

# Mobile Wireless Networks

## Chapter 1: Wireless Transmissions

Prof. **Andrea Detti**  
Univ. of Rome Tor Vergata  
Electronic Engineering Dept.  
Andrea.detti@uniroma2.it



### Sources:

1. Vijay K. Garg, "Wireless Communications and Networking", available at <http://www.slideshare.net/shirazthegreat/wireless-communicationsandnetworking>
2. Prof. Dr.-Ing. Jochen H. Schiller [www.jochenschiller.de](http://www.jochenschiller.de), MC – 2015, available at [http://www.mi.fu-berlin.de/inf/groups/ag-tech/teaching/resources/Mobile\\_Communications/course\\_Material/](http://www.mi.fu-berlin.de/inf/groups/ag-tech/teaching/resources/Mobile_Communications/course_Material/)
3. Christopher Cox, *An Introduction to LTE: LTE, LTE-Advanced, SAE, VoLTE and 4G Mobile Communications: Second Edition*, 2

# **BACKGROUND ON WIRELESS COMMUNICATIONS**

# DIGITAL MODULATIONS

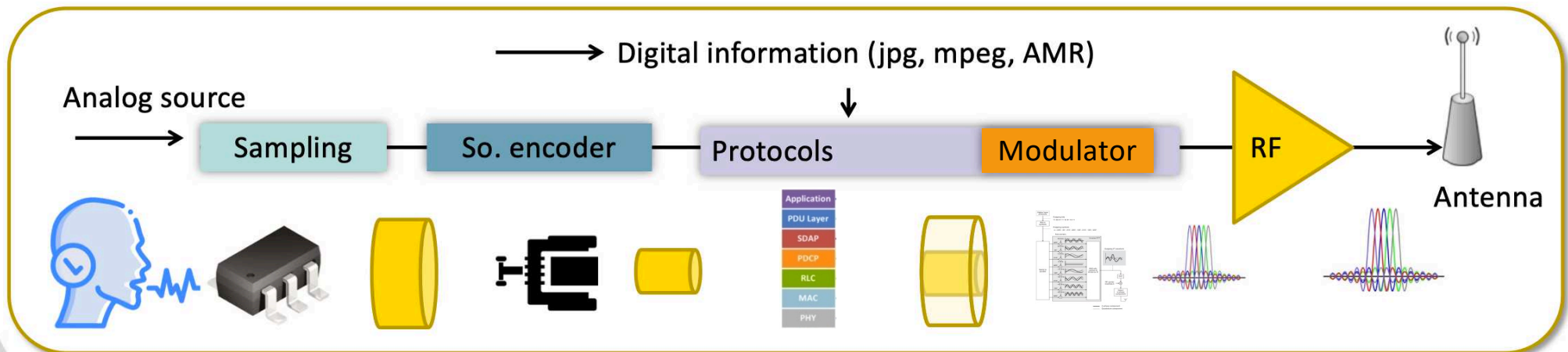
## Transmitter, channel, receiver

- Basic element of a wireless communication are a **transmitter**, a **receiver** and a **radio channel**
- Transmitter transforms digital data (bits) originated by a source to a radio signal
- Radio channel propagates the signal
- Receiver recovers digital data from the signal and sends it to the data sink
  - Receiver should also overcome distortions and disturbance that occurs over the communication channel



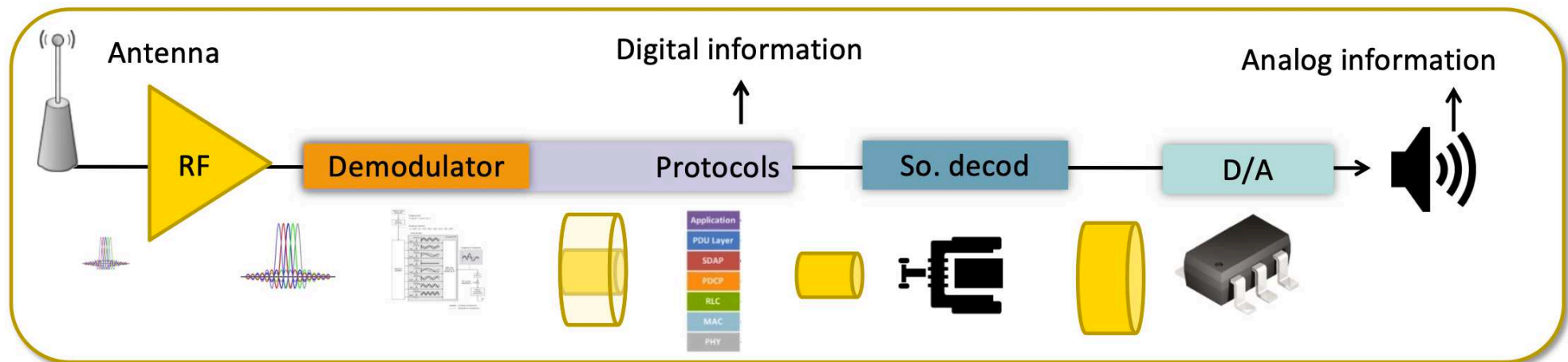
## Transmission chain

- A/D sampling converts from analog to digital
- Source encoder makes a kind of data “compression”
- Protocols provide *communication features* (networking, reliability, security, etc.)
- Modulator converts bits in analog waveforms
- RF section amplifies and shifts the signal to the required frequency (e.g. 2GHz)



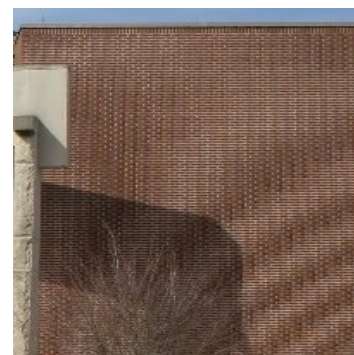
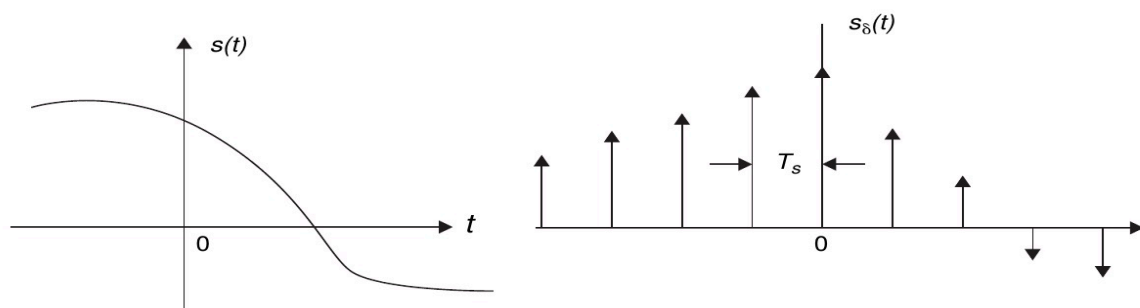
## Reception chain

- RF section filters, amplifies and moves signal to baseband
- Demodulator converts waveforms to bits
- Protocols provide *communication features* (networking, reliability, security, etc.)
- Source decoder decompress information
- D/A sampling converts from digital to analog



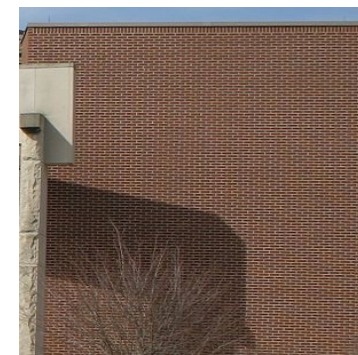
## Sampling and Quantization Process

- Periodical extraction of values from a continuous-time analog signal  $s(t)$  to generate a discrete-time signal  $s_\delta(t)$
- Nyquist–Shannon sampling theorem: if the bandwidth of  $s(t)$  is  $B$  (Hz) then perfect reconstruction occurs if the sampling rate  $1/T_s \geq 2B$ . Otherwise aliasing occurs which degrades the quality of reconstructed signal
- Quantization: sampled values are approximated to a finite number of levels (quantization error)



Aliasing

Spatial frequency bandwidth and sampling density:  
sampling theorem applied to image

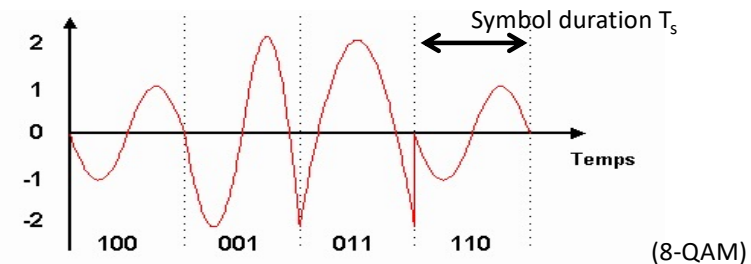


No aliasing

## Modulation: from bits to waveforms

- The sequences of bits arriving to the modulator are grouped in blocks of  $b$  bits, forming a finite symbol set of  $M (=2^b)$  symbols.
- A system using  $M$  symbols is referred to as an  $M$ -ary system ( $M$  aka mod. order)
  - $b = 1, M=2$ , binary system
  - $b = 2, M=4$ , quaternary system
- For each symbol the modulator uses a different waveform

...100001011110  $\rightarrow$  (100)(001)(011)(110)

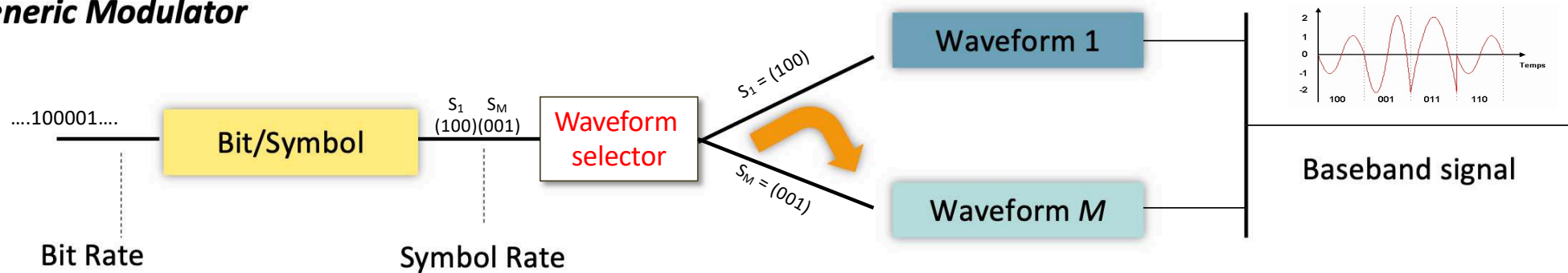




# Modulation

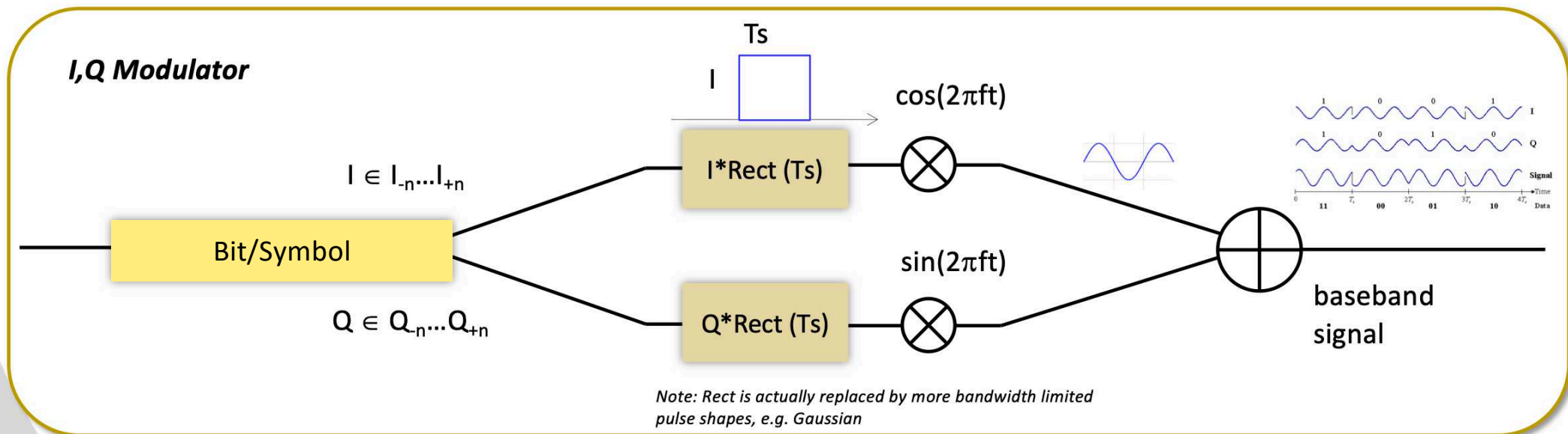
- An ideal modulator generates symbols from bits and then a selector triggers the transmission of the waveform associated to the symbol
- Modulation schemes differ on the used waveforms
- And different waveforms are more or less robust versus transmission impairments

## Generic Modulator



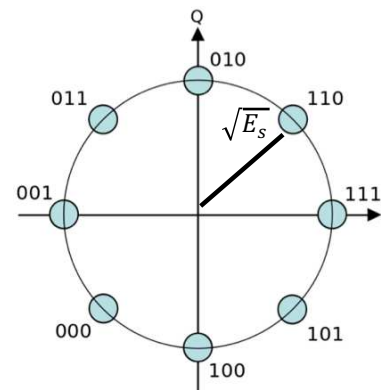
## I,Q Modulation

- In-phase and quadrature modulation is a family of modulations whose waveforms are characterized by:
  - Constant central frequency( $f$ ) and symbol duration ( $T_s$ )
  - Variable sine ( $Q$  - Quadrature amplitude) and cosine amplitude( $I$  – In phase amplitude)

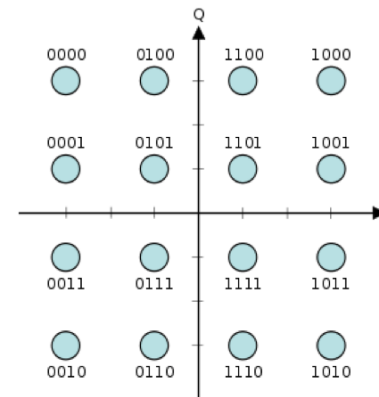


## I,Q Symbol representation

- The symbols of a I,Q modulation can be arranged on a 2D plane
- Points of the X axis represents the in-phase (I) amplitude; points of the Y axis represents the quadrature (Q) amplitude
- The possible (X,Y) point forms the “**constellation diagram**”
- Distance of a point from the center is a measure of the square root of the energy associated to the symbol ( $E_s$ )



8-PSK



16-QAM

- Typical I,Q modulation schemes are
  - **PSK**: Phase Shift Key,  $M$  symbols are located around a circle
  - **QAM**: Quadrature Amplitude Modulation,  $M$  symbols are uniformly located in a square area
  
- Frequency Shift Keying Modulation (FSK)
  - Base waveforms transmitted at difference frequency
  - Minimum Shift Keying (MSK): continuous phase frequency shift key with minimum frequency separation in order to have waveform orthogonality
  - GMSK: MSK with Gaussian shape
  
- Differential modulation (e.g. DPSK, DQAM): like PSK/QAM but symbols are associated to the difference between current bit block and previous block. E.g. difference=0 symbol n.1, difference=1, symbol n.2, etc.

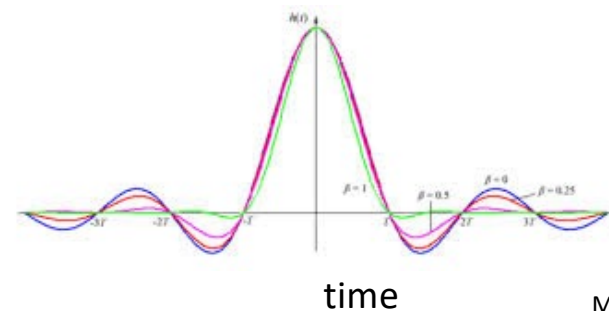
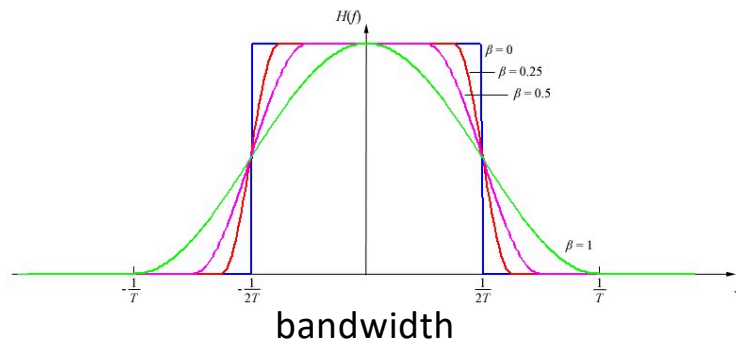
## Nyquist Bandwidth

- Theoretical minimum baseband bandwidth  $B_w$  required for transmission of  $R_s$  symbols per second (no noise)

$$B_w = \frac{(1 + \beta)}{T_s} = R_s(1 + \beta)$$

One symbol per sec requires  
(at minimum) 1 Hz

- $\beta$  roll-off factor due to the pulse shape (e.g. raised cosing). Trade-off between bandwidth and time extension of the signal
  - The more the time duration, the lower the bandwidth (small beta), but the higher inter-symbol interference



## Spectral Efficiencies

- In case of a modulation using  $M$  symbols the number of bit per symbol is  $\log_2 M$
- Relation between bit rate and symbol rate  $R_b = R_s \log_2 M$
- **Modulation spectral efficiency:** how much modulation exploits the Hz

$$\frac{R_b}{B_w} = \frac{\log_2 M}{(1 + \beta)} \text{ bit/s/Hz}$$

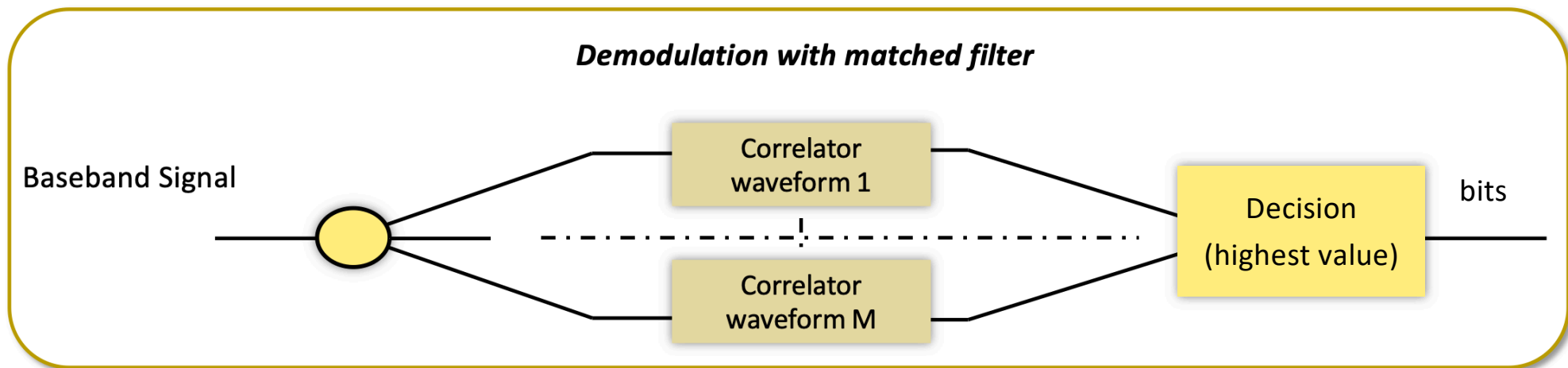
Modulation Efficiency can be increased  
increasing the number of symbols

- Link spectral efficiency of a cellular system usually takes into account also for physical layer overheads (unused times, channel coding, etc.)
  - 4G LTE (MIMO 4x4, 20 MHz)  $\rightarrow$  16.32 (peak)
  - 5G NR (MIMO 8x8, 20 MHz)  $\rightarrow$  30 (peak)

- On the receiving side a demodulator transforms back waveforms to digital symbols and bits, but
- The channel introduces degradations
  - Power attenuation, waveform distortion
  - Interference (from other transmitters)
  - Noise (thermal noise, amplifier noise, etc.)
- And these impairments leads to errors after the demodulation process
  - Bit Error Rate (BER)

## Demodulation

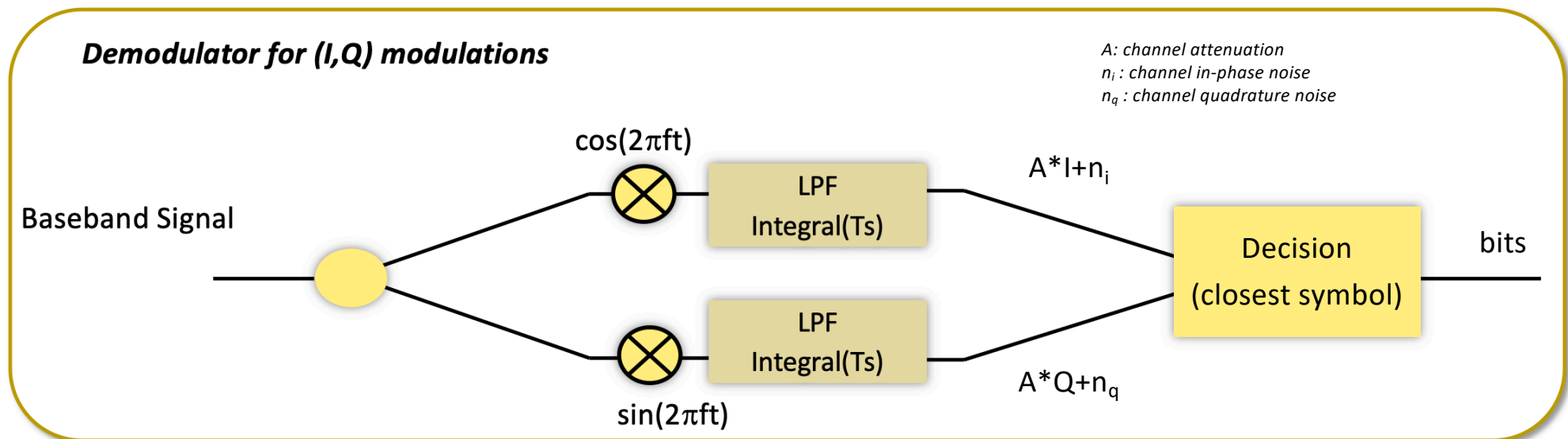
- The optimum demodulation process in case of AWGN channel and symbol with equal probability is realized by means of a bank of matched filters
  - Correlation of the incoming signal with a waveform
  - Highest output means best similarity





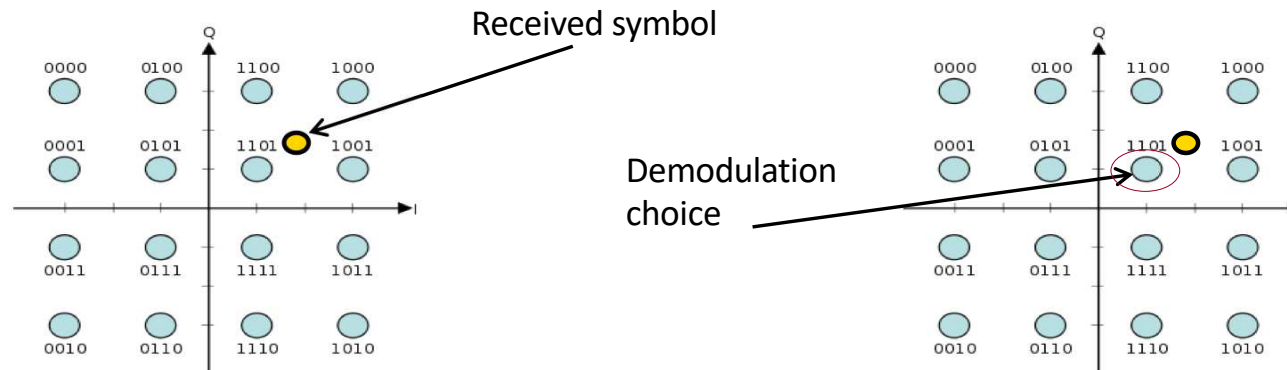
## Demodulation

- In case of (I,Q) modulation scheme, the matched filter demodulator can be implemented in a simple way....



# Demodulation

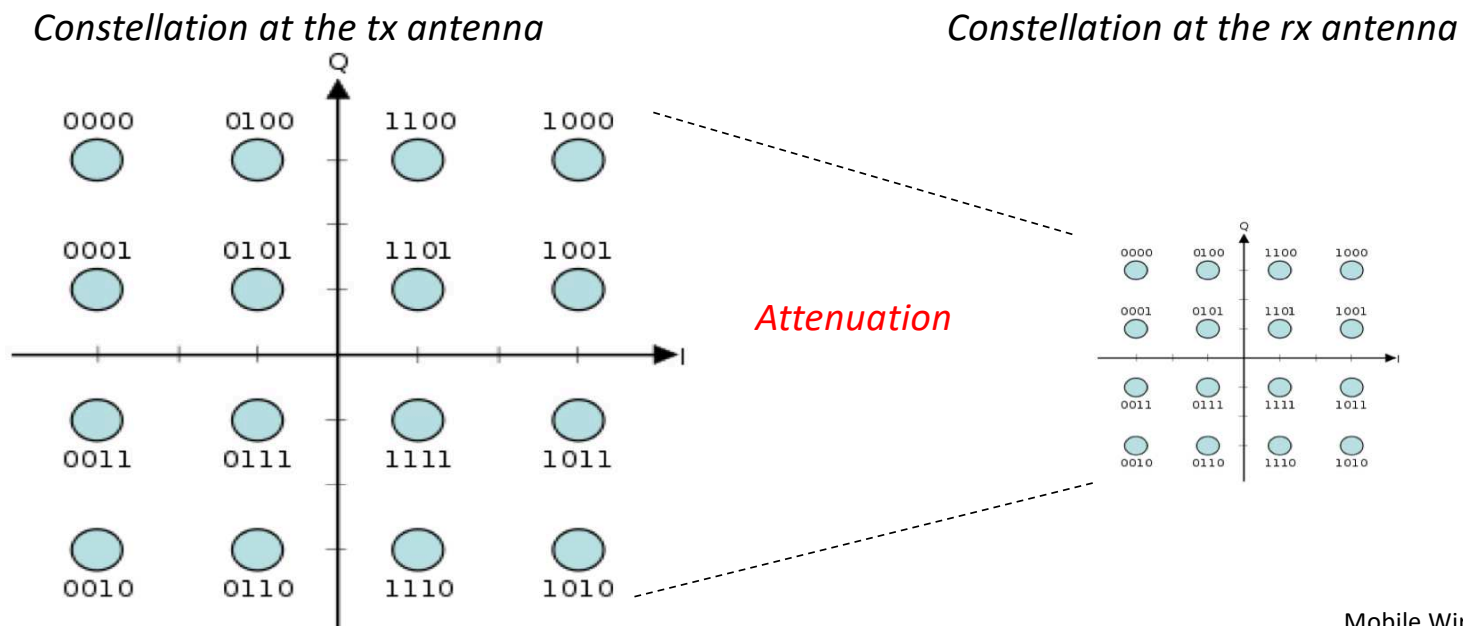
- For I,Q modulation the optimal demodulation process can resemble the following rule:
  - Report the received symbol in the constellation plane
  - Select the closest symbol among the  $M$  possible ones
- A similar reasoning could be repeated for other modulating schemes



<https://web.stanford.edu/group/cioffi/book/chap1.pdf>



## Demodulation

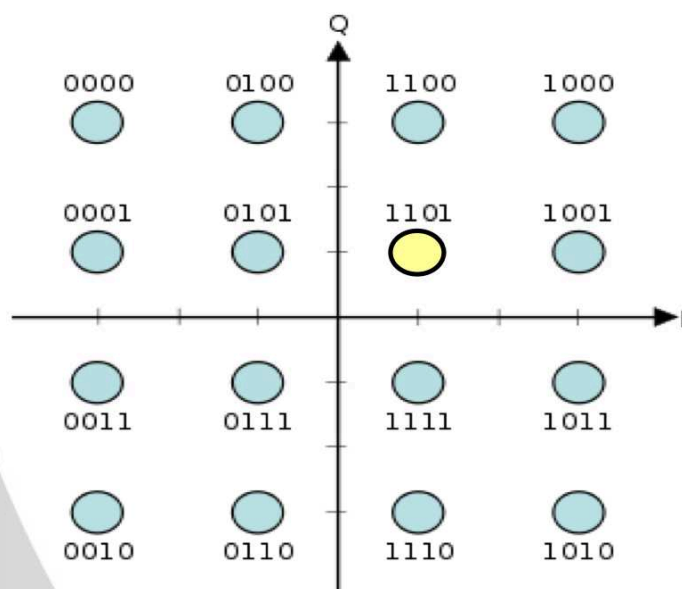
- Channel introduces signal attenuation and noise (which include waveform distortion for simplicity)
- Let us see these effects on the constellation plane
- Attenuation: reduce energy, i.e. distance of the symbols from the origin. Symbols in the constellation get more closer



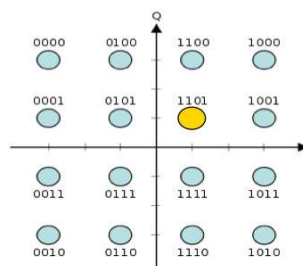
## Demodulation

- Noise: the noise randomly moves the transmitted signal on the constellating plane, indeed change the I,Q values in a not uniform way.

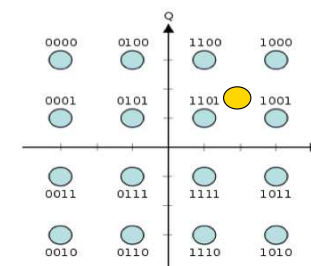
 Tx symbol  
 Rx symbol



*Attenuation*

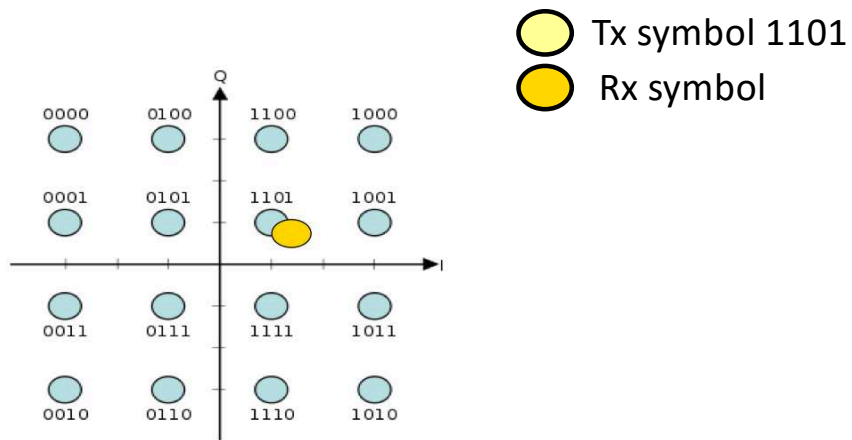


*Noise*

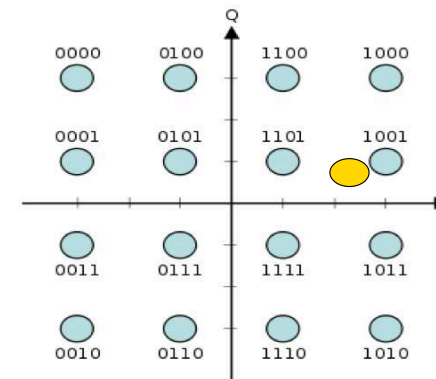


## Demodulation

- If the symbol movement leads the Rx symbol to be however closer to the transmitted one, we have perfect demodulation: no error
- But otherwise the demodulator will select another symbol that is different from the transmitted one: we have a bit error



No error  
demod bits 1101



error  
demod bits 1001

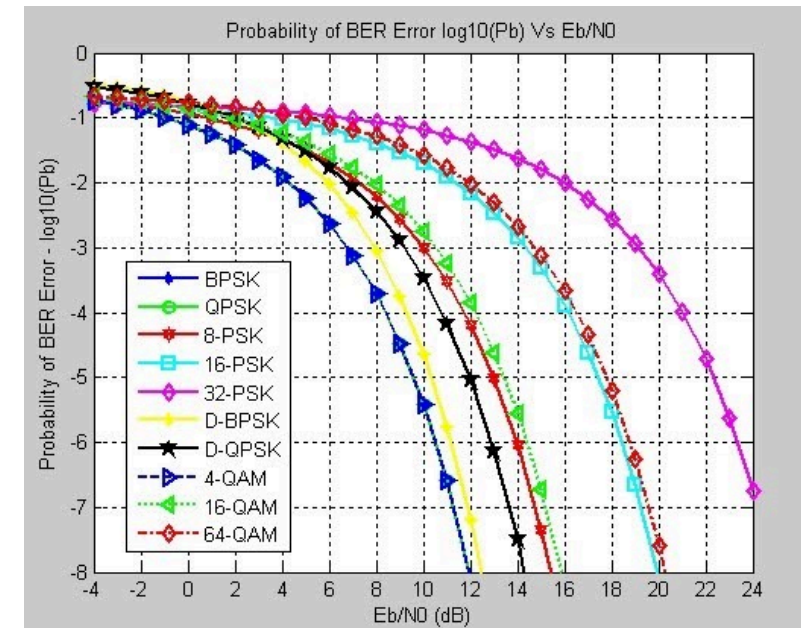
## Bit Error Rate

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- Average number of wrong bits over the total number of transmitted bits
- It is a critical dimensioning parameter to make feasible communication service
  - Different services need of different maximum allowed BER
- For a given modulation scheme, BER at the demodulation output depends on the ratio between the average energy per symbol measured at the demodulator ingress ( $E_s$ ) and the spectral noise ( $N_0$ ).
  - The energy determines the distance among symbols of the constellation: the smaller the energy, the closer the symbols, the higher the error probability of a given amount of noise
  - The noise determines how much the rx symbol is moved with respect to the location of the transmitted one. The more the noise, the more the movement, the higher the error probability

## Bit Error Rate

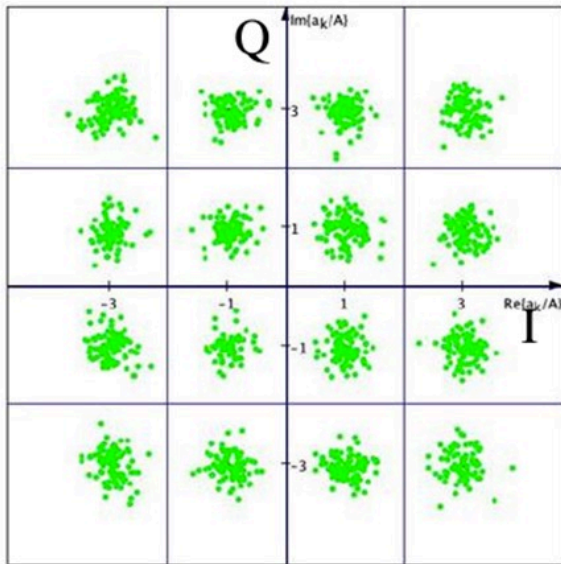
- For different modulation schemes, usually
  - the more the number of constellation symbols,
  - the higher the spectral efficiency,
  - the lower the resilience versus channel noise
- Indeed, the more the symbols, the closer they are in the constellation plane.
- In case of a channel only introducing an Additive White Gaussian Noise (AWNG), the BER for different modulation scheme versus  $E_b/N_0$  is reported in figure:
- $E_b = E_s / \log_2(M)$



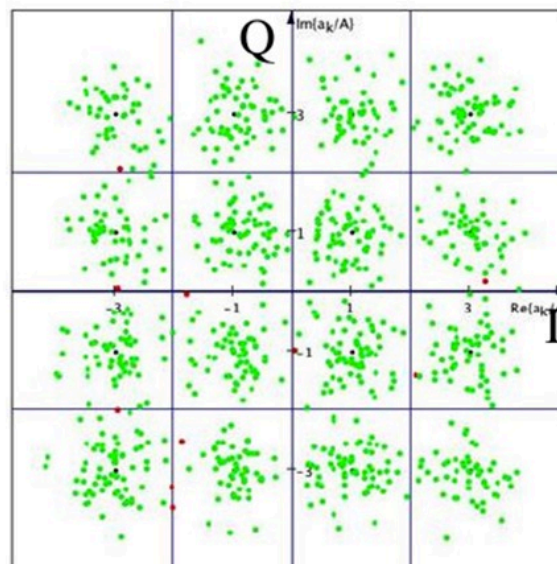
Modulation	Detection Method	Bit Error Rate ( $P_b$ )
BPSK	Coherent	$0.5 \operatorname{erfc}(\sqrt{\frac{E_b}{N_0}})$
QPSK	Coherent	$0.5 \operatorname{erfc}(\sqrt{\frac{E_b}{N_0}})$
$M$ - PSK	Coherent	$\frac{1}{m} \operatorname{erfc}\left(\sqrt{\frac{mE_b}{N_0}} \sin\left(\frac{\pi}{M}\right)\right)$
$M$ - QAM ( $m$ = even)	Coherent	$\frac{2}{m} \left(1 - \frac{1}{\sqrt{M}}\right) \operatorname{erfc}\left(\sqrt{\frac{3mE_b}{2(M-1)N_0}}\right)$
D - BPSK	Non-coherent	$0.5 e^{-\frac{E_b}{N_0}}$
D - QPSK	Non-coherent	$Q_1(a, b) - 0.5 I_0(ab) e^{-0.5(a^2+b^2)}$ where $a = \sqrt{\frac{2E_b}{N_0} \left(1 - \frac{1}{\sqrt{2}}\right)}$ $b = \sqrt{\frac{2E_b}{N_0} \left(1 + \frac{1}{\sqrt{2}}\right)}$ $Q_1(a, b)$ = Marcum Q-function

## Bit Error Rate

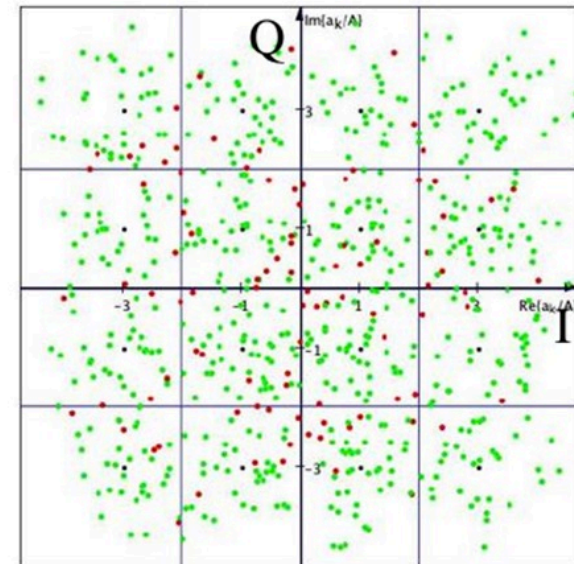
Bit errors in red



16-QAM with  $E_b/N_0 = 11$  dB



16-QAM with  $E_b/N_0 = 6$  dB



16-QAM with  $E_b/N_0 = 2$  dB



## Bit Error Rate

**Ideal Symbol Point**



**Random Noise**



**Phase Noise**



**AM Distortion**



**PM Distortion**



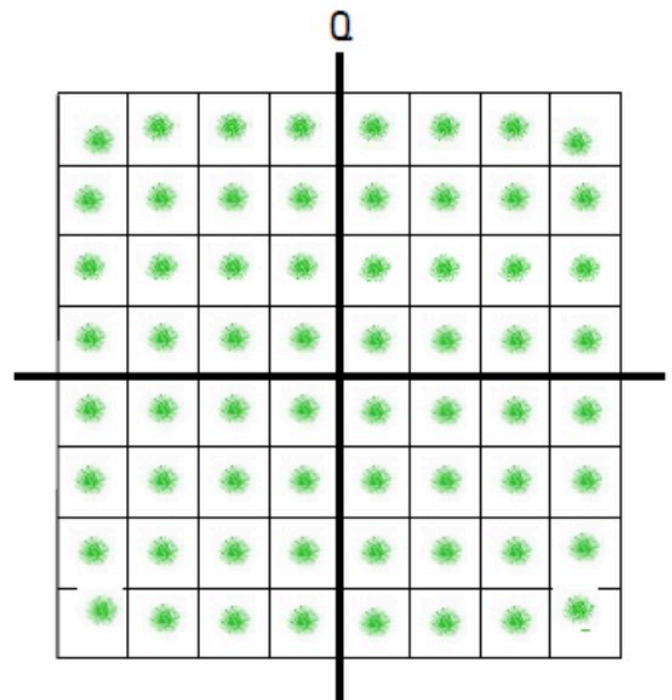
**Delay Distortion/ISI**



**Interference**



## 64 QAM Constellation



## Receiver Sensitivity

- Receiver Sensitivity  $P_{RS}$  is the minimum average received signal power  $P_r$  to satisfy a certain bit error rate (BER)
- For voice transmission, a BER of  $10^{-3}$  is usually used. We should also know under which conditions (additive white Gaussian noise (AWGN) or fading) the BER is evaluated.
- For a given BER we have  $(E_b/N_0)_{min}$
- Then we can compute receiver sensitivity as follow

$$P_{RS} = \left( \frac{E_b}{N_0} \right)_{min} N_0 R_b$$

*Energy per bit multiplied for bits per second returns energy per second i.e. power*

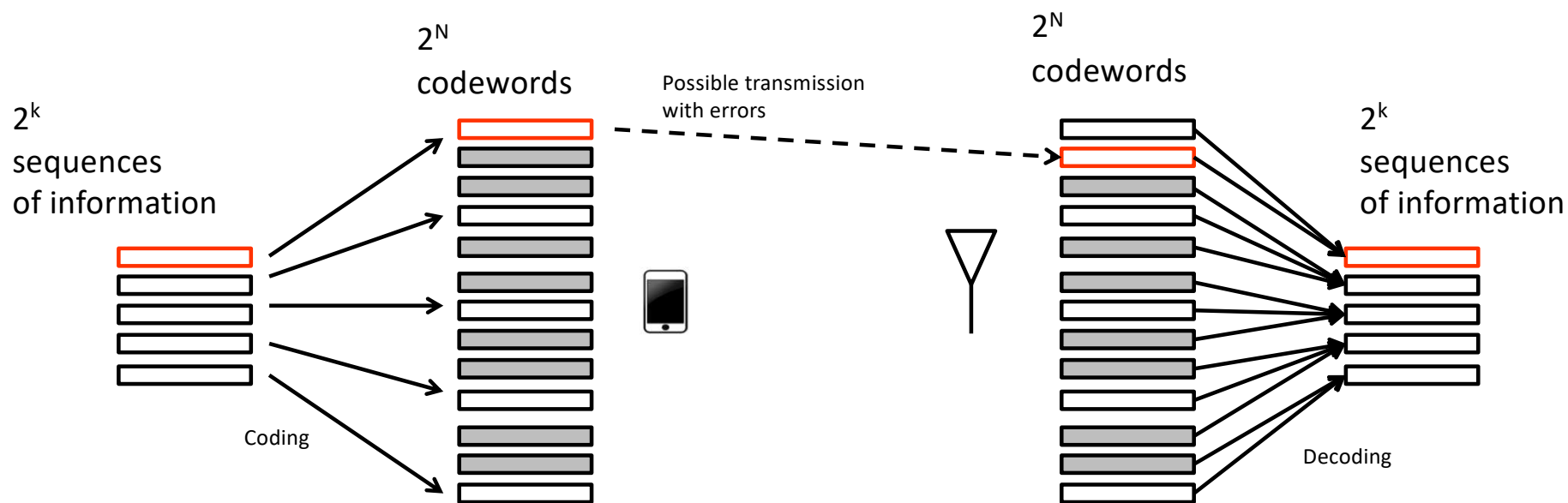
- Where  $R_b$  is the bit-rate
- Reasonable value for 4G/5G cellular systems are in the order of -110 dBm

# ERROR RECOVERY

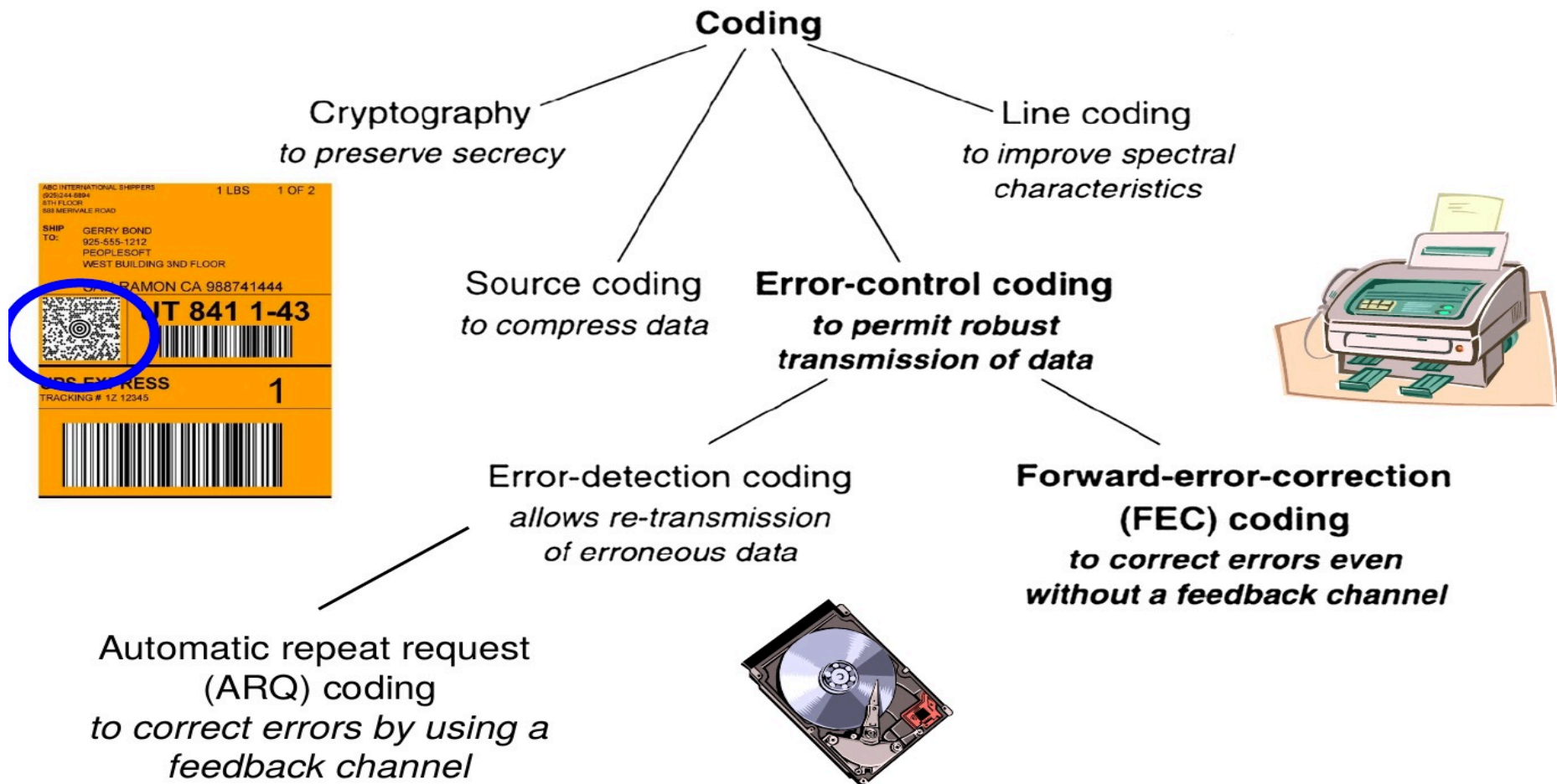
(CHANNEL CODING)

## Forward Error Correction (FEC) – aka Channel coding

- Transmitted information is represented using a codeword that is typically two or three times as long.
- The extra bits supply additional, redundant data that allow the receiver to recover the original information sequence
- E.g.  $K$  bits mapped on a codeword of  $N$  bits with  $N > K$ , coding rate  $K/N$

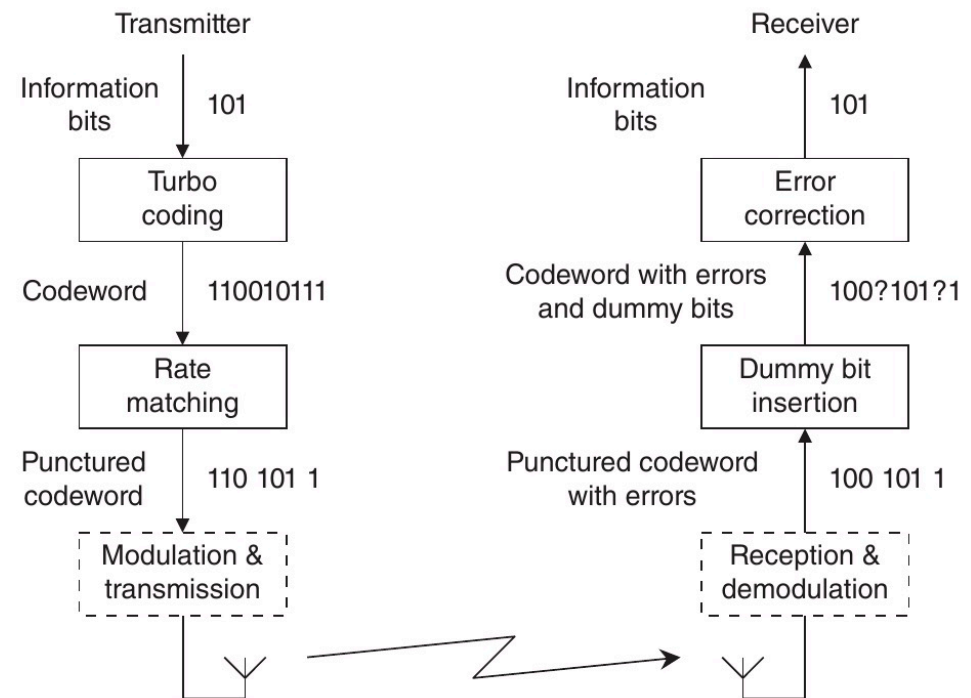


# Coding in Telecommunications



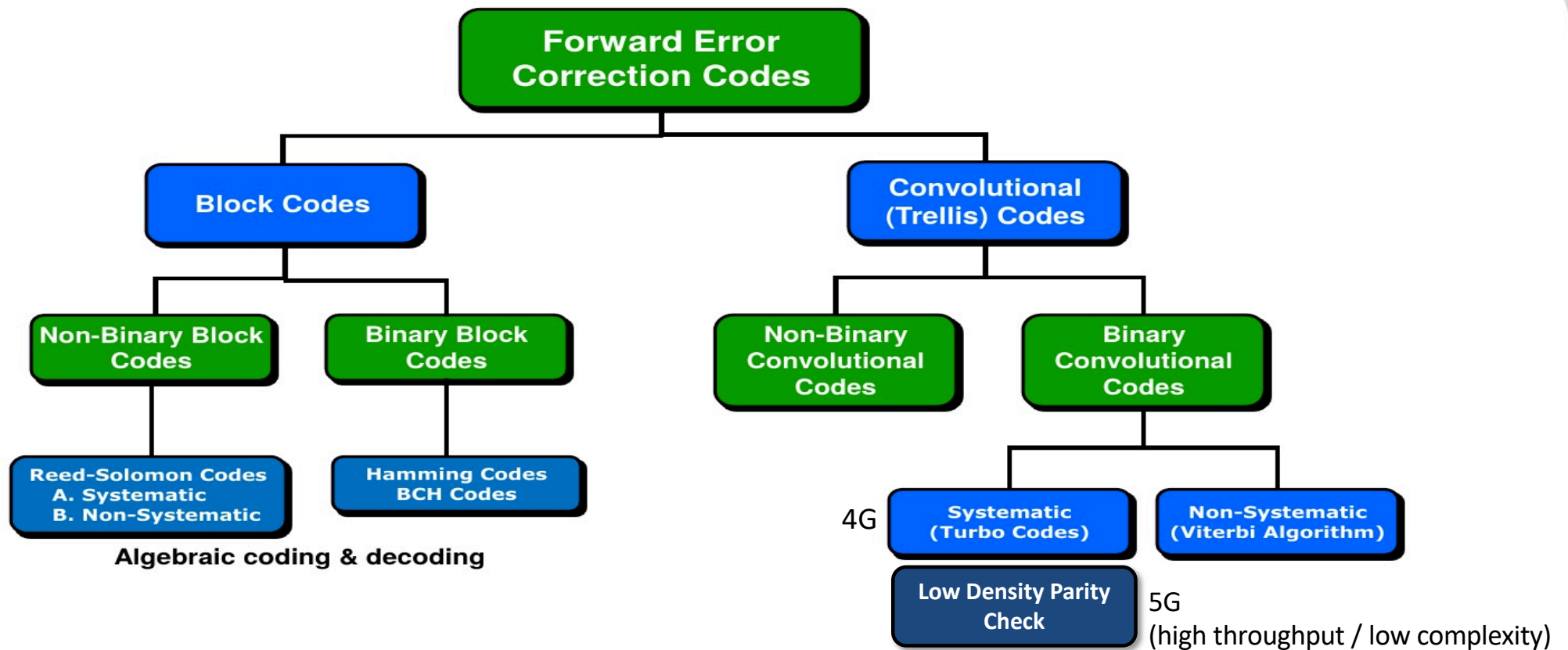
## FEC Rate Matching - Puncturing

- Mismatch between transmission blocks and bits coming out from coding
- Some of the coded bits are selected for transmission, while the others are discarded in a process known as *puncturing*.
- The receiver has a copy of the puncturing algorithm, so it can insert dummy bits at the points where information was discarded.



- Forward Error Correction (FEC) coding techniques are classified as either **convolutional** or **block codes**. The classification depends on the presence or absence of memory.
- A **block code** has *no memory* and works with blocks of bits. Each output codeword of an  $(N,K)$  block code depends only on the current buffer  $K$ .
- A **convolutional coder** has *memory* and work with *stream* of bits. Each bit in the output stream is not only dependent on the current bit, but also on those processed previously. This implies a form of memory.
  - The more the convolution length, the more the decoding delay, the more the coding gain

## FEC Type



Interesting Video on LDPC:

<https://medium.com/5g-nr/ldpc-low-density-parity-check-code-8a4444153934>



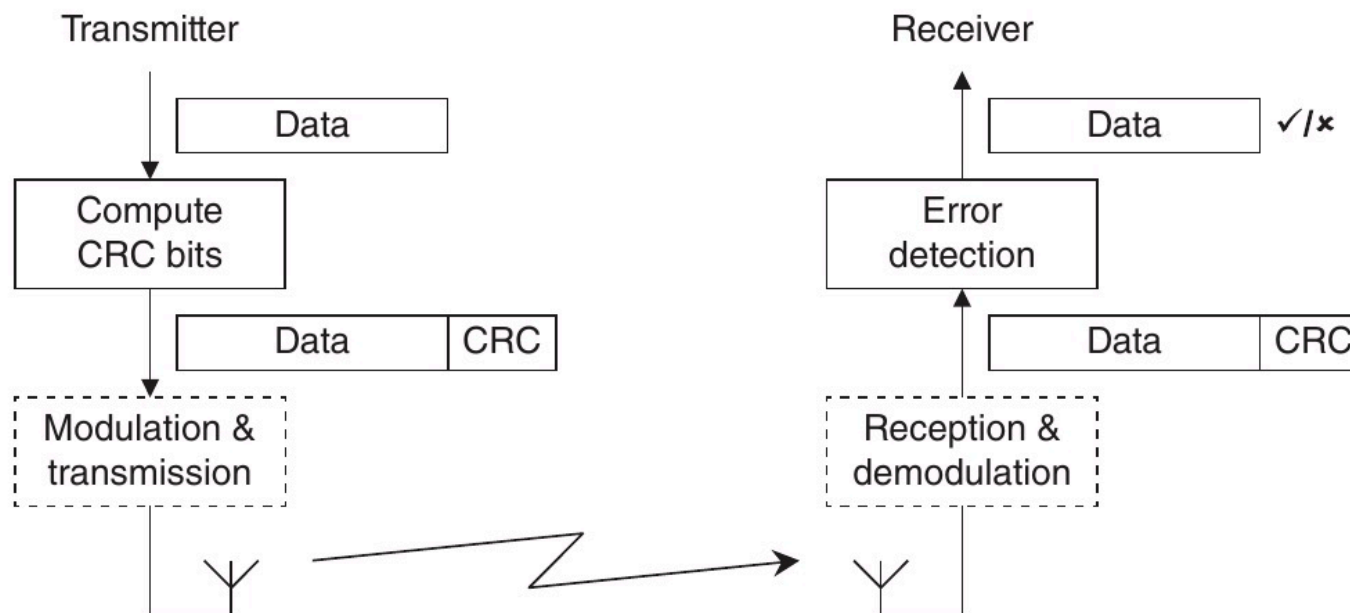
## FEC Multi-rate

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- **Multi-rate services can be achieved changing the FEC rate**
- Changes in the coding rate have a similar effect to changes in the modulation scheme (constellation points).
- Low number of points or low coding rate (many additional bits) implies low bit rate, higher reliability, and vice versa
- If the coding rate is low, then the transmitted data contain many redundant bits. This allows the receiver to correct a large number of errors and to operate successfully at a low received power, but at the expense of a low information rate.

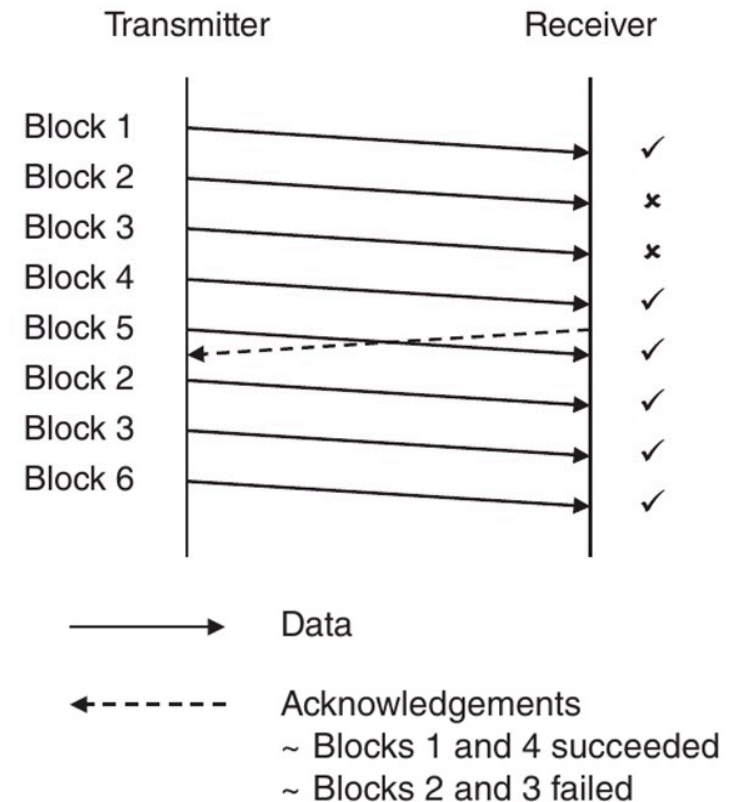
## Automatic Repeat Request (ARQ)

- The transmitter takes a block of information bits and uses them to compute some extra bits that are known as a cyclic redundancy check (CRC), appended to the information block and then transmitted



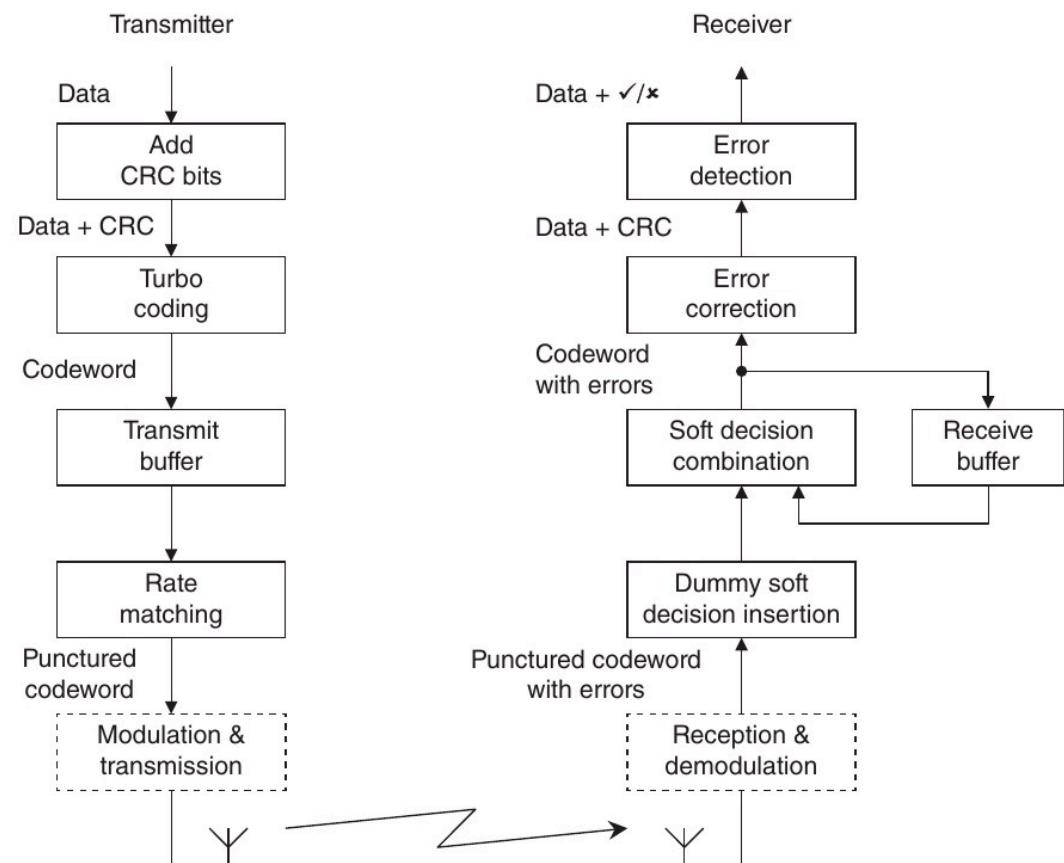
## Automatic Repeat Request (ARQ)

- They are a form of flow control schemes (Stop-and-wait, Go-back-N, Selective-Repeat)
- Retransmissions take time, this technique is only suitable for non-real-time streams such as web pages and emails.
- Selective Repeat (aka selective re-transmission) with cumulative ACK/NACK
  - In case of NACK the transmitter retransmit only the not Acked (NACK) packet



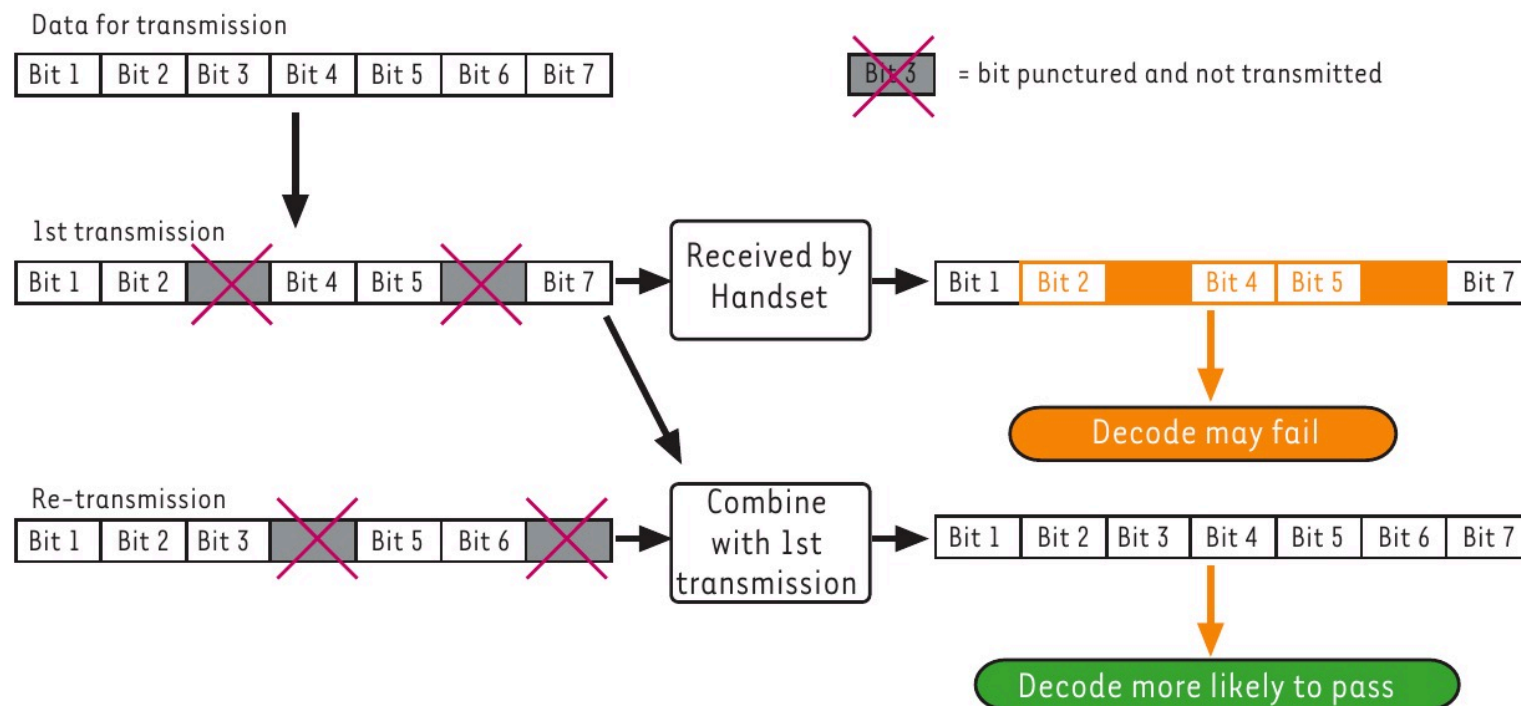
## Hybrid ARQ (HARQ) : ARQ + FEC

- FEC for foreseen error condition + ARQ as a fall-back for unexpected error conditions
- Hard Decision:
  - Decode FEC, check CRC, if error clean received buffer and request retransmission
- Soft Decision:
  - Exploit energy of any re-transmissions
  - In case of error combines re-transmissions in the receive buffer



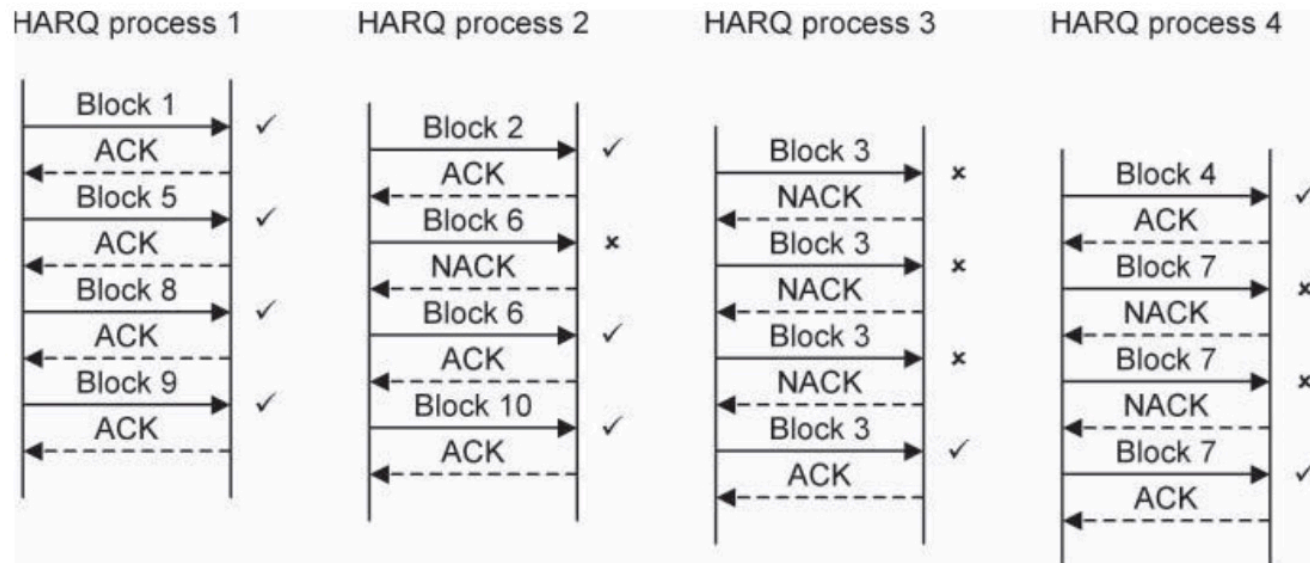
## Soft Decision with Incremental Redundancy

- Incremental Redundancy: a different puncturing pattern for retransmission.



## Parallel HARQ

- Normally, hybrid ARQ uses a stop-and-wait
  - Simple design but transmitter has to pause while waiting for the acknowledgement
- To improve the throughput, the transmitters exchange data amongst several hybrid ARQ parallel processes



# ADAPTIVE MODULATION AND CODING

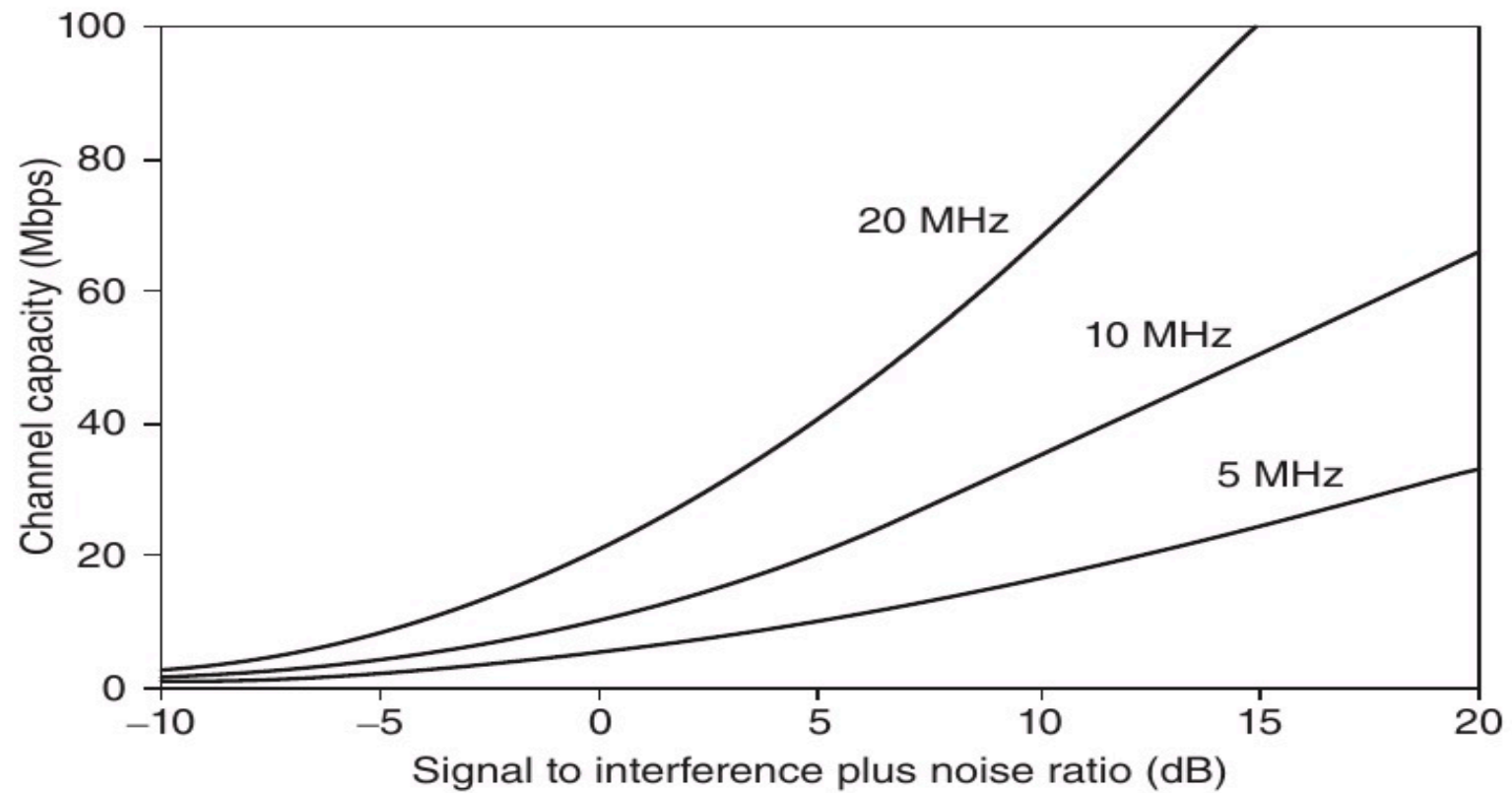
## Shannon–Hartley theorem (1948)

$$C = B \log_2 \left( 1 + \frac{S}{N + I} \right) = B \log_2(1 + SINR)$$

- SINR = Signal to Interference plus Noise Ratio
- AWGN Gaussian noise (N) and interference (I)
- Theoretical limit on data rate
  - $C$  bit/sec,  $B$  (Hz) used channel bandwidth ,  $S$  signal power,  $N+I$  = noise + interference power
- Power-Bandwidth exchange
  - Need to transmit  $C$  bit/s
  - Limited bandwidth --> increase the power
  - Limited power --> increase the used channel bandwidth  $B$ , e.g. with FEC
  - Increase bandwidth is more effective than power

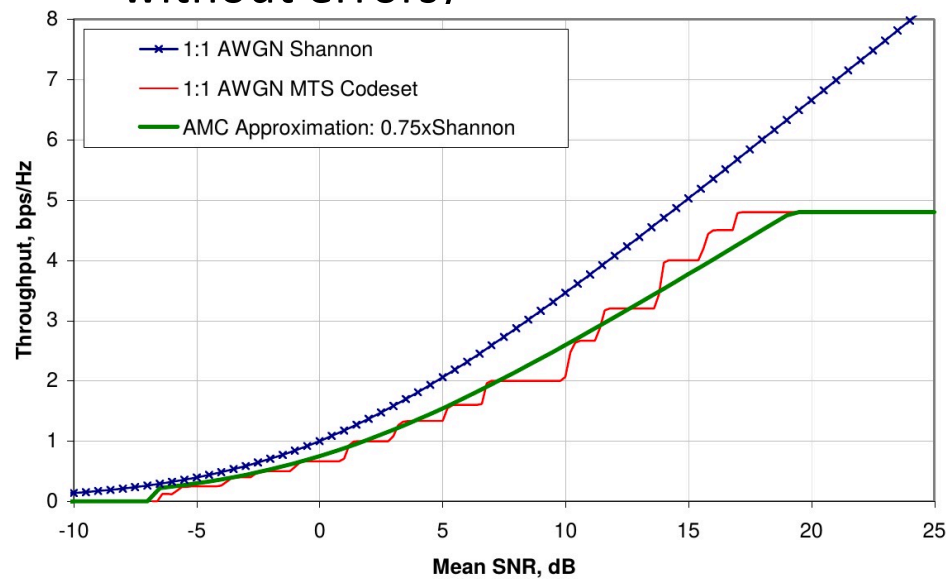


## Shannon–Hartley theorem (1948)



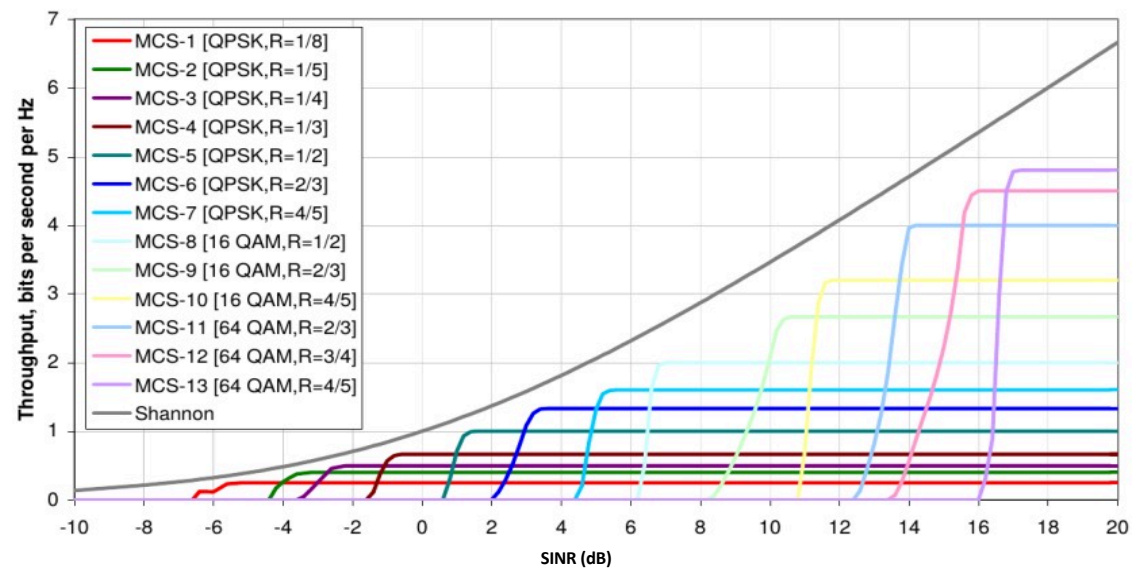
## Adaptive modulation and coding

- Modern wireless systems dynamically adapt the modulation and the coding rate to the channel condition in order to reach the maximum throughput (bits received without errors)



$$C = \alpha B \log_2(1 + SINR)$$

**Alpha-Shannon:** approximates the result of Adaptive modulation and coding with a linear formula. Alpha value close to 0.75



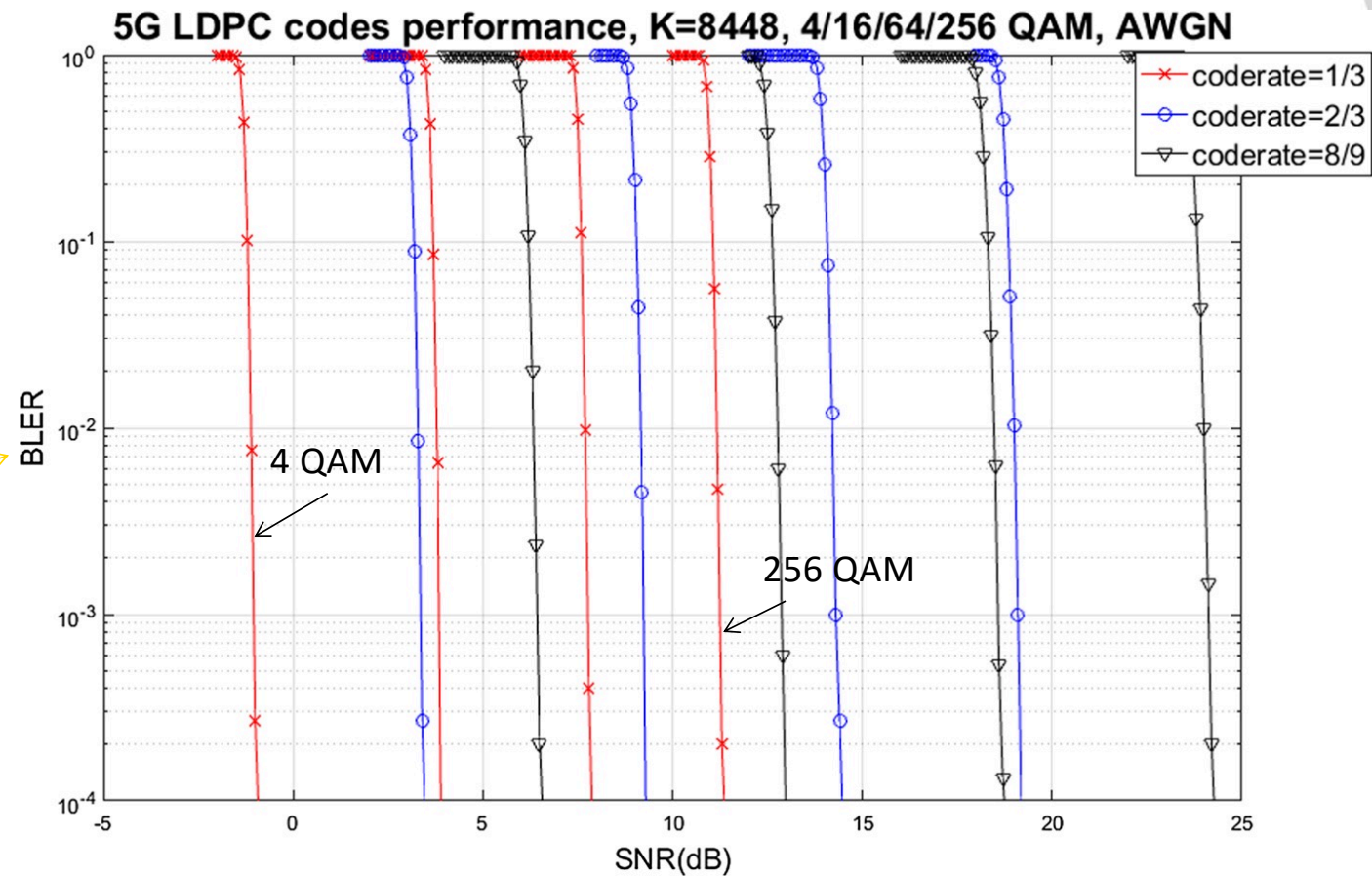
## Adaptive modulation and coding scheme : Lookup Table

**Table 7.1** Lookup table for mapping SINR estimate to modulation scheme and coding rate

CQI index	Modulation	Coding rate	Spectral efficiency (bps/Hz)	SINR estimate (dB)
1	QPSK	0.0762	0.1523	-6.7
2	QPSK	0.1172	0.2344	-4.7
3	QPSK	0.1885	0.3770	-2.3
4	QPSK	0.3008	0.6016	0.2
5	QPSK	0.4385	0.8770	2.4
6	QPSK	0.5879	1.1758	4.3
7	16QAM	0.3691	1.4766	5.9
8	16QAM	0.4785	1.9141	8.1
9	16QAM	0.6016	2.4063	10.3
10	64QAM	0.4551	2.7305	11.7
11	64QAM	0.5537	3.3223	14.1
12	64QAM	0.6504	3.9023	16.3
13	64QAM	0.7539	4.5234	18.7
14	64QAM	0.8525	5.1152	21.0
15	64QAM	0.9258	5.5547	22.7

## Adaptive modulation and coding scheme : BER

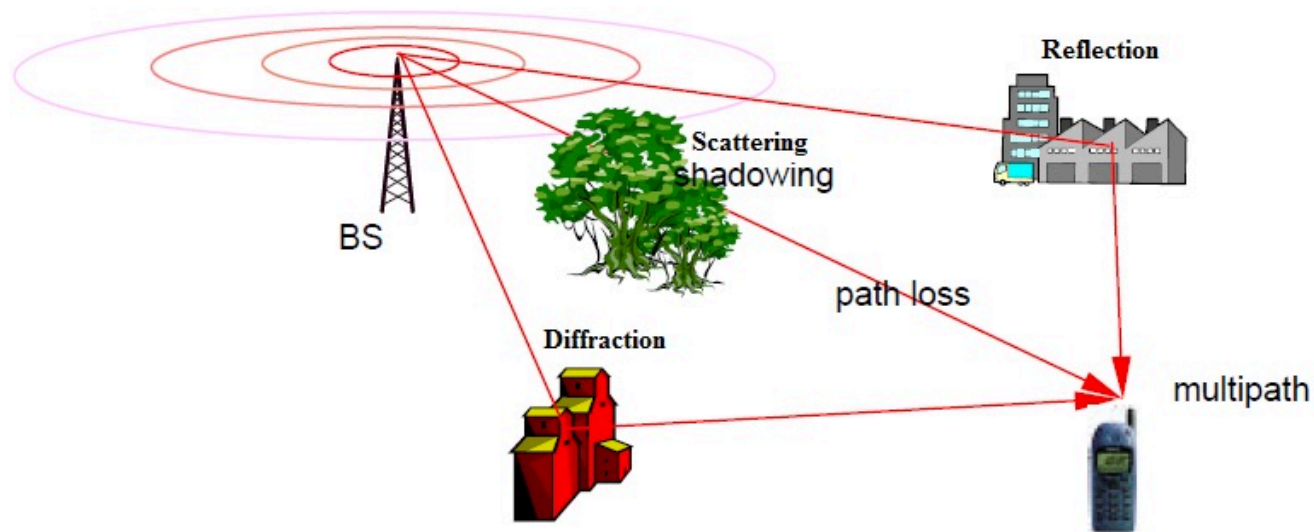
- In mobile networks transmissions are made in blocks of symbols
- BLER : Block Error Rate
  - *Residual* BLER: BLER after decoding



# RADIO PROPAGATION AND FADING

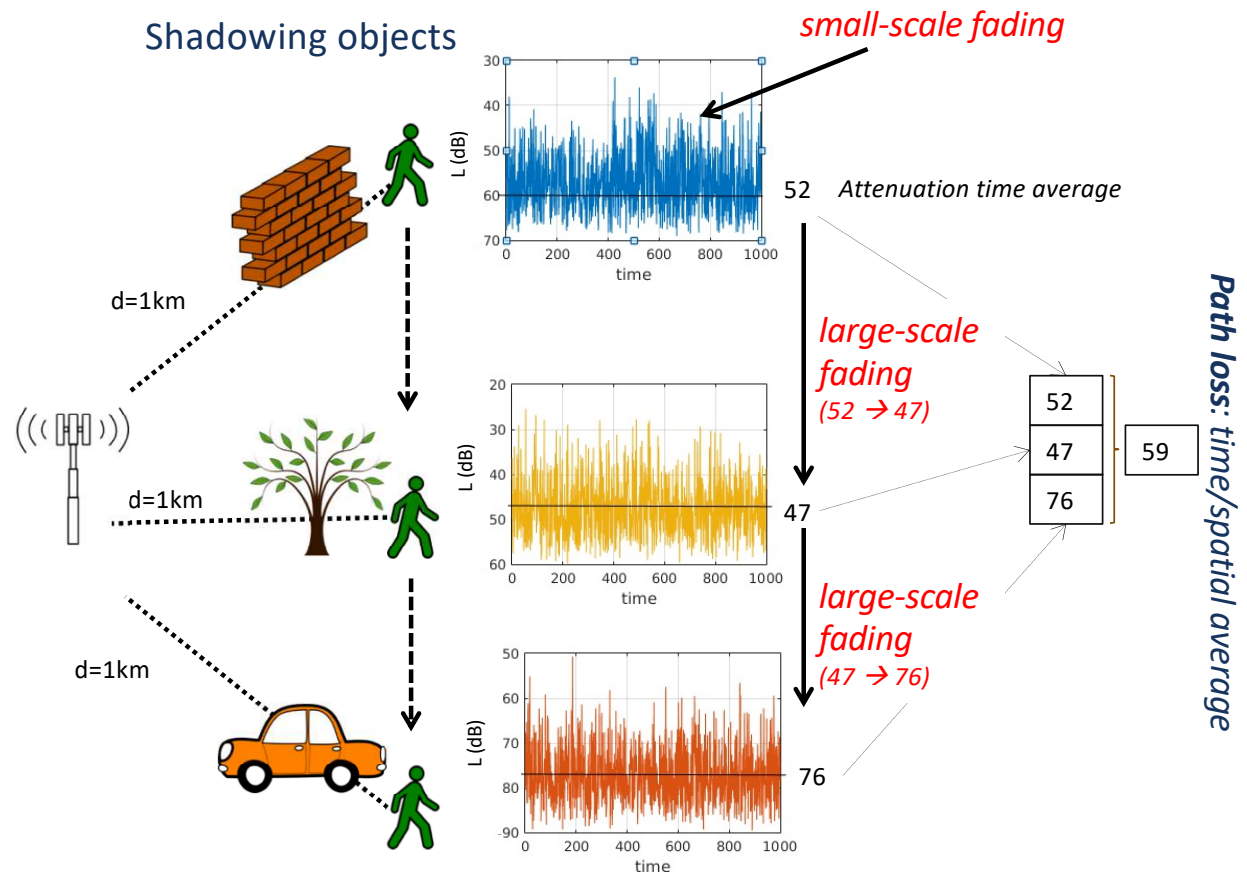
# Radio Propagation

- Radio propagation is the behavior of radio waves when they are transmitted, or propagated from one point on the Earth to another
- Fundamental for planning purposes understanding how signal power is reduced



## Path loss and fading

- **Path Loss** : average attenuation (time and space) of the signal power during its propagation at a given distance
- **Fading**: additional random attenuation versus time  $t$  and distance  $d$  from the transmitter
- Two kinds of fading
  - **Small-scale** fading
  - **Large-scale** fading





## Modeling path loss and fading

$$L(d, t)_{dB} = PathLoss(d)_{dB} + \chi_{\sigma}_{dB} + \theta_{dB} \quad [dB]$$

Attenuation at distance  $d$  and time  $t$  can be modeled as a **random variable**

Average value of the attenuation over space and time

Zero mean random variable modeling large scale fading

Zero mean random variable modeling small scale fading

$$x[dB] = 10\log_{10}(x[\text{linear scalar}])$$

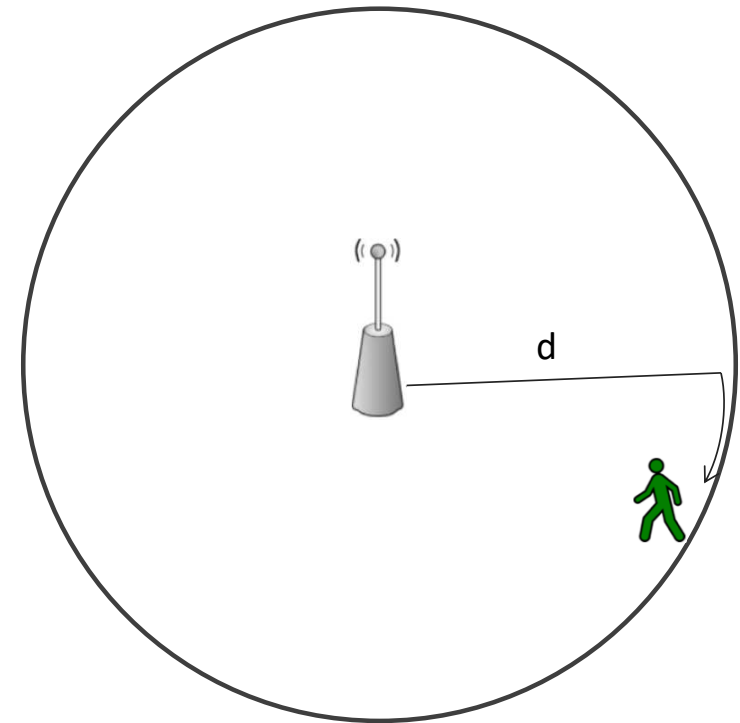
$$x[dBm] = 10\log_{10}(x[mW])$$

dB and dBm representation useful because  
multiplications/divisions becomes sums/subtractions



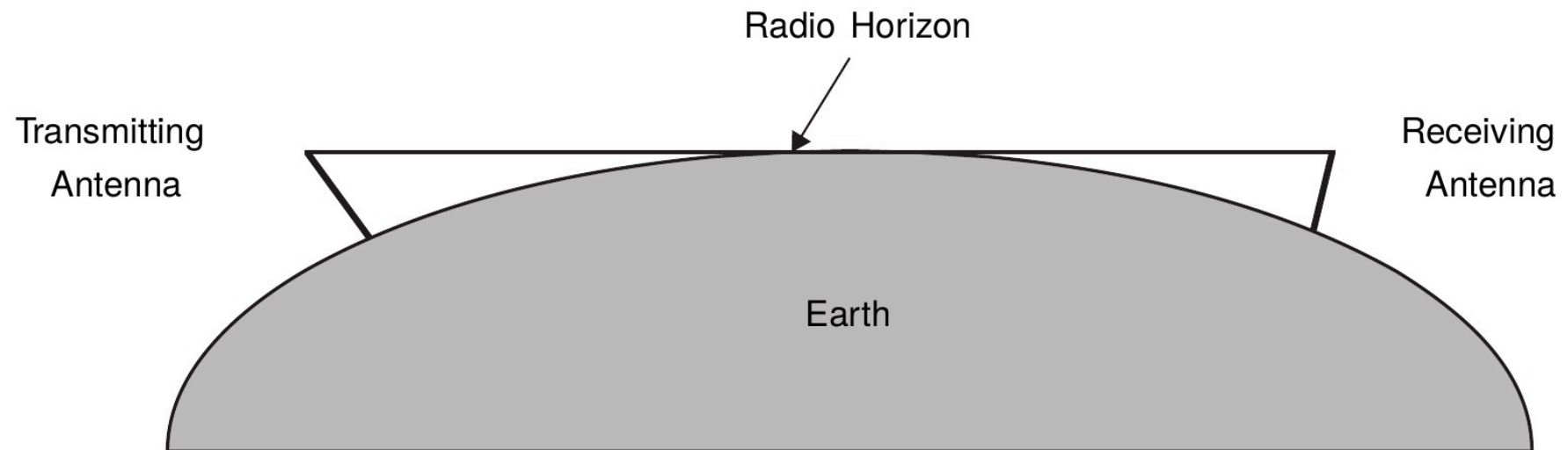
## Path-Loss models

- Path-loss is the **average** attenuation of the signal power due to the radio propagation
- It is a function of the distance  $d$  and it is a **space-time average** value: average of the time averages gathered at different locations at distance  $d$
- Some path-loss models were derived by statistical measurements (aka empirical model) and others were derived by analytical representation of the propagation environment



## Free-Space Path Loss model

- Free space is the simplest Line of Sight (LOS) path loss case



## Free-Space Path Loss

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- 6 dB for path length duplication
- 6 dB for frequency duplication

$$(L_p)_{\text{free}} = 32.44 + 20\log f + 20\log d \text{ dB}$$

where:

$f$  = carrier frequency in MHz

$d$  = separation distance in km ( $> 1$  km)

## Log-distance Path Loss model

$$L_p = \overline{L_p}(d_0) + 10 \gamma \log\left(\frac{d}{d_0}\right) \text{ dB}$$

$d_0$  = reference distance for which the path loss  $\overline{L_p}(d_0)$  is known (dB)

$\gamma$  = path-loss exponent, variable comprised between 2 and 6 :

Environment	Path-loss Exponent
Free-space	2
Urban area cellular radio	2.7-3.5
Shadowed urban cellular radio	3-5
In building LOS	1.6 to 1.8
Obstructed in building	4 to 6
Obstructed in factories	2 to 3

## Okumura/Hata Path Loss model

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- Model for cellular system
- Okumura analyzed path-loss characteristics based on a large amount of experimental data collected around Tokyo
- Hata derived empirical formulas for the median path loss ( $L_{50}$ ) to fit Okumura curves. Hata's equations are classified into three models
- Typical Urban:

$$L_{50}(\text{urban}) = 69.55 + 26.16 \log f_c + (44.9 - 6.55 \log h_b) \log d \\ - 13.82 \log h_b - a(h_m)(\text{dB})$$

## Okomura/Hata – Path Loss

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- $a(h_m)$  = correction factor (dB) for mobile antenna height as given by:

For large cities

$$a(h_m) = 8.29[\log(1.54h_m)]^2 - 11 \quad f_c \leq 200 \text{ MHz}$$

$$a(h_m) = 3.2[\log(11.75h_m)]^2 - 4.97 \quad f_c \geq 400 \text{ MHz}$$

For small and medium-sized cities

$$a(h_m) = [1.1 \log(f_c) - 0.7]h_m - [1.56 \log(f_c) - 0.8]$$

## Okomura/Hata – Path Loss

### *Typical Suburban*

$$L_{50} = L_{50}(\text{urban}) - 2 \left[ \log \left( \frac{f_c}{28} \right)^2 - 5.4 \right] \text{dB}$$

### *Rural*

$$L_{50} = L_{50}(\text{urban}) - 4.78(\log f_c)^2 + 18.33 \log f_c - 40.94 \text{ dB}$$

where:

$f_c$  = carrier frequency (MHz)

$d$  = distance between base station and mobile (km)

$h_b$  = base station antenna height (m)

$h_m$  = mobile antenna height (m)

The range of parameters for which the Hata model is valid is:

$$150 \leq f_c \leq 2200 \text{ MHz}$$

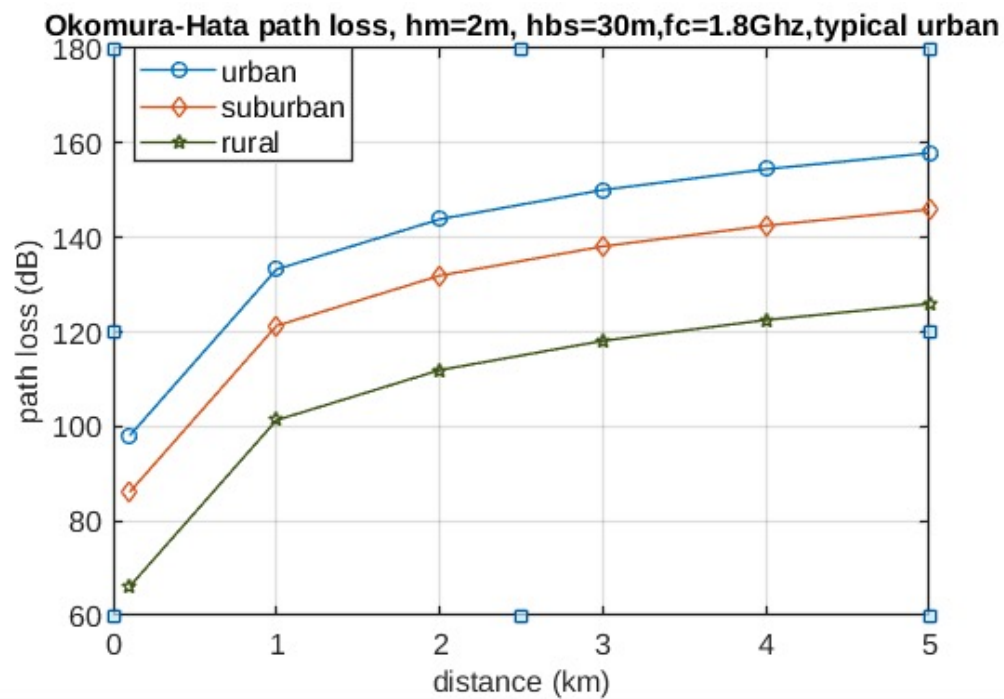
$$30 \leq h_b \leq 200 \text{ m}$$

$$1 \leq h_m \leq 10 \text{ m}$$

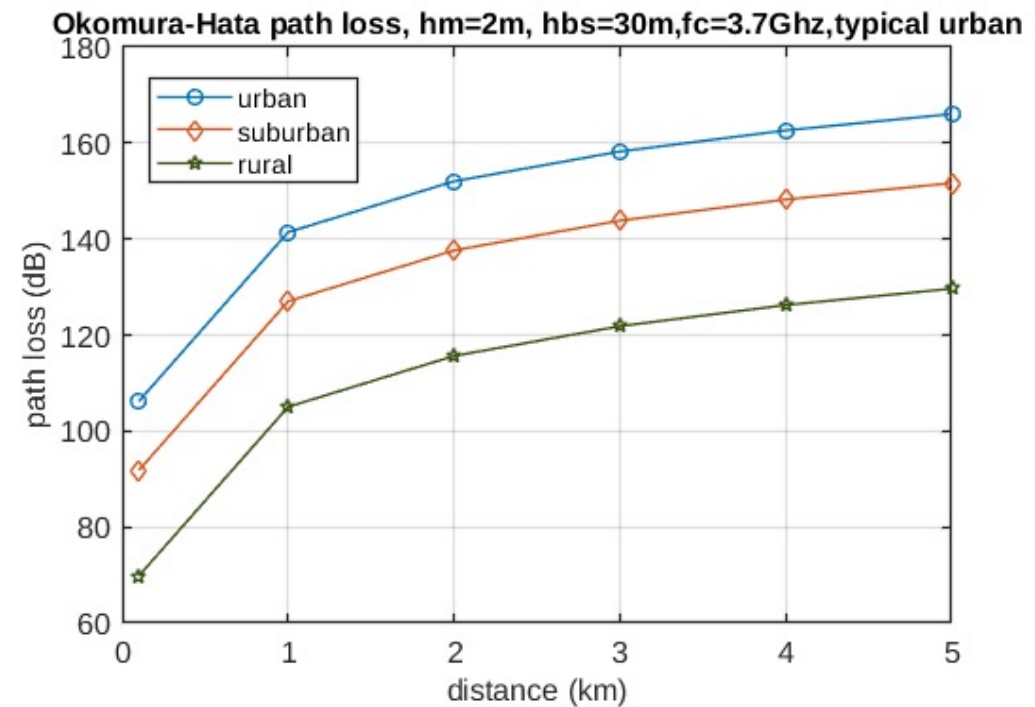
$$1 \leq d \leq 20 \text{ km}$$

## Okomura/Hata Path Loss model

Higher Frequencies, Higher attenuation



LTE (1.8GHz)



5G (3.7GHz)



# LARGE SCALE FADING

(SHADOWING, LONG-TERM FADING)

Sklar, Bernard. "Rayleigh fading channels in mobile digital communication systems. I. Characterization." Communications Magazine, IEEE 35.7 (1997): 90-100.

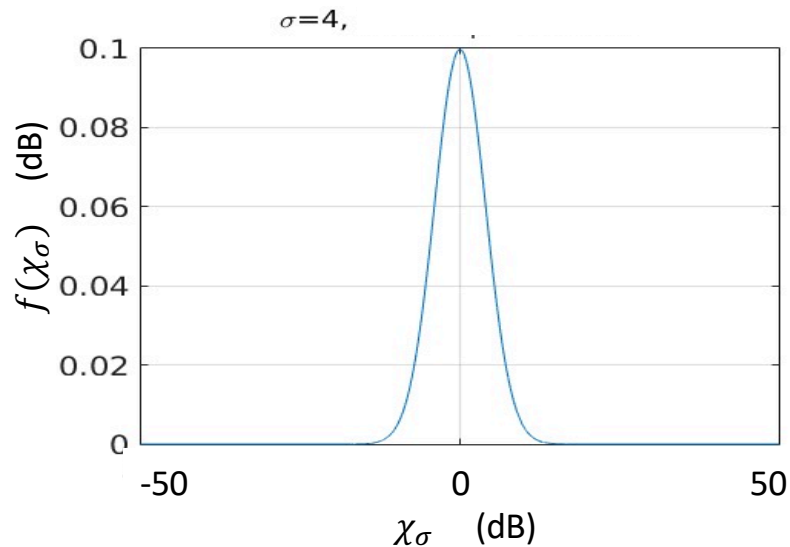
Sklar, Bernard. "Rayleigh fading channels in mobile digital communication systems. II. Mitigation." Communications Magazine, IEEE 35.7 (1997): 90-100.

- [illegible]

## Modeling large-scale fading

- Empirical studies have shown a good fit with log-normal distribution
- Log-normal means that the pdf in the logarithmic scale (i.e. dB) is a Gaussian
- $\sigma$  depends on the environment
  - Indoor 10-12 dB
  - Outdoor 6-8 dB

$$L(d, t)_{dB} = PathLoss(d)_{dB} + \chi_{\sigma} dB + \theta_{dB} \quad [dB]$$



$$f(\chi_{\sigma}) = \frac{1}{\sigma\sqrt{2\pi}} \exp\left[-\frac{\chi_{\sigma}^2}{2\sigma^2}\right]$$

# SMALL SCALE FADING

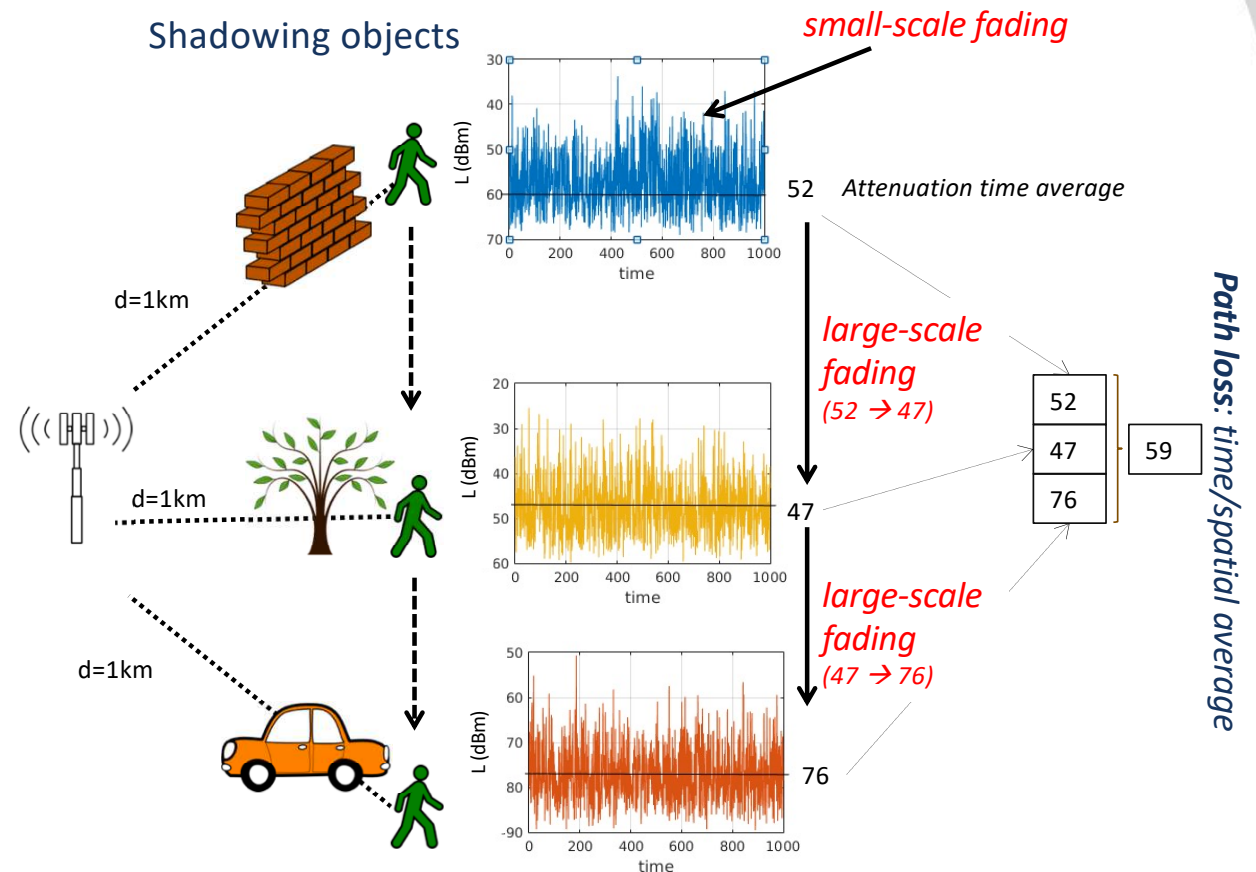
(SHORT-TERM , MULTIPATH, RAYLEIGH FADING)

Sklar, Bernard. "Rayleigh fading channels in mobile digital communication systems. I. Characterization." Communications Magazine, IEEE 35.7 (1997): 90-100.

Sklar, Bernard. "Rayleigh fading channels in mobile digital communication systems. II. Mitigation." Communications Magazine, IEEE 35.7 (1997): 90-100.

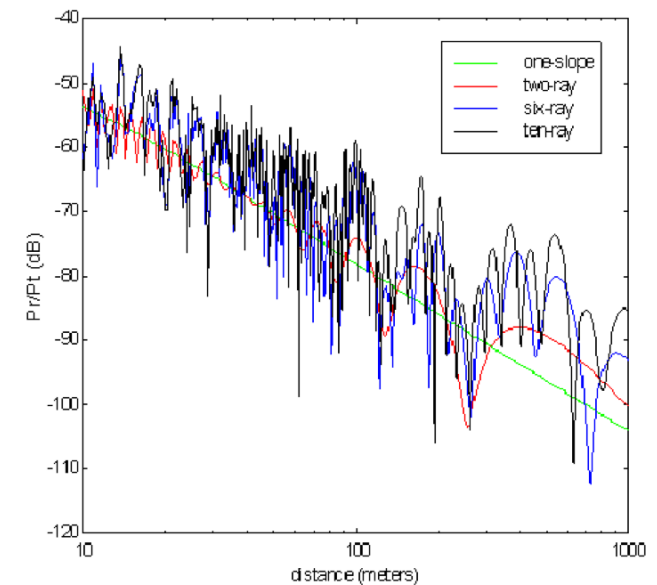
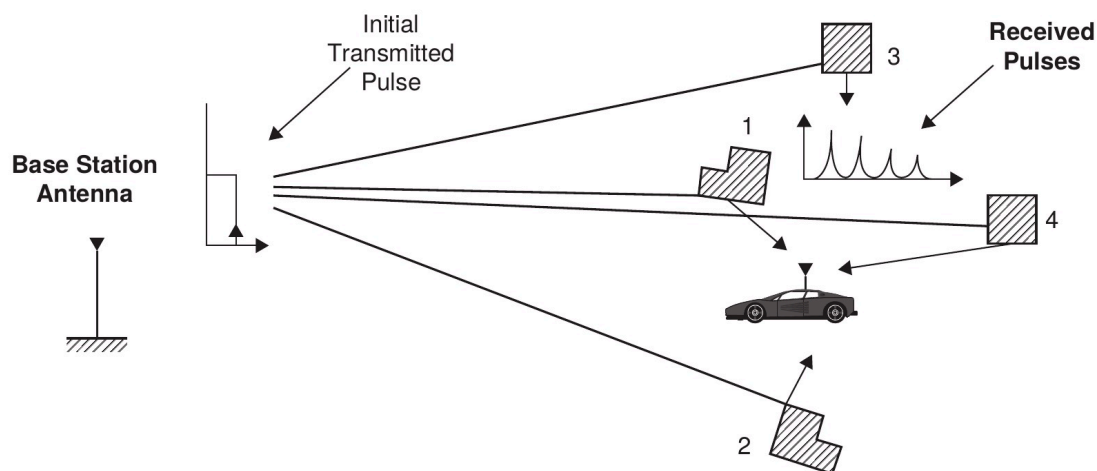
## Small-scale fading

- Dramatic random fluctuation (20,30 dB) of the attenuation around the path loss value due to **microscopic movements** (small scale) of the environment
  - Head movements
  - Hands movements
  - Etc.
- Small-scale fading is mainly due to **multipath phenomenon** whose propagation condition can completely change with movements of an half of a wavelength
  - @ 2Ghz it means 7.5cm
- In the example, the path loss @ 1km behind a wall is 52 dB, considering also the *shadowing* (large scale fading) provided by the wall. The user moves a bit while using her phone and this creates rapid fluctuation of the attenuation



## Multipath propagation

- Signal can take many different paths between sender and receiver due to reflection, scattering, diffraction
- Reflected signals arrive at the receiver with a random phase offset. Their combinations can reduce or amplify average signal power



Ray tracing plots of received signal power indicator  $20 \log |\sum_{i=0}^N p_i|$  as a function of  $\log d_0$  for  $N \in \{0, 2, 3, 5\}$  for a typical suburban case with street width of 20 feet, and average distance from street to home of  $w_t = 10$  feet (so  $w_s = 40$  feet)

Source: <http://morse.colorado.edu/~tlen5510/text/classwebch3.html>

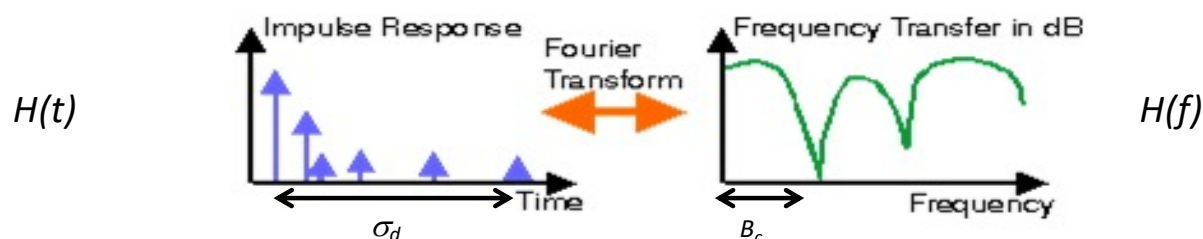
## Modeling small-scale fading

$$L(d, t)_{dB} = PathLoss(d)_{dB} + \chi_{\sigma} dB + \theta_{dB} [dB]$$

- The rapid variations (small-scale) in signal envelope caused by multipath is modeled by
  - **Rayleigh** distribution in absence of LOS
  - **Rician** distribution in presence of a LOS ray
  
- In addition to the fluctuation of the attenuation, multipath determines two other channel characteristics which have a dimensioning impact
  - **Delay spread**
  - **Coherence time**

## Delay spread

- Multipath creates time spreading of a signal

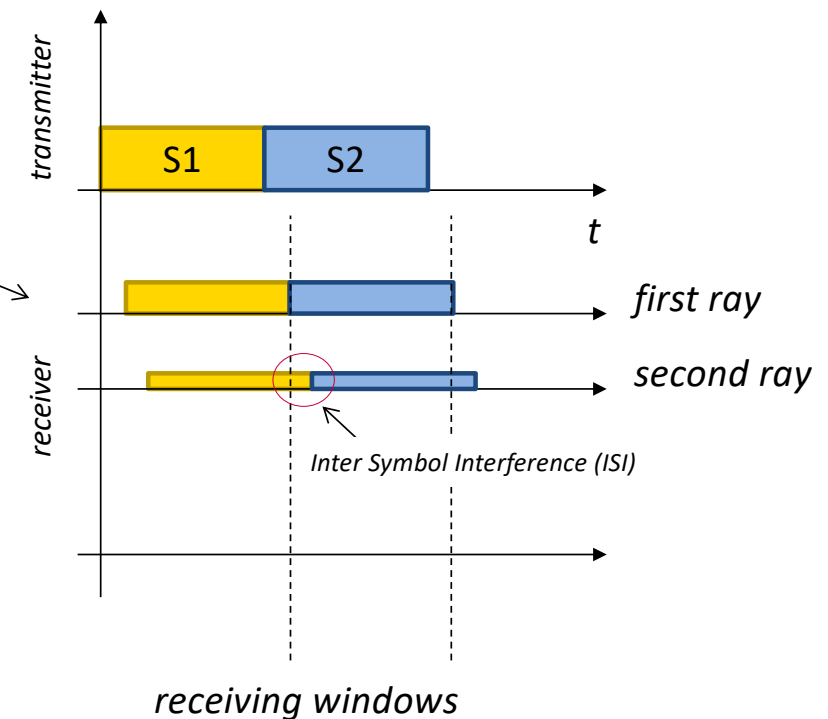
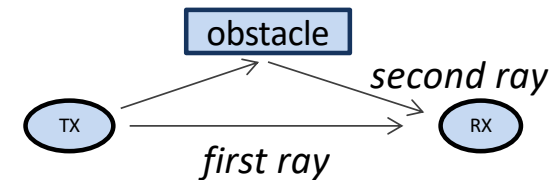
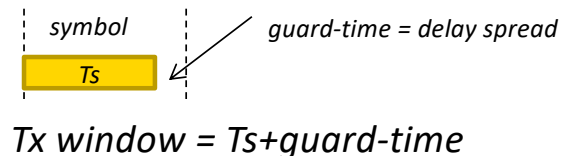


- **Delay spread:** total time interval during which reflections with significant energy arrive. It is usually measured through its root-mean-square (rms)  $\sigma_d$ .
- Multipath profile can be represented through the channel impulse response
- In the frequency domain, delay spread provides frequency selective fading
- **Coherence bandwidth** ( $B_c$ ): the frequency band over which the attenuation remains constant  $B_c \approx \frac{1}{5\sigma_d}$ 
  - If modulation bandwidth  $\gg B_c$  the channel is said to be a **flat fading channel**
  - Otherwise we have a **frequency selective fading channel**, equalization needed to limit distortion



## Delay spread

- In macro-cellular mobile radio, the rms delay spread  $\sigma_d$  is in the range from 100 nsec to 10 microsec.
- In indoor and micro-cellular channels, the delay spread is usually smaller
  - E.g. reported delay spreads in four European cities are less than 8 microsec in macro-cellular channels, less than 2 microsec in micro-cellular channels, between 50, 300 nsec in pico-cellular/indoor channels
- Delay spread creates the issue of Inter Symbol Interference (ISI)
- To avoid ISI a possible solution is to introduce **guard-times**, i.e. void transmission periods after the symbol, whose duration is equal to the delay spread.
- Guard times is an overhead that reduces the achievable bit rate. The overhead is limited if symbol duration is much greater than the guard time, i.e. the delay spread



## Coherence time

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- Channel Impulse Response  $H$  changes over time as a result of user's mobility, which changes the multipath conditions
- Coherence time is defined as the  $T_c$  time interval over which the channel is essentially invariant
- When symbol duration  $T_s < T_c$  we have a slow-fading channel
- When symbol duration  $T_s > T_c$ , the channel varies during a symbol transmission and we have a fast-fading channel
- Fast-fading condition are very difficult to be faced and often leads to a irreducible error rate; however, most of modern wireless system has  $T_s < T_c$
- Note: many books and literature papers wrongly call slow-fading the shadowing/large-scale fading, and fast-fading the multipath/small-scale fading. But being fast or slow only depends on the relationship between symbol duration and coherence time.

## Coherence time and doppler effect

- Intuitive considerations about what impacts the coherence time  $T_c$ :
  - the higher the user speed is, the shorter is  $T_c$ , because user is changing more rapidly the environment around him
  - the higher the carrier frequency ( $f_c$ ) is, the shorter is  $T_c$ , because it is lower the space to be covered with a given speed to change completely the propagation conditions
- Time changing of the channel impulse response can be modeled with the space-time correlation function  $R(\Delta T)$ , i.e. the autocorrelation function of the channel response time
- If channel does not vary then  $R(\Delta T)$  is a constant, e.g. equal to 1
  - For  $\Delta T < T_c$ ,  $R(\Delta T)$  is about constant
- If channel vary after  $\Delta T^*$  seconds, then  $R(\Delta T^*)$  is less than one and the more is the un-correlation (i.e. amount of change), the lower the value of  $R(\Delta T^*)$
- In the frequency domain, the Fourier transform of  $R(\Delta T^*)$  is the Doppler Power Spectral Density, which provides an idea about how the energy of a sinusoid is spread in the frequency domain
- Max Doppler spread  $f_d = \text{speed } (v) / \text{wavelength } (\lambda)$
- There is an important time-frequency relationship:

$$T_c \approx \frac{1}{f_d} = \frac{\lambda}{v} = \frac{c}{vf_c}$$

## Comfort zone of cellular systems

Limited  
guard-time  
overhead

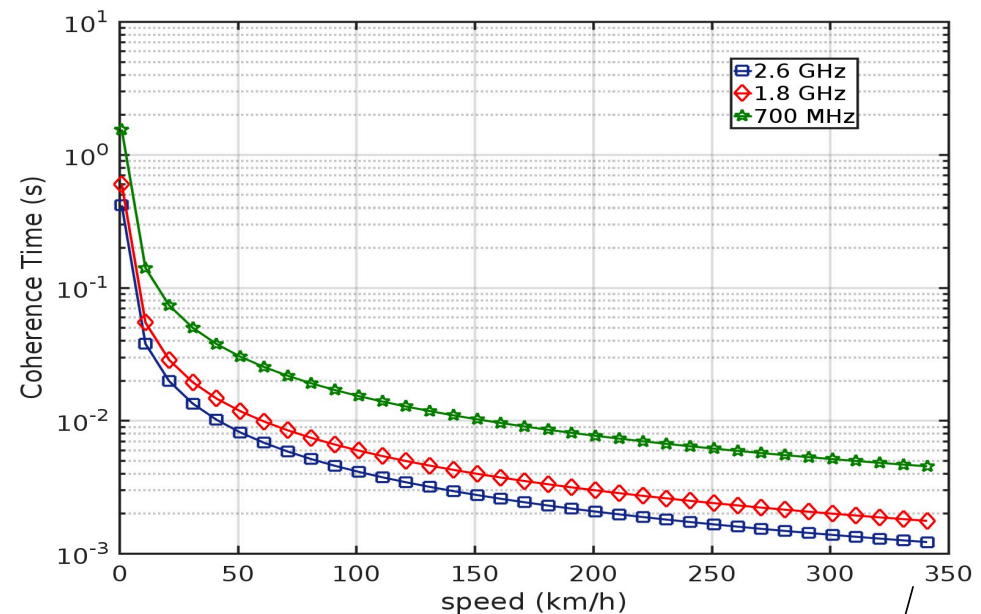
Slow-fading  
condition

$$\sigma_d \ll T_s \ll T_c$$

e.g. delay spread  
10 microsec

e.g. Coherence  
time 1 ms

e.g. Symbol duration 0.667 ms  
(LTE/5g @ 15kHz scs)



A system designed for a mobility up to 350 km/h must have a symbol duration much lower than this value (1.12 ms)