An HD Non-Linear Edit Suite

by Eric Wenocur

Since the launch of the Discovery HD Theater network in June of 2002, Discovery Communications has begun to edit, mix and create High Definition programming at their Technical Center in Bethesda, MD. One of the first steps in preparing to process HD shows was the design of their first HD non-linear edit suite. The chosen platform was the Avid|DS HD. Aside from it's own particular interface and operational quirks, and the massive amount of storage required, the DS is quite similar to other Avid products so this article will focus on how it was integrated into an edit system which accomodates both High Def and Standard Def editing, including an assortment of complex audio requirements.

Discovery's existing facility was built to handle fairly straightforward post-production needs: Composite and component analog video, component digital video (SDI), and analog audio. Adding the HD edit suite (and an assortment of equipment to support up/down-conversions, dubs and quality control evaluation) created several complications. These included the need to add patchbays for HD signals (with related patches for the down-converted versions); addition of tri-level sync and the need to patch reference signals into VTRs (and the Teranex Xantus HD/SD converter); and the need to handle audio in a mix of stereo, surround, analog, AES and Dolby-E flavors.

Interestingly, the video aspects of these additions were relatively simple. A Tri-Level Sync board was added to one of the facility's two Trilogy Mentor Plus master sync generators, with outputs at various frame rates available on the HD patchbays. Similarly, HD and SD ins and outs from VTRs pass through these patchbays, as do signals related to the QC suites. Naturally, this HD "island" also requires waveform and picture monitoring that can handle mixed HD formats. Discovery's house HD rate is 1080/59.94i, but production is also being done in 1080/24p, as well as PAL variations.

General Suite Applications

What was most complex about integrating the HD edit suite turned out to be audio, and the need for great flexibility in use of the room. It was decided that this edit suite should be able to function in three distinct modes:

1) Edit any programs in Standard Def, with normal stereo soundtrack. Output to Digital Betacam.

2) Edit new programs in High Def (with audio elements in place, but mixing done in audio post). Output video to HDCam master, audio tracks transferred via OMF or other means.

3) Modify existing HD programs, including 5.1 surround soundtracks (audio mix unchanged). Output to HDCam master with 5.1 soundtrack encoded in Dolby-E on two channels, plus stereo soundtrack on two channels.

Incoming material would be delivered on HDCam (Discovery house format), D5-HD, Digital Betacam (SD programs), DA-88 (audio stems for SD programs or 5.1 surround mixes), DAT, and other miscellaneous formats.

Audio would be particularly quirky due to the requirements of bringing as many as eight tracks into the Avid|DS, from all the above formats, in a plant with a primarily analog audio infrastructure. In this case, it was decided to handle audio for SD production as embedded AES (which was already working in ten other non-linear rooms) and to handle audio from DA-88 or DAT as analog into the suite mixer. Tracks from HDCam or D5 tapes could come into the mixer as discrete AES (wired through the HD patchbays specifically for this suite).

Video Strategy and Signal Path

The video portion of this system is comparable to most non-linears using serial component digital (SDI) signals. The Avid|DS has separate inputs and outputs for SD (SMPTE 259) and HD (SMPTE 292) signals, and a single input for Reference. Patches for these signals, and to the monitors in the suite, were provided on the HD patchbays previously mentioned.

A video distribution amp supplies reference to all edit system devices. The normalled input to this DA is NTSC blackburst, which is accepted by the Avid (and many other devices) for editing in 59.94 frame rate formats, but Tri-Level Sync in 1080/23.98p and 24p flavors can be patched as well.

Video monitoring in the edit suite is via a Videotek VTM-420SD/HD "rasterizing scope" (which displays waveform and metering on a standard SVGA computer display) and a Sony BVM color monitor. The design intent was to give the editor a monitor selection (and corresponding patch) to see an incoming VTR, and another to see the Avid. Although the Sony monitor has separate input cards for SD and HD signals, since the Videotek has only two discrete, non-looping inputs (which auto-detect SD or HD) there could be only two patches to feed both devices with either HD or SD signals. The simple approach appeared to be: Take the feed from a patch point, pass (active loop) through the BVM HD card, then the SD card, then terminate at the VTM. Unfortunately this proved to be unfeasible because the Sony HD card would not pass SD signals, and vice-versa. The problem was solved with some HD DAs from Aja Video which will pass anything from 270 Mb on up. So, each of the two monitoring patches is split with a DA, and fed to the VTM and both types of BVM inputs. The BVM is programmed with two "channels" for SD and two for HD--one of each format for each of the two patch points.

Operationally, the editor must patch reference to the suite (or leave the normalled black) and patch the desired VTR into both the Avid and the "A" monitor patch. The Avid output is normalled to the "B" monitor patch. The VTM is programmed with several presets which associate a video input (A or B) with the ncessary audio to be metered (see below).

Audio Strategy and Signal Path

The audio portion of this system is where things really get fun! Even though the room would not be creating "finished" surround audio, it was felt that the editor should always be able to listen to a mix in context--meaning stereo or surround--while editing the picture. It was also necessary to give the editor some control over audio going into (digitizing) and out of the Avid, for building tracks or working on an SD project. This combination proved to be exceedingly difficult to accomodate with a modest-sized digital mixer, and eventually required the addition of a "surround monitoring processor" to help manage all the options.

First, a very quick primer on surround audio. For the purposes of Discovery's HD Theater, we are concerned only with surround in the Dolby Digital 5.1 format (also known as AC-3 when encoded for consumer delivery). Dolby Digital specifies channels for left, right, center, stereo surrounds and LFE (low-frequency effects, the so-called ".1" channel). It also specifies a library of metadata information that can be carried with the audio stream and used to control functions in the viewer's home decoder. Typically, the audio mix is created in a conventional audio-post room, with the metadata added during this process, and the final result is dubbed onto a pair of VTR channels using Dolby-E encoding (Dolby's format for "transport" of up to eight AES channels on a single AES pair). At the transmission end, the Dolby-E tracks are decoded back to discrete 5.1, and then re-encoded into AC-3 for the consumer. The metadata is passed along in the AC-3 stream and is used by the decoder at home. (Extensive information on all these topics is available on the Dolby website.)

In the Avid|DS suite, AES audio passes between the patchbays and the Avid via a Yamaha 01V mixer (many of which were already used in the facility). The 01V, with an optional card, can handle eight channels of AES input and output, plus sixteen analog inputs, and has, effectively, ten internal buses for routing. This arrangement was sufficient to handle four channels of AES from patch to mixer to Avid (digitize), and four channels from Avid to mixer to patch (output), while leaving a pair of buses for monitoring stereo--which was fine for editing in SD. However, it would be impossible to ALSO monitor six channels of surround audio, since only two buses were available. In addition, there would be situations where it was desired to pass eight channels of audio from the Avid into the Dolby-E encoder and, again, this would preclude the ability to monitor in surround.

Compromise was needed. It was decided to limit some mixer functionality in certain cases, and to add a Martinsound MultiMax monitor processor. The MultiMax provides a variety of "wide" (8-channel) and stereo inputs which can be selected and routed to various multi-channel speaker systems. Although it does not handle Dolby Digital metadata, it does provide the ability to audition a 5.1 mix "downmixed" to stereo or mono. It also provides speaker mute and solos, volume control and dim, thus becoming the clearinghouse for all listening audio in the room.

It was also important to keep an already complicated system as operator-friendly as possible. One way to help was to avoid changing the function of mixer faders, whenever possible. Therefore, the first four mixer channels were designated as "From the VTR" (either AES or analog), the next four became "From the Avid" (AES), and the last eight were fixed as analog returns from the Avid used solely to feed the MultiMax. By carefully arranging which buses fed where (and writing seven mixer scene presets) it was possible to handle all required functions with a minimum of variation or need for the editor to fuss with the mixer routing.

The editor can recall mixer presets which reconfigure various internal settings. These include digitizing from AES or analog sources, outputting 4-channels to tape (while monitoring returns from the VTR), outputting 8-channels to tape or Dolby-E, and variations of these which provide either stereo or surround feeds from the mixer (depending on how many buses are available). The MultiMax has input selections for hearing six mixer bus outputs (surround monitoring), the mixer Main stereo bus (stereo monitoring), the Dolby-E decoder (for confidence while laying Dolby-E to tape), the mixer analog input patches (to verify a DA-88 patched in for digitizing), and the six "emulator" outputs of a Dolby DP-570 Audio Tool (in case metadata needs to be programmed in the edit suite).

As for audio metering, presets were written into the VTM unit to select appropriate audio formats when associated video was selected. The VTM can display the embedded audio in an SDI signal, so that was used as much as possible when digitizing from a VTR. In other cases, the unit's eight discrete AES inputs are used to display mixer outputs (for discrete 5.1) or the output of the Dolby-E decoder. The VTM also has a handy "Cinesound" display which shows surround audio in a visual layout that suggests the speaker locations.

Does This Really Work?

Once the room was put into operation it took several days to try various types of projects and debug the workflow process. Feedback from the editors suggested a few improvements to the original design, but it basically works the way it was intended. No question, though, it's complicated.

Despite having written a fairly detailed user guide, it goes without saying that this suite requires an editor who is technically savvy about both video and audio, and able to think clearly through the digitize/edit/output process. Every job requires making determinations about frame rates, sync, VTR selection, incoming audio formats, monitoring and output formats. The editor must have a sense of what he or she *should* be seeing and hearing at each stage, in order to be confident that they have selected the correct presets and controls, particularly with regard to the mixer and MultiMax.

In another scenario it might be valuable to have a larger digital mixer in the system. However, this would probably not eliminate the need for a device like the MultiMax for monitoring; the beauty of which is that it clearly delineates the available listening sources and makes it easy to verify that tracks have the correct content (such as by auditioning individual speakers). Unfortunately, there is no way to make working in HD much easier overall. And it is only going to get more complicated as HD becomes more mainstream, which will increasingly require engineers to find novel ways of solving peculiar problems!