Collection of Setups for Measurements with Audio Analyzers UPL and UPD

Application Note 1GA36_1E

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Subject to change

Products:

Audio Analyzer UPL

Audio Analyzer UPD

ROHDE&SCHWARZ

Contents:

1. Introduction

Being faced with a wide variety of standards and manufacturer's specifications, test engineers may often find it difficult to keep track of necessary audio measurements. This is aggravated by the fact that modern audio analyzers offer a multitude of settings. This application note serves as an aid offering a collection of typical setups that make it possible to get started with measurements immediately. In addition, information is given on associated standards, on the adaptation of the setups to specific measurement tasks as well as on the evaluation of results.

2. Purpose of Setup Collection

A variety of measurements is performed in audio engineering in order to ensure transmission quality. There is a vast number of standards defining measurement conditions, and the wide range of measurements to be performed is further expanded by manufacturers' specifications. With the widespread digital audio techniques, new sources of error and consequently new demands on the measurements are additionally created. An audio analyzer suitable for all these tasks will, therefore, incorporate a multitude of functions, resulting in a correspondingly large number of settings.

This application note is to help test engineers using Audio Analyzers UPD and UPL for the first time. It presents setting examples for all basic audio measurements and thus allows measurements to be performed with UPD and UPL immediately. For each setup, information is given on the type of measurement and underlying standards, and on how to modify the graphic display and interpret results.

All setups described here are stored on a floppy disk available from your local Rohde & Schwarz representative. Where appropriate, separate setups for use on analog and on digital interfaces are presented.

3. Notes on Setups

3.1. Installing the Setups on Audio Analyzer UPD or UPL

Installation of the setups on Audio Analyzer UPD or UPL is made by means of an external keyboard.

The installation floppy contains the file SETINST.BAT, which must be loaded from the MS-DOS interface of Audio Analyzer UPD or UPL. Directory C:\UPD\SET_EXAM or C:\UPL\SET_EXAM is to be generated and all setups copied into this directory. The setups are organized in four subdirectories according to their interface configuration.

To activate the setups, Audio Analyzer UPD must have firmware version 3.02 installed, and Audio Analyzer UPL firmware version 1.01 or higher.

Many measurements, mostly those at analog interfaces, can be performed with the UPD or UPL basic model. All measurements on digital interfaces require Option UPD-B2 or UPL-B2 (Digital AES/EBU Interface). Where further options are required, appropriate information is given in each case.

3.2. Structure of File Names

All setups are in the form of "actual setups". This means that, in contrast to "complete setups", the setups described here contain only the settings for the analyzer/generator used for a specific task and the related settings for the DISPLAY, FILE and OPTION panels. The advantage is that these setups are loaded in a considerably shorter time than complete setups, which contain all UPL/UPD settings.

To make it easier to find a required setup, the file names are organized according to a defined structure, which is described below:

FFFDS_OI.SAC

FFF: The first three or four characters describe the measurement function, for example: LEV = level measurement, PHA = phase measurement, THD = distortion measurement, etc

D: "D" placed after the measurement function stands for difference or deviation, for example: PHAD = phase difference, LIND = nonlinearity

S: "S" indicates a swept measurement, for example: LEVS = measurement of level versus frequency (frequency response), THDNS = THD+N value versus frequency

OI: These two characters describe the analog and/or digital interfaces used. Possible combinations are AA, AD, DA and DD.

SAC: This extension identifies the actual setups.

3.3. Basic Settings

Although the setups can basically be used on both UPD and UPL, separate groups of setups were generated for this application note to utilize the instrument functions optimally.

The basic settings for the analog inputs and outputs are as follows (see Fig.1):

Both the generator and the analyzer channel are active. The balanced XLR connectors with minimum generator impedance or maximum analyzer impedance are used.
All input and outputs are floating. The autorange function is on.
Frequency sweeps are normally logarithmic from 20Hz to 20 kHz, the OPERATION parameter in the DISPLAY panel is set to CURVE PLOT, the x axis is set to automatic scaling, the y axis is set to fixed, commonly used values.
Voltages are mainly indicated in V; the generator output voltage is set to 1V.
Distortion and intermodulation are indicated in dB.

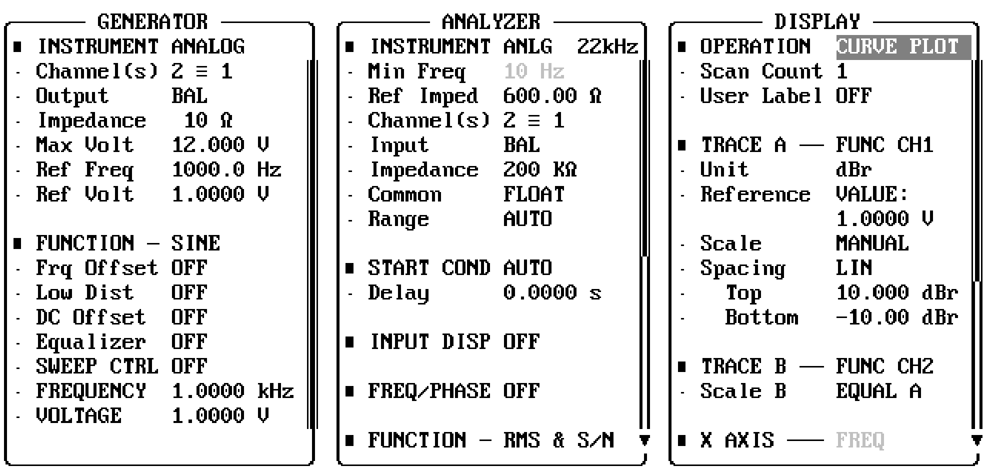


Fig. 1: Basic settings for analog interfaces

The basic settings for the digital inputs and outputs are as follows (Fig.2):

Both channels are active. The generator supplies 24-bit words, the analyzer evaluates 24 audio bits. The sampling frequency is 48 kHz; the professional format to AES 3 is used. The digital pulse amplitude of the generator is 1 V at the BNC outputs; this corresponds to 4V at the XLR connectors. Levels are mainly indicated in dBFS; audio signals are mostly generated with a level of -20 dBFS.

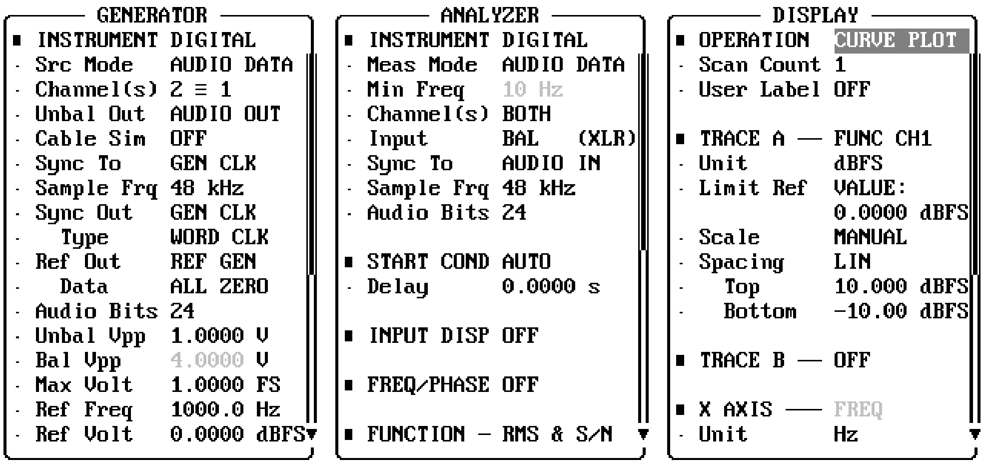


Fig. 2: Basic settings for digital interfaces

Basic settings in the FILE and OPTION panels:

C:\UPD\SET_EXAM or C:\UPL\SET_EXAM is entered as working directory.
The parameter link function is activated, it takes on the parameters of the function concerned.
Hardcopies are generated on the default printer. External monitors, if any, are served.

The lines with the most important settings are marked in each panel and can be displayed in the STATUS panel in addition to the graphic display.

The basic settings can be adapted as required for a specific measurement task. To this effect, the setup in question must be loaded and the desired changes made in the respective panels. Then the setup is to be stored under the same name or a user-defined name.

3.4. Notes on Measurements

For each application, the measurement conditions and procedure in accordance with the relevant standards are described. In addition, information is given on the purpose of a measurement and on expected results.
Under "Graphic display", the representation of results is described. The user will find hints on how to adapt the display to his specific requirements. Description of the setups includes modifications of measurements for the purpose of adaptation to specific measurement tasks. The user can thus generate, from the setups given here, any setups to suit his requirements. Any information relating to commands or command lines of UPD/UPL are *in italics* in this application note.

3.5. Standards

Most of the measurements described in the setups given here are defined in DIN IEC 268, "Sound System Equipment, Part 3: Amplifiers". This standard defines the measurements to be made on amplifiers for professional and domestic applications. The standard, however, refers to equipment with analog interfaces only.

As regards components with digital or analog and digital interfaces, many measurements are the same as for components with analog interfaces, but with digital interfaces effects will occur that call for modified or extended measurements. This is taken into account by AES17, "Measurement of Digital Audio Equipment".

Wherever possible, the setups described here are in line with the above standards.

3.6. Nominal Conditions - Standard Test Conditions

Basically, the **nominal conditions** defined by IEC 268 are to be observed in all measurements. These conditions essentially include operation of the equipment in accordance with the intended use, ie observance of the operating temperature range, appropriate power supply, etc.
The **standard test conditions** define the conditions under which measurements are to be performed. For example, the amplitude frequency response of an amplifier is to be measured at 10 dB below the full-scale amplitude since it is assumed that the level of commonly used speech or music signals will on average not exceed this level (see IEC 268-3).

As an important point, the correct input and output impedances at the DUT must be observed. In professional sound-studio measurements, power matching has been in use for a long time - this means the same impedance (usually 600 Ω) at the source and load -, voltage matching is preferred today. In the latter case, the source is operated at a low impedance (<30Ω) and the load at a high input impedance (>20 kΩ for balanced lines, >100 kΩ for unbalanced lines). The setups described here use voltage matching. However, if for example amplifiers intended for operation with loudspeakers are to be measured, appropriate load resistors must be connected to the outputs as otherwise the high input impedance of the measuring instrument would not reflect the true operating conditions of an amplifier.

4. Linear Distortion Measurements

4.1. Amplitude Frequency Response

Definitions and test conditions:

Measurement of the amplitude frequency response is the classic measurement task. Since this type of measurement is much more frequent than phase frequency response measurement, it is often described simply as the frequency response.
The frequency response of amplifiers is measured in accordance with DIN IEC 268-3 at 10 dB below the full-scale amplitude by sweeping an input signal of constant level over the frequency range. The output level is plotted against the frequency.
With digital components, the frequency response is measured in accordance with AES17 at -20 dBFS.

Graphic display:

In accordance with IEC 268-1, the frequency response is represented by displaying the rms output level in dB along the frequency axis using a logarithmic scale. The x- and y-axis scalings should be chosen such that a frequency decade corresponds in size to a level difference of 50dB (10 dB and 25 dB are also permissible). As with modern audio equipment, level differences are often very small, the scalings stated above may sometimes not be appropriate for revealing the fine structure of the frequency response. In the setups described, therefore, the y-axis has been scaled for ±10 dB, which more closely reflects practical requirements.

If the set y-axis scaling is inappropriate or if results are outside the displayed range, it is best to switch from *MANUAL* to *AUTO ONCE* in the *SCALING* line in the DISPLAY panel. As a result, the graphic display on Audio Analyzers UPD and UPL will be scaled such that, after a single sweep, all results are represented on the display.

IEC 268-1 defines a reference frequency of 1 kHz for the level values. This applies however only to analog systems. In accordance with AES17, a reference frequency of 997 Hz is to be used for digital as well as analog and digital systems.

Notes on measurements:

The amplitude frequency response can be measured in different ways with Audio Analyzers UPD and UPL. The differences are described in detail in the following.
In both cases, levels are represented in dBr. The user must, however, refer levels to 1kHz or 997 Hz. This is easiest done by placing a cursor on the reference frequency (select cursor in GRAPH panel) and by transferring the cursor value into the *Reference* line in the option window of the DISPLAY panel (see Fig. 3). Depending on the test points selected, it may not be possible to place the cursor precisely on the reference frequency. In most cases it will however suffice to place it on an adjacent point. If there are large variations of the frequency response in the vicinity of the reference frequency, interpolation has to be performed or the point determined exactly by way of measurement.

4.1.1. Sweep Measurements Using Signals from UPD/UPL Generator

Setups:	**LEVS_AA.SAC**	**LEVS_AD.SAC**
	LEVS_DA.SAC	**LEVS_DD.SAC**

This measurement is the standard measurement, as it were. The generator supplies a sinewave signal which is swept logarithmically at a constant level over the frequency range 20Hz to 20 kHz. The rms output level of the DUT is measured and displayed graphically.

The above setups provide for a sweep with 50 frequency points. The set measurement speed is *GENTRACK*, ie the analyzer adjusts the measurement time for each point to the cycle time of the generator signal, which results in very high measurement speeds.

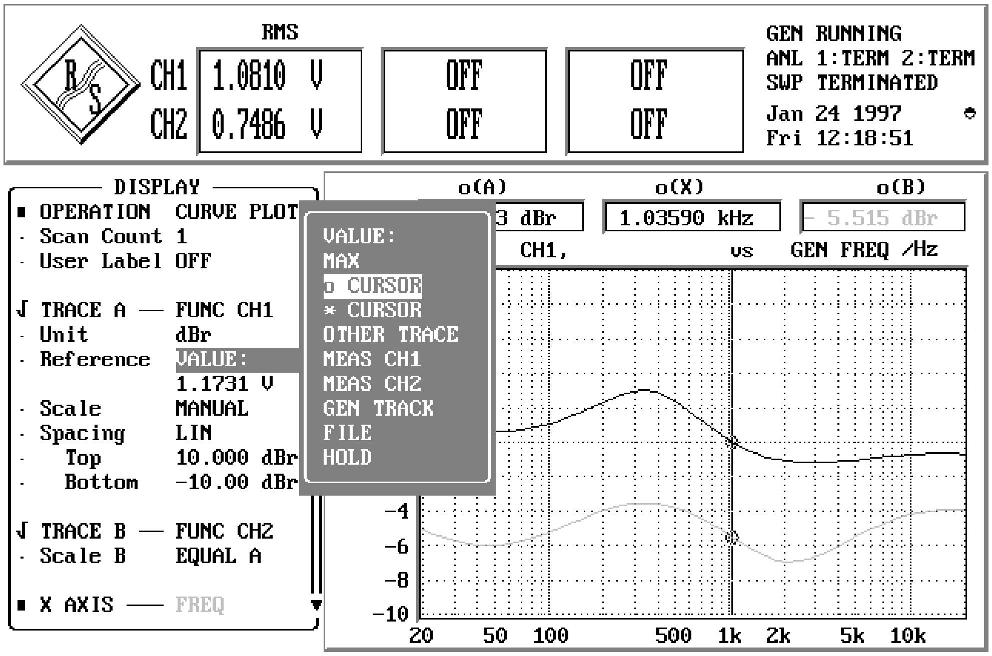


Fig. 3: Display of amplitude frequency response; the reference value is transferred by means of the cursor function

4.1.2. Sweep Measurements Using Signals from External Source

Setups: **LEVSE_AA.SAC** **LEVSE_DD.SAC**

Measuring the amplitude frequency response of a two-terminal DUT, eg a CD player, is not possible by means of the UPD/UPL generator. Such measurements are performed using a stored test sequence (eg from a test CD). To determine the test points, the analyzer measures not only the level but also the corresponding frequency of the test signal. With UPD/UPL, this is done by means of the external sweep function, which is selected under *START CONDITION* in the ANALYZER panel.

In the setup described here, the parameters of the external sweep were selected for use with the UPA-CD test CD from Rohde & Schwarz. Measurement values are collected on a specific frequency change detected in input channel 1. The *FRQ FST CH1* function used here makes for very fast frequency measurements but requires clean signals. When making measurements on signal sources with high noise content (eg cassette recorders), the slower *FRQ CH1* measurement function must be used instead.
In this setup, the frequency response between 20Hz and 20 kHz is determined with the frequency points spaced 5%, ie after a frequency change of 5% referred to the input signal a new level measurement is started.

The first test point is recorded when UPD/UPL for the first time measures a frequency higher than the set *START* frequency; the measurement sequence is terminated when the *STOP* frequency is attained. To ensure that the sweep function from the test CD is recorded right from the start, the sweep function must be activated on the analyzer by pressing the START key before the test sequence for the DUT is started.

The setups provide for frequency response display only for channel 1 since the test CD supplies a single-channel signal. By activating channel 2 in the ANALYZER panel and *TRACE B* in the DISPLAY panel, the second channel too will be displayed.

For standardized measurements of the frequency response of CD players, it is necessary, as an initial step, to measure for both channels the rms output voltage of the CD player while playing the signal with the recorded reference level (at 1 kHz, track1 of test CD) and store the results as reference values for both channels (DISPLAY panel, *Reference* line). To this effect, the scaling for channel 2 must be set to *NOT EQUAL A* and the reference value for *Trace B* entered separately.
Please note that the frequency response for each channel is thus referred to the level at 1 kHz. Any level differences between the two channels will not be recognized from the frequency response traces.

For detailed information on this type of measurement refer to Application Note 1GA12_1E, "External Sweep and Adaptive Measurement of DUTs with Extreme Transients Using Settling Function of UPD" available from your Rohde & Schwarz representative.

4.1.3. Fast Frequency Response Measurements Using FFT

Setups: **FFLEV_AA.SAC** **FFLEV_DD.SAC**

Although UPD and UPL provide extremely fast level measurements, swept frequency response measurements do not always satisfy speed requirements, for example in alignments or production. To avoid measurement errors caused by the windowing of the FFT, a rectangular window is used. This however requires the use of a special test signal. Audio Analyzers UPD and UPL generate a pseudo noise signal consisting of many discrete frequency lines, each line being an integer multiple of the analysis time window and thus being precisely matched to the frequency lines of the FFT analysis. Moreover, the test signal used should have a small crest factor to avoid overdriving of the DUT input by high peak voltage levels, which would be the case with white noise.

In this setup, FFT analysis with 2k points has been selected; this results in frequency response measurement with over 960 points with constant frequency spacing.
The generator signal is produced by means of the *RANDOM* function. This yields a multi-frequency signal whose frequencies are matched to the FFT lines of the analyzer (setting: *ANL TRACK*) and whose phases are optimized relative to one another for the smallest possible crest factor.
For this calculation, the generator requires a few seconds; the process is indicated in the status display in the upper right-hand corner of the screen. The measurement itself is performed at the speed of a single FFT. When the frequency response of the DUT is varied, the variation can be observed on the screen quasi in realtime since all test points are determined simultaneously.

The setup can be adapted to a finer or coarser resolution of the frequency points by selecting a different FFT size. However, the higher the number of frequency lines selected, the longer computation time of the generator prior to the start of the first measurement.

The voltage values of the test points shown in the graphic display are far below the rms values of the test signal. The latter, however, are the rms values of the bins of the FFT analysis, and when adding the squares of the discrete level values the rms value of the total signal is obtained. In this, FFT measurements differ from swept measurements, where the DUT is driven at one frequency only, whereas in measurements using a pseudo noise signal the total energy of the signal is distributed (broadband).

For this type of measurement, too, an Application Note is available: "Fast Frequency Response Measurements with Audio Analyzer UPD", 1GA04_1E.

4.1.4. Frequency Response Measurement at Different Levels

Setups: **MLEVS_AA.SAC** **MLEVS_DD.SAC**

In testing tape recorders with noise suppression, the frequency response must be determined at different levels since the Dolby method, for example, operates level-dependent. But in other cases too the variation of a parameter as a function of frequency and level is of interest. Fig.4 shows as an example the level-dependent frequency response of a filter with limiter.

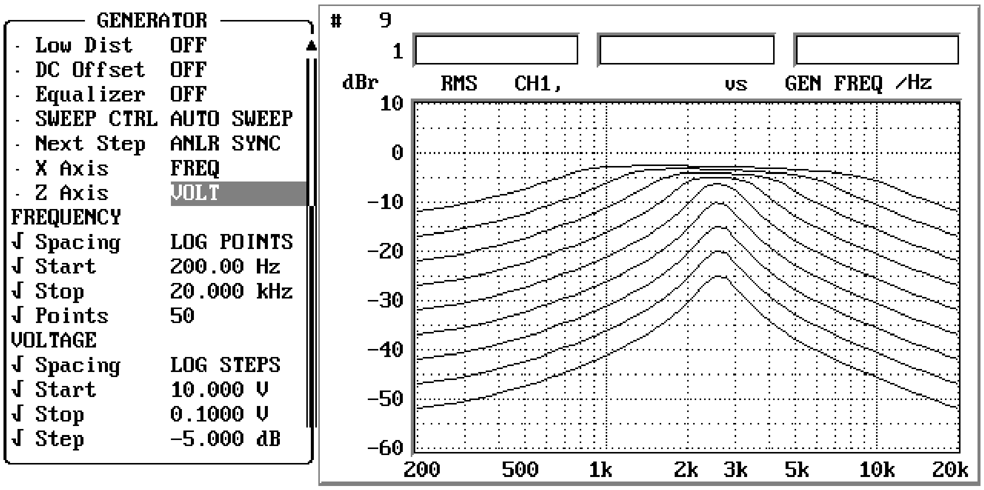


Fig. 4: Display of frequency response of filter with limiter at different levels

As mentioned under 4.1.1, the generator signal is swept over a frequency range from 20Hz to 20 kHz. Moreover, a second sweep parameter can be activated in the GENERATOR panel; for this setup, the parameter *Z Axis VOLT* has been selected. The start level is 1V, the stop level 0.1 V and the stepwidth -5 dB. The first frequency sweep is performed at a level of 1V, in each following sweep the level is reduced by 5 dB.
As a result, five traces are obtained. Single-channel representation has been selected for this setup in the interest of a clear-cut display. The second channel can be displayed by activating *Trace B* in the DISPLAY panel.

In this way, any number of traces can be displayed. For each channel, the last 17 traces can be stored. The measured data of the last 17 traces can further be evaluated using the cursors; switchover between traces can be made by means of the Page up / Page down keys.

4.1.5. Level Difference between Two Stereo Channels

Setups: **LEVDS_AA.SAC** **LEVDS_AD.SAC**
LEVDS_DA.SAC **LEVDS_DD.SAC**

With stereo equipment it is sometimes of interest to display channel unbalance as a function of frequency.

The measurement procedure is the same as for the dual-channel frequency response measurement described under 4.1.1, "Sweep Measurements". However, *Trace A* shows the frequency response for channel 1, whereas *Trace B* shows the level difference for channel 2 referred to channel 1. This is possible by setting the reference value for channel 2 not to a fixed value but taking the current measured value of channel 1 as a reference (setting: *Reference MEAS CH1*).

It is also possible to display the differential trace alone. To this end, simply switch off *Trace A*.

As another application, this setup can be used for determining the difference between the current and the previous frequency response of a DUT. For this, the stored results of previous measurements are required; the stored data will be taken as a reference for the current measurement when the corresponding file is called under *Reference*.

4.2. Phase and Group-Delay Measurements

Definitions and test conditions:

Measurement of the phase frequency response of amplifiers is also defined by DIN IEC 238-3.
Same as amplitude frequency response, phase frequency response is measured under standard test conditions. The input signal is swept over the frequency range at a constant level; results are graphically displayed versus frequency.

In the standard, a differentiation is made between two measurements:

- Determination of the **phase frequency response,** the phase difference between the input and the output of a DUT is measured and displayed versus frequency.
- Determination of the **phase difference,** the phase difference between the two stereo output channels of a DUT is measured and displayed graphically.

4.2.1. Measurement of Phase Frequency Response

Setup: **PHAS_AA.SAC**

With this setup, a logarithmic sweep with 50 frequency points is performed from 20 Hz to 20 kHz. UPD/UPL always measures the phase difference between its two input channels. To avoid any recabling of the DUT, channel 2 of the analyzer has for this measurement been internally connected to the channel 1 output of the generator. As a result, the phase difference between the input and the output of the DUT is measured in channel 1 of the DUT. Fig. 5 shows the setting for this setup.

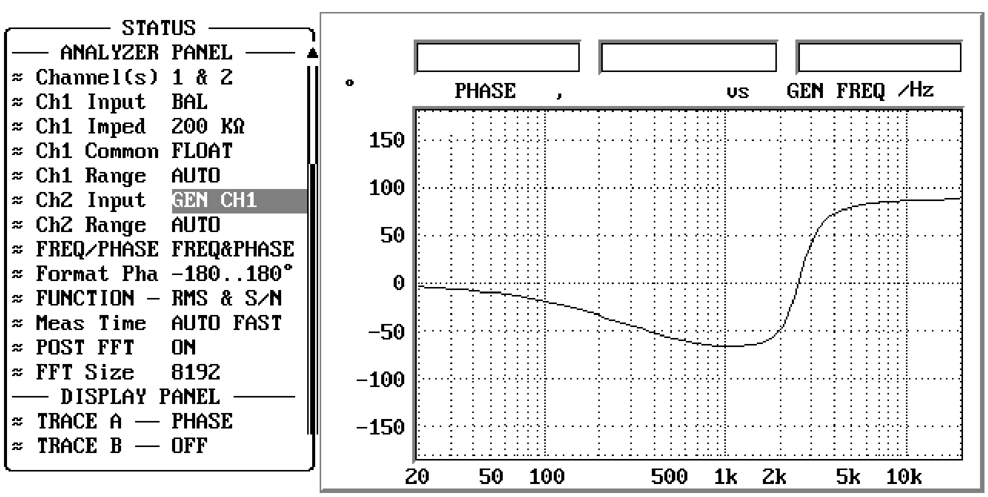


Fig. 5: Setting for measurement of phase frequency response

To measure the phase frequency response of the two stereo channels of a DUT, proceed as follows: after measuring channel 1, switch over to *HOLD* under *TRACE A* in the DISPLAY panel. Activate the display of phase frequency response results for channel 2 under *TRACE B*. Then reconnect the analyzer inputs: connect *Ch1 Input* to the internal generator and *Ch2 Input* to the DUT. When the sweep is restarted, the phase frequency response for the second channel will be displayed.

4.2.2. Measurement of Phase Difference between Two Stereo Channels

Setups: **PHADS_AA.SAC** **PHADS_DD.SAC**

The procedure is similar to that described under 4.2.1 except that in this case both output channels of the DUT are connected to the inputs of UPD/UPL. The phase difference between the two stereo channels is displayed with channel 1 taken as a reference.

4.2.3. Measurement of Group Delay Versus Frequency

Setup: **GRPS_AA.SAC**

For measuring the group delay, the information given under 4.2.1, "Measurement of Phase Frequency Response", applies analogously. Instead of phase measurement, the *FREQ&GRPDEL* setting is selected in the *FREQ/PHASE* line of the ANALYZER panel.

4.3. Combined Measurements

In the following setups, the results of amplitude and phase frequency response measurements are combined in one graphic display. For measurement procedures see relevant sections above.

4.3.1. Amplitude and Phase Frequency Response in One Display

Setup: **PHLVS_AA.SAC**

Combined display of amplitude and phase frequency response for channel 1.

4.3.2. Phase Difference and Level Difference between Two Stereo Channels in One Display

Setups: **PDLDS_AA.SAC** **PDLDS_DD.SAC**

Combined display of phase and level difference between two stereo channels referred to channel 1.

4.3.3. Group Delay and Amplitude Frequency Response in One Display

Setup: **GRLVS_AA.SAC**

Combined display of group delay and amplitude frequency response for channel 1.

5. Nonlinear Distortion Measurements

Nonlinear distortion is a variation of the signal shape caused by amplification in the transmission system as a function of the amplitude. In contrast to linear distortion, frequency components that are not contained in the input signal are generated.

Notes on measurements of A/D converters:

Modern A/D converters have a very high resolution, so that great importance is attached to the quality of analog signal generation. It may therefore be necessary, for all nonlinear distortion measurements, to use the optional Low Distortion Generator UPD-B1 or UPL-B1 for test signal generation. The setups described here use the universal generator incorporated in the audio analyzers as standard.

5.1. Total Harmonic Distortion (THD)

Setups:	**THD_AA.SAC**	**THD_AD.SAC**
	THD_DA.SAC	**THD_DD.SAC**

Definitions and test conditions:

Distortion is defined by DIN IEC 268-2. To measure distortion, an amplifier is driven with a sinusoidal signal under standard test conditions.

To determine total harmonic distortion, the amplitudes of the harmonics at the output of the DUT are measured, their rms values added and a ratio is formed to the total signal.
The result is indicated as distortion in % or as total harmonic distortion in dB.

Total harmonic distortion as a function of amplitude or frequency is measured analogously.

Measurement of nth-order distortion is performed in the same way except that in this case it is not the rms value of all harmonics that is determined but only individual harmonics are determined or combinations of specific harmonics used for calculating distortion.
An example of such a measurement is the 3rd harmonic specified for tape recorders.

Harmonic distortion or THD is a measure of quality mainly in the lower and middle frequency ranges. For a fundamental frequency of 8 kHz, for example, the 2nd harmonic of 16 kHz is already at the limit of hearing. The 3rd harmonic of 24 kHz is outside the audio transmission range. Harmonic distortion is therefore not suitable for describing nonlinear characteristics at higher frequencies.

Graphic display:

Total harmonic distortion can be indicated by means of a single measured value. With UPD/UPL, however, the spectral distribution of intermodulation products can be displayed, see Fig. 6.
Distortion as a function of frequency, for example, will be shown as a graph same as for frequency response.

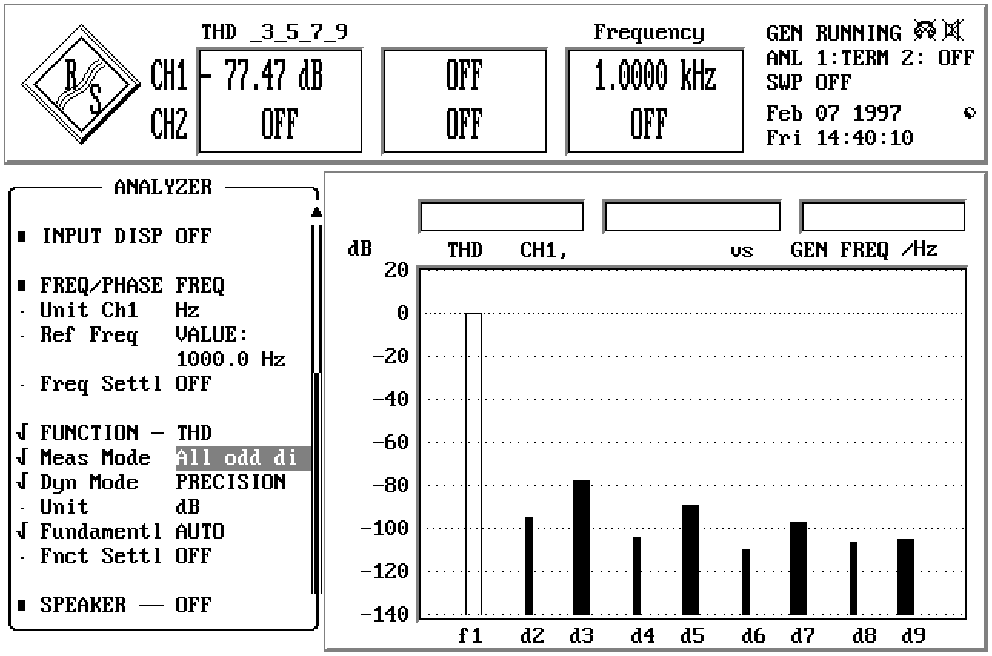


Fig. 6: THD measurement with display of distortion products

Notes on measurements:

In practice, DUTs frequently have a quadratic or cubic characteristic. This means that even-numbered or odd-numbered distortion products are predominant in the spectrum. This allows conclusions to be drawn as to the cause of harmonic distortion:

- a quadratic characteristic is obtained with unsymmetric distortion. Example: different gain for positive and negative halfwaves of a push-pull stage

- a cubic characteristic is obtained with symmetric distortion; this is typical of any type of overdriving. Examples: saturation with tape recorders, max. deflection of loudspeaker coils.

Audio Analyzers UPD and UPL allow distortion measurement up to the 9th harmonic as shown in the setups presented here. If a single harmonic is to be taken into account, this is selected in the *Measurement Mode* line. In the spectral display, the selected harmonics are shown as wide bars, the remaining harmonics as narrow bars. The components used for measurement are also indicated in the measured-value display.

5.2. THD+N

Setups: **THDN_AA.SAC** **THDN_AD.SAC**
THDN_DA.SAC **THDN_DD.SAC**

Definitions and test conditions:

Same as THD measurements, THD+N measurements use a sinusoidal signal to drive the DUT. However, in THD+N measurements, all spurious signals are taken into account in the result. This means that, in addition to harmonic distortion and noise, other signal components such as mixture products formed with the clock frequency in digital signal processing are taken into account in the result. To evaluate such spurious signals, spectral analysis must be performed in addition to THD+N measurement.

When comparing measurements the bandwidth must be taken into account.

In accordance with AES 17, THD+N are to be performed at a level of -1 dBFS and -20 dBFS. The measurement bandwidth is limited to half the sampling frequency and must not exceed 20 kHz.

In the setups described here a measurement bandwidth of 100Hz to 20 kHz has been selected, the analog output level is 1 V, the digital level -1 dBFS.

Graphic display:

The THD+N value can be indicated by means of a single measured value. With UPD/UPL, however, the spectral distribution of output products can be displayed using the post-FFT function and harmonics marked automatically as shown in Fig. 7. This enables nonharmonic signal components to be detected very easily.

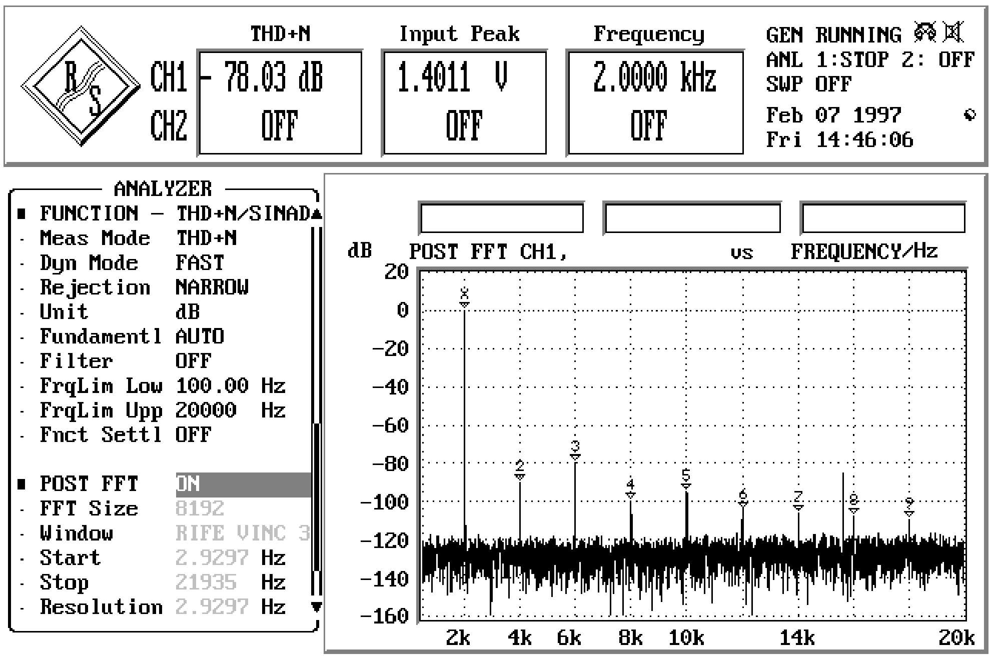


Fig. 7: THD+N measurement with distortion products marked

Notes on measurements:

This parameter too can be measured as a function of frequency or level.

With setups **THDNS_AA.SA** **THDNS_DD.SAC**

a linear frequency sweep from 20 Hz to 20 kHz is performed and the THD+N value displayed versus frequency.

5.3. Intermodulation

Setups: **MOD_AA.SAC** **MOD_DD.SAC**

Definitions and test conditions:

Instead of a single sinusoidal signal, a signal composed of two frequencies, f_1 and f_2, is used, which yields not only the harmonics mf_1 and nf_2 described above but also combination signals with the frequencies $(mf_1 \pm nf_2)$. The occurrence of these signals is referred to as intermodulation.

To determine the modulation distortion in accordance with DINIEC 268-3, an amplifier is operated under standard test conditions and driven with a two-tone signal. The frequencies of the two sinusoidal input signals should be such that f_1 is 0.5 to 1.5 octaves above the lower limit and f_2 0.5 to 1.5 octaves below the upper limit of the transmission range. The level ratio is 4:1. To calculate modulation distortion, the squares of the four mixture products formed by the 2nd-order intermodulation distortion ($f_2 + f_1$ and $f_2 - f_1$) and the 3rd-order intermodulation distortion ($f_2 + 2f_1$ and $f_2 - 2f_1$) are added up and the result referred to the level of signal f_2 with the higher frequency.
The result is indicated in % or in dB.

Graphic display:

As in the case of distortion measurements, the spectral distribution of the components can be displayed in addition to the measured value.

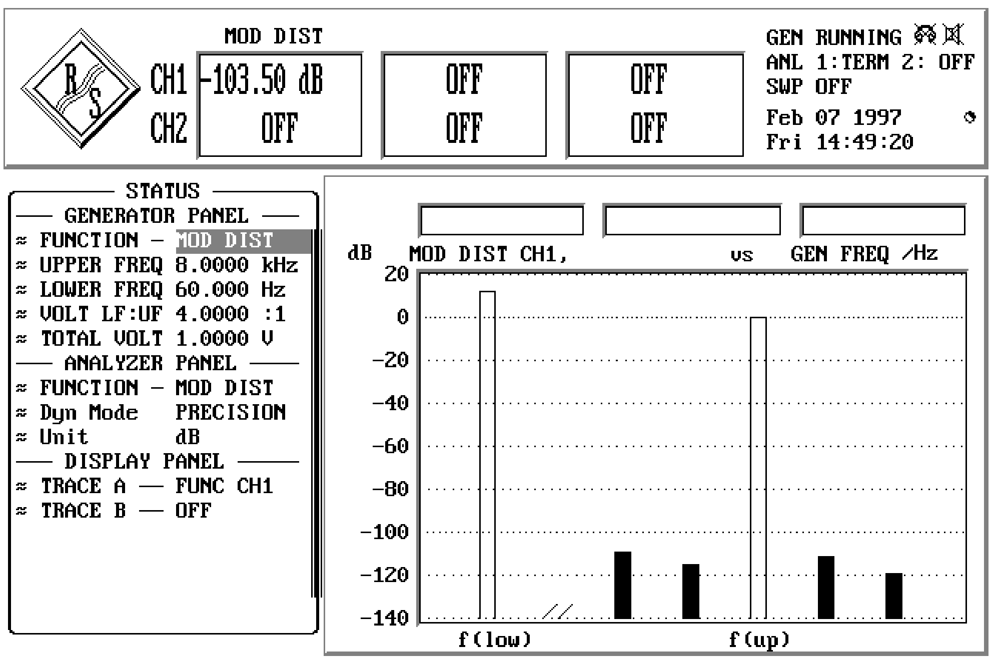


Fig. 8: Measurement of modulation distortion with graphic display of intermodulation products

Notes on measurements:

In the setups, a level ratio of 4:1 of the sinusoidal signals is selected. Setup MOD_AA.SAC uses 60 Hz and 8 kHz with -10 dB level. Setup MOD_DD.SAC uses 41 Hz and 7993 Hz with full-scale amplitude of the DUT, the latter in compliance with AES17.

If other test signals are to be used, the relevant lines in the GENERATOR panel are to be changed accordingly. No modifications are required in the ANALYZER panel; the analyzer automatically adjusts to the test signal.

5.4. Difference Frequency Distortion (DFD)

Setups: **DFD_AA.SAC** **DFD_DD.SAC**

Definitions and test conditions:

The difference frequency distortion is determined in a similar way as modulation distortion but using a test signal composed of two sinusoidal frequencies f_1 and f_2 of equal amplitude. The difference between the two frequencies is smaller than the lower frequency value. The voltage of the difference frequency $f_2 - f_1$ is measured whose position in the spectrum does not change as long as the frequency difference remains constant (2nd-order DFD). The 3rd-order DFD is determined from mixture products $2f_1 - f_2$ and $2f_2 - f_1$.

Because of the small frequency differences used here, great demands are made on the selectivity of the instrument regarding the measurement of 3rd-order DFD, especially when bandpass filters are used. Modern audio analyzers employ FFT analysis for this measurement, results are calculated automatically in line with standards.

Measurement of difference frequency distortion is defined by various standards that differ as follows:

- For measurements on amplifiers, DIN IEC 268-3 defines the test signals on the basis of a fixed frequency spacing (mainly 80 Hz) and the arithmetic mean frequency.
  Results are referred to twice the output voltage of f_2, the absolute values of the two components $2f_1 - f_2$ and $2f_2 - f_1$ being added for determination of the 3rd-order DFD.

- IEC 118 defines the DFD for measurements on hearing aids. Here, the upper frequency and the difference frequency are specified.
  Results are referred to output voltage f_2, and the 3rd-order DFD is determined by means of component $2f_1 - f_2$ only.

  The results obtained with the two standards thus differ by 6dB for d_2 and are equal for d_3 provided the levels of $2f_1 - f_2$ and $2f_2 - f_1$ do not substantially differ from each other.

- For measurements on digital components, differential frequency distortion measurement is defined by AES 17, the measurement being in this case referred to as intermodulation measurement. The standard defines as test frequencies the upper limit frequency based on the selected sampling rate as well as the frequency 2 kHz below the limit frequency. The peak value of the total signal is to be adjusted such that it is equal to the peak value of a sinusoidal signal at full-scale amplitude. As with IEC 268, results are referred to the total output signal of the DUT.

Graphic display:

Same as modulation distortion analysis.

Notes on measurements:

The setup for measurements on purely analog components is in line with IEC 268-3, the center frequency is 10 kHz, the difference frequency 80 Hz. If other frequencies are to be used, the relevant lines in the GENERATOR panel are to be changed. No modifications are required in the ANALYZER panel; the analyzer automatically adjusts to the test signal.
For measurements to IEC 118, the relevant lines in the GENERATOR and ANALYZER panels are to be changed.

The setups for measurements on digital components give a test signal with a 20-kHz and an 18-kHz tone at full-scale amplitude.

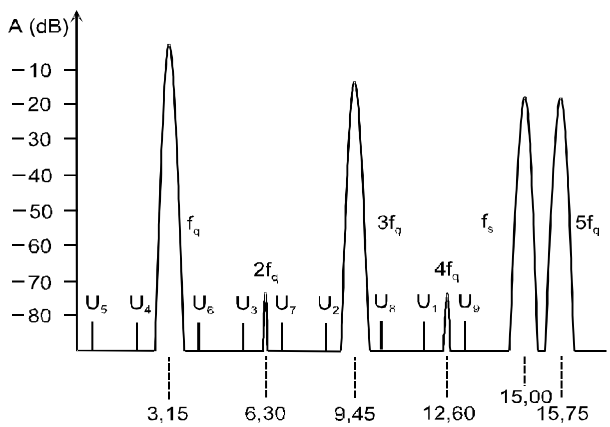
5.5. Dynamic Intermodulation (DIM)

This measurement function is implemented in Audio Analyzer UPD only. In addition, Option UPD-B1 (Low Distortion Generator) is required for measurements on analog interfaces.

Setups: **DIM_AA.SAC** **DIM_DD.SAC**

Definitions and test conditions:

To determine dynamic intermodulation distortion in line with DINIEC 268-3, the amplifier is operated under nominal conditions (ie at full-scale amplitude) and driven with an input signal consisting of a rectangular and a sinusoidal signal. The rectangular signal has a fundamental frequency of 3.15kHz and is band-limited to 30 kHz by means of a single-pole lowpass filter (100kHz are also permissible). The sinusoidal signal has a frequency of 15kHz and a level 12 dB below that of the rectangular signal.
The nine intermodulation products in the audible range are measured selectively, the sum of the squares of the products is formed, referred to the rms value of the sinusoidal signal, and indicated in % or dB.



Intermodulation components Frequency	Voltage	Frequency kHz
$5f_q - f_s$	U_5	0,75
$f_s - 4f_q$	U_4	2,40
$6f_q - f_s$	U_6	3,90
$f_s - 3f_q$	U_3	5,55
$7f_q - f_s$	U_7	7,05
$f_s - 2f_q$	U_2	8,70
$8f_q - f_s$	U_8	10,20
$f_s - f_q$	U_1	11,85
$9f_q - f_s$	U_9	13,35

Fig. 9: Test signal for DIM distortion and intermodulation products to be measured

DIM distortion is obtained due to the short rise time of the rectangular signal which causes the amplifier to be driven dynamically to the limit of its slew rate. This test signal is to provide a better correlation between subjective hearing tests and measured results since in this case, similar as with music containing pulses, the amplifier output voltage changes very rapidly. Low-distortion amplification of the 15kHz signal must be performed at the same time. Especially amplifiers with unfavourably designed negative feedback will respond with distortion.

Graphic display:

Audio Analyzer UPD can display the spectral distribution of the components in addition to the measured value.

Notes on measurements:

The setups for measuring dynamic intermodulation are based on IEC268-3. In sound broadcasting, a rectangular/sinewave signal combination of 2.96/14kHz is used instead of the 3.15/15-kHz frequency combination. The 2.96/14 kHz test signal can also be generated and analyzed with UPD.

5.6. FFT Analysis

Setups: **FFT_AA.SAC** **FFT_AD.SAC**
FFT_DA.SAC **FFT_DD.SAC**

FFT analysis is used where the spectral composition of a signal is to be examined. Audio Analyzers UPD and UPL provide a highly efficient tool for this purpose.

The setups described here generate a 1-kHz or 997-Hz test signal with a level of 1 V or -20 dBFS which can be applied to the DUT. The FFT analysis itself requires a minimum off settings. *FFT Size* defines the number of samples on which calculation is based. Higher *FFT Size* values give higher frequency resolution but at the same time entail longer measurement times. The setups were generated for 8k FFT; due to the high measurement speed of UPD/UPL it will only very rarely be necessary to select a lower number of points and thus obtain an even faster measurement.

Various *windows* are available to accommodate for a wide variety of applications. The setups use the Rife-Vincent window, which is characterized by steep slope of the bell lobe and excellent far-off interference suppression.

With noisy signals, spectral averaging may be useful sometimes. This can be performed using the *Average Mode* function; the type and number of averaging measurements can be entered.

Very closely spaced frequency components can be analyzed by means of the *Zooming* function. In contrast to the zooming function of the graphic display, the FFT zooming function actually yields higher measurement resolution since the signal is preprocessed in the time domain before the FFT calculation takes place. By entering *Center* and *Span,* the center frequency and the spread range for the zoom FFT are defined.

6. Measurement of Interference and Wow & Flutter

6.1. S/N Ratio

Setups: **SNRA_AA.SAC** **SNRA_AD.SAC**
SNRA_DA.SAC **SNRA_DD.SAC**
SNRC_AA.SAC **SNRC_AD.SAC**
SNRC_DA.SAC **SNRC_DD.SAC**

Definitions and test conditions:

The S/N ratio is the ratio in dB of the nominal output voltage to the sum of the broadband or weighted measured output voltages with the source EMF set to zero.
To determine the S/N ratio, the output voltage of the amplifier is measured under nominal conditions (ie the nominal output voltage V_{2ref} at full-scale amplitude of the DUT is measured). Then the source EMF is reduced to zero and the noise voltage V_2' is measured. The result is indicated as noise level V_2' or as S/N ratio 20 lg (V_{2ref}/V_2') db.
Audio Analyzers UPD and UPL provide S/N ratio measurements as automatic test sequences.

S/N ratio measurements are covered by a variety of test standards and procedures. These differ mainly in:

- the type of weighting filter used for simulating hearing sensitivity as a function of frequency,
- the type of detector used.

For linear audio noise voltage measurements, the unweighted rms noise voltage is measured in accordance with DIN 45412 "Noise voltage measurements on sound broadcast receivers and related equipment". In this measurement, a bandpass filter of 22.4Hz to 22.4 kHz is used for limiting the measurement bandwidth approximately to the range of audibility.

DIN 45412 further defines a commonly used method of S/N ratio measurement in which an A filter is used and the rms noise voltage determined.
The steep roll-off of the A curve with decreasing frequencies results in a strong attenuation of hum components, which is expedient for this measurement as it truly reflects hearing conditions.

The standard prescribes the use of an rms detector, so the average noise power is measured. However, the ear is very sensitive to sound containing pulses (noise peaks, clicking noise). Therefore, increasing use is made of a quasi-peak detector to CCIR 468-4 or DIN 45405.

The standard DIN 45405 "Noise voltage measurement in sound engineering" technically coincides with CCIR Recommendation 468 "Measurement of audio-frequency noise in broadcasting, in sound-recording systems and on sound program circuits". It defines, for example, filter curves for weighted and unweighted measurements.
For unweighted noise level measurements, the same bandpass filter as defined by DIN 45412 is used.

Measurement of noise as defined by DIN IEC 268-3 (amplifiers) provides for measurements using A filters and rms weighting as well as measurements to CCIR 468-3 (corresp. to DIN 45405).

Notes on measurements:

Setups SNRA... measure the S/N ratio as a weighted rms value using an A filter, setups SNRC... use a quasi-peak detector and a CCIR filter.
It should be noted that the quasi-peak detector requires a settling time of approx. 3 s to supply valid results. This time is set in the setup.

Apart from the filters used in the above S/N ratio measurements, a wide variety of other weighting filters is in use in practice. In digital applications, for example, weighting is performed with a CCIR ARM filter which is also known as CCIR 2k filter. It differs from the CCIR weighted filter in its reference frequency of 2 kHz (normally 1 kHz). Moreover, this measurement is rms-weighted. In the ANALYZER panel, any other weighting filter can be selected in the *Filter* line. The automatic S/N test sequence can be switched on or off in the *S/N Sequ* line.

A comparison of results of noise voltage measurements is possible only if the test conditions regarding detectors, weighting filters and measurement bandwidth are observed. Depending on the type of measurement, deviations of more than 10 dB may be obtained.

6.2. Crosstalk

Setups:	**CRSS_AA.SAC**	**CRSS_AD.SAC**
	CRSS_DA.SAC	**CRSS_DD.SAC**

Definitions and test conditions:

In accordance with DIN IEC 268-3, the level difference between the output signal of a fully driven channel and the output signal of a channel that is not driven is measured. The measurement is prescribed for both directions, and the results may differ due to asymmetries of the setup. The measurement is mandatory at the reference frequency and optional at further frequencies. Often measurements are made over the entire frequency range and results displayed graphically.

Both broadband and selective measurements are possible. Since with high-quality DUTs, the crosstalk level is in the vicinity of noise, only selective measurements are expedient in this case.
For crosstalk, measured values will always be <1 (negative dB value) since the measured voltage is referred to the nominal output voltage.
For crosstalk attenuation, which is likewise specified, values >1 (positive dB values) will be obtained since the reference used is reversed.

Crosstalk is measured at a level of -20 dBFS in accordance with AES 17.

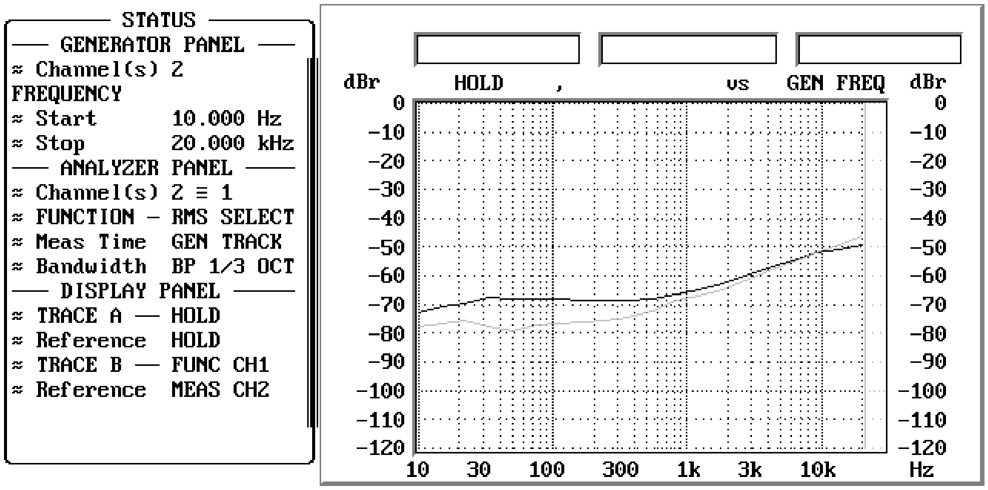


Fig. 10: Example of crosstalk measurement with required settings

Notes on measurements:

Crosstalk measurement is not a separate measurement function on Audio Analyzers UPD and UPL, it is performed as a level measurement, with results being referred to the level values obtained for the other channel in each case.

The setups determine crosstalk in both directions. Measurements are made selectively with the aid of the sweep function from 10 Hz to 20 kHz. The following additional settings are to be made:

- In the basic setting of the setup, the test signal is output to channel 1 only. The level for the "analog" setup is at 1 V and must be set to maximum level of the DUT.
  For all other setups, the level is to be set to -20 dBFS.
- The sweep is started and crosstalk coupled into channel 2 displayed graphically. In the measurement, the level of channel 2 is determined and continuously referred to the level measured for channel 1 (setting: *Reference MEAS CH 1*).
- Now set *TRACE A* to *HOLD* in the DISPLAY panel and activate *TRACE B* to display the results of channel 1.
- Set the reference for the results of channel 1 to *MEAS CH 2* in the *Reference* line.
- Set the *Channel(s)* line in the GENERATOR panel from *1* to *2* so that the other channel will be driven.
- When the sweep is restarted, the crosstalk from channel 2 to channel 1 will be displayed in the diagram.

Since settings in various panels have to be modified when changing from one channel to the other, the use of the status panel is expedient in this case. In this panel the key command lines for an application can be combined and settings made in the status panel. In the setups, this feature has been taken into account, see Fig. 10.

6.3. Stereo Separation

Setups: **SEPS_AA.SAC** **SEPS_AD.SAC**
SEPS_DA.SAC **SEPS_DD.SAC**

Definitions and test conditions:

Measurement of the stereo separation is very similar to crosstalk measurement. In the latter, one channel is driven and the levels measured for the two channels and correlated to one another. In stereo separation measurement, on the other hand, only one channel is measured and the test signal switched between the two channels. If, as is usually the case, the input levels for the two channels are equal, identical results will be obtained for stereo separation and crosstalk measurements. Crosstalk measurements can however easier be integrated into a sweep and are therefore performed almost exclusively today. It has become common practice however to use the term "stereo separation" instead of "crosstalk".

For digital applications to AES 17, stereo separation measurements are performed the same as crosstalk measurements. For stereo separation, positive dB values are obtained.

Notes on measurements:

For the measurement procedure, the information given under 6.2 "Crosstalk" applies analogously.

6.4. Wow & Flutter

Setups: **WFI_AA.SAC** **WFN_AA.SAC**
WFJ_AA.SAC

Definitions and test conditions:

When storing analog sound signals on moving media, the sound quality depends on the mechanical precision of the transport mechanism used. Short-term variations in speed will result in frequency fluctuations of the sound signal.
To measure the frequency fluctuations, which are referred to as wow & flutter, a sinusoidal tone is played, FM-demodulated and the signal measured.
Since hearing sensitivity is greatest at modulation frequencies of 4Hz, wow & flutter measurements are frequently performed by means of a 4-Hz weighting filter.

Wow & flutter measurements are covered by various standards differing in the test signal and the detector used:

- **DIN 45507 / IEC 386 / CCIR 409-2**
  Reference frequency: 3.15 kHz Evaluation: quasi-peak detector
- **NAB Recommendation**
  Reference frequency: 3 kHz Evaluation: average detector
- **Japan Industry Standard**
  Reference frequency: 3 kHz Evaluation: rms detector

Notes on measurements:

Three setups matched to standards DIN/IEC, NAB and JIS are available for wow & flutter measurements. Not only the analyzer but also the generator is set, for instance for the recording of signals on test tapes.

7. Measurements on Analog/Digital Interfaces

7.1. Clipping Level

Setup: **CLIP_AD.SAC**

Definitions and test conditions:

Components using internal digital signal processing must not be overdriven since any loading in excess of the digital level range would result in strong distortion of the signal (clipping level). The full-scale amplitude therefore plays a far more important role in digital than in analog applications.
The clipping level must be determined for all digital components with analog input stage. If digital outputs are accessible, this is accomplished by increasing the level of a 997-Hz sinusoidal input signal until the peak value of the digital output signal equals the largest data word (full scale).

The level thus obtained defines the full-scale amplitude of the digital system and is used as a reference value in a variety of measurements.

Notes on measurements:

The setup supplies an analog output signal of 997Hz. The level is set to 1 V.

As the clipping level serves as a reference in a variety of other measurements, it is expedient to use the *Ref Volt* function of UPD/UPL. The data can then be entered in dBr in the *VOLTAGE* line of the GENERATOR panel, which does away with the need for constantly converting the levels to the clipping level.

To determine the full-scale amplitude as described above, the level of the generator signal is increased in the *Ref Volt* line until the analyzer indicates the peak value of 0dBFS. In doing this, it must be ensured that the full-scale value is not exceeded in none of the channels.
The clipping level thus obtained can then be transferred to all setups used for measurements on that particular DUT; any other level entries are made in dBr in the *VOLTAGE* line.

7.2. Linearity of A/D Converters

Setup: **LINS_AD.SAC**

Definitions and test conditions:

A 997-Hz sinusoidal signal is applied to the input of the DUT. The level of this signal is decreased in steps of 5 dB starting from the full-scale amplitude. The output signal is measured and represented graphically versus the input signal. As the signal disappears in the noise with decreasing level, narrowband measurement using a third-octave bandpass filter is performed.

In the setup described here, converter linearity is measured by means of a level sweep from 0 dBr to -120 dBr. With a linear response of the converter, a diagonal is obtained as shown in the graphic display of Fig. 11.

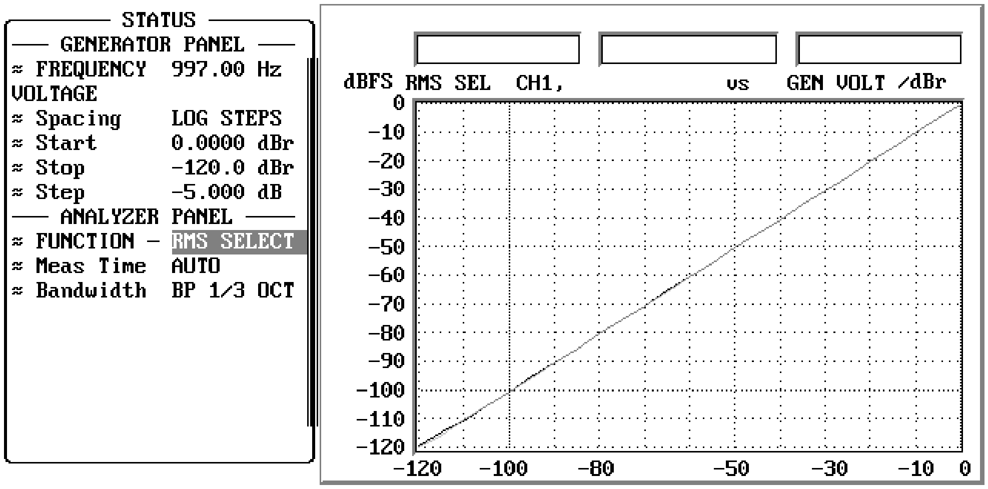


Fig. 11: Linearity measurement of A/D converter

Since deviations from the nominal characteristic are difficult to recognize in the above type of representation, level nonlinearity is measured in most cases, which is described in the next setup.

Notes on measurements:

To drive the DUT at full-scale level, the clipping level determined in the previous measurement is to be entered into the *Ref Volt* line of the GENERATOR panel. This level serves as a reference for all level values defined in dBr in the sweep lines.
The reference level used for x-axis scaling in the graphic display shown above is in this setup automatically transferred from the GENERATOR panel into the *Reference* line under *x Axis*.

7.3. Nonlinearity of A/D Converters

Setup: **LINDS_AD.SAC**

Same as previous setup, but showing deviation from ideal characteristic.

Definitions and test conditions:

This type of measurement is defined by AES17, the test parameter being referred to as level-dependent logarithmic gain. A linearity measurement is performed, the first result is however recorded only at -5 dBFS. For each test step, the logarithmic gain, ie ratio of output amplitude to input amplitude, is to be determined and represented graphically versus the input level. The resulting diagram shows the deviation of the converter transmission characteristic from the nominal linearity characteristic.

Measurements are to be performed selectively using a third-octave bandpass filter.

Notes on measurements:

For this measurement, too, the clipping level determined in accordance with 7.1 "Clipping Level" is to be entered into the *Ref Volt* line of the GENERATOR panel. This level serves as a reference for all level values defined in dBr in the sweep lines.

In the ideal case, a straight line is obtained in the graphic display, any deviation from the ideal characteristic of the converter can be read in dB.

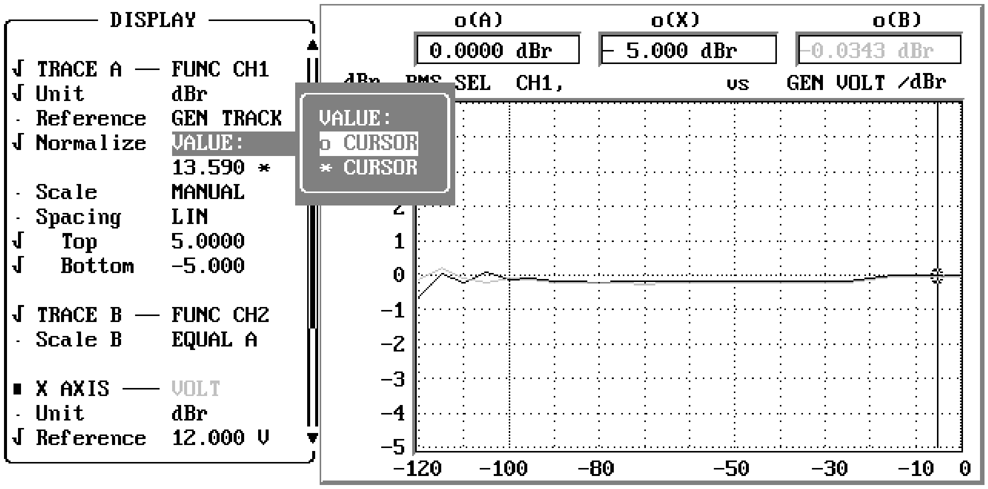


Fig. 12: Nonlinearity of A/D converter, the cursor value is transferred by means of the *NORMALIZE* function

This type of measurement however involves a physical problem, ie referring the digital output voltage of the converter to the analog input voltage at every test point. Audio Analyzers UPD and UPL have an internal "conversion factor" of 1 FS $\triangleq$ 1 V. With this factor, a straight line would be obtained but it would not coincide with the zero line. The gain factor of the DUT must, therefore, be taken into account in addition. This is done by means of the *NORMALIZE* function, which is included in the DISPLAY Panel (see Fig. 12). Here the gain can be entered directly, it is however easier in most cases to transfer this value from the graphic display. To this end, a cursor is placed on the linear section of the curve and the cursor value is transferred to the *NORMALIZE* line by selecting the item *o Cursor*.

7.4. Linearity of D/A Converters

Setup: **LINS_DA.SAC**

Definitions and test conditions:

For linearity measurements of D/A converters, the information given under 7.2 "Linearity of A/D Converters" applies analogously.

In addition it should be noted that for this measurement the digital input signal should contain a dither with a triangular probability density function and a level of 1 LSB. This dither is set in the setup.

Notes on measurements:

The reference value used for graphic display of the level values in dBr is obtained from the gain ratio of the converter, ie the ratio of digital input amplitude to analog output amplitude. With this setup, the reference value is easiest taken from the *Reference* line by selecting the *MAX* item. The maximum value measured will thus be taken as a reference. In this setup, this value corresponds to maximum level of the DUT since the generator sweep is started at 0dBFS.

7.5. Nonlinearity of D/A Converters

Setup: **LINDS_DA.SAC**

Definitions and test conditions:

This measurement too is defined by AES17. The information given under 7.3 "Nonlinearity of A/D Converters" applies analogously.

For this measurement too the digital input signal should contain a dither with a triangular probability density function and a level of 1 LSB. This dither is set in the setup.

Notes on measurements:

The measurement procedure is the same as described under 7.3 for "Nonlinearity of A/D Converters". In this case too the *NORMALIZE* function is needed; the gain of the converter is easiest transferred from the graphic diagram as described for the A/D converter above.

7.6. Signal Delay in Analog and Digital Systems

This measurement function is available only in Audio Analyzer UPL.

Setups:	**DEL_AA.SAC**	**DEL_AD.SAC**
	DEL_DA.SAC	**DEL_DD.SAC**

Definitions and test conditions:

This measurement is used to determine the signal delay between the input and the output of a digital system. In accordance with AES17, a pulse-shaped signal is applied to the DUT. The input and the output signal are displayed on an oscilloscope from which the delay can be read. The measurement is used whenever digital signal processing takes place, also on DUTs with analog or analog/digital interfaces.

Notes on measurements:

The measurement is performed with Audio Analyzer UPL in compliance with AES17. Compared with conventional dual-channel oscilloscopes, UPL offers the advantage that the two stereo channels can be measured simultaneously, thus allowing any delay between the two channels to be detected immediately.

This is possible because of the fact that UPL, using the *Waveform* function, can be triggered not only to the measurement channels but also to a burst signal supplied by the generator. This measurement function ensures that the measurement is started for the two channels exactly time-synchronously with the issue of the test signal. Since the test signal is applied to the input of the DUT, exact triggering to the input signal of the DUT is performed. Internal delays of UPL are taken into account and do not affect results.

For this test, a sine burst with a level of -20 dBFS is generated as prescribed by AES17. The burst consists of a 1-kHz signal which is output 10 times, followed by an interval of 90ms.

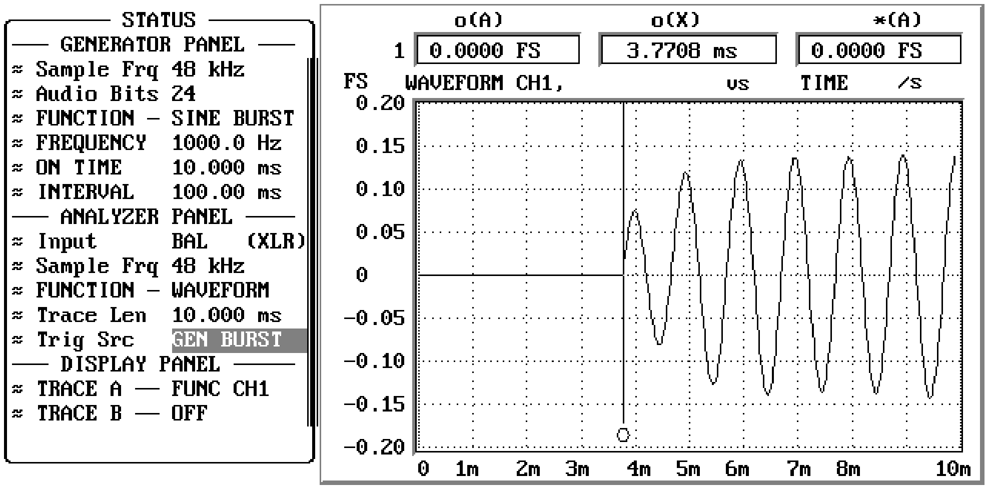


Fig. 13: Measurement of signal delay in digital systems

The signal delay is measured with the aid of the cursor. To this end, the cursor is placed at the point at which the signal departs from the zero line. The graphic window shows the level measured for each cursor position, so this procedure is very easy to perform. The delay is then indicated directly in the second cursor display window (the zero point in the graphic display corresponds to the start of the test signal).

If the measurement is to be performed for both channels, *TRACE B* is to be set to the *FUNC CH 2* measurement function. The second cursor can then be used for channel 2. It is further possible to display the time difference between the two cursors directly by appropriate setting using the softkeys at the bottom of the screen.

With equipment performing filtering of the signal, it may occur that the test signal is attenuated while settling to steady state. To be able to observe this effect in greater detail, a burst with several signal periods is used. Fig. 13 shows an example of such a measurement.

In addition to signal delay, AES 17 describes determination of the polarity between the input and the output signal. This is likewise performed with the setup described here. Polarity reversal is indicated if the displayed output signal does not start with the positive half-wave as is the case with the test signal from the generator.

8. Protocol Analysis

For digital data transmission to AES 3 or IEC 958 additional bits are included in the data stream. Channel status and user data can be generated and evaluated by means of Audio Analyzers UPD and UPL. Transmission errors can be displayed as well, and UPD even allows errored protocols to be generated.

For protocol analysis UPD has to be fitted with Option UPD-B2 (AES/EBU Interface) only whereas UPL requires Option UPL-B21 (Protocol Analysis) in addition to Option UPL-B2 (Digital Interfaces).

8.1. Binary Data Protocol

Setup: **PROTB_DD.SAC**

With the aid of this setup the user can generate the entire protocol by entering binary numbers and have it displayed in binary format as well. This facility is above all used by circuit design engineers to generate specific bit patterns.

8.2. Channel Status Data in Professional Format (AES 3)

Setup: **PROTP_DD.SAC**

This setup generates and evaluates the entire data protocol in line with AES 3 specifications. All information is displayed in plain text, ie decoded form. If the generator panel is switched to *ENHANCED*, the current CRC and time code can be calculated and output. With *STATIC* (on UPL) these data are calculated once and are not updated for subsequent output.

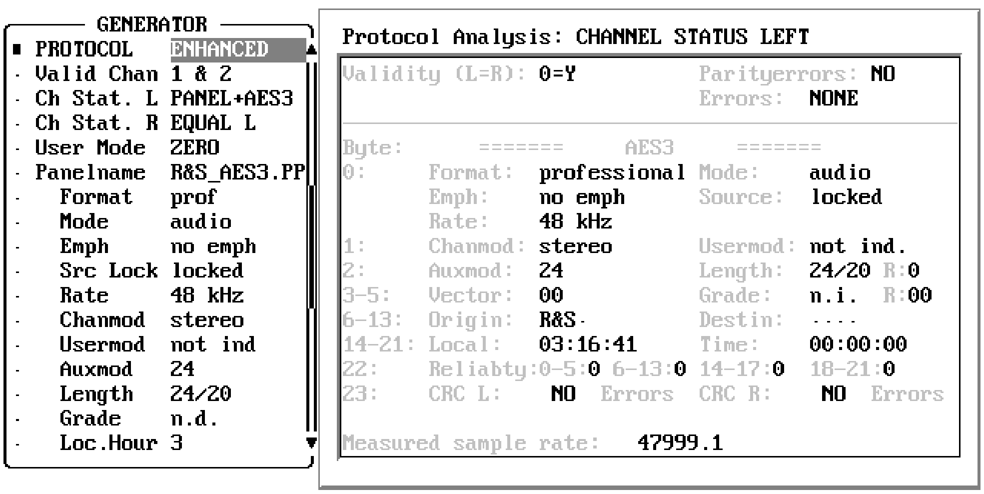


Fig. 14: Protocol analysis evaluated to AES3

8.3. Channel Status Data in Consumer Format (IEC 958)

Setup: **PROTC_DD.SAC**

Section 8.2 "Channel Status Data in Professional Format (AES3)" apply analogously, generation and evaluation of protocol data is in line with IEC958.

9. Digital Interface Tests

For error-free transmission of digital audio data, the physical digital signal must be investigated in addition to the digitally coded audio signal. For this purpose a physical digital signal with analog parameters such as pulse amplitude and clock frequency is used.

To be able to perform digital interface tests UPD has to be fitted with Options UPD-B2 (AES/EBU Interface) and UPD-B22 (Jitter and Interface Test), and UPL with Options UPL-B2 (Digital Audio Interfaces) and UPL-B22 (Jitter and Interface Test).

For these tests the measuring instrument has to be configured as follows:

- In the *Src Mode* line of the digital generator panel the user determines the tasks for which the built-in generators are to be employed. These signal sources can generate audio data and/or operate as jitter modulator and superimpose a balanced common-mode signal onto the audio data stream.
- The application of the built-in measurement functions is specified in the*Meas Mode* line of the digital analyzer. The measurements functions allow not only the audio contents to be analyzed but can also be used for jitter and digital phase measurements or for determining other physical signal parameters such as pulse amplitude or common-mode signals.

9.1. Common-Mode Interference, Digital Pulse Amplitude and Sampling Frequency

Setup: **DCOM_DD.SAC**

In the following example the generator is used to superimpose a sinewave common-mode signal of 100 mV onto the balanced audio line. Signal shape, level and sweep functions can be set. This setup allows the degree of immunity of the DUT to such type of interference to be checked.
The analyzer measures the incoming common-mode components. Level measurement, FFT and waveform function can be selected, the setup shows the rms value of the common-mode signal.

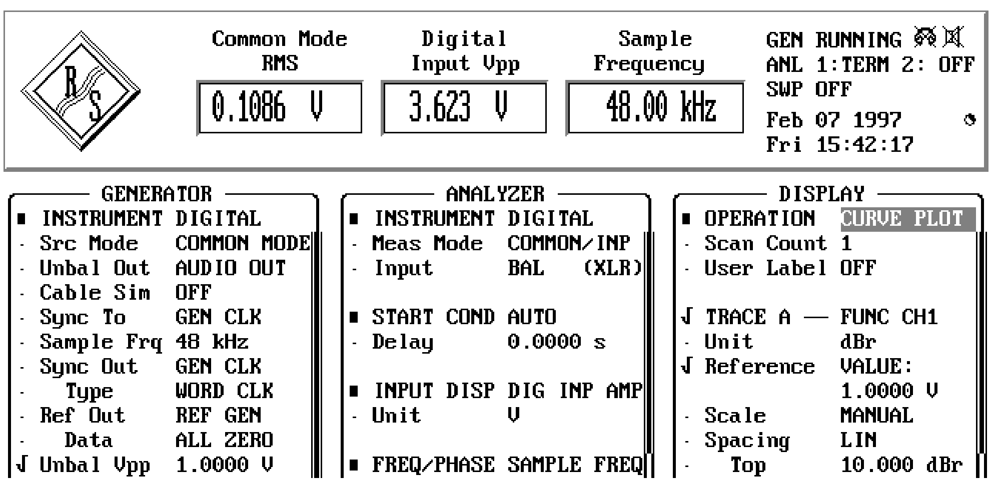


Fig. 15: Setup example and display of physical signal parameters

To find the digital input levels to which a DUT synchronizes correctly, it is possible to set the pulse amplitude of the digital data stream in the generator panel. The switchable cable simulator simulates the lowpass behaviour of a 100m cable by smoothing the signal edges.
With this setup the analyzer indicates the pulse amplitude as a peak value. Cable attenuation can be determined by referring the measured value to the generator level.

Digital input stages must also be able to handle clock frequency offsets. To investigate the capture range the user can vary the clock frequency in the generator, the analyzer displaying this frequency.

9.2. Jitter Amplitude

Setup: **JITAM_DD.SAC**

In this example the generator is used as jitter source. A sinewave jitter of 100Hz with an amplitude of 0.1 UI has been selected. If necessary, the jitter signal can also undergo sweeping.

The analyzer indicates the jitter amplitude in all common units.

9.3. Jitter Spectrum

Setup: **JITSP_DD.SAC**

Setup of generator as described above. FFT analysis is performed. The jitter spectrum reveals the cause of the jitter if, for example, a frequency used in the switching power supply is included in the jitter spectrum.

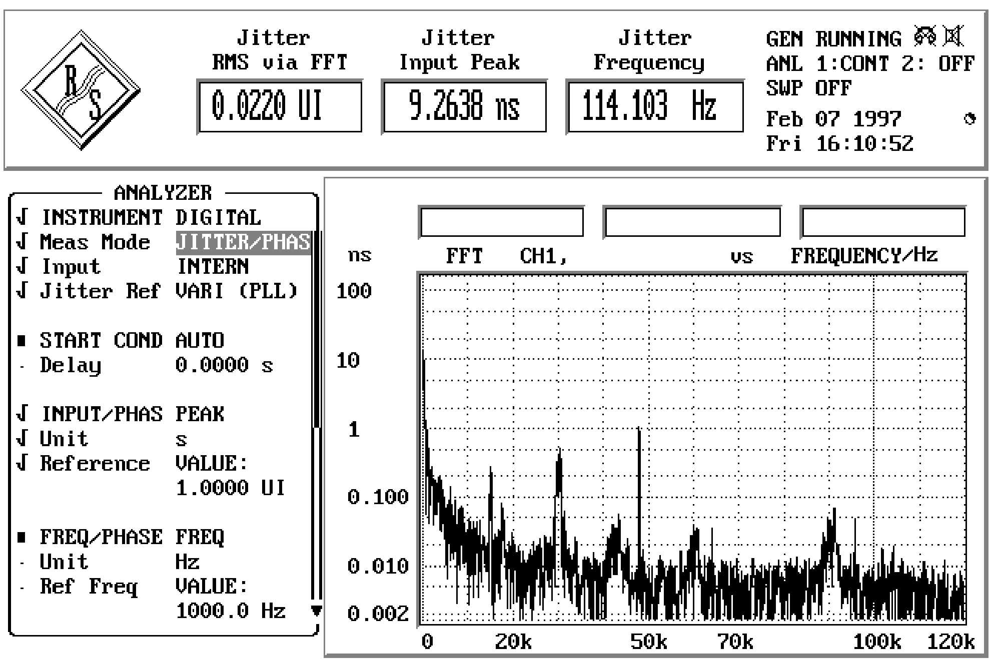


Fig. 16: Jitter spectrum with clear frequency components

9.4. Jitter Waveform

Setup: **JITWA_DD.SAC**

With this setup the generator produces a noise-like jitter signal, the waveform function of the analyzer shows this signal in the time domain.

9.5. Jitter Susceptibility

At present, this measurement function is only available in Audio Analyzer UPL.

Setups: **JITSU_DA.SAC** **JITSU_DD.SAC**

Definitions and test conditions:

The measurement of jitter susceptibility is at present being incorporated into AES17. It describes the effects of jitter at the digital audio or reference input on the quality of the audio contents. The digital input receives a sinewave signal of 1/4 of the sampling frequency at a level of -3dBFS, this signal being at the same time subject to jitter. The jitter frequency is swept from 80Hz to 20 kHz at a jitter level of 40 ns. The audio output signal is examined by graphically displaying the THD+N value versus the jitter frequency.

Notes on measurements:

The measurement of jitter susceptibility was not possible with Audio Analyzers UPD and UPL up to now. UPL with firmware version 1.01 and Option UPL-B1 (Low Distortion Generator) can now perform this measurement which involves the simultaneous generation of an audio signal and of jitter. UPL-B1 provides the second generator required for this purpose.

9.6. Phase between Audio Data Signal and Reference Signal

Setup: **DPHA_DD.SAC**

Definitions and test conditions:

If a digital audio equipment is synchronized using an external clock, the audio frames and the reference clock must be within a specific time range. As regards the phase shift between synchronization input and audio output of digital components, AES3 specifies a limit of 1/4 of the frame length (=32UI).

Notes on measurements:

Audio Analyzers UPD and UPL feature comprehensive synchronization facilities. In the generator section the phase shift between audio frames and reference clock can be selected between -64UI and +64 UI to determine the synchronization range of the DUT.

The *JITTER/PHAS* menu in the *Meas Mode* line of the analyzer allows the phase between reference input and audio input of the analyzer to be measured, which corresponds to the phase shift between synchronization input and audio output of the DUT assuming appropriate cabling.

Annex: Overview of Setups Used

Linear Distortion Measurements

Amplitude frequency response

Sweep measurement using signal from internal generator

LEVS_AA.SAC **LEVS_AD.SAC** **LEVS_DA.SAC** **LEVS_DD.SAC**	Logarithmic frequency sweep from 20 Hz to 20 kHz at constant level, graphic display of amplitude frequency response

Sweep measurement using signal from external source

LEVSE_AA.SAC **LEVSE_DD.SAC**	External frequency sweep from 20 Hz to 20 kHz, measurements made on 5% frequency change, graphic display of amplitude frequency response

Fast frequency response measurement using FFT

FFLEV_AA.SAC **FFLEV_DD.SAC**	Display of frequency response using 2k FFT

Frequency response measurements at different levels

MLEVS_AA.SAC **MLEVS_DD.SAC**	Logarithmic frequency sweep from 20 Hz to 20 kHz with additional level sweep from 1 V to 0.1 V, graphic display of amplitude frequency response curves

Level difference between two stereo channels

LEVDS_AA.SAC **LEVDS_AD.SAC** **LEVDS_DA.SAC** **LEVDS_DD.SAC**	Logarithmic frequency sweep from 20 Hz to 20 kHz, graphic display of level difference between the two stereo channels referred to channel 1

Phase and group-delay measurements

Measurement of phase frequency response

PHAS_AA.SAC	Logarithmic frequency sweep from 20 Hz to 20 kHz at constant level, graphic display of phase frequency response

Measurement of phase difference between two stereo channels

PHADS_AA.SAC **PHADS_DD.SAC**	Logarithmic frequency sweep from 20 Hz to 20 kHz, graphic display of phase difference between the two stereo channels referred to channel 1

Measurement of group delay versus frequency

GRPS_AA.SAC	Linear frequency sweep from 20 Hz to 20 kHz at constant level, graphic display of group delay versus frequency

Combined measurements	
Amplitude and phase frequency response in one display	
PHLVS_AA.SAC	Combined display of amplitude and phase frequency response for channel 1; parameters as indicated above
Phase difference and level difference between two stereo channels in one display	
PDLDS_AA.SAC **PDLDS_DD.SAC**	Combined display of phase and level difference between the two stereo channels; referred to channel 1
Group delay and amplitude frequency response in one display	
GRLVS_AA.SAC	Combined display of group delay and amplitude frequency response for channel 1

Nonlinear Distortion Measurements

Total harmonic distortion (THD)	
THD_AA.SAC **THD_AD.SAC** **THD_DA.SAC** **THD_DD.SAC**	Measurement of THD; simultaneous display of spectrum up to 9th harmonic
THD+N	
THDN_AA.SAC **THDN_AD.SAC** **THDN_DA.SAC** **THDN_DD.SAC**	Measurement of THD+N value; simultaneous display of spectrum with harmonics marked
THDNS_AA.SAC **THDNS_DD.SAC**	Linear frequency sweep from 20 Hz to 20 kHz, graphic display of THD+N value versus frequency
Intermodulation	
MOD_AA.SAC **MOD_DD.SAC**	Measurement of intermodulation; spectral display of intermodulation products
Difference frequency distortion (DFD)	
DFD_AA.SAC **DFD_DD.SAC**	Measurement of difference frequency distortion; spectral display of 2nd-order DFD
Dynamic intermodulation (DIM) - UPD only	
DIM_AA.SAC **DIM_DD.SAC**	Measurement of dynamic intermodulation distortion to IEC 268-3; spectral display of intermodulation products
FFT analysis	
FFT_AA.SAC **FFT_AD.SAC** **FFT_DA.SAC** **FFT_DD.SAC**	Spectral display by means of FFT analysis; generator supplies 1-kHz / 997-Hz test signals

Measurement of Interference and Wow & Flutter

S/N ratio

SNRA_AA.SAC **SNRA_AD.SAC** **SNRA_DA.SAC** **SNRA_DD.SAC**	Display of S/N ratio weighted with A filter; measurement by means of rms detector
SNRC_AA.SAC **SNRC_AD.SAC** **SNRC_DA.SAC** **SNRC_DD.SAC**	Display of S/N ratio weighted with CCIR filter; measurement by means of quasi-peak detector

Crosstalk

CRSS_AA.SAC **CRSS_AD.SAC** **CRSS_DA.SAC** **CRSS_DD.SAC**	Graphic display of crosstalk from channel 2 to channel 1; frequency sweep from 20 Hz to 20 kHz

Stereo separation

SEPS_AA.SAC **SEPS_AD.SAC** **SEPS_DA.SAC** **SEPS_DD.SAC**	Graphic display of stereo separation; channel 1 used as reference; frequency sweep 20 Hz to 20 kHz;

Wow & flutter

WFI_AA.SAC **WFN_AA.SAC** **WFJ_AA.SAC**	Measurement of wow & flutter; simultaneous generation of required test signal; three setups for standards DIN/IEC, NAB and JIS

Measurements on Analog/Digital Interfaces

Clipping level

CLIP_AD.SAC	Setup for determining clipping level of A/D converters

Linearity of A/D converters

LINS_AD.SAC	Graphic display of converter linearity, determined with level sweep 0 dBr to -120 dBr

Nonlinearity of A/D converters

LINDS_AD.SAC	Same as linearity of A/D converters, but with display of deviation from ideal characteristic

Linearity of D/A converters

LINS_DA.SAC	Graphic display of converter linearity, determined with level sweep 0 dBr to -120 dBr

Nonlinearity of D/A converters

LINDS_DA.SAC	Same as linearity of D/A converters, but with display of deviation from ideal characteristic

Signal delay in analog/digital systems - UPL only

DEL_AA.SAC **DEL_AD.SAC** **DEL_DA.SAC** **DEL_DD.SAC**	Measurement of signal delay in analog and digital systems using the *Waveform* function; determination of delay and polarity of the two stereo channels

Protocol Analysis

Binary data protocol

PROTB_DD.SAC	Generation and analysis of digital auxiliary data; display in the form of binary numerals

Channel status data in professional format (AES 3)

PROTP_DD.SAC	Generation and analysis of digital auxiliary data; display evaluated to AES3

Channel status data in consumer format (IEC 958)

PROTC_DD.SAC	Generation and analysis of digital auxiliary data; display evaluated to IEC 958

Digital Interface Tests

Common-mode interference, pulse amplitude and sampling frequency on digital line

DCOM_DD.SAC	Measurement of digital pulse amplitude and sampling frequency as well as analysis of common-mode interference on balanced digital lines; thus also generation of common-mode signals

Jitter amplitude

JITAM_DD.SAC	Display of jitter amplitude and generation of jitter

Jitter spectrum

JITSP_DD.SAC	Display of jitter spectrum and generation of jitter

Jitter waveform

JITWA_DD.SAC	Display of jitter waveform and generation of jitter

Jitter susceptibility - at present UPL only

JITSU_DA.SAC **JITSU_DD.SAC**	Display of jitter susceptibility and generation of jitter

Phase between audio data signal and reference signal

DPHA_DD.SAC	Setup to determine phase shift between audio signal and reference clock in case of external synchronization of DUT

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