In its work, the SMPTE-EBU Task Force has determined to use an object model for control. A similar model follows for content. "Object model" implies that all devices have an address, and the Task Force is working on a higher level to develop the specifications and standards for this model.

The group determined that it should work to identify equipment that needs to be modeled, and to identify the functionality that will require control and appropriate protocols for use with these devices. It is understood that present equipment manufacturers are currently working at lower level, and several protocols exist to communicate among devices, often in a proprietary manner. The group's efforts were directed towards identifying a level of commonality, which must exist among devices to enable common control over them.

Some devices will require time-aware control, where rapid communication and response is a necessity. Higher level control will be less time-significant, where information may be provided well ahead of activation. The control model was consequently divided into three temporal levels of control:

1. Level 1 – Systems dissociated from time, which communicate to time-aware devices (time frame of minutes or greater)
2. Level 2 –Time-aware devices (time frame of one video frame, or so)
3. Level 3 –Instantaneous response devices (time frame of microseconds).
4. Each class of device at any level will have its own set of attributes, and manufacturers may choose to implement proprietary interfaces among devices in any level.

See Appendix 4 for currently identified devices and aspects which require control.

Issues Pending

This group expects that over the next 12-18 months there will be proprietary solutions and perhaps an approach of using a monitor/keyboard per device until integrated approaches are fully identified and standardized. The group agrees that object models are preferred for implementation as described in the report of the Joint SMPTE / EBU Task Force.

Action Items

Further work is needed to identify devices and their attributes in the "Baseband" and "Input" classes. The attributes of all these devices then should be placed into an object model for control. These elements, devices and their control requirements then need to be integrated into a schedule and resource manager for high-level control.

Timing Plane

Frame Rate

For the foreseeable future, stations will have an obligation to continue NTSC service. In the near term, this service will provide a preponderance of the station's revenue. It is impractical to operate a single facility at 59.94 Hz and 60.00 Hz-based frame rates simultaneously. Therefore, IS expects that all broadcast facilities will operate at 59.94 Hz-based frame rates, and they will expect program material to be supplied to them at these rates.

Reference Frequency

At a base frame rate of 59.94 Hz, drop-frame time code is used. The drop-frame algorithm provides first-and second-order correction of frame numbering to clock time, which was judged sufficient for editing and program timing when the algorithm was developed. However, there is a residual error of slightly more than 2-1/2 frames per day.

For some time, numerous parties have expressed an interest in adding a third-order correction to remove this residual error. There is general consensus within SMPTE, the organization which developed the time code standard, on how this should be achieved. Briefly, this involves changing the algorithm to drop an additional 2-frame group once a day, and changing the reference subcarrier frequency by about 0.8 Hz. Together, these equalize drop-frame time and clock time over a 24-hour period, so daily jam-syncing would no longer be required. The slight shift in the reference subcarrier frequency would, of course, imply equivalent changes in the reference frequencies used by MPEG, principally the 27 MHz clock. However, the shift would be well within the tolerances specified.

Changing the frequency reference is not a step to be taken lightly. However, it is the consensus within SMPTE that the increasing use of automation and the advantages of having clock time and time code track without discontinuities are so compelling as to outweigh any shortcomings of the slight deviation from nominal frequency. The timing group discussed the pros and cons, but did not take a position pending further input, including the details of the modification to the drop-frame algorithm which were not available for discussion.

Reference Signals

SMPTE is in the process of revising RP-154, its reference signal specification, and upgrading it to a standard. The revised reference signal will be analog NTSC blackburst with vertical interval time code and a 5-frame count for deriving or synchronizing signals with a 5-frame periodicity with respect to 59.94 Hz. The group endorsed this approach and urges SMPTE to complete this work as rapidly as possible.

Timing information derived from the station reference must feed not only traditional television equipment, but also the station's data systems. Data systems normally distribute time via NTP, so provision to generate this should be included in the station's master sync and time distribution system. This is essential in order to keep time-dependent data in sync with audio and video. See Attachment F for timing critical elements.

Audio/Video Synchronization

There was considerable discussion of methods to keep audio and video in lip sync. There are two approaches. The first is at each processing block to delay the unprocessed component, audio or video, to match the processing latency of the other. The second is to timestamp all components at a point where they are known to be in sync, and to resolve any timing differentials at point of use by buffering as necessary. The advantage of the former is that it does not require any passing of timing data between processing steps. The advantage of the latter is it minimizes the amount of buffering necessary, helpful not only in its own right but also because delays at HD resolution are expensive.

Analog audio and uncompressed AES-3 digital audio cannot convey timestamps, so the first approach outlined above will be necessary for systems which employ them. Compressed audio systems, however, can include timestamps, and it is recommended that this be done. All video formats, analog or digital, compressed or not, can be timestamped, and it is recommended that this be done as well. Use of analog audio is discouraged. In any case, it is not expected that analog audio will be employed for systems with more than two discrete channels.

Common Reference

Frame synchronization will be possible in digital TV systems, but may have unintended consequences in the event of synchronizer buffer over- or underflow. For example, in the event of buffer overflow, a frame must be dropped, and it would require extra care to ensure that this is not a decoded I-frame. Upon subsequent re-encoding, loss of the I-frame could cause a noticeable transient change in the quality of the received signal. Other adverse effects could occur with repeat or loss of a frame's worth of audio.

It is now practical and inexpensive to use the Global Positioning system (GPS) as a common, accurate time reference. Although the instantaneous accuracy of GPS is degraded intentionally by dithering, averaging over several hours from a fixed location yields a clock which has the same intrinsic accuracy of the base GPS system, which is that of a cesium clock regularly corrected to UTC. Stations will be required to have an external standard time reference in any case in order to generate PSIP. It is strongly urged that all facilities, both stations and program originators such as networks and live syndicators, lock to GPS. Elastic buffering would still be required at the station to pull incoming signals back into frame phase, to compensate for satellite wobble, and to allow for GPS outages or failures; but buffer over- and underflow would be eliminated. Note that in order for this to work, there must be universal acceptance of the shift of the reference frequency as described above.

Cueing – Mix-Minus and IFB

Humans cannot contend with hearing their own voices returned to them with more than a few tenths of a second delay. Until now, local stations have not had to deal with this, because recovered audio from an off-air receiver has been well within this limit and has been entirely suitable for cueing on-air talent in the field.

With the advent of ATSC television, this will no longer be the case. The combined latency of the emission coding and decoding equipment may be a second or more. Not only will the talent not be able to use the program audio for cueing, the video will no longer be reliable as an indication of precisely when they are on the air.

National program providers have had to contend with these issues for years, and have developed techniques for dealing with them. Foremost among these is the dedicated IFB, or program interrupt. This provides a means to get a feed of the audio program, less the talent's own voice (a "mix-minus") to the talent, with provision for production personnel to interrupt the feed to cue him or her. Larger stations, particularly those who do a lot of live outside broadcast, are already aware of these requirements and may be equipped to provide them. Stations in smaller markets may need to be educated. Moreover, communications channels back to field talent will be required. In the simulcast scenario, of course, the existing NTSC facilities will serve, but after the demise of NTSC other means will be necessary (broadcast auxiliary, cell phone, etc.) Multiplexing of private audio within the 8-VSB signal will probably not be workable because of the necessary characteristics of an IFB receiver (wearable, omnidirectional antenna, usable beyond range of station).

Issues Pending

Assess the desirability of changing the station reference subcarrier frequency.

Monitor Plane

Introduction

The sub-group's goal was to examine the Top-Down system map and identify monitoring, test, and signal injection points and associated equipment to support DTV station operation and maintenance.

Methodology

The Monitoring Sub-Group reviewed the Top Down system map in order to determine and/or refine the monitoring/test points, to conduct a reality check to determine practicality and usefulness of the points, to determine which points should be used for operational testing and/or maintenance/test, and to determine which points should serve as test signal injection points.

Findings

All signals and possible test points were considered in this examination, including satellite signals, 'baseband' signals, and emission signals. The following is a summary of the various test points and signals, and test conditions as recommended by the sub-group. Through the deliberations we determined that manual and automated operations will continue as today, but in many cases with new test signals, devices and test points as may be needed with new systems. One should refer to the Monitoring Layer Diagram (Attachment G) when reviewing the following findings. The numbers in parenthesis refer to the test point locations noted in diamonds on the Monitoring Layer diagram.

Satellite signal monitoring will largely remain as the current traditional testing and measurement practices. Included will be test points at the output of the LNB (1) fed from the satellite dish using a spectrum analyzer and/or test receiver. Under normal conditions this is not an operational test point nor is a test signal injected here point.

A test point (2) could exist at the output of the satellite receiver (labeled DEMOD in the drawing) with appropriate bit error rate test equipment for the satellite transmission format employed (e.g., MPEG transport stream, other systems which assume little or no knowledge of the payload with the transmission link, DS-3, etc.). This is an operational test point, and test signals may be injected at this test point as appropriate for the device following.

Test point (3) requires a packetized data analyzer as appropriate to the bits stream at that point (e.g., in the case of an MPEG-2 Transport Stream delivered via satellite an MPEG-2 Transport Stream Analyzer would be appropriate, in the case of a DS-3 signal a DS-3 data stream analyzer would be appropriate). Audio/Video time stamp differential analysis may also be measured at this point. This test point is repeated in both the standard definition section (top) and the high definition section (bottom).