AV Streaming Design Guide

For Professional AV Systems

3RD EDITION

Extron Electronics

INTERFACING, SWITCHING AND CONTROL

Extron AV Streaming Design Guide

The publication of this third edition of the AV Streaming Design Guide, comes at a time when we see expansion of AV streaming into more and more applications. Streaming has become ubiquitous in government, education, corporate and everyday consumer use.

Streaming stands out as the next big step as the AV industry continues to innovate. Streaming can enhance AV system capabilities by: 1) Extending visual communication beyond a room or building, to any destination globally, 2) Increase the scalability of AV distribution systems, 3) Simplify cable infrastructure management, and 4) Provide the ability to record, document and analyze audio, video, and graphic data and produced by a variety of AV sources. Each one of the opportunities described here can increase the operating efficiency within an organization, or reduce capital expenditure and operating costs for audiovisual installations.

Extron offers a mature range of streaming products that support a diverse set of applications. The SME 100 encoder and SMD 101 decoder are H.264 streaming products developed to provide video processing, control, and networking capabilities for professional AV systems. They are ideal for streaming within an enterprise and can provide compatibility with a variety of decoding devices. For demanding, quality-critical applications, Extron VN-Matrix high resolution streaming and recording products offer visually lossless quality, very low latency, and error resilience, exceeding the performance characteristics of standards-based compression systems.

These products are just the start. Look for Extron to continue to produce new streaming products that fulfill the requirements and challenges encountered by AV systems. Additionally, Extron's in-depth training programs, system design, and support provide an essential element to success in this evolving technology space.

Proficiency in AV streaming will be essential to future success in the AV industry. This guide has been created as a starting point for Extron customers to achieve that goal. It provides technical reference data, and real-world system designs that illustrate practical and effective streaming applications using Extron streaming products.

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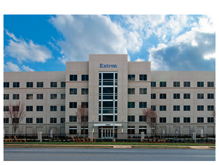
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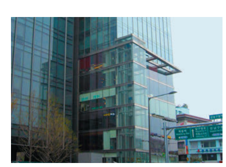
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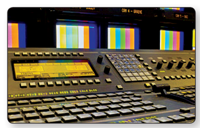

AV Streaming System Designs




Extron AV Streaming Product Solutions

Glossary

AV Streaming over IP Networks

> Continued advancement of networking and streaming protocols, hardware, and network infrastructure has made AV streaming a practical solution in more and more applications.

Streaming applications have been around for over a decade, and we encounter them every day in our personal and professional lives. Streaming is an powerful new technology to apply in professional AV systems; transporting media over the network where it can be extended to distant locations or recorded for reuse. Streaming makes many exciting, new applications possible, and in some cases, offers an alternative method to managing audio and video signals using traditional AV equipment.

The word “streaming” can be applied to many different industrial and consumer applications where audio, video, and computer-based media are delivered over IP networks. Streaming video continues to gain popularity because it delivers pre-recorded or live content to users over a network. Dedicated AV cables are not required between sources and display systems for live viewing, nor is there a requirement to download or store large content files prior to playback. Network connections, reaching virtually any location, can deliver content to a range of devices, including computers, mobile devices, media players, and dedicated video displays.

Customer expectations for streaming continue to increase as users become more and more accustomed to the delivery and consumption of visual content over networks in their daily lives. Continued advancement of networking and streaming protocols, hardware, and network infrastructure has made AV streaming a practical solution in more and more applications. The continued rise in popularity is also attributable to video being easily accessible on personal computers, laptops, iPads®, tablets, smartphones, and many other platforms in both consumer and professional applications.

Critical Factors to Streaming

Streaming applications require varying degrees of attention to a range of device, system, and user requirements such as:

1) Network environment and available bandwidth
2) Streaming bit rate
3) Latency
4) Scalability
5) Device compatibility
6) Application workflow

Figure 1-1. Each of these environments may use AV streaming, but the requirements and systems used to fulfill these applications can be very different

While bandwidth may be plentiful on some purpose-built, private networks, the public Internet is bandwidth-constrained and the quality of service cannot be quantified and assured. Network paths from source origination points to viewing endpoints may have bottlenecks, and not all content may pass easily in every private or public environment. A variety of techniques are used in streaming products to circumvent these problems, including encoding schemes that compress content to ultra-low bit rates, and buffering systems that allow reliable presentation and playback of video while operating on unreliable networks. However, latency, or delay is typically introduced into content that has been compressed and encoded to stream at low bit rates, which can affect its usefulness in some applications. The relevance of latency is often not well-understood until it is personally experienced.

Streamed content may originate as a live signal or in a stored format retrievable for viewing. Perhaps the most common example of live content is sporting events, where footage is streamed live to desktops or mobile devices. Consumer-focused Video on Demand - VoD services, including Netflix®, YouTube®, and Hulu®, all offer content that is intended to fit different consumption and viewing habits. These services thrive on their ability to make content widely accessible to a broad range of users anywhere over the Internet. The content is highly channelized with streaming offered in formats suitable to meet the needs of most viewers. These services deliver video that is stored for the viewer to play back at their convenience. The primary focus of these services has been on high availability and accessibility from public networks.

Streaming is used in many commercial and educational applications. IP security cameras, media content servers and live AV encoders are all examples of devices whose primary function is to transmit video content as packets of information across a network. All of these devices have different encoding, bandwidth, and bit rate characteristics, and they can all easily coexist as accessible resources on a network.

Two important questions to ask when considering an application for video streaming are:

- What are my application needs?
- What defines quality in my streaming application?

Video streaming is largely experiential. “Quality” is a relative term that depends on how streamed video is first encoded and then viewed. What constitutes an acceptable viewing experience for a consumer application will not always meet the needs of a professional application. For example, a viewer watching the news on a mobile device has a very different expectation of quality from a medical professional participating in a remote surgical procedure. While both expect to be delivered video they would define as being “high quality,” expectations and requirements for each application are different.

Not All Streaming Is Equal

The consumer world of video content is focused on requirements for accessibility to an infinite range of video experiences presented on consumer devices including PCs, smartphones and tablet PCs. Most of the streamed content used in these applications must use very low bandwidth because it is transported over the public Internet or wireless networks. In this environment, users are more interested in the availability and immediacy of video programming, and are less concerned with the latency. A modest amount of faults can be tolerated because the consumer is viewing content that is free or is sold at prices that are applicable to discretionary spending for personal entertainment. The user interface and control over the application will typically have a limited set of options and must not be too complicated to manage.

Moving beyond the consumer channels of streamed content and into professional and educational applications, the benchmarks for

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This design guide reviews subjects critical to understanding technical details relevant to streaming applications including fundamental characteristics of video signals, compression, encoding, network environments, streaming protocols and system scalability.

video transport and delivery take on a very different set of requirements.

Meeting spaces and classrooms for professional and educational users have long been designed and installed to produce high resolution images for viewing by many participants. The expectation in these environments is that content will not only be broadly accessible, but must be very high resolution as it will be presented on large displays. Some of these professional applications may also require low latency to support real-time communication. Use of private networks in professional environments usually offers greater bandwidth and quality of service - QoS, making higher performance possible using streaming equipment with a flexible set of controls. In these environments, attention must still be placed on keeping the interface simple for the user. Figure 1-2 identifies a series of streaming applications and categorizes them into one of two broad categories: Performance and Accessibility.

Consider the needs of large, multi-national organizations with global operations. Their desire is to integrate operations and facilitate communication between far-flung staff. Low-delay video transport can be a requirement for many of their streaming applications, so that individuals can interact, collaborate, or control equipment located across a country, or around the world. While network designs can support high bandwidths, IP networks cannot always guarantee AV signals can be delivered in the same way a directly connected cable can.

Table 1-1 identifies a variety of performance expectations that exist for different streaming applications. Each application can be referred to as "streaming" but they all require a different class of performance.

This design guide reviews subjects critical to understanding technical details relevant to streaming applications including fundamental characteristics of video signals, compression, encoding, network environments, streaming protocols and system scalability. Focus is given to streaming in professional AV environments, enterprise video streaming, distance collaboration systems and real-time mission critical applications. Each of the following topics will receive specific attention:

The nature of video & graphics: Video from cameras and production systems is different from graphic material that originates on a computer. Supporting these two different formats with quality in streaming applications requires that compression systems preserve both their common and unique attributes.

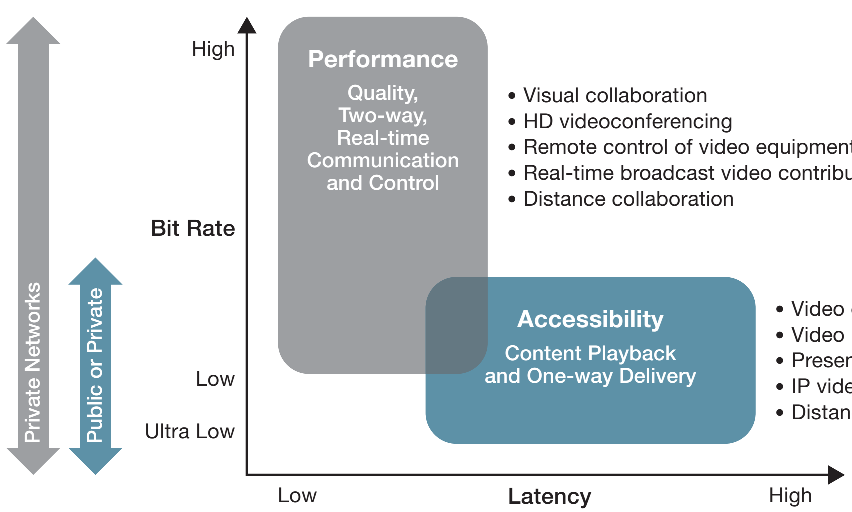

Figure 1-2. Streaming applications categorized into real-time Performance or content Accessibility applications

The need for compression: IP networks are limited in the amount of data they can pass between two points in real-time. Various degrees of compression are required to fit within the bandwidth available on different types of communication links. Different classes of streaming applications can accept various degrees of quality from lossy to visually lossless.

The relevance of delay: Low delay or low latency streaming of audio and video is essential to support natural, two-way communication, collaborative interaction or real-time control using video. However, low delay is not a requirement for every application and various degrees of latency are acceptable for different applications.

Realities of operating on IP networks: Network environments offer varying bandwidth and reliability. Many different methods exist for overcoming quality of service – QoS limitations across commonly-used switched and routed IP networks.

Compatibility: Standards-based and proprietary technologies are applied in compression, streaming transport protocols, encoding and decoding systems. Supporting compatibility and interoperability between streaming devices requires a holistic examination of all devices that will be used in a system.

Workflow and Scalability: A variety of workflow and scalability requirements exist in streaming systems based on the environment and application they are used in.

Each of the topics identified here contribute to the architectures that must be used to deliver scalable solutions. ■

Low delay or low latency streaming of audio and video is essential to support natural, two-way communication, collaborative interaction or real-time control using video. However, low delay is not a requirement for every application and various degrees of latency are acceptable for different applications.

Category	Consumer Delivery	Video on Demand	Public & Enterprise Safety	Web Broadcast	Real-Time Enterprise Communication
Application	Internet radio IPTV Cable TV	YouTube Hotel entertainment Distance learning Education	Security Surveillance	Road and weather Webinars	Videoconferencing Telepresence VoIP Collaboration Audiovisual
Content	Audio, low resolution video	Low resolution and high resolution video	SD and HD video	Audio, low res video, presentations, maps, and graphics	Audio, video, computer graphics, control
Delivery	One-way	One-way	One-way	One-way	Interactive communication
Input and Output	One to many	One to many	Few to few	One to many	Few to few
Presentation	Small screen	Small and large screen	Small and large screen	Small screen	Small and large screen
Network	Public	Public and private	Private	Public	Private & virtual private
Application Environment	Open	Open	Closed	Open	Closed
Control	Media subscription	Media subscription	Media subscription and camera control	Media subscription, voice, and keyboard chat	Near and far-end device control

Table 1-1. Requirements for streaming products vary by application

AV applications that use 4K displays are becoming more common. The Ultra HD - 3840x2160 and Digital Cinema Initiatives - DCI 4K standard – 4096x2160, both of which are referred to as 4K can be presented on an increasing number of projection and flat panel display products.

The Nature of Electronic Images

Electronic images are delivered in many forms. For the sake of clarity, here are a few definitions that apply to the topics covered in this guide as well as discussion of certain attributes of that imagery that is relevant to video streaming applications.

Standard definition video is defined as full-motion video images, with a frame rate and resolution, around 704×480 or less, running at 30 frames per second, or 60 fields per second interlaced. Analog NTSC and PAL composite video, S-video, and component video were once the most common signal formats used, but are now replaced by high definition signals.

High definition video is also full-motion, but with higher resolution, and generally conforming to parameters defined by the ATSC or DVB-T standards. Use of high definition video with a resolution of 1280x720 or 1920×1080 running at 50 or 60 frames per second, is a commonplace in consumer and professional applications.

The amount of motion and detail in an image can be just as important as its resolution when assessing the type of streaming product or bit rate that may be suitable for an application. Two different examples are presented in Figure 1.3. Fast-moving content collected outdoors at a live sporting event exhibits different attributes from talking head video collected indoors in a meeting room, which has much less motion and detail. Streaming products can take advantage of the lower motion and detail that exists in the talking head video and apply more aggressive compression techniques to stream at lower bit rates. Low latency may also be required to support real-time communication. Video from the outdoor sporting event will have much more detail and higher motion that is present in the environment and activity as well as panning and zooming by cameras. The combination of high motion and detail is also referred to as high entropy. The more entropy in an image, the more challenging it is to compress and preserve quality.

AV applications that use 4K displays are becoming more common. The Ultra HD - 3840x2160 and Digital Cinema Initiatives - DCI 4K standard – 4096x2160, both of which are referred to as 4K can be presented on an increasing number of projection and flat panel display products. Content presented on 4K displays may end up originating from computers, AV processing systems and digital media players using content transported over a network rather than from physical storage media such as DVDs and Blu-ray discs.

The term **graphics** refers to computer images which are typically, but not always, high resolution images. Resolutions of 1920x1080 or 1920x1200 are commonly used, and devices that can create 4K imagery such as computers and videowall processors have existed for some time. Video and high definition video usually operate in the component YUV color space with 16 bits per pixel, while computer graphics images normally utilize the RGB color space with 24 bits per pixel providing greater color detail than standards-based video formats.

Computer graphics differ from video in that the extent of motion in graphic images can vary widely. Graphic images may be entirely in motion, partially in motion, or stationary, even though the content is transported using a signal with a refresh rate, typically 60 frames per second. Three categories of motion are discussed here:

Real-time – 60 frames per second: Used in multimedia computer presentations with complex

Figure 1-3. The amount of detail and motion present in a video signal can vary dramatically by application

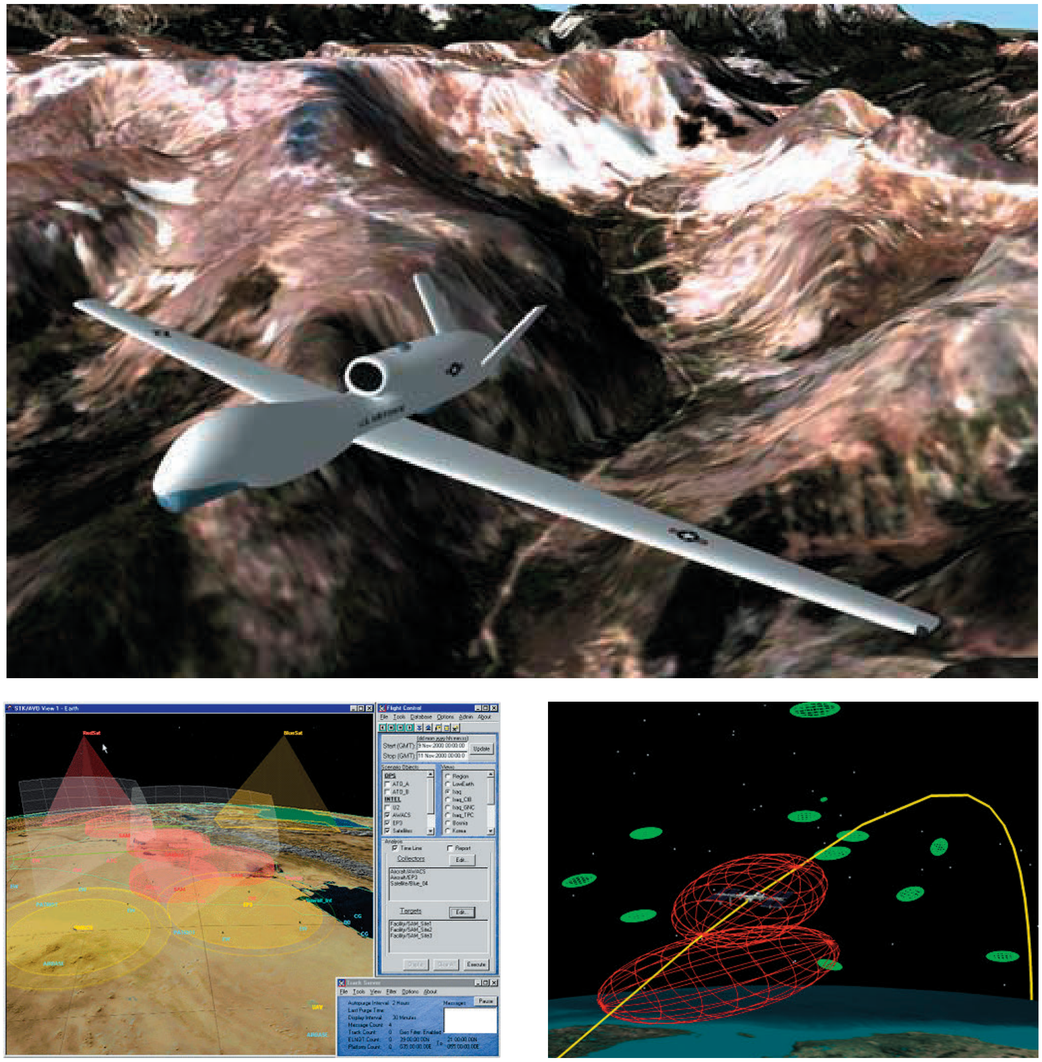

Figure 1-4. Examples of synthetically rendered fast-moving animations. Images from Analytical Graphics, Inc. www.stk.com

training and simulation content. Examples of real-time imagery include:

- Synthetically rendered computer animations which may contain high detail and motion. See Figure 1-4.

- Computers presenting video on the desktop at 30 frames per second. The native video frame rate is converted to 60 frames per second for the display. A single image or multiple images may be presented. See Figure 1-5.

Static images – One frame per second and slower: Maps, data screens and presentation slides which have very different amounts of detail may remain static for long periods, and the image update rate may be as slow as once every one to five seconds, and sometimes slower. See Figure 1-6.

Moderate change – 15 to 30 frames per second: There is a common intermediate requirement in which the great majority of the image remains static, but small items, such as

Computer graphics differ from video in that the extent of motion in graphic images can vary widely. Graphic images may be entirely in motion, partially in motion, or stationary, even though the content is transported using a signal with a refresh rate, typically 60 frames per second.

Figure 1-5. Motion video on the desktop. Multiple images (left) and a single MPEG window (right).

The ability to distinguish between video content with low entropy – low motion and detail and high entropy – high motion and detail can help identify the types of performance and streaming bandwidth that may be required to deliver quality video in different classes of streaming applications.

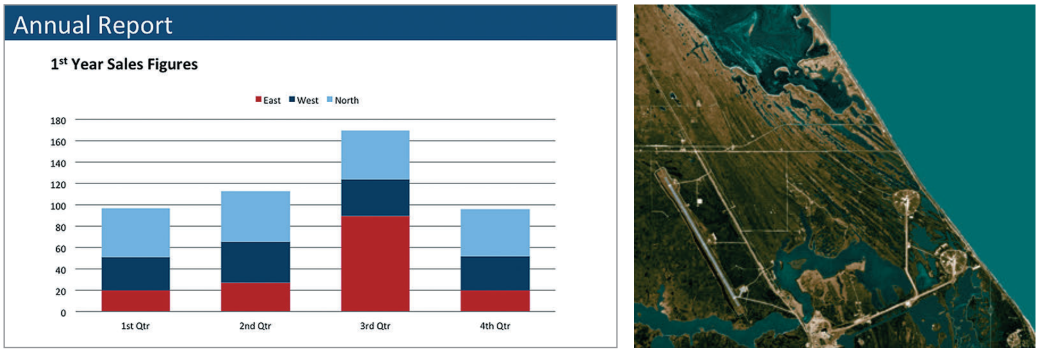

Figure 1-6. Examples of slow changing presentation slides and photographs

Figure 1-7. Examples of moderately changing images where status data or cursor movements may change frequently, but the vast majority of the background does not

text, status indication, alarm indication, and cursor movements require faster update. In the special case of cursor tracking, the update of the cursor movement must be 15 to 30 frames per second or equivalent, even if the remainder of the image has a slower frame rate. See Figure 1-7.

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Defining Networks

For the purposes of this guide, networks are defined as computer networks conforming to IEEE 802.3 Ethernet standards. Four data rates are currently defined for operation over twisted pair and fiber optic cables:

- 10 Mbps – 10Base-T Ethernet (IEEE 802.3)
- 100 Mbps – Fast Ethernet (IEEE 802.3u)
- 1000 Mbps – Gigabit Ethernet (IEEE 802.3z)
- 10 Gbps – 10 Gigabit Ethernet (IEEE 802.3ae)
- 40/100 Gbps – 40 and 100 Gigabit Ethernet (IEEE 802.3ba-2010)

The availability of 10 Gbps network equipment is growing, but broad use is not yet commonplace. The bandwidth available for streaming on enterprise and public network connections can vary dramatically depending on the age of deployed equipment, budgetary, and environmental factors. Connections made available by service providers can also vary dramatically based on geography and the cost of making last mile connections. Examples of the wide range of bandwidth available include:

- ISDN: 64-128 kbps
- T1 connection speeds of 1.54 Mbps and E1 connection speeds of 2.05 Mbps
- Cable broadband: Connection speeds from 30 Mbps to 100 Mbps are available in select locations, but the most common connection speeds available to consumers will support incoming traffic from 7 to 15 Mbps and outgoing traffic from 1 to 2 Mbps.
- LAN switching backbones from 10 Gbps to 2 Tbps

802.11 Wireless networks are also used for accessing streamed content, most frequently by consumer devices. Streaming to wireless consumer devices introduces additional network transport and quality of service - QoS challenges.

Traditional Means of Connecting Images from Source to Display

Until recent years, the traditional means of connecting images from source to display has used continuous analog or digital signals carried over AV cables dedicated to this single purpose. Standard definition video was historically distributed using a single coaxial cable carrying a composite signal. Graphics and professional video images expanded on this scenario using three to five coaxial cables; for example YCbCr or YUV for professional video and RGBHV for computer graphics.

Long distance transmission of video historically relied upon RF modulation techniques. As technology has matured, fiber optic distribution has become increasingly popular for long distance transmission of both video and graphics signals. Multimode fiber can be used to transmit signals up to 2 kilometers (1.25 miles) and singlemode fiber can be used to transport signals over 30 kilometers (18.75 miles).

In the broadcast field, well-established digital standards including 3G-SDI, HD-SDI and SDI are used, supporting distances up to 1,000 feet (300 meters) on a single coaxial cable for SDI and up to 330 feet (100 meters) for 3G-SDI and HD-SDI.

DVI is a signal format that is used for delivering images from computers and other AV sources. However the complex, multicore cables that are

The bandwidth available for streaming on enterprise and public network connections can vary dramatically depending on the age of deployed equipment, budgetary, and environmental factors.

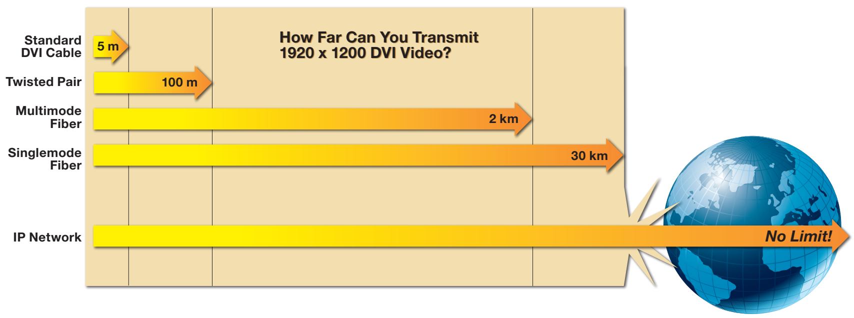
Figure 1-8. Medical applications make use of streaming to transport high resolution video and audio from surgical procedures for educational purposes. Audio and video was streamed over ten miles between the operating theater and auditorium in the photo to the left.

Many corporate, education, government, security, industrial, and defense applications require AV sources be accessible on multiple displays within a large area or spread over several locations.

used to transmit DVI signals are limited to very short distances, typically 5 meters (16 feet). Recent years have seen a rapid transition to the use of the HDMI signal to transmit digital AV signals. Conservative cable lengths for HDMI signals are in the same range as DVI, but equalization and extension products can be used to transmit HDMI signals up to 45 meters (150 feet). Frequently HDCP encryption is applied to the HDMI signal, providing copy protection for consumer playback products such as Blu-ray players or cable and satellite receivers. Once any HDMI source in an AV signal is encrypted, the entire system must be capable of maintaining encryption for that signal, preventing it from being presented on displays which are not HDCP-Compliant. HDCP encryption represents a unique challenge to overcome for an AV streaming application.

In many applications, structured cable such as Category 6 – CAT 6 is used to transport digital and analog audio and video signals using twisted pair transmitters and receivers. AV signals are extended greater distances using a cable that is easier to install while maintaining a high image quality. In addition to transmitting audio and video, control signals such as RS-232 and IR can be sent along with the video and audio over the same twisted pair cable eliminating the need for separate cable runs. Another advantage of using twisted pair cable is that the same contractors that are installing the house network cables can also pull additional Category cable for the AV system, often saving AV integrators time and money. Twisted pair transmission of digital and analog signals is ideal for distances up to 100 meters (330 feet).

Figure 1-9. Image transmission distance limits



Note: Cable equalization techniques support longer transmission distances with DVI/HDMI signals

A mature range of products are available for extending, switching and distributing AV content using all of the signals and cable formats identified here. Extron is a well-known supplier of AV distribution amplifiers, interfaces, scalers and matrix switchers for the transport of signals in AV systems. Figure 1-9 identifies conservative transmission distances for different types of cable used in AV applications. The Extron **Digital Design Guide** provides a comprehensive reference for digital AV signal formats, products and solutions managing and controlling modern AV systems.

AV Distribution Systems

Many corporate, education, government, security, industrial, and defense applications require that AV sources be accessible on multiple displays within a large area or spread over several locations. Some applications may require hundreds of inputs and outputs.

AV distribution systems are diverse, ranging from a series of flat panel displays, to simple or sophisticated presentation systems, to control rooms used in transportation, public safety, or military applications.

In all these applications, the conventional distribution of AV signals has required use of dedicated coaxial cable, twisted pair cabling, or optical fiber, as well as customized switching and distribution equipment.

Why Use an IP Network for AV Signal Distribution?

There are several potential economic and operational advantages that can be achieved streaming AV signals over computer networks. One or many of the following advantages can be realized:

- **Extend the reach of signals to destinations not practical to connect using traditional methods.** Destinations could be within a building, across a campus, throughout a country, or around the world.

- **Reduce the need to install a dedicated cable infrastructure,** particularly to devices that may change their location over time.

- **Increase flexibility in the choice of images.** Distributing video and graphic signals on the same cable offers greater flexibility for the endpoint displays.

- **Increase flexibility for the location of displays.** Displays can be added or moved at any time to any location where a network connection is available. AV signal formats can change over time.

- **Increase accessibility by allowing display of content at any network location on a variety of devices.**

- **Control distribution of content to specific, authenticated users.**

- **Record and play back content streamed over the network.**

- **A networked solution can reduce the amount of conduit, cable, weight, and resulting energy use.** ■

Traditional signal distribution technologies continue to provide quality and cost-effective connections for audio, video, and computer graphics. Streaming solutions expand system capabilities, increase the flexibility of endpoint locations, and extend geographic reach. In many applications with a pre-existing network infrastructure, the benefits of real-time collaboration over long distances, and scalability requirements necessitate the use of IP streaming as the primary video distribution infrastructure. Successful AV professionals will be proficient at integrating both traditional signal distribution and IP streaming solutions in combination to provide the best current and future value for customers.

Streaming solutions expand system capabilities, increase the flexibility of endpoint locations, and extend geographic reach.

Table 1-2. Comparison of signal transmission methods

	Coax	Twisted Pair	Fiber	IP Networks
Installation	Intra-building	Intra-building	Intra or extra building	Intra or extra building
Cabling	Fixed plant	Fixed plant	Fixed plant	Flexible plant
Scalability	Limited	Limited	Limited	Unlimited
Maximum transmission RGBHV Analog	1,000 ft (300 m) - with peaking amplifier	330 ft (100 m)	30 km	Unlimited
I/O node locations	Fixed	Fixed	Fixed	Flexible
Integrated recording	No	No	No	Yes
Air-gap possible	No	No	No	Yes
Max. distance (RGBHV)	Up to 300 m with DA's	300 m with CAT 5	> 30 km singlemode fiber	Unlimited
Transmission	Analog	Analog	Digital	Digital
System	Passive	Active	Active	Active
Influence from EM fields	Yes	Yes	No	No
Radiation	Yes/Un-secure	Yes/Un-secure	No/Secure	Twisted pair, Yes/Un-secure Optical, No/Secure
Remote Power	No need	Possible	Not possible	Possible
Cost	Function of cable length	Fixed cost for T & R	Fixed cost for T & R	Encoder and decoder cost, plus network

Data compression can broadly be defined as lossless, in which 100% of the original information is recovered after compressing and then reconstructing the data, or lossy compression, in which some data is permanently lost through the data reduction process.

The Need for Compression

When video and graphic signals are distributed over computer networks, their data rates must fit within the available bandwidth capacity of the network. An uncompressed image with a resolution of 1920x1200 that updates at 60 fps would require a data rate of 3.3 Gbps. See Table 2-1. However, this is three times the capacity of a Gigabit Ethernet network, and in fact, data rates for any resolution greater than SVGA 800x600 exceed 1 Gbps, as shown in Table 2-2. Even for VGA, and standard definition video, the data rate exceeds the capacity of a standard 100 Mbps Ethernet network.

The unrealistically high bandwidths required to deliver uncompressed video or computer graphic images across a network underscores the need for a digital image compression mechanism that substantially reduces bit rates while delivering high quality graphic or video imagery. For example, to reduce WUXGA computer-video to 50 Mbps, which could fit over a standard Ethernet network, a compression ratio of approximately 66:1 is required.

Variable	Value for WUXGA	Value for other resolutions
Color parameters	3	(e.g. RGB or YUV)
Bits per color	8	(or, e.g., 4, 10, 12)
Horizontal pixels	1920	(or e.g., 720, 1024, 1600, 1920)
Vertical pixels	1200	(or e.g., 480, 768, 1080, 1200)
Frames per second	60	(or e.g., 24, 25, 30, 85)
3x8x1920x1200x60 = 3,317,760,000 (3.3 Gbps) bits per second		

Table 2-1. Calculating the base bandwidth at WUXGA

Lossless vs. Lossy Compression

Real-time video, graphics, audio, and computer files can be compressed to reduce the bit rates necessary to transport or store them. Data compression can broadly be defined as **lossless** when 100% of the original information is recovered after compressing and then reconstructing the data, or **lossy** compression, in which some data is permanently lost through the data reduction process. Lossless compression is commonly used for reducing computer file sizes into the ZIP format, and for still image files using PNG or TIF formats. While it guarantees full integrity of the original files, lossless compression does not provide nearly enough data reduction to fulfill motion image applications on commonly used networks.

To efficiently compress graphic or video data, a variety of lossy compression techniques are employed that utilize sophisticated algorithms, which take advantage of our visual perception abilities to discard or greatly reduce data for parts of the image considered imperceptible or redundant to the human eye. High performance lossy compression can yield images that appear to be **visually lossless** by a trained observer,

Signal	RGB color	Bits per color	Horizontal pixels	Vertical pixels	Frames per sec	Base image Bandwidth*	Approximate data rate	Multiple of NTSC rate
VGA	3	8	640	480	60	442,368,000	442 Mbps	3.5
SVGA	3	8	800	600	60	691,200,000	691 Mbps	5.5
XGA	3	8	1024	768	60	1,132,462,080	1.1 Gbps	9.0
SXGA+	3	8	1400	1050	60	2,116,800,000	2.1 Gbps	16.8
UXGA	3	8	1600	1200	60	2,764,800,000	2.8 Gbps	21.9
WUXGA	3	8	1920	1200	60	3,317,760,000	3.3 Gbps	26.5
NTSC**	1.5	8	720	486	30	125,971,200	126 Mbps	1

* Base Image Bandwidth is for the active picture only. The blanking period is not included. Blanking can add 10 to 15% to the bandwidth listed above.
** NTSC video uses a 4:1:1 color scheme providing 1/4 the color information of RGB.

Table 2-2. Examples of the bandwidth of common computer signals compared to standard definition video

indistinguishable from the original. Lossy compression methods are typically implemented with low bit rate as a priority over image quality. In such cases, lossy images are acknowledged to be of lower quality than the original, but are usually adequate or better than adequate for the intended application. The H.264 compression format is typically applied to lossy video applications.

Lossy compression formats for audio, such as MP3 and AAC, capitalize on characteristics of our aural perception to reduce the data needed to transmit and store digital audio. This Guide will focus on video image compression and its application in AV streaming applications.

Signal	Data Rate
SDI (SMPTE 259M)	270 Mbps
HDSDI/HD-SDTI (SMPTE 292M)	1.485 Gbps
3G SDI (SMPTE 424/425 M)	2.97 Gbps
DV (SMPTE 314M) Uses light compression	25 or 50 Mbps

Table 2-3. Examples of data rates for serial digital video signals used in broadcast applications

Basics of Image Compression

There are two approaches to compressing motion images that are in common use today:

1. **Spatial compression**, which reduces the amount of information needed to describe a single image frame. Spatial compression is also known as **intra-frame compression**. Still image formats such as JPEG or JPEG 2000 apply spatial compression.

2. **Temporal compression**, or **inter-frame compression**, which reduces the need to send full frame data for every frame, while still retaining essential motion details in the reproduced image. The MPEG-2 and H.264 compression systems apply temporal compression in the vast majority of streaming applications.

Spatial Compression

Figure 2-1 summarizes the process of spatial compression for encoding a single frame.

The first step is to apply a "**transform**" to the pixel data. This process transforms, or changes, the image data from the spatial domain to the frequency domain. This means that instead of being described by a series of individual pixel values, the image is described by reference to a series of patterns that make up the image. This is similar to the way that, with audio signals, any sound can be made up by adding different proportions of multiple sinewave frequencies. By expressing image data in the frequency domain,

The MPEG-2 and H.264 compression systems apply temporal compression in the vast majority of products in which they are used.

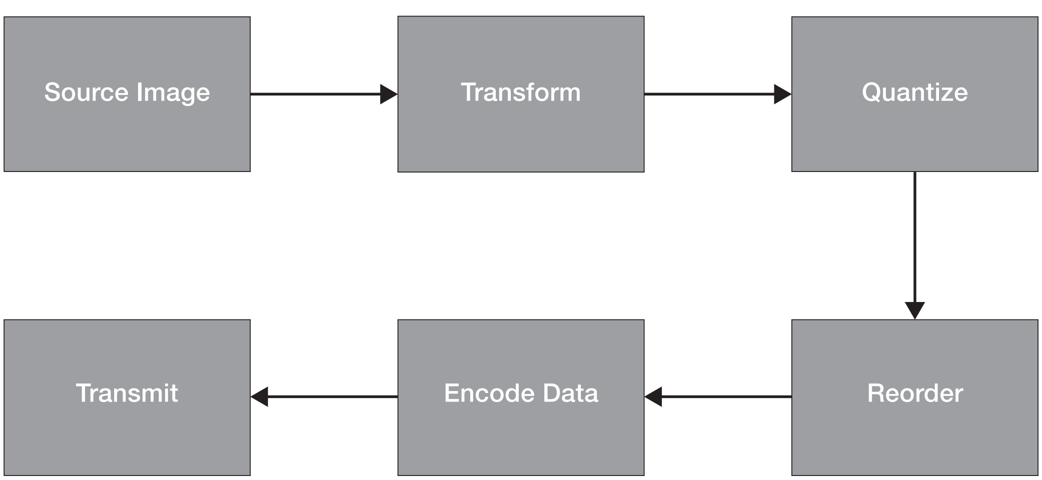


Figure 2-1. The processes used to encode a compressed image. Compression takes place during "quantize" and "encode".

This transform does not reduce the amount of data that makes up the image. What it does is make it easy to identify those spatial patterns that are most significant, and those that can be omitted based on characteristics of our visual perception.

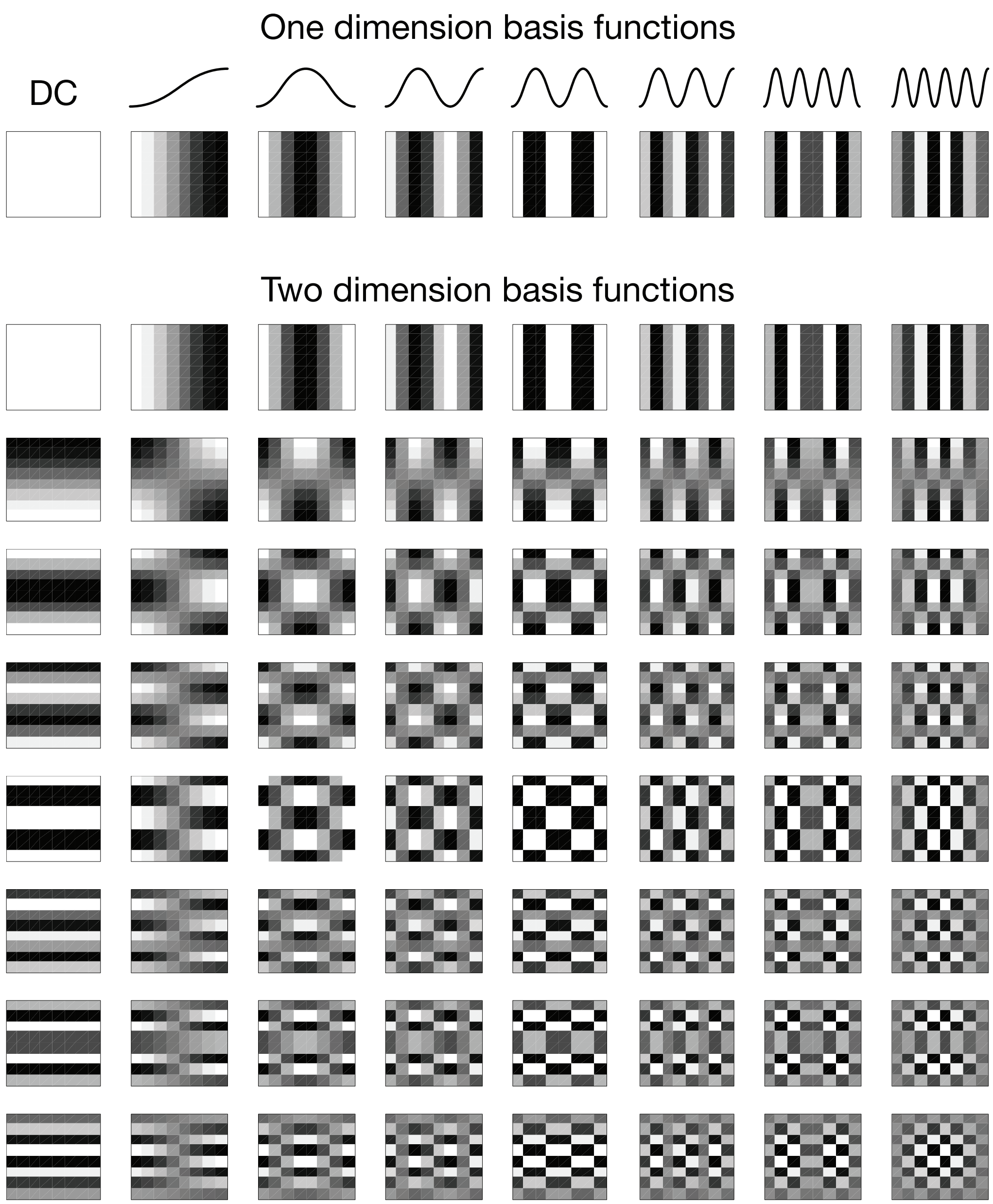


Figure 2-2. The "Basis Functions", or patterns, used in the DCT. These patterns are being related to the most commonly used block of 8×8 pixels.

an image can be made up by adding patterns in various proportions ranging from low to high spatial frequencies. See Figure 2-2. The transform is applied in blocks of pixels, usually 8x8, and attributes a coefficient to the mathematical function for each pattern, which essentially denotes the pattern's relative contribution to the image.

This transform does not reduce the amount of data that makes up the image. What it does is make it easy to identify those spatial patterns that are most significant, and those that can be omitted based on characteristics of our visual perception. Patterns corresponding to high frequency spatial components are often negligible, since their contribution to the image usually is very small and unnoticeable to the human eye. They are also often masked by other details in the image. Therefore, the transform coefficients for the higher frequency patterns can be set to zero, significantly reducing the data.

The entire transform process must be reversible, since when the image is finally decoded, it will be pixel data that is required to drive the display. Typically, a very good reconstruction of the image can be achieved by using as little as 5% to 10% of the original coefficient data.

The "**quantize**" step produces a reduced set of unique coefficients, and these make the most significant contribution to the image. The majority of the remaining coefficients round to zero.

The "**reorder**" step prioritizes the most significant coefficients first. This improves the efficiency of techniques applied in the encoding process.

The "**encode data**" step employs mathematical techniques that reduce the amount of data transmitted without losing image information. They include Run Length Encoding, which codes a long string of bits of the same value as simply the length of the run rather than the individual bits and Variable Length Coding, which assigns short codes to frequently occurring values.

The most widely used transform method is **Discrete Cosine Transform - DCT**. The DCT is popular because it is simpler to implement than other transforms and it can provide a high compression efficiency. The DCT has been employed by JPEG, MPEG, and H.264 compression systems. The DCT, having been applied in the MPEG standards and H.264, is the most popular transform in use today.

The **Discrete Wavelet Transform - DWT** is another transform that has been applied in a select class of streaming products. It provides high efficiency compressing still frames. The DWT is applied in JPEG 2000 and the Extron PURE3® codec, which will be examined further.

Temporal Compression

The primary strategy for temporal compression is to compare a sequence of frames, and apply techniques that identify and estimate only the change and motion. The greatest compression can be achieved for images that are or appear to be stationary. If a sequence of images depicts, for example, a ball as it traverses in space while the rest of the picture is fixed, the data will be limited to the ball and how its position changes in successive frames.

When using temporal compression, full frame information must periodically be sent, to ensure that a full reference frame exists to continue delivering accurate images if there is a disruption in the data stream. Such frames are referred to as **intra-coded** or **I frames**. Frames between the I frames are referred to as P and B frames. P frames use forward prediction, and are coded relative to the nearest previous I or P frame, which can also be referred to as key frames. B frames use bidirectional prediction and use the closest past and future I and P frames for reference. I frames are not dependent on other frames for decoding.

Use of I, B and P frames provides for high compression ratios, but it also leads to the complication that frames need to be transmitted out of order. In Figure 2-3, an original sequence of ten frames is shown, applying a typical sequence of I, B and P frames. The transmitted sequence is reordered because frame 2, a B frame, cannot be reconstructed without the full frame 4 information. The pattern of frames between any two I-frames in the original frame sequence is known as a **GOP - Group of Pictures**. Streaming products will use different GOP lengths and apply different combinations of I, P and B frames based on the applications they intend to serve. Low delay streaming products that operate at higher bandwidths may use a GOP of four in an IBBP sequence. Low delay, low bandwidth products may use only I and P frames and apply a GOP that may range from 30 to 200 frames.

The primary strategy for temporal compression is to compare successive frames, and to identify and limit the data transmitted to only the information that changes from one frame to the next.

Implementation of a standards-based codec does not necessarily guarantee interoperability.

Inter-frame prediction increases compression efficiency using P and B frames, applying a technique referred to as motion estimation. Each frame of video is divided up into a structure of 16x16 pixel "macroblocks." As each block is being encoded a reference frame is examined for a similar or matching block. If one is located, the new block can be encoded as a vector that points to the position of the matching block in the reference frame. The two blocks will most likely not be identical so the difference between the two blocks, the predictive error data will also be sent along with the motion vector. P frames will apply this process to a reference frame in the past and a B frame will apply this process to reference frames in the past and future.

More advanced compression techniques for spatial and temporal compression exist and some will be identified in greater detail as part of an examination of the H.264 standard.

Proprietary and Standards-Based Compression

Many different compression methods exist for still and motion images. They have evolved for various technical and commercial reasons and as technology has advanced the underlying techniques have become better understood. Some methods are proprietary, and in many applications, can deliver performance that sets them apart as the most practical option. Others have been introduced by the **ISO - International Organization for Standardization** and **ITU - International Telecommunication Union** standards organizations.

Standards for Still Image Compression

For still images, there are two leading standards, both developed by the Joint Photographic Experts Group - JPEG of the ISO. Both are intended for images of any size.

- **JPEG** is the original standard, and is based on the DCT. The DCT transforms blocks of 8x8 pixels.

- **JPEG 2000** is a newer standard, and is based on the DWT to achieve more efficient compression. This standard employs variable pixel block or tile sizes as opposed to standardized blocks. The tile size can be defined differently for each application or image within the source image creation application.

Both standards require symmetrical processing, which means equal computational power to encode and decode the information. Although primarily intended for the exchange of photographic still image files, JPEG and JPEG 2000 are used for motion video, simply by encoding each frame separately.

Motion JPEG or M-JPEG has been widely used for video, both in semi-professional video and in the security surveillance field. Because each frame

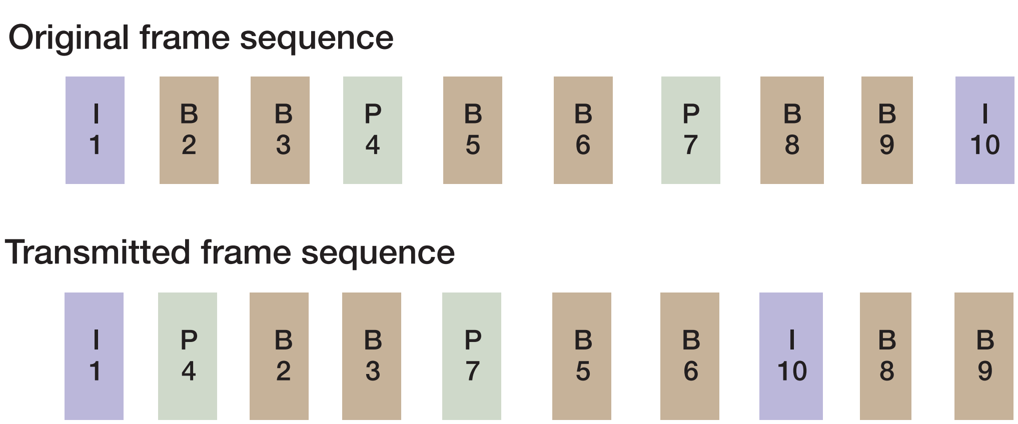


Figure 2-3. Encoded I, P and B frames are transmitted out of order to simplify the decoding process.

is encoded separately, there is no frame inter-dependence which facilitates video editing and random access. Consumer and semi-professional digital video work on the same basis, but are not fully JPEG compliant.

JPEG 2000 has been chosen by the Digital Cinema Initiatives - DCI group within the motion picture industry as the preferred method of distributing digital cinema first-run features to theaters. Digitally encoded films are stored, and played back from hard disk-based storage on specialized video players where high data transfer rates can be supported internal to the player. JPEG 2000 has also been used in video streaming products.

Standards for Motion Image Compression

Motion image compression standards have evolved from two major standards organizations, ITU and ISO. The ITU and ISO codec standards families have much in common. They both use the DCT as the transform and primary basis of spatial compression. They also employ inter-frame or temporal compression to take advantage of the redundancy between successive frames.

- The ITU has developed many standards for audiovisual systems and video encoding. The ITU H.26X standards were originally developed for videoconferencing using the public telephone network. The most commonly applied codec today is H.264, which was developed to improve upon the compression efficiency of H.263.

- The ISO developed standards within their Moving Picture Experts Group - MPEG with the aim of meeting the needs of broadcast television and consumer products. The compression techniques developed by MPEG are the most relevant to this guide.

The MPEG compression standards evolved as technology permitted new applications.

- **MPEG-1**, the original MPEG standard, was designed for video on a standard CD. It delivered progressive scan images with bit rates up to around 1.8 Mbps. The common resolution was 352×240 at 29.97 fps for an NTSC image. MPEG-1 was the compression standard for the VCD or Video CD format.

- **MPEG-2** addressed the limitations of MPEG-1 and became a huge success. It served as the basis for digital satellite and cable, HDTV broadcasts, and DVD-Video. In its most common implementation, MPEG-2 supported both progressive and interlaced standard definition video (e.g. 720×480 at 30 fps) with bit rates up to 15 Mbps. In practice, excellent results were obtained using 2-6 Mbps for standard definition video. "Higher Level" variants of MPEG-2 were applied to high definition video at resolutions up to 1920×1080 and bit rates from 19 to 80 Mbps.

- **MPEG-4** built upon the previous MPEG standards with a priority of achieving quality video compression at very low bit rates, improved performance when errors are experienced, and meeting consumer demand for multi-platform support. MPEG-4 is supported in a variety of devices and software media players. One part of the standard, MPEG-4 Part 10 or MPEG-4 AVC, is the same as H.264.

H.264, or its equivalent name, MPEG-4 Part 10 or AVC, was the result of a joint effort between the ITU and the ISO Joint Video Team - JVT. Competing video compression formats and standards exist which apply similar techniques to H.264 including SMPTE VC-1 from Microsoft, Google VP9 and Theora by the Xiph.Org Foundation. Each of these formats was produced to meet different technical or commercial objectives of their developers.

The H.264 standard applies an asymmetrical processing system, meaning that the bulk of the

Motion image compression standards have evolved from two major standards organizations, ITU and ISO.

The H.264 standard applies an asymmetrical processing system, meaning that the bulk of the computational load is on the encoder, allowing for a relatively simple decoder design.

computational load is on the encoder, allowing for a relatively simple decoder design. The H.264 standard defines the encoded bit stream and how the decoder must work, with the aim of enabling a simple, inexpensive decoder, which in turn would lead to affordable consumer products that decode content such as HDTVs and DVD players.

Implementation of each manufacturer's encoder is proprietary and a range of performance criteria can be pursued within a product design. The H.264 standard includes 21 profiles and 17 levels for video encoding. Each profile applies different encoding techniques and complexity and each level specifies the maximum resolution, frame rate, and bit rate that a decoder may use. An H.264 decoder may be compatible with one or many different profiles.

H.264 dramatically increased the efficiency of motion compression in many ways, and some of them are described here. It added intra-frame prediction for encoding I-frames in a method that reduces bandwidth while maintaining quality. Intra-frame prediction takes advantage of the spatial redundancy within a frame. It checks the macroblocks to the left and above the macroblock currently being encoded to determine if there is a close match. Once it finds a match, intra-frame prediction uses a vector to point to it and encode the difference. See Figure 2-4. For example, a frame showing a background with a large surface area will benefit from intra-frame prediction because the spatial redundancy can easily be exploited, resulting in increased compression.

H.264 divides frames into regions called **slices**, each of which contains a collection of macroblocks that are coded without reference to other slices. H.264 then applies P and B frame coding on these slices instead of entire frames. Each slice can be decoded independently of other slices, providing the additional benefit of confining any errors that are experienced to a slice rather than allowing them to propagate across an entire frame.

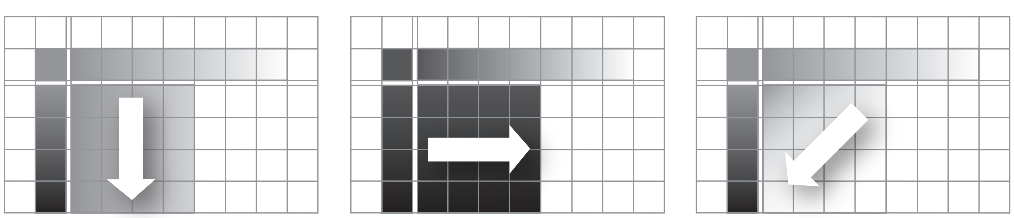
H.264 offers motion compensation by dividing the 16x16 pixel macroblocks into smaller sub-blocks and tracking their individual motions. This allows P and B frame prediction based on up to 16 successive or preceding frames for reference, rather than just a single past or successive frame as in MPEG-2.

The H.264 standard has also reduced the "blocking" or "tiling" artifacts that were frequently observed in MPEG-2 making the edges of macroblocks less visible. See Figure 2-5.

H.264 and the Future

H.264 is now clearly the most popular method for streaming AV signals in consumer and professional applications, among them Blu-ray Disc, digital satellite broadcasting, video-on-demand, Web or network video streaming, and videoconferencing. It offers the ability to encode to bit rates that range from 64 Kbps to 240 Mbps supporting many different classes of applications.

In January 2013, the High Efficiency Video Codec – HEVC also known as H.265, was approved by the ISO/IEC and ITU Joint Collaborative Team on Video Coding – JVT-VC. The primary objective for developing H.265 was to improve the bit rate efficiency of H.264 by 50%. It also supports use



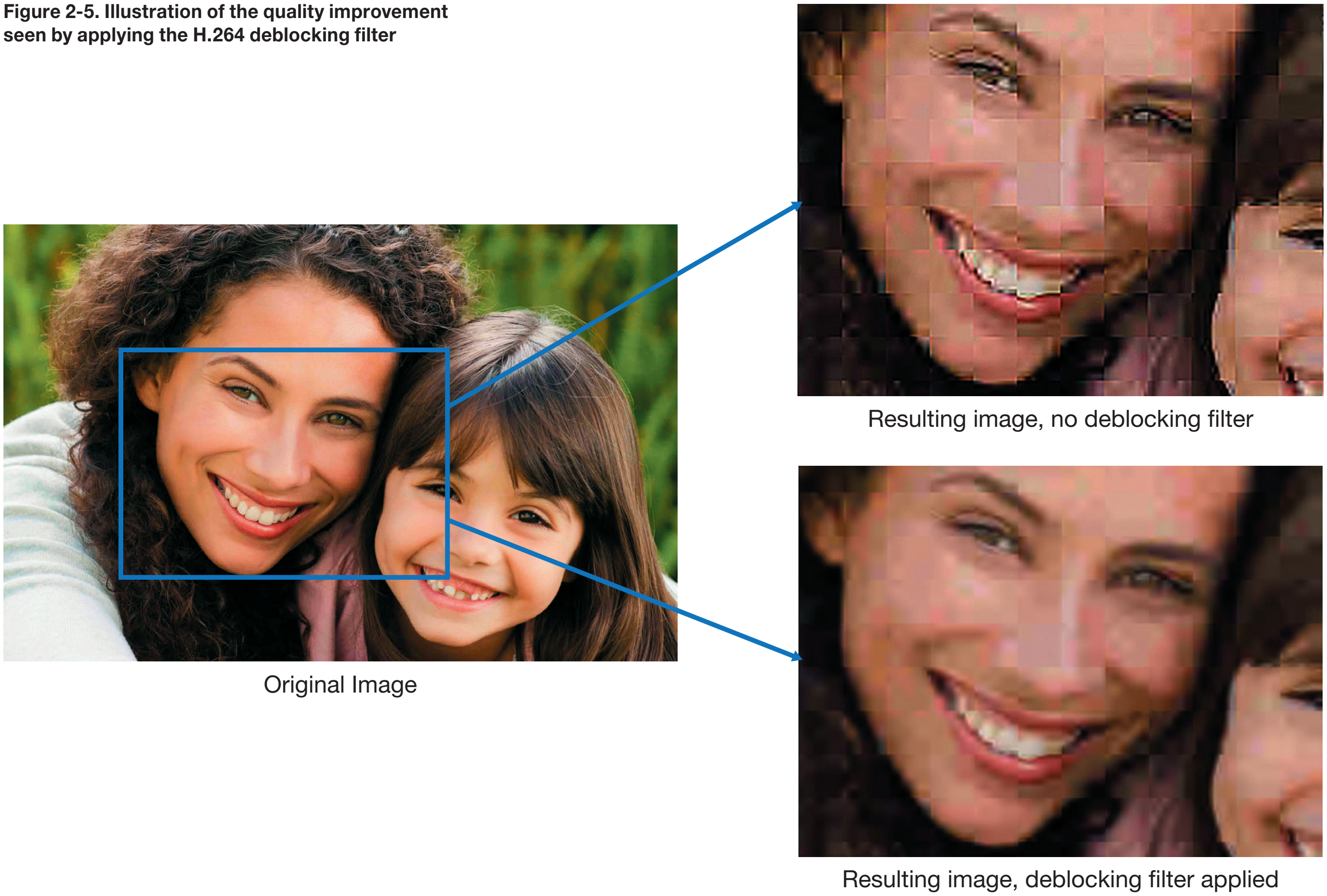
Pixels from the bottom row of the vertically adjacent macro block are copied into the intra-coded block

Pixels from the right side of the macro-block to the left are copied into the intra-coded block

Pixels from the adjacent macroblocks above and to the right are copied diagonally into the intra-coded block

Figure 2-4. Illustration of three of nine prediction modes applied to 4x4 luma macroblocks in H.264 intraprediction

Figure 2-5. Illustration of the quality improvement seen by applying the H.264 deblocking filter



with video resolutions up to 8K – 8192x4320 and provides methods for applying parallel processing. HEVC encoder designs can trade off computational complexity, compression rate, robustness to errors, and encoding delay, depending upon the target application. The 8x8 macroblock structure used by H.264 has been replaced with coding tree units – CTU which can be applied to larger block sizes up to 64x64 allowing the picture to be partitioned into various structural sizes, producing greater coding efficiency. This, among other improvements to coding and compression techniques produce bit rates that are 40% to 50% lower than those produced using H.264. Similar to H.264, specifications exist for multiple profiles and levels that provide different quality, bit rate, and coding complexity within the standard. Provisions exist for extensions to the defined profiles to support improvements to the standard and support future application requirements.

Since H.265 provides a dramatic improvement in compression efficiency, products implementing the standard will probably be first adopted in applications where network bandwidth is most scarce. Examples include streaming over wireless networks or Internet conncections. Encoders for streaming new, high resolution standards such as 4K and 8K may also adopt H.265, since there is so much more original pixel data to compress and the higher efficiency will be valuable. Applications that adopt H.265 first will also be those where end-to-end solutions can be supplied.

Data compression is not the only consideration for sending motion images over networks. The characteristics of computer networks introduce a number of practical problems that must also be addressed by choosing the right network architecture and transport protocol for successful AV streaming. ■

In January 2013, the High Efficiency Video Codec – HEVC also known as H.265, was approved by the ISO/IEC and ITU Joint Collaborative Team on Video Coding – JVT-VC

An Ethernet switch intelligently routes internode traffic so that nodes only receive traffic addressed to them.

There are a number of characteristics of networks that have to be taken into account when streaming AV signals. When a conventional analog or digital video signal is sent from a source to a display, the image is transmitted in real-time without delay or latency. The signal itself is continuous, and in general, it is not subject to any unpredictable degradation.

Packetized Data

If a digital image data stream is sent across a network, it must be packetized. When transmitted on the network, the nature of the data is, for the most part, inconsequential and all data, whether audio, video, text, or files, is treated in the same way. Before it can be sent over a network, however, the data must be formatted into IEEE MAC - Media Access Control frames, also known as packets, as illustrated in Figure 3-1.

For any IEEE MAC frame:

- Data carried within must be no larger than 1,500 bytes.
- Data must include additional overhead from the applied network protocol, for example, TCP/IP or UDP.

An Ethernet Local Area Network, or LAN, will have a number of nodes or connection points, and in principle all nodes can communicate with each other. These nodes access the network through Carrier Sense Multiple Access/Collision Detection or CSMA/CD. This means that when not transmitting, all nodes are listening. When a node transmits, no other node attempts transmission. However, signal speed variations mean that another node may send before transmission is completed, increasing the probability of a data collision on the network. Such collisions are detected and the competing parties "back off" for another attempt. This method of using a hub connection point for a network would function well for a small network comprised of only a few devices, but becomes inefficient if the network has high traffic. Therefore, networks are, in practice, managed by the use of various switching and routing devices.

The following technical points apply when considering use of networks for transporting real time AV signals:

- An Ethernet hub simply allows nodes to be connected together and CSMA/CD applies.
- An Ethernet switch intelligently forwards internode traffic so that nodes only receive traffic addressed to them. This reduces or eliminates bus contention at the local level. A switch also allows a node to simultaneously transmit and receive in duplex mode.
- An Ethernet bridge is a two-port switch used for segmenting networks or joining dissimilar media.
- An Ethernet router connects multiple networks, and connects to networks of different types.

Routers and switches use routing tables to determine how traffic is directed. These can be dynamic, in the sense that they are generated as

Synchronizes internal clock generator

Indicates type of payload

Preamble	Dest address	Source address	Length of Data	IEEE802.2 Header Optionally with snap extensions	Data 46 - 1500 Bytes at 10 MHz	CRC
8	6	6	2	3 or 8	Variable	4

Number of bytes

Figure 3-1. The IEEE802.3 Media Access Control Frame

needed by examining network traffic. However, they can also be static, imposing strict rules about how traffic is directed. This factor is of great importance with respect to transmitting image data streams over networks. In practice, network devices such as routers and firewalls may block streamed traffic if not pre-programmed to pass along certain streaming protocols.

Network Protocols

For effective communication over networks, there must be some level of formality regarding the way communication is conducted to ensure interoperability between different systems. The ISO proposed a model for this in its Open Systems Interconnection - OSI model. OSI defines seven different layers for any intercommunication protocol, starting at the bottom with a physical layer, which might be Ethernet, wireless, or some standard serial communications method. At the top is the application layer, which delivers a final output to the user or a specific application. See Figure 3-2.

Layer 1 defines the physical standards for interfacing between devices and transmission mediums such as cables, and includes Ethernet hubs. Layer 2 defines methods used to transmit data between two physical connections. In the case of Ethernet, MAC addresses are used to identify senders and intended receivers. The simplest, off-the-shelf Ethernet switches and bridges used by consumers typically fall into the Layer 2 classification. Layer 3 protocols define methods by which data is transferred across multiple networks. This is also referred to as internetworking, and IP - Internet Protocol addresses are used at this layer. At Layer 4, reliable, transparent data transfer or transport service is provided. Layer 5 manages sessions and connections between devices. At Layer 6, data is transformed or converted to a form that the application layer can use. Finally, at Layer 7, network applications interact with the user or specific devices and applications.

The OSI model is used as a reference and, while some systems follow the full model, others simplify it by combining the functions of certain layers. In particular, the TCP/IP protocol stack, also known as the Internet Protocol Suite, which is the basis of standard Ethernet communication, has only four layers, as indicated in Figure 3-3. TCP/IP stands for Transmission Control Protocol / Internet Protocol.

In the TCP/IP Protocol Stack, the Network Interface layer combines the functions of the Physical and Data Link layers of the OSI model. The TCP/IP Protocol Stack also combines the Session, Presentation, and Application layers of the OSI Model into a single Applications layer.

Use of Internet Protocol 4 - IPv4, which uses a 32-bit address, is prevalent in AV systems. IPv6, with 128-bit addressing, has been introduced

Layer 3 protocols define methods by which data is transferred across multiple networks. This is also referred to as internetworking.

Layer 7	Application
Layer 6	Presentation
Layer 5	Session
Layer 4	Transport
Layer 3	Network
Layer 2	Data Link
Layer 1	Physical

Figure 3-2 The OSI Screen Layer Model

Applications e.g. RTP HTTP SMTP FTP
Transport TCP UDP
Internetwork IP
Network Interface e.g. Ethernet, Fiber, WiFi

Figure 3-3 The Internet Protocol Stack

Figure 3-2. The OSI model defines seven layers of interconnection. The system behaves as if, at each layer, there is direct connection between each node; however, communication is through the layers. (at far left)

Figure 3-3. With Ethernet and most data networks, the preferred model is the TCP/IP four layer protocol stack.

UDP, by contrast, does not require a formal handshake between nodes. It applies a "best effort" protocol in which the packets are transmitted without the error-checking to guarantee reliable delivery.

to solve the inevitable problem of exhausting use of available public IP addresses, as well as simplifying packet routing, addressing for multicast applications and other enhancements. Until IPv6 completely supplants IPv4 for use everywhere, methods exist for allowing IPv6-only hosts to reach IPv4 services and isolated IPv6 hosts and networks to reach each other over IPv4-only infrastructure. Operating within private networks, AV systems remain effectively manageable using IPv4.

The TCP/IP protocol stack defines two protocols for the transport layer, TCP and UDP - User Datagram Protocol. See Table 3-1. The most significant difference between them is that UDP is "best effort" and TCP is "guaranteed delivery." TCP is used in most consumer and enterprise network applications for such tasks as exchanging file information between nodes. One common example is downloading files from an Internet Web site.

TCP requires establishment of an active connection session between nodes. It applies error-checking measures to ensure that all packets arrive reliably, and missing, duplicated, defective, or out-of-order packets are resent as necessary. While this ensures reliable packet delivery, the tradeoff is that the re-delivery of packets to the destination node may result in some latency.

UDP, by contrast, does not require a formal handshake between nodes. It applies a "best effort" protocol in which the packets are transmitted without the error-checking to guarantee reliable delivery. This eliminates any delay that would result from re-sending packets, but also means that packets transmitted via UDP may arrive out of order, have errors, or not reach their destination. The advantage of UDP, however, is that it is ideally suited for applications such as live video streaming that require speed and scalability, delivering packets to many endpoints. In UDP applications, packets are referred to as datagrams.

Table 3-1 lists the important attributes of TCP and UDP. They are applicable to the three major classes of network applications as follows:

- Unicast, where messages pass between two nodes. Can be TCP or UDP.
- Broadcast, where messages go from one node to all nodes on the network. Must be UDP.
- Multicast, where messages go from one node to many nodes, each subscribed to a multicast address group. Must be UDP.

The attributes of the TCP and UDP transport protocols as listed in Table 3-1 can lead to the following conclusions:

- Many applications will involve transmission of an image from one node to many nodes. Multicast operation would be desired to use network bandwidth efficiently, providing a motivation to use UDP.
- Applications requiring minimum latency will be better served using UDP.

Table 3-1. Comparison between the two commonly used protocols, UDP and TCP

UDP	TCP
Connectionless (no handshaking required)	Connection oriented (handshaking required)
Datagrams must be formatted in application	Automatically generated datagrams from bitstream
Multiple applications using ports	Multiple applications using ports
Unreliable (best effort) communication	Reliable (guaranteed) communication
No flow control (must be in application if required)	Flow control (deals with out-of-order data and error corrections)
No error recovery	Error recovery
Multicast possible (one to many)	One to one only
Minimum latency	Significant latency

- UDP does not provide reliable communication, so any application has to take into account the effect of lost data packets, out-of-order packets, and errors.

- The use of TCP will be desirable in network environments where low quality of service exists and latency is not a concern.

Various streaming transport protocols and system architectures have been developed to overcome the shortcomings that TCP and UDP have for transporting AV streamed content. One example is Real-time Transport Protocol – RTP, which time-stamps UDP packets and provides some measure of datagram flow control. RTP will be examined further along with other streaming transport protocols.

Many potential users of video streaming systems will be interested in scalability – the ability to expand a network in terms of the number of nodes on it, and the amount of traffic being handled.

A practical choice when the lowest latency is required, network bandwidth availability is predictable, and there is open access to endpoints is use of a handshake-free UDP protocol, along with provisions to handle errors at the display node. This environment can exist on private networks.

However, many applications do not require the lowest delay possible and must use the Internet or wireless networks where bandwidth and quality of service is less certain. In these environments, TCP transport over unicast connections provides reliable transport. A system solution supporting multiple unicast connections from a media server is illustrated in Figure 3-4.

The network transport protocol selected for AV streaming transport over a network is a significant consideration when designing scalable systems.

Intelligent, Programmable Network Switching and Routing

Commonly used enterprise switching products can provide the infrastructure to support many different types of AV streaming applications. These programmable network switches, also referred to as managed switches are available from many different manufacturers offering a diverse range of capabilities.

A summary of the types of capabilities that managed switches may support include:

Commonly-used enterprise network switching and routing products can provide very high backbone bandwidth or “switched fabric” capacity for streamed video traffic.

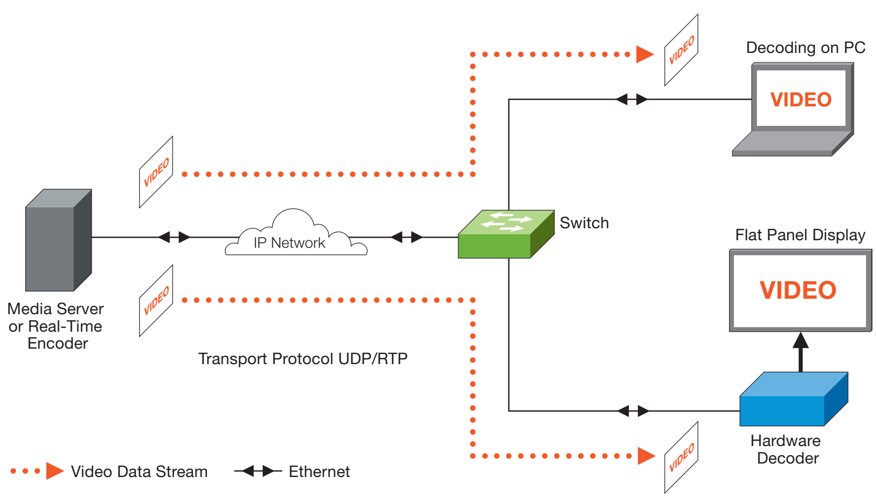


Figure 3-4. Unicast streaming produces one stream for every decoding device in the system

Determining if multicast traffic can be used on a network is one of the most important facts to establish when developing a scalable streaming solution.

- Switched fabric for managed switches can range from 10 Gbps to over 2 Tbps for high capacity enterprise aggregation switches. These switches can be stacked together and programmed to function as one large, high capacity system as illustrated in Figure 3-5. Network designs for data centers can support core switching bandwidths that exceed 80 Tbps.

- The tight arrangement of RJ-45 connectors on network switches provide a very high density for system wiring as can be seen in Figure 3-6.

- Static routing tables can be programmed, preventing certain types of traffic from reaching areas of a network based on the origination or destination IP address as well as traffic type, e.g., voice, data, and video.

- Dynamic routing tables can be automatically developed and "learned" by the routers and switches over time, as they work to efficiently distribute packets across multiple switches or router hops.

- Redundancy can be designed into switched networks. Multiple delivery paths ensure packet delivery if faults are experienced.

- Spanning Tree Protocol - STP and other methods exist for preventing undesired loops and "flooding" of traffic that could develop in architectures with multiple routing points capable of forwarding packets. Correct application and configuration of STP is critical to avoiding potential congestion or slowdowns.

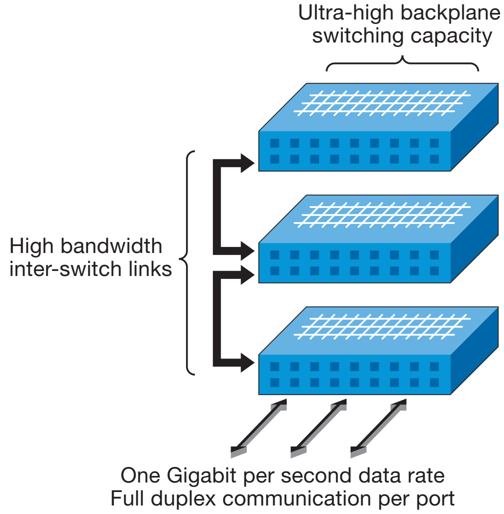


Figure 3-5. Network switches can be stacked and linked together providing a high bandwidth infrastructure for streaming applications.

- Virtual LANs - VLANs can be established by segmenting specific capacity on the same physical network to create different virtual networks. Defined amounts of bandwidth can be set aside in switches for different VLANs based on user groups, IP addresses, or traffic type. An example of this would be splitting capacity of a network equally between voice, data, and video traffic.

- Multicast support ensures that only a single layer of bandwidth is distributed across the network from the image source to multiple destinations regardless of the number of locations subscribing to it. Network traffic is pruned, or removed, from segments of the network where it is not required. Figure 3-7 illustrates multicast streaming. A standard Layer 2 switch, by default, will flood all of its ports with multicast traffic. Application of Internet Group Management Protocol - IGMP snooping and querying properly across routers and managed

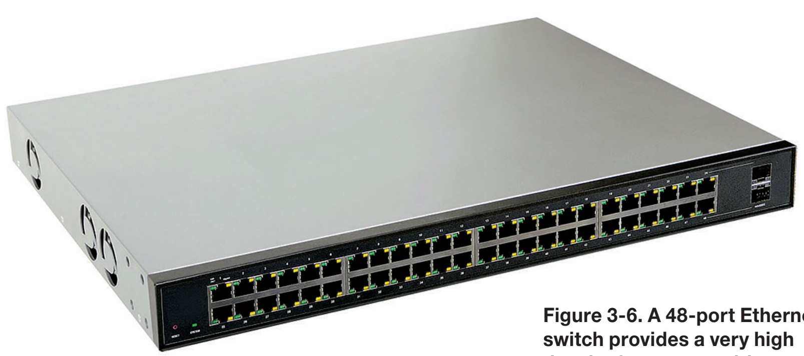


Figure 3-6. A 48-port Ethernet switch provides a very high density for system wiring.

Layer 2 switches ensures that nodes that have not subscribed to a multicast source will not be flooded with unnecessary multicast traffic.

Determining if multicast traffic can be used on a network is one of the most important facts to establish when developing a scalable streaming solution. Multicast streaming provides the fastest method for transporting real-time video and voice data, while ensuring low latency and using the lowest system bandwidth. It does however require that all switching and routing devices that will encounter the multicast traffic are programmed to efficiently transport that traffic. In the case of a stand-alone switch, the switch itself is the only device that must be programmed. If multicast is applied across a facility or enterprise network, the entire system may need to be programmed to support multicast traffic and some network hardware may need to be updated as well.

The Challenges of Compressing and Streaming AV on IP Networks

IP networks offer the promise of an AV signal distribution infrastructure that is convenient, scalable and cost effective. However, in practice, there are reliability and quality challenges that result from using IP networks.

Unlike conventional AV cabling installations, the actual performance of a network connection cannot always be predicted.

Transmission errors are remarkably rare within a dedicated or well-structured Ethernet installation that is working below full capacity. It is feasible to transmit video over such networks without any special precautions, provided that the video stream itself represents only a small percentage of the network traffic.

However, once there is traffic between different network segments and different network types, the likelihood of errors increases. This is especially true with public networks. QoS is the ability to guarantee data throughput as needed for a particular application, user, or performance requirement. QoS for public networks is currently defined in ITU Recommendation Y.1541, "Network performance objectives for IP-based services."

Table 3-2 identifies QoS classes in terms of performance target parameters, such as Internet Packet Transfer Delay - IPTD, Internet Packet Delay Variation - IPDV, Internet Packet Loss Ratio - IPLR, and Internet Packet Error Ratio -

Transmission errors are remarkably rare within a dedicated or well-structured Ethernet installation that is working below full capacity.

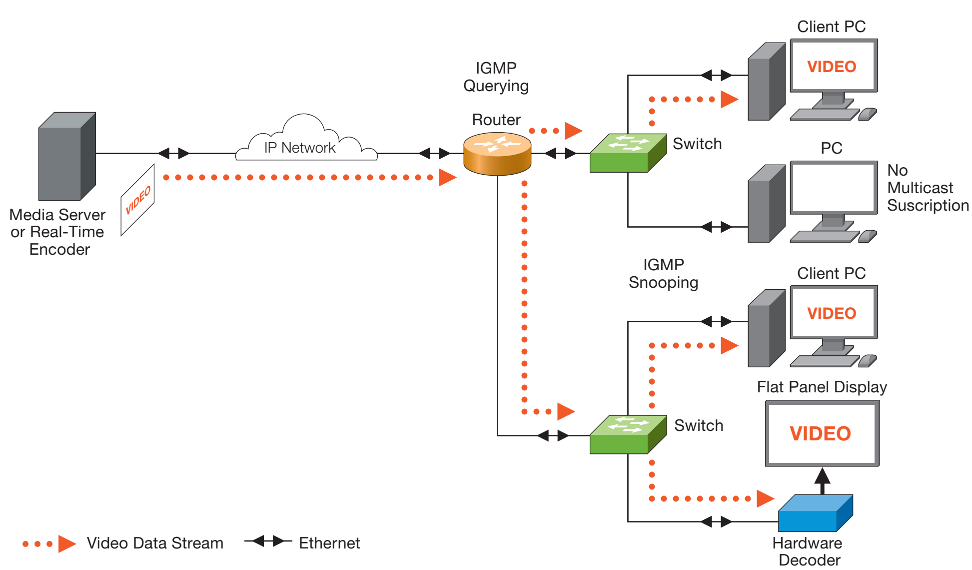


Figure 3-7. Multicast streaming makes the most efficient use of network bandwidth

In practice, private networks can be specified with better IPLR and IPER performance and network services do offer reliability with an upper bound IPLR of 1 x 10-4 and higher.

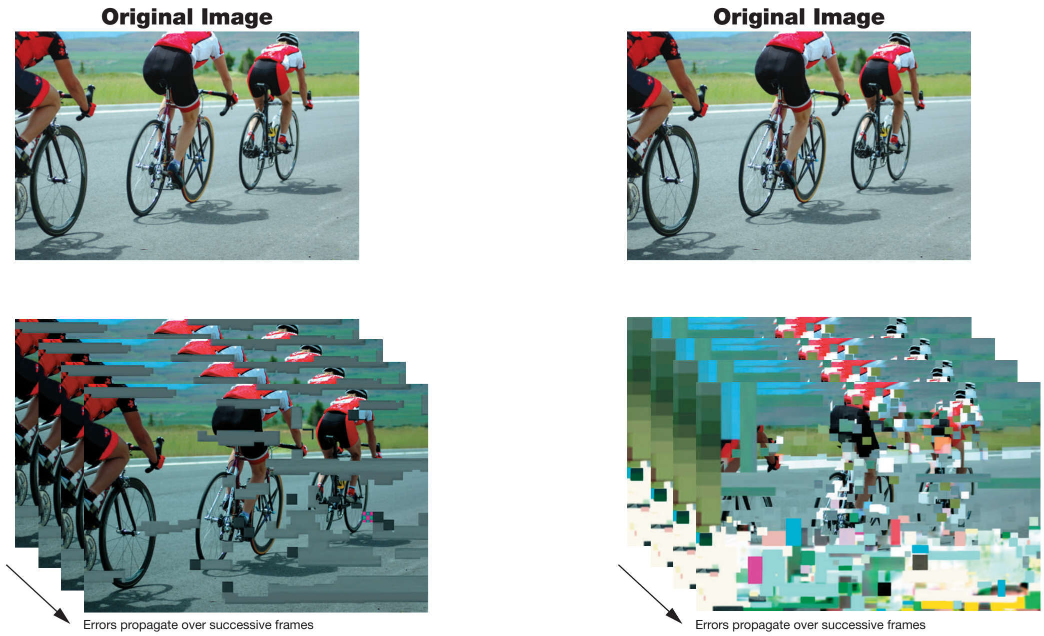


Figure 4-1. Examples of visual errors that may be observed in video streaming when network errors are experienced

IPER. The upper bound on packet loss probability identified in QoS Class 3 of 1x10-3 represents notable uncertainty for supporting quality video streaming. Quality business or mission-critical applications should not operate upon a platform with one or more lost packets per thousand, without robust error handling mechanisms in place. Two new provisional network QoS classes have been identified by the ITU to provide higher performance in ranges that are appropriate for video streaming. An upper bound IPLR of 1 x 10-5 or one packet lost per one hundred thousand has been proposed. However, the objectives identified in table 3-2 remain in effect until the ITU conducts further study and agrees on the provisional classes for QoS.

In practice, private networks can be specified with superior IPLR and IPER performance and network services regularly offer reliability with an upper bound IPLR of 1 x 10-4 and higher. However challenges remain building cost-effective network infrastructures that support broad enterprise streaming, purchasing affordable network services for streaming over vast geographic distances or applying streaming on networks that were originally designed to transport data and voice traffic. ■

Table 3-2. The network performance objectives under Y.1541 including provisional QoS classifications 6 and 7

	Specified QoS					Proposed QoS	
Parameter	**Class 0**	**Class 1**	**Class 2**	**Class 3**	**Class 4**	**Class 6**	**Class 7**
Internet Packet Transfer Delay - IPTD	100 ms	400 ms	100 ms	400 ms	1s	100 ms	400 ms
Internet Packet Delay Variation - IPDV	50 ms	50 ms	Unbounded	Unbounded	Unbounded	50 ms	50 ms
Internet Packet Loss Ratio – IPLR	1x10^{-3}	1x10^{-3}	1x10^{-3}	1x10^{-3}	1x10^{-3}	1x10^{-5}	1x10^{-5}
Internet Packet Error Ratio - IPER	1x10^{-4}	1x10^{-4}	1x10^{-4}	1x10^{-4}	1x10^{-4}	1x10^{-6}	1x10^{-6}
Note: Class 5 is undefined							

Network Errors

As discussed earlier, the UDP and TCP transport protocols handle errors in two completely different manners. TCP transport guarantees delivery of data. This performance can be exploited in conditions with uncertain QoS conditions, but scalability limitations are introduced due to use of unicast connections. The data acknowledgement process consumes additional bandwidth, and when errors are experienced, the encoder or server must resend the packet, introducing latency, which interferes with real-time performance. If a data resend is required for multiple devices, the encoder or server may not be able to keep up with demand. Examples of streaming systems that use TCP transport that support scalable solutions will be identified later.

When UDP transport is used, precautions must be taken to prevent or minimize the effects of network errors that are illustrated in Figure 4.1.

Some methods seek to combine UDP with the error-checking attributes of the TCP protocol; however, there are limits to what can be done without introducing unacceptable latency.

RTP can be implemented along with UDP to apply headers to the packets with time-stamp information, which allows a receiver to re-order packets received out of sequence and to identify missing packets.

Video encoders that make use of select H.264 profiles can apply certain coding techniques which re-order macroblocks and slices as well as partition image data based the degree of importance it plays within the video image. These coding techniques offer a level of error concealment that will be unique to each manufacturer's product.

Differentiated Services – Diffserv, can be applied to IP packets to classify and manage different types of network traffic. AV traffic can be identified with a request for higher priority than file transfers, and if routing equipment is oversubscribed, the video traffic can be forwarded before other data classes such as a file transfer which are not time sensitive. Diffserv can reduce the likelihood that errors are experienced if each of the services such as voice, data and video are classified with the appropriate priority and all devices in the system are programmed to support this system. Diffserv is therefore a practical method that can be deployed on a private network.

Various user groups and suppliers have introduced standard methods of error correction when UDP transport is used for streaming. Among these are those introduced by the Pro-MPEG Forum for the professional broadcast video industry. This group has introduced various Codes of Practice - COPs for the transmission of digital video over Ethernet networks. The following are the best-known among them:

- COP-3, for use with the MPEG-2 transport stream protocol.

- COP-4, for the transmission of uncompressed SDI video, including standard definition video at up to 270 Mbps and high definition video at up to 1.485 Gbps.

Both use FEC - Forward Error Correction, a method of transmitting redundant data packets along with the original packet data. When the receiver detects an error in any part of the data within a packet, the redundant data is used to correct the error and recover the original data.

FEC organizes the original packetized data stream into a matrix of columns and rows, for example, 100 packets arranged as 10×10. For each column and row, an additional FEC packet is transmitted. Missing packets can be reconstructed by comparing the FEC packet data with the remaining correct data. Correction can be one dimensional, where either row or column data only is used, or two dimensional, where both are used. One dimensional correction can recover single missing packets, but cannot guard against multiple packet loss or loss of FEC

RTP can be implemented along with UDP to apply headers to the packets with time-stamp information, which allows a receiver to re-order packets received out of sequence and to identify missing packets.

Forward error correction, if applied due to uncertainty in the network quality of service, will also introduce delay.

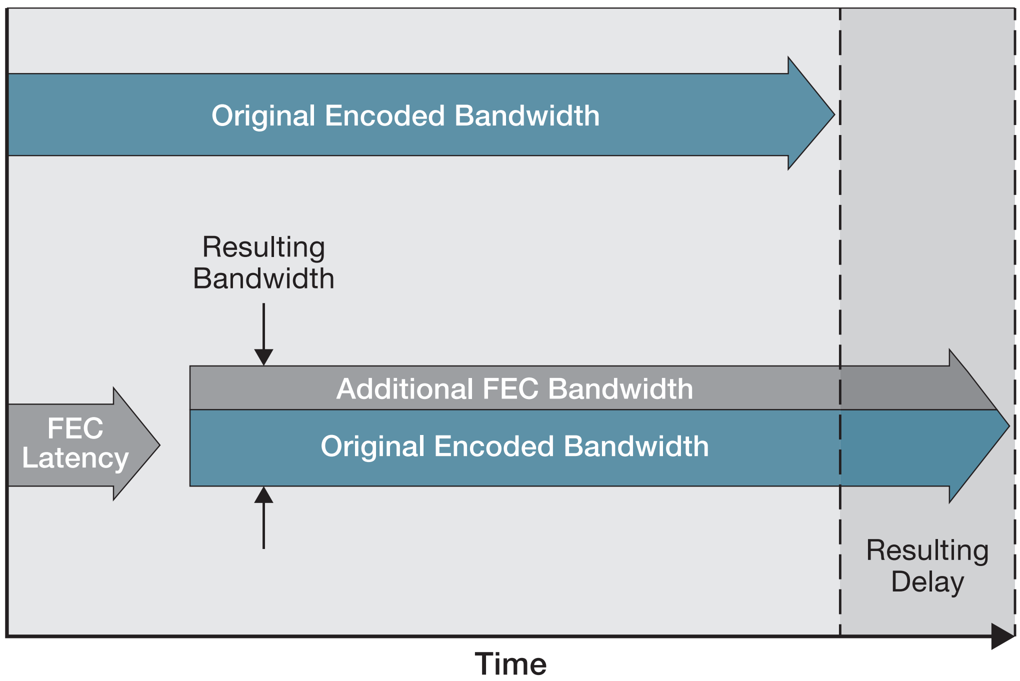


Figure 4-2. Illustration of the delay and bandwidth contributed by FEC

data. Two dimensional correction can protect against multiple packet loss, and loss of individual FEC packets.

The FEC method used by the COPs is identified in RFC 2733 - Request for Comments, the method by which Ethernet practice is promoted. The advantage of FEC is that it does not require retransmission of packets as with the TCP transport protocol. However, latency and redundant data is added to the streaming bandwidth. Broadcast streaming applications may use as much as 10% to 30% additional data. This is illustrated in Figure 4-2.

While the QoS that networks support continues to improve, many business applications must support error free, reliable video transport and one or more of the error concealment or correction methods identified here must be used.

Streaming Latency and Applications

Delay or latency can be described from an application perspective as well as an encoding perspective. As mentioned earlier, in order to compress video to low bit rates, inter-frame or temporal compression is necessary so that sequential frames can be compared to determine what redundant information need not be sent. Each frame used in this process increases the encoding delay. If an application operates on a private network, ample bandwidth may be available so that less temporal compression is required, resulting in lower latency. However, on public networks, more aggressive compression will be used to achieve bit rates that will fit low bandwidth connections to a majority of endpoint devices. This will increase delay or reduce quality. Forward error correction, if applied due to uncertainty in the network quality of service, will also introduce delay. Other system elements such as router hops in the network and buffering of video in displays or image processors can also contribute to the total system latency experienced.

However, delay will not be critical to many video streaming applications such as VoD and Webcasting of training presentations, for which the priorities are accessibility and easy consumption of information. Both of these applications are primarily one-way, and a delay of several seconds is acceptable. The main concern in these applications is establishing the initial connection and providing the user with a prompt indication that the streamed content will be presented shortly. Public networks are adequate for these one-way applications where low or ultra-low bit rates are used.

Other applications such as video surveillance may require a user to view streaming video and then control equipment located at the far end such as a pan tilt zoom camera. In such cases, latency should not exceed one second so that a user can make prompt operational decisions based on the video imagery.

On the other end of the spectrum, there are two-way interactive applications, including videoconferencing, distance collaboration, and remote device control where low latency is essential. If a face-to-face conversation is taking place across a network, the natural rhythm of the conversation will be impaired if just a moderate delay exists in the system. Individuals at each endpoint may frequently speak at the same time and noticeable latency will make communication awkward and ineffective. Distance collaboration with high-resolution visualizations may include requirements for real-time, remote control of a software application or device using a touch panel or keyboard and mouse. Any latency introduced by video streaming will soften the feedback of the man-machine interface that ideally provides a very tight, tacit control experience; the more latency that exists, the more difficult it is to control the equipment. Table 4-1 provides a reference for various amounts of latency that will be appropriate for different activities. ■

Distance collaboration with high-resolution visualizations may include requirements for real-time, remote control of a software application or device using a touch panel or keyboard and mouse.

Acceptable Latency by Application		
Application	**Delay**	**Latency requirement**
Surgical equipment control	0	No potential error is acceptable
Tactile device control	<50 ms	Application requires tight man-machine interface
Device control	<100 ms	Application has lower man-machine control accuracy
Videoconferencing	<200 ms	Face to face two-way communication, voice and video
Visual collaboration	<200 ms	Viewing of identical imagery with voice communication
Real-Time video contribution	<500 ms	Keep video production chain reasonably short
Surveillance camera control	<500 ms	Camera pan-tilt-zoom control to coarse positions
Surveillance camera monitoring	<1s	Security service has visibility into situation
Webcast	1-5s	View simple visuals and slides with voice, one-way delivery
Low bit rate video contribution	5-10s	Application driven by network limitations
Video on Demand	5-10s	Primary concern is download buffering time
Consider system latency introduced by all devices from input to display.		

Table 4-1. Acceptable Latency by Application

Pull streaming is managed from the decoding device, which initiates events using a session management protocol such as Real Time Streaming Protocol – RTSP.

Pull and Push Streaming

Two primary methods are used to manage streaming systems; pull and push. In pull streaming, the session is initiated from the decoding device. In push streaming, the encoding device initiates the session. Use of pull or push streaming may be determined based on a number of practical considerations. Requirements that contribute to use of pull or push methods include scalability, decoding devices, network architecture, and quality of service conditions:

- Decoding devices: Hardware decoders or software players may only be compatible with streaming transport protocols that use pull or push session management. For instance, commonly used set-top boxes require use of push streaming. If decoding devices are managed as part of a streaming system, it is more likely that push streaming can be used. If the decoding devices are not managed as part of a system, pull streaming can be used.

- System scalability requirements: The streaming capacity of an encoder and the number of decoding devices that must be served can also be a determining factor. For instance, push multicast streaming makes efficient use of network bandwidth, streaming to many decoders.

- Network architecture and policies: Multicast streaming may be supported on private networks, but unicast streaming may be required due to hardware limitations or operating policies. Multicast streaming applications are rarely supported on the Internet, WANs, or VPNs. Wireless LANs – WLANs require engineered designs to support multicast streaming.

- Quality of Service: AV streaming on low QoS network conditions will be better served using streaming transport protocols that use TCP transport and pull streaming session management.

Pull streaming is initiated from the decoding device, using a session management protocol such as Real Time Streaming Protocol – RTSP. Pull streaming uses a unicast connection and is frequently chosen when decoding devices are not managed as part of a system. Media players in wireless tablet PCs and smartphones will normally use streaming transport protocols based on TCP transport and therefore will use unicast pull streaming.

Push streaming is managed centrally from an encoder or streaming server. Encoders are programmed to push streams to a unicast or multicast address. Streams may be either pushed to a decoding device's defined IP address or streaming session management protocols such as Session Announcement Protocol – SAP, and Session Description Protocol – SDP, may allow decoding devices to identify and join a push multicast session. SDP and SAP also provide an automated method for decoding devices to build channel lists of available streams.

Each method has different strengths. Pull streaming is advantageous in ad hoc applications. For example, all PCs in a building may require access to a streaming source, but they do not all need to be managed as part of a system. Pull streaming can be compared to accessing content on the Internet. Web servers offer information to virtually any PC that can connect to them, but content is only supplied when a client requests it.

Push streaming is ideal for applications when the streaming session should be managed from

Table 5-1. A comparison of Pull and Push streaming management

Comparison Point	Pull streaming	Push streaming
Session Management Protocols	RTSP	SAP, SDP
Streaming Management Point	Decoding device	Streaming encoder or server
Multicast implementations	Primarily Unicast	Unicast or Multicast
Streaming system management	Managed or Unmanaged	Managed
Common network environments	Private or Public	Private

the encoder. The decoder does not need to initiate the session. The encoder can push to one decoder or multicast to many decoders.

A comparison of pull and push streaming can be seen in Table 5-1.

Streaming Transport Protocols

One of the most overlooked and misunderstood areas of AV streaming is the topic of streaming transport protocols. A common assumption about H.264 streaming products is that they are all compatible with each other. However, the H.264 standard only defines how compressed video data is decoded. Different streaming transport protocols may be required to deliver AV streaming content between devices. They are one part of a larger relationship of application, device, device platform, and media player that determine compatibility between streaming encoders and decoders.

Streaming transport protocols deliver encoded video and audio data across networks from hardware encoders or media servers to hardware decoders or software players operating on PCs, tablets, or smartphones. They provide varying combinations of speed, accuracy, and compatibility to fulfill various platform or application requirements.

As identified earlier, data is transported across a network using either the TCP or the UDP protocol; each of which offers unique strengths. Applied alone, neither TCP nor UDP transport protocols can fulfill the following requirements that exist for reproducing motion video, which include:

- Consistent data rates, which support decoding of live, moving pictures
- Accurate, sequential data for image decoding
- Synchronized data to support accurate AV decoding
- Varying latency requirements, which differ by application
- System scalability, which can range from one to thousands of endpoints

There are many different hardware devices and software players that decode AV streams, with each applying a different operating system or software player. Manufacturers of decoding devices and developers of decoding applications support different streaming transport protocols to meet various technical performance or commercial objectives. See Figure 5-1, which illustrates the variables that exist between these system elements.

Streaming Transport Protocols for Different Requirements

Streaming applications may use either standards-based or proprietary protocols. Commonly used standards-based protocols include:

One of the most overlooked and misunderstood areas of AV streaming is the topic of streaming transport protocols.

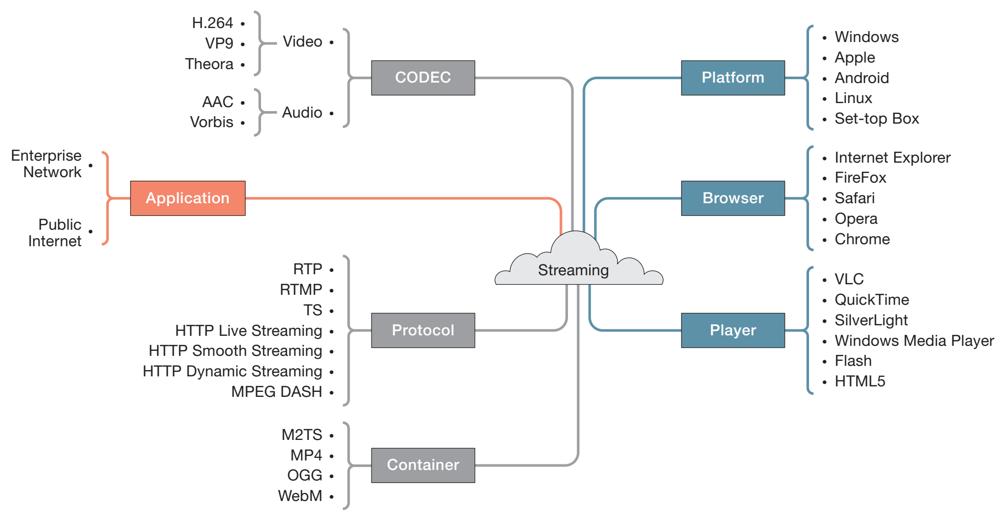


Figure 5-1. The variables that exists between streaming applications codecs, protocols, containers, decoding platforms, browsers and players

MPEG-2 Transport Stream – TS: Established by the Moving Pictures Expert Group, TS has been applied in broadcast transmission standards such as the Advanced Television Systems Committee – ATSC and Digital Video Broadcasting – DVB.

- RTP – Real Time Transport Protocol: RTP is broadly used by hardware encoding and decoding products, but it may not be applied in the same manner on every streaming product. H.264 video may be transported directly using RTP, or RTP may be used to carry MPEG -2 Transport Streams. RTP used along with UDP network transport supports high speed transport on networks.

- RTSP – Real Time Streaming Protocol: RTSP is a session management protocol, but is also applied as a non-standard method to deliver AV streaming payloads.

Both RTP and RTSP are identified as part of the Internet Protocol Suite.

- MPEG-2 Transport Stream – TS: The TS was established by the Moving Pictures Expert Group – MPEG and it is frequently used in streaming applications. It is a container format that uses RTP and UDP protocols for transport in streaming applications. It has been applied in broadcast transmission standards such as the Advanced Television Systems Committee – ATSC and Digital Video Broadcasting – DVB. It is used extensively in broadcast, IPTV, cable television and digital signage applications. Set-top box decoders typically use TS and many H.264 hardware encoders and decoders use it as well.

Proprietary streaming protocols are developed for use on specific software operating systems and media players. Common examples include:

- Microsoft Media System – MMS: MMS was developed by Microsoft Windows Media Services. MMS is frequently deployed in enterprise-wide streaming applications that use Windows Media Player to decode and present AV streams from PC desktops.

- Real Time Messaging Protocol – RTMP: Adobe has developed RTMP, RTMPS, RTMPE, RTMFP, and HTTP Dynamic Streaming protocols for use with Adobe Flash Player. Each protocol offers unique security, encryption, latency, and firewall traversal capabilities. Flash player was once supported on smartphone and tablet PCs using the Android operating systems and RTMP protocol. This is no longer the case.

- HTTP Live Streaming – HLS: The Apple HLS streaming protocol is used extensively for streaming to iPad and iPhone devices. Based on the TS protocol, HLS runs on Apple devices that use the iOS operating system and the HTML 5 media player. New smartphone and tablet PCs with the Android operating systems also support the HLS protocol.

- Smooth Streaming, an extension of Microsoft IIS Media Services, is another Microsoft

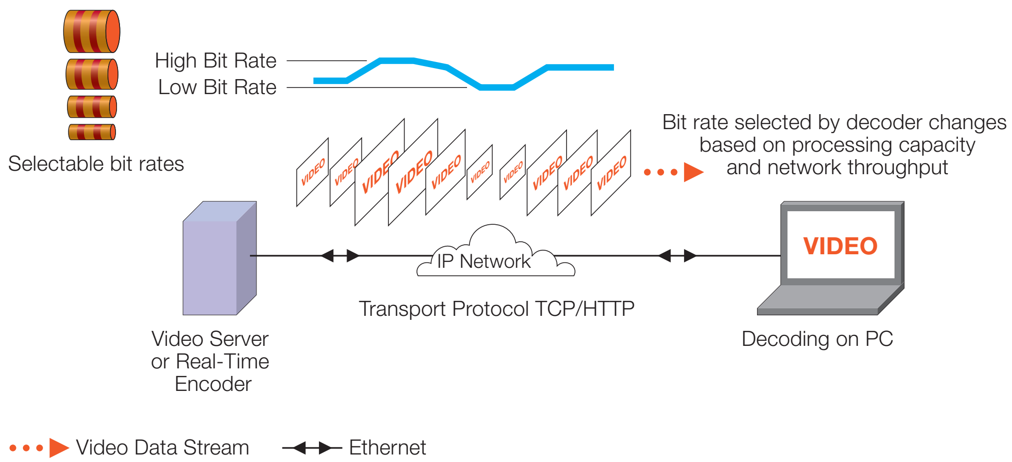


Figure 5-2. Adaptive bit rate streaming allows a decoder to select different bit rates based on varying operating conditions

streaming protocol, which has been developed for use with the Microsoft Silverlight player and other media clients.

These proprietary protocols are the preferred streaming transport methods for use on smartphones, tablet PCs, and media player applications identified earlier. As consumer devices such as tablet PCs and smartphones evolve over time, the player and streaming transport protocol they use may change. Some standalone streaming players and plugins developed for proprietary or standards-based protocols offer compatibility. However, device platforms and applications may not meet requirements for reliability in commercial applications. For instance, an RTP-compatible application on a tablet PC may not adequately manage the processing required for high bit rate streams or provide recovery mechanisms to address network errors.

- Hypertext Transfer Protocol – HTTP and TCP network transport protocols are used by HTTP Dynamic Streaming, HLS, and Smooth Streaming protocols. TCP network transport provides reliable transport, and HTTP permits tunneling of video streams through firewalls using port 80, left open for HTTP Web traffic. These protocols also support Adaptive Bit Rate – ABR capabilities that enable decoding devices to dynamically request higher or lower streaming bit rates in response to changes in network throughput or processing capacity. These protocols are ideal for streaming content across the Internet or WLANs where fluctuating bandwidth and QoS conditions exist. See Figure 5-2 for an illustration of how adaptive bit rate streaming works.

- A new adaptive bit rate protocol, Dynamic Adaptive Streaming over HTTP – DASH also known as MPEG-DASH was published as an international standard in April 2012. Broad implementation of the DASH standard may make streaming to wireless consumer devices and Internet destinations less complicated. However, until MPEG-DASH is applied broadly, AV professionals will still need to consider streaming protocols as an important variable to consider for different decoding devices that must be supported in streaming system designs.

As consumer devices such as tablet PCs and smartphones evolve over time, the player and streaming transport protocol they use may change.

Media Servers

Many applications require high scalability, delivering unicast AV streams to many decoding devices when networks cannot support IGMP multicast traffic. At some point, a single AV encoder will reach a limit to the number of unicast streaming sessions it can support. Many of the same applications that require high scalability also require the ability to deliver AV streams using different streaming transport protocols at different bit rates and resolutions. Media servers provide this capability. Media servers can transmux,

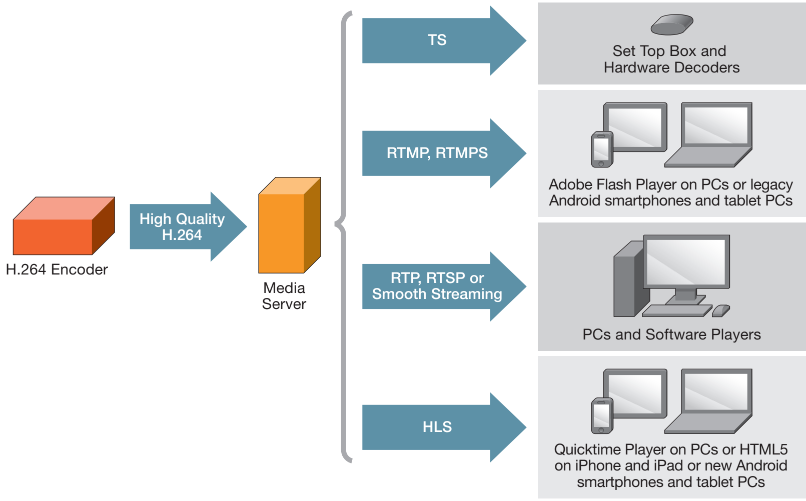


Figure 5-3. Media Servers convert streaming protocols, providing compatibility streaming to different types of devices

converting streaming transport protocols to make them compatible with different types of decoding devices. Media servers may also support transcoding, a process that produces lower resolution, lower bit rate streams for more efficient processing on different types of decoding devices. The term "multi-screen delivery" is frequently used to describe applications where the same content is streamed at different resolutions and bit rates for use on devices with different screen sizes.

When encoding for live, multi-screen streaming applications, Wowza Media Systems, a media server developer, recommends the following: For the greatest compatibility with decoding devices, apply standards-based compression such as H.264, and use commonly used streaming transport protocols such as RTP or RTMP. AV streams should be encoded at the highest resolution possible to preserve image quality for higher quality applications, leaving down-conversion and transcoding to other devices such as the media server.

How Should AV Designers Manage Streaming Protocols?

Streaming protocols are an important variable in H.264 applications. Failure to consider their role can result in significant compatibility gaps.

Where streaming protocols are considered as a part of a complete system design, choices will often need to be made between performance and compatibility. Small, point-to-point applications using specific devices are afforded the luxury of using products and streaming transport protocols that offer the best performance without the need to make compromises.

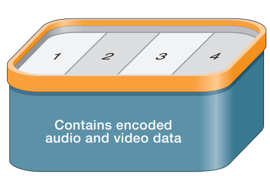
Projects that serve a large variety of decoding devices, including personal computers, tablets, or smartphones, should take into consideration the fact that the streaming protocols these devices are compatible with can change over time. Wherever possible, a system design should apply a standards-based protocol for AV encoding and distribution over a wired LAN. Media servers can be used to transcode AV streaming for use on wireless consumer devices or the Internet. Platform design choices based on proprietary protocols for specific media players or wireless consumer devices may be costly to maintain and support as technology changes.

Container Formats

One streaming subject that often produces confusion is the difference between codecs and container formats. Codecs are used to encode video and data for digital storage or live transport over a network. H.264 is an example of video codec and AAC is an example of an audio codec. Container formats package, or wrap, encoded audio and video together into a file format that is read by media players. One or many compressed audio and video data streams are held within a container file along with other data, such as metadata, and synchronization information necessary for cohesive playback. Container formats may also include other data, like subtitles, chapter information, or closed captioning information.

Container formats are notated in file extension names, and are supported by one or many media players. AVI, MKV, FLV, WMV, MOV, M2TS, and MP4 are all examples of container formats. AV media players will be compatible with container formats when they support the audio and video codecs carried within, and if Digital Rights

Container Formats Describe			
Video Codec Examples	Audio Codec Examples	Metadata Content	Video Captioning Examples
• H.264 • VC-1 • Theora • Dirac 2.1 • H.263	• AAC • MPEG-1 • WMA • VORBIS • PCM	• Author • Title • Location • Date • Copyright • License	• SAMI • SMIL • Hi-Caption • CMML • DXFP • 3GPP TS 26.245 • MPSub
1	2	3	4



Example Container Formats:
AVI, MP4, MOV, VOB, 3gP, TS

Figure 5-4. Illustration of the contents held within a AV media file container format

Streaming protocols are an important variable in H.264 applications. Failure to consider their role can result in significant compatibility gaps.

Application	Developers	Container Format
PC Media Playback	Microsoft, Apple	AVI, WMV, MOV
PC and Web Media Playback	Adobe Systems, RealNetworks, WebM Project, Xiph.org, DivX, Inc., CoreCodec, Inc.	FLV, RMVB, WebM, OGG, DivX, MKV
Disc Media Playback	DVD Forum, Blu-ray Disc Association, DivX, Inc.	VOB, M2TS, DivX
Data interchange	SMPTE, MPEG	MXF, MP4, TS
Mobile Device Playback	3GPP, 3GPP2	3GP, 3GPP2

Table 5-2. The developers and applications of various container formats

Container formats package, or wrap, encoded audio and video together into a file format that is read by media players.

Management - DRM usage rights are validated by the computer or playback device.

Numerous container formats have been created by a variety of developers to meet the needs of specific applications and playback media. Examples of different applications, developers, and container formats are identified in Table 5-2. Standards-based container formats such as MP4 and TS have been developed by the Moving Picture Experts Group, and proprietary formats such as AVI and WMV were developed by Microsoft. MOV was developed by Apple.

Ideally, AV systems that create or play back media files will use container formats that support usage across a wide range of devices. However, over time, new container formats will emerge, and playback applications and devices change, particularly consumer devices. Deciding to use the most popular, standards-based container format will help ensure that content is accessible by the greatest variety of devices and media players, and provide the greatest opportunity for future repurposing of content. When a special-purpose playback device is selected, or a unique container format is required, a transcoding application can convert standards-based container formats into new formats.

Network and Streaming Services

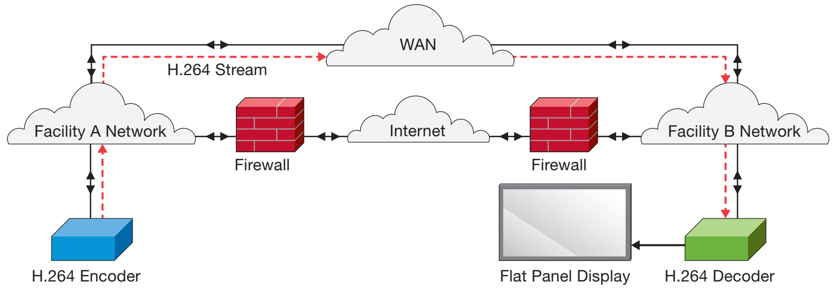
Many applications require AV streaming to be transported outside of a facility or LAN. With this requirement comes the need to determine if the network and streaming system can adequately support the streaming application or the cost of a network service for transporting content. Different methods are appropriate based on the application.

The points to consider when streaming outside a facility LAN:

- Scalability requirements
- Types of decoding devices and streaming transport protocols
- Location of destinations; inside or outside the enterprise
- Types of network connections available for streaming
- Existing bandwidth and quality of service conditions

The answers to these questions will frequently provide a guide to practical choices and considerations when streaming outside a facility LAN.

Figure 6-1. WANs provide a direct network connection between enterprise facilities



Service providers that provide WAN connections between facilities will offer a service level agreement – SLA, which identifies the class of service it will provide.

Streaming inside the Enterprise

A WAN that can support the cumulative bandwidth of streaming traffic between endpoints provides a practical streaming platform inside an enterprise. WANs normally provide the simplicity of operating inside a managed system, without the concerns of firewalls or encryption devices. Service providers that provide WAN connections between facilities will offer a service level agreement – SLA, which identifies the class of service it will provide. Examples of different classes of service that may be offered by a network are identified in Table 6-1.

Low latency, time sensitive streaming applications that apply push streaming will be more sensitive to network errors, requiring attention to class of service. Pull streaming applications that use TCP transport provide robust transport, and require less attention to a class of service.

WANs may be based on dedicated leased lines or more cost-effective packet switched networks such as multiprotocol label switching – MPLS, Carrier Ethernet, or legacy-oriented technologies such as synchronous optical networks - SONET, asynchronous transfer mode – ATM, or Frame Relay. MPLS is a popular, packet switched technology capable of providing a very high quality of service. Carrier Ethernet is known for offering high bandwidth at greater economy.

Each network transport technology has its own strengths and weaknesses relating to bandwidth, QoS, and the provider's SLA as well as cost. Establishing the last mile connection may also introduce a significant investment that must be considered when selecting a network service.

Table 6-1. Examples of different classes of service offered by a network service provider

Class	Availability	Latency (Round Trip)	Jitter	Packet Loss
1	99.999%	50ms	<10ms	<0.1%
2	99.99%	60ms	<15ms	<0.15%
3	99.9%	85ms	<20ms	<0.7%
4	99.5%	110ms	<25ms	<1.0%

We experience streaming as a consumer every day, watching video on demand, or communicating with friends and family using applications such as Skype. These applications are designed to transport AV streaming across consumer grade network services and they often leave business users with the impression that development of streaming systems should be easy because they use applications at home that appear on the surface to be the same. Business-grade networks are required when streaming critical communications within an organization. Exceptions exist, but consumer network services from the typical cable provider do not guarantee bandwidth, but rather provide a range that may not be sustainable. The bandwidth provided on consumer networks is often asymmetric, providing greater capacity for reception vs. transmission, fulfilling the typical usage requirements for consumption rather than production. QoS provided on home Internet service will also be lower than that provided by a business grade network.

Streaming over the Internet

The phrase, "I want to stream over the Internet" can be interpreted to mean a lot of things. It could mean AV streaming content is intended to be transported to consumers across the Internet, or it could mean that the Internet is being utilized for the transport of video between a few select, endpoints within an organization, because an internal WAN connection is not available. It is important to distinguish if an application will be streaming between private users or to consumers located outside of the organization.

As the Internet is a public network, streaming to "internal" enterprise destinations often requires encryption devices such as virtual private network – VPN to allow the data to be encrypted and "tunneled" to its destination. In these cases, the streaming encoder should support adjustments to the maximum transmission unit – MTU of

Ethernet frames or allow them to be split into two separate pieces. This will make room for encryption data to be carried alongside the streaming data. Absence of an MTU adjustment can prevent streaming traffic from passing through VPN and encryption devices.

The Internet is not a network that should be used as a platform for mission-critical, real-time streaming applications. HTTP-oriented streaming transport protocols that use TCP network transport can be used to support reliable transport for one-way streaming.

Content Delivery Networks

As streaming systems get larger, and the number of endpoints grows, at some point it becomes impractical to support the total demand directly from a single device. The number of subscribers may exceed the encoder or server's ability to efficiently support the demand or the network path may not support the bandwidth requirements. In these cases, overlay network services such as Content Delivery Networks - CDNs provide a solution for broad delivery of AV content across the Internet.

A CDN is a network of distributed servers that deliver video, web pages, or other media, to users efficiently, based on the user's geographic location and network distance from the streaming server. CDN servers are located at the edges of the Internet in strategic locations, enabling efficient delivery of AV streaming to users at any location. When a user requests streamed video, the CDN directs the request to a server in its network that is closest to the user. When using a CDN, producers of AV content need only deliver one stream to the CDN, and the CDN will fulfill demand from all users. CDNs charge a fee based on demand; it varies based on the volume of traffic or scope of services.

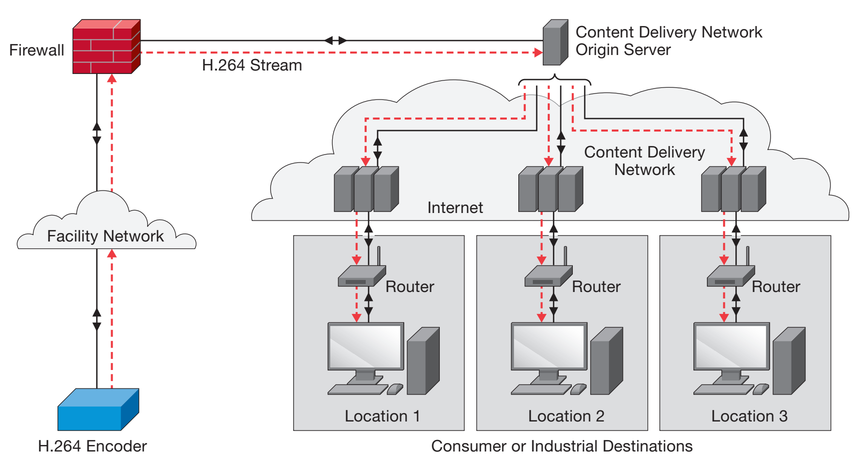
There are many different types of CDN providers. Many telecommunication providers have added media and video distribution to their offering to make greater use of their infrastructure. Other CDNs are Internet service providers that specialize in managing video content and other services. They contract with the network service providers to provide the infrastructure to transport the streamed traffic.

The needs that can exist for Internet streaming will vary greatly by customer. The usage pattern and demand for streaming by an international consumer goods manufacturer will be very different from a city government or a house of worship. The consumer goods manufacturer will probably be a high volume streaming user, seven days a week, 24 hours a day. An average city government and house of worship will have much lower demand for streaming and usage periods will be focused around certain events or specific days of the week.

Each CDN will have different strengths for serving customers. Large CDNs with global reach will offer cost-effective service for high volume customers. Other CDNs may offer unique programs aimed at serving tightly defined geographic areas or specific usage patterns. CDNs may offer detailed statistical usage data, transcoding, recording, and video-on-demand services, or readily accessible technical support. The best CDN for an application will depend upon the range of requirements, technical skills, experience and resources a customer has available to manage their part of the streaming application. ■

A CDN is a network of distributed servers that deliver video, web pages, or other media, to users efficiently, based on the user's geographic location and network distance from the streaming server.

Figure 6-2. Architecture of a content delivery network



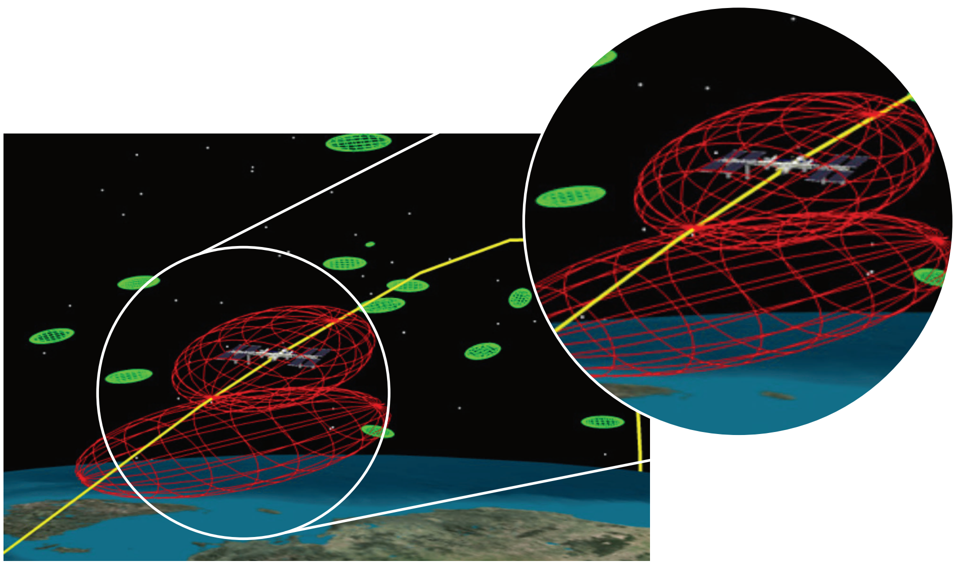
PURE3 was designed specifically for AV streaming, to optimize delivery of high resolution video and graphics with minimal latency, and within the bit rates acceptable to commonly used IP networks.

The Extron PURE3® Compression Codec

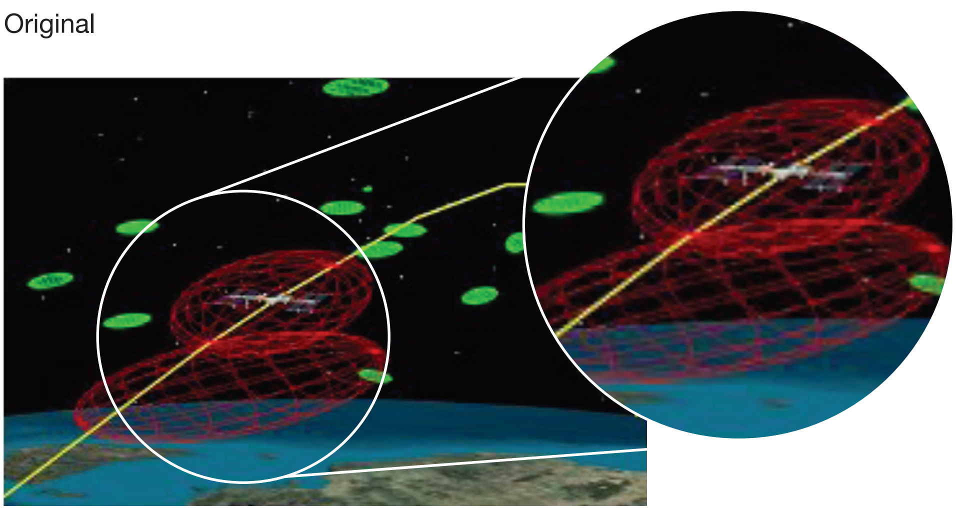
The Extron PURE3 codec addresses the AV streaming challenges identified previously, where interactive, real-time communication with high resolution video and graphic images is required. The PURE3 codec is designed to provide equal support for both video and computer graphic signals and offers a unique combination of:

- High image quality
- Low latency
- Bit rate efficiency
- Reliable delivery over networks with high immunity to network errors

Existing compression technologies discussed earlier, such as MPEG-2 and JPEG 2000, each had an application focus that made them unable to fulfill all these performance requirements; therefore, the PURE3 codec was developed. For example, MPEG-based systems provide a very low bit rate, but at the expense of latency and compromised image quality. JPEG 2000 provided the desired picture quality, but required high bit rates. Both standards are vulnerable to network errors unless error correction measures are applied, and that adds latency and bandwidth overhead, which is not acceptable for real-time



Original



Poor compression

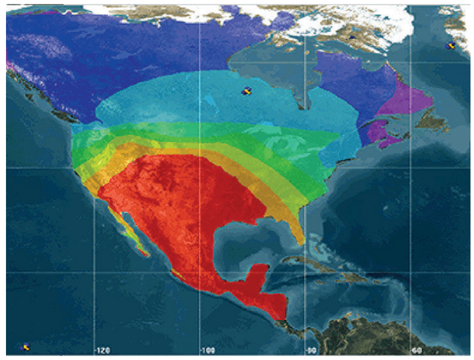
Examples of Poor Compression of Computer Imagery

1. The detail in the grid of the green sphere is lost (compression)
2. The color of the yellow line alternates between yellow and white (4:2:0 decimation)
3. The pixels of the red line are now a larger and different size (different resolution)

(original image: AGI)

Figure 7-1. Image quality differences between video processed with 4:4:4 and reduced color information

Video



Graphics

Figure 7-2. Video and Graphic images are different

AV streaming applications that require the highest quality, high resolution images.

PURE3 was designed specifically for AV streaming, to optimize delivery of high resolution video and graphics with minimal latency, within the bit rates acceptable to commonly used IP networks. The PURE3 codec provides a unique combination of performance that makes it ideal for use in demanding, quality critical applications.

Image Quality

A color space transformation is made from the RGB to the luminance/chrominance domain, also known as component video or YCbCr, but full 4:4:4 color information is retained. The DWT - Discrete Wavelet Transform was chosen to achieve the best coding efficiency for both motion and still graphic images. The wavelet transform is carried out with the highest possible input image quality, maintaining 4:4:4 luma/chroma information so that image detail is not lost in the transform process. Coding of the wavelet transform coefficients is carried out using an approach that exploits the nature of the wavelet data to provide highly efficient coding of images, that enables identification of redundant data, which can easily be discarded while yielding visually lossless compression.

Color System

Standards-based video compression systems typically process component video with 4:2:0 or 4:2:2 color subsampling, which can distort image detail through incomplete processing and distortion of color information. This becomes particularly evident when these compression systems are applied to computer imagery, which frequently has single pixel lines, small characters, and graphic detail. See Figure 7-1. Maintenance of 4:4:4 color information is typically not supported by commonly used compression products, but is supported in PURE3.

Optimized for Graphics and Video

The PURE3 codec supports capture and preservation of both video and computer graphic formats at their native resolution, aspect ratio, and frame rate, maintaining all of the pixel detail and

The PURE3 codec supports capture and preservation of both video and computer graphic formats at their native resolution, aspect ratio, and frame rate, maintaining all of the pixel detail and motion. This ensures natural, lifelike reproduction of any input format.

Table 7-1. Comparison of Video and Computer Graphic Images

Description	SD/HD Video	Computer
Origination	Naturally produced image in real-world / camera or sensor	Synthetically produced on a computing device
Motion	High motion	High to low motion
Signal interface	Digital or analog	Digital or analog
Pixel transition	Graduated (smooth)	Discrete (sharp)
Color space	YUV (luminance and color difference)	RGB (red, green, blue)
Color resolution	4:1:1, 4:2:0, or 4:2:2 typical	4:4:4
Common resolutions	720x486, 720x625, 480p, 525p, 720p, 1080i, 1080p	SVGA, XGA, WXGA, SXGA, SXGA+, UXGA, WUXGA and many more
Interlaced or progressive signals	Interlaced and progressive	Typically progressive
Frame rates	24 Hz, 25 Hz, 30 Hz, 50 Hz, 60 Hz	60 Hz, 70 Hz, 72 Hz, 75 Hz, 85 Hz, and more

PURE3 achieves low latency image compression by processing images in a single pass.

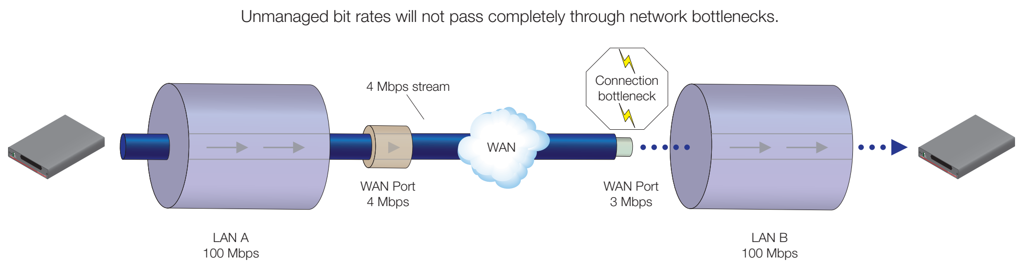


Figure 7-3. Illustration of potential network bottlenecks

motion. This ensures natural, lifelike reproduction of any input format.

The requirements for compression of video and computer graphic images are different. A comparison of the two image types is presented in Table 7-1.

The differences listed in the table help identify why a product designed for encoding video will probably provide sub-optimal results encoding detailed computer graphics.

Low Latency

PURE3 achieves low latency image compression by processing images in a single pass. This contrasts with motion-based codecs such as MPEG, in which multiple passes or expanding the Group of Pictures - GOP are required to achieve a target bit rate. For PURE3, the time required to execute the compression is determinate, and is independent of the image content. A proprietary, programmable temporal compression scheme provides “absolute” coding that relies only on absolute differences in content between frames. It does not require either image history or forward prediction, and therefore imposes no increase in processing latency.

Once the image content has been compressed by PURE3, low latency is further ensured through the use of UDP transport. When UDP is used together with RTP and RTCP protocols, streaming products that use the PURE3 codec provide high immunity to network transmission errors, without the need to carry redundant or overhead packet data which would increase latency. TCP transport is available for use in specialized applications.

Highly Efficient Compression

PURE3 uses DWT - Discrete Wavelet Transform as the transform method which offers an important advantage over DCT - Discrete Cosine Transform in that the coefficients tend to be statistically distributed within a small group of values, and therefore can be coded very efficiently. Various “compression profiles” can be applied to optimize coding efficiency. Each compression profile uses a weighting method for quantizing the wavelet coefficients that exploits the visual perception characteristics of the human eye, while minimizing the compressed image data. The use of compression profiles helps the user to determine the degree and nature of compression, so that it may be optimized for both the data bandwidth available and the application.

PURE3 further achieves low bit rates through a proprietary temporal compression scheme. When the image content has low motion, the PURE3 codec is capable of achieving incredibly high compression ratios while maintaining visually lossless performance. Compression ratios for full-motion, high entropy video sources result in more moderate compression ratios. The PURE3 temporal compression scheme is also capable of separating real-image motion from signal noise or naturally occurring noise.

Controlling the rate at which data is released to the network is just as critical as providing a

Streaming bit rates must be tightly managed to pass completely from sender to receiver.

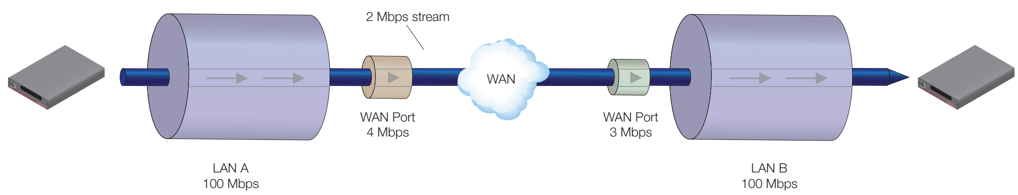


Figure 7-4. Illustration of tight bit rate management

wide variety of compression controls. Short, high bursts of data can quickly overload buffers in switching and routing equipment, resulting in lost data, particularly equipment used to manage thin communication links.

Options are available to select constant quality, constant bit rate, or peak bit rate, to control the amount of streaming traffic delivered to the network. Bit rate ranges for typical PURE3 applications and other codecs are presented in Tables 7-2 and 7-3.

PURE3 Error Concealment

As discussed earlier, real-world, switched IP networks ultimately will produce unpredictable errors including out-of-order, dropped packets, or bit errors. The PURE3 codec includes an error concealment system, which makes it highly immune to network errors. Picture data from previous frames are held at the decoder as a reference. If packet errors are experienced, this data is used to conceal the missing information. Error concealment is usually isolated to a small area, and successive video frames update with correct information as the next frame is received. The results of the PURE3 error concealment system are impressive. Video image integrity is maintained under heavy packet loss, and errors are rarely visible. When identified, only a small portion of the picture is affected and the duration is very short. An illustration of PURE3 error concealment in action over a sequence of video frames can be seen in Figures 7-5 and 7-6.

The PURE3 codec includes an error concealment system, which makes it highly immune to network errors.

Typical PURE3 Codec Bit Rates	Low	High
High Definition Surveillance (720p or 1080i): Indoor / Outdoor Monitoring	5 Mbps	20 Mbps
SXGA Computer Visualizations (1280x1024 @ 60 Hz): Maps or Models	1 Mbps	15 Mbps
WUXGA Simulations (1920x1200 @ 60 Hz): Lifelike Synthetic Animations with High Motion	15 Mbps	50 Mbps
High Definition Camera Transport (720p or 1080i), Broadcast Detail/Motion	40 Mbps	90 Mbps

Table 7-2. Typical bit rates for applications using VN-MATRIX, visually lossless with full-motion support

Typical Standards Based Encoder Application Bit Rates	Low	High
H.264 HD Internet Streaming	2 Mbps	6 Mbps
H.264 HD Telepresence (720p or 1080i): Talking Head Video Content	3 Mbps	10 Mbps
H.264 HD Camera Transport (720p or 1080i): Broadcast Detail/Motion	6 Mbps	25 Mbps
MPEG I-Frame HD Camera Transport (720p or 1080i): Broadcast Detail/Motion	100 Mbps	150 Mbps
JPEG2000 HD Camera Transport (720p or 1080i): Broadcast Detail/Motion	100 Mbps	200 Mbps
High Definition Video (HD-SDI), SMPTE 292M: Uncompressed	-	1.485 Gbps

Table 7-3. Typical bit rates reported by encoding products using standards-based compression systems

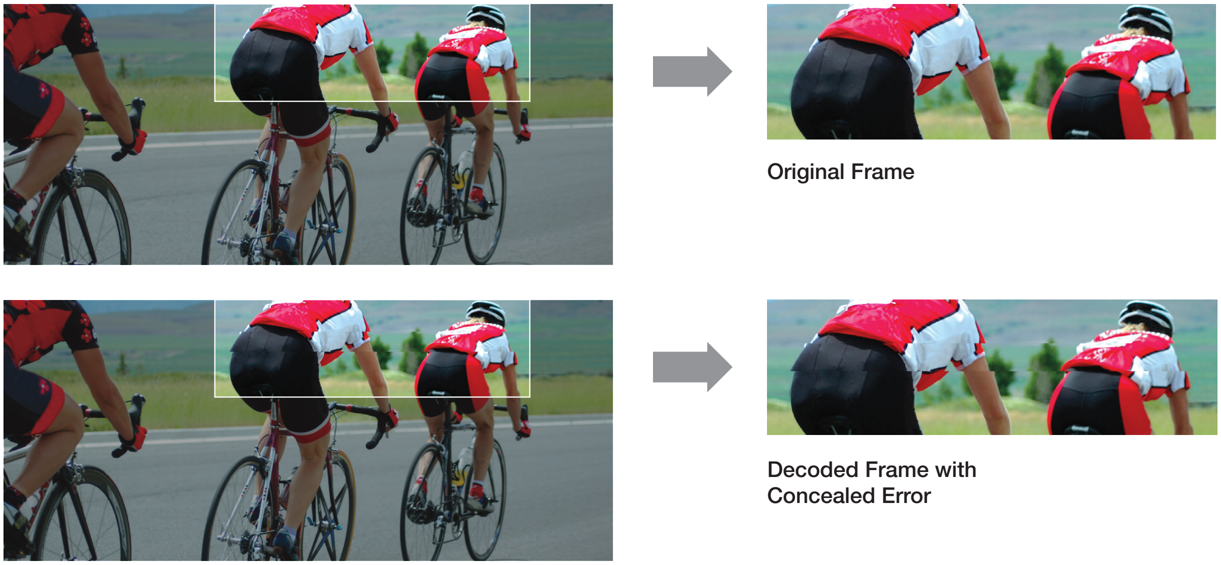


Figure 7-5. Illustration of PURE3® error concealment on a single frame

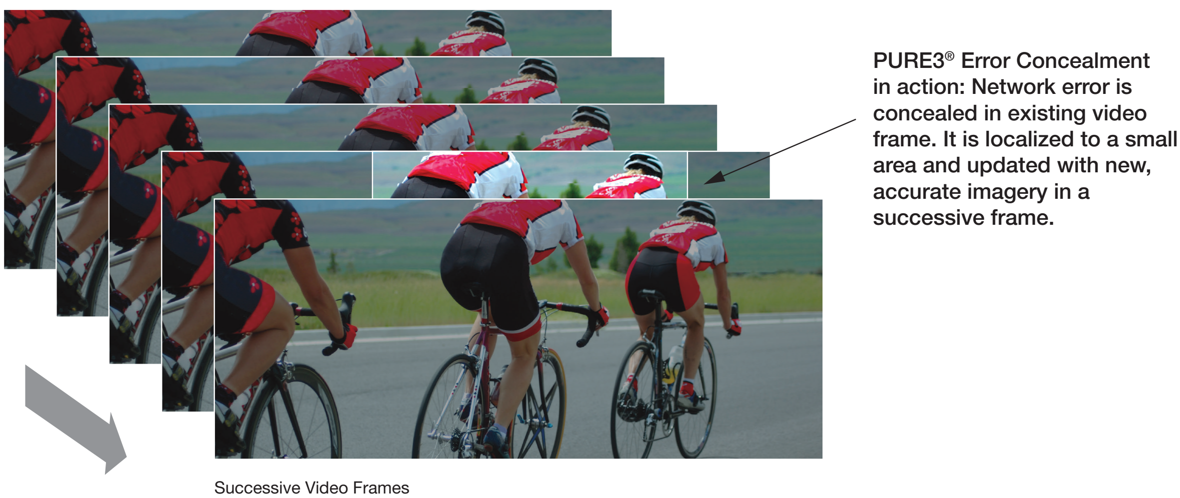


Figure 7-6. Illustration of PURE3® error concealment on a sequence of successive motion video frames

As applications continue to migrate delivery of real-time video from dedicated connections to switched and routed IP networks, understanding the stability that an error correction or concealment system provides to AV streaming systems is important.

Scalable Solutions

AV streaming solutions with PURE3 are highly scalable through the combination of efficient compression and support of multicast streaming, together with an error concealment system that does not require additional bandwidth or retransmission of data.

Stream Synchronization

Products applying PURE3 encode AV information on an absolute frame basis. Use of absolute frame information and not estimated data along with timecode in RTP allows video, audio and data streamed between encoders and decoders to maintain synchronization. This provides the ability to support several demanding application requirements, including:

- Audio/video lip sync
- Synchronization – genlock and framelock of images across multiple presentation screens

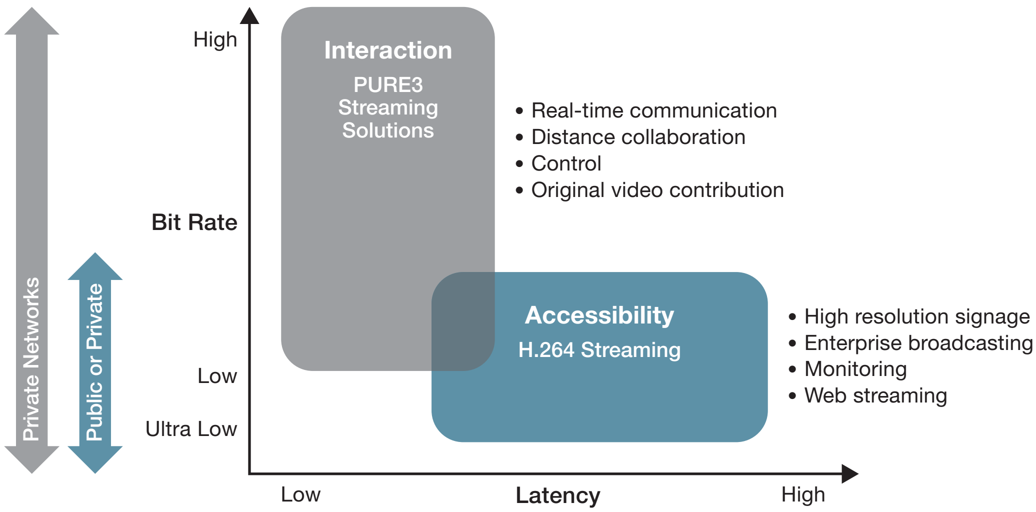


Figure 7-7. Streaming applications by bit rate and latency

- Ability to scale synchronized playback solutions across multiple storage servers
- Synchronize recording of external ancillary data to audio and video streams
- 4K and 3D streaming applications

Comparing Codecs

Video codecs for streaming are typically applied in one of the two application categories presented in Figure 7-7. The following generalizations can be made when comparing them:

- H.264 is positioned well to serve low and ultra-low bit rate video applications where interactivity is not required. Product implementations with low-delay H.264 codecs have also been developed, such as for videoconferencing. Error correction or high QoS network design is often required to ensure video is delivered reliably when using RTP protocol streams, or adaptive bit rate management control with HTTP and TCP protocols can be used. The H.264 codec is well-positioned to serve both enterprise and public networks where interoperability, low bit rate, and connection-oriented or connectionless protocols are used.

- The PURE3 codec is positioned to serve very high quality video and computer graphic mission-critical streaming applications that require very low delay to support interactive, real-time communication. It includes an error concealment system that eliminates the need for error correction or implementation of high-cost, high QoS networks. Products with the PURE3 codec will typically be used in enterprise applications where the network is managed.

- JPEG 2000 is positioned to deliver very high quality video applications with low delay, but uses higher bit rates. Error correction systems, which increase bandwidth and delay, and high QoS network designs, are typically required. Products with JPEG 2000 will be used in enterprise solutions where the network environment can be managed. As the bandwidth available from networks increases, use of JPEG 2000 streaming products may become more common.

Extron continually evaluates codecs and produces products that incorporate each one of the technologies identified here. As technologies mature and emerge, look for Extron to apply compression and streaming technologies that will produce the best results for professional AV applications.

H.264 well-positioned to serve both enterprise and public networks where interoperability, low bit rate, and connection-oriented or connectionless protocols are used.

Streaming System Designs

Where is Extron AV Streaming over IP used?

Extron manufactures streaming products that support a wide variety of applications. Potential streaming applications are organized into two primary classes:

H.264 Standards Based Streaming

Streaming products based on the H.264 standard provide the flexibility to deliver video over a variety of dedicated, shared use, or public networks. They support interoperability to allow PCs and many other H.264 devices to decode and present AV streaming. Extron's H.264 streaming products are used in these applications:

Streaming Presentations to Overflow Rooms

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AV Presentation Room Monitoring

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Live Streaming Over the Internet for Local Governments

Page 50

Distributed Patient Monitoring System

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Streaming of School Announcements and Central Media Assets

Page 54

Corporate Enterprise H.264 Streaming

Page 56

High Resolution Signage

Page 58

PURE3 High Resolution Video & Graphic Streaming

Some streaming applications have unique performance requirements, including low delay for collaboration, lossless compression, immunity to network errors for quality-critical applications, and preservation of native resolution for computer-video graphics. The Extron PURE3 codec fully supports all of these performance requirements.

HDCP-Compliant Streaming

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Distance Collaboration across an Enterprise

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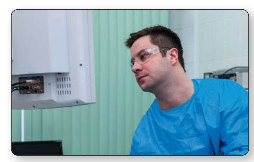
Campus Security Center Extension to Situation Room

Page 64

Low Delay Streaming in a Gross Anatomy Lab



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After Action Review

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Remote Control of Real Time Video Production Equipment

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Virtual Switching of AV Sources over Networks

Page 72

Cross-Country Production Collaboration

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Overview

Presentation rooms and auditoriums may occasionally feature presentations that must reach more people than they can accommodate. To solve this challenge, AV streaming can be used to provide a flexible and cost-effective solution to extend presentations to Overflow rooms or other locations anywhere on the network

Room Needs Assessment

Source Inputs	Sources in the auditorium include a PC, HD-SDI video camera, and a microphone.
Display - Near End	A high definition projection system is installed in the auditorium.
Physical Location	Presentations are viewable in the auditorium, in pre-determined Overflow rooms, or other locations either in the same facility or across campus, as required.
Control System	An Extron TLP Pro 720T in the Auditorium and a TLP Pro 1020T in Overflow rooms provide convenient control of the AV system
Network	A shared use Facility LAN - local area network connects the Auditorium with Overflow locations.
Displays - Far End	Overflow rooms are equipped with projectors or flat panel displays. Additional viewing may occur on laptop or PC displays, as required.
Functional Requirements	Audio and video must be streamed at bit rates appropriate to the capacity of the LAN, but high enough to provide image quality access for viewing streaming presentation content such as live camera, power point presentations or other AV content on the Overflow rooms' high resolution projection systems. A simple touchpanel interface is required to manage AV system devices.

System Design Solution

Streaming Encoder

An Extron **SMP 351** Streaming Media Processor accepts the HDMI output of the computer and the HD-SDI output of the camera.

Network

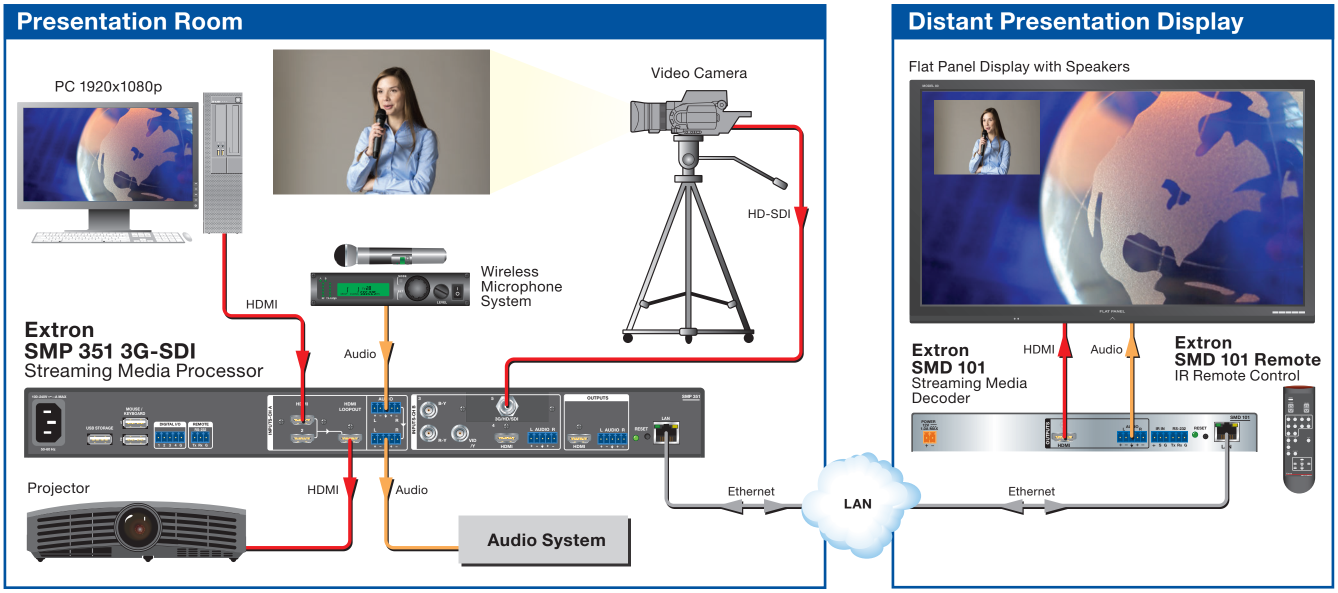
The shared-use facility network uses pull streaming to extend presentations from the Auditorium to Overflow rooms. If many connections are required, multicast push streaming can be deployed if the network is equipped with routers and switchers configured to support IGMP - Internet Group Management Protocol for multicast streaming. The SMP 351 supports both multicast and unicast streaming. Multicast is preferred when streaming to many endpoints, minimizing bandwidth, and efficiently distributing streams to end points. Streaming bit rates of 5 Mbps to 7 Mbps are sufficient to support high definition streaming. Very large displays such as those located in the Overflow rooms would benefit from a higher bit rate, such as 10 Mbps.

Decoding

Decoding is performed an Extron **SMD 101** located in each Overflow room to support streaming resolutions up to 1080p/60.

Control

In the Presentation room, an SMP 351 encoder is connected to the network, where users access the SMP 351's internal Web pages to remotely set up encoding before presentations, or make adjustments during presentations, to optimize streaming performance. In the Presentation room and Overflow rooms, Extron TLP Pro TouchLink® touchpanels provide Ethernet-based control of AV presentation systems, displays, and the encoder, allowing users to easily initiate or terminate streaming as needed.


Presentation Room
PC 1920x1080p
Video Camera
HD-SDI
Wireless
Microphone
System
HDMI
Audio
Extron
SMP 351 3G-SDI
Streaming Media Processor
Projector
HDMI
Audio
Audio System
Ethernet
LAN
Ethernet
Distant Presentation Display
Flat Panel Display with Speakers
Extron
SMD 101
Streaming Media
Decoder
HDMI
Audio
Extron
SMD 101 Remote
IR Remote Control

Overview

Many organizations with presentation environments want the ability to monitor and support live presentation sessions from a remote location, such as a helpdesk. AV technical staff can use Ethernet-based system control to access AV system and device functions, but often have to be in the presentation room to see what's on the display. This places demands on their resources by limiting their availability to concurrently address other support needs. Adding video streaming capability to presentation systems conveniently allows staff to view live on-screen content in high definition on their PCs. With video streaming, personnel can better manage and support multiple live presentation systems from the helpdesk or other locations.

Solution Needs Assessment

Source Inputs	Each AV presentation room includes AV connections at the lectern to support a presenter's laptop, with VGA and stereo audio. Every room also has two desktop PCs, a DVD/VCR combo player, and a CCTV tuner. A multi-format presentation switcher interfaces all sources with the projector.
Displays - Near End	A single projector is installed in each presentation room.
Physical Location	AV support staff must be able to view on-screen content for any presentation room on their PC, at the helpdesk, or anywhere they can use a PC on the network.
Control System	A touchpanel control interface allows the presenter to select the source to be displayed on the screen and adjust or mute the volume. The video streaming encoder will be controlled with the switcher to ensure the correct source is being streamed at the optimal quality and bit rate.
Network	The facility has a shared use network. Streaming traffic must be kept to a minimum, at an ideal maximum of 2 Mbps according to IT policy.
Encoder Capabilities	The encoder must be compatible with various source signal formats and integrated into an existing system without the need for extra AV equipment. It also should allow encoding bit rate adjustments whenever necessary.

Solution Needs Assessment

Displays - Far End	The streamed output will be viewed on PCs at the helpdesk or elsewhere in the facility. Staff should be able to quickly and easily configure any PC to receive streamed video.
Additional System Requirements	An audio system for playback of voice and program audio content. A microphone is to be provided for the presenter.

System Design Solution

Streaming Encoder

For each presentation room, an Extron **SME 100 HD** Streaming Media Encoder will be interfaced into the existing AV system between the **MPS 409** Media Presentation Switcher and the projector. The input loop-throughs of the SME 100 HD allow it to be installed without affecting AV system functions. An integrated three-input switcher on the SME 100 HD enables selection between the DVI, VGA, and standard definition outputs from the MPS 409 for streaming. EDID emulation and Auto-Image™ allow the SME 100 HD to be integrated into the system without extensive setup procedures. The SME 100 HD also features integrated video and graphics scaling to optimize source signal conversion for streaming.

Control System

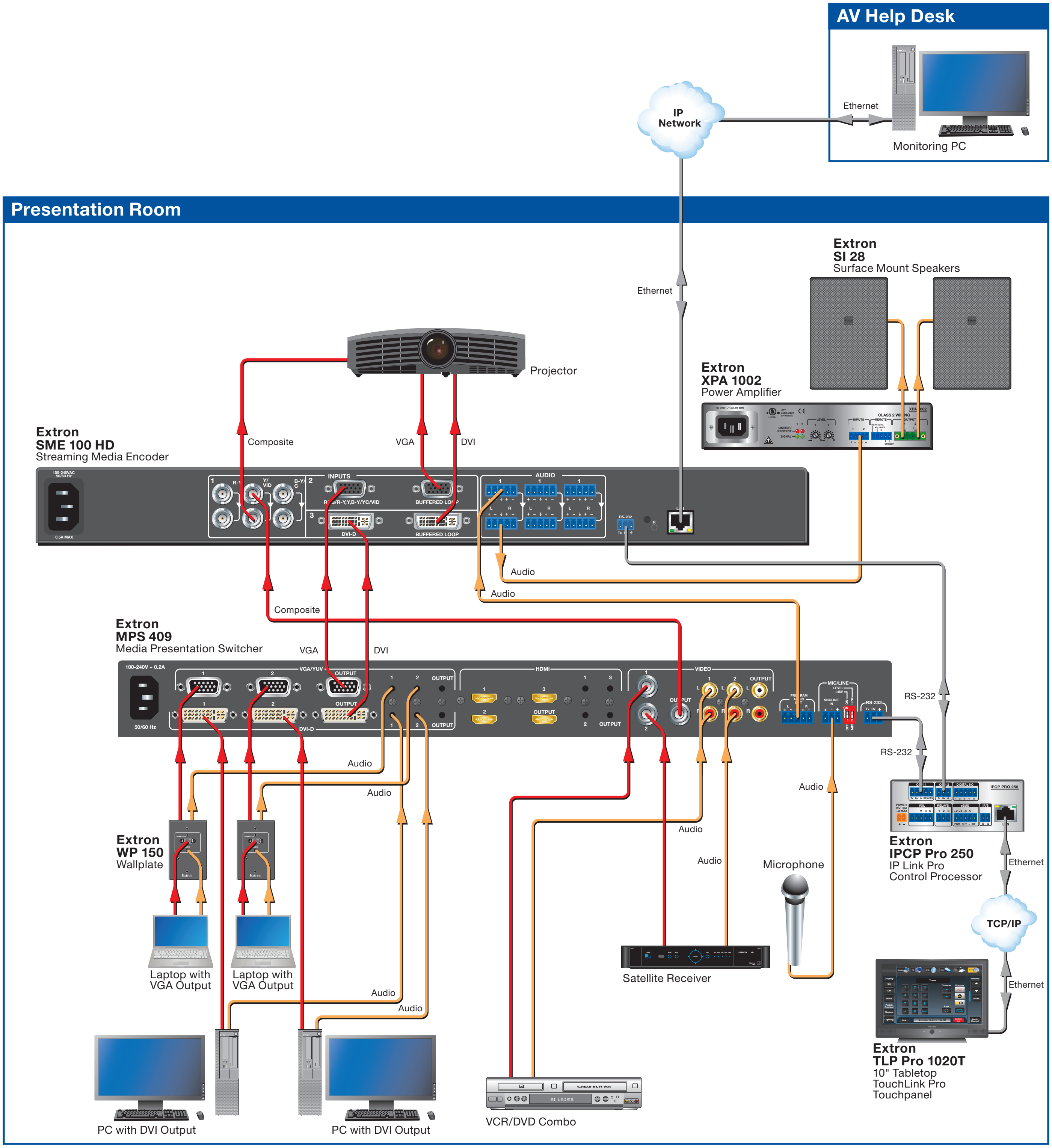
An Extron **TLP Pro 1020T** TouchLink® Pro Touchpanel is the user interface to the system, and works together with the **IPCP Pro 250** Ethernet Control Processor to control AV sources, the MPS 409, and the projector. The source switching in the SME 100 HD will be programmed to work together with the MPS 409 so that on-screen presentation content is streamed at the same time. Support staff will have additional control capabilities for the SME 100 HD from a Web browser on any PC, including the ability to stream any source in the system, and adjust streaming video quality including resolution, bit rate, frame rate, and picture controls.

Network

By combining scaling and selectable encoding quality control, the SME 100 HD is able to meet low bit rate targets in a converged network environment. Streaming presets are available so that staff can quickly adjust video resolution and encoding settings to respond to changes in IT policy, or examine a source at or near its native resolution and quality.

Decoding

The AV staff will be able to monitor presentations or perform troubleshooting from their office connected to the enterprise network. A PC can connect through a Web browser to the SME 100 HD, which automatically delivers an application capable of decoding the H.264 video streams. Third-party media players such as QuickTime® or VLC can also be used to easily view content on any PC in the facility.



AV Help Desk
Ethernet
Monitoring PC
IP Network
Presentation Room
Ethernet
Extron SI 28 Surface Mount Speakers
Extron XPA 1002 Power Amplifier
Projector
Composite
VGA
DVI
Extron SME 100 HD Streaming Media Encoder
Audio
Audio
Composite
Extron MPS 409 Media Presentation Switcher
VGA
DVI
RS-232
RS-232
Audio
Audio
Audio
Extron WP 150 Wallplate
Audio
Audio
Microphone
Extron IPCP Pro 250 IP Link Pro Control Processor
Ethernet
TCP/IP
Ethernet
Laptop with VGA Output
Laptop with VGA Output
Satellite Receiver
Audio
Audio
Extron TLP Pro 1020T 10" Tabletop TouchLink Pro Touchpanel
PC with DVI Output
PC with DVI Output
VCR/DVD Combo

Overview

Many local state and city governments would like to be more transparent with their citizens. One way to accomplish this is to make meeting or committee sessions easily accessible to them over the Internet. Being able to access these sessions via live or on-demand streaming will make it convenient for citizens to become more aware and involved with the political process, and informed of current affairs. For local governments, streaming may be a viable alternative to broadcasting on public access or local television.

Room Needs Assessment

Source Inputs	Professional cameras and real-time production equipment provide a high definition component video output plus stereo audio. The cameras are situated throughout the central meeting room of the government building. The production equipment is installed in the AV control room, adjacent to the meeting room.
Displays - Near End	A professional LCD flat-panel display allows the AV production manager to monitor the video capture and production. Streamed video will be monitored on a laptop or desktop PC.
Geography	There is technically no limit as to the size of the geographical region where streamed content may be viewed, but typically viewers would be located within the city, county, or state.
Control System	The AV/IT manager would like to be able to remotely configure and operate the streaming encoder as well as other equipment that will be part of the online media delivery system.
Network	Citizens will access online video from their homes or wherever they may be, through a network connection, Wi-Fi, or 3G/4G mobile service.
Displays - Far End	Internet users accessing streaming video may be viewing on their laptops, desktop PCs, televisions, tablets, or smartphones.
Functional Requirements	The user experience accessing online content should essentially be as easy as accessing streaming video from major sites such as Hulu or YouTube. A simple method will be required for internal or external users to subscribe to video streams over the Internet through their Web browsers from PCs or mobile devices.

System Design Solution

Streaming Encoders

Audio and video camera feeds from the local government building are input to real-time production equipment to select and monitor multiple camera feeds, and incorporate titling and transition effects. The HD component video output from the production equipment is connected to the Extron **SME 100 HD** Streaming Media Encoder and looped through to the monitoring display. The SME 100 HD will not be used to stream directly out to the Internet, but instead will be used to interface with the production video, provide high quality video processing, and deliver a master 720p stream that will be delivered to a Content Delivery Network - CDN service located outside the facility. The CDN will manage and administer Internet user access to the streaming content.

Network

Shared use network bandwidths ranging between 5 and 7 Mbps should be made available for the SME 100 HD to stream HD video and audio to the CDN and perhaps serve some internal locations. CDN content servers then forward the stream throughout the CDN service system. Internet users requesting streamed video are connected to the closest CDN content server, which delivers the video stream to their devices. Push streaming is applied transporting video to the CDN. Pull streaming is used to transport video to a Media Server, which fulfills requests for overflow streaming within the facility.

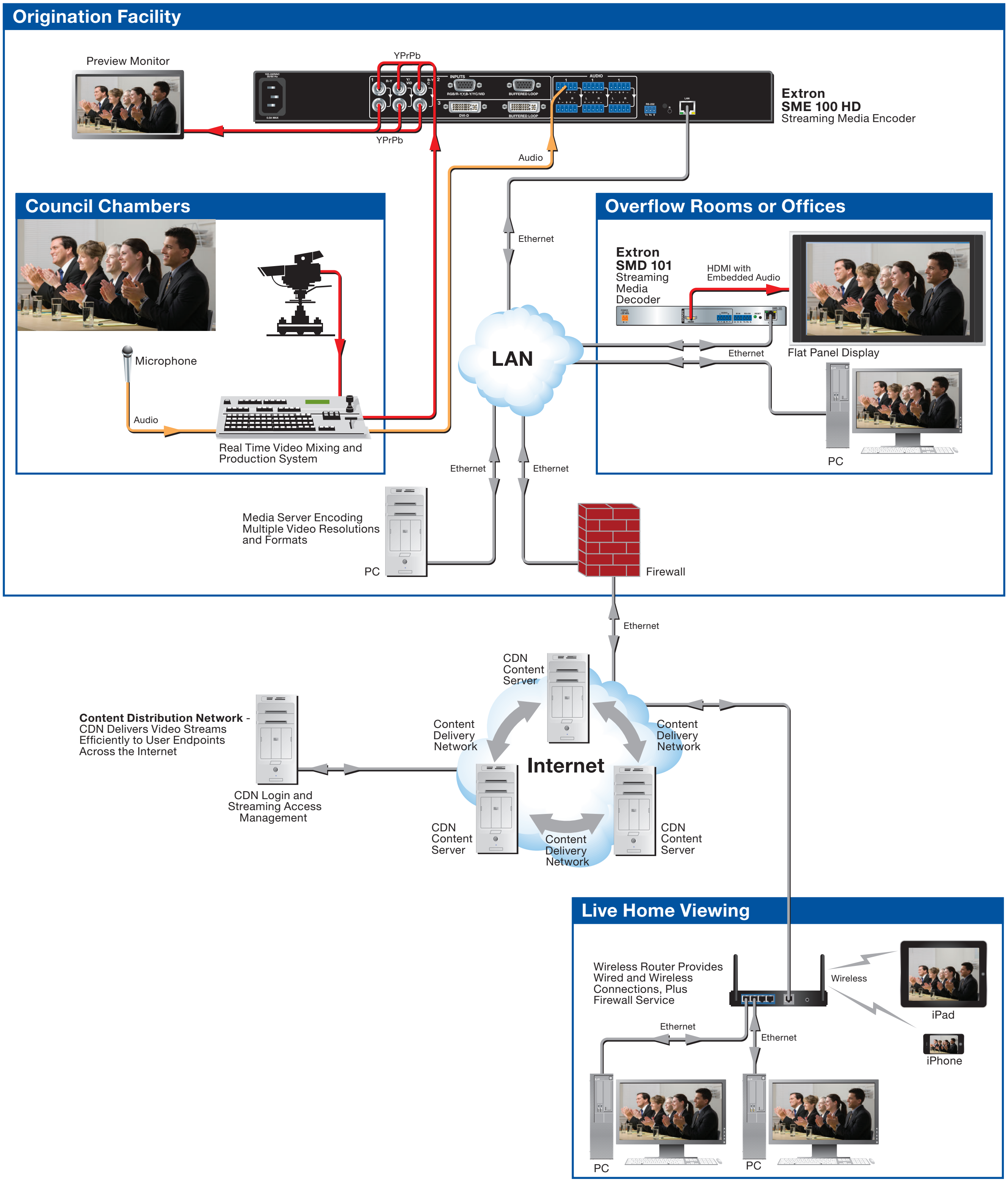
Decoding

As with other streaming or on-demand video resources on the Internet, users will be able to access online content with a wide variety of devices. Careful selection of services offered by a CDN ensures that video streaming can be converted to the streaming protocols required by various decoding devices such as PCs, tablet computers and smartphones.

An Extron **SMD 101** decoder presents streamed AV content on a flat panel display in the Overflow Room, and the AV/IT manager can quickly set up PCs across the facility to receive streaming video.

Control

The AV/IT manager will configure and control the SME 100 HD over the network by accessing the embedded Web pages. The embedded player allows the streams to be seen as changes are made.



Origination Facility
Preview Monitor
YPrPb
YPrPb
Extron
SME 100 HD
Streaming Media Encoder
Audio
Council Chambers
Microphone
Audio
Real Time Video Mixing and Production System
Ethernet
LAN
Overflow Rooms or Offices
Extron
SMD 101
Streaming Media Decoder
HDMI with Embedded Audio
Ethernet
Flat Panel Display
PC
Ethernet
Ethernet
Media Server Encoding Multiple Video Resolutions and Formats
PC
Firewall
Ethernet
CDN Content Server
Content Distribution Network -
CDN Delivers Video Streams Efficiently to User Endpoints Across the Internet
Content Delivery Network
Content Delivery Network
Internet
CDN Login and Streaming Access Management
CDN Content Server
Content Delivery Network
CDN Content Server
Live Home Viewing
Wireless Router Provides Wired and Wireless Connections, Plus Firewall Service
Wireless
iPad
Ethernet
Ethernet
iPhone
PC
PC

Overview

To improve patient care, hospitals can use AV streaming to distribute a visual summary of patient vital statistics to staff throughout the facility. This helps a limited number of staff efficiently monitor patient status and quickly identify urgent patient needs.

Room Needs Assessment

Source Inputs	Vital statistics and other information originate from medical equipment in patient rooms, presenting data and graphic information at 1080p resolution. Identification numbers are presented in lieu of patient names, preserving anonymity.
Displays - Near End	Patient data is presented on a flat panel display near medical source equipment.
Physical Location	Patient monitoring equipment is centrally located in the facility. Flat panel displays presenting patient information are located at working stations throughout the healthcare facility.
Control System	The streaming distribution system will be configured to operate in a fixed operational state. Healthcare facilities such as this operate 24 hours per day, seven days per week.
Network	Sections of the facility network streaming patient data are equipped to support 100/1000BaseT connection speeds and can support streaming traffic in the 5 to 10 Mbps range. The network is configured to support IGMP multicast streaming.
Displays - Far End	Large, 1080p flat panel displays are strategically placed in areas where staff members regularly conjugate or their normal working areas.
Functional Requirements	The 1080p video content is streamed at very high quality, preserving the text, characters, and graphical information so that it can be accurately interpreted by healthcare workers.

System Design Solution

Streaming Encoders

An **SME 100 HD** encoder supplied with a 1080p DVI input signal from medical equipment is streamed at 1080p, preserving very high picture quality. The bit rate used to transport this high quality imagery does not exceed 10 Mbps and is typically as low as 5 Mbps. Multicast push streaming is used for transport across the facility LAN.

Network

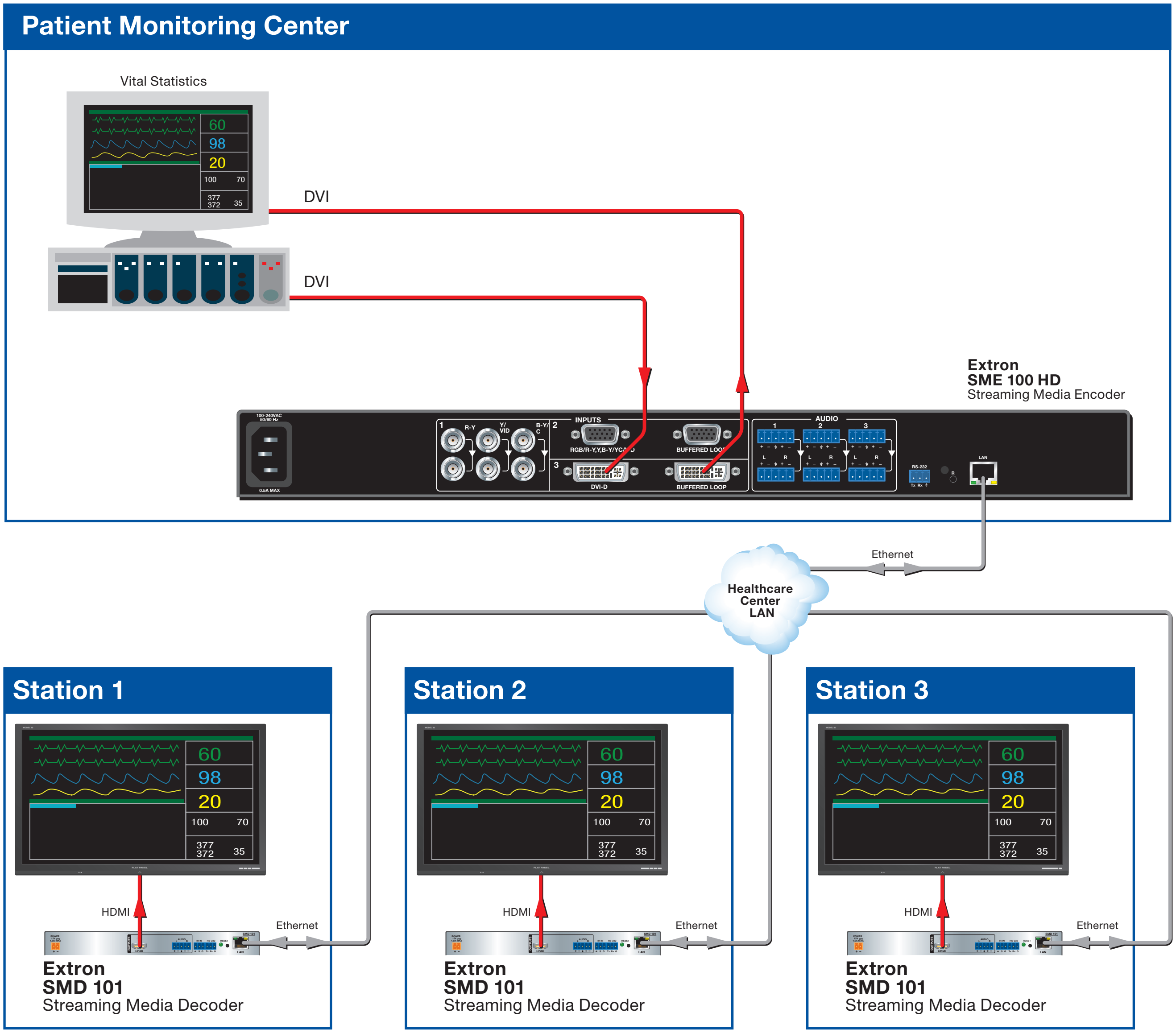
10/100/1000 BaseT Ethernet network switches are connected to the Extron SME 100 HD encoder and every SMD 101 decoder throughout the facility. The facility network is configured to support IGMP multicast traffic, ensuring that high definition AV streams are distributed efficiently.

Decoding

Compact **SMD 101** decoders are conveniently mounted behind, or in close proximity to, flat panel displays. Extron mounting kits are available to secure SMD 101 decoders into equipment racks, under tables or desks, or onto poles. HDMI cables connect SMD 101 decoders to flat panel displays using Extron **Lock-it**® cable lacing brackets, which ensure that reliable connections are maintained. Each SMD 101 decoder has its adjustable buffer set to 2 seconds, ensuring continuous decoding, even in situations of network contention and errors. Decoded video preserves the original 1080p source resolution, providing an accurate reproduction of the original data and graphic information which is presented upon the 1080p flat panel displays.

Control

The SME 100 HD encoder and SMD 101 decoders have been configured using their embedded web page interfaces to continuously stream one 1080p video source from the SME 100 HD encoder to every SMD 101 in the system. As there is no need to change this continuous stream, the system is left in a fixed operating state.


Patient Monitoring Center
Vital Statistics
DVI
DVI
Extron
SME 100 HD
Streaming Media Encoder
Ethernet
Healthcare Center LAN
Station 1
HDMI
Ethernet
Extron
SMD 101
Streaming Media Decoder
Station 2
HDMI
Ethernet
Extron
SMD 101
Streaming Media Decoder
Station 3
HDMI
Ethernet
Extron
SMD 101
Streaming Media Decoder

Overview

Many schools broadcast morning announcements to provide students, teachers, and administrators with the day's schedule and other important information. This information can be delivered in many different ways, including the school PA system, e-mail, digital signage, or a dedicated internal video broadcast system. Video streaming technology and classroom AV systems can be used to deliver announcements, providing high quality video and audio to students and staff. AV streaming can also connect centrally located media assets to specific classrooms on an as-needed basis.

Room Needs Assessment

Source Inputs	Sources include facility announcements from a camera and microphone as well as messaging originating from a PC in the administration offices. Devices such as cable/satellite receivers and DVD/VCR players are located in a Media Center.
Displays - Near End	Flat panel displays provide a confidence view of source content. Alternatively, encoded sources may be previewed by decoding them from a PC prior to streaming to classrooms.
Physical Location	AV source equipment and streaming encoders are located in a studio within the administration offices. Other AV sources, such as DVD/VCR player and cable/satellite receivers are located in the Central Media Center.
Control System	A control system with touch panel manages the availability of streaming sources to each of the classrooms. Streamed announcements are only made at select times, and content is streamed from the Central Media Center based on scheduled bookings. Each classroom needs to switch between presenting locally controlled sources and streamed content.
Network	The campus network requires 100BaseT network speeds and must support IGMP multicast traffic.
Displays - Far End	Each classroom has a projector that can be switched between local sources and streamed inputs.
Functional Requirements	Instructors in each classroom can select streaming channels, similar to the way they use a television at home and the streaming decoder input is selected in the same manner as other locally connected AV sources.

System Design Solution

Streaming Encoders

Extron **SME 100 HD** encoders stream centrally located AV sources across the campus LAN. The administration office includes two SME 100 HD encoders, one which streams announcements from a camera and microphone, and a second unit, which streams campus messages prepared on a PC.

A Central Media Center maintains a combination DVD/VCR player and a cable/satellite tuner, which streams content to one or many classrooms upon request for educational purposes only. Streaming can be scheduled in advance.

720p HD video is streamed across the campus at a maximum of 5 Mbps including stereo audio. Use of multicast streaming ensures that 5 Mbps is the maximum bandwidth each source contributes to any portion of the network.

Network

10/100/1000 BaseT Ethernet network switches are distributed throughout the campus. Each classroom includes an **SMD 101** decoder which is connected to the campus network. The network is configured to support IGMP multicast traffic supporting efficient distribution of streamed content.

Decoding

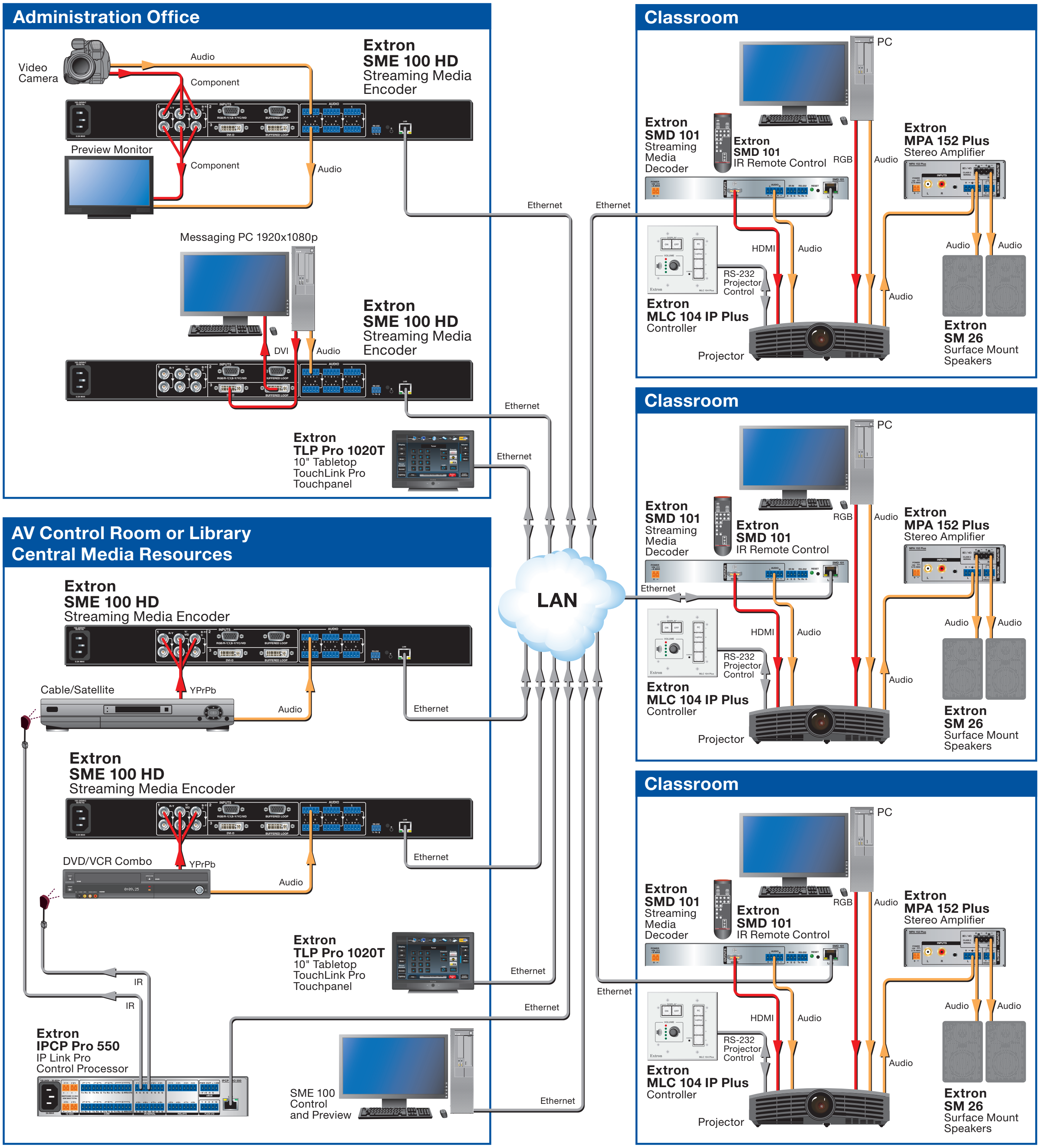
Extron SMD 101 decoders installed in each classroom supply an HDMI and analog stereo audio signal to the projection system. Audio delivered to the projector is switched out to an Extron **MPA 152** amplifier. Alternatively, HDMI-embedded audio can be supplied to the classroom's audio system.

Control

A channel list is configured on each SMD 101 identifying available SME 100 HD AV streamed sources. This makes channel selection with the **SMD 101 Remote**, an IR handheld remote control, straightforward. Instructors can select channel numbers directly or use the channel up/down button.

An Extron **MLC 104 IP Plus** controller manages input source selection and power to the projector. The SMD 101 can either be controlled by the SMD 101 Remote, or integrated into the classroom's local AV control system. The MLC 104 IP Plus can also be programmed to select streaming channels.

The administration offices and Central Media Center use Extron **TLP Pro 1020T** touch panels and **IPCP Pro 550** control processors to select various streaming presets for the encoding and streaming of content from the SME 100 HD encoders.



Administration Office
Video Camera
Audio
Component
Extron
SME 100 HD
Streaming Media Encoder
Preview Monitor
Component
Audio
Messaging PC 1920x1080p
Extron
SME 100 HD
Streaming Media Encoder
DVI
Audio
Extron
TLP Pro 1020T
10" Tabletop
TouchLink Pro
Touchpanel
Ethernet
AV Control Room or Library
Central Media Resources
Extron
SME 100 HD
Streaming Media Encoder
Cable/Satellite
YPrPb
Audio
Ethernet
Extron
SME 100 HD
Streaming Media Encoder
DVD/VCR Combo
YPrPb
Audio
Ethernet
Extron
TLP Pro 1020T
10" Tabletop
TouchLink Pro
Touchpanel
IR
IR
Extron
IPCP Pro 550
IP Link Pro
Control Processor
SME 100
Control
and Preview
LAN
Classroom
PC
Extron
SMD 101
Streaming
Media
Decoder
Extron
SMD 101
IR Remote Control
RGB
Audio
Extron
MPA 152 Plus
Stereo Amplifier
HDMI
Audio
RS-232
Projector
Control
Extron
MLC 104 IP Plus
Controller
Projector
Audio
Audio
Audio
Extron
SM 26
Surface Mount
Speakers

Overview

The broadly accepted H.264 standard offers great compatibility for the transport of live streamed content or playback of AV media files over an enterprise network. Use of existing network infrastructures is a cost-efficient and practical way to extend content to distant viewing locations.

In this example, a large corporation streams presentations to overflow meeting rooms as well as ad hoc destinations on PCs connected to the Campus LAN. In addition, pre-produced promotional and training content is played back on demand from network shares.

Room Needs Assessment	
Source Inputs	Sources include a PC, DVD player, and an SMD 101 H.264 streaming media decoder, which decodes pre-produced AV content from network shares.
Display - Near End	A 1080p projector in the Presentation Room presents content encoded for streaming over the network, including local sources and decoded content from network shares.
Physical Location	The Presentation Room is located in the main corporate facility. Meeting rooms are distributed across the corporate campus, and are too distant for dedicated AV cable to be economically feasible.
Control System	Touchpanel controllers provide simple control in the Presentation Room and Meeting Rooms.
Network	The existing campus network is used to transport streaming content. A dedicated or segmented network cannot be installed.
Displays - Far End	Large, flat panel displays in meeting rooms present content streamed from the Presentation Room or from network shares.

System Design Solution

Streaming Encoder

An Extron **SME 100 HD** streaming media encoder in the Presentation Room provides high quality streaming for both motion video and computer-video signals. Switches between sources are clean and glitch-free, even between different signal formats.

Network

The campus LAN provides transport for streaming from an Extron SME 100 HD encoder and network attached storage. The SME 100 HD consumes sufficiently low bandwidth that streaming traffic can be mixed with data and voice traffic over the campus network.

Decoding

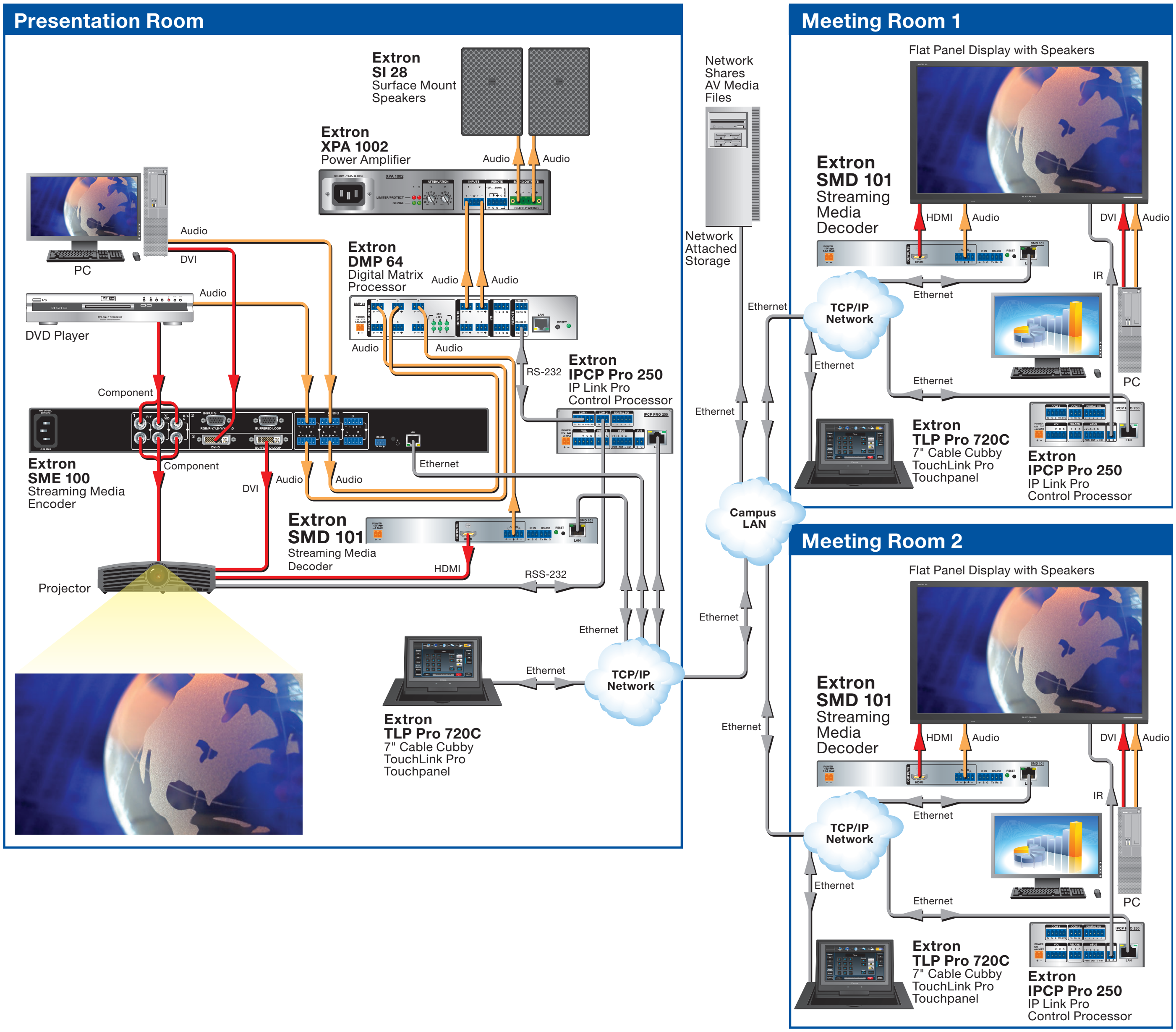
Extron **SMD 101** units decode live H.264 streams from the Presentation Room and play back AV media files available from network shares. The AV media files may be MP4 or MPEG-2 Transport stream container format files. Playlists and channel lists are prepared in SMD 101 decoders to make selection of live streams or defined AV media files simple.

Control

Extron **TLP Pro 720C** touchpanels and **IPCP Pro 250** control processors in the Presentation Room and Meeting Rooms provide simple control over playlists and channel lists pre-defined in SMD 101 decoders.

Monitoring

AV or IT staff can easily monitor live streaming from any PC in the facility. Access to the SME 100 encoder is password protected.


Presentation Room
Extron
SI 28
Surface Mount
Speakers
Extron
XPA 1002
Power Amplifier
Audio
Audio
PC
Audio
DVI
Extron
DMP 64
Digital Matrix
Processor
Audio
Audio
DVD Player
Audio
Audio
Audio
RS-232
Extron
IPCP Pro 250
IP Link Pro
Control Processor
Component
Extron
SME 100
Streaming Media
Encoder
Component
DVI
Audio
Audio
Ethernet
Extron
SMD 101
Streaming Media
Decoder
HDMI
RSS-232
Projector
Ethernet
Ethernet
TCP/IP
Network
Extron
TLP Pro 720C
7" Cable Cubby
TouchLink Pro
Touchpanel
Network
Shares
AV Media
Files
Network
Attached
Storage
Ethernet
Ethernet
Campus
LAN
Ethernet
Ethernet
Meeting Room 1
Flat Panel Display with Speakers
Extron
SMD 101
Streaming
Media
Decoder
HDMI
Audio
DVI
Audio
IR
TCP/IP
Network
Ethernet
Ethernet
Ethernet
PC
Extron
TLP Pro 720C
7" Cable Cubby
TouchLink Pro
Touchpanel
Extron
IPCP Pro 250
IP Link Pro
Control Processor
Meeting Room 2
Flat Panel Display with Speakers
Extron
SMD 101
Streaming
Media
Decoder
HDMI
Audio
DVI
Audio
IR
TCP/IP
Network
Ethernet
Ethernet
Ethernet
PC
Extron
TLP Pro 720C
7" Cable Cubby
TouchLink Pro
Touchpanel
Extron
IPCP Pro 250
IP Link Pro
Control Processor

Overview

IP networks can provide a cost-effective delivery infrastructure for AV signals. In this example, a facility shares company news, promotes events, and showcases new products and services to visitors and employees with a high-resolution signage system managed over the facility's network.

Room Needs Assessment	
Source Inputs	Sources include pre-produced content from a messaging PC and a cable/satellite receiver in the Central Media Studio.
Display - Near End	A local desktop flat panel display presents content encoded for streaming over the network.
Physical Location	Source equipment and a streaming encoder are located in the Central Media Studio.
Control System	Source streams may be selected either centrally from the Media Studio or locally from far end displays using an IR remote control.
Network	The facility network is segmented by IT staff for streaming of high resolution AV content from the Central Media Studio to the viewing locations.
Displays - Far End	Large flat panel displays in the Lobby and Cafeteria present pre-produced high resolution content streamed over the facility network. Two displays are presented in this application, but many more displays may be integrated into this distribution system.

System Design Solution

Streaming Encoder

In the Central Media Studio, an Extron **SME 100 HD** unit encodes standard high definition video and audio from a cable and satellite receiver for streaming across the facility LAN. A second SME 100 HD unit encodes and streams pre-produced facility messages from a messaging PC. Video is typically encoded to 1080p for streaming over the facility network.

Network

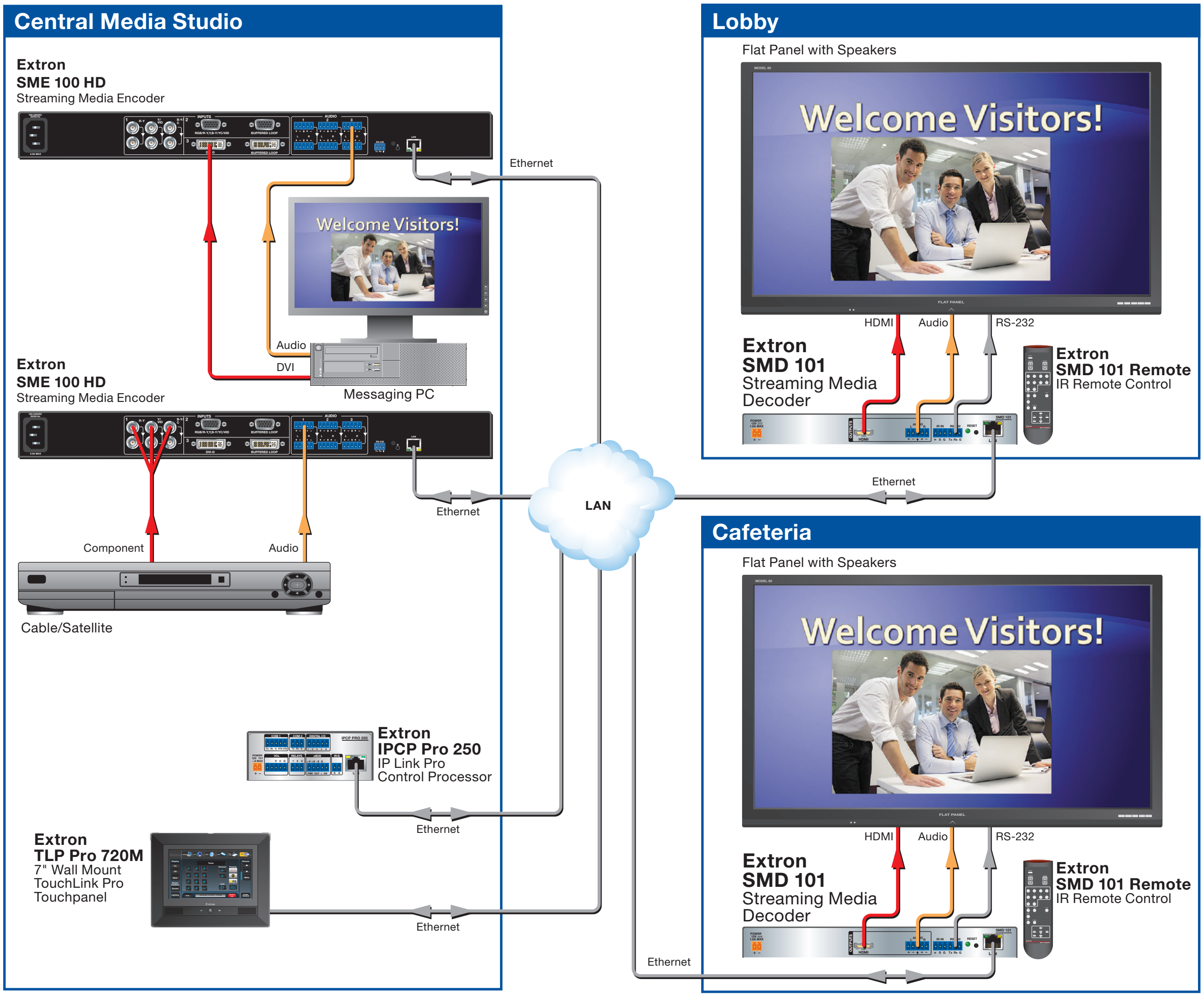
The facility LAN supports streaming bit rates from 5 to 7 Mbps from each SME 100 HD encoder. The network is configured by IT staff to support IGMP multicast traffic, which limits streaming traffic to the sections subscribed to each stream. This increases available bandwidth, allowing for efficient future growth. Many more displays and **SMD 101** decoders can be integrated into this system using multicast transport for audio and video.

Decoding

Large flat panel displays are installed in the lobby and employee cafeterias. Extron SMD 101 decoders are connected to the displays and decode selected AV sources. Where necessary, the SMD 101 scales output signals to match the display resolution. Analog stereo audio is supplied to embedded speakers in the flat panel displays. HDMI-embedded stereo audio is available when analog audio inputs are not available.

Control

Facility staff can select sources for local display with the Extron **SMD 101 Remote**, a handheld IR remote controller in the lobby or cafeteria. In the Central Media Studio, an Extron **TLP Pro 720M** Touchlink® Pro Touchpanel and accompanying **IPCP Pro 250** IP Link Pro control processor provide central management over the high-resolution signage streaming system. RS-232 control transported from the Central Media Studio to the SMD 101 streaming media decoders manage far end display functions, including power on/off.


Central Media Studio
Extron
SME 100 HD
Streaming Media Encoder
Ethernet
Welcome Visitors!
Audio
DVI
Messaging PC
Extron
SME 100 HD
Streaming Media Encoder
Ethernet
Component
Audio
Cable/Satellite
Extron
IPCP Pro 250
IP Link Pro
Control Processor
Ethernet
Extron
TLP Pro 720M
7" Wall Mount
TouchLink Pro
Touchpanel
Ethernet
LAN
Lobby
Flat Panel with Speakers
Welcome Visitors!
HDMI
Audio
RS-232
Extron
SMD 101
Streaming Media
Decoder
Extron
SMD 101 Remote
IR Remote Control
Ethernet
Cafeteria
Flat Panel with Speakers
Welcome Visitors!
HDMI
Audio
RS-232
Extron
SMD 101
Streaming Media
Decoder
Extron
SMD 101 Remote
IR Remote Control
Ethernet

Overview

Many AV systems today have completely digital processing and switching architectures. Use of just one HDCP-encrypted source, such as a Blu-ray player, satellite receiver, or PC with an HDMI output can mandate that every component along the signal path is HDCP-compliant. In addition, extending this content to other viewing areas will require HDCP-compliance in these locations as well. This corporate boardroom has an HDCP-compliant system that presents between one and four images on a large projection display. This content must be available to other meeting rooms; however it is impractical to use a continuous cable to connect these locations.

Room Needs Assessment

Staffing	The board room will be operated by company directors and managers who are not AV specialists, requiring a user-friendly operating interface.
Source Inputs	A Blu-ray player, satellite receiver, PC, and laptop/BYOD connection are fed into an MGP 464 Pro DI HDCP-compliant multi-window processor. An additional PC provides background content. The MGP 464 Pro DI presents various combinations or one to four images over the background source as a single, composited image. An audio system mixes the analog audio signals into a single stereo format. The video and audio from this system will be streamed to other locations.
System Reach	Video must be extended to reach three remote meeting rooms.
Network	The company network has been designed to support a variety of data, voice, and video applications. Sections of the network may be segmented or provisioned to support video streaming traffic between the defined endpoints.
Streaming Quality	Imagery comprised of multiple high-definition video sources and computer visualizations create a single high resolution image. Quality must be preserved, delivering a pixel-for-pixel reproduction of the board room's presentation to remote meeting rooms.
Functional Requirements	High quality imagery, including HDCP-encrypted video, will be extended from a board room to up to three remote meeting rooms.

System Design Solution

Streaming Encoder

An Extron **VNE 250** HDCP-compliant encoder employing the **PURE3®** codec is in the corporate board room, receiving a 1920x1200 HDCP-encrypted HDMI signal from an **MGP 464 Pro DI** processor. The loop out of the VNE 250 feeds the board room's flat panel display. If this display were not HDCP compliant, a green screen would be presented, indicating non-compliance.

The VNE 250's input signal is rapidly encoded in just 35 ms at the MGP 464 Pro DI's original resolution of 1920x1200, with 24 bit, 4:4:4 color sampling. This high-quality sampling ensures preservation of single pixel detail for small font, lines, and video/computer graphic signals.

Analog audio from the room's sound system is input into the VNE 250 encoder, which simultaneously passes it to the flat panel display. Audio is synchronized to video in VN-Matrix streaming systems.

Network

Local Area Network switches with 100/1000BaseT network connections are interfaced with the VN-Matrix encoder and decoders. The network switches can be segmented using a VLAN to separate streaming traffic from voice and data as it is transported between from encoders to decoders.

Streaming Decoder

An Extron **VND 250** decoder at each of the remote locations rapidly decodes the video/graphic stream in just 35 ms, and outputs an HDMI signal with embedded audio to an HDCP-compliant flat panel display. The PURE3 codec's error concealment system preserves a reliable, stable picture even during situations of high bit errors, jitter, or lost packets. Latency is extremely low, with video delivery between the sending and receiving locations taking less than 100 ms.

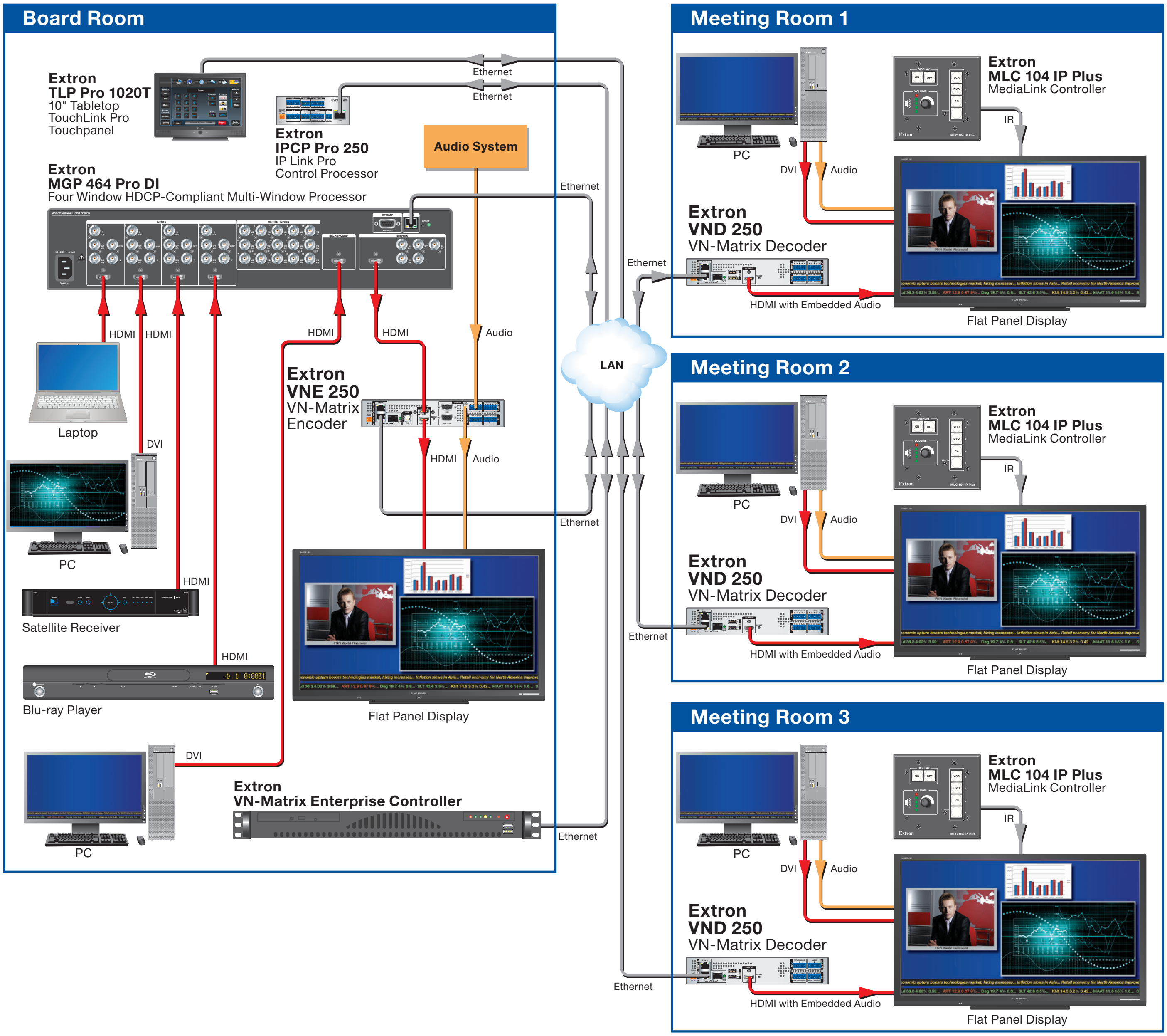
Decoded video is displayed at original source resolution, or scaled down when necessary to fit the display. If a display is not HDCP-compliant, a green screen is presented, aiding troubleshooting.

Control

An Extron **VNM Enterprise Controlle**r manages the VNE 250 encoder as well as the VND 250 decoders for configuration from one simple web interface.

The MGP 464 Pro DI processor and the streaming system are both managed from the board room by an Extron TouchLink® TouchPanel system interfaced with the MGP 464 Pro DI and the VNM Enterprise Controller.

The board room has master control over the streaming products, and can manage access to streamed connections from the meeting rooms. Each meeting room is equipped with an Extron **MLC 104 IP Plus** controller for selection of the display's input source and adjustment of volume.



Board Room
Extron
TLP Pro 1020T
10" Tabletop
TouchLink Pro
Touchpanel
Ethernet
Ethernet
Extron
IPCP Pro 250
IP Link Pro
Control Processor
Audio System
Extron
MGP 464 Pro DI
Four Window HDCP-Compliant Multi-Window Processor
Ethernet
HDMI
HDMI
HDMI
HDMI
Audio
Laptop
Extron
VNE 250
VN-Matrix
Encoder
DVI
HDMI
Audio
PC
Ethernet
HDMI
Satellite Receiver
HDMI
Blu-ray Player
Flat Panel Display
DVI
PC
Extron
VN-Matrix Enterprise Controller
Ethernet
LAN
Meeting Room 1
PC
Extron
MLC 104 IP Plus
MediaLink Controller
IR
DVI
Audio
Extron
VND 250
VN-Matrix Decoder
Ethernet
HDMI with Embedded Audio
Flat Panel Display
Meeting Room 2
PC
Extron
MLC 104 IP Plus
MediaLink Controller
IR
DVI
Audio
Extron
VND 250
VN-Matrix Decoder
Ethernet
HDMI with Embedded Audio
Flat Panel Display
Meeting Room 3
PC
Extron
MLC 104 IP Plus
MediaLink Controller
IR
DVI
Audio
Extron
VND 250
VN-Matrix Decoder
Ethernet
HDMI with Embedded Audio
Flat Panel Display

Overview

A major U.S. defense contractor manages large aeronautical, defense, space, and IT programs for the United States Federal Government Agencies and the Department of Defense. This application brings together situation awareness and operational imagery from real-time collaborative experiments, training experiments, and operations from multiple sites across the United States. Critical to this application is the ability to collaborate, viewing high definition imagery with identical quality simultaneously at locations that are thousands of miles apart.

Room Needs Assessment

Staffing	Specially trained staff manages command and control, simulation, and AV equipment at each location.
Source Inputs	Each site is equipped with computers and image generating equipment that create sophisticated visualization and simulation imagery. Inputs include high definition video, multi-graphic windowing processors, that may operate using analog or digital signal formats using a variety of resolutions such as 1024x768, 1280x720, 1280x1024, 1920x1080 or 1920x1200.
Displays Near End	A variety of flat panel, large screen projection, and videowall display systems may be used at each site.
Geography	The operating sites are situated at various locations across the country.
Control System	AV control touch panels will be used at each location to provide straightforward and simple user control over the system.
Network	The defense contractor's IT department manages network traffic, security, encryption, and usage policies over the enterprise WAN.
Displays Far End	The flat panel, projection, and videowall display systems vary by site, and may or may not offer resolutions that are identical to the original sources.
Documentation	Audio and video/graphic imagery presented on multiple screens will need to be recorded, preserving synchronization from screen to screen, and maintaining the original resolution, detail, and quality.

System Design Solution

Streaming Encoders

Extron **VNE 250** encoders and **VND 250** decoders employing the **PURE3**® codec, stream high resolution situation awareness and operational imagery between collaboration sites located throughout the United States. The PURE3 codec offers low-latency, visually lossless compression at very efficient bit rates, with a high immunity to network errors. A variety of compression and bit rate management tools provide control over streaming bandwidth. Many video/graphic signals are limited to 1-5 Mbps, while some high-motion video or simulator imagery may require 15-25 Mbps for optimal viewing.

Recording

Extron **VNR 100** recorders record and playback imagery captured from live streams sent across the Enterprise WAN. Multiple recorders can playback synchronized streams, allowing multi-source presentations to be recorded and played back in a fashion that reproduces the original viewing experience.

Network

Local Area Network switches with Layer 3 switching and routing capabilities interface with VNE 250 encoders, VND 250 decoders, and VNR 100 recorders. Network encryption is applied to IP traffic that exits secure facilities over the Enterprise WAN.

Streaming Decoders

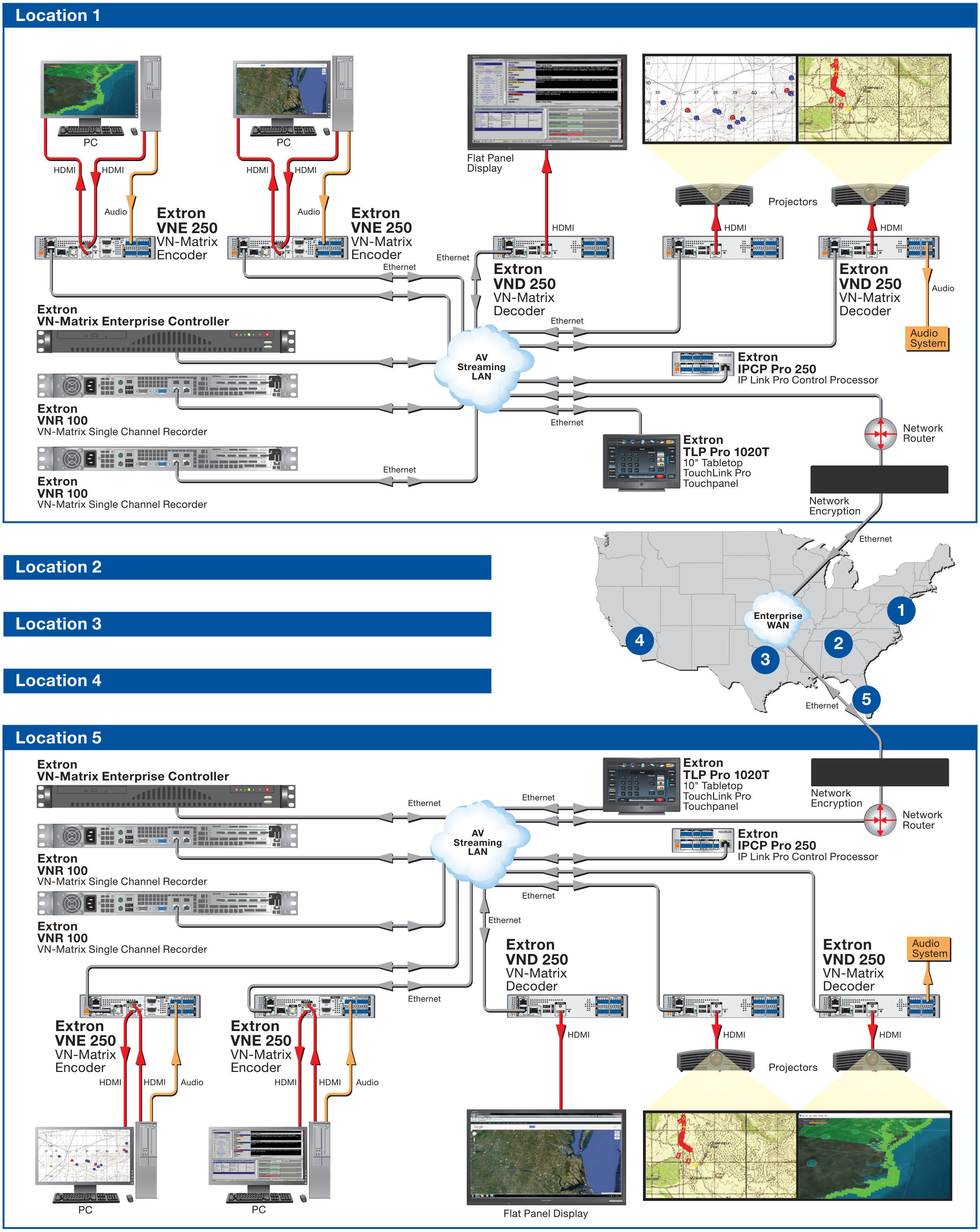
VND 250 units decode live or recorded content. Visually lossless quality is preserved. Images can be decoded at native resolution or scaled to match the display.

Error concealment ensures that stable, artifact free images are presented if network errors are experienced.

Extron VND 250 units rapidly decode live or recorded content in just 35 ms, either at native resolution or scale video to match the resolution of connected displays. PURE3 error concealment delivers reliable, artifact-free video even over unstable networks with bit errors, jitter, and lost packets.

Control

VNE 250 encoders, VND 250 decoders, and VNR 100 recorders are configured and monitored using **VNM Enterprise Controllers**. VNM Enterprise Controllers can operate together as one large system or independently in any local site. Preset configurations defining streaming system connections can be saved and quickly recalled from Extron **IPCP Pro 250** control processors and **TLP Pro 1020T** touchpanels. When pre-defined system streaming presets do not provide the desired streaming connections, manual connections can be established between encoders and decoders.



Location 1
PC
HDMI
HDMI
Audio
Extron
VNE 250
VN-Matrix
Encoder
PC
HDMI
HDMI
Audio
Extron
VNE 250
VN-Matrix
Encoder
Ethernet
Flat Panel
Display
HDMI
Ethernet
Projectors
HDMI
HDMI
Extron
VND 250
VN-Matrix
Decoder
Extron
VND 250
VN-Matrix
Decoder
Audio
Audio
System
Ethernet
Extron
VN-Matrix Enterprise Controller
AV
Streaming
LAN
Extron
IPCP Pro 250
IP Link Pro Control Processor
Extron
VNR 100
VN-Matrix Single Channel Recorder
Ethernet
Network
Router
Extron
TLP Pro 1020T
10" Tabletop
TouchLink Pro
Touchpanel
Extron
VNR 100
VN-Matrix Single Channel Recorder
Ethernet
Network
Encryption
Ethernet
Location 2
Location 3
Location 4
Enterprise
WAN
1
2
3
4
5
Ethernet
Location 5
Extron
VN-Matrix Enterprise Controller
Extron
TLP Pro 1020T
10" Tabletop
TouchLink Pro
Touchpanel
Network
Encryption
Ethernet
Ethernet
Network
Router
AV
Streaming
LAN
Extron
IPCP Pro 250
IP Link Pro Control Processor
Extron
VNR 100
VN-Matrix Single Channel Recorder
Ethernet
Extron
VNR 100
VN-Matrix Single Channel Recorder
Ethernet
Extron
VND 250
VN-Matrix
Decoder
Extron
VND 250
VN-Matrix
Decoder
Audio
System
Ethernet
Extron
VNE 250
VN-Matrix
Encoder
HDMI
HDMI
Audio
Extron
VNE 250
VN-Matrix
Encoder
HDMI
HDMI
Audio
HDMI
HDMI
HDMI
Projectors
PC
PC
Flat Panel Display

Overview

Surveillance cameras and video management systems combine to present imagery in control rooms and security centers, like this one on a university campus. While this imagery may be valuable to use in other locations, it is impractical to transport the many sources to other sites. AV streaming connects visual data produced in these environments to other locations across standard, routable IP networks.

Room Needs Assessment	
Staffing	University IT staff manage network and AV equipment across the entire campus.
Source Inputs	Surveillance cameras, facility maps, and infrastructure visualizations are routed through the Video Management System Workstation and presented on displays in the campus security center.
System Reach	Visual data may be streamed hundreds of yards or a few miles using the campus network.
Network	The campus network is intended to support a variety of data, voice, and video applications. Sections of the network can be segmented or provisioned to support video streaming traffic between defined endpoints if required.
Streaming Quality	Imagery built up from multiple video cameras, computer maps, and visualizations create high resolution, detailed images. Detail must to be preserved, delivering a pixel-for-pixel reproduction for critical analysis at the receiving location.
Functional Requirements	Extend high quality visual data from selected screens in the Campus Security Center to the Situation Room located at the Executive Campus offices several blocks away. Low delay streaming will be valuable to ensure Executors can see the same content as the security center.

System Design Solution

Source Input

A Video Management System Workstation with multiple 1920x1080 outputs displays a combination of security camera feeds, maps, and infrastructure data. Some of these signals feed the large, flat panel displays directly, while others pass through streaming encoders for distribution across campus before connecting to the displays.

Streaming Encoder

In the Campus Security Center, two Extron **VNE 250** HDCP-compliant encoders employing the **PURE3®** codec each receive a 1920x1080 HDMI signal from the Video Management System Workstation. Loop through outputs from both VNE 250 encoders feed two flat panel displays.

The VNE 250 encoders' input signals are rapidly encoded in just 35 ms at the Video Management System's original resolution of 1920x1080 per output, with 24 bit, 4:4:4 color sampling. Preservation of color information and streaming at native resolution ensures that imagery from video cameras and fine detail presented in maps and data screens will be preserved.

Network

Local Area Network switches transport the streaming traffic from the VN-Matrix encoder to decoders. A VLAN can be configured if there is a desire to segment the streaming video from the voice and data traffic. Streaming bit rates may range from 20 to 50 Mbps or higher to preserve the highest image quality presented in the campus security center to present in the situation room. A variety of bit rate controls are available to fine tune to bandwidth and fit a desired streaming bit rate.

Streaming Decoder

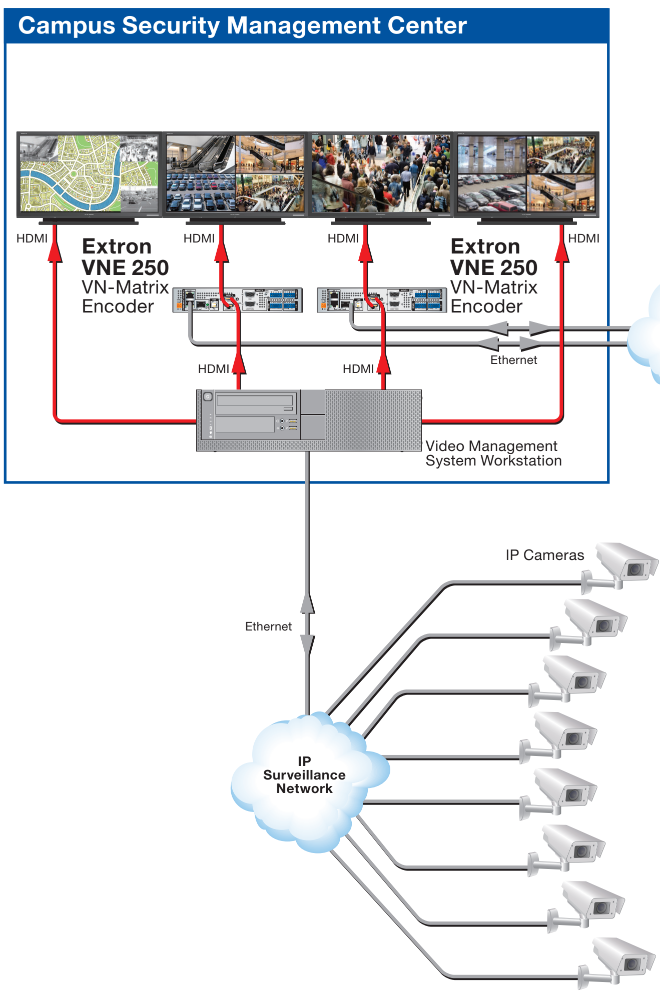
Two Extron **VND 250** decoders located in the situation room at the Executive Campus offices decode the video/graphic stream in 35 ms and outputs the decoded streams to two flat panel displays.

The PURE3 codec's error concealment system preserves a reliable, stable picture, even during situations of high bit errors, jitter, or lost packets. Latency is extremely low, with video delivery between the Campus Security Center and the Situation Room taking less than 100 ms, ensuring that the two locations are simultaneously viewing the same content during telephone discussions.

Decoded video is displayed at original source resolution, or scaled to match a display if necessary.

Control

The university's IT staff manages the system with one of the two VNE 250 encoders, which is configured to serve as the system controller. The two encoders are set in a fixed configuration to stream continuously without user input. Executive staff can select different sources for viewing at any time, seeing a real-time view of surveillance video from across the campus.


Campus Security Management Center
HDMI
Extron
VNE 250
VN-Matrix
Encoder
HDMI
HDMI
Extron
VNE 250
VN-Matrix
Encoder
HDMI
HDMI
HDMI
Ethernet
Video Management
System Workstation
Campus
Network
Campus Executive Situation Room
HDMI
HDMI
Extron
VND 250
VN-Matrix
Decoder
Extron
VND 250
VN-Matrix
Decoder
Ethernet
IP Cameras
Ethernet
IP
Surveillance
Network

Low Delay Streaming in a Gross Anatomy Lab

Overview

Audio visual technology can aid the effectiveness of instructional laboratory environments which often include workbenches and equipment that prevent students from clearly observing subject matter at the instructor's working station.

In this design, a Gross Anatomy Lab utilizes the existing network to stream high definition video to student stations that allows students to see a clear visual reference of the instructor's working subject. Low latency streaming of high definition video to personal student displays provides a visual reference of instructor's gestures pointing at working subjects identifying detail while ensuring the timing of the video aligns with the audio which is heard overhead or from a reinforcement system.

Room Needs Assessment

Source Inputs	High definition video will be collected from an overhead camera at either 720p or 1080p resolution.
Physical Location	An instructor points and gestures at specific physical areas of the subject being studied. Each student has their own subject to examine at their own personal working station. Individuals are dispersed across the laboratory.
Audio	The instructor's voice will be heard directly by students or through an overhead reinforcement system. Video streamed to working stations must have extremely low latency in order for the gestures of the instructor to be clearly correlated to audible instructions.
User Control	The instructor and students access presentations, online data, and real-time, streamed video from flat panel displays connected to desktop PCs located at each working station.
Network	The laboratory network that is used by the working station PCs will deliver real-time streaming video from the instructor's working station to the PCs used at the student working stations.
Displays Far - End	Large flat panels are mounted above student stations for easy viewing of network accessible reference data or streaming video.
Unique Requirements	All PCs receive streamed video and other data via the IP network; there are no dedicated AV cable runs to student stations.

System Design Solution

Streaming Encoder

A **VNE 250** encoder accepts 1080p resolution high definition video as an HDMI signal from an overhead camera. Compression and bit rate management is optimized to present natural motion by software decoding. Low latency encoding ensures that video is streamed quickly to the student stations.

Network

All streamed video destinations reside within the Gross Anatomy Lab and are served from a common network switch. This managed network environment allows IGMP multicast streaming to be used and has sufficient bandwidth to support bit rates in the 25 Mbps range. All workstation PCs have sufficient processing to decode VN-Matrix streams with full quality.

Video Decoding

Streamed video is decoded at student workstation PCs using **VNM Software Decoder**, which decodes with a very low latency. Total latency, from video capture to presentation on displays, is less than 100 ms, ensuring that the instructor's gestures correspond with audio heard directly in the laboratory.

Audio

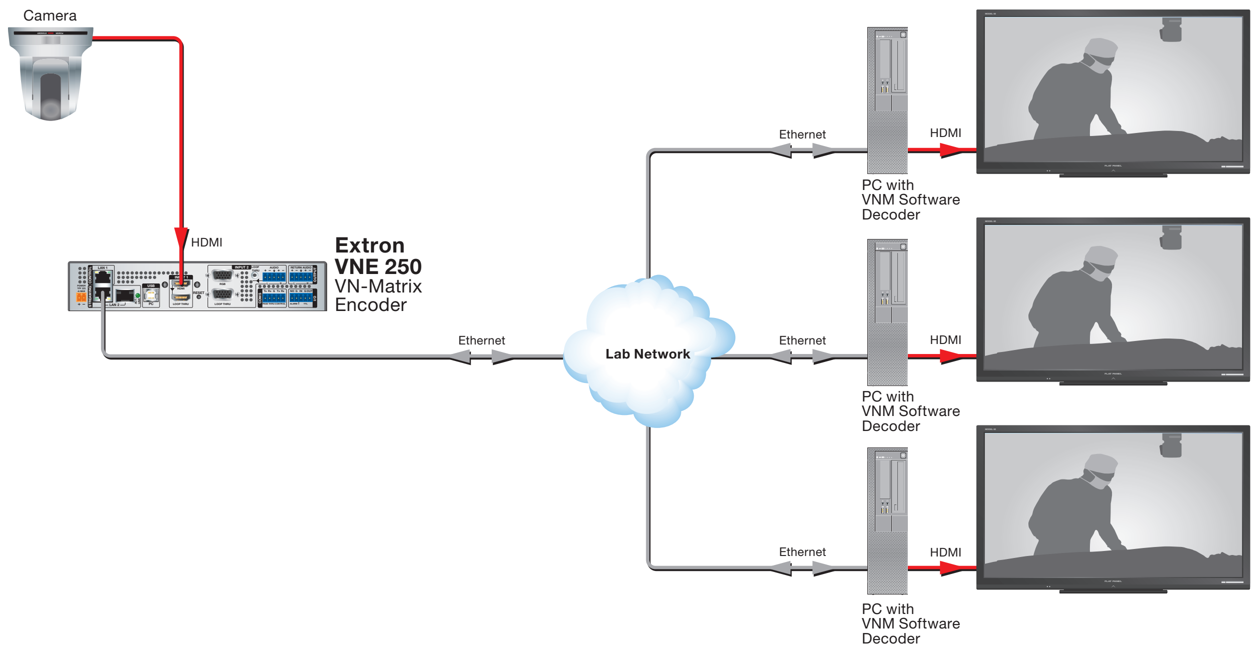
The instructor may be heard directly or from an overhead reinforcement system.

Control

To view the live video stream, students select the VNM Software Decoder desktop icon with their workstation PC mouse. They can size the application to fill the entire screen or only part of the desktop.

Instructor Station

Students Stations



Overview

Simulation systems provide virtual land, sea, and air joint mission environments for the training of military personnel.

Simulated missions train participants, while reducing or eliminating the personnel, fuel use, and risk associated with use of actual military vehicles. AV recording and streaming systems provide a cost effective alternative to reusing simulator equipment to replay missions.

Room Needs Assessment	
Staffing	Specially trained staff manages image generators along with IT and AV equipment used by pilots at each simulation station, which are referred to as “Ships.” A presentation manager directs each mission and manages recording and playback of the sessions.
Source Inputs	Each Ship contains four medium resolution image generators, each producing an instrumentation panel composited into a single high-resolution signal by an MGP 464 Pro DI, and an additional high-resolution image generator produces the heads-up-display - HUD. A single high-resolution image generator creates a visual mission overview for the entire session, and generates distributed interactive simulation – DIS data that is available to other simulation devices on the network.
System Reach	An archiving system is located in the same facility as the simulation lab and after action review – AAR room. The organization’s WAN supports collaborative after-action-review sessions at remote sites.
Network	A segmented network is used for the data, voice and higher bandwidth AV traffic that is used by the simulation system and AV streaming system.
Streaming Quality	This application requires the best-possible image quality, as precise detail is critical for accurate AAR analysis. Single-line, color detail must be preserved to effectively analyze pilot decisions and instrument readings.
Functional Requirements	The archiving system must be able to capture and playback content, maintaining source resolution supporting simultaneous playback and recording. The system must also support synchronization of all streams, and variable speed replay with single frame advance and reverse.

System Design Solution

Streaming Encoders

The simulation station incorporates two Extron **VNE 250** HDCP-compliant streaming encoders. One VNE 250 unit encodes the 1920x1200 HDMI output of the **MGP 464 Pro DI**, providing a composited view of each simulation station’s instrument panels, and another encodes the 1920x1200 HDMI output of the station’s HUD workstation. One additional VNE 250 encodes the 1920x1200 output of the system’s Mission Computer, and captures the DIS UDP mission data and audio.

The Extron VNE 250 encoder employs the **PURE3®** codec encoding in 35 ms at native resolution, with 24 bit, 4:4:4 color sampling. This high-quality sampling ensures that the single pixel detail of small font, lines, and detail from the video and computer graphic inputs is preserved.

Network

Local Area Network switches with Layer 3 switching and routing capabilities and 100/1000BaseT network connections are interfaced to the VNE 250 encoders and VND 250 decoders, **VNR 100** recorders, and **VNM Enterprise Controller**.

High network bandwidth is supported; ensuring preservation of critical detail from both low bandwidth maps and instrumentation streams, as well as high-bandwidth full motion renderings.

Streaming Decoder

Three Extron **VND 250** HDCP-compliant decoders located in the AAR room decode instrumentation, HUD, and mission control video/graphic streams and the Mission Computer imagery with audio in 35 ms, and output HDMI signals to three high-resolution front projectors and the audio system. Total delivery time between sending and receiving locations is well below 100 ms, and synchronization is maintained between each of the streams. Decoded video is displayed at original source resolution, or is downscaled, if necessary, to fit the display.

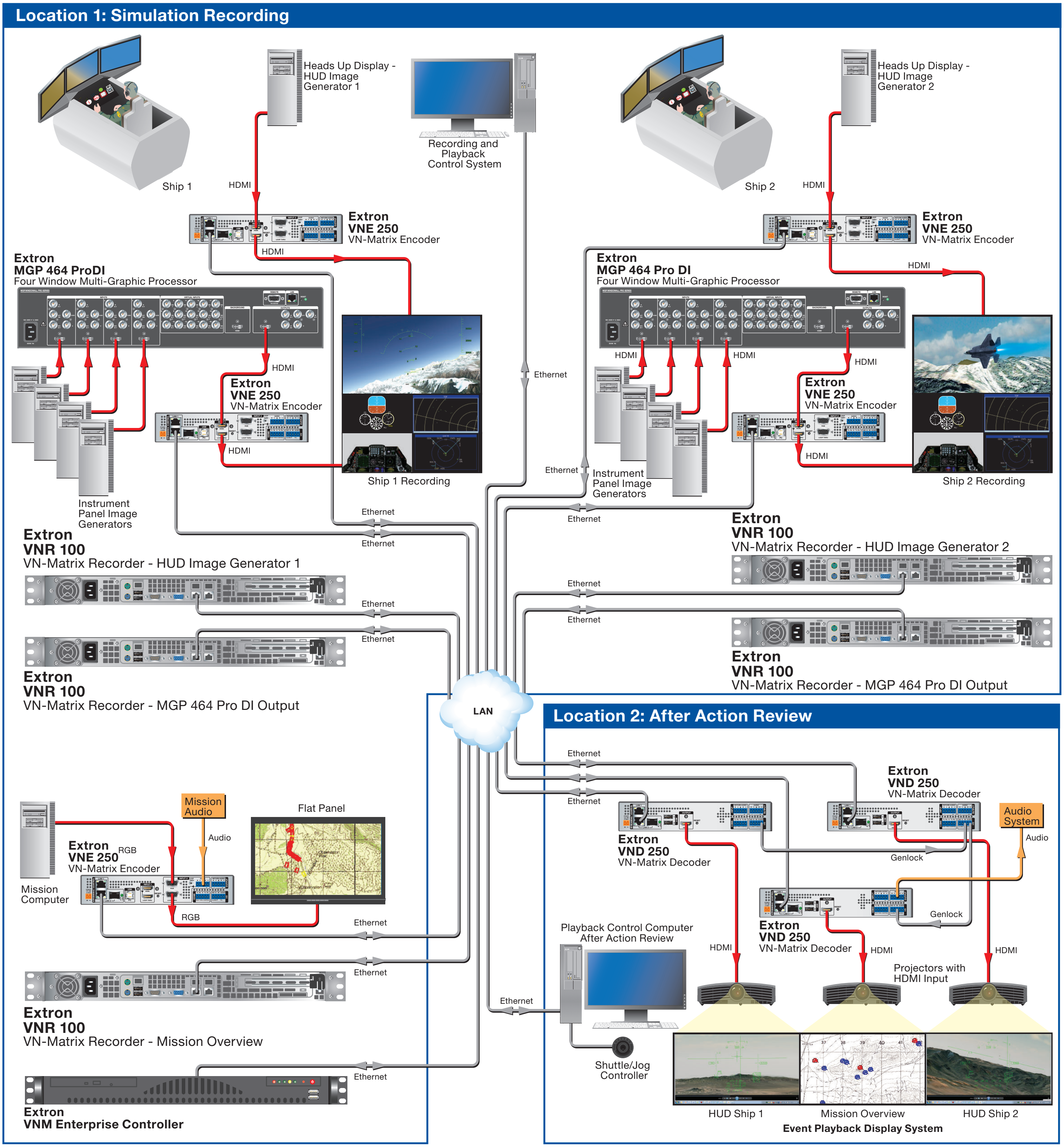
Recorder

Three VNR 100 units record the streams originating from the VNE 250 encoders, DIS metadata and audio each storing approximately 40 hours of video, based on a 50 Mbps bitrate. When played back together, the genlock connection across the VND 250 decoders ensures synchronization across the three video streams, audio, and DIS metadata during playback.

Control System

An Extron VNM Enterprise Controller provides system control over the VNE 250 and VND 250 units as well as the VNR 100 recorders.

The recording and playback control computer accesses the VNM Enterprise Controller’s web interface, giving the presentation manager start and stop control over recording sessions. A playback control computer, located in the AAR room, accesses the VNM Enterprise Controller’s web interface to play back selected missions.


Location 1: Simulation Recording
Ship 1
Heads Up Display - HUD Image Generator 1
HDMI
Recording and Playback Control System
Extron VNE 250 VN-Matrix Encoder
HDMI
Extron MGP 464 ProDI Four Window Multi-Graphic Processor
HDMI
Extron VNE 250 VN-Matrix Encoder
HDMI
Ship 1 Recording
Instrument Panel Image Generators
Ethernet
Extron VNR 100 VN-Matrix Recorder - HUD Image Generator 1
Ethernet
Extron VNR 100 VN-Matrix Recorder - MGP 464 Pro DI Output
Ship 2
Heads Up Display - HUD Image Generator 2
HDMI
Extron VNE 250 VN-Matrix Encoder
HDMI
Extron MGP 464 Pro DI Four Window Multi-Graphic Processor
HDMI
HDMI
HDMI
Extron VNE 250 VN-Matrix Encoder
HDMI
Ship 2 Recording
Instrument Panel Image Generators
Ethernet
Extron VNR 100 VN-Matrix Recorder - HUD Image Generator 2
Extron VNR 100 VN-Matrix Recorder - MGP 464 Pro DI Output
LAN
Mission Computer
Mission Audio
Audio
Extron VNE 250 VN-Matrix Encoder
RGB
Flat Panel
RGB
Ethernet
Extron VNR 100 VN-Matrix Recorder - Mission Overview
Extron VNM Enterprise Controller
Location 2: After Action Review
Ethernet
Extron VND 250 VN-Matrix Decoder
Extron VND 250 VN-Matrix Decoder
Audio System
Audio
Genlock
Extron VND 250 VN-Matrix Decoder
Genlock
Playback Control Computer After Action Review
HDMI
HDMI
HDMI
Projectors with HDMI Input
Shuttle/Jog Controller
HUD Ship 1
Mission Overview
HUD Ship 2
Event Playback Display System

Overview

Real-time streaming can centralize the production of high definition video collected from dispersed locations. In this application, a national soccer league streams high definition post-match interviews to a central production studio. This ambitious project remotely manages several small, high definition production systems in stadiums across the country from one central production studio.

Room Needs Assessment

Staffing	On-air television talent interviews coaching staff, athletes, and analysts at regional soccer stadiums. An effort is made to minimize technical staffing at stadiums.
Source Inputs	An HDMI signal supplies an image that presents multiple camera inputs to the production equipment along with audio levels and other system data.
Geography	A centralized facility site is manned with production staff that will remotely control production equipment located at sixteen regional soccer stadiums across the country.
Network	A WAN connects the central production site to remote production equipment at the regional stadiums.
Streaming Quality	Video decoded at the national production site must maintain original source detail and motion. Real time production control requires low latency for natural keyboard and mouse movement.
Functional Requirements	A video production engineer at the central control site must be able to see the detail in the multiple source image produced by the real-time production systems at the original source resolution and with very low latency. Excessive latency will interfere with tactile control of the man-machine interface.

System Design Solution

Source Input

Remote site editing equipment produces an 1920x1200 resolution HDMI signal, which includes a combination of video sources and graphical system control data.

Streaming Encoder

VNE 250 encoders encode the HDMI signal produced by real-time production equipment at studios in distant stadiums and stream the video/graphic image information to the central production studio. Visually lossless encoding is made possible by the **PURE3**® codec, which preserves both the fine detail of computer graphic data and video motion.

Network

A Wide Area Network supports streaming bandwidths from 15 to 100 Mbps as necessary between the remote studios and the central production site.

Streaming Decoder

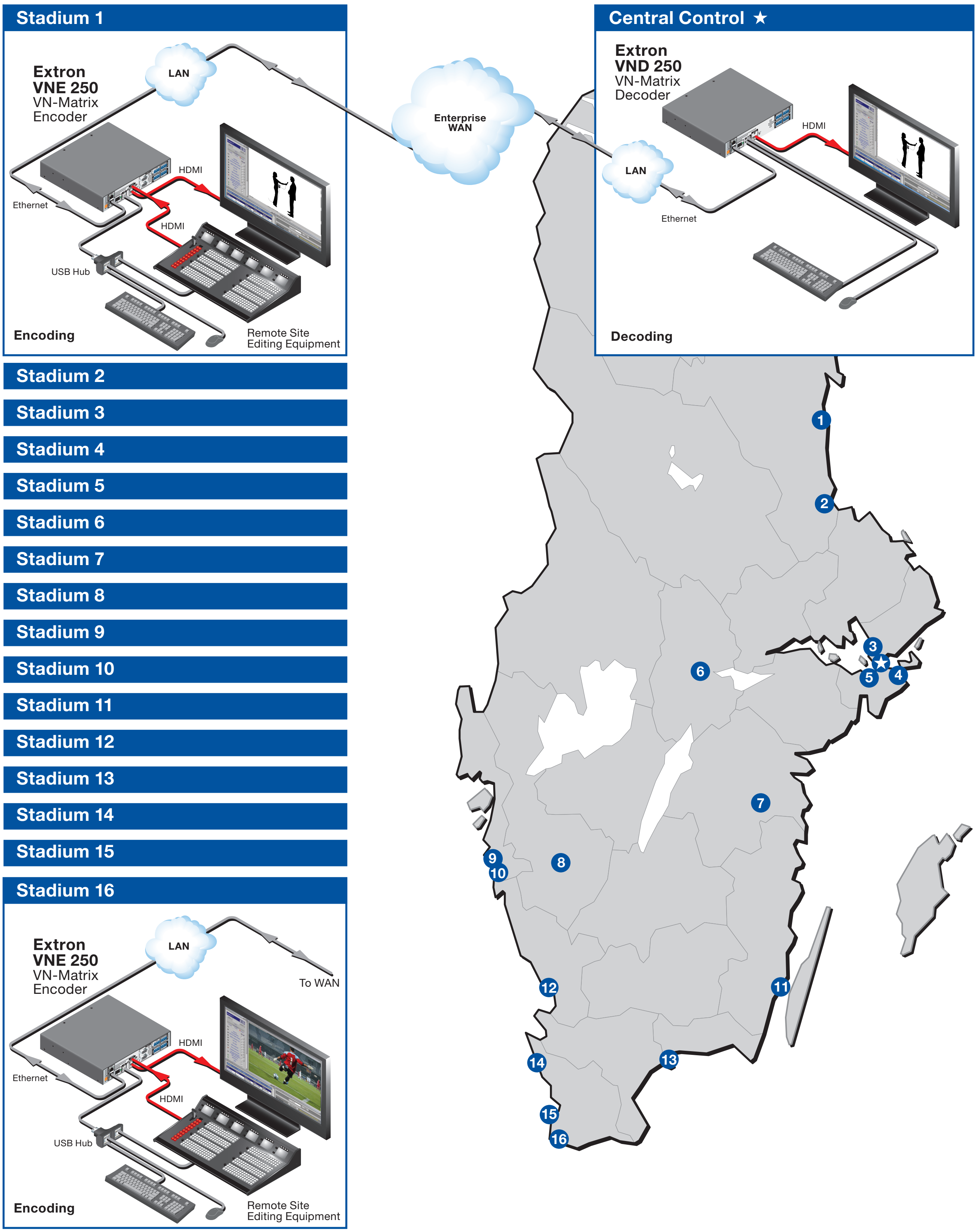
A **VND 250** decoder at the central production studio decodes video/graphic image information streamed from one of the live interviews at the soccer stadiums. Video is delivered with very low latency, under 100 ms, giving local engineers natural control of mixing and effects at remote sites. Low latency is critical to preserve tight tactile control in this long distance remote control application. PURE3 error concealment ensures a reliable picture even under conditions of heavy packet loss.

Far End Display

A VND 250 unit decodes the HDMI signal streamed in real-time from the remote stadium. An engineer views the multi-viewer production display on a large 1920x1200 resolution flat panel display at the central production studio.

Control

USB keyboard and mouse connections are made from remote production equipment to VNE 250 encoders and a USB keyboard and mouse are connected to the VND 250 decoder. This USB connection from decoder to encoder allows a keyboard and mouse at the central production studio to manage the real time production application interface at remote stadiums.


Stadium 1
Extron
VNE 250
VN-Matrix
Encoder
LAN
HDMI
Ethernet
HDMI
USB Hub
Encoding
Remote Site
Editing Equipment
Enterprise
WAN
Central Control ★
Extron
VND 250
VN-Matrix
Decoder
HDMI
LAN
Ethernet
Decoding
Stadium 2
Stadium 3
Stadium 4
Stadium 5
Stadium 6
Stadium 7
Stadium 8
Stadium 9
Stadium 10
Stadium 11
Stadium 12
Stadium 13
Stadium 14
Stadium 15
Stadium 16
Extron
VNE 250
VN-Matrix
Encoder
LAN
To WAN
HDMI
Ethernet
HDMI
USB Hub
Encoding
Remote Site
Editing Equipment
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16

Overview

IP networks can frequently be used as a cost-effective and scalable switching solution for large systems. In this scenario, a large defense organization utilizes the IP network infrastructure to route multiple sources on demand to various viewing locations.

Room Needs Assessment

Staffing	Trained technical staff configure and manage the IT and AV equipment. Non-technical meeting participants select sources and windowing configurations using TLP Pro 1020T touchpanels.
Source Inputs	Sources include a high-definition PTZ video camera and a mixture of modern and legacy computers that output digital and analog RGB signals at various resolutions.
System Reach	Imagery is streamed to destinations hundreds of yards away to a few miles away over the facility's local area network. Other destinations can be reached across a wide area network.
Network	The facility's LAN supports a variety of data, voice, and video applications. The network is segmented and provisioned to support video streaming traffic between defined endpoints.
Streaming Quality	The video camera and computer visualizations create high resolution, detailed imagery, which must be preserved at native resolution for critical analysis at receiving locations.
Functional Requirements	High-definition video and computer video signals are extended to viewing rooms equipped with two flat panel displays. One display presents a single source, while the larger display presents either one source or four sources simultaneously. Touchpanel systems provide simple control over source selection. The system features rapid, clean source switching even when selecting between sources with different resolutions.

System Design Solution

Source Input

High-definition video security cameras and computers of varying resolutions from 640x480 up to 1920x1200 generate maps, radar and strategic data for analysis.

Streaming Encoder

Extron **VNE 250** HDCP-compliant encoders employing the **PURE3®** codec are located throughout the facility. One VNE 250 encodes a digital high-definition camera feed, and three VNE 250 units encode a combination of analog and digital video graphic signals from workstation PCs. Loop outputs from workstation encoders feed local desktop monitors.

Systems with hundreds of encoders and decoders can also be integrated into large networked switching systems.

Input signals are rapidly encoded in 35 ms at source resolution, with 24 bit, 4:4:4 color sampling. This high-quality sampling ensures that the single pixel detail of small font, lines, and detail from video and computer graphic inputs is preserved.

Video Decoding

Local Area Network switches with Layer 3 switching and routing capabilities and 100/1000BaseT network connections are interfaced to the VNE 250 encoders and VND 250 decoders. The network switches are segmented with dedicated VLANs to support the bandwidth requirements of the high definition video and graphic streams between the encoders and viewing locations.

Streaming Decoder

One Extron **VND 250** decoder in each viewing room decodes a video/graphic stream with a rapid 35 ms decode process, and presents the high resolution signal upon a flat panel display.

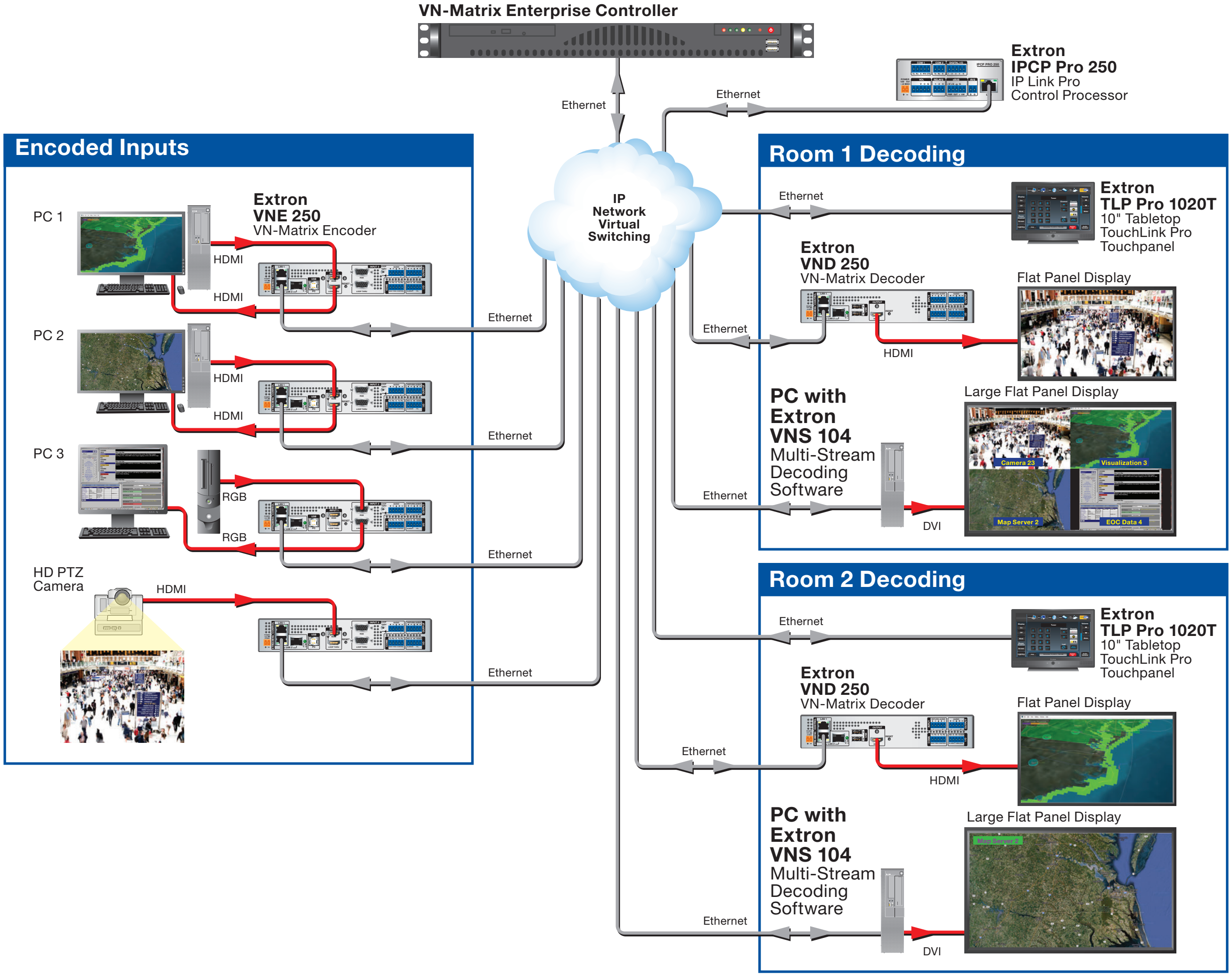
Another video/graphic stream is decoded by a PC running Extron **VNS 104** multi-stream decoding software, which decodes either one stream or four streams simultaneously, for the monitoring of images at slower frame rates on a single, large flat panel display.

Total video delivery between the source encoders and the viewing room decoders takes less than 100 ms, ensuring the simultaneous presentation of imagery across all viewing rooms, which is essential to critical analysis and decision making. Decoded high resolution imagery is presented at the native source resolution or scaled to meet viewing requirements.

Control

An Extron **VNM Enterprise Controller** provides access to VNE 250 encoder and VND 250 decoder units for configuration and management of the entire streaming system from one simple web interface.

The VNM Enterprise Controller interfaced with Extron **TLP Pro 1020T** TouchLink® Pro touchpanels provide control over source selection at each location.



VN-Matrix Enterprise Controller
Extron
IPCP Pro 250
IP Link Pro
Control Processor
Ethernet
Ethernet
IP
Network
Virtual
Switching
Encoded Inputs
PC 1
Extron
VNE 250
VN-Matrix Encoder
HDMI
HDMI
Ethernet
PC 2
HDMI
HDMI
Ethernet
PC 3
RGB
RGB
Ethernet
HD PTZ
Camera
HDMI
Ethernet
Room 1 Decoding
Ethernet
Extron
TLP Pro 1020T
10" Tabletop
TouchLink Pro
Touchpanel
Extron
VND 250
VN-Matrix Decoder
Flat Panel Display
Ethernet
HDMI
PC with
Extron
VNS 104
Multi-Stream
Decoding
Software
Large Flat Panel Display
Camera 23
Visualization 3
Map Server 2
EOC Data 4
Ethernet
DVI
Room 2 Decoding
Ethernet
Extron
TLP Pro 1020T
10" Tabletop
TouchLink Pro
Touchpanel
Extron
VND 250
VN-Matrix Decoder
Flat Panel Display
Ethernet
HDMI
PC with
Extron
VNS 104
Multi-Stream
Decoding
Software
Large Flat Panel Display
Ethernet
DVI

Overview

Organizations offering highly skilled production and animation services often have facilities located great distances from each other. Many times, customers are often not located in the same region where the creative work is prepared. This streaming solution allows the account management and customer to review creative work that has been prepared at locations a great distance away. Work is carried out collaboratively in real time; the customer and production company complete work quicker and more efficiently than before.

Solution Needs Assessment

Work Flow	Creative staff in the production facility have prepared the video or animation material and are ready to play back the material for review. Account staff at the presentation suite manage the customer discussion and review workflow using audio conference equipment.
Source Inputs	Video or animation production equipment located in the studio output HD-SDI video with embedded audio.
Geography	The production staff and customer review facility may be located at opposite ends of a state or, continent or across the globe.
Network	Network switches supporting Layer 3 switching in the local area networks and an enterprise WAN will connect the two facilities. Sustained bandwidth of 100 Mbps is required through the full connection path to support streaming of the video at full fidelity.
Control System	The encoders and decoders must be configured to operate within defined bit rates when in use.
Functional Requirements	A very low delay must be maintained so that as individuals are discussing and working at each location, they are both referring to identical material. IP networks do not guarantee 100% packet delivery, so the streaming solution must maintain a stable picture, with reliable picture quality, if packet loss is experienced.

System Design Solution

Source Input

Video production equipment is used to prepare and play back high definition video content with embedded audio, which is output and presented on accurate-color HD-SDI flat panel displays. The displays are capable of presenting 10-bit video resolution and 4:2:2 color. The monitors are frequently color corrected to ensure the truest color is presented.

Streaming Video Encoders

Extron **VN-Matrix**® **325** codecs employing the **PURE3**® codec interface the video production equipment. HD-SDI with embedded audio is encoded with low, 35 ms delay and the encoder preserves the 10-bit, 4:2:2 video quality contained in the serial digital video signal, critical to preserving the image quality that will be delivered to the far location. The VN-Matrix 325 codec is interfaced to a local area network to deliver the audio/video streams. A variety of compression and bit rate controls exist to allow delivery of the best picture given the available network bandwidth.

Network

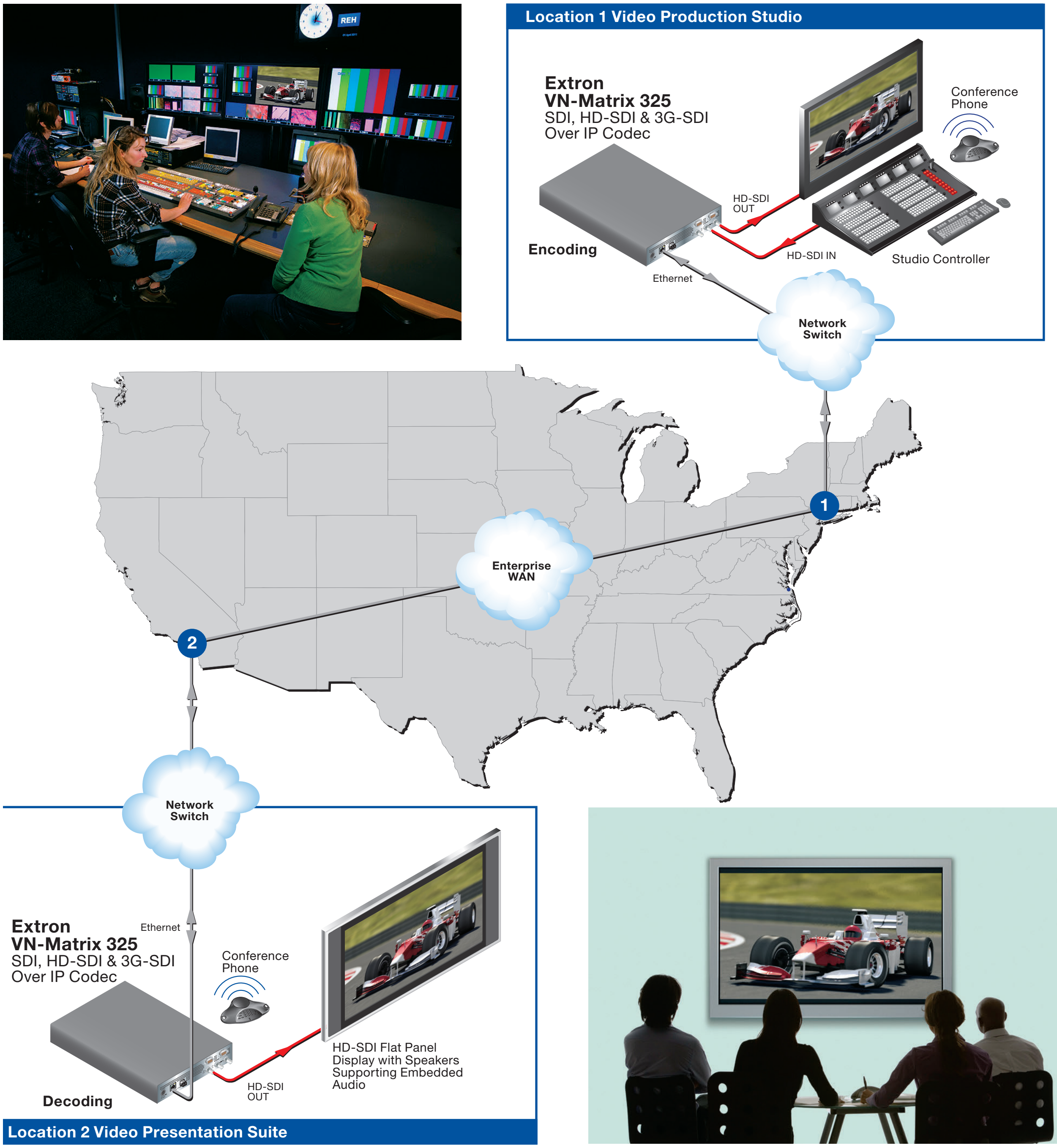
Professional, local area network switches with Layer 3 switching and routing capabilities and 1000BaseT network connections are interfaced to the VN-Matrix codec. A firewall exists at each location and an enterprise WAN ensures that the HD video can be delivered with a sustained throughput of 100 Mbps. Network bandwidth for HD video may range from 50 to 90 Mbps and a block of 4 audio channels will require 16 Mbps. SD video may require 15 to 20 Mbps and a block of four uncompressed audio signals will require 8 Mbps.

Streaming Video Decoders

Extron VN-Matrix 325 codecs decode the audio and video signals rapidly with a 35 ms decode process. Audio and video are synchronized and the low delay of the total encode to decode path ensures that when individuals at both sites discuss the material, they are both referring to the same piece of content. An error concealment system in the PURE3 codec preserves a reliable, stable picture even when bit errors, jitter, or lost packets are experienced on the network. Visually lossless image compression by the VN-Matrix 325 codecs preserve the 10-bit depth and 4:2:2 color information. The post production color grading application in particular is very sensitive to accurate color reproduction of the original signal.

Displays

HD-SDI flat panels are positioned at the far end Video Production review suite. The monitors are capable of presenting a full 10-bit color depth and are color corrected to ensure accurate reproduction of the decoded video signal.



Location 1 Video Production Studio
Extron
VN-Matrix 325
SDI, HD-SDI & 3G-SDI
Over IP Codec
Conference
Phone
HD-SDI
OUT
Encoding
HD-SDI IN
Studio Controller
Ethernet
Network
Switch
1
Enterprise
WAN
2
Network
Switch
Extron
VN-Matrix 325
SDI, HD-SDI & 3G-SDI
Over IP Codec
Ethernet
Conference
Phone
HD-SDI Flat Panel
Display with Speakers
Supporting Embedded
Audio
HD-SDI
OUT
Decoding
Location 2 Video Presentation Suite

Extron AV Streaming Products

Extron offers two classes of products for use in AV systems. Extron H.264 products include the **SMP 300 Series** Streaming Media Processors, **SME 100** Streaming Media Encoder, and the **SMD 101** and **SMD 202** Streaming Media Decoders; ideal products for integrating streaming into professional AV systems. They support the wide range of signal formats and resolutions common to AV sources and displays, providing advanced signal processing such as scaling and aspect ratio management. Extron H.264 products offer signal processing and control options that make them ideal for monitoring, overflow and signage applications. Extron **VN-Matrix®** products excel in applications with demanding quality requirements, streaming high resolution signals with visually lossless quality and low latency. Both Extron streaming product classes include signal processing and control capabilities that provide superior quality and simplify AV integration. They are applied in simple point-to-point configurations or systems that require high scalability streaming of AV signals to many endpoints over LANs or WANs.

H.264 AV Streaming

The **SMP 352** is a high performance streaming and recording processor for capturing and distributing AV sources and presentations as live streaming or recorded media. It can create independent recordings from two different sources simultaneously. The SMP 352 accepts HDMI, component, composite, and optional 3G-SDI signals and applies two-window processing to selected sources. It can record and stream simultaneously and can stream at two different resolutions and bit rates concurrently using a range of transport protocols and session management options. With no recurring licensing fees and comprehensive control and configuration features, the SMP 352 is a cost-effective, integration-friendly solution for delivering presentations to a larger audience.

H.264
MPEG-4 AVC

The **SMD 202** is a compact, high performance media player and live stream decoder that can present a locally connected AV signal, decode a live streaming source, or play back media files. The SMD 202 supports a wide range of media file container formats and streaming protocols, making it adaptable for use with a variety of encoded media. Advanced signal processing, scaling, and aspect ratio management supply high quality signals to AV displays. The SMD 202 can be controlled using Ethernet, RS-232, IR, or wired IR.

PURE3 High Resolution Video & Graphic Streaming

The PURE3 codec applied in **VN-Matrix** provides visually lossless quality streaming and recording high resolution video signals over IP networks with low delay and a high immunity to network errors. This makes VN-Matrix streaming products ideal for use in high quality, mission critical applications. Two series of products are available for streaming HDMI, RGB and audio signals, or 3G-SDI, HD-SDI or SDI signals with embedded audio.

PURE 3® CODEC
OPTIMIZED FOR NETWORKS

VN-Matrix 250 Series for HDCP-Compliant HDMI and RGB Streaming

The **VN-Matrix 250** Series encodes and streams video or graphic sources at resolutions up to 1920x1200 or 2048x1080, and decodes content at native source resolution. HDMI embedded stereo audio or analog stereo audio can be streamed as well as RS-232 control and USB keyboard and mouse data. The VN-Matrix 250 Series offers real-time performance and low latency, making it ideal for remote collaborative use. It is popular for use in high level conferencing, signal routing, and switching over IP networks.

VN-Matrix 325 for 3G-SDI, HD-SDI, and SDI Streaming

The **VNC 325 3G-SDI** codec streams 3G-SDI, HD-SDI, or SDI video with embedded audio over IP networks. It provides excellent image quality at highly efficient bit rates with low latency. The VNC 325 3G-SDI codec is ideal for use in applications transporting production quality video and audio in real-time.

SMP 300 Series

H.264 Streaming Media Processors

The SMP 300 Series of products are high performance streaming and recording processors for capturing and distributing AV sources and presentations as live streaming and recorded media. They incorporate Extron's FlexOS®, a flexible platform for automating system operation. Accepting HDMI, component, composite, and optional 3G-SDI signals, SMP 300 Series processors can record and stream simultaneously and can stream at two different resolutions and bit rates concurrently using a range of transport protocols and session management options.

- The **SMP 351** creates a composited two-window stream and recording from its available sources.
- An optional **LinkLicense**® upgrade unlocks SMP 352 functionality within the SMP 351.
- The **SMP 352** can create composited or independent recordings and streams from two different sources with independent settings for each channel, with Advanced Audio DSP and streaming presets.

Comprehensive control and configuration features make SMP 300 Series processors integration-friendly and easy to control and operate. Requiring no recurring licensing fees, these H.264 processors have a low cost of ownership, making them a cost-effective solution for delivering presentations to a larger audience.

SMP 300 Series processors are ideal for use in virtually any professional environment where AV sources can be streamed live or recorded for future reference, especially when combining multiple AV sources will enhance the message. Streaming and recording AV presentations allows an organization to communicate and train employees and students that cannot be present at an event. Event recording provides everyone with the opportunity to review and gain insight into the live experience. SMP 300 Series processors can be adapted to many applications, documenting virtually any meeting, conference, or activity that uses an AV source as a reference. They are ideal for use in corporate, education, government, healthcare, courtroom, house of worship, and rental and staging applications.

FlexOS

FEATURES

- Process two high resolution AV sources from up to five available input signals
- Dual recording and streaming - SMP 352 only
- Stream and record simultaneously
- High quality scaling with flexible two-window management - SMP 351 or SMP 352 in Single Channel mode
- Produces MP4 media files that are compatible with virtually any media player
- Enhanced Audio Digital Signal Processing - SMP 352 only
- Save recordings to internal solid state drive and external USB storage
- Automated transfer of recordings to network storage
- Stream concurrently at multiple resolutions and bit rates
- HDMI, component, composite, and optional 3G-SDI input
- Easy to configure and operate from the front panel or external control system
- Integrates with Entwine Enterprise Media Platform
- Directly compatible with Kaltura Video Hosting
- Compatible with third party Content Management Systems - CMS
- Schedule streaming and recording using Microsoft Exchange, iCalendar and others

MODEL	VERSION	PART#
SMP 351	Standard Version - 80 GB SSD	60-1324-01
SMP 351 3G-SDI	with 3G-SDI Input - 80 GB SSD	60-1324-02
SMP 351	Standard Version - 400 GB SSD	60-1324-11
SMP 351 3G SDI	with 3G-SDI Input - 400 GB SSD	60-1324-12
LinkLicense	SMP 351 80 GB Dual Recording Upgrade	79-2547-01
LinkLicense	SMP 351 w/ 3G-SDI 80 GB Dual Recording Upgrade	79-2547-02
LinkLicense	SMP 351 400 GB Dual Recording Upgrade	79-2547-03
LinkLicense	SMP 351 w/ 3G-SDI 400 GB Dual Recording Upgrade	79-2547-04
SMP 352 - 400 GB SSD	Dual Recording - 400 GB SSD	60-1634-11
SMP 352 3G-SDI - 400 GB SSD	Dual Recording w/3G-SDI - 400 GB SSD	60-1634-12

SME 100

RGB, DVI & Video Over IP H.264 Streaming Media Encoder

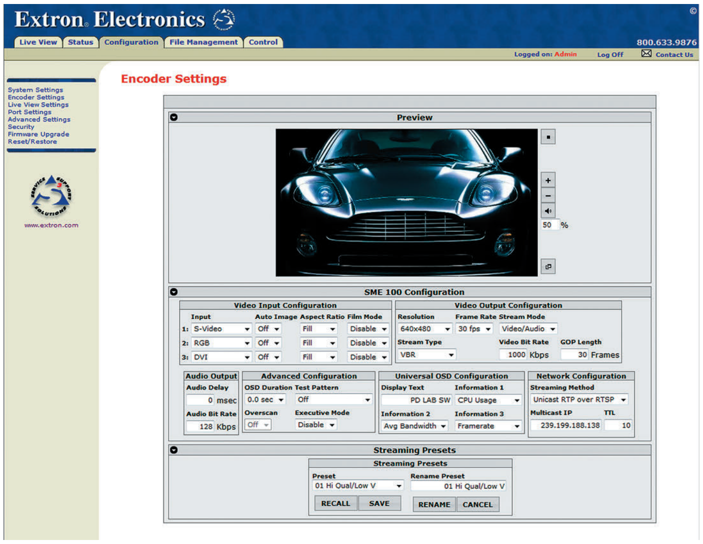
The Extron SME 100 is a live streaming media encoder that interfaces with DVI, RGB, HDTV, and standard definition signals for delivering media over IP networks. The SME 100 can be used with the Extron SMD 101 H.264 decoder to provide complete end-to-end streaming systems. It features an integrated three-input switcher with audio, plus buffered loop-throughs for simplified integration into AV systems. The SME 100 employs standards-based H.264 / MPEG-4 AVC and AAC encoding, and outputs an IP stream that can easily be decoded and viewed on desktop or laptop PCs. High performance Extron signal processing scales and optimizes video input signals for the intended viewing application. Encoding controls also provide adjustments for bit rate and quality. By extending AV signals over networks, the SME 100 significantly expands AV system capability.

FEATURES:

- Streams DVI, RGB, HDTV, and video signals with audio over IP networks
- Use with the SMD 101 H.264 decoder to provide complete end-to-end streaming systems
- Compatible with many third-party H.264 devices including set-top box decoders
- Supports input signals up to 1920x1200, including HDTV 1080p/60 — The SME 100 supports a wide range of input resolutions, from standard definition up to the high resolutions commonly used for computer-video and HDTV.
- Streams at selectable resolutions from 166x120 up to HDTV 720p/30 and 1080p/30
- Inputs: DVI-D with loop-through; universal 15-pin HD input with loop-through for RGB, HD component video, S-video, or composite video; BNCs with loop-throughs for component video, S-video, or composite video
- Outputs: Ethernet for streaming H.264 / MPEG-4 AVC-encoded video
- Standards-based H.264 / MPEG-4 AVC video compression
- Integrated three-input AV switcher
- Buffered input loop-throughs for video and audio
- DVI, RGB, HDTV, and standard definition video upscaling and downscaling
- Auto Input Format Detection
- Auto-Image™ setup
- Push and pull streaming session management
- Pull streaming transport protocols
- Push streaming transport protocols
- Session Announcement Protocol - SAP and Session Description Protocol - SDP
- H.264 compression profiles and level selection
- AAC audio encoding
- Encoding quality controls including video resolution, video bit rate, frame rate, constant or variable bit rate control, GOP length, and audio bit rate
- Encoding presets for quick recall of specific compression settings
- Streaming presets for quick recall of specific system configurations
- Auto Input Memory
- EDID Minder® automatically manages EDID communication between connected devices
- Glitch-free switching
- EIA-608B closed captioning support
- Power Save Mode
- Audio switching
- Audio input gain and attenuation
- Audio breakaway
- User-adjustable audio delay
- Picture controls for brightness, contrast, color, tint, detail, as well as horizontal and vertical positioning, sizing, and zoom
- Advanced deinterlacing
- Aspect ratio control
- Quad standard video decoding
- Internal test patterns for setup
- Front panel security lockout
- Ethernet monitoring and control
- Embedded Web interface with live preview window
- Network traffic prioritization
- RS-232 control port
- Rack-mountable 1U, full rack width metal enclosure
- Internal universal power supply

H.264
MPEG-4 AVC

MODEL	VERSION	PART#
SME 100 HD	H.264 HD Encoder	60-1061-01



SME 100 Embedded Live View Web page allows the user to view video streams while making adjustments to the unit

Control

Extron H.264 encoders and processors can be quickly configured from the front panel, through the USB port, using a Web browser to access embedded HTML pages, or through Ethernet or serial RS-232 ports. Video streams can be viewed live from the Extron embedded HTML interface while making encoder adjustments.

Several user controls are available to adjust AV encoding and streaming transport configurations. These include video resolution, video bit rate, frame rate, bit rate control, and audio bit rate. The GOP length can also be adjusted, allowing the user to specify use of fewer or more frames in the compression process. The bit rate can be configured to be constant or variable. Ultra-low bit rates less than 1 Mbps can be used with a low streaming resolution, or can be set to deliver HD video at a 10 Mbps limit to deliver very high resolution streams. Encoding presets can define different resolutions and bit rates and streaming presets allow unicast or multicast push configurations to be defined using different streaming transport protocols based on network conditions or decoding requirements. The encoding and streaming presets make quick recall of diverse encoder settings quick and easy.

Ideal for use in applications that require:

- The flexibility to decode audiovisual media streams using devices such as hardware decoders, set-top boxes or PCs which can quickly and easily support H.264 decoding.
- The ability to interface and stream many different computer-video, HDTV, or video resolutions or signal formats with audio.
- Streaming on converged networks where use of lower bandwidth is desired – less than 10 Mbps or 1 Mbps where ultra-low bit rates are needed.
- Accessibility to audiovisual information through one-way streaming.

Applications:

- **AV System Monitoring:** Monitor an AV system from a remote location by streaming the video and audio output to any PC in the facility with network access.
- **Meeting/Lecture Overflow:** Quickly add overflow rooms or areas by streaming AV presentations to a PC for output to a display.
- **Corporate or Enterprise Broadcast:** Centralize media sources such as PCs, DVD players, cameras, or satellite receivers into one location, and stream them across an enterprise network to individual rooms as needed. This can simplify AV system design, integration, and programming.
- **Multiple User Access:** Use with a media server to deliver streaming to different types of decoding devices within an enterprise, or stream to Content Delivery Networks for broad distribution to users across the Internet.

Presentation or
Videoconference Overflow
Page 46

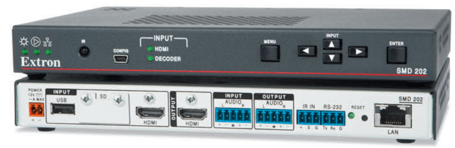
AV Presentation
Room Monitoring
Page 48

Live Streaming Over the
Internet for Local Governments
Page 50

Streaming of School Announcements
and Central Media Assets
Page 54

SMD 202

H.264 Streaming Media Player and Decoder

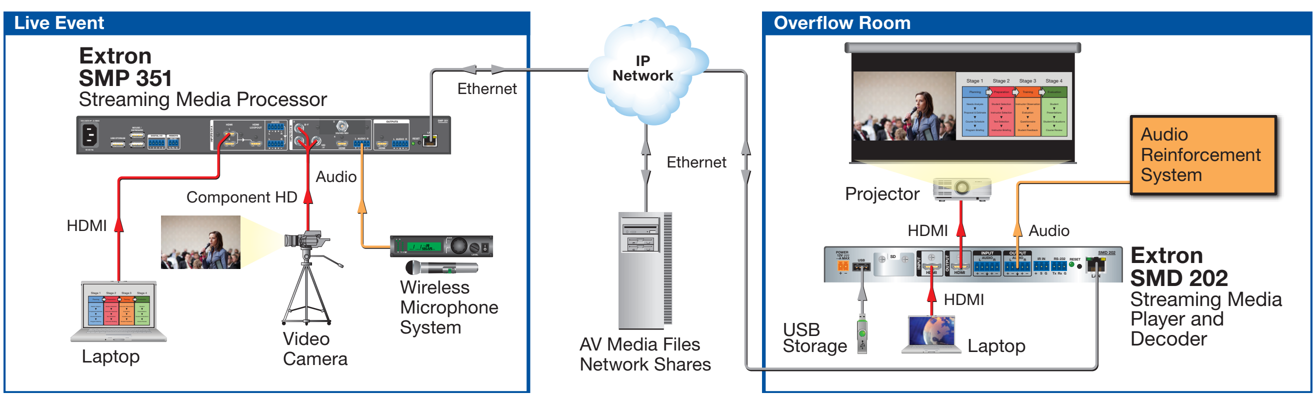


The Extron SMD 202 is a compact, high performance media player and live stream decoder used in H.264 streaming applications. It provides the flexibility to present a locally connected AV signal, decode a live streaming source, or play back media files from internal memory, removable SD card, local USB, or network storage. The SMD 202 supports a wide range of media file container formats and streaming protocols, making it adaptable for use with a variety of encoded media. Advanced signal processing, scaling, and aspect ratio management supply high quality signals to AV displays. An intuitive, interactive on-screen menu makes setup and source selection using front panel buttons or the optional handheld IR remote control easy. Designed for pro AV applications, the SMD 202 can be controlled using Ethernet, RS-232, IR, or wired IR.

FEATURES:

- Plays back media files from internal memory, removable SD card, USB storage, or network shares
- Decodes live H.264 streams using a variety of streaming protocols
- Local HDMI input with embedded stereo or analog stereo audio
- Selectable output resolutions from 640x480 to 1920x1200 including 1080p/60
- Supports streaming resolutions from 480x320 up to 1080p/60
- Multi-language, interactive on-screen display for setup and source selection
- Control from front panel, IR remote, wired IR, RS-232, Ethernet, or embedded Web interface
- On-screen presentation of streaming source information
- Play back media files in a loop or as part of a playlist
- Seamless media file transitions and looped playback
- Elegant still frame transition effects
- Play, pause, stop, and seek transport control
- Interactive On-Screen Display
- Optional SMD 202 Remote IR remote control
- HDCP-compliant HDMI input and output signal support with Visual Confirmation
- Selectable audio output format: HDMI-embedded stereo audio or analog stereo audio
- Ethernet to RS-232 pass through control
- Compatible with unicast and multicast push, or unicast and multicast pull streaming applications

MODEL	VERSION	PART#
SMD 202	H.264 Player and Decoder	60-1306-01
SMD 202 Remote	Handheld IR Remote Control for SMD 202	70-1059-01



SMD 101

RGB, DVI & Video Over IP H.264 Streaming Media Decoder

The SMD 101 is a compact, high performance H.264 decoder used with Extron SME 100 encoders to provide complete end-to-end AV streaming systems. The SMD 101 is used in H.264 enterprise streaming applications to decode live streams from SME 100 encoders or play back network-accessible AV media files available from network shares. It is compatible with streaming resolutions and refresh rates up to 1080p/60. The output resolution is selectable from 640x480 to 1920x1200. The SMD 101 offers integration-friendly control features such as IR remote, wired IR, RS-232, or Ethernet and an easy-to-navigate Web interface providing simple, flexible control and management options. This compact, energy-efficient decoder is an ideal counterpart to the SME 100 encoder in overflow, monitoring, multi-channel streaming systems, high resolution signage, and messaging applications.

FEATURES:

- Supports live IP video stream decoding
- Supports streaming resolutions from 480x320 up to 1080p/60
- AV media file playback from network shares
- Compatible with MP4 and MPEG-2 Transport Stream container formats
- Selectable audio output format: HDMI-embedded stereo audio or analog stereo audio
- Integrated scaler offers selectable output resolutions from 640x480 to 1920x1200
- Decode at native resolution
- EDID defined scaling
- Ethernet to RS-232 pass through control
- Fill/Follow/Fit Aspect Ratio Management
- Control from IR remote, wired IR, RS-232, Ethernet, or embedded Web interface
- Compatible with the full range of SME 100 streaming transport protocols
- Compatible with unicast and multicast push, or unicast pull streaming applications
- Optional SMD 101 REMOTE IR remote control

MODEL	VERSION	PART#
SMD 101	H.264 Streaming Media Decoder	60-1305-01

Ideal for use in applications that require:

- A hardware appliance for decoding live streamed content that can be managed and controlled as part of a professional AV system.
- The ability to manage and play back AV media files in playlists, and manage live stream decoding and media file playback as part of a channel list.
- Advanced AV signal processing, including high quality scaling, and aspect ratio management, provides superior video decoding and presentation quality.
- Support of diverse network topologies and conditions that require different streaming transport protocols based on network quality of service, using unicast pull and unicast or multicast push streaming.

Applications:

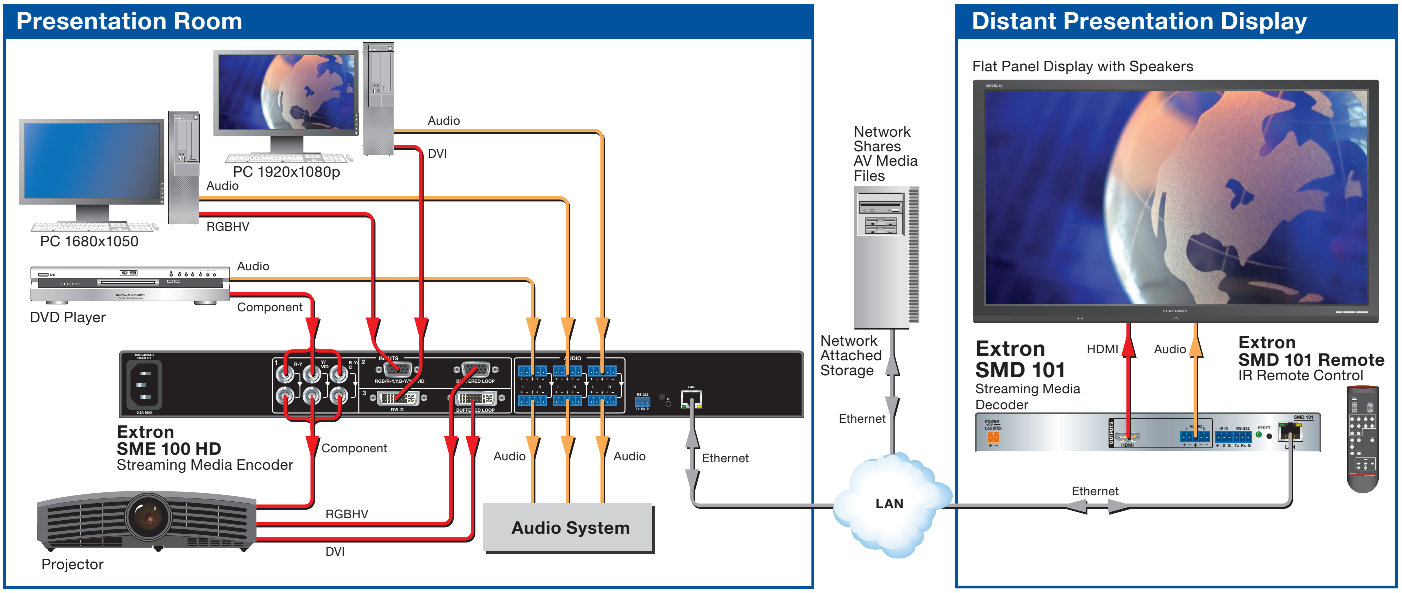
- High Definition Enterprise Streaming: Stream live video and audio or playback shared AV media files from AV presentation systems or displays located throughout an enterprise.
- High Resolution Signal Distribution: Distribute AV signals over a network when use of dedicated cable is impractical.
- Play back of Enterprise AV Media Assets: Play back network-accessible media files with promotional or educational content.
- High Resolution AV Signage Systems: Decode live streams or play back informational media presentations on public area displays throughout a facility.

Distributed Patient Monitoring System
Page 52

Streaming of School Announcements and Central Media Assets
Page 54

Corporate Enterprise H.264 Streaming
Page 56

High Resolution Signage
Page 58

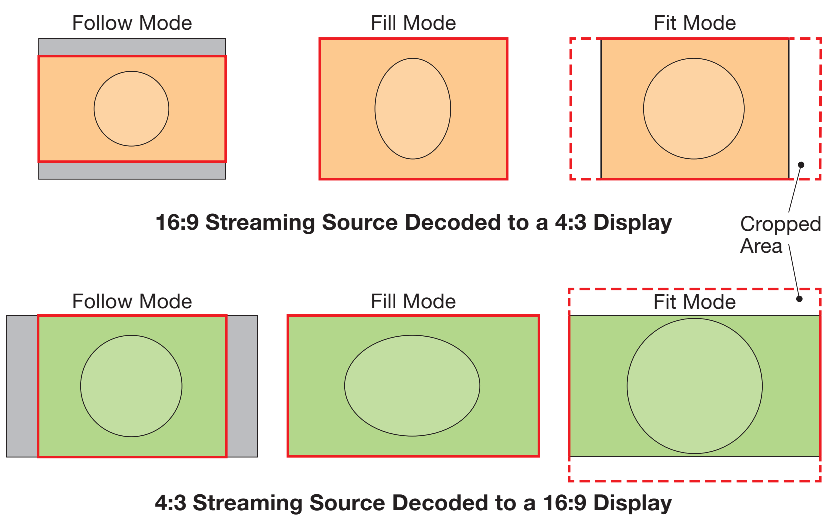


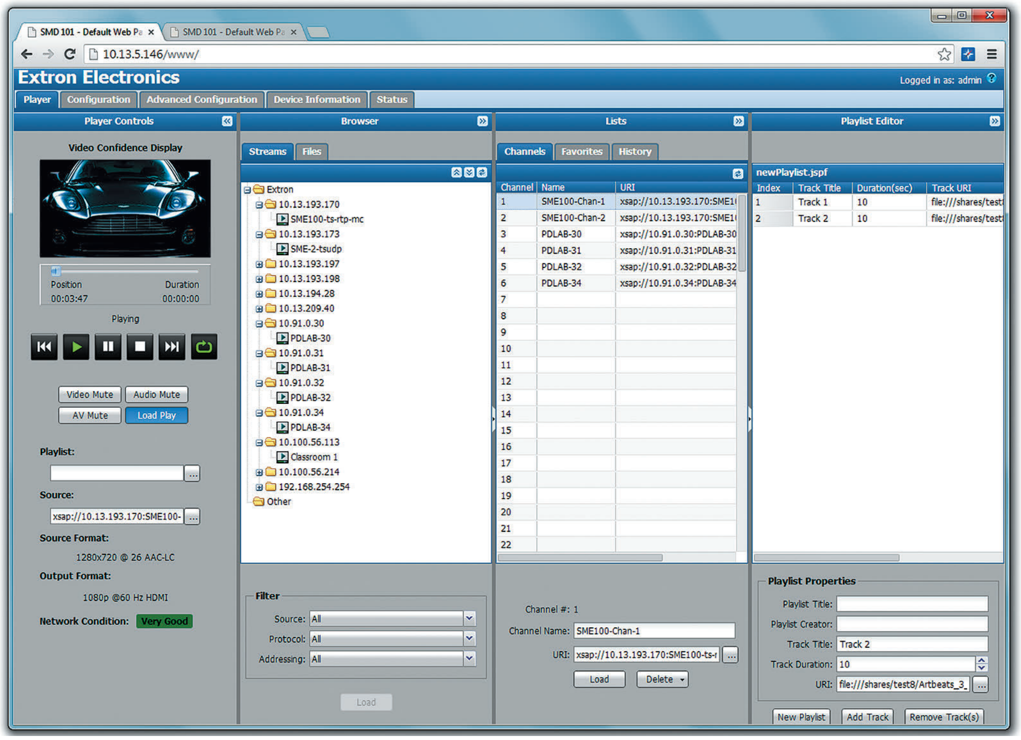
Live AV Stream Decoding and AV Media File Playback

The SMD 101 can decode live streams from SME 100 encoders and play back AV media files from network-accessible storage with shared access to networked devices – network shares. Compatible AV media file container formats include MP4 and MPEG-2 Transport Streams. This feature makes the Extron SMD 101 an AV media playback device as well as a live, H.264 streaming decoder.

Advanced Video Processing

The SMD 101 includes advanced video processing features that provide superior quality and simplify AV integration. It scales to a wide range of refresh rates and output resolutions from 640x480 to 1920x1200. The output rate and resolution can be set to a defined resolution or automatically selected based on EDID communication with the connected display. FILL, FOLLOW, and FIT aspect ratio control options provide choices for managing AV streaming channels where the aspect ratio of the content does not match the aspect ratio of the display. Video processing features such as these produce superior quality when presenting live streaming or on-demand content in AV systems.





The SMD 101 embedded web page provides the means to create playlists and channel lists. This simplifies selection of streaming sources from external control systems using Extron Simple Instruction Set - SIS commands.

Control and Management

SMD 101 and SMD 202 decoders can be managed and configured from their intuitive, easy-to-navigate embedded Web pages, which include a video confidence display window for verification of AV streaming sources. This embedded Web page clearly identifies features such as audio format, video format, resolution, user access, network, and control settings for straightforward configuration. The interface can also be used to select live AV streams or AV media files for playback. A progress bar presents the file path, position, and duration of AV media files.

Scalable System Deployment

Systems with many decoders that use common playlists and channel lists can be programmed and configured quickly and efficiently. Programmed source definition URLs and associated playlist and channel list data can be imported and exported as device configuration files. Once programmed, this configuration data can be uploaded to additional decoders, simplifying and streamlining system programming activities.

SMD 101 Remote

Handheld IR Remote Control, SMD 101

The Extron SMD 101 Remote is used with the Extron SMD 101 to select streaming source channels, which may include live streams, AV media file clips, or playlists. It provides control over audio volume, audio and video mute, and on-screen display of the AV media file playback progress bar or device information.

FEATURES:

- Provides infrared remote control of the SMD 101
- Streaming channel selection of live streams, AV media file clips and playlists
- Playback control of AV media files
- Analog audio volume
- Audio and video mute
- On-screen display of AV media file playback progress bar and device information
- Power on/off/standby
- Approximate range: 30 feet (9 meters)

MODEL	VERSION	PART#
SMD 101 Remote	Handheld IR Remote Control for SMD 101	70-1058-01

VN-Matrix® 250 Series

HDCP-Compliant HDMI and RGB Video Over IP Encoders and Decoders

The VN-Matrix® 250 Series provides real-time transmission of high resolution HDCP-compliant HDMI, DVI, or RGB video across standard IP networks for use in real-time streaming, recording, and playback applications. The VN-Matrix 250 Series accepts HDMI and RGB signals at resolutions up to 1920x1200 and 2048x1080, streams video and audio over an IP network, and decodes content back to original source resolution. Designed specifically to support mission critical AV applications, the VN-Matrix 250 Series supports streaming of commonly used AV signals, including HDCP-encrypted video. Stereo analog and HDMI-embedded stereo audio signals are both supported, providing compatibility with embedded display speakers or existing audio systems. The VN-Matrix 250 Series is ideal for applications with the most demanding quality requirements, such as command and control, simulation, medical, and distance collaboration with complex computer visualizations.

FEATURES:

- Streams HDMI, DVI, or RGB video and stereo analog or HDMI-embedded audio
- HDCP-compliant streaming
- Stream at native resolutions up to 1920x1200 and 2048x1080
- Low latency streaming - encode and decode are 35 ms each
- Extensive bit rate management
- High immunity to network errors
- PURE3® Codec
- 100/1000 BaseT Ethernet port available for AV control devices to interface with the VN-Matrix system
- Optional SFP port available as alternate streaming port for use with fiber-optic transceivers
- 10/100 BaseT Ethernet port available for AV control devices to interface with the VN-Matrix system
- Two-way audio streaming
- Analog stereo or HDMI-embedded stereo audio
- Audio breakaway streaming
- EDID Emulation
- Key Minder® continuously verifies HDCP compliance
- HDCP Visual Confirmation provides a green screen when encrypted content is sent to a non-compliant display
- Auto-Image

VNE 250 Encoder

Encoder for HDCP-Compliant HDMI and RGB Video Over IP

UNIQUE FEATURES:

- Encoder-only model
- Supports resolutions up to 1920x1200 and 2048x1080
- User-definable analog video source capture
- Accepts HDMI-embedded stereo audio and analog stereo audio
- EDID Minder® automatically manages EDID communication between connected devices
- USB host connection for keyboard and mouse data

MODEL	VERSION	PART#
VNE 250	Encoder for HDMI, RGB, Audio, USB K&M......	60-1274-01

VND 250 Decoder

Decoder for HDCP-Compliant HDMI and RGB Video Over IP

UNIQUE FEATURES:

- Decoder-only model
- Decode at native resolution or scale to display
- Analog stereo audio or HDMI-embedded stereo audio output
- Genlock connection for synchronized decoding
- USB keyboard and mouse interface
- Videowall magnification scaling
- Aspect ratio control

MODEL	VERSION	PART#
VND 250	Decoder for HDMI, Audio, USB K&M	60-1275-01

VN-Matrix 325 Codec

3G-SDI Over IP Codec

The VN-Matrix® 325 streams 3G-SDI, HD-SDI, or SDI video and embedded audio over IP networks to meet the emerging need for low delay, production quality transport of SD and HD video over enterprise networks. The VN-Matrix 325 codec uses Extron's PURE3® codec which exceeds many of the performance characteristics of standards-based compression formats, delivering visually lossless imagery with low latency, and a high immunity against network errors. The VN-Matrix 325 codec is switchable between encode and decode functionality and video decoding can be genlocked to an external SDI reference. VN-Matrix 325 is ideal for quality-critical applications such as live video delivery across a campus, production collaboration, studio-to-studio media exchange, and original video source contribution.

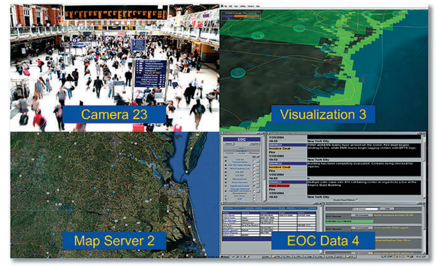
FEATURES:

- Streams serial digital video with embedded audio
- Supports 3G-SDI, HD-SDI, or SDI signals
- Low latency streaming - 35 ms encode and 35 ms decode
- 10-bit YCrCb 4:2:2 encoding
- Supports resolutions up to 1080p/60
- Embedded audio
- Streaming data rates from 6 Mbps to 150 Mbps
- Codec switchable between encode and decode operation
- Decoding is genlockable to an external SDI reference

MODEL	VERSION	PART#
VNC 325 3G-SDI	Codec for 3G-SDI	60-1249-01

VNS 104

Multi-Stream Decoding Software for VN-Matrix 250, 225, or 200 Series



VNS 104 Multi-Stream Decoding Software decodes one or four video streams and one stereo audio stream from VN-Matrix 225 real-time encoders and VN-Matrix Recorder playback channels. It operates on a Windows PC and is managed from a VN-Matrix Enterprise Controller. The VNS 104 is used in monitoring, remote presentation viewing, distance collaboration, and data visualization, in a variety of environments including command and control, after action review, training and simulation, medical and geological visualization.

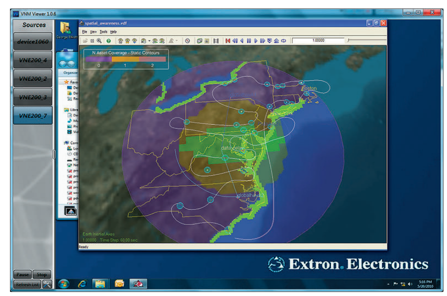
FEATURES:

- Decodes and displays one or four VN Matrix PURE3® video streams on a single display from a Windows PC platform
- Decodes one audio source which is selectable from the four decoded VN Matrix streams
- Managed and controlled from VN-Matrix Enterprise Controller
- Monitor four VN-Matrix streams from one display as a cost effective alternative to multiple hardware decoders and displays
- Scaling and aspect ratio control
- Select display mode and switch streams from an external control interface

MODEL	VERSION	PART#
VNS 104	Four Stream Decoding Software License	29-108-01

VNM Software Decoder

Software Decoder for VN-Matrix 250, 225, or 200 Series



The VNM Software Decoder application lets users view live PURE3® streams on a PC. The streams can originate from VN Matrix 225 or 200 encoders as well as any active playback streams from VNM Recorder. The VNM Software Decoder application includes a plugin that operates with Windows Media Player® to decode PURE3 streams. VNM Software Decoder identifies and lists available VN Matrix encoders and any active playback streams from VNM Recorders in buttons which can be selected for viewing in the application window. Connections to VN-Matrix 225, 200, or PURE3 streams can also be initiated from the embedded VN-Matrix Web browser interface.

FEATURES:

- Operates in conjunction with Microsoft® Windows Media Player
- Software decoder application provides basic source selection
- Decodes streams from VN-Matrix 250, 225, or 200 Series products
- Compatible with Microsoft® Windows
- Installs quickly and simply on any standard PC
- Install decoder on as many endpoints as desired
- Number of live stream decodes is licensed by system
- For use on LANs or private networks supporting multicast traffic

MODEL	VERSION	PART#
VNM Software Decoder	Single Stream Decoding Software License	29-098-01

VNR 100

VN-Matrix Single Channel Recorder

The VNR 100 digitally records and plays back high-definition computer graphics, video, audio, and data streamed in VN Matrix systems. It can record and play back at the same time, increasing duty cycles for expensive source equipment and presentation systems by utilizing both recording and playback features simultaneously. The time-slip feature, allows a live event to be recorded while a previous event is played back, and the chase-play feature allows a recording in progress to be streamed with a time-shifted delay. VN-Matrix systems can be configured with multiple VNR 100 units to record and play back synchronized multi-source AV presentations or multi-screen display systems. The VNR 100 is ideally suited for AV streaming and recording applications with the most demanding quality and performance requirements.

FEATURES:

- Simultaneously record and play back VN-Matrix AV streams
- Time-shift capabilities support time-slip or chase-play applications
- Transport controls include: play, pause, and variable speed playback at 2x 4x 8x speeds in forward or reverse as well as single frame advance in forward or reverse
- System scalability – create multi-channel recording systems using multiple VNR 100 units
- System synchronization – synchronize playback across multiple VNR 100 units
- Replacement media drive and operating system drive available
- Compatible with VN-Matrix 200, 225, 250 and 300 Series and VN-Matrix Software Decoder

MODEL	VERSION	PART#
VNR 100	VN-Matrix Single Channel Recorder	60-1291-01
VNR 100 OS	VNR 100 OS Drive 80 GB in Caddy	70-1002-01
VNR 100 MD	VNR 100 Media Drive 1 TB in Caddy	70-1003-01

VNM Recorder

VN-Matrix Multi-Channel Recorder

The VNM Recorder is a network appliance used to digitally record and play back high-definition computer graphics, video, audio, and data streams encoded to the IP network via VN-Matrix codecs, encoders, and decoders. The VNM Recorder is ideally suited for any VN-Matrix application requiring the documentation, archive, review, and playback of highly-sophisticated or demanding imagery.

Up to five PURE3® streams can be recorded or played back as a group per Recorder. PURE3 streams recorded together will maintain tight synchronization on playback. Link multiple Recorder units together for applications requiring more than 5 streams. Playback controls available to external control systems offer the following capabilities: search, locate, variable speed playback - forward and reverse, single frame advance.

FEATURES:

- Records multiple VN-Matrix PURE3® encoded audio and video IP streams
- Select up to five PURE3 streams to record and play back later as a group
- Configure systems with multiple recorders for applications that must record and play back more than five streams
- Playback controls available to external control systems include: search, locate, variable speed playback - forward and reverse, single frame advance
- Export sequential JPEG or Targa frames for use in media players or video productions
- 2 Terabyte RAID5 formatted storage with hot spare hard drive

MODEL	VERSION	PART#
VNM Recorder	Recorder	60-1121-01
VNMR Drive Set	Recorder Drive Set	70-852-01

VNM EC 200

Enterprise Controller for VN-Matrix Systems

The VNM EC 200 is a compact, dedicated enterprise controller for VN-Matrix® systems. It simplifies management of large VN-Matrix deployments, efficiently configuring, managing, and dynamically controlling large VN-Matrix systems from a single user interface. VN-Matrix devices can be selected in groups, and common configuration properties can be applied to all devices. Firmware can be uploaded to all devices or any group of devices in one action. Multiple VNM EC 200 controllers can be deployed in VN-Matrix systems, providing control over the entire system or independent clusters of VN-Matrix devices, or creating redundant systems for mission-critical applications.

Every VN-Matrix system includes a basic embedded Web browser interface for the configuration of encoders and decoders. This embedded interface is useful in small systems limited to a few units or systems that will be left in a fixed operational state. The VNM EC 200 provides greater processing capacity for efficient management, configuration, and dynamic control over large VN-Matrix systems. Its embedded web interface quickly organizes and sorts all devices in a system based on properties such as: unit status or type, operating mode, source, controller, or firmware version. Multiple VNM EC 200 controllers can be applied in VN-Matrix systems, providing control over the entire system or independent clusters of VN-Matrix devices.

The VNM EC 200 can also create streaming and recording system presets that can be recalled directly from an external control system. Each preset captures all device settings and defines specific streaming connection and recording actions. Recall of various streaming and recording presets greatly simplifies management and control of VN-Matrix systems.

Two VNM EC 200 units can be configured to operate together as a redundant system for mission-critical applications, with one configured as a primary unit, and the other as a secondary unit. System data is continuously synchronized between the redundant pair. The secondary unit continually monitors the primary unit's system health and seamlessly takes control in the event of a failure. Control is automatically returned to the primary unit if required, maintaining transparent system communications and control.

The VNM EC 200 is required when:

- More than 10 VN-Matrix units are configured into a system
- VNR 100 or VN-Matrix Recorder is integrated into a system
- A system of VN-Matrix units are interfaced to an external control system requiring dynamic control of the units in a switching solution
- Preset streaming and recording configurations must be prepared and recalled
- Multiple domains of VN-Matrix units must be operated as into one large system or independent clusters

FEATURES:

- Manage, configure, and control all VN-Matrix units as a system
- Create system presets for stream routing and recording configurations
- Control VNS 104 Multi-Stream Decoding Software display configurations and streaming connections
- High-level interface provides single point of control from external control systems
- Manage multiple VN-Matrix systems in combined or independent domains
- Provide redundant control for mission-critical applications
- Removable Solid State Drive for reliability or temporary removal of sensitive system data

MODEL	VERSION	PART#
VNM EC 200	Enterprise Controller for VN-Matrix Systems....	60-1516-01

Streaming AV Over IP Glossary

The new language of the IP-Internet Protocol era is used throughout this Guide. This lexicon of words, phrases, acronyms, and abbreviations appropriate to AV streaming over IP technologies, distribution methods, and the products is defined in the following Glossary of Terms.

PURE 3® CODEC

OPTIMIZED FOR NETWORKS

PURE3® Codec

A codec that is capable of encoding and streaming both video and computer graphic inputs and a wide variety of resolutions, preserving equal quality for both signal formats. It preserves a balance between three performance factors: low latency, low bandwidth, and high image quality. The PURE3® Codec has been optimized for use on IP networks which are acknowledged to be lossy. The codec includes an error concealment system which is highly resistant to network errors without using forward error correction.

Layer 7	Application
Layer 6	Presentation
Layer 5	Session
Layer 4	Transport
Layer 3	Network
Layer 2	Data Link
Layer 1	Physical

OSI Open System Interconnection Reference Model

The OSI Reference Model is a definition for layered communications and computer network protocol design. It was developed as part of the Open Systems Interconnection (OSI) initiative. The OSI model divides the network architecture into seven layers starting from the bottom up: Physical, Data Link, Network, Transport, Session, Presentation, and Application Layers.

2K
The Digital Cinema Initiatives consortium - DCI - established the resolution 2048x1080 for use with digital projection systems and monitors that were targeted to replace motion picture film systems.

3G-SDI
The new signal standard for serial digital, high definition video with 1920x1080 resolution and a 50Hz or 60Hz progressive frame rate. Up to 16 audio channels can be carried in the ancillary data. The 3G stands for 3 gigabits per second, which is 2 times the bit rate of a 1.485 Gbps HD-SDI signal.

4:1:1 color space
Chroma, or color information, is sampled at one-fourth the horizontal resolution of the luminance, or black and white information.

4:2:0 color space
Chroma, or color information, is sampled at half the vertical and half the horizontal resolution of the luminance or black and white information.

4:2:2 color space
Color information is sampled at half the horizontal resolution of the luminance, black and white information. 4:2:2 color sampling is popular in high-quality broadcast video systems.

4:4:4 color space
Color information is sampled at the same rate as the luminance, black and white information. Video systems designed for capturing real images typically quantize color information at one-fourth to one-half the detail of luminance information. This is acceptable for real images, where sharp, on-off transitions between colors do not occur. Computer graphic pictures contain sharp, pixel transitions and require maintenance of 4:4:4 color space; otherwise, information is lost.

4K
The Digital Cinema Initiatives consortium - DCI - established 4096 × 2160 (8.8 megapixels, aspect ratio ~17:9) as the standard resolution for 4K film projection. 4096x2160 is the native resolution for DCI-compliant 4K digital projectors and monitors. The DCI 4K standard has twice the horizontal and vertical resolution of DCI 2K, producing four times as many pixels overall.

10Base-T
An Ethernet standard for transmitting data packets at 10 Mbps over twisted pair cable. 10Base-T is a shared media. When used with a hub, all network nodes must share the same 10 Mbps capacity. When used with a switch, each connection supports a 10 Mbps duplex capacity.

100Base-T
An Ethernet standard for transmitting at 100 Mbps over twisted pair wire. 100Base-T was also called “Fast Ethernet” when first deployed in 1995. Officially the IEEE 802.3u standard, it is a 100 Mbps version of 10Base-T. Like 10Base-T, 100Base-T is a shared media LAN when used with a hub and 100 Mbps duplex when used with a switch.

1000Base-T / Gigabit Ethernet
An Ethernet standard that transmits at 1 Gbps over twisted pair wire. Use of Gigabit Ethernet is becoming commonplace and will eventually be used as frequently as 100Base-T connections.

A

AAC (Advanced Audio Coding)
Advanced Audio Coding - AAC - is a standardized, lossy compression and encoding scheme for digital audio commonly used in streaming applications. Designed to be the successor of the MP3 format, AAC generally achieves better sound quality than MP3 at similar bit rates. AAC has been standardized by ISO and IEC, as part of the MPEG-2 and MPEG-4 specifications. AAC is the default or standard audio format for YouTube, iPhone, iPod, iPad, Nintendo DSi, Nintendo 3DS, iTunes, DivX Plus Web Player and PlayStation 3.

ADSL (Asymmetrical Digital Subscriber Line)
Asymmetrical Digital Subscriber Line - ADSL - one of a number of DSL technologies for communications over copper telephone lines, and the most common one. ADSL is designed to deliver more bandwidth downstream (from the central office to the customer site) than upstream.

Analog
A continuous range of values to represent information. An infinite resolution of values can be established in an analog system.

Animations
Animations consist of motion image sequences produced synthetically on video processing or computing systems.

Artifacts
Any error in the perception or representation of any visual or aural information introduced by the involved equipment. Image artifacts appear as deviations from the original in the delivered image in video streaming systems.

ATM (Asynchronous Transfer Mode)
Asynchronous Transfer Mode - ATM - a standardized digital data transmission technology that is a cell-based switching technique and uses asynchronous time division multiplexing. This is the core protocol used over the SONET/SDH backbone of the ISDN (Integrated Services Digital Network).

B

B-Frame
Bi-directionally predictive coded picture. Contains predictive, difference information from the preceding and following I- or P-frame within a GOP. Data preceeding or following the B-frame are required to recreate video information in a B-frame.

Bandwidth
The capacity or available bandwidth in bit/s, which typically means the net bit rate, channel capacity, or the maximum throughput of a logical or physical communication path in a digital communication system.

BER (Bit Error Rate)
Bit Error Rate - BER -the rate at which bit errors are experienced across a data connection.

Best Effort
Describes a network service in which the network does not provide any guarantees that data is delivered or that a user is given a guaranteed quality of service level or a certain priority.

Bidirectional
The ability to move, transfer, or transmit in both directions.

Bit Depth
The number of bits used to represent the luminance and chrominance of a single pixel in a bitmapped image or video frame buffer. This concept is also known as bits per pixel (bpp), particularly when specified along with the number of bits used. Higher color depth gives a broader range of distinct colors.

Bit Error
Bit error indicates the number of bits of a data stream over a communication channel that have been altered. A bit error can result in unusable data or the corruption of an image in video streaming solutions.

Bit Rate
The number of bits that are conveyed or processed per unit of time. The bit rate is quantified using the bits per second (bit/s or bps) unit, often in conjunction with an SI prefix such as kilo- (kbit/s or kbps), mega- (Mbit/s or Mbps), or giga- (Gbit/s or Gbps).

Bridge
A device that connects two network segments together. These network segments may be similar or dissimilar, such as Ethernet and Token Ring. A bridge is inserted in the network to keep traffic contained within the segments to improve performance.

Broadcast
The operation of sending network traffic from one network node to all other network nodes.

Buffer
A region of memory used to temporarily hold data while it is being delivered from one process to another.

Burst
A sequence of data delivered in a short period of time. Network designs must account for both predictable data traffic and bursts of traffic.

Burst Error
Consecutive data errors that occur suddenly. If errors spanning several bytes occur, complete decoding at the receiving end may not be possible even if error correction is applied. As a measure against burst errors, methods such as interleaving are used. Errors occurring on real world networks are typically burst errors.

C

CET (Carrier Ethernet Transport)
Carrier Ethernet Transport - CET - wide-area Ethernet services used for high-speed connectivity within a metropolitan area, nationwide, or even internationally.

Chromaticity
An objective specification of the quality of a color regardless of its luminance. The quality is determined by its hue and colorfulness (or saturation, chroma, intensity, or excitation purity).

Chrominance
The measurement of the color value or color difference value in a pixel.

Collision
Two devices on a network attempt to use the physical media at the same time. The data from the two devices “collides.”

Color Depth
Describes the number of bits used to represent the color of a single pixel in a bitmapped image or video frame buffer. A common bit depth applied to computer graphic signals is 8-bits each for Red, Green, and Blue. An 8 bit depth will produce 256 levels and 256 raised to the third power results in a resolution of over 16 million colors.

Color Quantization
Color quantization defines the resolution, or number of colors used in a system. This is important for displaying images that support a limited number of colors and for efficiently compressing certain kinds of images. For example, reducing the number of colors required to represent a digital image makes it possible to reduce its file size or streaming bit rate.

Color Space
A technique for describing a color or a group of colors mathematically. A way to define a grouping with the entire range of chromaticities, often represented as a triangle within the CIE 1931 chromaticity diagram. Different image systems may apply different color spaces. The color space applied to broadcast video standards is different than the RGB color space used by computer systems.

Communication Bandwidths
Below are listed commonly available bandwidths for network switching equipment and connections made available for public and private networks.

All communication bandwidths presented below are listed in Megabits/s (Mb/s). 1,000 Mb/s equals 1 Gigabit/s (Gb/s).

LAN Connections	Mb/s
Ethernet (10 BaseT)	10.00
Fast Ethernet (100 BaseT)	100.00
Gigabit Ethernet (1000BaseT)	1,000.00
10 Gigabit Ethernet	10,000.00

WAN and MAN	Mb/s
ISDN4	0.51
DS1/T1	1.54
DS1C/T-1C	3.15
DS2/T-2	6.31
DS3/T3	44.74
DS3D/T-3D	135.00
DS4	274.18
E-1	2.05
E-2	8.45
E-3	34.37

E-4	139.26
E-5	565.15
OC-1	51.84
OC-3	155.52
OC-12	622.08
OC-24	1,273.86
OC-48	2,547.71
OC-192	10,240.00
OC-256	13,589.50
OC-768	40,768.51

Remote Wireless	Mb/s
Satellite Internet	0.51
Broadband Satellite Internet	2.02
Microwave 4 DS1	6.18
802.11b	11.00
IPDirect-Sat	20.00
802.11g	54.00
Microwave OC3	155.31
10G Laser	10,000.00

Please Note: *Full-duplex switched fabric capacity is typically specified by manufacturers. Half-duplex capacity is typically more relevant to multicast video applications as it identifies the one-way sustained throughput directionally. Switched network architecture and intelligent switching features, including hardware or software routing, multicast routing protocol support, latency, and other factors, can be far more critical to consider than switched fabric capacity when designing switched networks.*

Compression
The art and science of reducing the amount of data required to represent a picture or a stream of pictures and sound before sending or storing it. Compression systems are designed to eliminate redundant or repeated information to the desired data level while allowing the original information to be reproduced to the desired quality.

Congestion
Occurs when a link or node is carrying so much data that its quality of service deteriorates. Typical effects include queueing delay, packet loss, or the blocking of new connections. A consequence of this is that increases in offered load lead to only small increases in network throughput, or to an actual reduction in network throughput.

Constant Bit Rate (CBR)
Constant bit rate encoding means that the rate at which a codec's output data should be consumed is constant. CBR is useful for streaming multimedia content on data communication channels, which operate more efficiently or require the bit rate to remain within a tight tolerance. Typically the constant bit rate is created by stuffing bits into a variable bitrate signal that has a defined peak or maximum limit.

Container Format
Container formats package encoded audio and video together in a file format that can be used in media players. One or many audio and video data streams will be held in a container file along with other data including the audio and video codecs and synchronization information that is necessary to play back the streams cohesively. Other data that may also exist in a container format includes subtitles, chapter information and other metadata.

Constant Quality
The quality output from a process, such as video encoding, remains constant, while the output, such as a bit rate, may vary. Constant quality encoding will result in a variable bit rate if the nature of the video material changes.

Content Delivery Network (CDN)
A content delivery network - CDN - is an interconnected system of computers on the Internet that provides Web content rapidly to numerous users by duplicating content on multiple servers and directing it to users based on proximity. CDNs can deliver static or dynamic Web pages, and are particularly well suited to streaming audio and video. It provides efficient delivery of media and data over the Internet to diverse destinations.

Contention
The media that network devices use to deliver data is overused and "contention" for the media is experienced.

COP-3
A Code of Practices established by the MPEG Forum for the transmission of MPEG-2 transport streams which applies a technique known as Forward Error Correction to protect the enclosed data.

COP-4
A Code of Practices for the transmission of uncompressed standard video at up to 270 Mbps and High Definition video at up to 1.485 Gbps which applies a technique known as Forward Error Correction to protect the enclosed data.

CoS (Class of Service)
Class of Service - CoS - method of classifying traffic on a packet-by-packet basis using information in the type-of-service (ToS) byte to provide different service levels to different traffic. See also QoS.

CRC (Cyclic Redundancy Check)
Cyclic redundancy check - CRC - or polynomial code checksum is a method used to detect changes or errors in raw data, and is commonly used in digital networks, data communications, and storage devices.

CSMA/CD (Carrier Sense Multiple Access/Collision Detection)
Carrier Sense Multiple Access/Collision Detection - CSMA/CD - the Media Access Control method applied in Ethernet networks. When a device wants to gain access to the network, it checks to see if the network is quiet (senses the carrier). If it is not, it waits a random amount of time before retrying. If the network is quiet and two devices access the line at exactly the same time, their signals collide. When the collision is detected, they both back off and each waits a random amount of time before retrying.

D

DASH aka MPEG-DASH (Dynamic Adaptive Streaming over HTTP)
Dynamic Adaptive Streaming over HTTP - DASH or MPEG-DASH - enables adaptive bitrate, high quality streaming of media content over the Internet leveraging conventional HTTP web servers. It is the first adaptive bit-rate HTTP-based streaming solution to become an international standard.

Data Compression Ratio
The ratio representing the data output from a compression system relative to the original data. A computer-science term used to quantify the reduction in data-representation size produced by a data compression algorithm.

Data Services
A telecommunications service that transmits high-speed data rather than voice. Internet access is the most common data service, which may be provided by the telephone and cable companies as well as cellular carriers.

Decoder
A device that converts encoded information to a desired interface. The same method used to encode is usually reversed in order to decode. Video over IP decoders accept IP data streams and output an analog or digital video signal.

Digital
A data technology that uses discrete (discontinuous) values.

Discrete Cosine Transform (DCT)
A Fourier-related transform that is used to convert an image from a spatial domain to a frequency domain. Video systems then process the information in the frequency domain. Typically, more signal energy is located in the lower frequencies than the higher frequencies. The DCT is used in many video compression codecs including JPEG, MPEG, MPEG-2, MPEG-4, and H.264.

Discrete Wavelet Transform (DWT)
A transform used to convert an image from a spatial domain to a wavelet domain. Two filters are involved. The first a "wavelet filter" is a high pass filter, and the second a "scaling filter" is a low pass filter. The DWT provides more efficient image compression than the DCT due to advantages from analyzing signals with sharp discontinuities or spikes.

DSL (Digital Subscriber Line)
Digital Subscriber Line - DSL - a generic name for a family of digital lines (also called xDSL) provided by telephone carriers to businesses and consumers.

Dual-link HDSDI
A method applying two HDSDI signals 1920x1080 video at 50 or 60Hz as progressive frames at 12 bit depth or with 4:4:4 color quantization.

DVI (Digital Visual Interface)
Digital Visual Interface - DVI - the digital video connectivity standard that was developed by the DDWG – Digital Display Working Group. This connection standard offers two different connectors: one with 24 pins that handles digital video signals only and one with 29 pins that handles both digital and analog video. This standard uses TMDS – Transition Minimized Differential Signal from Silicon Image and DDC – Display Data Channel from VESA – Video Electronics Standards Association.

DVI-D
DVI connector that supports digital signals only.

DVI-I
Connector that supports both digital and analog signals.

E

Encoder
A device, circuit, or algorithm that converts information from one format to another. Video over IP encoders take analog or digital video input signals and convert them to IP data streams which are transmitted over IP networks.

Error Concealment
A method of concealing and hiding the impact of data lost during transmission. In video streaming systems, error concealment prevents lost network packets from disrupting a video frame or sequence of video frames.

Error Correction
A method of detecting errors and reconstructing the original information using extra, redundant information sent along with the original data.

Error propagation
A single error experienced produces a knock on effect to sequential information. In video streaming solutions, decoding products should provide a method by which a single error encountered affects only a small area of a picture and not affect an entire frame or sequential frames of video.

Ethernet
A Local Area Network (LAN) standard officially known as IEEE 802.3. Ethernet and other LAN technologies used for interconnecting computers, printers, workstations, terminals, servers, etc. within the same building or campus. Ethernet operates over twisted pair and over coaxial cable at speeds starting at 10 Mbps. For LAN interconnectivity, Ethernet is a physical link and data link protocol reflecting the two lowest layers of the OSI Reference Model.

Ethernet MAC Frames
A digital data transmission unit or data packet that includes frame synchronization information and a data payload. The synchronization data makes it possible for the receiver to detect the beginning and end of the packet in the stream of symbols or bits.

F

Firewall
A device that manages access of devices outside a network into a network, typically into a building or an enterprise. A firewall prevents unauthorized access to a network. It is also used to check on data delivered to and from a network to ensure the information is non-damaging.

Forward Error Correction (FEC)
A system of error control for data transmission, whereby the sender adds redundant data to its messages, also known as an error-correction code. This allows the receiver to detect and correct errors (within some bound) without the need to ask the sender for additional data. The amount of FEC required to guarantee delivery is not certain. Each application must consider the predictability of the network and the amount of protection that is desired.

Flash Player
Flash Player is a proprietary media player originally created by Macromedia, and developed and distributed by Adobe Systems since its acquisition of Macromedia late 2005. The Adobe Flash Player is free media player for playing multimedia files and streamed video and audio content created on the Adobe Flash platform. It is compatible with a large number of streaming protocols including RTMP, RTMPT, RTMPS, RTMPE, RTMPTE, RTMFP, and HTTP - progressive, HDS, and HLS.

Frame Lock
Multiple video sources delivered together, that maintain frame synchronization are frame locked. Frame lock is required for delivery of stereoscope 3D imagery consisting of two locked signals or 4K resolution images, which are built up using four, synchronized HD video signals.

Frame Rate
The frequency at which an imaging device produces unique, consecutive images called frames. The term applies equally to computer graphics, video cameras, film cameras, and motion capture systems. Frame rate is most often expressed in frames per second (FPS) and sometimes in progressive scan monitors as hertz (Hz). It can also be seen as refresh rate or vertical scan rate.

Frame Relay
Public, connection-oriented packet service based on the core aspects of the Integrated Services Digital Network. It allows private networks to reduce costs by using shared facilities between the end-point switches of a network managed by a Frame Relay service provider. Individual data-link connection identifiers (DLCIs) are assigned to ensure that each customer receives only its own traffic.

G

Gateway
A network node equipped for interfacing with another network that uses different protocols. Also can be described as an entrance and exit into a communications network.

Genlock
A common technique where the video output of one source, or a specific reference signal, is used to synchronize other television picture sources together. Video sources that are genlocked have vertical sync pulses, which are synchronized together.

GOP (Group of Pictures)
A Group of successive pictures within a coded video stream. MPEG, MPEG-2, and H.264 compression products apply a GOP structure to their video compression systems. Each coded video stream consists of successive GOPs. The visible frames are generated from the pictures contained in it. A GOP begins with an I-frame containing the full temporal resolution of the video frame. A series of predictive information is calculated between I-frames. P-frames are predictive and estimate forward. B-frames apply bidirectional prediction and estimate forwards and backwards. Products will apply GOP structures in different manners to support the needs of different applications, whether: low delay, low bit rate, or error resilience.

H

H.264 (MPEG-4 AVC)
A block-oriented, motion-compensation-based codec standard developed by the ITU-T Video Coding Experts Group (VCEG) together with the ISO/IEC Moving Picture Experts Group (MPEG). It is the product of a partnership effort known as the Joint Video Team (JVT). H.264 is used in such applications as Blu-ray Disc, videos from YouTube and the iTunes Store, DVB broadcast, direct-broadcast satellite television service, cable television services, and real-time videoconferencing.

H.265
See “High Efficiency Video Coding.”

HDS (HTTP Dynamic Streaming)
HTTP Dynamic Streaming - HDS - is a proprietary HTTP adaptive bit rate streaming protocol by Adobe Systems for use with Adobe Flash Players.

HD-SDI (High Definition SDI)
The high-definition version of SDI specified in SMPTE-292M. This signal standard transmits audio and video with 10 bit depth and 4:2:2 color quantization over a single coaxial cable with a data rate of 1.485 Gbit/second. Multiple video resolutions exist, including progressive 1280x720 and interlaced 1920x1080 resolution. Up to 32 audio signals are carried in the ancillary data.

HEVC (High Efficiency Video Coding)
High Efficiency Video Coding - HEVC - sometimes referred to as H.NGVC - Next-generation Video Coding - or H.265, is the successor to H.264/MPEG-4 AVC developed by the Joint Collaborative Team on Video Coding - JCT-VC. HEVC is said to double the data compression ratio of H.264/MPEG-4 AVC with the same quality. It can support 8K UHD and resolutions up to 8192×4320. HEVC replaces macroblocks used previously, with Coding Tree Units - CTUs - which can use larger block structures of up to 64×64 pixels and can better sub-partition into variable sized structures.

High-Definition Video
Refers to any video system of higher resolution than standard-definition (SD) video, and most commonly involves display resolutions of 1280×720 pixels (720p) or 1920×1080 pixels (1080i/1080p).

HLS (HTTP Live Streaming)
HTTP Live Streaming - HLS - is an HTTP-based media streaming transport protocol implemented by Apple Inc. as part of their QuickTime and iOS software for devices such the IPhone and iPad. HLS breaks a stream into a sequence of small HTTP-based file packets. As an HLS stream plays, a client may adapt the session to changing conditions by selecting from a number of alternate streams with varying data rates, Unlike UDP-based protocols such as RTP, an advantage of HTTP Live Streaming is its ability to traverse firewalls and proxy servers that allow standard HTTP traffic.

Hop
In a packet-switching network, a hop is the trip a data packet takes from one router or intermediate point to another in the network.

Hop Count
On the Internet (or a network that uses TCP/IP), the number of hops a packet has taken toward its destination.

HTML5
HTML5 is a markup language used to structure and present content for the World Wide Web. It is the fifth revision of the HTML standard. HTML offers greater support for multimedia, making it easier to embed video, audio, and graphical content without the need for proprietary plugins and APIs. It is the AV media player used in iOS operating system devices such as iPads and iPhones.

HTTP Tunneling
HTTP Tunneling encapsulates data within in the HTTP transport protocol. It is often used to transport RTP and RTSP streaming data through firewalls. HTTP traffic passes freely through port 80, the normal port used by web browsers, so it has the advantage of not requiring the opening of additional firewall ports.

Hub
A shared transmission media to which devices on a network are interfaced. Ethernet hubs have mostly given way to Ethernet switches.

Huffman Coding
A method of entropy encoding used in lossless data compression where the most frequently occurring values use the shortest codes.

I

IGMP (Internet Group Management Protocol)
Internet Group Management Protocol - IGMP - Host-to-router signaling protocol for IPv4 to report their multicast group memberships to neighboring routers and determine whether group members are present during IP multicasting. Similarly, multicast routers, such as E-Series routers, use IGMP to discover which of their hosts belong to multicast groups and to determine if group members are present.

IGMP Snooping
IGMP snooping, as implied by the name, is a feature that allows a switch to “listen in” for multicast join requests on a network and deliver to end-point network devices when requested. A switch which supports IGMP snooping will not flood all of its ports with multicast traffic. IGMP snooping is supported in Layer 3 switches and some Layer 2 switches.

IGMP Query
IGMP multicast streaming requires that layer 3 routing devices send out IGMP group membership queries. This allows network ports subscribing to a multicast to be identified in group membership tables through snooping by network switches. IGMP snooping will not work without presence of the querying activity.

Image Noise
The random variation of brightness or color information in images produced by the sensor and circuitry of a scanner or digital camera.

Interlace
In TV, each video frame is divided into two fields with one field composed of odd numbered horizontal scan lines and the other composed of even numbered horizontal scan lines. Each field is displayed on an alternating basis.

Interleaved RTSP
Interleaved RTSP is a technique in which an RTP stream is transported within an RTSP session. As RTP streams rely on use of UDP packets, which are unreliable, Interleaved RTSP leverages RTSP’s greater reliability due to its use of TCP transport.

Inter-Frame Coding
A compression technique that spans multiple frames of video and eliminates redundant information between frames.

Internetworking
The practice of connecting networks together using gateways which route packets between the networks.

Intra-Frame Coding
A method of video compression that reduces the information required to represent a single frame.

Intra-prediction
Intra-prediction is an advanced compression technique applied in H.264 which takes advantage of the spatial redundancy within a frame to reduce the amount of data required to encode an I-frame.

IP (Internet Protocol)
Internet Protocol - IP - defines addressing methods and structures for datagram encapsulation, allowing delivery of packets from a source to a destination across an internetwork based purely on addressing. It is the primary protocol that establishes the Internet.

IP Address
A numerical label using the Internet Protocol assigned to devices in a network. The IP address for the source and destination are included in an IP datagram.

IPv4
Internet Protocol version 4. The current version of the Internet Protocol, which is the fundamental protocol on which the Internet is based. It is a connectionless protocol for use on packet-switched Link Layer networks (e.g., Ethernet). It operates on a best effort delivery model, in that it does not guarantee delivery, nor does it assure proper sequencing or avoid duplicate delivery.

IPv6
Internet Protocol version 6. This new Internet Protocol is designed to replace and enhance the present protocol, which is called TCP/IP, or officially IPv4. IPv6 has 128-bit addressing, auto configuration, and new security features and supports real-time communications and multicasting. The primary equipment to apply IPv6 to is routing equipment, not source equipment.

J

JPEG (Joint Photographic Experts Group)
Commonly used method of lossy compression for photographic images using a discrete cosine transfer function. The degree of compression can be adjusted, allowing a selectable tradeoff between storage size and image quality. JPEG typically achieves 10:1 compression with little perceptible loss in image quality. JPEG compression produces blocking artifacts as the compression rate is increased.

JPEG 2000
A wavelet-based image compression standard and coding system. JPEG 2000 provides an increased compression performance over JPEG. An advantage offered by JPEG 2000 is a significant flexibility of the codestream, which allows for representing the image at various resolutions.

Jumbo Frame
Ethernet frames with more than 1500 bytes of payload. Network switches typically process packets with a maximum transfer unit, MTU of 1500 bytes. Use of jumbo packets can increase transmission efficiency by reducing the network transmission overhead used for the Ethernet datagram wrapper, which includes items such as the source and destination address.

L

LAN (Local Area Network)
Local Area Network - LAN - a computer network covering a physical area such as in an office building, a school, or a home. A LAN is useful for sharing resources including files, printers, games, or other applications. A LAN often connects to other LANs and to the Internet or other WAN - Wide Area Network.

Latency
A measure of time delay experienced in a system, the precise definition of which depends on the system and the time being measured. In video processing or encoding products, it is a measure of the amount of time used to process an input signal. In a packet-switched network, it is measured either one-way (the time from the source sending a packet to the destination receiving it), or round-trip (the one-way latency from source to destination plus the one-way latency from the destination back to the source).

Layer 2 Switch
Layer 2 switches support functions of the second layer of the ISO model, and provide hardware switching. They are capable of switching packets between devices physically connected to the switch. A table is built in the switch based on the physical MAC address of the connected devices. A Layer 2 switch does not examine IP address.

Layer 3 Switch
Layer 3 functionality of the third layer of the ISO model is provided by these switches. Layer 3 switches examine network packets and make switching and routing decisions based on information in the Ethernet packets. They are used in networked audio and video network delivery systems and large or complex internetworks, such as the Internet. Layer 3 switches support packet routing, VLANs, IGMP-snooping, and multicast data stream delivery.

Lip Sync
A technical term for matching lip movements seen in a video picture with voice. Audio and video is synchronized when lip sync is maintained.

Lossy Compression
Lossy Compression is a compression method that compresses data to fit within a targeted size. Significant information is lost, but the data remains similar enough to the original data to be useful and fits within a specific communication link or file size.

LPAC (Lossless Predictive Audio Compression)
Lossless Predictive Audio Compression - LPAC - an improved lossless audio compression algorithm developed by Tilman Liebchen, Marcus Purat, and Peter Noll at Institute for Telecommunications, Technical University Berlin (TU Berlin), to compress PCM audio in a lossless manner, unlike conventional audio compression algorithms, which are lossy. It is no longer developed because an advanced version of it has become an official standard under the name of MPEG-4 Audio Lossless Coding.

Luminance
The measurement of the black to white value for a pixel.

M

MAC (Media Access Control)
Media Access Control - MAC - the Media Access Control data communication protocol sub-layer provides addressing and channel access control mechanisms that make it possible for several terminals or network nodes to communicate within a multi-point network, typically a local area network (LAN). Access to the media may be spread out over time, or as in Ethernet, a mechanism is developed which allows random access, but provides a method for re-attempting use of the media if a collision is experienced.

Mathematically Lossless Compression
Allows the exact original data to be reconstructed from the compressed data. Data compacting in mathematically lossless processes is between 2:1 and 3:1. The term lossless is in contrast to lossy compression, which only allows an approximation of the original data to be reconstructed in exchange for better compression rates and smaller file .

Media Player
A software application used for the playback of audio and video files.

Media Server
A media server refers either to a dedicated computer appliance or to specialized application software, ranging from an enterprise class machine providing video on demand, to a small personal computer or NAS (Network Attached Storage) dedicated to storing a variety of digital media such as digital videos/movies, audio/music, or picture files. In streaming media applications, a media server accepts a live stream, encodes the content for single or multiple screen delivery, and delivers multiple unicast or multicast streams. A media server may also convert streaming transport protocols.

Metadata
The term metadata refers to "data about data". It provides information about one or more aspects of its content such as, the means of creation, its purpose, time and date of creation, applicable standards. Metadata included in AV media files often includes the author, title location, date, copyright and licensing information.

M-JPEG
Motion JPEG or M-JPEG video compression applies the discrete cosine transform to each video frame independently. No temporal compression is applied in M-JPEG and no frame interdependence exists as with MPEG compression. Each video frame is encoded as though it is an MPEG I-frame. Editing and random access are easily facilitated in product designs applying M-JPEG.

MPEG (Moving Picture Experts Group)
Moving Pictures Experts Group - MPEG - is a working group of experts that was formed by ISO and IEC. It is comprised of suppliers, users, and designers responsible for developing the motion picture video standard for commonly used audio and video compression and transmission.

MPEG-2
The second generation standard for video compression of audio and video applying the discrete cosine transform. The standard includes a combination of lossy video and audio compression methods which permit storage and transmission of movies using currently available storage media and transmission bandwidth. Commonly used for digital television transmission, DVD, and other similar equipment.

MPEG-2 Transport Stream
MPEG transport stream, also known as MPEG-TS or TS, is a standard format for transmission and storage of audio, video, and Program and System Information Protocol - PSIP - data. TS is commonly used in broadcast, IPTV, cable and professional streaming products to transport MPEG-2 or H.264 video and audio in streaming applications.

MPEG-4
A patented collection of methods defining compression of audio and visual (AV) digital data. Uses of MPEG-4 include compression of AV data for Web (streaming media) and CD distribution, voice (telephone, videophone), and broadcast television applications. MPEG-4 absorbs many of the features of MPEG-1 and MPEG-2 and other related standards, adding new features such as (extended) VRML support for 3D rendering, object-oriented composite files (including audio, video, and VRML objects), support for externally specified Digital Rights Management, and various types of interactivity.

MPLS (Multiprotocol Label Switching)
Multiprotocol Label Switching - MPLS - is a mechanism in high-performance telecommunications networks, which directs and carries data from one network node to the next. MPLS makes it easy to create "virtual links" between distant nodes. It can encapsulate packets of various network protocols.

MTU (Maximum Transfer Unit)
Each network has a maximum transfer unit or MTU, the maximum size for an Ethernet frame payload. Typically the MTU for a network is 1500 bytes. Routers break up data segments into two or more segments if the MTU is smaller than the payload in an Ethernet frame.

Multi-Pass Transform
Multi-pass transforms return to a data set to carry out a process. Multi-pass transforms are often capable of supporting greater compression ratios, but use a greater amount of time to process the data.

Multi-Purpose Transform
A multi-purpose transform is capable of converting more than one type of input format. The PURE3® codec is a multi-purpose transform in respect to its ability to process both video and computer graphic inputs which are different with respect to resolutions, color space, and color information.

Multicast
Multicast addressing is a network technology for the delivery of information to a group of destinations simultaneously using the most efficient strategy to deliver the messages over each link of the network only once, and creating copies only when the links to the multiple destinations split. A single stream is sent from the source to a group of recipients.

N

NAS (Network Attached Storage)
Network Attached Storage. One or more storage devices associated with a single server, which exist as a node on a LAN (Local Area Network).

NAT (Network Address Translation)
Network Address Translation - NAT - method of concealing a set of host addresses on a private network behind a pool of public addresses. It allows conservation of registered IP addresses within private networks and simplifies IP address management tasks through a form of transparent routing, and increases network privacy by hiding internal IP addresses from external networks.

Native Resolution
The native resolution of a LCD, LCoS, or other flat panel display refers to its single fixed resolution. It is the resolution at which an image was originally produced.

NTSC (National Television System Committee)
National Television System Committee - NTSC - the analog television system used in most of North America, South America, Japan, South Korea, Taiwan, Burma, and some Pacific Island nations and territories.

O

Optical Ethernet
An optical connection for delivering Ethernet packets. Ethernet signals have been traditionally interfaced on twisted pair cable. Optical Ethernet connections are used to preserve quality delivering the same signal over a greater distance, and for security concerns.

OSI Open System Interconnection Reference Model
OSI Reference Model is a definition for layered communications and computer network protocol design. It was developed as part of the Open Systems Interconnection (OSI) initiative. The OSI model divides the network architecture into seven layers, starting from the bottom up: Physical, Data Link, Network, Transport, Session, Presentation, and Application Layers.

Out of Order Packet
In computer networking, the delivery of data packets in a different order from which they were sent. Video decoders must account for out of order packets which may be experienced.

Overbooking
In the telecommunications industry, overbooking -- such as in the frame relay world -- means that a telephone company has sold access to too many customers which basically floods the telephone company's lines, resulting in an inability for some customers to use what they purchased.

Overhead
Any data which is transferred on a communication link which is in addition to the content or data that is delivered. In IP networks, overhead includes: addressing, control, routing, redundant, error-checking, and error concealment data.

Overlay Network
An overlay network is a computer network built upon another network. Overlay networks were initially built upon telecommunications networks and now are also built upon IP network services such as MPLS or encrypted and tunneled services over the Internet. Overlay networks connect networks over great distances. They provide a service upon which AV content can be distributed broadly over the Internet.

P

P-Frame
Predictive coded picture. Contains predictive information required to recreate a video frame.

Packet
A block of data that is transmitted over a network in a packet-switched system. A packet is also referred to as a frame or datagram.

Packet Jitter
The term jitter is used as a measure of the variability over time of the packet latency across a network. In real-time applications such as VoIP and video, variation in the rate at which packets in a stream are received that can cause quality degradation. Video decoders must account for jitter, which may be experienced delivering packets across a network.

Packet Loss
Occurs when one or more packets of data traveling across a computer network fail to reach their destination. Packet loss is distinguished as one of the three main error types encountered in digital communications; the other two are bit error and spurious packets caused by noise. Packet loss is typically experienced in the real world as a random burst of packet loss.

PAL (Phase Alternate Line)
Phase Alternate Line - PAL - an analog television encoding system used in broadcast television systems primarily in Europe, Asia, Africa, and Australia.

Pixel
Picture Element. The smallest unit or area of a video screen image that can be turned on or off, or varied in intensity.

Plug-in
A program of data that enhances, or adds to, the operation of a parent program. Software decoders often use a plug-in provided in media players.

Private Network
A communication network owned by one or more firms for their exclusive use.

Pro-MPEG Forum
An association of broadcasters, program makers, equipment manufacturers, and component suppliers with interests in realizing the interoperability of professional television equipment, according to the implementation requirements of broadcasters and other end-users. The Forum has been in existence for approximately eight years and has over 130 members.

Progressive
A method for displaying, storing or transmitting moving images in which all the lines of each frame are drawn in sequence.

Public Network
A network established and operated by a telecommunications provider, for specific purpose of providing data transmission services for the public. The Internet is a public network.

Pull Streaming
Pull streaming describes network communication where the initial request for data or streaming session originates from the client, or decoder, and then is responded to by the server. The reverse is known as push streaming, where the encoding device initiates, or pushes, data to clients.

PURE3® Codec
A codec that is capable of encoding and streaming both video and computer graphic inputs and a wide variety of resolutions, preserving equal quality for both signal formats. It preserves a balance between three performance factors: low latency, low bandwidth, and high image quality. The PURE3® Codec has been optimized for use on IP networks that are acknowledged to be lossy. The codec includes an error concealment system which is highly resistant to network errors without using forward error correction.

Push Streaming
Push streaming describes network communication where the streaming session is managed centrally from the encoding device. It is contrasted with pull streaming, wherein the request for the transmission is initiated by the decoding software or device. In push streaming applications the encoder defines the streaming destination. Such services are often based on preferences set in advance via a subscription model.

Q

QoS (Quality of Service)
Quality of Service - QoS describes the performance, such as transmission rates and error rates, of a communications channel or system. A suite of features that configure queuing and scheduling on the forwarding path of an E-Series router. QoS provides a level of predictability and control beyond the best-effort delivery that the router provides by default. (Best-effort service provides packet transmission with no assurance of reliability, delay, jitter, or throughput.) See also CoS.

Quantization
The procedure of converting a signal from one set of defined values to a new discrete set of values. An analog to digital conversion quantizes a continuous or infinite set of values to a smaller set of discrete, digital values.

R

Random Error
Errors in measurement that lead to measured values being inconsistent when repeated measures of a constant attribute or quantity are taken.

Real Images
Collected from the real world through image sensors. Video collected from film or electronic cameras can be considered real images.

Real-time
A system is said to be real-time if the operation delivers a correct value in the time and frequency in which it is required. The video system applied in North America, NTSC, requires a real-time system capable of delivering 30 frames per second.

Redundancy
Repeated data or equipment that provides a backup if the primary data or equipment fails.

Refresh Rate
Also called “Vertical Scan Frequency” or “Vertical Scan Rate”. The number of times in a second that display hardware draws a new video frame.

Router
A network device that forwards packets from one network to another. Routing is a Layer 3 function. Routers forward packets based on programmed or “learned” routing tables. Each incoming network packet is examined and a decision is made where to forward it. The destination address in the packets determines the port where outgoing packets are needed. In large-scale enterprise routers, the current traffic load, congestion, line costs, and other factors determine which line to forward to. Routers enable internetworking or the connection of many networks.

RTP (Real Time Transport Protocol)
Real Time Transport Protocol - RTP - an IETF standard for streaming real-time multimedia over IP in packets.

RTSP (Real Time Streaming Protocol
Real Time Streaming Protocol - RTSP - a network control protocol designed for use in audio visual and communications systems to control streaming media.

Run Length Encoding
Simple form of data compression in which runs of data are stored as a single data value and count, rather than as the original sequence. This is most useful on data that contains repetitive information.

S

Scalability
The property of a system, a network, or a process, which indicates its ability to handle growing amounts of work in a graceful manner with a limit that is unlikely to be encountered.

Scaler
A device for converting video signals from one size or resolution to another: usually “upscaling” or “upconverting” a video signal from a low resolution (e.g. standard definition) to one of higher resolution (e.g. high definition television).

Scaling
A conversion of a video or computer graphic signal from a starting resolution to a new resolution. Scaling from one resolution to another is typically done to optimize the signal for input to an image processor, transmission path or to improve its quality when presented on a particular display.

SDH (Synchronous Digital Hierarchy)
Synchronous Digital Hierarchy - SDH - see SONET.

SDI (Serial Digital Interface)
Serial Digital Interface - SDI - standard definition video is carried on this 270 Mbps data transfer rate. Video pixels are characterized with a 10-bit depth and 4:2:2 color quantization. Ancillary data is included on this interface and typically includes audio or other metadata. Up to eight audio channels can be transmitted. Audio is organized into blocks of four stereo pairs.

SDSL (Symmetrical Digital Subscriber Line)
Symmetrical Digital Subscriber Line - SDSL - offers bandwidth of up to 2.3 Mbps upstream and downstream over a single twisted pair copper phone line, over distances up to about 10,000 feet on an unrepeatered basis.

SFP (Small Form-factor Pluggable)
Small Form-factor Pluggable - SFP - the SFP is an interface used in fiber optic connections for direct signal connections or packet switched networks.

Signal Noise
A random fluctuation in an electrical signal, a characteristic of all electronic circuits.

Silverlight
Silverlight is an application framework created by Microsoft for the creation and presentation of Internet streaming media, multimedia, graphics, animation and audio. Its run-time environment is compatible with Microsoft Smooth Streaming and is available as a plug-in for web browsers running Microsoft Windows and Mac OS X. Silverlight supports H.264 video, Advanced Audio Coding, WMV, WMA, MP3, and VC-1 content across all supported browsers without requiring Windows Media Player.

Single Pass Transform
Transformation process that is carried out making only one examination of a data set. A single pass transform is required to maintain a low delay.

SLA (Service Level Agreement)
Service Level Agreement - SLA - an agreement between a network service provider and the user defining an established set of metrics to measure the service delivered relative to the service delivered. An SLA typically identifies the bandwidth delivered, Quality of Service, and service response time.

Smooth Streaming
Microsoft Smooth Streaming is a proprietary IIS Media Services extension that enables adaptive bit rate streaming of media to clients over HTTP. Smooth Streaming is supported with Silverlight on Windows Phone 7, and can be ported for operation on other client operating systems, such as Apple iOS, Android, and Linux.

Software Decoder
A software decoder provides a means to decode audio/video streams in software without requiring use of a dedicated hardware appliance. Software decoders are typically used on PCs using a browser page, media player, or special purpose application.

SONET (Synchronous Optical Networking)
Synchronous Optical Networking - SONET - a standardized multiplexing protocol that transfers multiple digital bit streams over optical fiber using lasers or light-emitting diodes (LEDs).

Spatial Resolution
A measurement of the resolution in a single frame of video. The horizontal resolution multiplied by the vertical resolution.

Spanning Tree
IEEE 802.1d. is a protocol that allows networks to prevent loops, or multiple paths, from developing between a source and a destination. Network routers communicate with each other using spanning tree protocol to prevent traffic from reaching unnecessary destinations. Spanning tree and other routing protocols prevent multicast video traffic from flooding networks with unnecessary, disruptive traffic.

Sub-frame Compression
Compression that is not carried out on an entire frame of video, but only a part of a video frame.

Subnet Mask
Number of bits of the network address used to separate the network information from the host information in a Class A, Class B, or Class C IP address, allowing the creation of subnetworks. In binary notation, a series of 1s followed by a series of contiguous 0s. The 1s represent the network number; the 0s represent the host number. Use of masks can divide networks into subnetworks by extending the network portion of the address into the host portion. Subnetting increases the number of subnetworks and reduces the number of hosts.

SVGA (Super VGA)
Super VGA - SVGA - a screen resolution of 800x600 pixels and above.

Switch
A device that cross-connects network devices. Today, switches are broadly deployed on modern industrial and consumer networks. Switching is a Layer 2 function. Ethernet frames are delivered between MAC addresses connected to network switches.

Switched Fabric
A network topology where network nodes connect with each other via one or more network switches (particularly via crossbar switches, hence the name). The term is in contrast to a broadcast medium, such as early forms of Ethernet.

SXGA (Super XGA)
Super XGA - SXGA - a standard screen resolution of 1280x1024 pixels.

SXGA+ (Super Extended Graphics Array Plus)
Super Extended Graphics Array Plus - SXGA+ - Commonly used on 14 inch or 15 inch laptop LCD screens with a resolution of 1400 × 1050 pixels.

Symmetrical Processing
Two processes are symmetric if the input process is of equal magnitude and complexity to the output process. The encoding and decoding processes in the PURE3® codec are symmetric.

Synchronization
Timekeeping that requires the coordination of events to operate a system in unison. Synchronization in video systems can refer to a number of items. Lip-sync is the synchronization of audio and video. Genlock refers to alignment of vertical sync in video signals. Frame-sync or framelock refers to the alignment of video frames in systems with multiple video sources.

Synthetic Images
Synthetic images are produced in artificial processes, for example in video processing or computing systems.

T

Telepresence
A set of technologies that allows individuals to feel as if they were present, to give the appearance that they were present, or to have an effect, at a location other than their true location. Telepresence solutions include the delivery of audio, video, data, and computer graphic information over IP networks using video over IP encoders and decoders.

TCP (Transmission Control Protocol)
A connection-oriented protocol designed to provide a reliable end-to-end data delivery over an unreliable internetwork.

TCP/IP Model
A set of communications protocols used for the Internet and other similar networks. It is named for two of the most important protocols in it: the Transmission Control Protocol (TCP) and the Internet Protocol (IP). The eight functions of the OSI model have been combined into only four layers in the TCP/IP model.

Temporal Resolution
The sampling frequency applied in a system. In video streaming applications, if an encoder accepts a video source at 60 frames per second, but streams video that updates the image six times per second, 90 percent of the temporal resolution has been removed.

Thin Client
A computer or a computer program that depends heavily on some other computer (its server) to fulfil its traditional computational roles.

Transform
A method applied to convert a data set from one domain to another. The rationale for transforming the data into a new domain is typically to make handling and processing the information easier. One common example is the RGB to YUV color space transformation. Imagery collected from the real-world using sensors is done in an RGB color space. The RGB information is then transformed to a component YUV domain allowing independent processing of luminance and color information.

Transformation
A change or alteration. In the context of still image compression, a picture frame is input as a fixed resolution of rows and columns of pixels and transformed into a frequency domain applying the Discrete-Cosine Transform.

Transcode
A conversion from one format to another, typically to make the content compatible with specific devices or applications. The transcoding of AV streaming content is usually performed to produce material with a different resolution, most frequently a lower resolution and bit rate from the original. Transcoding may include transmuxing.

Transmux
A change to the streaming transport protocol or container format. Transmuxing may or may not accompany a change the underlying audio and video data. For example, H.264 encoded video and AAC encoded audio carried within Real-time Transport Protocol - RTP may be transmuxed so that it is contained within the MPEG-2 Transport Stream, making it compatible with set-top box decoders.

Transrate
To convert to a different data rate or bit rate using the same streaming transport protocol or container format.

Transsize
Convert to a different resolution using the same streaming transport protocol or container format. Transsizing is used when the output resolution differs from the original content resolution.

Transport Stream
A defined package for delivering data. Transport Streams are multiplexes of audio, video, and other content that are usually broadcast over-the-air, although they can be streamed over IP networks too.

TTL (Time To Live)
Time To Live - TTL - multicast streaming traffic is typically programmed with a TTL value indicating the number of router hops that are permissible for the packet.

U

UDP (User Datagram Protocol)
A connectionless protocol providing “best effort” delivery of packets across networks. UDP is frequently used in real-time streaming applications where best-effort delivery is acceptable and the network devices and applications manage data flow control and errors.

Ultra HD
Ultra high definition television - UHDTV - is also frequently referred to as 4K or 4K UHD, is a digital video format defined and approved by the International Telecommunication Union (ITU). The Consumer Electronics Association (CEA) announced that the official term “Ultra HD” would be used for any display with a 16 x 9 ratio with at least 1 digital input cable carrying a minimum resolution of 3,840 x 2,160 square pixels. 4K Ultra HD (2160p) has a resolution of 3840 × 2160 (8.3 megapixels), which is roughly equivalent to 4K cinema or 4 times the number of pixels in Full HD format (1080p).

Unicast
The sending of messages to a single network destination host on a packet switching network. N clients of a unicast stream will require that a server produces N streams of unicast data.

UXGA (Ultra Extended Graphics Array)
Ultra Extended Graphics Array - UXGA - a screen resolution of 1600x1200 pixels.

V

Variable Bit Rate (VBR)
Varies the amount of output data per time segment. VBR allows a higher bit rate or storage space to be allocated to more complex segments of video and a lower bit rate to be allocated to less complex segments.

VANC (Vertical Ancillary Data Space)
Vertical Ancillary Data Space - VANC - the blanking interval within an SDI signal used to carry non-display data such as timecode and closed captioning data.

Vertical Frequency
See "Refresh Rate."

VGA (Video Graphics Array)
Video Graphics Array - VGA - a widely used analog interface between a computer and monitor that uses a 15-pin plug and socket. The original VGA resolution was 640x480 pixels.

Video
A format for transmitting and storing moving pictures. Video is transmitted and stored in various analog and digital physical formats.

Visually Lossless Compression
Allows the reproduced image to appear to human vision to be identical to the original image.

VLAN (Virtual LAN)
Virtual LAN - VLAN - a group of devices on a network with a common set of requirements that communicate as if they were attached to the same broadcast domain, regardless of their physical location. VLAN and their network traffic will be segmented from other devices that may be connected to the same physical system. A VLAN is a Layer 3 network function.

VLC Media Player
VLC media player is a portable, free and open-source, cross-platform media player and streaming media server written by the VideoLAN project. VLC media player supports many audio and video compression methods and file formats, including DVD-Video, video CD and streaming protocols. It is able to stream over computer network and to transcode multimedia files.

VOD (Video on Demand)
Video on Demand - VOD - unicast streaming video offered by service providers that enables the reception of an isolated video session per user with rewind, pause, and similar VCR-like capabilities.

VPN (Virtual Private Network)
Virtual Private Network - VPN - a method of providing a private network connection via a secure communications tunnel over a public network such as the Internet. VPNs maintain privacy applying tunneling protocol, encryption, and security procedures.

W

WAN (Wide Area Network)
Wide Area Network - WAN - a computer network that covers a broad area such as a link across a metropolitan, regional, or national boundary.

WUXGA (Wide Ultra Extended Graphics Array)
Widescreen Ultra Extended Graphics Array - WUXGA - a screen resolution of 1920x1200 pixels.

X

XGA (Extended Graphics Array)
Extended Graphics Array - XGA - a screen resolution of 1024x768 pixels.

Y

YUV
A component color system that organizes the black and white luminance information separately from the color or chrominance information. YPbPr and YCbCr are also component color systems. YPbPr is the analog version of the YCbCr color space; the two are numerically equivalent.

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