Instructional Evaluation

Class members filled in questionnaire. Talked about Student Learning Outcomes a bit.

Quiz

First of quizzes to come.

Digital Audio

ATSC (Advanced Television Systems Committee) selected Dolby Digital (AC-3) as a standard for DTV. US cable television industry has also adopted Dolby Digital for DTV applications. Most television facilities are not equipped to produce 5.1 channel sound. For this reason, many DTV programs use two-channel sound. The 5.1 channel sound used primarily for theatrical films on pay-per-view channels and at theaters.

Dolby Digital (AC-3) is the chosen audio compression scheme for the digital television transmission system. Allows up to 5.1 channels of discrete digital audio to be compressed into a single 384 kb/sec data stream. Why .1? Subwoofer bandwidth; at most 100 or 200 hertz the highest frequency. 48 kHz sampling per channel. 16-bit min to 24-bit max amplitude resolution. Locked to 27 MHz system clock. System flexible – allows AC-3 surround sound signal to be decoded as mono, stereo, and Pro Logic, as well as the full 5.1 channels. Pro Logic is a device that created a 4-channel effect. The word stereo means three-dimensional. Quadraphonic (2 channels in front, 2 in rear) joined positive on left rear and left front, positive on right rear and right front, hooked up other legs through a variable resistor between front and back to change cancellation.

AC-3 Audio (Dolby Digital)

Chosen since provides decoding flexibility and very high compression ratio with excellent quality retention. Wide dynamic range. At home, users can adjust dialog level, dynamic range. Seen dialog level adjustment, not seen much of dynamic range adjustment. Dialog level adjustment handy for hearing impaired.

AC-3 is a perceptual coding scheme that breaks the original signals into spectral components and in simple terms realigns the audio data to maximize the use of the gaps and imperceptible information found in the original audio recording spectrum. This spectral realignment allows for significant bit rate reduction, while retaining sufficient data to allow very high quality playback once decoded. AC-3 never designed to be anything but a distribution technique. It is not meant to be decoded and then re-encoded.

Almost impossible to re-encode to AC-3 properly due to spectral realignment that took place during first encoding. Double encoding AC-3 at best results in poor quality audio and at worst no discernible audio signal at all.

In a receiver, separate the AC-3 bitstream and the MPEG-2 bitstream, then send them to their respective decoders.

AC-3

AC-3 serial coded audio bit stream made up sequence of synchronization frames. Each synchronization frame contains 6 coded audio blocks (AB), each of which represent 256 new audio samples. A synchronization information (SI) header at the beginning of each frame contains information needed to acquire and maintain synchronization. A bit stream information (BSI) header follows SI, and contains parameters describing coded audio service. Coded audio blocks may be followed by auxiliary data (Aux) field. End of each frame is error check field includes CRC word for error detection. Additional CRC word in SI header, optional.

Synchronization frame contains SI (with first CRC), BSI, AB0 through AB5, Aux, CRC (second CRC).

Why 6 audio blocks? This is how 5.1 is carried. How to get stereo out of these 6 things? Associate left and right with separate audio blocks, then you can add them together if needed.

Audio Compression

AC-3 uses sophisticated coupling strategies among channels. The strategies work on one or more critical bands at a time. May for example allow phase of signal in one or more critical bands to be common to all channels while preserving only the relative amplitude. Spectral envelope and quantized mantissas for 6 audio blocks (1536 samples, 32 msec) are formatted into an AC-3 frame. Each block may contain information for up to 6 channels. An AC-3 bit stream is a sequence of AC-3 frames.

Sub-band Masking

System relies on masking of tones on either side of high-power tone. If you can't hear it, don't code it. **Sub-band masking** takes 3 Mbit/sec and reduces it to about 200 kbit/sec.

Two types of audio coding exists: MPEG-2 and Dolby AC-3. For audio, US uses AC-3, DVB (Digital Video Broadcast – the European standard) uses MPEG-2. DVB is the European equivalent of ATSC.

Sub-band Coding

Splits audio signal into four frequency bands that are each subsequently processed separately. By altering the short-term coding resolution in each band according to the energy of the signal in the band, spectral redundancies in the full-band audio signal are exploited by allowing sub-bands which contain high energy signals to be coded at a higher resolution than the sub-bands that contain low energy, higher frequency signals.

Linear Prediction

Another form of compression used in AC-3. Further removes spectral redundancies from sub-band signals. Reduced by comparing level of incoming sub-band sample with a prediction that the algorithm had made for that sample. The prediction (whether greater or smaller than the actual sample) creates a difference or error signal which can then be re-quantized. Success of linear predictive coding is highly dependent on the short-term periodicity of most signals and can be optimally matched to the non-linear auditory response of the human ear.

Recall that sampling at 48 kHz, while highest hearing is at 20 kHz. Voice frequency range is 300 to 3000 Hz. Thus, get at least 16 samples per cycle at 3000 Hz. This is apparently what gets compressed.

Adaptive Quantization

Exploits relatively slowly time-varying energy fluctuations which audio exhibits by continually adjusting the size of the quantizer step to match the signal level. Exploits phenomenon of temporal masking in which sharp impulse-like audio signals known to mask out other audio signals which appear before and after the impulse signal (for a substantial period of time either way).

Quantization efficiency for signals immediately following impulsive sounds can be reduced without audible effect. End result of algorithms: compressed audio signals significantly more robust in that can withstand numerous encode/decode cycles, while being able to retain the integrity of the initial audio signal. As compressed audio signal passes through broadcast facility, may be necessary to regenerate the original audio signals for manipulation or special processing. Some common operations cannot be readily accomplished in the compressed domain, such as audio-overs. Robust multi-pass operation critical to ensure continued integrity of signal as it is repeatedly decoded and re-encoded.

Splicing Issues

Ideally, splicing should occur at audio frame boundaries. If splicing performed mid-frame, audio decoder will mute. Outage on order of 2 frames (64 msec for ATSC AC-3), decoder will fade in/out at mute boundary.

Dolby E

Format not heard of much; maybe got abandoned. Interesting format.

CBTE Test Review

The test you have to study for after you get your CBT. This is one level up. Video category.

Gamma correction is used to compensate for the non-linear transfer curve of the kinescope (old word for picture tube / CRT).

Diagonal lines on a resolution chart will appear segmented if a camera has lost interlace. What happens if a camera loses horizontal scanning? The beam stops in one place on the gold mesh on the pickup tube and burns out the gold which evaporates and damages the tube.

The talent needs to do a scene setting piece to intro the rest of the video tape on a story. The best shot for technical reasons would be to have him stand near the entranceway to a building with slightly more light on him than on the background.

To measure a differential phase shift the engineer could use a modulated stairstep signal generator and a vectorscope. The stairstep is modulated with color subcarrier (3.579545 MHz). Could be a ramp or a stairstep, since get rid of the luminance anyway since using a vectorscope. Measure the change in phase as compared with burst. Checking to see if differential phase changes as amplitude changes. Read in degrees. Determines accuracy of an entire system when injecting this signal early on, can inject at different points to see what is causing the problem.

In a color monitor in which the individual colors can be turned off, can adjust hue with only blue color on.

A video processing amplifier can be used for correcting video levels, correcting setup, and adjusting color saturation and hue.

In 1 inch type C helical-scan VTRs, when recording stereo audio, the left channel is recorded on audio 1 and the right channel is recorded on audio 2. This is recording audio like an audio tape recorder.

Camera geometry can be set up by using linearity chart.

The chip chart is used to obtain chroma-free black, white, and in-between grays.

The three primary colors used in NTSC system are red, green, and blue.

The nominal pulse width for NTSC horizontal sync is 4.7 microseconds.

Chrominance subcarrier refers to the carrier which is modulated by the chrominance information.

In the NTSC color encoder, burst phase is set at 180 degrees.

In NTSC color television transmission, I and Q signals are chrominance signals.

If the target of a pickup tube is switched to an underscanned condition, the effect is similar to going to a narrow angle lens.

The test pattern used to make scanning linearity adjustments for a NTSC color camera is the registration chart.

Blanking level refers to the level of signal during blanking interval, except the interval during the scanning sync pulse and the chroma subcarrier sync pulse.

The NTSC color burst is placed in the back porch. Before there was color, there was only front porch and back porch. When adding color to the black and white NTSC signal, would add the burst to the back porch. The breezeway was created as the result of putting the burst within the back porch.

A NTSC sync generator provides a NTSC reference to equipment for proper video signal timing.

Lunch