Major Courses

RADIO COMMUNICATIONS

Exercise 1.

For a radio link find:
a) the maximum LOS distance between the transmitter and the receiver, if the antennas are placed at heights: $h_T = 2.13\ m$, and $h_R = 1.8\ m$;
b) the transmitted power imposed to obtain a received power $P_R = 0.1\ mW$, at this distance, if link frequency is $f = 2.4\ GHz$, and antennas gains are $G_T = G_R = 35\ dB$.

SOLUTION---

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

b) $\lambda = \dfrac{c}{f} = \dfrac{3 \cdot 10^8}{2.4 \cdot 10^9} = 12.5cm$

b1) all parameters are expressed in decibels:

$$\left[\frac{P_T}{P_R}\right]_{dB} = 20 \cdot \lg \frac{4 \cdot \pi \cdot D_{LOS}}{\lambda} - G_T[dB] - G_R[dB] = 20 \cdot \lg \frac{4 \cdot \pi \cdot 10^4}{12.5 \cdot 10^{-2}} - 35dB - 35dB = 50dB \Rightarrow$$

$$\Rightarrow P_T = 10^{-4} \cdot 10^{50/10} = 10W$$

b1) all parameters are expressed in absolute values:

$$G_T = G_R = 10^{35/10} = 10^{3.5} \approx 3162$$

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

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

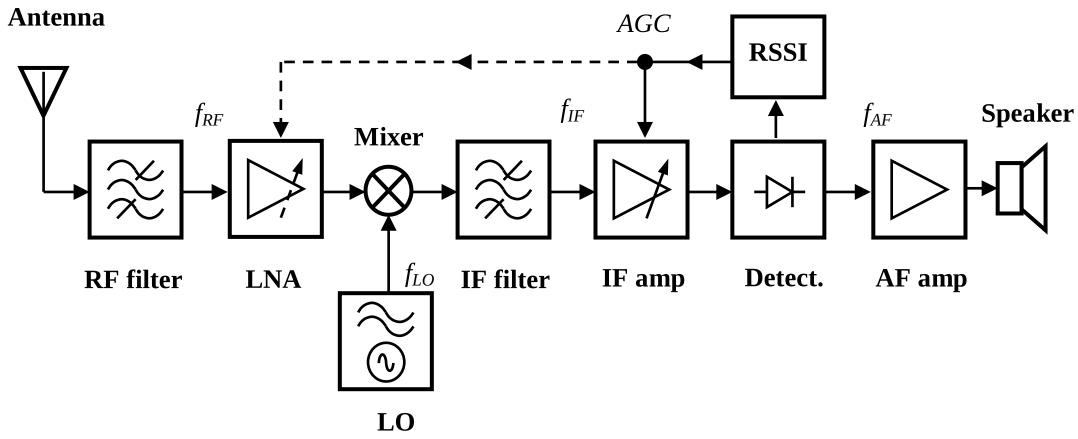
Exercise 2.

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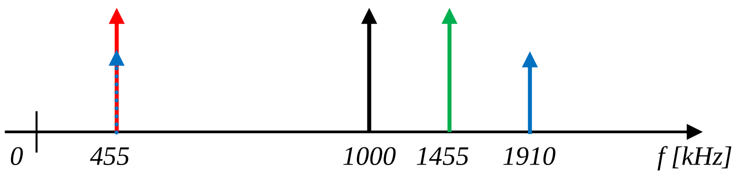
a) $f_{IF} = f_{LO} - f_{RF} \Rightarrow f_{LO} = f_{RF} + f_{IF}$

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

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

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

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



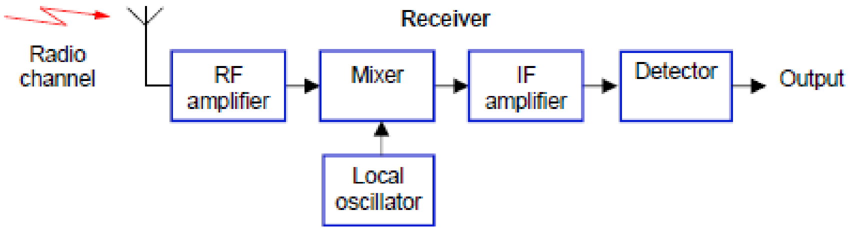
Theory 1.

Draw and explain the main blocks of a radio receiver.

Course no.1 (2016), slides 13-14.
Site address: https://intranet.etc.upt.ro/~RADIOCOM/

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).

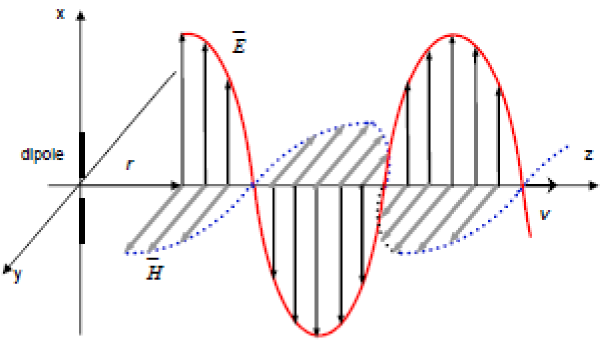
The user information is obtained after detection.

Theory 2.

Represent and give expressions for electric / magnetic fields intensities, *E(t, r)* and *H(t, r)*, with parameters signification (for a vertical dipole).

Course no.2 (2016), slides 11-12.
Site address: https://intranet.etc.upt.ro/~RADIOCOM/

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:

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

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

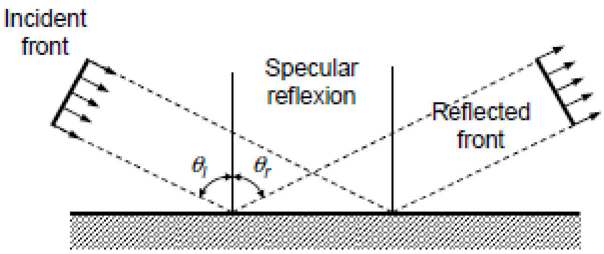
where:
v = EM wave velocity [m/s],
$\omega = 2\cdot\pi\cdot f$ = wave angular frequency,
$\lambda = v/f$ = wavelength,
$k = 2\cdot\pi/\lambda$ = phase constant of the wave.

Theory 3.

Draw and explain specular and diffuse reflections.

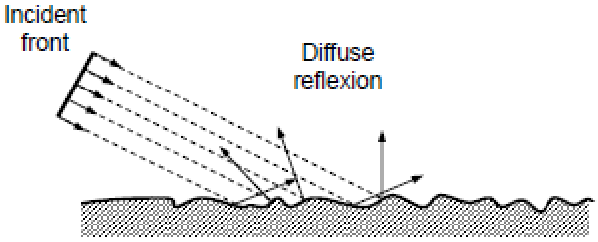
Course no.2 (2016), slides 29-30.
Site address: https://intranet.etc.upt.ro/~RADIOCOM/

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.



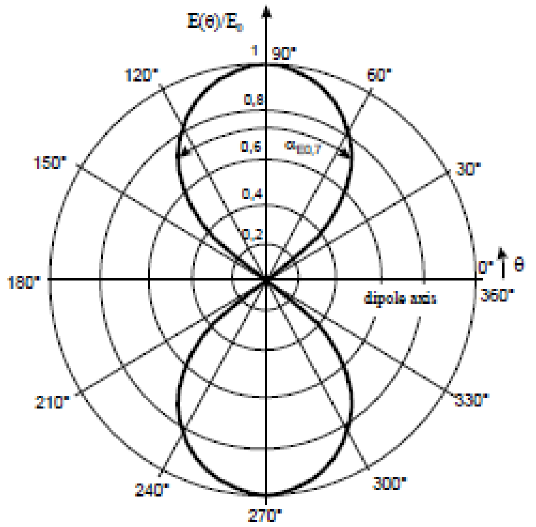
Diffuse reflection, which occurs mainly at shorter radio wavelengths, is the case where objects (walls and so on) can be considered as very rough on the scale of the wavelength.

Theory 4.

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

Course no.4 (2016), slides 37-38.
Site address: https://intranet.etc.upt.ro/~RADIOCOM/

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.

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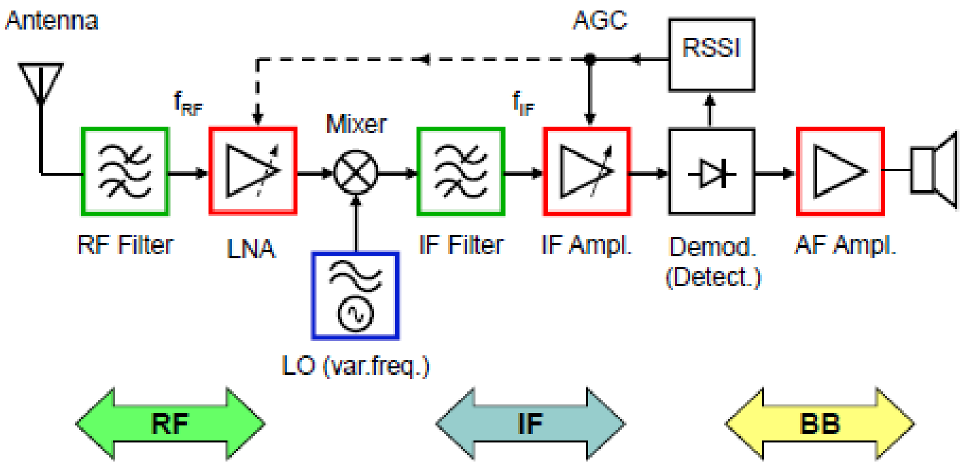
Theory 5.

Draw and explain the main blocks of an AM superheterodyne receiver.

Course no.6 (2016), slide 12 and some more.
Site address: https://intranet.etc.upt.ro/~RADIOCOM/

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.

POWER ELECTRONICS

1. Define the *total harmonic distortion* coefficient (*THD*) for a periodic signal *x(t)* and the *power factor* (*PF*) at a port with periodic voltage *v(t)* and current i*(t)*. *Power factor* formula for *sinusoida*l input voltage and nonlinear load.

Solution

The *total harmonic distortion coefficient* (THD) is defined as:

$$THD = \frac{rms\ value\ without\ \ fundamental}{rms\ \ fundamental} = \frac{\sqrt{X_0^2 + \frac{1}{2}\sum_{n=2}^{\infty} X_n^2}}{\frac{X_1}{\sqrt{2}}} \qquad (1)$$

where X_0 is the dc component and X_n are the amplitudes of the harmonics in the *x(t)* signal.

The *power factor* (*PF*) is a *figure of merit* that measures how efficient energy is transmitted through a port. It is defined as the ratio between *average* power and *apparent* power:

$$PF = \frac{P}{S} \cdot \frac{P}{V_{rms} \cdot I_{rms}} = \frac{\int_{t_0}^{t_0+T} v(t) \cdot i(t) dt}{\sqrt{\frac{1}{T}\int_{t_0}^{t_0+T} v(t)^2\, dt} \cdot \sqrt{\frac{1}{T}\int_{t_0}^{t_0+T} i(t)^2\, dt}} \qquad (2)$$

where t_0 is an arbitrary time moment in steady-state.

With a *sinusoidal voltage*, current harmonics do not lead to average power but *only* the *fundamental*. However, current harmonics increase the rms current and hence they decrease the power factor. As in a sinusoidal voltage only the fundamental is present, then $v(t) = V_1 \cos(2\pi ft - \varphi_1)$ and following that $P = \frac{1}{2} V_1 I_1 \cos(\varphi_1 - \theta_1)$; $V_{rms} = \frac{V_1}{\sqrt{2}}$; $I_{rms} = \sqrt{I_0^2 + \frac{1}{2}\sum_{n=1}^{\infty} I_n^2}$. Using these relationships in (2), it results that:

$$PF = \frac{\frac{I_1}{\sqrt{2}}}{\sqrt{I_0^2 + \frac{1}{2}\sum_{n=1}^{\infty} I_n^2}} \cdot \cos(\varphi_1 - \theta_1) = K_{di} \cdot K_\theta$$

with

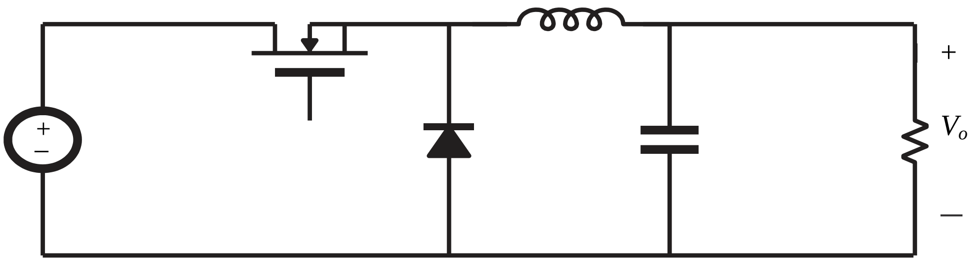
$$K_{di} = \frac{\frac{I_1}{\sqrt{2}}}{\sqrt{I_0^2 + \frac{1}{2}\sum_{n=1}^{\infty} I_n^2}} = \text{current distortion factor}$$

$$K_\theta = \cos(\varphi_1 - \theta_1) = \text{displacement factor}$$

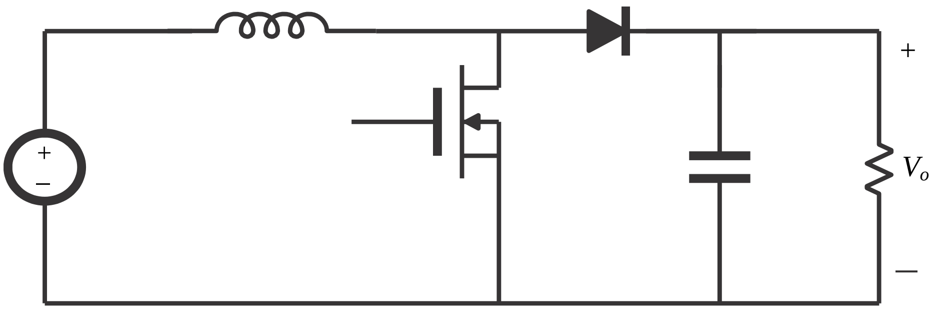
where φ_1, θ_1 denote the initial phase shift of the sinusoidal voltage and current fundamental, $i_1(t) = I_1 \cos(2\pi ft - \theta_1)$, respectively.

2. 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.

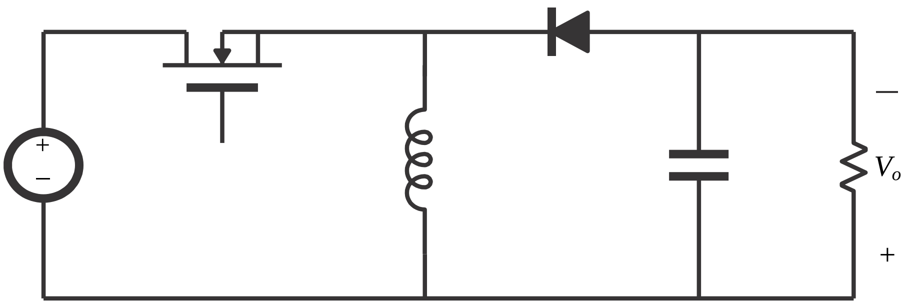
Solution



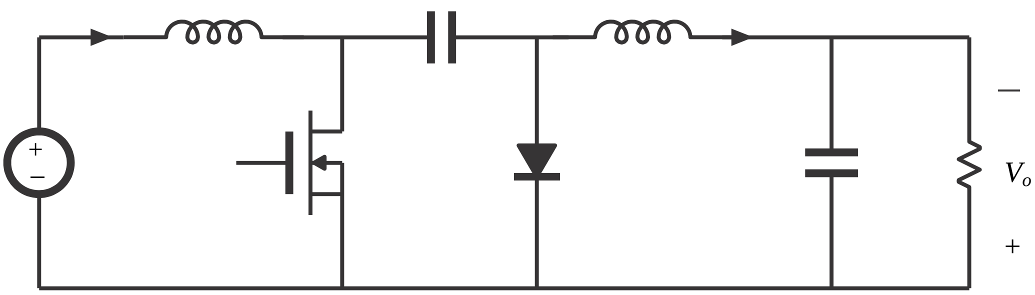
The buck converter. $M(D) = D$.



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



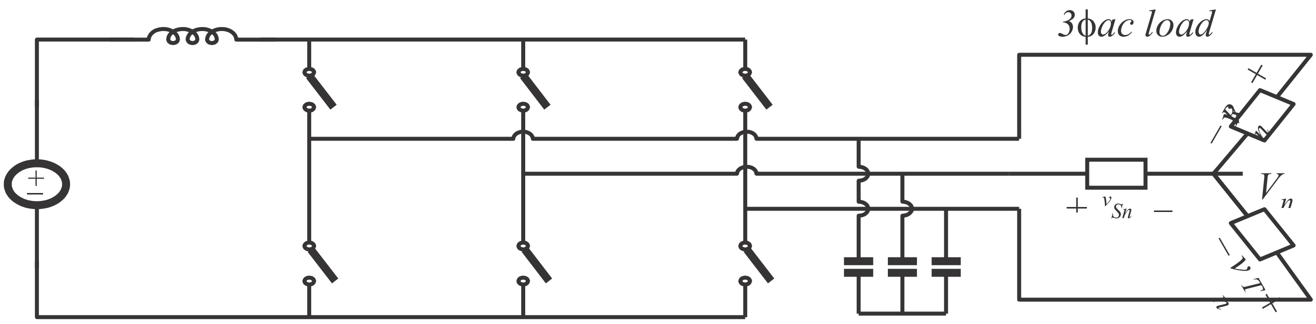
The buck-boost converter. $M(D) = \dfrac{D}{1-D}$.



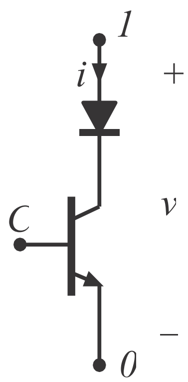
The Ćuk converter. $M(D) = \dfrac{D}{1-D}$.

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

Solution



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:



4. 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.

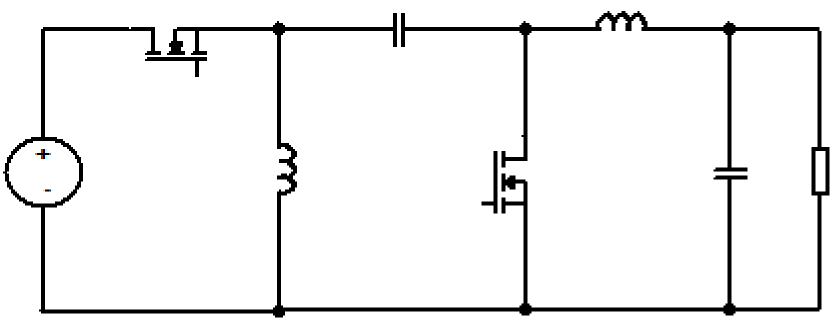
   Solution

   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:

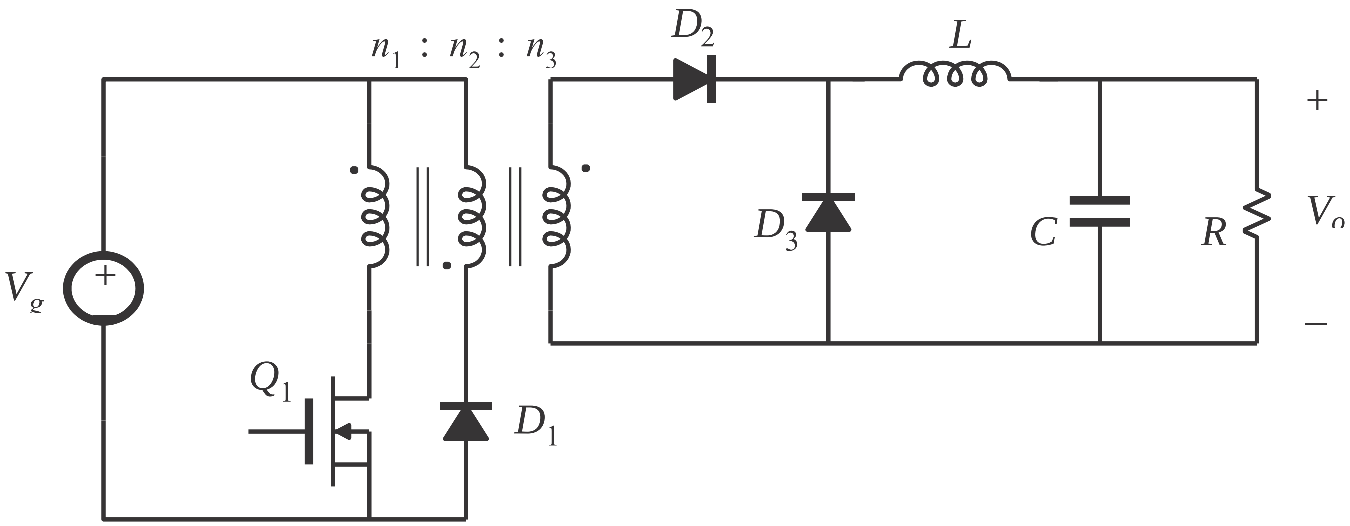
   - *High efficiency* because the conduction losses are reduced.
   - *Absence of discontinuous conduction mode*, because the diodes are the cause of DCM occurrence and in synchronous rectification they are missing.

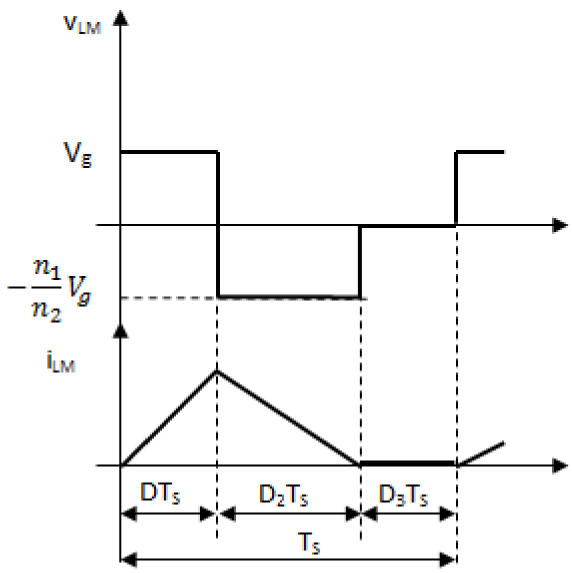
   The *synchronous Zeta converter* is presented below



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

<u>Solution</u>





From magnetizing inductor volt-second balance we get that

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

Solve for D_2:

$$D_2 = \frac{n_2}{n_1} \cdot D \tag{2}$$

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:

$$1 - D_2 - D \geq 0 \tag{3}$$

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

$$1 - \frac{n_2}{n_1} D - D \geq 0 \tag{4}$$

from which it follows that:

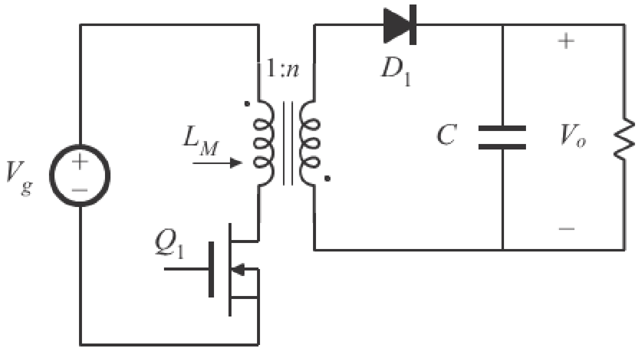
$$D \leq \frac{1}{1 + \dfrac{n_2}{n_1}} \tag{5}$$

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

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

6. 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.

<u>Solution</u>



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

$$I_g = D \cdot I_{L_M} \tag{3}$$

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

$$I_g = \frac{P_o}{V_g} \tag{4}$$

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

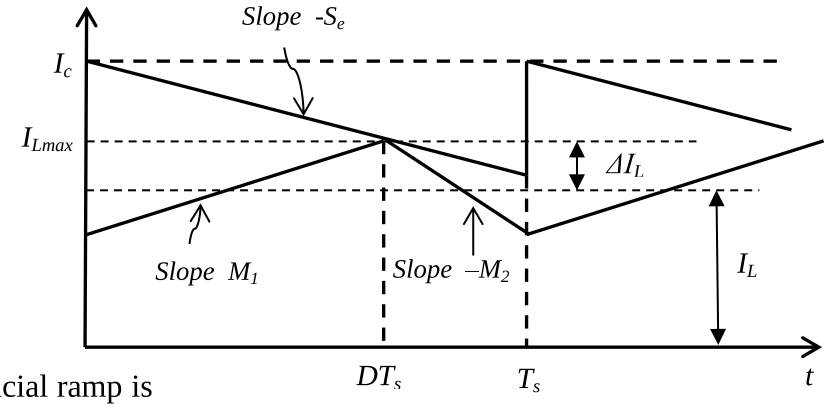
a) The rms transistor current is given by $I_{Qrms} = \sqrt{D} \cdot I_{L_M} = \sqrt{0.4} \cdot 0.83 = 0.53A$

b) Transistor voltage stress is $V_Q = \frac{V_g}{1-D} = \frac{300}{1-0.4} = 500V$. Estimated in a different way, this voltage stress is, $V_Q = V_g + \frac{V_o}{n} = 300 + 5 \cdot 40 = 500V$, the same result.

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

7. In order to convert a 9V voltage to a 5V voltage on a 10Ω load, a *peak current controlled buck* converter employing *deadbeat* control is used. Converter inductor is

$L = 400\mu H$, while the switching frequency is $f_s = 80kHz$. The current transducer transresistance is $R_f = 2\Omega$. Determine the value of the *control voltage* needed to achieve this voltage conversion.



Solution

In an ideal buck converter the dc output voltage is $V_o = D \cdot V_g$. Then the duty cycle results in:

$$D = \frac{V_o}{V_g} = \frac{5}{9} = 0.56 \quad (1)$$

As $D > 0.5$ it results that an artificial ramp is needed in order to get a stable system. In deadbeat control the slope of the artificial ramp, S_e, is equal to the absolute value of the falling slope, M_2, of the inductor current.

In a buck converter $M_2 = \frac{V_o}{L}$, hence

$$S_e = \frac{V_o}{L} \quad (2)$$

In peak current control the transistor is turned off when the inductor current equals the signal obtained by subtracting the artificial ramp from the current control I_c – see the figure. As a consequence we have:

$$I_c - S_e \cdot DT_s = I_{L\max} \quad (3)$$

But $I_{L\max} = I_L + \frac{1}{2}\Delta I_L$, where I_L is the dc inductor current and ΔI_L is the peak to peak inductor current ripple. In a buck converter $I_L = \frac{V_o}{R}$ and $\Delta I_L = \frac{(1-D)\cdot V_o}{Lf_s}$.

Replacing S_e from (2) and I_L and ΔI_L from the above relationships in (3), one obtains:

$$I_c - \frac{V_o}{L} \cdot DT_s = \frac{V_o}{R} + \frac{(1-D)\cdot V_o}{2Lf_s} \quad (4)$$

following that:

$$I_c = \frac{D \cdot V_o}{Lf_s} + \frac{V_o}{R} + \frac{(1-D)\cdot V_o}{2Lf_s} = \frac{0.56 \cdot 5}{400 \cdot 10^{-6} \cdot 80 \cdot 10^3} + \frac{5}{10} + \frac{(1-0.56)\cdot 5}{2 \cdot 400 \cdot 10^{-6} \cdot 80 \cdot 10^3} = 0.62A \quad (5)$$

In practice, voltages are those which are compared converting the current into a voltage. If the transducer transresistance is known, it immediately follows that the necessary control voltage is:

$$V_c = R_f \cdot I_c = 1.22V \quad (6)$$

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

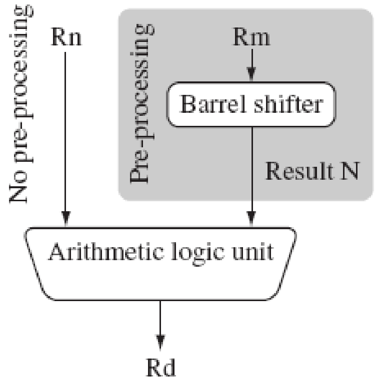
Reduced Instruction Set Computing (RISC) Characteristics:

- 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?

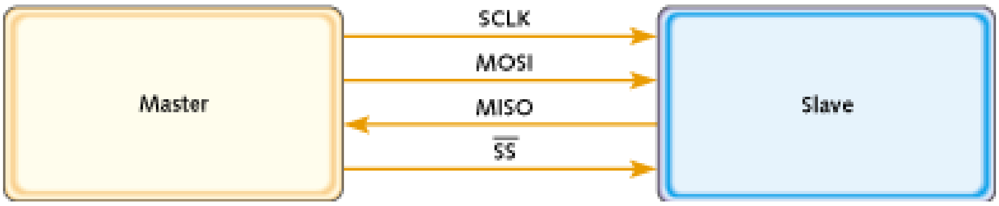
Answer: C.-D. Căleanu, Embedded Systems. Course Notes, 2017
https://intranet.etc.upt.ro/~EMBEDDED_SYS/Course , CHAPTER 5, slide 10-12.

- 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).

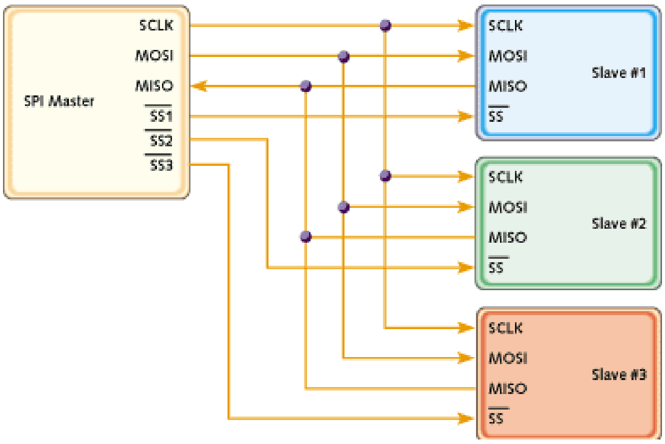
Answer: C.-D. Căleanu, Embedded Systems. Course Notes, 2017
https://intranet.etc.upt.ro/~EMBEDDED_SYS/Course , CHAPTER 6, slide 27, 28.

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. I^2C



- 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

Theoretical subjects

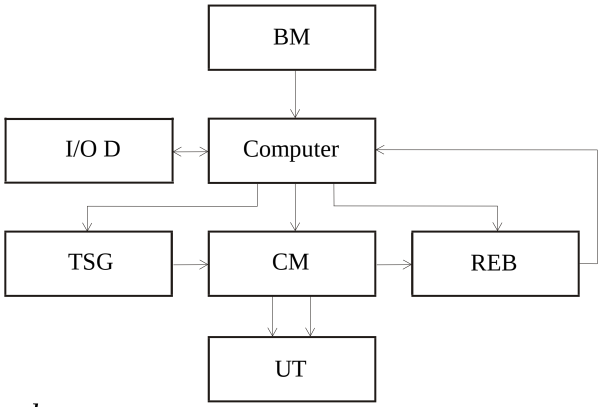
1. Test levels: definition and description of each level's characteristics

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:



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*

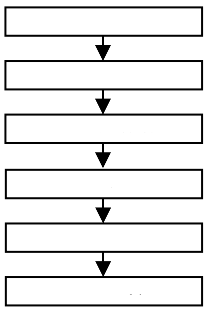
They are stable during the test. A time should be given from powering the IC, in order to allow stabilisation of the transition factors.
Test should be carried out under the less favourable conditions: minimum power supply value, maximum circuit's charge, etc.
The Test Signal Generator is formed by a 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}

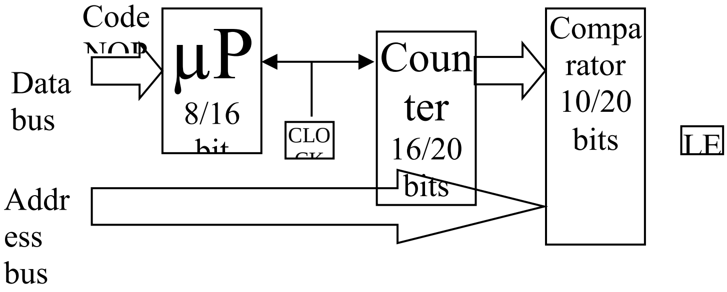
Those parameters might be defined by fixed or percentage thresholds. The realisation of the testing structure depends on that definition.
The evaluation should be done in the worst functional case: power supply, charge, etc
The Test Signal Generator uses digital programmable pulse generators: fronts, 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:



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:

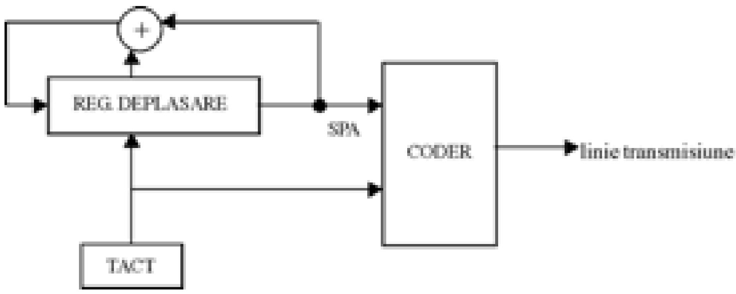
- *Clock circuit*
- *Initialization circuit*
- *Program counter*
- *Address bus*
- *Data bus – partially*
- *Instruction decoder – partially*
- *Instruction register – partially*

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:

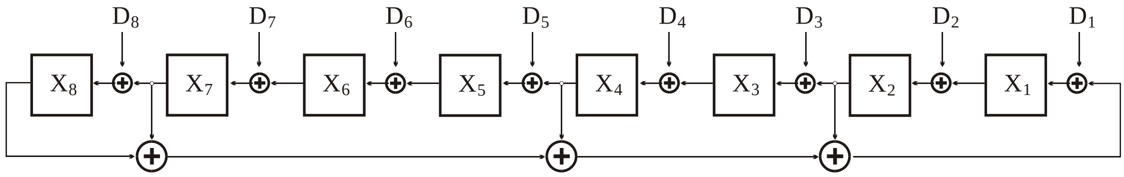


The problem is: avoiding the lock on 0 of the shift register.
At the reception end, there is a similar signal generator that provides the same sequence as at the emitter end, that is compared to the one received through the telecom network.
The problem is to ensure synchronization between the two generators.

Applications

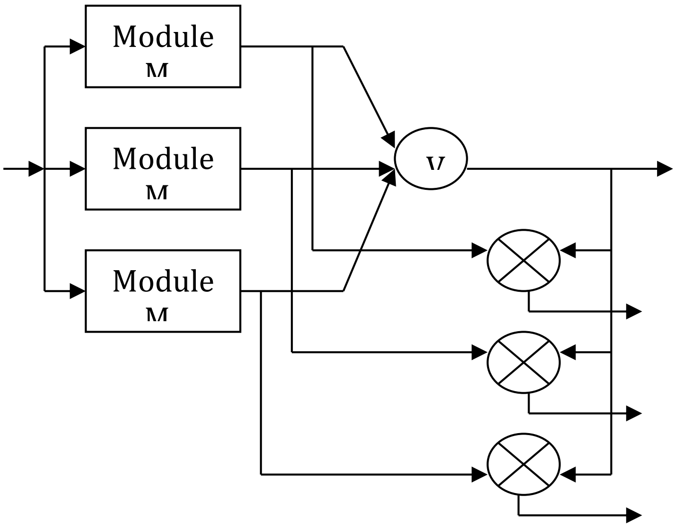
1. Build-up a random sequence generator starting from implementing the polynomial $G(x) = x^8 + x^7 + x^4 + x^2 + 1$ for use in a parallel signature analyser.

Solution:



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

Solution:



1. Principle of positive justification.

https://intranet.etc.upt.ro/~DIG_INT_NET/course/3_Hierarchical TDM.pdf, 10, 12, 14

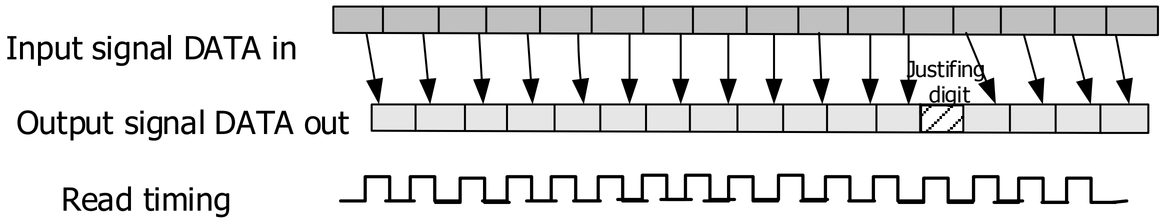
PDH multiplexing solutions

…

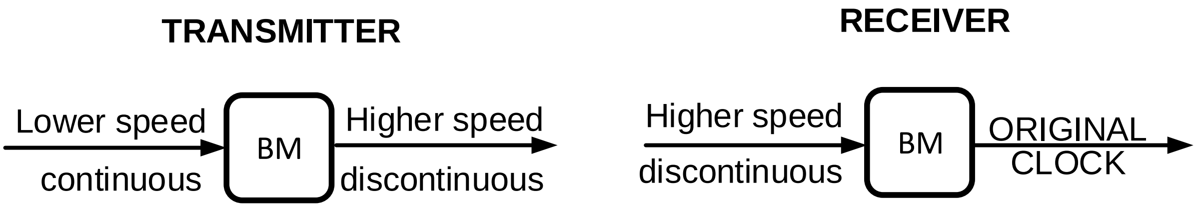
- Better solution – equalize the clock for all tributaries so that the bit rates are the same
  - Use faster clock rate than any of the tributaries
  - From time to time on each tributary a bit is doubled
  - Requires signaling for doubled bits (so that the receiver ignores them)

$$f_{\text{TRIBUTARY CLOCK}} < f_{\text{SEC. MUX CLOCK}}$$

Principle of positive justification – Transmitter



Principle of positive justification – Receiver

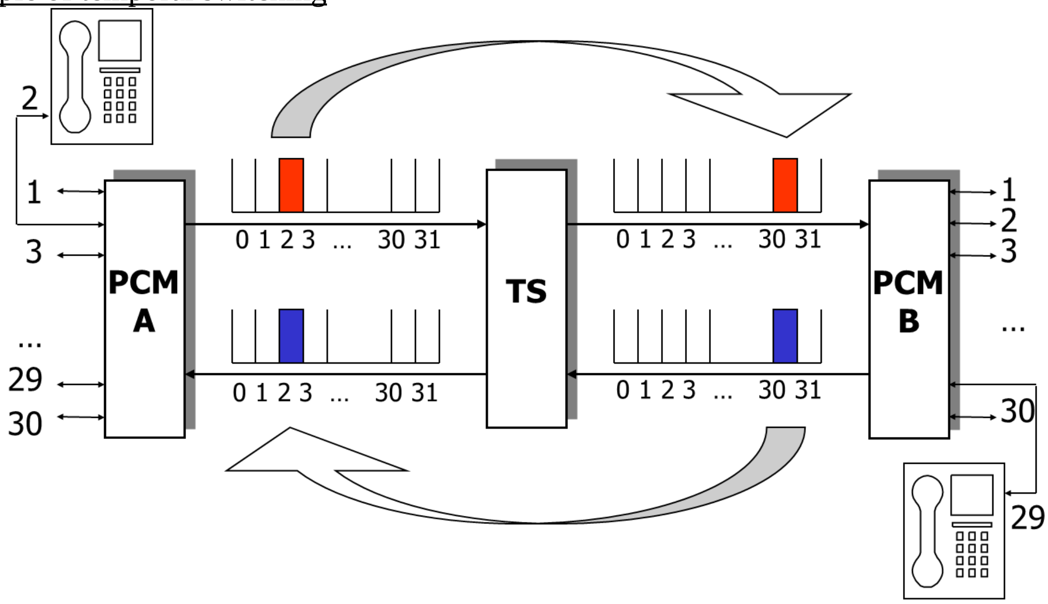


BM – buffer memory

2. **Digital switching – definition, the principle of the temporal switch, the principle of the spatial switch**

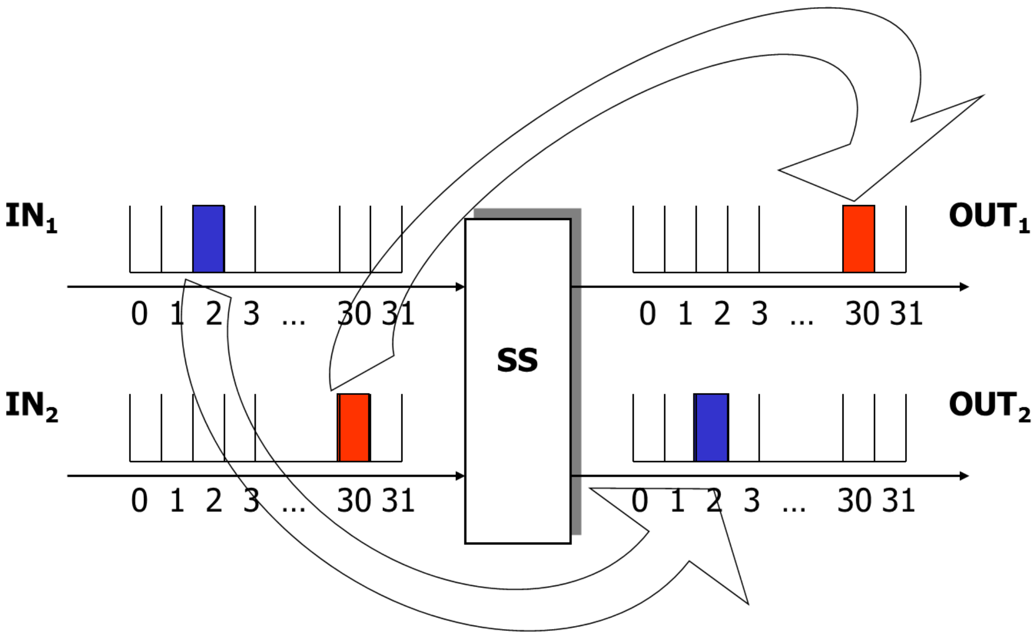
https://intranet.etc.upt.ro/~DIG_INT_NET/course/4_Digital_switching.pdf, 1, 3, 8-10

Principle of temporal switching



any TS of **any** data flow to **any** TS of the **same** data flow

Principle of spatial switching



any TS of **any** data flow to the **same** TS of **any** data flow

Digital switch

DIGITAL SWITCH

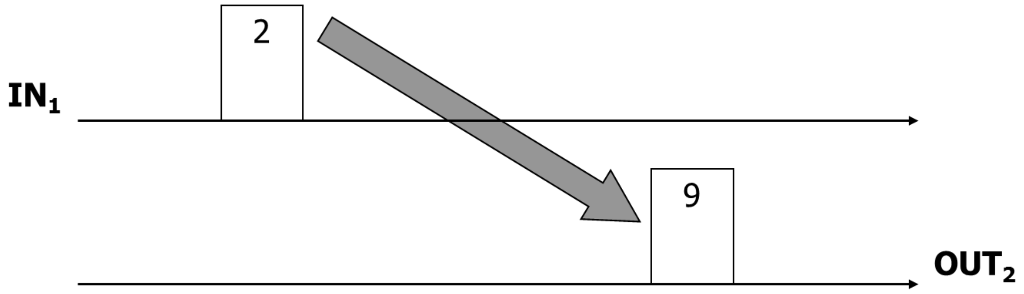
- any TS of any INPUT (data flow) to any TS of any OUTPUT (data flow)

TEMPORAL SWITCH

- any TS of any data flow to any TS of the same data flow

SPATIAL SWITCH

- any TS of any data flow to the same TS of any data flow



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3. Handover types in 3G networks.

https://intranet.etc.upt.ro/~DIG_INT_NET/course/6_Mobile networks_16.pdf, 46, 50, 51

<u>Handover</u>

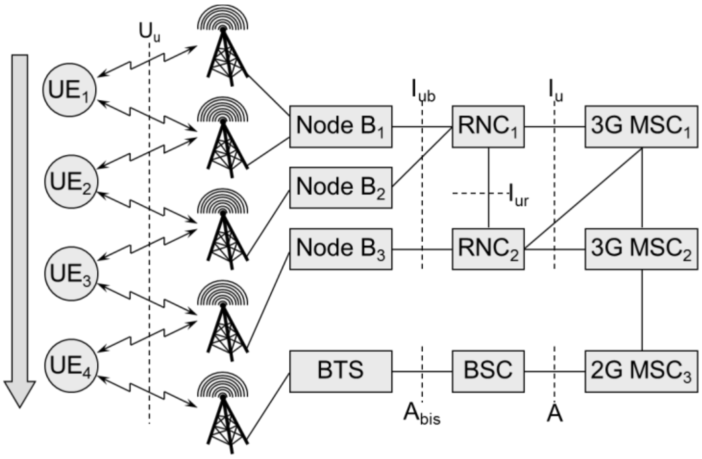
- Transfer of a link between 2 neighboring cells/antennas
- UMTS main handover classes:
  - Hard handover
    - Similar to GSM handover
    - Includes
      - Inter-frequency handover (change carrier frequency)
      - Inter-system handover (between UMTS and other systems)
  - Soft handover (new in UMTS)
    - Only available with FDD
    - Uses macrodiversity
      - fundamental characteristic of CDMA systems
      - mobile equipment communicates with up to 3 antennas simultaneously

<u>Handover types in 3G</u>

- Intra-nod B, intra-RNC (softer handover)

- UE_1 moves between 2 different antennas of the same Node B (Node B_1)
- Node B_1 combines and splits the data streams

- Inter-nod B, intra-RNC (soft handover)
  - UE_2 moves from Node B_1 to Node B_2
  - RNC_1 supports the soft handover combining and splitting the data streams



- Inter-RNC
  - UE_3 moves from Node B_2 toward Node B_3=> 2 possible situations
    - Internal (soft) inter-RNC handover, with RNC_1 acting as SRNC and RNC_2 acting as DRNC
    - External (hard) inter-RNC handover with relocation of the I_u interface

- Inter-MSC
  - MSC_2 takes over the connection and realizes a hard handover

- Inter-system (hard handover)
  - UE_4 moves from the 3G network to a 2G network
  - Important for areas with no 3G coverage

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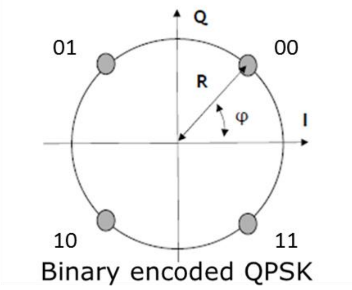
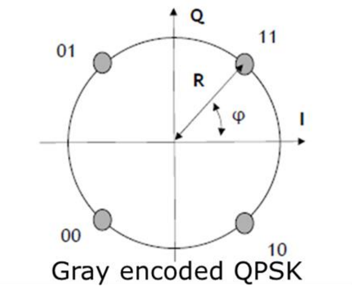
4. Digital modulation techniques used in LTE – list, characteristics and comparison.

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

<u>Digital modulations in LTE</u>

- 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





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

5. OFDMA – principle of sub-carrier orthogonality, application in LTE.

https://intranet.etc.upt.ro/~DIG_INT_NET/course/7_LTE.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)

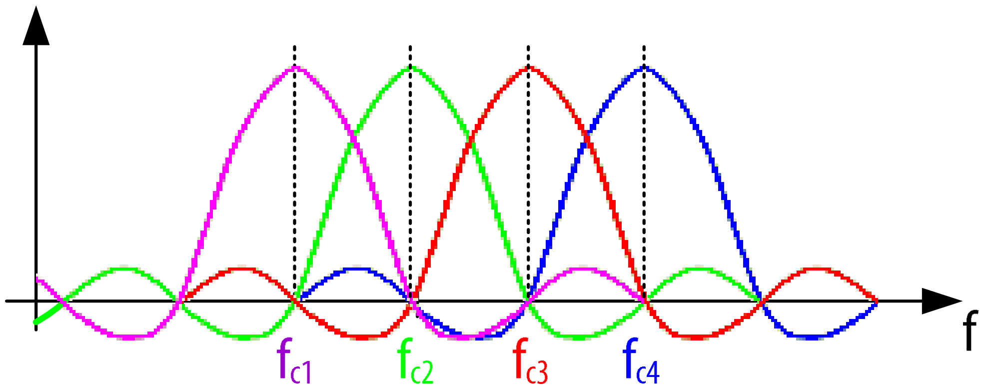
    - lower PAPR (Peak to Average Power Ratio) compared to OFDMA
    - used on UL
    - Unappropriate for DL
        - eNB uses all available sub-carriers
        - eNB transmits to multiple UEs at the same time

<u>OFDMA</u>

- 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 => => orthogonality (no interference)



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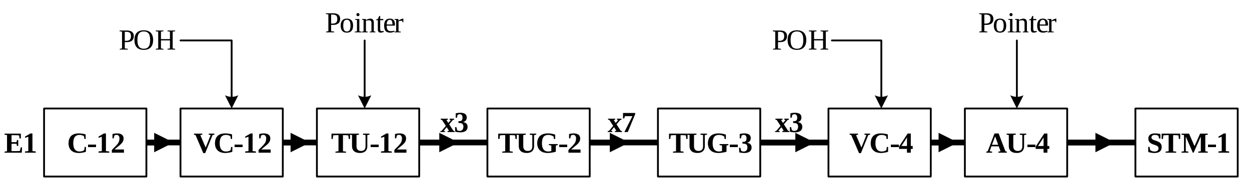
6. **An STM-1 multiplex carrying 48 E1 tributaries transits a node in the SDH network where 12 of the E1 tributaries are dropped, then other E1 tributaries are added in order to fill the STM-1 multiplex. Considering the multiplexing diagram below, the following are required:**

    a. **Calculate how many E1 tributaries are added to the STM-1 multiplex in the transit node.**

    b. **Calculate the total number of virtual containers inside the STM-1 frame before and after the transit node.**



https://intranet.etc.upt.ro/~DIG_INT_NET/course/3_Hierarchical TDM.pdf, 42, 47

Hints:

a. The total number of E1 tributaries that fill an STM-1 multiplex can be calculated using the given multiplexing diagram:

The payload area of a STM-1 multiplex is filled by an AU-4, which contains 1 VC-4.

The payload of a VC-4 is filled by 3 TUG-3.
Each TUG-3 is filled with 7 TUG-2.
A TUG-2 is filled with 3 TU-12.
Each TU-12 is filled from one E1 tributary (mapped through C-12 and VC-12).
Given that 12 of the initial E1 tributaries are dropped, only 48 – 12 = 36 of the initial E1 tributaries pass further. Since the STM-1 multiplex is filled, the number of added tributaries is the difference between the maximum number of E1 tributaries and 36.

b. There are two types of VC in the STM-1 multiplex for the given multiplexing diagram: multiple VC-12 (the number of VC-12 is equal to the number of E1 tributaries) and one VC-4 (which contains all the VC-12).

---

7. 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.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.

DATA COMMUNICATIONS

References: https://intranet.etc.upt.ro/~DATA_COM/2016-2017/

Theory

1. Multiplexing. Definition. Types of multiplexing methods.

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 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:

- Time-division multiplexing (TDM) ;
- Frequency-division multiplexing (FDM) ;
- Code division multiplexing ;
- Wavelength multiplexing.

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 and has the following characteristics.

- Uniformly distributed in frequency, generally modelled as additive white noise.
- Present in all electronic devices and transmission media.
- Its power is a function of temperature.
- Thermal noise is characterized by a near uniform distribution of energy over the frequency spectrum.

The amount of thermal noise in the band of 1 Hz is N_0=kT, where:

- N_0 is the noise power spectral density [Watts/Hz] –independent of frequency,

  k-is Boltzmann constant =1,38 .10^{-23} J/°K,
- T –is the absoulte temperature in Kelvin degrees.

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.

3. 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:

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

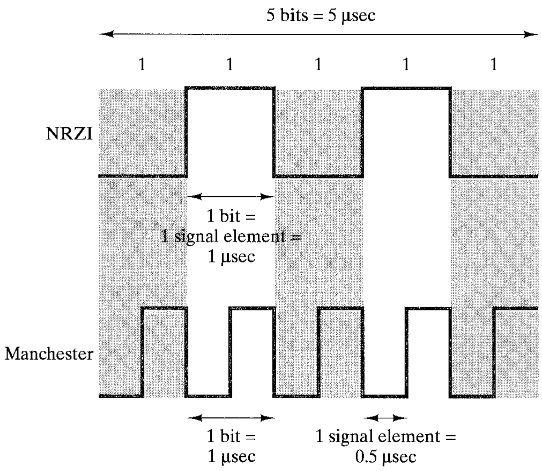
where:

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

Shannon’s formula expresses the theoretical maximum rate that can be achieved referred to as the error free capacity. In practice much lower rates are achieved. One reason is that only white noise is considered (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 this may determine an increase in intermodulation noise. Also notice that increasing B means that thermal noise will be increased too and this decreases the S/N ratio.

4. Compare a NRZI encoding with a Manchester encoding for the sequence 11111.

The results of the two encoding methods are presented in the following figure.



It can be observed that:

- In case of a NRZ code, during one bit representation, the signal stays on a constant voltage level (e.g. +1V or -1V).
- In case of Manchester code, during one bit representation, the signal can stays half of the bit period on a certain voltage level (e.g. +1V) and the other half on the opposite level (-1V). The modulation rate is twice the data rate D=2R, consequently a reduced efficiency comparative to NRZI is achieved. The increased number of transitions aids the synchronization process which is relaxed in case of Machester coding.

5. The ideal low pass filter and Nyquist criterion for zero ISI. Drawbacks of using the ideal low pass filter.

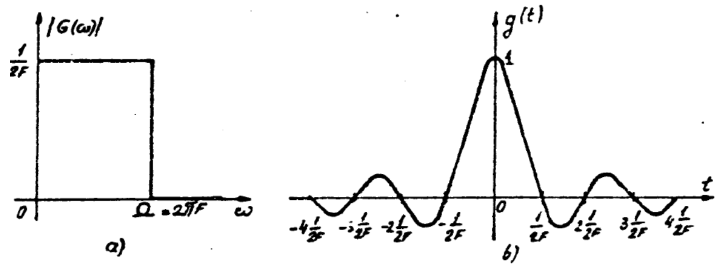
The Nyquist criterion for zero ISI is the following:

$$g_k = \begin{cases} 1, \text{if } k = n \\ 0, \quad \text{otherwise} \end{cases}$$

where g(t) represents the impulse response of the system obtained by connecting in series the output filter of the transmitter, the communications channel and the input filter of the receiver. g_k represent the samples of g(t) taken with a sampling step equal with T.

A system which satisfies the Nyquist criterion for zero ISI is the ideal low-pass filter.

The frequency response and the impulse response of the ideal low-pass filter are the followings:



In this case the function g(t) is a cardinal sine (sinc).

Drawbacks of using sinc pulse shaping filter:

- A sinc waveform is of infinite length, having tails of energy which extend to infinity. In practice, one can only design a finite duration waveform, which is an approximation of the real sinc pulse. Its time-domain "windowing" leads to a spectrum which is not an ideal low-pass function any more. Consequently, truncating the sinc waveform will increase its sensitivity towards the ISI phenomenon.
- The side-lobes of the cardinal sine are still important (even if they have a significantly lower energy compared to the main lobe). Actually, the pulse tails that fall in the adjacent symbol times, decay at the rate of 1/t. Consequently, if some timing error occurs (e.g. a receiver synchronization error), signals corresponding to adjacent symbols will significantly affect the value of the sample corresponding to the current symbol, thus creating ISI.

Problems

P.1 Write the frequency response of a $\cos^2$ shapping filter when T=6 s. Plot this frequency characteristic.

Solution:

The cut-off pulsation of the ideal low-pass filter is $\omega_0 = \frac{\pi}{T}$, and the expression of the frequency response of the $\cos^2$ shapping filter is:

$$G(\omega) = \begin{cases} \frac{\pi}{\omega_0} \cdot \cos^2 \frac{\pi\omega}{4\omega_0}; & |\omega| \le 2\omega_0 \\ 0; & \textit{in rest.} \end{cases}$$

where $\omega_0 = \frac{\pi}{6}$ in our case.

So, the required frequency response has the following expression:

$$G(\omega) = \begin{cases} 6 \cdot \cos^2 \frac{3\omega}{2}; & |\omega| \le \frac{\pi}{3} \\ 0; & \textit{in rest.} \end{cases}$$

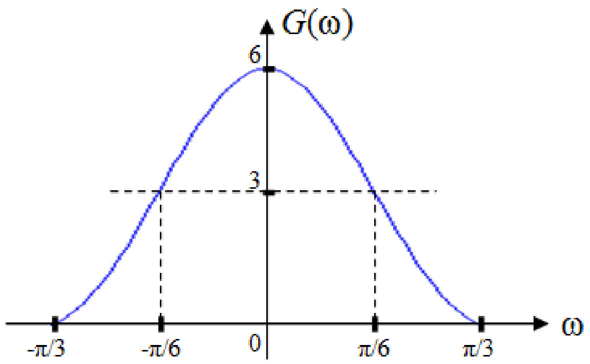
To plot this frequency response, we compute it in three different points:

- For $\omega = 0 \Rightarrow \cos^2 0 = 1 \Rightarrow G(0) = 6$;

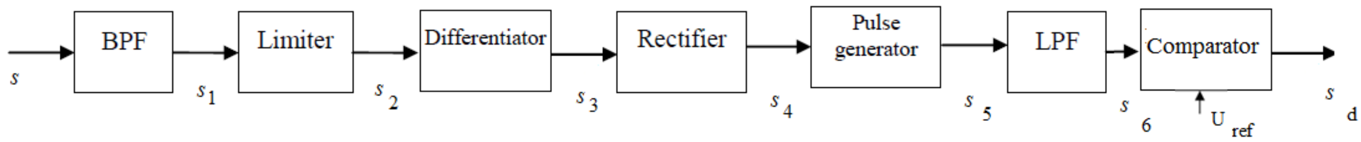
- For $\omega = \frac{\pi}{6} \Rightarrow \cos^2\left(\frac{3}{2}\cdot\frac{\pi}{6}\right) = \cos^2\frac{\pi}{4} = \left(\frac{\sqrt{2}}{2}\right)^2 = \frac{2}{4} = \frac{1}{2} \Rightarrow G\left(\frac{\pi}{6}\right) = 6\cdot\frac{1}{2} = 3;$

- For $\omega = \frac{\pi}{3} \Rightarrow \cos^2\left(\frac{3}{2}\cdot\frac{\pi}{3}\right) = \cos^2\frac{\pi}{2} = 0 \Rightarrow G\left(\frac{\pi}{3}\right) = 0.$

  So, we obtain the following plot.



P.2 Let's consider the frequency demodulator from the figure below:



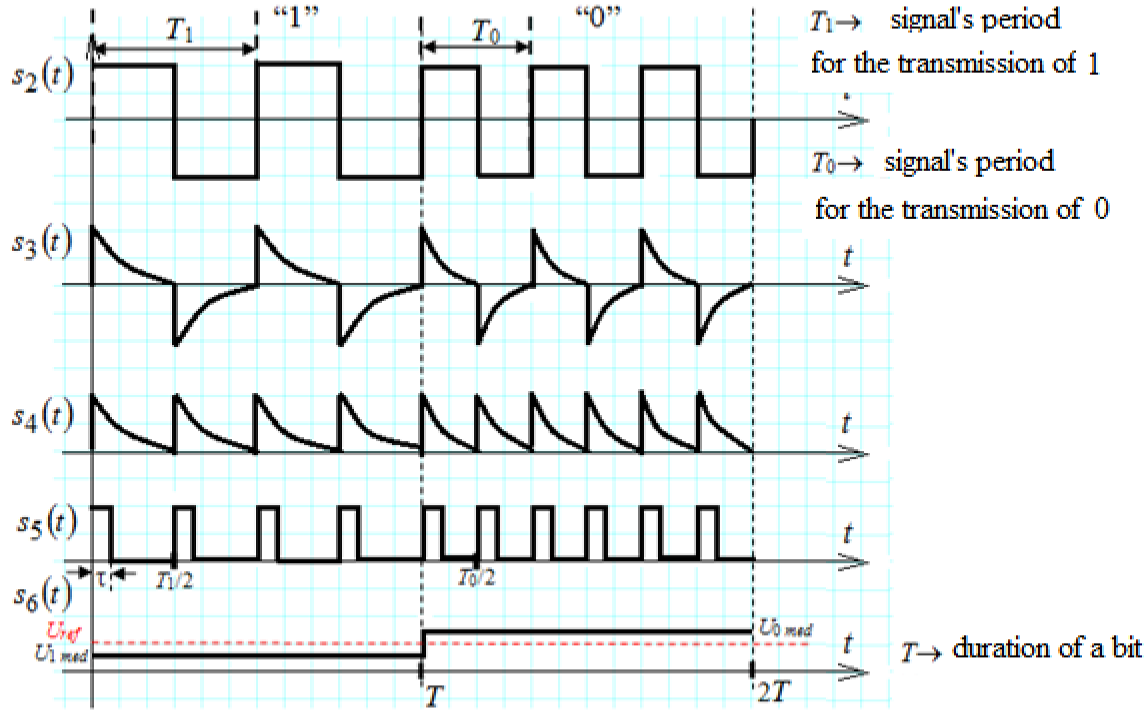
The modulation rate is of 300 Baud. The frequency used to transmit a bit of "1" is f_1=1650 Hz. The frequency used to transmit a bit of "0" is f_0=1850 Hz.

a. Plot the waveforms of the signals $s_2(t)\div s_6(t)$.

b. Compute the maximum duration τ_{max} of the rectangular pulses generated by the pulse generator from the demodulation chain.

Solution:

a. The limiter forms rectangular pulses, $s_2(t)$, which are derivated $s_3(t)$ and next rectified $s_4(t)$. The pulses generated have the duration τ, $s_5(t)$. The Low Pass Filter extracts the continuous component (the mean) of those pulses, $s_6(t)$. The first pulses, having a smaller frequency have a smaller mean ($U_{1\ med}<U_{ref}<U_{2\ med}$).



b.

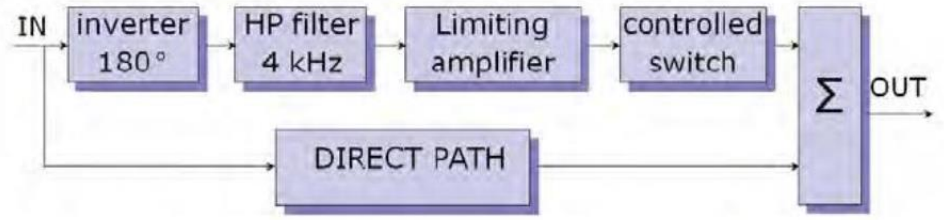
$\tau_{\max} = \dfrac{T_{\min}}{2}$, because $T_{min}/2$ represents the rythm of pulses generation.

$$\tau_{\max} = \frac{T_{\min}}{2} = \frac{1}{2 \cdot f_{\max}} = \frac{T_0}{2} = \frac{1}{2 \cdot f_0} = \frac{1}{2 \cdot 1850} = 270\,\mu s\,.$$

AUDIO AND VIDEO SYSTEMS

T1. Define and compare the two principles for noise reduction: DNL and Dolby.

Noise = low level signal and medium to high frequency -> can be identified and rejected (nose gate)
Example: Dynamic Noise Limiter (DNL) from Philips



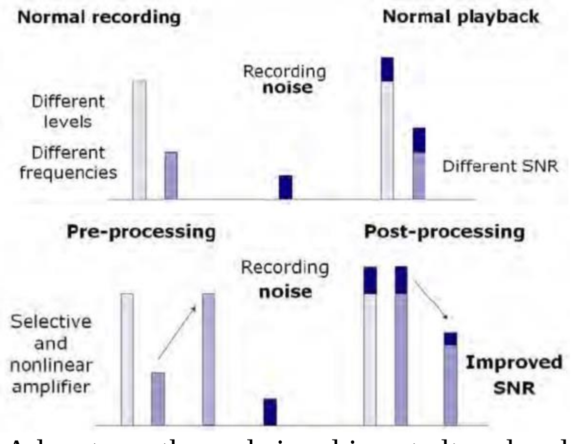
IN: noisy signal
OUT: signal with a SNR improved by 8 dB
Advantage: compatible with any recording system and any playback system

How it works:

1. during the pause between melodies
2. high level recorded music
3. low level recorded music

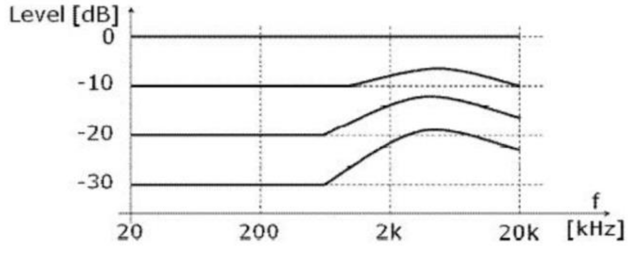
Disadvantage: it cannot make the difference between noise and real signal

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

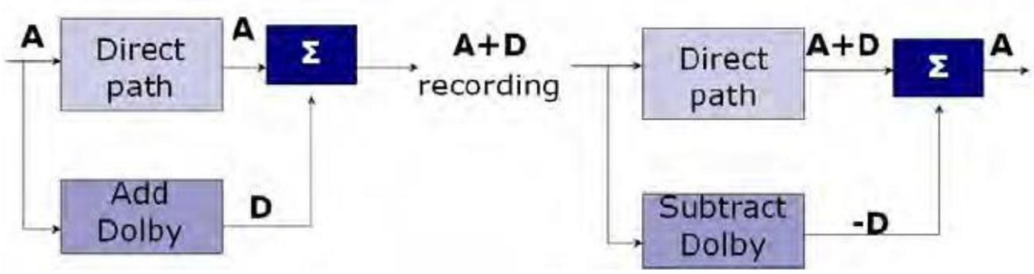


Noise:

- 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



T2. Define and compare the uniform and non-uniform quantization techniques.

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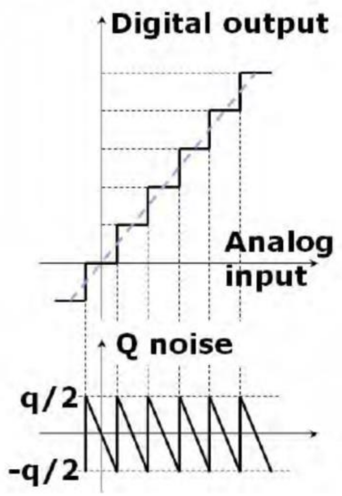
<td>
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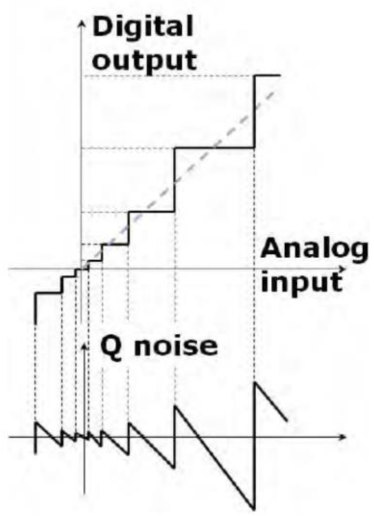
<td>
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<tr>
<td>
<u>Uniform quantization:</u>
■ decision levels (analog input)
□ uniform
■ quantization levels (digital output)
□ uniform
■ quantization steps (q) are constant:
□ for low level signal
□ for high level signal
■ quantization noise (error): -q/2 ÷ q/2

RESULT
■ low level signal with constant quantization error ⇨ low SNR

■ high level signal with constant quantization error ⇨ high SNR

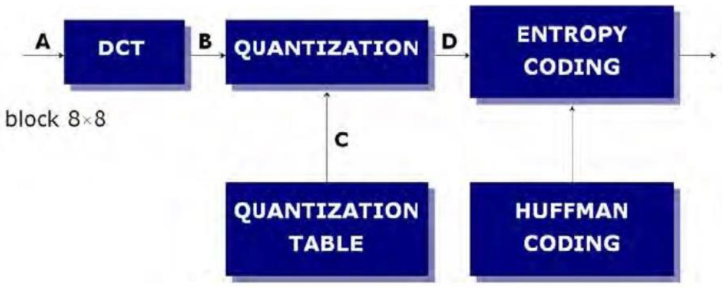
CONCLUSION
■ low general SNR (⇔ low quality)
</td>
<td>
<u>Nonuniform quantization:</u>
■ decision levels (analog input)
□ non-uniform
■ quantization levels (digital output)
□ non-uniform
■ quantization steps (q) are different:
□ for low level signal
□ for high level signal
■ quantization noise (error):
□ non-constant

RESULT
■ low level signal with low quantization error ⇨ high SNR

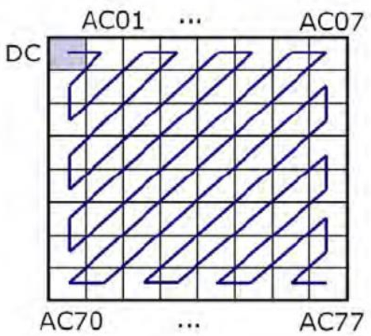
■ high level signal with high quantization error ⇨ high SNR

CONCLUSION
■ high general SNR (⇔ high quality)
</td>
</tr>
</table>

T3. Present the principles of JPEG compression (general block diagram and presentation of each component block).



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)



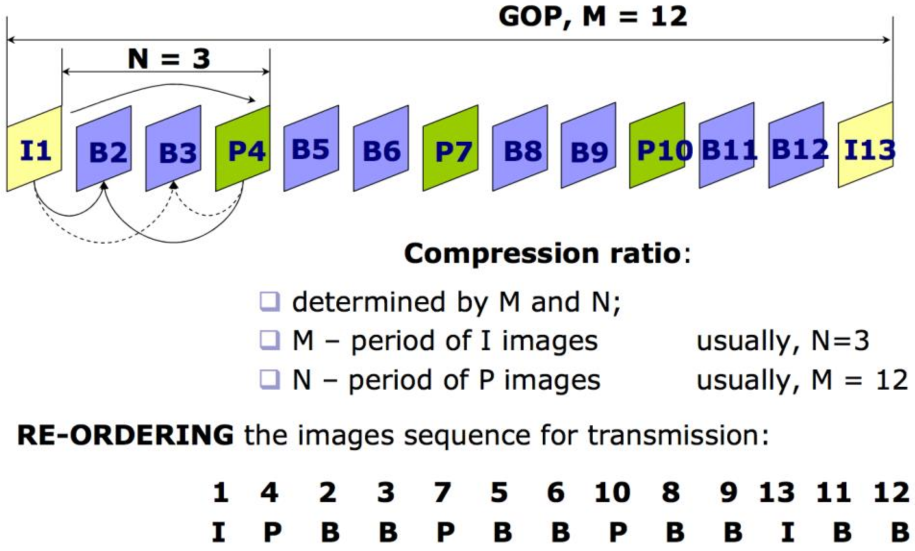
T4. Define the image types involved in MPEG compression and explain the compression principle.

- 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



T5. Present the color TV signals.

- Luminance of a (color) image is used in black-and-white television:

  $Y = 0.3 \times R + 0.59 \times G + 0.11 \times B$
- Using R, G, B signals would be incompatible with the old TV system.
- Compatible color TV systems use:
  - Y – luminance (for correct processing by black-and-white TV sets)
  - 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:

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

Luminance

$EY = 0.3 \times ER + 0.59 \times EG + 0.11 \times E\,B = 0 \div 1$

Color difference

$ER\text{-}Y = 0.7 \times ER - 0.59 \times EG - 0.11 \times E\,B = -0.7 \div 0.7$

$EG\text{-}Y = -0.3 \times ER + 0.41 \times EG - 0.11 \times E\,B = -0.41 \div 0.41$ (not transmitted)

$EB\text{-}Y = -0.3 \times ER - 0.59 \times EG + 0.89 \times E\,B = -0.89 \div 0.89$

P1. 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

<u>Hi-fi:</u>

t = 4*60+30 = 270 s

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

<u>MP3 compression, D=128 kbps:</u>

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