



Edge-based methods

- Edge-based methods is based on contour detection: their weakness in connecting together broken contour lines make them, too, prone to failure in the presence of blurring.



(a) Prewitt · Operator

-1	0	1
-1	0	1
-1	0	1

-1	-1	-1
0	0	0
1	1	1

(b) Sobel · Operator

-1	0	1
-2	0	2
-1	0	1

-1	-2	-1
0	0	0
1	2	1

Derivative Images (Edges)



(a) 原画像



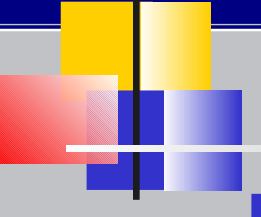
(b) ロバーツ・オペレータ



(c) プリューウィット・オペレータ



(d) ソーベル・オペレータ



A region-based method usually proceeds as follows: the image is partitioned into connected regions by grouping neighboring pixels of similar intensity levels. Adjacent regions are then merged under some criterion involving perhaps homogeneity or sharpness of region boundaries.

Connectivity-preserving relaxation method



- The main idea in connectivity-preserving relaxation-based segmentation method, is to start with some initial boundary shape represented in the form of spline curves, and iteratively modify it by applying various shrink/expansion operations according to some energy function.

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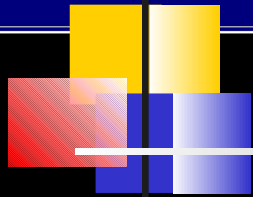
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Exaggerated view
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Speech Coding



- Speech coding is a technique sometimes referred to as **lossy coding**.
- The **input** and **output** signals are not mathematically equivalent but they are **perceptually similar**.
- Differences can be heard, but are hopefully not annoying or are acceptable for the application. Traditionally speech coding is used for communication applications using telephony bandwidth speech (**200 Hz - 3.5 kHz**).
- However, changes in the communication infra-structure have opened the door for new exciting algorithms targeting all types of bandwidths from 3.5 kHz all the way up to CD quality sound.



Speech Coders

Designing speech coders is a balancing game between

- Quality,
- bit rate,
- delay and complexity.

The quality is a function of the bit rate. For telephone quality speech the standard is 8 bits mulaw per sample. Using a 8 kHz sampling rate this results in 64 kb/s.

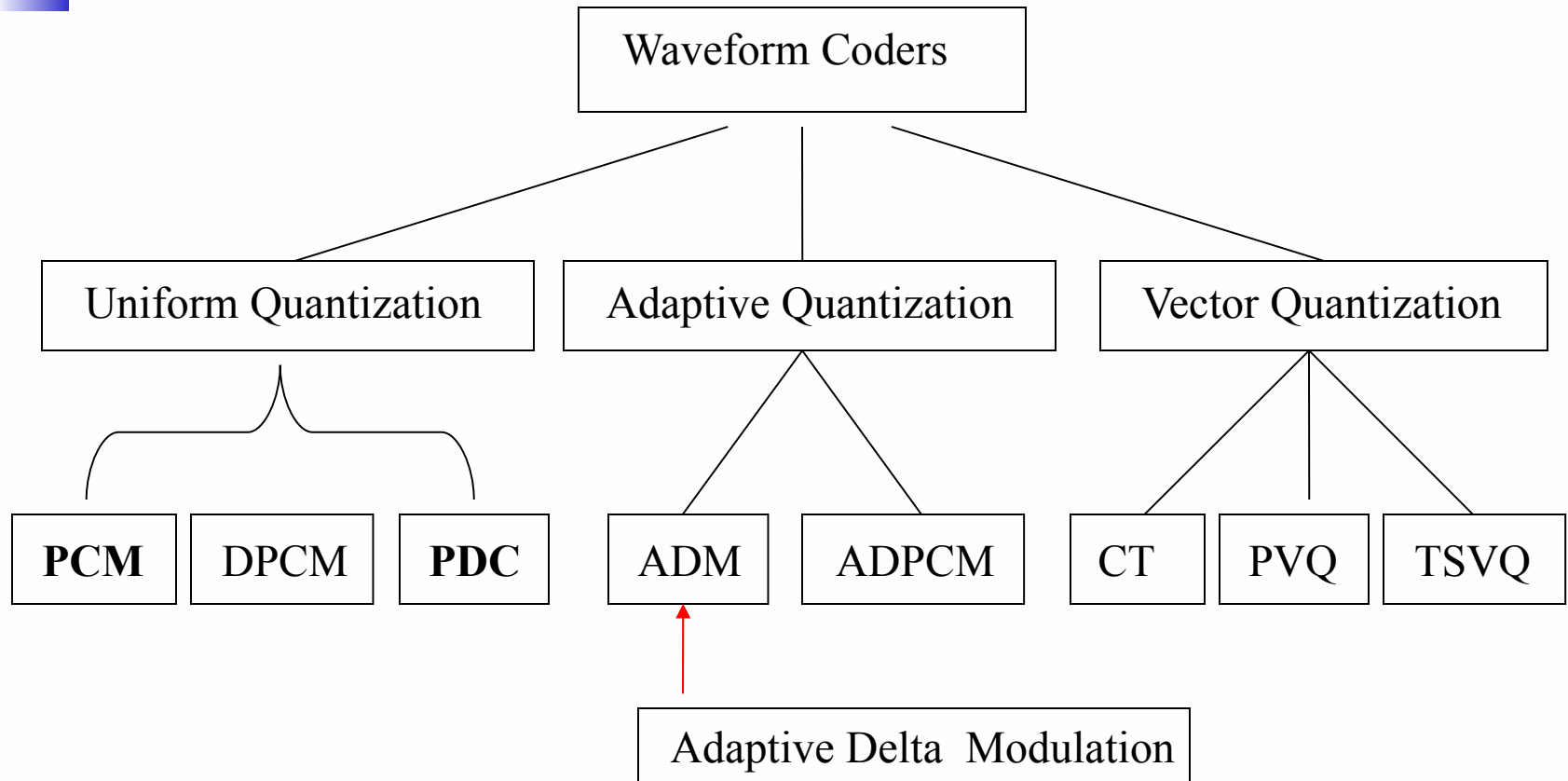
Speech coding algorithms can maintain this quality at substantially lower rates all the way down to 16 kb/s. At lower rates there will be some loss in quality, but even to rates as low as 1200 bits/s the speech is still quite intelligible



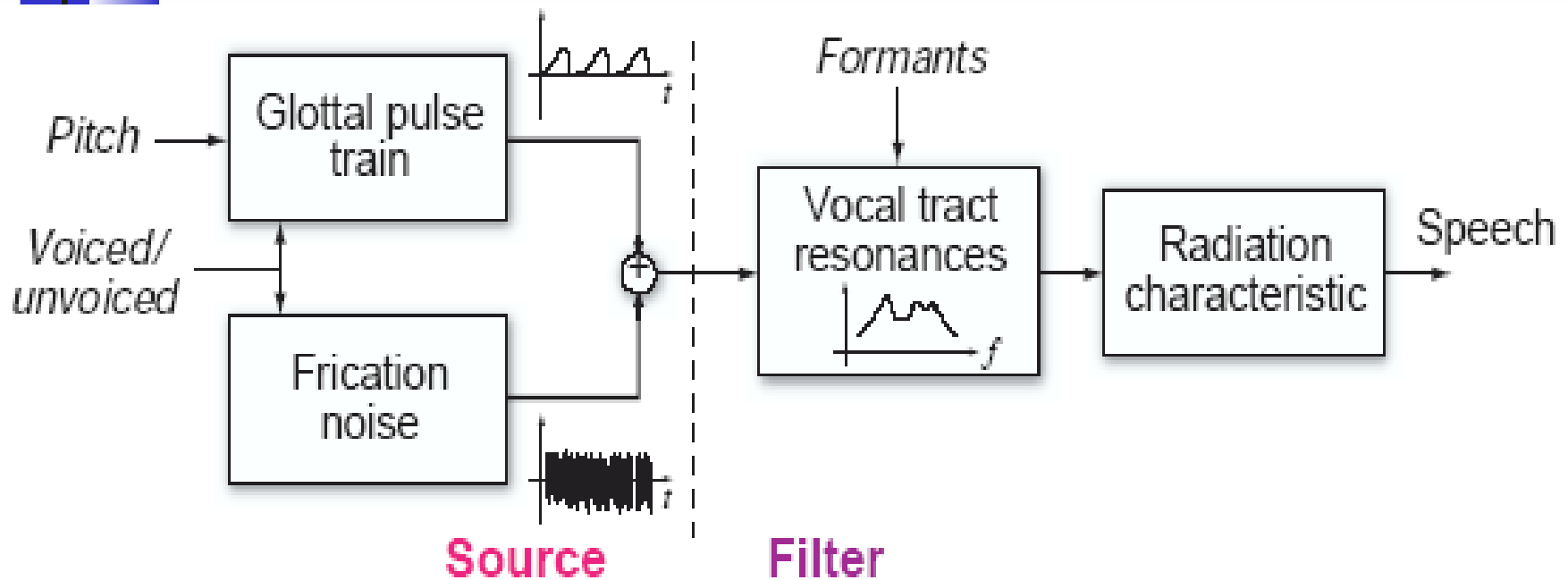
- Speech and audio compression has advanced rapidly in recent years spurred on by cost effective digital technology and diverse commercial application. It is including:
 - 1. Waveform coding
 - 2. Voice coding



Waveform Coders



Vocal tract and sound source modeling





Acoustic

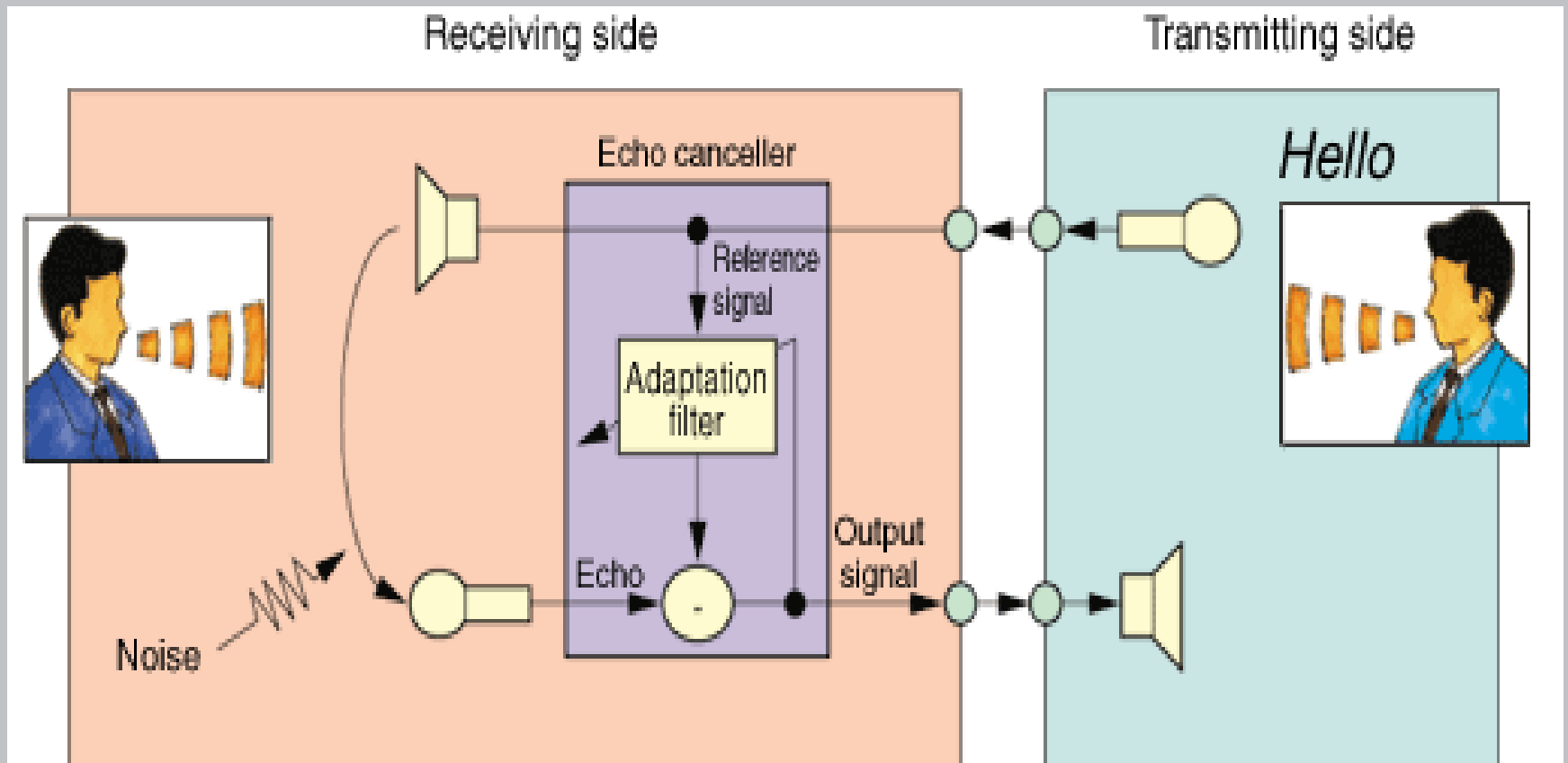
- *Acoustics* is the science of sound and the study of sound production and propagation.
- *Electro-acoustics* focuses on the transfer of a signal between acoustical and electrical form. It includes microphones and loudspeakers, echo cancellation, acoustic noise control, 3D audio and virtual acoustic audio rendering.
-



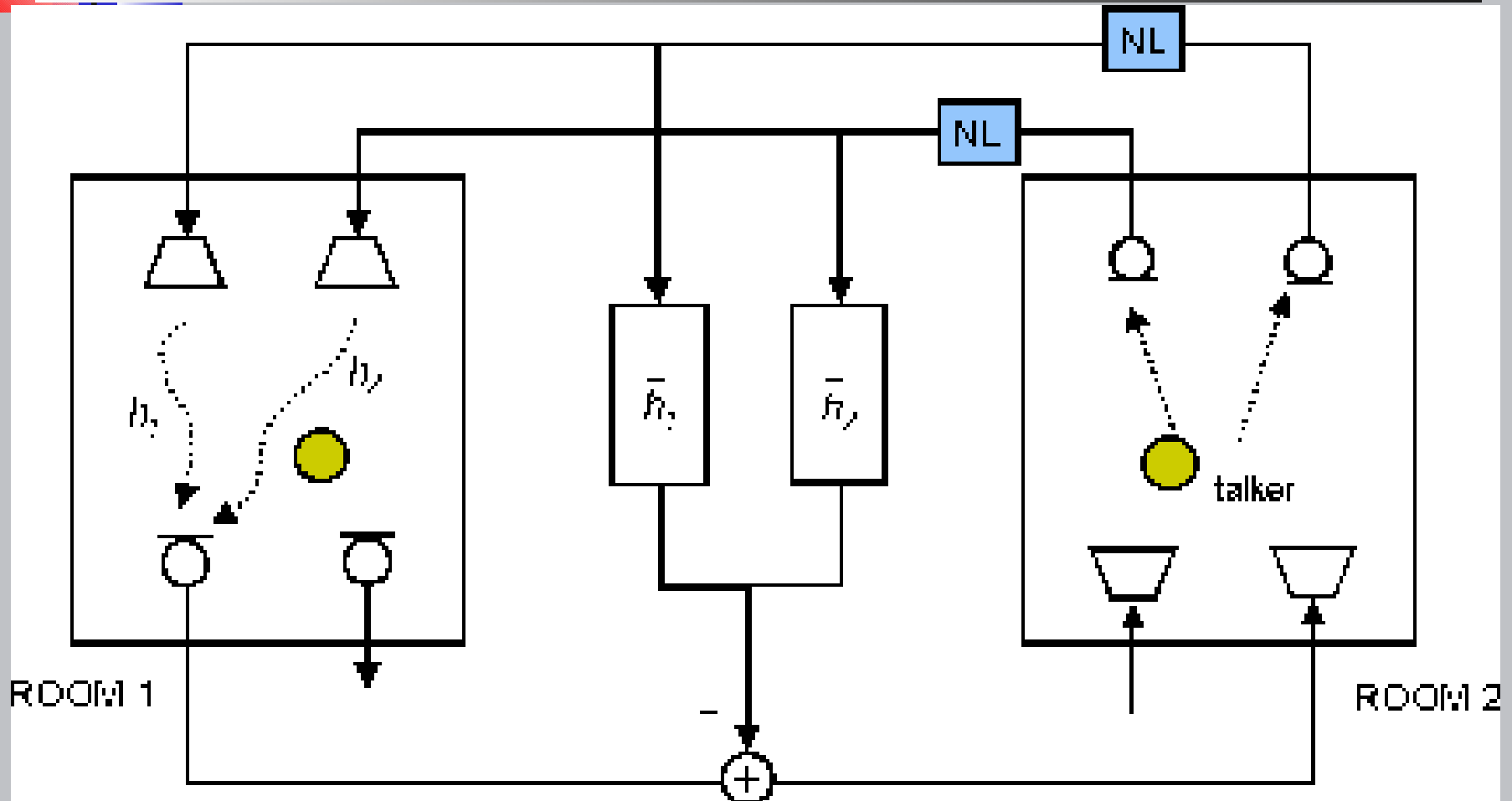
Echo Cancellation

- Echo cancellation was invented at Bell Labs in 1965 and research on network and acoustic echo cancellation continues. We have extended our investigations to the multi-channel problem and have successfully demonstrated real-time stereo acoustic echo cancellation in a teleconferencing system.

Echo Cancellation



Structure of the stereophonic echo cancellation

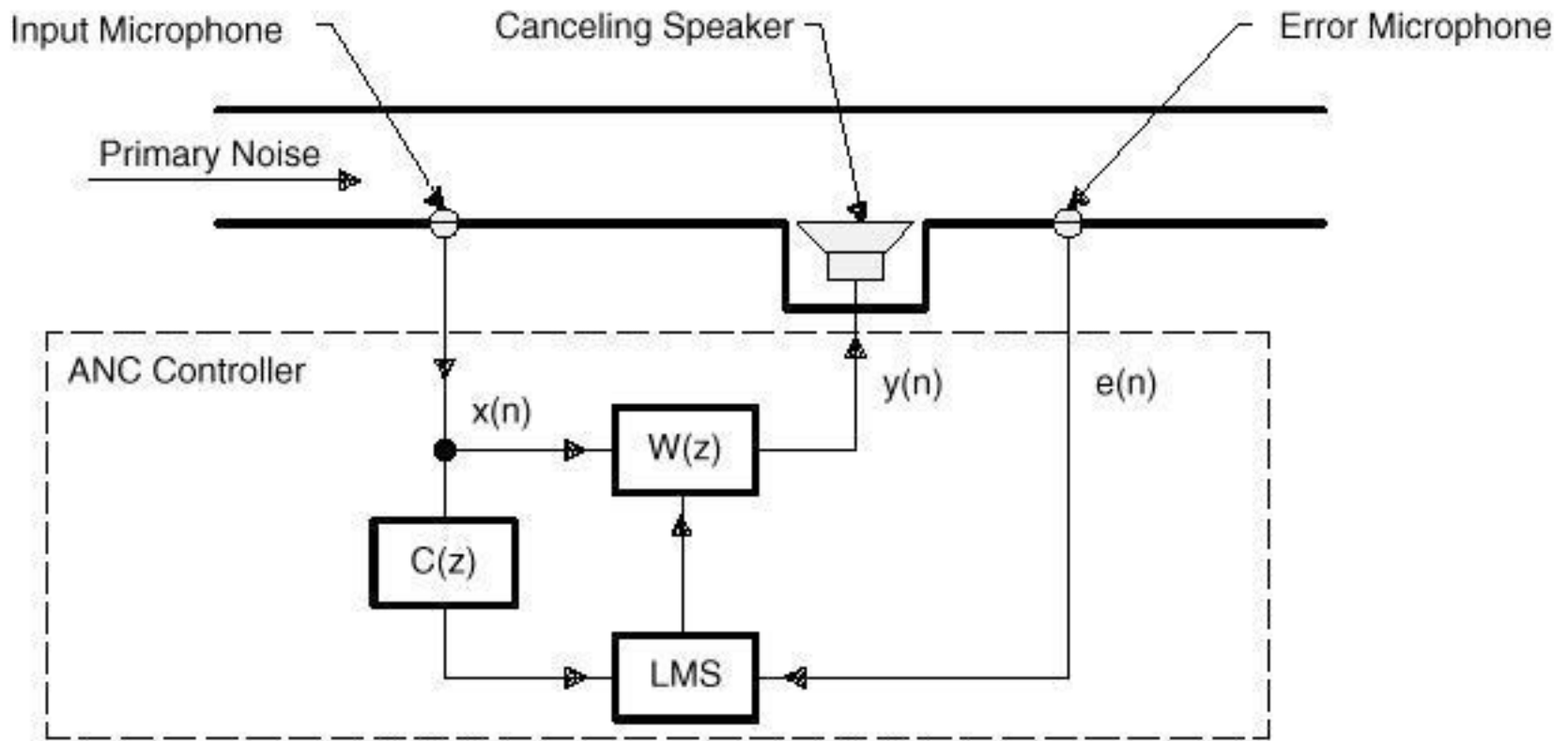




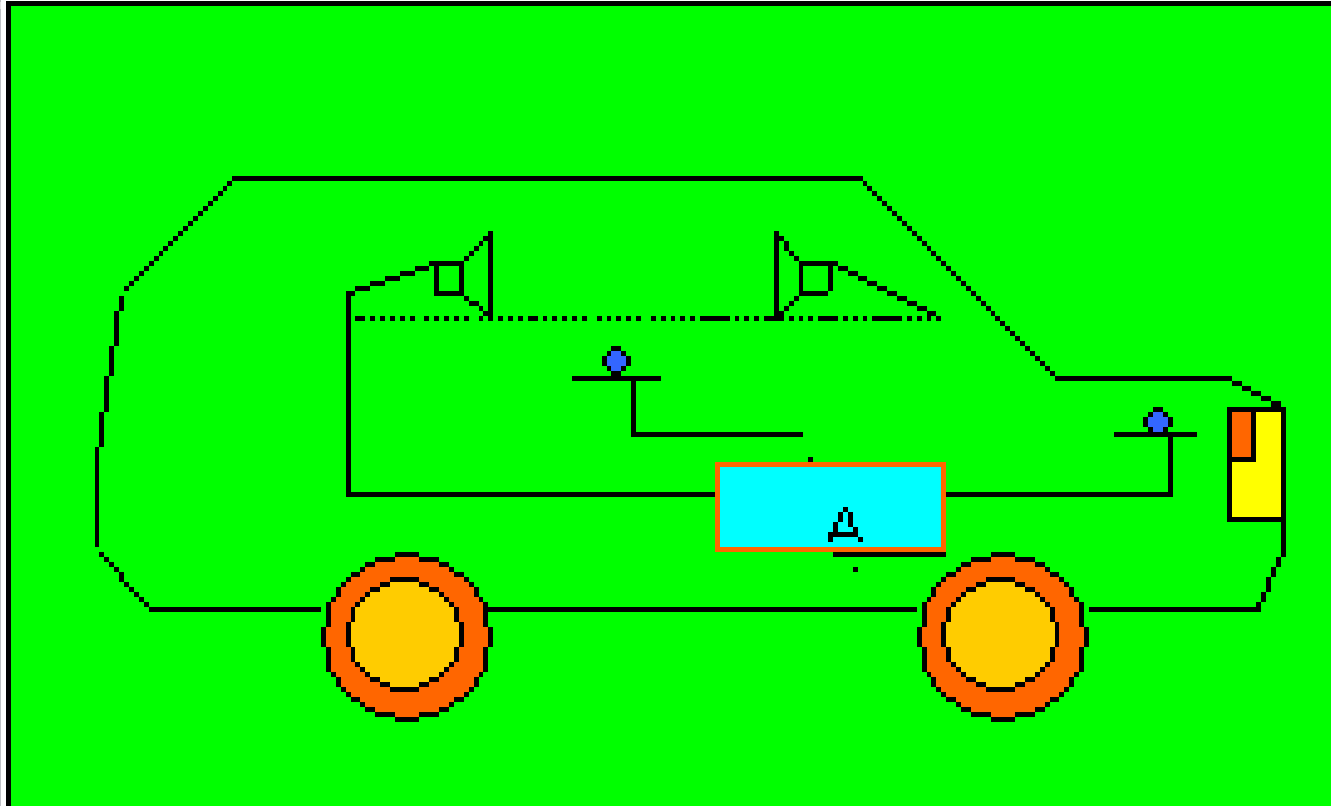
- Active noise control generates an **opposing wave** that is **equal in amplitude** but **out of phase** with the acoustic noise to be reduced.
- **Active noise control** makes use of adaptive digital filters in conjunction with reference and error sensing transducers and a secondary source, usually a loudspeaker.

This technology has been used in the control of noise generated in heating, ventilation, and air conditioning (HVAC) ducts, automobile exhaust noise, and aircraft engine and propeller noise, to mention just a few applications.

FXLMS Algorithm



- Acoustic noise control using the FXLMS algorithm



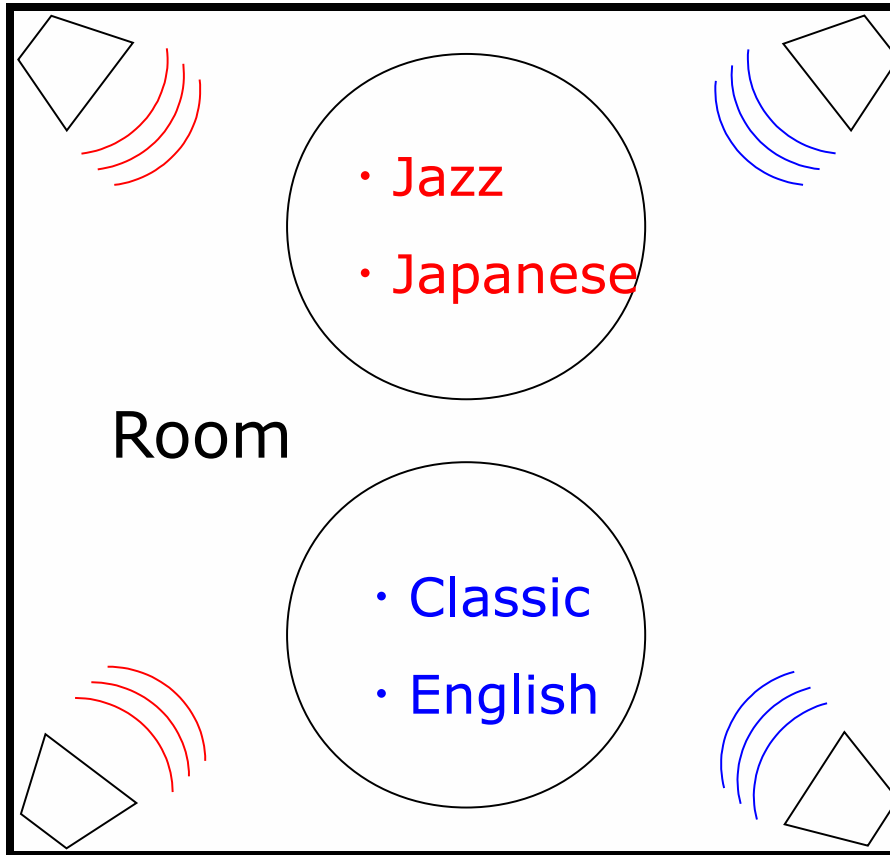
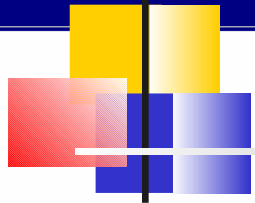
- Acoustic noise control in cars

Smart Acoustic Room (SAR)



- Smart Acoustic Room (SAR) is defined the acoustic response between two (or more) points could be controlled smartly. By control, we mean to have a well estimation of the acoustic path between two points and then to make the appropriate signal to cancel an unwanted noise or to emphasis to a desired signal (speech or music).

Application of SAR



When there are the peoples who want to listen to Jazz or Classic in a room, we don't want to use headphone as it totally isolate the person from surrounding.

In a conference room or big hall, we have two kinds of audiences that want to listen to the Japanese or English speech. If we can give two audiences the desire location, just by seating in the right place one can hear to desire language.



Digital Communications

- Digital communication techniques deal with transporting digital information (e.g. quantized or/and compressed speech, audio, image and video) reliably from a source to a destination.



- Digital audio broadcasting
- Digital cellular communication
- Storage in multilevel memory cells
- Efficient algorithm mapping and impleme

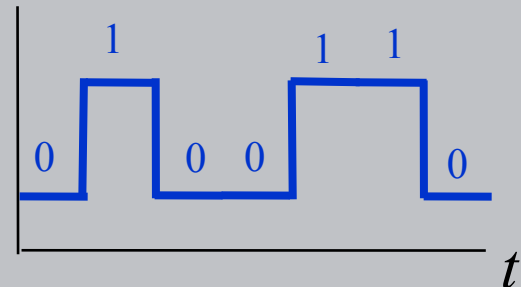
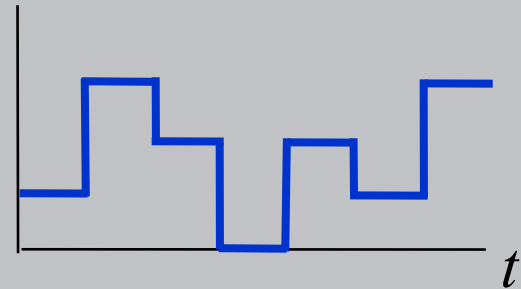
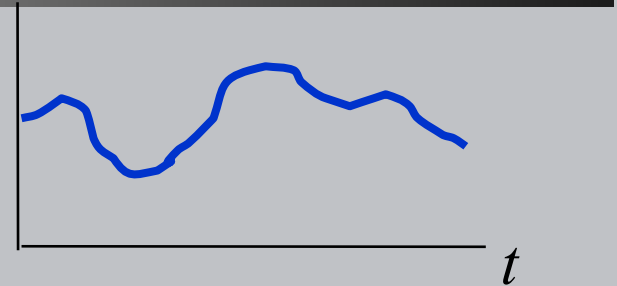
Can we classify signals?



- Messages or signals can be classified:
- Analog
 - A physical quantity that varies with “time”, usually in a smooth or continuous fashion
 - Fidelity describes how close is the received signal to the original signal. Fidelity defines acceptability
- Digital
 - An ordered sequence of symbols selected from a finite set of discrete elements
 - When digital signals are sent through a communication system, degree of accuracy within a given time defines the acceptability



- Analog Signals
 - Values are taken from an infinite set
- Digital Signals
 - Values are taken from a discrete set
- Binary Signals
 - Digital signals with just two discrete values



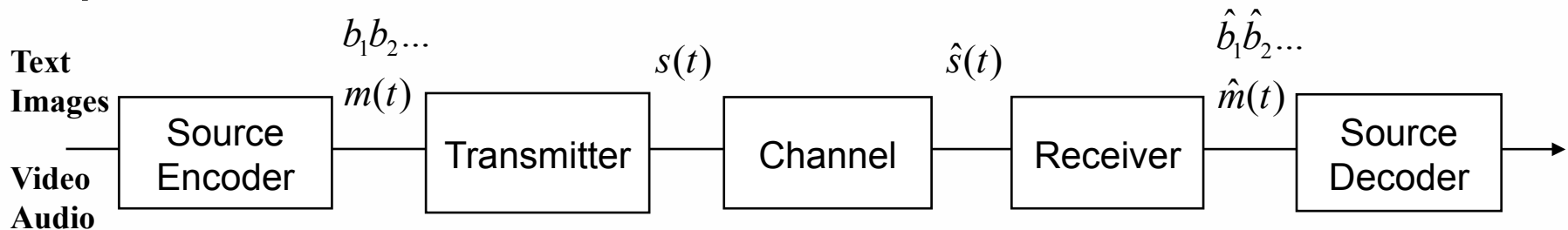


Elements of Communication Systems

- Transmitter
 - Modulation
 - Coding
- Channel
 - Attenuation
 - Noise
 - Distortion
 - Interference
- Receiver
 - Detection (Demodulation+Decoding)
 - Filtering (Equalization)



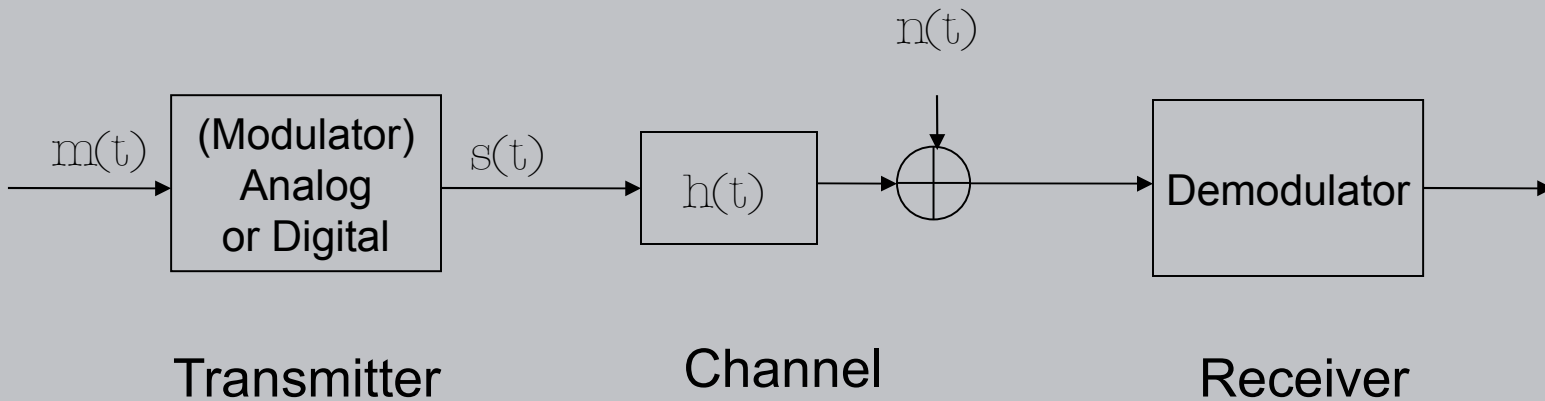
Elements of Communication Systems



- Encoder: Message \rightarrow Message Signal or bits
- Transmitter: Message signal \rightarrow Transmitted signal
- Channel: Introduces noise, distortion, interference
- Receiver: Received Signal \rightarrow Message Signal
- Decoder: Message Signal \rightarrow Original Message

Example: Microphone -----> Speaker

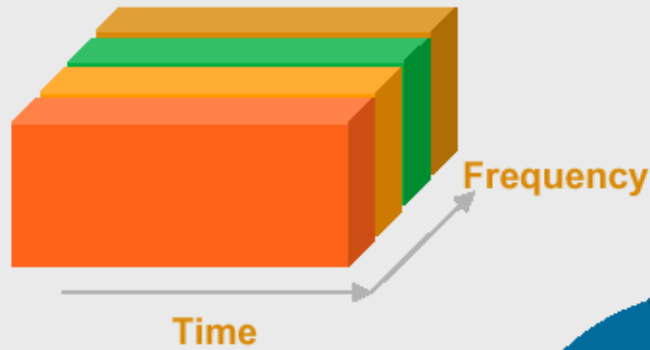
General Block Diagram of a Communication Transceiver





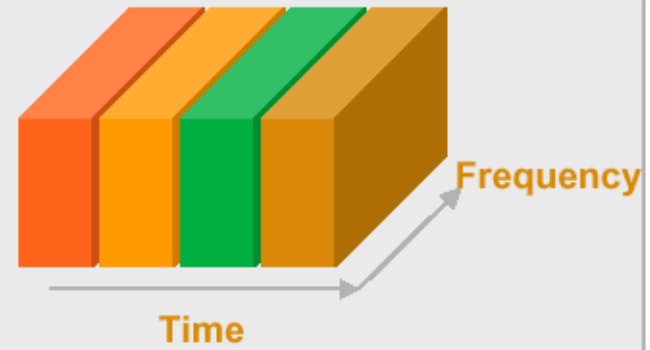
Generations of Digital communication

FDMA



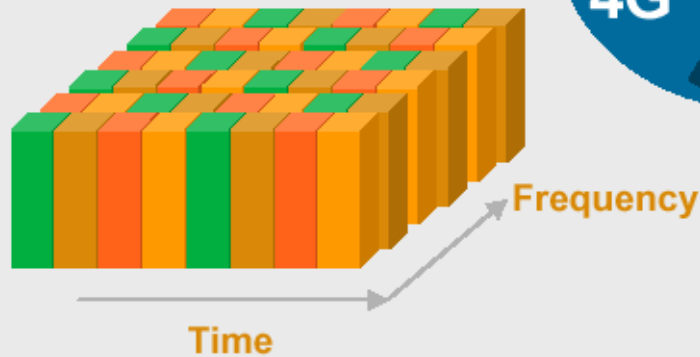
AMPS, TACS, NMT

TDMA



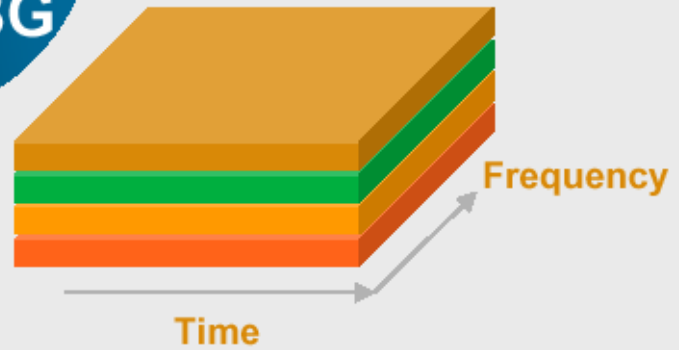
TDMA, PDC
GSM → GPRS/EDGE

OFDMA

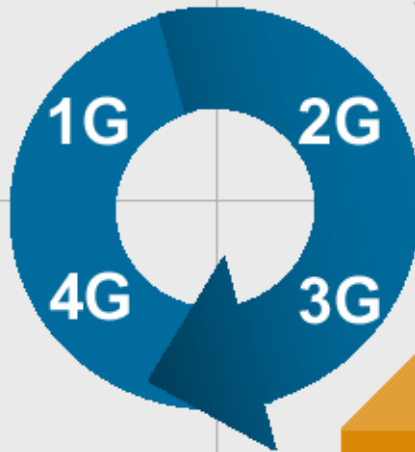


LTE, Rev-C, WiMAX

CDMA



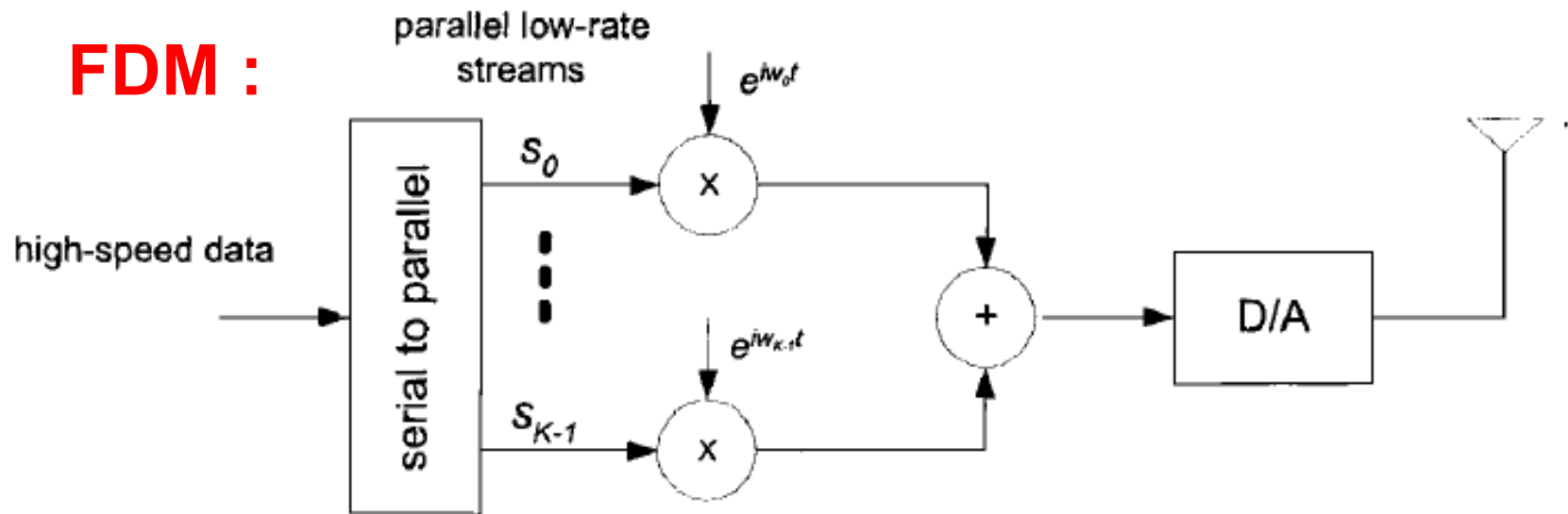
CDMA → EVXDO
UMTS → HSDPA
TD-SCDMA



Comparison between OFDM and FDMA Spectrum



FDM :



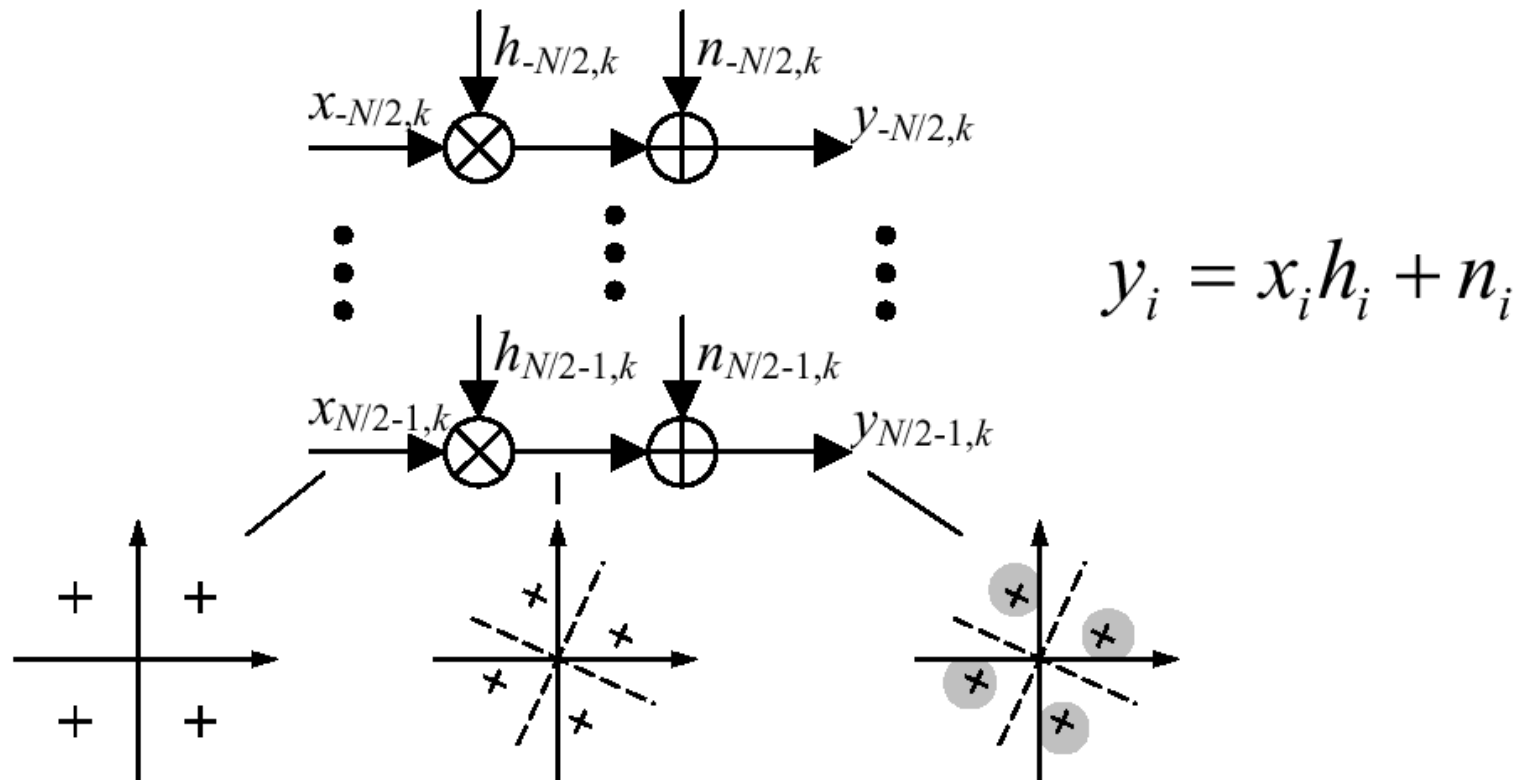
FDM scheme for high-speed transmission



Comparison between OFDM and FDMA Spectrum

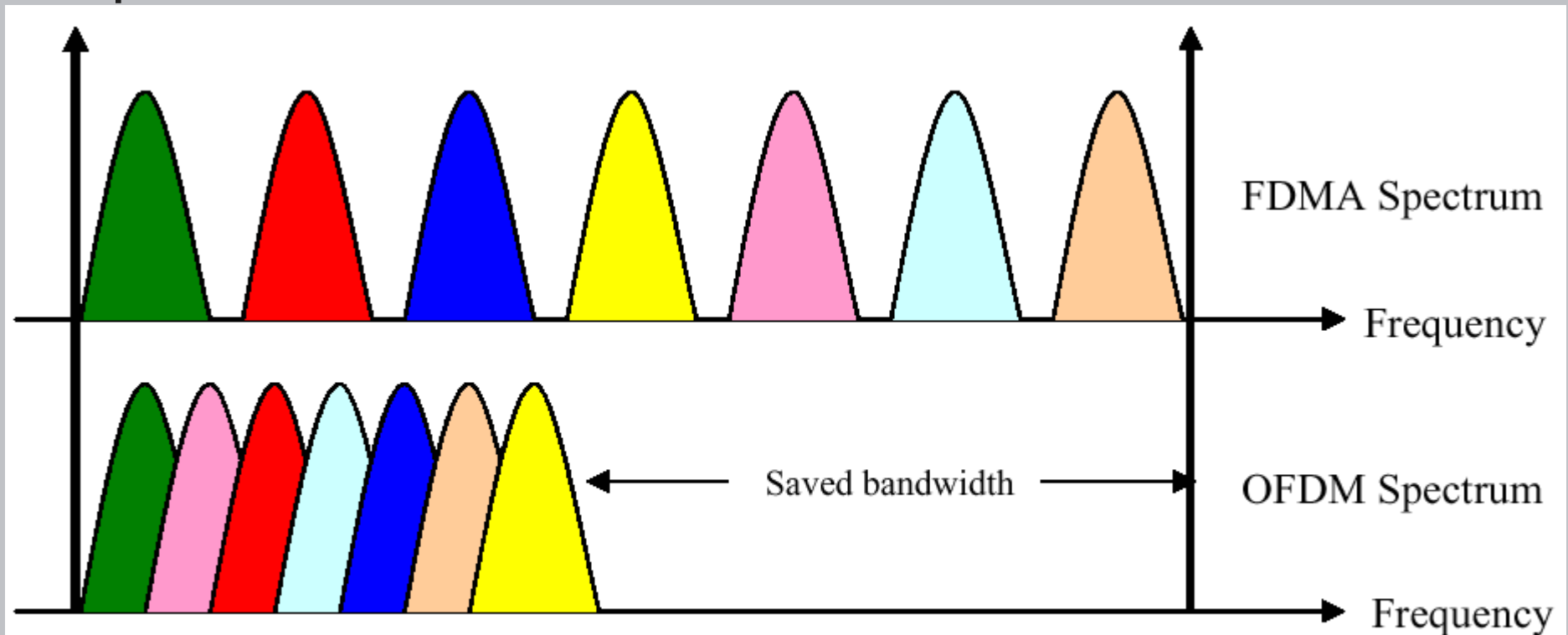
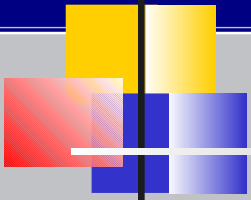
OFDM System Model

- Multiplication of data symbols with (complex-valued) channel transfer-function:





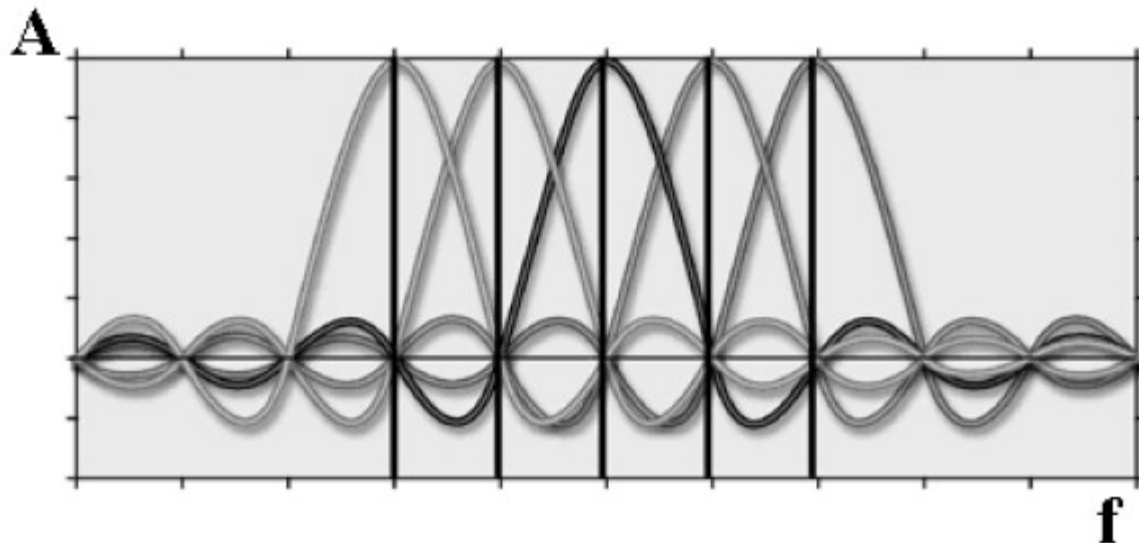
Comparison between OFDM and FDMA Spectrum



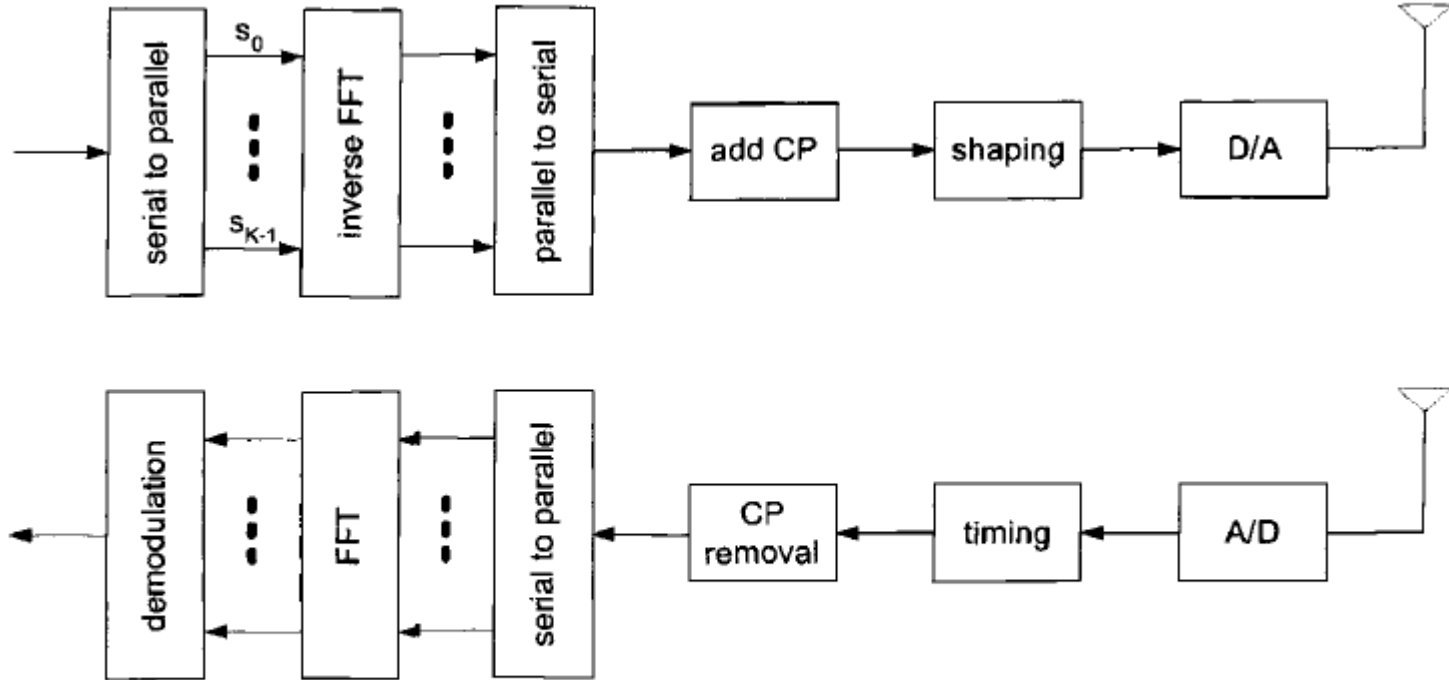
Bandwidth comparison of OFDMA and FDMA

OFDM overview

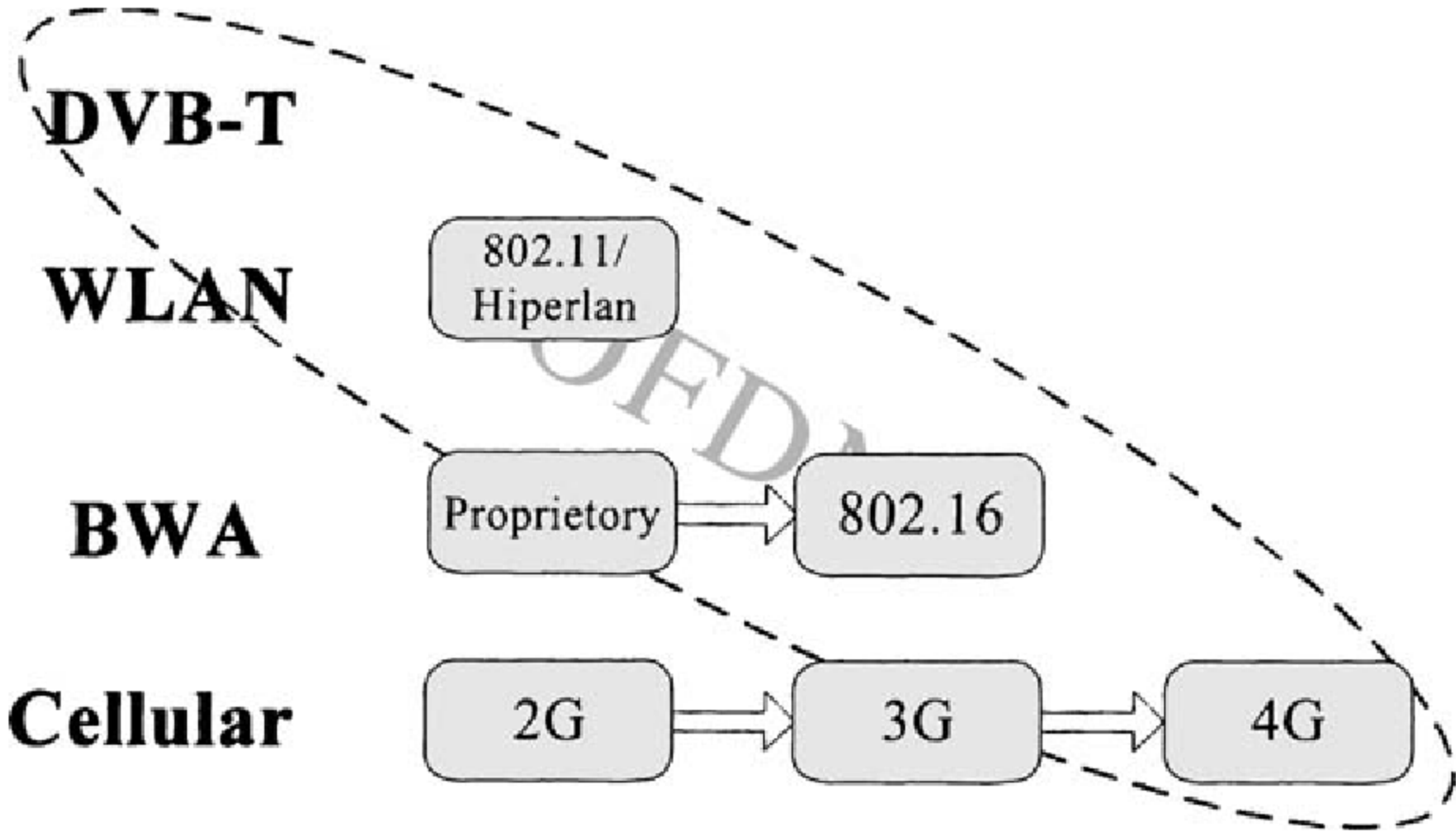
- Available bandwidth divided into subcarriers
 - Subcarriers overlapping but orthogonal with respect to each other
- at the peak of each subcarrier, the other subcarriers have zero amplitude

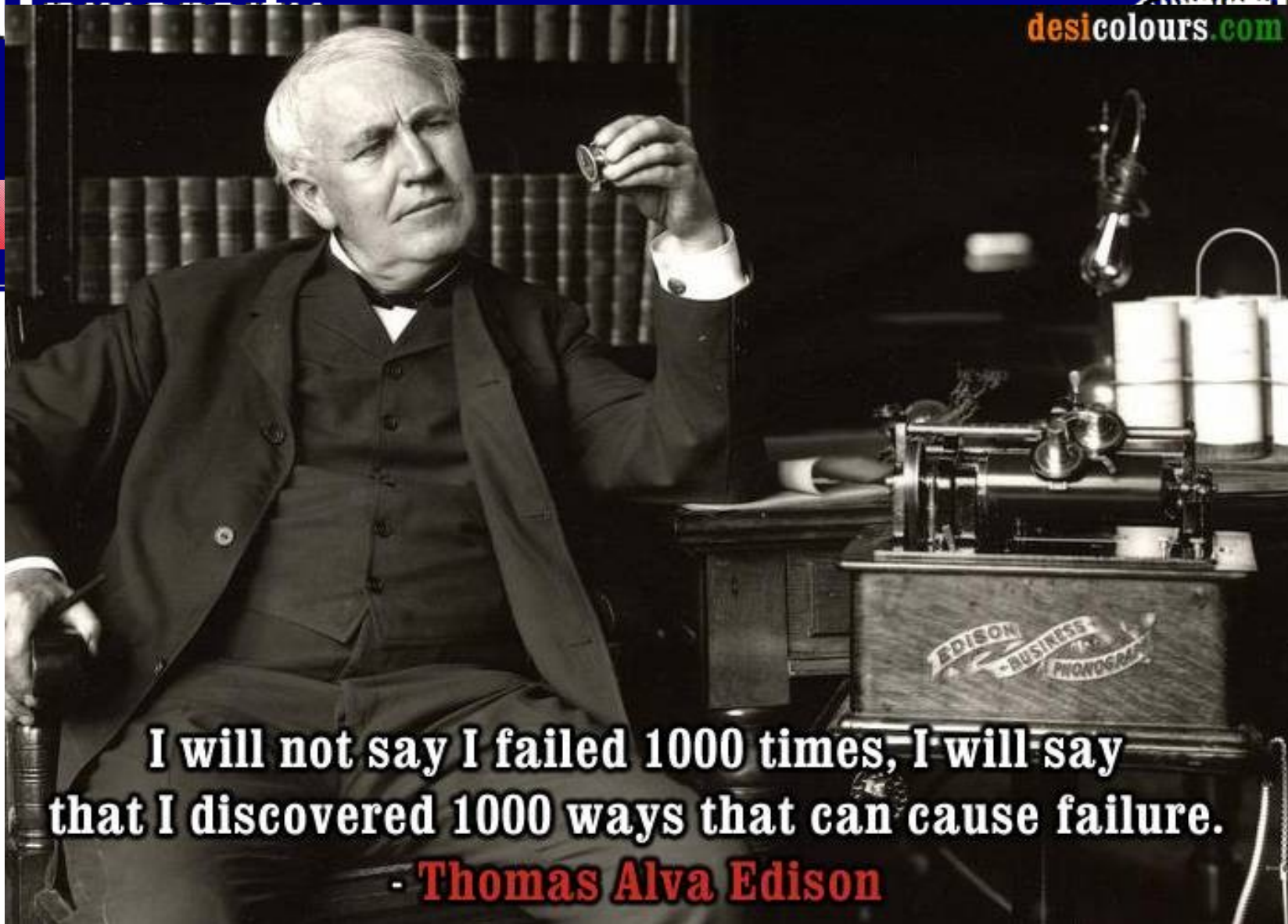


OFDM transceiver block diagram



OFDM transceiver block diagram





**I will not say I failed 1000 times, I will say
that I discovered 1000 ways that can cause failure.
- Thomas Alva Edison**

Thank you!