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# **Measurements on A/D Converters with Audio Analyzer UPL or UPD**

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Application Note 1GA30-0E

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Subject to change

Products:

**Audio Analyzer UPL**

**Audio Analyzer UPD**



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## **1. Summary**

Present-day sound studios are hardly conceivable without A/D converters. A/D converters are essential components when music is recorded and they directly influence the recording quality. The current Application Note describes a test program permitting automatic measurements of all parameters of sound studio equipment with the aid of Audio Analyzers UPD or UPL. Measuring converter chips with mostly serial interfaces that are not to standard are not the subject of this Application Note.

## **2. Introduction**

Since an increasing amount of digital equipment is used in sound studios, A/D converters have become essential components. When connected directly to the recording microphone, they considerably influence the recording quality. All renowned suppliers of professional studio equipment offer A/D converters. The trend is towards always higher resolutions and 24-bit converters are now available on the market. The greater the number of bits used for the conversion, the wider the available dynamic range. However, an increasing number of bits also places higher demands on the measuring equipment. Extremely pure test signals have to be generated and analyzed with an appropriate solution at the converter output. Audio Analyzers UPD and UPL fully comply with these requirements.

For a comprehensive characterization of A/D converters, a number of measurements has to be performed, which may of course be triggered manually one after the other. However, in many cases it may be much more convenient to run program-controlled test sequences. Such test sequences can be performed with the aid of Automatic Sequence Controllers UPD-K1 and UPL-B10 which are available as options. This capability is made use of in this Application Note.

A BASIC program is part of the present application. The measurements on A/D converters are performed automatically, results can be printed or stored for further processing.

## **3. Preparation and Start of Application Software**

### **3.1. Instruments and Auxiliaries Required**

Audio Analyzer UPD or UPL with a Low-Distortion Generator UPD-B1 or UPL-B1 and Digital Audio I/O option UPD-B2 or UPL-B2 is required for these measurements. Option -B2 comprises the standardized digital interfaces AES/EBU and S/PDIF as well as an optical interface. Option -B1 is an extremely low-noise sinewave generator which guarantees the required signal quality. An external keyboard is needed to make entries for generating reports. With the UPD, a keyboard is also required for running the test program; with Audio Analyzer UPL (as of firmware version 1.01) test sequences may also be called up without the use of a keyboard.

The BASIC program required for performing automatic measurement sequences has been stored on a floppy disk and is available from your local Rohde & Schwarz representative. The audio analyzer has to meet the following firmware requirements:

- UPD firmware version 3.04 or higher,
- UPL firmware version 1.21 or higher,
- Automatic Sequence Controller UPD-K1 / UPL-B10 installed,
- UPD/UPL configured for automatic sequence control with a 64 Kbyte program memory and a 64 Kbyte data memory (select setting 5 with configuration tool UPDSET or UPLSET).

### **3.2. Software Installation**

The application software is installed with the aid of the installation program ADCINST.BAT, which is also on the supplied floppy:

- Quit the measurement software by pressing the SYSTEM key on the instrument or Ctrl F9 on the keyboard.
- Insert accompanying disk.
- Select floppy-disk drive (enter A:)
- Call up the installation program (enter ADCINST)
- Return to UPD/UPL program (enter C:\UPD or C:\UPL)

The ADCINST program generates the C:\ADC directory in the audio analyzer (if it is not yet available) and copies the BASIC program and the setups required for the application into this directory.

### **3.3. Test Setup**

A printer may be required in addition to Audio Analyzer UPD or UPL for logging results. Connect the A/D converter input for the left channel to generator output 1 of UPD/UPL, that for the right channel to output 2. The converter output is connected to the digital input of UPD/UPL.

### **3.4. Starting the Application Software**

The application program is performed with the aid of the automatic sequence controller. By pressing the F2 key (external keyboard) the analyzer is switched to the automatic control function (with UPL switchover is also possible by means of the BACKSPACE key).

It is important that the logging function is switched off. In this case "logging off" is displayed at the bottom right of the screen. With the logging function switched on, all manually entered commands would be added to the program and use up valuable memory capacity. The log mode can be switched on and off with the F2 key of the external keyboard.

Since all program routines and setups are searched for in path C:\ADC, the application programs will have to be called up in this path. The path can be changed in one of the following ways:

- in the manual mode with command "Working Dir" in the FILE panel
- by calling up a setup required for the measurements on A/D converters
- from the automatic sequence controller with BASIC command line  
UPD OUT "MMEM:CDIR \"ADC\"", or  
UPL OUT "MMEM:CDIR \"ADC"
- under BASIC by entering CD ADC and then EXIT via the SHELL command
- at the DOS level by entering CD ADC.

The program floppy comprises the BASIC program ADCTEST.BAS for measurements on A/D converters. It is loaded and started by entering:

- LOAD"ADCTEST"
- RUN

This can also be done with the respective softkeys displayed at the lower screen edge when the automatic sequence controller is called up.

### 3.5. Configuring the Application

The measurement is performed with a number of setups which are also stored on the application floppy. All setups have the following designation:

AD\_XXXX.SAC

The measurement function can be identified from the abbreviation under XXXX, ie FREQ stands for frequency response, THDN for THD+noise measurement, CROS for crosstalk, etc. Only "actual setups" are used. Contrary to "complete setups", actual setups store only the analyzer and generator settings required for the application and the settings of the DISPLAY and FILE panels. The advantage of actual setups is that they can be called up more quickly than complete setups which include all the settings of the measuring instrument.

Upon delivery of the setups, "Default printer" is set in the OPTIONS panel. This means that the printer configuration is not determined by the setup but the configuration used by the audio analyzer for the last printout is maintained. New settings by the user are not required. The UPD/UPL screen is set for colour representation and an external monitor can also be used. The settings may of course be adapted to the user's requirements. They can be changed in the OPTIONS panel.

Since this Application Note is mainly intended for users of professional studio equipment, the balanced analog converter inputs and the balanced digital AES/EBU input will normally be required. Other interfaces may also be used, of course. Analog levels are indicated in dBu, measurement results for THD+N displayed in dB. All measurements are carried out on two channels.

If other settings are to be changed as well, the setups have to be called up and stored again under the same name after the modification. The following additional changes can be made:

- Entry of a comment in the graphics display (COMMENT in the DISPLAY panel)
- Definition of tolerance curves/values (LIMIT CHECK in the DISPLAY panel)
- Selection of settings to be displayed in the STATUS panel (ticked items)

**IMPORTANT:** If other settings except those listed here are changed, error-free operation of the software cannot be guaranteed.

### 3.6. Setup Conversion during Firmware Update

When the UPD/UPL firmware is updated, the setups may also have to be converted. The conversion is performed automatically when a setup is loaded, but it may cause short delays during loading. This can be avoided by converting and restoring the setups before the Application Software is started. Conversion can be performed in two ways:

- by calling up the UPD/UPL converter program in DOS:  
DO\_CONV \UPD\ADC or DO\_CONV \UPL\ADC.  
In this case all setups in the ADC directory are converted.
- by loading and storing each setup in the measurement mode.

## 4. Operating Concept

Softkeys permitting the user to intervene with the program are displayed at the lower screen edge. Since the softkey functions are also assigned to the external keyboard, the individual program routines can also be selected from the keyboard.

After program start the opening mask "Measurement of A/D converters with Audio Analyzers UPD or UPL" and softkeys F5: CONFIG, F6: DUT, F11: RESULTS and F12: RUN are displayed.

### 4.1. Softkey F5: CONFIG

Prior to starting the first measurement, a few entries have to be made for the test sequence. Press softkey F5: CONFIG to obtain the following menu:

Configuration for A/D Measurements	
Digital Interface:	<i>balanced</i>
Sample Rate:	<i>48 kHz</i>
Reference Level in dBu:	<i>21.2</i>
Display as Linearity Error	<i>Yes</i>
Autorun	<i>Yes</i>

Fig. 1: Entry menu CONFIG

"Digital Interface" informs on the interface to which the converter is connected. A choice can be made between balanced, unbalanced or optical (Toslink) interfaces by means of the page-up and page-down keys.

One of the three standard sample rates 32kHz, 44.1 kHz or 48 kHz can be selected. Selection is made by means of the page-up and page-down keys.

Under "Reference Level" the analog input level for fully driving the converter is entered in dBu. The entry is made using the numeric keys on the instrument or the external keyboard and has to be terminated with ENTER. This entry is for orientation only, the clipping level is defined more accurately under CLIPP before the actual measurement is started (see section 5.3, Determining the Clipping Level). The value determined under CLIPP is used as a reference for all measurements.

The converter linearity can also be displayed as a deviation from the ideal curve. The respective item in the CONFIG menu is to be answered with Yes or No.

Under "Autorun" an automatic test sequence can be selected (enter Yes) or manually triggered measurements (enter No). The Yes/No decision is also made with the page-up/page-down keys.

After the entries are completed, softkeys F6:OK and F7: CHANGE are displayed. With CHANGE the CONFIG menu can be opened again to modify the settings.

#### 4.2. Softkey F6: DUT

With softkey F6: DUT, additional information on the DUT may be entered, which is then printed in the test report, ie type and serial number of the A/D converter to be tested and the name of the person carrying out the test. Entries are made on the external keyboard. Date and time are set automatically by the system clock of the measuring instrument. Fig. 2 shows the entry menu:

Device under Test	<i>Type 12345</i>
Manufacturer	<i>Audio Company</i>
Serial No.:	<i>999999</i>
Tested by	<i>Mike Measureman</i>

Fig. 2: Entry menu DUT

Upon completion of the entry, the main menu can be called up again with softkey F6:OK or new entries can be made by pressing softkey F7: CHANGE.

Press softkey F12: RUN when all entries have been completed or if the settings used for the last measurement should be used. If an automatic measurement sequence has been selected in the CONFIG menu, the measurement sequence is started.

With single-shot measurements selected, softkey F12: RUN triggers the measurement which determines the clipping level (see section 5.3).

After this the following softkeys

F5: LINEAR,                      F6: FRQ-RESP,                      F7: ALIASING,                      F8: NOISE,  
 F9: SPECTRUM,                      F10: THDN-FRQ,                      F11: THDN-LEV and                      F12: CRSSTLK

are displayed for calling up the required measurement.

### 4.3. Softkey F11: RESULTS

With softkey F11: RESULTS, stored measurement results can be called up. All stored result files are listed in the current directory and the desired file can be entered. Graphs and result lists are displayed on the screen successively. Switchover between the displays is possible by means of the page-up/page-down keys.

## 5. Measurements

### 5.1. Standard Specifications

Measurements on components using analog/digital interfaces are performed in line with the AES17 standard, "Measurement of Digital Audio Equipment". As far as possible, the measurements described in this Application Note are performed in compliance with this standard.

### 5.2. Notes on Measurements

The individual measurements are described in the order in which they are performed in the automatic measurement sequence.

To start a measurement, it is called up with the associated softkey or the corresponding function key on the external keyboard. With the measurement completed, the results remain on the display and softkey F6: CONTINUE is displayed so that the selection softkeys can be called again. A return to the measurement menu for calling up the next measurement is also possible by means of the SPACE key on the external keyboard or with the SELECT key of the audio analyzer.

### 5.3. Determining the Clipping Level

Audio components performing digital signal processing must not be overdriven, as exceeding the digital value range (clipping level) leads to considerable signal distortion. The drive limit of digital modules is therefore far more important than that of analog components. For this reason particular attention has been attached to this point in the AES 17.

The clipping level must also be determined for A/D converters. Since the digital outputs are accessible, the clipping level is determined by applying a 997 Hz signal and increasing the input level until the peak value of the digital output signal has reached that of the longest data word (full scale).

The level thus obtained defines the full-scale drive of the digital system and is used as a reference in all further measurements.

Each time the measurement program is started - irrespective of whether a test sequence or as single measurement is to be performed - the program branches to a routine which automatically determines the clipping level. Starting at a level 10 dB below that of the level entered in the CONFIG menu, the input signal for the two channels of the A/D converter is increased step-by-step and the digital output level at each step measured until the clipping level is reached. The lower one of the clipping levels in the two channels is used as a reference.

The two clipping levels and the reference level are shown as in Fig. 3.

Differences between the clipping levels of the two input channels indicate that analog components of the converter are driven with a different level.



Measurement of Clipping Level	
Clipping Level Left	10.85 dBu
Clipping Level Right	10.42 dBu
Reference Level for Measurements	10.42 dBu

Fig. 3: Determining the clipping level

#### 5.4. Converter Linearity/Linearity Deviations

To measure the converter linearity, apply a 997 Hz sinewave signal to the inputs. Starting at the full-scale level, the signal level is reduced to -130 dB in 5 dB steps. The output signal is measured and graphically displayed versus the input signal. Since the signal is masked by the noise as the level gets smaller, a narrowband third-octave bandpass filter is used for the measurement.

With a linear converter characteristic, a diagonal line is displayed as shown in Fig. 4.

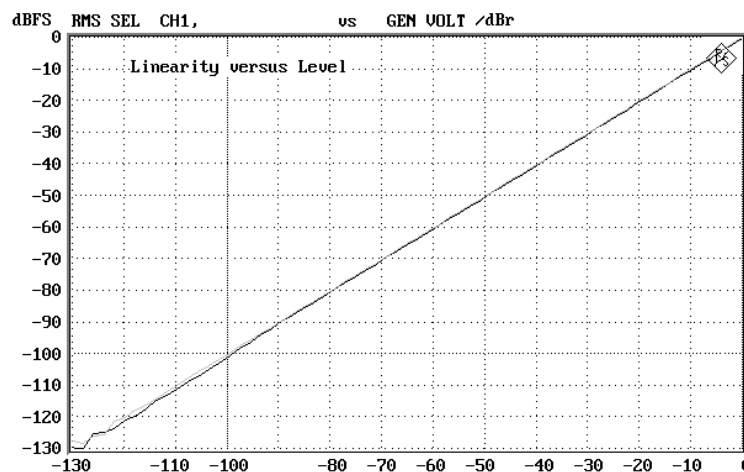


Fig. 4 Measuring the converter linearity

The linearity measurement is not described in the AES17 standard but is commonly used for determining the converter quality. However, with this kind of representation, deviations from the ideal characteristic are difficult to see. A better way is to measure the level linearity error. This measurement is described by AES 17 as "level-dependent logarithmic gain". The measurement is performed in the same way as the linearity measurement but the logarithmic gain is determined at each measurement step, ie the ratio of output and input amplitude. The result is graphically displayed versus the input level.

The diagram therefore shows the deviation of the converter characteristic from the nominal linearity curve. Contrary to the measurement performed under this application, AES 17 specifies -5dBFS as the first test point to be recorded. The linearity error is also measured selectively with the aid of a third-octave bandpass filter.

For the display of the linearity error, the described application program does not start a new measurement; the characteristic is derived from results previously obtained in the linearity measurement. In this case "Display as Linearity Error" in the CONFIG menu must be answered with Yes. The linearity measurement is started with softkey F6:LINEAR. After this measurement, display of the linearity error can be called up with the SPACE key of the external keyboard, the SELECT key on the audio analyzer or else with softkey F6: CONTINUE.

Fig. 5 gives an example. In the ideal case a straight line is obtained, the deviation from the ideal converter characteristic can be read in dB.

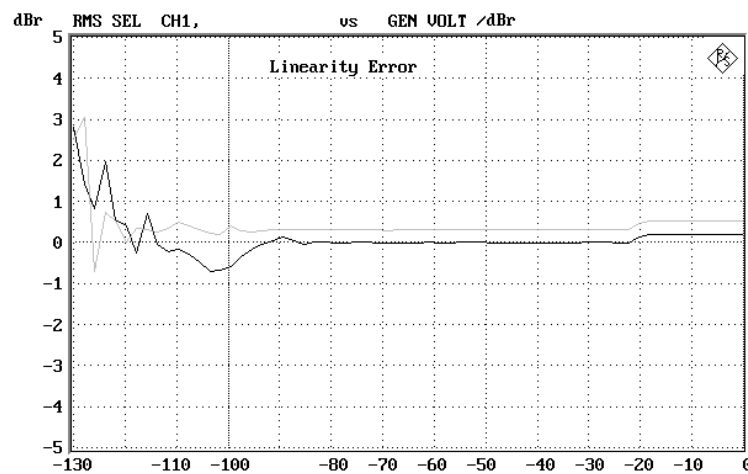


Fig. 5: Measuring the linearity error

A/D converters in state-of-the-art studio equipment normally use cascaded converter circuits with two overlapping converter ICs in the level range, ie as the input level goes down, a switchover is made to the second IC. If the gain range between the two converter ICs is not accurately adjusted, this switchover is visible as a step in the linearity error diagram. This type of switchover is also used by the converter measured in Fig. 5. The switchover to the second IC is made at approx. 20 dB below the full-scale level.

### 5.5. Frequency Response

Measuring the amplitude frequency response is a standard measurement in audio technology. In most cases the term "frequency response" is simply used as the amplitude frequency response is measured much more frequently than the phase frequency response. According to AES 17, the frequency response of digital components is measured at 20 dB below full level. The input signal is swept over the frequency range at constant level, the output is displayed in dB versus frequency referred to the level at 997 Hz.

In the present application program the measurement is performed between 20 Hz and 20 kHz (softkey F7: FRQ-RESP), the sweep is logarithmically switched in steps of 100. Fig. 6 gives an example. Here too, the measurement is performed on two channels. As the same reference value is used for both curves, the different drive levels of the two stereo channels can be seen. Since the frequency response is measured 20 dB below full level, the value determined as the clipping level less 20 dB is displayed.

State-of-the-art A/D converters do not generally show frequency response irregularities, but this measurement is practically always performed.

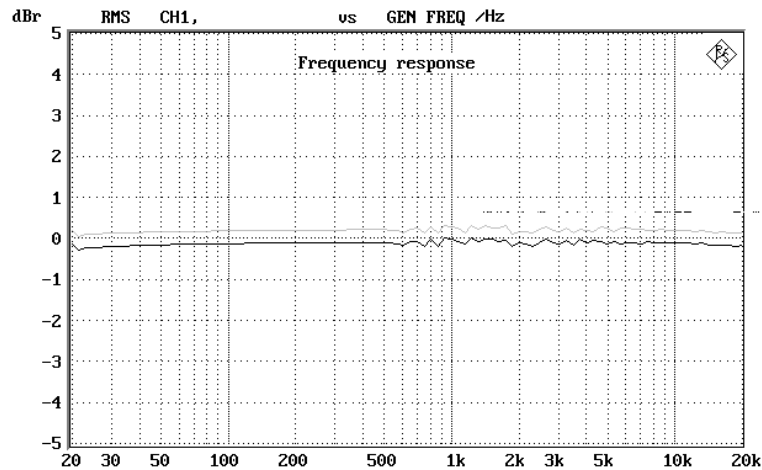


Fig. 6: Frequency response of A/D converter

## 5.6. Attenuation in the Stopband

With this measurement the converter's capability to suppress signal components above the aliasing frequency is determined. As specified by AES17, first a 997 Hz sinewave signal is applied with full-scale level to the DUT to determine the reference value for the output voltage. Then the input signal is swept between 20 kHz or the upper limit frequency of the DUT (depending on which one is lower) and four times the sample frequency but max. 192 kHz. The remaining signal components are measured as true rms value and displayed in dB relative to the previously determined reference value versus the input frequency.

The measurement is started with softkey F8:ALIASING. The Audio Analyzer sweeps the test signal between 10 kHz and 110 kHz logarithmically in 100 steps, the measured values of the output signal are graphically displayed as shown in Fig. 7. Determining the reference value at 997Hz (as described in AES 17) is not necessary in this case, as this reference has already been determined for measuring the clipping level.

A sufficient attenuation of high-frequency input signals is important because these signals may produce intermodulation products with the sample frequency, which then appear in the audible range. Such aliasing products cannot be separated from the original audio signal and thus impair the quality of the music played. The attenuation of the A/D converter in the stopband range should be at least 60 dB and not deteriorate towards higher frequencies.

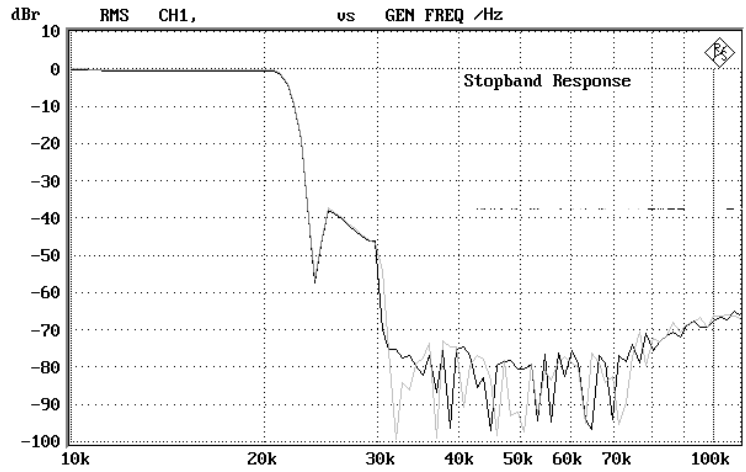


Fig. 7: Stopband attenuation of antialiasing filter at converter input

### 5.7. Idle-Channel Noise

AES 17 prescribes the following idle-channel noise measurement. The converter is operated without termination. The remaining output signal is measured with a standard AES 17 weighting filter connected ahead. The level is specified in dBFS.

The frequency response of the above-mentioned standard weighting filter corresponds to that of a weighting filter to CCIR 468-3 however with the gain set so that the 0 dB point is at 2 kHz and not at 1 kHz. Furthermore, the weighting filter is used together with an rms detector and not with the quasi-peak detector specified by CCIR. This kind of weighting filter is normally called CCIR-ARM filter or CCIR-2k filter.

Noise measurement		
RMS A-weighted	Left <i>-114.1 dBFS</i>	Right <i>-115.5 dBFS</i>
RMS unweighted	Left <i>-110.4 dBFS</i>	Right <i>-110.3 dBFS</i>
RMS CCIR-2k-weighted	Left <i>-110.9 dBFS</i>	Right <i>-111.3 dBFS</i>
Quasi-peak CCIR weighted	Left <i>-100.4 dBFS</i>	Right <i>-101.0 dBFS</i>
Quasi-peak unweighted	Left <i>-106.7 dBFS</i>	Right <i>-107.3 dBFS</i>
Continue with < SPACE >		

Fig. 8: Idle-channel noise - results obtained with different test methods

The definition of a weighting filter which is in conformity with the perception of the human ear has still not been finalized. The CCIR-2k filter from Dolby was proposed as a possible solution. BBC have come up with results indicating that rms detectors approach the perception of the human ear better than quasi-peak detectors and that the CCIR curve is not the closest approximation. As a compromise AES used a CCIR-2k filter with rms weighting but this decision has not lead to a greater uniformity either so that a large variety of weighting methods is still being employed. To allow the user to determine the result that suits him best, the present application program performs the idle-channel measurement with all commonly used filters and detectors. The example in Fig. 8 shows the greatly varying results in dB obtained with different test methods.

After starting the noise measurement with softkey F9:NOISE, the idle-channel noise spectrum is first displayed. The above results are displayed after a switchover with theSPACE key of the external keyboard, the SELECT key of the audio analyzer or softkey F6:CONTINUE. It should be remembered, however, that the noise measurements will take up some time as, according to CCIR, the quasi-peak detector alone requires a settling time of 3 s for each measurement.

## 5.8. Output Spectrum at Different Input Levels

For this test a 997 Hz sinewave signal is applied to the converter with the levels -0.5dB, -60 dB and -100 dB. The output spectrum obtained in each case is displayed; switchover is performed with the SPACE key of the external keyboard, the SELECT key of the audio analyzer or with softkey F6: CONTINUE.

AES 17 does not describe the display of the output spectra. However, since this display also shows any distortions or other nonlinearities, it supplements the measurement of distortion versus level.

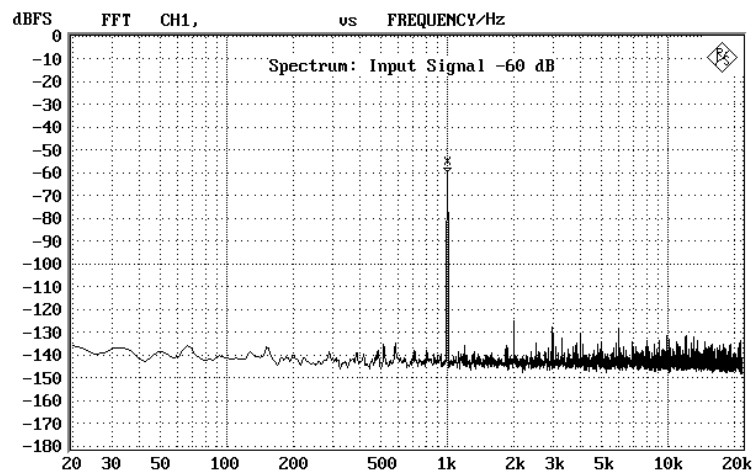


Fig. 9: Output spectrum at -60 dB signal level

State-of-the-art A/D converters have excellent S/N ratio and distortion characteristics. For this reason the generation of the test signal is of particular importance. Converter output spectra always show the spurious generated by the converter but also the distortion and noise components caused by the test signal. For this reason the application program uses the low-distortion generator of Audio Analyzers UPL or UPD for all critical measurements. This analog generator produces extremely pure sinewave signals with noise components that are clearly below those of modern converters.

## 5.9. THD versus Frequency

As specified by AES 17, a -1 dBFS sinewave signal is applied to the DUT and the THD+N value is measured. The measurement should be repeated at -20dBFS.

The test signal in the output signal is suppressed for the measurement, the bandwidth of the residual signal should be band-limited to the upper frequency limit or to 20kHz. The level of this filtered signal is measured and specified as a ratio of the unfiltered signal level. The test signal is swept between 20Hz and half of the upper frequency limit.

The present application program performs the described measurement above half the upper frequency up to 20 kHz. This ensures that also those distortions are detected which mainly show up at higher frequencies because of image frequencies.

Due to band-limiting of the measured signal, less distortion products are measured as the test frequency increases. For instance, at a signal frequency of 8 kHz, only the 2nd harmonic at 16 kHz is considered in the measurement, all others are outside the 20 kHz measurement bandwidth. The THD+N value therefore goes down as the signal frequency decreases; above 10kHz no harmonics are measured. If this drop does not show in the curve, the noise and/or other interfering products are greater than the distortion itself. A rise of the curve above 10 kHz is normally caused by mixture products of the signal harmonics and clock frequency, and this is caused by insufficient antialiasing filtering at the converter input.

## 5.10. THD versus Level

AES 17 describes the measurement of the THD+N value versus the input level. Basically the measurement is performed in the same way as the one described above. A 997 Hz sinewave signal is applied, the test signal is to be swept from 0dBFS to -80 dBFS.

Since the filtered output signal is always referenced to the unfiltered signal, THD+N levels increase as the signal level decreases. The noise is not reduced with the level so that a poorer S/N ratio is obtained.

The present application program does not display the THD+N value relative to the unfiltered signal but the THD+N level. Thus variations of the mixture products can be more easily detected. The test range is also extended; the program performs a sweep between 0dBFS and -100 dBFS.

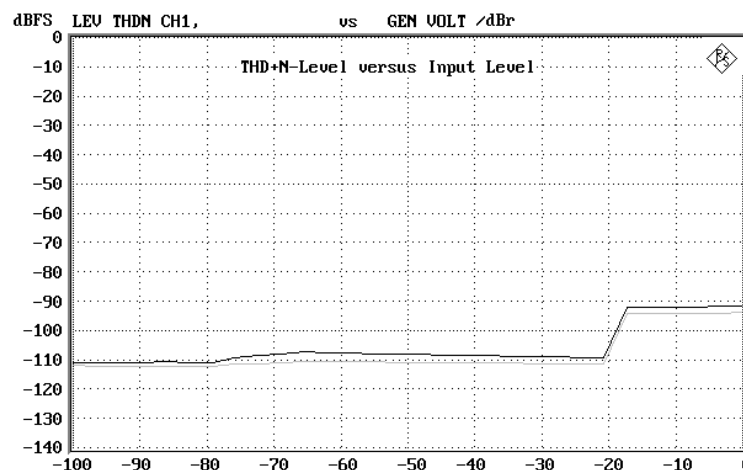


Fig. 10: THD+N level versus input level

To be able to cover the whole dynamic range required for measurements on studio equipment, cascaded converter circuits are frequently used. In this case two overlapping converter ICs are employed. With the aid of this circuit known as "gain ranging", the signal at lower levels is switched to a converter stage of higher gain. In the above example this switch point can be clearly seen by a sudden rise of the THD level. Since in the case of music signals this switch point is frequently crossed as the level varies and may become perceptible, the switching threshold is smoothed in more up-to-date circuits.

## 5.11. Crosstalk

For measuring the crosstalk between the two stereo channels, a level of -20 dBFS is applied to one channel only. The output level of this channel is then measured. The output level of the unused channel is measured by means of a third-octave bandpass filter with reference to the channel used. AES 17 prescribes this measurement from 10Hz to the upper frequency limit of the converter.

The present application program performs this measurement in line with AES17. The measurement is carried out in both directions, ie the crosstalk from channel 1 to channel 2 and from channel 2 to channel 1 is measured. As the circuit is normally not symmetrical, the measurements may yield different results. An example is given in Fig. 11.

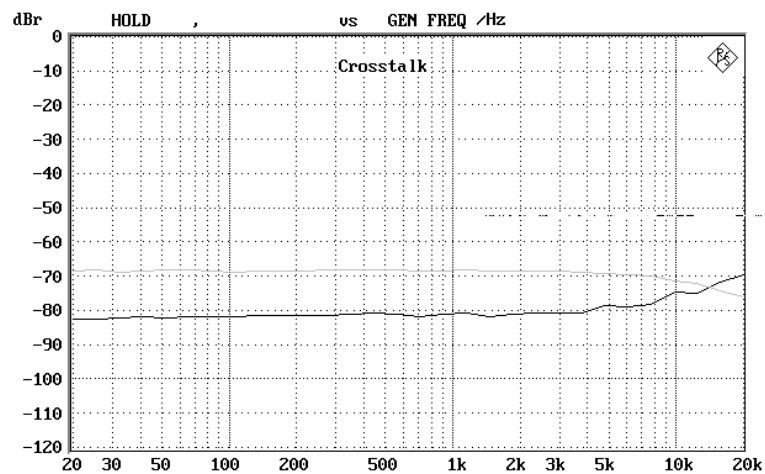


Fig. 11: Crosstalk measured in both directions

## 6. Processing of Measurement Results

### 6.1. Printout, Storage and Display of Results

Measurement results can be processed in many ways. When the automatic test sequence is completed, a selection menu is displayed permitting the results to be printed or stored. Individual results may also be displayed again one after the other. Switchover between the displays is performed with the page-up/page-down keys.

After single-shot measurements, branching into the selection menu is also possible using softkey F7: RESULTS.

### 6.2. Switchover between Automatic Sequence Control / Manual Control

At the end of a single-shot measurement, softkey F5:GOTO UPL is displayed. It permits switchover to UPD/UPL manual control without the need to terminate the BASIC program. This switchover may be used, for instance,

- to change the graphics scale,
- to shift the graphics cursor or to switch it on/off ,
- to add a comment to the graphics display,
- to print a display window with a modified STATUS panel,
- for a tabular display of measurement results or limit violations.

Return to the application program is by means of the F3 key on the external keyboard (for the UPL also by means of the BACKSPACE key). The BASIC program is accessed at the same location at which it was quit.

This possibility of interrupting the test run is not offered in automatic A/D converter testing.

## 7. Terminating the Application

Pressing the ESC key on the external keyboard or the CANCEL key on the UPD/UPL causes a return to the next higher menu level. At the highest menu level, the keys terminate the program, ie by pressing the keys repeatedly the program can be terminated from any menu level.

To prevent the program being terminated inadvertently, a query is displayed for confirmation.

The software can be aborted any time under BASIC using the key combination CTRL BREAK. The CONT key causes the program to be continued, RUN triggers a restart.



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