NEUTRIK TEST INSTRUMENTS

*Application Note*

# A2-D


## *AES/EBU Digital Audio Signal Analysis*

*V 3.01 / Sept. 2000 / PhS, MBe*

# ABSTRACT

**Only few years ago, in broadcasting all transmitted and processed information were strictly analog, either as microphone-, line level- or speaker-signals, or as signals in 100V technology. But the last years have brought a major change into the field of audio communication: the digital signal transmission and processing, which has been standardized by the AES.**

**However, the transition to exclusively digital signal transmission and processing in broadcast- and sound studio applications will take some years, during which both systems will be present in parallel, requiring interfaces to convert signals from analog to digital mode and vice versa.**

**This paper presents an easy-to-understand introduction into A/D signal conversion and explains the digital standards and their terminology, as well as some digital measurement capabilities of the A2-D.**

# DIGITAL SIGNALS AND A/D CONVERTERS

In order to represent an analog signal in a digital format or to process an analog signal in the digital domain, it is necessary to convert it into a digital signal by means of an analog to digital (A/D) converter which assigns a binary number proportional to the incoming analog signal (conversions other than binary are possible but of no practical significance).

Obviously, the accuracy of the digital representation depends on how many bits are used to represent the digital number. Several factors influence the choice for this number of bits such as conversion speed, the demands of the application, memory consumption and others. The number of bits for the conversion is called resolution. The number of steps that can be represented with a certain resolution can easily be calculated by putting the resolution to the power of two. With a currently common converter for professional applications providing 16-bit resolution, the number of steps is $2^{16}$=65536 bits.

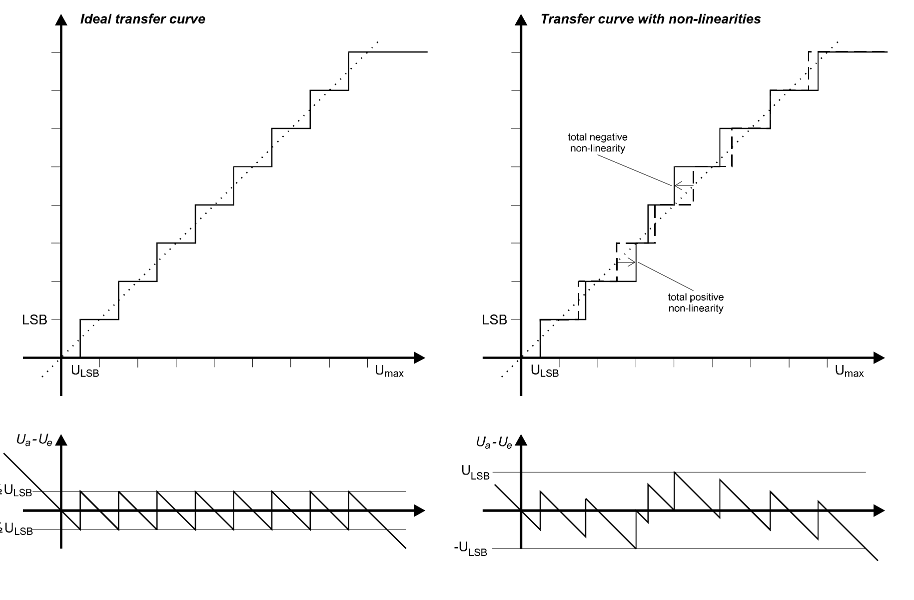



*Fig 1 Transfer Curves of an A/D Converter*

## *Linearity*

The A/D transfer curve on the far left represents the theoretical ideal. The resulting maximum conversion error is shown below as a deviation from the straight-line transfer curve. The absolute error never exceeds ½ LSB.

The adjacent diagram illustrates a real, non-ideal transfer curve. The max. absolute error is twice as high - due to non-linearity - and can be seen in the lower diagram; it may attain values up to 1 LSB.

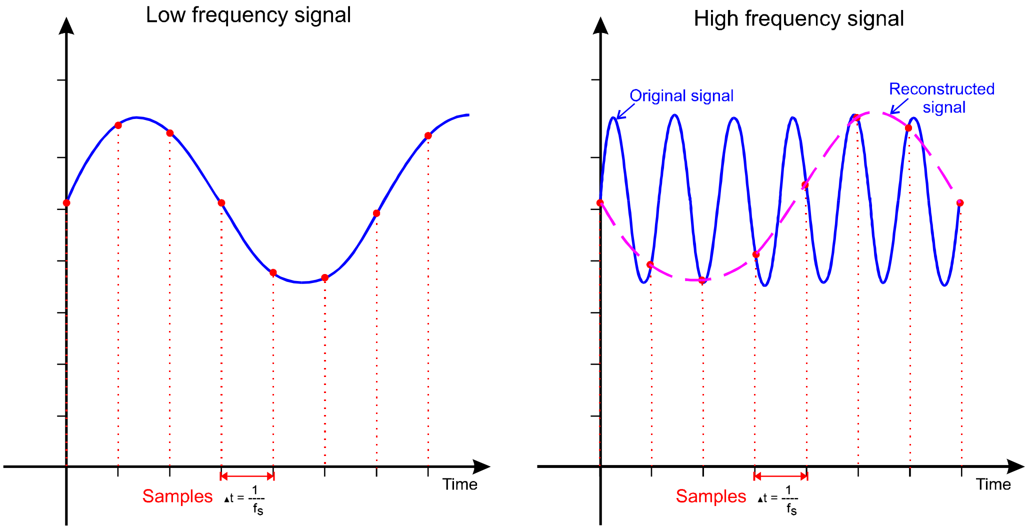



*Fig 2 Signal Sampling & Reconstruction*

### *Sampling*

The first signal has a frequency much lower than the sampling frequency, thus resulting in a correct signal reconstruction.

This means that signals with a level smaller than the last bit (called least significant bit or *LSB*) cannot be measured and are therefore registered as noise only. For instance, with a 16-bit converter, this noise floor theoretically is 1/655636 = 0.0015% or -96dB. Keep in mind that this dB value is the theoretical dynamic range of the converter, too.

***Each additional bit theoretically increases the dynamic range of an ADC by 6 dB (x 2). Consequently, each bit less decreases the resolution by 6 dB (÷ 2).***

In practice, the theoretical dynamic range cannot never reached due to imperfections of the measuring devices. A minimum loss of 3dB always has to be considered.

If the analog signal to be converted varies with time, its conversion into numeric values must also be done sufficiently fast in order always to represent the analog signal accurate enough. If the time variation is fast, the samples also have to be taken more frequently than with a slowly varying signal. Refer to *Fig 2* for a graphical illustration.

The number of measurements that are taken per second is called the *sampling frequency* ($f_s$). To measure a rapidly varying time signal such as a high frequency sine wave, at least two measurements per period of the sine wave signal have to be taken to allow reconstruction of the original signal. This fact is defined in the *sampling theorem*.

***The sampling rate has to be at least twice as high as the highest frequency in the signal.***

Or, expressed in another way, the highest signal frequency one may apply to an A/D converter must never be higher than half of the sampling frequency. If this theorem is violated, the higher frequencies are 'mirrored' at ½*$f_s$ and folded back into the signal band resulting in non-linear distortion which may be audible and thus must be avoided (see *Fig 3*).

For this reason, almost all A/D converters use anti-aliasing filters to make sure that no frequencies higher than half of the sampling frequency can pass through the converter.

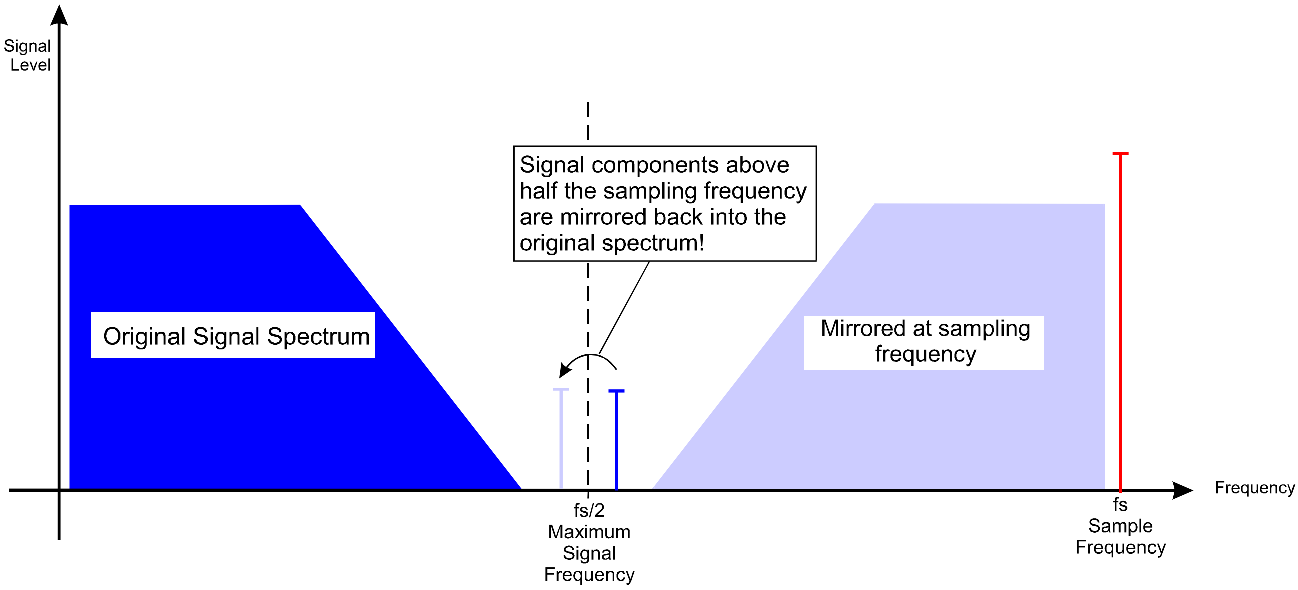



*Fig 3 Mirroring of Frequencies*

Professional audio equipment has to be able to handle frequencies up to 20kHz, so the minimum sampling rate required for processing all frequencies in this spectrum is 40kHz. However, due to the finite edge steepness of anti-aliasing filters in the rejection band, a sampling frequency of 44kHz or even 48kHz is necessary. Consequently, these two sampling frequencies have been standardized by the AES. A third sampling rate of 32kHz has also been standardized to cover the 15kHz analog bandwidth of broadcast quality.

## *Converter Parameter*

When characterizing a converter, two further, closely related, parameter of importance are the linearity and the accuracy, expressing the manner and the degree of deviation of the binary output from the ideal values. The inaccuracy of a converter is commonly expressed in LSB, and states by how many bits the output may be off (refer also to *Fig 1*).

Stability of the digital signal, particularly around the zero crossing line, is another important parameter of a converter. Especially temperature effects can influence it. During a transition from one number to another, many of the internal switches of the converter could change simultaneously. In reality this is not achievable and the switches change their state one slightly after the other. This leads to the fact, that during the sampling period an invalid number is present at the output. With a D/A converter, undefined output voltages between two steps may occur. These are called *glitches*. In practice, a hold stage keeps the output at the old value until the new sample has stabilized.

Again - particularly with D/A converters - it may happen that the input signal changes so fast that the internal amplifiers - even if they switch fast enough - cannot provide the required voltage change in the short period of time available, thus producing non-linear distortion. This effect is called the *slew rate limitation* and must not be underestimated.

# AES/EBU AND IEC958 STANDARD

Once the analog signal is converted into a digital signal, its further treatment can be done in parallel or serial. Parallel processing, in which all *n* bits of the *n*-bit wide sample are treated simultaneously (where *n* can be as high as 24 for audio signals), has the advantage of being fast compared to serial processing, but requires *n* wires or signal paths. Practical implementation of such circuits would often lead to either a lack or an excess of copper, depending on the application that might utilize fewer bits.
For this reason the standardization committee chose a serial format for digital audio data transmission. Besides the pure audio signal, a lot of additional information has to be transmitted, e.g. in order to enable synchronization between receiver and transmitter, to increase reliability and to encode the type of signal, application and parameter.
All these facts resulted in the AES/EBU standard for digital signal transmission. At the same time, the earlier de-facto standard, introduced by Sony and Philips for digital interfacing (dubbed SPDIF), has been standardized in IEC958 for unbalanced consumer links.

The main differences of the physical transmission between the AES/EBU and IEC958 standard are level, impedance of the cable and the type of connection (see also *Table 1*).
Both transmission standards are - besides the afore-mentioned differences - more or less identical, except that the AES/EBU standard defines the professional format only, while the IEC standard covers both the professional and the consumer format. The difference between these formats will be explained later.

In most applications, audio signals comprise two channels for stereo reproduction. The standards define the serial transmission of both channels one after the other, completed with additional information.

## *Definition*

The audio data of each channel is packed into 32 bit packages, called words. These start with a synchronization preamble, lasting four bits, followed by 4 bits auxiliary data. Follows the audio data with the LSB bit first. 20 bits are reserved for the transmission of the audio signal; if fewer are used, the unused bits are set to zero. After that follows the validity flag, a user bit, the channel status bit and finally the parity bit.

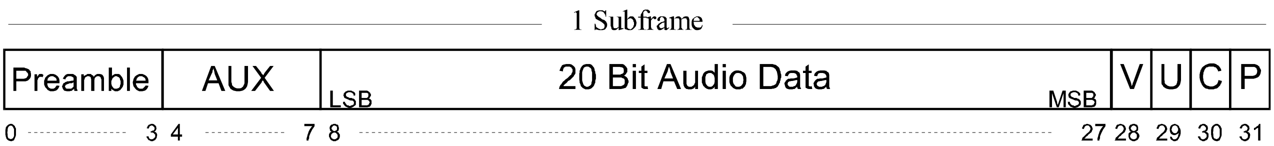



*Fig 4 Subframe for 1-Chn. Audio Data (32 bit)*


*Fig 5 Blockstart with X, Y and Z Preamble*

The *auxiliary* data bits can be used to extend the audio word to 24 bits, or - as the name implies - for transporting auxiliary information. The *validity* flag indicates whether the audio sample in that word is valid or not. The *user* bit can be used freely, although there exists an AES recommendation (AES18 / ANSI S4.52-1992) for the format of the user bit.

As explained below, the words are always grouped together into blocks of 192 words. The meaning of the *status* bit becomes clear in the context of a group of 192 status bits from a block, arranged in subgroups of 8 bits (called bytes); a detailed interpretation of these bytes will be given later. The *parity* bit is a checksum over the 32-bit word to find single bit errors (see also *Fig 5*).

One block of transmission contains the aforementioned 192 words of each channel as illustrated in *Fig 5*. Each word is headed by an individual channel preamble; the "X" preamble marks the A-channel and the "Y" preamble the B-channel. A special preamble, the "Z" preamble, replaces the "X" preamble at each beginning of a block to indicate the start of a new block.
This allows receiving devices to trigger to the data stream and find the beginning of a block, which is clearly necessary for the proper interpretation of the status.

## Transmission

Signal transmission is performed in the so-called *BiPhase* mark mode. This mode has been chosen to minimize the DC-component of the transmitted signal. The illustration below (*Fig 6*) shows that a logical "1" is represented by a polarity change in the middle of the source signal, while a logical "0" does not show this transition. Furthermore, each bit ends with another transition, independently whether its value is "0" or "1". Note that this requires the clock of the digital bitstream to be double the sampling rate times 64 (2 channels with 32 bits) which leads to a clock of ~6MHz at a sampling rate of 48kHz.

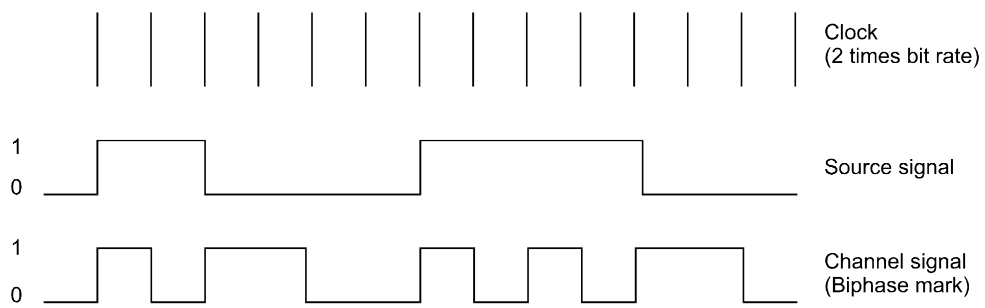



*Fig 6 BiPhase Mode*

## Channel Status Information

Every block of the transmission contains 192 frames, each with one "Channel 1" word and one "Channel 2" word. Each word contains one channel status bit. This leads to 192 channel status bits per channel and block. With the start of the next block, the channel status definition recommences. These 192 bits are grouped together as bytes (8 bits) that can also represent ASCII characters. The meaning of the bits and bytes in the channel status are different for the professional and consumer format. Actually, only bit 1 and bit 2 of the channel status in the two formats are identical to ensure proper discrimination between these two formats.

### Professional Format

The illustration of *Fig 7* shows the encoding of the channel status block for the professional format.

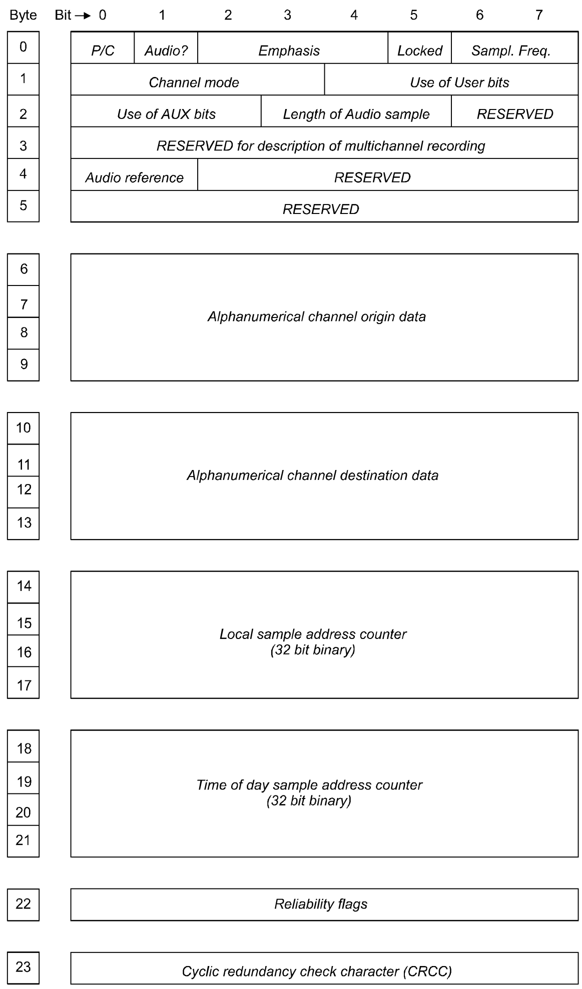



*Fig 7 Professional Channel Status*

### Byte 0

The first bit designates the type of format. A logical "1" indicates the professional format, while "0" represents the consumer format. Bit 2 indicates whether the transmitted data is audio data. The three "emphasis" bits define the type of emphasis used. Of the eight possibilities, only 4 are used and the others are reserved.
Bit 5 indicates whether the source sampling frequency has been locked onto or not. If this bit is "unlock" then the audio data is not reliable and may contain garbage. The following bits 6 and 7 contain the information about the sampling rate: 32kHz, 44.1kHz, 48kHz and "no indication" are defined. The sampling rates are not measured but set by the transmitter.

### Byte 1

The bits 0 to 3 encode the relation of the two channels. It ranges from "not indicated" to "two channel", "single channel" and "stereophonic". All further combinations are reserved for future applications. The second four bits are reserved for the "user bits management", defining the default mode, the "Z-preamble" mode and a user defined application. All others are reserved.

### Byte 2

Byte 2 defines the use of the auxiliary bits and the length of the audio data. The first three bits code the maximum length of the audio data to "Not defined", "24 bits", "20bits", or "User defined". Bits 3 to 5 designate how long the audio data in the previously defined frame really is. Reductions by 0, 1, 2, 3 and 4 bits are defined. The bits 6 and 7 are reserved.

### Byte 3

Byte 3 is completely reserved for future application.

### Byte 4

The first two bits define whether the audio signal is a reference signal and of what type, while the rest of the bits in that byte are reserved.

### Byte 5

Completely reserved for future applications.

### Byte 6 to 9

Theses bytes are interpreted as a set and represent the alphanumeric channel origin data. They are normally represented as ASCII characters. There exists no restriction on the characters that may be represented. Examples are "CD", "RDAT", "GEN" or "A2".

### Bytes 10 to 13

These are again ASCII data representing the channel destination data. The same restriction as for the channel origin data applies.

### Bytes 14 to 17

These form an ongoing sample counter 32 bits wide. Even though the counter increases by 48000 units per second, the maximum number that can be represented is 232 = 4, 294, 967, 296. The counter restarts at zero approximately every 25 hours.

### Bytes 18 to 21

These bytes represent in 32-bit binary format the time-of-day of the sample address. Together with knowledge of the sample address, the time difference between any frames can easily be calculated.

### Byte 22

This byte carries information about the confidence of the data in the channel status bits. Bits 0 to 3 are again reserved. A "1" in bit 4 signifies that uncertainty exists in bytes 0 to 5; a "1" in bit 5 indicates a possible error in bytes 6 to 13; in bit 6 bytes 14 to 17; and in bit 7 bytes 18 to 21.

### Byte 23

This byte contains a 32-bit checksum (CRC) over all channel status information to indicate possible bit changes during transmission. The polynomial is

$$G(x) = x^8+x^4+x^3+x^2+x+1.$$

## Consumer format

The consumer format was formed out of the earlier Sony and Philips standard, where copy protection was a major concern. The consumer status block defines much less information than the professional format. Only the first bytes are defined, the rest is unused.

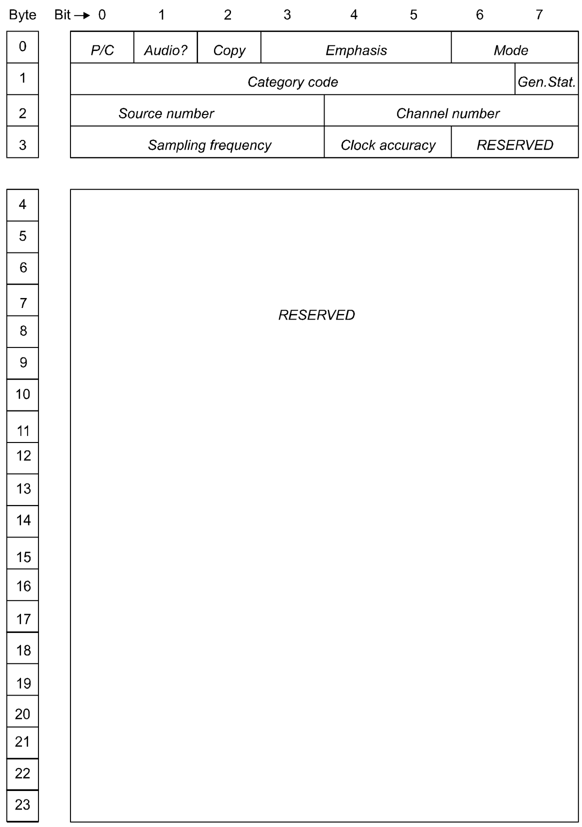



*Fig 8 Consumer Channel Status*

### Byte 0

The use of the first bit is identical with the professional format to ensure accurate identification of the formats. The bit 1 defines audio or non-audio use. Bit 2 - already different from the professional - is the copy protection bit; with this bit set to zero, no copy of the audio material is allowed. Bits 3 to 5 define the type of audio data, while bits 6 and 7 are always set to zero. The scheme for the copy protection is not yet fixed but under development.

### Byte 1

The first three bits define the category code that determines the interpretation of the next bits in that byte.

### Byte2

This byte defines in its first four bits the source number with LSB first. The next four bits define the channel number from "A" to "O".

### Byte 3

This byte decodes the sampling rate in bits 0 to 3. Bits 4 and 5 define the accuracy of the clock, while bits 6 and 7 are reserved.

### Byte 4 to 23

The content of these bytes is reserved for future use.

# ELECTRICAL REQUIREMENTS

Besides the type of information contained in the various bytes of a block, there also exist major differences between the professional and consumer format on the electrical signal level. The AES/EBU standard defines a balanced electrical signal transmission, while the IEC standard defines an unbalanced electrical or optical link. Further important parameters for the transmission are listed in the table below.

| *Type* | *AES/EBU* | *IEC 958* |
|---|---|---|
| *Connection* | XLR | RCA / optical |
| *Bal. / Unbal.* | Balanced | Unbalanced |
| *Impedance* | 110Ω | 75Ω |
| *Level* | 0.2V to 5Vpp | 0.2V to 0.5V |
| *Rise / Fall time* | 5 to 30ns | 0 to 10% / 0 to 20% |
| *CMRR* | 7VDC | > 30dB |
| *Clock accuracy* | not specified | I: ± 50ppm<br>II: 0.1%<br>III: Var pitch |
| *Transmission* | BiPhase | BiPhase |
| Jitter | ± 20ns | not specified |

*Table 1 Difference between AES and IEC Standard*

The list is not comprehensive and reflects some major differences only. For detailed explanation please refer to the respective standards.

## *Balanced Transmission*

The balanced transmission operates with 110Ω impedance and allows signal transmission distances of up to a few hundred meters. The acceptable carrier level has to be between 0.2Vpp and 5Vpp. The lower limit was initially defined at 2V but has been adjusted to accept also signals coming from an unbalanced transmission without the need for a re-amplification.

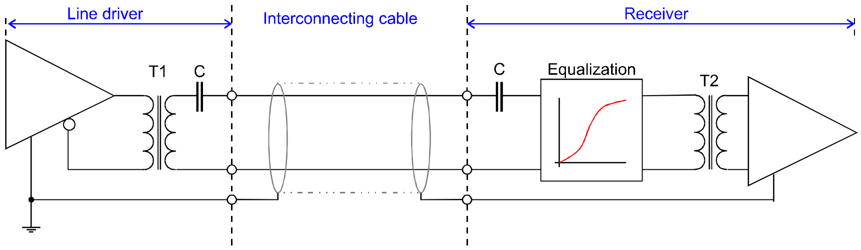



*Fig 9 Balanced Connection*

An early publication of the standard allowed connection of up to four receivers - each with the 110Ω input impedance - to one transmitter. The load impedance could in this case vary from 110Ω down to some 28Ω, certainly influencing the quality of the link (any reflection increases the probability of errors). However, for more reliable splitting, interface circuitry offering terminated outputs is recommended.
Please notice that for good and reliable RF shielding, **both** ends of the shield have to be connected, as opposed to the practice in audio connections, where - to avoid ground loops - sometimes one end only may be connected.

## *Unbalanced Transmission*

The IEC standard recommends an unbalanced or optical link with a permitted length of up to ten meters. Be aware that the optical connection with TOSLINK connectors and a non-mono mode fiber can introduce quite a remarkable amount of jitter into the digital system.

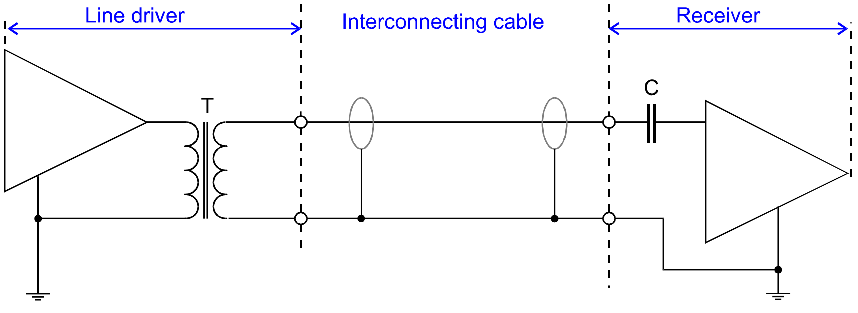



*Fig 10 Unbalanced Connection*

## *Jitter*

Jitter is defined as the time variance between the ideal and actual position of an edge of a digital signal. Most important for audio quality is the jitter between the edges that trigger the A/D conversion, i.e. the edges of the X-, Y-, and Z-preamble. This type of jitter is called word jitter. Jitter between neighbored bits is much less critical.

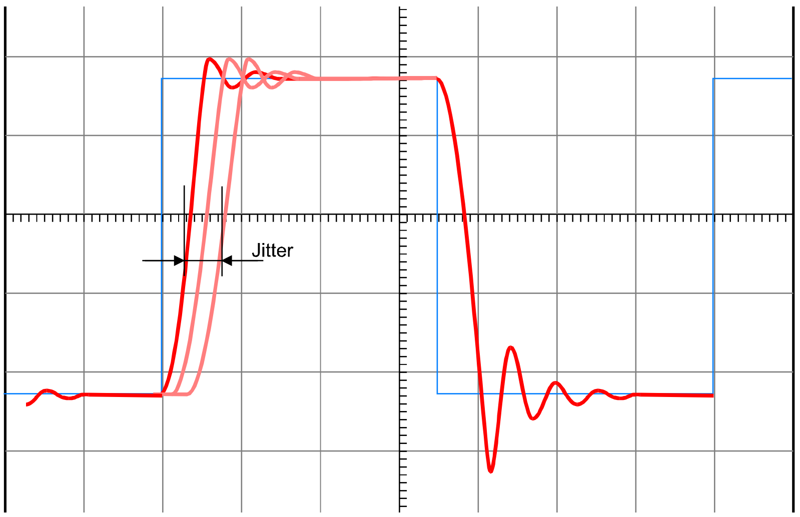



*Fig 11 Jitter*

The AES standard defines only the normal jitter of ~20ns and states that the clock jitter has to be specified more tightly! This allows for considerable leeway in interpreting compliance with the standard.
Jitter can arise or be "created" at any point in the signal path, especially on lines which carry signals with similar clock frequencies; the two signals might intermodulate leading to a time variance of the clock signal. Many professional equipment manufacturers therefore provide jitter attenuators or clock regenerators at the digital input, where the time variance is minimized or eliminated. To test these input stages, a digital signal with defined jitter has to be applied to find the limit where synchronization fails. The resulting jitter will be noticed first as increased distortion at the output of the converters due to the time base non-linearity of the conversion.

Jitter can be expressed either in time (mostly in ns), describing the total variance of the transition or in "Unit Interval [UI]", where the amount of jitter is set in relation to the sampling frequency.

## *Eye Pattern Test*

The eye pattern test is the only standardized audio quality test in the AES standard. It requires an analog oscilloscope with external trigger input and a digital audio analyzer with trigger output.

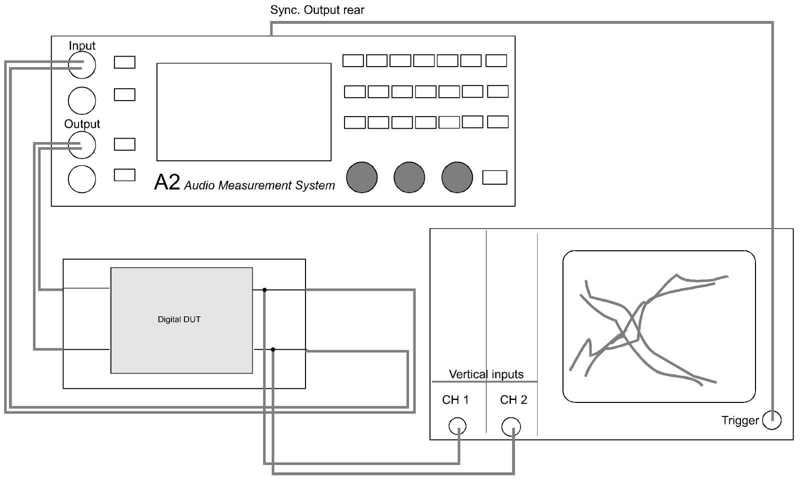



*Fig 12 Wiring for Eye Pattern Test*

### Wiring for Eye Pattern Test

The signal to be tested is fed to the vertical axis of an oscilloscope. Since the signals is transmitted in biphase mark, each input has to be fed to one channel of the scope. The signal of interest is the difference between the two traces, which can be obtained by subtraction.

The trigger output of the transmitter (i.e. of the A2) has to be connected to the trigger input of the scope; the trace restarts at each clock transition.

The reading of the results requires additional interpretation and calculation. Every transmission line has a low-pass characteristic, distorting the shape of an ideal square-wave signal. Time instability additionally varies the zero crossing of the signal.

The illustration in *Fig 13* explains how the eye pattern is created. The ideal square-wave signal is distorted in a way that it resembles a lowpass filtered signal. The trigger is synchronous with the positive edge of the square-wave signal. The trace starts with the positive slope. The scope is re-triggered on the negative edge of the square-wave signal to draw the trace of the negative shape. If the beam intensity is high enough, the screen will show several of the traces, forming one half of the oval eye pattern.

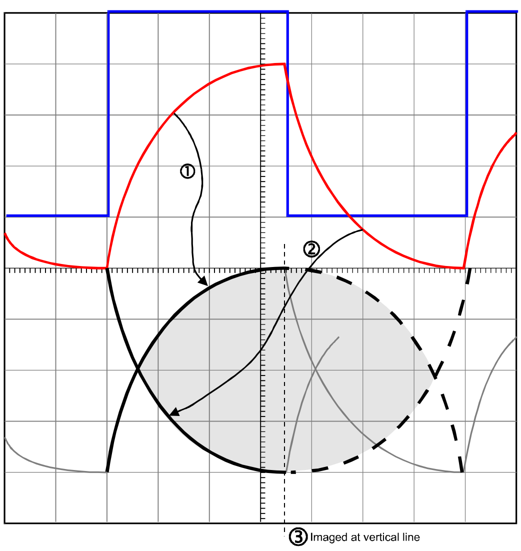



*Fig 13 Forming the Eye Pattern*

The opening of the eye indicates the level attenuation and degradation, while the width of the eye is influenced by the "1" and "0" as well as by jitter. The eye height shall be high enough to let the input circuitry safely decide between a logical zero and a logical one. Below a certain hysteresis the input device won't be able to make the decision and starts reading errors. This is the minimum opening height.

The AES standard specifies the minimum voltage to be bigger than 200mV for at least 50 % of the nominal eye width (see *Fig 14*).

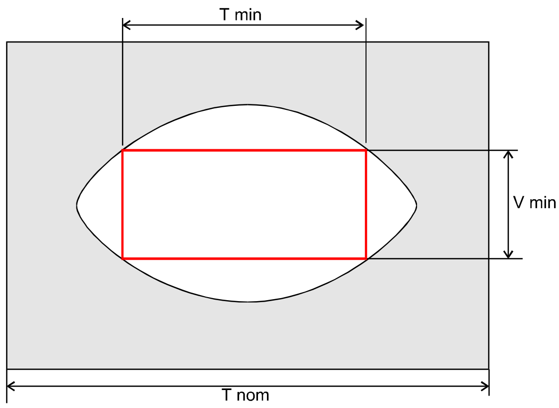



*Fig 14 Eye Diagram*

## MEASUREMENTS WITH A2 & AO10

The problems with digital signals are far more complex than those with analog signals. Furthermore, One cannot simply listen to the digital signal to verify the correctness of the selected channel and information.
The digital option AO10 now extends the capability of the NTI Measurement System A2 by enabling the generation, analysis and processing of digital signals according to the AES/EBU and IEC958 standards. With the A2-D, NTI has introduced the first portable, lightweight and very price competitive dual domain analyzer for research, service and maintenance.

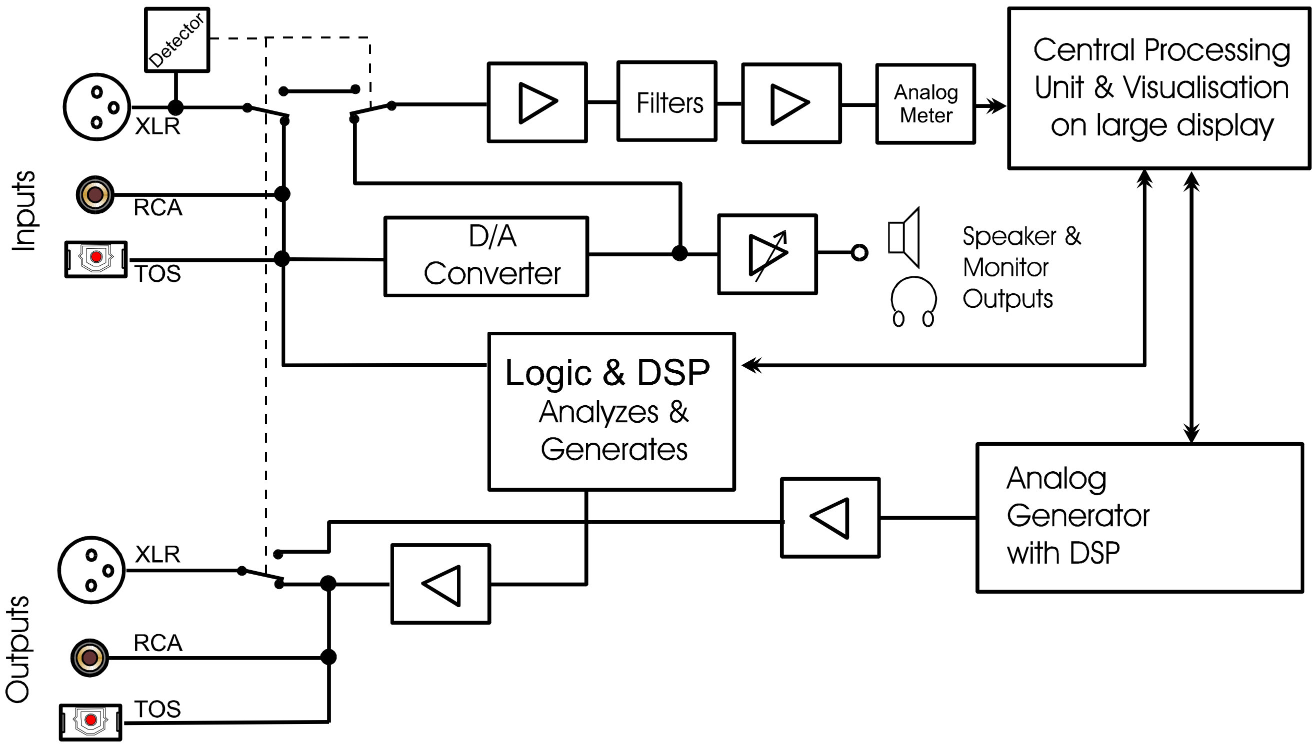



*Fig 15 Block Diagram of Digital Option*

One of the basic design ideas concerning the handling of the AO10 option was to keep the operation of the digital functions - especially the interpretation of the packed status information - as simple as the handling of analog functions.

In practice, the digital mode can be entered via the <PHASE> key. Simply press this button twice and the digital functions are enabled and displayed.
To return to the analog mode, simply press any of the purely analog function keys.

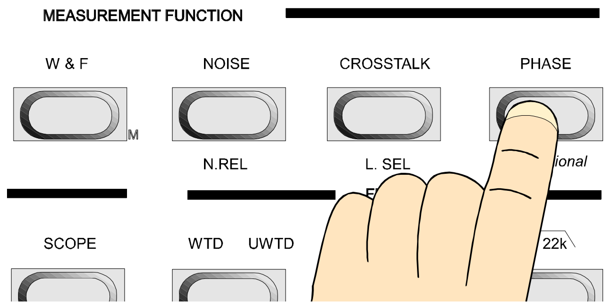


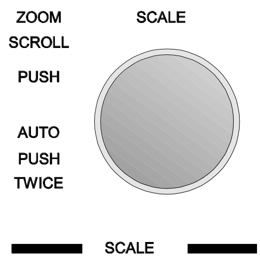

*Fig 16 Activation of the Digital Mode*

> ***Note: Activation of the digital mode is automatically performed if an AES/EBU signal is applied at the XLR connector of analyzer channel A (auto detector can be disabled).***

The digital functions present themselves in a screen that looks as illustrated in *Fig 17*. The structure is similar to the METER screen of the analog functions. The top two lines are reserved for the generator and its status, followed by the digital measurement result, and mostly accompanied by the digital bargraph. The last few lines are reserved for the status of the analyzer input.

## Operation

Operation of the digital features is very simple. All selections, alterations and modifications are performed through the <SCALE> wheel on the front panel of the A2. Each increment of the wheel moves the cursor (shown as inverted video) to the next selectable item.
A press of the <SCALE> wheel changes the cursor to normal video. Now, each turn changes the selected value. A second press of the wheel confirms the setting. The value becomes active immediately after the confirmation.




The only exception in this operation is the setting of the frequency and level of the audio data. They are - as in analog mode - altered with the wheels <LEVEL> and <FREQUENCY>.

> ***Note: Some selections are arranged as circular lists, allowing turning endless through the values and some are arranged as linear lists, with a beginning and an end.***

## Measurement Functions

### Peak Level

The measurement function "PeakLvl" measures the digital peak level of the audio signal. This is a completely digital function. The available units, % and dBF, can be selected through the <UNIT> key. The reference (100%, 0dBF) is always the upper limit of the converter. Percentage is the linear representation of the dynamic range while dBF shows the logarithmic scale.

## RMS Level

This measurement function uses also the analog features of the A2 by converting the digital audio signal through a high performance D/A converter and feeding it into the analog analyzer. The RMS value is obtained in the common units Volts, dBu and dBV.

## THD

Like RMS, the measurement function THD uses the analog circuitry of the A2 to measure the total harmonic distortion of a digital signal by converting it to an analog signal. By this way, dropouts in the bit stream as well as hanging bits and other, previously explained non-linearity of the DUT are easily detected.
Also, the influence of jitter on THD can be monitored with great sensitivity.

## (Channel) Status

This measurement function focuses on the status of the incoming signal. It illustrates and lists all the meanings and settings. With the given information on the screen, together with the knowledge of the status given in this application note, the contents of the status information can be perfectly deciphered.

Take care to note that the appearance and interpretation of the Status display in the professional and consumer modes differ completely.

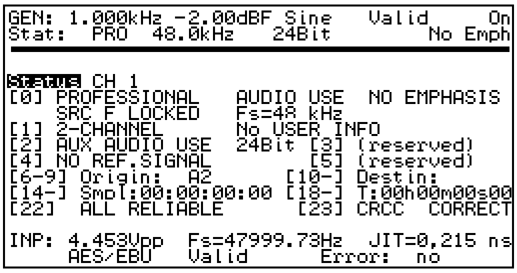



*Fig 17 Channel Status in PRO Mode*

## Bit Statistics

Bit statistics provide an immediate verification of which bits in the data stream contain changing information, and which bits remain always stable at zero or one. The screen shows both channels at the same time. Steady "zero" bits are marked with a "0" and the stable "one" bits with a "1". Continuously changing bits (usually the audio information, the status and the parity bit) are marked with the special "up-down" arrow.

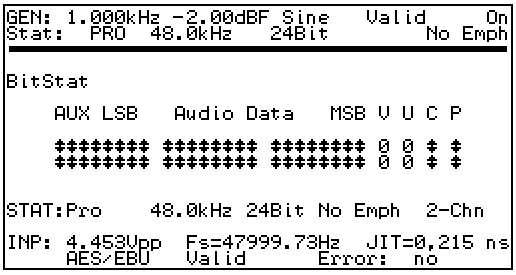



*Fig 18 Bit Statistics*

## Varying the Carrier Level

In order to test the behavior of the D.U.T when applying a degraded carrier signal with reduced carrier levels the generator of the A2-D can vary the amplitude of the AES3 carrier signal in the range from the maximum 5V down to levels of 0.15V (see *Fig 19*).

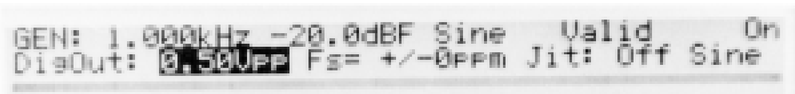



*Fig 19 Carrier Level, Sampling Frequency & Jitter*

## Jitter Measurements

Jitter measurement is continuously active in all digital measurement screens because of independent circuitry steadily monitoring the jitter. The AO10 measures the - for the D/A conversion - critical word-jitter with resolution of <0.1ns.

Note that the generator provides also a jitter generation mode, allowing the output of digital signals with defined jitter values, ideal for testing jitter acceptance. It furthermore allows changing the modulating jitter source to have either a sinusoidal- or a white noise characteristic. See *Fig 19* above. The last two entries in the second line control the amount of jitter and the modulating signal.

## De-Tuning of the Sampling Frequency

Depending on the circuit design of the AES signal receiver it might have or not the ability to follow variations of the sampling frequency of the source. The A2-D has therefore a feature implemented that allows varying the standardized sampling frequency in the range of ±1500ppm to easily check in the D.U.T the PLL's ability to follow the changed sampling frequency.

## Mixed Signal Operation

As expected and required for thorough tests of a dual domain device, the A2- has the full ability to operate in the purely analog-, purely digital as well as in the mixed Analog/Digital and Digital/Analog mode. This means that whatever analysis mode has been selected, the generator can be alternatively run in the other domain. The dual domain capabilities are of course also available for all sweep types. This ensures that with one single instrument the performance of A/D and D/A converters can be tested.


*Fig 20 Printout of Mixed Signal Sweeps*

*Fig 20* shows a mixed signal sweep printout. The generator was sweeping over its amplitude in the analog mode from -80dBu to 0dBu. The analyzer was set to the digital mode reading the AES3 bitstream and performing a THD+N analysis. Two traces have been recorded, one with the converters limiter active and the second one with no limiter.

Obvious on the graph is that the THD value is steadily decreasing with an increasing level, showing nicely that the S/N ratio is constantly increasing. But at analog levels of ~-24dBu the limiter becomes slowly active and, as a side effect, shows unwanted distortion. The trace without the limiter still remains decreasing but as the full scale of the converter is reached, it rapidly increases the distortions. The absolute, value where the distortion start to raise, depends on the gain setting (Level control) of the D.U.T. and might vary from device to device.

The two traces have been stored in the non-volatile memory of the A2-D and were printed out later by simply connecting the instrument to a printer through the A2-D's Centronics interface.

## *Spectrum Analysis (FFT)*

With the DSP capabilities of the digital option PCB in the instrument and by adding the inexpensive FFT option (article code AO12), the A2-D features also spectral display of the analog analyzer signals. The spectral display is offered as an alternative display method to the SCOPE mode. The button becomes a toggle function where the user can switch between time- or frequency-domain display.

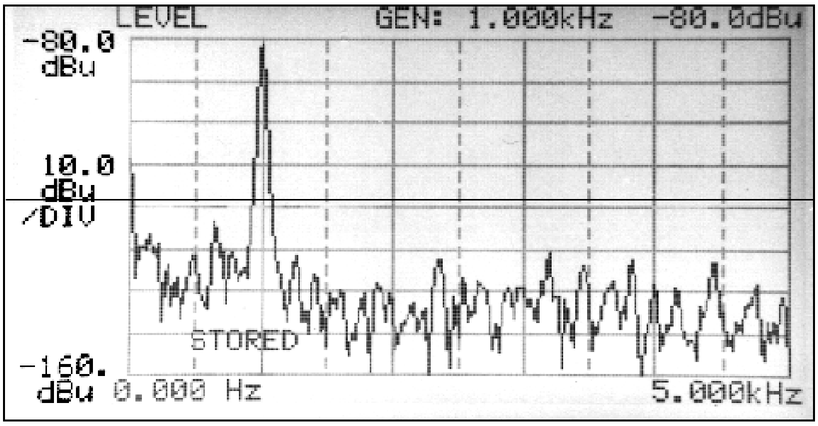



*Fig 21 Spectrum of a Low Level Input Signal*

The frequency resolution of the spectral display can be selected either from 0Hz top 5kHz or in a wideband range from 0Hz to 20kHz, thus covering the entire audio band. The screen is continuously updated with more than two readings per second. Sensitivity of the display is controlled - as usual in the A2-D operation - with the SCALE wheel, either automatically or manually with ZOOM and SCROLL.

## *Remote Control through AS04 Software*

If measurement processes have to be automated in order to minimize the amount of labor and the potential of false operation, a comprehensive software package to completely remote control the A2-D is available. The software runs on all 486 or Pentium processors under Windows 3.11 or 95. A powerful programming tool, called AMSL (Audio Measurement System Language) allows writing automated sequences with BASIC like commands.

Ask your local NTI representative for a free demo package.

# SPECIFICATIONS OF AO10

## *Inputs*

*AES3 via XLR A at front with 110W termination*
*IEC958 via RCA and TOSLINK with 75Ω termination*
*Sample clock through BNC at back*

| | |
|---|---|
| *Sampling frequency* | *32kHz, 44.1kHz, 48kHz or external* |
| *External clock:* | *25kHz to 52kHz (locking range)* |

## *Measurement Functions*

*Digital input signals (biphase)*

| | |
|---|---|
| *Peak level of data carrier* | *100mVpp to 10Vpp* |
| *Word clock frequency* | *25kHz to 50kHz* |
| *Resolution* | *0.05Hz* |
| *Accuracy* | *2ppm calibrated* |
| *Word clock jitter* | *0 to 40ns* |
| *Rectifier* | *Peak* |
| *Bandwidth* | *700Hz to 20kHz* |
| *Level measurement* | *-120dBF to 0dBF<br>0 to 100%* |

*Bit statistics of complete 32-bit word*

| | |
|---|---|
| *Status display with decoded information* | *for AES and IEC format* |

*D/A converted signal*

| | |
|---|---|
| *THD+N* | *See analog specs. of A2.* |
| *Res. of D/A converter* | *< -94dB* |
| *RMS Level* | *RMS calibrated to 0dBr with 0dBF (Sine)* |
| *Mixed signals* | *A-D or D-A* |

## *Outputs*

*AES3 via XLR A at front. 110Ω termination*
*IEC958 via RCA and TOSLINK with 75Ω termination*
*X, Y, Z trigger output via BNC at the back*

| | |
|---|---|
| *Sampling frequency* | *32kHz, 44.1kHz, 48kHz<br>or input clock<br>or analyzer clock.* |
| *Accuracy* | *10ppm (2ppm on requ.)* |
| *Detuning sampling freq.* | *± 1500ppm (steps of 100ppm)* |
| *Carrier output level* | *100mVpp to 5Vpp (loaded into 110Ω)* |
| *Signal waveforms* | *Sinusoidal, Sinusoidal with Dither, Pass Mode, Pass-Mode plus Dither* |
| *Frequency range* | *10Hz to fs/2* |
| *Level range* | *-120dBF to 0 dBF (0% to 100%)* |
| *Resolution of Audio word* | *4, 8, 12, 16, 18, 20, 24 bit* |
| *Dithering probability* | *1LSB TPDF (Triangular density function)* |
| *Internal jitter generator* | *2ns to 40ns (-84dBUI to -54dBUI @48kHz)* |
| *Modulation signal* | *Sinusoidal 2.5kHz or White noise 20Hz to 20kHz* |
| *Output impedance* | *110Ω with AES3 signals* |

## *Ordering Information*

*The digital measurement extension board AO10 fits in all A2 instruments with serial numbers >400. For retrofit of older units, please contact your local NTI agency.*

*A FFT option for the spectral analysis of a 1-channel analog signal is available for all A2-D units.*