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# An Overview of Channel Coding For OFDM System

Lokesh C. [1], Dr. Nataraj K. R.[2], Dr. Rekha K. R [3].

[1] Assistant Professor,Department of E & E E.VVCE,Mysore, India.
[2 & 3] Professor, Department of E & C E ,SJBIT, Bangalore, India.

*Abstract*— **Using a time domain equalizer it is possible to obtain an, branch diversity if the channel Consists of *M* resolvable paths. In OFDM the equalizer does not give you any diversity since all sub-channels are narrowband and experience flat fading. However the structure of OFDM offers the opportunity to code across the subcarriers. In it is shown that with An *M* - path channel it is possible to obtain an M - branch diversity through coding. Hence, in this diversity context a multicarrier system is comparable to a single carrier system besides Combating fading, coding has also been proposed to deal with long echoes that cause ISI Between subsequent OFDM symbols.**

**The design of codes for OFDM systems on fading channels follows many of the standard Techniques. That is special about OFDM is the time-frequency lattice and the possibility to Use two dimensions for interleaving and coding. To illustrate how to use this structure we Give an overview of two systems: the European DAB standard and a trellis-coded system by Hoher. The DAB system uses differential modulation, which avoids channel estimation, while the other system uses a multi-amplitude signal constellation, which requires channel estimation.**

*Index Terms – Digital Audio Broadcasting(DAB), Discrete MultiTone(DMT), Trellis-Coded Modulation (TCM), Pulse Amplitude Modulation (PAM), Viterbi Algorithm (VA), Soft Output Viterbi Algorithm (SOVA), Digital Video Broadcasting(DVB), Signal-to- Noise Ratio(SNR)*

## I. INTRODUCTION

A Major contribution to OFDM is given by DFT to perform baseband modulation and demodulation. This method did not focus on “perfecting the individual channels”, but rather on introducing efficient processing, eliminating the banks of subcarrier oscillators. To assure ISI and ICI the method used a guard space between the symbols and raised cosine windowing in the time domain. The system did not obtain perfect orthogonality between subcarriers over a dispersive channel, but it was still a major contribution to OFDM.

Another important contribution to OFDM is given by *cyclic prefix (CP)* or cyclic extension, with which we can resolve the problems of orthogonality problem. Instead of using an empty guard space, they filled the guard space with a cyclic extension of the OFDM symbol. OFDM system is usually designed with rectangular pulses, but recently there has been an increased interest in pulse shaping. By using pulse other than rectangular, the spectrum can be shaped to be well-localized in frequency, which is beneficial from an interface point of view.

OFDM is currently designed with digital audio broadcasting standard. Several DAB systems proposed for OFDM and its applicability to digital TV broadcasting is currently being investigated. OFDM, under the name DMT has also attracted attention as an efficient technology for high speed transmission on the existing wireless network.

## II. CHANNEL COADING FOR WIRELESS SYSTEM

### A. Digital Audio Broadcasting

Digital broadcasting to mobile receivers was under considerable investigation before the concept of OFDM in wireless communication. The standard for DAB was set to OFDM while it is still under investigation [1]. The DAB system uses *differential quadrature phase Shift* (DQPSK) to avoid channel estimation. The channel encoding process is based on punctured convolutional coding, which allows both equal and unequal error protection. As a source code, a rate ¼ convolutional codes with constraint length 7 and octal polynomials is used. The puncturing procedure allows the effective code rate to vary between 8/9 and ¼.

Interleaving is performed in both time and frequency. The former is a kind of block interleaving after which the bits are mapped to QPSK symbols. The frequency interleaver, working with the QPSK symbols, follows a permutation rule of the 2048 subcarriers. In one of the three transmission modes, this permutation rule is used.

$\Pi(0) = 0$; $\Pi(n) = 13\,\Pi(n-1) + 511 \pmod{2048}$, $n = 1, 2 \ldots 2047$.

This permutation defines the set
$\{\Pi(0), \Pi(1), \Pi(2) \ldots \Pi(2047)\} = (0, 511, 1010, \ldots\ldots 1221\}$

According to which the interleaving pattern is chosen. After the frequency interleaving, the QPSK symbols are differentially modulated on each subcarrier.

### B. *Trellis- Coded OFDM*

As an example of multi amplitude, coherent, and coded OFDM system, we consider the system investigated by Hoher. The investigation focuses a digital audio broadcasting scenario, but the concept is more general. The proposal given focuses on power and bandwidth efficient concatenated coding system for data transmission on time and frequency selective channels. An overview of the system is depicted in Figure 1.

Figure 1: Overview of the system investigated by Hoher.

The outer codes are *rate-compatable punctured codes* (RCPC) derived from the rate ½ codes with constrained length 7. The outer interleaving scheme is applied to break up error bursts from the inner coding system. The inner code is binary *trellis-coded modulation* (TCM) with one-dimensional signal constellation. The reason for this choice is that they were found to provide a good diversity factor at a very low decoder complexity. The code used is a one dimensional 8 state code with 4-level (uniform) *pulse amplitude modulation (*4-PAM). The 4-PAM output symbols are then combined to a 16 symbol *quadrature-amplitude modulation (16-QAM)* constellation and inter Leaved to break up channel memory. In the receiver the *viterbi algorithm* (VA) is used for decoding. This algorithm is capable of using the channel state information obtained from a pilot sequence.

$$J=\Sigma|h|^2|y-x|^2 \qquad \ldots\ldots\ldots\ldots \text{Eq(1)}$$

The decoding is performed by minimizing the metric where $h$ is the channel estimate, $y$ is the de-interleaved observation after equalization (the real or imaginary part) and $x$ is a potential codeword. Because of the outer code, the VA should be modified to provide reliability estimates together with the decoded sequence. This enables soft decoding of the outer code as well. By applying a *soft output viterbi algorithm* (SOVA). An improvement of 2 dB is obtained. Together with the inner coding, multicarrier signaling and slow frequency hopping, the interleaver provides dual time/frequency diversity.

### C. *Other Systems*

There has been other coded OFDM system proposed and analyzed, in an OFDM/FM system is investigated and simulated. OFDM is proposed as the transmission technique for the new DVB system [2, 8], where a multi resolution scheme is used together with joint source/channel coding. This allows several bit-rates and Thereby a graceful degradation of image quality in the fringes of the broadcast area.

### D. *Coding on Fading Channels*

Usually performance analysis of codes assumes perfect knowledge of the channel. However, in an analytical method was introduced that allows non-ideal channel information. This was later generalized to include non-ideal interleaving. This method has been used to analyze a coded OFDM system with pilot-based channel estimation on Rayleigh-fading Channels. A Major benefit of using analytical methods for evaluation of coded systems is that a coded bit-error rate can be obtained quickly after system modifications, without Time consuming simulations.

## III. CHANNEL CODING FOR WIRED SYSTEMS

An important difference between a wired and a wireless system is the characteristics of the channel [3, 4]. In the wired case the channel is often considered stationary, which facilitates a number of techniques to improve the communication system. Channel coding in combination with a technique called bit loading is often employed for this purpose. Multidimensional trellis codes are well suited for the channel coding. When using bit loading the sub channels are assigned individual numbers of bits according to their respective SNR's. An OFDM-based communication system using bit loading is often referred to as a DMT system.

### A. *Bit Loading*

Bit loading is a technique that is used for multicarrier systems operating on stationary channels. A Stationary channel makes it possible to measure the SNR on each sub-channel and assign Individual numbers of transmitted bits. A Sub channel with high SNR thus transmits more bits than a sub-channel with low SNR. Figure 2 shows a schematic picture of SNR and how the number of bits on each sub-channel varies accordingly.

Figure 2: Channel SNR and corresponding number of bits on each subcarrier.

When performing bit loading one usually optimizes for either high data rate, low average Transmitting energy, or low error probability. Typically two of these are kept constant and the Third is the goal for the optimization.

Which parameter should be optimized depends on the System, its environment, and its application.

When there is only one system operating on a cable, this system neither interferes with nor is interfered by other systems. This means that controlling the transmitting power to reduce Crosstalk is not necessary. Given a data rate and a bit error probability, whatever transmission Energy needed to achieve these goals can be used (within reasonable limits). In a multisystem Environment, where there are several systems transmitting in the same cable, the problem is More complicated, since the systems experience crosstalk. The level of crosstalk is proportional to the transmitting power in the systems. It is therefore desirable to have an equal transmission power in all systems, to obtain equal disturbance situations. In a multi System environment the average transmitting power is usually fixed, and the optimization is for either high data rate or low bit error rate.

### *B. Bitloading Algorithms*

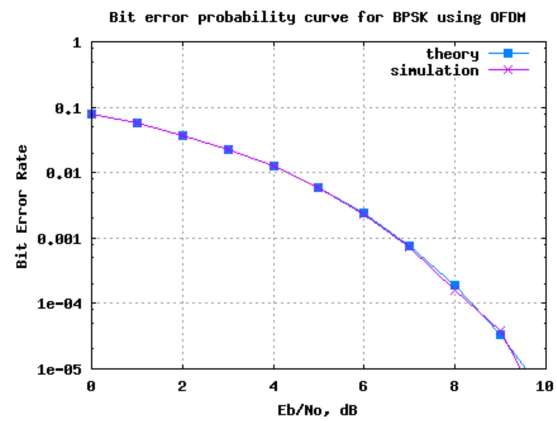
There are several techniques for bit loading in DMT systems are there in which several parameters that one can optimize for. Most algorithms optimize for high data rate or low bit error rate. Given a certain data rate and an energy constraint, the Hughes-Hartogs algorithm provides the bit loading factors that yield minimal bit-error rate. The idea behind the Hughes Hartogs algorithm is to assign one bit at a time to the sub-channels. The algorithm calculates the energy cost to send one bit more on each sub-channel. The sub-channel with the smallest energy cost is then assigned the bit. This procedure is repeated until a desired Bit rate is obtained [5].

### *C. Channel Coding*

A Typical coding scheme for DMT consists of an outer code, an interleaver, and an inner code [10]. Due to the type of applications that DMT Is designed to carry, it is appealing to keep a low delay between transmitter and receiver. This limits the interleaving depth and affects the choice of error correcting codes. The coding scheme uses an outer Reed Solomon code and an inner trellis Code. The Reed Solomon code and interleaving are designed to reduce errors due to impulse Noise. By using only one coder that codes across subcarriers. The delay is small compared to the case where one trellis code is used for each subcarrier. This is due to the Viterbi decoder's Need for a certain decision depth to make a good decision. In the investigated system the, Amount of data sent in one DMT frame is enough for the Viterbi algorithm to make a decision.

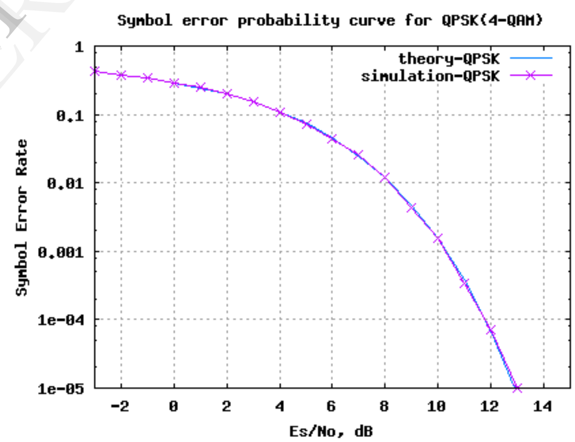
Multidimensional codes are well suited for DMT systems. By using several 2D constellations on different sub-channels it is easy to create multidimensional constellations. Where multidimensional codes allow fractional bits to be transmitted, which reduces the Granularity of the bit loading factors.

## IV. Results




Bit error curve for BPSK using OFDM

The curve shows relation between Eb/No (Bit to Noise ratio) and compute the bit error rate with BPSK. The curve shows that the signal energy is spread over a bandwidth of 16.250MHz, whereas noise is spread over bandwidth of 20MHz (-10MHz to +10MHz) [9].




Symbol Error Probability curve for an OFDM

The curve shows the relation between Es/No (Signal to Noise ratio) and we can observe that as the SNR is increased, the simulated symbol error rate intersects and then drops below the theoretical error curve.

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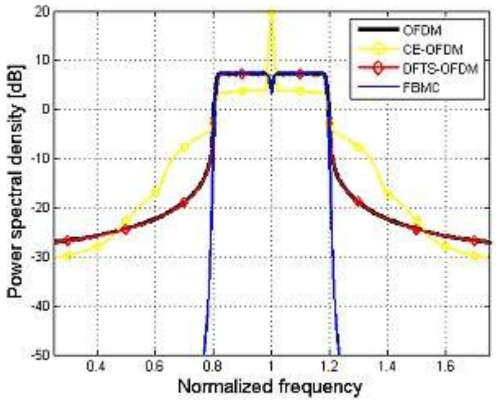
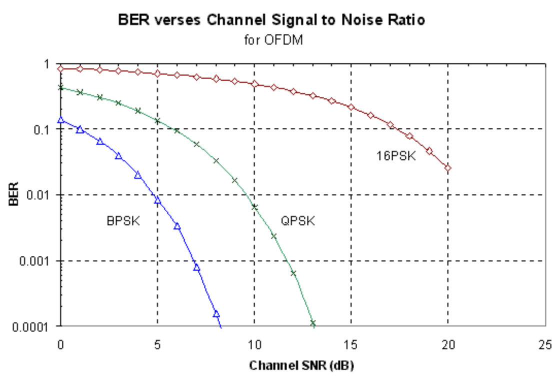


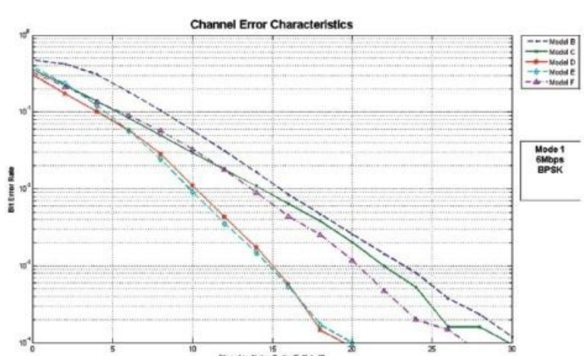

SDFT analysis of OFDM

The curve shows the spectral estimation. Stop band gains of various spectral are plotted. DFTS OFDM and CE OFDM will have wide stop band [7].



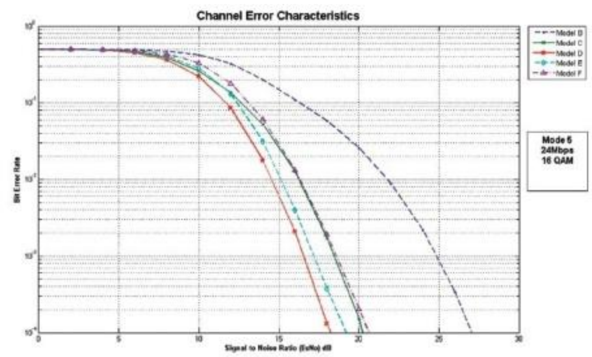

BER vs. Channel SNR curve

The curve shows the relation between BER and channel SNR. From the curve it gives an inference that as the stop band gain increases, the BER reduces and the stop band gain can have a maximum of 20 dB for a maximum of 16 QPSK.



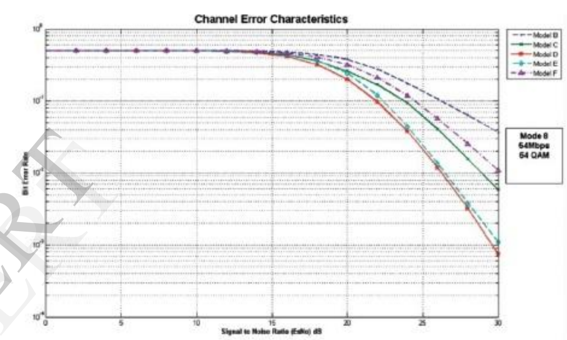

BER vs. SNR characteristics (6Mbps BPSK)

The curve shows the relation between BER and channel SNR by considering 6Mbps BPSK. We can infer that the BER is linear because of lower value of bit rate.




BER Vs SNR characteristics (24Mbps BPSK, 16 QAM)

The curve shows the relation between BER and channel SNR by considering 24Mbps BPSK and 16 bit QAM. From the curve we can infer that the BER is exponential because of higher value of bit rate (more than 24Mbps and 16 bit QAM).




BER Vs SNR characteristics (54Mbps BPSK, 64 QAM)

The curve shows the relation between BER and channel SNR by considering 54Mbps BPSK and 64 bit QAM. From the curve we can infer that the BER is exponential because of higher value of bit rate (more than 54Mbps and 64 bit QAM). The stop band gain of the OFDM system will be more than 30dB.

## V. CONCLUSION

This paper provides an inference that it is favorable to use OFDM in broadcasting applications, such as DAB and DVB. In wired system the structure of OFDM offers the possibility of efficient bit loading. By allocating a different number of bits to different sub channels, depending on their individual SNR's, efficient transmission can be achieved. OFDM often goes under the name DMT when used in wired system with bit loading. In DVB the data rate is much bigger and low bit-error rates are difficult to obtain with different PSK. A natural choice for DVB is therefore multi-amplitude schemes.

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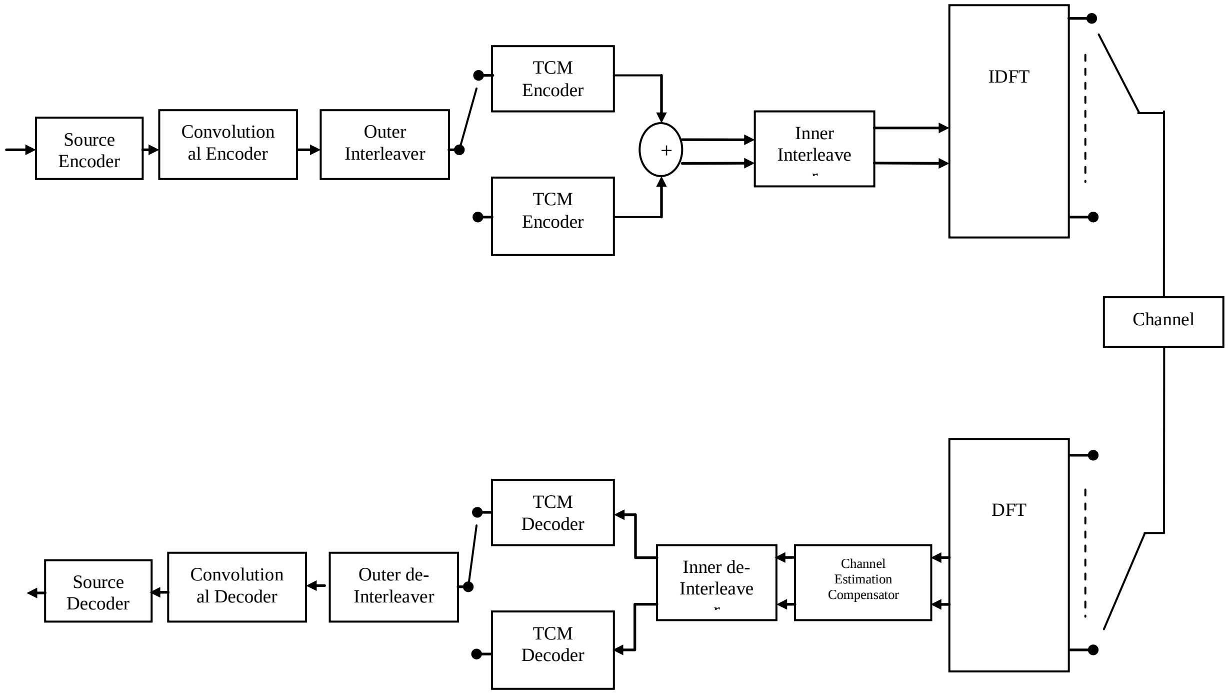


Figure 1: Overview of the system investigated by Hoher.



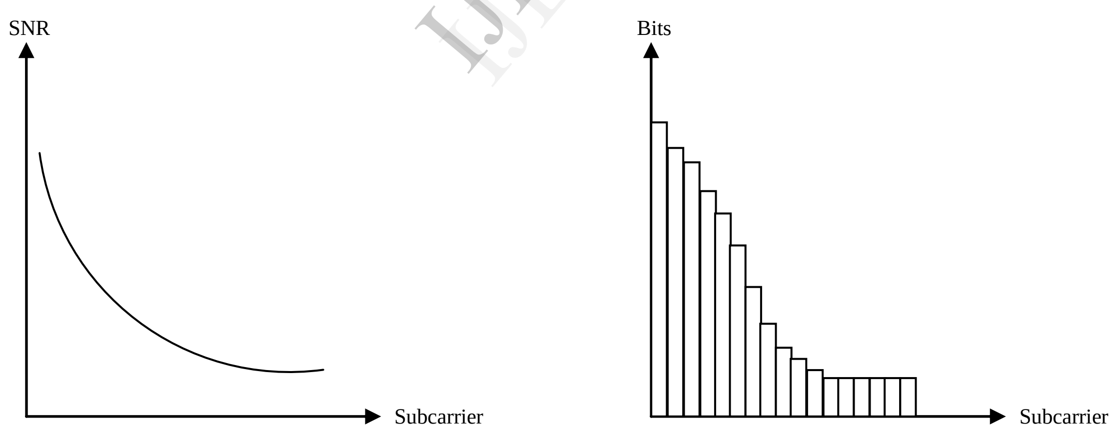


Figure 2: Channel SNR (left) and corresponding number of bits on each subcarrier (right)