Digital Audio and Broadcast Studio to Transmitter Link Interfacing Issues and Solutions

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Abstract

With the conversion of many studio-to-transmitter links (STLS) from analog to digital format there are many factors to consider in the new digital implementation. Digital links have the advantage of better audio performance, the addition of auxiliary channels and the ability to send and receive data. Early attempts at providing digital service on a studio to transmitter link employed a wide band analog transmitter whereby a digitally compressed signal was applied at a lower data rate to provide two audio channels in AES3 digital format and some low speed data. Every effort was made to restrict the data rate in order to meet the RF mask demanded by the channel allocation for the service. Methods employed included eliminating certain non audio bits that appear in the AES3 word format. These included the user bit, Validity bit for determining audio/non audio etc. The AES clock rate for these services is typically 32 KHz sample rate which is below the more commonly used 44.1 KHz rate used by modern equipment. Newer links use the more standard data rate of 44.1 KHz but still conserve bandwidth by restricting certain non audio bits. These factors and others can create compatibility issues with downstream equipment. This paper discusses methods developed by Broadcast Devices, Inc. to deal with these issues and others.

While certain references to the AES standard will be made it is not the intention of this paper to be a tutorial on the standard per se but how to deal with certain difficulties when trying to interface equipment to STLs that do not completely comply with the AES3 standard format.

Issue #1 - Bits not passed by STLs

A. Validity Bit

In the AES3 format there are several bits that are non audio in nature and are used for various purposes by AES formatted equipment. A critical bit for some AES receivers is the validity bit which is referred to as the V bit for abbreviation. This bit is intended to inform the AES receiver of what type of data is being transmitted in the preceding word as to whether it is audio or non audio information. If this bit is set to 0 then the preceding sample word is audio data. If it is set to 1 then the sample word contains non audio data. Examples of non audio AES formatted data include Dolby AC3 and Dolby E formats, Motion Picture Experts Group II Layer 3 audio format is also an example of non audio data. While all three examples are typically intended to be used as audio they are compressed formats that a conventional AES3 receiver and D/A converter cannot readily decode. The intention of this bit is to allow an AES receiver to send a mute command to a digital to analog converter that is being fed by the receiver when non audio data is being sent. Otherwise the D/A converter will output noise which can sometimes be an undesired effect. Certain AES formatted audio streams sent through broadcast STLs will eliminate this bit and certain AES3 receivers expecting to see the presence of a 0 in the V bit place and will mute the succeeding D/A converter or stage.

B. User Bit

The user or U bit is available for user data. The data rate is defined as twice that of the sample frequency employed. For example if the sample frequency is 48 KHz then a possible user data rate of 96 KB/sec is available for use as non audio data. This bit can generally be dropped with no ill effect on the decoded audio.

C. Channel Status bit

The C bit is present in every AES3 sub frame and is used by AES receivers to configure themselves for the incoming audio data. This is done so that decoded audio will be the same format as transmitted. Examples of channel status data are sample frequency, word length, and audio mode (stereo, mono, or two independent channels). This bit must be present or the AES receiver will not know how to format the decoded audio.

The BDI solution to the problem of dropping the Validity bit

Broadcast Devices, Inc. manufactures three related digital audio switcher/distribution products intended for general purpose broadcast use including STL switching and interface to downstream equipment including broadcast processors and exciters. The ATB/GPM/DAB series switchers are designed to pass at user discretion AES3 streams that do not contain the Validity bit. This is a user definable parameter in that if strict compliance to the AES operating standard is desired the units can be configured to recognize the presence or absence of the Validity bit. If the bit is missing and it is desired to pass the digital audio stream and/or decode it the units can be set to do this. These series of products will insert the proper Validity bit upon transmission of the AES3 stream output so that devices downstream that require the presence of this bit will be able to handle the audio stream presented. At the same time the analog output of these products will be decoded audio.

Issue #2 - Sample Rate Conversion

Early digital studio to transmitter links employed sample frequencies of 32 KHz. This sample frequency is generally undesired for modern audio processors and broadcast exciters. 44.1 KHz is typically the standard employed by modern equipment. If an older link is being used some sort of sample rate conversion must be performed to up sample from 32 to 44.1 KHz. Of course it may also be desirous to down sample from a higher rate to a lower one and a sample rate converter is also suitable for this purpose.

Generally, sampling up from 32 to some higher rate like 44.1 or 48 KHz is straightforward. What is not trivial is to down sample from a higher rate such as 48 KHz to 32 KHz which sometimes is still desired. The sample rate converter in this case needs to make sure that no audio above the 32 KHz Nyquist bandwidth is presented to the 32 KHz path. 48 KHz sampled audio can theoretically have components up nearly 24 KHz analog bandwidth exceeding the 32 KHz Nyquist bandwidth analog bandwidth of 16 KHz by 8 KHz. If not filtered out these components will alias in the 32 KHz path causing artifacts from 8 to 16 KHz to be injected into the analog spectrum. A good sample rate converter manages this seamlessly by digitally filtering the incoming higher sample rate content to the lower Nyquist bandwidth of the lower sample frequency.

Issue #3 Synchronous Switching

The AES3 audio format consists of blocks, frames and sub frames. Each block consists of 192 consecutive frames; each frame consists of two 32 bit sub frames. Each sub frame consists of 32 bits numbered 0 to 31. Each sub frame consists of a 4 bit preamble, up to 24 audio/data bits and four other bits including the Validity bit, User bit, Channel Status bit and Parity bit. The V, U, C bits have already been discussed. The Parity bit is sent for error correction as with most data formats. The parity bit is sent to detect an odd number of errors received. When switching digital audio sources the point at which the switch is made must be at the beginning of a frame, sub frame or word. Broadcast Devices, Inc. synchronous switchers switch at the trailing edge of the clock cycle just at the beginning of a digital word. We take an additional step to insure that switching is click free in that we use cross fade algorithm to insure transitions are smooth particularly in the case of non contiguous audio feeds. Take the example of two feeds that are non contiguous that need to be switched between. It is quite possible to switch at an interval where the signal level is say -10 dBfs and the second source is -50 dBfs. This is a difference of 40 dB and could create a discontinuity even though the switch was done synchronously. The rapid cross fade helps to insure that a discontinuity of level between sources is not noticeable.

Summary

Management of audio resources in an all digital and hybrid digital/analog plant needs careful consideration. This paper was intended to concentrate on three principle areas of difficulty approached with broadcast studio to transmitter links. Management of certain bits passed in the AES3 format was discussed and solutions for these problems were offered. Synchronous audio switching was also discussed and the problems associated with some codecs that do not handle non synchronous switching well. Sample rate conversion was also discussed. By use of downstream products that manage the presence or absence of the V bit, sample rate conversion and synchronous switching an effective switching and distribution system can be implemented for STL to transmitter interface.

References:

AES information document for digital audio engineering – Engineering Guidelines for the Multi Channel Digital Audio Interface AES10 (MADI) -Audio Engineering Society 2005

Broadcast Engineering – Digital Audio Details II June 15th 2008