DTS:
BRIEF HISTORY AND TECHNICAL OVERVIEW

This article takes a look at how DTS got started in the cinema and in the home, with a brief overview of how the systems work.

The cinema system came first. One of the first people to examine the feasibility of digital sound for motion pictures was Terry Beard, founder and president of Nuoptix, a company that makes high-performance analog optical sound recording equipment. His study was done for Todd-AO in 1972. In the mid-1980s, Terry joined with Jim Ketcham, then at Lorimar, to develop a system.

Their work was guided by two convictions, both at odds with the preponderance of industry opinion: one, that digital sound was for all theatres, not just showcase houses, and would become a standard release format; and two, that delivering the audio data on media designed for data would be inherently easier, less expensive, more reliable, more flexible, and higher in quality than trying to place the data directly on the film print.

Early experiments used DAT tapes synchronized with the projector, and the Nuoptix 2:1 compression system. But the tapes could not quickly re-sync past large edits. Jim proposed CD or CD-ROM as potential carriers, but these had inadequate capacity and transfer rates. "CD quality" was the target; thus 16-bit words at 44.1 kHz for 5.1 channels were the specs.

The concept became practical through the use of more compact audio compression, namely a system called apt-X100 devised by Stephen Smyth and his colleagues at Audio Processing Technology. This system yields 4:1 data reduction, making it possible to fit 100 minutes of multichannel audio on each CD-ROM. The data rate is 882 kbit/s. (CD-ROM was chosen over audio CD because the audio is handled as files and sections of files; and error correction is superior.) This meant that two discs would be sufficient even for movies longer than three hours. To synchronize the discs to the projector, time code is placed on the film.

By 1990, patents were filed, and a demonstration was given to SMPTE members. In early 1992, the system was demonstrated for Steven Spielberg, using a hard disk for playback, which resulted in Spielberg considering the technology for his upcoming picture, *Jurassic Park*.

Jurassic was a Universal release, and the studio needed to be convinced that the system would be practical and trouble-free in real operating theatres. Two "smaller" Universal pictures were used for the tests. On the first, unbeknownst to the producer, time code was added to the prints to verify that the code would not interfere with playback of the adjacent analog optical soundtrack. There was interference—on misaligned projectors. Once properly aligned, they worked fine, presumably with better analog sound too.

On the second picture, both time code and discs were prepared for unannounced shows in Westwood and Universal City (Los Angeles). These went smoothly. Universal provided funding for the roll-out, and Universal and Spielberg became part owners with Terry, Jim and others in Digital Theater Systems, as the new company was called. In the space of four months, between the company's founding on February 1, 1993 and the opening of Jurassic Park on June 11,* 876 systems were installed. As of this writing there are approximately 19,000 in use.

An industry unaccustomed (since Vitaphone) to sound on disc worried that the CD-ROMs would not show up with the prints. Thus a key invention in the system is the reel-size caddy that holds two discs and is packed in the standard can with the film. The caddy was conceived and sketched by Terry on the fly during a product presentation meeting in Las Vegas.

Originally, both stereo ("DTS-S") and six-track ("DTS-6") versions were planned. DTS-S carried the same L_T/R_T stereo matrix as the SVA track, but unlimited and in digital form, along with an LFE channel. It was believed that a certain number of theatres might want digital sound, but would be unwilling to install a six-track B-chain. Early industry reaction proved otherwise, and the small number of two-track versions produced were quickly upgraded to six-track.

In the DTS-6 system, the three screen and two surround channels are discrete. The LFE channel occupies the lower 80 Hz of the left surround and right surround channels. This was done to increase disc playing time, and works because the surround speakers in theatres are not designed for the 20 - 80 Hz range. Also, there had been a successful precedent for surround/LFE sharing in 70mm magnetic soundtrack practice.

* The original release date had been June 22. The advance in date was announced in May —quite a surprise to the crew scrambling to get the systems built and installed!

For “special venues,” meaning theme parks, motion simulators, etc., fully discrete six-track and eight-track DTS systems are available, and by adding players an unlimited number of tracks is possible.

In the standard 35mm system, the time code lies on the film between the analog soundtrack and the picture. This is an area largely protected from wear, and was covered by pre-existing printers without modification. The analog optical track and the time code are recorded at the same time.

A time code reader is mounted on the projector, which is a simple device compared to the scanners in data-on-film systems. The time code consists of a single row of “dashes” of varying length which are, individually, easily seen by the naked eye and thus highly robust. For 24 fps, 4-perf, the shortest dash is 5 mils by 12.5 mils. Wear sufficient to spoil DTS time code would obliterate the picture.

The DTS time code is a proprietary 30 Hz code which includes ID specifying the title and reel, to ensure that a disc can never be played with a mismatched reel. (SMPTE time code can be used for special installations.) For a given film speed, variations of ±10% can be tracked, and running a 24 fps film at 25 fps is no problem.

A further advantage of having only time code on the film is that the code can readily be adapted to any placement, pulldown, frame rate, or film gauge. DTS is thus the only one of the three major digital systems used on 70mm (including 5-perf, Iwerks, IMAX, and Showscan) and 16mm film.

In 35mm, for special setups, the code can be placed outside the perforations, allowing the full perf-to-perf width to be used for picture. In 70mm, the time code always lies outside the perforations, again allowing perf-to-perf picture, as cannot be done with magnetic since two of the six mag tracks are inside the perfs.

Having the audio data on discs allows using various soundtrack versions (languages, ratings) to be played with the same print, when the print is moved to another location; or even simultaneously in the same location (with a second player). And the time code can trigger in-theatre effects, subtitles, narration for the visually impaired, etc.

Edits, splices, missing footage, and changeovers are handled smoothly. As time code is read, the corresponding audio data is loaded off the disc into a delay line and is played out when the corresponding picture reaches the gate. After a splice or changeover, the disc seeks the new audio while the delay line continues to output its contents. If time code is interrupted, the system is programmed to “freewheel” and output audio for up to four seconds until time code resumes. Ultimately, the system will default to analog in case of a larger problem.

On a 70mm print, there is no default analog track. However, because of the large area available, the time code is oversize and all but indestructible.

Following the success of the theatrical DTS system, the company looked at the issue of premium sound for home theatres. Laserdisc was the premier home video medium of the time, delivering two channels of 16-bit, 44.1 kHz. Terry asked the question whether a 5.1-channel system could occupy the laserdisc’s digital tracks, each channel being superior to CD in quality.

Stephen Smyth and his colleagues, having formed AlgoRhythmic Technology, developed a very-high-performance codec ideally suited to this task, called Coherent Acoustics. When it was demonstrated successfully, AlgoRhythmic and DTS formed DTS Technology (since absorbed into the larger DTS) to bring the system to market.

Among the first people to audition Coherent Acoustics were recording engineer Tom Jung, editor Gary Reber, and music producer Brad Miller. Through testing with musical material, it became apparent to all that the system should be used for multichannel music, and so DTS offered its new codec on LD and CD, movies and music, in parallel.

Typically, a compression system reduces an audio signal into a smaller space or bandwidth, with the hope that quality will not deteriorate beyond a level deemed acceptable. The aim of Coherent Acoustics was fundamentally different: to take *all* of an existing bandwidth and use more efficient coding to *improve* quality. Consider CD and laserdisc, which share the same data rate. In the space where, with linear PCM, only two channels of 16-bit audio can reside, Coherent Acoustics can pack 5.1 channels of 24-bit audio, affording dramatically better sound. In this system, the six channels are fully discrete and the five main channels are all full-range. The LFE channel is band-limited.

Using almost all (1.235 Mbit/s) of the CD/LD bandwidth (1.411 Mbit/s), the compression ratio ranges from 2.9:1 (16-bit) to 4.3:1 (24-bit).

On DVD-Video, the data rate is slightly higher (1.509 Mbit/s), since 48 kHz sampling is used. In addition, a rate of 754 kbit/s is available for projects where space is limited by program length or other factors.

It is often assumed that the DTS systems use perceptual coding, but they do not. First, of course, “perceptual coding” must be defined, since, in a sense, any audio code is perceptual. Linear PCM, for example, reduces the real world of sound to limited bit depth and sampling rate based on assumptions about what the ear will find sufficient. There are other, and more efficient, ways to represent sound.

There are two categories of data which might be targeted by a bit-rate-reduction system. First, there is objectively redundant data, which can be removed with no loss. As a conceptual example, the number “0057” can be represented as “57” in a smaller space, with no loss of information. For computer files, PKZIP is an example of such a codec.

The second category is “perceptually irrelevant” data, valid to measuring instruments, but which cannot be heard. For example, in the presence of sound at a given frequency and level, other sounds lower in level and slightly lower or higher in frequency are masked by the original sound; that is, they can be heard less well, or not at all. Likewise, sounds lower in level and slightly after or before (!) the louder sound are masked.

The term “perceptual coder” usually means that the coder calculates and exploits masking thresholds. The idea is that the masked sounds need not be coded, or might be coded with fewer bits as a function of their audibility. Another perceptual technique involves combining channels at higher frequencies, which the ear has difficulty localizing. Envelope information may be maintained for the separate channels to allow the channel amplitudes to be somewhat reconstructed, but phase information is sacrificed.

Such strategies can work well, but are unnecessary at lower compression ratios, where coders can concentrate on removing objective redundancy in the signal. In this sense, the DTS systems have more in common with lossless coders than with competing perceptual coders.

In the DTS theatrical system, apt-X100 achieves its 4:1 reduction by using subband coding with linear prediction and adaptive quantization, which identify and remove redundancy and pack the data efficiently.

The first step, the division of the spectrum into subbands, itself offers an opportunity for coding efficiency, since the energy in the bands will be unequal at any instant and coding resolution can be adjusted accordingly by band.

Within each band, a prediction is made based on the recent history of the signal, and the prediction is subtracted from the actual signal. The difference is quantized. If the prediction is accurate, the difference will be smaller than the actual signal and so can be coded more compactly.

Adaptive quantization refers to the fact that the step size of the quantizer is adjusted dynamically to match the signal level, which maintains a constant signal to quantization-noise ratio.

The other algorithm used by DTS, Coherent Acoustics, is highly scalable and can operate anywhere in the range 32 kbit/s to 4 Mbit/s. It can handle up to 24 bits, and up to eight channels. (The decoder population implements 5.1 channels.) Again contrary to popular assumption, Coherent Acoustics is not a perceptual coder at the data rates used on CD, LD, or DVD. When operating at lower bit rates, the perceptual techniques are enabled.

Coherent Acoustics uses more subbands (32) than apt-X100 (4). Again, linear prediction and adaptive quantization (ADPCM) are used. The effectiveness of prediction is signal-dependent, and in each subband, if the prediction process does not offer a coding gain, it is disabled. At low bit rates, masking thresholds are calculated and bits are allocated in accordance with the psychoacoustic model. Finally, variable-length coding is used: the code words generated by the ADPCM are mapped to another set of words such that the most frequently occurring are given the most compact codes, and so on.

There are many other details about and differences between the algorithms (forward vs. backward adaption, etc.), and for interested readers, technical papers are available from DTS.

Looking to the future, DTS has demonstrated Coherent Acoustics in 24-bit, 96 kHz, for several years, and can deliver this in a form backward-compatible with existing decoders. Likewise, DTS has experimented with additional channels. Channel proliferation and coding methods are also lively topics in planning for digital cinema (and future film-based cinema, for that matter). So the present systems are only steps along the way in the evolution of multichannel sound for cinema and for the home. The challenge remains to think outside the conventional assumptions, as Terry Beard did when confronting the original goal of digital sound for movies.

—LORR KRAMER

V8