ING'S College **IONDON**



6CCS3COS Communication Systems: Chapter 1

Osvaldo Simeone

Evolution to 4G



[Neocific] ³

Work in progress on a 5G standard...



[ETSI]

4

Work in progress on a 5G standard...



[Qualcomm '17]⁵

Work in progress on a 5G standard...



Work in progress on a 5G standard...



 $[Qualcomm '17]^7$





Illustration 1.1 Mobile penetration, or number of cell phones per person, in selected countries as of June 2015. Six countries have a penetration of at least 100%, meaning there are more phones than people.

[Brinton and Chiang '17]

Quiz: When was the word "bit" invented?

1948

Reprinted with corrections from The Bell System Technical Journal, Vol. 27, pp. 379-423, 623-656, July, October

A Mathematical Theory of Communication

By C. E. SHANNON



The choice of a logarithmic base corresponds to the choice of a unit for measuring information. If the base 2 is used the resulting units may be called binary digits, or more briefly *bits*, a word suggested by J. W. Tukey. A device with two stable positions, such as a relay or a flip-flop circuit, can store one bit of information. N such devices can store N bits, since the total number of possible states is 2^N and $\log_2 2^N = N$.

Before 1948



[B. Rimoldi, Principles of Digital Communications, Cambridge University Press]

Pre-cellular mobile systems:

- 1948: Mobile Telephone Service from Bell Telephone
- Analog voice
- FDMA
- \$330 + per-call cost



https://smartphones.gadgethacks.com/news/from-backpack-transceiver-smartphone-visual-history-mobile-phone-0127134/

Pre-cellular mobile systems:

- 1956: Mobile System A (MTA) from Ericcson
- Analog voice
- FDMA
- Weighs as much as 300 iPhones!



https://smartphones.gadgethacks.com/news/from-backpack-transceiver-smartphone-visual-history-mobile-phone-0127134/

Pre-cellular mobile systems:

- 1964: Improved Mobile Telephone Service (IMTS)
- Analog voice
- FDMA



https://smartphones.gadgethacks.com/news/from-backpack-transceiver-smartphone-visual-history-mobile-phone-0127134/

Enter cellular networks





Illustration 1.6 This is a diagram of a cellular network. Each cell is a hexagon, with multiple mobile stations (MSs) and base stations (BSs). The shading of a cell indicates the frequency band that the cell is using. No two neighboring cells have the same shading, and so use different frequency bands to prevent interference.

[Brinton and Chiang ⁴7]



[Qualcomm '1⁷]







[MathWorks]

- Overview
 - 1. One-shot digital communications: Fundamentals
 - 2. One-shot digital communications: Passband Systems
 - 3. Stream digital communications

Main reference

- J. Cioffi, Lecture notes, Stanford Univ., Chapters 1, 2, 3

Additional reference

- B. Rimoldi, Principles of Digital Communications, Cambridge University Press

Expectations of inclusive behaviour

The Department of Informatics is committed to providing an inclusive learning and working environment.

Staff and students are expected to behave respectfully to one another – during lectures, outside of lectures and when communicating online or through email.

We won't tolerate inappropriate or demeaning comments related to gender, gender identity and expression, sexual orientation, disability, physical appearance, race, religion, age, or any other personal characteristic.

If you witness or experience any behaviour you are concerned about, please speak to someone about it. This could be one of your lecturers, your personal tutor, a programme administrator, the Informatics equality & diversity lead (Elizabeth Black), or any other member of staff you feel comfortable talking to.

The College also has a range of different support and reporting procedures that you might find helpful: <u>kcl.ac.uk/harassment</u>

ING'S College **IONDON**



6CCS3COS Communication Systems: Chapter 2

Osvaldo Simeone

- Overview
 - 1. One-shot digital communications: Fundamentals
 - 2. One-shot digital communications: Passband Systems
 - 3. Stream digital communications

Main reference

- J. Cioffi, Lecture notes, Stanford Univ., Chapters 1, 2, 3

Additional reference

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What Is One-Shot Digital Digital Communication?

- The transmitter (TX) selects one message from a finite set (e.g., a bit string) and sends a corresponding signal (or "waveform") through the communication channel via coding and modulation.
- The receiver (RX) decides the message sent by observing the channel output via demodulation and decoding.
- Optimum detection minimizes the probability of an erroneous receiver decision.



What Does the Transmitter Do?



- **Encoder**: Message $m \in \{0, ..., M = 2^b 1\}$ (*b* bits) \rightarrow symbol (real vector) x_m in the signal space
- **Modulator**: Symbol $x_m \rightarrow$ transmitted waveform (analog and continuous-time) $x_m(t)$ of duration *T* seconds and bandwidth *B* Hz

What Does the Transmitter Do?



- **Symbol rate:** 1/T messages per second
- Number of bits per symbol: $b = \log_2 M$
- **Data rate:** R = b/T bits per second

What Does the Transmitter Do?

- A modulator uses a bandwidth *B* to encode *R* bit/s
- Spectral efficiency

 $\eta = \frac{R}{B}$ (bit/s/Hz)

- Ex.: If R=10 Mb/s and B=10 MHz, the spectral efficiency is $\eta = 1$.
- The spectral efficiency ranges from values smaller than 1 (wireless channels with low SNR) to values larger than 10 (e.g., wireless broadcasting).

What Is the Channel?

- Adds noise due to interference and receiver circuitry.
- Attenuates the signal.

What is the Channel?

- Adds noise due to interference and receiver circuitry.
- Attenuates the signal.
- Distorts the signal due to multiple propagation paths.



http://www.wica.intec.ugent.be/research/propagation/physical-radio-channel-models

What is the Channel?



- Frequency selectivity: Non-uniform frequency response
- Time selectivity: Time-variability

What Does the Receiver Do?



- **Demodulator**: Channel output waveform (analog and continuoustime) $y(t) \rightarrow$ channel output vector **y** in the signal space
- Decoder: Channel output vector y → estimate m̂ of the message m (b bits)

How Much Power to Transmit?

- A basic question in the design of transmitter and receiver is: how much power should the transmitter use in order to ensure a given reliability (probability of correct detection)?
- This is known as **link budget**.
- The starting point is that the receiver needs to be guaranteed a certain signal-to-noise ratio (SNR)

$$SNR = \frac{\text{received power (mW)}}{\text{noise power (mW)}} = \frac{P_r \text{ (mW)}}{\sigma^2 \text{ (mW)}}$$

despite the attenuation caused by the channel.

How Much Power to Transmit?

• If the channel gain is $L \leq 1$, then the received power is

$$P_r = L \times P_x$$

where P_x is the transmitted power.

• Therefore, the required transmitted power is given as

$$SNR = \frac{P_r}{\sigma^2} = \frac{L \times P_x}{\sigma^2}$$
$$\Rightarrow P_x = \sigma^2 \frac{SNR}{L}$$

How Much Power to Transmit?

• Ex.: If SNR = 100, $L=10^{-8}$, and $\sigma^2 = 10^{-11}$ mW, then the required power is

$$P_x = 10^{-11} \times 100 \times 10^8 = 0.1 \text{ mW}$$

- These are realistic values and demonstrate the need to deal with large numbers (e.g., 10⁸) and very small numbers (e.g., 10⁻¹¹).
- To this end, communication engineers work with decibel (dB) measures.
• The decibel (symbol: dB) is a unit of measurement used to express a ratio of powers on a logarithmic scale:

power ratio (dB) = 10 log₁₀(power ratio)

dB	Power ratio									
100	10 000 000 000									
90	1 000 000 000									
80	100 000 000									
70	10 000 000									
60	1 000 000									
50	100 000									
40	10 000									
30	1 000									
20	100									
10	10									
6	3	.981 ≈ 4								
3	1	.995 ≈ 2								
1	1	.259								
0	1									

0	1
-1	0.794
-3	0 .501 ≈ ½
-6	0 .251 ≈ ¹ ⁄ ₄
-10	0.1
-20	0 .01
-30	0 .001
-40	0 .000 1
-50	0 .000 01
-60	0 .000 001
-70	0 .000 000 1
-80	0 .000 000 01
-90	0 .000 000 001
-100	0 .000 000 000 1

- Link budget calculations can be carried out fully in the logarithmic scale.
- Key properties: $\log(a \times b) = \log(a) + \log(b)$ $\log(a/b) = \log(a) - \log(b)$

(note that log(b^c)=clog(b) and hence the second property follows from the
first)

- Link budget calculations can be carried out fully in the logarithmic scale.
- Key properties: $\log(a \times b) = \log(a) + \log(b)$ $\log(a/b) = \log(a) - \log(b)$

(note that log(b^c)=clog(b) and hence the second property follows from the
first)

• To this end, powers are measured in dBm:

Power (dBm) =
$$10 \log_{10}(\frac{\text{power}(\text{mW})}{1 \text{ mW}})$$

and we have

$$P_{\chi} (dBm) = \sigma^2 (dBm) + SNR (dB) - L (dB)$$

- Ex.: For the same example above, working in the logarithmic scale, we have:
- If SNR = 100 = 20 dB,

 $L = 10^{-8} = -80 \text{ dB},$ and $\sigma^2 = 10^{-11} \text{ mW} = -110 \text{ dBm}$, then the required power is

$$P_{\chi} (dBm) = \sigma^2 (dBm) + SNR (dB) - L (dB)$$

= -110 + 20 + 80 = -10 dBm

• In wireless channels, the attenuation is given as a function of the distance *d* (m) between transmitter and receiver as

$$L (dB) = L_1 (dB) - \gamma \ 10 \ \log_{10}(d)$$

where L_1 is the attenuation at 1 m and γ is the path loss exponent (between 2 and 5 depending on the environment).

• In wireless channels, the attenuation is given as a function of the distance *d* (m) between transmitter and receiver as

$$L (dB) = L_1 (dB) - \gamma \ 10 \ \log_{10}(d)$$

where L_1 is the attenuation at 1 m and γ is the path loss exponent (between 2 and 5 depending on the environment).

• In wired channels, e.g., fiber optics cables, the attenuation can be typically written as

$$L (dB) = L_0 \left(\frac{dB}{km}\right) \times d$$

where L_o is the attenuation in dB per km (e.g., -0.1 dB/km for fiber).

• Ex.: Indoor WLAN (e.g., Wi-Fi)

Assuming $\gamma = 2$ and $L_1 = -50$ dB, and $\sigma^2 = -110$ dBm, what is the maximum distance at which can we ensure an SNR of 10 dB if the transmitted power is 0 dBm?

• Ex.: Indoor WLAN (e.g., Wi-Fi)

Assuming $\gamma = 2$ and $L_1 = -50$ dB, and $\sigma^2 = -110$ dBm, what is the maximum distance at which can we ensure an SNR of 10 dB if the transmitted power is 0 dBm?

In order to ensure SNR = 10 dB, the channel gain must be at least

$$P_{\chi} (dBm) = \sigma^{2} (dBm) + SNR (dB) - L (dB)$$

$$0 = -110 + 10 - L (dB)$$

$$\rightarrow L (dB) = -100 dB$$

and hence the maximum distance satisfies $-50 - 20 \log_{10}(d) = -100 \log_{10}(d) = 50/20 = 2.5 \rightarrow d = 10^{2.5} = 316 \text{ m}$

• Ex.: Fiber optics cable

Assuming L_0 =-1.5 dB/km, and σ^2 = -110 dBm, what is the maximum distance at which can we ensure an SNR of 10 dB if the transmitted power is 0 dBm?

In order to ensure SNR = 10 dB, the channel gain must be at least L(dB) = -100 dB, and hence the minimum distance is

-1.5d = -100

$$\rightarrow$$
 d = 100/1.5=67 km

How to Encode and Decode Information?



- Why is the performance dependent on the SNR?
- How much information can be transferred for a given SNR?
- How is the information encoded and decoded?

• Encoder:

message
$$m \in \{0, ..., M-1\} \rightarrow$$
 symbol (real vector) $\boldsymbol{x}_m = \begin{bmatrix} x_{m,1} \\ x_{m,2} \\ \vdots \\ x_{m,N} \end{bmatrix}$

- The signal space is \mathbb{R}^N
- The set of all symbols x_m , m=1,...,M, is the signal **constellation**
- We expect that a constellation with symbols that are further apart will lead to a smaller probability of error.

Examples: N=1



M=2 (b=1) Binary Phase Shift Keying (BPSK) *M*=4 (b=2) 4-Pulse Amplitude Modulation (PAM)

Examples: *N*=2



M=8 (b=3) 8-Phase Shift Keying (PSK)

M=16 (b=4) 16-Quadrature Amplitude Modulation (QAM)

https://en.wikipedia.org/wiki/Constellation_diagram

Examples: N=2

000_100	001_100	011_100	010 100 110 100	111_100	101_100	100_100	10	0 101 (103	102	98	99 •	97	96	• 32	• 33	• 35	•34	• 38	• 39	• 37	• 36
000 101	001_101	011_101	010 101 110 101	111 101	101_101	100 101	12	40 125 human human	127 •	126	122	123	121	120	• 40 • 56	• •1	• 43 • 59	• 42 	• 40 • 62	• 47 • 63	• 43 • 61	• 44 • 60
·	•	·	+5	•	•	•	316	6 117 8 85 6	119 	118 •	114 	115 •	113 •	112 .	• 48	• 49 • 17	• 51	• 50	• 54	• 55	• 53	• 52
000 111	001 111	011 111 •	010111 + 3	•	101 111 •	100 111	82	83 C	95 •	94 •	90	91 •	19 •	88	• 24	• 25	• 27	• 26	• 30	• 31	• 29	• 28
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000_000	001_000	011_000	010,000 110,000	111,000	101_000	100,000	236	• 237 •	239 8 1000	238	234	235 •	233 •	232 •	• 168	 169 	• 171	1 70	•174	• 175	• 173	●172
•	•	•	• _7	•	•	•	22 8 11	● 229 ● III III III III	231	230	226	227 •	225	224	• 160	• 161	● 163	●i62	• 166	• 167	• 165	• 164

M=256 (b=8) 256-QAM

• Average energy of a constellation: If each message m is selected with probability p_m , the average energy of a constellation $\{x_m\}$ is defined as

$$E_x = \mathbf{E}[||\mathbf{x}||^2] = \sum_{m=0}^{M-1} p_m ||\mathbf{x}_m||^2$$

where

$$||\mathbf{x}_{m}||^{2} = \sum_{n=1}^{N} |x_{m,n}|^{2}$$

is the energy of the *m*th symbol

• Average power:

$$P_x = \frac{E_x}{T}$$

• When M=2 (or b=1), the energy per bit is

$$E_b = E_x$$

• More generally, we have

$$E_x = bE_b$$

- *E_b* is an important metric, since it relates cost (energy) to performance (bit rate).
- Note: E_x is also often referred to as the symbol energy (and denoted as E_s).

What Happens When the Energy Increases/ Decreases?

• Quiz: BPSK energy

If N=1, M=2, the symbols are equally likely and the average energy is Eb<1, what are valid constellations?

A) -1,+1
B)
$$-\sqrt{E_{b}}$$
, $+\sqrt{E_{b}}$
C) 0 , $+\sqrt{2E_{b}}$
D) $-2\sqrt{E_{b}}$, $+2\sqrt{E_{b}}$

What Happens When the Energy Increases/ Decreases?

• Quiz: BPSK energy

Considering that the distance between constellation points determines the probability of error, which constellation would you choose?

A)
$$-\sqrt{E_b}$$
, $+\sqrt{E_b}$,
B) 0 , $+\sqrt{2E_b}$

What Happens When the Energy Increases/ Decreases?

• If all symbols have the same probability, we have the condition

$$\frac{1}{M} \sum_{m=0}^{M-1} ||\boldsymbol{x}_m||^2 = bE_b$$

• Examples:



What Does the Transmitter Do?



- **Encoder**: Message $m \in \{0, ..., M = 2^b 1\}$ (*b* bits) \rightarrow symbol (real vector) x_m in the signal space
- **Modulator**: Symbol $x_m \rightarrow$ transmitted waveform (analog and continuous-time) $x_m(t)$ of duration *T* seconds and bandwidth *B* Hz

Why Modulation?

- To adapt transmitted signals to the channel
- Note that the probability of error depends on the distance between symbols in the constellation *after* the distortion caused by the channel (to be discussed).

How Does the Modulation Adapt to the Channel?

Example:

• Channel frequency response: passband in the interval 100 Hz and 200 Hz with 150 Hz having the largest gain



- b=1, or equivalently M=2 (binary transmission)
- Encoder: $0 \rightarrow x_0 = -1$ and $1 \rightarrow x_1 = 1$
- Modulator: $x_0 \rightarrow x_0(t) = -1$ and $x_1 \rightarrow x_1(t) = 1$ for all t
- Quiz: Is this a good choice for the modulator?

How Does the Modulation Adapt to the Channel?

Example:

• Channel frequency response: passband in the interval 100 Hz and 200 Hz with 150 Hz having the largest gain



• Quiz: Which waveforms are suitable to be chosen by the modulator for this channel?

A) $x_0(t) = -\cos(2^*pi^*t)$ and $x_1(t) = \cos(2^*pi^*t)$ B) $x_0(t) = -\cos(300^*pi^*t)$ and $x_1(t) = \cos(300^*pi^*t)$ C) $x_0(t) = -\cos(100^*pi^*t)$ and $x_1(t) = \cos(100^*pi^*t)$ D) $x_0(t) = -\cos(240^*pi^*t)$ and $x_1(t) = \cos(240^*pi^*t)$

• Modulator:

symbol $x_m \rightarrow$ (analog and continuous-time) waveform $x_m(t)$ of duration *T* seconds (symbol period)

- The corresponding set of modulated waveforms {*x_m(t)*}, *m = 0, ..., M 1*, is a **signal set**.
- To map symbol to waveform, we need to associate each axis of the signal space R^N with a basis function in a set of **orthonormal basis functions**.



Some Math

- Inner product or correlation:
 - between real vectors

$$\langle \mathbf{v}, \mathbf{u} \rangle = \sum_{n=1}^{N} v_n u_n$$

- between real functions

$$\langle v(t), u(t) \rangle = \int_{-\infty}^{\infty} v(t)u(t)dt$$

Some Math

- Squared Euclidean norm or energy:
 - for a real vector

$$||\mathbf{v}||^2 = \langle \mathbf{v}, \mathbf{v} \rangle = \sum_{n=1}^N v_n^2$$

- for a real function

$$||v(t)||^{2} = \langle v(t), v(t) \rangle = \int_{-\infty}^{\infty} (v(t))^{2} dt$$

Some Math

Orthogonality:

- for real vectors: vectors ${\bf v}$ and ${\bf u}$ are orthogonal if

$$\langle \mathbf{v}, \mathbf{u}
angle = 0$$

- for real functions: functions v(t) and u(t) are orthogonal if

$$\langle v(t), u(t) \rangle = 0$$

Preliminaries:

- In \mathbb{R}^N the vectors defining the coordinate axes are $\mathbf{e}_m, m=1,...N$.
- \mathbf{e}_m contains all zeros except for a 1 in the *m*th position, i.e., $\mathbf{e}_{m,m}=1$ and $\mathbf{e}_{m,n}=1$ for *m* different from *n*.



Preliminaries:

• These vectors constitute an orthonormal basis:

$$\langle \mathbf{e}_m, \mathbf{e}_n \rangle = 0$$
 for $m \neq n$ (orthogonal)
 $\langle \mathbf{e}_m, \mathbf{e}_m \rangle = ||\mathbf{e}_m||^2 = 1$ (normalized)



Preliminaries:

• In \mathbb{R}^N we can write

$$\mathbf{x}_m = \sum_{n=1}^N x_{m,n} \mathbf{e}_n$$



A set of N function {φ_m(t)}, m = 1, ..., N is an N-dimensional orthonormal basis (and the functions are orthonormal basis functions) if it satisfies the conditions

$$\langle \varphi_m(t), \varphi_n(t) \rangle = 0$$
 for $m \neq n$ (orthogonal)
 $\langle \varphi_m(t), \varphi_m(t) \rangle = ||\varphi_m(t)||^2 = 1$ (normalized)



• Quiz: Which ones of the sets below is an orthonormal basis?



• General block diagram of a modulator:



$$x_m(t) = \sum_{n=1}^N x_{m,n} \varphi_n(t)$$

• Example: N=1



Constellation (E_b =1)



normalized basis function



• Example: N=2



constellation (E_b =1)



orthonormal basis functions




• Example: N=2



constellation (E_b =1)



orthonormal basis functions



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Some Math

Invariance of the inner product:

Theorem 1.1.1 (Invariance of the Inner Product) If there exists a set of basis functions $\varphi_n(t)$, n = 1, ..., N for some N such that $u(t) = \sum_{n=1}^{N} u_n \varphi_n(t)$ and $v(t) = \sum_{n=1}^{N} v_n \varphi_n(t)$ then

$$\langle u(t), v(t) \rangle = \langle u, v \rangle .$$
 (1.8)

where

$$u \stackrel{\Delta}{=} \begin{bmatrix} u_1 \\ \vdots \\ u_N \end{bmatrix} \quad and \quad v \stackrel{\Delta}{=} \begin{bmatrix} v_1 \\ \vdots \\ v_N \end{bmatrix} \quad . \tag{1.9}$$

Correlation in signal space = correlation between modulated signals

Is the Energy Modified by Modulation?

- Energy of the *m*th constellation point: $E_x = ||\mathbf{x}_m||^2$
- Modulation does not change the energy:

 $||x_m(t)||^2 = \langle x_m(t), x_m(t) \rangle$ = $\langle \mathbf{x}_m, \mathbf{x}_m \rangle$ (by invariance of inner product) = E_m

- The signals in the previous examples do not have good spectral properties for typical band-limited channels.
- Bandlimited channel: Channel that passes only a limited range of frequencies
- In fact, consider

$$\varphi_1(t) = \frac{1}{\sqrt{T}} \operatorname{rect}\left(\frac{t}{T}\right)$$



The Fourier transform is \sqrt{T} sinc(*fT*), which has an infinite bandwidth.

• The Fourier transform represents any (energy-limited) signal as the sum of an infinite sum of complex sinusoidal signals

$$e^{j 2\pi ft} = \cos(2\pi ft) + j \sin(2\pi ft)$$

with different amplitudes and phases



$$x(t) = \int_{-\infty}^{+\infty} X(f) \, e^{j \, 2\pi f t} df$$

- We will be mostly interested in the energy spectrum $G_x(f) = |X(f)|^2$, which describes how the energy of a signal is distributed in the frequency domain.
- Rayleigh theorem

$$E_x = \int_{-\infty}^{+\infty} |x(t)|^2 dt = \int_{-\infty}^{+\infty} G_x(f) df$$

• Computation of the Fourier transform

$$X(f) = \int_{-\infty}^{+\infty} x(t) e^{-j 2\pi f t} dt = \mathcal{F}\{x(t)\}$$

• The Fourier transform of non-energy limited signals can also be defined by using the impulse function.

• Example:

$$x(t) = e^{j(2\pi f_c t + \theta)}$$



Recall: Impulse function



$$X(f) = e^{j\theta} \delta(f - f_c)$$

(check by substituting in $x(t) = \int_{-\infty}^{+\infty} X(f) e^{j 2\pi f t} df$ and using the sifting property)

$$\delta(t - t_0) = 0 \text{ for all } t \neq t_0$$
$$\int_{-\infty}^{+\infty} f(t) \,\delta(t - t_0) \, dt = f(t_0)$$
(sifting property)

• Example:

$$x(t) = \cos(2\pi f_c t + \theta) = \frac{1}{2} e^{j(2\pi f_c t + \theta)} + \frac{1}{2} e^{-j(2\pi f_c t + \theta)}$$

$$\frac{1}{2} e^{-j\theta} \qquad X(f) = \frac{1}{2} e^{j\theta} \delta(f - f_c) + \frac{1}{2} e^{-j\theta} \delta(f + f_c)$$

$$X(f) = \frac{1}{2} e^{j\theta} \delta(f - f_c) + \frac{1}{2} e^{-j\theta} \delta(f + f_c)$$

Note that

$$\sin(2\pi f_c t) = \cos\left(2\pi f_c t - \frac{\pi}{2}\right)$$

• Example:

$$x(t) = \operatorname{rect}\left(\frac{t}{T}\right)$$



$$X(f) = \int_{-T/2}^{T/2} e^{-j \, 2\pi f t} dt = T \operatorname{sinc}(f \, T)$$



• The energy spectrum is typically measured in dB/Hz:

 $G_{\mathrm{x}}(f)|_{\mathrm{dB}} = 10\log_{10}G_{\mathrm{x}}(f)$



[Banelli et al '14]

• Example:



• Mnemonic trick: Energy of a sinc = squared value at peak × time of first zero

Properties:

1) Hermitian symmetry: If x(t) is real

 $X(f) = X^*(-f)$ or equivalently:

$$\begin{cases} |X(f)| = |X(-f)| \\ \arg(X(f)) = -\arg(X(-f)) \end{cases}$$
 Hermitian symmetry

Ex.: Rectangular function, sinc

2) Frequency translation:

$$\mathcal{F}\left\{x(t)e^{j\,2\pi f_c t}\right\} = X(f - f_c)$$

2') Corollary: $\mathcal{F}\{x(t)\cos(2\pi f_c t)\} = \frac{1}{2}(X(f - f_c) + X(f + f_c))$

Proof (corollary):
$$\cos(2\pi f_c t) = \frac{1}{2} (e^{j 2\pi f_c t} + e^{-j 2\pi f_c t})$$

Example:

 $x(t) = \operatorname{sinc}(t)$





- We now provide examples of practically used waveforms over bandlimited channels.
- Baseband signals with *N*=1:

$$\varphi_1(t) = \frac{1}{\sqrt{T}}\operatorname{sinc}\left(\frac{t}{T}\right)$$

Bandwidth limited to B=1/(2T).



- We now provide examples of practically used waveforms over bandlimited channels.
- Passband signals with N = 1 and carrier frequency $f_c \gg \frac{1}{T}$:

$$\varphi_1(t) = \sqrt{\frac{2}{T}} \operatorname{sinc}\left(\frac{t}{T}\right) \cos(2\pi f_c t)$$

Bandwidth limited to B=1/T around carrier frequency f_c .



• Passband signals with N=2 and carrier frequency $f_c \gg \frac{1}{T}$

$$\varphi_1(t) = \sqrt{\frac{2}{T}} \operatorname{sinc}\left(\frac{t}{T}\right) \cos(2\pi f_c t)$$
$$\varphi_2(t) = \sqrt{\frac{2}{T}} \operatorname{sinc}\left(\frac{t}{T}\right) \sin(2\pi f_c t)$$

Bandwidth limited to B=1/T around carrier frequency f_c .



 G_{φ_1}

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Bar Note: The cos and sin carriers are orthogonal and said to be in quadrature. More generally, we can choose any two carriers whose phases differ by $\pi/2$.

$$(f) = G_{\varphi_2}(f)$$

$$(f) = G_{\varphi}(f)$$

$$(f) = G_{\varphi}($$

• Passband waveforms with any *N*:

Orthogonal Frequency Division Multiplexing (OFDM), used in 4G, Wi-Fi,...

$$\varphi_m(t) = \sqrt{\frac{2}{T}} \operatorname{rect}\left(\frac{t}{T}\right) \cos\left(2\pi (f_c + \frac{m-1}{T} - \frac{N}{4T})t\right) \text{ for } m = 1, ..., N/2 \text{ and}$$
$$\varphi_m(t) = \sqrt{\frac{2}{T}} \operatorname{rect}\left(\frac{t}{T}\right) \sin\left(2\pi (f_c + \frac{m-(\frac{N}{2}+1)}{T} - \frac{N}{4T})t\right) \text{ for } m = N/2 + 1, ..., N$$

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Note: Carriers spaced by 1/T in the frequency domain are orthogonal.
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$$f$$
Remark: OFDM also includes a cyclic prefix that allows it to simplify the operation over frequency selective channels

• Passband waveforms with any *N*:

filtered-OFDM (f-OFDM), candidate waveform for 5G

 $\varphi_m(t) = \varphi_m^{\text{OFDM}}(t) * h(t)$



http://www.sharetechnote.com/html/5G/5G_Phy_Candidate_fOFDM.html

• It can be proven that the maximum number of orthogonal dimensions that can be accommodated in a bandwidth *B* over a time *T* is

$$N=2BT$$

- It follows that, except for $\varphi_1(t) = \sqrt{\frac{2}{T}} \operatorname{sinc}\left(\frac{t}{T}\right) \cos(2\pi f_c t)$ (*N*=1), the presented orthonormal basis functions use the available bandwidth and time in the most efficient way.
- In practice, it is generally preferable to choose orthonormal basis functions with N < 2BT that are easier to realize (the sharp spectral transitions of the sinc can only be approximated.)

• Quiz:

Which of the following functions are suitable for transmission on a passband channel that has a bandwidth of 1 MHz around a center frequency of 1 GHz?

A)
$$f_{4}(t) = \sqrt{2 \times 10^{7}} \operatorname{sinc}(t \times 10^{7}) \operatorname{cos}(2\pi \cdot 10^{9}t)$$

B) $f_{4}(t) = \sqrt{2 \times 10^{6}} \operatorname{sinc}(t \times 10^{6}) \operatorname{cos}(2\pi \cdot (1.001) \times 10^{9}t)$
C) $f_{4}(t) = \sqrt{2 \times 10^{6}} \operatorname{sinc}(t \times 10^{6}) \operatorname{cos}(2\pi \cdot 10^{9}t)$

How Effectively Is Bandwidth Used by a Transmitter?

• Recall that the spectral efficiency is defined as

 $\eta = \frac{R}{B}$ (bit/s/Hz)

• Ex.: The modulator uses the waveform $\varphi_1(t)=1/\sqrt{T} \operatorname{sinc}(t/T)$ with a bandwidth of 1 MHz and the encoder uses either BPSK or 4-PAM. What is the spectral efficiency?

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- We have

$$\eta = \frac{R}{B} = \frac{b/T}{1/2T} = 2b$$

and hence $\eta = 2$ for BPSK and $\eta = 4$ for 4-PAM with this choice of waveform.

• Ex.: Assume that we would like to transmit a file of size 2 Gbits. The modulator uses the baseband waveform $\varphi_1(t) = \frac{1}{\sqrt{T}} \operatorname{sinc} \left(\frac{t}{T}\right)$ with a bandwidth of 1 MHz and the encoder uses either BPSK or 4-PAM. How long does it take to complete transmission of the file (assuming that there are no errors)?

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The symbol period is T = $1/(2B) = 0.5 \times 10^{-6}$ s. The time need to download the file is given as

$$\frac{\text{file size}}{R} = \frac{2 \times 10^9}{b/T} = \frac{10^3}{b}$$

Hence, it takes 1000 seconds to download with BSK and 500 seconds with 4- PAM.

• Ex.: Assume that we would like to transmit a file of size 2 Gbits. The modulator uses the passband waveform $\varphi_1(t) = \sqrt{\frac{2}{T}} \operatorname{sinc}\left(\frac{t}{T}\right) \cos(2\pi f_c t)$ with a bandwidth of 1 MHz and the encoder uses either BPSK or 4-PAM. How long does it take to complete transmission of the file (assuming that there are no errors)?

• Ex.: Assume that we would like to transmit a file of size 2 Gbits. The modulator uses the passband waveform $\varphi_1(t) = \sqrt{\frac{2}{T}} \operatorname{sinc}\left(\frac{t}{T}\right) \cos(2\pi f_c t)$ with a bandwidth of 1 MHz and the encoder uses either BPSK or 4-PAM. How long does it take to complete transmission of the file (assuming that there are no errors)?

The symbol period is $T = 1/B = 10^{-6}$ s. The time need to download the file is given as

$$\frac{\text{file size}}{R} = \frac{2 \times 10^9}{b/T} = \frac{2 \times 10^3}{b}$$

Hence, it takes 2000 seconds to download with BSK and 1000 seconds with 4- PAM.

What Does the Receiver Do?



- **Demodulator**: (continuous-time analog) channel output signal $y(t) \rightarrow$ channel output vector **y** in the signal space
- **Detector:** channel output vector $\mathbf{y} \rightarrow \text{estimate } \hat{m}$ of the message m

What Does the Receiver Do?

• Useful observations:

1) vector projection: for a vector

$$x = \begin{bmatrix} x_1 \\ x_2 \\ \vdots \\ x_N \end{bmatrix} = \sum_{n=1}^N x_n e_n$$

the *n*th component can be obtained via correlation with the basis vector e_n (projection of *x* into e_n)

$$\langle x, e_n \rangle = x_n$$
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$$\langle x, e_n \rangle = x_n$$

2) modulating e_n yields $\varphi_n(t)$

3) **invariance of inner product:** correlation is equal in the signal space and on the modulated signals

• Assume that the received signal y(t) is noiseless and hence

$$y(t) = x(t) = \sum_{n=1}^{N} x_n \varphi_n(t)$$

• Therefore, recovering each constellation coordinate x_n is equivalent to projecting y(t) into $\varphi_n(t)$

• Assume that the received signal y(t) is noiseless and hence

$$y(t) = x(t) = \sum_{n=1}^{N} x_n \varphi_n(t)$$

• The demodulator can recover the symbol **x** as follows:



$$\langle y(t), \varphi_n(t) \rangle = x_n$$

• Proof:

$$\langle x(t), \varphi_n(t) \rangle = \langle x, e_n \rangle = x_n$$



Signal set:

$$X_{0}(t) = \sqrt{E_{b}} q_{1}(t) = \frac{1}{\frac{T}{2}} \frac{1}{\frac{T}{2}} t$$

$$X_{1}(t) = \sqrt{E_{b}} q_{2}(t) = \frac{1}{\frac{T}{2}} \frac{1}{\frac{T}{2}} t$$

· What is the ophincal receiver un the obsence of noise?



& Check that the receiver works when m=0.

$$\begin{split} \mathcal{J} &= 0, \quad \forall |t| = \sqrt{E_b} \varphi_1(t) \\ \implies &< \forall |t|, \quad \varphi_1(t) > = \sqrt{E_b} \quad \| \varphi_1(t) \|^2 = \sqrt{E_b} \\ &< \forall |t|, \quad \varphi_2(t) > = \sqrt{E_b} \quad < \varphi_1(t), \quad \varphi_2(t) > = 0 \end{split}$$



• In time domain:

$$y(t) = x(t) * h(t) = \int_{-\infty}^{+\infty} x(\lambda)h(t - \lambda)d\lambda$$
$$= \int_{-\infty}^{+\infty} h(\lambda)x(t - \lambda)d\lambda$$
$$= h(t) * x(t)$$
convolution

• In frequency domain

Y(f) = H(f) X(f)

and hence

$$G_{\mathcal{Y}}(f) = |H(f)|^2 G_{\mathcal{X}}(f)$$

output energy spectral density input energy spectral density

Examples:

a) $h(t) = \delta(t)$ H(f) = 1 Y(f) = X(f)

y(t) = x(t)



c) Same low-pass filter as above, but with input $x(t) = cos(4\pi t)$



→ Y(f) = H(f) X(f) = 0y(t) = 0

d) Repeat for $x(t) = \cos(\pi t)$



Y(f) = X(f)y(t) = x(t)

- Instead of using correlations, the demodulator can use the matched filter-based architecture.
- A filter with impulse response

 $h(t) = \varphi(-t)$

is said to be matched to the waveform $\varphi(t)$.

• Why "matched"?

$$\varphi(t)*h(t)=\varphi(t)*\varphi(-t)=\int_{-\infty}^{+\infty}\varphi(\lambda)\varphi(\lambda+t)d\lambda$$

 Hence, at *t*=0 (symbol peak), the matched filter recovers the energy of the pulse

$$\varphi(t) * h(t)|_{t=0} = \langle \varphi(t), \varphi(t) \rangle = ||\varphi(t)||^2$$



matched filter demodulator

• Proof of equivalence between the two demodulators

$$< x(t), \varphi_n(t) >= \int_{-\infty}^{+\infty} x(\lambda)\varphi_n(\lambda)d\lambda$$
$$= \int_{-\infty}^{+\infty} x(\lambda)\varphi_n(\lambda+t)d\lambda|_{t=0}$$
$$= x(t) * \varphi_n(-t)|_{t=0}$$

• Quiz: Evaluate the matched filters for the waveforms below.



How to Detect on a Noisy Channel?

• Consider first the vector (or discrete) channel model



- The channel is described by the conditional probability distribution $p(\mathbf{y}|\mathbf{x})$

How to Detect on a Noisy Channel?

• Consider first the vector (or discrete) channel model



- The channel is described by the conditional probability distribution $p(\mathbf{y}|\mathbf{x})$
- Ex.: Gaussian channel, *N*=1

$$y = x + z$$
 with $z \sim \mathcal{N}\left(0, \frac{N_0}{2}\right)$

• Gaussian distribution



- Probability density function: $f(x) = \frac{1}{\sqrt{2\pi\sigma}} e^{-\frac{(x-m)^2}{2\sigma^2}}$
- Computing probabilities:





• Tail probabilities:



$$\int_{a}^{+\infty} f(x) dx = Q\left(\frac{a-m}{\sigma}\right)$$

$$\int_{-\infty}^{\beta} f(x) dx = Q\left(\frac{m-\beta}{\sigma}\right)$$

• Q function



[B. Rimoldi, Principles of Digital Communications, Cambridge University Press]¹⁹⁸

• Q function



[B. Rimoldi, Principles of Digital Communications, Cambridge University Press¹⁹⁹

- Sensor measuring temperature
- If temperature > threshold γ -> alarm;
 if temperature < threshold γ -> no alarm



- Define m=0/1 (no fire/fire) and y=temperature
- Conditional probability density function (pdf)



 x_1 = average temperature when there is fire

• Conditional probability of error:

$$\Pr[\widehat{m} \neq m | m = 1]$$

• Conditional probability of error:



*x*_o = average temperature when there is no fire

- We are interested in the overall probability of error.
- By the law of total probability

$$P_e = \Pr[\widehat{m} \neq m]$$

= $\Pr[m = 1]\Pr[\widehat{m} \neq m|m = 1] + \Pr[m = 0]\Pr[\widehat{m} \neq m|m = 0]$



- Minimizing P_e is equivalent to minimizing the colored area
- Intuitive arguments show that the optimal threshold should be such that the two curves take the same value.



• This yields the optimal rule

$$p(y|m=0)\Pr[m=0] \ge p(y|m=1)\Pr[m=1]$$

• Optimal rule

$$p(y|m = 0)\Pr[m = 0] \ge p(y|m = 1)\Pr[m = 1]$$

a priori probabilities

• Optimal rule



a priori (from prior knowledge) × likelihood (from data)

• Optimal rule



log-prior (from prior knowledge) + log-likelihood (from data)

- The derived optimal rule is known as Maximum a Posterior (MAP).
- Posterior probability of the message given the received signal (Bayes theorem)

$$p(m|y) = \frac{p(y|m)p(m)}{p(y)}$$

• The optimal rule can be then equivalently expressed as

$$\hat{m} = \underset{m \in \{0,1\}}{\operatorname{argmax}} p(m|y)$$
$$= \underset{m \in \{0,1\}}{\operatorname{argmax}} p(y|m)p(m)$$

which justifies the name MAP.

 $\frac{E \times ample}{Pr[m=1] = 0.05}$ $-p(y|m=1) = \mathcal{N}(y|10,3)$ $p(y|m=0) = \mathcal{N}(y|1,1)$



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· If y = 4, should the alarm go off? First, note that $\log N(X|\mu,\sigma^2) = \log \left(\frac{1}{\sqrt{2\pi\sigma^2}}\right) - \frac{(X-\mu)^2}{2\sigma^2}$ Therefore, the MAP rule can be written as $log(0.05) + log(\frac{1}{\sqrt{2\pi^2}}) - \frac{(4-10)^2}{2}$ $\geq \log(0.95) + \log(\frac{1}{\sqrt{201}}) - (\frac{4-1}{201})^2$ \Rightarrow 7.01 < -5.47 \Rightarrow no fire $(\hat{m}=0)$ · Repeat for y=5

How to Detect on a Noisy Channel?



• Problem: Choose a decision rule $y \rightarrow \hat{m}$ that minimizes the probability of error

How to Detect on a Noisy Channel?

• Using again the law of total probability, we have

$$\min_{\hat{m}(\mathbf{y})} P_e = \min_{\hat{m}(\mathbf{y})} \Pr[\hat{m} \neq m]$$

=
$$\min_{\hat{m}(\mathbf{y})} \int p(\mathbf{y}) \Pr[m \neq \hat{m}(\mathbf{y}) | \mathbf{y}] d\mathbf{y} \text{ (by the law of total prob.)}$$

=
$$\int p(\mathbf{y}) \left(\min_{\hat{m}(\mathbf{y})} \Pr[m \neq \hat{m}(\mathbf{y}) | \mathbf{y}] \right) d\mathbf{y}$$

• And hence the optimal decoding rule is

$$\min_{\widehat{m}(\mathbf{y})} \Pr[m \neq \widehat{m}(\mathbf{y}) | \mathbf{y}] \Longleftrightarrow \max_{\widehat{m}(\mathbf{y})} \Pr[m = \widehat{m}(\mathbf{y}) | \mathbf{y}]$$
- It follows that the decision that minimizes the probability of error P_{e} is the MAP rule

$$\hat{m}(\mathbf{y}) = \underset{m \in \{0,...,M-1\}}{\operatorname{argmax}} p(\mathbf{x}_m | \mathbf{y})$$

where the posterior probability of m is

$$p(\mathbf{x}_m | \mathbf{y}) = \frac{p(m)p(\mathbf{y} | \mathbf{x}_m)}{p(\mathbf{y})} \propto p(m)p(\mathbf{y} | \mathbf{x}_m)$$

a priori (from prior knowledge) × likelihood (from data)

• As seen above, the rule can also be written in terms of log –probabilities.

• When p(m)=1/M, the MAP rule reduces to the Maximum Likelihood (ML) rule

$$\hat{m}(\mathbf{y}) = \underset{m \in \{0, \dots, M-1\}}{\operatorname{argmax}} p(\mathbf{y} | \mathbf{x}_{m})$$

likelihood (from data)

• Both ML and MAP rules partition the space of the received signal **y** into decision regions, one for each message *m*

MAP:
$$\mathcal{D}_m = \left\{ \mathbf{y} \in \mathbb{R}^N : m = \operatorname*{argmax}_{m' \in \{0, \dots, M-1\}} p(\mathbf{x}_{m'} | \mathbf{y}) \right\}$$



 Both ML and MAP rules partition the space of the received signal y into decision regions, one for each message m

$$\mathcal{D}_m = \left\{ \mathbf{y} \in \mathbb{R}^N : m = \operatorname*{argmax}_{m' \in \{0, \dots, M-1\}} p(\mathbf{y} | \mathbf{x}_{m'}) \right\}$$



• Consider now the important case of the additive Gaussian noise channel



• Consider now the important case of the additive Gaussian noise channel



$$\mathbf{y} = \mathbf{x} + \mathbf{z}$$
 with $z_n \sim \mathcal{N}\left(0, \frac{N_0}{2}\right)$ i.i.d.

• The likelihood of the message *m* for data **y** is

$$p(\mathbf{y}|\mathbf{x}_m) = p_{\mathbf{z}}(\mathbf{y} - \mathbf{x}_m) = \frac{1}{(\pi N_0)^{\frac{N}{2}}} \exp\left(-\frac{1}{N_0} \sum_{n=1}^N (y_n - x_{m,n})^2\right)$$
$$= \frac{1}{(\pi N_0)^{\frac{N}{2}}} \exp\left(-\frac{1}{N_0} \|\mathbf{y} - \mathbf{x}_m\|^2\right)$$

and hence the log-likelihood is

$$\log p(\mathbf{y}|\mathbf{x}_m) = -\frac{1}{N_0} \|\mathbf{y} - \mathbf{x}_m\|^2 + \text{const}$$

• For the Gaussian channel, the ML rule is simply a minimum distance rule

$$\hat{m}(\mathbf{y}) = \operatorname*{argmin}_{m \in \{0, \dots, M-1\}} \|\mathbf{y} - \mathbf{x}_m\|^2$$

and hence the decision regions are

$$\mathcal{D}_m = \left\{ \mathbf{y} \in \mathbb{R}^N : \ m = \operatorname*{argmin}_{m' \in \{0, \dots, M-1\}} \|\mathbf{y} - \mathbf{x}_{m'}\|^2
ight\}$$

Examples: N=1

N=2





M=2 BPSK

M=4 4-PSK

Example: *N*=1, *M*=4 (4-PAM)



BPSK transmission

%parameters Eb=1; No=0.01;%noise variance – try changing this parameter! L=1000; %number of bits

```
%simulation

m=randi(2,L,1)-1; %generate independent bits

x=sqrt(Eb)*2*(m-1/2); %generate signal vector

plot(x,zeros(size(x)),'o'); %plot transmitted constellation points

z=randn(L,1)*sqrt(No/2); %generate noise

y=x+z; %received signal

hold on; plot(y,zeros(size(y)), 'x');

mhat=(sign(y)+1)*1/2; %decoded bits

error_rate=sum(m~=mhat)/L
```



N0=0.1



error_rate =

0

No=1



error_rate =

0.0630

• For the Gaussian channel, the MAP rule is

$$\hat{m}(\mathbf{y}) = \underset{m \in \{0,\dots,M-1\}}{\operatorname{argmin}} \|\mathbf{y} - \mathbf{x}_m\|^2 - N_0 \log(p(m))$$

and the decision regions are accordingly defined.

• Example: For BPSK, the new threshold becomes (try to prove it!)

$$\frac{N_0}{4\sqrt{E_b}}\log\left(\frac{p(0)}{p(1)}\right)$$

• Using the equality

$$\|\mathbf{y} - \mathbf{x}\|^{2} = \|\mathbf{y}\|^{2} + \|\mathbf{x}\|^{2} - 2\langle \mathbf{x}, \mathbf{y} \rangle$$

we can rewrite the MAP rule also in terms of correlations only as

$$\hat{m}(\mathbf{y}) = \operatorname*{argmax}_{m \in \{0,...,M-1\}} \langle \mathbf{x}_m, \mathbf{y} \rangle + c_m$$

where

$$c_m = \frac{N_0}{2} \log(p(m)) - \frac{\|\mathbf{x}_m\|^2}{2}$$

• Block diagram of a MAP decoder



correlative decoder

Matlab: Implementing an ML Correlative Decoder

8-PSK transmission

%parameters

Eb=1;

No=1;%noise variance – try changing this parameter!

L=1000; %number of symbols

%simulation m=randi(8,L,1)-1; %generate independent symbols x(:,1)= sqrt(3*Eb)*cos(pi*(2*m+1)/8); x(:,2)= sqrt(3*Eb)*sin(pi*(2*m+1)/8); %generate signal vector plot(x(:,1),x(:,2),'o'); %plot transmitted constellation points z=randn(L,2)*sqrt(No/2); %generate noise y=x+z; %received signal hold on; plot(y(:,1),y(:,2), 'x');

Matlab: Implementing an ML Correlative Decoder

Xmat(:,1)= sqrt(3*Eb)*cos(pi*(2*[0:7]+1)/8); Xmat(:,2)= sqrt(3*Eb)*sin(pi*(2*[0:7]+1)/8);

for l=1:L %for each transmitted symbol
score=Xmat*y(l,:)'; %compute the score
[smax,imax]=max(score); %find the maximum score
mhat(l)=imax-1;
end

```
error_rate=sum(m~=mhat')/L
```

Matlab: Implementing an ML Correlative Decoder





0.3440

Example: QPSK, Eb = 1 J, No = 0.3 J $p(m) = \frac{1}{4}, m = 0, 1, 23$ · What is the output of the ophical decoder of y= 0.1 ? Solution 1: Since we have equipoloable messages, the decision regions of the ophimal ML decoder con be easily obtained: D, x, - + - 1 x, D,



For example,
$$||y-x_0||^2 = (0.1-1)^2 + (0.2-1)^2 = 1.45$$



· Repeat with p(0)=0.1, p(1)=0.5, p(2)=0.2, p(3)=0.2 Solution 1 : Apply MAP decoder voiring minimum disbuce decoder: _0.3 lg (0.1) $\frac{\|y - x_0\|^2}{-03 \log(0.5)} = 2.14$ Y $||y - x_1||^2$ m = 1_0.3 log (0.2) 3.13 min [1y-x2]2 -0.3 log (0,2) 2.73 $||y - x_3||^2$



Additive White Gaussian Noise Channel (AWGN)



Additive White Gaussian Noise Channel (AWGN)



$$y(t) = x(t) + z(t)$$

with WGN
$$z(t)$$
: $E[z(t)z(t-\tau)] = \frac{N_0}{2}\delta(\tau)$

- A random process *X*(*t*) is a collection of random variables indexed by time *t*.
- Another way to thinking about is in terms of random functions.
- We will be interested in stationary random processes: the probabilistic description of the process does not change with time.



- Two key quantities that characterize a stationary random process *X*(*t*) are:
- 1) Correlation Function
 - $R_x(\tau) = E[X(t)X(t-\tau)]$



• $R_x(\tau)$ tells us how predictable X(t) is based on $X(t - \tau)$:

 \uparrow |*R*_{*χ*}(*τ*)| → ↓ "randomness"

2) Power spectral density

- $S_{\chi}(f) = \mathcal{F}\{R_{\chi}(\tau)\}$
- Real and positive
- Symmetric $(S_x(f) = S_x(-f))$

• White Noise:

Uniform power spectral density



and impulsive correlation function

 $R_x(\tau) = \text{const} \times \delta(\tau)$

Additive White Gaussian Noise Channel (AWGN)



$$y(t) = x(t) + z(t)$$

with WGN z(t):
$$E[z(t)z(t-\tau)] = \frac{N_0}{2}\delta(\tau)$$

Μ



$$\langle y(t), \varphi_n(t) \rangle = y_n$$

correlative demodulator



matched filter demodulator

• Demodulated signal vector:

$$y_n = \langle y(t) = x(t) + z(t), \varphi_n(t) \rangle = \langle x(t), \varphi_n(t) \rangle + \langle z(t), \varphi_n(t) \rangle$$
$$= x_n + z_n$$

• Noise components are zero-mean Gaussian (linear combinations of Gaussian variables are Gaussian) with correlation:

$$\begin{split} \mathbf{E}[z_n z_{n'}] &= \mathbf{E}\left[\int \int z(t) z(t') \varphi_n(t) \varphi_{n'}(t') dt dt'\right] \\ &= \int \int \mathbf{E}[z(t) z(t')] \varphi_n(t) \varphi_{n'}(t') dt dt' \\ &= \frac{N_0}{2} \int \varphi_n(t) \varphi_{n'}(t) dt \\ &= \frac{N_0}{2} \delta_{nn'} \end{split}$$


How to Demodulate on an AWGN Channel?



How Does an Optimal Receiver Work?

• As a result, the cascade of optimal demodulator and decoder at the receiver can be summarized as



or equivalently with the matched filter demodulator in lieu of the correlative demodulator.

How Does an Optimal Receiver Work?

• Using invariance property of the inner product, the optimal cascade of demodulator and decoder can also be implemented directly as



• Note that this architecture requires *M*, which is typically much larger than *N*, analog correlators or matched filters.

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Are we Forgetting Anything?

• The demodulator captures the signal in full in the sense that

$$x(t) = \sum_{n=1}^{N} x_n \varphi_n(t)$$

but this is not the case for the noise component since

$$z(t) \neq \sum_{n=1}^{N} z_n \varphi_n(t)$$

• Can any other noise component obtained from

$$\tilde{z}(t) = z(t) - \sum_{n=1}^{N} z_n \varphi_n(t)$$

be useful for decoding the signal?

Are we Forgetting Anything?

- The short answer is: No, because the other noise components are independent of the signal and of the noise components **z** that affect the signal.
- The longer answer follows.

Which Parts of the Received Signal Can Be Neglected?

• Partition the received signal vector components into two parts as

$$\mathbf{y} = \left[\begin{array}{c} \mathbf{y}_1 \\ \mathbf{y}_2 \end{array} \right]$$

$$p(\mathbf{y}_2|\mathbf{y}_1,\mathbf{x}) = p(\mathbf{y}_2|\mathbf{y}_1)$$

then \mathbf{y}_2 is said to be irrelevant for the estimate of \mathbf{x} when \mathbf{y}_1 is given, while \mathbf{y}_1 is said to be a sufficient statistic for \mathbf{x} .

- Irrelevant components can be neglected with no loss of optimality, or, equivalently, the detector can focus solely on sufficient statistics.
- Proof:

If

$$p(\mathbf{y}|\mathbf{x}_m) = p(\mathbf{y}_1|\mathbf{x}_m)p(\mathbf{y}_2|\mathbf{y}_1,\mathbf{x}_m) \quad \text{chain rule of prob.}$$
$$= p(\mathbf{y}_1|\mathbf{x}_m)p(\mathbf{y}_2|\mathbf{y}_1) \quad \text{chain rule of prob.}$$

Which Parts of the Received Signal Can Be Neglected?

• As an application of the previous result, consider the discrete channel below:



 \mathbf{n}_1 and \mathbf{n}_2 are mutually independent and independent of \mathbf{x}

• We have

$$p(\mathbf{y}_2|\mathbf{y}_1, \mathbf{x}) = p(\mathbf{n}_2|\mathbf{x} + \mathbf{n}_1, \mathbf{x})$$
$$= p(\mathbf{n}_2)$$
$$= p(\mathbf{y}_2)$$
$$= p(\mathbf{y}_2|\mathbf{y}_1)$$

independence of \mathbf{n}_2 and $(\mathbf{x},\mathbf{n}_1)$

independence of $\mathbf{y}_{\scriptscriptstyle 2}$ and $(\mathbf{y}_{\scriptscriptstyle 1})$

and hence \mathbf{y}_2 is irrelevant and can be neglected.

Which Parts of the Received Signal Can Be Neglected?

• As another application, show that \mathbf{y}_2 is irrelevant also for the discrete channel below.



 \mathbf{n}_1 and \mathbf{n}_2 are mutually independent and independent of \mathbf{x}

Are we Forgetting Anything?

- No, because $\tilde{z}(t)$ is independent of $\sum_{n=1}^{N} z_n \varphi_n(t)$ and of x(t), and hence any component extracted from it is irrelevant.
- Proof: We need to show that $\tilde{z}(t)$ is independent of $\sum_{n=1}^{N} z_n \varphi_n(t)$:

$$\begin{split} \mathbf{E}\left[\tilde{z}(t')z_{n}\right] &= \mathbf{E}\left[\left(z(t') - \sum_{n'=1}^{N} z_{n'}\varphi_{n'}(t)\right)z_{n}\right] \\ &= \mathbf{E}\left[z(t')z_{n}\right] - \mathbf{E}\left[\left(\sum_{n'=1}^{N} z_{n'}\varphi_{n'}(t)\right)z_{n}\right] \\ &= \int \mathbf{E}[z(t)z(t')]\varphi_{n}(t)dt - \sum_{n'=1}^{N} \mathbf{E}\left[z_{n'}z_{n}\right]\varphi_{n'}(t) \\ &= \frac{N_{0}}{2}\int\delta(t-t')\varphi_{n}(t)dt - \frac{N_{0}}{2}\sum_{n'=1}^{N}\delta_{n,n'}\varphi_{n'}(t) \\ &= \frac{N_{0}}{2}\left(\varphi_{n}(t') - \varphi_{n}(t')\right) = 0 \end{split}$$

How Do We Design a Coding Scheme?

- In order to design the constellation, we need to evaluate the performance of a given constellation in the presence of an optimal receiver.
- As seen, we can concentrate on the discrete-time additive Gaussian model



$$\mathbf{y} = \mathbf{x} + \mathbf{z}$$
 with $z_n \sim \mathcal{N}\left(0, rac{N_0}{2}
ight)$ i.i.d.

How Do We Design a Coding Scheme?

• We will focus on the case of equiprobable messages for which MAP (optimal decoder) equals ML.

• Consider first *N*=1 and *M*=2 (binary communications)



• d_{ij} = distance between constellation points x_i and x_j

• Consider first *N*=1 and *M*=2 (binary communications)



• Consider first *N*=1 and *M*=2 (binary communications)



$$P_{e|1\to0}^B = \Pr[\widehat{m} = 0|m = 1]$$
$$= Q\left(\frac{d_{01}}{\sqrt{2N_0}}\right)$$

• The probability of error with N=1 and M=2 is hence given by

$$P_{e} = \frac{1}{2} P_{e|0\to1}^{B} + \frac{1}{2} P_{e|1\to0}^{B}$$
$$= Q \left(\frac{d_{01}}{\sqrt{2N_{0}}}\right)$$

- As we have seen, the distance depends on the transmission energy
- Recall: $E_{\rm b}$ = energy per bit
- Ex.: BPSK

• The probability of error is hence

$$P_e = Q\left(\sqrt{\frac{2E_b}{N_0}}\right)$$

• Ex.: On-Off Keying (OOK)

$$P_e = Q\left(\sqrt{\frac{E_b}{N_0}}\right)$$

• Since we need double E_b to obtain the same probability of error, ON-OFF keying has a loss of 3dB (= $10\log_{10}2$) with respect to the optimal modulation

Matlab: Plotting the Probability of Bit Error

BPSK and OOK

EbNodB=linspace(-5,10,100); %x axis in dB EbNo=10.^(EbNodB./10); %x axis in linear scale

```
for s=1:length(EbNo)
```

```
E=EbNo(s);
Pebpsk(s)=qfunc(sqrt(2*E));
Peook(s)=qfunc(sqrt(E));
```

end

```
semilogy(EbNodB,Pebpsk);
hold on
semilogy(EbNodB,Peook,'--');
```



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• Generalizing to any *N*, if M=2 (b=1), the probability of error is given by

$$P_e = Q\left(\frac{d_{01}}{\sqrt{2N_0}}\right)$$
 where $d_{01} = d_{10} = ||\mathbf{x}_0 - \mathbf{x}_1||$

for any N.

• Illustration:



• Note that, by the invariance of the inner product, we can compute the distance directly between the analog waveforms:

$$d_{01} = d_{10} = ||\mathbf{x}_0 - \mathbf{x}_1|| = \sqrt{\int (x_0(t) - x_1(t))^2 dt}$$

• Ex.: BPSK



$$d_{01} = d_{10} = 2$$

How Far Can You Go From a Wi-Fi Access Point?

• Ex.: Consider a Wi-Fi access point with a transmission power $P_x = 0$ dBm operating over a channel with attenuation at 1 m equal to $L_1 = -50$ dB and path loss $\gamma = 2$. Assume that the access point uses BPSK or OOK with modulator $\varphi_1(t) = \sqrt{(2/T)} \operatorname{sinc}(t/T) \cos(2\pi f c t)$ (*N*=1) with bandwidth 1 MHz and that the power spectral density of the noise is $N_0 = -170$ dBm/Hz. How far can you be if you wish to receive at a probability of error no larger than 10^{-2} ?

How Far Can You Go From a Wi-Fi Access Point?

• From the plot, using tables, or qfuncinv in MATLAB, we compute the required Eb/No:



How Far Can You Go From a Wi-Fi Access Point?

• It follows that the required received energy per bit is

 $E_{\rm b} = 4.3 - 170 = -165.7 \, \text{dBm}$ for BPSK

 $E_{\rm b} = 7.3 - 170 = -162.7$ dBm for OOK

• The required received power is

$$P_{\rm r} = \frac{E_{\rm b}}{T/b} = \frac{E_{\rm b}}{T} = -165.7 - (-60) = -105.7 \,\,\text{dBm for BPSK}$$
$$P_{\rm r} = \frac{E_{\rm b}}{T/b} = \frac{E_{\rm b}}{T} = -162.7 - (-60) = -102.7 \,\,\text{dBm for OOK}$$

• The maximum distance is obtained as

$$P_{\rm r} = P_{\chi} + L_1 \ ({\rm dB}) - \gamma \ 10 \ \log_{10}(d) = P_{\chi} - 50 - 20 \ \log_{10}(d)$$

= 0 - 50 - 20 \log_{10}(d) = -50 - 20 \log_{10}(d)

and so d = 609.5 m for BPSK and so d = 431.5 m are the maximum distances.

- Let's consider now any *M*
- Probability of error:

$$P_{e} = \frac{1}{M} \sum_{m=0}^{M-1} P_{e|m}$$
$$= \frac{1}{M} \sum_{m=0}^{M-1} \sum_{m' \neq m} P_{e|m \to m'}$$

where

$$P_{e|m} = \Pr[\hat{m} \neq m|m]$$

$$P_{e|m\to m'} = \Pr[\hat{m} = m'|m]$$

Computing

$$P_{e|m\to m'} = \Pr[\hat{m} = m'|m]$$

is generally difficult.

• We hence consider instead an upper bound that is easy to compute

$$P_{e|m\to m'} \le P^B_{e|m\to m'}$$

- The upper bound is obtained by assuming that only messages *m* and *m*' exist, and hence the system is binary.
- Having an upper bound is useful, since the real probability of error is smaller than the bound.



$$P_{e|m \to m'} \le P_{e|m \to m'}^B = Q\left(\frac{d_{mm'}}{\sqrt{2N_0}}\right)$$

• Ex.: 8-PSK



Union bound:

$$P_e \le \frac{1}{M} \sum_{m=0}^{M-1} \sum_{m' \ne m} Q\left(\frac{d_{mm'}}{\sqrt{2N_0}}\right)$$

Approximate union "bound":

$$P_e \cong \frac{1}{M} (\text{num. pairs at min. distance}) Q\left(\frac{d_{\min}}{\sqrt{2N_0}}\right)$$

 $d_{\min} = \min_{i \neq j} d_{ij} \qquad \text{minimum distance}$

 Important: By the invariance of the correlation, distances can be computed either in the signal space or on the waveforms of the signal set,

• Ex.: 4-Pulse Width Modulation (PWM)



Ex.: 4-PWM (b = 2)

1) Conditional squared distance spectrum:

for m = 0

$$d_{01}^{2} = \int_{T/4}^{T/2} \frac{16}{5} \frac{E_{b}}{T_{P}} dt = \frac{4E_{b}}{5}$$

$$d_{02}^2 = \frac{8E_b}{5}$$

$$d_{03}^{2} = \frac{12E_{b}}{5}$$
$$\Rightarrow \left[\left\{ \frac{4E_{b}}{5}, 1 \right\}, \left\{ \frac{8E_{b}}{5}, 1 \right\}, \left\{ \frac{12E_{b}}{5}, 1 \right\} \right]$$

for m = 1 and m = 2

$$\left[\left\{\frac{4E_b}{5},2\right\}, \left\{\frac{8E_b}{5},1\right\}\right]$$

for m = 3

$$\left[\left\{\frac{4E_b}{5},1\right\}, \left\{\frac{8E_b}{5},1\right\}, \left\{\frac{12E_b}{5},1\right\}\right]$$

2) Squared distance spectrum

$$\left[\left\{\frac{4E_b}{5}, 6\right\}, \left\{\frac{8E_b}{5}, 4\right\}, \left\{\frac{12E_b}{5}, 2\right\}\right]$$

3) Union bound:

$$\begin{split} P_e &\leq \frac{1}{4} \left(6Q\left(\sqrt{\frac{4E_b}{10N_0}}\right) + 4Q\left(\sqrt{\frac{8E_b}{10N_0}}\right) + 2Q\left(\sqrt{\frac{12E_b}{10N_0}}\right) \right) \\ &= \frac{3}{2} Q\left(\sqrt{\frac{2E_b}{5N_0}}\right) + Q\left(\sqrt{\frac{4E_b}{5N_0}}\right) + \frac{1}{2} Q\left(\sqrt{\frac{6E_b}{5N_0}}\right) \end{split}$$

4) Approximate union bound:

$$P_e \simeq \frac{3}{2} \operatorname{Q}\left(\sqrt{\frac{2E_b}{5N_0}}\right)$$



The squared distance spectrum is then given as

$$\left[\left\{\frac{8E_b}{5}, 6\right\}, \left\{\frac{32E_b}{5}, 4\right\}, \left\{\frac{72E_b}{5}, 2\right\}\right]$$

and the union bound is

$$P_{e} \leq \frac{1}{4} \left(6 \operatorname{Q}\left(\sqrt{\frac{4E_{b}}{5N_{0}}}\right) + 4Q\left(\sqrt{\frac{16E_{b}}{5N_{0}}}\right) + 2\operatorname{Q}\left(\sqrt{\frac{36E_{b}}{5N_{0}}}\right) \right)$$

$$\simeq \frac{3}{2} \operatorname{Q}\left(\sqrt{\frac{4E_{b}}{5N_{0}}}\right)$$
Union bound approximation

Matlab: Plotting the Union Bound

Union bound for 4-PWM and 4-PAM

EbNodB=linspace(-5,10,100); %x axis in dB EbNo=10.^(EbNodB./10); %x axis in linear scale

```
for s=1:length(EbNo)
```

E=EbNo(s); Pubpwm(s)=3/2*qfunc(sqrt(2/5*E))+qfunc(sqrt(4/5*E))+1/2*qfunc(sqrt(6/5*E)); Papppwm(s)=3/2*qfunc(sqrt(2/5*E));

Pubpam(s)=3/2*qfunc(sqrt(4/5*E))+qfunc(sqrt(16/5*E))+1/2*qfunc(sqrt(36/5*E)); Papppam(s)=3/2*qfunc(sqrt(4/5*E)); end

Matlab: Plotting the Union Bound

```
semilogy(EbNodB,Pubpwm,'b');
hold on
semilogy(EbNodB,Papppwm,'b--');
semilogy(EbNodB,Pubpam, 'r');
semilogy(EbNodB,Papppam, 'r--');
```
How Do We Compute the Union Bound?



How Do We Compute the Union Bound?

- The union bound approximation is increasingly accurate for large SNR values to the exponential decay of the Q function.
- The gain/ loss of a constellation as compared to another can be well approximated by considering only the arguments of the Q function in the union bound approximation.
- Ex.: Loss of 4-PWM as compared to 4-PAM

$$10\log_{10}\left(\frac{4/5}{2/5}\right) = 3 \text{ dB}$$

How Do We Evaluate the Performance of a Coding Scheme?

• Ex.: Try with 4-Pulse Position Modulation (PPM)



 $A = \sqrt{\frac{8 E_b}{T}}_{_{197}}$

t

What About the Bit Error Rate?

• The bit error rate (or probability is given as):



which can be bounded by using the union bound.

• Unlike the probability of error, the probability of bit error depends on the mapping between bits and symbols.

What About the Bit Error Rate?



Union bound on the probability of bit error:

$$\begin{split} P_b &\leq \frac{1}{4} \left(6\frac{1}{2} \operatorname{Q}\left(\sqrt{\frac{4E_b}{5N_0}}\right) + 4\frac{2}{2} \operatorname{Q}\left(\sqrt{\frac{16E_b}{5N_0}}\right) + 2\frac{1}{2} \operatorname{Q}\left(\sqrt{\frac{36E_b}{5N_0}}\right) \right) \\ & \simeq \frac{3}{4} \operatorname{Q}\left(\sqrt{\frac{4E_b}{5N_0}}\right) \end{split}$$

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6CCS3COS Communication Systems: Chapter 3

Osvaldo Simeone

What is This Course About?

- Overview
 - 1. One-shot digital communications: Fundamentals
 - 2. One-shot digital communications: Passband Systems
 - 3. Stream digital communications

Main references

- J. Cioffi, Lecture notes, Stanford Univ., Chapters 1, 2, 3



[http://www.extremetech.com]

• 1) Frequency Division Multiplexing (FDM)

...different information streams (e.g., radio stations) modulated on different carriers

Each carrier corresponds to a different passband channel



System	Carrier Frequency <i>f</i> _c
AM radio	530-1600 kHz
FM radio	88-108 MHz
Cellular	~900 MHz, ~1-2 GHz
Wi-Fi	2.4 GHz
Satellite	~3-6 GHz
Fiber optics	200 THz

legend: $k \rightarrow 10^3, M \rightarrow 10^6, G \rightarrow 10^9, T \rightarrow 10^{12}$, $Hz = {\rm cycles}/s$

2) The antenna size depends on the wavelength

$$\lambda = \frac{c}{f_c} \quad (c = 3 \times 10^8 \text{ m/s})$$

System	Wavelength λ
AM	~ 300 m
FM	~ 3 m
Cellular	~ 0.3 m
Wi-Fi	~ 0.1 m





• Note: The passband filter in practice is implemented at the receiver, but it is convenient to think of it as being part of the channel.

How To Carry Out Link Budgets for Passband Communications?

• The path loss at 1 m is typically given by Friis formula

$$L_1 (dB) = G (dB) + \gamma \ 10 \log_{10}(\frac{\lambda}{4\pi})$$

where G is the antenna gain and $\lambda = c/f_c$ ($c = 3 \times 10^8$ m/s).

• The overall path loss is given as

$$L (dB) = L_1 (dB) - \gamma \ 10 \ \log_{10}(d)$$

with *d* measured in meters



- Passband signals are difficult to process directly due to the large carrier frequency, especially in the digital domain.
- Operating the modulator and demodulator directly on passband signals requires tuning to the specific carrier frequency.
- Solution: Perform modulation and demodulation on baseband signals and carry out upconversion and downconversion separately.
- Upconversion/ downconversion are done via multiplication with sinusoidal signals at the carrier frequency.







Why is the Baseband Equivalent Complex?

• Passband signal: Fourier transform is non-zero only in a bandwidth *B* around the carrier frequency $\pm f_c \left(\frac{B}{2} < f_c\right)$



... suitable for transmission on a passband channel

• Passband signals are sent on a physical channel, and hence they are real. Therefore, their Fourier transform has Hermitian symmetry.

Why is the Baseband Equivalent Complex?

• **Baseband signal:** Fourier transform is non-zero only in a bandwidth *B* around the zero frequency



... not suitable for transmission on passband channel

• Given the generally asymmetric Fourier transform (as in the figure), baseband equivalent signals are complex in the time domain.

• Mathematically, a bandpass signal is defined as:

$$x_{pb}(t) = \sqrt{2} \ a(t) \cos(2\pi f_c t + \theta(t))$$

- Information is encoded by two baseband signals: amplitude a(t) and phase $\theta(t)$.

- Note: The additional term $\sqrt{2}$ as compared to the expression in J. Cioffi's notes is introduced to simplify some of the later derivations.

• Example: Amplitude modulation ($\theta(t) = 0$)



• Example: Phase modulation (a(t) = 1)



• Example: Amplitude and phase modulation



• Alternative form of the bandpass signal

$$x_{pb}(t) = \sqrt{2} x_I(t) \cos(2\pi f_c t) - \sqrt{2} x_Q(t) \sin(2\pi f_c t)$$
in-phase, or I,
component
$$x_{pb}(t) = \sqrt{2} x_I(t) \cos(2\pi f_c t) - \sqrt{2} x_Q(t) \sin(2\pi f_c t)$$
quadrature, or Q,
component

- Information is encoded by the two baseband signals $x_I(t)$ and $x_Q(t)$
- $-x_I(t)$ modulates the in-phase carrier $\cos(2\pi f_c t)$
- $-x_Q(t)$ modulates the quadrature carrier $-\sin(2\pi f_c t)$

• From amplitude/phase representation to in-phase quadrature representation:

Recall that $\cos(a + b) = \cos a \cos b - \sin a \sin b$

Using this formula, we obtain

$$x_{c}(t) = \sqrt{2} a(t) \cos(2\pi f_{c}t + \theta(t))$$

= $\sqrt{2} a(t) \cos \theta(t) \cos(2\pi f_{c}t) - \sqrt{2} a(t) \sin \theta(t) \sin(2\pi f_{c}t)$
 $x_{I}(t)$
 $x_{O}(t)$

and hence

 $x_{I}(t) = a(t) \cos \theta(t)$ $x_{Q}(t) = a(t) \sin \theta(t)$

- From in-phase / quadrature representation to amplitude / phase representation:
 - From the equations on the previous slide, we get

$$a(t) = \sqrt{x_I(t)^2 + x_Q(t)^2}$$

$$\theta(t) = \arg(x_I(t) + jx_Q(t))$$

Why Do We Use Complex Numbers to Represent Passband Signals?

• Illustration:



Information signal described by $(x_I(t), x_Q(t))$... cartesian coordinates $(a(t), \theta(t))$... polar coordinates

Remark: As *t* increases, the point • moves on the I-Q plane Remark: $\cos\left(2\pi f_c t + \frac{\pi}{2}\right) = -\sin(2\pi f_c t)$

- Based on the discussion from the previous slide, the information signal is completely described by the pair of baseband signals $(a(t), \theta(t))$ or $(x_I(t), x_Q(t))$.
- Baseband equivalent signal

$$x_{bb}(t) = x_I(t) + j x_Q(t) = a(t)e^{j\theta(t)}$$

Complex baseband representation



• $x_{bb}(t)$ is complex and baseband

• Exercise: Draw the baseband equivalent signals in the complex plane for the examples in slides 17, 18 and 19.

$$c(t) = x(t) + iy(t)$$

= $\left(\cos(t) + \frac{\cos(6t)}{2} + \frac{\sin(14t)}{3}\right)$
+ $i\left(\sin(t) + \frac{\sin(6t)}{2} + \frac{\cos(14t)}{3}\right)$
= $e^{it} + \frac{e^{i6t}}{2} + \frac{ie^{-14t}}{3}$. (2)

[Pei and Chang '17]



- Baseband signals are much easier (cheaper) to process than passband signals given the smaller frequencies involved.
- Baseband signals can be processed in the digital domain with Analog-to-Digital and Digital-to-Analog converters operating at feasible frequencies.
- Baseband signals are independent of the carrier frequencies and hence changing the carrier frequencies only requires to modify the up/down-converters.

What Does the Up-Converter Do?



LO = Local Oscillator

What Does the Down-Converter Do?



• Note: Up- and down-converters are fixed and need not be designed.
What Does the Down-Converter Do?

• Why do we need LPF? Consider the noiseless case y(t)=x(t)

$$y(t)\sqrt{2} \cos(2\pi f_c t) = 2x_I(t) (\cos(2\pi f_c t))^2$$

from L0
$$= x_I(t) + x_I(t) \cos(4\pi f_c t) - x_Q(t) \sin(2\pi f_c t) \cos(2\pi f_c t)$$

$$= x_I(t) + x_I(t) \cos(4\pi f_c t) - x_Q(t) \sin(4\pi f_c t)$$

$$\cos a \cos b = 1/2 (\cos(a - b) + \cos(a + b))$$

Removed by LPF
$$\cos a \sin b = 1/2 (\sin(a + b) - \sin(a - b))$$

And similarly for $y(t) \left(-\sqrt{2} \sin(2\pi f_c t)\right) =$
$$= x_Q(t) - x_Q(t) \cos(4\pi f_c t) - x_I(t) \cos(4\pi f_c t)$$

Can We Write Directly x_{pb}(t) as a Function of x_{bb}(t)?

• Define the analytic equivalent signal

 $x_{\rm A}(t) = x_{bb}(t)e^{j2\pi f_{\rm C}t}$

• We then have:

$$\begin{aligned} x_{pb}(t) &= \sqrt{2} a(t) \cos(2\pi f_c t + \theta(t)) \\ &= \sqrt{2} x_I(t) \cos(2\pi f_c t) - \sqrt{2} x_Q(t) \sin(2\pi f_c t) \\ &= \sqrt{2} Re\{x_A(t)\} \end{aligned}$$

- To summarize, a passband signal can be represented in the following ways:
 - 1. magnitude, phase $a(t), \theta(t)$ 2. inphase, quadrature $x_I(t), x_Q(t)$ 3. complex baseband $x_{bb}(t)$ 4. analytic $x_A(t)$

How To Represent Passband Signals?

Example: Compute in-phase and quadrature components, as well as amplitude of

 $x(t) = \operatorname{sinc}(10^6 t) \cdot \cos(2\pi 10^7 t) + 3\operatorname{sinc}(10^6 t) \cdot \sin(2\pi 10^7 t)$

How To Represent Passband Signals?

Example: Compute in-phase and quadrature components, as well as amplitude of

 $x(t) = \operatorname{sinc}(10^{6}t) \cdot \cos(2\pi 10^{7}t) + 3\operatorname{sinc}(10^{6}t) \cdot \sin(2\pi 10^{7}t)$

$$x_{I}(t) = \frac{\operatorname{sinc}(10^{6} t)}{\sqrt{2}}$$
$$x_{Q}(t) = -\frac{3\operatorname{sinc}(10^{6} t)}{\sqrt{2}}$$
$$x_{bb}(t) = \frac{\operatorname{sinc}(10^{6} t)}{\sqrt{2}} - j\frac{3\operatorname{sinc}(10^{6} t)}{\sqrt{2}}$$
$$a(t) = \sqrt{5}|\operatorname{sinc}(10^{6} t)|$$

Matlab: Plotting Baseband and Passband Signals

• A signal x(t) is represented as the vector

$$x = [\dots x(-2T_s), x(-T_s), x(0), x(T_s), x(2T_s), \dots]$$

by sampling



• In order to guarantee that no information loss is incurred by sampling, the sampling rate needs to satisfy the condition of the Nyquist-Shannon theorem:

$$\frac{1}{T_s} \ge 2 \times \text{ highest frequency of } X(f)$$

Matlab: Plotting Baseband and Passband Signals

Plotting signal from the previous example

```
Ts=10^-8; %sampling interval (it satisfies Shannon-Nyquist)
t=[-5*10^-6:Ts:5*10^-6];
```

```
x=sinc(10^6*t).*cos(2*pi*10^7*t)+3*sinc(10^6*t).*sin(2*pi*10^7*t);
plot(t,x)
hold on
a=sqrt(5)*abs(sinc(10^6*t));
plot(t,sqrt(2)*a, 'r--');
```

```
figure
xi=sinc(10^6*t)/sqrt(2);
xq=-3*sinc(10^6*t)/sqrt(2);
plot(t,xi);
hold on; plot(t,xq, 'r');
```

Matlab: Plotting Baseband and Passband Signals





• Using the frequency translation property of the Fourier transform, we can calculate

$$\begin{split} X_{pb}(f) &= \mathcal{F}\{x_{pb}(t)\} \\ &= \sqrt{2} \,\mathcal{F}\{x_{I}(t)\cos(2\pi f_{c}t)\} - \sqrt{2} \,\mathcal{F}\{x_{Q}(t)\sin(2\pi f_{c}t)\} \\ &= \frac{1}{\sqrt{2}} \Big(X_{I}(f - f_{c}) + X_{I}(f + f_{c}) \Big) - \frac{1}{\sqrt{2} \, j} \Big(X_{Q}(f - f_{c}) - X_{Q}(f + f_{c}) \Big) \\ &= \frac{X_{I}(f - f_{c}) + jX_{Q}(f - f_{c})}{\sqrt{2}} + \frac{X_{I}(f + f_{c}) - jX_{Q}(f + f_{c})}{\sqrt{2}} \\ &= \frac{1}{\sqrt{2}} \, X_{bb} \, (f - f_{c}) + \frac{1}{\sqrt{2}} X_{bb}^{*}(-f - f_{c}) \end{split}$$

where the last equality follows from the Hermitian symmetry of $X_I(f)$ and $X_Q(f)$, since

$$X_{bb}^*(-f - f_c) = X_I^*(-f - f_c) - j X_Q^*(-f - f_c) = X_I(f + f_c) - j X_Q(f + f_c)$$

• We have shown that

$$X_{pb}(f) = \frac{1}{\sqrt{2}} X_{bb}(f - f_c) + \frac{1}{\sqrt{2}} X^*_{bb}(-f - f_c)$$

• Remark: $X_c(f)$ satisfies Hermitian symmetry

• The relationship between $X_{bb}(f)$ and $X_{pb}(f)$ consists of an upconversion operation that guarantees Hermitian symmetry.



• The relationship between $X_{bb}(f)$ and $X_{pb}(f)$ consists of an upconversion operation that guarantees Hermitian symmetry.





Why Did We Add That sqrt(2) Term Again?

- In this way, the energy of $x_{pb}(t)$ is the same as the energy of $x_{bb}(t)$.
- This can be seen using Rayleigh theorem, as illustrated below.



Why Did We Add That sqrt(2) Term Again?



What Does This Mean for Coding and Modulation?

- The signal set produced by the baseband modulator is given by complex functions $x_{bb,m}(t)$ for m = 0,1, ..., 2^{M-1} .
- Therefore, both constellation and orthonormal basis functions are generally complex.

What Does This Mean for Coding and Modulation?



How to Encode?

• Encoder:

message $m \in \{0, ..., M-1\} \rightarrow$ symbol (**complex** vector) $\boldsymbol{x}_m = \begin{bmatrix} x_{m,1} \\ x_{m,2} \\ \vdots \\ x_{m,N} \end{bmatrix}$

- The signal space is C^N
- The set of all symbols x_m , m=1,...,M, is the signal constellation

How to Encode?



- More generally, any constellation that we have encountered earlier can be reinterpreted by considering every pair of real dimensions as a complex dimension.
- The reason for the (I,Q) labeling will be detailed shortly.

• Modulator:

symbol $x_m \rightarrow$ (analog and continuous-time) **complex** waveform $x_m(t)$ of duration *T* seconds (symbol period)

• Signal set:

$$x_m(t) = \sum_{n=1}^N x_{m,n} \varphi_n(t)$$

• The set $\{\varphi_n(t)\}, n = 1, ..., N$ is an *N*-dimensional **orthonormal basis**:

$$\langle \varphi_m(t), \varphi_n(t) \rangle = 0$$
 for $m \neq n$ (orthogonal)
 $\langle \varphi_m(t), \varphi_m(t) \rangle = ||\varphi_m(t)||^2 = 1$ (normalized)

Some Math

- Inner product or correlation:
 - between complex vectors

$$\langle \mathbf{v}, \mathbf{u} \rangle = \sum_{n=1}^{N} v_n u_n^*$$

- between complex functions

$$\langle v(t), u(t) \rangle = \int_{-\infty}^{\infty} v(t) u^*(t) dt$$

Some Math

- Squared Euclidean norm or energy:
 - for a complex vector

$$||\mathbf{v}||^2 = \langle \mathbf{v}, \mathbf{v} \rangle = \sum_{n=1}^N |v_n|^2$$

- for a complex function

$$||v(t)||^{2} = \langle v(t), v(t) \rangle = \int_{-\infty}^{\infty} |v(t)|^{2} dt$$

Some Math

Orthogonality:

- for complex vectors: vectors ${\bf v}$ and ${\bf u}$ are orthogonal if

$$\langle \mathbf{v}, \mathbf{u} \rangle = 0$$

- for complex functions: functions v(t) and u(t) are orthogonal if

$$\langle v(t), u(t) \rangle = 0$$

• Note that orthogonal pairs of real functions are also orthogonal when interpreted as complex functions (with zero imaginary part).

Examples:

1) N=1

$$\varphi_1(t) = \frac{1}{\sqrt{T}}\operatorname{sinc}\left(\frac{t}{T}\right)$$

2) Any N: Stream modulation

$$\varphi_n(t) = \frac{1}{\sqrt{T}} \operatorname{sinc}\left(\frac{t-n}{T}\right) \text{ for } n=1,...,N$$

3) Any N: Orthogonal Frequency Division Multiplexing (OFDM)

$$\varphi_n(t) = \frac{1}{\sqrt{T}} \operatorname{rect}\left(\frac{t}{T}\right) \exp\left(j2\pi\left(\frac{n}{T} - \frac{N}{2T}\right)t\right)$$
 for $n=1,...,N$



 $B \approx \frac{N}{T}$

Example: Given the 4-PSK constellation and the basis function indicated below find the signal set. Explain why the axes of the constellation are labeled as I and Q.

Example: Given the 4-PSK constellation and the basis function indicated below find the signal set. Explain why the axes of the constellation are labeled as I and Q.

$$x_{bb,0}(t) = -\sqrt{\frac{E_b}{T}}\operatorname{sinc}\left(\frac{t}{T}\right) - j\sqrt{\frac{E_b}{T}}\operatorname{sinc}\left(\frac{t}{T}\right), \ x_{bb,1}(t) = \sqrt{\frac{E_b}{T}}\operatorname{sinc}\left(\frac{t}{T}\right) - j\sqrt{\frac{E_b}{T}}\operatorname{sinc}\left(\frac{t}{T}\right)$$
$$x_{bb,2}(t) = -\sqrt{\frac{E_b}{T}}\operatorname{sinc}\left(\frac{t}{T}\right) + j\sqrt{\frac{E_b}{T}}\operatorname{sinc}\left(\frac{t}{T}\right), \ x_{bb,3}(t) = \sqrt{\frac{E_b}{T}}\operatorname{sinc}\left(\frac{t}{T}\right) + j\sqrt{\frac{E_b}{T}}\operatorname{sinc}\left(\frac{t}{T}\right)$$

As shown in the previous example, with a real basis function, the real part
of the constellation point is proportional to the I-component of the
baseband modulated signal and the imaginary part is proportional to the
Q-component.



• The same applies for stream modulation for each symbol, as well as to OFDM for each subcarrier.

- The maximum number of complex dimensions is given as *N*=*BT*, where *B* is the bandwidth of the passband signal, or equivalently the bandwidth of the baseband equivalent including also the negative frequencies.
- This is consistent with the general results in Chapter 2 since one complex dimension amounts to two real dimensions. In other words, the number of real dimensions is still *N*=2*BT*.

What Does This Mean for Decoding and Demodulation?



What Does This Mean for Decoding and Demodulation?

- Following the same reasoning as in the previous chapter, the optimal receiver can be obtained as the cascade of
 - Down-converter
 - Correlative or matched filter baseband demodulator (with complex correlations or impulse responses)
 - MAP decoder (or ML in case of equally likely messages)

How to Demodulate?

• After down-conversion, we have

$$y_{bb}(t) = x_{bb}(t) + z_{bb}(t)$$

with complex baseband noise

$$z_{bb}(t) = z_I(t) + j z_Q(t)$$

 The in-phase and quadrature components of the noise are independent WGN processes with power spectral density No/2 (to be discussed).

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How to Demodulate?



correlative demodulator

How to Decode?



$$c_m = \frac{N_0}{2} \log(p(m)) - \frac{\|\mathbf{x}_m\|^2}{2}$$

How to Demodulate and Decode?



$$c_m = \frac{N_0}{2} \log(p(m)) - \frac{\|\mathbf{x}_m\|^2}{2}$$

How to Demodulate and Decode?

• When the messages are equally likely, we can also use minimum distance decoding.


• **Problem:** Prove that the scores of the four messages in 4-PSK obtained by the receiver with a correlative decoder are the same as those obtained by the optimal receiver studied in the previous chapter in the absence of noise. Consider as an example the transmission of message *m*=0.

Solution: 1) Following Chapter 2: encoder: modulator : $\begin{array}{c} \chi_{1} & \left(\frac{1}{2} \right) \\ \chi_{2} & \left(\frac{1}{2} \right) \\ & \left(\frac{1}{$ $\varphi_2(t) = -\sqrt{\frac{2}{T}} \operatorname{sinc}\left(\frac{t}{T}\right) \operatorname{sin}\left(2\pi f c t\right)$ 0 X2 X2° (N=2) receiver $y(t) = \sqrt{E_b} \varphi_1(t) + \sqrt{E_b} \varphi_2(t)$ VEB $< y(t), y_1(t) >$ FVED 7 y(t) √ē, $\hat{m} = 0$ SKY, X1 < y(t), B max > <4, ×3

2) Using baseband equivalent transmitter and receiver modulator : encoder: $X_{1_{0}}$ $v_{X_{0}}$ $v_{X_{0}}$ V_{2} $v_{X_{3}}$ $(f_1(t) = \frac{1}{\sqrt{T}} \operatorname{sinc}\left(\frac{t}{T}\right)$ (N=1)receiver: $y(t) = (VE_{b} + jVE_{b}) \varphi_{1}(t)$ -XY,Xo>FRe Ztb $\dot{m} = 0$ you(t) < you(t), 42(t)> 4 = JEbrjJEb Ret-2JEB -> <4, ×2>)max -> {<4, ×3> -Re

$$for endowe: \langle \Psi, X_0 \rangle = (\overline{VEb} + j\overline{VEb})(\overline{VEb} - j\overline{VEb})$$
$$= (\overline{Eb} + \overline{Eb}) + j(\overline{Eb} - \overline{Eb}) = 2\overline{Eb}$$
$$\langle \Psi, X_1 \rangle = (\overline{JEb} + j\overline{VEb})(-\overline{VEb} - j\overline{VEb})$$
$$= (-\overline{Eb} + \overline{Eb}) + j(-\overline{Eb} - \overline{Eb}) = -2j\overline{Eb}$$

• **Problem:** Consider 8-PSK and that all symbols have the same probability. If Eb=1 and NO=0.1, what is the optimal decision if the received signal after demodulation is y=3+j0.01?

Matlab: Implementing a MAP Correlative Detector

8-PSK transmission (equal probability)

%parameters

Eb=1;

No=0.1;%noise variance – try changing this parameter!

L=1000; %number of symbols

%simulation m=randi(8,L,1)-1; %generate independent symbols x= sqrt(3*Eb)*cos(pi*(2*m+1)/8)+j* sqrt(3*Eb)*sin(pi*(2*m+1)/8); %generate signal vector plot(real(x),imag(x),'o'); %plot transmitted constellation points z=randn(L,1)*sqrt(No/2)+j*randn(L,1)*sqrt(No/2); %generate noise y=x+z; %received signal hold on; plot(real(y),imag(y), 'x');

Matlab: Implementing a MAP Correlative Detector

Xmat=sqrt(3*Eb)*cos(pi*(2*[0:7]+1)/8)'+j*sqrt(3*Eb)*sin(pi*(2*[0:7]+1)/8)';

for l=1:L %for each transmitted symbol
score=real(Xmat*y(l,:)'); %compute the score
[smax,imax]=max(score); %find the maximum score
mhat(l)=imax-1;
end

error_rate=sum(m~=mhat')/L





- To design the baseband modulator and demodulator, it is useful to operate directly with a baseband equivalent channel model.
- What is the equivalent baseband noise *z*_{bb}(*t*)? What is the equivalent baseband filter *H*_{bb}(*f*)?

Why Would We Want to Use An Equivalent Baseband System Again?

- As discussed, the main processing steps take place at the baseband modulator and demodulator. Therefore, for a communication engineer focusing on signal processing, up- and down-converters need not be included in the design.
- The baseband equivalent is carrier frequency-independent.
- (In practice, the channel model may depend on the carrier frequency.)

- A passband filter has a non-zero frequency response only around the carrier frequency $f_{\rm c}$.



$$x_{pb}(t) = \sqrt{2Re\{x_{bb}(t)e^{j2\pi f_{c}t}\}}$$
input passband
signal with
baseband
equivalent $x_{bb}(t)$
Passband filter
 $H_{pb}(f)$ or $h_{pb}(t)$
output passband
signal with
baseband
equivalent $x_{bb}(t)$
passband
frequency
response
response
$$y_{pb}(t) = \sqrt{2Re\{y_{bb}(t)e^{j2\pi f_{c}t}\}}$$
output passband
signal with
complex baseband
equivalent $y_{bb}(t)$



- How to choose the baseband filter so that we have the equivalence at the previous slide?
- It can be easily seen that we need

$$H_{pb}(f) = H_{bb}(f - f_c) + H_{bb}^*(-f - f_c)$$

• Remark: Unlike for signals, there is no $\frac{1}{\sqrt{2}}$ term.

Consider the bandpass filter in the figure below.



a) If the input signal is

 $x_{pb}(t) = \sqrt{2} \ x_I(t)\cos(2\pi 10^7 t)$ with $x_I(t) = \cos(2\pi 10^6 t) + \cos(6\pi 10^6 t)$

find the output $y_{pb}(t)$.

Input and filter:





$$\Rightarrow y_{pb}(t) = \sqrt{2} \left(2 \cos(2\pi 10^6 t) \right) \cos(2\pi 10^7 t)$$

$$y_I(t) = y_{bb}(t)$$

b) Calculate the complex baseband equivalent of the output $y_{pb}(t)$:

$$y_{bb}(t) = 2\cos(2\pi 10^6 t)$$

c) Calculate the equivalent baseband filter $H_{bb}(f)$ and $h_{bb}(t)$:



 $\Rightarrow h_{bb}(t) = 8 \times 10^6 \times \text{sinc}(4 \times 10^6 t)$

d) Using the baseband filter, calculate $y_{bb}(t)$. Compare with the results at point b).

Input and filter:



d) Using the baseband filter, calculate $y_{bb}(t)$. Compare with the results at point b).

Input and filter:



 $\Rightarrow y_{bb}(t) = 2\cos(2\pi 10^6 t)$, as at point b).

How Do We Obtain the Baseband Equivalent Noise?

• The bandpass noise z(t) is a WGN with power spectral density $S_z(f) = \frac{N_0}{2}$ within the bandwidth of the channel



How Do We Obtain the Baseband Equivalent Noise?

• The baseband noise is the output of the following blocks:

$$Z(t) \longrightarrow H_{pb}(f) \longrightarrow Down-converter \qquad z_{bb(t)} = z_I(t) + jz_Q(t)$$

• Assuming that $H_{pb}(f)$ is an ideal passband filter, it can be proved that

a) $z_I(t)$ and $z_Q(t)$ are WGN with power spectral densities

$$S_{z_{I}}(f) = S_{z_{Q}}(f) = \frac{N_{0}}{2}$$

within the bandwidth $\left(-\frac{B}{2}, \frac{B}{2}\right)$

b) $z_I(t)$ and $z_Q(t)$ are independent

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How Do We Obtain the Baseband Equivalent Noise?

Remark:

$$E[z(t)^{2}] = \frac{N_{0}}{2} \cdot 2B = N_{0}B$$
$$E[z_{I}(t)^{2}] = \frac{N_{0}}{2}B \qquad E[|z_{bb}(t)|^{2}]$$

$$E[z_{I}(t)^{2}] = \frac{-6}{2}B \\ E[z_{Q}(t)^{2}] = \frac{N_{0}}{2}B \\ B = N_{0}B \\ E[z_{Q}(t)^{2}] = \frac{N_{0}}{2}B \\ C = N_{0}B \\ C = N_{0}B$$

The power of the bandpass noise $E[z(t)^2]$ is hence equal to the power of the baseband signal $E[|z_{bb}(t)|^2]$.

This relationship is akin to that between the energy for bandpass and baseband signals.

Matlab: Generating Noise

• Generating a baseband WGN $W_z(t)$ with power $P_{W_z}(=N_0B_R)$ in MATLAB

 $W_I = sqrt(P_{W_Z}/2)^*$ randn (N,1); % generates N samples of $W_I(t)$

 $W_Q = sqrt(P_{W_Z}/2)^*$ randn (N,1); % generates N samples of $W_Q(t)$

$$W_z = W_I + j * W_Q;$$

% generates N samples of $W_z(t)$





• If the signals are bandlimited within the bandwidth *B* and the filter is ideal in this bandwidth, we can just write

$$y_{bb}(t) = x_{bb,m}(t) + z_{bb}(t)$$

where $z_{bb}(t)$ is AWGN

- From the viewpoint of modulator and demodulator, we have then obtained an AWGN channel as studied in the previous chapter with the only caveat that symbols and functions can be complex.
- From the viewpoint of encoder and decoder, the channel is additive Gaussian as studied in the previous chapter with one complex number representing two real dimensions.
- The optimal receiver is hence the one discussed in the previous slides.
- The probability of error can be computed as seen in the previous chapter.



LO = Local Oscillator



Phase asynchronism with phase offset θ

• Consider the noiseless case y(t)=x(t)

 $y(t)\sqrt{2}\cos(2\pi f_c t + \theta) = 2x_I(t)\cos(2\pi f_c t)\cos(2\pi f_c t + \theta)$

 $-2 x_Q(t) \sin(2\pi f_c t) \cos(2\pi f_c t + \theta)$

$$= x_{I}(t)\cos(\theta) + x_{I}(t)\cos(4\pi f_{c}t + \theta) - x_{Q}(t)\sin(4\pi f_{c}t + \theta) + x_{Q}(t)\sin(\theta) = x_{I}(t)\cos(\theta) + x_{Q}(t)\sin(\theta) \text{ after LPF} = y_{I}(t)$$

And similarly for $y(t) \left(-\sqrt{2}\sin(2\pi f_c t)\right)$: $y_Q(t) = x_Q(t)\cos(\theta) - x_I(t)\sin(\theta)$

 $\cos a \cos b = 1/2 (\cos(a - b) + \cos(a + b))$ $\cos a \sin b = 1/2 (\sin(a + b) - \sin(a - b))$

• Lack of phase synchronization hence causes interference between I and Q components:

$$y_I(t) = x_I(t)\cos(\theta) + x_Q(t)\sin(\theta)$$
$$y_Q(t) = -x_I(t)\sin(\theta) + x_Q(t)\cos(\theta)$$

• The previous relationship can also be written as

$$y_{bb}(t) = x_{bb}(t)e^{-j\theta}$$

• Equivalent baseband system



- How to detect a lack of phase synchronization?
- Consider BPSK. We have

$$y_{bb}(t) = x_m \varphi_1(t) e^{-j\theta}$$

• Hence, after demodulation we obtain

$$y = x_m e^{-j\theta}$$

• This implies that the received constellation is rotated.



- As a result, a decoder applying the decision regions of BPSK would fail completely if the phase offset is 90 degrees.
- A coherent decoder tries to estimate the phase offset and then compensates it before performing decoding.
- Estimation of the phase offset based on the reception of the data only has an irreducible ambiguity of 180 degrees.
- To resolve this ambiguity, we need to transmit pilot symbols prior to the data. Pilot symbols are known to the receiver in advance and do not carry any information.

- The same applies to other constellations, but the ambiguity is more pronounced.
- **Problem:** What is the maximum phase offset detectable without training symbols for 4-PSK? How about 8-PSK?
- **Problem (Non-coherent transmission):** What is the impact of a phase offset on on-off modulation? Can you design a coherent decoder with no pilots? Can you design a non-coherent decoder that works irrespective of the phase offset for equally likely messages?




LO = Local Oscillator



Frequency asynchronism with frequency offset Δf

• Consider the noiseless case y(t)=x(t)

 $y(t)\sqrt{2} \cos(2\pi (f_c + \Delta f)t)$ = $x_I (t)\cos(2\pi\Delta ft) + x_Q(t)\sin(2\pi\Delta ft)$ after LPF

And similarly for
$$y(t) \left(-\sqrt{2} \sin(2\pi (f_c + \Delta f)t) \right) =$$

= $x_Q(t) \cos(2\pi \Delta f t) - x_I(t) \sin(2\pi \Delta f t)$

• Equivalent baseband system



- How to detect a lack of frequency synchronization?
- Consider BPSK After downconversion and demodulation, the constellation looks as follows



• The constellation rotates at a frequency equal to the frequency offset. The frequency offset can hence be estimated in the frequency domain by observing received training symbols.

ING'S College **IONDON**



6CCS3COS Communication Systems: Chapter 4

Osvaldo Simeone

What is This Course About?

- Overview
 - 1. One-shot digital communications: Fundamentals
 - 2. One-shot digital communications: Passband Systems
 - 3. Stream digital communications

Main references

- J. Cioffi, Lecture notes, Stanford Univ., Chapters 1, 2, 3

What Have We Learned So Far?

• We have mostly focused so far on "one-shot" transmission with N=1 complex dimension (or N=2 real dimensions).



• Time-domain transmission:

$$\varphi_n(t) = \varphi(t - nT)$$
 for $n=1,...,N$

for some unitary-energy waveform $\varphi(t)$

• Time-domain transmission:

$$\varphi_n(t) = \varphi(t - nT)$$
 for $n=1,...,N$

for some unitary-energy waveform $\varphi(t)$

• Frequency-domain transmission: Orthogonal Frequency Division Multiplexing (OFDM)

$$\varphi_n(t) = \varphi(t) \exp\left(j2\pi \left(\frac{n}{T} - \frac{N}{2T}\right)t\right)$$
 for $n=1,...,N$

for some unitary-energy waveform $\varphi(t)$

• Each symbol carries one complex dimension.

• Time-domain transmission:



- *T* must be larger than 1/B since each symbol carries one complex dimension.
- The choice

$$\varphi(t) = \frac{1}{\sqrt{T}} \operatorname{sinc}(t/T)$$

maximizes the symbol rate, i.e., uses the minimum *T* for a given *B*.

• Frequency-domain transmission (OFDM):



- The spacing between subcarriers must be at least 1/T, since each symbol carries one complex dimension.
- The choice $\varphi(t) = \frac{1}{\sqrt{T}} \operatorname{rect}\left(\frac{t}{T}\right)$ uses the spectrum in the most efficient way, i.e., it minimizes the subcarrier spacing for a given *T*.

• Frequency-domain transmission (OFDM):



In practice, OFDM symbols, of duration *T*, are sent back to back with guard periods or cyclic prefixes

- The spacing between subcarriers must be at least 1/T, since each symbol carries one complex dimension.
- The choice $\varphi(t) = \frac{1}{\sqrt{T}} \operatorname{rect}\left(\frac{t}{T}\right)$ uses the spectrum in the most efficient way, i.e., it minimizes the subcarrier spacing for a given *T*.



Frequency-Time Representative of an OFDM signal

http://rfmw.em.keysight.com/wireless/helpfiles/89600b/webhelp/subsystems/wlan-ofdm/content/ofdm_basicprinciplesoverview.htm

- In this chapter, we will talk about two key aspects of data streaming.
- 1. Effect of channel distortions:

How does the channel affect the reception of successive symbols in time or frequency domain?

• 2. Coding over many dimensions:





- Let us start with time domain transmission.
- The design and analysis considered up to now applies only if the successive symbols do not interfere with one another.
- We know that this is the case if the delayed symbols are orthogonal.
- We will see that orthogonality in practice depends also on the channel and not only on the modulator.
- When orthogonality does not hold, the interference between successive transmissions is called **intersymbol interference (ISI)**. ISI can severely complicate the implementation of an optimum detector.

How to Stream Data in Time Domain?

- The message transmissions are separated by *T* units in time, where *T* is called the **symbol period**.
- 1/T is called the **symbol rate**.
- The **data rate** is

$$R = \frac{\log_2 M}{T} = \frac{b}{T} \text{ (bit/s)}$$

Introductory example: eucober: BPSK m[1] -> ×[1] Zud m[2] -> ×[2] modulator: $q_1(t) = \frac{1}{\sqrt{T}} \operatorname{rec}(\frac{t}{T}) = q(t)$ $\frac{1}{\sqrt{T}} \xrightarrow{P_1(t)} \frac{1}{\sqrt{T}} \xrightarrow{P_2(t)} 1 \xrightarrow{1}{\sqrt{T}} \xrightarrow{T}$



The optimal decoder, inving equally likely symbols, is BPSK dec .: $\xrightarrow{3} \overbrace{=}^{0} \overbrace{+}^{1} \xrightarrow{m} (=) \rightarrow \operatorname{Re} \rightarrow \overbrace{=}^{1} \xrightarrow{m} \xrightarrow{n}$ \Rightarrow decoding can be corried out symbol by symbol without loss of ophimality and the probability of bit error is $P_b = \mathcal{R}\left(\int_{N_b}^{2t_b}\right)$





let us fry to apply the symbol-by-symbol decoder for the second symbol: $< x(t), \varphi(t-T) > = < x[1] p(t) + x[2] p(t-T), \varphi(t-T) >$ = $x[1] < p(t), \phi(t-T) > + x[2] < p(t-T), \phi(t-T) >$ $= \times [1] + \times [2]$ => we have inter-symbol interference (ISI)!

Probability of bit error : Pb=Pr[m[]=0] × Pr[m[2] # m[2] m[1]=0] $+ \Pr[m[1]=1] \times \Pr[\hat{m}[2] \neq m[2] m[1]=1]$ " if m[1]=0 => ×[1]= JED and the equivalent condellation is -215 0 $\Pr\left[\hat{m}[2] \neq m[2][m[i]=0] = \frac{1}{2} \times Q\left(\frac{2\sqrt{b}}{\sqrt{m/2}}\right) + \frac{1}{2} \times \frac{1}{2}$ $= \frac{1}{2} Q \left(\sqrt{\frac{8 + b}{N}} \right) - \frac{1}{4}$ · serve probability if m[1]=1 $\Rightarrow l_b = \frac{1}{9}Q\left(\left(\frac{8Eb}{Nb}\right) + \frac{1}{4}\right)$



How to Stream Data in Time Domain?



• Note: With rectangular waveforms: no ISI (with an ideal channel), but generally unacceptable spectrum.

How to Stream Data in Time Domain?

• The baseband equivalent transmitted baseband signal can be written as (dropping the subscript "bb"):

$$x(t) = \sum_{l} x[l] \varphi(t - lT)$$



• Frequency selectivity: If the transmitter uses waveform $\varphi(t)$, the effective waveform is given by the convolution $p(t) = \varphi(t) * h(t)$.

• As a result of the channel, the received signal is

$$y(t) = \sum_{l} x[l] p(t - lT) + z(t)$$

where $p(t) = \varphi(t) * h(t)$.

• Example: As seen, if

 $\varphi(t) = 1/\sqrt{T} \operatorname{rect}(t/T)$ $h(t) = \delta(t) + \delta(t - T)$

we have

 $p(t) = 1/\sqrt{2T} \operatorname{rect}(t/(2T))$

... the effective waveform suffers from ISI while the original waveform does not. In other words, the ISI is created by the multipath channel.

• **Example:** ISI can limit the transmission rate $p(t) = 1/(1+t^4)$; BPSK; x(-T)=-1 and x(0)=1







eye diagram



eye diagram (for 4-PAM)

How Can We Avoid ISI?

• In the absence of ISI, the optimal correlative demodulator would operate as follows:



- Note that each symbol is encoded separately and hence it can also be decoded separately if there is no ISI.
- Note also that, unlike the simpler demodulator considered in the introductory example, this demodulator requires knowledge of the channel and is hence a coherent decoder.
• In the absence of ISI, the optimal correlative demodulator would operate as follows:

• Let us calculate *y*[*k*] assuming no noise for simplicity:

$$y[k] = \sum_{l} x[l] < p(t - lT), p(t - kT) >$$

$$= x[k] < p(t - kT), p(t - kT) >$$

$$E_{p}$$

$$+ \sum_{l \neq k} x[l] < p(t - lT), p(t - kT) >$$
inter-symbol interference (ISI)

• In the absence of ISI, the optimal correlative demodulator would operate as follows:

demodulator

• Let us calculate *y*[*k*] assuming no noise for simplicity:

$$y[k] = \sum_{l} x[l] < p(t - lT), p(t - kT) >$$

= $x[k] < p(t - kT), p(t - kT) >$
 E_{p}
+ $\sum_{l \neq k} x[l] R_{p}((l - k)T)$

inter-symbol interference (ISI)

autocorrelation function

$$R_p(\tau) = < p(t), p(t - \tau) >$$

- As a first observation, the useful signal x[k] is multiplied by the energy of the waveform Ep. Therefore, this decoder is able to collect all the energy created by the channel through multiple propagation paths.
- In order to have zero ISI, we need to ensure that the effective waveform *p*(*t*) is such that its correlation function satisfies

$$R_p(kT) = 0$$
 for all $k \neq 0$

• Nyquist criterion for zero ISI: *p*(*t*) should be orthogonal to all its time shifts at multiples of the symbol time *T*.

- As a first observation, the useful signal x[k] is multiplied by the energy of the waveform Ep. Therefore, this decoder is able to collect all the energy created by the channel through multiple propagation paths.
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 for all $k \neq 0$

- Nyquist criterion for zero ISI: *p*(*t*) should be orthogonal to all its time shifts at multiples of the symbol time *T*.
- More generally, ISI from one symbol l to another symbol k depends on how far two symbols are through the autocorrelation Rp((l-k)T).





$$R_p(\tau) = \mathcal{F}^{-1} \left\{ \begin{array}{c} G_p(f) \\ T \\ -1/2T \end{array} \right\} = \operatorname{sinc} \left(\frac{\tau}{T} \right)$$



does not satisfy the Nyquist criterion

since:



- d) We can reduce the impact of ISI by choosing waveforms with lower sidelobes than the sinc.
- A typical example is given by raised-cosine waveforms

$$p(t) = sinc\left(\frac{t}{T}\right) \cdot \left[\frac{\cos\left(\frac{\alpha \pi t}{T}\right)}{1 - \left(\frac{2\alpha t}{T}\right)^2}\right] \qquad \alpha = \text{roll-off factor}$$





• **Equalization** methods are used by communication engineers to mitigate the effects of the intersymbol interference.



- The equalizer attempts to compensate for the channel
- Equalizers can be linear or non-linear.

· Following the entroductory example anue that Y[k] = x[k] + x[k-1] + 2[k]· Linear equalizer: $\hat{y}[k] = -\hat{y}[k-i] + \hat{y}[k]$





=> Signal-to-noise - plus- uiterference natio (SINR):
- Without equalization:
SINR =
$$\frac{E_x}{E_x + N_0} = \frac{1}{1 + N_0/F_x}$$

- With equalization
SINR = $\frac{E_x}{2N_0}$

• Note: The SINR of the linear equalizer scheme decreases for successive symbols: equalization causes noise to accumulate (try!).

· Non-linear equalizer: - from y[k-i] decode x[k-i] -> x[k_i] - cancel x [K-1]: $\hat{g}[k] = -x[k-i] + \hat{g}[k]$

· What does this do? $\hat{y}[2] = -xT_1] + y[2]$ - if the decision x[1] is correct: $\hat{y}[2] = -x[n] + x[2] + x[n] + 2[2]$ = (x[2]+2[2])no 151! no incresse in noise !

• What does this do?

$$\hat{y}[2] = -\hat{x}[1] + \hat{y}[2]$$

$$- \hat{i}f$$
 the decision $\hat{x}[1]$ is not correct

$$\hat{y}[2] = -\hat{x}[1] + \hat{x}[2] + \hat{x}[1] + \hat{z}[2]$$

$$= \hat{x}[2] + (\hat{x}[1] - \hat{x}[1]) + \hat{z}[2]$$

$$= \hat{x}[2] + (\hat{x}[1] - \hat{x}[1]) + \hat{z}[2]$$

$$= \hat{x}[2] + (\hat{x}[1] - \hat{x}[1]) + \hat{z}[2]$$





How to Stream Data in the Frequency Domain?

• Frequency-domain transmission:



$$x(t) = \sum_{n=1}^{N} x[n] \varphi_n(t)$$

$$\rho_n(t) = u(t) \exp\left(j2\pi \left(\frac{n}{T} - \frac{N}{2T}\right)t\right)$$

where u(t) has energy equal to one and its Fourier transform guarantees no ISI in the frequency domain

How to Stream Data in the Frequency Domain?

• Frequency-domain transmission:



$$x(t) = \sum_{n=1}^{N} x[n] \varphi_n(t)$$

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where u(t) has energy equal to one and its Fourier transform guarantees no ISI in the frequency domain

• Waveform u(t) can be chosen as a rectangle with duration *T* or as the Fourier transform of a raised-cosine waveform.

How to Stream Data in the Frequency Domain?

• Frequency-domain transmission:



 $B \approx \frac{N}{T}$

- Convolutive channels (i.e., filters) *H*(*f*) do not affect the orthogonality of OFDM subcarriers.
- This is one of the key advantages of OFDM, which has motivated its adoption in most modern systems.
- Each OFDM subcarrier f_n is merely multiplied by the value $H(f_n)$.

Example: Assume en OFDM system with symbol of duration 1 us. We have N=4 subcorriers Autat are the channel gains on the four subcorriers of the Channel is $h(t) = \delta(t) + \delta(t_{-T})$

Subcarriers: , f[MHz] channel gains . $H(f) = 1 + e^{-2\pi fT}$ $= 1 + e^{-1} 2\pi f \sqrt{10^{-6}}$ $\Rightarrow |H(f_1)| = |H(-10^6)| = |1 + e^{+j^2 T}| = 2$ $H(f_2) = |H(o)| = |1 + e^{-j0}| = 2$ $|H(t_3)| = |H(10^6)| = 2$ $|H(f_4)| = |H(2 \times 10^6)| = |1 + e^{-14\pi}| = 2$ No 181 and increased chample gain!



How about with the channel $h(t) = S(t) + S(t - \frac{1}{2})?$

Channel gains : $H(f_1) = |\Lambda_+ e^+)^{2\# lo^6} \frac{10^{-1}}{2} | = |\Lambda_- \Lambda| = 0$ Zero channel H(fo) = |1+ej°| = 2 $H(f_2) = |1 + e^{-j2 + 10^6} \frac{10^{-6}}{2}| = 0$ $H(f_3) = |1 + e^{-j 4\pi 106 \sqrt{5}} | = |1 + e^{-j 2\pi}| = 2$ the frequency response can be zero in Some subcarriers,



• The available bandwidth depends on the carrier frequency.



• Subcarrier spacing increases (and hence *T* decreases) with the overall bandwidth in order to avoid *N* being too large for complexity reasons.

[Bhushan et al '17]

μ	Δf = 2 ^μ ·15 kHz
0	15 kHz
1	30 kHz
2	60 kHz
3	120 kHz
4	240 kHz
5	480 kHz



[Keysight Technologies '18]



[Keysight Technologies '18]

- A slot can be:
 - All downlink
 - All uplink
 - Mixed downlink and uplink
 - Static, semi-static or dynamic
- Slot aggregation is supported
 - Data transmission can be scheduled to span one or multiple slots



• Different services will share the same time-frequency resources.



- MBB = Mobile BroadBand
- D2D = Device-to-Device

[Bhushan et al '17]

Can We Use a Large N to Improve the Performance?

- The encoders and decoders studied up to now operate on one symbol, i.e., on two real dimensions or one complex dimension, at a time.
- The probability of error is hence determined by the minimum distance between constellation points in the two dimensional signal space.
- This type of systems are known as being **uncoded**.
- Can we improve the probability of error by **coding** and decoding over multiple symbols (i.e., over N>1 complex dimensions)?
- In other words, can we improve the system performance by coding?

How Do We Define Performance?

• 1) Bandwidth or spectral efficiency

$$\eta = \frac{R}{B}$$
 (bit/s/Hz)

• Since N=BT (complex dimensions), T=1/B is the minimum symbol durations for a complex dimension or two real dimensions. Therefore, we also write

$$\eta = RT = \frac{R}{B} \text{ (bit/2D)}$$
How Do We Define Performance?

• 1) Bandwidth or spectral efficiency

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 (bit/s/Hz)

• Since N=BT (complex dimensions), T=1/B is the minimum symbol durations for a complex dimension or two real dimensions. Therefore, we also write

$$\eta = RT = \frac{R}{B} \text{ (bit/2D)}$$

• 2) Power or energy efficiency

$$\frac{E_b}{N_0} \qquad \text{for a given } P_e$$

• 3) Complexity

Can We Use a Large N to Improve the Performance?



Can We Use a Large N to Improve the Performance?

• Energy efficiency:

$$d_{c} = \min m dispuce between foints in the constellation
$$d_{min} = \min dispuce between encoded
symbols?
$$d_{m,m'}^{2} = \sum_{k=1}^{N} [x_{m}[n] - x_{m'}[n]]^{2}$$

$$d_{m,m'}^{2} = N d_{c}^{2}$$

$$\Rightarrow P_{e} \approx Q\left(\frac{Nd_{c}^{2}}{2N_{o}}\right)$$$$$$

Can We Use a Large N to Improve the Performance?

For example, for BPSK Pe = Q(2EbxN => the every efficiency has improved by N

- We have concluded that repetition coding offers:
 - a coding gain (that is, an improvement of the energy efficiency) of 10log₁₀N dB
 - a reduction of spectral efficiency by 1/N



By C. E. SHANNON

Who is Claude Shannon Again?

- He's the most important genius you've never heard of, a man whose intellect was on par with Albert Einstein and Isaac Newton.
- At the age of 21, he published what's been called the most important master's thesis of all time, explaining how binary switches could do logic. It laid the foundation for all future digital computers.
- At the age of 32, he published "A Mathematical Theory of Communication," which has been called "the Magna Carta of the information age." Shannon's masterwork invented the *bit*, or the objective measurement of information, and explained how digital codes could allow us to compress and send any message with perfect accuracy.

Who is Claude Shannon Again?

• He worked on the top-secret transatlantic phone line connecting FDR and Winston Churchill during World War II and co-built what was arguably the world's first wearable computer. He learned to fly airplanes and played the jazz clarinet. He rigged up a false wall in his house that could rotate with the press of a button, and he once built <u>a gadget</u> whose only purpose when it was turned on was to open up, release a mechanical hand, and turn itself off. Oh, and he once had a photo spread in *Vogue* magazine.



Excerpts from this post



@ D. Costello



@ D. Costello



@ D. Costello

















































How to Encode over Multiple Dimensions?



How Do Convolutional Codes Work?



- Assume BPSK, and encode the information bits m in a stream *b*[1], *b*[2], ... where *b*[i]=1 if *m*[i]=1 and *b*[i]=-1 if *m*[i]=0.
- Example: m=(m[1]=0, m[2]=1, m[3]=1, m[4]=0,..., m[k]=1) message b= (b[1]=-1, b[2]=1, b[3]=1, b[4]=-1,..., b[k]=1) uncoded (signed) bits
- A convolutional encoder takes as input *k* information bits and outputs *n* encoded bits.
- The rate of the code is defined as r=k/n.
- Note that, with BPSK modulation, the spectral efficiency is computed as follows:

$$R = \frac{k}{nT} = \frac{kB}{n} = rB$$
 $\eta = \frac{R}{B} = r$

• As seen, this can be improved to

$$\eta = 2r$$

with QPSK.

• **Example**: r=1/2 and constraint length (memory) = 3



convolutional encoder

• **Example**: r=1/2 and constraint length (memory) = 3



• All registers are initialized to 1.

• State diagram description



state diagram

• State diagram description



state diagram

b=[1, 1] - 1, -1, 1, 1, 1, x = [1, 1,]-1, -1. -1, -1 -1, -1, 1, -1, -1, 1, -1, -1,



lower transitions correspond to b[j]=1 and higher transitions to b[j]=-1

trellis diagram



• The last two transmitted bits are fixed so as to end at state (1,1).



trellis diagram

Minimum distance decoding would require the computation of 2^k distances

$$\min_m \sum_j ||\mathbf{y}[j] - \mathbf{x}_m[j]||^2$$

or correlations

$$\max_{m}\sum_{j} \langle \mathbf{y}[j], \mathbf{x}_{m}[j] \rangle$$

where $x_m[j]$ is the jth encoded symbol for message m=0,1,...,2^k - 1.

• The Viterbi algorithm allows us to solve the problem with a complexity that is linear in *n*.





 $\mathbf{y} = [(1,3), (-2,1), (4,-1), (5,5), (-3,-3), (1,-6), (2,-4)]$



 $\mathbf{y} = [(1,3), (-2,1), (4,-1), (5,5), (-3,-3), (1,-6), (2,-4)]$

• 2) Feedforward pass: for each state compute path with maximum metric (survivor)



















- choose maximum between
 -1+5 and 5-5
- prune non-surviving path (dashed)



and so on...



• 3) Backward pass: Backtracking to find the decoded sequence



What About Eb?

- In the previous discussion, we have chosen the coded BPSK symbols to take the values +1 and -1.
- In practice, the values of the encoded symbols *x* depend on Eb. How?
- Set x to be +A and -A.

What About Eb?

- In the previous discussion, we have chosen the coded BPSK symbols to take the values +1 and -1.
- In practice, the values of the encoded symbols x depend on Eb. How?
- Set x to be +A and -A.
- Note that we have

$$kE_b$$
 = total energy

• We conclude that

$$nA^2 = kE_b \Rightarrow A = \sqrt{rE_b}$$

• To compute the probability of error, and hence the energy efficiency, we can use the union bound.

$$P_e \cong (\text{number of pairs at the min distance})Q\left(\sqrt{\frac{d_{min}^2}{2N_0}}\right)$$

• How to compute the minimum distance?

• Each codeword x_m , encoding a message m, corresponds to a path on the trellis.



• Each codeword x_m , encoding a message m, corresponds to a path on the trellis.



• The distance between two codewords is equal to

$$d^{2}(\mathbf{x}_{m}, \mathbf{x}_{m'}) = \sum_{j} ||\mathbf{x}_{m}[j] - \mathbf{x}_{m'}[j]||^{2}$$
$$= 4rE_{b}d_{H}(\mathbf{x}_{m}, \mathbf{x}_{m'})$$

Hamming distance between two codewords = number of differences

• To compute the probability of error, and hence the energy efficiency, we can use the union bound.

 $\left| \frac{d^2_{min}}{2N_0} \right|$

$$P_e \cong \frac{1}{2^k}$$
 (number of pairs at the min distance) $Q \left(1 \right)$

- How to compute the minimum distance?
- Consider one codeword as a reference, namely the all-1 codeword.
- We are interested in finding the **detour** on the trellis that has the minimum Hamming distance.



a detour

reference path



a detour

reference path

$$d_H(\mathbf{x}_{ref}, \mathbf{x}_{det}) = 5$$



a detour

reference path

$$d_H(\mathbf{x}_{ref}, \mathbf{x}_{det}) = 5$$

• It is not difficult to see that there is no detour at a smaller Hamming distance (why?).



a detour

reference path

$$P_{e} \cong \frac{1}{2^{k}} (\text{number of pairs at the min distance}) Q\left(\sqrt{\frac{4rE_{b} \times 5}{2N_{0}}}\right)$$
$$= \frac{1}{2^{k}} (\text{number of pairs at the min distance}) Q\left(\sqrt{\frac{5E_{b}}{N_{0}}}\right)$$

In Summary...



 This convolutional code has spectral efficiency of ¹/₂ with BPSK and 1 with QPSK and a coding gain with respect to BPSK of

$$10\log_{10}\left(\frac{5}{2}\right)\cong 4 \text{ dB}$$

• This improves over repetition code with the same spectral efficiency, which has a coding gain of 3 dB

More Examples of Convolutional Codes



Figure 12.7: A feedforward r = 2/3 encoder.

[Moon '05]

More Examples of Convolutional Codes



Figure 12.2: A systematic r = 1/2 encoder.

[Moon '05]

More Examples of Convolutional Codes



[Moon '05]

Turbo Codes



How to Increase the Spectral Efficiency?

- In order to increase the spectral efficiency, we need to use large constellations.
- In fact, if we used a constellation carrying *b* bits, the spectral efficiency would be *br*.
- However, with a constellation that is different from QPSK, the distances between codewords would **not** be proportional to the corresponding Hamming distances.

How to Increase the Spectral Efficiency?


How to Increase the Spectral Efficiency?

- Three main solutions:
 - Trellis Coded Modulation (TCM)
 - Multilevel modulation
 - Bit Interleaved Coded Modulation (BICM)

How to Increase the Spectral Efficiency?



code only chooses subset at the first level

Additional Material



- Modern base stations and access points have multiple antennas.
- Mobile devices at higher frequencies can also pack multiple antennas.



• Massive MIMO, i.e., the deployment of base stations with hundreds of antennas, is a key technology for 5G.



- Multiple antennas at the transmitting end of the communication link enable:
 - **Diversity**: transmission from multiple antennas can be leveraged so as to reduce the change of deep fades in the channel
 - **Space-Division Multiple Access**: serving multiple users in the same time-frequency resources by using beamforming (see figure)



- Multiple antennas at the receiing end of the communication link enable:
 - **Diversity**: reception from multiple antennas can be combined so as to reduce the change of deep fades in the channel

• Multiple antennas at both ends of a communication link enable transmission of multiple streams between a transmitter and a receiver.



• Consider a single-antenna device communicating to a multi-antenna base station.



• Consider a single-antenna device communicating to a multi-antenna base station.

uniform linear antenna array



transmitter positioned at an angle θ

• Consider a single-antenna device communicating to a multi-antenna base station.

uniform linear antenna array $\frac{\theta}{D \sin \theta}$ $\frac{1}{2D \sin \theta}$

• Relative propagation delay for antenna *n*

D

$$\tau_n = \frac{D\sin(\theta)}{c}(n-1)$$

• Passband signal received at first antenna

$$y_1(t) = \sqrt{2} \operatorname{Re}\{x_{bb}(t) \exp(j2\pi f_c t)\}$$

• Passband signal received at the *n*th antenna

$$y_n(t) = \sqrt{2} \operatorname{Re} \{ x_{bb}(t - \tau_n) \exp(j2\pi f_c(t - \tau_n)) \}$$

$$\cong \sqrt{2} \operatorname{Re} \{ x_{bb}(t) \exp(j2\pi f_c(t - \tau_n)) \}$$

$$= \sqrt{2} \operatorname{Re} \{ x_{bb}(t) \exp(-j2\pi f_c \tau_n) \exp(j2\pi f_c t) \}$$

• Baseband signal received at the nth antenna

$$y_{bb,n}(t) = x_{bb}(t) \exp\left(-j2\pi f_c \tau_n\right)$$

= $x_{bb}(t) \exp\left(-j2\pi f_c \frac{D\sin(\theta)}{c}(n-1)\right)$
= $x_{bb}(t) \exp\left(-j2\pi \frac{D\sin(\theta)}{\lambda}(n-1)\right)$

• Define the steering vector (N_a = number of antennas)

$$\mathbf{a}(\theta) = \begin{bmatrix} 1 \\ \exp\left(-j2\pi\frac{D\sin(\theta)}{\lambda}\right) \\ \exp\left(-j4\pi\frac{D\sin(\theta)}{\lambda}\right) \\ \vdots \\ \exp\left(-j2(N_a-1)\pi\frac{D\sin(\theta)}{\lambda}\right) \end{bmatrix}$$



• Consider transmission in the absence of ISI and communication using a time-domain waveform

$$x_{bb,m}(t) = x_m \varphi(t)$$

• Received signals on the antennas

$$\mathbf{y}_{bb}(t) = x_m \varphi(t) \mathbf{a}(\theta) + \mathbf{z}_{bb}(t)$$

where

$$\mathbf{y}_{bb}(t) = \begin{bmatrix} y_{bb,1}(t) \\ y_{bb,2}(t) \\ \vdots \\ y_{bb,N_a}(t) \end{bmatrix}$$

• Maximum ratio combining maximizes the signal-to-noise ratio



• Consider transmission in the absence of ISI and communication using a time-domain waveform

$$x_{bb,m}(t) = x_m \varphi(t)$$
 $\mathbf{y}_{bb}(t) = \mathbf{a}(\theta) x_m \varphi(t) + \mathbf{z}_{bb}(t)$

• Maximum ratio combining maximizes the signal-to-noise ratio

$$\mathbf{a}^{\dagger}(\theta)\mathbf{y}_{bb}(t) = \mathbf{a}^{\dagger}(\theta)\mathbf{a}(\theta)x_{m}\varphi(t) + \mathbf{a}^{\dagger}(\theta)\mathbf{z}_{bb}(t)$$
$$= ||\mathbf{a}(\theta)||^{2}x_{m}\varphi(t) + \mathbf{a}^{\dagger}(\theta)\mathbf{z}_{bb}(t)$$
$$= N_{a}x_{m}\varphi(t) + \mathbf{a}^{\dagger}(\theta)\mathbf{z}_{bb}(t)$$

• It can be proved that

$$Re\{\mathbf{a}^{\dagger}(\theta)\mathbf{z}_{bb}(t)\} \sim \mathcal{N}(0, N_a \times N_0/2)$$
$$Im\{\mathbf{a}^{\dagger}(\theta)\mathbf{z}_{bb}(t)\} \sim \mathcal{N}(0, N_a \times N_0/2)$$

since the power of independence random variables is equal to the sum of the individual powers.

What is the Performance Gain?

• It can be proved that

$$Re\{\mathbf{a}^{\dagger}(\theta)\mathbf{z}_{bb}(t)\} \sim \mathcal{N}(0, N_a \times N_0/2)$$
$$Im\{\mathbf{a}^{\dagger}(\theta)\mathbf{z}_{bb}(t)\} \sim \mathcal{N}(0, N_a \times N_0/2)$$

since the power of independence random variables is equal to the sum of t the individual powers.

• Hence, the SNR is improved by a factor equal to the number of antennas:

$$\frac{N_a^2 E_b}{N_a N_0} = N_a \times \frac{E_b}{N_0}$$

What is the Performance Gain?



What Can We Gain with MIMO?

- With MIMO with N_a antennas at the transmitter and receiver, we can communicate N_a data streams simultaneously.
- In this way, the spectral efficiency is multiplied by N_a .

What Can We Gain with MIMO?

• MIMO provides a multiplexing gain and not merely a coding gain.

