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Managing a Real World Dolby E Broadcast Workflow

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Abstract

BBC HD's surround sound programming brings the viewer a more immersive audio experience than television has previously offered, while audio metadata allows greater control over audio reproduction in the home. However the technology required to produce and deliver 6 channels of audio with associated metadata gives broadcasters a new set of challenges. The Dolby E[1] data stream format allows transport of the audio and metadata within the distribution infrastructure, but its use requires careful consideration of system timing, audio-video synchronisation, metadata control and monitoring. Additionally, the increasing popularity of LCD and plasma displays in the home can cause viewers problems with audio-video synchronisation due to processing delays found in such displays. The BBC set out to examine and resolve these problems, and the results are presented in this paper as a useful knowledge base for anyone handling Dolby E and/or Dolby Digital to deliver multichannel audio for television

[1] "Dolby E" and "Dolby Digital" are registered trademarks of Dolby Laboratories Inc.

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Authorisation of the Head of Research is required for publication.

1 Introduction

High Definition television (HD) is becoming increasingly popular around the world, and with the higher resolution pictures often comes multi-channel audio, or surround-sound. The BBC launched a trial HD channel in May 2006, which became a full service in December 2007. BBC HD is currently available on satellite (via Sky and Freesat) and cable (via Virgin Media), and in common with other HD services in the UK (such as Sky's own HD channels), uses Dolby Digital as the emission codec[2] (the codec used to broadcast audio to the home). This applies both to stereo programmes and where available, 5.1 surround sound. In developing the audio infrastructure for the broadcast and production environments, there are two primary considerations which make the challenge greater than that of stereo:

- 5.1 surround sound requires 6 discrete audio channels.
- The use of Dolby Digital for broadcast means metadata is required in addition to audio.

In order to transport both the larger number of audio channels and the associated metadata, technologies such as Linear Acoustics' StreamStacker and Dolby E are available to enable the distribution of multi-channel audio without the need for 6 channels or more of standard AES3 audio infrastructure. Dolby E is used by the BBC for delivering audio through the production and distribution signal chain. Some months after beginning broadcast however, the technology was still proving temperamental in use; while generally the channel was broadcast without major problems, small troubles such as audible artefacts at junctions and audio-video synchronisation issues were cropping up, and occasionally larger problems caused more high-profile on-air difficulties. A review of the use of Dolby E throughout BBC HD workflows was therefore initiated with the aim of examining the problem areas and resolving the issues which were being encountered.

Additionally, viewers who have invested in high quality HD receiving equipment may have problems with audio-video synchronisation in the home due to the processing delays introduced by modern LCD and plasma displays. Set-top boxes often provide controls to manage this, but viewers have no way to calibrate the settings easily, and the BBC set out to help them by broadcasting a synchronisation test signal.

This white paper examines the process of reviewing the multichannel audio infrastructure for BBC HD, many of the lessons learned from this review, and the associated work on providing an audio-video synchronisation (lip-sync) test signal for viewers.

1.1 Broadcast Infrastructure in the BBC

Much of the difficulty in managing the implementation of a new technology across widespread areas of the broadcast infrastructure lies in co-ordinating the efforts of disparate technical departments. In the modern BBC, as with many other broadcasters, the problem is compounded by the fact that large swathes of the distribution infrastructure are handled by external organisations. The path of a live high definition outside broadcast (OB) is summarised in figure 1, while the corresponding companies involved are illustrated by figure 2.

[2] MPEG stereo is used for some additional services such as Audio Description, however main programme audio can generally be assumed to always be Dolby Digital.

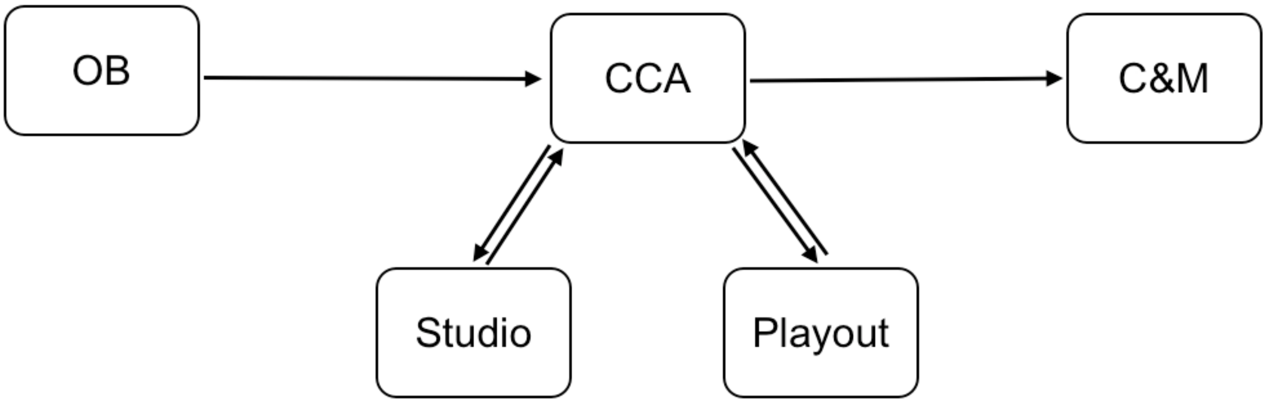

Figure 1 - An overview of the signal path followed by a BBC HD OB

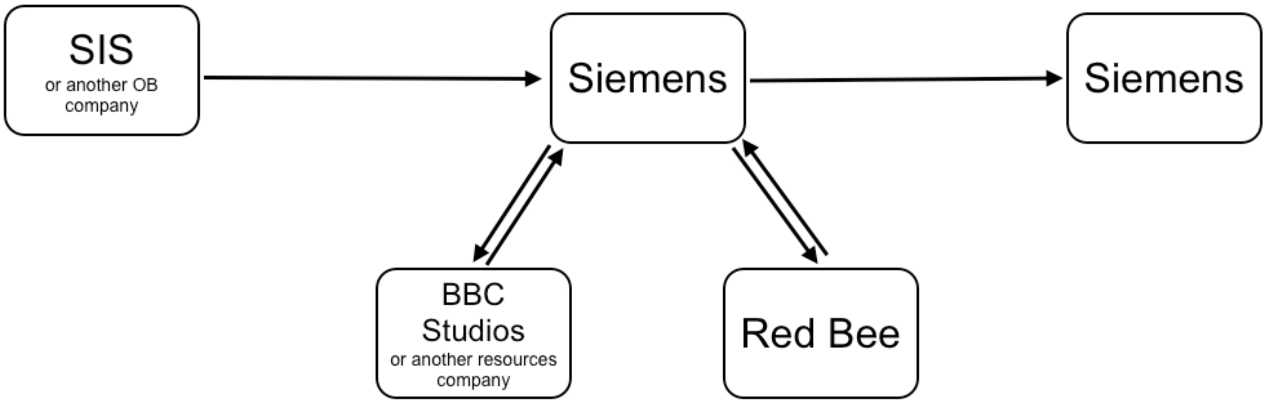

Figure 2 - The companies who run the various parts of the signal path

The major areas can be summarised as follows:

- **Outside Broadcast** - run by one of many OB companies, one common example being SIS Live, formerly BBC Outside Broadcasts. Additionally a separate links provider may be involved.
- **CCA** - the Central Communications Area is in BBC Television Centre (TVC) and acts as the central routing point of the distribution. (What many broadcasters call a Master Control Room.) Operated by Siemens.
- **Studio** - in the case of an outside insert into a studio programme, a studio in BBC Television Centre or elsewhere may be involved.
- **Playout** - Red Bee Media run the BBC’s playout area, where schedules are managed, junctions are mixed, continuity announcements are added and pre-recorded programmes are played out.
- **Coding & Multiplex (C&M)** - the coding and multiplex operation for digital TV is also run by Siemens and located in TVC.

Only through working with Siemens, Red Bee, BBC Resources and outside broadcast companies could a coherent strategy be formed for the correct handling of Dolby E to avoid problems. Additionally, Dolby themselves provided advice and support, as did Tektronix, whose monitoring equipment is used by the BBC along with devices from other manufacturers.

1.2 The BBC's Dolby E Review

In the autumn of 2008, a review of the use of Dolby E within the BBC HD production and distribution systems was initiated. There were 2 principal components of this review, along with other smaller investigations. The primary elements were:

- An examination of areas which have caused problems in the past, such as system timing and metadata loss.
- An investigation into audio-video synchronisation both in the distribution chain and the home.

The signal path was mapped from start to finish, and a custom notation developed to make examining synchronisation issues easier. The details of this notation are explained in section 4.2. A series of tests was conducted to ensure correct audio-video sync, and problems such as audible artefacts at programme junctions investigated in conjunction with the technology partners involved and - in some cases - the equipment manufacturers.

The complexities of managing a Dolby E based infrastructure led to some recommendations being made on the use and configuration of particular equipment, many of which are also summarised in this white paper. Finally, some training requirements were identified to aid operational staff in troubleshooting problems. Whilst these requirements are not discussed in detail here, the references section provides links to some helpful material for those requiring additional information on multichannel audio or more detailed information on Dolby E and its use.

2 An Introduction to Dolby E

Dolby E is a data stream designed by Dolby Laboratories which is used to carry up to 8 channels of digital audio within a standard stereo channel (AES3), as well as transporting metadata which describes the audio and affects its reproduction at the consumer's decoder. It is a professional data stream designed for use within production and broadcast infrastructure, not for use as an emission codec or by consumers. It employs comparatively light data rate reduction, ensuring that multiple encode-decode cycles are possible. Dolby suggests that at least 10 such cycles will result in no audible degradation.

Because Dolby E is carried within an AES3 stream, the bit depth of that stream is important. Using the top 16-bits of an AES3 stream, Dolby E can carry 6 channels, whereas in 20-bits, 8 channels are available. The *bit depth* setting of Dolby E equipment refers to the AES bit depth and hence the number of channels available in the Dolby E stream, not the bit depth of the Dolby E encoded audio.

Dolby E is often used in conjunction with Dolby Digital, a consumer data stream also carrying multiple channels of audio (up to 6) and associated metadata which used as the emission codec by many broadcasters. The *consumer metadata* used in a Dolby Digital stream is carried by the Dolby E stream, along with some additional *professional metadata*. Therefore the consumer metadata can be transferred from a Dolby E decoder to a Dolby Digital encoder, allowing metadata continuity from studio to home.

Because it is an encoded data stream and not linear audio, Dolby E frames may not be modified in any way such as gain adjustment, sample rate conversion or equalisation. Such modifications will alter the data stream and cause corrupt Dolby E frames. A corrupt frame will likely cause a mute for the duration of the frame, or be interpreted as PCM audio, causing a loud audible "splat" in the output.

2.1 Timing Basics

Dolby E divides audio into frames at a rate aligned with the associated video. In the the BBC's installation for example, the rate is 25Hz. A Dolby E frame may not be split; an incomplete frame will not reproduce the intended audio, but instead create an invalid Dolby E frame, with undesirable effects as described above. For this reason, a Dolby E frame is slightly shorter than a video frame, allowing a *guard band* to be used. This means that the Dolby E frame starts slightly after the video frame and ends slightly before it, leaving some unused time at the start and end of the frame. The switching point used by mixers and routers falls in this guard band, allowing switching between video sources without the corruption of the associated Dolby E.

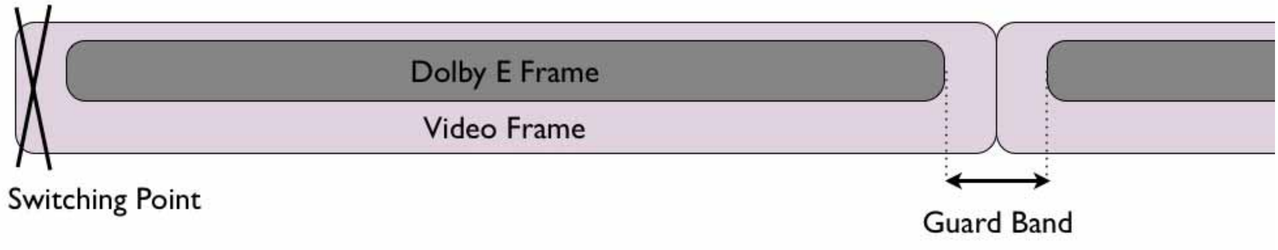

Figure 3 - Simplified Dolby E timing (not to scale)

The *Dolby frame location* or *line position* is the point at which the Dolby frame starts with relation to the video frame. It is often measured in video lines, though milliseconds are also used. There is some flexibility in acceptable Dolby line position. See section 3.1 for further detail.

Dolby E encode and decode cycles incur a fixed delay of 1 video frame each (e.g. 40ms in a 25Hz system). This means that delays have to be carefully managed, as significant audio-video synchronisation problems would otherwise occur, however the 'round number' nature of the processing delay makes the effects easier to manage.

2.2 Frame Structure

Dolby E frames are transmitted within an AES3 carrier, whether on a discrete connection or (most often) embedded in HD-SDI or SDI. The Dolby E data is encapsulated in AES3 using the SMPTE 337M format for non-audio data in AES3. A Dolby E frame therefore consists of the standard 337M headers, followed by the Dolby E data. Dolby E decoders (and other compliant equipment, where necessary) uses the 337M header as an indication that the data is Dolby E. The four 337M header words, Pa, Pb, Pc and Pd contain information about the bit depth, data type and length of the Dolby E frame. The data type code for Dolby E is 28.

2.3 Standards

In order to gain a full understanding of Dolby E from a systems perspective down to a bit stream level, one must be aware of the many layers involved. Dolby E itself is a way of packaging audio and metadata into a data stream, and it then sits as SMPTE 337M formatted data packets within an AES3 carrier. This is frequently embedded in serial digital video (SDI), making for a complex structure of data formats to understand. The standards involved are illustrated below for those wishing to gain a more detailed understanding than is presented in this document. Those blocks which are most relevant to the BBC implementation discussed here are highlighted.

<table>
<tr><td colspan="2">SDI
SMPTE 259M</td><td colspan="2">HD-SDI
SMPTE 292M</td><td colspan="2">MPEG-2
ISO/IEC 13818-1</td></tr>
<tr><td colspan="2">Digital Audio in SDI
SMPTE 272M</td><td colspan="2">Digital Audio in HD-SDI
SMPTE 299M</td><td colspan="2">Digital Audio in MPEG Transport Stream
SMPTE 302M</td></tr>
<tr><td colspan="6">AES3 Digital Audio</td></tr>
<tr><td colspan="6">Non-PCM Audio Data in AES3
SMPTE 337M & SMPTE 338M</td></tr>
<tr><td>AC-3</td><td>MPEG Audio</td><td>Dolby E</td><td>Captioning</td><td>User Data</td><td>Etc…</td></tr>
</table>

Table 1 - Standards involved in an embedded Dolby E installation

3 System Timing

Because Dolby E is co-timed to video, a video reference (analogue black and burst) is required by each piece of Dolby equipment. The precise timing of the Dolby E signal relative to the associated video is critical.

3.1 Dolby E Line Position

Dolby Laboratories provide the following guidelines on the line position of Dolby E signals. The table expresses the positions in TV lines for a 1080/I/25 system as used by the BBC. (i.e. 1080 lines at 25 interlaced frames per second, or 50 fields per second - Dolby documentation refers to this as 1080i50.) For further guidance in other video standards, contact Dolby Laboratories.

	TV Lines	µS
Earliest Position	**13**	**450**
Ideal Position (±80µs)	**21**	**730**
Latest Position	**53**	**1860**

Table 2 - Dolby Line Position for 1080/I/25

Audible splats or mutes are heard after Dolby E decoding suggest malformed or broken Dolby E packets are reaching the decoder. One possible cause of this is incorrect Dolby line position, for example by causing video cuts to occur during the Dolby frame rather than the guard band. In such situations, the line position can be monitored on various Dolby E aware monitoring tools as described in 6.3.

It is important to ensure that the measurements made are fully understood. Measuring in µs is relatively safe, however when measuring in video lines, it is imperative to know which TV standard is being referenced. The Dolby's DM100 bitstream analyser for example provides readings only in standard definition lines, so when compared to the lines of a 1080 line signal the readings will be misleading.

During the investigation of such symptoms around the BBC's playout centre, the line position of Dolby E encoded material leaving a playout server was found to be suspect, so testing such equipment can be useful. In the worst case where an individual device is particularly unreliable, a Dolby-realigning frame synchroniser may used to correct the problem, as described in 3.2.

Another useful measurement provided by some measuring devices is the *Pa spacing*. This refers to the time, in audio samples, between the *Pa* words of the SMPTE337M headers. In other words, it is the measurement of the spacing between two adjacent Dolby E frames. Because of the co-timed nature of Dolby E, this should be the same as one video frame; 1920 samples in a 25Hz PAL system with a 48KHz audio sampling rate.

3.2 Video Synchronisers

Given that timing is so critical, special attention must be paid to any device which alters the timing. Video frame synchronisers adjust the timing of a video signal to a local reference, and may drop or repeat video frames in doing so. However if the Dolby E guard band is not aligned correctly with the SDI frame boundaries, this dropping or repeating of video frames will leave the embedded Dolby frames broken. Where PCM is used, an small number of missing or repeated samples will likely not cause a noticeable effect, however because a Dolby E frame is a single continuous element of encoded data rather than a series of individual samples, a broken frame is rendered useless. In the worst case the frame will not be recognised by downstream equipment as Dolby E and will be reproduced as PCM samples, creating an audible splat. At best, a mute will occur for the duration of the frame.

Figure 4 illustrates a frame being dropped from a correctly-timed video signal with embedded Dolby E and an incorrectly-timed signal. In the latter case, the red blocks show partial Dolby E frames which are rendered useless.

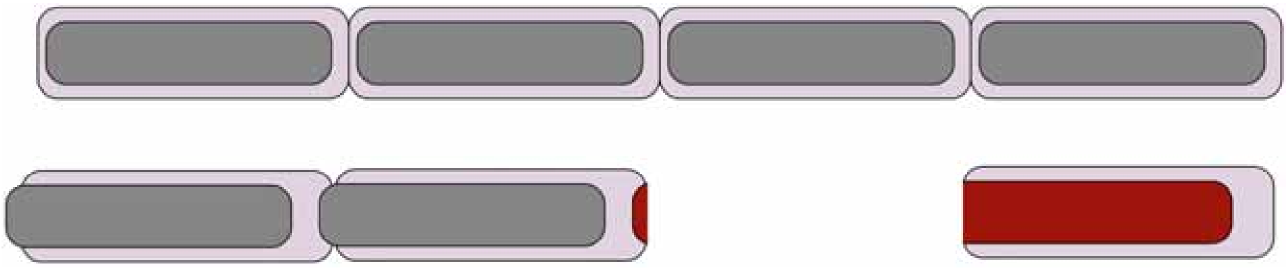

Figure 4 - The effects of dropping a frame with correctly- and incorrectly-timed Dolby E

If the timing of the Dolby E stream relative to the video is correct, no degradation will occur through most frame synchronisers, though it is always wise to check with manufacturers or perform your own tests. However, audio and video propagation and processing delays through links and processing devices may not be exactly the same, even with embedded audio. So if there is a possibility that a synchroniser will receive misaligned Dolby E at its input - for example after a long contribution link - a more robust option is the use of a frame synchroniser with *Dolby Re-alignment*. This process ensures that the Dolby E frame is correctly aligned with the video frame (i.e. the line position is correct and the guard band lies over the SDI frame boundary) before synchronising the video to the reference. This in turn ensures that Dolby frames are not corrupted and no problems occur as a result.

Cost considerations may be a factor in choosing a frame synchroniser for a Dolby E system. For example, Axon's HFS10 does not include a Dolby align process, and so whilst it will transparently pass correctly aligned Dolby E, it will potentially cause problems where the Dolby E is misaligned. The HES20 on the other hand includes a Dolby align process, making it more robust, but more costly. In the BBC's case, the short connecting links between some areas may be seen to be at a low risk of causing Dolby alignment problems, and as such less costly synchronisers are the best option. On outside source inputs however, where incoming signals may have undergone long journeys and complex processing, Dolby alignment is a necessary process. In migrating to a new Coding and Multiplexing infrastructure for digital television, a discussion was undertaken about the appropriate hardware to use, and given the short distance from the preceding Dolby E encoder, non-realigning synchronisers will likely be used. In a problem situation, there are other means to ensure a good Dolby E signal arrives into the coding centre, so the risk is low.

3.3 Contribution Links

The timing of Dolby E contribution links and other long-distance links requires extra care. In general, the approach is reasonably simple:

- At the outside broadcast or remote site, all equipment should be locked to a common reference, usually generated by a local sync pulse generator (SPG).
- At the receiving end, it is most likely that a local reference will be used for all equipment, produced by a different SPG. Therefore the re-synchronisation of the incoming feed must be undertaken with great care.

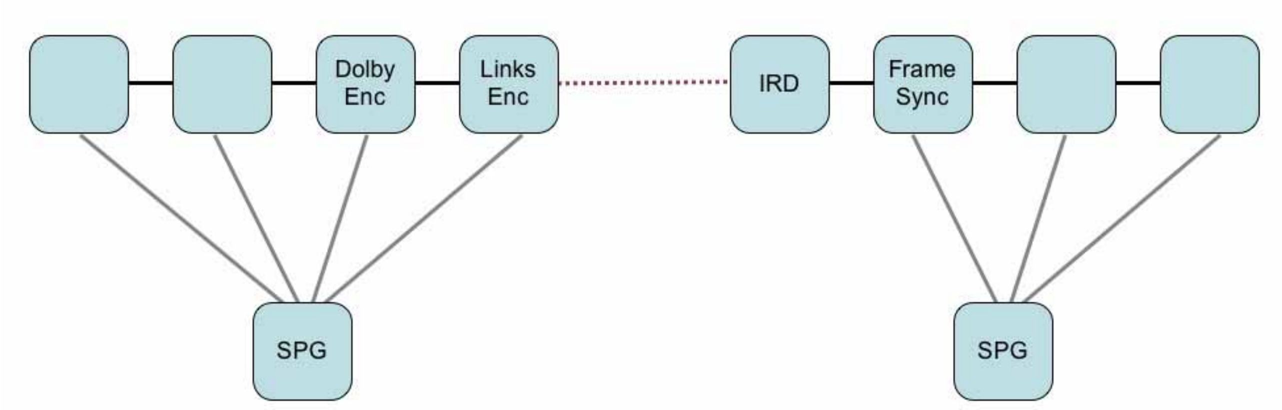

Figure 5 - Contribution link timing, with an IRD locked to incoming video

It is wise wherever possible for all equipment on a single site to be locked to the same SPG. In an outside broadcast situation it is wise to use a local SPG, as any problem with an incoming reference signal from a remote location would result in significant problems with the Dolby E signal chain due to misaligned Dolby E frames. It is also vital to ensure that all equipment is configured to use the synchronisation reference presented to its sync input.

In particular, the BBC have encountered problems with some MPEG links encoders, which may default to using an internal clock reference even when an external video reference is connected. This is not generally problematic in a PCM audio installation, but with Dolby E this will potentially cause Dolby frames to be misaligned to video within the transport stream. The problem is easily avoided by ensuring that the sync source of the encoder is set to external. An alternative can be to use the incoming video as a sync reference, removing the requirement for a separate black & burst sync reference, however this would cause tears or glitches in incoming video to adversely affect audio, whereas a separate reference signal removes this risk.

At the receiving end, all Dolby E decoding equipment and many other devices will need a video reference signal too. In many cases this will again be sourced from a local SPG. Therefore the incoming signal must be carefully synchronised to that local reference, taking into account the Dolby alignment issues discussed in 3.2. Often, an Integrated Receiver/Decoder (IRD) will be used to decode the incoming MPEG transport stream used to carry the video and audio across the fibre, satellite or other link from the remote site. Many IRDs have a sync input and synchronise the video internally to match the local reference. However in most cases, the synchronisers are not Dolby E aware, and do not account for Dolby frame alignment, meaning that misaligned Dolby E - a significant risk after a contribution link - could lead to corrupted Dolby E frames further down the signal chain. In the BBC's Central Communications Area, a Harris device is used which does perform Dolby re-alignment before video synchronisation, however in cases where the IRD does not, another solution is available. It is possible to sync the IRD to the incoming transport stream's Program Clock Reference (PCR) and not the local reference, and then externally perform a frame synchronisation with Dolby re-alignment, as shown in figure 5.

Another alternative suggested by Tandberg [1] is to clock the Dolby E decoder from the incoming video. In this scenario, an HD to SD down-converter is used to derive standard definition black & burst reference from the incoming HD video feed, and this reference is connected to the Dolby E decoder. However one must be aware of any potential phase difference created by the down-converter, as this could cause problems in using the resultant SD reference. As such, Tandberg recommend against the use of down-converters built into IRDs, which may introduce significant phase offsets. The BBC does not use this method in its infrastructure for accepting remote feeds, but a similar setup is implemented in the coding and multiplex area, where the sync source for the Dolby E decoder is derived from a down-converted version of the incoming SDI video.

3.4 Mixing in a Dolby E Environment

The video-synchronous nature of Dolby E is designed to enable cut transitions to be possible without decoding. Provided the Dolby frames are correctly aligned such that the cut takes place within the Dolby E guard band (figure 3), simple cuts should be problem free. A Dolby E frame actually includes a little audio data which overlaps with both the previous and following frame, meaning that Dolby E decoders are able to decode cuts as short cross-fades, making such transitions smoother. However if the Dolby E is not properly aligned, cuts may cause frames to be corrupted, resulting in mutes or splats.

To perform longer cross-fades, U- or V-fades, or to add voiceovers and other material, the Dolby E must be decoded to PCM, mixed or processed and the re-encoded to Dolby E. This is what happens in the BBC's playout and continuity systems, and this has been one of the areas of greatest difficulty in implementing the Dolby E infrastructure.

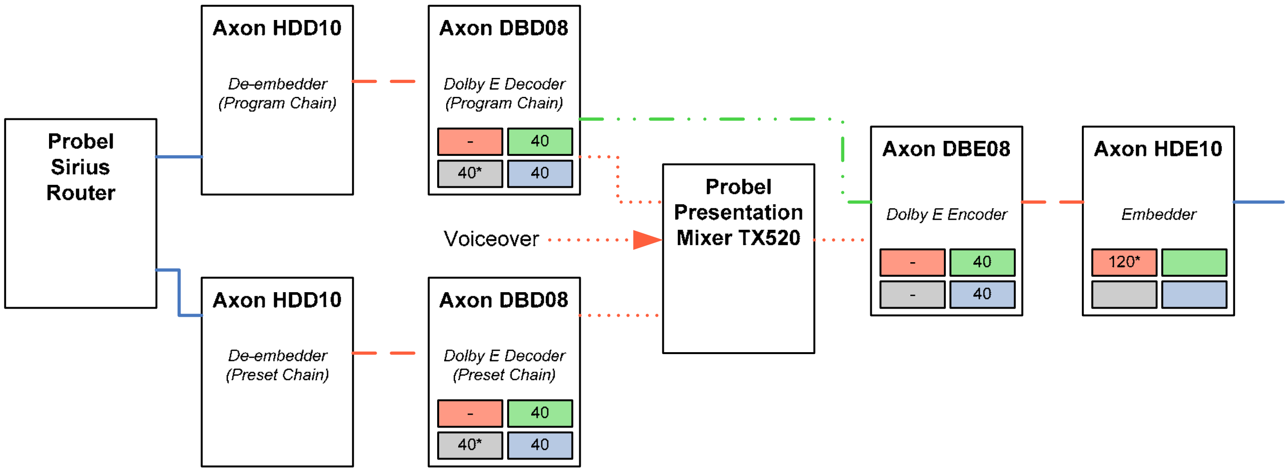

Figure 6 - Typical mixing infrastructure

Figure 6 shows a simplified schematic of one mixer chain, similar to that used by the BBC's partner, Red Bee Media. It can be seen that an audio mixer provides the transitions and addition of voiceovers, in the PCM domain. The router and mixer are controlled in tandem by the automation system, such that the outputs of the router are used to select a program (PGM) and preset (PST) input to the mixer. The main types of transition are as follows:

- In a cut transition, the router simply switches the source on the PGM path. The mixer performs no transition, and the PST path is not used.
- In a V-fade, the mixer reduces the volume of the outgoing audio source to silence, then the router switches the new source onto the PGM bus, and the mixer ramps up the volume again. The PST path is not used.
- A U-fade is similar to a V-fade, except that silence is maintained for a certain amount of time at the middle of the transition.
- A cross-fade uses both the PGM and PST paths. The volume of the PGM source is reduced as the PST volume is simultaneously increased until only the PST source is heard. A short while later, the incoming source is switched onto the PGM path and the mixer cuts back to the PGM path; this is referred to as the *switchback*.

It can be seen from figure 6 that there is a de-embedder and a Dolby E decoder between the router and the mixer. Previously there has also been a frame synchroniser, though this has now been removed as it was not necessary. It must be understood that these devices introduce a delay, which has caused significant problems in mixing. In a V-fade, the router changes the source at the point at which the mixer has faded down the outgoing source to silence. The mixer then immediately starts fading up the PGM bus again, in order to fade up the new source, with the intended effect that the outgoing programme fades to silence and then the new programme fades up. However, the delays introduced by the de-embedder and decoder mean that the outgoing source is actually still be present on the PGM bus for some time after the fade-up begins. (This is now slightly over a frame, though was previously up to 3 frames.) The outgoing audio source will therefore be heard again briefly, before switching to the incoming source as the fade continues. Looked at from the point of the mixer, it appears as though the switching of sources has happened too late.

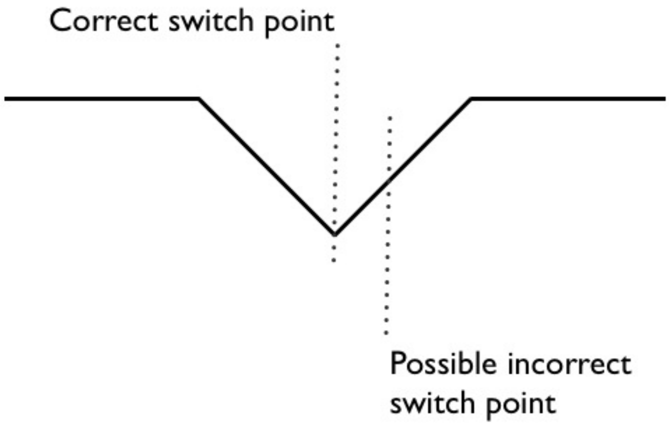

Figure 7 - The effects of decoding latency on V-fades

There only viable solution to this problem at the present time is to change the V-fade into a short U-fade, adding some silence in the middle to allow time for the new source to propagate to the mixer. The Pro-Bel mixer used in Red Bee has a “switchback delay” parameter which has exactly this effect. It delays the mixer by a variable amount before beginning the fade up. This setting also applies to cuts, adding a short amount of silence between the outgoing and incoming source, which is an acceptable compromise and as such this solution is now used by Red Bee Media on behalf of the BBC.

Pro-Bel provide their own white paper on mixing in Dolby E, which may be of additional interest [2].

Another problem that has been experienced at programme junctions is format switching causing instability in consumer decoders. It was found that, whilst all programmes are broadcast using Dolby Digital, the switch between 2.0 and 5.1 configurations (or vice versa) causes many receivers to produce audible glitches. BBC HD experimented with signalling all programmes as 5.1 to mitigate this problem, however viewers with Dolby Pro Logic decoders found this unacceptable. This is because such decoders would normally attempt to up-mix stereo programmes by routing some audio to the surround speakers, but cannot do so when the programme is signalled as 5.1. (It is also confusing when an amplifier indicates 5.1 but sound only comes from 2 speakers.) The solution currently employed is to signal all programmes in the correct format, but to dip to silence in junctions where the change is necessary. By making the format switch during a brief period of silence, the effect is hidden and decoders do not produce audible splats. This solution is analogous to the widescreen switching employed by the BBC, whereby changes between 4:3 and 16:9 programmes are made by dipping to black for a few frames, allowing consumer displays to settle on the new format without any visible effects to the viewer.

An additional complication in mixing Dolby E comes from the metadata carried within the Dolby E stream. When the audio is decoded to PCM, the metadata must be carried separately from the decoder to the encoder where it is reunited with the associated audio. There are problems to be aware of when separating metadata, which are outlined in section 5.3.

4 Audio-Video Synchronisation

Dolby E encode and decode cycles each incur a fixed delay of 1 video frame (e.g. 40ms in a 25Hz system). This means that delays have to be carefully managed, as significant audio-video synchronisation problems would otherwise occur, however the ‘round number’ nature of the processing delay makes the effects easier to manage.

Also, while delays caused by processing devices may be a long-standing problem for broadcasters, it is now becoming a factor for consumers too with the widespread use of LCD and plasma displays. The BBC set out to ensure that audio-video sync was correct throughout the production and distribution chains, and then to help viewers align their own equipment.

4.1 Delivery Requirements

In a PCM environment, it is expected that programmes will be delivered to a broadcaster with audio in sync with the video. However in a Dolby E system, the choice is less obvious. There are two main options:

- In Sync Encoded: The encoded audio lies on tape (or appears on the link) in sync with the video. The decode delay must be compensated for at the decode site by use of an equivalent video delay.
- Advanced (decode-compensated): The encoded audio appears a frame ahead of video, meaning that after the decode delay, the audio is in sync.

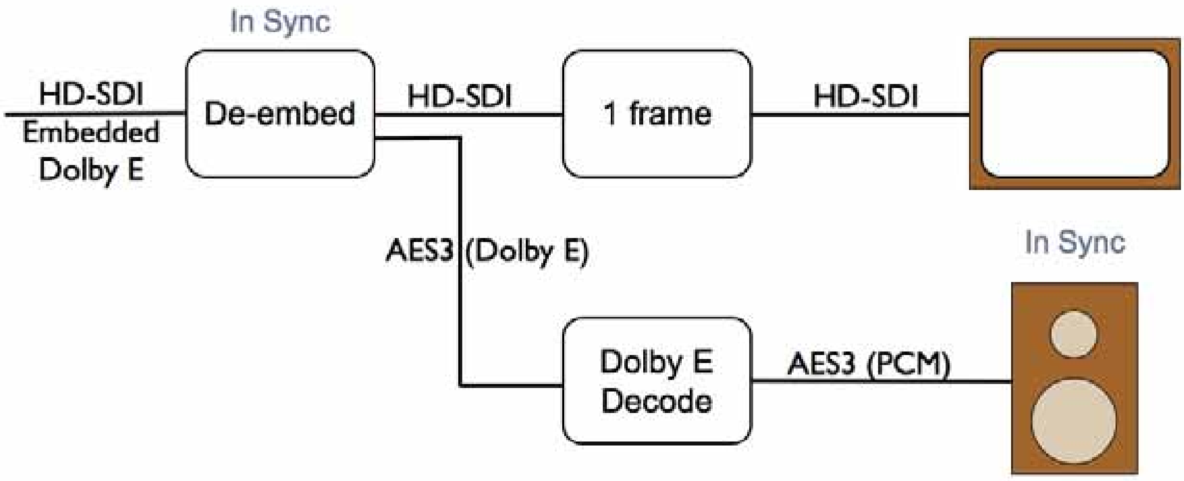

Figure 8 - The BBC's chosen delivery and delay solution for Dolby E

The BBC began by using the latter option in order to keep monitoring simple and to reduce the number of delays required, but in conjunction with other UK broadcasters chose to change to in-sync encoded delivery. This change was motivated by a variety of factors, including the ability to make cut edits without an audio offset. However the review of Dolby E found that an administrative error had meant that one technical area was still working to the older standard, causing BBC HD to have one frame of constant lip-sync error. This was quickly corrected, but is a cautionary tale to those managing Dolby E; delays must be carefully considered!

4.2 Synchronisation in the Delivery Chain

It is important to note that most broadcasters' distribution chains accept mixed content. The BBC's playout area must accept programmes incoming with 5.1 Dolby E audio and those with stereo PCM, outputting all programmes encoded as Dolby E (5.1 or 2.0) with appropriate metadata and correct sync. Therefore the different delays through devices must be considered for Dolby E, PCM audio, video and sometimes separate metadata. In attempting to track these delays, it quickly became apparent that traditional schematics would not present the required information concisely enough. As such a simple but effective notation was developed whereby any device can be represented as a generic block with delays indicated for the previously mentioned 4 elements.

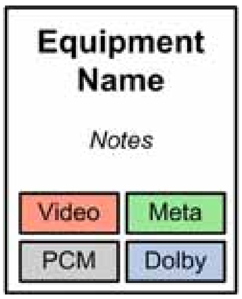

Figure 9 - A generic schematic block identifying delays

Figure 10 shows this block in use to describe a section of the BBC HD playout systems. This area is an excellent example of the challenges involved with managing sync. The Dolby E decoder, for example, must be carefully configured such that it passes through PCM audio unaltered but delays it by the same amount as the device would delay a Dolby E stream i.e. 40ms. If the decoder is incorrectly configured, it will leave PCM audio out of sync with the video because the rest of the system is designed to keep Dolby E in sync.

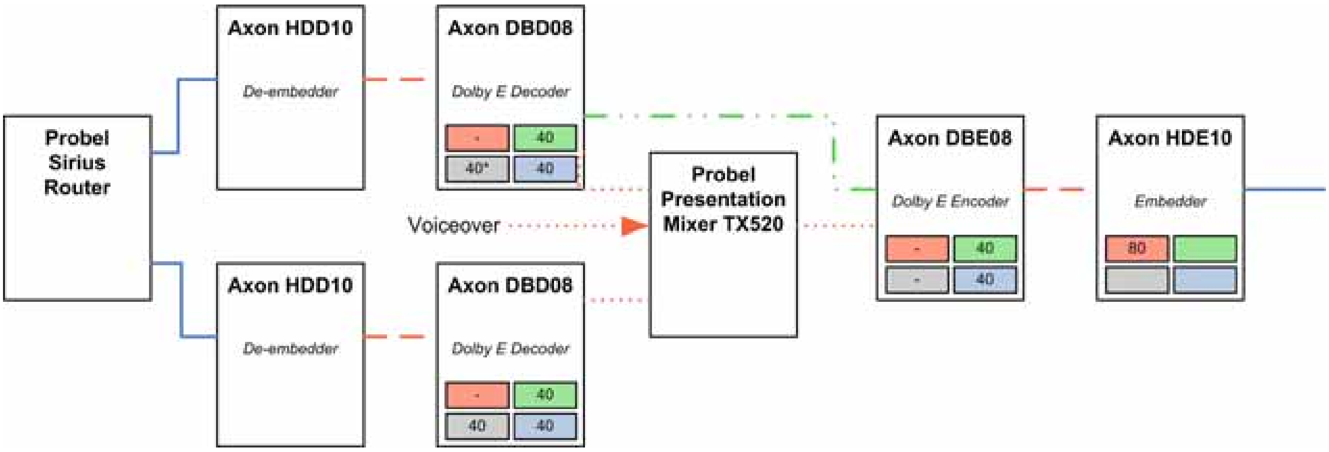

Figure 10 - A section of the BBC's playout infrastructure

Meanwhile the metadata for a Dolby E programme is taken from the decoder to the encoder on a serial link, so any audio delays that occur in between these devices - i.e. in the mixer - must be considered as they must not put the audio out of sync with the metadata. This could cause the metadata to be re-associated with the wrong frame of audio, causing difficulties at junctions. (The mixer's delay is low enough to be negligible in this particular case.) Finally the video must be delayed to match the delays in the audio chain. This delay is applied in the embedder, such that the embedded output is in sync. Whilst these issues may initially seem complex, the notation used allows simple arithmetic checks to ensure that the delays for PCM, Dolby E, video and metadata all match.

There is no substitute for testing, however. An appropriate sync test (such as Pro-Bel's VALID/VALID8, similar tests from signal generators such as Harris' Gen-Star or a tone & flash test from various devices including Tektronix monitoring equipment) should be used to ensure correct sync. When doing so, if the test signal needs to be encoded to Dolby E after generation and/or decoded to PCM for measurement, the delays involved in this process must be carefully accounted for so that the readings obtained are fully understood. The diagrams produced of the BBC's infrastructure and the delays involved were used to check and verify that test results were as expected, never as a substitute for testing.

Another potential pitfall is avoiding double-compensation. Many devices which are Dolby E aware "helpfully" include a frame of video delay to compensate for the audio decoding delay. Such devices include de-embedders, MPEG video links encoders and more. Clearly one must ensure that all such delays are known about. Generally these delays will be configurable, but the settings should be checked, as if a de-embedder has a frame of delay turned on and then a discrete frame delay is also used, the video will be double-delayed and hence out of sync with the audio. This situation has occurred on BBC outside broadcasts before, such as *Electric Proms* in 2009, where a links encoder inserted a frame of delay without the on-site staff realising; they had carefully added a video delay to compensate for the Dolby E encode. The lip-sync error was noticed, however the cause was only later discovered by careful examination of the links encoder manual. This kind of careful examination of the properties of each device used is necessary when dealing with Dolby E, and should always be done in advance.

VT machines such as HDCamSR devices have an *audio advance* setting, which allows audio to be advanced a frame on playback. This is very useful for accounting for Dolby E decode delays at playback time, however once again the settings must be understood and checked carefully to ensure that sync is maintained.

The BBC often uses video delays in embedders and other equipment to compensate for audio encode/decode delays rather than using discrete delay devices, but crucially the corresponding audio and video delays are always co-sited. This ensures that a signal leaving any individual technical area (a studio, CCA, playout etc) is in sync. It is then easier to test the sync of a particular area as there is a simple rule to work to; if it arrives in sync, it should leave in sync. The BBC share an approach with some other UK broadcasters, that no area should compensate for a separate area's delays.

4.3 Synchronisation in the Home

Whilst mixed audio formats in the delivery chain create lip-sync challenges for the broadcaster, the viewer has a challenge of his or her own in the home. In a situation where a set top box (STB) is separately connected to a display and an audio receiver or amplifier (by HDMI and optical, for example) the audio and video signal chains become separated. Processing delays in one chain can therefore cause synchronisation problems, and with modern LCD and plasma displays routinely adding anywhere up to 100ms of delay due to de-interlacing and other processing, the user can be left with significant synchronisation errors.

Worse still, the video is late with respect to the audio. With some simple consideration of the properties of sound and light waves, it can be seen that this is more disconcerting for the viewer than audio being late. Sound travels slower than light, so humans are used to the sound of an event reaching them fractionally after the visual. Anyone who has been to a concert in a stadium, park or other large venue will know that the sound of the singer and the sight of their lips moving on stage (or on video screens) are noticeably out of sync, with the audio being late. So introducing a situation in the home where the reverse is true - the video lags behind the audio - creates a highly unnatural effect which is confusing to the brain.

Work by BBC Research & Development has investigated the difference in perception of audio-video synchronisation between standard definition and high definition. Early results seem to indicate that some subjects are more sensitive to lip sync in HD, highlighting the need to minimise sync problems [3].

Many STBs have a configurable delay on their optical audio outputs, allowing the audio to be delayed by increments of 20ms (half a frame). However with no reference signal upon which to base their judgements, viewers are left relying on lip synchronisation in television programmes, which is subjective, inaccurate and of course could even be wrong at the point of broadcast. The BBC has strict guidelines as to acceptable lip-sync, but by necessity they are finite. Anything within 10ms audio early to 20ms audio late is allowable; this is unnoticeable to the viewer in normal circumstances, but insufficient for sensitive alignment of equipment.

Given that BBC HD currently only broadcasts programming at certain times of the day, a decision was taken that a sync test could be broadcast as part of the promotional loop which plays at other times. Once per repetition of the promotion (i.e. every two hours during the daytime), 90 seconds of a sync reference is played in order to allow viewers to adjust their equipment. The design of this test, and the process of ensuring that it was broadcast in sync to much tighter tolerances than usual, was the subject of considerable effort.

There were some unique requirements for a sync test to be broadcast in this manner. First and foremost, it should be as easy as possible to perform the test "by eye" without specialist equipment. Additionally, in order for BBC engineers to be able to perform precise measurements, automated millisecond-accurate measurements were a requirement. Finally, it would be ideal if the device required to perform these precise measurements was electrically simple. This would mean that testing devices could be produced in-house, and potentially the details for how to build such devices could even be made available to the public.

The starting point was work from BBC Research & Development, who had previously developed a sync test sequence referred to as the *digital clapperboard* [3]. Figure 11 shows a still frame from the modified BBC HD version of this sequence.

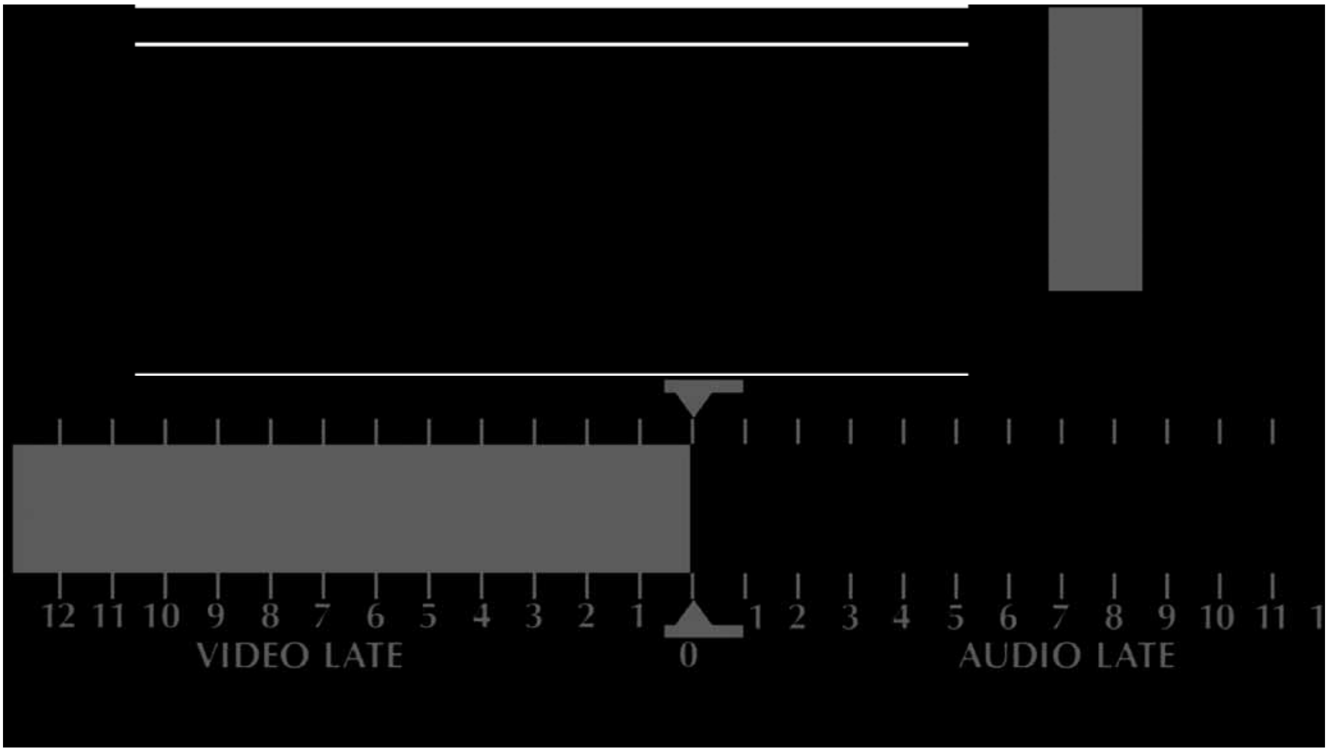

Figure 11 - The BBC HD sync test

The signal consists of 3 primary elements. The top right of the image is a bar which extends down from the top of the screen until it touches a static line just above the centre of the image. The extending bar touches the line at the same time as an audio “snap” is heard, then retreats back to the top of the display. The second element consists of 3 white horizontal lines, 5 video lines each, which appear for one frame at the same time as the audio snap. Finally there is a horizontally expanding bar moving across a time scale marked in video frames. When correctly synchronised, it reaches the centre of the screen at the time of the audio snap.

The horizontal scale is best used for measuring by eye. One can examine the right-hand tick mark (11 frames audio late) and note whether the snap has been heard before the bar reaches this point. Assuming it has, move on to the next tick mark to the left, and perform the same test. Continuing in this manner, eventually a point will be reached where the visual bar touches the tick mark at approximately the same time as the audio snap, or slightly before it. Most users can identify the sync offset to a precision of half a frame (20ms) using this method.

It is worth mentioning at this point the origin of the audio snap. The sound used is a recording of two blocks of wood being banged together. Following a series of tests at BBC Research & Development, this was found to be the sound which was most effective for subjective sync tests, largely due to the strong transients at the start and rapid but audible decay. Had the primary objective of this test been automated measurement, it would perhaps have been easier to use an artificially generated tone for example, however the requirement for “by eye” measurement made this sound the best choice.

Clearly, the broadcast of a sync test on one of the BBC’s flagship channels requires careful planning to ensure that the test signal reaches the viewer in sync. It would have been an embarrassing error to broadcast a sync test which was itself out of sync. As such, careful testing of the entire delivery chain was undertaken to ensure millisecond-accurate timing of this signal throughout.

An electronic test device was used to provide precise measurements of the sync signal. Using an analogue audio input to measure the audio snap and a “light pen” to measure the horizontal white lines which flash on a display, the offset in milliseconds between the two is derived and displayed. Because this measuring method uses a display and a decoded version of the audio, the delays of these devices must be carefully understood. A CRT display must be used due to the delays involved even in professional LCD and plasma displays, and the frame of delay from the Dolby E decoder (if the signal is Dolby E encoded) must be accounted for.

The first challenge was to get the test signal itself inserted into the promotional loop with perfect synchronisation. The sync test signal was produced and laid to tape. It was then ingested in the edit suite, and inserted into the promotion by the editor. Once the video edit was completed, the promotion was transferred to the dubbing suite for the audio finishing. From there the whole loop was Dolby E encoded and laid back on to tape (with a frame of video delay used to compensate for the Dolby E encode, of course). This tape was taken back to the edit suite where it was played and measured, producing a sync error value which was taken back to the dub in order for the audio offset to be adjusted accordingly. The reason for the back-and-forth was that an HD CRT was only available in the edit suite (so reliable measurements could only be made there) whereas the only place where sub-frame timing adjustments could be made was the dubbing suite, whose systems allow the audio to be moved in samples. Eventually a full copy of the promotional loop was available on-tape and measured to be in sync.

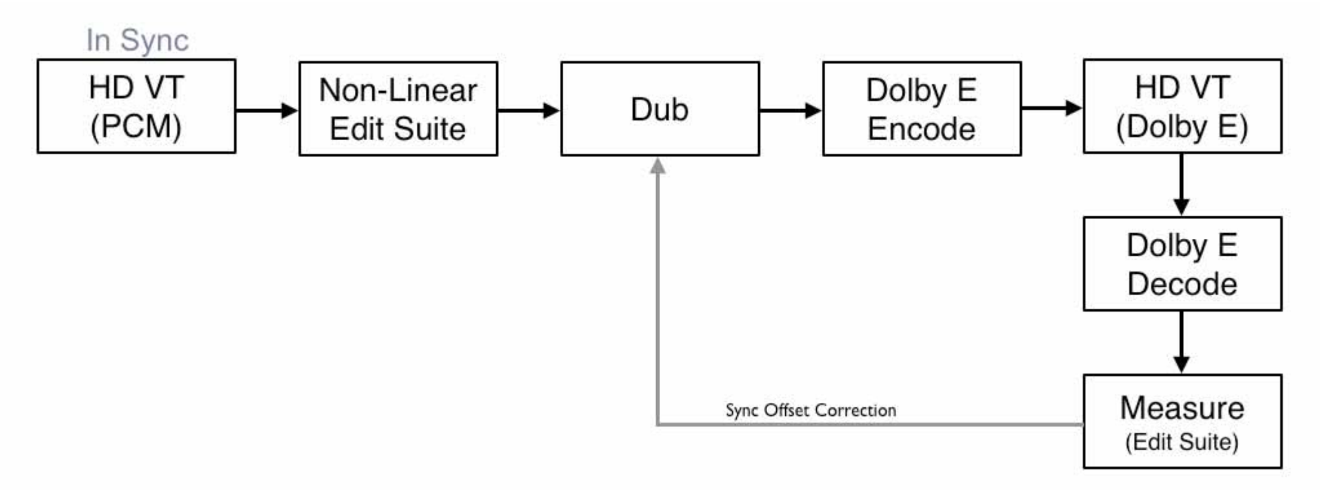

Figure 12 - Producing an in-sync version of the HD promotional loop

Next, the playout chain was tested. The tape was ingested into the playout server, and various measurements taken along the playout signal chain, within the playout centre and down-stream in the Central Communications Area. The emission coders could not be tested on air as this would potentially involve broadcasting the signal out of sync, so a duplicate set of encoders at BBC Research & Development's Kingswood Warren (KW) site were used to perform measurements. The playout and distribution chain was found to be in sync to a tolerance of less than 2ms; however the emission coders did introduce some offset. The measured value of this offset was used to adjust the video delay applied in the on-air coder, and finally the test was ready for broadcast.

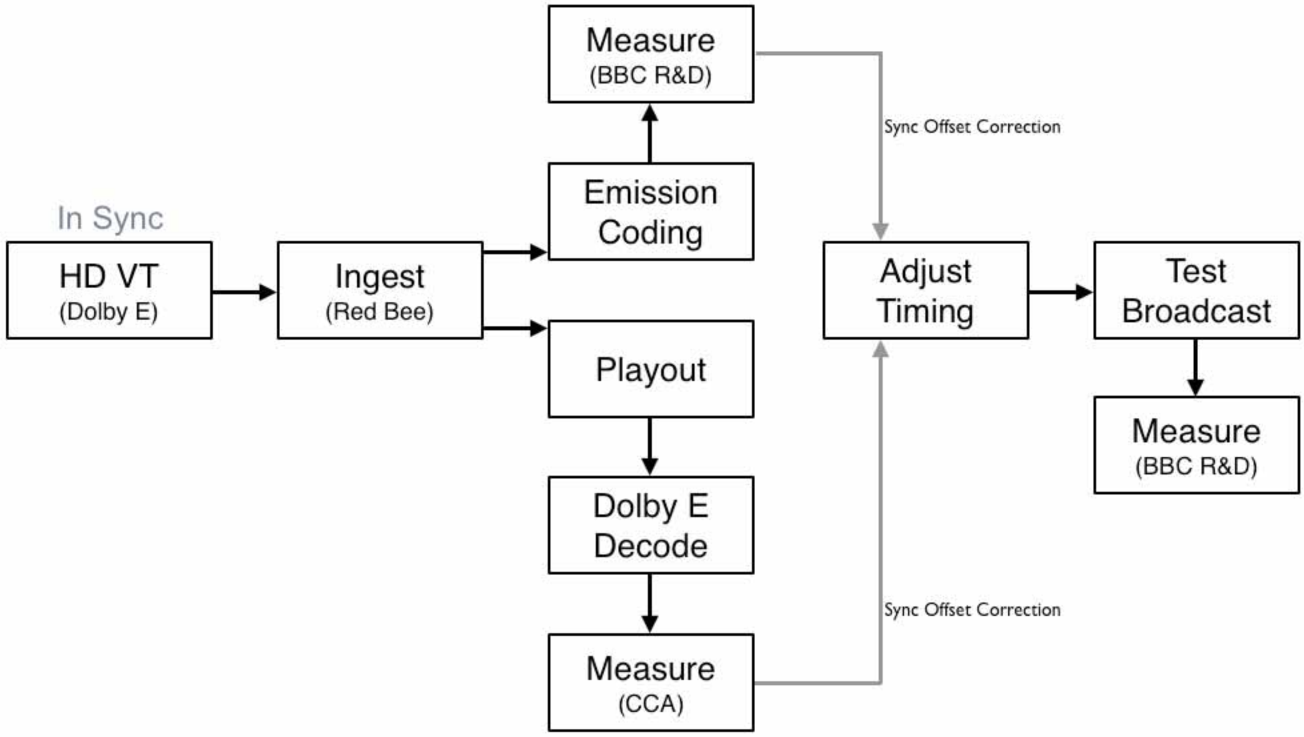

Figure 13 - Measuring sync in the broadcast chain

There is an old adage in broadcast that people claim a signal was “alright leaving me”, implying that any error was caused at the receiving end or in the link. The desire for the BBC HD sync test was to go a step further and ensure it was “alright arriving at you”. In other words, the signal was required to be correctly timed when received off air. To verify this, the sync test signal was broadcast, and measured off air at Kingswood Warren. This test used the most reliable and precise measurement possible; a transport stream analyser was used to examine the presentation time stamp (PTS) values of the MPEG transport stream, allowing identification the exact timing of the video flash. A PTS covers only one video frame but multiple audio samples, so the PTS of the start of the audio snap was not sufficient to make the corresponding audio measurement. However by examining the decoded waveform, the exact audio sample number within the PTS block could be found, and (knowing the audio sampling rate used) the time difference between the snap and the flash could be precisely measured. The off air test revealed this offset to be 0.9ms, an excellent result.

BBC HD continues to broadcast this sync test multiple times daily, so audiences now have a way to test the synchronisation of their home equipment. The test has been explained in blogs by both Rowan de Pomerai [4] and Andy Quested [5], (attached in Appendix I & II) and the response from viewers has been overwhelmingly positive. Additionally, it has generally led to a reduction in the number of comments received about the sync of the channel, and while problems on individual programmes do of course occur occasionally, there is now a confidence that the channel’s broadcast chain is in sync, and the BBC continue to push technology partners to verify synchronisation any time any change is made to devices in that chain, so that the synchronisation remains correct in the future.

5 Metadata

Dolby E includes audio data rate reduction (compression), but it is a mistake to treat it simply as a compression system. The data stream also carries metadata which describes the audio and controls downstream equipment, notably Dolby Digital encoders. The metadata included in the Dolby Digital stream controls the output level, dynamic processing and down-mixing performed in the receiver or set-top box, so setting the metadata correctly is vital.

5.1 Dolby Digital Metadata Overview

It is beyond the scope of this document to explain the detail of every metadata parameter used by Dolby Digital and Dolby E, however substantial information is provided by Dolby themselves [6] and the SMPTE [7]. The most important parameters to consider here are those which affect the final audio output in the home, primarily the “3 Ds” of dialnorm, dynamic range control and down-mixing. They are carried by Dolby E through the broadcast infrastructure and then via Dolby Digital to the home.

Dialogue level, also known as *dialogue normalisation* or *dialnorm*, is an indication of the loudness of the audio material. Its background is in film, where the level of the centre channel dialogue is a good indicator as to the value to be used, however the dialnorm value is still applicable in television, including programmes with no dialogue. By indicating the average loudness of the programme, the set-top box can adjust the level of all material to maintain an approximately constant loudness across programmes, junctions and where applicable, commercials. The decoder makes a volume adjustment to normalise all programmes to -31dBFS. For example a programme with a dialnorm of -27dB will be reduced by 4dB, whereas a programme at -22dB would be reduced by 9dB. It is very important to set dialnorm correctly, as an incorrect dialnorm will cause a programme to be reproduced too loudly or too quietly.

Dynamic Range Control or *DRC* applies dynamic processing in the set-top box to reduce the dynamic range of the audio. Expansion is applied to increase the level of quiet material while compression reduces the level of loud sections. The DRC metadata settings allow the selection of one of a number of DRC *presets* for each of two *profiles*. The profiles are *line mode* and *RF mode*, named after the type of outputs which to they are generally applied in consumer equipment such as DVD players or set-top boxes. The RF output will be capable of a small dynamic range, and must be protected against over-modulation, while line-level outputs are capable of a greater dynamic range. As a result, the RF mode will generally apply stronger range reduction than line mode, and peak-limiting is applied to prevent over-modulation.

Some decoders additionally include a mode which allows users to apply RF mode DRC to all outputs, reducing the dynamic range on the line outputs to allow quieter listening without quiet parts of the programme becoming inaudible. This is suitable for listening at night without disturbing neighbours, and hence is often called “midnight mode”.

The DRC preset for each mode chooses how much range control is applied. The available presets are *Film Standard*, *Film Light*, *Music Standard*, *Music Light*, *Speech* and *None*. Details of each preset are available from Dolby [6]. The presets adjust the amount of boost and cut applied and the ranges over which they are applied. However the basic form of each preset is the same, and is illustrated in figure 14.

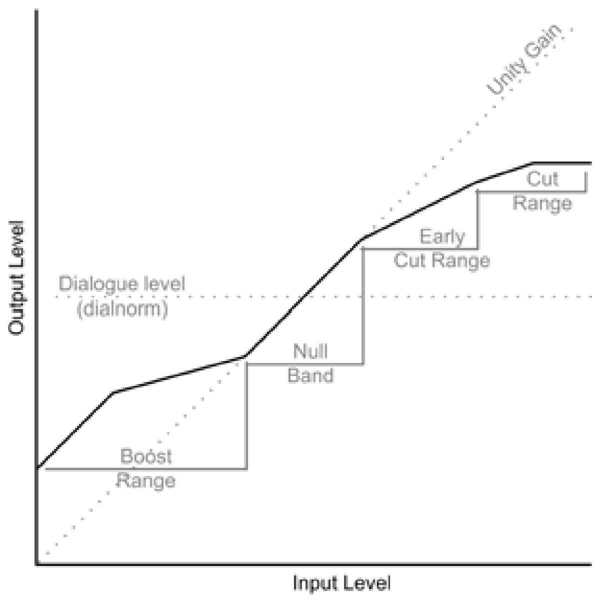

Figure 14 - Dynamic Range Control in a Dolby Digital decoder

Notice that the DRC behaviour is centred on the dialogue level as defined by the dialnorm parameter. This specifies the centre of the *null band* in which no DRC is applied. This relationship between dialnorm and DRC illustrates how critical it is to set the dialnorm correctly; incorrect dialnorm will cause incorrect DRC effects.

Finally the down-mix parameters define how a viewer listening in stereo hears a programme which is originated in 5.1. The surround sound programme is *down-mixed* to produce a stereo output by combining the contents of the surround and centre channels with the front left and right. Once again, the set-top box performs this processing under the control of metadata, so the down-mix parameters are important. Two types of down-mix are available, Lo/Ro and Lt/Rt. The former stands for “left only, right only” and the latter for “left total, right total”. Lt/Rt is a Dolby Surround mix which enables Dolby Pro Logic decoders to reproduce a rear channel, whereas Lo/Ro is a simpler down-mix which is compatible with further down-mixing to mono.

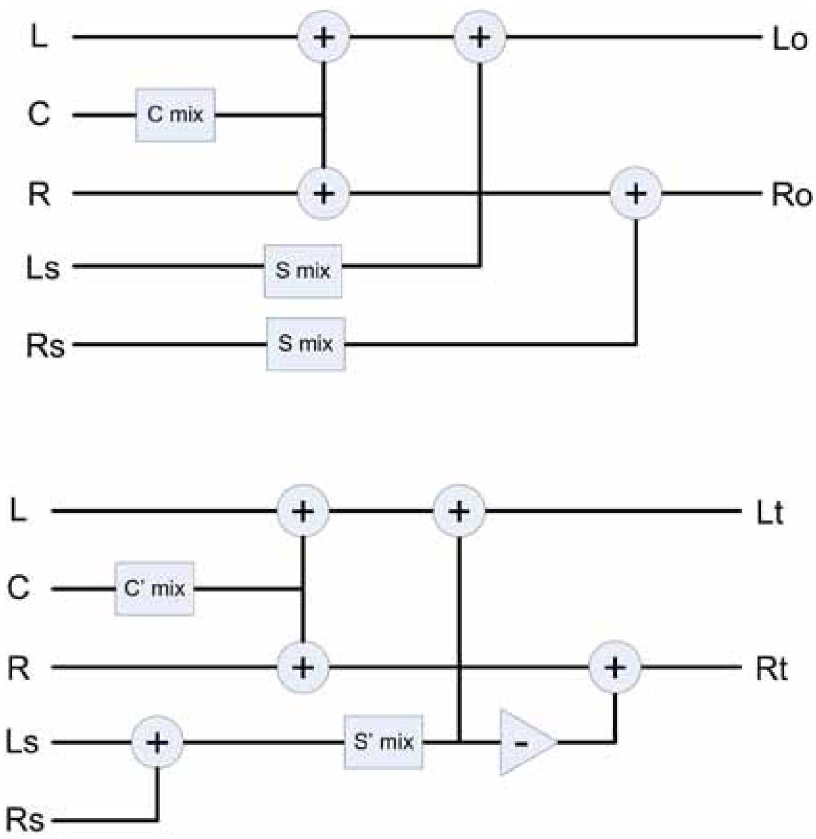

Figure 15 - Lo/Ro and Lt/Rt down-mixing

The metadata parameters allow control of the attenuation of surround and centre channels when down-mixing. The BBC generally use both a centre channel down-mix level and surround channels down-mix level of 0.707 (-3dB).

Additional parameters are available for decoders which support extended bitstream information (BSI), which provides extensions to the Dolby metadata. For these decoders, separate parameters are available for the down-mix levels used in Lo/Ro and Lt/Rt down-mixes. Hence if a user chooses a particular down-mix type, different down-mix parameters could theoretically be used, however the BBC’s standard parameters are the same for both down-mixes. It can be confusing however that the Dolby metadata set allows 3 different sets of down-mix parameters, the standard down-mix parameters used by decoders without support for extended BSI, and separately the two sets used for extended BSI decoders. One must remember that extended BSI encoders will simply ignore the standard down-mix parameters if separate Lo/Ro and Lt/Rt parameters are available.

Extended BSI also allows the broadcaster to specify a preferred down-mix, and the BBC chooses Lo/Ro for mono compatibility. However other broadcasters make different choices, for example Sky, whose history of providing Dolby Surround mixes before 5.1 was available means they prefer Lt/Rt to maintain Dolby Pro Logic compatibility. It must also be noted that some set-top boxes ignore this preference and default to one particular down-mix anyway, so broadcasters cannot assume one particular down-mix will be used.

One final metadata parameter that controls the audio output level of the decoder is *Surround 3dB attenuation*. This parameter originates from film dubbing theatres which historically have surround (i.e. rear) speakers at 3dB lower than the front speakers. Because films are mixed to this level, the decoder applies the same 3dB attenuation to the surround channels; however television is not mixed in the same environments, so this setting should usually be off.

5.2 Authoring Metadata

For Dolby E originated programming - meaning all 5.1 currently delivered to the BBC - the metadata is authored at the encoding site, be that a studio, outside broadcast or dubbing theatre. Two options are available:

- Author in the Dolby E encoder, e.g. Dolby's DP571.
- Author externally, e.g. in Dolby's DP570 multichannel audio tool, and then pass the metadata to the encoder via serial link.

Dolby's *Standards and Practices for Authoring Dolby Digital and Dolby E Bitstreams* [8] suggests using a DP570 to author the metadata, because this device can be remotely controlled from a user-friendly graphical interface (GUI) on a PC, and the device itself can also *emulate the metadata*, meaning it can provide monitoring with simulation of the effects of dialnorm, DRC and other metadata parameters. However the BBC currently prefers authoring directly in the DP571 (or other manufacturer's encoder). The reasons for this preference are primarily that the DP570 GUI makes it simple to change metadata parameters incorrectly, and some non-technical production staff find it confusing. Generally only a small number of parameters should differ from programme to programme, and the DP571's simple front-panel display interface makes it more difficult to accidentally change settings. By using only one device, confusion is also removed surrounding the source of the metadata. The concern about this largely stems from an incident during 2008's Eurovision Song Contest where incorrect metadata caused viewers only to hear the centre channel as the programme began. The cause was that the DP571's *AC3 Metadata* setting was disabled, telling the downstream Dolby Digital encoder to disregard the other metadata it was receiving from the Dolby E stream. This meant that the otherwise correctly set metadata did not reach the home and viewers were left with very odd results! Andy Quested explains the story in his BBC blog [9] (Appendix III). The AC3 Metadata setting has been removed from more recent versions of DP571 firmware.

It should be noted that all currently available authoring solutions are best suited to fixing a metadata set for the duration of a programme, whereas Dolby E's frame based nature does in fact make it possible to change the metadata many times per second if desired. While there is potential for future developments in the use of dynamically changing metadata, problems still exist in parts of the distribution, including in file-based systems as discussed in 7.2. However, the current structure (whereby metadata is set on a per-programme basis) works well, so there is not significant need to move to a more dynamic system. Potentially the most useful improvement could be made around programme junctions and voiceovers, where there may be audio coming from a variety of sources (e.g. the end credits of a programme plus a voiceover from a presentation suite) and better systems could be developed for automatically choosing appropriate dialnorm in this situation.

5.3 Metadata Over Serial Link

Dolby E decoders such as the DP572 have the ability to output metadata on a serial port (RS-485), for connection to another Dolby E device. Devices such as the DP570 multichannel audio tool can also author metadata for delivery to an encoder via serial link, as described in 5.2. The device outputs a burst of data on the serial port, co-timed with the start of the frame from the video reference, which describes the metadata for the corresponding Dolby E frame. The burst length is variable based on the program config (one 5.1 program has one set of metadata, where as for example a 5.1+2 config has 2 separate metadata sets, so a longer data burst). This facility can be used to pass external metadata to an encoder either for origination or to maintain metadata continuity throughout a decode-recode cycle. The latter situation occurs around the mixer in the BBC's playout system, as explained in 4.2.

When separating metadata from the associated audio, one must be aware of delays. Should the audio be delayed significantly before being recombined with the metadata at the Dolby E encoder, the metadata could be associated with the wrong audio frame. To avoid this, ensure that the audio is delayed by no more than half a frame. It would theoretically be possible to delay the metadata to compensate if necessary, however metadata delay devices are not commercially available at present.

5.4 Metadata for PCM-Delivered Programmes

Discussion of metadata so far has centred on a sound supervisor authoring metadata for his or her programme to best fulfil the artistic requirements of that programme. The distribution chain should therefore be completely transparent to this metadata, allowing continuity from studio to home. However, stereo programmes are generally delivered with linear PCM audio, not Dolby E. As such they have no metadata, and metadata must be applied in the distribution chain. This is done at the Dolby E encoder in Red Bee Media, after the presentation mixer.

Depending on the manufacturer's equipment used, metadata applied to incoming PCM will either be set by the decoder (which authors the metadata and outputs it on the serial link for transfer to the encoder) or the encoder. A decoder would detect that incoming audio is PCM and not Dolby E, switching to metadata generation mode, whereas an encoder would switch to internal metadata generation when no metadata is received on the metadata input. In either case the end result is the same; outgoing Dolby E has metadata included, and that metadata must be chosen by the broadcaster.

In the BBC's case, the metadata to be used is based around the recommended metadata settings in the HD Delivery Specification [10]. The program config parameter defines what *programs* (sets of associated channels, e.g. a stereo pair or a 5.1 set) are included in the Dolby E stream. For example a television programme with 5.1 audio would have a program config of "5.1". It would also be possible to have "5.1+2" if the 8 Dolby E channels contained both a surround mix and a stereo mix. The program config applied to PCM programmes is "2+2" which indicates two separate stereo programs, because a simple "2" option is not available. What this means is that the first one contains the programme audio and the second is discarded.

The down-mix parameters and other settings are the same as those suggested in the delivery requirements; however the dialnorm and DRC settings are chosen specifically for stereo programmes. The dialnorm used by the BBC for stereo programming is -23dB, a value very similar to that used by Sky (-22dB). This is considered to most closely match the loudness levels to which stereo programmes are generally mixed. It is important to get this setting correct, as an incorrect dialnorm would cause stereo programming to be reproduced too loudly or too quietly compared with 5.1 programmes, and the transition from one to another would be jarring for the viewer.

Some discussion has taken place around DRC settings applied to 2.0 material. Stereo programmes mixed to the standards required by analogue channels (as all BBC HD stereo programmes are) should not exceed the headroom available in the Dolby Digital system, and so should not require further dynamic processing. However some broadcasters prefer to apply some DRC in order to avoid activation of overload protection in the decoders should the signal peak too high due to an error.

Standardisation work is currently underway in the Audio Engineering Society (in the SC-02-02 group) on a simple method to convey metadata within PCM, using the embedding technique of AES41. It is anticipated that this method will allow the carriage of any of the complete Dolby metadata set, a restricted subset (the "3 Ds"), or AAC metadata. This will allow the creation of simple metadata-generation products where Dolby E is not being used.

5.5 Program Description

Each *program* (group of associated channels) within a Dolby E stream can have a *program description* (PD) which is an alphanumeric identifier. There are various schools of thought as to the most useful scheme for choosing PD values. Early BBC delivery specifications required a description of the programme with series title and episode number. This remains the case for tape-delivered programmes, however for live material the BBC now uses a scheme also implemented by other UK broadcasters whereby a description of the source encoder is used as the PD. For example, an encoder in Television Centre's studio 1 would use a PD of *BBC TC1*. In cases where multiple encoders exist (e.g. a main and reserve) a suffix is generally added, e.g. *BBC TC1-M* for the main encoder. Stereo programming will generally have a PD which signifies the fact that its metadata is authored in Red Bee Media, as described in 5.4.

Using this PD scheme enables downstream examination of a problematic stream to ensure that the metadata source is known, which can aid in finding problems. Additionally it avoids staff having to change the PD of an encoder for each different programme filmed in the same studio.

5.6 Metadata Reversion

A number of Dolby E and Dolby Digital devices have *reversion modes* for metadata. These settings allow control of the behaviour of a decoder or encoder when it receives invalid, corrupt or no metadata. For example a decoder may generate metadata internally to output on its serial link, or an encoder may disregard incoming metadata and generate its own instead. Because metadata directly affects the way the viewer hears a programme's audio, ensuring that these reversion settings are sensible is essential. Furthermore, the use of program description to indicate that metadata reversion has taken place - and where - can aid in troubleshooting problems.

The BBC issued metadata reversion guidelines in spring 2009 in order to co-ordinate the settings throughout the delivery chain. The metadata suggested for use in a reversion setting is (as with metadata for PCM programmes) based on the recommended settings from the BBC HD delivery specifications [10]. Because the reversion is done automatically in an error situation, the values used may not match the programme material as well as the values that would have been chosen by the sound supervisor, so an attempt was made to choose settings that would cause the least degradation to the reproduction of the most programmes. In particular, program config and dialnorm were critical choices.

The program config is specified as 5.1. In this mode, the consumer's decoder will reproduce a surround or stereo programme, and the worst case will be a stereo programme being incorrectly signalled, meaning Dolby Pro Logic decoders will not produce a rear channel. If 2.0 signalling were used, 5.1 audio would be seriously degraded, with some of the audio not being reproduced in the home at all, which is a much worse situation for the viewer.

A dialnorm of -23dB most closely matches the level of stereo programming, whereas for example -27dB (Dolby's factory default) would reproduce such programmes far too loudly. The worst case here is that a 5.1 programme will be reproduced too quietly due to having too high a dialnorm; however this is likely to be less objectionable for the viewer than a stereo programme being too loud.

The program description requirements specify that in a device whose internal metadata will only be used in a reversion situation, the description should indicate the location of the device and use the additional *MDR* suffix to enable engineers downstream to easily see that metadata reversion has occurred.

5.7 Handling Metadata Problems

In response to the high-profile problems that have occurred on some live outside sources such as 2008's Eurovision Song Contest, the BBC has worked with technology partner Siemens to implement an extra line of defence in the Central Communications Area (CCA). As described in 1.1, CCA is a central routing point for the distribution chain, and is the delivery point for outside broadcasts. As such it is an ideal place to correct problems with Dolby E streams, such as bad Dolby line position or corrupt metadata. A sidechain has been installed which allows a decode-encode cycle to be applied to any Dolby E stream, outputting a well timed Dolby E signal with locally generated metadata if necessary.

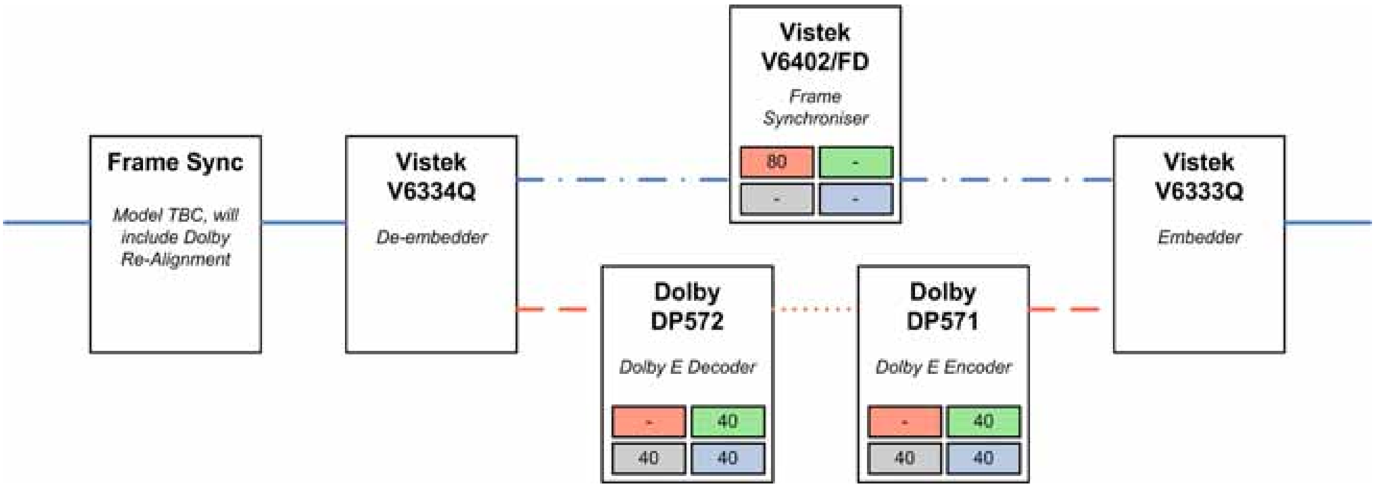

Figure 16 - CCA Dolby E sidechain

As well as a technically malformed Dolby E signal, other reasons for the use of this sidechain could include artistic decisions. If, for example, the dialnorm for an incoming outside broadcast is deemed to be considerably wrong, or other metadata is causing problems, a first solution would be to speak to the outside broadcast directly and ask them to amend their metadata settings. However there could be situations where language constraints make it difficult for staff to communicate complex requests, making it a challenge to make the request quickly and successfully. Alternatively, an international OB may not be prepared to modify their settings as this would affect other broadcasters too. In such a situation it may be necessary for the BBC to initiate use of the CCA Dolby E sidechain with manual metadata settings in the Dolby E encoder, such that the incoming metadata is overwritten with BBC-chosen settings.

In the Eurovision situation previously mentioned, a metadata setting in the source encoder was to blame for viewers only hearing the centre channel of the 5.1 audio. Had the sidechain been available at that time, it could have been used to overwrite the metadata with a known good set, enabling a quicker resolution to the problem.

6 Monitoring in a Dolby E Environment

In a traditional stereo environment, audio monitoring consisted of little more than listening to the stereo audio; perhaps one would occasionally check for CRC errors (cyclic redundancy check errors) in the data stream, or spot-check the mono down-mix. However in a Dolby E environment, matters are more complex.

6.1 A Layered Approach

The key to thorough monitoring with Dolby E is to understand the myriad elements that affect the final audio output. The effects of metadata have been discussed in section 5, so it should be clear that careful checks of the metadata are necessary. The Dolby E stream itself is an encoded data stream, so as well as the frame location and timing checks, it is sensible to monitor for CRC errors in the stream. The Dolby E frames are carried by an AES3 carrier, another data stream with its own CRCs and that in turn is frequently embedded in SDI or HD-SDI, yet another digital data stream with its own CRCs. There may also be other layers such as MPEG transport streams or other links encoding mechanisms. An error in any of these layers can cause disrupted audio output, so monitoring becomes quite a challenge. An error in the video stream directly affects the audio, so the two should not be thought of as completely independent.

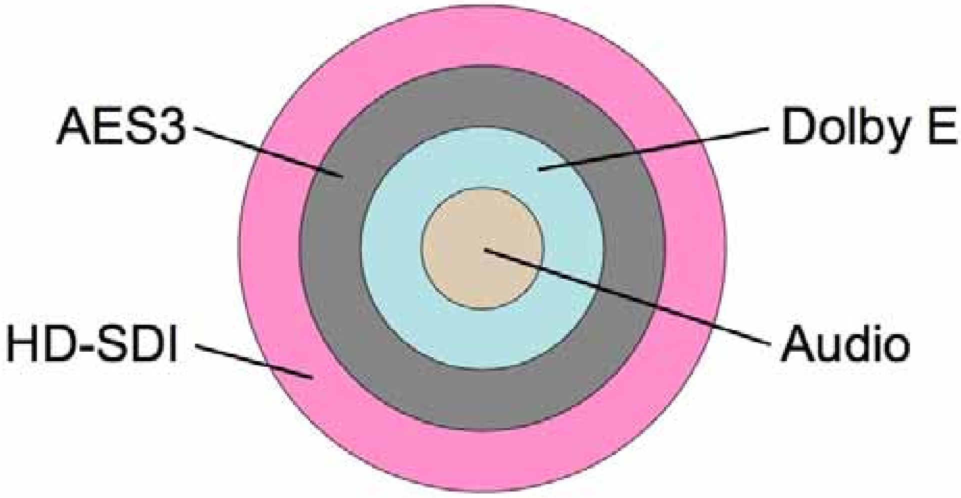

Figure 17 - The layers of a Dolby E system

6.2 Audio, Metadata and Down-mixes

Clearly as well as data integrity checks on the various data streams, one should listen to the audio itself. However even this is more complex than it may at first seem. Using a Dolby E decoder to produce the 6 discrete audio channels of a 5.1 mix may seem sufficient, but Dolby E decoders do not *emulate the metadata*. In other words, they produce the simple decoded audio, and do not apply the effects of dialnorm, DRC and so on. In order to experience the same audio as the consumer, one should use a device (such as the Dolby DP570 multichannel audio tool) which does emulate the metadata.

The stereo down-mix is just as important, as a very large number of viewers will listen this way. Once again, the effects of the metadata should be applied. In the case of programmes produced for both BBC HD and a standard definition BBC channel, a separate stereo mix is often still produced, so the down-mix and the separate stereo mix should sound as close to identical as possible. It is expected that in future, the stereo for SD channels will be derived from a down-mix of the 5.1, removing this complication.

It may also be wise to at least spot-check the Lt/Rt and Lo/Ro down-mixes separately, as viewers may experience either one depending on the set-top box. The mono mix is generally not checked because the process of down-mixing from stereo to mono is well known and reliable, and few people will listen to HD in mono. However it is certainly possible to do so, and this should be kept in mind.

The final component is off-air monitoring. In a live programme, the sound supervisor or someone on site should ideally have the ability to monitor the off-air audio and certainly technical areas such as CCA or playout should have this ability. It has become easier to implement more widespread off-air monitoring for BBC HD since the launch of *Freesat*, the UK's free-to-air satellite service, allowing monitoring to be implemented anywhere with a satellite feed simply by installing a set-top box. Previously, subscriptions to pay-tv services were required. The planned future launch of BBC HD on free-to-air digital terrestrial television will additionally improve the situation.

6.3 Monitoring Devices

For producing the various audio outputs to be monitored, the Dolby DP570 is a good solution. However simpler audio monitoring units (AMUs) from manufacturers such as TSL and Wohler can provide some of the functionality, including simple examination of basic metadata parameters. TSL's AMU2-HD is in use both in BBC Television Centre and Red Bee Media's playout suites for basic checks of dialnorm, program config and other metadata parameters.

Devices such as Tektronix's WFM7120 and WVR7120 with Dolby options provide thorough monitoring of the metadata, CRC errors and other properties of the Dolby E stream, AES3 carrier and HD-SDI, providing a single unit from which all the engineering-level settings can be checked. They can also provide decoded audio outputs, including metadata emulation. For these reasons they are in use in areas such as CCA and playout at the BBC. However they are designed to provide the maximum information to engineers, and as such do not provide the simpler at-a-glance type display of basic information (such as metadata) that production staff need. For these situations a simpler AMU is more appropriate. Tektronix provide a guide to monitoring surround sound, which is useful for anyone implementing such monitoring, though especially those using Tektronix devices.[11]

Dolby's DM100 analyser provides information about the Dolby E stream such as line position, Pa spacing, CRCs and more and their LM100 provides some information such as CRC errors and metadata parameters. This makes them good companions to HD-SDI waveform monitors which do not have inbuilt Dolby E decoding and monitoring. Off-air monitoring can be done with any HD set-top box.

Some work has been suggested in CCA on the integration of monitoring devices into existing broadcast control systems. The aim is to have a single screen which could be called up to display a selection of the most important metadata and audio parameters for an incoming stream (extracted from multiple monitoring devices if necessary) which would allow staff to quickly check for key problems in order to improve the line-up and test process.

7 Further Work

The BBC HD multichannel audio review has been largely successful in its aims of investigating current known problems, however to assume that the work is finished would be a mistake. Concerns still remain around some areas of the multichannel workflow, including multichannel line-up and Dolby E in file-based systems, while broadcasting developments on the horizon - including the impending arrival of HD on digital terrestrial television (DTT) - mean that there is much still to be done.

7.1 Dolby E and Multichannel Line-up

As explained in section 6, monitoring the many layers of a Dolby E installation is an extensive process. It is therefore desirable to have a line-up process for outside broadcasts and other contribution links which allows simple checking of as many basic tests as possible. Ideally, checks would include:

- Channels are in the correct order.
- Phase relationships are correct.
- A/V sync is correct.
- Metadata is being received.
- Metadata values are within expected ranges.
- The data streams are all intact (i.e. few CRC warnings on the Dolby E stream, the AES3 stream and the SDI).
- Dolby E is correctly aligned (line position is good).

All these checks are possible using current monitoring devices, but to execute them all is time-consuming. The BBC currently advocates the BLITS [12] line-up tones for surround, developed by Martin Black and Keith Lane of Sky. These tones are designed for operational line-up not engineering-level tests, but are an excellent way to ensure the channels are identified, the phase relationships are correct and more. If down-mixed with metadata emulation and examined on an AMU, BLITS can even identify the down-mix parameters specified in the metadata. However, it is an audio-only solution, and the lack of a video component means no sync tests are included, nor is any metadata testing. Separate sync tests such as Pro-Bel's VALID8 can be used for testing lip-sync, but an integrated solution would be the ideal.

The BBC's sync test signal (as described in 4.3) could be used for verifying synchronisation, but an integrated testing device which accepts a digital video signal with embedded audio would be necessary to make the process of measuring sync easy and quick; the current light-pen and analogue audio solution would not be sufficient. With commercial alternatives already available, the likelihood of this test emerging as another standard seems small. However, by building BLITS into other equipment which also provides sync testing or other facilities, a useful compromise could perhaps be reached. The BBC has discussed such possibilities with some equipment manufacturers, however no products exist currently. BLITS generating and testing equipment is already available from manufacturers so integrating this technology into other test signal generators and/or monitoring products is certainly technically feasible.

The other major point to consider is metadata checking. Previously, production staff have been seen to change a metadata parameter during line-up and ask the engineer at the receiving end to confirm the change. This ensures correct continuity of the metadata, but is again a laborious process. The most basic way to simplify this problem is to provide Dolby E encoded test signals, which make known metadata parameter changes at known times in the test signal. In this manner, the engineer can watch for the changes, ensuring they arrive correctly and are associated with the correct stream. A more complex solution might involve the test signal generator (TSG) controlling the metadata input of a Dolby E encoder such that the OB's own encoder is used to produce the Dolby E stream but with metadata chosen by the TSG when performing line-up.

Finally, a trade-off must be made between convenience and scope of the tests. An ideal test and line-up regime would work from end to end, starting at the camera and microphones so as to include those devices (as well as mixers etc) in the test. Using a TSG as described here will generally only test the link, but is quicker and easier to set up. Work is still ongoing at the BBC to decide which tests should be mandatory for all HD line-up, and which tests are required only if a fault is suspected. It is hoped that the future improvements in testing equipment will make it simpler to perform a useful series of checks on incoming signals than at the current time.

7.2 Dolby E in File-based Workflows

Currently, major non-linear edit systems do not natively support Dolby E. For many programmes, this is not a problem as the programme can be edited using uncompressed audio and then encoded to Dolby E for delivery. However, where material has been contributed using Dolby E (including material which was originally delivered live from an OB), a problem occurs. The Dolby E can be decoded and the material ingested with uncompressed audio, so that it can then be edited and finally encoded before delivery. However in this process, any metadata which was present with the incoming Dolby E stream is lost, which is far from ideal. In some cases within the BBC, this problem is overcome by manually entering metadata settings from the decoder into a database, where they can be stored until re-encoding takes place. Clearly this is a far from optimal solution, being labour intensive and prone to error. It is also only really suitable where one set of metadata applies to a whole video clip, which is currently the case for most if not all clips, but potentially the metadata could change more dynamically during a clip, making it very difficult to record using this method.

Software Dolby E decoders and encoders are starting to become available in the form of standalone applications and/or plugins for editing systems such as Pro-Tools and Final Cut Studio. These may allow the ingest of Dolby E without decoding, performing the decode in software and allowing the seamless storage of metadata. However this is an emerging field, so further work to integrate such processes into production workflows will be necessary. The BBC is currently investigating options including Neyrick SoundCode for Broadcast [13], which provides encoding and decoding of Dolby E in software. Other software available includes Minnetonka's SurCode [14].

7.3 DVB T2

The Digital Video Broadcasting for Terrestrial specification (version 2), along with video coding advancements (specifically MPEG4 H.264 or AVC), mean that the UK will have terrestrial broadcasts of high definition content soon. The specifications allow for the use of not only Dolby Digital but also High Efficiency Advanced Audio Coding (HE-AAC), so should the BBC or other broadcasters choose to use the latter for some or all broadcasts, the metadata requirements could change. Given the continued existence of Dolby Digital for satellite platforms, it would be unlikely that a major change in the production or distribution workflows would occur, however some kind of metadata translator may be needed in the coding and multiplex area, and the implications of any metadata differences will require careful study to ensure that the viewer experience across platforms remains the same. One solution is Dolby Pulse, a system from Dolby which combines HE-AAC coding with Dolby metadata.

8 Conclusions

The use of multichannel audio systems brings surround sound to the viewer, providing a better audio experience to complement the video improvements seen in HD. Technologies such as Dolby E allow the broadcaster to use existing infrastructure to provide both the extra audio channels and the metadata required for Dolby Digital broadcasts, however the challenges are considerable. System timing becomes even more important, and video synchronisation equipment may need to be chosen carefully to ensure Dolby E compatibility. Delays must be carefully considered to avoid audio-video synchronisation errors, and monitoring must be thorough to ensure audio problems are noticed and resolved quickly.

Audio metadata allows greater artistic control of audio output, as well as offering the consumer additional control of dynamic range in decoders with appropriate functionality. However with this control comes destructive power; incorrectly set metadata can cause incorrect loudness, sub-optimal down-mixes, or even a failure for the decoder to reproduce some channels of audio. As such, the broadcaster must ensure that the delivery chain is transparent to incoming metadata where appropriate, and also that the metadata produced for stereo programmes or that used in a reversion situation is chosen to best effect.

Guidance has been provided on many of the issues found by the BBC and some best practice has been shared on how to solve or avoid problems. However if one lesson has been learned, it is that a simple “plug and forget” attitude towards this technology is not enough. In an area such as HD which is still developing, a good understanding of the technologies used is necessary; equipment manufacturers, Dolby and others provide extensive documentation on the use of Dolby E, and the advice they provide should be heeded. With the necessary planning and testing, a broadcaster can implement a high definition infrastructure that provides viewers with a more engaging and exciting experience than ever before. In years to come, HD will undoubtedly become the norm, and the knowledge gained now should help the transition to be as smooth as possible.

9 Acknowledgements

The work described in this paper required input from a wide variety of individuals and organisations. The author would therefore like to thank those most closely involved; Andy Quested, Head of Technology for BBC HD commissioned and supported the work throughout, while many colleagues in BBC Research & Development provided assistance and expertise, especially Andrew Mason and Trevor Ware.

The BBC’s technology partners Red Bee Media (particularly David Godwin and Steve Scorer) and Siemens (particularly Jonathan Clinkscales and his colleagues) were helpful and supportive throughout, and have implemented many of the actions described here for the benefit of BBC HD. Additionally, input from BBC Studios and Post Production (especially Danny Popkin) was gratefully received. Sky’s Martin Black as well as representatives of ITV and Channel 4 have collaborated with the BBC on many of the issues described here, and Mr Black – along with Keith Lane - kindly make the BLITS tones available for use by the BBC and other broadcasters.

The detailed understanding of Dolby E was considerably aided by Mick Dwyer, Will Kerr and others from Dolby Laboratories, and Lee Ballinger of Tektronix provided significant information on monitoring Dolby E.

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Appendix I

Don’t Forget The Kitchen Sync

a blog post by Rowan de Pomerai, 29/11/2008

A couple of weeks ago I told you about the work we’ve been doing on the synchronisation of audio and video (lipsync) in our surround sound signal chain. However, no matter how much work we do, there’s one thing we can’t control, and that’s the equipment in your front room. You might not know this, but your shiny new flat-screen TV (LCD or plasma) introduces somewhere in the region of 40 to 100 milliseconds of delay, which means that if your audio isn’t delayed to match, the sync between the two is quite considerably wrong. Worse still, the audio is ahead of the video, which is much more noticeable than the sound being late.

Why? Well think back to your GCSE (or O-Level!) Physics lessons. Sound travels much slower than light, so you are used to hearing the sound of an event slightly after seeing it, particularly if it happens far away. Just try going to a gig in a stadium, and you’ll notice that if you’re at the far end of the stadium from the stage, the singer’s lips will be moving well before you hear the sound. So audio being late, whilst undesirable on TV, is much more tolerable than the video being late. The latter scenario looks very unnatural indeed with even quite a small error.

Most HD set-top boxes allow you to adjust the audio/video sync on the outputs, usually in 20ms steps. However just by watching TV programmes, it’s hard to judge when the sync is correct. As such, we wanted to do something to help our viewers to get their own setups correct – some of you have been asking for this in comments on our blog posts, and we suspect more will be interested in trying it when given the opportunity. So we’ve been devising a sync test which will be broadcast a few times during each day on BBC HD, starting in a week or two. The test signal is based on work from Andrew Mason, Oliver Haffenden, David Kirby and Alastair Bruce at BBC R&I, and it should allow you to adjust your set-top box to an acceptable level of sync just by watching the sequence and tweaking your settings.

If you want to get even more precise, you can make your own sync test device! The signal has been designed to be easy to monitor with a simple electronic circuit, using a light probe to sense a flash which appears on the screen once a second and an audio input to listen for a ‘clap’ which happens at the same time. The device can then tell you the time offset between the two and hence you can adjust your set-top box to compensate. Details will be posted on the BBC website. The ‘clap’, incidentally, is a recording of two bits of wood being snapped together; believe it or not we found this at least as effective as more ‘high-tech’ alternatives such as a brief burst of tone.

My job in all this has been to ensure that the test signal is in-sync when it gets broadcast, which is trickier than it sounds. Of course we aim to ensure that everything we transmit is in-sync however it’s absolutely impossible to ensure that sync is perfect, not least because the delays introduced by some equipment vary over time. Within a small tolerance of a few milliseconds, the difference is imperceptible anyway so it’s not a problem. The EBU defines standards for sync that say that when a programme is delivered to a broadcaster like ourselves, audio should be within +10 to -20ms (i.e. no more than 10ms ahead or 20ms behind the video). Of course, to ensure that the signal is within sensible tolerances when a programme actually gets broadcast, each step of the chain has to have sync errors much smaller than this, otherwise the delays could add up to a much larger a total error. When transmitting a signal whose express purpose is to be in-sync we’d like the sync to be even tighter than usual, which means making sure that every step of the chain is as close to ideal as it can be.

The signal started life on an edit suite, and so job number one is to ensure that when it comes off the edit suite and on to tape it stays in sync. Along the way we have the audio encoded to Dolby E, so we’ve already got 3 bits of equipment in play: the edit system, the Dolby encoder and the tape deck.

So we do that, check for any offset, correct it in the edit suite and then get back on to tape again. But wait… how do we check the sync? I'm armed with our sync-checker, but in order to use it, we have to decode the Dolby E and play the video out to a monitor, both of which could introduce their own delays! So we have to ensure we know the delays of each component in the chain, isolating them one by one, before we can rely on our measurements. Rule number 1 of testing is that you must ensure that your test equipment isn't affecting the results, or at least that the effect is known. In this case we can't test without having an effect, but isolating that effect allows us to get accurate measurements. It means we have to be really careful though – forget one source of sync error and you mess up all your results! That's my justification for the terrible pun that titles this post…

Then there's the signal chain to get the test sequence to air. We ingest the tape onto our playout servers (this process could, of course, introduce sync error), then play it out though a presentation mixer and some processing gear then down some fibre optics to Television Centre. From there it goes on through to the Coding and Multiplex centre where the signal is prepared for digital broadcast. All that could introduce a sync error, as could the coding process itself; the audio and video are coded separately and multiplexed back together, then multiplexed in with other channels for broadcast. Argh! So I'm going to work with staff from Red Bee and Siemens (our technology partners in these areas) to run the signal through the off-air chain, a backup set of equipment and connections used if the main signal chain goes down, and also in testing situations like this. We'll measure the sync error through this whole set of equipment, and if necessary adjust the offset at the encoders to get everything back into sync.

I'm fairly confident that there shouldn't be a big sync error in this signal chain, as it's been tested before and we're broadcasting with it every day, so major errors would be noticeable. However as I mentioned previously, our tolerances for broadcasting this test signal are much tighter than usual, so it will be a great chance to check that everything's working as well as it can possibly be. Hopefully all will go well, but you never know – I'll let you know next week!

Appendix II

A Christmas Present From The HD Channel

a blog post by Andy Quested, 17/12/2008

So, the channel is just over a year old - and what a year it's been! After looking back at some of the blogs and posts I can see how much we've done and worry about how much there's still to do.

Many of you have been asking for a test signal to help line up your own HD TVs, we have been listening but it's taken a while to get it sorted.

From this week the HD promo has two test signals and I want to talk about how to find them and how to use them to line up and check your home systems. I also wanted to share a fascinating mathematical proof that some people (Heroes style) can change the flow of time!

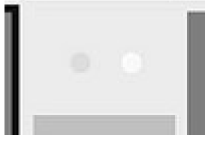
As many of you have noticed BBC test card has been going out for a couple of weeks, this has now been joined by an Audio/Video Sync test signal. The test card seems to have been given the name "Test Card X" but not by us, it is in fact a modified high definition version of test card W (named because it was widescreen!) and for those interested in the history of test cards, there is an interesting "romp" through it here - it even includes the current incarnation!

The HD version uses the very famous picture of Carole (George Hersee's daughter) re-scanned in high definition and added to an HD version of Richard Russell's well known widescreen test card.

Now for the purists there's a bit of a disappointment coming. No, not the fact the test card's only there for 90 seconds every two hours! Talking of that, I was with a group of people looking at the promo last week when the test card came up - they all said "does this have to up for so long" and "what's that noise on the sound track" I did attempt to explain how much it was wanted but it just made things worse! I said I had wanted 5 minutes and many of the posts had asked for up to 30 minutes - at that point I felt like I came from another planet and decided to get on with other things! But there is a test card going out and I hope we can all celebrate its reappearance after many years!

The disappointment is a technical one. I am going to admit I have doctored the test card - much to the disgust of many of my Research colleagues. Why? Two reasons actually. A high quality test signal like the HD test card is a very valuable asset and unlike the SD transmission chain the HD one is quite good and quite capable in the being "purloined"! Already some of the posts on digital spy have already gone into great detail with the exact measurements of the card.

This version of the test card can be easily identified as it's the only version with the HD DOG logo at the bottom. Now, I want no DOG posts in this blog, I will ignore them as the DOG debate goes on elsewhere.



What have I done and how useful is this version of the test card? First, white level has been reduced so the peak white box is not 100% (level 235 or 0.7v). The super white spot is now 100% and the linearity of the grey scale is now slightly inaccurate. However no domestic displays have the level of adjustment we expect a broadcast monitor to have, so I this does not affect the usefulness of the test card to help you line up a "normal" TV. Also the colour bars are slightly lower in colour level. My apologies go out to people like Richard Russell and all the others who made these test charts possible - but this does protect the value of the work.

The second reason is to help protect screens from burn in. The full level test card will burn a screen in quite a short period so please heed this warning:

DO NOT leave the test card on screen for more than 2 minutes if your screen is less than three months old or more than 5 minutes on older screens. Make sure you go back to the promo for several minutes before using the test card again.

If you want more detail of the changes there is a very good post on Digital Spy. If you do have a broadcast style display at home it is quite easy to calculate the offsets to apply to a colorimeter to make sure the readings are correct.

Now for a bit of an explanation about the test card and how to use it to line up your TV, I have done this at home so can say it does work. But before you start to line your set up please take note of the following:

1. Make sure you have the user manual and know where the controls are.

2. Do not do this if you are unsure of any of the controls or there effect on your television picture.

3. It is best to do this in a darkened room, it doesn't need to be completely dark but if it's too bright or there is a lot of light falling on the screen the results will not be good.

4. Many modern flat screen televisions have presets for sound and picture. Write down which one you use so if you get lost you can always go back and start again.

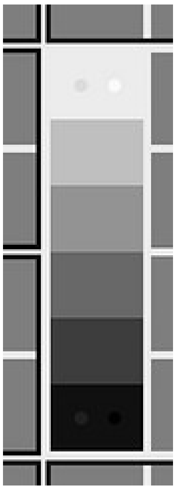
5. If you have a PVR it would be a good idea to record the test card section of the promo. Most of the line up can be carried out on a freeze frame of the test card. If you do this please be mindful of the warning above about screen burn.

6. If your TV has it, change the picture settings mode to "manual" or the equivalent, so any inactive controls become active allowing you to change the settings on the TV.

7. Turn the sharpness setting to off or zero. If there are any picture enhancing options, make sure they are turned off or to zero (if you can). Remember, on some TVs the sharpness control has a centre zero allowing you to soften pictures - please don't do that!

So to start:

BRIGHTNESS AND CONTRAST



There is a GREY SCALE to the left of the picture on the test card. It's there to show the correct black and white levels of the picture. Broadcast displays have the ability to adjust the grey level independently so there is a linear grey scale between the black and white blocks. I am not going into how to use this here if you are interested start with this.

The top white block has two spots. As I said earlier, usually the white block is peak white with the right spot higher (super white) and the left spot slightly lower. On our test card, the levels are slightly reduced.

The bottom black bock has two spots, the right hand one is below black level (sub black) and the left is slightly above black. The modifications to the test card have not change these levels.

BRIGHTNESS

To set the brightness:

1. Turn your brightness control up until you can see both spots.

2. Turn the brightness down until the sub-black spot disappears but make sure you can still see the left slightly brighter spot.

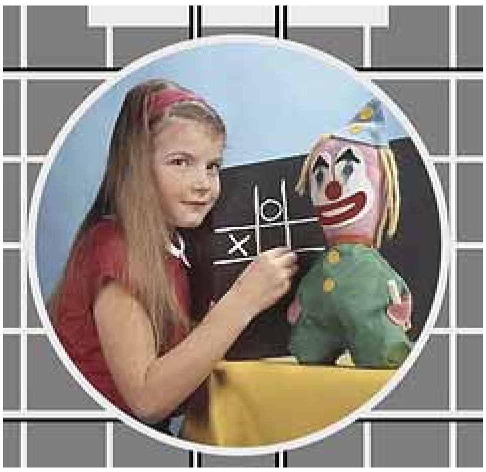
CONTRAST

When a broadcast monitor is lined-up properly, we use a meter to check the white level however on a domestic television contrast is more a matter of personal choice and will be different on different types of display (LCD, Plasma, Projector etc.)

Adjust the contrast until you like the overall look of the test card while you are doing this, keep an eye on the spots in the top white block to make sure you can still see the left hand one. It doesn't matter if you cannot see the super white spot so don't worry if it's not there.

COLOUR

Again colour level is very much down to personal taste but most TVs have too much of it! Too much colour makes pictures look very odd. It will also make some colours bleed into each other or appear to move so the colour smears over the edge of the object - in other words someone wearing bright colours clothes may have the colour slightly off to one side! The best bit of the test card to use to set colour is the picture of Carole.



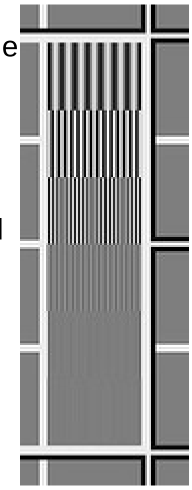
The centre of the test card has all you need to get the colour right. Carole's face should look natural and the primary colours in the picture (red dress and green and blue of the clown) should not be very bright. Colour is a subjective setting so just make sure you like it. Remember, if your colour setting was previously set very high you may not like the correct level until you get used to it!

One of the experts at BBC Research suggested another way to adjust colour level.

Get some Lee Lighting Filters No.181 Congo Blue and place it over the screen. This has the same effect as turning off the Green and Red leaving the Blue component of the picture. Looking at the colour bars around the edge to the test card, adjust the colour control until they all look the same brightness. There are some commercially available line up DVDs that use this method.

When you have adjusted the BRIGHTNESS, CONTRAST and COLOUR have a look at the promo again to see what you think. Watch it for some time so you get used to the new settings and see several different type of programme.

SHARPNESS



I have the sharpness control on my TV set to zero but some of you may want to add a little bit if the picture looks very soft.

To the right of the picture of Carole is a set of "frequency gratings" The frequencies are:

1. 5MHz 2. 10MHz 3. 15MHz 4. 20MHz 5. 25MHz 6. 30MHz

The BBC HD transmission system will pass frequencies 1 - 4. Most domestic displays will show 1-3 correctly but the 4th might not look quite right. A 50' 1080p display should be able to resolve the 4th grating satisfactorily.

PICTURE SIZE AND POSITION

Not all TVs offer menu setting that allow you to change picture size and position. Even if your TV does allow you to adjust size and position, it's not a good idea not to make anything but small changes unless you know what you are doing. Make a note of the current setting BEFORE you change anything!

Most displays lose a small amount of picture all round. This is called "overscan", it is perfectly normal and programmes have always been made taking this into account.

Some flat screens do have the option to either turn overscan off or reduce the picture size.

It is perfectly safe to use the "overscan off" option on you TV but you should not use the picture size controls for anything more than small changes.

The full test card should look like this, with the diamond points just touching the edge of the screen all the way round.

As a mater of interest, the cross on the Noughts and Crosses game is the centre of the picture!

You should now have a picture that looks fairly close to the one we see before transmission. Again watch some of the promo to get used to the new settings. Also if you have turned overscan off, you might want to look at some SD channels to make sure you don't see extra bit of the picture you don't like. You may see some white lines at the top of the screen on some News programmes for example. This happens when signals are brought back that don't fully meet the broadcast standards, but have to go to air too quickly (if not live) so it isn't possible to correct them.

AUDIO VIDEO SYNC

The second test signal is there to help you check and adjust audio/video synchronisation. AV sync has been the bane of my life ever since the test channel started. Remember we have rebuilt the HD Channel infrastructure round a service running 24 hours a day, 7 days a week so occasionally we have had to put new sections into service without being able to fully test them.

A couple of months ago, Rowan stated working with me to try and clear up our surround sound and AV sync problems. It has been a joy watching him dive headlong into the issues and some of you may have visited Rowan's blog to see how his work is progressing.

The last time I saw him, he was waist deep in diagrams of installations and programme signal routes to see if each video process had a suitable audio delay and each Dolby process had an appropriate video delay in circuit. The idea audio processing "takes time" is relatively new to television and we have to remind people to compensate for processing delay in the appropriate place so we can make sure everything is correct it if anything needs to be changed - I hope the message is getting through!

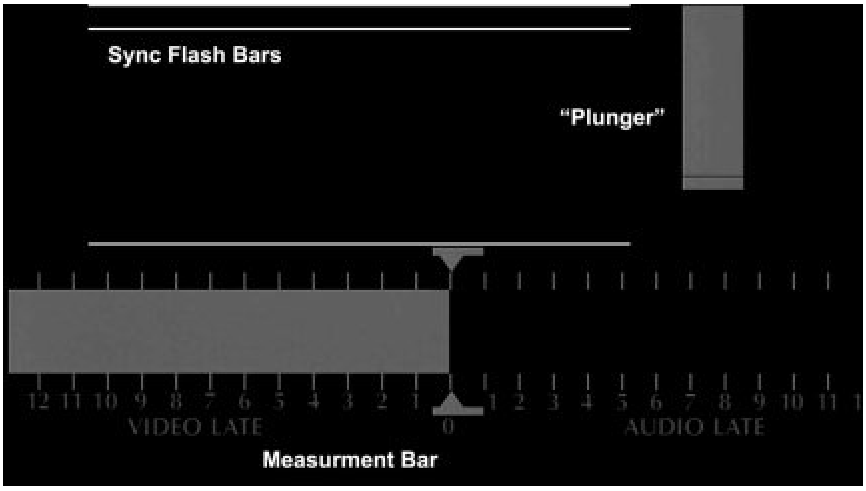
However even when we get it completely correct, some home setups can cause A/V sync problems of their own. The second test signal should help you check and adjust the sync timing of your AV system. This does not work on audio fed through the HDMI cable to the televisions own speakers. Any delays in that situation should be compensated for inside the TV. A/V sync is only adjustable when you use a AV systems connected by the optical/SPDIF output of a set top box or for AV amps that can us the HDMI output and have their own delay controls.

As some systems could have two ways to adjust AV sync - the set top boxes will have an audio delay option in the set up menus and good AV amplifiers may also have audio delay options, you need to start by setting all delays to zero. Again, please make a note of the settings BEFORE you start.

Why is there a need for A/V sync adjustment now? Most flat screen displays introduce a delay while they process the picture before it's displayed. Inside the TV the audio is delayed to match the processing delay but if you connect your set top box to an external audio system, the sound can be one or two frames ahead of the picture. In nature this is not normal and we can detect sound ahead of vision very quickly and it is "just not right"!

Our transmission system can also introduce delay to both audio and video signals. Some of the delay is obvious e.g. if we send the audio through a Dolby E decode/recode process, the sound is delayed by 2 frames so we must add a 2 frame video delay. Other process aren't so easy to check as the delay occurs inside a device that's processing audio and video together so the reason the A/V sync signal was not transmitted two weeks ago was to allow us time to test our whole system to make sure what we send you is actually in sync!

So it's time to introduce the BBC Research sync check signal...

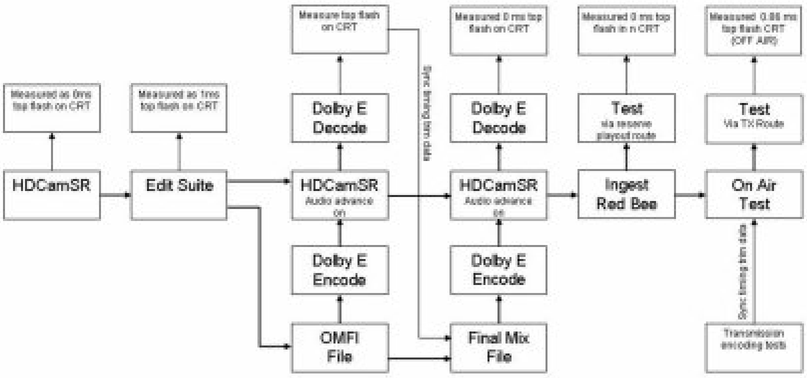


The audio is actually two blocks of wood being banged once a second - nothing to beat the real thing! The video is made up from three components:

1. A travelling bar marked in frames starting 12 frames before the audio clap and going on for 12 more frames after. 2. Three Sync Flash Lines. 3. A sync "plunger" or "clapper bar" (acting like a clapper board)

Before I tell you how to use the signal, you might like to know what we did to make sure the signal itself was synchronised and the transmission system did not put it out of sync. We - or I should say Rowan - have been measuring the signal at every point in the chain to make sure it is as accurate as possible when it arrives at your set top box.

This is the task I set Rowan to about three weeks ago!



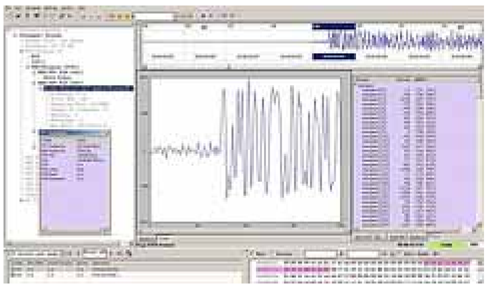
Why so complex? I needed to make sure the signal followed the routes many of our programmes do through post production and audio mixing, playout and transmission.

What is sync though? If you think about the speed of light vs. the speed of sound it's fairly obvious that sound arrives a lot later that the image of the "thing" making it. A rough rule is audio takes slightly less than 3ms to travel 1 metre so if you sit 2 and a half metres from your TV the audio takes nearly 7.5ms to reach you - nearly a quarter of a frame.

The effect of AV sync has been measured and tested quite extensively by the international broadcast standards bodies and we usually work to a tolerance of +20ms to -40ms (+half to - 1 frame) for a programme delivered to the BBC. This tolerance has been well tested in SD but there has not been enough work done to see if it's still OK in HD. To make sure there were no major surprises we have tightened the delivery specification to +10ms to -20ms while we do more tests.

In one of my previous blogs I explained that during the trial we found sync varied during a programme, especially live programmes, depending on how hard some of the early equipment was working. Now we are a lot more stable and have had a chance to go through the system from end to end to make sure it's sync. We have just finished testing the chain with an "off air" test of the signal and have a timing error of 0.86ms or 0.0125 of a frame!

Those of you who work or have worked in the business know the phrase "it's alright leaving me". A translation of that is "I'm OK, it's your problem"! To make sure you can use the signal to check your home system, we have gone one step further and made sure the signal is "alright arriving at you", not just "alright leaving me".



The final measurements of the off-air signal were made by looking at the digital signal from the transmitter - or in this case, the one received from the satellite.

This is what the test signal looks like in the transport stream from the satellite.

Rowan has a more detailed explanation of how and why we did this in his blog.

Part 1: Don't forget the kitchen sync

Part 2: Testing the test

How do you use the signal to check audio/video (AV) sync?

Remember this only works if you are connected to an external AV system. Check the audio delay setting in the set top box is 0ms and if your AV amplifier has a delay check that is set to zero.

Method 1

Look at the travelling bar at a point before the centre - look say at 10 on the left of screen. Listen to the clap and see if you think the bar has passed this point before you hear the sound. You might want to mask the right side with a bit of paper or put your finger on the number to help.

If the audio seems to happen after the bar has passed, move on a number and repeat until you think the audio and the point the bar passes your maker coincide. The sync point could be between two numbers but most devices only make corrections in half frame increments you will have to decide if you think it's closer to a number or closer to a half way position.

You should still be on the "video late" side of the zero mark. Read the number (or closest half number) and multiply by 40.

If your number is 3, the audio delay you need is 3 x 40 = 120ms. If your number is 1 and a half, the audio delay you need is 1.5 x 40 = 60ms

If your sync point is on the right hand side of the zero mark, I'm sorry to say there is nothing we can do to help. Before panicking - check you have no audio delay set then wait for the test signal to come round again.

Method 2

For those who like a challenge there is an electronic method. The white lines flash for 1 frame at the start of the audio waveform.

I am sure you can think of many ways to use this information to measure AV sync accurately, but here is a simple option for all of you with a dual beam oscilloscope, a photocell and a microphone lying around!

If you place the photocell over the top sync flash line and the microphone on one of your front speakers, connect them to separate input of the scope (with any amplification devices needed to boost or power them) you will get two spikes. Make sure your scope is configured to display the two traces at the same time and measure the difference in ms. Apply this delay to either the set top box or the AV system. If you have a very good AV system you may be able to get this exactly right instead of the nearest 20ms.

If you can't decide between two settings, it is always better to make the audio slightly late than have it in front of the pictures.

Why are there three sets of white flashes? The top line is the reference line, i.e. 0ms A/V offset when measured on an HD CRT. However sometime it is difficult to accurately measure the very top of active picture, possibly because the TV's casing gets in the way of the photocell. The second line is 1ms later (as measured on a CRT) and is usually easier to get an accurate reading from. The third line is in the centre of the active picture so should read 10ms A/V offset on a CRT.

On the various LCD and plasma displays we have tried this on, some show a difference between the three lines and some don't - not much help to you, but I would go for the second line if you can and minimise the delay there!

Please let me know how you get on.

How to record the BBC HD test signals

The HD test card is just over 1 hour into the promo and the AV sync signal is 50 minutes later. To record both signals, check the time the last programme finishes and add 1 hour. So if the last programme ends at 01:30 set your PVR to record from 02:25 to 02:45 for the test card and 03:15 to 03:35 for the AV sync signal.

My last thoughts this time are around phenomena I have discovered that manifests itself around my daughter and what used to be my phone. When she is on the phone I have discovered time slows down!

How do I know this? Simple maths!

Let's take a telephone billing period - call it Tm. A Tm can only have 4 values 28, 29 (every fourth year) 30 or 31. Each Tm is made up from telephone charging units, let's call them Tu.

Each Tu is charged at various rates but I am going to use the maximum UK rate £UK (premium numbers etc are barred).

So my maximum phone bill can only be Tu x £UK. As the Tm can have four values the total bill can vary between each Tm. So why have my bills been consistently two to three times this amount?

My thought are, as the rate £UK is fixed in any Tm (but in general is always rising) and the periods of charging Tm are fixed to one of four values the only thing that can change is the Tu! As this is measured in time, I can only conclude time around my daughter changes as she uses the phone. Oh! I forgot to say, we are on the free weekends and evenings tariff so this time dilation effects occurs in the brief period between the end of school and 18:00!

I think Douglas Adams spotted this change to the laws of maths in the Hitch Hikers Guide - other explanations welcome!

Have a very merry, in sync and well lined-up Christmas and we will speak again in the New Year.

Andy

P.S. When she is on the phone to us each Tu is very long period of time of course.

Appendix III

The BBC’s Bold Trial of Reverse Karaoke

a blog by Andy Quested, 17/07/2008

I'm sure that, like me, you've been to at least one Eurovision Song Contest party where you either had to dress up as one of the qualifiers, take a dish or bottle of the country you've been allocated, or been given a song sheet so you can sing along. This year, due to a technical fault, we had the ideal opportunity to try something new, so for around half an hour we sent the vocals for everyone to gather round the telly and provide their own instrumental accompaniment! From the comments in the many HD forums, I gather not many people took the opportunity, oh! well.

A couple of weeks ago, I was speaking to the technical director of Discovery: he said that over 75% of their quality control failures are due to surround sound problems.

Then you realise that Discovery are at least five years further down the HD road than usand they don't do that many live programmes! No excuse for getting it wrong, but it does put it into perspective.

So why and how does it go wrong and why isn't everything in surround?

Audio is a lot more complex than video and, for live programmes, the complexity is multiplied many times over. When we do live HD transmissions, the HD video feed goes directly to the HD channel and a down converted signal is sent to send to the SD channel for SD transmission. The SD channel has stereo audio, but we don't mix it down from the surround audio (yet); it is actually a separate mix at the moment.

It's worth remembering that well over 95% of the audience of any live programme is watching in standard definition with stereo sound and the standard definition audience must get the best possible service. This means, however, that we've just multiplied the problems by two.

On the day of the Song Contest, I had been providing my usual daughter taxi service and arrived home around 20:15. It did take a few moments to realise that something was wrong and that there should be a least a bit of music to go with the vocals.

My AV amp was saying 1.1 and this was a bit of a surprise because the transmission chain is supposed to be locked to 5.1 at the moment. This is another area for forum discussion and the BBC has been the subject of much derision, accusations that we don't know what we're doing and "what would Dolby say?" statements. So before going on with the Song Contest story, I'll give a bit of an explanation.

The HD channel only transmits one audio stream, and this is encoded either as 2.0 for stereo or 5.1 for surround. When we transmit a surround sound programme, the set-top box does a down-mix to stereo for anyone who listens via the analogue outputs or HDMIvia the TV speakers. We send metadata in the audio stream to control the mixdown. It's worth remembering the metadata as they will come back to haunt us later!

When the trial service launched, we only had a few surround sound programmes - but I noticed, as we switched between 2.0 and 5.1, there were a few clicks on the audio when I listened at home. We tracked these for a while to try and find out what was happening, but they weren't serious and I initially put it down to my cheap AV system. Also there were no comments from the audience and even DigitalSpy was quiet. Anyway, we had bigger audio issues to deal with at the time.

As we started to get more surround sound programmes and were switching between live, pre-recorded, surround and stereo in every direction, the clicks began to get a bit more annoying and there were a few forum comments, so now I knew others were hearing them too. We started doing some investigation and I even recorded every junction on my home set-top box over a weekend - sad or what? I had to programme it all manually as the guys at Sky wouldn't do an upgrade to allow me to record every programme junction, but not the programmes - advertisers' dream option, I thought!

Connecting my box to a few home cinema systems produced some very interesting results. Every box did something different - the more expensive ones actually muted the output for 2-3 seconds over each mode change; this was something we just couldn't tolerate. At the time, we were talking to Dolby about the problem, trying to find out whether our switching technology was at fault, but it looked like this was a home cinema system issue.

Dolby has also done some investigation and there seems to be a problem with the way some home cinema systems switch between surround modes. It's very similar to the aspect ratio switching issues we had in the early days of widescreen, i.e. unpredictable and variable between manufacturer and model. Dolby kindly sent us a release that explains the issues in more detail and we are looking at possible solutions to the problem. However, in the meantime we are staying locked to 5.1 - well, that's what I thought until the Song Contest came along!

I said earlier, I was distracted by other audio problems. Domestic audio systems use something called Dolby Digital or Dolby D to receive and decode multi-channel audio. We use another Dolby format, Dolby E, to carry up to 8 audio tracks in the space of a stereo pair. It's a very useful compression system, especially for outside broadcasts where there's limited capacity on the link back to BBC Television Centre. The Dolby E stream also carries all the metadata the sound supervisor has added to control levels and the stereo down mix. We can pass this onto the Dolby D encoder and then on to the decoder in a Dolby home cinema system or set-top box.

Audio signals are usually embedded into the video stream now, and Dolby E uses virtually every available bit in a PCM stereo pair - sending any measuring devise straight into the red zone! Because of this, every device we use to either pass or process the signal has to be properly aligned, timed and to carry the signal as specified by the standards bodies.

During the first summer of the trial, we had been trying surround sound on different events - not only to get experience capturing, but also to try moving the signal around our infrastructure. Wimbledon had been a great success, and we were looking forward to the Last Night Of The Proms being the climax to a summer of live HD.

The way we get signals back to TVC is now critical not just for the quality of the video but also for the successful delivery of surround audio. During Wimbledon, we used a fibre link and, if you remember my last post, our issues were all about getting the fibre into the switching centre and were nothing to do with the quality of the signal it carried! To get signals back from the Royal Albert Hall, we used a microwave link to the PO Tower and then our permanent connections back to TVC. Not much of a problem - we've been doing it in SD for years and it only needs a few more Mbs for the video and an additional Dolby E stream for the surround sound. What we discovered during tests, though, was that the video and the stereo went in and popped out the other end with no problems but the Dolby stream just didn't go anywhere.

It was during these tests my Sky programme guide dropped through the letterbox at home, and it was advertising (on the same night as Last Night Of The Proms, no less), a live HD concert by a beat combo led by someone called "Robbie Williams". This was to come from Manchester, if I remember correctly, and I would have laid money that it would be in surround and the whole package would come down to Osterley via a satellite link.

After a few phone calls and a couple of beers, I discover that just like us, the Dolby E stream was going into Sky's sat link and resolutely refusing to come out at the other end! Both of us had identified the broadcast MPEG2 encoder as the culprit but we were not getting anywhere with a solution. Needless to say, we pushed very hard and a few days before the Saturday night, an updated version of the firmware arrived and the surround burst out of the other end of the chain.

By now (I hope), you have forgotten what this post is all about! But just in case you still want an explanation about the Song Contest... At 20:20, I was on the phone to the duty engineers asking why I hadn't been told we were doing a karaoke trial! They were very well aware of what was going on and were desperately trying to find the problem, but they did put me through to the HD network director and we decided to go immediately to an up-converted BBC ONE feed - this gave us properly mixed stereo audio, but of course we lost the HD. I said I would call the engineers back in ten minutes to see how the fault-finding was going. [Photo by johnthurmon Flickr]

Now, the next question you want to know is: "Why couldn't you take the HD pictures with stereo sound?". For that, you'll need to read on...

At 21:00, I was talking to the Red Bee duty engineers who had traced the audio back through the chain, but could find nothing wrong. It is worth a bit of time to tell you how the Eurovision Song Contest was passed through the chain from the venue to you.

The UK is a voting member so we have to feed ourselves back to the venue for the voting section of the programme as well as mixing Terry into the main clean feed coming in. To do this, we use a studio gallery. The engineers had traced the audio all the way back to the studio and listened to the direct feed from the venue, and at every point it was all there and correctly mixed. Everyone at every monitoring point could hear perfectly mixed surround sound and because of this, everyone was convinced it was a problem with the Dolby-E-to-Dolby-D conversion - the very last process in the chain. Meanwhile, the studio team (the very first point in the chain) was sure that all the settings were correct and that they could decode the Dolby E properly to prove it - but the Dolby D encoders were telling us we were sending 1.1 and nothing we could say would convince them otherwise!

Time (and contestants) were passing fast, so it was time for a change of tack and I am not one to believe the obvious - especially when two devices are saying something different to everyone else!

Earlier, I asked you to remember metadata, and by now I was thinking metadata not audio. I went back through the chain, but this time asked about the metadata that were nothing to do with operations (eg mix-down settings, dial-norm, etc).

I was getting grief all through this, as my daughter had now decided the Song Contest was not on her viewing list for a Saturday night and how she was to call several boys she had promised to call back if I was on the phone and she had to use the landline as I had deliberately let her mobile credit run out!

Anyway, and despite this going on in the other ear, several areas reported an inconsistency in some of the set up metadata readings but were not sure if it was a fault or a misreading as the results were different when monitored in different areas.

Before we did anything else, I needed to know what the actual metadata settings were leaving TVC and I also need to check lip sync because each Dolby E process delays the audio by 1 frame and we have to compensate by delaying the stereo and video to match. We had been checking and changing so much, by now there was no guarantee even if we did correct the fault anything useful would come out at home!

Lip sync is probably the second biggest issue we've had during the last two years. Each device or process that a signal passes through introduces some delay from a few lines or microseconds up to several frames. This is one of the reasons embedded audio is so useful but even embedders and de-embedders introduce delay.

From the comments on some of the forums, you would think we had people sitting back watching out-of-sync feeds and doing nothing about it. We do take sync very seriously and every sync complaint is always investigated. We have an off-air recording of the HD channel so that we can go back and check if we need to. But what is sync really?

A while ago, C4, five, ITV, Sky, Virgin and the BBC met and agreed how we would define sync. "Easy", you say - "lips and voice should match!". Balls should hit rackets; boots should hit footballs etc at the right time! Not quite as easy if one of your signals is encoded and it takes one frame to decode it to see whether it's sync. Also, in a chain that has known video and audio delays, do you add all these together and compensate at the front of the chain, or at the back? Or, as we decided, do you insist that all audio is sync with the video as it leaves each area, no matter how it's processed?

We define sync as "in sync encoded" for Dolby E signals, so if you cut an embedded stream on a frame boundary, the Dolby E and the PCM stereo will cut in exactly the same place in the dialogue.

Each area is responsible for correctly aligning signals to meet this requirement, including any material recorded to or played back from tape.

One other thing we've noticed is sync drifts in some devices depending on how hard the device is working. In a complex chain, there could be ten or more processing units and if they're all working flat out, a signal path that was in sync during line up can suddenly be noticeably out of sync. Also, it's worth noting that you probably have to include how hard the home set-top box is working too when looking at end to end AV sync issues!

The EBU and other standards bodies are looking into sync specifications to see if the current standards are good enough for HD. We hope to see a tightening of the AV tolerance in all equipment soon.

Back to the Song Contest! To check exactly what our current state was, I asked the network director to switch back to the HD output so I could see what my AV amp did. That's why some of the forums reported a few seconds HD with mono audio! The metadata were still saying 1.1 on one of my boxes and 1.0 on another (please don't ask why I have two).

Now I had someone in the studio looking at the main Dolby E encoder, I was looking at a set up guide on the Dolby website (no, I don't have manuals at home) and I was talking through the settings I wanted to check. I really wanted them to check what the Dolby D metadata setting was, but no one could find it in the menu. Then one engineer asked if it could be the setting "AC3 metadata" - same thing I said, "What's it set to?". "Disable", was the reply - a quick change to "enable" and a phone chase back up the chain where everyone reported consistent metadata reading the Dolby D encoder said 5.1. The rest was easy, but the paperwork was hell.

So why did it ever get to air, what was wrong and why couldn't we do HD video with stereo audio?

Well, to start with, there was nothing wrong with the signal or the metadata and because of that the system did exactly what it as told to do - what was that? Simply put, we had told the system to ignore everything we told it to do! The embedded audio and metadata were not faulty, so the very simple system we put in place for the trial would not allow a non-standard change.

During the trial, we had to put in place a simple setup, but it was very well engineered to do a very simple job without intervention. Now we've been given permission for the HD Channel, we can spend the money needed to upgrade the system and to put in place all of the safeguards, monitoring and protection that a full channel requires. This has been going on around the "live" channel, and one of my first thoughts had been that some part of the upgrade was interfering with the main output.

The target for completion (including testing and acceptance) of the new infrastructure is the opening of the Olympics in August, which I gather is 8:08:08 pm on the 8/08/08! But we have just about got everything in place now, so from now on, we have the same type of transmission resilience in the chain as any other BBC channel. We also have the capacity to carry out the testing and trial we need to do whilst still on air.

It's my sincere hope that we are at the end of the first stage of our HD migration and that we can start to consolidate and to improve. So maybe this is not the sort of post you expected, but I hope it's given a bit more of the background of what we are up to and I would welcome comment or suggestions for what to do next.