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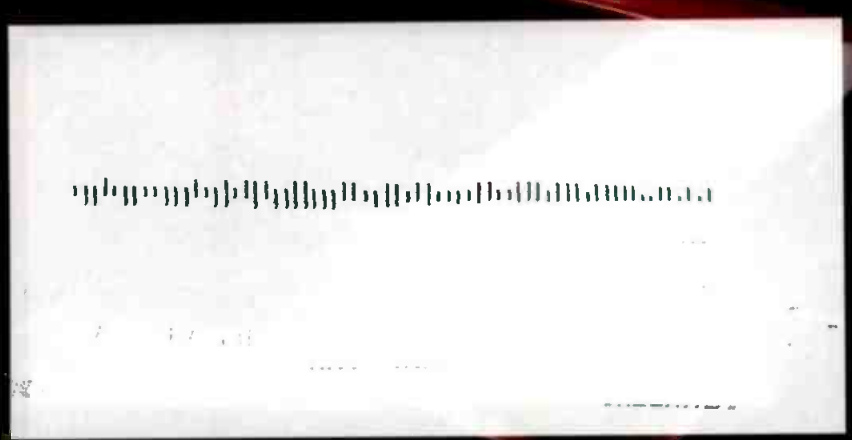

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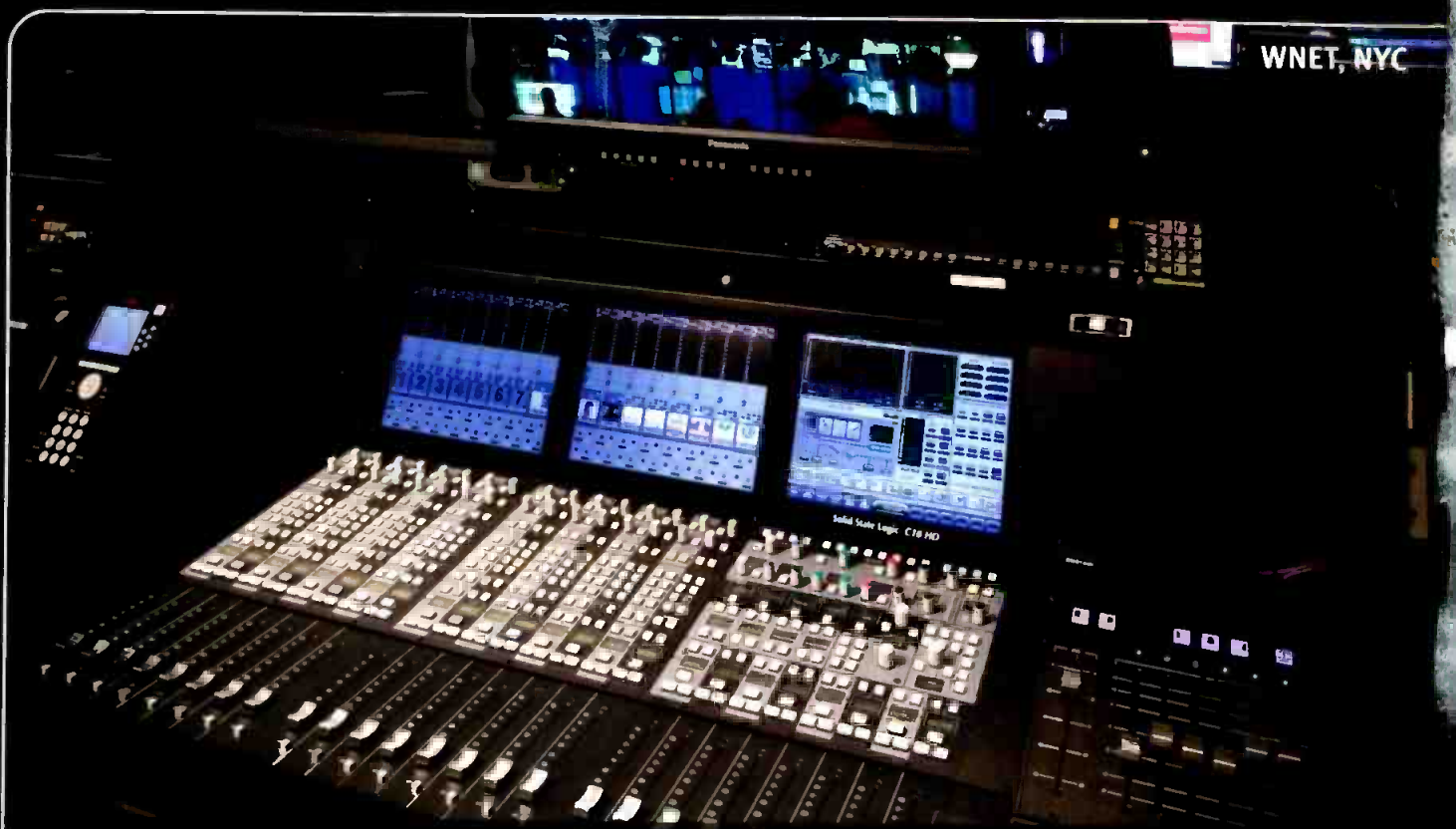

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TABLE OF CONTENTS

VOLUME 54 | NUMBER 9 | SEPTEMBER 2012

Broadcast Engineering®

SPECIAL REPORT

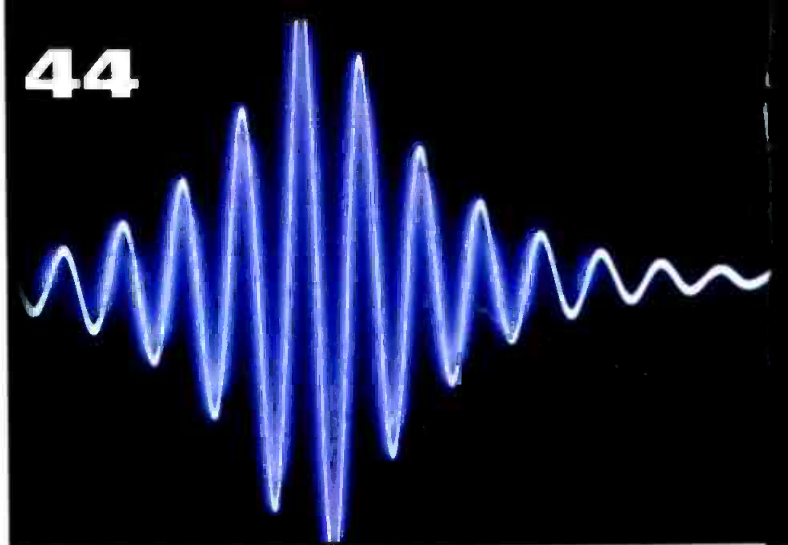
BEYOND THE HEADLINES

DIGITAL HANDBOOK

Continued on page 6

32

ON THE COVER:
In its new audio suite, WRAL-TV in Raleigh, NC, employs a Wheatstone D-10, a 5.1 surround television audio console. Photo courtesy Bill Burch, WRAL.



44

57

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TABLE OF CONTENTS (CONTINUED)

VOLUME 54 | NUMBER 9 | SEPTEMBER 2012

SYSTEMS INTEGRATION

NEW MEDIA NETWORKS

NEW PRODUCTS & REVIEWS

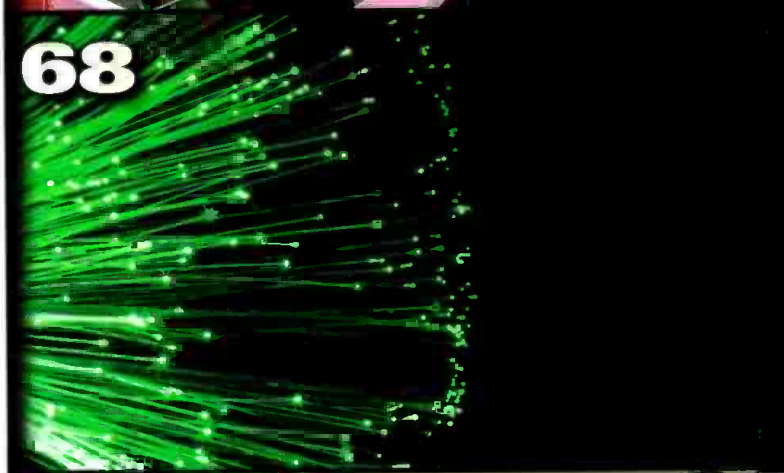
FIELD REPORTS

APPLIED TECHNOLOGIES

TECHNOLOGY IN TRANSITION

NEW PRODUCTS

DEPARTMENTS



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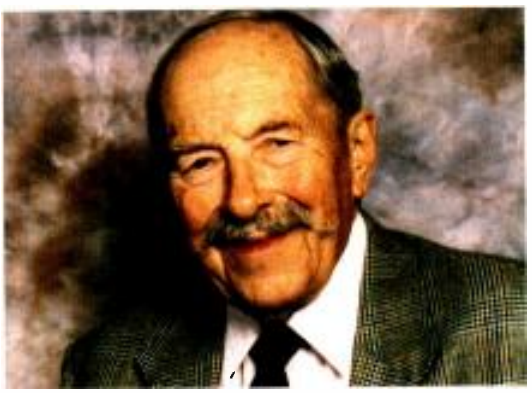
Remembering John Battison

We recently lost a good friend and fellow engineer. Long-time broadcast consultant and *Broadcast Engineering* writer John Battison passed away on Aug. 28, 2012. Just shy of turning 97, John spent more than 20 years writing for this publication and its sister version, *Radio* magazine. He retired from writing in 2009.

Battison worked as a broadcast consultant for more than 52 years. He was the author of 15 technical books and more than 500 articles.

In 1998, Battison was awarded NAB's Radio Engineer of the Year Award. Additionally, in 2006, John received the Society of Broadcast Engineers Lifetime Achievement Award. And, in 2002, he received an SBE award for Best Technical Article.



John Battison, broadcast consultant, writer for *Broadcast Engineering* magazine, founder and first president of the SBE
September 1915 – August 2012

Those awards might not be a surprise because many knew of John's life-long passion for broadcast and sharing his knowledge with others. But, many readers may not realize that he also was the founder of the SBE. In the December 1961 issue of *Broadcast Engineering*, John suggested the formation of a society for radio and television engineers, which became the SBE.

"Possibly a new institute is needed for broadcast engineers, one started in just the same way as the IRE [Institute of Radio Engineers] was originally," he wrote. "Perhaps it should be called the IBE, Institute of Broadcast Engineers, and be presided over by one of the great broadcast engineers of a few years ago."

With help from his family, Battison sent membership invitation letters to thousands of television and radio engineers in the United States and Canada. Based on the feedback from *Broadcast Engineering* readers, Battison decided to start the organization himself, running an application form in the magazine.

And so began the steps to build what we know today as the Society of Broadcast Engineers, with John as its first member and president. As witness of his efforts, SBE today has more than 5500 members from around the world.

"The Society of Broadcast Engineers was conceived in my office in Washington, D.C., in 1961," Battison reminisced during a 2004 SBE speech that celebrated the SBE's 40th anniversary. "It was born in 1964 in The Willard Room 'C' in the Chicago Hilton, courtesy of NAB, and today, on its 40th birthday, it is a strong and hearty force in broadcast engineering thanks to the members who followed after me."

John's many friends will recall his jovial personality and radiant smile, ever ready with an "I remember when ..." story. At the drop of a hat, John would regale friends with stories of piloting planes in WWII or building radio and TV facilities, sometimes in desolate locations around the world. He called AM "ancient modulation," and FM was the "forgotten medium."

In his 1998 NAB Engineer of the Year Award acceptance speech, John joked that he knew personally many of the fathers of broadcast electronics, including Lee de Forest and Major Armstrong. Sometimes, he'd remind the listener, "Speaking as an RF engineer, I still maintain that audio is something that messes up a nice, clean carrier."

John's "carrier" is now quiet, and he will be greatly missed. BE

Brad Dick

EDITORIAL DIRECTOR

Send comments to: editor@broadcastengineering.com

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Comparing JPEG 2000 and MPEG

JPEG 2000 excels in contribution applications.

BY DR. HELGE STEPHANSEN

Back in 1988, the first work began on video compression standards that would ultimately result in MPEG. The Motion Picture Experts Group, in cooperation with the International Organization for Standardization (ISO), created multiple standards for video compression, now known as MPEG-1, MPEG-2 and MPEG-4/AVC. MPEG was devised as a means of compressing broadband video into a small bit stream that could fit in extremely narrow broadcast or satellite transmission channels. As such, it is well-suited for distribution purposes, not for applications that require internal switching and processing.

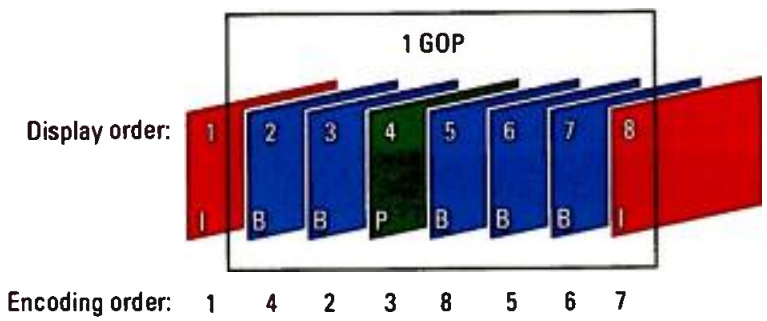

Figure 1. MPEG GOP structure

MPEG

Because MPEG is a motion image compression technology, it works on a sequence of video frames, known as a group of pictures (GOP). (See Figure 1.) A processor examines several frames of video and assigns one frame as the reference frame for that group (the I-frame). The GOP also has several predictive frames, or P frames, which use information from the I frame and previous P frames to construct images. Finally, there are bidirectional predictive frames, or B frames, which look at preceding and following I and P frames. Motion is analyzed and the motion vectors, which predict the offset from the current frame to the reference frame, are estimated down to a quarter pixel. The motion vectors and the difference between the actual images are used to reconstruct video that looks good at low bit rates.

Problems with MPEG encoding arise when there is a scene change or where a large percentage of the image changes at once. The processor can't predict the movement, and the system creates a distortion known as macroblocking. (See Figure 2.) Raising the bit rate adds more detail in each 8 x 8 block of pixels, which can

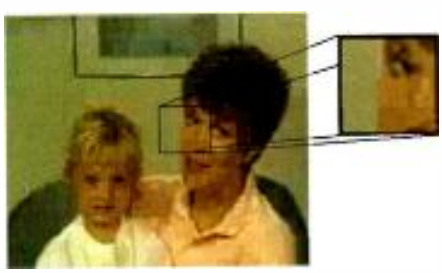

Figure 2. Macroblocking effects

FRAME GRAB *A look at the issues driving today's technology*

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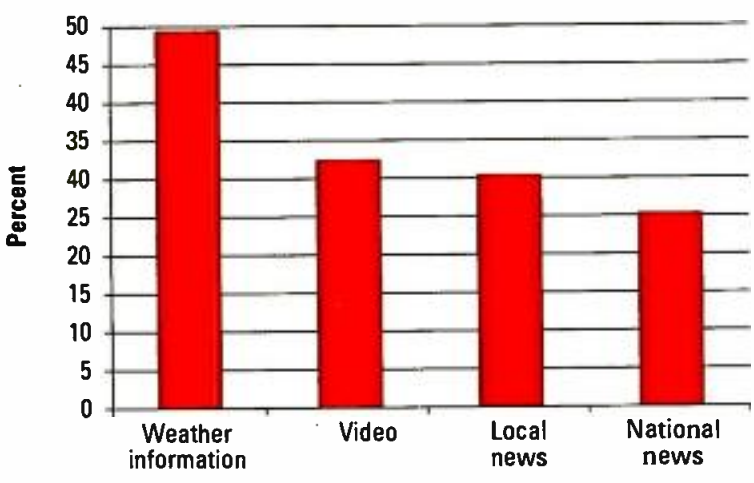

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reduce the number of P frames. But there will always be P frames — and the potential for macroblocking — in all implementations of MPEG.

Another major failing of MPEG compression is that it is difficult to edit and switch cleanly. With P and B frames, editing systems and switchers have trouble finding a clean frame on which to edit. MPEG also doesn't react well to being repeatedly encoded and decoded. It works best when the video is encoded once and decoded at the viewing location.

	MPEG-4/AVC	JPEG 2000
Quality (image to image)	Varying	Same
Latency	High	Low
Visual impairments	Blocking	Blur
Operating point for HD video format	15Mb/s-60Mb/s	60Mb/s-125Mb/s

Table 1. JPEG 2000 and MPEG-4/AVC attributes for broadcast contribution

JPEG 2000

Meanwhile, JPEG — short for Joint Photographic Experts Group — was originally developed as a compression standard for still images. With modern processors, however, it is now possible to take a video signal and save each frame as a separate JPEG file, creating a video format known as Motion-JPEG.

Several years after the JPEG team created its first compression scheme, it reconvened to develop an even more efficient and flexible system. Released in 2000, its new JPEG compression was called JPEG 2000. The original JPEG used discrete cosine transform (DCT), the same techniques used in MPEG, to reduce data into blocks of 8 x 8 pixels. JPEG 2000 uses discrete wavelet transform (DWT), which performs simultaneous multiresolution image analysis.

The resulting wavelet coefficients are gathered in sub-bands, each of which is partitioned into small "code blocks" and independently coded by an adaptive binary arithmetic encoder. Finally, the output of the arithmetic encoder is organized as a compressed bit stream that offers a significant degree of flexibility.

Wavelet compression provides the same processing for all pixels, and with each video frame being compressed individually, there is no error propagation from one picture to the next. As a result, there is no macroblocking at low bit rates; instead, there is blurring on the images. JPEG works on a single image, stripping away redundant data and encoding each video frame independently for consistent high-quality images. As a result, editing may be performed on any frame in a sequence of images because each video frame contains its own picture information.

With no dependency on other images, JPEG 2000 has low latency, less than 1.5 frames encode and/or decode, and maintains sync between the video and audio. In contrast, MPEG and other compression schemes that rely on predictive frames and motion estimation algorithms have high latency.

JPEG 2000 and its underlying wavelet compression can allocate 10 bits or even 12 bits at 4:4:4 quality, a level in line with the demands of digital cinema post production. In fact, there are several systems that use wavelet compression techniques similar to JPEG 2000. REDCODE and CineForm, for example, are used successfully for digital cinema production, which demonstrates the quality and robustness of wavelet compression.

Contribution

There are significant differences between MPEG-4 and JPEG 2000 for broadcast contribution. (See Table 1.) Further development of video and still image compression standards continues, and there will no doubt be future advancements for both standards. For example, High Efficiency Video Coding (HEVC), unofficially called H.265, promises improved video quality and data compression, as well as support for higher resolutions.

Broadcast contribution links seek to have the highest video quality because the signal may have to go through post-production editing. The output video quality from the contribution stage is propagated to the distribution link. As a consequence, the quality and efficiency from the distribution link is highly dependent upon the level of quality experienced during contribution. Broadcast contribution signals should be lightly compressed — almost equal to the original uncompressed signal — so there is minimal quality loss under multiple generations of encoding.

JPEG 2000 can compress HD video to a significantly lower bit rate, and then extract the video with no loss of information. When peak signal to

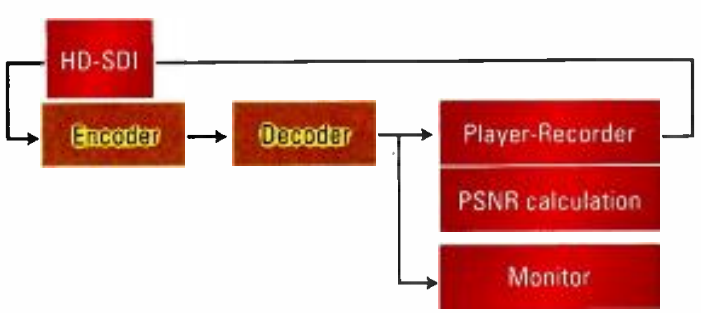

Figure 3. PSNR simulation setup

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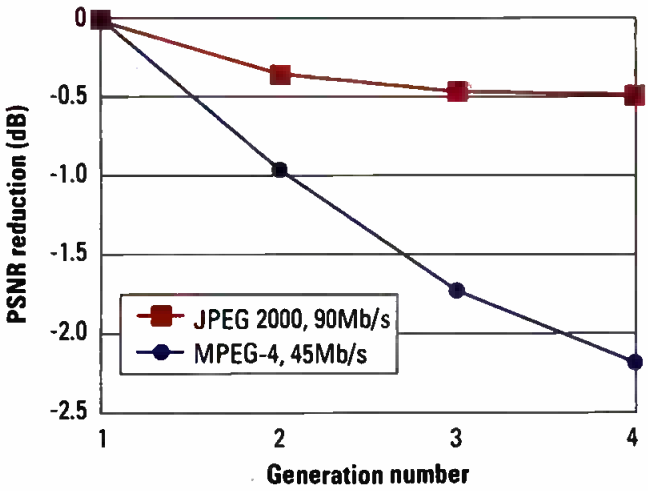

Figure 4. Multigeneration encoding

noise ratio (PSNR) performance is used to compare original footage with video compressed with JPEG 2000, the degradation is close to visually lossless and deviation from the original is low. The PSNR of footage encoded with MPEG, however, can vary greatly from picture to picture due to different methods of prediction.

Performance comparison

Recently, the performance of JPEG 2000 and MPEG-4/AVC under multigeneration encoding was evaluated using a single vendor's encoders and decoders. (See Figure 3 on page 12.) Video quality was measured in terms of PSNR for different HD interlaced sequences at the operating points of the encoders. The streams were encoded at 90Mb/s for JPEG 2000 and 45Mb/s for MPEG-4/AVC streams. These bit rates were used throughout the testing because they are common operating points.

MPEG-4/AVC provided high-quality video at 45Mb/s. Under multigeneration encoding and decoding, the PSNR reduction was sharp — roughly 2.2dB after the fourth generation. (See Figure 4.) JPEG 2000 also delivered high-quality video (at 90Mb/s), but it had less than 0.5dB loss after the fourth generation of encoding and decoding, which was less than one quarter of the reduction experienced by MPEG-4. In addition, the predictive coding of MPEG resulted in an end-to-end latency of approximately 1 second, which is roughly 10 times more than the latency experienced with JPEG 2000.

For another round of tests, designed to evaluate the influence of contribution encoding on end-user quality, a distribution encoder was used to encode the incoming signal using 4:2:0 8-bit at 6Mb/s, 8Mb/s and 10Mb/s (common settings for IPTV applications). The tests showed that because of JPEG 2000's high video quality, it is possible to increase the compression ratio for the final compression to broadcast without quality loss, saving from 5 percent to 15 percent of the total bit rate, depending on the final bit rate used for distribution. By spending more money on contribution, it is possible to save on broadcasting to the home, because more content can be placed in the available bandwidth.

By spending more money on contribution, it is possible to save on broadcasting to the home, because more content can be placed in the available bandwidth.

To illustrate the bandwidth savings that can be achieved when employing JPEG 2000 video compression, consider an uncompressed HD-SDI signal, which has a bit rate of about 1.5Gb/s. (See Table 2.) A JPEG 2000 encoder/decoder can maintain the quality of the original signal with a bit stream of 500Mb/s to 800Mb/s, depending on the content. At 600Mb/s, virtually any type of high-quality HD content will be transported mathematically losslessly with a JPEG 2000 system. Reducing the bit rate from there will increase the possibility of mathematical image loss, but any errors will be visually imperceptible even at far lower bit rates.

Stereo 3-D

Bandwidth-intensive formats like 3-D TV raise the bar even higher for quality and precision. The creation of 3-D video and particularly the need for the best possible picture quality has ramifications for how content is captured, produced and broadcast. In the contribution phase, left- and right-eye information must be transported along with other metadata required to generate the highest quality 3-D images and must be compatible with all 3-D compression and display systems. To ensure a high-quality viewing experience, it is critical that all the data required to display the left- and

Video format requirements	Typical IP bandwidth
HD-SDI uncompressed	1.5Gb/s
HD-SDI mathematically lossless JPEG 2000 mode	600Mb/s
HD-SDI visually lossless JPEG 2000 mode	120Mb/s-150Mb/s

Table 2: Bandwidth savings

right-eye images correctly is conveyed throughout the transport chain with minimal visual impairment.

JPEG 2000 ensures that the horizontal resolution, critical to the reconstruction of 3-D images in our brains, is preserved. A JPEG 2000 encoder can transport two 2K or HD-SDI channels, or the two synchronous left-eye/right-eye video streams that comprise stereo 3-D. Even after the JPEG 2000 video is compressed or decompressed, and throughout real-time transport over Metro Ethernet (metropolitan-area Ethernet), IP-MPLS (multiprotocol label switching) or SDH/SONET (synchronous digital hierarchy or synchronous optical network), the video quality stays true to the original imagery.

Applications

JPEG 2000 video compression is ideal for venues where high-quality programming regularly originates, such as stadiums, arenas and remote studios. While prices vary between service providers, commonly available IP links that provide high bit rates are universally less expensive than legacy systems, which translates to significant long-term operational cost savings. Before connecting an encoder, however, it is critical that the service provider understands the quality of service (QoS) requirements of high-quality video and can ensure sufficient bandwidth.

JPEG 2000 requires a high-bit-rate IP network and usually works best with both a primary and redundant link. If you need to do a remote shoot from a non-fixed location, i.e., in front of a burning house, then you probably need to use a microwave or satellite truck with MPEG compression.

Final signal distribution to viewers is a perfect application for MPEG. JPEG 2000 excels as a contribution technology, but is not intended as a distribution technology. That said, stadiums, city halls, remote studios, OB trucks and fixed locations where you need the highest quality video are good choices to consider using JPEG 2000 to maintain optimal video quality. As the cost of dedicated IP networks comes down and bit rates increase, JPEG 2000 video will help control costs when compared to traditional high-end video transport systems. Careful consideration of your outside broadcast requirements, access to high-speed IP networks and JPEG 2000 capabilities provide another choice in your video transport toolkit. BE

Dr. Helge Stephansen is chief technology officer at T-VIPS.

Regulatory fees due

The commission has announced the final 2012 regulatory fees.

BY HARRY C. MARTIN

The final 2012 regulatory fees have been announced by the commission. Table 1 shows the principal fees that apply to commercial television, plus LPTV, television translators and Class A television stations.

• *Ups and downs.* Fees for VHF television licensees in markets 26-50 have dropped nearly $2000 since the commission's initial proposals in May. VHF licensees in top-10 markets were granted a similar reduction of $7350 — about 8.4 percent — from the May proposals. UHF licensees in the top-20 markets were assessed increases over the May proposals in the range of 2 percent, or about $1000 each. By reducing fees for VHF full-power television stations, the FCC implicitly recognizes the parity of UHF and most VHF frequencies for digital broadcasting when compared to the much lower regulatory fees the FCC has historically assessed against UHF stations.

• *Fee window.* The FCC established Sept. 13, 2012, as the 2012 regulatory fees deadline. Anyone not paying by then will be assessed a 25 percent penalty and risks "red light" status, which means their applications and reports will not be processed until paid.

• *Multicasting.* For Class A, LPTV and television translator stations, a single fee will be assessed for each license, regardless of whether they transmit in analog or digital mode, or simulcast in both modes.

• *Online refund and waiver requests.* Any request for a refund, waiver, fee reduction or deferment of any annual regulatory fee, and of any application fee, must be submitted electronically. The commission's order provides no information about which online filing system might be used to apply for refunds or waivers. The order specifies no effective date for the new online system; presumably, the OMD will issue a public notice when its refund and waiver processing site is established.

• *Bills will not be mailed.* Again this year, the commission will not send bills reflecting each station's fee obligations. To ascertain what fees have been calculated per station, each station must go to www.fccfees.com, enter its call sign or facility ID, and proceed from there. When a dollar figure appears, check it carefully, as the FCC's postings are sometimes erroneous.

• *Paying fees for auxiliaries.* The FCC's fee calculator does not include fees for auxiliary licenses such as TV pickups, STLs, TV boosters and TV ICRs that may be associated with the main license. Thus, licensees should check their own and the FCC's records to be sure fees for such authorizations are included on their fee remittance forms (Form 159s) and reflected in the ultimate fee payment. BE

Harry C. Martin is a member of Fletcher, Heald and Hildreth, PLC.

? Send questions and comments to: harry.martin@penton.com

Dateline

- On or before Oct. 1, 2012, noncommercial TV and Class A stations in Alaska, Florida, Hawaii, Puerto Rico, Oregon, the Virgin Islands, Washington, and the Pacific Islands must file their biennial ownership reports.
- On or before Oct. 1, 2012, television stations, Class A TV, LPTV stations and TV translators in Florida, Puerto Rico and the Virgin Islands must file their license renewal applications.
- On Oct. 1, 2012, television and Class A TV stations in Alabama and Georgia must begin their pre-filing renewal announcements in anticipation of a Dec. 3, 2012, renewal application filing date.
- On Oct. 1, 2012, television and Class A TV stations in the following locations must post their 2012 EEO reports on the FCC's website: Alaska, Florida, Hawaii, Puerto Rico, Oregon, the Virgin Islands, Washington and the Pacific Islands.

2012 television regulatory fees

	VHF TV	UHF TV
Markets 1-10	$87,425	$34,650
Markets 11-25	$72,925	$31,950
Markets 26-50	$41,675	$21,875
Markets 51-100	$20,725	$12,625
Remaining markets	$5800	$3425
Unbuilt CPs	$5800	$3425
LPTV, TV translators and Class A TV (All markets)	$385	
Broadcast Aux. (STLs, TV pickups, Boosters, TV ICRs)	$10	
Satellite uplinks	$275	

Table 1. Shown here are the fees that apply to commercial television, plus LPTV, television translators and Class A television stations.

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Automation & Media Management

Digital distribution technology

Modulation schemes are increasing bandwidth efficiency.

BY ALDO CUGNINI

The distribution of content involves the need to adapt digital information to its means of carriage — what we typically call transmission — and to understand that fully, we need to understand the process of digital modulation.

Content distribution and transmission are implemented differently on wireless, wired and physical media, where each one uses modulation schemes tailored to carry content most efficiently and robustly. Wireless transmission includes terrestrial, satellite and wireless LAN channels, with VSB and OFDM serving the first, phase shift keying (PSK) for the second and OFDM again for the third. Wired transmission to the home is mostly by coaxial cable, using quadrature amplitude modulation (QAM), or by twisted pair, using OFDM over DSL. Physical media, i.e., DVD or Blu-ray, use an optical channel coding called eight-to-fourteen modulation (EFM).

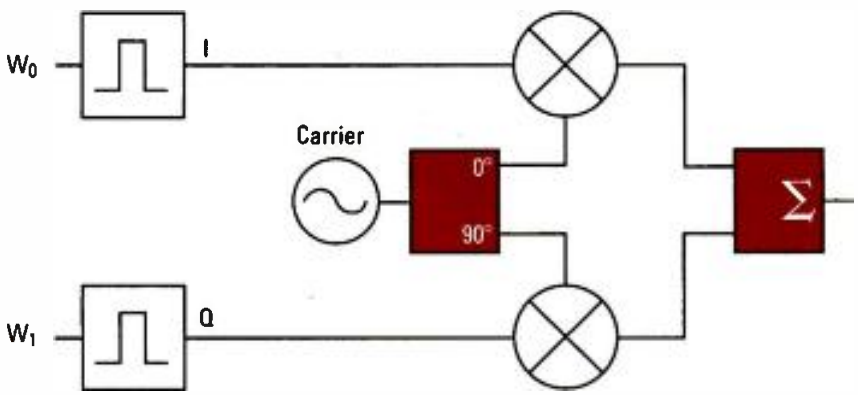

Figure 1. QAM modulator

Modulation schemes have different characteristics for different media

One of the simplest digital modulation schemes, QAM, takes two subsets of each binary word and applies the magnitude of each subword "value" to one leg of a multiplier (modulator), where the two multipliers are 90 degrees out of phase. (See Figure 1.) For example, a 4-bit word is split into two 2-bit words, each having a decimal value of 0, 1, 2 or 3. These values are then offset by the average (1.5 in this case, when the values are evenly distributed), resulting in a symbol input to the multiplier of -1.5, -0.5, 0.5 or 1.5, depending on the corresponding 2-bit word. By making the new average zero, there is no DC component to the symbol stream, and thus the output signal spectrum will have no power at the carrier, i.e., it is a suppressed carrier signal. On the receiving end, the digital demodulator can recover the carrier by one of various nonlinear means.

Because a "hard" transition between words would cause a large amount of redundant sideband energy, the input streams to the multiplier are sent through a raised-cosine filter that "smooths" the edges and results in a tight signal spectrum. Our

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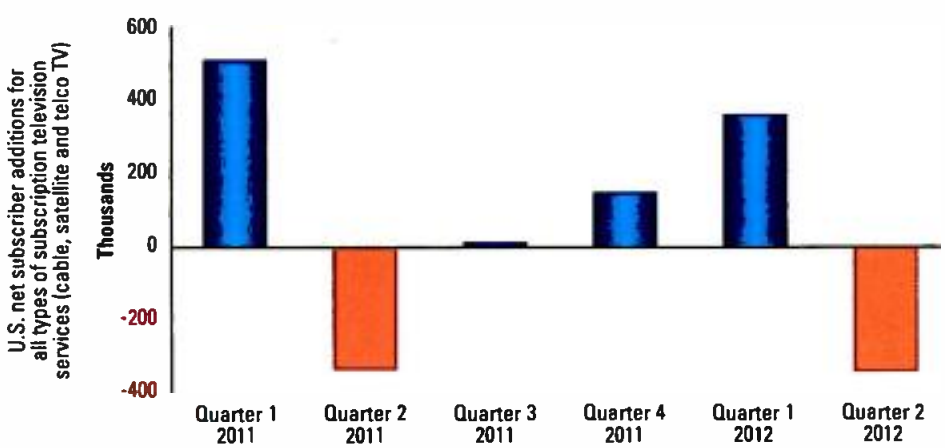

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4-bit example results in a 16-QAM signal "constellation" of 16 symbols ($16 = 2^4$), forming a 16-point grid when viewed on a modulation analyzer. Typical digital transmission systems use a 16-QAM, 32-QAM or 64-QAM constellation, depending on the required transmission bit rate and the desired C/N at a serviced receiver.

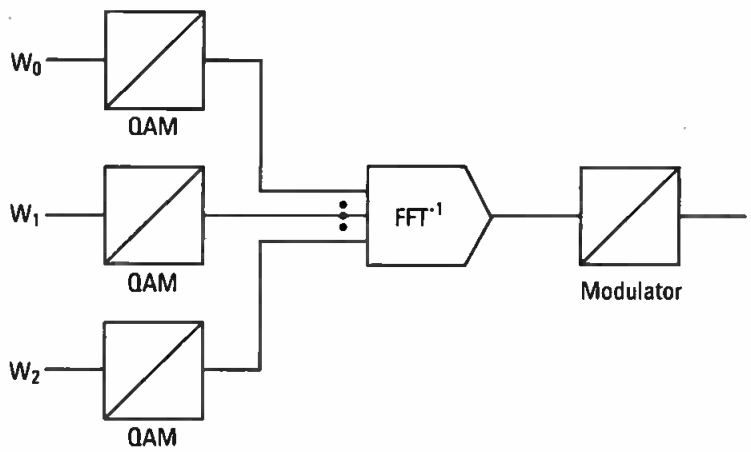

Figure 2. OFDM modulator (simplified)

VSB is a special case of amplitude modulation, wherein the output signal is filtered with a special form of high-pass filter (vestigial sideband or Nyquist) that attenuates one of the modulation sidebands. The absence of part of one of the sidebands necessitates the reinsertion of the carrier (or pilot) signal at a lowered level.

PSK is another modulation scheme that maps information symbols onto a transmitted carrier signal. With this scheme, however, the symbols are distributed onto a constellation that forms a circle, i.e., the amplitude is always constant, but the phase of the carrier is modulated by the symbols. While a 16-PSK transmission, with 4 bits per symbol, would be mapped onto one of 16 possible phases, it can be seen that a 4-PSK system (commonly called QPSK) is identical to a 4-QAM system. Having a constant amplitude is a benefit in systems with large linear distortion, such as that encountered on satellite links.

OFDM is a complex form of modulation that combines QAM and frequency-division multiplexing (FDM), the latter being the method by which different signals can simultaneously occupy a frequency band by using separate carriers at different frequencies.

An OFDM modulator first converts a single high-data-rate stream into multiple lower-rate streams. These parallel streams are then modulated onto orthogonal carriers that minimize the mutual interference that the data symbols could create during impaired channel conditions. This modulation is carried out through a combination of multiple QAM modulators followed by an inverse fast Fourier transform (FFT) that maps the individual streams onto a broadband signal. The resultant signal is then amplitude modulated onto the final RF carrier. OFDM, due to its strong performance in a heavy interference environment, is an integral part of the 802.11 standards used for Wi-Fi and now WiMAX.

In any of these systems, there is always a trade-off between bit rate and desired error-free performance, and these are set by two limits: Nyquist and Shannon. Nyquist sets the channel bandwidth needed to carry a train of symbols, and Shannon sets the C/N level that is needed for error-correctable performance. To understand this at the conceptual level, imagine a system that maps input data onto two amplitude states (i.e., two symbols). Increasing the data rate, according to Nyquist, requires increasing the channel bandwidth to allow for carrying the information without inter-symbol interference.

But if we take the same input data and map it onto four possible amplitude states, we now need only half of the channel bandwidth compared to the two-state system, for the same data rate. It would seem as though we got "free" channel capacity. But Shannon tells us that, given the same transmitter power, the need for twice the number of possible states means that the signal has now lost signal-to-noise power, since the distance between adjacent symbols is now half of the original distance.

Thus, Nyquist and Shannon together set the spectral efficiency of a system, which we can measure in bits per second of data per Hertz of channel bandwidth, commonly described as "(bits/s)/Hz" (or erroneously as bits/Hz). Thus, a modulation system with a "raw" data rate of 32Mb/s over a 6MHz wide channel results in a spectral efficiency of 5.3(bit/s)/Hz. In practice, the net bit rate efficiency will be lower, due to FEC and other overhead.

In any of these systems, there is always a trade-off between bit rate and desired error-free performance, and these are set by two limits: Nyquist and Shannon.

Because passband filters cannot have an infinite cutoff, spectral efficiency

also depends on the band-edge roll-off, which varies among systems. The net bit-rate efficiency of various modulation systems in use today ranges from about 3.23(bit/s)/Hz for ATSC to about 6.65(bit/s)/Hz for DVB-T2. (These numbers, however, should not be directly compared, as they represent different values of C/N threshold.)

New technologies on the horizon

It's conceivable that a digital distribution system that exceeds 10(bit/s)/Hz of efficiency is nearing our grasp, considering the current development of various promising technologies. Multiple-input multiple-output (MIMO) is one such solution, which uses multiple antennas and front ends to resolve reliability issues while maintaining or improving bandwidth efficiency. Combined with a video compression system such as High Efficiency Video Coding (HEVC), a new broadcast system could support more than 20 channels of high-quality audio and video within a 6MHz or 8MHz RF channel.

It's conceivable that a digital distribution system that exceeds 10(bit/s)/Hz of efficiency is nearing our grasp, considering the current development of various promising technologies.

Executives from television broadcast organizations around the world have already formed the global Future of Broadcast Television (FOBTV) Initiative, establishing a framework for cooperation to chart the future course of terrestrial television broadcasting. DVB and ATSC are among the groups working to develop and specify new compression and channel coding technologies that are expected to increase the data throughput and functionality of digital television broadcast. The sky's the limit! BE

Aldo Cugnini is a consultant in the digital television industry and a partner in a mobile services company.

Send questions and comments to: aldo.cugnini@penton.com

System administration for broadcasters

Training is a must for broadcasters who wear this hat.

BY BRAD GILMER

When professional media systems are delivered, they frequently contain computers and servers, and before we know it, because we installed them and we maintain them, we are the go-to guys and gals for all things computer-related. There are several problems with having computer system administration sneak up on you: 1) You probably are not trained to do the job. 2) The job may involve critical on-air or production systems. 3) You may be unaware of architectural patterns and anti-patterns. 4) Management may not be aware that you are performing in this capacity. 5) There is probably already someone in your organization who is the "real" system administrator; he or she is likely located in the IT department.

Bad things can happen to good broadcast engineers when they start heading down this road. On the other hand, it is inevitable that highly skilled broadcast engineers will end up performing system administration tasks. Let's dig a little deeper into the issues raised above.

Training is key

You probably are not trained to do the job. As a professional media engineer, there is no doubt that you have worked quite a bit with computers. You may have even taken some courses on computer architecture, programming and so on when you were at school.

But being a system administrator is a different animal. You are not just using computers; you become responsible for ensuring that others can use them as well. You will have to understand server operating systems (OS) at a whole new level. In some ways, a server OS is similar to a desktop OS. On the other hand, there are special utilities and tools that help you control access to the resources on the server, monitor security and configure the server for different applications.

As a system administrator, you may need to delve into the world of *NIX (Linux, Unix, Ubuntu, BSD, Red Hat, etc). There are several challenges with *NIX administration. First, the most efficient way to manage these servers is through the command line. This means learning a whole vocabulary of commands such as SED, LN, GREP and so on, many of which do not have equivalents in a Windows or MAC world. Second, many of the versions of *NIX are just slightly different from each other. So, you may learn where system files are kept in a Linux system, but when you move to BSD, things are in a different place. These are just a few points where a lack of training can hurt you.

Bad things can happen to good broadcast engineers when they start heading down the road to system administration.

Although professional training as a system administrator may not take care of all these issues, training is a wonderful thing, and it should be a planned, budgeted and expected part of the job.

Broadcast engineers often end up assuming computer system administration duties, which come with unique challenges ranging from gaps in training to possible clashes with management.

The job may involve critical on-air or production systems. When you combine this with the fact that you probably aren't trained for the job, you are headed for trouble. After all, if you are not trained, and then you are asked to maintain critical systems, how are you going to learn? I have frequently heard it said that the way someone becomes accomplished at anything is to have worked at it long enough to have made a number of mistakes, and to have learned from them. If you are maintaining critical on-air systems, how many mistakes can you make before your manager removes you from the opportunity to learn and

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go back for more experience? Not many; I can tell you from personal experience. And a related question: How can you make changes to a critical system without having confirmed those changes in a test environment ahead of time? Frequently, when we are thrust into a system administration role, we do not have time to think these issues through.

Patterns and anti-patterns

You may be unaware of architectural patterns and anti-patterns. Again, this relates back to the first point. If all you have is on-the-job training, how can a manager expect that you will be following best practices for system administration? After all, these best practices evolved through study and analysis of system administration tasks over years. Without training, even the sharpest knife in the drawer is going to make mistakes, simply because no one can think of everything.

Training may help you learn about patterns and anti-patterns without incurring the scars of practical experience.

Architectural patterns are ways of doing things that have proven to work well in most cases. An example of this is two-factor authentication (e.g. something you have, and something you know). Many different approaches have been tried to authenticate a user, and two-factor authentication is a successful pattern. Think of your ATM card; it involves something you have (the ATM card) and something you know (the PIN). There are many successful architectural patterns for system administrators, and books have been written on best practices.

As you might expect, anti-patterns are designs that do not work well. They are designs where you might say, "Given the opportunity, I would never do that again!" An example of an anti-pattern, although you see it all the time, is a watch folder. Watch folders — folders that are watched by an application such as a transcoder — are frequently used as a way to tell a downstream device to do something with a piece of content. For example, drop content into a folder, and magically, a transcoder will pick up the content, transcode it to another format and drop it in a pre-determined location. What's not to love? They are easy to configure, easy to understand and easy to change.

The big problem with watch folders is that you are using a file system as a messaging system. I could write an entire article on this topic (and maybe sometime soon I will), but this is a bad practice. If you want to tell a transcoder to do something, wouldn't it be better just to send it a message? Then it could acknowledge the message, and you would have an audit trail. What if someone puts a file in a watch folder, and then deletes it after the transcoder has started the job? How does the transcoder know you are done writing the file into the watch folder?

Getting back to issue number one, training may help you learn about patterns and anti-patterns without incurring the scars of practical experience.

Communication is key

Management may not be aware you are serving in this capacity. This is a problem because you are essentially performing two jobs for the price of one. Not only that, but when you make requests related to system administration, your manager may have a difficult time understanding what is behind your request. This might not only hurt you, but it may hurt the manager and the company (the law of unintended consequences). It is much better if everyone, especially your manager, understands that the business has changed, and that system administration is now a formal part of your job description. If you have found that system administration has crept into your job over the past several years, be sure your manager is aware of this.

As stated earlier, there is probably already someone in your organization who is the "real" system administrator; he or she is likely located in the IT department. This can cause no end of problems. Again, communication is key. I have seen many a good broadcast engineer get caught up in the turf war between IT and broadcast technology. If the organization is not clear about roles, then top management may question why in the world two IT departments are needed. As a broadcast engineer, it may fall to you to help educate your manager about the role you have found yourself in and how that differs from regular office IT.

Lastly, if you find yourself in the role of system administrator, please remember you are there to serve the needs of the organization; don't become a pain. Sometimes people who are in charge of systems treat these systems as their own personal kingdom. You will do yourself and your career a huge favor if you keep in mind that, without the creative and business types who frequently use the systems we administer, we would be out looking for work. BE

Brad Gilmer is president of Gilmer & Associates, executive director of the Advanced Media Workflow Association and executive director of the Video Services Forum.

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Grab your gear

News photographers need to balance their gear loads.

BY BRYAN C. FRANK

I am a news photographer. I go out with a camera and a microphone to record images and sound for a television newscast in the Los Angeles market.

It has always been my experience that my job is easier when I have the right tools. That said, a problem I face is that every story is different, and sometimes bringing the right tools in addition to the camera and microphone is a difficult choice to make. As much as it might be nice to have a magic bag that can hold all of your specialized widgets, gadgets, cables and other assorted goodies collected over the years, it (in my experience) doesn't exist.

So, it pains me to admit, but we're physically limited as far as how much we can carry. In other words, while it's not an absolute, get used to the idea that you can't always take it with you. Carry what you might need, but you have to find your own sweet spot.

Today's video journalists range wide on the "gear spectrum." Some travel with just a camera and microphone. Others fit the "kitchen sink" mold. Multi-tools and packing just the "needs" can help achieve a gear balance.

The gear spectrum

It is possible to travel light with just a camera and microphone. At the opposite end of the spectrum, however, you'll find the "kitchen sink" photographers. They are the human embodiment of "be prepared." They drive the overloaded van, carry an oversized fanny pack and fear being caught with only eight BNC barrel connectors. That said, naturally, I'd never poke fun at them since I've been saved once or twice (OK, maybe a dozen times) by the shooters who do carry everything.

Certainly, there are items that are not optional. The tripod is optional (no, Rembrandt, you don't *have* to have one), but the camera and a microphone are not. Also, don't forget an extra disc, tape or memory card (or whatever you're recording on), and spare batteries for the camera and wireless mic transmitter. (I think we've all learned this from experience.)

The further the walk back to the truck, the more precaution one should take not to lose recording ability. Wireless audio technology is pretty reliable, but never is it 100-percent guaranteed. XLR audio cable has saved more than one interview.

If you look into my bag, you'll find a pocket multi-tool — the kind that has knife blades, screwdrivers and unfolds into a pair of pliers. Just be careful about carrying it all the time. I always stop carrying it each time I lose one to an airport security checkpoint. And, of course, I'll buy another one when I'm caught without and need it in the field. Also, I keep an LED mini-flashlight as well as a headband hands-free LED light.

Digital age

If you've looked around recently, you might have noticed the broadcasting industry has its feet firmly planted in the digital age. My station converted to Sony XDCAM disc-based acquisition more than five years ago. We've also slowly begun migration towards SD card solid-state recording media. Shooting on disc and having the availability of a nonlinear edit system in the field has prompted my own purchase of two items that people cutting tape-to-tape never needed — a USB thumb drive and a portable 500GB external USB hard drive.

Both drives allow me to transfer and archive edited stories without having to hook up a laptop. They also give me the ability to copy and transfer visual digital material from other computers. I first used a 4GB thumb drive that would get me through most transfers, but I recently picked up a 32GB replacement because, apparently in storage, size matters.

As long as I'm talking about the more gadgety items in my life, I might as well admit that even though I'm a long-time PC user, Apple products have creeped into my world.

Yes, an iPhone and iPad have become nearly indispensable to me.

They're like the Swiss army knives of the digital age. I use them for GPS and general information on a daily basis. They're good for taking notes and offer another option for text or social media communication. In a pinch, the iPhone can (and has) serve as a backup video camera. They're not mandatory, but they can sure make the day easier.

A multi-format USB memory card reader also is handy to have. I've taken digital still shots and video into our computer from pocket cameras and even cell phones. I also still carry a set of cables that adapt RCA (standard consumer audio/video connectors) to BNC/XLR (standard grownup connectors). I'm just finding that I use cables less frequently.

Not everything one carries is a technical piece of gear, however. Should you ever get stuck on a fire or snow story miles from home, you'll quickly learn to keep handy a toothbrush.

Travel

I have my passport with me at all times because I really hope to someday hear the news director yell for someone to jump on a plane somewhere (heck, anywhere, I'm not picky). On the travel front, I keep separate bags packed — one with fire and safety gear and one with an assortment of clothes for different climates. We happen to cover a wide geographic area. It might not be snowing in downtown Los Angeles, but we have mountains and ski resorts within easy driving distance.

Again, you can't take it all with you, but sometimes a breaking news story isn't over by the end of a shift; i.e., you may not expect to spend a night away from home, but it happens.

You can try to plan for every scenario, but there's always going to be some limit on what you can carry.

I've shopped at grocery stores, gas stations, convenience stores, truck stops and just about anywhere open 24 hours for all things personal-hygiene related. I've also found myself to be much less picky about truck-stop fashion in single-digit weather.

You can plan, pack and try to anticipate every possible scenario, but there's always going to be some practical limit on what you can carry.

At the end of the shift, I think it's not so much about the tools and gadgets we use to get our jobs done; it really depends on the resourcefulness of the person doing the job. Regular people like me go out and make it happen. It's not always easy. Sometimes, we have the right tools, and other times we have to do what we can with the equivalent of two sticks, chewing gum and rubber bands. The point I'm making is that good news photographers will make air regardless of the gear they have or don't have.

Summary

In closing, I am not the guy who carries a ton of gear to every story. But, I also don't want to be the guy missing needed tools. As a photographer, what I pack is up to me, and I just want to have what I need to handle most situations. It's called the sweet spot, and it is where I think you'll find the best news photographers. BE

Bryan C. Frank is a news photographer in the Los Angeles television market.

An introduction to LTFS for digital media

The open standard can benefit content producers.

BY RAINER RICHTER

We live in an age where content is king. For the entertainment and media industry, the majority of this content is now produced in digital form, and virtually all of this content has digital distribution; that content is now digital data. Protecting that content, the lifeblood of this industry, with the right data storage system is more important than ever.

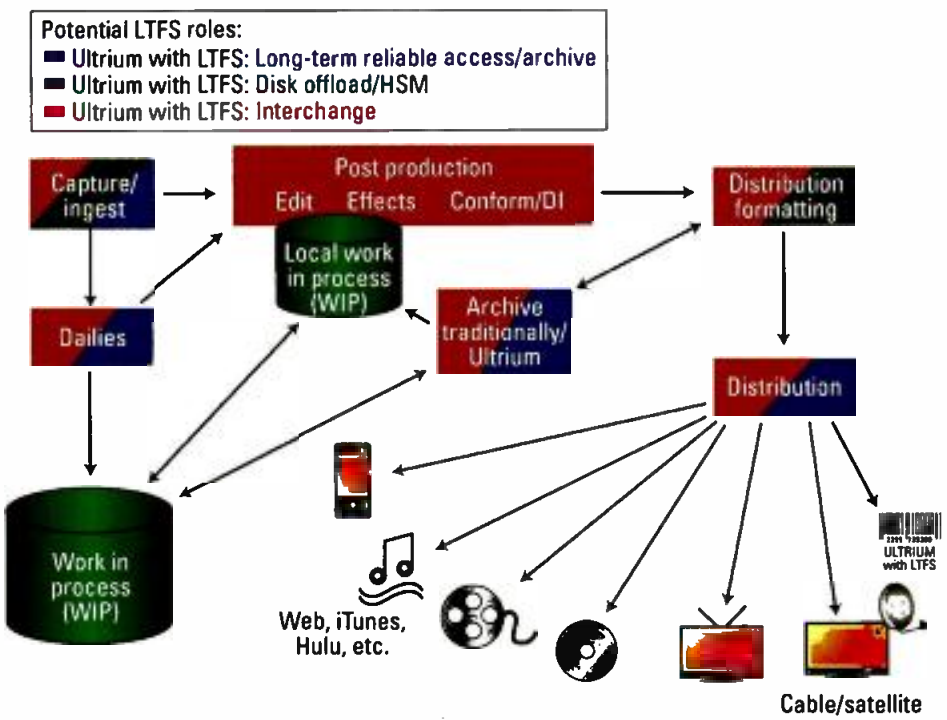

Figure 1. LTO tape is already firmly established in the media production environment. For production, LTFS efficiently supports several important requirements such as camera media resuse, backup, transport, direct access to data and archiving.

Tape is already firmly established in the media production environment whether to secure on-set content or for long-term archiving. LTFS broadens LTO technology usefulness by being easier to use and more robust. The open and self-contained LTFS format is useful if the tapes are to be sent offsite, archived or shared with a variety of recipients.

The LTFS standard was adopted by the LTO Program in April 2010. It is an open format and software specification that supports simpler and new ways to access data on tape. Although the tape model hasn't changed dramatically over the years, the speed, storage density and features of data tape have improved significantly, ultimately providing reliable and inexpensive storage as a sequential storage medium. With LTFS, accessing files stored on the LTFS-formatted media is similar to accessing files stored on other forms of storage media, such as disks or removable USB flash drives. It has no application software dependencies, offers support for large and numerous files, and often can have a lower total cost than traditional managed tape storage.

How LTFS works

LTFS consists of a software driver and the format specification. Drivers, some free and open source, are available for various operating and tape systems from the tape hardware vendor websites.

The format is a self-describing tape format and defines the organization of data and metadata on tape. Within the partitions, the tape contents are still stored as usual, as blocks of data and file marks. Files are mapped by LTFS to a hierarchical directory structure.

LTFS uses media partitioning, where tape is logically divided "lengthwise" into two partitions:

- *Index partition:* Contains file system info, index, metadata.
- *Content partition:* Contains the files/content bodies.

The standard LTO-5 tape cartridge is segmented into two partitions, one for the index and one for the data, so the index partition can be modified as needed without affecting the append-only data partition. This creates a self-describing tape where a user can see the tape cartridge and its contents in the operating system directory tree browser and can copy, paste or drag and drop files/folders to and from the tape. Similarly, applications can access the data on tape directly, unaware they are using tape, though there may be differences in latency inherent to tape.

When the tape is inserted and mounted in a tape drive, the

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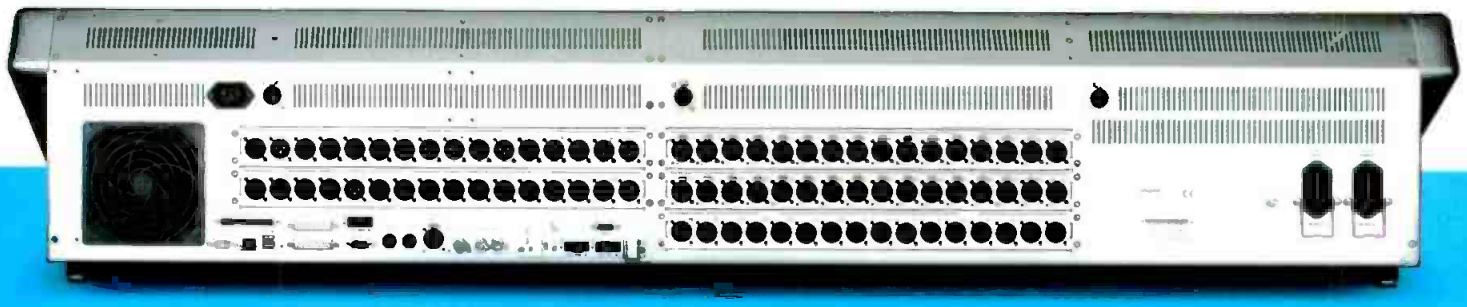

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information in the index partition is read and cached in the workstation's memory. From that point on, the index is accessed and updated in the workstation memory for fast performance and so the tape head can stay positioned in the content portion of the tape to more quickly access files or write new ones.

As the tape is used, the index is updated in the workstation memory for fast performance, eliminating the need to go back to the beginning of the tape. To protect the index, LTFS periodically copies the index from the workstation's memory to the data partition. When the tape is done being used (unmounted), the index is copied one more time at the end of the data partition, the tape is rewound and then the index is written twice to the index partition. Essentially, there are multiple copies of the index on tape for restoring in the unlikely event the index partition index is not usable or if a user wants to roll back the tape to a previous version.

For instance, you may want to restore it to the way the tape appeared last Monday by choosing an index from that date. In some tape library implementations, the tape cartridge's indexes are cached to a server disk for fast searching without having to remount the tape cartridges. LTFS also supports extended attributes, which enable custom file metadata.

Workflow

LTFS opens up new opportunities for media and entertainment storage and distribution, providing support to media workflows. LTFS directly affects two major trends having significant impact on media and entertainment companies and digital media producers: the shift to file-based workflows and increasing storage demands.

With the advent of digital technologies, moving image (video and film) content producers and distributors are transitioning from analog/linear workflows based on film or videotape technology to digital/nonlinear workflows based on the manipulation of data files. Many organizations have already completed this transition. This has been commonly referred to as "moving to a tapeless workflow," though it is more accurate to call it moving to a "*video*tapeless" workflow.

The other fact of life in media and entertainment is the demand of ever-increasing visual resolution and complexity (HD, 4K, 3-D, etc.) creating more and larger files that must be managed. Keeping hours of such media online on disk quickly becomes cost prohibitive. LTO tape, especially with LTFS, can address the challenge. LTFS can work with LTO hardware-based lossless compression, which can provide bandwidth and capacity benefits depending on the data content.

Production

LTO tape is already firmly established in the media production environment. (See Figure 1.) For production, LTFS efficiently supports several important requirements.

- *Camera media reuse:* Digital cameras encode motion images directly to SSDs or removable disks in the camera. These media are quite expensive (three to more than 350 times the equivalent media cost on LTO). Fast transfer of their contents to tape with LTFS enables reuse of this expensive media, reducing the number of SSDs or disks that must be purchased or rented.
- *Backup:* Backup of daily footage to LTO tape is a common requirement as the loss of a day's worth of production is costly. LTFS facilitates backup by enabling small portable independent systems to easily write daily content to tape.
- *Transport:* The density and cost of LTO-5 tapes with the self-describing capabilities of LTFS combine to create an effective transport medium. Large amounts of data can be sent more quickly and economically than network-based transmission methods. This is especially compelling for digital productions, which can produce terabytes of data for every day of shooting. The encryption features of LTO tape help secure the data in transit.
- *Economic direct access to data:* For any file-based production workflow, an LTFS-enabled tape drive can feed workstations or networks with content directly and relatively quickly, similar to a disk and unlike most traditional tape systems. An application via the operating system always has a direct and persistent view of a mounted LTFS tape and the files it contains. Consequently, in a workflow where access to a file is expected to be fast but not instantaneous, such as a stock footage collection or archive footage of an ongoing news story, an LTFS tape is an effective and economical choice for storage.
- *Archive:* LTFS-formatted tapes can be easily imported into an LTFS-compatible archive by simply reading the index and adding the file metadata to an archive manager's catalog. Conversely, traditional systems that use separate media for transport and archive require all the data be recopied. With LTFS, there is no need to read the much larger data partition or transfer the data to other storage media. The transport media and the archive storage media are one and the same under this scenario. The "import bandwidth" of tapes being added directly to a library *en masse* far exceeds any system that requires movement of the actual data.

These are just a few of the ways that LTO tape technology and LTFS are transforming production and storage in the media and entertainment industry. These excerpts and additional details are available in a white paper titled "LTFS Hits the Mark in Media & Entertainment: An In-Depth Introduction to LTFS for Digital Media" provided by the LTO Program at *www.ultrium.com/whitepaper* or at *www.mediatechmarketpartners.com.* BE

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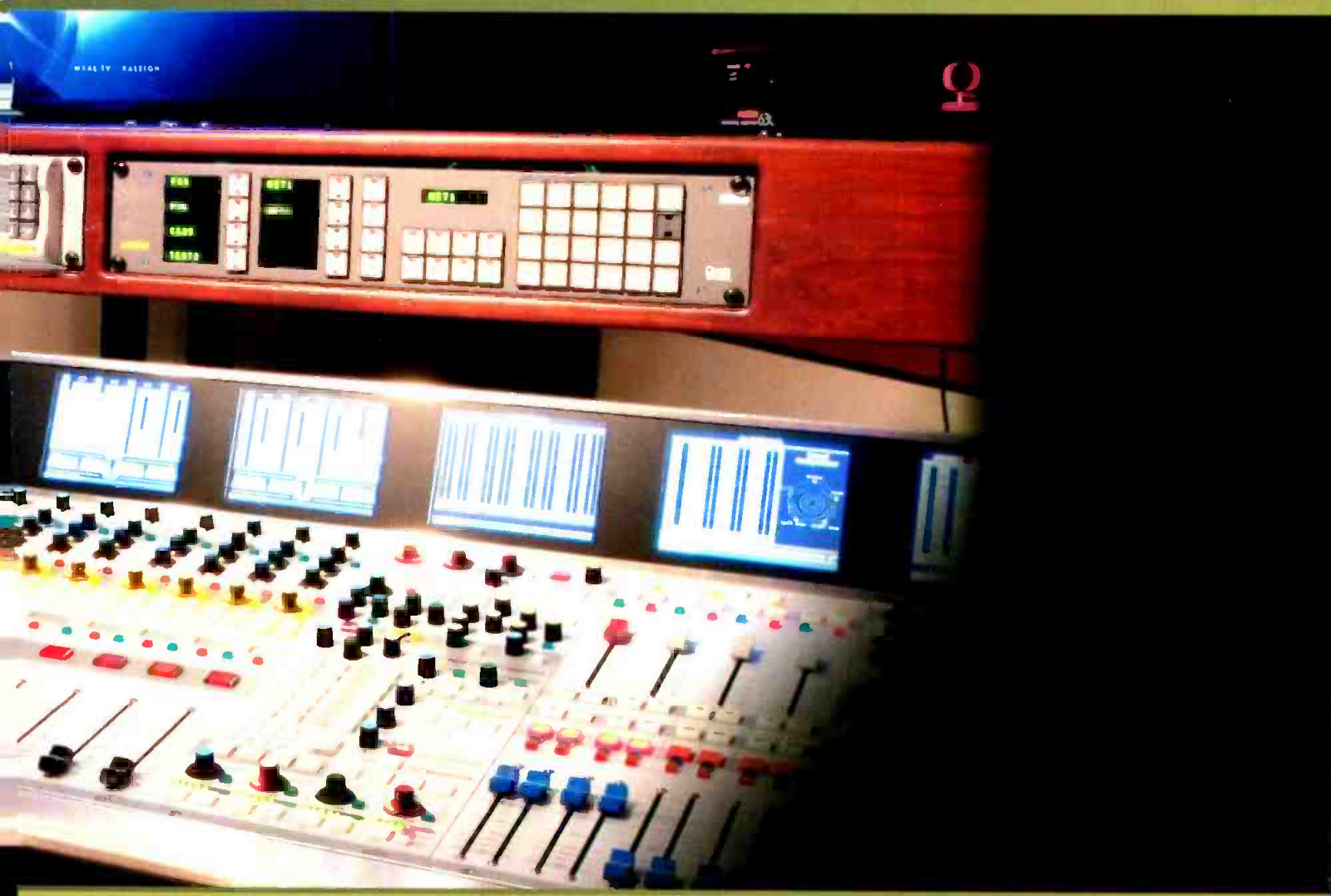

Consoles and mixers

BY GARY ESKOW

Shown here aboard a Telegenic OB truck, the Calrec Apollo console allows users the option to output LtRt phase-encoded signals directly from the console's main outputs.

Consoles sit at the center of every recording and broadcast project — in the studio, in the field, routing and processing audio. But, although technology has developed significantly, the role of the console has remained fundamentally consistent for the last 75 years or so.

As with every other class of equipment manufacturers, however, companies that build boards must keep listening to their clients. Are audio engineers asking for new feature sets? What will the next generation of consoles look like? Representatives from some of the top console manufacturers spoke with *Broadcast Engineering* and shared where this key industry is in 2012 and what changes consumers and the industry might look forward to in coming years.

"Analog snakes in TV broadcast are almost an extinct reptile. Multichannel digital connections over fiber and copper, such as MADI, are the evolution of the species." — Piers Plaskitt, Solid State Logic CEO

Broadcast Engineering: In the 1990s, digital audio workstations began including consoles as part of software packages. Are we beginning to see a push back in the area of console design? Specifically, can we expect to see consoles that host third-party plug-in processors?

Piers Plaskitt, Solid State Logic CEO: "It's arguable that a digital audio workstation and a digital console are built from the same pieces; i.e. a control surface, I/O and digital processing. However, the console is optimized for scale, performance and control, with short latencies and good hands-on operability. A digital audio workstation is designed to be screen-, keyboard- and pointer-controlled and should be optimized for editing and cost, with a modest amount of I/O. The technologies may be similar between each, but the applications and performance of each are quite different."

Phil Owens, Wheatstone Eastern U.S. sales manager: "Only in a pure production environment. Live boards typically have feature sets (mix minus, IFB, delays) that music production boards lack, and music production is where things like WAVES plug-ins are most helpful. Some of our competition implements some control over DAWS through the Mackie HUI protocol, but that doesn't allow for plug-in control. So, the distinction hasn't really blurred all that much. It's still a case of the right tool for the job."

Andy Trott, Studer by Harman vice president and general manager, mixers, microphones, headphones: "It's a question of workflow and personal choice. DAW systems are obviously used in the creation process, while the console is used alongside for production, but standalone in delivery."

Broadcast Engineering: Miles of analog snake being replaced by a digital matrix has brought large change to live audio production. What impact has this had on console manufacturing?

Henry Goodman, Calrec Audio head of sales and marketing: "Network technologies allow hundreds of audio signals to be transferred in both directions along a single connection. At its simplest, a densely packed I/O box can be located anywhere it is required and

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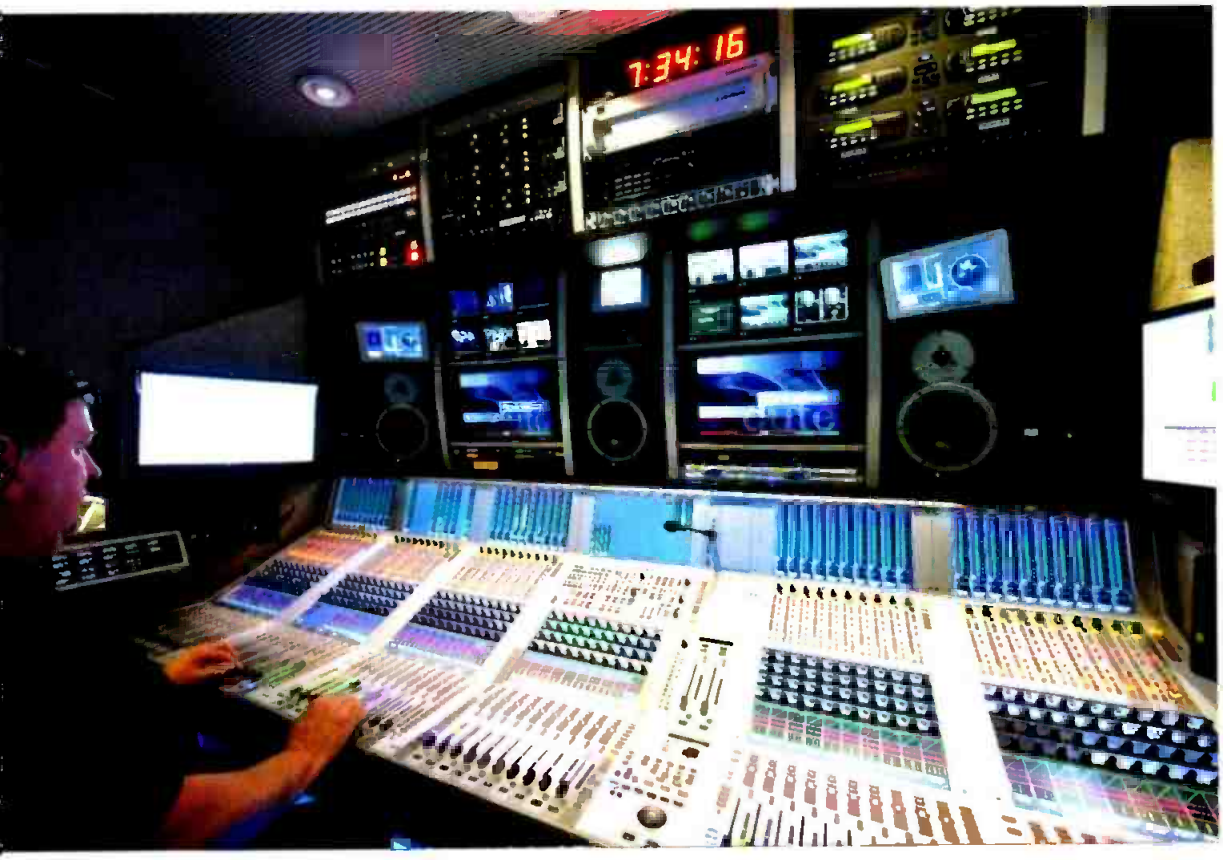


Shown here in the DutchView DV8 OB van, a Studer Vista 9 console is part of the production package built for both low- and high-end productions.

only one single connection (or two for redundancy) needs to be made back to the audio desk in order to send and receive signals.

"A smart system design allows us to create massively scalable networks where all resources are available to any client on the network. For the user, it's fantastic. You simply find a source on the network and connect it to a destination with a few button presses. It doesn't matter where on the network the source and destination are, or how many matrices the signal has to pass through. All of the internal routes are managed for you and created instantly.

"There is still some resistance to moving away from a traditional audio operator mixing on a console because of the nature of live TV. Moving to automation doesn't wave a magic wand over many day-to-day audio level problems." — Phil Owens, Wheatstone

"But, this technology goes way beyond simple routing. Complex access rights management, I/O aliasing for effortless movement between studios, flexible virtual patchbays and integration with broadcast control systems all contribute to incredibly powerful systems that were either not possible, or simply unmanageable with traditional patch bays.

"In terms of the impact on console design, it makes sense to integrate all of this smart network technology into the heart of each mixing desk. The desk becomes a client of the network, and user operation is designed around accessing shared, remote resources. All of this smart network technology is purely an enabling technology, though. A mixing desk's primary job is to process and mix audio; it should behave like an audio mixer. It should not be an audio router with some processing capability bolted on. It needs to be designed around the workflow and requirements of a human audio engineer, not an IT technician."

Andy Trott: "In a broadcast scenario, the sources all feed a central location where the stagebox would be located — where all the XLRs and other format connections are made. There are already numerous digital audio formats reaching this point. Much of a broadcast infrastructure may well be already using AES or SDI for signal distribution, so analog snakes are already rare, except in a fly-away system used for location live production, for example."

Piers Plaskitt: "Analog snakes in TV broadcast are almost an extinct reptile. Multichannel digital connections over fiber and copper, such as MADI, are the evolution of the species."

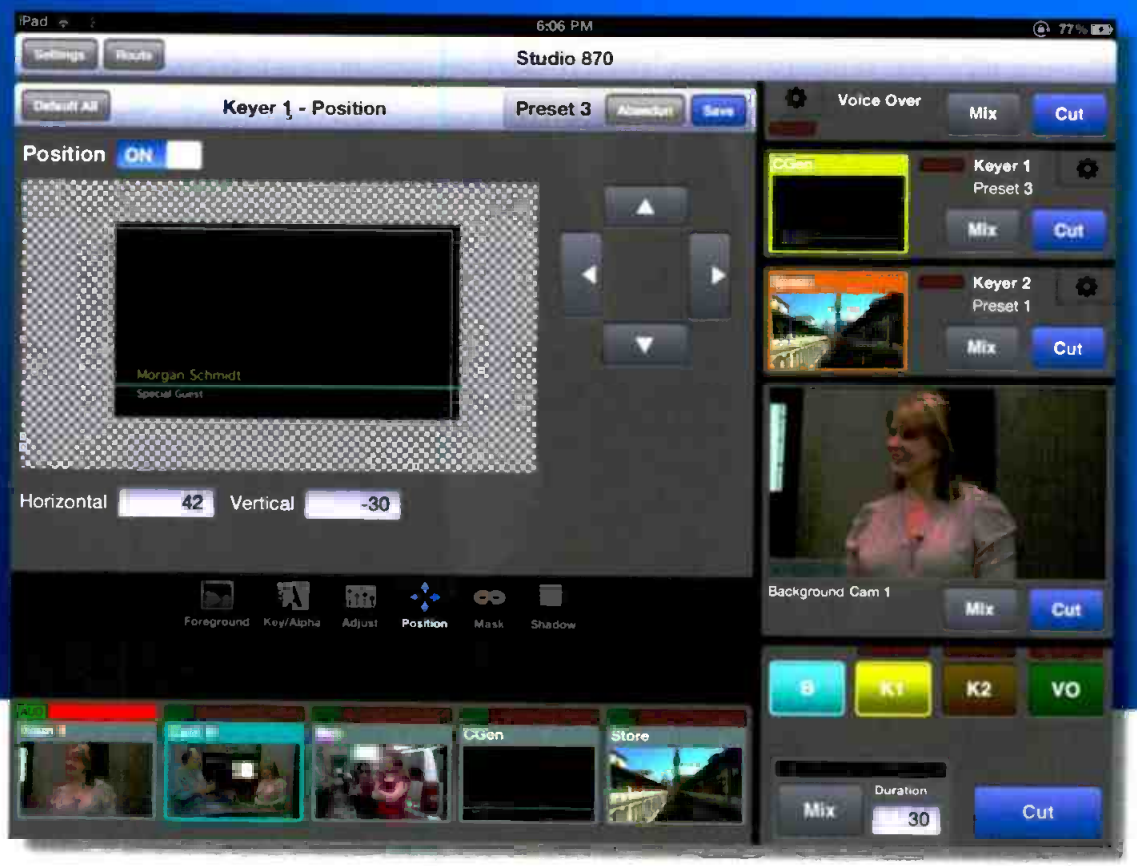
Broadcast Engineering: What new features are users asking for?

Andy Trott: "The obvious topics these days are based around workflow, loudness monitoring, remote control and AoIP."

Piers Plaskitt: "They are asking for more 'intelligent' products that are simpler to integrate and cost less money."

Phil Owens: "We've been asked for the

New Keying and Layering Technology



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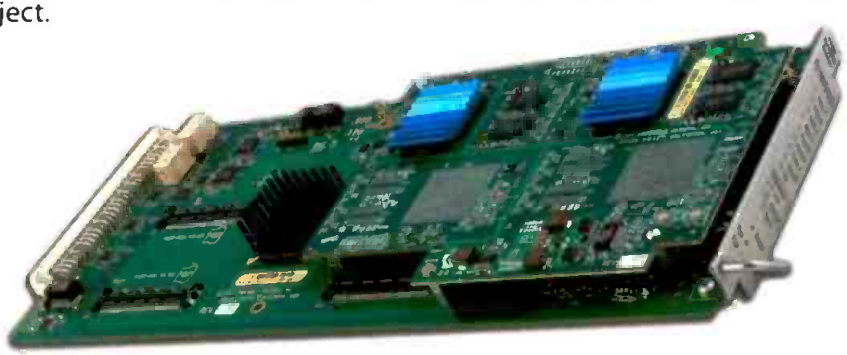

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The "multipurpose" room at MSG Media's facility, in New York, houses the Solid State Logic C100 HDS console. The board's primary use is for live audio mix, voice-over and SAP.

ability to de-embed SDI audio, and also for MADI I/O. With the approaching CALM Act deadline, we are also seeing requests for integrated loudness metering solutions using the LKFS scale."

Henry Goodman: "More often, users are requesting features that simplify their workflow. They want facilities to provide clear and concise information about the state of the system and the ability to quickly track down any issues. Integration with external control systems are important, as is remote control of desk parameters. Labor-saving devices such as automixers are becoming more popular, allowing engineers to focus on the more complex and artistic qualities of a mix."

Broadcast Engineering: Have you noticed any changes in broadcast production, and if so, do they influence console design? What differences are there between live mixing in surround and stereo, and how has the advancement of surround sound changed the way consoles are built?

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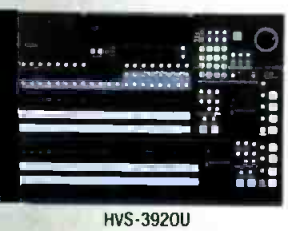
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Continuous Innovation

Phil Owens: "Many small to mid-market stations have moved to production automation with varying degrees of success. There is still some resistance to moving away from a traditional audio operator mixing on a console because of the nature of live TV. Moving to automation doesn't wave a magic wand over many day-to-day audio level problems.

"All broadcast consoles have to handle 5.1 content in combination with stereo and mono

"Productions are far more complex with more remote location feeds, multi-platform delivery and tri-media for TV, radio and online, and adding surround brings more challenges." — Andy Trott, Studer by Harman

sources. 5.1 content usually comes from network feeds, while local content is typically stereo. A typical broadcast day will have a combination of both."

Henry Goodman: "A critical point in mixing for different multichannel formats is in the correct monitoring of the full width and various downmixed versions of the program. Integration of the monitoring system with the various decoders is vital in understanding how a mix will translate to the viewer in different listening configurations.

"In terms of constructing the mix, working with multichannel signals should, in general, be as simple as working with mono or stereo signals."

Piers Plaskitt: "Probably the biggest recent change in TV broadcast production, from a console perspective, has been the adoption of SDI and its associated embedded audio/video routing infrastructure. This means that audio consoles often need to de-embed and embed audio to/from SDI streams.

"The difference with live mixing in surround and stereo is that there is sound coming directly from behind your ears, and a console needs a way to allow the operator to manage working with and listening to these extra channels. This increases the complexity of the console's channel panning, bussing and central monitoring sections."

Andy Trott: "Productions are far more complex with more remote location feeds, multi-platform delivery and tri-media for TV, radio and online, and adding surround

brings more challenges. Surround means six or more discrete channels to handle on the fader surface."

Broadcast Engineering: Have new technologies been developed recently to simplify the mix process? If so, how do they affect console design?

Piers Plaskitt: "To create a mix-in surround, the console needs a way to position sounds across multiple busses. It must also be able to conveniently handle incoming surround premixes and monitor all the various mono, stereo and surround listening permutations. To this end, 'surround ready' consoles are equipped with 5.1 and 7.1 channels, as well as complex panning features on every channel. Consoles must also fold down from surround to stereo, to cater for stereo feeds. Designing a console to manage all of these complexities is mandatory if you're going to do multi-format mixing properly. So, yes, surround audio affects console designs considerably.

"Another development is automated balance control, which allows multiple console channels to be automatically adjusted so the levels deliver a consistent mix. This makes balancing a fast-paced talk show a much more straightforward process."

Henry Goodman: "Downmixing is an integral part of contemporary broadcast workflows. Various versions of the final mix are sent to different broadcasters and are used in different contexts, be it host transmissions or clean feed to other countries, for example. Some of these feeds are not required in

"Designing a console to manage all of these [mixing] complexities is mandatory if you're going to do multi-format mixing properly." — Piers Plaskitt

the original 5.1 mix, so a quick and effective method of turning this into a stereo mix had to be developed within the console.

"Left only/right only downmixes are mainly used for local monitoring — people in and around the facility who have stereo or mono monitors.

"Left matrix total/right matrix total is used for transmission. As LtRt can be listened to as stereo, or decoded back to surround, the same path can be sent to all consumers. If they have

Shown here, the Dimension One unit installed at KOMO-DT, in Seattle, WA, is one of Wheatstone's flagship consoles.

5.1 decoders, it will decode it. If not, they hear stereo.

"Another benefit to LtRt is the ability to send 5.1 over two channels. Until relatively recently, it has been difficult to send more than two channels and keep them all time-aligned together, which is why DolbyE became popular in the pro environment. Transmitting two encoded channels also takes up less bandwidth than transmitting six discrete channels."

Phil Owens: "Advances in DSP silicon have enabled us to design extremely powerful mixing and processing cores that use significantly less power and occupy about one-third the space of previous designs. Today's consoles must do upmixing and downmixing: 5.1 content must be downmixed for feeds to stereo program and IFB outputs. Stereo and mono content must be upmixed and positioned for inclusion in 5.1 program outputs."

Gary Eskow is a composer, journalist and project studio owner.

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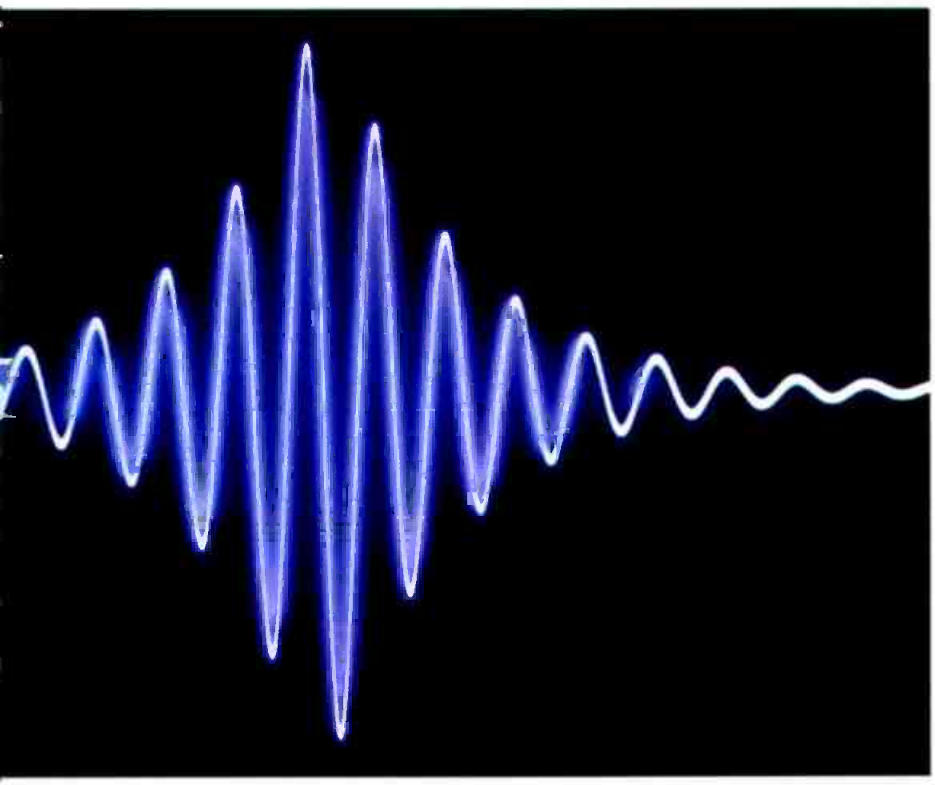
THE FORUM: Comparing loudness meters, part 2

BY ROBERT ORBAN



Different loudness controllers do not provide equally good subjective results even if they produce identical measurements on a loudness meter.

This article is the second and concluding part of an article in our Broadcast Forum focusing on an examination of loudness and measurement. Part 1 may be found in the August issue and on the *Broadcast Engineering* website at *www. broadcastengineering.com*.

We continue our discussion with an examination of the results after automatic loudness control. Figures 1 (on page 46) and 2 (on page 48) summarize that data. (To present the data with optimum graphic resolution, the loudness scales are narrower than in last month's graphs.)

Both the loudness vs. time graphs and the histograms show the Orban 8685 controls loudness well, although the details of the meters' indications are different. Both the BS.1770 and CBS measurements indicate that most of the data points are in a ±1dB/LK window.

The peak CBS readings fit within a ±2dB window. The BS.1770 readings also fit within a ±2 LK window — except for four short intervals, which appear as low-probability outliers in the left side of the histogram. These intervals correspond to dialog without background music and in the author's opinion illustrate a weakness in BS.1770-2: Based on our extensive listening tests, we have concluded that the meter does not effectively lock onto the A/85 "anchor element" (almost entirely dialog in the test material used to prepare this paper) and instead indicates that loudness increases when dialog level is held constant while underscoring or effects are added to the mix.[10]

Problems with low peak-to-RMS ratio material

In the subjective testing to validate the BS.1770 meter, there were outliers as large as 6dB (i.e., the meter disagreed with human subjective perception by as much as 6dB[11]). The subjective testing to validate the CBS meter found outliers up to 3dB, although fewer items were used in this testing. We hypothesize that the fact that the worst-case error of the BS.1770 meter was substantially larger than that of the CBS meter is caused by the BS.1770 meter's not modeling loudness summation or the loudness integration time constants of human hearing. BS.1770-2 states:

It should be noted that while this algorithm has been shown to be effective for use on audio programs that are typical of broadcast content, the algorithm is not, in general, suitable for use to estimate the subjective loudness of pure tones.

We have noted that the meter tends to overindicate the loudness of program material that had been subject to large amounts of "artistic" dynamic compression, as is often done for commercials and promotional material. In other words, the meter over-indicates the loudness of program material having an unusually low peak-to-average ratio, which, at the limit, approaches the peak-to-average ratio of a pure tone.

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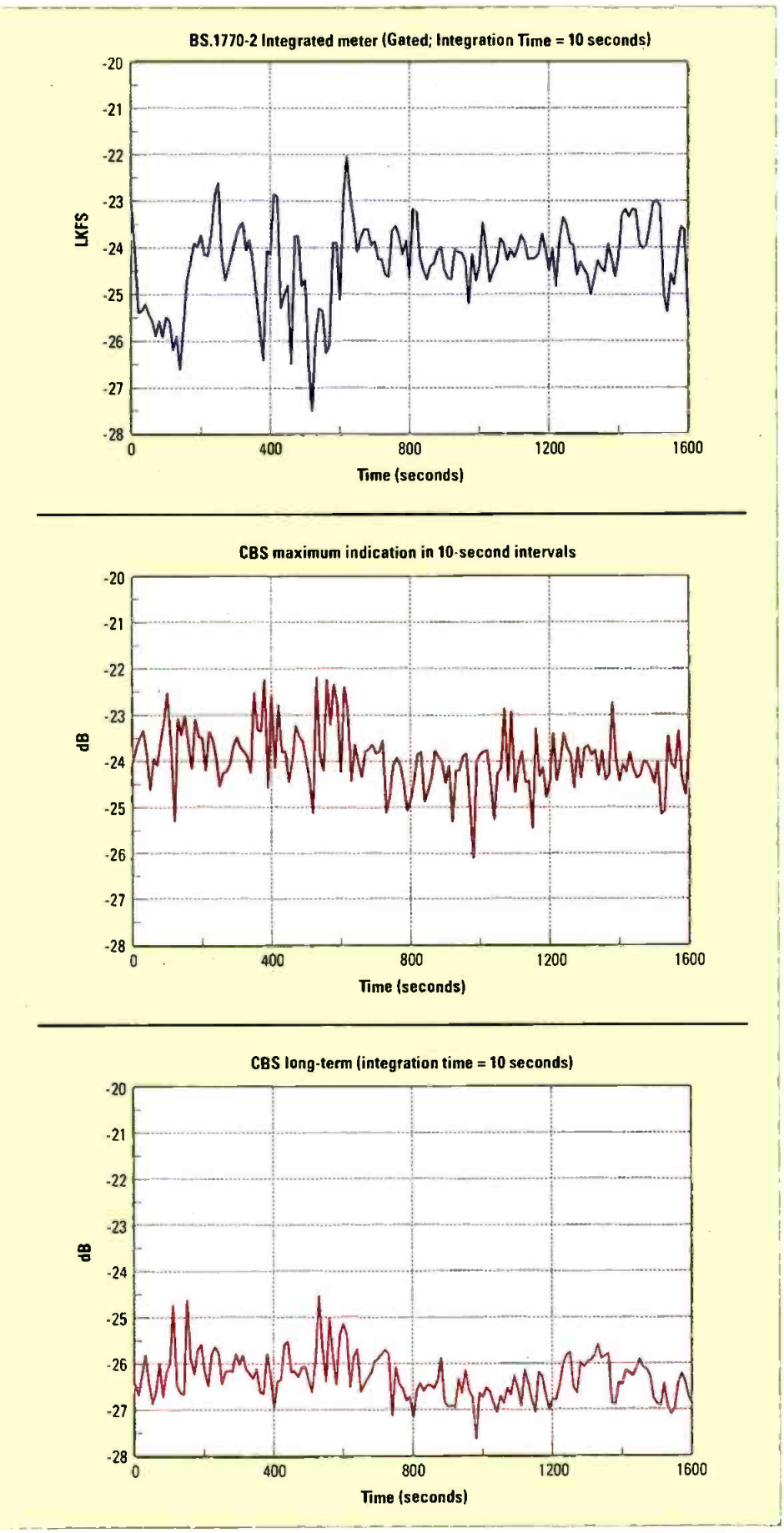


Figure 1. Shown here is the peak output of the CBS and BS.1770 loudness meters in each 10-second interval as a function of time.

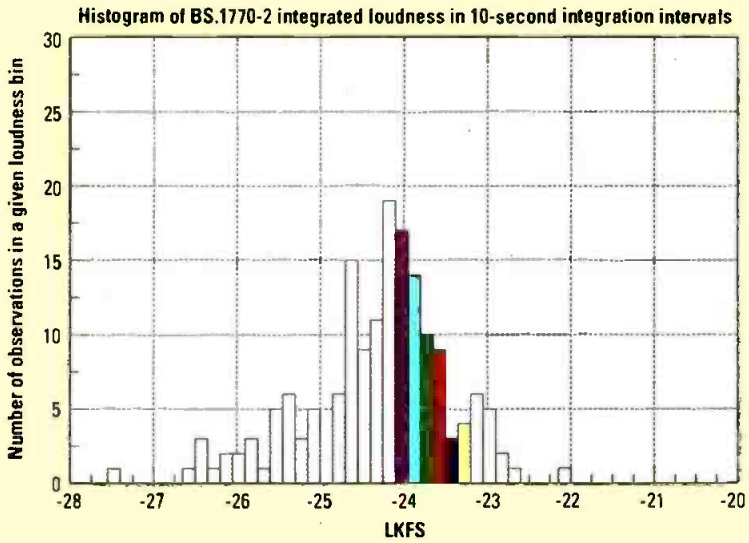
We have encountered heated complaints by mixers[12] and producers who stated that such material, when "matched" to the loudness of the surrounding program material via the BS.1770 meter, is considerably quieter in subjective terms. In turn, this has constrained the ability of producers to specify the type of audio processing they had previously used to give this material excitement and punch. We hypothesize that this problem is related to the fact that BS.1770 does not accurately indicate the loudness of pure tones.

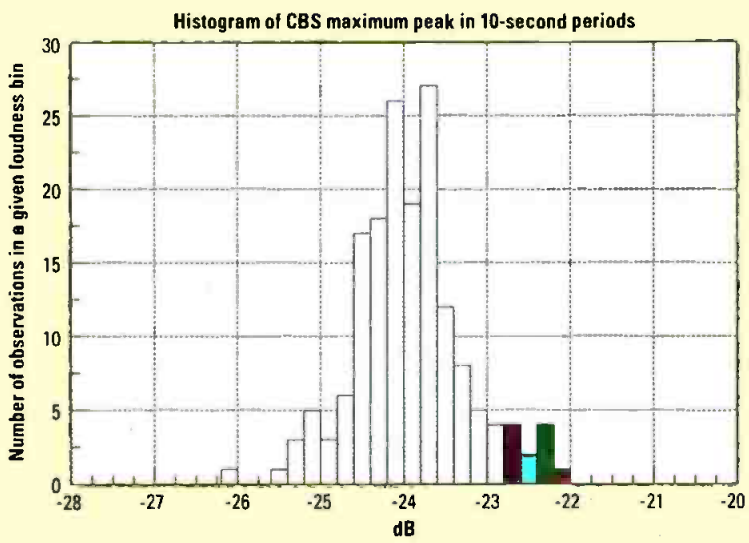
Some studies have indicated that when people are asked to assess the loudness of a given piece of material, they state that it sounds louder when underscoring or effects are added to constant-level dialog. The EBU has used these studies to justify the position taken in R 128 that a listener's impression of total loudness is more important than dialog level[13]. In our opinion, this misses the point. A more relevant question is whether viewers would want to turn down their volume controls to make dialog quieter when underscoring and effects appear. (In other words, whether effective TV commercial loudness control requires nothing more than applying gain control to commercials such that the BS.1770-2 "short-term" loudness[14] is always limited to 0 LK.)

Dialog is the most imprtant element in most television audio, and listeners do not want to turn down their volume controls every time that underscoring or effects appear under the dialog.

Orban and Dolby Labs hold similar views. We believe that dialog is the most important element in most television audio and that listeners do not want to turn down their volume controls every time that underscoring or effects appear under the dialog. The popular Dolby LM100 loudness meter[15] in its current revision uses the same Leq(RLB) algorithm as BS.1770 but adds gating to eliminate non-speech material, including silence. The author has used the Dolby LM100 to measure the output of the Orban 8685 with a wide variety of speech material, and has observed that this material is almost always controlled within a ±1dB window as measured on the LM100.

This demonstrates the benefits of a dialog-centric measurement. Moreover, the author





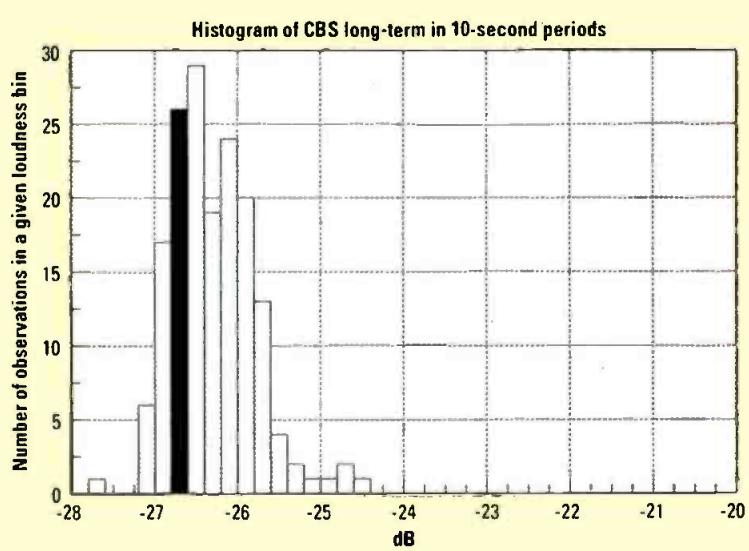


Figure 2. Shown here are histograms sorting loudness measurements into 0.25dB bins.

believes it is unwise to rely on a BS.1770 measurement to set the on-air loudness of unadorned dialog because this can cause the dialog to be too loud with respect to other material. The author has experimented with "inverse short-term BS.1770 loudness control" and believes that it sounds unnatural, pumping dialog loudness up and down in a subtly inartistic way as underscoring and effects come and go.[16]

Studies indicating that BS.1770 is inaccurate at very low frequencies

Another weakness of BS.1770 is that, unlike the CBS loudness controller and meter as implemented in Orban products, the BS.1770 algorithm does not take into account the loudness contributed by the LFE channel, for good reason. Nacross and Lavoie[17] tried to extend the BS.1770 algorithm to include the LFE channel by summing the K-weighted LFE channel's power into the current BS.1770 algorithm, where the gain is weighted for the fact that LFE channel receives a 10dB gain boost on playback, per Dolby's standards.

This modified BS.1770 algorithm failed to agree with the judgments of a subjective listening panel unless a 10dB attenuation "fudge factor" was applied to the LFE channel prior to its power summation with the other channels. Nacross and Lavoie concluded:

A problem exists, however, should ITU-R BS.1770 be modified to simply include an attenuated version of the LFE channel. Because the LFE channel receives a 10dB boost on playback, the low-frequencies on this channel would contribute differently to a loudness measure if they were moved to one of the other main channels, even though the perceived loudness would not appreciably change. This suggests that while LFE content does contribute to the perceived loudness, Equation (2)[18] does not sufficiently predict how that content should be included.

An Australian study may shed light on the failure of BS.1770 when program material contains considerable energy at very low frequencies.[19] The authors used octave-band noise in subjective listening tests with the goal of verifying the K-weighting curve used in BS.1770. The authors state:

Comparison of the test results with an image of the filter curve currently specified in ITU-R Recommendation BS.1770 shows good agreement at 250Hz and above 500Hz, reasonable

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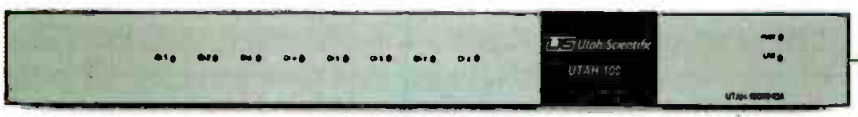

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agreement at 500Hz, but marked difference in the bottom two octaves.

The relatively good performance of the BS.1770 algorithm in ITU trials suggests that, in partial loudness terms, there was probably not much test content in the 125Hz band or below. While the existing BS.1770 filter curve is probably a good choice in applications where the program is dominated by speech, and it is certainly an improvement on the A and B curves in that application, it is likely to give significant errors in measuring the loudness of other programs with more partial loudness in the lower frequencies, such as movie soundtracks and popular music. It is, therefore, desirable to improve on this filter for more general measurement of program loudness.

Discussion and conclusions

Several studies have shown that the loudness "comfort range" for typical television listening is +2, -5dB[10]. Beyond this range, a viewer is likely to become annoyed, eventually reaching for the remote control to change volume (or worse, from the broadcaster's point of view, to mute a commercial). Whether measured via the CBS or BS.1770 algorithms,

It is important to carefully assess the audio quality and side effects that an automatic loudness controller produces so that one can choose a device that controls loudness effectively without producing objectionable and unnatural artifacts that can fatigue audiences.

the CBS loudness controller algorithm in Orban's current products effectively controls subjective loudness to much better than this +2, -5dB window.

In the original version of this paper, we had assumed that results using BS.1770 metering would be more consistent if that algorithm

employed gating to prevent unadorned dialog from reading low compared to music and dialog with substantial background music or effects. However, this did not prove to be true with the program material we used for testing; the results from the BS.1770-1 (ungated) and BS.1770-2 (gated) measurements were similar when measuring material that had been processed by the CBS loudness controller. It is likely that the loudness-controlled material seldom caused the gate to act. (The CBS algorithm does not need silence gating because it is a "short-term" loudness measurement that incorporates cascaded models of the "instantaneous" and "short-term" loudness time constants of human hearing[21], which the BS.1770 algorithm does not.)

Controlling loudness to a standard such as BS.1770 says nothing about the subjective acceptability of the loudness controller's action. We have found that a simple loudness controller that uses the inverse of the BS.1770 short-term meter's output to control loudness by gain reduction can cause unnatural-sounding gain pumping of dialog when underscoring and effects appear under the dialog.

More complex automatic loudness controllers can produce all of the well-known artifacts of dynamics processing. Improperly designed multiband compressors can reduce dialog intelligibility[22]. This is why it is important to carefully assess the audio quality and side effects that an automatic loudness controller produces so that one can choose a device that controls loudness effectively without producing objectionable and unnatural artifacts that can fatigue audiences. Different loudness controllers do not provide equally good subjective results even if they produce identical measurements on a loudness meter.

Based on extensive experimentation with typical broadcast material, we believe that the CBS loudness meter locks onto dialog more effectively than does BS.1770, particularly when the dialog is accompanied by underscoring and/or effects. Unlike the BS.1770 meter,

the CBS technology does not unnaturally penalize material having a low peak-to-RMS ratio, so it allows mixers and producers to freely use "artistic compression"[23] and other well-established production techniques with the knowledge that such material will be neither too loud nor too quiet when compared to the surrounding program.

Robert Orban is chief engineer, Orban.

FOOTNOTES:

[10] In the first published version of the paper, we observed the similar dips in the BS.1770-1 (ungated) loudness and hypothesized that they were caused by lack of gating on silence and low-level material. For this reason, we were surprised that BS.1770-2 gating made little difference in the measurements of this material.

[11] Refer to the scatter plots in Figs. 11, 12 and 13 of the ITU-R BS.1770-2 standard.

[12] For example: "I did a -24 [LKFS] piece for Fox that was wall to wall singing and music for two minutes. Because of the overall loudness and continued full audio signal, I had to bring it down and when it aired, it was 3db too quiet even though it matched the magic LKFS number. I have no problem using these meters or meeting specs, but they are faulty." —"wheresmyfroggy," AVID board, 3-28-2011

[13] Dash, Ian; Bassett, Mark; Cabrera, Densil, "Relative Importance of Speech and Non-Speech Components in Program Loudness Assessment," AES Convention Paper 8043, 128th AES Convention (May 2010).

[14] EBU R 128 specifies short-term loudness as a BS-1770-1 (ungated) measurement with a three-second integration time.

[15] http://www.dolby.com/professional/products/broadcast/test-and-measurement/lm100.html

[16] See Begnert, Fabian; Ekman, Håkan; Berg, Jan, "Difference between the EBU R-128 Meter Recommendation and Human Subjective Loudness Perception," AES Convention Paper 8489, 131st AES Convention, (October 2011). This paper states, "These loudness-equalized signals gave rise to a perceived maximum loudness difference of 2.8dB." This is very close to the 3dB number that has come up in other discussions. While the authors of this paper consider 3dB to be insignificant, others do not necessarily share this view, particularly advertisers who hear their expensive commercials aired 3dB quieter than surrounding program material!

[17] Norcross, Scott G; Lavoie, Michel C., "Investigations on the Inclusion of the LFE Channel in the ITU-R BS.1770-1 Loudness Algorithm," AES Convention Paper 7829, 127th AES Convention (October 2009)

[18] $$Leq(w) = \left[\frac{1}{T} G_{LFE} \int_T^0 \frac{x_w^2}{x_{ref}^2} \, dt + \sum_i \frac{1}{T} \int_0^T \frac{x_{w,i}^2}{x_{ref}^2} \, dt \right], dB$$

$i = L, R, C, L_S, R_S$

[19] Cabrera, Densil; Dash, Ian; Miranda, Luis, "Multichannel Loudness Listening Test," AES Convention Paper 7451, 124th AES Convention (May 2008).

[20] ATSC A/85:2009 Annex E, "Loudness Ranges"

[21] For example, see Glasberg, B.R. & Moore, B.C.J. (2002) "A Model of Loudness Applicable to Time-Varying Sounds," J.AES, vol.50:5, pp.331-342, May 2002.

[22] Stone, Michael A.; Moore, Brian C. J.; Füllgrabe, Christian; Hinton, Andrew C., "Multi-channel Fast-Acting Dynamic Range Compression Hinders Performance by Young, Normal-Hearing Listeners in a Two-Talker Separation Task," J. AES Volume 57 Issue 7/8 pp. 532-546; July 2009.

[23] It appears that the group that created R 128 may be biased against this style of production: "Again, this does NOT mean that within a program the loudness level has to be constant, on the contrary! It also does NOT mean that individual components of a program (for example, pre-mixes or stem-mixes, a Music & Effects version or an isolated voice-over track) have all to be at the same loudness level! Loudness variation is an artistic tool, and the concept of loudness normalization according to R 128 actually encourages more dynamic mixing!" EBU TECH 3343, op. cit., p. 17.

White space management

BY CHRISTOPHER LYONS

Broadcasters are innately familiar with the need to share spectrum. For many years, TV stations in the same market have been coordinating the wireless microphone frequencies used by their ENG crews to prevent interference when covering a breaking news event. But soon, a new class of commercial and consumer devices will begin sharing the same "white spaces" used by those wireless microphones. To continue delivering the kind of audio quality that viewers expect, broadcast engineers who use wireless microphones will need to take advantage of some new tools and techniques.

Meet the new neighbors

The new TV-band devices (TVBDs) come in two flavors. Fixed devices are typically mounted on a pole or other structure, and provide broadband access to other devices nearby. They can transmit with up to 1W of power and use an antenna with up to 6dB of gain, for an effective output power of 4W. Permitted power levels decrease as the antenna height increases.

Personal/portable devices could be used inside a home or carried like a smartphone. Their output power is limited to 100mW, and it is reduced to 40mW on channels that are adjacent to one occupied by a TV station.

These new TVBDs have only been deployed on a limited basis so far, with the first installations limited to one county near Wilmington, NC, and one county near Roanoke, VA. It is expected that wider deployment will begin later this year. Only two fixed devices have been certified by the FCC as of this writing, but approvals for additional products are in the pipeline.

Unlicensed TVBDs are only allowed to operate on TV channels that aren't occupied by broadcast stations or other authorized users (like public safety agencies) in a particular location. The device is told which channels it can use by an FCC-authorized TV Bands Device Database. The device contacts a database (through a cellular connection, for example), tells the database its precise location, and the database sends back a list of available channels. Alternatively, an installer can set the initial operating channel manually.

ENG crews using wireless audio systems will need to learn about the new white spaces databases to help prevent interference from new TV-band devices.

Once operating, if the device is unable to contact a database for more than 24 hours, or if no channels are available, the device cannot transmit. If a portable device moves more than 330ft, it has to contact the database again to see if the original list of channels is still valid.

Clearly, these new TVBDs could potentially interfere with wireless microphones used by broadcasters in the field. Depending on your location, TVBDs could go on-air in your town next month or next year. Casual users of wireless mics may be content to take their chances with interference, but broadcasters and production companies can't tolerate unpredictable audio dropouts.

Dodging bullets

Fortunately, there are two ways to avoid interference from TVBDs. First, the FCC has set aside at least two TV channels in every market exclusively for wireless microphone operation. These are off limits to TVBDs at

Permitted users	Wireless mics Fixed TVBDs							Wireless mics Fixed and portable TVBDs																														
Local assignments	Public safety	Public safety	TV	TV	Other available	TV	TV	TV		TV		TV		TV		TV	TV	TV	TV	TV	TV	Reserved	TV	Radio astronomy	TV	TV	Reserved	TV		TV	TV	TV		TV			TV	TV
TV channel	14	15	16	17	18	19	20	21	22	23	24	25	26	27	28	29	30	31	32	33	34	35	36	37	38	39	40	41	42	43	44	45	46	47	48	49	50	51
Frequency range (MHz)	470-476	476-482	482-488	488-494	494-500	500-506	506-512	512-518	518-524	524-530	530-536	536-542	542-548	548-554	554-560	560-566	566-572	572-578	578-584	584-590	590-596	596-602	602-608	608-614	614-620	620-626	626-632	632-638	638-644	644-650	650-656	656-662	662-668	668-674	674-680	680-686	686-692	692-698

Figure 1. TV channels available exclusively for wireless mic operation at Shure headquarters

all times, providing a safe haven for wireless mic operation during broadcast events, including unplanned activity like breaking news coverage. The two reserved channels are the first open channels above and below channel 37 that are not assigned to a station in that market.

At Shure's headquarters in Niles, IL, for example, the reserved channels are 35 and 40. (See Figure 1.) With current technology, about 15 wireless microphones can operate in one 6MHz TV channel. Up to 30 can operate in two channels. That's enough to support the needs of multiple ENG crews.

The databases currently synchronize their data every hour, but later that will be nearly real time. Since TVBDs only need to check the database once a day, it's wise to register at least one day before the event.

In addition to the two reserved channels, there may be other available channels for wireless microphone operation that are also off limits to TVBDs due to restrictions on their use. Fixed devices are not permitted to operate on a TV channel that is adjacent to one that is assigned to a TV station or other authorized user in that market. Portable devices are not permitted to operate below channel 21. This means that adjacent channels between 2 and 20 are off limits to TVBDs and available for wireless microphone use at all times. At our headquarters, these are channels 5, 8, 11, 13 and 18.

Advanced planning required

What if you're covering a big event that requires more wireless mics than will fit into those channels? In cases like these, you can register in the same database that tells the TVBDs which channels they can use. The database will create a protected zone around your event and tell any TVBDs nearby not to use the TV channels that your wireless mics are using if they're within 0.6mi for fixed devices or 1300ft for portable devices.

Registering in the database is fairly simple for broadcasters and other licensed wireless microphone users. First, access one of the FCC-approved databases, of which there are currently two. (See the "Meet the databases" sidebar below.) Enter the date, time, duration and location of your event and the specific TV channels on which you want to operate your wireless mics. You'll need to enter your station's FCC call sign or, if you're a network or an independent production company, your Part 74 wireless microphone license call sign. (Unlicensed wireless microphone users can also register a special event in the database, but must apply to the FCC for approval and temporary access.)

The databases currently synchronize their data every hour, but later that will be nearly real time. Since TVBDs only need to check the database once a day, it's wise to register at least one day before the event. If things change — say a rain delay forces a golf tournament to be rescheduled for the next day — you can

Meet the databases

To ensure a competitive market, the FCC will appoint multiple database administrators. Two have been approved so far, and more are expected to follow. Each database contains the same information, but the appearance and features offered may differ.

Spectrum Bridge: *whitespaces.spectrumbridge.com/whitespaces*

Telcordia: https://*prism.telcordia.com/tvws/home/contour_vis.shtml*

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update things like the event dates or times in your database registration easily. Also, if you need to change which TV channels will be protected for you, you'll need to revise your registration. The only catch is that you need to use the same database administrator that you used to register originally.

Defining your location

Most of the time, the location where wireless microphones are used can be described as a single point, defined by latitude and longitude. But some types of broadcast events take place over a wider area, requiring a larger protection zone. A golf course or race track, for example, would easily exceed 1300ft in size.

In those situations, you can enter four distinct coordinates that define a polygon up to 1.8mi x 1.8mi in size. TVBDs will not operate on your protected TV channels when they're inside that protection zone or within 1300ft or 0.6mi of the edge of it. For events that cover an extremely large or irregularly-shaped area, you can register multiple polygons. It would be worth the effort to identify the coordinates in advance of potential venues where you might need to use large numbers of wireless microphones.

Thankfully, the system developed to protect wireless microphone operation in the coming age of TVBDs was crafted with detailed input from both broadcast users and wireless manufacturers, and it represents a thoughtful attempt to protect incumbent operations while opening the door to new creative uses of spectrum. While in some cases a few extra steps may be required to ensure interference-free operations, it is hoped that they will be easily adopted into the content production workflow and enable broadcasters to maintain the standards of production quality for which they are known.

Chris Lyons is Manager, Technical & Educational Communications, with Shure.

Improving loudness control

BY THOMAS LUND

2012 has been a CALM year. However, chances are that 2013 will be even CALMer, as the much-debated law on loudness in broadcast becomes effective in just a few months. But is it, in fact, already due for a revision?

No doubt, the CALM Act is most welcome and represents a huge step in the right direction with regard to eliminating annoying jumps in loudness in television. But, the broadcast standard that the CALM Act relies on may possibly be improved, as technology has evolved since the CALM Act was signed by the President. Therefore, the focal point of this article will be ATSC's A/85 standard as it applies to broadcasters in the U.S., and not least how and in which areas it may be able to be updated for an even better performance with regard to measuring and controlling loudness in television.

However, two additional broadcast standards will also be mentioned: The International Telecommunication Union's (ITU) BS.1770 will be discussed, as it is in fact the "mother standard" upon which all other broadcast standards are built, and the European Broadcast Union's (EBU) R 128 will be discussed to examine whether applying specific R 128 tools to A/85 would be able to enhance ATSC's A/85.

Loudness monitors such as the TM9 by TC Electronic owe their existence to the Communications Research Centre (CRC) in Canada, which designed a simple, yet effective, model for measuring "loudness."

The starting point

In 2006, ITU introduced BS.1770, which was the first international loudness standard. It defined the core principle upon which virtually all other broadcast standards now rely. Until then, the audio industry at large had been struggling with various peak-level-based meters that made commercials and new pop/rock music appear systematically loud. Thanks are owed to Communications Research Centre (CRC) in Canada for designing a simple, yet effective, model for measuring "loudness." After verification in numerous independent studies, the CRC model made it into the ITU BS.1770 standard, transparent and free for the world to use.

Without getting into all of the technical details, the measurement was extended from its original mono to also work with stereo and 5.1 programs. A so-called K-weighted filter curve (defined by the above-mentioned research results) is applied to each audio channel, which, in fact, builds a bridge between subjective impression and objective measurement.

Both A/85 and R 128 build on BS.1770, but in 2011, ITU updated its standard to version 2, also known as BS.1770-2, which integrates further improvements to loudness-measuring tools. These improvements had already been proven efficient by Japanese, Brazilian and European broadcasters. However, ATSC decided that A/85 should continue to rely on the previous version — let's refer to it as BS.1770-1 — which is a bit unusual, as it is common scientific practice to refer to the latest revision of any standard. Normally, it would not be necessary to distinguish between 1770-1 and 1770-2, but simply refer to the standard as BS.1770 (in its latest version, whatever number that might be).

No doubt, the CALM Act is most welcome and represents a huge step in the right direction with regard to eliminating annoying jumps in loudness in television.

Different approaches

One specific part of the BS.1770-2 revision is essential — the gating scheme when measuring program loudness. This method prevents, e.g., long periods of silence or atmospheres in a movie, to affect the overall measurement undesirably. In short, this means that it is now possible to align different types of (television) programs in terms of loudness. News, sports, commercials, concerts, talk shows and movies can actually co-exist without viewers having to adjust the volume over and over again, which is exactly what caused the complaints that led to the CALM Act in the first place.

The gate, however, is not the only difference between A/85 and R 128. While R 128 measures the audio signal in full with the employed gating scheme, A/85 aims at detecting the speech part of the signal and using that as an anchor point for the measurements. Although this method can work for determining gain offsets between dialogue-based programs of a certain genre, it has proven to be ineffective when it comes to aligning many different types of programs.

For instance, one obvious, potential problem is that not all programs contain speech, or use it like it is used in movies. Furthermore, who determines what is "speech" and what is not? The proprietary and patent-protected dialogue-detection algorithm A/85 relies on can sometimes interpret a violin as being a human voice, and conversely, not recognize a Swedish dialect. Under the "speech ruling," it would also be easy for commercials to become even louder by just keeping dialogue softer than other elements of a mix.

Faced with these challenges, ATSC recognized that the dialogue-based approach was not suitable for aligning interstitials on TV, and consequently new annexes (J and K) added in July 2011 stated that commercials were no longer to be measured using the speech-anchoring approach, but that all sources had to be taken into account. However, regular programs are still recommended to be measured using speech anchoring, and consequently, broadcasters will have to switch back and forth between different ways of measuring whenever a commercial appears.

To avoid this, it might be worth considering switching to one transparent measurement method, based on open standards, that works across all types of program material.

The gating scheme

Obviously, the one measurement suggested above is available already, namely in BS.1770-2. Its key feature is the gating scheme, so let's have a closer look at how that actually works.

The gate is activated when programs with a wide loudness range are being measured. In such situations, the measurement hones in on foreground elements and disregards the rest.

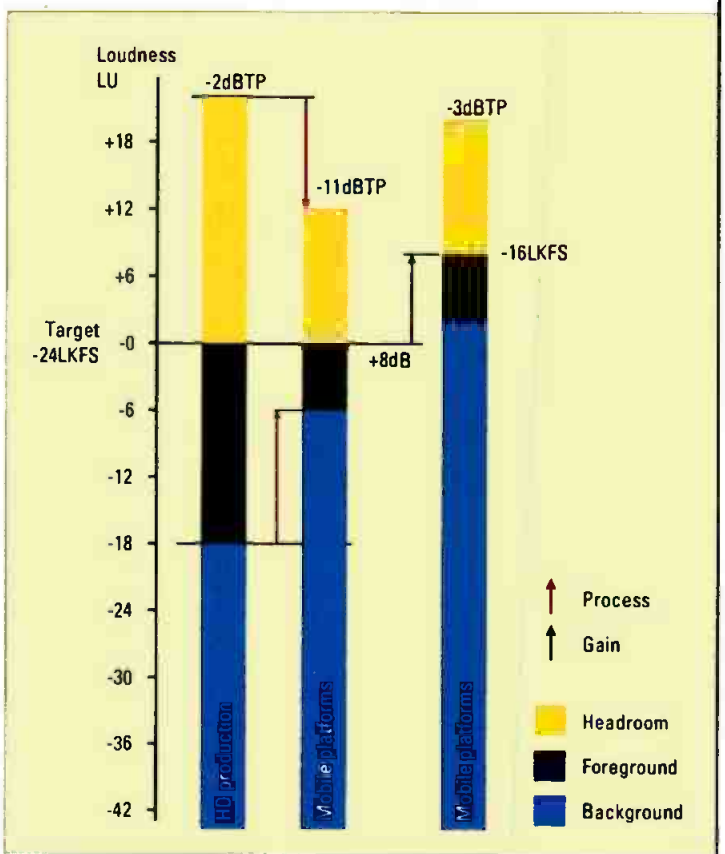


Figure 1. BS.1770-2, with its gating scheme, represents a giant step forward in audio quality — both in terrestrial broadcast and in mobile TV, podcast and analog distribution.

In practice, the gate takes long passages of silence or background audio into account by pausing the measurement of parts dropping below -10 loudness units (LU) relative to a measurement of the same program material without the gate.

Note how the BS.1770-2 gate is not set at an absolute loudness level (such as -34LKFS), which would be impractical and necessitate a new measurement in case a level offset was performed. Measurement gating is also a good help on the application side: With the previous, ungated technique (BS.1770-1), random parts of silence before and after a program could influence the result, making it virtually

impossible to obtain the same number twice.

As a net result of BS.1770-2, a station is now able to gain-offset (normalize) programs based on their foreground loudness, which is what the audience prefers to hear. Furthermore, with optimum gain-offset taking place at the station, little or no dynamics processing is needed, and it becomes easy to cater to various broadcast platforms at a high audio quality. Finally, BS.1770-2-based normalization grants programs more transmission headroom than if normalization was based on BS.1770-1, or if it was based on speech.

Conclusion

Compared to previous peak-based or speech-based attempts of controlling level in broadcast, the BS.1770-2 standard is a remarkable improvement. As shown in Figure 1 on page 59, even if only digital television (like ATSC A/85) is considered, and mobile TV, podcast and analog distribution are disregarded, the new standard is a giant step forward for audio quality. A fully transparent loop, based entirely on open technology, may now be created between production, ingest, transmission and the home listener, as shown in Figure 2.

Consequently, "sausage processing" at the point of transmission should be considered a thing of the past. Instead, BS.1770-2-compliant metering in production, and at subsequent stages, allows transparent handling and normalization of audio in the chain.

For broadcast platforms based on AC3, the BS.1770-2 measurement also enables a more precise and cheaper setting of dialnorm metadata than ever before. Remember, AC3 decoders are not dialogue-specific, neither with regard to normalization (dialnorm), nor with regard to processing (DRC).

In an ideal world, no matter what the program type, the perceived loudness level would stay about the same throughout a full day of broadcast.

For the plethora of other platforms — mobile TV, IPTV, podcast and counting — BS.1770-2 provides easy and audio-conscious answers also, especially in combination with complementary measurements such

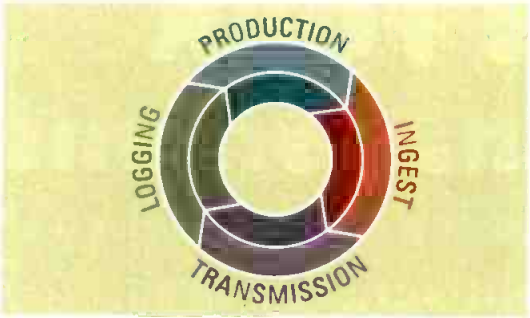


Figure 2. The new BS.1770-2 standard creates a fully transparent loop between production, ingest, transmission and the home listener.

as momentary loudness, short-term loudness and loudness range.

Under the new order, programs remain untouched, as long as their loudness range isn't excessive for the audience of a given platform. This is also an improvement over today, because AC3 processing (DRC) is typically routinely enabled during DTV transmission. Where this kind of domestic processing was supposed to be non-destructive, DRC has ironically become anything but: The home listener cannot turn off a primitive processor that is not even BS.1770-compliant.

In an ideal world, no matter what the program type, the perceived loudness level would stay about the same throughout a full day of broadcast, across channels, across platforms. We're close to that goal, and it all starts with the BS.1770-2 standard. Further complementary tools should not be dismissed, but the good news is that progress is being made constantly as users everywhere gain experience with the audio revolution that is taking place before our ears. The mere fact that broadcasters, as well as legislative assemblies, across the globe are now focusing on loudness solutions is a positive and welcome development that is sure to make the viewing — and, not least, listening — experience more enjoyable for everyone.

Thomas Lund is the HD development manager at TC Electronic.

Primestream's FORK Media Asset Management

The MAM system provides Discovery Latin America with a file-based workflow and automation.

BY JC SIERRA

Discovery Latin America, based in Miami, is unique among Discovery Communications' subsidiaries in that we do a little bit of everything. We cut our own promos and commercials, create our own graphics, and broadcast content to almost a dozen countries in Latin America and Brazil. Our facility's primary function is to take the programs produced for U.S. markets by Discovery and its affiliated networks (such as Animal Planet and TLC), and localize them for Latin American markets.

This means that a show that will air in Argentina, for example, has slightly different content from the same show that airs in Mexico or Brazil. This is because there are different cultures even among the Spanish-speaking countries. So we recut content to match the broadcast timing for the destination country, while the voice dubbing is performed in the destination country itself. Texted versions of Discovery's U.S. shows are retitled with Spanish and Portuguese graphics.

Teams of producer-editors (commonly called "preditors") within each regional group produce and edit their own spots on Final Cut Studio 3, Final Cut X or Avid Media Composer, taking in and sending out more than 100 hours of content per week. Managing all of that content is challenging, so an initiative is under way to streamline the content flow, starting with the adoption of a completely file-based workflow and system automation using Primestream's FORK Media Asset Management system.

With the adoption of Primestream's FORK Media Asset Management system, Discovery Latin America has streamlined its flow of content. Shown here is the facility's master control room.

Tapeless workflows

The customizable platform provides automatic scripting capabilities, which enable tapeless workflows. With the product's MAM production server and production client software, now everyone has a tape machine on their desktop.

With FORK's MAM production server and production client software, now everyone has a tape machine on their desktop.

The company worked closely with us to understand our workflow and determine exactly what we needed, and then developed and implemented a solution that's tailor-made for us.

Traditionally, preditors had to QC their spots from a tape machine, which usually meant a monopoly on the content or playback machine.

The production client is a versatile cross-platform software application that allows users to view the clips and modify their metadata, as well as perform a number of powerful and highly customizable action scripts available within the server application. Primestream customized our action scripts to perform not only all

automated actions in the background, but also actions initiated by the users as well. We have no need to touch any of the scripts because they simply work. If a script needs to be rewritten, a Primestream engineer can access our system remotely to make the changes.

Now every file that comes in, no matter what format, goes through a script. Using the "Send To" command, files are distributed to unique folders to be archived, transcoded or tagged before moving to a 378TB Xsan SNA or Spectra T950 LTOL5 tape archive. Because we still operate in SD video, some of the HD content from our two HD channels is first transcoded to SD.

A script registers the full-resolution files with Edit2Playout, a watch folder and media verification application. A render farm creates a proxy for everyone to view — as well as an SD MPEG-2/IMX 50 and DNx 145 file. Next, a video file transfer is created by triggering an action script. Discreet audio tracks are needed for multi-language subtitling or dubbing and once completed, craft editors insert finalized audio into the original programming.

Web client

Another feature that is exceptionally useful to us is the FORK Web-based Xchange client. This allows preditors and technicians to view, edit, add metadata and create markers on the proxy content through a Web interface. Because no client is needed, the user simply launches a desktop browser.

The Web client also provides mobile device support (Android and iOS) via an HTML5 media player. It has preset query fields that will automatically organize media into formatted, downloadable documents. Using a Web-based interface means that system maintenance and upgrades can easily be performed on the main server with little to no effect on the working environment. There are no installations. We simply do an upgrade on the backend and use the same Web link to access the server.

In my nearly two decades at Discovery, we've had to adopt several new workflows in order to adapt to changes in media technology. In FORK, we've found a media management platform that's flexible enough to adapt with us. BE

The FORK Web-based Xchange client allows preditors to view, edit, add metadata and create markers on the proxy content through a Web interface.

JC Sierra is Digital Media Engineering Manager for Discovery Latin America.

Gallery Sienna's MediaVortex

The system allows assets to be ingested locally, but logged and edited from thousands of miles away.

BY MARK GILBERT

Sporting events happen everywhere, all the time. Covering them can be an enormous task for broadcasters who are challenged with allocating resources to handle major international sporting events.

Sports broadcasters dream of remotely covering sporting events from their home base facilities, sending just a small set of equipment with a small crew to an event and leaving all the editorial staff back at their own desks. A new generation of technology leveraging fast global Internet connections and clever software solves the technical challenges and enables savings in cost and logistics, making or breaking budget-conscious outfits' plans to cover any given event.

Broadcast engineers in Televisa's studio used AJA and Sienna equipment to streamline the network's Olympic coverage.

A British developer of broadcast media infrastructure, Gallery Sienna, has created a solution to these challenges, in the form of the Sienna MediaVortex WAN server protocol and enhancements to the Sienna media asset management. Sienna MediaVortex allows assets to be ingested locally, but logged and edited from thousands of miles away. Ingest channels based on AJA Io XT portable Thunderbolt video I/O devices can also be controlled remotely. The finished edit can be delivered at locations on either end as a file transfer or via playout under remote control through an Io XT HD-SDI/HDMI interface.

MediaVortex from Gallery Sienna solved this problem using a UDP rather than TCP-based protocol, which is much less affected by latency, creating the equivalent of a file server over UDP. The system also

Using a Sienna web interface, operators at Televisa loaded a finished, conformed package of content into a baseband playout channel at the London Olympic site, and control real-time playout to air, or across a video link, using an AJA Io XT sent back to Televisa's London studio (pictured).

The solution

The main technical challenge is overcoming the inherent latency of a long-distance WAN connection, which in the case of a recent project at the London Olympics for Televisa, was around 250ms from London to the facility in Mexico. Since TCP messaging requires a reply to each packet before the next one can be sent, this latency is enough to bring TCP data protocols to their knees, dropping transfer speed from 100Mb/s to less than 1Mb/s.

MediaVortex from Gallery Sienna solved this problem using a UDP rather than TCP-based protocol, which is much less affected by latency, creating the equivalent of a file server over UDP. The system also understands the media being used, and intelligently handles QuickTime reference movies that are still being ingested, where the files being transferred are still growing in real time during the transfer.

This allows files being ingested via the Io XT to be propagated as proxies across the WAN during capture, enabling loggers and editors at the

remote end to start their work within seconds of the start of ingest.

The logistical task of managing media assets across multiple locations is helped with Sienna's "conjoined asset" concept. Each end of the WAN link has a Sienna MAM server, connected together in a Sienna distributed media cloud, where each side is aware of the other. An asset ingested in London, controlled remotely, is "propagated" to Mexico where it becomes a conjoined asset, with high resolution and proxy in London, but just a proxy in Mexico.

Both sides share a common key that conjoins them. Whenever metadata such as real-time logging markers are added to either side, the common key ensures the metadata is replicated at the other sites where the conjoined asset exists. This allows fully collaborative remote and local logging and editing. When remote proxy editing with Sienna's ImpulsEdit web-based editor is complete, a single click triggers an intelligent conforming process that seeks out the high-resolution media in the conjoined assets and performs the media assembly at the high-resolution site.

In the same way that the initial ingest process was remotely controlled from Mexico, so too can the playout. Using a Sienna web interface, remote operators can load a finished, conformed package into a baseband playout channel at the event site, and control real-time playout to air, or across a video link, using an Io XT. Alternatively, the package can be compressed using H.264 and sent back to the site in Mexico, where Sienna VirtualVTR can play it natively through an Io XT without the need to convert back into an interframe format.

Summary and benefits

In the example given for the London Olympics, using a 100MB Internet connection, Televisa was able to reduce head count from around 250 (mix of talent and editors/engineers) for Beijing to around 100 for London (mostly talent), and reduce hardware on site from six to two racks. The rest of the engineers and hardware worked from their permanent positions back in Mexico.

New technology delivers a solution to an old problem, opening up opportunities to save money and increase diversity. The benefits go beyond cost savings, by giving broadcasters a tool they can use to include niche sports which could not be justifiably covered with traditional on-site broadcast technologies. BE

Mark Gilbert is CTO of Gallery Sienna.

Appear TV's Multiscreen system

Multiplatform distribution expertise is key to survival.

BY CARL WALTER-HOLST

In the competitive world of broadcasting, being able to address a multitude of distribution formats has become mandatory for survival. In the beginning, there was terrestrial broadcast for widely distributed programming, although through a maze of different international standards. Next in the technical line was cable, followed in short order by satellite. Although broadcasters had to address two additional formats to distribute content, the end target was still a home television screen.

With digital broadcast standards in place and the emergence of Internet, Wi-Fi and, now, 4G programming pathways, standard computers and smart devices such as the iPhone and iPad are now capable of receiving content from any broadcaster anywhere on the planet. The problem with this avalanche of potential receiver units is the corresponding flood of distribution formats required to make each device happy, combined with the expectation of instantaneous availability of content on these distribution formats from its users. Clearly, solving the multi-screen dilemma is of the utmost importance for the modern broadcaster.

Go to the headend of the class

Although a number of manufacturers have introduced new dense, software-based transcoding and encoding platforms into the OTT/multi-screen marketplace, they are still unable to keep up with the large volume of incoming content to be processed. Furthermore, transcoding in itself doesn't address the additional functions needed to prep a program to be pushed to a segmenter, such as content acquisition, descrambling and the encoding of baseband signals, leaving a broadcaster to implement additional equipment. This workflow also brings with it the complexities of integrating different manufacturers' units into a workable and reliable system.

Solving the multi-screen dilemma is of the utmost importance for the modern broadcaster.

The fully integrated Appear TV Multiscreen system simultaneously prepares multiple signals from multiple sources in multiple formats for distribution to an HD television in the home, for a high-resolution computer screen, and lower-resolution Web- and mobile-based profiles, all at different bit rates optimized for each destination device. This is possible through the new multi-screen encoding and transcoding cards for

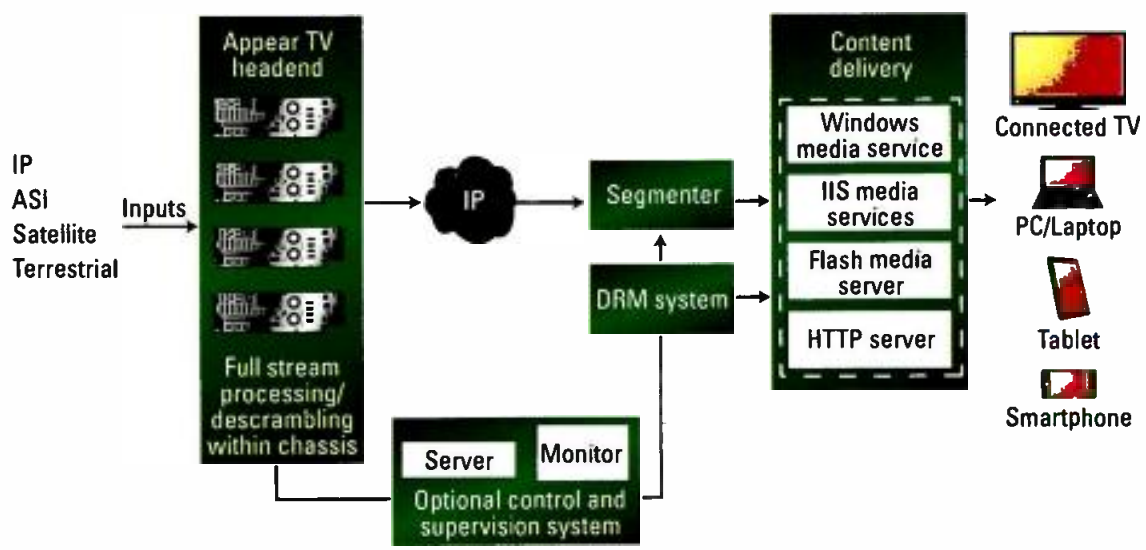


Figure 1. The Appear TV Multiscreen's open architecture allows it to receive signals through a variety of interfaces — including XML and SOAP — and deliver them to devices ranging from connected TVs to smartphones.

the Appear TV Unified headend system chassis. The cards, along with the rest of the system, offer a scalable option that can be integrated into an existing system to expand capabilities or implemented from the ground up for a new headend.

The future is in the cards

The Multiscreen encoding and transcoding cards provide a dense transcoding capability, supporting multiple profiles and bit rates simultaneously for multi-screen delivery. From one card, it is able to handle much of the preparation needed before content is sent from the main broadcast hub to local affiliates or online outlets. Its functions include input service replication, resolution change, interlace-to-progressive conversion, graphics rescaling and key-frame alignment. The end results of this setup are key-frame-aligned outputs in transport-stream format that are easily pushed along the broadcast chain.

Many similar solutions in the market offer software-based dense transcoding platforms, but only have IP interfaces with no provision for accepting baseband SDI/HD-SDI inputs or to turn around services from any input source. The Multiscreen system also provides the capability to support baseband inputs for encoding, as well as transcoding while maintaining high density and optimal systems integration.

For those looking for a complete system, the Unified headend chassis can hold a range of purpose-built cards, in addition to the encoding and transcoding cards, to handle switch IP/IO control, descrambling, audio leveling, DVB-S/S2, terrestrial, cable demodulating, multiplexing, modulation and general-purpose IP-IO, among many other capabilities.

Although current software-based alternatives focus on dense transcoding, they lack flexibility and functionality in other key areas.

Keeping IT simple

In terms of its core operations, all functionality within the headend can be configured and monitored using its integrated web-based user interface. Additionally, the system has a completely agnostic, open architecture with interfaces that include XML, SOAP and other industry-standard protocols. In this way, the Multiscreen system can work in conjunction with a range of installed systems, making it easy to implement enhanced capabilities for little investment.

With segmentation playing a big part in the distribution of these new streaming formats, the transcoding card can be configured to be compliant with a number of third-party segmentation devices. This allows broadcasters to maintain the segmentation devices already in place while still expanding their capabilities.

Multi-function solution

Broadcasters are in need of a single, multi-function platform that is capable of offering significant transcoding density, while providing comprehensive coverage of the other critical pieces required for a multiformat, adaptive-stream headend system. Although current software-based alternatives focus on dense transcoding, they lack flexibility and functionality in other key areas, forcing facilities to combine systems from multiple manufacturers. The new Multiscreen system is designed to resolve all of these problems. BE

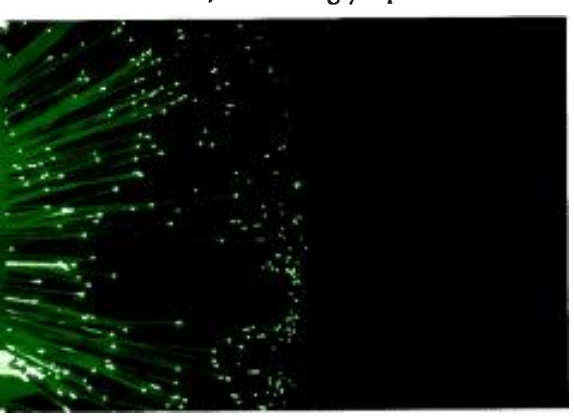
Carl Walter-Holst is CEO of Appear TV.

Fiber optics

The industry is shifting toward fiber as the preferred choice for connecting television signals.

BY JOHN LUFF

Bandwidth is not infinite, at least not in our restricted media technology enterprise. Cost and innovation are drastically affected today by our ability to interconnect signals of many types over distances no shorter than what we have been used to with coaxial video interconnections. While copper-based interconnection technology has key roles in modern facilities, increasingly optical fiber has important attributes that facilitate technology that traditional wired infrastructures cannot duplicate easily, or at least over practical distances.



Fiber is immune to electromagnet interference and, hence, works well in high RF environments where coax is at best difficult to use.

An evolution

From a technology standpoint, we are evolving away from infrastructures in which a single wire (or, more properly, cable) carries a single signal. Most current wiring diagrams show real devices connected by point to point, almost exclusively unidirectional pathways. This is because the signals carried are analog in physical nature, even if signals like SDI and AES digital interconnections are considered.

To make those baseband digital signals work requires sending analog modulation on the bearer, whether coax, paired wiring or fiber-optic cable. As we increasingly embrace IT-based technology, network technology allows us to overcome key aspects of the current interconnection model. First, to fall is the unidirectional nature of baseband interconnections. Second, and particularly affected by optical interconnection methods, more than one signal may be carried on a single cable. In fact, multiple signals can be carried in both directions on a single cable. This has broad implications for the topology of interconnection in single-site operations, as well as in multiple-site operations.

It is important to note that while analog and digital baseband signals occupy significant bandwidth in a cable, optical techniques are not normally amplitude modulation of the light, but rather FM, or more properly pulse modulation of FM. A second important factor is that the "carrier" frequency, i.e. the frequency of the light, is much higher. How high? A typical 1300 nanometer laser is about 230,000GHz. That means a lot of information can be modulated onto one beam.

Using CWDM, you can put about 18 beams in a single fiber, and with DWDM, you can put upwards of 40 and up to 128 wavelengths in a fiber. If you combine the ability to handle high data rates with potentially many video and audio signals in a single data stream, plus DWDM technology, it is easy to see how a single fiber has capacity that is quite astounding.

On my desk I have a short sample of fiber that is now more than 10 years old. One PVC "pipe" carries 864 fibers, each of which can be DWDM modulation. By my simple math, that is 110,592 wavelengths in a bundle that could run short or long distances. That makes any coax look puny.

Benefits

There are other reasons why fiber is attractive for connecting television signals. First, fiber is immune to electromagnet interference and, hence, works well in high RF environments where coax is at best difficult to use. It is lighter in weight, does not carry an electrical current (or ground current either), does not radiate energy, is quite small and can be used in corrosive environments.

But, there is always a catch. There are mitigating factors that in some applications make fiber a difficult choice. In some applications, particularly where high capacity and short distance connections are needed, fiber can be much more expensive than copper interconnection.

Copper is easier to install where "splices," or connectors, are needed. Copper can also carry power where fiber cannot, which is particularly important in some applications. SMPTE standardized fiber-optic camera cable for HDTV applications many years ago, but specified to power conductors and two twisted pairs in the cable to facilitate both power and communications. That hybrid camera cable has been quite successful at providing the best of both worlds — the convenience of copper and the capacity of fiber connections.

Other considerations

There are other questions that enter into any decision tree when considering fiber interconnections. Does the "cable" need to carry bidirectional signals copper losses? If long distance or high capacity improves the business case for the investment,

fiber wins. If resistance to mechanical stress, like under foot or around tight bends, is needed, then copper might be better.

To be sure, there are "tactical" fiber solutions invented for military installations and routinely used on the battlefield. Though not quite as dangerous, ENG and EFP applications commonly use tactical fiber interconnections, which can be as rugged as copper and work at much longer distances.

But to me the question always comes back to this: Where is technology headed in the long term? I strongly believe that the evolution of the industry is away from point-to-point signal routing and is moving toward networked topology for all kinds of facilities. TCP/IP (or UDP/IP) predominate today, and likely will for some time into the future as media networking schema. Their longevity is only enhanced by the Audio Video Bridging (AVB) standards initiative.

It won't be long before video server ports won't have separate input, control and output connectors, but rather a single connection to the network. Cameras won't need return feeds, teleprompter feeds, and RGB/YRyBy and monitoring outputs. Rather, they'll need a single optical connection that carries everything, including communications, metadata from the robotically controlled pan head and focus data from an operator in a remote location.

The potential implications of this for our industry cannot be overstated. Many hardware platforms would likely move to software-centric, and manufacturers of IT hardware would play even more strongly in broadcast and media facilities. Because of movement towards 4K imaging, 3-D applications and the huge bandwidth needed for high bit depth at high resolution, it is fair to assume that copper interconnections will continue to wane, and fiber will be strongly ascendant. Our industry will be vastly different, and the change will be for the better. BE

Fiber is lighter in weight, does not carry an electrical current, does not radiate energy and is quite small.

John Luff is a digital television consultant.

? Send questions and comments to: john.luff@penton.com

Harmonic — Spectrum MediaDeck 7000 SSD

Media server system is designed to offer the exceptional reliability, low power requirements and lower weight of solid-state disk storage, while providing the broadcast quality and cost-effective design of the MediaDeck 7000; equipped with four 480GB SSD drives; provides 57 hours of 50Mb/s storage capacity; packs up to four SD or HD video channels and GigE connectivity into its 1RU chassis; features multiple-codec playback, up/down/crossconversion and HD/SD simulcast capabilities.

www.harmonicinc.com

Blackmagic Design — UltraStudio4K

Compact, single-rack unit "breakout box" capture and playback system for Mac and Windows includes a wide range of rear video and audio connectors; front panel features an integrated color LCD, as well as video and audio input buttons; power supply is built-in; rear panel includes virtually every type of video and audio connection that exists, all using standard connectors; supports all video formats, including SD, HD, 2K and 4K playback; supports the new Dual Link 3Gb/s SDI connection for handling 4:2:2-based 4K; can take advantage of long Thunderbolt cables that can be up to 98ft away from the host computer.

www.blackmagicdesign.com

Litepanels — 1x1 LS

Newest fixture in the 1x1 LED panel series; designed for those who do not need DMX control; offers many of the features found in the 1x1 panel fixtures but at an entry-level price; available in daylight color balance, either flood (50° beam) or spot (30° beam).

www.litepanels.com

T-VIPS — CP524

TS adapter supports multi-stream (any input to any output), remultiplexing and flexible format conversion; designed to meet the requirements of operators and service providers for repacking and delivery of content for multiple end points; its filtering and remultiplexing features enable operators to save valuable bandwidth; equipped with reliability features such as hitless switching with diversity reception, service fallback and T-VIPS Embedded Redundancy Control to ensure 100-percent uptime even in the case of severe packet or link loss.

www.t-vips.com

BHV Broadcast — Syntax Essence

Low-cost version of the Syntax up/cross/downconverter; offers the same broadcast specifications as it predecessor but with a 50-percent price tag; based on Super-Resolution Bandlet Technology; brings the performance advantages of motion-compensated processing without the associated disadvantages of high cost and occasionally severe artifacts; full Syntax range offers options such as analog video inputs and ARC, which are available as extras on Syntax Essence.

www.bhvbroadcast.com

Front Porch Digital DIVAdirector V5.1

Media asset management system is a permission-based Web application that enables complete access to file-based content stored by DIVArchive CSM systems; contains new features that make it even easier for any media organization with a DIVArchive system to manage its digital files, especially if that organization is using the LYNX[sm] platform for moving, archiving and online video publishing to the cloud; available as a cloud service or licensed locally; offers enhanced integration with other components in the DIVASolutions product line.

www.fpdigital.com

Leader Instruments — LV 5837

Audio monitor accepts up to eight AES/EBU stereo digital audio feeds at sampling frequencies from 32kHz to 190kHz; has a wide range of measurement capabilities, including audio level metering in dBFS, Nordic, BBC and DIN scales; up to 16 audio channel bargraphs can be viewed simultaneously on the instrument's 6.3in TFT LCD color screen; peak hold can be switched between true-peak, PPM Type 1 or PPM Type II modes and zero to five seconds duration in 0.5 or one-second steps.

www.leaderamerica.com

Calrec Audio — Artemis Light

Newest member of the Artemis family of audio consoles; new feature is a compact processing rack dedicated to delivering digital signal processing and routing capabilities in a 4U enclosure; like all Artemis consoles, Artemis Light incorporates Bluefin2 high-density signal processing and Hydra2 networking technologies in the same compact yet scalable control surface used by Artemis Shine and Beam; employing the same hardware and software architecture, the console can be fully integrated with any existing Hydra2 network.

www.calrec.com

Primestream FORK Xchange Suite 1.5

Gives broadcasters instant Web access to content on their FORK Production servers from any Windows, Mac or iOS device; contains new capabilities for creating subclips and markers, as well as an upload manager, extended metadata functionality, multitrack audio control, and more new features that make it even easier for media enterprises to manage content efficiently among multiple FORK Production suites; supports multiple frame rates according to both NTSC and PAL standards.

www.primestream.com

Hamlet — Precision HDW7

Portable monitor's small, lightweight enclosure houses a 7in In-Plane Switching screen, designed to offer vivid, clear pictures and a stable response time; displays consistent and accurate color from all viewing angles; adjustable clip level, under and over luminance, and false color capability are enhanced further by the built-in clear and absolute waveform, RGB and vector traces; HDMI input is complemented by a 3G, HD, SD-SDi input and loop and headphone audio output.

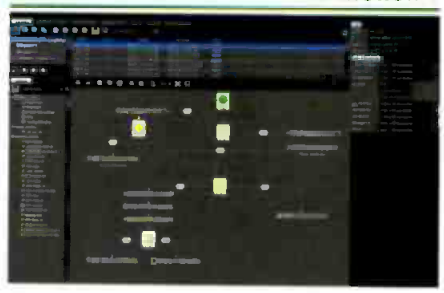
www.hamlet.co.uk

Ericsson — SVP 5500

HEVC/H.265 encoder is designed for the delivery of live and linear TV over mobile networks to mobile devices; capable of real-time encoding at resolutions up to HD; reduces bandwidth requirements for current and future services; enables operators to deliver consistent, high-quality TV experiences on mobile devices and to meet growing consumer expectations for TV Anywhere services, whether in the home or on the move.

www.ericsson.com

Harris — Invenio Motion 4



Designed to represent a reinvention of media asset applications from a simple workflow engine to a comprehensive media management system that integrates people, applications and devices into a holistic workflow; allows users to add, configure and manage devices, as well as assign tasks to personnel and track projects from inception to completion; consolidates multiple asset management tasks within its engine, bringing together many transcoding and storage options.

www.broadcast.harris.com

Orad — Maestro 6.7

Enterprise graphics system addresses end-to-end graphics creation and distribution workflows for real-time, pre- and post-broadcast production environments; seamlessly integrates with industry-standard newsroom, automation, traffic and editing systems; key new feature is the GMAM system, a graphics-oriented media asset management technology fully integrated within Maestro; additional new features include an order management system that better manages asset requests for video clips and images, and a SceneEdit module, which lets users create a completely new graphic design or modify the look of existing graphics from the playout station

www.orad.tv

NUGEN Audio — ISL

Inter-sample true-peak limiter is based on the standardized true-peak algorithms of ITU-R B.S. 1770 and related standards; suitable for the control of audio for post production and broadcast applications; true-peak limiting can also be used to ensure that downstream codecs (mp3, AAC) do not introduce distortion into the signal; although ISL has been designed for limiting relatively dynamic high-quality audio, it can also be used to hard limit and reduce dynamic range considerably if this is required.

www.nugenaudio.com

NOA — *ingest*LINE FrameLector

Video ingest module for typical SD material such as betacam SP; ingest tool provides scene detection, lossless compression on open source standards, remote control and an infinitely scalable, multi-stream parallel solution, with complete previewing.

www.noa-audio.com

Grass Valley — STRATUS

Latest version of the media workflow application framework includes archive, conform and metadata management enhancements; this functionality, when combined with ENPS, Octopus or iNEWS newsroom computer systems, permits users to access all of their STRATUS tools within the NRCS — streamlining the entire news production process; these news tools also facilitate the unlimited sharing of clips and content creation tools between collaborative workgroups or individual users.

www.grassvalley.com

Chyron — Axis World Graphics

Newly enhanced mapping system provides a more interactive experience for producers, as well as faster renders and many new features; order management tools offer new point allocation systems that enable more efficient management of graphics teams; custom order forms offer flexibility for handling many different asset types; additional new features include integrations with Apple Final Cut Pro and Adobe Premiere Pro video editing.

www.chyron.com

Small Tree GraniteSTOR TITANIUM4

Four-drive video editing shared storage system supports 2TB, 3TB or 4TB disk drives; its unique design enables video editors with limited resources to focus on editing video projects rather than trying to locate critical files off portable storage drives; equally effective in studios or on the move with a mobile production crew; designed to be simple to set up and manage; storage is available within minutes of start-up; is configurable from 8TB to 16TB of storage capacity, while driving nine streams of ProRes 422.

www.small-tree.com

Thomson Broadcast — Futhura GreenPower

Transmitters are now equipped with an increased operating range from 1.6kW to 11.6kW – UHF wideband; cooling system has been redesigned to provide further energy savings and ability to adapt to any environmental conditions, offering a 50-percent improvement in efficiency over current standard transmitters on the market.

www.thomson-broadcast.com

Utah Scientific — UTAH-100/UDS Universal Distribution System

Signal distribution system combines the flexibility of a multirate digital routing switcher with the economy of simple distribution amplifiers; modular system is based on I/O modules with 16 ports, interconnected by a cross-spoint fabric that allows any input signal to feed any number of output ports.

www.utahscientific.com

Shotoku Broadcast Systems — **SX300VR**

Tracking pan-and-tilt head is designed as the perfect companion for the TP200VR pedestal; features the company's reliable and accurate VR tracking technology; designed to offer a continuously adjustable perfect-counterbalance system; VISCAM ultimate fluid drag system ensures smooth, adjustable pan-and-tilt drag with enhanced torque.

www.shotoku.co.uk

Jampro — **RCCC**

Combiner family has a compact modular design that can be configured to fit into the smallest transmitter rooms; additional frequencies can be easily added when the time comes; uses temperature-compensated bandpass filters, which have an integrated heat sink top; this keeps the filters cool and locked on their frequencies; various models are available for different channel spacing.

www.jampro.com

EVS — **C-Cast**

Content delivery platform allows viewers to use their smartphones, tablets or laptops to access extra content and view clips the way they want; latest features include the ability to replay clips at variable speeds and the ability to import third-party statistics and analysis information integrated into a timeline; offers advanced integration with third-party Web systems.

www.evs.tv

Broadpeak — **BkM100 Mediator**

CDN management software enables network service providers to deliver video content to any screen, offering viewers a superior quality of experience over any network; includes an advanced caching mechanism that optimizes storage at edge levels; using the new content priority management tool, users can prioritize pay and free content to distribute CDN video services more efficiently.

www.broadpeak.tv

Crystal Vision — **Safire 3**

Real-time chroma keyer works with 3Gb/s, HD and SD sources; new processing features add more realism to the key, with shaping and softening of edges, key noise reduction, color-spill processing and 2D compensation for uneven illumination of backdrops with separate corrections for linear and radial lighting problems.

www.crystalvision.tv

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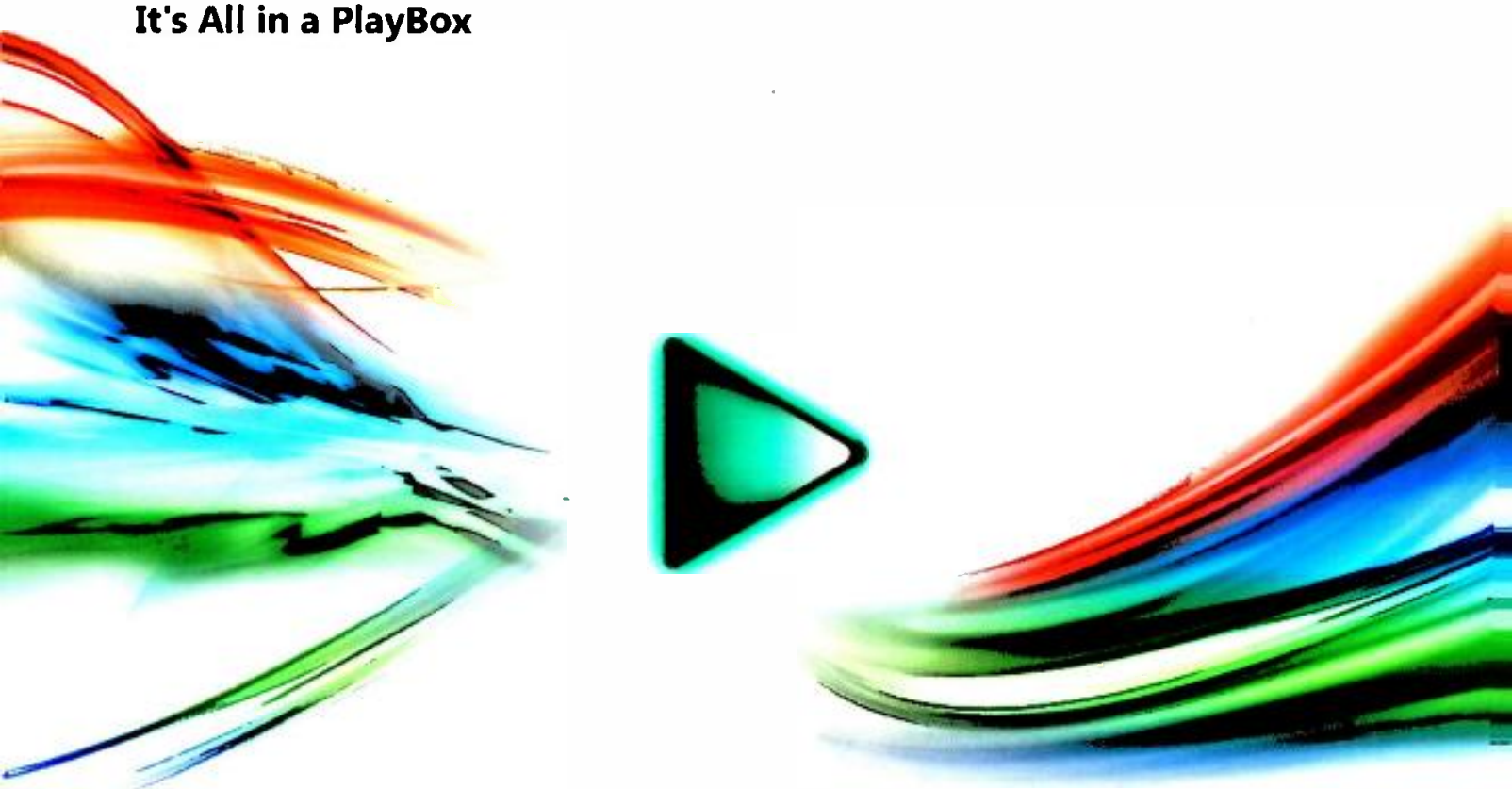

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