3.4 ORTHOGONAL FREQUENCY DIVISION MULTIPLEXING

Orthogonal Frequency Division Multiplexing (OFDM) is a modulation scheme that is especially suited for high-data-rate transmission in delay-dispersive environments.

It converts a high-rate data stream into a number of low-rate streams that are transmitted over parallel, narrowband channels that can be easily equalized.

Principle of Orthogonal Frequency Division Multiplexing

OFDM splits a high-rate data stream into N parallel streams, which are then transmitted by modulating N distinct carriers (henceforth called subcarriers or tones). Symbol duration on each subcarrier thus becomes larger by a factor of N.

In order for the receiver to be able to separate signals carried by different subcarriers, they have to be orthogonal.

Figure 3.4.1 shows this principle in the frequency domain. Due to the rectangular shape of pulses in the time domain, the spectrum of each modulated carrier has a sin(x)/x shape.

The spectra of different modulated carriers overlap, but each carrier is in the spectral nulls of all other carriers.

Therefore, as long as the receiver does the appropriate demodulation (multiplying by $\exp(-j2\pi f n t)$ and integrating over symbol duration), the data streams of any two subcarriers will not interfere.

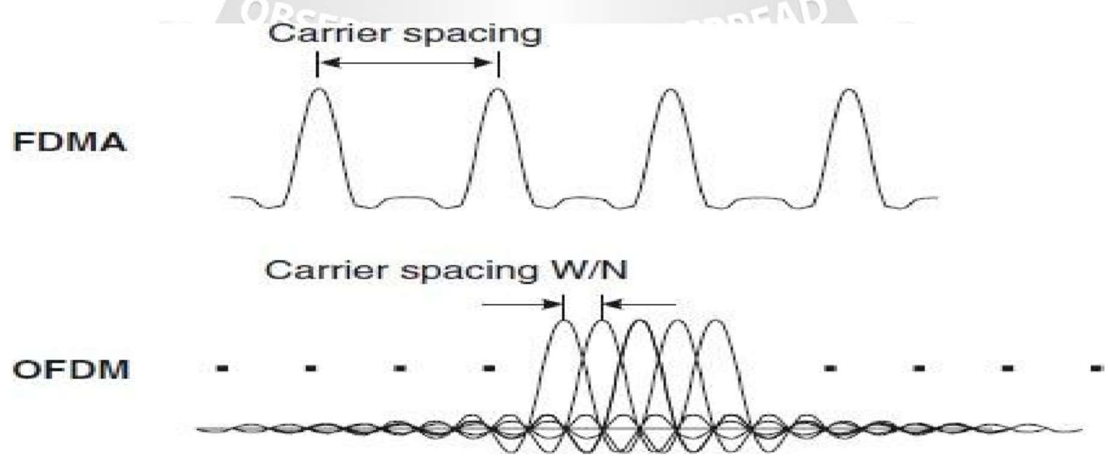

Fig3.4.1: Principle behind OFDM

[Source : "Wireless communications" by Andreas F.Molisch,Page-418]

Implementation of Transceivers

OFDM can be interpreted in two ways: One is an "analog" interpretation following from the picture of Figure 3.4.2.

First split the original data stream into N parallel data streams, each of which has a lower data rate. And have a number of local oscillators (LOs) available, each of which oscillates at a frequency fn = nW/N, where n = 0, 1, . . .,N − 1.

Each of the parallel data streams then modulates one of the carriers. This picture allows an easy understanding of the principle.

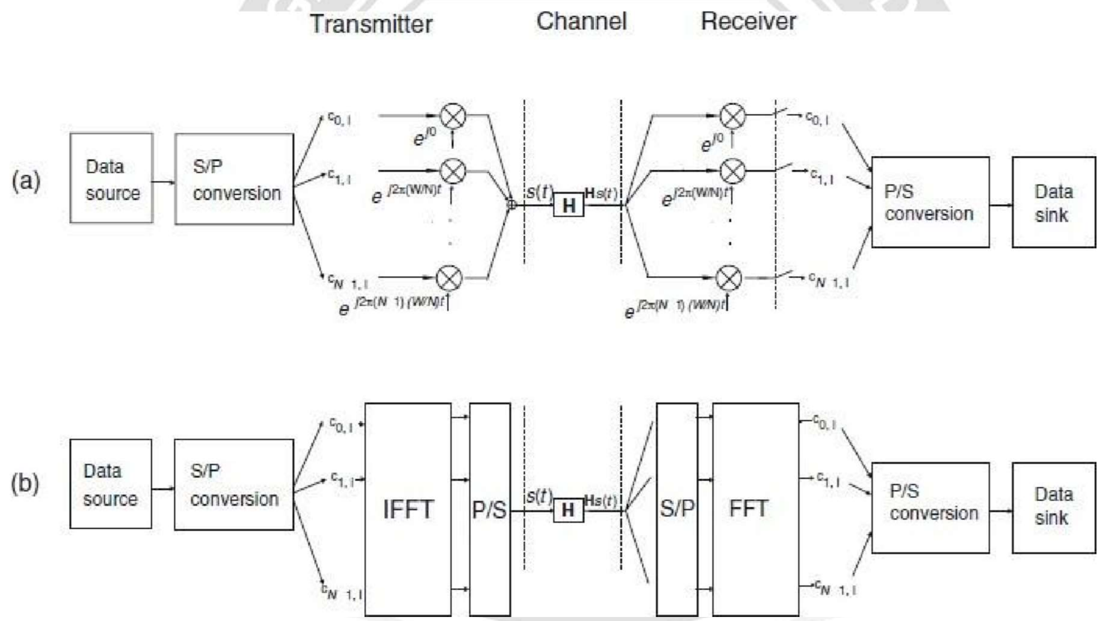

Fig 3.4.2. Transceiver structures for OFDM
[Source : "Wireless communications" by Andreas F.Molisch,Page-419]

An alternative implementation is digital . It first divides the transmit data into blocks of N symbols. Each block of data is subjected to an Inverse Fast Fourier Transformation (IFFT), and then transmitted .

This approach is much easier to implement with integrated circuits.

Let us first consider the analog interpretation. Let the complex transmit symbol at time instant i on the nth carrier be cn,i .

The transmit signal is then:

$$s(t) = \sum_{i=-\infty}^{\infty} s_i(t) = \sum_{i=-\infty}^{\infty} \sum_{n=0}^{N-1} c_{n,i} g_n(t - iT_S)$$

where the basis pulse gn(t) is a normalized, frequency-shifted rectangular pulse:

$$g_n(t) = \begin{cases} \frac{1}{\sqrt{T_S}} \exp\left(j2\pi n \frac{t}{T_S}\right) & \text{for } 0 < t < T_S \\ 0 & \text{otherwise} \end{cases}$$

Consider the signal only for i = 0, and sample it at instances tk = kTs/N:

$$s_k = s(t_k) = \frac{1}{\sqrt{T_S}} \sum_{n=0}^{N-1} c_{n,0} \exp\left(j2\pi n \frac{k}{N}\right)$$

This is the inverse Discrete Fourier Transform (DFT) of the transmit symbols.

Therefore, the transmitter can be realized by performing an Inverse Discrete Fourier Transform (IDFT) on the block of transmit symbols (the block size must equal the number of subcarriers).

In almost all practical cases, the number of samples N is chosen to be a power of 2, and the IDFT is realized as an IFFT.

Note that the input to this IFFT is made up of N samples (the symbols for the different subcarriers), and therefore the output from the IFFT also consists of N values. These N values now have to be transmitted, one after the other, as temporal samples – this is the reason why we have a P/S (Parallel to Serial) conversion directly after the IFFT.

At the receiver, sample the received signal, write a block of N samples into a vector – i.e., an S/P (Serial to Parallel) conversion – and perform an FFT on this vector.

The result is an estimate $\overline{C_n}$ of the original data C_n.