Application Note AN-0011

AES Router Considerations

The AES/EBU digital audio standard is probably the most popular digital audio standard today. Most professional devices (VTRs etc) and consumer products (CD players, DAT decks, etc.) that feature digital audio I/O support the AES/EBU standard.

AES/EBU provides both "professional" and "consumer" modes. The big difference is in the format of the channel status bits. The professional mode bits include alphanumeric channel origin and destination data, time of day codes, sample number codes, word length, and other goodies. The consumer mode bits have much less information, but do include information on copy protection. Additionally, the standard provides for "user data", which is a bit stream containing user-defined (i.e., manufacturer-defined) data.

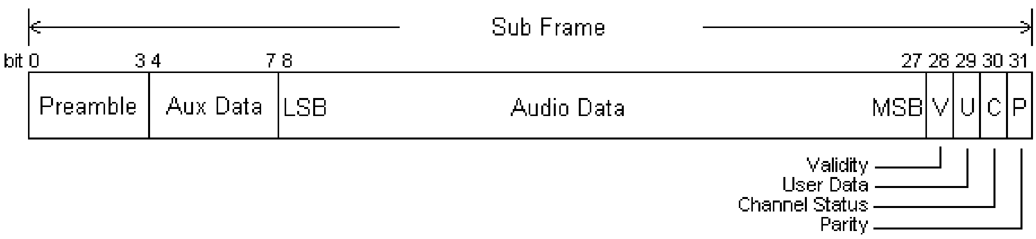
The physical connection media that are commonly used with AES/EBU:

Balanced (differential), using two wires and shield in three-core or multi-core cable.
Unbalanced (single-ended), using video coax cable with BNC connectors.

The Digital Audio AES Signal

The AES/EBU signal is a bit-serial communications protocol for transmitting digital audio data through a single transmission line. It provides two channels of audio data (up to 24 bits per sample), a method for communication control and status information ("channel status bits"), and some error detection capabilities. Clocking information (i.e., sample rate) is derived from the AES/EBU bit stream, and is thus controlled by the transmitter. The standard mandates use of 32 kHz, 44.1 kHz, or 48 kHz sample rates, but some interfaces can be made to work at other sample rates.

The data packet or sub-frame structure is shown below.



Audio samples can be up to 24 bits, with samples above 20 bits using the Aux Data bits. Therefore the Aux Data is only available for other uses with sampling systems using 20 bits or less. The serial bit rate varies depending on the sample rate that was used for the original analogue audio signal.

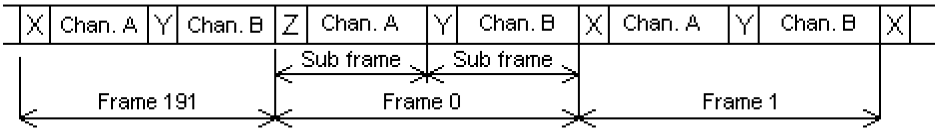
Broadcast (Nicam)	32Khz	2.0480 Mb/s
CD players	44.1KHz	2.8224 Mb/s
Professional equipment	48KHz	3.0720 Mb/s

The Validity bit (low level active) indicates that the audio sample is ready for conversion.

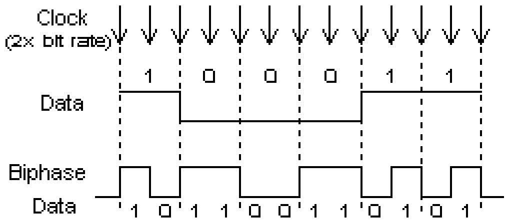
User bits and channel status are sent one bit per sample and must be processed after accumulation during a complete block transmission. User bit usage is undefined, and can be used for any purpose. Channel status bits give important information about audio data and the transmission line. Each audio channel has its own structure, and every 192 samples a channel status block can be recovered. See **Appendix A: Channel Status Block** at the end of this document for more details.

The Parity bit is set for even parity.

Two sub-frames joined together to form a frame. 192 frames are then combined into a block.



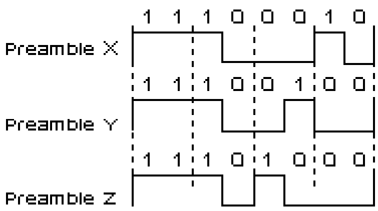
The serial stream uses bi-phase encoding, which means that the polarity of the signal is irrelevant to the decoder. In practical terms this means that the pair of wires used to carry the AES/EBU signal can be crossed over with no ill effects (with one exception in routing). The use of bi-phase was a deliberate design aim of the original AES specification as it prevents wiring errors (+ and - swapped) causing any audio problems.



The Z,Y,X referred to above are pre-amb le codes that identify the Channel A and channel B audio data.

Pre-amble	Biphase mark	Channel
X	11100010 or 00011101	Channel A
Y	11100100 or 00011011	Channel B
Z	11101000 or 00010111	Channel A & start of block

The pre-amble sequence is an illegal bi-phase signal, shown below, that is detected at the receiver.



As the signal is of a relatively high data rate, some high frequency elements will be lost in long cable runs. This requires the receiver to have some cable equalisation capability. Also, many receivers have a phase locked loop (PLL) to help recover the clock signal.

Audio Routing/switching Problems

The switching of audio signals in the analogue domain can lead to two types of switch click.

(1) The digital switch signals to the analogue crosspoint circuitry can crosstalk on to the analogue lines and produce a click. This is usually heard when switching between silence on two inputs with the amplifier volume turned up high. This effect is minimised in Quartz equipment by the use of router output silence switches, or by keeping the switch pulse very narrow so it is 'out of band' and has a very low energy content.

(2) The second type of click is caused by the transition from a minimum amplitude to maximum amplitude signal at the switch point. This generates an edge with a fast rise or fall time that causes the click. As most modern audio routers are designed to be wideband i.e. 100KHz or more, they are able to pass this transient.

However, many other pieces of studio equipment will have more typical bandwidths of 30Khz, and a sequence of such equipment will filter out the transient edge.

The analogue problems above have similar problems in the digital world.

(1) The switch point can corrupt the AES data stream sequence or an individual audio sample, which causes a click on the final digital-to-analogue converter (D-to-A) output. One corrupted sample will either be interpolated by the D-to-A converter or, at 48KHz would normally be 'out of band' and may be filtered by the D-to-A converter output filtering. However, the use of over sampling allows reductions in output filtering which tends to counter this argument. The error could also be removed by other AES processing equipment in the transmission chain, such as AES re-framers.

(2) The minimum to maximum transition problem exists in the AES world as well as the analogue world. This transition will be passed through the system as a full bandwidth transition (unless digitally filtered). The only standard filtering will be that in the D-to-A output stage.

Standard Routers (no Re-Synchronising option)

Un-locked Signals
These are signals of different sample rates, or signals from equipment that is not locked to a video or AES reference. As the timing of these signals will drift with respect to each other, it must be assumed that the router switch point will corrupt the pre-amble sequences. As a result of this, it may take downstream equipment a long time to re-establish lock.

All further discussions will assume that signals are locked to station reference and are operating at a sample rate of 48KHz.

Co-Timed Signals
When AES/EBU signals arrive at the inputs to a router they will be co-timed with respect to each other if the cable runs differ by less than approx 25m. When the router switches between these inputs, the router output signal will contain part of one AES signal that then changes at a random point (the switch point) to become the new input signal. Although the switch point is random, the pre-ambles will be aligned, and so only the audio data part of the AES sub-frame will be corrupted. This will produce a parity error 50% of the time. This corrupted audio sample will travel through the TV studio until a D-to-A converter or some other processing device (mixer, etc) cleans up the error. Many D-to-A converters detect the error and interpolate the last sample, i.e. fill in the gap, until the next valid sub-frame is received.

Mis-Timed Signals
When AES/EBU signals arrive at the inputs to a router they will be mis-timed with respect to each other if the cable runs differ by more than approx 25m. Mis-timing can also occur due to other equipments introducing processing delays. When the router switches between these inputs, the router output signal will contain part of one AES signal which then changes at a random point (the switch point) to become the new input signal. Depending on the size of the timing error, a pre-amble sequence may be corrupted, and this may cause downstream equipment loose lock to the AES signal for some time. This can appear as a complete loss of the AES signal for fractions of a second, which a D-to-A converter will find difficult to interpolate.

A standard or non-re-sync router would normally be used for pre-assignment of audio sources, or where the other studio equipment is not adversely affected by a corrupted audio sample. It must be used where the AES signals un-locked or of differing sample rates.

Special note on switching identical signals
It is possible for two identical AES data streams to be bi-phase inverted with respect to each other. When the AES signal is decoded this bi-phase inversion has no effect on the analogue audio or auxiliary data. Some equipment with multiple outputs may even give both inverted and non-inverted forms of the same signal, or not be clear about signal polarity.

This does not cause a problem until both the inverted and non-inverted bi-phase streams are switched through a router. What happens now is that the equipment after the router gets a corrupted bi-phase sequence, and therefore an illegal AES sub-frame, caused by the switch point and this can cause an audio click. To check if this is happening with two otherwise identical signals, take one of the router inputs and change the + and - wires and see if the audio click goes away. If the audio clicks remain then the signals are not co-timed, check on a dual channel scope at both 1us/div and 1ms/div for pulse edge alignment.

Re-Synchronising Routers

Re-Synchronising routers use special circuitry on each input to reprocess all signals such that their frame boundaries are co-timed with a reference input. Most modern routers achieve this by using an SRC (Sample Rate Converter) and this will also convert any input sampling rate to a common standard within the router.

A re-synchronising router would normally be used for 'clean' or 'live' switching of audio sources.

Appendix A: The AES/EBU Signal Channel Status Block

There are two Channel status blocks, one for channel A the other for channel B. Because this block is made of 192 frames, the channel status block is 192 bits long. These bits are arranged into a 24-byte block. Blocks can have two different formats: the professional format and the consumer format. The first bit of the block indicates the format (0=CONSUMER et 1=PROFESSIONAL).

Professional Standards

byte/bit	0	1	2	3	4	5	6	7
0	PRO=1	$\overline{\text{Audio}}$	Emphasis			$\overline{\text{Lock}}$	Fs	
1	Channel Mode				User Bit Management			
2	Aux Use			Word Length			Reserved	
3	Reserved							
4	Reference		Reserved					
5	Reserved							
6–9	Alphanumeric Channel Origin Data							
10–13	Alphanumeric Channel Destination Data							
14–17	Local Sample Address code (32bits binary)							
18–21	Time of day code (32 bits Binary)							
22	Reserved				Reliability flag			
23	Cyclic redundancy check character							

Channel Status Block Structure

Domestic Standards

byte/bit	0	1	2	3	4	5	6	7
0	PRO=0	$\overline{\text{Audio}}$	Copy	Emphasis			Mode	
1	Category Code							L
2	Source Num.				Channel Num.			
3	Fs.				Clock Acc..		Reserved	
4–23	Reserved							

Channel Status Block Structure (SP/DIF)

Appendix B: Glossary

AES: Audio Engineering Society.

ANSI: American National Standards Institute.

CCIR: Comite Consultatif International des Radio-communications. A UN regulatory body covering all forms of communication.. It both sets mandatory standards and makes recommendations.

EBU: European Broadcasting Union. This is a "club" of European broadcasters. It does not include representatives from manufacturers or post production. The EBU makes recommendations to the CCIR.

S/P-DIF: Sony/Philips Digital Interface Format. Refers to the AES/EBU signal operated in either consumer mode over unbalanced RCA plugs/cable or the professional signal operated over video coax cable.

SRC: Sample Rate Converter, a device that converts from one sample rate such as 44.1KHz to another rate such as 48KHz.