通訊系統 (II)

國立清華大學電機系暨通訊工程研究所
蔡育仁
台達館 821 室
Tel: 62210
E-mail: yrtsai@ee.nthu.edu.tw

Chapter 8
Multichannel Modulation

Capacity of AWGN Channel

Capacity of AWGN Channel

- According to **Shannon's information capacity law**, the capacity of an AWGN channel is defined by

$$C = B\log_2\left[1 + P/\left(N_0 B\right)\right] = B\log_2\left[1+\mathrm{SNR}\right] \quad \text{bits/sec}$$

  – where B is the channel bandwidth in hertz and SNR is measured at the **channel output**

- Equivalently, the capacity C in bits per channel use is

$$C' = \frac{1}{2}\log_2\left(1 + P/\sigma^2\right) = \frac{1}{2}\log_2\left(1+\mathrm{SNR}\right) \quad \text{bits/transmission}$$

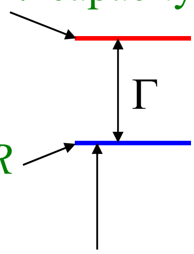
- In practice, we usually find that a **physically realizable encoding system** must transmit data at a rate R **less than** the maximum possible rate C for **reliable** reception.

Signal-to-Noise Ratio Gap

- For an **implementable** system operating at a certain low enough probability of symbol error
  - We introduce an **signal-to-noise ratio gap** (or just **gap**), denoted by Γ, which is defined by

$$\Gamma = \frac{2^{2C}-1}{2^{2R}-1} = \frac{\text{SNR}}{2^{2R}-1}$$

  - C: the capacity of the **ideal encoding system**
  - R: the capacity of the corresponding **implementable encoding system**



- It is a function of the **permissible probability of symbol error** P_e and the **encoding system** of interest
- It provides a measure of the "**efficiency**" of an encoding system
  - with respect to the **ideal transmission system**

Signal-to-Noise Ratio Gap (Cont.)

- A **small** (**large**) gap corresponds to an **efficient** (**inefficient**) encoding system
- Then, we have the attainable transmit data rate

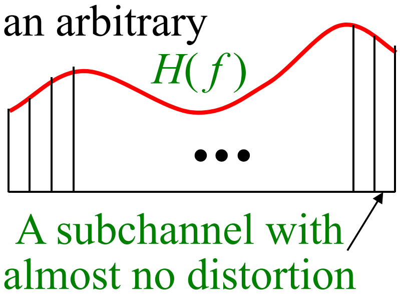
$$R = \frac{1}{2}\log_2\left(1+\text{SNR}/\Gamma\right) \quad \text{bits/transmission}$$

- For example: the desired **probability of symbol error** $P_e = 10^{-6}$
  - For an uncoded PAM or QAM system, the gap is **8.8 dB**
  - Through the use of channel coding (e.g., trellis codes), the gap may be reduced to as low as **1 dB**
- Because SNR = P/N_0B, the attainable **data rate** is defined as

$$R = \frac{1}{2}\log_2\left(1+\frac{P}{\Gamma N_0 B}\right) \quad \text{bits/transmission}$$

Continuous-Time Channel Partitioning

Continuous-Time Channel Partitioning



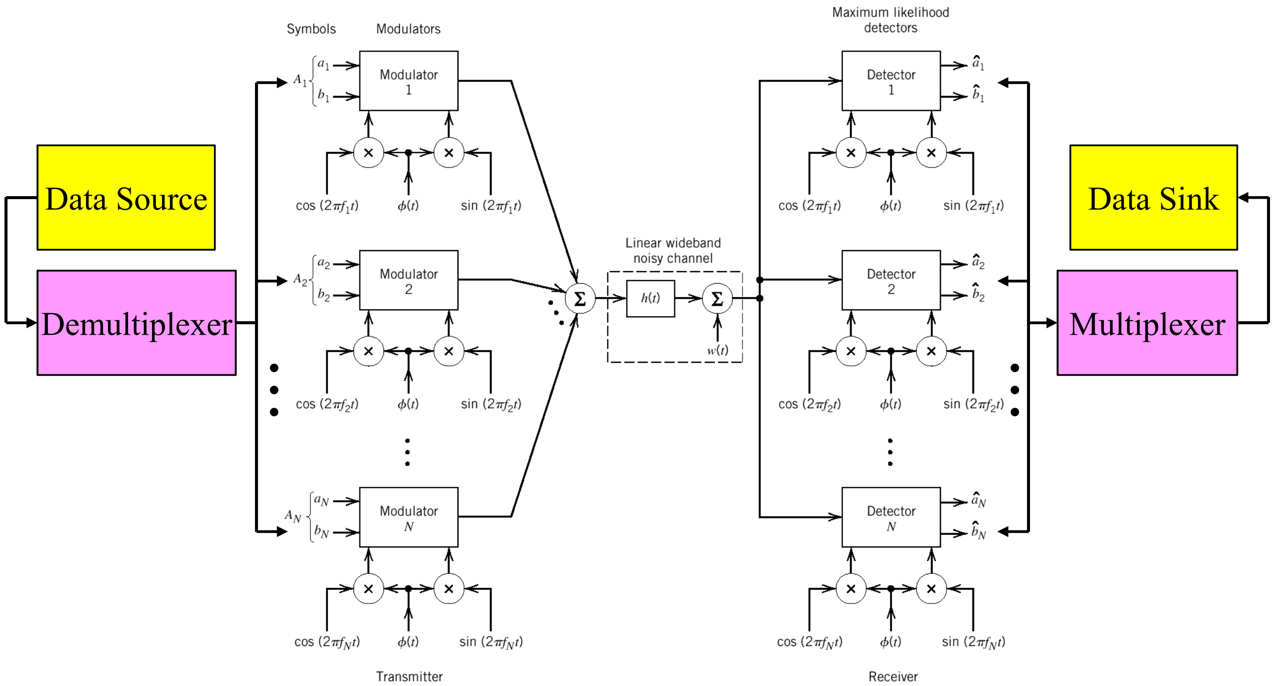
- Consider a linear **wideband channel** with an arbitrary frequency response $H(f)$.
  - The magnitude response $|H(f)|$ is approximated by a **staircase** function
  - Δf: the width of each **subchannel**
- In each step, the channel may be assumed to operate as an AWGN channel **free from inter-symbol interference**.
  - Transmitting a **wideband signal** is transformed into the transmission of a set of **narrowband orthogonal signals**
  - Each orthogonal **narrowband signal**, with **its own carrier**, is generated using a modulation technique, e.g., M-ary QAM
    - AWGN is the only transmission impairment (with a **constant response** for each subchannel)

Continuous-Time Channel Partitioning (Cont.)

- Data transmission over each subchannel can be **optimized** by invoking **Shannon's information capacity law**
  - The optimization of each subchannel is performed **independently** of all the others
- The need for **complicated equalization** of a **wideband channel** (due to the **non-constant** response) is replaced by
  - The need of **demultiplexing** and **multiplexing**
    - **Demultiplexing**: Demultiplex the incoming data stream into multiple subchannels
    - **Multiplexing**: Multiplex the demodulated data from multiple subchannels to a single data stream

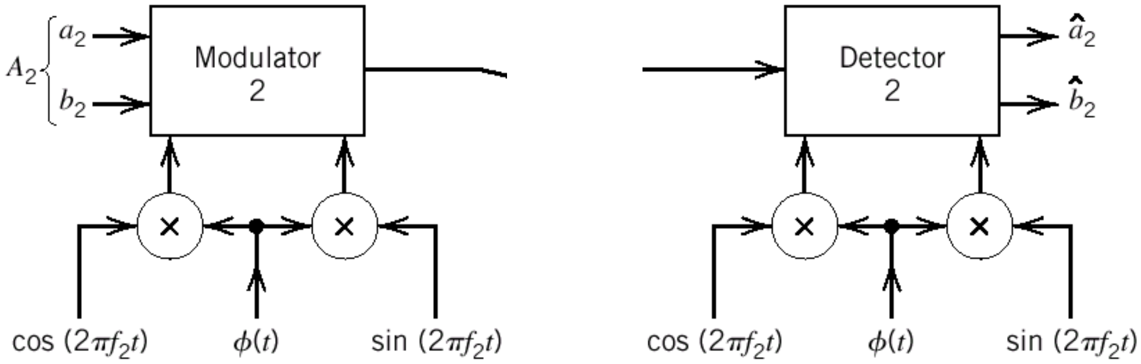
Continuous-Time Channel Partitioning (Cont.)

- A block diagram of the **multichannel** data transmission system



Continuous-Time Channel Partitioning (Cont.)

- The incoming data stream is first applied to a **demultiplexer**
  - Produce a set of N **substreams**
  - Each substream represents a sequence of **two-element** subsymbols, (a_n, b_n), $n = 1, 2, \cdots, N$, for **QAM modulation**
- The detected data of the N substreams are finally applied to a **multiplexer** to restore an output data stream



Geometric Signal-to-Noise Ratio

- In the **multichannel** transmission system, **each subchannel** is characterized by an SNR of its own.
  - However, it is highly desirable to derive a **single performance measure** of the **entire system**
- We assume that all of the subchannels are represented by one-dimensional constellations
  - The average channel capacity of the entire system is

$$R = \frac{1}{N}\sum_{n=1}^{N} R_n = \frac{1}{2N}\sum_{n=1}^{N}\log_2\left(1+\frac{P_n}{\Gamma\sigma_n^2}\right) = \frac{1}{2N}\log_2\left[\prod_{n=1}^{N}\left(1+\frac{P_n}{\Gamma\sigma_n^2}\right)\right]$$

$$= \frac{1}{2}\log_2\left[\prod_{n=1}^{N}\left(1+\frac{P_n}{\Gamma\sigma_n^2}\right)\right]^{1/N} \quad \text{bits/transmission}$$

Geometric Signal-to-Noise Ratio (Cont.)

- Let $(\text{SNR})_{\text{overall}}$ denote the **overall SNR** of the entire system.
  - Then, we may express the rate R as

$$R = \frac{1}{2}\log_2\left[1+\frac{(\text{SNR})_{\text{overall}}}{\Gamma}\right] \quad \text{bits/transmission}$$

- Accordingly, the overall SNR is

$$(\text{SNR})_{\text{overall}} = \Gamma\left[\prod_{n=1}^{N}\left(1+\frac{P_n}{\Gamma\sigma_n^2}\right)^{1/N} - 1\right]$$

- If the SNR is large enough, we have the approximation

$$(\text{SNR})_{\text{overall}} \approx \prod_{n=1}^{N}\left(\frac{P_n}{\sigma_n^2}\right)^{1/N}$$

  - It is the **geometric mean** of the SNRs of the individual subchannels and is **independent** of the gap Γ.

Loading of the Multichannel Transmission System

Power Loading

- Define the magnitude response $g_n = |H(f_n)|$, $n = 1, 2, \cdots, N$
- Assuming that the number of subchannels N is large enough
  - g_n is a **constant** over the entire bandwidth Δf
  - The average channel capacity is

$$R = \frac{1}{2N}\sum_{n=1}^{N}\log_2\left(1+\frac{g_n^2 P_n}{\Gamma\sigma_n^2}\right) \quad \text{bits/transmission}$$

  - where g_n and Γ are usually fixed; noise variance is $\Delta f\, N_0$, $\forall n$
- Goal: Optimize the overall bit rate R through a proper allocation of the **total transmit power** among the various subchannels
  - Subject to the total transmit power constraint

$$P = \sum_{n=1}^{N} P_n$$

Power Loading (Cont.)

- **Maximize** the bit rate R through an **optimal sharing** of the total transmit power P between the N subchannels
  - Subject to the total transmit power constraint P
- Through the **method of Lagrange multipliers**, the solution to the **constrained optimization problem** is

$$P_n + \frac{\Gamma\sigma_n^2}{g_n^2} = K, \quad n = 1, 2, \cdots, N$$

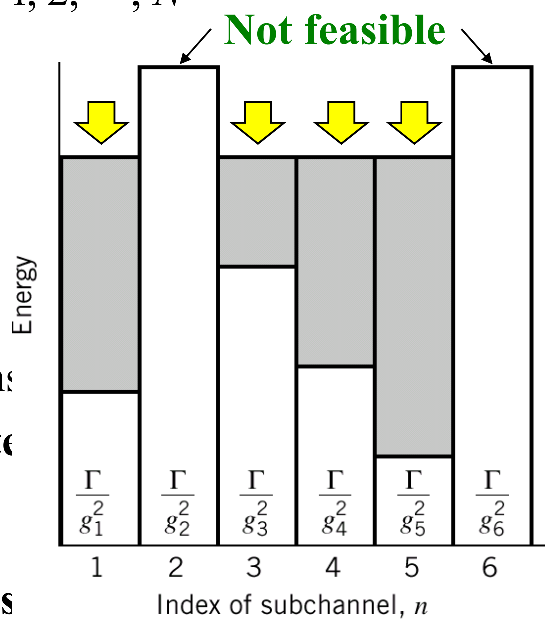
  - where K is a prescribed constant to meet the total transmit power constraint P
- The process of allocating the transmit power P to the individual subchannels is called **loading**.

Water-Filling Interpretation

- The **optimal** power allocation must satisfy the condition

$$P_n + \frac{\Gamma \sigma_n^2}{g_n^2} = K, \quad n = 1, 2, \cdots, N$$

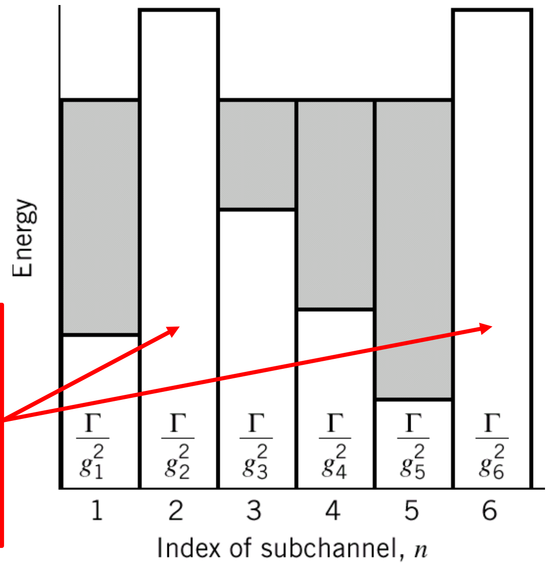
- Consider the case with $N = 6$
  - The gap Γ is assumed to be constant over all subchannels
  - The average noise power is set to $\sigma_n^2 = N_0 \Delta f = 1$
- We make the following observations
  - With $\sigma_n^2 = 1$, the sum of **allocated power** P_n and the **scaled noise power** Γ/g_n^2 is equal to a constant K for **four subchannels**



Water-Filling Interpretation (Cont.)

- The sum of power allocations to these four subchannels consumes **all the available transmit power** P.
- The remaining two subchannels have been eliminated from consideration
  - Because they would each require **negative power** to satisfy the condition (i.e., $P_n < 0$)

The channel response g_n is **too small**
$\Rightarrow$ The **scaled noise power** Γ/g_n^2 is **very large**
$\Rightarrow$ Allocating power to these two channels is **inefficient**



Water-Filling Interpretation (Cont.)

- The optimum solution for **loading** is referred to as **water-filling solution**
- This terminology follows from analogy of our optimization problem with
  - **A fixed amount of water**—standing for transmit power
  - Being **poured into a container** with a number of connected regions
  - **Each having a different depth**—standing for noise power
- In such a scenario, the water distributes itself in such a way that
  - A **constant water level** is attained across the whole container, hence the term "**water filling**"

Process of Loading

- The allocation of the fixed transmit power P among the various subchannels can be formularized as follows:
  - There are a total of $(N + 1)$ unknowns and $(N + 1)$ equations

$$P_n + \frac{\Gamma\sigma_n^2}{g_n^2} = K, \quad n = 1, 2, \cdots, N$$

$$\sum_{i=1}^{N} P_n = P$$

$$\boxed{\textbf{Unknowns: } \boldsymbol{P_n} \textbf{ and } \boldsymbol{K}}$$

$$\begin{bmatrix} 1 & 1 & \cdots & 1 & 0 \\ 1 & 0 & \cdots & 0 & -1 \\ 0 & 1 & \cdots & 0 & -1 \\ \vdots & \vdots & \ddots & \vdots & \vdots \\ 0 & 0 & \cdots & 1 & -1 \end{bmatrix} \begin{bmatrix} P_1 \\ P_2 \\ P_3 \\ \vdots \\ K \end{bmatrix} = \begin{bmatrix} P \\ -\Gamma\sigma^2/g_1^2 \\ -\Gamma\sigma^2/g_2^2 \\ \vdots \\ -\Gamma\sigma^2/g_N^2 \end{bmatrix} \Rightarrow \mathbf{Mu} = \mathbf{c}$$

Process of Loading (Cont.)

- Multiplying the inverse of **M** on both sides of the equation
  - The unknowns P_1, P_2, $\cdots$, P_N, and K can be obtained

$$\begin{bmatrix} P_1 \\ P_2 \\ P_3 \\ \vdots \\ K \end{bmatrix} = \mathbf{u} = \mathbf{M}^{-1}\mathbf{c}$$

  - K is always positive
  - It is possible for some of the P_n values to be negative
    - In such a situation, the negative P_n values are discarded

Example

- Consider a linear channel whose squared magnitude response $|H(f)|^2$ has the **piecewise linear form**
- To simplify the example, we have set the gap $\Gamma = 1$ and the noise variance $\sigma_n^2 = 1$
- Under this set of values, we have

$$P_1 + P_2 = P$$

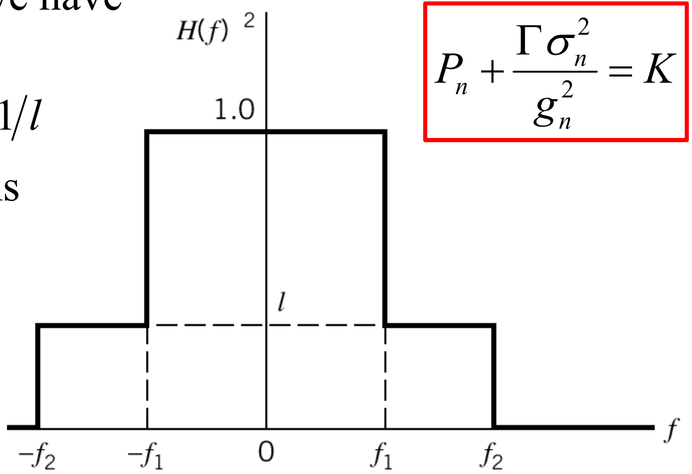
$$P_1 - K = -1; \quad P_2 - K = -1/l$$

- Solving the three equations for P_1, P_2, and K

$$P_1 = \left(P - 1 + 1/l\right)/2$$

$$P_2 = \left(P + 1 - 1/l\right)/2$$

$$K = \left(P + 1 + 1/l\right)/2$$

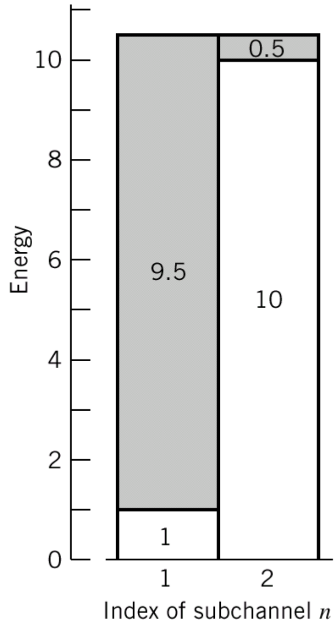


$$\boxed{P_n + \frac{\Gamma \sigma_n^2}{g_n^2} = K}$$

Example (Cont.)

- Since $0 < l < 1$, it follows that $P_1 > 0$
- But it is possible for P_2 to be **negative**
  - It happens if $l < 1/(P + 1)$
  - Correspondingly, P_1 exceeds the transmit power P ($P_1 > P$)
- Therefore, it follows that, in this example, the only acceptable solution is to have $1/(P + 1) < l < 1$.
- Let $P = 10$ and $l = 0.1$
  - The desired solution is

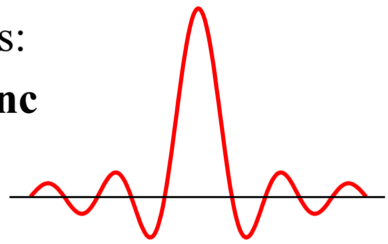
$$P_1 = 9.5$$
$$P_2 = 0.5$$
$$K = 10.5$$



Discrete Multitone

Discrete Multitone

- In the **multichannel modulation system**, **orthogonality** of the basis functions and the **channel partitioning**, is preserved.
- However, the system has two shortcomings:
  - The **passband basis functions** use a **sinc function** (rectangular function in the frequency domain)



    - Which is nonzero for an **infinite time interval**, resulting in a **non-causal** system
    - Practical considerations favor a **finite** time interval
  - For a **finite** number of subchannels N, the system is **suboptimal**
    - Optimality is assured only when N approaches **infinity** (to fit in with the frequency response $H(f)$)

Discrete Multitone (Cont.)

- We may overcome these shortcomings by using **discrete multitone** (**DMT**), which transforms a **wideband channel** into a set of N subchannels **operating in parallel**.
- The transformation of DMT is performed in **discrete time** as well as **discrete frequency**.
- The transmitter **input–output behavior** of the entire system admits a **linear matrix representation**
  - The **discrete Fourier transform** (**DFT**) can be used for implementation of DMT
  - The DFT is the result of discretizing the **Fourier transform** both in **time** and **frequency**.

Signal Generation

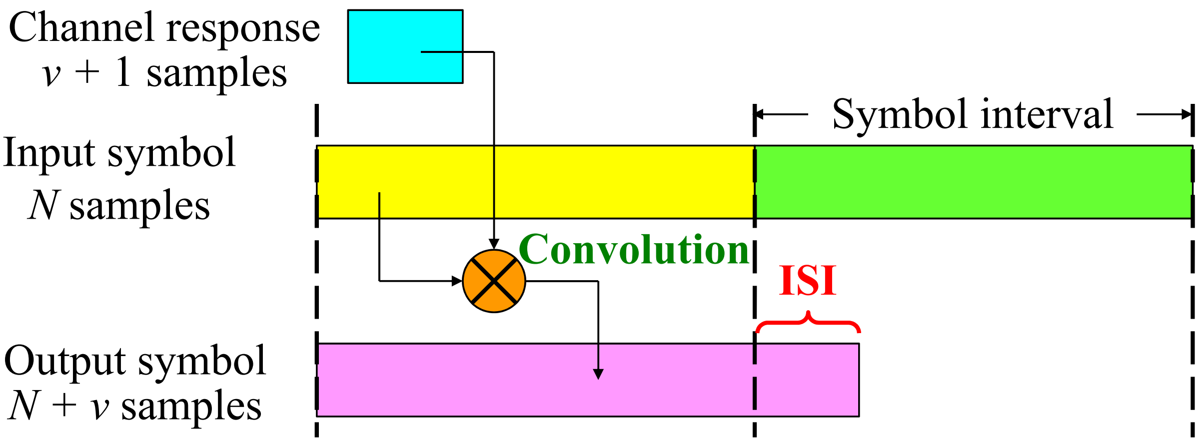
- Let the **channel impulse response** be $h(t)$, confined to a finite interval $[0, T_h]$
  - The sequence $h_0, h_1, \cdots, h_v$ denote the **baseband equivalent impulse response** of the channel sampled at the rate $1/T_s$

$$T_h = (1+v)T_s$$

- The sampling rate $1/T_s$ is chosen to be greater than **twice the highest frequency component** of interest (Nyquist rate)
  - In accordance with the **sampling theorem**
- Let $s[n] = s(nT_s)$ denote a sample of the transmitted symbol $s(t)$, $w[n] = w(nT_s)$ denote a sample of the channel noise $w(t)$, and $x[n] = x(nT_s)$ denote the sample of the channel output.

Signal Generation (Cont.)

- By **linear convolution**, the input symbol sequence $\{s[n]\}$ of length N produces an output sequence $\{x[n]\}$ of length $N + v$.
  - The extension of v **samples** is due to the **inter-symbol interference** produced by the channel (the channel impulse response $h(t)$)

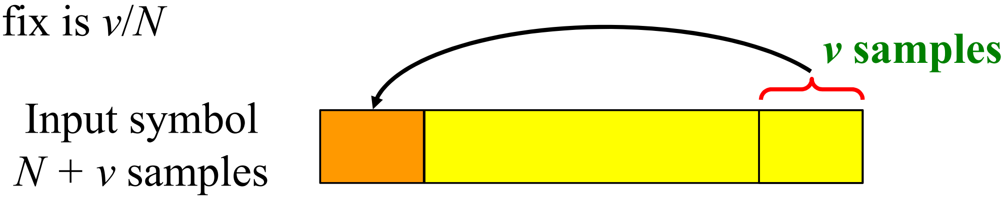


Signal Generation (Cont.)

- To overcome the effect of ISI, we create a **cyclically extended guard interval**
  - Each symbol sequence is **preceded** by a **periodic extension** of the sequence itself
  - The **periodic extension** is called as a **cyclic prefix** (**CP**)
- Specifically, the **last** v samples of the symbol sequence are repeated at the beginning of the sequence being transmitted

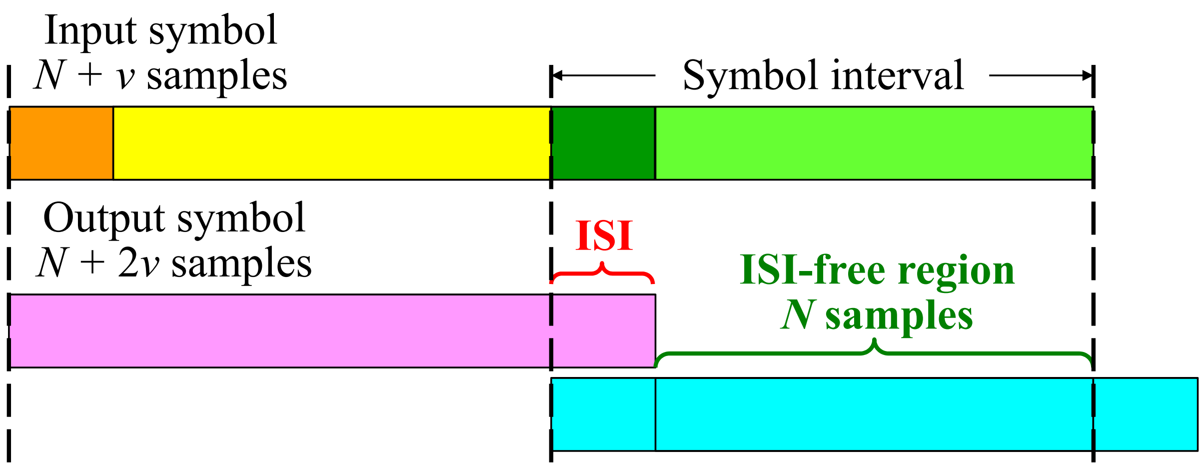
$$s[-k] = s[N-k], \quad k = 1, 2, \cdots, v$$

- The **excess bandwidth factor** due to the inclusion of the cyclic prefix is v/N



Signal Generation (Cont.)

- With insertion of the v samples of CP, we have
  - The output symbol contains $N + 2v$ samples
  - The length of ISI is still v samples
  - The ISI-free region contains N samples

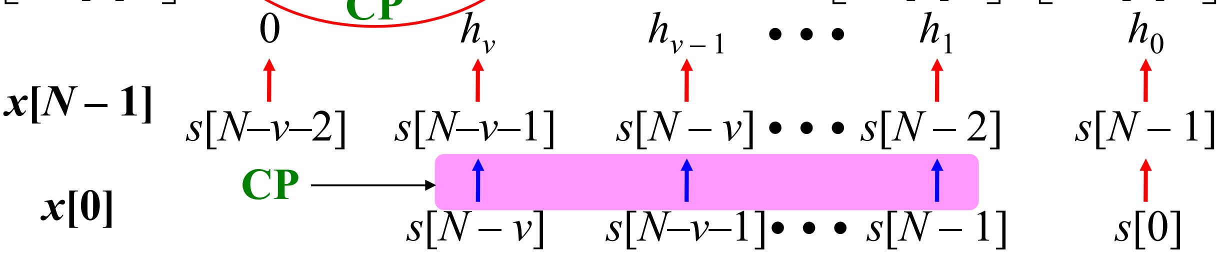


Signal Generation (Cont.)

- With insertion of the CP, the matrix description of the channel output takes the new form:

$$\begin{bmatrix} x[N-1] \\ x[N-2] \\ \vdots \\ x[v] \\ x[v-1] \\ \vdots \\ x[0] \end{bmatrix} = \begin{bmatrix} h_0 & h_1 & h_2 & \cdots & h_{v-1} & h_v & 0 & \cdots & 0 \\ 0 & h_0 & h_1 & \cdots & h_{v-2} & h_{v-1} & h_v & \cdots & 0 \\ \vdots & \vdots & \vdots & \ddots & \vdots & \vdots & \vdots & \ddots & \vdots \\ 0 & 0 & 0 & \cdots & 0 & h_0 & h_1 & \cdots & h_v \\ h_v & 0 & 0 & \cdots & 0 & 0 & h_0 & \cdots & h_{v-1} \\ \vdots & \vdots & \vdots & \ddots & \vdots & \vdots & \vdots & \ddots & \vdots \\ h_1 & h_2 & h_3 & \cdots & h_v & 0 & 0 & \cdots & h_0 \end{bmatrix} \begin{bmatrix} s[N-1] \\ s[N-2] \\ \vdots \\ s[v] \\ s[v-1] \\ \vdots \\ s[0] \end{bmatrix} + \begin{bmatrix} w[N-1] \\ w[N-2] \\ \vdots \\ w[v] \\ w[v-1] \\ \vdots \\ w[0] \end{bmatrix}$$

CP



Signal Generation (Cont.)

- The discrete-time representation of the channel output is

$$\mathbf{x} = \mathbf{H}\mathbf{s} + \mathbf{w}$$

  - **s**: the transmitted symbol vector; **w**: the channel noise vector; **x**: the received signal vector; all are N-by-1 vectors
- The channel matrix **H** is a **circulant matrix**:
  - Constructed by the **channel response** $[h_0, h_1, \cdots, h_v, 0, \cdots, 0]^T$
  - Every **row** of the matrix is obtained by **cyclically** applying a **right-shift** to the **previous row** by **one position**

$$\mathbf{H} = \begin{bmatrix} h_0 & h_1 & h_2 & \cdots & h_{v-1} & h_v & 0 & \cdots & 0 \\ 0 & h_0 & h_1 & \cdots & h_{v-2} & h_{v-1} & h_v & \cdots & 0 \\ \vdots & \vdots & \vdots & \ddots & \vdots & \vdots & \vdots & \ddots & \vdots \\ 0 & 0 & 0 & \cdots & 0 & h_0 & h_1 & \cdots & h_v \\ h_v & 0 & 0 & \cdots & 0 & 0 & h_0 & \cdots & h_{v-1} \\ \vdots & \vdots & \vdots & \ddots & \vdots & \vdots & \vdots & \ddots & \vdots \\ h_1 & h_2 & h_3 & \cdots & h_v & 0 & 0 & \cdots & h_0 \end{bmatrix}$$

Discrete Fourier Transform

- Consider an N-by-1 vector $\mathbf{x} = [x[N-1], x[N-2], \cdots, x[0]]^T$
- Let the DFT of $\mathbf{x}$ be $\mathbf{X} = [X[N-1], X[N-2], \cdots, X[0]]^T$
  - where the element is defined by

$$X[k] = \frac{1}{\sqrt{N}} \sum_{n=0}^{N-1} x[n] \exp(-j2\pi kn/N), \quad k = 0, 1, \cdots, N-1$$

  - The term $\exp(-j2\pi kn/N)$ is the **kernel** of the DFT
- Correspondingly, the IDFT (i.e., inverse DFT) of $\mathbf{X}$ is

$$x[n] = \frac{1}{\sqrt{N}} \sum_{k=0}^{N-1} X[k] \exp(j2\pi kn/N), \quad n = 0, 1, \cdots, N-1$$

Channel Representation

- For a circulant matrix, it permits the **spectral decomposition**:

$$\mathbf{H} = \mathbf{Q}^{\dagger} \mathbf{\Lambda} \mathbf{Q}$$

  - where the superscript † denotes **Hermitian transposition**
- The matrix $\mathbf{Q}$ is a **square matrix** defined in terms of the kernel of the N-point DFT as

$$\mathbf{Q} = \frac{1}{\sqrt{N}} \begin{bmatrix} \exp\left[-j\frac{2\pi}{N}(N-1)(N-1)\right] & \cdots & \exp\left[-j\frac{2\pi}{N}2(N-1)\right] & \exp\left[-j\frac{2\pi}{N}(N-1)\right] & 1 \\ \exp\left[-j\frac{2\pi}{N}(N-1)(N-2)\right] & \cdots & \exp\left[-j\frac{2\pi}{N}2(N-2)\right] & \exp\left[-j\frac{2\pi}{N}(N-2)\right] & 1 \\ \vdots & \ddots & \vdots & \vdots & \vdots \\ \exp\left[-j2\pi(N-1)/N\right] & \cdots & \exp\left[-j2\pi\times 2/N\right] & \exp\left[-j2\pi/N\right] & 1 \\ 1 & \cdots & 1 & 1 & 1 \end{bmatrix}$$

$$q_{kl} = \frac{1}{\sqrt{N}} \exp\left[-j\frac{2\pi}{N}kl\right], \quad k, l \in \{0, 1, \cdots, N-1\}$$

Channel Representation (Cont.)

- The matrix **Q** is an **orthonormal matrix** or **unitary matrix**

$$\mathbf{Q}^{\dagger}\mathbf{Q} = \mathbf{I}$$

- The matrix Λ is a **diagonal matrix** that contains the N **DFT values** of the sequence $h_0, h_1, \cdots, h_v$
- Denoting these transform values by $\lambda_{N-1}, \cdots, \lambda_1, \lambda_0$
  - $[\lambda_{N-1}, \cdots, \lambda_1, \lambda_0]^{\mathrm{T}} = \mathrm{DFT}([h_0, h_1, \cdots, h_v, 0, \cdots, 0]^{\mathrm{T}})$

$$\mathbf{\Lambda} = \mathrm{diag}\left(\lambda_{N-1}, \cdots, \lambda_1, \lambda_0\right) = \begin{bmatrix} \lambda_{N-1} & 0 & \cdots & 0 \\ 0 & \lambda_{N-2} & \cdots & 0 \\ \vdots & \vdots & \ddots & \vdots \\ 0 & 0 & \cdots & \lambda_0 \end{bmatrix}$$

Frequency–Domain Representation

- Define the **time-domain** transmit signal vector as

$$\underset{\text{Time domain transmitted samples}}{} \longrightarrow \mathbf{s} = \mathbf{Q}^{\dagger}\mathbf{S} \longleftarrow \underset{\text{Frequency domain parallel signals}}{}$$

Time domain transmitted samples → $\mathbf{s} = \mathbf{Q}^{\dagger}\mathbf{S}$ ← **Frequency domain parallel signals**

  - where **S** is the **frequency-domain** transmit signal vector
  - Each element of the N-by-1 vector **S** is a **complex-valued** point in a two-dimensional QAM signal constellation.
- Given the channel output vector **x**, we define its corresponding **frequency-domain** representation as

$$\mathbf{X} = \mathbf{Q}\mathbf{x}$$

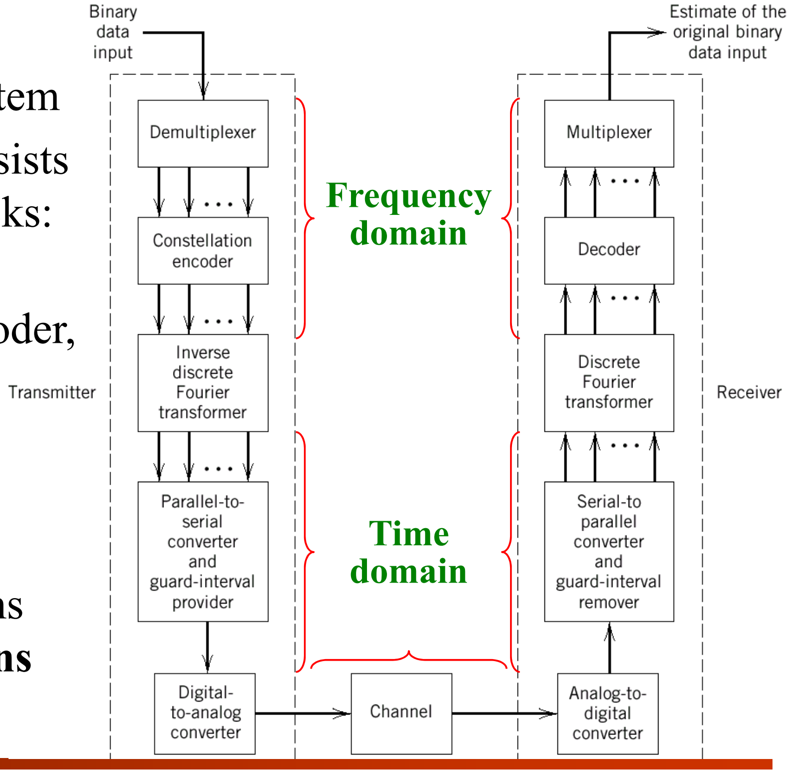
- Then, we may rewrite **X** in the equivalent form

$$\mathbf{X} = \mathbf{Q}\mathbf{x} = \mathbf{Q}\left(\mathbf{H}\mathbf{s} + \mathbf{w}\right) = \mathbf{Q}\left(\mathbf{Q}^{\dagger}\mathbf{\Lambda}\mathbf{Q}\mathbf{Q}^{\dagger}\mathbf{S} + \mathbf{w}\right) = \mathbf{\Lambda}\mathbf{S} + \mathbf{W}$$

  - where $\mathbf{W} = \mathbf{Q}\mathbf{w}$ is the **frequency-domain** noise vector
- That is, $X_k = \lambda_k S_k + W_k, k = 0, 1, \cdots, N-1$ ← Parallel subchannels

DFT-Based DMT System

- Block diagram of the DFT-based DMT system
- The **transmitter** consists of the functional blocks:
  - Demultiplexer, Constellation encoder, IDFT, Parallel-to-serial converter, Digital-to-analog converter (DAC)
- The **receiver** performs the **inverse operations** of the transmitter



DFT-Based DMT System (Cont.)

- **Demultiplexer**: converts the incoming **serial data stream** into **parallel form** (N constellations in the **frequency-domain**)
- **Constellation encoder**: maps the parallel data into **all multibit subchannels** with each subchannel being represented by a **QAM signal constellation**
  - **Bit allocation** among the subchannels is performed in accordance with a **loading** algorithm.
- **IDFT**: transforms the **frequency-domain** parallel data at the constellation encoder output into parallel **time-domain** data
  - For efficient implementation of the IDFT using the **fast Fourier transform** (**FFT**) algorithm, we need to choose $N = 2^k$, where k is a positive integer

DFT-Based DMT System (Cont.)

- **Parallel-to-serial converter**: converts the **parallel** time-domain samples into **serial** form (discrete time samples)
  - Guard intervals stuffed with **cyclic prefixes** are inserted into the serial data on a periodic basis before conversion into analog form.
- **Digital-to-analog converter (DAC)**: converts the digital data into analog form ready for transmission over the channel

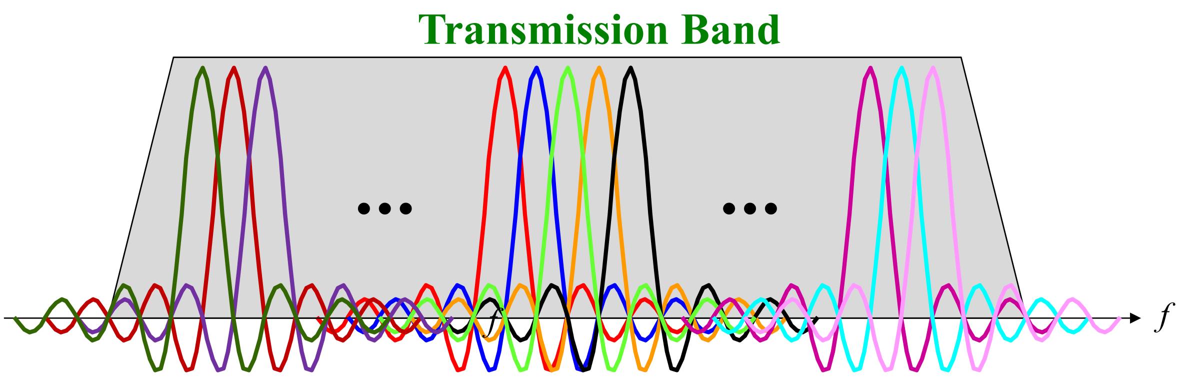
Orthogonal Frequency Division Multiplexing (OFDM)

OFDM Concept

- **Orthogonal frequency division multiplexing** (**OFDM**) is a form of **multi-carrier modulation**.
  - OFDM is particularly well suited for **high data-rate transmission** over **delay-dispersive channels**.
- Specifically, a large number of closely spaced **orthogonal subcarriers (tones)** is used to support the transmission.
  - The incoming data stream is divided into a number of **low data-rate sub-streams**, one for each subcarrier
- In addition, two other changes have to be made for OFDM:
  - In the transmitter, an **upconverter** is included after the DAC to **translate** the signal to the **transmission band**
  - In the receiver, a **downconverter** is included before the ADC to **translate** the signal to the **baseband**

OFDM Concept (Cont.)

- **Orthogonal frequency division multiplexing (OFDM)** is a promising technique because of its
  - High bandwidth efficiency and
  - Resistance to multipath fading
- **Orthogonality** is maintained among the subcarriers
- **Narrowband** transmission for each digitally modulated signal

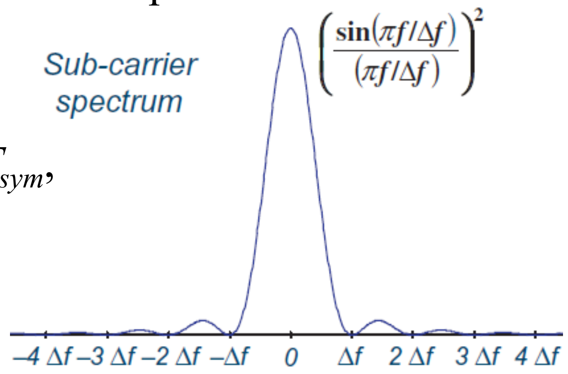


OFDM Concept (Cont.)

- However, the following characteristics distinguish OFDM from a straightforward multi-carrier extension:
  - The use of a typically **very large number** of relatively narrowband subcarriers (e.g., several hundred subcarriers)
  - Simple **rectangular pulse shaping** is used

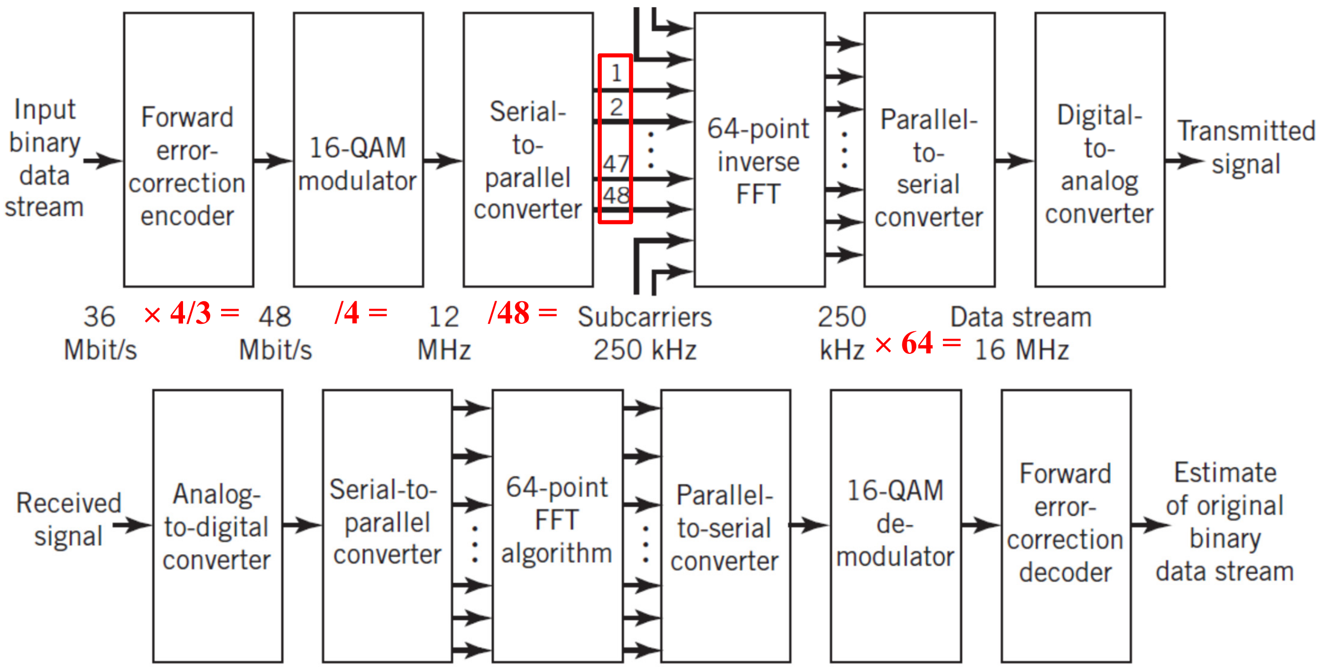
  $\Rightarrow$ A sinc-square-shaped per-subcarrier spectrum
  - **Tight frequency-domain packing** of the subcarriers

  $\Rightarrow$ A subcarrier spacing $\Delta f = 1/T_{sym}$, T_{sym} is the per-subcarrier modulation-symbol duration



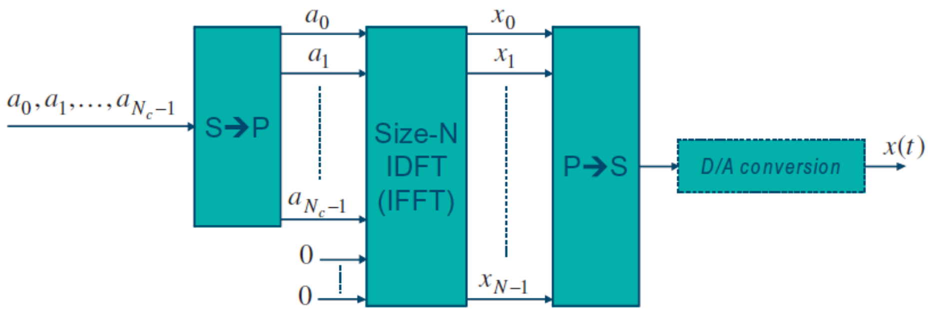
OFDM Transmitter/Receiver

- Block diagrams of transmitter/receiver for a 36 Mbits/s system

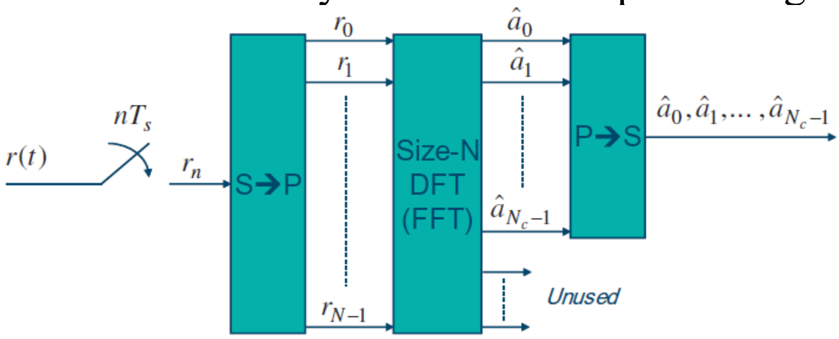


OFDM Implementation

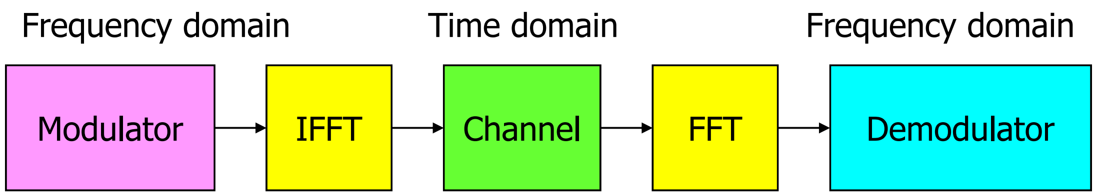
- OFDM **modulation** by means of **IFFT** processing

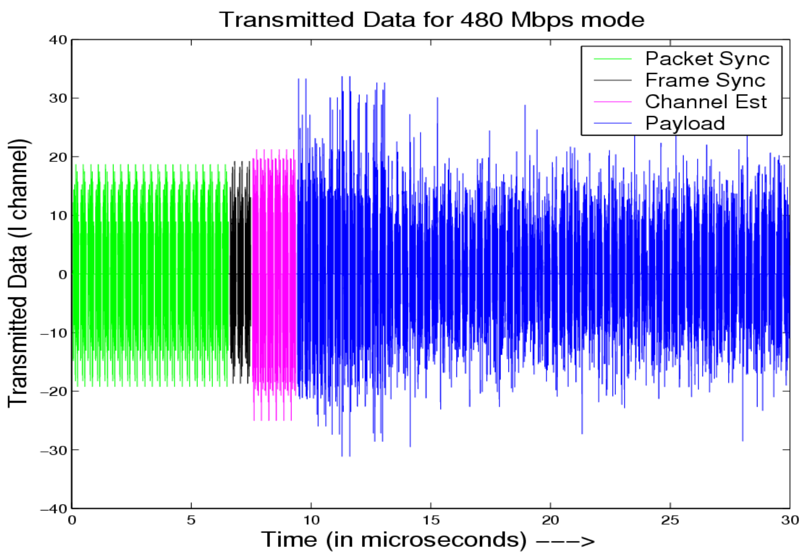


- OFDM **demodulation** by means of **FFT** processing



OFDM Transmission





Homework

- **You must give detailed derivations or explanations, otherwise you get no points.**
- Communication Systems, Simon Haykin **(4th Ed.)**
- 6.43;