APPLICATION OF THE FAST FOURIER TRANSFORM TO DIGITAL AUDIO ELECTRICAL AND ACOUSTICAL MEASUREMENT TECHNIQUES

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APPLICATION OF THE FAST FOURIER TRANSFORM TO DIGITAL AUDIO ELECTRICAL AND ACOUSTICAL MEASUREMENT TECHNIQUES.

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0. ABSTRACT

This paper discusses applications of single and dual channel FFT analyzers to evaluate digital audio devices, such as Compact Disc players and PCM audio processors.

With an appropriate choice of excitation and FFT instrument a very detailed analysis of the analog output signal is possible and surprisingly easy to perform. The information obtained in most cases allows the user to diagnose the function of both digital and analog circuitry involved with the least possible interaction with internal hardware.

1. INTRODUCTION

It is not immediately obvious that an off-the-shelf FFT-analyzer can be used in a straightforward way for very detailed analysis of Digital Audio products. Some initial points of doubt one has to consider are for example:

a) Today's FFT analyzers employ 12 bit quantization. Isn't that too crude for looking at 14-16 bits Digital Audio devices?

b) Will looking at one sampling device with another sampling device cause problems due to different sampling rates?

c) Looking at anti-image and anti-alias filters found in Digital Audio products with a device that in itself incorporates a 20 or 25.6 kHz anti-alias filter (not necessarily any better than the one in the measured device) may seem somewhat dubious.

As we shall discuss in the following, these objections are rather easily overcome. Numerous practical measurement examples, theoretical results regarding quantization distortion and optimal FFT sampling rates and suggestions regarding improved test signals on Compact Discs and optimal FFT-based instrumentation are presented.

2.1 TIME VS. AMPLITUDE RESOLUTION - DITHER CASE

The essence of applying FFT-analyzers to Digital Audio lies in the fact that they operate on blocks of time-data rather than individual samples. This possibility of exchanging listening (measuring) time for amplitude resolution is already well known in Digital Audio from the use of dither [1]. The function of dither can be summarized in the following way:

> "If the listener is given enough listening time and the relevant information is present - even at a level below 1 LSB - during all that time, he (i.e. the listener or a measuring device) will - by averaging - be able to recover the information from (dither) noise."

In other words by randomizing the quantization error, dither allows accurate recovery of the original analog waveform, if given sufficient time and consecutive signal repetitions.

If we step aside for a moment from the main scope of this paper, there is a very important, "BUT..." to the dither's function that we would like to highlight here:

Obviously time averaging and hence dither work only for signals that are repetitive (periodic) and sustain in time. These kinds of signals (typically sinusoids) are relevant and easy to produce as test signals. "Real life" program material (music) is, however, often of quite a different breed. A singer following, say, a guitar solo within a few milliseconds is quite a dramatic change in waveforms. The listener's possibility to "lock on to" and time average these signals is only limited to a few hundred samples. Dither or no dither, quantization artifacts on transients remain a "staircase distortion" rather than average out as harmless noise.

In particular, this may offer an explanation for the often expressed impression that digital recordings, even in mono, render a subjectively more "dead" room sound than an equivalent analog recording. If we concentrate on the mono case, so that we can disregard the possible differences in stereo channel separation and cross channel mixing, we may imply that the room information is then encoded in the reproduction of reverberation "tails" following each orchestra chord. These relatively short, low level "tails" will become highly distorted in a digital quantization process. In opposition to this an analog recording yields low level signals virtually distortion free. Although the broadband noise floor is typically higher than for a digital recording medium, the "analog medium" noise is uncorrelated to the signal hence presumably allowing the human brain to track the reverberation "tails" all the way down to insignificance. The incurable digital quantization distortion presumably inhibits this tracking process at an earlier point. Consequently it makes sense to assume that the brain interpretes this as a shorter reverberation time, i.e. the ambience is perceived more "dead", the acoustic surroundings appear smaller.

2.2 TIME VS. AMPLITUDE RESOLUTION - FFT-CASE

A 16 bit linear digital encoding has a dynamic range between the highest and the smallest resolvable sine amplitudes of 90.3 dB.

For a typical 12-bit FFT analyzer, which in the following has been applied to the analog output of various digital audio devices, the theoretical dynamic range is correspondingly 66.2 dB. Too little? Not necessarily, because:

The analyzer operates with blocks containing from 1,024 to 10,240 time samples and the Fourier Transformation involves averaging (weighted integration) over all these values. This on the whole provides a resolution much better than inherently present in each individual sample and is the reason for the fact that 12-bit FFT analyzers typically display spectra over an 80 dB span. Most of the limitation lies in the accuracy of the averaging routines and correctly applied dither (at the FFT input) rather than the 12 bit quantization itself.

90 dB is still more than 80 dB. But where the 16 bits of digital audio are locked to cover a fixed analog voltage range, the 12 bits of the FFT analysis can via the input attenuators be "zoomed" to provide detailed analysis of fine structure of the analog output voltage. When the dynamic span between a test tone and its distortion products becomes too wide the full power of FFT analysis can be aimed at the distortion products by employing an analog notch filter in front of the FFT. With a typical 60 dBs attenuation of the test tone, FFT analysis of distortion down to a comfortable -140 dB can be performed.

It must be noted that since a 12 bit ADC of the FFT analyzer can be manufactured far more linear than 14 or 16 bits DACs & ADCs of Digital Audio and since an analog notch filter is not capable of creating non-harmonically related distortion, the obtained distortion pictures will be fully descriptive of the digital device under test and NOT the measuring chain. The following measurements on an oversampling (Unit D) and a non-oversampling (Unit A) compact disc players will illustrate this.

2. TEST TONE FREQUENCIES

In order to get the optimum amplitude/frequency resolution out of an FFT time record consisting of 1,024 samples, all of these samples should be equally employed in the Fourier transformation. In other words this means that a rectangular time weighting (a.k.a "Flat" or "No Weighting") should be employed rather than the generally used Hanning weighting.

To users of FFT analyzers it is well known that use of the rectangular time weighting is only permissible for a limited range of signals, namely those that only contain frequencies exactly coinciding with frequency lines of FFT analysis. Since for a given FFT analyzer $\Delta f = \frac{1}{T}$, where Δf is the spacing between frequency lines and T is the time record length in seconds, the preferred test tone frequencies are Δf, $2\Delta f$, $3\Delta f$, ..., $n\Delta f$.. etc.

Let us consider what this means in terms of practically available CD test discs and FFT analyzers:

Typical test frequencies found on test disc are 20 Hz, 1 kHz and 16 kHz.

A 20 kHz 400 lines FFT analyzer, like for example Bruel and Kjaer Type 2033 that has been employed in the following, is hence with a Δf of 50 Hz perfectly applicable to all 3 of these frequencies.

On the other hand a 25.6 kHz 800 lines FFT analyzer, like for example Bruel and Kjaer type 2032 that also has been employed in the following, however, for other measurement types, with its Δf of 32 Hz doesn't contain a 20 Hz nor a 1 kHz line in its baseband mode.

By resorting to Hanning weighting when rectangular weighting isn't possible, the effective frequency resolution is compromised from equaling the line spacing Δf to approx. $1.5\Delta f$ [2]. Using a type 2032 one could hence perform analysis of 20 Hz or 1 kHz tones with the same resolution of $1.5 \times 32 = 50$ Hz as with type 2033 and the 16 kHz tone can be analyzed with $\Delta f = 32$ Hz.

The remarks above apply to the analyzers operating in the so called "Baseband" mode from 0 Hz to Full Scale Frequency. When operated in Zoom mode frequency resolution in the fraction of a Hertz range is achievable, at the expense of longer measuring time.

Fig. 1. A - 80 dB 1 kHz test tone replayed by a CD-player and analyzed 0-20 kHz with a Flat weighting.

Fig. 2. "Philips Test Sample" test disc contains test tone at prime number - frequencies, here: 997 Hz. For analysis of such frequencies non-flat window types as for example Kaiser-Bessel, Flat-Top or like here, Hanning, must be employed. Compare to Fig. 1. and note the loss of frequency resolution.

3. ANTI-ALIAS FILTER OF THE FFT

If the anti-alias and the anti-image filters of Digital Audio do their job of bandlimiting well, it is advisable to bypass the input anti-alias filter of the FFT analyzer when analyzing with a bandwidth of 20 kHz or more.

This will allow the greatest accuracy in analysis of amplitude and phase relationships. Should, however, the digital test object leak out spurious frequencies above 20 kHz, switching of the FFT's anti-alias filter in and out can be used as a simple way of detecting such a leakage. See figs. 3-8.

Fig. 3. The noise floor of an oversampling Compact Disc Player with de-emphasis filter OFF. The anti-alias filter of the FFT-analyzer is IN.

The noise floor level is NOT -125 dB as could be implied by uncritical number reading. Remember that this broadband noise is measured here with a 50 Hz bandwidth (Type 2033, 20 kHz span, 400 lines, Flat weighting).

The total 0-20 kHz noise rms value is hence
$-125 \text{ dB} + 20 \log \sqrt{400} = -125 \text{ dB} + 26 \text{ dB} = -99 \text{ dB}$

Fig. 4. Same as Fig. 3. but with the anti-alias filter of the FFT analyzer OFF. The "5.6 kHz" component is in fact the second harmonic of the oversampling frequency 176.4 kHz, i.e. 352.8 kHz.

5.6 kHz equals 352.8 kHz - 7x(51.2 kHz), where 51.2 kHz is the sampling rate of the analyzer. The amplitude measured is affected by the specifications of the analog front end circuitry of the FFT above 20 kHz. Please note also that only frequencies lying within ±20 kHz (or whatever other bandwidth of FFT analysis is employed) from FFT's sampling frequency or it's harmonics will become "visible" due to aliasing.

Fig. 5. & 6. 1 kHz tone at 0 and -10 dB relative to full scale respectively, analyzed without FFT's anti-alias filter. All spurious components except the one at "10.25 kHz" "track" the -10 dB fall in level of the 1 kHz tone. This indicates that they are intermodulation products between 1 kHz and some spurious frequencies from the digital signal treatment in the player. The "10.25 kHz" component, which is also seen on Fig. 4., is however signal independent and must be created by some other mechanism, presumably the tracking servo.

Fig. 7. & 8. Same as Figs. 3 and 4 however for the non-oversampling CD-player. 14700 Hz is one third of the sampling frequency and is used for frame merging inside the player (1 frame = 6 left + 6 right samples). The origin of the other leaking frequency of 13240 Hz in Fig. 7 is more obscure.

Most evident in Fig. 8 is the aliasing of the sampling frequency of 44.1 kHz at "7100 Hz" (=51200 - 44100 Hz).

4. SIGNAL-TO-QUANTIZATION NOISE RATIO?

The theoretical formula for Signal to Quantization Noise Ratio for an N-bit linear PCM system is well known (see f. ex. [3]):

$$\text{SNR (dB)} = 6.02 \cdot N + 1.76$$

A practical measurement of the quantization noise level is however not an easy task:

If there is no signal there will be no quantization noise!

[Obviously figures 3 and 7 do not represent quantization noise. Such measurements are however commonly used for obtaining a "Signal-To-Noise Ratio" for CD-players and "compared" to the theoretical value above.]

The preferred test signal: a sine wave, is not applicable since it does not induce random quantization noise but a very deterministic harmonic (quantization) distortion.

Add dither to the sine wave and by notching down the sine wave at the output it is now possible to measure the sum of dither-plus quantization noise rms value.

This is probably the most reasonable way, since dither should be an inherent part of all digital audio. When it is there one often does the test without the sine wave, letting the dither-noise generate a one-bit random oscillation of the DAC. This test is however of questionable value also since in a practical DAC or ADC the quantization steps are not equidistant, so looking at just a selection of few adjacent quantization intervals may yield a result, nonrepresentative of the full range of the DAC/ADC.

5. THEORETICAL DISTORTION IN AN IDEAL LINEAR 16 BIT PCM SYSTEM

The steady state distortion in digital audio results from one of the following four mechanisms:

Type 1 "High Level Distortion"	Which is clipping due to overload of either the digital or the analog paths.
Type 2 "Medium Level Distortion"	Which is distortion due to nonequidistant quantization steps in the transfer characteristics of ADC and DACs.
Type 3 "Small Level Distortion"	Which is the fact that small level analog signals only use few bits and hence come out as "square waves". This kind of distortion is present at all levels of the input signal. It however becomes more significant relative to smaller signal levels.
Type 4 "Slewing and timing distortion"	Which is due to imperfections in exact timing of the sampling and deviations from the ideal "infinite" transition speed by the DACs.

Whereas the other three types can be minimized by employment of appropriately well functioning hardware, type 3 distortion represents an inherent artifact of the technology itself. It is however to a certain extent curable with dither, as discussed in paragraph 2.1.

Assuming all other things to be ideal, we have simulated how much Type 3 distortion is to be expected on a 1 kHz tone sampled with 16 bits at 44.1 kHz. The calculation was performed on an HP 15-C pocket calculator (!) using the Discrete Fourier Transform formula:

$$S(k \times 1\ \mathrm{kHz}) = \frac{1}{441} \sum_{n=0}^{440} y(t_n) \exp\left(-j \frac{2\pi \times n \times k}{441}\right)$$

With a 0 dB y(t) represented as $32767 \times \sin(2\pi \times 1000 \times t)$ the effects of ideal quantization are simulated simply using "Integer part" function of the HP 15-C. Dither can be added using the "Random #" function. 441 point Fourier Transform is employed in order to avoid windowing problems. The results are shown in table 1.

Most striking in table 1 is the way in which the level of the fundamental after quantization becomes less and less faithful to the original level of the recorded sine wave for small signals. Since "room ambience" or reverbertion, as discussed in paragraph 2.1 is heard as a "small" signal in-between the high level, direct sound signal, we see that quantization will increase the dynamic span between the two in favor of the high level, direct sound. So, once again, we have an argument for the digital recording sounding less "ambient" than an analog recording.

Also at this point we should note that, -60, -70, -80 dB or -90 dB 1 kHz (level of the fundamental) signals found on many test CDs obviously must be encoded using a "master" sine wave of a successively higher amplitude than the nominal. This means that while an 0 dB test track can only be obtained by digitizing an 0 dB sine wave, a -90 dB test track can be obtained from a range of sine waves anywhere between -85 and -90 dB by different combinations of phase and DC-offset relationships (see paragraph 6) resulting in quite different distortion spectra. Figs. 9. and 10. illustrate this ambiguity in available -90 dB "test" signals. "Computer generated to an accuracy of 99.999...%, as it states on many test discs, indeed doesn't mean identical!

Fig. 9. Spectrum of the "-90 dB test tone" of "Super Audio check CD". Replayed by an analog filter CD-player.

Fig. 10. Same as Fig. 9. except the test disc is now "SONY 3". A much better distortion characteristics of this disc over the previous one is evident. It is possible that some of the improvement is due to use of dither on this record. However, no mentioning of such is found on the sleeve of the record.

We often see test reports on CD-players in different publications displaying the FFT spectrum of -60, -80, or -90 dB test tones. At these test tone levels the tone itself and its distortion components are easily accommodated and displayed by an FFT analyzer without the need of a notch filter at the front end. However, based on the discussion above, we feel it is advisable to go through the trouble of using a notch filter, so that distortion measurement of tones at 0 dB is performed. These tones are much more consistent on different makes of test discs. At lower levels results from different CD-player are only comparable if performed with the same test disc make.

Another valid point is that an 0 dB tone exercises all 65,536 quantization levels available, whereas, a -60 dB tone only goes through 65 of them and a -90 dB tone only 3-4!

TABLE 1. THEORETICAL QUANTIZATION DISTORTION OF A 1 kHz SINE WAVE

Sampled with 16 bits at 44.1 kHz

All levels in dBs ref. maximum amplitude sine wave of 32767 x LSB

Signal	Fundamental	2. harm	3. harm	4. harm	5. harm	6. harm	7. harm	THD
0 dB	- 0.01	-119.08	-104.11	-130.56	-108.27	-133.45	-109.38	1.5 10^{-5}
-60 dB	-60.18	-129.78	-104.95	-128.29	-106.41	-122.03	-113.06	1%
-80 dB	-81.58	-136.17	-100.34	-133.90	-123.40	-136.59	-105.94	19%
-84 dB	-87.15	-134.67	-101.80	-163.13	-118.62	-136.13	- 99.51	46%
-90 dB	-99.89	-143.14	-100.72	-136.39	-102.47	-137.18	-105.44	222%

6. FURTHER RESULTS FROM THE DISTORTION MODEL

Table 2 lists some other results obtained from our HP 15-C simulation. Line A shows a test run without any quantization, i.e. it is just a check of the numerical accuracy of the Fourier Transform employed. It is satisfactory.

In line B, using the "Random #" function, we have simulated the effect of adding dither. As anticipated its effect on just 441 samples is negligible. Many more samples (i.e. a longer "time record") would be needed to see the distortion-reducing effect of dither.

In line C, we removed dither and changed the function of the simulated analog-to-digital converter to be symmetrical, i.e. rather than converting the analog interval [0;1[into quantization value 0, it is now converting [-1/2; 1/2[into 0. Somewhat surprisingly this increases the THD from 19% to 24%.

For the -90 dB tone in line D on the other hand, a symmetrical ADC improves the performance quite drastically from 222% to 112%.

TABLE 2.

	Signal	Fundamental	2. harm	3. harm	4. harm	5. harm	6. harm	7. harm	THD
A	-80dB no quantization	-80.00	-265.40	-260.74	-267.64	-240.76	-258.97	-251.48	$15\ 10^{-8}$
B	-80dB dither added	-81.82	-111.13	-106.14	-112.23	-113.86	-121.28	-113.76	19%
C	-80dB symmetrical ADC	-82.05	-110.66	-109.52	-111.07	-104.91	-111.78	-117.90	24%
D	-90dB symmetrical ADC	-95.38	-101.73	-127.64	-107.13	-110.06	-127.66	-111.75	112%

7. PRACTICAL DISTORTION MEASUREMENTS ON TWO CD-PLAYERS

Using a tunable notch filter (B&K Type 2120) followed by a 400 line, 20 kHz FFT analyzer (B&K Type 2033) we have investigated 0 dB distortion of two "first generation" CD players: Unit A and Unit D.

The Unit A player employs a conventional analog anti-image filter, whereas the Unit D player utilizes oversampling and digital filtering.

The anti-alias filter of 2033 was ON in all the following measurements.

Fig. 12. This is the -60 dB, 1 kHz test tone replayed by Unit A and analyzed without a notch filter.

Fig. 13. 0 dB, 1 kHz test tone replayed by Unit A. The notch filter provides 68.8 dB attenuation of the 0 dB fundamental. Note that presence of the notch filter raises up the noise floor of the measuring situation by some 10 dB relative to fig. 12. This filter is hence too noisy for measurement of quantization noise as discussed in paragraph 4, but provides us with sufficient resolution for distortion measurements.

Fig. 14. Same test tone as in fig. 13., but replayed by Unit D. Except for the 3rd harmonic there is noticeably more high order distortion than in fig. 13.

Fig. 15. This is the difference between fig. 13. and 14., i.e. between an analog filter and a digital filter unit, displayed using the "Instant divided by Memory" - mode of the FFT analyzer.

We can see that the player with the analog filter exhibited just second and third order distortion, whereas, the oversampling unit created a multitude of harmonic components with the second and eighth through tenth as the dominating plus a range of non-harmonically related discreet frequencies in the level range -94 to -108 dB. This distortion picture is indicative of "crosstalk" between the digital and the analog signal treatment circuits via the internal DC-power supply and possibly slew rate limiting in the very fast operating DAC.

This diagnosis complies well with the fact that a British re-make of the here tested oversampling unit has addressed the insulation of the internal DC-power supply with a clearly improved sound as the result.

Although a -90 dB distortion of a sine pure tone is definitely inaudible as such, the measurement results indicate imperfections in the player causing (possibly) audible distortion of more complex signals.

Fig. 16. 0 dB, 20 Hz tone replayed by Unit A.

Fig. 17. 0 dB, 20 Hz tone replayed by Unit D. Again non-oversampling unit performs better.

20 Hz is typically the lowest test frequency found on test discs.

The two CD-players were compared both subjectively and objectively*) using a piece of recorded music containing subsonic frequencies. A measurable difference in the rendition existed only below 20 Hz and "should" hence be inaudible. However the player outputting more subsonic frequencies sounded "fuller" over a pair of headphones. This presumably indicates that the subsonic frequencies although not transmitted by the headphones created distortion components in the headphones in the lower frequency range hence increasing the perceived low frequency - "listening pleasure". Unfortunately no test CD containes a single subsonic test frequency allowing a more accurate estimation of the created distortion.

Fig. 18. 0 dB, 16 kHz test tone causes in the analog filter player a -90.9 dB spurious at 12.1 kHz and a -102.5 dB spurious at 3.9 kHz. These are clearly second and third order harmonic distortion (32 and 48 kHz respectively) created during the digital signal processing and undergone aliasing back into the audio range. They reflect the non-linearity (not equidistant transition steps) in the employed 16 bits DA-converter.

Fig. 19. 0 dB 16 kHz test tone. For the oversampling player, because of a 4 times higher sampling frequency (176.4 kHz) the first 9 distortion components of 16 kHz will not be aliased back into the audio frequency range. This measurement confirms this.

8. MAGNITUDE AND PHASE OF THE FREQUENCY RESPONSE. CD-PLAYERS.

By far the best method of measuring group delay distortion is by FFT analysis. Here one can artificially create a reference spectrum of any test signal and compare this reference to the spectrum of the signal actually replayed by the CD player under test. Any broadband signal can be used, but since the reference spectrum of a single impulse has zero phase and flat magnitude, catching a single one of them will directly give us what we want without further postprocessing. The phase of the complex instantaneous spectrum will reflect the group delay distortion that the impulse has undergone in the filter, and its magnitude will show ripples in the filter passband characteristics.

But alas! No test disc with just a single impulse is available. Many discs contain a series of fast repeated impulses, which is good for providing a stable display on a non-storage oscilloscope but is non-optimal for FFT analysis. Closely spaced impulses - even if one artificially manages to "window out" just one single of them, ring into one another and cause erroneous ripples in the calculated spectrum.

In order to be able to see the roll-off behaviour of the filters above 20 kHz we have used the 25.6 kHz, 800 lines B&K type 2032 FFT Analyzer for the following measurements. Its anti-alias filter was switched off in order not to introduce extra phase and magnitude modification and off course no notch filter etc. was employed.

*) See Appendix on page 14.

Fig. 20. Repeated 0 dB impulses as they come out of Unit A.

Fig. 21. Using the TRANSIENT window of the FFT analyzer to isolate only one impulse, the instantaneous spectrum yields the magnitude and phase of Unit A's analog anti-image filter.

Fig. 22. As fig. 20. but for the oversampling Unit D.

Fig. 23. As fig. 21. but for Unit D.

Where the Unit D is superior to Unit A in terms of phase response, its magnitude response on the other hand is inferior in roll-off steepness. At 23 kHz it provides about 12 dB less attenuation than the analog filter in Unit A.

Also this has been corrected in the British re-make of the oversampling unit. The softly off-rolling Bessel filter at the output stage has been replaced by a sharper Butterworth filter with improved sound quality as a result.

Please note in the figures above that the ripples on the shown frequency responses are due to:

a) Limited time window width (T=2.9 msec gives a $\Delta f \approx \frac{1}{T} \approx$ 350 Hz)

b) Previous impulses on the test record keep on ringing and hence interfere with the impulse which we pick out for spectrum calculation.

Therefore they should not be interpreted as ripples in the passband characteristics of the tested anti-image filters (they are flat within $\pm$0.5 dB).

9. MAGNITUDE AND PHASE OF THE FREQUENCY RESPONSE. RECORD/REPLAY-DEVICES. DUAL CHANNEL FFT MEASUREMENTS.

Whenever we have access to both the analog input and the analog output of a digital record/reproduce device, dual channel FFT methods can be employed.

Using the dual channel Bruel and Kjaer Type 2032 analyzer we have performed such measurements in order to evaluate the function of a Unit F audio processor alone and in conjunction with an analog "phase correction" filter.

As previously discussed, the optimal signal for FFT-analysis is one that only contains frequencies coinciding with the lines of analysis. If all those frequencies are present simultaneously and with equal strength we have one special type of test signal, often employed in dual channel analysis: the so called Pseudo Random Noise. (For a more thorough discussion of different excitation signals pros et cons see for example [4]).

For the 25.6 kHz, 800 lines analyzer employed, the pseudo random noise signal from the built-in signal generator, hence contains frequencies of 32, 64, 96 Hz... etc. all they way up to 25.6 kHz. It is thus repetitive with a period length of 1/32 Hz = 31.25 ms. Any analog test object to which we would apply such a signal would after a while stablize and produce an output signal with exactly the same repetition rate of 31.25 ms. This is however, not the case for a digital test object, like for example Unit F. Since 31.25 ms. does not equal an integer amount of 44.1 kHz sampling periods but exactly 1378 + 1/8 of them, the Unit F will repetitively keep producing 8 different responses to the same input signal!

This effect of "time variance" is similar to the effect of wow and flutter in analog recording and it's effect upon dual channel FFT analysis of such a device is thoroughly discussed in [5]. In short:

a) The coherence will drop and the magnitude of the frequency response will be underestimated with higher frequencies

b) The averaged frequency response will not necessarily converge into a smooth curve, but will keep looking "noisy".

Fig. 24. The measurement setup and the phase response for the digital audio processor.

Fig. 25. The corresponding magnitude of the frequency response. The measurement object is NOT -1.7 dB down at 20.000 Hz as indicated in the upper right corner! This number is an underestimation due to "wow and flutter" - like effects caused by non-optimal matching of the FFT analyzer's bandwidth and the sampling rate of the audio processor.

Fig. 26. The coherence of the above measurement confirms problems at higher frequencies.

Fig. 27. Phase response of the analog Group Delay Corrector. Note how smooth the curve is compared to measurements above on a digital device.

Fig. 28. The combined response of the Group Delay Corrector and the digital audio processor. The phase response is greatly improved throughout most of the audible range.

Pseudo Random noise-type signals provide the best utilization of an FFT analyzer, due to the fact that a rectangular time weighting can be used and hence optimum frequency resolution obtained.

In order to avoid discrepancies between the FFT analyzer and the test object, such as we have encountered above, a very simple requirement can be imposed on the sampling rate of the FFT analyzer:

<u>One FFT time record must contain an integer amount of digital audio sampling periods</u>.

Based on this simple requirement we can hence "design" an optimal FFT analyzer aiming particularly at analyzing both "44.1 kHz" and "48 kHz" digital audio:

The necessary numbers are:

$44100 = (2 \times 3 \times 5 \times 7)^2$ $\qquad$ $48000 = 2^7 \times 3 \times 5^3$

and if we want an 800 lines frequency resolution we need 2048 samples ($=2^{11}$) in each FFT time record. It is easily shown that only four FFT sampling rates satisfy the compatibility criterion above. They are:

f_1: $2^{13} \times 3 = 24,576$ Hz (too low for practical application)

f_2: $2^{13} \times 5 = 40,960$ Hz (too low)

f_3: $2^{13} \times 3 \times 5 = 122{,}800$ Hz (better)

f_4: $2^{13} \times 5 \times 5 = 204{,}800$ (best)

It is our suggestion that an optimal FFT based analysis of "44.1 kHz" and "48 kHz" digital audio should be based on an FFT sampling rate of 204800 Hz or 122800 Hz. To the best of our knowledge, at present date none of the commercially available FFT analyzers utilize any of these sampling frequencies. However, by using an external sampling frequency source many of them can be controlled to do it.

10. CONCLUSION

We have demonstrated and discussed methods of applying discreet Fast Fourier Transform to analyze digital audio products.

Based on the many applications of todays FFT analyzers to digital audio, optimal future instrumentation for this purpose can be designed and an improved understanding of the audible effects of digitizing audio signals gained.

APPENDIX

The objective comparison of the two CD units using music was done in the following manner:

We connected a pair of 600 ohms headphones in parallel with a type 2033 FFT analyzer (1 Megaohm input resistance) to the line outputs of the unit under test. The analyzer was hence monitoring the electrical signal going to the headphones, using which we conducted a subjective comparison. The piece of music we used was "Satin Doll", track 16 of "Super Audio Check CD".

Fig. 29. Average of 20 consecutive music spectra as replayed by Unit A. 0 to 100 Hz.

Fig. 30. Average of 20 consecutive music spectra as replayed by Unit D. 0 to 100 Hz.

Fig. 31. The difference between Unit A and Unit D (fig. 29 - fig. 30) is most obvious in the subsonic range. The combination of 600 ohms load and 22 uF output capacitance of unit D yield a -3 dB point at 12 Hz.

Fig. 32. Connection of a 600 ohms load (i.e. the headphones) directly to the line outputs was convenient in order to obtain the same listening level in the headphones, as well as when using type 2033 in triggered mode to ascertain that the triggering takes place at same instants in time in both cases. 600 ohms is, however, on the low side of allowable load. The removal of the headphones, as seen here, yields the two units virtually identical, -also in the subsonic range.

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4) "Dual Channel FFT Analysis Parts I and II", H. Herlufsen
Bruel and Kjaer Technical Review nos. 1 and 2 - 1984

5) "Dual Channel FFT Analysis for the Development and Evaluation of Tape Recorders", Andre Perman
Bruel and Kjaer Application Note BO 0098-12

Brüel & Kjær

Time Function Start: seconds End: seconds Not Expanded: Expanded:

Full Scale Level: -50 dB

F. S. Frequency: 20 kHz

Weighting: Flat

Average Mode: Linear

No. of Spectra: 40

Comments:

1 kHz, -80 dB

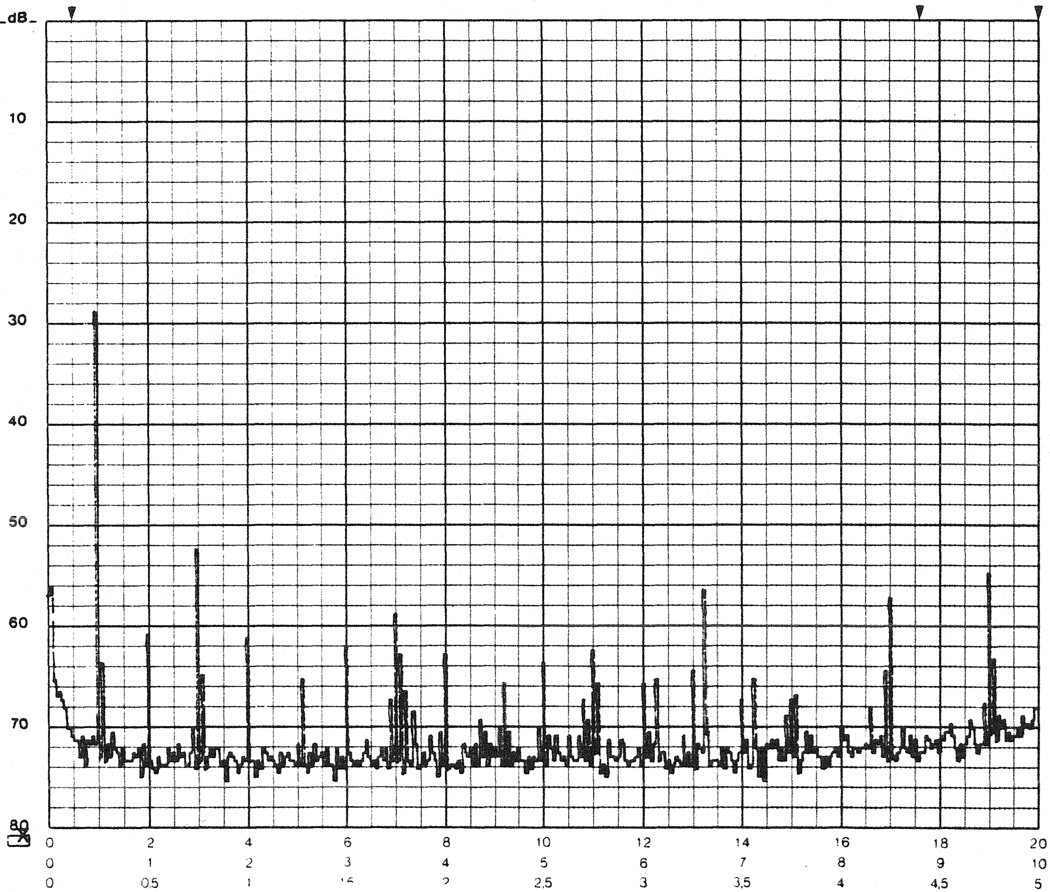
CD: "Super Audio Check" CBS/SONY

Player: Unit A

Record No.: 1

Date: 11-21-1983

Sign.: AP



QP 1002

Fig. 1

Measuring Object: Unit A

Brüel & Kjær Time Function Start: seconds End: seconds Not Expanded: Expanded:

Full Scale Level: -50 dB dB

F. S. Frequency: 20 kHz

Weighting: Hanning

Average Mode: Linear

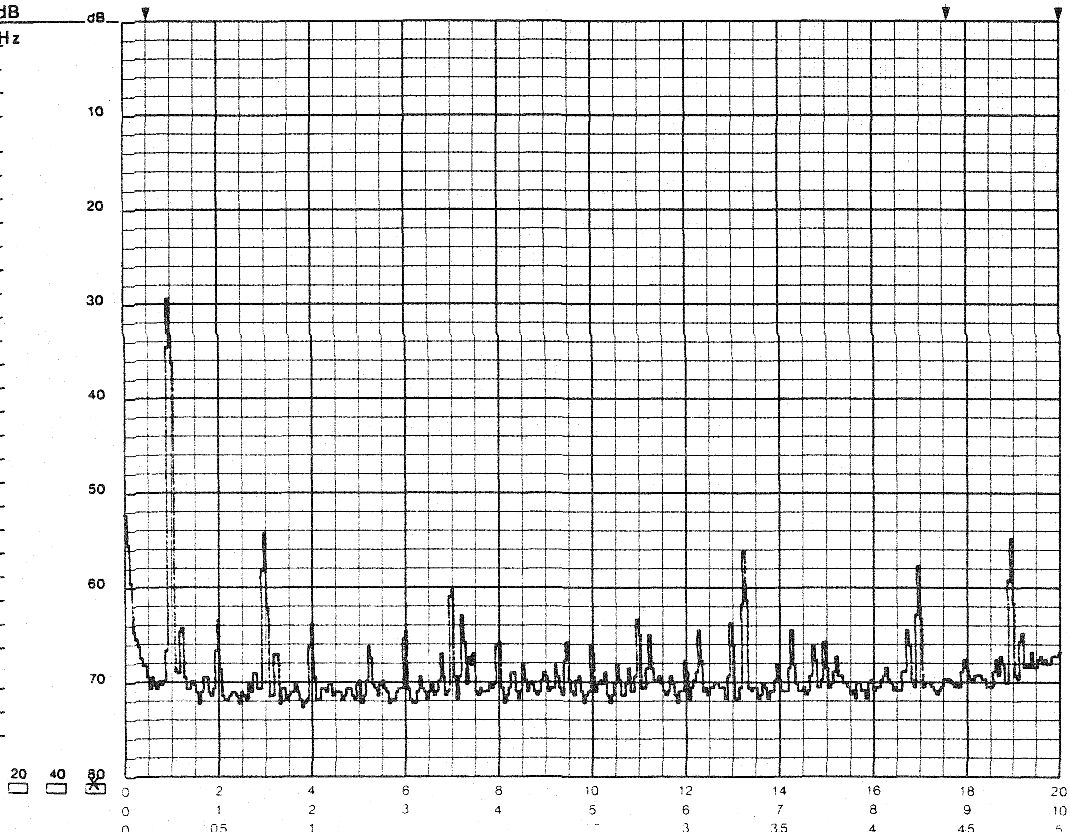
No. of Spectra: 57

Comments:

997 Hz, -80 dB

CD: "Philips Test Sample"

Player: Unit A



Record No.: 2

Date: 11-21-1983

Sign.: AP

Measuring Object: Unit A

Fig. 2

QP 1002

Brüel & Kjær

Time Function Start: seconds End: seconds Not Expanded: Expanded:

Full Scale Level: –50 dB

F.S. Frequency: 20 kHz

Weighting: FLAT

Average Mode: LIN

No. of Spectra: 128

Comments:

" Infinity zero without emphasis"

CD: Sony test 3, track 23

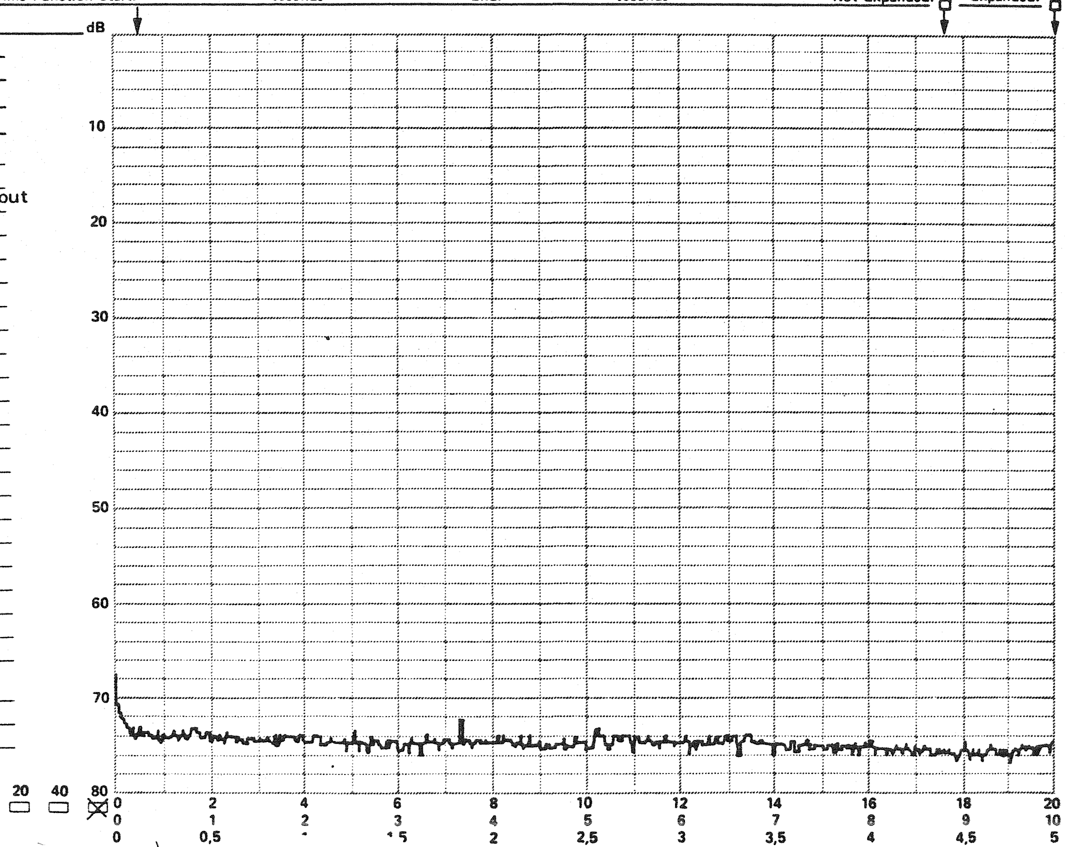
Player: Unit D

2033 anti-alias filter IN

Record No.: 3

Date: 11-23-1983

Sign.: AP



QP 1002

Fig. 3

Measuring Object: Unit D

850491

Brüel & Kjær

Time Function Start: seconds End: seconds Not Expanded: Expanded:

Full Scale Level: -50 dB

F. S. Frequency: 20 kHz

Weighting: Flat

Average Mode: Linear

No. of Spectra: 128

Comments:

"Infinity zero without emphasis"

CD: Sony test 3, track 23

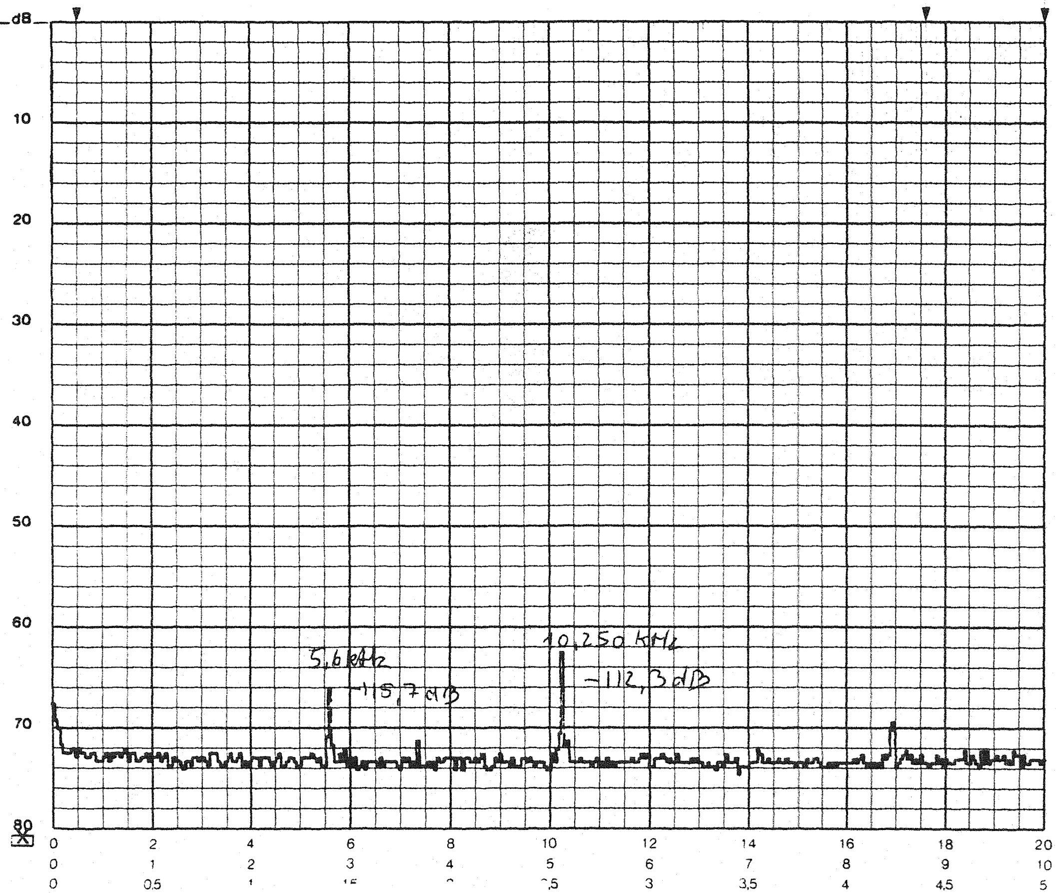
Player: Unit D

2033 anti-alias filter OUT

Record No.: 4

Date: 11-23-1983

Sign.: AP



20 40 80

Fig. 4

Measuring Object: Unit D

QP 1002

Brüel & Kjær

Time Function Start: seconds End: seconds Not Expanded: Expanded

Full Scale Level: 0 dB

F. S. Frequency: 20 kHz

Weighting: Flat

Average Mode: Linear

No. of Spectra: 128

Comments:

1 kHz, 0 dB

CD: Sony test 3

Player: Unit D

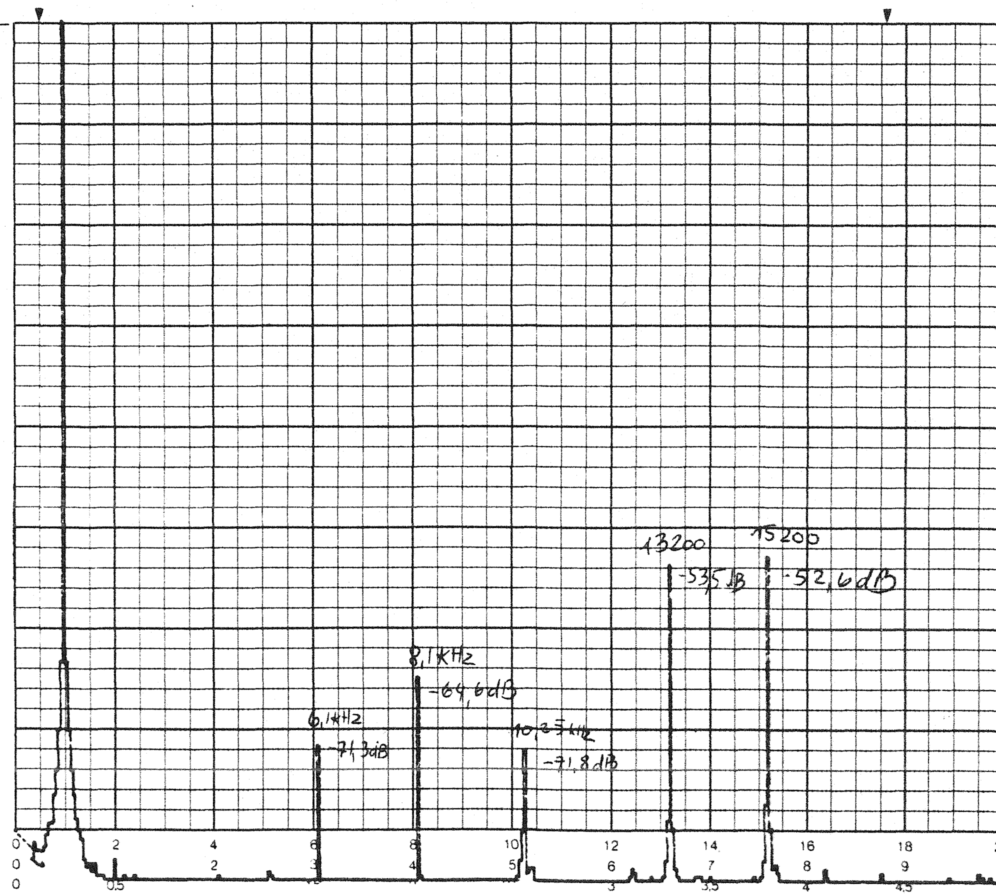
2033 anti-alias filter OFF

NO notch filter

Record No.: 5

Date: 11-23-1983

Sign.: AP



20 40 80

QP 1002

Fig. 5

Measuring Object: Unit D

Brüel & Kjær

Time Function St... seconds End seconds Not Expanded Expanded

Full Scale Level: 0 dB dB

F. S. Frequency: 20 kHz

Weighting: Flat

Average Mode: Linear

No. of Spectra: 128

Comments:

1 kHz, -10 dB

CD: Sony test 3

Player: Unit D

2033 anti-alias filter OFF

NO notch filter

Record No.: 6

Date: 11-23-1983

Sign.: AP

20 40 80

Fig. 6

Measuring Object: Unit D

QP 1002

Brüel & Kjær

Time Function Start: seconds End: seconds Not Expanded: Expanded:

Full Scale Level: –50 dB = 890 μVolt

F.S. Frequency: 20 kHz

Weighting: Flat

Average Mode: Lin 128

No. of Spectra:

Comments:
"Infinity zero without emphasis"

CD: Sony 3

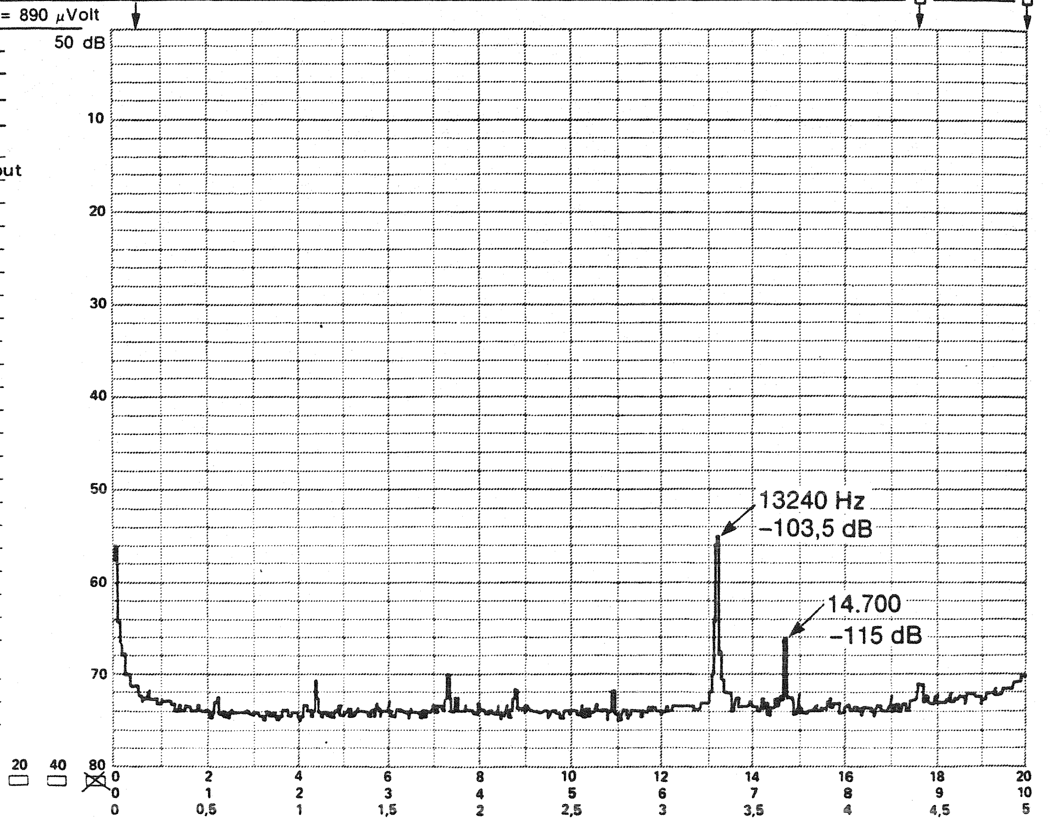
Player: Unit A

2033 anti-alias filter IN

Record No.: 7

Date: 11-21-1983

Sign.: AP



QP 1002

Fig. 7

Measuring Object: Unit A

850493

Brüel & Kjær

Time Function Start: seconds End: seconds Not Expanded: Expanded:

Full Scale Level: -50 dB

F. S. Frequency: 20 kHz

Weighting: Flat

Average Mode: Linear

No. of Spectra: 128

Comments:
"Infinity zero without emphasis"

CD: Sony 3

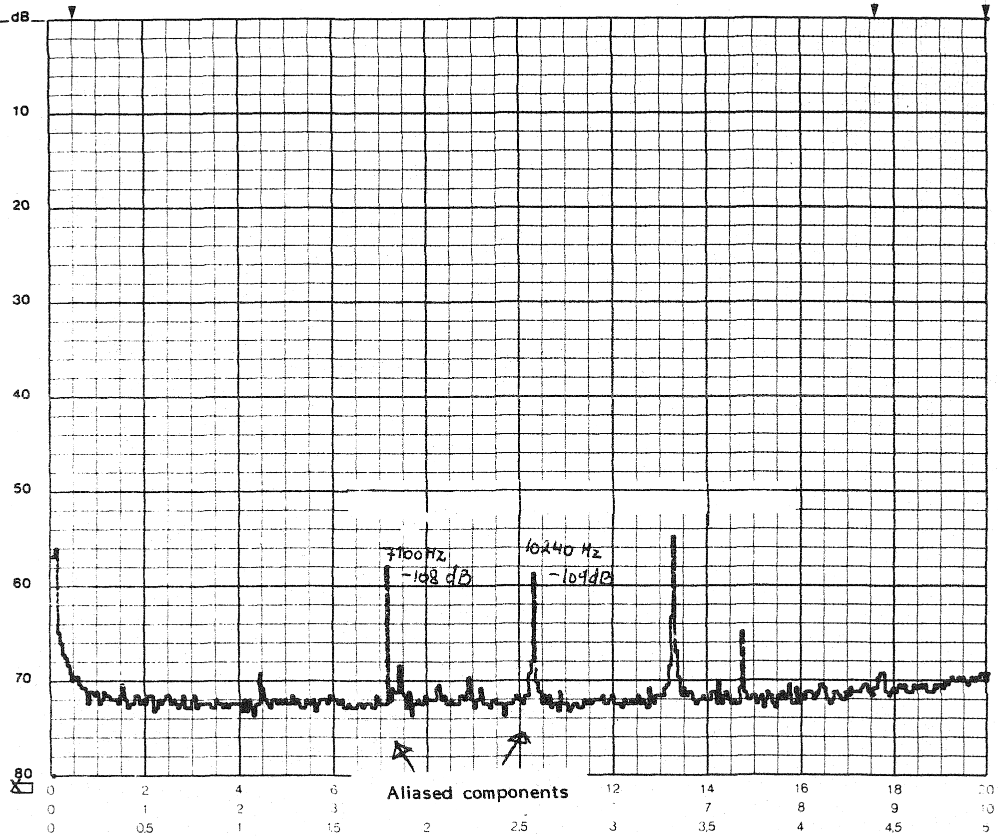
Player: Unit A

2033 anti-alias filter OUT

Record No.: 8

Date: 11-21-1983

Sign.: AP



20 40 80

Fig. 8

QP 1002

Measuring Object: Unit A

Brüel & Kjær

Time Function Start: ______ seconds End: ______ seconds Not Expanded: ☐ Expanded: ☐

Full Scale Level: –50 dB

F.S. Frequency: 20 kHz

Weighting: FLAT

Average Mode: LIN

No. of Spectra: 30

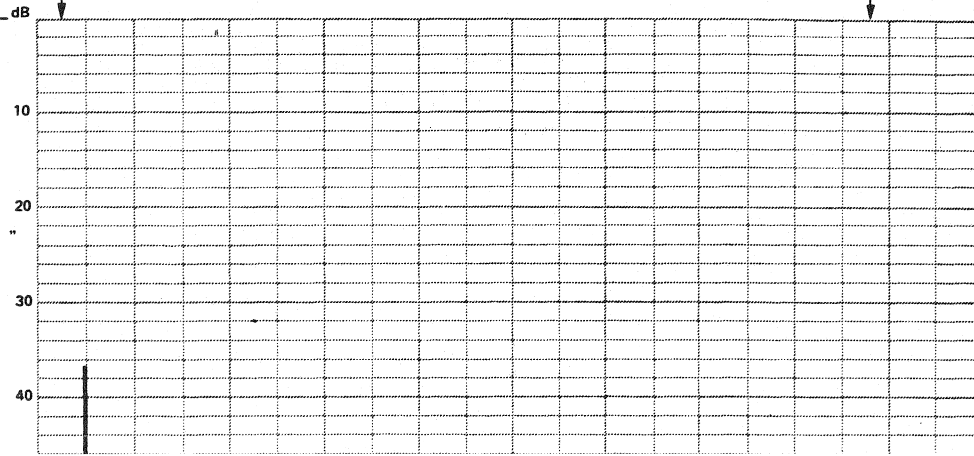
Comments:

1 kHz, -90 dB

CD: Super Audio Check

Player: Unit A

2033 anti-alias
filter IN



Brüel & Kjær

Time Function Start: ______ seconds End: ______ seconds Not Expanded: ☐ Expanded: ☐

Full Scale Level: -50 dB dB

F. S. Frequency: 20 kHz

Weighting: Flat

Average Mode: Linear

No. of Spectra: 128

Comments:

1 kHz, -90 dB

CD: Sony 3

Player: Unit A

2033 anti-alias
filter IN

Record No.: 10

Date: 11-21-1983

Sign.: AP

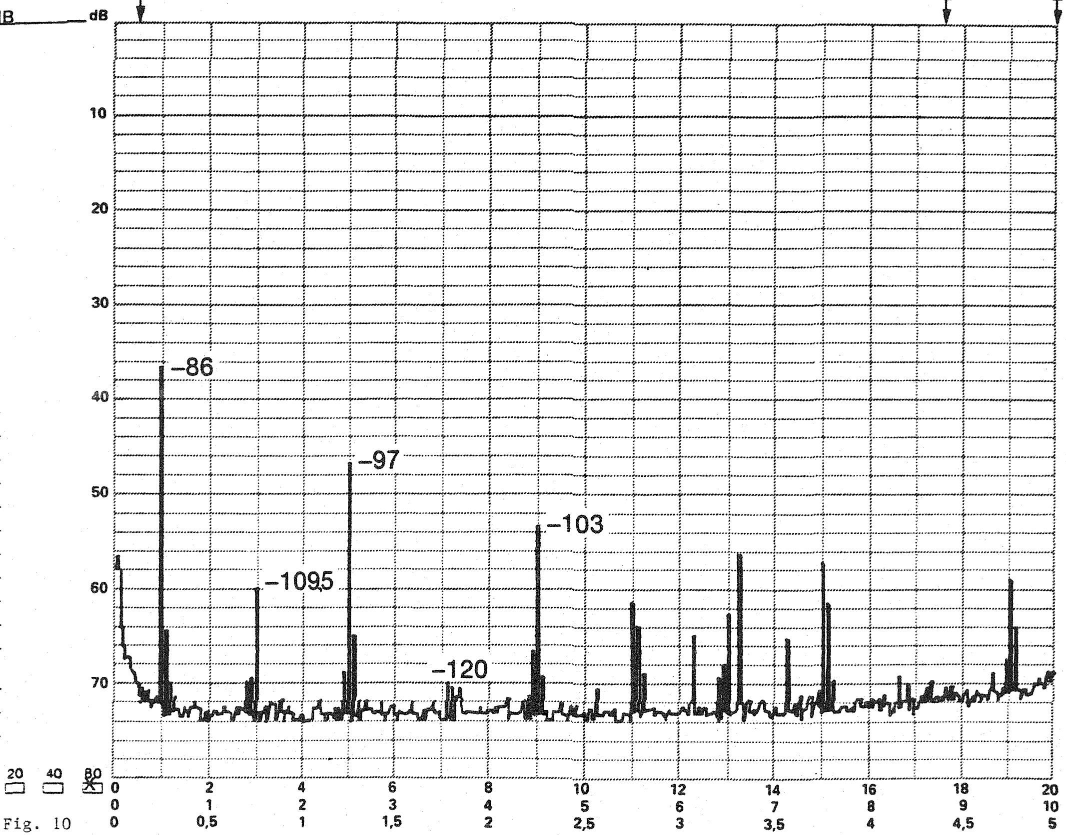


Fig. 10

QP 1002

Measuring Object: Unit A

850477

Brüel & Kjær Time Function Start seconds End seconds Not Expanded: Expanded

Full Scale Level: dB

F. S. Frequency: 2 kHz

Weighting: Hanning

Average Mode: Linear

No. of Spectra: 512

Comments:

Characteristics of

1⅛ Notch filter

Type 2120 positioned

at 1 kHz

Measured using

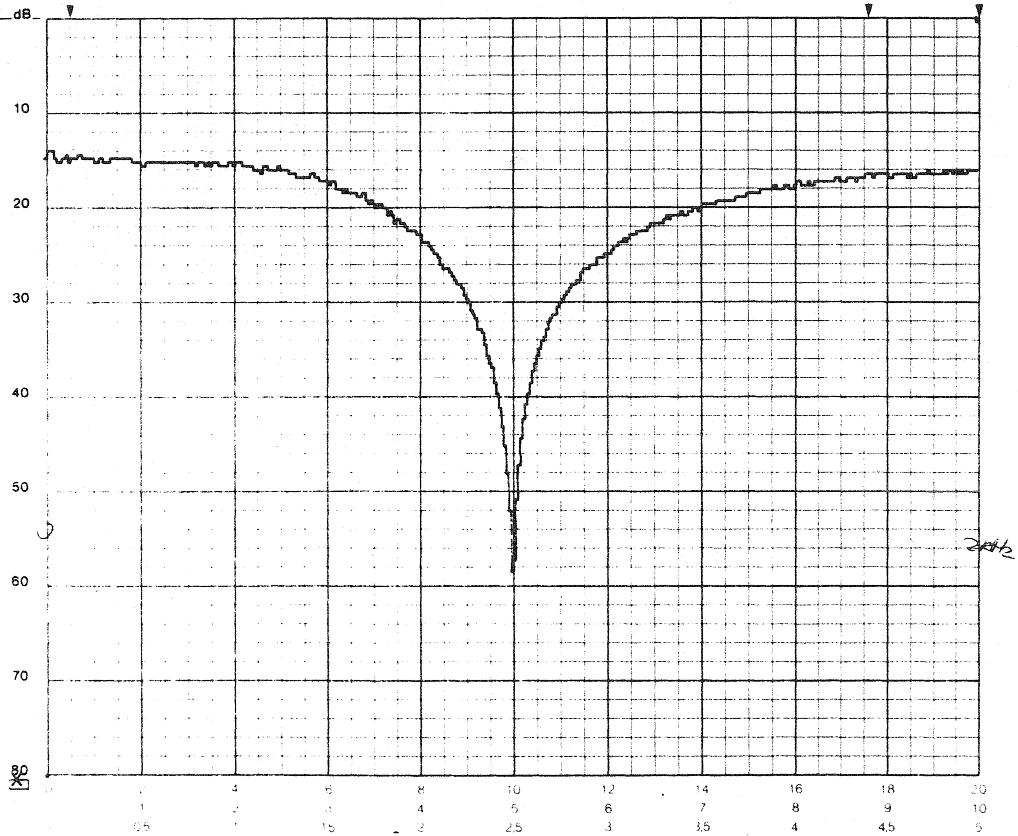
" -20 dB white noise:

Record No.: 11

Date: 11-22-1983

Sign.: AP

20 40 80



QP 1002

Fig. 11

Measuring Object

Unit A

Brüel & Kjær

Full Scale Level: -50 dB

F. S. Frequency: 20 kHz

Weighting: Flat

Average Mode: Linear

No. of Spectra: 225

Comments:

1 kHz, -60 dB

CD: Sony 3

Player: Unit A

Record No.: 12

Date: 11-22-1983

Sign.: AP

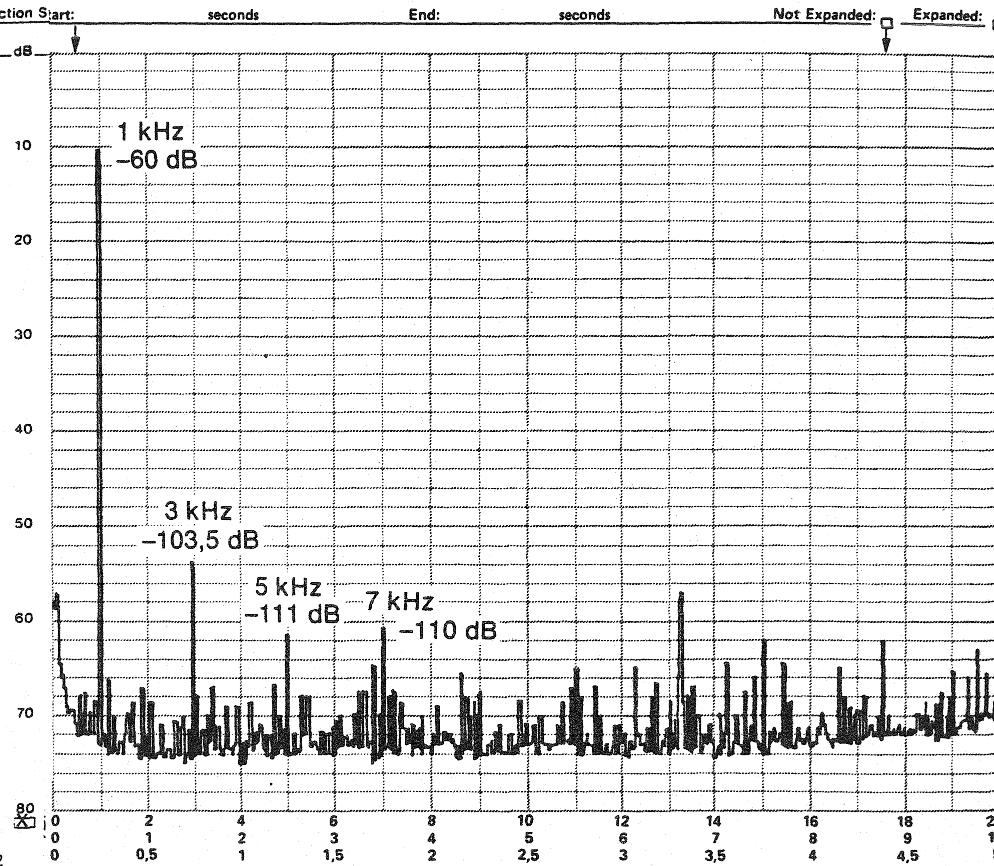


Fig. 12

QP 1002

Measuring Object: Unit A

850479

Brüel & Kjær

Time Function Start: seconds End: seconds Not Expanded: Expanded:

Full Scale Level: –60 dB dB

F.S. Frequency: 20 kHz

Weighting: FLAT

Average Mode: LIN

No. of Spectra: 256

Comments:

1 kHz, 0 dB

CD: Sony 3

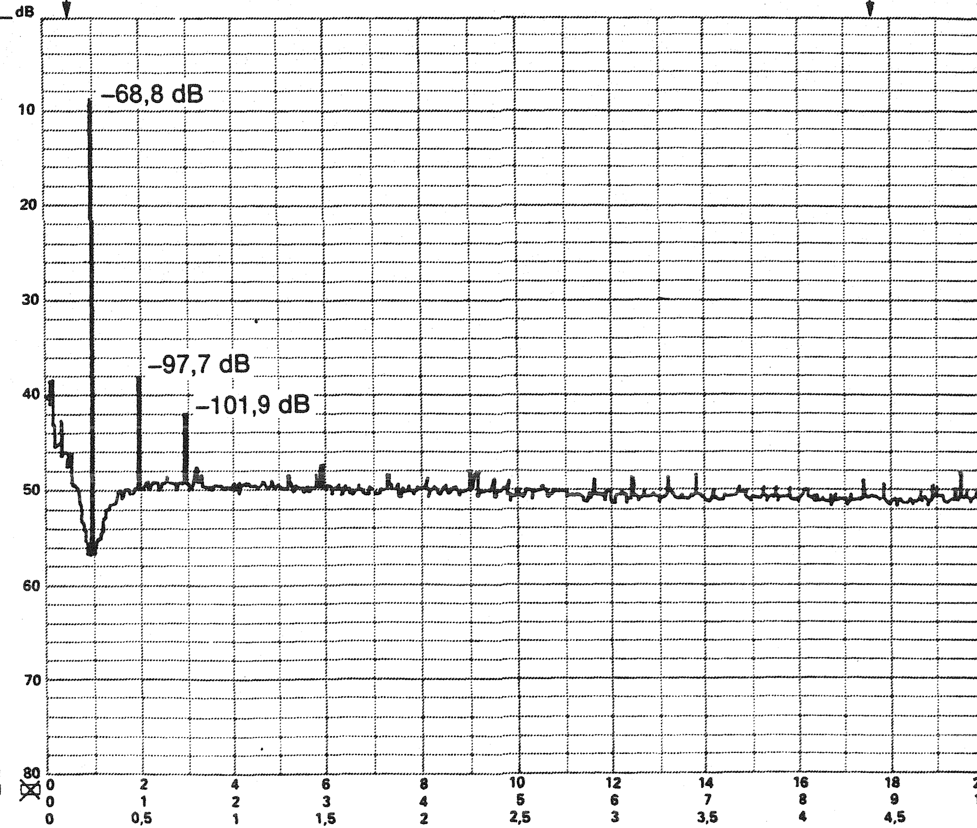
Player: Unit A

Notched by 2120

Record No.: 13

Date: 8-12-1983

Sign.: AP



QP 1002

Fig. 13

Measuring Object: Unit A

850486

Brüel & Kjær

Time Function Start: seconds End: seconds Not Expanded: Expanded:

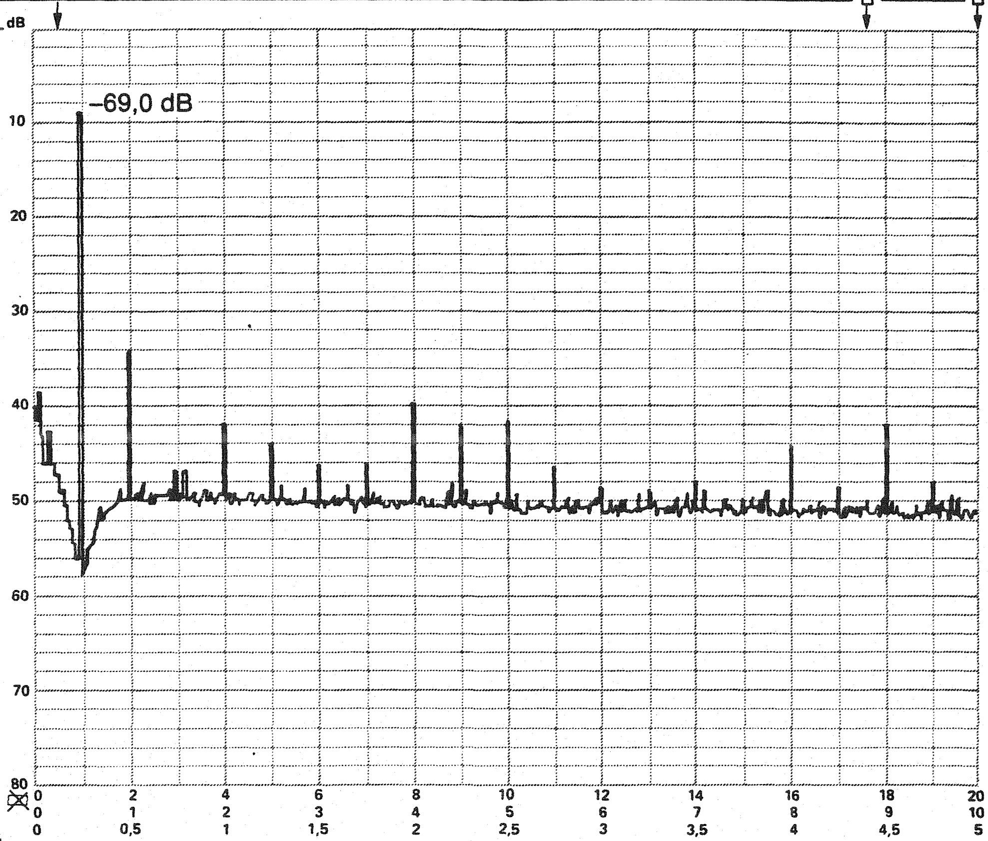
Full Scale Level: -60 dB
F.S. Frequency: 20 kHz
Weighting: FLAT
Average Mode: LIN
No. of Spectra: 256

Comments:
1 kHz, 0 dB

CD: Sony 3

Player: Unit D

Notched by 2120



Record No.: 14
Date: 8-12-1983
Sign.: AP

20 40 80

Fig. 14

QP 1002

Measuring Object: Unit D

850487

Brüel & Kjær

Time Function Start: seconds End: seconds Not Expanded: Expanded:

Full Scale Level: dB

F. S. Frequency: 20 kHz

Weighting: Flat

Average Mode: Linear

No. of Spectra: 256

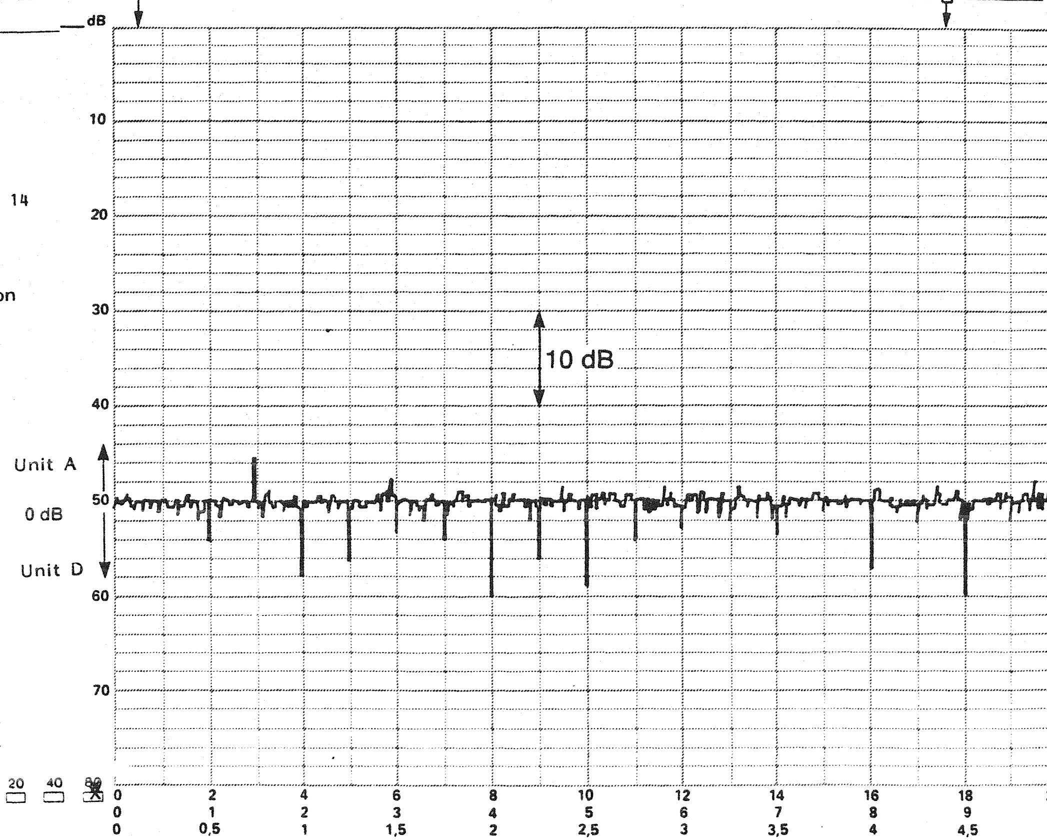
Comments:

Measurement 13

minus measurement 14

Unit A/Unit D

10 dB/major division



Record No.: 15=13/14

Date: 8-12-1983

Sign.: AP

QP 1002

Fig. 15

Measuring Object:

850488

Brüel & Kjær

Full Scale Level: -60 dB

F.S. Frequency: 1 kHz

Weighting: FLAT

Average Mode: LIN

No. of Spectra: 164

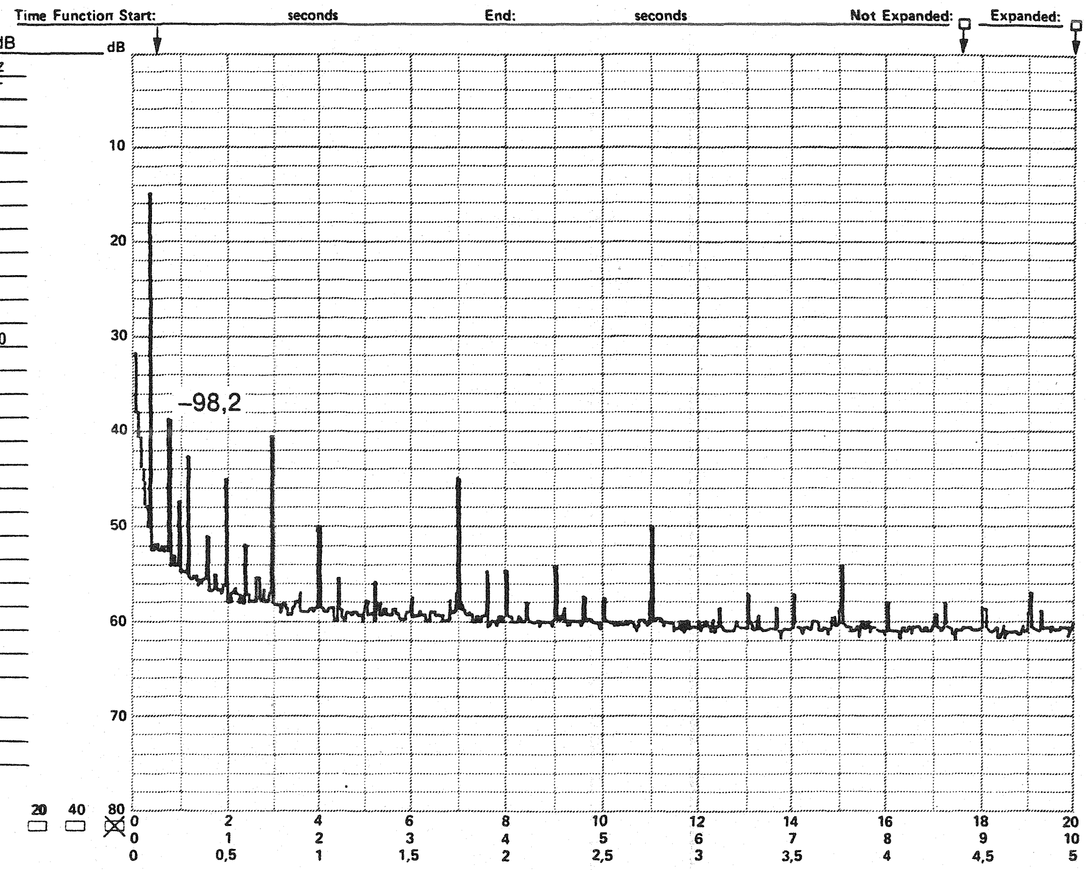
Comments:

20 Hz, 0 dB

CD: Sony 3

Player: Unit A

Notched by 2120



Record No.: 16

Date: 8-12-1983

Sign.: AP

QP 1002

Fig. 16

Measuring Object: Unit A

850489

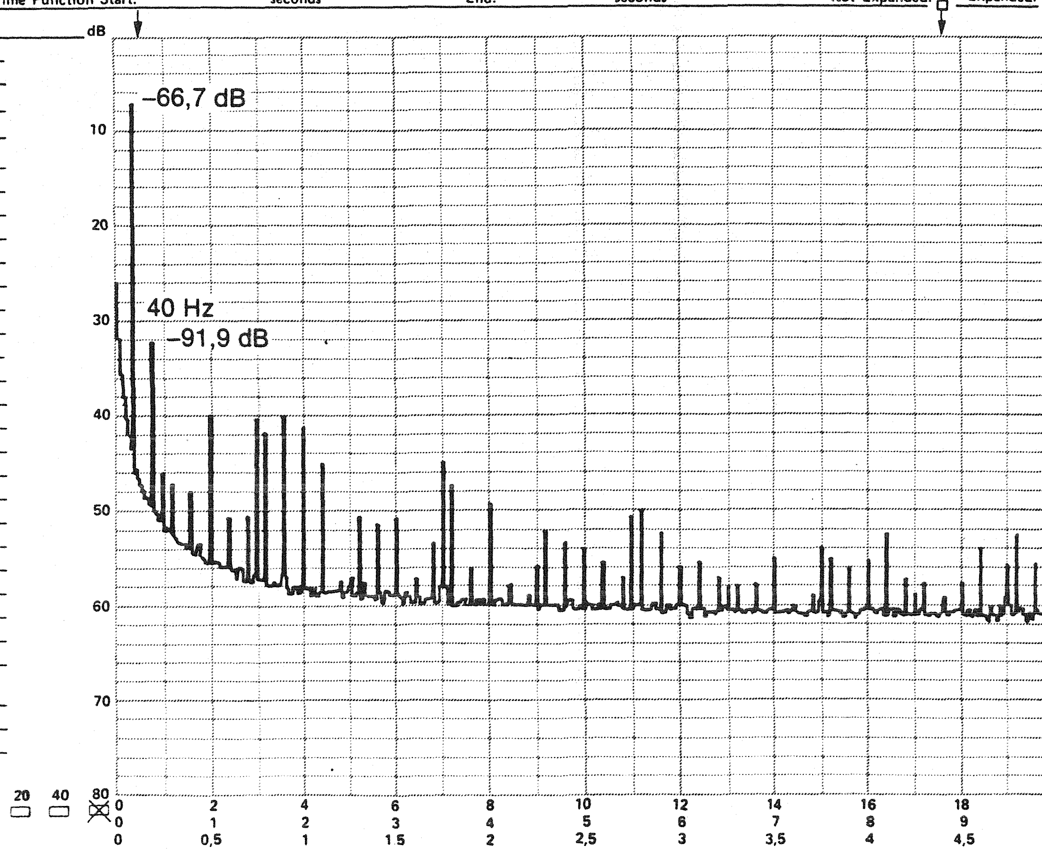
Brüel & Kjær

Time Function Start: seconds End: seconds Not Expanded: Expanded:

Full Scale Level: -60 dB dB
F.S. Frequency: 1 kHz
Weighting: FLAT
Average Mode: LIN
No. of Spectra: 215

Comments:
20 Hz, 0 dB
CD: SONY 3
Player: Unit D
Notched by 2120

Record No.: 17
Date: 8-12-1983
Sign.: AP



QP 1002

Fig. 17

Measuring Object: Unit D

850490

Brüel & Kjær

Time Function Start: ______ seconds End: ______ seconds Not Expanded: ☐ Expanded: ☐

Full Scale Level: -60 dB dB

F.S. Frequency: 20 kHz

Weighting: Flat

Average Mode: LIN

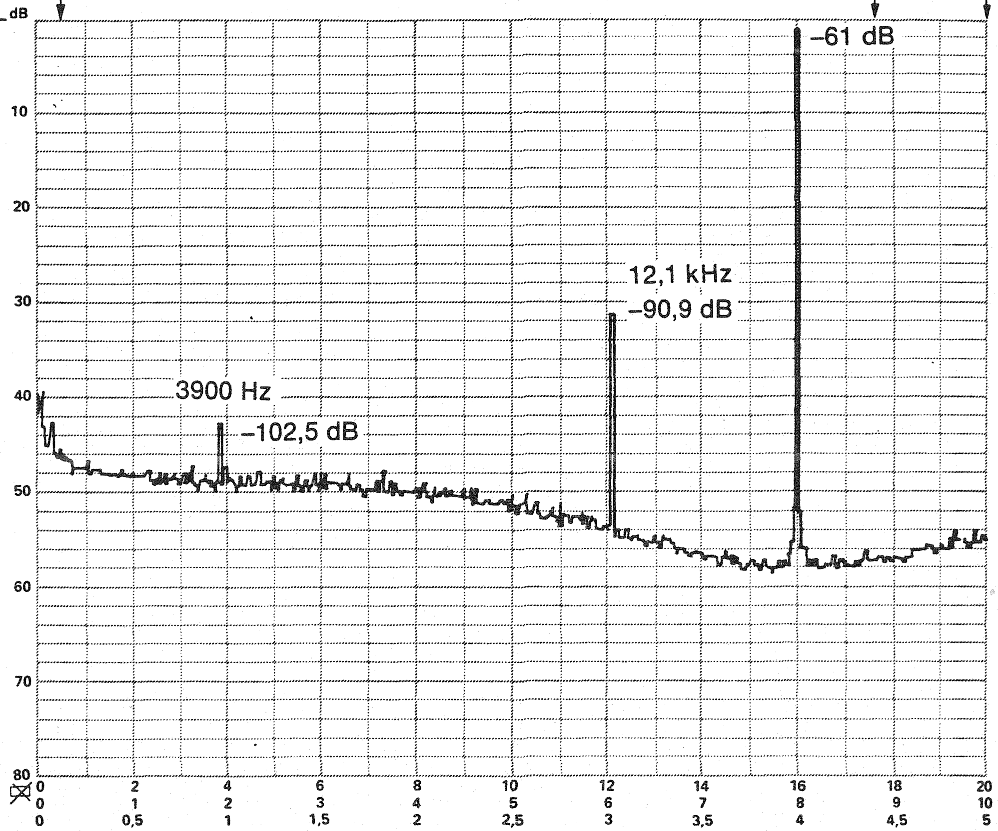
No. of Spectra: 256

Comments:

16 kHz, 0 dB

CD: SONY 3

Player: Unit A



Record No.: 18

Date: 8-12-1983

Sign.: AP

20 40 80 ☐ ☐ ☒

QP 1002

Fig. 18

Measuring Object: Unit A

850484

Brüel & Kjær

Time Function Start: ______ seconds   End: ______ seconds   Not Expanded: ☐   Expanded: ☐

Full Scale Level: -60 dB

F.S. Frequency: 20 kHz

Weighting: FLAT

Average Mode: LIN

No. of Spectra: 256

Comments:

16 kHz, 0 dB

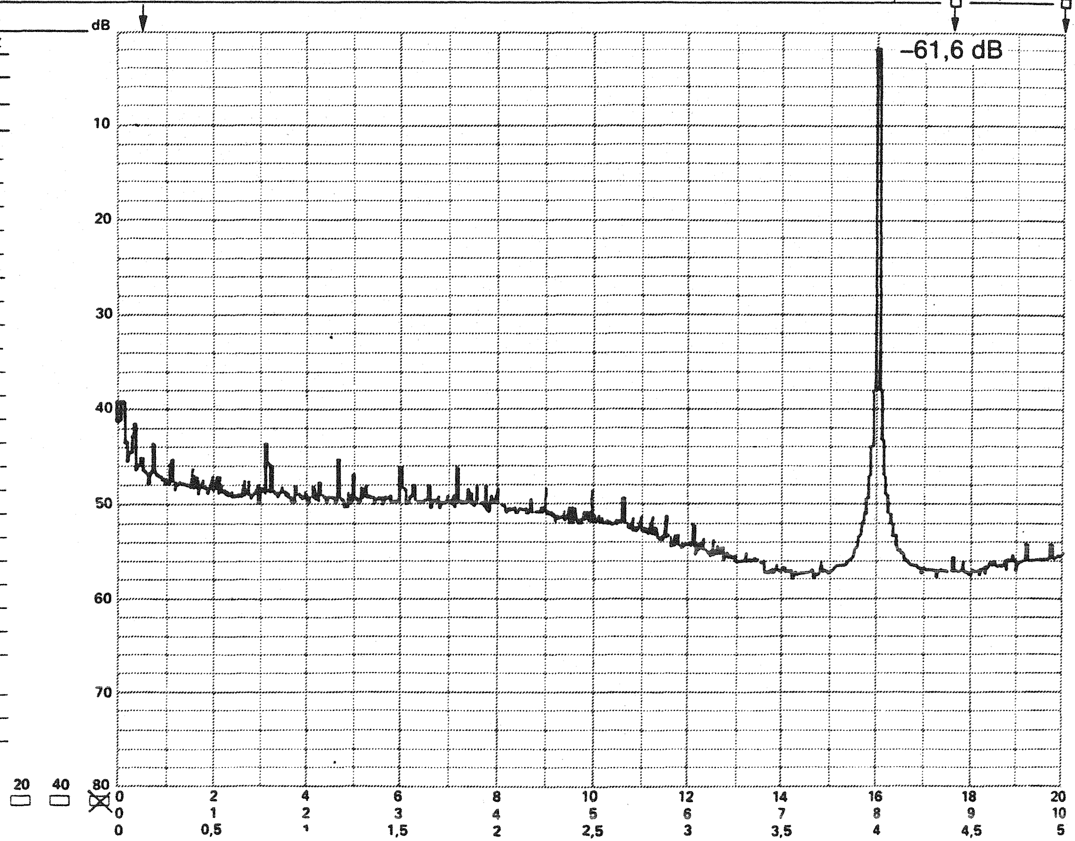
CD: SONY 3

Player: Unit D

Record No.: 19

Date: 8-12-1983

Sign.: AP



QP 1002

Fig. 19

Measuring Object: Unit D

850485

Brüel & Kjær

Time Function Start: 0 seconds 7,81 ms End: seconds Not Expanded: Expanded:

Full Scale Level: 2,10 V

F.S. Frequency: 25,6 kHz

Weighting: Trans SH: 2,792 m L: 2,914 m

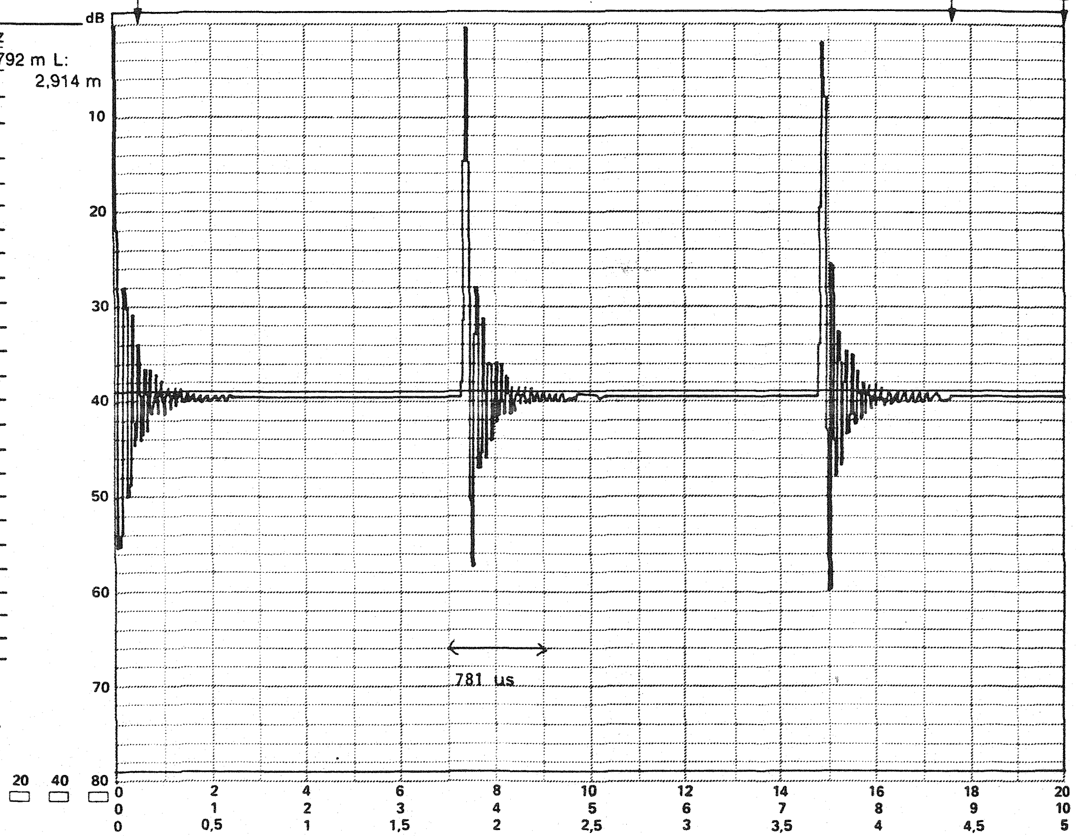
Average Mode:

No. of Spectra:

Comments:

Track 23 off

"Philips Test Record"



Record No.: 20

Date: 6.12.83

Sign.: A. Perman

QP 1002

Fig. 20

Measuring Object: Unit A

850481

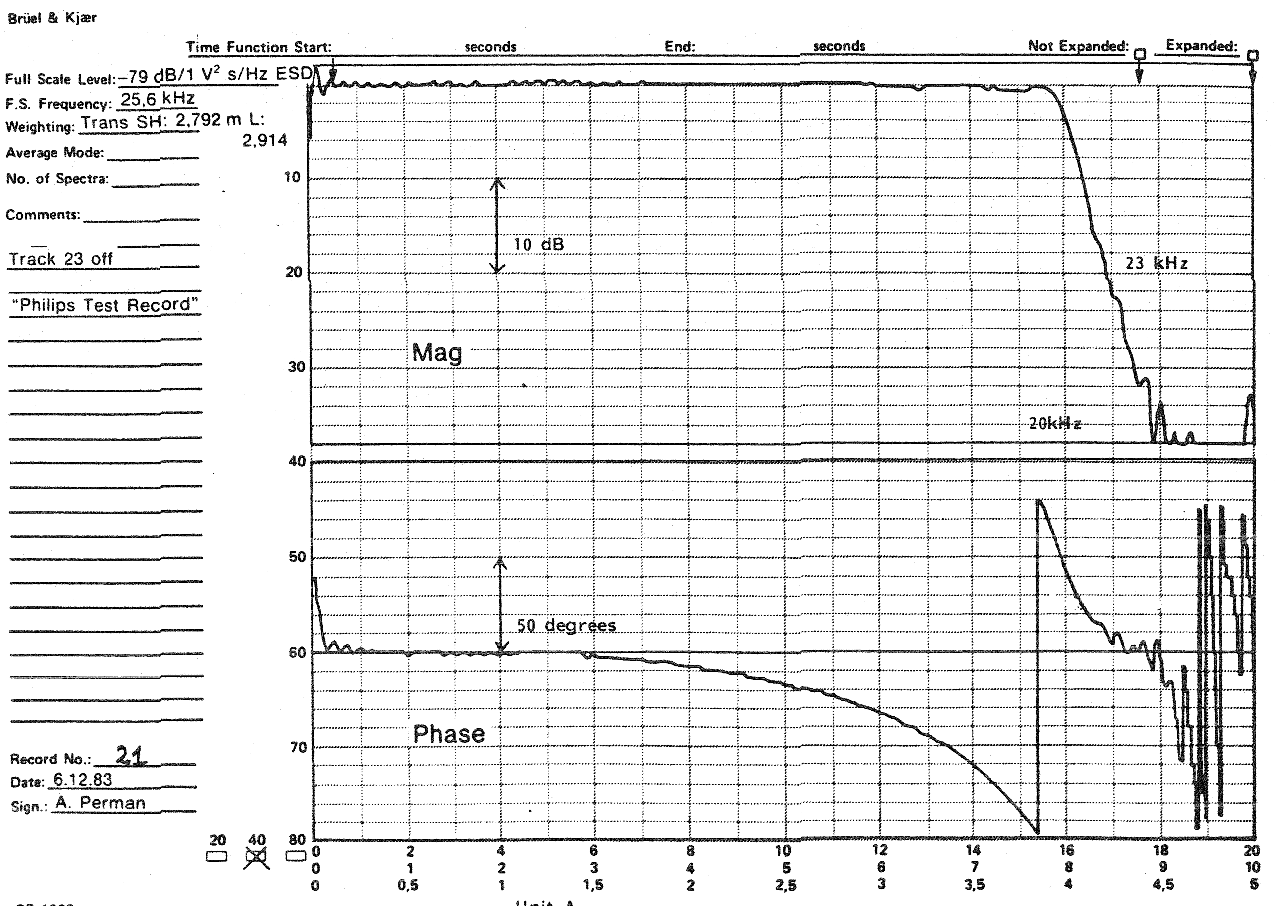


Fig. 21

Measuring Object: 850483

Brüel & Kjær

Time Function Start: 0 seconds 7,81 ms End: seconds Not Expanded: Expanded:

Full Scale Level: 2,73 V

F.S. Frequency: 25,6 kHz

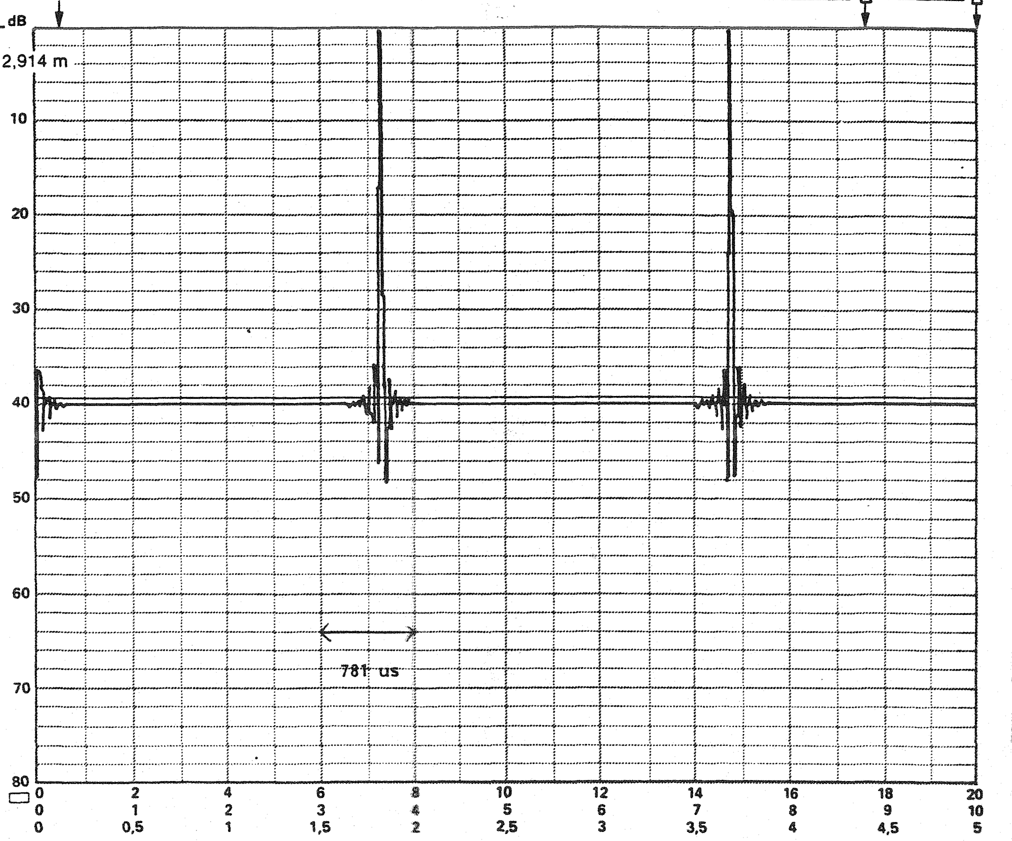
Weighting: Trans SH: 2,029 m L: 2,914 m

Average Mode:

No. of Spectra:

Comments:

Filter Off (2032)



Record No.: 22

Date: 6.12.83

Sign.: A. Perman

QP 1002

Fig. 22

Measuring Object: Unit D

850480

Brüel & Kjær

Time Function Start: seconds End: seconds Not Expanded: Expanded:

Full Scale Level: −79,0 dB/1 V^2 s/Hz ESD

F.S. Frequency: 25,6 kHz

Weighting: Trans SH: 2m029 m L: 2,914 m

Average Mode: Instant

No. of Spectra: 1

Comments:

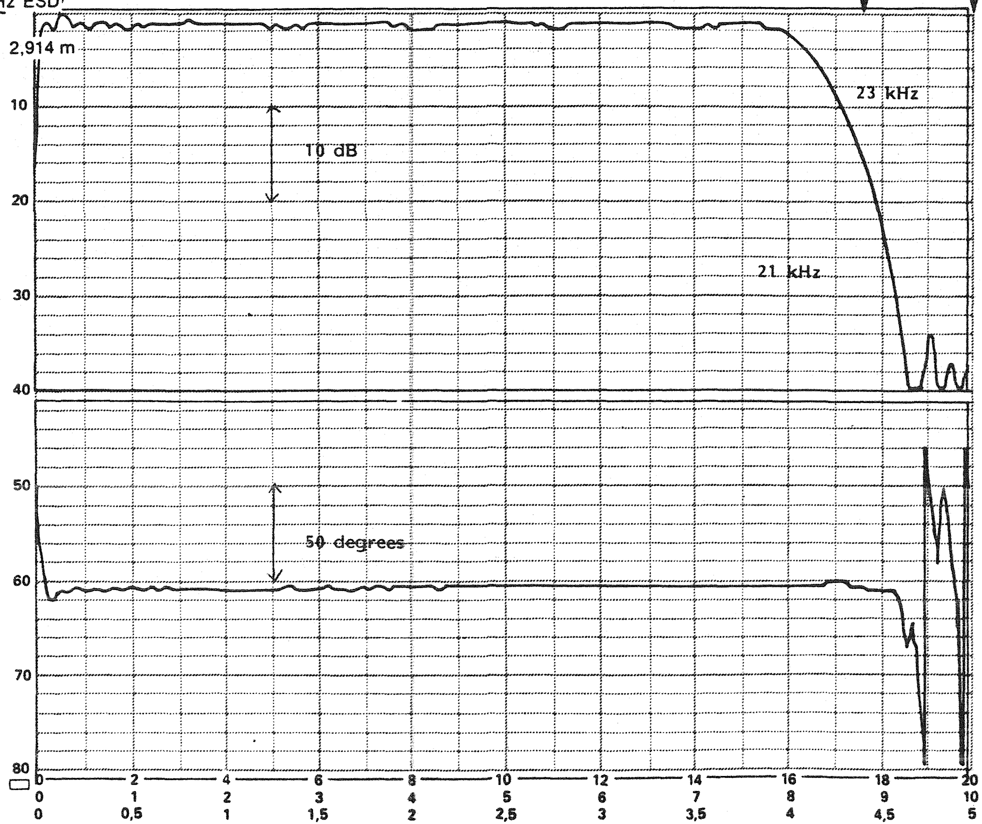
0 dB impulse

Record No.: 23

Date: 6.12.83

Sign.: A. Perman

20 40 80



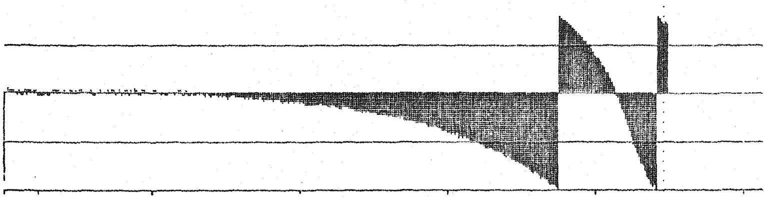
QP 1002

Fig. 23

Measuring Object: Unit D

850482

W2 FREQ RESP H1 PHASE INPUT REF Y: 145.7DEG
Y: -200 TO +200 DEG CMP:-0.40792ms X: 22304Hz
X: 0Hz + 25.6kHz LIN ΔX: 21248Hz
#A: 1123 ΔY : 141.9DEG



SETUP W3 **Unit F** IN/OUT. LEFT CHANNEL -20 dB VU.

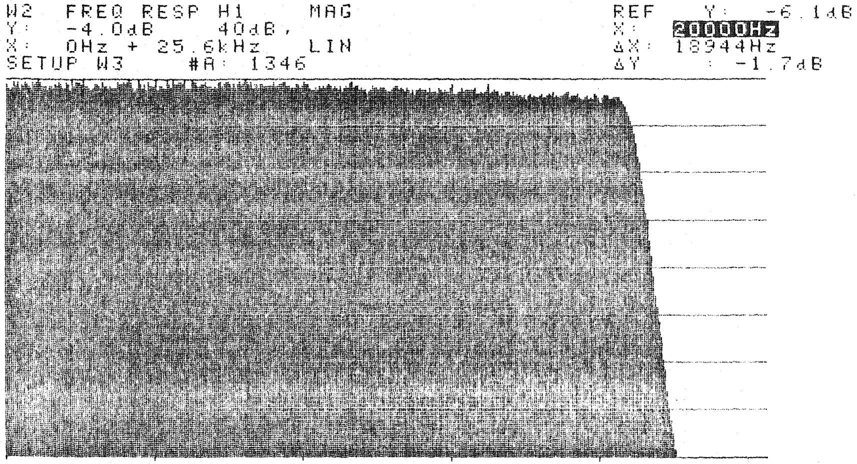
MEASUREMENT: DUAL SPECTRUM AVERAGING
TRIGGER: GENERATOR
DELAY: TRIG→A: -0.534ms CH.A→B: 11.306ms
AVERAGING: LIN 32767

FREQ SPAN: 25.6kHz ΔF:32Hz T:31.3ms ΔT:15.3µs
CENTER FREQ: BASEBAND
WEIGHTING: RECTANGULAR

CH.A: 800mV + 3Hz DIR FILT:25.6kHz 1V/V
CH.B: 1.5V - 3Hz DIR FILT:25.6kHz 1V/V
GENERATOR: PSEUDO RANDOM NOISE

--TRIGGERED-- -0

Fig. 24



Unit F IN/OUT. LEFT CHANNEL -20 dB WU.

Fig. 25

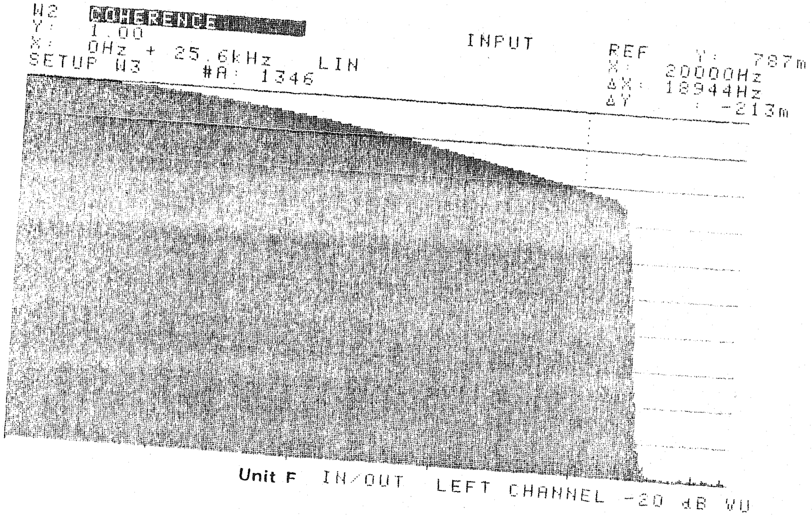
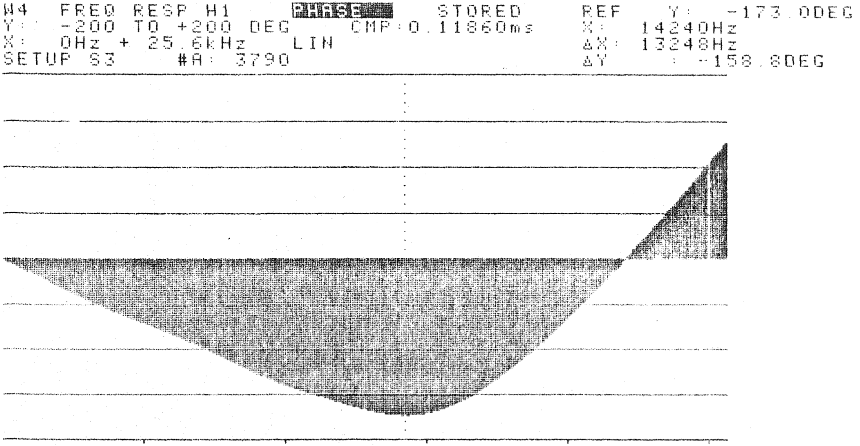
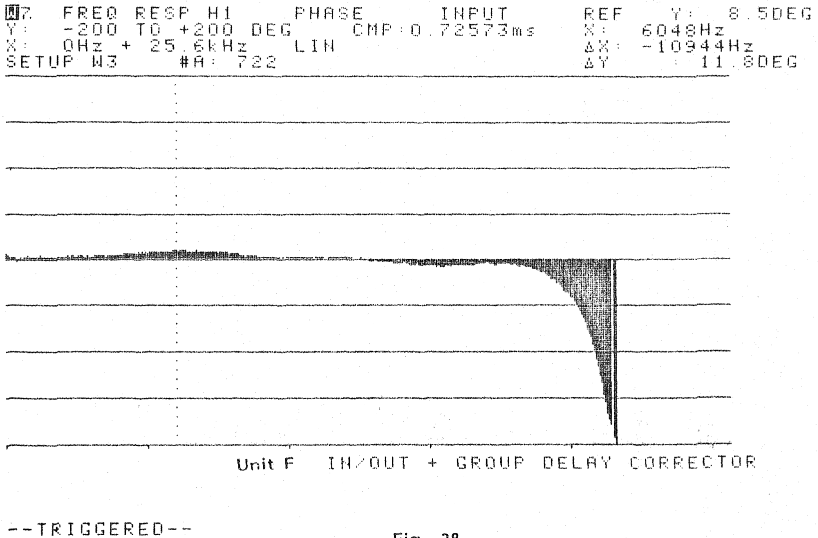

W2 COHERENCE
Y: 1.00
X: 0Hz + 25.6kHz LIN
SETUP W3 #A: 1346
INPUT
REF Y: 787m
X: 20000Hz
ΔX: 18944Hz
ΔY: -213m
Unit F IN/OUT LEFT CHANNEL -20 dB VU

Fig. 26



RESPONSE OF THE GROUP DELAY CORRECTOR FOR **Unit F**

Fig. 27



A7 FREQ RESP H1 PHASE INPUT REF Y: 8.5DEG
Y: -200 TO +200 DEG CMP:0.72573ms X: 6048Hz
X: 0Hz + 25.6kHz LIN ΔX: -10944Hz
SETUP W3 #A: 722 ΔY : 11.8DEG
Unit F IN/OUT + GROUP DELAY CORRECTOR
--TRIGGERED--

Fig. 28

Brüel & Kjær

Time Function St

Full Scale Level: ______ dB

F. S. Frequency: 100 Hz

Weighting: Hanning

Average Mode: Linear

No. of Spectra: 5x4 avg

Comments:

Music: "Satin Doll"

CD: "Super Audio Check", track 16

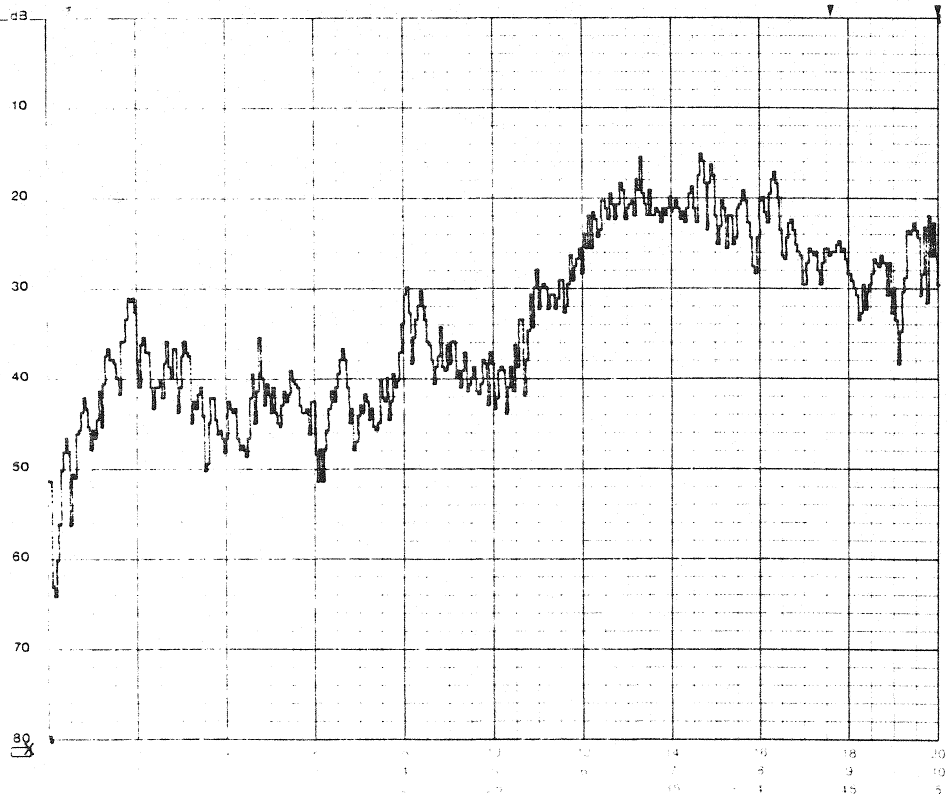
Player: Unit A

Load: 600 ohms

Record No.: 29

Date: 24.11.1983

Sign.: AP



QP 1002

Fig. 29

Unit A

Brüel & Kjær

Time Function Start ______ seconds   End ______ seconds   Not Expanded   Expanded

Full Scale Level: ______ dB

F. S. Frequency: 100 Hz

Weighting: Hanning

Average Mode: Linear

No. of Spectra: 5x4

Comments:

Music: "Satin Doll"

CD: "Super Audio Check", track 16

Player: Unit D

Load: 600 ohms

Record No.: 30

Date: 24.11.1983

Sign.: AP

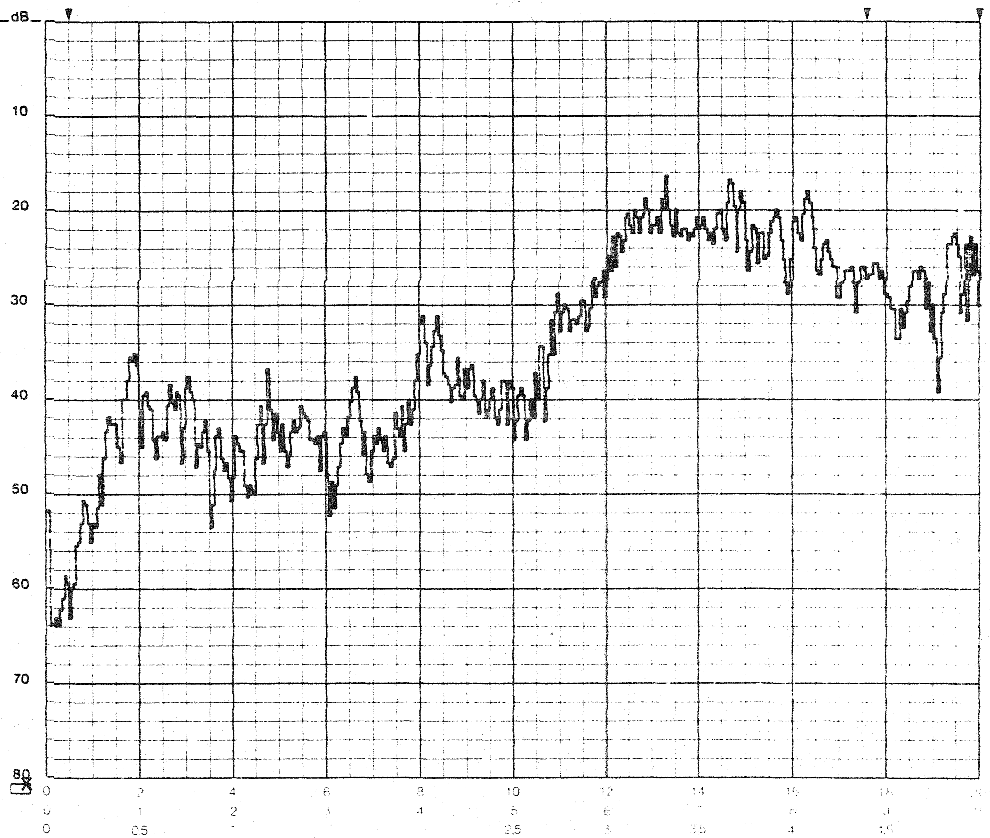


Fig. 30

Measuring Object: Unit D

QP 1002

Brüel & Kjær

Time Function

Full Scale Level: ______ dB

F. S. Frequency: 100 Hz

Weighting: Hanning

Average Mode: Linear

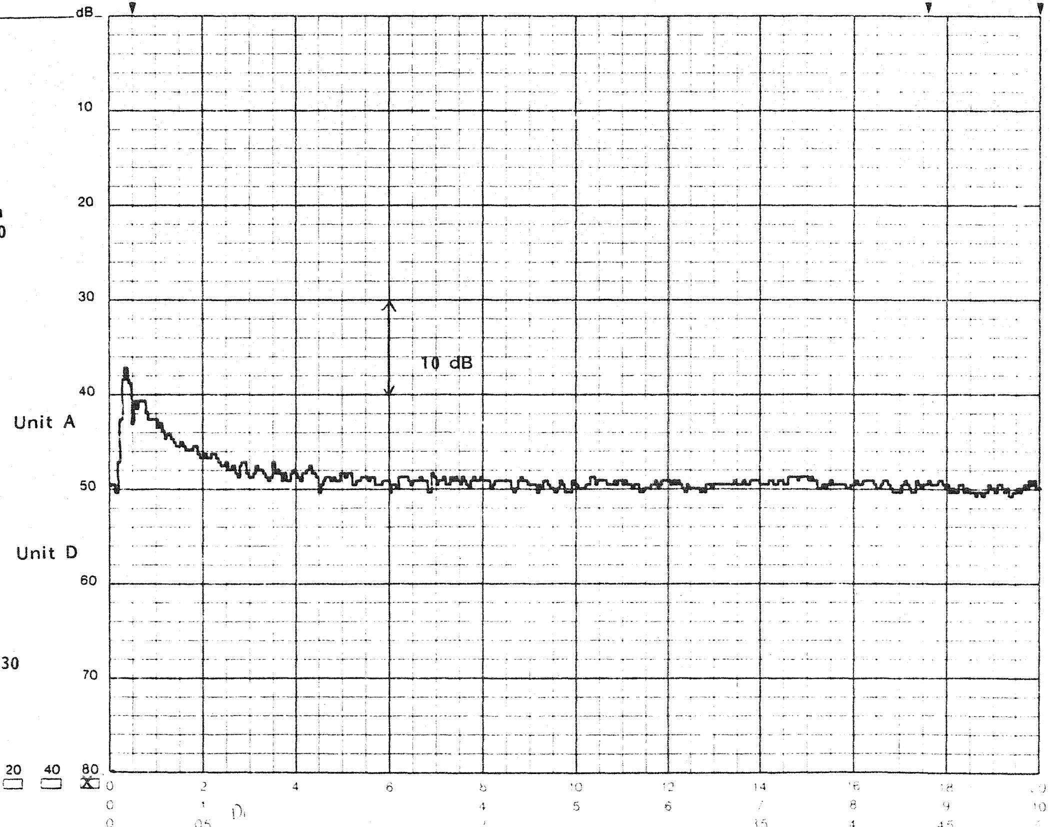
No. of Spectra: 5x4

Comments:

Difference between fig. 29 and fig. 30

Unit A vs. Unit D

Load: 600 ohms



31 = 29/30

Record No.:

Date: 24.11.1983

Sign.: AP

QP 1002

Fig. 31

Measuring Object: Unit A ÷ Unit D

Brüel & Kjær

Time Function Start: seconds End: seconds Not Expanded: Expanded:

Full Scale Level: dB

F. S. Frequency: 100 Hz

Weighting: Hanning

Average Mode: Linear

No. of Spectra: 5x4

Comments:

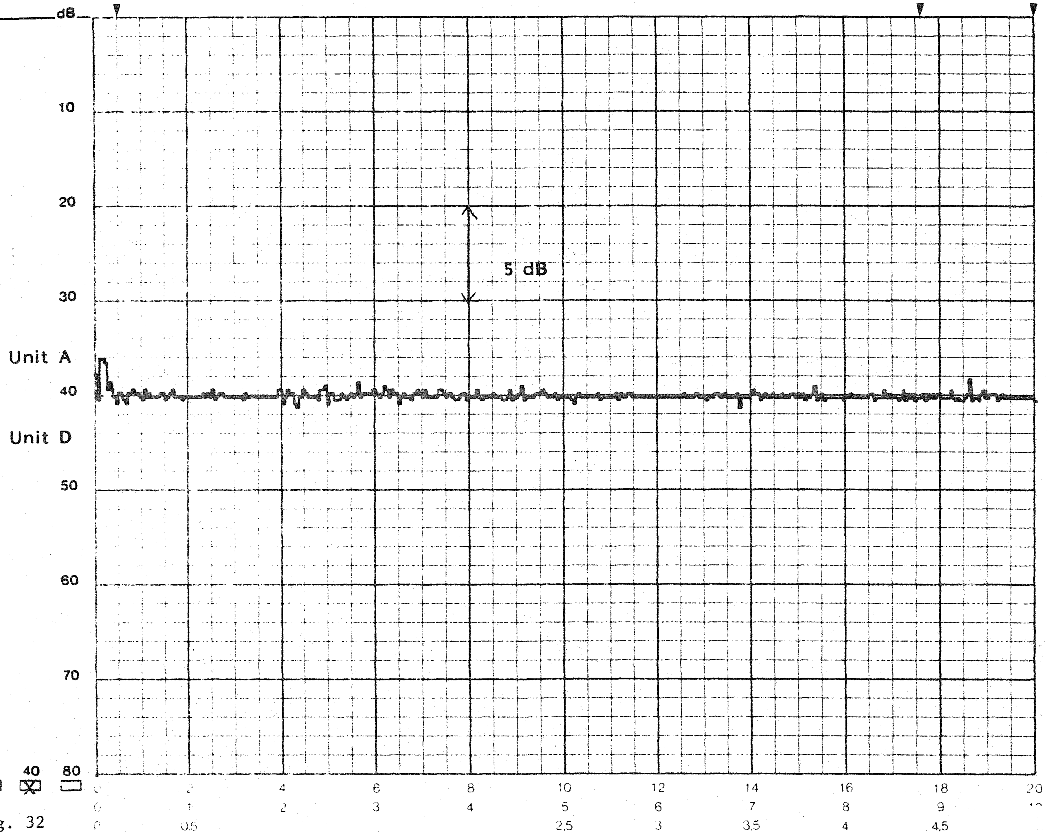
Unit A vs. Unit D

Load: 1 Megaohm

Record No.: 32

Date: 8.12.1983

Sign.: AP



20 40 80

Fig. 32

QP 1002

Measuring Object: Unit A ÷ Unit D