Broadcast Engineering

SEPTEMBER 2013
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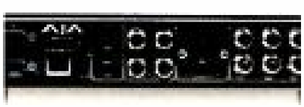

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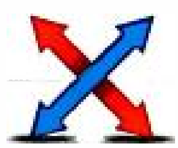

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VIDEO SYSTEMS

VOLUME 55 | NUMBER 9 | SEPTEMBER 2013

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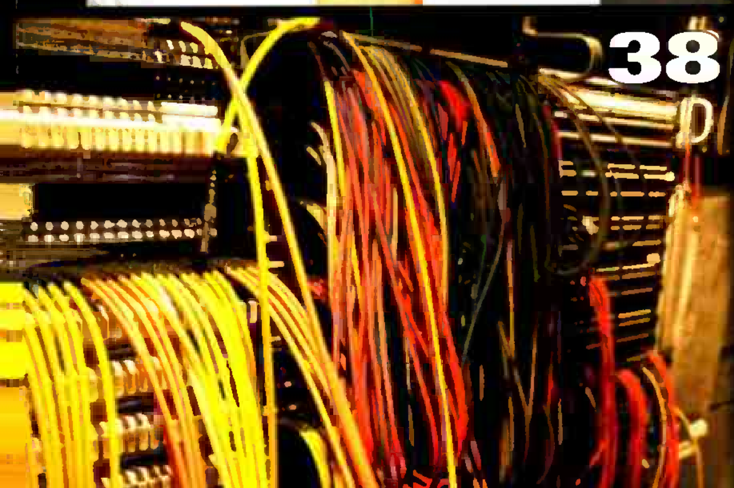
ON THE COVER:

In preparation for the upcoming AES Convention in October, *Broadcast Engineering* brings you several in-depth tutorials offering insights into broadcast audio.



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DIGITAL EDITION EXCLUSIVE FEATURES!

Check out the additional features and content found only in our digital edition:

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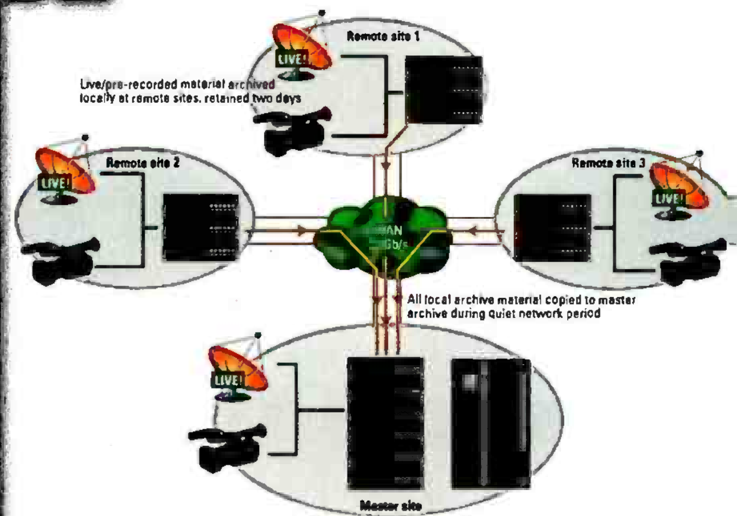

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The cloud isn't cheap

Cloud storage and software is huge in the IT space. And, as NAB and IBC illustrated, broadcast vendors are increasingly offering similar versions for media clients. Even so, engineers have yet to fully embrace the idea of handing off their content to a server in an unknown location, hosted by an IT company that claims "no downtime and perfect security," yet cannot spell "broadcast."

Let's examine some of the benefits and potential drawbacks of a cloud solution.

First of all, cloud storage is cheaper, right? Not according to Sameer Manas at CloudPlugged.com. In his article "Cloud Storage is cheap? Think again, Cloud Storage vs. Local Storage compared," he makes the case that when

compared to local DAS/SAN/NAS solutions, the CAPX cost of storage is less than the OPEX of cloud when all operating feature costs are included. A 1TB, 10,000rpm drive now costs less than $100. And costs will continue to fall as SSD usage increases, helping HD prices further decrease.

A second point is that cloud solutions require an Internet connection. He claims that access times will be 2X slower, even with an enterprise-grade solution because of the Internet. He suggests that any near-real-time application will require a Google Fiber-like connection.

A third point, often echoed by others, is that cloud-stored files must always be small in size. One cloud storage vendor limits file sizes to 5Gb, although it offers a work-around where larger files must first be segmented.

The often-made claim that cloud is "green" may be misleading. Google alone uses about 2 billion KWH/yr. Today, server farms account for 14 percent of all carbon emissions and consume 2 percent of all the power used in the U.S. By 2020, data centers will consume more than 25 percent of all the power generated in the U.S. Perhaps the only green here is what is being paid to the power companies.

Another issue: All clouds aren't alike. It's like the old days of selecting a tape format. The two most common choices were BetaSP or P2, Sony or Panasonic. Having made the decision, changing to the other platform became expensive and time-consuming. Changing cloud vendors will represent similar challenges.

Current cloud storage costs about $0.10/Tb/month, but expect prices to drop. While that's good, it means that signing a long-term contract may not be in your best interest. Some experts predict that once a client uploads content to a particular cloud solution, moving it becomes so problematic that even a 50 percent drop in costs won't offset the labor and hassle of changing cloud solution vendors. If you select vendor X, and your costs are higher than expected, or you are offered a "better deal" by vendor Y, can you afford the costs of time, manpower and training to change? Reminds me of the days of standards — when we had lots of them.

Finally, the cost of actual storage space may be the smallest part of your monthly bill. Some cloud vendors charge for every file upload and download. If you need a cloud archive, those costs may be insignificant. However, such access costs could be a huge factor if you're looking at a collaborative newsroom system where hundreds of files are transferred daily.

Choosing a cloud solution may require more research that first thought. And right now, there is considerable controversy over technology, security and cost. Perhaps the adage of "baby steps" applies. BE

Brad Dick
EDITORIAL DIRECTOR
Send comments to: editor@broadcastengineering.com

The best broadcast quality mini converters now in both regular and heavy duty models!

The world's most popular converters are now available in two families, for the studio or heavy duty for live outside broadcast! The new heavy duty models are machined from solid aluminum so they look beautiful and are super tough! There are 14 models including HDMI, analog, optical fiber, audio embedding/de-embedding and up, down, cross conversion. Mini Converters are even available as OpenGear cards for when you need a rack mount solution.

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Hybrid routing

The approach can save space, power and installation costs.

BY CHUCK MEYER

Hybrid routing describes routers providing integrated, simultaneous switching of audio and video signals. Given the widespread adoption of embedded audio equipment, the vast majority of broadcast facilities have been designed and installed with minimal, if any, audio routing capacity.

Hybrid routers switch audio and video signals together, with nearly perfect time coherence — something most easily done in a single frame. Any contemporary router control system provides breakaway control of mono audio signals, stereo AES-3 signals, Dolby E and Dolby Digital signals. Using the hybrid approach, these control systems should become easier to configure and use.

The problem

Requirements to produce theater-quality surround sound have grown simultaneously with consumer adoption of HDTV programming. Providing this capability in video production facilities involves adding audio-specific mixing equipment, production switchers or even recording consoles. (See Figure 1.) It also requires monitoring equipment and listening areas to evaluate surround-sound quality, and the ability to switch and control mono audio channels, discrete AES-3 signals, AES-3 signals with non-PCM payload and possibly even MADI signals used for bulk audio transport. (See Figure 2.) This is a significant expenditure of time, capital and equipment room space if audio is not embedded.

Modular equipment for embedding and de-embedding significantly reduces audio signal cabling but does not eliminate separate AES. This additional equipment also adds cost and consumes rack space.

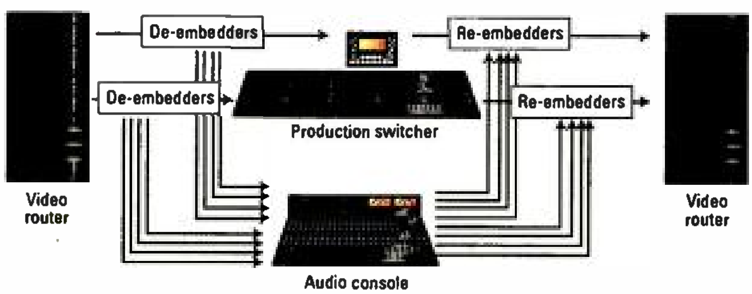

Figure 1. With the move to HDTV and surround sound, video production facilities often require production switchers, audio-specific mixers or recording consoles.

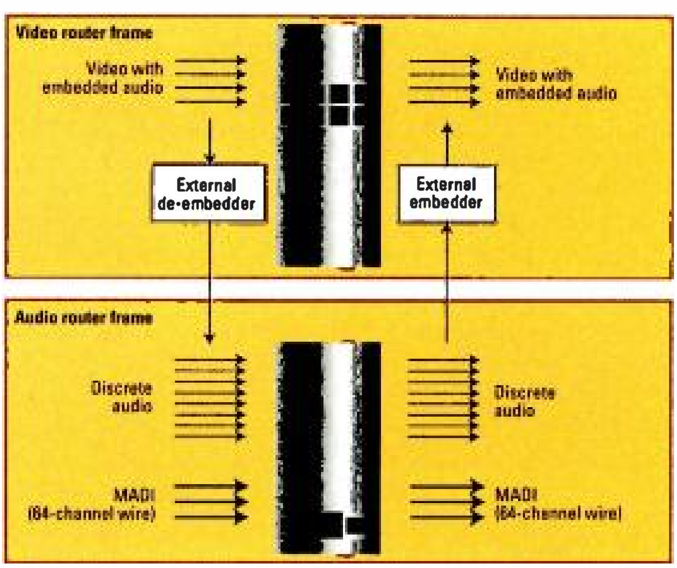

Figure 2. Routing de-embedded discrete audio or bulk audio transport via MADI can take a significant amount of capital and rack space.

Exact timing is required

Audio processing requires exact phase alignment of digital audio samples at 20.8µsec boundaries for 48kHz audio. Put differently, one sample of offset between two nominal 1kHz tones is equivalent to 7.5 degrees of phase offset error. HD-SDI embedded signals carry up to 16 mono channels of audio. MADI can carry 64. A final surround-sound program mix of six channels feeds the encoder. Often, 24 channels of audio, or more, are mixed down to the final six. Field recording for stage events requires audio consoles with multiple MADI I/O streams. Compared with traditional television sound, audio channel count has grown nearly tenfold and must all be perfectly aligned.

Ethernet is not perfect

Live production of audio and video signals requires near perfect alignment of audio with video and very low latency. Audio/video timing should be held to less than one video line of slip. Latency needs to be

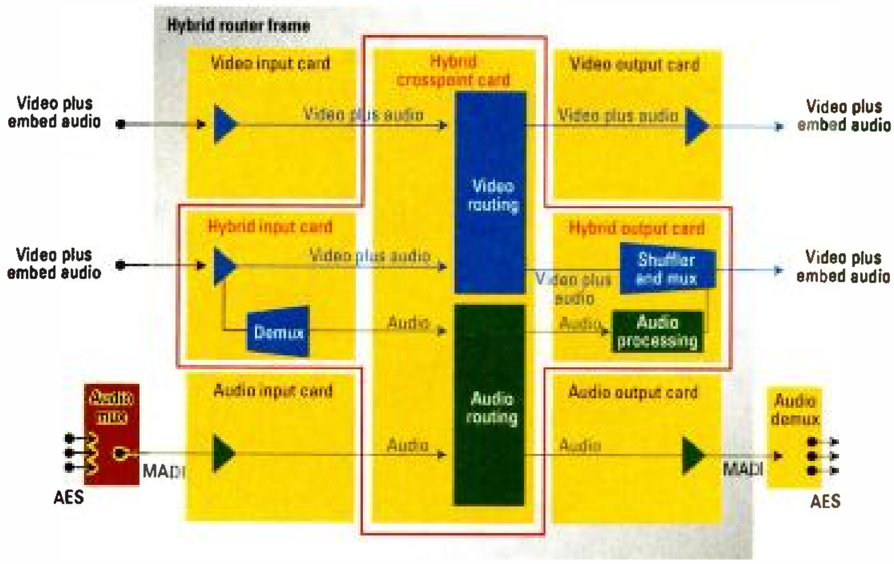

Figure 3. A hybrid router differs from a traditional router beause it integrates signal switching. Crosspoint cards switch audio and video while maintaining precise timing.

much, much lower than 1ms so that multipass processing does not generate lip-sync errors.

Hybrid routers typically range from 200 to 400 inputs by 500 to 1000 outputs. While Ethernet AVB approaches the timing metrics discussed, it presently does not scale to the sizes required. SMPTE 2022 has been recently expanded to include high-bit-rate video mapping into IP, but it fits better with contribution where FEC is necessary to correct errors from a noisy channel, rather than real-time production.

Baseband routing is a great solution

The perfect switch fabric is synchronous and deterministic for both audio and video. It preserves the installed, embedded infrastructure of existing facilities and trucks while accommodating AES-3, MADI and analog audio signals with minimal additional equipment. It is easily controlled by an intuitive router control system using traditional control panels supporting audio breakaway, which is exactly what baseband routers do.

How hybrids work

Hybrid router architecture follows the same general approach as traditional routers. What's different is integrated signal switching. For example, consider a router frame with input, output and crosspoint cards. (See Figure 3.)

Input cards receive embedded SDI video. After cable equalization, signals are distributed to "on-card" de-embedders. They are also fed to the crosspoint. The de-embedders rapidly extract audio signals. Video inputs are assumed to be mutually timed to less than plus or minus one-half a line relative to each other and house clock. Each video input requires its own de-embedder. All de-embedded audio is then assembled and organized into a time-division multiplex (TDM) signal. For example, eight HD-SDI signals, each with 16 audio channels, yield one TDM stream with 128 mono audio channels. This audio TDM stream is then fed to its corresponding audio switch on the crosspoint card.

The result is that one crosspoint card now switches both the video and audio signals. Video is switched traditionally with a crossbar chip. Audio signals are switched in the time domain using shared memory architecture, typically implemented in an FPGA. It is critical that audio delay is minimized in this switch because video signals and audio TDM streams both feed the corresponding output card.

The switch card provides complete configuration flexibility. It receives either SDI or TDM audio and ensures they are routed to the correct switch chips based on signal type. Router input cards may be populated in any order or combination without respect to card type: hybrid, standard video or MADI. A crosspoint card with both video and audio switching on one module is easily and efficiently protected with an n+1 crosspoint protection scheme. One crosspoint card with both switch fabrics in a common audio-video frame ensures all timing requirements while providing robust, reliable switching.

It's the output module that does the hard work. In addition to switched video signals, it receives TDM audio streams, which are demultiplexed and subsequently embedded into each video output. Each video signal is demultiplexed just enough to enable new audio signals to be switched into the correct time slot in compliance with SMPTE-standard embedding. This results in a combination of audio switching and embedding. The video signal is then scrambled, channel coded, reclocked and driven out of the router.

Ancillary data information for audio and video must be preserved. AFD and Dolby E burst preamble are two key examples.

Critical timing parameters

Minimizing delay is essential. With careful output embedder design, video delay can be just a few pixels. Conceptually, video data stays in the serial domain. Audio data is embedded bit by bit on the fly. Short video delays of much less than one-half a line greatly simplify plant timing, which is important for production. Audio delay is ideally as short, but the SMPTE standard dictates that audio samples always be embedded at the beginning of ancillary data space at the start of the video line. Audio samples are, therefore, delayed by one line.

The maximum audio delay results from different audio sample distributions across multiple lines of video. Differing buffer depths must

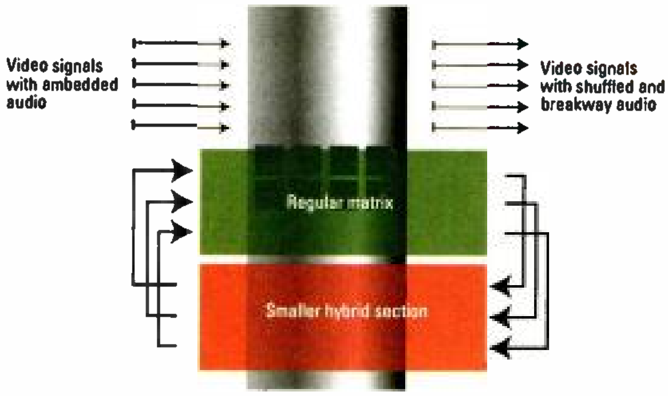

Figure 4. Dynamic hybrid pathfinding (DHP) uses a smaller router partition for hybrid routing, and takes the place of discrete embedders and de-embedders.

be managed for each de-embedder and embedder in the router so that any mono audio signal de-embedded from any video signal may be embedded into a common video output. This results in a one-line minimum to two-line maximum delay. And, there is plus or minus one-half video line signal variance. The maximum delay is, therefore, a short three lines. Even multiple passes will not affect lip sync.

Production applications

Audio production benefits from hybrid routing. MADI is widely deployed and now supports 64 synchronous channels of 48kHz multiplexed into one 125Mb/s data stream. Today, production facilities lock all audio to 48kHz, and MADI provides signal timing coherence and minimal delay. For all these reasons, hybrid routers offer interoperability for embedded video, audio and MADI. The audio router switch is nonblocking, mono, full broadcast and covers every input from de-embedders, AES-3 and MADI to any output with audio capability.

Care must be taken to preserve multichannel phase coherence or audio image because embedded audio sample distribution will vary between video signals. Audio sample timing slips might be generated when switching audio from embedded inputs into embedded outputs. One audio sample time slip is a significant phase error, which degrades the surround sound image of program audio. Sixteen-channel audio embedders provide coherent image-accurate audio transport within one video signal. If more than 16 channels must be phase coherent, MADI is better.

A number of MADI products support sound stages, or remote microphone fields. These MADI signals are routed to an audio mixer, switched between MADI streams or switched into video as groups of 16 or eight channels. These switches are accurate. Phase coherence can be lost by switching audio from different videos into a common video signal. Fortunately, using multiple SDI video signals to transport image-accurate audio is not standard practice.

Dolby E is a common signal for production and needs to be handled correctly within a router. Ancillary data must be preserved, and it is essential that router switch points comply with SMPTE RP-168. Hybrid router internal audio delays as described are short enough to comply with Dolby E guard band specifications.

Hybrid ingest with DHP

Hybrid routing is popular for ingest because it affords complete flexibility to shuffle audio and actually route any de-embedded mono input to any other embedded mono output. Sixteen-channel audio alignment and assignment is, therefore, much simplified.

Dynamic hybrid pathfinding (DHP) provides affordable hybrid routing when only part of the router requires audio switching. Using this technique, one portion of the router contains hybrid inputs and outputs connected to sources that require frequent channel reassignment. Another smaller router partition is populated with more hybrid input and output modules, but these modules are fed by, and back into, the router. (See Figure 4.) This "pooled resource" topology is the same as if discrete de-embedders and embedders were configured in an external modular equipment frame. The balance of the router is filled with standard video and MADI input and output cards. This hybrid pooled resource is similar to tie lines between multiple routers and is managed with DHP.

TRADITIONAL ROUTER			
TYPICAL RACK SPACE	RACK SPACE	WEIGHT	POWER
Video router 576 x 576 - 38RU	38RU	265lbs	5900W
2 layers AES router each 512 x 512, full mono	33RU	275lbs	900W
100 embedders 100 de-embedders	30RU	200 lbs	3000W
TOTALS	101RU	740LBS	9800W

HYBRID ROUTER		
RACK SPACE	WEIGHT	POWER
38RU (62% less)	409lbs (45% less)	5700W (42% less)
38RU	409LBS	5700W

Figure 5. A hybrid router can save space, weight and power compared to a traditional router.

It's been shown that DHP can reduce router costs by 20 percent, partly by reducing the hybrid card count for the primary router. Additionally, the pathfinding hybrid cards are significantly less expensive than external equipment to achieve the same result. The resulting pooled resource provides mono audio routing between the Pool and Core routers. This is simply not possible with external embedders and de-embedders. That said, there is no free lunch. The Pool can be blocking, as is the case with tie lines, and the size of the pool is sized for expected use, which might occasionally be too small.

With DHP, the audio breakaway route is automatically executed between signals on standard input and output cards. The router control system finds an available output/input path that re-enters the pooled router hybrid cards and generates the additional takes for these cards. Four manual takes are now one.

Determining the requirements for this style of routing is straightforward. Decide on the number of inputs and outputs requiring hybrid capability and the additional number of de-embedding inputs and embedding outputs for peak load. Then add up the remaining standard SDI inputs and outputs. The input and output total determines the frame size and the number of modules needed for each format and processing type. Generally, DHP is most effective when the Pool is less than 20 percent of the router's capacity.

Summary

Hybrid routing is compelling technology for new router installations. It dramatically reduces the equipment required by eliminating external embedders, de-embedders and audio routers. It also reduces cable count, space requirements and installation costs. It provides accurate switching, ultra-low delay and high-quality audio. For those in existing facilities, a hybrid router approach can instantly free up rack and cable space. And I haven't even touched on power savings, which are realized in every case. (See Figure 5.) The sum total of hybrid routing is not just that video and audio routing can be achieved in a single unit, but that the resultant streamlining of operational activities saves time and money. BE

Chuck Meyer is CTO, Core Products, Miranda Technologies.

+ ADDITIONAL RESOURCES +

The following are available on the *Broadcast Engineering* website:

- IP for signal management
- Video processing
- Router fundamentals you should know

Metadata

Describing broadcast content is more complicated than it seems.

BY ALDO CUGNINI

Because digital transmission systems distribute data, one can say that metadata is "data about the data." However, as we are usually concerned in this column with broadcasting of program content, a better description would be "information about the content." The forms of defining, handling and relaying metadata are quite varied, and this month we'll examine a few, from ATSC to DVB and onto the Internet.

Mobile receivers

One thing metadata provides is program information to fixed and mobile receivers. The well-known Program and System Information Protocol (PSIP) has long been the prime carrier of metadata using the ATSC A/53 standard. PSIP carries several important data tables, including timekeeping and channel information, and the Event Information Table (EIT), which supplies titles and program guide data for each program event associated with a virtual channel. Each EIT is limited, however, to a period of three hours, so broadcasters must routinely update the EITs when new or more accurate information becomes available.

In the ATSC Mobile Standard (A/153), more extensive program metadata can be carried within various components: Signaling, Announcement and Non-real-time (NRT) file transfer. The Transport Signaling System is a signaling layer that uses the Fast Information Channel (FIC), in combination with the Service Signaling Channel (SSC), to deliver critical information. This information allows for rapid program acquisition by the receiver. Although they do not provide detailed program descriptions, the FIC and SSC provide enough structural information to allow video and audio decoding to initialize in the receiver. The information carried within the SSC is similar to the high-level PSIP information carried within ATSC A/53.

Announcement, which is optional, provides the framework for an Electronic Service Guide (ESG), using components from the Open Mobile Alliance (OMA) Broadcast Services Enabler Suite (OMA BCAST). An ESG is delivered as a file consisting of several XML sections, using File Delivery over Unidirectional Transport (FLUTE), a scheme that ensures quality of file transfer over the potentially lossy, one-way broadcast medium.

ESGs can carry information for upcoming as well as current programs, including start times and duration of events, channel icons, program titles and descriptions, genre, and ratings. Multiple ESGs can be carried simultaneously, and an aggregated ESG across providers could be downloaded via an out-of-band (interactive) channel, such as by 3G or Wi-Fi.

The concept of metadata can be extended hierarchically, using what is called an XML schema, a document that describes the structure of other XML documents. In that respect, an XML schema can be considered "data about the metadata." XML schema can be used to define extensions that add ATSC NRT-specific metadata; one example could be to define a grouping of files into content items within the FLUTE sessions used for fixed or mobile NRT content delivery. Standard methods for constructing an XML schema include the document content description (DCD).

Metadata requires an infrastructure through which it can be generated, processed and delivered to the transmission point. For this purpose, there is an ATSC standard (A76B) called Programming Metadata Communication Protocol (PMCP). PMCP is specifically defined to provide a consistent means to handle metadata in systems and equipment supporting the production and delivery of PSIP and Announcement tables. Target applications include traffic, PSIP generation and automation systems at broadcast centers and program listing services.

Metadata requires an infrastructure through which it can be generated, processed and delivered to the transmission point.

Using PMCP, advance program information as well as last-minute scheduling changes can be quickly delivered to the universe of digital receivers. Examples of elements within PMCP include the "ShowData" and "PsipEvent" elements, which can communicate metadata about a program, independently of its scheduled

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Live Production

PmcpMessage xmlns:xsi="http://www.w3.org/2001/XMLSchema-instance"
PsipEvent action="add" duration="PT30M"
EventId channelNumber="27-1"
InitialSchedule startTime="2013-09-15T10:00:00-05:00"
ShowData
Name lang="eng"
Transition to Digital
Description lang="eng"
Understanding digital technology
ParentalRating region="1"
Rating dimension="Entire audience" value="TV-G"
Audios
Ac3Audio audioId="1" lang="eng"
Ac3Audio audioId="2" lang="eng"
serviceType="visually_impaired"
Captions
Caption708 service="1" lang="eng"

Figure 1. PCMP uses XML to provide program metadata.

broadcast air time, affecting both current and future ESG information. Figure 1 shows one such example.

Audio also carries associated metadata, including several factors that can be controlled for each program, such as Bitstream Mode, Dialogue level (Dialnorm), Dynamic range control (DRC) and Downmixing. Bitstream Mode defines the arrangement of discrete or associated services, the most commonly used being: Complete Main (CM) supports from one to 5.1 channels of audio; Main M&E is similar to CM, but omits the dialogue channel, which can be carried separately as Associated Dialogue (D); descriptive audio and increased-intelligibility audio can be sent as Associated Visual Impaired (VI) and Associated Hearing Impaired (HI), respectively; and Associated Emergency (E) can be sent to override all other audio.

Dialogue level sets the average level of speech in the program audio at playback time, referenced to a known sound level. This parameter can now be used to help assure compliance with the CALM Act. Dynamic range control allows the user to optimize the dynamic range of the content, essentially setting it to a pre-calibrated compression curve. Downmixing allows for appropriate reproduction in the home environment, so that every user can enjoy a compatible experience regardless of the presence or absence of multiple speakers.

Room Type is another audio parameter, which describes the equalization used during the final production mixing session. The "Large room" parameter emulates a dubbing stage with the industry-standard X-curve equalization; the "Small room" has flat equalization. This parameter allows a home audio system to be set to the same equalization.

Although various audio metadata parameters are reserved for professional use, the ones mentioned here could all be provided to the consumer. Audio metadata in broadcast and production facilities has its own carriage interface, either by SMPTE RDD 06-2008 over an RS-485 serial connection or via an HD-SDI connection.

Service information over DVB provides extensible metadata. DVB similarly provides metadata for video, audio and program information, supported through various structures and XML schema for live, on-demand and file-based content; DVB-SI (Service Information) is codified in ETSI EN 300 468 and ETSI TR 101 211. DVB now includes a metadata profile defined by the TV-Anytime Forum (ETSI TS 102 323 and others), with XML schemas and profiles adapted for enhanced PVRs. A program guide for broadband (Internet) content is also in the works, together with associated metadata definitions.

Support for screen formats

Because the transition to digital broadcasting has required the support of new display formats, the adequate presentation and cross-compatibility of content authored in the 4:3 and 16:9 aspect ratios has created a need for supporting metadata. To fill this need, various organizations have developed the Active Format Description (AFD), a standard set of codes defining the aspect ratio and active picture characteristics of a video program.

Together with optional bar data (indicating the size of top, bottom, left and right bars), DTV receivers can be instructed to crop, letterbox, pillarbox, or pan and scan images for best viewing compatibility. ATSC, ETSI (for DVB) and SMPTE have all published similar but non-identical versions of AFD codes, in A/53, TS 101 154 and 2016-1-2007, respectively. BE

Aldo Cugnini is a consultant in the digital television industry.

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

ADDITIONAL RESOURCES

The following are available on the *Broadcast Engineering* website:

- Metadata
- Metadata and MXF, part 1
- Metadata and MXF, part 2

Software development

Sometimes we find ourselves doing jobs that are very different from what we signed up to do.

BY BRAD GILMER

The "accidental system administrator" is one of the most popular columns I have ever written. In that article, I talked about becoming an accidental system administrator — how over the years I had gradually become the go-to person for server and network issues in the company.

There is another trend I want to cover this month, and that is the fact that many media companies have developed small software development shops within their engineering departments. This is good and bad. The good part is that adding the ability to write code to support media operations has become a core capability for many of us. The bad part is, as with the accidental system administrator, it is a role we take on out of necessity, and many of us, myself included, have never really worked in a small software development shop where best practices were being used.

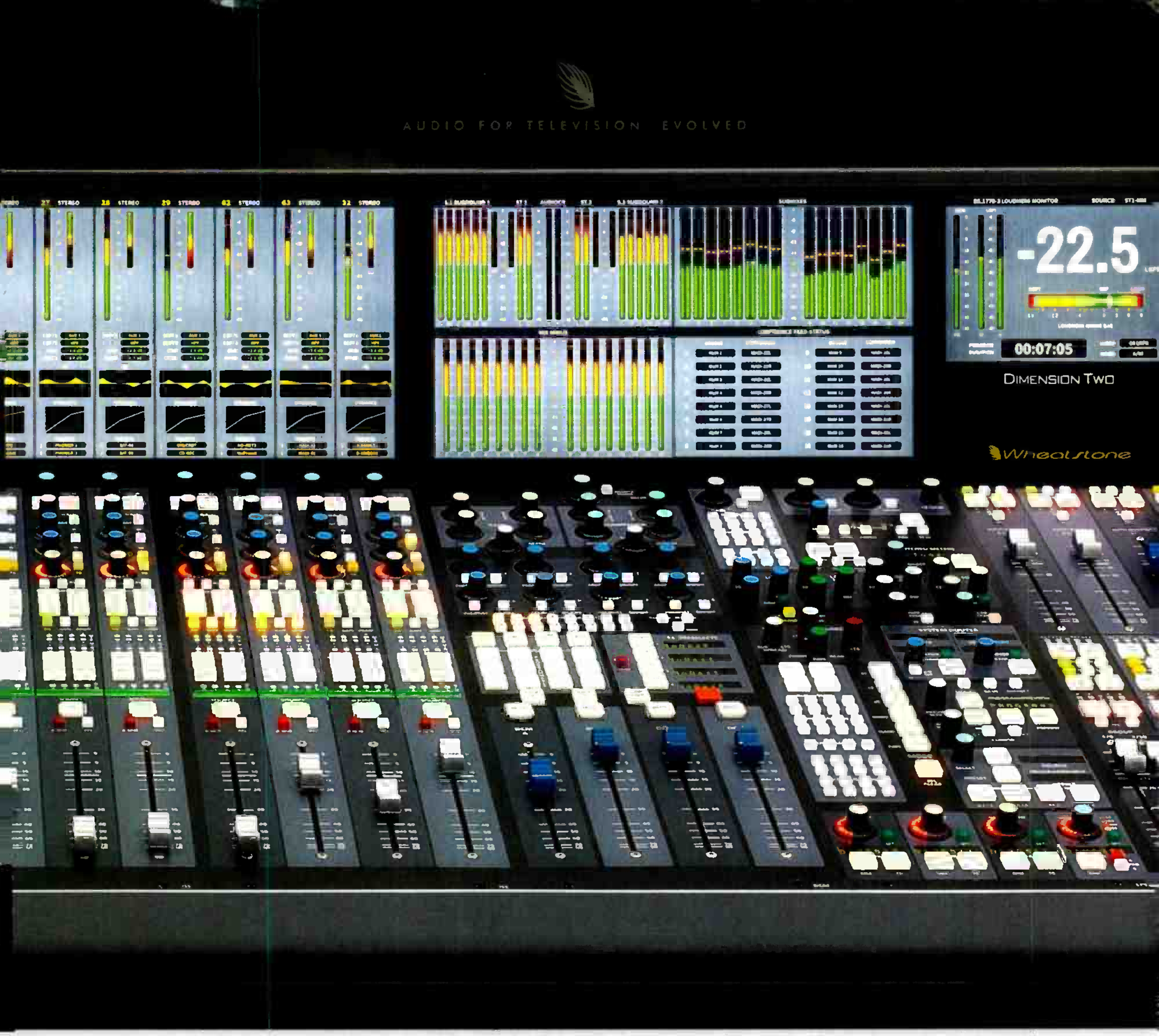
I thought it would be useful to look at small-shop best practices, so I asked three of the most talented software architects I know — Jim Trainor at Trainor Engineering, Dan Shockley at Turner Broadcasting System and Richard Cartright at Quantel — to give me their advice for those who find themselves in a small software shop. The sidebars below and on page 20 summarize what they said.

Agile

Agile software development is an extremely popular method for developing software. It establishes specific roles for people during development, makes it clear who is responsible for which decisions, enforces good communication within teams and helps mitigate the feeling that many of us have that we have to get it absolutely perfect the first time. Agile is a great way to get stalled projects moving, and it is also a great way to allow team managers and business managers gauge the progress or lack of progress a team is making.

It can take time and training to properly implement agile in your shop. It does have overhead, and it is not the right solution for every situation. Plus, many people equate agile with iteration. Iteration is just one component of the agile process,

SUMMARY OF SMALL-SHOP BEST PRACTICES FROM THE EXPERTS	
Small shops	"Lots of small teams with specific projects work well within large organizations, as small teams tend to communicate better. The small shop is just like a small team in a larger organization, and with the right people, can achieve surprisingly large projects."
Agile	"If you think you need agile, then be aware: Agile is more than just iterating — a lot more. But iteration seems to get all the focus. If you cannot implement a meaningfully complete set of agile best practices, then just avoid the hype altogether and do not do agile. Just do whatever works for you to stay organized. If you can do agile well, then you will be better off, but it is multifaceted, and success depends on all the facets — not just iteration."
Software patterns	"Michael Jackson (not the singer — *http://mcs.open.ac.uk/mj665/*) recommends: Make it work, make it right, make it fast. Start by making prototypes that demonstrate the concept, get some feedback, and then throw it away! Use the learning to build a version with a good architecture (that passes quality assurance tests). If and only if it is not fast enough, throw it away! Then use the learning to make an optimized version. Moral of the story? Throwing away code is good!"
	"Avoid truck numbers of one. One truck hits one developer and ends the development of a critical piece of software. Pair programming, peer review and transferring developments at various stages to different team members. This improves code quality and reduces risk."
	"Read other people's source code. Research APIs and tools that can help you beyond the obvious ones. Only implement what you need ... what your business specializes in ... Otherwise, use open source or buy libraries."
	"Fewer lines of code mean less errors and less maintenance. Actively seek out languages and APIs that are concise."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Wheatstone

SUMMARY OF SMALL-SHOP BEST PRACTICES FROM THE EXPERTS	
Change	"Resistance to change can lead to entrenched unproductive practices that produce low-quality software — a double whammy. Do not stick with antiquated development practices, tools, languages, etc., just because everyone is comfortable with them and thinks it would be too costly to change; that is just 'resistance to change' speaking. Recognize when opportunities for change are available. Then do it. If you do not move on, you will be left behind. Speaking quite frankly, you need to be realistic about people. Changing development practices sometimes requires changing people. That's life."
Tools	"Use the tools, Luke! Many excellent software development tools and frameworks for source code management, unit testing, regression testing, continuous integration, continuous deployment, issue tracking and team collaboration (e.g. wikis, agile tracking aka Pivotal) are available. They take time and expense to set up, and developers need to buy into using them. In my experience, every second spent doing this is worth it!"
Testing	"You cannot succeed without automated testing. If it is missing, then go back and fix that. Once testing is in place, development becomes a test-driven process. Get quality in your team's head, and keep it there from that point forward."
Testing	"Plan on how you are going to test the software before and as you write it, using automated testing frameworks. The code does not exist until the tests are written, the tests can be run automatically, and your code has passed."

albeit an important one. Iterating your code allows you to get software out there that delivers "a unit of business value," meaning that the release does something positive for the business. So you do not have to do everything prior to getting value out of the time you have spent developing the code. But if you iterate your code and think you are doing Agile, you are missing out on a lot of the value that the framework can bring to your shop and to your business.

Software patterns

Look for software patterns. Software patterns are practices that have been tried in the past and worked well. The sidebar on page 18 lists some good patterns.

And, of course, there are specific software coding practices that work well too. Look for practices that work well in your organization, and actively seek to repeat them.

Not only should you look for software patterns that are successful, you should also actively study anti-patterns. This helps you to avoid making mistakes — either making mistakes that others have made, or repeating mistakes you have made yourself. Studying failure can be very enlightening. The anti-pattern I see over and over is Watch Folders. I could write a whole article on why these are a bad idea, and perhaps one day I will.

Tools

The Open Source community is an absolutely invaluable resource for small software shops. Just about any tool you can imagine for the development, testing and deployment of code is available free of charge. (See *http://sourceforge.net*, for example). While you can pay a lot of money for commercial development tools, there is almost always a free tool available that will do the same thing. Check for Open Source tools that have a good following and are well supported, and then implement them.

Testing

As the testing comments in the sidebar above make clear, a critical step for any shop is testing your software. You could argue that if you cannot describe how you would automatically test your software, you do not know what your software is doing in the first place — a dangerous proposition for sure!

Sometimes we find ourselves doing jobs that are very different from what we originally signed up to do. And for some reason, broadcast engineering is one of those fields where this happens rather frequently. Think about your job. Think about the advice of these software experts. If you now find yourself in a situation where you are writing code for your company, or you are managing people who do, you can benefit from the wise council of these experts. BE

Brad Gilmer is president of Gilmer & Associates, executive director of the AMWA and executive director of the VSF.

? Send questions and comments to:
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+ ADDITIONAL RESOURCES +

The following are available on the *Broadcast Engineering* website:

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- Media services: Services architecture enables functional flexibility
- System network core concepts structure overall performance

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Cameras and lenses

Higher resolutions push sensors and optics.

BY STEVE MULLEN

The camera industry is pursuing advanced — beyond HD — camera technology. These advances include both higher temporal resolution and higher spatial resolution. The latter, in turn, forces changes in the choice of a lens for use with an advanced camera.

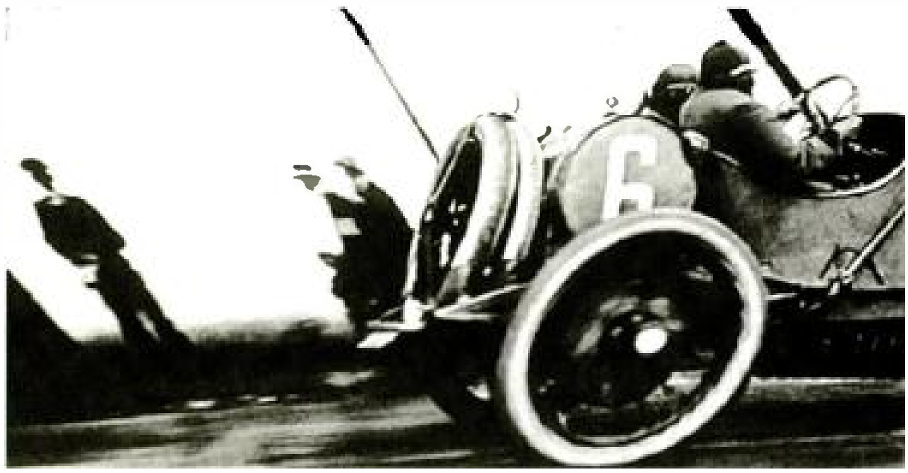

Figure 1. This rolling shutter artifact caused the car wheel to appear oval.

Sensors and processing

The earliest of the recent advances in camera technology is the switch from CCD to CMOS sensors. One problem, banished by CMOS, is vertical smear. Unfortunately, this step forward has been achieved with a step backward. Rolling shutter artifacts are present when there is motion by the camera and/or objects moving within the frame. Slight random movements by a handheld camera can create a wobbly gelatin look that is particularly disturbing.

Because a CMOS sensor's rows are processed — reset, integrated and output — in a sequence that occurs over time, a CMOS sensor exposes each frame in a top-to-bottom pattern. The row-exposure offset creates a rolling-shutter skew. (See Figure 1.) To date, the most common solution has been to read out multiple vertical slices of the image simultaneously. The more slices a chip can output, the faster a whole image is captured — and the less rolling shutter artifacts.

Most CMOS imagers today use active-pixel sensor (APS) technology that is implemented by three transistors. Each pixel has a reset transistor, an amplifier transistor and a row select transistor.

By adding a fourth transistor to each pixel's circuitry, it is possible to capture an image with all sensor pixels simultaneously. All sensor photodiodes are reset at the same instant. By beginning image capture at the same moment in time, there is no row-exposure offset to cause rolling shutter. The fourth transistor holds a photodiode's integration value until it can be read out. This *global shutter* design eliminates rolling shutter.

Frame rates and resolution

The multislice technology that enabled rapid sensor output, although not needed to reduce rolling shutter artifacts, still serves a purpose in today's cameras. Camera manufactures can use fast sensor output to support higher frame rates. The obvious use is to provide the temporal resolution of 720p60 at a 1920 x 1080 or 3840 x 2160 spatial resolution.

Many consumer HD cameras already support 1080p60. Like 4K cameras, there are no distribution systems that support 1080p60. However, when editing projects that employ motion effects, the 2X additional full-resolution frames enable smoother effects. And, 4K enables static and dynamic video crops.

Increasing camera resolution supports two options. First, many new cameras use only a single sensor. Thus, some form of demosaicing is required

Crop factor

A photographic lens' focal length (e.g., 50mm) is specified relative to the dimensions of 35mm still film, which are 35mm x 24mm with a diagonal of 43.3mm. In the digital world, this is called a full frame. When a 50mm lens is mounted on a full-frame camera, it provides a field of view that is similar to that provided by our eyes.

When a 50mm lens is used with a camera that has a sensor smaller than 35mm x 24mm (for example, a 22.2mm x 14.8mm APS-C sensor, which has a diagonal of 26.7mm), only the central portion of a scene is captured. The resulting capture has an angle of view that is necessarily smaller. The scene looks as though it were shot with a longer lens on a 35mm camera. The relation between the 43.3mm and 26.7mm diagonals is called the crop factor. In this case, the crop factor is 1.6. Therefore, a 50mm lens behaves like a 80mm lens when mounted on the smaller sensor.

VIDEO

by Lawo

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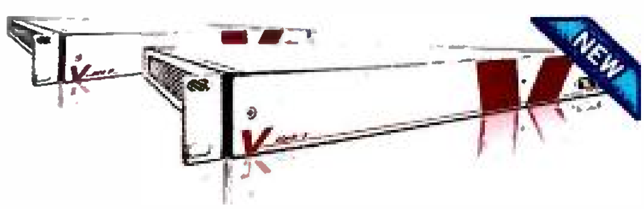

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LAWO

Figure 2. JVC's UHD GY-HMQ30 camcorder

to obtain RGB or Y'CrCb signals. A typical de-Bayer process yields a horizontal luminance resolution of approximately 78 percent of the sensor's horizontal resolution. However, by employing a sensor with 2400 (or more) pixels, after a de-Bayer, the potential horizontal luminance resolution will be 960TVL or 1080LW/ph.

These cameras are referred to as "2.5K" cameras. Potentially, they can provide the same horizontal luma resolution as a 2K-pixel three-chip camera. A single-chip camera can also provide equal chrominance resolution when recorded to a 4:2:2 format. However, the competence of the demosaicing process determines the amount of chroma artifacts captured. Chroma artifacts are not an issue with three-chip cameras.

When recording to a compressed HD format, horizontal resolution is real-time downsampled within a camera to 1920 columns. Cameras that capture RAW information record all pixels with the downsample, if desired, performed in post.

Second, 4K, and even 8K, camera resolution is possible. UHD (3840 x 2160) and 4K (4096 x 2160) are coming to market. These cameras include JVC's newly announced GY-HMQ30 (shown in Figure 2), which has a Nikon F-mount and records using 144Mb/s H.264, as well as RED Digital Cinema's 6K EPIC DRAGON (RAW-only recording) and Blackmagic Design's Production Camera 4K (RAW and log recording).

Sony's F65 is capable of what it calls "True 4K" plus 8K capture using its non-Bayer 20-megapixel Super 35 chip. (See Figure 3.)

Higher camera resolution demands higher lens modulation transfer function (MTF). A lens' MTF describes the relation between image contrast and spatial resolution. The higher the frequency, as represented by the X axis of Figure 4, at which roll-off begins, the more visible the fine detail that passes through a lens. Visibility is a function of contrast that is represented by the Y axis. Figure 4 presents a sample MTF for an HD curve. A 4K camera requires at least twice the performance.

Lenses

The quest for high-resolution lenses has led to equipping these new cameras with cinema and photo lenses. The F65 can use Sony's electrically coupled versions of Zeiss lenses. Other 4K cameras provide a PL-mount for cinema lenses. Other cameras support Canon EOS lenses or Micro Four Thirds (M43) lenses. EOS and M43 mounts can employ adaptors that enable use of F-, A- and E-mount lenses. (See the "Crop factor" sidebar on page 22.)

Modern lenses developed for still cameras do not have an aperture ring. Aperture control is via an electronic coupling. Thus, an adaptor must, itself, provide aperture control. This control will drive the aperture within the lens via a coupling pin. Obviously, the adaptor ring cannot have marked f-stops. Aperture is set by monitoring exposure on the camera's LCD.

Photo lenses typically alter aperture size in steps rather than continuously. That means you will not be able to smoothly adjust the aperture while shooting.

The lens and camera engineering trends we are currently experiencing are likely to continue because the next technology target is 8K video capture and recording resolution. **BE**

Steve Mullen is the owner of DVC. He can be reached via his website at http://home.mindspring.com/~d-v-c.

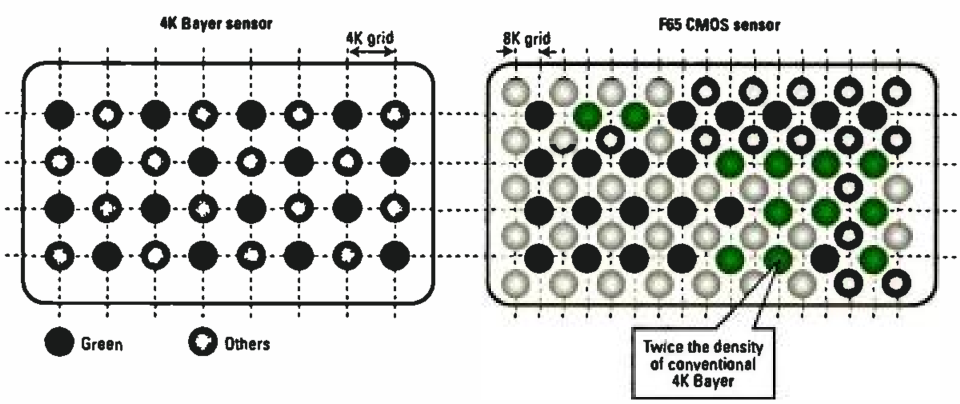

Figure 3. Sony F65 "True 4K" and 8K sensor

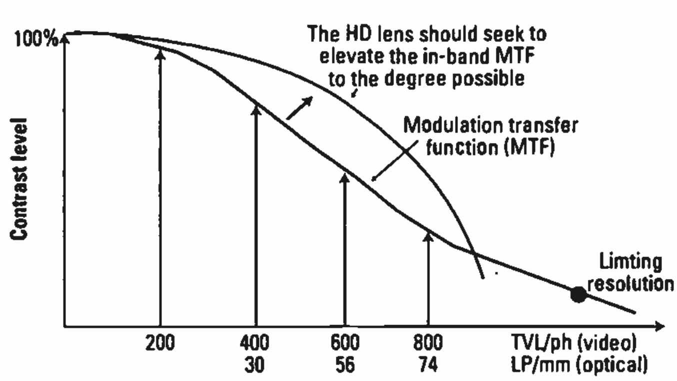

Figure 4. Example MTF curve for an HD camera


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A next-generation broadcast platform

A new standard should address three specific areas.

BY BRETT JENKINS

I get tired of people saying the broadcast industry is full of dinosaurs. It gets under my skin sometimes. But the thought occurs to me that maybe it bothers me so much because it just might be true.

The pace of technological change in today's world is astonishing, and being trapped by a 1990s paradigm of television might very well be the meteor crash that wipes us out. At the same time, the fact remains that viewers really like broadcast television. A simple, linear stream of high-quality content that is always on, easy to access and easy to use continues to provide great value for the vast majority of people. But I don't think it's a debate anymore that viewers' habits are changing. Viewing continues to be more and more fragmented, both in terms of the content people are watching and the devices they are watching on. And perhaps more importantly, time-shifted viewing is making up an ever increasing slice of the viewing pie. So we're left with a question about what we could change to better service our customers in today's world.

That's why so many in the broadcast technical community have been talking about what we could do from a technical standpoint to unshackle the chains so that we can continue to be players in the world of video information and entertainment. The idea of a next-generation broadcast platform using a new broadcast standard has been talked about for years, but what does that mean? If the industry is given a chance to reinvent itself technically, what should we strive to look like? For this article, I'll focus on three areas of advancement that I think should be included in any new platform: improvements in the radio link, changes in video resolutions and codecs, and the ability to introduce advanced services.

Radio link

As you look at the state of the art in today's world compared to when 8VSB was adopted by the ATSC, you'll discover that we actually have a very "bit-efficient" standard. That is, in terms of bits delivered per MHz used at a given C/N, the current system actually stands up surprisingly well. But the ATSC standard (ATSC A/53) has always had problems in reliably delivering those bits to a receiver when the propagation path has changing characteristics (multipath or Doppler). Performance under those conditions is significantly improved when using the newer DTV Mobile broadcast standard (ATSC A/153), but that standard requires you to give up much of the bit efficiency that was inherent in the 8VSB system to start with.

> **If the industry reinvents itself technically, what should it strive to look like?**

An ideal technology would give you great performance in terms of bits delivered, but also allow you to reach devices that are subject to multipath and mobility. For me, this is a critical point, and one of a few fundamental requirements that must be included in a next-gen system: Broadcast airwaves need to be able to reach unattached devices.

More and more viewing is taking place on screens other than the fixed TV in the living room of your home. A recent Council for Research Excellence study[1] has said that about 2 percent of all TV content viewing happens on mobile devices. That may be a small number, but it's really just getting started, and it's growing quickly. Still, one could debate whether a broadcast standard should reach such devices. I'm convinced that broadcast delivery (especially of live, high-demand content) is absolutely the most efficient, and therefore the most consumer friendly, way to deliver programming. It simply doesn't make sense to me to replicate the same data over and over and use limited spectrum resources to deliver programming on a one-to-one basis.

But what if you don't want to reach mobile devices and instead want to focus on delivering the most bits possible to a fixed antenna? A next-generation standard could allow for that case as well. Some of the most advanced broadcast standards today have modes that offer as much as double the bit capacity (somewhere around 40Mb/s) in the same channel.

That brings up another capability that should be part of a new standard: The standard could have multiple operational modes that are flexible enough to allow each broadcaster to optimize the delivery of bits based on how difficult the channel is or what their service model is. So if I want to be in the mobile broadcast business, I can. If the demand is for OTA delivery of 4K Ultra HD video at a

FOOTNOTE:

1. http://www.researchexcellence.com/documents/misc/TV_Untethered_ARF_FINAL.pptx

very high bit rate, I can choose that instead. Or even better, I can deliver multiple services each with their own bit rates and levels of robustness all at the same time, and the device itself can figure out which service is best suited for its environment.

Content compression

We know that the current ATSC system uses MPEG-2 for video compression. Over the past few years, codecs have advanced, and most newer video delivery systems are using tools from the MPEG-4 toolkit. That's a technology that delivers approximately the same video quality at about half the bit rate. International standards groups are working on an

An evolvable standard is one where you can make changes as time goes on but not break existing services.

even more advanced video compression standard called High Efficiency Video Coding (HEVC) that targets another 50 percent reduction in bit rate for the same quality. These advances in video coding are essential if you want to enable OTA delivery of higher resolutions such as 4K.

I can't say whether 4K will be the next big thing in television viewing or whether it will go the way of 3-D, or perhaps be something in between. I do believe that many television viewers care about high-quality video, and as a broadcaster, I know I need to be prepared to deliver what my viewers want, and I don't have that capability with today's system. So it seems obvious that a next-generation standard would need to include the most advanced compression schemes available. Beyond that, the industry should look for a way that the system can evolve as compression technology continues to advance. What we really need is an upgradeable system that allows new, future codecs to be introduced without stranding receivers. This implies a system that allows receivers to be upgraded via software, a capability that many video playback devices already have.

Advanced services

Finally, a new standard could and should enable advanced services with the hope that these services will create value for users and enable new revenue streams for broadcasters and other participants.

The types of services envisioned — targeted advertising, synchronization with other applications (second screen or first screen), non-real-time delivery of content, etc. — have actually been talked about for a long time. Many of the ideas can be enabled by incorporating two things: a flexible data transport mechanism based on IP, and a flexible application execution environment on the receiver.

The good news for the broadcast industry is that television sets are already beginning to have those capabilities as part of Smart TV features, but we need to have a broadcast delivery that is able to take advantage of the features. Once the delivery and execution environment are set up, you have created fertile ground that will allow for creativity and flexibility in what advanced applications actually do. You don't necessarily have to define exactly what all the applications are. That's the vision that I believe our industry should have when it comes to a next-generation platform — one where we can innovate and evolve what we are doing.

Evolving standard

One last point that overarches all these technical advancements is an idea that we need an "evolvable" standard so that we don't end up in the same place again 20 years from now.

An evolvable standard is one where you can make changes as time goes on but not break existing services. It encompasses both forward and backward compatibility. You can choose to stay with an old version of the standard and still have devices both old and new continue to operate, or you can move to a newer feature set and not break existing receivers, albeit they may not be able to take advantage of the new features.

Much of the discussion among industry technical experts revolves around how to accomplish that. It sounds like an obvious idea, but it's hard to come up with such a platform. The broadcast industry is hampered in two major areas where the telecommunication industry is not. First, we're not vertically integrated. That is, we don't control both the network and the user equipment. Second, we don't have two-way communication with user equipment, so we can't "negotiate" the communication link. Those challenges will be hard to overcome, but it will be important to do so if we want to make sure we don't repeat history with a next-generation platform.

The next-generation broadcast platform that I hope to see will have all those characteristics: an advanced and highly flexible radio link that will allow me to deliver the maximum bits in challenging conditions; a video compression system that allows me to send the highest quality video that is available on the market; a platform that allows for new and innovation applications; and a defined path that allows for change so we're not stuck with the same technologies forever. I'm happy that so many of my colleagues in the industry see this same possible future and are working together to make it happen. BE

Brett Jenkins is VP CTO, LIN Media.

? Send questions and comments to: editor@broadcastengineering.com

+ ADDITIONAL RESOURCES +

The following are available on the *Broadcast Engineering* website:

- ATSC 3.0 planning already under way
- Broadcast DTV
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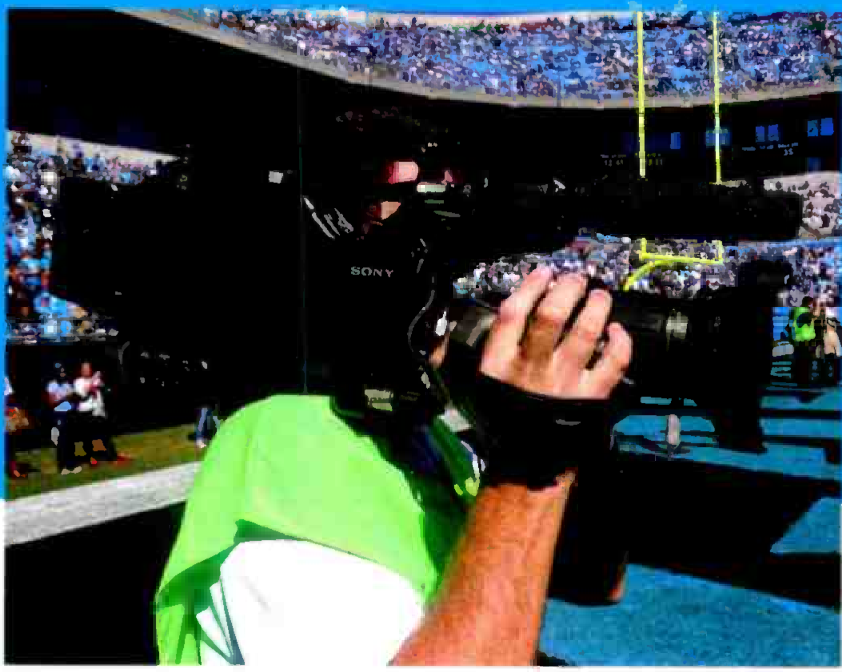

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Pros tell their DWX stories at sony.com/DWX

Microphone placement

Care must be taken to achieve the best sound quality.

BY EDDY B. BRIXEN

When you want to record or amplify a given sound source like a voice or a musical instrument, it is obvious to place the microphone in the near field close to the source. This sounds easy. However, there are some acoustical and technical matters to consider in order to make it sound right. This article will briefly outline some of the important issues.

The sound source

The first thing to consider is the sound source itself. What does it sound like? How loud is it? Is it possible to catch the essentials of this source just at one pickup point? Remember, many musical instruments only

The naturalness, timbre and intelligibility of the human voice is affected by microphone placement in both broadcast and theater productions. For instance, stage actors outfitted with head-worn lavalier mics, such as the those shown in the images above, might opt to wear them upside down to reduce the amount of noise generated by breathing.

Similarly, body-worn mics, such as the one worn by the anchor shown top left, raise the spectrum of the human voice 800MHz from neutral, and attenuate it in the 3kHz to 4kHz range.

sound right if all the radiated sound is summed up by the acoustical room. Selecting a position for a microphone is a subjective choice.

The voice

The voice should be picked up in a way that meets all demands for naturalness, timbre and intelligibility. Actually, what is perceived as a natural-sounding voice is what we hear in front of a speaking person, and at a distance of approximately 1m.

In broadcast and live sound environments, sound engineers will often opt for a lavalier microphone (chest worn) or a headset microphone (head worn), as these allow more freedom of movement for the artist, commentator or presenter. One should be aware of the fact that placing the microphone at this shorter distance results in a recorded spectrum that is different from the natural and neutral spectrum perceived at a normal listening distance. This difference is far from negligible.

From the curves shown in Figure 1 it can be seen that there is a general tendency of the spectrum of voice to raise around 800Hz, which basically must be considered and compensated for. However, the most important deviation to be aware of is the attenuation that causes reduced speech intelligibility.

The speech spectrum at the typical chest position has a lack of frequencies in the range of 3kHz to 4kHz.

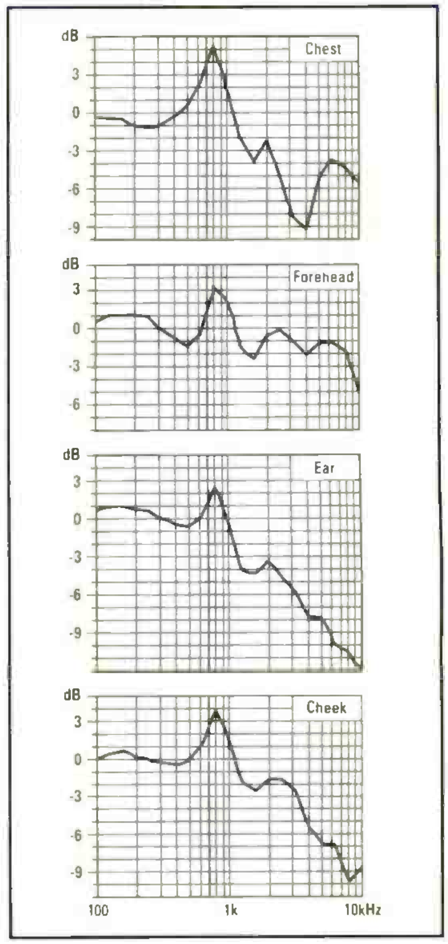


Figure 1. Four curves are shown. The upper curve quantifies the way the speech spectrum picked up at the chest differs from the spectrum of the same person's speech is picked up at 1m. [Ref: Brixen, Eddy B.: AES Convention no. 100, Copenhagen, Denmark. Preprint 4284.]

This frequency range is extremely important for obtaining speech intelligibility. If a microphone with a flat frequency response is positioned on a person's chest, the 3kHz-to-4kHz range should be boosted around 5dB to 10dB just to compensate for the loss.

In reality, there are two solutions: Use a microphone that is pre-equalized to compensate, or remember to make the right equalization in the editing process. Note that no ENG mixers or cameras automatically compensate for this, and no controls are provided to do so. In many cases ,this is not compensated. Hence, the intelligibility is often low.

If a microphone is positioned in a fixed position on the chest or mounted on a stand in front of a person, turning of the head may change the level and the timbre of the voice. This problem is solved by using a headset microphone. Also the level is approximately 10dB louder at the cheek compared to the chest position. So, there are many reasons to use headsets — i.e., the spectrum is less affected compared to the chest position. However, to some degree, a high-frequency rolloff has to be compensated.

Interestingly enough, a forehead position (close to the hairline) — as is often used in film and stage performance — is relatively neutral regarding speech intelligibility.

Handheld microphones are often used for interview/voxpop. If possible, the microphone should be kept in a fixed position. This prevents the microphone from being at the wrong position at the wrong time. However, if the background noise it is a problem, the obvious solution is to move the microphone closer to the speaking person.

Unwanted sources

Another solution for reducing unwanted background sounds is to choose a microphone with higher directivity — typically a cardioid type. Using a shotgun/interference tube microphone is rarely efficient, especially if the noise predominantly contains low frequencies, for instance, at car racing tracks and similar locations.

All cardioid microphones exhibit proximity effect, which is often useful when fighting background sound. The proximity causes a raise of low frequencies when the sound source is close to the microphone. On the other hand, when equalized for the right tonal balance, the same effect will provide a reduction of low frequencies of distant sound sources. By the way, remember that most cardioid types of microphones exhibit most proximity effect on axis. This is why handheld cardioids should not be addressed from the side. (Many untrained microphone users treat the microphone like an ice cream cone.)

When placing a directional microphone in front of large surface — like the front of a kick drum, the back of an upright piano or just in front of a person's cheek — it must be noted that reflected sound can "bypass" the microphone's rejecting direction. (See Figure 2.)

Noisy microphone

Under some conditions, the noise may be generated by the microphone. Electrical noise (microphones' self noise) should not be a problem, as the sound pressure level from the primary source is high. However, mechanical noise, such as handling noise, and also wind-generated noise, have to be considered.

It should be noted that, in general, omnis (pressure microphones) are less sensitive to mechanical vibrations and wind, compared to cardioids (pressure-gradient microphones).

Mechanical noise can even be transmitted by the cable. Also, noise from clothing can be a problem, for example, when hiding a miniature microphone. This is especially a problem with body-worn microphones in film production and explains why various gadgets have been developed to reduce the problem.

When using a cardioid headset microphone, it is advisable to use a windscreen, as this can reduce wind-generated and popping noises.

In general, omnidirectional miniature microphones have perfect directional characteristics. When it comes to wind sensitivity, they are more sensible on-axis. So for chest-worn omnis, it is an advantage to mount the microphone upside down, as the sound quality remains the same, but less noise is generated from breathing. (See Figure 3.)

RMS and peak level

Last — but not least — what kind of SPL can be expected from the sound source? When we speak about the level, we normally mean the SPL — the RMS value. That's a kind of mean value. However, what is more interesting are

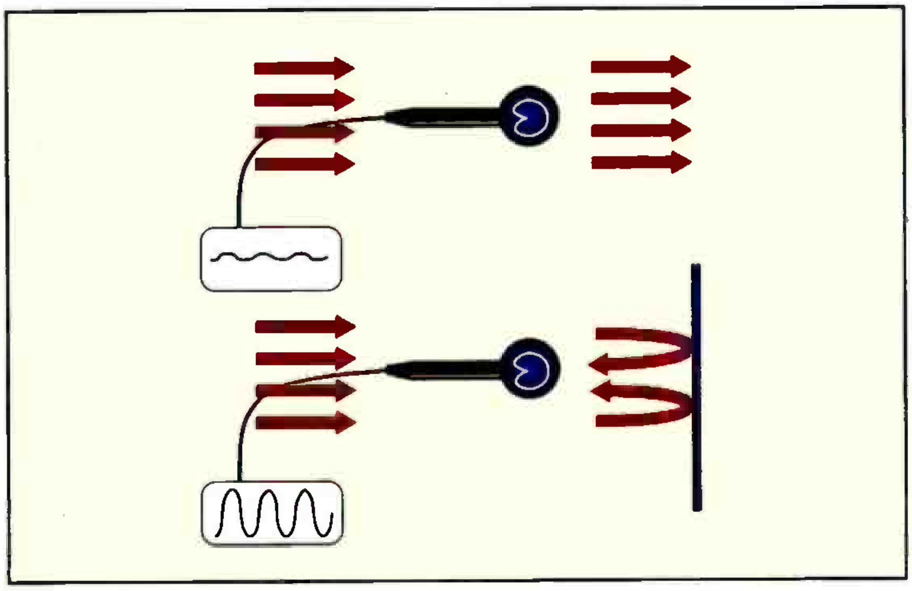


Figure 2. When placing directional microphones, it must be noted that reflected sound can "bypass" the microphone's rejecting direction.

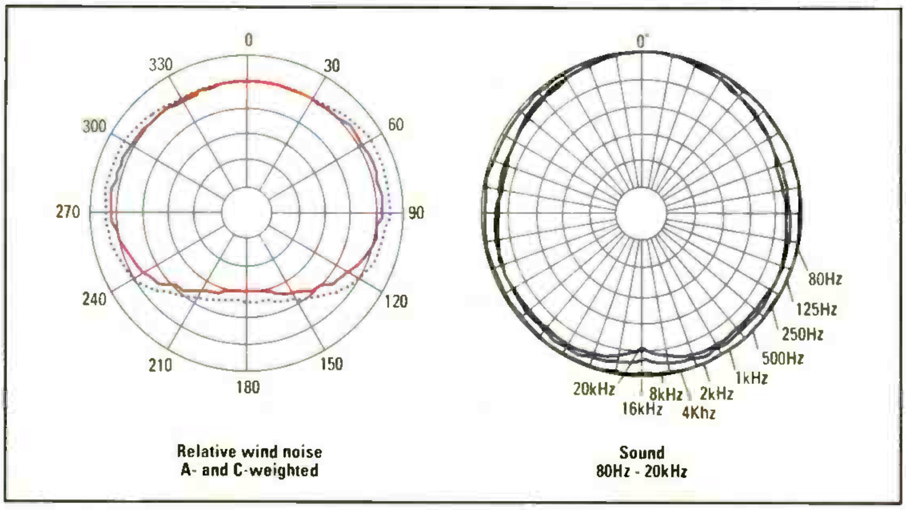


Figure 3. Varied factors have to be taken into account in close miking applications. For instance, it is desirable to mount chest-worn omnis upside down, because that way, less noise is generated from breathing.

the peak values, because these values may be 10dB to 20dB higher than the related RMS value.

For instance, if a (loud) singer is able to produce a SPL of 140dB at the lip plane, this may result in peak values far above 150dB.

Inside a kick drum, close to the bell of a trumpet, close to explosions, extremely high peaks can be expected. In the recording channel, "space" must be prepared for that. So, the microphone should be able to reproduce the high levels, and the input stage should be able to cope with signals that can be as high as 1V.

Using sound

Close miking has many applications. It is a good technique, and it provides good separation. However, care has to be taken in order to achieve the best sound quality. BE

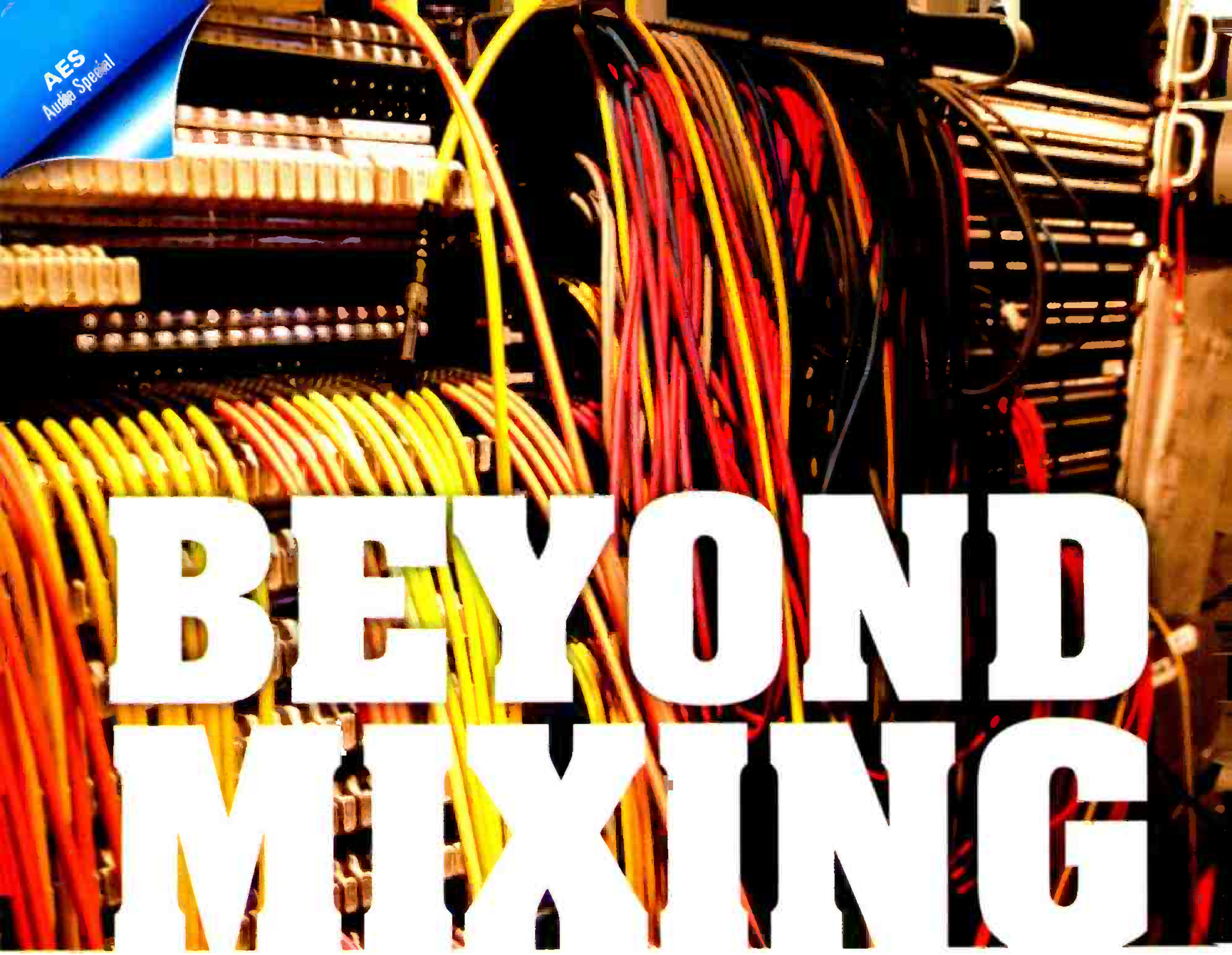
Eddy B. Brixen is an audio specialist at DPA Microphones.

Send questions and comments to: editor@broadcastengineering.com

ADDITIONAL RESOURCES

The following are available on the *Broadcast Engineering* website:

- Microphones
- DPA microphones used to create Golden
- Microphone tutorial



BEYOND MIXING

Digital matrix routers represent a paradigm change in audio broadcast production.

BY FELIX KRÜCKELS

From the manual switchboards and the early electro-mechanical automatic exchanges developed in the late 1800s, the telephone has evolved far beyond the imaginations of its inventors. A measure of this progress can be seen in the movies, where entire plots hinge on the telephone systems of different times, from the landline alibi of "Dial M for Murder," through the public hostage in "Phone Booth" to the virtual escape routes of "The Matrix."

Today, sports and events production comms systems are following the trail of telephone systems into IP-based routing, with the implication that the rest of our audio systems and operations will soon follow. But the truth is that we are rather further behind the telecoms curve than we would like to think.

Looking back, it is easy to see how close early telephone routing and sound recording were. Not only did our patch bays resemble the bays of a manual switchboard, but the plugs we used were the same — and subtly different from the standard quarter-inch TRS jacks found on musical instruments and mixing boards.

We suffered from the same patching problems too — dirty jacks giving high-resistance connections, fatigued joints, connections lost whenever patch cords were disturbed. While with big monitor speakers, the studio "spaghetti" patch bay alongside the mixing board impressed the record company A&R man, it often frustrated the best efforts of the recording engineer to do the best job. The same frustrations were equally common in broadcast studios and OB vehicles.

A new paradigm

Not to be confused with digital networking, routing was one of the benefits brought about by digital audio. While sound purists debated

Today, facilities often try to avoid the type of "rat's nest" shown here by relying on routers.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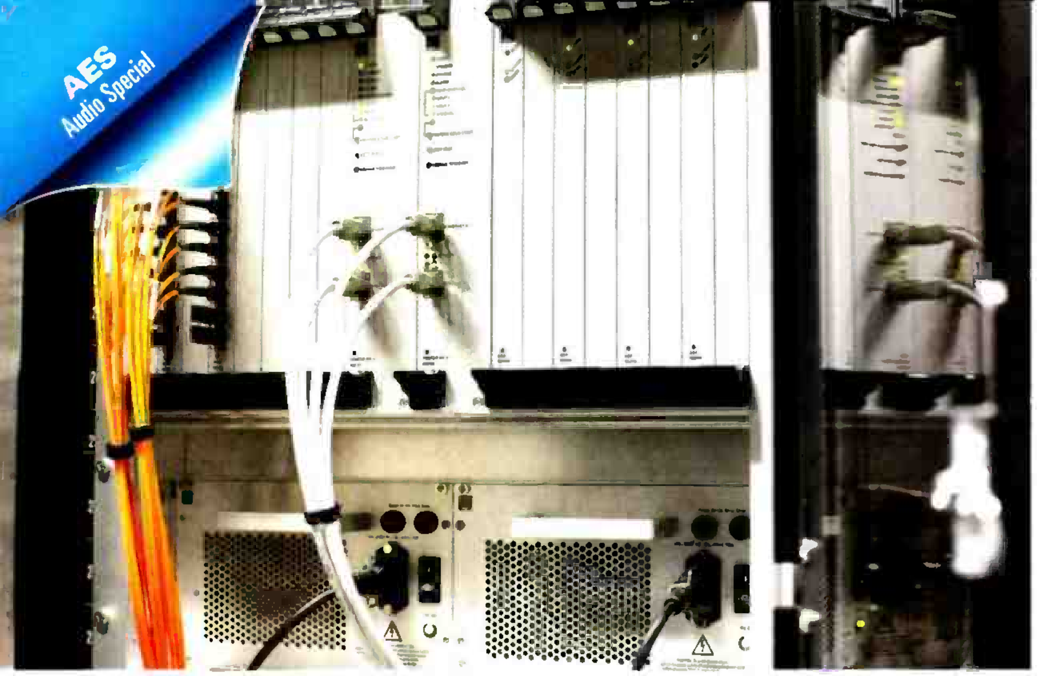


The digital router core still resides in a patch bay, but the "spaghetti" has been replaced with fewer physical connections and flexible internal configurations.

the merits of different sample rates and bit depths, the practical advantages of digital routing were clear, although some were more evident than others.

With digital matrices taking care of routing, the immediate failings of mechanical patching systems vanished. In their place, digital routers allowed their inputs to be assigned to their outputs (even multiple outputs) at the touch of a button. This is exactly the same principle that underlies the first electronic telephone matrix switches, actually, as is the concept of central control. Yes, we still have patch bays in audio, but the spaghetti plate has been replaced by an impressionist painting — a few arcs of color that only allude to the complexity of the tasks they perform.

The digital audio matrix not only brings more reliable routing, it allows I/O configurations to be stored in memory and almost instantly recalled. It also allows control of routing and processing from a central operator's position, eliminating the need to break from other operations to make changes or adjustments. To do this, its footprint has been transformed from a rack of jacks and cables to a panel in the console bay and pages on a system monitor. Apart from being tidier, it's infinitely easier to read.

Better yet, a programmable system allows configurations to be moved independently of the equipment they served. It even allows configurations to be prepared away from the control room or OB truck. Before we follow telephone systems into packet switching and IP addressing, we would do well to understand the full potential of digital matrices — both their practical advantages and the benefits they bring to broadcast and live events.

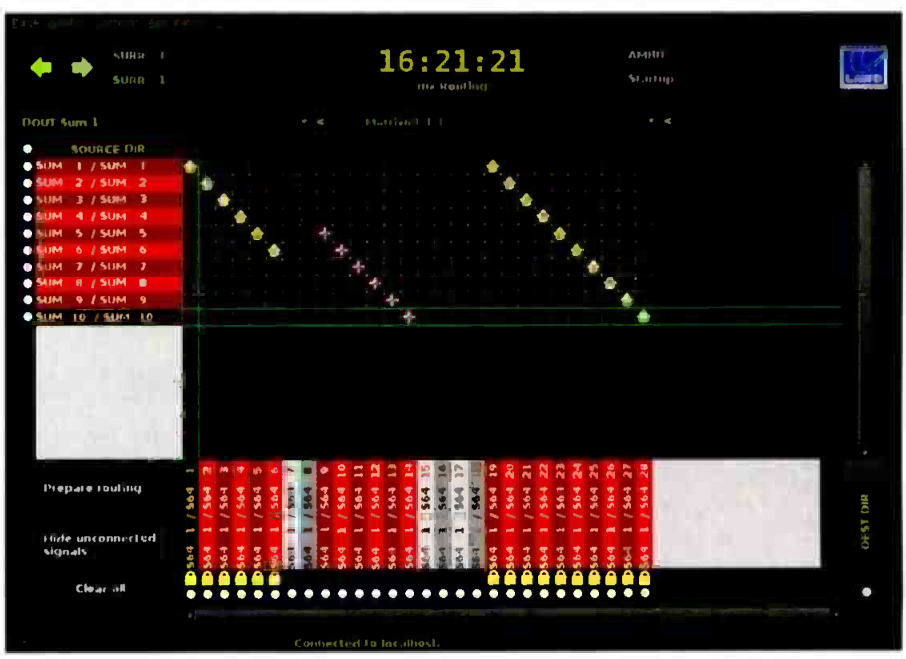
A simple study of time spent against gains made in setting up a live mixing console or OB truck shows how a given setup for a manual system comes at the cost of time and/or number of operators. The immediate gains of a simpler, quicker and more reliable digital matrix are just the first step.

If a setup needs to be repeated, it can be carried over fully and instantaneously, with cumulative savings in time and effort. This offers obvious applications to regular television broadcasts, theater productions and live sports events.

In addition, a broadcast studio or OB truck that can be readily configured to serve different duties can be more productive, and profitable to operate, than one that requires hours to configure each time it takes on a different production. A truck that can be quickly reconfigured to provide its full potential range services can be used more frequently and in a wider range of applications than one that cannot.

In broadcast, as in live sound, renting a digital mixing console locally and configuring it from presets

A programmable system allows configurations to be moved independently of the equipment they served.



The digital router allowed its inputs to be assigned to its outputs (even multiple outputs) at the touch of a button. Its footprint was transformed from a rack of jacks and cables to a panel in the console bay and pages on a system monitor.

carried on a memory stick removes the logistical and environmental issues of shipping a large, heavy piece of equipment. In the same way, a single console can be configured to meet different needs and then updated as required during a program or event.

Preparing a system configuration ahead of time and away from a specific location can save time on-site.

Stored configurations can even be passed from one sound engineer to another as they assume responsibility for a TV show, sports event, artist or tour.

Working offline is another concept that defies old analog systems. Preparing a system configuration ahead of time and away from a specific location can save time on-site and make good use of time that is necessarily spent in hotel rooms or traveling from location to location. This is equally true whether the travel is taking you between gigs or broadcasting a sports event from around the world.

Route of no return

So far, the advantages of digital matrix routing appear largely — although not exclusively — practical. There are, however, good business and peripheral gains here, too.

There are advantages in productivity and efficiency for audio operators, whether they are broadcast studios, OB services or live sound production companies, as well as cost savings.

There is the opportunity to review. After a broadcast ends, it can be reviewed offline to assess which aspects worked and which could be improved.

Finally, there are quality of work advantages. Freeing engineers from the physical duties of managing an analog patch bay offers savings in time, but freeing them from logistical duties and operational distractions also leaves them free to do a better job. From a company perspective, all of these considerations add up to better service for the client.

The future is likely to see IP step in, and the best use of the next development will follow proper understanding of this one. BE

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Chicago's BAM Studios produces audio for television, feature films, corporate presentations and commercial ventures.

AUDIO REFERENCE MONITORS

Innovations in audio monitoring deliver more reliable and consistent results.

BY GARY PARKS

The reference monitor speaker performs the same role for the audio portion of the program as the video monitor does for the images. It provides the neutral, uncolored "truth" about the clarity, quality, intelligibility, noise floor and other elements of the sounds that have been captured by the microphones. When mixing multiple audio tracks in post-production — blending dialog, ambience, sound effects and music to create the finished aural experience — these monitors allow the mix engineer to accurately place each of these elements using the tools of relative level, equalization for frequency shaping, spatial placement and reverb/effects to alter listener perception of the sources of those various sounds.

While circumstances differ somewhat between an engineer recording a band to create music tracks destined for airplay and one mixing dialog and other sounds for an interview, sporting event, documentary or commercial project, both trust the audio monitors on the console to serve as a reliable reference. For the former application, the audio track must stand by itself. The stakes are fully as high when audio accompanies and interacts with video images, creating an experience for the viewer where the sum is greater than its parts.

How have manufacturers advanced the art of monitoring beyond a woofer, dome tweeter and crossover in a rectangular box? What technical innovations have been introduced to enhance consistency, transparency and reliability? How have relatively recent breakthroughs in electronics, digital signal processing and computing been married with transducers and enclosures to create the latest audio reference monitors?

Performing a function

The key requirement is that the audio monitor is a reference and ideally reproduces any audio signal introduced through it without coloration, distortion or phasing anomalies, and with the transient characteristics and frequency balance of the original source (or at least as it was captured by the microphone). A sales engineer at one of the leading monitor manufacturers lamented

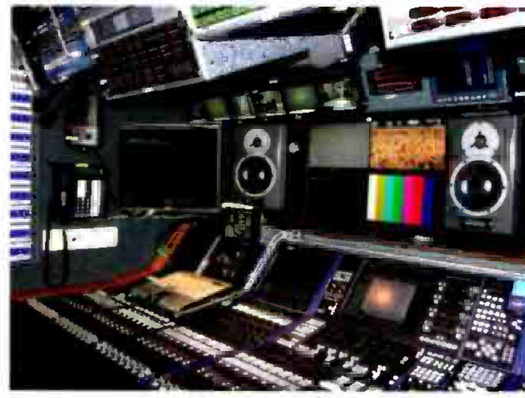
that, in contrast with the exciting adjectives that can be used when presenting an effects device or certain other audio goodies, the description for monitors are words such as neutral and transparent, and the main goal is "preventing coloration."

The audio suite at WestCom Media Group, Germany, produces daily content for private and public televison.

This fidelity to the source material is what gives an engineer the ability to make educated, critical judgments when building the soundscape. The neutral quality also can help prevent or delay listener fatigue as the engineer spends long hours evaluating minute changes to EQ, levels, panning and effects to best meld audio with video. And in a live setting, it can promote confidence.

An A1 audio engineer specializing in sports told me that in the rush to set up for an event, he doesn't have the time to test or calibrate the audio monitors above the mix console in the OB truck. The smaller near-field active monitors he typically encounters in the compact space devoted to audio are usually approximated toward the engineer's ears, and he accepts them as a neutral reference for his mix.

Depending on the size of the mixing location, the level of overall SPL



Shown here is the audio booth in one of Total Production Services' HD OB trucks.

required, and the amount and pitch of low-frequency energy within the program material, monitors of differing sizes are required. For near-field work with audio mainly in the vocal range, a pair of accurate two-way monitors with perhaps 4in to 5in low-frequency drivers would be sufficient. Large post-production facilities often have larger wall-mounted stereo or left-center-right monitors with subwoofers, and perhaps other mid-sized

monitors in surround-sound positions or in more near-field positions on the console bridge.

Even while using transparent monitors for critical mixing, more mundane speaker systems are often used to demonstrate the integrity of the mix to clients. A major Chicago post house keeps a pair of computer monitors and a consumer LCD-TV wired in to check the final results.

Active monitors

When amplification moved from using external commercial amplifiers to being integrated within the monitors, a major step was taken toward accuracy and consistency in audio reproduction. Self-contained systems allow more precise matching of the characteristics of the transducers and the amplifiers that are driving them. Though the first attempts at this integration occurred perhaps three decades ago, these efforts have become much more sophisticated — and are mirrored in many of the professional and even consumer loudspeaker systems that are reproducing that finished mix.

Even most of the smaller professional powered monitors have a separate channel of amplification for the LF and HF drivers, and one manufacturer's top-of-the-line control room monitor has two 1100W amps for each of the low-frequency cone speakers, a 600W amp for the midrange and a 300W amp for the high-frequency driver — with a frequency response of 21Hz to 20kHz, +/-2.5dB. A competitor's four-way system provides six channels of amplification with a control panel that resembles a powered mixer.

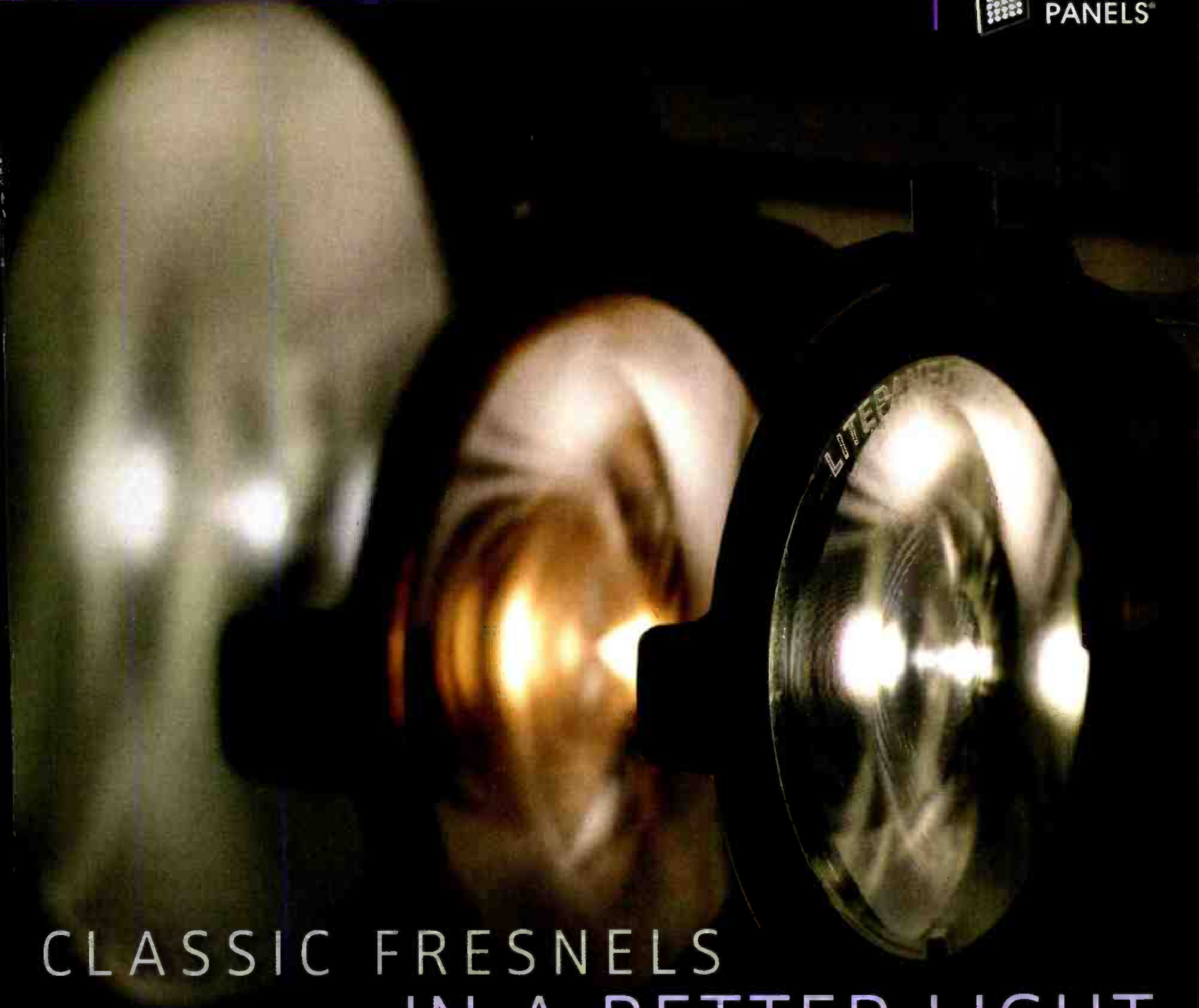
The development of Pulse Width Modulation (PWM or Class D) amplifiers with an efficiency of over 90 percent is the technology that allows such powerful amplification to be contained within relatively small enclosures with minimal heat-sinking. Coupling a channel of amplification with each driver means that their individual impedances and power requirements can be more closely matched, and the relative frequency bands and levels fed to each one makes it more effective to precisely balance and blend their outputs.

Digital Signal Processing

DSP algorithms for equalization, driver high-pass and low-pass for crossovers, delay compensation to time-align drivers within the enclosures and mains with subs, phase compensation, and other parameters are built into the circuitry and accessible via controls on the amp panel or remotely via control software. Many of the new monitors accept both analog inputs and digital signals adhering to the AES/EBU and S/PDIF standards, so the signal can remain digital from the console through the amplification. Some of the analog inputs contain precision analog-to-digital converters to process incoming signals into the digital domain.

Active crossover circuitry, in contrast to passive crossovers, responds more consistently to input signals at various levels and throughout a listening session, when the temperature of components can fluctuate — and produces less distortion. Steeper roll-off filters lessen the overlap between drivers, and the interaction of two drivers reproducing the same frequencies in differing physical locations in the enclosure. One system provides a movable 1/10th octave parametric filter to compensate for low-frequency room modes. Some monitors also include the ability to store and recall monitor settings, so that the particular parameters that are most useful for a certain room, type of mix or client can be consistently recalled. For film

LITE PANELS



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mixing, several monitors integrate a switchable Dolby X-curve setting that corresponds to the requirements for theatrical audio.

Transducer innovations

Many specialized materials technologies are used in reference monitors to achieve accurate linear response from a cone speaker or high-frequency device, and to optimize the weight-to-efficiency ratio of the transducers. Light yet rigid multi-layer cones mate with single or dual high-excursion voice coils powered by neodymium alloy magnetic structures. High-frequency drivers may be made of aluminum-magnesium or beryllium so that they are strong and rigid enough to accurately reproduce extremely high-frequency signals (even well above 20kHz) while providing extremely accurate transient response without breakup modes distorting their output.

One manufacturer specializes in inverted/concave HF dome drivers which in its experience yield an efficient, linear response with better dispersion and less "beaming" of the highest frequencies. Another's trademark is its pleated ribbon HF (and in some models, MF) drivers that achieve flat frequency and phase response, and because of their three-dimensional structure have a greater surface area than dome HF devices. These drivers often use high-flux neodymium magnets for increased output.

Stiff cones that resist breakup distortion are made from a variety of materials, ranging from a sandwich of honeycombed Nomex with Kevlar on the inner and outer surfaces, to glass fiber over a foam core, glass beads over dense paper with a specially damped surround, and so on. For precise excursion control, one manufacturer's latest models feature dual voice coils in a dual-magnetic structure. Several monitors offer time-aligned coaxial drivers, with their combined output emanating from the same location.

Advanced enclosure designs

Several monitor models have rounded edges to control the diffraction of sound waves around the cabinet and geometrically shaped baffles with integral horns to control the directivity of the transducers' outputs. The goal is to provide a flat on-axis and off-axis

ADDITIONAL RESOURCES

The following are available on the *Broadcast Engineering* website:

- Proper loudspeaker placement
- Speaker selection is important, but so too is where you place it
- Listening environment testing

frequency response, minimizing as best as possible the tendency for lower frequencies to spread more widely horizontally and vertically than higher frequencies — which can result in a less accurate sonic picture at the edge of the coverage pattern.

The baffles onto which the drivers are mounted are designed to be rigid, using cast aluminum or similar materials so that the enclosure is not excited into motion to color the frequency response. Other portions of the enclosure are similarly rigid so that they do not flex. Mounting points and feet damp any transferred vibration. To increase the low-frequency extension of these relatively small enclosures, innovative reflex ports use long internal tubes to reproduce bass without compression — sometimes with flares on the exit to minimize port noise.

Computer control and networking

Like so many other video and audio production systems, many audio reference monitors feature RJ45 connectors for networking and are supplied with control software. One model even has a USB connector to facilitate firmware updates.

The computer software allows the engineer to remotely adjust the configurations and parameters of multiple monitors and subwoofers, even across several different editing rooms. For example, the system can be remotely set for a stereo mix, LCR or 5.1 configuration. A pre-set for a particular client could be recalled and loaded, the LF or HF shelving applied or the X-curve placed across all monitors to audition the theatre mix.

Some reference monitor systems incorporate special software to perform room calibrations, using measurement microphones at the central listening position (and optionally at other locations). Test signals are sent to the individual monitors, and their frequency and phase response are measured. The software will then calculate the optimal delay settings and equalization to provide the most accurate overall response at the listening position.

Completing the mix

Like the decisions you make when selecting the best video capture, processing and monitoring equipment, finding the proper audio reference monitors for your application will enhance the quality of your production. A neutral reference lets you accurately hear the audio; detect and correct problems; evaluate changes to EQ, level and positioning; and determine the most satisfactory final mix. These innovations in audio monitoring will deliver more reliable and consistent results — and make you comfortable that the mix you hear is what you'll deliver to your viewers. **BE**

Gary Parks is a freelance writer, formerly with EV, Vega Wireless, Clear-Com and Meyer Sound.

Hitachi's SK-HD1200

The camera serves as Gearhouse Broadcast's go-to model.

BY MICHAEL GROTTICELLI

Video production companies and rental equipment houses are always careful when selecting their inventory to choose technology that will translate into a fast return on investment. When it comes to cameras, the promise of reliability and functional flexibility are certainly important. But, for the U.S. division of Gearhouse Broadcast, it needed to see it for themselves.

During production of the 2012 U.S. Open of Surfing, in Huntington Beach, CA (in July), a Gearhouse crew set up a Hitachi SK-HD1200 high-definition camera on a tripod among the wind-blown sand and punishing salt-water surf and left it running, without a camera cover, for three days.

"It's one of the worst environments to operate a camera in and, even after three days in the elements, the camera performed perfectly," said Marc Genin, Managing Director, Gearhouse Broadcast, LLC. "Needless to say, the engineers were pleasantly surprised with the camera's robustness, which is a de facto requirement in our business, capped off with excellent images."

Gearhouse Broadcast used more than 60 SK-HD1200 cameras to support its live coverage of the 2013 Australian Open tennis tournament.

As part of a global company, the U.S. headquarters (in Sun Valley, CA) of Gearhouse Broadcast operates a full-service facility that caters to live sports and entertainment production. The U.S. arm of the company has recently purchased 16 of the cameras (used mainly in 1080i/59.94fps mode, with an assortment of Canon HD lenses), with plans in place to buy several more by the end of the year. It uses them for a wide variety of live

"It's one of the worst environments to operate a camera in and, even after three days in the elements, the camera performed perfectly."

sporting events, studio shows and music events. Basically, when a client does not request another brand of camera, the SK-HD1200 is the company's new highest end go-to model.

"The Hitachi has turned out to be a rather interesting camera," Genin said. "As a global company, we are always looking for new cameras that perform well and offer our clients good value for the money. We have to buy what our customers want, but we also are interested in what we think is right for the job. We think the Hitachi holds up very well."

The camera is also used extensively by the "Home & Family" show, a long-standing live magazine show that is seen on the Hallmark channel by Gearhouse Broadcast in Los Angeles for 10 months of the year. Beginning this past January, the crews have left the cameras running through the month of July.

"The shader is old school and thought it best to leave the cameras on," Genin said. Considering, the cameras were new out of the box, switched on and left in the studio, "Not one camera has come back to us with any major issues since."

Features

Among the camera's many features, Genin's team particularly likes the cameras' head weight at 4.8lbs ("about a pound lighter than other cameras, yet its body is all metal"), and it's dockable recorder back, which allows the company to use it for a P2 job one day and an XDCAM project the next. It can also choose to operate the camera over a triax or fiber transmission path.

"We've already bought the head. Why not make the most advantage of it?" Genin said, adding that the camera also carries more power in the head than others, so that it can accommodate more accessories (like multiple LCD viewfinders, lights, etc.).

The camera also provides a low signal-to-noise (>61dB) HDTV image using three 2.3 million pixel, micro-lens array, native 1080p CCD imagers. Different DSP ICs are used independently for the HDTV camera head processing, the transmission system and the Camera Control Unit (CCU) processing. In addition, new power-efficient digital signal processor LSI chips offer dynamic processing in excess of 38-bits per pixel, per RGB channel. It also features motorized and remotely controlled optical filters and 16-bit A/D converters.

"The picture quality is incredible and not very noisy at all," Genin said. "Our internal tests have shown that the blacks are excellent and the low-light capabilities are superb. We don't just look at price. We want a quality camera that stands up to the elements. That's the priority. The Hitachi camera is basically bullet-proof."

Perhaps the most "refreshing" thing for Gearhouse Broadcast, Genin said, is that customers who use other Japanese-oriented cameras are immediately familiar with the menu structure of the SK-HD1200. It has never had a customer return one or say they need extra training.

Top-notch performance

More than 60 of the cameras were used by the Australian division of Gearhouse Broadcast to support live coverage of the 2013 Australian Open tennis tournament. The cameras were on hand once again for this summer's Vans U.S. Open of Surfing (July 20-28), where they delivered the same top-notch performance as they did at last year's event. **BE**

Michael Grotticelli is an online editor for Broadcast Engineering.

? Send questions and comments to: editor@broadcastengineering.com

Leveraging LTFS

A new file system allows broadcasters to take charge of their content.

BY RAJ PATEL

Since its launch in 2010, LTFS has attracted growing interest from systems developers working on behalf of broadcasters and media companies. It offers security and longevity for archive storage. Perhaps more importantly, it allows designers to put the archive at the center of the asset management system, playing an active role, rather than something added at the end of the process and requiring additional steps to store and retrieve content.

This paper briefly describes LTFS, explains why it is relevant to and valuable for media organizations, and outlines some of the ways in which it can be used in new workflows.

Linear Tape File System

Most will be familiar with the series of LTO archive tape formats. These were originally developed by Hewlett-Packard, IBM and Seagate (now Quantum) as an open archive standard. LTO stands for Linear Tape – Open. The three founders created the LTO Consortium, which develops the standards and makes them openly available to any vendor.

LTO is a tape format. As tape technology advances, new versions of LTO appear. The current iteration is LTO-6, although systems using earlier versions are still in widespread use.

LTFS, although it comes from the same organization, is not a tape format but a file system: Linear Tape Filing System (LTFS). Most importantly, it is a self-describing file system, using XML to define the organization of both data and metadata on each tape.

The important point to bear in mind is that it is not a wrapper, which requires each system to understand it to access the content. The tape itself is completely self-describing.

Tapes written in the LTFS format can, therefore, be used independently of any external database or storage system. An LTFS-format tape can be taken from one system and read by any other, without problems. Content can be taken from one organization's archive and delivered to another's asset management system without the systems having to think about how to manage the transfer. For the first time, assets are truly portable. (See Figure 1.)

Some argue that it is a compromise design. The emphasis has been

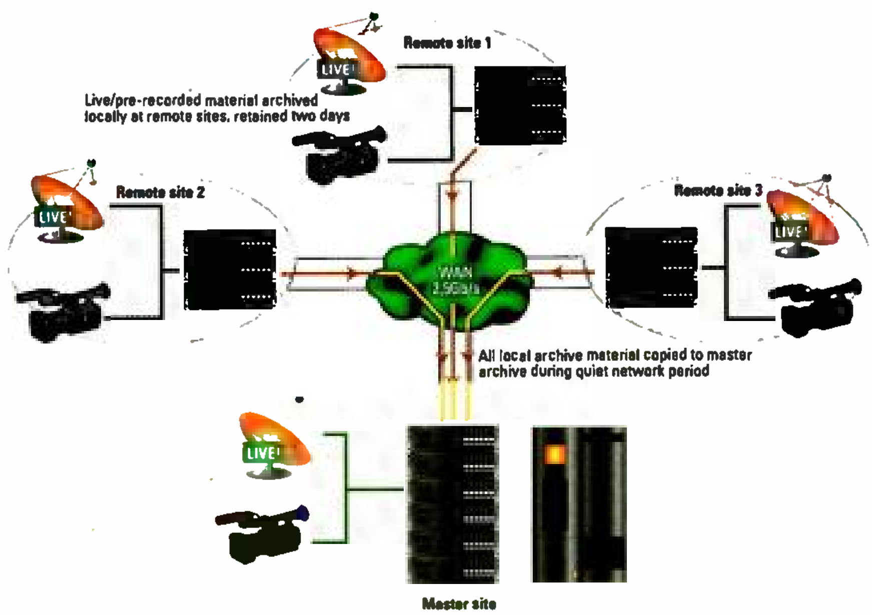

Figure 1. The strength of LTFS is its self-describing nature; an LTFS-format tape can be taken from one system and read by any other.

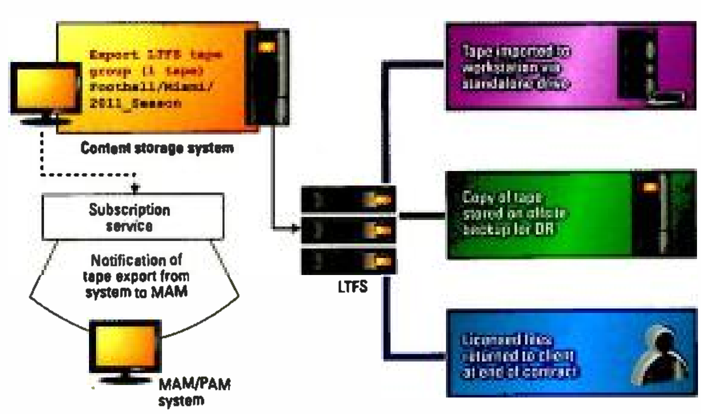

Figure 2. The ability of LTFS devices to be written to using any file management tool running on Windows, Linux or OS X, is an advantage in delivering materials to clients.

on the ability to take a tape from one system and read it in another. It means, for instance, that LTFS tapes cannot include permissions and ownerships: All content is accessed by the root. For media applications, which tend to have their own quite complex structure of access control, this is not a problem.

LTFS in the media industry

Although LTFS is an IT standard, the LTO Consortium chose to launch the new file system at NAB in April 2010. Broadcast and media applications have been seen as important from the start.

The headline benefits include portability and extreme resilience. Because LTFS is a self-describing file system, each tape is a complete and self-contained system of assets and related metadata. A tape can be written in one archive system and will be immediately readable in another. So rapid exchange of content is hugely simplified.

There is no need for translation utilities to go from one archive or content management system to another. The index information and metadata on the LTFS tape automatically populate the database receiving the tape.

Nor do you need specialized software to write the tape. Any file management tool, running on Windows, Linux or OS X, can write to an LTFS device in exactly the way it would write to a USB stick or a removable disk drive, as shown in Figure 2.

One of the key concerns for media enterprises is the ability to maximize the value of assets by monetizing content quickly and cost-effectively. The ability to deliver materials in a standardized format, without downstream processing to create a client-specific deliverable, is a major advantage.

The same advantage of portability makes it equally well suited to disaster recovery. We talk so often about "asset management" that it can be easy to forget we really mean that content is an asset to the business, which if lost or destroyed could have a catastrophic effect on its ability to continue. Content that is archived to LTFS can be reconstructed on a completely different system, even if the original asset management is completely destroyed. So LTFS represents an excellent archive medium, which is inherently future-proof and adds an extra layer of resilience over the already highly reliable LTO format by virtue of its portability. It also has the potential of providing a simple way of delivering content, particularly where there are multiple elements rather than just a finished program.

LTFS and new workflows

Consider sports production. The typical live sports event generates a huge amount of content, and an equally huge amount of descriptive metadata from the logging team marking each play and each interesting shot.

During the event, the fast-turnaround team is creating highlights packages, which in turn become more content and metadata to be logged. But you do not know until after the event which parts of the content will be relevant or valuable. It may be some considerable time later that the significance of a player's actions in one game become clear because of what happens in another.

The solution is to write the content, together with the logging metadata, to LTFS tape using a standalone drive, on location and as it happens. All of the material is preserved, so there is no worry about losing a vital sequence during a game or even a season.

The portability of the tapes means that they can easily be transported back to the broadcaster, the production company or the sport's body, or indeed all three. The outside broadcast company does not need to worry about different clients needing material in different formats. Anyone can read an LTFS tape, and an intelligent archive control system can store the material instantly without the need for further processing.

That intelligent archive could also generate proxies so that the content is instantly available to all who need it. The logging data is automatically transferred to the asset management system.

The same process could also be used in reverse. A large amount of archive content could be sent to the location on LTFS tapes. Should a situation arise where it would be good to use archive footage for context, it could be recovered without the need for a dedicated asset management system, effectively by dragging and dropping from the LTFS tape.

FIMS

The Framework for Interoperative Media Systems (FIMS), the joint development from AMWA and EBU, is seen as a significant enabling technology in the move toward service-oriented architectures and largely automated workflows. It is dependent upon open systems that respond predictably to standardized messages, allowing content to be pushed around a large-scale system.

LTFS is a natural companion to FIMS. Archive systems using LTFS can sit on the FIMS network, making it part of the live content repository rather than just a back-end archive. It is a simple step to put all the enterprise's

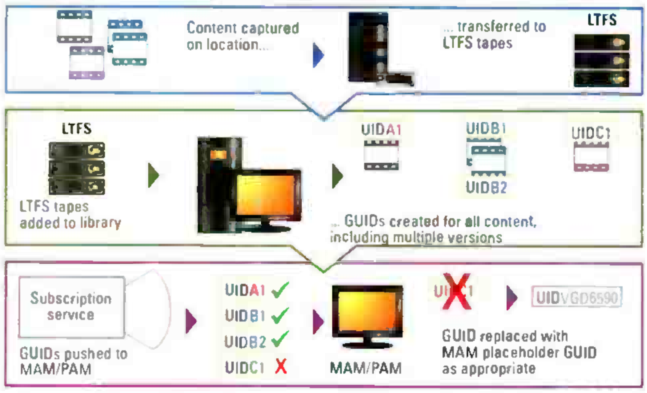

Figure 3. Using LTFS changes the archive process by allowing tapes to be written anywhere, archive-ready, and for the content to be easily exported.

content online at all times, without the need for offline search-and-restore processes.

Future of LTFS

To meet the challenges of more content, more deliverables and tighter budgets, the broadcast and media industry has to seize the advantages offered by off-the-shelf IT solutions. The scale of the IT market — with the consequent large R&D budgets and rapid commoditization — means that standard solutions will deliver better value, better reliability and better future evolution.

LTFS meets this model. It offers the media enterprise an architecture whereby the archive is also the primary source for content, accessible throughout the workflow.

In the future, as service-oriented architectures become more prevalent, many elements of the infrastructure will be replaced as required, rather than re-engineering the complete system. With content stored using LTFS, it will be practical to replace the asset management system as part of this ad-hoc upgrade cycle: The structure of the archive is not "owned" by a particular database.

Rather, you "own" the content, without reliance on a third-party development or application. You can take your content anywhere, confident that the essence and the metadata will be accessible by any system that has a tape drive.

Content can be written to, and read from, LTFS devices on standalone drives or large robotic libraries via multiple operating systems. Content can come and go simply by using the appropriate hardware for the task in hand. Ingest need no longer be centralized, as tapes can be written anywhere ready for the archive, as shown in Figure 3. Content can equally easily be exported for delivery to third parties or for "assets-to-go" workflows.

LTFS is being widely adopted in other industries, ensuring its success. Its adoption in the media industry will deliver simple interoperability, high resilience and a solid future-proofing strategy. BE

Raj Patel is product manager at SGL.

ONLINE For more information on these products, go to www.broadcastengineering.com.

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www.snellgroup.com

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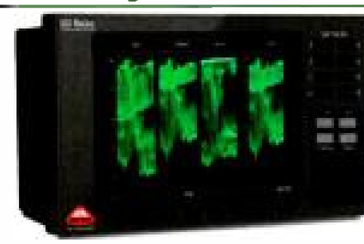

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WEIGEL BROADCASTING CO has an immediate opening for a SYSTEMS ENGINEER. The ideal candidate will have proven experience with large scale video storage and asset management systems, Grass Valley or Harris Video Servers, file transcoders, iNews systems, broadcast systems integration, and automated transmission playout and production equipment. Direct experience with Grass Valley Stratus and Telestream Vantage systems a plus. A solid understanding networking concepts is required. Candidates with knowledge of scripting and programming knowledge a plus but not required. Must have a minimum of 5 years experience in broadcast or cable network engineering. This is a F/T, salary position. Work schedule includes one day per weekend and part of an on-call rotation schedule.

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If interested, please send your resume and cover letter to:

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