RADIO COMMUNICATIONS

Exercise 1

For a radio link, are known: the link frequency is $f = 2.4\ GHz$; the antennas gains are $G_T = G_R = 35\ dB$; the antennas are placed at heights: $h_T = 2.13\ m$, and $h_R = 1.8\ m$; the minimum power value for the receiver is $P_R = 0.1\ mW$.

a) Find the maximum LOS distance between the transmitter and the receiver, D_{LOS}.

b) Find the minimum transmitted power, P_T, expressing all parameters in absolute values, in order to accomplish the radio link.

SOLUTION---

a) $D_{LOS}[km] = 3.57 \cdot \left(\sqrt{h_T} + \sqrt{h_R}\right) = 3.57 \cdot \left(\sqrt{2.13} + \sqrt{1.8}\right) = 3.57 \cdot \left(1.46 + 1.34\right) \approx 10km$

b) $\lambda = \dfrac{c}{f} = \dfrac{3 \cdot 10^8}{2.4 \cdot 10^9} = 12.5cm$
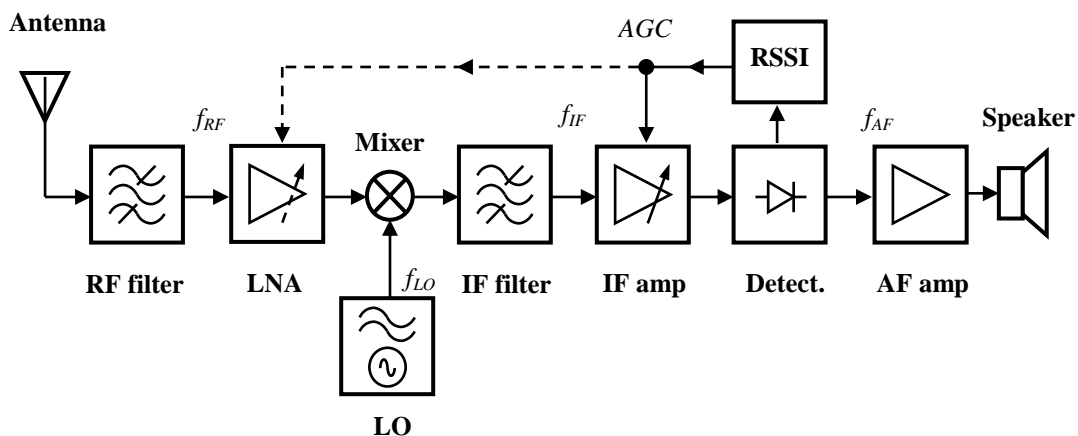
$G_T = G_R = 10^{35/10} = 10^{3.5} \approx 3162$

$\dfrac{P_T}{10^{-4}} = \dfrac{1}{G_T \cdot G_R} \cdot \left(\dfrac{4 \cdot \pi \cdot r}{\lambda}\right)^2 = \dfrac{1}{(3162)^2} \cdot \left(\dfrac{4 \cdot \pi \cdot 10000}{0.125}\right)^2 \approx 10^5 \Rightarrow$

$\Rightarrow P_T = 10^5 \cdot 10^{-4} = 10W$

Exercise 2

The superheterodyne AM receiver presented below ($f_{IF} = 455\ kHz$) is designed to receive the MW radio band (*526.5 kHz … 1606.5 kHz*).

a) Find the frequency range for the local oscillator.

b) Find the image frequency range.

c) Represent $f_{RF}, f_{LO}, f_{IF}, f_{img}$ in frequency domain for $f_{RF} = 1000\ kHz$.

SOLUTION---

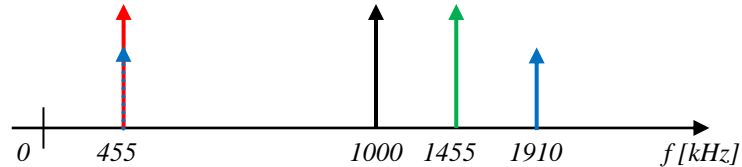**a)** $f_{IF} = f_{LO} - f_{RF} \Rightarrow f_{LO} = f_{RF} + f_{IF}$

$$\Rightarrow \begin{cases} f_{LO.min} = 526.5 + 455 = 981.5 kHz \\ f_{LO.max} = 1606.5 + 455 = 2061.5 kHz \end{cases} \Rightarrow f_{LO} = 981.5 kHz...2061.5 kHz.$$

b) $f_{img} = f_{LO} + f_{IF} = f_{RF} + 2 f_{IF}$

$$\Rightarrow \begin{cases} f_{img.min} = 526.5 + 2 \cdot 455 = 1436.5 kHz \\ f_{img.max} = 1606.5 + 2 \cdot 455 = 2516.5 kHz \end{cases} \Rightarrow f_{img} = 1436 kHz...2516.5 kHz.$$

c) $f_{RF} = 1000\ kHz, f_{IF} = 455\ kHz, \Rightarrow f_{LO} = 1455\ kHz, f_{img} = 1910\ kHz.$

Theory 1

Draw and explain the main blocks of a radio receiver.

ANSWER---

The receiver operates on the output signal from the radio channel in preparation for delivery to the transducer at the destination.
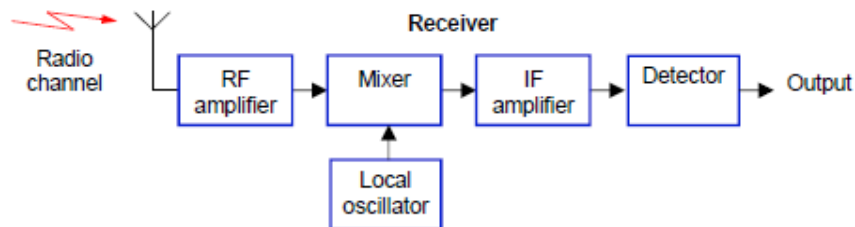
Receiver operations include amplification to compensate for transmission loss, and demodulation and decoding to reverse the signal-processing performed at the transmitter.

Filtering is another important function at the receiver.

The receiver oscillator is called the local oscillator as it produces a local carrier within the receiver which allows the incoming carrier from the transmitter to be modified for easier processing within the receiver (frequency translation by down conversion, on the intermediate frequency).

Site address: https://intranet.etc.upt.ro/~RADIOCOM/, Course no.1, slides 13-14.

Theory 2

Represent and give the expressions for electric / magnetic fields intensities, *E(t, r)* and *H(t, r)*, with parameters signification (for a vertical dipole).
ANSWER---
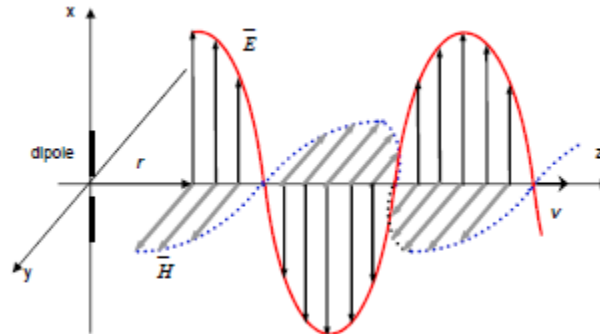
The two vectors *E* and *H* are orthogonal one against other but also against the propagation direction.

In any point of the space, their amplitudes are proportionally, and their oscillations are synchronized (in phase).

The expressions for the electric field intensity, *E*, and the magnetic field intensity, *H*, in a point at distance *r* from the antenna (dipole) are:

$$E(t,r) = E_{max} \cdot cos\left[\omega \cdot \left(t - \frac{r}{v}\right)\right] = E_{max} \cdot cos(\omega \cdot t - k \cdot r) \qquad [V/m]$$

$$H(t,r) = H_{max} \cdot cos\left[\omega \cdot \left(t - \frac{r}{v}\right)\right] = H_{max} \cdot cos(\omega \cdot t - k \cdot r) \qquad [A/m],$$

where:
v = EM wave velocity [*m/s*],
$\omega = 2 \cdot \pi \cdot f$ = wave angular frequency,
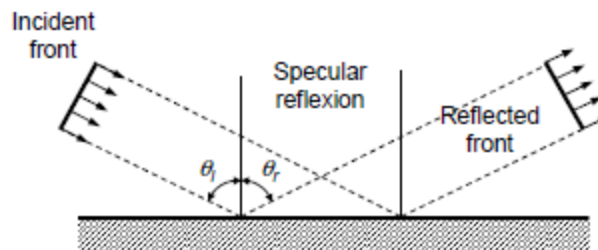$\lambda = v/f$ = wavelength,
$k = 2 \cdot \pi/\lambda$ = phase constant of the wave.

Site address: https://intranet.etc.upt.ro/~RADIOCOM/, Course no.2, slides 11-12.

Theory 3

Draw and explain specular and diffuse reflections.
ANSWER---

Specular reflection is the case where the surface can be treated as perfectly smooth at the frequency of interest, meaning that any surface irregularities are very small compared with a wavelength.

The angle of incidence θ_i is equal to the angle of reflection θ_r.
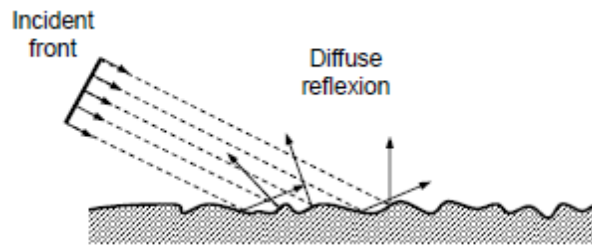
Diffuse reflection, which occurs mainly at shorter radio wavelengths, is the case where objects (walls and so on) can be considered as very rough on the scale of the wavelength.

Site address: https://intranet.etc.upt.ro/~RADIOCOM/, Course no.2, slides 29-30.

Theory 4.

Represent (E and H planes) and briefly describe the directivity of a half-wave dipole antenna.

ANSWER---
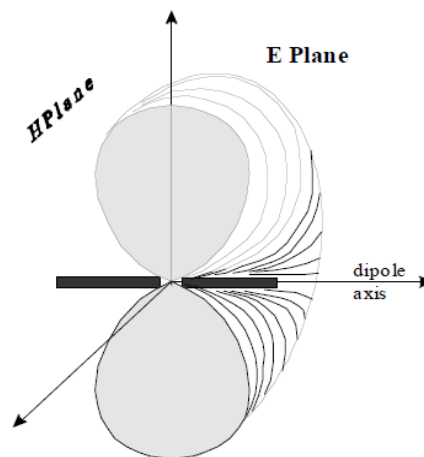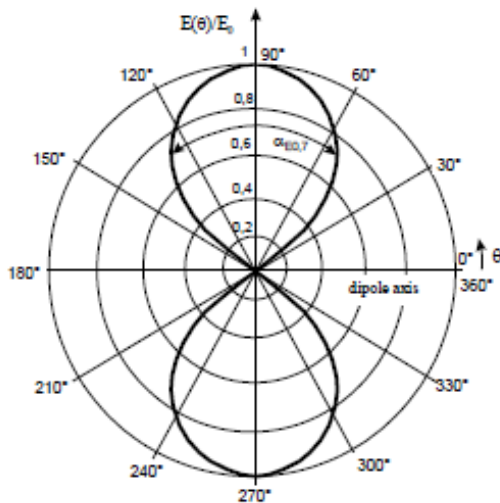
$$H(\varphi)/H_0 \equiv 1$$

(A half-wave dipole has an antenna gain of *1.64* or *G = 2.15 dBi*).

The beamwidth of half-wave dipole is a $E_{0.7} = 80°$ in E-plane (the half-wave dipole antenna has a bidirectional pattern in the E-plane).

The beamwidth of half-wave dipole is a $H_{0.7} = 360°$ in H-plane (the half-wave dipole antenna has an omnidirectional pattern in the H-plane)
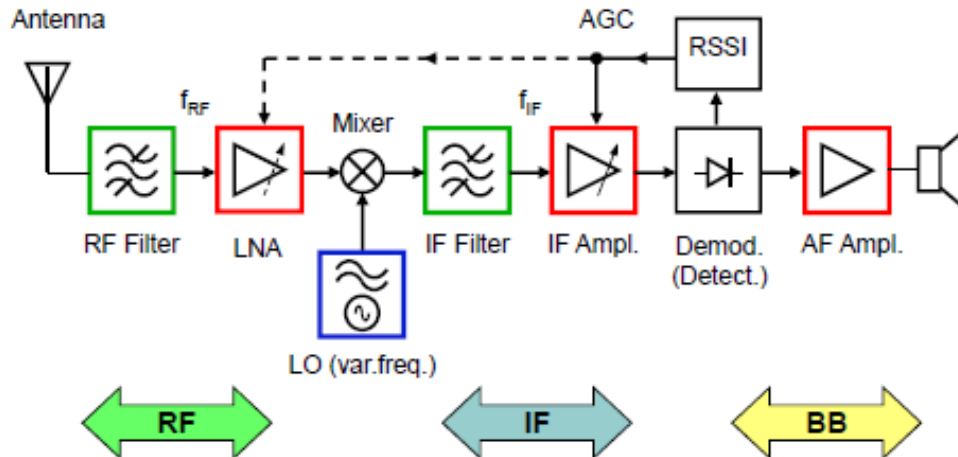
This makes the antenna useful for broadcast and wireless applications.

Site address: https://intranet.etc.upt.ro/~RADIOCOM/, Course no.4, slides 37-38.

Theory 5

Represent the structure of an AM superheterodyne receiver and briefly explain the functions of the main blocks.
ANSWER--


- after the receiving antenna,
- a **radio frequency front end** is needed, consisting in a band-pass filter (in order select the desired carrier frequency / radio station) and a low noise amplifier (due to very low power received signals);
- next, the local oscillator and the mixer followed by an IF filter are used for heterodyne process performing RF translation on the intermediate frequency;
- the **intermediate frequency** blocks amplify the signal to ensure the right level at the demodulator input (it is mainly done here);
- a detector extracts the initial information;
- the **baseband** blocks process the information (basically, audio frequency amplification);
- RSSI = received signal strength indicator (optional);
- AGC = automatic gain control loop is used to maintain a constant level in case of fading.

Site address: https://intranet.etc.upt.ro/~RADIOCOM/, Course no.6, slide 12 and some more.

POWER ELECTRONICS

1. The four *basic dc-dc nonisolated ideal converters*: *buck*, *boost*, *buck-boost* and *Ćuk*: schematics with MOS transistors and diodes and their static conversion ratio *M* in terms of transistor *duty cycle D*. Also indicate the output voltage polarity for each topology.
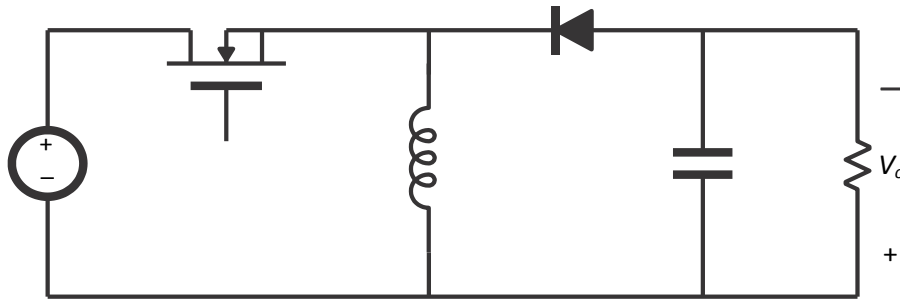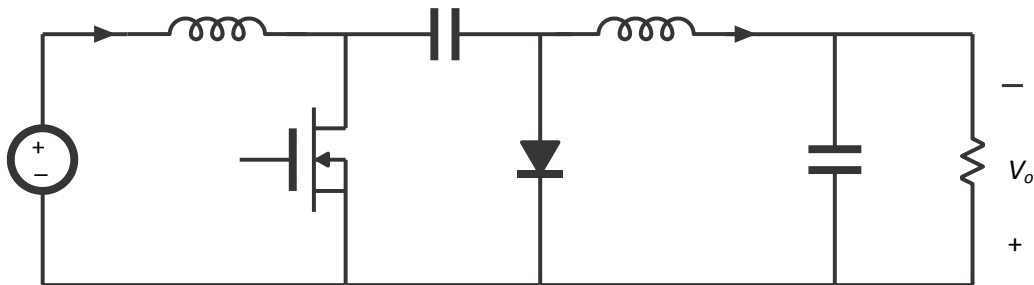
<u>Solution</u>
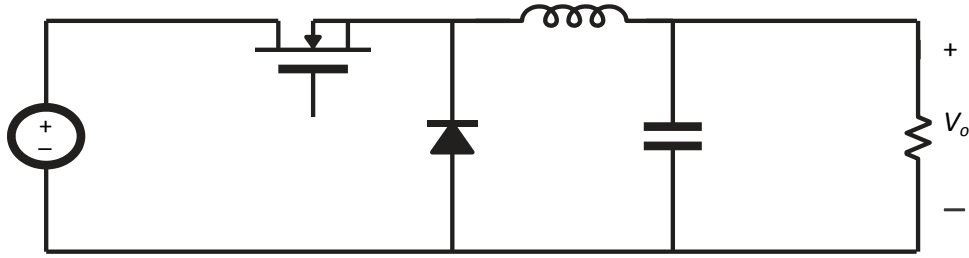
The buck converter. $M(D) = D$.
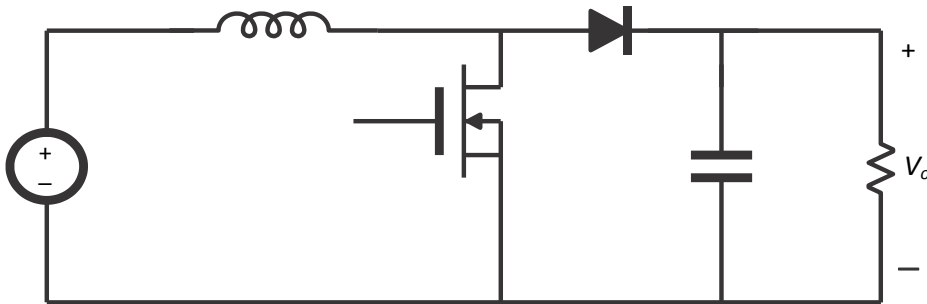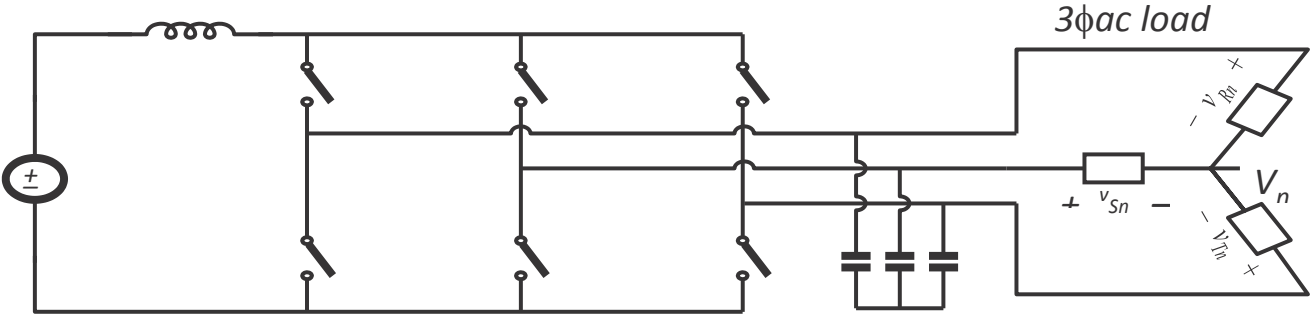

The boost converter. $M(D) = \dfrac{1}{1-D}$.

The buck-boost converter. $M(D) = \dfrac{D}{1-D}$.

The Ćuk converter. $M(D) = \dfrac{D}{1-D}$.

2. The *three phase current source inverter* – schematics, characteristic nature and practical switch implementation.

Solution

3φac load

This inverter exhibits a *boost-type* conversion characteristic. The switches are *current unidirectional* (voltage bidirectional) two quadrant switches, operating in quadrants I and II. A possible practical implementation contains a series diode and a transistor, as below:
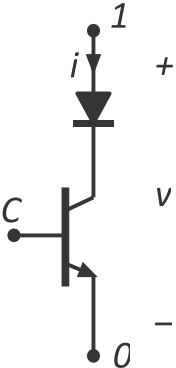
3. Explain when synchronous rectification is suitable, what synchronous rectification is and enumerate at least two of its advantages. Draw the schematic of a *synchronous Zeta* converter.
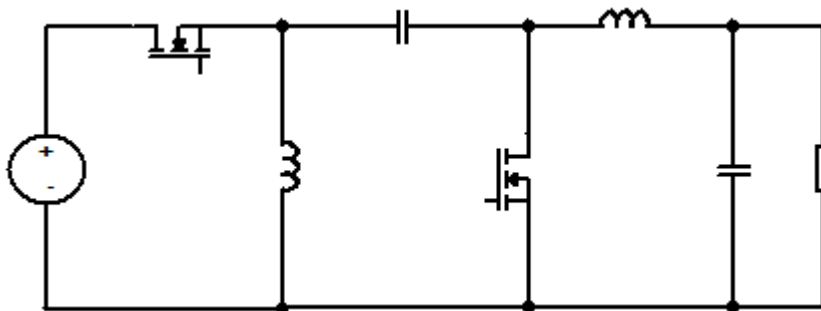
Solution

Synchronous rectification is used in *high current* applications, when the conduction losses in the diodes are high because of the high currents that flow through them. Synchronous rectification means to replace the diode by a MOS transistor, as the latter exhibits much lower conduction losses due to its low on resistance. The MOS is connected such that *its internal diode to play the same role and to be positioned similar to diode that is replaced*.

Advantages of synchronous rectification are:

- *High efficiency* because the conduction losses are reduced.
- *Absence of discontinuous conduction mode*, because the diodes are the cause of DCM occurrence and in synchronous rectification they are missing.
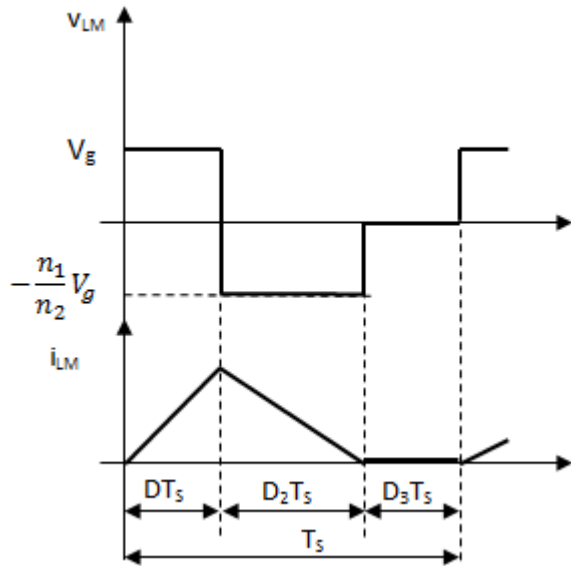
The *synchronous Zeta converter* is presented below:

4. The classical *single-transistor three windings forward* converter: schematics, magnetizing inductor voltage and current waveforms, maximum duty cycle formula.
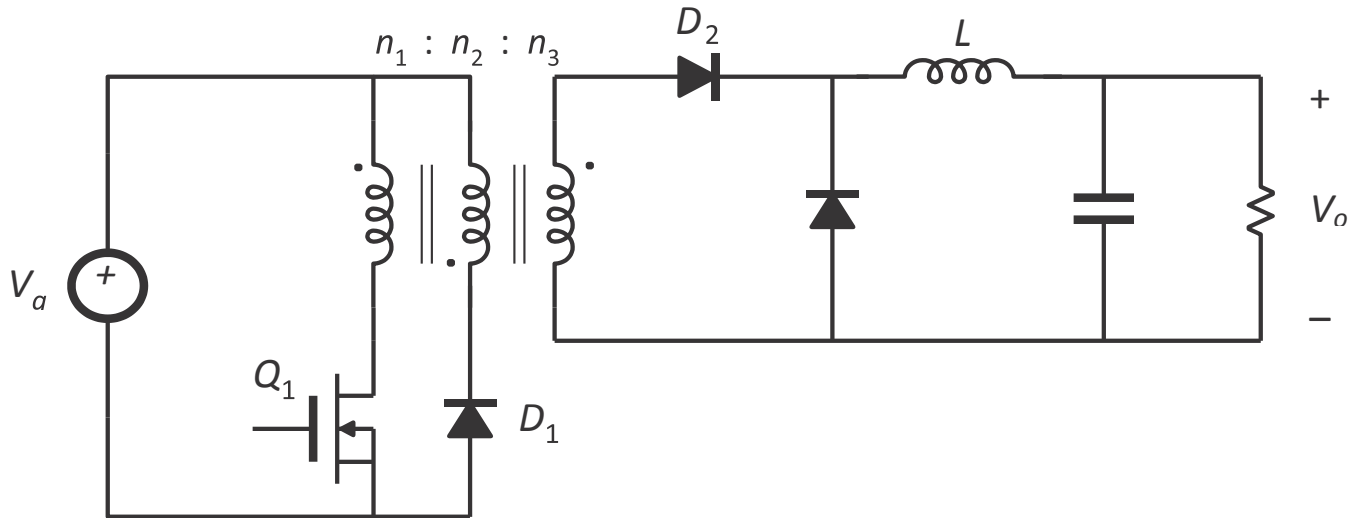
Solution

From magnetizing inductor volt-second balance we get that

$$D \cdot V_g + D_2 \cdot (-\frac{n_1}{n_2} \cdot V_g) + D_3 \cdot 0 = 0 \qquad (1)$$

Solve for D_2:

$$D_2 = \frac{n_2}{n_1} \cdot D \qquad (2)$$

On the other side, for correct core reset D_3 cannot be negative. Clearly $D_3 = 1 - D_2 - D$ and therefore it is necessary that:

$$1 - D_2 - D \geq 0 \qquad (3)$$

Using the value for D_2 given by (2), inequality (3) becomes:

$$1 - \frac{n_2}{n_1} D - D \geq 0 \qquad (4)$$

from which it follows that:

$$D \leq \frac{1}{1 + \frac{n_2}{n_1}} \qquad (5)$$

From (5) the maximum duty cycle value results as $D_{\max} = \frac{1}{1 + \frac{n_2}{n_1}}$.

<u>Note.</u> If the final result is provided without proof, the answer will also be accepted.

5. The ideal *full bridge buck*-derived converter – schematics, transistors control sequence, derive the static conversion ratio. In case of a nonideal converter specify at least one method that can be used to prevent core saturation due to nonidealities.

<u>Solution</u>

Converter schematic is represented below.
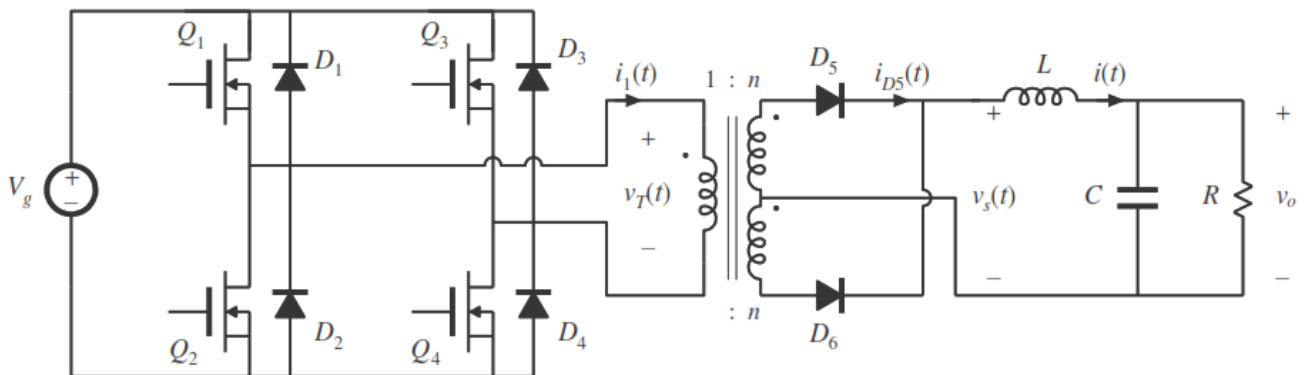
Transistors control sequence is the following:

Interval	ON transistors
$[0, DT_s]$	Q_1, Q_4
$[DT_s, (1-D)T_s]$	None
$[(1-D)T_s, T_s+DT_s]$	Q_2, Q_3
$[(1-D)T_s, 2T_s]$	None

Writing volt-second balance across L for a period T_s, one obtains taking into account that $V_C=V_o$:

$$D \cdot \left(nV_g - V_o\right) + (1 - D) \cdot (-V_o) = 0 \tag{1}$$

From (1) it immediately results that

$$V_o = nDV_g \tag{2}$$

and the static conversion ratio

$$M \stackrel{\text{def}}{=} \frac{V_o}{V_g} = nD \tag{3}$$

Saturation can be prevented either *placing a capacitor in series with the primary*, or using *current mode control*.

6. In a CCM operated nonideal *boost* converter supplied from a 5 V dc source and delivering power to a 10 Ω resistive load, the transistor exhibits an on resistance R_{on}=20 mΩ, while the forward voltage drop of the diode is V_D=0.5 V. All the other components are assumed to be ideal. The converter is controlled with a duty cycle D=0.475.

Calculate the dc output voltage and the efficiency of the conversion.

<u>Solution</u>

The equivalent circuit with the lossy elements replaced by their equivalent models is depicted below. In this schematic switch *S* and diode *D* are ideal. In order to evaluate the dc magnitudes the inductor current i_L and the capacitor voltage v_C are assumed to be constant, hence equal to their dc values, Moreover, it can be remarked that $v_o=v_C$. With these considerations, writing volt-second balance for the
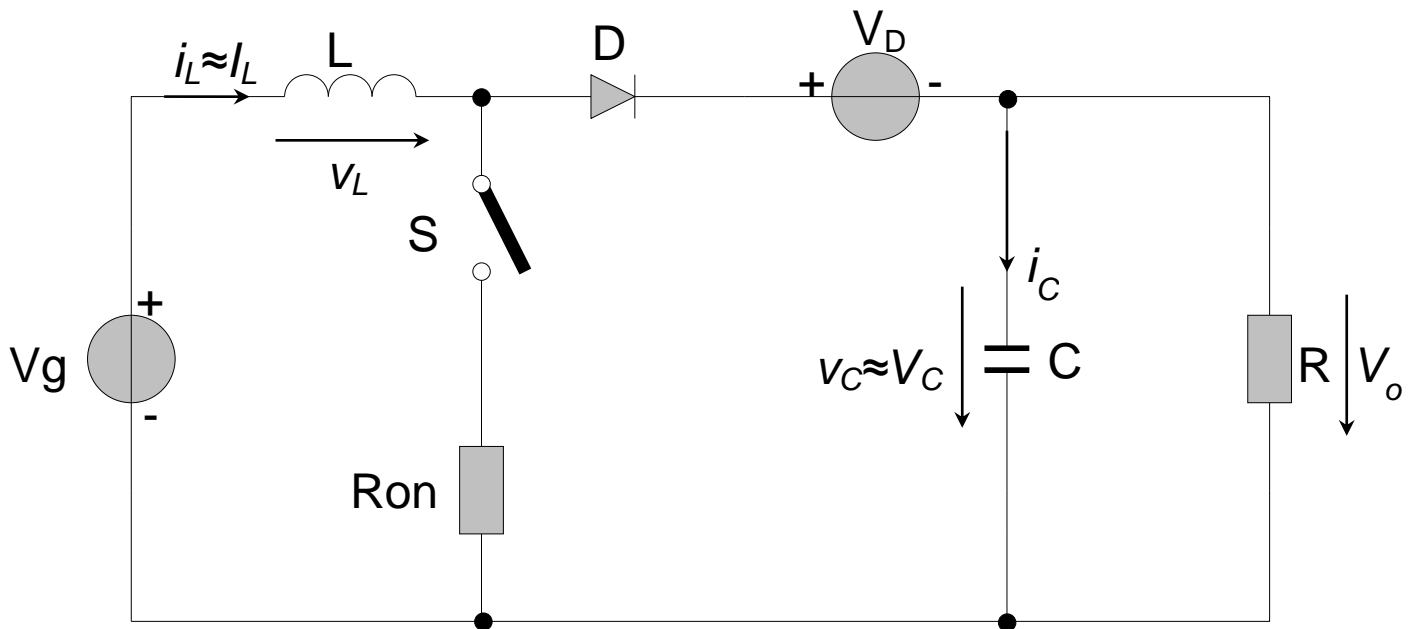
inductor and charge balance for the capacitor it results that:

$$D \cdot \left(V_g - I_L \cdot R_{on}\right) + (1 - D) \cdot \left(V_g - V_C - V_D\right) = 0 \tag{1}$$

$$D \cdot \left(-\frac{V_C}{R}\right) + (1 - D) \cdot \left(I_L - \frac{V_C}{R}\right) = 0 \tag{2}$$

Equations (1) și (2) provide a linear system with V_C and I_L as unknowns. Solving it one obtains:

$$V_o = V_C = \frac{\frac{V_g}{1-D} - V_D}{1 + \frac{D}{(1-D)^2} \cdot \frac{R_{on}}{R}} = 9V \tag{3}$$

$$I_L = \frac{V_C}{R(1-D)} = \frac{\frac{V_g}{1-D} - V_D}{1 + \frac{D}{(1-D)^2} \cdot \frac{R_{on}}{R}} \cdot \frac{1}{R(1-D)} = 1.71A \tag{4}$$

Conversion efficiency is given by:

$$\eta = \frac{P_o}{P_g} = \frac{\frac{V_o^2}{R}}{V_g \cdot I_L} = 0.9442 \text{ or } \eta = 94.42\% \tag{5}$$

7. An ideal CCM operated *flyback* converter is supplied from dc voltage source of 300V and delivers 5V at the output to a 100W load. The transformer turns ratio is defined in the figure below and equals $n = \frac{1}{40}$.

The reactive elements C and L_M are high enough such that to admit that the capacitive voltage and the magnetizing current are constant for dc analysis, which is typical for CCM operation. Determine:

a) The rms transistor current.
b) Transistor voltage stress.
c) Diode voltage stress.

Solution

It is known that the flyback converter comes from the buck-boost topology and therefore its conversion ratio equals that of the buck-boost multiplied by the transformer turns ratio *n*. Hence:

$$M(D) = \frac{V_o}{V_g} = n \frac{D}{1-D} \tag{1}$$
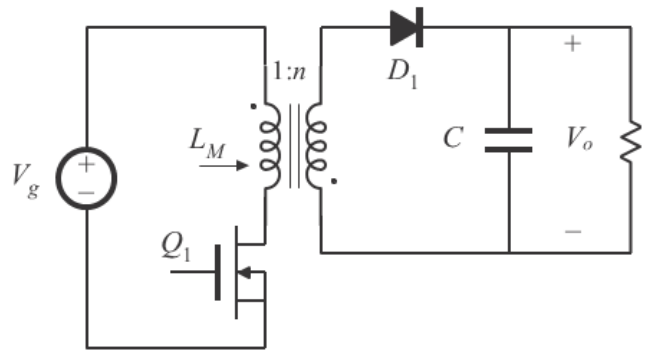
From (1) the duty cycle results as:

$$D = \frac{V_o}{V_o + nV_g} = \frac{5}{5 + \frac{1}{40} \cdot 300} = \frac{5}{12.5} = 0.4 \tag{2}$$

The dc current drawn from the supply voltage is

$$I_g = D \cdot I_{L_M} \qquad (3)$$

Using average power balance, $P_g = V_g \cdot I_g = P_o$, the same I_g current is:

$$I_g = \frac{P_o}{V_g} \qquad (4)$$

From (3) and (4) it follows that $I_{L_M} = \dfrac{P_o}{D \cdot V_g} = \dfrac{100}{0.4 \cdot 300} = 0.83A$

a) The rms transistor current is given by $I_{Qrms} = \sqrt{D} \cdot I_{L_M} = \sqrt{0.4} \cdot 0.83 = 0.53A$

b) Transistor voltage stress is $V_Q = \dfrac{V_g}{1-D} = \dfrac{300}{1-0.4} = 500V$. Estimated in a different way, this voltage

stress is, $V_Q = V_g + \dfrac{V_o}{n} = 300 + 5 \cdot 40 = 500V$, the same result.

c) Diode voltage stress is $V_D = V_o + n \cdot V_g = 5 + \dfrac{1}{40} \cdot 300 = 12.5V$

Embedded Systems

1. What are the relative advantages/disadvantages of RISC versus CISC architectures?

Answer: C.-D. Căleanu, Embedded Systems. Course Notes, 2017

https://intranet.etc.upt.ro/~EMBEDDED_SYS/Course , CHAPTER 3, slides 13-15.

- The most common types of general-purpose ISA architectures implemented in embedded processors are:
- Complex Instruction Set Computing (CISC) Model
- Reduced Instruction Set Computing (RISC) Model

Complex Instruction Set Computing (CISC) Characteristics:

- A large number of instructions each carrying out different permutation of the same operation
- Instructions provide for complex operations
- Different instructions of different format

- Different instructions of different length
- Different addressing modes
- Requires multiple cycles for execution

Reduced Instruction Set Computing (RISC) Characteristics:

- Fewer instructions aiming simple operations that can be executed in a single cycle
- Each instruction of fixed length – facilitates instruction pipelining
- Large general-purpose register set – can contain data or address
- Load-store Architecture – no memory access for data processing instructions

2. Which is the role of the barrel shifter? Present its block diagram and enumerate the basic operations which could be performed with it. Illustrate the concept with an assembly language example.

Answer: C.-D. Căleanu, Embedded Systems. Course Notes, 2017
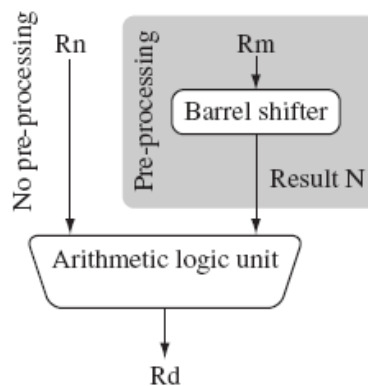
https://intranet.etc.upt.ro/~EMBEDDED_SYS/Course , CHAPTER 4, slide 43, 44.

Enables shifting 32-bit operand in one of the source registers left or right by a specific number of positions within the cycle time of instruction

Basic Barrel shifter operations: Shift left, right, rotate

Facilitates fast multiply, division and increases code density

Example: mov r7, r5, LSL #2 - Multiplies content of r5 by 4 and puts result in r7

3. Present possible implementations for the non-volatile memory. What could be store in it?

Answer: C.-D. Căleanu, Embedded Systems. Course Notes, 2017

https://intranet.etc.upt.ro/~EMBEDDED_SYS/Course , CHAPTER 5, slide 10-12.

- Mask ROM
   - Used for dedicated functionality

- – Contents fixed at IC fab time (truly write once!)
- ERPOM (erase programmable)
  - – Requires special IC process (floating gate technology)
  - – Writing is slower than RAM, EPROM uses special programming system to provide special voltages and timing
  - – Reading can be made fairly fast
  - – Rewriting is slow
    - Erasure is first required, EPROM – UV light exposure, EEPROM – electrically erasable
- Flash
  - – Uses single transistor per bit (EEPROM employs two transistors)
  - – A flash memory provides high density storage with speed marginally less than that of SRAM's
  - – Write time is significantly higher compared to DRAM
- On-chip non-volatile storage is used for storage of:
  - – Configuration information
  - – Executable code that runs on core processors
  - – Recorded data: repeated write

4. The SPI protocol (bus configuration, comparison with I2C).

Answer: C.-D. Căleanu, Embedded Systems. Course Notes, 2017
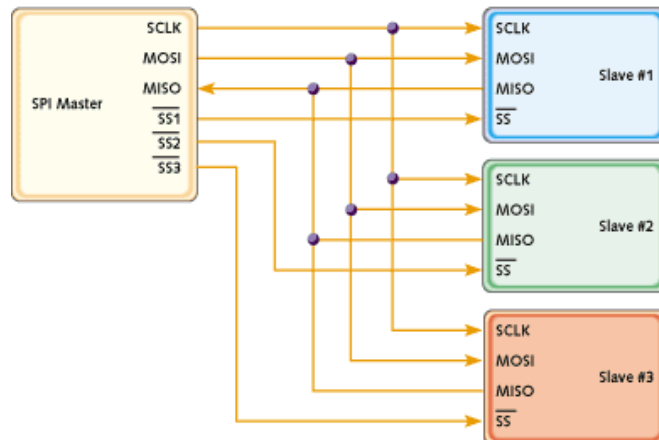
https://intranet.etc.upt.ro/~EMBEDDED_SYS/Course , CHAPTER 6, slide 27, 28.

SPI Bus Configuration
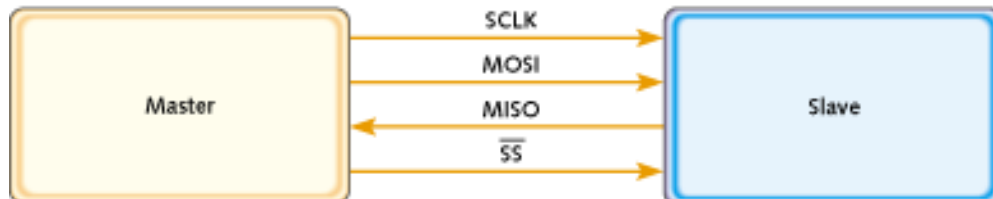
- Synchronous serial data link operating at full duplex
- Master/slave relationship
- 2 data signals:
  - – MOSI – master data output, slave data input
  - – MISO – master data input, slave data output
- 2 control signals:
  - – SCLK – clock
  - – /SS – slave select   (no addressing)

SPI vs. I²C

- For point-to-point, SPI is simple and efficient
    - Less overhead than I2C due to lack of addressing, plus SPI is full duplex.
- For multiple slaves, each slave needs separate slave select signal
    - More effort and more hardware than I2C

5. Define the kernel and its responsibilities.

Answer: C.-D. Căleanu, Embedded Systems. Course Notes, 2017

https://intranet.etc.upt.ro/~EMBEDDED_SYS/Course , CH. 8, slide 12, 13.

Kernel's definition

- Most frequently used portion of OS
- Resides permanently in main memory
- Runs in privileged mode
- Responds to calls from processes and interrupts from devices

Kernel's responsibility

- Managing Processes
- Context switching: alternating between the different processes or tasks
    - Various scheduling algorithms
    - Scheduling: deciding which task/process to run next
- Various solutions to dealing with critical sections
    - Critical sections = providing adequate memory-protection when multiple tasks/processes run concurrently

6. **Take the following mathematical formula and create a code sequence for it in assembly for the ARM microcontroller:**

$$y = \begin{cases} x + 1, & x < 3 \\ 5, & x = 3 \\ x + 2, & x > 3 \end{cases}$$

   R0 should contain variable x and R1 variable y.

Solution:

```
        MOV R0, #3          ;for testing, can be any value

        CMP R0, #3
        BLT less
        BGT greater
        MOV R1, #5
        B stop
greater ADD R1, R0, #2
        b stop
less    ADD R1, R0, #1

stop
     END
```

7. **Take the following assembly code for ARM microcontroller and translate it into the equivalent C code:**

```
                    MOV R0, #0
                    MOV R1, #1

        eti1    ADD R0, R0, R1
                ADD R1, R1, #1
                CMP R1, #10
                BLE eti1
```

Solution:

```
        int i, x;
        x= 0;
        for (i=1; i <= 10;  i++)
                x=x+i;
```

ELECTRONIC EQUIPMENT TESTING

<u>Theoretical subjects</u>

1. Test levels: definition and description of each level's characteristics
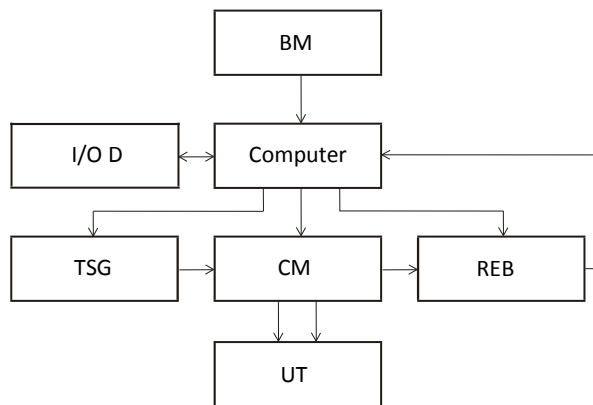
Solution:

There are three levels of test:

- *electronic component (including circuit board),*
- *electronic board (equipped circuit board - components),*
- *equipment.*

a) *failed test at the component level – the component should not be mounted on the circuit board, economically has been proven that this test (also done by the producer) should be repeated by the user*
b) *can be done through "nails bed testing" (complicated and costly for realizing the test equipment) or through "test stimulus generation" (simple connection, but costly for implementing the test program)*
c) *depends on the equipment: normally involves broken connections between functionally correct boards. Can involve some more sophisticated equipment, such as the signature analyzer.*

2. Architecture of an automatic test equipment and characteristics of each block.

Solution:

```
                    ┌──────────┐
                    │    BM    │
                    └────┬─────┘
                         │
                         ▼
   ┌──────────┐     ┌──────────┐
   │  I/O D   │◄───►│ Computer │◄──────────────┐
   └──────────┘     └────┬─────┘               │
         ┌───────────────┼───────────────┐     │
         ▼               ▼               ▼     │
   ┌──────────┐     ┌──────────┐    ┌──────────┐
   │   TSG    │────►│    CM    │───►│   REB    │─┘
   └──────────┘     └────┬─────┘    └──────────┘
                         ▼
                    ┌──────────┐
                    │    UT    │
                    └──────────┘
```

UT – unit under test

TSG – test signal generator

REB – responses evaluation block

CM – connection matrix

BM – back-up (external) memory

I/O D – Input/Output Devices

3. Principles of testing the static and dynamic parameters of a digital integrated circuit

Solution:

The static parameters of an IC are:

- *Input and output voltages*
- *Input and output currents*

They are stable during the test. A time should be given from powering the IC, in order to allow stabilisation of the transition factors.

Test should be carried out under the less favourable conditions: minimum power supply value, maximum circuit's charge, etc.

The Test Signal Generator is formed by a number of Programmable Voltage Sources (PVS) and Programmable Constant Current Generators (PCCG).

The Responses Evaluation Block is a simple measuring instrument.

The Connection Matrix is formed by a number of relays or commuting transistors.

The dynamic parameters of an IC are:

- *the transition times: t_{tLH}, t_{tHL}*
- *the propagation times: t_{pLH}, t_{pHL}*

Those parameters might be defined by fixed or percentage thresholds. The realisation of the testing structure depends on that definition.

The evaluation should be done in the worst functional case: power supply, charge, etc

The Test Signal Generator uses digital programmable pulse generators: fronts, length, amplitude, polarity, filling factor, etc

The Responses Evaluation Block is typically formed by a counter, with "start" and "stop" commanded by the discriminated fronts of the tested parameter.

4. Principles of the modular activation method for processors' functional tests
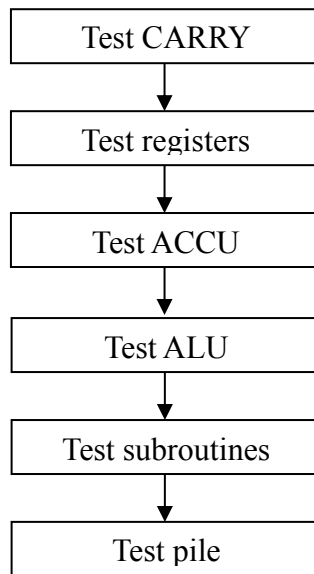
Solution:

The processor is divided on hierarchical levels (both functional and hardware).

The test is done based on a self-test program.

The program starts with the basic level

*Each tested level can be used for testing superior levels*

A possible testing strategy:

```
┌─────────────────────┐
│     Test CARRY      │
└─────────────────────┘
          │
          ▼
┌─────────────────────┐
│    Test registers   │
└─────────────────────┘
          │
          ▼
┌─────────────────────┐
│      Test ACCU      │
└─────────────────────┘
          │
          ▼
┌─────────────────────┐
│      Test ALU       │
└─────────────────────┘
          │
          ▼
┌─────────────────────┐
│   Test subroutines  │
└─────────────────────┘
          │
          ▼
┌─────────────────────┐
│      Test pile      │
└─────────────────────┘
```

In order to start, a minimum level of components and functions should be correct! - KERNEL

The kernel should include: initialization circuits, program counter, address bus, data bus, instruction decoder, clock circuit

The hardware test of the kernel might be done through the following strategy:
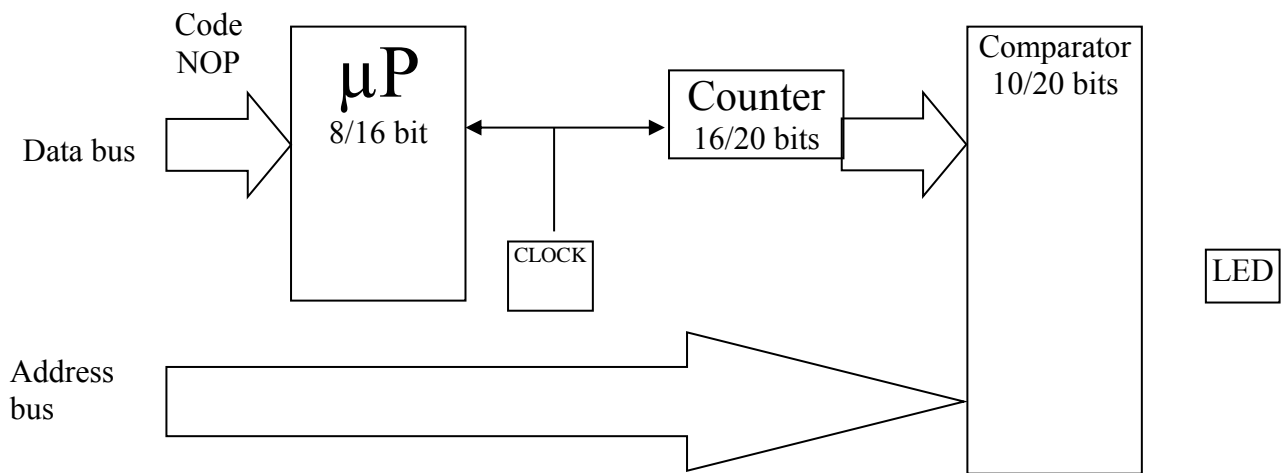
If the kernel test is passed, the following circuits are functionally corrects:

-   *Clock circuit*
-   *Initialization circuit*
-   *Program counter*
-   *Address bus*
-   *Data bus – partially*

- *Instruction decoder – partially*
- *Instruction register – partially*

5. Principles for testing the Bit Error Rate for a telecommunication digital network
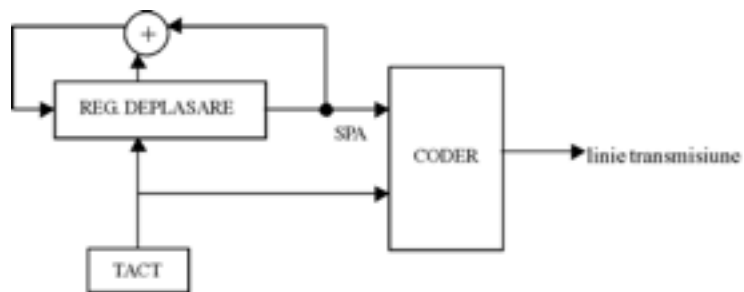

Solution:

The Bit Error Rate (BER) is the ratio between the number of error bits and the total number of transmitted bits

BER estimation can be done:

- *in-service*
- *out-of-service*

The test signal is a pseudorandom binary sequence, standardized through the 0.151 CCITT recommendation.

The principle of the test generator used at the transmission end is:

The problem is: avoiding the lock on 0 of the shift register.

At the reception end, there is a similar signal generator that provides the same sequence as at the emitter end, that is compared to the one received through the telecom network.

The problem is to ensure synchronization between the two generators.
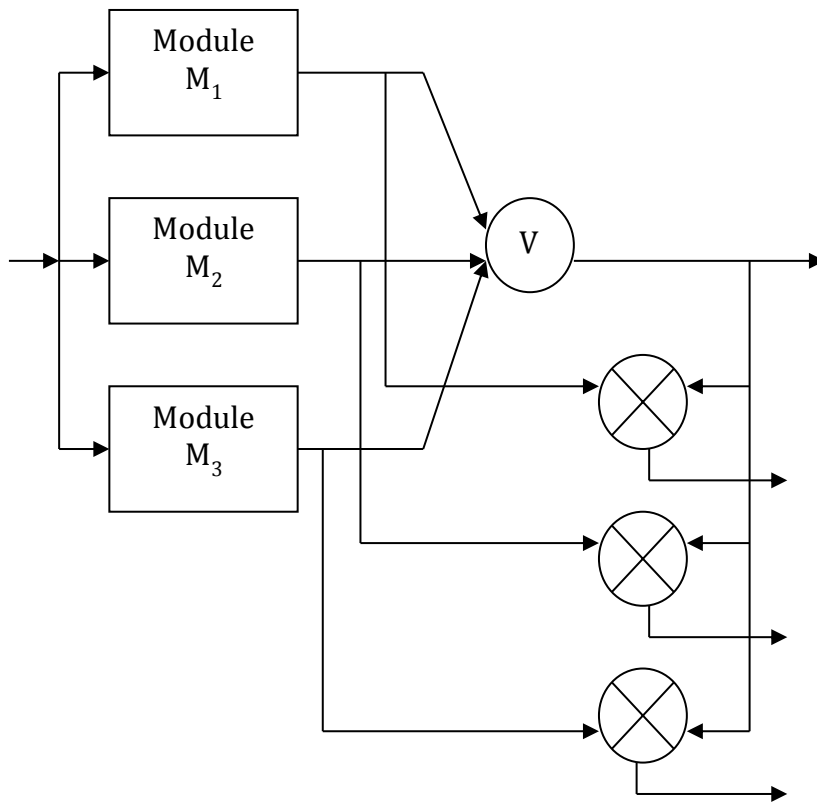

Applications

1. Build-up a random sequence generator starting from implementing the polynomial $G(x) = x^8 + x^7 + x^4 + x^2 + 1$ for use in a parallel signature analyser.

Solution:
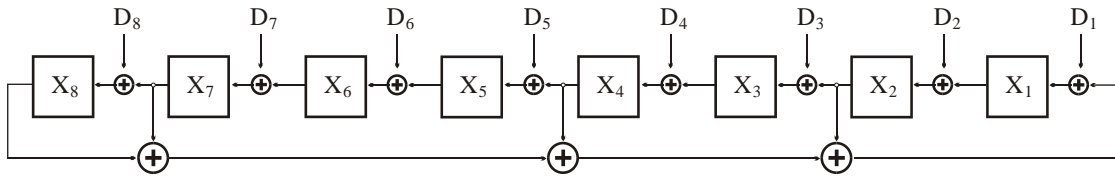
2. Imagine a schema for detecting the faulty module in a triple modular redundant electronic structure
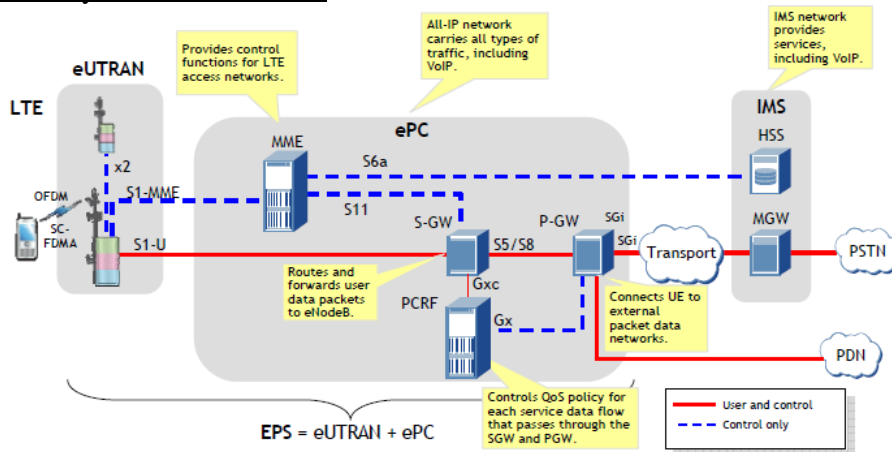
Solution:

INTEGRATED DIGITAL NETWORKS

1. LTE architecture – enumerate the functional blocks and briefly explain their main functions for 3 of them.
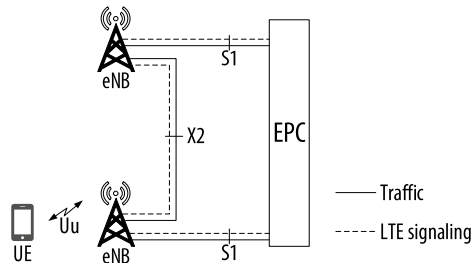
LTE – System Architecture

IMS - IP Multimedia Subsystem - delivers voice and other multimedia services over IP in mobile networks
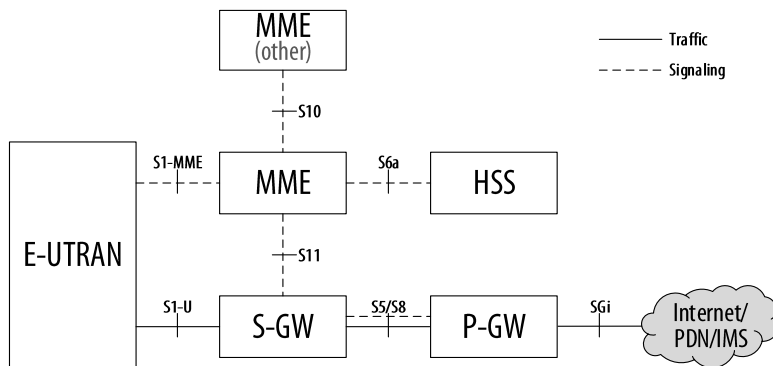EPS – Evolved Packet System

E-UTRAN
- Handles radio communications between mobile terminals and EPC

- Evolved Node B (**eNB**) – the only component of E-UTRAN
    - Base station which controls mobiles in one ore more cells
    - Uses analog and digital signal processing functions to
        - send radio transmissions to all mobile on DL
        - receive radio transmissions from all mobiles on UL
    - Controls low-level operation of all mobiles using signaling messages (e.g. handover commands)
    - Interfaces
        - S1 to EPC
        - X2 to nearby eNBs (optional)
    - used for signaling and packet forwarding during handover
   - A mobile communicates with only one eNB at a time – **serving eNB**

EPC

MME
(other)

———— Traffic
- - - - Signaling

S10

S1-MME

E-UTRAN

MME

S6a

HSS

S11

S1-U

S-GW

S5/S8

P-GW

SGi

Internet/
PDN/IMS

- HSS – Home Subscriber Server
  - central database with information about all network operator's subscribers
- P-GW – PDN (Packet Data Network) GateWay
  - exchanges data with one or more external devices or PDNs
    - servers of the network operator
    - internet
    - IMS (IP Multimedia Subsystem)
  - P-GWs which provide connection to PDNs to a mobile terminal do not change during the lifetime of the connections
    - when the mobile switches on, it is assigned to a default P-GW at to have always on connectivity to a default PDN (e.g.internet)
    - later, the mobile can be assigned to one or more additional P-GWs for connectivity to other PDNs (e.g. IMS, corporate network)
  - several P-GWs in a typical network
- S-GW – Serving GateWay
  - router forwarding data between enBs and P-GWs
  - a S-GW serves the mobiles in a given geographical area
  - a connected mobile is assigned to a single S-GW
    - the S-GW may change during the connection if the mobile moves to the coverage area of another S-GW (handover/cell reselection)
  - S-GW service area – geographical area covered by 1 or more S-GWs
    - mobiles moving within a S-GW service area don't need to change the S-GW
  - Several S-GWs in a typical network
- MME – Mobility Management Entity
  - controls high level operation of mobiles through signaling messages
    - security
    - management of data streams (not on the radio interface)
  - a MME serves the mobiles in a given geographical area
  - a connected mobile is assigned to a single MME (serving MME)
    - the MME may change during the connection if the mobile moves to the coverage area of another MME (handover/cell reselection)
  - MME pool area – geographical area served by 1 or more MME
    - mobiles moving within an MME pool area don't need to change serving MME
  - Several MMEs in a typical network

2. Digital modulation techniques used in LTE – list, characteristics and comparison.

https://intranet.etc.upt.ro/~DIG_INT_NET/course/7_LTE_20.pdf, 22, 26

Digital modulations in LTE

- QPSK
    - Robust, less efficient
- QAM
    - High efficiency, less robust
- Gray encoding
    - neighboring symbols in constellation
    - only 1 bit different
    - limits the number of bit errors

Binary encoded QPSK

Gray encoded QPSK
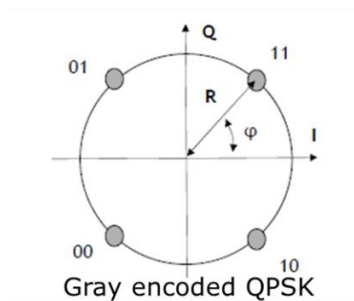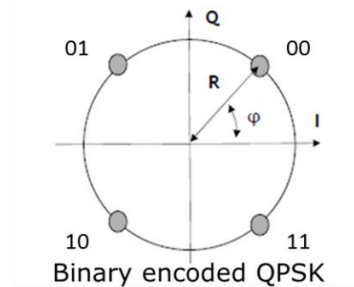
LTE modulation summary

Modulation	No. of symbols	Bits/ symbol	Bit rate/ Baud rate	Robustness	No. of amplitudes	No. of phases
QPSK (4QAM)	4	2	2/1	+	1	4
16QAM	16	4	4/1	+/−	3	12
64QAM	64	6	6/1	−	9	52

3. OFDMA – principle of sub-carrier orthogonality, application in LTE.
https://intranet.etc.upt.ro/~DIG_INT_NET/course/7_LTE_20.pdf, 30, 35, 38

LTE multiple access

- OFDMA (Orthogonal Frequency Division Multiple Access)
    - Multiple equally spaced orthogonal subcarriers
    - Data stream is split in multiple sub-streams
        - Each sub-stream modulates a subcarrier using 64QAM, 16 QAM or QPSK
    - used on DL
- SC-FDMA (Single Carrier Frequency Division Multiple Access)
    - Uses only a reduced number of sub-carriers (contiguous group)
    - lower PAPR (Peak to Average Power Ratio) compared to OFDMA
    - used on UL
    - Unappropriate for DL
        - eNB uses all available sub-carriers
        - eNB transmits to multiple UEs at the same time

OFDMA

- Data stream is split into multiple sub-streams
- Frequency bandwidth divided into multiple sub-bands (sub-carriers)
- Each data sub-stream modulates (QPSK, 16QAM or 64QAM) a sub-carrier
- Sub-carrier orthogonality
    - the signal sent on a carrier does not interfere with signals sent on other carriers
    - achieved by proper choice of sub-carrier spacing
        - $\Delta f = 1/T$ (T – OFDMA symbol period)
        - in LTE, $T = 66.7\ \mu s \Rightarrow \Delta f = 15$ kHz

Orthogonal carriers

- OFDM spectrum example:
    - 4 carriers spaced by $\Delta f = 1/T$ (= 15 kHz for LTE)
    - at each carrier frequency (e.g. f_{c2})
        - there is a maximum of the spectrum of the signal transmitted on that carrier (f_{c2})
        - all spectra of signals transmitted on other subcarriers (f_{c1}, f_{c3} and f_{c4}) are crossing 0 $\Rightarrow$ $\Rightarrow$ orthogonality (no interference)
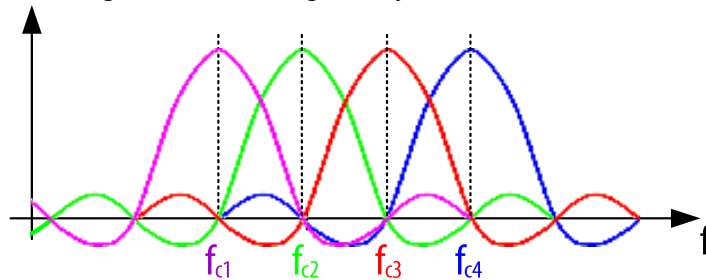
4. LTE uplink power control.

Need for power control
- Wireless channel conditions vary
  - UE position in the cell
  - time
- The transmitted signal is adjusted to compensate for the varying conditions
  - Power control
  - Rate control – MCS (Modulation and Coding Scheme)
- Power control
  - Transmit power is varied → constant data rate
- Rate control
  - Data rate is varied → constant transmit power
- On DL eNB transmits with maximum power on all PRBs
- On UL UE follows a power control strategy to adjust transmit power
  - reduce power consumption
  - reduce inter-cell interference
- UEs situated near the center of the cell use lower transmit power
- UEs situated near the cell edge use higher transmit power
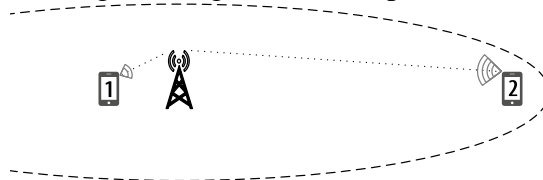
UL power control
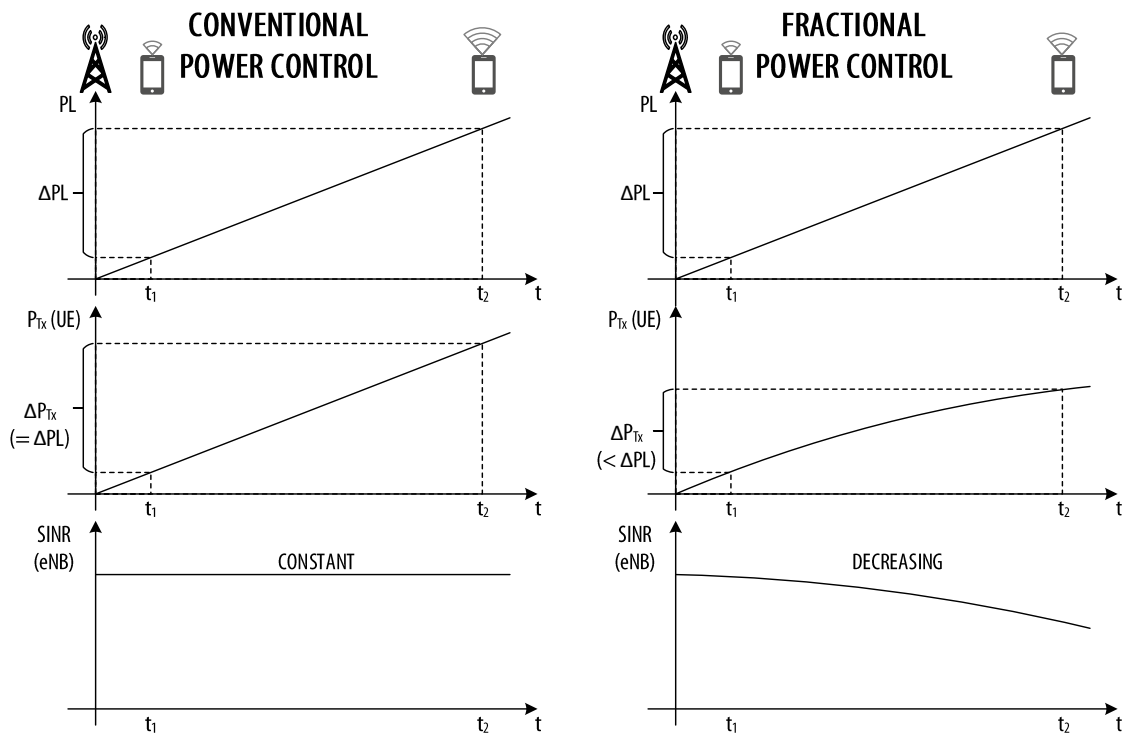
- Conventional power control
  - Target: fully compensate the PL (Path Loss)

- Amount of increase in UE transmit power = amount of increase in path loss
    - Advantage: maintains constant SINR at eNB side
    - Disadvantage: increased inter-cell interference
- Fractional power control
    - Target: partially compensate the path loss
    - Amount of increase in UE transmit power < amount of increase in path loss
    - Advantages:
        - reduced intercell interference
        - increased average cell throughput
    - Disadvantage: worse SINR from UEs near cell edge => lower data rates
- Open loop power control
    - UE calculates the DL path loss

    PL = Reference eNB transmit power − measured RSRP
    - UE uses conventional or fractional power control attempting to compensate the PL

   PTx = min(Desired eNB Power + αPL; Pmax)
        - α – fractional power control factor; $0 < \alpha \leq 1$
        - Pmax – UE maximum transmit power (typically 23 dBm)
    - UE receives no feedback from eNB for power control

1. UE measures RSRP
2. UE calculates PL
3. UE adjusts transmit power
conventional / fractional

- Closed loop power control
    - Allows dynamic UE transmit power adjustment based on TPC commands issued by eNB
    - TPC (Transmit Power Control) commands have up to 2 bits (1 QPSK symbol)
        - Absolute values
        - Accumulated values
        - Transmitted
            - inside UL scheduling grant
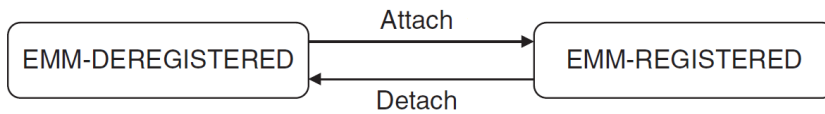   - on a separate power control channel
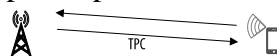
1. UE receives TPC command
2. UE adjusts transmit power
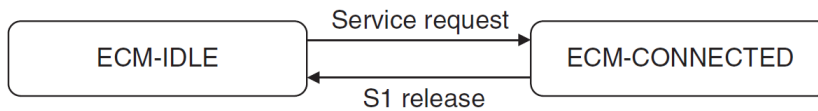based on TPC

5. LTE – state diagrams

- EPS Mobility Management (EMM) state diagram
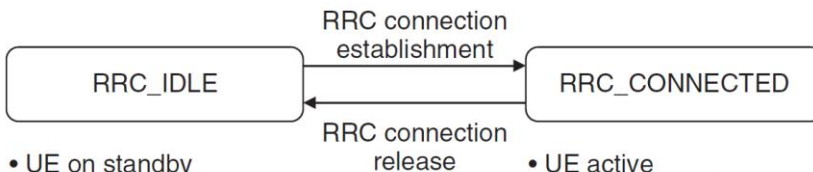
| EMM-DEREGISTERED | Attach → ← Detach | EMM-REGISTERED |

- UE switched off / out of coverage
- No serving MME or S-GW
- No IP address
- No default EPS bearer

- UE registered with EPC
- Serving MME and S-GW allocated
- IP address allocated
- Default EPS bearer exists

- EPS Connection Management (ECM) state diagram

| ECM-IDLE | Service request → ← S1 release | ECM-CONNECTED |

- UE on standby
- MME knows tracking area list
- No S1 or radio bearers
- Paging
  Service requests
  Tracking area updates

- UE active
- MME knows serving eNB
- All bearers exist
- Any communication possible

- Radio Resource Control (RRC) state diagram

| RRC_IDLE | RRC connection establishment → ← RRC connection release | RRC_CONNECTED |

- UE on standby
- No serving eNB
- No S1 or radio bearers
- Cell reselection
- Paging
  RRC connection requests

- UE active
- Serving eNB allocated
- All bearers exist
- Handovers
- Any communication possible

6. **Calculate the frequency bandwidth required for a LTE-advanced cell to achieve a data rate of 450 Mb/s on DL with the spectral efficiency of 15 bits/s/Hz. Find a solution to obtain the necessary bandwidth using standard LTE frequency bands.**

https://intranet.etc.upt.ro/~DIG_INT_NET/course/7_LTE_20.pdf, 5, 31, 32

Hints:

Use the rule of three to find the amount of necessary bandwidth

15 bits/s …………………………………………..1 Hz

450Mbits/s ………………………………………..B Hz

B=….

The standard LTE channel bandwidths are 1.4, 3, 5, 10 and 20 MHz.

If the value of B is lower or equal to 20 MHz, then it is possible to use 1 frequency band with the smaller standard bandwidth equal or higher than B.

If B exceeds 20MHz, then carrier aggregation must be used (obtain the desired bandwidth by combining up to 5 frequency bands having standard bandwidths). Examples of carrier aggregation to obtain 40 MHz:

2 bands of 20 MHz or

1 band of 20 MHz and 2 bands of 10 MHz.

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7. **Calculate the maximum theoretical DL data rate for a LTE cell with a bandwidth of 10 MHz, using 2 antennas.**

https://intranet.etc.upt.ro/~DIG_INT_NET/course/7_LTE_20.pdf, 60, 75

Hints:

- D = (no. of antennas) × (no. of symbols/s) × (no. of bits/symbol)

- Max. theoretical data rate
  - All REs are taken into account => **168 [RE/PRB]**
  - 10 MHz => 50 PRBs/ms => 50[PRB]×168[RE/PRB] = 8400 [RE/ms]
  - Each RE carries 1 symbol => **$8.4×10^6$ [symbols/s]**
  - The modulation is 64QAM (**6 bits/symbol**) for all symbols (in order to transmit as many bits/symbol as possible)
  - D_t = 2[antennas] × $8.4×10^6$ [symbols/s] × 6[bits/symbol]= **100.8 Mb/s**

DATA COMMUNICATIONS

References : https://intranet.etc.upt.ro/~DATA_COM/

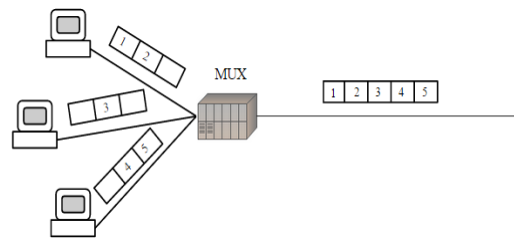https://cv.upt.ro/course/view.php?id=3241

Theory 1

Multiplexing. Definition. Types description: synchronous TDM, statistical (asynchronous) TDM, FDM, wavelength multiplexing.

Multiplexing is a method by which multiple data streams, coming from different sources, are combined and transmitted over a single communication channel. The device that combines multiple data streams into a single one is called multiplexer (MUX). The reverse process, called demultiplexing, extracts the original data streams at the destination. This task is realized by a device called demultiplexer (DMUX). Some methods used for multiplexing data are:synchronous time-division multiplexing (STDM), asynchronous time-division multiplexing (ATDM)- called also statistical multiplexing, frequency-division multiplexing (FDM), wavelength multiplexing
In synchronous TDM, shown in figure below, there is a static allocation for each source, the same interval of time, in the same order. Time slots are allocated to sources, whether they have or not information to transmit. In this way, the channel capacity might be wasted.
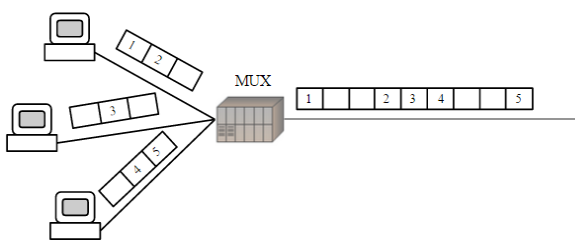
a)Synchronous time-division (STDM)   b) Statistical TDM

An efficient alternative of the synchronous TDM is the *Statistical TDM*, also called *Asynchronous TDM (ATDM).* In ATDM is allocated bandwidth to a variable number of users as needed, in a dynamical manner as shown in figure above. In this case the transmitted packets may not arrive in order and the demultiplexer cannot simply identify the source based on data packets reception. For this the sequence order and the explicit address of source must be specified in data packets. Static TDM is carried out at the physical layer in the OSI model and TCP/IP model. Statistical multiplexing is carried out at the data link layer and above.
In *Frequency Division Multiplexing (FDM),* the available bandwidth of a single physical medium is subdivided into several independent frequency channels. FDM assigns "frequency ranges" to each user on a medium, all signals being transmitted at the same time, each one using a different frequency. The most common examples of FDM are the radio/TV broadcasting, where multiple radio signals at different frequencies pass through the air simultaneously, or the cable television, where many TV channels are carried on a single cable, at the same time.

Wavelength division multiplexing (WDM) is a particular case of FDM used in optical fiber communications. The inputs and outputs of WDM are different wavelengths of light (colors).

Theory 2

What is thermal noise? Give the expression of the power spectral density of a thermal noise, indicating the meaning of each parameter and the measurement units.

Thermal noise is generated by the thermal agitation of electrons. It has the followings characteristics:

- Uniformly distributed in frequency, generally modelled as white noise, as it can be seen in nearby figure
- It cannot be eliminated
- It is present in all electronic devices and transmission media
- It is dependent on temperature
- It is particularly significant for satellite communications
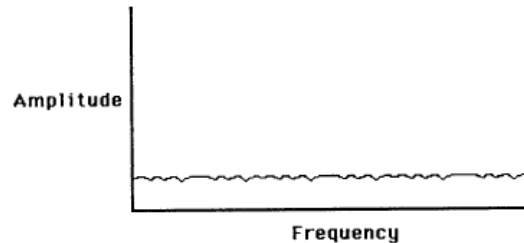
The amount of thermal noise in the band of 1Hz is $N_0=kT$.
Where:

- N_0 is the noise power spectral density [Watts/Hz] –independent of frequency
- k-is Boltzmann constant=$1,38 \cdot 10^{-23}$ J/°K
- T –is temperature in Kelvins degrees (absolute temperature)

The amount of thermal noise in a bandwidth of B Hz is dependent on frequency:
$$N = kTB$$

Notice that, larger the bandwidth, larger will be the amount of thermal noise "seen" by the receiver. We may therefore say that larger bandwidth transmissions are more affected by the thermal noise, compared to the narrower bandwidth transmissions. Expressed in dBW:

$$N = 10\log k + 10\log T + 10\log B =$$
$$-228,6 + 10\log T + 10\log B$$

Theory 3

What is multipath? In which types of communications can it appear? Which are its causes and its effects on signal transmission?

In radio communication, multipath is the propagation phenomenon that results in multiple copies of the original signal reaching the receiving antenna, by two or more paths.
Multipath appears in:

- in terrestrial, fixed microwaves communications
- in mobile communications

Causes of multipath include atmospheric ducting, ionospheric reflection and refraction, and reflection from water, bodies and terrestrial objects such as mountains and buildings.

There are three important propagation mechanisms of generating multipath:

- Reflection when a signal encounters a surface large relative to its wavelength (a ground reflective wave for mobile communications)
- Diffraction at the edge of impenetrable objects, large compared to signal wavelength
- Scattering when a signal encounters a surface of size on the same order of magnitude as its wavelength or less.

Due to the existent obstacles the signal can be reflected, diffracted, scattered so that multiple copies of the same signal, with varying delays and attenuation might be received. In extreme cases, the receiver may capture only the reflected signal and not the direct one.

Multipath propagation causes multipath interference, including constructive and destructive interference, and phase shifting of the signal; destructive interference causes fading. This may cause a radio signal to become too weak in certain areas to be received adequately, so multipath propagation can be detrimental in radio communication systems.

Effects of multipath:
- One or more delayed copies of a pulse may arrive at the same time, as the primary pulse for a subsequent bit. This may generate Intersymbol interference (ISI), constructive or destructive.
- Multiple copies of a signal may arrive at different phases. If phases add destructively, the signal level relative to noise declines, making detection more difficult.

As a conclusion , multipath propagation may be detrimental in radio communication systems.
To deal with this problem multipath signals may be reinforced and/or cancelled.
The OFDM technique was especially designed to deal with the multipath problem.

Theory 4

Write down the formula describing the Shannon capacity theorem for noisy channels, explain the meaning of each parameter and interpret the relation.

The Shannon capacity theorem widely used form is:

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

Where:
- C is the channel capacity, the rate at which data can be transmitted over a given communication path, under given conditions, expressed in bps.
- S/N is signal to noise ratio of power **expressed in absolute value, not expressed in dB!**

Shannon's formula expresses the theoretical maximum data rate that can be achieved referred to as the error free capacity. In practice much lower rates can be achieved. One reason is that only white noise is considered in the above relation. (not impulse noise, nor attenuation)

Shannon proved that if the actual information rate on a channel is less than the error-free capacity, then it is theoretically possible to use a suitable signal code to achieve error-free transmission through the channel. Currently the Shannon capacity cannot be achieved even in an ideal white noise condition of transmission, due to encoding issues, such as coding length and complexity environment.

For a given level of noise the channel capacity can be increased either by increasing S or B. But increasing signal may increase the nonlinearities in the system and and this may determine an increase in the intermodulation noise. Also notice that increasing B means that thermal noise will be increased too, and this decreases the S/N ratio.

Supposing that the error free capacity is achieved, the number M of signalling levels, symbols, needed can be determined considering the Hartley-(Nyquist) law : $C = 2 B \log_2 M$

Theory 5

What is Pulse Code Modulation? Which is the error introduced in PCM and which are the ways of reducing it?

PCM is an analog to digital signal digitizing technique, based on the sampling theorem which states: If a signal x(t) is sampled at regular intervals of time and at a rate higher than ! twice the highest significant signal frequency, then the samples contain all the information of the original signal. The function x(t) may be reconstructed from these samples by the use of a low-pass filter.
Ex for voice data considering B=4 kHz a data rate of R=8kbps is sufficient to characterize the voice signal.
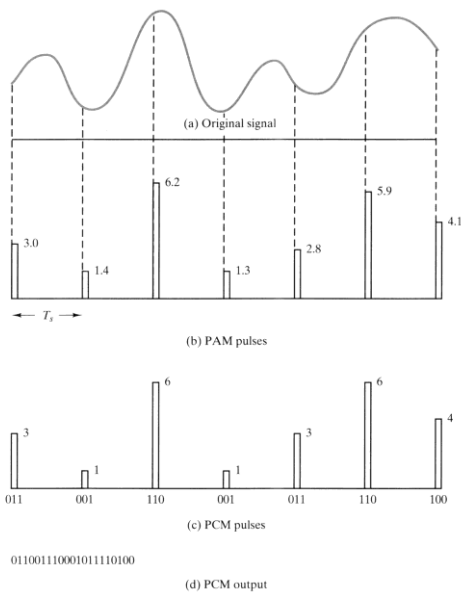
See the PCM example fom the figure:

•   **Original signal**
has continuous amplitudes in its dynamic range

•   **PAM** - **signal** is a discrete pulse train with constant amplitude values, which are periodically samples of the analogue signal

•   **Quantized PAM** signal has only the values that can be quantized by the words available (here by 3 bits words).To each analog PAM sample is assigned a binary code, which only approximates it. With a code having 2^n bits there will be obtained n quantization levels.

The digital signal consists of a block of *n* bits, where each *n*-bit number is the amplitude of a PCM pulse. With a code having 2^n bits can be obtained n quantization levels, as examples: 2 bits – 4 quantization levels; 3 bits – 8 quantization levels, 4 bits -16 quantization levels.

By quantizing the PAM pulse, original signal is only approximated. This leads to quantizing noise or quantization error **.** Quantization error/noise is unavoidable, regenerated analog data will not be 100% the same as the original

To reduce the quantization error: we have to increase the quantization number of levels, and/or to increase the sampling rate (to decrease sampling time interval).

The problem with equal spacing (the same step of quantization –uniform quantization) is that the mean absolute error for each sample is the same, regardless of the signal level. Consequently, lower amplitude values are relatively more distorted. By using a greater number of quantizing steps for signals of low amplitude, and a smaller number of quantizing steps for signals of large amplitude, an important reduction in the overall signal distortion is achieved. This type of PCM is called nonlinear quantization.

Problem 1

Let's consider a bipolar transmission in the baseband, with controlled ISI. After sampling the received signal, $r(t)$, the following samples values are obtained: r[1]=1, r[2]=0, r[3]=0, r[4]=2, r[5]=0, r[6]=-2, r[7]=0, r[8]=0, r[9]=-2, r[10]=0. The transmitter uses a precoding given by the following relation: $b_k = a_k \oplus b_{k-1}$

Consider that $b_0=0$. Find the data sequence at the output of the precoder b_k and the initial data sequence a_k.

Solution: Look over the following truth table implemented by the precoder:

a_k	b_{k-1}	$b_k = a_k \oplus b_{k-1}$
0 (-1)	0 (-1)	0 (-1)
1	0 (-1)	1
0 (-1)	1	1
1	1	0 (-1)

Observation Consider that bipolar signals are transmitted in the line.
Controlled ISI means that it is used a cosine filter in the transmission line and the received signal is given by the relation $r_k = b_k + b_{k-1}$

Case 1
Notice that if $a_k = 1$ (and known that $b_k = a_k \oplus b_{k-1}$).
From the table it can be observed that $b_k \neq b_{k-1}$
When it is used for shaping the cosine filter a controlled ISI can be achieved, because $r_k = b_k + b_{k-1}$ so it results $r_k = 0$

Case 2
If $a_k = 0$ and known that $b_k = a_k \oplus b_{k-1}$ .From the table it can be observed that $b_k = b_{k-1}$.
Because $r_k = b_k + b_{k-1}$ it results either $r_k = 1+1 = +2$ or $r_k = -1-1 = -2$
In conclusion, the decision on a_k will be done directly by using the folowing relation :

$$a_k = b_k \oplus b_{k-1} = \begin{cases} 0, & if \ r_k = \pm 2 \\ 1, & if \ \ \ r_k = 0 \end{cases} \quad (3.1)$$

The sequence b_k will be computed using the precoding formula.

$$b_1 = a_1 \oplus b_0 = 1 \oplus 0 = 1; \ b_2 = a_2 \oplus b_1 = 1 \oplus 1 = 0; \ b_3 = a_3 \oplus b_2 = 1 \oplus 0 = 1 \ ... \ .$$
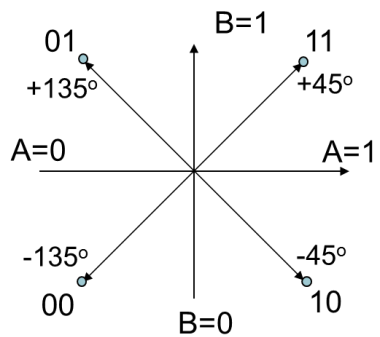
The following sequences will be obtained:

k	1	2	3	4	5	6	7	8	9	10
r_k	1	0	0	2	0	-2	0	0	-2	0
a_k	1	1	1	0	1	0	1	1	0	1
b_k	1	0	1	1	0	0	1	0	0	1

Problem 2

The binary sequence 1110010010 is brought to the input of a QPSK modulator. Choose the correct corresponding phases of the phase modulated signals. Determine the data rate R and the modulation rate D if the bit duration T_b=2ms.

Solution

In QPSK (Quadrature PSK), the modulation symbol is a group of two bits, as in the constellation diagram below:

Four possible signaling symbols M=4 can be observed. For each symbol (signaling level) a phase shift occurs, as in the figure.

 Splitting the data sequence in groups of two bits it is obtained the following corresponding phases of the phase modulated signals

45°; -45° ; 135°; -135 °; -45°;

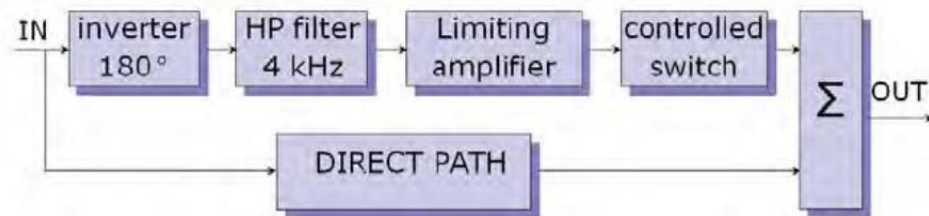$$R = \frac{1}{T_b} = \frac{1}{2.\,10^{-3}} = \frac{1000}{2} = 500bps$$

$$D = \frac{R}{log_2 M} = \frac{500}{log_2 4} = 250 \ Baud$$

TRANSMISII TELEFONICE

Noise = low level signal and medium to high frequency -> can be identified and rejected (nose gate)

Example: Dynamic Noise Limiter (DNL) from Philips

IN: noisy signal

OUT: signal with a SNR improved by 8 dB

 Advantage: compatible with any recording system and any playback system
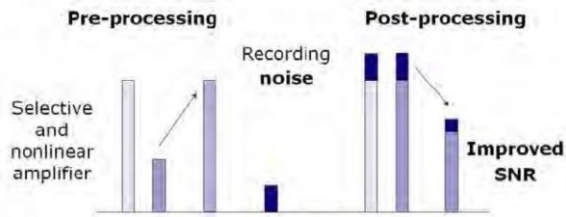
How it works:

　　1. during the pause between melodies

　　2. high level recorded music

　　3. low level recorded music

 Disadvantage: it cannot make the difference between noise and real signal

The system performs: - signal processing before recording

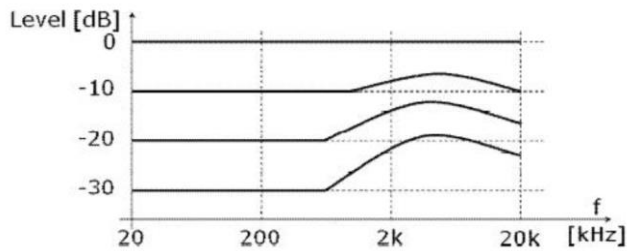　　　　　　 - opposite processing after playback

Advantage: the real signal is not altered and obtained with a high SNR

Disadvantage: works on the same system (recording and playback)
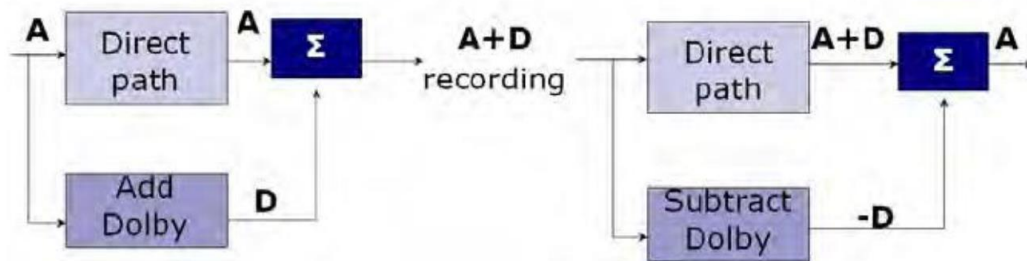
Dolby system

Noise:

- High frequency (1 ÷ 15 kHz)
- Low level (-20 ÷ -40 dB)

Dolby circuits:

- Amplify nonlinearly and selectively during recording
- Performs opposite processing when playing the signal
- Increases SNR with 9 dB

Uniform quantization:

- **decision levels** (analog input)
  - □ **uniform**
- **quantization levels** (digital output)
  - □ **uniform**
- **quantization steps** (q) are **constant:**
  - □ for low level signal
  - □ for high level signal
- **quantization noise** (error): -q/2 ÷ q/2

RESULT
- **low level signal with** constant quantization error ⇨ **low SNR**

- **high level signal with** constant quantization error ⇨ **high SNR**

CONCLUSION
- **low general SNR** (⇔ **low quality**)

Nonuniform quantization:

- **decision levels** (analog input)
  - □ **non-uniform**
- **quantization levels** (digital output)
  - □ **non-uniform**
- **quantization steps** (q) are **different:**
  - □ for low level signal
  - □ for high level signal
- **quantization noise** (error):
  - □ **non-constant**

RESULT
- **low level signal with** low quantization error ⇨ **high SNR**

- **high level signal with** high quantization error ⇨ **high SNR**

CONCLUSION
- **high general SNR** (⇔ **high quality**)
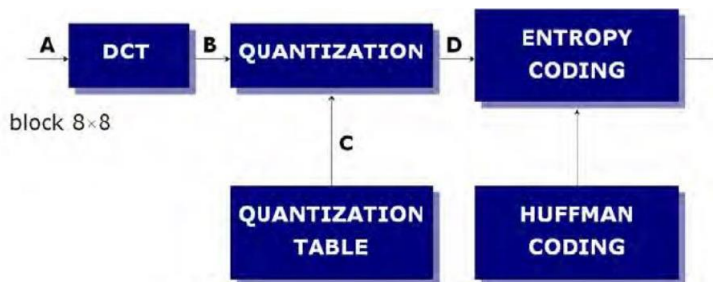
DCT

… transforms time representation block A (lot of data points)

… in frequency representation block B (few data points – few frequency components) QUANTIZATION

… reduces non-uniformly the accuracy of coefficients, D, according to the quantization table C (4 tables implemented in JPEG algorithm):

> … low frequency with higher accuracy (small steps, non-zero values)

> … high frequency with lower accuracy (big steps, most values equal to zero)

ENTROPY CODING

… is used to obtain data compression

… zig-zag scanning is used to obtain long sequences of "zero"

> „ RLE (Run-Length Encoding) offers an excellent compression

> „ Huffman coding is used to obtain higher compression factor

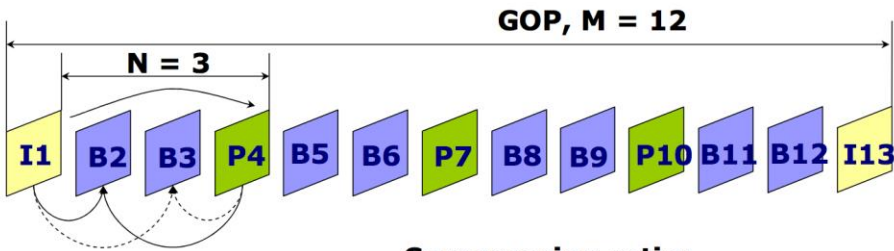„ DCT (similar to Fourier transform) converts data from

> „ from time domain

>> … 8×8 pixels block: … rows 0-7 and columns 0-7

> „ to frequency domain

> … 8×8 coefficient matrix: … 00 position - DC coefficient - average of the 8×8 block

>> … 01-77 positions - AC coefficients - low frequency in the upper left corner high frequency elsewhere

Zig-zag sequencing

„ starts with low frequency coefficients (non-zero),

„ then high frequency coefficients (zero);

„ results a long sequence of zeroes, after a few significant values, easy entropy coding (RLE, Huffman)

T4. Define the image types involved in MPEG compression and explain the compression principle.

- Spatial redundancy removal DCT
- Temporal redundancy removal
  - Motion compensated forward
  - Bidirectional prediction (interpolation)

MPEG uses three types of images

- Image I
  - JPEG coded
  - independent to the sequence of moving images
  - robust coding
  - independent to precedent errors
  - low compression factor

- Image P
  - a predicted image is estimated (motion-compensated forward)
  - the difference between actual and predicted image is coded
  - sequence of predictions may propagate possible errors
  - higher compression factor
- Image B
  - a bidirectional interpolated image is calculated, using I and P images
  - very good estimation
  - may propagate errors
  - best compression factor

GOP, M = 12

N = 3

I1 B2 B3 P4 B5 B6 P7 B8 B9 P10 B11 B12 I13

Compression ratio:

❑ determined by M and N;
❑ M – period of I images   usually, N=3
❑ N – period of P images   usually, M = 12

RE-ORDERING the images sequence for transmission:

1	4	2	3	7	5	6	10	8	9	13	11	12
I	P	B	B	P	B	B	P	B	B	I	B	B

T5.   Present the color TV signals.

o Luminance of a (color) image is used in black-and-white television:
$$Y = 0.3 \times R + 0.59 \times G + 0.11 \times B$$

o Using R, G, B signals would be incompatible with the old TV system.
o Compatible color TV systems use:
   o Y – luminance (for correct processing by black-and-white TV sets)
   o C – chrominance (color information only, no brightness information)
     => color difference signals: R-Y, G-Y, B-Y

From the 4 signals, only 3 are used:

o luminance
   o $Y = 0.3 \times R + 0.59 \times G + 0.11 \times B$
o chrominance (2 color difference)
   o $R-Y = 0.7 \times R - 0.59 \times G - 0.11 \times B$
   o $B-Y = -0.3 \times R - 0.59 \times G + 0.89 \times B$

Luminance

$$E_Y = 0.3 \times E_R + 0.59 \times E_G + 0.11 \times E_B = 0 \div 1$$

Color difference

$$E_{R-Y} = 0.7 \times E_R - 0.59 \times E_G - 0.11 \times E_B = -0.7 \div 0.7$$

$$E_{G-Y} = -0.3 \times E_R + 0.41 \times E_G - 0.11 \times E_B = -0.41 \div 0.41 \text{ (not transmitted)}$$

$$E_{B-Y} = -0.3 \times E_R - 0.59 \times E_G + 0.89 \times E_B = -0.89 \div 0.89$$

Compute the data rate for both stereo music standard (hi-fi), and voice standard (telephony). Consider a hi-fi tune of 4 minutes and 30 de seconds. What is the memory capacity (in bits [b] and in bytes [B]) needed to store this tune? The same tune is compressed in MP3 format, using a data rate of 128kbps. Compute the compression rate.

Standard for stereo music (hi-fi), audio band of 20kHz:

- 2 channels
- sampling frequency: 44,1kHz
- n=16 bits per sample
⇨ data rate: D = 2 * 44,1kHz * 16 = 1411200 bps

Standard voice (telephony), audio band of 3,4kHz:

- 1 channel
- sampling frequency: 8kHz
- n=8 bits per sample
⇨ data rate: D = 1 * 8kHz * 8 = 64 kbps = 64000 bps

C = D*t

C = memory capacity; t = duration; D = data rate

Hi-fi:

t = 4*60+30 = 270 s

D = 1411200 bps => C = 270 * 1411200 = 381,024 Mb = 381,024/8 MB = 47,628 MB

MP3 compression, D=128 kbps:

C = D*t = 128000 * 270 = 34,56 Mb = 34,56/8 MB = 4,32 MB

Compression rate = C(hi-fi)/C(MP3) = 47,628/4,32 = 11,025;

P2.   Compute the data rate for the digital TV signal for standard format (4:2:2) and for 4:2:0 format. Compute the number of samples broadcasted for a line in 4:2:0 format.

Digital signals data rate:

D = n * fS;   n = bits per sample; fS = sampling frequency

Digital TV standard:

n = 8

fS = 13,5 MHz

data rate of a digital TV signal:

D = D(Y) + D(R-Y) + D(B-Y) = n(Y)* fS (Y) + n(R-Y)* fS (R-Y) + n(B-Y)* fS (B-Y)

n(Y) = n(R-Y) = n(B-Y) = 8; <- same number of bits per sample for both luminance and chrominance

 Luminance is sampled with standard sampling frequency (luminance is processed for every pixel) => fS (Y) = fS = 13,5 MHz

Format 4:2:2:

Chrominance is sampled with half of the standard sampling frequency (chrominance is processed every second pixel both on x axis and y axis)

 => fS (R-Y) = fS (B-Y) = fS /2 = 13,5/2 MHz = 6,75 MHz

   ⇨ D(4:2:2) = 8 * 13,5 + 8 * 6,75 + 8 * 6,75 [Mbps] = 108 + 54 + 54 [Mbps] = 216 Mbps

Format 4:2:0:

- Alternate for rows: 4:2:2 format and 4:0:0 format
- For a row in format 4:2:2 the computation is the same as before
- For a row in format 4:0:0: D(4:0:0) = D(Y) = 108 Mbps
⇨ D(4:2:0) = (D(4:2:2) + D(4:0:0))/2 = (216 + 108)/2 = 162 Mbps (medium data rate)

Number of samples broadcasted, format 4:2:0:

Row in format 4:2:2:

N(4:2:2) = N(4:2:2,Y) + N(4:2:2,R-Y) + N(4:2:2,B-Y)

N(4:2:2,Y) = row length * fS (Y) = 64µs (standard) * 13,5MHz = 64*13,5 = 864 samples

N(4:2:2,R-Y) = N(4:2:2,B-Y) = row length * fS (R-Y/B-Y) = 64*6,75 = 432 samples

   ⇨ N(4:2:2) = 864 + 432 + 432 = 1728 samples
Row in format 4:0:0:

N(4:2:2) = N(4:2:2,Y) = 864 samples

N(4:2:0, medium) = (1728 + 864)/2 = 1296 samples