

Dual Channel FFT Analysis for the Development and Evaluation of Tape Recorders



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Introduction

This Application Note deals with measurements on tape recorders that can be performed using Brüel & Kjær Types 2032 (High Speed) and 2034 (Standard) Dual Channel FFT Signal Analyzers.

The combination of a tape recorder and a dual channel FFT analyzer is interesting both from the point of view of the testing device and the test object. A tape recorder exhibits so called time variance (wow and flutter) and has a substantial amount of spe-

What a dual channel FFT analyzer does

A linear time invariant two-port (input-output device) is fully characterized by its transfer function or frequency response H(f). H(f) is defined as the ratio of the output spectrum B(f) to the input spectrum A(f):

$$H(f) = \frac{B(f)}{A(f)}$$
 (no noise) [1]

Under ideal conditions one measurement would suffice. In practice, however, both spectra are contaminated with noise so the measurement is repeated and a complex average performed in order to reject random noise and any non-linearities. A complex average means that both the magnitude and the phase of the spectra are considered. At any given frequency, f, a (Fourier) spectrum has both a magnitude and a phase (or alternatively a real and an imaginary part). It can therefore be considered to represent a vector in the complex plane. We will use this analogy in the following. Thus, complex averaging means that a true vector addition is performed.

In contrast, a *power (or auto-) averaging* means that only the numerical lengths of the vectors are averaged and all phase information is discarded. Complex averaging is capable of rejecting random noise. Auto-averaging is not.

In practice, the averaging of H(f) is not done on the "raw" fraction given cific non-linearities associated with it (magnetic saturation, bias effects etc.). These effects violate somewhat the two basic assumptions of dual channel FFT analysis: time invariance and linearity. From the point of view of the analysis technique it is hence interesting to study how these effects influence the measured results. This is done in the first sections of this note.

Once one is aware of these influences, a dual channel FFT analyzer can be used in a very straightforward manner to obtain a vast range of measurements which are applicable both to the design as well as to performance checks and calibration of tape recorders. Especially worth noting are the measurements of phase response which are very easily performed using a dual channel FFT Analyzer but rather cumbersome to obtain with most other techniques. Numerous measurement examples are given in the latter sections of this note.

in equation [1] above. Instead, first the fraction is "prolonged" by the complex conjugate $A^*(f)$ of the input spectrum, i.e.:

$$H(f) + \text{noise} = \frac{B(f)}{A(f)}$$
$$= \frac{B(f) A^*(f)}{A(f) A^*(f)}$$
$$= \frac{A^*B(f)}{|A|^2(f)}$$
[2]

The averaging is then performed *in*dependently in the numerator and the denominator, resulting in the averaged results $G_{AB}(f)$ and $G_{AA}(f)$ respectively. $G_{AB}(f)$ is called the "cross spectrum" and $G_{AA}(f)$ is the "input auto spectrum". The division of the two yields an *estimate* of the transfer function, $H_1(f)$:

$$H_1(f) = \frac{G_{AB}(f)}{G_{AA}(f)}$$
[3]

These manipulations of equation [1] serve the following purpose: The measured output spectrum B(f) appears only in the cross spectrum where it is always "weighted" by the corresponding input spectrum A(f). In this manner all spectral components at the output which are not correlated to the input will be averaged out. Hence $H_1(f)$ rejects the noise at the output of the test device and $H_1(f)$ will in fact equal the "true" transfer function H(f) if the noise present in the measurement affects only the output of the test object. In the practical situation, where noise is present both at the input and the output, $H_1(f)$ gives us a lower limit for H(f), i.e.:

$$|H_1(f)| \le |H(f)| \tag{4}$$

Analogously, one may instead choose to "prolong" equation [1] by the complex conjugate $B^*(f)$ of the output spectrum. After averaging, this yields another estimate $H_2(f)$ of the transfer function:

$$H_{2}(f) = \frac{G_{BB}(f)}{G_{BA}(f)}$$
$$= \frac{G_{BB}(f)}{G_{AB}^{*}(f)}$$
[5]

Both $H_1(f)$ and $H_2(f)$ are implemented in the B&K Types 2032 and 2034 Dual Channel Analyzers. It can be shown that $H_2(f)$ exactly equals the "true" transfer function H(f) if all the noise originates from the *input* of the test object. In the practical situation, where noise is present both at the input and the output, $H_2(f)$ gives an upper limit for H(f):

$$|H_2(f)| \ge |H(f)|$$
 [6]

Note that for tape recorders, where most noise is tape hiss and hence only



Fig.1 Typical coherence of tape recorder measurements using: (a) impulse excitation (b) white noise excitation

affects the output, $H_1(f)$ seems the most reasonable estimate to use. Note also that for the phase response measurement, we will always have the condition:

$$\angle H_1(f) = \angle H_2(f) = \angle G_{AB}(f)$$
[7]

With the two estimates $H_1(f)$ and $H_2(f)$ at hand, a very simple measure of the "quality" of the measurement, the so-called *coherence* $\gamma(f)$ can be defined as follows:

$$\gamma^{2}(f) = \frac{H_{1}(f)}{H_{2}(f)}$$
$$= \frac{|G_{AB}(f)|^{2}}{G_{AA}(f) G_{BB}(f)}$$
[8]

Always, $0 \leq \gamma(f) \leq 1$ [9]

 $\gamma(f) = 1$ indicates a very good measurement. $\gamma(f) = 0$ indicates a rather poor and unreliable measurement. Factors decreasing the coherence are for example: external noise, distortion, too coarse frequency resolution of the analyzer and, as we shall see, time variance of the test object. Fig. 1 shows coherence in a typical tape recorder measurement using noise and impulse excitation respectively. Note the very typical monotonous decay with frequency. It is due to wow & flutter.

The two methods, H_1 and H_2 , for obtaining an estimate of the transfer function are so flexible that they will almost always yield a reasonable result no matter what kind of excitation signal is employed.* This is one of the major advantages of dual channel analyzers. One natural bond on the choice of allowable excitation signals, however, lies in the fact that equations [1] -[3] are not defined if the denominator, at some frequency f_o within the range of the analyzer, equals zero. Therefore (for transfer function measurements) only broadband excitation is permissible. Typically, impulse, random white noise or pseudo-random white noise is



used. These signals together with a variable sine are available from the built-in signal generator of Types 2032 and 2034, making the analyzers completely self-contained universal test-stations.

Fig.2 summarizes the basic definitions of frequency response measurements using a dual channel FFT analyzer.

* If a limitation to only one type of excitation is set beforehand, more devoted detection methods, utilizing the knowledge of the input signal more effectively, can be employed. Time Delay Spectrometry, utilizing only linear sine sweep excitation is a good example.



Fig.2. Summary of the basic definitions of dual channel FFT measurement of frequency response

The influence of wow and flutter upon dual channel analysis

Repetition of a measurement and subsequent averaging is a vital part of dual channel FFT analysis. However, it has full meaning only if the condition of time invariance is satisified, i.e. the test object stays basically "the same" throughout the measurement series. Generally this is not the case with tape recorders. The fluctuations in the tape speed make the tape transportation delay from the recording head to the replay head vary somewhat around a middle value, in addition to the time-stretching and timecompression of the signal itself (so called modulation noise).

The effect of this variation in tape transportation delay is illustrated in Figs.3 and 4. Here repetitive application of a short square impulse-excitation is considered. Since the excitation signal is very short in duration, the effects of time-streching and timecompression can be considered negligible.

Fig.3 illustrates how a differential deviation $\Delta \tau$ of the encountered delay from an average value τ_N is reflected in a phase rotation of the corresponding cross spectrum according to the formula:

$$\Delta\phi(f) = -2\pi f \,\Delta\tau \tag{10}$$

To a very good approximation, the magnitude is not affected.

Fig. 4 shows the situation after averaging many impulse measurements. Wow and flutter cause a spread of the encountered time delays between the recording and replay head around the nominal value τ_N . For simplicity a rectangular distribution of width $\Delta \tau_{MAX}$ is assumed.

According to equation [10] above this "delay spread" causes a more and more severe phase spread with increasing frequency. It is however striking to note that this does not in any systematical way influence the estimate of the average phase response, which is the average direction in which the cross-spectrum "vectors" are pointing. But the phase spread results, after complex addition and averaging of the vectors, in an underestimation of the magnitude of the cross spectrum. (Here the underestimation is relative to the same case without wow and flutter, i.e. no phase spread). $|G_{AB}|$, $|H_1|$ and coherence will decrease systematically with frequency and at a maximum frequency, f_{MAX} = $1/\tau_{MAX}$, where the phase spread reaches 180°, we will have $|G_{AB}|$ $|H_1|$ = coherence = ZERO, due alone to wow and flutter. See Fig.1

Beyond f_{MAX} the phase estimate also becomes incorrect. The Analyzers automatically blank out the parts of phase responses measured with too low a coherence. The auto-blanking can be switched off, if desired.

We can hence draw the following conclusions:

 "Don't trust the magnitude of H₁ if the system is not time invariant."



Fig.3. The influence of test object delay-variation on the "instantaneous" cross spectra



Fig.4. The spread of the encountered delays causes spread in the direction of the encountered cross-spectrum components

- "To obtain correct magnitude use power averaging and not complex averaging, i.e. use AUTO SPEC-TRA." This is in compliance with usual techniques for tape recorder measurements.
- "The estimate of the *phase* of the transfer function H_1 is, up to

Measurements

Choice of excitation signal

As previously mentioned, Analyzers Types 2032 and 2034 offer a choice of excitation signals: random white noise, pseudo-random white noise, impulse and variable sine^{*}.

The great advantage of broadband excitation (white noise and impulse) lies in the ease with which the *total* frequency response, including phase, can be monitored while, for example, adjusting bias, equalization or azimuth. Also, it can be implied that in many cases this type of excitation re f_{MAX} , not affected by wow and flutter."

The last conclusion is a rather unique feature of dual channel FFT analysis, as most other techniques have problems measuring the phase correctly and within reasonable time limits due to wow and flutter. The frequency f_{MAX} , the point where the coherence reaches zero in a tape recorder measurement can be considered a simple figure of merit for the device. However, it **cannot** in any straightforward way be translated into "wow and flutter" and "drift" values.

sembles to a higher degree the signals usually recorded. However, by tradition, sine excitation is the most commonly used and best-known test signal for tape recorders. Since it is a narrowband signal, sine excitation is not applicable to dual channel analysis as described here. It is applicable in other, less sophisticated modes of the Analyzer (channel A and channel B auto-averaging, statistical). Hence all but the phase response measurements can be performed with either broadband or narrow band excitation.

The minor differences between results obtained with sine excitation and broadband excitation are due to the different impact upon the signals of bias, "self-biasing", saturation, demagnetization, "head bumps" and other non-linearities associated with magnetic tape recording. See the section "Frequency Response (Magnitude)" for examples.

In choosing between random noise and impulse excitation (pseudo random noise should not be used) impulse excitation was chosen for most of the following measurements, as it minimizes the influence of wow and flutter.

One problem with impulse excitation is that the VU-meters of the tape recorders cannot be used for level monitoring (too slow). However, use of the Analyzer Δ -cursor, ensures that the recording takes place at a level below saturation (see Fig.5) and the absolute output level can be read out.



Fig. 5. Check of the linearity of the recording process when impulse excitation is used. The value " Δ TOTAL" specifies the amount of energy contained with the Δ -cursor range (Δx field). Note that the first increase of input level by 10dB (Fig. $a \rightarrow$ Fig. b) is accompanied by a 9dB increase of the output, whereas the next 10dB increase in input level (Fig. b) \Rightarrow Fig. c) results in only a 5,8dB increase at the output, indicating saturation of the magnetic components. The figures show the envelope of the time function. As the envelope is always positive, a logarithmic amplitude scale (here 40dB) can be employed for better visual dynamics



^{*} For variable sine excitation, the frequency can be set to any desired value within the range 0Hz to 25,6kHz.

Setting up the Instruments

Fig.6 shows the instrumentation set-up employed for the measurements. In all cases, the test object was an audio-type 3-head cassette tape recorder (Alpine AL-85) with simultaneous access to pre- and post-tape signals.,

The lowpass filter band-limits the excitation signal to cover only the *relevant* frequency range of the tape recorder. This is important since frequencies outside this relevant range will be added to the bias-field and cause the so-called self-biasing effect or — in the worst case — self-erasure of the signal. See Fig.7.





Fig. 6. (a) The instrumentation

- (b) The measurement set up
 - (c) If the spacing and hence the delay between the recording and replay heads of the tape recorder is not known beforehand, it can be measured by temporarily increasing the record time length T of the analyzer (here 1s) and using the "Reference" cursors to measure the delay (here 97,16ms)



Fig. 7. (a) The recorded impulse (bottom) when the input impulse (top) is band-limited to 20kHz (b) The recorded impulse without any band limitation of the input. Note the self-erasure

Phase Measurements

Dynamic Change of Phase — Speed Drift

Fig.8 shows a phase measurement which was performed at two different

places on a cassette tape using the setup shown in Fig.6. Note, how a small drift of the average tape velocity (in this case only 0,087 ms/96,679 ms = 0.9 promille) is reflected in a change of the average encountered delay between the recording and replay heads and hence, according to equation [10], a different *linear* tilt (with a slope of $-2\pi\Delta\tau$) of the phase response. In order to discern this linear tilt (which is audibly unimportant) from other peculiarities of the phase response, a *linear* frequency axis is employed.

The linear tilt of the frequency phase response constitutes an extremely sensitive method of measuring time differences. A time resolution which is far better than the length of the sampling interval ΔT can be achieved.

The "CMP." field of the Analyzer enables compensation of any linear phase tilt, for example to yield a phase display which is fully contained within the -200 to +200 degrees Y-axis range, without any "jumps".

Equalization of Average Phase

The simple pre- and post equalization circuits employed in most tape recorders generally compensate for flaws in the magnitude response, but the phase response, i.e. the resultant average phase response will not be the ideal linear phase.

A simple first order all-pass network can therefore be inserted to improve the phase response. The tuning of such a phase corrector is greatly simplified using the Analyzer and the results can be verified. See Figs.9 to 11.

In subjective audio terms, a linear phase results in better transient response (Fig.11), correct relation between the fundamental tone and the harmonics of music and a more precise stereo picture.

The all-pass function employed has the form:

$$H(f) = \frac{1 - j \frac{1}{f_0}}{1 + j \frac{f}{f_0}}$$
[11]

See Fig. 10.

Dynamic Change of Phase — Movement of Recording Zone

During the recording procedure the tape passes the recording head of the tape recorder. The actual recording on the tape, however, does not take place exactly at the recording head gap. There, the field is too strong and it only saturates the tape. The remanence on tape (i.e. the recording) actually takes place in a so called *recording* zone somewhat beyond the recording head gap (Fig. 12). The position of the recording zone depends on the tape used and on the total magnetic field level at the recording head (signal + bias). The stronger the field, the more the recording zone penetrates into the tape and extends out from the recording head. Fig. 13 shows how this movement can be measured using the Dual Channel Signal Analyzer, as the movement manifests itself as a small change in the delay from recording to replay head.

Unfortunately the variations in tape speed also influence this same delay, as seen in Fig.8. Therefore, although an instantaneous change in the phase tilt can be seen when varying bias and signal level, a *reliable* measurement of the movement of the recording zone requires that the *exact average tape speed* associated with each measurement series is known. This can, for example, be accomplished using one track of a multiple track recorder to monitor tape speed while varying bias or signal level only in the other track.

Azimuth

The relative azimuth misalignment between recording and replay heads







- Fig.8. (a) Record/replay phase response of a tape cassette towards the end of the cassette
 - (b) Record/replay phase response of the tape cassette near the beginning of the cassette
 (c) Using the "Equalize" function the difference between the
 - (c) Using the "Equalize" function the difference between the two responses up to some 14kHz is shown to equal a pure delay-difference of -0,08724ms. Beyond 14kHz the greater difference between the two is due to very low coherence, i.e. the measurement here is "unreliable". The automatic phase blanking is here inoperative



Fig. 10. Phase and magnitude responses of the all-pass phase correction network employed, with cross-over frequency at 3840 Hz for this specific tape/tape recorder combination. Experiments showed that basically the same phase response was obtained and hence the same phase correction could be applied for the following three tapes: TDK MA (metal), TDK SA-X (chrome) or TDK AD (normal)



Fig. 11. Impulse responses with and without phase correction. Linear phase response gives better transients. With phase correction the energy of the impulse is more symmetrically and tightly confined in time. Viz, the time interval of $\Delta x = 0,305$ contains relatively more energy than that contained in a 4 times as wide time span without phase correction. The so called "ringing" is reduced with linear phase

can be measured by modifying the instrumentation set-up shown in Fig.6.

First, the same signal (here, white noise) is recorded simultaneously on two tracks. Then it is replayed with one channel of the replay head connected to channel A of the Analyzer and the other channel connected to channel B.

Under ideal conditions the two signals will be identical and hence the calculated transfer function will equal unity (zero phase). Any azimuth misalignment is revealed as a phase tilt (Fig. 14).



Fig. 12. The recording zones present in front of a recording head at:

(a) High bias level

(b) Medium bias level. Optimum utilization of the full tape thickness

(c) low bias level

Deeper penetration of the recording field into the tape results in better signal-tonoise ratio and lower distortion. Conversely, as the thickness of the recording zone increases, and short wavelengths comparable with this thickness cannot be recorded, high frequencies are lost with deeper penetration (i.e. with higher bias). A thinner recording zone is obtained the more rapidly the magnetic field decays with distance outwards from the recording head. The shape of the field depends on the shape of the recording head tip. These phenomena can be investigated by measuring the movement of the recording zone as a function of bias level and hence the magnetic field level







Fig. 13. Movement of recording zone as function of bias level. Neglecting the influence of tape speed drift, the movement amounts to $23,5 \,\mu s \times 4,76 \, cm/s = 1,1 \,\mu m$. The nominal head gap-width is $3 \,\mu m$



Fig. 14. (a) Azimuth. There is a time difference of 3,8 µs between two stereo tracks. A time misalignment of 3,8 µs corresponds to a rotation of approximately

(b) The measurement set up(c) The instrumentation

Frequency Response (Magnitude)

Magnetic recording is fundamentally a non-linear process which is quasilinearized by various measures such as biasing and equalization. However, both the choice of input signal and its level will affect the performance.



Traditionally sine excitation is used as standard reference. It is most conveniently handled by Types 2032 and 2034 using peak detection. See Fig. 15. The built-in variable sine generator is manually or externally (via an IEC interface) stepped through a range of relevant values, their peak values being retained on the screen.



Alternatively broadband excitation may be used, giving, among other things, the advantage that the *total* response can be monitored while simultaneously adjusting, for example, the bias or equalization.

As can be seen in Figs. 15 and 16, a recorder adjusted for flat response using broadband excitation will also be basically flat when measuring with sine signals. By trial and error an empirical rule of thumb can be established for a given tape/tape recorder combination, relating the necessary noise or impulse level to obtain a response as similar as possible to one using a given *sine* excitation level. Such an empirical rule of thumb will work best at "low" levels (-20 dB and under) where the linearity of the recording process is high.

Note the much better correspondence between the auto-spectra of Figs. 15 and 16 and what one would intuitively call "the frequency response" of this tape recorder. The complex-averaged result of Fig. 9a is severely corrupted by wow and flutter at high frequencies. The magnitude of H_1 is only trustworthy at low frequencies. Fig.17 illustrates low frequency "head bumps" of the replay head. "Bumps" are due to the stray field of the recorded tape immediately to the left and right of the replay head penetrating the head and, depending on the wavelength, either adding to or partly cancelling the "desired" field originating from the tape exactly at the position of the replay head. Obviously "bumps" will be largely attenuated if impulse excitation is used.



- (b) Spectrum of the output signal from the tape recoder. (The level of -15dB is measured using the PEAK VU indication of the tape recorder)
- (c) Same as Fig. 16a., but with white noise excitation



- Fig. 17. (a) The measurement set up
 - (b) The so-called "bumps" of the replay head can be viewed by decreasing (or zooming) the frequency span of the analyzer accordingly (here to 400 Hz). Note that the displayed range is chosen to 200 Hz only. Alternatively, a logarithmic x-axis from 4 Hz to 400 Hz can also be chosen. Note also only 20 dB y-range



Noise

For evaluating tapes and noise reduction systems, the built-in facility for A-weighting the noise spectra is very useful. Lin. and A-weighted signal-to-noise measurements according to IEC 94–3 can hence be performed directly. See Figs. 18a and 18b. A comparison of the noise spectrum of a factory bulk-erased ("virgin") tape with the noise spectrum of a tape which has been erased using the tape recorder enables the erasure performance of the recorder to be judged. See Fig. 19. It should be noted that when the AC-bias of the recording head is active, the noise on tape is increased by up to some $3 dB^{[1]}$ (re-recording of tape noise) despite the fact that no audio signal is fed to the head, and hence the apparent efficiency of the erasure head is decreased.



Fig. 18. (a) The noise spectrum of a factory bulk-erased tape replayed with Dolby C noise reduction system. Shown here with logarithmic frequency axis. The total noise power can be read out in the "TOTAL" field
 (b) Same as Fig. 18a but A-weighted. The weighted noise power can now be read out in the "TOTAL" field. The A-weighting curve is

a standardized frequency weighting which reflects the subjectively perceived noise annoyance. It gives low importance to frequencies below some 400 Hz and above some 11 kHz



Fig. 19. (a) & (b) Factory bulk-erased "virgin" tape replayed with and without the Dolby C noise reduction system. A 20dB reduction of noise is achieved with Dolby C throughout most of the audio range. Note the linear frequency axis
 (c) & (d) Tape deck erased tape recorded and played back with and without the Dolby C noise reduction system. Note some 3dB higher noise level at 3kHz relative to the "virgin" tape



The basic tape noise is due to random orientation of the small magnetic particles in the tape causing a varying flux through the replay head. Generally the higher the density of particles, the less the flux variation and hence the less is the noise.

The steep increase in low frequency noise, seen clearly in Figs. 19a to 19d and analyzed in Fig. 20 mainly reflects the (lack of) smoothness of the tape surface. Each time a "thicker" particle in the surface passes the replay head it lifts the tape somewhat from the head causing a so-called "drop out". With no signal or a DC signal recorded on tape, this so called "DC modulation noise" or "surface noise" occurs.

With a pure sine wave recorded on the tape similar phenomena distort the reproduction, also causing modulation noise which is seen as "skirts" on the spectrum of the replayed sine wave (which ideally should consist of only one vertical line). See Fig. 21.

Wow and flutter also cause sidebands in the spectrum of a sine wave. However, these sidebands occur as discrete modulation frequencies at-



tributable to different rotation speeds of skew rotating parts associated with the tape transport. A faulty pressure roll can, for example, be identified by identification of the associated modulation line and possibly by its harmonics in the wow and flutter spectrum.

Noise distributions, drop-outs and other effects such as drop-ins and amplitude compression on tape can also be analyzed employing the statistical mode of the Analyzers as shown in Fig. 22.

| a. | 6 840 | 326 |
|------------------------------------------|---------------------------------------------------------------------------------------|-----|
| CH.A: CH.B: GENERATOR: | 1V + 3Hz DIR FILT:25.6kHz 1V/V 400mV + 3Hz DIR FILT:25.6kHz 1V/V REFERENCE SINE | |
| FREQ SPAN: CENTER FREQ: WEIGHTING, | 200Hz ^ ∆F:250mHz T:4s ∆T:3.91ms Z00M 3152Hz HANNING | |
| AVERAGING | LIN 100 OVERLAP: 0% | |
| MEASUREMENT: TRIGGER: | CH B SPECTRUM AVERAGING FREE RUN | |
| SETUP W2 | MODULATION NOISE AND WOW AND FLUTTER. SA-X CROS | 2 |

Fig. 21. Modulation noise and wow and flutter

- (a) A 3152Hz sine wave is recorded and the Analyzer "zoomed" for closer view of sidebands on the replayed signal. "GENERATOR: REFERENCE SINE" automatically gives a signal centred in the "zoom" range
 (b) Discrete components are due to wow & flutter. The
- more broadband "skirt" is modulation noise





Fig. 22. The Analyzer in its statistical mode

Harmonic Distortion

The 80 dB and 25,6 kHz frequency ranges of the Analyzers are well-suited for measurements of harmonic distortion.

The high frequency bias signal which is fed to the recording head together with the low frequency signal to be recorded, greatly linearizes the recording process. Adjusting the bias level, as is often the case, to obtain a maximum total output level with a recorded sine wave does **not**, however, yield an optimum recording. Viewing **both** the fundamental and the distortion components as the bias level is adjusted, a noticeable reduction in distortion will be seen at a bias level slightly lower than that for maximum output (Fig. 23b). This is the optimum bias level. At even lower bias levels, the distortion components increase once more.

Once the bias has been adjusted for minimum harmonic distortion, the equalization network may be adjusted to obtain a flat frequency response. As previously described, this is most easily accomplished using broadband excitation to monitor the total response while the adjustment is made. (For a "dynamic" display, *exponential* averaging of consecutive results is used on Types 2032 and 2034.) In theory^[2] and also in practice, magnetic recording exhibits *odd* harmonic distortion products only, with a predominant 3rd harmonic. Even order components do not exist in principle since the magnetic transfer curve (magnetic field strength versus remanence) is symmetrical. If a second harmonic component appears, something is amiss in the form of a DC component: magnetized heads or magnetized tape guides, leakage of DC-current into the recording head (faulty coupling capacitor), amplifier distortion or asymmetrical AC-bias waveform^[3].



Fig. 23. (a) The built-in sine generator of Types 2032 and 2034 is of good quality. Its distortion products lie generally below -60dB, which is sufficient for measurements on most tape recorders*
 (b) At optimum bias level the predominant 3rd harmonic distortion component equals some -35,7dB ~ 1,6%. TDK SA-X (CrO₂)

- b) At optimum bias level the predominant 3° narmonic distortion component equals some $-35,7aB \sim 1,6\%$. TDK SA-A (CrO₂) tape was used
- Fig. 24. For distortion and flutter measurements the Analyzer is equipped with special "HARMONIC" and "SIDEBAND" cursor functions



^{*} An external, higher quality sine generator may be employed to provide full utilization of the 80 dB dynamic range of the Analyzers. At frequencies above 12 kHz the harmonic distortion can no longer be measured. Twin tone difference frequency distortion can then be employed, using both the external sine generator and the built-in generator of Type 2032/34

Further Practical Considerations

Noise Reduction (NR) Systems

In this Application Note we have primarily been concerned with the evaluation and calibration of the magnetic recording-replay process as such and therefore any noise reduction (NR) system has, in the first instance, been deliberately omitted from the signal path (with the exception of Figs. 18 and 19). As a natural second stage in the total evaluation of a tape recorder, a NR system can be introduced to the signal path and the previously described measurements repeated. These measurements will provide a straightforward measure of the influence of NR at different levels and for different types of signals. In a few cases, especially when using impulse excitation, it can be found that nonlinear effects of NR (attack and decay times, limiting) will dominate. These considerations will not be pursued here.

Tape Recorders with a Combined Record/Replay Head

Only the measurements of tape speed drift and movement of the recording zone described in this note fully require a 3-head tape recorder and thus are not applicable for a tape recorder with a combined record/replay head. These measurements are largely dependent on the exact timing of the generator (channel A) and replayed (channel B) signals.

For a tape recorder with combined head, a good impression of the appearance of the average phase can be measured by first recording a series of impulses on tape using the instrumentation set-up shown in Fig.4 (including the low-pass filter). During the replay of this tape, the low-pass filtered generator signal should be fed to channel A of the Analyzer. This signal represents the originally recorded impulse and its phase will be subtracted from the phase of the replayed impulse in channel B when a H_1 calculation is performed.

Unless an extremely well-trimmed tape transportation is involved, the repetition rate of the impulses in the two channels will not be identical and thus averaging cannot be performed. Reasonable, although noisy, results can be obtained, however, using only one impulse.

Summary

A large part of the advantage of using Types 2032 or 2034 Dual Channel Signal Analyzers lies in the very userfriendly display of results and the convenience with which a comprehensive range of measurements can be carried out to obtain detailed knowledge about the tested tape recorder and tape. In addition, the dual channel FFT method for measuring the phase response offers performance which is uniquely superior to most other techniques.

Although the measurements described in this note have been performed on a high quality audio-type cassette recorder, the majority of the application possibilities, of course, apply equally well to tape recorders intended for other purposes within the 0-25 kHz range: studio-type multitracks, "Walkmen", dictaphones and instrumentation tape recorders (AM and FM).

References

 Jørgensen, F., "The Complete Handbook of Magnetic Recording", TAB Books Inc., Blue Ridge Summit, PA., USA, 1980. p. 222

[2] ibid., p. 320

[3] ibid., p.324



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