HANDBOOK
SNELL & WILCOX
SERIES

Your essential guide to digital

by John Watkinson

Your essential guide to digital

A Snell & Wilcox Guide by John Watkinson

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Other publications written by John Watkinson in the Snell and Wilcox Handbook series include: “The Engineer’s Guide to Standards Conversion”, “The Engineer’s Guide to Decoding and Encoding” and “The Engineer’s Guide to Motion Compensation.”

SNELL & WILCOX

Engineering with Vision

Contents

1. Why should I read this?

The world of television production and program delivery is going digital. There are good reasons for that which we will take a look at here. No-one goes digital for its own sake. Sure, it's glamourous, but there had better be more reasons than that; like economic reasons for instance. Quite simply, a carefully planned digital installation costs less to run today than a traditional analog plant doing the same job. It can run with less down time and needs less scheduled and unscheduled maintenance. The experimental phase is over and now that digital technology has matured to deliver reliable, cost effective hardware, it's analog that suddenly looks uneconomic.

It's not just the hardware which is changing, but the whole nature of television is under scrutiny. The nature of television changed when the home VCR was developed. Viewers could time-shift broadcasts, or purchase or rent cassettes of their choice - an early form of Video-on-Demand. Television changed again when competition from cable and satellite came along. Now it is set for a period of further change. The aspect ratio will change from 4:3 to 16:9. Higher quality pictures will reach the home via advanced signal standards. Economics says that advanced television will have to use compression which also means it will be digital. Mass storage technology from computers and video compression technology combine to allow true Video-on-Demand where the viewers choice is communicated to the server via a back-link. That back-link also allows interactive video and home shopping. Some personal computers can already receive television pictures and play compressed digital video disks. Video games are increasing in power so quickly that their manufacturers will soon be able to move into general computation and video processing. The boundaries of computing and video continue to overlap.

So unless you want to start a sheep farm, it's not a question of whether or not to go digital, more a matter of when and to what extent. If you are in the lucky position of starting afresh, going fully digital is the only way. However, most of us aren't in that category. We have heavy investments in equipment and staff who are familiar with it. Rush in and spend big bucks on a hastily conceived total digital refit and you could get your ass burned, especially if your staff haven't been trained to handle the new technology. A more careful approach says that you don't have to throw out all the existing gear. Replace a few key items and get most of the advantages of digital while getting good service out of your existing investment. In this guide we offer lots of tips on how to do just that. Like any new technology, digital has strengths and weaknesses and you have to understand the basics in order to create reliable systems. Digital does the same job as analog, but in quite a different way and with more operational freedom. Treat digital equipment like analog and you'll get nothing but trouble and miss out on the freedom. Treat it right and you'll reap the benefits. One of the jobs of this guide is to explain what right means.

Digital is complex and the mathematicians can have a great time spouting formulas and buzzwords which make a lot of sense to rocket scientists. This guide is not for them. It's for normal people who want to make decisions without getting buried in details. So there's no math here, and buzzwords are only put in so we can explain what they mean rather than to show off what a bunch of smart-asses we are. We've collected all the buzzwords together in a glossary at the end.

2. What is digital?

2.1 Digital is just another way of representing an existing television waveform. Fig.2.1a) shows that a traditional analog video system breaks time up into

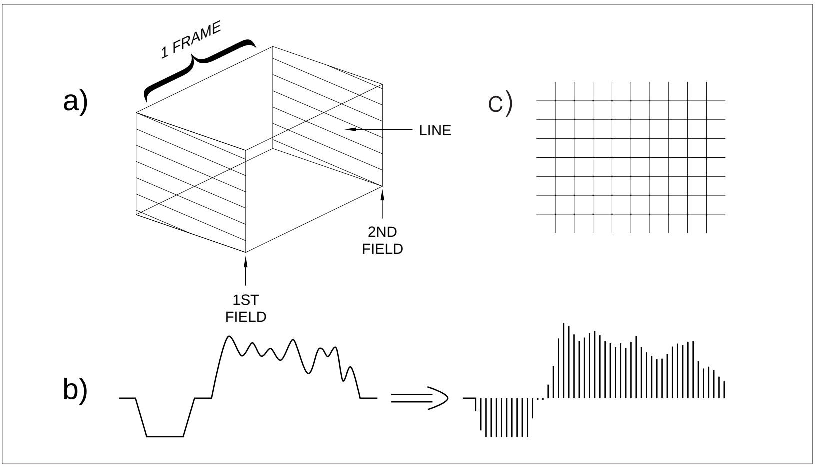
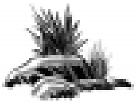


Fig 2.1a In analog television, time is sampled into fields, and each field is sampled vertically to make lines. If each line is also sampled, as in b), an array of pixels will be formed. Such an array can be made from a conventional analog television signal by sampling it a line at a time as shown in c).

fields and frames, and then breaks up the fields into lines. These are both sampling processes: representing something continuous by periodic discrete measurements. In digital we simply extend the sampling process to a third dimension so that the video lines are broken up into three dimensional point samples which are called pixels or pels. Both terms are what you get when you say "picture cells" in a hurry. Fig.2.1b) shows how a television line is broken up into pixels. A typical 525/60 frame contains around a third of a million pixels. In computer graphics the pixel spacing is often the same horizontally as it is vertically, giving the so called "square pixel". In broadcast video systems pixels are not quite square for reasons which will become clearer later in this section.

Once the frame is divided into pixels, the variable value of each pixel is then converted to a number. Fig.2.1c) shows one line of analog video being converted



to digital. This is the equivalent of drawing it on squared paper. The horizontal axis represents the number of the pixel across the screen which is simply an incremental count. The vertical axis represents the voltage of the video waveform by specifying the number of the square it occupies in any one pixel.

The shape of the waveform can be sent elsewhere by describing which squares the waveform went through. As a result the video waveform is represented by a stream of whole numbers, or to put it another way, a data stream.

2.2 Any waveform can be digitized in this way. With NTSC signals, there is only one waveform per channel. When NTSC is digitized, the entire waveform, chroma included, is just expressed numerically. Digital NTSC still has the same characteristics as analog NTSC, including the four field sequence and 0.1% frame rate offset because it is one and the same thing.

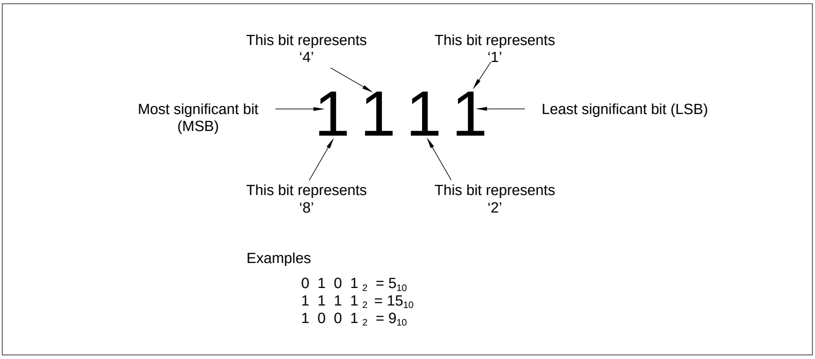


Fig 2.2 A binary conversion table allows any ordinary decimal number to be stored or transmitted by any system which can exist in two states.

In the case of component analog video there will be three simultaneous waveforms per channel. We need three convertors to produce three data streams in order to represent GBR or colour difference components.

By converting video (and audio) signals into data we have the freedom to use storage, processing and transmission techniques which have been developed for computers.

2.3 Digital transmission has the advantage over analog in that it can reject noise and jitter. If we convert the whole numbers representing the video waveform voltage into binary, using the conversion table shown in Fig.2.2, our resultant binary digits, or bits, have only two states, 1 and 0. These two states can then be represented by any electrical, magnetic, optical or mechanical system which can exist in two states. Fig.2.3 shows binary represented electrically by two different voltages, magnetically by the reversal

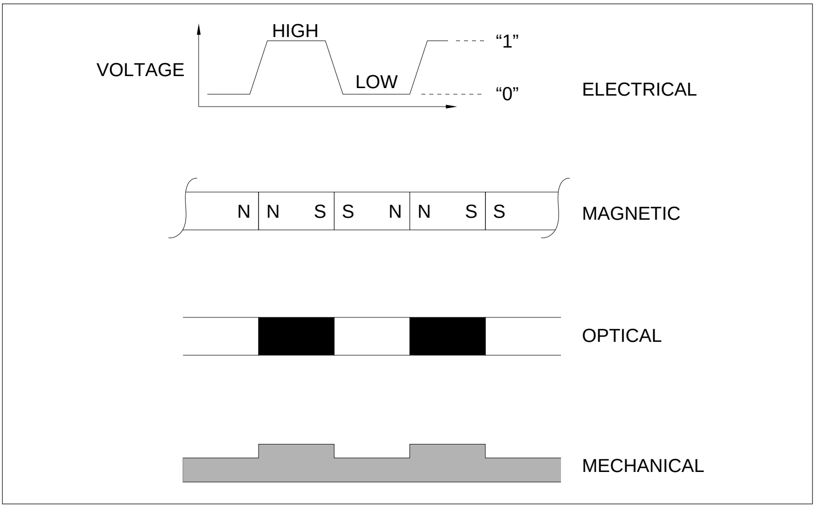


Fig 2.3 Several ways of handling binary data are shown here. In electronic circuits, two different voltages are needed, commonly achieved with a simple on/off switch. In magnetic recording, two flux directions are used. Optical recording may use alternating opaque or dark areas and mechanical storage may use raised or depressed areas on a surface.

of the direction of magnetisation, optically by alternate opaque and transparent areas of a recording, and mechanically by the presence of pits in the surface of a laser disk.

With only two states, more noise can be rejected than in any other system. Fig.2.4a) shows how using the example of an electrical interface. Although the signal transmitted is a clean, two-level waveform, by the time it reaches the receiver it will have picked up noise and jitter. The receiver compares the voltage of the signal with a reference which is mid-way between the transmitted levels in a process called slicing. Any voltage above the slicing

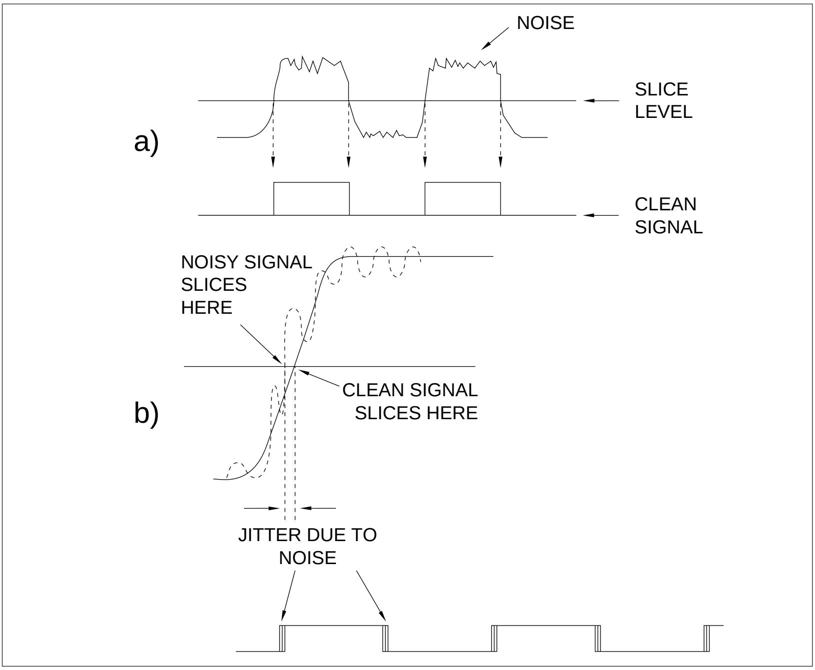


Fig 2.4 A binary signal has the advantage that a simple slicer shown at a) can be used to recreate a square signal. The result is that noise only causes jitter as shown in b). This can be removed with a phase locked loop.

level is considered a 1 and any voltage below is considered a 0. This slicing process will reject considerable amounts of noise and restore the signal to clean binary once more.

Slicing will fix the noise, but it cannot fix jitter. We need another process to do that. Fig.2.4b) shows that jitter has the effect of displacing voltage changes, or transitions, along the time axis in a random manner. However, the average timing of a large number of transitions is unaffected. A phase-locked loop (PLL) is an electronic circuit which can average the timing of many transitions to recreate a

stable clock from a jittery one. It acts like the flywheel on a piston engine which averages the impulses from the pistons to produce smooth rotation.

The combination of a slicer and a phase-locked loop is called a reclocker. The neat thing about a reclocker is that the waveform coming out of a reclocker is identical to the waveform at the transmitter. Noise and jitter have been rejected so there has been no loss of data due to the transmission. The really neat thing is that we can cascade or tandem as many reclockers as we like and the same data will come out of the end as is shown in Fig.2.5.

The reclocker is found everywhere in digital systems. Digital VTRs use it to eliminate tape noise and jitter. Hard disk drives use it, CD players use it,

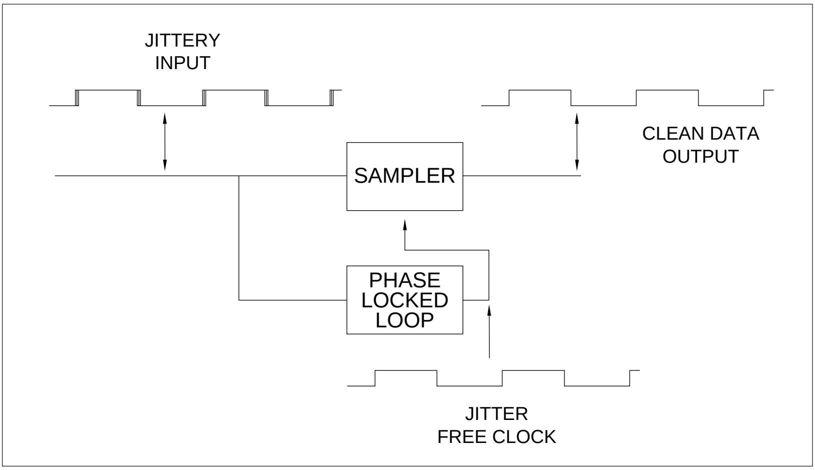


Fig 2.5 A reclocker uses a phase locked loop to recreate a clean clock which samples the input once per bit and produces a jitter free output. Any number of slicers and reclockers can be cascaded without quality loss.

routers use it. The universality of reclocking is due to the fact that it prevents generation loss. A device which contains a reclocker launches a clean signal which is just as robust as the signal launched from the previous device.

2.4 One of the reasons for the use of digital systems is that no matter how many generations of recording or transmission are used, there will be no generation loss. Note that this is only strictly true if we don't employ compression. Fig.2.6 shows a simple system with analog inputs and outputs, analog-to-digital (ADC) and digital-to-analog (DAC) convertors, and a full-bit-rate DVTR with digital inputs and outputs.

The data leaving the ADC are reclocked on receipt at the DVTR and recorded identically on the tape. On playback, the offtape data are reclocked to eliminate tape noise and jitter. Any data which are incorrect because of tape dropouts will be put back to their original values by the integral error correction system prior to transmission to the DAC. The DAC reclocks the received data so that it is identical to that leaving the ADC.

The result is that the picture quality of the system output is determined only by the ADC and the DAC. The ADC could be connected straight to the DAC, or the DVTR could be put into E-E mode, bypassing the tape, and the

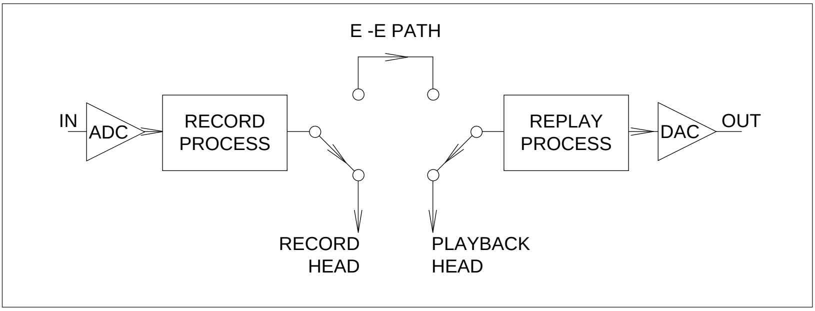


Fig 2.6 In a simple DVTR with effective error correction, the picture quality in playback is the same as in E-E.

picture would not appear any different. This is true provided that there is no compression system and that the data are not corrupted by noise in excess of what the reclockers can handle or by dropouts which are too big for the error correction system of the DVTR. Thus in digital systems there are two main areas of concern:

1/ ADCs and DACs determine picture quality and the best available should be used. Cheap convertors are false economy.

2/ Quality is determined by the convertors only if there are no data errors. Therefore the goal in digital systems is DATA INTEGRITY.

Data integrity cannot be measured with conventional analog monitoring equipment. Just how to measure and maintain data integrity will be explained in chapter 4.

3. How do I convert from analog to digital?

3.1 This section deals with the ADC and the DAC which together largely

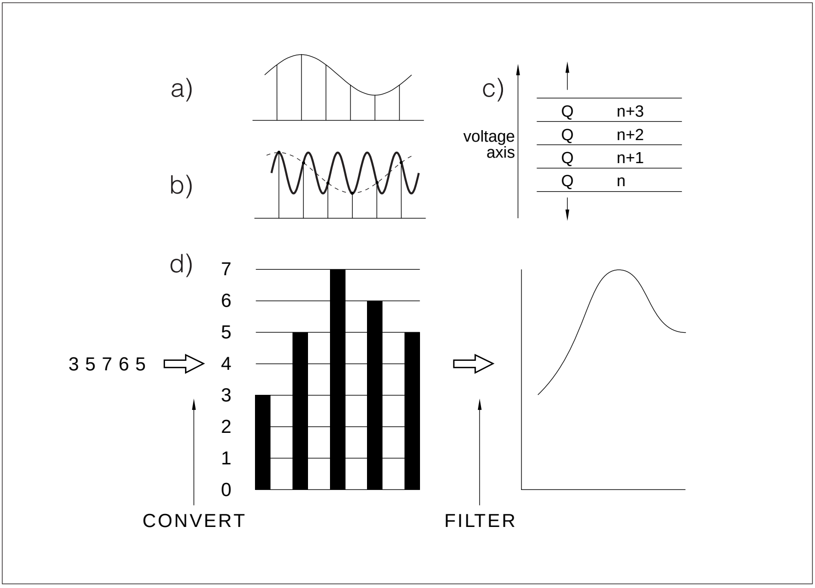


Fig 3.1 a) A high sampling rate is intuitively adequate, but if the sampling rate is too low, aliasing occurs as shown in b). c) Quantizing divides the voltage range up into equal intervals Q. The quantized value is the number of the interval in which the input voltage falls.
d) Numbers are converted back to voltage pulses, and these are filtered to produce a continuous signal.

determine the picture quality achieved in a digital system. Convertors use three fundamental processes: filtering, sampling and quantizing.

3.2 Fig.3.1a) shows what happens when sampling is done right. The original waveform is preserved in the envelope of the samples. Fig.3.1b) shows what happens if we screw up and put in a signal whose frequency is too great for the sampling rate in use. The envelope of the samples now carries an incorrect waveform. This is aliasing; the result of incorrect sampling.

Everyone has seen stagecoach wheels stopping and going backwards in cowboy movies. It's an example of aliasing. The frequency of wheel spokes passing the camera is too high for the frame rate in use. It is important to prevent aliasing in convertors and this is done by including a filter, called an anti-aliasing filter, prior to the sampling stage. Don't leave home without one. The anti-aliasing filter stops frequencies higher than one-half the sampling rate from entering the sampler and so aliasing cannot occur.

3.3 In addition to the sampling process the ADC needs a quantizer to convert the analog sample to a binary number. Fig.3.1c) shows that a quantizer breaks the voltage range or gamut of the analog signal into a number of equal-sized intervals, each represented by a different number. The quantizer outputs the number of the interval the analog voltage falls in.

The position of the analog voltage within the interval is lost, and so an error, called a quantizing error, can occur. As this cannot be larger than a quantizing interval the size of the error can be minimized by using enough intervals. In an eight-bit convertor there are 256 quantizing intervals because this is the number of different codes available from an eight bit number. This allows an unweighted SNR of about 50dB. In a ten-bit convertor there are 1024 codes available and the SNR is about 12dB better. Equipment varies in the wordlength it can handle. Older equipment and recording formats such as D-1 only allow eight-bit working. More recent equipment uses ten-bit samples.

3.4 Fig.3.1d) shows that to convert back to analog, two processes are needed. Firstly a voltage is produced proportional to the binary value of each sample, then these voltages are passed to a reconstruction filter which turns a sampled signal back into a continuous signal. It has that name because it reconstructs the original waveform. So in any digital system, the pictures we see have come through at least two analog filters. In real life we may have to convert a signal in and out of the digital domain several times for practical reasons. Each generation, another two filters are put in series and any shortcomings in the filters tend to get magnified. Thus the best filters available are the ones to use. There is no point in using low-priced filters as their defects will then be passed on perfectly by the digital system.

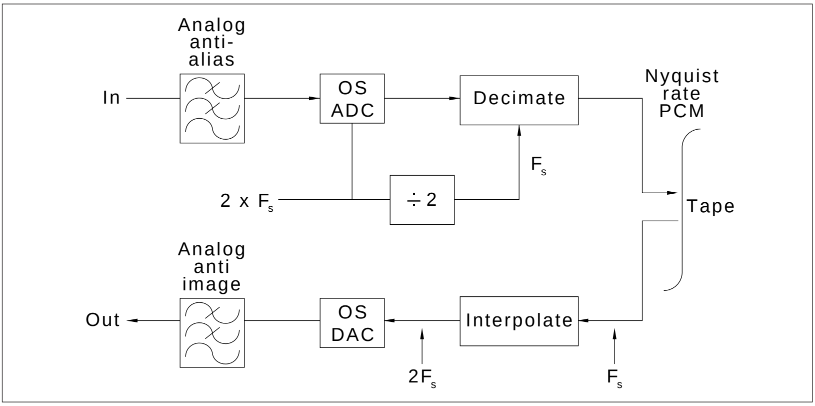
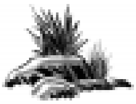


Fig 3.2 In an oversampling convertor the sampling rate is temporarily increased to ease the problems of filtering in the analog domain. The sampling rate is changed up or down in the digital domain where filtering is easier.

3.5 One way of overcoming the drawbacks of analog filters is to use oversampling. This can be applied to ADCs and DACs as Fig.3.2 shows. In an oversampling ADC, the sampling rate is temporarily doubled, so that the analog anti-aliasing filter need only stop signals of twice the video bandwidth. Following the ADC, the doubled sampling rate data are passed through a digital filter which performs a precise anti-aliasing function with better phase linearity than an analog filter could manage. Once this is done the sampling rate can be reduced to the normal figure.



In an oversampling DAC the sampling rate of the input data is doubled in an interpolator which is a digital filter that computes new values lying between the existing samples. The DAC itself now operates with twice the usual sampling rate and the analog reconstruction filter finds it easier to separate the doubled sampling frequencies from the video frequencies.

4. How do digital interfaces work?

4.1 This section discusses the standards which exist for interchanging and routing data representing both component and composite video signals. We also look at the problems of interfacing between composite and component signals.

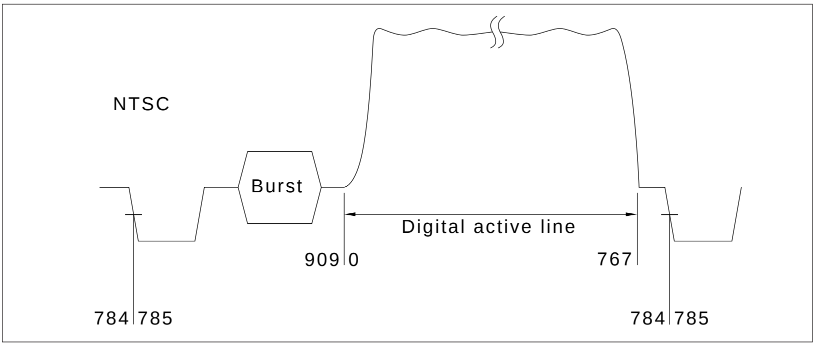


Fig 4.1 The sample numbering of digital NTSC.

4.2 Standardized composite digital video uses a sampling rate which is locked to four times subcarrier frequency (4FSc). The actual rate is about 14.32 MHz in NTSC. This makes life easier because the phase of samples is then stable with respect to the color burst and Y/C separation in digital filters becomes simpler. Fig.4.1 shows how NTSC is sampled at 4FSc. Note that the sample numbering starts at the beginning of the active line and wraps around into the next line. This is because composite digital VTRs like D-2 and D-3 don't record the horizontal blanking interval. They start recording at sample 1 and on replay the horizontal interval is recreated artificially. However, the digital interfaces carry the entire NTSC signal.

Fig.4.2 shows how the analog NTSC signal fits into the eight- and ten-bit quantizing ranges. The sampling is carried out on the I and Q axes and as a result samples do not coincide with burst peaks or crossings.

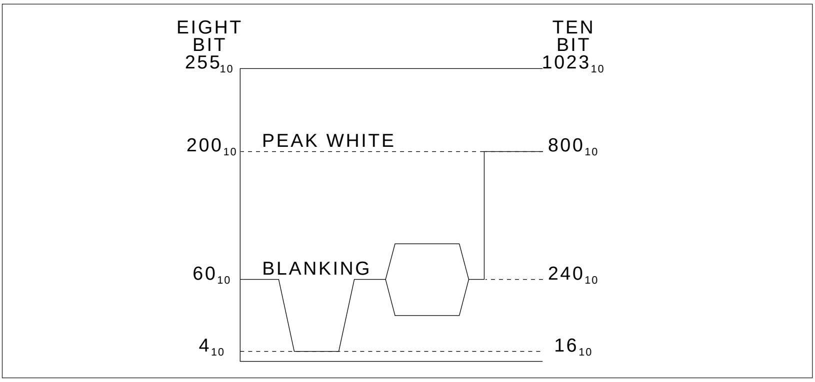


Fig 4.2 How the NTSC waveform fits into the eight- and ten-bit quantizing range.

4.3 Component signals are sampled differently to composite. A common sampling rate was arrived at which allows 525/60 and 625/50 video to be sampled at a common rate locked to horizontal sync. The figure most often used for luminance is 13.5MHz. The price of this commonality is that the pixels are not square. Fig.4.3 shows how the U.S. standard TV line

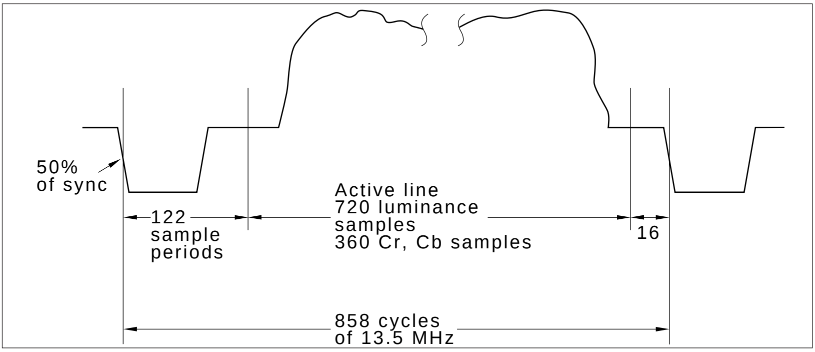


Fig 4.3 The sample numbering of digital component.

fits into 13.5MHz sampling. Note that only the active line is transmitted or recorded in component digital systems. The digital active line has 720 pixels and is a tad longer than the analog active line so the sloping analog blanking is sure to be included.

In component systems, the colour difference signals have less bandwidth. In analog components (from Betacam for example), the colour difference signals have one half the luminance bandwidth and so we can sample them

with one half the sample rate, i.e. 6.75MHz. One quarter the luminance sampling rate is also used, and this frequency, 3.375MHz, is the lowest practicable video sampling rate, which the standard calls 1.

So it figures that 6.75MHz is 2 and 13.5MHz is 4. Most component production equipment uses 4:2:2 sampling. D-1, D-5 and Digital Betacam record it, and the serial digital interface (SDI) can handle it. Fig.4.4a) shows what 4:2:2 sampling looks like in two dimensions. Only luminance is represented at every pixel. Horizontally the colour difference signal values

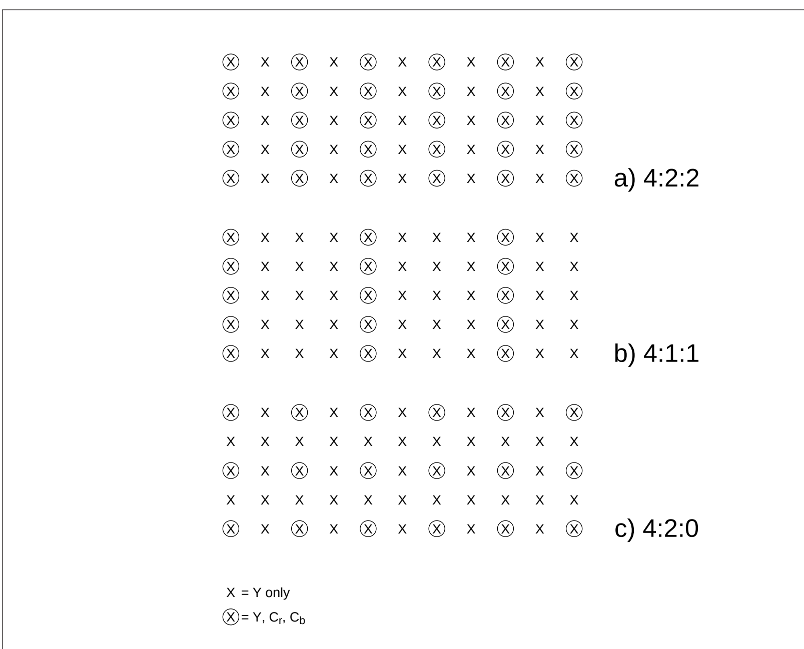


Fig 4.4 a) Post production standard 4:2:2 sampling in two dimensions.
b) 4:1:1 sampling cuts the horizontal colour bandwidth.
c) 4:2:0 sampling matches horizontal and vertical spacing.

are only specified every second pixel. Two other sampling structures will be found in use with compression systems. Fig.4.4b) shows 4:1:1, where colour difference is only represented every fourth pixel horizontally. Fig.4.4c) shows 4:2:0 sampling where the horizontal colour difference spacing is the same as the vertical spacing giving more nearly "square" chroma.

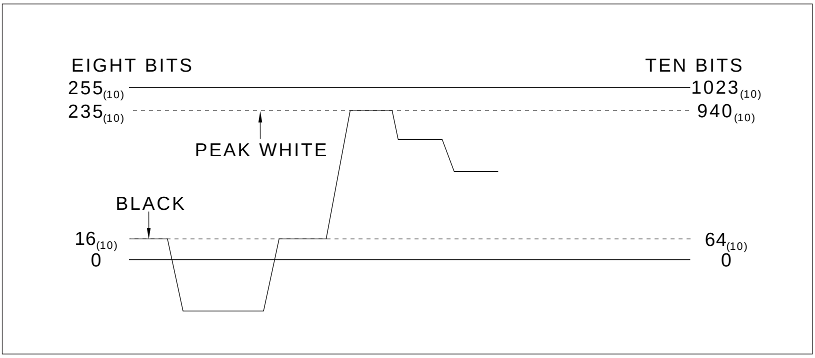


Fig 4.5 How component video fits into the quantizing range for eight-and ten-bit sampling.

Fig.4.5 shows how component digital fits into eight- and ten-bit quantizing. The analog syncs can go off the bottom of the scale because we only use the active line. Colour difference signals are offset upwards so positive and negative values can be handled by the binary number range.

4.4 Interfacing digital video can be done two ways: parallel and serial. Parallel interfaces use a 25-pin D-connector and a heavy multi-core cable. Parallel signals can't go more than about 150 feet before the bits all arrive at different times and confuse the receiver. Maybe the worst news is that a parallel router has to switch every bit and the clock separately, so it is actually an eleven-layer device - complex, expensive and impossible to make in large sizes. With the availability of serial chip sets parallel is now effectively obsolete, but only very recent equipment has serial inputs and outputs as standard. If you have some older digital equipment with only parallel connexions, don't panic, just add a parallel-serial conversion unit. Adaptors of this kind are referred to as "glue" because they help stick things together.

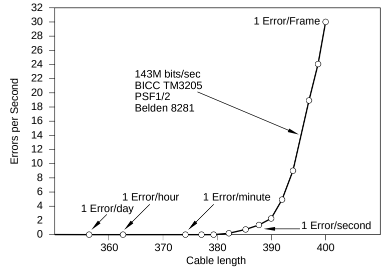
4.5 The serial digital interface (SDI) allows component or composite digital to be passed down regular coax cable. Instead of sending all the bits of a sample at once on different wires, SDI sends them in turn down the same wire. This needs high frequencies - 270 million bits per second for 4:2:2. At that frequency just serializing the bits won't work as some bit patterns produce signals that are difficult to handle. The solution is to use a technique called scrambling which breaks up these awkward bit patterns before sending. A descrambling process at the receiver completely restores the original bit patterns.

SDI signals have such a high frequency that they suffer from cable loss. The further they go, the smaller the signal gets and the more jitter it suffers. This is normal, and the system is designed around the reclocking technique mentioned in section 2.3 so that the losses and jitter are cut out. However, for reclocking to work reliably, the signal cannot be so small that the slicer can't decide whether it's high or low nor can it be so jittery that the phase locked loop can't figure out what to do. Fig.4.6a) shows what cable lengths can be used in SDI. These are not negotiable so don't exceed them. Fig 4.6b) shows what happens if you do. The performance of SDI drops scarily after the maximum design distance is passed. The change is so dramatic engineers call it a crash knee. For reliability you have to leave some margin in an SDI cable installation to keep away from the end of the world.

So how do you measure the margin? This is easy. When the system is first hooked up and working, simply extend the length of each cable in turn with another 50 feet or so of cable. If the system still works fine, the margin is OK. If the system fails the margin is too small. The cable will have to be upgraded, or a repeater installed. Before going to that expense, you might want to check that the phase locked loop in the SDI receiver is adjusted properly. Checking wherther the system is working is easy if you have EDH, as any bit errors will be flagged. Without EDH, you need a signature analyzer. This is a piece of test gear which incorporates a form of EDH. The signature analyzer is fitted after the receiver under test and calculates a)

System	Clock	Fundamental	Crash knee length	Practical length
NTSC Composite	143 MHz	71.5 MHz	400 m	320 m
PAL Composite	177 MHz	88.5 MHz	360 m	290 m
Component 601	270 MHz	135 MHz	290 m	230 m
Component 16:9	360 MHz	180 MHz	210 m	170 m
CABLE: BICC TM3205, PSF1/2, BELDEN 8281 or any cable with a loss of 8.7 dB/100 m at 100 MHz				

b)



Bit error rates (BER)	NTSC	PAL	270Mb	360Mb
1 bit error/field	4.2×10^{-7}	4.2×10^{-7}	4.2×10^{-7}	4.2×10^{-7}
1 bit error/second	4.2×10^{-9}	4.2×10^{-9}	4.2×10^{-9}	4.2×10^{-9}
1 bit error/minute	4.2×10^{-10}	4.2×10^{-11}	4.2×10^{-11}	4.2×10^{-11}
1 bit error/hour	4.2×10^{-12}	4.2×10^{-12}	4.2×10^{-13}	4.2×10^{-13}
1 bit error/day	4.2×10^{-14}	4.2×10^{-14}	4.2×10^{-14}	4.2×10^{-14}
1 bit error/month	4.2×10^{-15}	4.2×10^{-15}	4.2×10^{-15}	4.2×10^{-16}
1 bit error/year	4.2×10^{-16}	4.2×10^{-16}	4.2×10^{-16}	4.2×10^{-17}

Fig 4.6 a) the cable lengths which can be used with SDI. Don't exceed them.
b) the error rate of SDI rises dramatically at the crash knee.
Practical systems must stay to the left of the knee to avoid flaky symptoms.

and displays a check symbol from the incoming frame which is called a signature. The signal source has to be a fixed digitally generated pattern which is exactly the same every frame. If there are errors, the displayed signature will be different every frame. If the system is error free, the signature stays the same.

4.6 In serial digital, component and composite signals are as incompatible as they are in analog. However, the SDI interface signal is just a bit stream, and at that level the only difference is in the bit rate. So it's perfectly possible to build dual-standard routers which can work with composite or component SDI. These devices are available today. A quality router will contain a reclocker, implemented with the aid of a phase locked loop. Often all that is needed to

change from composite to component is to flick a switch which changes the center frequency of the loop. This means that you can buy a router today to upgrade a composite system to digital, and still use it when you change over to components in the future.

4.7 SDI handles digital video which has the same timing as analog video. Thus where analog video has blanking, component digital has a gap in the data. These gaps can be used to transmit ancillary data along with the video. Ancillary data passes through routers transparently and video-only devices ignore it. Smart devices detect the special codes which herald ancillary data and strip it out of the bitstream. Ancillary data can also be used with composite digital, although it is limited to the space available in sync tip because composite digital interfaces still carry syncs and burst.

4.8 Embedded audio is one of the most popular applications of ancillary data. In component digital up to eight channels are possible, composite can manage four. Embedded audio is most useful in tape operations, because many of today's DVTRs automatically include the audio in the SDI signals.

4.9 EDH stands for error detection and handling which is an optional system designed for SDI which detects bit errors and reports them. EDH is not an error correction system. Error correction has to hold up received data until the correction is done, and that delay is intolerable in a production environment. SDI is designed to be robust enough to function without error correction so it causes minimum delay. An SDI error is an unusual event in a well installed system. EDH simply detects errors and tells you about them.

Fig.4.7 shows how EDH works. An EDH equipped transmitting device calculates a check symbol from each frame of video and puts it in a special ancillary data block in the vertical interval after the frame to which it relates. Non-EDH equipped receivers ignore it, but an EDH receiver will make the same check calculation on the received frame, and compare the result with the check symbol which is extracted from the ancillary block. If the two are the same, all is well, but if there is a difference of as little as one bit, a

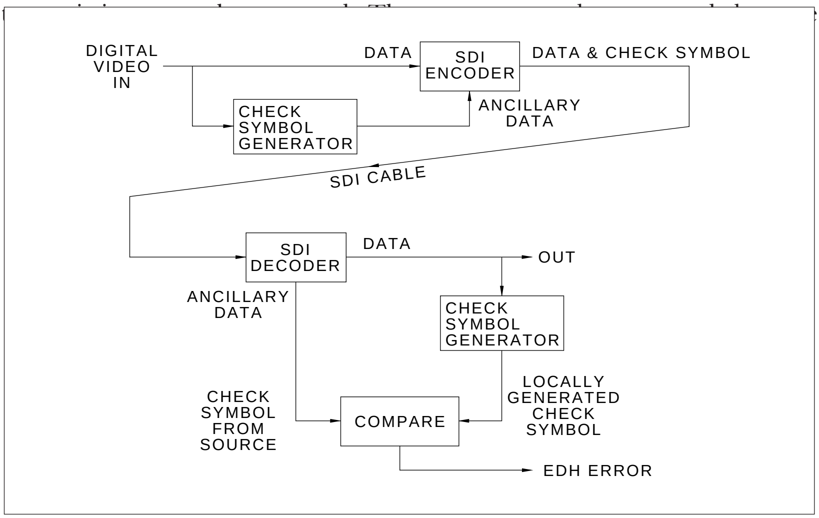
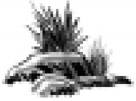


Fig 4.7 In an EDH system, a check symbol is put in the ancillary data by the sending device. The receiver compares this symbol with one it calculates for itself.

by the time the error is detected the frame has gone. Instead the EDH equipped receiver will flag the error. One way in which this can be done is by operating a contact closure which is wired in to your error reporting system. A remote control and status system may have the ability to pass all EDH errors back to a central display.

4.10 Today more and more installations are using components because of the ease with which complex manipulations can be carried out and because of the editing freedom. There is still a lot of composite gear around, however, and interfacing to that requires decoding.



4.11 Why is decoding so difficult?

In ideal NTSC the composite encoding process tightly frequency interleaves the chroma into the luminance in three dimensions: vertically, horizontally and temporally. The additional vertical frequencies are evident in the two-line sequence of NTSC. The additional temporal frequencies are evident in the four-field edit sequence. Real life composite signals are often non-ideal.

Cut edits along the time axis and colour transitions across and down the screen produce high frequencies which fail to interleave in the ideal manner. Such signals are theoretically and practically impossible to decode correctly because chroma frequencies have actually extended into luminance space and vice versa. Composite video is actually a form of compression because it allows colour pictures in the same bandwidth as monochrome. These occasional undecodable composite signals are literally compression artefacts.

Technicalities aside, the point to grasp is that precise composite decoding is not always possible on real signals. There is no one decoding technique which is ideal. Different decoding techniques have different strengths and weaknesses and so it is not surprising that a wide variety of hardware is available. The art of decoding is to employ the right decoder for the job within the available budget.

4.12 Decoding methods

Ideal composite video has a three dimensional frequency interleave. The ultimate separation of chroma and luminance can only be obtained by filtering in all three dimensions. Some decoders do just that. The temporal

filtering process, known as a field comb, requires simultaneous access to a number of different input fields and this can only be done with a large quantity of memory and consequently such decoders cannot be made at low cost. Such techniques cannot be used on unstable signals because instabilities destroy the phase relationships of the chroma from field to field. The inevitable delay is unacceptable in some applications. The ideal composite spectrum is only obtained in the absence of rapid motion and a cut edit disturbs the three-dimensional spectrum. It is interpreted by a field based decoder as enormous motion which causes difficulty.

If the three dimensional filtering approach is not appropriate, the next alternative is to use vertical filtering which depends upon the vertical cancellation of chroma from one picture line to the next. By providing a filter with a number of line delays, a vertical filter, known as a line comb, can be made having good performance without causing a substantial delay to the signal.

It is well known that such filters fail on vertical chroma changes, where the phase relationship from line to line is disturbed. This is the equivalent in the vertical axis of the cut edit in the time axis. The result on screen is called “hanging dots” which are caused by residual chroma which has broken through into the luminance.

Traditionally, line comb decoders adapt to this situation by reverting to steep-cut low pass filtering for the luminance signal. This avoids the chroma breakthrough, but results in a low resolution picture. Snell and Wilcox have developed a superior alternative known as a Gate circuit which is used after the line comb. This patented system contains a notch filter, centred on subcarrier frequency, whose depth and width can be varied over a wide range. The notch is controlled by the degree of chroma breakthrough in a separate comb filter. In this way high frequency luminance is not lost unnecessarily.

Y/C separation cannot be performed by a field or line comb in the case of unstable signals such as VCR replay. In these cases the usual filters must be bypassed and all Y/C separation is performed by the Gate alone.

4.13 Not just a decoder?

In the real world decoding is only one of the processes which need to be performed. Input signals will not necessarily be genlocked to station reference and a frame synchronising stage may be needed. Composite input signals are predominantly analog, but composite digital signals from D-2 or D-3 recorders may also need to be handled. Production steps are increasingly carried out using component digital 4:2:2 signals and it becomes logical to include the analog-to-digital conversion in a decoder. If the conversion to digital is at the input of the decoder, the filtering can all be done digitally. Digital inputs from D-2 or D-3 DVTRs are easily handled if the ADC is bypassed.

5. 4:3 or 16:9?

Now that 16:9 has been adopted for television working and is coming into increasing use, more and more people are coming across the problems of incompatibility and a degree of confusion has arisen. This section is

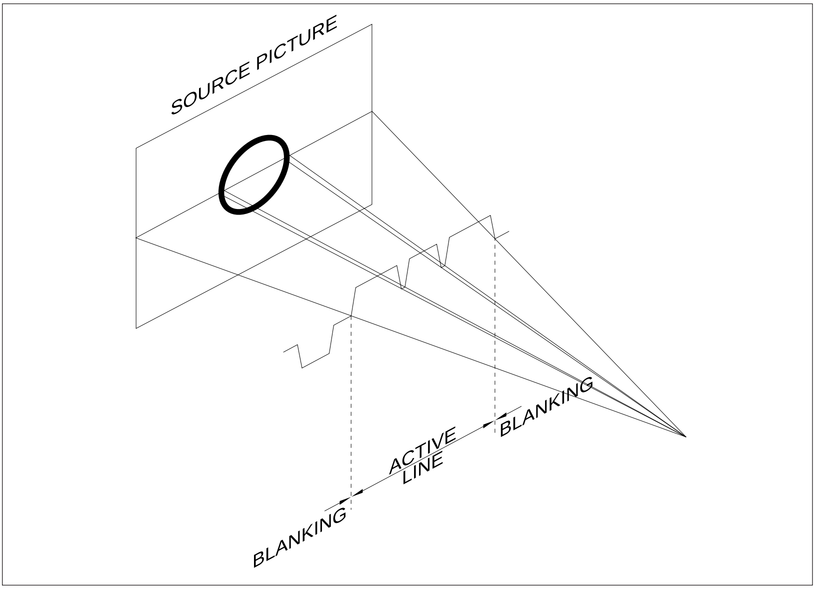


Fig 5.1 The horizontal dimension of the source picture is mapped onto the timespan of the active line.

intended to set out the basics of 16:9 production and illustrates how it can be done alongside existing 4:3 working with the use of aspect ratio convertors.

Fig.5.1 shows how a given picture is mapped onto an analog TV waveform. Neglecting the blanking interval which is needed for tube flyback, the distance across the picture is proportional to the time elapsed along the active line. The camera will break the picture into a standard number of lines, and again neglecting the vertical blanking interval, the distance down the picture will be proportional to the time through the frame in a non-interlaced system. If the format has a fixed number of lines per frame, the aspect ratio of the video format reflects in the ratio of the horizontal and vertical scan speeds. Neglecting blanking, the ratio of horizontal to vertical scan speed in an ideal 525 line system having a square picture would be 525 to 1. In a 4:3 system it would be more like 700 to 1. In a 16:9 system it would be about 950 to 1. A viewpoint of this kind is useful because it is size independent. The picture can be any size as both axes then scale by the same amount.

Clearly if the display is to be compatible with the resultant video format, it must have the same aspect ratio so that the vertical and horizontal mapping retains the correct relationship. If this is done, objects portrayed on the display have the same shape as they had in the original picture. If it is not done correctly there will be distortion. Most test cards contain a circular component to test for this distortion as it is easy to see non-circularity. If a circular object in front of a camera appears circular on the display, their scanning is compatible because both have the same aspect ratio. This test, however, does NOT mean that both camera and display are meeting any standard. For example both camera and display could be maladjusted to underscan by 10% horizontally, yet the circularity test would still succeed. Thus the aspect ratio compatibility test should be made by checking the

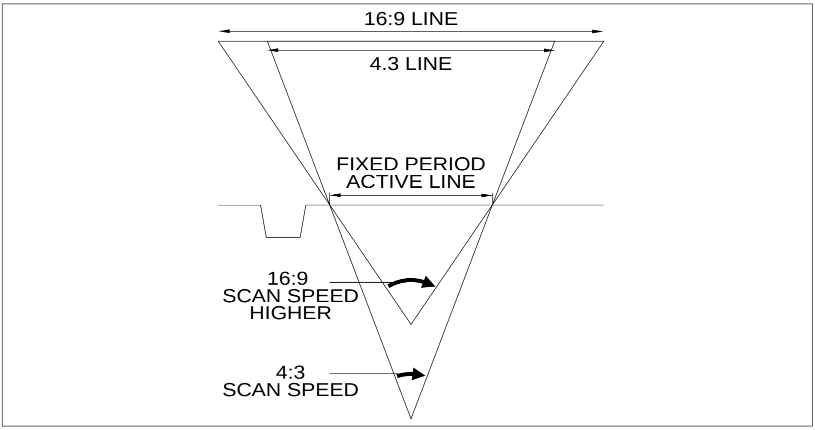
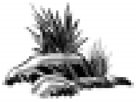


Fig 5.2 Using a higher scan speed, a longer source line can be mapped onto a standard active line

display with an electronically generated precision circle prior to assessing the camera output. Any discrepancy would then be removed by adjusting the camera.

Fig.5.2 shows how a 16:9 picture is mapped onto a video signal. If the frame rate and the number of lines in the frame is kept the same, the wider picture



is obtained by simply increasing the horizontal scan speed at the camera. This allows the longer line to be scanned within the existing active line period. It is obvious that a 16:9 CRT will display the resulting signal with correct circularity.

Any television camera can instantly be adapted to work in this way by fitting an anamorphic lens with a ratio of 1.333...:1 which maps a 16:9 picture onto a 4:3 sensor. Clearly the viewfinder will look rather peculiar so its vertical scan will have to be reduced to 0.75 of its former deflection.

Some tube cameras can be converted to 16:9 by changing the scan amplitudes. Fig.5.3 shows that as the tube is circular, the change is made by reducing the vertical scan a good deal and increasing the horizontal scan by a little. Clearly with a CCD camera this is impossible as the scanning is

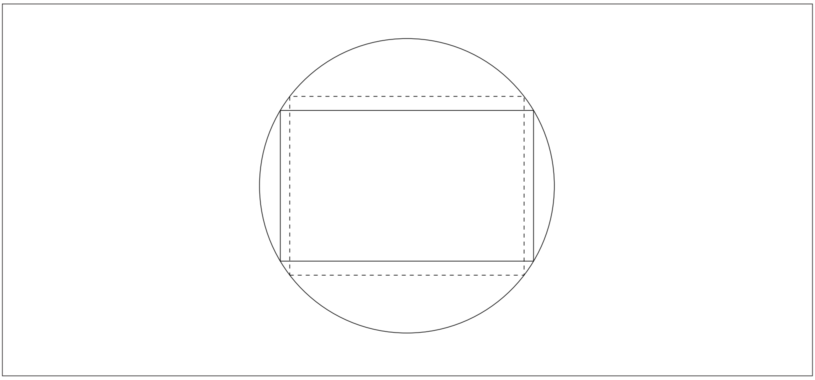


Fig 5.3 On a circular tube, changing to 16:9 requires the vertical scan to be reduced and the horizontal scan to be increased.

locked into the pixel structure. In CCD cameras a wider sensor will be needed if an anamorphic lens is to be avoided.

By redefining the scanning speed ratio at the camera we have produced a different video standard which is now incompatible with 4:3 displays even though a waveform monitor confirms that it meets all the timing specifications. By stretching the horizontal scan the video has been rendered anamorphic. Fig.5.4 shows the result of connecting 16:9 video to a 4:3 monitor. The incompatible mapping causes circularity failure. Circular

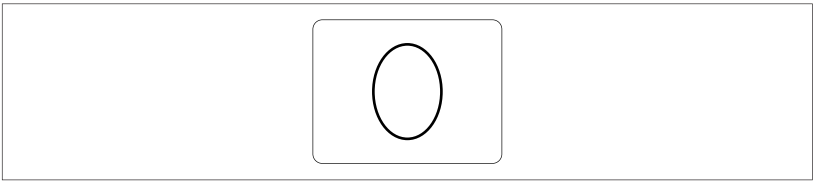


Fig 5.4 The result of displaying 16:9 video on a 4:3 screen is a horizontal compression.

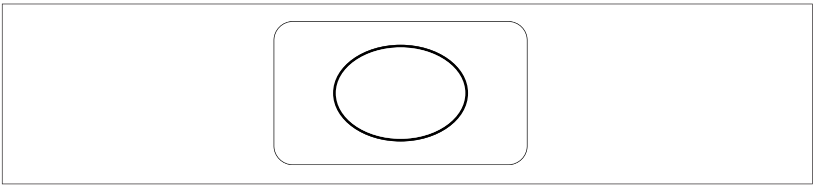


Fig 5.5 The result of displaying 4:3 video on a 16:9 screen is a horizontal stretch.

objects appear as vertical ellipses. Fig.5.5 shows the result of connecting 4:3 video to a 16:9 monitor. Circular objects appear as horizontal ellipses. What is needed is a form of standards convertor which will allow interchange between the two formats.

As 16:9 can use the same scanning parameters as 4:3 the resultant video signal looks the same. In practice 16:9 video will pass through most 4:3 production equipment quite well. As the line is physically longer, yet is scanned in the same time, a given level of detail in a picture will result in a 33% increase in bandwidth. Most analog switchers and routers can handle this without a problem. All that is needed is to re-program pattern generators in switchers so that a circular wipe can still be obtained. Normal analog VTRs will not have enough bandwidth and so the result will be a slight loss of resolution.

Wide-band analog VTRs have been made for 16:9 working, but it is not economic to consider analog recording in a future system.

In a digital component VTR, the standard 13.5 MHz sampling rate used in most DVTRs gives a maximum theoretical video bandwidth of 6.75 MHz. If used anamorphically for 16:9, the horizontal resolution obtained would be the equivalent of a 4:3 system with three quarters of that bandwidth, or about 5 MHz. In practice this bandwidth cannot be met with real anti-aliasing filters and 4.5 MHz is a more realistic figure. This is enough for 525/60 working but marginal for 625/50. An alternative is to use 18 MHz sampling which gives the same resolution in 16:9 as is obtained in 4:3 because the pixel spacing is exactly the same. Fig.5.6 shows that the active line contains 960 pixels instead of 720. Currently only the D-5 format can record such a signal, but disk based recorders will be able to accommodate it quite easily as the disk format is virtually independent of the video standard.

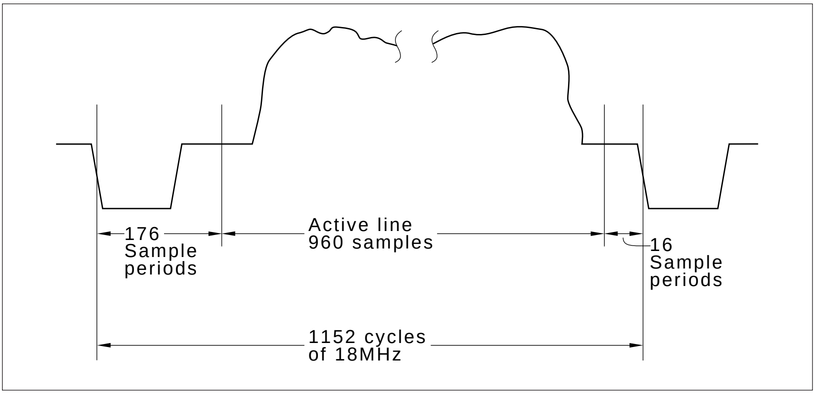


Fig 5.6 Raising the sampling rate to 18MHz allows the same pixel spacing with the faster scan of 16:9. There are now 960 samples per active line instead of 720.

Standards conversion between 16:9 and 4:3 has come to be called aspect ratio conversion. An aspect ratio convertor allows interchange between 4:3 and 16:9 signals. There are two basic applications as can be seen in Fig.5.7. If 16:9 cameras and production equipment are used, an aspect ratio convertor is needed to view material on 4:3 monitors and to obtain a traditional broadcast output.

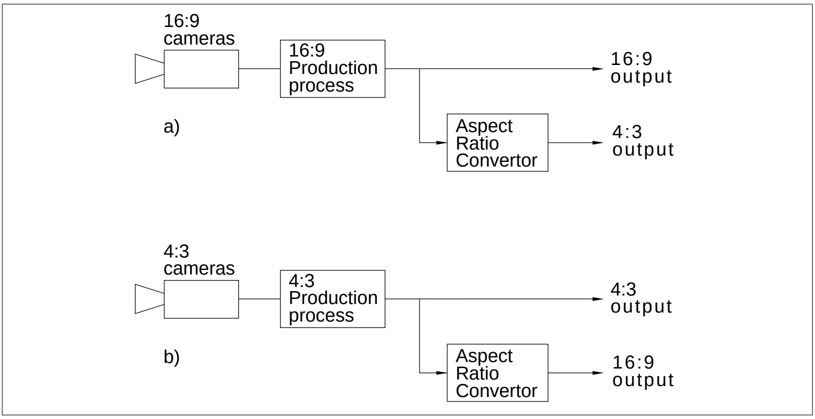


Fig 5.7 Whether production is in 16:9 or 4:3, a conversion process is necessary to allow dual standard output.

Alternatively, conventional cameras can be used with a large safe area at top and bottom of the picture. 4:3 equipment is used for production and the aspect ratio convertor is then used to obtain a 16:9 picture output.

The criterion for conversion must be that circularity has to be maintained otherwise the pictures will appear distorted. Thus an aspect ratio convertor must change the aspect ratio of the picture frame, without changing the aspect ratio of portrayed objects.

If circularity is maintained, something else has to go. Fig.5.8a) shows the result of passing 16:9 into a convertor for 4:3 display. If the screen must be filled, the convertor must perform a horizontal transform of 1.333...:1 to maintain circularity. With 18 MHz digital input the transform is trivial as it involves selecting 720 pixels out of 960 to make a 13.5 MHz output. The result of doing this alone is that the edges of the input picture are lost as they will be pushed outside the active line length. This may be acceptable if a pan/scan control is available. Alternatively, if no part of the image can be lost, the convertor must perform a vertical transform of 0.75:1. This will result in the vertical blanking area of the 16:9 input entering the 4:3 screen area and the result will be black bars above and below the picture.

Fig.5.8b) shows the reverse conversion process where 4:3 is being converted to 16:9. Again if "full screen" mode is required, the convertor must perform

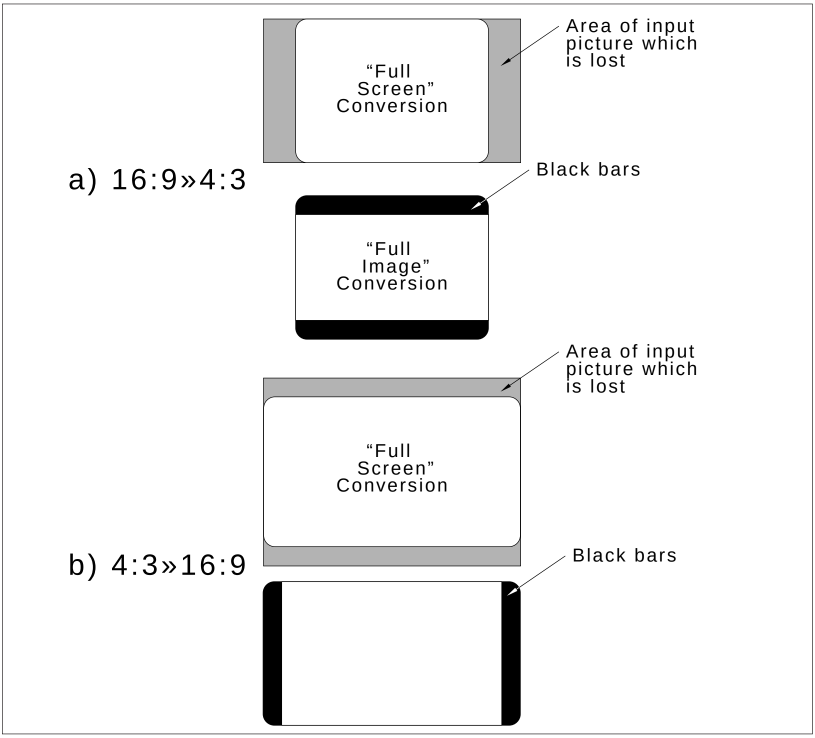


Fig 5.8

a vertical transform of 1.333..:1 to maintain circularity. This pushes the top and bottom of the input picture into 16:9 blanking. If the 4:3 material was shot with 16:9 safe areas this is no problem. However, if the input was intended for 4:3 it may have wanted detail near the top or bottom of the picture and a tilt (vertical pan) control may be provided to select the area

which appears in the output. If no part of the image can be lost, i.e "full image" mode is required, a horizontal transform of 0.75:1 is needed, and this must result in the horizontally blanked areas of the 4:3 input entering the 16:9 screen area.

The above steps represent the two extremes of full screen or no image loss. There is a scale between those extremes in which the black bars can be made smaller in one axis by an image loss in the other axis. Accordingly a practical aspect ratio convertor needs to perform vertical and horizontal transforms which may be magnification, translation or both. In order to maintain circularity, the ratio between the horizontal and vertical magnifications can only have three values, 1.333..:1 for 16:9 to 4:3 conversion, 1:1 for bypass and 0.75:1 for 4:3 to 16:9 conversion. Having selected the mode, a single magnification control would vary the conversion between “full screen” and “full image” modes. When not in full image mode, pan and tilt controls allow the user to select the part of the input image which appears in the output.

6. What is compression about?

In a general sense compression, also known as data reduction or bit-rate reduction, is a high tech way of getting more out of less. The sound and picture still arrive but the bit rate or bandwidth required is less than if the signals are sent in their original form. From a purely black box viewpoint, compression is accountant heaven. Every process which relies upon storing or conveying large quantities of data can be made more economical using compression. Existing processes get cheaper, new processes become viable. Digital Video Broadcasting (DVB) and Video-on-Demand (VOD) would be impossible without it. Non-linear editors would be hopelessly uneconomic.

Compression is a flexible technology because the degree of coding complexity and the degree of compression used can be varied to suit the application. Video contains redundancy because typical images contain areas which are similar. The actual information in video is known as the entropy which is the unpredictable or new part of the signal, and the remainder is redundancy, which is a part of the signal which is predictable. The sum of the two is the original data rate. If the degree of compression is so great that the resulting data rate is less than the entropy, then information will be lost. In theory, all of the redundancy could be removed, leaving only the entropy, but this would require a perfect algorithm which would be extremely complex.

In practice the compression factor should be less than this for production purposes so that some safety margin is available. This allows simpler algorithms to be used and where necessary also permits multiple generations without artifacts being excessive. Thus production DVTRs such as Sony's Digital Betacam and the Ampex DCT use only very mild compression of around 2:1.

In consumer equipment, the compression factor will be higher than for production and some of the entropy will be thrown out in the compression process.

For production recorders, only redundancy within the field is used, and the compression type is called intra-coding. No advantage is taken of the redundancy from one field to the next as this would compromise editing.

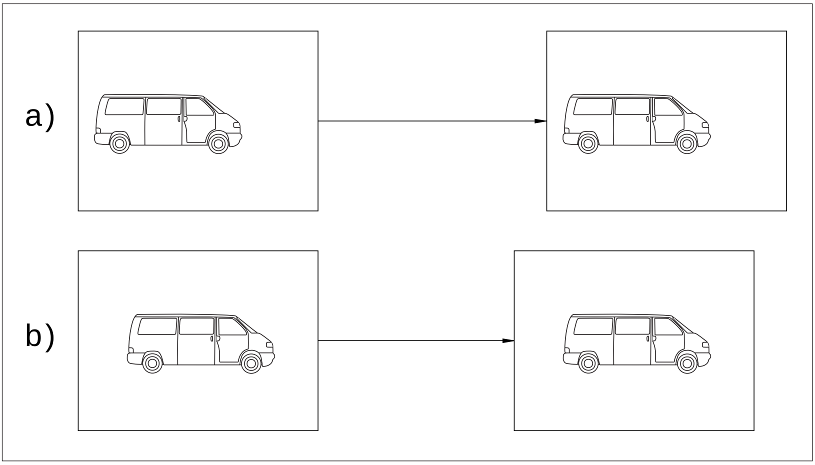


Fig 6.1 In an inter-coded system, picture differences are sent. In stationary areas, the differences are small, but on moving objects, large differences are generated as a) shows. Motion compensated compression is designed to reduce the difference by cancelling the motion as in b).

If editing is not a requirement higher compression factors are easier to obtain if advantage is taken of redundancy between successive images. In inter-coding, only the difference between images need be sent. With a still picture, successive images will be identical and the difference won't amount to much. In practice, movement reduces the similarity between successive images and the difference data increases. One way of getting over that is to use motion compensation. If the motion of an object between images is known and transmitted as a motion vector, the decoder can use that motion vector to shift the pixel data describing the object in the previous image to the correct position in the current image. The image difference for the object will then be smaller.

Fig.6.1 shows the principle. If two pictures are superimposed as shown, it will be seen that there is only redundancy or similarity in the stationary

background. However, if one of the pictures is shifted according to the calculated motion vectors prior to superimposition, it will be found that there is now a great deal of similarity in the data representing the moving object.

A motion compensated coder works as follows: A complete picture is intra-coded and sent, but is also locally stored so that it can be compared with the next input picture to find motion vectors for various areas of the picture. The first picture is then shifted according to these vectors to cancel inter picture motion, and compared with the second picture to produce difference data. It will be seen from Fig.6.2 that the difference data and the

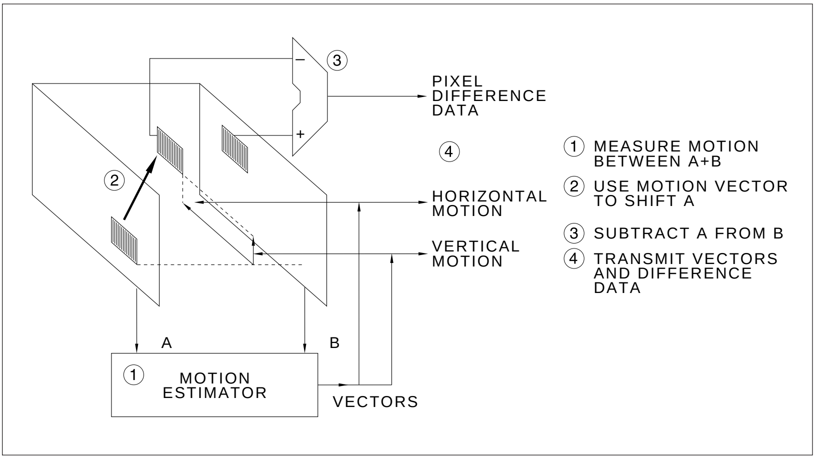


Fig 6.2 In a motion compensated system, vectors are sent telling the decoder how to shift an area of the previous picture prior to adding the difference data to make the current picture.

motion vectors are transmitted. At the receiver the decoded first picture is also held in a memory. It is then shifted according to the transmitted motion vectors and then the difference data are added to it to create the second picture. Any desired number of subsequent pictures can be sent as motion vectors and difference data with respect to the first picture before the process repeats. Clearly the difference data can be treated as pictures and can be subject to further compression techniques. The MPEG compression standards are based on processes of this kind.

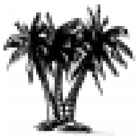
Motion compensated coding only works properly if the incoming signals are squeaky clean. Noise, dropout concealments, timebase error, film grain, camera shake, film weave and residual subcarrier all appear to a motion compensated system as extra differences between pictures. These differences gobble up data rate leaving less for the picture. If you bought a compressor which looked great on studio-shot 4:2:2 source material, don't be surprised if it looks like the pits on input from a VCR or a satellite. If you can't guarantee consistent high signal quality, think about getting a noise reducer at the same time as your compression system.

Compression algorithms intended for transmission of still images in other applications such as wirephotos can be adapted for intra-field video compression. The ISO JPEG (Joint Photographic Experts Group) standard is such an algorithm. Inter-field data reduction allows higher compression factors with infrequent artifacts for the delivery of post-produced material to the consumer. With even higher compression factors, leading to frequent artefacts, non-critical applications such as videophones, CD-ROM and games are supported where a low data rate is mandatory. The ISO MPEG (Moving Picture Experts Group) standards address these applications.

MPEG 1 is a simple system which gets over the complexity of interlace by simply throwing away every other field before compression begins. As the vertical resolution and the motion portrayal performance are effectively halved by this move, then for consistency the horizontal resolution is also halved. Thus the compressor proper begins with one quarter of the input bandwidth and not surprisingly achieves a high compression factor even if the picture does look as if it had been in one of those hydraulic things the wreckers yard uses to crush junked automobiles.

MPEG 2 recognises interlace and does not need to subsample before compression because it does not attempt such a high compression factor.

As a result the pictures are quite good if the application is restricted to final delivery of material which has already been post produced.



If post production is going to be done, then a contribution quality compression will be needed, allowing a comfortable performance margin. Around 30-40 megabits/sec delivers contribution quality.

You don't have to understand the complexities of compression if you stick to the following rules:-

1. If compression is not necessary don't use it.

2. If compression has to be used, keep the compression factor as mild as possible; i.e. use the highest practical bit rate.

3. Don't cascade compression systems. This causes loss of quality and the lower the bit rates, the worse this gets. Quality loss increases if any post production steps are performed between compressions.

4. Compression systems cause delay.

5 Compression systems work best with clean source material. Noisy signals, film grain and weave or poorly decoded composite video give poor results.

6. Compressed data are generally more prone to transmission errors than non-compressed data.

7. Only use low bit rate coders for the final delivery of post produced signals to the end user.

8. Compression quality can only be assessed subjectively.

9. Don't believe statements comparing codec performance to "VHS quality" or similar. Compression artefacts are quite different to the artefacts of consumer VCRs.

10. Quality varies wildly with source material. Beware of "convincing" demonstrations which may use selected material to achieve low bit rates. Use your own test material, selected for a balance of difficulty.

7. How do digital VTRs work?

This section looks at how DVTRs work, how to check their data integrity, and compares the available formats. The digital VTR is actually a data recorder which is optimised for working with data representing video. Production DVTRs need to do much more than record and play. They must

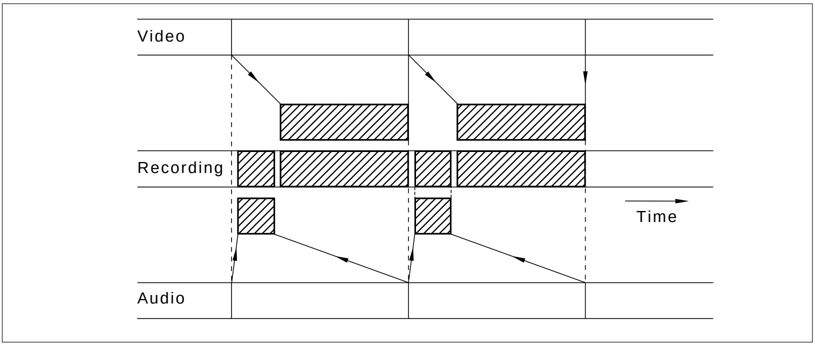


Fig 7.1 In time compression, data are squeezed so that several signals can share the same track.

be able to record several independent audio channels, they must work in slow motion and they must be able to edit using time code.

As digital audio and digital video are just different kinds of data, DVTRs record both using as much common hardware as possible. The same heads and tape tracks are used to record both. The tracks are simply divided by edit gaps so that part of the track can be updated without disturbing the rest. Time compression is used to get audio and video data into the same track. Fig.7.1 shows how time compression works. Audio and video data are temporarily stored in RAM prior to recording. When the RAM is read, the clock frequency is raised so that the data come out in less than real time. The video data are squeezed up by a few percent to make room for the audio. The audio data are heavily time compressed to have the same data rate as the video so that the same read/write circuitry can be used.

Tape is imperfect and suffers from dropouts where some of the magnetic coating is missing. Dust gets between the head and the tape momentarily blanking out the signal. DVTRs use error correction and concealment to deal with most of these problems. It is important to appreciate the difference. Error correction actually computes the values of pixels which are missing and restores the data to exactly its original form. Thus error

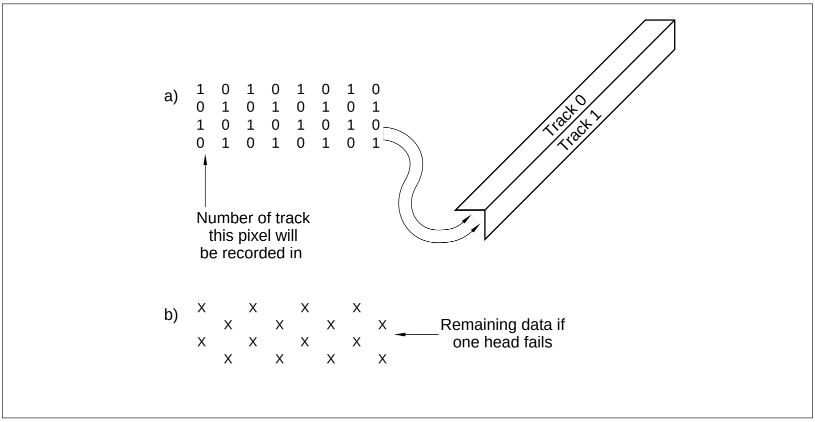


Fig 7.2 Distributing samples over two heads in the pattern shown allows concealment in the case that one head clogs.

correction is indetectable in operation. However, some errors are just too big to be corrected, such as when a head clogs up. In this case concealment is used. Concealment cannot compute exact pixel values, but instead

estimates probable values from correct pixels nearby. Thus concealment is potentially visible, but in most cases the estimate is remarkably good and goes unnoticed.

The error strategy of DVTRs is based on combinations of the following ideas:

1/ Distribution. Data are distributed over two or more heads and tracks working in parallel. Fig 7.2a) shows that the distribution is done by feeding a particular head with alternate pixels in rows and columns. If one head clogs, it can be seen in Fig.7.2b) that what is left is the original picture, but sampled less often. Interpolation can be used to estimate values of the missing pixels.

2/ Shuffling. Fig.7.3 shows that a dropout usually causes an area of severe damage in the picture, surrounded by intact areas. If the dropout is correctable this is no problem, but if concealment has to be used, the

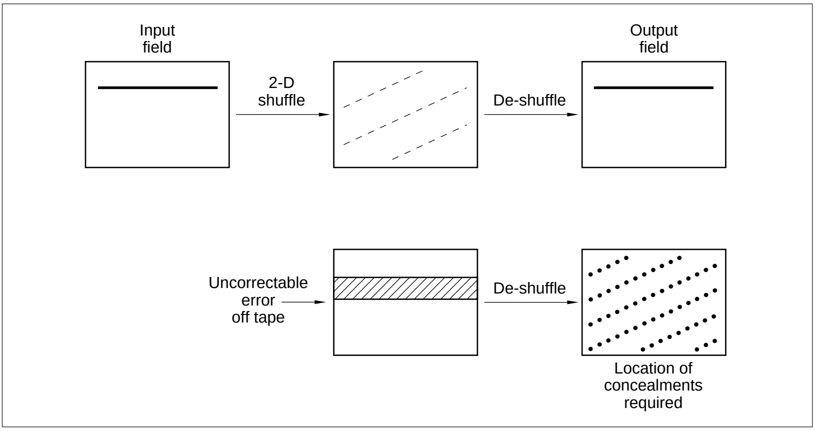


Fig 7.3 In shuffling, pixels are pseudo-randomly moved around the picture on recording and moved back on replay. This splits up regular defects due to uncorrected data blocks and instead, easily concealable single errors are dotted around the screen.

concealed area will be obvious to the viewer. The solution is a shuffle process which messes up the order of the pixels before recording and then straightens them out on playback. When shuffling is used, an uncorrectable dropout results in pixels requiring concealment which are randomly spread around the screen rather than all together. Concealment is then more successful.

3/ Product codes. Product codes are an error correction strategy where pixels are formed into a rectangular array prior to recording.

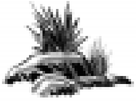


Fig.7.4 shows that check words are then added both to the rows and the columns of the array prior to recording.

On replay there is a two stage process. The rows are checked for error, and small errors are corrected. Large errors, however, are used to produce flags which go into the array. When columns are read from the array, the second stage of correction is told by the flags where the errors are.

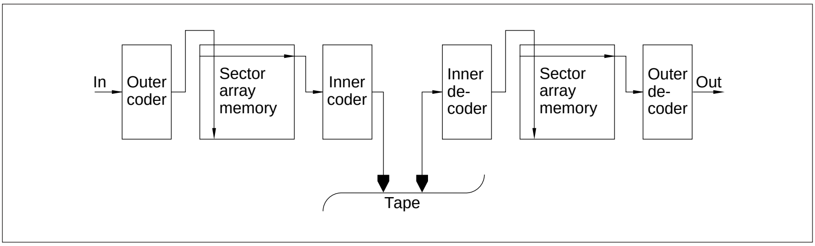


Fig 7.4 In a product code, error correcting codes cross in two dimensions in a data array. Errors detected in one code can be used as pointers by another code crossing it.

In the presence of error correction and concealment, it is impossible to assess how well a DVTR is working by looking at the replay picture. The tape and heads may be new and clean, or so worn and dirty as to be within a whisker of exceeding the power of the error correction system but a picture monitor will show the same quality.

The only sure way of establishing the performance margin of a DVTR is to enter the error rate menu and observe the rate at which error correction is being performed. If this is higher than the normal figure, maintenance is needed. Some machines allow the concealment and/or error correction system to be disabled for testing purposes. The error detection still works, but a pixel in error is replaced by a mid-grey pixel so it is visible on the monitor screen. The error rate can then be estimated visually. This method is quite good for checking a tape which is suspected of having dropouts as individual frames can be inspected with the jog function, but don't make the mistake of going on air with it!.

The amount of data which is needed to represent a field is so great that it is not possible to record it all in a single tape track as analog VCRs do. The solution is segmentation, where several tracks are required to record a whole

field. Although there are more tracks, the tape used is less because the tracks are so narrow.

All modern DVTRs use azimuth recording to increase efficiency. Fig.7.5 shows that in azimuth recording the heads are twisted alternately in azimuth so that alternate tracks have magnetic patterns which have a different

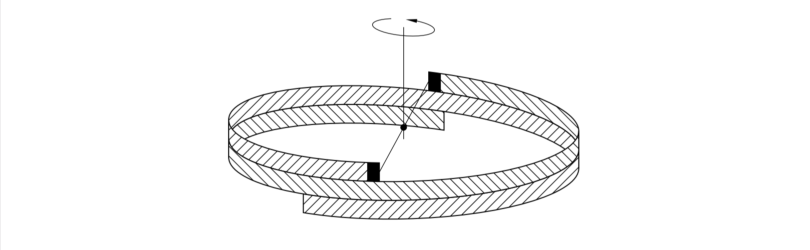


Fig 7.5 Modern VTRs use azimuth recording in which the heads are twisted to record alternate tracks having two different azimuth angles. No gaps between tracks are then needed.

alignment. Heads with the same alignment can read the tracks without difficulty, but heads of the opposite azimuth type can't. This means that the tracks can be put side by side without a guard band.

Fig.7.6 shows a typical DVTR tape footprint. Note the alternating azimuth, segmentation and edit gaps between the audio and video sectors.

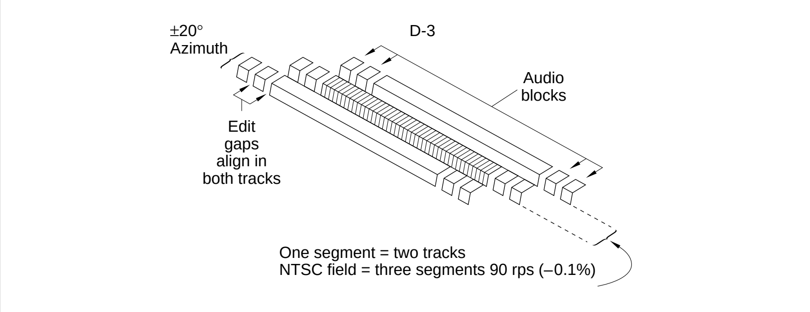


Fig 7.6 A typical DVTR footprint on tape. See text.

The first production DVTR, launched in 1987, used the D-1 format which recorded colour difference data according to CCIR-601 on 3/4 inch tape. D-1 was too early to take advantage of high energy tapes and its recording density was quite low, leading to large cassettes and high running costs. The majority of broadcasters then used composite signals, and a component recorder could not easily be used in such an environment. Where component applications existed, the D-1 format could not compete economically with Betacam SP and M-II analog formats. D-1 found application only in high-end post production.

D-2 came next, but this was a composite digital format, handling conventional PAL and NTSC signals in digital form, and derived from a format developed by Ampex for an automated cart. machine. The choice of composite recording was intended to allow broadcasters directly to replace analog recorders with a digital machine. D-2 retained the cassette shell of D-1 but employed higher energy tape and azimuth recording to improve recording density and playing time. Early D-2 machines had no flying erase heads, and difficulties arose with audio edits. D-2 was also hampered by the imminent arrival of the D-3 format.

D-3 was designed by NHK, and put into production by Panasonic. This had twice the recording density of D-2; three times that of D-1. This permitted the use of 1/2 inch tape, making a digital camcorder a possibility. D-3 used the same sampling structure as D-2 for its composite recordings. Coming later, D-3 had learned from earlier formats and had a more powerful error correction strategy than earlier formats, particularly in the audio recording. Note that as D-2 and D-3 are composite machines, they still require color framing systems and must edit in four-field lock.

By this time the economics of VLSI chips had made made compression viable, and the first application was the Ampex DCT format which used approximately 2:1 data reduction so that component video could be recorded on an updated version of the 3/4 inch cassettes and transports designed for D-2.

When Sony were developing the Digital Betacam format, compatibility with the existing analog Betacam format was a priority. Digital Betacam uses the same cassette shells as the analog format, and certain models of the digital recorder can play existing analog tapes. Sony also adopted compression, but this was in order to allow the construction of a digital component VTR which offerred sufficient playing time within the existing cassette dimensions and which would be as robust as its analog predecessor.

The D-5 component format is backward compatible with D-3. The same cassettes are used and D-5 machines can play D-3 tapes. Compression is not used; the tape speed is doubled in the component format in order to increase the bit rate.

D-6 is a digital high definition format based on the D-1 cassette shell.

The DVC (digital video cassette) format uses quarter inch tape and a compression factor of about 5:1 to record component video in a cassette which is extremely small by consumer or professional standards. When hardware to this format becomes available the ENG scene will be transformed.

In the future recording technology will continue to advance and further formats are inevitable as manufacturers perceive an advantage over their competition. This does not mean that the user need slavishly change to every new format, as the cost of format change is high. The wise user retains his current format for long enough to allow a number of new formats to be introduced. He will then make a quantum leap to a format which is much better than the present one, missing out those between and minimising the changeover costs.

8. What is non-linear editing about?

8.1 Non-Linear editing takes advantage of the freedom to store digitised image data in any suitable medium and the signal processing techniques developed in computation. In all types of editing the goal is to output the appropriate sequence of material at the appropriate time. In an ideal world the difficulty and cost involved in creating the perfect edited work are discounted. In practice there is economic pressure to speed up the editing process and to use cheaper media. Editors will not accept new technologies if they form an obstacle to the creative process, but if a new approach to editing takes nothing away, it will be considered. If something is added, such as freedom or flexibility, so much the better.

When there was only film or video tape editing, it did not need a qualifying name. Now that images are stored as data, alternative storage media have become available which allow editors to reach the same goal but using different techniques. Unlike today's DVTRs, in all other digital editing, samples from various sources are brought from the storage media to various pages of RAM. The edit is viewed by selectively processing two (or more) sample streams retrieved from RAM. Thus the nature of the storage medium does not affect the form of the edit in any way except the amount of time needed to execute it.

8.2 Tapes only allow serial access to data, whereas disks and RAM allow random access and so can be much faster. Editing using random access storage devices is very powerful as the shuttling of tape reels is avoided. The technique is called non-linear editing because the time axis of the storage medium is made non-linear.

Fig.8.1 shows what's in a hard disk based workstation. The screen shows a montage of many different signals, each of which appear in windows. In addition to the video windows there will be a number of alphanumeric and

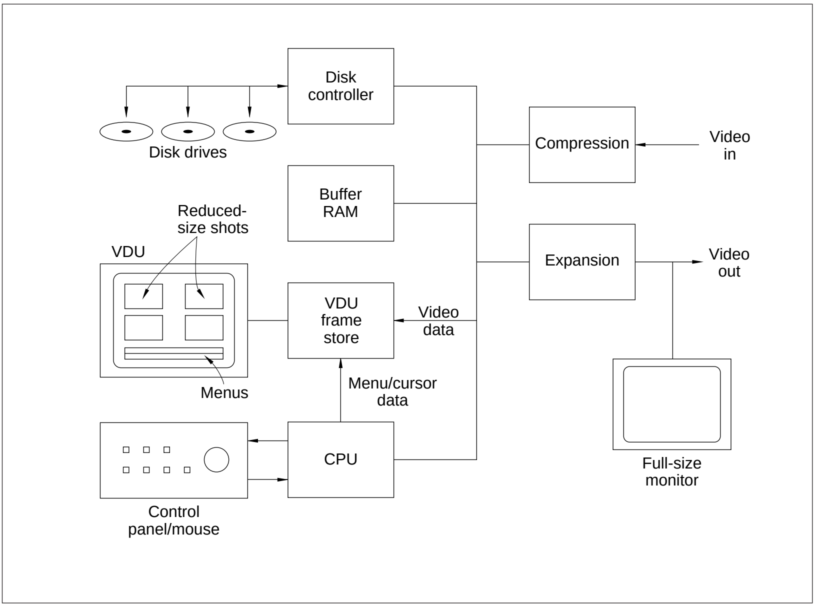


Fig 8.1 The components of a hard disk workstation. The hardware is basically a computer. It's the human interface and the software that makes it into an editing machine.

graphic display areas required by the control system. There will also be a cursor which can be positioned by a trackball or mouse. The screen is refreshed by a framestore which is read at the screen refresh rate. The framestore can be simultaneously written by various processes to produce a windowed image. There may also be further screens to reproduce full size images for preview purposes.

Digital inputs and outputs are provided, along with optional convertors to allow working in an analog environment. In many workstations, compression is employed, and the appropriate coding and decoding logic will be required adjacent to the inputs and outputs. With mild compression, the video output of the machine may be used directly for some purposes. This is known as on-line editing. Alternatively a high compression factor may be used, and the editor is then used only to create an edit decision list (EDL).

This is known as off-line editing. The EDL is then used to control automatic editing of the full bandwidth source material, most likely on tape.

8.3 Disk based workstations fall into two categories depending on the relative emphasis of the vertical or horizontal aspects of the process. High end post-production emphasises the vertical aspect of the editing as a large number of layers may be used to create the output image. The length of such productions is generally quite short and so disk capacity is not an issue and data reduction will not be employed. In contrast a general purpose editor used for television program or film production will emphasise the horizontal aspect of the task. Extended recording ability will be needed, and the use of compression is more likely.

The machine will be based around a high data rate bus, connecting the I/O, RAM, disk subsystem and the processor. If magnetic disks are used, these will be Winchester types, because they offer the largest capacity. Exchangeable magneto-optic disks may also be supported.

8.4 Before any editing can be performed, it is necessary to have source material on line. If the source material exists on magneto-optical (MO) disks with the appropriate file structure, these may be used directly. Otherwise it will be necessary to input the material in real time and record it on magnetic disks via the compression system. In addition to recording the compressed source video, reduced size versions of each field may also be recorded which are suitable for the screen windows.

Digital editors must simulate the "rock and roll" process of edit-point location in VTRs where the tape is moved to and fro by the action of a shuttle knob, jog wheel or joystick. Whilst DVTRs with track following systems can work in this way, disks cannot. Disk drives transfer data intermittently and not necessarily in real time. The solution is to transfer the recording in the area of the edit point to RAM in the editor. RAM access can take place at any speed or direction and the precise edit point can then be conveniently found by monitoring signals from the RAM. In a window based display, a source recording is attributed to a particular window, and will be reproduced within that window, with timecode displayed nearby.

Before the edit can be performed, two edit points must be determined: the out point at the end of the previously recorded signal, and the in point at the beginning of the new signal. The second edit point can be determined by moving the cursor to a different screen window in which video from a different source is displayed. The jog wheel will now roll this material to locate the second edit point while the first source video remains frozen in the deselected window. The editor's microprocessor stores these in an edit decision list (EDL) in order to control the automatic assemble process.

8.5 It is also possible to locate a rough edit point by typing in a previously noted timecode, and the image in the window will automatically jump to that time. In some systems, in addition to recording video and audio, there may also be text files locked to timecode which contain the dialog. Using these systems one can allocate a textual dialog display to a further window and scroll down the dialog or search for a key phrase as in a word processor. Unlike a word processor, the timecode pointer from the text access is used to jog the video window. As a result an edit point can be located in the video if the actor's lines at the desired point are known.

8.6 Using one or other of the above methods, an edit list can be made which contains an in point, an out point and a filename for each of the segments of video which need to be assembled to make the final work, along with a time code referenced transition command and period for the vision mixer. This edit list will also be stored on the disk. When a preview of the edited work is required, the edit list is used to determine what files will be necessary and when, and this information drives the disk controller.

Fig.8.2 shows the events during an edit between two files. The edit list causes the relevant blocks from the first file to be transferred from disk to memory, and these will be read by the signal processor to produce the preview output. As the edit point approaches, the disk controller will also place blocks from the incoming file into the memory. In different areas of the

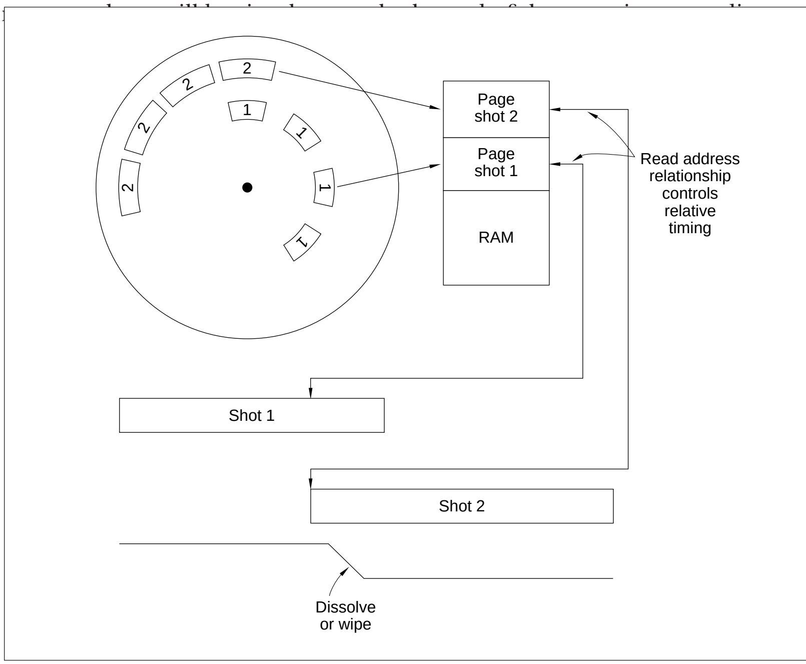


Fig 8.2 During an edit the disk system must supply data relating to two video signals at once so that a dissolve or wipe may be made between them.

the beginning of the incoming recording. Before the edit point, only pixels from the outgoing recording are accessed, but as the transition begins, pixels from the incoming recording are also accessed, and for a time both data streams will be input to the vision mixer according to the transition period required. The output of the signal processor becomes the edited preview material, which can be checked for the required subjective effect. If necessary the in or out points can be trimmed, or the crossfade period changed, simply by modifying the edit decision list file. The preview can be repeated as often as needed, until the desired effect is obtained. At this stage the edited work does not exist as a file, but is recreated each time by a further execution of the EDL. Thus a lengthy editing session need not fill up the disks.

8.7 It is important to realize that at no time during the edit process were the original files modified in any way. The editing was done solely by reading the files. The power of this approach is that if an edit list is created wrongly, the original recording is not damaged, and the problem can be put right simply by correcting the edit list. The advantage of a disk-based system for such work is that location of edit points, previews and reviews are all performed almost instantaneously, because of the random access of the disk. This can reduce the time taken to edit a program to a fraction of that needed with a tape machine.

During an edit, the disk controller has to provide data from two different files simultaneously, and so it has to work much harder than for a simple playback. If there are many close-spaced edits, the controller and drives may be hard-pressed to keep ahead of real time, especially if there are long transitions, because during a transition a vertical edit is taking place between two video signals and the source data rate is twice as great as during replay. A large buffer memory helps this situation because the drive can fill the memory with files before the edit actually begins, and thus the instantaneous sample rate can be met by allowing the memory to empty during disk-intensive periods.

8.8 Once the editing is finished, it will generally be necessary to transfer the edited material to form a contiguous recording so that the source files can make way for new work. In off-line editing, the source files already exist on tape or film and all that is needed is the EDL; the disk files can simply be erased. In on-line editing the disks hold original recordings they will need to be backed up to tape if they will be required again. In large broadcast systems, the edited work can be broadcast directly from the disk file server. In smaller systems it will be necessary to output to some removeable medium, since the Winchester drives in the editor have fixed media.

9. Does EMC affect me?

9.1 EMC stands for electromagnetic compatibility. A piece of electrically powered equipment or a system is considered electromagnetically compatible if it does not cause interference with other equipment and if it does not suffer from interference due to other equipment. The growth of electronics and the marginal performance of lots of available equipment has been such that problems have been occurring and legislation has had to be introduced to enforce standards. This comes into force in Europe 1996 but it is only a matter of time before something similar appears in the USA.

Today electronic equipment is everywhere and is used in applications where a failure can cause loss of life or serious disruption. There are also more potential sources of interference.

9.2 Another effect which has gradually grown is the distortion of the ac supply by electronic loads. The ideal load for an ac power system is resistive; the current is in phase with the voltage and proportional to it. Unfortunately much electronic equipment behaves in a manner far from ideal. Many power supplies contain transformers to step down the line voltage to that needed by the circuitry. A lightly loaded transformer is almost purely inductive. The current is nearly in quadrature to the voltage so that appreciable current flows even though little power is delivered. The measure of this phenomenon is called the power factor. The power lost in the distribution network is proportional to current, so equipment with an adverse power factor is in practice less efficient because a larger proportion of the power it uses is wasted in transmission. The growing use of electronic equipment means that power stations are seeing increasingly inductive loading and suffering more transmission loss than ever. This wasted power translates directly into pollution.

A simple electronic power supply contains a bridge rectifier to obtain dc from the ac line. Whilst this works as far as the equipment is concerned, it is not too good for the power transmission system as it sees a load which varies throughout the cycle. The diodes only conduct when the instantaneous voltage of the ac input exceeds the voltage of the reservoir capacitor. Thus load current is only drawn at the peaks of the ac cycle.

9.3 Electronic equipment now forms a significant fraction of the load seen by power stations. In addition to the adverse power factor they are now seeing waveform distortion where the peaks are pulled down by rectifier conduction. The waveform distortion causes harmonic generation, resulting in increased radiation and losses. Consequently another aspect of the impending EMC regulations is to improve the load behaviour of electrical equipment.

Although electromagnetic compatibility is common sense, it also adds to the cost of equipment. Improving the power factor costs money. Preventing the emission of unwanted radiation requires extra suppression components. Reducing the sensitivity of equipment to interference needs extra screening and filtering.

There has been a great deal of opposition to the EMC regulations, but they are necessary, timely and beneficial to society. Legislation is actually quite a fair way of avoiding problems. If all manufacturers have to comply, they all have the same increased costs and thus remain equally competetive.

9.4 The implications for professional audio and video equipment manufacturers are not significantly worse than elsewhere, although the smaller sales volume in professional equipment will result in a higher proportion of compliance testing costs to be recouped on each unit.

Preventing emission or unwanted pickup by electronic circuitry is not too difficult as radio frequency interference (RFI) is stopped by sheet metal. The ideal totally steel encased device is impracticable because this conflicts with other requirements such as power input, maintenance access, cooling airflow and signal inputs and outputs. Access doors require flexible metal “fingers” to bridge the joints and effectively prevent RFI leaking through the cracks. Airflow is arranged to pass through a metal maze.

Inputs and outputs are more difficult because cables can act as antennas and bring RF into the equipment from outside or radiate internally generated RF outside. The common solution is to filter and/or decouple all terminals of a connector using a small parallel capacitance and a series inductance such as a ferrite bead. When these are used, the impedance of the wiring into the equipment rises and the parallel capacitor appears as a short circuit to chassis at frequencies above those used by the wanted signal. This still allows differential video inputs using floating BNC connectors.

Screened cables are complemented by metal bodied connectors which are electrically connected to the screen and which mate with their sockets over their whole circumference to prevent RF leakage in or out.

The BNC connector is no problem as it was designed from the outset as an RF device and can only be wired in this way. Not so the XLR audio connector which was designed way back primarily for durability. The XLR connection standard uses one pin for the cable screen, and the body may be plastic on more recent examples, effectively holding an open house for RF. Analog audio equipment seldom contributes to the generation of RF, but it is difficult to see how its susceptibility to RF can be managed when such unsuitable connectors are used.

10. Practical ways to go digital

If you start from nothing, going digital is quite easy. As long as the system is sensibly designed and installed with attention to EMC and data integrity, there should not be any problems over and above the usual hassle of building a system. In fact digital could be quicker to install because there is less setting up to do.

However, most people have an existing facility which is unlikely to be uniformly worn out. Typically there will be a collection of gear of different ages with different amounts of economic life left. It makes no sense to scrap perfectly good equipment for the hell of it.

Starting from that assumption, we suggest here some existing scenarios and ways in which digital technology may be introduced in a less traumatic manner.

1. No matter what your existing recording/editing equipment is, it can be made more efficient by putting one or more offline non-linear editors alongside it. If you are currently editing on analog composite or component tape, a lot of head wear takes place simply jogging around edit points. With a non-linear system, the tape system only has to conform to the EDL and this takes a lot less time and causes less wear. So you have the choice of doing the same amount of work to the same quality, and extending the life of your VTRs, or increasing your throughput. This approach has the advantage that the existing system remains intact to keep you going if you have teething problems. If you do a lot of news work, consider an on-line non-linear editor.

2. Suppose you have a large analog component system with VTRs, an analog router, an analog switcher, and various 4:2:2 DVEs, paint systems etc. Chances are the VTRs still have some life in them but you spend a lot of time and effort adjusting the component analog signals to get the Y/C timing and levels right. It's not difficult to upgrade analog component recorders so they have serial digital inputs and outputs. Consider replacing the router with an SDI router which is virtually maintenance free. Any 4:2:2 equipment can be connected to SDI with a simple "glue" adaptor. Keeping the analog switcher would need

lots of ADCs and DACs, so it would probably be better to have a 4:2:2 switcher as well. Remember that some digital switchers offer integral routeing or can control an external router instead of having an internal one. Either approach could save you money. Going this route allows you simply to upgrade to 4:2:2 DVTRs in the future.

3. You replaced your analog composite VTRs with D-2 or D-3 recently, but retained the analog routing and switchers. You will get better quality with less maintenance by replacing the router with an SDI device. Adapting D-2 or D-3 to composite SDI is easy; the internal convertors are bypassed. Buy a composite digital switcher, or keep the analog one and fit it with convertors. Although this sounds self defeating, it isn't. Effectively the analog switcher gets an input from a local DAC, and all routeing is done with loss free SDI. The DAC in the DVTR is bypassed, so there is still only one DAC in the line, but cable losses and analog router maintenance have been eliminated. If you intend to go component digital at some time later, the SDI router is a good investment because if you buy the right model it will also work on 4:2:2 SDI.

4. You recently upgraded your composite installation to D-2 or D-3 VTRs and SDI routeing, but the amount of post business is increasing and you keep getting requests for component output. Consider using the COM^3 encoding system with your existing hardware. This system capitalises on the fact that D-2 and D-3 VTRs have 4FSc sampling which gives them more bandwidth than NTSC needs. COM^3 encoders take conventional component signals and produce an NTSC-like signal which actually uses the whole of the available bandwidth. The signal is compatible with composite analog switchers and monitors, so you can post produce as normal, but when the result is decoded to component in a COM^3 decoder, the result is far better than an NTSC decode. The chroma bandwidth is higher, and there are less cross-products. As a result you get near 4:2:2 performance but using composite DVTRs. If the output you produce from COM^3 is going to be compressed, no-one would know it hasn't come from 4:2:2, whereas residual cross products from regular NTSC are bad news in compression.

5. You have a composite analog system with a few digital devices like DVEs, but you want to go component digital. Converting every composite device to 4:2:2 with encoders and decoders would be financial suicide. The solution is to use the approach shown in Fig.10.1. Here a 4:2:2 SDI router is installed alongside the existing composite router. The two routers talk via gateways which contain decoding, encoding and conversion. As there are only a few of these,

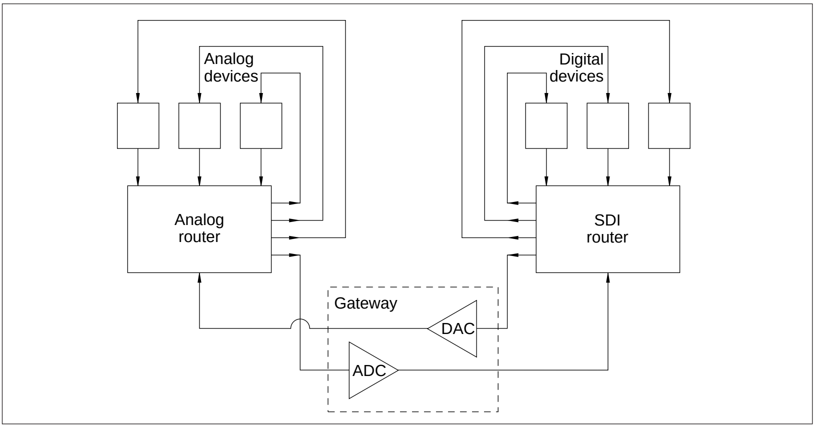


Fig 10.1 Using a gateway, a new component SDI router can exist alongside an existing analog component or composite router. This is much more cost effective than fitting all analog equipment with convertors (and possibly composite encoders/decoders)

they can be top grade devices. All existing digital hardware is adapted to SDI and goes on the digital router. All existing analog gear stays on the old router. Any time you retire a piece of analog gear, it is disconnected from the analog router, but its replacement will be SDI and goes on the digital router. Thus over a period of time, you can transition to 4:2:2 as funds allow. You might want to retain the old analog device in circuit until you're sure the new piece of digital hardware is settled in.

11. Systems of the future

11.1 Future digital systems will rely on storage just as much as today's. Right now, no single digital recording technology has dominated the others because no single one is best in all circumstances. Digital recording only requires some parameter to be stored in one of two states and takes place on a wide variety of media including RAM, magnetic and optical disk, and tape. In computer land, media have primarily been compared on three factors. The access time, the cost per bit and the transfer rate. Subsidiary considerations include exchangeability, reliability, and reaction to power loss.

11.2 RAM has the fastest access time because it has no moving parts (except for electrical charge). Magnetic disks come next because the whole recording area is exposed to a two-dimensional access mechanism (rotation and radial address). Optical disks have the same access principle, but the pickup is heavier and slower. Tape comes last in this race because it has to be shuttled to expose the wanted area to the pickup.

However, compare the cost per bit and the picture is different. Here magnetic tape is supreme because it is such a simple medium to manufacture. Rotary head tape comes top because it currently offers higher recording density than stationary heads allow. Magnetic disk drives need an air film between the disk surface and the head to eliminate wear so they can stay on line for years at a time. This causes a spacing loss and limits the recording density. Also the precision metal disk substrate costs more to make than plastic film. These factors push the cost per bit up. Optical disk are also expensive to make because of the complex construction. Most expensive is RAM which is extremely intricate, with every bit having its own wiring inside a chip.

The best medium on one scale is the worst on the other! Thus there is no overall best storage technology, and this will continue to be true in the future, because improvements will occur to all media in parallel until physical limits are reached. Therefore don't expect to find broadcast quality camcorders using hard disks for a long while yet. By the time they are ready, DVC and son of Digital Betacam will be out there. ENG cameras using disks will require high compression factors to get enough playing time and the cost of the media will be something else. The sheer low cost of tape means that it will not go away readily, and will remain indefinitely for purposes such as archiving.

11.3 In the past the high data rate of digital video was a headache but the inevitable increase in performance of all media will mean that in the future the actual recording step becomes easy and we will compare equipment using other criteria. One of these will be the transfer rate. We have become accustomed to the limitations of analog production equipment, where real time operation was the norm. In analog video production, all dubbing was done at normal speed so that it could be monitored. With read after write and error correction, digital media can transfer data reliably without human intervention; they can be designed to monitor themselves better than we can! There is no longer a constraint to use real time transfer, and when time costs money, the best recorder may be the one which dubs fastest, as it is in computer land. In addition to media which can operate at high speed, there will also be a need for an interface standard for high speed transfer between units.

11.4 Another consequence of increased storage capacity is that compression will no longer appear so attractive for recording. It may be just a phase we are going through. One justification for compression is that it helps in faster than real time transfer. If for the sake of argument 4:1 compression is used, the data rate is 1/4 the original. If the original data rate is maintained, the material can be transferred at 4 X real time.

11.5 The relative merits of different storage media will not change greatly in the future, so current computer land solutions will still be applicable. For a long time, computers have combined storage media in real applications to extract the best of each. Fig.11.1 shows how this approach can be applied to solve video production problems. The computer processor is replaced or supplanted by a DSP device (a computer optimised for signal processing

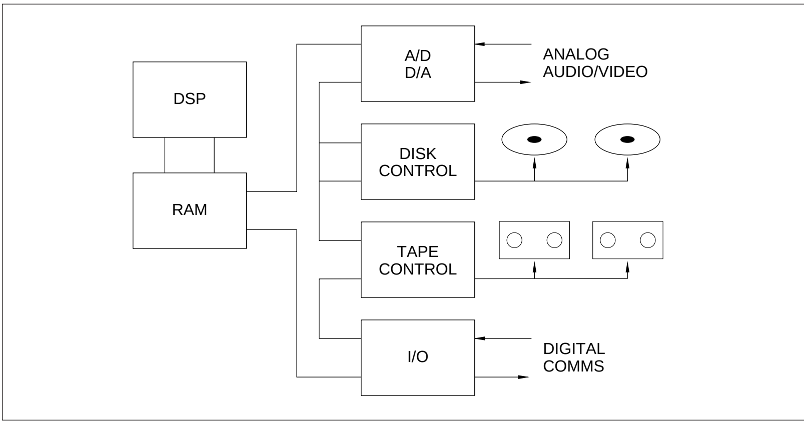


Fig 11.1 A hybrid approach to digital production employs all storage media as required rather than being committed to only one type.

rather than general calculation), but the usual computer arrangement of RAM, disk and tape is retained. The communications ports are replaced by digital video (and audio) interfaces. The disk drive here would use Winchester technology because it does not need to be removeable as there is a tape cassette for that purpose. The disk might well use parallel transfer where each head has its own circuitry and all can move data in parallel where required. The tape deck might use stationary heads using thin film technology and narrow tracks requiring a tracking servo. The adoption of stationary heads is designed to allow operation at several speeds. Alternatively a rotary head transport may be used which has a single high transfer rate, but which is buffered by RAM and works intermittently if a lower rate is required. The tape deck does not need to be able to edit to

field accuracy because it only writes large data blocks which have been edited on disk or in RAM. Taking away the edit requirement puts up the recording density.

Such a general purpose system is extremely flexible, but one example of its use will be given here. Consider that a video production is to be made using a number of sketches where several takes will be made of each one.

Subsequently the best sections of each take are combined in a final edited program. During each take, audio and video from the input convertors is recorded in full bandwidth on the tape cassette in the second of two partitions. At the same time the video and audio are compressed in the DSP, and recorded on the hard disk. At the end of each take the compressed file from the disk is transferred to the first tape partition along with a complete record of the control panel setup. At the end of the recording session the tape contains a full bandwidth version of everything, but at the beginning of the tape is a browsing file which contains a compressed version. The tape can be taken away and edited in a different machine, or brought back to this machine. Upon installing the cassette, the compressed file is transferred to the disk, and the console setup is re-loaded. If 4:1 compression is used, and the tape plays a 10 X speed, this transfer occurs at 40 X speed. However, this process need not be completed before editing begins, because the disk controller supports multiple access and the user can see and hear the beginning of the recording before the end is transferred.

Editing is now performed using the compressed disk files, but only for the purpose of creating an edit decision list. When the final editing is finished, the user can leave the machine and go fishing. In his absence, the EDL is executed on the full bandwidth recording on the cassette and stored on disk. On his return, the user can play the full bandwidth recording from the disk, and check that everything is as expected. After any last minute changes, the disk contents can be streamed back to a new cassette which becomes a transmission master. Such a device could be assembled today from existing components.

11.6 Interlace is a form of compression because it halves bandwidth by sending only half of the picture lines in any one field. As a result it can handle high resolution in the absence of motion and it can handle motion in the absence of high resolution. Put them together and it dies. Recalling what we said above about not connecting compression systems in tandem, it should be clear that if you are going to use digital compression, interlaced video is a bad place to start from. The resultant interlace artifacts confuse the hell out of the compressor, more so if it is motion compensated. We also said above that the input to a compressor has to be squeaky clean, and interlace prevents us fully reaching that.

In fact, for the same quality, a compressor will achieve a lower bit rate when it starts with progressive scan than if it starts with interlace, because it does not have to second guess what the interlace missed out. Thus in the future when compression becomes more widespread, interlace may fade away. There will be further pressure for it to do so from the computer community who have never used it.

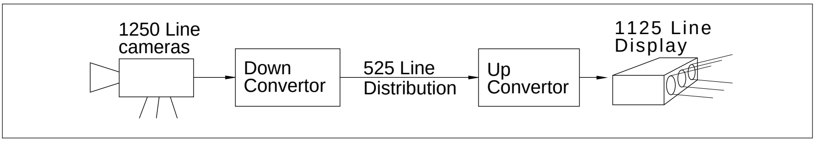


Fig 11.2 A future transmission system could advantageously use different line structures for capture, transmission and display. Here, downconverting for transmission and upconverting for display gives the same subjective results as HDTV.

Spin-offs from standards conversion technology include down convertors from High-Def. to Standard Def. and up convertors to return. Down converted SD pictures look better than pictures from an SD camera because using more lines and down converting is a form of oversampling. The same is true at the display. Fig.11.2 shows a system with oversampling camera and display, but using an SD signal in between. Because the aperture effects of both camera and display are eliminated, the result looks far better than SDTV, and subjectively as good as HD. Future transmission standards need to consider that the optimum number of picture lines is different for capture, transmission and display.

12.Further reading

The following reading list builds on the introduction given here, but covers all the necessary theory and practice of digital production.

From the Snell and Wilcox Handbook series:

The Engineer's Guide to Decoding and Encoding - essential reading from those faced with the problem of interfacing today's component equipment with composite signals.

The Engineer's Guide to Standards Conversion - shows that standards convertors are not all alike and how to compare them.

The Engineer's Guide to Motion Compensation - companion to the Standards Conversion handbook which shows how motion compensation transforms standards conversion.

From Focal Press:

Introduction to Digital Audio

Introduction to Digital Video

The Art of Digital Audio

The Art of Digital Video

The Digital Video Tape Recorder

The Digital Interface Handbook

Audio and Video Compression

13.Glossary

Active line: Lines of the TV frame carrying picture information; all frame lines except the vertical blanking interval.

Aliasing: Generation of incorrect samples due to input frequencies exceeding one-half the sampling rate.

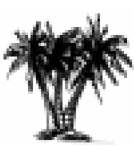
Anamorphic format: Viewed picture format with geometric deformation of the widescreen picture aimed to achieve full vertical screen occupation while using the conventional TV display.

Anti-aliasing filter: Filter which restricts the frequency range of an analog signal to less than one half the sampling rate.

Azimuth recording: Twisting alternate magnetic heads left and right to record adjacent tracks which can be read without crosstalk. Also called guard-band-less recording.

Blanking: Process of periodical setting of video signal values to some predefined value during some predifined tome intervals; usually to the black level during fly-back.

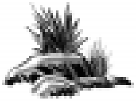
Buzzword: A specialist term which performs two functions:
a) to those who understand its meaning it makes technical conversations briefer.
b) to those who do not understand its meaning it is a way of preventing communication or of being put down.



CCIR Rec 656, CCIR-656: CCIR Recommendation 656 “Interfaces for Digital Component Video Signals in 525-line and 625 line television system”. A companion document to CCIR-601 which specifies the signal format to be used and the particular characteristic of both serial and parallel digital interfaces.

Colour difference signal 601, CCIR Rec. 601, Rec. 601, CCIR-601, ITU (CCIR)-601: Set of digitisation and interface formats, first given in the CCIR Recommendation 601 and based on the idea of common sampling rate for both 625/50 and 525/59.94 scanning standards. To describe the family of sampling rates special notation was introduced, where 3.375 MHz frequency is used as unit of measurement: e.g. '4:2:2' means that luminance signal Y is sampled at 13.5 MHz and both P1 and Pb are sampled at 6.75 MHz. Although it has not formally approven the up-dated version of ITU (CCIR) Rec. 601 will allow the users to operate with 360 Mbit/s (18 MHz sampling rate) or 270 Mbit/s (13.5 MHz sampling rate) as they choose. Hence the 10-years old Rec. 601 reterring to 4:3 operation becomes 601 Part A, and the new wide-screen modification will be known as Rec. 601 Part B.

Color framing: A process which synchronizes the subcarrier and sync timing in composite VTRs to allow edits without jumps or breakup on replay.



Compression: Also called bit-rate reduction or data reduction. Process which allows pictures to be represented with less data. Picture quality may suffer if used to excess.

Concealment: Process used when error correction is not possible. Interpolates or estimates missing pixel values from those nearby.

Contribution A form of video signal transmission where the destination is not the ultimate viewer, and where some processing (e.g. in a vision mixer) is expected before the signal reaches the ultimate viewer.

Contribution quality: Describes a signal which is going to be further post-produced and so must have high quality.

Cr, Cb: Digital color difference signals, e.g. signals conforming to CCIR Rec 601.

Data integrity: General term for any action or strategy which minimizes the proportion of data bits in a system which are corrupted.

Digital active lines: Same as analogue active lines plus few lines above and below to prevent distortions caused by digital filtering.

Distribution: Sharing data between two or more heads in a DVTR allows concealment if one head clogs.

Edit gap: In a DVTR, a space left on the recorded track allowing data on one side of the space to be edited without corrupting data on the other side.

E-E: Electronics to electronics; a mode in a VTR where the tape and heads are bypassed but the signal passes through everything else.

EMC: Electromagnetic compatibility. General term for set of regulations and procedures to ensure that electronic equipment neither suffers from nor generates interference.

Entropy: The unpredictable part of a signal which has to transmitted by a compression system if quality is not to be lost.

Four-field sequence: A repetition rate which occurs in NTSC due to the subcarrier frequency having a half line offset. The subcarrier can only return to a given phase after an even number of lines and this requires two frames or four fields.

Horizontal editing: Edit process which primarily involves the time axis. e.g. cuts. Contrasts with vertical editing.

Jitter: Instantaneous timing errors in the position of signal changes or transitions. See phase-locked loop.

Motion Vector: Parameter sent by compression system which tells the decoder how to shift a previous picture so it more nearly resembles the current picture.

Non-linear: An editing system in which random access storage is used so that non-linear access to the material is possible.

NTSC: Never Twice the Same Color. A television system in which the colors are a function of where the hue control was left by an unskilled viewer.

Nyquist frequency: Nyquist Rate, Nyquist Limit. Terms used in connection with sampling. The Nyquist frequency is considered the minimum sampling frequency for correct digital reproduction of a signal. Because of the engineering reasons, sampling of video signals is often performed at the rate higher than its Nyquist rate.

Off-line: System where low quality images are used for decision making purposes. Low quality is not seen by end viewer.

On-line: System where the quality seen by the operator is the same as that seen by the end viewer.

Oversampling: Temporary use of a higher than necessary sampling rate in convertors in order to simplify analog filters.

Pel: See pixel.

Phase-locked loop: An electronic circuit which extracts the average phase from a jittery signal in a manner analogous to a flywheel. See reclocker.

Pixel: Short for picture cell. A point sample of a picture. Also called a pel. See also square pixel.

Power Factor: In electrical supplies, power factor measures the efficiency of transmission. If current is out of phase with voltage, the power factor is poor and transmission losses increase.

Product code: Error correction strategy where pixels are formed into a rectangular array with check words calculated on both rows and columns.

Quantizer: A device which breaks an analog signal's voltage range into even intervals and outputs the number of the interval in which the analog input lies.

Random access: Storage device like a disk where contents can be output in any order. Contrasts with serial access.

Reclocker: A combination of a slicer and a phase-locked loop which can remove noise and jitter from a digital signal.

Reconstruction: Filtering process which converts a series of samples back to a continuous waveform.

Redundancy: a) in error correction, extra check bits appended to the wanted data.
b) in compression, that part of a signal which can be predicted and so need not be sent.

Reed-Solomon code: Powerful error correcting codes named after their inventors.

RFI: Radio frequency interference. Interference which is radiated rather than conducted.

Sampling: A process in which some continuous variable is measured at discrete (usually uniform) intervals.

Segmentation: In VTRs the use of several parallel tracks to record one field.

Serial access: Storage system such as tape where data comes out in a fixed sequence. Contrasts with random access.

Shuffling: Random pixel reordering process used in DVTRs which spreads uncorrected pixels over a large area to aid concealment.

Signal to noise ratio, SNR, S/N: Measure of relative amplitude of random noise in a signal: usually expressed in dB with reference to the nominal video signal level.

Slicer: Electronic circuit which judges an input to be above or below a threshold. Used to clean up binary signals. See reclocker.

SMPTE-125: SMPTE document equivalent to CCIR Rec 656.

Spatial frequency: Value inverse to the period of the pattern in the TV picture.

Square pixel: An image sampling process in which the vertical and horizontal sampling spacing is the same.

Time compression: Process used to squeeze the time taken to send a given quantity of data by raising the data rate.

Vertical editing: Editing where manipulations take place within the picture. e.g. layering, chroma key. Contrasts with horizontal editing.

Weighted noise: Noise put through special weighting filter to take into account the subjective visibility of different frequency components.

Weighted SNR: Ratio of the nominal signal amplitude to the weighed noise mean square value expressed in dB: usually is 10-16 dB higher than unweighted SNR.

Winchester disk: Disk drive having heads and disks sealed in one unit allowing higher speed and capacity.

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