THE BOOK

An Engineers Guide to the Digital Transition

An NVISION® Guide

Acknowledgments

This book is a distillation of the experiences and writings of many prominent research engineers and system designers. We would like to thank the host of chief engineers and industry professionals, have been of tremendous help in identifying the difficulties and special needs of designing large hybrid television facilities. We would also like to thank Chuck Meyer, VP of engineering at NVISION, for his many hours of work in preparing the Digital Audio sections of this book. Mention should also be made of John Watkinson who's prolific writings have provided us with a wealth of reference materials.

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Introduction

Since our beginnings in 1989, we at NVISION have been committed to helping video system designers to fully understand the nuances and likely pitfalls that would be encountered as our industry embarked upon its journey from analog to digital. Naturally, our first task was to identify the areas of greatest danger by spending many hours of discussion and investigation with those Broadcasters and Post Production Company's who had taken the first steps along the path.

In 1994 we published our first catalog that contained application and engineering notes for the incorporation of digital audio within video system design. This publication included our product literature, but was heavily biased toward technical design notes rather than the normal glossy sales presentation.

The response from its recipients was overwhelming, so in 1995 an updated version of the catalog was published following the introduction of new products and requests for more information from our customers. Both of these catalogs can be found on the shelves of engineering offices all around the world, normally 'dog eared' and definitely well read. To further aid our clients, in 1995 we commissioned the well known author, John Watkinson, to write a 'How To' book entitled 'The Video Engineers Guide to Digital Audio', which has been equally well received.

In talking to the readers of these publications, it was discovered that sharing this information has helped many avoid the trauma and expense that had inevitably been experienced by the earlier pioneers and has also helped combat the ‘three wise monkey’ (hear no evil, see no evil, speak no evil) attitude frequently adopted by major video equipment manufacturers. We realize that ‘we’ (you the reader and us the manufacturer) are all in this together. You need products to solve problems and to make interconnection easier and we need to provide them at a price you can afford. If we can achieve this, then ‘we’ all succeed.

Although digital video and audio equipment has been available for more than a decade, it has only recently become the standard rather than the exception. In fact it is now becoming very difficult to purchase analog equipment. Therefore making the transition to digital equipment and techniques is inevitable. It need not be painful or more expensive, but it is necessary to learn from the errors and misconceptions of the past.

Therefore, by popular demand, we have created this latest offering. It contains all of the application notes that Chuck Meyer wrote for the original catalogs (updated where necessary), as well as information on video formats, standards and interconnection considerations. It is heavily biased towards digital audio, as this still seems to be the area of greatest uncertainty in mixed format or all digital facility design. Naturally, it also promotes our own products, but hopefully you’ll forgive us for our tendency for promoting the things that we do well.

We trust that you will find this book helpful, and we would be grateful for any comment regarding its content or suggestions for subject matter that you would like to see in future publications.

Nigel Spratling

Vice President - Marketing

Chapter 1. Making the Transition

As most people who have recently purchased equipment have discovered, it is becoming more difficult to find analog equipment. Manufacturers find it easier to build digital products, as good analog circuit design and performance is somewhat of a ‘black art’. Analog equipment manufacture is more difficult than digital. It is frequently necessary to hand select components or specify high tolerance parts to ensure that design performance is maintained. Of course, good digital equipment design is also an art, but manufacturing processes are greatly improved when no analog circuits are involved. Digital equipment can be produced with consistent quality and is generally far more reliable than its analog counterpart.

As a result, analog interfaces on VCR’s, Effects Systems and other signal processing devices are now becoming expensive optional extras. However, most television broadcast facilities currently maintain analog interconnection for 90% of the existing equipment. Therefore, for some time to come, analog and digital interfaces must coexist in established facilities. New system design must of course accommodate product that only has analog I/O’s, but by utilizing digital interconnection, reliability can be improved while keeping costs to a minimum.

The net result of these facts, is that engineers in every Broadcast station, Production facility or Post house, are actively involved in making the transition. Those of us who have designed, engineered and maintained analog systems for years, need to be aware of the rules that have changed since the introduction of digital formats.

Video system design requirements change very little. Although more care needs to be taken of cable quality and length when transporting digital video, timing considerations remain much the same. In an analog video system, audio has no specific timing requirement except the need to maintain 'lip sync'. Digital audio, like video, is a stream of data that must contain timing information so that it can be correctly received at its destination. This requirement for correctly timed audio data is new to video system design and adds a level of complexity that needs to be well understood.

Designers should not be disturbed by this fact. Digital audio does not necessarily cost any more, in fact if no analog is involved, it probably costs less. In general, digital audio has better quality, is more consistent and reliable. However, there are several issues that can cause engineers much consternation, late nights and possible changes in vocation. To help you avoid this, we have dedicated the greater portion of 'The Book' to this subject.

So you have to make the transition, but what's the right way? Scrap all of your analog DA's, Routers, Edit Suites and Mixers and replace them with new digital equivalents? Well, unless the current facility is quite small and you are planning to significantly increase its size, this is probably not a financially acceptable proposition. At some point those analog components will need to be replaced, therefore this needs to be considered when designing the inclusion of digital components.

The Digital Island

One of the most common short term methods of introducing digital equipment, has been to build small digital suites with analog I/O's via appropriate A to D and D to A converters. This method obviously allows existing analog equipment to be fully utilized, but care should be taken with the quality of conversion equipment chosen as the conversion processes are most likely to introduce unwanted artifacts into the final result. Ideally, once the signals have been passed to the digital suite, they should be processed entirely in the digital domain and not converted back to analog until the work is completed. Each A to D / D to A pass will multiply any error introduced by conversion. If several conversions occur, the resultant product could have a lower technical quality than required.

Many digital products are available with analog I/O's, but often the A to D / D to A converter circuits are the weakest link. Outboard converters often offer better performance and lower cost. Even when the costs are a little higher, the performance benefits can be significant enough to warrant the expense.

The Hybrid System

The next step on from the Digital Island is to introduce digital interconnection. As a cost exercise, A to D / D to A converter costs should be compared to the cost savings of digital DA's and routing equipment. The average cost of good quality A to D / D to A converters for audio and video is in the $5-7K range. Compare this to the average crosspoint cost of a 32^2 digital video/audio routing system at about $1.5K. Naturally, you need enough digital signals to warrant this router size and you will still need the use of some converters. Most new digital routers can interface to existing control systems (depending on age and control port availability), so you should not need to replace your current router control.

However, by investing a little more at the outset, the system will maintain a higher overall performance and have room for expansion at a considerable cost advantage.

The All Digital Facility

If a new facility or a major reconstruction is being planned, it should have all digital interconnect. A common misconception is that the routing, DA's and other terminal equipment required, cost more than the analog equivalent. This is not the case. Good digital equipment can be built more easily and with greater reliability. Initial prices may be on a par but maintenance costs should be far less.

Digital television, computer and communication systems are rapidly merging. Over the next few years, we will see some radical changes in the hardware and methodology we employ to perform our tasks. The common denominator will be the interconnection method. As you will see in the following chapters, we have tried to identify standards and compatibility issues, as well as provide useful information to assist in designing the digital interconnect.

Chapter 2. Designing a Digital Audio System

Providing a reliable interconnection for transferring analog signals between two pieces of equipment requires more than a length of copper and two mating connectors. Digital audio connections are not any different. To maximize the benefits offered by digital technology, an understanding of new procedures and formats is required. In addition, there are two interconnect choices available to the user, and the criteria, which needs to be considered in deciding which to use are defined and explained below. Signal impedance matching and careful cable selection are two key areas which must be understood. Digital audio data formats will also have an impact on the design process. Finally, details for grounding, shielding and distribution of common signals form the basis for actually wiring up a facility.

A Brief History of Digital Formats

A large number of digital audio formats have been used to some degree over the past decade. The four primary signal formats likely to be encountered in the video environment are MADI; the Multi-Channel Audio Digital Interconnect developed as an Audio Engineering Society (AES) standard for the interconnect of 56 channel digital audio between consoles and multi-track recorders, SDIF-2; the Sony Digital Interface Format developed for multi-track recorder and CD mastering equipment, SPDIF; the Sony Philips Digital Interface developed for serial transmission of digital audio information between consumer products and AES3. AES3 has become the dominant standard for the interconnection of digital audio signals between equipment; audio and video. The AES3 standard is jointly supported by both the AES and EBU societies, and is often referred to as the AES/EBU standard for digital audio. The standard defines the baseband data format for two channels of audio and respective overhead information, the transmission data format and the electrical interface for the signal. The AES3 standard was first ratified and published in 1985 and was subsequently enhanced and amended in 1992. Officially, the correct title is AES3-1992, ANSI S4.40-1992.

Historically, SDIF-2 was the first widely used digital audio format. An SDIF-2 interface consists of 3 coaxial cables; one each for left channel data, right channel data and a timing signal. Multi-track recorders feature a balanced SDIF-2 interface using twisted pair ribbon cable and 50 pin D-type connector. The timing signal is often referred to as the SDIF-2 word clock, or word clock for short. It is a square wave signal oscillating at the digital audio sample rate; the rate at which analog signals are sampled for conversion into a digital format. Word clock is still very popular for timing in audio only facilities. AES3 is the most popular format in video facilities. Digital audio program distribution is simplified with the AES3 format. First, both left and right channel audio data are placed into one serial data stream, left data first then right. Second, data is coded for transmission into a bi-phase signal, a self-clocking data format. With these enhancements, the AES3 signal transmits 2 channels of audio with timing, as one balanced signal, over one twisted pair cable. Digital audio program production typically requires independent channels of audio. For this reason, the AES3 format allows the 2 channels of audio data to be monaural. Digital audio production mixers then break the AES3 signal apart into two separate channels of audio before the mix. After production, the material is formatted into the desired number of AES3 signals with the correct channel assignments for distribution. Digital audio production work is still carried out with the unbalanced SDIF-2 format in many audio only facilities.

It is important to note that the AES3 format is intended to be independent of the audio conversion sample rate. However, the net data rate is exactly sixty-four times the sample rate. Since 48 kHz is the most frequently used sample rate in the video environment, the most frequently encountered bit rate for AES3 data is 3.072 Mb/s, (megabits per second). This is clearly a much larger bandwidth signal than traditional analog audio and failure to recognize this fact can lead to problems which are discussed later in this paper and in more detail in Chapter 3. Video engineers recognized that the AES3 signal had a similar bandwidth to analog video and pushed for standardization of a low level voltage signal format for coaxial AES3 data transmission. In fact, there are two nearly identical proposed guidelines developed by the AES and the SMPTE to transmit AES3 formatted data in single ended coaxial cable.

AES3-1992 for Twisted Pair

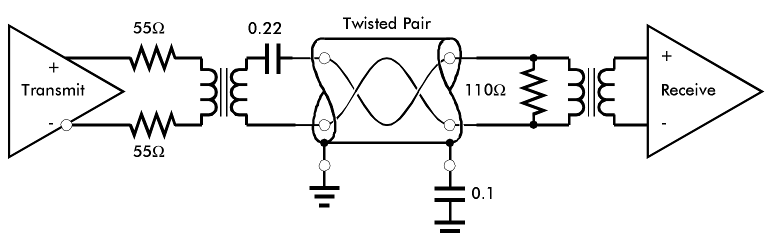


Figure 2-1: Recommended AES3 Interconnect Circuitry

The AES3 twisted pair interconnect is well defined in the AES3 -1992 standard document. The signal is bi-phase coded, transformer coupled and transmitted in a balanced format, on twisted pair copper. The output level may be 2 to 10 volts peak to peak with a source impedance of 110 ohms. The receiver should have a sensitivity of at least 200 mV and an input impedance of 110 ohms. The interconnecting cable should also exhibit a nominal 110 ohm characteristic impedance. The standard connector is specified to be the XLR type. A typical AES3 interconnect and electrical path are shown in figure 2-1. Please note the shield bypass capacitor at the receiver. This is recommended for increased suppression of high frequency emissions and is not part of the AES3 specification.

AES3-ID for Coaxial Cable

The AES3 and SMPTE committees have established electrical interface guidelines for the transmission of AES3 data on coaxial cable, sometimes referred to as AES3-ID. The interface is single ended. The signal level is 1.0 volt, +/- 20 percent, peak to peak, when terminated with 75 ohms. The source impedance is also 75 ohms. It is not required that the signal be transformer coupled, but most implementations are adaptations of existing AES3 circuits so the transformer remains. This interface format is perceived by many video engineers as offering greater compatibility in their operational environment. The AES3-ID and SMPTE guidelines specify the BNC connector as standard. A typical AES3-ID interconnect and electrical path are shown in figure 2-2. Please note the shield bypass capacitor at the receiver. This is recommended for increased suppression of high frequency emissions and is not part of the AES3-ID specification.

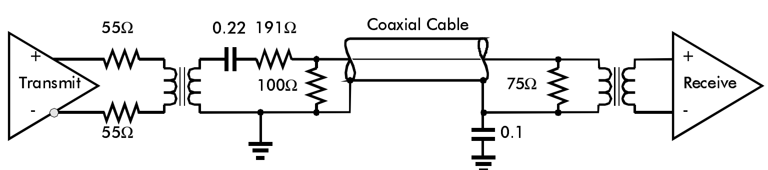


Figure 2-2: Recommended AES3-ID Interconnect Circuitry

As mentioned, these guidelines were developed at the request of users who wished to use video equipment to route and distribute digital audio signals. Unfortunately, the bandwidth of the AES3 signal can exceed that of many video DAs and routers. In particular, the high frequency energy generated by the fast digital edges of the AES3 signal can lead to ringing, oscillation and crosstalk in analog video equipment. These fast edges are often slew rate limited by slow analog DAs or other video distribution equipment. These distortion effects may lead to jitter generation at later stages in the signal path. Careful consideration should be given to the use of video equipment for AES3 signal distribution. Equipment designed for the AES3 standard and the AES3-ID and SMPTE guidelines should provide a clean, artifact free signal. For more details, see Chapter 3.

Suggested Considerations for Choice of AES3 Interconnect

An easy procedure to evaluate which interface type to use follows.

First, consider cable costs. In new facilities, this is easy. Determine the number of feet of each type required and get competitive bids. In existing facilities, twisted pair and coaxial cable may already be installed in floors, ducts and ceilings. Analog audio cable is not recommended for digital applications, but coaxial cable is and may provide a great savings. The costs of cable removal, replacement and downtime must all be considered in addition to the price of the cable itself. High quality digital audio twisted pair cable varies greatly in price, however the typical range is from one-fifth to three-fourths the cost of high quality video coax.

Second, include cable termination costs. BNC connections are arguably easier to make and provide higher density packaging than XLR connections. Router and terminal equipment with BNC connector options provide attractive cost benefits for a facility. Be sure to look for this option in product data sheets.

Third, the cost of supporting existing equipment must be considered. Since XLR connectors are standard, virtually all professional audio and video equipment uses them. If the BNC option is considered, include the costs of converting all existing XLR and twisted pair equivalent connections to the BNC format. Quotes for this equipment, which is usually supplied on a per connector basis, will be required. Be sure to specify the 1 volt operation level per AES3-ID or SMPTE guidelines since more than one voltage option is available for these converters.

Fourth, evaluate your equipment needs and the ability of existing analog equipment to truly meet the specifications required for accurate transmission of digital signals. Compare operational and maintenance costs also. As an example, an inexpensive coaxial interface digital AES3 router designed specifically for digital audio data, requires considerably less space and power than an existing analog video router, it will "drop in" to the existing pre-cabled facility and provide both reduced operational costs and peace of mind.

Electrical Properties of the AES3 Signal

It is important to treat digital audio as a high frequency signal. It's 3.072 Mb/s data rate requires a bandwidth similar to that of analog video. Digital audio signals should be treated with the same care in cabling and installation. To this end, a well executed interconnect in either interface format will have matched source, destination and cable impedance. Unfortunately, equipment manufactured to the AES3 standard prior to 1992 violated this rule. The standard specified a 110 ohm source and a 250 ohm load resistance. Fortunately, this can easily be corrected. Equally fortunate is the fact that the new 1992 AES3 Standard replaces the mismatched 250 ohms with a matched 110 ohm load. Chapter 3. provides the exact remedy.

Cable Types

Cable selection is important for AES3 applications. For coaxial use, select a good, 75 ohm characteristic impedance cable. Any cable which provides acceptable analog video transmission performance should work. Many twisted pair options are available which meet the exact 110 ohm impedance requirement. If one of these is not available, or does not meet your budget, consider a good quality data cable. A cable exhibiting an impedance value within ten percent of 110 ohms and a low capacitance per foot rating, such as 12 or 13 pF/ft, will work quite well. More detail can be found in Chapter 3.. Using matched impedance and good quality cable, transmission distances of up to 1000 feet can be achieved for either format without equalization.

Signal Distribution

A common signal often needs to be distributed throughout a plant. Typical examples are synchronization, digital test tone and digital quiet. These signals are discussed later in this chapter and in Chapter 15. This process requires strict attention to detail since this is a high frequency signal. Typically, one or more distribution amplifiers are used in a tree topology to insure network reliability and constant signal phase between various pieces of equipment. A conceptual picture is shown in figure 2-3. Chapter 3 describes AES3 signal distribution with loop through techniques and when phase is important. It also discusses network reliability. While not defined as part of AES3 standard practice, loop-thru topologies work if good transmission line techniques are practiced.

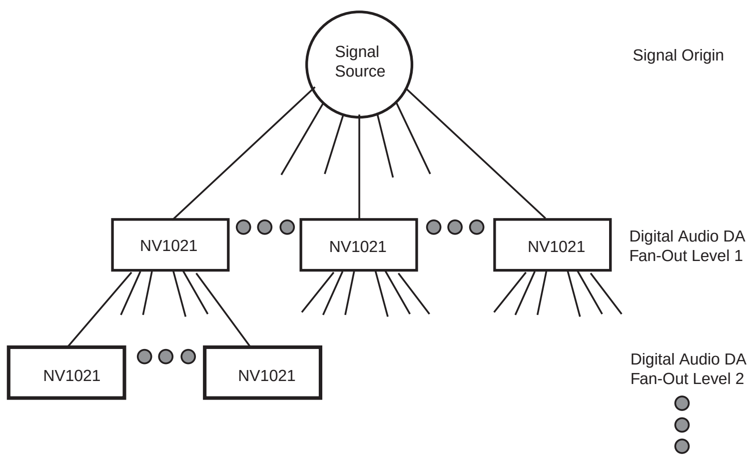


Figure 2-3: Recommended Distribution Topology for AES3 Signals

Details for Shield and Ground Connections

Shield and ground connections are still important for data signals. Figures 1 and 2 demonstrate a preferred technique for connecting the receive cable shield in order to minimize signal to noise, and RF emissions without creating low frequency or DC ground loops which cause various ills ranging from HUM to fires.

Other Standards to Be Aware of and to Anticipate

AES 10, more popularly known as MADI; the Multi-Channel Audio Digital Interface, and AES 11; Digital Audio Synchronization, are two standards increasing in importance. AES10 defines a serial format for the transmission of 56 channels of digital audio data. MADI is intended for local area connection of two pieces of equipment with either fiber or coax. It is not a telecommunications compatible standard. AES11 is the synchronization standard. It defines jitter and long term stability for a digital audio reference signal. Proposed guidelines recommending a fixed phase relationship for audio and video reference signals may be included as part of AES 11. As of this writing, PAL and AES3 signals will be phased so that AES3 block is aligned to the start of PAL line 1 within +/- 5% of an AES3 frame. An absolute, operationally convenient AES3 relationship to NTSC is much more difficult to define because of the 1000/1001 ratio in the NTSC frame rate. Finally, the popularity of low cost 4 and 8 channel digital audio recorders will likely result in an intermediate standard with a channel count between that of AES3 and MADI.

A Design Strategy for Plant Interconnect

The above information can be put to use immediately. Whether designing a new facility, or refurbishing an old one, an interconnect strategy is crucial to the success of the final design.

- Choose a common plant standard for interconnect. AES3 is the obvious choice, however in some audio only facilities SDIF-2 may still be preferred. Remember that any material not originated in the standard format must be converted.

- Choose an electrical interface, BNC or Twisted Pair. This is thoroughly discussed above.

- Select cable. Consider cost per foot, cross-section diameter, ease of termination, existing cable and associated labor costs.

- Consider the need for multi-channel audio interconnect between devices such as multi-track recorders and consoles. Also, consider the interconnect between facilities. Local transmission of digital audio may be less expensive than repetitive equipment or delivery services.

Synchronization of Digital Audio Facilities

Virtually all pops and clicks which plague digital audio production are eliminated when equipment is synchronized. In fact, the only way to achieve direct digital audio transfers between machines without pops and clicks or the expense of sample rate converters, is to synchronize them to a common reference. No digital audio transfer can be completed if the source and destination equipment are not locked. Any digital plant design, must include a strategy for synchronization of all equipment. Other benefits are derived from this approach. A common reference removes the effects and idiosyncrasies of poorly designed data and clock recovery circuits. Configuration time is decreased, and machine dependent operational considerations are removed, reducing the number of red herrings and the amount of wasted maintenance time. If considered early in the design process, the costs of synchronization are small. When implemented as an afterthought, the costs can be prohibitive, both in time and materials.

A digital audio signal such as an AES3 data stream or an SDIF-2 word clock may serve as a reference. NTSC or PAL video signals work equally well. These sync signals can be used in a local island, a suite or across an entire facility. Large facilities will most likely benefit from an integrated audio video synchronization network. Including audio in this network is a new concept; previously not implemented for analog audio.

This section describes various techniques for synchronizing islands, suites or layers and plants. It will define and discuss frequency only synchronization as well as combined phase and frequency synchronization and when to be concerned about each. Some equipment does not allow for external synchronization. This section describes two methods for integrating this equipment into a synchronized facility. The importance of phasing audio and video signals is also discussed.

Frequency Synchronization

All 48 kHz oscillators are not created equal. Each is specified to oscillate at 48 kHz within some tolerance, typically +/- 25 parts per million or +/- 1.2 Hz for professional audio equipment. Now consider NTSC and PAL digital video tape recorders. When digital audio material is transferred from one machine to another, pops and clicks occur regularly even though both recorders nominally produce 48 kHz equivalent sample rates. The PAL and NTSC video timing references are not locked to each other. If the PAL and NTSC signals could be locked to a common clock, pops and clicks disappear. This is the answer. Synchronize all equipment to a common reference. PAL and NTSC video plants have been synchronized within common video formats, for years. It is required for color accurate editing, among other things. Audio has been without this burden until now. The emergence of digital audio as a primary format requires that all equipment must be locked to a common clock. NTSC, PAL and digital audio equipment must be synchronized together.

Fortunately, nearly all professional video equipment generates a 48 kHz digital audio sample rate when locked to a standard video reference. NTSC equipment operating at 59.94 fields per second and PAL machines operating at 50 fields per second both output 48 kHz audio. The chart shown in figure 2-4 illustrates that PAL and NTSC are not the only signals which can be locked to a common clock. Film and CD players can be locked together as well. In fact almost any piece of equipment can be locked to this timing chain. Once locked, digital audio material can be transferred between equipment of nearly any video standard and format transparently. This is how digital technology is intended to work. This is why a common time base is required. Chapter 15 provides more details of this process.

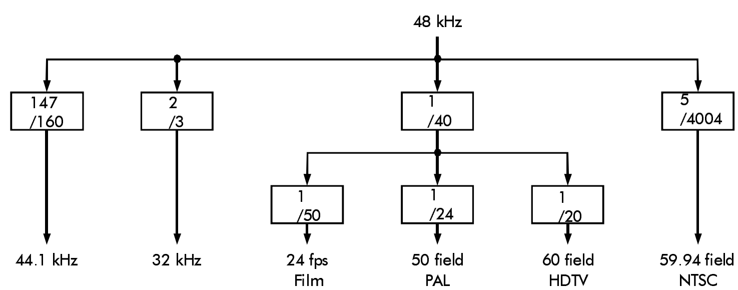


Figure 2-4: Integer relationships for audio and video frequencies

Phase Synchronization

Digital audio reference signals must provide phase and frequency information. While a video signal can provide for frequency synchronization, it can not always provide for phase accurate audio referencing. Phase alignment is extremely important for audio particularly when the analog signal is converted to digital. Any audio processing and recording equipment will force the alignment of all AES3 inputs to a common AES3 frame phase. Any difference in the frame phase of analog to digital converters will generate a proportional phase error between the audio signals when an AES3 frame phase alignment is executed. A quick calculation reveals that if a 48 kHz

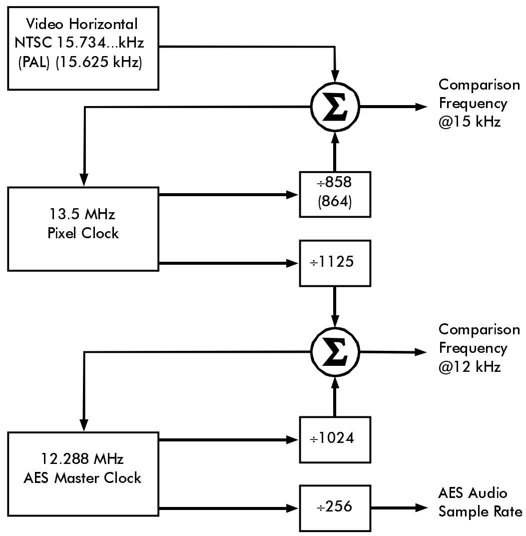


Figure 2-5: Typical topology for locking audio to video horizontal line rate

sample rate is used, a one-half sample time error in phase is equivalent to 75 degrees of phase shift for a 20 kHz analog tone, a serious degradation in the audio image. The only conclusion which can be made is that digital audio phase must be preserved when converting multiple channels of audio between the analog and digital domains. This is almost impossible to achieve with a typical video reference. Figure 2-5 shows two phase lock loops which, when used in cascade, lock to horizontal line rate and generate an audio reference frequency. Unfortunately, these two phase locked loops include four digital dividers which start at random states on power up. Hence the output phase of the audio reference is random; incoherent. The only way to provide an absolute phase reference is to use an SDIF-2 word clock or an AES3 signal as the common master clock for conversion. Chapter 4 and Chapter 15 contain more details.

Strategies for Synchronization

The Local Island

One simple and effective technique for locking a piece of audio equipment to a piece of video equipment, a DVTR for example, is to use an extra AES output of the DVTR. This output is a standard AES signal operating at the correct video referred sample rate. The down side is that this connection will probably need to be made uniquely for every session via a patch bay or jumper cables. It is important to note that it is virtually impossible to lock the DVTR to the audio equipment. Chapter 4, Chapter 5 and Chapter 7 show some examples.

A Small Suite

For a suite of audio equipment, including a rack frame full of analog to digital converters (ADCs), which is going to be used for video applications, use a local digital audio reference locked to video. All audio and video equipment will be frequency locked and the conversion equipment will be running with complete phase coherence, provided the digital inputs of the video equipment are used. The audio image will be completely preserved and the transfer of digital audio between equipment in this suite is assured. Phase coherence generates savings when a production mixer is part of the suite. Since all the inputs are in phase, no input by input delay adjustment is required to correct the audio image.

Video Facilities, Existing and New

The biggest synchronization gain is achieved by locking an entire video facility to a common reference for both NTSC and PAL formats. All audio is running at exactly the same 48 kHz rate. Audio program material may be transferred universally, transparently throughout the facility. The signal feels exactly like analog. The exciting part is that both new and existing facilities can be configured this way with little effort. Simply generate a common timing signal which feeds both the PAL and NTSC master sync generators. This will lock all video equipment to a common time base. Downstream video timing is preserved. For configurations where audio phase is crucial, the suite approach described above is useful. For facilities which already operate in either PAL or NTSC and are expanding into the opposite format, this same topology may be used, or the timing network for the new format may be locked downstream of the existing video network. In either case, all digital audio sample rates can be locked to a common reference. Chapter 15 provides detailed examples.

The Fine Points of Plant Synchronization

The most stable oscillator available should exist at the top of the timing chain. Typically, this is a 5 MHz ovenized crystal oscillator, however some facilities may choose to use a 5 MHz rubidium or cesium reference. These three options provide stability which exceeds the NTSC and PAL standards. If a PAL facility is to be slaved to an NTSC facility, beware that NTSC stability requirements are less stringent than those for PAL. When either video format is slaved to the other, more phase locked loops are introduced into the timing chain than in the case where two video formats are generated from a common starting point. This may cause some increase in jitter in video equipment referenced to the slaved generator.

Audio Sources without Sync Inputs

Converting asynchronous audio to a standard audio sample rate will simplify plant operations. This is particularly true in larger facilities. Asynchronous material exists, it is inescapable. Digital audio equipment without synchronization inputs has been manufactured and used for years. The biggest offender is equipment which offers a vari-speed option. This allows the sample rate to vary by +/- 12.5%, slightly more than one musical step. For example, a 48 kHz nominal sample rate can vary between 54 kHz and 42 kHz. When confronted with this material, there is only one course of action: digital audio sample rate conversion.

A digital audio sample rate converter operates analogously to a video standards converter. The signal to be converted is fed to the input of a dynamic low pass filter which continuously adjusts its output phase, producing interpolated sample values which occur at a rate determined by an external timing reference. Sometimes, the input and output sample rates may be locked together through an integer relationship; for example, 48 kHz and 44.1 kHz are related by the ratio of 160 to 147. This type of conversion is called a synchronous sample rate conversion. Other times, there is no integer relationship between the two rates. This is called an asynchronous rate conversion and is often required when conversions between video formats or video film transfers are made. It is always required for vari-speed applications. Chapter 7 provides further insight into the process of sample rate conversion.

Sample rate converters are widely available, and units which are specifically tailored to the 4 channel architecture of digital video are available. Converters can be installed as shared devices, accessed by a router, or as stand alone devices which are patched in when and where required. Sample rate converter outputs may be synchronized to a reference using all three techniques described above. Detailed examples are shown in Chapter 5 and Chapter 7.

Signal Path Timing

Digital audio delay lines are essential for preserving the phase relationship between audio and video. Loss of Lip Sync provides easy indication that phase has slipped. The relative timing of audio and video signals varies as video processing equipment is inserted into the signal path. Equipment such as DVEs and color correctors insert delay into the video path which much be compensated for to maintain the original phase relationship between audio and video signals. An adjustable digital audio delay provides this functionality. Chapter 5 describes both digital audio and time code delay in a video processing environment.

A Design Strategy for Synchronization

No plant should be without this strategy. What follows are some key points to bear in mind as the plant design develops.

- All digital audio transfers require at least two pieces of equipment be synchronized.
- Digital video equipment does not usually lock to digital audio signals, so audio equipment must be locked to a video reference if both audio and video equipment are in use.
- Video timing signals can all be locked to one common reference.
- Existing video facilities can be synchronized without disturbing any downstream video timing alignment.
- A phase accurate digital audio reference should be used for conversion between analog and digital formats. This reference should be frequency locked to video.

Routing Digital Audio Signals

AES3 digital audio routers may be sorted into three classes: embedded, synchronous and asynchronous. Each has its benefits and weaknesses. All are intended to provide automated interconnect of AES3 digital audio signals within a facility, or between suites. Routing digital audio data is not as simple as one might think. It is necessary to understand the signal format sent to the router, and any change the router may make in the output format, and the effect any changes may have at the receiver. To make an educated decision between the three routing types, it is important to first understand the AES3 serial data format.

The AES3 format defines a sub-frame; one audio channel, a frame; an ordered pair of sub-frames and a block; 192 frames. Two audio channel sub-frames make up an AES3 frame as shown in figure 2-7. If the channels are stereo, Left comes first and is called channel A or 1 and Right is channel B or 2. The C,V,U and P bits indicate channel status, sample validity, user data and sample parity, respectively for each channel. Validity and Parity refer to sub-frame channel data; the immediate audio sample. Channel status bits accumulate on a block basis to form a 192 bit, or 24 byte, data header which contains unique information about the audio data in each channel. Useful information contained in the channel status bits indicates sample rate, emphasis coding, stereo format and professional mode. A CRC is also included for the 24 bytes of channel status data. The channel status data headers in a given AES3 data stream are independent. As an example, A channel audio may be emphasized and B channel audio not. Remember, the two channels are not required to be stereo. The User bit does not have a defined application and is not often used. Four Auxiliary bits are reserved for additional audio services. By default, they can provide room for 24 bit audio data however, these

4 bits can also be used to encode a third audio channel into the data stream. The channel status bits indicate exactly what information these 4 bits contain. Finally, 4 bits are reserved for framing. In fact these four bits contain three unique bi-phase code violations. These violations indicate if the subsequent data is channel A , B or the start of block; a special case of channel A. Figures 6 and 7 show the AES3 format in detail.

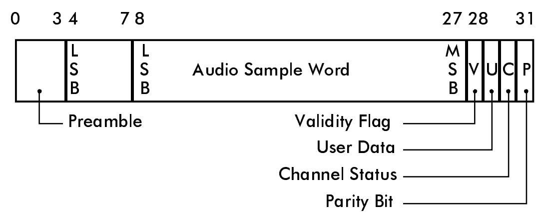


Figure 2-6: AES3 subframe format

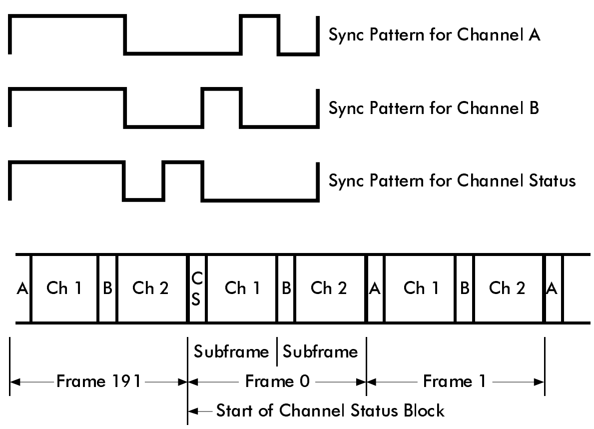


Figure 2-7: AES3 frame format

Embedded Audio Routing

The SMPTE 259M-1993 standard allows digital audio signals to be embedded (multiplexed) within the serial video data stream. The specification provides for up to 4 AES audio channels (4 stereo pairs or 8 mono) to be coded in a single video channel. Further, 272M-1994 specifies the maximum number of audio channels permissible in the video ancillary data space as 16.

This feature seems to offer tremendous advantages over traditional methods of running separate audio/video systems, particularly in broadcast facilities where limited audio breakaway is required. At first look, utilizing embedded audio offers some very attractive benefits: simplified system design, reduced cable requirements and distribution amplifier count, single routing system and of course excellent cost savings. However, although these advantages are theoretically very real, available product and current technologies do not necessarily allow these advantages to be fully utilized.

Various articles, quotations and statements extolling the virtues of embedded audio systems have been made. Many of these statements have been made by manufacturers of devices designed to embed and de-embed, but the real issue is: Can a working embedded system be built that makes use of the advantages, cost effectively?

Assuming that such systems work properly, it must be assumed that anyone building a system will need to interface with numerous pieces of equipment that do not provide embedded audio (most VCRs and VTRs with the exception of the latest machines). Each of these devices will need an embedder, such devices currently cost between $2 - 4K each, plus the cost of the equipment frame & supply, resulting in an average cost of about $3K per source.

Comparison with separate AES 'On Air' Router Number of non-embedded sources = 32	Cost per Source	Total Cost $
AES - Embedders	3,000	96,000
Less - Cable savings, 6000 meters @ $3 per meter		18,000
Less - Cable installation labor @ $3 per meter		18,000
Less - AES **Synchronous** audio routing system 64^2		20,000
Total Savings/(Cost)		(40,000)

Comparison with separate AES 'Pre-select' Router Number of non-embedded sources = 32	Cost per Source	Total Cost $
AES - Embedders	3,000	96,000
Less - Cable savings, 6000 meters @ $3 per meter		18,000
Less - Cable installation labor @ $3 per meter		18,000
Less - AES **Asynchronous** audio routing system 64^2		15,000
Total Savings/(Cost)		(45,000)

So on this point alone, unless an all new facility is being constructed using the latest equipment, the following cost comparisons seem reasonable:

These comparisons do not include any equipment for de-embedding, which is essential for audio replacement, overs and mixing in master control. It also excludes audio A to D converters, as they are required for both systems.

Regardless of cost, does the system work properly?

Unfortunately, very little care has been taken by video equipment manufacturers to ensure that AES audio is synchronous with video signals.

The AES3 signal, like video, is made up of frames of data (see figures 2-6 and 2-7). At 48KHz there are 5 AES3 blocks during each PAL video field and 4.170833 blocks for an NTSC field (see fig. 8). Therefore, if a video signal is used as a genlock source for AES3 signals, the frame alignment and phase relationship between audio signals is arbitrary. In this circumstance, regardless of whether a signal is embedded or not, clean audio transitions are difficult to achieve.

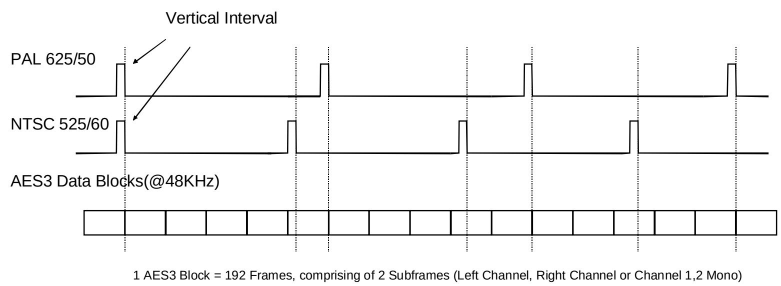


Figure 2-8: AES3, 525/60 and 625/50 timing relationships

If it is assumed that embedders are used that can be locked to a standard AES3 reference and that the audio inputs will be re-synchronized to that reference (a big assumption), it would be possible to ensure that all audio embedded by these devices would be synchronous. Therefore, when the video streams carrying these signals are switched, the audio should transition without error (provided that the switch used is transparent to the areas of the video signal carrying the ancillary data information).

These assumptions do not take into account any video equipment in the path that strips and re-inserts sync, may change blanking and possibly introduce line delays. As audio data is embedded in every line of video, realignment of video data can be detrimental to the audio information. It may still be recoverable, but it will certainly no longer be synchronous with other signals and it will no longer be possible to provide error free switching with another signal that did not take the exact same path.

The net result of all this is simple, it is not currently practical to build a system with embedded audio that can be switched or mixed with 100% reliability, without including an elaborate cross-fade mechanism in the switching system design (see Fig.9). Furthermore, an embedded system that includes existing equipment will probably cost more than separate systems, even though it will be simpler in design.

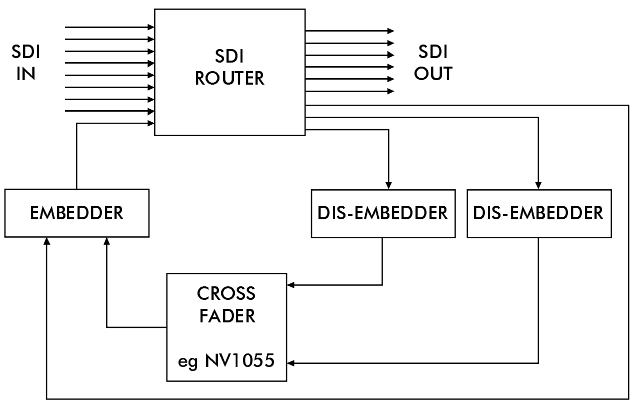


Figure 2-9: Error-free embedded AES switching

No mention has been made of operational flexibility; if an embedded system is chosen, design consideration must be given to the level of flexibility required. Once audio and video are combined, audio or video insertion and mixing is no longer possible without first de-embedding audio. Even inserting a station ident via a video presentation switcher will disturb the embedded audio data. Therefore limiting the ability to manipulate any signal without first routing it to the appropriate de-embedding and embedding devices.

Embedded audio has its place, it is a valuable cost saver in new systems that do not need much flexibility and have limited switching requirements. However, in the transition from analog to digital, there is a need to maintain a hybrid system for some time to come. Therefore, research should be conducted before embedded systems are considered seriously.

For all but the simplest systems, separate audio and video routing systems currently offer the best cost and performance advantages.

Asynchronous Routing

Asynchronous routing is the simplest in form and the most versatile. This type of router is truly an electronic switch which from the outside operates as if only wires were used to connect outputs to inputs. Because this signal is not processed, an asynchronous router will accept digital audio inputs at any sample rate. The destination equipment must of course be able to lock to the sample rate of the data routed to it. Most equipment will mute if the sample rate is not compatible for its selected mode of operation. The sample rate of asynchronously routed digital audio signals is allowed to change dynamically. This feature allows full vari-speed operation, an important feature for audio production. Asynchronous routers can also switch signals which are synchronized to a common reference. There is one potentially significant drawback to both embedded and asynchronous routing: when a switch is made between inputs, corruption of the output AES3 serial data format is virtually guaranteed. An AES3 frame will be truncated. Destination equipment may have to reacquire lock. Equipment must be realigned to the AES3 frame and block boundaries in order to resume normal operation. This process takes a perceptible amount of time, sometimes as long as one second. Some equipment behaves gracefully during this time, that is, it mutes. Not so well behaved equipment may continue to generate audible pops and clicks.

An asynchronous router is effective, easily integrated into any facility and efficiently operated. Asynchronous routers work well in environments where circuits are pegged for the current job and not changed for the duration. They are the only choice for environments where many different audio sample rates are used. Telecine applications, or professional audio production are good examples. They do not work well as the final stage for ON-AIR switching or where video frame accurate audio editing is required.

Asynchronous routers are always implemented as space matrices. They are also the least expensive option. Price and flexibility make this option very popular.

Synchronous Routing

The synchronous router is much more complex. It is designed to provide an AES3 frame accurate switch, predictably timed to the start of a video field if desired. Any transition between two inputs is made at an AES3 defined frame boundary. This insures that the destination equipment sees no discontinuity in AES3 frame or block data formats. The synchronous router requires each input be frequency locked to a common digital audio reference. Techniques for this are described in this text and in Chapter 9. Synchronous routing is similar to field accurate video routing. A properly timed video switch is visible only as a cut. If not correctly timed, color shift, H-shift and even vertical roll can result. A correctly timed AES3 switch provides a clean cut. The alternative is pops, clicks and mutes.

A synchronous router must receive each AES3 input and align its frame phase to that of a local reference, usually the house AES3 digital audio reference. When a take occurs, audio data is removed from the serial bit stream of the new source and striped into the correct position in the serial bit stream of the output. Each AES3 input is frame aligned and the switch transition also occurs at an AES3 frame boundary. The resulting switch preserves stereo phase and channel assignment. The destination machine receives a synchronous router output data stream which contains no framing interruptions. The result is a clean, instant take; no pops, clicks or pregnant pauses. An analog type audio break-away is achieved. This transition can be aligned to video. It is also possible to achieve subframe routing with similar synchronous techniques. Chapter 6 and Chapter 9 provide further insight into these operations.

The synchronous router is the perfect answer for clean "HOT" takes and "ON-AIR" applications. This makes it an excellent choice for video facilities. It provides a guaranteed video timing relationship for audio transitions. Synchronous routers are manufactured as both space and time matrices. Space matrices offer unbounded expansion. Time matrices are limited by internal bus bandwidth considerations. Synchronous routing is typically fifty percent more expensive than asynchronous, but less expensive than embedded.

Analog Routing

Analog audio signals still exist. There are two approaches to supporting both signal formats in one facility. First, keep the two formats separate, either by suite or by layer. A separate digital router slaved to the existing control system is the least expensive option and the easiest to install. Routers which allow both analog and digital cards in the same frame offer another approach, however, they are typically more expensive and require significantly more effort to install. This two layer topological option has one major drawback. The digital and analog layers cannot share signals. This drawback is eliminated with analog to digital converters (ADCs) and digital to analog converters (DACs).

The second approach is to use ADCs and DACs to make each analog device appear to the plant as a digital device.

Then only one digital router is required. Some ADCs and DACs are necessary for crossover in the two layer topology discussed. Unless this crossover requirement is small, and it usually is not, the incremental cost of the remaining conversion equipment is offset by the savings realized with an all digital router. As existing analog machines are made obsolete and replaced with digital equipment, ADC and DAC are eliminated or reassigned. Router ports become available for use and the transition to digital is simplified.

A Design Strategy for Routing

The following steps should help clarify what type of routing is best for a given need and what variables should be analyzed as part of a thorough design analysis.

- Embedded routing is clearly best if the audio is never broken out of the data stream. If this is not the case, another routing option may be indicated. Determine the size of the audio break-away matrix needed. Include the cost of embedders and disembedders to support this matrix. Then compare to a separate audio matrix feeding embedders only.

- "ON-AIR" or clean "HOT" switching requires synchronized AES3 signals. If this feature is required or desirable, synchronous routing is necessary.

- Asynchronous routing is the most flexible, and can always be used unless a clean transition must be guaranteed.

- Asynchronous routing is the least expensive option. Synchronous routing is typically priced fifty percent higher.

- Embedded routing must be priced as a combination of both audio and video routers. Do not forget to include the price of embedders and disembbeders for audio only inputs and outputs.

- Think hard before buying a new analog audio router. Digital routers are less expensive, use less power and require less space. Use ADCs and DACs as needed to bridge between the analog and digital formats.

- When synchronous routing is used, all digital audio sources must be synchronized. This is recommended for any routing option.

Analog Audio Considerations

Analog equipment such as human ears, human vocal tracts, microphones and speakers have not been eliminated. Analog machines which cannot be disposed of for economic or archival reasons still need to be supported. For these applications, ADC and DAC equipment will be required as an interface between the analog and digital facilities. ADC and DAC equipment must be properly aligned to prevent unwanted gain or attenuation. Two factors need to be considered; Full Scale Digital level and input/output impedance settings.

A Full Scale Digital (FSD) input is that analog signal level which, when converted to digital, results in the largest possible digital code output. This is analogous to the clip level of an analog signal. Full scale digital level is usually set to the analog clip level, or maybe 1 dB above. It is important to standardize this level on all equipment which provides conversion between the analog and digital domains. It is possible to find machines which are calibrated to different absolute analog levels. Transfers between these pieces of equipment may generate unwanted gain or attenuation. Avoid transfers of digital material in the analog domain if at all possible. If unavoidable, Chapter 4 provides some possible solutions for Full Scale Digital Level problems.

Input and output impedance selection is also important. Audio signal distribution has traditionally been implemented with a matched 600 ohm source and destination impedance. Advances in equipment design have made it possible to use low source impedance, nearly zero ohms, and high destination impedance, twenty to forty kilo-ohms. This has become the standard practice in the majority of facilities. because it does not introduce 6 dB of attenuation at every interconnect. Chapter 4 provides more details on how to accommodate matched impedance environments.

Summary

Digital Audio technology may be readily integrated into any facility, existing or new. Attention to detail and good design practice will result in a reliable, efficient facility. Synchronization and routing are the foundation upon which the plant will be built. An investment of design energy in these areas will yield a substantial return over many years. Plant interconnect requires more detail than with analog, HUM is gone, but jitter and reflections have taken its place. Careful selection of cable and transmission line wiring practice, eliminates virtually any opportunity for jitter or signal reflections to disrupt the transparent exchange of digital audio signals within the plant. Digital technology is ready for use.

Chapter 3. AES3 Interconnect and Distribution

Part 1: Interconnection

Pops, clicks and other unacceptable sounds are often generated when data is transferred between two pieces of equipment designed to meet the AES3 standard. In a large facility, three or more pieces of equipment configured in series may produce different results. AES3 digital audio is fast becoming the only format available on many machines, therefore a solution must be found for these problems. Larger, all digital facilities with centralized routing and distribution are being designed and installed. Techniques must be developed which permit reliable transfers of digital audio data between various pieces of equipment.

These symptoms are common to three unique problems: impedance mismatch, jitter, and frequency inaccuracy. Any combination of these may be present simultaneously. The key is to eliminate as many of these problems as possible, if not all. Each problem, its causes and some solutions follow.

	AES3 1985	AES3 1992	AES3 ID
Signal Level	3 to 10 Volts	2 to 7 Volts	1 Volt +/- 20%
Source Impedance	110 Ohms	110 Ohms	75 Ohms
Input Impedance	250 Ohms	110 Ohms	75 Ohms
Cable Impedance	NO SPEC	110 Ohms	75 Ohms
Connector	XLR	XLR	BNC

Table 3-1: Electrical interconnect specifications for AES3 data

Impedance Mismatch

Table 1 compares the essential features of the AES3 1985, AES3 1992 and AES ID Standards and guidelines. The bandwidth of the AES3 signal is much higher than analog audio, 5 to 6 MHz versus 20 kHz. AES3 data interconnections must be treated as transmission lines. Successful connection of equipment requires careful practice of techniques used for high frequency signals: source, input and cable characteristic impedance should all match.

Look at table 3-1 again. Two sources of mis-match are readily apparent in the AES3 1985 standard: input impedance and cable impedance.

Correct the major problem first. Convert all AES3 1985 equipment input to be compatible with AES3 1992. Use a 196 Ohm resistor, a standard 1 percent value, as shown in figure 3-1. With care, this resistor fits inside the mating XLR plug. This change should be made throughout your facility. The easiest way to verify equipment impedance as either 110 or 150 Ohms is by schematic examination. Transformer coupling prevents DC measurement of internal resistance.

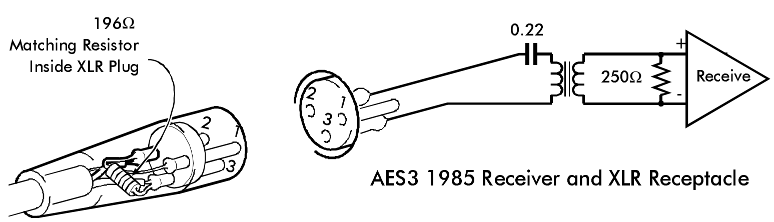


Figure 3-1: Impedance Correction of AES3 1985 Receivers

Alternatively, isolate AES3 1985 equipment with a digital audio distribution amplifier such as the NV1021. Place the NV1021 as close as possible to the destination equipment, allow a maximum of 10 feet for cable between the NV1021 and the AES3 1985 equipment. This configuration provides a matched termination to the original source and minimizes path length of the reflected signal at the load. Accurate data recovery by the receiver is now possible. Figure 3-2 shows this connection.

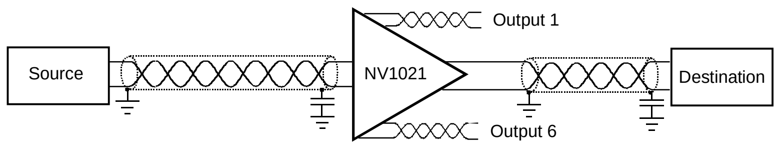


Figure 3-2: Load isolation

Correcting cable impedance is not as easy. Ideally, an impedance matching resistor is connected as shown in figure 3-1 so that the combined value is equal to that of the cable. Instead of 196 Ohms, a 45 Ohm analog audio cable requires 54.9 Ohms to match a 250 Ohm input or 76.8 Ohms to match a 110 Ohm input. Fortunately, for short cable runs where reflection amplitude is the greatest, the NV1021 DA will accurately recover the data with its input impedance selected to 110 Ohms. Ideally, existing analog audio wiring is replaced with 110 Ohm, low capacitance cables, however, it may be desirable to salvage long runs of pre-installed analog audio twisted pair. A cable equalizer enables reliable transmission for these applications and provides even longer transmission distances for 110 Ohm cable. A cable equalizer corrects cable transmission loss with a filter that typically provides more gain at higher frequencies than lower frequencies. Once properly equalized, the data may be accurately recovered by the receiver. Cable equalization is provided by digital audio distribution amplifiers (DAs), such as the NV1021, at affordable prices. Using DAs is often less expensive than pulling new cable runs. Figure 3-3 shows this application and Table 3-2 provides some typical results for various types of cable.

The NV1021 DA will equalize cable for both twisted pair and coaxial applications. The difference is transmission distance. A typical analog audio cable can be equalized for up to 1000 feet, digital audio cable for 2000 feet and coaxial cable for 4000 feet or more. These results are a general guide. Cable type and topology determine the exact transmission distance for any given cable run. Table 3-2 shows various cables types and their typical transmission distances.

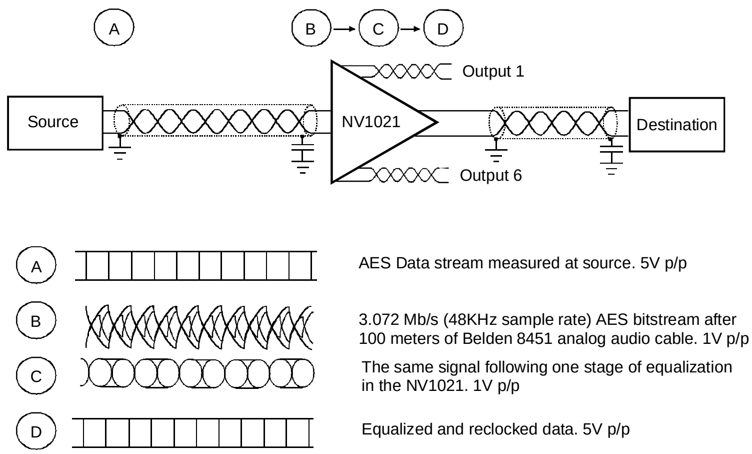


Figure 3-3: Cable equalization

	NV1021 w/ No EQ	max EQ
45 Ohm twisted pair, 30pf/ft	250 feet	1000 feet
100 Ohm twisted pair, 12pf/ft	500 feet	2000 feet
110 Ohm twisted pair, 12pf/ft	800 feet	2500+ feet
75 Ohm coax, 20pf/ft	2000 feet	4000+ feet

Table 3-2: Cable Types and Typical Transmission Distances

BNC vs. Twisted Pair: The NV1021 Accommodates Both

A number of video plant engineers prefer coaxial cable for AES3 data transmission. AES and SMPTE standards groups have proposed guidelines for 75 Ohm, 1 Volt BNC interconnect. Impedance match is correctly specified from the start, eliminating many potential problems. Some of these engineers have considered using analog video distribution and routing equipment for AES3 signals. This practice should be discouraged. AES3 signals contain fast digital edges which generate high frequency energy that exceeds the bandwidth and slew rate of many analog video products. Signal edges become distorted. This distortion generates jitter in receiving equipment. High frequency signal energy generates crosstalk which causes bit errors. Jitter and crosstalk related bit errors are data pattern dependent and difficult to isolate and diagnose. For these reasons, equipment specifically designed for AES3 signals is strongly recommended.

Different signal levels have been used for coaxial AES3 data transmission. Early adapters which only matched the 110 or 250 impedance to 75 Ohms yielding a signal level of 3 or 4 volts. The 1 volt level as proposed by the AES3 ID and SMPTE guidelines has gained the widest acceptance. If the coaxial interface is used, AES3 to BNC adapters must be purchased for AES3 equipment conversion. Be sure to specify 1 Volt operating level. Consistent use of 1 volt levels insures compatibility with future equipment. Also, specify the receiver adapters for either 110 or 250 Ohm matching. NVISION provides virtually all of its AES3 equipment with coaxial interface options. This eliminates adapters and provides an easily installed 1 volt, 75 ohm interface.

Jitter

Audible pops and clicks are caused by bit errors generated when data clock recovery circuits cannot follow the short term frequency variation in the input data stream. This frequency variation is jitter. Jitter is generated by a receiver when its input is distorted as described in this chapter. Jitter may also be generated and amplified by a poorly designed phase locked loop (PLL). As equipment is cascaded and long signal runs introduced, jitter increases, destination equipment no longer recovers the data correctly and bit errors occur. Conversely, a well designed PLL attenuates jitter, making it easier for downstream equipment to accurately recover data. Jitter is rarely a problem for two pieces of equipment. Once three or more pieces are connected in series, the potential for problems will arise.

Eliminate jitter by using a common synchronization signal or with jitter attenuating DAs. Synchronizing all AES3 equipment to a common reference insures that AES3 output signal timing is derived from the reference clock, not just the recovered data clock. Chapter 15. describes synchronization in more detail using the NV5500 and NV1080 family of reference generators. If external synchronization is not possible, or equipment count is small, the NV1021 Digital Audio DA provides jitter attenuation. Insert the NV1021 in the signal path after the “Last Straw” piece of equipment, that piece which when inserted into the signal path caused the problem. The NV1021 is very robust, reducing jitter to acceptable levels and insuring accurate data recovery by following equipment.

Finally, some equipment does not use PLL clock recovery. Use of this equipment should be avoided whenever possible. Figure 3-4 shows a typical signal chain and the NV1021.

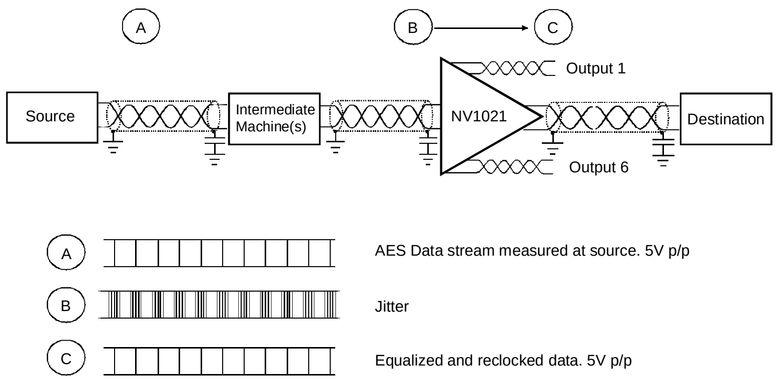


Figure 3-4: Jitter attenuation

Frequency Accuracy

Digital audio sample rates for two interconnected pieces of equipment must be identical. If two pieces of equipment are operating at a nominal 48 kHz digital audio sample rate, they must be synchronized one to the other, or to a common reference. Chapter 15 offers more details on equipment synchronization.

Part 2: Distribution

Those facilities converting to digital audio discover that digital silence and synchronization must be distributed throughout the plant. Furthermore, it is often discovered that audio image is lost, even though a synchronization network is used.

It has become apparent that affordable distribution networks must be designed to provide maximum signal fan-out with accurate phase when needed. Equipment which converts signals from Analog to Digital must be aligned in sampling phase and frequency to preserve the audio image. A signal such as digital silence or alignment tone typically has no phase requirement.

Digital Audio distribution amplifiers can be used to distribute AES3 reference signals, digital quiet and alignment tone. Proper distribution of an AES3 synchronization signal insures the preservation of audio image. Affordable, reliable distribution networks are easily designed and phase accurate networks require no additional expense. Pop free digital audio, phase accurate analog conversions and digital quiet distribution are provided with one AES3 distribution network carrying digital quiet.

Figure 3-5 shows a preferred distribution topology for AES3 signals. Specific design rules for the NV1021 and NV1022 fan out DA's are also provided. This configuration provides the best reliability as well. If one output fails, only the dependent signal branch is adversely affected. A failed output in other topologies will affect a greater percentage of the sync network.

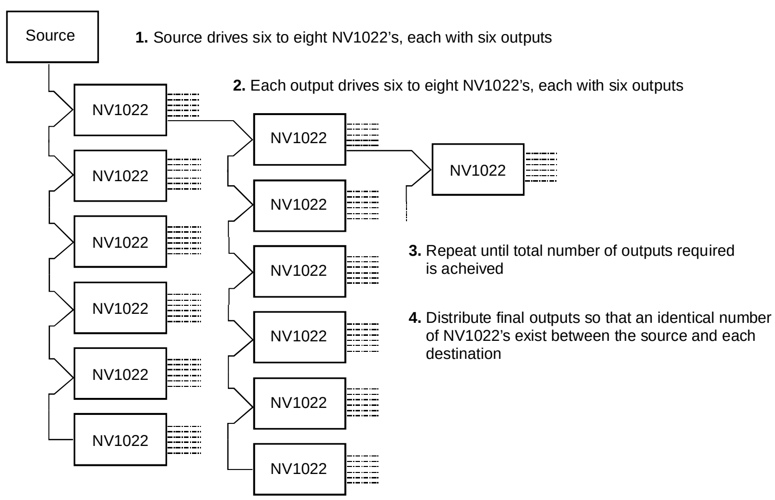


Figure 3-5: Phase Accurate, High Fan-Out Topology

These rules guarantee that equipment referenced to the outputs of this network are phase and frequency aligned. Figure 3-6 shows two methods for accurate phase synchronization of A/D converters. NV1020 CODECs are used in these examples.

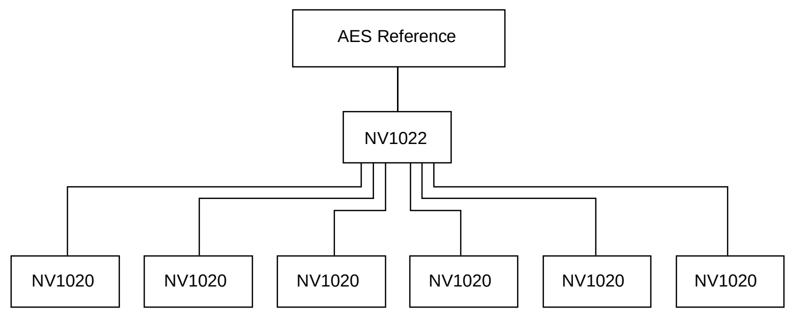


Figure 3-6a: Accurately Synchronized CODECs without Looping

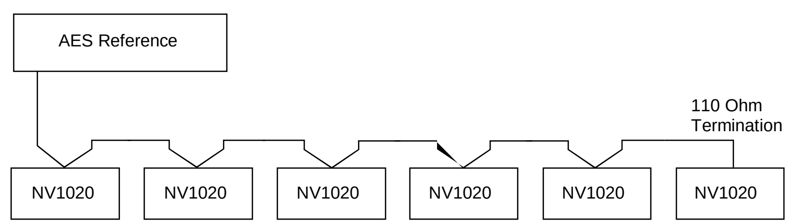


Figure 3-6b: Accurately Synchronized CODECs with Looping

Loop-Thru

Costs are reduced when AES3 signals are looped as part of the distribution technology. While the AES3 1992 standard does not provide for signal loop-thru, NVISION has designed the NV1000 Series of Digital Audio products to sensibly support signal loop-thru. Figure 3-5 includes recommended design rules for looping NVISION equipment. Figure 3-7 shows a preferred layout technique which eliminates the adverse effects of transmission line stubs. Many other pieces of equipment on the market do not offer loop-thru capability.

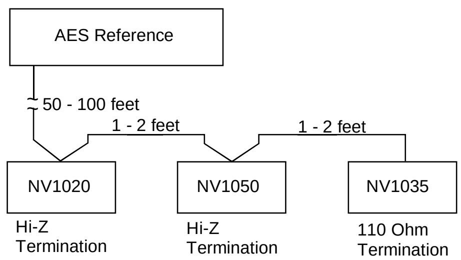


Figure 3-7: Preferred Topology for Loop-Thru Connections

Chapter 4. Audio Conversion Between Analog and Digital

Unexplained audio level changes, pops and clicks and loss of multi-track audio image are problems encountered in converting between analog and digital audio formats.

Digital audio equipment is fast replacing analog machines in production and broadcast facilities. However, ears and mouths are still analog. Speakers, microphones, archived material and essential analog equipment must be conveniently and reliably integrated into the digital facility.

NVISION analog to digital converters (A/Ds), digital to analog converters (D/As) and synchronization products provide features which eliminate these problems. Level adjustment, impedance selection and synchronization are all key elements to be included in facility design.

Audio Level Correction

A/D and D/A converters use digital values which are proportionately related to the absolute analog audio levels of the signals they are converting. The ratio between analog signals and their equivalent digital representation is not standardized. The term Full Scale Digital has been defined as that peak analog signal level, usually clip, which corresponds to the largest possible digital value. As an example, NVISION A/D conversion products allow jumper selectable FSD levels from +12 dBu to +28dBu (see page 51). Analog clip occurs at this level and the analog signal is scaled at the A/D input so that this level generates the largest possible digital output value. Similarly, a full scale digital signal input to a D/A converter set to +28 dBu operation will provide a +28 dBu analog signal output. If +24 dBu operation is selected, the converter will generate a +24 dBu analog output.

How does an unexpected level change occur? Consider a 0.0 dBu tone recorded as digital material with an FSD of +24 dBu. Play it back through a D/A set for +28 dBu. Feed this analog signal to an A/D converter set for an FSD of +24 dBu. The digital level of the material is now 4.0 dB higher than it was before. Additionally, digital headroom is reduced by 4 dB because the maximum analog input level of the A/D is 4 dB lower than D/A. The reverse process generates 4 dB of attenuation. Figure 4-1 shows a typical transfer process which generates unexpected gain.

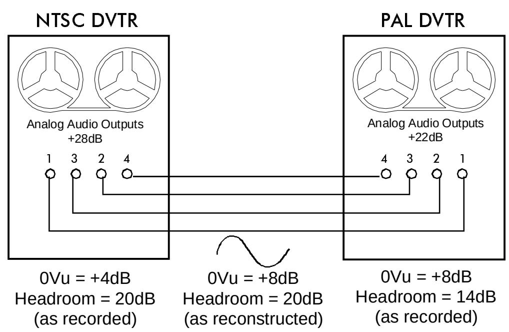


Figure 4-1: FSD Level and Unexpected Gain

A/D and D/A converters are integral assemblies in many D1 and D2 format Digital Video tape recorders. This equipment is notorious for "generating" level changes. The EBU attempted to remedy this problem by providing technical guidelines which were published as "EBU Technical Recommendation R68-1992". Unfortunately, a third operational practice was defined. Table 4-1 shows typical operating levels and FSD levels encountered in professional audio and video production. The inconsistencies clearly show why level changes occur when digital material is transferred via the A/D and D/A converters in this equipment.

	Analog	Digital for Audio	Digital for Video	Digital for EBU Video
FSD Nominal Headroom	No Spec 0.0 Vu No Spec	+24 dBu +4 dBu 20dB	+24 or +28 +4 or +8 20dB	dBu +22 dBu dBu +4 dBu 18 dB

Table 4-1: Common Operating Levels for Audio Production

Reduced EBU head room creates an additional problem. Simple analog scaling of the signal cannot provide 20 dB of headroom. A digital attenuation of 2 dB is necessary to correct this condition. The NV1055 AES3 Mix/Minus with Router is a perfect solution for this problem. Chapter 6 provides more details on the NV1055. Once headroom is correctly adjusted, the EBU output can be resistively attenuated by 2 dB and fed to an NVISION A/D set for +20 dB of operation. The signal is now correctly scaled, and ready for digital distribution.

Unexpected gain is easily avoided by following three simple rules:

1) Set a plant standard for A/D and D/A converter full scale levels. We recommend +24 or +28 dBu.

2) Do not transfer digital material between digital equipment using A/D and D/A converters.

3) If you must violate rule #2, be sure FSD operating levels of D/A and A/D converters match. EBU equipment may require physical modification as opposed to jumper selection.

The NV1020 CODEC provides easy selection of either +24 or +28 dBu FSD levels. The NV1035, NV1045, NV1036 and NV1046 are even more versatile. They allow level adjustment in 1 dB steps between +16 and +28 or +12 and +24 dBu FSD respectively. See figure 4-2.

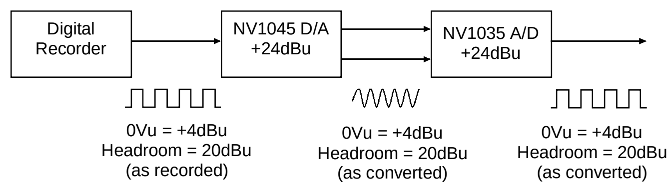


Figure 4-2: Correct FSD alignment for digital transfers via analog

Impedance Matching

Source and input impedance values of analog audio inputs and outputs are treated differently by manufacturers and users alike. The alternatives are matched terminations or high impedance inputs and low impedance sources. If matched terminations are incorrectly used consistent audio levels will be impossible to maintain. NVISION strongly recommends that matched impedance techniques be abandoned for analog audio distribution.

Historically, audio was distributed in a matched impedance environment with 600 or 150 Ohm impedance levels. But audio is a low frequency signal. Transmission line theory predicts that matched impedances are not required. Another argument states that power transfer is maximized when source and load impedances are matched. Again, this argument only holds for matched characteristic impedance environments; transmission theory again. Analog audio cable typically exhibits a characteristic impedance of 35 to 70 Ohms, clearly not a matched impedance environment for either 600 or 150 Ohms.

Matched terminations require 6 dB of additional gain and headroom to compensate for 6 dB of attenuation which is generated when the source is terminated. This is an extremely inefficient use of power. Ideally, audio input impedances should be set to a high impedance value.

Most modern equipment provides a 20 to 40 kohm input impedance. Output source impedance should be matched to the cable impedance. Additionally, long, highly capacitive cable runs cause oscillations in output line drivers if no purely resistive load is present. To provide a partial line match, prevent oscillations and provide short circuit current limit protection, an output source resistance of 65 to 75 ohms is typically used. Hi-Z, Low-Z terminations do not affect signal levels. No gain variations are introduced. Power is not wasted since the source does not have to generate an additional 6 dB of signal level.

Use of a 75 Ohm output impedance to drive a 600 ohm load will result in an attenuation error of approximately 1.0 dB. This can be corrected on any D/A module or CODEC module which must drive a 600 Ohm load. NVISION NV1000 series modules provide jumper selection of 600 Ohm or 40 kilo-ohm input impedance on A/D converters and 75 Ohm or 600 Ohm source impedance on D/A converters.

Synchronization

Pops and clicks are generated when A/D converters are not locked to the destination of their digital signal outputs. For small islands, the A/D converter can be locked directly to the destination using a spare output of this same destination equipment. Figure 4-3 shows a typical DVTR rear connector panel and how to connect it with a pair of NV1020 CODECs.

Larger groups of converters can be synchronized to a common reference which is often locked to house video sync. Figure 4-4 shows a NV1080 AES reference generator and some NV1020 CODECs connected in this topology. Synchronization networks are easily implemented with reference loop-thru connections provided on all NVISION A/D and CODEC conversion modules.

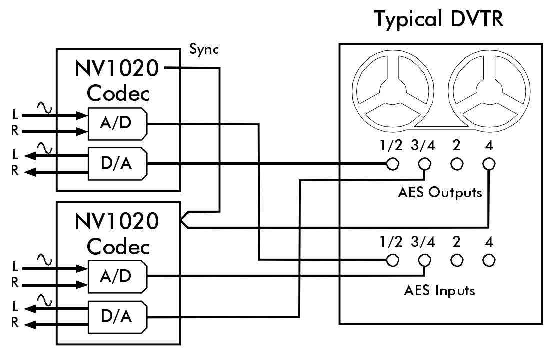


Figure 4-3: Small island synchronization

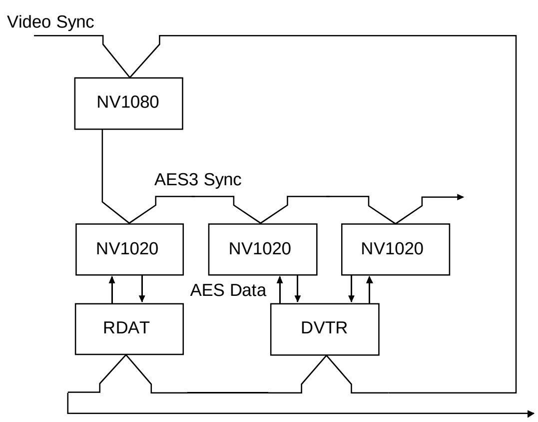


Figure 4-4: Large facility synchronization

Equipment simply has to be synchronized if glitch free conversion is desired. Chapter 15. provides extensive discussions of frequency synchronization.

Audio Image

The live session sounded great but the digital playback sounds murky, the audio image is lost. The most likely cause for this occurrence is loss of audio phase. When more than one AES3 digital audio signal is fed to a common destination, such as a mixer, digital phase alignment occurs. Each input is aligned to a common AES3 frame boundary so that signals can be filtered, shaped, equalized and summed. If A/D conversion equipment used to generate digital master material is not locked in phase as well as frequency, the original analog phase relationship of these audio channels is irrevocably lost.

NVISION designs all of its A/D and D/A conversion equipment so that the sampling instant of conversion is defined by an AES3 frame boundary. A/D converters use an AES3 reference input, not video. A composite video signal can generate an accurate frequency reference as shown in figure 4-5. However phase is another story. Four digital dividers help generate the output frequency. At power on, these dividers are initialized to a random state. A guaranteed phase relationship between sampling points defined by the output of two circuits of this type is usually not provided.

An AES3 reference is highly recommended to guarantee phase accurate sample and audio image preservation. All NVISION A/D converters support an AES3 reference with a loop-thru input connection. This assures phase accuracy and easy, cost effective installation of synchronization signals. Figure 4-4 shows one implementation of this topology.

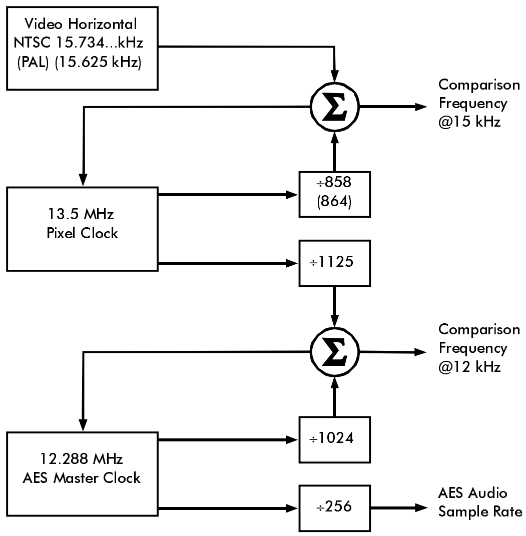


Figure 4-5: Locking Digital Audio to Video Horizontal

Chapter 5. Digital Audio Delay Compensation

You hear the words and then the lips move. Everything was perfectly locked when you started. What went wrong?

Loss of lip sync between audio and video in program production is not a new phenomenon. Yet, with all of its power and sophistication, digital technology has not alleviated the problem. In fact, the situation is worse. Digital audio and video processing and distribution equipment are rife with delay. One line delays, 2 field delays, 4 field delays, 10 frame delays; the list goes on. In order to maintain lip sync, the differential delay between audio and video should be held to less than ½ of a video field It is equally important to preserve this phase relationship for time code as well.

Video path delay is often longer than audio path delay. A delay inserted in the audio path compensates for the difference, restoring the original phase relationship. Processes such as color correction and real time digital video effects generation benefit from audio and time code delay compensation. Converting analog material into a digital format also generates differential delay which needs compensation. Complex applications such as standards conversion are simplified with audio delay however, time code is typically re-striped in the destination format. For instance, an NTSC to PAL transfer usually discards SMPTE time code and re-stripes EBU time code.

Loss of Lip Sync is clearly unacceptable. Fortunately there is an alternative. Digital audio delay is easily installed, provides extremely accurate compensation and is quite affordable. Loss of Lip Sync, the alternative, is not an option. The following applications provide topologies and detailed discussions for the examples just described.

Existing analog material must be converted to a digital format.

Video noise reduction and image enhancement are desired as part of the conversion process. This type of 2D spatial processing typically inserts 2 or more fields of delay in the video signal path. Audio and time code phase must be accurately preserved. Figure 5-5-1 shows this process. The differential delay between the audio A/D and the combined video A/D and noise reduction equipment is compensated for by the NV1060 AES and the NV1061 Time Code Delay modules. The NV1080 Digital Audio Reference generator is used to synchronize the audio A/D converter to the sample rate of the destination DVTR. A local video sync network is used to time the video A/D and NV1080 module.

The NV1060 and NV1061 are easily configured for the desired delay which is typically 2 or 4 fields.

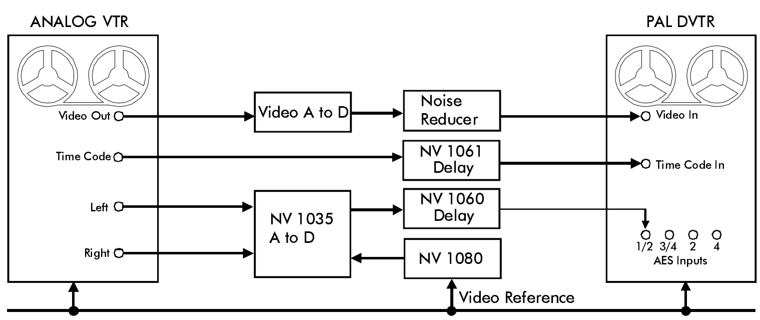


Figure 5-1: Delay Compensation for Analog to Digital Conversion with image enhancement

Synchronized delay compensation is also possible.

Figure 5-2 shows an NV5500 synchronizing PAL and NTSC DVTRs. The NV1060 compensates for the delay of an all digital standards conversion between the two formats. Since the NTSC and PAL recorders generate identical 48 kHz sample rates, digital audio rate conversion is not required.

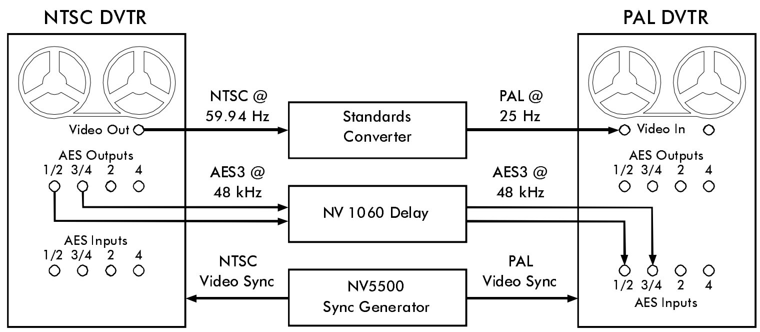


Figure 5-2: Synchronous Delay Compensation

Compensation for standards conversion delay.

Figure 5-3 shows an NV1060 4 Channel AES3 Delay and an NV1050 4 Channel AES3 Sample Rate Converter used to compensate the delay generated in an NTSC to PAL conversion. The source material is NTSC format video transfered from film with a 2 – 3 pulldown. It is fed to a standards converter executing a 3 – 2 drop followed by 2 or 4 field interpolation. The PAL video field rate is 4.004% slower than normal at the standards converter output. The designation video recorder locks to this rate generating a 4.004% slower audio sample rate. The NV1050 provides digital audio sample rate conversion from 48 kHz to 46 kHz. An AES3 output of the destination DVTR backtimes the NV1050.

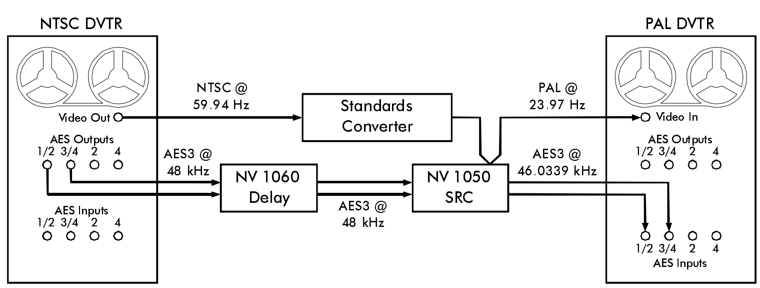


Figure 5-3: Delay Compensation for Standards Conversion

The NV1060 delay module can also be used for audio only delay compensation as well. In its standard configuration the NV1060 delay provides 0.01 to 0.341 seconds of delay for two AES3 signals. Using industry standard SIMM SRAM modules, this delay can be extended to 2.67 seconds. If longer delays are required, either cascade NV1060 modules or if only 1 AES3 signal is used, loop it from the first output back to the second input for 5.33 seconds of delay. Either choice offers plenty of delay for applications such as satellite audio transmission, digital reverb, echo generation, and the like. The possibilities are endless.

Chapter 6. AES3 Mixing and Subframe Routing

Boss: The client wants Left on channel 1, Right on channel 2, M&E on Channel 3 and an M&E over an LR mono mix on channel 4.

You: Anything else?

Boss: Oh yeah, I want it now.

You: Great, the client's tape doesn't look anything like this, Left on 3, Right on 1, M&E on 2, ... Yeech.

Part 1: Digital Audio Layback, Dubs and Channel Swapping

Digital audio production and distribution are standard practice and AES3 is the format of choice, particularly in the 4 channel environment. Many operations such as tape duplication and dubbing require simple audio processing. Phase inversion, input gain adjustment, stereo mix and voice overs are just a few of the basic operations that must be performed. Feature laden, digital audio production mixers provide this functionality, but just are not efficient for these simple applications. Equipment and operational costs are steep and the feature set is overkill. Conversion to the analog domain costs less, but the degradation in performance associated with A/D and D/A conversion make this alternative less appealing.

Analog and SDIF-2 digital audio, transmit each channel as a separate signal. AES3 digital audio transmits two channels as one signal. For digital video applications, 4 channels of audio, or two AES3 signals, are used. In order to manipulate audio signals on a channel by channel basis, the AES3 frame must be taken apart. Further processing or channel swapping between AES3 signals necessitates phase alignment of these data sub-frames. Figure 6-1a shows two AES3 signals as they might appear to inputs of a production mixer. Figure 6-1b shows a preferred format and phase alignment for processing.

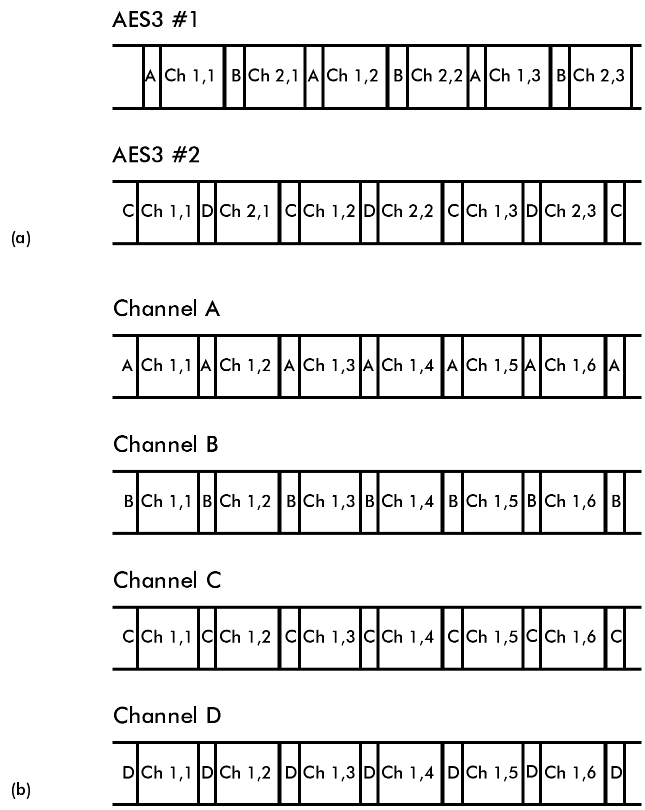


Figure 6-1: AES3 Digital Audio Framing

Using this technique, it is now possible to swap data between A, B, C and D channels. It is also possible to create simple linear mix effects, such as stereo or quad mix downs. All channels are aligned, so stereo phase and audio image are preserved. Four channel processing is usually all that is required for typical video duplication and dubbing processes. Figure 6-2 shows an internal signal path for a 4 layer, 4 x 1 digital audio mixer. Each AES3 output channel is generated as a 4 to 1 linear mix of the inputs. Mixing with zero coefficient values provides channel swapping. Independent phase inversion and gain adjustment is provided for each input. Four unique 4 x 1 mix setups are easily defined and each finished mix is assigned to either channel of either AES3 output. Final gain adjustment for each output channel is also provided. The NV1055 provides exactly this functionality.

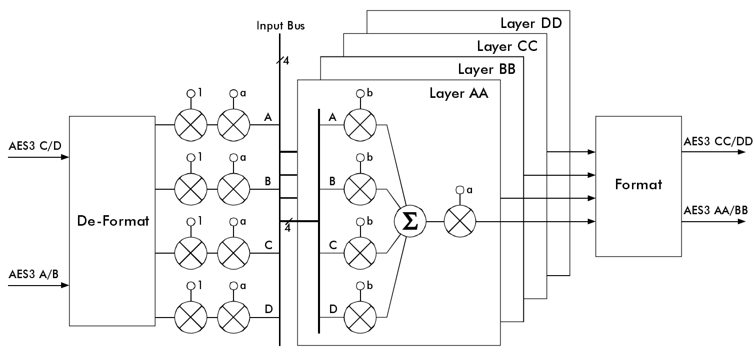


Figure 6-2: NV1055 conceptual internal architecture

Figure 6-3 shows the NV1055 connected between two DVTRs. All necessary signal timing is derived from the signal inputs, enabling clean glitch free signal manipulation. The NV1055 can be configured with front panel switches, or a full featured 1 RU remote control, for the specific task described in this chapter or for countless other operations as supported by the architecture shown in figure 6-2.

The use of perfectly sized, 4 x 4 linear mixers actually increases the efficiency of large, full featured audio production mixers. Simple tasks are removed from the schedule of the production tool. In fact, small mixers can be strategically installed in the tape room itself.

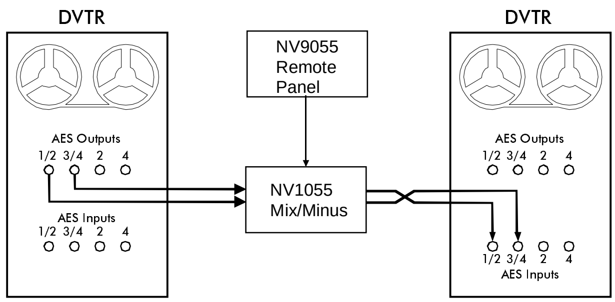


Figure 6-3: NV1055 with remote

The inescapable situation described in this chapter occurs all too frequently. The NV1055's architecture is one that allows an unending variety of production operations and configurations as well as the ability to get the job done now.

Part 2: Digital Audio Breakaway

Serial digital video with embedded audio is gaining popularity, particularly with the broadcast community where easy interconnect and reduction in cabling and connector costs are perceived as great advantages. Digital audio breakaways, however, result in pops, clicks or glitches rather than the desired clean audio transition, but the "ON AIR" digital router must transition cleanly.

For clean transitions to occur, all AES3 data must have the same sample rate. This is usually the case for video broadcast facilities. Additionally, all AES3 signals must have uninterrupted framing and exact phase alignment. Serial digital routers do not guarantee this relationship when a switch is made. Therefore, it is necessary to extract the digital audio from the serial data stream, align it, execute the transition and insert the resulting data back into the serial video data stream. Equipment known as embedders and disembedders handle the multiplexing and de-multiplexing of the serial audio and video data, but not the audio transition. Unfortunately, most disembedders corrupt the AES3 fame so even using analog transitions results in less than satisfactory results. The costs associated with audio breakaway also deserve careful analysis in any plant design. Audio embedders and disembedders are expensive as are the additional costs of A/D and D/A conversion if analog transitions are used. Figure 6-4 shows the conceptual audio break–away process.

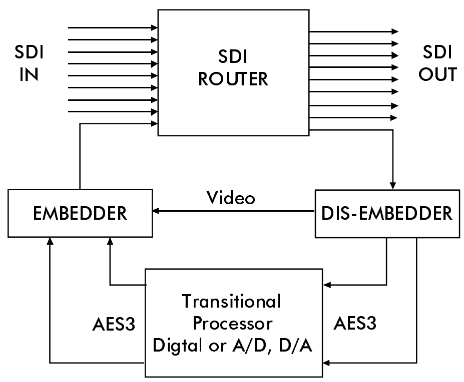


Figure 6-4: Digital audio breakaways

If any of these three conditions is violated, a transition between AES3 signals generates a glitch. Synchronization is easily provided as described in Chapter 15. Phase alignment is achieved as described in this chapter. Unfortunately, the disembedder violates the most important rule. AES3 framing is interrupted when a video switch occurs. It is still possible, however, to implement clean audio transitions with another approach.

The NV1055 can be used as a separate layer of audio routing in parallel with the video router, AES3 transitions and breakaways are executed on non-embedded audio inputs. After processing, its outputs are fed to embedders and then on to the serial video router for distribution. Because the NV1055 has digital mixing capabilities, a true transition with fade is possible. The audible result is just like analog. For larger applications, the NV3512SA Synchronous AES3 router provides uninterrupted AES3 framing and increased connectivity as well as the opportunity to share the transition and channel swapping capability of NV1055 modules. More information about Synchronous AES3 routing is provided in Chapter 9.

Clean "ON AIR" transitions. A transition between two, 4 channel DVTRs is implemented as shown in Figure 6-5. The digital audio outputs of each DVTR can be swapped and mixed in the first layer of NV1055s. The second layer executes the A/B transition.

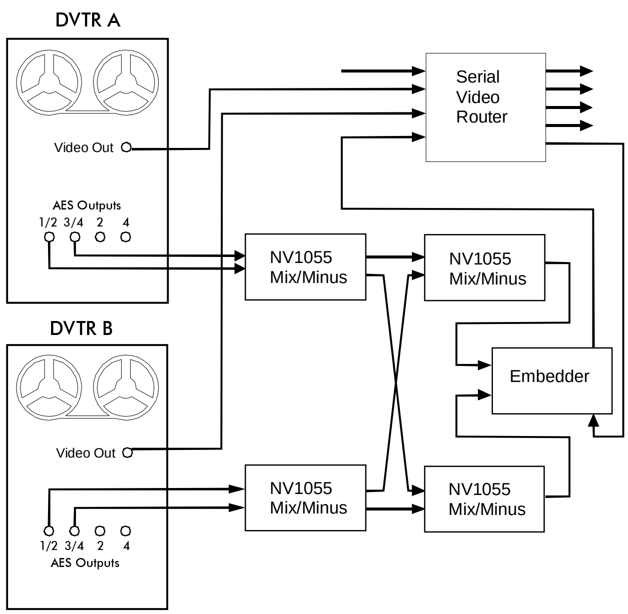


Figure 6-5: Clean audio breakaways

Larger configurations are also possible. Figure 6-6 shows an NV3512SA and a cluster of 4 NV1055's configured as in the previous application. The transition unit now becomes an easily allocated, shared resource. The 512 square matrix size of the NV3512SA enables the construction of extremely large routing systems with a sizable number of shared resources. Clean, digital audio transitions are possible, even in a serial video environment.

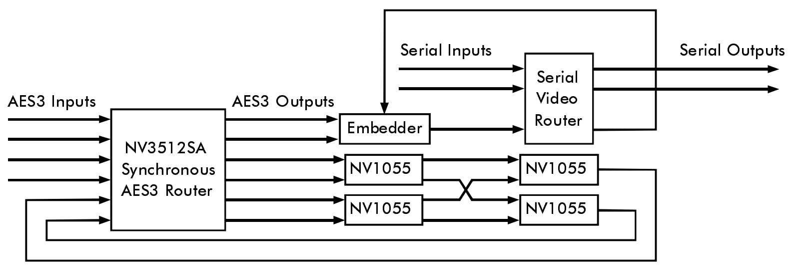


Figure 6-6: A large audio breakaway configuration

Chapter 7. Audio Sample Rate Conversion

The boss has demanded that all audio transfers be digital. After all, he says, digital technology is the best, our clients demand it, and we cannot afford to lose our business to the "all digital" guys down the street. Unfortunately, clean, glitch free transfers of digital audio aren't easy unless source and destination equipment are frequency locked to a common reference; an option that simply is not available in every situation. Some equipment locks only to video, other equipment locks only to audio. Worse, is equipment that cannot lock to any external reference. When frequency locking is not possible, audio sample rate conversion provides clean, high quality, digital transfers.

Video production facilities meet the locking issues head on when they tackle multi-format video transfers. These transfers between NTSC, PAL and Film generate incompatible digital audio sample rates. Audio for video production provides more incompatibilities. CD players and RDATs may not lock to video and video equipment does not lock to digital audio signals or references. Audio production is often carried out at 44.1 kHz. Even when locked to a common video reference, something must be done to make 44.1 kHz material compatible with the 48 kHz sample rate used for professional video applications. Digital audio rate conversion transforms input signals ranging from 28 kHz to 54 kHz into output signals anywhere in the same frequency range; a range large enough to guarantee successful transfers for virtually any application. The NV1050 provides all these features and full digital processing quality that surpasses the analog alternative.

Digital audio sample rates generated by video equipment are locked to input video reference timing. For sample rates to be identical among video equipment, a common video reference must be shared. The NV5500 provides this capability for multi-standard video equipment. Chapter 15. offers more details. But even the NV5500 is not a complete solution. The examples that follow show two different types of NTSC to PAL transfers.

A Simple NTSC to PAL Conversion. The facility shown in figure 7-1 primarily produces NTSC material. Only infrequently are standards conversion to PAL performed. While an NV5500 could synchronize the PAL recorder to an NTSC reference, the NV1050 offers a less expensive, more versatile solution for the occasional user. The NV1050 converts all 4, 48 kHz digital audio outputs of the NTSC recorder to the slightly different 48 kHz sample rate required for input to the PAL recorder. An AES3 output of the PAL recorder precisely backtimes the NV1050 with the correct rate. The NV1060 shown in this application, compensates for the processing delay of the standards converter.

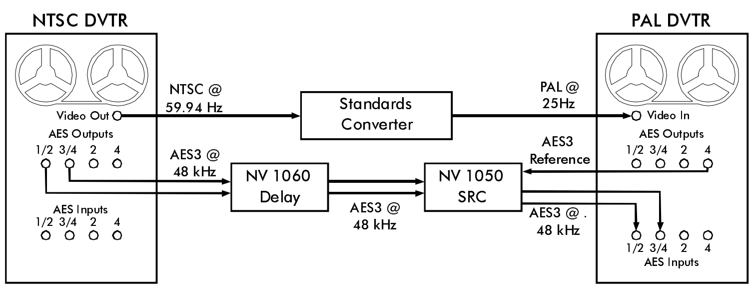


Figure 7-1: A standard NTSC to PAL transfer

A Film to Video to Video Conversion. A previous telecine transfer generated an NTSC tape with 3/2 video pull-down. A normal standards conversion of this material to PAL results in excessive judder, picture "jitter". The video converter shown in this example executes a 2/3 drop followed by either 2 or 4 field interpolation to complete the video conversion process. The PAL output of the converter operates at 24 Hz, less 0.1%, a shift which results from starting with 59.94 Hz NTSC. The corresponding digital audio rate for 23.97 Hz PAL is 46.0339 kHz. These are the rates at which the PAL material is recorded. Play back then occurs at normal speed, 25 Hz and 48 kHz. This process is very similar to a direct telecine transfer from film to PAL. The telecine, however, is not required, saving significant expense. Figure 7-2 shows the NV1050 and an NV1060 connected between two DVTRs. The NV1050 converts the 48 kHz audio outputs of the NV1060 delay to 46.0339 kHz for the PAL recorder. The NV1060 compensates for the delay of the standards converter. Notice that the configuration is identical to that shown in figure 7-1 except for the backtiming. Because 23.97 Hz PAL sync generators are not easy to find, this application only works when the NV1050 and the PAL recorder are locked to the “slow” PAL output of the standards converter. The NV1050 offers extremely flexible backtiming support.

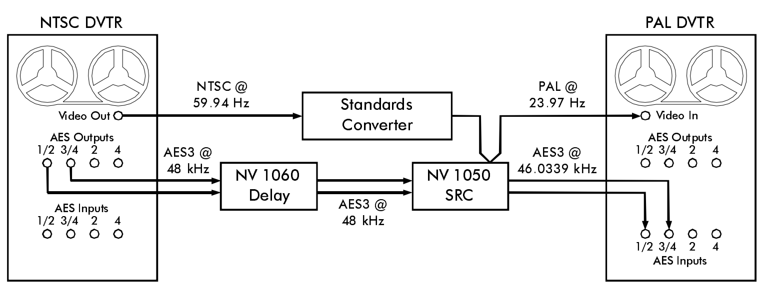


Figure 7-2: Sample rate conversion for complex video transfers

Multiple source rate conversion is often encountered in audio production. In figure 7-3, a non-lockable RDAT master contains field recorded audio which must be combined with CD sound library material as part of the production process. The CD operates at 44.1 kHz, the RDAT at 48 kHz and the DVTR at 48 kHz also. None of the equipment, however, is locked. The NV1050 can separately lock to each input and convert them to a common sample rate determined by a backtiming reference from the DVTR. In this case, an AES3 signal is shown for this purpose, however, a plant master video reference, such as the NV5500, could also be used. The NV1055 following the sample rate converter provides a 4 x 4 linear mix to finish the production. Chapter 6. contains more information on the NV1055 and its capabilities.

Clean, efficient, digital audio sample rate conversion solves numerous transfer problems easily and economically. Rate conversion is an inescapable process when using digital audio in video environments. With all digital performance, automatic configuration and 4 channel I/O, the NV1050 will pay its way in an extremely short period of time.

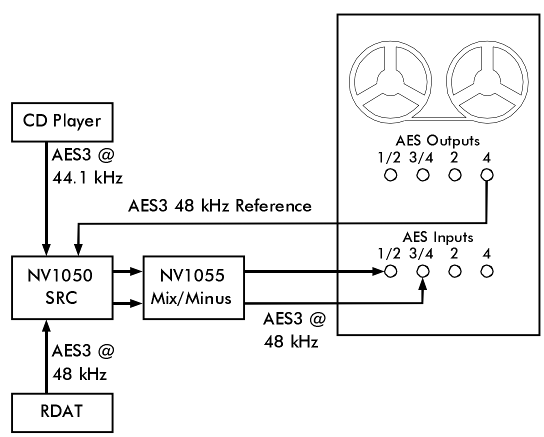


Figure 7-3: Multiple source rate conversion for audio production

Chapter 8. Slaving Routers to Existing Control Systems

"Why should I have to junk my existing router control system just because I want to upgrade to digital routing technology?"

An excellent question. An existing router control system is a significant investment indeed. Routers are installed and configured. Addresses are assigned. Panels are programmed, racked and cabled together and setups are stored. In addition to this investment of time and energy, the existing router control system is familiar. Operators use it to get the job done. It is a productive tool. The installation of digital routers should not mandate the installation of a new control system. NVISION supports this philosophy.

That is why NVISION happily provides all of its routers with a Control Interface Module. (NVISION AES, Time Code and Data Routers are described in chapters 9 through 12.) This module is capable of interpreting the commands of any number of router control systems. Serial RS-422 and hardware parallel interfaces are supported, allowing NVISION routers to function on any level, within any address space you choose.

"Many hours are invested in programming panels with 17 layers of configuration menus and cryptic command lines with weeks lost waiting for new EPROMs from the factory which are needed to re-configure your router. There must be a better way."

There is. NVISION routers all include a software utility package; NVUtils. This software operates on any IBM PC compatible computer and communicates with NVISION routers through the RS-232 diagnostics port located at the front of the frame on the Control Interface Module. An exciting program contained in NVUtils is NVMap. It allows the configuration of all NVISION router options.

1. Use NVMap to configure the level of the router. If two partitions are desired, two levels can be created. This is very useful in creating AES3 ½ and ¾ layers. It is also helpful when time code and asynchronous AES3 routing modules share a common frame and control interface module.

2. Use NVMap to assign logical addresses to the actual physical addresses labeled on the back of the router. This is useful when the existing routers and control system are capable of executing a multi-level or layer salvo with one single take command. Since the new digital router will most likely have physical addresses which don't match the existing, corresponding controller map, the new router ports will have to be logically reassigned to match the existing. NVMap lets you do it right now. No factory proms, no waiting.

3. Use NVMap to logically link layers. Again using a 2 layer, AES3 example, NVMap can assign a link between pairs of inputs and outputs, logical as well as physical. When one output is addressed as part of a take command, its matching output is recalled from memory. Similarly the matching source is also found. Both takes are automatically executed by the NVISION control interface module. A second virtual layer has been created. This is extremely useful for audio follow video operations. Additionally, the combined use of logical reassignment and links allows the creation of two new layers within an existing router control layer. Specifically, some router control systems only allow 4 layers. Using the combination technique just described, two layers of 64 x 64 digital audio can exist with one analog layer of 64 x 64 audio in one

level. No special hardware or upgrade to your existing control system is required.

4. Use NVMap to create internal virtual layers. This is really HOT. For example; feature 3 is used, creating a linkage between AES ½ and AES ¾, however, flat access is also required. The flat router address space can be configured as one level, the linked address as a second. The priority is then selected for the structure that is preferred, should a conflict arise. The NVISION control interface module sorts out the conflict, generates the takes and properly configures the matrix in its new, clean state. As a specific example, assume an NV3064A provides two layers of 32 x 32 AES3 which are logically linked most of the time. Occasionally, the matrix needs to be a flat 64 x 64 for audio production. Configure the linkages for the two 32 x 32 layers. Set this layer for priority 1. Then assign a second layer to the same addresses. When a take to the second layer is executed, the linkage search is ignored. Only one source is taken to one destination. If the destination is in conflict, the route doesn't happen, its priority is too low. Conversely, when a linked take encounters a conflict with either of its destinations, the lower priority router is undone and the new routes are executed.

5. Use NVMap for DATA router broadcast connections. NV3128D/NV3256D routers allow many destinations to listen to one host with only one destination responding. Use NVMap to create a virtual layer much like that described for feature 4. In this case however, the NVISION control interface module uses command stream context to determine the source, the listening destinations and the acknowledging destination. Because it has dynamic ports, the NV3128D/NV3256D is easily configured exactly this way. If the take to the new level is in conflict with standard configurations, the command is not executed. If a take to a standard level conflicts with a broadcast route, the

broadcast route is undone and all crosspoints and ports are returned to their correct state. The standard level take automatically executed as well.

NVMap also provides router diagnostics. The internal state of the router can be dumped from the control interface module into the PC host for analysis. Command response status is provided as indication of control system activity and viability.

“What do I do for control if I never owned a router before?”

The other feature of NVUtils is NVTake. This IBM PC software provides complete router control. Single takes and salvos can be executed. Salvos can be written and stored. When ready for use, recall the salvo and execute it with a single key stroke. Alternatively, for larger installations, the NV9000 series of networked control panels are available to provide total control over NVISION routers, delay cards and mix/minus modules.

Slaving routers to existing control systems is often the smartest thing to do. It saves time, money, effort and sometimes provides features not available with new router control systems.

NV3064, NV3512, NV3128 and NV3256 Control Interface Module Specifications

All NVISION control interface modules are designed with a common feature set independent of router family or command stream format. Following are the key features of these modules.

The entire router matrix status as well as configuration options selected with NVMap are stored in SRAM with capacitor backup. If the control system fails, the last known map is continuously refreshed within the router.

This information is shown in table 8-1.

Matrix Refresh Time	512 outputs in less than 16 msec, continuous.
Matrix Storage Element Retention	20 minutes guaranteed, 12 hours typical.
Matrix Initialize on Power Up	Last known state/hard coded default, selectable.

Table 8-1: Matrix memory dynamics

The timing of router transitions is often critical and every effort has been made to insure that NVISION routers always execute the requested transition. Possible reference inputs and their priority are shown in descending order in table 8-2.

Reference Priority 1	Video Vertical, NTSC or PAL
Reference Priority 2	AES3 External Frequency Reference
Reference Priority 3	Internal Free Running AES3 Oscillator
Reference Priority 4	Software Watchdog, Execute each Command as Received

Table 8-2: Transition timing priority structure

Many existing audio/video router command streams are supported. Odds are that the one you need is listed in table 8-3. If not, please contact our Technical Support Department. NVISION is always interested in helping you maximize the potential of your equipment investment. Table 8-3 shows the router protocols currently supported. When ordering an NVISION router, always be sure to specify the desired protocol.

Adding the enhanced features of digital routing shouldn't necessitate a grinding halt to your productivity, or consume precious hours of operators valuable time for training. The economy and convenience as well as the feature enhancements available with command interpretation may well extend the lifetime of your existing control system.

Serial RS-422

MFG.	FORMAT
BTS	Remote Binary Protocol ES-Bus
ProBel	RS-422/RS-485
Utah	AVS1, AVS2 PL320, PL160
Vistek	RS-422

Serial or Parallel Hardware

MFG.	FORMAT
GVG	Horizon
PESA	37 Pin Parallel

Table 8-3: Currently supported router protocols

Chapter 9. Routing AES3 Signals

"Audible glitches disrupt my plant whenever the router switches. I read the NVISION notes on synchronization and I know that my facility is correctly timed. What am I doing wrong?"

You probably aren't doing anything wrong. The question you should be asking yourself is: "Do I need Synchronous or Asynchronous Routing?".

Equipment synchronization guarantees transparent, direct digital transfers in operations where AES3 framing is uninterrupted. Once equipment such as async routers and patch bays are inserted in the signal path, pops and clicks are sure to appear.

AES3 digital audio is now the standard format for video facilities, nearly all of which incorporate routing switchers for flexible interconnect of devices and sharing of expensive resources. Since there are a number of routing options available; synchronous, asynchronous, embedded and analog, understanding the advantages and disadvantages of each is paramount, whether the design task at hand is building a new facility, adding a digital layer to an existing facility or something in between.

Asynchronous describes digital audio routers and other equipment operating without a common frequency reference and/or phase relationship.

Embedded describes a serial data stream typically containing 4 channels of digital audio (2 AES3 data streams) and one digital video signal.

Synchronous describes AES3 equipment capable of locking to a common reference frequency. Sometimes, phase aligned operation is included as an operational characteristic. Synchronous routers have both properties. They are, in a word, isochronous.

Asynchronous routing of AES3 data with products such as the NV3064A or the NV3512A currently enjoys the greatest popularity, indeed many manufacturers only supply this type of router. Its X-Y architecture is the least expensive alternative, accommodates any audio sample rate and has the least impact on plant integration. There are, however, some drawbacks. AES3 frames are truncated during the router transition. This interruption of the AES3 framing pattern generates audible glitches in downstream equipment. Equipment with incompatible sample rates may be connected together, resulting in incorrect operation. Some equipment types, such as RDATs and CD players, use internal oscillators with a large frequency range, other equipment, such as D/A or A/D converters, use crystal oscillators with a narrow frequency range. Connecting a large range source to a narrow band destination is not reliable, the destination may not acquire lock. Operators will have to contend with these situations. Synchronization of equipment connected by the router reduces the occurrence of this phenomenon to true mistakes; connecting a 48 kHz device to 44.1 kHz machine for example.

Embedded routing is much like asynchronous routing. When the serial video data stream is switched, the AES3 framing information is interrupted. Embedded formats are popular in broadcast environments where reduced cabling and connectorization costs are perceived as great benefits. Unfortunately, the glitches associated with AES3 frame interruptions are not so well received. Broadcast facilities, however, operate in a synchronous environment. Figure 9-1 shows a suggested circuit for the execution of clean audio transitions within an embedded system. Embedders, a synchronous AES3 router and often a small mixer are used for this purpose. Embedded routers are also X-Y space matrices, however, their cost exceeds that of asynchronous routing by 2 or 3 to 1. Chapter 2 provides more detail on embedded audio break-aways.

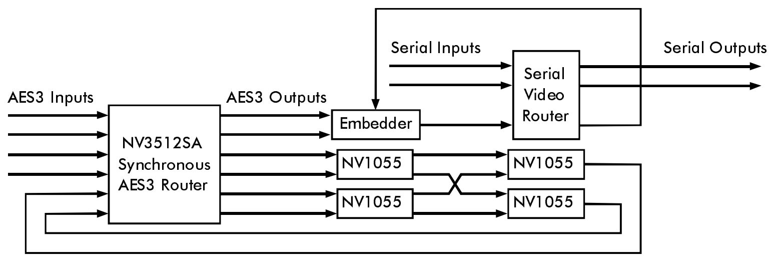


Figure 9-1: Clean embedded audio breakaways

Synchronous AES3 routing, as provided by the NV3512SA, NV3064SA and NV1308SA, eliminates pops and glitches provided that all inputs to the router are synchronized to a common sample rate. Chapter 15 shows how easy this can be. A few frequency synchronized input signals, shown in figure 9-2a, are phase aligned relative to an internal AES3 router reference. The resulting signals, shown in figure 9-2b, may now be routed without disruption of AES3 framing. Audio is essentially stripped out of one signal and inserted into another. The output framing format is uninterrupted eliminating downstream pops and clicks. Clean router transitions are executed at video vertical as shown in figure 9-2c. Vertical interval is used as a setup strobe for the next AES3 frame boundary. Video slaved takes occur within one AES3 frame, 20 microseconds for a 48 kHz audio sample rate, of the vertical interval pulse introducing an acceptable maximum delay of 1/3 of a video line. Synchronous AES3 routers are either X-Y space matrices or Time Division Multiplexed (TDM) matrices, sometimes called bus routers. TDM router architectures are practically limited to 256 square by bus bandwidth economical

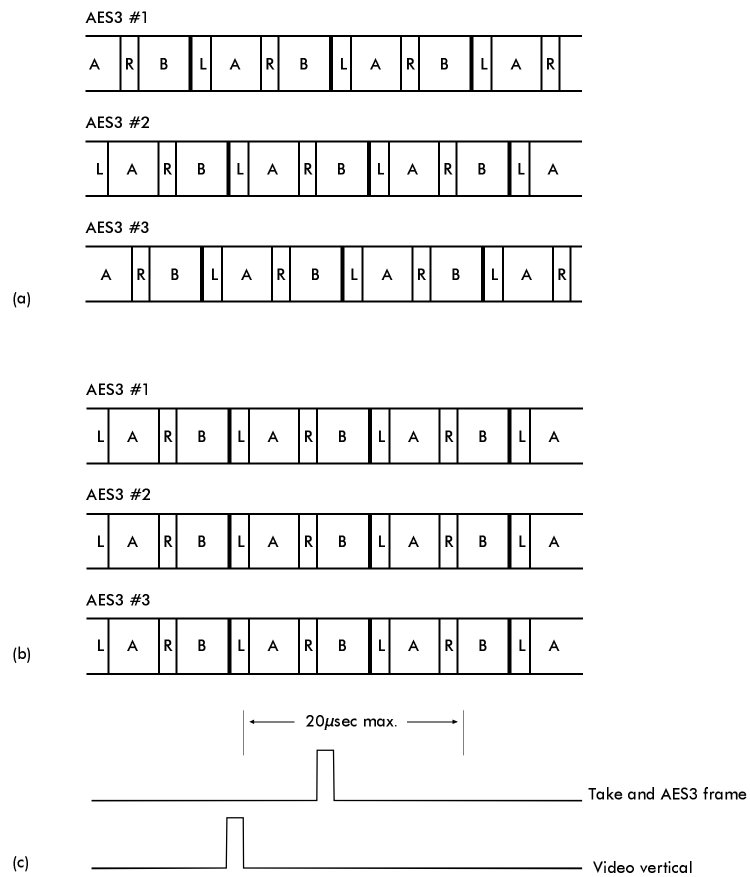


Figure 9-2: Phase and timing relationships for AES3 sync routing

considerations. The NVISION routers described in this section are all X-Y matrices. Synchronous routers are typically 50% more expensive than the Asynchronous option.

The key features, pros, cons and primary application for the three digital alternatives are:

1. Asynchronous

Key: Accommodates any sample rate

Pro: Inexpensive

Con: Pops and clicks on HOT switches

Use: Pro audio, audio post, no live audio switching

2. Embedded

   Key: Digitally multiplexed Audio and Video Data

   Pro: Reduces Cabling and Interconnect

   Con: Expensive, Pops and clicks, No internal breakaway.

   Use: Broadcast, distribution, Very ineffective for audio post

3. Synchronous

   Key: Uninterrupted AES3 Framing

   Pro: Clean, live switching

   Con: Operates at only one sample rate

   Use: All applications, except multi-frequency audio post

Since virtually all new digital audio and video equipment uses the AES3 format, purchasing a new analog router is the least progressive choice, particularly with the handsome savings in space, power and cost offered by digital matrices. Analog material, however, is inescapable. Figure 9-3 shows the integration of an analog and digital router. A/D and D/A conversion equipment is included to bridge the two formats. The number of A/D and D/A converters required depends on the population count of shared machines. For new installations consider a single digital matrix and converters rather than two routers. The tradeoff is purely economic. Router crosspoint count grows geometrically as port count grows linearly. Converters are not inexpensive, still, some will be required in any case. Rare indeed is the facility which does not share material across the analog and digital boundaries. The true cost of conversion equipment is the difference of all the converters used for an all digital implementation less those needed for a hybrid, analog digital configuration. A digital router costs far less than its analog counterpart. Include the costs of space, power and cooling and the savings are significant.

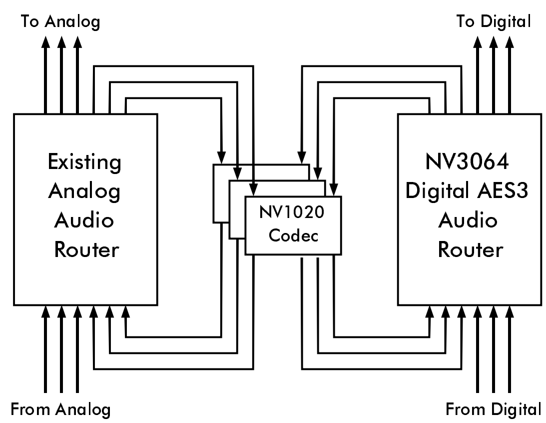


Figure 9-3: Mixed analog and digital audio routing

Adding an independent layer of digital audio into an existing facility is simple. Determine the dimensions of the router you need, find the best combination of price and features, and slave it to your existing control system as an additional layer. Chapter 8 describes how all NVISION routers are slaved to virtually any new or existing router control system.

Planning Router Requirements

Step 1. Determine the router size needed for analog equipment, digital equipment and the analog/digital crossover. If the crossover exceeds 50%, use an all digital router. For calculation, use stereo audio cross points. AES3 digital accommodates stereo in one signal.

Step 2. Add the number of crossover inputs and outputs to the dimensions of the analog and digital router matrices in step 1.

Step 3. Calculate the cost of the stereo analog and digital routing matrices.

Step 4. Add the cost of Stereo A/D and D/A conversion used to bridge between the two formats. Use one A/D and one D/A for each stereo crossover.

Step 5. This is the raw cost of the mixed format solution.

Step 6. Add the number of stereo analog inputs and outputs to the number of digital audio inputs and outputs.

Step 7. Calculate the cost of this digital audio router.

Step 8. Add the cost of Stereo A/D and D/A conversion for all analog equipment.

Step 9. This is the raw cost of the digital solution.

Step 10. Compare the total in step 9 with that in step 5.

The result depends heavily on crossover size and router expansion costs. Hidden costs are physical router size, power consumption, and control system complexity. One flat router is the easiest to control and the most versatile for plant use. Even if the all digital approach is modestly more expensive, the all digital approach offers five exceptional advantages:

1 Complete accessibility to virtually any piece of equipment or material anywhere in the facility.

2 Absolute forward compatibility. New digital equipment offers AES3 I/O.

3 Future expansion costs are minimized. Digital crosspoints are less expensive than analog.

4 Future expansion is simplified. Replace an old 1 inch with a DVTR and simply bypass the converters. No router control system changes are necessary.

5 Operating costs are lower; less power means less cooling, which means less power again. A significant savings in rack space is also provided with digital matrices.

Your router solution is specific to your operation. No single topology or configuration ever seems applicable to any other facility or operation. The best approach is to carefully analyze your facility and operational practice in order to determine your optimal configuration.

Chapter 10. Small AES3 Asynchronous Router Building Blocks

Small 8 x 8 AES3 asynchronous routers are a great starting point for project studios and small suites, but, how can these building blocks be used to make larger routers?

This question seems innocuous enough, yet, AES3 signals must be handled with care. The high frequency nature of the signal necessitates that interconnections be designed with transmission line practice in mind. A concept which, if not considered from the outset, typically yields equipment designs without the appropriate features to allow easy, reliable matrix expansion by cascading multiple 8 x 8 routers.

Let's start with the basics. Conceptually, four 8 x 8 routers are connected as shown in Figure 10-1 to form a 16 x 16 router matrix.

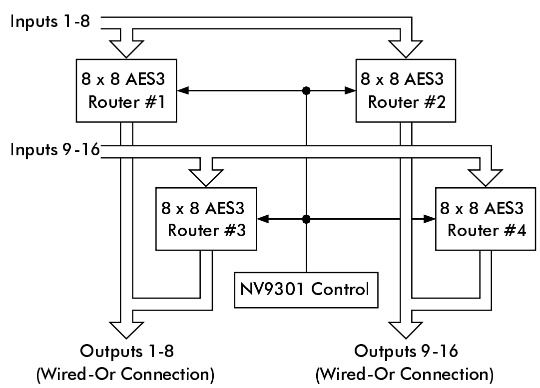
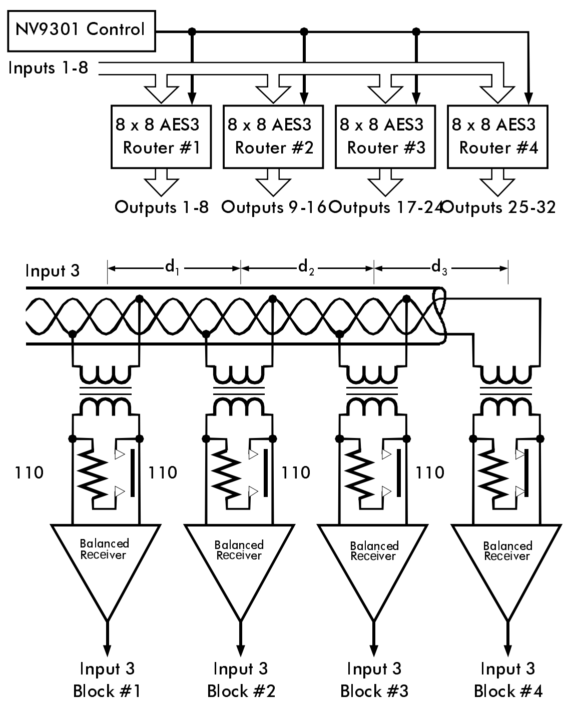


Figure 10-1: A 16 x 16 matrix with 8 x 8 building blocks

Note the "Wired-Or" connection at the outputs of the building blocks. Used in place of a 2 x 1 crosspoint, the wired-or connection provides similar functionality with lower cost. This type of connection requires each output of the

8 x 8 building blocks to have a high impedance state when the driver is turned off. Certainly, this feature is easy enough to include, however, it requires an extra level of control sophistication. Before discussing control, it is important to consider the design of the input and data busses which bind the 8 x 8 building blocks together. For this discussion, Figure 10-2 shows four, 8 x 8 modules configured as an 8 x 32 array and Figure 10-3 shows four, 8 x 8 modules configured as a 32 x 8 array.

(a)



(b)

Figure 2: a) A conceptual 8 x 32 matrix b) A detail of one AES3 input bus

Figure 10-2, an 8 input by 32 output router matrix, is used to demonstrate the concept of input bussing. Figure 10-2a shows the conceptual construction of the 8 x 32 matrix with 8 x 8 blocks. Figure 10-2b shows the detail of the receiver for input signal number 3. Transmission line practice requires that the termination for the input be at the end of the bus and that it be equal in value to that of source and the transmission line itself. Notice that inputs with selectable input impedance provide exactly this functionality. Typically, the distance between inputs, *di*, should be kept small compared to the length of the cable run from the source to routing matrix. This minimizes the effect of any reflections generated by the finite input impedance of each receiver and other impedance mis-matches in the cable and driver.

Figure 10-3, a 32 input by 8 output router matrix, is used to demonstrate the concept of output bussing. Figure 10-3a shows the conceptual construction of the 32 x 8 matrix with 8 x 8 blocks. Figure 10-3b shows the detail of the output driver for output signal number 3. Transmission line practice requires that a constant source impedance be provided at the origin of the signal driving the transmission line. Since each driver has the ability of operation in a Hi-Z mode, the source impedance of the active output is always 110 ohms. However, each output is tied to the bus with a piece of cable whose length is *di*, and all but the last output have a stub of wire with a high impedance load to contend with. This stub is not impedance matched and a significant reflection is generated at the end of the stub. In fact, this reflection is capable of causing bit errors in the data stream. In theory, it is possible to place a fixed impedance at the end of the bus, adjust the impedance of each section of connecting cable, and select the source impedance of each driver so that the entire transmission line is matched regardless of which driver is active. In practice this is often impossible due to the availability of cable and the signal levels required on the transmission line. A practical solution is to keep the cable length small. For the example shown, the total length of all the *di* should be kept to less than 2 feet for 48 kHz AES3 signals.

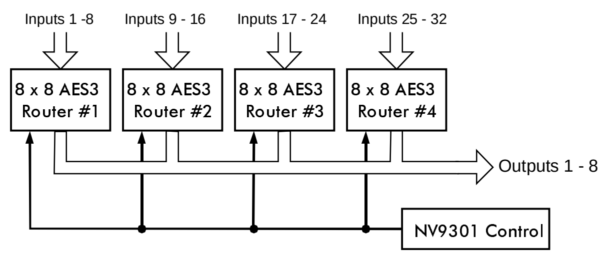


Figure 3a: A conceptual 32 x 82 matrix

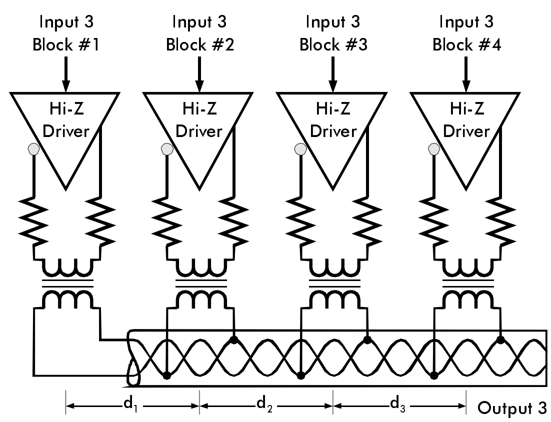


Figure 3b: A detail of one AES3 output bus

The principals shown in these two examples may be applied to most any expansion dimension desired.

Each of the conceptual drawings indicated a common control signal for router configuration. While it is possible to have separate control for each of the four blocks shown in the previous three examples, it is cumbersome to manually execute 2 or 3 takes, plus Hi-Z control, in order to direct a new signal input to a given output. For this reason, it is highly desirable that each 8 x 8 building block be uniquely addressable. Each block is then configured based on the inputs and outputs associated with its own address. This permits an external intelligent controller to execute all the necessary commands to configure the desired router output while only necessitating one simple command input from you, the operator.

Using small building blocks to make larger routers is possible, provided that input impedance selection is available and output bus lengths are kept short. Additionally, the use of the composite matrix is greatly simplified with addressable building blocks and a single point of external control.

The techniques described in this section only apply to asynchronous routing. In the case of synchronous routers 'Wired - Or' connections are not possible due to signal timing considerations. The NV1308SA 8 x 8 synchronous router, is supplied as standard with a 16 x 8 backplane, so that it can be readily expanded to this dimension. For larger synchronous routing applications, consider the NV3064SA (32^2 to 64^2) or the NV3512 (8 x 32 to 512^2).

Chapter 11. Time Code Routing with Signal Processing

If Time Code is so easy to route, how come I'm having such a problem?

Time Code signals, vital to the effective operation of any video facility, are often so badly degraded by the time they reach their destination, that accurate recovery of the signal is impossible. Traditionally, analog audio routers are used to distribute time code. However, the space and power they consume could be more effectively used particularly since they really fail to accommodate the varying electrical interface requirements of time code signals.

The real culprit is apathy. Until it's broken, users always have something more important to spend their money on. Equipment manufacturers seem to have decided that since normal speed time code, uses less than 20 kHz of bandwidth, analog audio routing equipment is acceptable for this application, thus saving the expense of designing a product just for time code. Even the SMPTE standard specification for longitudinal time code, 12M-1986, fails to adequately define the electrical interface of the signal. So manufacturers of various types of video equipment use either single ended BNC or balanced XLR interconnection circuitry. To further aggravate the situation, most VTRs, as a cost savings, provide raw time code output signals directly off the tape. Instead of the trapezoidal waveform defined in SMPTE 12M, a sinusoidal signal is supplied. To make accurate recovery even more challenging, the amplitude of this signal varies with tape play speed; higher when faster, lower when slower. Unfortunately, the result is that power hungry, space consuming equipment, incapable of consistently routing accurate time code has been the only available alternative. Until digital technology provided one.

Longitudinal time code is really a data signal. Its data rate is 2.4 KB/s which when coded as a bi-phase signals yield nominal frequencies of 1.2 and 2.4 kHz. As mentioned, these frequencies vary proportionately with tape play speed as does signal amplitude. The electrical interface is implemented in two different ways, but these characteristics can be accommodated with cleverly designed, digital interface circuitry.

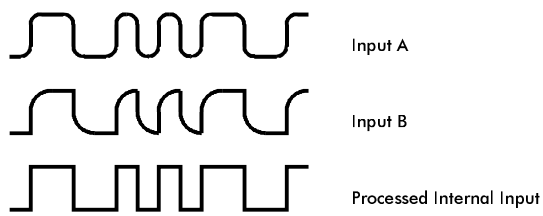


Figure 11-1: Time Code Input Signal Processing

Two step processing of the signal allows an “analog” input to be routed internally in a digital format and produced once again as an “analog” signal at the output. Figure 11-1 shows two typical time code signal wave shapes. The amplitude of these signals may be as low as 200 mV peak to peak or as high as 10 V peak to peak. The frequency can vary from 30 Hz for tape jog to 100 kHz for high speed tape shuttle. The key is to convert these input signals to a square wave as shown. All timing information, the only information needed, is preserved in this digital representation.

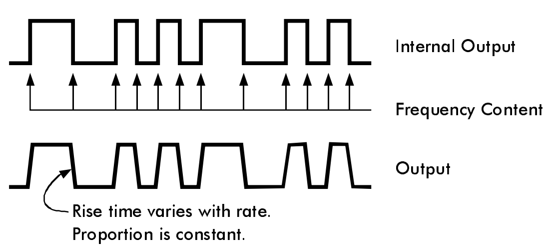


Figure 11-2: Time Code Output Signal Processing

After routing, the signal must once again "feel" analog. Figure 11-2 shows this conversion process. The signal frequency is actually measured for each output. This information is then used to generate a fixed amplitude, trapezoidal waveform with proportionately controlled rise times at any tape speed from 1/30 normal to 100 times normal play speed; a signal which is always textbook perfect.

Seemingly incompatible interconnect requirements really aren't, if the proper interface circuitry is used. A wide bandwidth, balanced input amplifier accommodates either single ended or balanced signals. For single ended inputs, tie the shield and inverting connections together, this will not affect the input sensitivity. Output circuitry incorporates electronic virtual transformers. Tie the inverting output and common together without reducing signal amplitude. Using a router which incorporates this type of interface circuitry provides the additional advantage of equipment isolation. The chance of accidentally damaging equipment by cross connecting the ground and signal pins of mismatched machines is eliminated. A simple example of these connections is shown in figure 11-3.

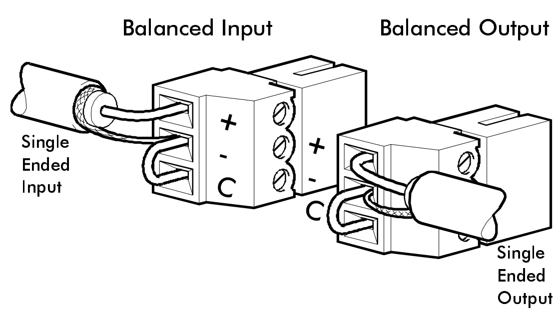


Figure 11-3: Balanced and Single Ended Time Code Connections

The NV3064TC and the NV3512TC contain all the features described. The NV3064TC provides a 64 x 64 matrix in 6

RU and consumes 45 Watts. An analog audio router requires up to 18 RU. and 500 Watts. Digital routing provides a 10 fold savings in power costs. If the associated costs of air conditioning equipment and its operation are included, the benefit is even greater.

The NV3512TC router is designed for the largest facilities. Sizes ranging from 8 x 32 to 512 x 512 in 40 RU are provided with the same powerful feature described in this chapter. The NV3064TC and NV3512TC are easily slaved to your existing router control systems, simplifying installation and eliminating operator training. Digital time code routing offers an exceptional alternative to previous practices.

Chapter 12. Machine Control Routing

"Let's see now. The tape machine is controlled, so, it is connected to the router controlled output, which, because its an output, is really controlling. The editor driving the tape machine is controlling, and, as such, is connected to a controlling input which is called controlled because it is receiving the data. Therefore, the tape machine is controlling the editor."

"That's what's wrong. Hey, I better run down to the machine room and get this edit finished."

As difficult as it may seem, patching and even routing RS-422 data connections is a daily routine in any facility. Changing the sense of machines so that they can be controlled by an editor for one session and then be the boss for duplication that night is no easy task. Special cables which swap pin pairs must frequently be located and carefully installed. Switches or cable positions on the back of routers and equipment, must be set or moved for the exceptional job, and then returned to their normal positions. Any special tasks such as multiple tape laybacks or controlling more than one machine with a common edit port require special daisy chain cables that are always lost, and extra hours of configuration time spent in awkward positions behind equipment racks.

The simple solution: use relay routers. There aren't any directional electronics in the signal path to get in the way.

In 1960 perhaps, this was the answer, but not in the 90's. Relays are expensive, consume lots of real estate and use even more electricity. Digital electronics offer better functionality at 1/4 the cost, in 1/3 the space with 1/10 the power. Relay routers still require 2 matrix layers, one for the commamd data direction and one for the response.

Additionally, many relay routers fail to eliminate bus contentions which occur when both RS-422 connectors of a machine, typically a VTR, are connected to a router.

No, relay routers are not a solution. The real answer to the problem lies in understanding data routing requirements and machine interactions, then designing a router matrix to fit the needs of the application. Virtually every other data router on the market tries to modify an existing X-Y space matrix approach. The NV3128D/NV3256D is different.

A control data communications interface complying with SMPTE 207M is bi-directional. One data stream allows the master to control the slave, the other carries the slave's response back. In order to avoid cables which swap signal pairs, the output pins of one connector become the input pins of its mate. This yields the often confusing terminology of controlling and controlled which refers to the directionality of the signal on the two active pin pairs defined in SMPTE 207M. Figure 12-1 shows this definition.

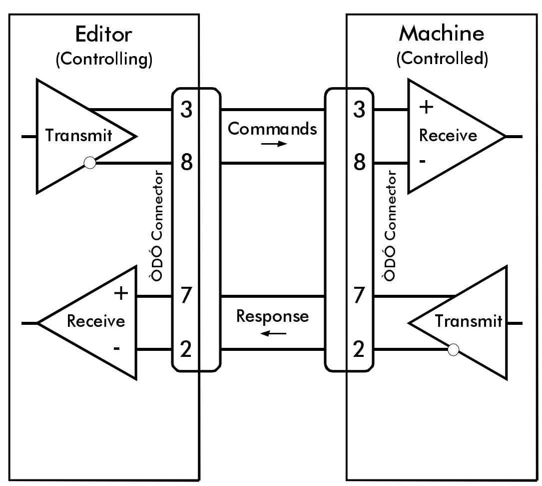


Figure 12-1: SMPTE 207M Port Definition and Pin-Out

Editors typically provide 8 to 16 connectors for controlling machines such as VTRs. Since editors are nearly always the master, signal direction and pin function don't change. VTRs behave differently. They have control panels, and so, while most of the time they are indeed slaves, they can also be the master. This means that pin functionality is swapped. Early BVU-era equipment provided only one 9 Pin D connector for control connections. Since the sense of the connector, or port, changed, routing was impossible. Drivers would be connected to drivers and equipment would be damaged. So a second connector was provided, one with a reverse pin out of the first. Now, both connections could be made to two router layers wired in opposite. One for the controlling case, the other for the reverse.

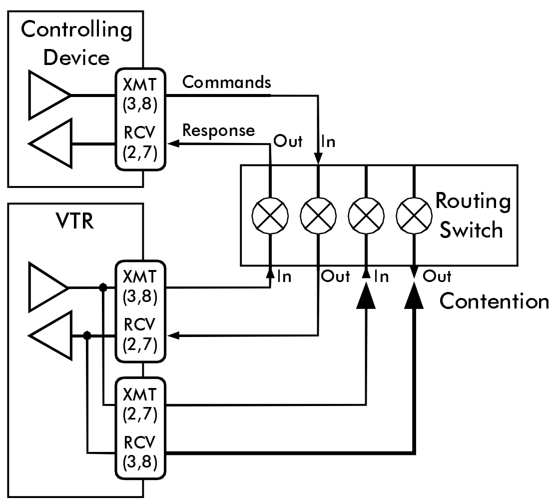


Figure 12-2: RS-422 Data Contention with Standard Routers

Receiver and transmitter circuitry inside the VTR is also capable of either controlling or controlled operation, so this approach simply had to work. Wrong again. The reversing connection between the two ports on the VTR is hard wired. When both VTR ports are connected to their respective layers of the router, the desired output signal of the router is fed to the VTR and looped right back out to the other router layer. Unless the router is capable of presenting a Hi-Z impedance to either VTR port, bus contention causes the equipment to lock up as shown in figure 12-2.

The interesting part about all this is that the electronics inside the VTR is capable of operating in either a controlling or controlled mode on either connector. Circuitry enabling this operation is shown in Figure 12-3. So, if a router can provide Hi-Z impedance to machine ports, as well as controlled or controlling configurations, bus contention is eliminated and only one connection need ever be made between the VTR and the router. This is a powerful concept. In fact it is the basis for developing the NV3128D/ NV3256D Dynamic Port Architecture.

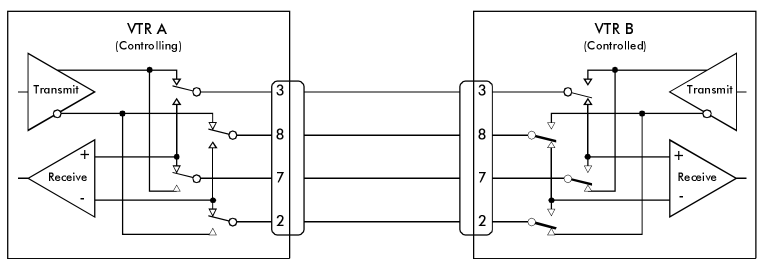


Figure 12-3: Controlling / Controlled VTR Circuitry

Using Dynamic Ports reduces the number of connectors for VTRs from 2 to 1, significantly reducing the space and cost requirement. Additionally, this technique eliminates fixed port configuration of router connections. Previously, data routers were square, two layered dinosaurs; one physical matrix used for controlling connections, and another used for controlled signals. The two layers are permanently configured as is the equipment attached to them. While this scheme works, the versatility of modern equipment is lost. Routers of this type use twice as many crosspoints as those with dynamic ports. So, dynamic ports save money again. In addition, asymmetric routers waste large numbers of crosspoints, and their attendant space and power. With dynamic ports, as in the case of the NV3128D/NV3256D, there are simply a number of devices connected to the router. At any moment the actual configuration could be 17 by 239 or 69 by 187. Crosspoint utilization is optimized because the dynamic port circuit converts two, twisted pair data streams into fixed direction receive and transmit signals. With this powerful architecture, VTRs can be inputs or outputs to the router at any given point in time while editors are always inputs. There is no penalty, particularly since the cost of the dynamic port is quite small.

While other routers claim to offer dynamic ports, they really provide rear panel switches or two cross-wired connectors. This is not a time saver. Dynamic ports need to be automatic. This feature saves the time, pain, agony and angst associated with implementing the proper router configuration for a critical edit session. The NV3128D/NV3256D uses the router control stream commands in context to configure the state of each port. The source is configured to accept a controlling data stream and the destination is configured to provide this information to the controlled device. The reverse connection is also made: one command, two takes. In fact, the first step taken by the router is to remove all previous takes made to the ports addressed by the new take command. This eliminates the potential for bus contention or hung equipment. Again, this is all done automatically.

Timing is very important too. When a given editor port switches from one machine to another, if the switch happens too fast, the editor doesn't realize that it should be speaking AMPEX, not SONY. The result is hung equipment because the protocols do not match. The NV3128D/NV3256D automatically inserts a time out between the drop of the old device and the connection to the new machine. This tells the editor that an equipment change has occurred and a new communications protocol needs to be initialized. The result is painless operator interaction. Figure 12-4 shows a dynamic port structure capable of generating controlled, controlling or Hi-Z configurations.

Figure 12-5 shows a conceptual implementation of a dynamic port oriented router. It provides all the features described and one more: broadcast. The NV3128D/ NV3256D allows many devices to be listeners while only one answers back. Dynamic ports prove their value once again.

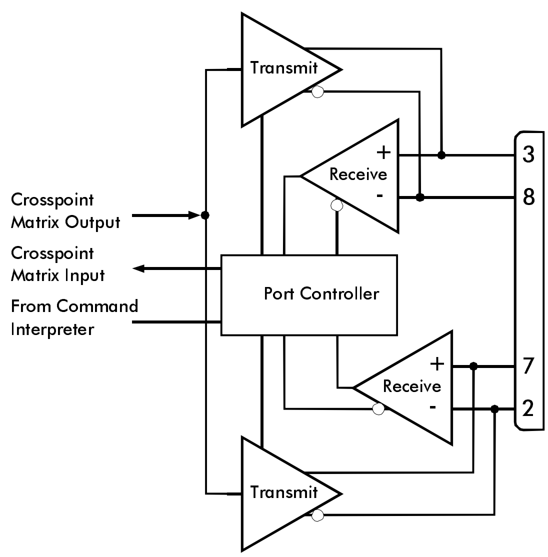


Figure 12-4: Dynamic Port Detail

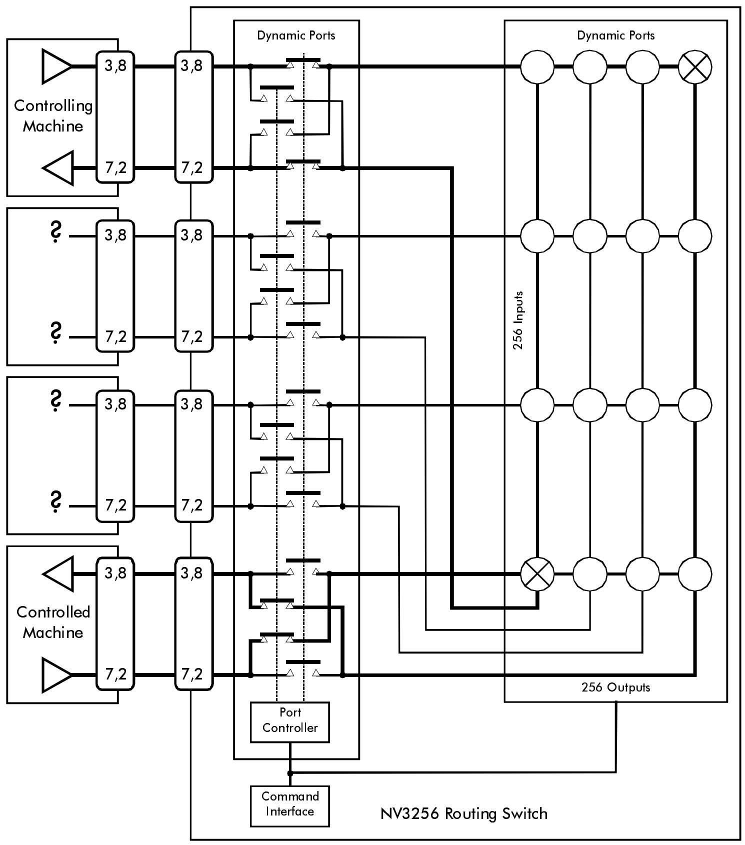


Figure 12-5: Dynamic Ports in an X - Y Architecture

Unfortunately, RS-422 is not the only electrical interface to be found in today's production and broadcast facilities. Personal computers and other peripheral devices such as a laser disc player, or CD Juke box, use RS-232 connections. RS-232 is the interface mainstay of the computer and data modem industry. Its similarity to RS-422 is a bi-directional connection. Its differences, however, are many. RS-232 transmit and receive data use a single conductor, rather than a twisted pair. Additionally, RS-232 equipment also supports a more sophisticated protocol than SMPTE 207M. Additional lines are allocated for handshaking, a process which is used to initiate data transfer from the master to the slave as well as provide status indication of the transmission channel. RS-232 also has a special name for the master and the slave: Data Terminal Equipment and Data Communication Equipment (DTE, DCE). And, just like SMPTE 207M, the direction of signal flow changes if the equipment, or data port, is DTE or DCE. Figure 12-6 shows a basic RS-232 connection for a DTE to DCE link.

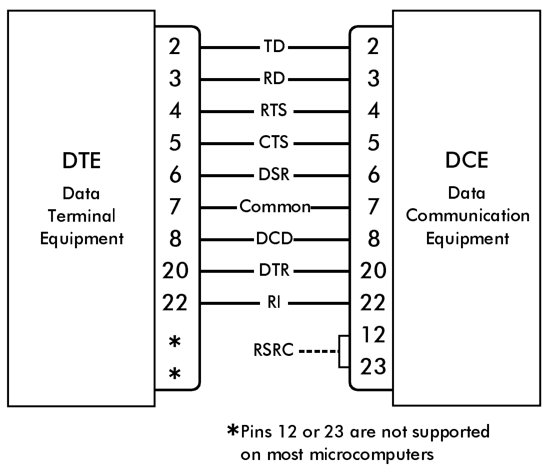


Figure 12-6: Industry Standard 25 Pin D RS-232 Connection

RS-232 ports do not change dynamically. This is simultaneously good and bad news. The good news is that once the port is connected correctly for either DTE or DCE operation, no further intervention is necessary. The bad news is that the definition of DTE and DCE is quite vague, a given piece of equipment is usually never the sense you think it should be. Additionally, in a video production environment, it is desirable to connect two DTE devices to each other. Fortunately, dynamic ports readily accommodate this mode of operation. Dynamic ports can be easily configured to provide a fixed, logical orientation of either DTE or DCE functionality. The port configuration is set with the NVUtils router configuration software. There are no switches to set and no swapper cables to assemble or purchase. Once set, the information is stored in non-volatile memory. The configuration remains fixed until you change. Once the port is configured, transmited and received data become two signals with a pre-defined direction inside the router, just as in the RS-422 dynamic example described earlier. Figure 12-7 shows an RS-232 dynamic port.

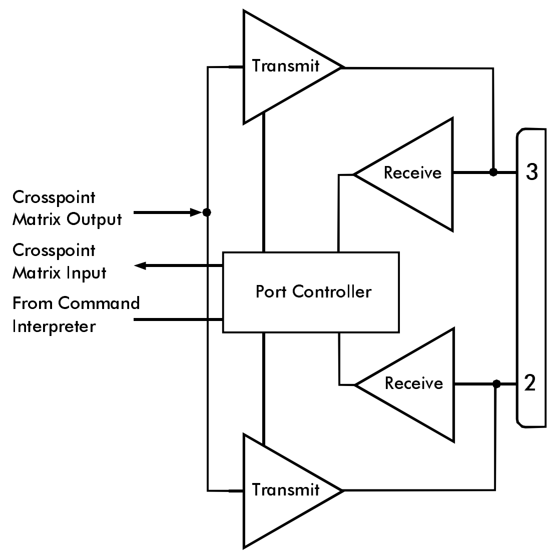


Figure 12-7: An RS-232 Dynamic Port

Wait a minute. Figure 12-4 and figure 12-7 look very similar. It seems as if this type of structure allows a connection between RS-232 and SMPTE 207M signals. This is true provided the RS-232 equipment is capable of "SMART MODEM" operation. In SMART MODEM operation, additional handshaking is not used to initiate a communications session. Fortunately, if the connections on DTE and DCE equipment are made as shown in figure 12-8, most modern RS-232 equipment will function correctly. This is easily verified by observing the fact that computer platforms are routinely used to control VTRs which use SMPTE 207M without any handshaking.

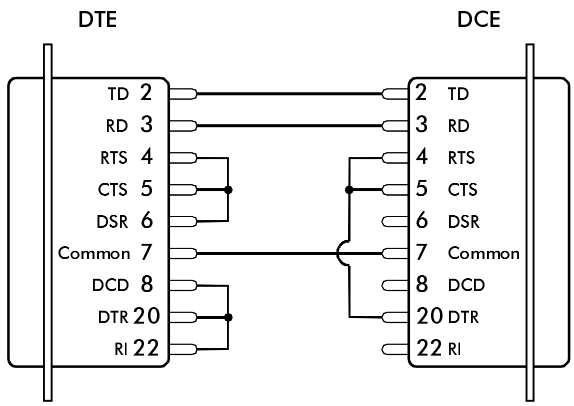


Figure 12-8. An RS-232 Smart Modem Connection

The SMART MODEM connections shown in figure 12-8 are used to deceive the DTE and DCE devices into thinking that the handshake is being executed. In fact, each end is actually using its request line or ready line to generate an acknowledgment which then allows the transmission of data for a DTE device, or the reception of data for a DCE device. The connections shown should be made in the D connector shell at the equipment end of the cable connected to the router. It is important to make the common connection shown in figure 12-8. RS-232 is a single ended signal, it may not work reliably without it.

Figure 12-9 shows the SMPTE 207M, RS-422 to SMART MODEM, RS-232 router connection with conversion.

Full handshake RS-232 may also be routed but at the expense of breakout type Y cables and multiple ports for each machine. These routes will only work between

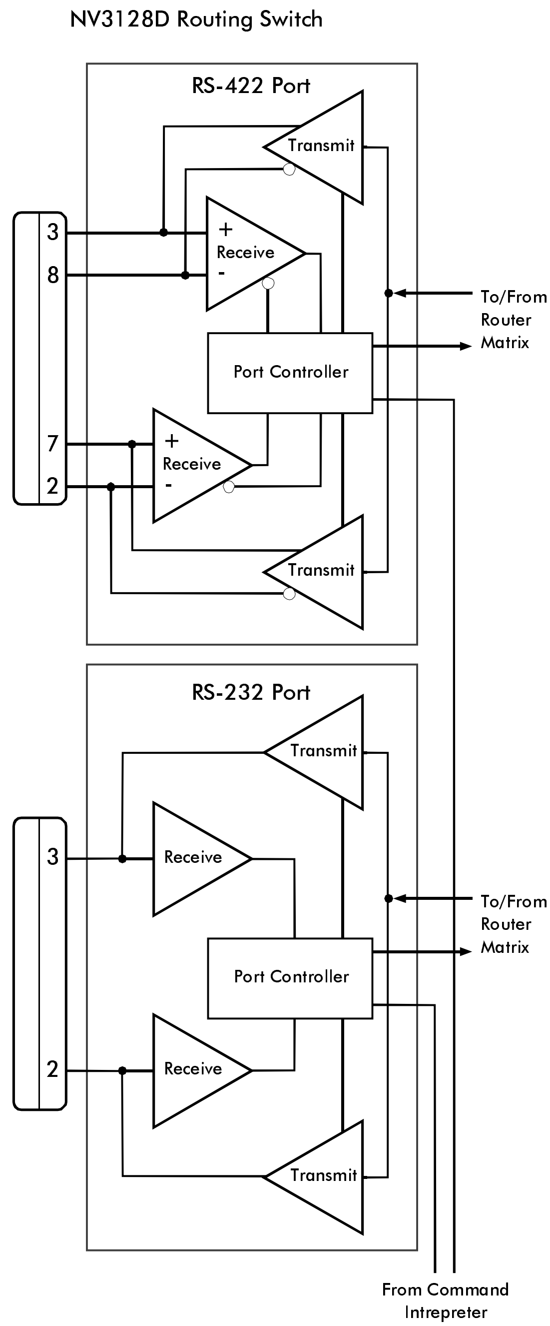


Figure 12-9: Routing and Converting SMPTE 207M with RS-232

compatible pieces of RS-232 equipment. For these applications, the NVUtils software allows linkages to be made between ports so that only one take command is required from the router control system. Chapter 8. provides the details of how virtually any control system can be used to access all the functionality of the NV3128D/ NV3256D router. This includes DTE and DCE configuration, RS-422 Broadcast connections, SMPTE 207M to RS-232 conversion and full handshake RS-232 router linkages. Save power, space, time and anxiety with digital, dynamic port routing for RS-422 and RS-232 control data.

Chapter 13. Designing a Digital Video System

Video Evolution

All currently deployed terrestrial broadcasting systems still use analog transmission techniques, albeit several digital approaches have been proposed, recommended, and otherwise lobbied for. Satellite broadcasters have deployed several all-digital transmission systems designed around the Motion Picture Experts Group (MPEG) recommendations. The television post production community has been using digital techniques for over ten years, and many post production houses are now nearly all-digital for both video and audio. Broadcasting facilities, however, are still largely analog.

In analog production environments, video signals must arrive at switchers, mixers and keying systems with a precise timing relationship. In a digital environment, the equivalent receiving devices are equipped with input buffers that compensate for differences in cable delays. This one feature, makes digital production systems much easier to deploy than the analog counterpart.

Non-Linear editing is another step in the evolution of digital video production. As technology progresses, random access to full quality video will gradually become practical on a larger and larger scale. Note that “gradually” and “full quality” are important parts of this observation. Devices that provide random access to substantial libraries of compressed video material are available today. Devices that provide random access to full bandwidth standard definition video (at a D1 or better quality level) are still limited to relatively short storage times. At the high end of the production triangle, there is currently a demand for better than D1 quality. certain organizations are working with specialized 4:4:4 and 8:4:4 equipment, and some of the highest quality post production is being performed in the 22:11:11 (HDTV) format.

Although, computer technology in general, and disk technology in particular are moving rapidly, video image quality expectations are also moving, at least at the high end.

ATV Systems

Advanced television technology has moved forward steadily ever since the term HDTV was coined in the late seventies. One advanced TV delivery system has been deployed in Japan for many years. The MUSE (Multiple Sub-sampling Encoding) system was in use only a short period of time before aggressive development of compressed digital delivery systems began. In November 1995, ACATS (Advisory Committee on Advanced Television Service) in the US finished 8 years of work and presented a recommendation to the FCC. Initially 23 different systems were submitted for evaluation. After a number of years of analysis and evaluation, four all digital systems remained.

The proponents of those four systems were encouraged to form an alliance (known as the Grand Alliance or GA for short) and produce a final system. This was accomplished, and the final system outperformed all previous submissions in every way. The Grand Alliance system is a digital transmission system using MPEG-2 video compression. It is capable of transmitting one HDTV signal or 4 Standard Definition Television (SDTV) signals in one 6MHz channel.

The GA system is based on a component video signal that accommodates frame rates of 23.976 Hz, 24 Hz, 29.97 Hz, 30 Hz, 59.94 and 60 Hz interlace (I) or progressive scan (P) video. Standardized video inputs can be at 525/ 59.94 (I or P), 525/60 (I or P), 750/59.94(P), 750/60(P), 1125/59.94(I), or 1125/60(I). Additionally, the system recognizes film origination at 24 Hz (and 23.976Hz) or 30 Hz (and 29.94 Hz) and uses only the relevant progressively scanned frames for compression coding (i.e. the third field in 3/2 pull-down is discarded. The 525 line capability of the system supports either 4:3 or 16:9 aspect atio. The 750 and 1125 line capability of the system supports only 16:9 aspect ratio. Receivers will display in their native scanning format, and convert all inputs to match. The exception being that receivers will have to track the .1% offset for signals that were originated at 23.976, 29.97, or 59.94 fps. This allows for a graceful transition from NTSC to a truly 60 Hz system. The following documents are normative references for input signals.

SMPTE 274M (1995), *Standard for television, 1920 x 1080 Scanning and Interface*.

SMPTE S17.392 (1995), *Proposed Standard for television, 1280 x 720 Scanning and Interface*. (Currently being released as SMPTE 296M)

ITU-R BT.601-4 (1994), *Encoding parameters of digital television for studios.* (currently being updated to ITU-R BT.601-5

Other reference documents describing the recommended system in detail are available for downloading from the ATSC (Advanced Television Systems Committee) on the world wide web at http://atsc.org/standard.html.

Table 1. SDTV Video System Relationships

System/ Tape Format	Field Rate	Fields/ Frame	Lines/ Frame	Line Rate	Line Integer H=	Subcarrier	Luminance Sample Rate	Pixels/ Line
NTSC	59.94 Hz	2	525	15734. Hz	$\frac{2.25MHz}{143}$	$\frac{455H}{2}$	Analog	Analog
NTSC D2/D3	59.94 Hz	2	525	15734. Hz	$\frac{2.25MHz}{143}$	$\frac{455H}{2}$	910H=4SC = 14.318-MHz	760 Typ
525/59.94 D1/D5/ Digital	59.94 Hz	2	525	15734. Hz	$\frac{2.25MHz}{143}$	Component	858H = 13.5 MHz	720 Max 704 Min
PAL	50Hz	2	625	15625Hz	$\frac{2.25MHz}{144}$	$\left(\frac{1135H}{4}\right)+25$	Analog	Analog
PAL D2/D3	50Hz	2	625	15625Hz	$\frac{2.25MHz}{144}$	$\left(\frac{1135H}{4}\right)+25$	4SC = 17.734-MHz	940 Typ
PAL D1/D5/ Digital	50Hz	2	625	15625Hz	$\frac{2.25MHz}{144}$	Component	864H = 13.5MHz	720 Max 704 Min

System/ Tape Format	**Active Lines/ Frame**	**Chroma Sample Rate**	**Bits/ Sample**	**Serial Data Rate**	**48 kHz Audio (A) Vertical Relationships**	**Other 48kHz Audio (A) Relationships**
NTSC	483	Analog	Analog	Analog	$\frac{V}{5} = \frac{A}{4004}$	1144H = 375A
NTSC D2/D3	483	Encoded	8	9100H = 143.18-Mb/s	$\frac{V}{5} = \frac{A}{4004}$	1144H = 375A
525/59.94 D1/D5/ Dig	483	429H = 6.75MHz	D1=8 D5=10 Dig =10	17160H = 270Mb/s	$\frac{V}{5} = \frac{A}{4004}$	1144H = 375A
PAL	576	Analog	Analog	Analog	$V = \frac{A}{960}$	1152H = 375A
PAL D2/D3	576	Encoded	8	40SC = 177.34-Mb/s	$V = \frac{A}{960}$	1152H = 375A
PAL D1/D5/ Dig	576	432H = 6.75MHz	D1=8 D5=10 Dig =10	17280H = 270Mb/s	$V = \frac{A}{960}$	1152H = 375A

Table 2. ATV Video System Relationships

System/ Tape Format	Field Rate	Fields/ Frame	Lines/ Frame	Line Rate	Line Integer H=	Subcarrier	Luminance Sample Rate	Pixels/ Line
ACATS HDTV 1080x1920 60Hz	60Hz	2	1125	33750Hz	$\frac{(2.25MHz)(3)}{200}$	Component	2200H = 74.25Mhz	1920
ACATS HDTV 1080x1920 59.94Hz	59.94 Hz	2	1125	33716. Hz	$\frac{(2.25MHz)(15)}{1001}$	Component	2200H = 74.17 MHz	1920
ACATS HDTV 720x1280 60Hz	60Hz	1	750	45000Hz	$\frac{2.25MHz}{50}$	Component	1650H = 74.25MHz	1280
ACATS HDTV 720x1280 59.94Hz	59.94 Hz	1	750	44955 Hz	$\frac{(2.25MHz)(20)}{1001}$	Component	1650H = 74.17 MHz	1280
ITU-R BT.1120 1250/50	50Hz	2	1250	31250Hz	$\frac{225MHz}{72}$	Component	2304H = 72.00MHz	1920
24 Frame Film	24	1 (Repeated)	N/A	N/A	N/A Frame Rate= 2.25MHz/93750	N/A	N/A	N/A

System/ Tape Format	Active Lines/ Frame	Chroma Sample Rate	Bits/ Sample	Serial Data Rate	48 kHz Audio (A) Vertical Relationships	Other 48kHz Audio (A) Relationships
ACATS HDTV 1080x1920 60Hz	1080	1100H = 37.125MHz	10	44000H = 1.485Gb/s	$A = 800\ V$	$45A = 64H$
ACATS HDTV 1080x1920 59.94Hz	1080	1100H = 37.08-MHz	8 or 10	44000H = 1.483-Gb/s	$\frac{V}{5} = \frac{A}{4004}$	
ACATS HDTV 720x1280 60Hz	720	825H = 37.125MHz	8 or 10	33000H = 1.485Gb/s	$A = 800V$	$45A = 48H$
ACATS HDTV 720x1280 60Hz	720	825H = 37.08-MHz	8 or 10	33000H = 1.483-Gb/s	$\frac{V}{5} = \frac{A}{4004}$	
ITU-R BT.1120 1250/50	1152	1152H = 36MHz	8 or 10	230400H = 1.44Gb/s	$V = \frac{A}{960}$	$576H = 375A$
24 Frame Film	N/A	N/A	N/A Up to 14 bits Equiv Dyn Range	N/A	$V = \frac{A}{200}$	

The central notion of ATV is higher video and audio quality delivered to viewer. The price of this quality improvement is the need for higher signal bandwidth. This increase in bandwidth is compensated for within delivery systems by the utilization of the compression scheme described above. In the studio, the need for greater bandwidths is likely to substantially extend the lifetime of linear production techniques, and real time distribution concepts.

Full bandwidth ATV signals consume nearly 1.5 Gigabits/sec of continuous bandwidth. Even with 2 to 1 compression (a logical extension of digital Betacam technology), a studio quality signal would require nearly 700 Mb/s of continuous bandwidth. This kind of bandwidth is technologically feasible today in an updated version of a classical video distribution system. Looking at the computer technology curve, the probability of large volumes of studio quality ATV material being accessible on disk is likely to be several years away.

The audio side of ATV includes real multi-channel surround sound. Even with high quality 525 line delivery, and certainly with 750 or 1125 line delivery, larger and larger screens for the home will become available. As the screen size increases, the value of surround sound to the consumer will become obvious. The upside of multi-channel sound (5.1 channels in the case of the GA system) is that excellent audio imaging is possible over a large listening area, rather than the center "sweet spot" associated with stereo. The movie industry has widely accepted multi-channel sound for theatrical repease. The challenge for television production will be to provide the ability to produce an accurate audio image. Many of the techniques currently used in film audio production to locate the source of a sound relative to the screen, will need to have a cost reduction to fit into the economic model of television production.

Integer Relationships

A very important, and seldom clarified aspect of widely deployed television systems, is that all of the system parameters are integer related. Additionally, all of the world's current and proposed broadcast systems are interrelated by a few key integers. The first color system, NTSC, was bounded by a pre-determined and treaty-bound sound carrier offset frequency. To minimize sound to color interference (and vice-versa), the line frequency was adjusted to be a sub-harmonic of the sound carrier offset. The color subcarrier was set to be an odd harmonic of ½ the line frequency. This arrangement produced a spectral interleaving that reduced the interference between the sound and color signals. The key integer that fell out, was a line frequency equal to 1/286 times the 4.5MHz sound carrier offset from the visual carrier. Similar constraints faced other early television system designers.

When work began in earnest on component digital studio systems, the fact was observed that 2.25MHz (4.5MHz/2) was a line harmonic of nearly all the world's television systems deployed at that time. After much negotiation, agreement was reached on digital component sampling for both 525 and 625 line systems. The resulting document, CCIR-601 (now ITU-R BT.601-4 currently being updated to ITU-R BT.601-5) defined the sampling of component video signals at 13.5MHz for luminance and 6.75MHz for chrominance (both frequencies are multiples of 2.25MHz) for both 525/59.94 and 625/50 systems.

As High Definition TV development progressed, constant pressure has been applied to keep 2.25MHz in mind when defining system parameters. At present, the 50Hz and 60Hz implementations of HDTV keep a low integer relationship with 2.25MHz. The exceptions are the transition systems that allow 750 and 1125 line television to operate at the NTSC field rate of 59.94Hz, thus allowing up-conversion of pre-recorded material and for simulcasting with a single production operation. The latter case may be much better served by a well designed 60 to 59.94Hz down-converter for the NTSC simulcast.

In the last few years, the sampling rate for digital audio associated with video production has been well standardized on 48kHz. To fully understand video system design, the 48kHz number needs to be added into the integer relationship mix. Tables 1 and 2 show how television systems inter-relate, and how they relate to the 48kHz audio sampling rate. This information is useful for both systems designers, and equipment designers, as it shows how systems can be locked together in an optimal fashion.

Current Formats

In both the analog and digital realms, there are two basic signal formats, component (Y',P_R',P_B', Y',C_R',C_B' & $R'G'B'$) and composite (NTSC, PAL, PAL-M,PAL-N & SECAM). Analog composite signals have an advantage over component when cable requirements are considered, as component requires 3 cables for each video source (4, in the case of RGB & Sync). However, composite signals suffer from quality loss due to coding system bandwidths, and encoding and decoding artifacts. In the digital domain, both formats can be transmitted serially via a single cable, thus allowing component systems to be constructed that take advantage of better signal quality, while benefiting from the interconnection methods used in a composite analog system.

There is essentially one digital interconnection method currently utilized for standard definition television, which is described in the SMPTE 259M - 1993 standard. This allows for the serial transmission of either 8 or 10 bit 4:2:2 coded Digital Component Video or 10 bit Digital Composite Video. Component serial data is transmitted at up to 270Mb/sec and composite at 143Mb/sec (NTSC) or 177Mb/sec (PAL).

'D' numbers signify tape format, although they are often (and incorrectly) used to describe serial format (D1 = serial component, D2 = serial composite). There are several 'D' VCR formats; D1,D2,D3 & D5, as well as Digital Betacam, DVC-Pro and Sony SX. The D2 and D3 machines are composite formats and the others component.

Table 13-3: Current Digital VCR formats

Format	**Tape Width**	**Signal Format**	**Sample Rate**	**# of Bits per Sample**
D1	19mm	Component 4:2:2	13.5MHz Y' 6.75MHz C'_R,C'_B	8
D2	19mm	Composite (PAL or NTSC)	4 x Subcarrier Frequency	8
D3	‰	Composite (PAL or NTSC)	4 x Subcarrier Frequency	8
D5	‰	Component 4:2:2 or Composite (PAL or NTSC)	13.5MHz Y' 6.75MHz C'_R,C'_B	10
DCT	19mm	Component 4:2:2 Compressed	13.5MHz Y' 6.75MHz C'_R,C'_B	8
Digital Betacam	‰	Component 4:2:2 Compressed	13.5MHz Y' 6.75MHz C'_R,C'_B	10
DVC-Pro	…	(preliminary) Component 4:1:1 Compressed	13.5MHz Y' 3.375MHz C'_R,C'_B	8
Sony SX	‰	(preliminary) Component 4:2:2 Compressed	13.5MHz Y' 6.75MHz C'_R,C'_B	8

All of these DVCR's have a standard Serial Digital Interface (SDI) based on SMPTE 259M. You cannot of course, connect a digital composite machine to a component machine without the use of a Rate Converter (sample rate converter with decoder / encoder). But the SDI connection is designed to work regardless of digital tape format, provided that the line/field rates are compatible. Some older digital component equipment had parallel interfaces and did not incorporate an SDI I/O, however, inexpensive serializers / deserializers are readily available that provide interconnection to these devices.

As mentioned previously, CCIR 601 (now ITU-R BT.601-4) is the sampling standard for digital component video that was set by a committee made up of members from SMPTE and the EBU. It specifies a common sample rate for 525 and 625 line systems of 13.5MHz for Luma and 6.75MHz for each color component. CCIR 601 is <u>not</u> an interconnect standard.

Video Standards to be aware of

Regardless of signal type (analog or digital), all current world standards for SDTV are based on 525 line / 59.94 fields or 625 line / 50 fields. Therefore, component signals (being free from encoding) originate in either of these formats.

A variety of encoded standards exist that are derived from three basic coding methods; NTSC (National Television Systems Committee) a 525/59.94 system with a 3.58MHz subcarrier, PAL (Phase Alternating Line) and SECAM (Sequential Couleur avec Memoire), both 625/50 systems with PAL having a 4.43Mhz subcarrier. SECAM has a more complicated modulated subcarrier based on two frequencies, 4.44MHz D_B and 4.25MHz D_R.

Subsets of these standards include PAL - N, PAL - M, SECAM - H and SECAM - V. It should be noted that digital composite products are only available in the NTSC or PAL format.

Signal Distribution and Interconnection

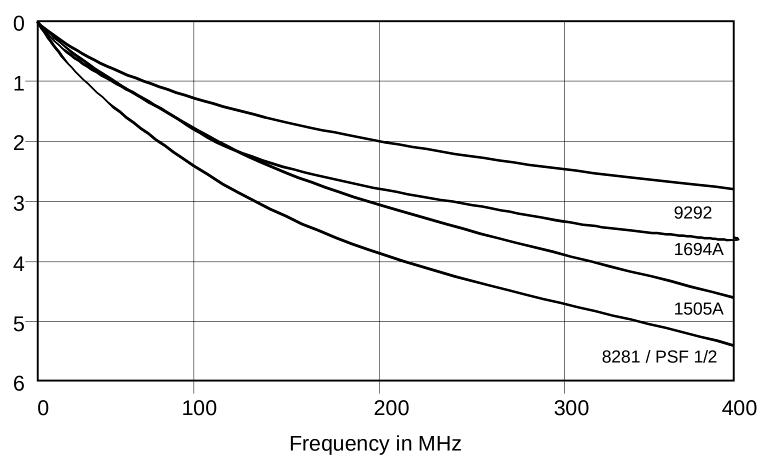
When wiring a facility for distribution of digital video signals, there is no significant change in a schematic sense, from the distribution of analog signals. Of course, digital video has a much higher frequency range (270Mbits/sec vs. 4.2MHz for NTSC and 5.5MHz for PAL) so cable quality and accurate termination is of paramount importance.

Signal level accuracy is less critical, but high frequency digital signals cannot be satisfactorily transported over long distances without intermediate equalization and re-clocking.

When considering adding digital video distribution within a current analog system, attention must be given to the quality of existing cable and more importantly, the type of BNC connectors used. For many years, the BNC connectors and patch panels supplied for use with 75 ohm video cable have had a 50 ohm impedance. In the analog realm this has little or no effect, as signal wavelength is much longer than the connector length. At digital video frequencies, this impedance mismatch can induce transmission line reflection that results in signal jitter. This unwanted effect can cause unstable reception at the destination and make reliable operation difficult. Tracking down the intermittent loss of signal in systems using existing cables, connectors and patch bays can be very difficult, if not impossible. Therefore, paying extra attention to the connectors and patching systems used during installation, can save a lot of time and effort de-bugging a system later.

Serial data can be transported via most existing 75 ohm coaxial cable, however attenuation losses can be much greater in cable designed for analog video. Belden 8281, commonly used for many years, has a 5.5dB per 100ft loss at 400MHz, whereas cables designed for digital signals exert losses of almost half that (see figure 13-1). Please note that the maximum frequency of an NRZI coded 270Mbit signal is 135MHz, however, its third harmonic is 405MHz, which is important for accurate data recovery.

Utilizing high quality digital video cable, lengths for 270Mbits serial data should still be kept within 300 meters, without intermediate EQ and reclocking. You will read some equipment specs that state a maximum cable length of 300 meters. This is based on a theoretical ideal that is very difficult to achieve without perfect impedance matching, and excellent PLL design in the receiver.



Synchronization

Engineers experienced at designing analog video systems are often painfully aware of the time and effort required to ensure that video signals are correctly timed. For analog switchers (mixers) and keying systems, it is of critical importance. However, digital equipment incorporates reclocking and retiming of input data, so cable propagation delay is much less significant.

Of course every system component requires a common reference signal to ensure vertical alignment, and frame synchronization of non-synchronous inputs is necessary.

Some sync generators now provide a digital reference signal as well as the analog 'Black & Burst' reference. For component video, the burst reference is not normally used, but black & burst is still the most widely employed reference signal. When considering reference signals, please note that all currently available product can accept analog reference. Digital reference can be provided, if necessary, by a correctly locked digital bar generator.

Therefore system designs will continue to use analog reference for some time to come, naturally allowing the re-deployment of analog DAs for sync distribution.

All digital receivers require a finite time to provide accurate buffering and reclocking. This time varies from a few clock cycles, to a number of video lines or a complete frame (dependent on received signal timing and product function). These delays will normally have no effect on the final video output, but impact the audio delay requirement to ensure that 'Lip Sync' is maintained. Please refer to chapter *5* for information on digital audio delays.

Routing Digital Video

Analog video routers, no matter how well designed, tend to suffer from levels of induced noise and crosstalk. The digital equivalent does not suffer from either of these problems. However, good input phase locked loop design is essential to provide accurate reclocking and jitter reduction. A digital router with the appropriate reclocking circuits also allows the maximum cable lengths to be used on both its inputs and outputs.

Some SDI routers employ wideband analog crosspoints, please be aware that this type of router may not perfectly preserve the ancilliary data and original signal timing. This type of switch may handle the video component of the SDI signal acceptably, but embedded audio data will be subject to potentially large transition errors. Therefore, switch architecture should be investigated when the utilization of embedded audio is considered.

Chapter 14 . Video Format Conversion

Composite to Component

Most new digital installations utilize the digital component format, therefore it is necessary to decode any existing analog or digital composite sources. The source signal bandwidth (i.e. VTR format) determines the quality and cost of the decoding system required to retain the best signal performance.

Decoding NTSC signals is a little easier than PAL, but recovering artifact free component signals at maximum bandwidth is somewhat of a 'black art'. Simple 'notch' decoders will provide limited luminance bandwidth. The filtering process will remove luma frequencies that occur in the chroma subcarrier region and above (see figures 14-1a and 14-1b.).

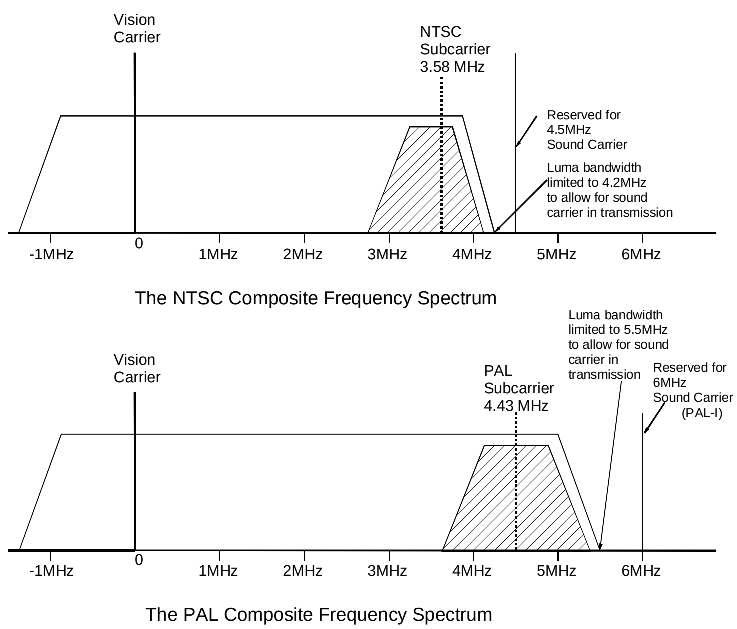


Figure 14-1a: Composite PAL and NTSC frequency spectrums

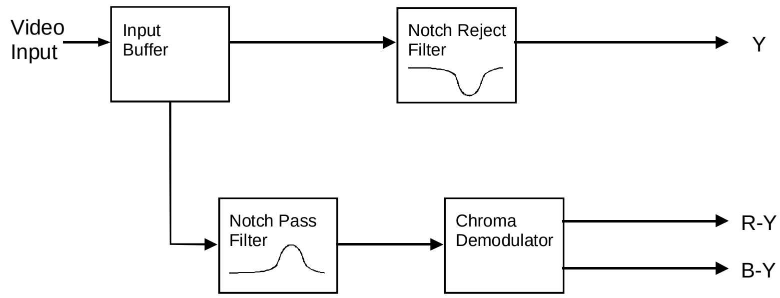


Figure 14-1b: Simple video decoder using notch filtering

Comb filter techniques can be applied to analog decoders with reasonable success. The process of comb filtering is simple in concept; The incoming signal follows two paths, one having a notch filter to extract chroma signals, the other a line delay (2 lines of delay are required for PAL). The chroma signal is inverted and added to the delayed path which cancels the chroma content in the luma channel (see figure 14-2.). If the chroma content changes significantly from line to line, this process can cause significant errors in the final signal.

Adaptive comb filter decoders have been designed to detect significant vertical transitions in chroma content and switch to a notch filter until the transition has passed. This technique overcomes chroma errors at transitions and provides a good quality output, nominally at the full recoverable bandwidth (see figure 14-3.). Design response of the tratransition detector is critical in providing the best compromise between filtering methods. Diagonal transitions will cause unwanted chroma artifacts, particularly in PAL, where the error is amplified by the extra line of delay.

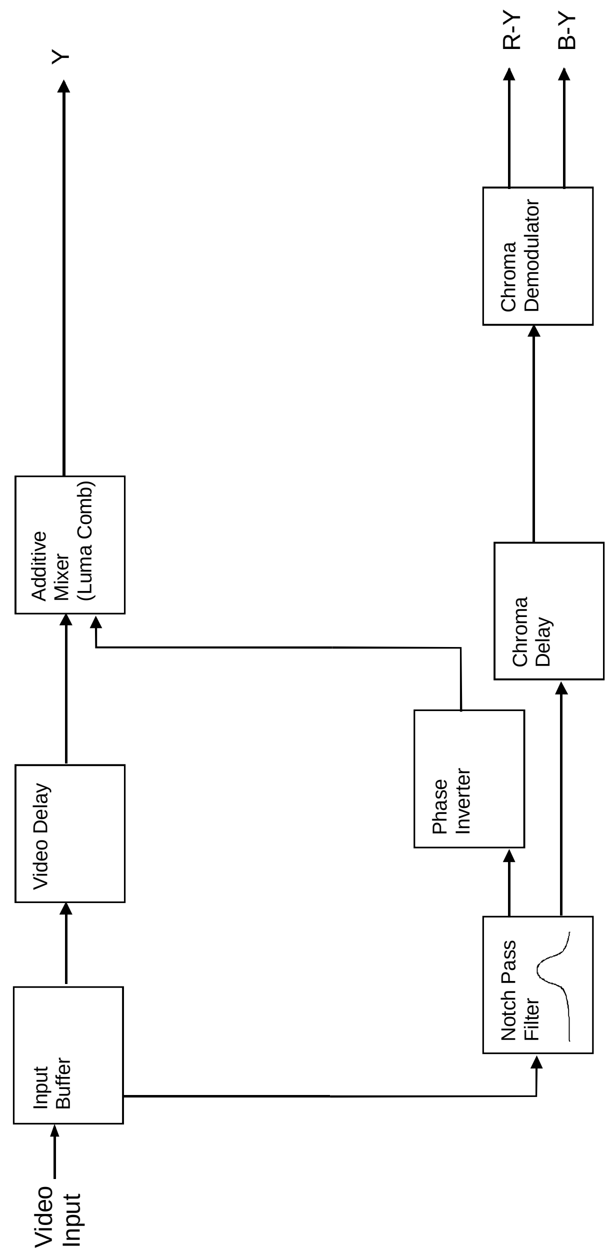


Figure 14-2: Line Based, Comb Filter Decoder
(Simplified Block Diagram)

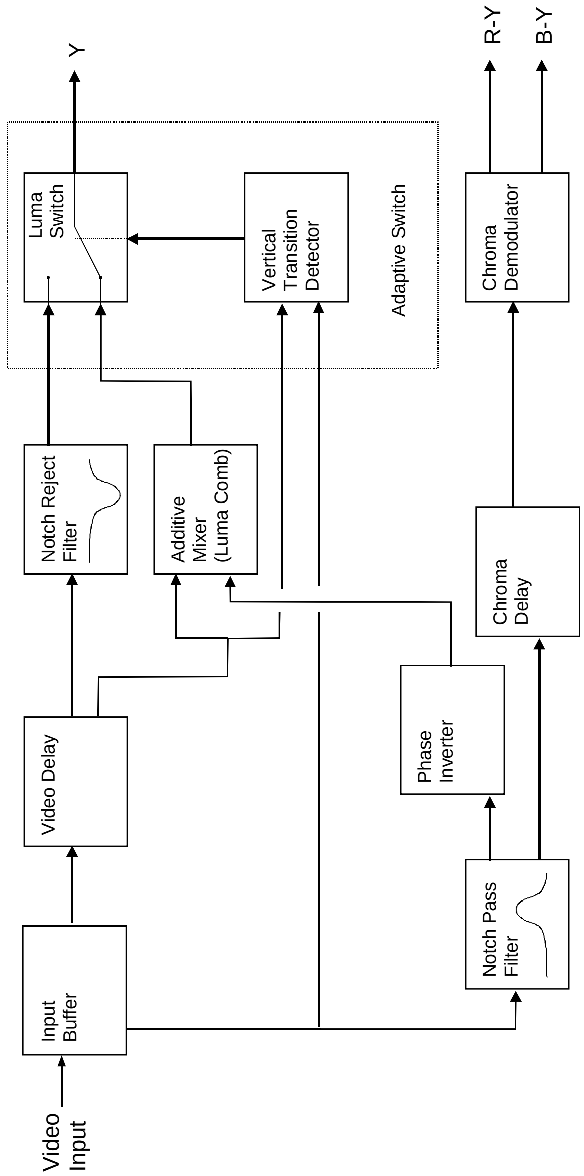


Figure 14-3: Line Based Adaptive Comb Filter Decoder
(Simplified Block Diagram)

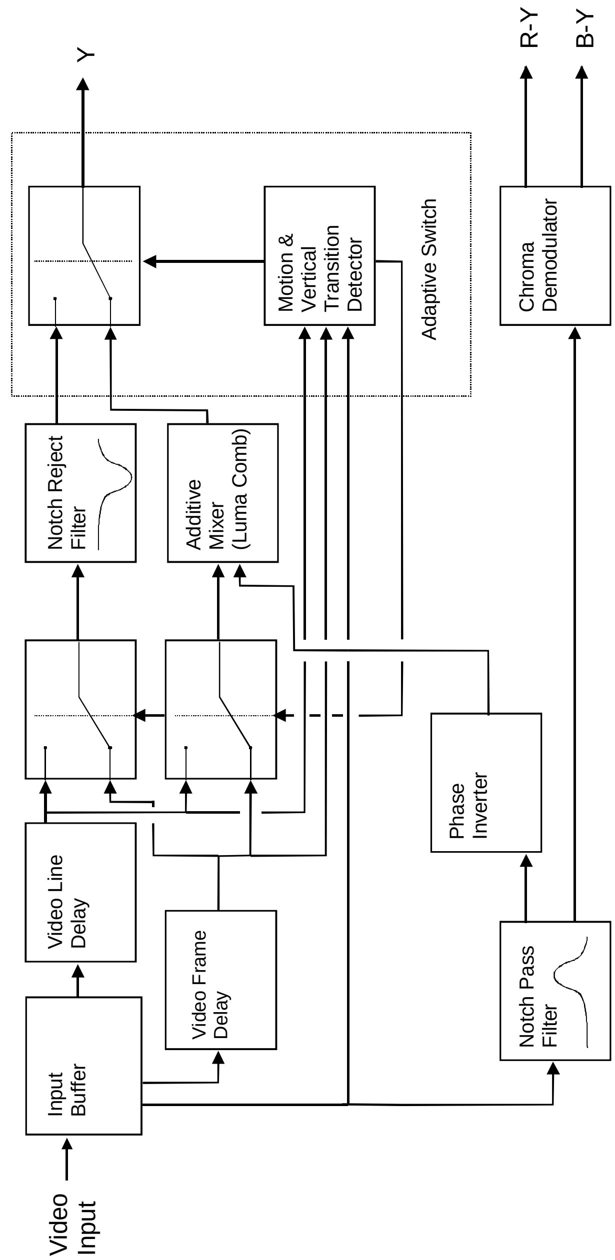


Figure 14-4: Frame Based Adaptive Comb Filter Decoder
(Highly Simplified Block Diagram)

By digitizing the signal, it is possible to add spatial decoding to the adaptive comb technique. In a 'frame based' adaptive comb decoder, chroma information between fields or frames can be compared, allowing the elimination of decoding error at diagonal transitions of still images. However, the design must now include a motion detector to determine motion elements so that the decoder can 'adapt' its filters to use line combing or notch filtering where motion edges are detected (see figure 14-4.). This process avoids creating chroma errors induced by picture movement in a frame based comb. A further problem occurs when a scene change occurs, the frame based comb must delay chroma by 1 field or more. Therefore, when a scene change occurs, chroma from the last field will be present in the first field of the new scene.

All digital decoding (digital composite to component), presents yet another level of complexity due to the different sample rates employed. Therefore it is also necessary to provide sample rate conversion from 4 times subcarrier to 13.5MHz.

From the above you can see that decoding video can be a very complex task and a 'perfect' decoding system is almost impossible to achieve. However, many good, practical decoders are available to suit the various applications found in day to day operations. Please refer to table 14-1. Which identifies decoder limitations for the type of input source.

Table 14-1: Decoder limitations

Decoder Type	Decoded Luma Bandwidth	Limitations	Application
Notch	approx. 3MHz NTSC approx. 4MHz PAL	limited Bandwidth NTSC Cross-color artifacts visible in Luma	Good for VHS , Umatic & other 'color under' source material
Simple Comb	Available recorded or transmitted bandwidth	Vertical shift in chroma. Gross Chroma errors at Vertical transitions	Will process full bandwidth input. Not suitable for Graphics or material with high saturation levels
Adaptive Line Comb (2D)	Available recorded or transmitted bandwidth	Diagonal Chroma aberrations likely. Cross-color artifact at vertical transitions	Very good for most applications. Highly saturated graphics may be unacceptable at vertical edges.
Adaptive Frame Comb (3D)	Available recorded or transmitted bandwidth	Motion artifacts likely. Chroma temporal artifacts likely at scene changes	Very good for graphics and highly saturated video. Chroma artifacts may be objectionable at scene changes.

Analog to Digital Conversion

As mentioned previously, most distributed analog signals are composite, and the digital format becoming the norm is component. Therefore the process of analog to digital conversion requires three steps.

1. a) Decode composite to component or

   b) Convert from Analog To Digital.

2. a) Convert analog to digital or

   b) Decode Composite to Component.

3. Serialize the parallel data that results from step 2.

High end digital decoders are available that incorporate all three processes, but A to D converters for serial digital component usually have analog component inputs that require separate decoders. If the input is composite, the output quality is typically limited by the type of decoder employed (see Table.1). If the input is component, the conversion quality is only limited by the anti-aliasing filters and the accuracy of the internal A to D device chosen.

For composite digital formats, the A to D process, samples at 4 times the subcarrier frequency of the signal standard. Provided that the filter circuits are well designed, the digital signal should be an accurate representation of the analog original. As with all digital signals, composite digital has the advantage of multiple recording passes without degradation, as well as auto EQ and input retiming.

Component A to D converters are available with either 8 bit or 10 bit quantizing. For most broadcasting applications, 8 bit signals are adequate However, there is a significant difference in signal to noise ratio (described later in this chapter). For digital production and post production, 10 bit quantizing offers the advantage of a much deeper noise floor and a significant reduction of visible artifacts caused by the rounding errors generated during multiplication (mixing).

If an 8 bit device receives a 10 bit signal it will simply ignore the bottom two bits. 8 bit and 10 bit signals are completely compatible, therefore the choice of A to D products becomes a normal cost vs. performance exercise.

Encoding

The task of creating a composite signal from a component original is far less complex than the reverse operation. The main consideration is in the filter technique chosen for chroma bandwidth reduction (necessary to ensure that chroma fits within the designated spectrum for each coding system). A simple notch (bandpass) filter can be used for this application. However, a digital chroma comb can be more effective in reducing the cross-color artifacts, that result as a function of luma/chroma frequency cross-talk within the subcarrier spectrum. High quality analog and digital encoders are available from many vendors at reasonable cost.

4:4:4, 4:2:2 etc.

ITU-R BT.601-5 is the current revision of the CCIR 601 document that specifies the sample rate and structure of digital component signals. This specification is based on a universal sample rate of 13.5MHz for SDTV, regardless of line or frame rate.

Readers are sometimes confused by the 4:4:4,4:2:2, 4:1:1 etc. classifications for digital signals, so here's a straight forward explanation (hopefully!).

13.5MHz was chosen as the sampling frequency for the highest bandwidth signal of the analog components, this is of course the luminance signal. As the bandwidths required for color difference signals are typically lower, using a lower sample rate conserves valuable data space. Therefore, the 601 recommendation was to sample the color difference signals at a rate that was an integer division of 13.5MHz.

Initial evaluation of the sample rate required to digitize video signals, determined that a good choice was 4 times the subcarrier frequency, naturally this rate is dependent on coding system. When CCIR debated sample rates for component, the reference to 4 x *fs* was retained, and even though the final choice was 13.5MHz, it is still referred to as 4. Thus, 1 = 3.375MHz, 2 = 6.75MHz and 3 = 10.125MHz.

Therefore a 4:2:2 system describes a component system having a mixed sample rate of 13.5MHz Y¢, 6.75MHz P¢$_{R}$ and 6.75MHz P¢$_{B}$. From this it can easily be seen that the sampling frequency ratio states the effective chroma channel bandwidths.

An R'G'B' signal, having the full luminance bandwidth in all three channels, requires a 4:4:4 sample rate ratio. Whereas signals destined for transmission via composite systems have more than adequate chroma bandwidth at 4:1:1.

For post production applications, 4:2:2 allows for a good compromise in chroma bandwidth. Thus, tasks such as chroma keying and graphics generation can be carried out more effectively.

The number of bits per sample has no reference to sampling rate ratio. An 8 bit system can resolve 256 levels for each sample and a 10 bit system 1024 (for either luma or chroma). The real benefits of a 10 bit system are that the signal to noise ratio is 12dB better than the 8 bit version (70dB vs. 58dB) and when multiplying data, the resultant number has rounding errors of 4 times less significance. Therefore, mixing and layering with soft key edges is much cleaner, and the 'step' pattern obvious in low level 8 bit mixes is no longer visible.

Chapter 15. Bringing it all Together

What is that SNAP, CRACKLE and POP anyway? I thought we finished the Rice Krispies® cereal sound bite.

Unfortunately, you were right. The sound bite was done days ago. Equally distressing is the knowledge that the problem needs to be fixed because Kellogg's® cereal sound bites does not a business make.

Clean, transparent, direct digital transfers between two pieces of equipment can only occur when they are operating at exactly the same digital audio sample rate. Audible glitches generated during this process are invariably the symptom of poor synchronization. Even though two pieces of equipment provide 48 kHz sample rates, these frequencies are not identical if they are not locked to a common clock. Digital audio samples which are slipped or dropped and the bit errors that occur without synchronization cause the audible pops and clicks.

Video equipment with a common format is easily synchronized by house video sync. Either NTSC or PAL audio capable equipment generates a 48 kHz sample rate when locked to this reference. Digital audio equipment is typically synchronized with AES3 or SDIF-2 word clock signals, not video. But video equipment doesn't lock to audio reference signals. A digital audio reference providing AES3 and SDIF-2 references locked to video eliminates this dilemma.

Multi-format Video Production poses a different problem. Even though PAL and NTSC equipment is synchronized to house references for their respective formats, they are not frequency locked to each other. Different video formats must be synchronized to a common frequency to insure identical sample rates; typically 48 kHz for audio used with video. Figure 15-1 shows that any number of formats and sample rates may be synchronized to a common clock.

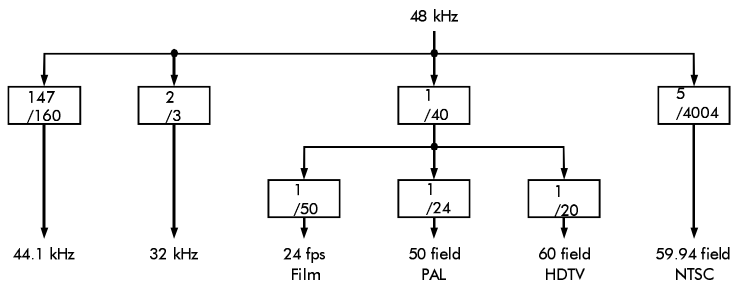


Figure 15-1: Multi-format synchronization relationships

The relationships shown in figure 15-1, indicate that it should be possible to synchronize NTSC video, PAL video, AES3 and SDIF-2 word clock to a common time base, guaranteeing frequency lock. NTSC at 59.94, PAL at 50, HDTV at 60, Film at 24 and AES3 and SDIF-2 at 48 kHz, 44.1 kHz and 44.056 kHz can all be generated simultaneously, synchronizing virtually every piece of equipment in any facility.

New and existing two format facilities are easily synchronized starting at the top of the video timing chain. Figure 15-2 shows this application. Existing NTSC and PAL sync generators are locked to compatible outputs of the NV5500. Clean, direct digital audio transfers between PAL and NTSC video equipment are now possible because 48 kHz is identical for all referenced equipment. All downstream video timing is unaffected. Audio reference outputs synchronize A/D converters and other audio equipment to the 48 kHz sample rate. The result is truly powerful.

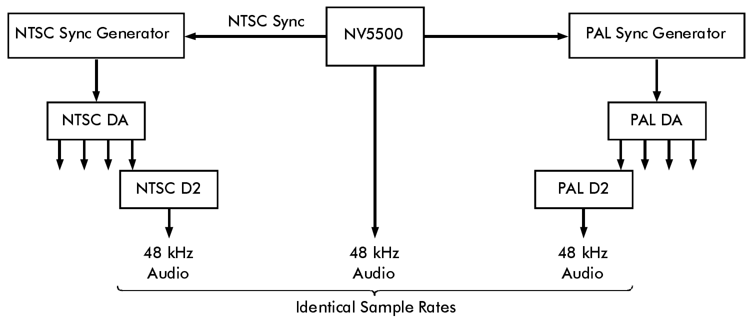


Figure 15-2: NV5500 master timing for plant synchronization

Slave Plant Synchronization is also possible by locking either NTSC to PAL to the opposite format. This application arises as single format video facilities add additional formats. Figure 15-3 shows one implementation of this topology. The new video format is frequency locked to the existing one and all the digital audio reference signals are available. Because NTSC and PAL do not have a meaningful phase relationship, the NV5500 family does not include video genlock capability. Most new video equipment has genlock built in, so downstream network timing is simplified.

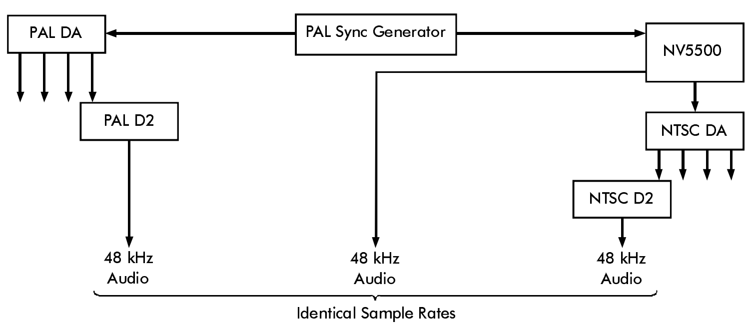


Figure 15-3: NV5500 slave timing for plant synchronization

PAL B,G & H	Subcarrier = 4.43361875MHz +/- 5Hz Stability = +/- 1.13 ppm
PAL I	Subcarrier = 4.43361875MHz +/- 1Hz Stability = +/- 0.225 ppm
NTSC - M	Subcarrier = 3.579545MHz +/- 10HZ
AES11 Grade 1	Stability = +/- 1 ppm
AES11 Grade 2	Stability = +/- 10 ppm

Table 15-1: Comparison of accuracy requirements for various standards

When slave locking video formats, frequency accuracy should be considered. Table 15-1 shows the accuracy requirements for AES3 references and a number of composite video signals. The table shows that an NTSC signal locked to PAL is sufficiently accurate, a PAL signal locked to NTSC, however, may not be.

For Single Video Format Facilities, a video locked, digital audio reference generator provides cost effective synchronization. Digital audio references are used for plant wide or local island applications. The NV1080 family provides video referenced AES3 and SDIF-2 digital audio synchronization signals. Figure 15-4 shows a sync network topology composed of one NV1080 and any number of NV1022 Distribution amplifiers. This type of topology is discussed in detail in Chapter 3.

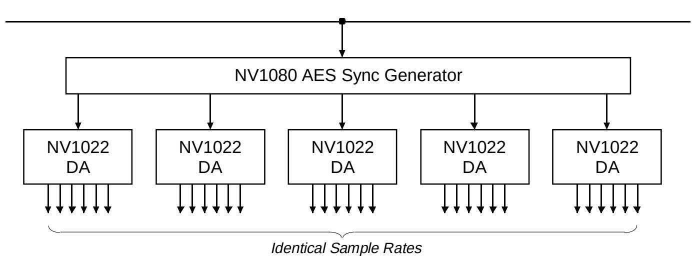


Figure 15-4: NV1080 Digital Audio Sync Network

For Local Island Applications such as a small audio production suite, an NV1082 is used as shown in figure 15-5. It provides timing for a modest number of A/D converters, a production mixer, an RDAT and a DVTR. SDIF-2 signals are converted to the AES3 format with the NV1071.

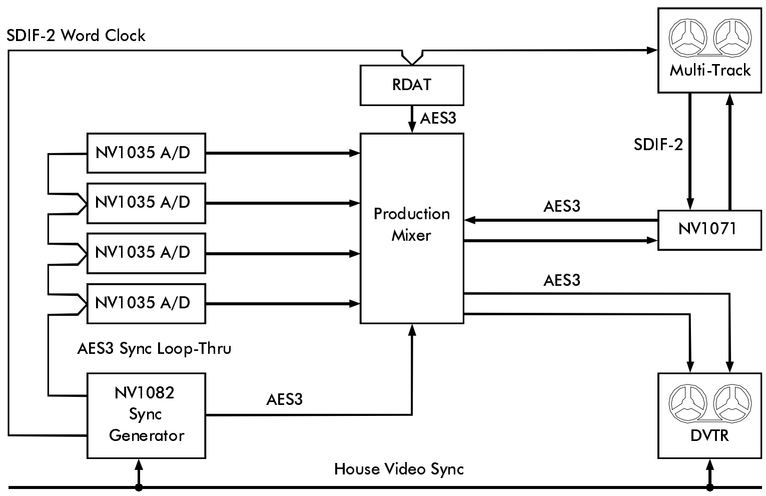


Figure 15-5: The NV1082 and Local Island Synchronization

An AES3 reference signal is not restricted to be digital quiet, tone, or anything else. Quiet is useful as a router input. Tone is helpful for plant continuity checks and equipment alignment. The NV5000 or NV1080 families provide easy AES3 reference configuration. You select digital quiet or tone, at 1 kHz or 500 Hz, with a peak signal level of FSD or 20 dB below FSD.

No synchronization technique prevents glitches if asynchronous digital audio equipment is inserted in the signal path. For example, asynchronous AES3 routers generate glitches when a switch is made. Once equipment re-locks, the glitches are gone. Synchronous AES3 routing completely eliminates these artifacts, provided that all interconnected equipment is synchronized. This is discussed in more detail in Chapter 6 and Chapter 9.

Plant synchronization is inexpensive, easily implemented and extremely powerful as demonstrated in all of the applications discussed.

I wonder if Cap'n Crunch® needs a sound bite? Now where did I put that phone number......

Further Reading

Introduction to Digital Audio. Focal Press
Introduction to Digital Video. Focal Press
Audio and Video Compression. Focal Press
The Art of Digital Video. Focal Press
The Art of Digital Audio. Focal Press
1995 Product Catalog. NVISION
The Video Engineers Guide to Digital Audio. NVISION