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SERIES

The Engineer's Guide to

Compression

By John Watkinson

The Engineer's Guide to Compression

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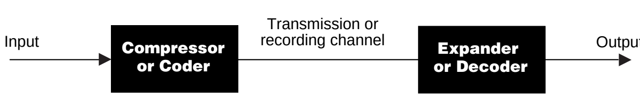
Section 1 - Introduction to Compression

In this section we discuss the fundamental characteristics of compression and see what we can and cannot expect without going into details which come later.

1.1 What is compression?

Normally all audio and video program material is limited in its quality by the capacity of the channel it has to pass through. In the case of analog signals, the bandwidth and the signal to noise ratio limit the channel. In the case of digital signals the limitation is the sampling rate and the sample wordlength, which when multiplied together give the bit rate. Compression is a technique which tries to produce a signal which is better than the channel it has passed through would normally allow. Fig.1.1.1 shows that in all compression schemes a compressor, or coder, is required at the transmitting end and an expander or decoder is required at the receiving end of the channel. The combination of a coder and a decoder is called a codec.

Figure 1.1.1



There are two ways in which compression can be used. Firstly, we can improve the quality of an existing channel. An example is the Dolby system; codecs which improve the quality of analog audio tape recorders. Secondly we can maintain the same quality as usual but use an inferior channel which will be cheaper.

Bear in mind that the word compression has a double meaning. In audio, compression can also mean the deliberate reduction of the dynamic range of a signal, often for radio broadcast purposes. Such compression is single ended; there is no intention of a subsequent decoding stage and consequently the results are audible.

We are not concerned here with analog compression schemes or single ended compressors. We will be dealing with digital codecs which accept and output digital audio and video signals at the source bit rate and pass them through a channel having a lower bit rate. The ratio between the source and channel bit rates is called the compression factor.

1.2 Applications

For a given quality, compression lowers the bit rate, hence the alternative term of bit-rate reduction (BRR). In broadcasting, the reduced bit rate requires less bandwidth or less transmitter power or both, giving an economy. With increasing pressure on the radio spectrum from other mobile applications such as telephones developments such as DAB (digital audio broadcasting) and DVB (digital video broadcasting) will not be viable without compression. In cable communications, the reduced bit rate lowers the cost.

In recording, the use of compression reduces the amount of storage medium required in direct proportion to the compression factor. For archiving, this reduces the cost of the library. For ENG (electronic news gathering) compression reduces the size and weight of the recorder. In disk based editors and servers for video on demand (VOD) the current high cost of disk storage is offset by compression. In some tape storage formats, advantage is taken of the reduced data rate to relax some of the mechanical tolerances. Using wider tracks and longer wavelengths means that the recorder can function in adverse environments or with reduced maintenance.

1.3 How does compression work?

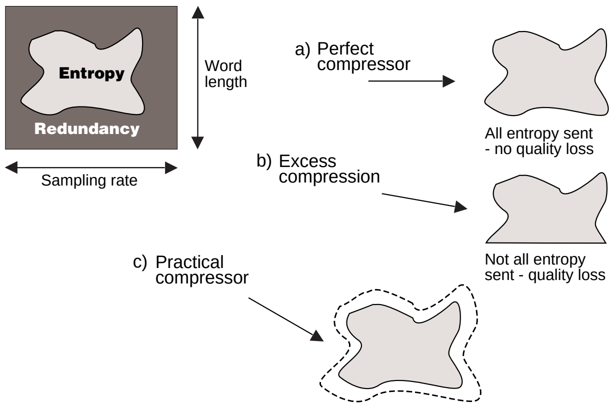
In all conventional digital audio and video systems the sampling rate, the wordlength and the bit rate are all fixed. Whilst this bit rate puts an upper limit on the information rate, most real program material does not reach that limit. As Shannon said, any signal which is predictable contains no information. Take the case of a sinewave: one cycle looks the same as the next and so a sinewave contains no information. This is consistent with the fact that it has no bandwidth. In video, the presence of recognisable objects in the picture results in sets of pixels with similar values. These have spatial frequencies far below the maximum the system can handle. In the case of a test card, every frame is the same and again there is no information flow once the first frame has been sent.

The goal of a compressor is to identify and send on the useful part of the input signal which is known as the entropy. The remaining part of the input signal is called the redundancy. It is redundant because it can be predicted from what the decoder has already been sent.

Some caution is required when using compression because redundancy can be useful to reconstruct parts of the signal which are lost due to transmission errors. Clearly if redundancy has been removed in a compressor the resulting signal will be less resistant to errors. unless a suitable protection scheme is applied.

Fig.1.3.1a) shows that if a codec sends all of the entropy in the input signal and it is received without error, the result will be indistinguishable from the original. However, if some of the entropy is lost, the decoded signal will be impaired in comparison with the original. One important consequence is that you can't just keep turning up the compression factor. Once the redundancy has been eliminated, any further increase in compression damages the information as Fig.1.3.1b) shows. So it's not possible to say whether compression is a good or a bad thing. The question has to be qualified: how much compression on what kind of material and for what audience?

Figure 1.3.1



As the entropy is a function of the input signal, the bit rate out of an ideal compressor will vary. It is not always possible or convenient to have a variable bit rate channel, so many compressors have a buffer memory at each end of a fixed bit rate channel. This averages out the data flow, but causes more delay. For applications such as video-conferencing the delay is unacceptable and so fixed bit rate compression is used to aviod the need for a buffer.

So far we have only considered an ideal compressor which can perfectly sort the entropy from the redundancy. Unfortunately such a compressor would have infinite complexity and have an infinite processing delay. In practice we have to use real, affordable compressors which must fail to be ideal by some margin. As a result the compression factors we can use have to be reduced because if the compressor can't decide whether a signal is entropy or not it has to be sent just in case. As Fig.1.3.1c) shows, the entropy is surrounded by a "grey area" which may or may not be entropy. The simpler and cheaper the compressor, and the shorter its encoding delay, the larger this grey area becomes. However, the decoder must be able to handle all of these cases equally well. Consequently compression schemes are designed so that all of the decisions are taken at the coder. The decoder then makes the best of whatever it receives. Thus the actual bit rate sent is determined at the coder and the decoder needs no adjustment.

Clearly, then, there is no such thing as the perfect compressor. For the ultimate in low bit rates, a complex and therefore expensive compressor is needed. When using a higher bit rate a simpler compressor would do. Thus a range of compressors is required in real life. Consequently MPEG is not a standard for a compressor, nor is it a standard for a range of compressors.

MPEG is a set of standards describing a range of bitstreams which compliant decoders must be able to handle. MPEG does not specify how these bitstreams are to be created. There are a number of advantages to this approach. A wide variety of compressors, some using proprietary techniques, can produce bitstreams compatible with any compliant decoder. There can be a range of compressors at different points on the price/performance scale. There can be competition between vendors. Research may reveal better ways of encoding the bit stream, producing improved quality without making the decoders obsolete.

When testing an MPEG codec, it must be tested in two ways. Firstly it must be compliant. This is a yes/no test. Secondly the picture and/or sound quality must be assessed. This is much more difficult task because it is subjective.

1.4 Types of compression

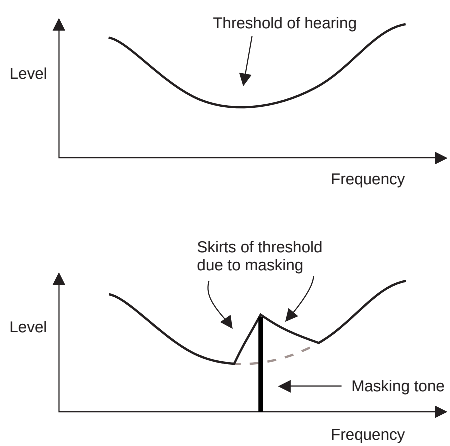
Compression techniques exist which treat the input as an arbitrary data stream and compress by identifying frequent bit patterns. These codecs can be bit accurate; in other words the decoded data are bit-for-bit identical with the original. Such coders, called lossless coders, are essential for compressing computer data and are used in so called 'stacker' programs which increase the capacity of disk drives. However, stackers can only achieve a limited compression factor and are not appropriate for audio and video where bit accuracy is not essential.

In audio and video, the human viewer or listener will be unable to detect certain small discrepancies in the signal due to the codec. However, the admission of these small discrepancies allows a great increase in the compression factor which can be achieved. Such codecs can be called near-lossless. Although they are not bit accurate, they are sufficiently accurate that humans would not know the difference. The trick is to create coding errors which are of a type which we perceive least. Consequently the coder must understand the human sensory system so that it knows what it can get away with. Such a technique is called perceptual coding. The higher the compression factor the more accurately the coder needs to mimic human perception.

1.5 Audio compression principles

Audio compression relies on an understanding of the hearing mechanism and so is a form of perceptual coding. The ear is only able to extract a certain proportion of the information in a given sound. This could be called the perceptual entropy, and all additional sound is redundant. The basilar membrane in the ear behaves as a kind of spectrum analyser; the part of the basilar membrane which resonates as a result of an applied sound is a function of frequency. The high frequencies are detected at the end of the membrane nearest to the eardrum and the low frequencies are detected at the opposite end. The ear analyses with frequency bands, known as critical bands, about 100 Hz wide below 500 Hz and from one-sixth to one-third of an octave wide, proportional to frequency, above this. The ear fails to register energy in some bands when there is more energy in a nearby band. The vibration of the membrane in sympathy with a single frequency cannot be localised to an infinitely small area, and nearby areas are forced to vibrate at the same frequency with an amplitude that decreases with distance. Other frequencies are excluded unless the amplitude is high enough to dominate the local vibration of the membrane. Thus the membrane has an effective Q factor which is responsible for the phenomenon of auditory masking, in other words the decreased audibility of one sound in the presence of another. The threshold of hearing is raised in the vicinity of the input frequency. As shown in Fig.1.5.1, above the masking frequency, masking is more pronounced, and its extent increases with acoustic level. Below the masking frequency, the extent of masking drops sharply.

Figure 1.5.1



Because of the resonant nature of the membrane, it cannot start or stop vibrating rapidly; masking can take place even when the masking tone begins after and ceases before the masked sound. This is referred to as forward and backward masking.

Audio compressors work by raising the noise floor at frequencies where the noise will be masked. A detailed model of the masking properties of the ear is essential to their design. The greater the compression factor required, the more precise the model must be. If the masking model is inaccurate, or not properly implemented, equipment may produce audible artifacts. There are many different techniques used in audio compression and these will often be combined in a particular system.

Predictive coding uses circuitry which uses a knowledge of previous samples to predict the value of the next. It is then only necessary to send the difference between the prediction and the actual value. The receiver contains an identical predictor to which the transmitted difference is added to give the original value. Predictive coders have the advantage that they work on the signal waveform in the time domain and need a relatively short signal history to operate. They cause a relatively short delay in the coding and decoding stages.

Sub-band coding splits the audio spectrum up into many different frequency bands to exploit the fact that most bands will contain lower level signals than the loudest one.

In spectral coding, a transform of the waveform is computed periodically. Since the transform of an audio signal changes slowly, it need be sent much less often than audio samples. The receiver performs an inverse transform.

Most practical audio coders use some combination of sub-band or spectral coding. Re-quantizing of sub-band samples or transform coefficients causes increased noise which the coder places at frequencies where it will be masked. Section 4 will treat these ideas in more detail.

If an excessive compression factor is used, the coding noise will exceed the masking threshold and become audible. If a higher bit rate is impossible, better results will be obtained by restricting the audio bandwidth prior to the encoder using a pre-filter. Reducing the bandwidth with a given bit rate allows a better signal to noise ratio in the remaining frequency range. Many commercially available audio coders incorporate such a pre-filter.

1.6 Video compression principles

Video compression relies on two basic assumptions. The first is that human sensitivity to noise in the picture is highly dependent on the frequency of the noise. The second is that even in moving pictures there is a great deal of commonality between one picture and the next. Data can be conserved by raising the noise level where it cannot be detected and by sending only the difference between one picture and the next.

Fig.1.6.1 shows that in a picture, large objects result in low spatial frequencies (few cycles per unit distance) whereas small objects result in high spatial frequencies (many cycles per unit distance). Fig.1.6.2 shows that human vision detects noise at low spatial frequencies much more readily than at high frequencies. The phenomenon of large-area flicker is an example of this.

Figure 1.6.1

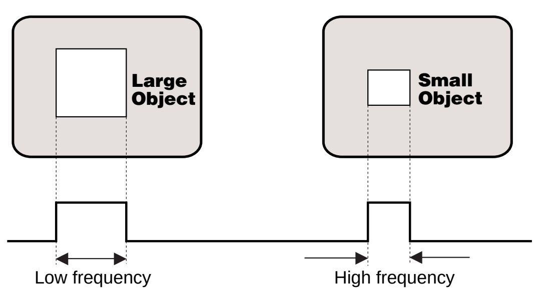
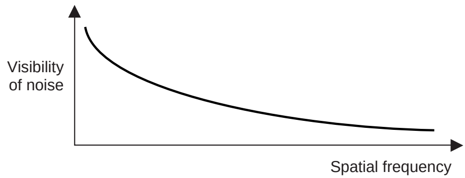
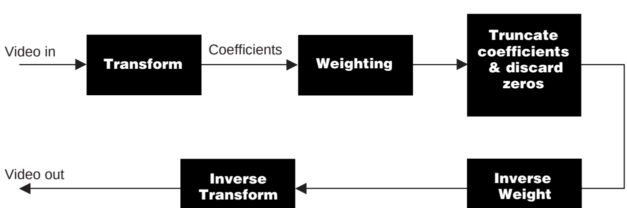


Figure 1.6.2



Compression works by shortening or truncating the wordlength of data words. This reduces their resolution, raising noise. If this noise is to be produced in a way which minimises its visibility, the truncation must vary with spatial frequency. Practical video compressors must perform a spatial frequency analysis on the input, and then truncate each frequency individually in a weighted manner. Such a spatial frequency analysis also reveals that in many areas of the picture, only a few frequencies dominate and the remainder are largely absent. Clearly where a frequency is absent no data need be transmitted at all. Fig.1.6.3 shows a simple compressor working on this principle. The decoder is simply a reversal of the frequency analysis, performing a synthesis or inverse transform process. Section 3 explains how frequency analysis works.

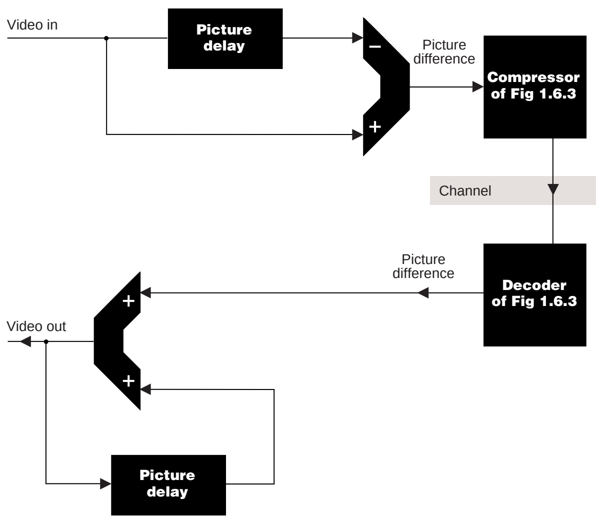
Figure 1.6.3



The simple concept of Fig.1.6.3 treats each picture individually and is known as intra-coding. Compression schemes designed for still images, such as JPEG (Joint Photographic Experts Group) have to work in this way. For moving pictures, exploiting redundancy between pictures, known as inter-coding, gives a higher compression factor.

Fig.1.6.4 shows a simple inter-coder. Starting with an intra-coded picture, the subsequent pictures are described only by the way in which they differ from the one before. The decoder adds the differences to the previous picture to produce the new one. The difference picture is produced by subtracting every pixel in one picture from the same pixel in the next picture. This difference picture is an image in its own right and can be compressed with an intra-coding process of the kind shown in Fig.1.6.3.

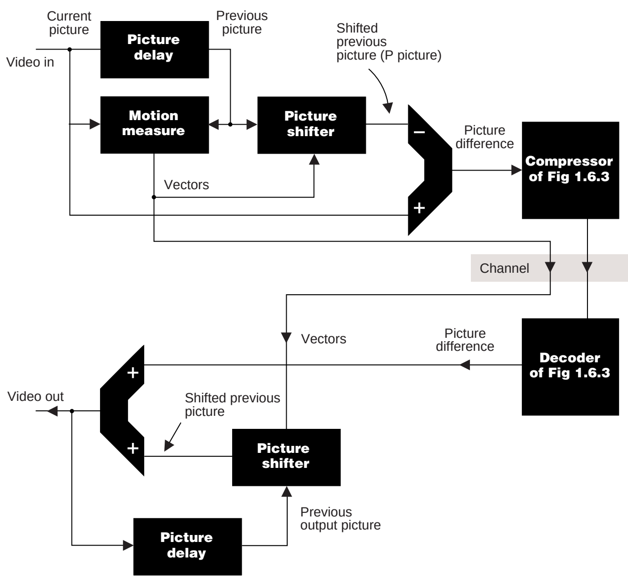
Figure 1.6.4



There are a number of problems with this simple system. If any errors occur in the channel they will be visible in every subsequent picture. It is impossible to decode the signal if it is selected after transmission has started. In practice a complete intra-coded or I picture has to be transmitted periodically so that channel changing and error recovery are possible. Editing inter-coded video is difficult as earlier data are needed to create the current picture. The best that can be done is to cut the compressed data stream just before an I picture.

The simple system of Fig.1.6.4 also falls down where there is significant movement between pictures, as this results in large differences. The solution is to use motion compensation. At the coder, successive pictures are compared and the motion of an area from one picture to the next is measured to produce motion vectors. Fig.1.6.5 shows that the coder attempts to model the object in its new position by shifting pixels from the previous picture using the motion vectors. Any discrepancies in the process are eliminated by comparing the modelled picture with the actual picture.

Figure 1.6.5



The coder sends the motion vectors and the discrepancies. The decoder shifts the previous picture by the vectors and adds the discrepancies to produce the next picture. Motion compensated coding allows a higher compression factor and this outweighs the extra complexity in the coder and the decoder. More will be said on the topic in section 5.

1.7 Dos and don'ts

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 codecs.
4. Compression systems cause delay and make editing more difficult.
5. Compression systems work best with clean source material. Noisy signals, tape hiss, 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. If a very low bit rate is required, reduce the bandwidth of the input signal in a pre-filter.
8. Compression quality can only be assessed subjectively.
9. Don't believe statements comparing video codec performance to "VHS quality" or similar. Compression artifacts are quite different to the artifacts 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.

Section 2 - Digital Audio and Video

In this section we review the formats of digital audio and video signals which will form the input to compressors.

2.1 Digital basics

Digital is just another way of representing an existing audio or video waveform. Fig.2.1.1 shows that in digital audio the analog waveform is represented by evenly spaced samples whose height is described by a whole number, expressed in binary. Digital audio requires a sampling rate between 32 and 48kHz and samples containing between 14 and 20 bits, depending on the quality. Consequently the source data rate may be anywhere from one half to one million bits per second per audio channel.

Figure 2.1.1

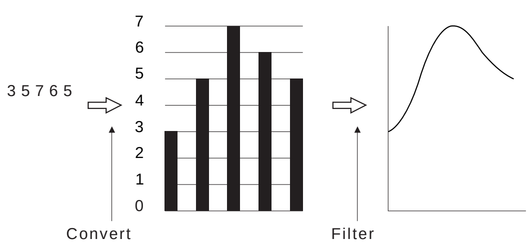
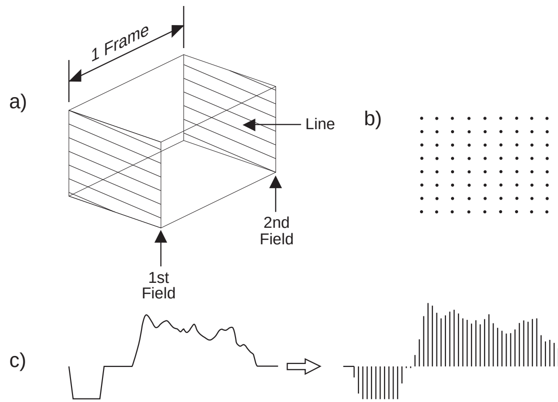


Fig.2.1.2a) shows that a traditional analog video system breaks time up into fields and frames, and then breaks up the fields into lines. These are both sampling processes: representing something continuous by periodic discrete measurements. Digital video simply extends the sampling process to a third dimension so that the video lines are broken up into three dimensional point samples which are called pixels or pels. The origin of

these terms becomes obvious when you try to say “picture cells” in a hurry. Fig.2.1.2b) shows a television frame broken up into pixels. A typical 625/50 frame contains over 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.2.1.2c) 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.

Figure 2.1.2



In the case of component analog video there will be three simultaneous waveforms per channel. Three converters are required to produce three data streams in order to represent GBR or colour difference components. Composite video can be thought of as an analog compression technique as it allows colour in the same bandwidth as monochrome. Whilst digital compression schemes do exist for composite video, these effectively put two compressors in series which is not a good idea. Consequently the compression factor has to be limited in composite systems. MPEG is designed only for component signals and is effectively a modern replacement for composite video which will not be considered further here.

2.2 Sampling

Sampling theory requires a sampling rate of at least twice the bandwidth of the signal to be sampled. In the case of a broadband signal, i.e. one in which there are a large number of octaves, the sampling rate must be at least twice the highest frequency in the input. Fig.2.2.1a) shows what happens when sampling is performed correctly. The original waveform is preserved in the envelope of the samples and can be restored by low-pass filtering.

Figure 2.2.1a

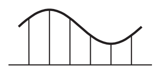
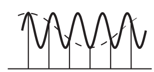


Fig.2.2.1b) shows what happens in the case of a signal whose frequency more than half the sampling rate in use. The envelope of the samples now carries a waveform which is not the original. Whether this matters or not depends upon whether we consider a broadband or a narrow band signal.

Figure 2.2.1b



In the case of a broadband signal, Fig.2.2.1b) shows 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 essential to prevent aliasing in analog to digital converters wherever possible and this is done by including a filter, called an anti-aliasing filter, prior to the sampling stage.

In the case of a narrow-band signal, Fig.2.2.1b) shows a heterodyning process which down converts the narrow frequency band to a baseband which can be faithfully described with a low sampling rate. Re-conversion to analog requires an up-conversion process which uses a band-pass filter rather than a low-pass filter. This technique is used extensively in audio compression where the input signal can be split into a number of sub-bands without increasing the overall sampling rate.

2.3 Interlace

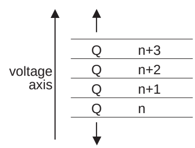
Interlace is a system in which the lines of each frame are divided into odd and even sets known as fields. Sending two fields instead of one frame doubles the apparent refresh rate of the picture without doubling the bandwidth required. Interlace can be considered a form of analog compression. Interlace twitter and poor dynamic resolution are compression artifacts. Ideally, digital compression should be performed on non-interlaced source material as this will give better results for the same bit rate. Using interlaced input places two compressors in series. However, the dominance of interlace in existing television systems means that in practice digital compressors have to accept interlaced source material.

Interlace causes difficulty in motion compensated compression, as motion measurement is complicated by the fact that successive fields do not describe the same points on the picture. Producing a picture difference from one field to another is also complicated by interlace. In compression terminology, the difficulty caused by whether to use the term “field” or “frame” is neatly avoided by using the term “picture”.

2.4 Quantizing

In addition to the sampling process the converter needs a quantizer to convert the analog sample to a binary number. Fig.2.4.1 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 minimised by using enough intervals.

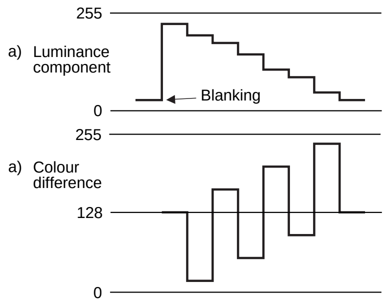
Figure 2.4.1



In an eight-bit video converter 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 converter 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.

Fig.2.4.2 shows how component digital fits into eight- and ten-bit quantizing. Note two things: analog syncs can go off the bottom of the scale because only the active line is used, and the colour difference signals are offset upwards so positive and negative values can be handled by the binary number range.

Figure 2.4.2



In digital audio, the bipolar nature of the signal requires the use of two's complement coding. Fig.2.4.3 shows that in this system the two halves of a pure binary scale have been interchanged. The MSB (most significant bit) specifies the polarity. Fig.2.4.4 shows that to convert back to analog, two processes are needed. Firstly voltages are produced which are 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 on screen and the sound have come through at least two analog filters. In real life a signal may have to be converted 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 will be magnified.

Figure 2.4.3

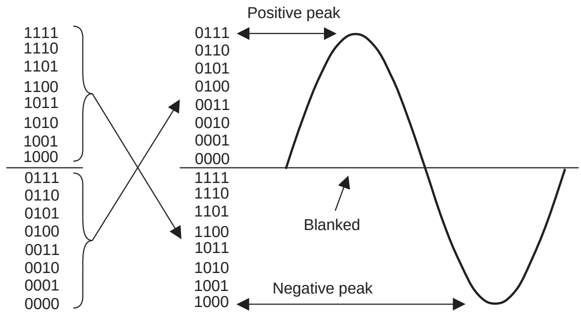
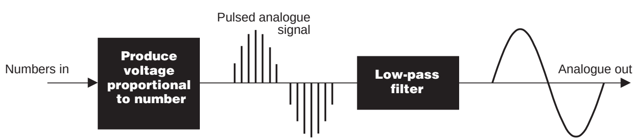


Figure 2.4.4

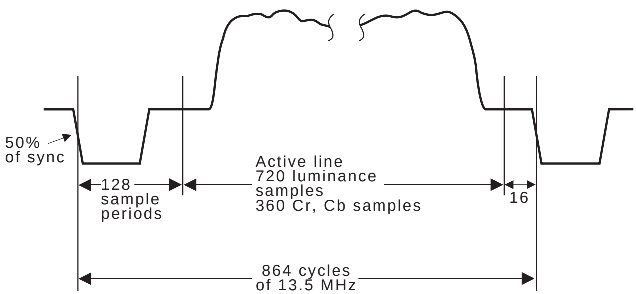


2.5

Digital video

Component signals use a common sampling rate which allows 525/60 and 625/50 video to be sampled at a rate locked to horizontal sync. The figure most often used for luminance is 13.5MHz. Fig.2.5.1 shows how the European standard TV line 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 slightly longer than the analog active line so the sloping analog blanking is always included.

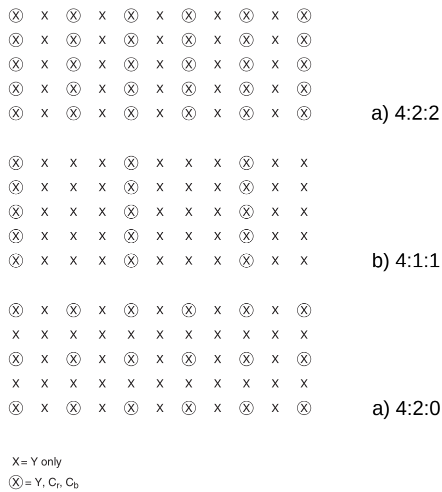
Figure 2.5.1



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.2.5.2a) shows what 4:2:2 sampling looks like in two dimensions. Only luminance is represented at every pixel. Horizontally the colour difference signal values are only specified every second pixel.

Figure 2.5.2

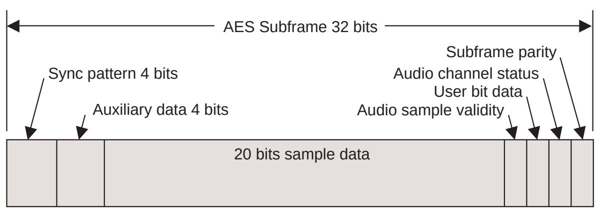


Two other sampling structures will be found in use with compression systems. Fig.2.5.2b) shows 4:1:1, where colour difference is only represented every fourth pixel horizontally. Fig.2.5.2c) shows 4:2:0 sampling where the horizontal colour difference spacing is the same as the vertical spacing giving more nearly "square" chroma. Pre-filtering in this way reduces the input bandwidth and allows a higher compression factor to be used.

2.6 Digital audio

In professional applications, digital audio is transmitted over the AES/EBU interface which can send two audio channels as a multiplex down one cable. Standards exist for balanced working with screen twisted pair cables and for unbalanced working using co-axial cable. A variety of sampling rates and wordlengths can be accommodated. The master bit clock is 64 times the sampling rate in use. In video installations, a video-synchronous 48kHz sampling rate will be used. Different wordlengths are handled by zero-filling the word. Two's complement samples are used, with the MSB sent in the last bit position. Fig.2.6.1 shows the AES/EBU frame structure. Following the sync. pattern, needed for deserializing and demultiplexing, there are four auxiliary bits. The main audio sample of up to 20 bits can be seen in the centre of the sub-frame.

Figure 2.6.1



Section 3 - Compression tools

All compression systems rely on various combinations of basic processes or tools which will be explained in this section.

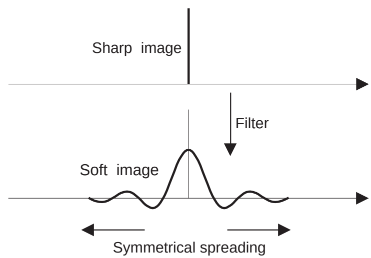
3.1 Digital filters

Digital filters are used extensively in compression. Where high compression factors are used, pre-filtering reduces the bandwidth of the input signal and reduces the sampling rate in proportion. At the decoder, an interpolation process will be required to output the signal at the correct sampling rate again.

To avoid loss of quality, filters used in audio and video must have a linear phase characteristic. This means that all frequencies take the same time to pass through the filter. If a filter acts like a constant delay, at the output there will be a phase shift linearly proportional to frequency, hence the term linear phase. If such filters are not used, the effect is obvious on the screen, as sharp edges of objects become smeared as different frequency components of the edge appear at different times along the line. An alternative way of defining phase linearity is to consider the impulse response rather than the frequency response. Any filter having a symmetrical impulse response will be phase linear. The impulse response of a filter is simply the Fourier transform of the frequency response. If one is known, the other follows from it.

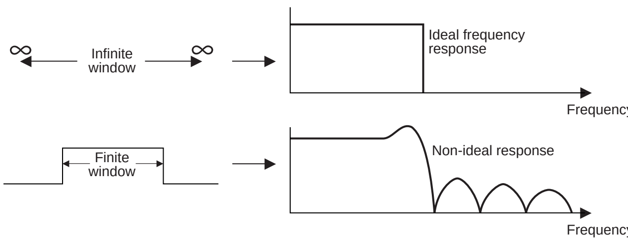
Fig.3.1.1 shows that when a symmetrical impulse response is required in a spatial system, such as a video pre-filter, the output spreads equally in both directions with respect to the input impulse and in theory extends to infinity. However the scanning process turns the spatial image into a temporal signal. If such a signal is to filtered with a phase linear characteristic, the output must begin before the input has arrived, which is clearly impossible. In practice the impulse response is truncated from infinity to some practical time span or window and the filter is arranged to have a fixed delay of half that window so that the correct symmetrical impulse response can be obtained.

Figure 3.1.1



Shortening the impulse from infinity gives rise to the name of Finite Impulse Response (FIR) filter. A real FIR filter is an ideal filter of infinite length in series with a filter which has a rectangular impulse response equal to the size of the window. The windowing causes an aperture effect which results in ripples in the frequency response of the filter. Fig.3.1.2 shows the effect which is known as Gibbs' phenomenon. Instead of simply truncating the impulse response, a variety of window functions may be employed which allow different trade-offs in performance.

Figure 3.1.2



3.2 Pre-filtering

A digital filter simply has to create the correct response to an impulse. In the digital domain, an impulse is one sample of non-zero value in the midst of a series of zero-valued samples. An example of a low-pass filter will be given here. We might use such a filter in a downconversion from 4:2:2 to 4:1:1 video where the horizontal bandwidth of the colour difference signals are halved. Fig.3.2.1a) shows the spectrum of a typical sampled system where the sampling rate is a little more than twice the analog bandwidth. Attempts to halve the sampling rate for downconversion by simply omitting alternate samples, a process known as decimation, will result in aliasing, as shown in b). It is intuitive that omitting every other sample is the same as if the original sampling rate was halved. In any sampling rate conversion system, in order to prevent aliasing, it is necessary to incorporate low-pass filtering into the system where the cut-off frequency reflects the lower of the two sampling rates concerned. Fig.3.2.2 shows an example of a low-pass filter having an ideal rectangular frequency response. The Fourier transform of a rectangle is a sinx/x curve which is the ideal impulse response. The windowing process is omitted for clarity. The sinx/x curve is sampled at the sampling rate in use in order to provide a series of coefficients. The filter delay is broken down into steps of one sample period each by using a shift register. The input impulse is shifted through the register and at each step is multiplied by one of the coefficients. The result is that an output impulse is created whose shape is determined by the coefficients but whose amplitude is proportional to the amplitude of the input impulse. The provision of an adder which has one input for every multiplier output allows the impulse responses of a stream of input samples to be convolved into the output waveform.

Figure 3.2.1

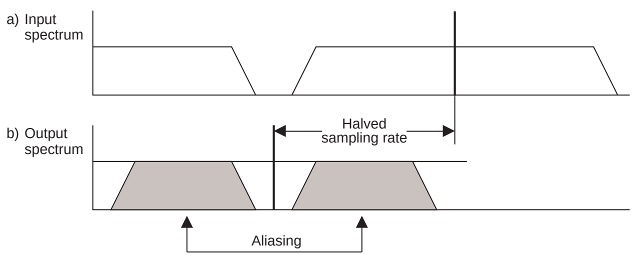
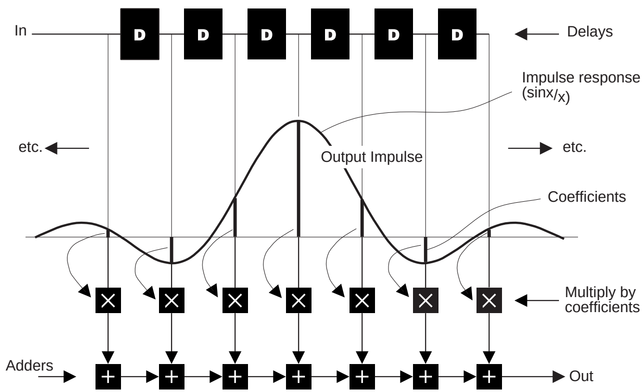


Figure 3.2.2



Once the low pass filtering step is performed, the base bandwidth has been halved, and then half the sampling rate will suffice. Alternate samples can be discarded to achieve this.

There are various ways in which such a filter can be implemented. Hardware may be configured as shown, or in a number of alternative arrangements which give the same results. The filtering process may be performed algorithmically in a processor which is programd to multiply and accumulate. In practice it is not necessary to compute the values of samples which will be discarded. The filter only computes samples which will be retained, consequently only one output computation is made for every two input sample shifts.

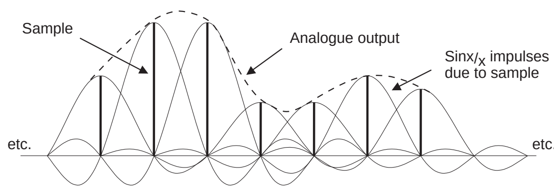
3.3 Upconversion

Following a compression codec in which pre-filtering has been used, it is generally necessary to return the sampling rate to some standard value. For example, 4:1:1 video would need to be upconverted to 4:2:2 format before it could be output as a standard SDI (serial digital interface) signal.

Upconversion requires interpolation. Interpolation is the process of computing the value of a sample or samples which lie off the sampling matrix of the source signal. It is not immediately obvious how interpolation works as the input samples appear to be points with nothing between them. One way of considering interpolation is to treat it as a digital simulation of a digital to analog conversion. According to sampling theory, all sampled systems have finite bandwidth. An individual digital sample value is obtained by sampling the instantaneous voltage of the original analog waveform, and because it has zero duration, it must contain an infinite spectrum. However, such a sample can never be seen or heard in that form because the spectrum of the impulse is limited to half of the sampling rate in a reconstruction or anti-image filter. The impulse response of an ideal filter converts each infinitely short digital sample into a sinx/x pulse whose central peak width is determined by the response of the reconstruction filter, and whose amplitude is proportional to the sample value. This implies that, in reality, one sample value has meaning over a considerable timespan, rather than just at the sample instant. A single pixel has meaning over the two dimensions of a frame and along the time axis. If this were not true, it would be impossible to build a DAC let alone an interpolator.

If the cut-off frequency of the filter is one-half of the sampling rate, the impulse response passes through zero at the sites of all other samples. It can be seen from Fig.3.3.1 that at the output of such a filter, the voltage at the centre of a sample is due to that sample alone, since the value of all other samples is zero at that instant. In other words the continuous time output waveform must join up the tops of the input samples. In between the sample instants, the output of the filter is the sum of the contributions from many impulses, and the waveform smoothly joins the tops of the samples. If the waveform domain is being considered, the anti-image filter of the frequency domain can equally well be called the reconstruction filter. It is a consequence of the band-limiting of the original anti-aliasing filter that the filtered analog waveform could only travel between the sample points in one way. As the reconstruction filter has the same frequency response, the reconstructed output waveform must be identical to the original band-limited waveform prior to sampling.

Figure 3.3.1



4:1:1 to 4:2:2 conversion requires the colour difference sampling rate to be exactly doubled. Fig.3.3.2 shows that half of the output samples are identical to the input, and new samples need to be computed half way between them. The ideal impulse response required will be a sinx/x curve which passes through zero at all adjacent input samples. Fig.3.3.3 shows that this impulse response can be re-sampled at half the usual sample spacing in order to compute coefficients which the express the same impulse at half the previous sample spacing. In other words, if the height of the impulse is known, its value half a sample away can be computed. If a single input sample is multiplied by each of these coefficients in turn, the impulse response of that sample at the new sampling rate will be obtained. Note that every other coefficient is zero, which confirms that no computation is necessary on the existing samples; they are just transferred to the output. The intermediate sample is computed by adding together the impulse responses of every input sample in the window. Fig.3.3.4 shows how this mechanism operates.

Figure 3.3.2

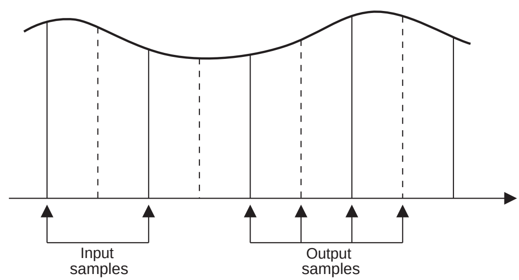
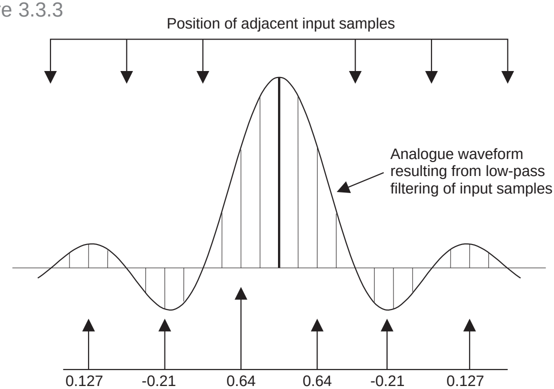
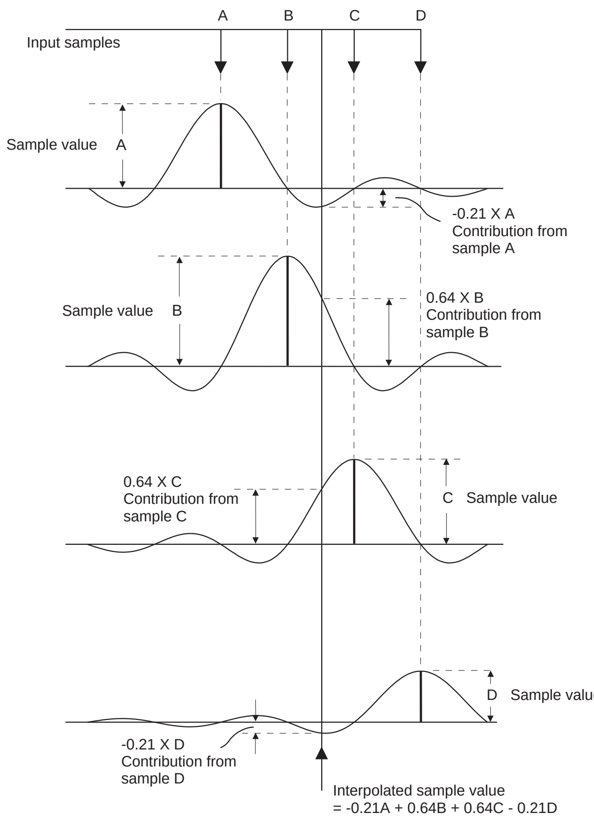


Figure 3.3.3




A
B
C
D
Input samples
Sample value
A
-0.21 X A
Contribution from
sample A
Sample value
B
0.64 X B
Contribution from
sample B
0.64 X C
Contribution from
sample C
C
Sample value
D
Sample value
-0.21 X D
Contribution from
sample D
Interpolated sample value
= -0.21A + 0.64B + 0.64C - 0.21D

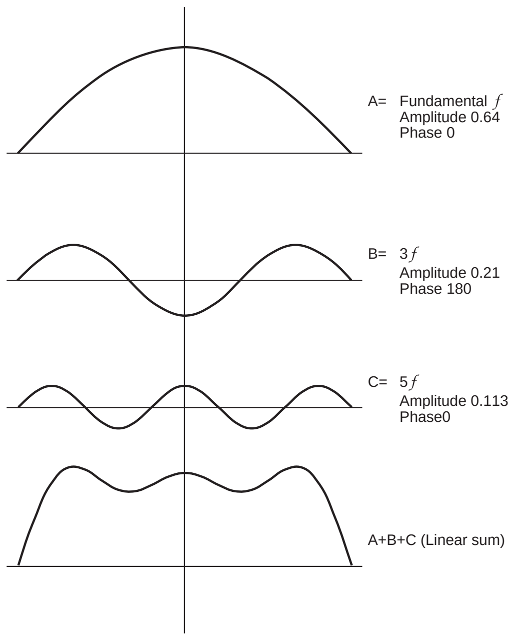
3.4 Transforms

In many types of video compression advantage is taken of the fact that a large signal level will not be present at all frequencies simultaneously. In audio compression a frequency analysis of the input signal will be needed in order to create a masking model. Frequency transforms are generally used for these tasks. Transforms are also used in the phase correlation technique for motion estimation.

3.5 The Fourier transform

The Fourier transform is a processing technique which analyses signals changing with respect to time and expresses them in the form of a spectrum. Any waveform can be broken down into frequency components. Fig.3.5.1 shows that if the amplitude and phase of each frequency component is known, linearly adding the resultant components results in the original waveform. This is known as an inverse transform.

Figure 3.5.1



In digital systems the waveform is expressed as a number of discrete samples. As a result the Fourier transform analyses the signal into an equal number of discrete frequencies. This is known as a Discrete Fourier Transform or DFT. The Fast Fourier Transform is no more than an efficient way of computing the DFT.

It is obvious from Fig.3.5.1 that the knowledge of the phase of the frequency component is vital, as changing the phase of any component will seriously alter the reconstructed waveform. Thus the DFT must accurately analyse the phase of the signal components. There are a number of ways of expressing phase. Fig.3.5.2 shows a point which is rotating about a fixed axis at constant speed. Looked at from the side, the point oscillates up and down. The waveform of that motion with respect to time is a sinewave.

Figure 3.5.2

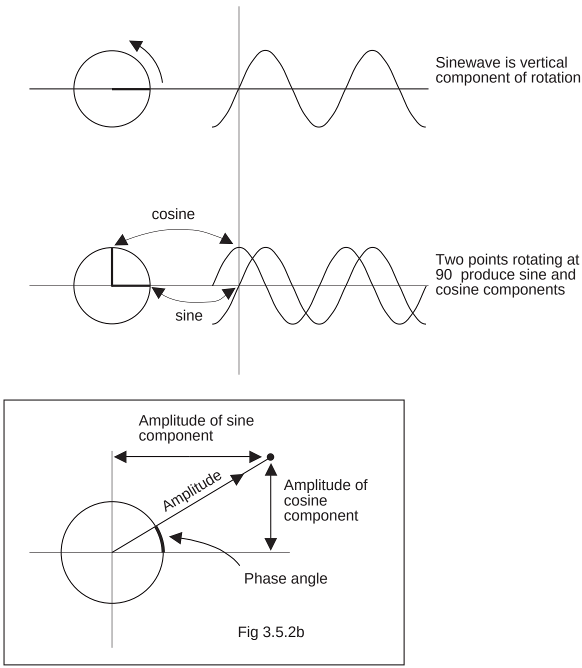


Fig 3.5.2b

One way of defining the phase of a waveform is to specify the angle through which the point has rotated at time zero (T=0). If a second point is made to revolve at 90 degrees to the first, it would produce a cosine wave when translated. It is possible to produce a waveform having arbitrary phase by adding together the sine and cosine wave in various proportions and polarities. For example adding the sine and cosine waves in equal proportion results in a waveform lagging the sine wave by 45 degrees.

Fig.3.5.2b also shows that the proportions necessary are respectively the sine and the cosine of the phase angle. Thus the two methods of describing phase can be readily interchanged.

The Fourier transform spectrum-analyses a block of samples by searching separately for each discrete target frequency. It does this by multiplying the input waveform by a sine wave having the target frequency and adding up or integrating the products. Fig.3.5.3a) shows that multiplying by the target frequency gives a large integral when the input frequency is the same, whereas Fig.3.5.3b) shows that with a different input frequency (in fact all other different frequencies) the integral is zero showing that no component of the target frequency exists. Thus a from a real waveform containing many frequencies all frequencies except the target frequency are excluded.

Figure 3.5.3

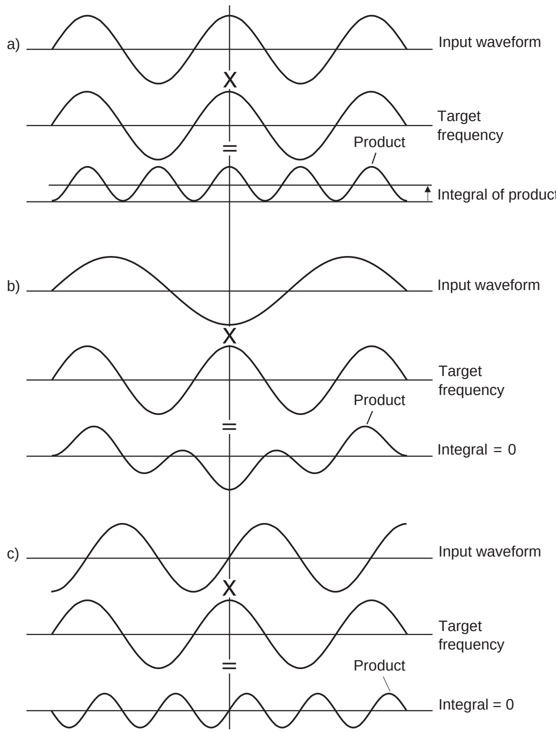


Fig.3.5.3c) shows that the target frequency will not be detected if it is phase shifted 90 degrees as the product of quadrature waveforms is always zero. Thus the Fourier transform must make a further search for the target frequency using a cosine wave. It follows from the arguments above that the relative proportions of the sine and cosine integrals reveals the phase

of the input component. For each discrete frequency in the spectrum there must be a pair of quadrature searches.

The above approach will result in a DFT, but only after considerable computation. However, a lot of the calculations are repeated many times over in different searches. The FFT aims to give the same result with less computation by logically gathering together all of the places where the same calculation is needed and making the calculation once.

The amount of computation can be reduced by performing the sine and cosine component searches together. Another saving is obtained by noting that every 180 degrees the sine and cosine have the same magnitude but are simply inverted in sign. Instead of performing four multiplications on two samples 180 degrees apart and adding the pairs of products it is more economical to subtract the sample values and multiply twice, once by a sine value and once by a cosine value.

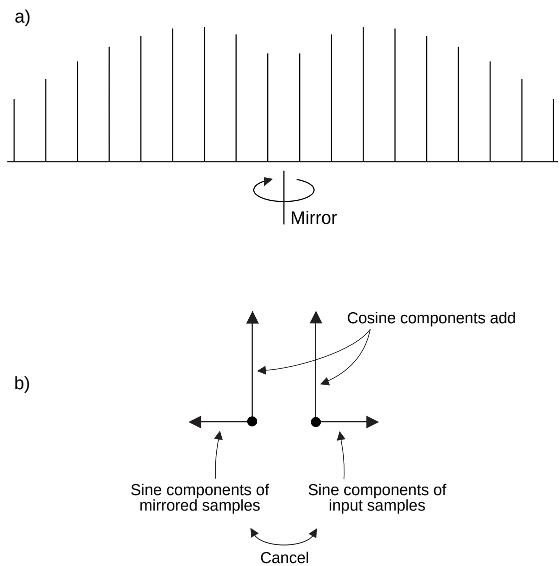
As a result of the FFT, the sine and cosine components of each frequency are available. For use with phase correlation it is necessary to convert to the alternative means of expression, i.e. phase and amplitude.

The number of frequency coefficients resulting from a DFT is equal to the number of input samples. If the input consists of a larger number of samples it must cover a larger area of the screen in video, a longer timespan in audio, but its spectrum will be known more finely. Thus a fundamental characteristic of transforms is that the more accurately the frequency and phase of a waveform is analysed, the less is known about where such frequencies exist.

3.6 The Discrete Cosine Transform

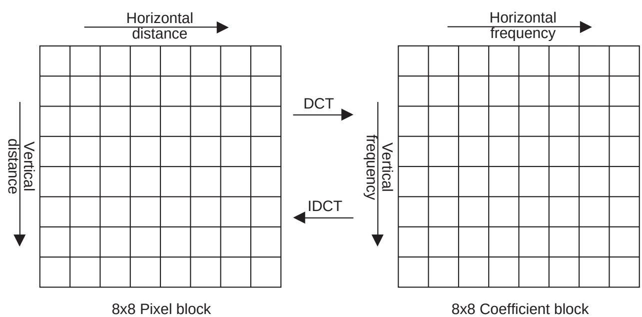
The two components of the Fourier transform can cause extra complexity and for some purposes a single component transform is easier to handle. The DCT (discrete cosine transform) is such a technique. Fig.3.6.1 shows that prior to the transform process the block of input samples is mirrored. Mirroring means that a reversed copy of the sample block placed in front of the original block. Fig.3.6.1 also shows that any cosine component in the block will continue across the mirror point, whereas any sine component will suffer an inversion. Consequently when the whole mirrored block is transformed, only cosine coefficients will be detected; all of the sine coefficients will be cancelled out.

Figure 3.6.1



For video processing, a two-dimensional DCT is required. An array of pixels is converted into an array of coefficients. Fig.3.6.2 shows how the DCT process is performed. In the resulting coefficient block, the coefficient in the top left corner represents the DC component or average brightness of the pixel block. Moving to the right the coefficients represent increasing horizontal spatial frequency. Moving down, the coefficients represent increasing vertical spatial frequency. The coefficient in the bottom right hand corner represents the highest diagonal frequency.

Figure 3.6.2



3.7 Motion estimation

Motion estimation is an essential component of inter-field video compression techniques such as MPEG. There are two techniques which can be used for motion estimation in compression: block matching, the most common method, and phase correlation.

Block matching is the simplest technique to follow. In a given picture, a block of pixels is selected and stored as a reference. If the selected block is part of a moving object, a similar block of pixels will exist in the next picture, but not in the same place. Block matching simply moves the reference block around over the second picture looking for matching pixel values. When a match is found, the displacement needed to obtain it is the required motion vector.

Whilst it is a simple idea, block matching requires an enormous amount of computation because every possible motion must be tested over the assumed range. Thus if the object is assumed to have moved over a sixteen pixel range, then it will be necessary to test 16 different horizontal displacements in each of sixteen vertical positions; in excess of 65,000 positions. At each position every pixel in the block must be compared with the corresponding pixel in the second picture.

One way of reducing the amount of computation is to perform the matching in stages where the first stage is inaccurate but covers a large motion range whereas the last stage is accurate but covers a small range. The first matching stage is performed on a heavily filtered and subsampled picture, which contains far fewer pixels. When a match is found, the displacement is used as a basis for a second stage which is performed with a less heavily filtered picture. Eventually the last stage takes place to any desired accuracy.

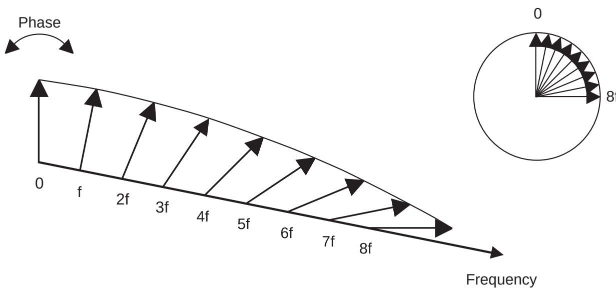
Inaccuracies in motion estimation are not a major problem in compression because they are inside the error loop and are cancelled by sending appropriate picture difference data. However, a serious error will result in small correlation between the two pictures and the amount of difference data will increase. Consequently quality will only be lost if that extra difference data cannot be transmitted due to a tight bit budget.

Phase correlation works by performing a Fourier transform on picture blocks in two successive pictures and then subtracting all of the phases of the spectral components. The phase differences are then subject to a reverse transform which directly reveals peaks whose positions correspond to motions between the fields. The nature of the transform domain means that if the distance and direction of the motion is measured accurately, the area of the screen in which it took place is not. Thus in practical systems the phase correlation stage is followed by a matching stage not dissimilar to the block matching process. However, the matching process is steered by the motions from the phase correlation, and so there is no need to attempt to match at all possible motions. By attempting matching on measured motion only the overall process is made much more efficient. One way of considering phase correlation is that by using the Fourier transform to break the picture into its constituent spatial frequencies the hierarchical structure of block matching at various resolutions is in fact performed in parallel.

The details of the Fourier transform are described in section 3.5. A one dimensional example will be given here by way of introduction. A row of luminance pixels describes brightness with respect to distance across the screen. The Fourier transform converts this function into a spectrum of spatial frequencies (units of cycles per picture width) and phases.

All television signals must be handled in linear-phase systems. A linear phase system is one in which the delay experienced is the same for all frequencies. If video signals pass through a device which does not exhibit linear phase, the various frequency components of edges become displaced across the screen. Fig.3.7.1 shows what phase linearity means. If the left hand end of the frequency axis (zero) is considered to be firmly anchored, but the right hand end can be rotated to represent a change of position across the screen, it will be seen that as the axis twists evenly the result is phase shift proportional to frequency. A system having this characteristic is said to have linear phase.

Figure 3.7.1



In the spatial domain, a phase shift corresponds to a physical movement. Fig.3.7.2 shows that if between fields a waveform moves along the line, the lowest frequency in the Fourier transform will suffer a given phase shift, twice that frequency will suffer twice that phase shift and so on. Thus it is potentially possible to measure movement between two successive fields if the phase differences between the Fourier spectra are analysed. This is the basis of phase correlation.

Figure 3.7.2

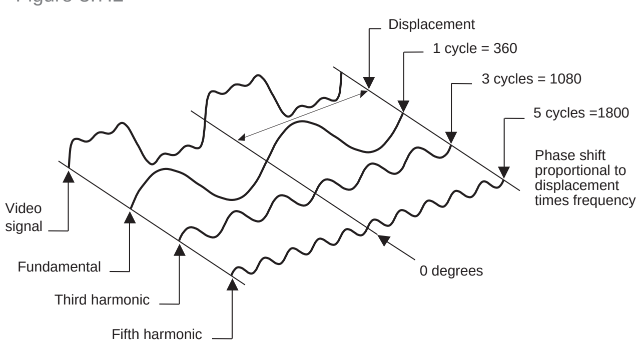
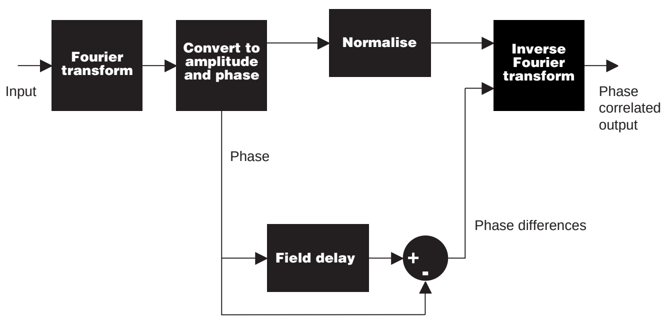


Fig.3.7.3 shows how a one dimensional phase correlator works. The Fourier transforms of pixel rows from blocks in successive fields are computed and expressed in polar (amplitude and phase) notation. The phases of one transform are all subtracted from the phases of the same frequencies in the other transform. Any frequency component having significant amplitude is then normalised, or boosted to full amplitude.

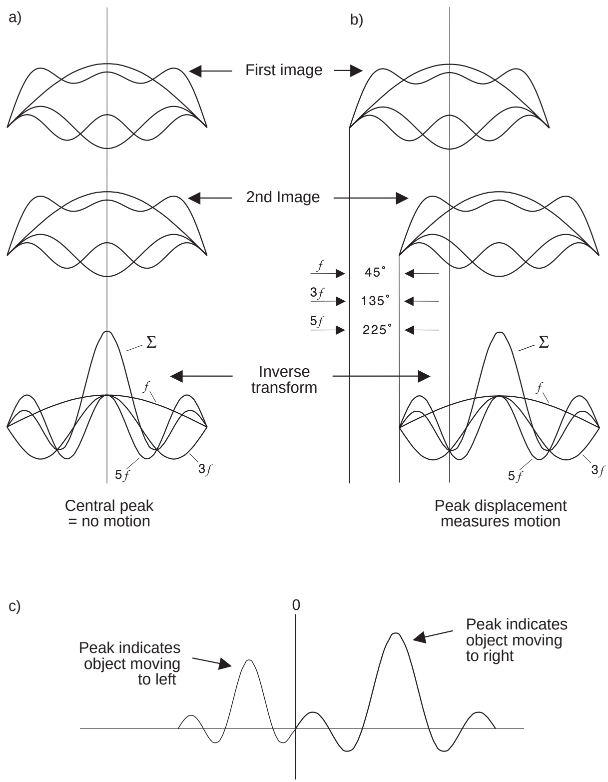
Figure 3.7.3



The result is a set of frequency components which all have the same amplitude, but have phases corresponding to the difference between two blocks. These coefficients form the input to an inverse transform.

Fig.3.7.4 shows what happens. If the two fields are the same, there are no phase differences between the two, and so all of the frequency components are added with zero degree phase to produce a single peak in the centre of the inverse transform. If, however, there was motion between the two fields, such as a pan, all of the components will have phase differences, and this results in a peak shown in Fig.3.7.4b) which is displaced from the centre of the inverse transform by the distance moved. Phase correlation thus actually measures the movement between fields.

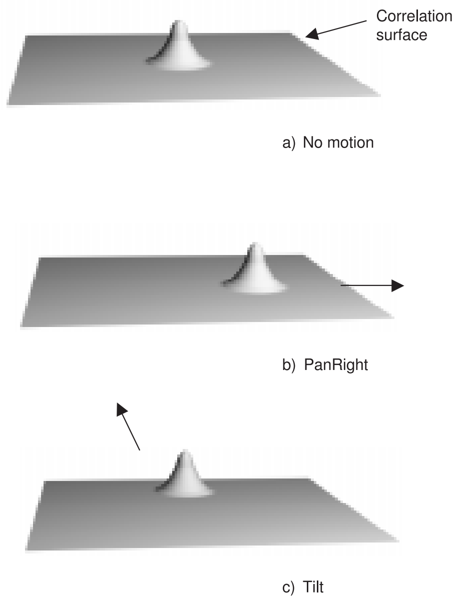
Figure 3.7.4



In the case where the line of video in question intersects objects moving at different speeds, Fig.3.7.4c) shows that the inverse transform would contain one peak corresponding to the distance moved by each object. Whilst this explanation has used one dimension for simplicity, in practice the entire process is two dimensional. A two dimensional Fourier transform of each field is computed, the phases are subtracted, and an inverse two dimensional transform is computed, the output of which is a flat plane out of which three dimensional peaks rise. This is known as a correlation surface.

Fig.3.7.5 shows some examples of a correlation surface. At a) there has been no motion between fields and so there is a single central peak. At b) there has been a pan and the peak moves across the surface. At c) the camera has been depressed and the peak moves upwards. Where more complex motions are involved, perhaps with several objects moving in different directions and/or at different speeds, one peak will appear in the correlation surface for each object. It is a fundamental strength of phase correlation that it actually measures the direction and speed of moving objects rather than estimating, extrapolating or searching for them.

Figure 3.7.5



However it should be understood that accuracy in the transform domain is incompatible with accuracy in the spatial domain. Although phase correlation accurately measures motion speeds and directions, it cannot specify where in the picture these motions are taking place. It is necessary to look for them in a further matching process. The efficiency of this process is dramatically improved by the inputs from the phase correlation stage.

Section 4 - Audio compression

In this section we look at the principles of audio compression which will serve as an introduction to the description of MPEG in section 6.

4.1 When to compress audio

The audio component of uncompressed television only requires about one percent of the overall bit rate. In addition human hearing is very sensitive to audio distortion, including that caused by clumsy compression. Consequently for many television applications, the audio need not be compressed at all. For example, compressing video by a factor of two means that uncompressed audio now represents two percent of the bit rate. Compressing the audio is simply not worthwhile in this case. However, if the video has been compressed by a factor of fifty, then the audio and video bit rates will be comparable and compression of the audio will then be worthwhile.

4.2 The basic mechanisms

All audio data reduction relies on an understanding of the hearing mechanism and so is a form of perceptual coding. The ear is only able to extract a certain proportion of the information in a given sound. This could be called the perceptual entropy, and all additional sound is redundant. Section 1 introduced the concept of auditory masking which is the inability of the ear to detect certain sounds in the presence of others.

The main techniques used in audio compression are:

* Requantizing and gain ranging

These are complementary techniques which can be used to reduce the wordlength of samples, conserving bits. Gain ranging boosts low-level signals as far above the noise floor as possible. Requantizing removes low order bits, raising the noise floor. Using masking, the noise floor of the audio can be raised, yet remain inaudible. The gain ranging must be reversed at the decoder

* Predictive coding

This uses a knowledge of previous samples to predict the value of the next. It is then only necessary to send the difference between the prediction and the actual value. The receiver contains an identical predictor to which the transmitted difference is added to give the original value.

* Sub band coding.

This technique splits the audio spectrum up into many different frequency bands to exploit the fact that most bands will contain lower level signals than the loudest one.

* Spectral coding.

A transform of the waveform is computed periodically. Since the transform of an audio signal changes slowly, it need be sent much less often than audio samples. The receiver performs an inverse transform. The transform may be Fourier, Discrete Cosine (DCT) or Wavelet.

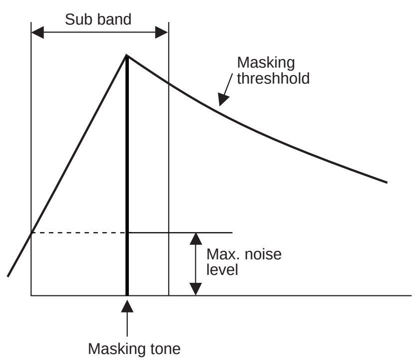
Most practical compression units use some combination of sub-band or spectral coding and rely on masking the noise due to re-quantizing or wordlength reduction of sub-band samples or transform coefficients.

4.3 Sub-band coding

Sub-band compression uses the fact that real sounds do not have uniform spectral energy. When a signal with an uneven spectrum is conveyed by PCM, the whole dynamic range is occupied only by the loudest spectral component, and all other bands are coded with excessive headroom. In its simplest form, sub-band coding works by splitting the audio signal into a number of frequency bands and companding each band according to its own level. Bands in which there is little energy result in small amplitudes which can be transmitted with short wordlength. Thus each band results in variable length samples, but the sum of all the sample wordlengths is less than that of PCM and so a coding gain can be obtained.

The number of sub-bands to be used depends upon what other technique is to be combined with the sub-band coding. If used with requantizing relying on auditory masking, the sub-bands should be narrower than the critical bands of the ear, and therefore a large number will be required, ISO/MPEG Layers 1 and 2, for example, use 32 sub-bands. Fig.4.3.1 shows the critical condition where the masking tone is at the top edge of the sub band. Obviously the narrower the sub band, the higher the noise that can be masked.

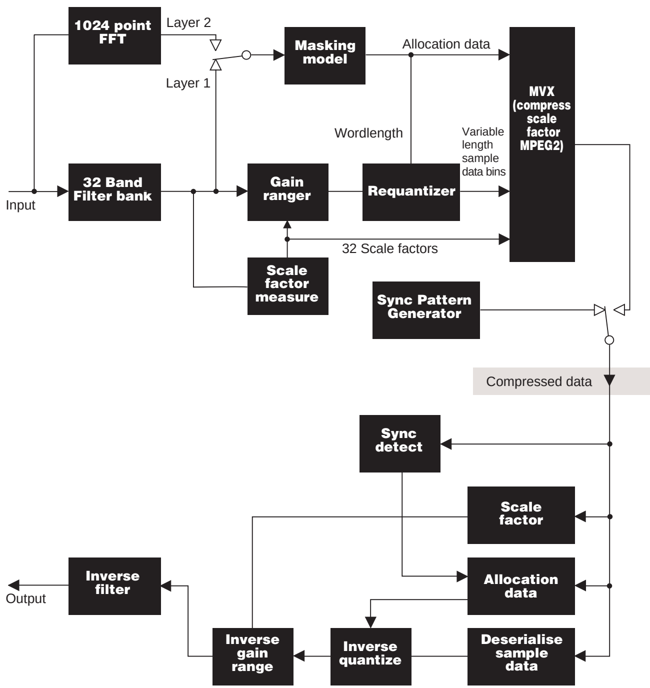
Figure 4.3.1



The band splitting process is complex and requires a lot of computation. One bandsplitting method which is useful is quadrature mirror filtering. The QMF is a kind of double filter which converts a PCM sample stream into to two sample streams of half the input sampling rate, so that the output data rate equals the input data rate. The frequencies in the lower half of the audio spectrum are carried in one sample stream, and the frequencies in the upper half of the spectrum are heterodyned or aliased into the other. These filters can be cascaded to produce as many equal bands as required.

Fig.4.3.2 shows the block diagram of a simple sub band coder. At the input, the frequency range is split into sub bands by a filter bank such as a quadrature mirror filter. The decomposed sub band data are then assembled into blocks of fixed size, prior to reduction. Whilst all sub bands may use blocks of the same length, some coders may use blocks which get longer as the sub band frequency becomes lower. Sub band blocks are also referred to as frequency bins.

Figure 4.3.2



The coding gain is obtained as the waveform in each band passes through a requantizer. The requantization is achieved by multiplying the sample values by a constant and rounding up or down to the required wordlength. For example, if in a given sub band the waveform is 36 dB down on full scale, there will be at least six bits in each sample which merely replicate the sign bit. Multiplying by 64 will bring the high order bits of the sample into use, allowing bits to be lost at the lower end by rounding to a shorter wordlength. The shorter the wordlength, the greater the coding gain, but the coarser the quantisation steps and therefore the level of quantisation error. If a fixed data reduction factor is employed, the size of the coded output block will be fixed. The requantization wordlengths will have to be such that the sum of the bits from each sub band equals the size of the coded block. Thus some sub bands can have long wordlength coding if others have short wordlength coding. The process of determining the requantization step size, and hence the wordlength in each sub band, is known as bit allocation. The bit allocation may be performed by analysing the power in each sub band, or by a side chain which performs a spectral analysis or transform of the audio. The complexity of the bit allocation depends upon the degree of compression required. The spectral content is compared with an auditory masking model to determine the degree of masking which is taking place in certain bands as a result of higher levels in other bands. Where masking takes place, the signal is quantized more coarsely until the quantizing noise is raised to just below the masking level. The coarse quantisation requires shorter wordlengths and allows a coding gain. The bit allocation may be iterative as adjustments are made to obtain the best masking effect within the allowable data rate.

The samples of differing wordlength in each bin are then assembled into the output coded block. The frame begins with a sync pattern to reset the phase of deserialisation, and a header which describes the sampling rate and any use of pre-emphasis. Following this is a block of 32 four-bit allocation codes. These specify the wordlength used in each sub band and allow the decoder to deserialize the sub band sample block. This is followed by a block of 32 six-bit scale factor indices, which specify the gain given to each band during normalisation. The last block contains 32 sets of 12 samples. These samples vary in wordlength from one block to the next, and can be from 0 to 15 bits long. The deserializer has to use the 32 allocation information codes to work out how to deserialize the sample block into individual samples of variable length. Once all of the samples are back in their respective frequency bins, the level of each bin is returned to the original value. This is achieved by reversing the gain increase which was applied before the requantizer in the coder. The degree of gain reduction to use in each bin comes from the scale factors. The sub bands can then be recombined into a continuous audio spectrum in the output filter which produces conventional PCM of the original wordlength.

The degree of compression is determined by the bit allocation system. It is not difficult to change the output block size parameter to obtain a different compression factor. The bit allocator simply iterates until the new block size is filled. Similarly the decoder need only deserialize the larger block correctly into coded samples and then the expansion process is identical except for the fact that expanded words contain less noise. Thus codecs with varying degrees of compression are available which can perform different bandwidth/performance tasks with the same hardware.

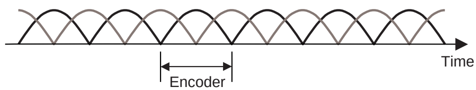
4.4 Transform coding

Fourier analysis allows any periodic waveform to be represented by a set of harmonically related components of suitable amplitude and phase. The transform of a typical audio waveform changes relatively slowly. The slow growth of sound from an organ pipe or a violin string, or the slow decay of most musical sounds allow the rate at which the transform is sampled to be reduced, and a coding gain results. A further coding gain will be achieved if the components which will experience masking are quantized more coarsely.

Practical transforms require blocks of samples rather than an endless stream. One solution is to cut the waveform into short overlapping segments or windows and then to transform each individually as shown in

Fig.4.4.1. Thus every input sample appears in just two transforms, but with variable weighting depending upon its position along the time axis.

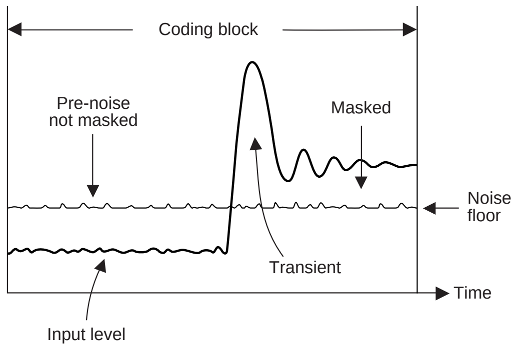
Figure 4.4.1



The DFT (discrete Fourier transform) requires intensive computation, owing to the requirement to use complex arithmetic to render the phase of the components as well as the amplitude. An alternative is to use Discrete Cosine Transforms (DCT) in which the coefficients are single numbers. In any transform, accuracy of frequency resolution is obtained with the penalty of poor time resolution, giving a problem locating transients properly on the time axis. The wavelet transform is especially good for audio because its time resolution increases automatically with frequency.

The wordlength reduction or requantizing in the coder raises the quantizing noise in the frequency band, but it does so over the entire duration of the window. Fig.4.4.2 shows that if a transient occurs towards the end of a window, the decoder will reproduce the waveform correctly, but the quantizing noise will start at the beginning of the window and may result in a pre-echo where a burst of noise is audible before the transient.

Figure 4.4.2

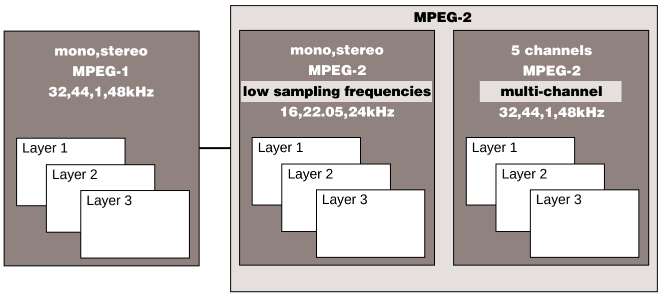


One solution is to use a variable time window according to the transient content of the audio waveform. When musical transients occur, short blocks are necessary and the frequency resolution and hence the coding gain will be low. At other times the blocks become longer and the frequency resolution of the transform rises, allowing a greater coding gain.

4.5 Audio compression in MPEG

The layers and levels of MPEG audio coding are shown in Fig.4.5.1. In MPEG-1 audio inputs have sampling rates of 32, 44.1 or 48KHz. There are three coding algorithms of ascending complexity, known as Layers 1,2 and 3. There are four operating modes in MPEG-1 which are shown in Fig.4.5.2. In mono mode, only one audio signal is handled. In stereo mode, two audio signals are handled, but the data are held in a common buffer so that entropy variations between the two channels can be exploited. In dual mono mode, the available bit rate is exactly halved so that two independent unrelated audio signals, perhaps a dual language soundtrack, can be handled. In joint stereo mode, only the lower half of the input audio spectrum is transmitted as stereo. The upper half of the spectrum is transmitted as a joint signal. This allows a high compression factor to be used.

Figure 4.5.1



Audio in MPEG-1 was intended primarily for full-bandwidth music applications. When high compression factors are required, the noise floor will inevitably rise and a better result will be obtained by curtailing the audio bandwidth, an acceptable solution for speech and similar applications. Whilst MPEG-2 decoders must be able to decode MPEG-1 data, MPEG-2 allows as an option 3 additional lower sampling rates which are exactly one half of the MPEG-1 rates so that downsampled audio can be used as input. This is known as the Low Sampling Frequency (LSF) extension. MPEG-2 also allows a multi-channel option intended for surround-sound applications.

4.6 MPEG Layers

Layer 2 is designed to be the most appropriate Layer for everyday purposes. Layer 1 requires a simpler coder and therefore must use a higher bit rate or lower quality will result. Layer 3 is extremely complex and consequently allows the best quality consistent with very low bit rates.

Layer 1 uses a sub-band coding scheme having 32 equal bands and works as described in section 4.3. The auditory model is obtained from the levels in the sub-bands themselves so no separate spectral analysis is needed.

Layer 2 uses the same sub-band filter, but has a separate Transform analyser which creates a more accurate auditory model. In Level 2 the processing of the scale factors is more complex, taking advantage of similarity in scale factors from one frame to the next to reduce the amount of scale data transmitted. Layers 1 and 2 have a constant frame length of 1152 audio samples.

Layer 3 is much more complex because it attempts a very accurate auditory modelling process. A 576-line Discrete Cosine Transform is calculated, and various numbers of lines are grouped together to simulate the varying width of the critical bands of human hearing. The audio is transmitted as transform coefficients which have been requantized according to the masking model. Huffman coding is used to lower the data rate further. The decoder requires an inverse transform process. Layer 3 also supports a variable frame length which allows entropy variations in the audio input to be better absorbed.

Mono Stereo Dual Mono Joint Stereo	**All layers**
M-S Stereo Intensity Stereo Intensity & M-S Stereo	**Layer 3 only**

Layer 3 supports additional modes than the four modes of Layers 1 and 2. These are shown in Fig.4.5.2. M-S coding produces sum and difference signals from the L-R stereo. The S or difference signal will have low entropy when there is correlation between the two input channels. This allows a further saving of bit rate.

Intensity coding is a system where in the upper part of the audio frequency band, for each scale factor band only one signal is transmitted, along with a code specifying where in the stereo image it belongs. The decoder has the equivalent of a pan-pot so that it can output the decoded waveform of each band at the appropriate level in each channel. The lower part of the audio band may be sent as L-R or as M-S.

Section 5 - Video compression

In this section the essential steps used in video compression will be detailed. This will form an introduction to the description of MPEG in section 6.

5.1 Spatial and temporal redundancy

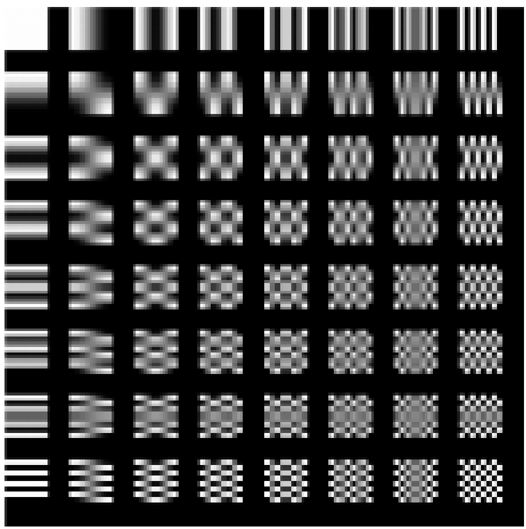
Video compression requires the identification of redundancy in typical source material. There are two basic forms of redundancy which can be exploited. The first of these is intra-frame or spatial redundancy, which is redundancy that can be identified in a single image without reference to any other. The second is inter-frame or temporal redundancy which can be identified from one frame to the next.

5.2 The Discrete Cosine Transform

Spatial redundancy is found in all real program material. Where a sizeable object is recognisable in the picture, all of the pixels representing that object will have quite similar values. Large objects produce low spatial frequencies, whereas small objects produce high spatial frequencies. Generally these frequencies will not be present at high level at the same time. Normal PCM video has to be able to transmit the whole range of spatial frequencies, but if a frequency analysis is performed, only those frequencies actually present need be transmitted. Consequently a major step in intra-coding is to perform a spatial frequency analysis of the image.

In MPEG the Discrete Cosine Transform or DCT (see section 3) is used. Fig.5.2.1 shows how the two-dimensional DCT works. The image is converted a block at a time. A typical block is 8 x 8 pixels. The DCT converts the block into a block of 64 coefficients. A coefficient is a number which describes the amount of a particular spatial frequency which is present. In the figure the pixel blocks which result from each coefficient are shown. The top left coefficient represents the average brightness of the block and so is the arithmetic mean of all the pixels or the DC component. Going across to the right, the coefficients represent increasing horizontal spatial frequency. Going downwards the coefficients represent increasing vertical spatial frequency.

Figure 5.2.1



Now the DCT itself doesn't achieve any compression. In fact the wordlength of the coefficients will be longer than that of the source pixels. What the DCT does is to convert the source pixels into a form in which redundancy can be identified. As not all spatial frequencies are simultaneously present, the DCT will output a set of coefficients where some will have substantial values, but many will have values which are almost or actually zero. Clearly if a coefficient is zero it makes no difference whether it's sent or not. If a coefficient is almost zero, omitting it would have the same effect as adding the same spatial frequency to the image but in the opposite phase. The decision to omit a coefficient is based on how visible that small unwanted signal would be. If a coefficient is too large to omit, compression can also be achieved by reducing the number of bits used to carry the coefficient. This has the same effect; a small noise is added to the picture.

5.3 Weighting

Visibility of spatial frequencies by the human viewer is not uniform; much higher levels of noise can be tolerated at high frequencies. Consequently weighting is used so that any noise which is created is concentrated at high frequencies. Coefficients are divided by a weighting factor which is a function of the position in the block. The DC component is unweighted whereas the division factor increases towards the bottom right hand corner. At the decoder an inverse weighting process is needed. This multiplies higher frequency coefficients by greater factors, raising the high frequency noise.

After weighting, the value of certain coefficients will be even smaller. On typical program material, large value coefficients are mostly found in the top left corner, the remainder of the coefficients are often negligible or zero. It is an advantage to transmit the coefficients in a zigzag sequence starting from the top left corner. When this is done, the non-zero coefficients are typically transmitted soonest, leaving typically zero coefficients at the end. At some point when all subsequent coefficients are zero, it is more efficient to terminate the transmission with a simple character which tells the receiver that there are no more non-zero coefficients.

5.4 Variable length coding

Analysis of typical source data shows that the probability of low value coefficients having many leading zeros is higher than that of large value coefficients sent as 4,2. Where the probability of such values is known it is more efficient to use a coding scheme which takes advantage of such a distribution.

The Huffman coding scheme allocates shorter codes to more probable values and longer codes to values which are less probable. When used with source data having assumed probability an improvement in transmission efficieny is obtained.

In DCT coding it is often the case that may coefficients have a value of zero. In this case run length coding simply tells the decoder how many zero valued coefficients were present instead of sending each separately.

5.5 Intra-coding

Fig.5.5.1 shows a complete intra-coding scheme. The input picture is converted from raster scan to blocks. The blocks are subject to a DCT. The coefficients are then zigzag scanned and weighted, prior to being requantized (wordlength shortening) and subject to run-length coding. In practical systems a constant output bit rate is required even though the source entropy varies. There are two solutions. One is to use a buffer to absorb the variations. The other is to measure the quantity of output data and compare it with the required amount. If there is too much data, the requantizer is made to shorten the wordlength of the coefficients more severely. If there is insufficient data the requantizing is eased.

Figure 5.5.1

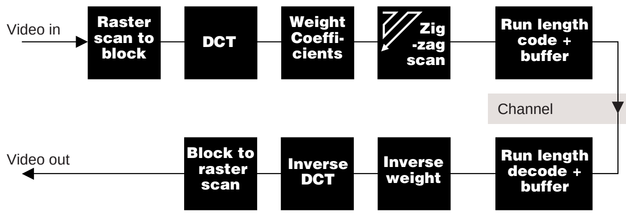
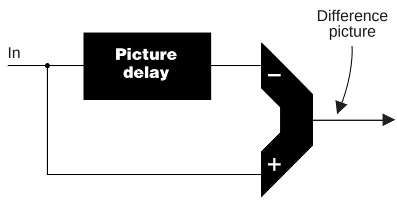


Fig.5.5.1. also shows the corresponding decoder. The run length coding is decoded and then the coefficients are subject to an inverse weighting before being assembled into a coefficient block. The inverse transform produces a block of pixels which is stored in RAM with other blocks so that a raster scanned output can be produced.

5.6 Inter-coding

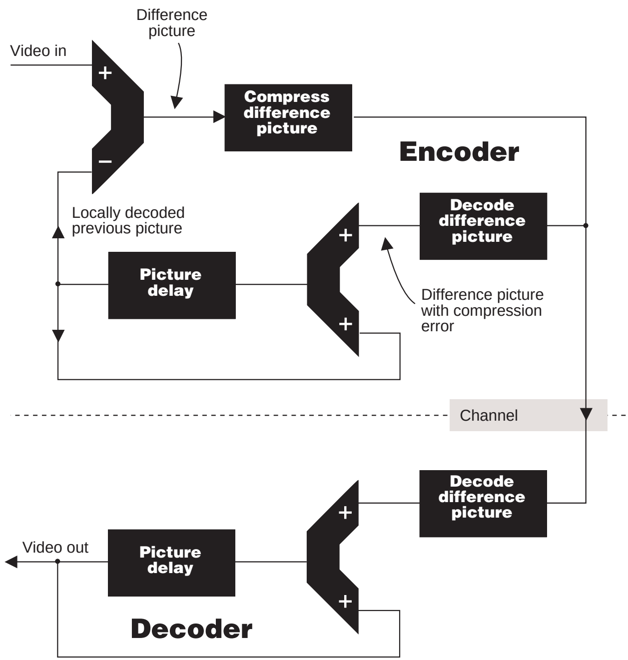
The system of Fig.5.5.1 is basically that used in JPEG which is designed for still images. Intra-coding takes no advantage of redundancy between pictures. In typical source material one picture may be quite similar to the previous one. Fig.5.6.1 shows how a differential coder works. A frame store is required which delays the previous picture. Each pixel in the previous picture is subtracted from the current picture to produce a difference picture. The difference picture is a two-dimensional image in its own right and this can be subject to an intra-coding compression step using a DCT. At the decoder there is another framestore which delays the previous output picture. The decoded difference picture is added to the previous picture to make the current picture which not only forms the output, but also enters the pircture delay to become the basis of the next picture.

Figure 5.6.1



In differential coding it is important to prevent the build up of small errors which might accumulate over a succession of pictures. One such source of error is the compression of the difference picture. Fig.5.6.2 shows how this error is removed. Both the encoder and the decoder contain an identical difference picture decoder. When the encoder subtracts a previous picture from a current picture, it actually subtracts a locally decoded compressed picture identical to that which the real decoder will have. Consequently any small errors due to the compression of earlier difference pictures are removed.

Figure 5.6.2

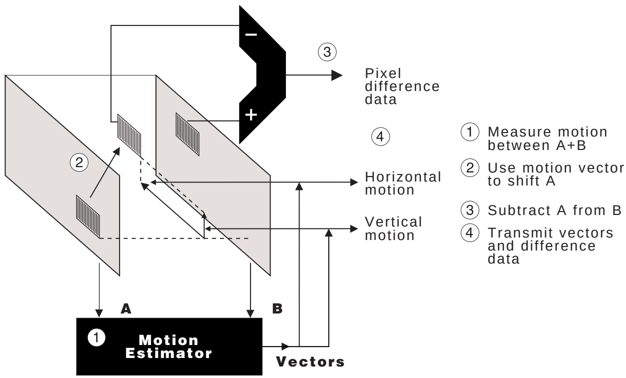


5.7 Motion compensation

Differential coding works well on largely static picture material, but falls down heavily when there is motion. In the extreme case of a pan, the value of every pixel changes and the difference picture contains nearly as much data as the original picture. However, a pan does not change many pixel values at all, the values are just present in a different place. Consequently motion compensation is required to obtain high compression factors in differential coding schemes.

Fig.5.7.1 shows that in a motion compensated compressor, areas of two successive pictures are compared in order to determine the direction and distance of relative motion between the frames. These are expressed as a two dimensional vector. The encoder uses the motion vector to shift pixels from the previous picture to make what is called a predicted picture prior to subtracting them from the current or actual picture. If the motion measurement has been accurate, the shifting process will have aligned the objects in the predicted frame with their position in the actual frame and only a small amount of difference data will result. Clearly if the motion vectors are transmitted along with the difference data, the decoder can make the same shift of the previous frame to produce its own predicted frame. Adding the difference data will produce the decoded current frame.

Figure 5.7.1



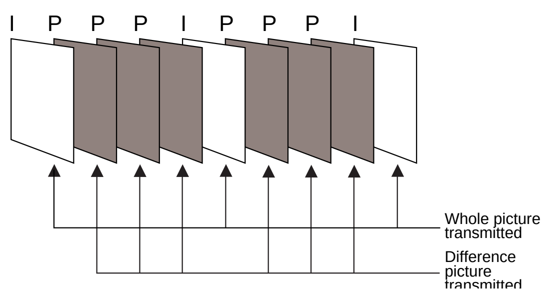
The picture is broken up into rectangular areas called macroblocks. Each of these macroblocks has its own motion vector which applies to the whole block. Clearly if the edge of a moving object lies across a macroblock, some of the macroblock is moving and some is not. This situation may be handled by setting the motion vectors to zero and handling the moving part with difference data, or by using non-zero motion vectors and handling the stationary part with difference data. A smart compressor might try both to see which approach yielded the smaller bit rate.

As motion in real program material tends to be relatively constant across several pictures, the motion vectors are transmitted differentially. Consequently a motion vector parameter of zero is interpreted as the same motion continuing. The vectors from the previous picture are copied. If the motion slows down or changes direction, vector differences are sent.

5.8 I pictures

Pure differential coders cannot be used in practice because the decoder can only make sense of the transmission if it is received from the beginning. Any transmission errors will affect the picture indefinitely afterwards. In practice differential transmission systems punctuate the difference pictures with periodic absolute pictures which are intra-coded. Fig.5.8.1 shows the principle. The intra-coded or I pictures are created by turning off the prediction system so that no previous data are subtracted from the input. The difference picture becomes an absolute picture. In between I pictures, the motion compensated prediction process operates and P pictures, consisting of motion vectors and DCT compressed difference values are created.

Figure 5.8.1



In the case of film originated material which has been played on a telecine, knowledge of the scanning process is useful to an encoder. For example a 60Hz telecine uses 3:2 pulldown to obtain 60 Hz field rate from 24 Hz film. In the three field frame, the third field is a direct repeat of the first field and consequently carries no information. Only the first two fields are transmitted by the coder. The decoder can repeat one of them to recreate the 3:2 sequence.

In both 50Hz and 60 Hz telecine, there can be no motion between the fields which come from the same film frame. Consequently there is no point in measuring motion or transmitting motion vectors for the second field.

There is no requirement for absolute regularity in the I pictures. Picture types are labelled so the decoder can automatically handle whatever it receives. Cut edits in the incoming material cause large difference data. With a suitable pre-processor, a compressor can anticipate the arrival of a cut in the source material and make it coincide with an I picture so that the most effective use of the bit rate is made.

It should be clear that editing of the bitstream is likely to be problematical in a differential coding scheme. If the bitstream is switched at an arbitrary point the decoder will be unable to produce pictures until the next I picture arrives. Consequently editing is restricted to switching at I pictures. This will be too restricting for production purposes and where editing is anticipated, the two fields of an interlaced frame can be compressed as one I picture and one P picture. Editing to frame accuracy is then possible, Clearly the compression factor which can be obtained will be relatively poor because of the frequent I pictures. This is consistent with the cautions expressed in Section 1.

Section 6 - MPEG

In this section all of the concepts introduced in earlier sections are tied together in a description of MPEG. MPEG is not a compression scheme as such, in fact it is hard to define MPEG in a single term. MPEG is essentially a set of standard tools, or defined algorithms, which can be combined in a variety of ways to make real compression equipment.

6.1 Applications of MPEG

Unusually, MPEG does not define how the encoder should carry out compression. Instead MPEG defines precisely how a decoder should attribute meaning to a variety of compressed bitstreams. MPEG does not suggest how to transmit the bitstream, as this will depend on the application. The reason for this approach is that it allows a great deal of flexibility whilst retaining compatibility. The manufacturers can use proprietary encoding algorithms yet produce a standard bitstream. As algorithms improve with time, the signals will still be compatible with existing decoders.

The uses of video compression go beyond the requirements of broadcast television. Applications range from small moving images on personal computers to large high definition displays. Picture quality requirements range from relatively poor for security and videophone purposes through CD-I to transparent quality. Sometimes the same transmission needs to be decoded to different standards. For example, there is no point in decoding a high definition signal for display on a 12 inch TV set. Different applications can afford different investments in encoder and decoder complexity. In addition to video, MPEG caters for associated compressed audio and means to multiplex audio and video together in a way which retains synchronisation.

6.2 Profiles and Levels

The wide range of performance and complexity required is handled by subdividing MPEG into Profiles and then dividing each profile into Levels. Fig.6.2.1 shows that a Profile is a technique whereas a level is a constraint such as picture size or bit rate used with that technique. Fig.6.2.1 also shows the Profiles available.

Figure 6.2.1

Levels → / Profiles ↓	Low level up to 352x288 pixels 4 MBit/s max.	Main level up to 720x576 pixels 15 MBit/s max	High 1440 level Up to 720x576 pixels 60 MBit/s max.	High level Up to 1920x1152 pixels 80 MBit/s max.
Simple		.		
Main		M.P.M.L.		
S.N.R scaleable				
Spatial scaleable				
High				

The Simple Profile, as its name suggests, allows minimal complexity at encoder and decoder. Basically a simple profile system uses no more than the I and P pictures described in section 5.

The Main Profile is designed to be suitable for a large proportion of uses and has received a lot of attention. The Main Profile will be described in detail later in this section.

The SNR Scaleable Profile is a system in which two separate data streams are used to carry one video signal. Together, the two data streams can be decoded to produce a high quality picture with a high signal-to-noise ratio. However, if one of the data streams is not received, the result is a picture having the same spatial resolution but with a reduced signal-to-noise ratio. Instead of producing one compressed bitstream, the coder is designed to produce a two streams. One of these carries a noisy picture, whereas the other carries a "helper" signal which when added to the first signal eliminates the noise. An anticipated use of the technology in DVB (digital video broadcasting) where transmitters would radiate an SNR scaleable signal where the first bitstream is transmitted with more power. Fixed receivers with good antennas could receive both bitstreams and produce a quality picture. In the case of poor reception only the more powerful signal would get through allowing a poorer picture in preference to no picture at all.

The Spatial Scaleable Profile is a system in which more than one data stream can be used to reproduce a picture of different resolution depending on the display. One data stream is a low-resolution picture which can be decoded alone for display on small portable sets. Other data streams are "spatial helper" signals which when added to the first signal serve to increase its resolution for use on large displays.

The High Profile is temporally scaleable, allowing a transmission which can be decoded into two different frame rates according to the complexity of the decoder.

Within the Profile structure, a number of levels are available as shown in Fig.6.2.1. The levels differ primarily in the resolution available and the bit rate required. The Main Level is appropriate for Standard Definition Television (SDTV). Consequently the majority of broadcast and production interest is in the MPML (Main Profile Main Level) subset of MPEG.

In general, an MPEG decoder with a given Profile and Level must be capable of decoding inputs corresponding to lower profiles and levels as well. Thus a scaleable decoder would continue to operate if it received a main level or a simple level input.

Most of the hierarchy of MPEG is based on 4:2:0 format video where the colour difference signals are vertically subsampled. For broadcast purposes, it is widely believed that 4:2:2 working is required and a separate 4:2:2 profile has been proposed which is not in the hierarchy.

6.3 MPEG-1 and MPEG-2

MPEG-1 was the first document to be published. This described standards for low-bit rate coding up to 1.5 megabits/sec. Interlace was not supported and the resolution was limited, allowing a maximum of 352 x 288 pixels. Up to 192kbit/sec could be used for stereo audio. The bit rate was largely determined by that of a standard Compact Disc so that it could be used for A/V applications such as CD-I and CD-ROM.

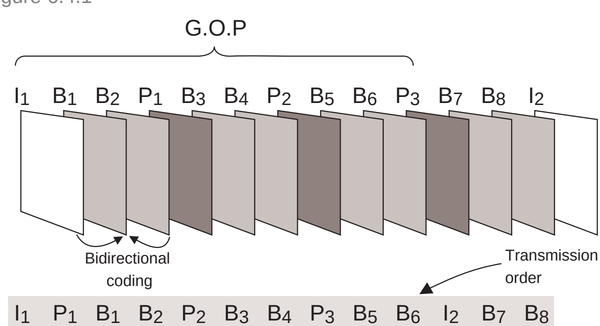
Clearly the uses of MPEG-1 in broadcasting are extremely limited, and MPEG-2 was designed to rectify the problem. MPEG-2 supports interlace and codes SDTV at between 3 and 15 megabits/sec or HDTV at 15-30 megabits/sec. The audio capability is extended to support surround sound. Because of the hierarchical structure of MPEG, an MPEG-2 decoder will be able to decode MPEG-1 signals, but not vice versa.

6.4 Bi-directional coding

One of the most powerful features of MPEG is the concept of bi-directional coding. Section 5 introduced the concept of motion compensated prediction and P pictures which were decoded by using the previous picture as a basis, shifted by motion vectors and corrected with difference data. Bi-directional coding takes this idea a step forward by allowing prediction of part of a picture from earlier or later pictures.

Fig.6.4.1 shows an example of a picture sequence used in MPEG. The sequence begins with an I picture as an anchor, and this and all pictures until the next I picture are called a Group of Pictures (GOP). Within the GOP are a number of forward predicted or P pictures. The first P picture is decoded using the I picture as a basis. The next and subsequent P pictures are decoded using the previous P picture as a basis. The remainder of the pictures in the GOP are B pictures. B pictures may be decoded using data from I or P pictures immediately before or afterwards or taking an average of the two. As the pictures are divided into macroblocks, each macroblock may be forward or backward predicted. Obviously backward prediction cannot be used if the decoder does not yet have the frame being projected backwards. The solution is to send the pictures in the wrong order. After the I picture, the first P picture is sent next. Once the decoder has the I and P pictures, the B pictures in between can be decoded by moving motion compensated data forwards or backwards. Clearly a bi-directional system needs more memory at encoder and decoder and causes a greater coding delay. The simple profile of MPEG dispenses with B pictures to make a complexity saving, but will not be able to achieve such good quality at a given bit rate.

Figure 6.4.1



The use of bi-directional coding becomes clear upon considering the boundaries of a moving object. At the leading edge of an object, the background is being concealed, consequently less data are required in future frames. Background data are already known at the decoder and forward prediction is the most efficient. However, at the trailing edge the background is being revealed and background data will not be available in past frames. However, more of the background will be available in future frames so it is more efficient to use data from a later frame and discard some of it. In practice a coder may perform both forward and backward codings but only send the one which results in the smallest difference data.

6.5 Data types

The result of an MPEG compression of a single video signal will be a number of different data types as follows:

1/ The decoder must be told the profile, the level and the size of the image in macroblocks, although this need not be sent very often.

2/ The sequence of I,P and B frames must be specified so that the decoder knows how to interpret and reorder the transmitted picture data. The sequence parameters consist of one value specifying the number of pictures in the GOP and another specifying the spacing of P pictures.

3/ The time to which a picture relates is indicated by sending a form of time code.

4/ The weighting used in the compression process is variable and it may be necessary to send the weighting matrix values if a default value has not been used.

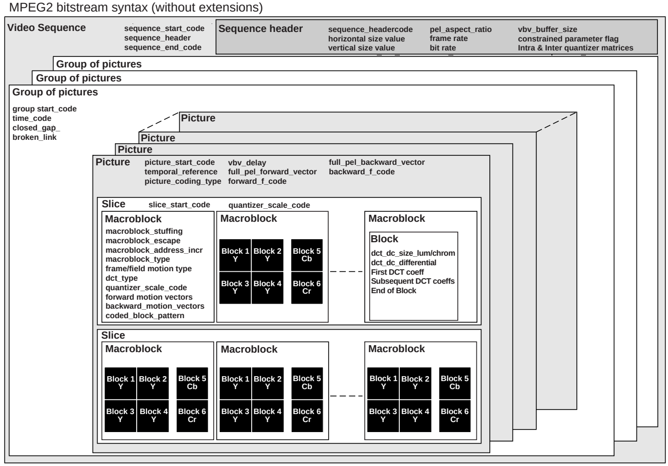
5/ The data allowing three types of picture to be decoded.

6.6 MPEG bitstream structure

The structure of a single MPEG bitstream begins with a Video Sequence as shown in Fig.6.6.1. A Video Sequence contains a number of GOPs. The sequence header specifies the profile and level, the picture size and the frame rate. Optionally the weighting matrices used in the coder are carried.

Figure 6.6.1

MPEG2 bitstream syntax (without extensions)



Within the VideoSequence a number of GOPs are sent sequentially. Each GOP begins with a GOP header containing the time code stamp of the first picture.

Within each GOP, the first picture is always an I picture, whereas subsequent pictures could be P or B. The picture header contains a code specifying the type of picture and its time location within the GOP. As pictures are sent out of sequence for bi-directional coding the picture header is an important step in getting the display order correct.

Each picture is divided in two dimensions into 8 x 8 pixel blocks. As the chroma has been subsampled by a factor of two, for every four luminance blocks there is only one R-Y block and one B-Y block. A set of four luminance blocks in a square and the two corresponding colour difference blocks is the picture area carried in a macroblock. A macroblock is the unit of picture information to which one motion vector applies and so there will be a two-dimensional vector sent in each macroblock, except in I pictures where there are by definition no vectors. In P pictures the vectors are forward only, whereas in B pictures it is necessary to specify whether the vector is forward or backward or whether a combination of the two is used.

A row of macroblocks across the screen is called a slice. It is not essentialfor a slice to extend across the full width of a screen, although the edge of the picture myst conicide with the edge of a slice. In the case where the full width of the screen is occupied by a slice in 601 luminance there are 720 pixels per line so there will be 45 macroblocks per slice as 45 x 16 = 720. Each slice carries its own address in the picture for synchronising purposes. A number of slices makes up a whole picture. For example in the Low Profile picture there is a maximum of 288 lines. There would be 18 slices in the picture as 18 x 16 = 288.

6.7 Systems layer

The structure described so far is known as an elementary stream as it carries only one channel of video. Other elementary streams will carry the associated audio. The next layer of coding concerns the multiplexing of the video and audio of several programs. This is known as the Systems layer.

A Program Stream consists of the video and audio for one service having one common clock. Program Streams have variable length data packets which are relatively unprotected and so are only suitable for error free transmission or recording systems.

A Transport Stream is an alternative to a Program Stream. This uses fixed length packets with error correction. Fig.6.7.1 shows a Transport Stream packet which consists of 187 bytes of data. Although not part of MPEG, it is common to protect such a packet with 16 bytes of redundancy. The Systems layer provides program allocation tables which list all of the elementary streams being carried and the services they relate to. As each service may have a different clock source, independent clocking is provided for. The use of variable length coding means that the data buffer contents at decoder and encoder rise and fall dynamically effectively changing the encode and decode delays. The systems layer uses time stamps embedded in the data to ensure that decoded audio and video signals are output in sync. If conditional access is used, the system layer provides control of scrambling and encryption.

Figure 6.7.1

Sync	187 Bytes Data	16 Bytes Redundancy

Compression is a technology which is important because it directly affects the economics of many audio and video production and transmission processes. The applications have increased recently with the availability of international standards such as MPEG and supporting chip sets. In this guide the subject is explained without unnecessary mathematics. In addition to the theory, practical advice is given about getting the best out of compression systems.

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