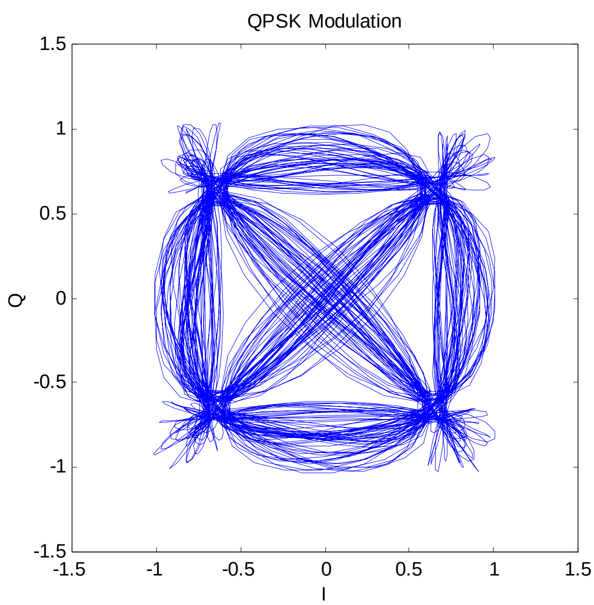
JAMES COOK UNIVERSITY

Electrical
and
Computer
Engineering

Digital Communication Systems and their Modulation Techniques



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ACRONYMS

ADSL	Asymmetrical Digital Subscriber Line
AM	Amplitude Modulation
APCO-25	Association of Police Communication Officers, Specification No 25
ARR	Automatic Repeat Request
ASK	Amplitude Shift Keying
BCH	Bose-Chaudhuri-Hocquenghem (Coding)
BPSK	Binary Phase Shuft Keying
CDMA	Code Division Multiple Access
CELP	Codebook Linear Prediction (Vocoder)
CNR	Carrier to Noise Ratio
COFDM	Coded Orthogonal Frequency Division Multiplexing
CQPSK	Controlled Quadrature Phase Shift Keying
CT2	Cellular Telephone 2
CVSDM	Continuously Variable Slope Delta Modulation
D-AMPS	Digital- Advanced Mobile Phone System
DAB	Digital Audio Broadcasting
DC-CDMA	Direct Sequence Code Division Multiple Access
DCS-1800	Digital Cellular System (1800 MHz) (Same as PCN)
DECT	Digital European Cordless Telephone
DHTV-T	Hierachical Digital Television Transmission
DMCA	Digital Multi- Channel Access
DOQPSK	Differential Offset Quadrature Phase Shift Keying
DPSK	Differential Phase Shift Keying
DQPSK	Differential Quadrature Phase Shift Keying
DSP	Digital Signal Processing
DSRR	Digital Short Range Radio
DVB	Digital Video Broadcasting
ERMES	European Radio Message System
FFT	Fast Fourier Transform
FH-CDMA	Frequency Hopping Code Division Multiple Access
FM	Frequency Modulation
FSK	Frequency Shift Keying
GMSK	Gaussian Minimum Shift Keying
GPS	Global Positioning System
GSM	Group Special Mobile (Global System for Mobile Communications)
HD-Divine	High Definition- Digital Video Narrow band Emission
HDTV	High Definition Television
HIPERLAN	High Performance Local Area Network
I-Q	Inphase- Quadrature

IMBE	Improved Multi-Band Excitation (Vocoder)
ITU	International Telecommunications Union
JCT	Japanese Cordless Telephone
JDC	Japanese Digital Cellular Radio (Same as PDC)
MPEG	Motion Picture Expert Group
MSK	Minimum Shift Keying
NADC	North American Digital Cellular Radio
OFDM	Orthogonal Frequency Division Multiplexing (Same as COFDM)
OQPSK	Offset Quadrature Phase Shift Keying
PCN	Personal Communications Network (Same as DCS-1800)
PDC	Personal Digital Cellular (Same as JDC)
PHP	Personal Handy Phone (Same as JCT)
POCSAG	Post Office Code Standardisation Advisory Group
POTS	Plain Old Telephone Service
PRS	Pseudo Random Sequence
PSK	Phase Shift Keying
QAM	Quadrature Amplitude Modulation
QPSK	Quadrature Phase Shift Keying
RAKE	No acronym, Receiver Block Diagram looks like a garden rake
RPE-LTP	Rectangulat Pulse Excitation-Long Term Prediction (Vocoder)
SDTV	Standard Definition Television
SNR	Signal to Noise Ratio
SPECTRE	Special Purpose Extra Channels for Terrestrial Resolution Enhancements
TETRA	Trans European Trunk Radio
TFTS	Terrestrial Flight Telephone System
VSELP	Vector Sum Excited Linear Prediction (Vocoder)

Introduction

There is a trend in communication system design and implementation to replace analogue techniques with digital techniques. This document shows that for both voice and video, transmission coding procedures exist which result in the digitally transmitted signal occupying the same or less bandwidth than the corresponding analogue signal. For example, a digital cellular radio system can transmit two to three times the number of calls per MHz allocated bandwidth than an analogue system. For Video transmissions, the same frequency allocation can be used throughout the country for the same programme material, so that up to approximately 350 PAL quality programmes can be broadcast on the existing UHF band.

Digital transmission provides the capability for error correction, security of transmission and a tolerance to multipath fading. In addition, digital systems in general are more immune to interference, permitting a better frequency reuse in broadcasting and mobile applications.

This document restricts itself to digital transmission systems. Analogue transmission systems will however continue to exist for many years, but will eventually be replaced by digital transmissions.

There are several recent digital transmission developments, which are having a significant effect on communication technology. These techniques are:

1 Data coding to firstly rely on the limitations of human hearing and vision and secondly to closely relate to the way the signals are produced.

> I Vocoders. Vocoders permit good quality voice transmission at very low data rates due to them transmitting the time varying model of the vocal tract and its excitation, rather than digitising a waveform. Data rates of 4.8 kbps give good quality speech.
>
> II Musicam. Music coding which relies on the fundamental properties of the ear. Signal components which cannot be detected are removed. CD quality music is obtained at 256 kbps compared with 1.4 Mbps for a CD.
>
> III MPEG picture coding. MPEG coding considers: what can be observed by the eye, does part of the image move across the picture, is part of the present image the same as before. The MPEG Video coding technique uses this to minimise the amount of information that needs to be stored. DVD disks can store a whole movie at a much better quality than VHS tape. Typical compression ratios are more than 25:1.

2 Digital TV and Audio Broadcasting.

> Digital TV transmission started in Australia on 1 Jan. 2001 for Capital Cities and before 1 Jan. 2004 for all regional areas. The PAL transmissions are scheduled to be terminated 8 years after the digital TV introduction into an area. Digital TV has been operating in the USA, UK, Sweden, Finland and Spain since 1998. It is currently being introduced in many countries around the world.
>
> Digital Audio Broadcasting has been in operation in the UK since 1995 and trial braodcasts ate taking place in many countries. Since the cost of the receivers is high the use has been very low.

Digital Modulation Techniques

The digital data produced by the mobile radio, audio or television programmes, needs to be modulated onto a carrier for transmission. The simplest modulation technique is Amplitude Shift Keying (ASK), where a carrier is simply turned on and off. ASK is not normally used for Radio Frequency transmissions. The fibre optic cables, which typically operate at bit rates of 500 Mbps, use ASK. The LED generates an infrared or visible carrier when it is on and no carrier when it is off.

In Frequency Shift Keying (FSK), the frequency of the carrier is varied with the data. FSK is thus the same as Frequency Modulation but since the binary data only has 2 levels, only 2 frequencies are used. FSK is used in modems at speeds up to and including 300 bps.

Four level FSK (4FSK) is a technique where the digital data is changed to a 4 level signal allowing 2 bits to be transmitted at a time. This technique provides a reasonable spectral efficiency and is implemented cheaply. 4FSK is thus primarily used in cost conscious applications. The European Radio Message System (ERMES) uses this technique and modulates a 169 MHz carrier with 6.25 kbps data at a ± 1.5625 kHz or with a ± 4.6875 kHz frequency deviation.

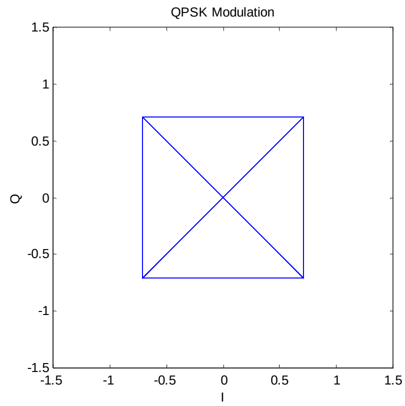
Quadrature Phase Shift Keying

In Phase Shift Keying (PSK), the phase of a carrier is changed with the binary data. In Binary Phase Shift Keying (BPSK), the digital data causes a 0° or 180° phase shift. BPSK is thus exactly the same as Double Sideband Suppressed Carrier Modulation. In Quadrature Phase Shift Keying, 4 possible phase locations are used at a time. Two data bits can thus be transmitted simultaneously. One of the data bits produces the In phase (I) component and the other data bit produces the Quadrature (Q) component. The phase plane diagram of QPSK is shown in Figure 1A. To avoid having to transmit an absolute phase reference to the receiver, most systems use a differential transmission technique where the input data causes a change in phase shift. This is called Differential Quadrature Phase Shift Keying (DQPSK). For 2400 baud modems conforming to the ITU V.26 standard, the Dibit (2 bits) cause a phase shift as indicated in Table 1. Note that a 180° phase shift is possible for QPSK.

In many cases the binary data entering the QPSK modulator is filtered to reduce the bandwidth required for the system. The trajectory from one location on the I-Q plane to the next location, is thus a continuous path rather than a jump. For the 180° phase shift that path can go through the origin of the I-Q plane, resulting in a momentary decrease in signal amplitude. If saturated, Class C, amplifiers are used in the transmitter, nonlinearities can occur as the amplifier is momentarily turned off. Figure 1B shows an I-Q diagram for a QPSK system as is used in the base station of a Qualcom CDMA [1] mobile radio system, which uses a QPSK system. It can be clearly seen that some trajectories pass through the origin of the I-Q plane. A Class A

Dibit	Phase Change QPSK	$^{\pi}/_{4}$ QPSK
00	0°	+45°
01	+90°	+135°
11	+180°	+225°
10	+270°	+315°

Table 1. ITU V.26 Phase Shifts



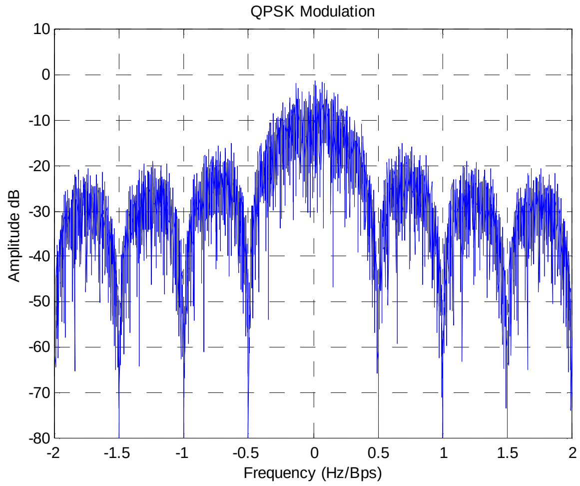
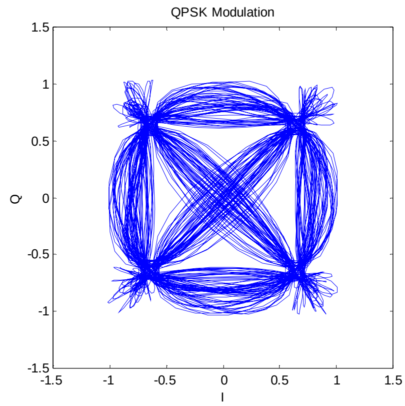


Figure 1A. Unfiltered QPSK I-Q Diagram and Spectrum



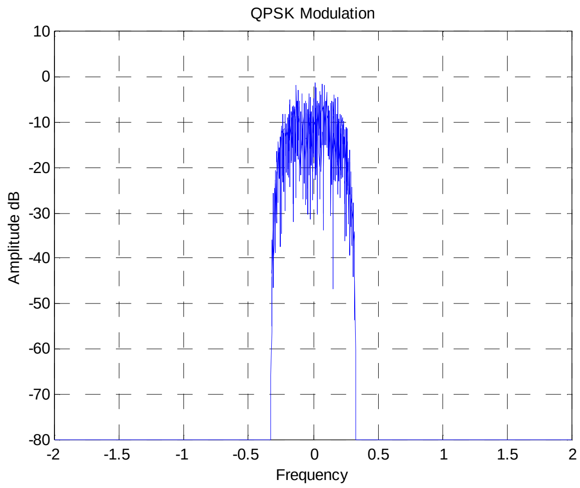
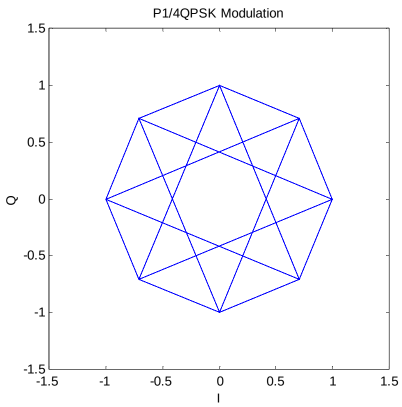


Figure 1B. Filtered QPSK I-Q Diagram and Spectrum, Raised Cosine, r = 0.3.



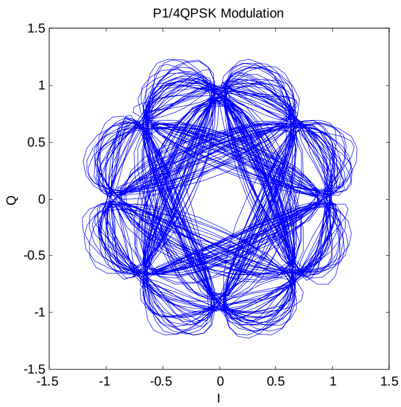
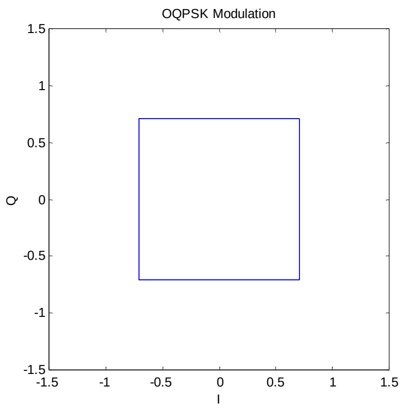


Figure 1.C. $^{\pi}/_{4}$QPSK I-Q Diagram.

Left: Unfiltered, Right: Raised Cosine, r = 0.3.



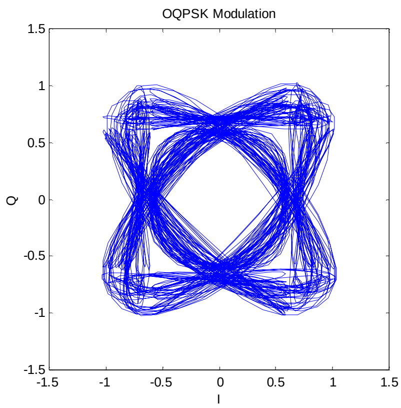


Figure 1.D. OQPSK I-Q Diagram.

Left: Unfiltered, Right: Raised Cosine, r = 0.3.

amplifier stage is thus required for the transmitter. That situation is avoided by using $^{\pi}/4$QPSK modulation. $^{\pi}/4$QPSK and $^{\pi}/4$QDQPSK occupy two sets of 4 locations, like QPSK, with the second set rotated by 45 degrees, each set is used every alternate transmisssion bit. $^{\pi}/4$QPSK occupies 8 locations on the I-Q plane, the positions lying on a circle and having 45° spacing. Differential $^{\pi}/4$QPSK is used in many mobile radio and digital cordless telephone systems. Figure 1.C. shows the unfiltered and filtered I-Q diagrams for $^{\pi}/4$QPSK.

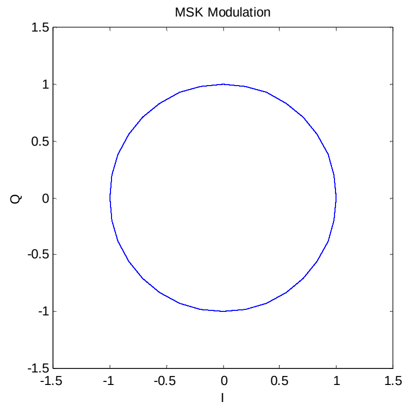
By changing the timing such that one bit of the Dibit changes half a clock pulse before the other one, 180° phase changes can also be avoided. Only one bit changes at a time, but twice as many changes occur. The phase changes are then 0° and 90° for one of the bits and 0° and -90° for the other. This system is known as Offset Differential Quadrature Phase Shift Keying (ODQPSK). If the Dibits provide absolute rather than differential phase information, the system is Offset Quadrature Phase Shift Keying (OQPSK). OQPSK and ODQPSK do not have any 180° phase transitions and the phase trajectories will thus avoid the origin of the I-Q plane. Figure 1D. shows an I-Q diagram OQPSK. The absence of trajectories passing through the origin of the I-Q plane in Figure 1C can clearly be seen. A class C amplifier can thus be used in the mobile transmitter. OQPSK occupies 4 locations on the I-Q plane, the positions lying on a circle and having 90° spacing. QPSK, DQPSK, OQPSK, ODQPSK, $^{\pi}/4$QPSK and $^{\pi}/4$DQPSK all have the same spectrum as shown in Figure 1.

In practical I-Q modulators, the signals are generated using Digital Signal Processing (DSP) Techniques driving the actual I-Q modulator directly. The difference between QPSK, OQPSK, DQPSK and ODQPSK is then simply a difference in the DSP software. The differences between these modulation techniques are thus becoming blurred. Differential QPSK is used in the NADC and JDC mobile radio systems. QPSK and Offset QPSK is used by the Qualcom CDMA mobile radio system.

Quadrature Phase Shift Keying (QPSK) and the many variants theoretically require a bandwidth of 0.5 Hz/bps. In practice the bandwidth is larger. The bandwidth required for transmitting the data can be reduced by filtering the digital data presented to the I-Q modulator. The filtering can be of several forms. Firstly Nyquist filtering [2, 6,7], also known as Raised Cosine filtering can be used. In the NADC, JDC and JCT mobile radio systems the Root Raised Cosine Filters are used. Having one such filter in the Transmitter and one in the receiver results in an overall Raised Cosine Filter. The Raised Cosine Filter is also known as the Nyquist filter and restricts the bandwidth, whilst minimising the intersymbol interference. The second type of filtering is using a conventional low pass filter. Since the transient response is very important, filters with a good time domain response, such as Bessel and Gaussian filters are normally used.

Minimum Shift Keying

It is possible to control the phase changes, so that the rate of change of phase is uniform and that the correct I-Q location is reached just at the end of the data period. That system is known as Minimum Shift Keying (MSK), since the minimum rate of phase shift is used. A constant rate of change of phase



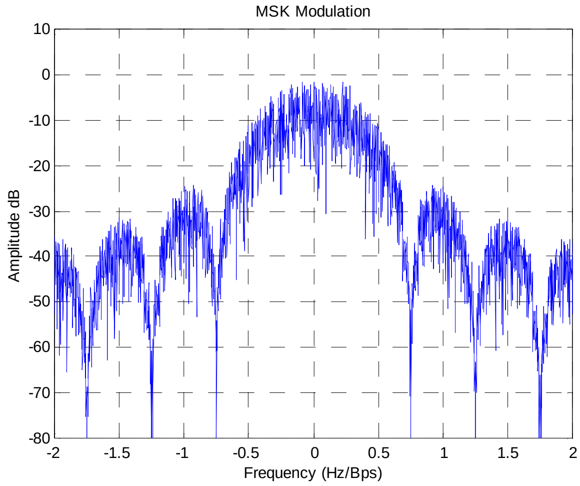


Figure 1E. Unfiltered MSK I-Q Diagram and Spectrum

corresponds to a constant frequency, MSK can thus also be seen as a special case of FSK, where the change of frequency is such that a 90° phase change is obtained after one data period. Minimum Shift Keying (MSK) produces slowly changing phase transitions and has less out of band energy than QPSK systems. MSK has 99% of its energy within 1.2 Hz/bps. Filtering the data going into the modulator will further reduce the out of band components. By filtering the input data to an MSK system with a Gaussian filter, Gaussian Minimum Shift Keying (GMSK) is produced. The GSM mobile radio system uses GMSK. The Gaussian filters have a Bandwidth 81.25 kHz (0.3 BT) [3,4,5]. Since GSM has a data rate of 270.833 Kbps with an allowable 200 kHz bandwidth per GSM channel, this corresponds to a 0.738 Hz/bps normalised bandwidth. GMSK is being used for the GSM and PCN mobile radio systems, the CT2, CT3 and DECT cellular phones and is proposed for the DSRR short range radio system. In many instances GMSK will be produced by using an I-Q modulator and generating the input signals to that modulator with digital signal processing techniques. ROM look-up tables are used to generate the I and Q vector locations in practice.

Some systems specify the modulation as either FSK or QPSK. The APCO-25 trunk mobile radio system being tested in the USA, specifies its modulation either as Controlled QPSK (CQPSK) or Controlled FSK (CFSK). CQPSK consists of a conventional $^{\pi}/4$QPSK modulator with the input data being filtered with a Nyquist Raised Cosine Filter with r = 0.2. The modulation can also be produced by having 4FSK with frequency deviations of ± 600 Hz and ± 1.8 kHz. The input to the FSK modulator is filtered with the same Nyquist filter.

The GMSK specifications for GSM require coherent phase detection and a maximum phase error of ± 20° over one data slot is specified. The CT2 and DECT systems expect the GMSK to be generated using frequency modulation and a tolerance of -10% and +40% on the frequency deviation is specified.

Quadrature Amplitude Modulation

In Quadrature Amplitude Modulation, the data is directly mapped into In-Phase (I) and and Quadrature (Q) signals. Since the I and Q signals are othogonal, the I and Q data can be controlled independently. By having more than two levels on each direction, a large number of data bits can be transmitted at the same time. The QPSK signals of Figure 1A can also be seen as 4 QAM (4 level Quadrature Amplitude Modulation). Having 4 levels in the I direction and 4 in the Q direction results in 16QAM and allows 4 bits to be transmitted at one time. 64QAM permits 6 bits to be transmitted at one time and provides even better channel utilisation at the expense of noise immunity. QAM modulation is used for telephony modems operating at rates of 4800 kbaud and above (ITU V.32.). 16QAM is also used by the Digital Multi-Channel Access (DMCA) system used by trucks in Japan. 64QAM is one of the possible modulation techniques for each carrier in the Digital TV system used in Australia. Figure 2 (a) shows the signal constellation for 64 QAM. Gray code coding is used, such that an error will affect one bit only.

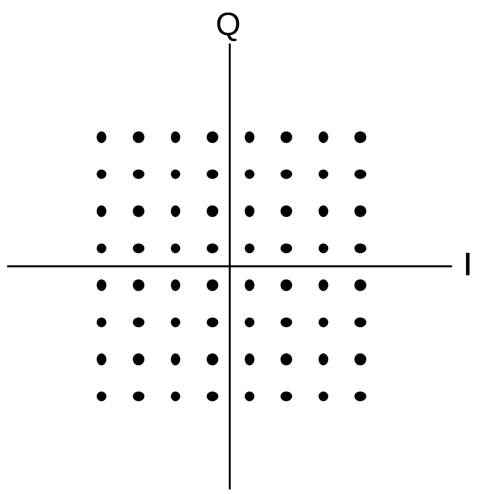


Figure 2 (a). 64 QAM Constellation, with

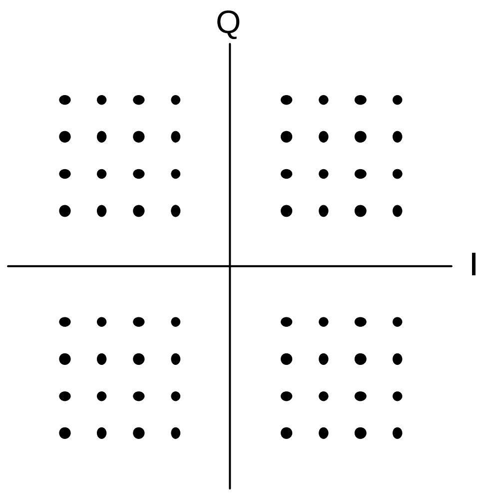


Figure 2 (b). 64 QAM Constellation, with Hierarchical Coding.

Hierarchical Coding

The more levels are used, the more data is transmitted, but the smaller the noise immunity. The number of levels used its thus a compromise. In some systems like TV, it is desirable to have a very good performance if possible, but an acceptable and much lower picture quality should be able to be obtained under most circumstances. This can be achieved by coding the most significant data bits in a robust manner and the lesser important data bits in a less robust manner, making them more susceptible to errors. A 64 QAM coding scheme, with hierarchical coding is shown in Figure 2 (b). Because each of the quadrants have been separated, it is less likely that an error will result in the wrong quadrant being used. The robust data is thus the quadrant (2bits) and the less robust data is the data inside each quadrant (4 bits).

Error Perfomance

Haykin [6] shows that coherent detection produces the same BER for a 3 dB worse SNR than noncoherent detection. The detection technique is thus important to maximise the BER. Figure 3 shows the bit error rate that can be expected for different coding techniques as the received bit energy to noise ratio is varied. It can be seen that for 64QAM without error correction a CNR greater than 20 dB is required for an error rate of less than 1%.

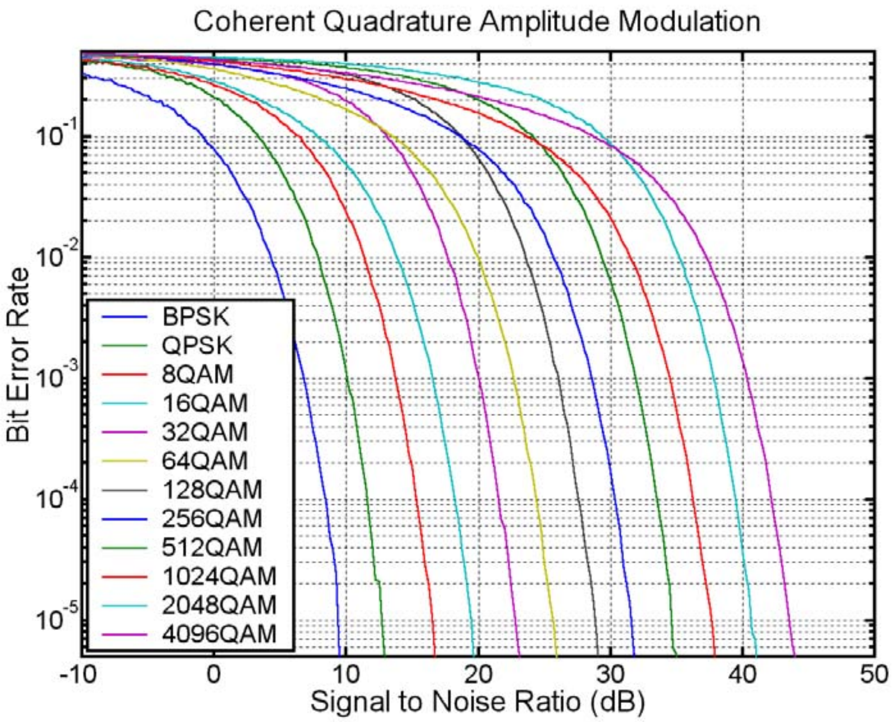


Figure 3. Bit Error Rates Error for different Coding Schemes.
(Author E.Lawrey, with permission)

Error Correcting Coding

Most digital communication techniques rely on error correcting coding to achieve an acceptable performance under poor carrier to noise conditions. In this paper only a brief summary is given, further details can be found in Chapters 5 and 6 of Digital Communications by B. Sklar [7] or other Communications texts. Error correcting coding techniques can be split into three basic techniques: Automatic Repeat Request, Forward Error Correction and Trellis Coding.

Automatic Repeat Request

With Automatic Repeat Request (ARR) the receiver checks for transmission errors and if detected requests the message to be re-transmitted. Many modem protocols such as Kermit, Xmodem, Ymodem and Zmodem use this technique. ARR is used for data communications using mobile radio, but it cannot be used for voice applications since the delay involved in re-transmission is to long. The technique can also not be used for Audio or Video Broadcasting since not every receiver will receive the same errors.

Forward Error Correction

With Forward Error Correction extra data bits are added to the message during transmission, allowing up to a certain number of errors to be corrected. This technique is used by all the systems discussed in this paper. The forward error correcting techniques can be split up into two groups:

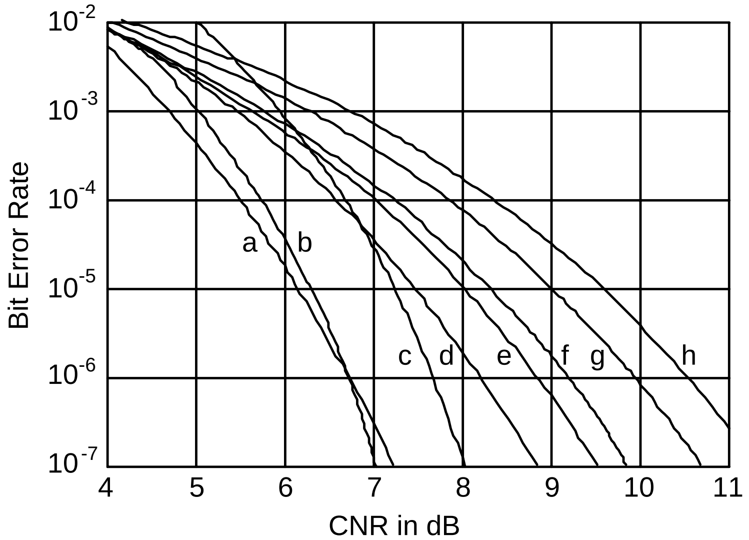


Figure 4. Error Correcting Coding and Channel Capacity [Ref 7, Fig 5.23]
a Vitterby half rate, K=7. b BCH (127,64), e = 10. c BCH (127,36), e = 15.
d Extended Golay (24,12), f Hamming (15,11), e = 1. h Uncoded.

Block Codes

With Block Codes a block of data has error detecting and correcting bits added to it. One of the simplest error correcting block code is the Hamming Code, where parity bits are added to the data. Three parity bits added to four data bits allows for one error to be corrected in the total of 7 bits. This is called the Hamming (7,4) Code. The coding efficiency increases as the blocks become larger and 6 parity bits with 57 data bits, allow one error to be corrected in resulting 63 bits. This is the Hamming (63,57) Code. It is very difficult to produce coding schemes using Hamming Codes for correcting multiple errors and other codes are normally used. The Golay (23,12) Code allows up to 3 errors to be corrected in a total of 23 bits, corresponding to 12 data bits and 11 check bits. Bose- Chaudhuri- Hocquenghem (BCH) codes allow a large number of errors to be corrected without too heavy a computational load. BCH codes are used in high speed telephony modems. Reed Solomon Codes are a special group of BCH codes. They are used in Compact Disk players to correct for the bursts of errors caused by scratches on the CD.

By adding the error correcting bits to data, transmission errors can be corrected. However since more data has to be squeezed into the same channel bandwidth the more errors will occur. As a consequence there is an optimum number of errors that should be corrected. Figure 4 shows the final bit error probability for BPSK modulation in a representative communication channel of fixed bandwidth. It can be seen that the BCH (127,36) code which can correct 15 errors has a worse performance than the BCH (127,64) code which can correct 10 errors.

Convolutional Coding

In Convolutional coding the input bits are passed through a shift register of length K. N output bits are generated by modulo 2 adding selected bits held in different stages of the shift register. For each new data bit N output bits are produced. This coder is called a $^1/_N$ rate coder since N output bits are produced for every one input bit. The output bits are influenced by K data bits, so that the information is spread in time.

As an example, the GSM mobile radio system uses 3 separate coding schemes depending on the importance of the data. The most important data words from the RPE-LTP coder (Class 1A) have parity bits added, they are then combined with important data words (Class 1B) and convolutional coding with a half rate coding is applied. The resulting data is then combined with the least important data, which has no error correction applied to it. At the receiver, the inverse process is performed. The Viterbi algorithm allows the data to be recovered in a maximum likelihood manner with an acceptable computational load.

Trellis Coding

With Trellis Coding the modulation and the error correcting coding is done in one operation. QAM is used for the modulation and additional space is allowed. For example in the CCITT V32 modem standard, trellis coding is used at the 9600 baud rate. The I-Q plane is shown in Figure 5. The 32 locations allow 5 data bits to be transmitted at one time. However only 4 data bits are used. Two of the data bits are applied to a convolutional coder, causing one additional bit to be added. The other 2 bits are coded on the I-Q plane such that a maximum possible distance is obtained. The locations marked with a * denote the vectors corresponding to those bits. It can be seen that the distance is maximised. There are 8 such groups of widely spaced signal sets. Trellis coding is very efficient in maximising the information capacity of a channel.

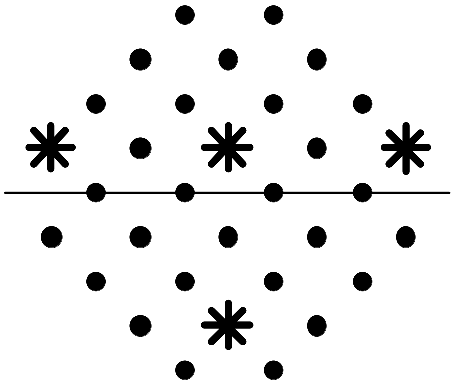


Figure 5. ITU V.32 Trellis Diagram.

It is possible to use one or more of these error correcting schemes. Modems at a 14400 baud rate use BCH coding and Trellis coding, while the file may be transmitted using the Ymodem protocol.

Digital Television Receivers using COFDM, use Block Coding and interleaving is used to correct for multipath fading, with a Rate Compatible Punctured Convolutional (RCPC) code, which has been defined as in the DVB Satellite standard, being applied to the resulting data to correct for Rayleigh fading.

To adapt the error protection to the actual transmitting conditions, several code rates can be chosen. The following coderates are specified in the DVB-T system [9]: 1/2, 2/3, 3/4, 5/6, 7/8

The code rate 1/2 has the highest redundancy, but the highest transmission safety. This mode should be applied to strongly disturbed channels. On the other hand a code rate of 7/8 has a low redundancy but a very weak error protection. Therefore, it should be used for channels with only low interference.

The hierarchical coding used in DVB as shown in Figure 2, is closely related to trellis coding in that I-Q vector space techniques are used to improve the robustness of the system.

Spread Spectrum Techniques

Code Division Multiple Access (CDMA)

CDMA modulates the data with a set of orthogonal codes such that the resulting spectrum is spread over a wide range of frequencies. This spreading can be achieved using one of two ways. Firstly, a pseudo random sequence or a Walsh function can be used to determine the selection of a carrier frequency. That modulation is then Frequency Hopping CDMA (FH-CDMA) which in this report is simply called Frequency Hopping and is treated separately. Secondly, a pseudo random sequence or a Walsh function can be used to directly modulate digital or analogue data. That technique is called Direct Sequence CDMA (DS-CDMA), is normally referred to simply as CDMA and is treated here. Finally Qualcom in the USA has developed a cellular mobile radio system that they call CDMA as it uses DS-CDMA techniques for modulating the data. In this report the pseudo random sequence or a Walsh function, which is used to modulate the data is called the Coding Sequence. The block diagram of a DS-CDMA system is shown in Figure 6.

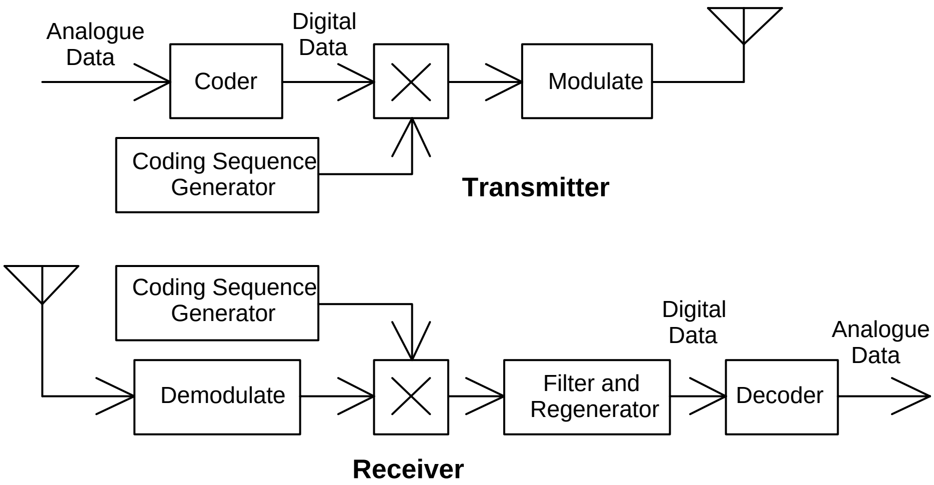


Figure 6. Block Diagram of a DS-CDMA system

At the receiver the signal is demodulated by multiplying it with the same Coding Sequence. If the received and transmitted Coding Sequences are in time synchronism, they will correlate and the demodulated data is the same as the input data. The “chip rate” is the clock rate of the Coding Sequence and that is normally many times that of the digital data. The chip rate determines the spectral bandwidth, which is thus many times that of the data alone as is shown in Figure 7. Code Division Multiplexing also known as Code Division Multiple Access, together with Frequency Division Multiplexing and Time Division Multiplexing form the 3 basic means whereby multiple users can be combined onto the same communication channel. Some communication systems like the GSM mobile radio system use a combination of two or more of these techniques.

Since the modulation process and the demodulation process in a CDMA system are the same, the signal spectrum from other users and interferers are spread

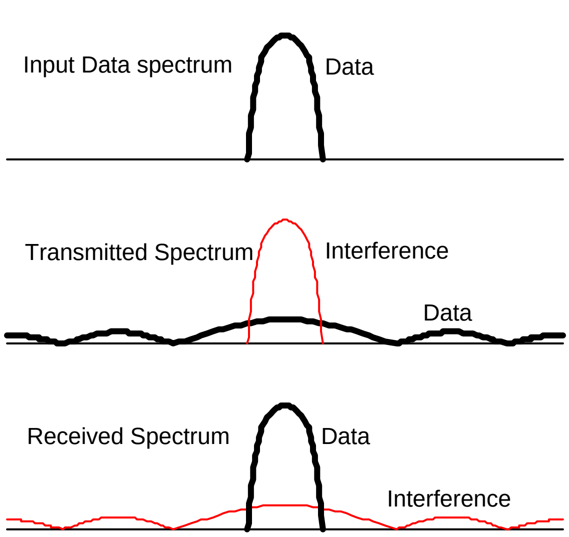


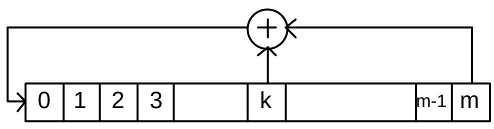
Figure 7. Spectrum of DC-CDMA system.

during demodulation. The wanted signal however correlates and its bandwidth is reduced. If there are N Coding Sequence bits per data bit, over one data bit period, the wanted signal produces an output of N units while an uncorrelated signal produces an output of one unit. During the demodulation process the spectrum occupied by the signal is reduced by a factor N and since the same signal power now occupies a reduced bandwidth, the Signal to Noise Ratio is increased by the same factor N. This is called the Process Gain and is illustrated in Figure 7.

The coding sequences, used for the different users, should be orthogonal. That is when one coding sequence is multiplied by another coding sequence the result should be zero, or as close to it as possible. During the demodulation process the wanted signals correlate, while the unwanted signals do not. Several possible codes can be used. The most common ones are briefly described below:

Pseudo Random Sequences (PRS)

By taking feedback from a number of flipflops in a shift register, a pseudo random code is produced, as shown in Figure 8. The code is random in that the probability of a logic one or logic zero occurring is close to 0.5 and is independent of the previous data bit. The code is known since one obtains the same output code by loading the shift register with the same data. For an M bit register the maximum length sequence repeats after 2^M-1 clock pulses. In addition there exist many different maximum length sequences, so that it is unlikely that another user would use the same sequence. For example for a 15 bit register the maximum length sequence is 32767 bits long and there are 1800 different feedback connections all of which will cause different maximum length sequences [8], one is included in figure 9.



M	K	Tap Positions	M	K	Tap Positions
2	3	2,1	10	1023	10,3
3	7	3,1	11	2047	11,1
4	15	4,1	15	32767	15,1
5	31	5,2	18	262 143	18,7
6	63	6,1	20	1 048 575	20,3
7	127	7,1	21	2 097 151	21,2
8	255	8,4,3,2	22	4 191 303	22,1
9	511	9,4	23	8 388 607	23,5

Figure 8. Pseudo Random Sequence Tapping Points.

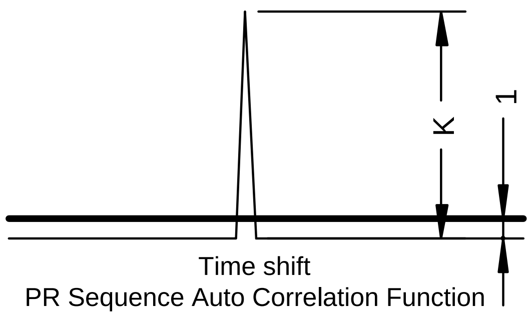


Figure 9. Auto-correlation Function.

Figure 9 shows the autocorrelation function of an M register maximum length PRS. It can be seen that if the sequence is in phase, an output of 1 is obtained, so that the transmitted data stream is recovered without loss. Other sequences or other data produces an output of 1/K, since the PRS is an odd number of bits long and thus has one more 1 than 0 or vice versa. The longer the sequence, the more the unwanted signals are suppressed. If the clock rate of the receiver Coding Sequence is slightly different from that of the transmitter, a demodulated output waveform like the one shown in Figure 9 results. This can then be used to determine the correct synchronisation of the Coding Sequences. It can be seen that if a multipath signal arrives one PRS bit later than the direct signal and the receiver is locked onto the direct signal, the multipath signal will produce very little output. CDMA receivers are thus very tolerant to multipath propagation. If the multipath signal has a delay of less than one PRS bit, then there will be some correlation between the multipath signal and the receiver's coding sequence. Depending on the phase shifts produced by the reflections making up the multipath signal, the multipath signal may reinforce or cancel the direct signal. For the Qualcom CDMA receiver, the chip rate is 1.2288 Mbps, so that a multipath signal that has travelled 244 m more, corresponding to a 1 chip delay will be ignored. It can be seen that high chip rates are preferred to minimise the effect of multipath propagation. A higher chip rate causes a wider transmitted spectrum. In many systems, the permitted bandwidth is limited, thereby setting a limit on the multipath immunity.

The Qualcom IS95 CDMA cellular radios use a 42 bit register code producing a "Long Code" which is used as the master clock to synchronise all the CDMA radios. This long code is ANDed with a user assigned long code mask, which allows each user to be uniquely identified. The IS95 CDMA cellular radio also uses other important Digital Communication Techniques. The block diagram of its forward link, from the base station to the mobile, is shown in Figure 10 [Ref 1, Fig 8]. The reverse link uses a different configuration.

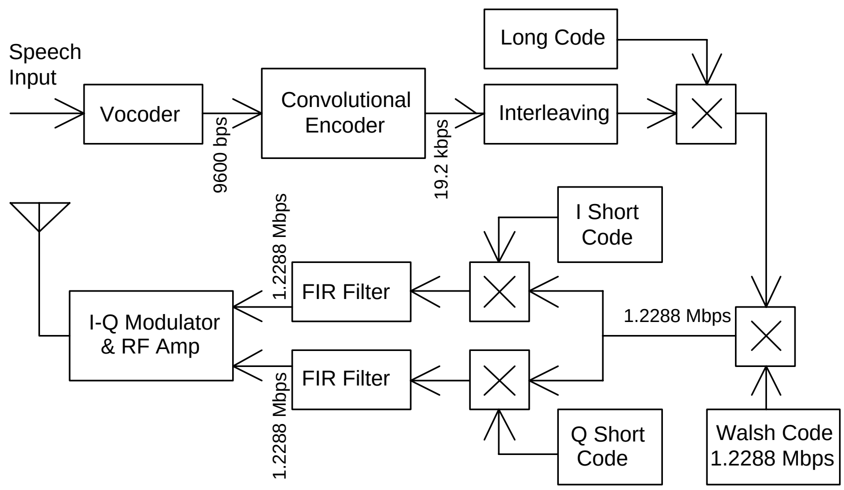


Figure 10. Qualcom CDMA phone block diagram.

Gold Codes

Gold Codes are produced by modulo-2 adding two different maximal length pseudo random sequences of the same length. Shifting the phase, that is the starting position, of one of the sequences produces a different pseudo random sequence. Two 5 bit register can produce a maximum length sequence of 31 bits each and can produce 31 different 31 bit random sequences by simply shifting the phase of one of the sequences. Gold codes can be chosen such that the cross correlation between the codes is uniform and bounded. Gold codes are thus very useful in CDMA system design. Gold codes are used in the GPS global navigation system, where DS-CDMA techniques are used to provide immunity from interferers.

Walsh Functions

Walsh Functions or Walsh Codes have the property that they are orthogonal to each other and orthogonal to the logical NOT of each other. Walsh codes are available with lengths of integer powers of 2, i.e. 1, 2, 4 8, etc. Walsh codes are generated by the recursive expansion:

$$W_{2n} = \begin{bmatrix} W_n & W_n \\ W_n & \overline{W_n} \end{bmatrix} \qquad \text{where} \qquad W_1 = 1$$

The set of Walsh codes to be used would normally be evaluated and stored in a ROM for subsequent use. The Qualcom CDMA cellular radio system uses 64 bit Walsh Codes to provide for 59 user channels. In addition there are the following 3 Channels types:

1) A Pilot Channel, containing the final I and Q spreading sequences thereby enabling all radios to obtain a coherent phase reference.
2) A Synchronisation Channel, which transmits the time of day information, which is required for the mobile and the base station to align their clocks, so that the coding sequences are in phase.
3) One or more Paging channels.

Frequency Hopping

In Frequency Hopping, the data is modulated onto many carriers, covering a wide frequency spectrum. There are two basic types of frequency hopping:

Slow hopping

Slow hopping where the carrier frequency is changed less than or comparable to the data rate. The bandwidth around each carrier is thus the same as the bandwidth of the data. A typical example of this use is in a secure analogue or digital military communications system using shortwave or UHF radio receivers. SCIMITAR, SINCGARS and JAGUAR are such commercial products. The GSM mobile radio system standard allows for slow frequency hopping to be used in conjunction with Time Division Multiplexing. The mobile radio needs to respond to 224 hops per second, with a required settling time of less than 1 millisecond. Slow frequency hopping systems have basically the same noise and interference immunity properties as single frequency systems except that an interferer on one of the carriers will only affect the system when that carrier is used.

Fast hopping

With fast hopping the carrier frequency is changed at the same or faster rate than the data rate. In practice a small odd number typically 3 or 5 frequency hops are used per data bit. The data is demodulated for each chip interval and a majority decision is made to minimise the effect of interference on one of the frequencies used. The faster the hopping rate, the better the noise immunity and the better the tolerance to multipath fading. In many fast hopping systems, the frequencies and the data rate are selected such that the centre of the channel occurs at the first null of the data modulated on the adjacent channel.

OFDM and COFDM

Orthogonal Frequency Division Multiplexing (OFDM) or Coded Orthogonal Frequency Division Multiplexing (COFDM), is a system where the individual data bits of a word are coded onto individual carriers. The carrier frequencies are chosen to be mutually orthogonal over one symbol period. For two frequencies to be orthogonal over one symbol period, their product must be zero when integrated over one symbol period. This is satisfied when an integral number of carrier cycles to fit into the symbol period. Those carriers are then modulated using standard digital communication techniques such as FSK, PSK or QAM. Multipath fading will tend to cause a notch in the received spectrum at one of the carrier frequencies, causing a consistent error in one of the data bits. To make the system tolerant to these errors, error correcting codes are normally included as part of the transmission scheme. All practical systems thus use Coded OFDM (COFDM) rather than the uncoded OFDM. Some literature uses the term OFDM when discussing COFDM systems and the terms OFDM and COFDM can be used interchangeably, with COFDM being the more correct term.

COFDM is used for the Digital Video Broadcasting (DVB) or Digital Television in Europe and Australia, where it allows full utilisation of the available spectrum

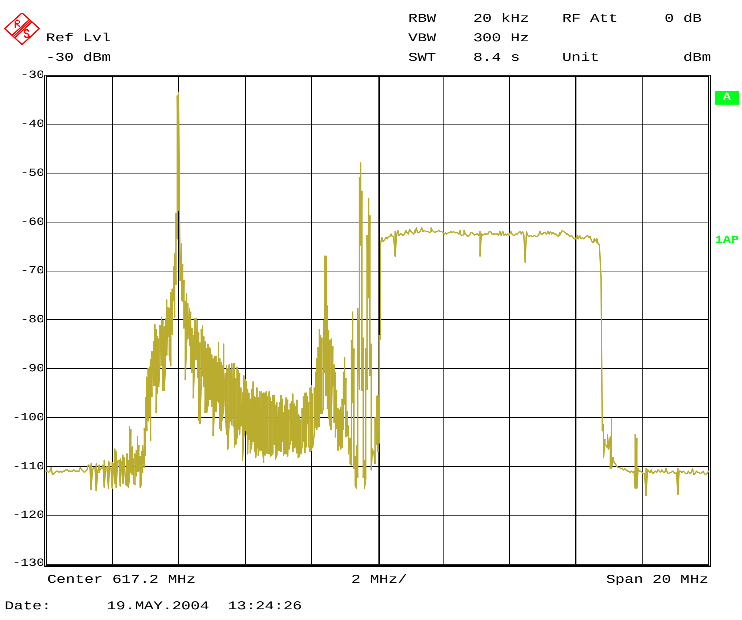


Figure 11. Spectrum of Adjacent TV channels with PAL (Left) and DVB (Right).

for each channel and the avoidance of frequency carriers that would cause interference to or from other services. The major power in a PAL broadcast is concentrated near the Vision, Colour and Sound subcarrier frequencies. These frequencies would not normally be used for the COFDM carriersduring the initial transmission phase when both the PAL and the DVB are used and some DVB channels will have distant PAL transmiters, which can cause interference to the DVB signal. A typical spectrum of a COFDM - DVB system is shown in the right channel of Figure 11, where 1705 carriers are used to fill this spectrum. Notice the better spectrum utilisation of DVB compared to the adjacent PAL transmission.

For single frequency transmission, the transmitters at different locations must operate at a frequency difference less than 0.1% of the inter carrier spacing, ie for an 8 k mode at a frequency error of less than 1Hz for a transmitter frequency of up to 800 MHz. This is a significant limitation and requires external synchronisation control.

COFDM is used for the Eureka 147 Digital Audio Broadcasting (DAB) system which was introduced in the UK during 1995. In Australia test transmissions are taking place, using frequencies at 1.5 GHz. However with the added cost of the receiver, it remains to be seen if the service becomes very popular.

Multipath Propagation

Introduction

Multipath Propagation occurs when signals arrive at the receiver both direct from the transmitter and via a reflection from or transmission through an obstacle. Depending on the dielectric constant and conductivity of the reflecting surface or transmitting medium, the reflecting signal will undergo a phase change and a polarisation change. Depending on the amplitude and phase of the reflected signal, these reflected signals can cancel the direct signal, causing no signal to be received. For a fixed receiver, these reflections cause nulls in the received spectrum to occur at frequencies where the difference in the path length is $^{n\lambda}/_{2}$. Figure 13 shows such a spectrum. In the time domain the multipath signal can be represented by the reception of one direct signal together with one or more slightly delayed signals being received from the reflecting objects. Depending on the type of modulation used, either the frequency domain or the time domain representation will most clearly illustrate

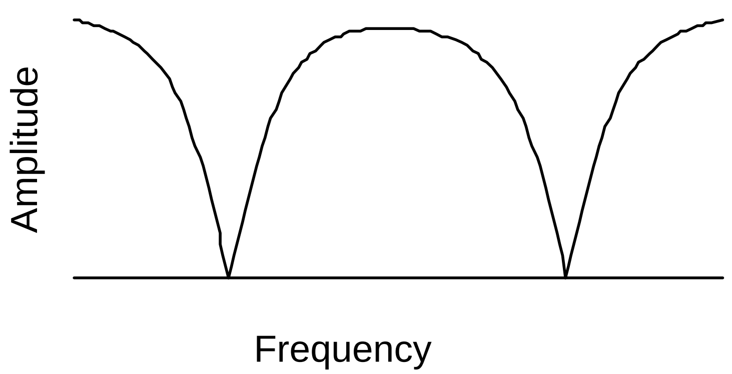


Figure 12. Spectrum caused by Multipath Fading.

the effects of the multipath signals and the susceptibility or immunity of the modulation system to multipath fading.

For a moving receiver and a fixed carrier frequency, the multipath signals cause an interference pattern with a spacing larger than the wavelength of the carrier. At 1 GHz the nulls of interference will thus be 300 mm apart in worst case. The distance can be larger depending on the geometry. The area where the attenuation due to multipath is more than 10 dB is 1%. As the receiver travels, the received amplitude is thus Amplitude Modulated with the multipath signal. The frequency of that modulation depends on the change of path lengths of the direct and the reflected signals and it thus depends on the geometry. The maximum frequency is:

$f_{carrier}$ *speed*/c where c is the speed of light.

As shown below, spread spectrum techniques offer some immunity from multipath propagation.

Multipath with Frequency Hopping

In slow frequency hopping systems, the interference pattern for each of the carrier frequencies is different. There is thus 0.01% chance that two frequencies being used both have an attenuation in excess of 10 dB. GSM cellular mobile radio systems can use frequency hopping with up to 26 frequencies. As a consequence the probability of enough data to be corrupted to make the system inoperable due to multipath fading is negligible when frequency hopping is used. The number of carrier frequencies that can be used for frequency hopping in each mobile radio cell is governed by the available frequency band, the number of adjacent cells and their allocated frequencies. The number of frequencies available for the GSM system is at present limited and no frequency hopping is used by the GSM operators. As a consequence the GSM system has no immunity from multipath fading.

In fast frequency hopping systems the immunity to multipath propagation is even better. Firstly, for a 3 chip per data bit system, the probability of 2 of the 3 chip bits being affected, causing the majority decision to give the wrong result is very small. Secondly, if the reflected signal arrives after the receiver has switched to the next carrier frequency, the reflected signal will be rejected by the IF chain of the receiver. It can be seen that the faster the hop rate, the more chips per data bit and the shorter time the transmitter is tuned to each carrier frequency, the better the immunity from multipath fading. Unfortunately increasing the chip rate also increases the bandwidth, so that in practice a compromise is required.

It should be noted that the nearly complete removal of certain frequencies is not the only effect of multipath fading on frequency hopping systems. The additional reflected signals cause some phase shifts to all the other carriers. Depending on the modulation techniques, those phase shifts can cause errors. For analogue input signals such as audio and amplitude or frequency modulation, these phase shifts do not cause any problems. For digital transmissions and phase modulation, the resulting constant phase shift must be allowed for. Differential coding techniques, such as DQPSK achieve this.

Multipath with CDMA

For CDMA systems the effects of multipath are best described in the time domain. The reflected signal will arrive at the receiver with a slight time delay compared with the direct signal. In the receiver the signal is correlated with the local Coding Sequence. If the transmitted and received coding sequences are exactly in phase they correlate to give the correct data output. If the reflected signal arrives more than one chip time after the direct signal, the reflected signal will not produce a correlated output and is ignored by the receiver. For a chip rate of 10 MCps (MChips/sec), a transmission bandwidth of 20 MHz is required. Any multipath signal that has travelled 30m farther than the direct signal will be one chip bit delayed and will thus be ignored. Multipath signals with shorter delay times will produce an interference with the direct signal. The higher the chip rate the shorter the path difference required to produce interference and the better the immunity from multipath fading. Increasing the chip rate also increases the bandwidth required to transmit the signals, so that a compromise is required.

It is possible to demodulate each of the multipath signals individually, using separate correlators for each multipath signal. In this case several different demodulated outputs are obtained. Provided the differences in path lengths in the multipath signals are less than a quarter of the data bit rate, these signals can then simply be added to improve the signal to noise ratio. The compromise is that the SNR can be increased at the expense of hardware complexity. For 2 marginal signals, the demodulated signals when added result in a 3 dB increase in SNR. As shown in Figures 3 and 4, for a typical digital communication system without error correction, that can result in a change in error rate from one error in 800 bits to one error in 100 000 bits. For a system with error correction the improvement is even more dramatic. Using the multipath signals to obtain useful data is very worthwhile.

The RAKE receiver passes the received signal through a multi-tapped delay line. The signals from these taps are weighted and combined. The process is most easily illustrated by considering a direct signal and one reflected signal. The direct signal will pass through the whole delay line and is normally added directly to the output. Reflected signals will have a delay with respect of the direct signal and the time signal from the direct path that is at the end of the delay line occur at one of the taps in the delay line, as shown in Figure 13. By then setting that weight to ±1 according to the phase of the reflected signal, that reflected signal will then add in phase with the direct signal. The direct signal component at the tapped output will not correlate and appear as noise.

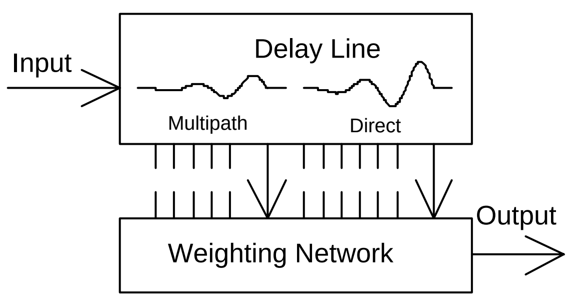


Figure 13. Block Diagram of a Rake Receiver.

By making the weights other values than just ±1, the SNR can be maximised, rather than simply maximising the signal. All other weights are zero, so that no additional noise is included. Multiple reflections can be accommodated by setting the appropriate weights. The receiver has to adjust the weights at each of the taps, in order to maximise the SNR. A simple optimisation process can achieve this.The Rake receiver is primarily used to minimise the multipath effects that are more than one chip clock delay from the direct path.

Multipath with COFDM

In COFDM systems the data is coded onto many carriers and each of those carriers is modulated using PSK or QAM at a low data rate. The more carriers, the lower the modulation rate and the narrower the bandwidth of each of these carriers. The carriers become thus virtually like pure sine wave sources. During one of the bit periods the received signal will contain 3 separate intervals as shown in Figure 14. The First time interval, occurs during the latter part of the Guard Interval. The direct signal is present but the multipath signals have not yet arrived. The received signal is thus exactly as expected. This time interval is normally very short, too short to accurately demodulate the signal. The second time interval occurs during the Measure Interval, both the direct and multipath signals are received in a steady state manner. The received I-Q vector has been shifted a fixed amount from the ideal locations due to the multipath signal. The multipath signals will change the phase and amplitude of the individual carrier sine wave. Since the carriers are nearly constant sine waves, the data modulation does not change the phase shift between the direct and the multipath signals. The multipath signals will thus cause a fixed change in the received phasor, with the whole I-Q pattern being rotated and changed in size by a fixed amount. The standard DQPSK demodulator can thus be used to demodulate the data without being affected by the multipath signal. This time interval is the largest and the signal detection takes place during this time. The receiver simply allows for this phase shift and if differential transmission and detection techniques are used this phase shift simply cancels. The third interval occurs during the first part of the Guard Interval. The direct signal is not transmitted but the multipath signals still exist. At the end of this period the multipath signals will have disappeared as

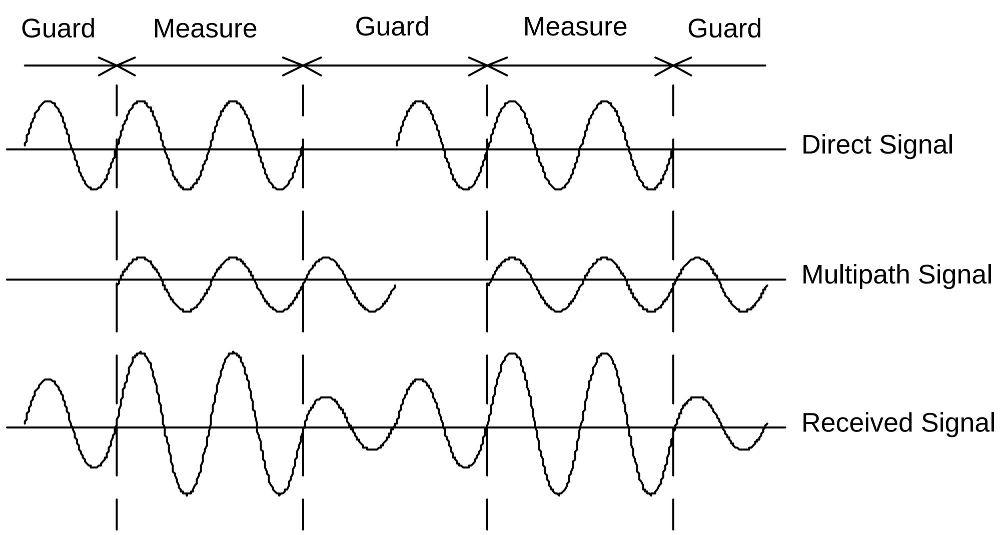


Figure 14. COFDM Received Waveforms.

well. This time interval is again too small to make any useful measurements. The multipath signal can be compensated for during the measurement period and will thus not cause any serious errors to the COFDM system. Since the first and the third periods are transient in nature and no measurements are made during this time, they are called a guard period. The guard period should be long enough to accommodate the delay from all the multipath signals, but short enough not to reduce the data rate too much. The two parts of the guard period are normally merged, so that the new carrier data is already transmitted while the multipath signals are still decaying. The higher the number of carriers, the bigger the guard period can be without affecting the data rate and the more tolerant the system is to multipath signals. The Guard period is normally 5% to 20% of the total symbol period. The length of the Guard pariod is exaggerated in Figure 14.

Some of the carriers will be affected more by multipath propagation. If the reflected signals cancel the direct signal exactly, corresponding to the nulls in the spectrum of Figure 12, then no signal is available to measure phase shifts on. Only 1% of all the carriers suffer more than 10 dB of attenuation due to multipath. For 99% of all the carriers the phase can thus be determined accurately and the correct data determined. Error correcting techniques are used to compensate for the attenuated carriers. COFDM receivers can through the digital signal processing carried out in the receiver determine which carriers are being reduced in amplitude, making their phase measurements more prone to errors. The receiver can then reduce the weighting of that particular subcarrier and hence reduce the resulting error rate.
With the error correcting techniques, the COFDM systems can tolerate multipath signals that are of the same amplitude as the direct signal without degrading the data. The multipath signals can in fact be generated by a transmitter in an adjacent cell, provided the same programme is transmitted and the carriers are all synchronised. It is thus theoretically possible to use the same allocated frequency for all the transmitters in a broadcasting network. The tolerance does not extend to other carriers which are at the same frequency but are not in synchronism with the desired signal. If the unwanted signals are 20 dB below the required signal, they will cause less than 6° phase shift to the required carrier and the data will still be able to be demodulated without error. An outdoor antenna will achieve a selectivity greater than this, so that the interference affects of adjacent transmitters are virtually eliminated. Even with different programmes being transmitted the frequencies should be able to be repeated on a chessboard type pattern, i.e. a 2 cell structure, rather than the 7 cell structure commonly used in cellular radio.

Communication Systems

Digital Video Broadcasting DVB

COFDM is used for the Digital Video Broadcasting (DVB) or Digital Television in Europe and Australia, where it allows full, constant power, utilisation of the available spectrum for each channel. Theoretically this will give the best performance possible.

In Australia Digital Television was introduced in the capital cities on 1 January 2001, with the progressive introduction into other arteas during 2001 and 2002. Simultaneous PAL and DVB transmissions will occur for an 8 year period. Political pressure and the market penetration of DVB, which will depend on the receiver cost, will determine if this period will be extended. A summary OFDM modulation for the 8k and 2 k modes of DVB is shown in Table 2 [10].

The 8 k mode will have better multipath immunity, because of the lower data rate. Because COFDM can operate at high interference levels it is theoretically possible to allocate the same frequency to one TV channel throughout the country. In Australia, for Digital TV, Channels 0 to 5A will not be used, because of electrical noise, and other allocation requirements like FM. There are thus 8 channels available in the VHF band (Band III) and 40 channels available in the UHF band, (Bands IV and V), excluding channel 68 and 69. It will thus be possible to have 48 different HDTV programmes or a minimum of 192 standard definition TV (SDTV), ie PAL quality, programmes to be broadcast in the existing VHF band III and UHF bands IV and V. These figures show that there is a great incentive to convert the existing TV transmissions to a digital format as the spare spectrum can be auctioned off for Video on Demand etc.

Parameter	8 k Mode	2 k Mode
Number of Carriers	6 817	1 705
Data symbol	1024 µs	256 µs
Carrier Spacing	976.6 Hz	3 906 Hz
Total Carrier Spacing	6.65625 MHz	6.65625 MHz
Multipath Immunity 1/8 Guard	38.4 km, 128 µs	9.6 km, 32 µs
Multipath Immunity 1/16 Guard	19.2 km, 64 µs	4.8 km, 16 µs

Table 2. COFDM DVB Modulation Specs.

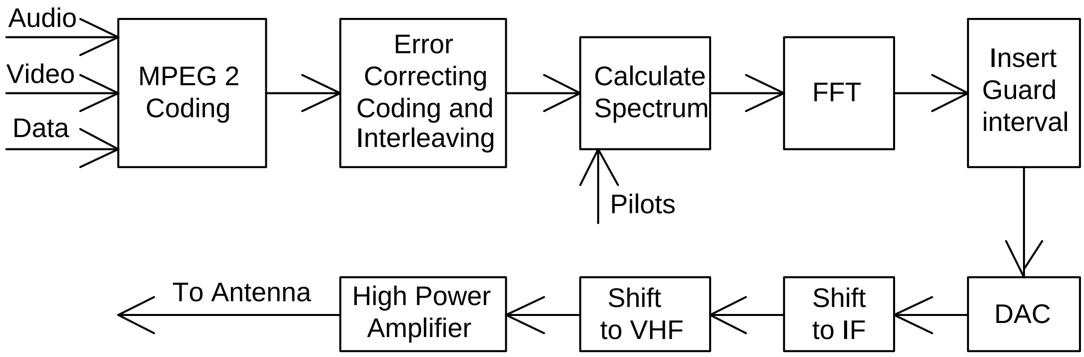


Figure 12. COFDM DVB Transmitter Block Diagram.

TV Type	Hor Pixels	Vert Pixels	Apect Ratio	Frame Rate	Sample Rate MSampl-e/s	Raw Data Rate MSample/s	Coded Data Rate Mbit/s
DVB-HDTV	1920	1080	16:9	25	51.840	1244	20
DVB-HDTV	1440	1080	16:9	25	38.880	933	20
DVB-HDTV	720	576	16:9 & 4:3	50	20.736	498	10
DVB-SDTV	720	576	16:9 & 4:3	25	10.368	249	5
ATSC-HDTV	1920	1080	16:9	30 & 24	62.208	1493	20
DVD	720	576 & 486	16:9 & 4:3	25 & 30	10.368 & 10.498	248 & 251	5 Ave 10 Max

Table 3. Video Formats and Data rates.

Figure 12 shows a typical block diagram of a Digital Television generator. In Australia [11], the best HDTV definition is 1920x1080 pixels at a frame rate of 25 pictures per second. As shown in Table 3, this corresponds to 51.84 MPixels/sec. At 8 bits per pixel, this corresponds to 414.72 MBit/sec per colour (RGB). This corresponds to 1.244 Gbit per second for the raw data. If required, the data rate can be reduced by sampling the colour information at half the rate, which then results in a raw rate of 829.44 Mbit/sec. For SDTV the screen definition is 720x576 pixels, at 25 pictures per second, giving a raw data rate of 10368 pixels per second. At 8 bits per pixel, this corresponds to 82.944 MBit/sec per colour (RGB). This corresponds to a 248.8 Mbit/sec raw data rate. That data is fed into the Digital Signal Processing (DSP) Unit. The DSP unit takes these RGB data streams and uses the MPEG compression techniques and the relevant transmission formatting to produce the coded video data. Audio data and Data to be transmitted are combined with this signal, to form one data stream of up typically 20 Mbps for HDTV.

That data stream is then coded onto the COFDM carriers, using QPSK, 16-QAM or 64-QAM, depending on the data rate requirements of the channel. When Hierarchical Coding us used using 64-QAM or 16-QAM then the QPSK data rate corrseponds to the reduced performance data rate. It can be seen that this is good enough for SDTV. All the carriers and their phases are generated inside the DSP unit. For a typical single program channel application, the data is split into 256 μsec blocks for a 2 k mode and 1024 μsec block for an 8 k mode. The data block is then used to calculate the phase of each of the many carriers

Modulation Type	Data Rate
DVB QPSK typ data rate 1/8 Guard, 2/3 Code	6.451 Mbit/s
DVB 16-QAM typ data rate 1/8 Guard, 2/3 Code	12.902 Mbit/s
DVB 64-QAM typ data rate 1/8 Guard, 2/3 Code	19.353 Mbit/s
ATSC 8-VSB, (4 level data, with trellis coding)	19.3 Mbit/s

Table 4. DTV Modulation Types and Data rates.

used in the system and generate the resulting time function. An FFT is then applied to that data, using many DSP chips operating in parallel. Very high speed Digital Signal Processors are required to perform this in real time. The resulting 6.65625 MHz bandwidth signal is then modulated onto the carrier using conventional frequency shifting techniques.

In Australia a 7 MHz channel spacing is used, and with DVT adjacent channels can be used. Under those conditions, the transmitted TV spectrum will thus look like a uniform spectrum with 6.65625 MHz carrier blocks and 343.75 kHz gaps between the carriers from adjacent stations. The Digital receiver must be able to handle the high adjacent channel power levels without distortion to the wanted signals.

For the single frequency transmission mode, all the transmitters for a given broadcaster, like the ABC, are operated at the same frequency. For this to occur without performance degradation, the transmitters at the different locations must operate at a frequency difference less than 0.1% of the inter carrier spacing, ie for an 8 k mode at a frequency error of less than 1Hz for a transmitter frequency of up to 800 MHz. This is a significant limitation and requires external synchronisation control, which can be achieved using GPS time standards.

With DVB, different modulation methods (QPSK 16-QAM and 64-QAM) , error correcting rates using 1/2, 2/3, 3/4, 5/6 and 7/8 convolutional coding and quard intervals of 1/4, 1/8, 1/16 and 1/323 of the symbol duration are permitted. Several different sound systems can also be used. This gives a very flexible system.

In Australia, the current requirements are for both SDTV and HDTV to be transmitted on the same channel. If the program material for the SDTV and HDTV is the same, hierarchical coding can be used to achieve this without any overheads. However a SDTV set top box would not be able to decode this signal. As a result two seperate data streams need to be coded. The SDTV program will require between 4 and 8 Mbit/s and the HDTV program will be about 16-19 Mbit/s. Using a guard period of 1/16 of the symbol duration and 2/3 convolutional coding results in a data rate of 20.5 Mbit/s and using 3/4 convolutional coding results in a data rate of 23 Mbit/s, both of which fit into the 7 MHz TV channel. It should be noted that even though the HDTV picture has twice the vertical resolution and more than twice the horizontal resolution, the data rate is slightly less than 4 times the SDTV image, since bigger compression ratios can be achieved on the larger picture. This requires a Carrier to Noise Ratio (CNR) of at least 15.4 dB for a BER of 0.02% after the Viterbi decoder.

At present, all stations in Australia use the 8 k mode of transmission with 64 QAB modulation, the SBS and channel 7 use a 2/3 FEC and a 1/8 guard interval, giving a data rate of 19.353 MBPS. The ABC, the Nine and Ten networks use a 3/4 FEC code and 1/16 guard period, giving a data rate of 23.053 MBPS. In line with the frequency planning proposals for Digital Television in Australia, currently available Digital Set Top Boxes only permit the reception of VHF band III and UHF. In Australia, the 88-108 MHz FM band will thus be free from TV signals once the PAL transmissions are terminated.

ATSC

In the USA, a system with a much simpler modulation technique is used, thus reducing the cost of home receivers, but sacrificing transmission quality. This system was proposed in 1993 by the Digital DHDTV Grand Alliance, a group of companies, including AT&T, General Instruments Corp, MIT, Philips, Sarnoff, Thomson and Zenith. The system used 4 level AM, which is trellis coded to give 8 levels in total. This 8 level AM is then VSB filtered to produce the transmitted waveform. Because the symbol rate is close to 10MSymbol per second, the system is much more susceptible to multipath propagation then the DVB system. As a consequence external antennas must be used with the ATSC system. Adaptive equalisation must also be used in the TV transmitter, to ensure the group delay remains constant during the VSB filtering and transmission. The ATSC system is more tolerant to Impulse noise than DVB. The cost of ATSC receiversshould be less than DVB receivers as no FFT needs to be performed inside the receiver. There is no choice of error correction system or sound system type, as this is all fixed in the standard.All ATSC transmissions are High definition 1920x1080 and this is providing an incentive for HDTV display development. The ATCS system is however much more susceptible to multipath propagation, so that an outdoor antenna must to be used. The takeup rate so far has been poor.

Sound and Video Coding for TV

For Video transmission the Motion Picture Expert Group (MPEG) has been working to produce standards for coding and multiplexing methods for video and audio. The MPEG-2 standards cover the coding and multiplexing of TV signals proposed in ITU-R recommendation 601 and coding for HDTV in a wide variety of applications. Both the DVB and the ASTV systems use MPEG2 coding. The MUSICAM (ISO/IEC 1172/3 MPEG Audio Layer II) process is part of this standard and is used by the DVB system. In the MUSICAM system, sounds which cannot be identifies by the ear, such as a low level frequency component close to a high power signal, are removed. This allows the raw CD stereo rate of

System Parameter	Mode 1	Mode 2	Mode 3
Frequency range	< 375 MHz	< 1.5 GHz	< 3 GHz
Frame Duration	96 ms	24 ms	24 ms
Null Symbol	1.297 ms	324 μs	168 μs
Guard Interval	246 μs	62 μs	31 μs
Data Symbol	1 ms	250 μs	125 μs
Total Symbol	1.246 ms	312 μs	156 μs
Number of carriers	1536	384	192
Carrier Spacing	1 kHz	4 kHz	8 kHz
Bandwidth	1.536 MHz	1.536 MHz	1.536 MHz

Table 5. Digital Audio Broadcasting Formats and Data rates.

1.41 Mbit/s to be reduced to 256 kbit/s without audible degradation.

The sound options allowed in the Australian standard are Mono, Stereo, 2 channel Dolby surround encoded stereo and Multichannel audio. The multichannel Audio is the AC-3, 5.1 channel sound. This is the same sound system used in movie theatres and the ATSC system, with three speakers in front, two in the back and a sub-woofer to give a sound that you can feel rather than hear. The AC-3 sound option initially was not part of the DVB standard but is now incuded.

During 2004 the cost of digital TV SDTV set top boxes dropped to less than $200, resulting in a fairly rapid takeup of digital TV, with more than 250 thousand set top boxes being sold in a six month period. The cost of HDTV set top boxes is less than $700. All these set top boxes have AC3 sound.

DAB

COFDM is used for the Digital Audio Broadcasting (DAB) system which was introduced in the UK during 1995. However it is only recently that receivers have become avaiable at a reasonable cost. In addition different frequency bands are allocated across Europe for DAB, restricting the universal appeal of DAB. The Eureka 147- DAB system has a bit rate is be approximately 1.28 Mbps for 5 stereo channels corresponding to 256 kbps for each stereo channel. Error correcting techniques bring this bit rate up to just over 2.4 Mbit/s, which fits inside the 1.54 MHz spectrum allocation for each channel. The data period is 1.246 ms, corresponding to a carrier frequency sapacing of 1kHz. As shown in Table 4, the allocated bandwidth of 1.54 MHz allows 1536 carriers at this frequency spacing.

DAB was initially due for introduction in Australia during 2001, using frequencies at 1.5 GHz. There is at present a number of transmission frequencies and digital radio broadcasting standards[12,13] around the world. This has resulted in a high receiver cost and lowuser acceptance. The latest information on the ABA website does not mention any introduction dates.It is likely that in the future some digital broadcasting system will be adopted. The acceptance of stereo AM was very poor worldwide, mainly due to 4 standards being allowed, a similar situation is likely to arise for DAB. In contrast to the introduction of Digital TV, where the ABA web site is updated very frequently, no new announcements on DAB have been made since 1998.

The 44.1 ksamples/sec 16 bit stereo data from a CD player will be coded using the MUSICAM process. This coder analyses the sound in separate frequency bands and removes sounds which cannot be detected by the ear. This allows the bit rate to be reduced from 1.41 Mbps for the raw stereo CD data to typically 256 kbps for the processed data with little audible difference. Up to 5 or possibly 6 such stereo signals will be combined into one RF signal for broadcasting. The resulting 1.28 Mbps, or 1.152 Mbps for six 192 kbps data streams, will be coded into a 1.5 MHz bandwidth signal using COFDM.
If the bit rate on each of the COFDM carriers is 1.25 ms, the same transmission frequency can be used for the same programme throughout the country and the signal from more than one transmitter is utilised in locations where both are received in a comparable signal strength.

ADSL

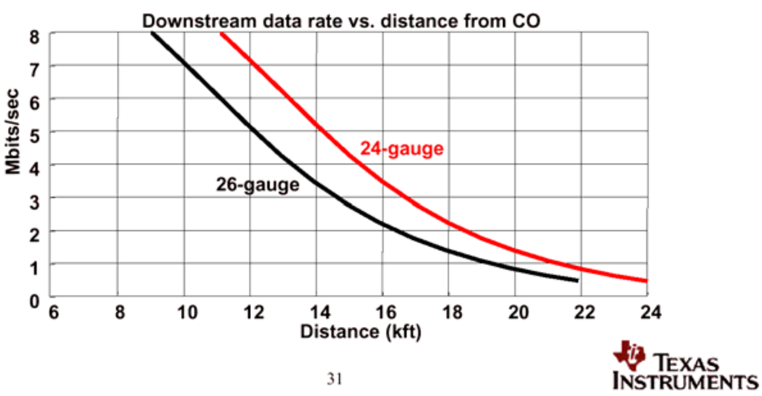


Figure 13. ASDL Data rates (Texas Instruments[14]).

Asynchronous Digital Subscriber Loop, is a technique where data signals are coded onto copper cable connecting the telephone exchange to the telephone subscriber. The bandwidth from 0- 4 kHz is taken up by the Plain Old Telephone Service (POTS).The frequency band from 25.875 kHz to 138 kHz is used as an upstream band, ie from the subscriber to the exchange. The frequency band from 138 kHz to either 552 kHz or 1104 kHz is used for the reverse link, ie the link from the exchange to the subscriber.

The bandwidth from 10 kHz to 1 MHz is divided up into 256 independent sub-channels, each slightly more than 4.3125 kHz wide. Each channel is QAM modulated with between 2-15 bit of data and trellis coding an option. By measuring the quality of each subchannel on a particular subscriber telephone line, the subchannel capacity can be determined and an appropriate modulation scheme can be assigned to it. The actual number of channels and the frequency of the channel used is thus variable and depends on the line condition. Figure 13 shows the typical data rates that can be achieved on copper cables as a function of the distance between the customer and the exchange. A data rate of 1 Mbps can be achieved for distances up to 6 km from the exchange.The cables have the least attenuation at low frequencies and for this reason the upstream data is allocated the lower frequency region, since that has a lower data rate, but is less tolerant to errors than say video on demand.

The first six channels are reserved for analogue telephony. The upstream digital communication link can occupy 24 subchannels, giving an upstream data rate of up to 640kbit/s. The downstream data takes up 248 channels if echo cancellation is used and 222 channels if it is not, giving a downstream data rate up to 6.144 Mbit/s.

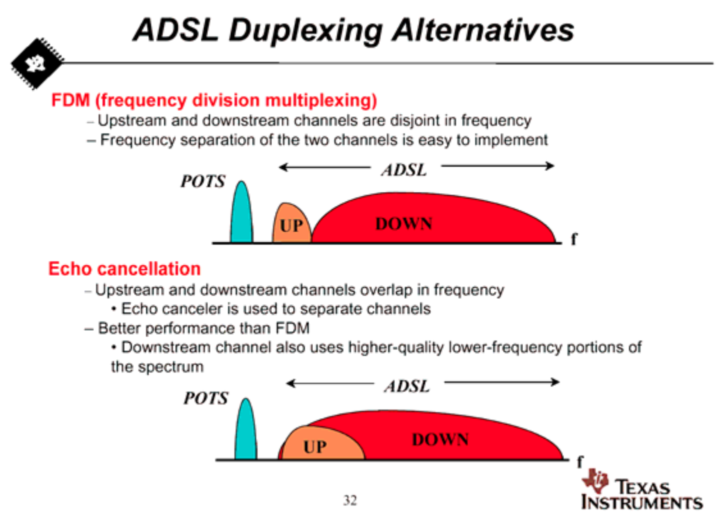


Figure 13. ASDL Duplexing (Texas Instruments [14]).

A simplified version ADSL-Lite has 512 kbit/s upstream capability and 1.5 Mbit/s downstream capability. The ADSL-Lite hardware does not require any splitter to seperate the POTS, and this reduces installation costs. ADSL lite is also known as G.Lite has 128 subchannels, with a modulation of 8 bits (256QAM) per carrier. ADSL-Lite is very suitable for Internet access, the full ADSL system is required for video on demand.

The operation of ASDL is very similar to OFDM, except that we have Multitone Channels, the carriers of which are not synchronised and since Multipath problems do not occur on cables, no guard period is used.These are several versions of xDSL available, HDSL, ADSL, VDSL. HSDL was the first version , is symmetriacl (same uplink and downlink speed) and is the slowest, VDSL is the fastest and is under development. VSDL has up to 26 Mbps upstream and 56Mbps downstream data speed. Even inside ASDL, there are several versions available the ASDL-Lite version is a popular one.

If no echo cancellation is used, the upstream data will produce a significant amount of crosstalk in the downstream channel. To avoid this the upstream (ie sent by the user) channel and the downstream (ie received by the user) channel are allocated different frequencies. If echo cancellation is used, the crosstalk is reduced sufficiently to permit the downstream data to occupy the same frequency region as the upstream data. Since this is a lower frequency region, less attenuation and noise results, giving rise to better communication at the expense of more complesx hardware. A good source for information on ADSL is available from the ADSL Forum web site[15].

In Australia ASDL was introduced late in 2000. The ADSL-Lite version available in Australia There are three data rates available depending on how much one pays. The lowers rate is 64 kbps upstream and 256 kbps downstream . The medium speed system is 128 kbps upstream and 512 kbps downstream and the high speed service is 256 kbps upstream and 1500 kbps downstream.

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Appendix A1 Digital Mobile and Pager System Summary

System	GSM	DCS-1800 PCN	NADC DAMPS	PDC (JDC)	CDMA	MobileSat	ERMES	POCSAG
Type	Cellular	Cellular	Cellular	Cellular	Cellular	Satellite	Paging	Paging
Location	World Wide	World Wide	North America	Japan	World Wide	Australia	Europe	World Wide
Service start	1991	1993	1993	1994	1995	1994		
Frequency Range	890-915 MHz (to mobile) 935-960 MHz (to mobile)	1.71-1.785 GHz (to mobile) 1.805-1.88 GHz (to mobile)	824-849 MHz (to base) 869-894 MHz (to mobile)	810-826 MHz 940-956 MHz 1429-1441 MHz 1453-1465 MHz 1477-1489 MHz 1501-1513 MHz	824-849 MHz (to base) 869-894 MHz (to mobile)	1545-1559 MHz (down) 1646-1660 MHz (up)	169.4125-169.8125 MHz	130 MHz to 570 MHz (depends on country)
Spectrum Allocation	50 MHz	150 MHz	50 MHz	80 MHz	50 MHz	14 MHz	400 kHz	typ 175 kHz
Users per Channel	8-16	8-16	3-6	3-6	118	1	1	1
Channel Spacing	200 kHz	200 kHz	30 kHz	25 kHz	1.23 MHz	7.5 kHz	25 kHz	25 kHz
Number of Channels	124	374	832	1600	10	2 x 500	16	typ 7
Number of Callers	992	2992 & 5984	2496	4800	1180	50 000		typ 1 Million
Mobile Output Power	20 mW - 20 W	2.5 mW-1 W	2.2 mW - 6 W	up to 2W	0.2 to 6.3 W	10W	NA	NA
Data Structure	T&FDMA/FDD	T&FDMA/FDD	T&FDMA/FDD	T&FDMA/FDD	CDMA/FDD	FDMA	TDMA	TDMA
Modulation	GMSK	GMSK	$^{pi}/_{4}$DQPSK	$^{pi}/_{4}$DQPSK	DQPSK/ ODQPSK	$^{pi}/_{4}$DQPSK	4 FSK, 1.5625 & 4.6875 kHz	FSK 4 kHz
Codec	RELP-LPT 13 kbps	RELP-LPT 13 & 6.7 kbps	VSELP 7.95 kbps	VSELP 7.95 kbps	QCELP (var) 8.55 kbps	IMBE 4.2 kbps	NA	NA
Modulation Data Rate	270.833 kbps	270.833 kbps	48.6 kbps	42 kbps	1.2288 Mbps	6.6 kbps	6.25 kbps	4.8, 2.4, 1.2, 0.512 kbps
Filter	0.3 Gaussian	0.3 Gaussian	RRCos r=0.35	RRCos r=0.5	FIR 615 kHz	RRCos r=0.4	Bessel 3.9 kHz	
Frame Duration	4.615 ms	4.615 ms	40 ms	40 ms	20 ms	20 ms		

Appendix A2 Digita Telephone and Fixed Radio System Summary

System	CT2	CT3	DECT	PHP (JCT)	TFTS	TETRA	APCO-25	DSRR
Type	Cordless Phone	Cordless Phone	Cordless Phone	Cordless Phone	SkyPhone	Trunk Radio	Trunk Radio	Short Range Radio
Location	World Wide	Europe	World Wide	Japan	World Wide	Europe	USA	Europe
Service start	1989	1990	1994	1994		1995		
Frequency Range	864-868 MHz	862-866 MHz and others at 1.8GHz	1.88-1.9 GHz	1895-1907 MHz	1.67-1.675 GHz (ground to air) 1.8-1.805 GHz (air to ground)	380-400 MHz 410-430 MHz 450-470 MHz 915-933 MHz 970-988 MHz	806-821 MHz Base Station 851-866 MHz Mobile	888-890 MHz 933-935 MHz
Spectrum Allocation	4 MHz	4 MHz	20 MHz	12 MHz	10 MHz	96 MHz	30 MHz	4 MHz
Users per Channel	1	8	12	4-8	2	4	1	78
Channel Spacing	100 kHz	1 MHz	1.728 MHz	300 kHz	30.3 kHz	25 kHz	12.5 kHz	25 kHz
Number of Channels	40	8	132				up to 600	77
Mobile Output Power	1 mW - 10 W	low power	250 mW	10 mW				<5W
Data Structure	TDMA/TDD	T&FDMA/TDD	T&FDMA/TDD	T&FDMA/TDD	TDMA/TDD	TDMA/TDD	TDMA	T&FDMA/TDD
Modulation	GMSK	GMSK	GMSK	$^{pi}/_{4}$DQPSK	$^{pi}/_{4}$DQPSK	$^{pi}/_{4}$DQPSK	C4FM/CQPSK	GMSK
Codec	ADPCM 32 kbps	ADPCM 32 kbps	ADPCM 32 kbps	ADPCM 32 kbps	VOCODER 9.6 kpps	VOCODER 7.3 kbps	IMBE 4.4 kbps	RELP-LPT 13 kbps
Modulation Data Rate	72 kbps	640 kbps	1.152 Mbps	384 kbps	44.2 Mbps	36 kbps	4.8 kbps	16 kbps
Filter	0.5 Gaussian	0.5 Gaussian	0.5 Gaussian	RRCos r=0.5	RRCos r=0.4	RRCos r=0.4		0.3 Gaussian
Frame Duration	2 ms	16 ms	10 ms	5 ms	80 ms	60 ms	20 ms	10 seconds max.