Coding and Modulation of Satellite DVB Systems

Various steps used by a satellite based DVB communication system

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The world is going wireless. In the last decade the explosion of wireless services has been phenomenal. Companies big and small have jumped on the bandwagon to provide one type of service or another. To insure interoperability, various working groups have been established to define system requirements and interfaces. The result of these efforts is a series of published specifications that developers must adhere to. One such service is satellite based digital video broadcasting (DVB). This article explores the modulation and coding required by one such specification for satellite based DVB broadcasting of HDTV services direct to home, and other applications. The specification described here is denoted by the reference EN 300 421 V1.1.2 (1997-08) which is produced by the European Telecommunications Standards Institute (ETSI). Figure 1 shows the basic block diagram of the system. While the development is specific to the standard, many of the techniques used are common among digital wireless systems.

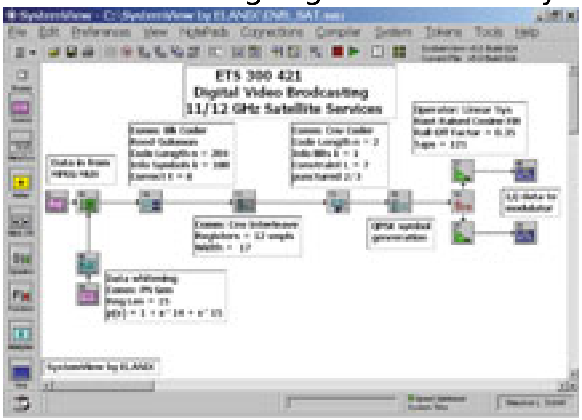

Figure 1. DVB modulation and coding block diagram

MPEG Packet

It is assumed that the data has already been converted to digital form via some encoding technique. The MPEG packet generator multiplexes a number of individual signals before transmission. The exact number is dependent on the bandwidth of the particular satellite transponder. An example is detailed below. The basic packet structure is show in Figure 2. The packet is 188 octets long with the first octet being a sync pattern 01000111. The packets are grouped in blocks of 8 bits (byte). For the first packet the sync word is inverted, for packets 2-8 they are not. Packet 9 inverts the pattern again etc.

Figure 2. DVB packet structure

Data Whitening

In most communication systems, transmission of a long sequence of 1's or 0's is not desired. First of all, the spectrum of the signal in this case reverts to a pure carrier as opposed to a fully modulated spectrum. In addition, the various synchronization loops in the receiver depend on the zero crossings of the data to generate an error signal for the tracking loop. The MPEG output data is first 'whitened' by an XOR operation with a pseudo random data sequence generated by the polynomial $1 + x^{14} + x^{15}$, it has a sequence length of 32767 bits before repeating. The synchronization segments of the packets are already chosen for their good cross-correlation properties, so the whitening is not applied to them. The whitening starts after the first sync byte and continues through the group of 8 packets. The length of the sequence is 7*188+187 = 1503 bytes = 12024 bits. The data sequence is reset to its initial state at this point.

Reed Solomon (RS) FEC

This is the first of two forward error-correcting (FEC) codes. The code is described by 3 numbers [N = 204, K = 188, t = 8]. This coder works on 8 bit (1 byte) symbols. It encodes 188 symbols into 204 symbols, and can correct up to 8 symbol errors in the 204 symbol word. The code is systematic in that the first 188 symbols of the encoded word are the 188 symbols of the input. The remaining 16 are called parity symbols.

Convolutional Interleaver

After the RS encoder, the data symbols are randomized by an interleaver that has dimensions 17 &#215 12 = 204 symbols wide. The symbols are now converted back to bits for the next step in the processing. The interleaver architecture used is known as a convolutional interleaver (having nothing to do with the convolutional FEC). It is more complicated than a simple row vs. column block interleaver. Figure 3 shows a block diagram of the structure. There is a series of 12 parallel shift registers. The first send the data directly through, the second has a 17 tap buffer, the third has a 34 tap buffer and so on. The data is sequentially entered into each bank, one per symbol. The data is read out in the same manner using a commutator switch. The advantage of this structure over block interleavers is a reduction by 2 in the memory and throughput delay. The 204 symbol depth of the interleaver is, of course, chosen to match the 204 symbol length of the RS encoded packet.

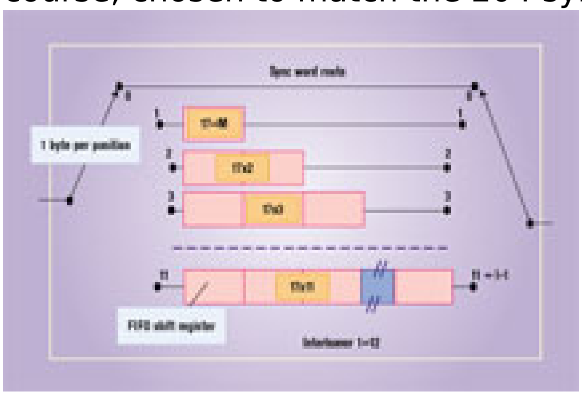

Figure 3. 17 &#215 12 Convolutional Interleaver

Punctured Convolutional FEC

To achieve the stringent BER performance, a second inner FEC is used. This a rate

1/2 constraint length 7 code punctured down to a rate of 2/3. A rate 1/2 encoder produces two output bits for every input bit. The puncturing operation 'deletes' one of every four output bits producing the final 2/3 rate. Convolutional coders that directly produce rate 2/3 codes exist, but it turns out that the decoding algorithm for the punctured approach is easier to implement.

The combination of the convolutional FEC (inner) and the RS coder (outer) is called a concatenated code. Let's explore why this architecture is so popular. The basis is the nature of the convolutional decoder structure. Figure 4 shows a typical error pattern out of a Viterbi decoder produced by the EDA tool SystemView by Elanix. Note that the errors come in bursts. This is due to the fact that the decoder has chosen an erroneous data path through the trellis decoding structure as the most likely answer.

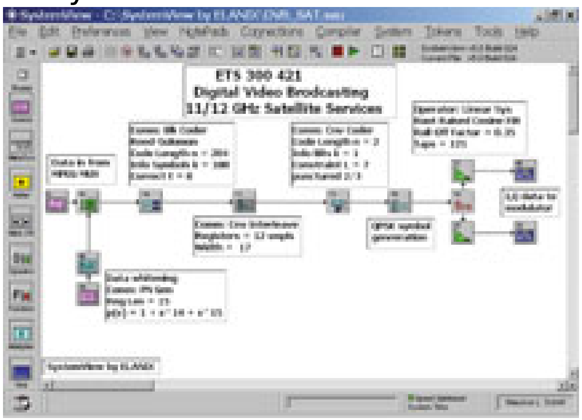

Figure 4. Burst Error Pattern of Convolutional Codes

Now consider the RS decoder. Suppose that there are (luckily) 8 contiguous bit errors that exactly match the span of one RS symbol. The decoder can correct 8 symbols. In short the RS decoder is an efficient processor when the input error pattern has bursts. This is generally not true for the convolutional Viterbi decoder.

QPSK Encoder

The next step is to generate the phase symbols for the QPSK modulator. Every two bits out of the convolutional encoder is converted into one of four phases according the Grey code rule {[00] = &#149/4, [01] = 7/4, [11] =5/4, [10] = 3/4}. QPSK modulation is commonly employed since it carries twice the data rate, or 1/2 the bandwidth for the same data rate, as compared to a BPSK system with the same error rate performance.

We now return to the issue of the system data rate in relation to the transponder bandwidth. For example, for a 54 MHz transponder bandwidth, the useful data rate out of the MPEG mux is 51.8 Mbps using a rate 2/3 convolutional encoder. The rate out of the RS encoder is 51.8*204/188 = 56.2 Mbps. The rate out of the convolutional encoder is 56.2*3/2 = 84.3 Mbps. Finally, the QPSK modulator works on two bits at a time reducing the transmitted bandwidth to 42.15 MHz, which is 54MHz/1.28 by the specification.

Root Raised Cosine (RRC) Filters

The RRC filters are used to limit the occupied bandwidth of the transmitted signal. The name derives from the fact that they are one half of a raised cosine (RC) filter in the frequency domain. At the receiver there is an identical RRC filter. Mathematically the relation can be written RRC*RRC = RC. The RC filter has a time impulse response

$h(t) = [\sin(x)/x]\cos(\beta x)/[1 - 4\beta^2 x^2$

x = Πt/T

and a corresponding spectral response
P(f) = 1 0>=f>=(1 - b)/2T
= 0.5[1 - sin{Π(fT - .5)ß}]
(1 - ß)/2T>=f>=(1 + ß)/2T
= 0 otherwise

The term ß is called the role off factor that is equal to .35 in this case. This RRC filter pair has two advantages. First they are a matched filter set between the transmitter and receiver. Second, the output RC waveform does not exhibit Intersymbol Interference. This is seen by noting that h (kT) = 0, k not equal to 0. One disadvantage of the RRC filters is that the final modulated signal does not have constant amplitude. Thus all amplifiers must be used in their linear range and therefore, can limit power efficiency or total power output.

Traveling Wave Tube Amplifier (TWTA)

The TWTA is a common element in communication satellite technology. For an input sinewave of frequency f and amplitude r, the TWTA is characterized by the relation

$$y(t) = A[r(t)]\sin(2\Pi ft + \varphi[r(t)]\}$$

The empirical relations

$$A(r) = a_r r/[1 + b_r r^2$$

$$\varphi(r) = a_\varphi r^2/[1 + b_\varphi r^2$$

describe A(r) and &#149(r). The first term is called AM/AM conversion, and the second is AM/PM conversion. The four contestants [a_r, b_r, $a^?$, $b^?$] can be determined from the actual TWTA tube measurements via a least square fit. The term A(r)/r is the nominal gain. A plot of A(r) shows the output power increasing with the input, and then leveling off and actually decreasing as the input power continues to increase. This is the saturation phenomena mentioned above. For DVB, the TWTA must be operated below the point where significant effects due to the saturation set in.

Conclusion

In this article described the various steps used by a satellite based wireless DVB communication system. Each step in the modulation and coding was presented along with discussions as to why the various processing steps were used.