



1988



TURKISH REPUBLIC OF NORTHERN CYPRUS

NEAR EAST UNIVERSITY

FACULTY OF ENGINEERING

DEPARTMENT OF ELECTRICAL & ELECTRONIC
ENGINEERING

DEGREE OF BSc

EE 400 GRADUATION PROJECT

DIGITAL TELEVISION PRINCIPLES

SUBMITTED BY : Mr.FATIH KIZILTAN
SUBMITTED TO : PROF.DR. K. HALIL ISMAILOV

NICOSIA-2000



CONTENTS

CHAPTER 1 : TELEVISION HISTORY

1.1	HISTORY OF TELEVISION.....	1
1.1.1	PREFACE.....	1
1.2	THE FIRST BROADCAST.....	2
1.3	COLOUR TELEVISION.....	4
1.3.1	ON THE PRIMARIES.....	5
1.3.2	COLOUR TELEVISION TRANSMISSION.....	7
1.4	DIGITAL TELEVISION.....	9
1.4.1	HOW DIGITAL SAMPLING WORKS.....	10

CHAPTER 2 : TELEVISION PRINCIPLES

2.1	BASIC TV PRINCIPLES.....	11
2.2	MONOCHROME TELEVISION TRANSMITTER	13
2.3	TELEVISION BROADCAST STANDARDS.....	13
2.4	DTV BY FIGURES.....	15
2.5	WHAT IS DIGITAL TV?.....	28
2.6	WHAT IS DTV (WHAT ELSE IT DOES) ?.....	29

CHAPTER 3 : THE NEW TELEVISIONS

3.1	DIGITAL TELEVISION : MAKING IT WORK.....	30
3.1.1	BEHIND THE LENS.....	30
3.1.2	THROUGH THE CABLES.....	32
3.2	DIGITAL EQUIPMENT FOR NTSC VIDEO.....	33
3.3	THE NEW TELEVISION.....	34
3.4	COMPRESSION IN DIGITAL TV STUDIOS.....	36
3.5	THE SPLICING OPERATION.....	37
3.6	DIGITAL TELEVISION BROADCAST.....	39
3.7	THE FULL-PRODUCTION STATION.....	40
3.8	THE PASS-THRU STATION.....	41
3.9	GLOBAL HDTV TRENDS.....	42
3.10	THE DIGITAL TELEVISION STANDARDS.....	42
3.11	VIDEO COMPRESSION AND DECOMPRESSION.....	44
3.12	DTV STANDARDS.....	44
3.12.1	DTV: JUST THE FACT.....	46
3.13	FREQUENTLY ASKED QUESTIONS.....	47
3.14	STANDARDS.....	48
3.15	UNANIMOUSLY RECOMMENDS.....	49

CHAPTER 4 : A-D & D-A CONVERSION

4.1	DIGITAL TECHMQUES IN DOMESTIC RECEIVERS.....	53
4.2	A-D & D-A CONVERSION AND ITS EFFECT ON TV SIGNALS.....	55
4.3	A-D CONVERSION.....	55
4.4	THE COMPO SITE VIDEO SIGNAL.....	56
4.4.1	HORIZONTAL BLANKING TIME.....	57
4.4.2	VERTICAL BLANKING TIME.....	57
4.4.3	RF TRANSMISSION OF THE COMPOSITE VIDEO.....	58
4.5	OVERVIEW OF THE VIDEO COMPRESSION.....	59
4.6	VIDEO PRE-PROCESSING.....	60
4.6.1	VIDEO COMPRESSION FORMATS.....	60
4.7	D-A CONVERSION.....	61
4.8	SAMPLING AND FREQUENCY CHARACTERISTICS.....	61
4.8.1	INPUT OUTPUT TRANSFER CHARACTERISTICS.....	64
4.9	METHODS OF QUANTISING ERRORS.....	64
4.10	METHODS USED FOR A-D & D-A CONVERSION.....	67
4.10.1	A-D CONVERSION TECHNIQUES.....	67
4.10.2	D-A CONVERSION TECHNIQUES.....	72

CHAPTER 5 : COLOUR TELEVISION

5.1	COLOUR TELEVISION TRANSMISSION AND RECEPTION.....	74
5.1.1	COLOUR TELEVISION TRANSMITTER.....	74
5.1.2	COLOUR CAMERA.....	74
5.1.3	COLOUR ENCODING.....	75
5.1.4	LUMINANCE SIGNAL.....	75
5.1.5	CHROMINANCE SIGNAL.....	76
5.1.6	COLUR BRUST.....	77
5.1.6	SCANNING FREQUENCIES FOR COLOUR TRANSMISSION.....	78
5.1.7	FREQUENCY INTERLACING.....	78
5.2	COLOUR TELEVISION RECEIVER.....	79
5.3	DIGITAL FILTERING OF TELEVISION SIGNALS.....	80
5.4	THE TWO DIMENSIONAL FOURIER TRANSFORM.....	80
5.5	THE THREE DIMENSIONAL FOURIER TRANSFORM.....	85
5.6	TUNER.....	86

ACKNOWLEDGEMENT

One of the pleasures of authorship is acknowledging the many people whose names may not appear on the cover but without whose efforts, cooperation and encouragement a work of this scope never have been completed

I am much indebted to

Prof. Dr. Khalil ISMAILOV

For this kind supervision generous advice clarifying suggestions and support during the whole scope of this work.

In the end, I would like to thank my family for their support during my education.

CHAPTER 1: TELEVISION HISTORY

1.1 HISTORY OF TELEVISION

1.1.1 PREFACE

Did you know there are more television sets in the world than there are telephones? Even the television professionals find it hard to believe. However the statistics prove it to be true; according to official figures from the International Telecommunication Union there were 565 million telephones in 1983, and 600 million television sets. Other figures are just as impressive: in Belgium, from 1967 to 1982, the average time spent watching television by children from 10 to 13 years, increased from 82 to 146 minutes per day. Stupefying in every sense of the word.

Our senses are assailed every day by the attraction of the visual message. Its all-pervasiveness and instantaneity are finally tuned to our way of thinking, whether we be hard-pressed or lazy. We expect from it effortless pleasure and hot news. A Chinese proverb tells us a picture is worth ten thousand words.

But the setup fiction takes its toll and we thirst for more. Images pour over us in a never ending torrent. Television has already modified our social behaviour. It fosters, for example, our taste for things visual the impact of the picture and its colours. It encourages in us a yearning for the big spectacle the razzmatazz and the forthright declaration. The effect can be seen in the way we react one to another and in the world of advertising. But television cannot yet be said to have enriched our civilisation. For that to happen it must become interactive, so the viewers may cease to be just absorbers.

In the flood of images from the silver screen the less good accompanies the best, just as in the cinema or in literature. The factor which distinguishes television from the cinema and books, however, is that the full quality range, down to the very worst, is offered to us round the clock, in our own homes. Unless we take particular care to preserve our sense of values, we let it all soak in. We have not yet become "diet consciences", as regards our intake of television fare, although this is becoming increasingly necessary as the number of channels available to the public steadily increases. Without this self-control our perception becomes blurred and the lasting impression we have ceases to be governed by a strict process of deliberate reflection.

Television cannot, on its own, serve as an instrument of culture. It has, to be appreciated that it is not well-suited for detailed analysis or in-depth investigation. The way it operates and its hi-tech infrastructure are such that it cannot do justice to the words of the poet. How fortunate that there are other media for that.

Television aims at our most immediate perception. Pictures to see almost to feel. It is a medium for multiple contact; it sets the whole world before us. It offers us entertainment games, sports and more serious programmes news. Eurovision was created for that very purpose. Television offers something of everything, and each viewer can pick and choose whatever he or she finds the most enlightening.

The cultivation of a diet-conscious viewing public will be easier if the viewers can become more familiar with the media and how they work if we can do away with the “telly” myth. Some attempts have already been made. The 50th anniversary of television affords an excellent opportunity to contribute to this movement and, by showing equipment and drawings, we hope to enlighten our visitors about the workings of this most consumed of consumer technologies.

This brochure will bring them closer still to understanding what happens behind the television screen. We have made every effort to make the essential features of television understandable to visitors without specialised scientific knowledge. We have restricted ourselves to aspects likely to be of particular interest to viewers, concentrating on systems or organisations which the public know to exist, but of which they have only a very meagre understanding.

We hope, therefore, that this brochure, like the exhibition it accompanies, will serve to bring the public and the media a little closer.

1.2 THE FIRST BROADCASTS

March 1935. A television service was started in Berlin (180 lines/frame, 25 frames/second). Pictures were produced on film and then scanned using a rotating disk. Electronic cameras were developed in 1936, in time for the Berlin Olympic Games.

Figure 1.1



Figure 1.2

2 November 1935. Television broadcasting began in Paris, again using a mechanical system for picture analysis (180 lines/frame, 25 frames/second)

That same year, spurred on by the work of Schoenberg, the EMI company in England developed a fully electronic television system with 405-line definition, 25 frames/second, and interlace.

The Marconi Company provided the necessary support regarding the development of transmitters.

The British government authorised the use of this standard, along with that of Baird, for the television service launched by the BBC in London in November 1936 (the Baird system used mechanical scanning, 240 lines, 25 frames/second and no interlace). The two systems were used in turn, during alternate weeks.



Figure 1.3

The 240-line mechanical scanning system pushed the equipment to the limit and suffered from poor sensitivity. The balance thus swung in favour of the all-electronic 405-line system which was finally adopted in England in February 1937.

The same year, France introduced a 455-line all-electronic system.

Germany followed suit with 441 lines, and this standard was also adopted by Italy. The iconoscope was triumphant. It was sensitive enough to allow outdoor shooting. It was by means of a monster no less than 2.2 m long, the television canon, (in fact an iconoscope camera built by Telefunken) that the people of Berlin and Leipzig were able to see pictures from the Berlin Olympic Games. Viewing rooms, known as Fernsehstuben were built for the purpose.

Equipment that was easier to manipulate was used by the BBC for the coronation of His Majesty King George VI in 1937 and, the following year, for the Epsom Derby. Public interest was aroused. From 1937 to 1939 receiver sales in London soared from 2 000 to 20 000.

Research in the United States (Zworykin and the RCA company) bore fruit at about the same time. The first public television service was inaugurated in New York in 1939 with a 340-line system operating at 30 frames/second.

Two years later, the United States adopted a 525-line 60 frames/second standard.

The first transmitters were installed in the capital cities (London, Paris, Berlin, Rome, New York) and only a small proportion of the population of each country was therefore able to benefit. Plans were made to cover other regions.

The War stopped the expansion of television in Europe. However the intensive research into electronic systems during the War, and the practical experience it gave, led to enhancements of television technology. Work on radar screens, for example, benefited cathode-ray tube design; circuits able to operate at higher frequencies were developed.

When the War was over, broadcasts resumed in the national standards fixed previously: 405 lines in England, 441 lines in Germany and Italy, 455 lines in France. Research showed the advantages of higher picture definition, and systems with more than 1000 lines were investigated. The 819-line standard emerged in France.

It was not until 1952 that a single standard (625 lines, 50 frames/second) was proposed, and progressively adopted, for use throughout Europe. Modern television was born.

1.3 COLOUR TELEVISION

The physical concept allowing the reproduction of colour is metamerism: the effect of any colour on the human eye can be reproduced by combining the effects of three other colours, known as primaries. Three simple colours can constitute primaries if none can be achieved with a combination of the other two.



Figure 1.4

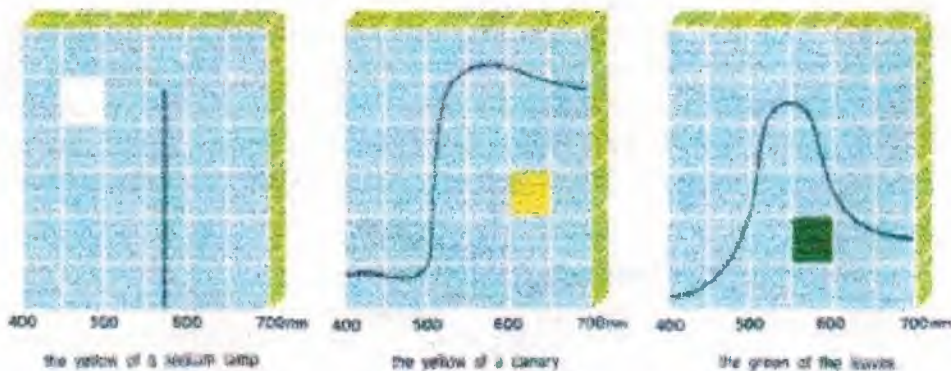


figure 1.5

In practice, we use red, green and blue, since this trio can match the greatest range of natural colours. In other words, we can define any colour by indicating the proportion of red, green and blue which have to be used for its reconstitution.

In physical terms, a colour corresponds to a series of electromagnetic radiations of different wavelengths. As primaries, we can select radiations of a single wavelength (monochromatic) or groups of several different wavelengths (polychromatic). The primaries used in modern television set are quasi-monochromatic.

The primaries used in television result from a compromise between the range of colours to be reproduced and what can in fact be manufactured with the available luminescent materials.

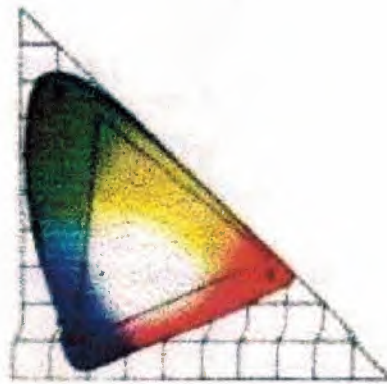


Figure 1.6

1.3.1 ON THE PRIMARIES

The triple nature of colour derives from a characteristic of human physiology, since colour vision depends on the absorption of light by the retina of eye, by three different photosensitive pigments.

The practical experience showing that three colours can, when brought together, equal a fourth, indicates that this principle can serve as the basis for colour reproduction. Experience shows also that the greater the differences between the three primaries, the greater will be the variety of colours that can be reproduced. That is why the primaries in traditional use are very saturated red, green and blue. These are the “analysis primaries”.

To transmit the corresponding electrical signals in the best possible way it is desirable to combine them to give three different signals. One represents the values of picture brightness (luminance) and the other two, taken together, represent the purely chromatic values of the picture. These are the “transmission primaries”.

In the camera, the colour is decomposed into primaries by means of prisms. Each primary illuminates a separate tube and therefore produces its own signal.



Figure 1.7

In receivers, the colour is reconstituted using a large number of luminescent spots arranged in red-green-blue triplets. The spots are close enough so that, from a reasonable viewing distance, a triplet appears as a single information source. In other words, the eye sees each triplet as one picture element.

The number of discernible colours, with television primaries, is around ten thousand.

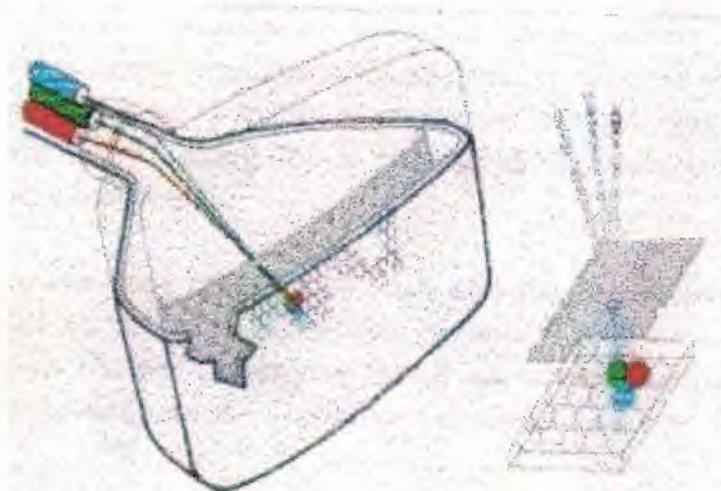


Figure 1.8

The red, green and blue primaries are only used in the camera and receiver. Between these, the constraints imposed by practical transmission systems are such that they must be cunningly converted into a different form, as we shall see.

1.3.2 COLOUR TELEVISION TRANSMISSION

The first practical demonstration of colour television was given back in 1928 by Baird; he used mechanical scanning with a Nipkow disk having three spirals, one for each primary. Each spiral was provided with a separate set of colour filters. In 1929, H.E. Ives and his colleagues at Bell Telephone Laboratories presented a system using a single spiral through the holes of which the light from three coloured sources was passed; the signal for each primary was then sent over a separate circuit.

As 1940 approached, only cathode-ray tubes were envisaged, at least for displaying the received picture.

In 1938, Georges Valensi, in France, proposed the principle of dual compatibility: programmes transmitted in colour should also be received by black and white receivers; programmes transmitted in black and white should also be seen as black and white by colour receivers.

In 1940, Peter Goldmark, of CBS in and the United States, demonstrated a sequential system for transmitting three primaries obtained using three colour filters placed in the light path before scanning.

The system was barely practicable. In addition, it required three times as large a range of frequencies (i.e. band-width) as compared to black-and-white transmission. Other researchers were looking for a non-mechanical solution which would not require such a large bandwidth.

In 1953, simultaneous research at RCA and the Hazeltine laboratories, in the United States, led to the first compatible system. This was standardised by the National Television System Committee, made up of television experts working in industry, and is known as the National Television Systems Committee (NTSC) system.

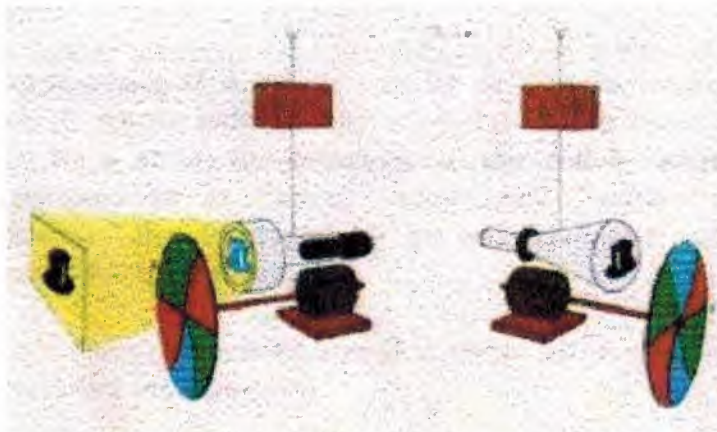


Figure 1.9



Figure 1. 10

The signal is no longer transmitted in the form primaries, but as a combination of these primaries. This provides a “luminance” signal Y which can be used by black and white receivers. The colour information is combined to constitute a single “chrominance” signal C . The Y and C signal are brought together for transmission.

The isolation of the chrominance and luminance information in the transmitted signal also allows bandwidth saving to be made. In effect, the bandwidth of the chrominance information can be made much smaller than for the luminance because the acuity of the human eye is lower for changes of colour than it is for changes of brightness.

The visual appearance of a colour can be defined in terms of three physical parameters for which words exist in our everyday vocabulary: the hue (which is generally indicated by a noun) the saturation (indicated by an adjective, with the extreme values referred to as “pure” colour and “washed-out” colours) the brightness or lightness (also indicated by an adjective, the extremes here being “bright” colours and “dark” colours).

The compatible colour television signal is made up such a way as to ensure that these parameters are incorporated.

The amplitude of the C signal corresponds to the colour saturation, and its phase corresponds to the hue.

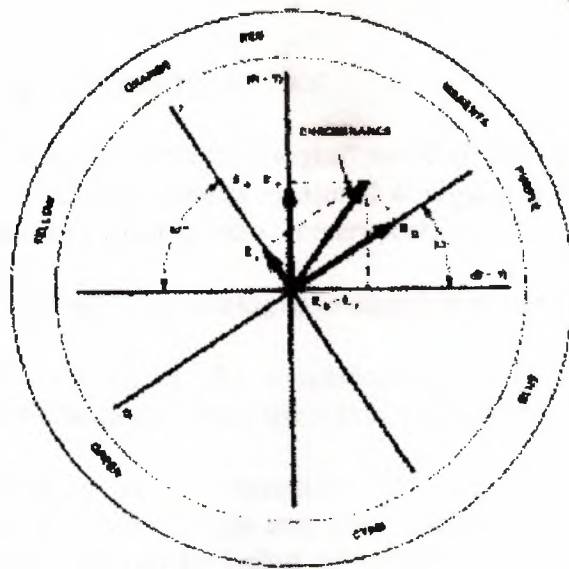


Figure 1.11

The system was launched in the United States as early as 1954

The first American equipment was very susceptible to hue errors caused by certain transmission conditions. European researchers tried to develop a more robust signal, less sensitive to phase distortions.

In 1961, Henri de France, put forward the SECAM system (Sequentiel Couleur a Memoire) in which the two chrominance components are transmitted in sequence, line after line, using frequency modulation. In the receiver, the information carried in each line is memorised until the next line has arrived, and then the two are processed together to give the complete colour information for each line.

In 1963, Dr. Walter Bruch, in Germany, proposed a variant of the NTSC system, known as PAL (Phase Alternation by Line). It differs from the NTSC system by the transmission of one of the chrominance components in opposite phase on successive lines, thus compensating the phase errors automatically.

Both solutions found application in the colour television services launched in 1967 in England, Germany and France, successively.

1.4 DIGITAL TELEVISION

The values of the brightness or colour of picture elements along a television line can be represented by a series of numbers. If these are expressed in base 2, each value can be transformed into a sequence of electrical pulses.

1.4.1 How digital sampling works

The operation which converts from the "analogue" world to the "digital" world comprises two stages: sampling in which the value is measured at regular intervals and quantification in which each measurement is converted into a binary number.

These operations are carried out by an analogue to digital converter.

The series of "1" and "0"s obtained after quantification can be modified (i.e. coded) to counteract more effectively the disturbances the signal will meet during transmission.

Digital television technology is an extension of computer and image processing technology. Advantages are easy storage and great scope for image processing. Each picture element is isolated and can be called up independently according to varied and complex. Since the signal has only two possible values (0 or 1), detection is based on the presence or absence of the signal. Hence the possibility of regenerating it. Advantage: the signal can be preserved during successive recordings or on noisy transmission paths.

This technique is already in wide-spread use for special effects on existing images. It lies at the root of computerised image synthesis systems.

CHAPTER 2 : TELEVISION PRINCIPLES

2.1 BASIC TELEVISION PRINCIPLES

The word ,television comes from the Greek word *Tele* (meaning distant) and the Latin word *vision* (meaning sight). Therefore, television simply means to see from a distance. In its simplest form, television is the process of converting images (either stationary or in motion) to electrical signals and, then, transmitting those signals to a distant receiver, where they are converted back to images that can be perceived with the human eye. Thus, television is a system in which images are transmitted from a central location and then received at distant receivers, where they are reproduced in their original form.

The idea of transmitting images or pictures was first ezperimented with in the 1880s when Paul Nipkow, a German scientist, conducted ezperiments using revolving disks placed between a powerful light source and the subject. A spiral row of holes was punched in the disk. which permitted light to scan the subject from top to bottom. After one complete revolution of the disk, the entire subject had been scanned. Light retlected from the subject was directed to a light-sensitive cell, producing current that was proportional in intensity to the retlected light. The fluctuating current operated a neon lamp, which gave off light in exact proportion to that retlected from the subject. A second disk exactly like the one in the transmitter was used in the receiver and the two disks revolved in exact synchronization. The second disk was placed between the neon lamp and the eye of the observer, who, thus, saw a reproduction of the subject. The images reproduced with Nipkow's contraption were barely recognizable, although his scanning and synchronization principle are still used today.

In 1925, C. Francis Jenkins in the United States and John L. Baird in England, using scanning disks connected to vacuum-tube amplifiers and photoelectric cells, were able to reproduce images that were recognizable. Although still of poor quality. Scientists worked for several years trying to develop effective mechanical scanning disks that, with improved mirrors and lenses and a more intense light source, would improve the equal of the reproduced image. However, in 1933, Radio Corporation of (RCA) announced a television system, developed by Vladimir K. Zworykin, that used an electronic scanning technique. Zworykin's system required no mechanical moving parts is essentially the system used today.

In 1941, commercial broadcasting of monochrome (black and white) television signal began in the United States. In 1945, the FCC assigned 13 VHF television low-band channels, I to 6 (44 MHz to 88 MHz), and 7 high-band channels 7 to 13 (174MHz to 216 MHz). However, in 1948 it was found that channel I (44 MHz) to 59 MHz caused interference problems; consequently, this channel was reassigned mobile radio services. In 1952, UHF channels 14 to 83 (470 MHz to 890 MHz) were the FCC assigned by the FCC to provide even more television stations. In 1974, the FCC reassigned to cellular telephone frequency bands at 825 MHz to 845 MHz and 870 MH2 to 890 MHz, thus, eliminating UHF channels 73 to 83 (however, existing licenses are renewable).

Table 2.1 shows a complete list of the FCC channel and frequency Corporation proposed the method of inter-carrier sound transmission for television broadcasting that is used today. In 1949, experiments began with color transmission, and in 1953 the FCC, adopted the National Television Systems Committee (NTSC) system for color television broadcasting. Which is also still used today.

TABLE 2.1 FCC CHANNEL & FREQUENCY ASSIGNMENTS

CHANNEL NUMBER	FREQUENCY BAND(MHz)	CHANNEL NUMBER	FREQUENCY BAND(MHz)	CHANNEL NUMBER	FREQUENCY BAND(MHz)
1*	44-50	29	560-566	57	728-734
2	54-60	30	566-572	58	734-740
3	60-66	31	572-578	59	740-746
4	66-72	32	578-584	60	746-752
5	76-82	33	584-590	61	752-758
6	82-88	34	590-596	62	758-764
7	174-180	35	596-602	63	764-770
8	180-186	36	602-608	64	770-776
9	186-192	37	608-614	65	776-782
10	192-198	38	614-620	66	782-788
11	198-204	39	620-626	67	788-794
12	204-210	40	626-632	68	794-800
13	210-216	41	632-638	69	800-806
14	470-476	42	638-644	70	806-812
15	476-482	43	644-650	71	812-818
16	482-488	44	650-656	72	818-824
17	488-494	45	656-662	73*	824-830
18	494-500	46	662-668	74*	830-836
19	500-506	47	668-674	75*	836-842
20	506-512	48	674-680	76*	842-848
21	512-518	49	680-686	77*	848-854
22	518-524	50	686-692	78*	854-860
23	524-530	51	692-698	79*	860-866
24	530-536	52	698-704	80*	866-872
25	536-542	53	704-710	81*	872-878
26	542-548	54	710-716	82*	878-884
27	548-554	55	716-722	83*	884-890
28	554-560	56	722-728		

* NO LONGER ASSIGNED TO TELEVISION BROADCASTING

2.2 MONOCHROME TELEVISION TRANSMITTER

Monochrome television broadcasting involves the transmission of two separate signals: an aural (sound) and a video (picture) signal. Every television transmitter broadcasts two totally separate signals for the picture and sound information. Aural transmission uses frequency modulation and video transmission uses amplitude modulation. Figure 2.1 shows a simplified block diagram for a monochrome television transmitter. It shows two totally separate transmitters (an FM transmitter for the sound information and an AM transmitter for the picture information) whose outputs are combined in a diplexer bridge and fed to a single antenna. A diplexer bridge is a network that is used to combine the outputs from two transmitters operating at different frequencies that use the same antenna system. The video information is limited to frequencies below 4 MHz and can originate from either a camera (for live transmissions), a video tape or cassette recorder, or a video disk recorder. The video switcher is used to select the desired video information source for broadcasting. The audio information is limited to frequencies below 15 kHz and can originate from either a microphone (again, only for live transmissions) from sound tracks on tape or disk recorders, or from a separate audio cassette or disk recorder. The audio mixer/switcher is used to select the appropriate audio source for broadcasting. Figure 2.1 also shows horizontal and vertical synchronizing signals, which are combined with the picture information prior to modulation. These signals are used in the receivers to synchronize the horizontal and vertical scanning rates (synchronization is discussed in detail later in the chapter).

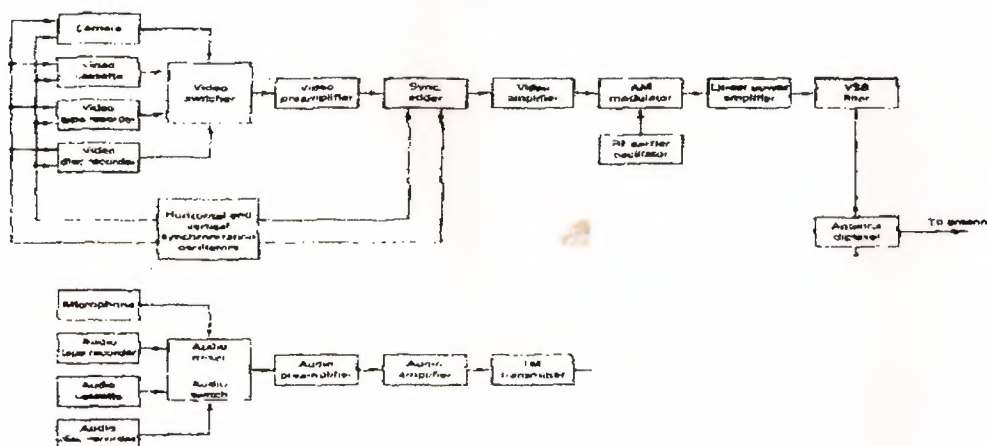


Figure 2.1

2.3 TELEVISION BROADCAST STANDARDS

Figure 2.2 shows the frequency spectrum for a standard television broadcast channel. Its total bandwidth is 6 MHz. The picture carrier is spaced 1.25 MHz above the lower limit for the channel, and the sound carrier is spaced 0.25 MHz below the

upper limit. Therefore, the picture and sound carriers are always 4.5 MHz apart. The color sub-carrier is located 3.58 MHz above the picture carrier. Commercial television broadcasting uses AM vestigial side-band transmission for the picture information. The lower side-band is 0.75 MHz wide and the upper side-band, 4 MHz. Therefore, the low video frequencies (rough outline of the image) are emphasized relative to the high video frequencies (fine detail of the image). The FM sound carrier has a bandwidth of approximately 75 kHz (± 25 -kHz deviation for 100% modulation). Both amplitude and phase modulation are used to encode the color information onto the 3.58-MHz color sub-carrier. The bandwidth and composition of the color spectrum are discussed later in the chapter. Also discussed is frequency interlacing, used to permit adding the color information without increasing the total bandwidth above 6 MHz.

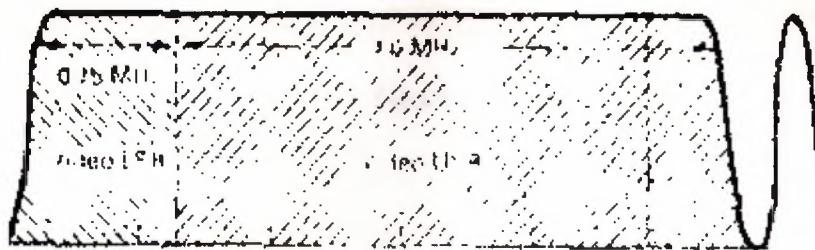


Figure 2.2

2.4 DTV BY FIGURES

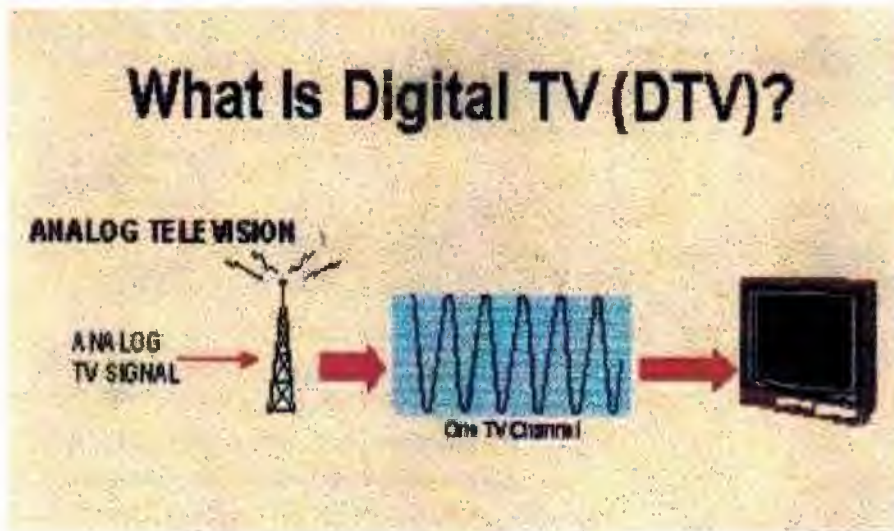


Figure 2.3

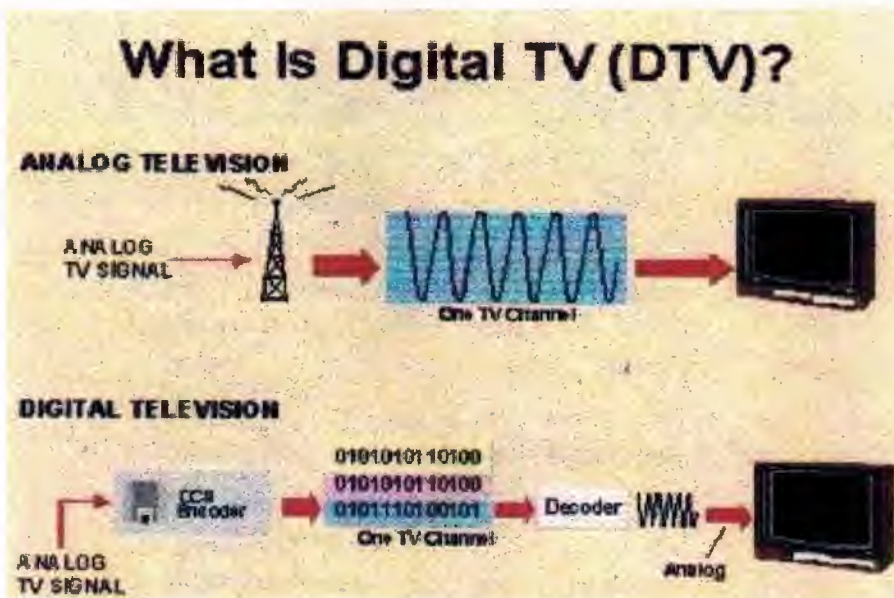


Figure 2.4

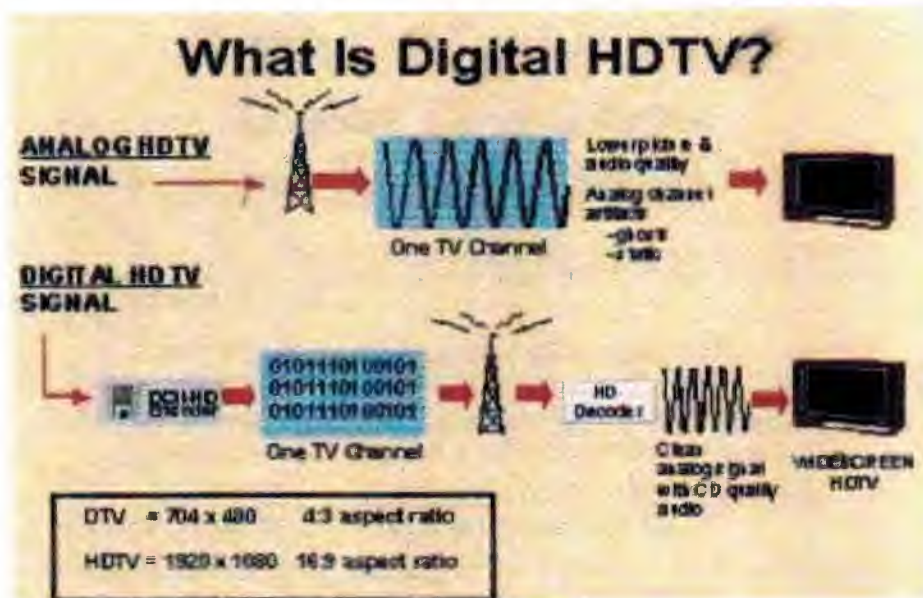


Figure 2.5

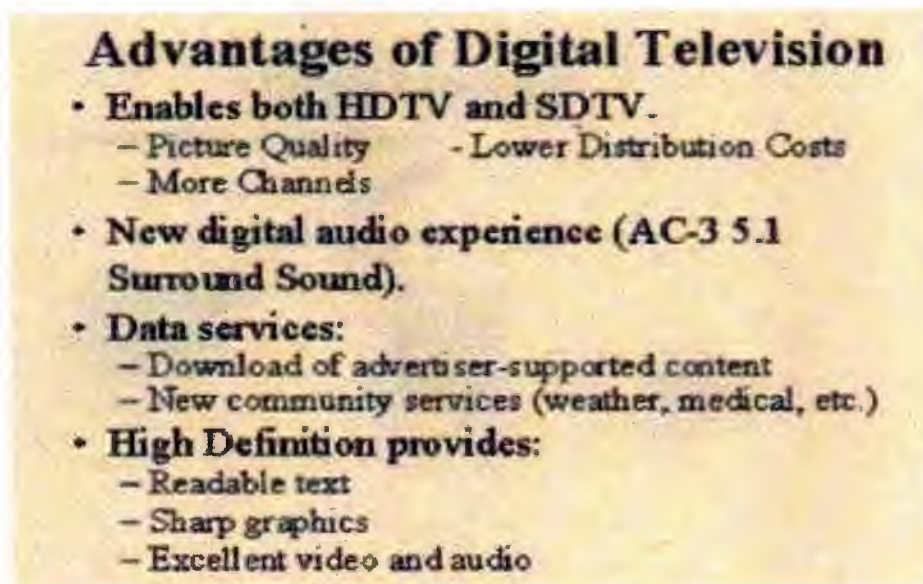


Figure 2.6

Digital TV is Simple

- Convert the analog signal to digital
- Remove the redundant information from the video frame.
- Remove the redundant information between frames
- Transmit the signal (which now consumes about 10% of the bandwidth of the analog signal).
- Convert the digital signal back to its original analog form at the receive location.

Figure 2.7



Figure 2.8

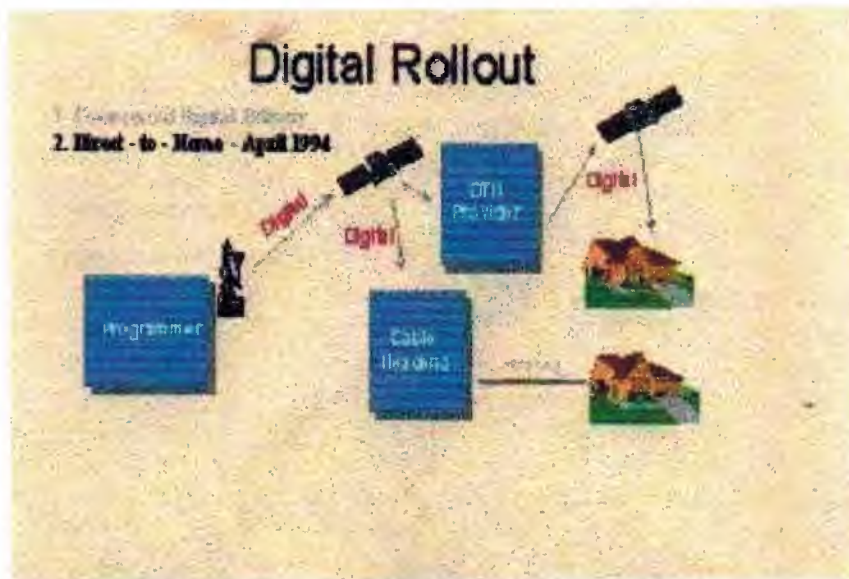


Figure 2.9

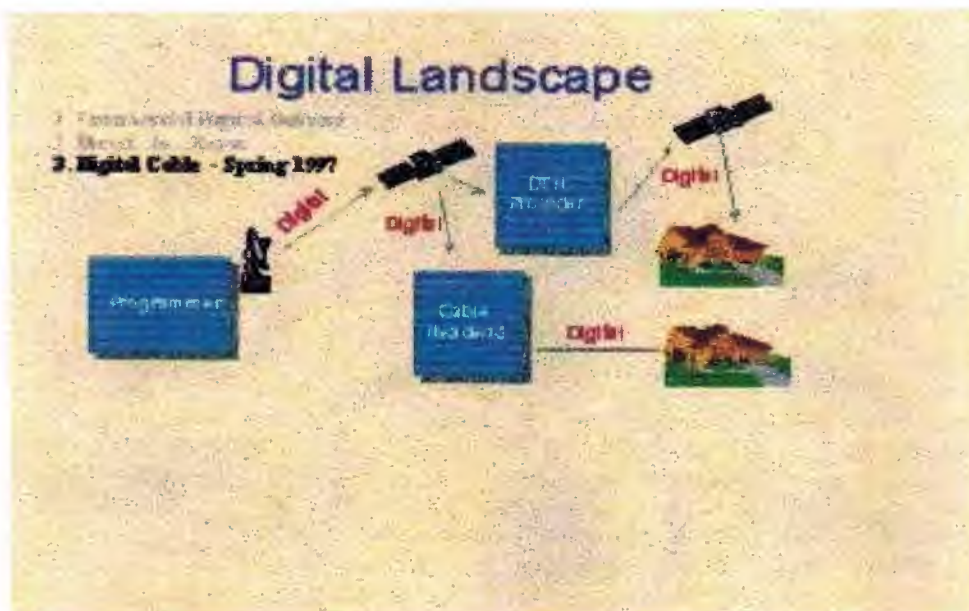


Figure 2.10



Figure 2.11

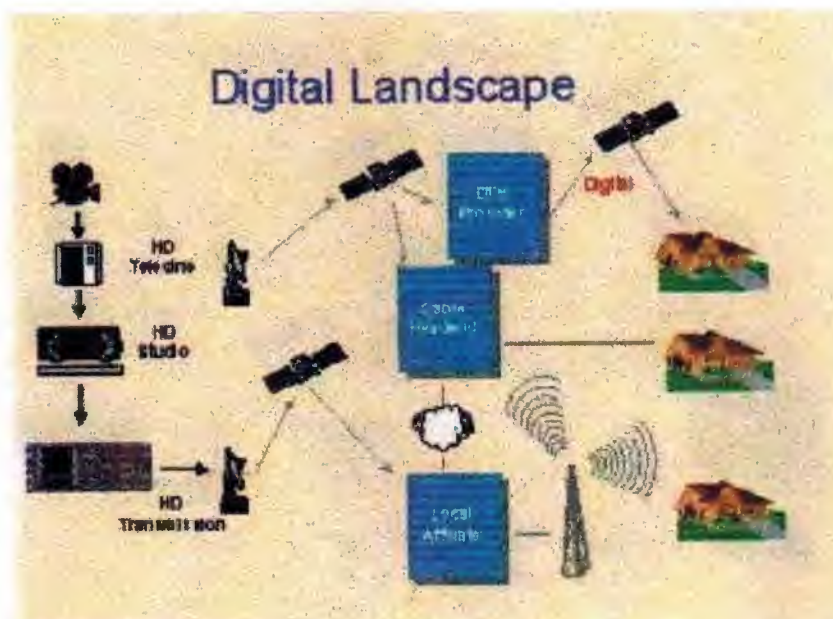


Figure 2.12

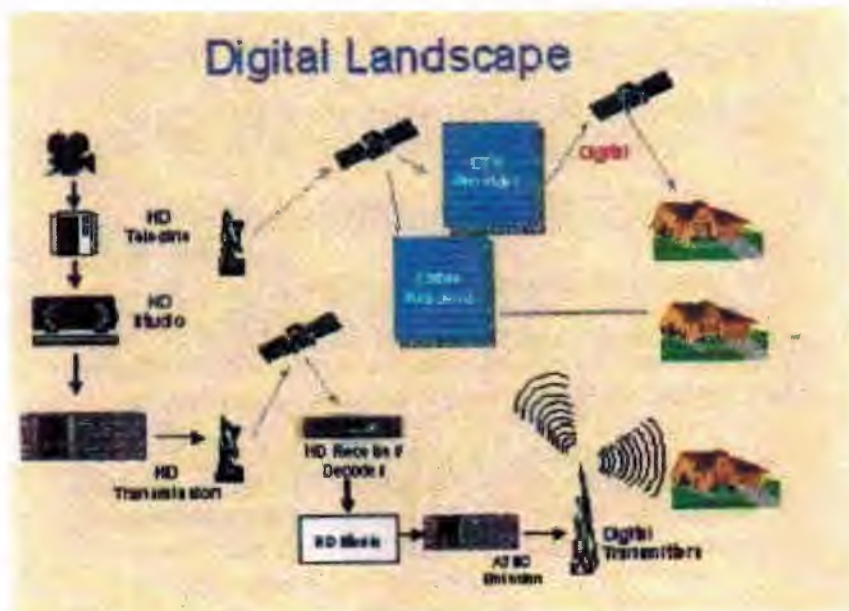


Figure 2.13

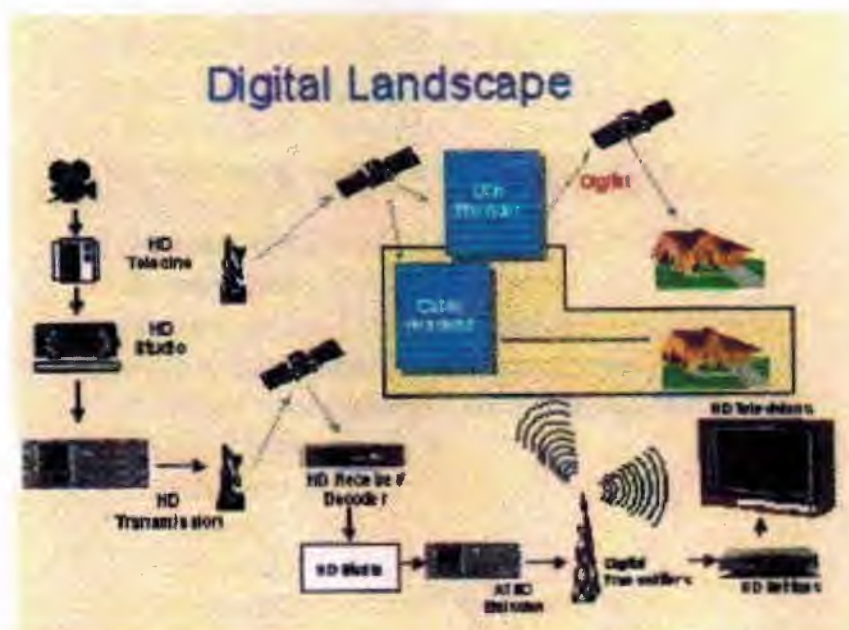


Figure 2.14

Digital Broadcast Chronology

- 1990-1991: GE develops first all digital HDTV system.
- Nov. 1991: World's first digital broadcast from KPBS, San Diego.
- 1994-1995: Grand Alliance develops commercially viable, end-end digital broadcast system on behalf of ATSC.
- Dec. 1996: FCC adopts the Grand Alliance/ATSC system as the standard for U.S. Broadcasting, and establishes the "fifth report and order" describing the mandate to broadcasters.

Figure 2.15

FCC 5th ATV Report & Order

- *Grants additional channel to existing* broadcasters for digital service
- **Requires one free TV service**
 - Equal or better resolution
 - Aired during same hours as analog service
- **Simulcasting**
 - 6th year: 50%
 - 7th year: 75%
 - 8th year: 100%
- **Broadcasters return "free" spectrum**
 - If 85% of viewers have made provisions to receive digital TV signals.

Figure 2.16

FCC 5th ATV Report & Order

- Grants additional channel to existing broadcasters for digital service
- Requires one free TV service
 - Equal or better resolution
 - Aired during same hours as analog service
- Simulcasting
 - 6th year: 50%
 - 7th year: 75%
 - 8th year: 100%
- Broadcasters return "free" spectrum
 - If 85% of viewers have made provisions to receive digital TV signals

Figure 2.17

On Air Requirements

- By May 1, 1999
 - ABC, CBS, Fox & NBC
 - Affiliates in top 10 markets
 - 40 stations, 30% of households
- By November 1, 1999
 - ABC, CBS, Fox & NBC
 - Affiliates in top 30 markets
 - Totals: 120 more stations, 53% households

Figure 2.18

On Air Requirements

- By May 1, 2002 - All Commercial
- By May 1, 2003- All Noncommercial
- May 1, 2006
 - Cease analog service
 - Repack spectrum
 - Recapture spectrum

Figure 2.19

Sales of Digital Television (CEMA)

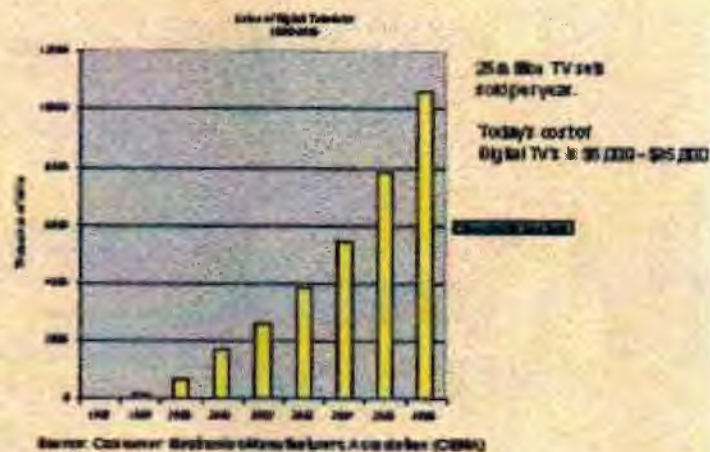


Figure 2.20

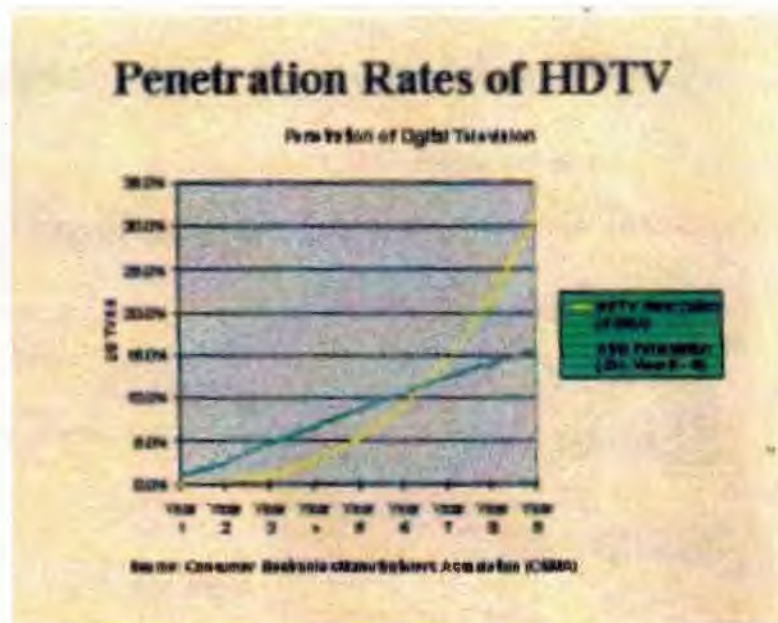


Figure 2.21

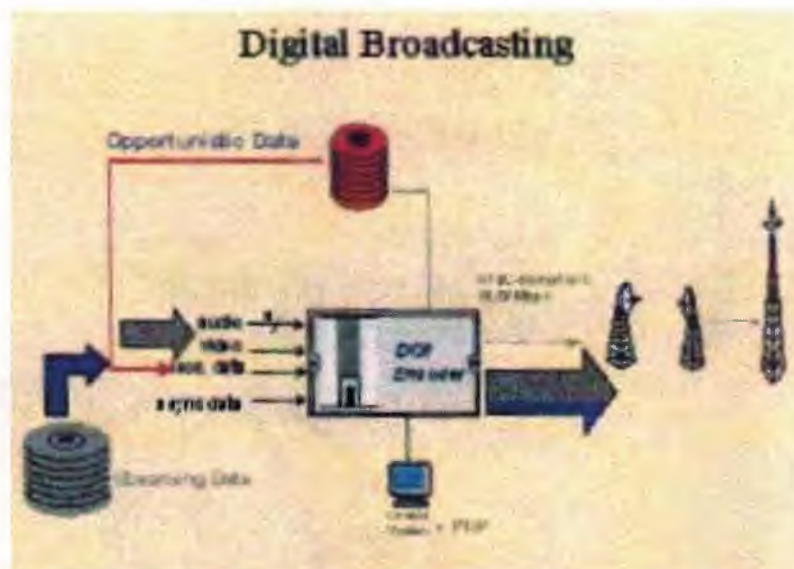


Figure 2.22



Figure 2.23

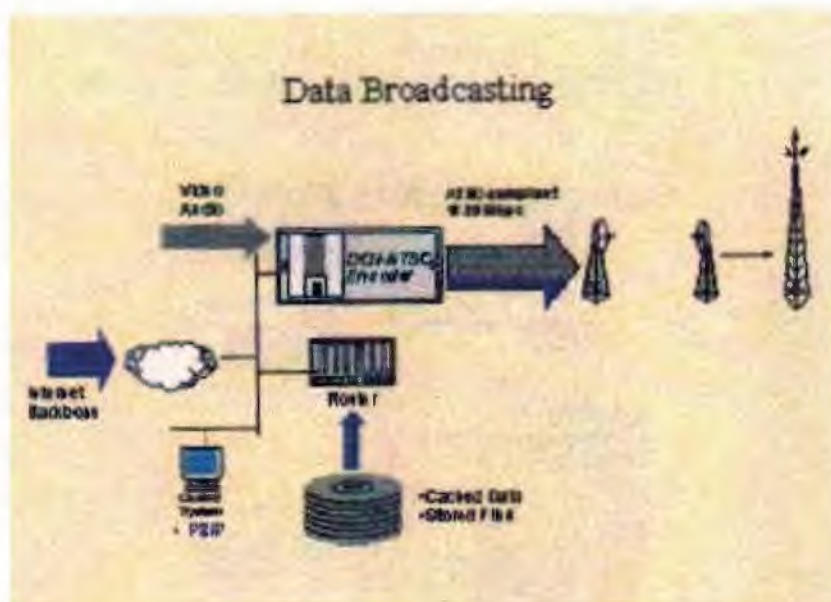


Figure 2.24

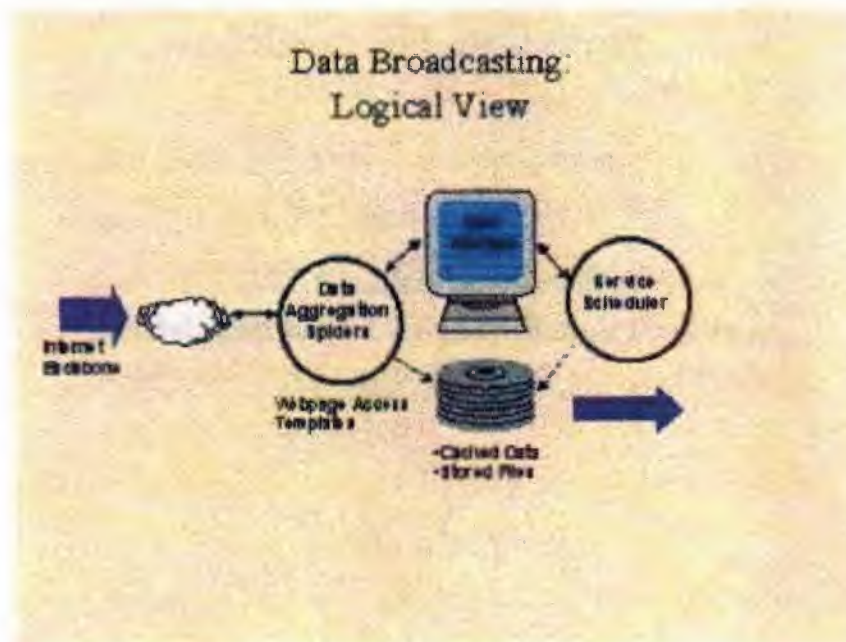


Figure 2.25



Figure 2.26

Hurdles and Challenges

- **Sub \$2,000 Digital Television Sets.**
- **Resolution of "Must-Carry" Rules.**
- **Completion of Standards (Copy Protection, Digital Interface to TV, On-Screen Display).**
- **Cable Compatibility**

Figure 2.27

Summary

- **DTV is more than just pretty pictures**
- **Multi-channel means more customer choices.**
- **Opportunistic data means more data pipes to the home.**
- **Convergence of TV and PC means opportunities for integrated applications.**

Figure 2.28

2.4 WHAT IS DIGITAL TV?



Figure 2.29

Digital Television will replace analogue TV broadcasting in the same way that CD replaced vinyl within sound reproduction largely getting rid of picture ghosting and other types of interference. As well as receiving a sharper and cleaner TV picture you will also be able to receive CD like sound quality alongside it. The more compact signal will offer the potential for more channels to be broadcast on the same bandwidth as one analogue broadcast channel. Digital Broadcasting started on November 1998 with all the current channels (TTV, BBC1, BBC2, and channels 4 and 5) digitally broadcasting with a further 16 channels being broadcast by the DBB (Digital British Broadcasting). To receive digitally broadcast channels you will either need a digital integrated TV or you will have to buy a top set receiver for your TV at the cost of around £200. Cable subscribers will be able to receive the new digital stations through their own equipment at an additional cost, but satellite subscribers will need a top set receiver and a new or upgraded dish to receive digital broadcasts.

Most households will be able to receive digital broadcasting through their TV aerials with the exception of a few, and digital broadcasting will not have immediate national coverage (around 75% of UK coverage at its launch) but with the intention of total coverage within 2 years. As to the broadcasting of analogue channels there is no immediate withdrawal of third broadcast but could be phased out within 10 to 12 years.

In Turkey: The Terrestrial Digital Broadcasting studies for both DVB-T and T-DAB have been started at the same time. Today, the concerned Administrations are preparing a concept which will include national broadcasting strategy and the timing of analogue shutdown. After this study, a plan will be developed for introduction of DVB-T.

2.6 WHAT IS DTV?

What else it does

HDTV IS 1/3 WIDER THAN NTSC

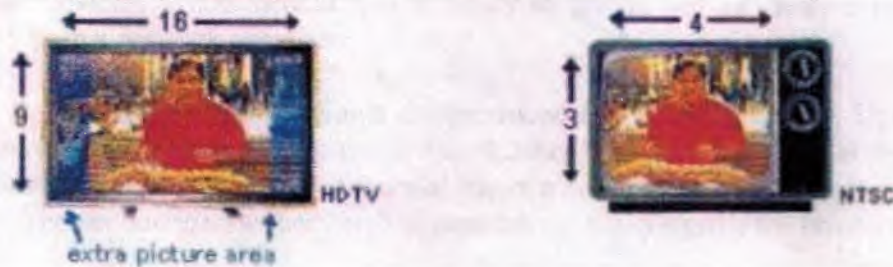


Figure 2.30

High definition isn't the only focus of the digital-TV development.

Multichannel broadcasting is a service broadcasters will offer in targeted markets by 1999. Viewers who own set-top box converters will then have access to a wider choice of TV programming. But whether or not viewers will find the extra programs worth up to thousands of dollars is a fundamental dilemma.

Datacasting is part of DTV receiving great support from public broadcasting including development of 10 short demos of what "enhanced digital broadcasting" looks like. The 10 works are featured in film festivals and in a travelling technology road show. Public broadcasting plans to sink another \$3 to 4 million in funding for digital projects in order to utilise the Datacasting capability of digital television.

Because of the educational value and widespread application of datacast programs the ma become more popular than high-definition programs in driving adoption of the technology by consumer in the future.

all this occasionally interrupted by commercials. The people before the camera (news anchors and reporters) are known in the industry as “talent”. The talent on the set is supported by makeup people, cameramen, and other assistants.

In the control room [Fig. 3.1], production engineers face a wall of video monitors. These include displays from several a cameras on the set, remote feeds from reporters (say, at a fire), graphic devices, a commercial that is ready to go on the air, and, if necessary, a satellite feed from a network provider.

Fig. 3.1 Live television programs are produced in the control room. Upon prompt from a time-keeper, the news anchor (or talent) finishes his or her sentence as the technical director fades the video from that camera and insert a commercial from a tape machine for transmission. The production swither hetps to seambllessy video signals for broadcast.

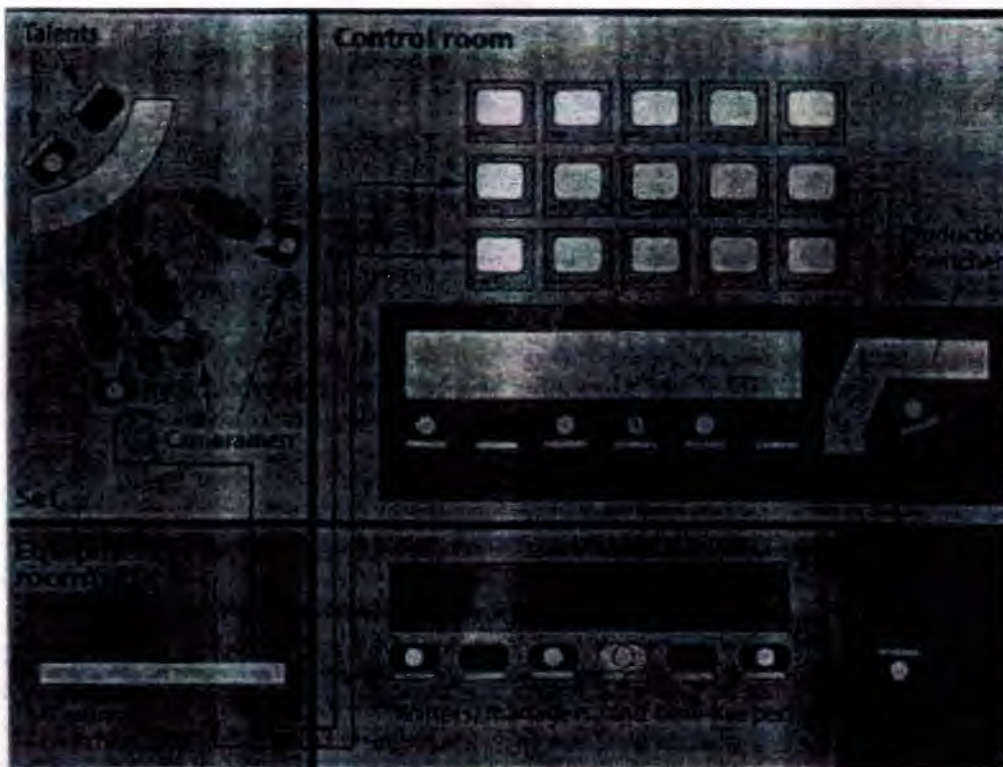


Figure 3.1

The NTSC signal contains horizontal and vertical synchronising (sync) signals so that the display device (a TV monitor or a projector) can properly scan the video onto the screen to form a picture. A television studio has a reference sync generator to which all the video in the plant must be synchronised. This synchronisation to local sync is also called ‘studio genlock,’ which is handled by a device called a frame synchroniser.

In the equipment room, a routing switcher (or router), which can have hundreds of video input and output ports, handles all the video in a TV station. At the push of a button, the routing switcher allows easy connectivity among the many video cameras, tape machines, and other studio equipment.

The production switcher is the main piece of equipment in the control room and is used to handle special effects, like video fades and wipes, and inserts commercials. Because an abrupt beginning of a commercial could annoy viewers, the immediately preceding video is slowly faded to black. (A wipe occurs when the new video gradually replaces the old video in a 'wiping' motion.)

The control room is where the producers of the show determine the flavour and the flow of the broadcast. In charge of the live broadcast is the technical director, who decides what goes on-air using a production switcher to select the appropriate video. Everyone in the control room as well as the set—including the talent—is on the intercom channel for voice communications among the crew. Sometimes, in case of some news breaking during the broadcast, the control room prompts the talent to finish a story immediately in order to convey the news to the audience. And the audience at home hears something like,

'This just in...'

Viewers following the weather reports see the weatherman every day, in all appearances standing in front of a large map as he or she makes predictions. In reality, the weatherman is standing in front of a blank green or blue wall. The video containing the map is selectively mixed with the video of the weatherman and wall so that the weatherman seems to lie in the foreground and the map in the background. This operation by the production switcher is called chroma-keying.

The time-keeper in the room keeps everyone informed of timing information, like the number of minutes or seconds left before a commercial break. Technicians in the audio room maintain the proper audio mix and audio levels.

3.2.2 THROUGH THE CABLES

In a conventional television studio NTSC signals are routed on coaxial cables from one piece of equipment to another. A typical TV station has thousands of cables connecting the equipment room, the control room, the set, and the audio room. There are in addition separate cables for intercom, computer network, and telephone connections [Fig. 3.2]. Television stations also have a sophisticated computer graphics department where artists create graphics for use in various programs.

Fig. 3.2. In a conventional television studio [top], NTSC signals are routed on coaxial cables from one piece of equipment to another: the frame synchroniser locks a remote feed to the studio sync before the video is sent to the routing switcher; the production switcher adds effects like chroma-keying, fades, and wipes.

The full-production HDTV studio [center] uses high-speed asynchronous transfer mode (ATM) routers for the routing of compressed bit-stream and has provisions for uncompressed production and storage. The studio also supports the existing studio equipment by up converting it to the HDTV format.

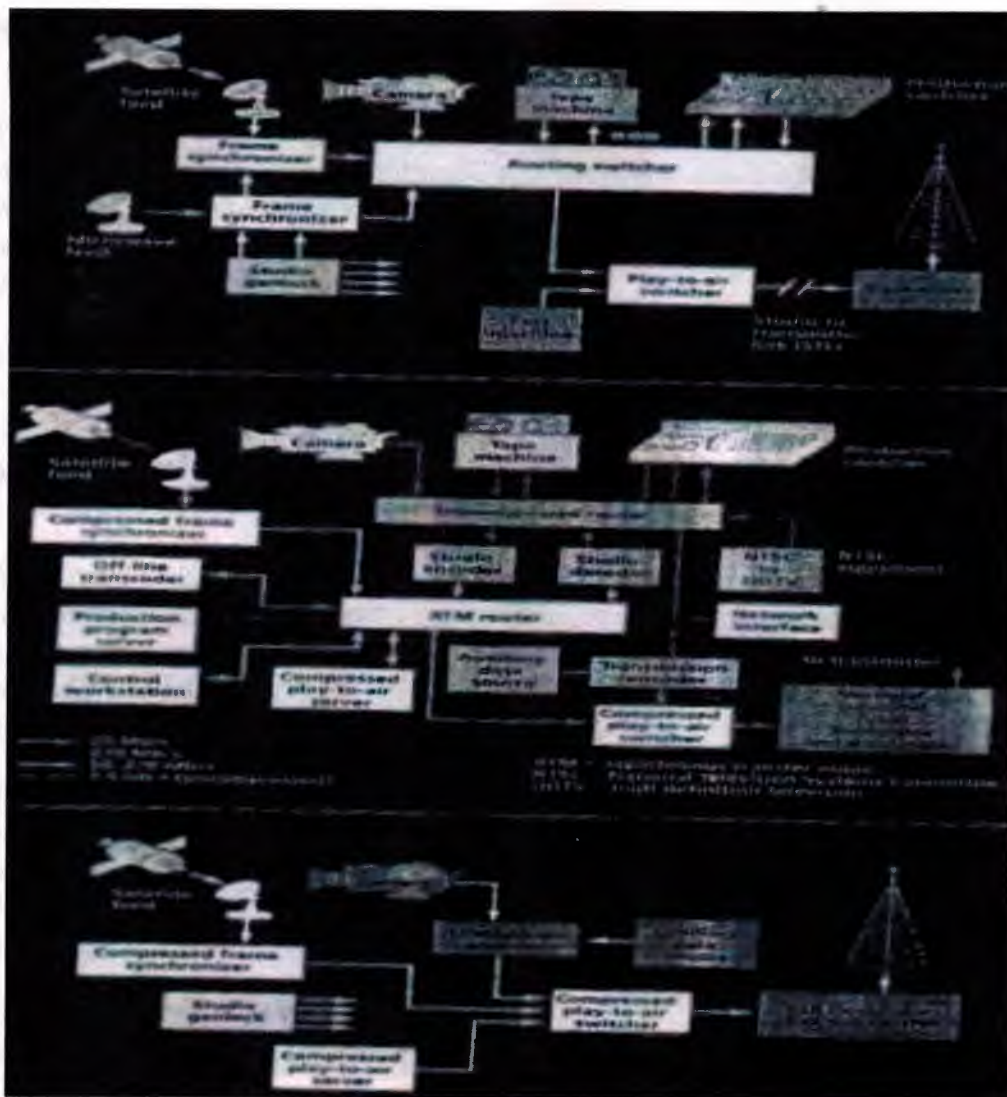


Figure 3.2

The HDTV pass-through station [bottom] takes an HDTV network feed and passed it through the studio without decoding. Local news production could be done with existing NTSC equipment and then encoded as a standard definition bit-stream. A play-to-air server stores compressed HDTV and standard-definition TV commercials and other information to be inserted by the play to air switcher.

Some TV stations have a remotely located transmitter. To transfer the NTSC program to transmitter, a studio-to-transmitter link (STL), is used. STL are usually implemented over a microwave link or a decimated land line.

3.2 DIGITAL EQUIPMENT FOR NTSC VIDEO

The raw video signal originating from a studio camera is considered a base-band component. In it the three colour components red, green, and blue are separate signals. But because early NTSC systems needed to maintain compatibility with the then black-and-white TV sets, the R, G, B colour space in video signals was converted to Y, U, V colour space.

Y stands for luminance information (lightness), U for blue minus Y, and V for red minus Y.

The Y, U, V colour space is also based on the human visual system: the eye's receptors for colour are fewer than those for luminance, and they have much less spatial resolution. So less bandwidth needs to be assigned to color difference signals U and V) than to the luminance signal. If RGB-to-YUV color space conversion is done maintaining the full bandwidth chroma-hue plus saturation-it is called a 4:4:4 sampling.

If the conversion is carried out on chroma samples every other pixel, then it is termed a 4:2:2 sampling scheme. The 4:2:2 sampling halves the chroma resolution horizontally, resulting in a 33 percent saving in bandwidth compared to a 4:4:4 sampling, yet with no perceptible loss in video quality. A 4:2:0 sampling reduces the chroma bandwidth even more, halving the overall bandwidth.

Rapid advances in digital ICs have made possible a new class of studio equipment, namely, digital equipment for NTSC video. New production switchers, routers, and tape machines all support digital component 4:2:2 video. For high-quality production purposes, there exist international standards, like, for instance, the International Telecommunication Unions Recommendation-601.

The 601 standard for broadcast video has an active resolution of 720 pixels (picture elements) by 485 lines, plus a 4:2:2 sampling scheme. There are in addition standards for parallel and serial video data interchange between equipment, standards like SMPTE-259D (for Society of Motion Pictures and Television Engineers), a 360-Mbit interface (between, say digital tape machines and routers).

Although the vast majority of TV studios still use the analogue NTSC equipment many have begun making the transition to toll digital facilities for producing NTSC programs.

3.3 THE NEW TELEVISION

The new ATSC high-definition standard defines four basic digital television formats [Table 3.1]. These formats are defined by the number of pixels per line, the number of lines per video frame, the frame repetition rate the aspect width-to-length ratio, and the frame structure (interlaced or progressive).

PICTURE FORMATS SUPPORTED ATSC STANDARD						
PICTURE SIZE	FRAME RATE				ASPECT RATIO	
1920x1080	60i	---	30p	24p	16:9	---
1280x720	---	60p	30p	24p	16:9	---
704x480	60i	60p	30p	24p	16:9	4:3
640x480	60i	60p	30p	24p	---	4:3

i=interlaced, p=progressive

Table 3.1

Interlacing is a technique the camera uses to take two snapshots of a scene within frame time. During the first scan it creates one field of video containing even-numbered lines, and during the second, it creates another containing the odd-numbered lines. This technique, which is used in NTSC video makes for reduced flicker and therefore higher brightness on the television receiver for the given frame rate and bandwidth. On the other hand most computer-generated video is scanned in progressive format, in which one frame of video contains all the lines in their proper order.

The ATSC standard includes both the interlaced and the progressive scanned video formats. Which of the two should be use is still under debate. Some computer companies argue for a lower resolution version of the progressive format, which is compatible with computer monitors Television manufacturers favour the inclusion or multiple formats, as they expect the use of interlaced formats to be initially more common.

The new-standard includes the two high-definition television (HDTV) formats. In one the 1920-pixel -by- 1080 line video is interlaced whereas in the other the 1280-pixel-by-720-line video is in progressive-scan format.

Most of the HDTV broadcast equipment emerging today from manufacturers (for example, camera and production switchers from Sony Corp, Tokyo) is designed for the 1920-by-1080 interlaced video format only. This is mainly due to a lack of progressive-scanning HDTV cam eras and monitors. (As of this writing, only Panasonic, a division of Matsushita Electric Industrial Co. Osaka, Japan, has announced the availability of progressive HDTV cameras.) Still, nearly all existing movies, which originate as 24 frame-per-second film, are in effect in a progressive format when translated into video Movies in HDTV format are likely to he the first wave of HDTV programming to reach viewers.

Among other ATSC video formats [Table 3.1 again] the progressively scanned 704pixel-by-480-line video will probably have an appeal for some television stations. Panasonic has announced a full line of production, storage, and display products supporting this format. Compared to the NTSC standard, it has a wider picture (an aspect ratio of 16:9, as compared to 4:3) and is devoid of ratio-facts typical of interlaced video (such as the line crawl that affects some scenes containing slow vertical motion).

And finally, the ATSC standard also supports the standard-definition television (SDTV) formats. Both are interlaced and are either 704-pixel-by-480-line or 640-pixel-by-480-line. As most of the current television studio infrastructure supports one or other of these two formats, most local production (local news, for instance) is likely to remain in one of the formats in the early years. (Even so, the studios may decide to convert the resulting NTSC video into HDTV before encoding and transmitting it.)

Because ATSC permits these four formats, the compressed bit-stream may abruptly change in video format even as it is being aired. For example, a commercial could be broadcast in the 704-by-480 progressive format, followed by a movie in the 1280-by-720 progressive format, followed by local news promo in the 640-by-480 interlaced format, The ATSC standard takes note of this likelihood.

It recommends that the receiver seamlessly and without loss of video, continue to display all these formats in the native format of the television receiver.

3.4 COMPRESSION IN DIGITAL TV STUDIOS

The ATSC standard in the United States specifies the Moving Pictures Experts Groups MPEG2 as the video compression standard [Fig. 3.3]. It also specifies AC-3 compression for audio. Full details of the standard are given in the ATSC Standard Documents A/53 for video and A/52 for audio and is available at the ATSC World Wide Web site at <http://www.atsc.org>. An uncompressed 10-second-long HDTV video clip sampled as 4:2:2, requires 1.2 GB of storage. Obviously, to store and route many hours of digital video in the studio requires compression.

Fig. 3.3. The Advanced Television Systems Committee standard may be viewed as a four-layer hierarchy. The picture layer handles multiple video formats and frame rates. The compression layer specifies MPEG 2 compression for video and the Dolby-designed AC-3 compression for audio. The transport layer supports multiple programs and ancillary data in a single TV channel. The transmission layer's 8-VSB modulation in 6 MHz of terrestrial bandwidth provides a net data rate of 19.39Mb/s.

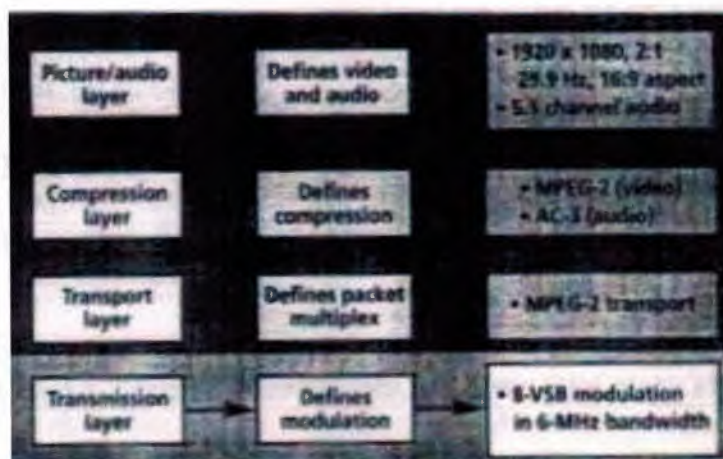


Figure 3.3

Note that the ATSC standard is a transmission standard for digital television that, by design, allows very high decoded-picture quality only once. If the same video is coded (compressed) and decoded several times, the picture quality rapidly drops. To overcome this problem, Tiered compression at various bit-rates may evolve for TV studio applications.

With current uncompressed production equipment... effects can be imposed on any NTSC video frame arbitrarily technical director who decides to cut away from a camera and go into a commercial can do it instantaneously. Any production equipment for compressed video must be equal to this requirement. An intra-coded system can certainly rise to the occasion: all intra-coded frames are independent frames, so that altering (or deleting) one intra frame would not effect others in the bit-stream.

Future advances in technology may enable production switchers that operate on compressed intra frame only) video, such as splicers that allow frame-accurate edit/inserts on compressed bit-streams. And so on. Research is being done on the feasibility of such devices at several laboratories around the world.

Compression in the TV studio not only reduces storage and archival costs but also allows stored video to be transferred to another destination faster than in real time. Table 3.2 shows various tiers of compression that may be needed in the studio. Compression can be optimised for the bit-rate required. The encode: decode latency that can be tolerated, and the quality that must be maintained.

If the bit-rate used within the studio is in the range of 200-270 Mb/s then the compressed data can be stored in and routed by uncompressed standard definition equipment that complies, with SMPTE 259. Both Sony and Panasonic have proposed systems like this [Table 3.2]. Their proposals allow D1 and D5 tape equipment currently in wide use to be used to store compressed data. The 270-Mb/s version of SMPTE 259 can be used to store and route up to 200-Mb/s data, and the 360-Mb/s version can be used for bit-rates up to 270 Mb/s. As more equipment supporting HDTV pictures becomes available. Digital SDTV will give way to HDTV programming.

TIERED COMPRESSION IN A TV STUDIO		
STUDIO APPLICATION	COMPRESSION RATIO	CODING FORMAT
VIDEO PRODUCTION	4:1	I-FRAMES ONLY
CONTRIBUTION LINK	10:1	IPPP FRAME STRUCTURE
STORAGE AND ARCHIVAL	25:1	IPIP FRAME STRUCTURE
TRANSMISSION	50:1	IPBBBBP FRAME STRUCTURE

I=INDEPENDENT FRAMES P=PREDICTED FRAMES		
B=BIDIRECTIONALLY DIRECTED FRAMES GOP=GROUP OF PICTURES		

Table 3.2

3.5 THE SPLICING OPERATION

When a video feed from one source is switched and replaced with video from another source, splicing occurs. It happens frequently in the broadcast environment when, for instance, a local station interrupts its network feed to insert locally generated programming. In the case of uncompressed video streams, it is relatively easy to splice two segments together. All that is required is that the two video sources be synchronized. The operator may perform a splice at any frame boundary because each frame in each stream contains the same amount of data and frame boundaries are synchronized.

But bit-streams of compressed video contain video frames of unequal length. The number of bits used to represent each frame can vary greatly depending on a number of things, including:

- The type of coding used. A predicted frame, coded using motion-compensated data from another frame, usually requires fewer bits to code than an intra-coded picture that contains all the information required to decode the original image.
- The variability in size of pictures of the same type. The greater the complexity of the image, then the greater the number of bits required to code it.
- The target bit-rate that video encoding must achieve. To do this, the encoder has the ability to arbitrarily change the number of bits that it assigns to a picture.

The 'From' stream makes reference to the stream being switched, and the "To" stream refers to the stream that the "From" stream is being switched to. Since both the streams contain I, P, and B frames, which always vary in size (and duration), arbitrarily switching from one to the other would cause severe disruptions in the decoder.

Using predicted frames also creates bit-streams in which the ability to successfully decode each frame requires the presence of other frames. The splice point must be carefully selected so that every frame in the resulting sequence around the splice has all the information required to decode it. A successful splice between these two sequences can occur only at the point indicated.

Splicing is simple on two uncompressed video streams-the hardware only has to wait for the vertical edge of the frame [Fig 3.4]. Splicing on a compressed bit-stream is more complicated because there are varying numbers of bits in pictures of a compressed video, and the two bit-streams are rarely aligned. Also, the splice operation cannot arbitrarily happen because of the use of predicted frames (P and B). The bit-stream must be created with the marked splice in-points and out-points, then the splice can wait proper location at which a splice is made.

Fig 3.4. The compressed audio (AC-5) sync frame of synchronisation information (SI), bit-stream information (BSI), six compressed audio blocks (32 ms duration), and a cyclic redundancy check (CRC).



Figure 3.4

Each video encoder must also create a bit-stream that maintains the integrity of a data buffer located in the decoder by preventing under-or overflow. This buffer, fixed in size in the MPEG specifications, is used to store the compressed representation of every video frame until it is time for that particular frame to be decoded.

To provide optimum picture quality, the video encoder takes full advantage of this buffer size. Two encoders separately encode the "To" and "From" streams. Individually, the two streams do not violate the buffer limits, but when the two are spliced, the resulting bit-stream violates the buffer limits. This violation will cause an exception in the decoder that may crash it momentarily.

Overcoming this problem can be achieved by placing stricter constraints on the rate control just prior to the splice point. The stricter control has the effect of pushing the buffer level to a common point in both encoders. The splice must identify splice points in each of its input bit-streams, and splice the two streams at the appropriate points.

In the case of a pre-encoded video stream, the splicing device may also be required to cue a bit-stream from tape or disk. It must also make changes in the transport layer of the spliced stream, to ensure that the splice is invisible to the decoder. Splicing also requires an encoding process that creates the splice point. The encoder must produce a picture coding structure and deal with the bit-stream constraints. As part of the act of creating the splice point, the video encoder must signal the presence of this splice point.

A SMPTE engineering group on packetized television is looking into the splicing of compressed bit-streams and has created a Recommended Practice that has been issued for ballot to its voting members.

3.6 DIGITAL TELEVISION BROADCAST

The early deployment of broadcast digital television DTV services will take place over the next year. The rollout will begin in the larger U.S. markets. The Federal Communications Commission (FCC) is assigning additional channels to broadcasters for digital transmission and has mandated a rapid build-out plan. The four top network affiliates in each of the 30 top markets must be on the air by 1 November 1998, with all the commercial stations on the air by 2002. TV broadcasters have committed to having major markets on-air (one-seventh of the population served) with digital TV by December 1998. Present scheduling from the FCC also calls for the return of a second (NTSC) channel to the Government by the year 2006.

These early services will be a combination of standard definition TV, high-definition TV, and data. The choice of high or standard definition will be made by the broadcaster on the basis of the material to be broadcast and its targeted audience. The data services that accompany the SDTV or HDTV broadcast may or may not be related to the program being transmitted. One example: the data services that will carry additional information like the Web page address of the product whose commercial is being aired. And the user, if needed, can access the Web page using the television remote control and display it on the TV set.

The Federal Communication Commission's approval of the DTV standard means that the clock for broadcasters to begin the transition to digital television has started ticking. A number of efforts are currently under way to create and test the equipment and techniques required for such services. Among them are:

- The NIST High Definition Broadcast Technology (HDBT) project. For this joint venture, the Advanced Technology Program provides participating companies with co-funding for high-risk research and development, in order to accelerate the development and commercialization of new technologies in the area of HDTV broadcast. The Advanced Technology Program is an arm of the Department of Commerce's National Institute of Standards and Technology (NIST). The companies, participating in the project include Advanced Modular Solutions, Comark, IBM, MCI, Philips, Sarnoff Corp., Sun Microsystems, and Thomson. Among the areas of research are tiered MPEG compression for different applications in the studio, asynchronous transfer mode (ATM)-based MPEG routing, compressed-domain processing, and compressed-domain splicing technologies.
- The Advanced Television Systems Committee. This is a cooperative effort on the part of manufacturers and broadcasters to promote the DTV standard and in addition to certify the new equipment as it comes on the market.
- The WHD-TV Model Station. Another cooperative effort, it is headed by Maximum Service Television (MSTV) and the Consumer Electronics Manufacturers Association (CEMA) and aims to promote the HDTV standard and to test new equipment as it becomes available.

Although no digital TV sets are yet available, several stations are on-line digitally. They are WHD-TV (Model Station, Washington, D.C.), WRAL (Raleigh, NC.), and WCBS-HD (New York City). The capabilities of these stations vary from transmission tests to the full production and transmission of HDTV programming. WHD-TV recently achieved the transmission of the first live NTSC and HDTV simulcast of a scheduled program.

As other broadcasters step up to convert to HDTV, probably two studio configurations will prevail: a full-production facility for the large markets and network production, and a facility known as a pass-thru station, the minimum needed to get on the air with digital content.

3.7 THE FULL – PRODUCTION STATION

The full-production HDTV facility must support existing NTSC equipment. When possible, it also must allow compressed operations (like storage and splicing) to avoid encoding and decoding penalties. The configuration resembles that being created as part of the NIST HDBT project. In this setup, a high-speed ATM computer network routes the compressed bit-stream around the studio.

Besides the compressed video, the ATM net routes intercom, digital audio (compressed or not), and data. All the equipment (servers, encoders, and the off line transcoder) interfaces to the ATM router, which is the studio's central switch, replacing the router of conventional studios.

The trans-coders job is to convert one compressed format into another for example a 155-Mb/s I-P-I-P format into the 45-Mb/s I-P-B-B Format). All the devices on the computer network are controlled by the studio control workstation. This architecture also allows connection to be made to other TV studios over existing telecommunications networks. A network interface device has this job.

In the early stages, video production will be performed on uncompressed video. As compressed technology advances, more production will be done on compressed video. Most studios will probably transition to compressed production but retain uncompressed elements to take advantage of some of its Features.

3.8 THE PASS-THRU STATION

The 1000 small network-affiliated TV stations in the United States depend heavily on the major networks for most of their television programming. They also add local commercials (at pre-assigned times) for revenue and provide local news and information.

Because of their special needs, these affiliates could benefit substantially from equipment for splicing compressed streams. The stations would receive a satellite feed from the network and without decoding (and re-encoding) the bitstream, be able to splice in the commercials. Technical hurdles, however, remain before the compressed splicing technology becomes commercially available. One problem needing solving is how to add a local stations logo to the network feed in compressed domain.

For local news production, the pass thru stations can continue to use existing NTSC equipment, whose output they can then encode for broadcast, using a standard definition encoder.

In this configuration, the network feed will be received by a satellite feed and will then pass through a local compressed-data switcher on the way to the transmit site. The switcher is in place to allow the insertion of local commercials into the network feed. Optionally there is also a video encoder to allow the encoding of locally generated content in either high- or standard definition TV. This configuration is open to allow further upgrades for local production.

3.9 GLOBAL HDTV TRENDS

- 1.) Australia: Australia like the one that is backed in the United States by the Advanced Television Systems Committee (ATSC) is currently under consideration.
- 2.) Brazil: At present, the ATSC standard itself is being weighed.
- 3.) China: A standard like Europe's digital video broadcast (DVB) is being considered.
- 4.) Europe: Countries in Europe have agreed to DVB, which like ATSC includes MPEG2 video compression and packetized transport but which has different audio compression and transmission schemes. The DVB standard now has guidelines for a 1920-pixel-by-1080-line high-definition television (HDTV) format (more information can be obtained on the World Wide Web from <http://www.dvb.org>).
- 5.) Japan: The country was the first to deploy analog HDTV, but that system never gained popularity. Plans for digital television transmission seem uncertain. Direct broadcast using satellite (DBS) service is based on MPEG2 and DVB.
- 6.) North America: Canada and Mexico appear likely to adopt the U.S. ATSC standard.
- 7.) South Korea and Taiwan: Each of these countries has government-funded research programs for digital television. Both governments are likely to decide on a system that is similar to either the DVB or the ATSC standard within the next six months.

3.10 THE DIGITAL TELEVISION STANDARD

The Digital Television Standard describes a system designed to transmit high quality video and audio and ancillary data over a single 6 MHz channel. The system can deliver reliably about 19 Mbps of throughput in a 6 MHz terrestrial broadcasting channel and about 38 Mbps of throughput in a 6 MHz cable television channel. This means that encoding a video source whose resolution can be as high as five times that of conventional television (NTSC) resolution requires a bit rate reduction by a factor of 50 or higher. To achieve this bit rate reduction, the system is designed to be efficient in utilising available channel capacity by exploiting complex video and audio compression technology.

The objective is to maximise the information passed through the data channel by minimising the amount of data required to represent the video image sequence and its associated audio. The objective is to represent the video, audio, and data sources with as few bits as possible while preserving the level of quality required for the given application.

Although the RF/Transmission subsystems described in the Digital Television Standard are designed specifically for terrestrial and cable applications, the objective is that the video, audio, and service multiplex/transport subsystems be useful in other applications.

System block diagram

A basic block diagram representation of the system. This representation is based on one adopted by the International Telecommunication Union, Radiocommunication Sector (ITU-R), Task Group 11/3 (Digital Terrestrial Television Broadcasting). According to this model, the digital television system can be seen to consist of three subsystems.

1. Source coding and compression,
2. Service multiplex and transport, and
3. RF/Transmission.

“Source coding and compression” refers to the bit rate reduction methods, also known as data compression, appropriate for application to the video, audio, and ancillary digital data streams. The term “ancillary data” includes control data, conditional access control data, and data associated with the program audio and video services, such as closed captioning. “Ancillary data” can also refer to independent program services. The purpose of the coder is to minimize the number of bits needed to represent the audio and video information. The digital television system employs the MPEG2 video stream syntax for the coding of video and the Digital Audio Compression (AC-3) Standard for the coding of audio.

“Service multiplex and transport” refers to the means of dividing the digital data stream into “packets” of information, the means of uniquely identifying each packet or packet type, and the appropriate methods of multiplexing video data stream packets, audio data stream packets, and ancillary data stream packets into a single data stream. In developing the transport mechanism, interoperability among digital media, such as terrestrial broadcasting, cable distribution satellite distribution, recording media, and computer interfaces, was a prime consideration. The digital television system employs the MPEG2 transport stream syntax for the packetization and multiplexing of video, audio, and data signals for digital broadcasting systems. The MPEG2 transport stream syntax was developed for applications where channel bandwidth or recording media capacity is limited and the requirement for an efficient transport mechanism is paramount. It was designed also to facilitate interoperability with the ATM transport mechanism.

“RF/Transmission” refers to channel coding and modulation. The channel coder takes the data bit stream and adds additional information that can be used by the receiver to reconstruct the data from the received signal which, due to transmission impairments, may not accurately represent the transmitted signal. The modulation (or physical layer) uses the digital data stream information to modulate the transmitted signal. The modulation subsystem offers two modes: A terrestrial broadcast mode (8 VSB), and a high data rate mode (16 VSB).

3.11 VIDEO COMPRESSION & DECOMPRESSION

The need for compression in a digital HDTV system is apparent from the fact that the bit rate required to represent an HDTV signal in uncompressed digital form is about 1 Gbps, and the bit rate that can reliably be transmitted within a standard 6 MHz television channel is about 20 Mbps. This implies a need for about a 50:1 or greater compression ratio.

The Digital Television Standard specifies video compression using a combination of compression techniques, and for reasons of compatibility these compression algorithms have been selected to conform to the specifications of MPEG2, which is a flexible internationally accepted collection of compression algorithms.

The purpose of this tutorial exposition is to identify the significant processing stages in video compression and decompression, giving a clear explanation of what each processing step accomplishes, but without including all the details that would be needed to actually implement a real system. Those necessary details in every case are specified in the normative part of the standards documentation, which shall in all cases represent the most complete and accurate description of the video compression. Because the video coding system includes a specific subset of the MPEG2 toolkit of algorithmic elements, another purpose of this tutorial is to clarify the relationship between this system and the more general MPEG2 collection of algorithms.

3.12 DTV Standards

The simplified chart below outlines the various DTV standards. Although there are numerous factors involved in evaluating technical quality, generally the greater number of lines the clearer the picture will appear. The listings in red are considered HDTV which represents a very noticeable improvement in picture clarity and quality.

Active LINES	1080	1080	720	720	480	480
Per Picture	1920	1920	1280	1280	704	640
Pixels Per Line	16:9	16:9	16:9	16:9	4:3, 16:9	4:3
Aspect Ratio	23.97 to 30*6	29.97 to 30	23.976 to 60	29.97 to 30	29.97 to 60	29.97 to 30
Frame Rate	Progressive	Interlaced	Progressive	Interlaced	Progressive	Interlaced

Table 3.4

*To simplify things the specific frame rate steps in some of these ranges have been eliminated.

Since all of the above options are based on digital electronics (and not the analogue NTSC system we're now using) all of the new systems will result in an improvement in video and audio quality. As we will see when we discuss audio, the new digital systems also represent a dramatic improvement in audio quality.

The progressive/interlaced issue (introduced in the last module) is a thorny one and one in which all the technical issues have yet to be resolved-especially with the 1080 standard.

The chart below lists the networks and the systems decided upon thus far. (All of these are subject to change.)

ABC	720 line progressive and 480 line progressive
CBS	1,080 line, interlaced (prime time) 480 line interlaced (off-peak viewing hours)
NBC	1,080 line, interlaced (prime time) 480 line interlaced (off-peak viewing hours)
FOX	480 line progressive (possibly 60 frames per second for sports; 30 frames per second for other programming)
PBS	Not announced
WB	1,080 line interlaced
SONY	1,080 line interlaced

You will commonly see the 1,080 line interlaced high-definition system listed as simply 1080i and the 420 standard definition progressive system listed as 420p.

Both the 720p and the 1080i formats are considered high-definition, whereas the 480p format is considered "standard definition" because it's similar to what we're seeing now.

Whatever formats come into wide use, the networks and local TV stations (and cable, satellite and postproduction services) are going to have to invest hundreds of millions of dollars to convert to the new technology. This will involve new studios new cameras, new tape machines, new switchers, new transmitters, and in many cases even new transmitter towers.

3.12 DIGITAL TELEVISION : JUST THE FACTS

Today's Television and Tomorrow's Digital Televisions Park Ridge, NJ, June 13, 1997 - The recent series of Federal Communications Commission (FCC) rulings has set in motion the transition from today's analogue over-the-air broadcasting to digital television broadcasting. Digital television, known as DTV, promises cinema quality video and audio, as well as entirely new digital data services.

Today's high-performance analogue televisions already deliver a digital television entertainment experience from digital video sources, such as DSS® satellite DVD video players, digital video (DV) camcorders, WebTV™ Internet terminals and Play Station® video game systems. Consumers can continue to enjoy those digital video entertainment sources, as well as their cable TV programs, on the analog sets they have now throughout the advent of digital television and beyond.

Consumers can be confident that tomorrow's digital television broadcasts will coexist with today's analog televisions for many years to come. The transition to DTV will not happen immediately. It will be phased in over at least a decade. DTV broadcasts are expected to remain somewhat limited for several years. Throughout this transition period and beyond, analog television sets can maintain their position as the cornerstone of the home entertainment system. Digital TV broadcasting will not make today's analog TVs obsolete.

Today's high-performance analogue televisions, such as Sony's Trinitron® and Videoscope® big screen sets, can be considered future-ready. The FCC's target is to complete the transition from analogue to digital broadcasts by 2006. That target is subject to change, based on market penetration of DTV sets. Consumers can experience a smooth transition from analogue to digital broadcasts by adding DTV converters to analogue television sets.

DTV converters will allow DTV signals to be displayed on today's analog TVs, much like Sony-brand DSS receivers convert satellite's digital signals for today's televisions.

With DTV converters, today's high-performance analog televisions can deliver even better picture and sound quality than they do now. Many televisions are capable of displaying better pictures than those now delivered through today's analog broadcasts. In other words, many sets do not display the best picture they can technically produce because they are receiving video from analog signals, which are inferior to digital signals.

Today's high performance analog televisions are a great value. Digital television prices will be substantially higher than today's big screen TV prices.

Consumers considering the purchase of a new television should look for the highest quality televisions with multiple video inputs especially S-video to accommodate current and future sources of digital video. Twenty-six Sony big screen Trinitron and Videoscope televisions have S-video and multiple A/V inputs and outputs.

3.13 FREQUENTLY ASKED QUESTIONS

1.) Should I wait to buy a new TV set until the digital TV's make their debut in stores next year?

No. While only DTV sets will enable consumers to fully enjoy the superior video, dramatic multi-channel audio and other benefits of digital TV broadcasts and services, consumers can still enjoy digital TV broadcasts on televisions being sold today for these reasons:

- Today's TVs are "future ready." High-performance TVs -- like Sony Trinitron and Videoscope big screen sets -- have multiple A/V inputs, including S-video, and are particularly well-suited for digital video sources.
- The transition from today's analogue sets to digital TV sets will take at least a decade. Initially, digital TV broadcast programs will only be available in 10 cities (about 30% of the U.S. population).
- Today's TVs are feature-rich and affordably priced. As with all new consumer electronics products, the first digital TV sets, which are slated for stores in late 1998, will be priced very much higher than comparably-sized analogue TV sets.

2.) Will DTV make my current TVs obsolete?

No. Today's TVs will display analogue or digital broadcast transmissions for many years to come. Even if analogue TV broadcast terrestrial transmissions are discontinued in 2006, as the FCC targets, DTV converters will enable consumers to watch digital TV broadcasts.

Consumers also can continue to enjoy watching VHS videotapes, DVD discs, and cable TV and DSS programs on their current TVs for many years to come.

3.) Will I be able to watch digital television broadcasts on my current TVs?

Initially, broadcasters are expected to transmit programs in both analogue and digital, known as simulcasting. If DTV broadcasts are available in your area, you will need to add a DTV converter to today's TVs to view digital TV broadcasts as they become available. These DTV converters will be available from a number of manufacturers. You can continue to enjoy cable, DSS, VHS, DVD and DV programs on today's TVs throughout the transition to digital TV broadcast programs and beyond.

4.) How much will the new digital TVs cost? How much will the DTV converters cost?

The new digital TVs are likely to be much higher in price than similarly sized TVs today. Some manufacturers have announced DTV sets beginning at \$7,000. DTV converters are expected to be available for a few hundred dollars, rendering today's TVs fully compatible with digital TV broadcast programs as they become available.

Sony will have digital TVs in the market as digital TV broadcast transmissions become available. We also expect to provide DTV converters for existing TVs as DTV broadcasts become more widely available.

5.) Will I be able to use my existing video source components with DTV sets?

While no company has released details for DTV set specifications, it is expected that DTV sets will have video inputs capable of accepting current video sources, such as VCRs, DVD players, DSS systems, etc.

6.) If I have cable or satellite TV, how will digital TV affect my service?

Consumers will be able to continue receiving cable or satellite programs on today's TVs. Ask your cable operator or program provider for more details.

7.) Will the new digital TVs work with current speakers and sound systems?

Yes. Consumers should consider adding Dolby® Digital surround sound components to take full advantage of digital TV broadcast transmissions (as well as DVD).

8.) If I want to buy a television set today, what features will help ensure I'm ready for DTV?

Consumer should consider buying televisions with the best possible picture quality. DTV converters and broadcasts will maximise performance of analogue sets.

Consumers should also look at sets that have multiple A/V inputs, especially S-video

3.14 STANDARDS

TV operations use much digital equipment operating in a mainly analogue environment. By the late 1970s it was quite usual for a typical BBC programme to have been through about 15 separate digital processes, some composite and some component. Thus, the signal would encounter many analogue-to-digital and digital-to-analogue conversions and several PAL to RGB or YcrCb decoding and encoding processes. In some cases where two pieces of digital equipment were adjacent in the sequence it was still necessary to go through an analogue process because there were no industry standards and the only way to interconnect them was by using analogue signals in the established composite or RGB format.

This situation broadcasted on both sides of the Atlantic to start work on defining standards for digital TV so that pieces of equipment could eventually be connected together without the A-D/D-A stages. In Europe the work was led by the European Broadcasting Union and in North America by the Society of Motion Picture and Television Engineers. Initially attention was concentrated on composite standards for the 625-line PAL and 525 line NTSC standards by the EBU and SMPTE respectively.

Digital components

The increasing use of component signals in digital equipment, coupled with the possibility of establishing a substantial area of common ground between component coded signals for the 625 and 525 line encouraged a change in direction. A joint committee of the EBU and the SMPTE chaired by Howard Jones from the BBC Research Department set out, in liaison with industry and broadcasting unions in other parts of the world, to establish a world-wide digital TV standard for component signals. It is worthwhile quoting from the introduction to the document which has now become the basic international digital TV standard: Recommendation 601 of the CCIR (the International Radio Consultative Committee of the International Telecommunication Union). Although the standard was not adopted until 1986, the introduction expresses clearly what prompted the initiative in 1979 which eventually produced the standard.

The CCIR.

CONSIDERING

- (a) that there are clear advantages for Television broadcasters and programme producers in digital studio standards which have the greatest number of significant parameter values common to 525-line and 625 line systems:
- (b) that a world-wide compatible digital approach will permit the development of equipment with many common features, permit operating economies and facilitate the international exchange of programmes:
- (c) that an extensible family of compatible digital coding standards is desirable. Members of such a family could correspond to different quality levels, facilitate additional processing required by present production techniques, and cater for future needs.
- (d) that a system based on the coding of components is able to meet some, and perhaps all, of these desirable objectives;
- (e) that the co-siting of samples representing luminance and colour-difference signals or, if used, the red, green and blue signals facilitates the processing of digital component signals required by present production techniques.

3.15 UNANIMOUSLY RECOMMENDS

That the following be used as a basis for digital coding standards for television studios in countries using the 525-line system as well as in those using the 625-line system;

The key to commonality is that, because the products 625×50 and 525×60 are almost equal the time taken to scan a TV line in the two systems is very nearly the same. Thus, in the terms discussed earlier, if one samples the analogue waveform described by Y at the same sampling frequency for both scanning systems, it is possible to

ensure that there are the same number of samples appearing in the same positions along the line in both the 625-line and 525-line raster. The circumstances would, of course, be similar for CR and CB but these would be sampled at a lower sampling frequency and would therefore, be spaced farther apart than the Y samples.

Most of the points in the 'Considering' quoted above are self explanatory. In (e) reference is made to the fact that by choosing values for the luminance and chrominance (CR, CB) frequencies which are simply related to each other and are integer multiples of 2.25 MHz (the lowest common multiple of the line frequencies in the 625/50 and 525/50 systems) it is possible to co-site the luminance and chrominance samples in the same static orthogonal pattern in both systems.

After much discussions, many experiments and international demonstrations comparing the relative merits of different sampling frequencies as well as different ratios between the sampling frequencies for luminance and chrominance, 13.5 MHz. and 6.75 MHz were finally chosen.

Table 1 gives the main encoding parameters for the 4:2:2 member.

The most important features to note in addition to the sampling frequencies are the choice of 8 bits per sample PCM for each of the three signals and the 720: 320:320 samples for Y: Cr:Cb respectively for the active part of the line, i.e. the part carrying the picture rather than "the part allocated to the analogue synchronizing waveform which does not form part of the digital time specification."


Table 3.6 shows the relationship between the 720 luminance sample and the sync periods. It also illustrates how the different sync-times for the 625/50 and 525/~60, systems have been adjusted to facilitate the common sampling structure for the active line. A similar table could of course be constructed for the chrominance signals by replacing the 720 by 360 in each case.

The Recommendation 601 document derives and defines in detail the relationship between the parameter in the Table 3.5. The normalized values for the boundary conditions.

The other matters defined in Recommendation 601 are aspects of quantization and the filter characteristics for the signal sampled at 13,5 and 6,75 MHz.

PARAMETERS	525-LINE, 60 FIELD PER SECOND SYSTEMS	625-LINE, 50 FIELD PER SECOND SYSTEM
1.CODED SIGNALS: Y, CR: CB	THESE SIGNALS ARE OBTAINED FROM GAMMA PRE-CORRECTED SIGNAL NAMELY: EY,ER-EY,EB-EY	
2.NUMBER OF SAMPLES PER TOTAL LINE: -LUMINANCE SIGNAL (Y) -EACH COLOUR DIFFERENCE SIGNAL CR:CB	858 412	1 864 432
3.SAMPLING STRUCTURE	ORHOGONAL,LINE FIELD AND FRAME REPETITIVE, CR AND CB SAMPLES CO-SITEDWITH ODD(1 ST ,3 RD ,5 TH ETC.) Y SAMPLES IN EACH LINE	
4.SAMPLING FREQUENCY -LUMINANCE SIGNAL -EACH COLOUR DIFFERENCE SIGNAL	13,5 MHz 6,75 MHz THE TOLERANCE FOR THE SAMPLING FREQUENCY SHOULD COINCIDE WITH THE TOLERANCE FOR THE LINE FREQUENCY OF THE RELEVANT COLOUR TELEVISION STANDARD.	
5.FORM OF CODING	UNIFORMLY QUANTISED PMC, 8 BIT PER SAMPLE, FOR THE LUMINANCE SIGNAL & EACH COLOUR DIF SIGNAL.	
6.NUMBER OF SAMPLES PER DIGITAL ACTIVE LINE: -LUMINANCE SIGNAL -EACH COLOUR DIFFERENCE SIGNAL	720 360	
7.ANALOGUE-TO-DIGITAL HORIZALTAL TIMING RELATIONSHIP -FROM END OF DIGITAL ACTIVE LINE TO OB	16 LUMINANCE CLOCK PERIOD	12 LUMINANCE CLOCK PERIOD
8.CORRESPONDENCE BETWEEN VUDEO SIGNAL AND QUANTISATION LEVELS: -SCALE -LUMINANCE SIGNAL -EACH COLOUR DIFFERENCE SIGNAL	0 TO 255 220 QUANTISATION LEVELS WITH THE BLACK LEVEL CORRESPONDING TO LEVEL 16 AND THE PEAK WHITE LEVEL CORRESPONDING TO THE LEVEL 235. THE SIGNAL LEVEL MAY OCCASIONALLY EXCURSE BEYOND LEVEL 235. 225 QUANTISATION LEVELS IN THE CENTRE PART OF THE QUANTISATION SCALE WITH ZERO SIGNAL CORRESPONDING TO LEVEL 128.	
9.CODE-WORD USAGE	CODE-WORD CORRESPONDING TO QUANTISATION LEVELS 0 AND 225 ARE USED EXCLUSIVE FOR SYNCHRONISATION. LEVELS 1 TO 254 ARE AVAILABLE FOR VIDEO.	

Table 3.5



525-line 60 field Per second System	122 T	720 T	16 T
OB (leading edge of the sync, half-amplitude reference)		Digital active line period	OB Next line
625-line 50 field Per second System	132 T	720 T	12 T

Table 3.6
Relation of digital active line analogue sync, refrence.

CHAPTER 4 : A-D & D-A CONVERSION

4.1 DIGITAL TECHNIQUES IN DOMESTIC RECEIVERS

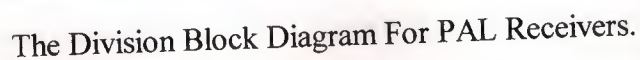
Ultimately digital techniques will be used to deliver TV to the public using the bandwidth reduction methods. At this stage it is not possible to obtain sufficient bit-rate reduction to enable a TV signal to be sent digitally in the channels allocated for terrestrial or direct broadcast satellite transmission. A hybrid approach using a combination of analogue and digital transmission is possible, provided that many of the digital processes described in this book could viably be carried out in a domestic receiver.

The feasibility of this was demonstrated in 1983 by ITT-Intermetally in Germany who produced a TV receiver for PAL, SECAM and NTSC, using seven VLSI 1 (very large scale integration) silicon chips to perform most of the receiver functions in digital form. The block diagram of this is shown in Figure 1. The key was the development of A/D and D/A converters for vision and sound in the video codec unit and audio processor unit respectively. The video A/D and D/A converters operate at a sampling frequency of 17.73 MHz. Whilst, of course, not up to the standard of the studio quality products. Their performance was surprisingly good for a mass-produced consumer product. The other significant chips were the video processor unit and the deflection unit using digital filtering techniques.

The main objective of the 'Digivision' approach introduced by ITT was to use digital television techniques to increase the versatility of the receiver under software control and to reduce the reducing the assembly times and production variables normally associated with analogue circuit boards. However they recognised that once the main hurdle of producing a domestic receiver with the signals in digital form had been overcome future enhancements were feasible in principle. These include noise reduction improved decoding, standards conversions (to remove flicker or line structure) and 'enlarging' parts of the picture, all based on the assumption that the cost of field storage will continue to fall to a level where it is no longer a significant fraction of the total cost of the receiver.

The next significant development was the establishment of a standard for DBS transmission which assumed that the studio output would be a Recommendation 601 digital signal rather than a composite PAL or SECAM signal. Thus, rather than encode this into a composite form one would capitalise on the somewhat wider bandwidth of the satellite channel to send a time compressed analogue component signal where the luminance and chrominance were transmitted one after the other during the line period [18,19]. This strategy relied on two factors. Firstly, that since the public would have to buy satellite dishes and microwave down-converters, they would accept the need to buy new receivers for DBS rather than expect to be able to use their existing PAL receivers (Alternatively, they would be prepared to buy 'set-top' adapters for their existing sets which would allow the establishment of a new transmission standard which would eventually lead to the elimination of composite coding at the receiver for DBS reception).

The first MAC/packet decoder chips which were reduced in 1988 by ITT-Intermetall using their 'Digivision' technology but operating in component form at the sampling frequencies of 13.5 and 6.75 MHz specified in Recommendation 601. Shows the layout of one of these VLSI chips. Whilst this is still some way from the complexity required to enable sophisticated bandwidth reduction and standards conversion circuitry to be incorporated in domestic TV receivers. It does indicate the feasibility of doing this well before the end of the 20th century. Unless the use of VLSI in TV receivers becomes viable at the decoding of bandwidth compressed signals could be delivered to the public at a level of availability which they have come to expect for TV.



54

4.2 ANALOGUE-TO-DIGITAL AND DIGITAL-TO-ANALOGUE CONVERSION AND ITS EFFECT ON TELEVISION SIGNALS

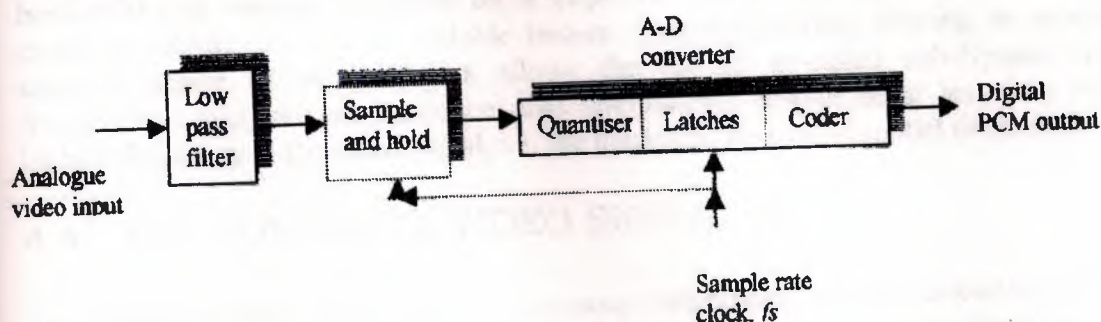
GENERAL

The introduction of digital equipment into basically analogue television studios has resulted in the tandem connection of relatively large numbers of video PCM codec. (the term 'codec' refers to a combination of an analogue-to-digital A-D converter and a digital-to-analogue converter (D-A) converter.) Furthermore, the number of codecs in tandem is likely to increase for some years before digital paths start to replace analogue connections between digital equipment. Thus a very high performance is required from individual converter units if the use of digital processing is not to cause a degradation in the picture quality obtained from a broadcasting network.

This topic discussed all aspects of the performance of video PCM coders, including requirements regarding number of bits per sample and sampling frequency, the meaning and relevance of data given by manufacturers, methods of measuring coding accuracy and acceptable limits for any inaccuracies. By describing test of several codecs operated in tandem, it is shown that performance is by no means solely dependent on the design of basic A-D and D-A circuitry. Other factors, such as the quality of the associated video low-pass filters and the use of good construction techniques in surrounding circuits, are also important.

4.3 ANALOGUE-TO-DIGITAL (A-D) CONVERSION

Analogue-to-digital conversion of video signals involves three main namely quantisation, sampling and coding, as illustrated in figure 4.2.



Basic elements of A-D converter and associated processes.

Figure 4.2

The quantiser measures the magnitude of the analogue input signals and divides it into 2^n parts where n is number of digits in the binary codewords conveyed by the digital output signal. The required $2^n - 1$ decision level are designed to be equally spaced over the conversion range, assuming that the video signals being processed have been gamma corrected.

For A-D conversion of linear video signals prior to gamma correction, a nonlinear quantisation characteristic providing greater accuracy near black level than near white level may be used as cameras and telecines. However the vast majority of applications where A-D conversion of video signals is currently employed occur in the gamma-corrected domain and it will be assumed that this is the case in the remainder of this topic.

The outputs of the level comparators are sampled at regular intervals by means of digital latches. The outputs of these latches are then Transcoded to obtain the n -bit digital signal indicating the highest decision level which has been exceeded at each sampling instant.

It is obviously necessary that the times at which the outputs of the different level comparators are latched must correspond to identical instants in the analogue input signal. The resulting timing problems can be caused by sampling the analogue signal prior to the A-D converters and holding these sampling values constant for a significant proportion of the interval between samples. The desirability of employing a sample-and-hold circuit depends on the method of A-D conversion as discussed later.

The sampling frequency f_s which is employed in an A-D converter must normally be at least twice the highest frequency component f_v in the input video signal. This is known as the Nyquist sampling criterion and the reason behind it is to avoid the generation of alias components having frequencies lying within the video frequency base-band of 0 to f_v . The purpose of the low pass filter shown in Figure 4.2. Is to ensure that the sampled video signal does not contain redundant components with frequencies above the specified video bandwidth. This enables the sampling frequency and hence the digital bit-rate to be kept to a minimum without aliasing distortion.

The application of the Nyquist criterion to video signals is complicated by the fact that their spectra can be resolved into three components corresponding to variations in the horizontal and vertical directions on a displayed picture and variations in time due to changing picture content. If suitable two-or three-dimensional filtering is applied, this complex nature of video spectra allows the use of so-called sub-Nyquist sampling frequencies which do not introduce alias components despite being less than twice the highest frequency in the video signal, i.e. the maximum horizontal spatial frequency, f_v .

4.4 THE COMPOSITE VIDEO SIGNAL

The composite video signal includes luminance (brightness) signals, horizontal and vertical sync pulses, and blanking pulses. Figure shows the composite video signal for a single horizontal scan line, 1H (63.5 μ s) (Notice that a positive transmission signal is shown.) The figure shows that the active portion of the scan line occurs during the positive slope of the scanning waveform and horizontal retrace occurs during the negative slope (the blanking time). The brightness range for standard television broadcasting is 160 IRE units peak to peak. 160 IRE units is generally normalized to V_{p-p} . The exact value of 1 IRE unit is unimportant; however, the relative value of a video signal in IRE units determines its brightness.

For example, maximum brightness (pure white) is 120 IEE units, and no brightness is produced for signals below the reference black level (7.5 IEE units). The reference black level is also called the pedestal or black setup level. The blanking level is 0 IEE units, which is below the black level or, in other words, blacker than black. Sync pulses are negative-going pulses that occupy 25% of the total IEE range. A sync pulse has a maximum level of 0 IEE units and a minimum level of -40 IEE units. Therefore, the entire sync pulse is below black and, thus, produces no brightness. The brightness range occupies 75% of the total IEE scale and extends from 0 to 120 IEE units, with 120 units corresponding to 100% AM modulation of the RF carrier. However, to ensure that over-modulation does not occur, the FCC has established the maximum brightness (pure white) level to be 87.5% or 100 IEE ($0.875 \times 160 = 140$ units, $-40 + 140 = 100$ units).

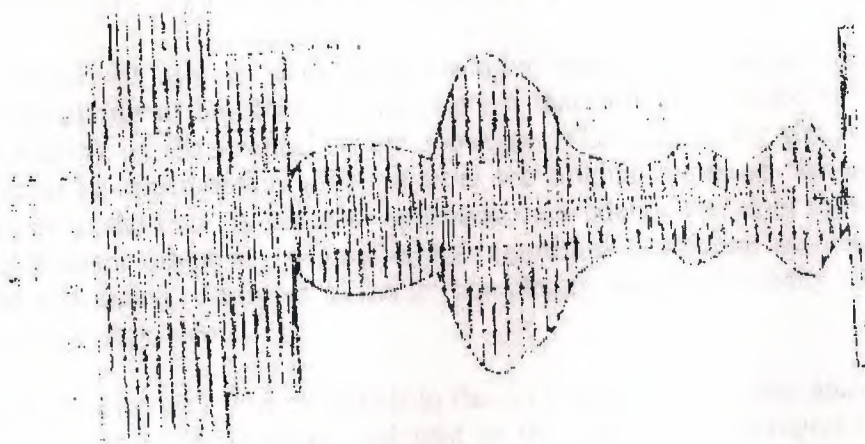
Figure 9 shows the composite video signal for the even field, which equals 1 V or 16.7 ms and is sufficient time for 262.5 horizontal scan lines (262.5H). However, the vertical blanking pulse width is between 0.05V and 0.08V or 833 micros to 1333 micros. Therefore, the vertical blanking pulse occupies the time of 13 to 21 horizontal scan lines which leaves 241.5 to 249.5 active horizontal scan lines. The figure also shows that most of the active scan lines occur during the positive slope of the vertical scanning waveform, and the vertical retrace occurs during the vertical blanking pulse.

4.4.1 HORIZONTAL BLANKING TIME

Figure 10 shows the blanking time for a single horizontal scan line. The total blanking time is approximately 0.1GH or 9.5 micros to 11.5 micros. Therefore, the active (visible) time for a horizontal line is approximately 0.84H or 52 micros to 54 micros. Figure 10 shows that the sync pulse does not occupy the entire blanking time. The width of the actual sync pulse is approximately 0.08H or 4.05 micros to 5.25 micros. The time between the beginning of the blanking time and the leading edge of the sync pulse is called the front porch and is approximately 0.02H with a minimum time of 1.27 micros. The time between the trailing edge of the sync pulse and the end of the blanking time is called the back porch and is approximately 0.06H with a minimum time of 3.81 micros.

4.4.2 VERTICAL BLANKING TIME

Figure 11 shows the first 10H of a vertical blanking pulse for a negative transmission waveform. The figure shows that the entire blanking pulse is below the level for reference black (below 7.5 IEE units). Each vertical blanking pulse begins with six equalizing pulses, a vertical sync pulse, and six more equalizing pulses. The equalizing pulses ensure a smooth, synchronized transition between the odd and even fields. The equalizing pulse rate is 31.5 MHz, which is twice the horizontal scanning rate.



RF ENVELOPE FOR NEGATIVE TRANSMISSION VIDEO SIGNAL

Figure 4.4

Therefore, each equalizing pulse takes $112H$, and the 12 pulses occupy a total time of $6H$. The structure vertical sync pulse occupies the time of $3H$. The serratis in the vertical sync pulse ensure that the receiver maintains horizontal synchronization during the vertical retrace time. A total time of nine horizontal scan lines [9H] is required to transmit the equalizing and vertical sync pulses. From the figure it can be seen that the first SH occur at the end of a vertical scan (at the bottom of the CRT). The following SH occur during the retrace time, and all 9H occur during the vertical blanking time and, therefore, are not visible. The exact vertical blanking time is determined by the transmitting station; however, it is generally $2H$. Horizontal lines 10 through 21 of each field are often used to send studio test signals an automatic color and brightness signals.

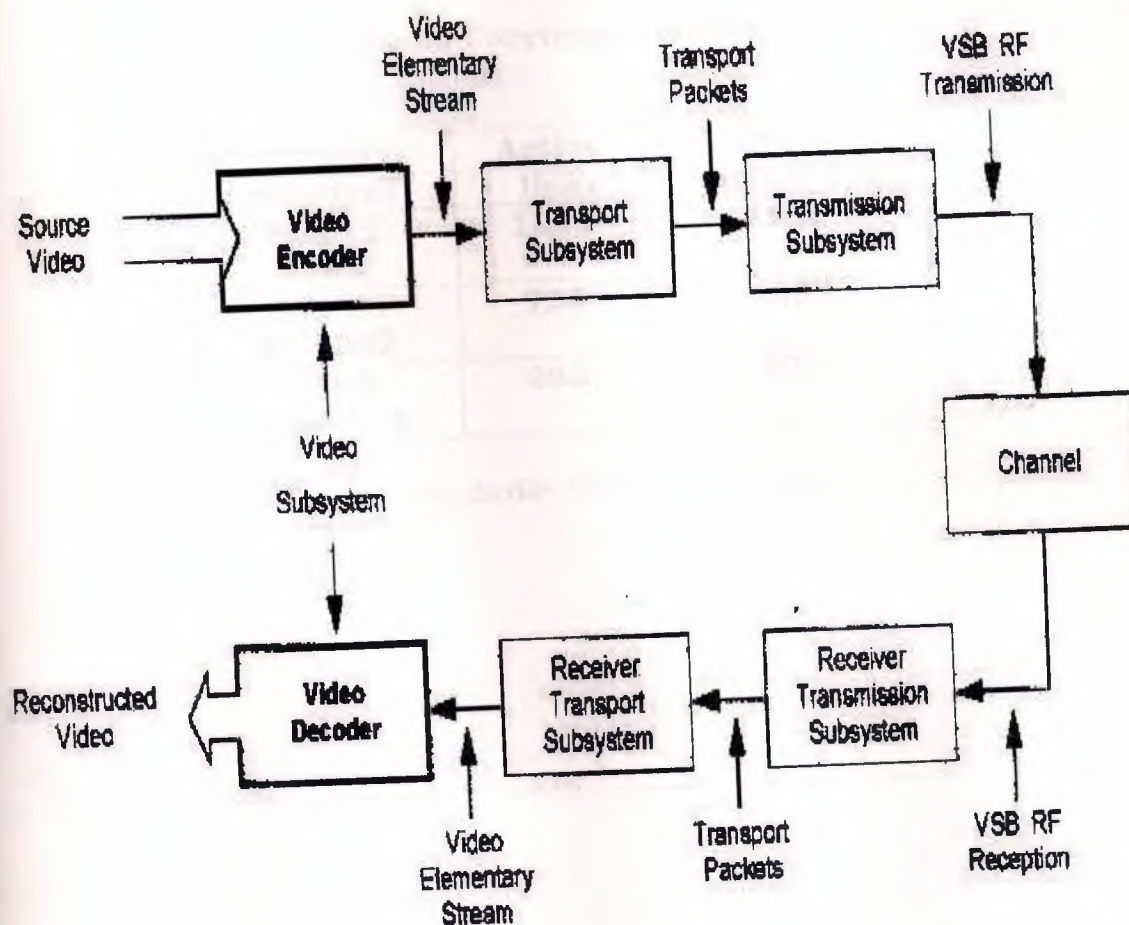
4.4.3 RF TRANSMISSION OF THE COMPOSITE VIDEO

The transmitted AM picture carrier is shown in Figure for negative polarity modulation, which is the FCC standard. Negative transmission of the RF carrier simply means that changes toward white in the picture decrease the amplitude of the AM picture carrier. The advantage of negative transmission is that noise pulses in the RF signal increase the carrier toward black, which makes the noise less annoying to the viewer the changes toward white. Also, with negative transmission, brighter images occur not often than darker ones: thus, negative transmission uses less power than positive transmission. Notice that the AM envelope shown in Figure has the shape of the composite video signal, and the luminance, blanking, and sync signals can be easily identified. Also, note that during the tips of the horizontal sync pulse there is no AM modulation, and the luminance signal never exceeds 87.5% AM modulation.

4.5 OVERVIEW OF VIDEO COMPRESSION

The video compression system takes in an analogue video source signal and outputs a compressed digital signal that contains information that can be decoded to produce an approximate version of the original image sequence. The goal is for the reconstructed approximation to be imperceptibly different from the original for most viewers, for most images, for most of the time. In order to approach such fidelity, the algorithms are fizable, allowing for frequent adaptive changes in the algorithm depending on scene content, history of the processing, estimates of image complecity and perceptibility of distortions introduced by the compression.

Figure 4.5 shows the overall flow of signals in the ATV system. Note that analogue signals presented to the system are digitised and sent to the encoder for compression, and the compressed data then are transmitted over a communications channel. On being received, the possibly error-corrupted compressed signal is decompressed in the decoder, and reconstructed for display.



Video Coding in Relation to the ATV system

Figure 4.5

4.6 VIDEO PRE-PROCESSING

Video pre-processing converts the analogue input signals to digital samples in the form needed for the subsequent compression. The analogue input signals are red (R), green (G), and blue (B) signals.

4.6.1 Video compression formats

Vertical line	Pixels	Aspect ratio	Picture rate
1080	1920	16:9	60I,30P,24P
720	1280	16:9	60P,30P,24P
480	704	16:9 and 4:3	60P,60I,30P,24P
480	640	4:3	60P,60I,30P,24P

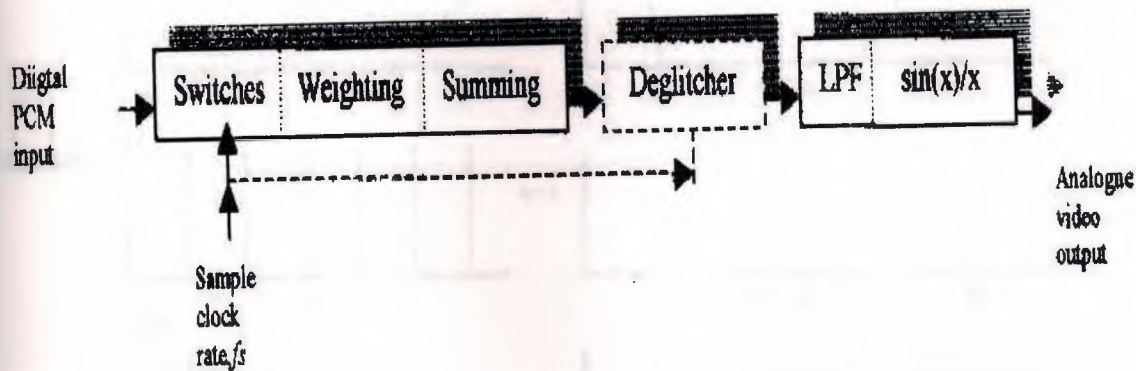
Table 4.1 Compression Formats.

Video standard	Active lines	Active samples/line
SMPTE 274M	1080	1920
SMPTE S17.392	720	1280
ITU-R BT.601-4	483	720

Table 4.2 Standardies Video Input Formats.

4.7 DIGITAL-TO-ANALOGUE (D-A) CUNVERSION

In video digital-to-analogue (D-A) converters, each incoming binary codeword is converted to a quantised analogue sample synchronous-of a group of switches operated by the digital input and associated weighting and summing networks, as indicated in Figure If the most significant bit (MSB) contributes voltages 0 or 12V to the output analogue sample, the lower significant bits contribute voltages 0 or $V/4, V/8, V$, etc. The resulting, analogue sample voltages obtained from video D-A converters are normally held substantially constant for an entire sampling period thus forming a stepwise waveform with each step of width. The required continuous video waveform is obtained from this stepwise waveform by passing the output from the D-A converter through a low-pass filter and a 'sin(x)/x' equaliser.



Basic elements of D-A converter and associated processes.

Figure 4.5

In practice, the leading edges of the steps corresponding to different code changes can have different overshoots or undershoots, known as glitches. These glitches can be removed by a difficult to label deglitcher in figure 4.5. With careful design, the energy in the glitches can be made insignificant, thus avoiding the need for a deglitcher.

4.8 SAMPLING AND FREQUENCY CHARACTERISTICS

The theoretical sampling frequency and filtering requirements of A-D and D-A conversion will be explained by considering the spectra of the video signal at various stages in the process. For convenience, it is assumed that the spectrum, of the video signal applied to the A-D converter is defined entirely by the preceding low-pass filter whose amplitude/frequency characteristic has a flat pass-band and sharp cut-off at frequency f_c , as illustrated in Figure 4.6. Sampling of the corresponding video signal $E_p(t)$ as shown in Figure 4.6

is equivalent to multiplying $E_i(t)$ by a train of constant amplitude pulses $E_p(t)$ as shown in Figure 3. the resulting sampled signals being shown in Figure 3e. In the frequency domain, the multiplication of $E_i(t)$ by $E_p(t)$ results in a spectrum which contains components at frequencies equal to the sum and difference of all the components in the spectra of $E_i(t)$ and $E_p(t)$. By Fourier analysis, the spectrum of $E_p(t)$ contains components of constant amplitude of all frequencies given by nfs where n is any integer. Thus the spectrum of video samples $E_s(t)$ is as shown in Figure 4.6.

As mentioned previously, the output from a D-A converter $E_o(t)$ is a stepwise waveform as shown in Figure 4.6. This waveform consists of a train of pulses of the same amplitude as the video samples $E_s(t)$ but having a constant width. The effect of the change on pulse shape sync the spectrum on the signal can be explained as follows.

Spectra of waveforms involved in A-d and D-a conversions.

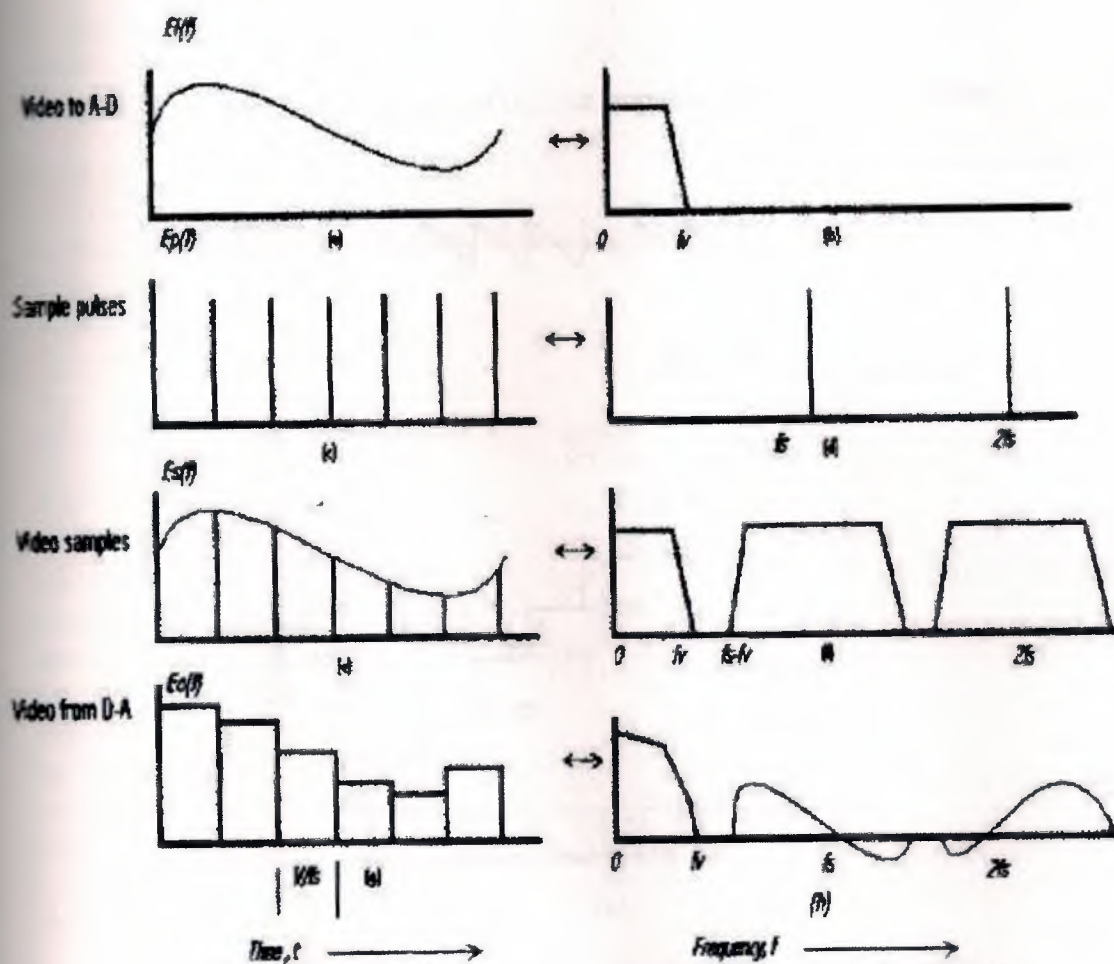


Figure 4.6

Each impulse in $E_s(t)$ has a flat spectrum at any frequency as shown in Figure 4(a) whereas each rectangular pulse in $E_o(t)$ has a spectrum $A(f)$ of the form shown in Figure 4(b) given by

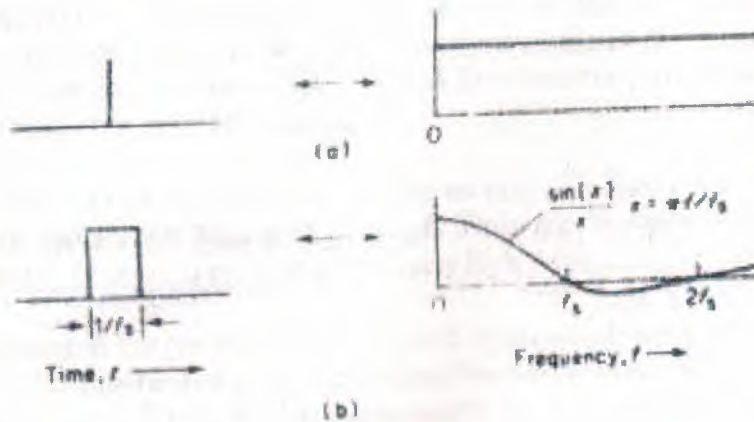
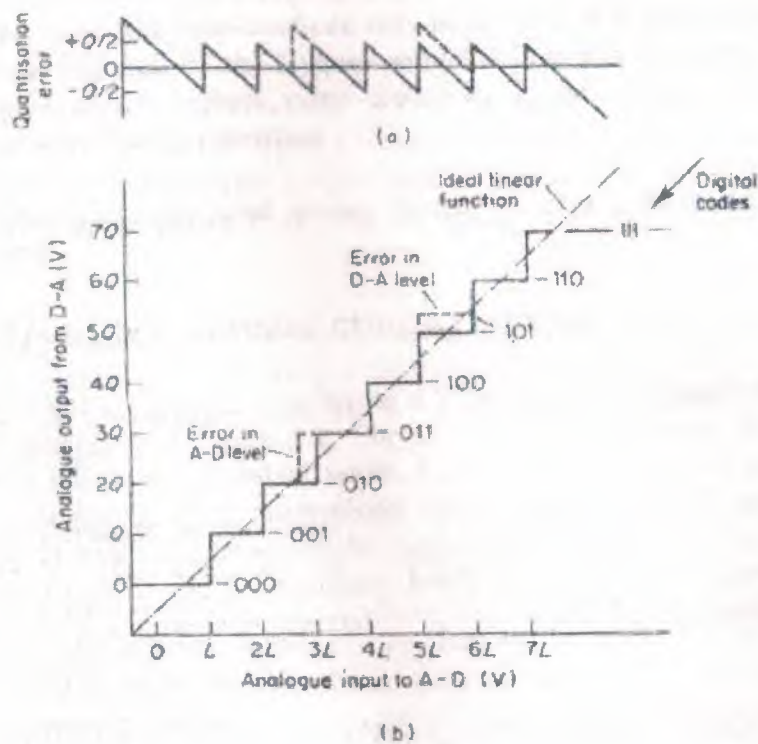


Figure 4.7 Spectra of (a) Unit Impulse, (b) Rectangular Pulse



Input/Output Characteristic of 3-bit A-D and D-A Converters.

Figure 4.7

It follows that the process of deriving the D-A converter output $E_o(t)$ using the video samples $E_s(t)$ is equivalent to low-pass filtering $E_s(t)$ using the $\sin(x)/x$ characteristic of equation (4.6). Thus the spectrum of the video output from the D-A converter is as shown in Figure 4.6(h).

By examining the spectrum of the D-A converter output, it can be seen that a signal with the same spectrum as that of the A-D converter input can be obtained by means of a low-pass filter which removes all components above frequency f_v followed by an equaliser whose frequency response over the range 0 to f_v is the inverse of the $\sin(x)/x$ characteristic given by equation. It follows that, if the spectra of the A-D converter input and D-A converter output are the same, the video waveform of the D-A converter output will be the same as the waveform at the input to the A-D converter prove that that no group-delay (phase) distortion has been introduced.

Note that the $\sin(x)/x$ characteristic of the rectangular pulses has nulls at the sampling frequency f_s and integer multiples thereof. Thus the low-pass filter following the D-A converter does not need to provide particularly high attenuation at those frequencies.

The components in the spectrum of a sampled signal which are centred on frequencies f_s , $2f_s$, etc, i.e. all components except the original base-band components, are often referred to as alias components. The term 'alias components' has no strict definitions however, as far as band-width is concerned and is sometimes used to refer to only those unwanted components which are within the base-band frequency range.

Since the lowest frequency alias component is at $f_s - f_v$, it can be seen that the presence of alias components within the base-band can only be avoided if $f_s - f_v$ is greater than f_v , i.e. if in other words, as required by the Nyquist sampling criterion there must be at least two samples per cycle of the highest video frequency if the original waveform is to be reconstructed without aliasing distortion.

In practice, f_s should be somewhat greater than $2f_v$ to cope with the finite rate of cut-off of realisable filters.

4.8.1 Input/output transfer characteristics

The input/output voltage transfer function of a PCM codec (combination of A-D and D-A converters) is illustrated in Figure 4.7. For an ideal A-D converter, all the steps in the transfer function will have a constant width L , and for an ideal D-A converter they will have a constant height Q . Thus, instrumental inaccuracies in the A-D and D-A converters cause variations in width and height of the steps respectively. For video signal processing, any constant dc offset is the A-D decision levels or all the D-A quantum levels are not important because it can be compensated by a corresponding dc shift in the applied or decoded video signals.

4.9 METHODS OF SPECIFYING QUANTISING ERRORS

It can be seen from Figure 4.7 that the quantising errors caused by an ideal codec, i.e. the differences between its input/output transfer characteristic and a straight line drawn to minimise these differences, have peak values of $Q/2$ and are uniformly distributed between these peak values. There is no magnitude of any set of quantising errors with these peak values and distribution is equal to $Q/\sqrt{12}$.

In expressions for signal-to-quantising-noise ratios, the peak-to-peak output of a D-A converter fed with n -bit code words is normally taken to be equal to Q .

although its precise value is 2⁻¹. Using this approximation, the peak-to-peak signal S_{p-p} to r.m.s. quantising noise Q_{rms} ratio of an ideal codec is given by

$$S_{p-p} = 20 \log_{10} (\sqrt{2^n} Q \times 12 / Q)$$

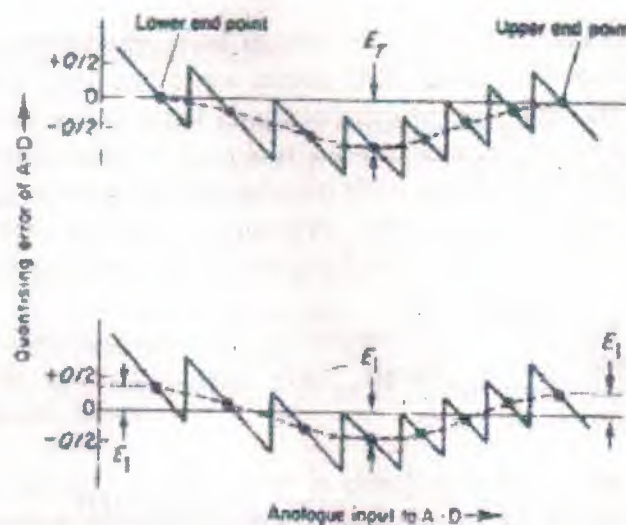
$$Q_{rms} = 6.02n + 10.8 \text{ dB}$$

This calculation does not take into account the bandwidth of the quantising noise and the effects of sampling. However, the magnitudes of quantising errors on different samples are uniformly distributed between $Q/2$ in a random manner. It can be shown that the spectrum of the quantising noise is flat and its r.m.s. Magnitude in the range 0 to $f_s/2$ remains equal to $Q/\text{square-root } 12$. The above conditions for quantising error distribution are approached more closely as the number of bits per code-word increases and they are substantially satisfied by the number of bits nominally used for coding video signals. Thus when the quantising noise bandwidth is reduced to a video bandwidth f_v less than $f_s/2$, the signal-to-quantising noise ratio is increased by $10 \log_{10} (f_s/2f_v)$ with respect to equation 3.

A further factor to be considered is that the magnitude of the video signal is conventionally taken to be equal to the difference in voltage between black level V_b and white level V_w . Thus, assuming that V_w and V_b all refer to the same point in the signal chain, the video signal r.m.s. quantising noise ratio of an ideal n -bit codec is given by Video signal-to-quantising noise ratio.

$$= 6.02n + 10.8 \pm 10 \log_{10} (f_s / 2f_v) - 20 \log_{10} [(V_w - V_b) / S_{p-p}]$$

A comparison of ideal and practical video signal-to-quantising noise ratio provides a very useful guide to the performance of video A-D and D-A converters. A particularly useful feature of signal-to-quantising-noise measurements is that they can be obtained in the presence of rapidly changing video signals. As a result, they take into account incorrect coding of high-frequency signals in addition to static errors in the quantisation characteristic.



Other methods of specifying the accuracy of converters are:

Statistic' errors (or low frequency) quantising errors are usually specified in terms of 'linearity' Figure 4.8 Integral based linearity errors of A-D converter errors which for A-D converters, are based on the deviation of the mid-points from their ideal ideal position are known as integral linearity error other terms used for defining linearity errors are given below.

Positions assuming perfect D-A conversion. The deviation of these midpoints from their ideal positions is known as signal linearity errors'. Other terms used for defining linearity errors are given below.

Terminal-based integral linearity error. $E(t)$ is maximum linearity error measured with respect to a straight line drawn through the end-points of the input/output characteristic as illustrated in Figure 4.8. This is the most easily measured and calibrated integral linearity error measurement.

Independent integral linearity error. $E(t)$ is the maximum linearity error measured with respect to a straight line drawn to minimise the peak error as illustrated in Figure 4.8. This is the most useful specification for video A-D converters where (absolute gain and offset are not critical).

Differential linearity error is the difference in the distance between two adjacent step mid-points and the step width of an ideal characteristic.

Code size error is the error in the distance between adjacent levels i.e. between adjacent vertical transitions

Quoted specifications for the linearity error of video A-D converters without further qualification e.g. linearity error is 0.2%, of full range normally refer to the independent integral linearity error.

Another specification of linearity often quoted for video A-D and D-A converter is their peak to-peak differential gain and phase errors. Differential gain specifications indicate the variation in slope of the input/output characteristic averaged over the range of levels given by the peak-to-peak amplitude of the colour sub-carrier used in the video test signal. As a result of the averaging process, differential gain specifications give a better guide to overall linearity than, is given by integral linearity error specifications which indicate the maximum error occurring at any single quantum level.

In addition to having a satisfactory static performance video A-D and D-A converters must be able to accurately convert rapidly video signals. Factors affecting this dynamic performance are discussed below.

If the analogue input of an A-D converter is changing rapidly, any time-error T in the sampling process causes a magnitude error T as indicated in Figure 4.9. For a sinusoidal input of frequency f and peak-to-peak magnitude xV_{fs} where V_{fs} is the full scale magnitude of the conversion range and g lies in the range 0 to 1, the maximum value of V is given by the expression.

$$V = n \times V_{fs} \Delta T \quad (5)$$

As an example of the significance of limiting errors, equation 5 shows that a value of T equal to 250 ps will cause a magnitude error V equal to about $V_{fs}/256$ when a full-screen 5 MHz video signal is being digitally encoded.

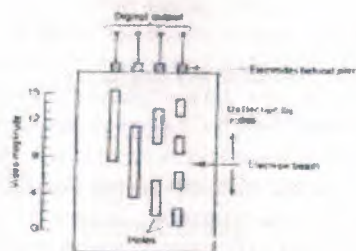
There are a number of terms containing the word 'aperture' which are used to refer to inaccuracies in A-D conversion obtained only in the presence of rapidly changing analogue signals. These coding errors may occur either because the signal is sampled at an incorrect time because of instrumental deficiencies, such as too slow a response time or slew-rate limitations in A-D circuit elements. The most commonly terms used are aperture-time, aperture-jitters, aperture uncertainty and aperture error. There does not seem to be any generally agreed precise meaning for any of these terms and different manufacturers of A-D converters use different terms for, apparently the same type of error. Difficulties arise because different measurement techniques can yield different results. Thus the application of equation 5, to a quoted aperture error T , or similar term, normally gives only a general guide to performance with high frequency input signals and it will not indicate, for example, that some codes are never generated under these circumstances although they may be generated with slowly changing input signals. In practice the author has found that the most useful specification of high frequency performance is given by signal-to-quantising-noise measurements made in the presence of high frequency video signals as discussed in figure 4.9

4.10 METHODS USED FOR A-D AND D-A CONVERSION

4.10.1 A-D conversion techniques

The earliest equipment capable of performing reasonably accurate PCM encoding of television signals was almost certainly that constructed in the Bell Research Laboratories in about 1948-9. The quantisation process in this equipment was performed by means of a specially constructed cathode ray tube in which the anode consisted of a plate containing a pattern of holes of the form shown in Figure This plate was scanned by a flat ribbon-shaped beam of electrons whose deflection was proportional to the magnitude of the analogue video signal. When the electron beam passed through any opening in the plate, it hit one of the electrodes situated behind the plate causing a digital 1 to appear at the output. Separate electrodes for each output digit were placed behind each vertical column of holes.

Figure 4.10



A 'Gray code' rather than binary code pattern of holes was selected because of this property that only one-digit changes in any transition between adjacent code levels. Thus quantising errors caused by coding uncertainty when the beam was

mid-way between two code levels were restricted one quantum level. With a binary pattern, large errors could easily be obtained under these conditions; for example an output of 0000 or 1111 could be obtained for a beam position mid-way between 0111 and 1000. The Gray code output from the electrodes was converted to binary code after it had passed through digital latches triggered by clock pulses. This early equipment was capable of providing 5-bit code words at a sampling rate of 10 MHz.

Little further work was carried out for the next ten years until the arrival of high-frequency transistors to replace thermionic valves provided the possibility of compact equipment capable of making effective use of digital video signals. An improved version of Q beam coding tube equipment, with transistors used at least partly in the associated circuitry, was completed in about 1959. This equipment gave 7-bit code words at a sampling rate of 10 MHz and was used successfully in experimental transmissions of digital video signals over telephone cable pairs. Further improvements led to the construction in about 1963 of high-accuracy 9-bit equipment capable of operating at sampling rates up to at least 12 MHz. In this last equipment employing the beam coding tube, associated circuitry was entirely solid state, discrete transistors being employed since integrated circuits were not yet available.

At about the same time that the final version of the beam coding tube was being developed, Bell Laboratories started work on the design of a solid state quantiser, a block diagram which is shown in Figure 4.11. Each 'folding' amplifier stage in this quantiser performed two functions.

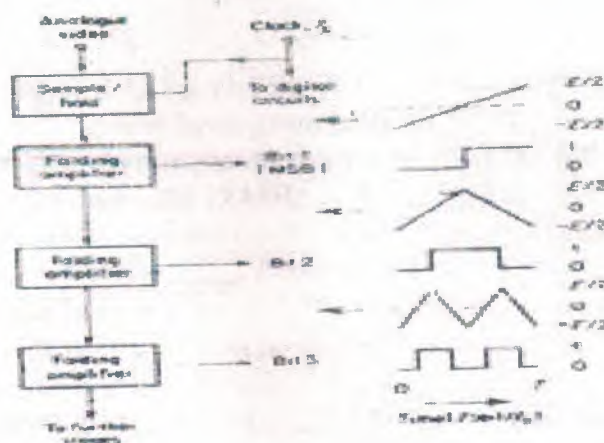


Figure 4.11

Serial A-D converter using folding amplifier.

- It gave out a digital one or zero depending on whether the analogue input was positive or negative.
- It delivered an analogue residue signal to the next stage which was a full-wave rectified version of the input, the output being non-inverted for negative inputs and inverted for positive inputs as shown in Figure 4.11. This rectification process was achieved by means of diodes in the feedback paths of operational amplifiers.

The residue was suitably biased and amplified, so that it ranged between the sample peak values as the input.

As in the seam coding tube equipment the initial digital signal obtained from the folding amplifiers was in the form of a Gray code. In addition to the benefits mentioned above provided by the use of a Gray code the folding process producing this code also had the useful feature that slowly changing video signals did not provoke large rapid transitions in the residue signal as in the binary code subtraction technique discussed below. Such transitions are difficult to transmit undistorted through a large number of cascaded amplifiers. Nevertheless the bandwidth of the amplifiers has to be much greater than that of the original analogue video signal in order to handle satisfactorily the rapid transitions obtained with rectified high frequency video signals. Difficulties in encoding high frequency video signals were overcome with the assistance of an initial sample-and-hold

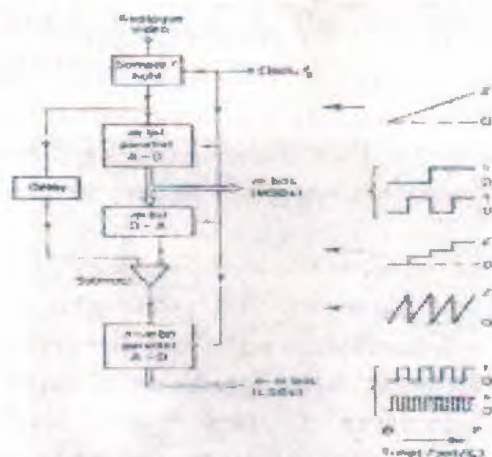


Figure 4.12 Parallel-Serial n-Bit A-D converter
(waveforms given for $n=4$, $m=2$).

circuit. This folding A-D converter provided 9-bit code-words and operated successfully at sampling frequencies up to at least 12 MHz.

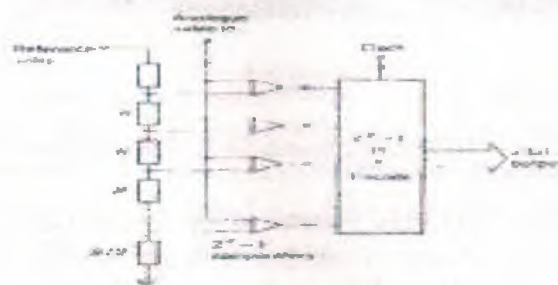


Figure 4.13
Parallel or 'flash' A-D converter

While this folding coder was being developed in the Bell Laboratory, work commenced in the BBC Research Department on the construction of a different type of solid state coder, a block diagram of which is shown in Figure 4.12. In this parallel-serial type of coder, the first parallel A-D section coarsely digitises the analogue input to give the m most significant bits of the required n -bit code-words. After the first m bits have been determined, they are converted back to an analogue signal having 2^m quantisation levels and this quantised signal is then subtracted from the unquantised input. Finally the resulting difference signal is fed in

a second parallel AD section giving the $n-m$ least significant bits. In a two-stage converter of this type, it is instrumentally convenient to make m to equal to n/Z .

As illustrated in Figure 4.13 parallel A-D section contains $2x - 1$ level comparators where x is the number of bits to be generated. These comparators are fed in parallel with the video signal on one of their inputs and reference voltages, uniformly spread over the conversion range, on their other inputs.

In the first parallel-serial converter, 3 bits were generated in each parallel stage giving a total of 6 bits per sample. In theory, a single stage parallel converter could have been used on its own to give all 6 bits, but the number of level comparators required, was inconveniently large in the days when all the circuitry had to be constructed from discrete transistors. Even the use of seven level comparators per stage led the somewhat unwieldy equipment as illustrated in Figure 4.14. The card being held by the author is a single comparator and tunnel diode latch.



Figure 4.14

Prototype BCC A-D converter constructed in 1965.

Another alternative would have been to employ only one level comparator in each of six stages. This approach was not adopted because of the difficulty of accurately transmitting the rapid transitions in the difference signal through five or more analogue subtraction connected in cascade.

The 3+3 bit equipment which was constructed operated satisfactorily at sampling rates up to at least 13 MHz, good results being obtained with composite PAL colour signals including high amplitudes of the 4.43 MHz colour sub-carrier.

The BBC parallel-serial equipment was re-engineered about four years later in 1969. This new equipment made use of small-scale integrated circuits which had recently become available and derived 4 bits per stage to give 8 bit code-words at sampling frequencies up to about 18 MHz. Further improvements enable correct 8 bit coding to be obtained with decision levels in the first stage significantly less accurate than 1 LSB ($1/256$ of full range)

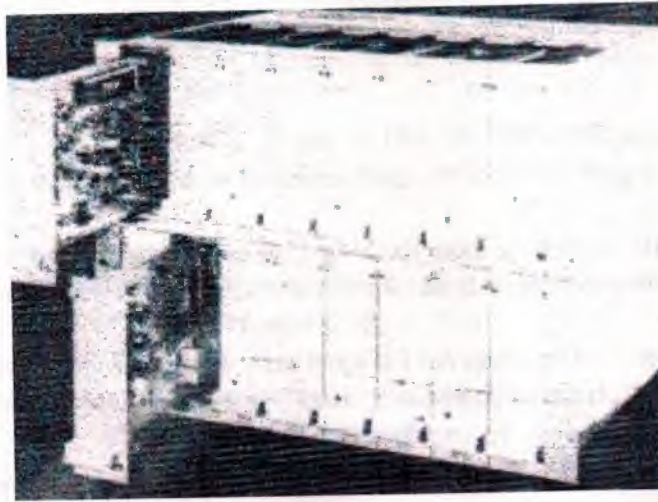


Figure 4.15

Collection of eight A-D and eight D-A units constructed in 1980 used for tests.

4.10.2 D-A conversion techniques

The practical difficulties involved in the D-A conversion of television signals are less severe than those involved in A-D conversion. All the well-known techniques include a number of electronic switches operated by the digital input together with analogue networks for weighting in summing the outputs of these switches.

Generally, the weighting and summing networks are of two types, namely; those where



Figure 4.16

weighted currents are added in a single resistor as shown in Figure 4.16 and those where the same magnitude currents are applied to different nodes of a resistor ladder network which is normally arranged as an R-2R ladder as shown in Figure 4.16. Variations of the R-2R ladder technique include the use of switches voltage rather than current sources. The voltage sources are applied via 2R resistors to a series chain of resistors with end values of 2R and intermediate value of R.

Bell Laboratories employed an ladder network in the early 1960s to provide accurate 9 bit D-A conversion at sampling rates up to at least 12 MHz. The first BBC D-A converter constructed a few years later employed the weighted current and single resistor technique. Modern single-chip converters still use one of these two techniques.

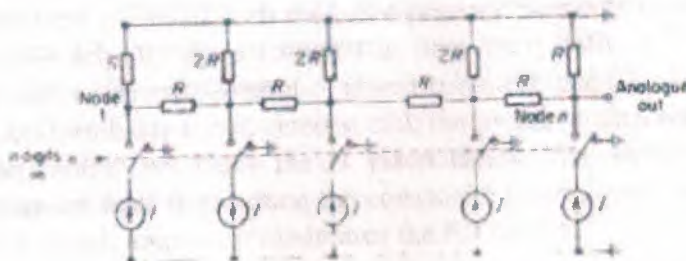
The switched ladder network is generally preferred because matching and tracking problems are eased by the fact that all resistors and current in the weighting network have similar values.

A desirable feature of the R/2R ladder network is that the input resistance at each node is constant ($=2R/3$). Thus, since each node is driven from an identical design of switch, the

delay and rise-time of the signals generated at each node should in theory be constant. Inequality in these delays and/or rise-times is undesirable as it causes overshoots or

figure 4.18

undershoots, known as glitches, in the output from a D-A converter at steps obtained when a number of digits change simultaneously. These glitches are particularly noticeable when



all digits change simultaneously at the middle of the conversion range e.g., when a 8bit digital input changes from 4111 to 1040. resulting effect is illustrated in Figure 4.18.

If the duration of the glitches is short compared to a sampling interval, they can be removed by a following sample and hold circuit which samples the analogue output during the flat part of each step and holds the signal level constant during the transitions. However, considerable care must be taken in the design of this deglitching circuitry and associated buffer amplifiers in order to ensure that no slew-rate distortion is introduced. Any such distortion causes inter-modulation between the base-band frequency video components and components centred on the sampling frequency and its harmonics. The resulting intermodulation products can readily have frequencies lying within the wanted base band and are, therefore, not removable by the following video low-pass filters. The best way to avoid these problems is to ensure that no significant glitches are generated so that no deglitching circuit is required and to apply video low-pass filtering to the D-A converter output before it passes through any analogue amplifier.

CHAPTER 5 : COLOUR TELEVISION

5.1 COLOUR TELEVISION TRANSMISSION & RECEPTION

5.1.1 Colour Television Transmitter

In essence, a color television transmitter is identical to the black-and-white transmitter, except that a color camera is used to produce the video signal. With color broadcasting, all the colors are produced by mixing different amounts of the three primary colors: red, blue, and green not to be confused with the three primary pigments cyan, magenta, and yellow. A color camera is actually three cameras in one, each with separate video output signals. When an image is scanned, separate camera tubes are used for each of the primary colors. The red camera produces the R video signal, the green camera produces the G video signal, and the blue camera produces the B video signal. The R, G, and B video signals are combined in an encoder to produce the composite color signal, which when combined with the luminance signal, amplitude modulates the RF carrier.

5.1.2 Colour Camera

Figure 5.1 shows a configuration of mirrors that can be used to split an image into the three primary colors. The chromatic mirrors reflect light of all colors. The dichroic mirrors are coated to reflect light of only one frequency (color) and allow all other frequencies (colors) to pass through. Light reflected from the image passes through a single camera lens, is reflected by the two achromatic mirrors, and passes through the delay lens. Dichroic mirrors A and B are mounted on opposing 45° angles. Mirror A reflects red light, while blue and green light pass straight through to mirror B. Mirror B reflects blue light and allows green light to pass through. Consequently, the image is separated into red, green, and blue light frequencies. Once separated, the three color frequency signals modulate their respective camera tubes and produce the R, G, and B video signals.



Figure 5.1

5.1.3 Colour Encoding

Figure 5.2 shows a simplified block diagram for a color television transmitter. The R, G, and B video signals are combined in specific proportions in the color matrix to produce the brightness (luminance) or Y video signal and the I and Q chrominance (color) video signals. The luminance signal corresponds to a monochrome video signal. The I and Q color signals amplitude modulate a 3.58 MHz color sub-carrier to produce the total color signal, C. The I signal modulates the sub-carrier directly in the I balanced modulator, while the Q signal modulates a quadrature (90 degree out of phase) sub-carrier in the Q balanced modulator. The I and Q modulated signals are linearly combined to produce a quadrature amplitude modulation (QAM) signal, C, which is a combination of both phase and amplitude modulation. The C signal is combined with the Y signal to produce the total composite video signal (T).

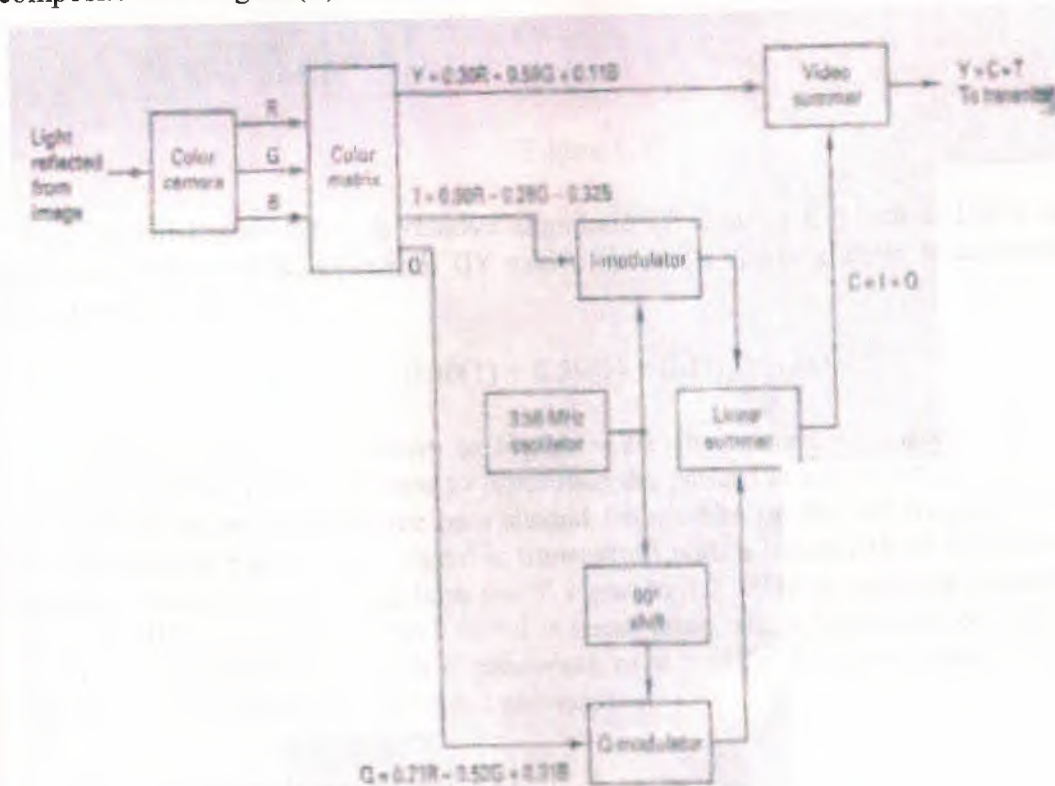


Figure 5.2

5.1.4 Luminance signal

The Y or luminance signal is formed by combining 30% of the R video signal, 59% of the G video signal, and 11 % of the B video signal. Mathematically, Y is expressed as

$$Y = 0.30R + 0.59G + 0.11B \quad (1)$$

The percentages shown in Equation 1 correspond to the relative brightness of the three primary colors. Consequently, a scene reproduced in black and white by the Y signal has exactly the same brightness as the original image. Figure 5.3 shows how the Y signal voltage is formed from several values of R, G, and B.

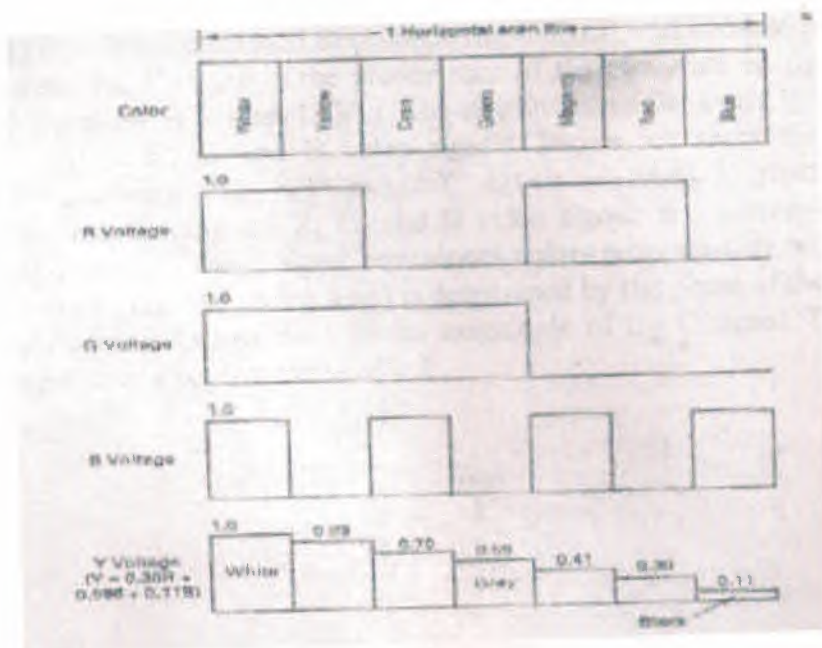


Figure 5.3

The Y signal has a maximum relative amplitude of unity or 1, which is 100% white. For maximum values of R, G, and B (1V each), the value for brightness is determined, from Equation 1 as follows:

$$Y = 0.30(1) + 0.59(1) + 0.11(1) = 1.00$$

The voltage values for Y shown in Figure ~ are the relative luminance values for each color. If only the Y signal is used to reproduce the pattern in a receiver, it would appear on the CRT as seven monochrome bars shaded from white on the left to gray in the middle and black at the right. The Y signal is transmitted with a bandwidth of 0 MHz to 4 MHz. However, most receivers band-limit the Y signal to 3.2 MHz to minimize interference with the 3.58-MHz color signal. The I signal is transmitted with a bandwidth of 1.5 MHz, and the Q signal is transmitted with a bandwidth of 0.5 MHz. However, most receivers limit both the I and Q signals to 0.5-MHz bandwidth.

5.1.5 Chrominance signal

The chrominance or C signal is a combination of the I and Q color signals. The I or in-phase color signal is produced by combining 60% of the R video signal, 28% of the inverted G video signal, and 32% of the inverted B video signal. Mathematically, I is expressed as

$$I = 0.60R - 0.28G - 0.32B \quad (2)$$

The Q or in-quadrate color signal is produced by combining 21 % of the R video signal, 52% of the inverted G video signal, and 31 % of the B video signal. Mathematically, Q is expressed as

$$Q = 0.21R - 0.52G + 0.31B \quad (3)$$

The I and Q signals are combined to produce the C signal, and because the I and Q signals are in quadrature, the C signal is the phasor sum of the two (that is, the magnitude of C is $\sqrt{I^2 + Q^2}$ and the phase is $\tan^{-1} Q/I$). The amplitudes of the I and Q signals are in turn proportional to the R, G, and B video signals. Figure 5.4 shows the color wheel for television broadcasting. The R-Y and B-Y signals are used in most color television receivers for demodulating the R, G, and B video signals and are explained later in the section. In the receiver, the C signal reproduces colors proportionate to the amplitudes of the I and Q signals. The hue (color tone) is determined by the phase of the C signal, and the depth or saturation is proportional to the magnitude of the C signal. The outside of the circle corresponds to a relative value of 1.0

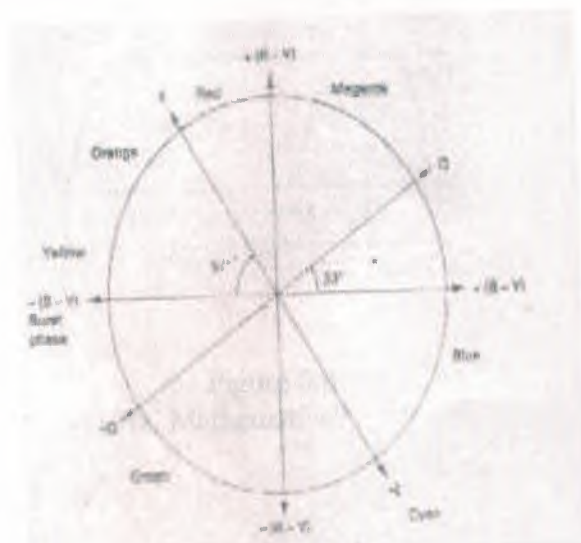


Figure 5.4

5.1.6 Colour burst

The phase of the 3.58-MHz color sub-carrier is the reference phase for color demodulation. Therefore, the color sub-carrier must be transmitted together with the composite video so that a receiver can reconstruct the sub-carrier with the proper frequency and reference phase and, thus, determine the phase (color) of the received signal. Eight to ten cycles of the 3.58-MHz sub-carrier are inserted on the back porch of each horizontal blanking pulse. This is referred to as the color burst. In the receiver, the burst is removed and used to synchronize a local 3.58-MHz color oscillator. The color burst is shown in Figure 5.5.

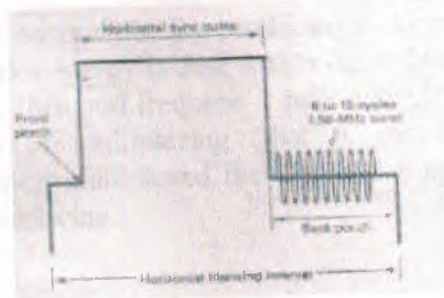


Figure 5.5

Figure 5.6 shows the composite RF frequency spectrum for color television broadcasting.

5.1.7 Scanning frequencies for colour transmission

Harmonic relations among the color sub-carrier and the horizontal and vertical scanning rates determine the frequency of the colour sub-carrier. The exact value for the color sub-carrier is 3.579545 MHz. The sound sub-carrier (4.5 MHz) is the 286th harmonic of the horizontal line frequency. Therefore the horizontal line rate (f_h) for color

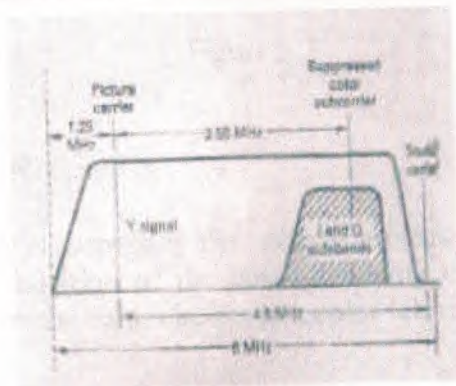


Figure 5.6

transmission is exactly 15.750 kHz. Mathematically, is

$$f_h = 4.5\text{MHz}/286 = 15,734.26\text{Hz}.$$

The exact value of the vertical scan rate (f_v) is,

$$f_v = 15,734.26/262.5 = 59,94\text{Hz}$$

The color sub-carrier frequency (C) is chosen as the 455th harmonic of one-half of the horizontal scan rate. Therefore.

$$C = 15,734.2612 \times 455 = 3,579545\text{MHz}$$

5.18 Frequency interlacing

The Y portion of the video signal produces cluster of energy at 15.73426-kHz intervals throughout the 4-MHz video bandwidth. By producing color signals around a 3.579545-MHz color sub-carrier, the color energy is distributed within the void intervals between the black-and-white information. This is clustered frequency interlacing or sometimes frequency interleaving and is a form of multiplexing (that is the color and black-and-white information is frequency-division multiplexed the total video spectrum). Figure 5.7 shows the spectrum for frequency interlacing.

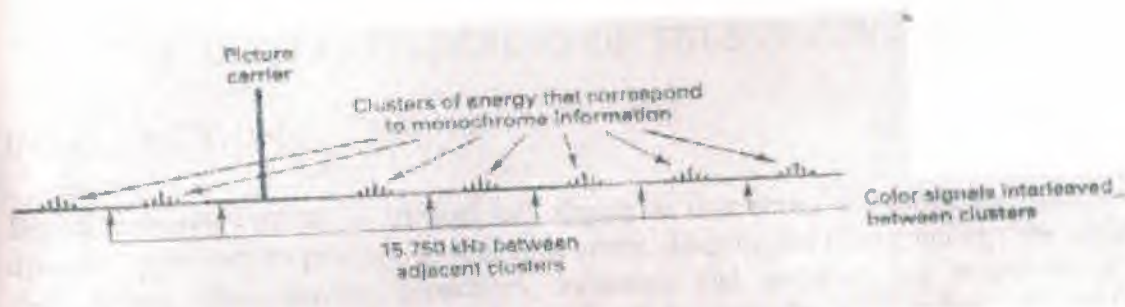


Figure 5.7

5.2 Colour Television Receivers

A color television receiver is essentially the same as a black-and-white receiver except for the picture tube and the addition of the color decoding circuits. Figure 5.8 shows the simplified block diagram for the color circuits in a color television receiver.

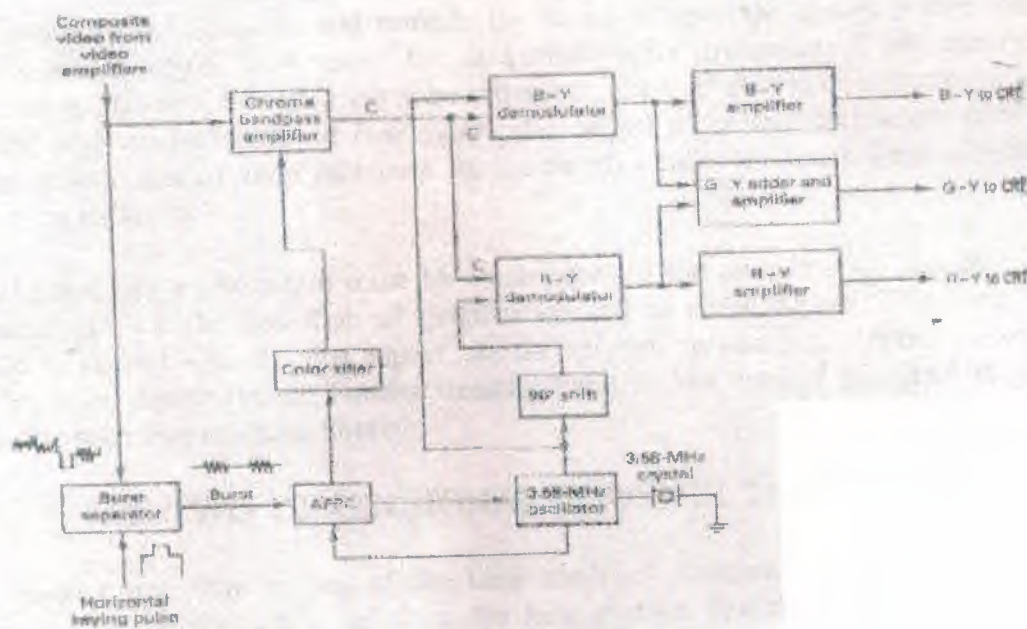


Figure 5.8

The composite video signal is fed to the chroma band-pass amplifier, which is tuned to the 3.58-MHz sub-carrier and has a band-pass of 0.5 MHz. Therefore, only the C signal is amplified and passed on to the B-Y and R-Y demodulators. The 3.58-MHz color burst is separated from the horizontal blanking pulse by keying on the burst separated only during the horizontal fly-back time. A synchronous 3.58-MHz color sub-carrier reproduced in the color AFC circuit, which consists of a 3.58-MHz color oscillator color AFPC (automatic frequency and phase control) circuit. The color killer shuts off chroma amplifier during monochrome reception (no colors are better than wrong color). The C signal is demodulated in the B-Y and R-F demodulators by mixing it with phase-coherent 3.58-MHz sub-carrier. The B-F and R-F video signals produced the R and video signal by combining them with the F signal.

5.3 DIGITAL FILTERING OF TELEVISION SIGNALS

INTRODUCTION

Digital technology allows us to deal with signals as time series and perform mathematical operations on them to produce further time series. Thus digital filters, unlike their analogue counterparts which involve capacitors, inductors and resistors, are based on precise mathematical relationships. As such, their behaviour is independent of hardware imperfections and their performance is predictable (within limits) and, necessary, can be simulated using a digital computer to model the design.

The term 'digital filter' is implicitly assumed to be a device which effects the spectral characteristic of a signal. Within this broad definition there is a large class of filters whose properties are independent of the signal values (linear) and of time (timeinvariant) and this chapter will be mainly concerned with these.

Over the past ten years the subject of digital filtering has been studied intensively and there is now an enormous body of literature concerned with it. This topic can do no more than introduce the concepts and provide the barest outline, the theory before introducing some practical applications. For a comprehensive treatment of the theory. One particular application, that of video noise reduction, although based on a particular kind of filtering, is a good example of how many other signal processing techniques need to be drawn upon to solve certain problems. As this merits a fuller treatment forms a substantial part of the topic.

At this point, a distinction must be made between the operation of sampling and its consequences and the operation of quantisation and its consequences. This topic will be mostly concerned with the first aspect. As the spectral domain is the domain of interest we will begin by examining the Fourier transform and, in the context of video filtering, its particular relevance to image filtering.

5.4 THE TWO DIMENSIONAL FOURIER TRANSFORM

The Fourier transform is one of the basic tools of communication engineering which allows us to transform a signal from the time domain into the frequency domain. In this domain the operation of filters then became far more trackable.

Just as a one-dimensional function, such as a sound, signal, may be expressed as a superposition of one-dimensional sinusoids, a two-dimensional function, representing the brightness of an image, may be expressed as a superposition of two dimensional sinusoids or spatial frequencies. These may be visualised as sloping sinusoidal gratings having integral numbers of cycles per picture width and height: Figure 5.9 shows the first few members of the set. The image brightness $E(x,y)$ is then related to the amplitudes and phases of the frequencies by the equation

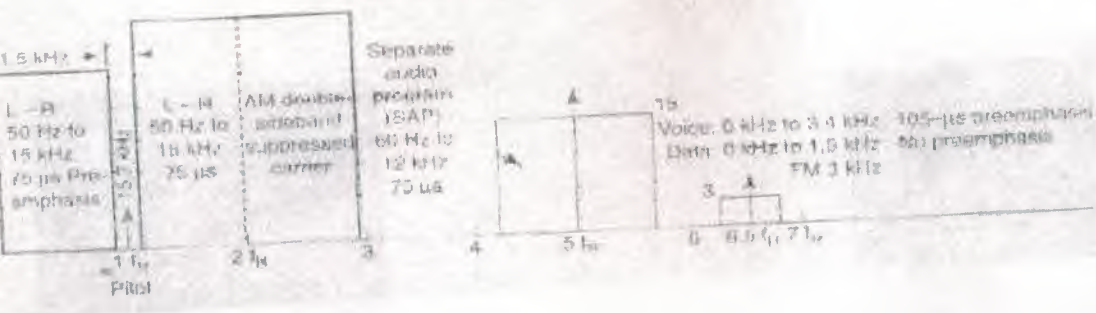


Figure 5.9

where a_{mn} and Δ_{mn} are the amplitude and phase of the frequency with components m and n cycles per picture width and height, and the picture dimensions are w g h . The function represented by equation 1 is two-dimensionally periodic in the picture dimensions as shown in Figure 5.10

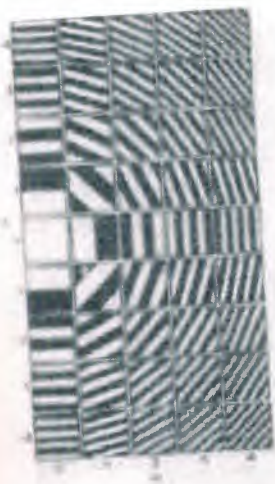


Figure 5.10

Just as with two-dimensional transforms, the physical frequency may be expressed in terms of conjugate exponential frequencies so that the amplitude and phase can be combined into a single complex number. Thus the relationship in equation 1 becomes

$$E(z,y) = \sum_{mn} a_{mn} \Delta_{mn} \exp j 2\pi (mx/w + ny/h) \sim \quad (2)$$

where

$$A_{mn} = 1/2 A_{mn} \exp j mn \text{ and } A_{-m,-n} = A_{mn}^*$$

The array A_{mn} , plotted as function of m and n , constitutes the two-dimensional Fourier transform of the image.



Figure 5.11

Note that A_{nn} is a complex quantity so that both an amplitude and a phase spectrum exist, the former having a centre of symmetry and the latter, a centre of antisymmetry. Figure 5.11, shows examples of the amplitude spectrum of three different images and Figure 5.12 shows the corresponding phase spectra. As can be seen it is possible to pick out certain features at the amplitude spectra. i.e. they are somewhat coherent whereas the phase spectra appear as an incoherent jumble.



Figure 5.12

The phase spectrum often gets forgotten although it is equally, if not more, important for images. For example, if the phase transform of one image is paired with the amplitude transform of another and vice versa then the inverse transforms closely resemble the originals whose phase transforms were used but have little similarity to the originals whose amplitude transforms were used, as shown in Figure 5.13. This shows that most scenes have much the same amplitude characteristic and that most of the significant information is carried by the phase transform. Figure 5.14 shows the recognition of an image from the phase transform alone where the amplitude transform is set to unity.

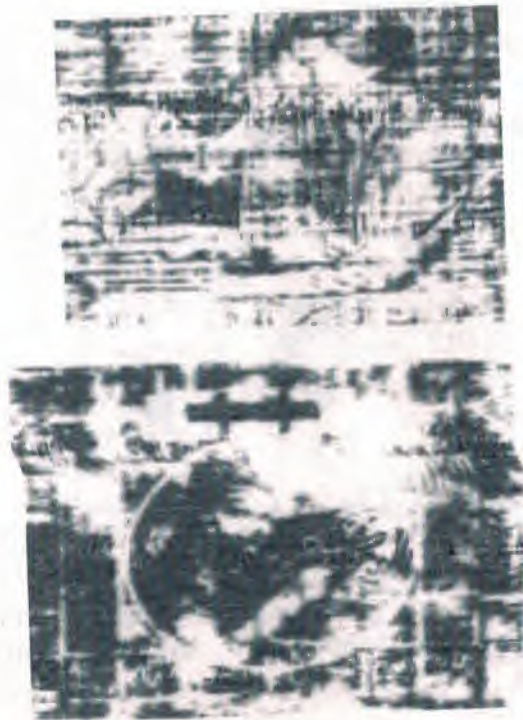


Figure 5.13



Figure 5.14

Typically, the amplitude characteristic peaks at the origin and dies away as the reciprocal of the spatial frequency magnitude in much the same way as the transform of an edge. There are exceptions, of course, when a scene contains a significant proportion of periodic detail such as test pictures or random detail such as a crowd scene, which tends to have a flatter spectrum. A further exception is where the three components of a coloured image are combined into a composite signal so that the colour information modulates a sub-carrier. The sub-carrier has a defined position in the two-dimensional transform space and develops side-bands having the same symmetry properties about the carrier as the overall spectrum.

Clearly, then, a filter designed to select or reject a particular region, or regions, of the space must have due regard to the phase characteristic, i.e. the phase transforms must be unaffected or, at worst, suffer no more than the addition of a linear function corresponding to a delay. Failure to observe this restriction results in undesirable.

Effects on edges. Figure 5.15 shows, for example, the effect of a filter which notches out a region of the spectrum near the sub-carrier and which has a poor phase response. Note the effects near vertical edges.

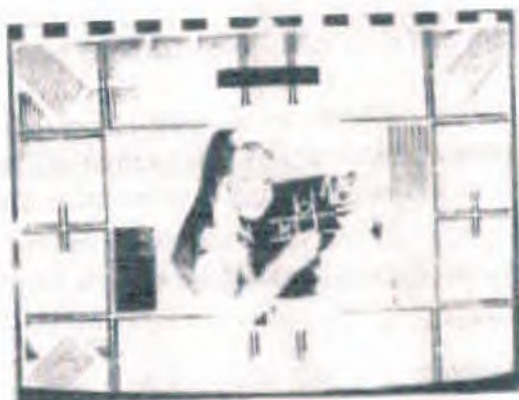


Figure 5.15

Another class of filters is designed to deliberately affect the phase transform whilst leaving the amplitude transform unaffected. This requirement usually arises in the context of signals modulated on a sub-carrier.

In the case of filters designed to pass or reject certain bands of frequencies the requirements of sharpness of cut and step-response overshoot suppression always conflict. Often the designer is faced with needing as sharp a cut as possible to satisfy some system requirement such as minimization of cross talk or loss due to repeated filter cascading. On the other hand an infinitely sharp cut causes a step to develop a slowly decaying sinusoid at the cut frequency, an effect known as 'ringing'. For example with a low-pass filter, the effect is for the edge to overshoot and undershoot by some 8.4% as shown in figure 5.16.

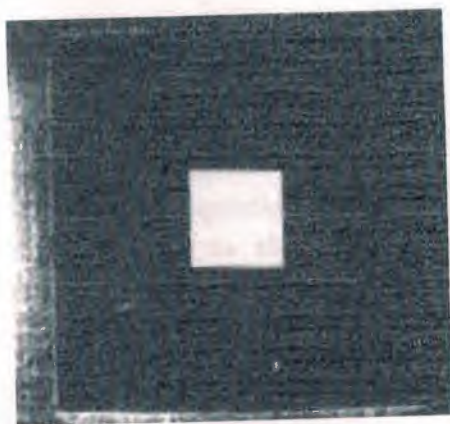


Figure 5.16

5.5 THE THREE-DIMENSIONAL FOURIER TRANSFORM

Television is concerned with two-dimensional pictures that move, so it is natural to introduce time as a third dimension. Then, in a way entirely analogous to the two-dimensional case, the three-dimensional brightness function of a moving image, $E(z,y,t)$ may be expressed as

physical function $\cos[2n(mx/w+ny/h+ft)mn]$ is a three-dimensional frequency and is thought of as a sloping grating which is moving so that crests pass a fixed point at temporal frequency of f Hz. The array A_{mn} plotted as a function of m , n and f represents the three-dimensional Fourier transform of the moving image whose amplitude phase parts have the same symmetry properties as in two dimensions.

When an image moves, the spatial frequencies of which it is composed must move in the same way. Supposing that the motion is pure translation with velocity components u and v the image brightness may be written, through equation 2 as

$$E(z-ut, y-vt) = \sum_{mn} a \sum_{mn} \Delta A_{mn} \exp j 2n (mx/w + ny/h - mut)w - nvt/h \quad (4)$$

Explaining equations 3 and 4 it is given that the relationship between the temporal frequency and the velocity is given by

$$F = -(m u/w + n v/h)$$

These equations is that of a plane in m, n, f space whose gradient is proportional to the magnitude of the velocity (if m and n are suitably scaled) and whose direction of slope is the direction of the velocity. Thus the transform of a translating image lies on a plane in the three-dimensional transform space as shown in Figure 5.17.



Figure 5.17



Figure 5.18

5.6 TUNER

The tuner, illustrated in Figure 5.19, as implemented in the prototype submitted for test, receives the 6 MHz signal (UHF or VHF) from the antenna. It is a high-side injection double-conversion type with a first IF frequency of 920 MHz. This puts the image frequencies above 1 MHz, making them easy to reject by a fixed front end filter. This selection of first IF frequency is high enough so that the input band-pass filter

selectivity prevents the local oscillator (978-1723 MHz) from leaking out the tuner front and interfering with other UHF channels, yet it is low enough for second harmonics of UHF channels (470-806 MHz) to fall above the first IF band-pass. Harmonics of cable channels could possibly occur in the first IF passband but are not a real problem because of the relatively flat spectrum (within 10 dB) and small signal levels (-28 dBm or less) used in cable systems.

The tuner input has a band-pass filter that limits the frequency range to 50-810 MHz, rejecting all other non-television signals that may fall within the tuner's image frequency range (beyond 920 MHz). In addition, a broadband tracking filter rejects other television signals, especially those much larger in signal power than the desired signal power. This tracking filter is not narrow, nor is it critically tuned, as is the case of present day NTSC tuners that must reject image signals only 90 MHz away from the desired channel. Minimal channel tilt, if any, exists due to this tracking filter.

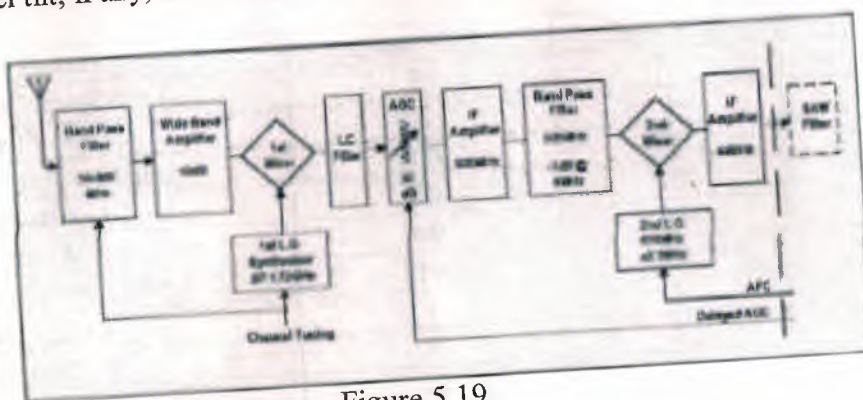


Figure 5.19

Tuner block diagram.

A 10 dB gain, wideband RF amplifier increases the signal level into the first mixer, and is the dominant determining factor of receiver noise figure 5.19 dB over entire VHF UHF, and cable bands. The first mixer, a highly linear doublebalanced design to minimise even harmonic generation, is driven by a synthesised low phase noise local oscillator (LO) above the first IF frequency (high-side injection). Both the channel tuning (first LO) and broadband tracking filtering (input band-pass filter) are controlled by microprocessor. The tuner is capable of tuning the entire VHF and UHF broadcast bands as well as all standard, IRC, and HRC cable bands.

The mixer is followed by an LC filter in tandem with a narrow 920 MHz band-pass ceramic resonator filter. The LC filter provides selectivity against the harmonic and sub-harmonic spurious responses of the ceramic resonators. The 920 MHz ceramic resonator band-pass filter has a -1 dB bandwidth of about 6 MHz. A 920 MHz IF amplifier is placed between the two filters. Delayed AGC of the first IF

Signal is applied immediately following the first LC filter. The 30 dB range AGC circuit protects the remaining active stages from large signal overload.

The second mixer is driven by the second LO, which is an 876 MHz voltage controlled SAW oscillator. It is controlled by the frequency and phase-locked loop (FPLL) synchronous detector. The second mixer, whose output is the desired 44 MHz second IF frequency, drives a constant gain 44 MHz amplifier. The output of the tuner feeds the IF SAW filter and synchronous detection circuitry.

The tuner is made out of standard consumer electronic components, and is housed in a stamped metal enclosure.

Receiver Planning Factors Used by PS/WP3

Planning factors	Low VHF	High VHF	UHF
Antenna impedance (ohm)	75	75	75
Bandwidth (MHz)	6	6	6
Thermal noise (dBm)	-106.2	-106.2	-106.2
Noise figure (dB)	10	10	10
Frequency (MHz)	69	1954	615
Antenna factor (dBm)	-111.7	-120.7	-130.7
Line loss (dB)	1	2	4
Antenna gain (dB)	4	6	10
Antenna F/B ratio (dB)	10	12	14

Table 5.1

Digital Television Standard Video Formats

Vertical line	Pixels	Aspect ratio	Picture rate
1080	1920	16:9	60I, 30P, 24P
720	1280	16:9	60P, 30P, 24P
480	704	16:9 & 4:3	60P, 60I, 30P, 24P
480	640	4:3	60P, 60I, 30P, 24P

Table 5.2

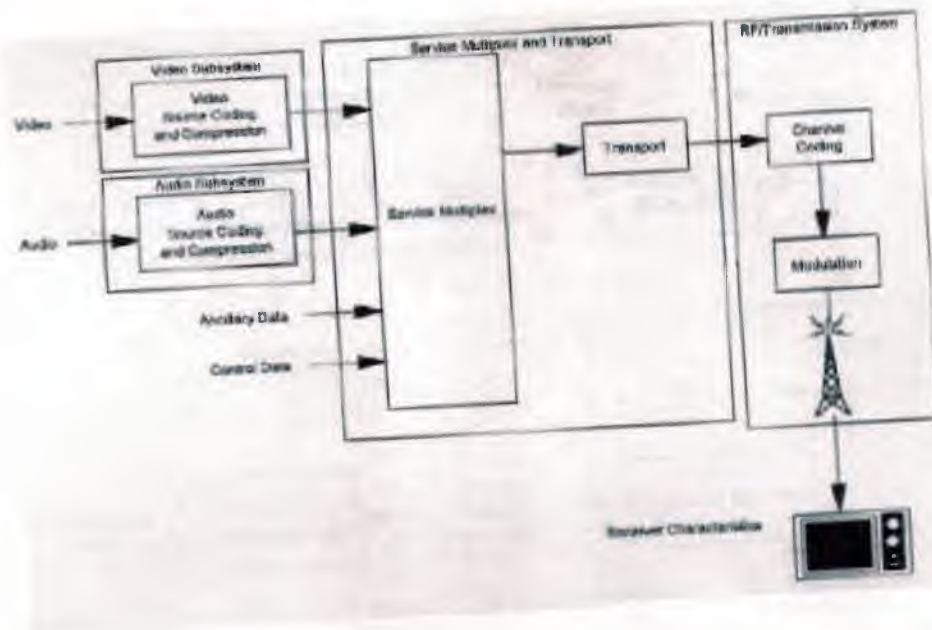


Figure 5.20

ITU-R digital terrestrial television broadcasting model.

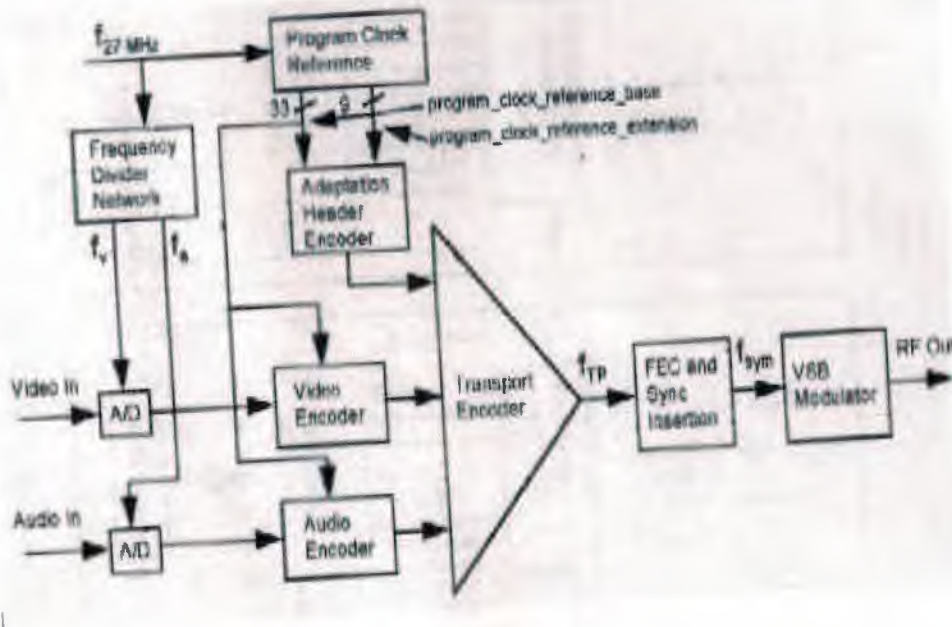
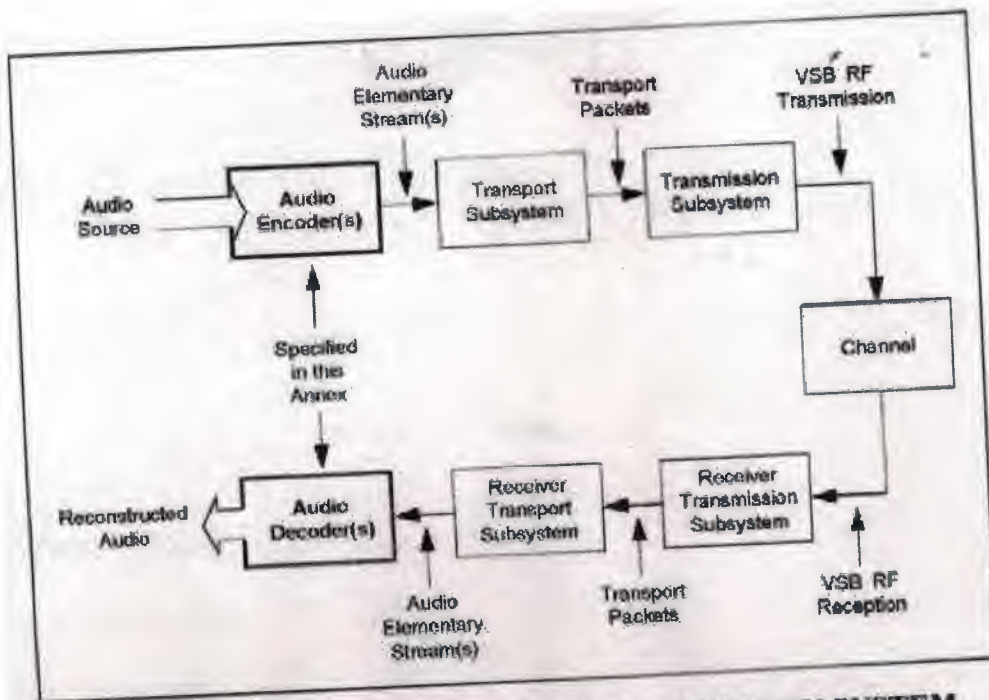


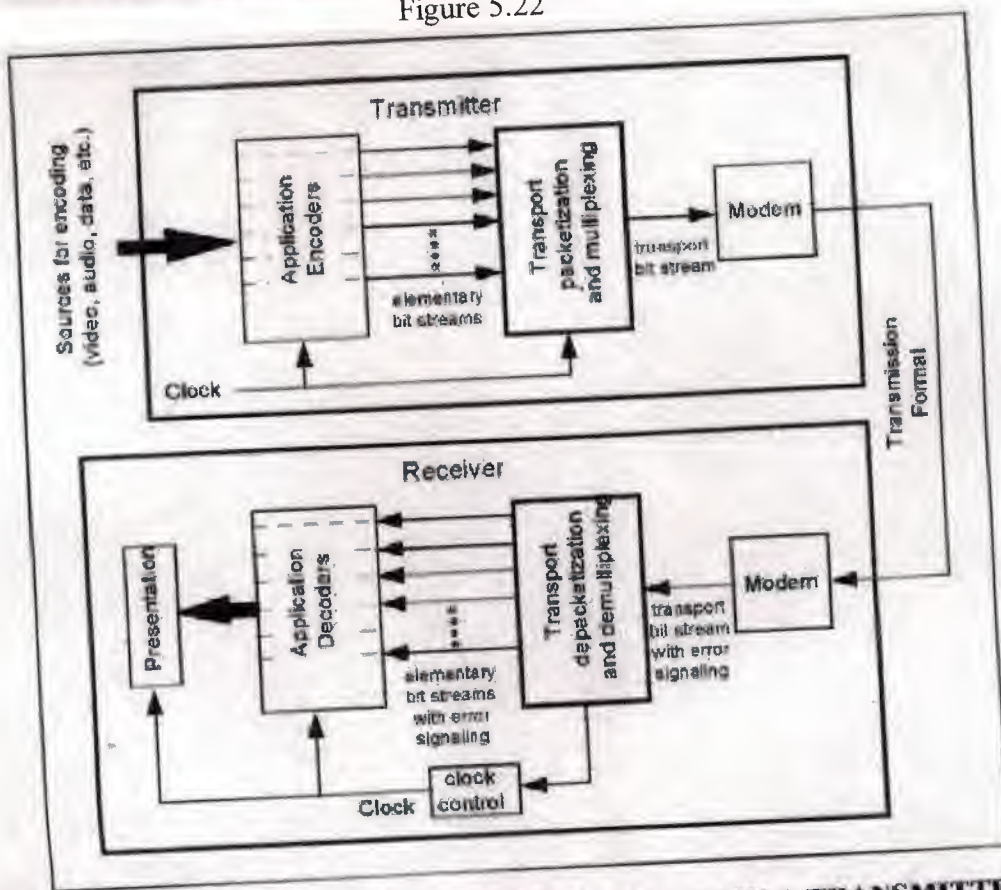
Figure 5.21

High level view of encoding equipment



AUDIO SUBSYSTEM IN THE DIGITAL TELEVISION SYSTEM.

Figure 5.22



SAMPLE ORGANIZATION OF FUNCTIONALITY IN A TRANSMITTER-RECEIVER PAIR FOR A SINGLE PROGRAM.

Figure 5.23

REFERENCES

ELECTRONIC COMMUNICATIONS SYSTEMS
THIRD EDITION..... WAYNE TOMASI

IEEE SPECTRUM.....OCTOBER 97

INTERNET ADDRESSES

[HTTP://WWW.SCARNOFF.COM](http://WWW.SCARNOFF.COM)

[HTTP://WWW.ATSC.COM](http://WWW.ATSC.COM)

[HTTP://WWW.DTV.COM](http://WWW.DTV.COM)

[HTTP://WWW.SONY.COM](http://WWW.SONY.COM)

[HTTP://WWW.IEEE.COM](http://WWW.IEEE.COM)