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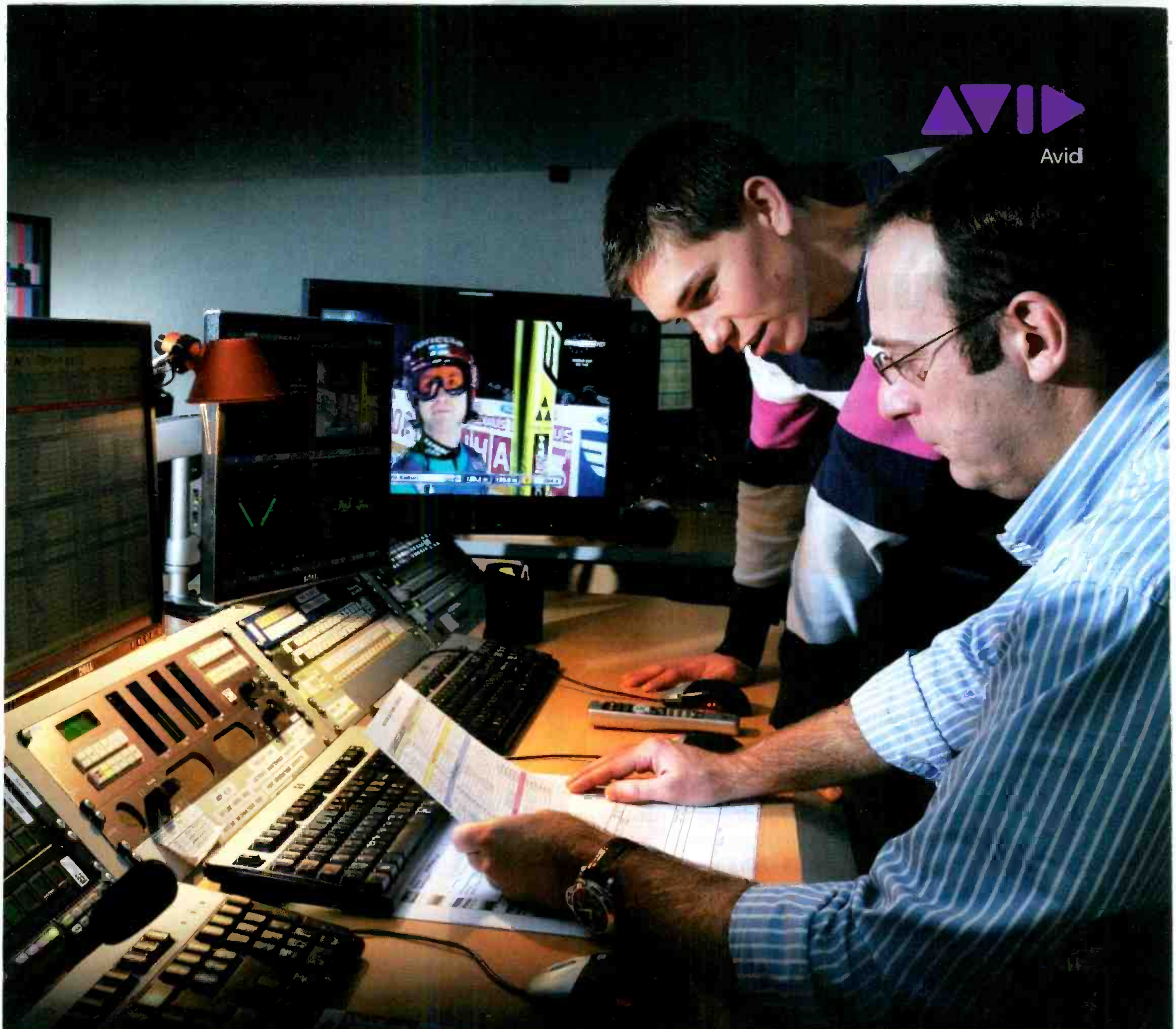

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TABLE OF CONTENTS

VOLUME 53 | NUMBER 9 | SEPTEMBER 2011

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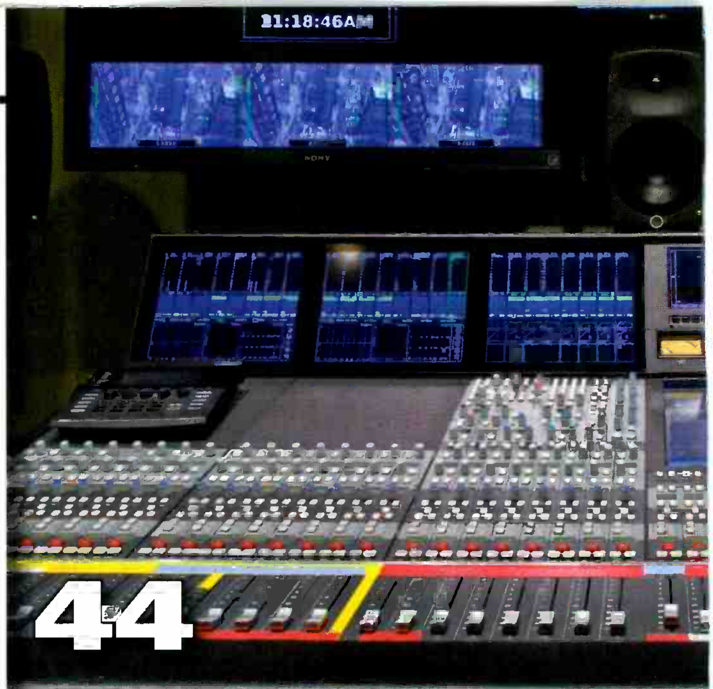
FEATURES

BEYOND THE HEADLINES

continued on page 6



ON THE COVER:
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TABLE OF CONTENTS (CONTINUED)

VOLUME 53 | NUMBER 9 | SEPTEMBER 2011

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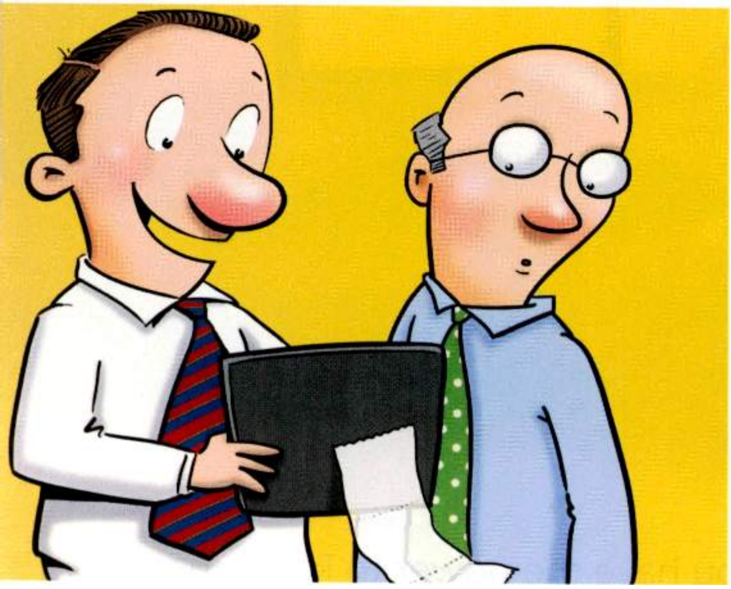
Tethered users and toilet tablets

A late August e-mail subject line captured my attention. While the e-mail appeared to have little to directly impact OTA broadcasting, the title could not be ignored: "Owners take their tablets to the restroom." As curiosity got the better of me, I clicked to read the entire press release.

It seems that Staples Advantage, a division of the Staples office supply company, conducted a survey to see how tablet owners were using their portable devices. The study asked 200 owners of the computers, across multiple companies and industries, about their use of the technology. The study revealed that 60 percent of tablet users admitted to taking their devices on vacation, and almost



80 percent said they use them in bed. A third of tablet owners said they take them to restaurants. Okay, I can believe these numbers, but it was the next study data point that caused me to pause.

The study showed that more than a third of tablet users, 35 percent, take their devices into the restroom. Clearly, one of the key advantages of these relatively new devices is their portability. One often sees tablets being used in restaurants and other non-office spaces. Combined with Wi-Fi, they can be an effective communication tool. And for parents, tablet games help address the issue of fidgety kids when dining out or traveling.

If broadcasters have their way, viewers soon will be able to connect USB receivers to tablets and enjoy OTA DTV while away from the home and even on-the-go. After all, connection-free OTA is one of broadcasters' several advantages over cable.

Tablets also enable a wide range of both business and gaming applications to be carried out on portable and larger screens. No external or folding keyboard is required; owners can type right on the screen. On the surface, a tablet solution seems ubiquitous and perfectly suited for today's busy and fast-paced executive.

But to me, it just seems creepy that some Americans have become so tethered to technology, they cannot even use the restroom without being tethered to an electronic gadget.

A telephone survey asked adults if they always washed their hands when in public restrooms. While 96 percent claimed they did, actual observations indicate otherwise.

A survey from the American Society for Microbiology and the American Cleaning Institute reported that in restroom observations, 93 percent of women washed their hands in public restrooms, but only 77 percent of men did so. But that's an improvement over the results from 2007, when only 66 percent of men did.

I've observed some of those non-washers in our company restroom. Call me a phobic, but I even employ the paper towels used to dry my hands to open the restroom door. If nothing else, such hygiene can help prevent the spread of colds and flu.

All this leads me to regard the use of any tablet computer, which may have been close to a recent flush, at a dinner or restaurant table less than appetizing.

The next time someone wants to share something with you on their tablet, let them hold it. Or, ask them where it's recently been. BE

Brad Dick

EDITORIAL DIRECTOR

Send comments to: editor@broadcastengineering.com

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Captioning's future

As platforms continue to expand, dubbing's alternative is an economic means to use and resell assets.

BY ED HUMPHREY

Generally, there are two reasons for captioning content: for accessibility — both for the hearing impaired and for when audio is unavailable (e.g. within public spaces such as at airports, gymnasiums, etc.) — and for language translation to enable access to content by foreign language audiences.

Particularly in the digital television, multichannel world, an increasing amount of content is reused across regions. Captioning provides an economical means to allow broadcasters and content providers to reuse and resell their assets across multiple platforms. (See Figure 1.)

Dubbing may make sense in some regions. However, in terms of workflow, cost and multi-language access, it does not provide the optimum solution.

The alternative to dubbing — captioning — provides a number of advantages that make it a highly attractive solution for reaching the greatest number of viewers. The reduced cost makes it more viable for a greater range of content. It is easier and more practical to offer captions using efficient state-of-the-art workflows, especially for content that is live-to-air or has completed production very close to time-of-air. And, captions provide the greatest accessibility not just to the main program content, but also to advertisements, which widens the demographic served by the advertiser. Equally, as noted above, captions allow for content to be accessible in public spaces.

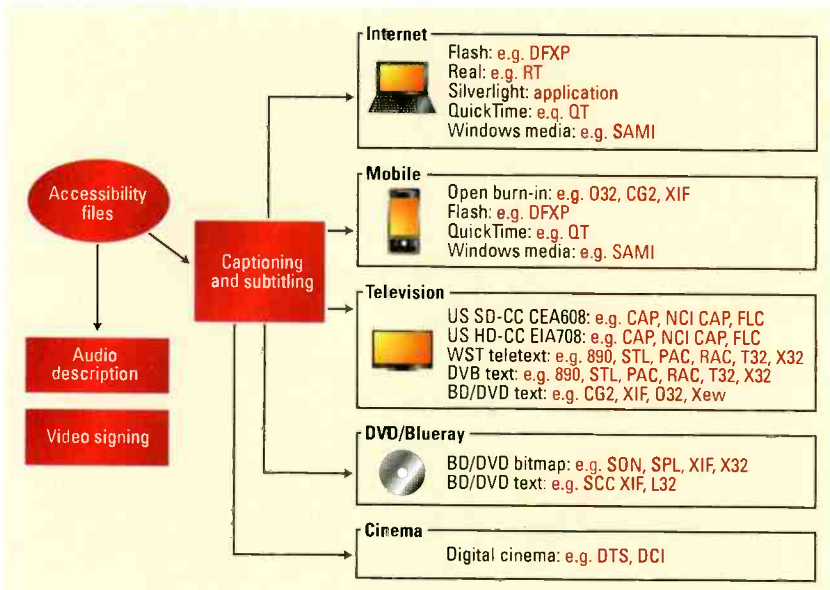


Figure 1. This chart shows the wide range of platforms to which captioning and subtitling can be deployed.

Multi-language captioning

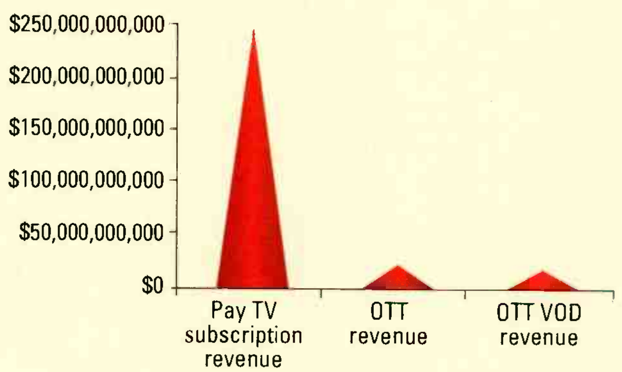
In addition to the necessity to comply with legislation in the target locations, captions provide an inexpensive way to open up content to the wider local and international audiences through language translation.

Depending on the format of captioning used, it is possible for multiple languages to be delivered to viewers who then can choose their preferred language. In formats such as CEA-708, WST Teletext and DVB, it is also

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

Looking ahead, traditional pay TV still tops

OTT products continue to evolve and expand, but projected revenues are still far behind.



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possible to switch on and off captions via remote control.

Captions also provide a rich source of content-relevant metadata for the video asset. Enhancing the asset with this additional data aids repurposing, as well as enables detailed searching and indexing of content. With integration into centralized media asset management systems, content can also be more easily monetized through clip resale, redistribution and syndication.

Figure 2. Once captions have been created, they must be bound to content. Options have increased as encoding technologies have grown to faster-than-real-time capabilities.

Evolving production and broadcast workflows

To aid format and resolution conversions for diverse distribution formats, many broadcasters want to store video assets as a single common mezzanine format. This represents the highest quality version. Thereafter, all subsequent broadcast and streaming versions will be derived from it.

To optimize repurposing, the storage of caption data should align with this principle and be stored as a high-level generic form of caption data. With this approach, there are two key methodologies to consider. One relies on the creation of a master caption, which has as much information as possible related to it. This allows less sophisticated derivatives to be readily produced. In effect, this becomes the mezzanine format caption and relies on informed choices being made during the creation/preparation phase for presentational aspects such as font, color, positional and alignment information, drop shadow, and character edging.

Alternatively, there is the transcode approach, which relies on a lowest common denominator format file (such as an CEA-608 compliant caption or Teletext caption) being created and effectively upconverted to the target format. This will optimize ease of use and speed, but does not take advantage of the sophisticated options available within higher-end standards (such as CEA-708, DVB ETS-300-743 and DVD Bitmap).

Many broadcasters choose to create a hybrid of the two. They implement some of the capabilities while limiting the overall time dedicated to creating the caption by constraining and automating some choices. When considering which approach to take, it is worth noting the variations that different standards offer in terms of levels of control and sophistication.

Captioning for file-based workflows

The caption creation technologies most commonly in use fully support nonlinear video — thereby allowing content providers to digitally send encrypted and/or watermarked video clips. This saves costs associated with tape-based methods.

The best creation systems allow for far greater caption operator productivity through technology enhancements such as shot change detection, semi-automated time-coding functions, advanced reading speed algorithms and so on. Modern caption authoring systems should also be Unicode-capable, in order to create and repurpose captions in almost any language. For complete flexibility, they also should fully cater for HD, VOD, Web, mobile, digital cinema and the burgeoning stereoscopic 3-D space.

The employment of a wide network of caption operators makes expedited delivery ever more possible. Often, these operators are located around the globe in order to maximize time zone benefits. As a result of this highly distributed mode of production, a substantial worldwide network of professional freelance caption operators has emerged, equipped with the latest creation workstations and using high-speed broadband links to receive secured video clips and job instructions.

Finally, creation is sped-up through caption agencies sending proofs back to their clients electronically. This is achieved by generating an all-digital approval, which has captions overlaid to video so that the client can quickly and easily assess placement, timing, font choice and other factors.

Binding captions to content

After the creation phase has been completed, the caption or subtitle data must then be bound to the content, enabling presentation to the viewer when they watch the programming.

This binding can be considered as occurring in one of three periods of time, as illustrated in Figure 2:

- *Early binding.* The pre-prepared file is linked to the program content well ahead of transmission.
- *Late binding.* Similar to early binding, but occurs closer to air time and only becomes possible due to faster-than-real-time encoding technologies.
- *Live binding.* For either truly live content or for pre-prepared content that only becomes available very close to airing, thereby eliminating the possibility of pre-binding captions.

In modern workflows, files are either sent for time-of-air transmission (a live bind), or are transcoded into a

file-based video asset (during early or late binding).

Driving time-of-air transmission will be a system that tightly integrates into the automated workflow of a master-control facility — with the caption playout system approving files in advance of airing and then airing the correct file at the right time automatically, either with or without external time-code.

The time-of-air system can also be used as a gate keeper for real-time captioning, where the system authenticates the caption operators and their work slot, prior to allowing pass through to air.

Additionally, the time-of-air system can extend the control of the automation system over the live captioning by switching the data source to the in-line caption encoder based on the automation schedule. This removes the dependency on the live caption operator to remember to manually switch control at the right times. This prevents situations such as open connections blocking playout of pre-prepared content during commercial breaks.

Increasingly, a hybrid of ingest and time-of-air methods is becoming the workflow of choice, resulting in a system that intelligently arbitrates between ingest to video servers whenever possible and time-of-air playout as appropriate. In this role, the time-of-air playout system is elevated to a central caption management platform.

The time-of-air caption system can also provide interfaces to other ancillary data signals and XDS information such as wide-screen signaling, vChip parental controls, Broadcast Flag information, DRM controls such as CGMS-A data, Digital Program Insertion (DPI) data, etc.

Transcoding captions for flexible asset management

As well as supporting different output distribution formats, modern captioning solutions support reversioning of video assets. This occurs when an asset is manipulated in the time domain or split into different program segments. The process often takes place within an NLE and can effectively destroy the captioning data as it becomes disassociated with the video and audio content. Modern transcoding solutions can circumvent this issue by using the edit decision list from the NLE (and other sources of data describing the differences between the original and the derived version of the video), bridging the caption data from the original to target version.

By using the mezzanine format for video and caption data described earlier, the caption data component can be transcoded appropriately at the same time as the video, ensuring the same quality or better of captions as for the broadcast version. The same also applies for situations where the lowest common denominator mezzanine route has been followed. It may not provide better presentation, but it still provides the other benefits of easier repurposing and greater efficiency through predictable output.

Seamless integration in next generation workflows

Any platform for caption playout and management must provide the stability needed to ensure confidence in data making it to the viewer. Therefore, it is vital to seek out technology providers who have the expertise and experience in developing and delivering the specialist systems required, as well as those who have proven their worth in the field. Using this expertise, it becomes possible to implement highly fault-tolerant solutions that integrate seamlessly into modern workflows. BE

Ed Humphrey is president of the Americas, Softel.

Analog LPTV end nearing

Some channels will cease LPTV operation this year.

BY HARRY C. MARTIN

In July, the FCC released its long-awaited decision announcing that Sept. 1, 2015, will be the drop-dead date for analog Class A TV, LPTV and TV translator operations (together, "LPTV"). However, analog LPTV operations on Channels 52-69 must cease by Dec. 31, 2011, regardless of whether the licensee has been able to find an available lower channel. But, anyone with a companion channel or flash-cut CP has until Sept. 1, 2015, to convert to digital. Here are some of the implications of the FCC's decision:

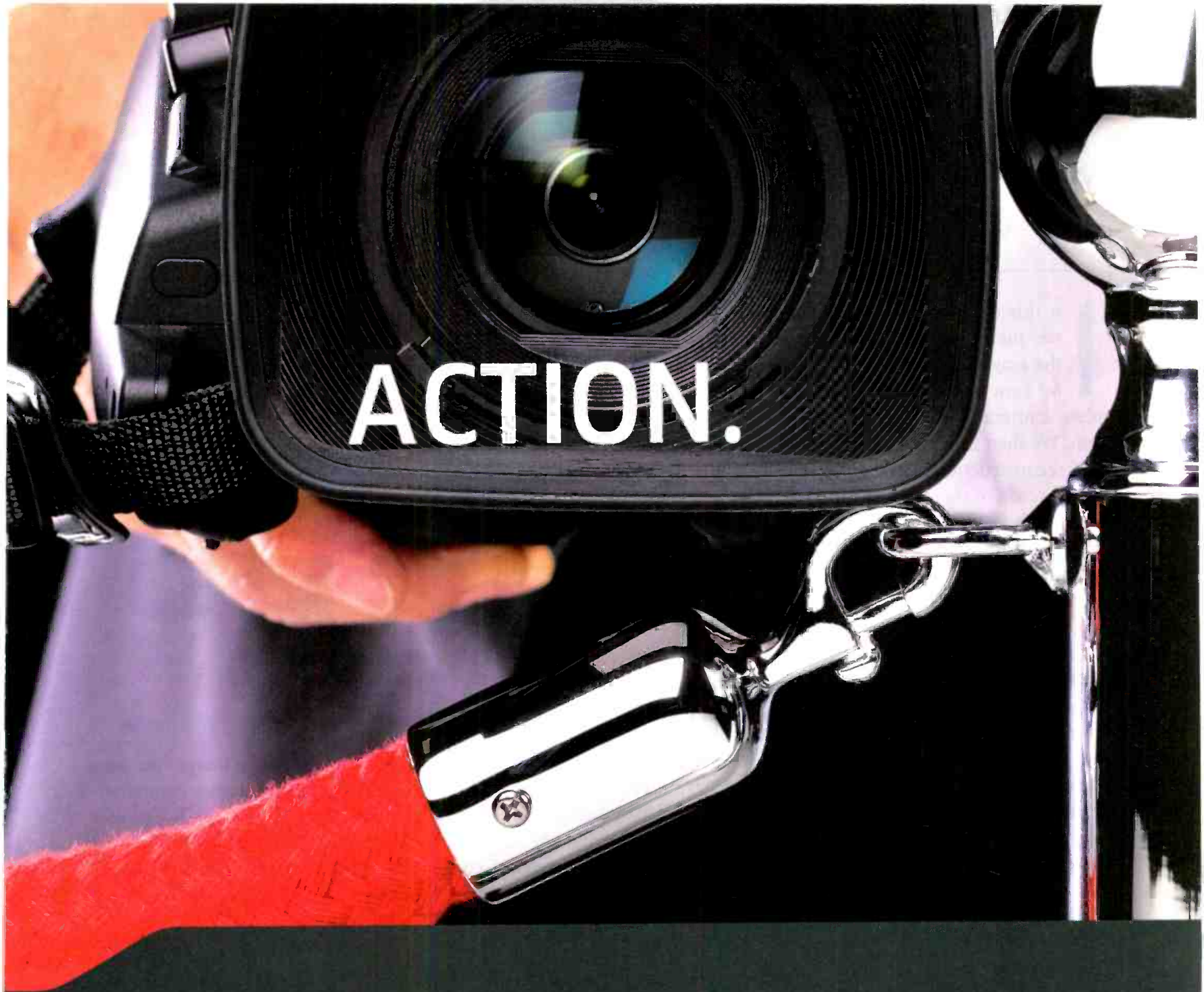
- *Repacking.* The FCC anticipates that a four-year transition period ending on September 1, 2015, should provide adequate time for LPTV stations to accommodate themselves to channel "repacking." The commission has embraced the notion of repacking the TV spectrum to make more space for broadband use by repurposing up to 20 broadcast TV channels and either relocating affected stations to other channels or partial channels, or by having the stations relinquish their frequencies through "incentive auctions" — a plan now being considered in Congress. (See "FCC Update" in *Broadcast Engineering*'s July issue.) The repacking process, if it moves forward as the FCC wants, is certain to affect a sizable number of full-power DTV allotments, which would in turn shrink the spectrum available for digital LPTV.
- *Federal money for the transition.* The FCC has encouraged NTIA to ask Congress to extend the existing program for reimbursing LPTV digital transition costs. Some $30 million remain in unspent funds in that program. The FCC does not discuss either: (a) the program's statutorily-mandated eligibility criteria, which strongly favor the most rural stations and completely exclude urban stations; or (b) the program's dollar limits of $6000 and $20,000, neither of which fully covers conversion costs.
- *Procedural relief.* The FCC has automatically extended all currently outstanding digital CPs for flash cuts (i.e., on-channel conversions to digital) or digital companion channels for existing analog stations. No matter when those permits were issued or how many extensions were previously requested, all of these have now been extended to Sept. 1, 2015. Permittees who can show extraordinary unforeseeable circumstances will be eligible for extensions until March 1, 2016. Construction permits for new stations have not been similarly extended; they are good only for their original three-year terms.
- *Notices to the public.* Migrating licensees must broadcast announcements 30 days before terminating analog operation if they have program origination capability. Stations lacking such capability must find another way to notify the public — e.g., newspaper notices.
- *"Minor" change redefined.* LPTV stations applying for displacement to a new channel are restricted to a 30mi change in transmitter sites. Other changes may exceed that distance and still be classified as "minor" as long as there is not any overlap of licensed and proposed protected service contours. The FCC will now impose the 30mi limit on all minor changes in addition to the contour overlap requirement.
- *Use of VHF channels.* The FCC's spectrum repacking proposal may involve moving LPTV stations to available VHF channels. To enhance the attractiveness of VHF, the FCC has increased the LPTV power limit to 3000W, on all VHF channels.
- *Class A gets option to choose channels.* Class A stations with both analog and digital operations will now be free to elect either of their channels for permanent digital operation. They can apply for a construction permit to flash-cut their analog channel or may migrate their Class A protected status to their digital channel without a CP by filing a Form 302-CA license application.
- *Fees for ancillary digital services.* Both full-power and LPTV licensed stations that provide non-broadcast digital data services in addition to video program streams are required to file Form 317 each December and pay 5 percent of their gross ancillary services revenues to the government. Digital LPTV stations operating under an STA without a license have been exempt, but no longer. If they provide no such ancillary services, they may say so and pay nothing. BE

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

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

Dateline

- Noncommercial TV stations in Iowa and Missouri must file their biennial ownership reports on or before October 1, 2011.
- By October 1, TV and Class A TV stations in the following locations must place their 2011 EEO reports in their public files and post them on their websites: Alaska, Florida, Hawaii, Iowa, Missouri, Oregon, Puerto Rico, Virgin Islands, Washington and the Pacific Islands.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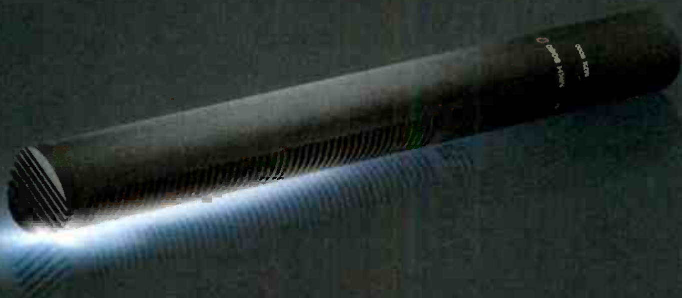


Digital video processing

Gain insight into how the parts work together.

BY ALDO CUGNINI

In this column earlier this year, we mentioned "light-to-light," the concept of an overall transfer function describing a complete camera-to-display video system. We then zeroed in on sampling and reconstruction, two elements of this cascade of signal processing elements. This month, we'll look at this concept from a holistic perspective, to gain insight as to how the various elements interrelate.

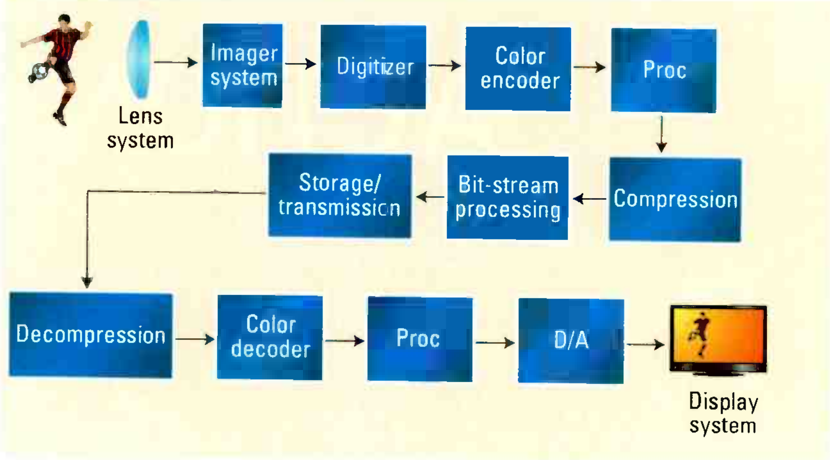


Figure 1. Camera-to-display video processing elements.

Characterization of video systems

In systems analysis, the behavior of a system with an input and output can be characterized by several means. With audio systems, it is customary to describe the frequency response of a system, i.e., the gain of the system from input to output as a function of the frequency of the signal applied to the input. In mathematical terms, this gain is a complex function that can be evaluated in terms of magnitude and phase, and in this form is generally called the transfer function of the system.

With video systems, a similar concept applies, wherein the input and output "signals" are comprised of two-dimensional (or more) images. In fact, as video is concerned with moving images, the signal can be considered as having three dimensions: vertical, horizontal and temporal. Breaking down the image into luminance and chrominance components adds a "dimension" as well. The complete description of a video system, from input image to output (display) image can thus be characterized by one or more transfer functions, and we are predominantly concerned with the amplitude aspect of those.

Transfer functions can be linear or nonlinear, meaning that the input-output amplitude relationship at any one frequency can be plotted as a straight line or as a curve, respectively.

When dealing with linear systems, transfer functions are commutative, i.e., cascading different systems will result in an overall transfer function that does not change when the order of the different systems is changed. But video systems tend to be nonlinear, as the transfer functions depend on many different characteristics of the images and often will vary over time as well.

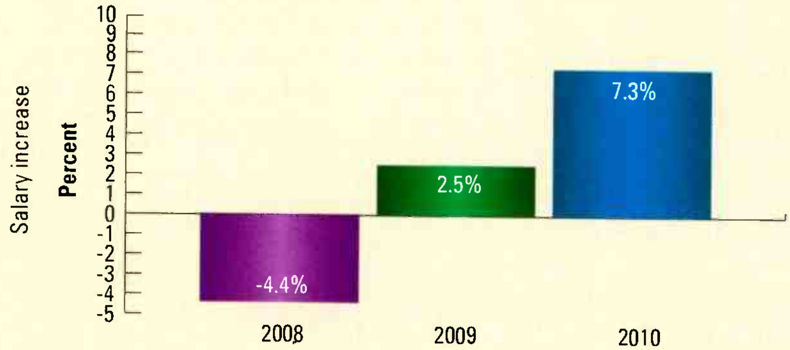
The typical elements of a complete video system are shown in Figure 1. A number of these processes can take place simultaneously, i.e., in real time, or can operate at different points in time, such as upon retrieval from a storage medium. Also, some of the

FRAME GRAB

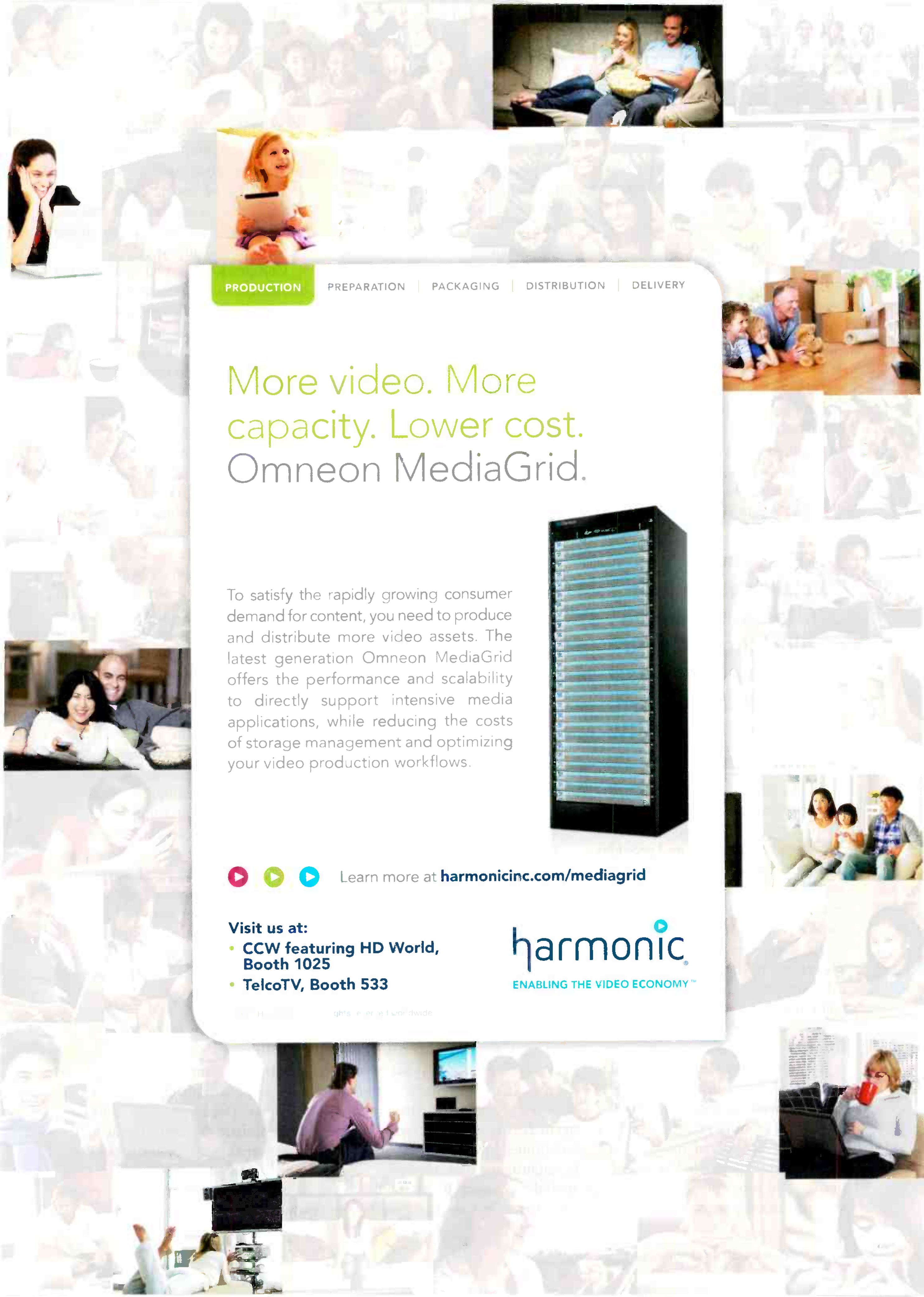
A look at tomorrow's technology

Local TV news salaries increase in 2010

News salaries climbed in TV newsrooms in 2010 by 7.3 percent, following two muddled years that saw a decline in 2008 and a modest increase in 2009.



Source: RTDNA/Hofstra University Annual Survey *www.rtdna.org*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elements shown can be omitted or interchanged in order; one example is a totally analog system, which would not contain a digitizer, or compression, or the inverse functions.

Lens system

We start with the optics of the video camera; the concept of transfer function applies to optics as well as to electronics. In the optical domain, we are generally concerned with the optical transfer function, which includes magnitude and phase. Because phase is usually of less interest when a wide spectrum of light is considered, with video systems we are more often interested in the modulation transfer function (MTF), which concerns amplitude only. The MTF describes the performance of an optical system as a function of spatial frequency, measured in cycles per unit angle, and provides a way to characterize the maximum spatial resolution of the system. Other measures of the behavior of optics include monochromatic and chromatic aberration, which are spatial distortions of the ability of an optical system to form well-focused and color-registered images.

Modern codecs also include loop filters that reduce blocking artifacts but may cause loss of subjective detail or objective fidelity.

Imager system

State-of-the-art cameras use CCDs or CMOS sensors to capture images. These devices inherently perform the sampling function, i.e., they convert a continuously valued image into a discrete spatially sampled representation. While most professional cameras use three imagers and a color separation system of optical prisms, other video cameras use a color separation Bayer filter that is incorporated in a single imager. The performance of an imager is defined by many characteristics, including maximum resolution (in two axes), minimum sensitivity, signal-to-noise ratio and linearity.

Digitizer

Imagers usually do not digitize the signal, i.e., convert the amplitude values into quantized digital values. That function resides in the digitizer, also called an analog-to-digital converter. (There are some imagers, however, that integrate the digitizer on the same device.) The Nyquist Theorem states that information signals must be sampled at a rate at least twice as high as the highest frequency component; otherwise, aliasing will result. The combination of imager and digitizer will determine the resolution, aliasing performance and pixel rate of the video.

Color encoding usually includes chrominance subsampling to reduce signal bandwidth, with contribution and distribution signals typically using 4:2:2 sampling, and transmission signals using 4:2:0 sampling; this subsampling will introduce its own aliasing. Color space modification can also be integrated into this process.

Post processing consists of a number of image transformation functions that provide special effects and image improvement. The simplest of these are luminance and color correction, which adjust for poor levels from cameras and make video consistent across content, and peaking, which applies a high-pass filter (both dimensions in a digital system) that accentuates image detail. Peaking alone can emphasize noise in the image, so noise reduction should accompany its use, by intelligent coring and/or filtering. These techniques lower noise by adaptively quantizing and changing select spatial frequency bands of the image, depending on the distribution of luminance and chrominance values. Format conversion is another processing element that involves a resampling of images in both the spatial and temporal dimensions, with subsequent changes to image resolution and aliasing.

Compression is used to lower the transmission bit rate and/or storage requirement of digital video. While the process can be lossless, i.e., the input/output transfer function approaches a "straight wire connection," high compression factors are commensurate with image distortion, generally causing a number of known artifacts, including blockiness and quantization noise. Modern codecs like MPEG4-AVC/H.264 also include loop filters that reduce blocking artifacts but may cause loss of subjective detail or objective fidelity (accurate pixel reconstruction). The effect of all of the preceding system elements on the performance of the compression system should not be underestimated. Poor processing and interfacing of signals in both analog and digital form can cause an increase in artifacts and a decrease in the efficiency of the compression system.

Compressed bit streams can be modified by numerous methods of bit-stream processing, such as recoding for multiplexing, splicing to switch program content and logo insertion. Storage and transmission, which in the ideal are lossless processes, can nonetheless delete information through errors in the media. While storage is usually designed in a way that makes errors nonexistent, transmission errors can result in irretrievable data corruption and loss.

The decoding side

The processes on the decoder side can be distributed among various devices or completely integrated in a display. Decompression is usually deterministic, within the accuracy of the arithmetic operations. Thus, any two decoders implementing the same fixed-point (or floating-point, when so encoded) mathematical operations should produce the same output. However, there may be some savings in processor power dissipation, memory size, etc., by taking some approximation shortcuts in the decoding algorithm, which can have an effect on output picture quality.

With most consumer displays based on either an LCD, plasma or small projection panel, the display can become part of the final digital-to-analog conversion process.

Color decoding is typically the straightforward inverse of the encoding process. Processing on the decoder side provides image improvement and allows user preferences, including sharpening (peaking) and noise reduction. Spatial down-resolution can occur either within this processing or as part of the decompression process. The D/A (digital-to-analog converter), while ideally providing an inverse to the digitizer function, always requires a low-pass filter (LPF) that can potentially degrade high-frequency response, i.e., the ability to reproduce image detail.

The display system attempts to construct a faithful reproduction of the original image. With most consumer displays based on either a large LCD or plasma panel, or a small projection panel (the latter usually involving optics), the display can become part of the final digital-to-analog conversion process. While signal reconstruction in the system sense usually requires a low-pass filter, and one will be used when driving analog video outputs, some digital display panels eliminate the D/A converter and LPF entirely. An optical spatial filter is usually absent in panel displays due to its complexity and the ability of the human eye to integrate the pixels into a continuous image. BE

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Navigating the cloud

Cloud computing brings opportunity, and challenge.

BY BRAD GILMER

If you are wondering how the "cloud" relates to professional media applications, you are not alone because one of the hottest topics in the IT world these days is cloud computing. So, this month we will look at several concepts behind cloud computing and explore both opportunities and risks for our industry.

What is a cloud?

Clouds are usually used as a symbol for a network — frequently, the Internet. Clouds were used prior to this by telephone companies to signify their nationwide switching network. The Internet cloud that keeps showing up on just about every facility diagram is just a lot of switches, routers and servers interconnected with fiber optic lines and Unshielded Twisted Pair (UTP) cables.

Early in the history of the Internet, the cloud idea was pretty straightforward. Packets transited the network from source to destination based upon the destination IP address in the header. As problems with the Internet were addressed and solved, however, equipment in the cloud became more complex. For example, content started being saved in router caches close to the consumer, rather than being immediately deleted, so that frequently accessed material could be served from the router rather than from the original source. Then, the concept of managed networks emerged, guaranteeing performance through the cloud. Since that time, the cloud gained and continues to gain capabilities as it morphs from being an interconnection of private computers to a collection of servers that are able to provide a wide variety of services to the public.

Whatever as a service

When you get into a conversation about the cloud, the services topic is not far behind. A quick Internet search for "as a service" returned the following: Storage as a Service (SaaS), Software as a Service (SaaS), Platform as a Service (PaaS), Computing as a Service (CaaS), Infrastructure as a Service (IaaS), Security as a Service (SaaS?), Data as a Service (DaaS), Communications as a Service (CaaS), and my favorite, Monetization as a Service (MaaS). The list seems endless. That said, some of these services may have applicability to the professional media industry, particularly Storage as a Service and Software as a Service.

The cloud has become a system of connected servers instead of computers. This growth expanded potential capabilities, but it has also compounded organizational complexity.

Storage as a Service

Storage as a Service (SaaS) has been around for some time. This service originated when some very large companies decided to try selling excess storage capacity on their Internet servers to the public. Early efforts at offering remote storage services faced several challenges. First, bandwidth bottlenecks, at both the server and consumer ends, presented a major problem. Over time, however, most users have been able to gain access to very high bandwidth Internet connections. Ironically, some successful remote storage services suffered from their own success: As they became popular, the bandwidth needed at their locations increased tremendously as the number of users with high-bandwidth connections increased. In the end, only very large companies with servers distributed all over the world remained viable.

A second challenge was providing a method of connectivity to the remote storage. Early implementations relied on File Transfer Protocol (FTP), Network File System (NFS), Common Internet File System (CIFS), SAMBA or some other remote file system implementation. Media professionals are well acquainted with the issues surrounding large file transfers and FTP. FTP's file size limit varies with operating systems, but even now it is not unusual to find limits of 4GB. Also, unless you are particularly careful, FTP's throttling mechanism will cause exceedingly slow transfers or complete failures of transfers if the network delay is too long.

Users experienced interoperability problems with NFS, CIFS, SAMBA and other remote file system implementations as well. Perhaps some of these

problems were due to configuration errors, but still, implementations of these shared file system architectures were less than plug-and-play. Firewalls also caused connectivity problems with some remote storage implementations. IT departments were unwilling to open up the communications necessary to make remote file systems work across corporate boundaries.

Later implementations relied on HyperText Transfer Protocol (HTTP) and Secure HyperText Transfer Protocol (HTTPS) for file transfer. In particular, HTTPS alleviated many of the configuration/interoperability issues and also provided relatively secure transit across corporate firewalls. But, HTTPS still did not allow the sort of flexibility and overall architectural design that many remote storage users were seeking. A new, simple, computer-friendly method was needed to access remote storage on the Internet.

Storing video in the cloud

Two powerful ideas combined to enable SaaS. The first is the concept of cloud services. In order to understand this better, it may help to look at a specific example. Amazon has been a major provider of cloud services. Amazon's Simple Storage Service (S3) offers large amounts of storage at very competitive prices. The main difference between remote storage and S3 is the way Amazon presents the storage to the user.

From a business perspective, Amazon is offering storage in the cloud; you are not connecting to a specific server. Instead, you send content to Amazon, and then they put it somewhere. Later, you can search for it or retrieve it. You interact with S3 using Representational State Transfer (REST). REST allows users to interact with services using many common Web programming languages, including Perl, Java and PHP. How you interact with the service is up to you; the results are the same. But the important point is that Amazon provides a storage service to you as a business. How they do it, where they store the content and how they maintain it is invisible to you.

The second concept that has been critical in introducing services in the cloud is the clever renaming of the Internet as "the cloud." To many people, the concept of a service existing in the cloud is a lot easier to understand than the concept of remote storage offered as a business over the Internet. Which would you rather buy? But, whenever you hear SaaS, realize that vendors are talking about a service being offered on distributed hardware connected to the Internet.

Security

I have attended several professional media conferences where SaaS has been introduced. Without question, the number one issue raised right off the bat is security. Vendors can say all they want about the security methods they use, but we users will always be concerned about whether content stored in the cloud is safe. In fact, I question whether we will ever be totally comfortable with a business where the essential premise is that you do not have to worry about how and where the content is stored because the vendor has you covered.

That said, major broadcasters use Virtual Private Networks (VPNs) for on-air video delivery every day. Perhaps over time, financial considerations will overcome our concerns for security, or better yet, perhaps vendors will offer professional media services such as SaaS on secure, managed networks.

Software as a Service

Another area that may ultimately be interesting for professional media companies is Software as a Service (also SaaS). Two professional media applications that immediately come to mind are video editing and video transformation. Imagine a video editor application available on the Web. Now, also imagine that you are willing to store content in the cloud. One can then quickly see a professional video application where you can work on your project from virtually any computer. You can give others access to your project, allow them to see rough-cuts in real time and interactively participate in an editing session — even though editors and clients may be hundreds or thousands of miles away. Editors can access the same content for different projects (a news story and a segment of a long-format news show, for example). Editing applications can be written to allow differing editing functionality based upon the capabilities of the end-user device (workstation vs. tablet) and bandwidth available.

Video transformation as a service involves giving content to a service and asking the service to transform the video in some way. This has been done for some time in the Hollywood community, although the concept of offering this as a service is relatively new.

Software as a Service challenges several long-standing practices. For example, the software is not owned or maintained by the end user; the functionality of the editor depends on the end-user device, and content is stored in the cloud. However, we may be willing to accept these changes given the flexibility and capabilities presented by SaaS.

Conclusion

New cloud services are coming, and security and viability concerns of these services for professional media exist. But, realizing them as Internet-based server options helps frame informed talks with vendors. BE

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Nonlinear editors

Here's a look at some of the technical distinctions of today's NLE systems.

BY L.T. MARTIN

These are heady days for high-end nonlinear editing system providers. But, we may have recently lost one of the biggest players in the game.

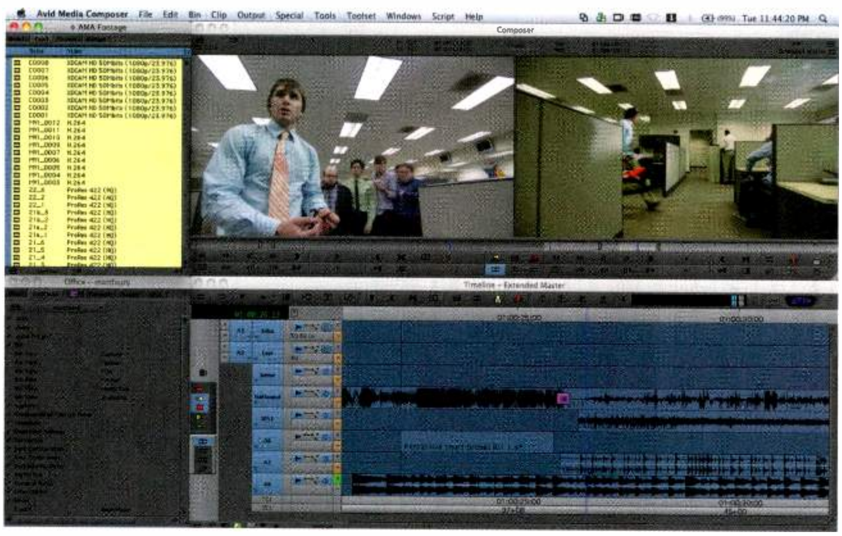
Despite the fact that Apple's Final Cut Pro 7 software is being used by 54.6 percent of the professional editing community, according to a report by market research firm SCRI International, in June Apple released a completely rewritten version called Final Cut Pro X. Much to the surprise of many, despite that this new incarnation is completely 64-bit, so many professional features were removed that it appears Apple may be switching its focus away from the high-end video editing market segment. Final Cut Pro X cannot import projects started in Final Cut Pro 7 or any other NLE. It cannot ingest from or output to tape and can't even be connected to a broadcast monitor. These noticeable (and important) missing features have more than a few editors scratching their heads.

So this makes it an apt time to take a look at some of the technical distinctions under the hood of the still broad spectrum of NLE systems from the major manufacturers who still want to offer solutions in this space.

Avid Media Composer

Avid's Media Composer is still the most used NLE on prime-time TV productions, being employed on up to 90 percent of evening broadcast shows. One reason is its design philosophy, called Avid Intelligent Architecture, that Avid instigated back in 2008 with the release of Media Composer 3 and the DX hardware line. Avid Intelligent Architecture is a way of constructing software that constantly looks at the Mac or Windows computer being used, the graphics card installed and whatever hardware is attached, and selectively sending a given task to the component best suited to accomplish it.

For example, if an editor needs to do something that is CPU-intensive such as decoding a codec, those files get sent directly to the CPU. But effects work will be directed to the GPU, and DNxHD encoding (short for Avid's proprietary Digital Nonlinear Extensible High Definition compressed file format) or thin raster format expansion will go to Avid's DX hardware. That enables the Media Composer software to constantly evaluate the resource that can most efficiently get the job done.



Avid's Media Composer is now compatible with Pro Tools hardware for greater workflow flexibility. This includes audio capture and monitoring of Media Composer projects through Pro Tools hardware and voice-over recording into the NLE to enable faster collaboration between sound and picture departments.

Avid has also been successful implementing phonetic indexing into its systems starting with the 2007 introduction of its ScriptSync feature. ScriptSync analyzes the audio tracks on clips in the Media Composer source bin and links them to the written script that has been imported into the system. Clicking on a line of script gives access to all the takes containing that specific dialogue.

This year, Avid introduced PhraseFind for unscripted productions such as documentaries or news packages. PhraseFind is a search engine plug-in that can find specific words or phrases in a production without reference to a script.

Autodesk Smoke

One of the most cost-effective and powerful nonlinear finishing systems is Autodesk's Smoke. Originally available only as a turnkey system on Linux, Smoke became available for the Macintosh in a software-only, 64-bit version in December 2009. The latest incarnation, Smoke for Mac 2012, is an online NLE with extensive graphics capabilities that can import EDLs in all major formats. Smoke gains its image quality by automatically converting every compressed incoming file to uncompressed RGB and performing all its graphics/effects work in 4:4:4:4 color space.

Smoke is also the only NLE that can use the platform-independent Autodesk FBX file format to load 3-D models and motion graphics in their original, unrendered native format. FBX, which originally stood for film-box, is similar to a PDF format for 3-D graphic objects.

For stereoscopic, dual channel 3-D work, Smoke for Mac 2012 can also scan each video track to detect differences between the left- and right-eye clips.

Grass Valley EDIUS

Grass Valley has been adding some amazing capabilities to the latest 6.02 version of its EDIUS software, which expands its usefulness in broadcast applications. The company acquired the system from Canopus in 2005 and is marketing EDIUS as a software-codec edit system for Windows, highlighting the efficiency of its constantly upgrading codecs. In fact, in November 2007, EDIUS was the first to crack the challenge of decoding the computationally-demanding but low bit rate AVC Intra HD format.

Thanks to its robust codec technology, EDIUS can now edit 4K files (4096 x 2160) in 8- and 10-bit. To edit these files, the system automatically generates proxy files during capture that are easier for the CPU to handle. This lets EDIUS perform all its tasks in real time without ever needing to render, even on platforms of lesser capabilities.

The system's Source Browser puts the functionality needed for importing all files into a single viewer, supporting from P2 to Blu-ray applications. EDIUS can handle a wide range of output options, and can even export back to an AVCHD card or an iPad/iPod.

Grass Valley has added a new live capture option to EDIUS. The option allows users to ingest live video on one EDIUS client while they simultaneously begin to edit it on a different EDIUS client.

Boris FX Media 100

After being spun out of Data Translation in 1996, the Media 100 NLE was eventually acquired by Boris FX in October 2005. Always based on the Macintosh platform, Media 100's core engine, called HAL for Hardware

Abstraction Layer, has never used Apple's QuickTime proprietary multimedia framework to play out its files.

Of course, the Media 100 system does use QuickTime for handling graphics formats and system-level codecs. But it does not use QuickTime's playback services because the Media 100 system was always intended for broadcast use and has become known for its ability to quickly play out its timeline without waiting for QuickTime intervention.

Although Media 100 can play out a sequence of shots directly from its timeline, it uses Adobe After Effects as its compositing system, and can access Apple's Color software for color correction. In fact, many editors use Media 100 and Apple's Final Cut Pro 7 software side-by-side. Media 100 can be faster when assembling a simple sequence thanks to its ability to use an A/B roll style of timeline, which gives the editor simple access to the pre- and post-roll handles of each shot. Even so, many editors may find that Final Cut Pro 7 provides greater effects and image manipulation capabilities.

Sony Creative Software Vegas Pro

The Vegas Pro NLE from Sony Creative Software boasts several firsts. It was the first editor to go 64-bit on Windows, and was the first NLE to incorporate a digital audio workstation within the NLE itself. With support for industry-standard VST plug-ins, Vegas Pro provides the ability to edit audio down to the sample level and perform 5.1 surround sound mixing.

The latest version, Vegas Pro 10, enables integrated stereoscopic 3-D editing and the ability to deliver 3-D projects either as two track files or single files with side-by-side, top/bottom or line-alternate encoding. It also provides ingest, preview and output of closed captioning data, supporting industry-standard closed-captioned file types for either broadcast or Web delivery.

Adobe Systems Premiere Pro

Many editors may suggest that Adobe Systems' Premiere Pro NLE benefits most from the presumed retreat of Final Cut Pro from the broadcast editing arena. As part of its Creative Suite 5.5 collection of software applications, Premiere Pro benefits from the development of the Mercury Playback Engine, which runs natively in 64-bit on both Windows and Mac platforms.

Adobe has optimized Premiere Pro for maximal use of a platform's GPU to enable it to handle file sizes up to 5K and beyond. It works with both the broadly accepted OpenCL processing language and also NVIDIA's proprietary Compute Unified Device Architecture (CUDA) parallel computing architecture. This design lets Premiere Pro offload many processing requirements from the CPU onto a qualified GPU.

The result is that Premiere Pro has sufficient power to handle multiple layers of Red Digital Cinema's 4K files complete with color correction. Adobe has pursued an aggressive release cycle of its Premiere Pro software upgrades. But unlike some other vendors, the company maintains a high priority to ensure each new version is completely backwards compatible with its predecessors. BE

L.T. Martin is a freelance writer and post-production consultant.

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Multiviewers

Increase your understanding of this important piece of broadcasting equipment.

BY RENAUD LAVOIE

Multiviewers have become ubiquitous in most broadcast control rooms. The flexibility of displaying multiple images in various resolutions in an active and situational-aware display has many benefits. Building on some of the key points learned in the last article, "Understanding blocking capacitor effects," (August 2011), let's continue our review of broadcast and production equipment design considerations, this time with respect to multiviewers.

Some multiviewers also have the ability to analyze each input signal to be sure the signals are correct and without errors.

As you may remember, my first task in the video broadcast industry was to build an optical-to-electrical converter and to evaluate small form-factor pluggable (SFP) integrated circuits from different vendors. That assignment was promptly interrupted due to a change in priority, and I moved to the multiviewer group to increase the manpower. At that time, I had little first-hand information about multiviewers, and Wikipedia didn't help much either. So I had to learn the hard way. Fortunately, as I look back on the experience, I can say that we had a talented team, and the product we developed turned out to be a great success.

A full discussion of the multiviewer by itself could take more space than this entire issue, so I will attempt to explain it from a 10,000ft view. If portions of this technology generate questions or you want to discuss these issues, please let me know, and in my blog or through *Broadcast Engineering*'s Disqus, I can explain things in more detail.

The global functionality of the multiviewer can be summarized like this: Take multiple video and audio signals and combine them to create attractive and flexible layouts. In addition, some multiviewers also have the ability to analyze each input signal to be sure the signals are correct and without errors.

Basically, multiviewers are composed of several types of common cards: input video card, input audio card, GPIO card, output card, interconnect card and, in some cases, router cards. (See Figure 1.)

The input video card

The input video card is responsible for the reception of the multiple video feeds. The number of inputs often varies from four to 16 inputs, depending of the size of the multiviewer. The first task of the input video card is to receive the signals and perform any needed equalization or light-to-electrical conversion. One of the new trends is to use SFP cards to handle this task. Because of the variety of signal types — coaxial SDI, optical SDI, composite, DVI, video component — and the pressure to release products faster and

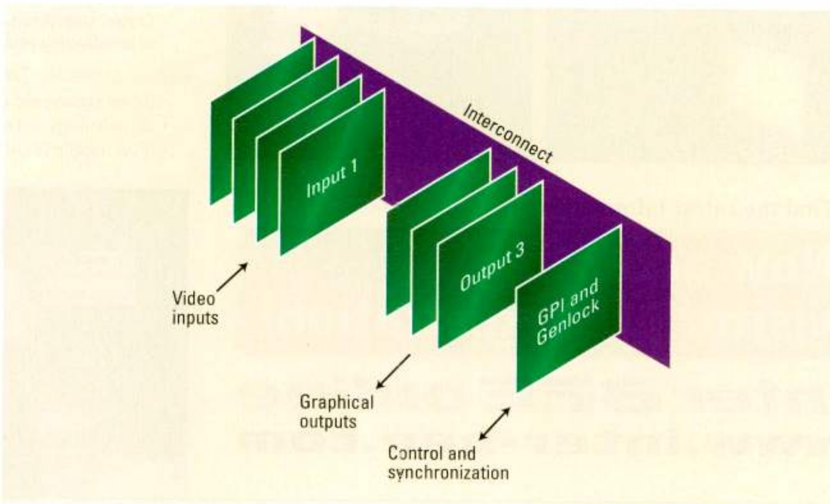

Figure 1. A multiviewer is typically built from only a few types of common cards, using as many as necessary to complete the design.

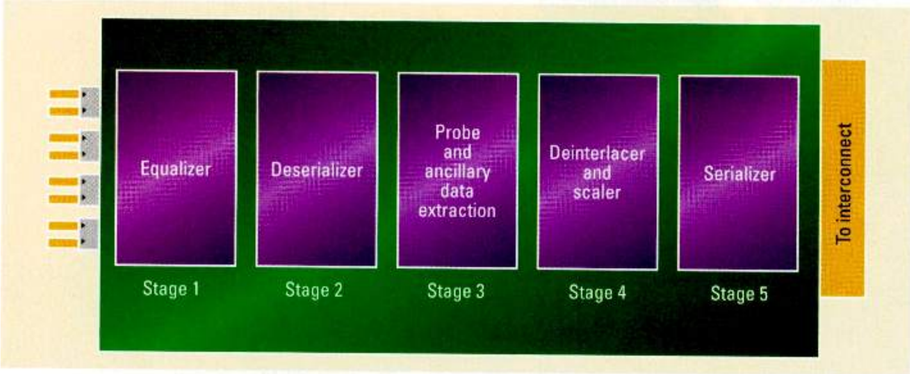

Figure 2. Shown here is an example video input card with multiple processing stages.

faster, it makes sense for manufacturers to use a generic connector (SFP cage), which enables them to support all video formats without the need to create a variety of different cards.

In the first stage, the shape of the signal is restored and converted to SDI. After Stage 1, the serial data is converted to parallel data for processing, but why? The reason lies in the speed of the SDI signal. To properly convert the data (with cost-justifiable technology), it is advantageous to first convert the serial data to parallel data. This is what the deserializer does — take the serial data and parallelize it to ease the further processing of the video.

The video can now be upscaled or downscaled more easily. We will soon have the technology to process the data without deserializing it, but for now, the data must be processed in parallel form. The next processing steps are now feasible: deinterlacing, upscaling, downscaling, color conversion, audio analysis, audio extraction, etc. All of these functions are possible on the parallel video data.

Depending on the multiviewer's architecture, the functions implemented by the input video card vary from deserialization only to full scaling. To better understand the input card functionality, let's reconsider the example multiviewer input card. (See Figure 2.)

In this case, each of the eight video feeds on each input card represents a 3G-SDI, 1080p signal. Each signal is first equalized/converted at Stage 1 and then deserialized at Stage 2. This represents 8 x 2.97Gb/s, a total of 23.76Gb/s of aggregate data.

$$\text{Total bandwith} = \sum_{input = 1}^{Nb\_input} Bandwith_{input}$$

The third stage is to probe and/or extract auxiliary data, which may include audio, teletext, closed captioning, etc. This is an optional step. The video data itself is processed at the fourth stage. In the fourth stage, the signal may be deinterlaced, which is not required for 1080p, and then scaled. This stage is crucial, and each multiviewer manufacturer has its own secret sauce to maintain the quality of the content.

Let's assume that we did downscale all the input video in stage four by a factor of eight to all inputs with the same size on one display. The aggregate data is now 8 x 2.97Gb/s / 8 = 2.97Gb/s between this input card and the output card.

$$\text{Scale bandwith} = \sum_{input = 1}^{Nb\_input} (Bandwith_{input} \times scale\_factor)$$

To send this data to the output card for the final scaling, color correction and picture-in-picture, along with closed captioning, VU meters, VITC, etc., the data channel needs to support from almost 6Gb/s (when SD-SDI signals are downscaled) up to 23.76Gb/s for nonscaled eight inputs. To accomplish this in a cost-effective manner, the input card usually reserializes the data (Stage 5) before sending it on the interconnect card. An important factor to consider when designing a multiviewer input card is to perform this step with uniform latency. Any delay induced by processing must be constant across all inputs. This becomes even more important when processing 3-D imagery that originates as two separate sources.

Interconnect card

In many cases, the interconnect card is a passive card. It can be seen as a large array of cables that connect every input card and every GPI card to every output card. This card also contains the communication links to control every other components in the multiviewer: input cards, output cards, GPIO cards and audio cards. One of

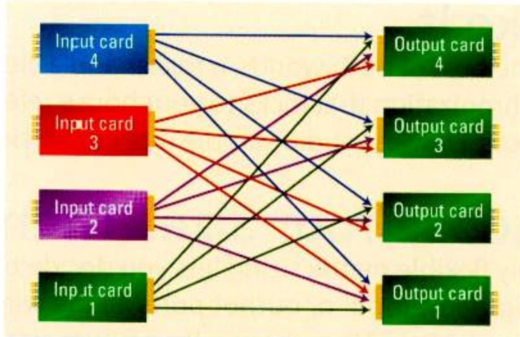

Figure 3. The point-to-point video path architecture has become the more common design for video links.

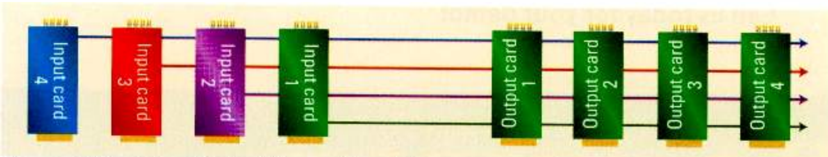

Figure 4. The daisy-chain video path architecture can be used for the communication channel.

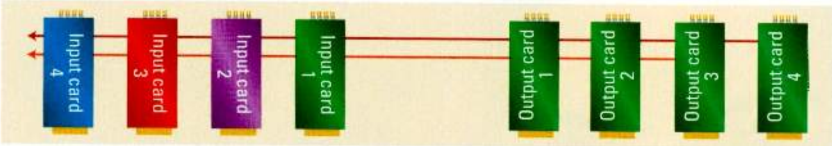

Figure 5. Shown here is an example communication link with the master path in red and the redundant path in orange.

the most important links on the interconnect card is the synchronization link. This includes Hsync, Vsync, Fsync, clock and proprietary links to synchronize all the cards together.

allow multiple users to share the same control path and statuses.

Finally, the synchronization link is composed of Hsync, Vsync and other synchronization signals. This link can

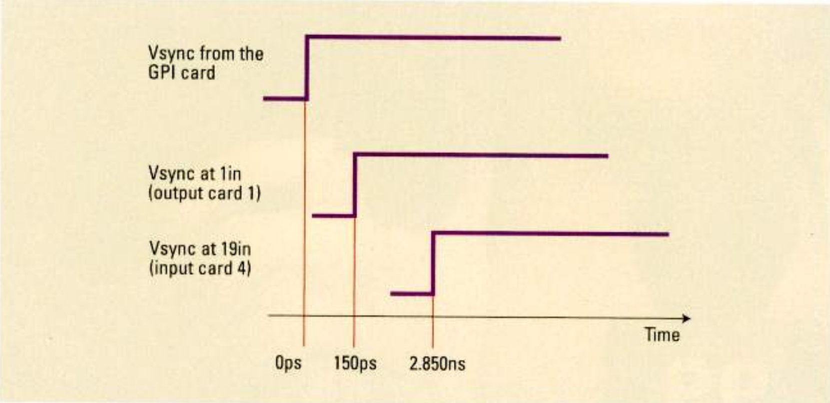

Figure 6. This graph shows how card length affects delay on the Vsync.

Different architectures have been implemented to connect inputs to outputs. These implementations are often thought of as daisy chain, bus or point-to-point. However, with the increase in signal speed, the point-to-point architecture has become the more common design for video links. (See Figure 3 on page 34.)

For the communication channel, the daisy-chain approach can be used. (See Figure 4 on page 34.) But like above, with the increased data speed, it has become more difficult to do. Usually, the internal communication link is composed of two or more communication paths to ensure system redundancy. In our example, (see Figure 5), output card 4 is the origin of the master path (colored red), and output card 3 is the origin of the redundant path (colored orange). The communication link also contains external communication to allow the user to take the full control of the multiviewer. Nowadays, the external communication link is typically Ethernet (with SNMP) to

be a daisy chain, like that shown with the blue line in Figure 4. However, the delay between the first card in the system to the last one should be almost the same. The delay on a printed circuit board is typically 150ps to 180ps per inch, which is not a significant factor, but buffers in the path are critical. Let's assume a standard rack unit chassis where the printed circuit board will be 19in long. (See Figure 6.)

If this delay is not acceptable (architecture dependant), a point-to-point architecture can be used instead.

For the current crop of multiviewers, each vendor does it differently, and the interconnect card can be created in a thousand different ways. Let's use Figure 3 as the reference for our multiviewer discussion.

Each input card sends one video link to each output card, and there are four output cards. This means the interconnect card is composed of 4 x 4 high-speed links that carry video signals. But this is theoretical, and this depends of the bandwidth between input cards and output cards. In practice, two or more links are used for video links. Let's now take a look at the output card.

Different architectures have been implemented to connect inputs to outputs. The point-to-point architecture has become the more common design for video links.

Output video card

In our example, each output card can receive video from four input cards. In this example, that means eight video inputs per input card. Under a worst-case scenario, each output card will receive the entire video stream from all input cards (a 32-input stream). Let's assume the output card is capable of driving one monitor with a 4K resolution maximum. In this example, the output card will drive one monitor capable of 4K resolution each (3840 x 2160 pixels).

Because we can have 4 x 1080p images displayed on this monitor, the worst-case aggregate bandwidth will then be 4 x 2.97Gb/s = 11.88Gb/s. We will ignore any saved bandwidth from removing the blanking period. (Only the active picture is usually transmitted between the input and output cards to save bandwidth.)

The question the system designer needs to answer is: Can the 4 x 1080p imagery come from one input card? If so, the bandwidth of each input card is now 11.88Gb/s maximum. With today's new programmable logic FPGA cards, 11.88Gb/s can be achieved with one high-speed link, commonly called serializer/deserializer or SerDes.

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Each input card can send one or two links to the output card for processing, for a total of each output card receiving up to eight links from the input's cards. The processing power of the output card therefore must be 4 x 11.88Gb/s, or 47.52Gb/s of video data. This is a large amount of data to process in real time. Often the video data is encoded to ease the deserialization process using a well-known encoding scheme like the 8b/10b. Using 8b/10b, the final data bandwidth becomes 47.52Gb/s x 1.25, or 59.4Gb/s. This requires that the output card be equipped with a powerful enough FPGA to be able to process this amount of video data.

The first stage performs the deserializing process. (See Figure 7.) Once this is done, further data processing can be done in Stage 2, such as final scaling and color conversion (from YCbCr to RGB). Following the second stage is the positioning and the graphic muxing. This is where the images are placed at the correct location in the final layout. Graphics are also muxed to allow the user to add transparency, pictures-in-pictures and fancy layouts like rotating images. Multiviewers also must be

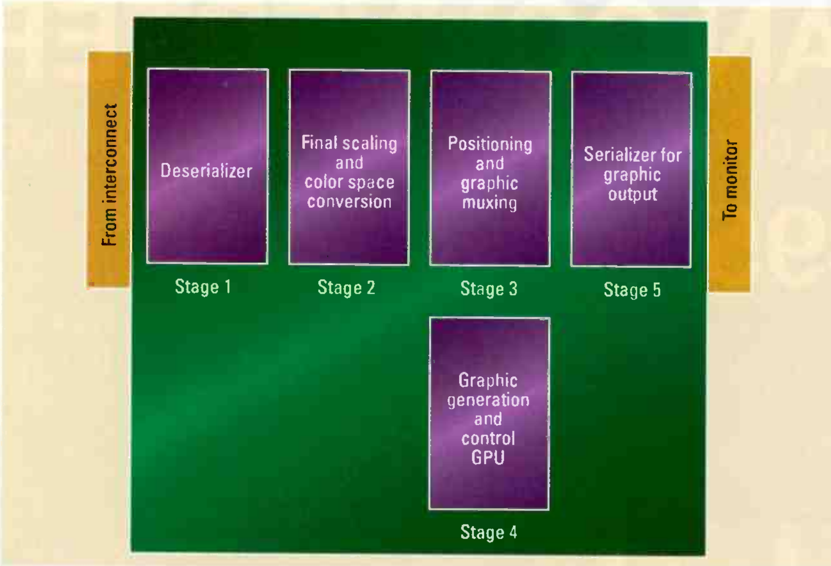

Figure 7. This block diagram shows an example of an output video card.

capable of displaying closed captioning, VU meters, tallies, and other signals and alerts.

The final stage is another serialization, this time to format the data to be received by the monitor or to send to another destination (SDI, streaming, etc.). The graphic objects are generally created by an internal CPU that controls the layout of the objects and the system behavior, SNMP, etc.

The software

The software is a key component of any multiviewer. This operational feature applies not only to the graphical objects and final layout, but also external devices the multiviewer may need to control. Such software should remain easy to understand and to use. With remote access, this software should run on different platforms and sometimes from different locations throughout the world, thanks to the Internet and the thumbnails!

Complementary cards

The multiviewer's complementary cards include any other supporting features needed to make this device work in the real world. While these cards are important, they are also often less complex. Let's review some of the more common cards used on multiviewers.

The GPIO card is used to add even more functionality to a multiviewer. These cards control the tallies and visually report alarms. The GPIO card is typically equipped with either 64 or 128 general-purpose inputs and outputs to control other devices in the studio (such as microphone mute, communication request, ring alert, start a play-out system, red-light, studio warning lights, etc). The inputs are tolerant to 24V, and the outputs are isolated by opto-couplers to permit easy interfacing.

The genlock card is used to synchronize the output of the multiviewer to the reference signal of the studio. The genlock card can also create all the clocks used in the multiviewer to ensure that all boards receive the same clock (remove the drift between clocks) and maintain clock phase.

The need for audio monitoring is obvious. A multiviewer provides a convenient platform to see visually what is happening on both video and audio sources. Also, multichannel operation precludes operators from being able to hear the audio from perhaps hundreds of channels. Visual displays, along with automated silence sensors, make the process far less cumbersome. Instead of requiring a separate device to analyze the audio sources, the multiviewer can do it and alert the operator only when necessary.

Multiviewers and IP

As IP technology intrudes into the traditional serial digital video master control room, multiviewers will need to adapt. Some multiviewers already support the encoding format; others are able to receive thumbnails for monitoring.

Another growing trend is to embed the multiviewer in routers and other equipment. Some vendors already embed the monitoring router function inside the multiviewer. For some applications, it may make sense to minimize the interconnect between the router by locating the devices close together, even by making the multiviewer a component inside the router. When this approach is taken, manufacturers will need to make this a robust device because the router is the most crucial device in any broadcast or production environment. BE

Renaud Lavoie is president and CEO of Embrionix Design.

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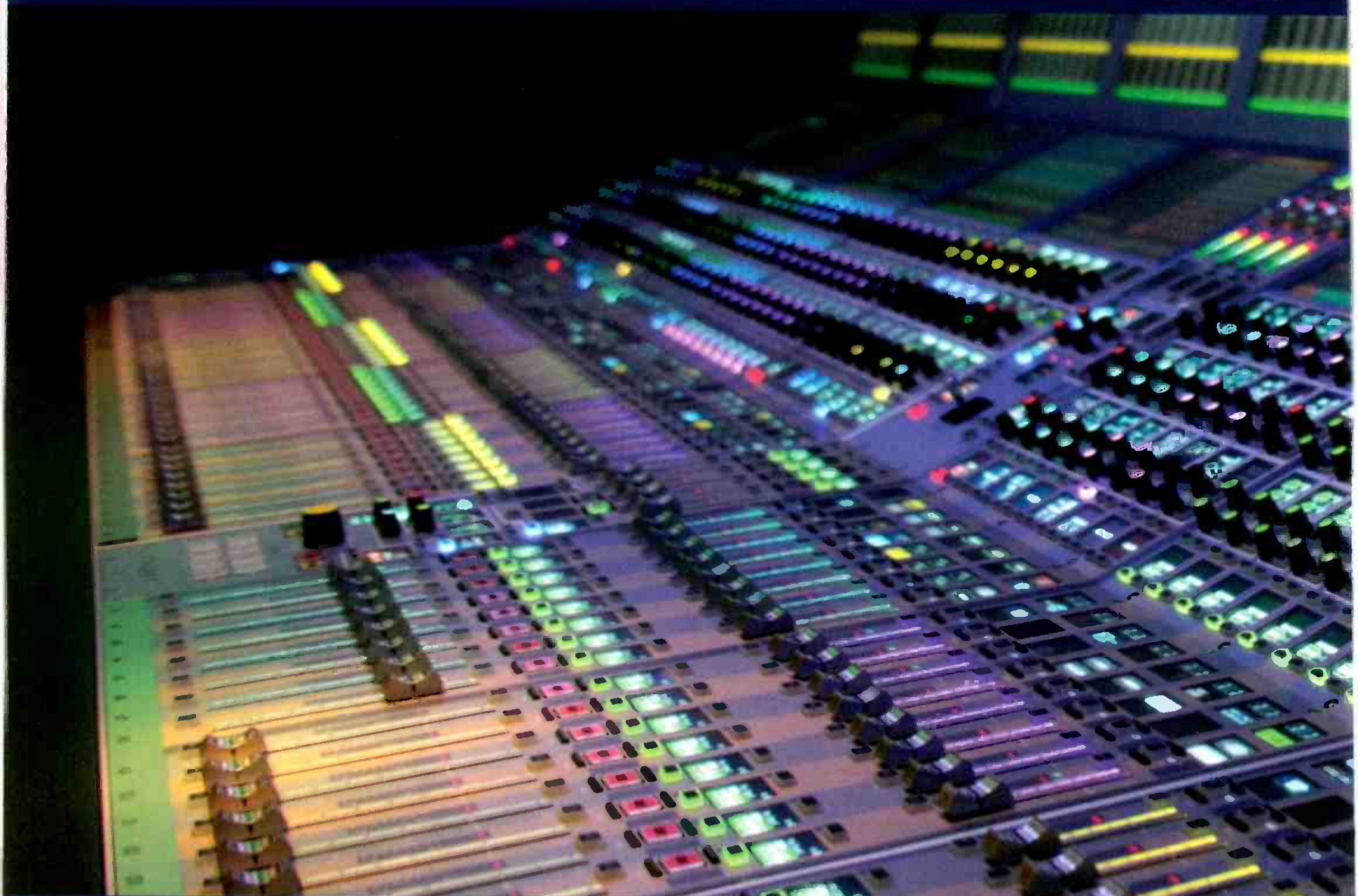
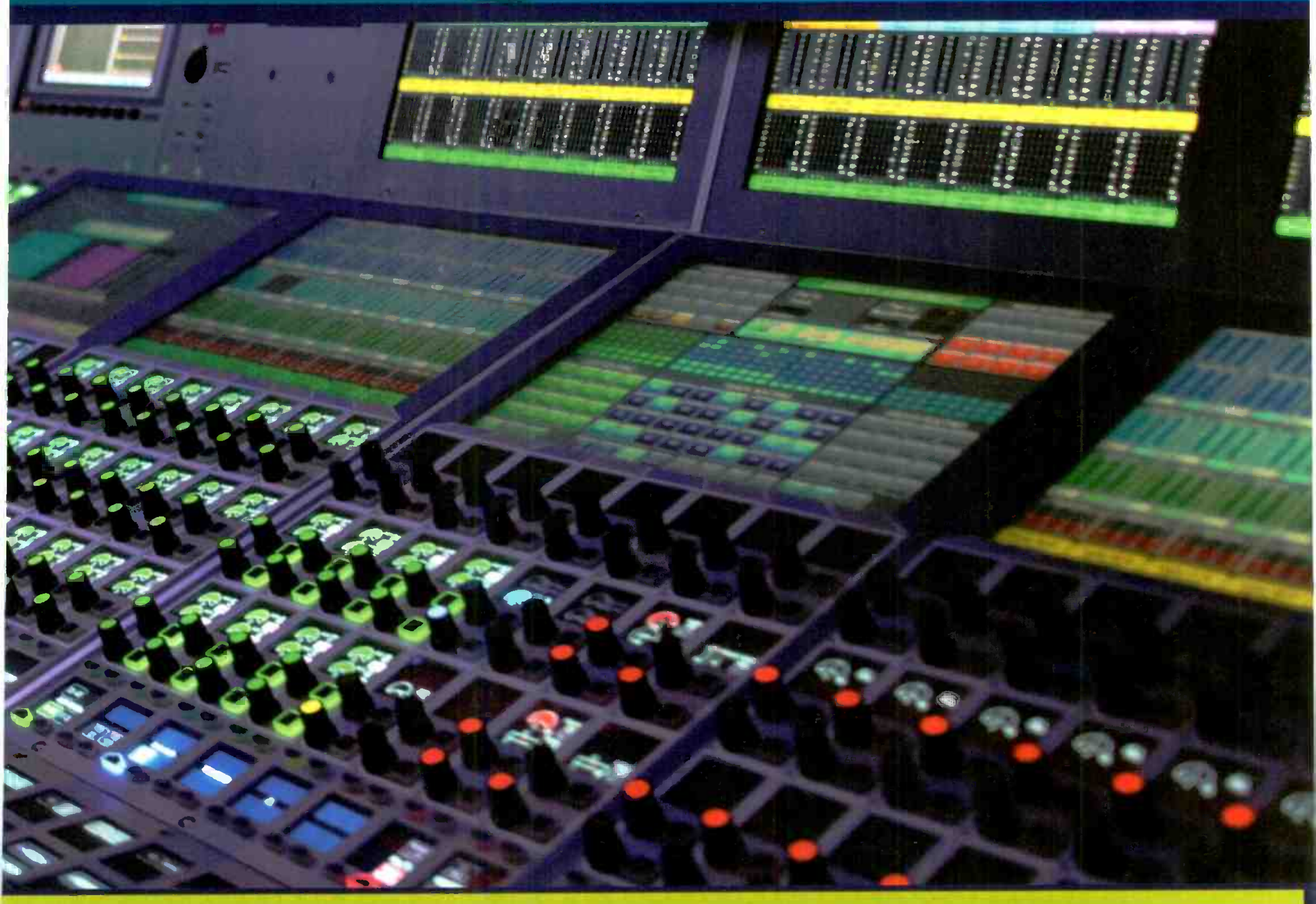


DIGITAL AUDIO SYSTEMS

Report



PERFECT FIT

FORM

Wheatstone's television audio consoles are designed to fit where you really need them – in VERY tight spaces. All our surfaces open from the front for complete access to the inner workings. All of the audio, I/O and logic functions are located outside the control surface in rack-mount cages that can be located anywhere in your network. This is a huge plus when planning a new facility or retrofitting an existing one. It also means these are a natural for production or on-air use in remote broadcast trucks.

FEEL

You need to sit down in front of a console and go to work. Immediately. No time for weeks of learning or getting used to it. Our consoles let you do exactly this, placing everything where you'd expect. Since the inputs are all 100% assignable to any source (and you have access to a LOT of sources), your efficiency in getting audio where you need it is greatly increased, letting you concentrate fully on mixing.

FUNCTION

Network First™ modular, networkable technology. It's the way Wheatstone consoles are designed and built. This means you can have the advantage of placing control surfaces, mix engines or interface boxes anywhere you want them, how ever you want them, without worrying about future access or network interaction/audio routing. No matter how many control surfaces or I/O points you need, just plug them in wherever you want and you are up and running.

D-8

SURROUND SOUND
TELEVISION AUDIO CONSOLE

WHEATSTONE IS TELEVISION AUDIO AT ITS FINEST

FEATURES

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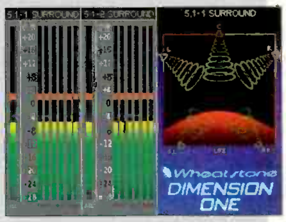

it's all engineered to fit your workflow. Because of our modular design, future features are easy to implement. We know that with changing technologies you'll need flexible designs that can handle feeds that may not exist today. We're ready.

FOLLOW-THROUGH

In a world where service is becoming secondary, our pledge is this: Wheatstone will be there from the beginning for as long as you need us, any time, day or night. We've been building television audio consoles, right here

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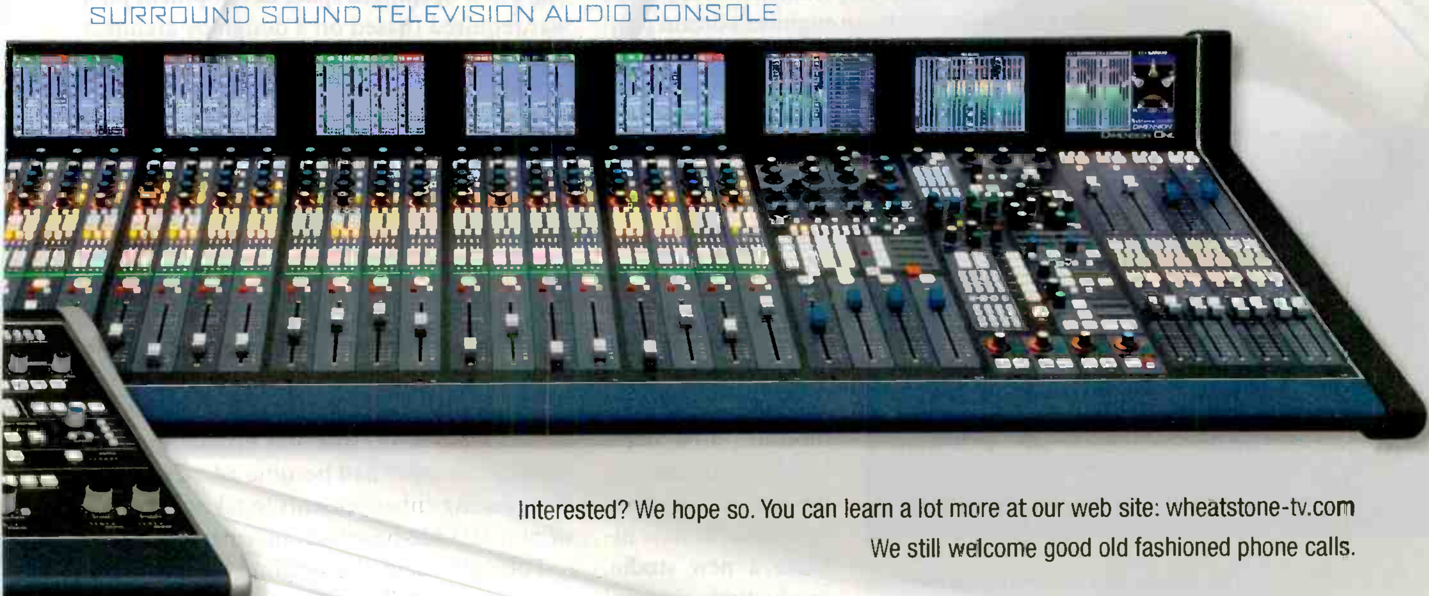
PRICE

It's challenging out there and you've got to make certain that every piece of gear you purchase is perfect for your needs at a price that's better than seems possible...AND that it's going to last. Our design initiative dictates that every piece of equipment we make fits that criteria. Our **D-8** pictured on the left starts at under $65K! And our **Dimension One** pictured below gives you a full set of big market features for a manageable mid-market price.

Regardless of the size of your operation (or your budget), Wheatstone is a perfect fit.

DIMENSION ONE

SURROUND SOUND TELEVISION AUDIO CONSOLE



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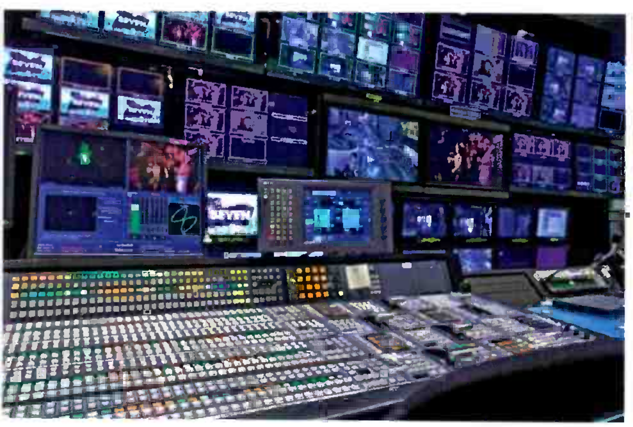
MTV does more with less

BY MICHAEL GROTTICELLI

To optimize the space for the production and operations staff, the control room features a three-tier console configuration that provides clear sight lines to the front monitor wall.

In the spring of 2010, Michael Bivona, vice president of engineering at MTV Networks, was faced with a perplexing challenge. He was tasked by the company's Music Group division with rebuilding the physical and technical production facilities of Viacom's MTV studio overlooking Times Square in New York City from SD to HD. While this might seem to be an engineer's dream project, Bivona had to do it with less space and a limited budget; and his staff still had to oversee production of a daily live show (the popular daytime show "The Seven") they had been doing previously.

Oh, and one other caveat: He had to launch the new HD studio on New Year's Eve 2011, a mere six months after first breaking ground on the new second-floor facilities that now include a new studio, production control room, audio mixing suite, ingest/playback and video control/transmission areas. Additionally, numerous HD updates and enhancements were also required for the post-production facilities, delay rooms and updated transmission facilities that are co-located on alternate floors of the building. They also updated two rooftop robotically controlled camera systems.



The production control room features a Sony MVS-8000A HD switcher and a monitor display wall composed of 20 flat-panel LCD monitors of various sizes.

Taking it in stride, Bivona said, "Our team is often asked to do the impossible with limited resources. The success of this project is a tribute to the great professionalism and expertise of the entire project team."

Considering all of these factors, and looking to partner with an established systems integration firm with a proven track record, Bivona called upon the expertise of Joseph Policastro, a founder and senior director of Broadcast Integration Services (BIS, Union City, NJ). The two had previously worked together in 1997 to build the original SD facilities in the previously larger space. They were now challenged to develop an integrated project program that could achieve more with fewer resources.

Since Viacom had surrendered half of its original space to a new tenant in the building, a new optimized space development plan was required (based on a design by architect Neil Tucker), and a totally new mechanical, electrical and plumbing (MEP) infrastructure was required (based on a design from AMA Consulting Engineers, NY). Most importantly, the broadcast technical infrastructure — which is based on HD and 3G-compliant equipment and systems — had to be carefully selected and integrated in a "tightly coupled" architecture in order to provide maximum flexibility, scalability and future expansion capability.

The design team also had to deliver the basic workflow processes that production and operations crews had become adept at while introducing enhancements to take advantage of new HD file-based advancements.

Policastro said the original design phase posed many challenges since there was much less space to work with, yet they had to accommodate all of the necessary systems and

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equipment requirements. The design plan that ultimately evolved is practical and retains its traditional production workflow, yet provides a highly flexible platform to accommodate new technologies as they become available. Several examples include the creative use of MADI systems to conserve wiring requirements and maximize the audio I/O channel capacity.

Production studio with a view

The new 4000sq ft production studio features Sony HDC1500 (1080i) cameras, used in various configurations. The studio camera system, which is wired to accommodate up to 14 cameras, is based on a predominantly SMPTE camera fiber backbone. A glass streetside view overlooking New York's Times Square, it provides a stunning backdrop for the studio, complete with an outdoor 45ft-wide electronic billboard in a large picture frame across the street that Bivona and his team often utilize to great effect as part of the show's imagery.

The audio control room houses a Solid State Logic C140 console mixing surface and several roll-around racking carts to accommodate a variety of outboard audio and peripheral support equipment.

To support the show production requirements, the team also had to implement more than 73 wireless channels for microphones, IFB and party line in one of the most active RF environments in New York City. Since the studio also accommodates audience provisions, and considering the addition of dynamic live music-based production components, there are also audio console mixing positions for both front of house (FOH) and monitor mix (MM).

The studio lighting system is based on high-efficiency LED fixtures, which solved the problem of suitably lighting a smaller studio space for HD production while simultaneously reducing power and heat loads.

HD production control room

The newly installed, state-of-the-art production control room features a Sony MVS-8000A HD switcher, Chyron graphics and a monitor display wall composed of 20 flat-panel LCD monitors of various sizes with multiviewer software for dynamically assigning sources, aspect ratio sizing and tally. In order to optimize the space for the production and operations staff, a three-tier console configuration features the necessary ergonomics while providing suitable sight lines to the front monitor wall. Analogous to a typical mobile production truck environment, these design efficiencies fully maximize the space while providing what Bivona said was a comfortable environment for the crew.

Adjacent to the control room is the audio mixing suite, which was acoustically designed to accommodate a conventional surround-sound monitoring environment. The audio control room houses a Solid State Logic C140 console mixing surface and several roll-around racking carts to accommodate a variety of outboard audio and peripheral support equipment. To optimize space and minimize cabling, the console interface engine, audio I/O frames and jackfields are all co-located in the adjacent equipment room racks. A Genelec GLM surround-sound monitoring system provides a software-based time delay and phase alignment application to optimize the room monitoring environment. Also, considering the recent passing of the CALM Act, Linear Acoustic loudness monitoring is also implemented in the audio system chain.

Audio communications

Implementing 64-channel MADI technology within the audio console core as well as throughout the entire plant facilitated the efficient distribution of multiple channels of audio (via a fiber-based infrastructure). The MADI transport stream allowed multiple audio mixer positions, routing switcher, intercoms and ancillary interface devices to share and process audio channels simultaneously. In some areas, discrete AES signal subsystem routing and support provisions were deployed as a backup.

The overall communications system is based upon an RTS ADAM Matrix with a combination of analog, MADI and IP-based RVON interface cards. The matrix, keypanels, telephone interface units and other gear are all seamlessly interfaced to provide a plethora of point-to-point, IFB, four-wire (e.g. cameras, etc.) and party-line communications requirements. Although the majority of the IFB and party line are interfaced to the talent and operations via wireless provisions; there

is also an outboard hard-wired RTS TW two-wire system as well.

Post production/playback

To support production and post capabilities, graphics and video files are shared over an extensive IP-based file sharing network.

The team has also implemented an EVS hard disk system for multicamera recording and playout production support. Studio and post-production teams share resources, linking the studio EVS system to the post-production Avid Media Composer editing workstations and ISIS Interplay media storage.

The Avid workstations are located on an adjacent floor and networked to an ISIS storage system that is tightly integrated with EVS XT[2] server and IP Director interface.

This provides for tapeless studio recordings, simultaneous Avid-based high-res editing and Avid Interplay-based ingest and low-resolution review of media from either the playback or producer areas in the studio facility located on the concourse level. Additionally, this provides seamless file-based capabilities from the numerous producers' offices and workstations situated on various floors throughout the building.

Although the facility has heavily segued to a tapeless workflow environment, there is a complement of Sony HDCam SR tape decks to supplement additional production requirements and legacy tape playback and record functions.

The equipment room, from both sides

The main equipment room posed several unique challenges due to the space constraints and required HVAC provisions. Again, taking cues from mobile production trucks and the team's skillful use of space, the room features two rows of 36in-deep racks, 8ft high, with equipment and components carefully installed on the front and rear sides. To provide suitable equipment cooling in such a dense environment, the HVAC system is implemented in a hot-aisle/cold-aisle configuration (e.g. racks oriented back-to-back for hot-aisle return). Maintaining the individual pieces of gear can be cumbersome at times, but Bivona and Policastro spent many hours planning which pieces of equipment were most likely to need more focused attention and which did not. Meticulously planned cable management practices were also implemented to best utilize the available cable routing paths.

Among the key equipment system components, the plant's routing infrastructure is based on a hybrid AV router from Evertz Microsystems. The router connects the facility's production equipment, audio mixers,

Project team

MTV Networks:
Michael Bivona, VP of engineering
Jim Brizzolara, director of studio engineering
Bill Anchelowitz, director of project management
Thayne Knop, director of content creation technologies
Tyler Marinello, studio engineering
Adrienne Bonfrisco, director of administration

Broadcast Integration Services:
Joseph Policastro, principal senior director
Kevin Henneman, senior design engineer
Adam Semcken, senior design engineer
Andy Morris, engineer
Robert Gilmartin, engineer
Judi Southard, logistics
Chris Butler, integration supervisor
William Frederick, lead tech
Javier Casilimas, engineer

Equipment list

ADC video and audio patching
Avid ISIS storage, Media Composer Nitris editing
Chyron HyperX graphics platform
Evertz EQX multiformat/hybrid router, VIP-X multiviewer, VistaLINK SNMP monitoring, test and measurement equipment
EVS XT[2] video servers
Genelec speakers and loudspeaker manager software
Gepco fiber systems
Image Video tally system
Joseph Electronics fiber systems
Linear Acoustic 5.1 monitoring and loudness systems
Middle Atlantic Products equipment racks
QTV teleprompters
RTS intercoms
Sony HDC1500 cameras, HDCAM SRW-5800 tape playback, LCD displays, MVS-8000A HD switcher
Solid State Logic C140 audio console
Sennheiser wireless microphones
TBC Broadcast consoles
Tektronix test and measurement gear
Telecast Fiber Systems SHED system
Telemetrics/Canon rooftop camera robotics
Vinten studio pedestals
Wohler audio monitoring

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and utility/external sources via fiber and copper cable.

Space savers

Looking to conserve real estate, the design team used onboard processing and fiber-optic and network connectivity for subsystems like production routing, audio mixing and intercom communications. Multiplexing typically handled by multiple trays of outboard devices and cabling with patching points were replaced with a common audio standard (MADI), as well as numerous embedders and de-embedders located in the router frame itself. This saved on rack space as well as reduced cooling and power requirements.

While some engineers might cringe at having to troubleshoot such a system, Bivona said the use of Evertz Vistalink SNMP monitoring software streamlines the trouble shooting process while monitoring system health and alerting them when a problem does arise.

Video control/transmission

The video control/transmission area contains all of the respective camera CCUs, OCPs and two SVO camera shading/QC positions. Although the system is currently wired to accommodate a total of 14 camera chains, only seven camera systems were installed initially to accommodate the current production requirements. Bivona indicated that for any ad-hoc and additional future production applications, it is simply a matter of purchasing or renting additional camera chain components.

There are also several broadcast service panels (BSPs) judiciously located in the studio as well as located in several alternate areas of the overall concourse studio facility, including support areas such as the green room and common corridors. This allows the additional flexibility for stand-up camera positions as required. All of the SMPTE camera fiber drops terminate back at a SMPTE patch panel in the equipment room. Operations then has the flexibility to assign any camera to any respective CCU.

In addition to the studio camera compliment, there are also two roof cameras with remote PTZ provisions from the SVO positions. These cameras provide a stunning view overlooking Broadway and Times Square. The interface panels on the roof areas include a Telecast Fiber Systems SHED system that converts SMPTE baseband signals to light (E/O) for distribution over single-mode fiber. HD signals are then converted back (O/E) at the receive end for interface to the core systems. Leveraging existing fiber drops on numerous floors within the building, the SHED allows HD camera signals to be transported over two ordinary strands of single-mode fiber to numerous locations throughout the building.

Additional provisions within the video control/transmission area also include utility color correctors and discrete routing switcher control panels to support MTV's release transmission path assignments.

Final comments and future applications

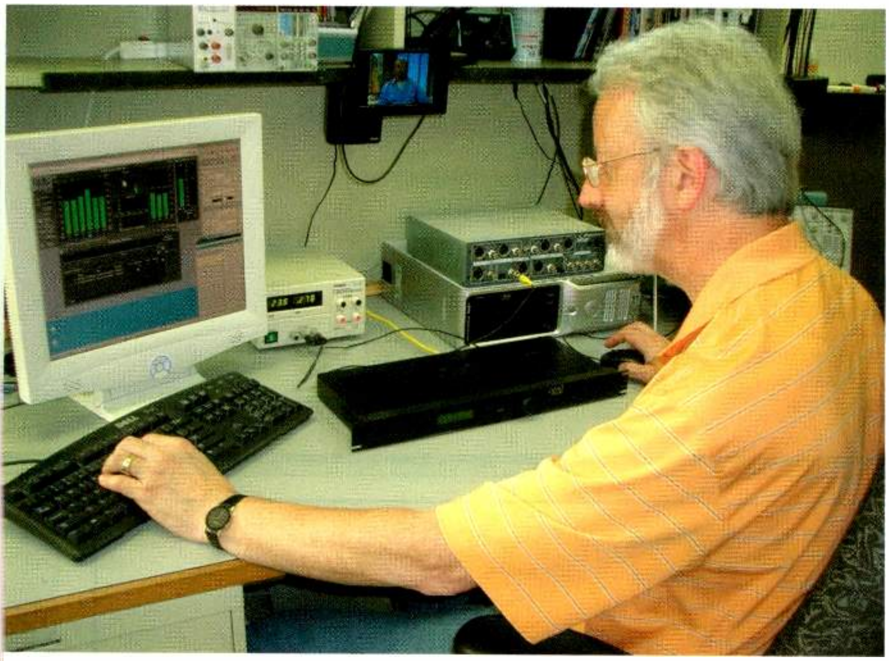
The original concept of ultimate flexibility while minimizing day-to-day operating costs has resulted in the ability of MTV Networks operators to quickly and easily recall preset show configurations and setups as necessary with minimal reconfiguring of the infrastructure in place.

With ongoing close attention to detail and the continual liaison of all of the project team members throughout all phases of the project, the new facility is one that MTV Networks can live with for many years to come, according to Bivona. As for other potential applications, the core infrastructure even provides the flexibility of producing 3-D content by adding some option cards if the network ultimately decides to experiment with a dual-stream 3-D format in the future.

Michael Grotticelli regularly reports on the professional video and broadcast technology industries.

New loudness metering standard

BY RICHARD CABOT AND IAN DENNIS



Dr. Richard Cabot uses a test suite developed for the Prism Sound dScope Series III audio test instrument. The test measures whether loudness meters conform to the new BS.1770 algorithm.

Loudness issues have been around since the beginning of broadcasting, or at least broadcast advertising, but they were exacerbated by the DTV transition and inconsistent use of the dialnorm metadata. Listener dissatisfaction increased, ultimately culminating in passage of the CALM Act.

In the design stages of DTV, considerable effort was put into development of loudness measurement technologies to support the dialnorm mechanism. This work culminated in the original version of the International Telecommunications Union (ITU) standard BS.1770, which describes a fundamental loudness measurement algorithm. It was validated through multiple sets of listening tests on various pieces of program material. The ITU standard also describes a true-peak meter for determining the peak amplitude expected when a digital audio signal is reproduced in the analog domain or transcoded into another digital format.

Several years ago, the ATSC developed recommended practice A/85, which describes how broadcasters should ensure a satisfactory listening experience for DTV viewers. The CALM Act mandates that the FCC enforce the loudness related portions of ATSC A/85, and its successors, through appropriate rule making. The loudness measurement portion of A/85 is based on BS.1770.

The CALM Act created an obvious opportunity for equipment manufacturers to provide loudness measurement tools. Although a few loudness measurement products were on the market by 2009, many more were introduced at NAB and IBC in 2010, and there are currently more than a dozen loudness measurement products of various forms on the market. Each is vying for a piece of the large market created as broadcasters equip their facilities for loudness measurement.

In late 2010, the ITU committee that maintains BS.1770 accepted (after much negotiation) changes submitted by the European Broadcasting Union (EBU). The result is a significant improvement in the calculation of loudness — one that makes the measurement much more sensitive to the loud portions of an audio segment. The effect is to prevent advertisers from significantly increasing the loudness of a portion of a commercial by manipulating the loudness elsewhere. Consider a hypothetical example where an announcer screams at widely spaced intervals throughout a commercial in an effort to get the viewers' attention. Though the average loudness might seem reasonable, the peak loudness will be quite annoying. The new method puts more emphasis on the louder portions and assesses this spot as louder than the original technique would have.

This new version of BS.1770 has just been published. Consequently, meters in use will need to conform to the revised specification. Most manufacturers of such products have been following the developments in the EBU and ITU and have upgraded their software to accommodate the change. Unfortunately, however, they haven't necessarily done it correctly.

The problem

As a user, how do you assess whether a meter you are considering meets the revised specification? How do you know if its software has been modified to conform to the new version standard? It's not as easy as testing a VU meter or a PPM. The EBU specifies some basic tests in its technical recommendations and provides the necessary waveforms on its website. However, these tests are basic and do not thoroughly test all aspects of compliance.

We will describe a suite of tests developed specifically to check every aspect of a meter's design. These tests also give diagnostic information about any implementation issues that exist. They are available at no charge as described at the end of this article.

Our test suite was developed by crafting signals whose parameters change dynamically so as to stress individual portions of the measurement in isolation. Each test can then maximize its sensitivity to the specific implementation errors it was designed to detect. The signals were developed using mathematical models of the algorithm, including models with various intentional implementation errors. The signals were optimized to give the largest difference between readings obtained by the correct model and those obtained by incorrect implementations.

The new BS.1770 algorithm operates on multiples of a basic 100ms interval, so readings differ slightly with variations between the start of the measurement and the start of the signal. These reading differences follow a cyclic pattern, with alignments 50ms apart creating maximal difference. Consequently, the test signals were evaluated at a reference alignment and at an alignment 50ms delayed. Signal characteristics were adjusted to minimize this difference, though sometimes this was in direct conflict with the desire to maximize the sensitivity to implementation errors.

Understanding the ITU standard

The original ITU loudness measurement algorithm is shown in Figure 1. The audio channels (except the LFE) are independently filtered with a low-frequency roll-off to simulate the sensitivity of the human ear and a high-frequency shelf to simulate head diffraction effects. The combined response of these filters is referred to as "K weighting" and is illustrated in Figure 2. (See page 54.) Surround channels are given a 1.5dB boost to account for the relative gain provided by their position on each side of the listener. The individual channel powers are summed to obtain the surround program's total power. This is averaged over the entire program, yielding a single number metric for the program loudness. If a "dynamic" indication of loudness is desired, a three-second moving average is typically used. Readings are reported in LKFS (Loudness, K-weighted, relative to Full Scale) which may be thought of as loudness dBFS.

The ATSC recommendation specifies that loudness measurements should focus on dialog or an alternate anchor element. The intent was that viewers would set the dialog loud enough to be intelligible in their environment, and that maintaining constant dialog loudness would maintain intelligibility.

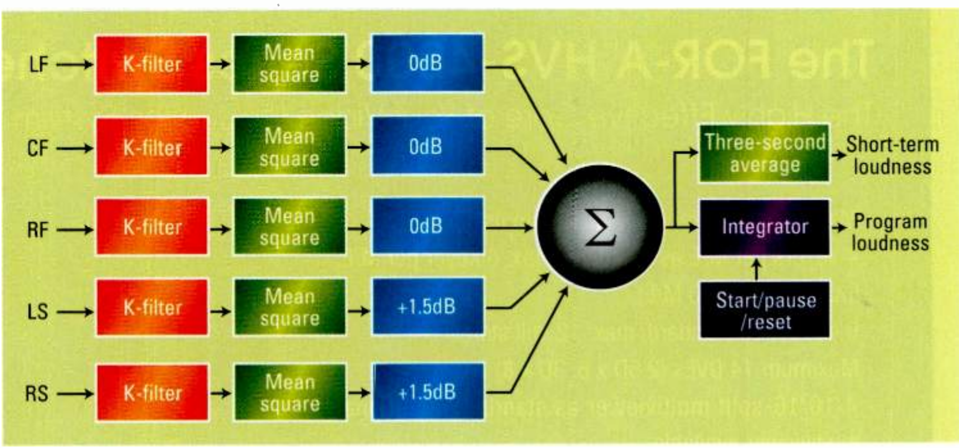


Figure 1. Shown here is the original ITU loudness measurement algorithm. Audio channels (except the LFE) are independently filtered with a low-frequency roll-off.

This assumed "well behaved" content (many commercials don't fit this description), and also depended on proprietary loudness measurement technology. In an effort to address these and other issues, the EBU PLOUD committee revisited BS.1770. Their work resulted in the 2011 revision of BS.1770.

This revision maintains the same filtering and power measurement method used in the original standard, but changes the way measurements are averaged and presented. The integrator stage of Figure 1 is replaced with

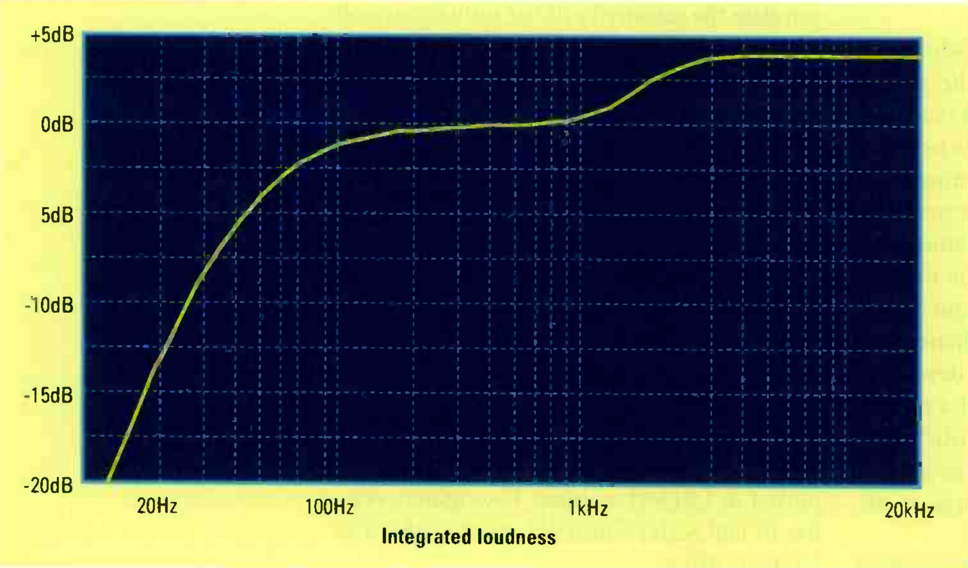


Figure 2. K weighting, illustrated here, is the combination of results between low-frequency roll-off and high-frequency shelf filter responses.

the processing shown in Figure 3. The channel power is summed over 400ms intervals. These intervals overlap by 75 percent, so a new value is obtained every 100ms. Results are gated with a start/stop control to allow selection of the audio segment to be measured. An absolute gate of -70LKFS is applied, which automatically eliminates lead-in and playout portions of isolated audio segments.

The algorithm focuses on the foreground portion of the audio by a two-step averaging procedure (the orange elements in Figure 3). The 400ms measurement values are averaged over the content being measured. The resulting LKFS value is decreased by 10 and used to gate the 400ms measurement values. This "relative gate" focuses the assessment on foreground sounds, the elements that generally dominate viewers' judgments of program loudness. The values that pass the relative gate are averaged to form the final reading called "Integrated" loudness (abbreviated "I").

The standards also specify a true-peak meter. This is a device that measures the peak value a digital audio waveform will reach when it is reproduced in the analog domain, or when it encounters many forms of digital processing. To understand the problem, recall that digital audio represents a continuous analog signal by a series of samples, taken at regular intervals determined by the sample rate. As Figure 4 on page 56 illustrates, there is no guarantee that samples will land on the audio waveform peak.

However, these samples do represent the underlying audio waveform, and when it is reconstructed, the peak will be restored. This peak can also occur when the samples are subjected to many types of processing — anything that introduces phase shift or time offset — such as sample rate conversion, filtering or delay. If this happens in the digital domain, the new samples may clip, even if the original samples did not reach digital full scale. Because many peak meters merely display the maximum audio sample, they incorrectly gauge the system headroom.

The EBU recommendation introduces other

If a peak happens in the digital domain, the new samples may clip, even if the original samples did not reach digital full scale. Because many peak meters merely display the maximum audio sample, they incorrectly gauge the system headroom.

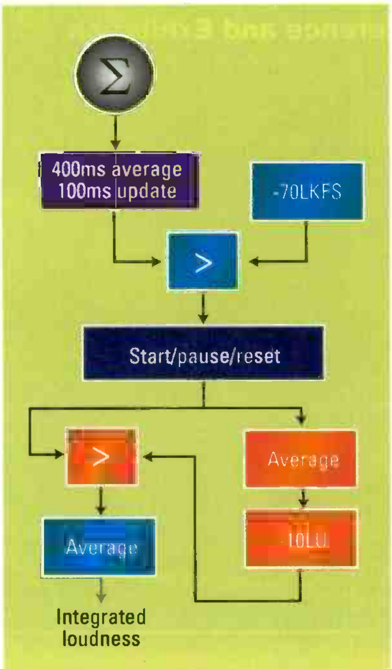


Figure 3. This chart reflects the EBU PLOUD committee's 2011 revision of BS.1770. The revision changes the way measurements are averaged and presented.

Listeners' perception of loudness is best described with a longer averaging time. The EBU recommends a running three-second average that it calls the "Short-Term" loudness.

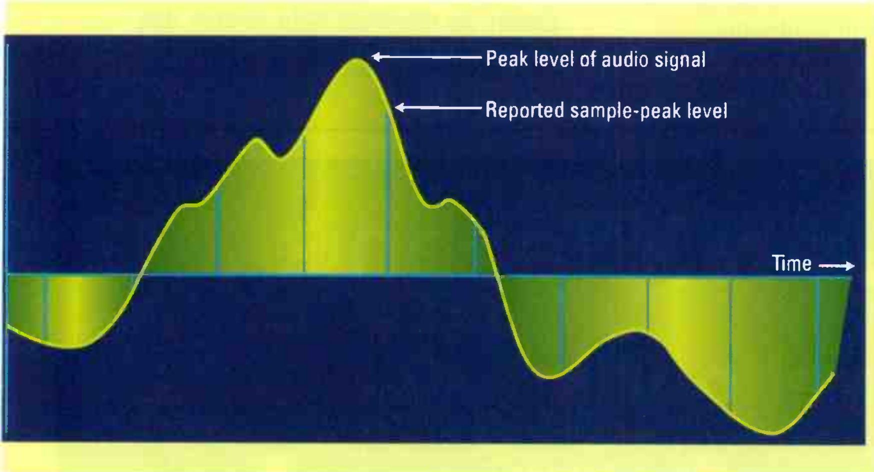


Figure 4. This graph shows there is no guarantee that samples will land on the audio waveform peak.

measures that are still under consideration by the ITU. Intended to assist mixers and program personnel in creating and characterizing content, their acceptance by the ITU is unlikely to impact CALM Act requirements. However, given their potential usefulness in production, it is helpful to understand them.

The EBU specifies "Momentary" loudness (abbreviated "M") as the stream of 400ms measurements that drive the gating mechanisms described earlier. When displayed on a meter, they look much like a VU display since the 400ms averaging time is close to the 300ms of a classic VU meter. Mix engineers watch the display for help estimating the program loudness of live productions. Listeners' perception of loudness is best described with a longer averaging time. The EBU recommends a running three-second average that it calls the "Short-Term" loudness (abbreviated "S").

The EBU also defines a measurement called "Loudness Range" (abbreviated LRA). This is derived from the Short-Term loudness using a relative gating process similar to that described above but with the gate set at -20. The LRA is the span from the 10 percent to 95 percent points on the distribution of Short-Term loudness values that pass the relative gate. The LRA is descriptive of the program material dynamic range. Using the 95 percent point allows occasional extremely loud events, while the 10 percent point ignores modest silent intervals during the program. If the LRA exceeds about 15, it is likely that viewers will be unable to find a single volume control setting appropriate for the entire program.

Loudness meter evaluation

The test suite described here is available for testing BS.1770 compliance of any loudness meter. It may be downloaded as a dScope III script or as .wav files from www.prismsound.com/loudness1770, and also as a series of wave files and documentation from www.qualisaudio.com. More complete documentation of the tests, their design and their expected results are included in the download package. Any new tests will be added as they are developed.

The current 16-test menu for the script-based implementation is shown in Figure 5. All tests, except Test 2, are stereo signals and should be applied to the LF and RF channels of a surround loudness meter. All tests, except the first three, comprise a 1kHz sinewave at varying amplitudes. When the expected result is a range rather than a specific target, this is due to the 100ms alignment uncertainty.

Test 1 checks the accuracy of the True-Peak meter. The initial waveform is a one-eighth sample rate, -6dBFS sinewave in which samples are chosen to correspond to the sinewave peaks. After three seconds, the frequency changes for one cycle to one-fourth sample rate, and the amplitude increases to -2dBFS.

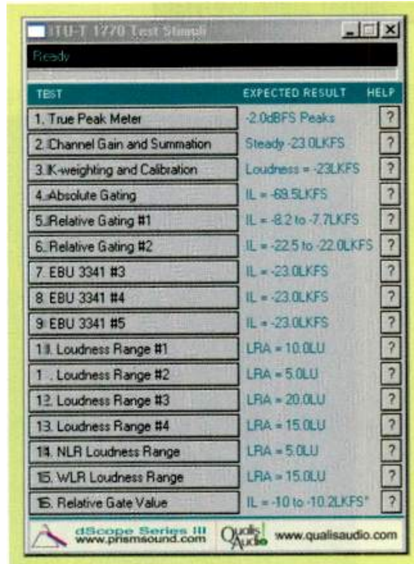


Figure 5. This image shows the menu for script-based implementation. The full suite for testing BS.1770 compliance of any loudness meter currently consists of 16 tests.

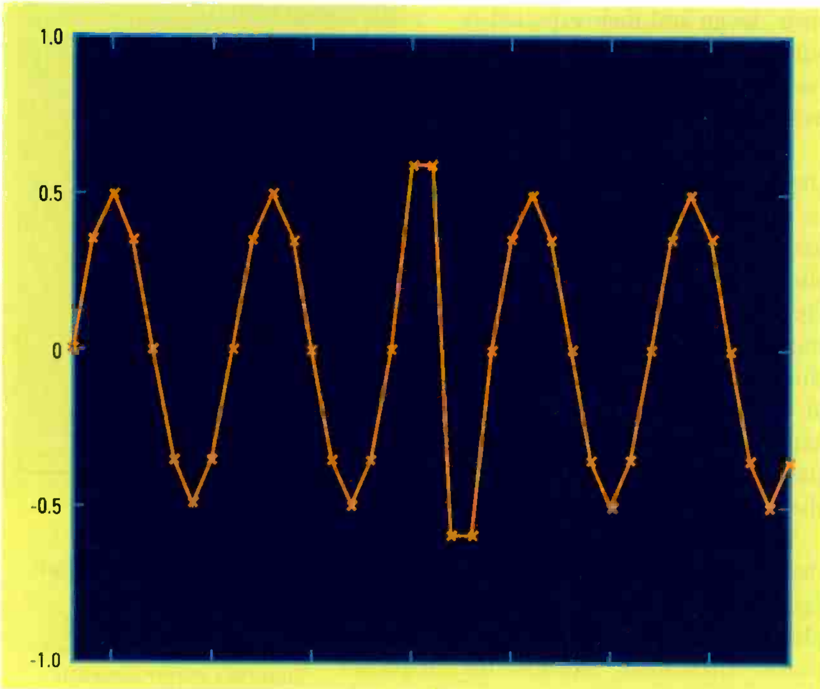


Figure 6a. This graph shows changes to the -6dBFS sinewave after the frequency changes for one cycle to one-fourth sample rate and the amplitude increases to -2dBFS. The samples are chosen to occur 45 degrees off the sinewave peaks.

The samples are chosen to occur 45 degrees off the sinewave peaks as shown in Figure 6a. When this waveform is properly interpolated (as would occur when reproduced in the analog domain), the result is the waveform shown in Figure 6b. All meters will read -6dBFS initially. After three seconds, the reading should increase to -2dBFS. Non-interpolating meters will increase to -5dBFS.

Test 2 is Dolby Digital-encoded and stimulates all channels simultaneously, including the LFE. Power summation is checked by using sinewaves of slightly different frequencies. A compliant meter reads -23LKFS. If the meter incorrectly sums in the time-domain, the reading will cycle. If the meter includes the LFE, the reading will be too high.

Test 2 is Dolby Digital-encoded and stimulates all channels simultaneously, including the LFE. If the meter incorrectly sums in the time-domain, the reading will cycle. If the meter includes the LFE, the reading will be too high.

Test 3 checks the filter response at six frequencies: 25, 100, 500, 1k, 2k and 10kHz using sinewaves of varying amplitudes to give a constant reading of -23LKFS.

Test 4 alternates between -69.5dBFS and -90dBFS to exercise the absolute gating function. A compliant meter reads -69.5LKFS. A meter that does not implement absolute gating reads -71.5LKFS to -72.7LKFS.

Test 5 steps the amplitude between -23 and -6dBFS at intervals between 0.5 and 1.4 seconds. A meter that correctly implements relative gating reads -7.7LKFS to -8.2LKFS; a non-compliant meter reads between -13.2 and -13.5.

Test 6 checks aspects of relative gating missed in test 5 by alternating between -36dBFS and -20dBFS. A compliant meter reads between -22LKFS and -22.5LKFS, whereas non-compliant meters read between -24.3LKFS and -24.7LKFS.

Tests 7, 8 and 9 are corrected implementations of EBU Tech 3341 Test Cases 3, 4 and 5 that evaluate basic loudness meter operation.

The versions created by the EBU have slight errors discovered in the test suite's development.

Tests 10, 11 and 12 are implementations of EBU Tech 3342 Test Cases 1, 2 and 3, which evaluate basic loudness range meter operation. Each test takes 40 seconds and spends half its time at each of two amplitudes 10dB, 5dB and 20dB apart, respectively. The loudness range meter should read these same values.

Test 13 implements EBU Tech 3342 Test Case 4, which evaluates loudness range meter gating. It spends 20 seconds at each of the amplitudes -50dBFS, -35dBFS, -20dBFS, -35dBFS and -50dBFS. A compliant meter will ignore the -50 amplitudes and read 15LU. If gating is not implemented, the meter reads 30LU.

Tests 14 and 15 are sinewave-based alternatives to EBU Tech 3342 Test Cases 5 and 6, which are narrow-loudness-range (NLR) and wide-loudness-range (WLR) program clips. The NLR test uses amplitudes of -50, -40, -25, -20dBFS, -15dBFS, -20dBFS, -25dBFS, -40dBFS and -50dBFS, while the WLR test replaces the -25 amplitudes with -35dBFS. In both tests, the -40dBFS amplitudes are maintained for three seconds and the -15dBFS amplitudes for two seconds, while the others have 23 one-second durations. The durations at -40 and -15 test the 10 percent and 95 percent statistical processing defined in the loudness range algorithm.

Test 16 measures the meter's relative gate threshold, displaying -8 or -10 as appropriate. This is useful to determine if your meter is designed around the old standard.

If a loudness meter gives the expected results for each of the tests above, the likelihood is high that the implementation is compliant with the latest version of BS.1770.

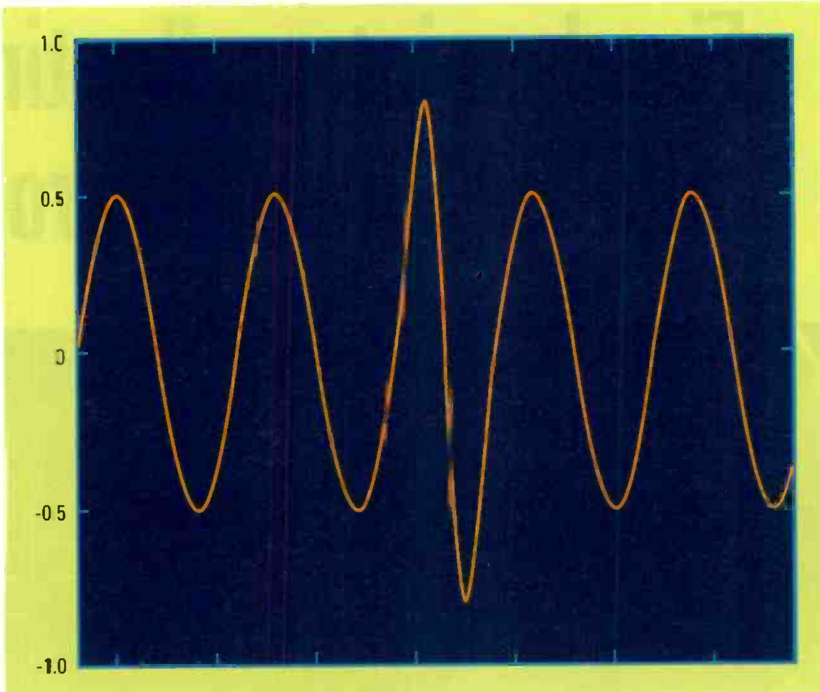


Figure 6b. This figure shows what happens when the waveform is properly interpolated, as would occur when it is reproduced in the analog domain.

Richard Cabot is the CTO of Qualis Audio. He was previously chairman of the AES digital audio measurement committee for the development of the AES-17 standard. Ian Dennis is technical director of Prism Sound and, as vice chair of the AES digital audio measurements committee, wrote the document which became the true-peak meter specification in BS.1770.

Fixed-point vs. floating-point numbers in audio processing

BY PATRICK WARRINGTON

In digital audio, using a fixed-point number format can yield benefits in resolution and dynamic range.

The traditional view is that the floating-point number format is superior to the fixed-point number format when it comes to representing sound digitally. In fact, while it may be counter-intuitive, there is a case to be made that the use of floating-point numbers yields lesser resolution than the use of fixed-point notation.

Floating-point numbers defined

Floating-point numbers are like scientific notation on calculators: They have a mantissa, the number part, and an exponent, a multiplier used to scale the number part. For example, 1.414 x 10^3 is a floating-point number with a mantissa of 1.414 and an exponent of 3. The attraction of this form of notation is that it can be used to express numbers over a much larger range than would be possible if the same number of digits were used in a fixed-point (integer) number.

To understand the impact number format has on digital audio systems, we must consider two properties: resolution and dynamic range. Resolution or numerical precision is determined by word length. As it increases, resolution improves. Dynamic range is also determined by word length, but in the floating-point format, it can be dramatically extended by the choice of exponent. Compare, for example, a 24-bit fixed-point number that has a dynamic range of about 144dB, to a 24-bit floating-point number where eight bits are designated as an exponent. The latter has a dynamic range of more than 1500dB.

Choosing the right number format

To choose the right number format for a digital audio system, the dynamic range and resolution must be large enough to afford faithful representation of all audio signals that may be encountered. Sound pressure level (SPL) is a logarithmic measurement of sound levels where 0dB represents the threshold of human hearing. (See Table 1 for real-world examples.)

In general, music, sports and drama do not demand the full range of sounds — from jet engine to gently rustling leaves. Nonetheless, the 24-bit fixed-point format remains unsuitable for digital audio systems because audio processing can introduce errors that are manifested as audible noise unless additional resolution is provided. Note that it is resolution, not dynamic range, that is needed. To illustrate this, let's take a look at the most important audio processes: gain, mixing and equalization.

Gain, mixing and equalization

Applying gain to a digital audio signal means multiplying by a big number for louder and a small number for quieter. The problem comes when the product is a number that doesn't fit neatly into the number of digits you have to represent it. There is usually an extra bit that you have to get rid of, a process called truncation. How you truncate affects sound quality. Simply rounding up or down introduces an unpleasant quantization noise, so a better idea is to add a random number to the leftover bits and then round up or down. This idea is known as dithering, and it makes low-amplitude signals sound much better. The downside is that the number format must carry additional resolution in the form of footroom bits to dither against. Here, floating point does not help because there is no need for extended dynamic range. In fact, any bits given over to carrying an exponent are extraneous and would be better deployed extending notation in the mantissa.

If multiple signals are to be mixed (added), then it is a good idea to provide some additional headroom bits to extend the dynamic range during the calculation, but this does not have to be a large number. If a floating-point number were used to generate intermediate headroom, as the mantissa is scaled up, footroom bits would be lost, affecting any subsequent dither calculation and introducing noise. Hence, in mixing calculations, the floating-point format is in fact a liability.

Equalization is a more complex case. To make it more convenient for processing, the ubiquitous biquad calculation used in digital filters may be arranged in different forms. For example, the popular direct form II is used to reduce the number of computations in systems where DSP cycles are at a premium. The tradeoff is that the resulting intermediate calculations have such a large dynamic range that floating-point format must be used, at least in a system constrained to a fixed word length such as a DSP chip. In other words, the need for using floating point is a consequence of cost-cutting rather than a consequence of the pursuit of high quality.

Limitations of a rigid floating-point format

The problem that arises from the use of a rigid floating-point format (for example, that found in ADSP SHARC chips) is that the resolution is fixed by DSP architecture, not by the requirements of the calculation. Most of the time, this is adequate, but there are certain filter configurations where it is not. The elevation of the noise floor that results from the resolution limit of the floating-point format is significant. (See Figure 1.)

If high-quality sound is the goal, the best approach is to first decide on the desirable level of performance and then select the number format to achieve it. In the case of EQ, a high level of resolution is needed in parts of the calculation to avoid generating the kind of noise evident in Figure 1. A flexible architecture allows word length to be increased so that it most precisely matches desired performance.

Using the same filter in an audio console that relies on fixed-point-based digital signal processing shows how the high word length, fixed-point approach has reduced the noise floor of the filter to almost exactly that of the test set. (See Figure 2 on page 62.)

Lack of resolution produces a secondary effect that also impairs filter performance. This is because you cannot add very big numbers to very small ones. Let me demonstrate this with an extreme case. Imagine that you have adopted a seven-digit floating-point format with a four-digit mantissa and a three-digit exponent. You can represent the number

Real-world sounds	Sound levels
Calm breathing, or gently rustling leaves	10dB
Normal conversation	40dB to 60dB
Passenger car at 10 meters	60dB to 80dB
Hearing damage (long-term exposure)	85dB
Vuvuzela	120dB
Level of sound that can cause physical pain	130dB
Jet engine at 30m	150dB
M1 rifle at 1m	168dB
Stun grenades	170dB to 180dB

Table 1. Here are some examples of the dynamic range of sounds in the real world.

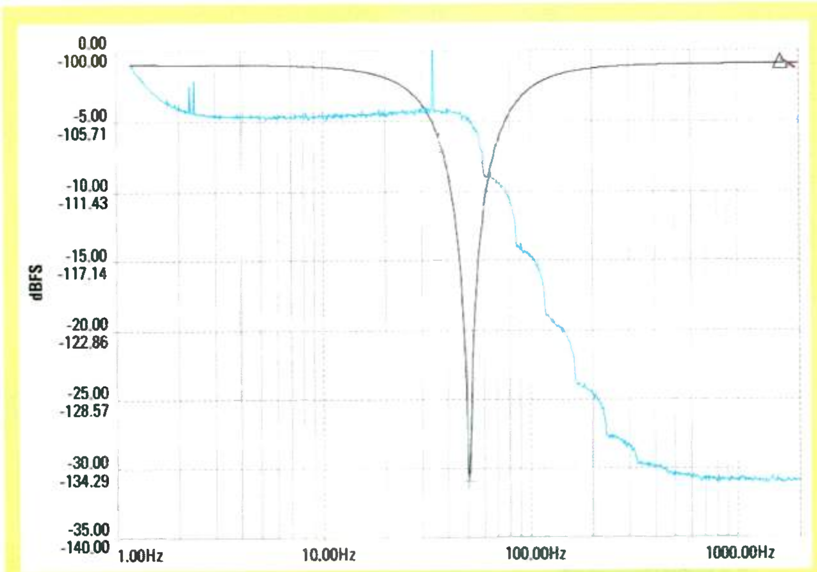


Figure 1. This plot shows the THD+N for a 30dB notch filter at 50Hz executed in 40-bit floating-point (direct form II).

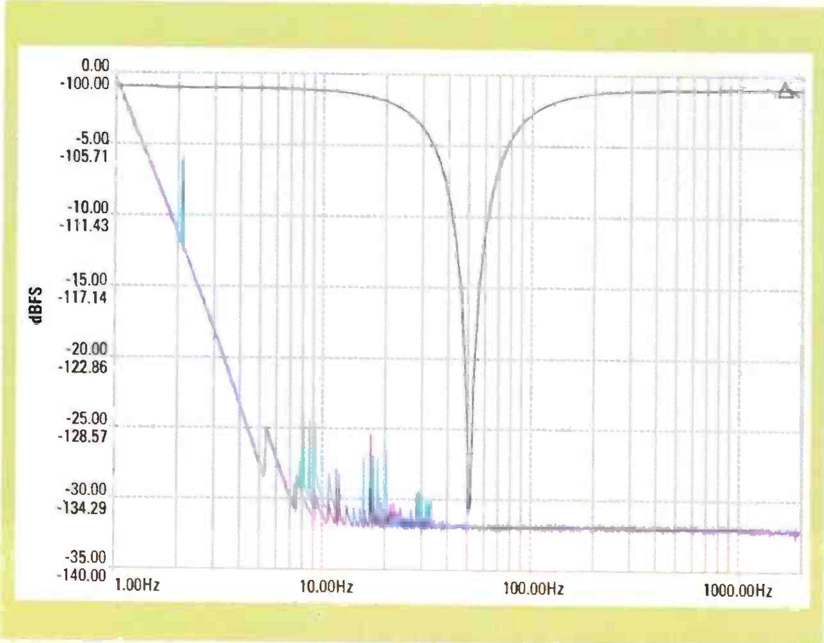


Figure 2. This plot shows the THD+N for a 30dB notch filter at 50Hz executed in fixed-point notation. The blue trace is the THD+N of Calrec's Bluefin2 processing. The red trace is the THD+N of the the test set looped output to input.

1 million by writing it as 1000 x 10^3. Now, add the number 999 to this. You should get 1,000,999, but since you have only four digits available, the result you end up with is 1000 x 10^3 — the same number you started with. In other words, adding 999 has no effect on the result, an illustration of the limitations of the floating-point system's lack of resolution.

In conclusion, it is inevitable that the use of floating-point numbers will deposit arithmetic errors when they are subject to the mathematics of audio processing. Candidly, these errors are small and often irrelevant, especially when compared to the many other threats encountered by audio quality on the path between microphone and living room. Still, if numerical precision is our goal, as it should be, then we should strive for objective analysis of the virtues of competing notational systems — floating point vs. fixed point — rather than falling prey either to intuitive explanations or the common wisdom.

Patrick Warrington is technical director at Calrec.

Processing audio for mobile DTV

BY MICHAEL GROTTICELLI

As mobile device usage grows, so has the need for high-quality sound.

Just when audio mixing engineers were finally getting comfortable with 5.1 surround sound and other types of multichannel audio formats for sophisticated home and public theater use, along comes the need to correctly process stereo audio signals for portable devices with 1in speakers.

Broadcasters have to understand the limitations of their one-fourth rate (or less) transmission channel and mix the content knowing it will be heard on tiny earplugs or little speakers. Reducing dynamic range is a way to ensure comfortable listening.

The goal for content distributors is to create a consistent and predictable audio experience without having to preprocess everything upstream of the device. Sending out the proper tags and metadata, the encoder will then instantly know it has to make sure the content is rendered (decoded) properly for the mobile device.

In most cases, the dynamic range needs to be more tightly controlled due to physical and electrical limitations of cell phones, tablet PCs and other devices. Couple these limitations with the fact that these devices are normally used in very noisy environments (trains and buses), and program audio can become unreliable and frustrating for users if not mixed right.

Dolby Labs has done some testing of portable mobile DTV reception devices, as part of its ongoing research into this area. The company's Dolby Mobile product (a suite of post processing technologies for mobile devices) includes a series of custom compression algorithms that are used to improve the audio experience on mobile devices. Some of these algorithms can take a surround-sound mix and "virtualize" the 5.1 signals for rendering over ear buds, headphones and even a larger device's compact internal speakers.

One of the things Dolby's engineers learned during their tests is that TV-based content (movies, sitcoms, etc.) are mixed with different production philosophies, in terms of peak levels and dynamic range, than MP3 music files. Modern music is typically produced with peak levels frequently at or just below clipping and the loudness averaging approximately 11dB below that. Therefore, it doesn't have a lot of dynamic range, whereas movies and TV content offer a variety of levels relating to dialogue, sound effects and ambient noise. In this case, the typical average loudness for this type of content is in the -22 to -29 below clipping — 12dB to 18dB lower. (Remember, a 10dB difference in loudness is approximately equivalent to a subjective doubling, or halving, in perceived loudness.)

Dolby's research in this area sought to establish a new reference level for what sounded good on small speakers, but also looked to leverage metadata computed at the encoder upstream (either in Hollywood or at a broadcast facility). The decoder at the device could then identify the incoming bit stream and activate different modes to get the best results.

The company has taken the two main legacy decoder operating modes, Line and RF (used for DVD and TV content), and developed a new reference level for portable devices that are based on the same principals. It's called

Portable Mode and is present in the latest generation of Dolby decoders and processing tools for mobile devices.

Jeffrey Riedmiller, Director, Sound Platform Group, at Dolby Labs, worked with a few of his colleagues last summer to develop a comprehensive white paper that was presented at the Audio Engineering Society conference in London in May 2010. It describes a nondestructive method for controlling playback loudness and dynamic range on portable devices and is based on a worldwide standard for loudness measurement as defined by the ITU.

They worked with a database of 25,000 songs, along with hundreds of TV and movie titles, and found that when looking at music, the mean or medium of that large data set ended up being about 11dB below maximum level. So, Dolby decided to establish -11dB as the new audio reference point for its decoders embedded within portable devices. Using this level, portable device manufacturers could leverage the metadata contained in the latest Dolby formats to non-destructively process the wide dynamic range content, such as TV shows and movies, in a manner optimized for these device types all while providing a better subjective match (in terms of loudness and dynamics) to music files being played. The result gives the consumer a much improved (and more consistent) experience in terms of loudness, intelligibility and peak level control.

With this metric as a baseline, most encoders can be used to control the level and dynamics of content processed with Dolby's technology and metadata generated with other companies' technology as well. This new metric is also being deployed inside Dolby's Mobile platform and its suite of core technologies that are used for Dolby Pulse and Dolby Digital Plus decoder products.

Riedmiller said this new Portable Mode also makes things easier for mobile video producers to create content that complies with new government mandates to protect against audio loudness. It does this while also bringing the dialogue out in front to make it more intelligible (and enjoyable) when viewing in noisy environments. (This is especially import for live sporting events, where the crowd can drown out the announcer if the mix is not done right.)

Meanwhile, the associated metadata is still attached to the media so that if a person plugs his or her iPhone into a docking station, it can play out that music or TV show on a large home theater speaker system and sound great, with the originally mixed full dynamic range.

Linear Acoustic's AERO.mobile can downmix and process separate audio streams to provide controlled, intelligible results for mobile devices.

However, there are a number of challenges with processing audio signals for mobile devices that engineers should be aware of, according to Tim Carroll, founder of Linear Acoustic. He said that audio programs may vary from mono to 5.1 channels, but the current mobile audio pipeline is stereo (and mono in many cases). This requires compatible and automatic downmixing that is stereo- and mono- compatible.

Linear Acoustic offers its AERO.mobile product, which handles up to two separate mobile audio streams. It combines downmixing with processing and psychoacoustic tools to produce a controlled and intelligible result for mobile devices. The mobile processing will soon be available in the company's AERO.file file-based product as well.

The Dynamic range processing has to be tailored not only for controlling range, but also for protecting dialogue, and this requires managing a different set of pyschoacoustic challenges. Carroll said that multiband processing, coupled with these new psychoacoustic tools, could work extremely well. Attention must also be paid to the codec itself as different versions and mixes of encoders and decoders can produce surprises.

Peter Poers, Director Sales & Marketing, at Jünger Audio-Studiotechnik, said broadcasters have to understand the limitations of their one-fourth rate (or less) transmission channel and mix the content knowing it will be heard on tiny earplugs or little speakers. Reducing dynamic range is a way to ensure comfortable listening.

Jünger makes a Television Audio Processor that combines audio conditioning and encoding in the same unit. It also performs dynamic range management, loudness and peak level control, consistent spectral balance (by Spectral Signature), and metadata control for the transport stream encoder.

Broadcasters also have to recognize the limitations of their legacy encoders. Poers said if a station arranges and mixes for a "consistent spectral energy balance," the encoding results would improve by estimating low transmission bit rates in the system.

Therefore, broadcasters launching new Mobile DTV services should talk with equipment vendors to ensure that they support new decoder operating modes, like Dolby Portable, and also the generation of metadata in their systems. The encoder also has to generate the proper metadata.

This should not be a problem as the original MPEG and DVB specs have had bit fields within it that carry audio metadata across decoders. That said, many manufactures have not implemented it to date.

Jünger's Television Audio Processor combines audio processing and encoding in the same unit.

Standards are always key to success in broadcasting, and right now consistent use of a way to process audio for mobile devices is still evolving. Yet, progress is being made in several places (including within the Advanced Television Systems Committee) as new ATSC A/153-compliant encoders and decoders become available to help ensure content sounds good and levels meet with consumer needs. A satisfying user experience is perhaps the most important element to a successful mobile TV service. Being able to hear it correctly is a big part of that.

Michael Grotticelli regularly reports on the professional video and broadcast technology indstries.

Adaptive, automated loudness control

BY PETER POERS

Today, we live in a world of automation. You see it in our industry, in our daily working lives and even at home. Everywhere you look, there is technology that is designed to make things happen faster by replacing human effort with machines.

The broadcast industry is no exception to this rule. Automation is everywhere, and the trend to speed things up with technology is apparent, especially now that the Internet has given us new ways to disseminate audio and video. The traditional technologies that we used to rely on for production and distribution are changing to encompass this new world and this, in turn, is further reducing the influence of human control. This begs the question: Will these changes alter our experience of moving pictures and audio? Should we preserve the human control of audio signals in such a highly automated broadcasting world?

Malaysian broadcaster Astro has installed more than 30 channels of Jünger Audio's LEVEL MAGIC automated audio processing at its facility in Cyberjaya.

Impressions of a broadcast facility

Whenever I imagine a totally automated television broadcast facility, two things immediately spring to mind. The first is what I call the "fleece jacket syndrome." No matter how warm it is outside, everyone inside is wearing a fleece jacket or a sweater because the air has been cooled down to such an extent that it feels like a permanent winter. No wonder broadcasters are so keen to grab one of those fleece jackets whenever a manufacturer is giving them away!

The second image I have is of the control room — a huge place with dimmed lights and pictures flickering away on massive video walls. No matter where you are in the world, these control rooms all look the same — incoming channels, transmission channels, leased lines, etc. In front of this, there is inevitably a large desk with a number of control panels and intercoms, as well as computer screens showing program lists, scheduling information, the condition of the transport streams and perhaps the satellite uplink configuration. Everything in there needs to be controlled, but where are all the people? Usually there are very few of them. The image of any individual channel on the multiscreen display gives the channel name and visual information about the video content. There may also be some bargraph-like visualization that tells you something about the running audio. The engineers who are present are surveying the operation of all the different technical gear involved in the process. There might be a station for quality monitoring of the audio and video, but you don't find this in every control room. This is what a totally automated broadcasting transmission center looks like. This image is, in fact, the reality for broadcasting centers, turn

> When I image a totally automated television broadcast facility, everyone inside is wearing a fleece jacket or a sweater no matter how warm it is outside because the air has been cooled down to such an extent that it feels like a permanent winter.

around facilities, uplink teleports, and cable and IPTV headends everywhere in the world.

If you look around one of these facilities and try to find an audio fader or a video switcher, you will be disappointed because there aren't any. Nor is there anyone there to control a switcher or fader. In modern facilities, staff are managing data that are being stored on massive servers with the unbelievable hard drive storage of many terabytes. Programmed schedulers are running the playout for hundreds of TV channels. The monitoring that is being carried out is primarily to control the basic condition of the running video and audio. There isn't any quality monitoring in place.

In such an environment, how does one ensure that certain quality standards for audio and video are being maintained for each dedicated transmission or distribution network? This is a delicate issue because, in an automated facility, there is a need to achieve optimum results while keeping human effort to a minimum.

In terms of picture content, this isn't too difficult because throughout the production chain, the video signal is subject to many dedicated controls in order to keep it within legal transmission requirements. Video format conversion is commonplace, and various levels of performance are available. Video is data that is delivered in a compressed format for the digital domain, and all those compression devices are performing a number of video filters and algorithms to guarantee the best possible signal quality within an acceptable bandwidth for the transport stream.

However, when it comes to audio, things are different. Of course, there is technology on the market that is designed to convert the audio into the format required by the transport stream and to meet the technical specifications needed to guarantee the best sonic performance. But that is where the similarity stops because, with audio, there is no common overall technical specification that is designed to check or legalize the content.

Audio engineers do have the benefit of some technical recommendations, but sometimes this doesn't solve the problem. For example, if audio is coded into the digital domain, the highest possible value in level should be 0dBFS. But that basic recommendation doesn't really reflect the wide variety of different, practical ways in which one can deal with digital audio. Take, for instance, CD audio. At 16-bit audio resolution, CD audio uses almost all of the available coding space (theoretically 96dB system dynamic). This means that if a broadcaster is getting music from a CD, it will be controlled and mastered to reach 0dBFS as its maximum value. That's very different from a typical broadcast signal, which now uses -18dBFS or -20dBFS as its alignment level in order to keep it in line with the recommendations issued by international regulatory authorities such as the ITU, the EBU and ATSC. What this means for broadcasters is that audio content coming from different sources can have very different level conditions. Content from CD audio can be more than twice as loud as audio coming from a standard TV broadcast.

Differences in program loudness are painful

As an industry, one thing we are all realizing is that the level of audio sources used in broadcast transmission can vary wildly. Of course, nothing is transmitted that hasn't already been processed and quality-controlled, but we still have a number of issues to contend with relating to audio levels and audio control.

Over the last 10 years, broadcasters have begun to understand that technical-oriented level control doesn't necessarily solve their problems when it comes to delivering better quality audio transmission. Audio loudness is now a hot topic, and there have been many articles published that discuss this issue and give the background to it. In simple terms, all audio sources that have been processed to control loudness should deliver the same overall loudness impression. And proper loudness control is definitely improving the quality of audio in digital broadcasting systems.

But what remains an issue is how loudness control is applied in today's world of automated broadcasting. Broadcasters have no choice but to use integrated or external audio processing to perform this control. I can already hear the complaints that some people, most notably skilled audio engineers, will make in response to that statement. I know they will be asking how automated online loudness control can be pleasant to the ears and be done in a way that isn't detrimental to the audio.

I can understand their concerns, but in a world where automation is king, there is no other choice because broadcasters are not likely to install an audio booth with proper monitoring and fader control where someone can sit and perform the task manually. We have to recognize that some kind of automated audio control is required if broadcasters are to comply with the new

True Visions, a cable and satellite television provider in Thailand and Southeast Asia, has installed LEVEL MAGIC automated audio processors to balance audio levels across its pay-TV channels.

One thing we are all realizing is that the level of audio sources used in broadcast transmission can vary wildly. Nothing is transmitted that hasn't already been processed and quality-controlled, but we still have issues to contend with relating to audio levels and audio control.

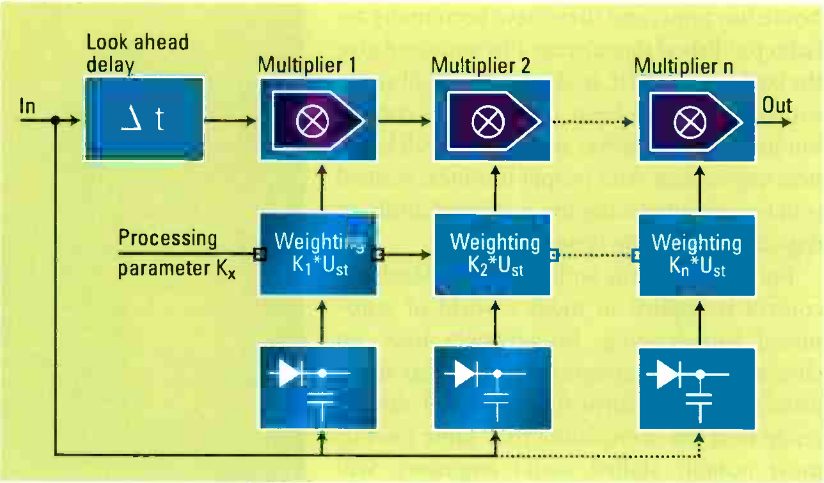


Figure 1. The digital implementation of the multiloop design permits a short delay time to be introduced into the audio signal path. This allows the gain changing elements to "look ahead" and determine the correction needed before applying it to the delayed signal just in time to control even the fastest transients.

loudness standards and recommendations and maintain the highest level of quality for their audiences.

Given that there is no other choice, all that remains to be discussed is what characteristics this online automated loudness control system should have.

Adaptive audio control

Unlike an ordinary line amp, the gain of a loudness control processor is not constant; it varies with time depending on the specific control algorithm of the loudness processor and the changing loudness and amplitude of the input signal. These variations in the gain, which represent the real control process, should take place without any bothersome side effects to the audio signal itself, effects such as pumping, signal distortion, sound coloration or noise modulation. In other words, they should be inaudible.

The setting of the attack time parameter of a loudness control element affects how the unit will react to rapid amplitude changes in the audio signal. A long attack time can lead to overshoots (and consequent distortion) because the system is not fast enough to reduce the gain. A short attack time minimizes the chance of overshoots, but the more rapid gain changes in such cases have audible side effects such as clicks and other modulation artifacts.

Single-band and multiband designs are being used to develop hardware and software control units and, depending on the architecture, both designs are capable of doing a perfectly good job. One advantage offered by a single-band design is that it doesn't touch the sound as there are no filters involved.

The question is: How can one guarantee the best possible performance if no one has the opportunity to realign the circuits while they are being used? In the absence of a human, what is needed is a system that is adaptive and performs like a human. Such an algorithm needs to change its behavior based on the resources available, and for this you need a multiloop principle.

I suggest the use of a multiloop approach because we believe this is the best solution.

The various loops each work over the entire frequency spectrum (wired band). They work in parallel, each with a different set of attack and release parameters. Each loop develops a control signal, which is then summed with the controls from the other loops to produce a single gain control signal applied to one final gain control element.

The way of summing the individual control signals requires unique technology. The digital implementation of the multiloop design permits a short delay time to be introduced into the audio signal path. This allows the gain changing elements to "look ahead" and determine the correction needed before applying it to the delayed signal just in time to control even the fastest transients. This is particularly important for the limiter, which provides a precisely leveled output signal absolutely free of overshoots (clipping). (See Figure 1 on page 70.)

The proprietary algorithms used in the multiloop design also allow the automatic adjustment of the attack and release times according to the evolution of the input signal over time. This is called adaptive loudness control. By analyzing the incoming audio, the system can set relatively long attack times during steady-state signal conditions but short attack times when there are impulsive transients.

Conclusion

Despite the views of some critics, adaptive multiloop technology does provide a way to integrate audio control systems into fully automated broadcasting systems. It is obvious that a file-based control philosophy will never cover 100 percent of all broadcast transmission content — not least because television still includes live content, and even when tapeless technologies are employed, this content is still live. In a typical broadcasting facility, we will continue to have a mix of file-based and stream-based content control in order to guarantee proper output signal condition anytime and independent on the source.

Regarding the tasks of the engineers in an automated multichannel control room environment, any kind of signal processing must be a set-and-forget solution because no one has the time or the resources to use a human for this task, especially not for audio. In this scenario, the only way to create "automated" and "quality" real-time online processing algorithms is to use adaptive technologies. Provided they meet the requirements of all the current different standards and recommendations, these adaptive technologies will always guarantee the best audio control performance in a fully automated system.

Regarding the tasks of the engineers in an automated multichannel control room environment, any kind of signal processing must be a set-and-forget solution because no one has the time or the resources to use a human for this task, especially not for audio.

Peter Poers is managing director of Jünger Audio.

Intercoms

Today's intercom systems are high-tech.

BY JOHN LUFF

What could be more fundamental than communications in an industry where individuals collaborate to create compelling content? But don't make the assumption that intercom systems are not high-tech these days. We have come a long way since the era of carbon microphones and Western Electric headsets, and what is fundamental is that function has not changed, only the sophistication with which we attack the technology. Of course, this is largely due to tools that simply were not possible in the early days of television production. It is, however, remarkable that we still plug headsets into the backs of cameras and use gooseneck microphones on rack-mounted communications panels as the user interface for communications just like we did more than a half-century ago.

Two drivers are largely responsible for the wholesale change in communications underlying the user interface mentioned above.

First, in the latter part of the 20th century, around 1970, a method was developed of using a duplex two-wire (actually three-wire) system for party-line communications that carried power along with the audio. Party-line communications was nothing new, and in fact was the basis of telephone communications for about 100 years before RTS and Clear-Com improved on the approach.

Around the same time, flexible systems were developed that used active and passive audio hybrids to allow effective communications on multiple party lines at the same time without joining all of them together. This development of the "matrix intercom system" may have had even more far reaching impact. Every control room today is built with private matrix communications channels for different parts of the crew to communicate selectively instead of using one large party line system. I am sure this saves many productions from devolving into chaos. It also allows problem solving to move on in private when it might otherwise have to wait for a quiet period. Best of all, matrix systems allow things to be centralized that otherwise would require multiple overlapping and redundant systems.

At the Eurovision 2011 song contest, organizers EBU and NDR used multiple Riedel systems, including Artist digital matrix intercoms, Performer party-line intercoms, and various digital and analog radios.

IFB

Take, for instance, IFB. IFB can be built without a matrix system and was done that way from day one in permanent studios. But by connecting talent earpieces through a matrix system, the production staff needs to have only one microphone for communications. Using software-definable systems for switching the audio signals allows them to be routed flexibly for monitoring, with interrupt or noninterrupt outputs easily created. It is the routing, né matrix, aspect of systems like this that make highly flexible and powerful systems possible. Many modern systems also adopt digital displays for control panels, which allows signals to be named uniquely for each production — for instance, labeling the IFB for the anchor with his name instead of tape labels or fixed electronic labels.

Sports and live production

Nowhere is this more important than in major sports production. Indeed, much of the advancement of matrix systems, and in particular the highly flexible control and programming systems that have evolved, are a direct result of systems designed to support the needs of major sports and location entertainment show production.

I had the fortune to work in that segment of the industry and saw the rise of an "A2" position whose responsibility was just communications. On large productions it is a full-time, high-pressure position. Over time, it has grown to be a complex part of a production as well. Major productions, like the Academy Awards, spend months planning every part of the installation of communications, which includes multiple production units for domestic and international program distribution, as well as facilities

Ukrainian broadcaster TV Company Ukraine uses two Riedel Artist 128 digital matrix intercom systems at its Kiev studio.

to support single-camera standups (with IFB) for affiliates around the world. Every department in a large production needs private, though overlapping, communications, including lighting, audio, producers, director's "camera" circuits, engineering channels and more.

Wide area systems

But there are applications today that are far more technically interesting and that make use of more complex digital transport mechanisms. Large, distributed broadcast organizations have complex communications needs. I was first introduced to wide area systems like this when I started a 30-year association with Eurovision in the late '70s. Even then, Eurovision (the European Broadcasting Union's "operations" arm) maintained a four-wire communications system that had drops in every control room in Europe that originated or received shared signals. At that time, it was analog and hard to keep levels balanced. Today, Eurovision, and any large operation, particularly those doing news production from many interconnected sites, have a crying need to connect local matrix intercom systems into a vast WAN intercom system that facilitates complex live breaking news production.

Solving this thorny problem today is made considerably easier by the ability to connect matrices together digitally to form a huge virtual matrix. Audio is often connected as VOIP trunks, and IP-based control systems allow configuration of complex productions without involving personnel in each site. It is possible to pass mnemonics for labeling panels, configurations of who can speak with whom, even mix minus selections for IFB to be flexibly built with only modest preplanning. Web-based programming permits graphical representations of the topology as well as simple Web service calls for some operations. Leveraging IT technology for voice communications is not unique to media operations, but the immediacy of our industry puts particular stress on systems for quality of network service and guarantees of service availability. The PSTN can always be a backup, but relying on voice circuits would severely limit the flexibility we have come to expect from modern communications.

I find it interesting that in an era of smart phones and wireless everything that we have not more fully adapted communications technology to wireless approaches. That is not to say that wireless intercom has not been available for a long time, for it has. But why aren't we using Bluetooth headsets instead of tying ourselves with copper leashes? Perhaps we give camera people a touch screen to adjust volume and mute their mics, or flash the name of someone calling a camera on a smart OLED tally indicator. Why not allow forwarding a call to an absent operator to their smart phone over Wi-Fi within the facility? A little creativity might make the sophistication of today's communications technology look simplistic in the future. **BE**

John Luff is a television technology consultant.

? Send questions and comments to:
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T-VIPS nSure

Transport stream monitoring range is designed to ensure the uninterrupted delivery of high-quality video content and simplify the day-to-day operations by increasingly complex video broadcast infrastructures; range is used by satellite, IPTV and terrestrial network operators for efficient monitoring and switching of transport streams; enables flexible interfacing with support for IP/Ethernet, SONET/SDH, DVB-ASI or SMPTE-310 networks; provides simultaneous, in-depth analysis of transport streams, services, PIDs, and DVB PSI/SI and ATSC PSIP tables.

www.t-vips.com

LYNX Technik yellobrik CHD 1802

Signal converter converts HDMI video into broadcast-quality SDI video, supporting SD, HD and 3G-SDI; supports 3-D video formats; includes two electrical SDI outputs, as well as an optional fiber-optic output; audio in the HDMI signal is embedded into the SDI output, and the two external analog audio inputs can be embedded into any AES channel; various fiber options are available, which include 6mi and 24mi fiber transmitters.

www.lynx-technik.com

Harmonic ProMedia

Multiscreen production suite performs a broad range of functions, including transcoding, packaging and origination to enable high-quality video creation and delivery of live streaming, live-to-VOD, and VOD services to TVs, PCs, tablets, smart phones, and other IP-connected devices; includes ProMedia Live real-time video processing and transcoding system, ProMedia Carbon file-based transcoder, ProMedia Package carrier-grade adaptive streaming preparation system, and ProMedia Origin HTTP and RTMP streaming video server.

www.harmonicinc.com

Miranda Kaleido-XQUAD

Quad-split multiviewer for truck and studio monitoring; can be logically linked to more quad-split processors for large monitor wall quad views and operated seamlessly as one system; integrates with NVISION and third-party routers, with source control and centralized mnemonic database access; offers tally support via direct serial interface; uses a 1RU, half 19in frame chassis; available with four 3Gb/s/HD inputs and a single multiviewer output or eight 3Gb/s/HD inputs and two multiviewer outputs.

www.miranda.com

Telecast Fiber Systems Thor series

Thor-CBL-HDMI uses multimode fibers for RGB and high-speed digital graphic signals, and copper wires for low-frequency HDCP control signals; Thor-CBL-DisplayPort (DP) provides an alternative to digital display interfaces such as HDMI and DVI; both systems harness optical technology for transmissions that stretch performance beyond the limitations of copper wire by spanning greater distances, providing data security and negligible RFI/EM, and eliminating costly analog distribution systems.

www.telecast-fiber.com

Harris NEXIO Farad

Online storage system is ideal for large-scale ingest, editing and playout for production, sports, news and live-event applications; designed to deliver the highest levels of bandwidth and storage capacity to support even the most demanding multiplatform media delivery workflows; capacity scales up to more than 512TB of usable storage and/or 32,000Mb/s.

www.broadcast.harris.com

Bridge Technologies Traffic Analysis

Option provides advanced monitoring and analysis of IP protocols and services across the complete range of VideoBRIDGE probes; enables operators to combine detailed monitoring and analytics of broadcast/IP streams and all OTT services under a single unified VideoBRIDGE environment with consistent display and reporting features.

www.bridgetech.tv

Ikan — VL5

5in HDMI on-camera monitor with resolution of 800 x 480; includes peaking, underscan, switchable aspect ratio and aspect ratio guides, blue gun, RGB adjustability, native Canon EOS 5D live view output, and 480i/p, 576i/p, 720p, 1080i/p signal input; comes with 12VDC adapter, as well as a choice of Sony, Panasonic or Canon DV battery adapter plate; includes a sun visor for clarity in the field.

www.ikancorp.com

Panasonic — AG-AC130

Camcorder features three 1/3in, 2.2-megapixel CMOS imagers to capture native 1920 x 1080 HD images; includes a new 22X HD zoom lens; features two SD memory card slots for relay or simultaneous recording, compatibility with Panasonic's new SDHC Class 10 UHS-1 memory cards, a color viewfinder and LCD display, and all-digital recording; weighs just over 5lb; power consumption is under 12W.

www.panasonic-broadcast.com

ViewCast — Osprey 260e/460e

260e video capture card provides one channel selectable between composite, component or Y/C video and balanced or unbalanced audio in a low-profile form factor; ideal for compact system enclosures; 460e video capture card can simultaneously capture four independent channels of analog video and unbalanced stereo audio signals and process them independently, minimizing internal PC space requirements; both are designed to work with industry-standard media encoding applications.

www.viewcast.com

Nevion — FCS250-PLUS

Compact, cost-effective monitoring system is designed to ensure that services achieve continuous target quality standards; features a built-in Web interface for local monitoring and configuration with SNMP and XML for remote access; monitors and assesses the integrity of up to four DVB-ASI or HD/SD-SDI signals or 256 video-over-IP streams; features automated SLA compliance monitoring and monitoring of DVB-ASI signals according to TR 101290, SD-SDI signals according toSMPTE-259, and HD-SDI signals according to SMPTE-292 standards.

www.nevion.com

Ensemble Designs — BrightEye Mitto 3G Fiber

Scan converter has DVI, VGA or HDMI inputs; provides SD, HD or 3GB/s SDI video outputs, as well as an optical SDI output; complete control of audio levels, channel mixing and audio delay is provided for the analog, AES and embedded audio inputs; can also be used with Apple's iPad and iPhone, enabling video from these devices to be converted to SDI.

www.ensembledesigns.com

Pilat Media — BMS Operational Cockpit

Management software ties together workflow processes and business information to monitor and display the status of a large number of key business processes; at-a-glance visuals and graphic displays quickly and efficiently highlight exceptions and alert business executives to items needing attention, such as a process that is likely to slip and miss its due date, available inventory, accounts receivables, missing content rights and permissions.

www.pilatmedia.com

Front Porch Digital DIVApublish mpx

Online publishing tool leverages content through an integrated workflow; delivers media in a simple, streamlined fashion; distributes content to myriad locations and devices while maximizing marketing results through integration with partners in adware, syndication, analytics and content delivery; is fully integrated with DIVArchive and DIVAdirector.

www.fpdigital.com

Utah Scientific FLEX I/O

UTAH-400 routers and UTAH-100/XFD fiber distribution frames are now available with the FLEX I/O signal module; users can now easily configure systems to implement coax and fiber I/Os on a port-by-port basis; provides enhanced ability to customize use of signals according to an application's specific needs; for the UTAH-400 router, option consists of input/output cards that carry eight or 12 signals and updated rear-panel assemblies to provide converter block access.

www.utahscientific.com

Egripment ProTraveller

Jib/crane system is designed for DSLR and HDV cameras with a maximum weight of 22lb; features smooth, high-quality crane movement in combination with a technical remote head; can be mounted on any 100mm bowl type of connection; weighs 82lb.

www.egripment.com

NewTek LightWave 10.1

Visual effects software features a new off-axis stereoscopic camera rig option, improved Viewport Preview Renderer, advanced import and export capabilities (including MDD and integrated Autodesk Geometry Cache support), and a new Skin Shader node; artists count on the features of LightWave to deliver stunning results; provides access to all major stereo camera rigs and the ability to dynamically correct for toe-in distortion in the animation pipeline.

www.newtek.com

Mosart Medialab Newscast Automation 3.0

Automation system features redesigned graphical interface with full user configurability; story elements such as cameras, clips and graphics can easily be assigned to buttons for simple and quick access; a range of interface overviews also can be assigned to buttons; story scripts from the newsroom computer system are shown directly in the GUI, and operators can store clips and graphical elements in a pool ready for execution on request; content repurposing/reuse and MAM integration are enhanced with the delivery of news as-run information to continuity control systems such as Snell's Morpheus.

www.mosart.no

Snell **Centra Vite**

Router control configuration tool provides fast setup and immediate router use; web control panel ensures instant operation; external snapshot application saves live status of all or a selection of routes; offers straightforward upgrade path to Centra system-wide control and monitoring system without compromising existing user interfaces and fundamental operation.

www.snellgroup.com

Volicon **Observer Mobile**

Web-based interface supports H.264 on Apple iOS using Safari Web browser along with native support for iPad, iPhone and other mobile devices; provides added convenience of accessing live Observer streaming along with back navigation of previously recorded content from both local and remote locations; can play, pause, search and create logged content on demand using smart devices.

www.volicon.com

Wowza — Media Server 3

Streaming solution is a single extensible foundation for "any screen done right" media delivery of the best possible viewing experience where and when it's wanted; designed to make simple and cost-effective formerly hard-to-implement features such as adaptive bit-rate streaming, time-shifted playback and integrated rights management.

www.wowzamedia.com

TSL — AVM-T-MIX

Rack-mount touchscreen-controlled audio mixer/monitor; touch-mix system delivers a unique combination of audio monitoring and channel mixing capabilities designed to simply workflow and operations; users can see channel activity, recognize the source and listen to resulting audio from individual, grouped or mixed channels; surround sound can be down-mixed instantly with the touch of a button.

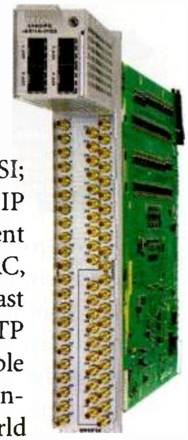
www.tsl.co.uk

Crystal Vision — Up-Down-S 3G

Synchronizing up/down/crossconverter allows flexbile, up/down/crossconversion between 3Gb/s, HD and SD sources; provides an output picture quality that broadcasters standardize on; includes special features to allow studios to easily operate in HD and SD at the same time — with its ability to perform two different conversions simultaneously and give out co-timed dual outputs that remain unchanged in format, even if the input changes.

www.crystalvision.tv

Evertz — 3080IPG-ASI-IPGE



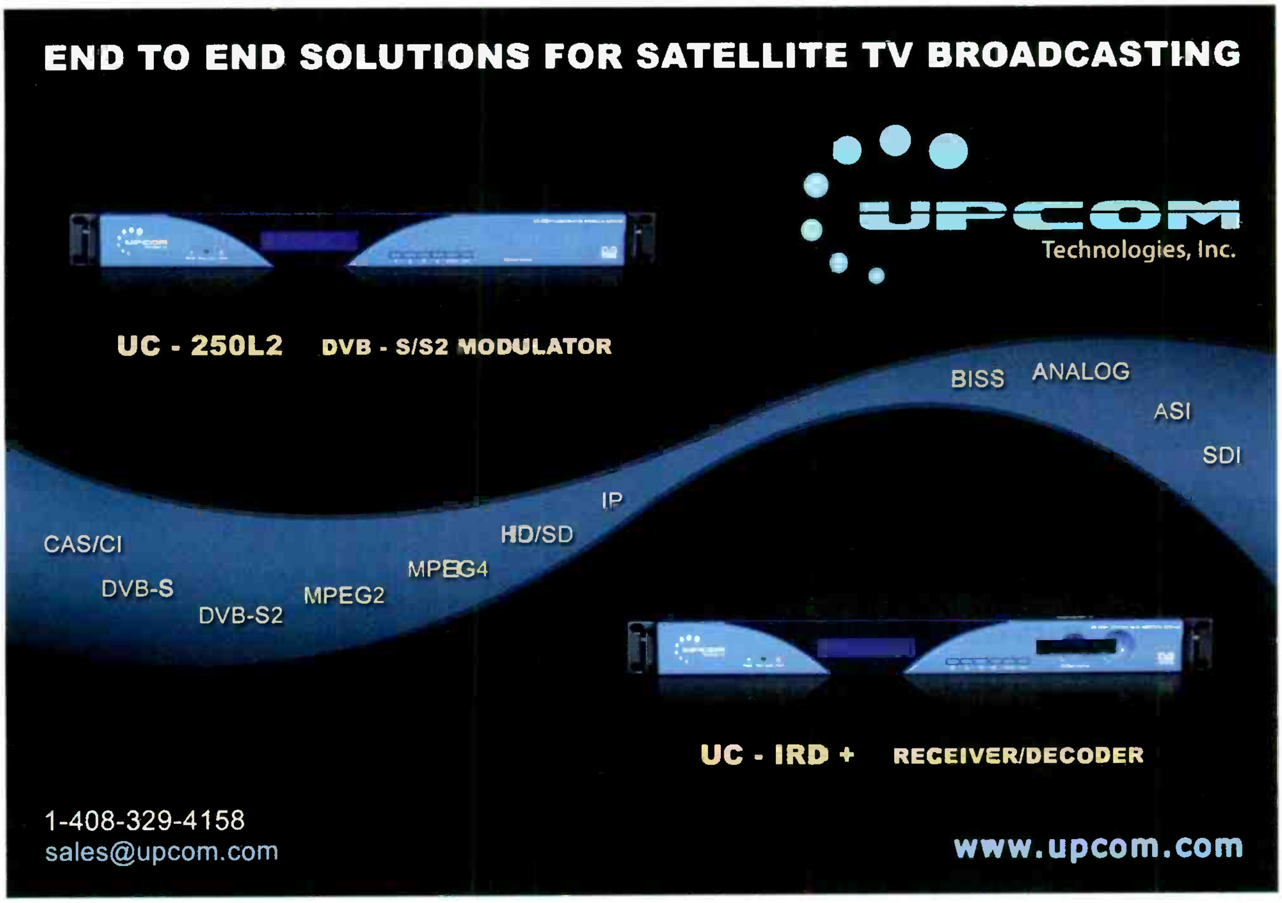
Bi-directional high-density IP gateway encapsulator/decapsulator concurrently transforms ASI to IP, IP to ASI; regenerates IP from IP input with different settings, such as MAC, IP Unicast/Multicast addressing, add RTP and ASI FEC; is capable of ridging traditional compressed world (ASI) with the emerging video over IP networks; can send and receive 32 ASI signals from a unicast or multicast IP stream.

www.evertz.com

Sonnet Technologies Echo Express PCIe 2.0

Expansion chassis enables users to plug in high-performance PCI Express 2.0 cards to any computer with a Thunderbolt port; available in two sizes; standard supports one half-length, double-width, x16 (x4 mode), PCIe 2.0 card, while XL model supports one full-length card; both models have integrated universal power supplies and cooling fans; two Thunderbolt ports support daisy chaining of up to six devices to a single port on host computer.

www.sonnettech.com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Workflow's new world

Some are braving it now; it's inevitable for the rest.

BY ANTHONY R. GARGANO

The world of high tech, it seems, is always rife with buzz words, and certainly broadcast is no exception. One of the buzz words we hear today constantly being bandied about is "workflow." Companies talk about providing workflow solutions, increasing workflow efficiencies, end-to-end workflow, workflow ad nauseum.

So, what is this new technospeak, workflow?

Don't be intimidated by the fancy term. Underneath it all it is quite simply the process that you undertake, from beginning to end, to accomplish what you need to get done. Break that process down into its individual steps, and now you have a new technospeak term: workflow elements. One such workflow element is transcoding or format conversion.

Today's broadcast facilities find themselves dealing with an increasing number of video formats. These formats run the gamut from network distributed satellite content to user-generated cell-phone video (and in quality levels that can range from HD to Skype), not to mention everything in between. In this multiformat environment, a common workflow element is the transcoding of that content to a different format. Format conversion may be required on the input side for incoming material in order to get it on-air. The days of station output being solely confined to "On Air" are long past. Today's broadcasters find themselves supporting the requirements of multiple output platforms and, as a result, may be required to recode that same content to make it available for Internet streaming, for use on the station's website or for transmission via the station's Mobile DTV service.

For this particular workflow element, there are a number of offerings with various levels of capability. Recently, I had the opportunity to look at several of what might be termed enterprise solutions for transcoding and format conversion. The full extent of options may perhaps seem like overkill for a small local broadcaster today, but given the ever proliferating applications for video (more and more of those are in a file-based infrastructure, thankfully), it might be just what you need tomorrow. And, at the broadcast network level or for station groups that use centralized aggregation and distribution points for content, it may not be overkill at all.

> **Should your routine not yet involve formal workflow products, be prepared because it will someday soon.**

Three of the systems I looked at were from RadiantGrid, Rhozet and Telestream. Some key parameters of interest in format conversion workflow are speed, quality and the number of formats handled. The best way to evaluate any product is by demonstration in your own facility, using the content you have to deal with on a daily basis and evaluating the results in the practical environment defined by your own needs and requirements.

Speed, for example, is a function not only of the workflow product processing algorithms, but also it is regulated by the computer hardware installed in a facility. Output quality is going to be largely driven by the quality level of codecs used in the processing software but can be influenced by the amount of time that is devoted to processing the content. Need a quick turnaround? Shorten the cycle, and normally one would need to be prepared to pay the cost in quality level. Perhaps not though, if you are using RadiantGrid's technology. Its grid processing technique can slice the content into bite-size chunks and essentially parallel process the transcoding effort across as many computer platforms as can be made available on the grid.

Then, there is the horsepower race of number of formats handled, which is strictly defined by the manufacturer. Give or take a few, Telestream can handle about 120 different formats. Rhozet claims upwards of 150.

RadiantGrid offers the option of containers, codecs, audio formats and ancillary data models. Customers can then mix and match in any way they see fit to satisfy the needs of the requirement. In essence then, the actual number could be hundreds of format combinations.

Should your daily routine not yet involve formal workflow products and systems, be prepared because they will someday soon be an integral part of your workday life. File-based workflow is already driving some automation systems from ingest to quality control to transfer to the play-out server.

If you would like to delve into a more in-depth treatment of workflow and workflow processes be sure to check out *Broadcast Engineering's* webcast series at: broadcastengineering.com/webcast. There is a range of excellent tutorials offered as live webcasts or on an on demand basis. BE

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