Title: Audio technology in the television broadcast studio and the realationship between analog and digital audio levels.

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Abstract

With the widespread adoption of digital infrastructures within Broadcast video and audio standards were implemented to ensure seamless equipment interoperability. For the video portion of the signal the transition was fairly straightforward. Strict standards exist for analog video therefore a predictable transition of the video signal into a digital world was possible. The audio transition has been far more complicated. While published standards exist for digital audio, no standards are in pace for analog audio in terms of nominal audio level, full scale digital audio level and audio headroom. What is used changes by market, geographic location and even application. Audio is still implicitly an analog I/O format. We cannot speak in (or listen to) digital bits, most audio originates in the analog domain and all audio has to be converted back to analog to hear it. With all the variables and no real standardization for analog audio there is an added layer of complexity which needs to be considered. Managing the A/D and D/A conversion of audio signals is critical to assure optimum audio quality within digital broadcast and Post Production infrastructures.

This white paper will help to understand the issues surrounding the A/D and D/A conversion of audio signals and explore how this relates to "standardized" digital audio methodology (and technology) used in broadcast today.

Analog Audio. Units, Levels and Connectivity

There is no published spec or standard practice for analog audio levels. The transition to digital audio requires the analog signal is converted into digital bits using an analog to digital converter. It's critical to correctly scale the analog audio input level into the analog to digital converter to use the maximum number of bits for the conversion to preserve quality and dynamic range. Likewise it's important to convert the signal back to an analog signal using digital to analog converters delivering analog signals at the correct level. Before we get into scaling audio correctly for conversion it's necessary to understand a little about analog audio levels and the units used to represent analog audio signal.

The unit of measure for analog audio levels in the Central European and Scandinavian market areas is dBu. "u" is referring to a voltage level of 0.775mv (rms) therefore this voltage level is referred to now as "0dBu".

Where did this originate? Well this value is derived from the voltage necessary to deliver 1mW of power into a 600 Ohm resistor (typical telephone headset speaker at normal volume level) and is referred to as "dBm" (the m standing for Milliwatt). Today 600 Ohm technology is somewhat obsolete and therefore technically speaking, not considered valid anymore.

Today, for Broadcast applications the typical analog audio studio level is + 6dBu (1.55v). On a peak meter this value is considered the 0dBu level.

> *Note. The use of dB with no further suffix means it's the representation of a voltage ratio according to the formula: p (in dB) = 20log U1/U2 (U1 being the voltage level as read on a voltmeter and U2 being the reference value, in this case +6dBu)*

In theory the maximum level in a completely analog world is +22dBu (but in practice this almost is never the case). Today the maximum is usually limited to +15dBu due to the scaling necessary into digital devices, but more on that later.

An analog audio input should come from a low resistance symmetrical source (typically XLR connector using twisted pair wire). The output impedance of the source should be ≤40Ω and the input impedance should be ≥10KΩ (typically 10…20KΩ) which permits adequate adjustment range. This means it's possible to connect several input stages to a single source if required (but in practice not a very good idea as a short circuit in one cable could impact all inputs).

> *The exception to the above are analog microphone sources whose output impedance is 50…250KΩ and the preamplifier input impedance is ≥ 1KΩ.*

Audio isolation transformers (or special electronics) are used to create floating connections between devices. While standard for all audio connections in the past it's not implicitly required today for smaller systems. But using floating connections (and therefore no grounding) can prevent losses in audio quality (signal to noise ratio) when a ground potential difference exists between systems. Therefore, for connectivity between distant systems (between facilities or buildings) it's good practice to use isolated inputs and outputs to prevent signal to noise problems.

At the consumer level there is another voltage unit used "dBV", here the reference level is not 0.775V but 1V. 1V = "0dBV". Typical input and output voltage levels for consumer devices (using asymmetrical unbalanced connections) is 0.5….2V rms. Output impedance is kept as low as possible; typically between 470Ω and 1KΩ. Input impedance is almost always ≥10kΩ (typically 47KΩ)

There is also another unit of measure typically associated with audio level meters. The prevailing metering system in Europe is PPM which uses an integration time of 10ms for the audio signal and then displays the result on a peak meter. In contrast the Anglo Saxon market area the "Volume Unit" or VU meter prevails. Here 0VU corresponds (most of the time) to 0dBu static level (using sine waves). Therefore the dynamic difference due to inertia (integration time 300ms) is just under 6dB. The VU response characteristic is considerably more inert than the PPM indication which, as we will see later, also leads to complications when setting analog audio levels for digital conversion.

As far as program material is concerned the voltage peak value "approximately" corresponds to European levels; 0dB = +6dBu = full recording level.

Sometimes the term "Dolby Level" us used in the film industry. This corresponds to a normally controlled volume (VU level) equivalent to 0dBu (set with sine wave signals). This equals a sound pressure level of 85dB (A) during level adjustments of acoustic systems (cinema). This acoustical volume approximately conforms to the standard level adjustment of loudspeakers in German broadcasting centers so to complete the chain: 0dB = full recording level = +6dBu = 92dB(A)volume.

Digital Audio

Digital audio did not originate in Television Broadcast and is in fact a fairly recent adaptation. A number of digital audio formats have been used over the past two decades. Here is brief review of the commonly known formats which you may come across in a production or broadcast environment.

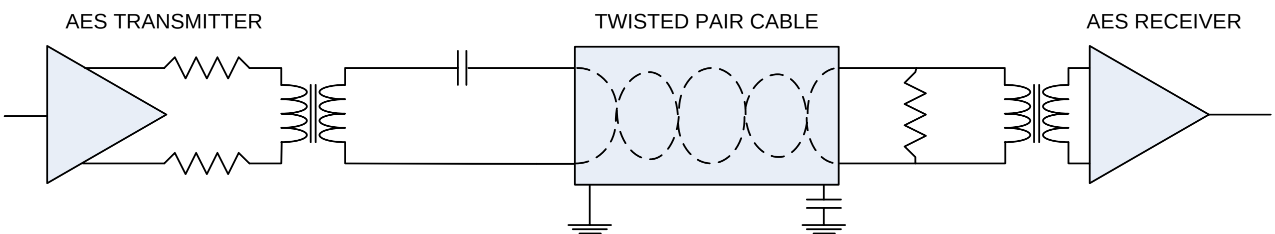
SDIF-2; which stands for Sony Digital Interface Format, also known as IEC 60958-3. This was developed by Sony and used primarily for multi-track audio CD mastering equipment. This was the first widely used digital audio format designed to carry stereo audio (left and right channel) audio over coaxial cable. Three cables are used for the interface one each for the left and right channel and a third for a reference signal or clock (referred to as Word Clock). For multi-track use twisted pair ribbon cable us used terminating in 50 pin SubD connections. Many audio only facilities still use multi-track SDIF-2 today.

MADI; which stands for Multi-Channel Audio Digital Interconnect, also known as AES10-2003 and is used for the interconnection of 56 channel digital audio between Multi-track recorders and digital audio consoles. This was developed as a standard by the AES (Audio Engineering Society).

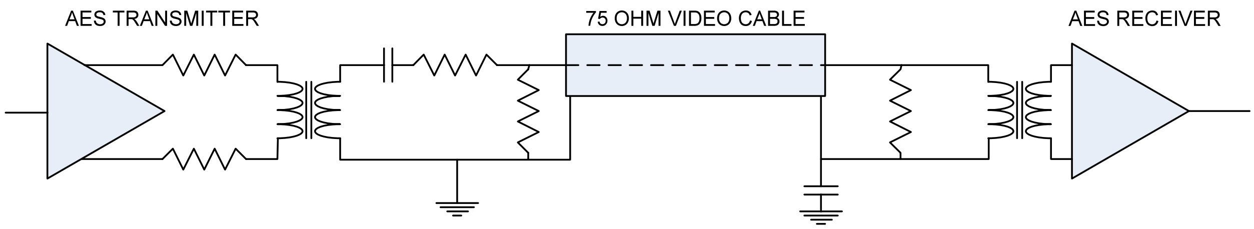
SPDIF; which stands for the Sony Philips Digital Interface. Developed by Sony and Philips this is a consumer format designed to move digital audio between various consumer products.

AES3; This is probably the most familiar and now widely used format for the transmission of digital audio signals between broadcast / professional audio visual equipment. The AES standard is jointly supported by the AES (Audio Engineering Society) and EBU (European Broadcast Union) and is often referred to as the joint AES/EBU standard. (Officially AES3-2003). The AES standard places both left and right audio channels in a single data stream (left data first and then the right) using a single physical connection. In reality these are two independent monaural signals so in today's multi-channel audio environment its better considered a 2 channel digital audio interface.

There are two ways to interface AES3 digital audio signals. One uses a symmetrical balanced interface much the same way analog audio was interconnected. This was initially adopted in the 1980s to make use of existing analog audio wiring and made the upgrade to a digital audio infrastructure easier. It soon became problematic because transmitting a high frequency signal (3Mbits/s @ 48KHz) using analog audio wiring hits its physical limitations very quickly, due to increased jitter, shock spots and undefined impedances. Cable lengths of 50m max were all that could be reliably achieved.



In the 1990s an asymmetrical version was added to the AES standard. Referred to as AES3-ID. This is far better suited to a video environment as is uses standard 75Ω connectors and video coaxial cable. Distances of 300m or more can be achieved with this interface. As it's a pure digital signal there is no performance degradation using this single ended un-balanced interconnection when compared to the symmetrical balanced version.



The Transition Between Analog and Digital Audio

For German speaking nations there is an explicit arrangement between broadcasting centers. Whenever there is conversion between a digital audio format (MADI, ADAT, AES3, embedded audio or whatever) to an analog audio signal then the following is valid:

"+ 6dBu in the analog domain (= 0dB on analog and digital VU meters) **is -9dBFS as measured in the digital domain** (with a static sine wave of 997KHz)**"**

> *dBFS = dB Full Scale. This is the analog level at which the A/D Converter reaches its maximum number range. Sometimes also referred to as "FSD" = Full Scale Digital. Therefore only negative numbers are possible because the maximum value is 0dBFS.*

This is a voltage value of +15dBu at 0dBFS (simply adding 9dB). The value +9dBFS should be avoided because it provides for no overhead and hence erroneous measurements are possible. It is better to take measurements at -9dBFS.

When adopted this fixed correlation is enough for all conversions.

This correlation allows for 9dB of headroom in a normally controlled audio signal. As previously mentioned the PPM indicators as used in Europe have an integration time of 10ms. The "real" peaks are 6dB above the indicated level. A 3dB reserve is factored so it's always possible to transmit a distortion free signal when the audio level is correct. Providing more headroom is not necessary and would only narrow the available dynamic range. During the early adoption of digital audio some broadcasters adopted 6dB of headroom because at that time only 16bit audio systems were being used.

Today almost all digital audio devices are matched as stated above. *With the exception of CD players: these have the correlation +6dBu = -6dBFS because the source material audio level is already optimized.*

Within the EBU market area sometimes +18dBu = 0dBFS which is typically used for program exchange. In fact only -3dBFS is ever reached, so adjustment into the German speaking markets is certainly possible.

In Anglo Saxon markets there is slightly higher headroom due to the more inert characteristics of VU meters used in these areas. +18dBu or +20dBu (SMPTE) = 0dBFS is typical.

There is a bigger extreme in the music industry. The level is often driven up to the "red range" on the VU meters during recordings to achieve maximum saturation of the magnetic tape used for recording. In some extreme cases the levels might even be: +24dBu = 0dBFS, but this only applies to audio systems providing a full 24 bit word width which is used with higher sampling rates (96/192KHz or DSD)

In Conclusion

There are a few basic rules to consider when making the transition to and from digital audio.

1. Know the nominal input level of your analog signal. If it's not specified then measure it. It's important to scale the analog signal as accurately as possible into the A/D converter. If the level is too high the A/D converter will overload and you will get distortion, if it's to low you will sacrifice dynamic range, quality and introduce noise.

   General Guidelines for different market areas and the "nominal" audio levels used.

Market Area	Nominal 0dB FS level
Germany (and most of Europe)	+ 15dBu
Anglo Saxon Markets (UK, USA ect)	+ 18dBu and + 20 dBu
Music Industry	Up to + 24dBu

2. Agree on an internal studio standard for audio levels used with A/D and D/A conversion, and stick to it throughout.

3. Do not transfer digital audio signals between equipment using D/A and A/D converters. The conversion process is where level mismatches and mistakes occur most often. Once the signal has be digitized, keep it digital and at the nominal level wherever possible.

   If conversion is unavoidable make sure the FSD operating levels of the A/D and D/A converters match.

LYNX Technik produces a range of high quality audio conversion products designed for professional and broadcast applications. These products feature user selectable FSD (Full Scale Digital) presets for the most common analog audio levels you will encounter in any most markets. More information on LYNX Technik Products can be found at **www.lynx-technik.com**

References and Standard References

AES3 (acc. IEC 60958-4)
AES3-id (Coaxial version of AES3)
SMPTE 272M (Embedded Audio)
IRT Requirement Specifications 3.5 and 3/1-8/2
Nordic Standard N10
RBT Directive Digital Audio (2001)