DIGITAL VIDEO BROADCASTING STANDARDS FOR SATELLITE, TERRESTRIAL AND CABLE TELEVISION TRANSMISSION (INVITED)

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ABSTRACT

The paper describes the Digital Video Broadcasting (DVB) technical standards used for the transmission of digital television over satellite, terrestrial and cable networks. The paper explains the principles of both the source and channel coding used in the family of DVB standards. The main focus will be on the DVB-T terrestrial standard together with the plans for starting digital television services in the United Kingdom (UK).

INTRODUCTION

In the UK, our early studies on the feasibility of transmitting digital television to the home began in the late 1980's. It came at a time when there was interest from both government and the general public in providing more choice in the television programme material that was currently available; this meant providing more television channels to the home in addition to the newly emerging analogue satellite services. Finding a new frequency band was not an option. All the useful spectrum capable of transmitting television right up into the low microwave frequency bands had already been allocated. The breakthrough was to find new digital modulation methods that were power, interference and spectrally more efficient than the vestigial sideband amplitude modulation (VSB-AM) system used by analogue television. The biggest breakthrough of all was to be able to show that these new digital modulation methods would enable additional television channels to share the same UHF spectrum as the old analogue channels without causing interference, see references 1 & 2.

MPEG SOURCE CODING

When broadcast quality television is digitised the resulting bit rate is at least 216 Mbits/s, which is too large to send within the bandwidth of current television channels. In order to reduce the bit rate to values which can be transmitted then large compression ratios need to be employed which reduce the bit rate to some 2-15 Mbits/s without noticeable picture degradation. The compression scheme which has been adopted on all broadcast transmission media is the world-wide ISO standard MPEG 2 compression algorithm, see references 3 & 4. The MPEG standard also defines a method of compressing the audio, from 1536 Kbit/s to 256 Kbits/s for stereo, using MPEG layer II coding.

The video compression algorithm uses three main methods to remove the temporal, spacial and statistical redundancy which exists in all television images. The temporal redundancy is removed by making use of the fact that in many areas of the picture there is little in the way of movement and pixels from adjacent television frames are very similar. Temporal redundancy is removed by subtracting the pixels in adjacent television frames to produce smaller difference

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values. In order to improve this process an estimate of the local motion is made and the second frame is shifted by these motion vectors in order to reduce the predicted difference values.

Spacial redundancy represents the redundancy in the scene texture and makes use of the fact that within a television frame many of the pixels are often similar, an example would be in large areas of sky. Spacial redundancy is removed by taking the two-dimensional Discrete Fourier Cosine Transform (DCT), eliminating the zero valued coefficients and severely quantising the DCT values that remain. This quantisation process causes some picture degradation which is designed to be at the limit of visual perceptibility.

Statistical redundancy is removed by using Huffman coding. This process takes in fixed length compressed bit streams, having different probabilities of occurrence, and produces variable length code words where the codeword length is inversely proportional to the input codeword probabilities.

Audio Compression is achieved by first splitting the signal into a set of frequency sub-bands. The sub-bands are used to take account of the fact that the signal does not use all the audio bandwidth all of the time. Hence, some of the sub-band outputs can be ignored and the others can be quantised to a level which only produces just noticeable distortion to the reconstituted audio signal. This quantisation distortion can be increased by the use of a Psycho-acoustic Model (PAM). This PAM process makes use of the fact that the human ear cannot hear soft sounds which occur at the same frequency as loud sounds. The soft sounds are essentially masked by the loud sounds of the same frequency. The PAM detects when this situation occurs and adjusts the quantisation process accordingly to reduce the resulting output bit-rate.

FREQUENCY PLANNING

In order to fit additional channels into the UHF television spectrum, without displacing the existing system, we need to design a new transmission signal format which meets the criteria that: the new signal must not cause interference into the existing analogue television signal; the new signal must have a similar service area to the existing analogue television signal, and; the new signal must not suffer interference from the existing analogue television signal.

With an all digital signal, the only way to fulfil the first criterion is to transmit the new digital television signal at much reduced power. Since the interference characteristic of the digital signal is noise-like into the existing analogue television signal, the co-channel protection ratio cannot really be altered from the just perceptible noise interference protection ratio of the analogue system. In order to achieve the non-interference transmissions the new digital television signal has to be transmitted at some 20dB lower power compared with its analogue VSB-AM equivalent.

Fortunately, the second criterion of similar coverage areas for the new and old television services can still be met for the digital signal even though it is transmitted at some 20 dB lower power. This arises because the improvements in C/N ratios achievable by digital systems employing modern error coding properties are in excess of the 20 dB reduction in power level. Typically the C/N for the analogue television system is some 40 dB whereas the C/N for a digital system is between 10dB and 20 dB, depending on the modulation scheme chosen.

The third criterion of non-interference into the digital system from the analogue system also results from the enhanced interference properties of digital signals. However, we need to look at the analogue television signal as an interferer in a little more detail. As an interferer, the terrestrial analogue VSB-AM signal has two main components, the vision carrier and the sound carrier, which are at well defined frequencies. The VSB-AM sidebands are at much lower power in comparison. The VSB-AM signal can then be approximated as two high power continuous wave (CW) signals with a much lower power broadband noise like interference signal represented by the sidebands. Provided the new digital signal can withstand the two CW jamming signals represented by the two carriers then the digital signal will be very robust to interference from the existing VSB-AM signal. It turns out that by using the Orthogonal Frequency Division Multiplexed (OFDM) signal, which comprises some 2000 narrow band overlapping carriers, then significant interference immunity can be provided to the digital signal. In the early days, this co-channel interference immunity was provided by omitting carriers which were at the same frequencies as the two VSB-AM carriers. In this way holes were cut in the digital spectrum to form two complementary matched spectra, as shown in reference 1. However, it was later found that by applying soft decision error coding and channel state estimation techniques to the OFDM signal, 'virtual holes' could be provided by the error decoding method in the receiver, and the use of actual spectral holes was unnecessary. Eliminating the actual holes increased the overall data capacity and the channel decoding performance was not compromised for receivers which were not located at the edge of the service area - where co-channel interference performance is at its most critical.

The reason why new digital television services can be inserted into the frequency plan, whereas it was not possible to introduce further analogue services, is because there is still some redundancy left in the existing plan - however, analogue services cannot make use of it. The main factor governing the number of channels that can be transmitted per transmitter is dependent on the frequency re-use distance. Since digital signals have superior interference properties they will have a shorter frequency re-use distance and more channels per transmitter can be accommodated. Another factor which is very important in allowing frequencies for these new digital signals to be found, in what appears to be a full analogue TV frequency plan, is that there exist channels which were sterilised by the analogue transmitter and receiving equipment performance specifications available in 1961, when the original plan was devised. These unusable frequencies are known as the 'taboo' channels and they basically consist of the adjacent, image and receiver local oscillator channels. It turns out that, within the UK, the channels that are actually found for digital television are mostly the upper and lower adjacent channels to the existing analogue television services. For a full description including diagrams of the theory of how additional digital channels are introduced into the analogue lattice plan see reference 1.

OFDM CHANNEL CODING

Orthogonal Frequency Division Multiplexing consists of a large number of carriers equally spaced in frequency at harmonics of the first carrier frequency. Each carrier is modulated by some digital modulation method such as 4-PSK or 64 QAM. The OFDM signal is like a traditional FDM signal but the sidebands of the individual carriers overlap each other. Since the multiplex is orthogonal the overlapping sub channels do not interfer with each other. The multiplex is constructed using the Fourier transform and the multpliplex is orthogonal because the sub-carrier frequencies are at harmonic intervals, the signal is mathematically orthogonal for the same reason that a Fourier series is orthogonal. Indeed, the modulation of large numbers of sub-carriers is only possible because the Discrete Fourier Transform (DFT) process can be applied to the incoming signal. In fact modulation and multiplexing are applied in one operation by using the inverse DFT. Demultiplexing and demodulation are performed at the receiver by using the opposite DFT process, see reference 5.

The reason for choosing OFDM for the digital terrestrial television transmission signal was two fold. Not only did it give a signal that could be tailored to reject interference from the analogue VSB-AM signal, as already described above, but it also gave excellent performance in the presence of multipath propagation. Multipath is a major feature of UHF terrestrial propagation and potentially very damaging to high bit rate digital signals. The OFDM signal, together with its associated channel estimation and error decoding strategy, is very insensitive to multipath propagation. In fact 0 dB echoes, i.e. the signal and echo are at the same power level, result in error free reception. Multipath propagation produces two mechanisms which damage digital signals. The first is intersymbol interference, which is a time domain phenomenon, which causes one digital symbol to spill over into the next symbol position causing destructive interference. OFDM is inherently robust to intersymbol interference because the time duration of each sub-channel symbol is very long, and in fact the number of carriers are chosen to give a symbol period which is very long compared to the delay spread of the naturally occurring multipath. The more carriers, the longer will be the symbol period of each sub-channel. The OFDM system is like a parallel data modem where the data rate of each sub-channel is the total data rate divided by the number of sub-channels.

The second multipath mechanism is frequency selective fading; this mechanism occurs in the frequency domain and is caused by the relative phases of the main and delayed signals producing either an addition or subtraction of the two signals. At frequencies where the two phases subtract the signal is said to have faded, at frequencies where the two phases add an enhancement in the level of the signal takes place. In the case of OFDM, this will mean that certain sub-channels are either enhanced or reduced into the background noise floor. However, since the signal contains channel estimation, the precise carriers which are affected can be detected in the receiver and the soft decision Viterbi error decoding strategy can make use of the reliability information provided by the respective carriers. This reliability information will enable the Viterbi error decoder to produce a better overall decoded bitstream, with fewer errors. The improved decoding occurs because the more reliable bits mitigate against the less reliable bits and cause the Viterbi decoder to follow a decoding path that is better than had the information about the more reliable bits not been present, or indeed the information about the unreliable bits just been ignored. In this way a surprising result occurs in that the multipath signal actually adds constructively to the original signal - the secret is in the error decoding strategy.

DVB-T

A full description of the DVB-T system is described in reference 6 which is the European Telecommunication Standards Institute (ETSI) standard for digital terrestrial broadcasting A detailed block diagram is shown in the ETS 300 744 standard. The main functional blocks for the transmit end consists of an energy dispersal scrambler, a Reed-Solomon (204,188,8) error correcting coder, a convolutional interleaver operating on bytes, an inner error correcting code which is a punctured Viterbi convolutional encoder (the puncturing allows all coding rates of n/(n+1) to be achieved), a further bit wise interleaver which gives the receiver improved multipath performance and then the OFDM multiplexing and modulation process (which is basically an inverse DFT operation).

The OFDM process can either produce around 2000 carriers or around 8000 carriers. The 8000 carriers enable the system to work with large Single Frequency Networks. Since OFDM has excellent multipath performance it is possible to have two adjacent transmitters on the same frequency provided the bit streams are identical. This is because the signal from the second transmitter just looks like an active multipath version of the signal from the first transmitter. Each carrier is modulated in either a QPSK, 16-QAM or 64 QAM constellation. There is also the possibility to use non-uniform combinations of these three schemes in order to provide a multi-resolution transmission system having hierarchical staggered failure points. The DVB-T signal is suitable for terrestrial broadcast television transmission within the VHF and UHF bands. It can also be used at 2.5 GHz for the microwave video distribution service (MVDS) .

DVB-S AND DVB-C

The DVB-S satellite modulator has a similar block diagram to the DVB-T modulator except that the system only uses one carrier modulated with QPSK, see references 7 & 8. Multipath is not a problem with satellite transmission and the non-linearity of the satellite at present limits us to QPSK modulation. The system uses the same concatenated Viterbi convolutional and Reed-Solomon error correcting codes but the bit-wise inner interleaver is not used. Since satellite bandwidths are much larger the output bitrate can be made to vary from 2 Mbits/s up to around 60 Mbits/s. The system is suitable for broadcast television transmission within the 12 GHz satellite bands and also for MVDS transmission at 40 GHz.

The DVB-C cable modulator is similar to the single carrier DVB-S modulator except that the modulation scheme on the single carrier is either 16-QAM, 32-QAM or 64-QAM, see reference 9. Furthermore the Viterbi convolutional encoder is omitted leaving the same outer interleaver and Reed-Solomon channel coding in place. The cable receiver does, however, contain an adaptive equaliser in order to deal with the short multipath propagation which exists in VHF and UHF cable systems.

DVB OPERATIONAL EXPERIENCE

In the case of MPEG 2 and DVB-S, my company NDS Ltd has significant experience of designing, manufacturing and installing these systems for major clients. Most of our customers over the last three years have been television service provides wishing to provide operational television services to cable headends, however, we have also supplied to several direct to home broadcasters as well. The system is very flexible and has full operational broadcast redundancy in the event of equipment failure during live transmissions. The system has proved to be very reliable. Major customers, who will be shortly be offering direct broadcast satellite television services to the home in the UK, are now installing the DVB-S system and digital satellite receivers should be in the shops within the next year or so.

Years of field trials in the UK, leading up to what are now experimental transmissions in London of the DVB-T system, have been conducted by both the Independent Broadcasting sector and the BBC. The necessary legislation enabling digital terrestrial broadcasting using MPEG 2 and the DVB-T system has been in place for some time now in the UK. There was a Frequency Planning Conference which covers protection ratios and transmitter frequency assignment for Europe held in Chester, England last year. An ITU-R recommendation for DVB-T now exists. Service providers have been appointed to begin the new digital terrestrial broadcasts in the UK, again new services should start within the next year or so. Receiver manufactures are currently designing the receivers and modulation equipment will soon be installed at major transmitter stations.

The application of DVB-T and DVB-C to MVDS services is still at the field trial stage. We have made several successful field trials of both the DVB-T and DVB-C systems at 2.5 GHz, in both Ireland and South Africa, last year.

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