TI 315
D12 AES/EBU Input/Output and Wiring (1.0EN)

d&b audiotechnik

Introduction

Unlike traditional power amplifiers, the D12 is able to directly accept an AES/EBU digital audio signal as an input alternative to ordinary line level analog audio. A dedicated XLR3F connector is provided for this purpose, marked "DIGITAL AES/EBU". The adjacent XLR3M connector, standard with all D12 amplifiers produced since 2005-09, is for linking to further amplifiers.

When connecting inherently digital audio equipment fitted with both analog and digital interfaces, the digital option is generally preferable. Connecting digitally avoids unnecessary conversions between the analog and digital domains. As these conversions are the main source of distortion, errors and delays in a digital system, maximum audio fidelity is maintained.

AES/EBU digital audio is data, not audio and therefore requires different handling and interconnection techniques from line level analog audio.

AES/EBU standard

AES/EBU stands for Audio Engineering Society / European Broadcasting Union and is the common term for the AES3 standard for serial transmission of digital audio over twisted pair cable, first published in 1985. It has been refined several times, and the present official definition is enshrined in AES publication AES3-2003. [1]

Electrically, the AES/EBU interface uses a three wire balanced connection in accordance with the RS422 standard for data transmission. See Fig. 1. Note the two signal wires are a twisted pair.

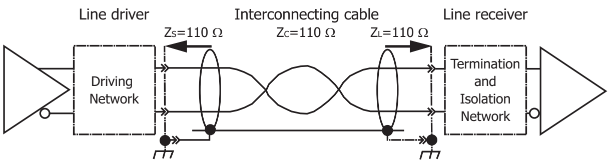


Fig. 1: AES/EBU – physically (Source [1] AES 3, Fig. 6)

Signal amplitude is 2...7 Vpp, source and load impedance is 110 Ω ± 20 %.

As connectors 3 pin XLR are being used.

The AES/EBU standard allows for two channels of audio data at up to 24 bits resolution. In addition, the AES/EBU signal also carries meta data which contains information on the used audio format.

Sampling rates and latency

Any audio signal in a digital audio system has a sampling rate associated with it. The two most common sampling rates are 44.1 kHz, used in consumer applications and 48 kHz, which is used in professional and broadcast applications. Multiples of these rates, such as 96 kHz, may also be encountered.

Care needs to be taken to ensure sampling rate compatibility between a digital audio source and the equipment it is driving, incompatibility will result in silence.

Recent equipment often uses an asynchronous sample rate converter (ASRC) on its input to adapt the external signal to the internal audio clock. This leads to a maximum flexibility in interfacing different sources, but is achieved at the expense of an increased overall latency (typically by approx. 2 ms) and some decrease in audio quality. For technical reasons, this ASRC can not be bypassed, even if the audio signal's sampling rate correspondents with the internal processing rate.

Therefore it is advisable when the main components of the digital audio system are operating with a uniform sampling frequency, passing on ASRC completely. In such a set up all devices are operating synchronously with a common clock provided by the digital audio source, which is usually the digital mixing console. Devices connected to an AES/EBU output of the source can recover the clock from the digital signal and synchronize their internal clocks on it.

Especially in live sound applications the sum of latencies within the signal chain has to be observed. Besides an ASRC, every analog to digital (ADC) and digital to analog (DAC) conversion also causes latencies and conversion artifacts. Wherever possible, interconnection of equipment should be maintained in the digital domain to avoid these effects. Additionally, a digital audio transmission avoids the decrease of audio quality caused by long cable runs, which is a common problem in analog audio wiring.

It is important to carefully consider the latency of each signal path within multi-channel systems. Signal path latencies add up with the delay caused by acoustical path differences of the sources. Undesired acoustical effects can therefore only be avoided when the complete electrical and acoustical paths are time aligned, for example by using the D12 delay function. A timing accuracy of between 1 and 2 ms is usually sufficient to avoid coherency problems between different arrays or groups of loudspeakers within one system. Within one array, however, a much higher accuracy is necessary which can only be achieved by using identical signal paths for feeding the amplifiers.

Please note that using the D12 digital signal input can reduce the latency in the signal path but will not eliminate the small internal latency of the D12 signal processing of about 0.3 ms.

Topology of an AES/EBU wiring

The AES/EBU interface allows, in contrast to CAN or DMX512, only a point-to-point connection. Every cable segment may only comprise one transmitter and one receiver. Signal distribution with Y-cables or simple loop through, which is common in analog audio wiring, is not possible here as it leads to impedance mismatch.

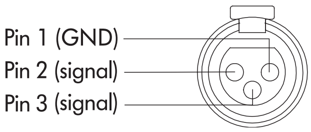
Further devices can only be supplied with the digital signal if an output with a buffered signal is provided, allowing a daisy-chain topology where all amplifiers are connected in a row. (see section . Application examples, Fig. 3).

In absence of such an output, external distribution amplifiers (DA) have to be used. These provide the input signal electronically refreshed on multiple outputs, leading to a star topology of the network. (see section . Application examples, Fig. 4). It may be possible to use a passive splitter instead of a DA under some circumstances. (see section Passive splitters)

AES/EBU INPUT of the D12

The AES/EBU XLR3F input connector is found on the D12's rear panel, below the analog inputs. The input is transformer-coupled for isolation; additional RFI suppression gives good immunity against external interference.

As the AES/EBU input is essentially a balanced input, the XLR pinout is standard:



Internally, the D12 uses a sampling rate of 96 kHz. The AES/EBU input accepts signals at either 48 kHz or 96 kHz. In the former case, a synchronous sample rate conversion to 96 kHz is performed by a software algorithm, resulting in a much lower latency compared to an ASRC.

The D12's AES/EBU input does not accept other sampling rates, direct connection of devices such as CD players is not possible as these use a sampling rate of 44.1 kHz.

The selection of the desired signal source is located under the item "Input" in the settings menu of the D12, where it is possible to switch between analog and digital input. When digital input is selected, the sampling rate of the input signal is displayed as soon as the D12 has locked on a valid signal. For sampling rates differing from the supported ones, the display shows the detected rate, followed by a '?', but the audio data will not be processed. If the sampling rate is not indicated, there is either no input signal at all or the signal is corrupted in such a way, that the D12 can not lock on it.

IMPORTANT: Level settings must be very carefully observed if a digital audio source, other than the output of a mixing console, is connected directly to the D12 input. Digital audio is recorded with reference to a maximum digital clip level of 0 dBFS (dB Full Scale), this corresponds to an analog input signal of +27 dBu at the D12. If the level of the D12 is not reduced adequately (by -30 dB, for example), the signal will be amplified at the maximum possible level (the GR and OVL indicators will illuminate). Always set the D12's gain to a minimum before connecting an AES/EBU source directly to the D12 for the first time, and then increase it gradually until the required volume is achieved. If the OVL indicator flashes after the gain has been trimmed, this is not a sign of an output overload. In this situation, the OVL indicator is showing an input level that is greater than -3 dBFS.

AES/EBU LINK of the D12

For the connection of further devices with AES/EBU input the rear I/O panel of the D12 is equipped with an XLR3M AES/EBU LINK connector, where the digital input signal is available electronically refreshed. An active circuit corrects the degrading by cable losses of the edges of the input signal and restores the standard amplitude.

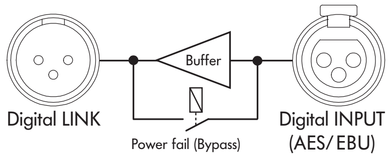


Fig. 2: D12 Digital INPUT and LINK

The internal connection also includes a failsafe bypass relay. In the event of power failure in the supply to the D12 amplifier, the relay connects the INPUT directly to the LINK output so that further equipment in the chain still receives an input signal.

Cable

Much greater attention must be paid to the choice of cables used to transmit AES/EBU digital audio than with analog audio. Because the digital audio signal is essentially high-speed data, cables suitable for a far higher bandwidth are needed. Instead of the 20 kHz for analog signals, a bandwidth of more than 12 MHz is required for transmission of an AES/EBU signal with 96 kHz sampling rate.

For a reliable transmission of these high-frequency signals the impedance of the cable has to be matched to the internal resistance of sender and receiver. A mismatch leads to reflections which overlay the original signal. Together with other interferences this will lead to higher jitter levels at the receiver. Jitter is the deviation from the ideal of the timing of an digital event. It disturbs clock recovery and decoding of audio data from the AES/EBU signal and is therefore one of the main reasons for transmission errors.

Because of their inadequate and unpredictable properties at the high frequencies needed for data transmission, standard microphone cables are not suitable for digital audio interconnections. Compared to a digital cable they have a higher capacity and therefore an impedance lower than required, leading to the previously described effects.

For transmission distances up to 100 m between the signal source and the last device in the chain, use a screened twisted-pair cable with an impedance of 110 Ω at all frequencies up to 128 times the used sampling frequency. Cable that meets these requirements will generally be marketed as being suitable for digital audio. Some cable types with lower HF attenuation may permit transmission distances of up to 200 m. These figures apply to an AES/EBU signal at 96 kHz sampling rate;

practical distances may be double at 48 kHz. Users are recommended to experiment with cable lengths before setting a system up.

For longer transmission distances the use of 75 Ω coaxial cable according to AES-3id [2] is recommended (see also section . Application examples, Fig. 5). These usually have narrower impedance tolerances and a lower attenuation and are therefore dedicated for the transmission of high-frequency signals. For interfacing the coaxial cable to devices with XLR connectors a format converter is needed (refer to section . Accessories under Format converters). Possible transmission distances are also dependent on the cable quality, but are usually greater than 500 m at 96 kHz sampling rate.

Choice of cable type will also depend on whether the system is portable or fixed. Cables designed for permanent installation are generally superior in performance at the expense of physical flexibility and ruggedness.

Additionally it is possible to transmit AES/EBU signals over computer network cables, provided they meet the CAT5 standard or higher. Note that common types are not suitable for portable use. Transmission quality is comparable to a good AES/EBU cable, even though the impedance of CAT5 cable is only 100 Ω. Only one pair is required for the AES/EBU signal. The remaining pairs can be used for other purposes like carrying further AES/EBU links or a CAN signal for remote control of amplifiers.

It should be noted that significant differences in performance exist between the various cables sold as suitable for digital audio. The figures for possible transmission distances given above are based on laboratory tests of multiple samples of several brands and types of cable and represent the values the majority of the samples have achieved or exceeded. For applications with higher requirements, please contact the d&b audiotechnik support for further advice.

The list given below is of those where samples have shown to provide reliable data communication at 96 kHz sampling rate over distances of 100 m (this list is not comprehensive):

- Belden 1696A
- Belden 1800B
- Cordial CDMX1
- Cordial CDMX234
- Draka AC10SS 24/7 1P
- Draka Mikro22 Outside AES/EBU 1P
- Gotham GAC-2/foil AES/EBU
- Kabeltronik DigiOne
- Klotz OTW204
- Klotz OT206
- Sommer Binary234

Accessories

As will be seen from the preceding paragraphs, much greater care needs to be exercised in distributing AES/EBU digital audio than analog audio. In some cases, some items of additional hardware may be required. Note that none of these items alter the audio data itself in any way.

Passive splitters

Passive splitters distribute the signal on multiple outputs by use of transformers. These do not compensate for cable losses and will always exhibit signal attenuation. Therefore the range of application for passive splitters is limited to set ups with an input signal of good quality, short cable runs and only a few devices to be fed.

Distribution amplifiers

Distribution amplifiers condition the input signal actively for multiple outputs.

There are two methods in use for refreshing the digital audio signal: Repeating and reclocking.

Repeaters use an active circuit to restore the waveform edges of the input signal and reset the output signal amplitude to the standard level. They do not correct for any inherent jitter in the input signal, and thus should only be used in situations where the received AES/EBU signal is known to be of good quality. The D12 LINK output is buffered using this method.

In a situation where the digital audio signal has suffered deterioration resulting in jitter, it may be necessary to use a distribution amplifier that reclocks the signal. Rather than reshaping the waveform, a reclocker effectively regenerates the signal as new, using a cleaned-up version of the signal's embedded clock to time the new data. With this method, the DA outputs are virtually identical to the original source signal.

Depending on the circuit design, reclocking may introduce a small additional amount of latency, this must be taken into account when calculating total system latency.

Format converters

If 75 Ω coaxial cable is used for distribution of AES/EBU digital audio, a format converter will be required to transform the unbalanced signal on the coaxial cable to the balanced 110 Ω signal required by an AES/EBU input, or vice-versa.

Format converters are available in both passive (transformer) and active (electronic) versions.

Application hints

With analog audio distribution, audio quality can be said to decrease linearly with distance; the longer the cable, the more the degradation.

Deterioration of digital audio is different. Digital audio transmission is lossless, as long as the digital carrier signal can be decoded without errors. With decreasing carrier quality, several clicks or short dropouts will occur, followed by a complete breakdown of transmission. The margin between a just error free transmission and complete breakdown is usually very small, so that minor additional disturbing factors (for example interferences from lighting dimmers or sharp bends in the cable) could interrupt a transmission which has already been close to the limit, but working without problems so far. Therefore it is essential to design digital audio connections with sufficient reserve, distances close to the limit should be avoided.

In absence of measurement capabilities for signal quality (for example an oscilloscope for checking the eye pattern of the carrier signal), the maximum possible distance can be found by practical tests. It is advisable to arrange an experimental set up with the same devices and cables which are going to be used later, where the cable length is increased until the first transmission errors occur. In practice not more than 75 % of this length should be utilized to have a reserve for potential interferences.

In order to avoid discontinuities in impedance, only one cable type should be used within a line segment. Every junction with different impedances at both sides will cause a part of the signal to be reflected back towards the transmitter, resulting in a decrease of carrier signal quality. Even when the cables have the same nominal impedance, the actual value might be different due to tolerances. With a repeater in between, different cable types may be used.

References / Literature

Further information on digital audio wiring can be found in the following documents:

[1] AES3-2003: AES Recommended practice for digital audio engineering - Serial transmission format for two-channel linearly represented digital audio data.

[2] AES-3id-2001: AES information document for Digital audio engineering - Transmission of AES3 formatted data by unbalanced coaxial cable.

[3] EBU User Guide: AES/EBU digital audio interface: engineering guidelines.

[4] AES Convention Paper 5915: Return Loss and Digital Audio.

Application examples

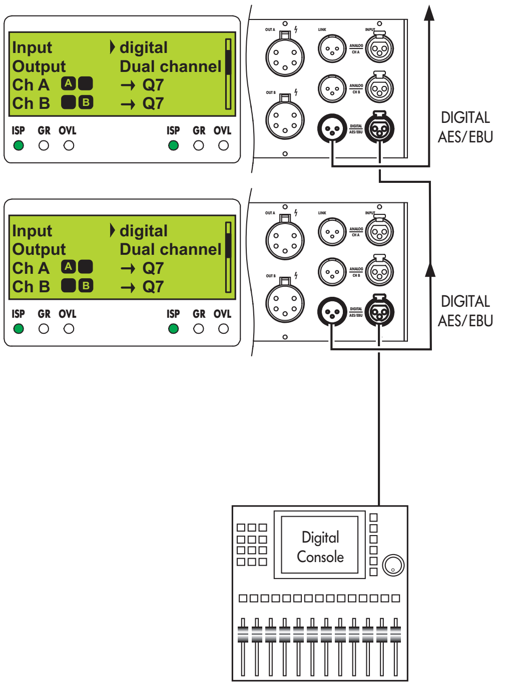


Fig. 3: AES/EBU wiring of D12 with LINK output

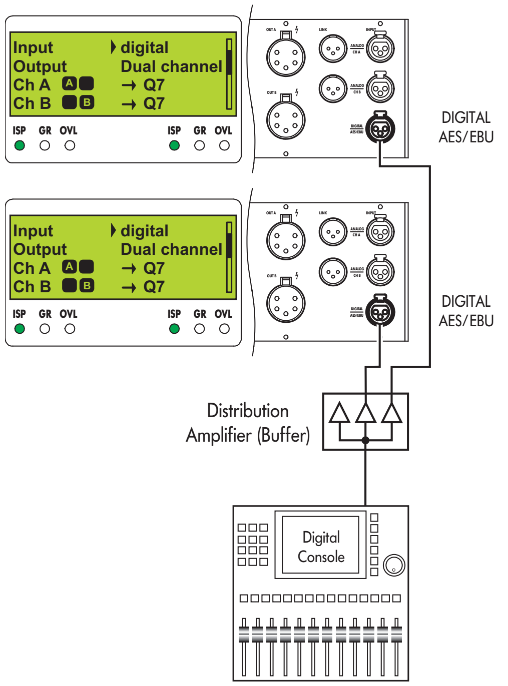


Fig. 4: AES/EBU wiring of D12 without LINK output

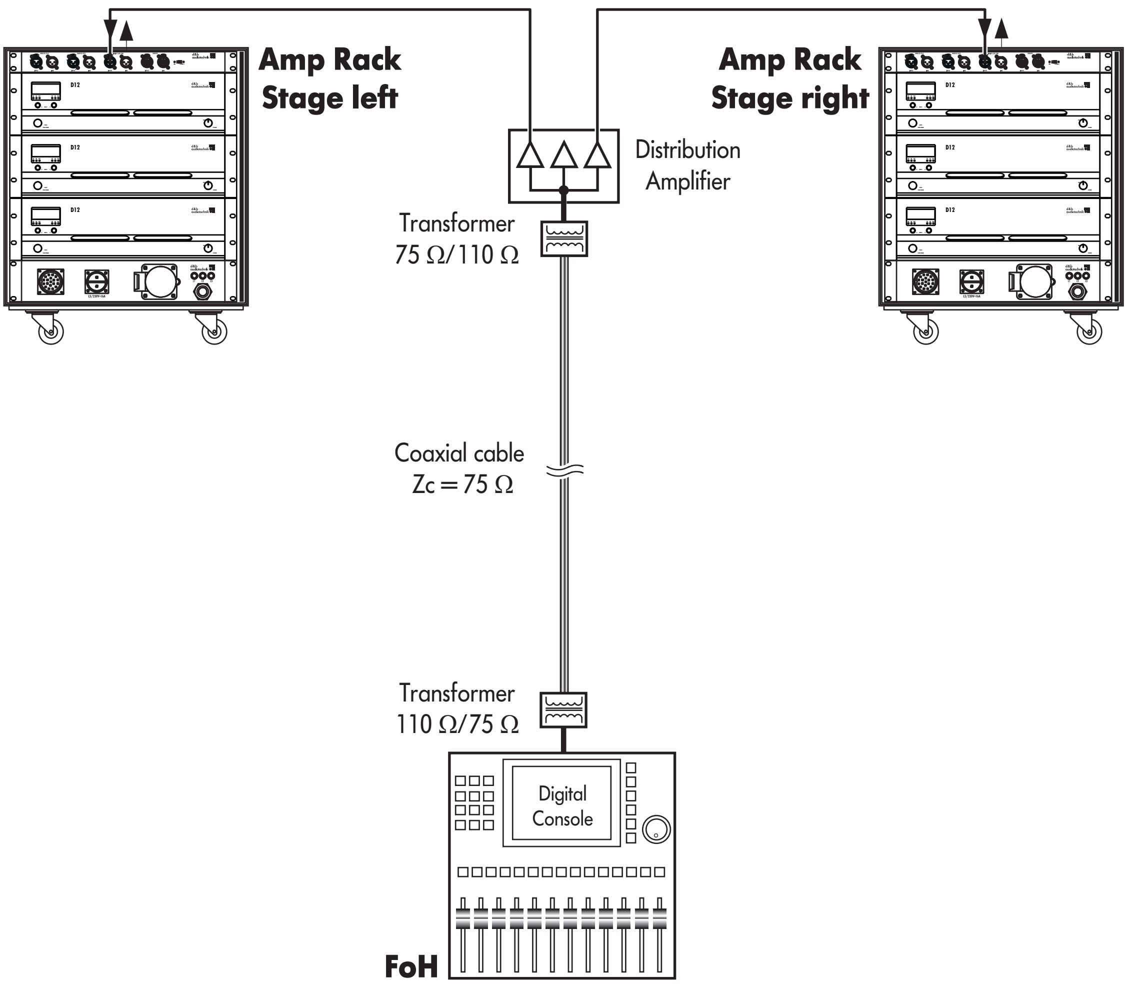


Fig. 5: AES/EBU wiring with coaxial cable and distribution amp for long distances

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