MARITIME UNIVERSITY OF SZCZECIN

ORGANIZATIONAL UNIT:
DEPARTMENT OF MARINE COMMUNICATION TECHNOLOGIES

INSTRUCTION

ELECTRICAL ENGEENERING AND ELECTRONICS
Laboratory
Exercise No 6: Modulation and detection

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6. MODULATION AND DETECTION

6.1. The purpose and scope of the exercise

The aim of the exercise is to learn about analog modulation, demodulation and frequency transformation.

Issues

1. Parameters and spectrum of basic electrical signals.
2. Types of analog modulations.
3. Amplitude modulation.
4. Phase modulation.
5. Frequency modulation.
6. Amplitude demodulation.
7. Frequency demodulation.
8. Frequency transformation.

Control questions

1. What is modulation and when is it used?
2. What are the types of modulation?
3. Discuss the amplitude modulation.
4. Discuss the frequency spectrum of the amplitude modulated signal.
5. What is the modulation depth *m*?
6. Discuss frequency and phase modulation.
7. What are the differences between frequency and phase modulation?
8. What are the basic differences between different types of modulation?
9. Discuss the idea of operation of basic modulation systems.
10. What is detection?
11. Discuss the principle of operation of a linear diode detector.
12. Discuss the principle of operation of the simplest frequency demodulation system.

6.2. Description of the program

The simulation program of amplitude and frequency modulation is started by selecting the "Modulation" icon on the desktop.

After starting the program, the program window will be set on the computer screen by default in the tab for simulation of amplitude modulation.

Change of modulation to phase and frequency modulation (PM and FM) is done by selecting the second tab in the upper part of the program window.

In both tabs, on the right side there are fields and sliders to change / specify the parameters of individual signals.

On the right below, there are fields for changing the scale of the charts.

After changing the parameters, the waveforms of all signals change automatically.

During the simulation, the amplitude modulation is carried out according to the formula:

$$U(t) = \left[U_n + m \cdot x(t)\right] \cdot \sin(\omega_n t)$$

where:

$U(t)$ - modulated signal,
U_n - amplitude of the carrier signal,
$x(t)$ - modulating signal,

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ω_n - pulsation of the carrier signal.

Phase modulation is carried out according to the formula:

$$U(t) = U_n \cdot \sin\left[\omega_n t + d_{ff} x(t)\right]$$

where:

d_{ff} - współczynnik dewiacji fazy.

Modulacja częstotliwości realizowana jest według wzoru:

$$U(t) = U_n \cdot \sin\left[\omega_n t + d_{fc} \cdot \int_0^t x(t)dt\right]$$

where:

d_{fc} - phase deviation factor.

The shape of the modulating signal may be as follows:

- one sine wave (in the simulation program – 1sin):
  $x(t)=U_s \cdot \sin(\omega_s t+a)$
- the sum of two sine waves (in the simulation program – 2sin):
  $x(t)=U_s \cdot \sin(\omega_s t+a)+ U_{s2} \cdot \sin(\omega_{s2} t+a_2)$
- the sum of three sine waves (in the simulation program – 3sin):
  $x(t)=U_s \cdot \sin(\omega_s t+a)+ U_{s2} \cdot \sin(\omega_{s2} t+a_2)+U_{s3} \cdot \sin(\omega_{s3} t+a_3)$
- rectangular signal:

$$x(t) = \begin{cases} U_s & \text{for } t \le \frac{T}{2} \\ -U_s & \text{for } t > \frac{T}{2} \end{cases}$$

- triangular signal:

$$x(t) = \begin{cases} 4 \cdot U_s \cdot f_s \cdot \left(t - \frac{1}{f_s}\right) - U_s & \text{for } t \le \frac{T}{2} \\ -4 \cdot U_s \cdot f_s \cdot \left(t - \frac{1}{f_s}\right) - 3 \cdot U_s & \text{for } t > \frac{T}{2} \end{cases}$$

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6.3. THE COURSE OF THE EXERCISE

6.3.1. AM modulation study

Set (check if they are already entered) the following data:

- time of the tested wave t_{max} = 1 ms
- the lower spectrum frequency f_d = 0 Hz
- the upper spectrum frequency f_g = 40 kHz

Warning! The above data should be left unchanged throughout the exercise.

Set (check if they are already entered):

- U_n = 20 V
- f_n = 20 kHz
- typ of modulation - AM
- m = 100 %
- U_s = 20 V
- f_s = 1 kHz

Change the shape of the modulating signal respectively:

- **1 sin** – one sine wave,
- **2 sin** – signal consisting of a sum of two sine waves,
- **3 sin** – signal consisting of a sum of three sine waves,
- **rectangular signal**,
- **triangular signal**.

For each set shape of the modulating signal it is necessary:

- redraw signal waveforms,
- redraw the spectrum of the modulated signal
- write the modulation depth *m* on the measurement card,
- write the carrier and modulation signal parameters on the measurement card.

6.3.2. Study of the influence of carrier and modulation signals on the spectrum of the modulated signal

Set the shape of the modulating signal to 1 sin.

a. Set the frequency of the modulating signal:

f_s = 1 kHz.

Change the frequency of the carrier signal in sequence:

- **f_n = 10 kHz,**
- **f_n = 20 kHz,**
- **f_n = 30 kHz.**

Write how the modulated signal and the modulated signal spectrum changed.
Redraw the spectrum only.

b. Set the frequency of the carrier signal

f_n = 20 kHz.

Change the frequency of the modulating signal in sequence:

- **f_s = 1 kHz,**
- **f_s = 2 kHz,**
- **f_s = 3 kHz.**

Write how the modulated signal and the modulated signal spectrum changed.

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<u>Redraw the spectrum only</u>.

6.3.3. Study of the influence of signal amplitudes on the AM modulation process

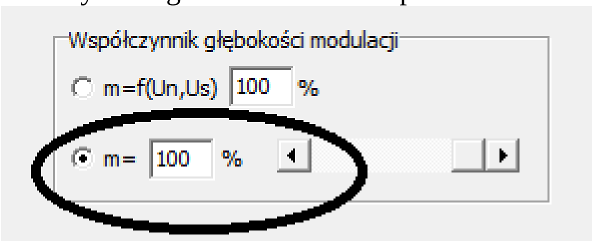
Set the frequency of the carrier signal and the frequency of the modulating signal:

f_n = 20 kHz f_s= 2 kHz.

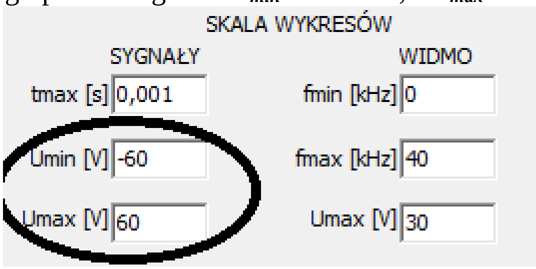
Set the amplitude of the carrier signal to:

U_n = 30 V.

Select the option of manually setting the modulation depth factor m:



Change the scale of the graphs for signals: **U_{min} = -60 V ; U_{max} = 60 V**



Write how the modulated signal has changed, how the spectrum has changed, and the amplitudes of the modulating signal U_s has changed.

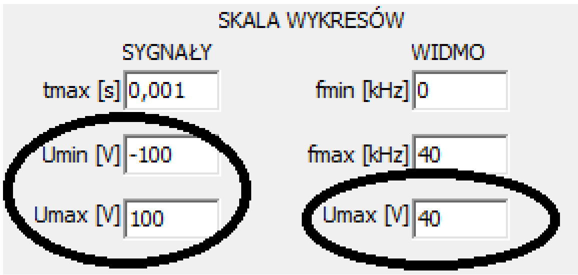
<u>Redraw the modulated signal and spectrum.</u>

Set the amplitude of the carrier signal again to:

U_n = 20 V.

Change the scale of the graphs:

– for signals: **U_{min} = -100 V ; U_{max} = 100 V**
– for spectrum: **U_{max} = 40 V**



Change the modulation depth factor to:

m = 150 %.

Write how the modulated signal has changed, how the spectrum has changed, and the amplitudes of the modulating signal U_s has changed.

<u>Redraw the modulated signal and spectrum.</u>

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Change the modulation depth factor to:

***m* = 200 %**.

Write how the modulated signal has changed, how the spectrum has changed, and the amplitudes of the modulating signal U_s has changed.

Redraw the modulated signal and spectrum.

Based on the observed processes, find the relationship between the amplitude of the U_n, carrier signal, the amplitude of the modulating signal U_s and the modulation depth factor *m*. Write the formula on the measurement card.

6.3.4. PM and FM modulation study

Go to the phase and frequency modulation simulation tab. Set the following data:

- U_n **= 20 V**
- f_n **= 20 kHz**
- d_{ff} **= 1**
- d_{fc} **= 6280**
- U_s **= 10 V**
- f_s **= 1 kHz**

Change the shape of the modulating signal to successively:

- **1 sin** – one sine wave,
- **rectangular**,
- **triangular**.

For each set shape of the modulating signal it is necessary:

- redraw signal waveforms,
- write the phase deviation and frequency deviation,
- save the parameters of the carrier and modulation signal.

Based on the observed processes, find the relationship between phase deviation and frequency deviation.

6.3.5. Study of AM and FM real modulation

Observe the waveforms on the oscilloscope demonstrated by the lecturer. Redraw the waveforms and spectra indicated by him. Redraw the spectrum for emissions: A3E, H3E, J3E.

6.4. Assessment conditions

The condition for assessment of the exercise is:

- to write a short test at the beginning of the class with a positive result,
- to do the exercise,
- preparing a report according to the instructions below,
- positive assessment of the report on the next class.

The report should include:

- formulas describing amplitude modulations as well as phase and frequency modulations,
- rewrited modulation waveforms,

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- explanation of the influence of a carrier signal on the modulated signal and spectrum in the AM modulation,
- explanation of the influence of a modulating signal on the modulated signal and spectrum in the AM modulation,
- explanation of how the phase modulated signal is generated and how the frequency modulated signal is generated,
- dependences described in points: 6.3.3 i 6.3.4.
- waveforms from point 6.3.5. indicated by the lecturer and emission spectra: A3E, H3E, J3E.
- own conclusions and observations.

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6.5 Theoretical part

6.5.1. Analysis of electrical signals

An electrical signal is the time course of an electrical quantity, i.e. voltage or current. The signals can be fixed or variable. The group of constant signals includes signals in which the value of current or voltage remains unchanged over time (Fig. 6.5.1.). From the electronics point of view, we will also include signals of a changing value in this group of signals, provided that they are slow changes and the current or voltage value will not change the polarity (Fig. 6.5.2.). A typical example of such a signal is the voltage at the terminals of the car battery while driving.

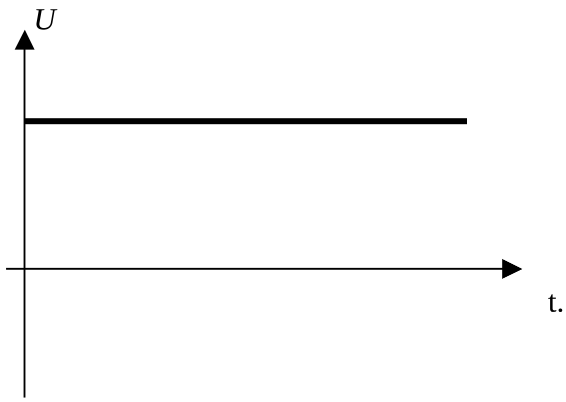

Fig. 6.5.1

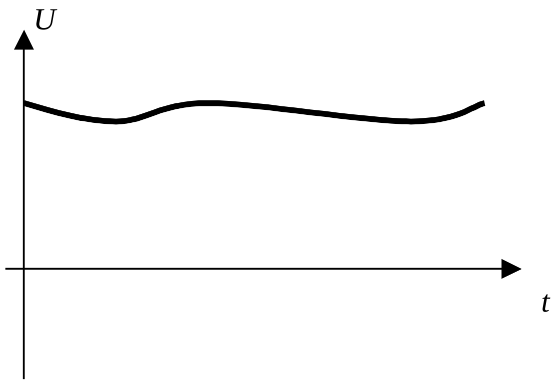

Fig. 6.5.2

The second group of signals are variable signals. These are the signals in which the direction of the current flowing or the voltage polarity changes (figure 6.5.3). Among the variable signals, periodic signals play a special role. The periodic signal is the electric signal in which we can distinguish a fixed time segment called the period T, after which the signal value is repeated (Fig. 6.5.4).

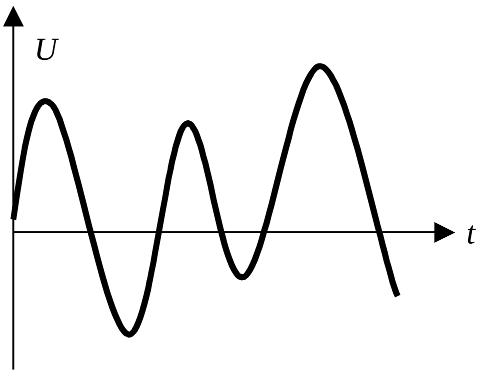

Fig. 6.5.3

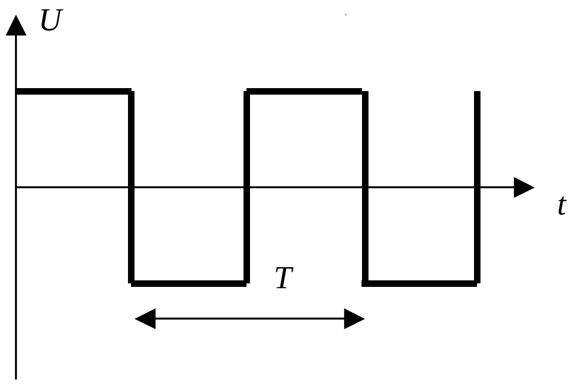

Fig. 6.5.4

Variable signals may exist without a constant component, as in Fig.5.5.4 or with a constant component (Fig.6.5.5). The constant component may be positive or negative and is simply the average signal value.

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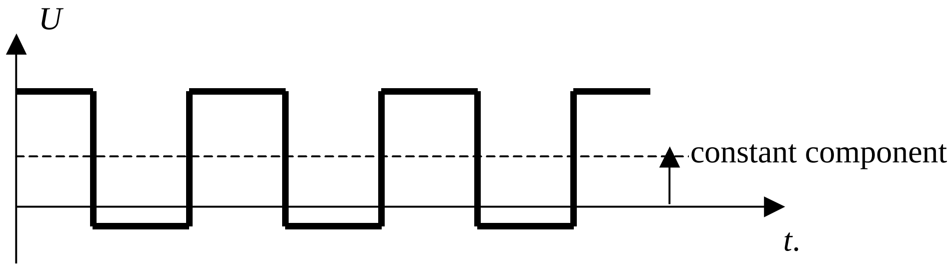

Fig. 6.5.5

Electrical signals, depending on what signal characteristics we want to emphasize, can be presented in various ways. In Figures 6.5.1 to 6.5.5, signals were presented graphically. This is the most natural presentation of signals, in this form we watch them, eg on an oscilloscope. With this presentation system on the ordinate, instantaneous current or voltage values are expressed in amperes or volts (or in mA, mV, etc.), and on the abscissa the time expressed in seconds (or in ms, μs, etc.).

In a situation where we particularly want to show the phase relationships between electrical signals, the vector presentation of signals turns out to be more useful. In this system of presentation, the lengths of vectors present the values of voltages or currents on the adopted scale, and the angles between them determine the phase relationships between particular voltages and currents. For example, let us analyze a simple electrical circuit consisting of the resistance *R* and the capacitance *C* supplied with a sinusoidal voltage *U* with frequency *f* shown in Fig. 6.5.6.

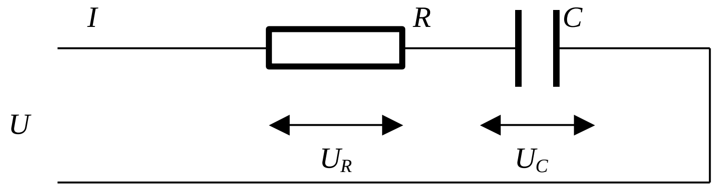

Fig. 6.5.6

Of course, both the current flowing in the circuit as well as the supply voltage and voltage drops on the resistance and capacity can be represented graphically as four sine waves offset from each other. However, this drawing would be difficult to read. The vector presentation of currents and voltages will be definitely more useful here (Figure 6.5.7).

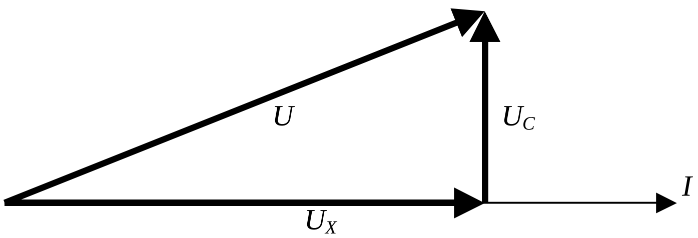

Fig. 6.5.7

Another method of signal presentation is to present them in the form of a frequency spectrum. Consider for this purpose a simple sinusoidal signal presented graphically in Fig. 6.5.8.

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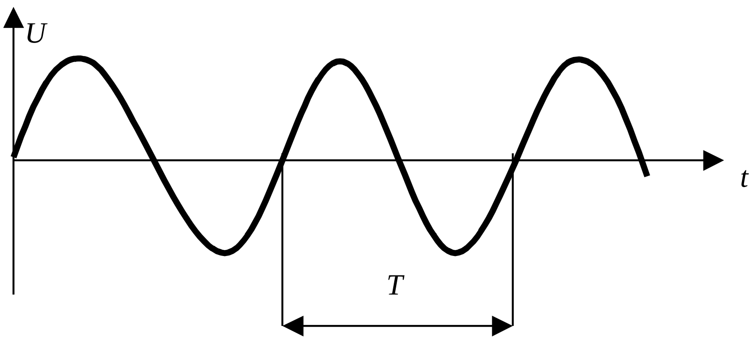

Fig. 6.5.8

We can write this signal analytically in the form:

$$u = U_o \sin \omega t$$

where:

– U_0 – signal amplitude

– $\omega = 2\pi f = \frac{2\pi}{T}$ – pulsation

In order to present this signal in a spectral form on an axis calibrated in frequency units, we draw a band with a height equal to the amplitude of the signal at the adopted scale. The position of the band on the axis determines its frequency (Fig.6.5.9).

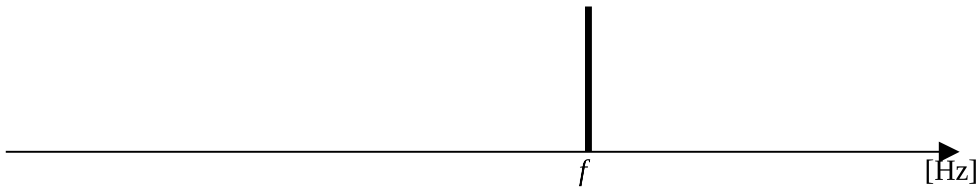

Fig. 6.5.9

It should be noted that Figures 6.5.8 and 6.5.9 represent the same signal, differ only in the way it is presented.

Spectral presentation is especially often used for complex signals appearing in issues related to radio communication. Consider the concept of complex signal. The aforementioned sinusoidal signal, also called harmonics, is referred to in electronics as a simple signal. Each other signal, in any shape, is a composite signal consisting of a finite or infinite sum of straight (sinusoidal) signals at different frequencies. This sum presented on the frequency axis in the form of a fringe arrangement is called the frequency spectrum of the complex signal. The distribution of the spectrum depends on the nature of the composite signal. We will consider frequency spectra for three groups of signals, namely for periodic, acoustic and pulse signals.

The periodic signal has a regular fringe spectrum consisting of a finite or infinite sum of straight (sinusoidal) signals with frequencies that are multiples of the basic frequency of the complex signal. This sum will contain a zero frequency band if a fixed component was present in the composite signal. Depending on the shape of the signal in the spectrum, only even, only odd or even odd harmonics may be present. As an example, Fig. 6.5.10 presents the spectrum of a rectangular signal without a constant component with the frequency f_0.

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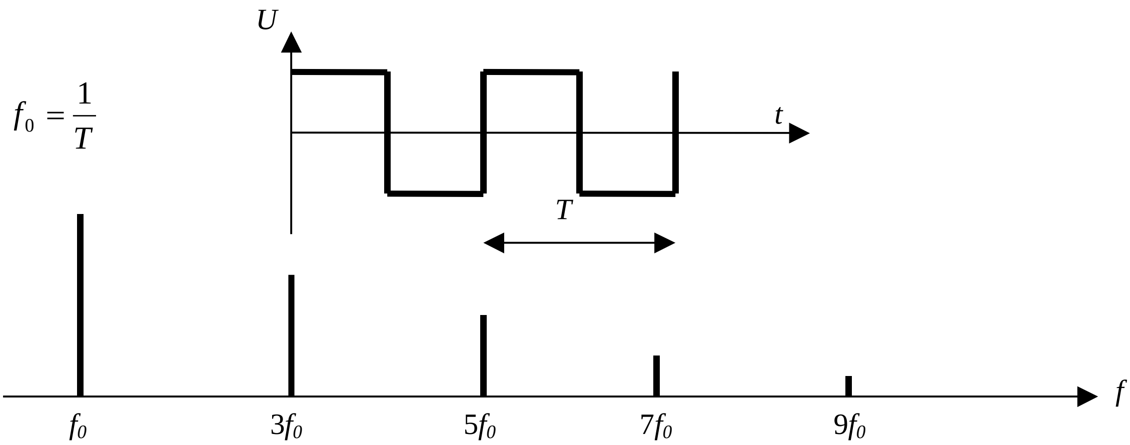

Fig. 6.5.10

As shown in the figure, in the case of a rectangular signal, the spectrum consists of odd harmonics. The lack of a constant component in the signal results in the lack of a zero frequency band. Although the number of bands in the spectrum is infinitely great, however, due to the fast-wavelengths of higher harmonics, it is enough to take into account the first ten harmonics. So the band occupied by this signal extends practically from f_0 to 9 f_0.

The spectrum of a triangular signal is similar in nature, and odd harmonics are also present in it. However, the signals on the outputs of one or two-wave rectifiers have a spectrum consisting of even harmonics. Of course, there are also periodic signals containing in the spectrum both even and odd harmonics.

The acoustic signal consists of many superimposed sounds. Each of these sounds has a specific frequency (pitch) and contains some harmonics that are multiples of the basic frequency. The spectrum of this signal will therefore consist of many irregularly distributed bands representing amplitudes of component sounds and their harmonics. In figure 6.5.11. presents an example spectrum of an acoustic signal consisting of two sounds, including their harmonics.

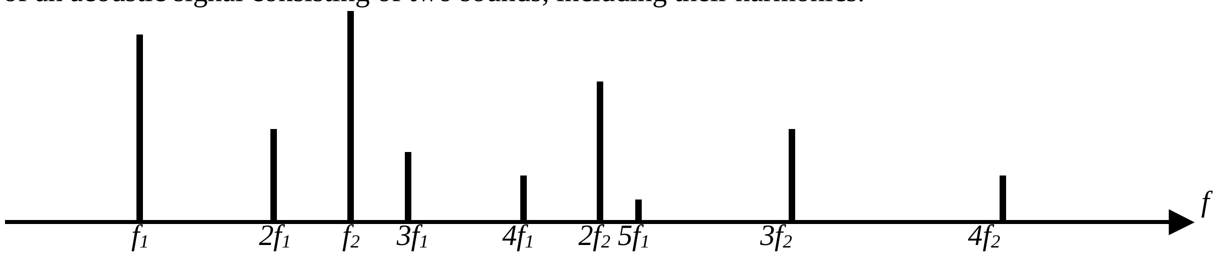

Fig. 6.5.11

The acoustic signal contains theoretical frequencies in the 20 to 20,000 Hz band. The maximum frequency of this signal is particularly important. In practice, it depends on the technical possibilities of recording and reproduction of sounds and certain legal regulations related to the transmission system:

- for VHF frequency-modulated signals f_{max} = 15 000 Hz
- for signals transmitted on MF and HF with amplitude modulation f_{max} = 4 500 Hz
- for signals broadcast under maritime communications f_{max} = 2 800 Hz

Of course, the wider the bandwidth of transmitted frequencies, the better the sound quality.

The pulse signal has a continuous spectrum extending theoretically from zero to infinity. In practice, the components of the spectrum at very high frequencies have so small amplitudes that they can be disregarded. As an example of this type of spectrum, Fig. 6.5.12 presents a single rectangular pulse of duration τ and its spectrum.

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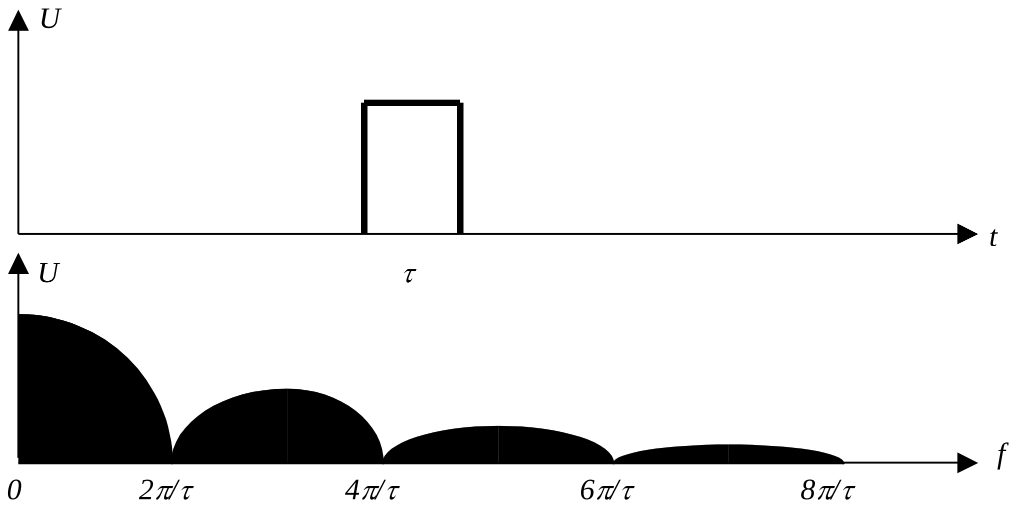

Fig. 6.5.12

The above figure shows four groups containing the spectrum frequencies of the rectangular pulse spectrum. In fact, there are infinitely many of these groups, but further parts of the spectrum have so small amplitudes that they cannot be taken into account. Since the width of the group is inversely proportional to the duration of the pulse τ, the entire range of the spectrum taken into account also depends on the duration of the pulse. Shortening the pulse, extends its frequency spectrum.

6.5.2 Modulation

The general principle of radiocommunication is to send to the transmitting antenna connected to the transmitter a current of such shape that the information to be transmitted is stored in it.

There is a field around the antenna, called an electromagnetic wave, in a shape corresponding to the current fed to the antenna. This wave propagates in space at a speed equal to the speed of light (c = 300,000 km / s) and reaching the receiving antenna causes the voltage in the shape corresponding to the shape of the electromagnetic wave to be induced in it. In this way, the signal from the transmitter reaches the receiver in the shape that was transmitted to him in the transmitter. For the time being we omit distortions that may occur during transmission. The theory of the construction of antennas shows that in order to obtain an effective transmit power, the geometrical dimensions of the antenna should be close to the length of the transmitted wave. The dependence of the wavelength on its frequency is expressed as follows:

$$\lambda = \frac{c}{f}$$

where: λ - wavelength,
c – speed of light,
f - frequency.

The above relationship shows that for acoustic frequencies (20 - 20,000 Hz), the antenna must have dimensions counted in hundreds or thousands of kilometers. Therefore, it is not possible to transmit these signals directly in the context of radiocommunication. In order to overcome this difficulty, a special carrier wave is generated in the transmitter with a relatively high frequency (above 150 kHz) and the signal that we want to transmit (acoustic, television, radar, etc.) is recorded in the modulation process.

Thus, we will call modulation the process of recording information on the carrier wave.

We can present a carrier wave that is a sinusoidal signal:

$$i = I_m \cdot \sin(\Omega t + \varphi)$$

where: i – instantaneous current value,
I_m – current amplitude,
Ω - pulsation $\Omega = 2\pi f$,

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φ - phase.

As it results from the given dependence in the modulation process, we can change either the amplitude value (AM modulation) or angle (angular modulation). Depending on the dependence of the angle on the modulating waveform, we obtain FM frequency modulation or PM phase modulation.

6.5.3. Amplitude modulation AM

Amplitude modulation is defined as a process in which the amplitude of the carrier wave is linearly dependent on value of the modulating signal, keeping frequency and phase constant. The carrier wave is a sinusoidal signal generated in the transmitter by a special, very stable generator. Its frequency is the nominal frequency of a given broadcast station. The modulating signal is the information signal that we want to transmit. It can be electricity from a microphone, TV camera, etc.

Let us now consider the simplest example of amplitude modulation when the current I with radio frequency *F* is modulated with a sinusoidal acoustic frequency *f*. To obtain an effective modulation, frequency *F* should be at least 10 times greater than *f*. Figure 5-13 shows the signal, initially without modulation, and then modulated with such a simple tone as a function of time. During modulation, the current amplitude changes according to the following:

$$I_m = I_{m0} + m \sin \omega t$$

where: I_m – instantaneous value of modulated current amplitude,

I_{m0} - amplitude of the carrier wave,

m - modulation depth,

ω - acoustic frequency pulsation equals $2\pi f$

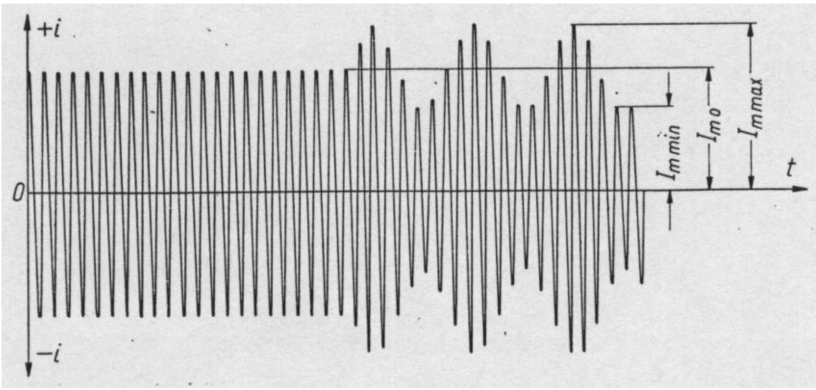

Fig. 6.5.13

The depth of modulation is called the ratio of the largest increase in carrier amplitude to the amplitude of the unmodulated carrier wave:

$$m = \frac{I_{m\max} - I_{m0}}{I_{m0}} = \frac{I_{m0} - I_{m\min}}{I_{m0}} = \frac{I_{m\max} - I_{m\min}}{2I_{m0}} = \frac{I_{m\max} - I_{m\min}}{I_{m\max} + I_{m\min}}$$

The depth of modulation can vary from zero to one (0 - 100%). Depth of modulation 0 means an unmodulated signal, depth above 100% would mean an overmodulated signal, i.e. distortion would occur. In practice, the modulation depth is usually within 30-70%. Fig. 6.5.13 shows that the current value of the modulated current can be represented by:

$$i = I_{m0}(1 + m \cdot \cos\omega t)\cos\Omega t$$

This expression takes the function of cosine instead of a sine because of its parity, which simplifies transformations and does not affect the substantive content of the expression. After the usual trigonometric transformation of the above formula, we get:

$$i = I_{m0}\cos\Omega \cdot t + \frac{1}{2} m I_{m0}\cos(\Omega + \omega)\cdot t + \frac{1}{2} m I_{m0}\cos(\Omega - \omega)\cdot t$$

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It follows from this formula that the current with straight-mode modulated amplitude consists of three cosine currents with frequencies *F, F+f, F-f* and amplitudes correspondingly equal to:

$$I_{m0} \qquad \frac{1}{2} m I_{m0} \qquad \frac{1}{2} m I_{m0}$$

In Fig. 6.5.14. these amplitudes are presented in a frequency function in the form of a spectrum. The middle band represents the current of the carrier wave, while the other two represent the currents of the lower and upper side bands. The frequencies of these currents are radio frequencies, and the difference between these frequencies and the carrier frequency *F* corresponds to the frequency of the modulation signal *f*. The currents with sideband frequencies can be separated by means of suitable filters.

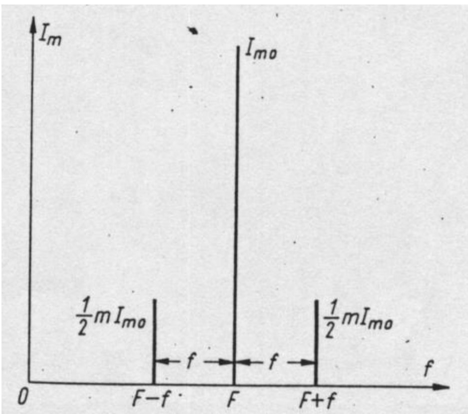

Fig. 6.5.14

In fact, the microphone currents generated when transmitting speech or music have a complex shape. They consist of many frequencies (see - spectrum of acoustic signal). Therefore, the actual spectrum of the amplitude modulated signal will include in the sidebands not one but a plurality of bands corresponding to the constituent frequencies of the actual audio signal. The recording of such a spectrum can therefore be presented in the form:

$$i = I_{m0}\cos\Omega\cdot t + \sum_{1}^{n}\frac{1}{2}\cdot m_n I_{m0}\cos(\Omega+\omega_n)\cdot t + \sum_{1}^{n}\frac{1}{2}\cdot m_n I_{m0}\cos(\Omega-\omega_n)\cdot t$$

In this formula, the first part corresponds to a carrier wave, the next parts being respectively upper and lower sidebands. The spectrum of the amplitude modulated signal with several simple tones (n = 5) is shown in Fig. 6.5.15.

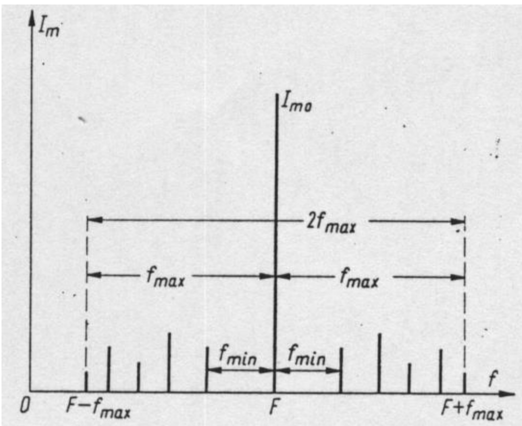

Fig. 6.5.15.

The drawing shows that the width of the entire frequency band occupied by the amplitude modulated signal is equal to the difference of the extreme lateral frequencies:

$$B = (F + f_{max}) - (F - f_{max}) = 2\, f_{max}$$

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Amplitude modulation is commonly used in radio broadcasting on medium and short waves. Because the bandwidth occupied by each broadcasting station depends on the maximum frequency of the broadcasted audio signal, it has been assumed that in broadcasting programs, sound frequencies higher than 4.5 kHz will not be transmitted. This results in not the best quality of the broadcast music, but narrows the bandwidth used by each broadcasting station to 9 kHz.

6.5.4. Single-sideband modulation (SSB)

From the definition of modulation depth, it follows that the coefficient m is directly proportional to the amplitude of the modulating signal. The height of the fringes in the sidebands thus carries information about the amplitudes or intensity of the individual sounds of the components of the modulating (acoustic) signal. In turn, the distances of the fringes from the carrier wave band determine the frequency of these sounds. All information about the modulating signal is therefore included in the sidebands. It is easy to notice (Fig. 6.5.15) that the side, bottom and top bands are symmetrical in relation to the carrier band and the information transmitted by them is thus mirrored. It is possible, without detriment to the information content of the signal, to cut one of the sidebands using a suitable filter. Also, the carrier wave band *F* does not carry any information about the modulation signal. Although it is necessary in the detection process, but it can be played directly in the receiver, without the need to transmit. The spectrum of a single-side modulated signal without a carrier wave is shown in Fig. 6.5.16.

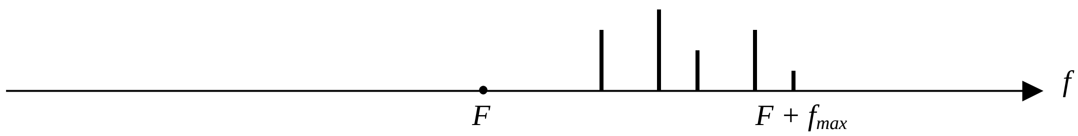

Fig. 6.5.16.

Single-band modulation, due to inferior quality, is mainly used to transmit speech on shortwave (in maritime radio also on intermediate waves). However, it has two very important advantages in relation to the full amplitude modulation: half the narrower bandwidth and less energy needed to transmit the signal. Since, as already mentioned, single-mode modulation is used only for speech transmission, the maximum frequency transmitted in the audio signal is 2800 Hz. The bandwidth is therefore for single-band modulation:

$$B = f_{max} = 2.8 \text{ kHz}$$

Single-side modulation without a carrier wave is denoted by the symbol **J3E.**

6.5.5. Frequency modulation (FM)

Frequency modulation is one of the ways to obtain angular modulation. In this modulation the carrier frequency depends on the value of the modulating signal amplitude. The method of frequency dependence in time in modulation with a simple tone is presented in Fig. 6.5.17. As in the case of amplitude modulation, the carrier wave with the frequency F_0 is generated by a very stable generator in the transmitter. In the modulation process, this frequency will vary in proportion to the value of the modulation signal; for positive values of the modulating signal, these will be changes to the values of $F_0 + \Delta F$ and for negative values to the values $F_0 - \Delta F$. The maximum frequency of the carrier frequency ΔF is called the frequency deviation. For frequency modulation to be effective, the carrier frequency should be at least 1000 times the frequency of the modulation signal. In practice, frequency modulation is used on ultrashort waves, above 30 MHz.

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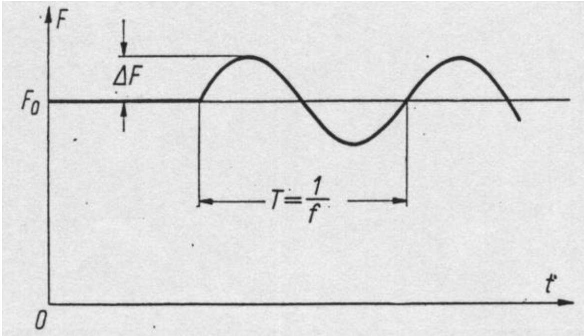

Fig. 6.5.17

Assuming that during the modulation the carrier frequency changes periodically according to the relationship:

$$F = F_0 + \Delta F \cos 2\pi f t$$

we get a formula for modulated current:

$$i = I_{m0} \sin(2\pi F_0 t + \beta \sin 2\pi f t) \qquad gdzie \qquad \beta = \frac{\Delta F}{f}$$

After the transformations we get a series of expressions determining the frequency currents of the spectrum of the frequency-modulated signal. The spectrum, like in the case of amplitude modulation, has a band with a carrier frequency and two side bands. The difference is that in the case of frequency modulation, the spectrum is infinitely wide, but the coefficients at the higher order elements decrease quickly and can be omitted. The practical spectrum width (99% of energy) is given by the formula:

$$B = 2(\Delta F + f_{max})$$

where f_{max} - maximum frequency in the modulation signal.

In Fig. 6.5.18, the waveforms for frequency modulation are shown: (a) it is a modulating signal with the frequency f, (b) is a modulated signal.

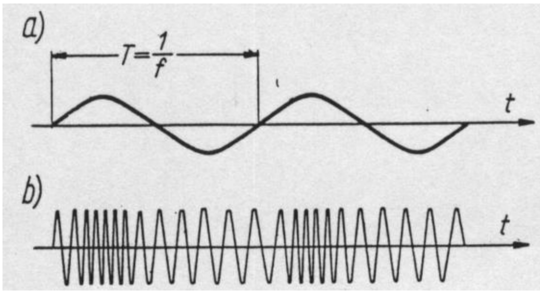

Fig. 6.5.18.

6.5.6. Phase modulation PM

In phase modulation, the phase angle of the carrier wave depends on the value of the modulation signal. When modulating the carrier phase with a simple sinusoidal tone with the frequency f, the angle changes periodically according to the relationship:

$$\varphi = \varphi_0 + \Delta\varphi \sin 2\pi f$$

where Δφ specifies the maximum change in the phase angle and is called the phase deviation. The phase modulated current will therefore be expressed by a formula:

$$i = I_{m0} \sin(2\pi F_0 t + \Delta\varphi \sin 2\pi f t + \varphi_0)$$

In Fig.6.5.19. the idea of phase modulation is presented.

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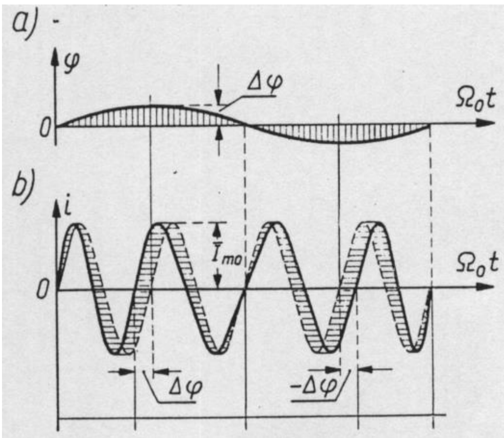

Fig. 6.5.19.

The fragment (a) of the figure shows the changes of the phase angle (proportional changes in the value of the modulating signal), and the fragment (b) shows the modulated current waveform (solid line). The current waveform before modulation is marked by a dashed line. As can be seen from this figure, not only the phase angle, but also the frequency changes during phase modulation. The maximum frequency change or frequency deviation is:

$$\Delta F = \Delta\varphi f$$

So we see that with the right choice of modulation parameters, frequency modulation and phase modulation give the same results. However, this only applies if the modulating signals are sinusoidal signals. In the case of modulation with other signals (triangular, rectangular etc.), these modulations differ substantially.

In Fig. 6.5.20. a sinusoidal modulation signal as well as frequency and phase modulation results are presented. In both cases, the modulated signals are identical. There was only their shift in relation to each other. When modulating the frequency, the maximum density of the modulated signal occurs for the highest value of the modulation signal. In the case of phase modulation, it occurs when the value of the modulating signal is zeroed in the direction of rising values.

When comparing the amplitude modulation with the angular modulation one can notice two basic advantages of the latter. Firstly, a wider frequency band is transmitted in the case of angular modulation, allowing for more accurate signal transmission. This is especially important for musical signals. Secondly, the angular modulation is much more resistant to interference. Most of the disturbances occurring during transmission are of an amplitude nature, i.e. they overlap with the amplitude of the signal.

Because in the amplitude modulation, the signal information is stored in the amplitude changes, each disturbance causes an undesirable change in the transmitted signal. With angular modulation, amplitude changes induced by interferences do not affect the signal value, because all information is contained in frequency changes. The harmful amplitude changes caused by disturbances can thus be removed by means of appropriate circuits in the receiver.

6.5.7. Detection

Detection, also called demodulation, is a process that is inverse to modulation. It consists in reproducing the information stored in it from the modulated signal. Depending on the modulation method used, we have the amplitude detection and frequency detection used to demodulate the angularly modulated signals.

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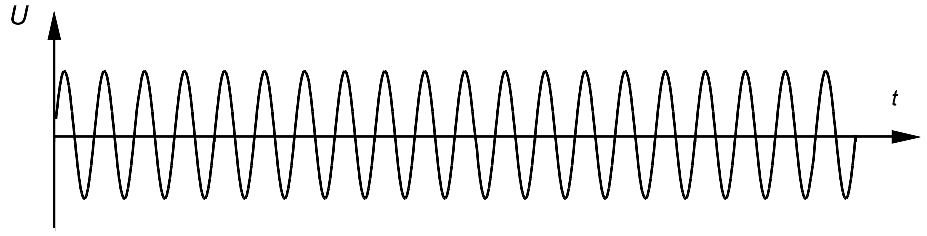
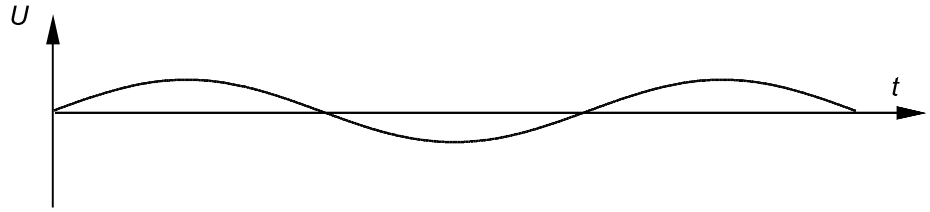
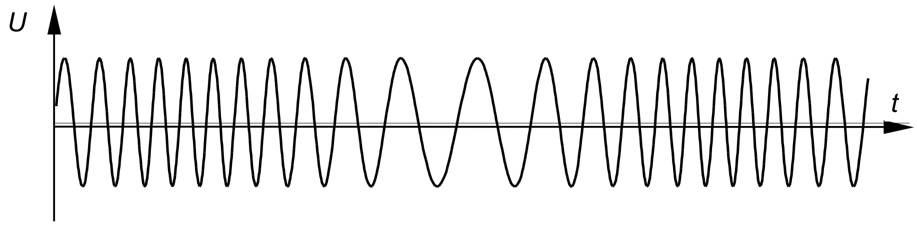


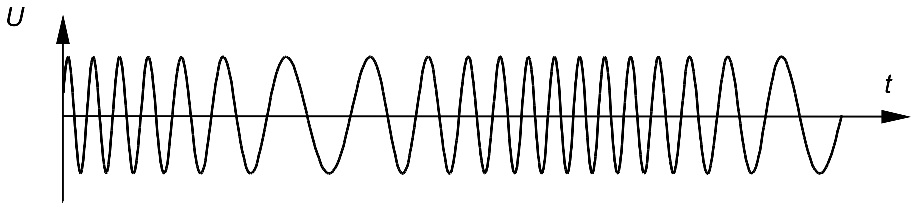
Carrier wave



Modulating signal



Frequency modulated signal



Phase modulated signal

Fig. 6.5.20.

6.5.8. Amplitude detection

The amplitude modulated signal has an average value of zero. Bringing such a signal to the loudspeaker will not cause any vibrations of the membrane, i.e. we will not hear the voice. The diaphragm is too inert to be able to vibrate to a very high radio frequency, and due to the zero mean value there is no other force that would affect it. The amplitude detection consists in such a transformation of the modulated signal, so that the average value proportional to the envelope, ie to the value of the modulating signal, appears in it. This can be achieved by passing the received signal modulated by a diode which, by conducting the current one-way only leaves the positive halves of the signal. The waveform obtained in this way has an average value with a shape similar to the envelope of the signal. This is shown in Figure 6.5.21.

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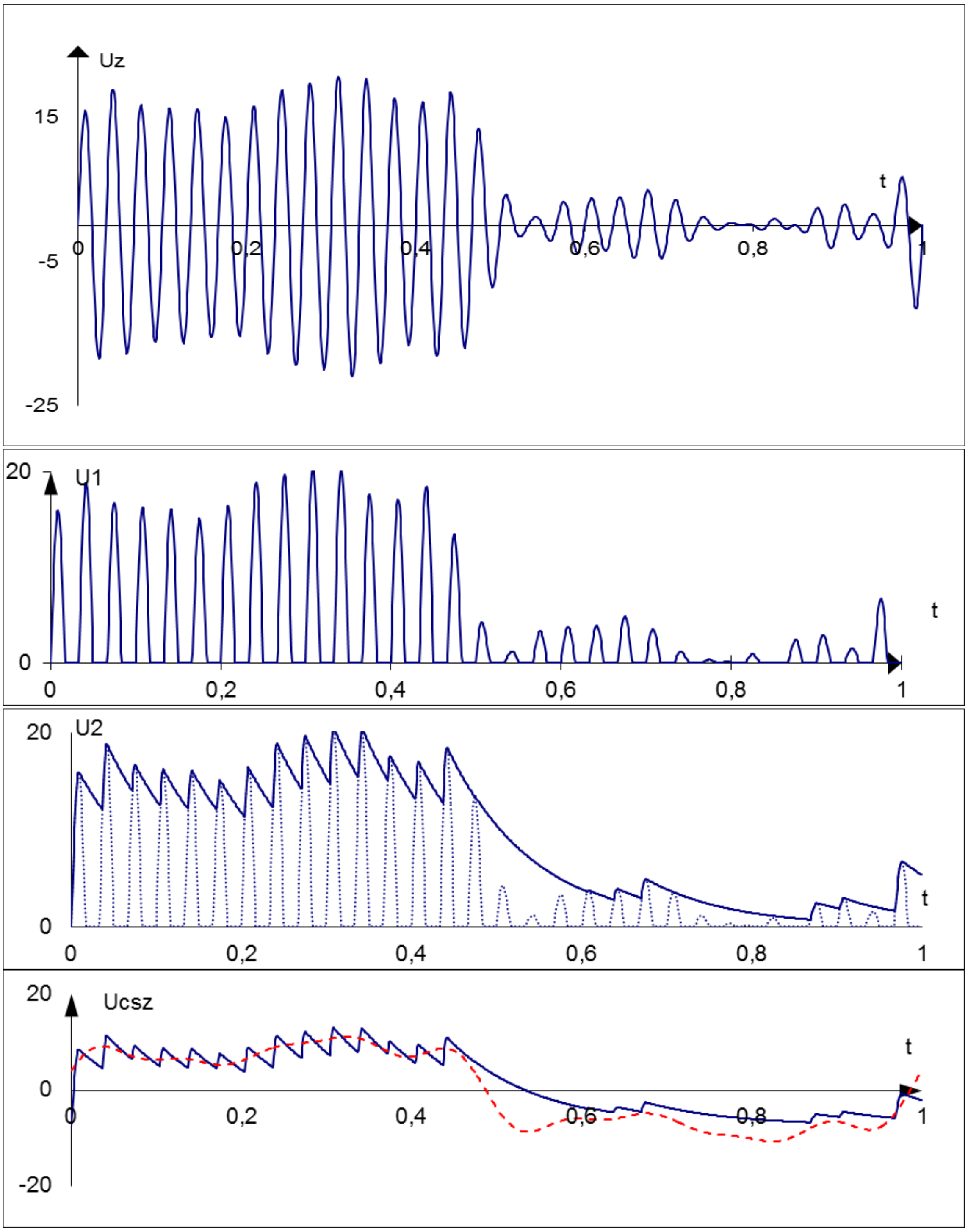

Fig. 6.5.21

The waveform *a* represents the current as a function of time before detection, and *b* the current flow after detection. Amplitude detection can be accomplished in several ways. One of the more common amplitude detector circuits is shown in Fig. 6.5.22.

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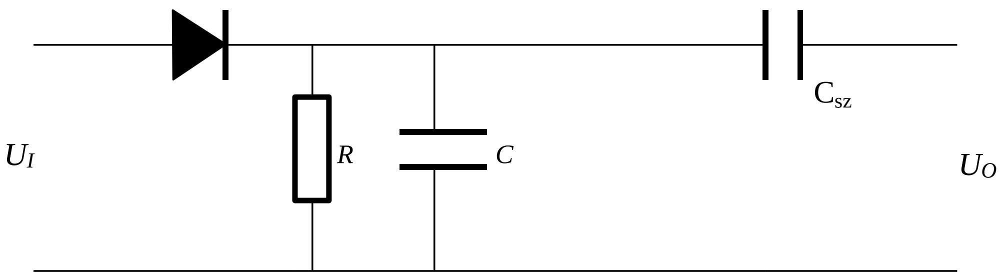

Fig. 6.5.22

A voltage such as the current in figure 6.5.21 b appears on the R resistor. The average value of this voltage has a shape similar to the envelope, but not identical. In order to achieve greater detection fidelity, the capacitor C is connected in parallel to the resistor. The resistor system with the capacitor works similarly to the half-wave rectifier system. At moments of voltage drop to zero, the capacitor maintains it at the envelope level due to accumulated charge.

The difference with the rectifier lies in the fact that in the case of a detector, the RC time constant must be optimally matched to the maximum frequency in the modulating signal, so that the system can keep up with envelope changes. Too high a time constant would cause distortions of the demodulated signal. The capacitor *C* has the task of eliminating the constant component present in the signal after detection. As a result, a signal of the same shape as the modulation signal is obtained at the detector output.

6.5.9. Frequency detection

The task of the frequency detector is to convert the received voltage of modulated frequency into an acoustic frequency voltage. The frequency detector circuit consists of three blocks: an amplitude limiter, a discriminator and an amplitude detector. The discriminator and detector are usually combined into one system, as shown in Figure 6.5.23. The figure also shows the signals at the inputs and outputs of individual blocks of the frequency detector.

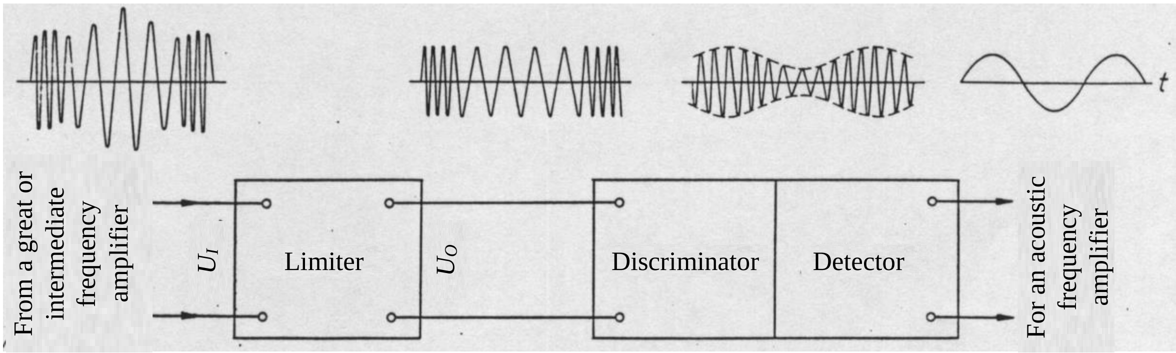

Fig. 6.5.23

The limiter's task is to eliminate amplitude disturbances that occur during signal transmission. The limiter operates on the basis of a properly polarized diode, which cuts the amplitude of the signal at a properly selected level, leaving the frequency modulation unchanged.

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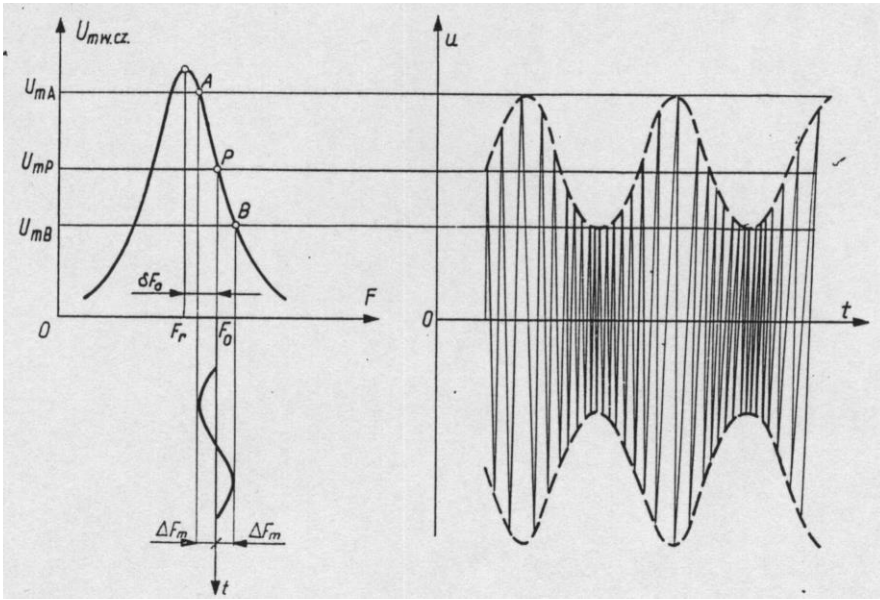

Fig. 6.5.24

The task of the discriminator is to transform the signal of a fixed amplitude and variable frequency into a signal with an amplitude proportional to the frequency, i.e. a signal conversion with frequency modulation to a signal of modulated amplitude. The simplest discriminator is the resonant circuit operating on the slope of the characteristic. When working on the *A-B* section, the slope of the resonance characteristic, which is approximately straight line, we get voltage changes approximately proportional to frequency changes. The operating point *P*, corresponding to the carrier frequency F_0 should be located in the middle of the straight section of the characteristic slope. Frequency fluctuations ΔF_m should not extend beyond the rectilinear part.

The discussed principle of the work of such a discriminator is presented in Fig. 6.5.24. In fact, the slope of the resonant circuit characteristic is not a straight line, so the amplitude changes will not be fully proportional to the frequency changes, i.e. there will be distortions in the detection process. Therefore, the actual systems of discriminators are more complicated. They use double resonant circuits with different resonance frequencies, so combined that one can obtain a resultant characteristic that is a difference in their characteristics. This is shown in Figure 6.5.25. The resultant characteristics have a long straight line that allows the discrimination process to be carried out without distortion.

The purpose of the detector is to convert the amplitude demodulated signal obtained after the discriminator into an acoustic signal. The detector that is part of the frequency detection system works on the same principle as the amplitude detector described in chapter 7.1. As a result, at the output of the frequency detector, we get the same signal that was saved on the carrier wave during the modulation.

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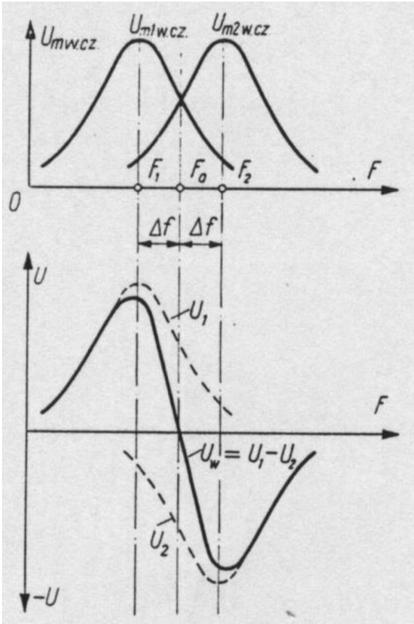

Fig. 6.5.25

6.5.10. Frequency transformation

The frequency conversion consists in changing the frequency of the carrier wave without changing the shape and nature of the modulation. Fig. 6.5.26 shows the amplitude modulated signal before and after the frequency conversion. Of course, it is also possible to perform frequency transformations for frequency-modulated signals. Then the nominal frequency of the carrier wave F_0 will change, and the frequency deviation ΔF will not change. There are many reasons for using frequency conversion. The most important of them are:

- Possibility to amplify signals received from different stations on one intermediate frequency (radio, television, etc.),,
- Enable undisturbed transmission of signals by means of radio links,
- Radical frequency reduction for enabling signal transmission and amplification using conventional devices.

We will discuss the reasons for using the frequency conversion. The first reason is that in the antenna of the receiver there are signals from many stations with different frequencies: F_1, F_2, F_3, ...F_n, at similar voltage levels. In order to select one station, a resonant circuit with an adjustable capacity is coupled to the antenna. Thanks to the resonant characteristics of the circuit, the voltage levels at the output will vary. A high voltage level is obtained for the selected station, and the remaining stations will be suppressed. The more the frequency of a given station differs from the frequency of the selected station, the greater the attenuation of the signal. This is shown in Figure 6.5.27.

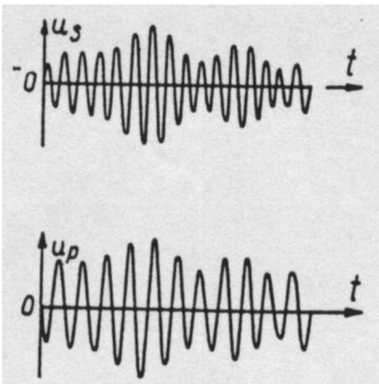

Fig. 6.5.26

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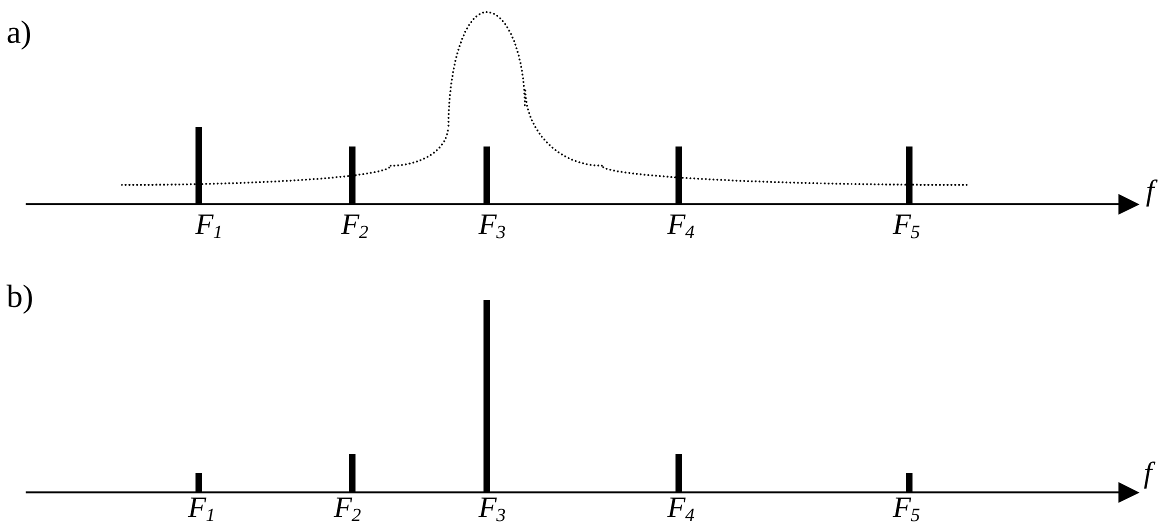

Fig. 6.5.27

In part "a" of the figure shows the level of signals in the antenna, and in part "b" the level of signals at the input of the resonant circuit. In order to tune to the desired station, eg F_3, the capacity in the resonance circuit should be selected so that the resonance frequency F_0 of the circuit is equal to the station frequency F_3.

$$F_0 = \frac{1}{2\pi\sqrt{LC}} = F_3$$

As shown in Fig. 6.5.27, at the output of the resonant circuit we have a clearly distinguished signal from the selected station. There are also, though suppressed, signals from unwanted stations. Therefore, in the further amplification process, selective amplifiers tuned to the frequency of the selected station must be used. The higher the number of amplifier stages, the higher will be the suppression of undesired signals in relation to the signal of the selected station. There are even ten-stage amplifiers in high-end receivers.

In the case of a change in the received station, it would be necessary to change not only the capacity in the resonant input circuit, but also to tune all stages of the amplifier to a new frequency. This activity would be so cumbersome that it is practically impossible. Therefore, a frequency conversion system is inserted between the input circuit and the amplifier. In this system, the frequency of the received signal F_1, F_2, F_3, ...F_n, is changed to a precisely defined frequency for a given receiver, frequency called the intermediate frequency F_P. All stages of the amplifier can now be tuned once and for all to this frequency.

The second reason for the use of frequency conversion is associated with the transmission of signals, especially in the VHF range. Because radio waves in this frequency range propagate in straight lines, the range of reception is limited, depending on the height of the antennas and the terrain up to 60 - 100 km.

If it is necessary to send a signal over a longer distance, a chain of relay stations is placed on the transmission route. Each such station consists of a receiver and a transmitter. A frequency transformation must take place between the transmitter and the receiver so that the signal is transmitted at a different frequency than was received. If no transformation was used, two signals shifted in time would reach the radio or television receivers between the two relay stations. The overlapping of these signals would cause reception interference.

The third reason for using frequency transformation is, for example, in radars. The principle of radar operation is generally based on the fact that the rotating antenna sends a very focused beam of signals. These signals, encountering an obstacle, bounce off and return to the antenna. The time measured from the moment the signal is sent to its return determines the distance from the obstacle, and the angle of the antenna's position towards the north at the time of reception determines the bearing on this obstacle. According to the theory of antenna design, in order to obtain a good focus of

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the beam, and therefore the proper radar image resolution, the linear dimensions of the antenna should be at least 100 times the wavelength of the transmitted wave.

Because the antennas on the ships can not be too big, therefore, very short waves, ie very high frequencies should be used. In sea radars, frequencies between 3 - 10 GHz are used. A signal with such a high frequency is thus sent and returns as a reflected echo. The level of the returning signal is so low that to be processed further it must first be appropriately reinforced. The problem is that conventional transistors or integrated circuits are not suitable for operation at such high frequencies.

Also, signals with such frequencies can not be transmitted with ordinary cables, expensive, large waveguides must be used for this. To avoid these problems, immediately after receiving the returning signal, its frequency is reduced several hundred times to the value of 50 - 60 MHz. Sending and amplifying signals with this frequency is no longer a problem.

There are many methods to realize the frequency transformation. The sum conversion will be discussed. To convert the frequency in this way, two signals are input to one common circuit: the received signal with the frequency F_s, and the signal from the local generator, called the heterodyne with the frequency F_h. In this circuit there must be a non-linear element, eg a diode. As a result, a series of signals with different frequencies, being the frequency combinations F_s and F_h, will be created in the circuit. The necessary signal with the frequency being the difference F_h - F_s is selected by means of a band filter tuned to this frequency.

The scheme of the frequency transformation system is shown in Fig. 6.5.28. Fig. 6.5.29 presents the characteristics of a diode, which is a non-linear element.

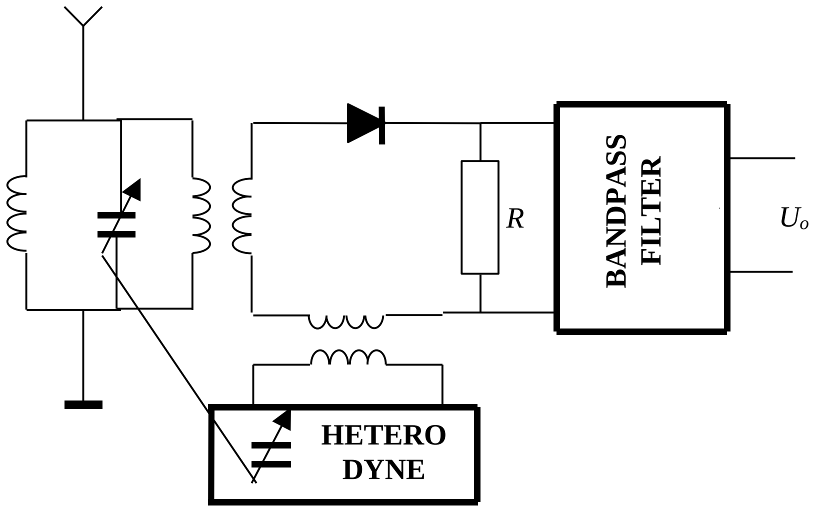

Fig. 6.5.28

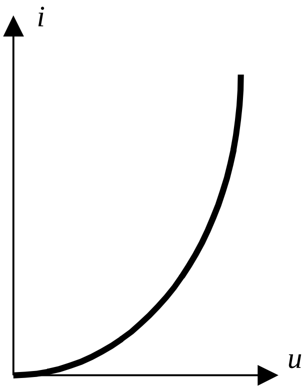

Fig. 6.5.29

A sum of signal voltages appears in the circuit:

$$u_s = U_{ms} \cos\Omega_s t = U_{ms} \cos 2\pi F_s t$$

And heterodyne:

$$u_h = U_{mh} \cos\Omega_h t = U_{mh} \cos 2\pi F_h t$$

the total voltage will be equal:

$$u_1 = u_s + u_h = U_{ms} \cos\Omega_s t + U_{mh} \cos\Omega_h t$$

Assuming that the diode's characteristic is approximately a parabola, the current flowing through the diode will be equal:

$$i = au_1{}^2 = a(U_{ms} \cos\Omega_s t + U_{mh} \cos\Omega_h t)^2 = aU_{ms}^2 \cos^2\Omega_s t + aU_{mh}^2 \cos^2\Omega_h t + 2aU_{ms}U_{mh} \cos\Omega_s t \cdot \cos\Omega_h t$$

using known trigonometric relationships:

$$\cos^2\alpha = \frac{\cos 2\alpha + 1}{2} \quad oraz \quad \cos\alpha \cdot \cos\beta = \frac{1}{2}\cos(\alpha+\beta) + \frac{1}{2}\cos(\alpha-\beta)$$

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we will get after transformations:

$$i = \frac{aU_{ms}^2}{2} + \frac{aU_{mh}^2}{2} + \frac{aU_{ms}^2}{2}\cos 2\Omega_s t + \frac{aU_{mh}^2}{2}\cos 2\Omega_h t +$$
$$+ aU_{ms}U_{mh}\cos(\Omega_h + \Omega_s)t + aU_{ms}U_{mh}\cos(\Omega_h - \Omega_s)t$$

As it results from the last expression, the current flowing through the diode in addition to the constant component, variable components with frequencies:

$$2F_s \qquad 2F_h \qquad F_h + F_s \qquad F_h - F_s$$

This current, flowing through the resistor *R*, will create proportional voltages in accordance with Ohm's law. The band filter connected to the resistor is tuned to the differential frequency F_h - F_s, so only the signal with this frequency will appear at the output of the filter.

$$u_{wy} = aRU_{mh}U_{ms}\cos(\Omega_h - \Omega_s)t$$

Amplitude of voltage from heterodyne is a constant value, so you can mark it:

$$aRU_{mh} = k$$

as a result, we have:

$$u_{wy} = kU_{ms}\cos(\Omega_h - \Omega_s)t$$

This voltage is just as modulated as the signal voltage. It differs from it only by frequency. Heterodyne is an *LC* type generator. Its frequency can be changed using the capacitor *C*. The capacitor is mechanically coupled to the capacitor in the input circuit. When changing the frequency of the received station, the heterodyne frequency changes at the same time. As a result, the differential frequency called the intermediate frequency will be a constant value.

$$F_p = F_h - F_s = \text{const.}$$

6.6. Report template

The report template starts from the next page. Contains 12 pages.

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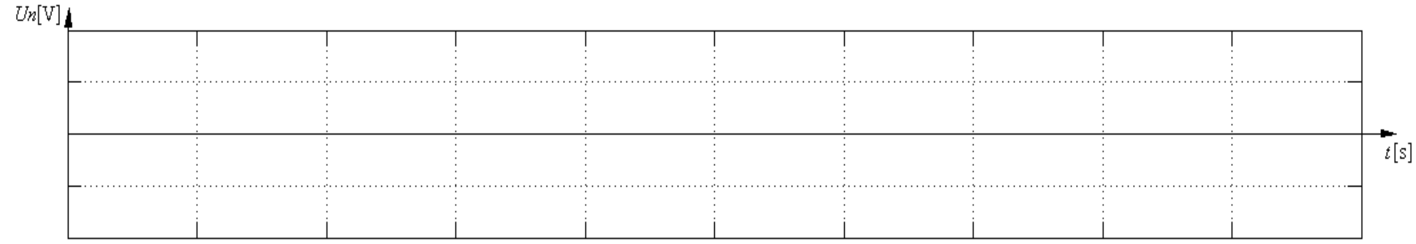
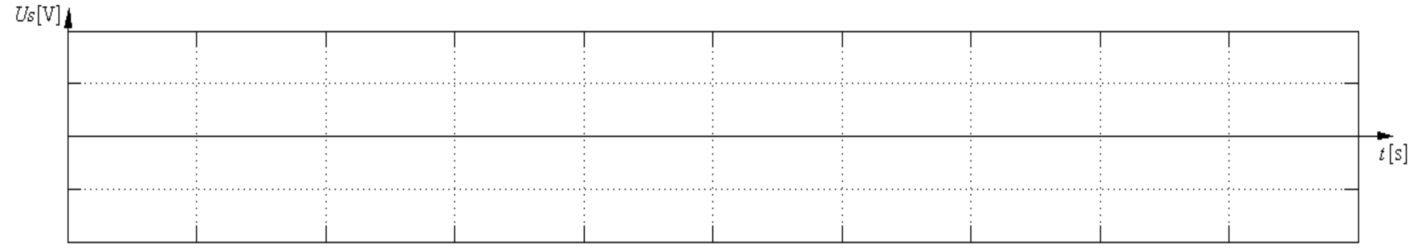
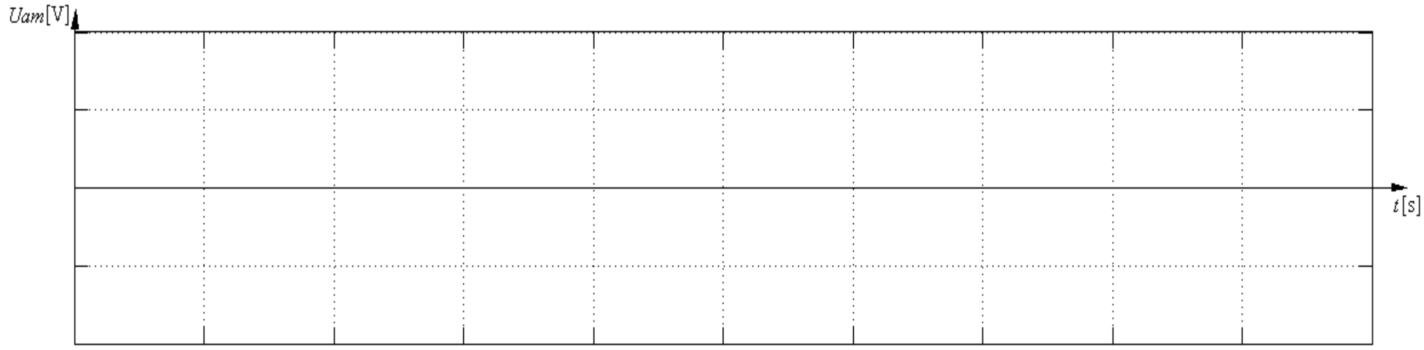
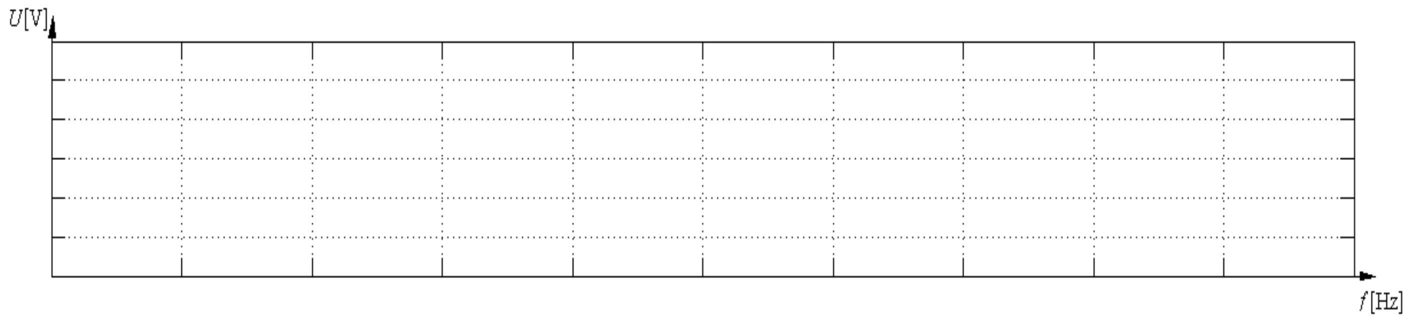
U_n [V]	f_n [kHz]	U_s [V]	f_s [kHz]	Shape of the modulating signal

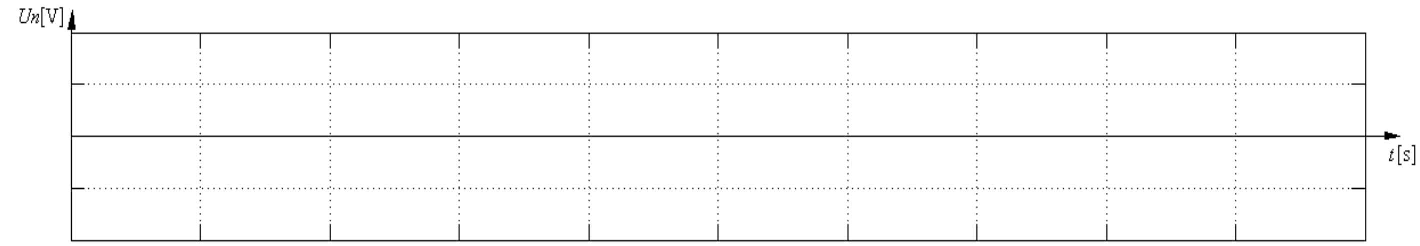


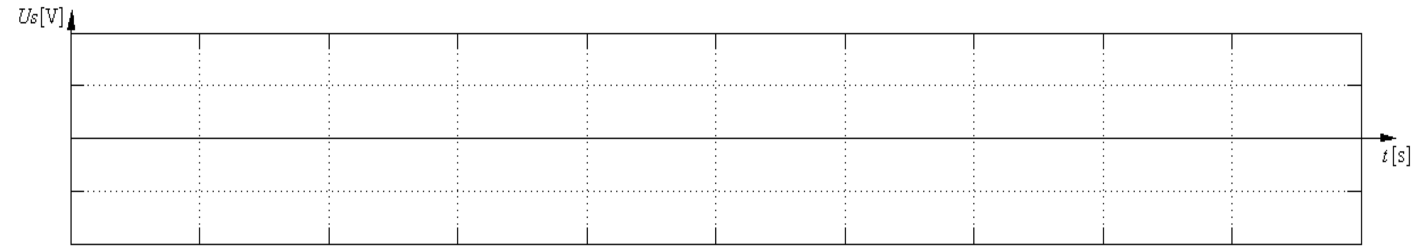


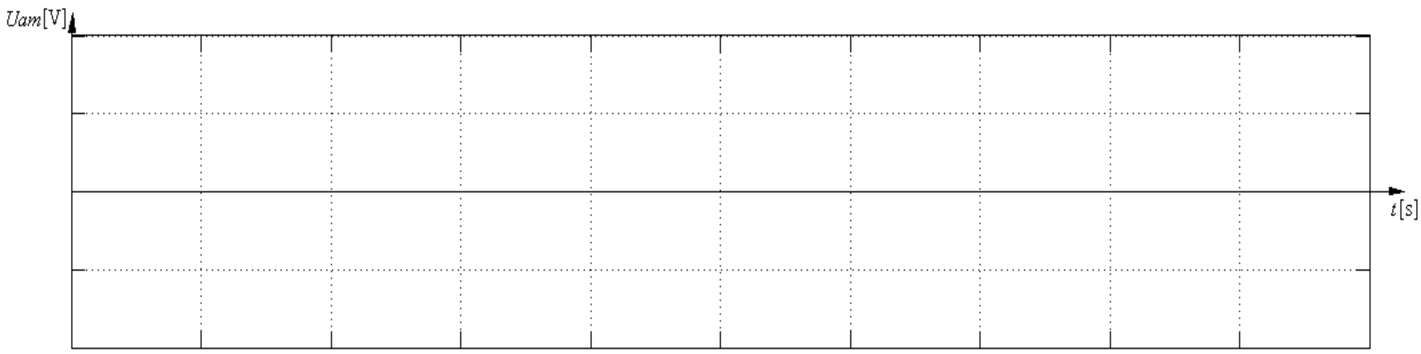


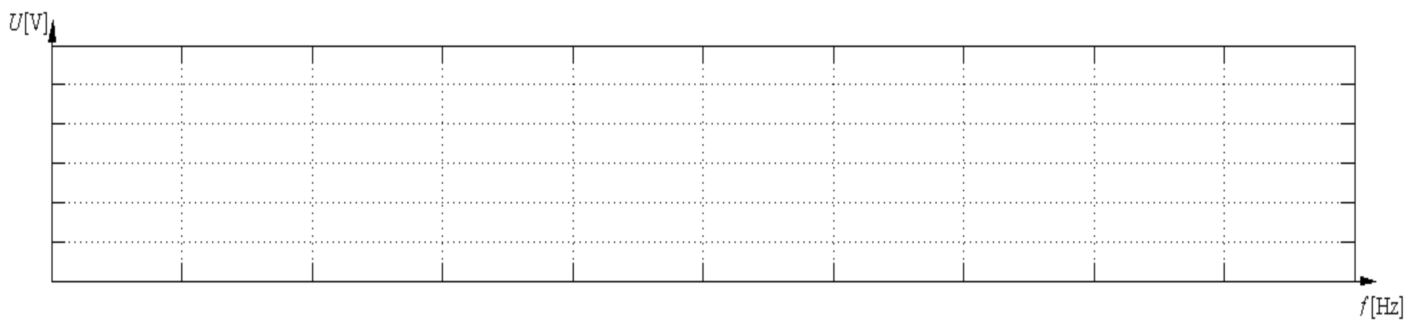


U_n [V]	f_n [kHz]	U_s [V]	f_s [kHz]	Shape of the modulating signal

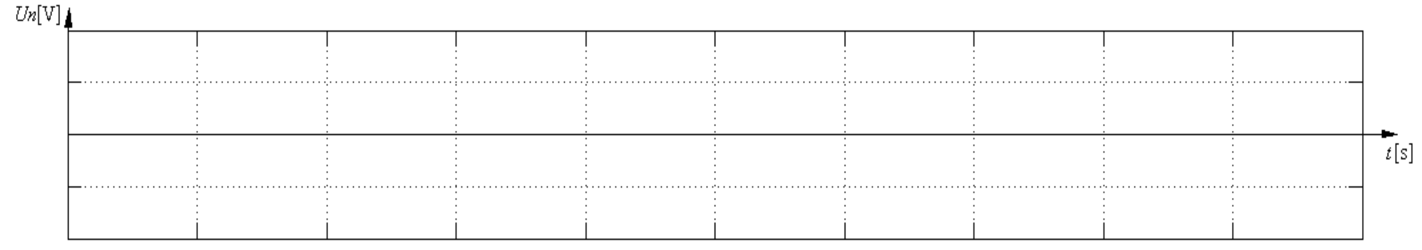


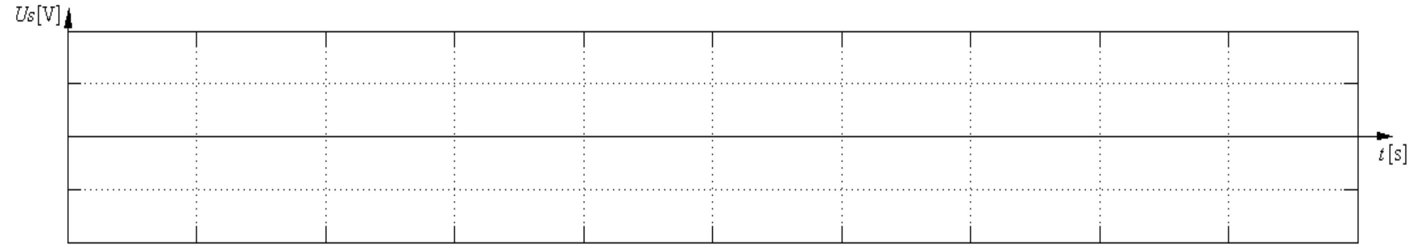


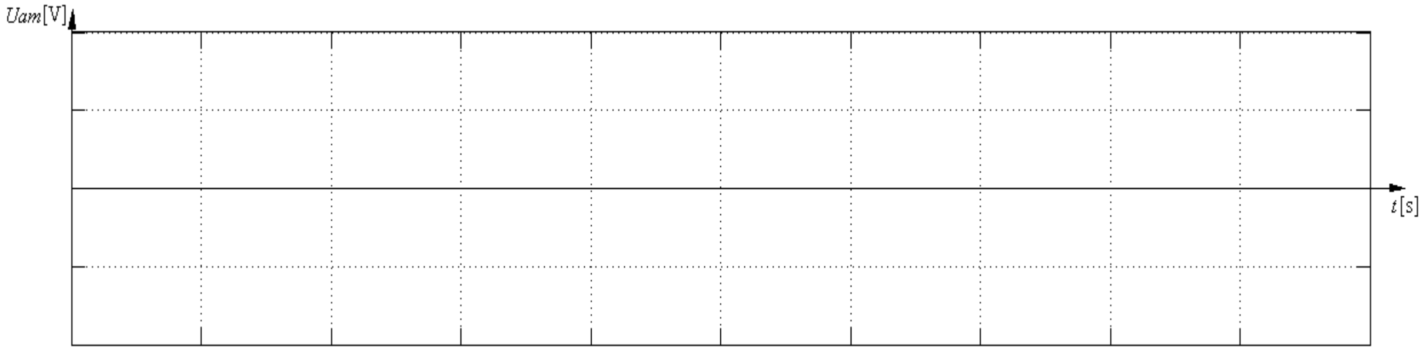


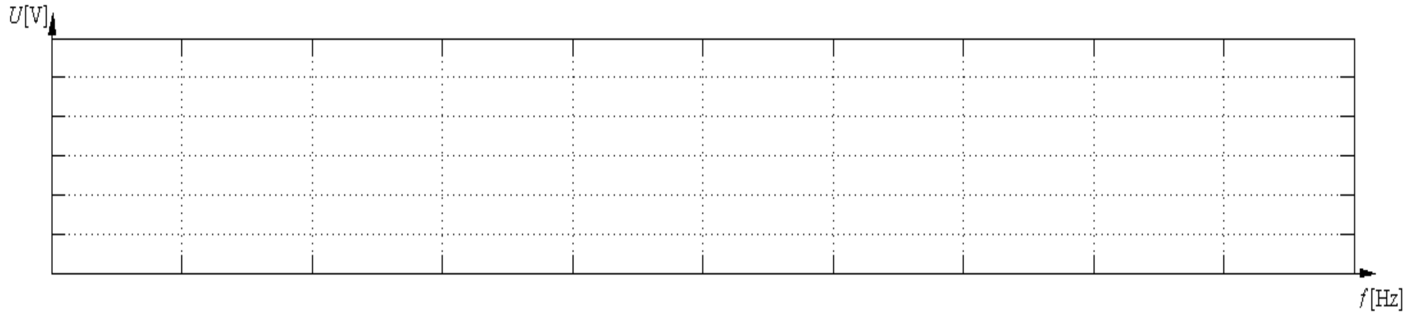


U_n [V]	f_n [kHz]	U_s [V]	f_s [kHz]	Shape of the modulating signal

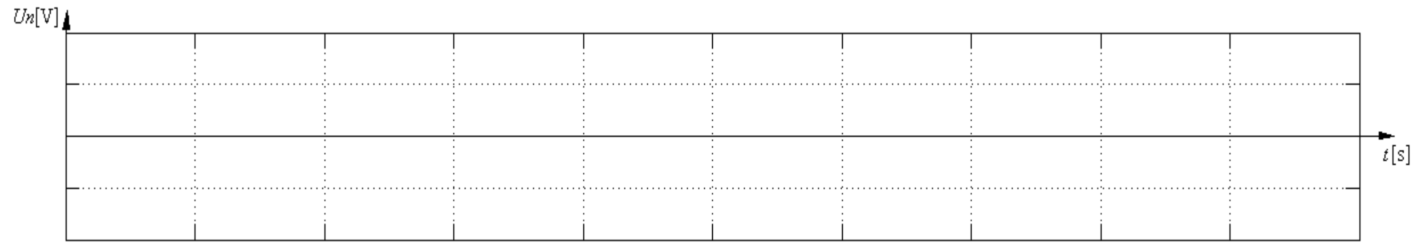


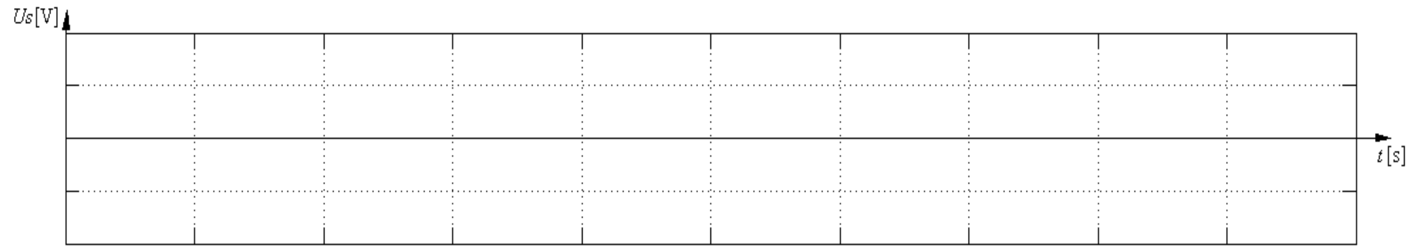


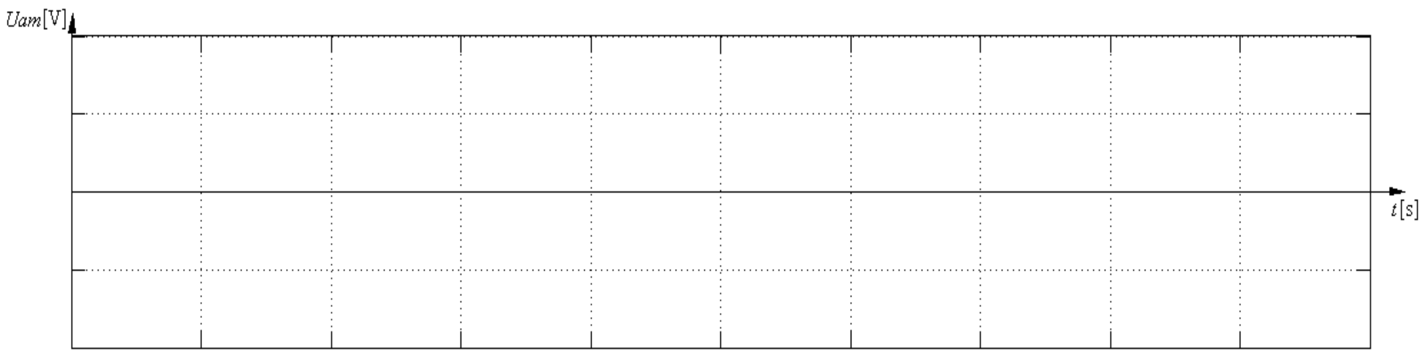


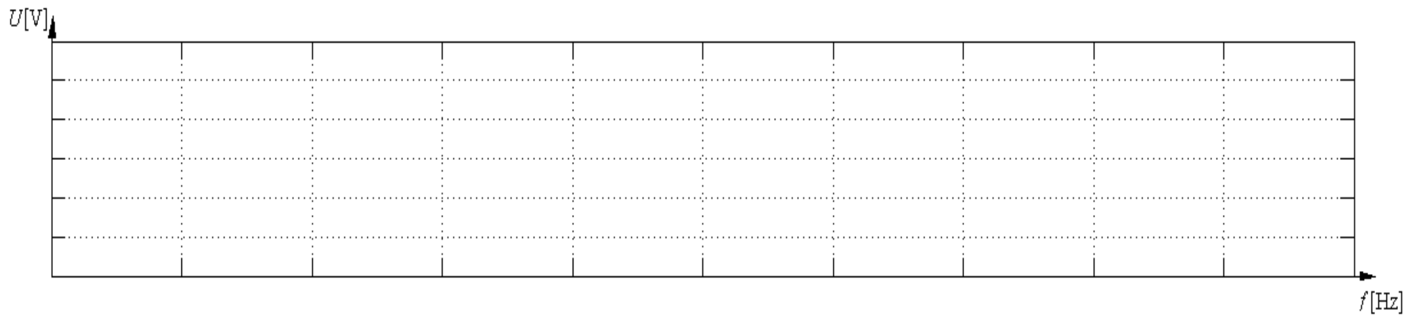


U_n [V]	f_n [kHz]	U_s [V]	f_s [kHz]	Shape of the modulating signal

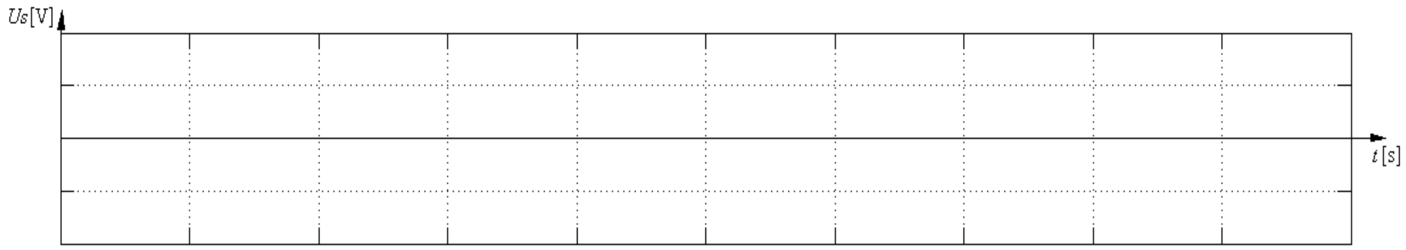


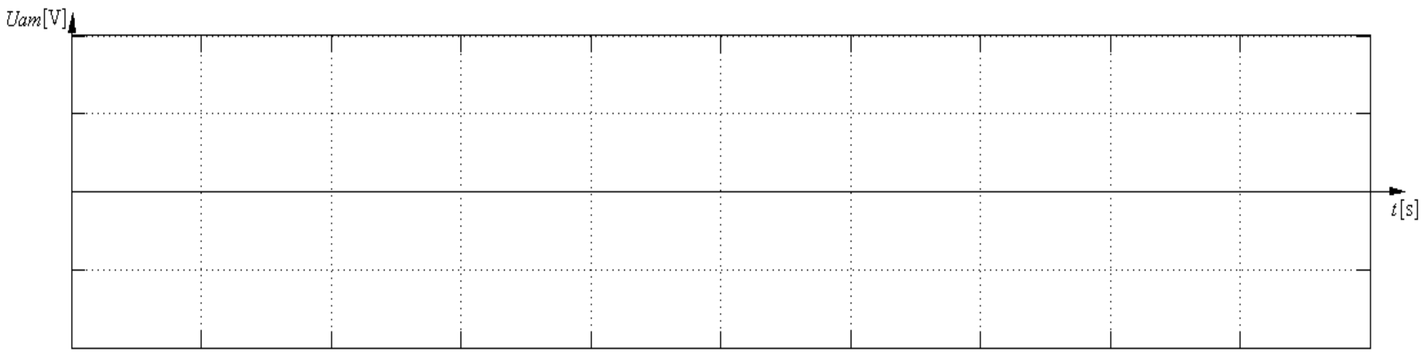


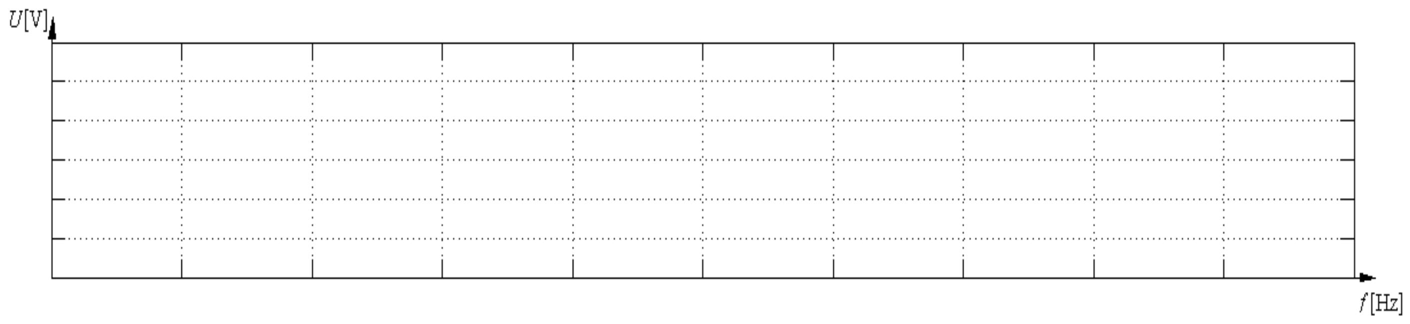




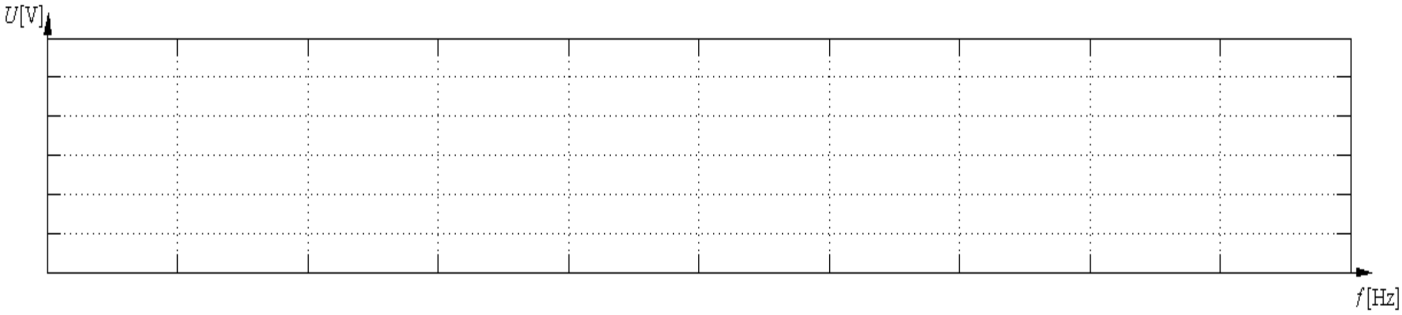
U_n [V]	f_n [kHz]	U_s [V]	f_s [kHz]	Shape of the modulating signal

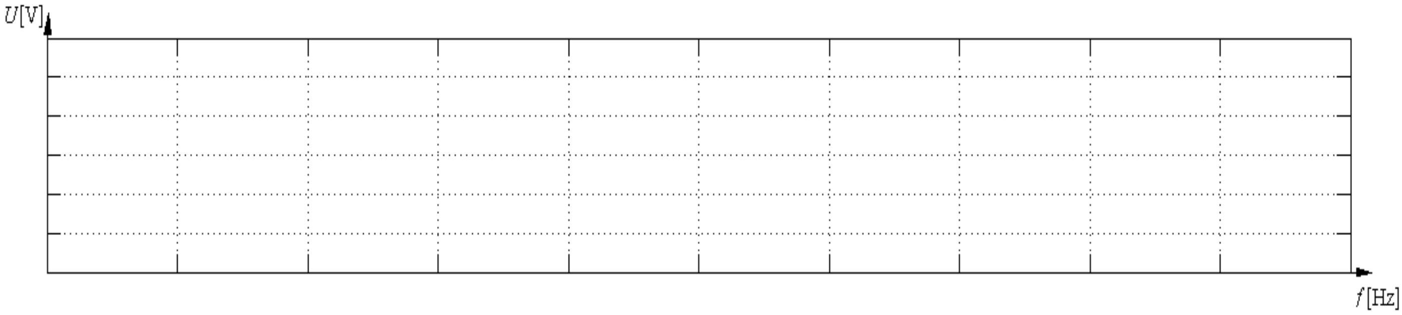


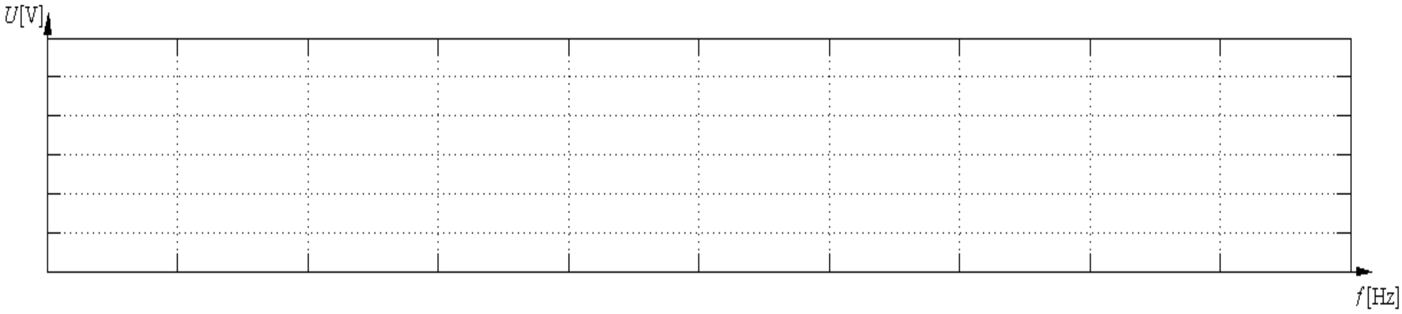




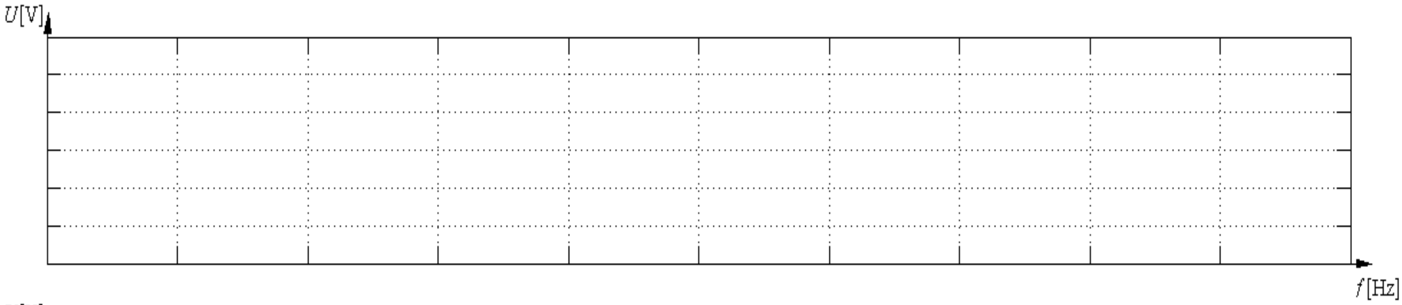
f_s [kHz]	f_{n1} [kHz]	f_{n2} [kHz]	f_{n3} [kHz]

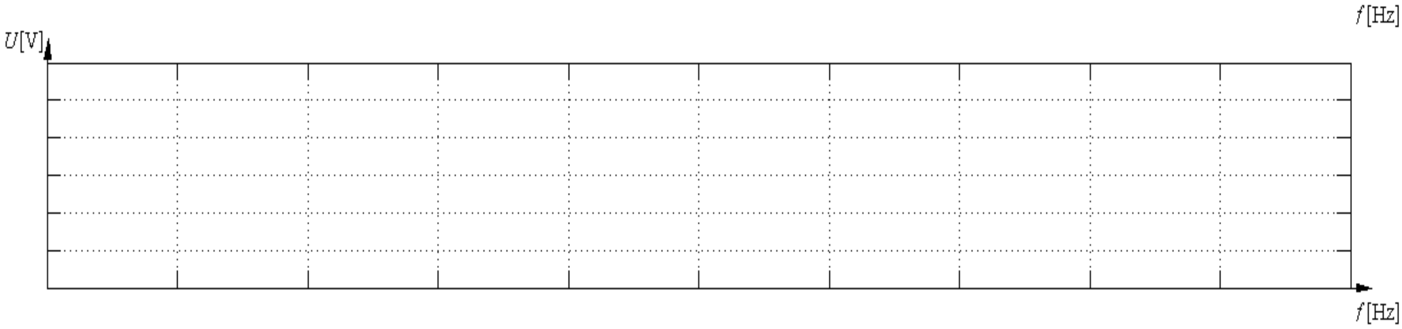


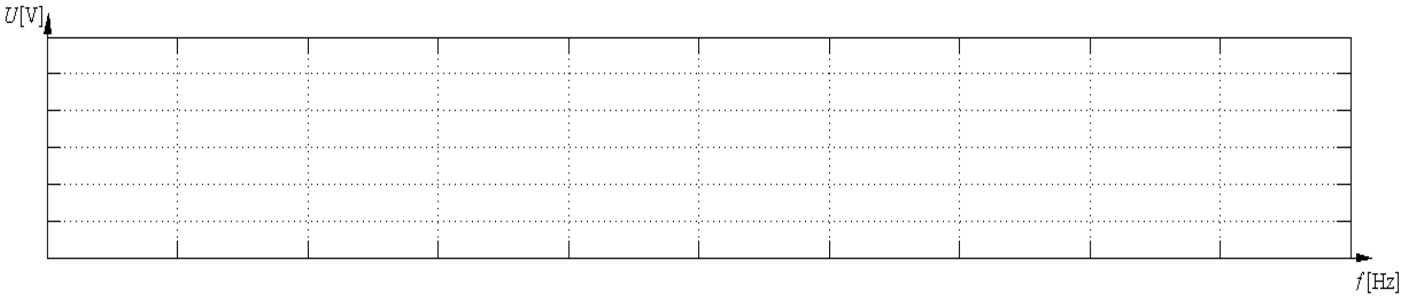




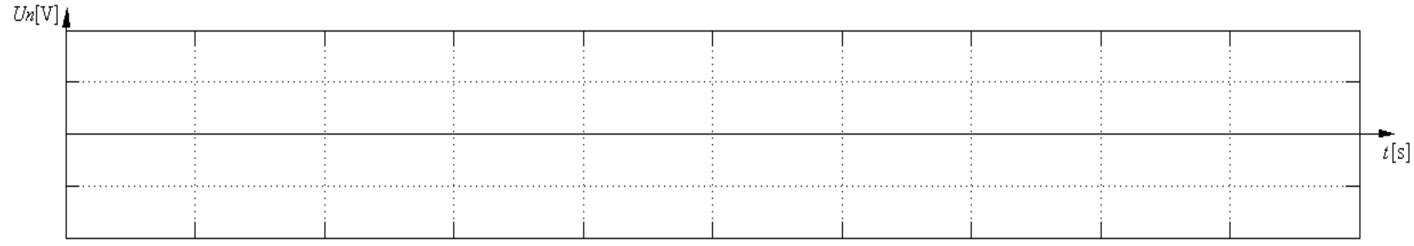
f_n [kHz]	f_{s1} [kHz]	f_{s2} [kHz]	f_{s3} [kHz]

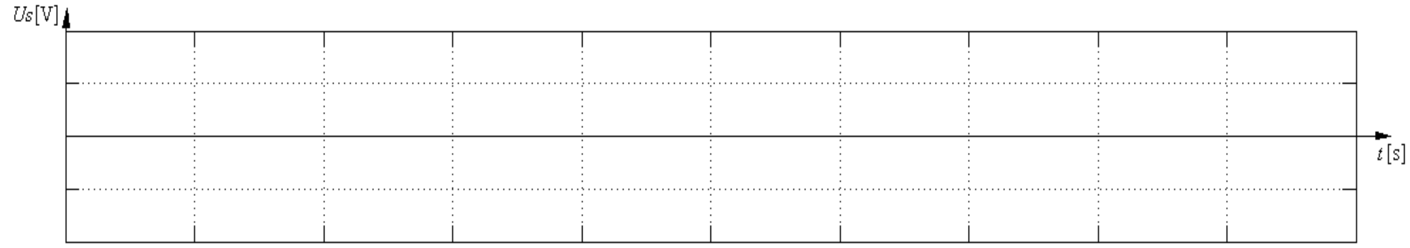


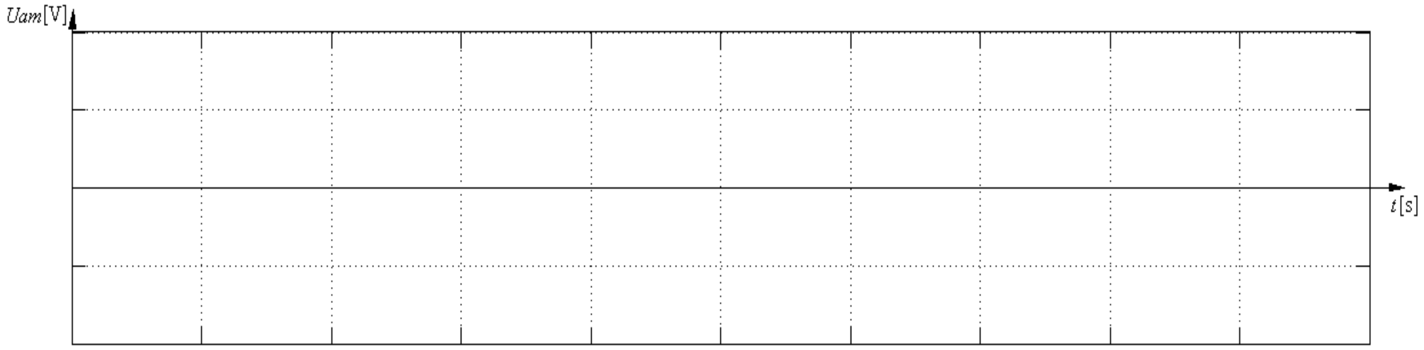


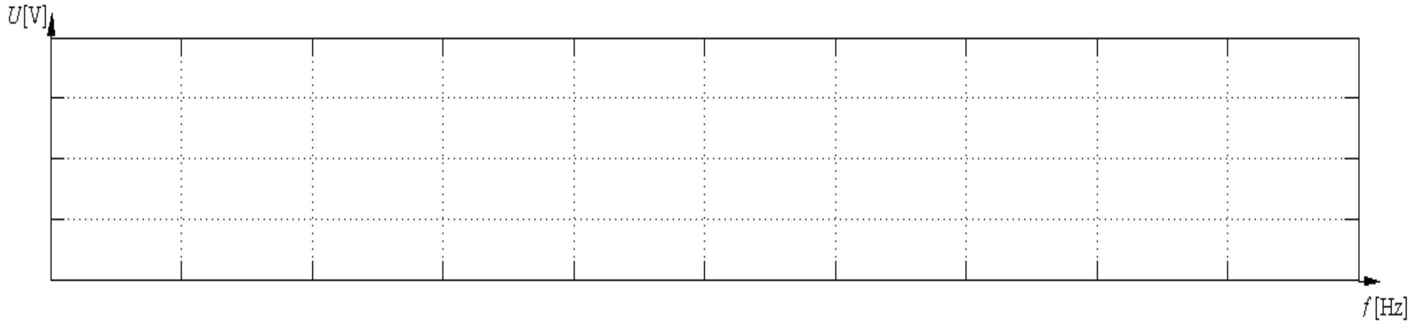


U_n [V]	f_n [kHz]	U_s [V]	f_s [kHz]	m

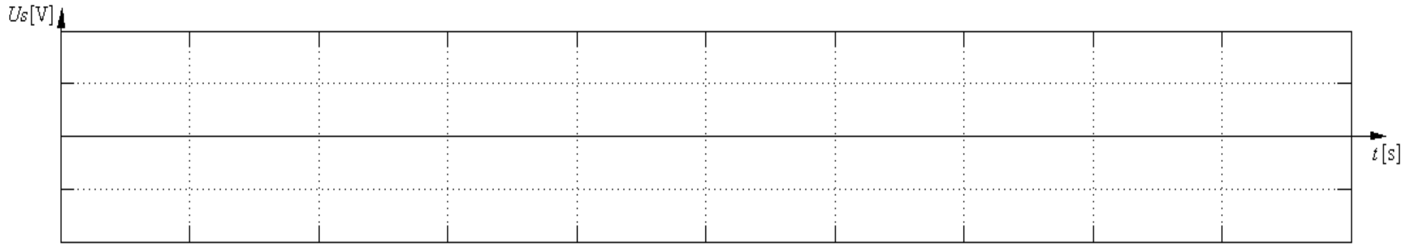


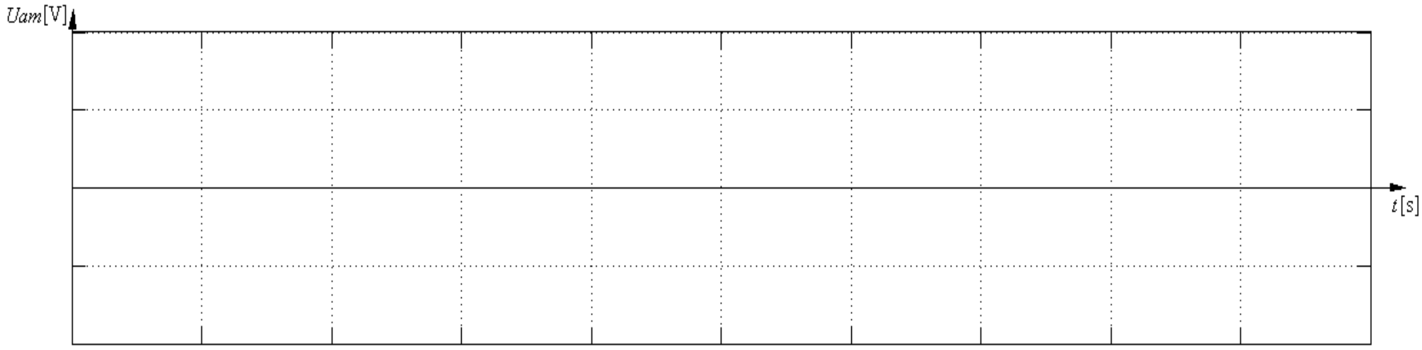


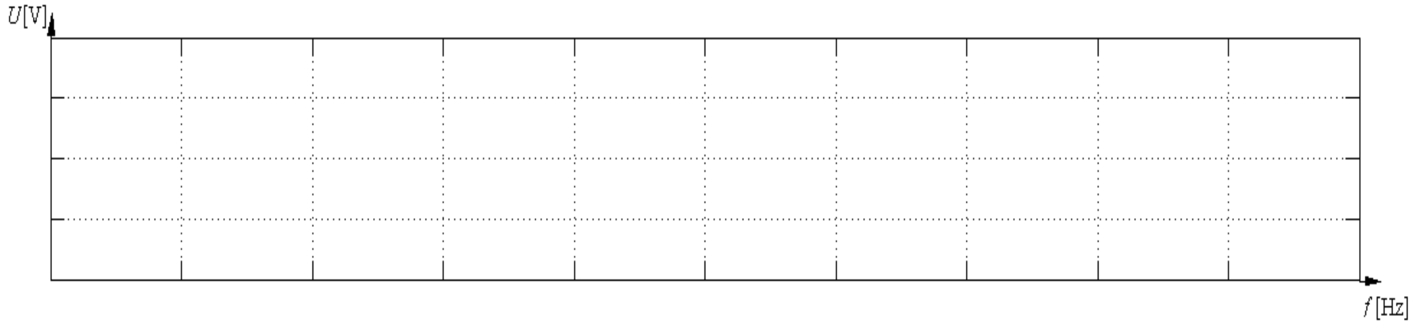




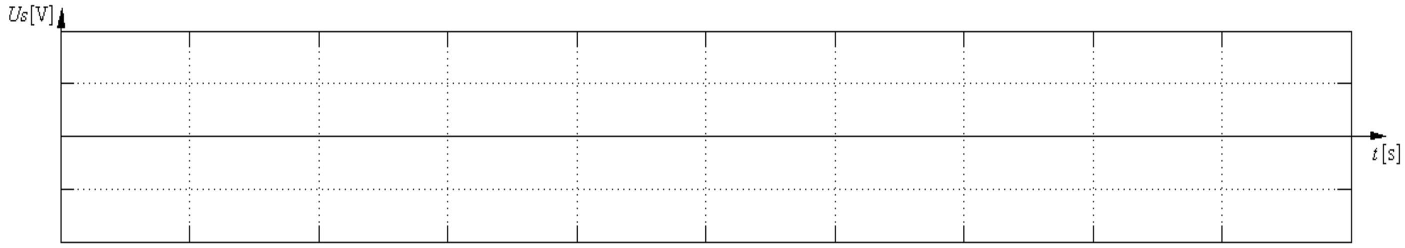
U_n [V]	f_n [kHz]	U_s [V]	f_s [kHz]	m

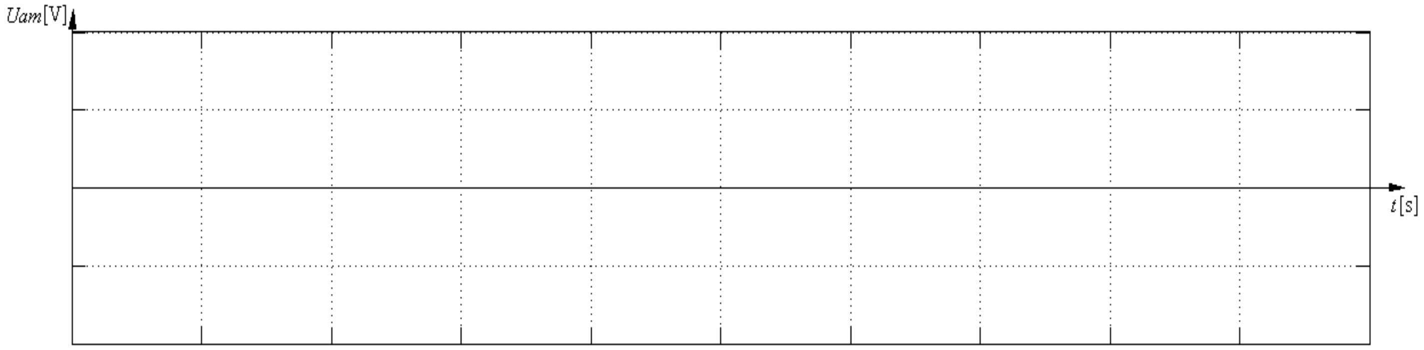


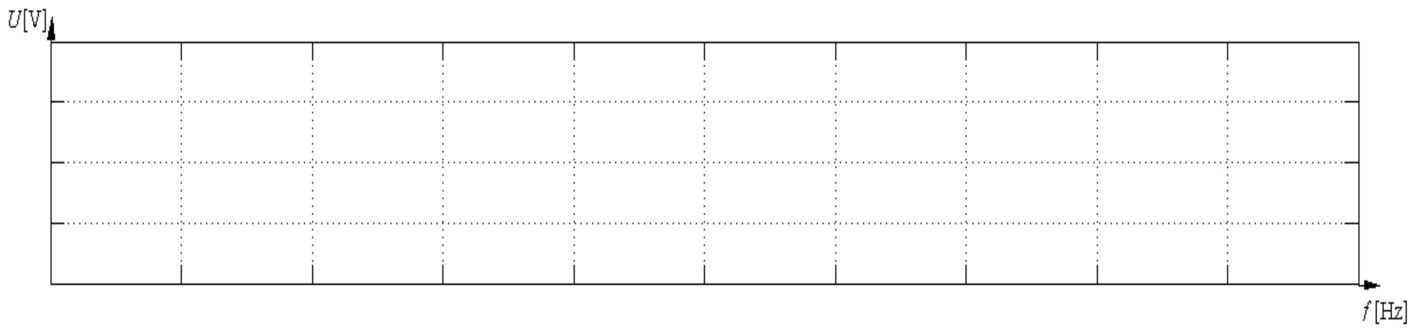




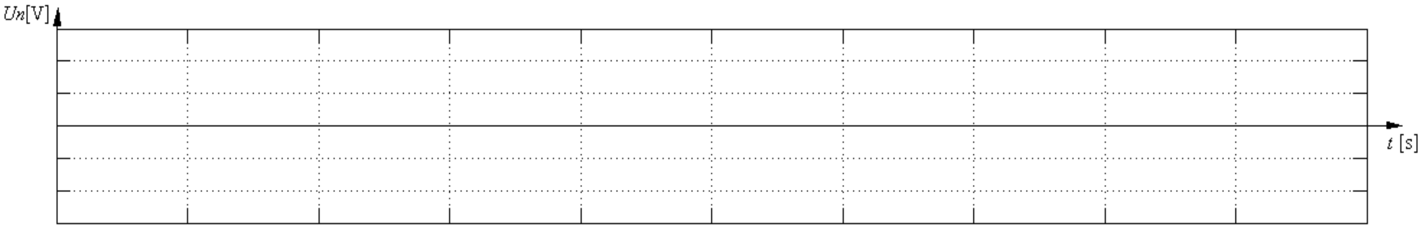
U_n [V]	f_n [kHz]	U_s [V]	f_s [kHz]	m

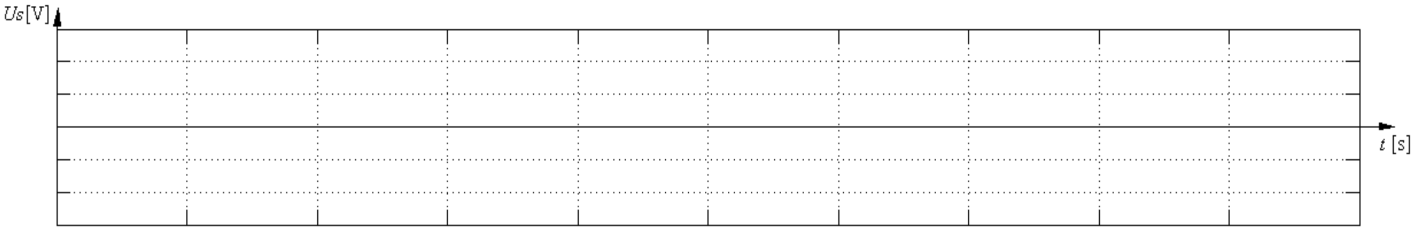


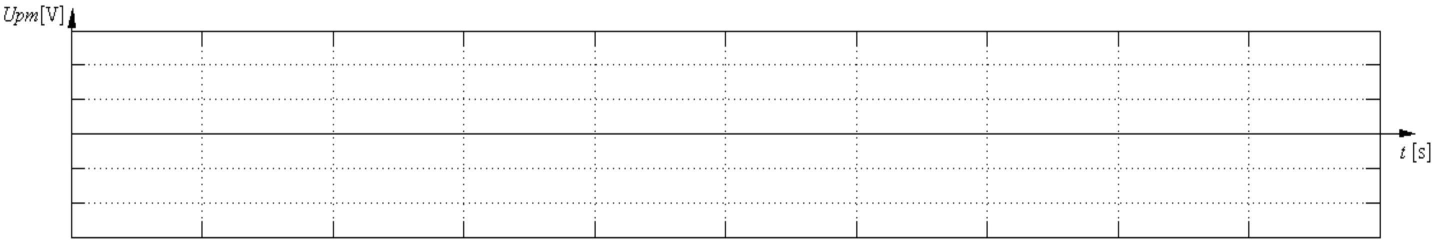


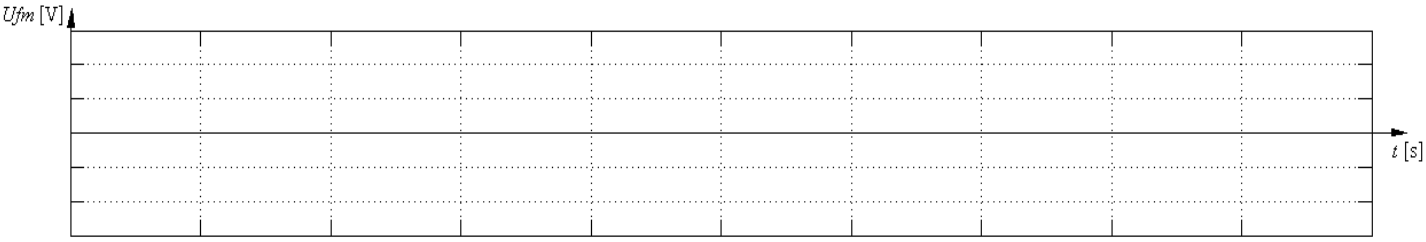


U_n [V]	f_n [kHz]	U_s [V]	f_s [kHz]	Shape of the modulating signal
d_{ff}	$\Delta\varphi$ [rad]	d_{fc}	ΔF [rad/s]	

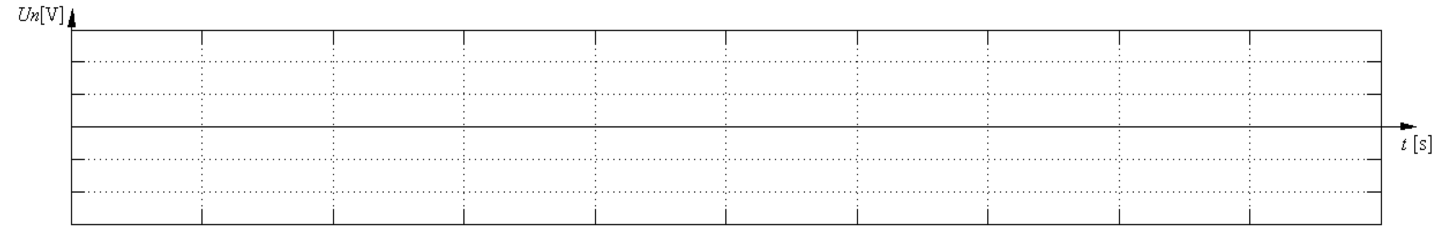


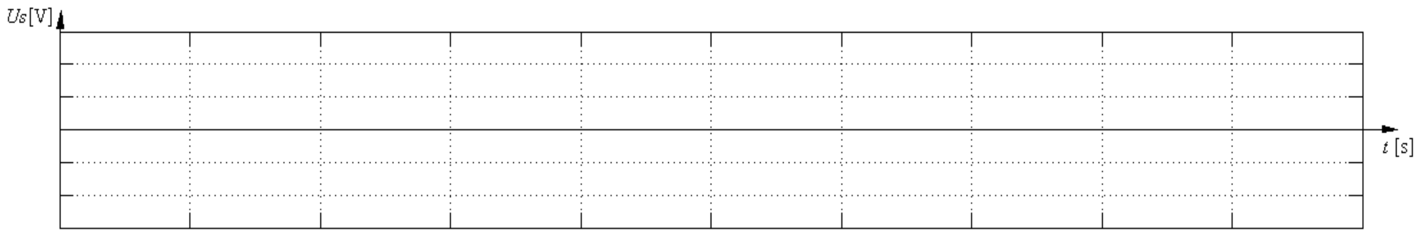


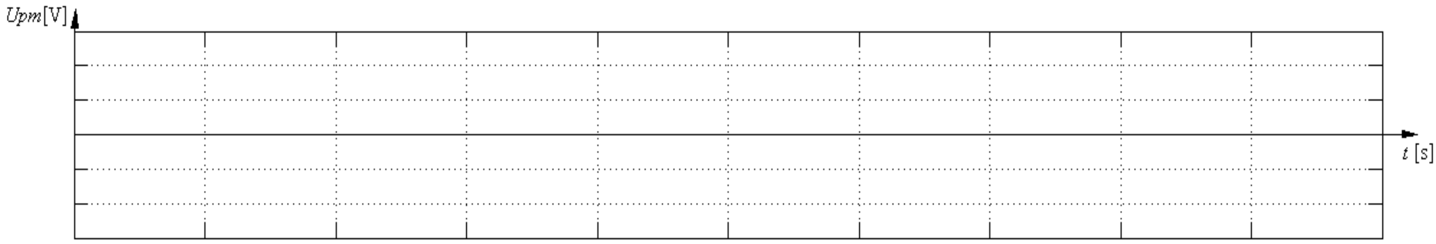


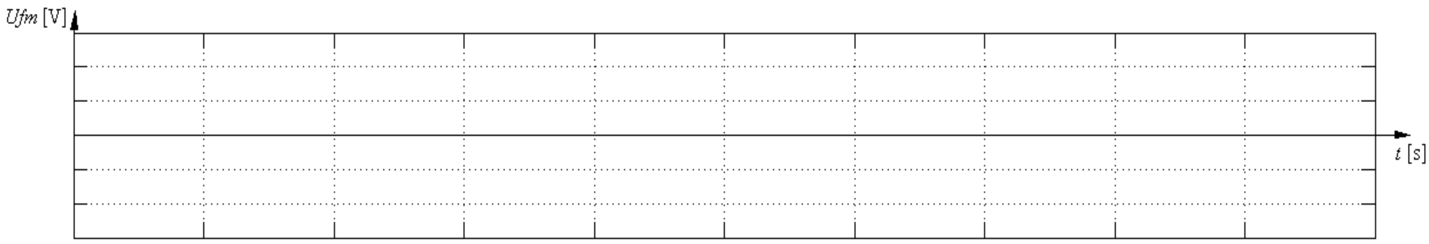


U_n [V]	f_n [kHz]	U_s [V]	f_s [kHz]	**Shape of the modulating signal**
d_{ff}	$\Delta\varphi$ [rad]	d_{fc}	ΔF [rad/s]	









U_n [V]	f_n [kHz]	U_s [V]	f_s [kHz]	Shape of the modulating signal

d_{ff}	$\Delta\varphi$ [rad]	d_{fc}	ΔF [rad/s]