Fig. 13. Complete TDS system used to obtain the results shown in Fig.11 & 12.

converted into a frequency shift this automatically compensates for the inverse square law provided the Mixer offset is in the "Cal." position and the tracking filter offset is set to zero.

Examples of Use

Time delay spectrometry (TDS) in itself has a vast range of applications. Moreover the B & K system, made up of standard instruments offers a wide choice of other measurements as each piece of equipment in the setup retains its original capabilities. For example, advanced distortion measurements and noise analysis can be carried out using parts of the complete TDS instrumentation.

The modular design also allows the B & K system to be tailored to any specific application. If the frequency phase response or the time magnitude response is not considered important, then the Phasemeter Type 2971 or the FFT Analyzer Type 2031/2033 respectively can be omitted. Often, however, both the complex frequency response and the time response yields valuable information.

The basic instrumentation together with the expanded versions are listed under the Specifications on page 16. A schematic arrangement of the complete TDS system, used in

this case for loudspeaker measurement, is shown in Fig.9. By exchanging the measuring microphone/preamplifier assembly and the power amplifier with the corresponding transducers and amplifiers of another system (e.g. hydrophones and acoustic emission transducers) the same arrangement can be used for measurements on another system.

Loudspeaker development and quality control

A typical application for the complete system is loudspeaker development.

The responses of loudspeakers normally display both frequency- and time-domain related irregularities. At high and low frequencies the response will drop off due to variations in the acoustical and electrical impedance. On the other hand diffraction at the edges of the cabinet can bring about a whole series of delayed signals most easily recognised in the time domain. Consequently, although expressing the same information, both frequency and time response are essential for a quick localization of the trouble.

Linear frequency phase response is important not only for audible reasons but also to ease design of crossover networks for multiway speakers. If the separate drivers are not time aligned the phase response will display sudden changes of slope at

the crossover frequencies. Also by comparing the frequency magnitude and phase responses it is possible to check the individual drivers for minimum / non-minimum phase behaviour.

The high sensitivity of the frequency phase response to changes in the transmission time proves very helpful for the initial adjustment of the delay compensation. Fig.10 illustrates the influence of a slight off-tune. Although only minor deviations are seen in the magnitude response, the phase response is radically changed.

Figs.11 and 12 show some results from measurements of the response of a commercial loudspeaker. The setup itself is illustrated in Fig.13. The response is measured with and without the influence of close reflections and frequency response is displayed both with a logarithmic and with a linear frequency axis. The logarithmic display can be recorded either as in Fig.11 and 12 where one division corresponds to one third-octave or on logarithmically calibrated recording paper.

The time response itself provides a very efficient method of improving the impulse response of a loudspeaker. The quality of the direct sound is very important for the listening results since it arrives first at the human ear and the first few milli-

seconds are typically unaffected by the room reflections. Fig.14 shows the first 2 ms time response from a high quality loudspeaker with and without a hood. The responses which are based on the total frequency spectrum clearly reveal the increased diffraction and reflection caused by the hood. The width of the added peaks are very narrow indicating a broad band phenomenon.

The low measuring time of typically 0,5 – 2 seconds is a great advantage for development work where the effect of electrical or mechanical modifications of speakers can be monitored in "real time".

For quality control it is often sufficient to check the magnitude of the frequency response. The reduced version of the TDS-system is ideally suited for this purpose. Influence from both reflections and the noise normally present in a production environment is eliminated even when using short measuring times. If the best possible signal to noise ratio is not required, the 1902 may also be omitted.

Hearing aid development and testing

One of the most demanding measurements in hearing aid development is that of determining the directional characteristics of aids equipped with pressure gradient microphones. Such microphones are extremely sensitive to pressure gradient variations especially for rear incident, low frequency sounds for which a 3 dB measurement error at 400 Hz typically occurs for only a 0,1 dB or 1° variation in the sound pressure at one of the input ports of a directional hearing aid. Consequently these measurements previously necessitated a large, anechoic chamber. With TDS such measurements can be performed in an ordinary room or even a reverberation chamber. Results of such measurements are shown in Fig.15 for four different angles of incidence of the sound. The deviations between these curves and those obtained in an anechoic chamber for frequencies less than 1 kHz are less than 1 to 2 dB.

Room Acoustics and Sound Insulation

For room acoustics and sound insulation both frequency and time re-

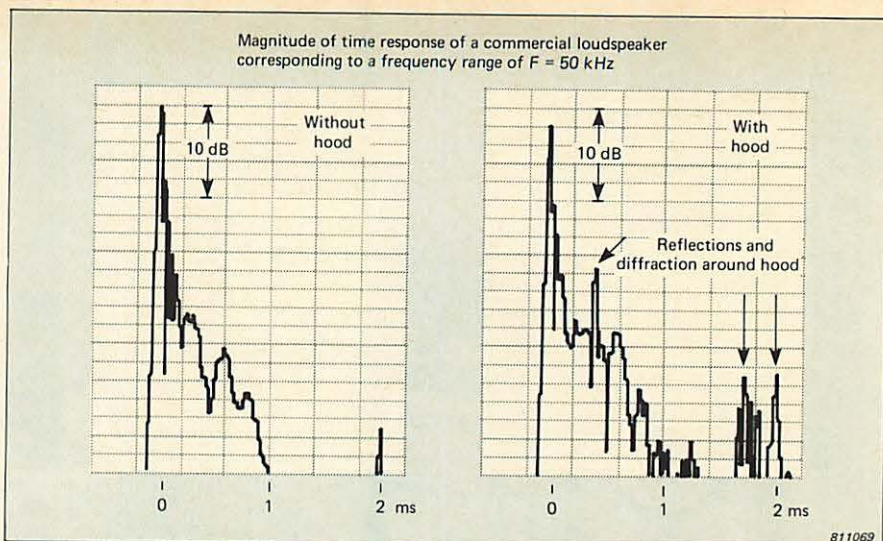


Fig. 14. Time response of a commercial loudspeaker with and without hood. Only the first two milliseconds are shown corresponding to the direct sound. The effects of reflections and diffraction around the hood are clearly seen in the right-hand response. To obtain a better time resolution a wider frequency range was swept for these measurements

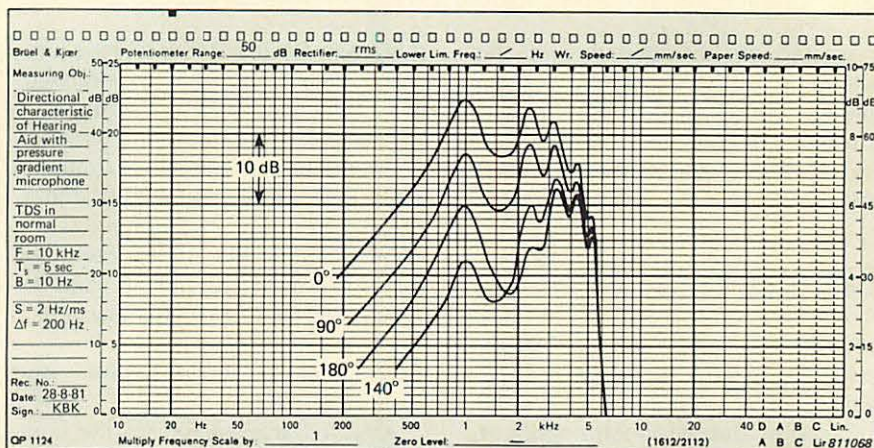


Fig. 15. Results of the very demanding measurements on a hearing aid equipped with a pressure gradient microphone. Using TDS the measurements are disturbed neither by background noise nor by pressure gradient variations in the sound field

sponse measurements are relevant. The time response is used to investigate the total reflection pattern of a room and to identify the sound paths through windows, doors, etc. Knowing the physical distances involved it is an easy matter to determine the sound path related to a certain peak in the measured time response.

Isolation of a specific part of the time response enables measurement of the corresponding frequency response. The reflective properties of e.g. the ceiling of a room can be evaluated by comparing the response of the reflection to the direct response. For this purpose the square distance attenuation and the directional properties of the loudspeaker must be taken into account.

Transmission loss and reflection coefficient of absorbing materials can be measured for various angles of incidence by inserting a piece of the material into the direct path or next to an ideally reflecting surface.

A special application is checking anechoic rooms for reflections. When testing highly directional transducers reflections even more than 30 dB lower in level than the direct sound can easily influence the measurements.

When basing the time response on a narrow frequency range the reverberation time of a room is simply derived from the slope of the response. The large memory of the High Resolution Analyzer Type 2033

is particularly suited for this use as it enables the analyzer to make an automatic scan of the reverberation decay versus frequency. It can also display the average reverberation decay for a wider frequency range (see Fig.16).

B & K Microphones used as sound sources

A condenser microphone can be used reciprocally i.e. as a sound emitter as well as a sound receiver. When correctly chosen and compensated a condenser microphone comes close to being an ideal sound characterised by a very flat frequency response over a wide frequency range and small physical dimensions (Fig.17). A major limitation of condenser microphone sources is the relatively low sound pressure levels and consequently the poor signal to noise ratio obtainable with most measuring techniques. However this limitation does not apply to the TDS technique due to its outstanding noise rejecting properties. In normal non-anechoic environments, TDS can provide signal to noise ratios of well above 40dB. Thus with TDS a condenser microphone sound source becomes a straightforward tool for a multitude of otherwise cumbersome two-port measurements such as scale model tests, acoustic impedance, investigations of absorbers and reflectors, calibration of other microphones and creation of well-defined sound fields.

Other Applications

The examples mentioned above have been selected from the electroacoustical and room acoustical fields and only give a brief idea of the possibilities of the Brüel & Kjær TDS equipment. Another electroacoustical application is the measurement of loudspeaker and hands-free telephones in an ordinary laboratory or on the production floor. The time selective frequency response and the time response has, however, just as many uses in the underwater, acoustical, mechanical and electrical fields. A few examples are given below.

Crack detection

When a sound wave propagates along an air filled tube or a solid structure holes and cracks will generate reflected waves. By measuring

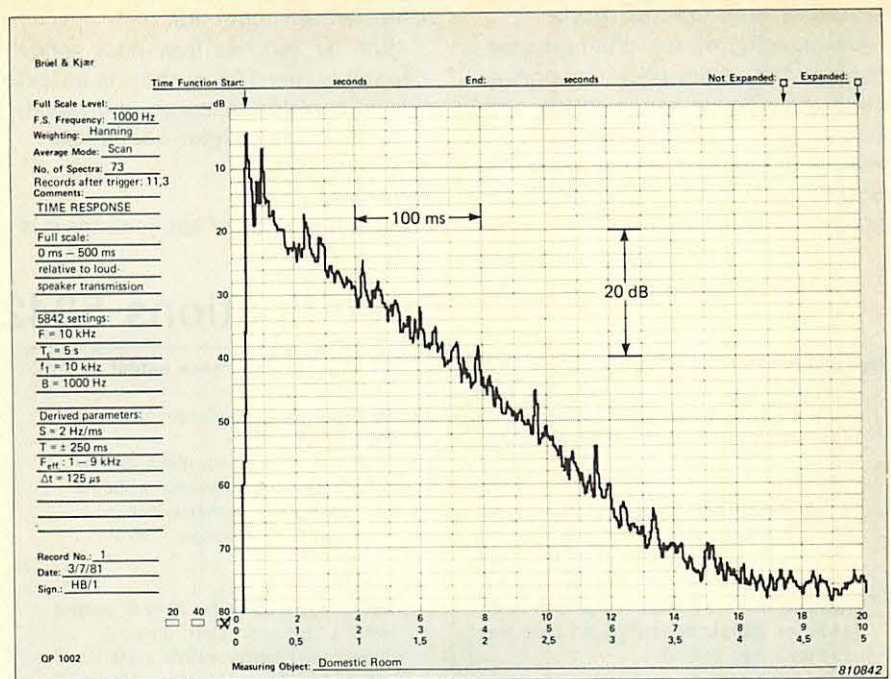


Fig. 16. Reverberation decay measured with the TDS system. The zoom facility of the 2033 can be used to obtain a longer time response, so that complete reverberation decays can be studied in detail. The scan average has been used to compress the 4000 line time response to 400 lines

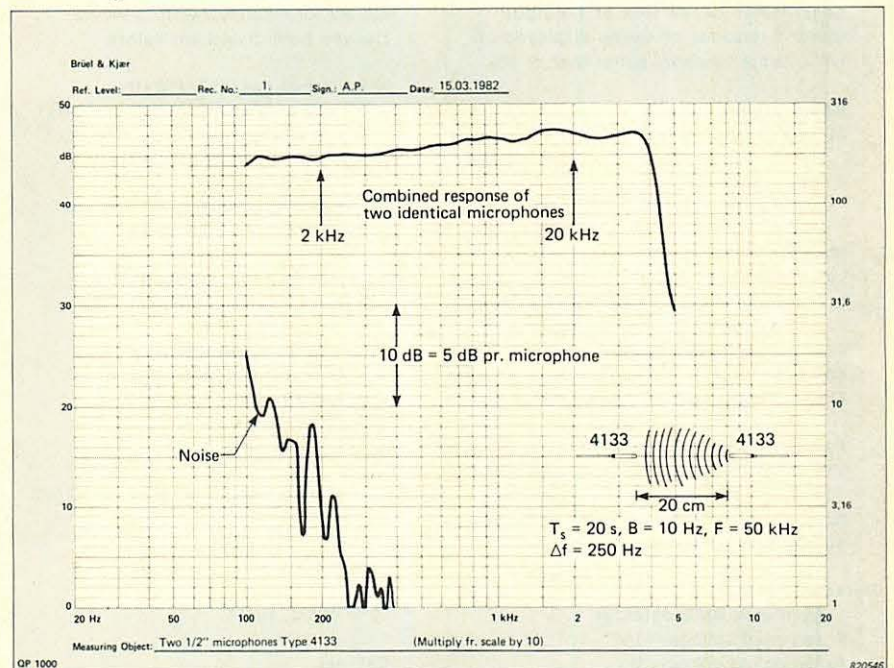


Fig. 17. Total frequency response of two 1/2" Microphones Type 4133, using one as an emitter and the other as a receiver. Deviations from the ideal flat response can be equally attributed to the emitter and the receiver thus making the 10dB division on the vertical axis equivalent to 5dB per microphone. For signal to noise evaluation the 10dB division is valid

the time response one or more places along the structure holes can be located and their sizes roughly judged.

Delay compensation in tape recorders

Tape recorders exhibit a significant delay between recording and playback heads. Normally this ne-

cessitates very long measuring times for selective swept measurements of frequency response and distortion. By compensating for the delay, bias optimisation can be carried out in "real time" by monitoring alternately the frequency response and the 3rd order difference frequency distortion.

Transient response of filters

Anti-aliasing filters, although possessing flat frequency responses within the audible range, often produce audibly different results. From the time response it can be seen that the filters will have very different responses to transients.

Underwater acoustics

With its 200 kHz frequency range capability, the TDS system is an excellent tool for research and development in underwater acoustics.

Summary

From the range of applications it is

seen that the TDS system is suited for fast measurement of transfer functions within the frequency range 2 Hz — 200 kHz. The responses can be presented in both time and frequency domains. For a given measuring time no other method offers comparable rejection of reflections and noise.

Specifications 5842

Ramp Generator Section

Sweep Time: 0,2s; 0,5s; 1,0s; 2,0s and 5,0s \pm 1%.

Time between sweeps (continuous mode): 0,2s

Ramp Amplitude (Sweep Width): 0,25; 0,5; 1,0; 2,5; 5,0 and 10,0 V_{pp} \pm 1%

Ramp Lower Limit (Sweep Start): Adjustable 0 - 10 V (10V corresponds to 2 kHz or 200kHz depending on 2010 frequency range setting)

X-Output for display and recording: -5V to +5V nominal. Adjustable \pm 50%. Minimum load impedance 5 k Ω

Logarithmic conversion of X-output: Upper 2 decades of sweep displayed on full X-range Linearity better than \pm 5%

Ramp trigger voltage:

300 mV positive slope

Trigger output:

Positive going pulse slope

Marker (Marker Switch "On"):

1 ms pulse appearing as

1: 0 V at marker output (normally + 15 V)

2: an increase of 0,5 V at Y output

Can be positioned anywhere within the sweep limits.

Marker output source impedance: 10 k Ω

Pen Lift (Marker Switch "Off")

TTL "0" during sweep (max. current drain 10 mA)

open output between sweeps (max. voltage 15 V)

Detector Section

Logarithmic RMS detector

Rated input voltage: 10 V

Rated output voltage: 0 V

Conversion ratio: 10 dB to 1 V

Minimum load impedance: 5 k Ω

Dynamic range: > 70 dB

Crest Factor capability: 3

Linearity of conversion: Better than 1 dB

Averaging time: 0,7 ms (2010 IF = 30 kHz)

7 ms (2010 IF = 750 Hz)

Phase Reference Section

Measuring signal output

Amplitude: max. 10 V rms

Minimum load impedance: 15 k Ω

Square wave reference output

Amplitude: 8 V_{pp}

Minimum load impedance: 15 k Ω

Frequency:

30 kHz for Filterbandwidth \geq 316 Hz

750 Hz for Filterbandwidth \leq 100 Hz

Derived from 2010 masterlock

Phase adjustment range: > 270°

Mixer Section

Mixer frequencies for 2010 IF output

Mixer offset pos. "Cal": 0 Hz

30 kHz for Filterbandwidth \geq 316 Hz

750 Hz for Filterbandwidth \leq 100 Hz

Derived from 2010 masterclock

Mixer offset pos. "1": 100 Hz

30,1 kHz for Filterbandwidth \geq 316 Hz

650 Hz for Filterbandwidth \leq 100 Hz

Derived from crystal oscillators

Mixer offset pos. "2": 250 Hz

30,25 kHz for Filterbandwidth \geq 316 Hz

500 Hz for Filterbandwidth \leq 100 Hz

Derived from crystal oscillators

Nominal output: 10 V

Minimum load impedance 5 k Ω

Noise and spurious frequencies

60 dB below nominal output

Distance compensation

+ 6 dB pr. octave filter

Frequency range: 0 — 250 Hz

Unity gain frequency: 50 Hz

General

Operating temperature: 10° to 40°C (50° to 104°F)

Power Supply:

100, 115, 127, 220, 240 V (\pm 10%)

50 to 60 Hz, 10 VA

Cabinet:

Supplied as model A (light-weight metal cabinet)

Dimensions and Weight:

Height: 132,6 mm (5,2 in)

Width: 209,5 mm (8,3 in)

Depth: 200,0 mm (7,9 in)

Weight: 2,7 kg

Accessories included:

2 Control cables AQ 0034

1 Power cable AN 0020

Spare fuses 125 mA VF 0030

Spare fuses 250 mA VF 0031

Control cable WL 0380

(for Type 2308 X-Y recorder

pen lift)

Kit for 2971 modification..... WH 1073

Basic instrumentation for TDS (not including transducers, preamplifiers etc.):

Time Delay Spectrometry

Control Unit..... Type 5842

Distortion Measurement Control Unit

Type 1902

Heterodyne Analyzer Type 2010

X-Y Recorder..... Type 2308

Power Amplifier, e.g. Type 2706

Oscilloscope (variable persistence screen and external X

input. Dual channel for simultaneous display of both mag-

nitude and phase)

1 Screened cable, B & K

plugs, 1,2 m AO 0014

5 Screened cables, B & K-

BNC, 1,2 m..... AO 0064

5 Screened cables, BNC

plugs, 1,2 m AO 0087

2 Control cables, 8 pin plugs,

1,5 m AQ 0034

1 Control cable, 8 pin plug-

BNC, 1,1 m..... WL 0380

2 Connectors, BNC-T piece JJ 0152

Recommended recording

paper QP 1002

Additional instrumentation for phase measurements:

Phasemeter (including Modifi-

cation WH 1073) Type 2971

3 Screened cables, BNC

plugs, 1,2 m AO 0087

1 Connector, BNC-T piece JJ 0152

Additional instrumentation for time response measurements

Narrowband Analyzer Type 2031

or

High Resolution Analyzer Type 2033

1 Screened cable, B & K-BNC,

1,2 m AO 0064

1 Screened cable, BNC plugs,

1,2 m AO 0087

1 Control cable, 8 pin plugs,

1,5 m AQ 0034