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Audio Advances Rapidly Since 1983

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From BTSC to Surround Sound, audio changed with the times

In honor of TV Technology's 20th anniversary, we will take a look back over the last 20 years of television audio. This is a perfect stretch of time because it seems that almost all the major audio advances we rely on today have happened since 1983. That being said, it feels like it has been a nonlinear increase with the last few years seeing an almost unbearably fast release of technologies, supporting products and accompanying confusion.

STEREO

Arguably the largest jump forward in television audio was the introduction of BTSC stereo broadcast technology to the NTSC system in 1984. To be fair, the work was done a bit earlier than this, but the standard was finished and published in 1984 thereby allowing stereo broadcasts to begin.

As we have discussed before, it was actually called the Multichannel Television Sound (MTS) standard as it defined not only the main stereo channels but also the monaural SAP and PRO channels. The 315-page standard created by the Electronic Industries Association (EIA) was quickly adopted as the standard for "multichannel" television sound by the FCC, but adoption by broadcasters was much slower. Thus far, approximately 725 stations broadcast in stereo, and of those, about 170 are noncommercial. This total is still less than half of the approximately 1,600 FCC licensed television stations currently on the air. Due to the impending switch to digital television, it seems unlikely that these numbers will increase much, and with the age of some of the equipment the count may actually decrease.

DIGITAL AUDIO

The advent of digital video recording began in 1988 with the release of the D-1 component VTR developed by Sony and BTS. For the first time, four independently editable channels of 16-bit, 48kHz audio were available. This obviously allowed a great deal of flexibility in post production, but more importantly it instantly cured inter-channel level and phase problems. Sony followed the D -- 1 with the composite D -- 2, and the portable Digital Betacam formats, both of which offered the same four channels of audio but with the resolution increased to 20 -- bits -- better than CD and standard DAT audio quality.

Panasonic was close behind with the release of its first digital VTR, the D-3, which also had four audio channels, albeit with 16-bit resolution. The company's follow-on machine, the D-5, matched the audio performance of D -- 2 and Digital Betacam with four channels of 20 -- bit, 48kHz audio.

The next machines to be developed were to support high definition video. One of the first was the Sony HDD1000, which records uncompressed HD video and eight channels of audio. This large, heavy, one-inch tape machine greatly helped during the early days of HDTV testing and demonstrations in the U.S. and abroad. Unfortunately, it was not inexpensive to own or maintain, and it required some skill to operate. Although some legacy material still requires these machines, they are becoming increasingly difficult to find.

BTS and Toshiba developed the D -- 6 in the late 1990s that is capable of 10 channels of 24 -- bit, 48kHz audio (12 channels in 50Hz mode). However, it took until the 21st century before more than four channels of audio were included in a popular, compact tape format. Eight audio channels are included as an added feature of the Panasonic HD D-5 format but currently only in 24P mode. The competing Sony HDCAM format has the same video features as HD D-5 but contains only four channels of audio. The very latest offerings from Sony are on the right path and finally do offer eight channels of 24-bit, 48kHz audio.

During this time, nonlinear audio editing systems from companies like Sonic Solutions and Pro Tools began to emerge. These systems dramatically improved the quality, speed and flexibility of audio in post production and were ready for multichannel sound almost from the start. Today there are even inexpensive systems, such as Cool Edit Pro, that allow for simple nonlinear operations and even support some very basic multichannel features. These packages are finding - scale nonlinear systems cannot be justified.

SURROUND SOUND

Because the BTSC system provides for stereo audio, by default it is capable of carrying matrixed Surround sound. As we have discussed ad nauseum in previous columns, Dolby Surround and other matrix-type systems encode multiple channels of audio into a stereo compatible format called LtRt (Left total Right total). In mono it sounds like mono (with any surround channel information canceling); in stereo it sounds like stereo, and with a surround decoder it will reproduce the multichannel audio. Beware of the stereo "enhancers" found on some television sets because they can overdo it with surround encoded audio and may negatively affect dialogue intelligibility.

Surround is also a part of the ATSC audio system, better known as Dolby Digital (AC-3), approved for use by the FCC in the mid-1990s. This flexible system can carry from one to 5.1 discrete channels of audio; and it contains a whole host of features to make the audio work with many different pieces of equipment in many different listening environments. The ATSC system is also capable of providing full-bandwidth channels for SAP or Descriptive Visual services, which can be stereo or even full 5.1 channels. This system has helped to spur on the added audio channels on VTRs and systems such as Dolby E to support multichannel audio from post production all the way to final broadcast.

LOUDNESS AND AUDIO PROCESSING

The loud commercial problem existed prior to 1983. As new audio processing systems from Orban, CRL and Modulation Sciences hit the market to support the new BTSC system, these problems began to diminish. They later developed dynamics processors that either incorporated the classic CBS loudness algorithm or used multiband processing, which proved to be very effective at controlling the two channel loudness problems.

One of the features of the Dolby Digital (AC-3) system is the ability to pass full dynamic range audio with low distortion and noise, but this has the potential side effect of bringing back the loudness problems. How do you support the benefit of having a wide dynamic range channel while not having large shifts in loudness? New loudness measurement systems such as the Dolby LM100 can help with program-to-program loudness variations by indicating the proper metadata values that need to be set in the Dolby Digital (AC-3) encoder. Another solution is the Linear Acoustic OCTiMAX 5.1, which is a multichannel dynamic range processor designed specifically to work with the ATSC system and Dolby Digital (AC-3).

A/V SYNC

At least one thing has become worse since 1983. The issue of audio-to-video synchronization (a.k.a. lip-sync) has grown into a gigantic mess. It seems that the more digital video and audio technologies that are introduced into a system, the more likely lip-sync errors will be. There is now a whole new market for both measurement and correction equipment. Tektronix developed a system called the AVDC100 that via a video watermark allows for automatic lip-sync correction. The company also has a system originally developed by Intera that allows lip-sync to be checked after transmission.

As we have discussed, the golden rule is that it is always best to correct the sync problems as soon as they are created. Sometimes delay is unavoidable, such as in certain digital video effects devices. The trick is that while the video is being processed, and therefore is being delayed, the audio must be delayed as well. To that end, Pixel Instruments has developed a system that allows for silent adjustment of audio delay that can actually track the changing video delay. I have tested the system and to my ears it does a remarkably good job masking delay adjustments and could prove to be very useful.

To summarize, the last 20 years have brought us from analog to digital, slowly from VTRs with four audio channels to those with eight or more, and from mono to 5.1 Surround Sound. We have also slipped in sync and gotten variably louder, but have figured out how to measure and correct both problems. The pace of audio developments seems to have reached breakneck speeds in the last few years, and I for one cannot wait to see (and of course hear) what will happen in the future.

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