AES3, AES/EBU

AES3: Audio Engineering Society Standard #3

EBU: European Broadcasting Union

The AES3 interface was already specified in 1985 and it was made a standard in 1992. Ever since the standard is recurrently updated and adapted to advanced requirements. From there the standard is very universal and it's applicability is manifold. On the other side this makes it somewhat complex.

Specifications

- 2 channels
- balanced transmission
- XLR connector
- audio data up to 24Bit / 192kHz
- cable length: 100m and more
- impedance: 110Ohm (± 20%)
- level: 2 - 7 Vpp at the output side of a unit (at 110 Ohm, without long wiring)
- large channel status information

Comparing AES3 and AES/EBU

The AES3 digital audio interface and the AES/EBU digital audio interface only differ in this one detail: the EBU (European Broadcasting Union) standard regulates that coupling transformers are mandatory at each interface be it sending or receiving side. This is only optional with the AES3 standard.

Functionality

One aim developing the AES3 standard was to allow for digital data transmission the reuse of the cable network well established for analog audio signal transmission, a network often hunderds of kilometres long in facilities as broadcast studios. These are balanced cables and they allow signal transmission with frequencies up to about 10 MHz with cable length up to 300m if appropriate signal equalization is performed. To transmit digital signals over these analog audio cables some conditions have to be met, but they can easy be fulfilled:

- The signal has to be free of a DC component as transformers may be within the transmission chain.
- The signal itself must carry clock information as there are no separate lines to feed the bit clock and sample clock.
- Reversal of polarity should not have any influence on retrieval of the audio information.

These conditions can be met by a bi-phase-mark coding scheme.

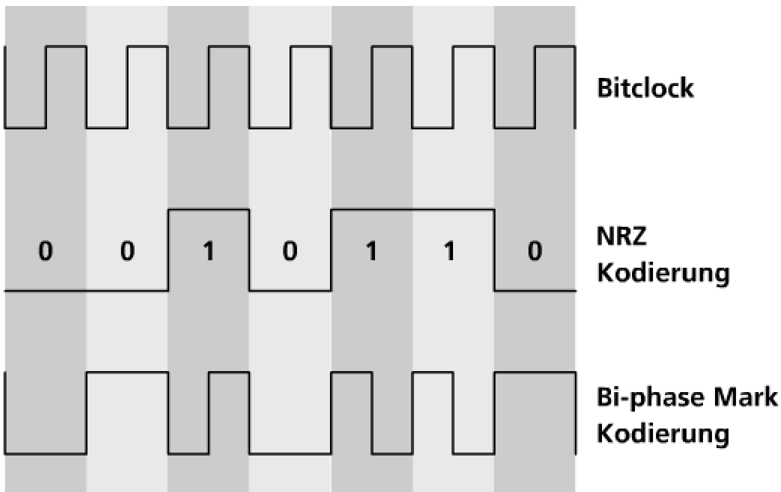


Figure:
With Bi-Phase-Mark coding each bit boundary is marked by switching over signal polarity. To distinguish a digital ´1´ from digital ´0´ an additional transition is inserted in the middle of a ´1´ bit. This code is proof against polarity reversal and it contains no DC component. So it can pass transformers. Even long ´0´ or ´1´sequencies within the bit stream contain recurrent alternations in signal state. So the bit clock is easily recovered.

With AES3 each sample of one audio channel is part of the so called subframe. A subframe consists of 32 bits arranged in this format:

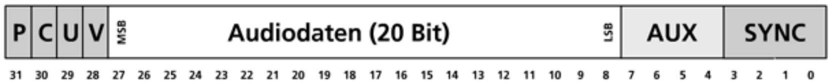


Figure:
The AES3 subframe. Transmission of bits takes place as numbered. One may imaginate the subframe be shifted out the paper to the right.

- Audio data: 20 bits are available for audio data. If more bits are required 4 Aux bits (auxiliary bits) can broaden sample resolution. Independent from sample resolution the most significant bit (MSB) is located at leftmost position (bit position 27). Bit positions not used are filled with zeros.
- AUX: Mostly the 4 Aux bits serve to broaden audio sample resolution to 24 bits or they are set to zero. Only occasionally the 4 bits serve to transmit an additional low quality audio signal, typically a speech (tally) signal.
- Parity Bit (P): This bit serves to detect transmission errors. It's digital value is set on the transmitting side the way to get an even number of ´1´s in the range of bits 4 -31. The receiver tests if an even number of ´1´ reached it's input. If not it messages a parity error.
- Validity (V): Primary this bit was intended to mark invalid or defective sample values. This bit when low indicates that the sample value is valid. Samples flagged invalid should not be taken into account for further processing e.g. for digital to analog conversion (D/A).

Common practice is a somewhat different disposal of the validity bit. A CD-player in most cases can perfectly correct a sample error but sometimes a correction is impossible. In this case the validity bit is set to ´1´.

Digital interfaces may also transport compressed audio data as MP3, Dolby Digital . It´s not possible to convert back such data directly to analog by digital to analog conversion . So also with compressed data the validity bit is set to ´1´.

- User Bit (U): With each subframe one userbit is transmitted (see paragraph "Important terms for further information).
- Channel Status (C): AES3 channel status comprises 24 byte or 24byte* 8 bit/byte = 192 bit of information. As each subframe includes just one channel status bit a block of 192 subframes is required until the full channel status information is transmitted. The start of a channel status block is flagged by a special SYNC pattern used to indicate the start of a subframe.

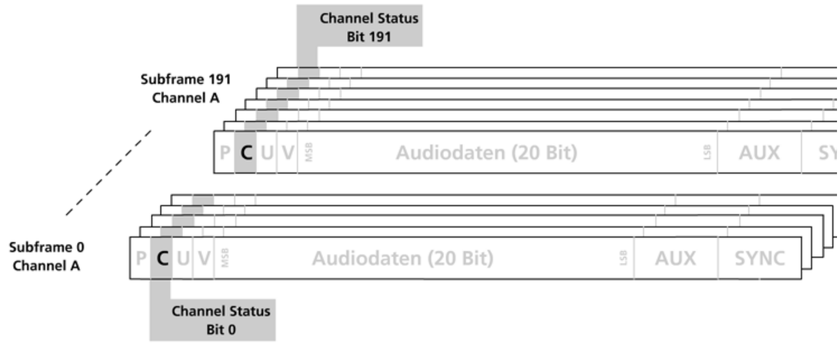


Figure:
Just one bit of 192 bits channel status information is transmitted per subframe. At a sample frequency of 48 kHz transmitting all this bits takes 4ms.

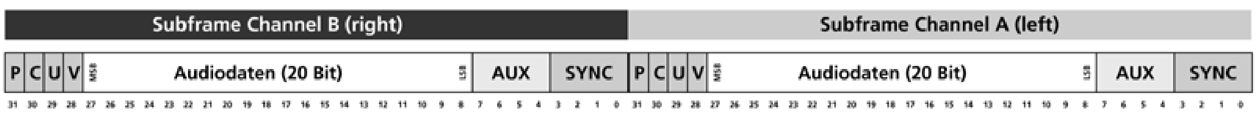
- Synchronisation (SYNC): Bi-Phase-Mark coding implies bit clock information, but subframe clock isn't part of this coding scheme. Now as only one (balanced or unbalanced) signal line is available also this clock must be extractable from the signal.

For this purpose each subframe starts with a SYNC header consisting of 4 bits. Three different types of headers are defined, named preambles in technical terminology, each indicating special positions in the serial data stream:

- X-Preamble: start subframe channel A
- Y-Preamble: start subframe channel B
- Z-Preamble: start subframe channel A and at the same time start of a 192 bit channel status block

Preamble bits are coded a way violating the bi-phase-mark coding rule. The receiver so is enabled to distinguish the preamble bits from ordinary coded bits.

Two subsequent subframes - we talked about subframes only until now - form a AES3-FRAME. As each subframe transports the signal of one channel the AES3-frame makes available two separate channels named A and B. So the AES3 framework can be charged with a stereo signal. Channel A takes the left part , channel B the right one of the stereo signal.



Both channels A and B can of course not only accept stereo signals but 2 totally independent mono signals of same sampling frequency. This causes the channel nomination to be different from L (left) and R (right). Likewise it is possible to combine the transmission capacity of both channels to transmit a single signal but now with advanced sampling rate (96kHz, 192kHz). This technology is called S/MUX mode, double wire mode or according the AES “Single Channel Double Frequency” mode. (for more details see the appendix “1C2F Mode” in the Digilyzer user manual)

Important to keep in mind: As each of the two frame channels is “embedded” in it’s appropriate subframe each disposes of it’s own channel status and user data!

Channel Status

Channel status information can be tailored for two different user groups: professionals and consumers. Applications within these two groups are very different and so are the channel status information associated. From there for each user group a subcode list (also called format) exists, defining how channel status bits have to be interpreted.

- Consumer Area: Channel status data serve to prevent copying of copyrighted audio material in excess of one copy.
- Professional Area: Here practice demands for reliable information about quality, type and source of the signal. Further information on synchronization is asked.

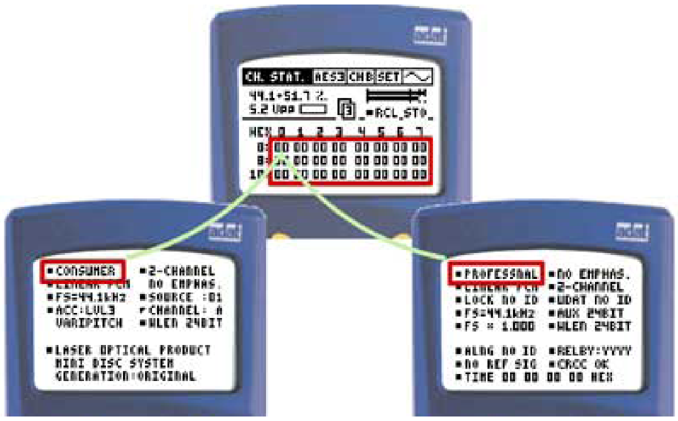
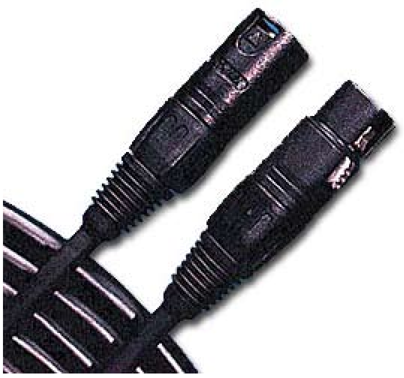


Figure:
The first bit within the block of channel status bits determines how further channel status information has to be interpreted, as consumer or as professional data. Devices as NTi Audio's Digital Audio Analyzer DL1 automatically display as plaintext the bit interpretation.

By agreement AES3 interfaces transmit channel status information in the professional format. Consumer format comes into operation with interfaces like S/PDIF and TOSLINK. Anyhow it is not made impossible to transmit channel status data via AES3 interfaces according to the consumer format. Consequence may be at the worst that a device will completely deny operation on the incoming audio data.

Cable



Standard balanced analog audio cable is widespread since a long time. One may come across very different types with impedances ranging from 40 Ohm to 110 Ohm, most of the older ones with impedances below 70 Ohm. So frequently one will encounter balanced cables not matching the 110 Ohm specification for digital audio cable.

Nevertheless it may be possible to transmit digital audio via up to 80m of those mismatched cables without cable equalization. Transmission up to 300m is possible in the case of good equalization. Greater distances can be bridged using cables complying with the 110 Ohm specification. Up to 240m can be bridged without and more than 750m with cable equalization. Connector pinning coincides between analog and digital cables: Pin1 = screen, PIN2 = signal (+), PIN3 = signal (-). .