FEBRUARY 2013

Broadcast Engineering®

www.broadcastengineering.com

A PENTON MEDIA PUBLICATION

SPECIAL REPORT: STORE AND MANAGE VIDEO

ALSO INSIDE:

UNDERSTANDING ETHERNET AND IP
Built on OSI hierarchy, the difference is layers

VIDEO AND DATA CONVERSION
Changing signals from A to D and from E to O

SATELLITE SYSTEMS
New Ka-band and IP-over-satellite solutions

VOLUME 55 | NUMBER 2 | FEBRUARY 2013

BroadcastEngineering®

SPECIAL REPORT

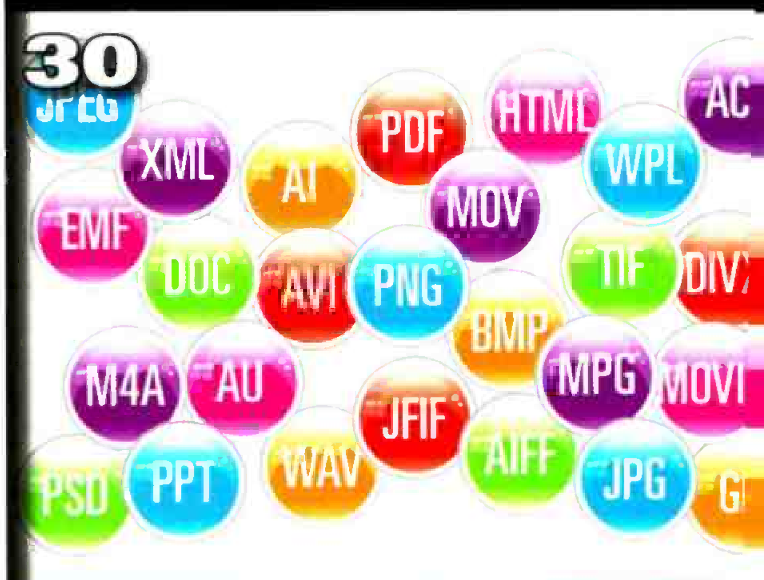
BEYOND THE HEADLINES

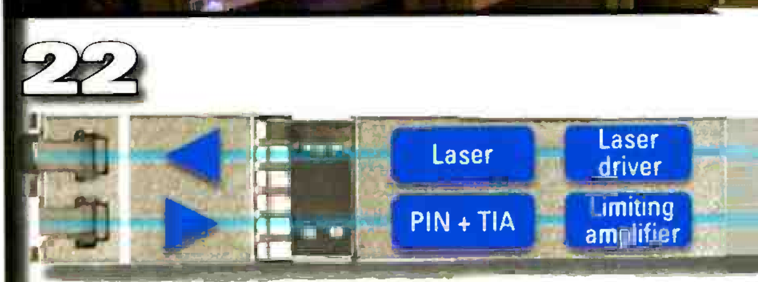
DIGITAL HANDBOOK

continued on page 6

ON THE COVER:

Asset management, storage and IP networking must be considered when planning for the future.





DIGITAL EDITION EXCLUSIVE FEATURES!

We're adding new interactive features to our digital edition! Look for icons in the table of contents to find out which articles have these special functions.

Innovation in the
Multi Screen World

snell

snellgroup.com/ICE

Automation & Media Management

VOLUME 55 | NUMBER 2 | FEBRUARY 2013

SYSTEMS INTEGRATION

NEW PRODUCTS & REVIEWS

DEPARTMENTS

LATEST NEWS!

A new group of broadcasters — calling itself the Expanding Opportunities for Broadcasters Coalition — has told the FCC it has 39 stations in the top 12 markets that will participate in the upcoming wireless spectrum auctions.

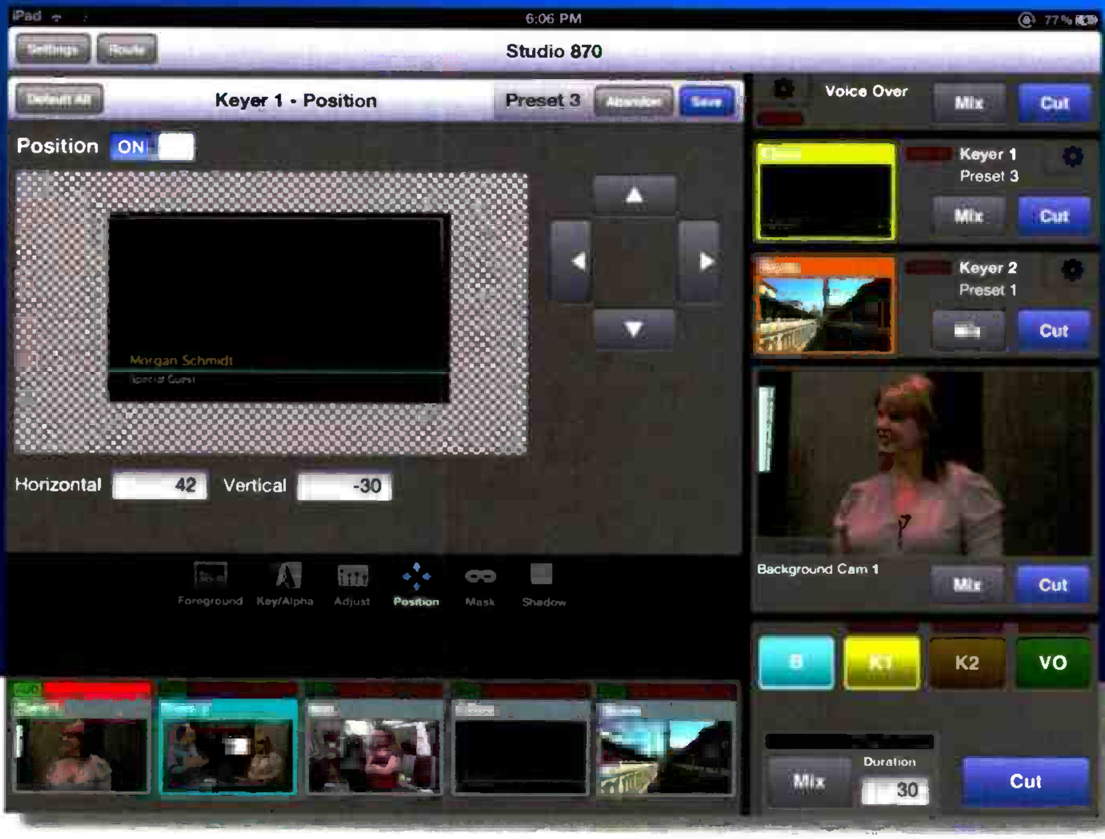
Learn more at *www.broadcastengineering.com*

SEE IT ONLINE!

Check out Brad On Broadcast, our editor's thoughts about the latest industry news. Are spinning disks really dying off?

Learn more at *www.broadcastengineering.com*

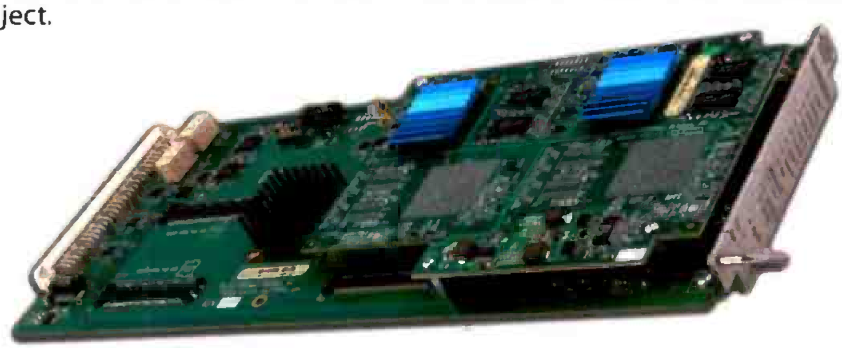
New Keying and Layering Technology



Introducing the new, award winning Avenue Layering Engine – multi layer keying that's just right for branding, small master control, flypacks and trucks. There's a frame sync on every input so go ahead and use sources that aren't genlocked. And the extraordinarily intuitive iPad™ control gives you new freedom in switching a show or event.

PICK HIT BroadcastEngineering 2012

Two, independent linear keyers, program/preset background transitions, audio mixing, voice over and breakaway, make it an agile and flexible solution for combining audio and video content. Inputs can be driven by SD, HD or 3G SDI signals from cameras, remote feeds, character generators, graphic and stillstore systems, and video servers. Realtime processing, low latency and serial control make it easy to integrate into your next project.



ENSEMBLE DESIGNS

+1 (530) 478-1830
www.ensembledesigns.com

Are spinning disks dead?

InfoWeek writer Paul Venezia wrote in his Jan. 14 column that storage would soon evolve into "... a world devoid of the painfully outdated yet ubiquitous spinning disk."

Yep, and the familiar phrase "Tape is dead" has been bandied around for 25 years. However, folks like StorageTek, Quantum, FOR-A, IBM and others make a living by selling hundreds of millions of dollars of that so-called "dead" technology every year. Content providers, including Major League Baseball, ESPN and virtually everyone with a library larger than a few thousand hours, still rely on tape for long-term storage.

I believe Venezia is too quick to write off spinning disks as effective storage mediums. He suggests it would be far better to even forgo the entire concept of "local storage." Um, I'd like to say something about that. "No friggin' way."

I've written plenty about the perils of local storage, disk crashes and data loss. But, that pales in comparison to pitching all of your content into the cloud and then realizing that you no longer have any control over where, when or how it's used. Before you store your precious new 13-part series on Amazon's EC2 or Microsoft's SkyDrive, better read the SLA. See who has access to that content and who can peek in your backroom.

Then, you might also want to read up on the litany of cloud failures. Do an Internet search of the term, and you'll develop cold shivers over the access to your cloud-based content.

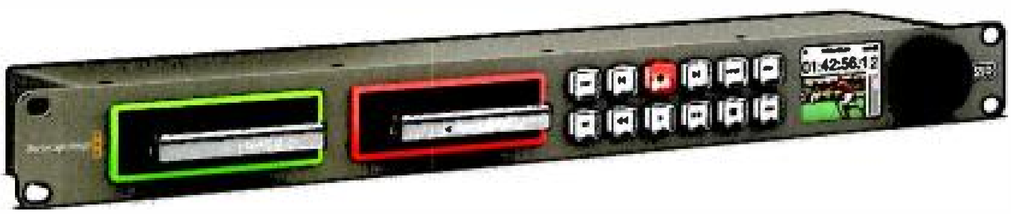
Venezia's visioned future isn't "a bunch of hot-swap disks eating up rack units and expelling a lot of heat." Instead, he believes users will rely on a "1U device that provides sufficient storage capacity, along with high performance and high reliability, and functions as a set-and-forget appliance." Hold on, wait a minute. Exactly what's behind that 1RU device that actually holds that content? Maybe a water bucket? He never fully explains. Watch my other hand.

One shouldn't be too directionally optimistic when it comes to technology. Solutions have a funny way of changing course when science offers better options. Who knows how we'll store bits 25 years from now? Certainly not some tech writer — including me.

That said, how do you think video will be stored by the time you retire? **BE**

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

Blackmagicdesign



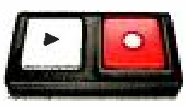
NOW WITH
ProRes
AND
DNxHD

Introducing HyperDeck Studio, the broadcast deck that uses solid state disks!

Now you can get the incredible quality broadcast deck that records and plays using low costs solid state disks. HyperDeck Studio works like a regular broadcast deck, and supports dual disk slots so the moment one disk is full, recording can continue to the next disk! Thats unlimited recording! HyperDeck Studio supports 10 bit uncompressed as well as compressed DNxHD™ and ProRes™ recording so you can simply mount the disks and edit media directly without any file copying! HyperDeck Studio is perfect for live production, instant replays and disk server use!

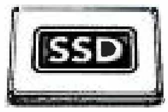
Absolute Perfect Quality

HyperDeck Studio works with both compressed and perfect quality uncompressed video. You can select between 10 bit ProRes, DNxHD MXF, DNxHD QuickTime™ and 10 bit uncompressed QuickTime recording and playback. With uncompressed you get a mathematically perfect "clone" recording between capture and playback so you can always trust your recording because there is absolutely no higher quality possible!



Familiar Broadcast Deck Design!

HyperDeck Studio has a familiar broadcast deck style design with transport buttons, jog/shuttle knob, deck control port and build in LCD! You get unlimited duration recordings because HyperDeck Studio includes dual SSD slots, so when recording, if your disk fills, it will automatically continue recording to the other disk.



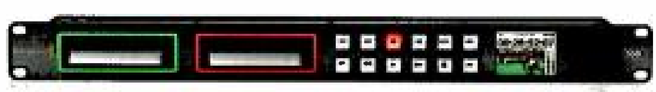
Solid State Disk (SSD) Recording

Simply plug in a fast 2.5" solid state disk into HyperDeck Studio and start recording! SSD's are used in desktop and laptop computers so prices are constantly falling while sizes are getting bigger! Plug the SSD into your computer and the disk will mount right on your computer's desktop! Files are stored in standard QuickTime or MXF file formats so you can use the media in Mac™, Windows™ and Linux software!

Use Cameras, Switchers and Monitors

With SDI and HDMI inputs and outputs, HyperDeck Studio works with virtually every camera, switcher or monitor! Plug into televisions or video projectors for instant on set preview or get exciting live action replay with ATEM production switchers. Even use it for digital signage. Just press play twice for loop playback! Imagine using pristine broadcast quality 10 bit recording on your next live event!

HyperDeck Studio
$995



Learn more today at **www.blackmagicdesign.com/hyperdeckstudio**

FIMS 1.0

A testing process is now available to the community.

BY IAN HAMILTON

The rapid pace of change in the media industry demands that leading media organizations implement efficient and agile digital media processing capabilities to control costs and capitalize quickly on new business opportunities. However, achieving the full promise of file-based media workflows and the efficiencies they bring requires seamless interaction among products from different vendors. Therein lies the challenge: Getting disparate products to interoperate is like getting a group of individuals each speaking a different language to understand each other. The effort requires standards for more than just the words or the dialogue — i.e. the file formats. Seamless interaction also requires standardizing the way the dialogues are handled — i.e. the approaches to coordinating and connecting the components of media processing systems.

Most modern businesses today leverage service-oriented architectures (SOAs) to assemble agile business systems. Indeed, SOAs offer tremendous advantages to processing media for playout and distribution. A key element of SOA involves exposing system components as shared network-based services. Connections with services can then be created and reconfigured quickly to respond to changing requirements and demand.

The Framework for Interoperable Media Services (FIMS) was initiated as a joint project between the Advanced Media Workflow Association (AMWA) and the European Broadcasting Union (EBU) to create a common framework for applying SOA principles to processing media. FIMS standardizes service interface definitions for implementing media-related operations in a SOA. As one example of how it can be used, a system that ingests and prepares content for playout and nonlinear delivery can leverage FIMS to interact flexibly with media capture, transfer and transform products from multiple vendors. Using FIMS, system components can easily be added, updated and removed in response to changing business requirements and demand.

The problem

Although workflows are commonplace for moving, processing and storing media using software-centric systems deployed on commodity IT infrastructure, many software-based media processing systems suffer from problems endemic to older hardware and physical media-centric film and video processing systems. Whenever unique media processing requirements must be addressed — and let's face it, what broadcaster does not have a unique requirement of some sort — there's an expensive custom software integration project needed to delivered bespoke hardwired media processing silos.

Worse yet, many software-based systems use watch folders to hand off media between components of the system. While watch systems can allow multivendor interoperability and loose coupling, they create a whole new set of quality, reliability and management problems.

The approach

FIMS focuses on the abstract service layer that connects applications and orchestration systems with services that perform operations on media like capture, transform and transfer. (See Figure 1.) FIMS addresses media

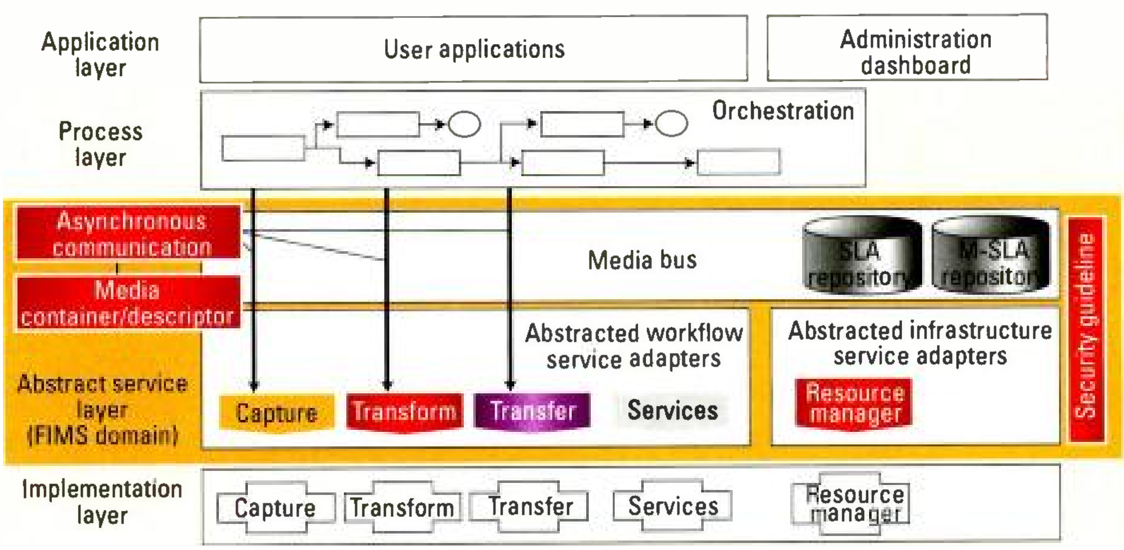


Figure 1. FIMS reference model

specific requirements associated with the resource-intensive nature of processing and storing media.

For example, FIMS accommodates long-running processes by incorporating an asynchronous calling pattern and resource prioritization through queuing. The FIMS 1.0 specification defines services interfaces for capturing, transforming and transferring media, with ongoing development efforts under way to define additional services. With FIMS, a workflow that receives media, transcodes it into the required formats and then transfers the result for linear playout and nonlinear distribution can easily be automated through standards-based service orchestration.

To those unfamiliar with SOA and Web services, the breadth of products and technologies marketed by big IT vendors can be intimidating and overshadow the underlying principles. The FIMS 1.0 specification avoids much of this complexity by defining a resource model as a central component of interaction between service providers and consumers. The resource model approach places more emphasis on what is being communicated between the service consumer and provider than on how it is being communicated. This allows a neutral position on Web services technical debates such as those concerning SOAP versus Representation State Transfer (REST).

The main resources in the FIMS 1.0 model are Services, Jobs and Media Objects. Jobs are processed by Services to perform operations on Media Objects. In the context of the resource model, the Media Objects are formally called Business Media Objects. This qualification is made because these objects only contain the metadata relevant to the business operation being performed along with information about how to access actual media content.

FIMS 1.0 also makes a clear distinction between Media Objects on the message bus and the media bus. The term bus is borrowed from computer architecture where a bus is a subsystem that transfers data between components of a system. The message bus is used to communicate information about media, including media processing instructions. The media bus is used to store, access and move master, mezzanine and finished media products. This model is analogous to the management of physical products where business systems track raw materials, components and inventory for associated real-world objects that are stored and transported using warehouse, ships, planes and trucks.

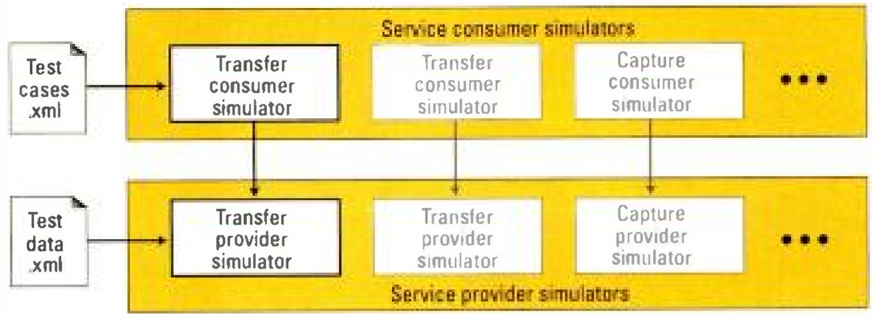


Figure 2. FIMS test harness components

FIMS Test Harness Project

A practical approach to standardization requires more than object models and interface definitions to gain serious traction. Consequently, the FIMS Technical Board recognized the need for a tool to help FIMS implementers validate their implementations. Signiant did as well, given the test-driven development methodology we use to develop software. Using such an approach, developers create and code tests to validate an implementation before coding it. This leads to higher quality products that can be evolved quickly.

Signiant initiated the FIMS Test Harness Project within the FIMS Technical Board and helped design and build a test harness for contribution to the FIMS effort. The ultimate goal of the test harness is to remove barriers to FIMS adoption and promote implementation of FIMS-compliant interfaces.

Through discussion among the technical board, it was agreed that the test harness should consist of a set of service consumer and service provider simulators. (See Figure 2.) Doing so facilitates interaction between third-party products, which is a key benefit of FIMS. The simulators allow FIMS implementers to independently test their service consumer or provider implementation. These independent tests give vendors confidence that their products will interact properly in real-world conditions.

Completely data-driven testing was another goal of the test harness project. As such, test cases and associated service behaviors are specified with test configuration data. Service consumer configurations specify a sequence of commands to invoke against a service provider. Service provider configuration specifies how the service responds to specific commands with mock behavior. This approach helps ensure that no coding is required to use the test harness and create new test scenarios.

In addition, the test harness project sought to leverage an existing Web services test framework. Broad availability of standards-based testing tools is a significant benefit of a standards-based SOA approach. Through a series of proposals and discussions in the technical board, the decision was made to use the soapUI test framework. SoapUI is an open-source Web service testing application for SOAs. The functionality of soapUI covers Web service inspection, invoking, simulation and mocking, functional testing, and compliance testing. As such, the soapUI framework covers most of the service consumer behavior, and new code was only required to implement mock service provider behavior.

Despite its name, the soapUI framework supports both SOAP and RESTful interactions between service providers and consumers. SOAP is a remote procedure call (RPC) standards-based approach to Web services that references a stack of related standards. The REST approach is conceptually simpler than SOAP and uses standard HTTP operations to interact with resources. FIMS 1.0 formally defines SOAP bindings for services and, although the resource-based approach allows for RESTful interactions, work is ongoing in the FIMS Technical Board to further specify a REST interface. The test harness design allows a REST interface to be easily plugged in on top of existing mock service implementations.

A first version of the FIMS test harness has been contributed to the FIMS initiative for free use by the FIMS community. The FIMS test harness is an easy way to gain practical experience with FIMS, including working with FIMS calling patterns and the FIMS resource model. It provides valuable feedback to both vendors implementing FIMS and users wanting to start working with FIMS. The FIMS community is encouraged to utilize and help expand the test harness through the implementation of additional test cases and behaviors.

Toward lower costs, greater agility

The full promise of file based workflows requires seamless interaction amongst products from multiple vendors. FIMS applies SOA principles to media to facilitate an important part of this interaction. The FIMS Test Harness Project provides a practical starting point for working with FIMS. Adoption and implementation of FIMS brings the media industry a big step closer to file-based workflow's promise of lowering costs and ultimately improving business agility. BE

Ian Hamilton is co-founder and chief technology officer, Signiant, and a member of the FIMS Technical Board.

ADDITIONAL RESOURCES

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

- IPTV subscribers to double by 2017
- IP video distribution
- Emerging markets sustain global IPTV boom

Next-generation broadcast

Evolving technology brings broadcasters better sound.

BY ALDO CUGNINI

The development of new broadcast technologies that offer higher efficiencies and/or performance is an ongoing quest, with researchers continuing to push on methods to achieve a higher data throughput. The result is that the hardware and software providing media delivery will continue to evolve. On the broadcast side, modulation and compression are key components of a next-generation broadcast system.

HEVC promises to increase coding efficiency by up to 50 percent compared with AVC/H.264.

Digital modulation

Digital modulation customarily takes the form of transmitting symbols that are mapped from groups of bits. These symbols can be sent using one of a variety of modulation schemes, including VSB, QAM and COFDM. Because reception is usually limited by channel noise, interference and distortion, there is a certain carrier-to-noise (C/N) ratio that is required for error-free reception. For this reason, the symbol rate is usually limited in a channel, regardless of the number of different symbols (or constellation points) that can be transmitted.

But there are ways of pushing this limit, either by clever channel coding, or by changing the necessary transmit power. New error correction schemes are constantly being developed — pushing the envelope to its theoretical limit and making higher-order constellations like 256QAM possible. But another way to maximize channel usage is by adopting a "cellular" approach, where multiple transmitters would blanket an area, each with a lower power than a single one, but able to get a stronger signal to receivers — yielding a higher C/N ratio and thus increasing the potential throughput.

Adaptive transmission technologies are also gaining interest in the research community. The idea is to get feedback of instantaneous channel-state information from a number of receivers, and use this to dynamically modify characteristics of the transmission system — including output power; transmission architecture (i.e., number and individual power of active transmitters); and even modulation parameters. Any one of these systems would represent a radical change in broadcasting infrastructure and federal regulations, but could potentially generate a sizeable increase in available bandwidth.

Video compression

As you read this, the latest MPEG compression specification, High-Efficiency Video Coding (HEVC), was due to be released as a final draft international standard, called MPEG-H Part 2 and ITU-T H.HEVC (and sometimes referred to as H.265). HEVC promises to increase coding efficiency by up to 50 percent compared with AVC/H.264.

The HEVC specification is organized into profiles, tiers and levels, allowing equipment of varying complexity and cost to fit into well-defined architectures. As with its MPEG predecessors, an HEVC profile is a subset of the entire bit-stream syntax, and a level is a specified set of constraints imposed on values of the syntax elements in the bit stream. The constraints may be limits on values, or arithmetic combinations of values (e.g., number of pixels per frame multiplied by number of frames per second). Currently, three HEVC profiles have been established: Main, Main 10 (with up to 10 bits per color), and Main Still Picture.

HEVC introduces the concept of tiers within each profile, which carry their own constraints; tiers were established to deal with applications that differ in maximum bit rate. A level specified for a lower tier is more constrained than a level specified for a higher tier, and levels are similarly constrained within the tiers.

Audio compression

High-Efficiency Advanced Audio Coding (HE-AAC), as shown in Figure 1 on page 16, has already been adopted into various transmission systems, including ATSC, DVB and ISDB. HE-AAC is based on the MPEG-2 AAC Low-Complexity profile in combination with Spectral Band Replication (SBR). MPEG-2 AAC is non-backwards-compatible with the other parts of MPEG-2 and MPEG-1 audio, including the so-called MP-3 format.

These earlier MPEG codecs, as well as AAC, use a high-level structure that combines a psychoacoustic


vimeo PRO
JOIN THE RESOLUTION
Be part of the world's leading network of creative film professionals and showcase your work exactly how you want
JOIN vimeo PRO NOW
VIMEO.COM/JOINPRO

model, spectral processing and quantization. MPEG-1 and MPEG-2 spectral processing used mapped samples that are either sub-band samples or DCT-transformed sub-band samples. A psychoacoustic model creates a set of data used to control the quantizer and coding, wherein the production of coding artifacts is modeled at the encoder. The model relies on characteristics of the human auditory system, whereby quiet sounds within "critical bands" are masked by louder sounds.

MPEG-2 AAC also uses a psychoacoustic model, spectral processing and quantization, but with newer tools. A modified discrete cosine transform (MDCT) is used in the spectral processing step, and noiseless coding is performed on the spectral data.

Used in HE-AAC, SBR is a tool that can save coding bandwidth by allowing the codec to transmit a bandwidth-limited audio signal. The process is based on a decoder resynthesis of the missing sequences of harmonics, previously truncated in order to reduce the data rate. A series of tonal and noise-like signal components is controlled by adaptive inverse filtering, as well as the optional addition of noise and sinusoids. Side-data in the bit stream can also be transmitted in order to assist in the reconstruction process. Typical data rates used for HE-AAC are 48kb/s to 64kb/s for stereo and 160kb/s for 5.1 channel signals.

HE-AAC version 2 (v2) is a combination of HE-AAC and Parametric Stereo (PS). The PS coding tool captures the stereo image of the audio input signal with a small number of parameters, requiring only a small data overhead. Together with a monaural downmix of the stereo input signal generated by the tool, the decoder is able to regenerate the stereo signal. The PS tool is intended for low bit rates.

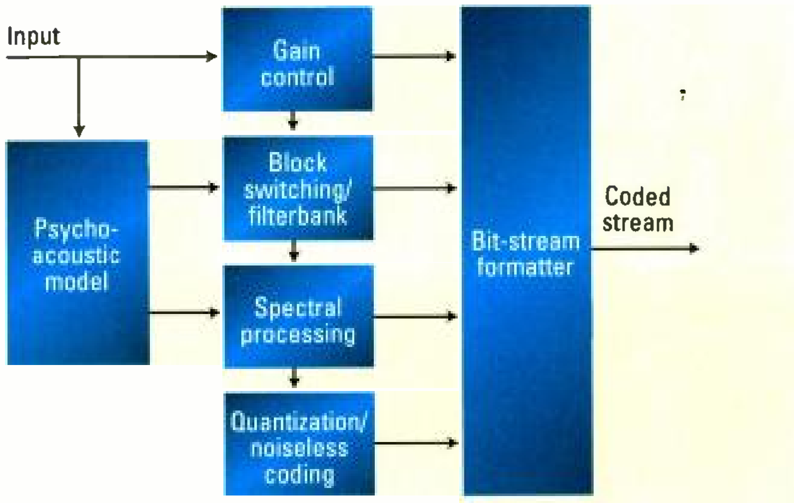


Figure 1. MPEG Advanced Audio Coding incorporates elements from earlier MPEG standards.

Spatial Audio Object Coding (SAOC), is a recent MPEG-D coding tool (which can be implemented independently), wherein the different sound sources within a transmission are treated as separate coding "objects." MPEG SAOC for Personalized Broadcast Systems is one potential application that can provide audio-related interactivity to consumers. Interactivity for broadcast applications can enable users to adjust TV sound according to individual preferences, or can even offer a hearing-impaired audio service without an additional audio channel.

Broadcasting with SAOC could allow users to adjust the volume of single audio objects within movies, TV shows or even live transmissions. For example, commentator and ambient sound levels during a sports broadcast could be adjusted by the consumer, as one means to improve intelligibility to suit one's own hearing capability.

Hybrid TV is yet another area that is developing quickly. Combining a high-bandwidth, free OTA path with an Internet side channel, Hybrid TV makes new services available to home and mobile viewers. Specifications for Hybrid Broadcast Broadband TV (HbbTV) and other systems have already been published, with some trial services already deployed in Europe and Asia.

New tools to stay competitive

Transmission, modulation and compression schemes are being updated at a pace that ongoing infrastructure planning is required to stay competitive with other forms of media delivery. The challenge is in adopting new technologies with as much backward-compatibility as possible, so as to not require consumers to constantly upgrade their hardware. From that standpoint, the entire media chain would benefit from those technologies that support "soft" equipment upgrades, and that requires close cooperation between broadcasters and consumer electronic companies. 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:

- Broadcast in transition
- ATSC issues reports on next-generation broadcast TV, 3-D TV broadcast
- ATSC to explore next-generation broadcasting spec that could be incompatible with today's technology

Ethernet and IP

Both on the OSI hierarchy, their difference is in layers.

BY BRAD GILMER

Ethernet and IP are terms broadcast engineers use many times every day. But what, exactly, is the difference between the two?

Layers

Many years ago, manufacturers needed to include, inside their application programs, software that was written to allow the program to interface to a specific network interface card. This meant that an application would only work with specific networking hardware; change the hardware, and you had to rewrite the application. Very quickly, vendors faced an escalating number of possible network devices and a number of different underlying physical network implementations (such as RG-11, RG-59 and UTP cable).

Manufacturers wanted to separate the development of their application from all of the chaos going on at the networking level. They also wanted to be able to sell their applications to different users who might have different networking technologies in their facilities. The seven-layer Open System Interconnection (OSI) data model was developed to address this issue in a standardized way. While it is not too important to understand all layers of the OSI model, it is good to understand the first four layers. (See Figure 1.)

Layer 1 is the physical layer, sometimes referred to as PHY. This is the physical network hardware, and it operates at the bit level. Layer 2 is called the data link layer and consists of a mixture of specifications describing the electrical signals on a cable and the way that data is encapsulated into frames. Ethernet is a Layer 2 protocol. Layer 3 is referred to as the network layer, and it is here that data is encapsulated into packets. IP packets are referred to as datagrams. Layer 4 is the transport layer. Here, we speak in terms of sessions. Transmission Control Protocol (TCP) and User Datagram Protocol (UDP) are Layer 4 protocols.

Each of these layers plays a role in networking. Importantly, Layer 2 is responsible for physical addressing, meaning that the network architecture can rely on the fact that a Layer 2 address is unique and permanently assigned to a specific piece of hardware. Layer 3 is responsible for logical addressing, meaning that addresses are assigned at this level by a network engineer in a way that organizes network clients into logical groups. But, neither of these two layers are responsible for anything more than "best effort" delivery of packets from one place to another. If the packets get lost, duplicated, rearranged or corrupted, neither Layer 2 nor Layer 3 will do anything about it. Layer 4 protocols are responsible for establishing end-to-end connections called sessions, and these protocols may or may not recognize the loss of packets and do something about it. TCP, for example, will request that lost packets be resent. UDP will not. (Remember that Ethernet operates at Layer 2, and IP operates at Layer 3.)

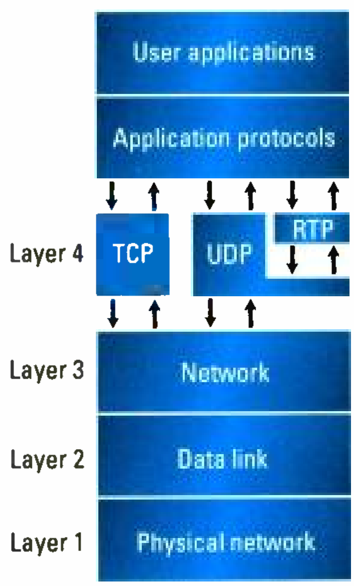


Figure 1. This shows the first four of seven layers to the OSI data model.

A hierarchy

Ethernet and IP are part of a hierarchy of interchangeable technologies in the seven-layer OSI stack. While most of the networks that broadcast engineers are likely to encounter today are Ethernet and IP, it is important to understand that, in the early days, Ethernet was just one of a large number of competing Layer 2 technologies. Other Data Link Layer technologies include ATM, Token Ring and ARCNET.

> **Most networks encountered today are Ethernet or IP. But, Ethernet once was just one of a large number of competing Layer 2 technologies.**

While Layer 2 does a great job of organizing data into frames and passing them on to a physical network, it is not capable of allowing network architects to group computers into logical networks and allowing messages from those computers to be sent (or routed) to a much larger campus or even worldwide network. This is where Layer 3 comes in. IP operating at Layer 3 organizes data into datagrams, which can be sent across any

kyrion

Accept nothing less
Your DSNG truck deserves it

Upgrade your fleet to Kyrion CM5000, the best in class contribution encoder, designed to help satellite up-link professionals navigate unexpected obstacles.

- Meets all DSNG requirements: SD & HD, MPEG-2 & MPEG-4, 4:2:0 & 4:2:2, 8-bit & 10-bit, up to 1080p60
- Complete front panel access with video monitoring
- Boots faster, whisper-quiet in operation, easy web graphic user interface
- Modular design with optional second video channel or built-in DVB-S/S2 modulator

ateme

Keep content looking great wherever it's played

Layer 2 networking technology. IP datagrams are the same size and the same format, regardless of whether these packets are sent across Ethernet, Token Ring or some other network. In fact, you might be surprised to learn that IP packets from your computer may first travel over Ethernet, then over a SONET for long-distance transmission, and then be put back into Ethernet for delivery on a local network at the destination.

IP is not the only Layer 3 protocol in common use today. There are a number of other critical protocols that operate at this layer, many of which have to do with configuring and maintaining networks. Internet Control Message Protocol (ICMP) and Open Shortest Path First (OSPF) are two examples of this.

In summary, Ethernet and IP are part of a hierarchy. IP packets are carried in Ethernet frames. IP packets that travel over long distances are likely to be carried in the payload portion of SONET frames. Furthermore, when you see the notation TCP/IP, remember that TCP is also part of this hierarchy. It is highly likely that every time you see this notation used in reference to traffic on a LAN, remember that you are actually talking about TCP over IP over Ethernet.

Other differences

There are many other differences between Ethernet and IP that are derived from the fact that they are on different layers of the OSI model.

Ethernet frames contain source and destination addresses. The frames also contain a payload area. IP packets are transported in the payload area of the Ethernet frames. The IP packets also contain source and destination addresses and a payload section. If both Ethernet and IP contain similar structures, why use Ethernet at all?

Remember that the point of adding IP as a layer on top of Ethernet is to allow the IP packet layout (and the application software above that layer) to remain constant while providing different underlying Layer 2 structures. Basically, if you change the network, then you change Layer 2 drivers, and perhaps Layer 1 hardware, but everything above that layer remains the same.

From a practical standpoint, there are significant differences between Ethernet addresses and IP addresses. Ethernet addresses are assigned to a network interface card or chip set at the factory. They are globally unique,

The first three octets of an Ethernet address convey meaning; they are an Organizationally Unique Identifier, or OUI.

cannot be changed (not true, actually, but this was the original assumption), and getting an Ethernet configuration up and running essentially is a plug-and-play process. IP addresses, on the other hand, are not assigned from the factory. These addresses need to be configured (sometimes dynamically), and while Dynamic Host Configuration Protocol (DHCP) works very well most of the time to automatically configure IP addresses, there are many cases where broadcast engineers must manually configure the IP address of a device before it can be used on the network.

Another difference is that the first three octets of an Ethernet address convey meaning; they are an Organizationally Unique Identifier (OUI). It is possible to look up an OUI and determine who assigned the Ethernet address to the hardware. IP addresses, however, have absolutely no meaning assigned to them. That is not to say that there are not special IP address ranges because there are. But, the numbers themselves do not convey any specific information.

Perhaps the most important difference between Ethernet and IP is that Ethernet frames are not routable, while IP packets are. In practical terms, what this means is that an Ethernet network is limited in terms of the number of devices that can be connected to a single network segment and the distance Ethernet frames can travel. Limits vary, but, as an example, Gigabit Ethernet on Cat 7 cable is limited to about 330ft. Gigabit Ethernet on single mode fiber (expensive) is limited to 43mi.

In order to extend the network beyond this length, you need to be able to route signals from one physical Ethernet network to another. This was one of the original design goals of IP. Network architects assign computers to IP addresses based upon physical location and corporate function (news vs. production, for example), and then IP routers automatically recognize whether packets should remain within the local network or whether they need to be routed to the Internet. BE

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

? Send questions and comments to: brad.gilmer@penton.com

+ ADDITIONAL RESOURCES +

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

- Turning a generic IP network into a video network
- Computer networks Part 1
- Computer networks Part 2

File-Based Transcoding

powered by **Kayak**®

Transcending Transcoding

Automated. Intelligent. Agile.

Digital Rapids Transcode Manager® 2.0, powered by the groundbreaking Kayak platform, blends high-volume media transformation and workflow processes so seamlessly it's almost unfair to call it transcoding. Unlock new revenue opportunities and reduce your costs with streamlined, intelligent workflows.

- Superior quality, flexibility and scalability
- Adaptive, logic-driven process automation maximizes efficiency and reduces errors
- Intuitive, visual workflow design tools
- Powerful Kayak platform enables easy integration of new technologies

Visit our website to learn more: www.digitalrapids.com/tm2

NABSHOW® *Where Content Comes to Life*

See us in booth SL5624

digital rapids®

www.digitalrapids.com

905-946-9666 x212

Connecting Content to Opportunity

Encoding • Transcoding • Workflow • Streaming • Broadcast

Video and signal conversion, part 1

Converters must keep pace with video format changes.

BY RENAUD LAVOIE

In previous articles, I have covered multiviewers, routers and some specific topics about AC-coupling capacitors, the evolution of SFP modules and fiber-optic theory. One important aspect I haven't covered yet is conversion.

In a perfect world, interfaces and standards would interact fluidly and perfectly. Unfortunately, this is not the case, and with history as a guide, new advances in technology will perpetuate this situation. A key component requirement is that an SFP enable absolute interoperability. A simple SFP cage allows easy interchange between multiple modules that bridge a wide number of standards.

This article is dedicated to the function of conversion, a broad topic. There are many options for these functions. There are rack-based products, card-based products, dongle products and, of course, SFP products.

Because this topic has great depth, it will be divided into two parts. This first section includes an introduction to signal conversion as opposed to format/video conversion. We'll look at techniques to convert between electrical and optical signals. This article will also review basic video standards conversion techniques.

In part two, we'll examine implementations of graphic format converters; VGA, DVI, HDMI and others; as well as the familiar up/down/cross-conversion systems.

Diversity

There are so many options for converter features, performance and packaging that it is impossible to describe all the functions and details in one article. A quick market survey would indicate that users prefer modular, or glue, products. While they can require time to reconfigure cables and firmware settings, they are easier to upgrade than rack-based products and usually cost less as well. As requirements change for video production and processing, the features for conversion can change as well. Upgrading to better features and functions is a never-ending process; nothing is ever perfect.

Optical converter

The optical converter is usually targeted towards video signals running at 270Mb/s, 1.5Gb/s and 3Gb/s, and soon 6Gb/s and 11.88Gb/s (10G-SDI), but they can support data rates from 15Mb/s to 11.88Gb/s depending on configuration of the internal signal processing blocks. (See Figure 1.)

This is the simplest optical-to-electrical and electrical-to-optical converter. The basic blocks for the electrical-to-optical converter are:

- *Laser driver.* Electrical digital signals feed the laser driver generating output currents that bias the laser for the correct average power output and modulate the laser to generate the digital 1s and 0s. The video pathological signal has a large DC component, so it is crucial to have a laser driver that supports this unbalanced power pattern. (See Figure 2.)
- *Laser or LED.* This device converts electrical currents into photons, or light. Optical output power is in direct proportion to the current level. An article covering the detailed physics of this conversion will be published later in *Broadcast Engineering.*

For the optical-to-electrical converter, the blocks are:

- *Photodiode.* It receives the photons and converts them to an electrical current output.
- *Trans-impedance amplifier (TIA).* It converts the photodiode current to a voltage output, which feeds a limiting amplifier. The photodiode current is small; it might only be in the pico-amp range. The TIA converts this current to a voltage and amplifies the signal, but

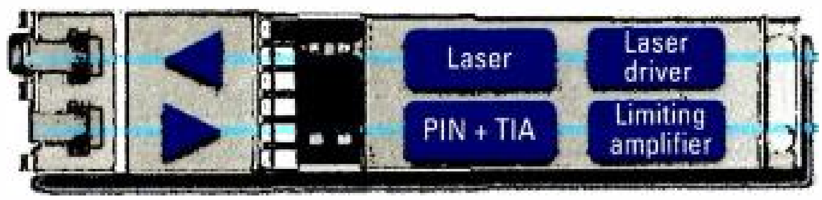


Figure 1. Electrical-optical transceiver–converter. Images courtesy Embrionix

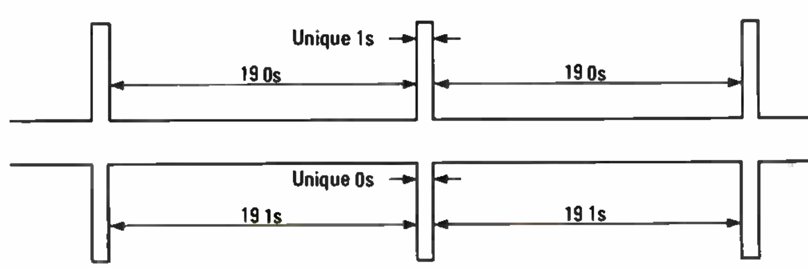


Figure 2. Pathological signal (See "Understanding blocking capacitor effects," *Broadcast Engineering* magazine, August 2011)

the gain is designed so TIA output is distortion-free. The high-gain limiting amplifier converts the low-level TIA output into digital 1s and 0s.

This processing is necessary to ensure distortion-free operation even with the video pathological signal. To reduce jitter, advanced optical converters integrate reclockers just before the laser driver or right after the limiting amplifier. This reduces jitter in order to achieve the best signal quality and the lowest possible bit error rate.

Diagnostic functions such as jitter measurement, or eye diagram analysis, can be included along with image and signal processing capability. These advanced features are available as options in some SFPs, providing engineers nonintrusive, real-time signal performance monitoring and status at the edge of their signal distribution network. Engineers can view all the signals in the studio and know exactly the point where any failure might occur.

Video standards converters

Video standards converters cover a broad range of formats: NTSC, PAL, interlaced and progressive, and computer graphics formats in addition to various signal transports such as DVI, HDMI, USB, DisplayPort and Thunderbolt. And even this list is a subset of existing equipment. Technology advances and the convergence of the worldwide video market have made it necessary to repurpose video assets that could have originated from any part of the world, from a video camera or a computer, for an application that is going to a different part of the world, being viewed by a different monitor, being stored on a computer disc, or any combination thereof.

One good example is the NTSC or PAL composite video baseband signal (CVBS) to digital decoder, sometimes called a video A/D converter, or simply a converter. The CVBS format has been used for more years than other current formats, decades before SDI, or any other digital format. The original digital transition started nearly 20 years ago, but some studios and production facilities still have large numbers of CVBS feeds, signals and tape assets.

Today, nearly every video signal is distributed digitally to the final viewer. Video production is digital, and playout is digital. But these legacy CVBS signals and tape assets must be converted to digital for compatibility with the new digital infrastructure.

What parameters are important in an analog to digital converter? Based on the application, some are more important than others. Typically, choices are made for resolution (8, 10 or 12 bits), linearity (differential gain and

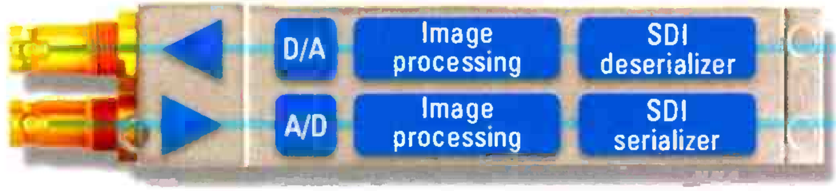


Figure 3. CVBS A/D and D/A converter with SDI I/O

differential phase), chroma, luma, gain, hue and brightness are important in nearly every application to preserve the original picture quality. A long or faulty coaxial cable can seriously degrade CVBS signal picture quality as can poor filters and converter chips. Today, digital transmission for video creates far fewer picture artifacts than analog modulation did. But, accurate anti-alias filters, linear converters and little or no differential delay between luma and chroma are still requirements for high-quality on-air pictures.

Today's semiconductor technology provides really small converters. For example, one SFP for CVBS includes two independent converters in a single SFP package. (See Figure 3.) These fully featured converter chips integrate more features into less space with less power than some modular products and card-based converters.

The basic blocks are:

- *A\D converter*. This critical block includes analog signal filtering. Resolution, linearity and analog noise are also controlled with this chip. A poorly designed semiconductor has a negative impact on overall performance. Composite NTSC or PAL conversion is still a common function. While component RGB signals are still used, computer graphics formats have rapidly diminished their use. Simultaneously, DVI, HDMI and other digital video-specific transports have replaced the three coaxial cables of RGB systems with a single cable, albeit with multiple twisted pairs inside.
- *Image processing*. This block could accomplish functions such as audio de-embedding, audio analysis, video analysis, time base correction, etc.
- *SDI serializer*. This block converts the parallel data from the A/D to serial data. It receives 10-bit or 20-bit video words as well as necessary synchronization and ancillary information such as audio data, or signal-specific metadata such as TRS, EAV and SAV. It adds framing data and converts the parallel data to serial. As the serial data is clocked out, a data scrambler is used to reduce DC content, and then NRZI channel encoding is used to ensure that data is invertible.
- *D/A converter*. This block receives the digital data in parallel in addition to video timing and framing signals. It converts it to an analog composite or component signal. It could use 8, 10 or 12 bits of resolution. The video D/A conversion, or encoding, is far easier to implement than the A/D, so this block is less critical than in the decoder.
- *SDI deserializer*. This is the opposite of the serializer. It receives the serial signal, recovers framing, deserializes the data and outputs the data in parallel words (10 or 20 bits). The sophistication and miniaturization of semiconductors allows this block to de-embed audio signals, provide output sync signals, analyze for CRC errors, extract other metadata and provide status flags and information about the video signal itself.

This concludes part one of this series. Be sure to visit the Web version of this article as it links to previous digital video signal tutorials. The second part of this article will appear in the April issue of *Broadcast Engineering*. BE

Renaud Lavoie is president and CEO of Embrionix.

+ ADDITIONAL RESOURCES +

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

- Video processing
- 3-D standards conversion
- Signal conversion

The IP revolution reaches playout

SDI still has its place, but change is coming.

BY SARA KUDRLE AND MICHEL PROULX

Television facilities are gradually evolving and are becoming more IP/networking-centric in many areas. IP networking infrastructure has been built up throughout facilities primarily to handle the exchange of files from one storage device or file-based system to another. Files have been successfully navigating the IP domain within television and playout facilities for some time.

IP networking makes perfect sense for file-based processes, but what about real-time program streams? There are many areas where real-time or linear program streams are in use. (See Figure 1.) Today, we commonly find IP networking used for real-time program streams on the edges of the facility, typically in the incoming feeds area or at the point of outbound transmission. At these points, the stream is encoded and compressed for transport. Take that farther, and here is the killer question: Could IP networking be used throughout a facility?

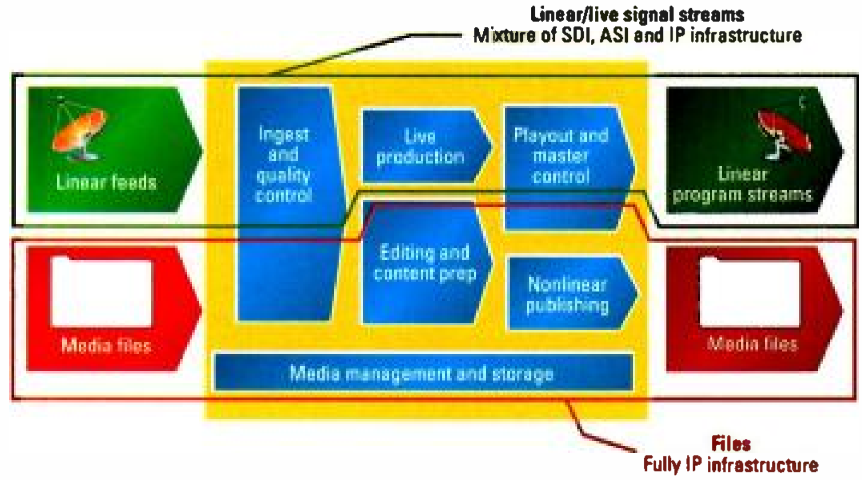


Figure 1. In many areas today, real-time or linear program streams are used, resembling a model like this one.

Headends take an IP plunge

Anyone unsure about the suitability of IP networking infrastructure to carry real-time program streams need only look to the radical transformation that has occurred in cable, DTH and IPTV headends in the past 10 years. (See Figure 2.) In the beginning, these headends would accept a combination of compressed and uncompressed signals, and employed a combination of SDI and ASI infrastructures. Now, these headends have transitioned to using IP networking infrastructures for several reasons. Namely, the signals they transport are now compressed when they arrive and remain so either all the way to the home or, for cable systems that still support analog tiers, are decoded close to the home where the signal is modulated to RF. But, why did TV delivery headends transition to IP?

With the functional integration of devices leading to the development and proliferation of compressed domain splicer systems, the transition to IP was facilitated. Today, traditional MPEG decoders and encoders are being replaced by integrated systems that feature IP inputs and outputs, and combine ad and graphics insertion and transcoding from one compression format to another, or from one bit rate to another, inside a single box. With these integrated transcoder systems, a signal can be demodulated at the entrance of the facility but stay

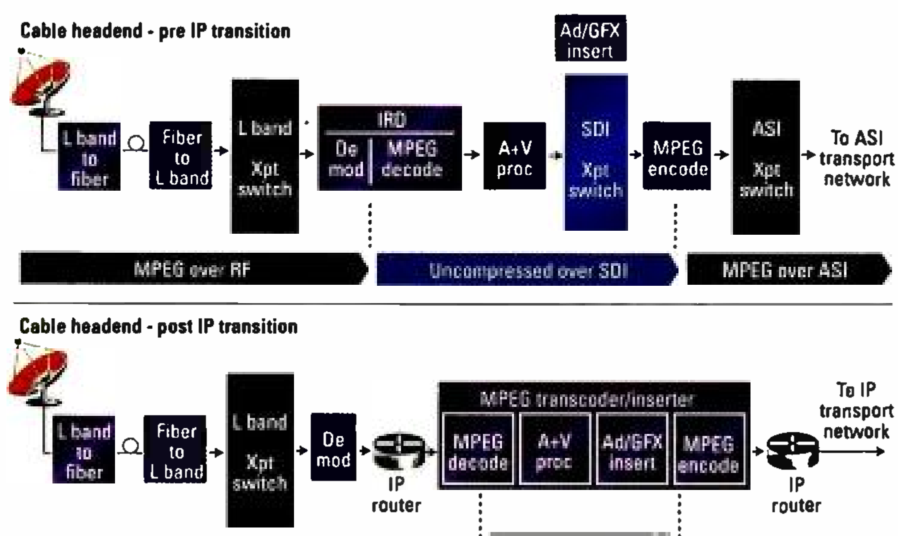


Figure 2. This shows the transition to IP infrastructure in headends.

IP until it hits the integrated processing chain. Here, it is internally transcoded, while allowing processing and insertions to take place, thus allowing routing to remain in the IP realm. This integration has allowed the scalability to more channels. Also, it allows for ease of redundancy within a flexible, routable environment, compatible with telecom networking gear already in use there. Having switched the video infrastructure to IP, headends have the ability to share the same infrastructure for TV, telephony and data for Internet services.

Could IP replace SDI?

In cable headends, the case was clear. But, what about a TV station or multichannel origination facility? Some in the industry, and many from outside the industry, think Ethernet/IP networking should be replacing SDI everywhere in a TV plant. Many think that the transition from SDI to Ethernet will happen gradually. In considering this transition, we follow something of a common sense rule that states, "If the real time video signal is already encoded/compressed, then it is practical to carry it over an IP infrastructure. But, if you are inside a facility, and you have to encode a signal just to switch and distribute it within the facility, then it may be less practical or economical to use an IP infrastructure."

Based on this rule, there are specific areas within a television production and origination facility where Ethernet/IP networking for real-time programs makes sense, and other areas where SDI/HD SDI makes more sense. (See Figure 3.)

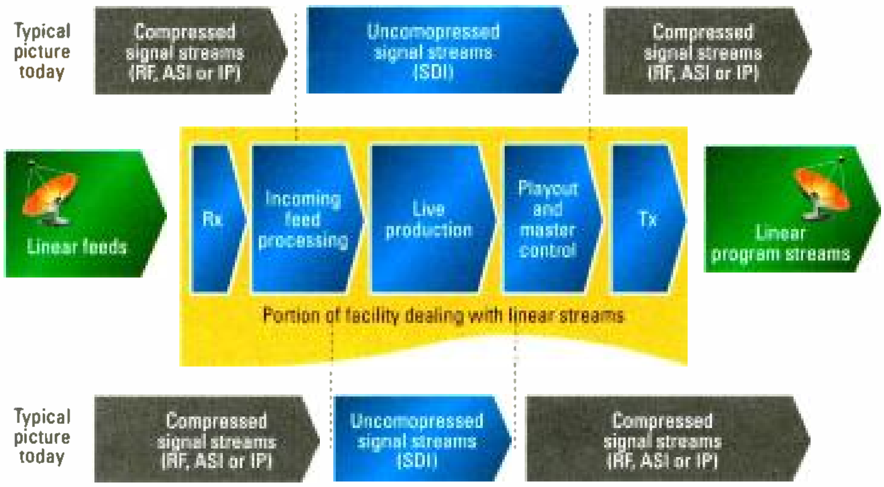


Figure 3. This shows the use of compressed and uncompressed linear signal streams in a typical television broadcast/production facility.

Production

There are several reasons why an IP/Ethernet infrastructure may be less practical today in a live production environment. Many sources in a live production environment are not natively compressed (cameras, graphics, etc.), and there is a desire to keep them uncompressed to maximize quality and, more importantly, to avoid encode/decode delays. One could consider leaving these sources uncompressed and routing them using Ethernet/IP equipment instead of SDI. But, the high bit rates associated with uncompressed video, and the large number of sources in a modern live production, make this impractical today.

The bit rates for uncompressed video are quite large, ranging from 270Mb/s for SD, 1.5Gb/s for HD and on up to 12Gb/s for 4K. (See Table 1.) Now, if we were only talking about a few uncompressed sources, then using IP/Ethernet equipment would be workable. But, it is common to have 1000 x 1000 matrices in a large production facility. And, some larger facilities are now at 2000 x 2000 matrices. The Ethernet networking gear currently available is just not economically viable at those rates and fabric sizes, but it will become more feasible as the price per 10GigE port decreases.

IP's next use in a facility

Based on our rule that if signals are compressed, then it is practical to use

	Uncompressed in plant	Contribution and recording	Distribution to affiliates and headends	Delivery to the home
SD	270Mb/s	25Mb/s-100Mb/s	8Mb/s-25Mb/s	1Mb/s-5Mb/s
HD	1.5Gb/s	50Mb/s-100Mb/s	19Mb/s-50Mb/s	5Mb/s-15Mb/s
1080p60	3.0Gb/s	est. 150Mb/s	TBD	10Mb/s-20Mb/s
4K/Ultra HD (future)	12Gb/s	est. 400Mb/s	TBD	TBD

Table 1. Typical bit rates for uncompressed video can be large, ranging from 270Mb/s to 12Gb/s.

IP infrastructure, we think and IP networking infrastructure begins to make a lot of sense in the playout area. This is the area where automation systems, playout servers, branding, subtitle insertion and delivery encoding combine together to deliver real-time program streams to increasingly diverse distribution platforms. Consider a typical multichannel playout system as it exists today in an SDI-centric world and a more IP-centric approach. (See Figure 4.)

Why migrate playout to IP?

Three main motivational factors influence migration to an IP infrastructure in playout. The first factor is integration. In the past, a playout chain consisted of many discrete devices each interconnected by SDI and supported by SDI routing to allow reassignment of resources and redundancy protection. Today, broadcasters, particularly multichannel broadcasters, use integrated playout systems, often called channels in a box, that combine most of the functionality of a channel chain in a single device, typically software on a standard computer server. These integrated systems accept one or two inputs for live programming but perform all other master control functions inside the box. The output of the integrated channel box typically is a delivery-ready real-time stream on SDI. A few vendors of these integrated channel boxes now provide an option for an encoded (compressed) output over IP.

The second factor enabling IP in playout is that multiple "deliverable" bit rates are becoming required as facilities need to feed secondary outputs for OTT formats along with primary outputs, such as main distribution systems using, for example, ATSC or DVB-T for terrestrial transmission or DVB-S for satellite transmission. Where a standalone encoder is used now to compress the channel's stream for one stream, in the future, encoders will be replaced by transcoders that support IP inputs and outputs, and provide multiple delivery formats. Like in cable headends, these IP in/IP out transcoders enable the transition to IP infrastructure.

From a topology perspective, there is often a separation between playout and uplink or delivery functions. With the availability of IP connectivity between those points, it creates a natural place to start migrating the interconnect between playout and uplink to IP.

Within this context, it is now possible and sensible to link the

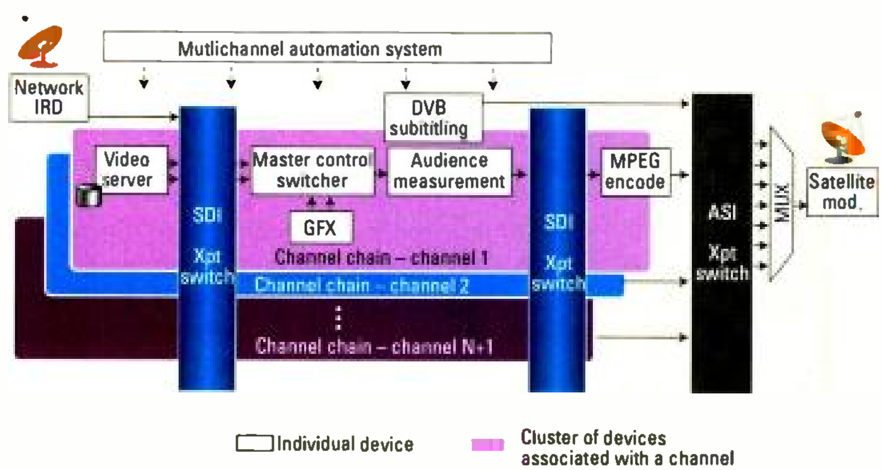


Figure 4. Shown here is a multichannel playout system in an SDI-centric world.

integrated playout device to the transmission transcoding devices using an IP infrastructure. The output of the channel chain is encoded within the playout chain and delivered to the transmission transcoding device over IP. But, rather than have the playout device provide a final delivery grade encode, it provides a higher bit rate mezzanine encode.

Mezzanine encoding

A mezzanine encode is a low-compression encode resulting is an intermediate bit rate. The ideal mezzanine rate is high enough to maintain maximum quality for multigeneration transcoding. At the same time, the rate should be low enough for efficient, cost-effective transportation around the facility and be transcoded easily to multiple output formats for playout distribution.

Preferably, it would be a common format used for recording the original video, which is good enough for editing. One example would be Sony XDCAM-HD. It is common and supports good-quality 4:2:2, 8-bit compression, and it is easy and economical to encode. XDCAM HD is also compatible with MXF, allowing carriage of multiple audio tracks and SMPTE 436M ancillary data. In the context of the playout application, the mezzanine encoding technique is sensible because it keeps encoding in the integrated playout device simple and provides a maximum base quality for the transcoder to work with.

Benefits and challenges

The benefits of an IP networking structure are abundant. First, the flexibility and absolute routability of all signals is simplified. All source signals are wrapped up and compatible with all destinations. We are able to create a truly redundant architecture, which allows protection against points of failure. We are able make use of generic IP switches, which are present already in many facilities and easy to acquire. Since the switches support IP inputs and outputs, they are easy to integrate with other IP systems, such as subtitle insertion devices.

Monitoring can be accomplished for many points with standard IP monitoring tools. Creating an IP infrastructure may eventually allow us to virtualize the playout device as a software application in a virtual machine, allowing several instances to run simultaneously and further enhance scalability.

There also will be challenges moving to an IP-based playout infrastructure, similar to those faced with the TV delivery transition to an IP infrastructure. (See Figure 5.) If we think of the structure physically, it will become much more complex. Before, it was simple when one wire carried one signal to one port. Now, with potentially several signals per wire, routing becomes less obvious. Where exactly is each signal? Can I just take this wire and plug it in elsewhere? No. There is not a simple way to just patch around a point of failure because it is no longer a single signal. If there is a failure, it is difficult to fix, meaning it fails big time.

Luckily, IP networks have the ability to provide excellent redundancy to prevent a potential massive failure. IP network topologies can ease the creation of truly redundant architectures, which prevent many catastrophes. By simply combining two channel chains along with two switches, every routable path becomes redundant. A failure in one chain or one switch will not prevent the signal passage. Actually, we could lose both a switch and a channel chain and still be OK. This redundancy allows the broken node to be serviced without disrupting normal operations.

Managing signal routing also becomes more complex. With SDI routing, it is common to find simple router control panels that allow operators with basic training to quickly make changes to signal routing. In an IP routing environment, routing is typically a system administration task. Tools will have to be developed to make setting of routes simpler and more operational.

Conclusion

In the future, as IP/networking technology evolves, Ethernet port speeds increase and port costs decrease, IP migration will naturally move to other areas. As technology becomes increasingly IP-centric, following this path will leave facilities prepared for whatever follows. Before then, however, now is the time to become more IP-centric and knowledgeable. **BE**

Sara Kudrle is a senior software engineer with Miranda, and Michel Proulx is the former chief technology officer for Miranda.

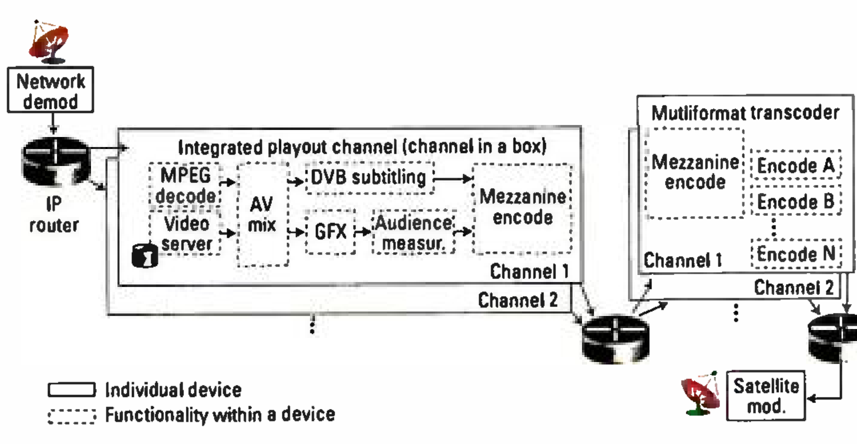


Figure 5. Challenges exist when moving to an IP-based infrastructure like this one.

ADDITIONAL RESOURCES

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

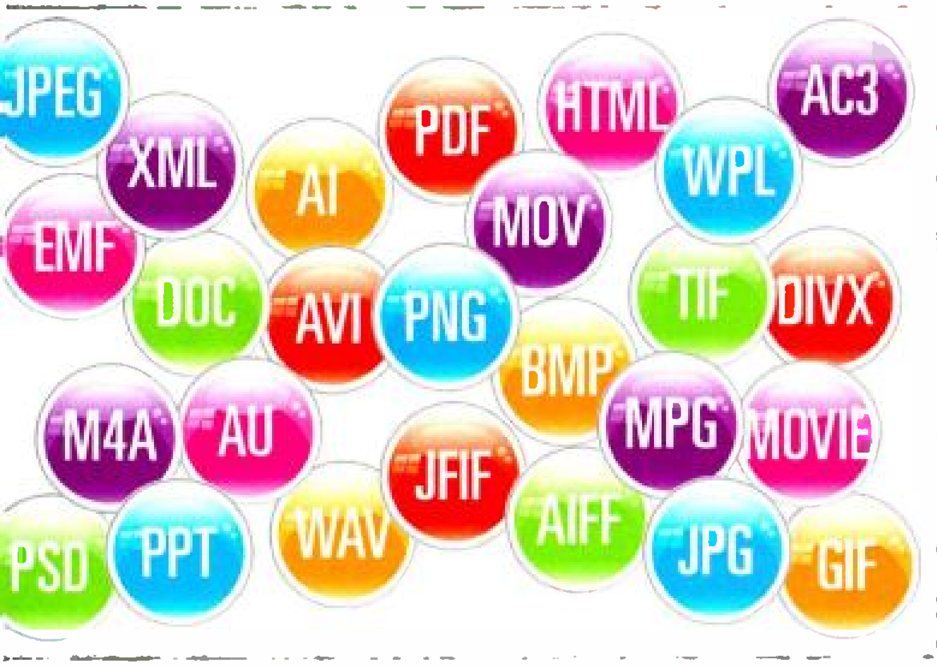
- Playout automation
- Migrating to IT-based playout
- A new world of multichannel automation

Special Report
From BroadcastEngineering

BUILDING FOR TOMORROW

Managing video assets

BY HELEN CHING, P.ENG.



The challenge of managing the number and size of digital assets is, at best, monumental. And as technology evolves, the amount of video content being generated is growing exponentially.

In today's digital landscape, video assets are constantly being created and delivered to be used by multiple departments within an organization. Once these assets are produced, proper management of the content proves a big challenge.

With the never-ending expansion of digital technology, the broadcasting industry has experienced a reduction in on-camera post and production costs. As technology evolves, the amount of video content being generated is growing exponentially — and there are no signs of slowing down.

Challenges

The implication of digital growth is substantially greater for the broadcast industry compared to others, given the following:

- Video file sizes are considerably larger than documents or images. Manipulating them throughout the creative processing cycle requires extensive computing power, broadband width and storage space.
- With thousands of frames for even a short video, it can be a time-consuming chore to locate a certain scene or a specific segment within a large video file if the information is not tagged with metadata.
- Videos can be wrapped in multiple formats (i.e. MOV, Flash, MXF) in combination with multiple compression formats (i.e. MPEG, H.264, ProRes) and need to be managed accordingly.
- With the pressure to deliver visual content quickly, along with the need for airing relevant and timely media clips, the video process from generation to post production is forced to become nonlinear. All departments spanning globally need to work in parallel and collaborate closely to reduce the completion time to finalize the clips.
- Consumers' insatiable appetite for more videos — from bloopers, to crew interviews, to movie-within-the-movie — drives the industry to produce more assets and turn them into tangential products to fulfill consumer needs.

Gone are the days of tape-based workflows. The end game, however, remains the same: gather, edit and implement footage as quickly as possible in order to include it in the final video and add value immediately.

Solution

To address these workflow challenges, organizations must leverage a powerful and strategic system of metadata categorization with each individual piece of video content. That way, the organization can automate the workflow and create a robust digital eco-system.

Gone are the days of tape-based workflows. The end game, however, remains the same: gather, edit and implement footage as quickly as possible in order to include it in the final video and add value immediately.

With automated workflows, content creators and distributors can easily share assets across platforms and collaborate wherever they may be geographically located. (See Figure 1 on page 32.)

To implement a robust digital asset management (DAM) solution, an organization

Beautiful video everywhere

Multiscreen is not a technology, it's an experience

Changing the way the world watches television
Amazing video quality, seamless operation, choice. Harmonic Multiscreen delivers the experience with a proven, complete solution. Compatible, tested and already in service. At scale, not in a lab. You may not see our name, but the experience will tell you that we're behind it.

Learn more at **harmonicinc.com/multiscreen**

harmonic®

should take a close look at how the video assets will be used and who will be using them. Factors to consider are the stakeholders, possible use cases, components of the workflow, the current inventory and the development of metadata schemas.

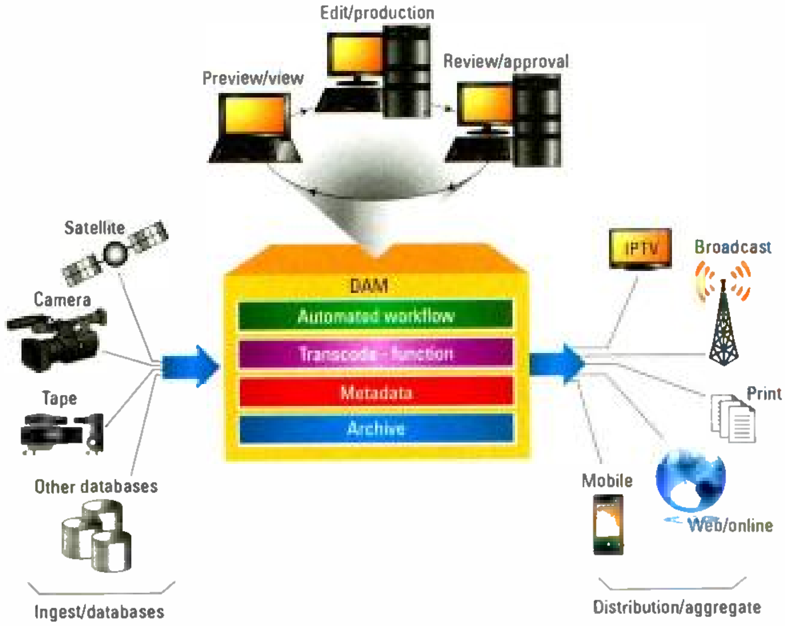


Figure 1. A robust digital asset management system enables content creators and distributors to easily share assets across platforms and collaborate wherever they may be geographically located.

Identify the stakeholders

Different user groups can have different requirements (clients, producers, marketers, writers, special effect wizards, editors, etc.). Identify the stakeholders that will be using the videos; this will ensure the necessary requirements can be met once the organizational structure is in place. Some stakeholders will only be adding content, while others will concentrate primarily on searching and digesting content to finalize programming.

Depending on the stakeholders, they may need to access various content from stills, to scripts, to storyboards and to movie schedules. All this information needs to be managed and requires careful consideration during implementation.

Determine all possible use cases

For a production company, one use case could be capturing live performances, rehearsals and practice sessions to be used for compiling a final 3-D high-definition movie that will eventually be distributed in theatres. The video shoots may happen at different locations around the world, with different on-site teams. Each of the teams would need to provide the scenes they have shot, as well as collaborate with editors to determine if additional footage is required.

Another use case could be delivering an hour-long show, five days a week. The show may consist of a main video, musical clips and audio files, all of which need to be compiled into a seamless package for distribution and ready for viewing by the audience at a certain hour each day. With the intense daily deadline, the production team needs to be extremely efficient during the assembly process, have the ability to quickly create Edit Decision Lists (EDLs) and manage the rights and licensing information for the music clips.

Each of these use cases reveals collaboration points that the DAM management implementation need to take into consideration.

Identify components of the workflow

What are the different components within the organization's workflow? Does it include creating, producing, editing, reviewing, approving and distributing the video, or are there additional components to be considered? For each component within the workflow, what information is required?

For example, for the creating component, date and time of video shoot, actors in the scene, act and scene number, and the cameraman that shot the scene could all be relevant details that the organization would want to capture. However, for the approving component, approver name, date and time of approval, and job ID could be a different set of details the organization would also want to store.

Take stock of current inventory

During its investigative stage, the organization discovers that its tape libraries were not well maintained or catalogued. Producers and editors use different methods to locate needed footage and identify clips, resulting in hours spent sifting through screening in order to isolate. What video formats are currently in existence, and what formats may be required in the future?

What is the anticipated plan for the old footage? Does the organization believe all vintage footage should be digitized from tape, or would it make more sense to have only a critical percentage of the inventory be converted? Would converting the whole inventory allow the organization to repurpose old footage, monetize the assets and aggressively grow additional revenue streams?

Develop metadata schemas

By pulling all the details highlighted in the above sections, an organization can then develop an all-encompassing metadata schema that can be used to tag relevant information. Once metadata contains relevant information, video clips can then be searched quickly and be repurposed when needed.

Instead of spending hours sifting through videos, broadcasters can now identify required footage in minutes. With correct artists tagged to music files or correct photographers tagged to image files, digital rights management becomes a no-brainer. With metadata, there is context, searchability and traceability — all reducing manual intervention.

Benefits

A powerful, centralized DAM environment, combined with a strategic metadata structure, offers a solution that removes much of the clutter from video asset management. By eliminating repetitive tasks, an organization can increase staff productivity while reducing operating costs at the same time.

The ability to repurpose existing content increases the value of and can be used to generate revenue numerous times. This also speeds up time-to-market, given that the organization doesn't need to recreate assets. Transitioning to a permanent digitized central repository containing all its digital assets helps an organization facilitate staff collaboration, regardless of time zones or geographical locations.

Helen Ching, P.Eng., is Director, Product Marketing at North Plains.

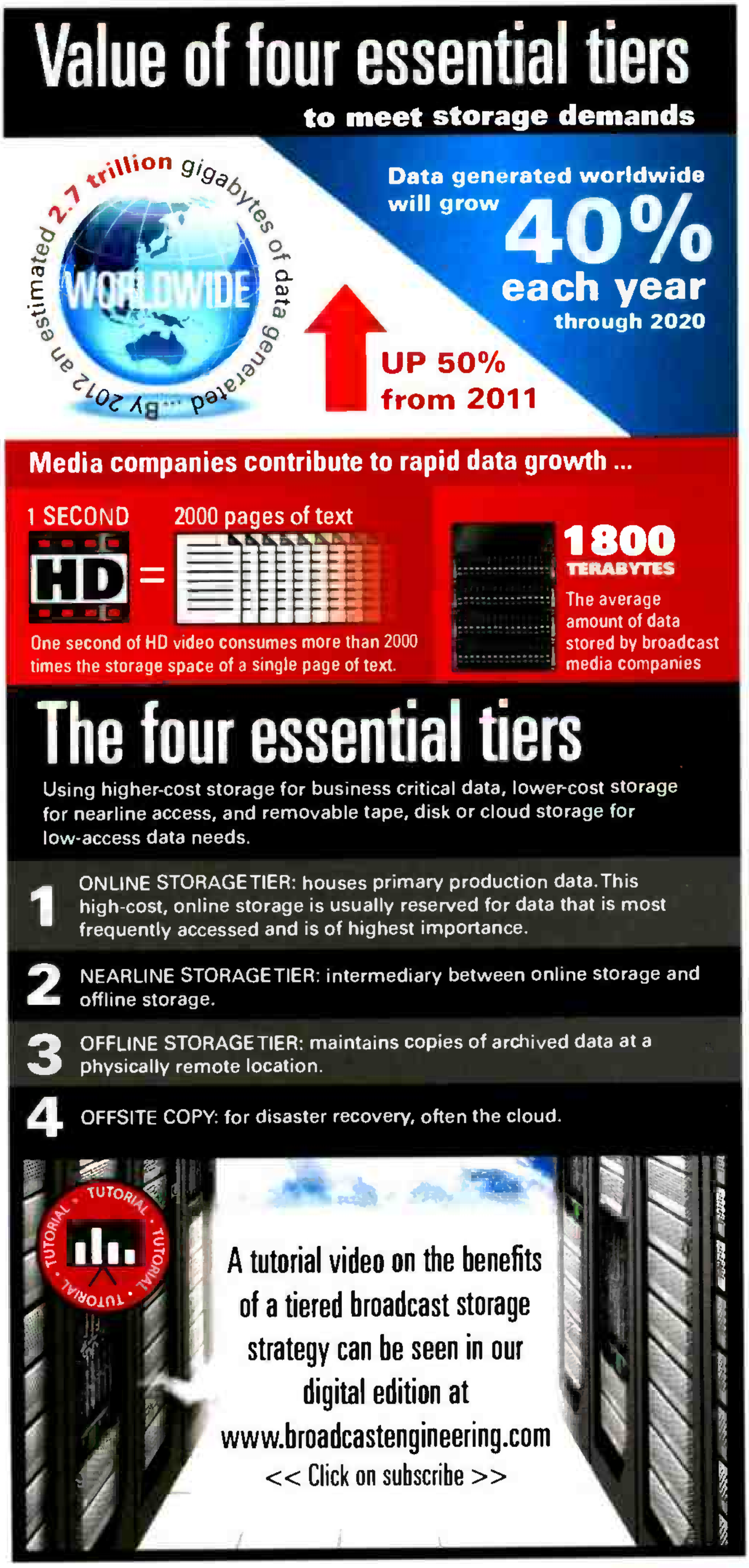
ADDITIONAL RESOURCES

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

- Branding: Keeping viewers glued to your channel
- Building a scalable MAM system
- Going tapeless

Broadcast storage strategy

BY RUSTY ROSENBERGER



The numbers can seem daunting: IDC said that by 2012, the world would generate 2.7 trillion gigabytes of data, up 50 percent from 2011. McKinsey & Co. predicts that the amount of data generated will grow 40 percent each year through 2020. In broadcast, a high percentage of the data produced is digital content, leaving organizations dealing with a faster than normal growth rate compared to many other industries.

As such, media companies are contributing more than their fair share to this rapid pace of data growth. In fact, McKinsey reports one second of HD video consumes more than 2000 times the bytes of a single page of data. The report further states that the average aggregate amount of data stored by a broadcast media company with more than 1000 employees is 1800TB. To put that into perspective, in 2009 the Library of Congress announced that the approximate amount of its collections that are digitized and freely and publicly available on the Internet is about 74TB. That means an average broadcast company stores more than 24 times the amount of data as the Library of Congress.

Rackspace provides another example that visually demonstrates the massive amount of digital data produced, using video content as an example. Assuming that the average video is 4 minutes long and 200MB in size, every day 405 Blu-ray discs are uploaded to YouTube.

As production technology (i.e. frame rates and resolution) continuously gets better and the need for content to be shared across multiple platforms (i.e. websites, social media) increases, broadcast companies are requiring both faster access and more efficient storage infrastructures. In many cases though, data-growth-driven storage needs are hindered, competing with other enterprise-critical expenses. Data is the most critical and valuable asset for this industry, yet many organizations often do not have an optimized, secure storage strategy in place capable of growing with their needs.

Many personnel responsible for making IT decisions in the broadcast and media industry deal with the data deluge by buying increasing amounts of primary storage. While this approach keeps data close at hand, it is excessively expensive and makes it difficult to find specific data — similar to looking for a needle in a haystack. A better solution, proven in enterprises, can be found in an approach called storage tiering, where data is stored in higher-performance, lower-cost storage, or longer-life types of storage over time, based on the current value of the data to the organization. This keeps the data readily available when it is most needed, while cost-effectively and securely storing it for use down the line.

Storage tiering is not a new concept, but it may be new to organizations that are being increasingly challenged by the volume of data they are creating. By learning about tiered storage, and by building data archiving, backup and other data management systems on top of a tiered storage infrastructure, broadcast institutions can reduce content storage costs, quickly transfer large data sets and safeguard valuable data. The tiered approach uses higher-cost storage for business-critical data, lower-cost storage for nearline access, and removable tape, disk or cloud storage for low-access data needs.

The online storage tier is where a company's active, primary production data resides. High-cost, online storage is usually reserved for data that is most frequently accessed and is of highest importance, yet many companies still use the same online storage technologies for backup. This approach is expensive and susceptible to failure.

The nearline storage tier represents an intermediary between online storage, which enables rapid data access, and offline storage, which is more affordable but requires more time for data access. To reduce storage costs without dramatically reducing speed of data access, companies can replicate data to a nearline data protection device. They also can set policies for types of data that should be stored online or nearline. This tier of data access can provide a copy of data in the event that online data is compromised.

The offline storage tier maintains copies of archived data at a physically remote location. The best data management systems enable multiple ways to do this, including removable media that can be transported to another site. RDX is a good example of a removable media format that delivers cost-efficient backup, recovery and archiving. RDX is a unique type of media that offers all of the advantages of disk-to-disk storage, including high performance and a low failure rate, plus the removability and portability of tape. As removable storage, RDX supports offsite storage for disaster recovery and offers a practical way to seed cloud storage. New secure versions of RDX media also feature encryption and cryptographic erase capabilities.

Broadcast IT managers need a system that will not only provide an onsite copy of data for fast restoration, but also an offsite copy if the disaster causes damage to the primary storage location. As confidence in cloud providers continues to increase, more companies will utilize cloud storage for offsite backup. The main benefits of cloud storage are threefold. First, the cloud provides true disaster recovery and business continuity. It adds critical offsite storage to ensure that a business' most important asset is accessible in the event of a disaster. Second, cloud providers offer pay-as-you-go options, which enable businesses to account for storage as an operational expense, not a capital expense. Finally, cloud storage is infinitely scalable, and additional capacity can be used when needed. However, it is important to keep in mind that scalability is not just about the size or amount of storage. The speed of access and throughput must also be scalable.

A four-tier, RDX-based storage approach offers scalability and agility as the amount of data being stored increases exponentially. Broadcast companies need to look for appliances that support increased-capacity media cartridges or the easy addition of new appliances or arrays. It can seem daunting for some companies to overhaul their storage infrastructure while adhering to regulatory constrictions and compliance efforts. However, implementing a four-tiered storage strategy is the best approach for cost-effective optimization and data backup, especially in an industry that produces large media files and an above average amount of data. ■

Rusty Rosenberger is global product management director, Scalable Storage Business, for Imation.

ADDITIONAL RESOURCES

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

- Dynamic storage tiering
- Considerations for IP storage technology
- Storage Primer

IP networking

BY MARTIN WALBUM AND HELGE STEPHANSEN

Danish content delivery services provider GigaContent recently installed Nevion's IP kit in its network.

Classic fears of IP for contribution-level broadcasting — including lack of control, end-to-end quality and security concerns — are slowly being overcome. Why? The advantages far outweigh the risks, given recent technology advancements, coupled with the revenue potential of implementing technology that can transport video and other media content in real time within the same network.

In fact, IP infrastructures may soon be essential to stay competitive in a future defined by consumers of content, not the content itself. Consumers are demanding content on the go, seamlessly in step with the way they live their lives — anywhere, any time, and looking as good on the small screen as the one in the living room. IP is already being used extensively for distributing content to the range of available devices in the secondary distribution realm, as shown in Figure 1. It's time to see why IP is ideally suited to transporting content in the primary distribution space.

Any-to-any connections

IP requires managing connections, scaling bandwidth, implementing smartly engineered management tools, and a lot in between. But it doesn't have to be daunting. It's important to keep the end goals in mind — more efficiency, both in terms of resources and hard costs (read: lower CAPEX and OPEX); economies of scale; and increased network capacity. On the broadest scale, the key to IP technology is the any-to-any connectivity it provides. From a single interface, it allow operators to securely transport video signals to one or more receivers in the network. Think of IP as one big virtual video router that spans the globe and runs on IP instead of traditional equipment.

But IP isn't suited just for the big scale. A considerable share of IP solutions in the professional video realm consist of simple networks such as studio-to-transmitter links. The technology allows easy and reliable connections, and when combined with Ethernet, is a universal solution for small systems to large, complex networks.

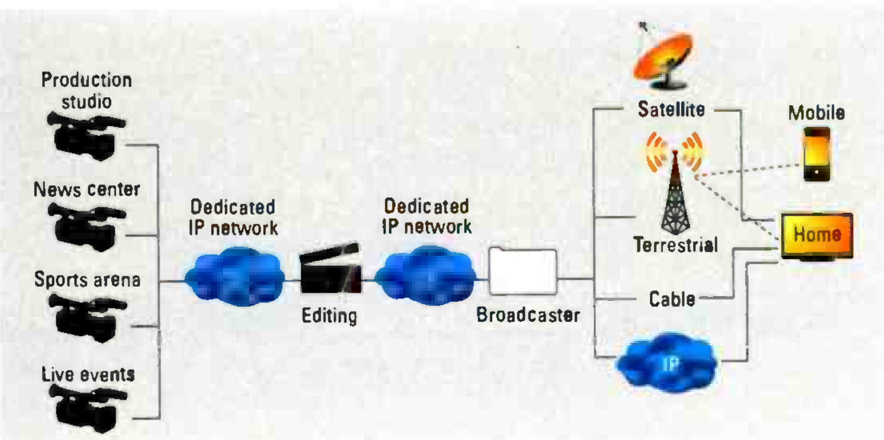


Figure 1. IP is becoming the common standard for media transport for backhaul, distribution and SMPTE 310 studio-transmitter links.

Bandwidth savings — challenges

Some in the industry believe they should migrate to IP to "save money," without really understanding why. As the amount and complexity of content increases exponentially, strategic use of bandwidth is critically important. Video over IP offers the flexibility to scale bandwidth depending on the required quality of the delivered content. With IP, operators can use bandwidth dynamically by shifting connections when and where they're needed.

The real benefit of IP is the ability to share bandwidth among many different applications.

When an IP link isn't being used for video transport, this same bandwidth can be used for other purposes. Although this is one of the biggest benefits of IP, it's also one of its biggest drawbacks. To save bandwidth, IP networks are typically over-provisioned — which means that the same bandwidth is planned for use by different applications.

When calculating the needed bandwidth on an IP link, it's assumed that not all applications need to be transported at once. If this does happen, excessive traffic is dropped on a random basis. For data applications such as e-mail and file transfer, this isn't a problem, as lost IP packets can simply be retransmitted without the end user noticing.

For live video streams, however, this kind of over-provisioning is completely unacceptable. To overcome this issue, IP networks maintain quality by differentiating between packet requirements with the goal of providing constant, guaranteed bandwidth to live services like video streams, while giving less priority to services that aren't time-critical.

So low-priority traffic is delayed to smooth out network traffic bursts. This is more complex than it sounds, but possible with good administration and control capabilities. It also requires a tool to help set up, schedule and tear down connections. These tools are available in the marketplace today, and systems are getting more sophisticated as they automatically manage these functions.

Typical SLA service definitions

Availability/uptime: Network, port or service availability (SA): Time (measured as a percentage) that the network, service or port is available for use and can be accessed by the customer. Network availability is usually measured from the originating service provider network switch node to the terminating switch node (backbone), otherwise known as the service interface point (SIP).

Latency, or network transit delay: Average time it takes a data packet to transmit across the service provider's network, measured in milliseconds. This can be measured as one-way or round-trip delay.

Throughput/committed information rate: Percentage of data frames successfully transported across the service provider's network. Ratio of frames delivered at the terminating SIP of a connection (egress of the network) over the number of frames offered from the originating SIP of that connection (ingress of the network).

IP/packet loss ratio: Percentage of IP data packets unsuccessfully transported across the network.

Utilization: Port-level determination of the ratio of average in/out traffic to the port speed.

Time to report/MTTN (Mean time-to-notify): Time between the failure of a connection from one SIP to another, and the time that the service provider begins to correct the problem.

Time to Respond/MTTR (Mean time-to-repair): Time interval between when the customer informs the service provider of a connection failure between SIPs and the time that service is restored.

On-time provisioning/installation: Time between when the customer places an order and when the service provider fulfills the order.

Table 1. It's important that ISPs and broadcasters share a common understanding of service terms when using IP networks for live video streams.

Service-level agreements (SLA) parameters

SLA	Sample range for parameters
Availability	• 100% availability with max 60 minutes downtime per month • 99.99% network availability for standard service and 99.999% for enhanced service
Latency or network transit delay	10ms - 80ms
Throughput or data delivery rate	99% - 100% (or measured in dropped packets)
Mean time-to-repair or restoration time	4 hours
Mean time-to-install	Next-day service
Credit if SLAs not met	Automatic account credit if SLAs not met.
Other services	• 24 x 7 network monitoring and technical support • Online reporting (daily and monthly usage reports) • Bandwidth availability (4 hours for phone requests/30min for Web requests)

Table 2. Shown here is a sample of SLA parameters for contribution video services.

Service-level agreements

Although some broadcasters are creating their own IP networks, many are taking advantage of existing service provider infrastructures. SLAs — contracts between service providers and customers that define mutually-agreed-upon service-level characteristics, as shown in Table 1 on page 37, and associated metrics assuring that these are met — are enormously significant now as customers demand bullet-proof networks and guaranteed performance.

• *SLA reports.* Reports provide a visual snapshot of the traffic performance indicators defined in SLA agreements. Service providers use these performance reports to: 1) determine what service levels are reasonable to offer customers, 2) provide visual documents to help customers understand performance levels, 3) troubleshoot problems on a network, and 4) prove that the service provided meets contractual obligations. Broadcasters and other customers use SLA reports to monitor network performance and verify their provider's SLA compliance.

• *SLA parameters.* IP and Ethernet services were introduced with basic SLAs in place covering performance metrics such as network availability and mean time-to-repair. These SLAs were typically target objectives, not guarantees. As service offerings increase to the 1Gb/s and 10Gb/s range for mission-critical applications within large networks, Ethernet SLAs are becoming more comprehensive, encompassing penalties if service targets aren't met. (See Table 2.)

These measurements must be carefully considered and understood in the context of contribution video transport. It's incumbent upon broadcasters and studio environments to put SLAs in place that ensure the right kind of use. Some key performance metrics are measured as average values per day or week. This works for data services, where end users don't know if an e-mail is delivered with a few seconds — or even minutes — delay, but live contribution services demand more accurate measurement within specific timeslots.

For example, it's not good enough to measure the average packet loss over a week if the peak value occurs during the 10s in which an athlete breaks the world record for the 100m sprint. SLAs for video-contribution services typically require ultra-short reaction times for a few defined hours during a week. Both the broadcaster and the service provider need to understand and build to specific network needs, as shown in Figure 2.

A case for education

If there's any moral to the story of rising IP adoption, it's that broadcasters must have basic knowledge of how IP networks function, especially when securing bandwidth through service provider networks. Broadcasters need to understand how their needs fit with established IP service parameters and translate

In fact, IP infrastructures may soon be essential to stay competitive and operate in a future defined by consumers of content, not the content itself.

those needs into concrete SLAs for continued high-quality service required by contribution video transport.

The reality is that if you're not using the bandwidth provided by your service provider, someone else probably is. Re-selling unused bandwidth is part of service providers' business models. We must ensure through careful monitoring and management that broadcast IP networks provide the same level of guaranteed bandwidth that SDH and other legacy point-to-point connectivity provides — as well as new benefits. That means knowing what to look for and how to get it. BE

Martin Walbum and Helge Stephansen are senior solutions architects for Nevion.

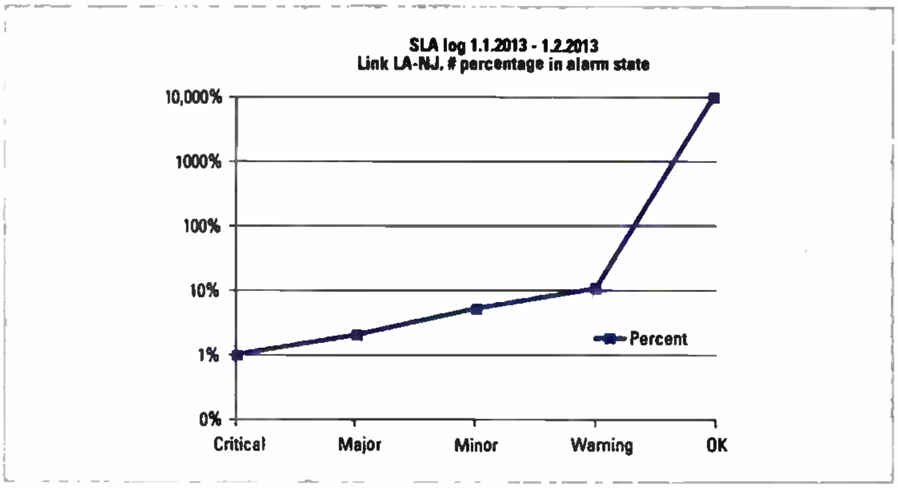


Figure 2. This chart shows an SLA performance graph of a cross-country link.

ADDITIONAL RESOURCES

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

- Router fundamentals you should know
- Expect new IP audio standards
- Considerations for IP storage technology

Tektronix's Cerify

Broadcasters automate QC to avoid playout errors.

BY RICHARD DUVALL

File-based workflows are increasingly used in post production and broadcast, supplanting earlier, less integrated digital workflow architectures. File-based environments can streamline content production, reduce operational expenses, and are critical for the delivery of OTT programming to PCs and mobile devices for multiscreen video services.

Given its importance, quality control (QC) is naturally critical for the successful use of file-based workflows. Historically, QC has relied on direct visual inspection, usually with personnel spot-checking a few minutes of a program's beginning, middle and end. However, this approach has never truly been sufficient for two reasons. The first is that humans tend to notice just two classes of technical impairments:

- Signal-level-related issues such as video luma and chroma, or audio loudness; and
- Obvious distortions and dropouts manifested in problems such as black sequences, frozen frames, blockiness, loss of audio and audio/video sync discrepancies.

Because of this, there are errors that are hard to recognize even when the program is closely watched.

"Slice order" errors in MPEG transport streams can cause blockiness and other picture distortions.

Scalability is the second reason visual inspection is insufficient; the industry is simply "running out of eyeballs" to check the tens of thousands of files being created for multi-screen services monthly.

It is this tremendous volume of content that must be managed and quality-checked that is driving the industry toward automated, process-spanning QC solutions.

Automating QC

Tektronix's Cerify is a fully automated system for verifying and checking file-based content prior to transmission or use. It can be deployed on stand-alone Windows PCs or as an enterprisewide system that interfaces with third-party automation or asset management systems.

It performs comprehensive QC of file-based video ingested from multiple sources and encoded at different bit rates, formats and compression standards for SD/HD, VOD and IPTV delivery. In particular, it checks file-based video for syntax issues, baseband problems, encoded content errors and structural issues:

- *Syntax testing.* Syntax testing is critical, as it ensures the integrity of the encoding of the file structure. Syntax issues can cause catastrophic problems, and without syntax checking, it is even possible that the content won't play out at all! Incorrect syntax can also put set-top boxes into a continual state of rebooting, or do just the opposite — lock them up and necessitate rebooting. Rigorous syntax testing can verify a compliant stream and save a truck roll.

Syntax errors are particularly damaging in VOD and other file-based services, given the growth of these media libraries. Spot checks simply cannot address the literally tens of thousands of programs that need to be checked, which drives the need for automated QC.

- *Baseband checking.* Cerify can look at the decoded baseband video and audio to evaluate image and audio performance. Looking more closely at baseband errors, it appears that most are part of the original acquisition, though some may be caused by edits made in the workflow.

Gamut violations are classic baseband errors that must be captured with an instrumented decoder, which can also capture other gamut problems with RGB color components. Black frames, frozen frames, letter boxes and pillar boxes, and dropouts also can be detected from the decoded video content.

Audio problems such as clipping are also observable in the decoded baseband stream. By looking at the signal level, it is possible to determine whether loudness limits, peak limits, instantaneous peaks and true peak value limits have been exceeded, as well as long-term loudness over the span of the content. With the arrival of government deadlines for audio loudness compliance, Cerify can go beyond audio checks by automatically correcting audio loudness in the audio essence(s). Working with all container types, audio level correction can be applied to the entire essence duration, and can be triggered by non-compliance to EBU R 128 or ITU 1770-2 Audio Loudness recommendations.

- *Checking encoded content.* Encoders, too, are a potential source of errors. A faulty encoder can produce syntax, similar to an editing application that has flaws of its own. A misconfigured encoder, or one with the bit rate set too low, may over-compress the material.

Encoder problems can appear as "slice order" errors in MPEG transport streams. These errors create block artifacts on the screen and degrade picture quality.

Additionally, interlaced videos are encoded either bottom-field-first or top-field-first and must be played out in the same sequence. A conflict in the field order will cause distracting motion artifacts — almost a zigzag motion when the subject should be moving smoothly across the screen. North American and European SD content have opposite field ordering, which is easy to overlook when transcoding or editing.

• *Structural checking.* Structural checks are as much a part of QC as are baseband and encoder checks. It is possible for significant (and required) elements to be absent from content that seems syntactically correct. The QC system must examine the container structure, the content and the codec headers, comparing the actual contents with the expected information to identify structural problems.

Checking video and audio codec headers may reveal unexpected essence formats and encoding with respect to profile and level, GOP structure, frame and sample rates, picture size and aspect ratio, interlaced or progressive scan, color depth, and color sampling. Ultimately, structural checks are to ensure that the content has the right format characteristics set up in the proper order.

Cerify addresses these issues and can be integrated with automation and asset management systems, as shown in Figure 1, thus feeding them with data using the CeriTalk API required to automatically decide on next workflow steps or drawing attention to assets that need expert review.

The XML-based test templates can be exchanged between Cerify systems, and applied as the definition of the required test standards between suppliers and broadcasters to establish SLAs and reduce costly churn.

Conclusion

Automated QC processes save time and resources and are always able to devote their undivided attention to the content.

In addition, automated QC is more thorough and objective than visual inspection. It is consistent and reproducible, and able to isolate errors that are encoded deep within the files. Problems with syntax errors or encoding parameters and mismatches with structural metadata are not normally perceptible to the human eye at the point of inspection. Nevertheless, they can cause serious problems later in the workflow.

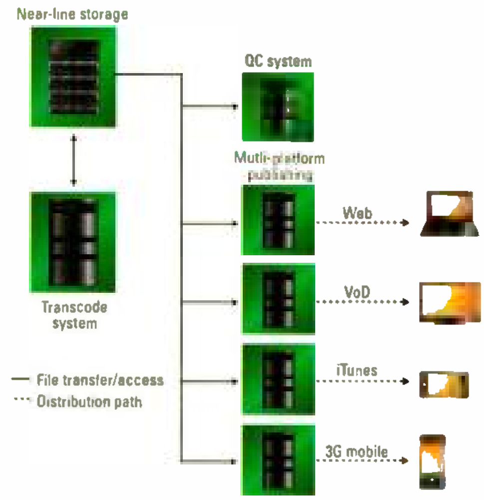


Figure 1. The transcode system converts content into formats compatible with diverse receiving platforms; the QC system checks the content quality before transmission and notifies the media asset management system.

Automated processes supplement human QC and can help sort out the most critical errors. This maximizes labor efficiency by focusing QC technicians on problems requiring urgent attention. BE

Richard Duvall is technical marketing manager for Tektronix.

+ ADDITIONAL RESOURCES +

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

• Automating Quality Control in File-Based Workflows

Ventuz's Ventuz Broadcast

The design tool delivers interactive newscasts.

BY ERIK BEAUMONT

As visual and interactive expectations of audiences grow, studios are faced with the challenge of creating and delivering news productions that adapt to and incorporate forward-thinking innovations. These may include such things as 3-D graphics and streams embedded with live video content, as well as interactive, in-studio video walls. The challenge is to produce real-time motion graphics and incorporate 2D/3-D graphics with live or archived video content, Web content or various other real-time streams — all while providing the talent with easy-to-use, multi-touch functionality to deliver compelling content.

This complex broadcast environment requires a tool that offers an open framework to seamlessly integrate with a newsroom graphics production infrastructure. This challenge also requires a graphics engine that delivers a nonlinear and high-end authoring environment paired with control functionality for a wide range of external components.

Interactive design

One such tool is Ventuz Broadcast, an on-air graphics and virtual design tool from Ventuz Technology. This open and flexible authoring environment provides the building blocks for the integration of 2D/3-D content (graphics, video, objects and text) and animation. Its nonlinear nature allows news production teams to make changes during any point of the news production cycle, in real time. This software is built around a concept that allows it to interact with any interface — whether it be touch screens, interactive video walls, USB devices or different control devices.

The open and flexible authoring environment of the Ventuz Broadcast enables the integration of 2D/3-D content (graphics, video, objects and text), and animation for interactive, in-studio newsroom applications.

The software runs on standard Windows hardware and integrates seamlessly with other software. Therefore, creative boundaries and complex newsroom presentations are limited only by hardware power.

Ventuz Broadcast graphics creation software is designed around Microsoft DirectX, a collection of APIs for handling a variety of multimedia tasks. The software development team chose this platform due to its ability to provide facilities with the freedom of individual hardware choice, thus removing technology barriers.

Each broadcast facility is able to take full advantage of the functionality of its existing professional equipment, while incorporating an updated broadcast graphics pipeline. Custom hardware options, bundled as a turnkey solution, are also available, depending on the size, desired capacity of power and scalability of the newsroom presentation. This type of software offers a wide range of possibilities to further modernize the way newscasters interact with each other, with their content and ultimately, their audience.

Interactive capability

Broadcasters can offer viewers more breadth and depth of news by integrating real-time content of local, national and international news into their on-air programming in addition to interactivity. The Ventuz tool allows the easy combination of common broadcast graphics with interactive technology. The nonlinear approach of the software provides the newsroom presenters the freedom to choose topics and corresponding imagery or video clips; they are not dependent on the control room personnel to make changes.

During news delivery, a news anchor now has the ability to use a newsdesk touch screen, which can be a mirrored version of what is visible on the video wall. This use of the touch screens in the newsroom enhances an already profitable business model for broadcasters, enabling them to introduce new forms of programming as they adapt content to provide viewers with a more immersive experience.

The complexities of a content-rich news production and presentation are reduced with software features. Production capabilities and composition tools integrate 2D or 3-D elements, fonts and multi-layer textures and text effects; import content (2D/3-D, audio, video, graphics); and crawl/roll/ticker elements. Also, real-time data imports of databases, live RSS feeds and time code are possible, as are asynchronous updates of images and XML.

In addition, the software allows reporters to state logic for creating animations: Instead of defining the animation on a timeline, this software enables users to create states of an object. For example, in state 1, the object is in position 1, with a scaling of 1 and opacity of 1; in state 2, the object is in position 2, etc. Users can create as many states as needed, and then link the states to certain triggers. Thus, complex animations can be created that work well with interactive setups.

The software also offers reporters keying, remote, export and publishing functions. It is configured with an SDI fill and key signal to perform live key creations routed to a digital switcher or external keyer. It also has the ability to pass content, commands, assets from external applications or data sources to the graphics software engine. This is ideal for use with standard templates — a regular occurrence in a news production. Its export and publishing functions can be used for preparing backup and runtime archives of newsroom productions. Additionally, a scripting feature grants access to a comprehensive scripting environment for newsroom engineers, editors and programmers to develop unique components or properties for their specific facility's newsroom presentation requirements.

The tool offers straightforward setup of interactive presentations in all its forms: single touch, multi-touch with as many touch points as the hardware can handle, touch translation and live drawing for telestrators. Multi-display on the product allows users to display content seamlessly across as many monitors as are required, and the software is resolution-independent for use on video walls of any size and any format.

Different versions

The Ventuz broadcast software is available in two versions — The Designer, which is an authoring and design tool, and Runtime, which serves as a player for 3-D graphics. Additionally, with the next software generation, to be released in the first half of this year, Ventuz Director will be introduced. Director is a content authoring and show control tool using timelines, templates and pages to enable established broadcast graphics workflows.

Software tools such as Ventuz Broadcast offer a wide range of possibilities to further modernize the way in which newscasters interact with each other, their content and the viewing audience. With a more flexible approach to the delivery of information comes the possibility of including viewer participation, such as text messages and social media feeds. With control over the graphics and imagery at their discretion, newscasters can easily cater to the audiences' interests, resulting in greater viewer engagement and retention. BE

Erik Beaumont is product manager at Ventuz.

Riedel Communications

The Red Bull Stratos mission relied on the company's communications technologies.

BY MICHAEL GROTTICELLI

Remotely controlled cameras and widely dispersed intercom communications have played a key role for different types of production and live broadcast applications for many years based on their ability to cover areas that are impossible or impractical for a human. But none to date has reached the heights of the system developed to document, in real time, the world's record for a freefall skydive and, it turns out, the first human to break the sound barrier without vehicular power.

Throughout the record-breaking jump, the mission control ground crew maintained complete two-way communications with Baumgartner and shared his experience in near real time with the world.

Felix Baumgartner's skydive from the edge of space in October 2012 captivated millions of viewers as they watched nervously via the Internet and other IP-connected devices as he stepped off his flying platform and plunged to Earth. Baumgartner, a highly experienced extreme base jumper and skydiver, ascended to 24mi in a stratospheric balloon that lasted about two hours and then fell 121,100ft in 4 minutes, 20 seconds.

Throughout the Red Bull-endorsed record-breaking jump, the mission control ground crew was able to stay in touch with him and share his experience with the world.

That's because several technical teams at Riedel Communications, based in Germany, worked closely with Baumgartner and his team in the U.S. and elsewhere during a series of test skydives over a three-year period prior to the historic leap. Riedel provided most of the technology used for the audio/video-transport system — including both wireless and wired digital intercom systems — mounted inside the Red Bull Stratos capsule that carried Baumgartner into the stratosphere, as well as the entire communications infrastructure for the Red Bull Stratos project.

The company's system engineers also designed and supplied fiber-based video and signal distribution and wireless video links that allowed the captivating pictures from the capsule's onboard cameras to be transmitted back to the ground with very low latency. (The signal is coded and decoded for the wireless transport, which naturally caused some latency in the video by the time it reached Earth.)

Matthias Leister, head of broadcast solutions at Riedel Communications, said his team outfitted a small pressure housing (about the size of a full beer keg) inside the capsule with enough equipment to satisfy a medium-size mobile production truck. This turned out to be a major challenge as size and weight restrictions limited them in a number of technical areas.

Really long-distance live HD production

In the Red Bull Stratos capsule, company staff mounted an onboard video control system for nine HD video cameras with specially developed telemetry control. This system was used to remotely control the cameras, which included three RED ONE cameras, three Canon EOS 5D DSLR still cameras, nine PENTA Studiotech LMP HD1200 micro HD cameras and 13 Panasonic P2 recorders. It also included a small digital video router to capture all of the ISO signals. This enabled complete remote control of the whole video system and featured three HD video downlinks that were dynamically assigned to the selected cameras and whose feeds were captured inside an on-site company trailer.

The company's engineers developed the software that allowed mission control to switch any of the nine HD cameras to any of the three video downlinks, allowing mission control operators on the ground to remotely trigger (start and stop) Panasonic P2 recorders inside the capsule, as well as adjust color balance, gain and shutter control on the cameras.

A single dual-digital radio link was employed to control all of the video downlinks from the capsule, Leister said, which allowed them, in the event of RF interference, to switch to another frequency and continue controlling a particular device. Switching so many different devices and brands from different manufacturers proved to be a major challenge in and of itself.

Ground control brings it all together

The communications infrastructure on-site included the entire compound, the mission control, the production offices, the media/press center, an HD edit suite and a mobile production truck (Lyon Video's MU-8) used as the main A/V control room. All these facilities and positions were networked together into a single communications infrastructure via the company's Artist Digital Matrix system.

The company also provided an on-site digital radio network with more than 100 radio receivers and 10 channels, which were seamlessly integrated into the wired matrix intercom system. This allowed radio users to directly talk to intercom users and vice versa. The company also supplied the radio system housed inside Baumgartner's chest pack and used to communicate live during the freefell.

Leister said that the free-fall radio system inside Baumgartner's chest pack was not as powerful and could not transmit a signal to Earth as well as the one radio system inside the capsule because they could not use a large enough battery system (due to weight and bad aerodynamics). Instead, they used steerable antennas on the tracking trucks and relayed the signal via a TV tower in the nearby town of Caprock, NM. That tower relayed the signal between mission control and the tracking truck.

Fiber-based networking in a ring

Once the video signals reached the ground, they were immediately routed with the company's MediorNet fiber-based network technology. Two dozen nodes were installed in a redundant ring topology across the compound to ensure reliability. In case of a potential connection loss between two nodes, the signals would have still been distributed due to the redundant topology. There could be no second chances to get it right.

The two-way connection to the launchpad of the capsule was also accomplished with the system, with two MediorNet Compact frames connected to the main system. All links in the system were tied together with the company's PURE cable, a tactical fiber cable for demanding mobile applications that is equipped with professional-duty Neutrik opticalCON QUAD connectors.

The MediorNet backbone transported all video signals of the Red Bull Stratos mission, as well as all signals from the production truck and from two FlightLine optical tracking trucks used to monitor the in-flight capsule. Riedel networking technology was also used to distribute intercom panels and IP-based broadcast audio signals between mission control and an off-site recording facility, and for the transfer of telemetry data featured on several broadcast applications, such as a moving map that documented Baumgartner's fall and his speed and oxygen levels.

In addition to the video transport, the MediorNet technology also served as a network backbone for the on-site intercom installation and the Internet connection, providing Ethernet connectivity to all areas of the ground-based compound.

The live feed of the Roswell, NM, jump, reportedly drawing a record 8 million concurrent streams on YouTube and airing on more than 80 TV outlets in 50 countries, was produced by Red Bull Media House. Once Baumgartner landed safely on the ground, after reaching a peak speed of 833.9mph (or 1.24 times the speed of sound), the technical feat — switching so many different devices and brands from different manufacturers through a single system — was said to be equally rewarding. BE

Michael Grotticelli covers the professional video and broadcast technology industry.

Ka-band

Ka-band IP-over-satellite systems are becoming an attractive alternative to SNG.

BY STUART BROWN

News remains a central genre for broadcasters, and as new markets and new delivery platforms emerge, broadcasters are expected to provide 24/7 coverage for rolling news channels and increasingly sophisticated websites. In an era of tight budgetary control, that means controlling deployment costs while responding ever faster.

Traditional methods for establishing live links — such as point-to-point microwave and satellite newsgathering (SNG) — continue to dominate. There are, though, newer methods based on IP delivery, and we are seeing these beginning to get a foothold in the contribution market.

Using an IP platform has a number of benefits, including:

- *Cost:* Because IP systems are used by pretty much everyone these days, hardware and connectivity are inexpensive and getting cheaper by the day.
- *Universality:* IP is a universal standard for transmitting data around the world, greatly simplifying operations from foreign countries.
- *Bi-directionality:* IP readily supports IFB and reverse communications.
- *Enhanced communications:* An IP platform can give journalists access to increasingly sophisticated newsroom IT systems and the Internet.

Today the most common IP platform is the range of 3G/4G bonding systems, which use multiple cell phone data channels to send and receive material. These are quick to set up, so they often work well as a first response solution. However, they can be unreliable, particularly at large events where the systems are competing for bandwidth with the general public. Bit rates are typically below those normally considered for even SD quality. The introduction of 4G has alleviated this a little, but as general uptake of 4G increases, the problem of network congestion is likely to re-emerge.

Another solution rapidly gaining traction is the use of Ka-band IP-over-satellite systems. Introduced to provide high-speed Internet connectivity to remote communities, these systems can also be used to provide HD broadcast-quality live paths, plus Internet and IT facilities, at a fraction of the cost of "regular" Ku-band SNG terminals.

Power consumption of a Ka-band system is in the range of 60W to 70W, so a journalist could carry it to a news location, rig it in a rental car and run it from the car battery.

Frequency re-use

Ka-band operates at extremely high frequencies — in the 30GHz range. This enables the satellite spot beams to be much smaller than those typical of Ku-band.

In turn, this enables the frequencies for each beam to be re-used in different spot beams without risk of interference. This significantly increases the capacity of the satellite.

The latest Ka-band satellites have 38 times the data capacity of a comparable Ku-band satellite. The extra capacity not only reduces costs, but also it minimizes contention, the number of people sharing access to the system and thus competing for its resources. This is important when streaming video.

Antenna size

The gain of a dish antenna is proportional to the frequency being used. At the high frequencies of Ka-band, you can use a smaller dish to get an equivalent power output up to the satellite. It also means the RF amplifier requirements are reduced.

This all combines to significantly reduce the size, weight and power requirements of the terminal, meaning it can be transported in a regular car or flown as checked-in baggage. Carrying the equipment rather than waiting for it to be shipped makes for faster deployment.

Line-up

Finding the correct satellite and then optimizing the antenna alignment is traditionally a skilled job and requires an engineer with complex and expensive equipment such as a spectrum analyzer. As the latest Ka-band systems are designed for home installation by unskilled personnel, much of the alignment process has been considerably simplified by the system design.

In Europe, Eutelsat provides a free iPhone app for locating its main Ka-band satellite, KA-SAT; holding the phone up to the sky shows the user exactly where the satellite is located. The user simply sets the dish elevation to the displayed figure and points to the area of sky indicated. The transceiver then emits a tone to guide the final optimization of elevation and azimuth. Ka-band services use circular polarization, so there are no issues with cross-polarization associated with traditional Ku-band systems.

Bookings

Once the terminal is correctly pointed at the satellite, it automatically logs on to the system and, in a matter of a few minutes, is ready to be used. Initially, it gives a low-bit-rate connection to the Internet, allowing access to a booking portal for reserving higher bit rates.

The traditional downside of Ka-band has always been its susceptibility to rain fade. This is because, as a general rule, the higher the frequency of an RF signal, the more it is absorbed by rain. The latest systems compensate for this by running with significant spare capacity in clear sky conditions. In severe rain, the system will automatically drop to a more robust modulation scheme. It means the user will notice a drop in available bit rate, but should still remain on-air.

Let's look, then, at a typical system designed for HD newsgathering. The minimum requirement should be for data rates of around 9Mb/s upload and 20Mb/s download, which gives comfortable headroom even when live to air.

The antenna will be in the 0.6m to 1.0m range. A 0.75m dish will give the required performance in most conditions. Most of the major manufacturers have Ka-band flyaway and vehicle mount antennas of this size in their product ranges. This links to the transceiver, which will probably be 3W to 4W. It will mount directly on the antenna and connect to the data modem by L-band RF cables, although some systems now use a single standard coax cable.

The IP-over-satellite modem will depend on the choice of satellite system. In addition to converting IP data to satellite RF, it will also supply power to the transceiver. The video data, in turn, comes from an IP video encoder, again widely established and proven technology.

The complete system is likely to weigh less than 44lb and pack into a single case, meaning it would not be a serious problem to take it as baggage on a commercial flight. Power consumption will be in the range of 60W to 70W, so the journalist could carry the system to a news location, rig it in a rental car and run it from the vehicle battery.

Most important, all this technology is available today and proven in use. Major broadcasters are taking advantage of the agility, flexibility and cost savings that IP over satellite delivers.

One broadcaster covered the Greek general elections using a single 70cm Ka-band antenna clamped to a railing on the roof of the journalists' hotel. Using the bi-directional IP platform, they were able to provide simultaneous live reports for domestic and international television bulletins; off-air feeds of competitor broadcasters and local Greek TV; FTP transfers of edited packages; live radio reports; FTP transfers of edited radio packages; full access to the newsroom IT system; and Web browsing and e-mail.

All this was accomplished using a transceiver and modem combination, which cost $500. It makes an attractive alternative to SNG! **BE**

Stuart Brown is Broadcast Systems Director of Cobham.

+ ADDITIONAL RESOURCES +

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

- A tutorial on modulation technologies, from ASTC, OFDM, QAM to DVB
- The cloud and connectivity
- Is the cloud really as green or cost-friendly as widely thought?

METAMORPHOSIS

The Changing Face of MEDIA & ENTERTAINMENT

FREE
Exhibits-only Pass
Use Code PA07

It's all about ME, and how the Mobile Experience is energizing the transformation of Media & Entertainment. Second screens, OTT and smart TVs are enhancing television with interactive content and applications.

Meeting Expectations to seamlessly deliver dynamic content that thrives in Multiple Environments will depend on how well you integrate new strategies, technologies and players. NAB Show® is a Marketplace Energized...addressing the toughest challenges for station engineers and distributors across all platforms, including news and sports coverage, studio production, high efficiency video encoding and workflow, big data, asset management and cloud. Give yourself something to smile about. Register today!

NEW in North Hall: CONNECTED MEDIA WORLD

CONFERENCES April 6–11, 2013 / EXHIBITS April 8–11
Las Vegas Convention Center / Las Vegas, Nevada USA

www.nabshow.com

NABSHOW®
Where Content Comes to Life

Join Us! #nabshow

A PENTON MEDIA PUBLICATION

BroadcastEngineering®

www.broadcastengineering.com

Editorial Director: Brad Dick, brad.dick@penton.com
Editor, World Edition: David Austerberry, editor@broadcastengineeringworld.com
Managing Editor: Susan Anderson, susan.anderson@penton.com
Production Editor: Andrew Ward, andrew.ward@penton.com
Production Editor: Curtis Kitchen, curtis.kitchen@penton.com
Production Editor: Laura Collins, laura.dixon@penton.com
Senior Art Director: Robin Metheny, robin.metheny@penton.com
Technical Consultants:
Computers & Networking – Brad Gilmer
Digital Video – Aldo Cugnini

Brand Manager: Carmen Lake, carmen.lake@penton.com
Sr. Marketing Manager: Kim Davidson, kim.davidson@penton.com
Director, Online Sales: Angie Gates, angie.gates@penton.com
Production Coordinator: Curtis Jansen, curtis.jansen@penton.com
Audience Development: Sonja Trent, sonja.trent@penton.com

Penton

Penton Media, Inc.
249 West 17th Street
New York, NY 10011
Chief Executive Officer: David Kieselstein, david.kieselstein@penton.com

MEMBER ORGANIZATIONS
Sustaining Member of: Society of Broadcast Engineers
Member, American Business Media,
Member, BPA International,
The Missouri Association of Publications

BE US/Canada Subscription Rates: Free/controlled circulation to qualified subscribers. Non-qualified persons may subscribe at the following yearly rates (subject to change): USA - $59; Canada - $69; Outside US and Canada - $99. Digital subscription $12 worldwide. Customer service can be reached at: broadcastengineering@pbinews.com or by calling 866-505-7173 or 847-763-9504

BE World Subscription Rates: Free/controlled circulation to qualified subscribers. Non-qualified persons may subscribe at the following yearly rates (subject to change): USA - $59; Canada - $69; Outside US and Canada - $99. Digital subscription $12 worldwide. Customer service can be reached at: broadcastengineering@pbinews.com or by calling 866-505-7173 or 847-763-9504

ARCHIVES AND MICROFORM: This magazine is available for research and retrieval of selected archived articles from leading electronic databases and online search services, including Factiva, LexisNexis and Proquest. For microform availability, contact National Archive Publishing Company at 800-521-0600 or 734-761-4700, or search the Serials in Microform listings at *napubco.com.*

REPRINTS: Contact Wright's Media to purchase quality custom reprints or e-prints of articles appearing in this publication at 877-652-5295, +00-1-281-419-5725, or penton@wrightsmedia.com

PHOTOCOPIES: Authorization to photocopy articles for internal corporate, personal, or instructional use may be obtained from the Copyright Clearance Center (CCC) at 978-750-8400. Obtain further information at *copyright.com.*

PRIVACY POLICY: Your privacy is a priority to us. For a detailed policy statement about privacy and information dissemination practices related to Penton Media products, please visit our Web site at *www.penton.com.*

EDITORIAL and BUSINESS OFFICE: Penton Media, 9800 Metcalf, Overland Park, Kansas 66212 • 913-341-1300 • *penton.com*

AD INDEX

Broadcast Engineering is not responsible for errors in the Advertisers Index.

SALES OFFICES

KEY ACCOUNTS SALES MANAGER
Carmen Lake
(772) 344-6035
Fax: (913) 514-7326
carmen.lake@penton.com

US/CANADA

East and classified sales
Julie Dahlstrom
(312) 840-8436; Fax: (913) 514-3684
julie.dahlstrom@penton.com

West
Carmen Lake
(772) 344-6035
Fax: (913) 514-7326
carmen.lake@penton.com

Customer Service:
866-505-7173 or 847-763-9504

Broadcast Engineering, February 2013, Vol. 55, No. 2 (ISSN 0007-1994) is published monthly and mailed free to qualified persons by Penton Media, Inc. 9800 Metcalf Ave., Overland Park, KS 66212-2216. Periodicals postage paid at Shawnee Mission, KS, and additional mailing offices. Canadian Post Publications Mail Agreement No. 40612608. Canada return address: Bleuchip International, P.O. Box 25542, London, ON N6C 6B2. POSTMASTER: Send address changes to Broadcast Engineering, P.O. Box 2100, Skokie, IL 60076-7800 USA. CORRESPONDENCE: Editorial and Advertising: 9800 Metcalf, Overland Park, KS 66212-2216 Phone: 913-341-1300; Edit. fax: 913-967-1905. Advert. fax: 913-967-1904. © 2013 by Penton Media, Inc. All rights reserved.

INTERNATIONAL

Europe
Richard Woolley
+44-1295-278-407
Fax: +44-1295-278-408
richardwoolley@btclick.com

Israel
Asa Talbar
Talbar Media
+972 (77) 562-1900
talbar@talbar.co.il

Japan
Mashy Yoshikawa
Orient Echo, Inc.
+81-3-3235-5961; Fax: +81-3-3235-5852
mashy@fa2.so-net.ne.jp

Asia Pacific
Tan May Ling
InterAct Media & Marketing
+65-9623-1405; Fax: +65-6265-3000
tmling@singnet.com.sg

REPRINTS
Wright's Media
877-652-5295
+00-1-281-419-5725
penton@wrightsmedia.com

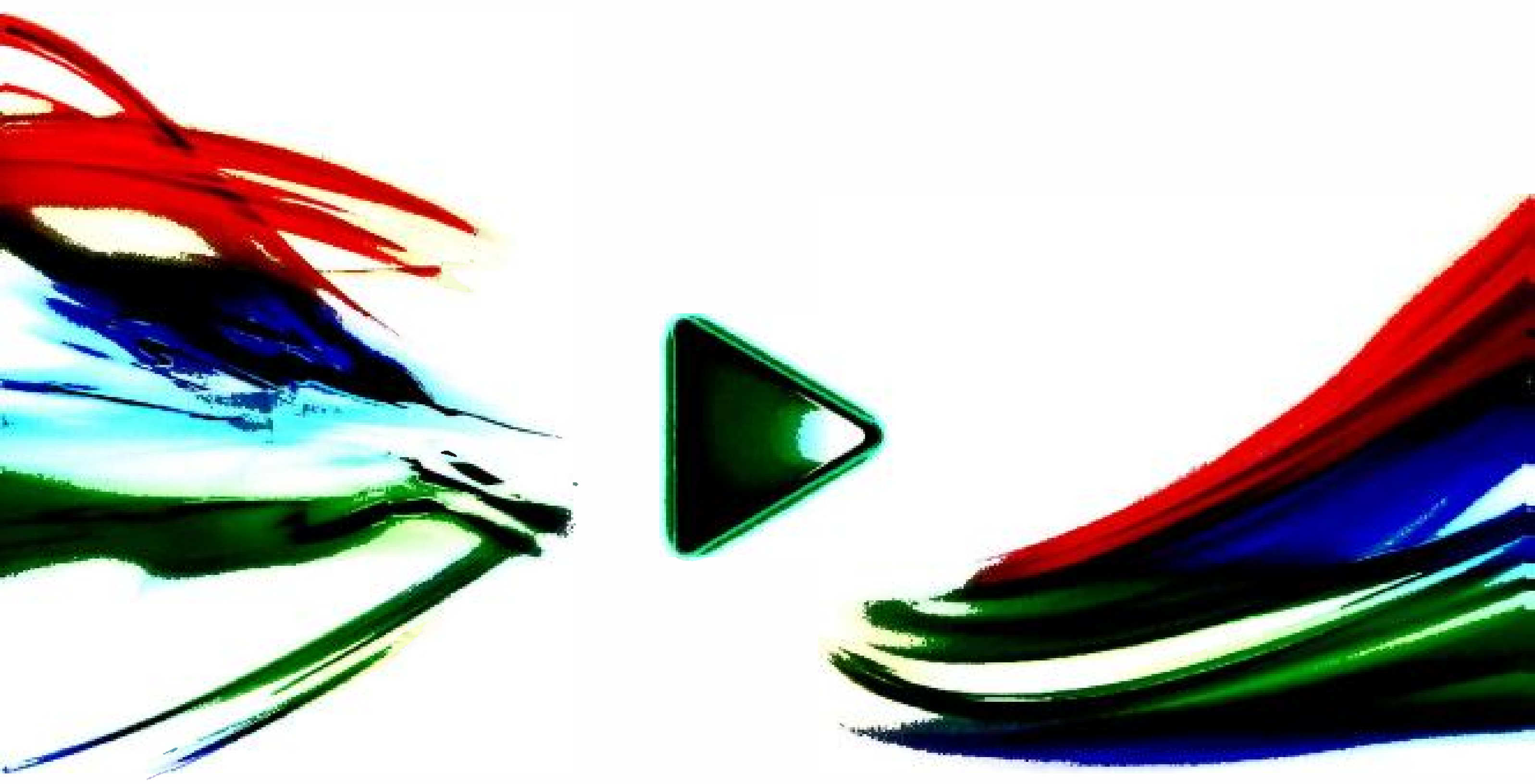
LIST RENTAL SERVICES
MeritDirect
Marie Briganti, (877) 796-6947
mbriganti@meritdirect.com
Kristen Killingback, (914) 368-1062
kkillingback@meritdirect.com
Chris Blohm, (914) 368-1031
cblohm@meritdirect.com


RIEDEL
See us at
NAB Booth C4937
WHEN IT REALLY MATTERS...
Real-Time Networks for Video, Audio, Data and Communications
ZENITH
SWISS WATCH MANUFACTURE
RED BULL STRATOS
MISSION CONTROL
ROSWELL
WUPPERTAL | BEIJING | BERLIN | GLENDALE | LONDON | MOSCOW | SÃO PAULO | SINGAPORE | SYDNEY | VIENNA | ZURICH
www.riedel.net

PLAYBOX TECHNOLOGY

From a Single Channel Playout Server to a Multi Channel Turnkey Broadcast Centre

It's All in a PlayBox



- Single and Multi Channel Playout Automation with interactive CG and Graphics
- Traffic Management • Media Asset Management and Archive • Newsroom System
- Remote Redundant Playout • Ad Insertion and DPI • Time Shift
- IPTV and Internet TV Solutions
- Available in HD / SD and DVB (ASI/IP)

NABSHOW Where Content Comes to Life

Join PlayBox on Booth N5435

General Sales Contacts: sales.usa@playbox.tv • **Website:** www.playbox.tv

- **PlayBox UK** +44 1707 66 44 44 • **PlayBox Bulgaria** +359 2 9703050 • **PlayBox Romania** +4 031106 5115
- **PlayBox Turkey** +90 216 3379402 • **PlayBox Adria** +381 600245901 • **PlayBox Albania** +355 44 80 45 52
- **PlayBox USA** +1 404 424 9283 • **PlayBox Asia Pacific** +603 7954 3400 z• **PlayBox India** +91 9811913333