AN-NAJAH NATIONAL UNIVERSITY

FACULTY OF ENGINEERING

TELECOMMUNICATION ENGINEERING DEPARTMENT

Communications Lab

10646328

Student Manual

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An-Najah National University
Faculty of Engineering and IT

جامعة النجاح الوطنية
كلية الهندسة وتكنولوجيا المعلومات

Department Name : Telecommunications Engineering Department
Course Name: Communications lab Number:10646328

Report Grading Sheet

Instructor Name:	Experiment #:
Academic Year: 2018/2019	Performed on:
Semester:	Submitted on:
Experiment Name:	
Students:	
1-	2-
3-	4-
5-	6-

Report's Outcomes

ILO _1_ =(12) %	ILO _2_ =(50) %	ILO _4_ =(38) %	ILO __ =( ) %	ILO __ =( ) %

Evaluation Criterion	Grade	Points
Abstract answers of the questions: "What did you do? How did you do it? What did you find?"		
Introduction Sufficient,Clear and complete statement of objectives.	1	
Theory Presents sufficiently the theoretical basis.		
Apparatus/ Procedure Apparatus sufficiently described to enable another experimenter to identify the equipment needed to conduct the experiment. Procedure sufficiently described.	2	
Experimental Results and Calculations Results analyzed correctly. Experimental findings adequately and specifically summarized, in graphical, tabular, and/or written form.	3	
Discussion Crisp explanation of experimental results. Comparison of theoretical predictions to experimental results, including discussion of accuracy and error analysis in some cases.	2	
Conclusions and Recommendations Conclusions summarize the major findings from the experimental results with adequate specificity. Recommendations appropriate in light of conclusions. Correct grammar.	1	
References Complete and consistent bibliographic information that would enable the reader to find the reference of interest.		
Appendices Appropriate information, organized and annotated. Includes all calculations and raw data Sheet.		
Appearance Title page is complete, page numbers applied, content is well organized, correct spelling, fonts are consistent, good visual appeal.	1	
Total	10	

Table of Contents

Department of Telecommunication Engineering

Communication Lab (10646328)

Total Credits	**1**

major compulsory

Prerequisites	P1 : Communication principles (10646322)

Course Contents

This course contains experiments that cover analogue and digital communication systems. At the beginning the students are introduced to different lab devices and software tools. Students will start with the analogue part where they will learn how to generate and demodulate different types of amplitude and frequency Modulations. Later in this lab, students will experiment digital communication operations such as Sampling, Quantization, Pulse Code Modulation, Pulse Time Modulation, Delta Modulation, data formats, Amplitude Shift Keying, Frequency Shift Keying and Phase Shift Keying.

	Intended Learning Outcomes (ILO's)	**Student Outcomes (SO's)**	**Contribution**
1	An ability to apply knowledge of communication theory and equations practically	B	30 %
2	An ability to design electronic component related to communication	C	40 %
3	Ability to simulate communication experiment using software tools	K	15 %
4	Ability to function in teams.	D	15 %

Textbook and/ or References

Communication Lab Manual.

Assessment Criteria	**Percent (%)**
Laboratory Work	70 %
Final Exam	30 %

Course Plan

Week	**Topic**
1	Introduction to communication lab (hardware and software tools)
2	Amplitude Modulation
3	AM Demodulation and Single Sideband AM
4	Frequency Modulation
5	Frequency Demodulation and Phase Modulation
6	Pulse Amplitude Modulation (PAM)
7	Pulse Code Modulation (PCM)
8	Delta Modulation
9	Pulse Time Modulation
10	Pulse Time Demodulation
11	Data Format, ASK & FSK
12	2PSK & 4PSK
13	Practical Exam
14	Theoretical Exam

Lab Safety Guidelines

1) Be familiar with the electrical hazards associated with your workplace.

2) You may enter the laboratory only when authorized to do so and only during authorized hours of operation.

3) Be as careful for the safety of others as for yourself. Think before you act, be tidy and systematic.

4) Avoid bulky, loose or trailing clothes. Avoid long loose hair.

5) Food, beverages and other substances are strictly prohibited in the laboratory at all times. Avoid working with wet hands and clothing.

6) Use extension cords only when necessary and only on a temporary basis.

7) Request new outlets if your work requires equipment in an area without an outlet.

8) Discard damaged cords, cords that become hot, or cords with exposed wiring.

9) Before equipment is energized ensure, (1) circuit connections and layout have been checked by a laboratory technician and (2) all colleagues in your group give their assent.

10) Know the correct handling, storage and disposal procedures for batteries, cells, capacitors, inductors and other high energy-storage devices.

11) Experiments left unattended should be isolated from the power supplies. If for a special reason, it must be left on, a barrier and a warning notice are required.

12) Equipment found to be faulty in any way should be reported to the laboratory technician immediately and taken out of service until inspected and declared safe.

13) Never make any changes to circuits or mechanical layout without first isolating the circuit by switching off and removing connections to power supplies.

14) Know what you must do in an emergency, i.e. Emergency Power Off

15) For microwave and antenna trainer:

   a. You should, whenever possible, remove the power from the gun oscillator before placing yourself in front of transmitting antenna.
   b. For your safety, do not look directly into the waveguides or horn antennas while power is being supplied by the gun oscillator. Because, although the microwave is invisible, it can be dangerous at high levels or long exposure times.

16) For fiber optics trainer:

   a. Do not look inside the connector of the Optical Sources when these are operating. Although nothing can be seen, as the emitted wavelength should be out of the visible range, it can be dangerous for your sight.

b. Do not bend the optical cables with too narrow curves, as the fiber inside should cut off or damage. The minimum curving ray is around 2 cm;

c. Sometimes clean the connectors' head with a cotton wad soaked with alcohol;

Electrical Emergency Response

The following instructions provide guidelines for handling two types of electrical emergencies:

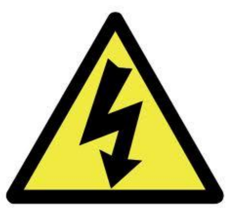
1. *Electric Shock*:

When someone suffers serious electrical shock, he or she may be knocked unconscious. If the victim is still in contact with the electrical current, immediately turn off the electrical power source. If you cannot disconnect the power source, depress the Emergency Power Off switch.

IMPORTANT:

Do not touch a victim that is still in contact with a live power source; you could be electrocuted.

Have someone call for emergency medical assistance immediately. Administer first-aid, as appropriate.



2. *Electrical Fire:*

If an electrical fire occurs, try to disconnect the electrical power source, if possible. If the fire is small and you are not in immediate danger; and you have been properly trained in fighting fires, use the correct type of fire extinguisher to extinguish the fire. When in doubt, push in the Emergency Power Off button.

NEVER use water to extinguish an electrical fire.

Lab Report Format

Following the completion of each laboratory exercise, a report must be written and submitted for grading. The purpose of the report is to completely document the activities of the design and demonstration in the laboratory. Reports should be complete in the sense that all information required to reproduce the experiment is contained within. Writing useful reports is a very essential part of becoming an engineer. In both academic and industrial environments, reports are the primary means of communication between engineers.

There is no one best format for all technical reports but there are a few simple rules concerning technical presentations which should be followed. Adapted to this laboratory they may be summarized in the following recommended report format:

- ABET Cover Page
- Title page
- Introduction
- Experimental Procedure
- Experimental Data
- Discussion
- Conclusions

Detailed descriptions of these items are given below.

Title Page:

The title page should contain the following information

- Your name
- ID
- Experiment number and title
- Date submitted
- Instructors Name

Introduction:

It should contain a brief statement in which you state the objectives, or goals of the experiment. It should also help guide the reader through the report by stating, for example, that experiments were done with three different circuits or consisted of two parts etc. Or that additional calculations or data sheets can be found in the appendix, or at the end of the report.

The Procedure

It describes the experimental setup and how the measurements were made. Include here circuit schematics with the values of components. Mention instruments used and describe any special measurement procedure that was used.

Results/Questions:

This section of the report should be used to answer any questions presented in the lab hand-out. Any tables and /or circuit diagrams representing results of the experiment should be referred to and discussed / explained with detail. All questions should be answered very clearly in paragraph form. Any unanswered questions from the lab hand-out will result in loss of points on the report.

The best form of presentation of some of the data is graphical. In engineering presentations a figure is often worth more than a thousand words. Some simple rules concerning graphs and figures which should always be followed. If there is more than one figure in the report, the figures should be numbered. Each figure must have a caption following the number. For example, *"Figure 1.1:DSB-SC "* In addition, it will greatly help you to learn how to use headers and figures in MS Word.

The Discussion

It is a critical part of the report which testifies to the student's understanding of the experiments and its purpose. In this part of the report you should compare the expected outcome of the experiment, such as derived from theory or computer simulation, with the measured value. Before you can make such comparison you may have to do some data analysis or manipulation.

When comparing experimental data with numbers obtained from theory or simulation, make very clear which is which. It does not necessarily mean that your experiment was a failure. The results will be accepted, provided that you can account for the discrepancy. Your ability to read the scales may be one limitation. The value of some circuit components may not be well known and a nominal value given by the manufacturer does not always correspond to reality. Very often, however, the reason for the difference between the expected and measured values lies in the experimental procedure or in not taking into account all factors that enter into analysis.

Conclusion:

A brief conclusion summarizing the work done, theory applied, and the results of the completed work should be included here. Data and analyses are not appropriate for the conclusion.

Notes

Typed Reports are required.   Any drawings done by hand must be done with neatness, using a straightedge and drawing guides wherever possible.
Freehand drawings will not be accepted.

Telecommunication Department
Communications Lab
EXP. 1 Amplitude Modulation

1. Introduction

Signals

In electrical telecommunications engineering, messages are usually in the form of time-dependent electrical quantities, for example, voltage *u(t)* or current *i*(*t*). These kinds of quantities which are described by time functions are called signals. In order to transmit messages a parameter of the electrical signal must be suitably influenced. In cases where a signal defined as a time function is known and the signal value can be determined exactly at any given point in time, then the signal is called deterministic. Examples of **deterministic signals** are:

- 1. Harmonic oscillation

$$u(t) = A \cdot \sin(2\pi ft + \phi) \qquad 1.1$$

- Symmetrical square wave

$$u(t) = u(t + nT) \qquad n = 1, 2, 3... \qquad 1.2$$

$$u(t) = \begin{cases} A \text{ for } 0 < t < T/2 \\ 0 \text{ for } T/2 < t < T. \end{cases}$$

Deterministic telecommunications is useless from the point of view of information theory. Only unknown, i.e. unpredictable messages are important for the message receiver. Nevertheless, when discussing modulation methods it is standard procedure to work with harmonic signals. The results which can be obtained are then clearer and more straightforward. If the signal value for any given point in time cannot be given because the signal curve appears totally erratic, then the signal is called stochastic. An example for a stochastic signal is noise. **Stochastic signals** can be described using methods of probability mathematics, but they will not be taken into consideration here. Signals are distinguished according to the characteristic curves of their time and signal coordinates. If the signal function s(*t*) produces a signal value at any random point in time, the signal function is called **time-continuous** (continuous w.r.t. time). In contrast, if the signal values differ from 0 only at definite, countable points in time, i.e. its time characteristic shows "gaps", then this is referred to as **time-discrete** (discrete w.r.t. time). What is true for the time coordinate, can also be applied to the signal coordinates. Accordingly, a signal is called **level-continuous**, if it can assume any given value within the system limits.

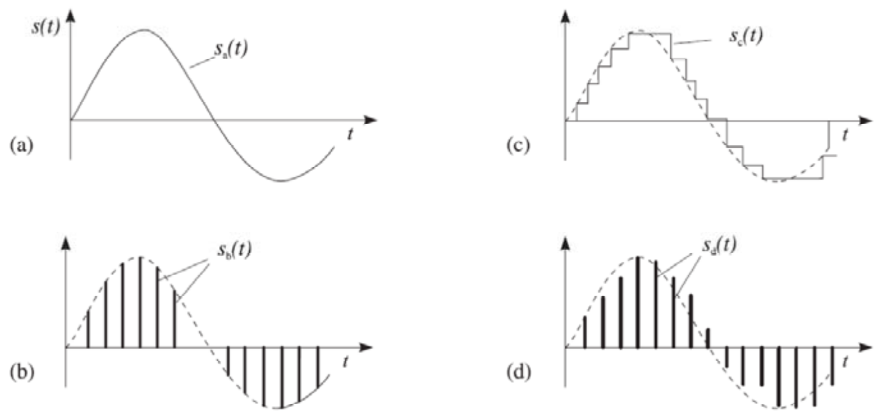


Fig. 1.1: Classification of signals (a) time and level-continuous, (b) time–discrete (samplled), level-continuous, (c) time-continuous, level-discrete (quantized), (d) time- and level-discrete

It is called **value-discrete or n-level**, if only a finite number of signal values are permitted. Two important signal classes can be defined using these 4 terms:

Analog signals
A signal is called analog if it is both time as well as value-continuous.

Digital signals
A signal is called digital, if it is both time as well as value-discrete. Fig. 1.1 shows the various kinds of signals.

Time and spectral domain
In the technical sciences there exists, in addition to the “time domain”, signal representation in the “frequency” or “spectral domain”. The equivalence of the two types of representation can be seen in the depictions in Fig. 1.2.

If you first consider the harmonic function as specified in the equation 1.1, then a display on time domain results in the familiar, dynamic (time) characteristic according to Fig. 1.2.a. The sinusoidal time function is described by the amplitude A and the period duration T. However, a totally equivalent representation of this function is reproduced when the variables A and $f = 1/T$ are used instead of the parameters A and T. If the amplitude is displayed on the frequency axis, then this form of representation is called the **amplitude spectrum**.

Thus, a single line can depict a harmonic function. Now, after Fourier, every **non-harmonic**, **periodic function** can be represented as the superimposition of harmonic oscillations with fixed amplitudes $\boldsymbol{S(n)}$. As an example Fig. 1.2.b presents a symmetrical square-wave signal with the amplitude A_R and the period of oscillation T_R. We can see from the corresponding amplitude spectrum $\boldsymbol{S_R(n)}$ in Fig. 1.2.d that the square-wave function is produced from the superposition of many (an infinite number of) harmonic oscillations. Their frequencies are odd numbered multiples of $f = 1/T_R$ and their amplitudes decrease as a function of the ordinal number n, see Table 1.1.

Note: Any precise and comprehensive discussion of the spectra not only takes the amplitude spectrum $\boldsymbol{S(n)}$ into consideration but also the phase spectrum $\boldsymbol{\varphi(n)}$. However, in many practical exercises it suffices to determine the amplitude spectrum.

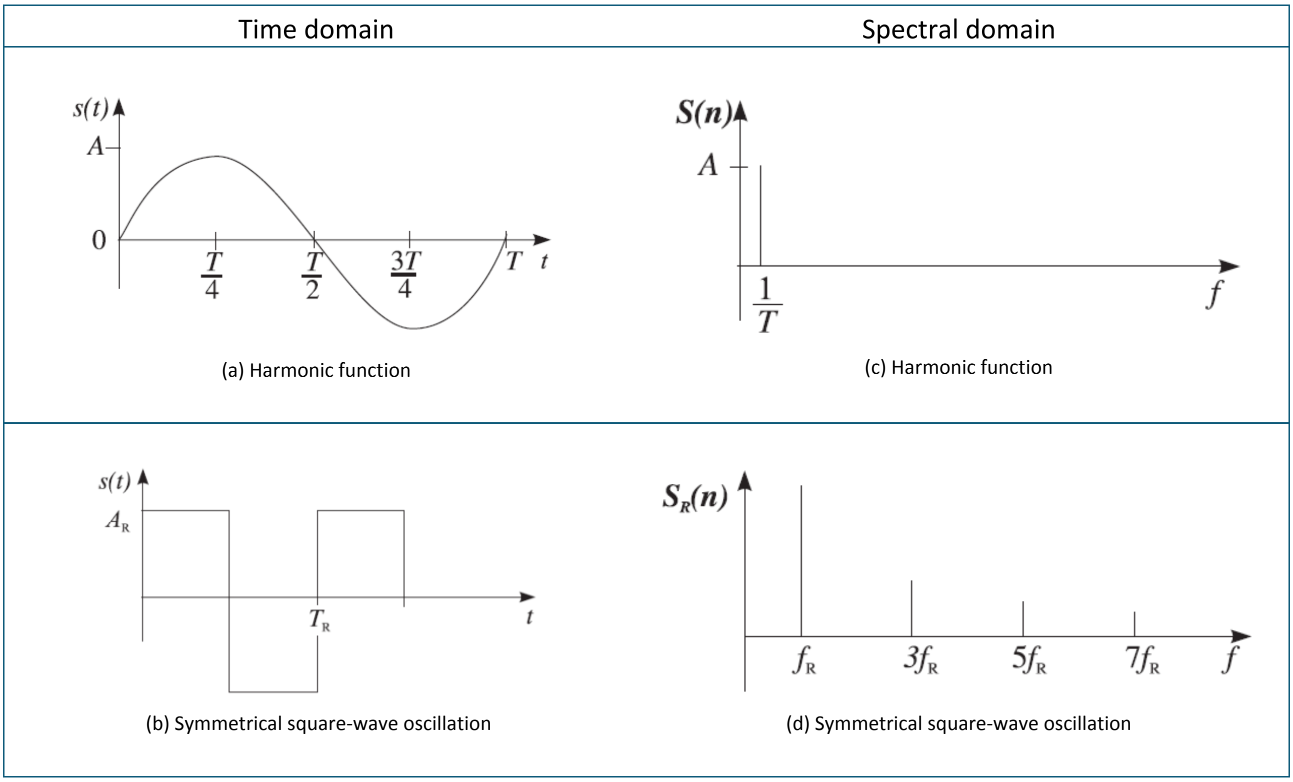


Fig. 1.2: Time and spectral representation

Harmonic	Frequency	Amplitude
1	$f_R = 1/T_R$	$S_R(1) = \frac{4}{\pi} A_R$
2	$3\,f_R$	$\frac{S_R(1)}{3}$
3	$5\,f_R$	$\frac{S_R(1)}{5}$
4	$7\,f_R$	$\frac{S_R(1)}{7}$
n	$(2\,n-1)\,f_R$	$\frac{S_R(1)}{2n-1}$

Table 1.1: Amplitude spectrum of a symmetrical square-wave signal n = 1,2,3,4, …

Modulation

When speaking of modulation, one generally refers to the conversion of a modulation signal $s_M(t)$ into a time function with altered characteristics using a carrier signal. The message signal influences a parameter of the carrier in a suitable fashion. Either harmonic oscillations or pulse trains are used as carrier signals. If, for example, a harmonic carrier is used with the form:

$$s_C(t) = A_C \cos(2 f_C t + \phi) \qquad 1.3$$

then the message signal $s_M(t)$ can have an effect either on the amplitude A_C, the carrier frequency f_C or the zero phase angle φ. These effects result in the **analog modulation methods**:

- Amplitude modulations (AM)
- Frequency modulation (FM)
- Phase modulation (PM)

In the case of analog modulation methods, the modulation process means a continuous conversion of the modulating signal $s_M(t)$ into a higher frequency band (frequency conversion). The modulating signal is shifted from the **baseband** (AF range, **original frequency band**), into an RF frequency band. It no longer appears in the spectrum of the modulated oscillation. A modulation always requires that the carrier and the modulation signal interact. Both of these signals are fed into a **modulator**. The original signal $s_M(t)$ is recovered from the modulated signal through demodulation. Consequently, modulation and demodulation are mutually related, inverse processes. The complexities involved in modulation and demodulation are considerable. The following reasons explain why modulation is worthwhile:

1. Modulation enables the matching of the modulating signal to the characteristics of the transmission channel (radio links e.g. are only possible above a certain frequency.).
2. Existing transmission channels can be multiply exploited using modulation, (frequency or time division multiplex systems).
3. Improved signal-to-noise ratios can be obtained using modulation.

The communication system according to Shannon

Electrical communications engineering is divided into three classical subfunctions:

1. **Transmission of the message**
2. **Processing of the message**
3. **Relaying the message** (telephone technology)

If only a single transmission channel is examined, (i.e. no telephone technology), then we can concentrate on the remaining functions illustrated by the scheme in Fig. 1.3.

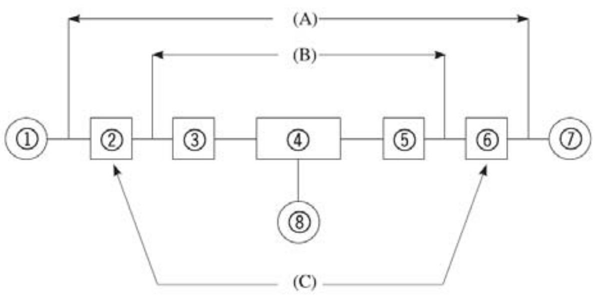


Fig. 1.3: The communications system. (A) The communications system, (B) Message transmission, (C) Message processing

1. Message source (human being, measurement sensor, etc.)
2. Converter (microphone, television camera, stran gauges, thermo sensor, etc.)
3. Tramsmiter
4. Transmission channel (radio link, transmission cable, data storage system)
5. Receiver
6. Converter
7. Message recipient
8. Interference source

The **telecommunication system** (A) consists of equipment used for message transmission (B) and message processing (C). The message source (1) generates the information, which is to be made available to the message recipient (7). The signals generated are of the most varied physical nature, e.g. sound, light, pressure, temperature, etc. It is the function of the converter (2) to convert the non-electrical signal of the source into an electrical one. The transmitter (3) converts the converter signal into one better suited for transmission via the channel. Thus the modulation process takes place in (3). The transmission channel (4) serves either to bridge a spatial distance, or to overcome a period of time. The modulated signal, generally distorted by the interference source (8), reaches the receiver (5), where it is then reconverted into its original electrical signal there (demodulation). Finally, the converter (6) transforms the electrical signal back into the physical signal required by the message recipient (7). The message recipient can take the form of the human being with eyes and ears or a machine in a process control loop.

2. Review of amplitude modulation

In amplitude modulation (AM) the momentary value of the message signal $s_M(t)$ has an immediate effect on the amplitude of the carrier oscillation $s_C(t)$. This takes place in a modulator, see Fig. 2.1.

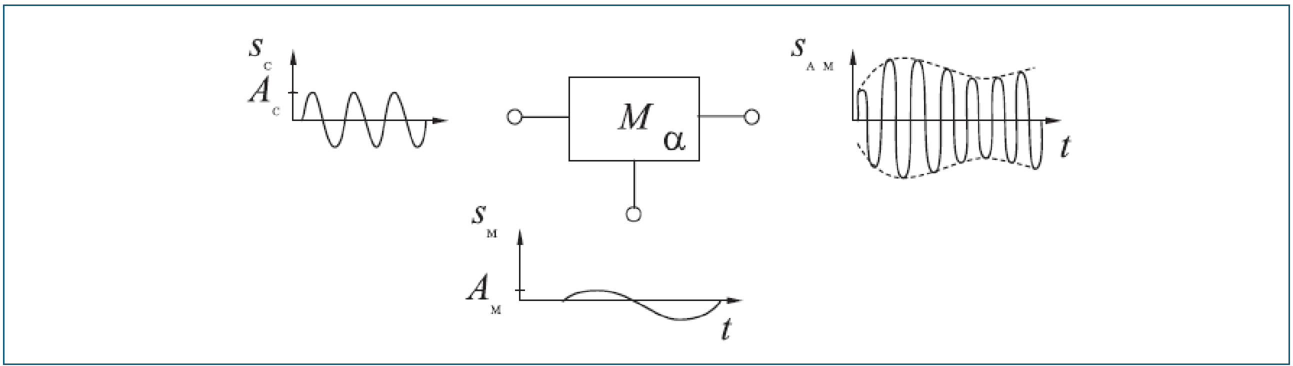


Fig. 2.1: Generation of amplitude modulation

Here it would be:

$s_C(t) = A_C \cos(2\pi f_C t)$ 2.1

for the high-frequency carrier and:

$s_M(t) = A_M \cos(2\pi f_M t)$ 2.2

for the low frequency message signal. The combining of the carrier and message signal in the modulator then provides the following modulation product:

$s_{AM}(t) = [A_C + \alpha\, s_M(t)] \cos(2\pi f_C t)$ 2.3

$= [A_C + \alpha\, A_M \cos(2\pi f_M t)] \cos(2\pi f_C t)$

Where α stands for the modulator constant, which expresses the affect of the message signal $s_M(t)$ on the amplitude *AC* of the carrier. Normally the equation 2.3 is described in more general terms. For this you need the following definitions:

$\Delta A_C = \alpha\, A_M$ Amplitude deviation 2.4

$M = \Delta A_C / A_C$ Modulation index 2.5

Amplitude deviation ΔA_C describes the maximum change away from the original value A_C in the carrier amplitude. The modulation index *m* reproduces the ratio of the amplitude deviation to the carrier amplitude. Thus it is possible to convert (2.3) as follows:

$$s_{AM}(t) = A_C\left[1+\frac{\Delta A_C}{A_C}\cos(2\pi f_M t)\right]\cos(2\pi f_C t)$$

$$= A_C\left[1+m\cos(2\pi f_M t)\right]\cos(2\pi f_C t) \qquad 2.6$$

Fig. 2.2 shows the amplitude modulated signal according to (2.6).

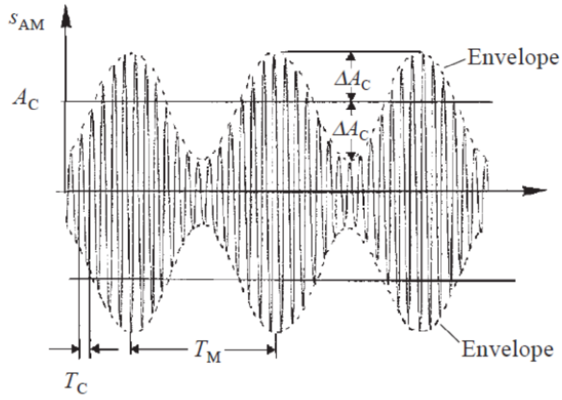


Fig. 2.2: Amplitude modulated signal

The modulating signal $s_M(t)$ can be recognized in the envelope curve. Normally the following holds true: $0 < m < 1$. The following limiting cases for m are interesting:

$m = 0$: no modulation effect
$m = 1$: full modulation, the envelopes bordering the modulating signal just touch at their minimum values
$m > 1$: overmodulation, the envelopes permeate each other, modulation distortion arises.

Special cases are depicted in Fig. 2.3.

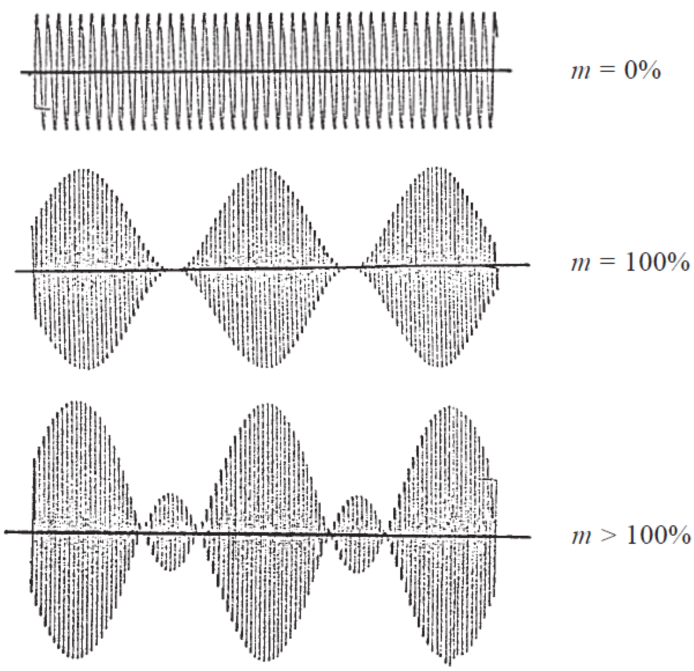


Fig. 2.3: Limiting cases for the modulation factor

The spectrum of amplitude modulation

The expression in the brackets of (2.6) describes the envelope of amplitude modulation. If you multiply the dynamic (time) characteristic of the carrier oscillation in this expression, you obtain:

$$s_{AM}(t) = A_C\,[\cos(2\pi f_C t) + m\cos(2\pi f_T t)\cos(2\pi f_M t)] \qquad 2.7$$

The application of the addition theorem:

$$\cos x \cdot \cos y = 1/2\,[\cos(x - y) + \cos(x + y)]$$

provides:

$$s_{AM}(t) = A_C \cos(2\pi f_C t) + \frac{m}{2}\cos[2\pi(f_C - f_M)t] + \frac{m}{2}\cos[2\pi(f_C + f_M)t] \qquad 2.8$$

From (2.8) you can see the spectral composition of amplitude modulation, see Fig. 2.4.

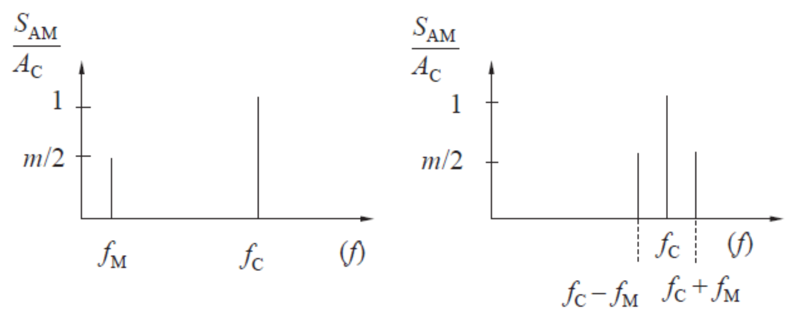


Fig. 2.4: The spectrum of amplitude modulation

From the spectrum we can see that besides the carrier oscillation with the frequency f_C, there are also 2 side oscillations with the frequencies $f_C + f_M$ and $f_C - f_M$ contained in $s_{AM}(t)$. According to (2.8) the amplitudes of the equal side oscillations depend on the modulation index. The oscillation with the lower frequency $f_C - f_M$ is called the lower sideline, the one with the higher frequency is the upper sideline $f_C + f_M$. The lower sideline (*LSL*) slips further into the range of lower frequencies as the signal frequency f_M increases. This frequency response of the *LSL* is referred to as **inverted position**. The upper sideline (*USL*) shifts into the higher range of frequencies with increasing signal frequency. It lies in the **normal position**.

In Fig. 2.5 the terms are in standard representation for the transmission of an information band, which extends from a lower frequency limit f_u up to the upper frequency limit f_o. The representation according to Fig. 2.5 is standard particularly in carrier frequency technology. The bandwidth requirement of AM equals twice the maximum message frequency f_{Mmax}:

$$b = 2\,f_{Mmax} \qquad 2.9$$

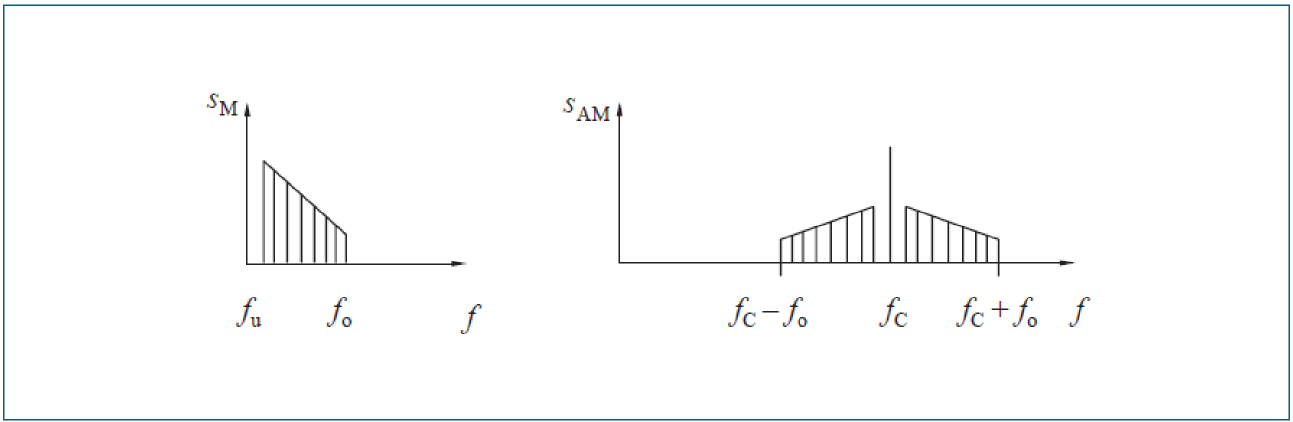


Fig. 2.5: Normal and inverted position

Questions

1.What is meant by modulation? Mixing?

2.Name the reasons for performing modulation?

3.In DSB the carrier's peak values are affected by the instantaneous value of the message signal, but the spectrum shows that the carrier amplitude remains constant! How do you explain the apparent contradiction?

4.Define amplitude deviation and the modulation index?

5.In radio links the carrier is normally attenuated to 5%...10%. What advantages does this have compared to transmission with 100% carrier amplitude? Why isn't the carrier completely suppressed?

6.How high is the maximum efficiency in DSB? How can the efficiency be increased?

7.Which methods of carrier suppression are there?

3. Double Sideband AM

The features of the DSB are summarized in Fig. 3.1.

$$s_{DSB}(t) = [A_C + \alpha s_M(t)] \cos(2\pi f_C t) \qquad 3.1$$

Demodulation methods : Envelope demodulation
: Synchronous demodulation
Bandwidth : $b = 2 \cdot f_{Mmax}$ 3.2
Application : Radio transmission

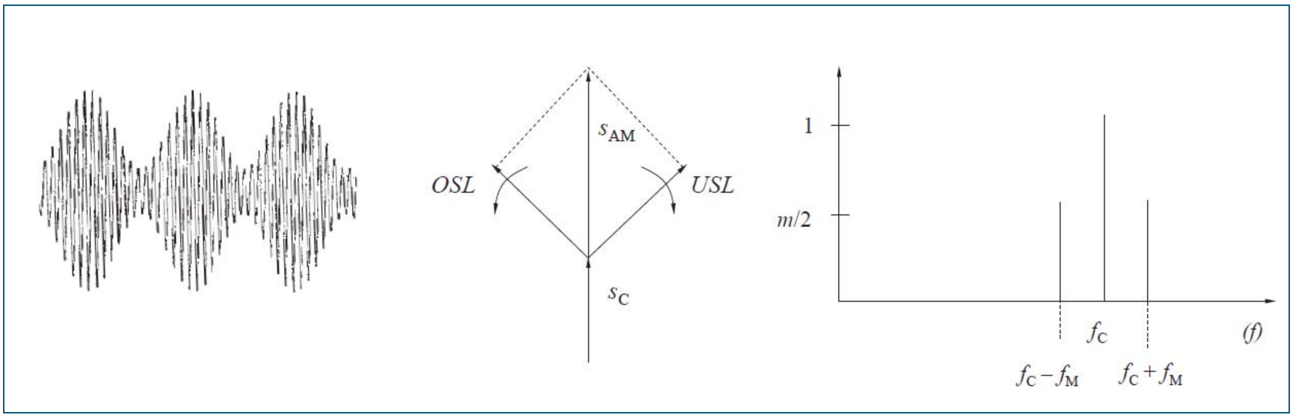


Fig. 3.1: Representation of the DSB

The DSB$_{SC}$

If in the equation 3.1 the constant component A_C inside the brackets is suppressed, then we obtain:

$$\begin{aligned} s_{DSBSC} &= \alpha s_M(t)\cos(2\pi f_C t) \\ &= \alpha A_M \cos(2\pi f_M t)\cos(2\pi f_C t) \\ &= \frac{\alpha A_M}{2}\cos[2\pi(f_C - f_M)t] \\ &\quad + \frac{\alpha A_M}{2}\cos[2\pi(f_C + f_M)t] \end{aligned} \qquad 3.3$$

The DSB_{SC} consists of the superimposition of 2 harmonic oscillations, whose frequencies $f_C + f_M$, resp. $f_C - f_M$ are in direct proximity due to the fact that $f_C >> f_M$. Therefore, the dynamic characteristic of the DSB_{sc} is a beat. Here the side oscillations arise on account of the **frequency conversion** from f_M to $f_C - f_M$ resp. $f_C + f_M$. Since there is no carrier, the modulation depth *m* cannot be defined. Overmodulation is not possible. The amplitude of the modulation product $s_{DSBSC}(t)$ is directly proportional to the instantaneous value of the modulating signal $s_M(t)$. The upper and lower envelope curves have the abcissa as a joint reference line, instead of the positive or negative carrier amplitude. The features of the DSB_{SC} are summarized in Fig. 3.2.

Clear to be seen in the dynamic characteristic is the abrupt phase change of 180° at the zero crossover of the envelope curve. The envelope curve contains sinusoidal halfwaves of double the signal frequency.

Demodulation methods	: Synchronous demodulation	
Bandwidth	: $b = 2 \cdot f_{Mmax}$	3.4
Application	: Radio transmission	

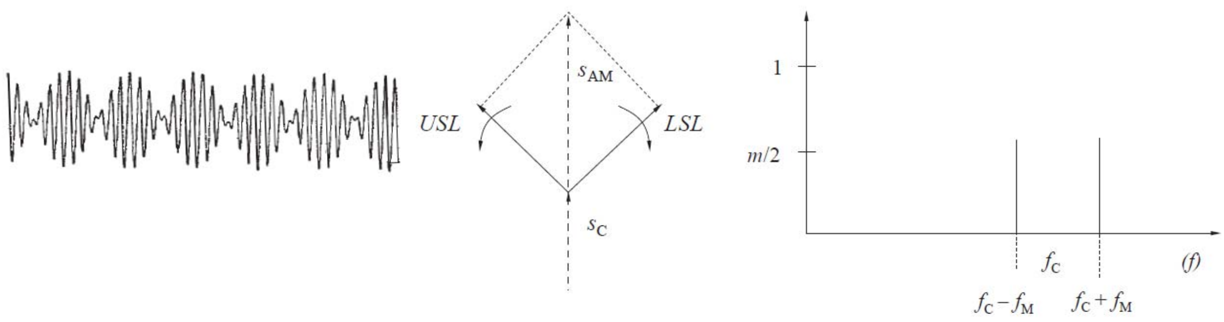


Fig. 3.2: Representation of the DSB_{sc}

3.1 Investigations on the dynamic characteristic of the DSB

Required accessories

To carry out the experiments the following accessories are required:

Qty.	Cat.-no.	Designation
1	736 201	CF-Transmitter 20 kHz
		Accessories
1	726 09	Panel Frame-T130, two Level
1	726 86	DC-Power Supply ± 15 V/3 A
1	726 961	Function Generator 200 kHz
1	501 46	Pair cables 100 cm, red/blue
2	501 461	Pair cables 100 cm, black
1	501 511	Set of 10 Bridging plugs, black
		Measuring instrument
1	524 013S	Sensor-CASSY 2 - Starter

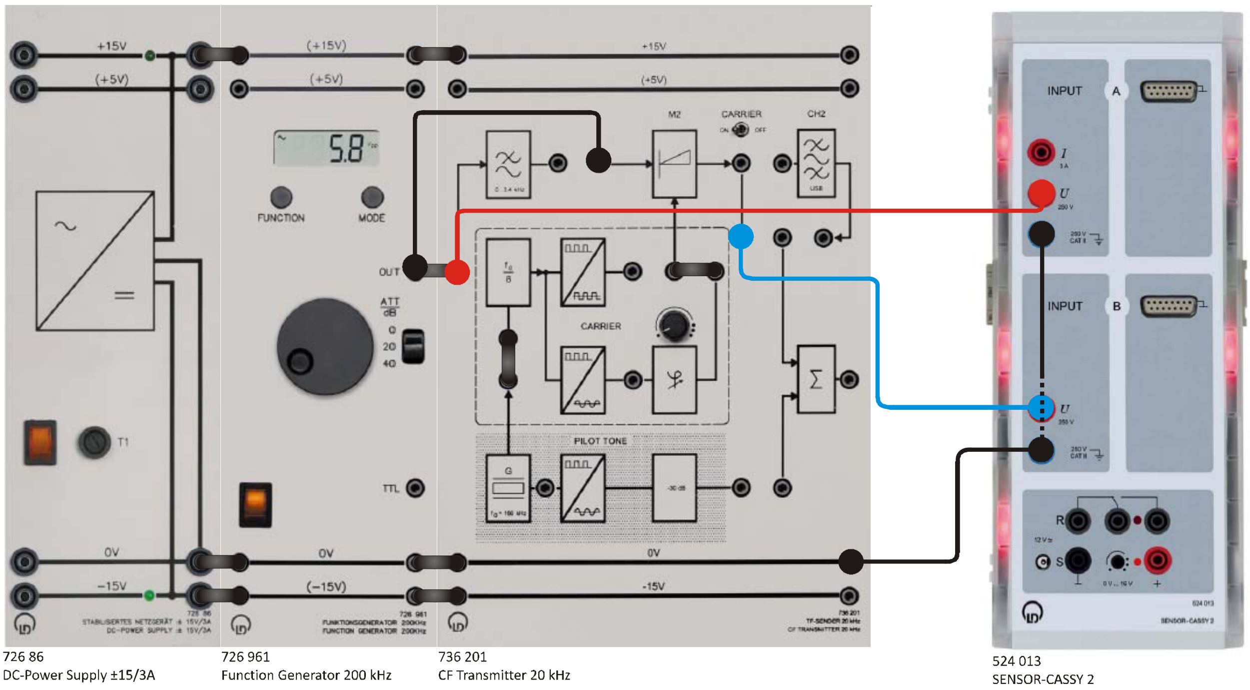
Part 1: Software Tool

Use the formula function in the CASSY LAB 2 software to draw the AM_DSB_LC /AM_DSB_SC.

Part 2: Procedure

Experiment set-up

Assemble the components as shown below. Set the function generator to: sine, A_M = 2 V and f_M = 2 kHz.



Experiment set-up: **Double Sideband**

DSB

- Set toggle switch to **CARRIER ON** setting.
- Start the measurement by pressing *F9*.
- Display the output signal of the modulator M2 on CASSY lab 2 (this signal is called the modulation product SAM(t)) and the modulating signal sM(t) of the function generator (Modulation product on channel B, modulating signal on channel A of Sensor - CASSY 2 - Starter). Shift the AF signal (modulating signal) to the upper or lower envelope curve of the AM signal. For this purpose see the CASSY lab 2 settings.
- Vary the frequency f_M and the amplitude A_M of the modulating signal of the frequency generator. What do you observe?

Diagram 3.1-1: Dynamic (time) characteristic of the DSB signal, Modulating signal $s_M(t)$, Modulation product at the output M2.

The envelope curve of the AM signal nearly coincides completely with the modulating signal and immediately follows its changes in frequency and amplitude.

Trigger to the modulating signal $s_M(t)$. Reduce the AM signal of function generator to approx. 1 V. Determine the modulation depth m. The following applies for the modulation depth m:

$$m = \frac{\Delta A_C}{A_C} = \frac{D-d}{D+d} \qquad 3.5$$

Where:
D: Peak-to-peak value of the maximum of the *AM* signal
d: Peak-to-peak value of the minimum of the *AM* signal.

Results

D =

m ≈

d =

Distortion can only be detected with difficulty when determining the modulation depth directly from the modulated signal. A better approach is to determine m from the modulation trapezoid. For this Sensor - CASSY 2 is operated in **XY modus** and the message signal $s_M(t)$ is used for horizontal deflection. The result obtained on the screen is a trapezoid which opens to the left. Set the function generator to: sine, A_M = 2 V and f_M = 2 kHz.

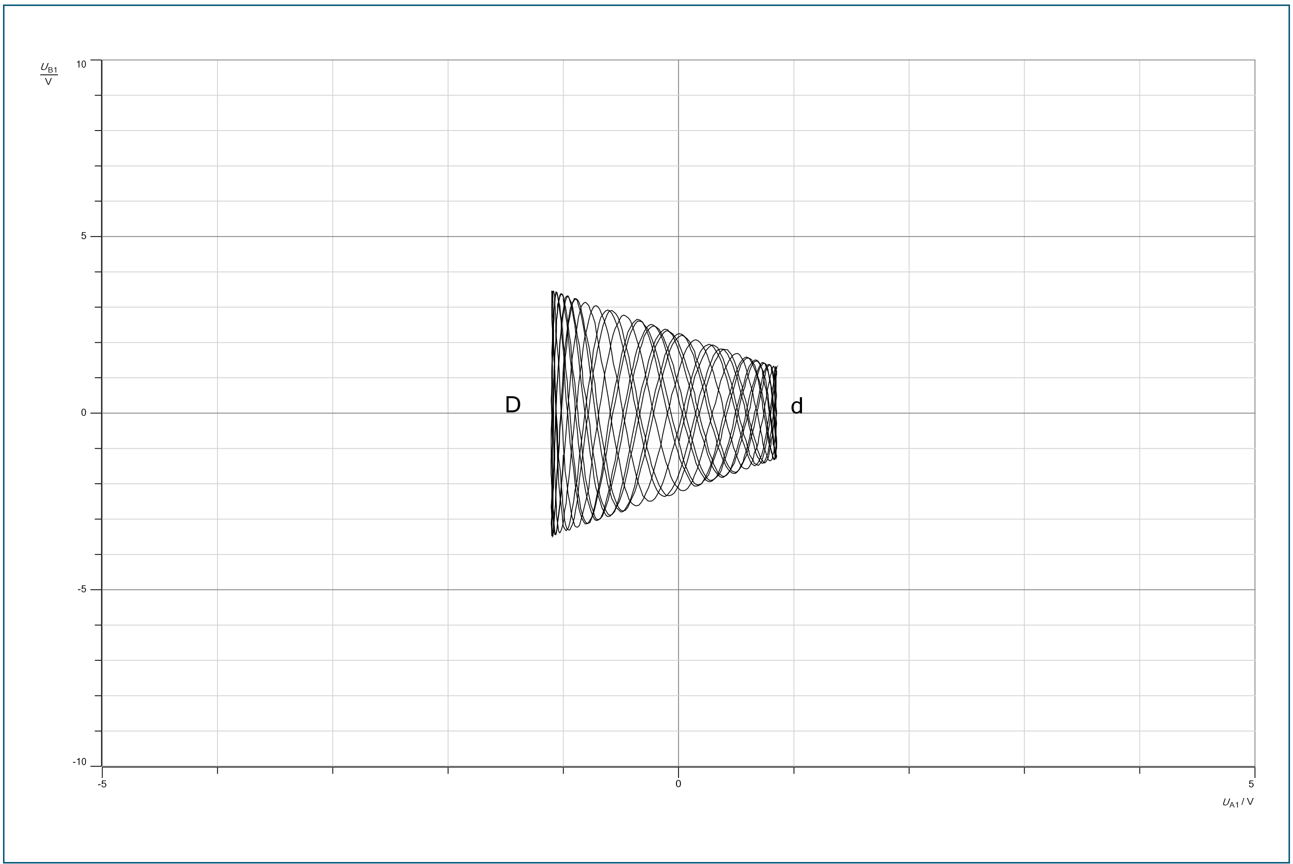


Diagram 3.1-2: The modulation trapezoid. Modulation product at the output M2

Sensor - CASSY 2 - Starter is operated in XY modus. Set the coordinate origin in the middle of the screen using the X-position and Y-position controllers. If the modulating AF signal reaches its negative maximum value, then the X-deviation is at the far left. From there it increases horizontally to the right with a rising modulation signal. At the same time the amplitude of the modulated signal drops. Consequently in XY display modus a trapezoid is produced which the its broad side on the left.

DSB_{SC}

- Set the toggle switch to **CARRIER OFF**.
- Set the function generator to: sine, A_M = 2 V and f_M = 2 kHz.
- Start the measurement by pressing *F9*. Save your result.

Proceed as described in the paragraph to "**DSB**". What is this kind of signal called? What characteristics does it have?

Diagram 3.1-3: Dynamic characteristic of the DSB_{SC} signal

The DSB_{SC} signal has the characteristic of a beat, i.e. it is the linear superpositioning of 2 harmonic oscillations with very close frequencies. The envelope curve of the beat shows zero crossings. There the beat signal experiences phase shifts of 180°. Also the DSB_{SC} signal follows the frequency changes of the AF signal without any visible phase delay. There is no overmodulation caused by amplitude changes in the modulating signal, as in the case of DSB.

3.2 Spectrum of the DSB

DSB

- Set the toggle switch to the **CARRIER ON** position.
- Use a sinusoidal signal with A_M = 2 V and f_M = 2 kHz of the function generator as the modulating signal $s_M(t)$. Feed the modulating signal into the input LP filter of the CF transmitter. Measure the AM spectrum in the range from approx. 15 kHz up to 25 kHz.
- For this Sensor - CASSY 2 - Starter is operated in ***FFT modus***. For this purpose see the CASSY lab 2 settings.
- Start the measurement by pressing F9.

Diagram 3.2-1: DSB spectrum

Repeat the experiment for $s_M(t)$: Sinusoidal, A_M = 1 V and f_M = 3 kHz. Feed the modulating signal $s_M(t)$ directly (without the LP filter) into the modulator M2 (why?). Keep the Sensor - CASSY 2 - Starter settings unchanged. Compare the results. How does the *USL* respond as a function of the signal frequency f_M? What about the *LSL*? What is the frequency response of the *LSL* and *USL*? Determine the transmission bandwidth of the *AM* signal based on the measurements. Generalize your results for a randomly taken modulating signal. Determine the modulation depth *m* from the various spectra.

Diagram 3.2-2: DSB spectrum (f = 3 kHz)

With f_M = 3 kHz the frequency of the modulating signal $s_M(t)$ already lies in the cutoff range of the LP filter. For that reason using a filter can lead to the attenuation of the amplitude at the modulator input and thus to a reduction in the modulation index. From the spectra it follows that:

– With increasing signal frequency f_M the USLs are shifted away from the carrier in the direction of higher frequencies. This frequency response of the USL is called the normal position, high signal frequencies also lie in the modulation spectrum at high frequencies.

– With increasing signal frequency fM the LSLs shift further away from the carrier into the lower frequencies. The frequency response of the LSLs is thus called the inverted position because high signal frequencies lie in the modulation spectrum at low frequencies. Basically the following applies: The upper sideband is in the normal position, the lower sideband is in the inverted position.

The modulation index m amounts to approx. 60%. Calculate the transmission bandwidths in DSB based on the spectra.

DSB_{SC}

- Set the toggle switch to **CARRIER OFF**.
- Use a sinusoidal signal with A_M = 2 V and f_M = 2 kHz of the function generator as a modulating signal.
- Measure the spectrum as described in paragraph “**DSB**”.
- Start the measurement by pressing F9.

Diagram 3.2-3: DSB_{SC} spectrum

The AM spectrum for modulation with a square-wave signal

The AM spectrum is linear. For that reason we can draw direct conclusions as to the AM spectrum based on our knowledge of the spectrum of the input signal. Now let us assume that the input signal *s(t)* consists of a frequency mix, whose spectrum ***S****(f)* is shown in the following Fig. 3.4-1

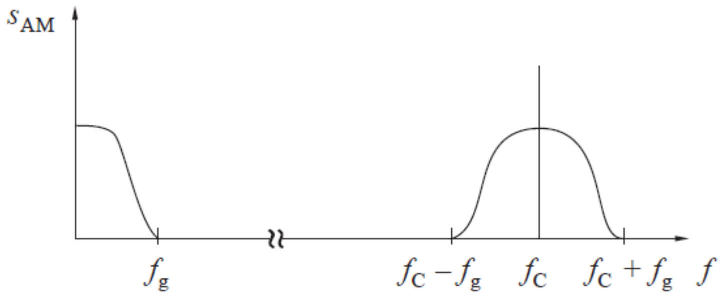


Fig. 3.2-1: AM for randomly taken spectrum of the modulating signal

What should the corresponding AM spectrum look like? Repeat the recording of the spectrum for a modulating **square-wave signal** with *A*M = 2 V and f_M = 2 kHz. Feed the square-wave signal directly of the frequency generator and the CARRIER into the modulator M2 . Explain your findings.

Diagram 3.2-4: AM spectrum for square-wave modulation

The USL1 is taken as a reference for the calculation of the spectrum. Corresponding to the known characteristic curve of the square-wave spectrum the LSL and USL have to be located symmetrically to the suppressed carrier, where the amplitudes decrease inversely to the ordinal number n. The deviations between the theory and the measurements increase with rising frequency due to the finite upper frequency cutoff of the modulator IC. Transmission bandwidth based on the spectrum b = (22.01 – 18.00) kHz ≈ 4 kHz = 2 f_M. The following is true in the general case of a modulating signal with the maximum frequency limit f_{Mmax}: b = 2 f_{Mmax}.

Part 1:AM demodulation

Envelope and synchronous demodulation

1. Envelope demodulation

First the AM signal is rectified, see Fig. 2.1.

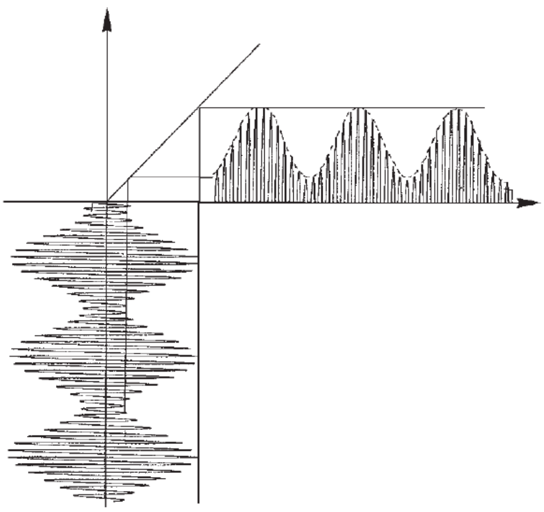


Fig. 2.1: Envelope curve demodulation

The dynamic characteristic of the current passing through the rectifier can be subjected to Fourier series expansion. It can be shown that a rectified AM signal contains the following signal components:

1. A DC voltage component
2. The original signal with the frequency f_M.
3. Components with higher frequencies f_C, $f_C + f_M$, $2 f_C + f_M$, etc.

Fourier expansion shows that rectification of the AM signal produces many new spectral components which are not present at the input of the rectifier. A suitable filter is used to suppress these unwanted spectral components. Envelope demodulation belongs to the so-called incoherent demodulation methods, as neither the carrier phase nor the carrier frequency are of any importance. Fig. 2.2 reproduces the possible circuit configuration of an envelope demodulator.

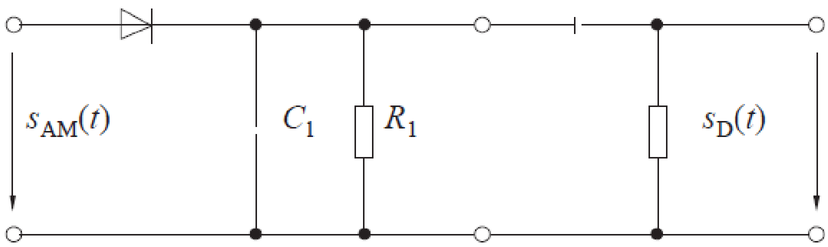


Fig. 2.2: Envelope curve demodulation

Envelope demodulation always requires the carrier. After rectification this produces a DC voltage, which establishes the working point of the diode. In practice envelope demodulation is frequently used in AM radio communications due to its simple circuitry.

2. Synchronous demodulation

In principle synchronous demodulation is simply another modulation process. To carry it out, you need an auxiliary oscillation in the receiver, which in terms of frequency and phase corresponds exactly to the carrier oscillation in the modulator. The auxiliary carrier $s_{Aux}(t)$ and the modulated signal $s_{AM}(t)$ are supplied to a circuit with multiplying capabilities, see. Fig. 2.3:

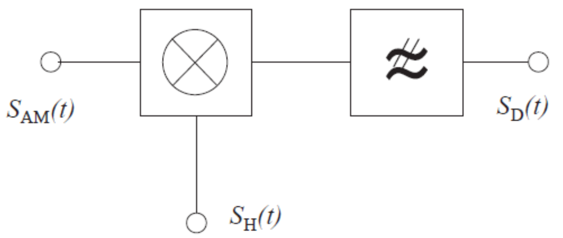


Fig. 2.3: Synchronous demodulation

Three cases can be distinguished *DSB*, *DSB*$_{sc}$ and *SSB*:

1. Demodulation of DSB

$$s_{AM}(t) = A_C\left\{\cos(2\pi f_C t) + \frac{m}{2}\cos\left[2\pi(f_C - f_M)t\right] + \frac{m}{2}\cos\left[2\pi(f_C + f_M)t\right]\right\}$$

$$s_{Aux}(t) = \cos(2\pi f_{Aux} t + \phi) \qquad 2.1$$

Fig. 2.9: Envelope curve demodulation

The amplitude of the auxiliary oscillation is negligible, furthermore it is true that $f_C = f_{Aux}$ (i.e. frequency equality prevails between carrier and auxiliary carrier). After lowpass filtering we obtain the demodulated signal:

$$s_D(t) = A_C \cos\phi + \frac{A_C m}{2} \cos(2\pi f_M t) \cos\phi \qquad 2.2$$

2. Demodulation of *DSB*$_{sc}$. The constant DC voltage component A_C cos φ is omitted:

$$s_D(t) = \frac{A_C m}{2} \cos(2\pi f_M t) \cos\phi \qquad 2.3$$

3. Demodulation of SSB$_{SC}$.

$$s_D(t) = \frac{A_C m}{4} \cos(2\pi f_M \pm \phi) \qquad 2.4$$

In the synchronous demodulation of *DSB* a phase error φ reduces the amplitude of the demodulated signal by the factor cos φ. In the case of *SSB,* the phase error leads to a shift in the demodulated signal. In both cases the phase between the carrier and the auxiliary carrier has a noticeable effect on the demodulation process. Due to this phase sensitivity synchronous demodulation is also called coherent demodulation.

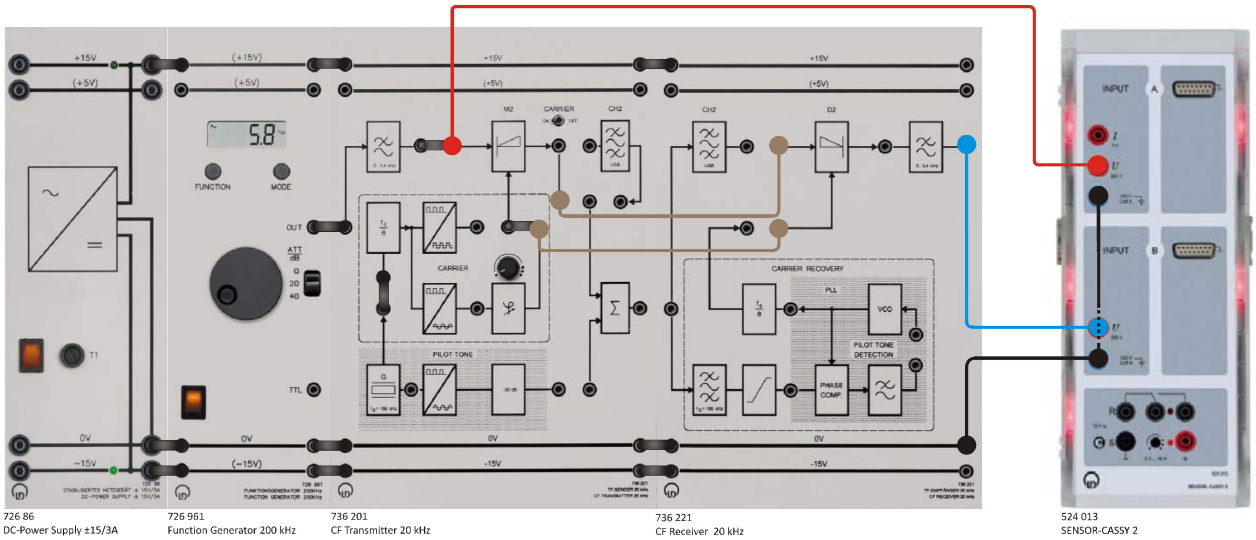
Questions

1.Which methods of AM demodulation are you familiar with and how do they differ?

2.Which demodulation method is used for AM with supressed carrier?

Experiment set-up

Assemble the components as shown below. Set the function generator to: sine, A_M = 2 V and f_M = 2 kHz.



Experiment set-up: **AM Demodulation (synchronous demodulation)**

DSB

- Set the phase controller on the CF transmitter to far left limit. Feed the *DSB* signal from the output of the modulator M2 directly into the demodulator D2 (do not use channel filter CH2!) Using a connecting lead feed the carrier signal (f_C = 20 kHz) of the CF transmitter into the auxiliary carrier input of the demodulator D2. What have you achieved by this?
- Display the modulating signal $s_M(t)$ on CASSY lab 2 as well as the demodulated signal $s_D(t)$ at the output of the LP filter of the CF receiver. Sketch the curve of the modulating signal and the demodulated signal in Diagram 3.3-1. Tap the auxiliary carrier for the demodulator D2 in front of the phase shifter of the CF transmitter.
- Start the measurement by pressing F9.

Diagram 3.3-1: Modulating and demodulated signal for the DSB.

With the exception of a constant amplitude factor the modulating signal $s_M(t)$ and the demodulated signal $s_D(t)$ are in agreement.

Carrier recovery

Carrier recovery is performed in the CF receiver using a PLL circuit. The PLL circuit is a control loop whose function is to match the frequency and phase of an oscillator to the reference oscillation. Fig. 4.2 illustrates the structure of a PLL circuit.

Let's assume that the input signal $s_1(t)$ is supplied with the frequency f_1 to the phase detector. You can be fairly certain that the VCO is not going to be so friendly as to oscillate precisely at the same frequency. So its frequency f_2 will initially differ from f_1. At the output of the phase detector an AC voltage is generated whose frequency is equal to the difference $f_2 - f_1$. This AC voltage is now supplied to the input of the VCO via the loop filter. The VCO will respond to an AC voltage at its input with a corresponding change in frequency. In turn the VCO's changing frequency is detected by the phase detector. With a little luck the PLL locks into the frequency of the input signal. The PLL corrects the VCO until the input frequency and the VCO frequency coincide. A voltage U_Φ arises behind the PD based on the phase shift. This is supplied to the VCO free of interfering AC components (U_F) through the loop filter. The following relationship prevails between the control voltage UF and the frequency f_{VCO} of the VCO:

$f_{VCO} = k_F \cdot U_F$

The control characteristic of the VCO (CF receiver)

Remove the bridging plug between the loop filter and VCO input at the PLL. Feed a variable DC voltage U1 from the function generator into the VCO input. Use this variable DC voltage to control the frequency of the VCO. Note down your measurement results in Table 5.3.2-1. Sketch the results in Diagram 4.2.

Attention: 0 V< UF < 5 V

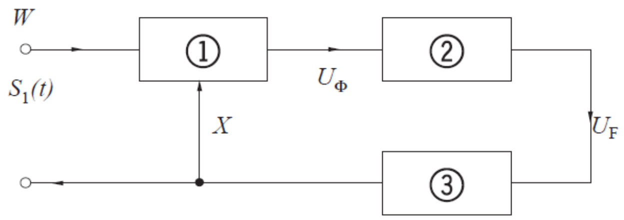


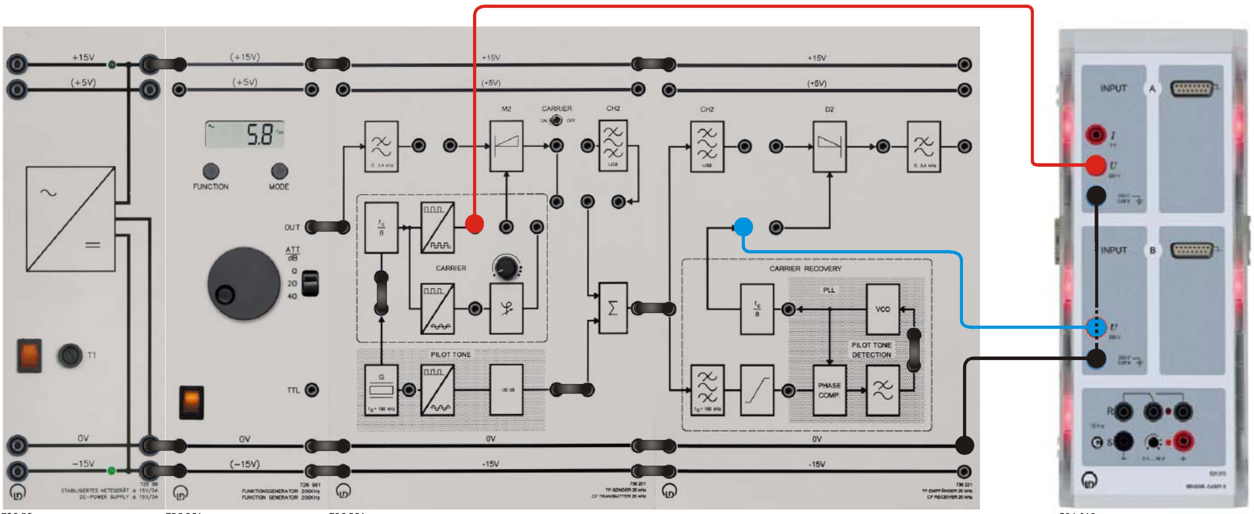
Fig. 3.3-1: Design of a PLL. (1): Phase detector, (2): Loop filter LF, (3): VCO

DC-voltage	Frequency
1	
2	
3	
4	
5	

Table1: variable DC voltage to control the frequency of VCO

Synchronous demodulation with the aid of a PLL-controlled VCO

Remove the cable connected to the auxiliary carrier input of the demodulator D2. For this insert the bridging plug between the CARRIER RECOVERY and auxiliary carrier input of D2. Use now for demodulation the recovered auxiliary carrier from the PLL CARRIER RECOVERY. Assemble the experiment set-up as shown below.



726 86
DC-Power Supply ±15/3A

726 961
Function Generator 200 kHz

736 201
CF Transmitter 20 kHz

736 221
CF Receiver 20 kHz

524 013
SENSOR-CASSY 2

Experiment set-up: **Synchronous demodulation with the aid of a PLL-controlled VCO**

Sketch the pilot tone of the transmitter and the recovered signal in the receiver at the output of the PLL circuit in Diagram 3.3-2.

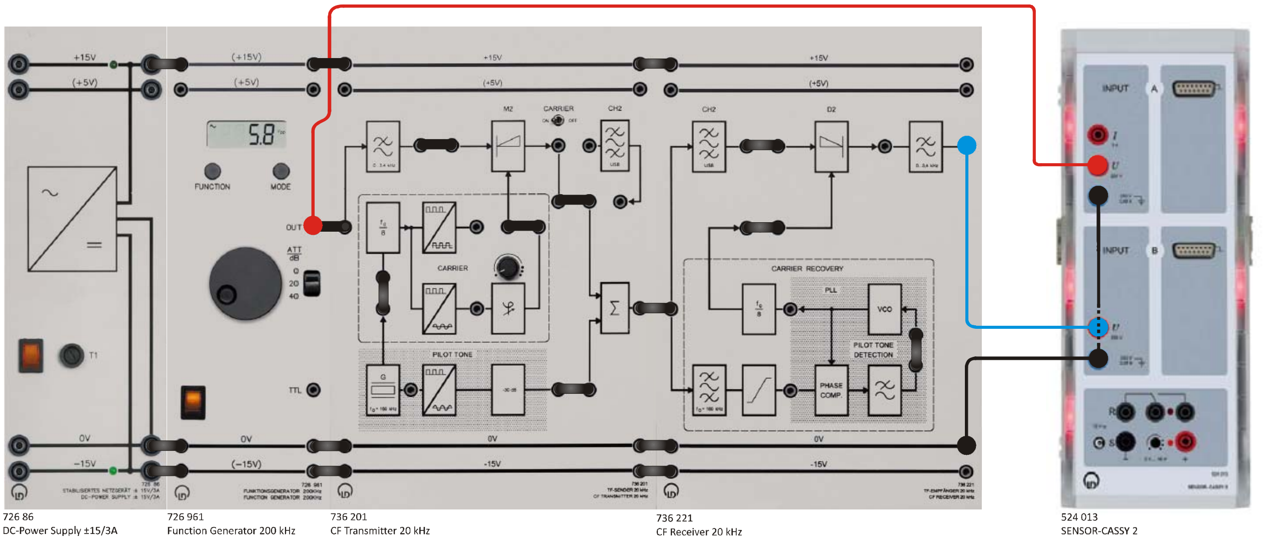
- Start the measurement by pressing F9.

Hint:
The required auxiliary carrier oscillation is generated out of the pilot tone recovered in the PLL circuit by means of frequency division *f*/8. Depending on the initial state of the frequency divider this creates a fixed phase shift between the auxiliary carrier and carrier oscillation. Consequently, the demodulated signal shows an amplitude error. (For the sake of testing connect and disconnect the bridging plug in the PLL-circuit of the CF receiver and observe the amplitude of the demodulated signal.

Diagram 3.3-2: Pilot tone and recovered signal of the receiver.

DSB_{SC} Demodulation

Assemble the experiment set-up as shown below.



Experiment set-up: **VCO DSB_{SC} demodulation**

- Set the toggle switch to **CARRIER OFF**. Repeat the experiment in accordance to the previous paragraph “**DSB**”.
- Start the measurement by pressing F9.
- Discuss the results.

Diagram 3.3-3: Modulating and demodulated signal in DSB_{SC}

Summarize the requirements made on the auxiliary carrier in synchronous demodulation.

The DSB_{SC} shows the same phase-dependency as the DSB. Requirements for the auxiliary carrier in synchronous demodulation:

1. Frequency stability and frequency equality with the original carrier frequency.

2. Constant phase angle < 90°. Ideally $\varphi = 0°$.

3. Amplitude stability of the auxiliary carrier

Part two : The single Sideband AM (SSB)

In DSB each sideband carries all of the information contents. The transmission bandwidth could thus be reduced by half, if one sideband is suppressed. It does not matter which sideband is used for transmission and which one is suppressed. The upper sideband appears in the normal position, the lower one in the inverted position. If, for example, we suppress the lower sideband in Fig. 2.5 then we obtain:

$$s_{SSB}(t) = A_C \cos(2\pi f_C t) + \frac{m}{2}\cos[2\pi(f_C + f_M)t] \qquad 2.5$$

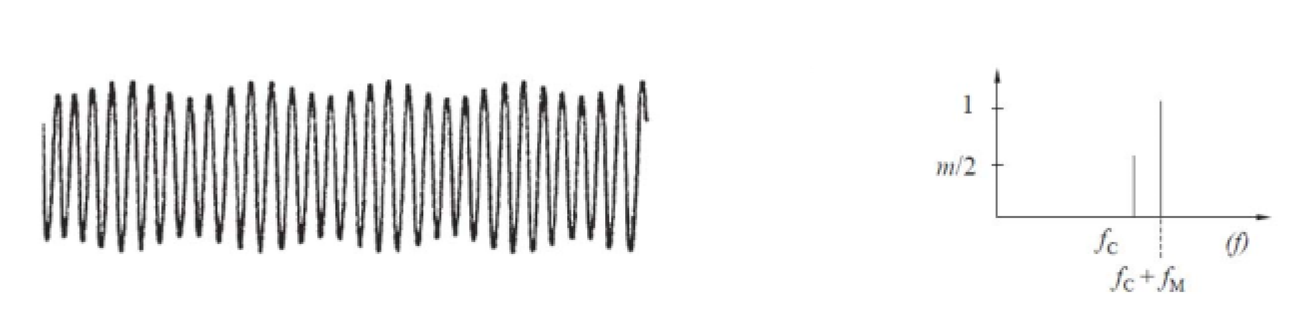


Fig. 2.5: Representation of SSB.

To suppress a sideband, a bandpass filter with sharp cutoff is used which only allows the desired spectral components to pass. The dynamic characteristic of *SSB* resembles that of the *DSB*$_{SC}$. However the envelope curve is somewhat more distorted.

Demodulation method : Synchronous demodulation
Bandwidth requirement : $b = f_{Mmax}$ 2.6
Application : Line-bound transmission of telephone signals in frequency division multiplex technology

The *SSB* with residual carrier

If instead of the unattenuated carrier only a defined fraction *k* of the carrier amplitude is transmitted, then you obtain the *SSB* with residual carrier:

$$s_{ESB,T}(t) = \left[A_C k \cos(2\pi f_C t) + \frac{m}{2}\cos[2\pi(f_C + f_M)t]\right] \qquad 2.7$$

Demodulation method : Synchronous demodulation
Bandwidth requirement : $b = f_{Mmax}$.
Application : *SSB* radio links

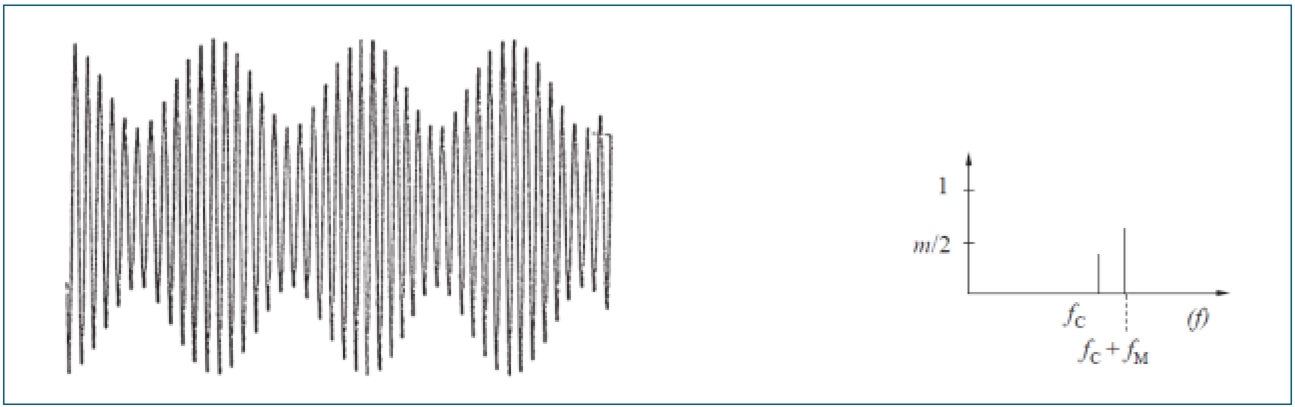


Fig. 2.6: Representation of SSB.

The vestigial sideband AM (*VSB)*

In message or information signals with very low frequency components bandpass filters with very sharp cutoffs are required for the filtering out of the unwanted sideband. Since this leads to phase distortion, part of the unwanted sideband is also transmitted. Then the filters used may have cutoffs which are less sharp, on the other hand, the slope characteristic has to be precisely defined (Nyquist slope).

The overlapping transmission range has to run symmetrically with respect to the carrier frequency.

Demodulation method : Synchronous demodulation
Bandwidth requirement : $f_{Mmax} < b < 2\ f_{Mmax}$
Application : TV technology

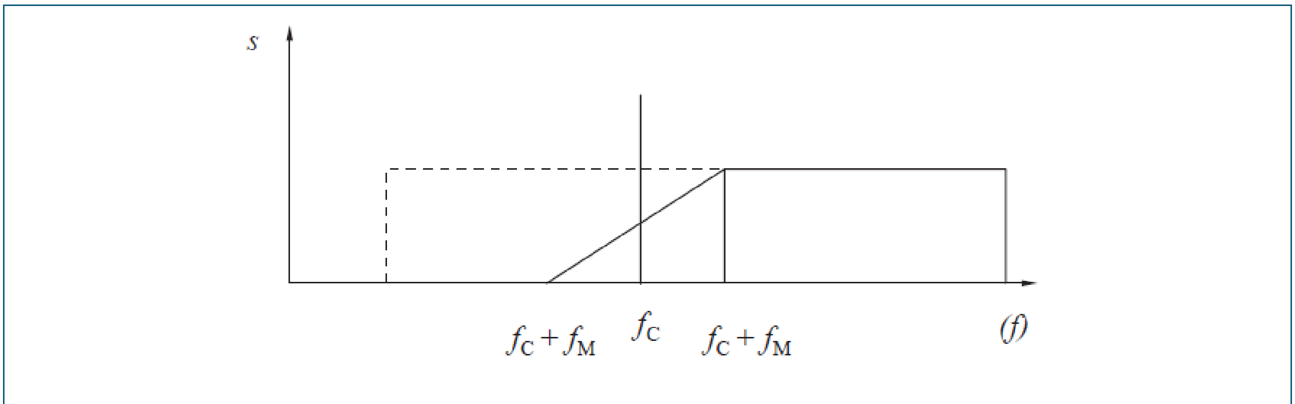


Fig. 2.7: Filter characteristic with the Nyquist slope for VSB.

Investigations on the dynamic characteristic of the SSB

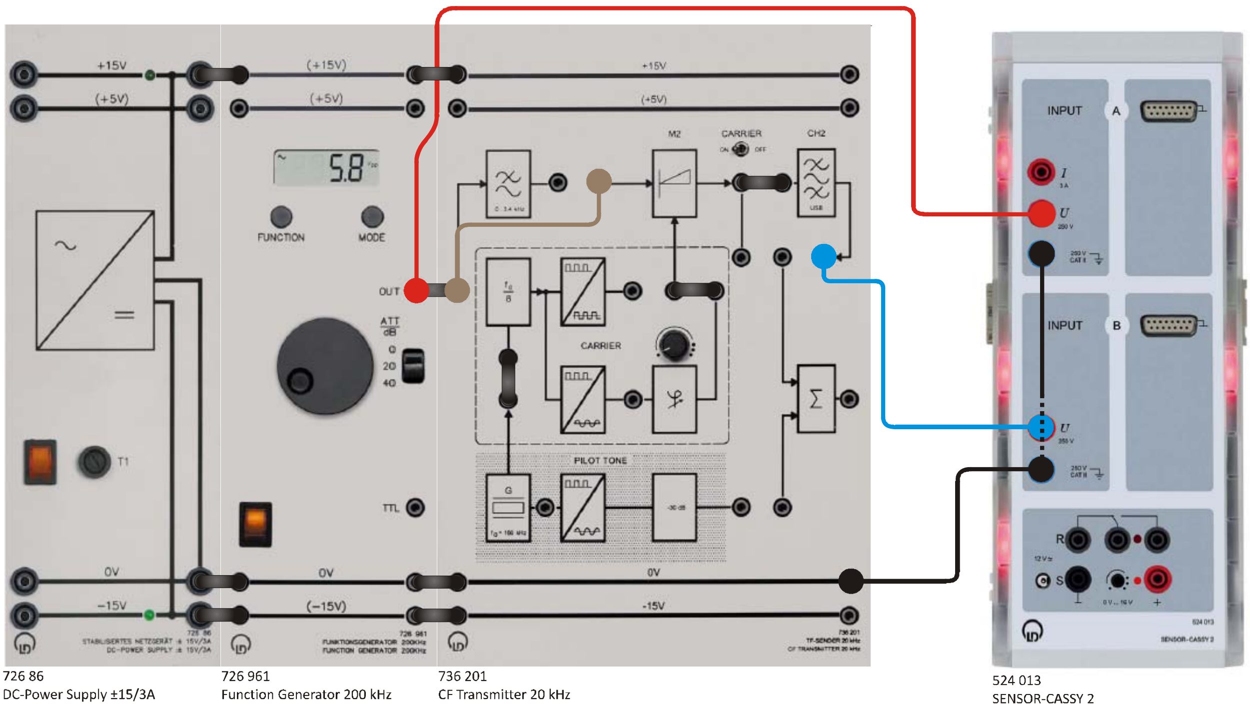
Required accessories

To carry out the experiments the following accessories are required:

Qty.	Cat.-no.	Designation
1	736 201	CF-Transmitter 20 KHZ
1	736 221	CF-Receiver 20 KHZ
		Accessories
1	726 09	Panel Frame-T130, Two Level
1	726 86	DC-Power Supply ± 15 V/3 A
1	726 961	Function Generator 200 kHz
1	501 46	Pair cables 100 cm, red/blue
2	501 461	Pair cables 100 cm, black
1	501 511	Set of 10 Bridging plugs, black
		Measuring instrument
1	524 013S	Sensor-CASSY 2 - Starter

Experiment set-up

Set up the experiment as specified below. Connect the output of the function generator directly to the AF input of the modulator M2. Set the function generator to: sinusoidal, A_M = 2 V and f_M = 2 kHz.



Experiment set-up: **Single Sideband**

SSB

- Set the toggle switch to **CARRIER ON**.
- Feed a sine signal from the function generator with f = 2 kHz, A = 1.5 V
- Display the output signal of the channel filter CH2 and the modulating signal $s_M(t)$ of the function generator on Sensor - CASSY 2 - Starter and sketch the signals. (Modulation product on channel B, modulating signal on channel A of Sensor - CASSY 2 - Starter).
- Start the measurement by pressing F9.

SSB_{SC}

- Set the toggle switch to **CARRIER OFF**.
- Proceed as described in the paragraph "***SSB_{RC}***".
- Start the measurement by pressing F9.
- What features does the *SSB_{SC}* signal have?

Diagram 4.1-1: Dynamic characteristic of the SSB_{RC}.

The SSB_{RC} signal resembles the DSB signal in terms of its dynamic characteristic.

Trigger to the modulating signal $s_M(t)$. Switch the modulating signal off. Measure the unattenuated carrier amplitude A_C at the input of the channel filter, as well as the amplitude A_{RC} of the attenuated carrier at the output of the channel filter. Calculate the ratio $k = A_{RC}/A_C$. Determine the carrier suppression t in dB according to the equation 2.8:

$$t = 20 \log \frac{m}{2k} \qquad 2.8$$

Diagram 4.1-2: Dynamic characteristic of the SSB_{SC}.

The modulated SSB_{SC} signal is a pure sinusoidal signal when the carrier and the unwanted sideband have been completely suppressed.

Spectrum of the SSB

SSB_{RC}

- Set the toggle switch to **CARRIER ON**. As the modulating signal use a sinusoidal signal with A_M = *2 V* and f_M = 1 kHz. Feed the modulating signal into the input LP filter of the CF transmitter.
- Measure the SSB spectrum in the range of approx. 15 kHz up to 25 kHz obtained from:

Label the spectral lines. Determine the transmission bandwidth of the AM signal based on the measurements. Generalize the results for the case of any modulating signals.

Diagram 4.2-1: SSB_{RC}. Spectrum.

SSB_{SC}

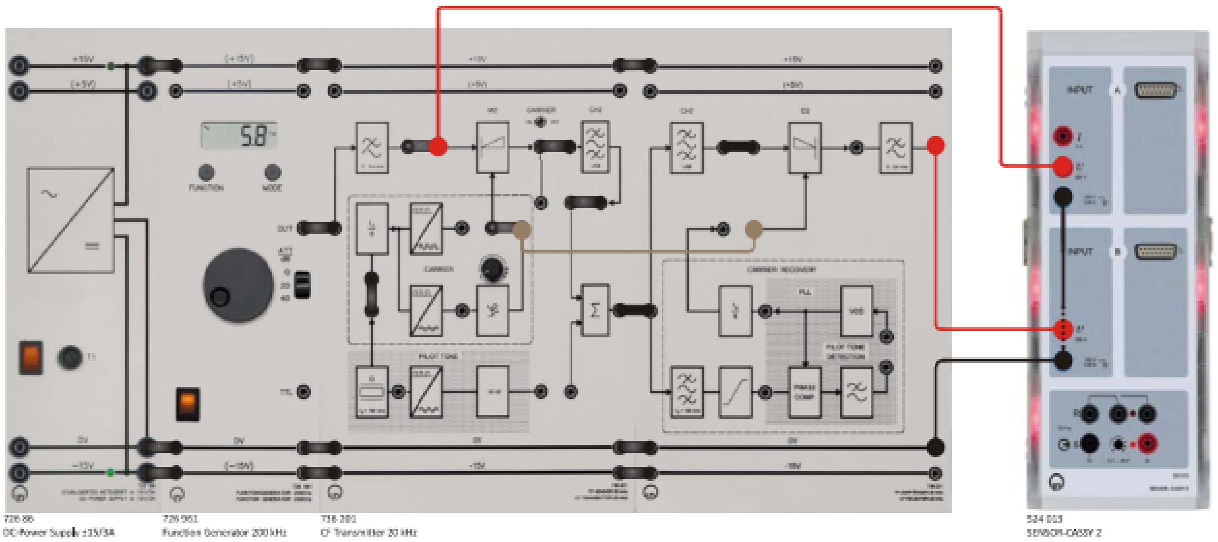
Set the toggle switch to **CARRIER OFF**. Use a sinusoidal signal with A_M = 2 V and f_M = 2 kHz as a modulating signal. Measure the spectrum as described in the previous paragraph "***SSB_{SC}*** ".

Diagram 4.2-2: SSB_{SC}. Spectrum.

SSB demodulation

Experiment set-up

Set up the experiment as specified below. Start with the settings of the Sensor-CASSY 2 - Starter . With the aid of a connecting lead feed the carrier signal (f_C = 20 kHz) of the CF transmitter directly into the RF input of the demodulator D2. What have you achieved by this? Set the function generator to: sinusoidal, A_M = 2 V and f_M = 2 kHz.



Experiment set-up: **SSB demodulation**

SSB_{RC}

- Display the modulating signal $s_M(t)$ on Sensor-CASSY 2 - Starter as well as the demodulated signal $s_D(t)$ at the output of the LP filter of the CF receiver.
- Start the measurement by pressing F9.
- Adjust the phase between the original carrier of the transmitter and the auxiliary carrier of the receiver. What do you observe?

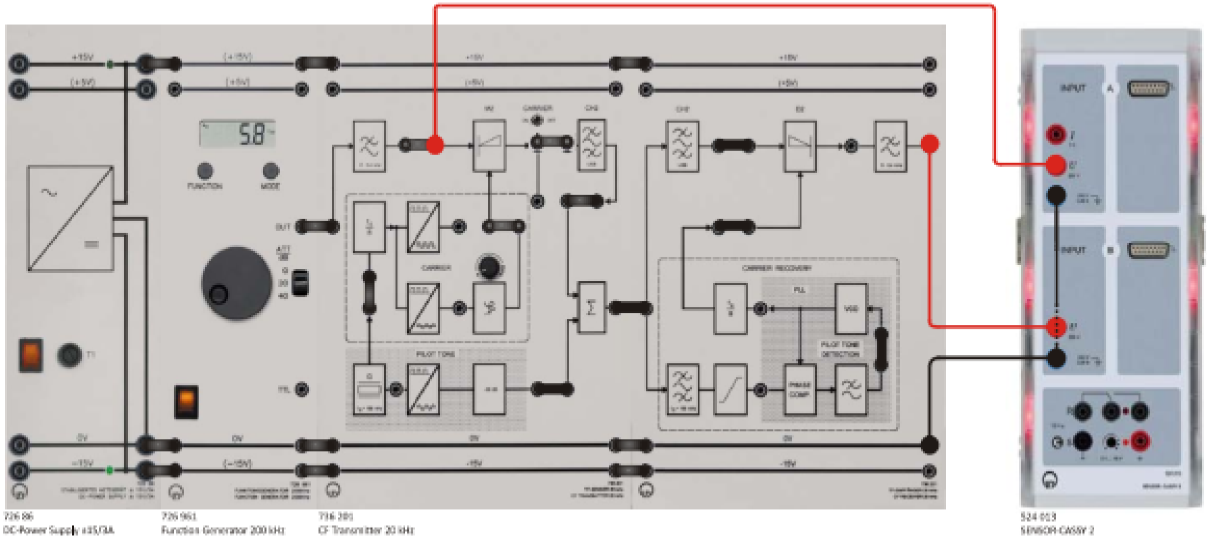
Diagram 4.3-1: Modulating and demodulated signal in SSB.

The synchronous demodulation of a SSB_{RC} signal provides a perfectly suitable demodulated signal. The phase-shift set by the phase controller of the CF transmitter occurs between the modulating signal $s_M(t)$ and demodulated signal $s_D(t)$. Any influence on the amplitude of the demodulated signal cannot be detected. The following applies for the demodulated signal:

$$s_D(t) = \frac{A_C m}{4} \cos\left(2\pi f_M t \pm \phi\right)$$

SSB_{SC}

Remove the connecting lead between the CF transmitter and the demodulator. Now for the demodulation use the recovered auxiliary carrier from the PLL circuit to recover the carrier. For this connect the bridging plug between CARRIER RECOVERY and the auxiliary carrier input of D2. Discuss your findings.



Experiment set-up: **SSB_{SC} demodulation**

- Set the toggle switch to **CARRIER OFF**. This time repeat the experiment for *SSB_{SC}*.
- Start the measurement by pressing F9.

Diagram 4.3-2: Modulating and demodulated signal in SSBSC.

Also in the case of SSB signals the carrier has no influence on synchronous demodulation. It can be switched on and off at will.

5. The Ring Modulator

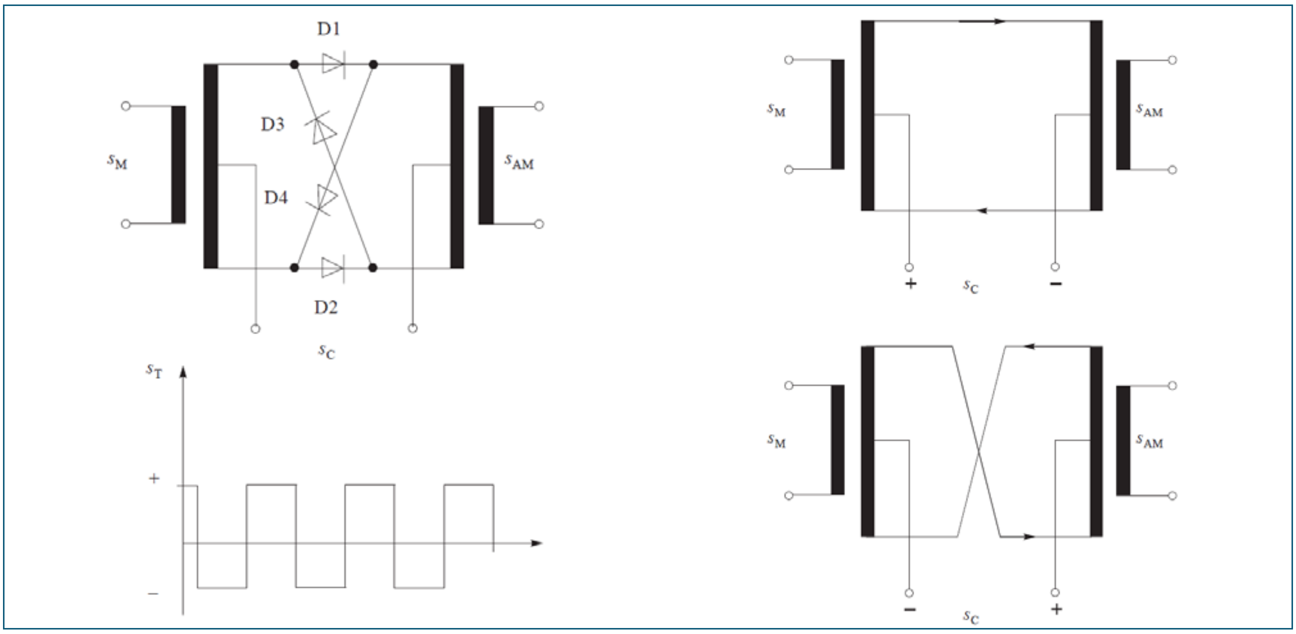


Fig. 2.8: Ring modulator in diode technology

The modulation process has been described as a multiplication of a harmonic carrier $s_C(t)$ with an equally harmonic message signal $s_M(t)$ using a multiplier IC. This is how the wanted modulation product is directly obtained without any undesired sidelines. However, in practice product modulators in the form of integrated circuits are of no importance because they can only be used at relatively low frequencies. Consequently, modulation is performed using discrete components with non-linear characteristics on account of the high frequencies used in communications engineering. A group of modulators important in practice is known under the name of balanced modulators. These types of balanced modulators include the push-pull and ring modulators.

1. Angle modulation

In the frequency and phase modulation of a harmonic carrier $s_c(t)$, the modulating signal $s_M(t)$ has an effect on the instantaneous angle:

$$\Psi(t) = 2\pi f_c t + \Phi$$

For this reason FM and PM are seen jointly as angle modulations.

Phase modulation

Given that the modulating signal is harmonic with zero-phase angle Φ = 0°:

$$s_M(t) = A_M \cos(2\pi f_M t) \qquad (1)$$

The message signal should have a direct impact on the instantaneous phase angle $\Psi(t)$. From this it follows that:

$$\Psi_{PM}(t) = 2\pi f_c t + A_M k \cos(2\pi f_M t)) \qquad (2)$$

where k stands for the modulation constant of the angle modulator. This equation describes the influence of the modulating signal $s_M(t)$ on the instantaneous phase angle $\Psi(t)$. For the dynamic (time) characteristic of a PM signal we obtain:

$$s_{PM}(t) = A_c \cos[2\pi f_c t + A_M k_{PM} \cos(2\pi f_M t)]. \qquad (3)$$

The variable $A_M k_{pM}$ specifies the maximum angular change which the modulating signal can evoke. This is called phase deviation $\Delta\Phi_{PM}$:

$$\Delta\Phi_{PM} = k_{PM} A_M \qquad (4)$$

The instantaneous frequency f_{PM} is obtained from the derivation of the phase angle with respect to time:

$$f_{PM} = \frac{1}{2\pi}\frac{d}{dt}\Psi_{PM}$$

$$f_{PM} = \frac{1}{2\pi}\frac{d}{dt}[2\pi f_c t + A_M k_{PM} \cos(2\pi f_M t)]$$

$$f_{PM} \quad f_c = A_M k_{PM} f_M \sin(2\pi f_M t) \qquad (5)$$

Since the frequency f_{PM} changes in time, it is called the instantaneous frequency. The maximum change in frequency is called the frequency deviation ΔF and in PM amounts to:

$$\Delta F_{PM} = A_M k_{PM} f_M = \Delta\Phi_{PM} f_M \qquad (6)$$

Normally the characteristic of the frequency deviation and phase deviation are graphically represented as a function of the modulation frequency f_M:

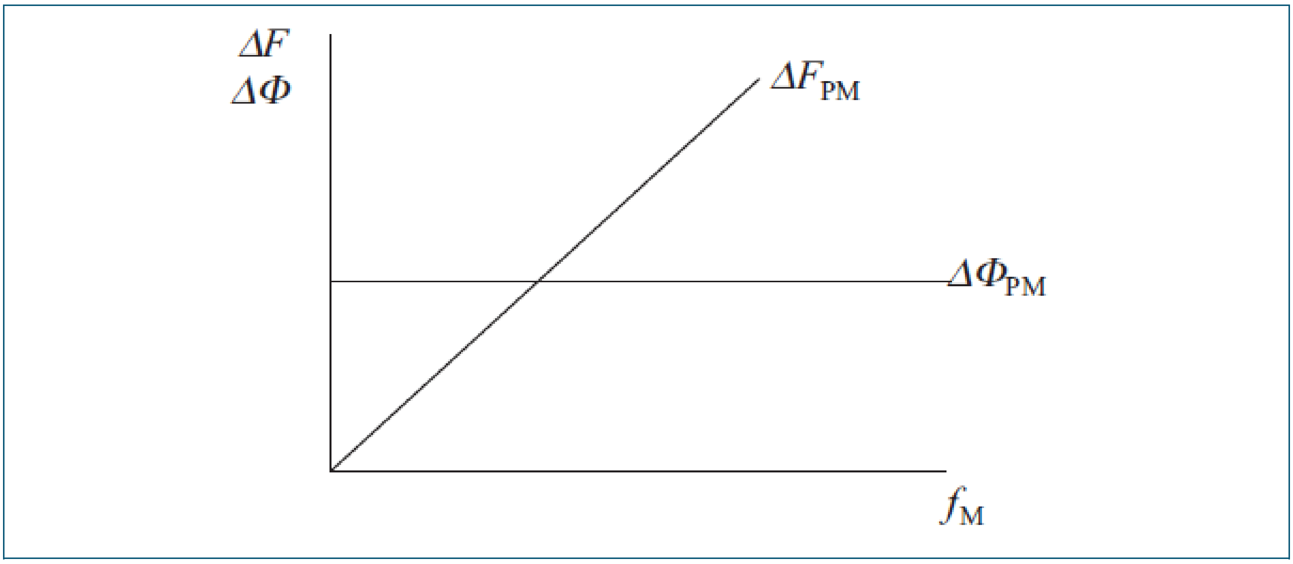


Fig. 1: Frequency and phase deviation for PM as a function of f_M

Frequency modulation

Another important angle modulation method used in practice is FM. Here the modulating signal $s_M(t)$ affects the instantaneous frequency $f_{FM}(t)$ of the carrier oscillation:

$$f_{FM} = f_c + \Delta_{FM} \cos(2\pi f_M t) \qquad (7)$$

where ΔF_{FM} means the frequency deviation for FM. For ΔF_{FM} the following holds true:

$$\Delta F_{FM} = k_{FM} A_M \qquad (8)$$

In order to determine the instantaneous phase angle $\Psi_{FM}(t)$, the instantaneous frequency $f_{FM}(t)$ has to be integrated with respect to time:

$$\Psi_{FM}(t) = 2\pi \int f_{FM}(t) dt$$

$$\Psi_{FM}(t) = 2\pi \int [f_c + \Delta F_{FM} \cos(2\pi f_M t)] dt'$$

$$\Psi_{FM}(t) = 2\pi f_c + \frac{\Delta F_{FM}}{f_M} \sin(2\pi f_M t) \qquad (9)$$

Thus in FM the phase deviation is:

$$\Phi_{FM} = \frac{\Delta F_{FM}}{f_M} = \frac{k_{FM} A_M}{f_M} \qquad (1\text{-}10)$$

For the modulated FM signal you obtain:

$$s_{FM}(t) = A_c \cos\left[2\pi f_c t + \frac{\Delta F_{FM}}{f_M}\sin(2\pi f_M t)\right] \tag{1-11}$$

If, as in the case of PM, you record the frequency and phase deviation over the signal frequency f_M you obtain the depiction according to fig. 1-2.

Note: Harmonic modulating signals $s_M(t)$ are the prerequisite when dealing with the definition of frequency and phase deviation.

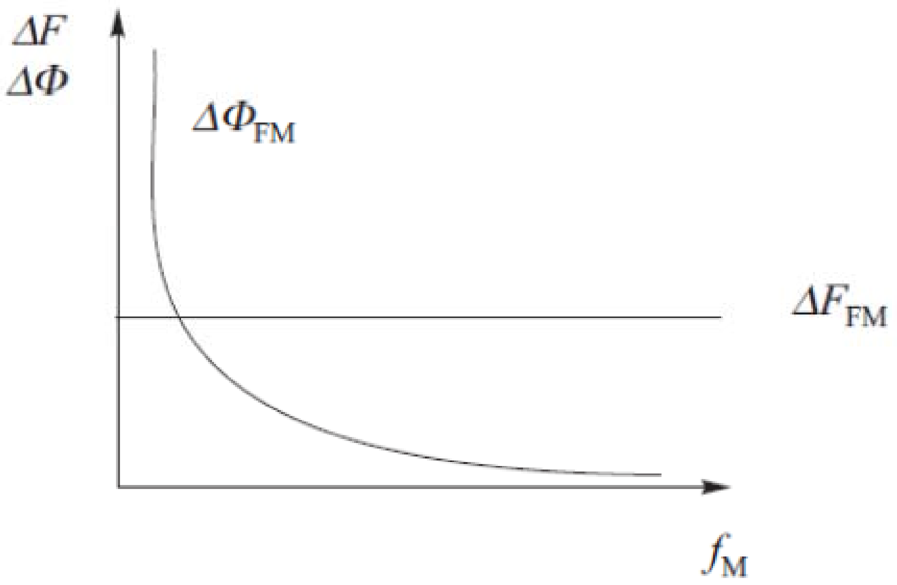


Fig. 2: Frequency and phase deviation in FM as a function of f_M

The modulation index η

The modulation index is an important variable in the discussion of modulation spectra and the noise of angle modulation methods. The following applies for this:

$$\eta_{FM} = \frac{\Delta F_{FM}}{f_M} = \Delta\Phi_{FM} \tag{1 2}$$

$$\eta_{PM} = \Delta\Phi_{FM} \tag{1-13}$$

The introduction of the modulation index permits us to discuss the spectra of FM and PM jointly.

The spectrum of an angle-modulated oscillation

When deriving the spectrum of an angle-modulated oscillation the starting point is the following equation:

$$s_{ang}(t) = A_c \cos[2\pi f_c t + \eta \cos(2\pi f_M t)] \qquad (1\text{-}14)$$

The discussion applies to the same extent for both FM and PM. As the mathematical procedures required for this are considerable, only the results shall be presented here. Since angle modulated signals are time periodic, they have discrete line spectras. The spectral lines occur in intervals of whole number multiples (integers) of the signal frequency f_M with respect to the carrier f_c. The amplitudes of the spectrum are expressed by the Bessel functions of the first kind of the order n.

$$J_n(\eta) = \frac{\eta^n}{[2G(n+1)]}$$

$$J_n(\eta) = \left[1 - \frac{\eta 2}{[2(2n+2)]} + \frac{\eta 4}{[2\cdot 4(2n+2)(2n+4)]} - \frac{\eta 6}{[2\cdot 4\cdot 6(2n+2)(2n+4)(2n+6)]} + \dots\right] \qquad (1\text{-}15)$$

Where G (n + 1): Gamma Junction

With the aid of the spectrum you can rearrange the signal equation according to (1-14):

$$s_{ang}(t) = A_c \sum_{n=-\infty}^{\infty} J_n(\eta) \cos[2\pi(f_c + \eta f_M)t] \qquad (1\text{-}16)$$

The characteristics of the Bessel functions for n = 0 ... n = 10 are reproduced in fig. (3,4). The characteristic curves look like attenuated oscillations. An arbitrarily drawn spectrum of angle modulation is depicted in fig. 5.

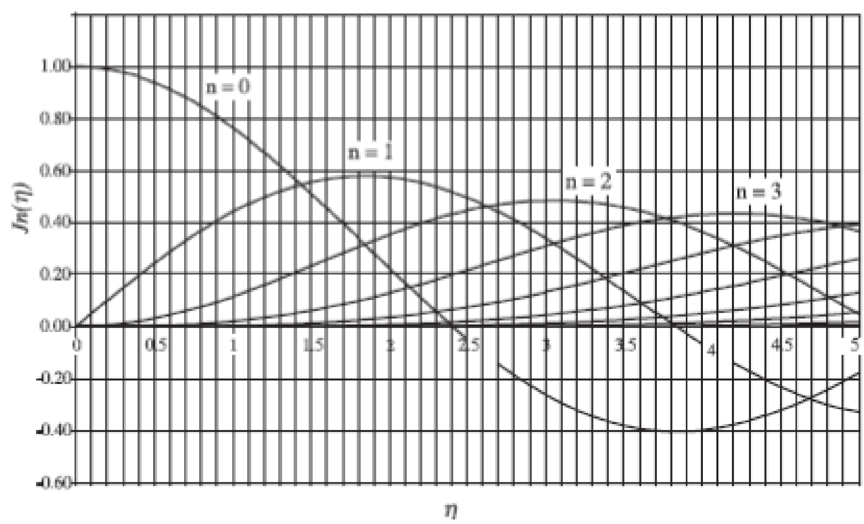


Fig. 3: The Bessel functions of the 1st kind, of the order *n = 0 ... 10*

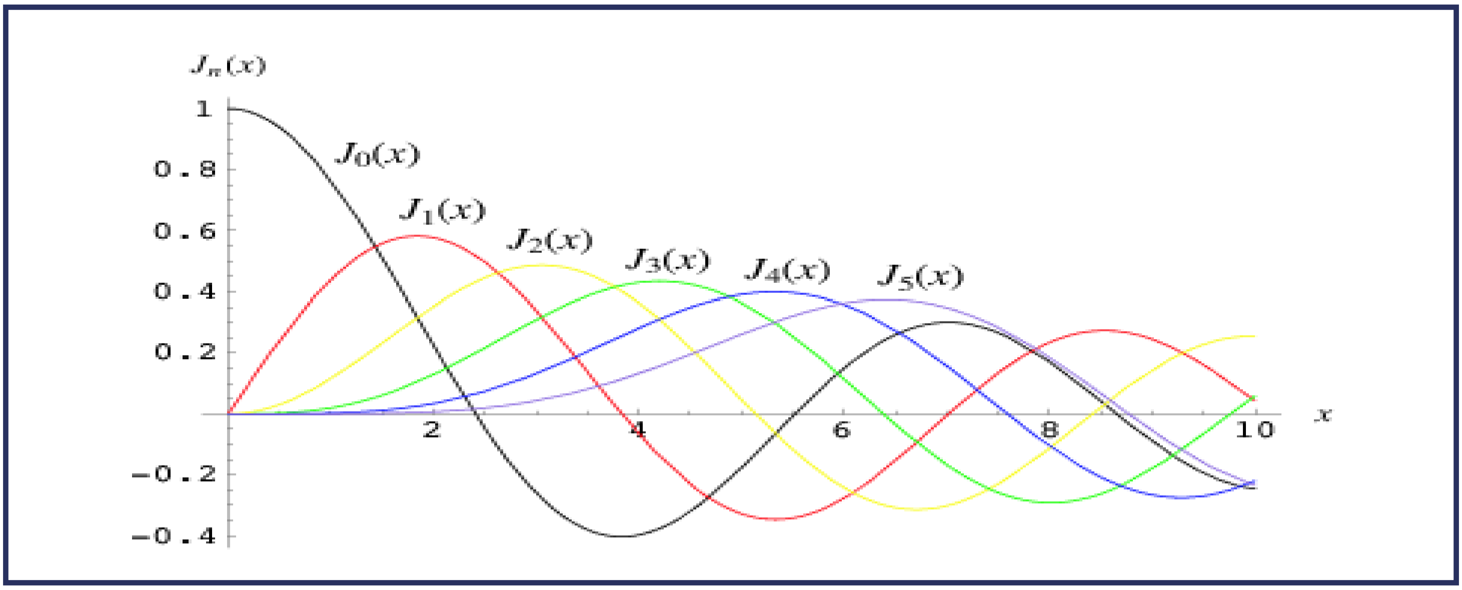


Fig.4 :The Bessel functions of the first kind of the order n=0 ... 10

The carrier amplitude is determined by the Bessel function $J_0(\eta)$, sidelines *at* $f_c \pm f_M$ by $J_1(\eta)$, the sidelines *at* $f_c \pm 2f_M$ by $J_2(\eta)$, etc. The angle-modulated oscillation has a spectrum which extends infinitely in principle. However, the amplitudes for the sidelines of a higher order rapidly drop off particularly for small modulation indices. In practice calculation is performed with the Carson bandwidth. Here, only those spectral components are taken into consideration with amplitudes exceeding a fixed fraction of the carrier amplitude (17):

$b_{ang} = 2(\Delta F + f_M)$ spectral lines > 10% of the carrier amplitude

$b_{ang} = 2(\Delta F + 2f_M)$ spectral lines > 1 % of the carrier amplitude (17)

The bandwidth requirement is thus dependent on the desired transmission quality. Obviously the modulation index is the decisive parameter in the angle modulation spectrum. Due to the oscillating curve of the Bessel functions, there might be individual spectral lines missing in the spectrum. For this to happen, a zero crossing has to exist in the corresponding Bessel function. This also applies for the carrier! This feature is exploited to determine the frequency or phase deviation experimentally. The following table provides an overview of the first five zeros in the Bessel functions $J_0(\eta)$… $J_2(\eta)$.

Zeros of the Bessel functions

$J_0(\eta) = 0$ η	$J_1(\eta) = 0$ η	$J_2(\eta) = 0$ η
2.4	*3.8*	*5.1*
5	*7.0*	*8.3*
8.7	*10.2*	*11.5*
11.8	*13.3*	*14.7*
14.9	*16.5*	*16.9*

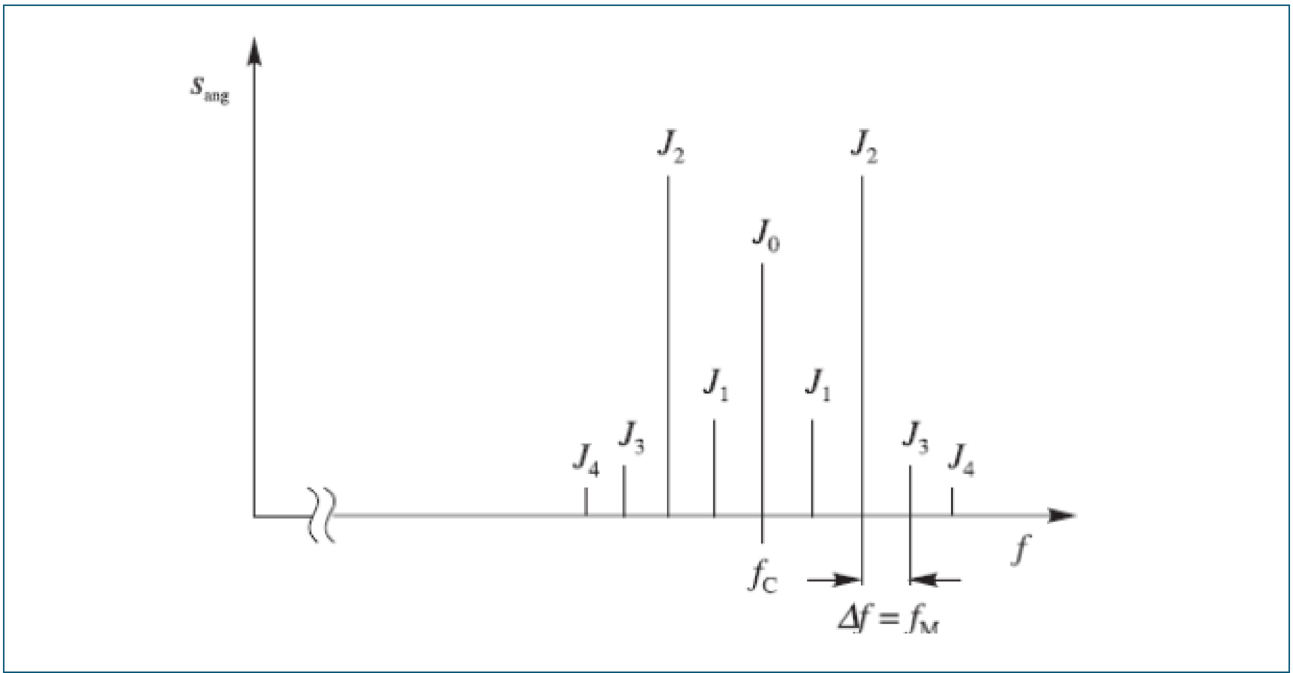


Fig. 5: The spectrum of an angle-modulated oscillation

Generation of angle modulation

The simplest method of generating an FM signal involves a controllable reactance, e.g. a frequency determining varicap-diode in a resonant circuit, see fig. 1-5. The resonance frequency f_R of the oscillation circuit is reproduced by Thomson's oscillation equation:

$$f_R = \frac{1}{\left[2\pi\sqrt{LC}\right]} \tag{18}$$

Here C makes up the total circuit capacitance, which is a function of the signal voltage $s_M(t)$. Consequently, the resonance frequency f_R also follows the variations of $s_M(t)$. FM is frequently derived from PM due to the better stability of the carrier's center frequency. PM can be realized using various methods; for example, it can be generated out of an AM signal using a method similar to QAM.

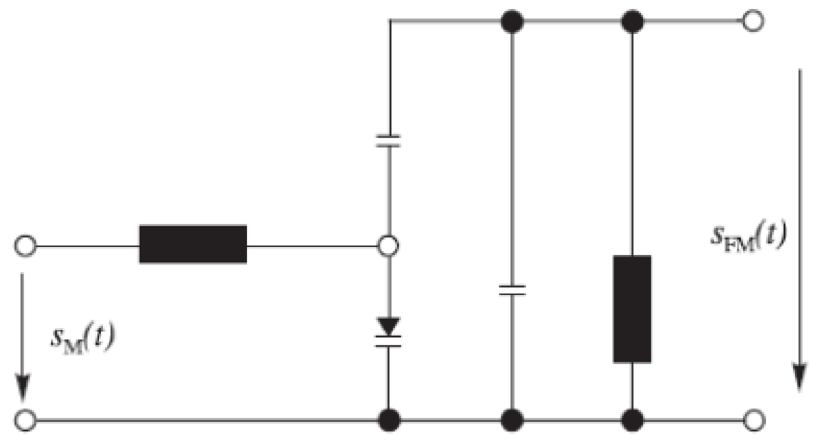


Fig. 6: Generation of FM using voltage-controlled capacitance diode $s_{FM}(t)$

In PM the frequency deviation ΔF_{PM} climbs linearly with the signal frequency f_M. While in FM the frequency deviation is independent of f_M. If you wish to convert PM into FM, all you have to do is compensate for the proportional rise of f_M by making sure that a suitable low pass filter is inserted into the signal path. The *LP* operates with integrating action. Its installation in the signal path is depicted in fig. 7.

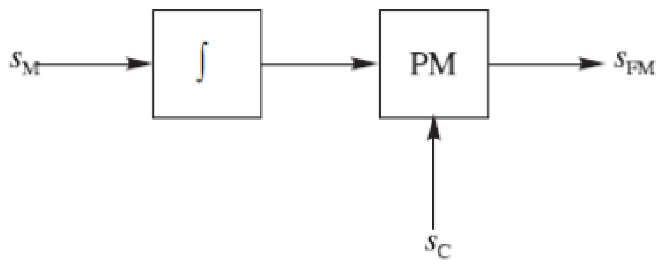


Fig. 7: Generation of FM out of PM

On the other hand you can also generate PM out of FM by inserting a differentiator element in the signal path, see fig. 8.

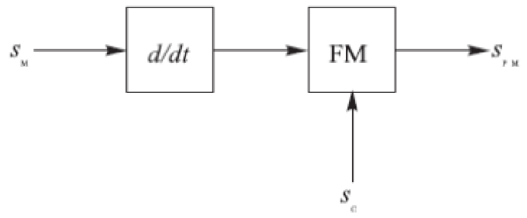


Fig. 8: Generation of PM out of FM.

Demodulation of angle modulation

There are a number of important methods used for the demodulation of FM or PM signals:

- Conversion of FM in AM with subsequent envelope demodulation (frequency discriminators)
- Conversion of FM into a pulse-width modulated oscillation with subsequent demodulation (coincidence demodulation)
- Use of a phase control circuit (PLL)

The training panel, 736 28, FM/PM demodulator uses a PLL circuit. For that reason we will take a closer look at the application of the PLL in FM demodulation. The explanations refer to fig. 9.

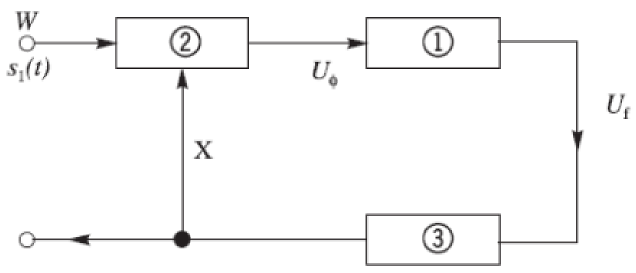


Fig. 9: Principle of the PLL 1:
Loop filter
2: Phase detector
3: VCO

The phase of the received signal is measured and the difference between it and the phase angle of the VCO is determined. This task is performed by the phase detector PD. The output signal of the PD is filtered and supplied to the VCO input. Here frequency correction of the VCO signal is performed. If the instantaneous frequency of the received signal changes, an AC voltage is produced at the output of the loop filter. This AC voltage is proportional to the modulating signal. Selecting the correct parameters for the loop filter is vital for interference-free demodulation.

Difference between the demodulation of FM and PM

When using frequency discriminators, the demodulated signal is proportional to the frequency deviation ΔF. In FM this is directly proportional to the amplitude A_M of the modulating signal according to (8):

$$\Delta F_{FM} = k_{FM} A_M \quad (8)$$

If you use these kinds of discriminators for the demodulation of PM, then in accordance with (6) you have to take into account that the frequency deviation the discriminator responds to increases proportionally with the modulation frequency f_M

$$\Delta F_{PM} = k_{PM} A_M f_M \quad (6)$$

This increase must be compensated for in the demodulator.

How angle modulation methods respond to interference

The response of modulation methods to interference is described by the modulation gain G:

$$G = \frac{SNR_{AF}}{SNR_{RF}} \quad (19)$$

While in FM the signal-to-noise ratio (SNR) and modulation gain G decreases with increasing modulation frequencies, in PM these variables remain constant for all f_M. In order to make FM more immune to interference at high signal frequencies, the amplitude A_M and therewith the ΔF_{FM} have to be correspondingly increased with the signal frequency. This process is called preemphasis. This is the equivalent of a differentiation of the modulating signal.

Questions

1 Which carrier parameters can be used for angle modulation?

2 Which angle modulation method is least affected by noise? Why?

3 What does preemphasis mean?

4 When is deemphasis used in FM demodulation? How does it function?

5 In FM with $\eta = 1.0$ how high are the amplitudes of the 1^{st} sideline compared to AM for *m* = 100%? Given that the amplitude of the unmodulated carrier is $A_C = 1$ V. Use fig. 1-3.

6 Assuming in FM that the spectrum for a harmonic signal $A_1 = 5$ V and $f_{M1} = 1000$ Hz is known. Can conclusions be drawn regarding the spectrum, which is produced for a different harmonic modulation signal, with, for example $A_1 = 3$ V and $f_{M1} = 100$ Hz? Compare this with AM!

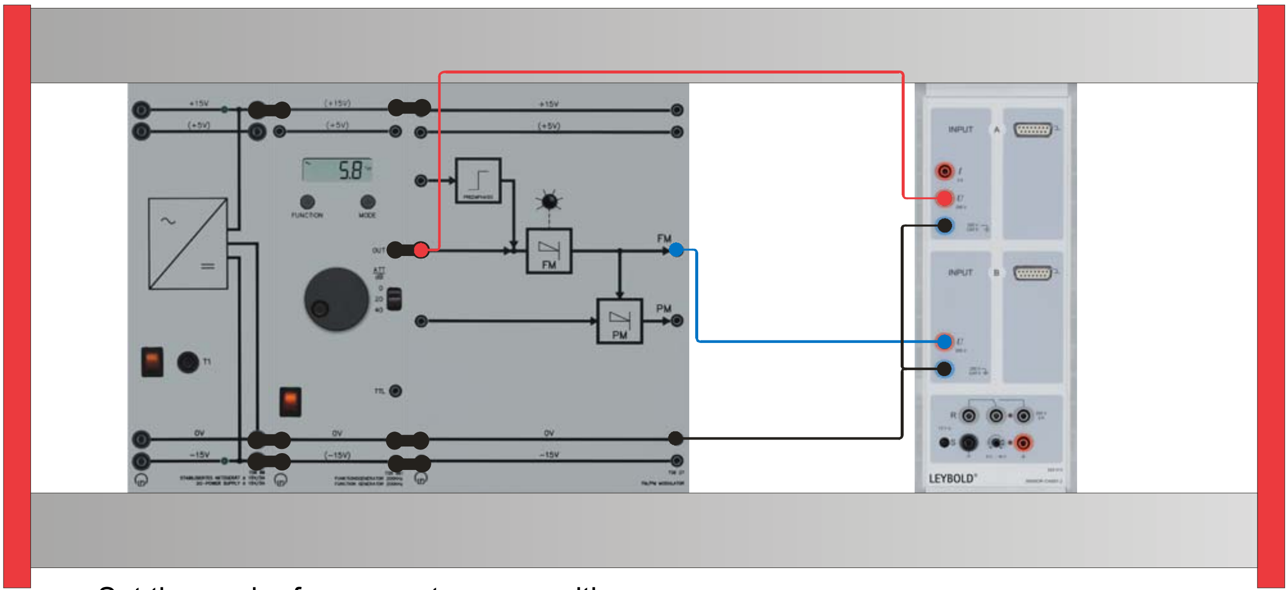
Equipment overview

Qty.	Cat.-no.	Designation
1	736 27	FM/PM-Modulator
1	736 28	FM/PM-Demodulator
		Accessories
1	726 09	Panel Frame-T130, two Level
1	726 86	DC-Power Supply ± 15 V/3 A
1	726 961	Function Generator 200 kHz
2	501 46	Pair cables 100 cm, red/blue
2	501 461	Pair cables 100 cm, black
1	501 511	Set of 10 Bridging plugs, black
1	501 512	Set of 10 Bridging plugs with tap, black
		Measuring instrument
1	524 013S	Sensor-CASSY 2 - Starter

Frequency modulation

Dynamic response of FM

Experiment set-up: Assemble the components as shown below.



- Set the carrier frequency to max position.
- 1.Use a square signal with AM = 20 Vpp and fM = 0.1 Hz of the function generator as the modulating signal sM(t). Feed the modulating signal into the input of the FM modulator,observe the spectrum.

  2.Use a sinusoidal signal with A_M = 20 V_{pp} and f_M = 1.0 kHz of the function generator as the modulating signal sM(t). Feed the modulating signal into the input of the FM modulator.
- Start the measurement by pressing F9.

- The amplitude of the FM modulated signals remains constant.
- Information is carried within the instantaneous change of the carrier frequency.

The characteristic of the FM modulator

- Start the measurement by pressing *F9*.
- With the potentiometer, set the frequency of the carrier line to max. postion.
- Use a DC-signal with -10 V of the function generator as input signal.
- Feed the DC-signal into the input of the FM modulator.
- Determine the frequency f_C of the carrier from the spectrum.
- Enhance the DC voltage in steps of 1 V and repeat each time the measurement of the carrier frequency fc = fVCO
- Note all frequency values in the table.
- Draw the characteristic carrier frequency versus the control voltage of the VCO.

U_1/V	f_{VCO}/kHz
-10	
-9	
-8	
-7	
-6	
-5	
-4	
-3	
-2	
-1	
0	

1	
2	
3	
4	
5	
6	
7	
8	
9	
10	

- Determine the coefficient of the FM modulator:

$$k_{FM} = \frac{\Delta f_{VCO}}{\Delta U_1}$$

The coefficient of the FM modulator is given by the slope of the characteristic curve:

Determination of the frequency deviation

The maximum frequency deviation depends of the coefficient of the FM modulator and the amplitude of the modulating signal. In many experiments, we use an amplitude of the function generator of A_M = 20 V_{pp}. Calculate the frequency deviation:

$$\Delta F_{FM} = k_{FM} \cdot \frac{20V}{2}$$

Result

Measurements in the frequency domain of FM

The spectrum of FM

- Use a sinusoidal signal with A_M = 20 V_{pp} (= amplitude 10 V) and f_M = 300 Hz of the function generator as the modulating signal sM(t).

Hints: - The function generator displays peak to peak values. - Reset the DC offset to 0 V=.

- Feed the modulating signal into the input of the FM modulator.
- Sketch the graph of the FM spectrum.
- Interpret the result.
- Repeat the measurement of the FM spectrum for a modulating signal A_M = 20 V_{pp} and f_M = 200 Hz.

In FM, the carrier amplitude is determined by the Bessel function $J_0(\eta)$. This Bessel function possesses zeros for certain values of the modulation index η. A zero of $J_0(\eta)$ means that the spectral line of the carrier is missing, it's amplitude becomes zero. Zero crossings of $J_0(\eta)$ arise for the following modulation indices:

$\eta_1 = 2.4$ $\eta_2 = 5.5$ $\eta_3 = 8.8$

- In the FM spectrum sidelines appear every $n*f_M$ around the carrier line. n = 1,2,3…
- The amplitudes of the sidelines are determined by Bessel functions and depend on the modulation index η.
- Sideline spacing is 300 Hz.
- The spectrum of the FM signal is infinitely large. However, the most important part of the signal's energy is concentrated within the Carson bandwidth.
- The Carson bandwidth limits the FM spectrum with regard to the unmodulated carrier amplitude.
- Small errors made determining the modulation index η, have considerable influence in the Bessel functions.

2.4.2 Determining the carrier zero crossings by varying the frequency deviation ΔF_{FM}.

- Use a sinusoidal signal with A_M = 0 V_{pp} and f_M = 100 Hz of the function generator as the modulating signal $s_M(t)$.
- Feed the modulating signal into the input of the FM modulator.
- Start the measurement by pressing *F9*.
- Slowly enhance the amplitude of the modulating signal A_M of the function generator.

  Observe the decay of the carrier line.
- Note the values of the amplitudes A_M of the modulating signal, when the carrier line disappears. Hint: the function generator displays peak to peak values.
- Enter your measurements into the table and calculate the modulation index η.
- Sketch the graph of the FM spectrum for the first carrier zero crossing.

$$\eta_{FM} = \frac{K_{FM}\ A_M}{f_M} = \frac{K_{FM}}{100} \cdot \frac{v_{pp}}{2}$$

With A_M the amplitude of the modulating signal in V and a fix modulating frequency f_M = 100 Hz.

Observation

Measurement		Theory
A_M/V_{pp}	**η**	**η**

2.4.4 FM spectrum for square wave modulation

Determine the carrier zero crossings by varying the modulating frequency f_M.

- Use a sinusoidal signal with A_M = 20 V_{pp} (amplitude 10 V) and f_M = 500 Hz of the function generator as the modulating signal $s_M(t)$. Feed the modulating signal into the input of the FM modulator.
- Start the measurement by pressing *F9*.
- Slowly reduce the frequency of the modulating signal f_M of the function generator. Observe the decay of the carrier line.
- Sketch the graph of the FM spectrum for the first carrier zero crossing.
- Note the values of the frequencies of the modulating signal f_M when the carrier line disappears.
- Enter your measurements into the table and calculate the modulation index η.

$$\eta_{FM} = \frac{K_{FM}\,A_M}{f_M} = \frac{K_{FM}}{f_M} \cdot \frac{20}{2}$$

Observation

Measurement		**Theory**
f_M/Hz	η	η

FM spectrum for square wave modulation

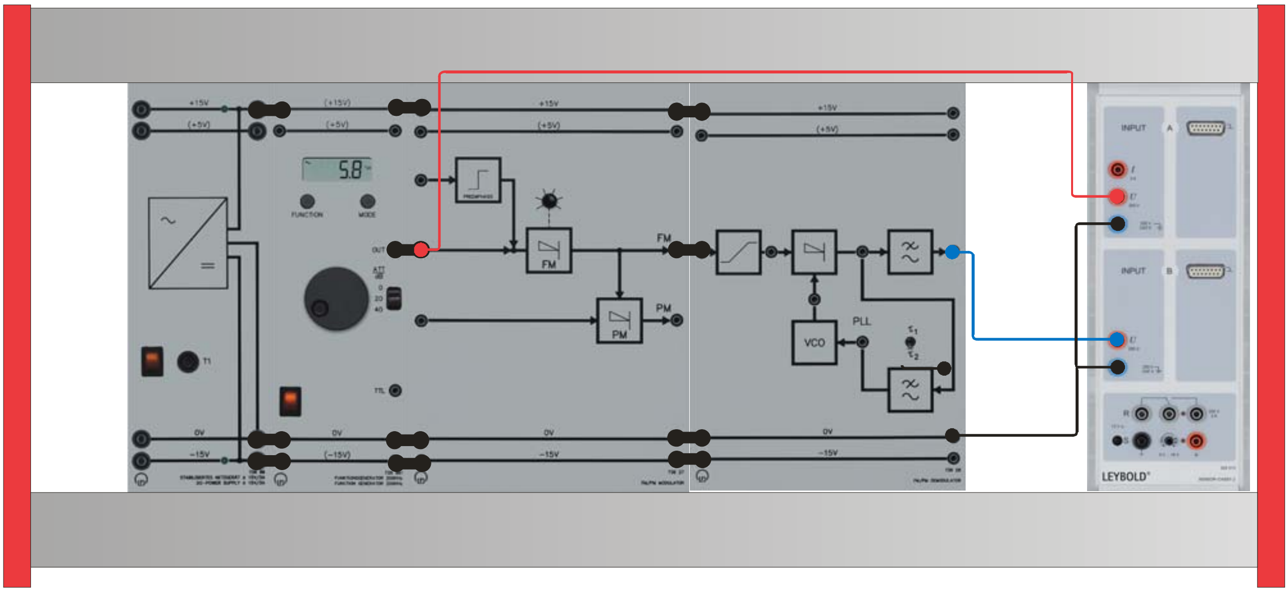
- Use a square wave signal with $A_M = 20\ V_{pp}$ (amplitude 10 V, duty cycle 50%) and $f_M = 200$ Hz of the function generator as the modulating signal $s_M(t)$.
- Feed the modulating signal into the input of the FM modulator.
- Start the measurement by pressing *F9*.
- Sketch the graph of the FM spectrum.
- Interpret the results.

- The spectrum of the FM is non linear.
- The FM spectrum is very extended. It contains a lot of sidelines.

FM demodulation

FM demodulation for loop filter τ_2

To carry out this experiment, assemble the components as shown below.



Time response of the FM modulating system

- Feed the modulating signal into the input of the FM modulator.
- Set the loop filter of the FM demodulator to τ_2.
- Use a sinusoidal modulating signal $s_M(t)$ with A_M = 10 V_{pp} and f_M = 500 Hz.
- Measure at the input of the FM modulator and at the output of the FM demodulator.
- Start the measurement by pressing *F9*.

Hint: If necessary, correct the frequency of the carrier line to f_C = 20.0 kHz (without modulating signal).

- Sketch the time responses of the modulating signal $s_M(t)$ and the demodulated signal $s_D(t)$.

- The amplitude of demodulated signal depends on frequency and loop filter setting τ_2.

Transfer characteristic of the FM modulating system

- Use a sinusoidal modulating signal $s_M(t)$ with A_M = 2 V_{pp} and initially f_M = 500 Hz. • Start the measurement by pressing *F9*.
- Determine the amplitude of the demodulated signal A_D with the spectrum analyser. • Note the value into the table.
- Enhance the frequency of the modulated signal in steps of 500 Hz and repeat the measurement.
- Sketch the results in a diagram.

f_M/Hz	A_D/V
500	
1000	
1500	
2000	
2500	
3000	
3500	
4000	
4500	
5000	

FM demodulation for loop filter τ_1

- Repeat the experiment.
- Sketch the time response of the FM modulating system for **τ_1.**
- Sketch the transfer characteristic of the FM modulating system for **τ_1.**

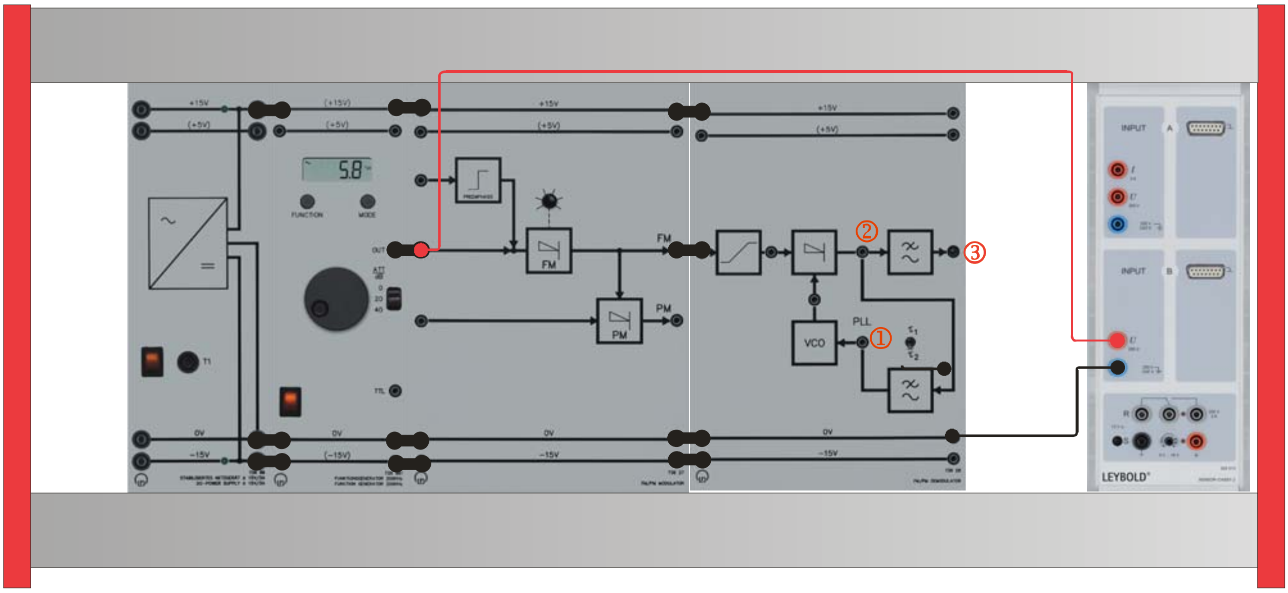
- The amplitude of demodulated signal depends on frequency and loop filter setting τ_1.

Transfer characteristic of the FM modulating system

f_M/Hz	**A_D/V**
500	
1000	
1500	
2000	
2500	
3000	
3500	
4000	
4500	
5000	

Reference Spectrum at the input of the FM modulator

Experiment set-up: Assemble the components as shown.



- Feed the modulating signal into the input of the FM modulator.
- Set the loop filter of the FM demodulator to τ_2.
- Set the function generator to a sinusoidal modulating signal $s_M(t)$ with $A_M = 5\ V_{pp}$ and $f_M = 1000$ Hz.
- Measure at the input of the FM modulator with channel B of the Sensor-CASSY2.
- Start the measurement by pressing *F9*.
- Sketch the spectrum of the modulating signal S_M.

Spectrum at the PLL demodulator, input of VCO

- Use the experimental set up of the “Reference Spectrum” .
- Set the loop filter of the FM demodulator to τ_2.
- Measure at the input of the VCO of the FM demodulator (①) with channel A of the Sensor-CASSY 2.

. • Start the measurement by pressing *F9*.

- Sketch the spectrum of the control signal of the VCO.

- Loop filter setting: **τ_2**
- The dc voltage tunes the VCO of the PLL circuit to the mean frequency of the carrier. The dc-components is strongly dependent of the settings of the carrier frequency on the FM modulator and may be subject to change individually.
- In the spectrum appears additionally a small line at the frequency of the modulating signal (here at f_M = 1 kHz).

Spectrum at the PLL-demodulator, output of the mixer stage

- Use the experimental set up of the “Reference Spectrum” .
- Measure at the output of the PLL circuit of the FM demodulator (②) with channel A of the Sensor-CASSY2.
- Start the measurement by pressing *F9*.
- Sketch the spectrum of the output signal of the PLL.

- The spectrum at the mixer output contains a lot of undesired spectral components.
- Using a suitable low pass filter, the undesired spectral components are suppressed.

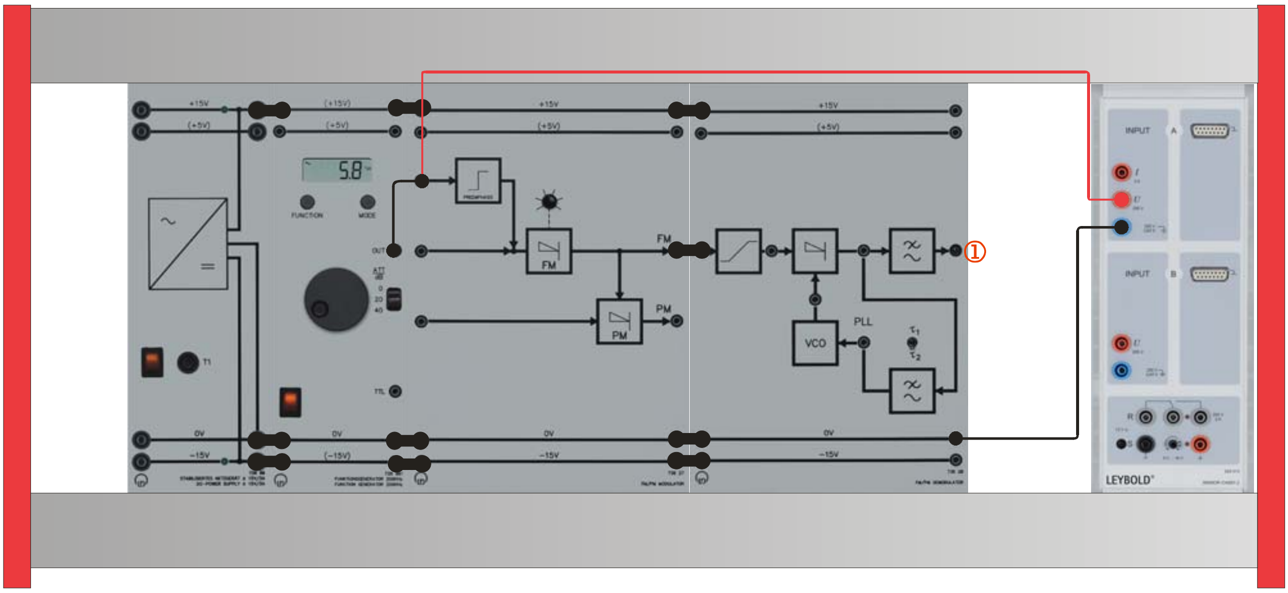
Spectrum at the output of the FM demodulator low pass filter

- Use the experimental set up of the reference measurement.
- Measure at the output of the low pass filter of the FM demodulator (③) with channel A of the Sensor-CASSY2.
- Start the measurement by pressing *F9*.
- Sketch the spectrum of the output signal of demodulator low pass.

- Loop filter τ_2
- At the output of the low pass filter, the spectral line of the modulated signal is measured.
- Undesired spectral components, due to the demodulation process in the PLL mixer, are suppressed.

FM preemphasis

Experiment set-up: Assemble the components as shown.



- Set the loop filter of the FM demodulator to τ_2.
- Set the function generator to a sinusoidal modulating signal $s_M(t)$ with A_M = 5 V_{pp} and initially f_M = 500 Hz.

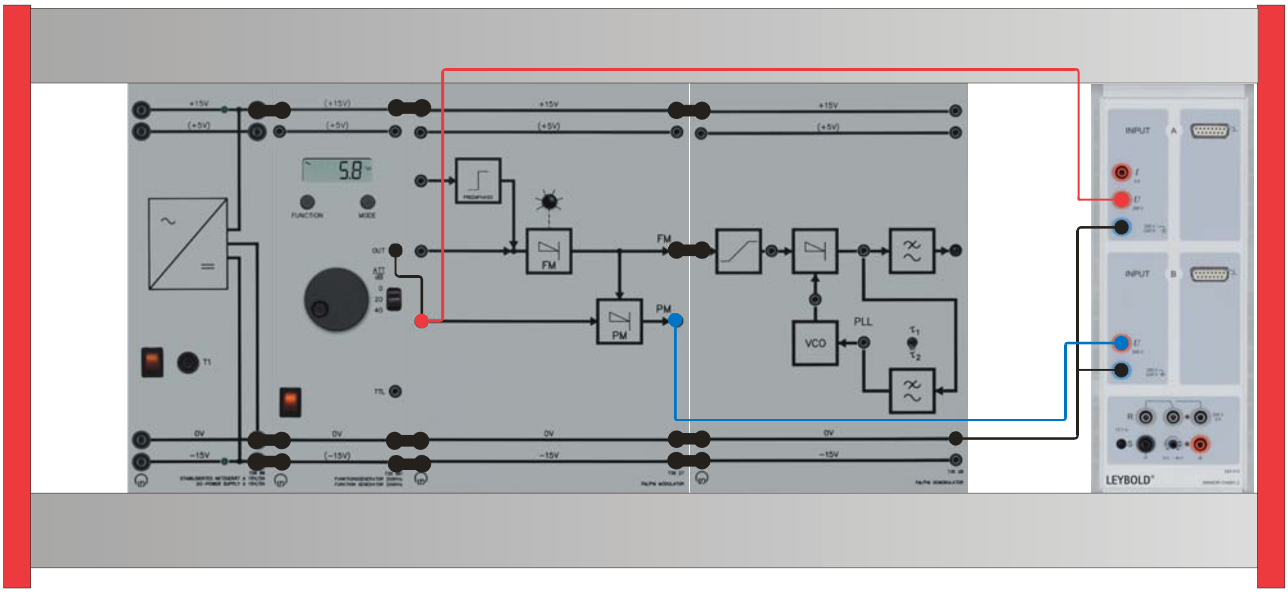
Time characteristic of the input signals

- Measure at the input of the preemphasis stage with channel A and the output of the FM demodulator low pass filter with channel B of the Sensor-CASSY2
- . Start the measurement by pressing *F9*.
- Sketch the time response of the modulating signal s_M(t) and demodulating signal sD(t)..
- Set the function generator sequentially to 1000 Hz and 2000 Hz.
- Sketch the time response of the input signal of the preemphasis stage and the output signal of the FM demodulator .

Phase modulation

Dynamic response of PM

Experiment set-up: Connect the components as shown below.



- Set the carrier frequency to max. position.

1.Use a square signal with AM = 20 Vpp and fM = 0.1 Hz of the function generator as the modulating signal sM(t). Feed the modulating signal into the input of the FM modulator,observe the time domain.

- 2. Set the function generator to a sinusoidal modulating signal $s_M(t)$ with A_M = 2 V_{pp} and f_M = 1000 Hz.
- Feed the modulating signal $s_M(t)$ into the input of the PM modulator.
- Measure the modulating signal with channel A of the Sensor-CASSY2.
- Measure the PM signal at the output of the PM modulator with channel B of the Sensor-CASSY2.

Start the measurement by pressing *F9*.

Interpret your results.

- The dynamic response (time characteristic) of the PM signal and the modulating signal are similar to FM.

3.2 Characteristic of the PM modulator

- Set the carrier frequency to max. position.
- Set the function generator to DC.
- Feed a DC voltage U_1 = -1.0 …+ 1.0V into the input of the PM modulator.
- Load the CASSY Lab 2 example PM_TD.labx.
- Start the measurement by pressing *F9*.
- Determine the phase shift ΔX/µs between the carrier oscillations of the FM and PM output. To evaluate your measurements right click Use Set Marker / Measure Difference of CASSY Lab2 (alt+D)
- Enter your results into the table and calculate the phase shift.
- Sketch the characteristic ΔX = f(U_1) of the phase modulator.
- Sketch the time characteristics for -1.0 V and +1.0 V.
- Interpret your results.

U_1/V	ΔX / µs	Degree /°
-1.0		
-0.9		
-0.8		
-0.7		
-0.6		
-0.5		
-0.4		
-0.3		
-0.2		
-0.1		
0.0		
0.1		
0.2		
0.3		
0.4		
0.5		
0.6		
0.7		
0.8		
0.9		
1.0		

$$\Delta\Phi = \frac{\Delta X}{T} = \frac{\Delta X}{50\mu s} \cdot 360°$$

Result

T = 50 µs for f_c =

The PM spectrum

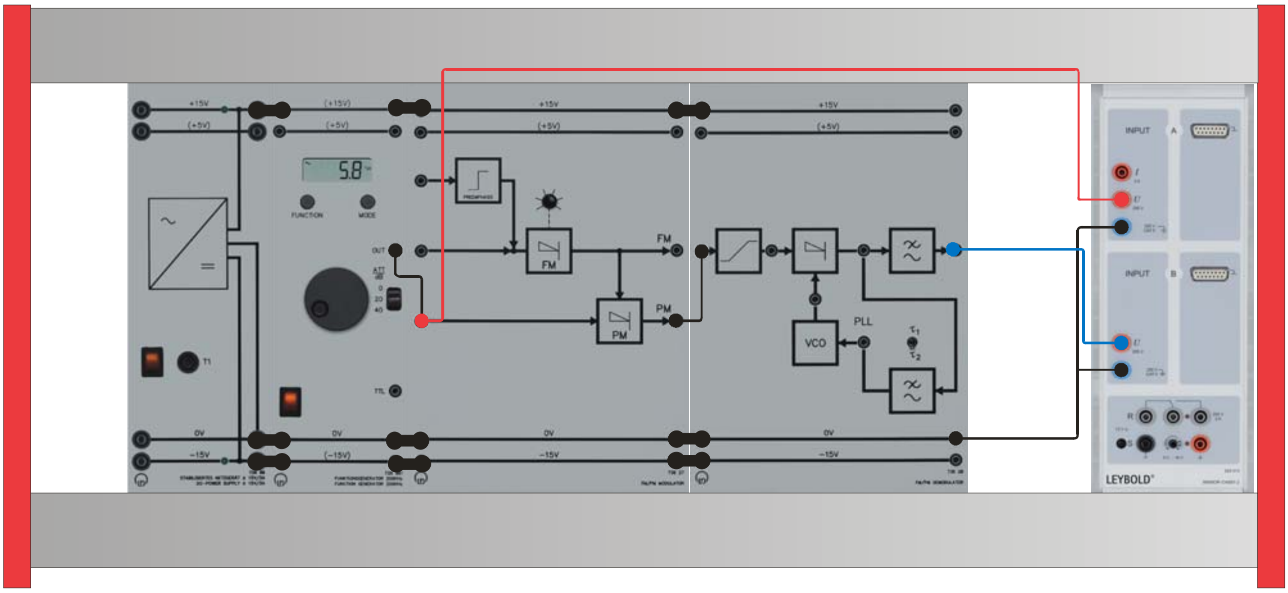
Use the experiment set-up in dynamic response of PM.

- Measure the spectrum at the PM modulator output with channel B of the Sensor-CASSY2.
- Start the measurement by pressing *F9*.
- Set the carrier frequency to 20.0 kHz with the potentiometer.
- Connect the function generator to the PM modulator input.
- Set the function generator to a sinusoidal modulating signal $s_M(t)$ with $A_M = 2\ V_{pp}$ and $f_M = 300$ Hz.
- Sketch the graph of the PM spectrum.
- Interpret the result.
- Repeat the measurement of the PM spectrum for a modulating signal $A_M = 2\ V_{pp}$ and $f_M = 200$ Hz.

PM demodulation

PM demodulation for loop filter τ_2

Experiment set-up: Connect the components as shown.Set the function generator to a sinusoidal modulating signal sM(t) with AM = 2 Vpp and fM = 1000 Hz .



- Set the toggle switch of the PM demodulator to $\boldsymbol{\tau_2}$.
- Start the measurement by pressing *F9*.
- Interpret your results.

PM demodulation for loop filter τ_1

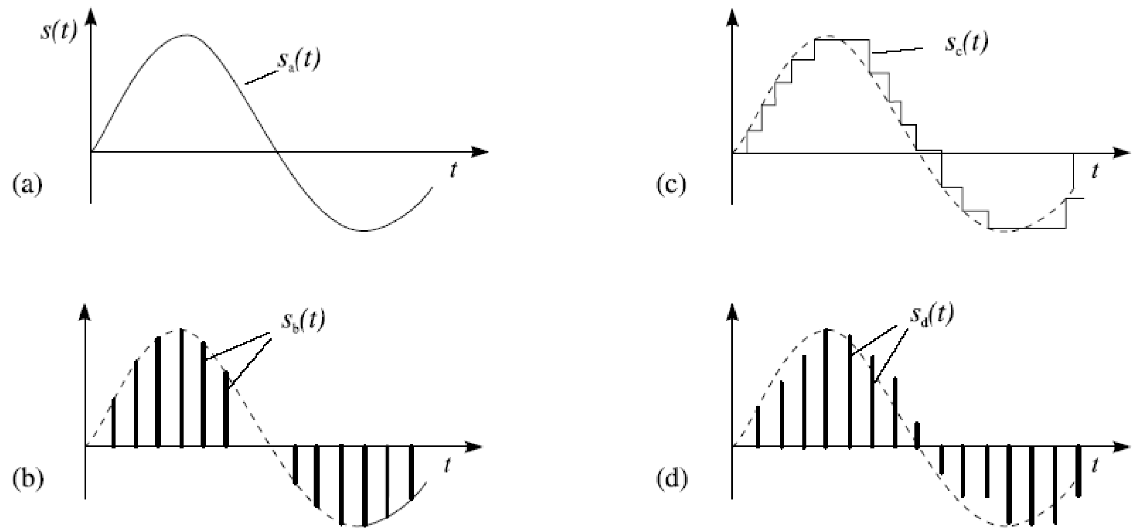
- Set the toggle switch of the PM demodulator to $\boldsymbol{\tau_1}$.
- Repeat the experiment.

Introduction

Signals

In electrical telecommunications engineering, messages are usually in the form of time-dependent electrical quantities, for example, voltage *u(t)* or current *i(t)*. These kinds of quantities which are described by time functions are called signals. In order to transmit messages a parameter of the electrical signals must be suitably influenced. Signals are distinguished according to the characteristic curves of their time and signal coordinates. If the signal function s(*t*) produces a signal value at any random point in time, the signal function is called **time-continuous** (continuous w.r.t. time). In contrast, if the signal has different signal values only at definite, countable points in time starting from 0, i.e. its time characteristic shows “gaps”, then this is referred to as **time-discrete** (discrete w.r.t. time). That which is true for the time coordinate, can also be applied to the signal coordinates. Accordingly, a signal is called **value-continuous**, if it can assume any given value within the modulation limits. It is called **value-discrete or n-level**, if only a finite number of signal values are permitted. Two important signal classes can be defined using these four terms:

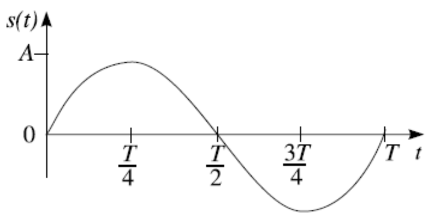
- **Analog signals**
  A signal is called analog if it is both time as well as value-continuous.
- **Digital signals**
  A signal is called digital, if it is both time as well as value-discrete.



Classification of signals
a: time- and level-continuous
b: time-discrete (sampled), level-continuous
c: time-continuous, level-discrete (quantized)
d: time- and level-discrete

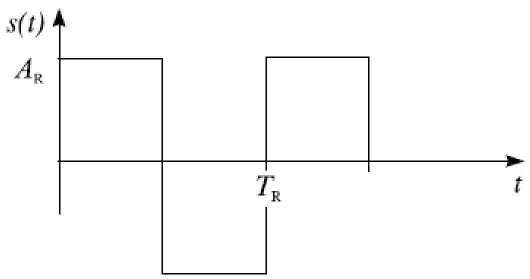
Time and spectral domain

In the technical sciences there exists, in addition to the “time domain”, signal representation in the frequency or spectral domain. Both representations are subsequently discussed.

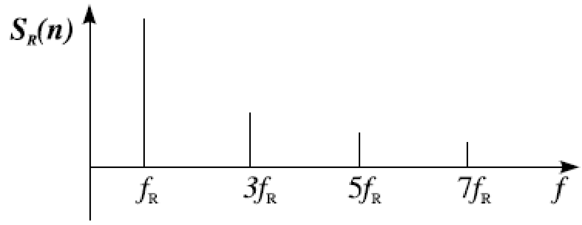


Harmonic function, time domain

Harmonic function, frequency domain



Symmetrical square wave, time domain



Symmetrical square wave, frequency domain

If you first consider the harmonic function, then a display on the oscilloscope results in the familiar, time characteristic. The sinusoidal time function is described by the amplitude A and the period duration T. However, a totally equivalent representation of this function is reproduced when the variables A and $f = 1/T$ are used instead of the parameters A and T. If the amplitude is displayed on the frequency axis, then this form of representation is called the **amplitude spectrum**. Thus, a single line can depict a harmonic function. According to Fourier every **non-harmonic**, **periodic function** can be represented as the superimposition of harmonic oscillations with fixed amplitudes ***S*(*n*)**. As an example a symmetrical square-wave signal with the amplitude A_R and the period of oscillation T_R is represented. It is seen from the corresponding amplitude spectrum ***SR*(*n*)** that the square-wave function is produced from the superimposition of (an infinite number of) harmonic oscillations. Their frequencies are odd numbered multiples of = $1/T_R$ and their amplitudes decrease as a function of the ordinal number *n.*

Harmonic	Frequency	Amplitude
1	$f_R=1/T_R$	$S_R(1)=4A_R/\pi$
2	$3f_R$	$1/3S_R(1)$
3	$5f_R$	$1/5S_R(1)$
4	$7f_R$	$1/7S_R(1)$
n	$(2n-1)f_R$	$1/(2n-1)S_R(1)$

n=1,2,3,4,...

Modulation

When speaking of modulation, one generally refers to the conversion of a modulation signal $s_M(t)$ into a time function with altered characteristics using a carrier signal. The message signal influences a parameter of the carrier in a suitable fashion. Either harmonic oscillations or pulse trains are used as carrier signals. **Digital modulations** work with pulse-shaped carriers. The modulation spectra arising are extremely extended. Frequently, the message signal is still received in its original frequency band. In these cases a simple low-pass demodulation can be carried out on the pulse-modulated signals. A modulation always requires that the carrier and the modulation signal interact. Both of these signals are fed into a **modulator**. In the analog procedure it suffices to have one element with multiplying characteristics. In digital methods the signal is sampled (time-discrete). The sampling is performed by a “switch”, which also possesses modulating characteristics. The original signal $s_M(t)$ is recovered from the modulated signal through demodulation. Consequently, modulation and demodulation are mutually related, inverse processes. The complexities involved in modulation and demodulation are considerable. The following reasons explain why modulation is worthwhile:

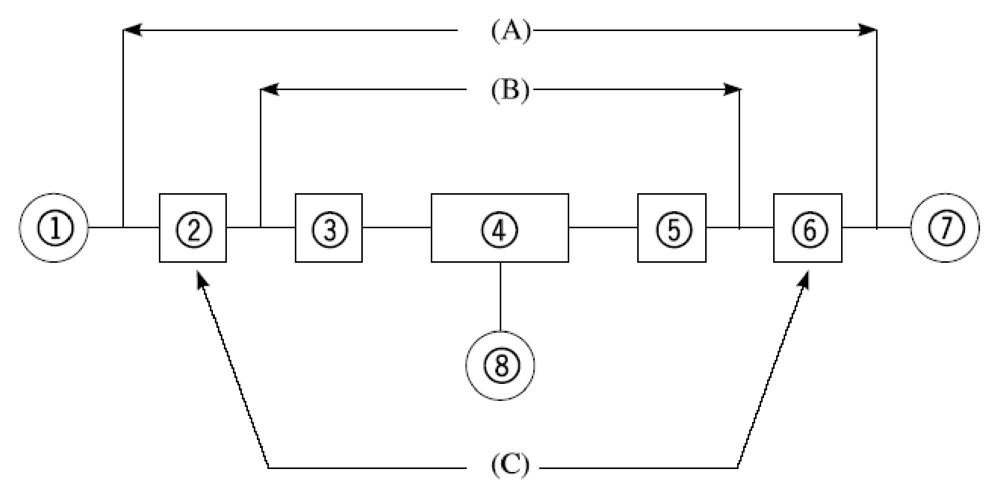
- Modulation enables the matching of the modulating signal to the characteristics of the transmission channel. (radio links e.g. are only possible for a certain frequency band.)
- Existing transmission channels can be multiply exploited using modulation, (frequency or time division multiplex systems).
- Improved signal-to-noise ratios can be obtained using modulation.

The communications system according Shannon

Electrical communications engineering is divided into three classical sub functions:

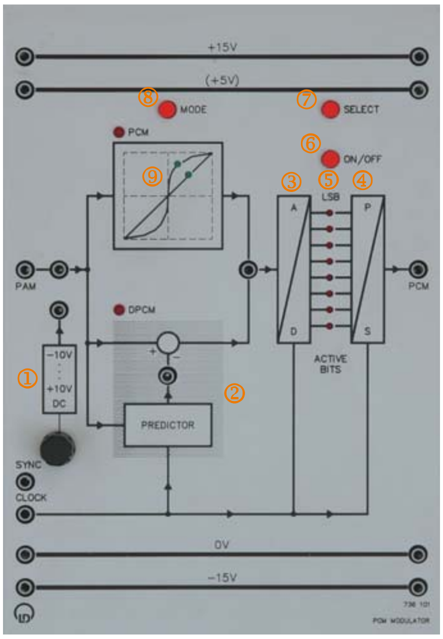
- Transmission of the message
- Processing of the message
- Telephone exchange technology

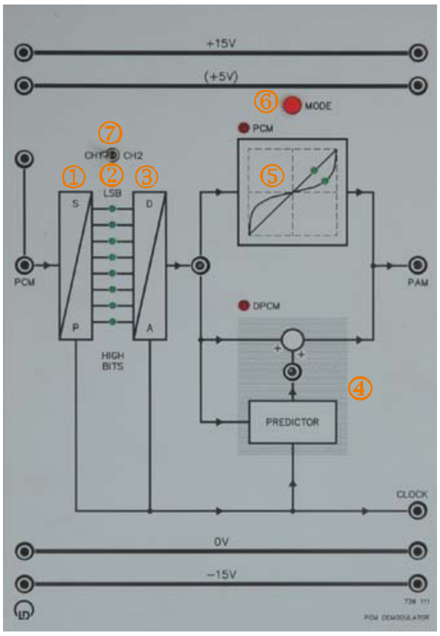
If a single transmission channel is considered, (i.e. no telephone technology), then we can concentrate on the remaining functions illustrated by the following scheme.



(A) The telecommunications system
(B) Message transmission
(C) Message processing
1 Message source (human being, measurement sensor etc.)
2 Converter (microphone, television camera, strain gauges, thermo sensor etc.)
3 Transmitter
4 Transmission channel (radio link, transmission cable, data storage system)
5 Receiver
6 Converter
7 Message recipient
8 Interference source

The telecommunication system (A) consists of equipment used for message transmission (B) and message processing (C). The message source (1) generates the information, which is to be made available to the message recipient (7). The signals generated are of different physical nature, e.g. sound, light, pressure, temperature. It is the function of the converter (2) to convert the non-electrical signal of the source into an electrical one. The transmitter (3) converts the converter signal into one better suited for transmission via the channel. Thus the modulation process takes place in (3). The transmission channel (4) serves either to bridge a spatial distance, or to overcome a period of time. The modulated signal, generally distorted by the interference source (8), reaches the receiver (5), where it is then reconverted into its original electrical signal there (demodulation). Finally, the converter (6) transforms the electrical signal back into the physical signal required by the message recipient (7). The message recipient can take the form of the human being with eyes and ears or a machine in a process control loop.





PCM-Modulator

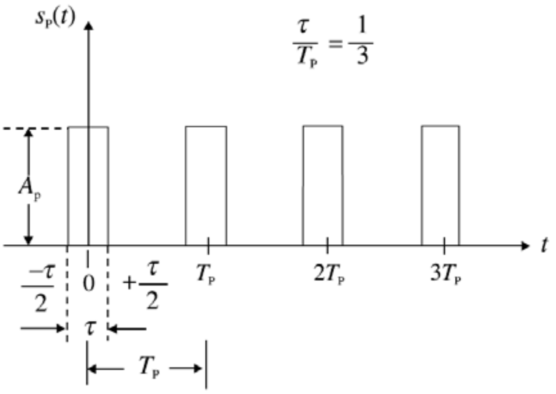
1. DC voltage source (10-turn potentiometer) for static modulation experiments
2. Predictor module to form prediction value in DPCM operating mode
3. A/D converter (8 bit, coding is carried out according to magnitude and polarity)
4. Parallel/serial converter. The data stream at the output of the PCM modulator contains all the synchronization signals needed for the receiver.
5. LEDs to display the selected bits
6. Pushbutton for on/off switching of a selected bit (permits introduction of bit errors, or the artificial reduction of the resolution).
7. Pushbutton for the selection of a bit. With each press of the button the selection changes to the next higher bit. The selected bit flashes for approx. 2s.
8. Pushbutton for the selection of the operating mode. Switching sequence: PCM linear quantization, PCM nonlinear quantization, DPCM. The selection is indicated via the LEDs
9. Compressor with display of the 13-segment characteristic

PCM-Demodulator

1. Serial/parallel converter
2. LEDs for the display of the higher order bits (investigations on binary coding with DC voltage source of the PCM modulator)
3. D/A converter (8 bit, encoding carried out according to magnitude and polarity)
4. Predictor module to form the prediction value in the DPCM operating mode
5. Expander with 13-segment characteristic
6. Pushbutton for operating mode selection. Switching sequence: PCM linear quantization, PCM non-linear quantization, DPCM. Selection is indicated via the LEDs.
7. Toggle switch to change display of the higher order bits from channel 1 to channel 2.

Pulse train

Theory



Time domain representation of a pulse train.

A_P	pulse amplitude
τ	Pulse duration
T_P	Pulse period

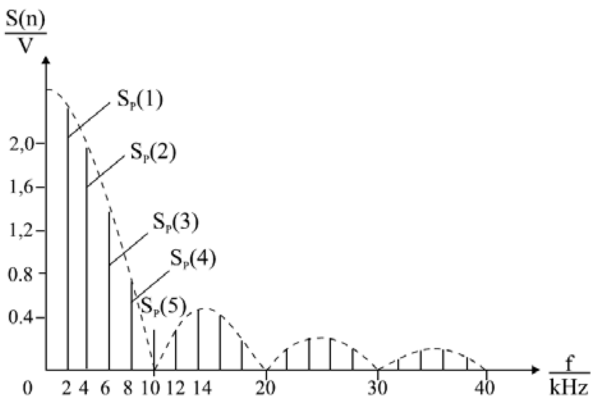
Instead of the pulse period T_P, its reciprocal value, the pulse frequency $f_P = 1/T_P$ can also be equally employed.

$$\frac{\tau}{T_P} = \tau \cdot f_P$$

The ratio between the pulse duration τ and pulse period T_P determines the duty cycle τ/T_P.

$$S_P(n) = 2A_P \frac{\tau}{T_P} \frac{\sin(\pi\tau n f_P)}{\pi\tau n f_P}$$

Amplitude spectrum and time characteristic are closely related. The amplitudes of the harmonics are proportional to the pulse amplitude A_P and the duty cycle τ/T_P.



As a periodic signal with respect to time the pulse train possesses a discrete line spectrum. Characteristic for the amplitude spectrum are the zero crossings arising in the envelope curve. The example shows the spectrum with envelope curve for a pulse train with the parameters: A_P=6V, τ/T_P=2/10, f_P=2kHz.

Digital modulations use pulse trains as carriers. That is why they are called pulse modulations. Examples include: PAM; PDM; PPM; PCM. Each pulse train is unmistakably characterized by the following 3 parameters:

- Pulse amplitude A_P (peak/peak value).
- Pulse period T_P (pulse frame).
- Pulse duration τ (pulse width).

In order to understand various forms of pulse modulation it is important to examine the spectral peculiarities of pulse trains. The pulse function $s_P(t)$ can be subjected to Fourier series expansion.

$$s_P(t) = A_P \frac{\tau}{T_P} + \sum_{n=1}^{\infty} S_P(n)\cos(2\pi n f_P t)$$

Fourier series expansion describes the pulse train as the superpositioning of an infinite number of cosine oscillations, whose frequencies are integer multiples of the pulse frequency f_P. Furthermore, there is a DC component $A_P*\tau/T_P$ present. Each of the cosine oscillations has a precisely defined amplitude $S_P(n)$. The following table reproduces the calculated amplitude values and the frequencies of the spectral lines.

Table: Pulse spectrum

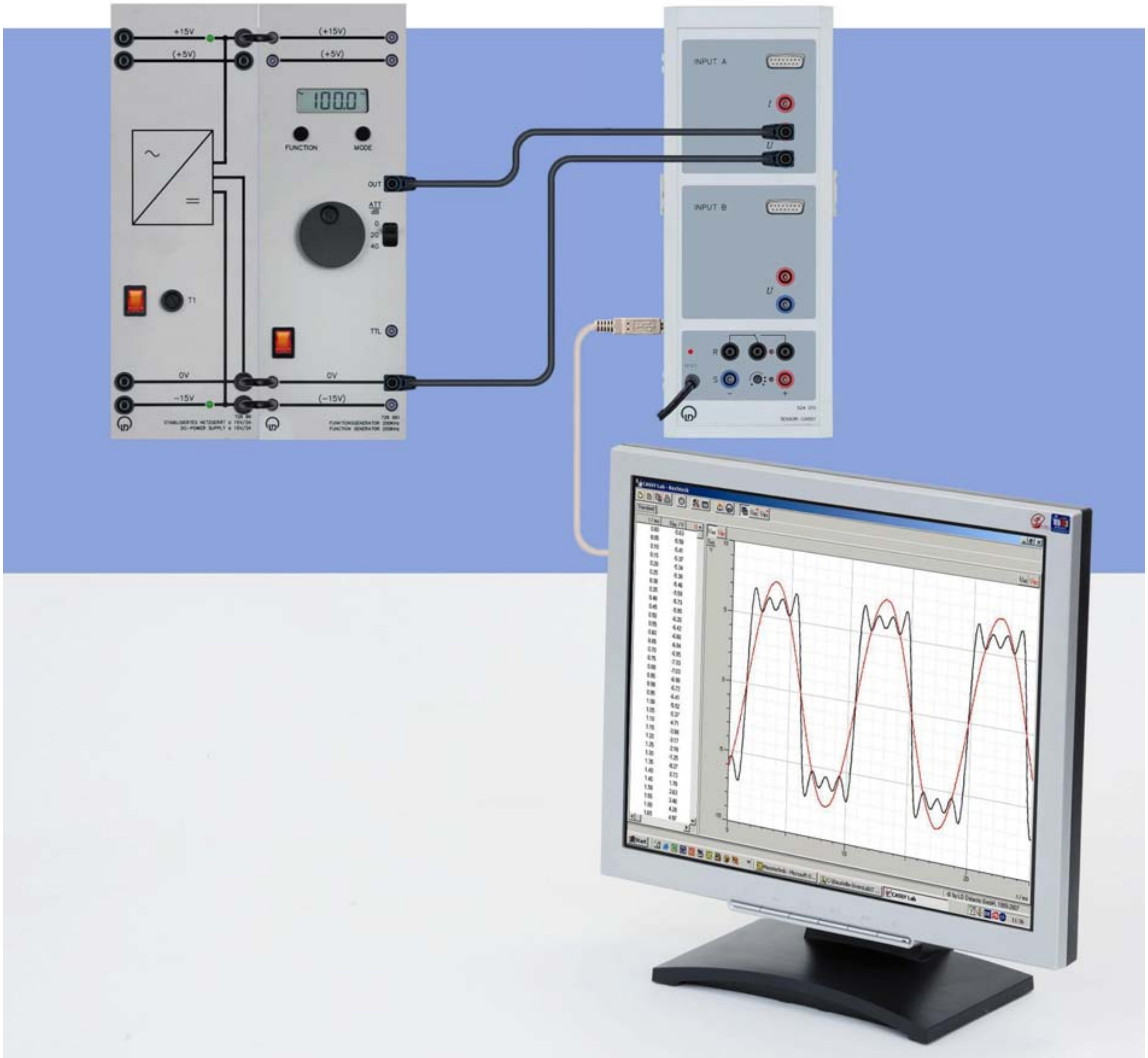
A_P=6V, τ/T_P=0.2, f_P=2kHz

n	f/kHz	S_P(n)/V
1	2	2.25
2	4	1.82
3	6	1.21
4	8	0.56
5	10	0.00
6	12	0.38
7	14	0.52
8	16	0.45
9	18	0.25
10	20	0.00
11	22	0.20
12	24	0.30
13	26	0.28
14	28	0.16
15	30	0.00
16	32	0.14
17	34	0.21
18	36	0.20
19	38	0.12
20	40	0.00

Material

1	524 013S	Sensor CASSY 2 Starter
1	726 961	Function generator 200 kHz
1	726 86	Stabilized power supply ±15 V, 3 A
1	726 09	Panel frame T130, two level
1	501 461	Pair of cables 100 cm, black
1	501 511	Set of bridging plugs, black
1	564 002	Book: Pulse Code Modulation
1		PC

Carrying out the experiment



- Set up the shown experiment.
- Select a pulse train at the function generator with f_P = 1 kHz, pulse amplitude A_P = 5 V (10 V_{pp}) and duty cycle τ_1/T_P = 1/10.

- Start the measurement by pressing *F9*
- Determine the time characteristic of the pulse train.
- Determine the spectrum of the pulse train.

Where are in general the zero crossings in the envelope of the pulse spectrum?

- How many spectral lines *l* arise between two zero crossings of the envelope (sync-function)?

- Repeat the measurement of the spectra and time characteristics for the same pulse frequency f_P = 1 kHz and pulse amplitude A_P for different duty cycles τ_4/T_P = 4/10, and τ_6/T_P = 9/10. Proceed as described above.

- Why do pulse trains require large transmission bandwidths?
- What is the structure of the spectrum of a pulse train?
- What kind of characteristic curve is the envelope curve of the pulse spectrum?

Results

Time characteristic of the pulse train
A_P = 5 V, f_P = 1 kHz, τ_1 = 1/10

FFT spectrum of the pulse train
A_P = 5 V, f_P = 1 kHz, τ_1 = 1/10
Number of lines in each sub spectrum: l =
1. zero crossing of the sync-function:

Time characteristic of the pulse train
A_P = 5 V, f_P = 1 kHz, τ_2/T_P = 4/10

FFT spectrum of the pulse train
A_P = 5 V, f_P = 1 kHz, τ_2/T_P =4/10
Number of lines in each sub spectrum: l =
1. zero crossing of the sync-function:

Time characteristic of the pulse train
A_P = 5 V, f_P = 1 kHz, τ_3/T_P = 9/10

FFT spectrum of the pulse train
A_P = 5 V, f_P = 1 kHz, τ_3/T_P = 9/10
Number of lines in each sub spectrum: l =
1. zero crossing of the sync-function:

Summary

The spectrum of the periodic pulse train consists of discrete lines. The amplitudes of the spectral lines are limited by an envelope, which is described by the sync-function:

$$\frac{\sin(\pi\tau f)}{\pi\tau f}$$

This function has zero crossings at:

$$\sin(\pi\tau f) = 0 \qquad \pi\tau f = m\pi \qquad \text{with m = 1,2,3}$$

Zero crossing m appears at the frequency:

$$f = \frac{m}{\tau} = f_{0m}$$

Expressed by τ/T_P and f_P, the zero crossings appear at:

$$f_{0m} = m f_P \frac{T_P}{\tau}$$

	f_P = 1 kHz	f_P = 2 kHz	f_P = 3 kHz
τ/T_P	**f_{01}/kHz**	**f_{02}/kHz**	**f_{03}/kHz**
1/10			
2/10			
3/10			
4/10			
5/10			

For narrow pulses with a small pulse period τ the position of the 1st zero point is shifted towards higher frequencies. In principle one would have to use infinite broadband channels for the transmission of pulses. These kinds of channels do not exist. For any acceptable pulse transmission a bandwidth is needed which reaches at least the 1st zero point of the pulse spectrum. This frequency lies at $f_{01} = 1/\tau T$. The narrower the pulse, the larger the bandwidths required for transmission.

The number of spectral lines between two zero points of the envelope curve is given by the natural number *l* for which the following holds true:

$$l \leq \frac{T_P}{\tau}$$

This means that for τ_1/T_P = 1/10 there are *l* = 9 lines between the two zero points. The 10th line has 0 amplitude, i.e. it coincides with the zero point.

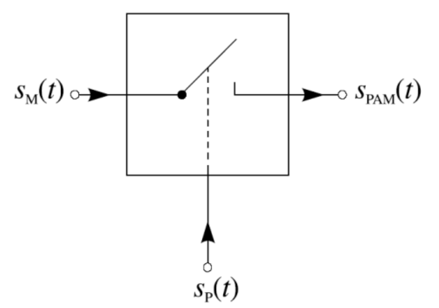
Reducing the duty cycle means:

- The amplitudes of the harmonics are getting smaller.
- The position of the zero crossings in the envelope is shifted towards higher frequencies.
- The number of spectral lines between two zero crossings increases.

The spectra for τ_1/T_P = 1/10 and τ_6/T_P = 9/10 are equal. The corresponding time signals only differ by a DC-component or a signal inversion. Both will not be represented in the spectrum.

Pulse amplitude modulation (PAM)

Theory



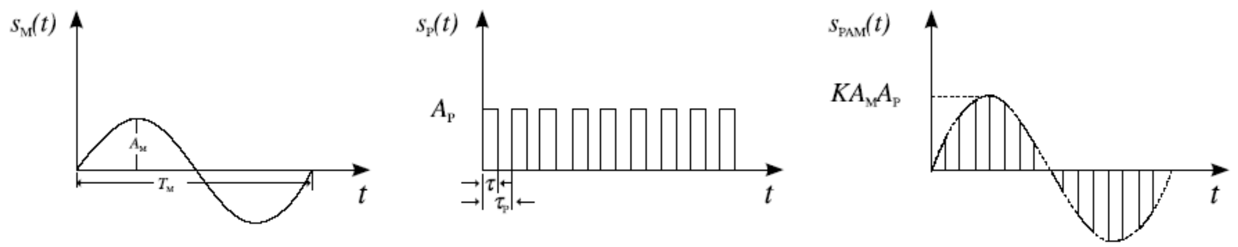
PAM means modulation by a switch. The conversion of the analog signal into a digital signal requires three steps:

- sampling
- quantization
- coding

Sampling gives the PAM signal. Using an electronic switch, which is triggered by a pulse train $s_P(t)$, the signal $s_M(t)$ present at the input is chopped into pulses with the width τ. This process is called time discretization. The PAM signal only arises at definite, discrete times. It is zero in the pulse intervals. Thus the following applies:

- The PAM signal is time-discrete and value continuous.
- The PAM signal is neither analog nor digital.

PAM is not suitable as a transmission method because it is very prone to distortion due to the nature of value-continuous signals. PAM achieves practical importance as an intermediate stage in the generation of many other kinds of pulse modulations. The figure shows how PAM modulation is produced for the special case of a harmonic input signal $s_M(t)$.



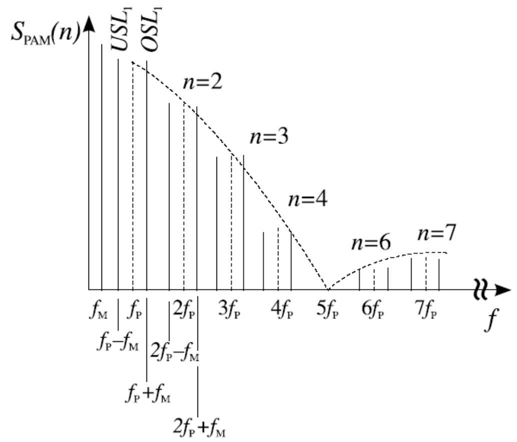
Generation of PAM

The PAM signal consists of pulses which have the curve of the input signal. The multiplication of $s_M(t)$ with $s_P(t)$ supplies the PAM signal $s_{PAM}(t)$ with the exception of constant factor *k*:

$$s_{PAM}(t) = k \cdot s_M(t) s_P(t)$$

In this case we are dealing with a bipolar PAM because both positive as well as negative signal values can arise. By superpositioning a DC voltage on the modulating signal *s*M(*t*) a unipolar PAM can be produced. The pulse amplitude of the switching pulse train has no effect at all on the PAM signal.

PAM spectrum



The time characteristic of the PAM signal represented in Fourier expansion is given by:

$$s_{PAM}(t) = A_M \frac{\tau}{T_P}\cos(2\pi f_M t) + \sum_{n=1}^{\infty} A_M \frac{\tau}{T_P} \frac{\sin(n\pi \frac{\tau}{T_P})}{n\pi \frac{\tau}{T_P}} \cos\left[2\pi(nf_P \pm f_M)t\right]$$

When modulation is performed with a cosine input signal, the spectrum of PAM contains an infinite number of harmonic oscillations. These lines group themselves in pairs around the suppressed carrier lines as lower and upper side lines ($nf_P \pm f_M$). The modulating signal $s_M(t)$) evaluated with the factor τ/T_P also occurs in the spectrum. For the amplitudes of the nth sub spectrum the following holds true:

$$S_{PAM} = A_M \frac{\tau}{T_P} \frac{\sin(\pi \cdot \tau \cdot nf_P)}{\pi \cdot \tau \cdot nf_P}$$

Comparison of the PAM signal with the Fourier expansion of the pulse train gives:

- Instead of the direct component $A_P \tau/T_P$ in the pulse train, in PAM the modulating signal $s_M(t)$ evaluated with the pulse-duty factor τ/T_P appears in the original frequency position. For that reason the input signal can be recovered through simple low pass filtering of the PAM signal (low pass demodulation).
- In the case of bipolar PAM there are no more carrier lines. The upper and lower sidelines USL, LSL are produced. This is similar to double sideband amplitude modulation without carrier.

Characteristic for the PAM spectrum is (at the pulse frequency f_P) the periodic repetition of the spectrum of the modulating signal $s_M(t)$. If the signal frequency f_M is increased at a constant pulse frequency f_P, then the sidelines of all the subspectra are shifted further away from their suppressed carriers. With $f_M = f_P/2$, the respective lower sidelines of the subspectra $n+1$ and the upper sidelines of the sub spectrum n coincide. If f_M is increased still further, then the subspectra even overlap! A low pass demodulation is now impossible. Not only the desired spectral components of the modulating signal but also the spectral lines of higher subspectra will pass through the filter's pass band (aliasing).

Sampling theorem

In order to avoid aliasing the following must hold true for the **sampling rate**:

$f_P > 2f_M$

Then at least two sampling values (samples) are apportioned to each period of the input signal. The receiver can then reconstruct the input signal $s_M(t)$ completely from these two samples per period.

The regulations described here for the time discretization of signals are normally summarized in the form of Shannon´s **sampling theorem**:

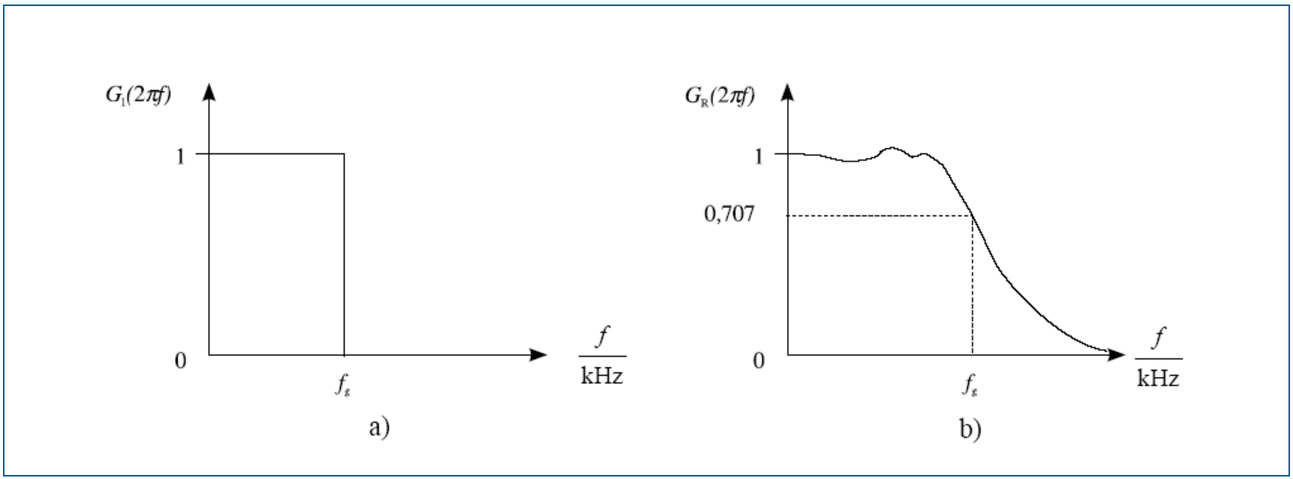
- Each time function limited to a particular bandwidth is specifically determined by its discrete sampling values if the sampling process supplies at least two samples per oscillation at the maximum occurring signal frequency.

The following concepts are useful with respect to the sampling rate f_P:

- Over sampling $f_P > 2f_S$. In the case of over sampling the reconstruction of the modulating signal $s_M(t)$ in the receiver is possible using a real low pass filter.
- Sampling with the Nyquist rate $f_P = 2f_M$. Demodulation is only possible with an ideal low pass filter with infinitely steep edges, (theoretical limiting case).
- Subsampling $f_P < 2f_M$. In the case of undersampling aliasing arises during reconstruction of the signal $s_M(t$ on the receiver side.

Aliasing

In real systems undersampling is avoided by employing bandwidth limitation. For economic reasons a financially feasible compromise is sought after for the sampling rate and thus for the highest signal frequency to be transmitted. In communications engineering it is standard procedure to limit the desired signal to the frequency range from 300 Hz to 3.4 kHz. According to Shannon a pulse frequency of $_{Pmin} = 2{*}f_{Mmax} = 6{,}8\ kHz$ is needed when sampling with the Nyquist rate. For demodulation an ideal low pass filter would have to be available.

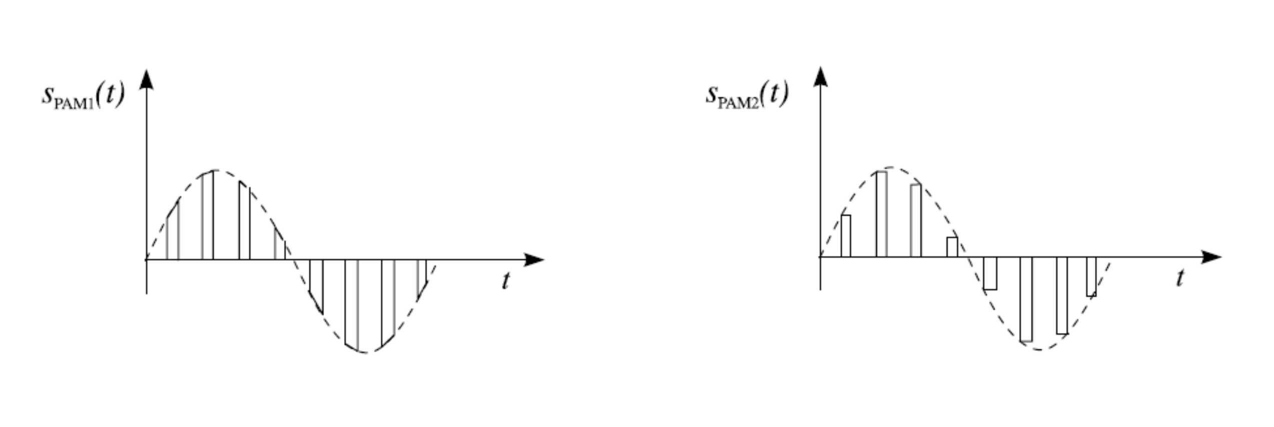


Amplitude response of low pass filters: a = ideal, b = real

Due to the finite slope steepness of real low pass filters it is impossible to carry out sampling with the Nyquist rate. For that reason commercial systems operate with a slight oversampling rate at the upper frequency limit of $f_{Mmax} = 3{,}4\ kHz$. They use a pulse frequency of $f_P = 8\ kH$.

Another form of PAM

PAM, which is generated solely with an electronic switch, is described by the multiplication of the modulating signal $s_M(t)$ with the pulse train $s_P(t)$. The curve of the original signal is contained in the pulse amplitudes of this kind of PAM_1 signal. Another type of the PAM is obtained if the curved pulse is converted into square-wave pulses with variable amplitude. The generation of this kind of PAM is performed using sampe & hold circuits (S&H).



Time characteristics of *PAM_1* and *PAM_2*

The varied time curve for the PAM_2 influences the corresponding spectrum.

Comparison of both types of PAM

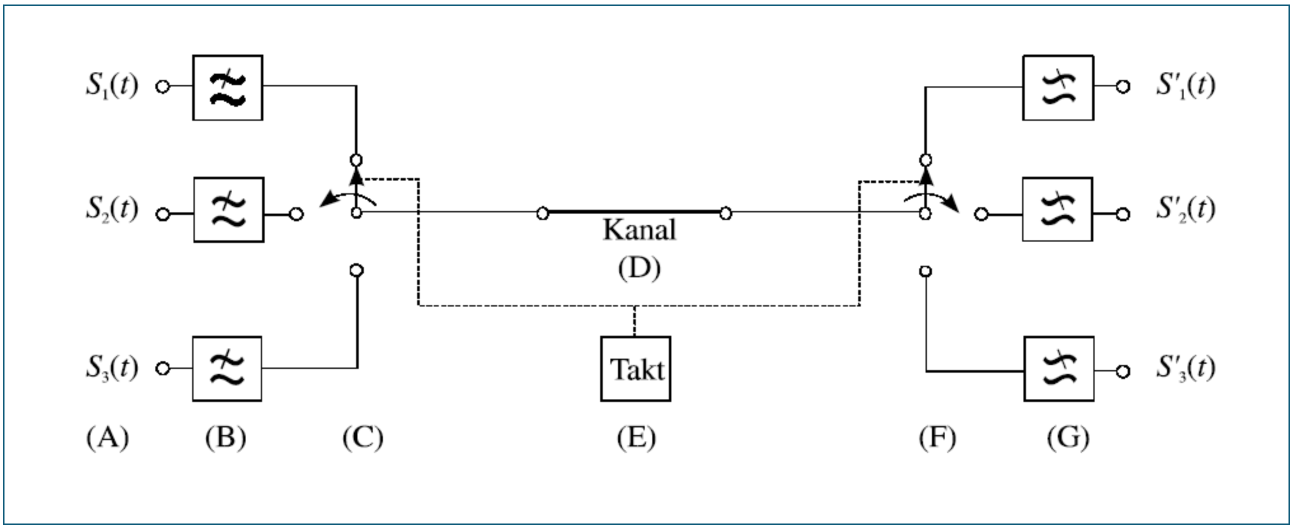
- Both PAM types contain the modulating signal $s_M(t) = A_M \cos(2\pi f_M t$. In PAM_1 this spectral component is only evaluated with the constant factor τ/T_P P. In the case of PAM_2 there is an additional evaluation factor $si(\pi f_M T_0)$. This brings about an additional, signal frequency dependent attenuation.
- In both types of PAM an infinitely extended line spectrum arises with sidelines for the frequencies $f = nf_P +/- f_M$.
- Both types of PAM are bipolar and thus suppress the carrier lines. Upper and lower sidelines are produced, which are evaluated with signal-frequency independent factors for the case of PAM_1. The PAM_2 also shows a signal-frequency dependent attenuation in the sidelines, which rises with increasing signal frequency f_M. In contrast to PAM_1, PAM_2 is distorted linearly.

Benefits of PAM_2

- Possibility of increasing the pulse-duty factor in the receiver. The shorter the pulse duration τ of pulse trains, the more communication channels can be accommodated by a single pulse frame of the duration T_P. If the aim is to bundle many channels in time-multiplex transmission, then pulses are needed with a small duty cycle τ/T. The amplitudes of the demodulated signals are also proportional to the duty cycle. However, the advantage to maintain many channels over one transmission link at a low τ/T is offset by the disadvantage of smaller receiving amplitudes. And it is precisely this disadvantage which can be corrected with PAM_2 by increasing the duty cycle on the receiver.
- Triggering the AD converter for PCM. The square-wave pulses being applied at the output of the S&H stage all have a time constant pulse amplitude, which is dependent on the instantaneous value of the modulating signal. This is the prerequisite for subsequent A/D conversion. PAM_2 is thus the precursor to PCM.

Time division multiplex

During the sampling of a signal time gaps arise in which no information is transmitted on the transmission channel. The time between any two samples of a signal source can be used to transmit information from other sources. By time shifting the samples of the different sources and placing them onto the transmission line in interleaved form a multiple exploitation of the transmission channel is obtained, known as time division multiplexing (TDM). The principle of TDM can be schematically depicted by two redating switches.



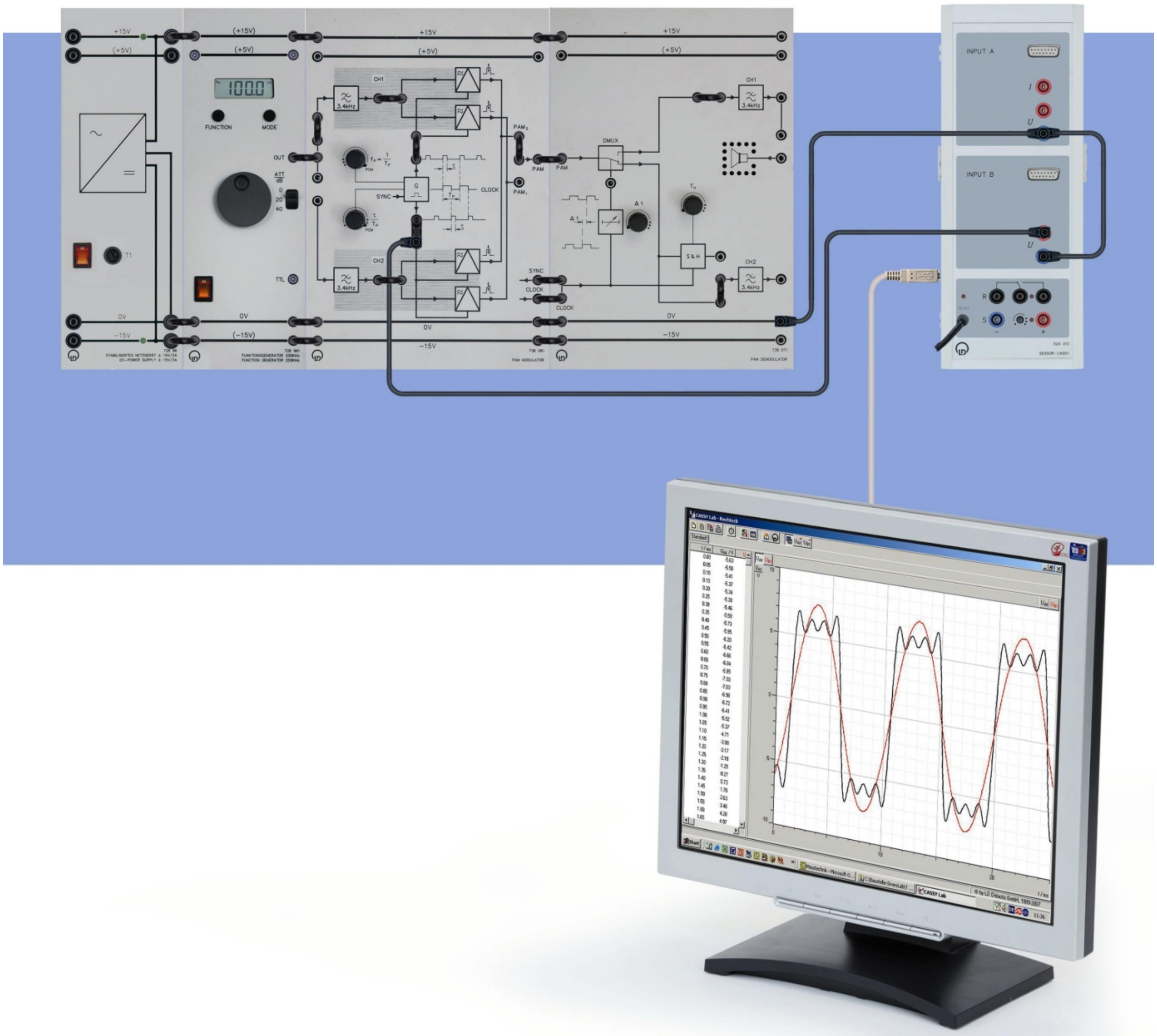
Principle of TDM with PAM
A: Signal sources
B: Band limiting filter
C: Multiplexer
D: Transmission channel
E: Synchronization clock
F: Demultiplexer
G: Low pass demodulator

The switch *C* on the transmitter side is called the multiplexer. It connects in sequence all the *n* sources to channel *D*. At the end of the transmission channel there is another redating switch *F*, the demultiplexer, which distributes the incoming samples to the *n*-receiver. Obviously both switches have to operate synchronously. Otherwise the messages will reach the wrong receiver, resulting in channel cross talk. The switching cycle of the multiplexer/ demultiplexer defines the pulse frame.

Material

1	736 061	PAM Modulator
1	736 071	PAM Demodulator
1	524 013S	Sensor CASSY 2 Starter
2	726 961	Function generator 200 kHz
1	726 86	Stabilized power supply ±15 V, 3 A
1	726 09	Panel frame T130, two level
2	501 461	Pair of cables 100 cm, black
1	501 441	Pair of cables 25 cm, black
2	501 511	Set of bridging plugs, black
1	501 512	Set of bridging plugs with tap, black
1		PC

Experiment set up



Adjusting the sampling frequency

- The sampling frequency f_P is set using the FFT analyzer. For that purpose set the PAM modulator:
  Controller for duty cycle $\tau/T_P \rightarrow$ PCM
  Controller for sampling frequency $f_P \rightarrow$ ●●● (max)
  CASSY UB1 $\rightarrow$ Clock generator *G*.
- Start the measurement by pressing *F9*
- Now slowly adjust the pulse frequency f_P, until the spectral line of the fundamental mode appears at f_0 = 5000 Hz ($3f_0$ = 15 kHz, etc). Don't change the sampling (pulse) frequency f_P anymore.

Time characteristic of the PAM

Measure the Input and output of the channel filter CH1.

- Function generator: Sine, 500 Hz, A = 10 Vpp.
- CASSY UA1 $\rightarrow$ Input of channel filter CH1.
- CASSY UB1 $\rightarrow$ Output of channel filter CH1.
- Start the measurement by pressing *F9.*

Display the time characteristic of the PAM.

- Function generator: Sine, 500 Hz, A = 10 Vpp.
- CASSY UA1 $\rightarrow$ Input PAM Modulator channel CH1.
- CASSY UB1 $\rightarrow$ Output PAM_1.
- Start the measurement by pressing *F9.*
- Repeat the measurement at the output PAM_2.

Measure the modulating signal $s_M(t)$ and the demodulated signal $s_D(t)$ as a function of the duty cycle.

- Now: Controller for the sampling frequency $f_P \rightarrow$ ●●● (max)
- Adjusting the duty cycle: (approximation :min 10 % , max 50%)
  CASSY UB1 $\rightarrow$ Clock generator *G*.
  . Start the measurement by pressing *F9.*
- Slowly readjust the duty cycle τ/T_P, until the display of the CASSY instrument shows τ/T = 50 %. Eventually correct the display, for that make a right click into the instrument *Duty Cycle* and match the factor *1.1* to your special situation. For the maximum position (PCM) is true: τ/T_P = 50 %.
- CASSY UA1 $\rightarrow$ Input of channel filter CH1 at PAM modulator.
- CASSY UB1 $\rightarrow$ Output of channel filter CH1 at PAM demodulator.

- Start the measurement by pressing *F9.*
- Repeat the measurement for τ/T_P = 30 % and τ/T_P = 10 %.
- Sketch your results.

Spectra of the PAM

The PAM_1 spectrum as a function of the frequency of the modulating signal.

- All measurement are made for f_P = 5000 Hz. Follow the hints **Adjusting the sampling frequency**.
- Function generator: Sine, 500 Hz, A = 10 Vpp.
- CASSY UA1 → Output PAM_1 at PAM modulator.
- CASSY UB1 → Output of the clock generator.
- Start the measurement by pressing *F9.*
- Repeat the measurement for f_M = 1 kHz and f_M = 2 kHz.
- Sketch your results.

Das PAM_1-Spectrum as a function of the duty cycle.

- Function generator: Sine, 1000 Hz, A = 10 Vpp.
- CASSY UA1 → Output PAM_1 at PAM modulator.
- CASSY UB1 → clock generator *G*.
- Setting of the duty cycle:

  Start the measurement by pressing *F9.*

  Set the duty cycle to τ/T = 30 %.

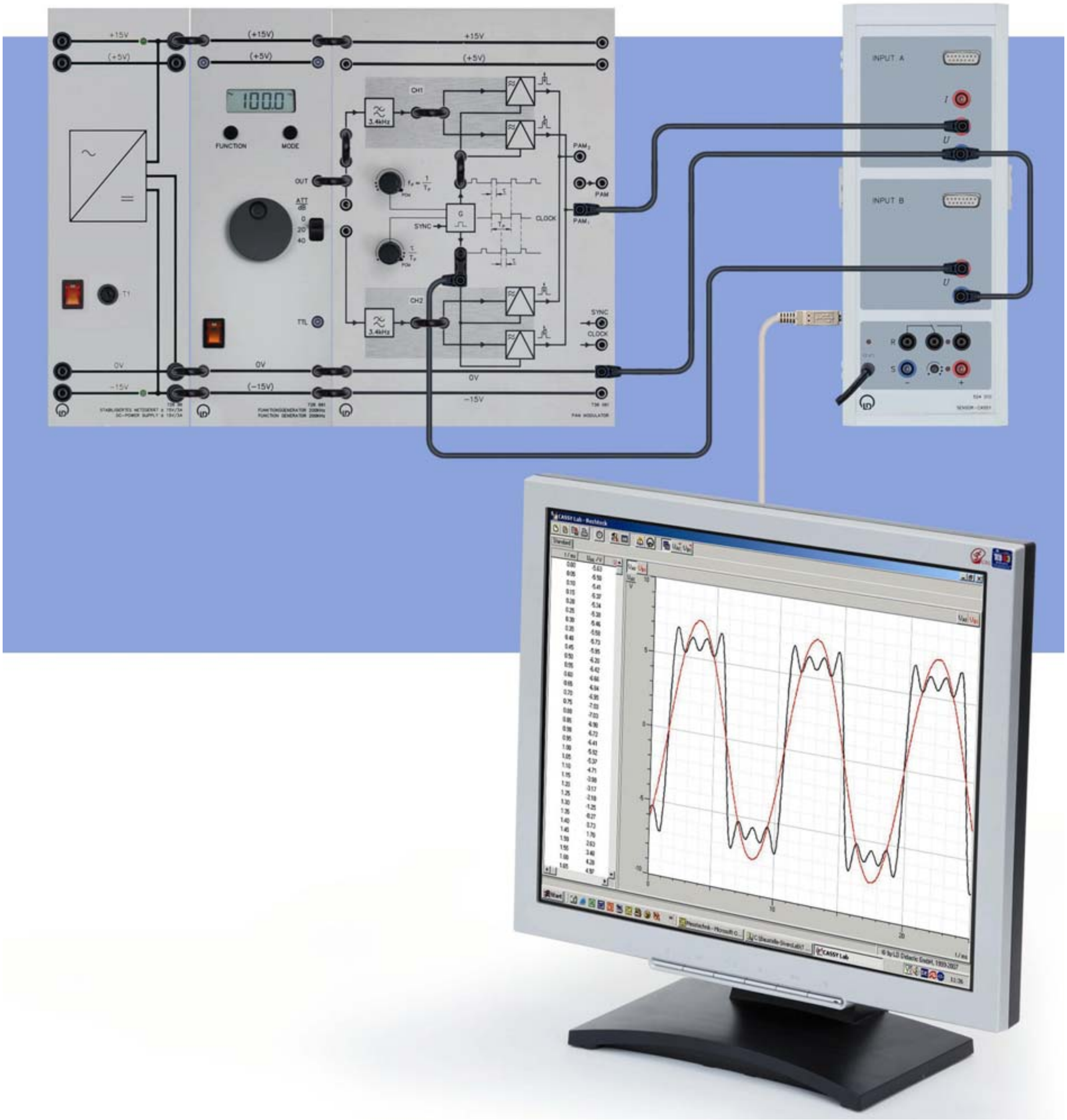
- Start the measurement by pressing *F9.*
- Sketch your results. Mark in the spectrum the position of the suppressed carrier lines.

  Compare the PAM spectra with the pulse spectra. What is the behavior of the upper side lines USL with regard to the frequency of the modulating signal f_M? What is the behavior of the lower side lines LSL?

The PAM_2 spectrum as a function of the frequency of the modulating signal.

- All measurement are made for f_P = 5000 Hz. Follow the hints **Adjusting the sampling frequency**.
- Function generator: Sine, 500 Hz, A = 10 Vpp.
- CASSY UA1 → Output PAM_2 at PAM modulator.
- CASSY UB1 → Output of the clock generator.
- Start the measurement by pressing *F9.*
- Repeat the measurement for f_M = 1 kHz and f_M = 2 kHz.
- Sketch your results.

Displaying aliasing



An undistorted demodulation of PAM signals is only possible, if the sampling theorem is fulfilled.

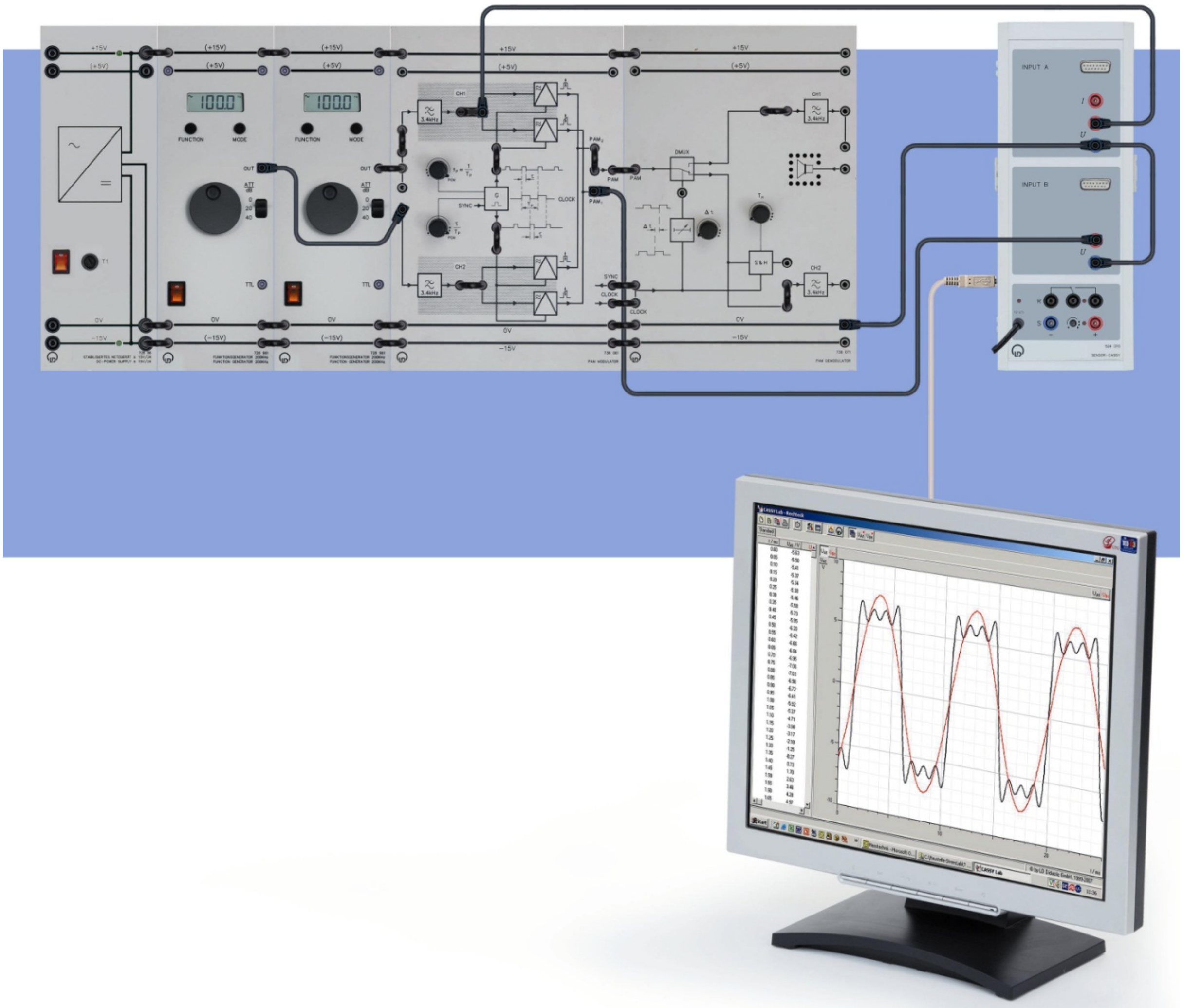
Subsampling in the frequency domain

- Function generator: Sine, 3000 Hz, A = 5 Vpp.
- CASSY UA1 → Output PAM_1 at the PAM modulator.
- CASSY UB1 → Clock generator *G*.
- For the setting of the sampling frequency f_P = 5000 Hz
- For the setting of the duty cycle τ/T_P= 20 %
- Start the measurement by pressing *F9.*
- Sketch the results.

Subsampling in the time domain

- Function generator: Sine, 3000 Hz, A = 5 Vpp.
- CASSY UA1 → Input of the PAM modulators
- CASSY UB1 → Output PAM_1 at the PAM modulator
- Start the measurement by pressing *F9.*
- Display the modulating signal $s_M(t)$ and the demodulated signal $s_D(t)$ at subsampling.

PAM time multiplex



Display the time characteristic of the time multiplex signal.

- Sampling frequency f_P = 5000 Hz, duty cycle maximal.
- Function generator 1: Triangle, f_{M1} = 200 Hz, A = 5 Vpp.
- Function generator 2: Sine, f_{M2} = 300 Hz, A = 10 Vpp.
- CASSY UA1 → Input PAM modulator channel CH1.
- CASSY UB1 → Output PAM modulator PAM_1.
- Start the measurement by pressing *F9.*

Results

Time characteristics of the PAM

Input signal (red)- and output signal (black) of the input filter of *CH1*.

Modulating signal and *PAM₁* signal A_M = 10 Vpp, f_M= 500 Hz f_P = 5000 Hz, τ/T_P = 50%

Modulating signal and *PAM₂* signal A_M = 10 Vpp, f_M= 500 Hz f_P = 5000 Hz, τ/T_P = 50%

Modulating signal $s_M(t)$ and demodulated signal $s_D(t)$ for τ/T_P = 50%

Modulating signal $s_M(t)$ and demodulated signal $s_D(t)$ for τ/T_P = 30%

Modulating signal $s_M(t)$ and demodulated signal $s_D(t)$ for τ/T_P = 10%.

Spectra of the PAM

PAM_1

$s_P(t)$: A_P = 5 V τ/T_P = 50% f_P = 5000 Hz
$s_M(t)$: A_M = 5 V f_M = 500 Hz

$s_P(t)$: A_P = 5 V τ/T_P = 50% f_P = 5000 Hz
$s_M(t)$: A_M = 5 V f_M = 1000 Hz

$s_P(t)$: A_P = 5 V τ/T_P = 50% f_P = 5000 Hz
$s_M(t)$: A_M = 5 V f_M = 2000 Hz

$s_P(t)$: A_P = 5 V τ/T_P = 30% f_P = 5000 Hz
$s_M(t)$: A_M = 5 V f_M = 1000 Hz

Summary

- The calculation of the PAM spectra makes use of:

$$S_{PAM} = A\frac{\tau}{T_P}\frac{\sin(\pi\cdot\tau\cdot nf_p)}{\pi\cdot\tau\cdot nf}$$

- The spectral amplitudes determined apply respectively for the **upper** and **lower sidelines** (*USL*n and *LSL*n).
- A double line appears with bipolar PAM instead of an individual carrier line. The frequency interval of the sidelines of the suppressed carrier is equal to the signal frequency f_M.
- The *USLs* are shifted with increasing signal frequency f_M into the range of higher frequencies in the PAM spectrum. The *USLs* are in the **normal position**. The *LSLs* are correspondingly shifted into the range of lower frequencies in the PAM spectrum. They appear in the **inverted position**

PAM_2

$s_P(t)$: $A_P = 5$ V $\tau/T_P = 50\%$ $f_P = 5000$ Hz
$s_M(t)$: $A_M = 5$ V $f_M = 500$ Hz

$s_P(t)$: $A_P = 5$ V $\tau/T_P = 50\%$ $f_P = 5000$ Hz
$s_M(t)$: $A_M = 5$ V $f_M = 1000$ Hz

$s_P(t)$: $A_P = 5$ V $\tau/T_P = 50\%$ $f_P = 5000$ Hz
$s_M(t)$: $A_M = 5$ V $f_M = 2000$ Hz

Measurements are taken at the PAM_2 output.

Summary

- The modulating signal $s_M(t)$ experiences a frequency-dependent attenuation when converted into a PAM_2 signal (S&H). Special equalizers or form filters have to be used wherever these attenuation distortions (= linear distortions) are disruptive.
- The effect of the frequency dependent attenuation distortion only appears clearly for longer pulse duration τ and at high signal frequencies f_M.

Displaying aliasing

$s_P(t)$: A_P = 5 V τ/T_P = 20% f_P = 5000 Hz

$s_M(t)$: A_M = 5 V f_M = 3000 Hz

PAM spectrum for subsampling

$s_P(t)$: A_P = 5 V τ/T_P = 20% f_P = 5000 Hz

$s_M(t)$: A_M = 5 V f_M = 3000 Hz

PAM signal for subsampling

in
V

$s_P(t)$: A_P = 5 V τ/T_P = 20% f_P = 5000 Hz

$s_M(t)$: A_M = 5 V f_M = 3000 Hz

Demodulated signal for subsampling

Summary

- Aliasing generates non-linear distortions, i.e. new spectral components appear in the output signal which were not contained in the modulating signal $s_M(t)$. Since these spectral components occur in the pass band of the demodulator low pass, an output signal is produced.

PAM time multiplex

$s_P(t)$:	A_P = 5 V	τ/T_P = 20%	f_P = 5000 Hz
$s_M(t)$:	A_M = 5 V	f_{M1} = 200 Hz	f_{M2} = 300 Hz

PAM time multiplex input

The envelopes of both channels are limiting the time multiplex signal

PAM demodulator time shift Δt $\rightarrow$ left/middle

- CASSY UA1 $\rightarrow$ Output PAM demodulator channel CH1.
- CASSY UB1 $\rightarrow$ Output PAM demodulator channel CH2.
- Start the measurement by pressing *F9.*

$s_{M1}(t)$:	A_{M1} = 5 V	f_{M1} = 200 Hz
$s_{M2}(t)$:	A_{M2} = 10V V	f_{M2} = 300 Hz

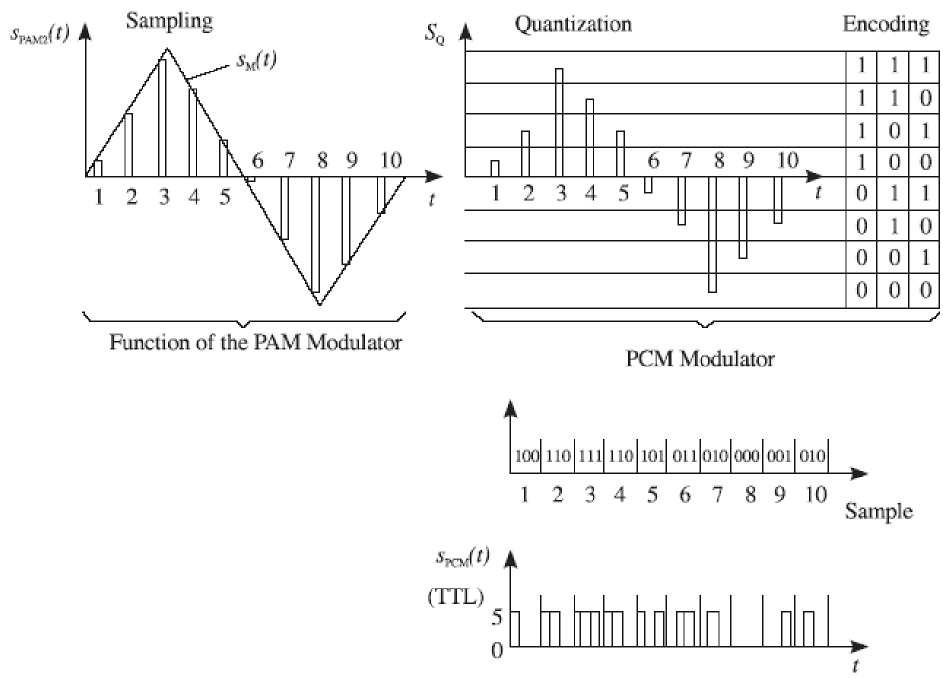
Cross talk at time multiplex. Time shift Δt $\rightarrow$ middle

$s_{M1}(t)$:	A_{M1} = 5 V	f_{M1} = 200 Hz
$s_{M2}(t)$:	A_{M2} = 10V V	f_{M2} = 300 Hz

Cross talk at time multiplex. Time shift Δt $\rightarrow$left

Pulse-Code-Modulation (PCM)

Theory



The PAM signal generated by sampling is time discrete and value continuous. To convert it into a digital signal quantization and encoding are still necessary.

Quantization

By quantization we mean the narrowing down of all possible signal values to a finite number. Whereas the PAM signal can assume any random signal value $s_{PAM}(t)$ between the modulating limits of the PAM modulator, the quantified signal $s_Q(t)$ generally demonstrates a stepped shape. The modulating range of the PCM modulator is broken down into a fixed number of intervals. Each interval is represented by assigning one signal value only. This is set representatively for all signal values, which occur in their respective interval. The quantization process takes an infinite number of all possible, continuous signal values and reduces them to a finite number. This quantization process also referred to as value discretization, always results in an additional error. This inevitable quantization error can be considered as a noise phenomena and is thus called quantization noise. The quantization distortion is influenced by the input signal type. In addition to this, the magnitude and number of quantization intervals also play a decisive role. The quantization intervals can be either equidistant discrete or logarithmic steps. In the case of equidistant quantization intervals this is referred to as linear quantization. In the case of logarithmic steps this is called non-linear quantization (see companding). The quantization becomes more precise with an increasing number of steps and there is a decrease in the quantization noise. However, small quantization intervals are more at risk from external noise causes. This primarily affects the intervals of small signal values.

Encoding

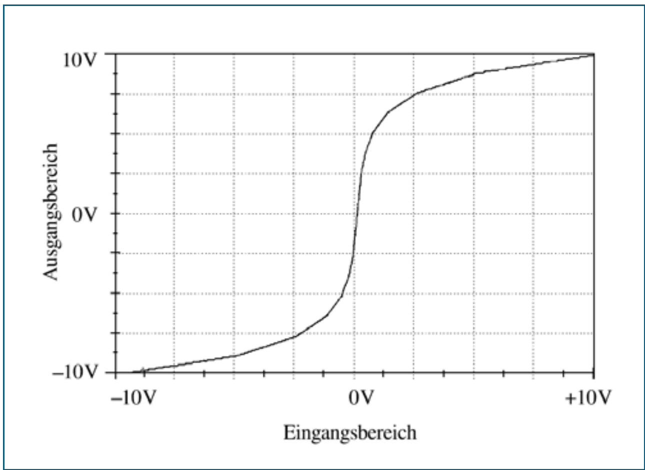
Evidently a sampled, discrete value signal is unsuitable for transmission via a noisy transmission channel. The digital signal at the output of the quantifier requires a shape better suited to the relationships prevailing on the channel. The conversion process required for this is called encoding. Coding constitutes the allocation of a specific mapping between the elements of two different character sets. Among the various types of codes the binary codes play an important role. These involve only the elements 0 and 1 (high and low) as character sets. These binary digit are also called bits. Technically the two characters are represented by two voltage levels (TTL), two frequencies, two amplitudes. A well-known binary code uses the binary system for the representation of numbers in the decimal system:

Decimal	Dual
0	0000
1	0001
2	0010
3	0011
4	0100
5	0101
6	0110
7	0111
8	1000
9	1001

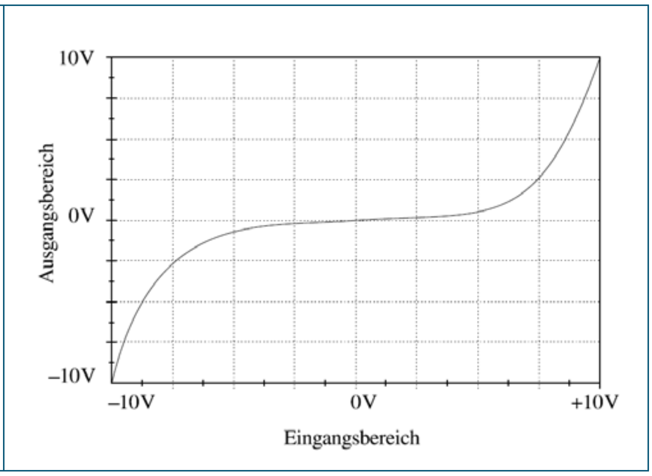
The dual code is called multi-stepped, because in the transition from one binary character to the next, it is possible that several bits have to be converted at the same time (see for example the transition from 3 to 4) in the decimal system. The sampling of the modulating signal $s_M(t)$ in the PAM modulator supplies the s_{PAM2} signal. Quantization is performed through the allocation of representative voltage values to the respective quantization intervals. In each case, these lie in the middle of the relevant intervals. One can see the quantization errors, e.g. in samples 2 and 5 or 7 and 10. In both cases the allocation is performed to the same representative value! A binary coding starting with the code word 000 for the negative peak value of s_{PAM2} as well as a conversion to TTL-level finally supplies the PCM signal. PCM demodulation takes place by means of reconversion into a PAM signal. This is followed by the well known PAM demodulation using a low pass filter. Each signal value s_Q of the quantified signal is converted into a sequence of 0 V and 5 V pulses. A subsequent PCM demodulator only has to distinguish between these two voltages. Let us assume that the critical threshold of the receiver is at 2.5 V. Then, noise continues to have no impact on the signal transmission as long as its level stays below this critical threshold.

Companding

The term companding is composed out of the words compression and expansion. The idea behind companding is the desire to improve signal to noise ratio (SNR). A signal-to-noise ratio kept constant over a large modulation range is of great advantage precisely in cases of fluctuating signal amplitude. Imagine music which alternates between low and loud volume levels. Distortion during low volume sections are perceived to be stronger than those occurring during loud passages. The principle of companding is based on increasing the low amplitudes on the transmitter end. High amplitudes, less subject to distortion, are decreased. The compression carried out in the transmitter has to be alleviated again on the receiver end. This so-called expansion process returns the low amplitude values back to their original levels. Companding is also a standard method of improving the signal-to-noise ratio in PCM systems. In PCM the noise arising through the transmission can be completely eliminated as long as it remains below the critical threshold. The advantage is offset by the quantization distortion. In the case of the pulse modulations, in particular PCM, companding is especially useful as it does not lead to an expansion of the required transmission band. A logarithmic characteristic fulfills to a great extent the requirement for a constant signal-to-noise ratio. It is frequently approximated by a 13-segment characteristic. The 13 segment compressor and expander characteristics are illustrated below. With the companding method the SNR is increased by 24 dB.



Compressor characteristic

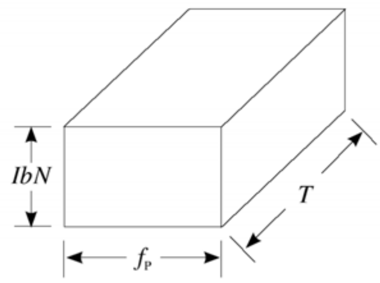


Expander characteristic

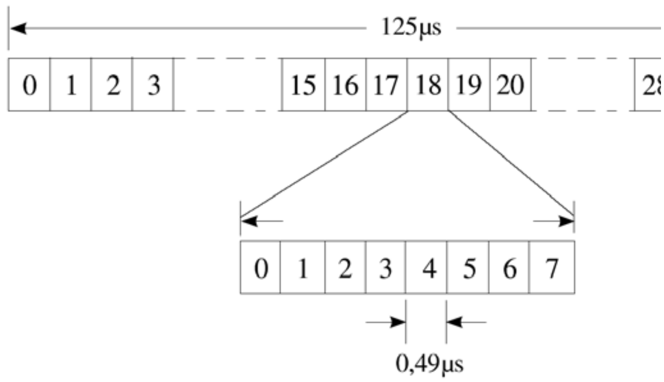
Time division multiplex with PCM

In addition to the PAM, PPM, PDM as well as PCM are also suitable for the time-division multiplex principle. The PCM method has a great practical significance and should therefore be introduced in brief using the example of the PCM 30/32 system. Here we are dealing with a commercially utilized time-division multiplex system for the transmission of 30 telephone channels as well as a synchronous signal and a telephone switching signal. Each telephone channel transmits signals in the frequency range between 300 Hz and 3.4 kHz. According to the sampling theorem this requires a minimum sampling rate of $f_P = 2f_{smax}$ = 6,8 kHz. Since there is no steep edged band limiting filter, the sampling rate for practical reasons has been fixed at f_P = 8 kHz. As a result the pulse frame has a duration of $T_P = 1/f_P$ = 125 µs. All 32 channels are sampled within this time period. The signals of all the channels are each quantified with 8 bits. Consequently there has to be $C = f_P*8*32$ bit transmitted per second. Accordingly the information flow *C* in PCM30/32 amounts to:

$C = f_P$ 8 32 Bit/s = 2.048 Mbit/s.



The pulse frame is broken down into 32 equally large time segments. The segment 0 alternately carries either an alarm signal or the synchronous signal required for frame recognition. The segments 1...15 and 17...31 transmit the telephone signals. The time segment 16 is reserved for the telephone switching signal.



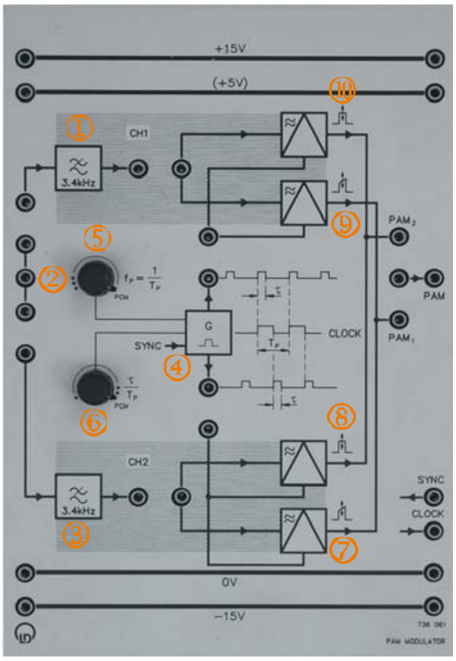
The pulse frame of the PCM 30/32

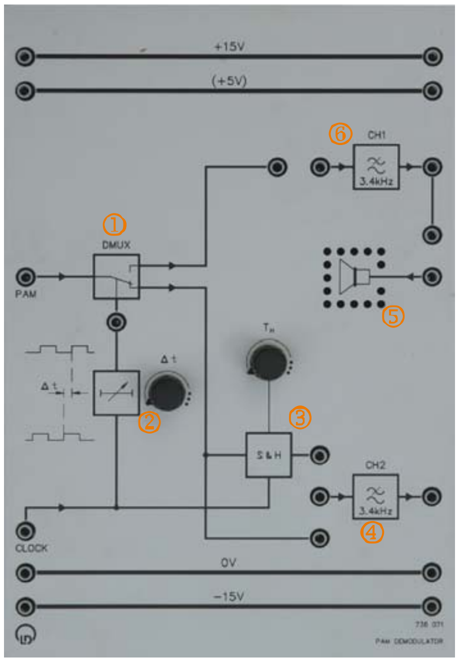
The advantage of PCM-TDM over PAM-TDM is in the greater disturbance insensitivity of the digital signals and the possibility of signal recovery or regeneration.

Equipment overview

Qty.	Cat.-no.	Designation
1	736061	PAM Modulator
1	736071	PAM Demodulator
1	736101	PCM Modulator
1	736111	PCM Demodulator
		Accessories
1	72609	Panel frame T130, Two Level
1	72686	DC-Power Supply ± 15 V/3 A
2	726961	Function Generator 200 kHz, 230 V
1	524013S	Sensor-CASSY 2 Starter
3	50059	Set of 10 safety bridging plugs, black
1	500592	Safety Bridging Plugs with Tap, black, set of 10
2	500614	Safety connection lead 25 cm, black
4	500644	Safety connection lead 100 cm, black
1	564002	LIT: Pulse Code Modulation T 7.2.2.1
		Additionally recommended
1	72610	Panel frame T150, Two Level

Training system





PAM Modulator

1. Input filter channel 1
2. Socket field for connection of the function generator
3. Input filter channel 2
4. Clock generator
5. Controller for f_P
6. Controller for duty cycle
7. Modulator for PAM1 channel 2
8. Modulator for PAM2 channel 2
9 Modulator for PAM1 channel 1
10. Modulator for PAM2 channel 1

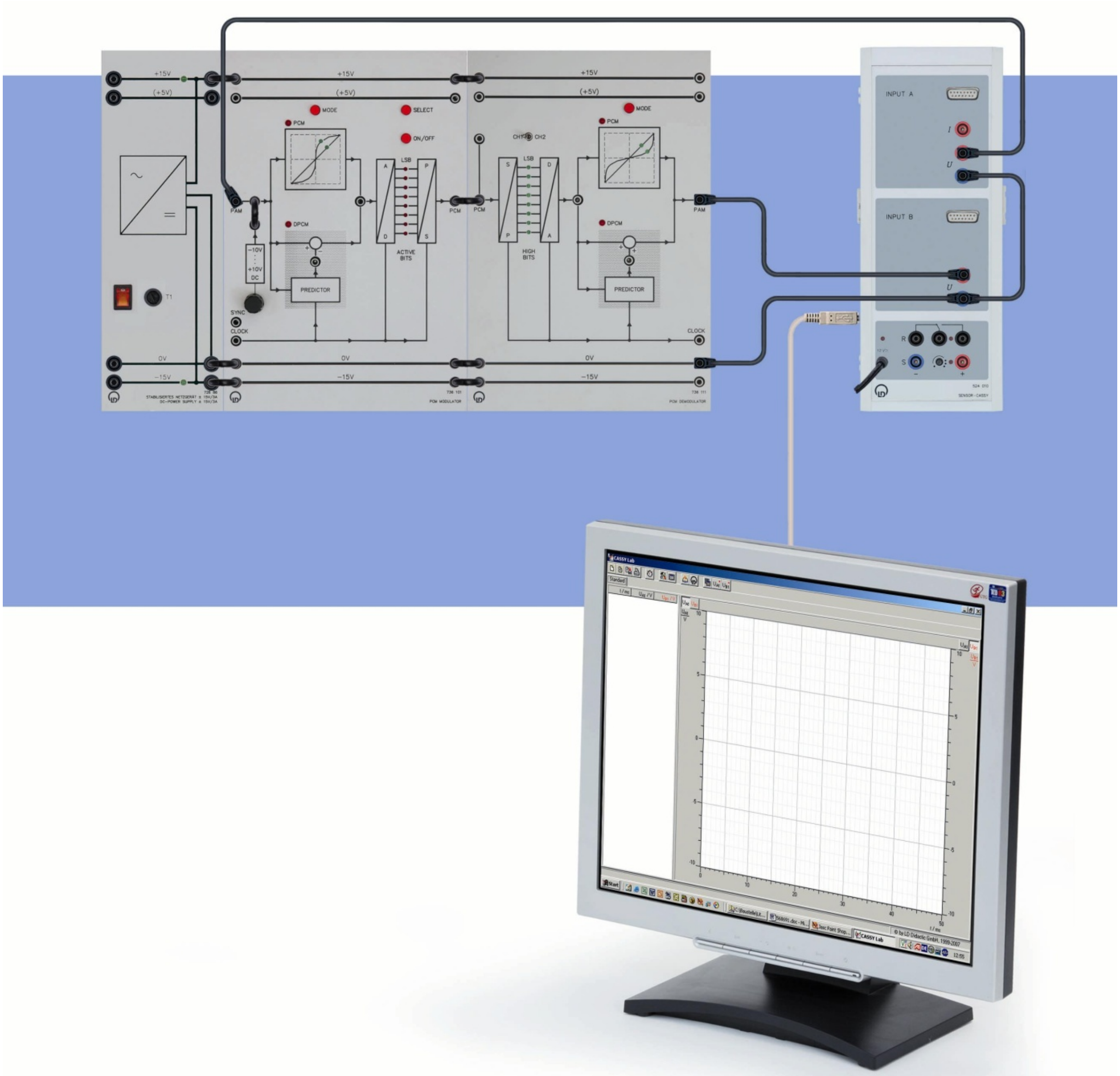
PAM Demodulator

1. Demultiplexer
2. Variable skew Δt
3. Sample and hold element with adjustable hold time T_H
4. Demodulator low pass channel 2
5. Loudspeaker with integrated push-pull stage
6. Demodulator low pass channel 1

Material

1	736 061	PAM Modulator
1	736 071	PAM Demodulator
1	736 101	PCM Modulator
1	736 111	PCM Demodulator
1	524 013S	Sensor CASSY 2 Starter
2	726 961	Function generator 200 kHz
1	726 86	Stabilized power supply ±15 V, 3 A
1	726 09	Panel frame T130, two level
2	501 461	Pair of cables 100 cm, black
1	501 441	Pair of cables 25 cm, black
3	501 511	Set of bridging plugs, black
1	501 512	Set of bridging plugs with tap, black
1		PC

Quantization



Linear quantization

- Set up the shown experiment and switch on the power supply.
- By pressing several times MODE set the PCM modulator and PCM demodulator to: PCM, linear quantization (watch the allocated LEDs).
- Activate all bits. For that press the button SELECT, until all (red) LEDs in the PCM modulator indicate ACTIVE. In the further course of the experiment: Deactivate bits from the LSB (button SELECT and ON / OFF, see below).
- At the PCM modulator: turn slowly the potentiometer for DC voltage. In the range of small inputs (< –10 V) overload of the A/D-Converter may occur. This means a sudden decrease of voltage 0 V —> –9,5 V. It is not critical eventually start your measurement from ca. –9,5 V.
- Turn the potentiometer completely to the left.
- Start the measurement by pressing *F9.*
- Turn the potentiometer to the right. This produces an input voltage at the PCM-Modulators (736 101) which is slowly rising from –10 V to +10 V. This input voltage is displays as voltage U_{A1}. The output voltage (after quantization) at the PCM Demodulator (736 111) is displayed as voltage U_{B1}.

- After recording the quantization characteristic, stop the measurement by pressing *F9*.

Non-linear quantization

- Press the MODE button of the PCM modulator and PCM demodulator one time. Now both systems are in the mode non-linear quantization (watch the allocated LEDs in the 13-segment characteristic). Repeat the measurement.

Compressor / Expander characteristic

- For plotting the compressor/expander characteristic only one device is operated in the non-linear mode, while the other device runs in the linear mode.

Variant

- Reduction of the resolution from 8 to 5 bits. For this deactivate the three least significant bits (LSB) of the PCM modulator by pressing of SELECT and ON/OFF. Repeatedly pressing SELECT leads to the position of the desired bit. ON/OFF toggles between active/inactive.
- Turn the potentiometer back to left and repeat the recording of the quantization characteristic.

software settings:
measuring time :100 sec
interval :100 msec
(off) trigger
(on) repeating measurements

x-y mode : display --> new --> add new curve --> x axis :ua1 (input of pcm modulator)
--> y axis :ub1 (output of pcm demodulator)

Results

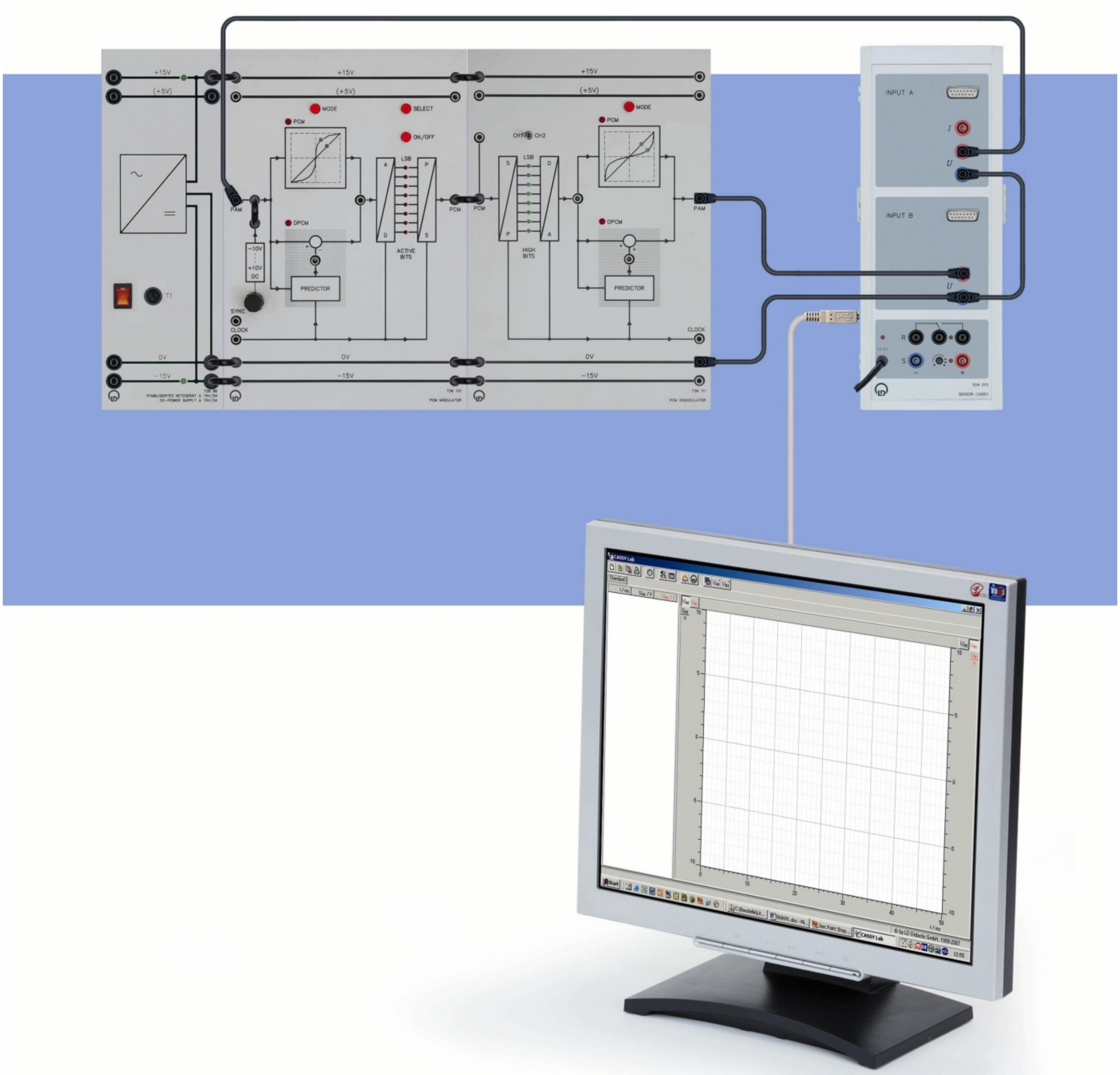
Quantization

Resolution: 8 Bit Linear quantization **Interpretation :**	
Resolution: 5 bit Linear quantization **Interpretation:**	
Resolution: 8 bit Non-linear quantization	

Resolution: 5 bit Non-linear quantization **Interpretation:**	
Resolution: 8 bit Compressor characteristic **Interpretation:**	
Resolution: 5 bit Compressor characteristic	

Resolution: 8 bit Expander characteristic **Interpretation:**	
Resolution: 5 bit Expander characteristic	

Encoding

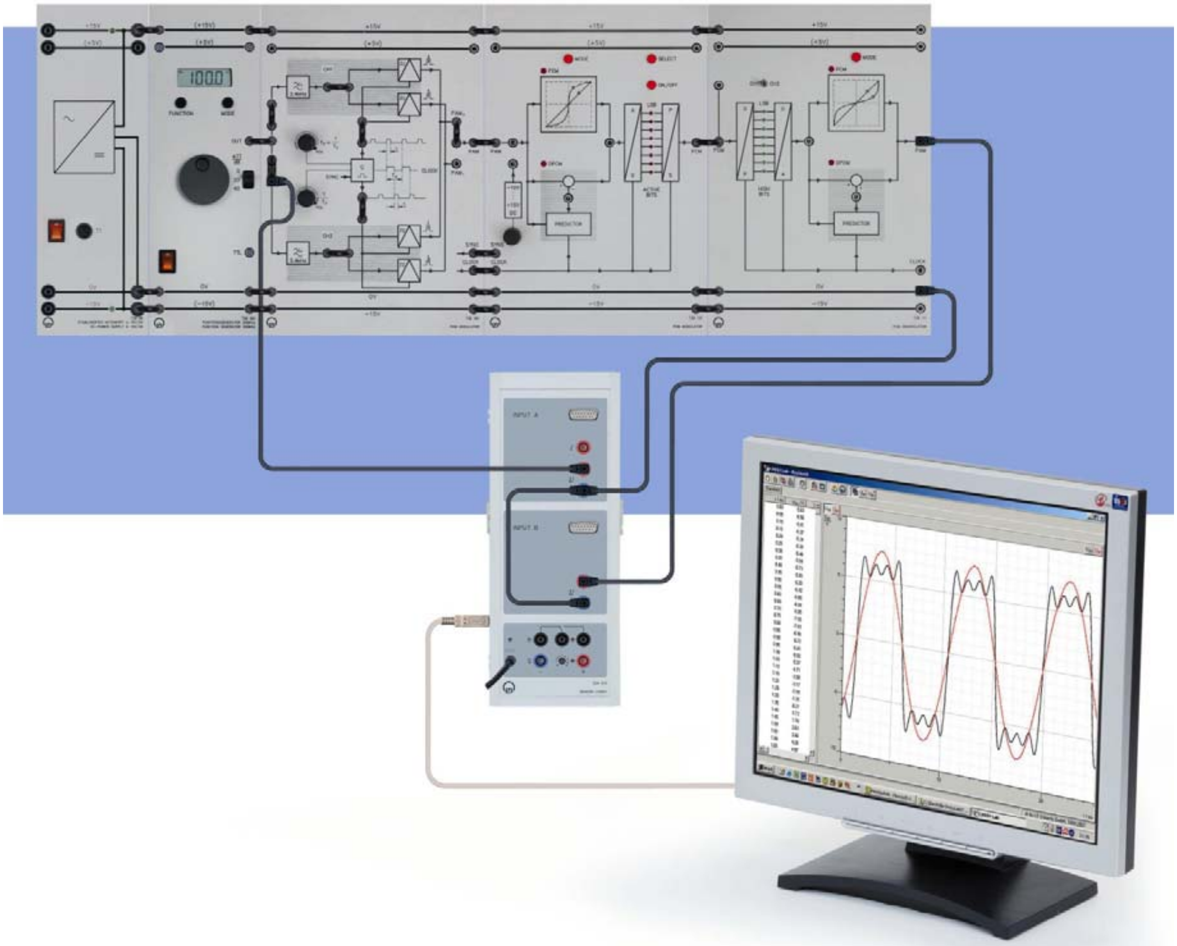


- Set up the shown experiment and switch on the power supply.
- By pressing several times MODE set the PCM modulator and PCM demodulator to: PCM, linear quantization (watch the allocated LEDs).
- Activate all bits. For that press the button SELECT, until all (red) LEDs in the PCM modulator indicate ACTIVE.

- Set the potentiometer to maximum left.
- Vary the DC voltage UA1 with the potentiometer according to the values in the table. Note the output voltage of the PCM demodulator UB1 and the corresponding bit pattern (green LEDs).
- Demonstrate the relationship between the formation of the serial data packets and the “high bits” display. Which of the bits is the LSB in the data packets? Which one is used for coding the polarity?

UA1/V	UB1/V	MSB LSB
-10.00		
-9.00		
-8.00		
-7.00		
-6.00		
-5.00		
-4.00		
-3.00		
-2.00		
-1.00		
0.00		
1.00		
2.00		
3.00		
4.00		
5.00		
6.00		
7.00		
8.00		
9.00		
10.00		

Quantization noise



- Set up the shown experiment and switch on the power supply. Connect both channels (*CH1* and *CH2*) of the PAM Modulator (736 061) with the function generator. This avoids time gaps at the output of the PCM demodulator.
- Function generator: Triangle, 30 Hz, 12 V_{pp}
- By pressing several times MODE set the PCM modulator and PCM demodulator to: PCM, linear quantization (watch the allocated LEDs).
- Activate all bits. For that press the button SELECT, until all (red) LEDs in the PCM modulator indicate ACTIVE.

- Start the measurement by pressing *F9.*
- Sketch the measurement and give an interpretation.
- Repeat the experiment for a resolution of 5 bit.
- Repeat the experiment for a resolution of 5 bit and a frequency of the triangle signal f_M = 300 Hz.

Quantization noise

Resolution 8 bit
Triangle 12 $V_{pp} \rightarrow A_M = 6$ V, $f_M = 30$ Hz

Curve	Color	Signal
U_{A1}	red	Input at PAM-Mod
U_{B1}	blue	*PAM* at PCM-Dem
$Q = U_{A1}-U_{B1}$	black	Quantization noise

Interpretation

Resolution 5 bit
Triangle: 12 $V_{pp} \rightarrow A_M = 6$ V, $f_M = 30$ Hz

Curve	Color	Signal
U_{A1}	red	Input at PAM-Mod
U_{B1}	blue	*PAM* at PCM-Dem
$Q = U_{A1}-U_{B1}$	black	Quantization noise

Interpretation

Influence of phase shift
Resolution: 5 bit
Triangle: 12 $V_{pp} \rightarrow A_M = 6$ V, $f_M = 300$ Hz

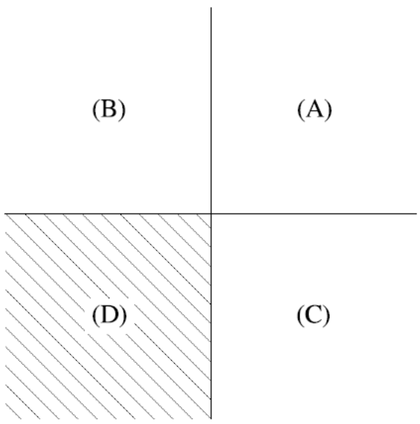
Curve	Color	Signal
U_{A1}	red	Input at PAM-Mod
U_{B1}	blue	*PAM* at PCM-Dem
$Q = U_{A1}-U_{B1}$	black	Quantization noise

Interpretation

Difference Pulse Code Modulation (DPCM)

Theory

A picture which is rastered with a line scanning pattern is formed on the screen of a monitor. Normally this has no detrimental effects because the human eye as the information receiver blends the closely positioned discrete lines into a homogenous whole picture. Any substantial increase in the number of lines has no marked advantage for the human eye. The picture rastering performed by the TV camera reduces the picture information to the minimum needed by the human eye. Unimportant information, i.e. information which the human eye is incapable of resolving, is not further processed. The selection process isolating the important (relevant) from the unimportant (irrelevant) information is called irrelevance reduction**.** Also the quantization process in the PCM modulator can be understood as irrelevance reduction. Apart from the relevancy and irrelevancy criteria there is still another signal property, which is part of signal information reduction. The signal characteristic most important here differentiates the signal values according to whether they are known to the receiver (redundant) or unknown (non-redundant). In fact only the unknown portion of a signal needs to be transmitted, the redundant components can be added again automatically by the receiver without any information loss occurring. In short a signal can be represented in the so-called message plane.

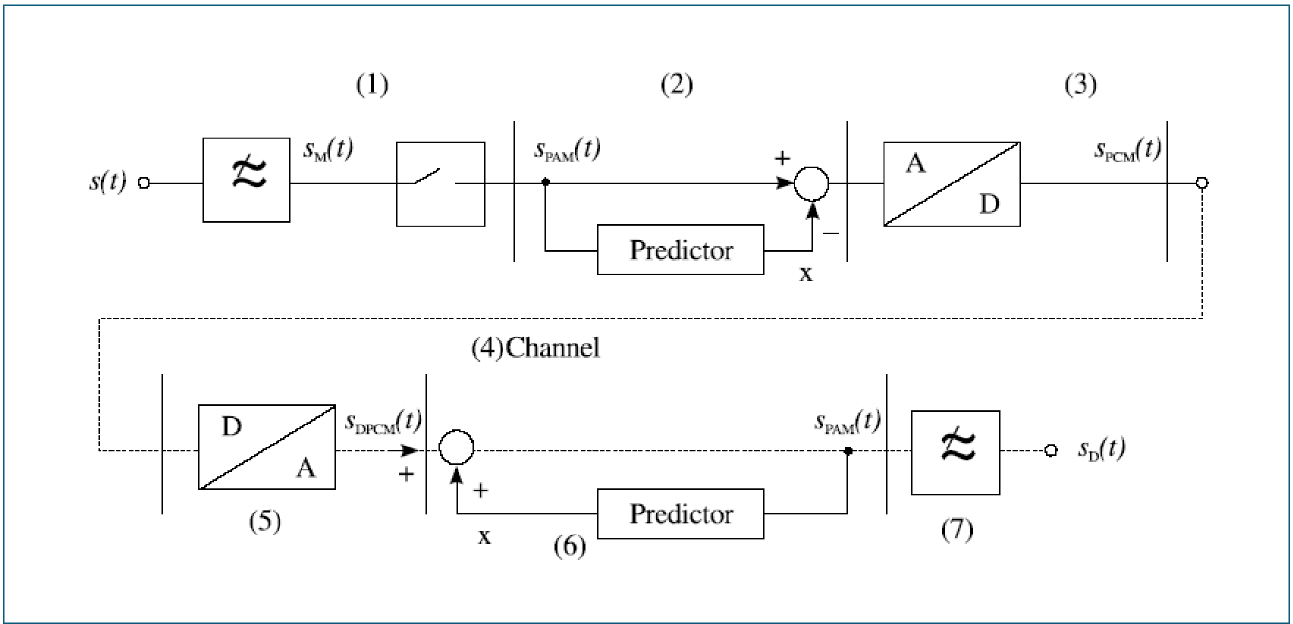


The message plane

A: redundant, non-relevant
B: redundant, relevant
C: non-redundant, non-relevant
D: non-redundant, relevant

The horizontal line divides the redundant part of the information from the non-redundant part. The vertical line distinguishes according to relevance and irrelevance. For the receiver only the shaded, unknown part of the information is important.

The DPCM constitutes a method of reducing the redundancy of the information signal. The figure demonstrates one possible operating principle for a DPCM link. The quantization is no longer carried out for each individual PAM value. Only the difference between the current PAM value and a predicted value *X*, which is formed in the DPCM modulator, is quantified. This predicted value also called estimated value or prediction value, is generated in a predictor from the previous PAM values. Thus the signal's past history goes into it.

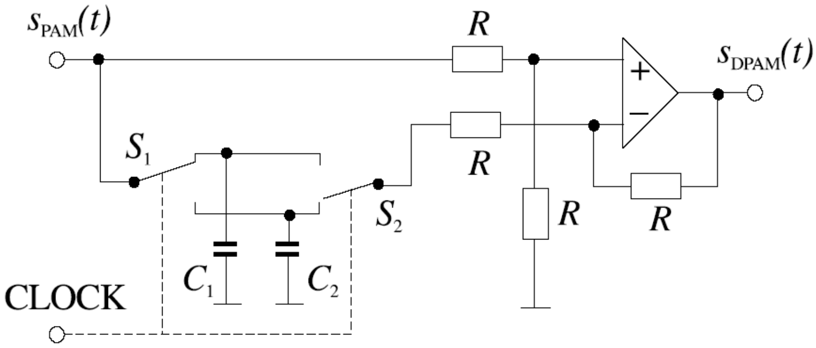


Basic setup of a DPCM system
1: Sampling (PAM modulator)
2: Decorrelation, formation of the predicted value (PCM modulator)
3: Quantization, coding (PCM modulator)
4: Channel
5: Decoding, D/A conversion (PCM demodulator)
6: Recovery of the redundancy (PCM demodulator)
7: Recovery of the time continuation (PAM demodulator)
AM Demodulator (Rückgewinnung der Zeitkontinuität)

In order for the predictor to be able to form the prediction value correctly, the statistics of the modulating signal have to be known. Consequently, the principle of the DPCM is based on the possibility of being able to make probability statements regarding the occurrence of particular PAM values. Since statistics is vital for the prediction, the DPCM can only work for input signals with known statistical behavior. Consequently, in the following experiment it only makes sense to work with triangular signals. The information important for the prediction of the next respective sample lies solely in the constant ascent of the triangular function. The only thing that must be decided is whether this ascent is negative or positive. An important area of application for DPCM, which is also called predictive source coding, is digital image processing. Due to the enormous volume of data, methods on bit rate reduction are indispensable in the area of digital image processing. There are various possibilities for the integration of the DPCM modulator. Redundancy reduction can also be integrated into the quantization process.

Operation of the predictors in the PCM modulator and the PCM demodulator

The PCM modulator and PCM demodulator training panels are microprocessor controlled. Their function in DPCM operation can be explained using the following analog equivalent circuit diagram.

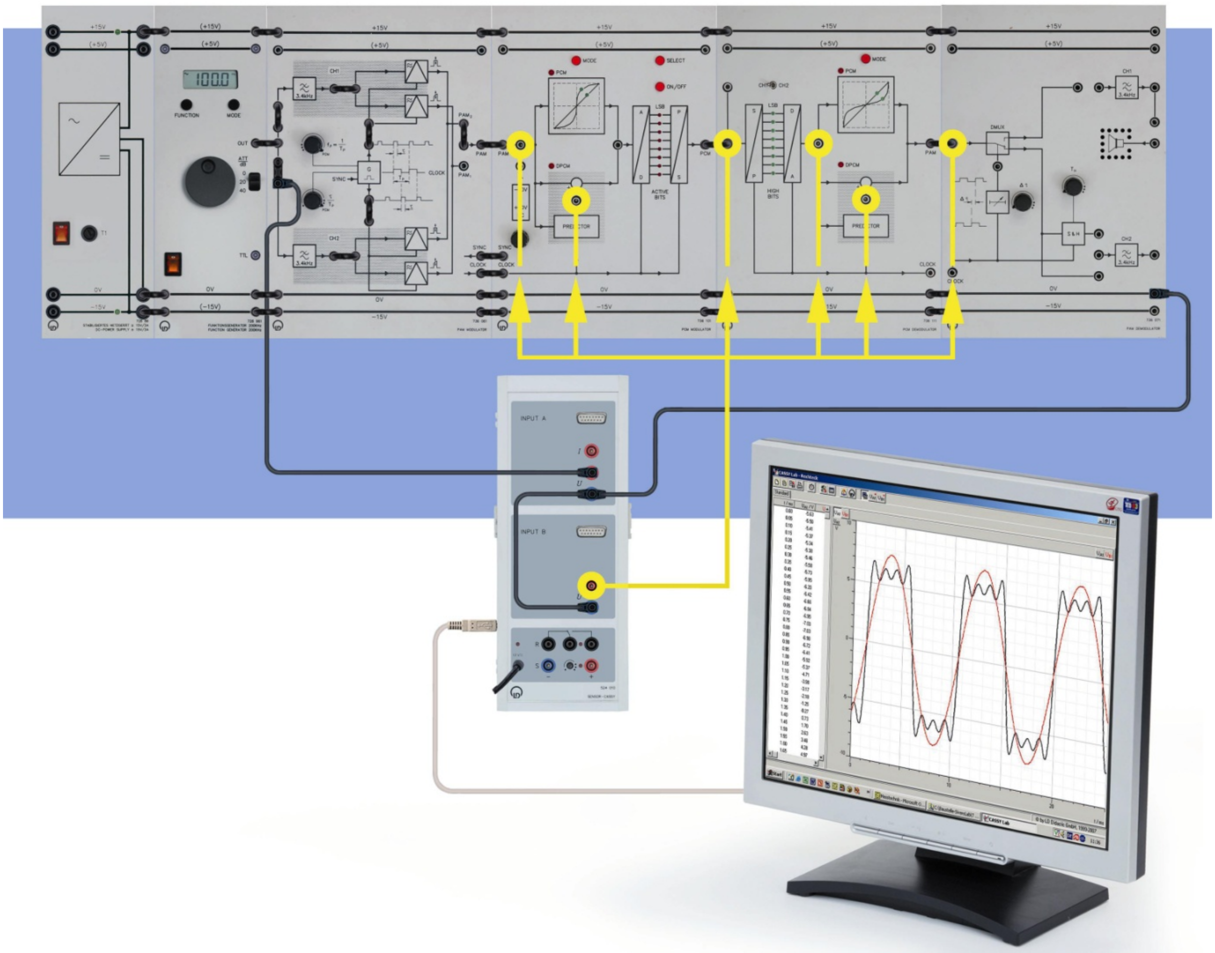


The DPCM modulator contains a differential amplifier and an analog memory. The switches S_1 and S_2 synchronized at the sampling frequency f_P operate in counter sense. Together with storage capacitors C_1 and C_2 they form the PREDICTOR. The formation of the prediction value is simple. The PAM pulses occurring in rhythm with the sampling frequency f_P are alternately charged in the storage media C_1 und C_2 by the switching operation of S_1. S_2 is used to read out the last respective PAM value. While in the shown switch position the current PAM value is read into the capacitor C_1 via the switch S_1, the last PAM value is read out of the capacitor C_2 via S_2. Consequently the differential amplifier can form the difference between the current sampling value n and the previous sample n–1. Since only one value (n–1) from the signal's history is used for the formation of the estimated value X, this method is called Previous Sample Prediction. The function of the DPCM demodulator is inverse with respect to the DPC modulator. Here the DPCM signal must be added to the prediction value.

Material

1	736 061	PAM Modulator
1	736 071	PAM Demodulator
1	736 101	PCM-Modulator
1	736 111	PCM-Demodulator
1	524 013S	Sensor CASSY 2 Starter
1	726 961	Function generator 200 kHz
1	726 86	Stabilized power supply ±15 V, 3 A
1	726 09	Panel frame T130, two level
2	501 461	Pair of cables 100 cm, black
3	501 511	Set of bridging plugs, black
1	501 512	Set of bridging plugs with tap, black
1		PC

Carrying out the experiment



- Set up the shown experiment and switch on the power supply. Connect both channels (*CH1* and *CH2*) of the PAM Modulator (736 061) with the function generator. This avoids time gaps at the output of the PCM demodulator.
- Function generator: Triangle, 30 Hz, 12 V_{pp}

- By pressing several times MODE set the PCM modulator and PCM demodulator to: PCM, linear quantization (watch the allocated LEDs).
- Activate all bits. For that press the button SELECT, until all (red) LEDs in the PCM modulator indicate ACTIVE.
- DPCM is a redundancy reducing method. In the predictor the difference to the previous value is transmitted. At the start of the transmission it is important that the predictors in the PCM modulator and in the PCM demodulator start from the same prediction value. During switch-on the prediction value is initialized with 0. But since the two systems cannot be switched on simultaneously, the following switch-on sequence has to be adhered to:

  1. Connect the PAM input of the PCM modulator to 0 V.
  2. Switch the PCM modulator to DPCM mode.
  3. Switch the PCM demodulator to the DPCM mode.
  4. Disconnect the PAM input of the PCM modulator from 0 V.
  5. Drop the amplitude of the modulation signal to 0 V (on the function generator).
  6. Feed the modulation signal into the PAM input of the PCM modulator and reset to the desired amplitude.

- Step 5 has to be performed every time before selecting the ACTIVE BITS. Afterwards the signal amplitude can be enhanced again.

Settings on the PAM system	
Sampling frequency	$f_P \rightarrow$ PCM
duty cycle	$\tau/T \rightarrow$ PCM
Time delay of the Demultiplexer	$\Delta t \rightarrow$ min

Settings on the PCM system	
PCM-Modulator	DPCM
PCM-Demodulator	DPCM
ACTIVE BITS	all on
Channel selection	CH1

- Start the measurement by pressing *F9.*
- Connect the channel UA1 of the CASSY with the input signal of the PAM modulator. With channel UB1 of the CASSY record successively the following signals:
  PAM input
  Predictor of the DPCM modulator
  Output of the DPCM modulator
  Input of the DPCM demodulator
  Predictor of the DPCM demodulator
  PAM output of the DPCM demodulator
- Sketch your measurements and give an interpretation.

Results

DPCM-Modulator

Red: Triangular signal from function generator
Black: PAM-Signal 8 bit

Red: Triangular signal from function generator
Black: Predictor signal 8 bit

Red: Triangular signal from function generator
Black: DPCM output 8 bit

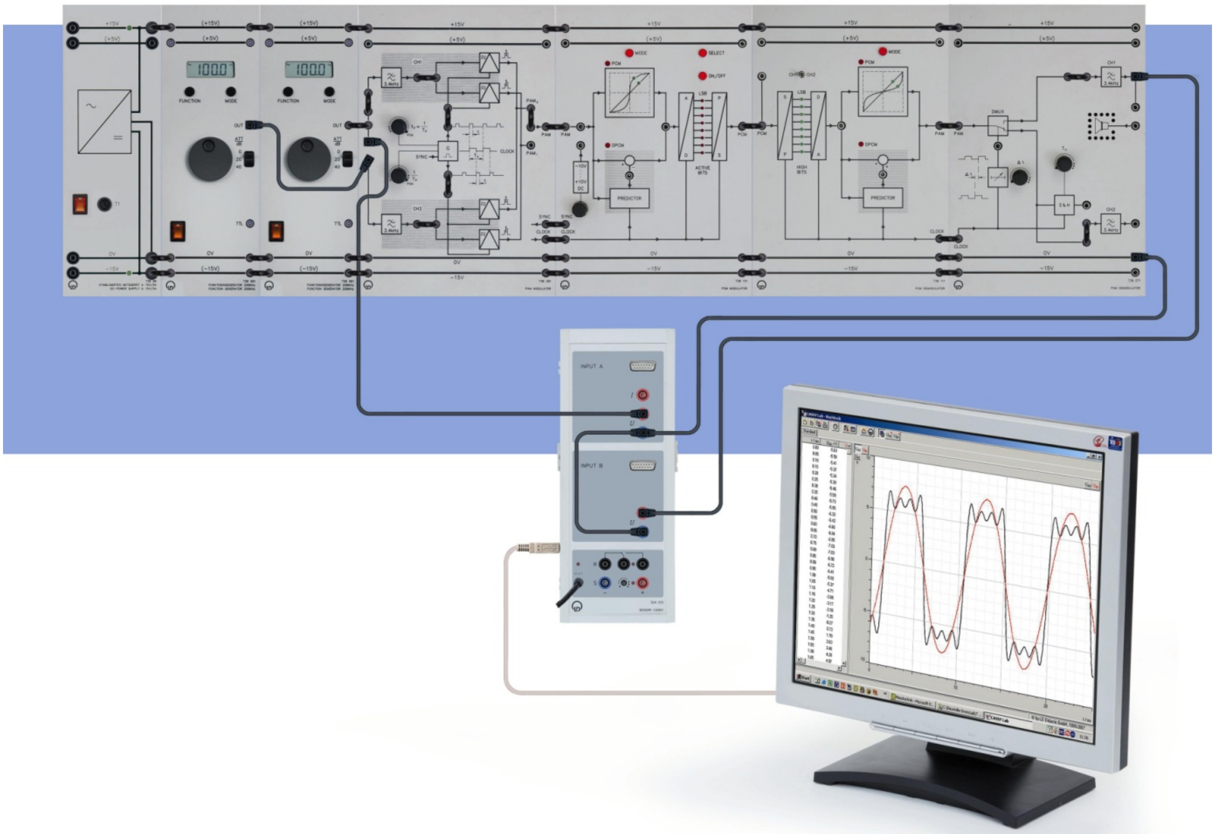
DPCM-Demodulator

Red: Triangular signal from function generator
Black: Input DPCM demodulator 8 bit

Red: Triangular signal from function generator
Black: Predictor signal, demodulator 8 bit

Red: Triangular signal from function generator
Black: PAM output DPCM dem. 8 bit

PCM transmission



- Set up the shown experiment and switch on the power supply.
- By pressing several times MODE set the PCM modulator and PCM demodulator to: PCM, linear quantization (watch the allocated LEDs).
- Activate all bits. For that press the button SELECT, until all (red) LEDs in the PCM modulator indicate ACTIVE.
- PAM Modulator
  Controller for the duty cycle $\tau/T_P \rightarrow$ PCM
  Controller for the sampling frequency $f_P \rightarrow$ PCM
- Function generator 1: Sine, f_{M1} = 300 Hz, A = 10 Vpp.
- Function generator 2: Triangle, f_{M2} = 200 Hz, A = 5 Vpp.
- PAM Demodulator
  Time shift $\Delta t \rightarrow$ links

1. Part of the experiment
- Start the measurement by pressing *F9.*

2. Part of the experiment(change CASSY-connections)
- CASSY UA1 $\rightarrow$ Input PAM Modulator Kanal CH2.
- CASSY UB1 $\rightarrow$ Output PAM Demodulator CH2.
- Repeat the measurement.

PCM transmission

Resolution: 8 bit PCM transmission Function generator 1: Sine, f_{M1} = 300 Hz, A = 10 Vpp.	
Resolution: 8 bit PCM transmission Function generator 2: Triangle, f_{M2} = 200 Hz, A = 5 Vpp.	

Summary

- The predictor value is the second to last input voltage value. Consequently the form of the prediction signal is the same as that of the input signal, but delayed by one sampling period.
- The DPCM signal is the difference of the input signal and the predictor signal. For a triangular shaped signal this difference is constant with respect to magnitude.
- Since no change occurs to the signal by means of transmission over the PCM link, the input signal of the DPCM demodulator is identical to the output signal of the DPCM modulator.
- The DPCM demodulator adds the transmission value to its prediction value. Since the transmission value is always the same in terms of magnitude, but the sign is reversed, the addition produces a triangular curve.
- The difference between the characteristics with 8 bit respectively with 4 bit quantization is non significant. The transmission with only 4 bit saves 50% of the transmission capacity.

Telecommunication Department
Communications Lab
EXP. 7 Delta Modulation/Demodulation

1. Theory

1.1 Linear delta modulation

In terms of circuitry delta modulation (DM) is a simple alternative to pulse code modulation (PCM). Here only one bit is used for sampling. With one bit only two states can be encoded. Consequently, in contrast to PCM, every single sampling value need not be quantized. As is the case in DPCM, only the difference between the momentary value $s_M(t)$and an estimated value (predicted value) X(t) is encoded. Therefore, DM belongs to the prediction based modulation methods. The design of a simple delta modulator is elaborated below.

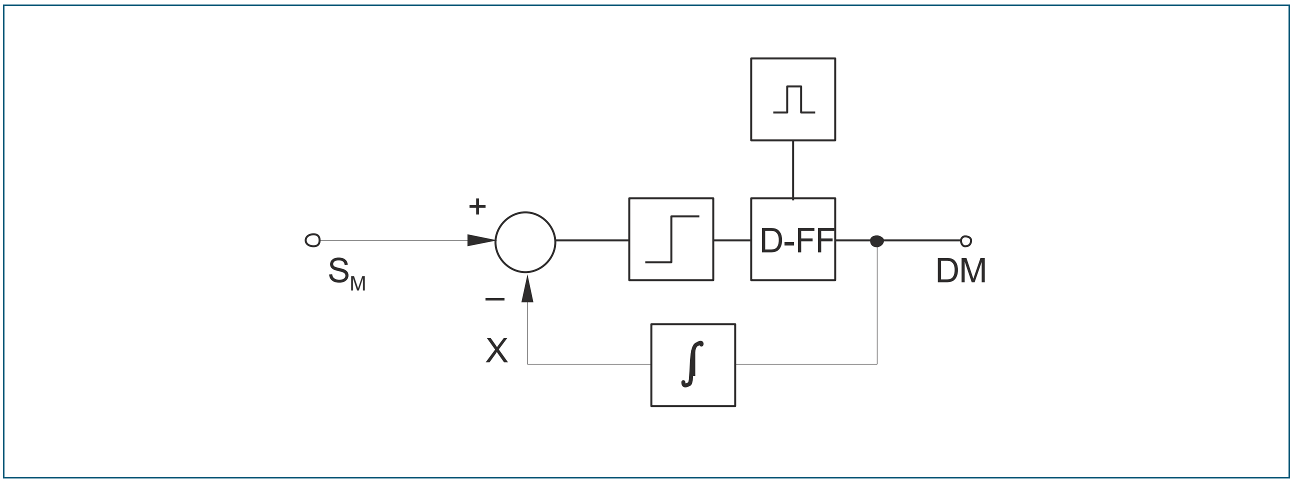


Fig. 1: Linear delta modulation (LDM)
(1): Comparator with differential input
(2): Clock generator
(3): Bistable element (D-FF)
(4): Integrating network (RC lowpass filter)

The modulating signal $s_M(t)$ is applied to the positive input of the comparator. The prediction signal X(t) is supplied to the inverting input of the comparator. The prediction signal forms a variable comparator switching threshold. If $s_M(t) > X(t)$, the comparator output inverts to logical 1. For $s_M(t) < X(t)$ the comparator inverts to logical 0. The inversion level of the comparator, which is a function of the modulating signal $s_M(t)$ and the estimated value X(t), is stored in a bistable element D-flip-flop (D-FF). With each clock pulse the D-FF element shifts the information contained in its D-input to its output and stores it there until the next clock pulse arrives. The DM signal is generated directly as the output signal of the D-FF element. As we see from the Fig. 1, the estimated signal X(t) is formed out of the output signal of the D-FF element using an integrating network. In the most elementary case, this network is a RC low pass filter. The individual steps leading up to the generation of DM are explained in the following. Let us first take the positive edge of the input signal $s_M(t)$ into consideration. Here the following applies: $s_M(t) > X(t)$. For that reason the comparator is inverted to logical 1. Consequently a logical 1 state also appears with the next pulse of the clock generator at the output of the D-FF element. If you wish to avoid DC offset during demodulation of the DM signal, then the D-FF has to operate in bipolar modus. In the DM modulator panel the following signal levels are chosen:

logical 1 = + 2 V
logical 0 = - 2 V.

A voltage pulse of +2 V is applied to the integrating network for a logical 1 state. The integral of this positive voltage pulse appears at the output of the LP. As long as the output signal X(t) of the LP remains smaller than the respective momentary value of $s_M(t)$, the comparator continuously maintains the logic 1 state at the output. Then the D-FF element continues to supply +2 V to the integrating low pass filter with each clock pulse. It is only when X(t) exceeds the value of the input signal $s_M(t)$, that the comparator switches back to zero and the D-FF supplies –2 V to the low pass filter. The prediction value X(t) then becomes smaller. In practice it is actually an RC low pass filter which frequently performs the function of the predictor. DM always requires a considerably higher sampling frequency than PCM. Information theory provides a cogent explanation for this. By applying the theory of information data can be made countable (measurable). Except when applying methods of information reduction, the amount of information M may not be changed during signal transformation, (provided there is the same transmission performance). Thus when you compare DM with an 8 bit PCM the following holds true:

$$M_{DM} = f_{CLDM} \cdot T \cdot lb2 \qquad 1.1$$

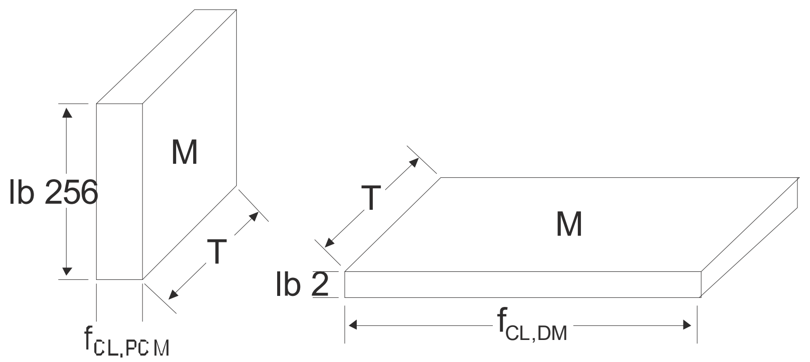


Fig. 2: Information cuboid in DM and PCM

$$M_{PCM} = f_{CLPCM} \cdot T \cdot lb256 \qquad 1.2$$

Where:

f_{CLDM}	sampling frequency in DM
f_{CLPCM}	sampling frequency in PCM
lb2 = 1 bit:	number of signal values to be coded with one bit
lb 256 = 8 bit:	number of signal values to be coded with 8 bit
T	transmission duration (identical for both modulation modes)

If the same amount of information is required for DM and PCM ($M_{DM} = M_{PCM}$) the following holds true:

$$f_{CLDM}\, lb2 = f_{CLPCM}\, lb\ 256 \qquad 1.3$$

$$f_{CLDM} = 8\, f_{CLPCM} \qquad 1.4$$

1.2 Slope overload and granular noise

In linear DM the prediction signal is always formed using pulses with the same step height, e.g. always with ±2 V pulses. This could lead to one characteristic error in DM, which is called slope overload. The error arises because the prediction signal only possesses a finite edge steepness. If the input signal $s_M(t)$ increases too quickly, then the predictor can never generate a signal which fulfills the conditions $X(t) > s_M(t)$, or $X(t) < s_M(t)$. The predicted signal at the low pass output then continuously "lags behind" the input signal. If the time constant τ of the integrating low pass is large with respect to the sampling period FCL, then the following dependency has to be maintained to avoid slope overload:

$$A_{M\max} = k \cdot A_p \frac{f_{CL}}{f_M} \qquad 1.5$$

Where:

A_{Mmax}:	maximum amplitude of a modulating, sinusoidal signal
A_P:	pulse amplitude at the integrator input
f_{CL}:	sampling frequency
f_M:	frequency of the modulating signal
k:	constant factor < 1

Apart from the above -mentioned dependency the maximum permissible amplitude A_{Mmax} is also affected by the critical frequency f_c of the integrator low pass filter. If f_M exceeds the cutoff frequency f_c, then A_{Mmax} drops with the transmission function of a low pass filter. The cutoff frequency of the integrator low pass filter is adapted to the requirements posed by the transmission of voice signals and amounts to f_c = 150 Hz.

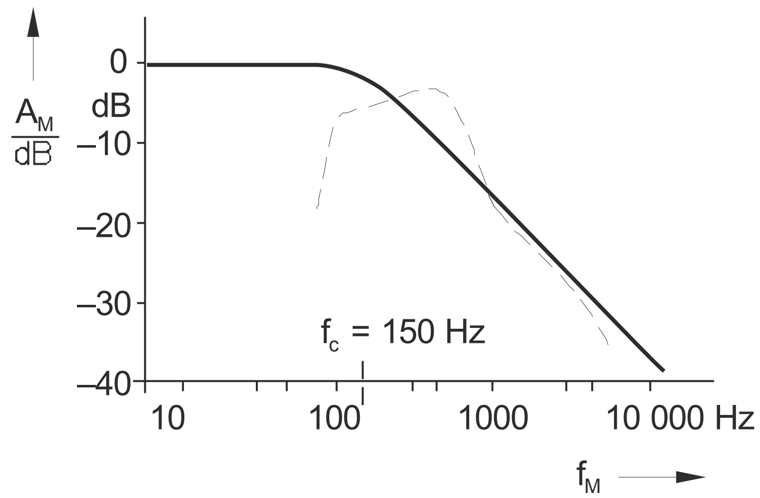


Fig. 3: Maximum amplitude of the modulating signal for LDM as a function of the signal Frequency f_M. Dashed line: Dynamic curve of a human voice signal.

Another typical error in delta modulation is granular noise. Let's assume that the input signal has a very low slope (DC voltage).

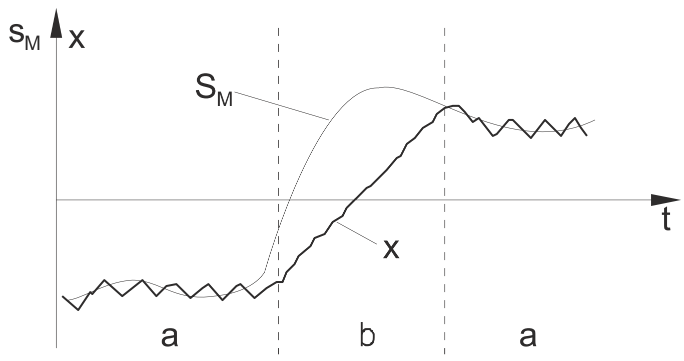


Fig. 4: Formation of the prediction signal X(t) in LDM
a: Range with granular noise
b: Range with slope overload

The prediction signal X(t) then always has to fluctuate around the DC voltage value, i.e. a kind of ripple is produced, whose amplitude decreases as the sampling frequency increases. The DM modulator no longer responds under minimum amplitude A_{Mmin}. The following relationship is cited in the literature for this minimum amplitude of the modulating signal $s_M(t)$:

$$A_{M\min} = \pi \cdot A_p \frac{f_c}{f_{CL}} \qquad 1.6$$

The dynamic range of linear DM is really modest on account of slope-overload and granular noise. For LDM it is specified as:

$$\frac{D}{dB} = 20\log \cdot \left[\frac{f_{CL}}{\pi \cdot f_g \cdot \sqrt{1+\left(\frac{f_M}{f_c}\right)^2}} \right] \qquad 1.7$$

The errors occurring during the generation of the prediction signal X(t) are depicted in the figure below.

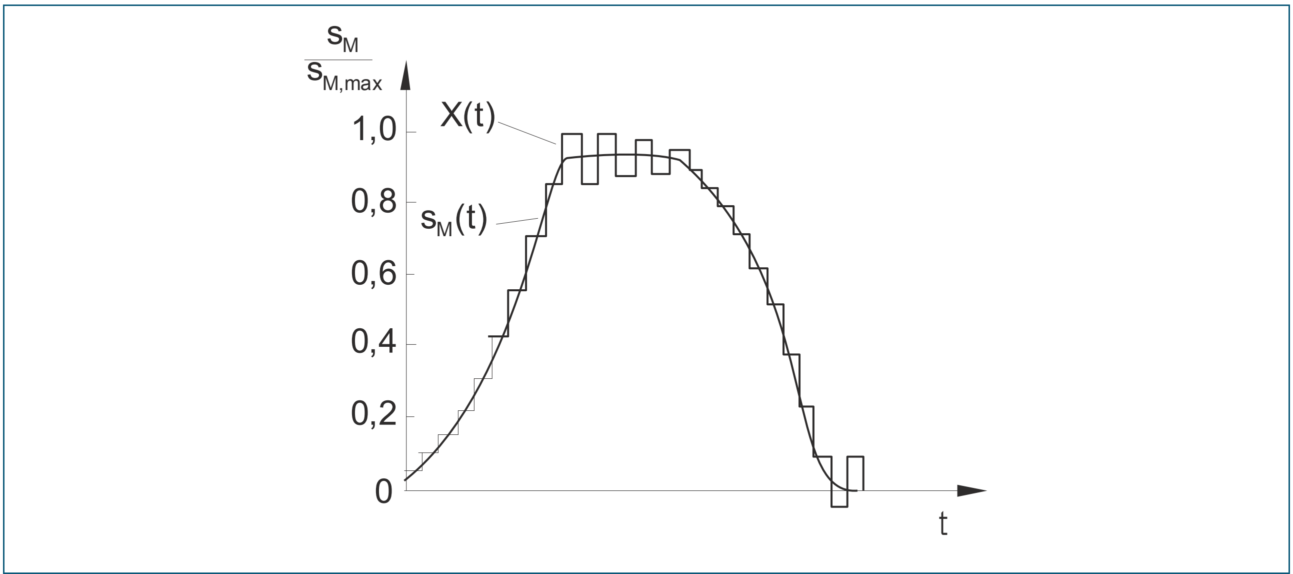


Fig. 5: Formation of the prediction signal X(t) in DCDM

1.3 Adaptive delta modulation

The slope overload error can be reduced using an adaptive procedure. Here pulses of variable step height are used to form the prediction signal. Adaptive delta modulation constitutes a form of non-linear DM. In this method the step height is formed using a control variable derived from the dynamic characteristics of the modulating signal $s_M(t)$. There exists a whole series of adaptive methods, e.g. High Information Delta Modulation (HIDM) or Continuously Variable Slope DM (CVSD). The methods of Digital Controlled Delta Modulation (DCDM) described here, obtain the control signal to adapt the step height from the DM digital signal. It is the foundation of the principle used in the training panels. In the method depicted below, the history of the DM signal is stored using a 3-stage shift register. Based on the appearance of a one bit sequence at the parallel outputs of the shift register we conclude that a voltage increase has occurred in the modulating signal $s_M(t)$. A zero bit sequence encodes a voltage drop while switchover from 0 to 1 stands for constant voltage. With this kind of regularity a companding law can be derived for the modulator and demodulator, which operates without additional synchronization. The formation of the prediction signal X(t) from pulses of variable step heights is illustrated in the block diagram.

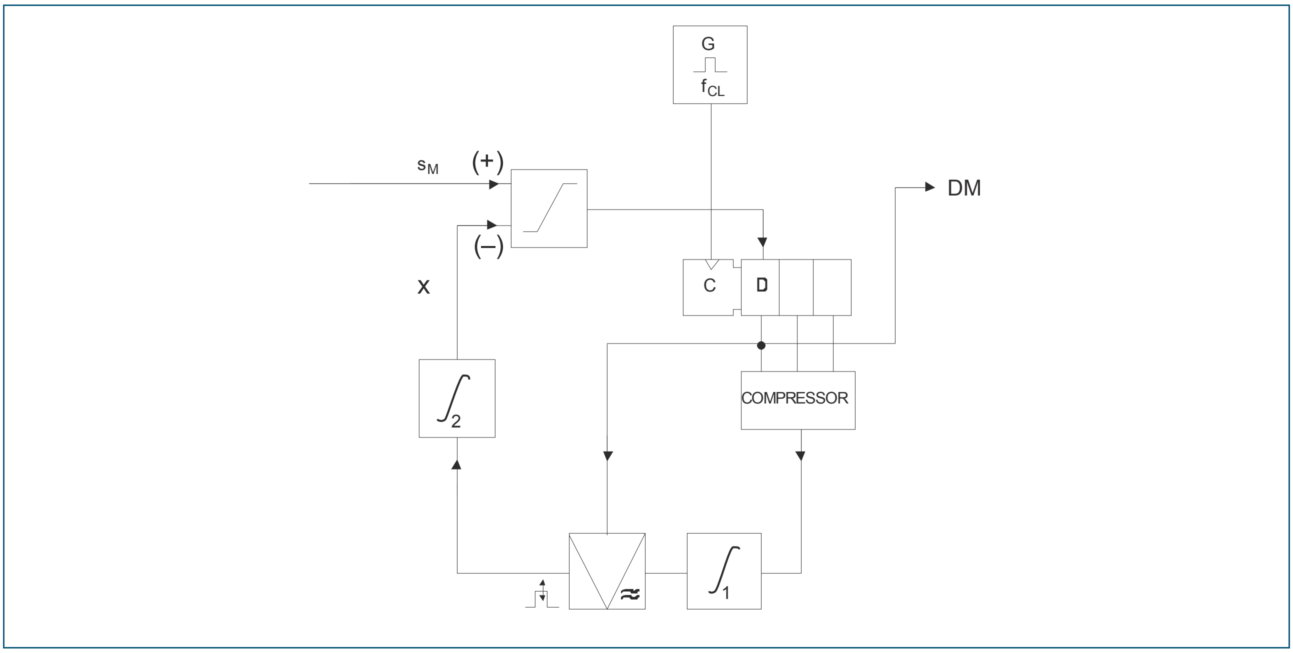


Fig. 6: Principle of DCDM

1.4 Delta-sigma modulation

Delta-sigma modulation is produced out of LDM by connecting an integrator in series. The integrator prevents rapid voltage increases at the input of the comparator. This is used to avoid slope overload. However, the dynamic of this operation is only minimal as is the case in LDM. Delta-sigma modulation is primarily used in top-quality audio signal processing.

1.5 The spectrum of the delta modulation signal

The clock-pulsed output signal of the D-FF element constitutes a binary random sequence. The amplitude spectrum of this kind of stochastic signal, in which both signal values occur with equal probability is white, i.e. it contains all frequencies with equal amplitudes. The case of permanent slope overload with its attendant period square-wave signal shall not be investigated here. Because of its high sampling frequency aliasing can easily be avoided in DM.

1.6 Demodulation methods

The demodulation of the LDM is frequently performed according to the double integrator method. This method uses two low pass filters connected in series. The prediction signal X(t) appears again at the output of the 1st low pass filter. With the exception of the prediction error, this prediction signal corresponds precisely to the modulating signal $s_M(t)$. The 2nd low pass filter is used to filter out the residual ripple. To demodulate DCDM signals, it is first necessary to recover the clock pulse. Afterwards the companding principle is used to determine the pulse amplitude for the integrating low pass filter. The DCDM demodulator then registers the arrival of more than three ones or zeros in succession to determine an increase in the pulse amplitude A_P. The pulse amplitude has to be switched to the integrator with the correct sign. Both demodulation methods are depicted in the diagram.

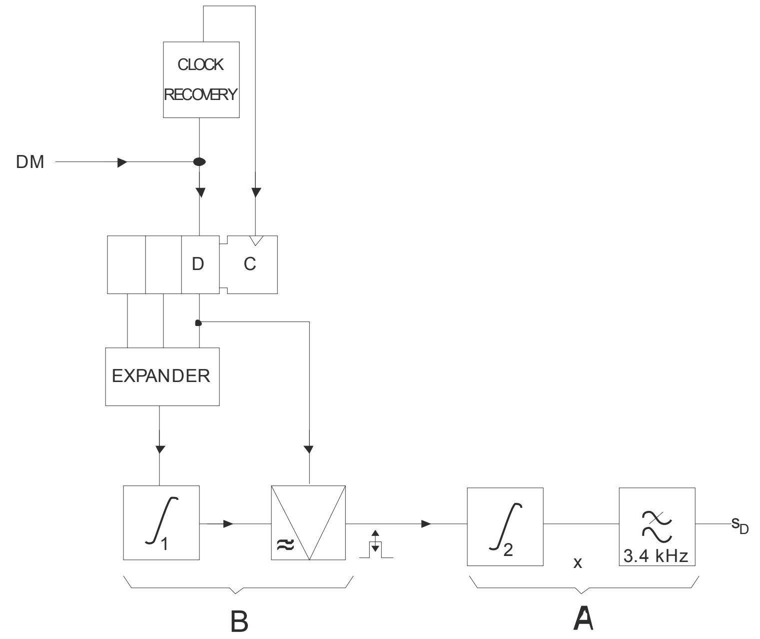


Fig. 7: Delta demodulation
A: LDM
A + B: DCDM

1.7 DM application

Due to its simple coding method DM is interesting for signal transmission at low bit rates, around 10... 40 kbit/s. It possesses a dynamic characteristic adaptable to human speech, a feature which is particularly interesting for telephone technology. Because of its simple hardware prerequisites, DM offers an alternative for the design of cost-effective codec facilities. Due to its high sampling rate it does not need a steep-edge filter for band limiting purposes. For that reason aliasing practically plays no role at all. Another feature of delta modulation is its low susceptibility to interference. Only equivalent bits are generated. In comparison, the bits of a PCM word possess entirely different valencies (MSB to LSB). The disadvantage of DM is its limited dynamic range. Any rapid change in the level of the input signal always leads to transmission errors, which do not occur in PCM.

1.8 Shift keying

By shift keying we mean the information transmission carried out through the mere turning on and off of a suitable carrier. There are three important types of shift keying for a binary, modulating signal:

ASK: Amplitude Shift Keying
In ASK the carrier is turned on and off in the rhythm of the modulating signal. Depending on whether the carrier is already premodulated with an AF signal or not, a distinction is drawn between audible and non–audible telegraphy.

FSK: Frequency Shift Keying
FSK means switching between two defined carrier frequencies f_1 and f_2. ASK has the disadvantage that the receiver cannot distinguish between line interruptions or transmitter breakdowns and the

transmission of the 0 binary state. FSK does not have this disadvantage because the information is contained in discrete frequencies f_1 and f_2. The absence of a transmission signal has to be attributed to transmission failure. Consequently it is possible to employ here automatic controlling facilities to monitor data transmission.

PSK: Phase Shift Keying
In PSK the information is located in the phase angle of the carrier signal. This is changed abruptly in time with the data signal, e.g. from $\Phi_1 = 0°$ to $\Phi_1 = 180°$. In addition to the advantages of FSK, PSK also enjoys greater interference immunity. Consequently this type of keying is gaining in popularity for data transmission. The keying types are depicted in the following figure.

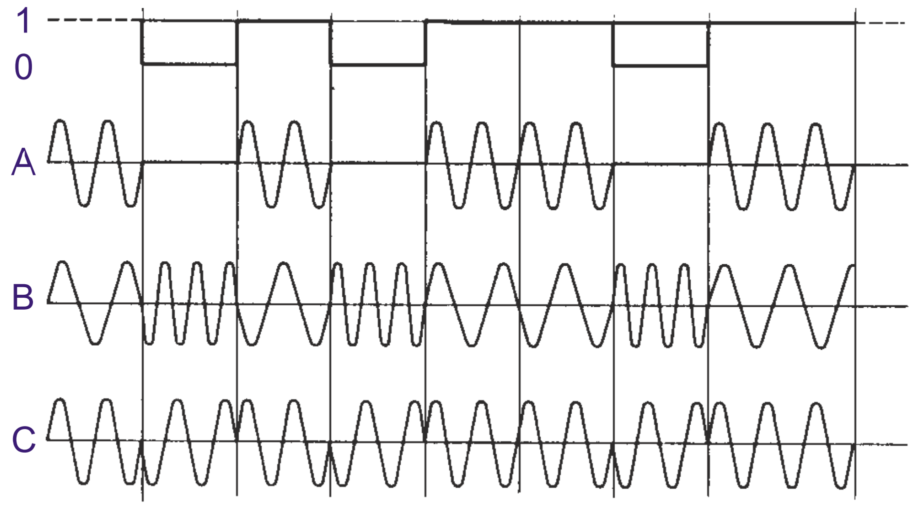


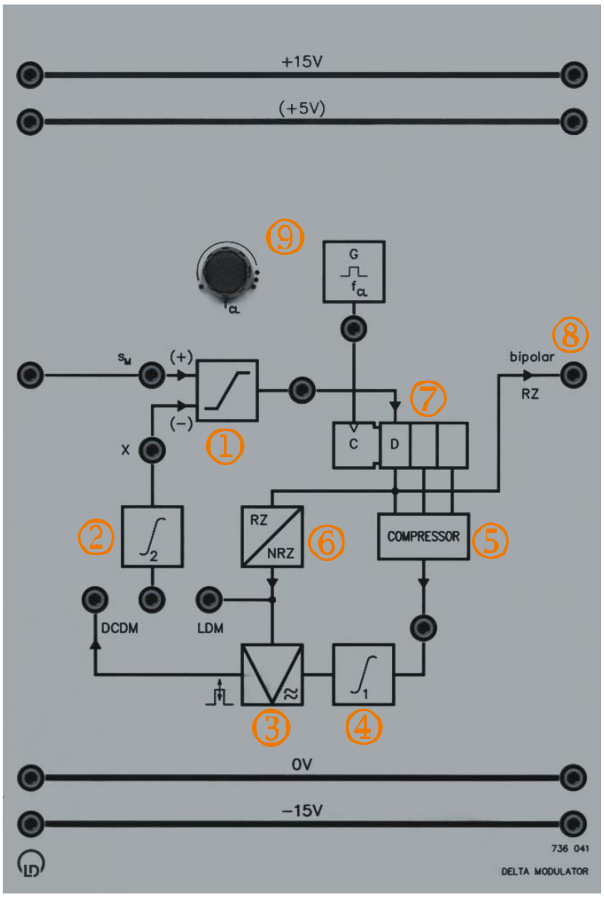
Fig. 8: Types of shift-keying
A: ASK
B: FSK
C: PSK

The information being transmitted through shift keying has to be in digital form. Originally shift keying was only used to transmit written information, which was converted into digital signals using the Morse code. The required conversion of the letters into Morse alphabet constituted a kind of coding process. However, the coding can also proceed with any random data signal or the output signals from the PCM or DM modulators.

1.9 Training objectives

- Information cuboid, measurability of information
- Interference immunity in DM
- Application of DM
- Determination of granular noise
- Determination of slope overload
- Dynamic response of LDM and DCDM
- Comparison between linear and adaptive DM
- Clock-pulse recovery and synchronization through quasiternary signals in RZ format
- Demodulation of adaptive DM
- Double integrator method

2. Equipment descriptions



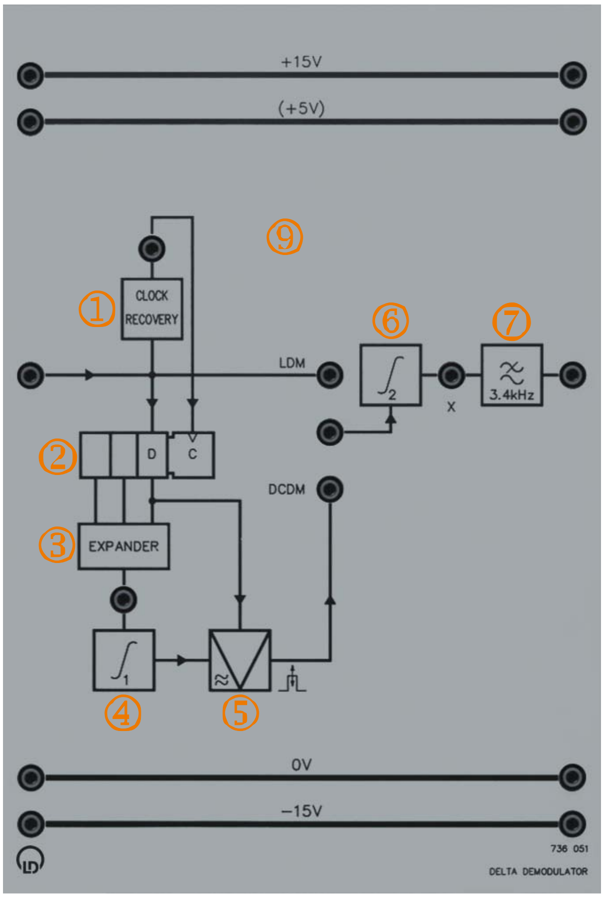
Delta Modulator (Cat. no. 736 041)

The Delta modulator consists of:

- ① Comparator
- ② Integrator for the formation of the prediction value X(t)
- ③ PAM modulator for the generation of adaptative pulse amplitude approx. ± 0.5 V … ± 10 V
- ④ Pre-integrator generation of the pulse amplitude (magnitude)
- ⑤ Compressor for the derivation of the control signal for the PAM modulator (TTL signal)
- ⑥ Converter RZ/NRZ format (LDM) Pulse amplitude ± 2 V
- ⑦ 3-stage shift register for storing signal history
- ⑧ Output for delta-modulated signals in RZ format, bipolar ± 5 V
- ⑨ Clock generator with adjustable sampling frequency (10 kHz … 100 kHz)

On the operation of the DCDM system
The control signals output from the compressor are converted into variable pulse amplitude signals in integrator 1. The PAM modulator connected downstream sets the polarity, with which the pulses are fed into integrator 2. The PAM modulator is triggered for this purpose directly by the DM output signal in NRZ format.

On the operation of the LDM system
Output D of the shift register operates like a single D-FF element. After RZ/NRZ conversion its output signals are fed directly into integrator 2 with the constant pulse amplitude ±2 V.



Delta Demodulator (Cat. no. 736 051)

The Delta demodulator consists of:

- ① Clock-pulse recovery (important in the demodulation of DCDM)
- ② 3 stage shift register
- ③ Expander (TTL signal)
- ④ Integrator 1 for recovery of the adaptive pulse amplitude (magnitude)
- ⑤ PAM modulator for recovery of the charge pulse with correct polarity
- ⑥ Demodulation integrator
- ⑦ Output lowpass with +1 gain, critical frequency 3.4 kHz
  Input signals: bipolar RZ/NRZ
  Input sensitivity for DCDM: 250 mV

Device for demodulation of linear and adaptive delta modulation (LDM and DCDM).

3. Test

1. Calculate the information flow $C = \frac{M_{DM}}{T}$ for a DM with f_{CL} = 100 kHz.

C = kBit/s

2. What does codec mean? Where are these kinds of facilities needed?

3. What are the advantages and disadvantages of DM?

Advantages:

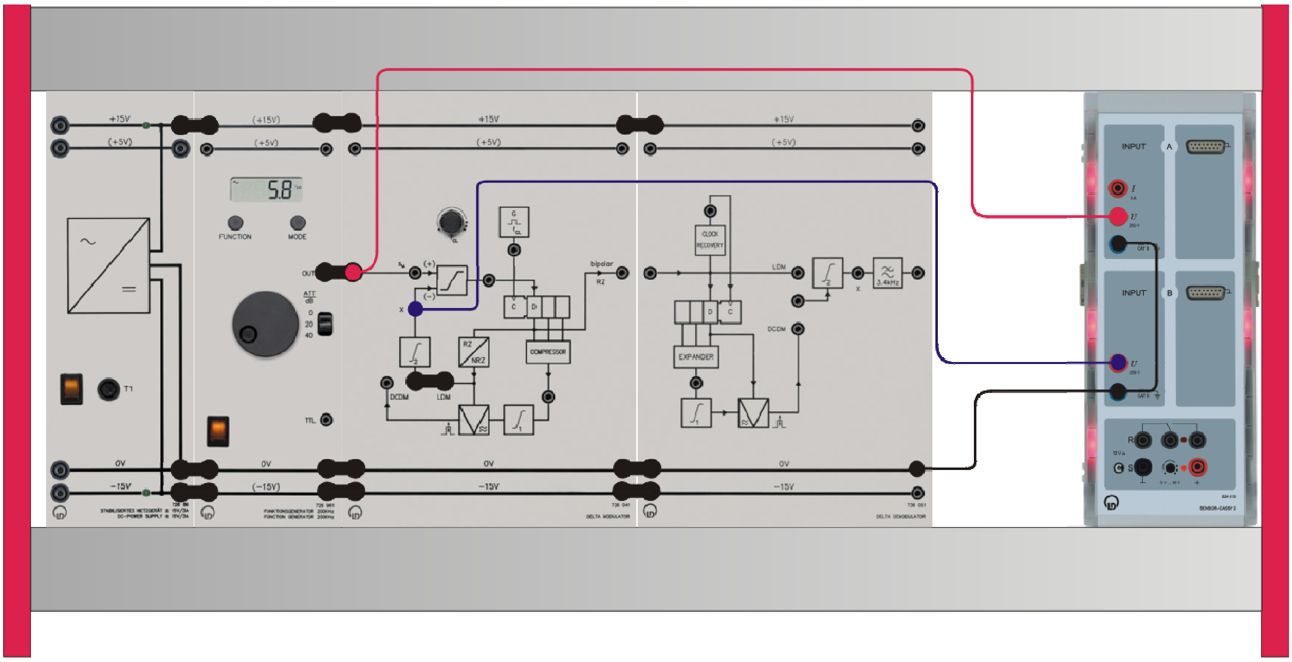
Disavantages:

4. Experiment procedure

Experiment 1: Prediction signals in LDM and DCDM

◆ **Prediction signals in LDM**

Experiment set-up:



Settings on the DM system / function generator

Clock frequency f_{CL}	100 kHz (max.)
Modulating signal s_M	sine, 100 Hz, 1 V_{pp}, ATT = 0 dB
Delta modulator	LDM, set the bridging plug
Delta demodulator	----

- Connect the Sensor-CASSY 2 inputs:
  Input A → s_M modulating signal
  Input B → X prediction signal at the output of the 2nd integrator

- Start the measurement by pressing F9.
- Jointly display on the monitor $s_M(t)$ and the prediction signal X(t).
- Sketch the signals and in diagram 1-1.

- The modulating signal $s_M(t)$ and the prediction signal X(t) are almost identical.

- Reduce the clock frequency to f_{CL} = 10 kHz (min.)
- Repeat the measurement.
- Sketch the signals and in diagram 1-2.

- Modulating signal $s_M(t)$and prediction signal X(t) in LDM, clock frequency f_{CL} = 10 kHz.
- The prediction signal X(t) deviates visibly from $s_M(t)$ at low sampling frequency.

◆ Prediction signals in DCDM

Settings on the DM system / function generator

Clock frequency f_{CL}	100 kHz (max.)
Modulating signal $s_M(t)$	sine, 100 Hz, 1 V_{pp}, ATT = 0 dB
Delta modulator	DCDM, set the bridging plug
Delta demodulator	----

- Repeat the measurements.

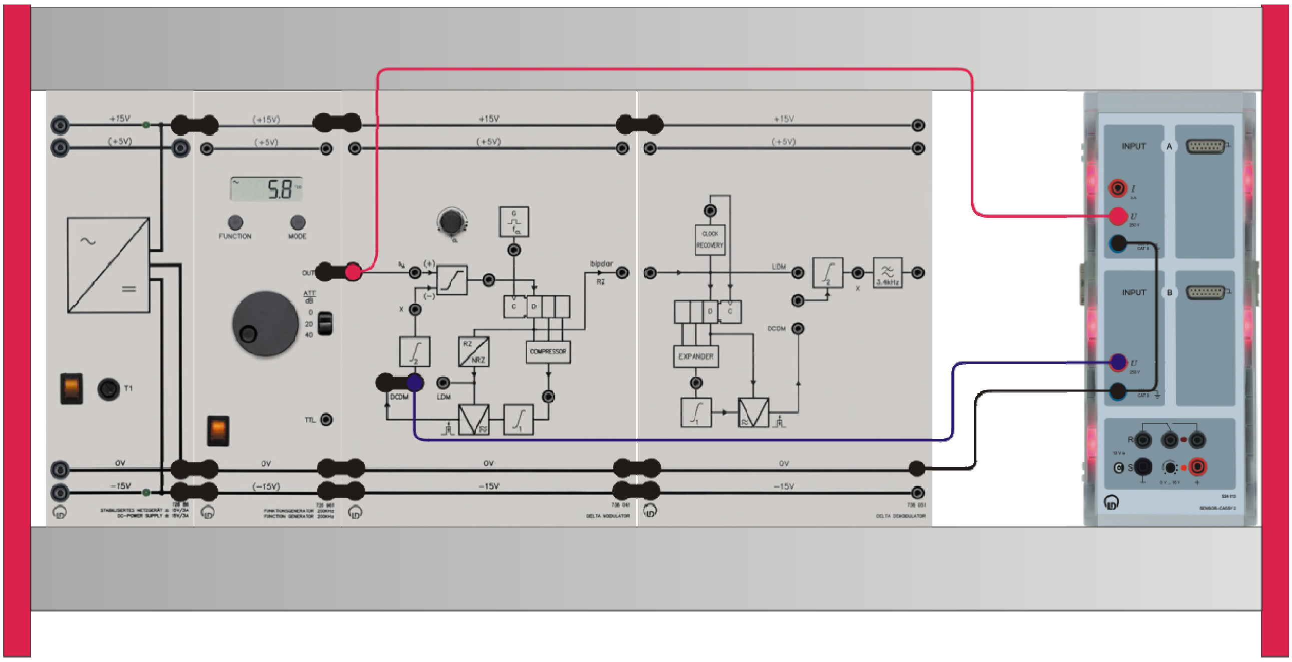
- Sketch the signals and in diagram 1-3.
- Reduce the clock frequency to f_{CL} = 10 kHz (min.)
- Repeat the measurement.
- Sketch the signals and in diagram 1-4.

- The dependency of the prediction signal X(t) on the sampling frequency is similar in DCDM.

Experiment 2: PAM pulse heights

◆ **Pulse height in DCDM**

Experiment set-up:



Settings on the DM system / function generator

Clock frequency f_{CL}	50 kHz, see below
Modulating signal $s_M(t)$	sine, 100 Hz, 7 V_{pp}, ATT = 0 dB
Delta modulator	DCDM, set the bridging plug
Delta demodulator	----

- Connect the Sensor-CASSY 2 inputs:
  Input A → s_M modulating signal
  Input B → DCDM signal at the output of the PAM modulator
- Start the measurement by pressing F9.
- Jointly display on the monitor $s_M(t)$ and the DCDM signal.
- Set the clock frequency to approximately $f_{CL} \approx 30\ldots50$ kHz, by setting the potentiometer into the middle position. Then, the pulse signals show the smallest eye aperture (smallest pulse heights in the waists).
- Sketch the signals in diagram 2-1.

> - The pulse height is dependent on the modulating signal.
> - It is at its maximum where $s_M(t)$ has its extremes and its minimum at the zero-crossovers of $s_M(t)$.

◆ **Pulse height in LDM**

Settings on the DM system / function generator

Clock frequency f_{CL}	50 kHz, same frequency as for DCDM
Modulating signal $s_M(t)$	sine, 100 Hz, 7 V_{pp}, ATT = 0 dB
Delta modulator	LDM, set the bridging plug
Delta demodulator	----

- Repeat the measurements.
- Sketch the signals in diagram 2-2.

- Modulating signal $s_M(t)$ and pulse signal in LDM
- Clock frequency approx. f_{CL} = 40 kHz
- The pulse height is independent of $s_M(t)$. The LDM modulator shows the highest rate of activity in the range of the greatest slopes of $s_M(t)$.

Results

Experiments 1 & 2

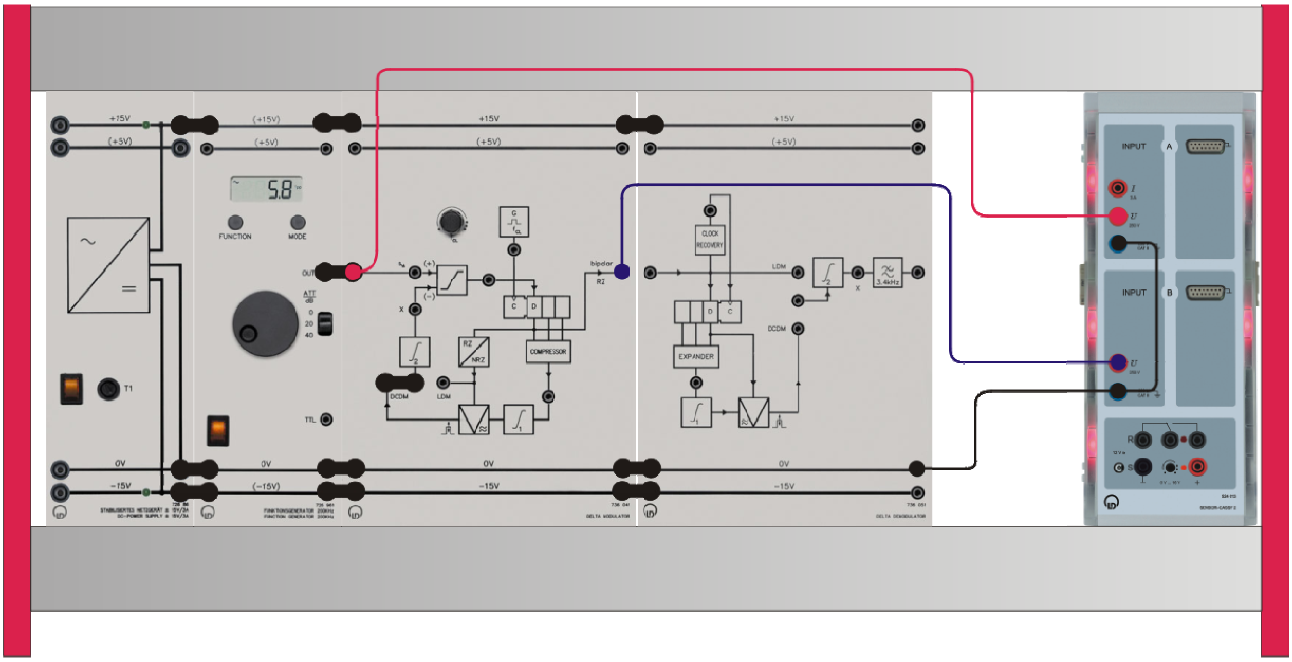
Diagram 1-1: Modulating signal sM(t) and prediction signal X(t) in LDM for clock frequency fCL=100 KHz	
Diagram 1-2: Modulating signal sM(t) and prediction signal X(t) in LDM for clock frequency fCL=10 KHz	
Diagram 1-3: Modulating signal sM(t) and prediction signal X(t) in DCDM for clock frequency fCL=100 KHz	

Diagram 1-3: Modulating signal sM(t) and prediction signal X(t) in DCDM for clock frequency fCL=10 KHz	
Diagram 2-1: PulseAmplitude_DCDM_50kHz clock frequency	
Diagram 2-2: PulseAmplitude_LDM_50kHz clock frequency	

Experiment 3: DM signals

◆ **Output signals of the DCDM modulator**

Experiment set-up:



Settings on the DM system / function generator	
Clock frequency f_{CL}	30 kHz (approx.)
Modulating signal $s_M(t)$	sine, 100 Hz, 4 V_{pp}, ATT = 0 dB
Delta modulator	DCDM, set the bridging plug
Delta demodulator	----

- Connect the Sensor-CASSY 2 inputs:
  Input A → s_M modulating signal, function generator
  Input B → DM signal at the output of DM modulator (bipolar, RZ)

- Start the measurement by pressing F9.
- Jointly display on the monitor $s_M(t)$ and the DM signal.

- Start the measurement by pressing F9.
- Display $s_M(t)$ and the bipolar output signal of the delta modulator on the monitor
. • Sketch the signals in diagram 3-1.

- Modulating signal $s_M(t)$ and bipolar DM signal in DCDM coding.
- Balanced switching between ±1 signals.
- The DCDM signal is generated in RZ format.
- The quasiternary DM signal permits simple clock pulse recovery with the aid of an EXOR gate in the delta demodulator.

◆ **Output signals of the LDM modulator**

Settings on the DM system / function generator

Clock frequency f_{CL}	30 kHz (approx.)
Modulating signal $s_M(t)$	sine, 100 Hz, 4 V_{pp}, ATT = 0 dB
Delta modulator	LDM, set the bridging plug
Delta demodulator	----

- Repeat the measurements.
- Sketch the signals in diagram 3-2.

- The LDM signal demonstrates gaps.
- The comparator apparently gets stuck in one state with the zero crossover of the modulating signal $s_M(t)$, because the prediction signal X(t) is always too small.
- For a certain time during strong positive or negative signals, there is no transition in the DM coded signal (saturation).

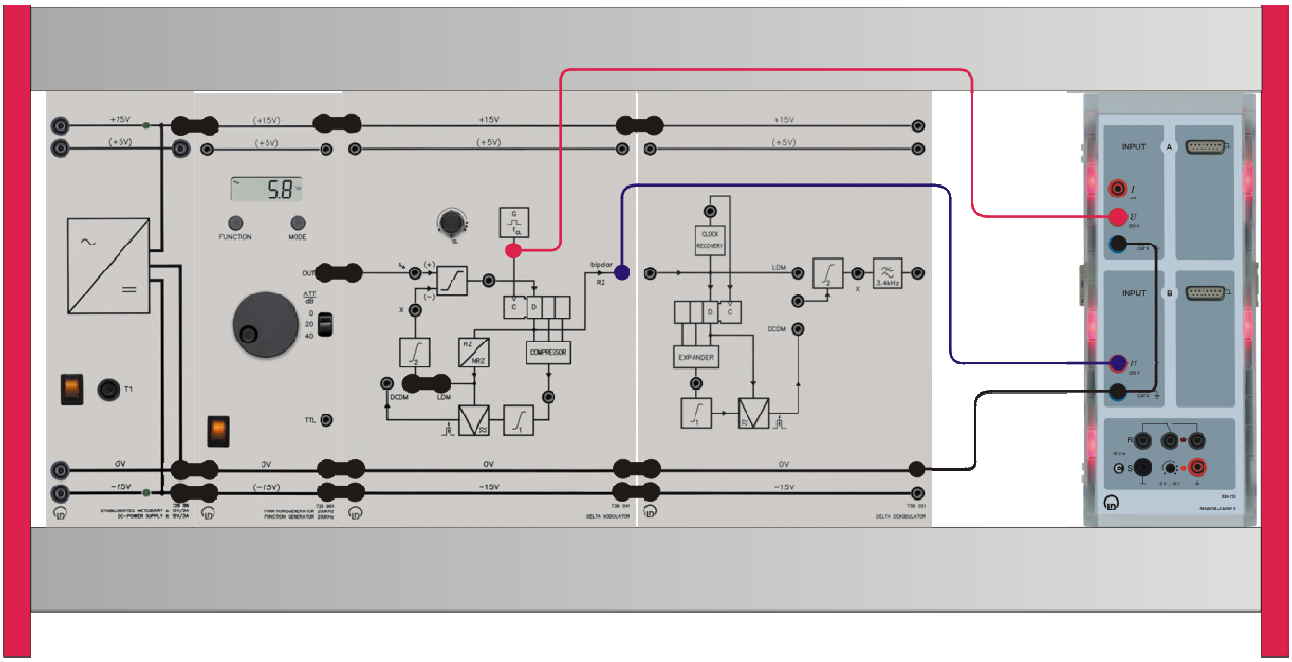
Experiment 4: Clock and DM signals

Settings on the DM system / function generator	
Clock frequency f_{CL}	10 kHz (min.)
Modulating signal $s_M(t)$	sine, 200 Hz, 2 V_{pp}, ATT = 0 dB
Delta modulator	LDM, set the bridging plug
Delta demodulator	----

The type of DM signals is not important in this experiment. Thus no distinction is made between LDM and DCDM mode. Consequently the experiments are carried out in LDM mode only.

◆ **DM signals in RZ format**

Experiment set-up:

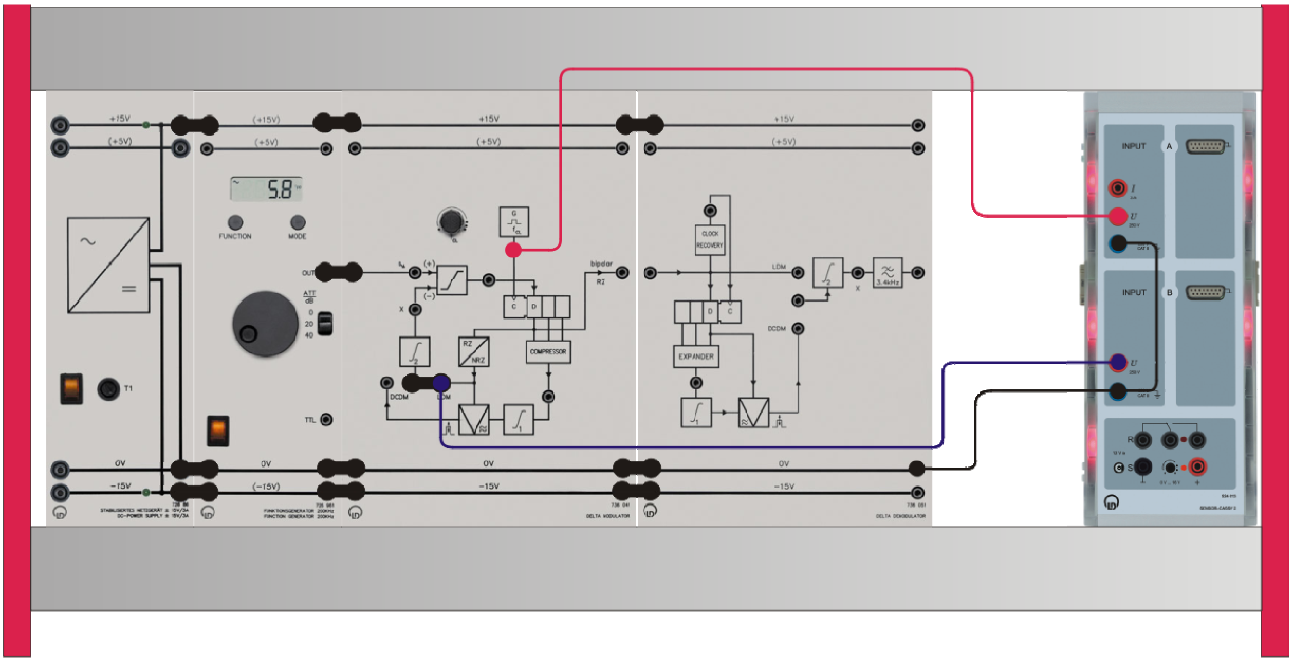


- Connect the Sensor-CASSY 2 inputs:
  Input A → Clock generator (G), f_{CL}
  Input B → DM output (bipolar, RZ)
- Start the measurement by pressing F9.
- Display the clock signal and the bipolar output signal of the delta modulator.
- Sketch the signals in diagram 4-1.

> - The DM signal is sampled at double the clock frequency $2f_{CL}$.
> - The RZ DM signal is with reference to ground.
> - This format is called Return to Zero (RZ)

◆ DM signals in NRZ format

Experiment set-up:



- Connect the Sensor-CASSY 2 inputs:
  Input A → Clock generator (G), f_{CL}
  Input B → LDM output at the RZ/NRZ-converter
- Display the LDM signal in NRZ format and the bipolar output signal of the delta modulator on the monitor.
- Sketch the diagrams of the signals.
- Sketch the signals in diagram 4-2.

- During the negative clock period, the DM signal maintains the value of the positive clock period

Results

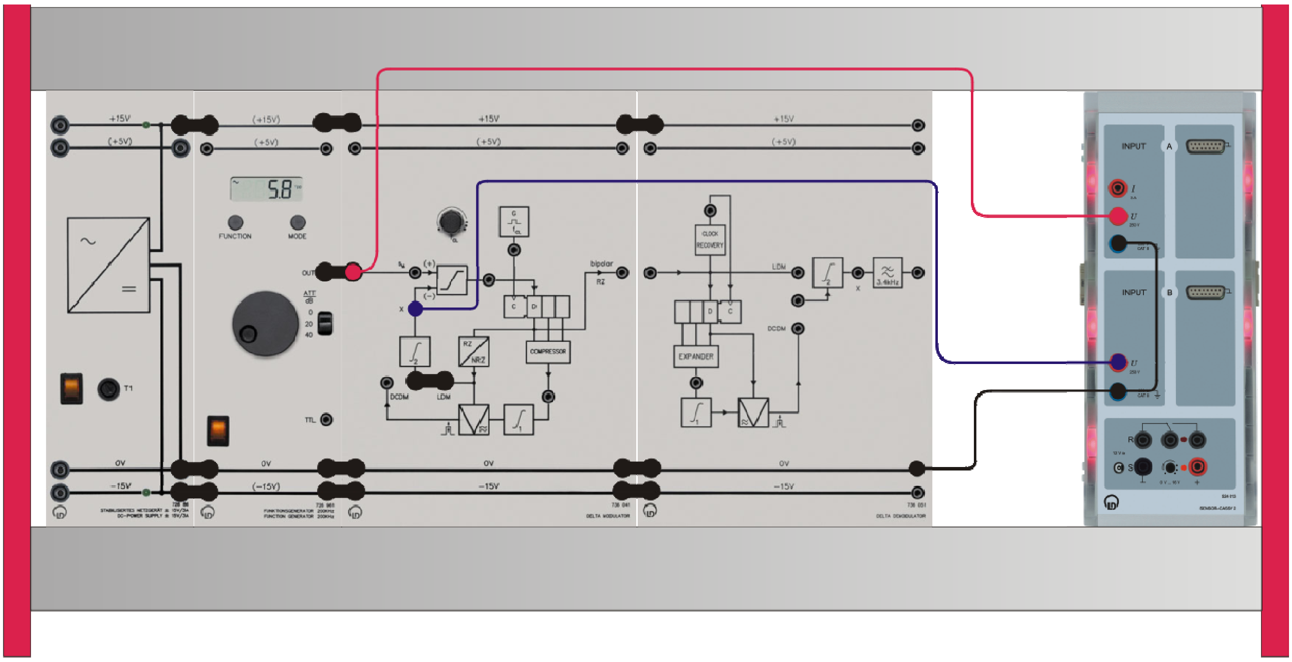
Experiments 3 & 4

Diagram 3-1: DCDMCode_30kHz.	
Diagram 3-2: LDMCode_30kHz.	
Diagram 4-1: LDM_Clock_RZ_10kHz.	
Diagram 4-2: LDM_Clock_NRZ_10kHz.	

Experiment 5: Granular noise

◆ Granular noise in LDM

Experiment set-up:



Settings on the DM system / function generator

Clock frequency f_{CL}	10 kHz (min.)
Modulating signal $s_M(t)$	square 50%, 200 Hz, 2 V_{pp}, ATT = 0 dB
Delta modulator	LDM, set the bridging plug
Delta demodulator	----

- Connect the Sensor-CASSY 2 inputs:
  Input A → modulating signal s_M
  Input B → prediction signal *X*
- Load the CASSY Lab 2 example

- Start the measurement by pressing *F9*.
- Display the modulating signal $s_M(t)$ and the prediction signal X(t) on the monitor. • Sketch the signals in diagram 5-1.
- Repeat the experiment for f_{CL} = 30 kHz and f_{CL} = 100 kHz.
- Sketch the signals in diagrams 5-2/3.

- The granular noise decreases with increasing sampling frequency.

◆ **Granular noise in DCDM**

Settings on the DM system / function generator

Clock frequency f_{CL}	10 kHz (min.)
Modulating signal $s_M(t)$	square 50%, 200 Hz, 2 V_{pp}, ATT = 0 dB
Delta modulator	DCDM, set the bridging plug
Delta demodulator	----

- Repeat the measurements.
- Sketch the signals in diagrams 5-4/6.

Notes for granular noise

- Granular noise decreases in DM with increasing sampling frequency.
- The prediction signal X(t) is saw tooth shaped in the absence of the modulating signal, $s_M(t) = 0$.
- The peak-to-peak value of the saw tooth voltage decreases with increasing sampling frequency.
- Due to the complicated frequency dependency of the DM system there is an optimum sampling rate with minimum granular noise. This sampling rate is smaller than at the maximum sampling rate f_{CL} = 100 kHz.
- Granular noise for $s_M(t) = 0$, is approximately equal in LDM and DCDM for the modulators designed on 736 041.

Results

Experiment 5

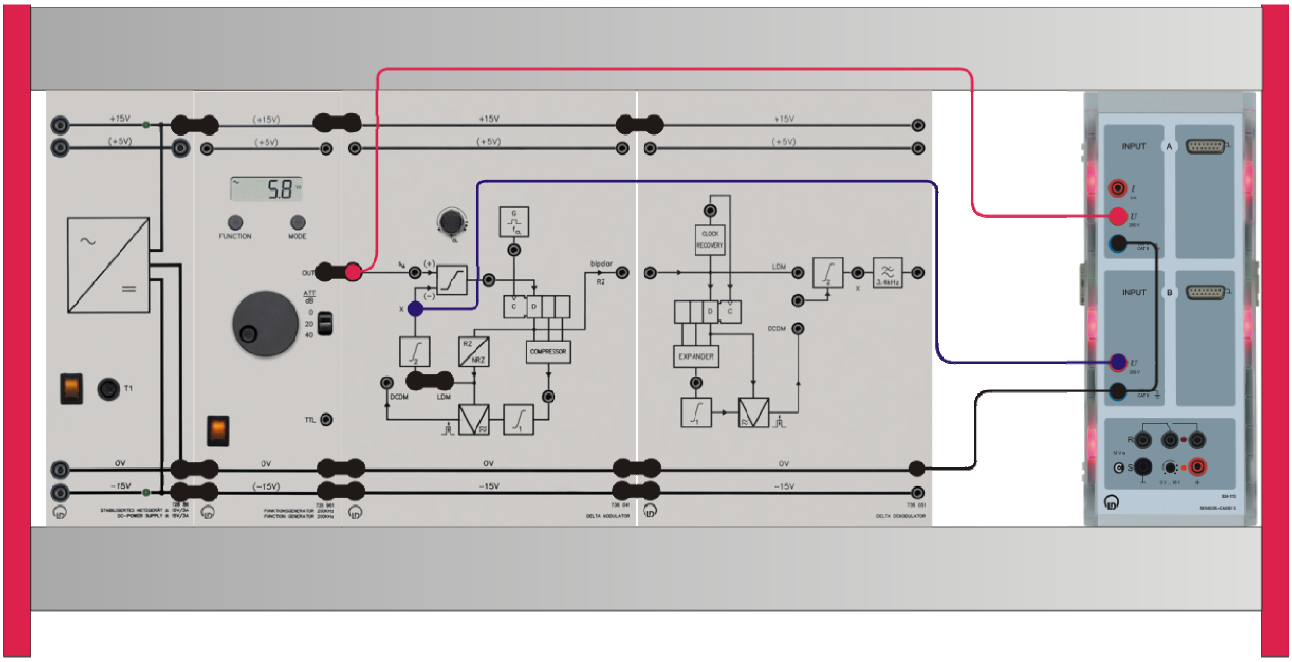
Diagram 5-1: LDM_Granular_10kHz	
Diagram 5-2: LDM_Granular_30kHz	
Diagram 5-3: LDM_Granular_100kHz	

Diagram 5-4: DCDM_Granular_10kHz	
Diagram 5-5: DCDM_Granular_30kHz	
Diagram 5-6: DCDM_Granular_100kHz	

Experiment 6: Slope overload

◆ Slope overload in LDM

Experiment set-up:



Settings on the DM system / function generator	
Clock frequency f_{CL}	100 kHz
Modulating signal $s_M(t)$	sine, 100 Hz, 7 V_{pp}, ATT = 0 dB
Delta modulator	LDM, set the bridging plug
Delta demodulator	---

- Connect the Sensor-CASSY 2 inputs:
  Input A → modulating signal s_M
  Input B → prediction signal X

- Start the measurement by pressing F9.
- Display the modulating signal $s_M(t)$ and the prediction signal X(t) on the monitor.
- Sketch the signals in diagram 6-1.

- Slope overload in LDM, sine modulation.
- The amplitude of the modulating signal is too high (A_M = 7 Vpp).
- Due to the simple low pass integration (Integrator 2: is realized as low pass filter without gain) the amplitude of the prediction value X(t) cannot become higher than the pulse amplitude at the input of the LP filter.
- Thus exponentially distorted charge and discharge curves of a LP filter can be clearly recognized in the prediction signal.

- Set the function generator to: square-wave, 50%, 100 Hz, 4 Vpp.
- Repeat the measurement.
- Use diagram 6-2.

- LDM, square-wave modulation.
- The amplitude of the modulating signal A_M is equal to the pulse height (2 Vpp).
- The frequency f_M of the modulating signals lies at such a low level that the prediction value can approximate the modulating signal.

- Increase the frequency of the square-wave signal to 300 Hz.
- Repeat the measurement.
- Use diagram 6-3.

- LDM, square-wave modulation.
- The amplitude of the modulating signal A_M is equal to the pulse height (2 Vpp).
- The frequency f_M of the modulating signals lies at such a low level that the prediction value can approximate the modulating signal.

◆ Slope overload in DCDM

Settings on the DM system / function generator

Clock frequency f_{CL}	100 kHz
Modulating signal $s_M(t)$	sine, 100 Hz, 7 V_{pp}, ATT = 0 dB
Delta modulator	DCDM, set the bridging plug
Delta demodulator	---

- Repeat the experiment.
- Sketch the signals in diagram 6-4.

> - DCDM, sinusoidal modulation.
> - Due to the pulse heights adapted to the dynamic characteristics of the modulating signal there is a considerably greater dynamic range at low frequencies f_M than is the case in LDM (higher amplitudes A_M are permitted).

- Set the function generator to: square-wave, 50%, 100 Hz, 4 Vpp
- Repeat the measurement.
- Increase the frequency of the square-wave signal to 300 Hz
- Sketch the signals in diagram 6-5/6.

> ***Interpretation*** (diagram 6-5)
>
> - Square wave 50%, 100 Hz, 4 Vpp.
> - DCDM, square-wave modulation.
> - The prediction signal X(t) is built up with considerably steeper slopes.
>
> ***Interpretation*** (diagram 6-6)
>
> - DCDM, square-wave modulation f_M = 300 Hz.
> - LDM was already completely in slope overload for this modulating signal.

Results

Experiment 6

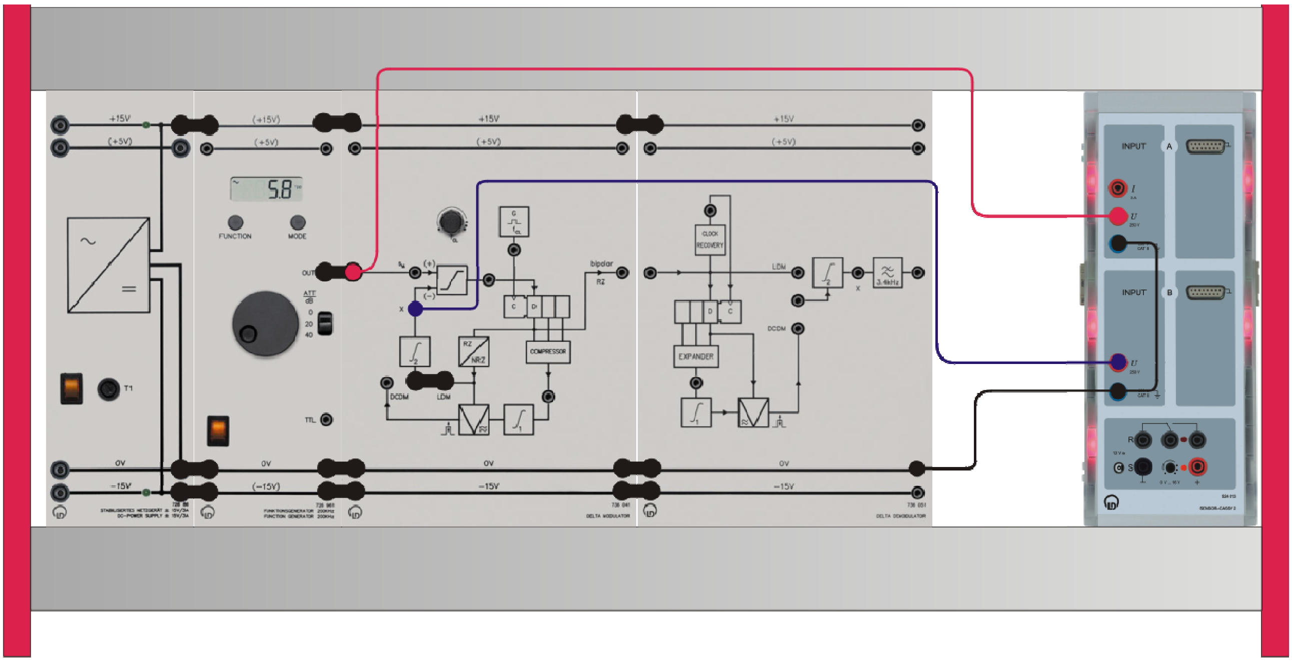
Diagram 6-1: LDM_SlopeOver_100kHz_sine	
Diagram 6-2: LDM_SlopeOver_100kHz_square_100Hz	
Diagram 6-3: LDM_SlopeOver_100kHz_square_300Hz	

Diagram 6-4: DCDM_SlopeOver_100kHz_sine	
Diagram 6-5: DCDM_SlopeOver_100kHz_square_100Hz	
Diagram 6-6: DCDM_SlopeOver_100kHz_square_300Hz	

Experiment 7: Dynamic of LDM and DCDM

The dynamic between LDM and DCDM is compared as a function of the frequency of the modulating signal.

Experiment set-up:



Settings on the DM system / function generator	
Clock frequency f_{CL}	100 kHz
Modulating signal $s_M(t)$	sine, f_M see table, A_M variable
Delta modulator	LDM / DCDM
Delta demodulator	

- Connect the Sensor-CASSY 2 inputs:
  Input A → modulating signal s_M
  Input B → prediction signal X
- Start the measurement by pressing F9.
- Display the prediction signal X(t) on the monitor.
- Set the frequency of the function generator according to the table.
- Vary the amplitude A_M of the modulating signal s_M, until X demonstrates the beginnings of slope overload (recognizable by the exponential distortion).
- Note the value of A_M in the column specified A_{Mmax}.
- Calculate the dynamic D. For A_{Mmin} use the value for granular noise (approx. 20 mVpp).
- Repeat the measurements for DCDM.

Results

	Start of slope overload		Dynamic response	
	LDM	DCDM	LDM	DCDM
f_M/Hz	**A_{Mmax}/mVpp**	**A_{Mmax}/mVpp**	**D/dB**	**D/dB**
100				
200				
300				
400				
500				
1000				
1500				
2000				
2500				
3000				

Clock frequency f_{CL}: 100 kHz

$$D / dB = 20 \cdot \log \frac{A_{M\max}}{A_{M\min}}$$

Results

Experiment 7

Diagram 7-1: Dynamic_LDM.	
Diagram 7-2: Dynamic_DCDM.	

Comparing the dynamics of LDM and DCDM

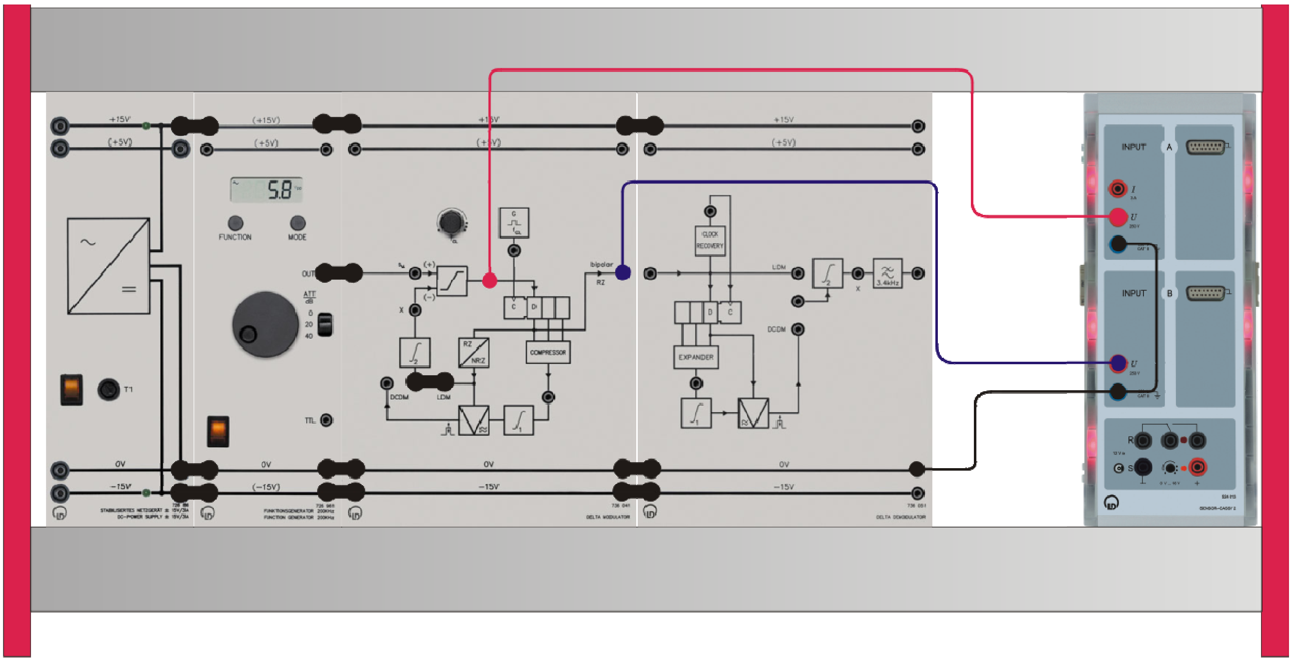
- The dynamic between LDM and DCDM is a function of the frequency f_M of the modulating signal.
- The maximum permissible amplitude A_{Mmax} characterizes the beginning of the slope overload.
- Slope overload additionally depends on the clock frequency f_{CL}.
- The minimum permissible amplitude A_{Mmin} is determined by the granular noise, which also depends on the clock frequency (20 mVpp is a good practical value).
- The dynamics are severely dependent on the value of the granular noise measured.
- The DCDM method implemented on the training panel improves the dynamics by approx. 10 dB over LDM.

Experiment 8: Coding and companding

◆ **LDM: Coding in the DM modulator**

In delta modulation the coding of the differential values is performed by an edge triggered D-flip flop, functioning as a sample and hold circuit. The differential values stem from the comparison of the modulating signal and the prediction signal. With each clock pulse the bit present at the D-input is shifted to the output and stored temporarily until the arrival of the next clock pulse.

Experiment set-up:



Settings on the DM system / function generator

Clock frequency f_{CL}	10 kHz
Modulating signal $s_M(t)$	sine, 100 Hz, 4 V_{pp}, ATT = 0 dB
Delta modulator	LDM, set the bridging plug
Delta demodulator	----

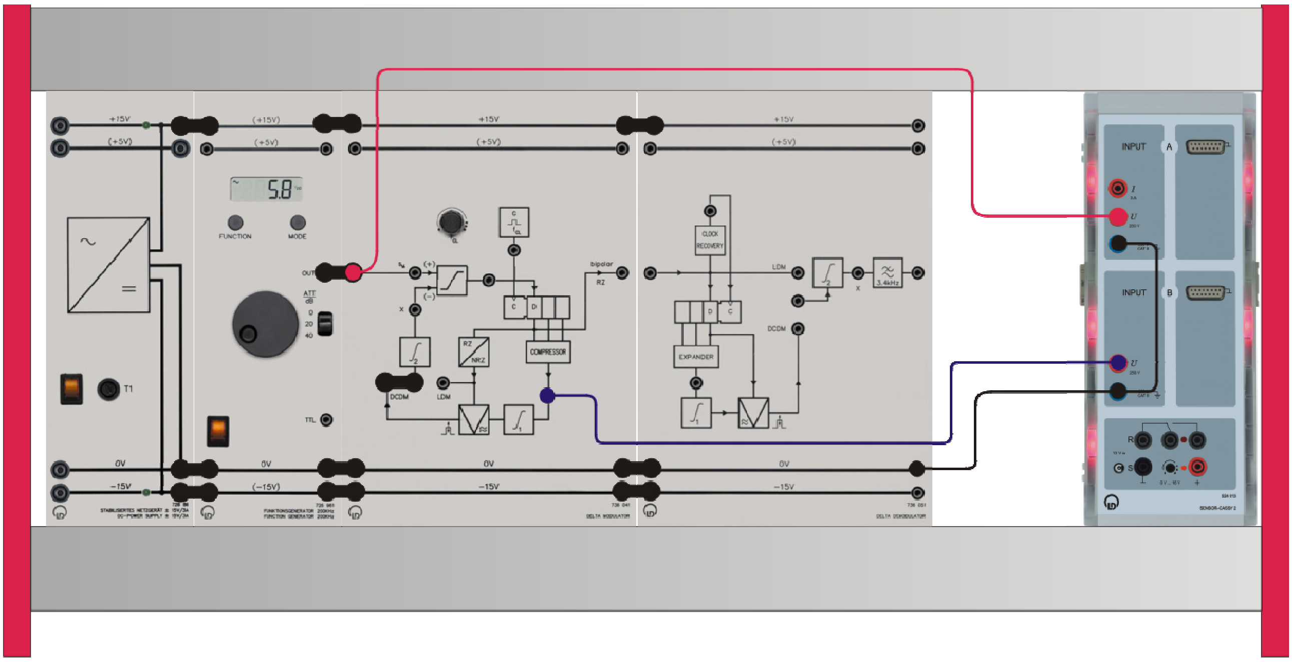
- Connect the Sensor-CASSY 2 inputs:
  Input A → Comparator out (input D of the D-FF)
  Input B → DM output (bipolar/RZ)

- Start the measurement by pressing *F9*.
- Display the DM signal (bipolar, RZ) and the signal at the comparator output on the monitor.
- Sketch the signals in diagram 8-1.

◆ DCDM: Companding in the DM modulator

In the case of DCDM the companding principle is derived from the 0/1-distribution of the bits at the shift register.

Experiment set-up:



Settings on the DM system / function generator

Clock frequency f_{CL}	50 kHz (approx. middle position)
Modulating signal $s_M(t)$	sine, 100 Hz, 15 V_{pp}, ATT = 0 dB
Delta modulator	DCDM, set the bridging plug
Delta demodulator	----

- Connect the Sensor-CASSY 2 inputs:
  Input A → prediction signal X(t)
  Input B → COMPRESSOR

- Start the measurement by pressing *F9*.
- Display the prediction signal X(t) and the signal at the output of the compressor on the monitor.
- Sketch the signals in diagram 8-2.

Interpretation (diagram 8-1)

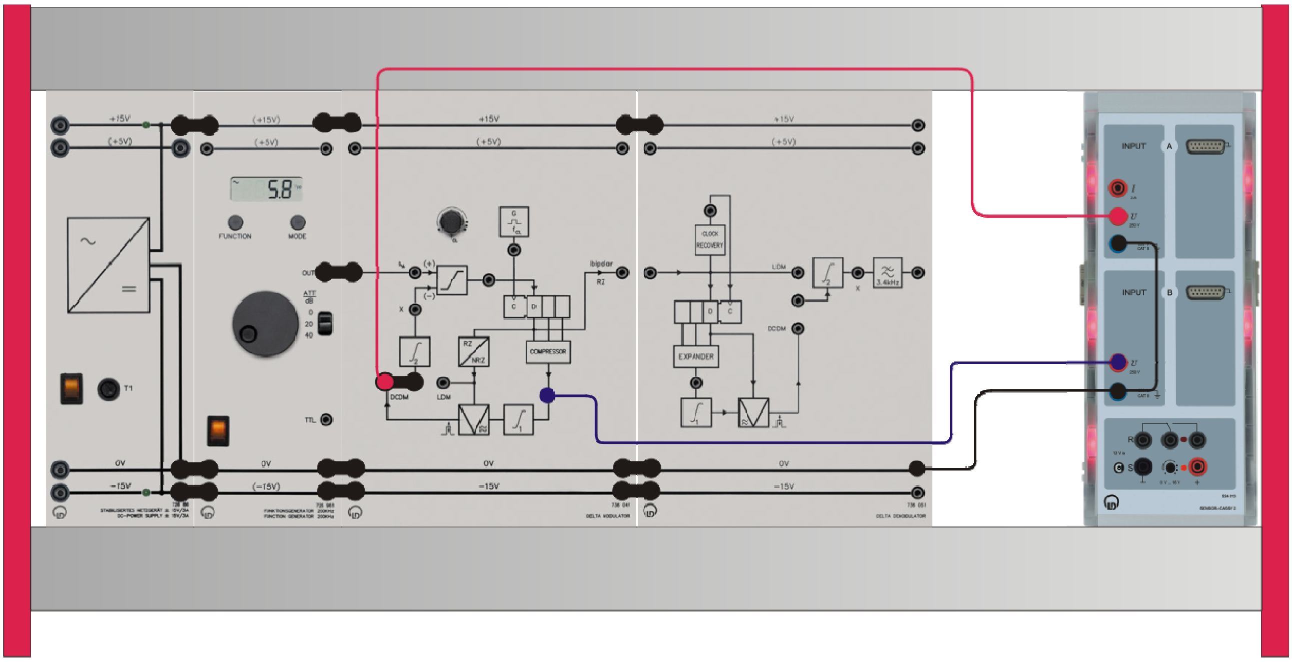
- Coding with the aid of bistable flipflop.
- Between the output signal of the comparator and the DM coded signal there is no fixed time reference.
- While the DM-coded signal is clock-synchronized, the comparator output is always switched over when the condition $s_M(t) \neq X(t)$ is fulfilled regardless of the sampling clock pulse.

Interpretation (diagram 8-2)

- The compression process in DCDM.
- In DCDM the companding law is derived from the 0/1-distribution of the bits at the shift register.
- In the range where the modulating signal $s_M(t)$ has steeper slope a continuous zero sequence is produced.
- Where the slope in $s_M(t)$ is less steep, the output signal of the compressor changes.

◆ DCDM: Compressor & DM signals

Experiment set-up:



Settings on the DM system / function generator

Clock frequency f_{CL}	50 kHz (approx. middle position)
Modulating signal $s_M(t)$	sine, 100 Hz, 15 V_{pp}, ATT = 0 dB
Delta modulator	DCDM, set the bridging plug
Delta demodulator	----

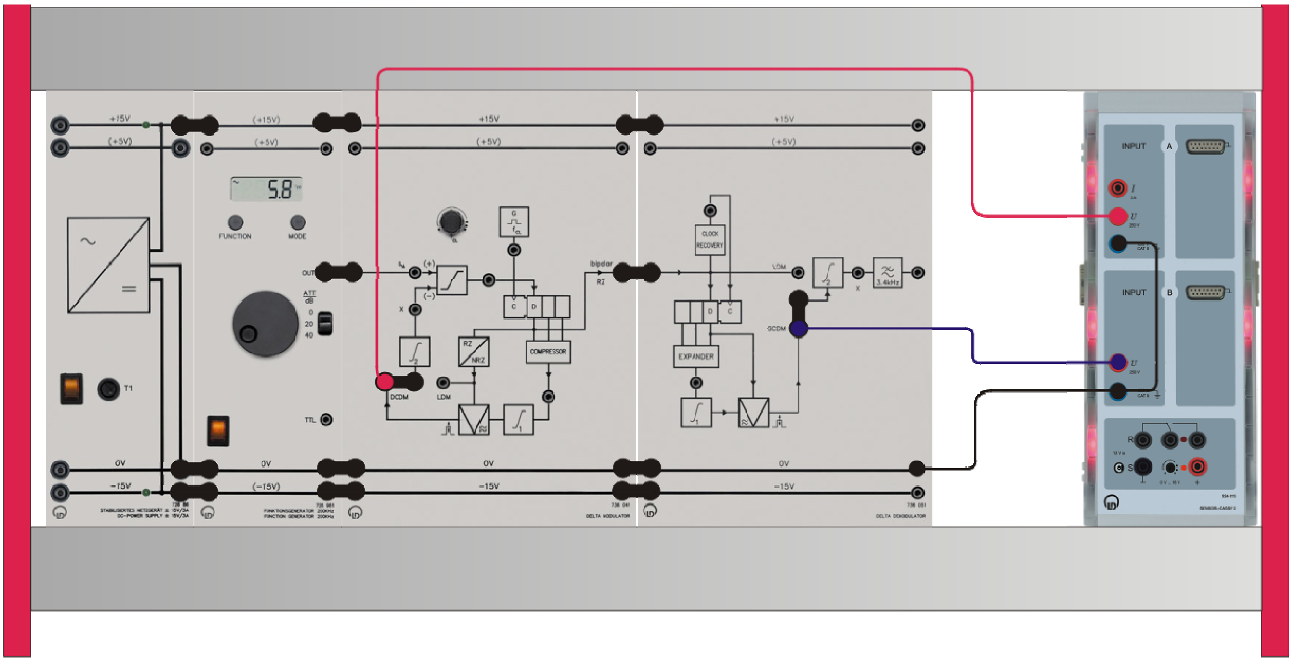
- Connect the Sensor-CASSY 2 inputs:
  Input A → DCDM
  Input B → COMPRESSOR

- Start the measurement by pressing *F9*.
- Display the signals at the outputs of the compressor and the PAM modulator (DCDM output).
- Sketch the signals in diagram 8-3.

> - Compressor output and·pulse signals in DCDM.
> - An increasing sequence of 1- signals at the compressor output is converted into rising pulse amplitudes by the PAM modulator.
> - The PAM modulator additionally has to determine the polarity of the pulses.

◆ DCDM: Recovery of the predicted PAM signals in DM modulator and DM demodulator

Experiment set-up:



Settings on the DM system / function generator	
Clock frequency f_{CL}	50 kHz (approx. middle position)
Modulating signal $s_M(t)$	sine, 100 Hz, 15 V_{pp}, ATT = 0 dB
Delta modulator	DCDM, set the bridging plug
Delta demodulator	DCDM, set the bridging plugs

- Connect the Sensor-CASSY 2 inputs:
  Input A → DCDM (DM modulator)
  Input B → DCDM (DM demodulator)

- Start the measurement by pressing *F9*.
- Display the PAM signals (DCDM) in the delta modulator and delta demodulator on the monitor

- Sketch the signals in diagram 8-4.

Results

Experiment 8

Diagram 8-1: LDM_Code_10kHz	
Diagram 8-2: DCDM_X_Compand_50kHz	
Diagram 8-3: DCDM_Compressor	
Diagram 8-4: DCDM_PAM_PAM_50kHz	

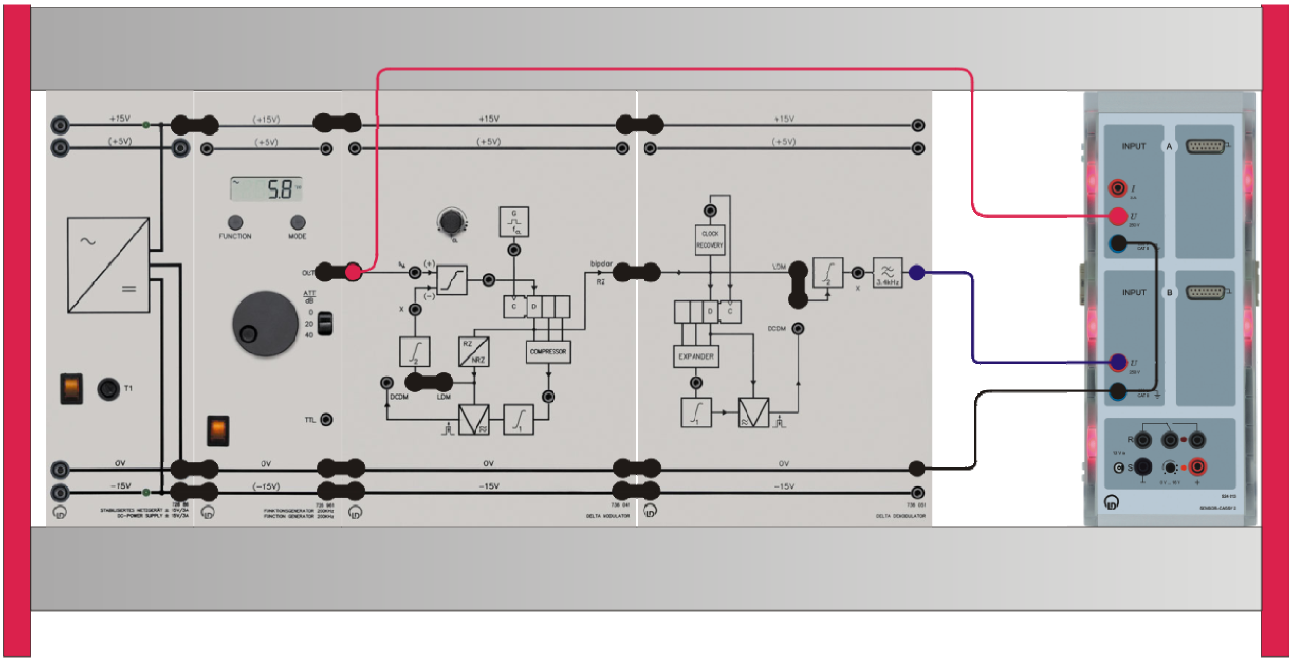
Note: Coincident traces!

- Jointly representation of the PAM signals in the DM modulator and DM demodulator
- The PAM signals (DCDM) in the delta modulator and the recovered pulse of the delta demodulator are identical.

Experiment 9: Demodulation

◆ **LDM demodulation**

Experiment set-up:



Settings on the DM system / function generator

Clock frequency f_{CL}	100 kHz
Modulating signal $s_M(t)$	triangular, 300 Hz, 4 V_{pp}
	square 10%, 100 Hz, 4 V_{pp}, ATT = 0 dB
Delta modulator	LDM, set the bridging plug
Delta demodulator	LDM, set the bridging plugs

- Connect the Sensor-CASSY 2 inputs:
  Input A → modulating signal s_M at the input of the DM modulator
  Input B → demodulated signal s_D at the output of the DM demodulator
- Start the measurement by pressing *F9*.
- Display the modulating signal $s_M(t)$ and the demodulated signal $s_D(t)$ on the monitor and sketch these in diagram 9-1.

- LDM demodulation according to the double integrator method, triangular modulation.
- In the demodulated signal there are slight distortions, which can be attributed to the onset of slope overload.

- Switch the function generator to square wave 10%, see above. Demodulation distortion can be displayed better with this signal.
- Display $s_M(t)$ and $s_D(t)$ on the monitor again and sketch the signals in diagram 9-2.

> - LDM demodulation according to the double integrator method, pulse modulation.
> - Demodulation distortion can be displayed better with a pulse-shaped, modulating signal.
> - The signal demodulated using the LDM demodulator shows severe amplitude errors

◆ DCDM demodulation

Settings on the DM system / function generator

Clock frequency f_{CL}	100 kHz
Modulating signal $s_M(t)$	square 10%, 100 Hz, 4 V_{pp}, ATT = 0 dB
Delta modulator	DCDM, set the bridging plug
Delta demodulator	DCDM, set the bridging plug

- Switch the delta modulator and demodulator to DCDM operation by reconnecting the bridging plugs at DM modulator and DM demodulator.
- Display $s_M(t)$ and $s_D(t)$ on the monitor and sketch the signals in diagram 9-3.

> - DCDM demodulation.
> - The transmitted pulse signal is reproduced considerably better in DCDM than in LDM.

Results

Experiment 9

Diagram 9-1: LDM_Dem_Trian_100kHz	
Diagram 9-2: LDM_Dem_Square_10_100kHz	
Diagram 9-3: DCDM_Dem_Square_10_100kHz	

Telecommunication Department
Communications Lab
EXP. 8 Pulse Time Modulation

1. Introduction

1.1 Pulse time modulation methods

Pulse duration modulation (PDM) and pulse position modulation (PPM) are jointly referred to as pulse time modulation methods (PTM). As in analog methods of angle modulation the wanted information is not contained in the amplitude of the modulated signal in PTM. Thus the susceptibility for distortion caused by noise on the transmission channel is reduced. The similarities between analog angle modulation (FM, PM) and pulse-time modulation also include the modulation spectra. In the fulfillment of the sampling theorem and suitability for time multiplexing the same prerequisites apply for PTM as for PAM.

1.2 Pulse duration modulation (PDM)

In PDM, the pulse duration T_0 is used as the variable parameter for the transmission of information. This can occur in three different ways:

1. The leading pulse edge is modulated.
2. The trailing edge is the information carrier.
3. Both pulse edges are altered by the modulating signal.

The figure gives a graphic depiction of the three methods. Of the various methods used to generate PDM signals, we shall limit our description here to the details of the saw tooth method implemented on the PTM modulator training panel, 736 081. The starting point for PDM generation is a PAM-signal.

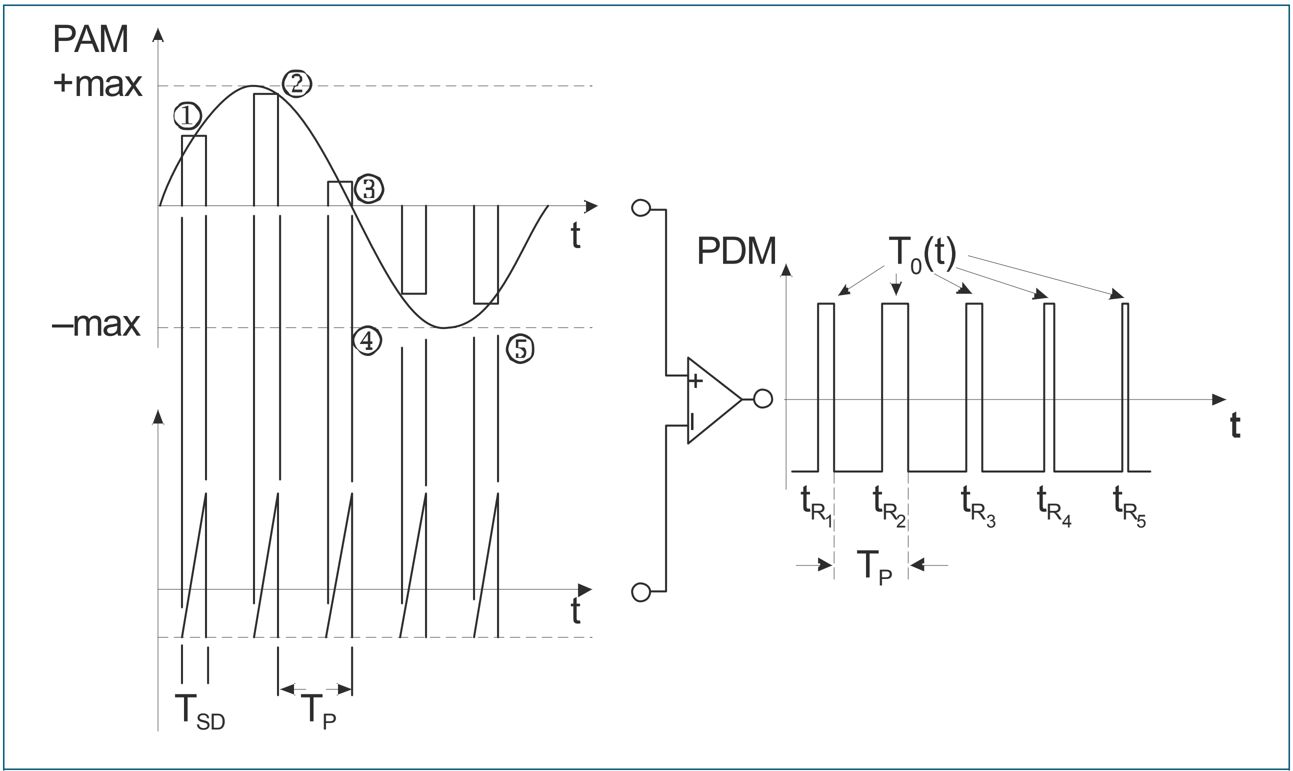


Fig. 1: Generation of PDM according to the sawtooth method
The PAM pulses ① … ⑤ are converted into positions of the trailing edges t_{R1} … t_{R5}

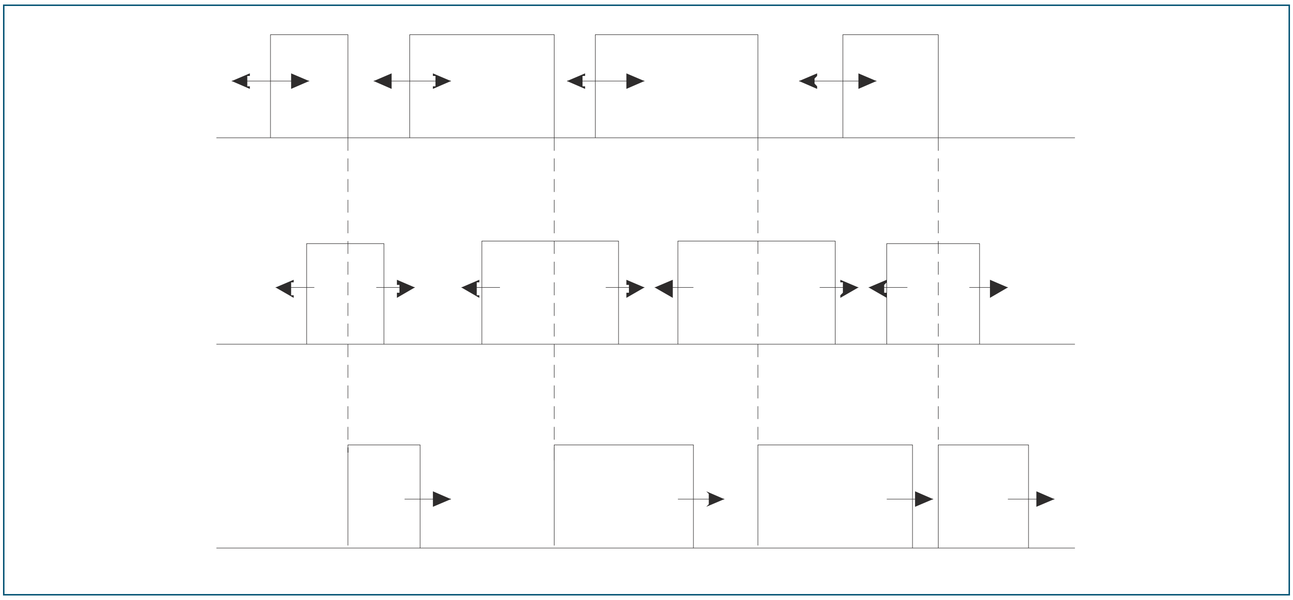


Fig. 2: Three methods for generating PDM signals

Three methods for generating PDM signals

This signal is applied to the non-inverting input of a comparator. The inverting input is connected to a ramp generator, which is started by the sampling process. CLOCK and RAMP signals are synchronized. The start level of the ramp signal coincides with the negative modulating limit of the PAM modulator. The positive modulating limit corresponds to the ramp amplitude. Depending on the momentary height of the PAM pulse, the comparator is tripped sooner or later by the time-linear rising voltage of the ramp signal. This is how pulses of varying width arise at the output of the comparator, whose trailing edges vary in synchronization with the modulating signal. The PAM signal serves here as a switching threshold for the comparator, controlled by the modulating signal $s_M(t)$.

- T_{SD}: Ramp duration
- T_p: Sampling period
- t_R: Time point of the trailing edge in PDM.

The following relationships can be derived:

- The leading edges of the PDM pulses are synchronous with the PAM pulses.
- The time position of the trailing edge t_R is a function of the momentary value of the PAM signal.
- The maximum duration of the PDM pulse is identical to the ramp duration T_{SD}.

The maximum or minimum pulse duration occurs with largest positive or negative PAM signals. The maximum variation which the pulse duration T_0 can experience is called time deviation Δτ. In the case of modulation with a harmonic input signal, the momentary pulse duration T_0 for PDM is:

$$T_0 = \tau + \Delta\tau \cos(2\pi f_M t_R) \qquad 1.2\text{-}1$$

The time deviation is proportional to the signal amplitude A_M:

$$\Delta\tau = k \times A_M \qquad 1.2\text{-}2.$$

1.3 Pulse Position Modulation (PPM)

Compared to PAM, PDM exhibits a lower susceptibility for noise. The relationships existing for PPM are even better and thus even more suitable for long-distance transmissions. Here the pulses are generated with both constant pulse amplitude A_p as well as fixed pulse duration T_0. The information of the modulating signal is contained in the relative time position of the pulses with respect to the pulse frame T_p. The PPM is derived from the PDM by means of differentiation, rectification and limiting. In PPM narrow pulses are used, which only require low transmission power. Furthermore, there are advantages for multiplex operation. The reference time is lacking in the PPM signal. For that reason the PPM pulses have to be transmitted in a pulse frame. An important parameter for the description of PPM is the phase deviation (not to be confused with the variable of the same name used in analog phase modulation). The phase deviation Δt reflects the maximum time shift of the pulses within the pulse frame T_p. The modulation index m_{PPM} can be defined using the phase deviation Δt:

$$m_{PPM} = \frac{\Delta t}{T_p} \qquad 1.3\text{-}1$$

1.4 Pulse amplitude modulation (PAM)

The generation of a PTM signal is performed via PAM formation as an intermediate step. In this process, a sample and hold element triggered by a pulse train $s_P(t)$ is used to "hack" the modulating signal $s_M(t)$ into pulses with the width τ. This process is called time discretization. The PAM signal only appears at certain discrete times. It is zero during the pulse intervals. Thus the following holds true:

- The PAM signal is time-discrete and level continuous.
- The PAM signal is not an analog signal, but it is al so not a digital signal.

In practical terms the PAM signal has significance only as an intermediate stage in the generation of other pulse modulations. The generation of a PAM modulation is often discussed for the particular case of a harmonic input signal $s_M(t)$. Thus, the PAM signal can be represented by the following Fourier series.

$$s_{PAM} = A_M \frac{\tau}{T_p} \frac{\sin(\pi f_M \tau)}{\pi f_M \tau} \cos(2\pi f_M t) + \sum_{n=1}^{\infty} A_M \frac{\tau}{T_p} \frac{\sin[\pi(nf_p \pm f_M)\tau]}{[\pi(nf_p \pm f_M)\tau]} \cos[2\pi(nf_p \pm f_M)t] \qquad 1.4\text{-}1$$

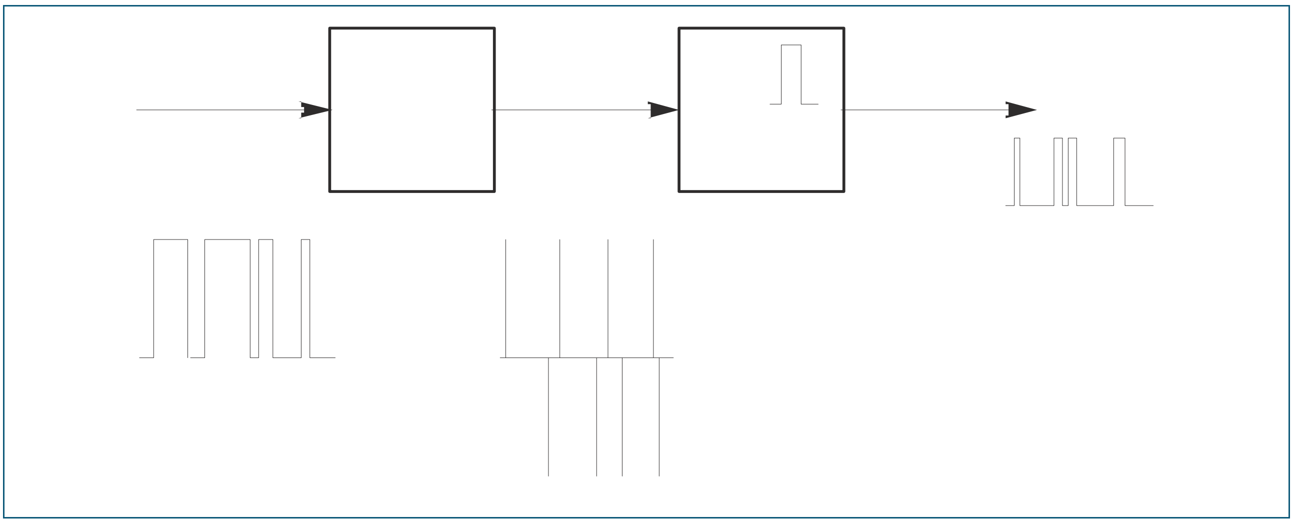


Fig. 3: Generating PPM on basis of PDM

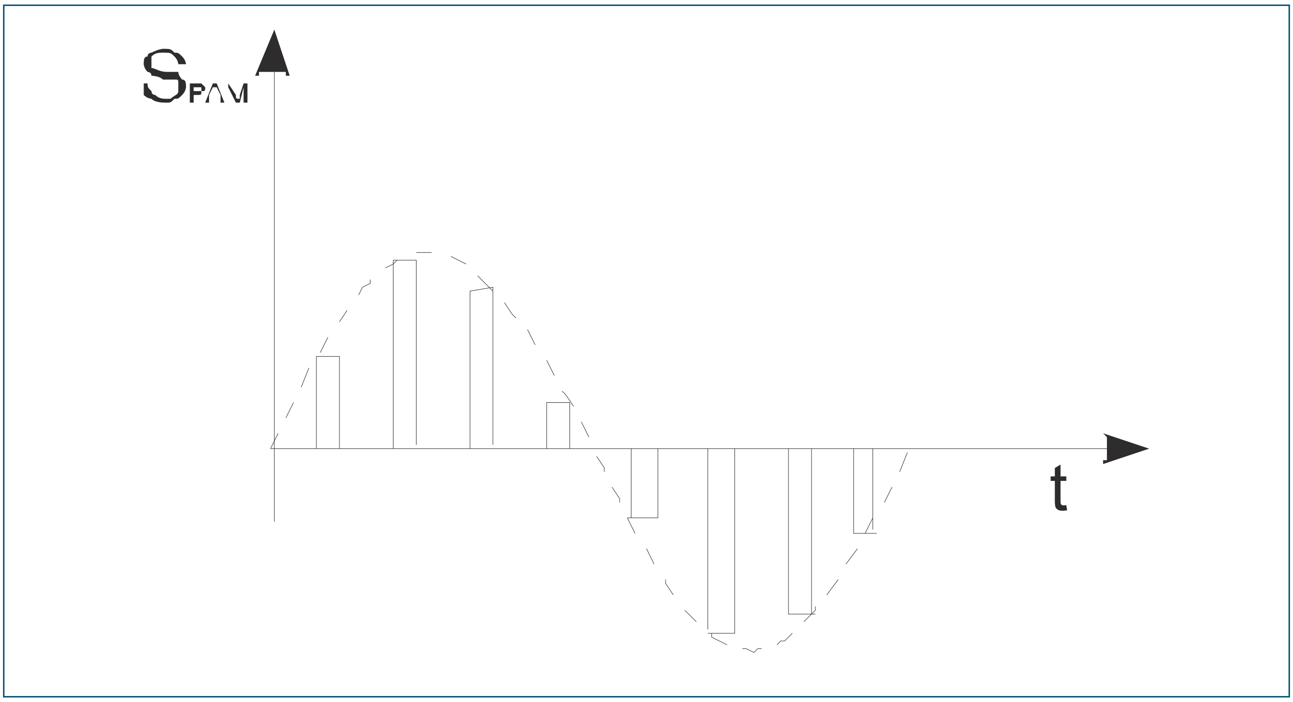


Fig. 4: Generation of PAM

The PAM spectrum contains:

- The modulating signal $s_M(t) = A_M \cos(2\pi f_M t)$ multiplied by the evaluation factor $\tau/T_p\, si(\pi f_M \tau)$. It causes an additional, signal frequency-dependent attenuation.
- Infinitely expanded line spectrum with side lines at the frequencies $f = nf_p \pm f_M$
- The bipolar PAM suppresses the carrier lines. There arise upper and lower side lines. The sidelines also demonstrate a signal frequency dependent attenuation, which rises with increasing signal frequency f_M.

Characteristic for the PAM spectrum is the periodic repetition (with the pulse frequency f_p) of the spectrum of the modulating signal $s_M(t)$. Now if the signal frequency f_M is increased at constant pulse frequency f_p, then the sidelines separate themselves from their suppressed carriers. When $f_M = f_p/2$ is reached, then the respective lower sideline of the partial spectrum *n+1* and the upper sideline of the partial spectrum *n* coincide. If f_M increases still further, then the partial spectra even overlap (aliasing)!

The sampling theorem
In order to avoid aliasing the following must apply for the sampling rate:

$$f_p > 2f_M \qquad 1.4\text{-}2$$

Then at least two samples are dropped for each period of the input signal. The receiver can reconstruct the input signal $s_M(t)$ completely from these two samples. Aliasing is prevented through the band limiting of the modulating signal $s_M(t)$.

1.5 The time-division multiplex method

During the sampling of a signal time gaps arise when no information is transmitted on the transmission channel. This particular time between two respective sampling pulses of a source can be used for transmitting the information of other sources. If the samples of various sources are subsequently sent to the transmission channel, then you have a multiple utilization of the channel (referred to as time division multiplex = TDM). PTM and PAM signals are both suitable for the time division multiplex method.

1.6 The PTM spectrum

A PTM spectrum has a complicated structure similar to that of a FM spectrum. For that reason only the general characteristics should be listed here.

- A PTM spectrum contains the spectrum of the modulating signal $s_M(t)$ = $A_M \cos(2\pi f_M t)$, i.e. there is a line appearing at the signal frequency f_M.
- A PTM spectrum contains the pulse carrier lines evaluated with the Bessel functions $J_0(m)$ each at $f = nf_p$.
- Additional sidelines of the modulating signal $s_M(t)$ spread out around each carrier line, i.e. lines appear at the frequencies $f = nf_p \pm qf_M$.

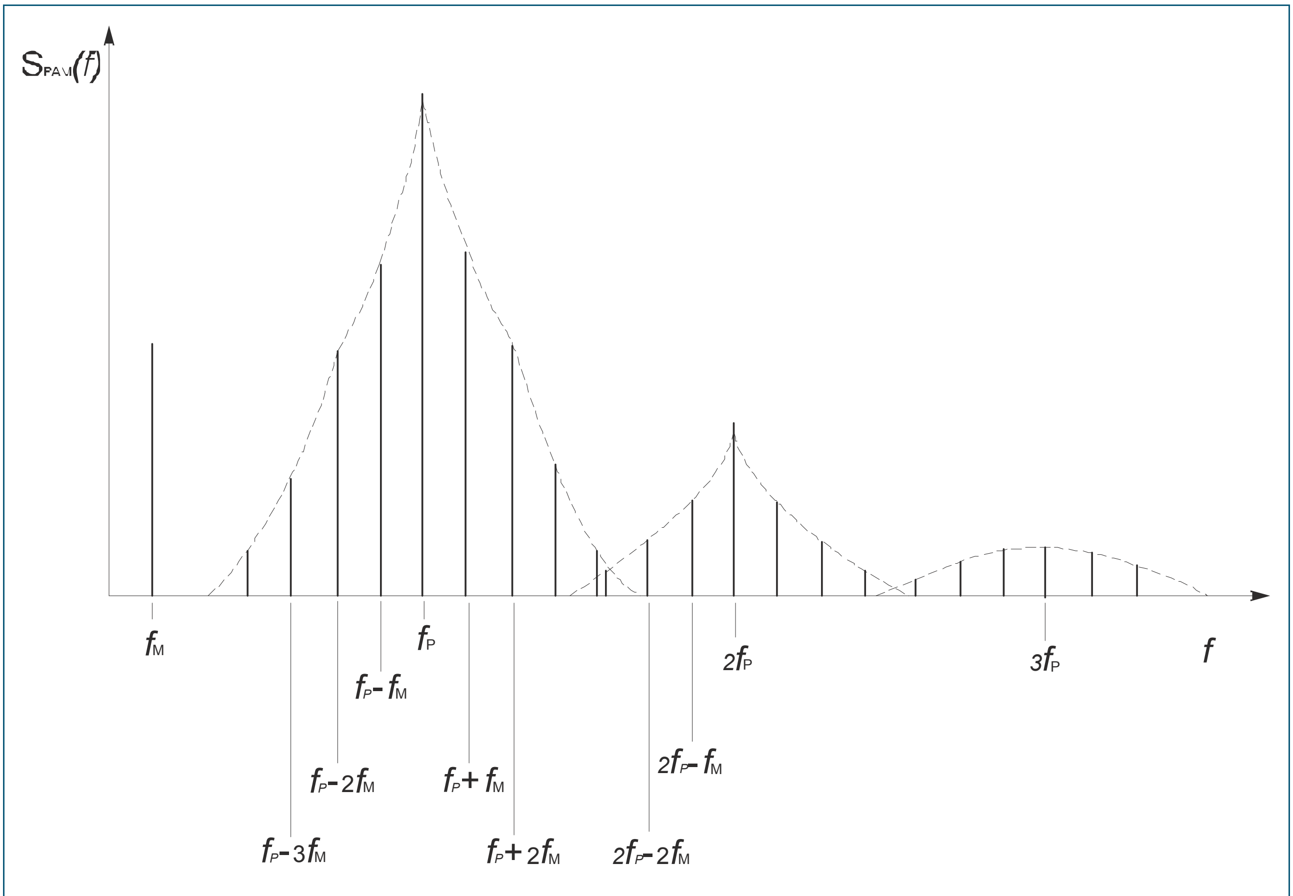


Fig. 5: PTM spectrum

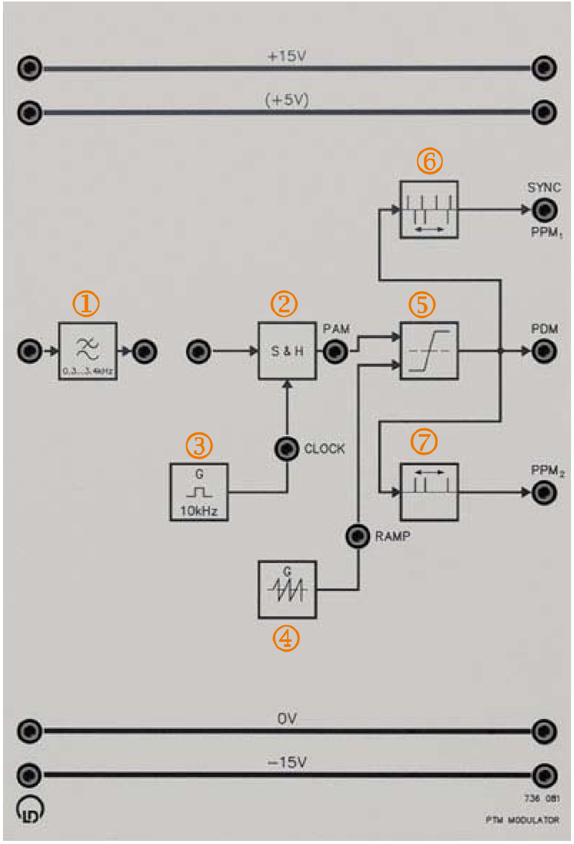
PTM spectrum

These statements apply for modulation with a harmonic signal $s_M(t) = A_M \cos(2\pi f_M t)$. The modulation index m is important for the determination of the PTM spectrum. For example with PDM we obtain the modulation index of PDM as the quotient of the time deviation $\Delta\tau$ and sampling period T_p:

$$m_{PDM} = \frac{\Delta\tau}{T_p} \qquad 1.6\text{-}1$$

The spectrum of the modulating signal $s_M(t)$ does not linearly influence the PTM spectrum. Instead it is expressed by the rather complicated Bessel functions. Thus pulse time modulation methods belong to the non-linear modulation methods like frequency and phase modulation. Due to the characteristics mentioned in points 2 and 3, there is always some overlapping of the various partial spectra in the PTM spectrum. This applies generally and cannot be attributed to aliasing, i.e. sampling errors. For that reason, low pass filtering as a demodulation method can only be used for small modulation indices $m \leq 3\%$, because then there are only less distinct interfering sidelines. If the prerequisite of small modulation indices is not given, then the demodulation has to be performed by reconverting the signal into a PAM signal. So-called sample and hold circuits are used for this.

2. Instruments



PTM-Modulator (Cat. no. 736 081)

The PTM modulator consists of:

- (1) Input filter 0.3... 3.4 kHz, V = 1
- (2) Sample and hold element
- (3) Clock generator, sampling frequency: 10 kHz
- (4) Ramp generator
- (5) Comparator
- (6) Differentiating element
  Positive pulses = PPM_1
  Negative pulses = synchronization
- (7) Pulse shaper
  Positive pulses = PPM_2 (no synchronization included)

This device is used to investigate the pulse time modulation methods PDM and PPM. Out of the modulating signal $s_M(t)$ pulse duration modulation (PDM) is achieved through the intermediate sampling step (PAM formation). A subsequent differentiation of the PDM gives the PPM signal. The ramp generator is connected to the negative input of the comparator. Thus, the modulating signal affects the trailing edge (negative slope). In the pulse time modulations realized here, the information needed for the synchronization of the receiver is transmitted together with the wanted data.

3. Questions

3.1 What is the structure of the spectrum of a pulse train?

3.2 What does the envelope curve of the pulse spectrum look like?

3.3 Why is the PDM method less susceptible to interference than PAM?

3.4 Is PDM one of the linear modulation methods?

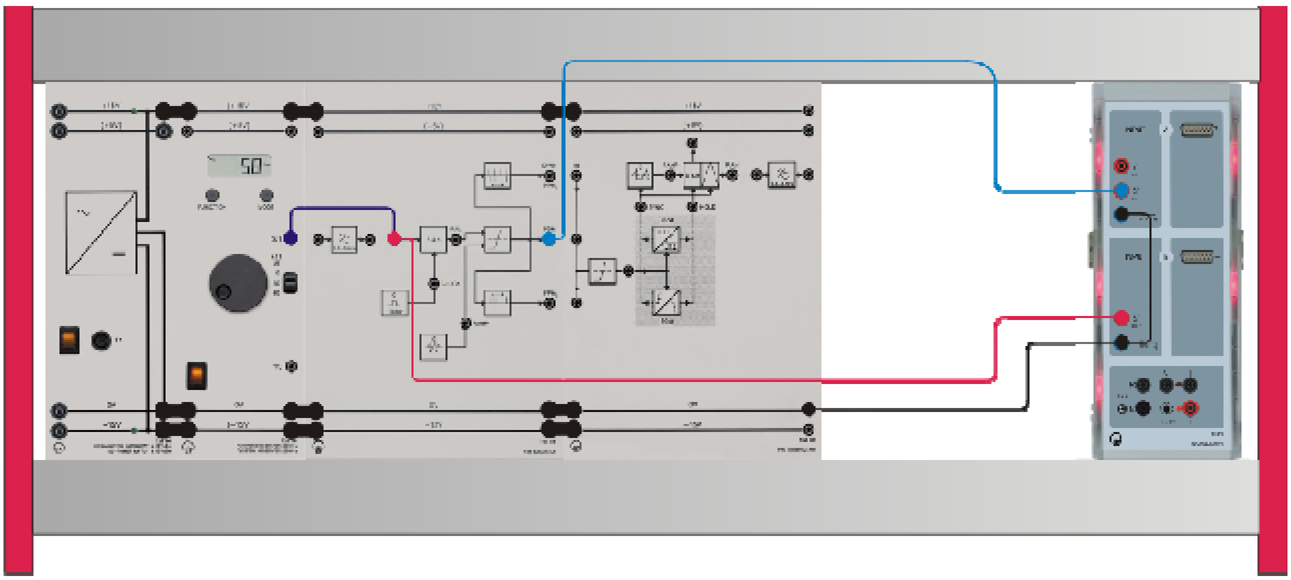
3.5 Under what circumstances is LP demodulation of the PDM signal possible?

3.6 How is PPM generated?

4. Experiment procedure

4.1 Investigating PDM

Experiment setup



4.1.1 PDM – time domain

- Set up the experiment as specified above.
- Set the function generator to: sine, 1.000 kHz, 20.0 Vpp.
- Start the measurement by pressing *F9*.
- Display the modulating signal $s_M(t)$ (S&H input) and the PDM signal on the monitor.
- Interpret your results

Results

PDM dynamic characteristics

Red: Modulating signal sM(t)

Black: PDM signal with instantaneous pulse duration T0

Interpretation

- The pulse duration T_0 is proportional to the modulating signal $s_M(t)$.
- It increases with positive values of $s_M(t)$ and becomes smaller with negative values $s_M(t)$.
- PDM generation is performed via the intermediate PAM formation step.
- The saw tooth generator is started synchronously with the appearance of the PAM pulse. Its start level is at -10 V. It establishes the negative modulation limits of the PTM modulator. The saw tooth signal can rise up to +10 V.
- This corresponds to the positive modulation limits of the PTM modulator.

- Depending on the instantaneous value of the PAM signal at the output of the S&H stage the comparator is tripped earlier or later by the positive saturation value (+5 V) with respect to the negative limit (-5 V). This means that the formation of the trailing edge of
- the PDM pulse is controlled by the instantaneous value of the PAM signal.
- An increase in the signal amplitude A_M leads to more pronounced changes in the pulse duration T_0 and vice versa; meaning the pulse duration could be at a maximum when the PAM signal is at a maximum.

4.1.2 PDM - spectrum

- Set up the experiment as specified in above.
- Set the function generator to: sine, 2.000 kHz, 12.0 Vpp.
- Start the measurement by pressing *F9*.
- Sketch the graph of the PDM.
- Interpret the result.

Results

Interpretation

- The spectral line of the modulating signal appears in the spectrum at the frequency $f = f_M$.
- That is why LP demodulation is possible.
- In addition to that, carrier lines appear at multiples of f_p.
- Sidelines appear as double lines around the carrier lines at a frequency interval of $\pm f_M$.

  The situation is similar to SSB-AM with a very slight degree of modulation.
- The PDM spectrum lines depend strongly on changes in the modulating signal $s_M(t)$.
- PDM has a non linear spectrum.

4.1.3 PDM modulator characteristic

The modulator characteristic graphically represents the relationship between the signal-dependent pulse duration T_0 and the modulating signal U_{DC}.

- Set the function generator to: DC, -10.0 V.
- Use the DC-voltage of the function generator as input signal of the S&H stage of the PTM modulator.
- Start the measurement by pressing *F9*.
- Determine the pulse duration T_0 of the carrier on the monitor.
- Enhance the DC voltage in steps of 1V and repeat each time the measurement of the pulse duration $T_0 = T_0(U_{DC})$.
- Note all pulse durations in the table.
- Reset the DC-voltage to 0 V by the end of the measurement.
- Draw the characteristic *pulse duration versus the control voltage* of the function generator.
- Determine the time deviation Δτ by measuring the minimum and maximum pulse duration:

$$\Delta\tau = \frac{T_{0\max} - T_{0\min}}{2}$$

- Determine the modulation index m_{PDM}: $m_{PDM} = \frac{\Delta\tau}{T_P} = \Delta\tau \cdot f_P$

Results

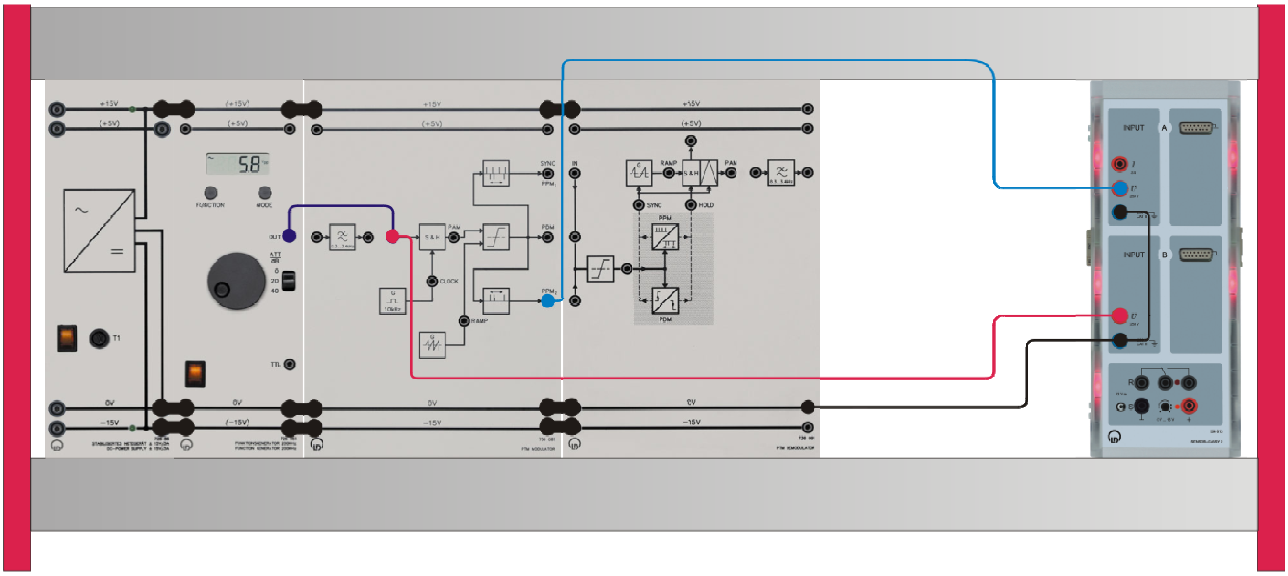
U_1/V	T_0/µs	U_1/V	T_0/µs
-10		0	
-9		1	
-8		2	
-7		3	
-6		4	
-5		5	
-4		6	
-3		7	
-2		8	
-1		9	
		10	

Time deviation Δτ: $\Delta\tau = \frac{T_{0\max} - T_{0\min}}{2} =$

Modulation index m_{PDM}: $m_{PDM} = \frac{\Delta\tau}{T_P} = \Delta\tau \cdot f_P =$

4.2 Investigating PPM

Experiment setup



4.2.1 PPM – time domain

- Set up the experiment as specified above.
- Set the function generator to: sine, 2.000 kHz, 20.0 Vpp.
- Start the measurement by pressing *F9*.
- Display the modulating signal $s_M(t)$ (S&H input) and the PPM2 signal on the monitor.
- Interpret your results.
- Repeat the measurement at the PPM1 output.
- Hint: Repeat the measurements, until you get a clear change in the pulse position.

Results

PPM dynamic characteristics
Red: modulating signal $s_M(t)$, f_M = 2000 Hz
Black: PPM2 signal

Interpretation

The PPM-pulses are shifted further apart with rising modulating signal $s_M(t)$. The signals (arbitrarily called *PPM2* here) are positive with TTL-level. In the literature they are commonly referred to as just PPM-signals.

PPM dynamic characteristics
Red: modulating signal $s_M(t)$
Black: PPM1 signal

Note: Measurements have to be repeated several times, until a significant change in the pulse positions occurs.

Interpretation

- When the signal amplitude AM is increased, a stronger time-shift in the PPM-pulse and vice versa is achieved.
- The positive pulses appear synchronously with the rising edges of the PDM-signal. They serve in the synchronization of the PPM-demodulator.
- The information of the modulating signal $s_M(t)$ is contained in the negative pulses at the trailing edges of the PDM-signal.
- The signal consisting of the positive synchronous signal and the negative, information carrying pulse is arbitrarily designated *PPM1* on the training panel.
- The PPM1 signal resembles a quasi-ternary signal in RZ format.
- Signals in RZ format carry wanted information in the positive and negative pulses.

4.2.2 PPM - spectrum

- Set up the experiment as specified above.
- Set the function generator to: sine, 2.000 kHz, 20.0 Vpp.
- Start the measurement by pressing *F9*.
- Interpret the result.

Results

PPM2 spectrum

Interpretation

- Carrier lines are smaller, less energy.
- No synchronization lines.

Results

PPM1 spectrum

Note

The specified spectrum amplitudes serve as orientation. Due to the complicated relationships between the spectral lines and the Bessel functions, even small deviations in the modulation indices lead to considerable changes in the overall spectrum. For that reason it is more important to take the measured frequencies than the absolute amplitude values.

- Spectral lines of the synchronization pulses appear at 20/30/40/50 kHz. They show almost constant amplitudes because of their small duty cycle.
- Check whether you can detect spectral components in integral multiples of the carrier (here multiples of f_p = 10 kHz) as well as lines around each carrier at an interval of whole numbered multiples of the signal frequency f_M .
- The spectrum of the PPM resembles that of the PDM signal. In PDM the high modulation index m causes very stretched partial spectra to form around each carrier line.
- PPM is better suited for an experiment-based investigation of the spectra than PDM.
- In PDM great differences between the carrier amplitudes and the sidelines are produced (at least for low modulation indices m_{PDM}).
- Due to the constant pulse duration, the PPM spectrum has both the sidelines at nf_M as well as the carrier with about equal amplitudes.
- Demodulation of the PPM signal with LP is only possible as long as there are no interfering spectral components in the band pass of the LP filter, in other words, until other partial spectra distort the baseband spectrum.
- Normally, demodulation is performed after the signal is reconverted into a PAM signal using a sample and hold circuit.

4.2.3 PPM modulator characteristic

The modulator characteristic graphically represents the relationship between the signal-dependent pulse position Δt and the modulating signal U_{DC}.

- In the experiment set-up above, connect Channel UA1 (of the Sensor CASSY 2) → PPM1 (PTM modulator)
- Set the function generator to: DC, -10.0 V.
- Use the DC-voltage of the function generator as input signal of the S&H stage.
- Start the measurement by pressing *F9*.
- Determine the pulse position Δt with respect to the positive going SYNC pulses of the PPM1 signal.
- Enhance the DC voltage in steps of 1 V and repeat each time the measurement of the pulse position Δt = Δt (U_{DC}).
- Note all pulse durations in the table.
- Reset the DC-voltage to 0 V by the end of the measurement.
- Draw the characteristic pulse position versus the control voltage of the function generator.

- Determine the time deviation Δt by measuring the minimum and maximum pulse position:

$$\Delta t = \frac{T_1 - T_2}{2}$$

Results

U_1/V	T_0/μs	U_1/V	T_0/μs
-10		0	
-9		1	
-8		2	
-7		3	
-6		4	
-5		5	
-4		6	
-3		7	
-2		8	
-1		9	
		10	

Interpretation

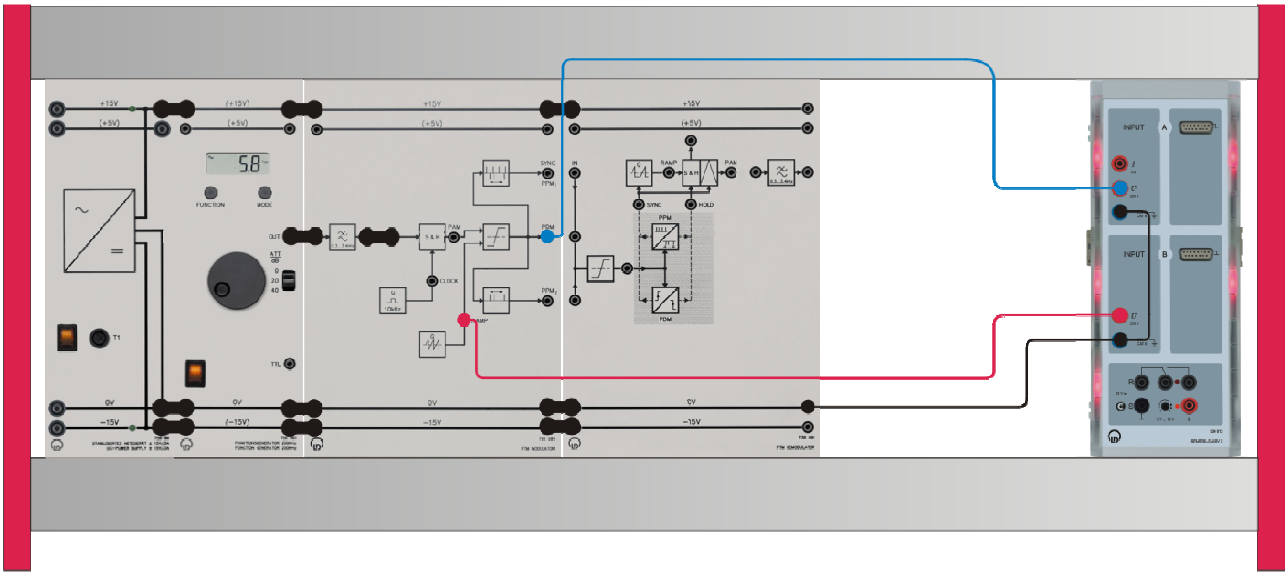
Phase deviation Δt:

Modulation index m_{PPM}:

Δt: measured between positive slope of the SYNC and negative slope of PPM

4.3 Generation of PTM with the ramp method

Experiment setup



- Set up the experiment as specified above.
- All measurements take place at the PTM modulator.
- Set the function generator to: sine, 2.000 kHz, 20.0 Vpp.
- Start the measurement by pressing *F9*.
- Display the signal at the PDM output of the comparator $s_{PDM}(t)$ and at the output of the ramp generator RAMP.
- For unchanged modulating signal $s_M(t)$ display the signals at the outputs PAM and PDM on the monitor.
- Repeat the experiment. Display the signals PAM and PPM1 on the monitor.
- Interpret your results.

Results

Display of PDM and ramp
Red: saw tooth signal
Black: PDM signal

Interpretation

- The PDM-signal is triggered with every second saw tooth.
- The gaps arising could be used to transmit the information of a second channel. (Capability of PDM for the time multiplexing method).
- The width of the PDM-pulses is proportional to the instantaneous value of the modulating signal $s_M(t)$.

Dynamic relationship between PAM and PDM

Red: PAM signal at the output of the S&H stage
Black: PDM signal

Interpretation

- As the PAM signal rises, the trailing edge of the PDM signal appears later.
- Trailing edge modulation

Display of PAM and PPM_1
Red: PAM signal at the output of the S&H stage
Black: PPM1 signal

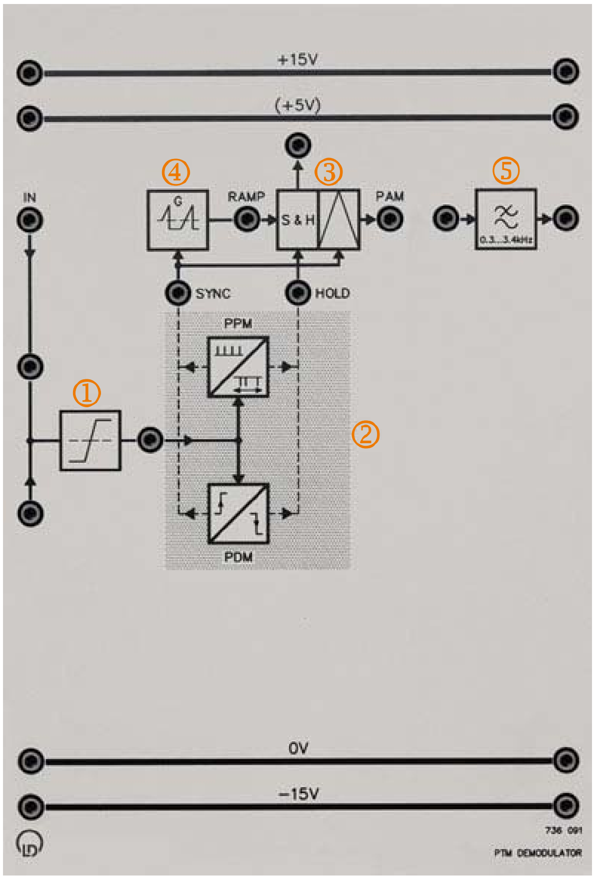
Interpretation

- The positive pulses in the PPM1-signal are only needed for the synchronization of the PTM- demodulator.
- The information of the modulating signal $s_M(t)$ is contained in the time position of the negative pulse with respect to the positive synchronous pulse.

PTM demodulation

As demodulation methods there are the low pass demodulation for PDM and the demodulation according to the ramp method for both, PDM and PPM. Demodulation according to the ramp method is a reversal of the corresponding modulation principle.

Instruments



PTM-Demodulator (Cat. no. 736091)

The PTM demodulator consists of:

(1) Input pulse former
(2) Isolation network for synchronization and wanted data

	Data	Synchronization
PDM	trailing edge	leading edge
PPM 1	negative pulse	positive pulse

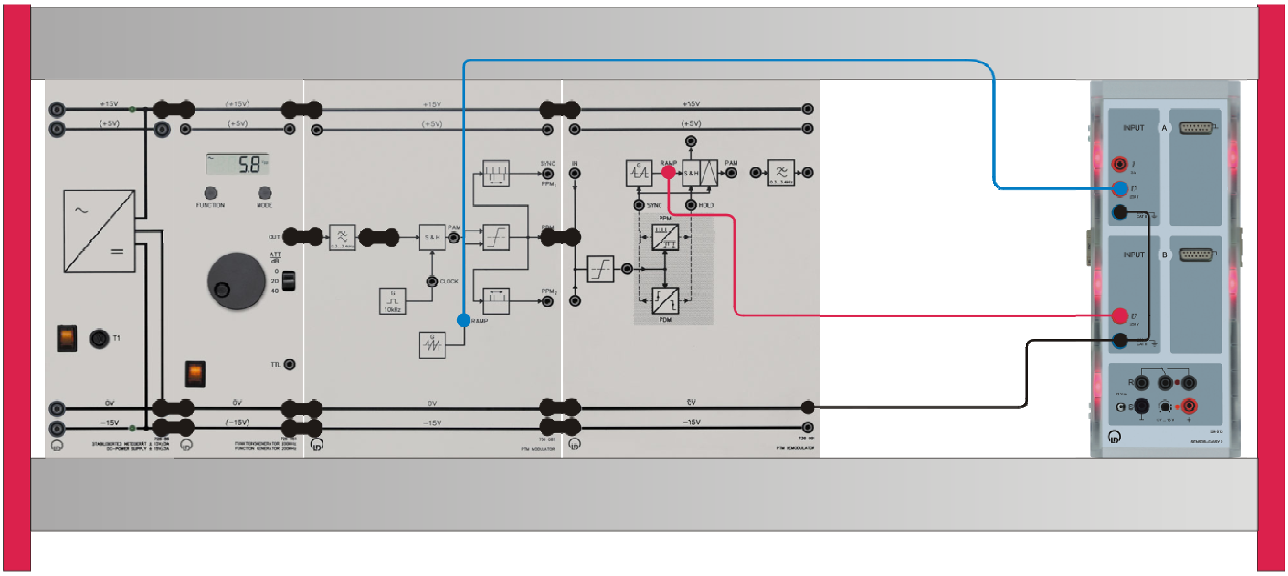
(3) Sample and hold element
(4) Ramp generator
(5) Demodulator filter 0.3... 3.4 kHz, V = 1

The PTM demodulator converts the pulse time modulation signals (PDM or PPM) back into PAM. Finally, using low pass demodulation the original modulating signal is recovered out of the PAM signal.

Demodulation of PTM

Demodulation according to the ramp method

Experiment setup



Ramp recovery

- Use the experiment set-up: Set the function generator to: sine, 2.000 kHz, 20.0 Vpp .
- Start the measurement by pressing *F9*.
- Display the signals at the RAMP outputs of the PTM modulator and demodulator.
- Subsequently connect the PPM1 and PPM2 output of the PTM modulator to the input of the PTM demodulator.
- Repeat the measurement.
- Interpret your results.

Results

Ramp recovery for PDM / PPM1
Black: Saw tooth signal of the PTM-modulator
Red: Recovered saw tooth of tile PTM-demodulator

Interpretation

- The ramp of the modulator covers the ±10 V modulating range.
- The recovered ramp of the demodulator (±5 V) is synchronous with the saw tooth of the modulator.
- Only every second ramp triggers the demodulator. Each rising edge of the PDM signal starts the ramp generator of the PTM demodulator.
- With PDM / PPM1, the ramp generator of the PTM demodulator runs synchronously with respect to the saw tooth generator of the PTM modulator.

Ramp recovery for PPM2

Notes for PPM2

- With PPM2, the ramp generator of the PTM demodulator runs arbitrarily with respect to the ramp generator of the PTM modulator.

Synchronization recovery in PDM mode

- On basis of the first experiment set-up carry out the following experiment.
- Connect the PDM output of the PTM modulator to the input of the PTM demodulator.
- Set the function generator to: sine, 2.000 kHz, 20.0 Vpp.
- Start the measurement by pressing *F9*.
- PTM-modulator: Display the PDM signal at the output of the comparator $s_{PDM}(t)$ on channel UA1.
- PTM demodulator: Display the sync signal at the output of the ramp generator on channel UB1.
- Interpret your results.

Results

Black: PDM-signal at the PTM-modulator
Red: SYNC-signal at the PTM-demodulator

Interpretation

- The PTM-demodulator has to be synchronized with the modulator.
- This synchronization signal can be transmitted by its own channel. But that is not a very elegant solution which also entails added costs. Normally, the synchronization transmitted jointly with the PDM-signal.
- In the training system, the synchronization is contained in the rising edge of the PDM-signal.
- The joint transmission of synchronization and the data signals permits simple further
- processing, e.g. transmission via microwave links or optical fibers due to the fact that the synchronous- and information signals are subjected to the exact same time lags.

Synchronization recovery in PPM1 mode

- On basis of the first experiment set-up carry out the following experiment.
- Connect the SYNC / PPM1 output of the PTM modulator to the input of the PTM demodulator.
- Set the function generator to: sine, 2.000 kHz, 20.0 Vpp.
- Start the measurement by pressing *F9*.
- PTM-modulator: Display the PPM1-signal at the output SYNC / PPM1 on channel UA1.
- PTM demodulator: Display the sync signal at the input of the ramp generator on channel UB1.
- Interpret your results.

Results

Black: PPM1-signal at the PTM-modulator
Red: SYNC-signal at the PTM-demodulator

Interpretation

- The synchronization signal is contained in the positive pulses. These pulses are isolated from the actual data (negative pulses) in the PTM-demodulator.
- The SYNC pulses trigger the ramp generator of the PTM demodulator.

Synchronization recovery in PPM2 mode

- On basis of the first experiment set-up carry out the following experiment.
- Connect the PPM2 output of the PTM modulator to the input of the PTM demodulator.
- Set the function generator to: sine, 2.000 kHz, 20.0 Vpp.
- Start the measurement by pressing *F9*.
- PTM-modulator: Display the PPM2-signal at the output PPM2 on channel UA1.
- PTM demodulator: Display the sync signal at the input of the ramp generator on channel UB1.
- Interpret your results.

Results

Black: PPM_2-signal at the PTM-modulator
Red: SYNC-signal at the PTM demodulator

Interpretation

- The PPM2-signal is a pure PPM-signal (positive pulse). It does not contain any synchronization information for the PTM-demodulator implemented here.
- The SYNC-signals are erroneously. They cannot synchronize the saw tooth generator of the PTM demodulator.

HOLD signals in PDM mode

- On basis of the first experiment set-up carry out the following experiment.
- Connect the PDM output of the PTM modulator to the input of the PTM demodulator.
- Set the function generator to: sine, 2.000 kHz, 20.0 Vpp.
- Start the measurement by pressing *F9*.
- Display the ramp at the PTM demodulator on channel UA1.
- Display the HOLD signal at the PTM demodulator on channel UB1.

- Interpret your results.

Results

The interaction of HOLD and RAMP in the recovery of the PAM-signal (PDM mode)

Black: Recovered ramp signal at PTM-demodulator
Red: HOLD pulse

- Display the HOLD signal at the PTM demodulator on channel UB1.
  Display the PDM signal at the PTM modulator on channel UA1.
- Interpret your results.

Results

The interaction of PDM HOLD and HOLD in the recovery of the PAM-signal (PDM mode)

Interpretation

- The HOLD pulses are triggered by the trailing edge of the PDM signal or the negative pulses of the PPM-signal.
- They terminate the charging of the storage capacitor in the S&H-stage.

HOLD signals in PPM1 mode

- On basis of the first experiment set-up carry out the following experiment.
- Connect the SYNC / PPM1 output of the PTM modulator to the input of the PTM demodulator.
- Set the function generator to: sine, 2.000 kHz, 20.0 Vpp.
- Start the measurement by pressing *F9*.
- Display the ramp at the PTM demodulator on channel UA1.
- Display the HOLD signal at the PTM demodulator on channel UB1.
- Interpret your results.

Results

Interaction of HOLD and RAMP in the recovery of the PAM-signal. (PPM1_Ramp_Hold.labx)

Black: Recovered ramp signal at PTM-demodulator
Red: HOLD pulse

- Display the HOLD signal at the PTM demodulator on channel UB1.
  Display the PPM1 signal at the PTM modulator on channel UA1.
- Interpret your results.

Results

The interaction of PPM1 and HOLD in the recovery of the PAM-signal (PPM1 mode)

Interpretation

- Same behavior as in PDM mode.

S&H signals and recovered PAM

- On basis of the first experiment set-up carry out the following experiment.
- Connect the PDM or PPM1 output of the PTM modulator to the input of the PTM demodulator.
- Set the function generator to: sine, 500.0 Hz, 20.0 Vpp.
- Start the measurement by pressing *F9*.
- Display the output signal of the S&H-stage of the PTM demodulator on channel UA1.
- Display the PAM signal at the PTM demodulator on channel UB1.

Results

Black: Output signal of the S&H·stage at the PTM·demodulator
Red: Recovered PAM-signal on the PTM-demodulator

- The SYNC-signal triggers the saw tooth generator in the PTM-demodulator.
- It charges the storage capacitor of the S&H-stage until the HOLD-signal arrives.
- The recovered saw tooth signal does not have to have the same amplitude as the original saw tooth signal of the modulator.

- Interpret your results.

Results

Black: S&H-signal
Red: PAM-signal

Interpretation

- The PAM-signal is generated through blanking during the hold-phase.

PAM signals in the PTM modulator and demodulator

- On basis of the first experiment set-up carry out the following experiment.
- Connect the PDM output of the PTM modulator to the input of the PTM demodulator.
- Set the function generator to: sine, 500.0 Hz, 20.0 Vpp.
- Display the PAM signal at the PTM modulator on channel UA1.
- Display the PAM signal at the PTM demodulator on channel UB1.
- Interpret your results.

Results

Black: PAM-signal at modulator
Red: PAM-signal at demodulator

PDM demodulation with the ramp method

- On basis of the first experiment set-upcarry out the following experiment.
- Connect the PDM output of the PTM modulator to the input of the PTM demodulator.
- Connect the output PAM with the input of the low pass filter of the PTM demodulator with a bridging plug.
- Set the function generator to: sine, 500.0 Hz, 20.0 Vpp.
- Display the output signal of the function generator on channel UA1.
- Display the signal at the output of the low pass filter of the PTM demodulator on channel UB1.
- Interpret your results.

Results

Black: modulating signal from the function generator
Red: demodulated signal

- The duty cycle of the PAM signal only amounts to 50% at the input of the LP filter.
- Thus the amplitude A_D of the demodulated signal only amounts to half that of the modulating signal $s_M(t)$.
- Phase shift between the modulating and demodulated signal.
- Frequency of the modulating and demodulating signal is equal.

LP demodulation of PTM signals

- On basis of the first experiment set-up carry out the following experiment.
- Connect the PDM output of the PTM modulator directly to the input of the LP filter of the PTM demodulator via cable.
- Set the function generator to: sine, 500.0 Hz, 20.0 Vpp.
- Display the output signal of the function generator on channel UA1.
- Display the signal at the output of the low pass filter of the PTM demodulator on channel UB1.
- Interpret your results.

Results

Black: modulating signal from the function generator $s_M(t)$
Red: demodulated signal $s_D(t)$

- Demodulated signal much smaller due to small duty cycle.
- LP demodulation of PDM.
- The demodulation performed by the LP out of the signal recovered from the PDM signal $s_D(t)$ demonstrates high frequency distortion components.
- The amplitude is considerably lower than is the case in demodulation according to the ramp method.

- Repeat the experiment for the LP demodulation of PPM2.
- Interpret your results.

Results

Black: modulating signal from the function generator $s_M(t)$
Red: demodulated signal $s_D(t)$

- For PPM2 only low pass demodulation is possible.
- Due to the small duty cycle, the amplitude of the demodulated signal is very small (0,02 V)!

Introduction:-

Using the course

Welcome to the COM3LAB course.Before you begin with the course, you should spend some time on the next pages with the COM3LAB System.

More detailed information can be obtained by clicking in the individual areas in the adjacent image.

Theory Filed

This field has explanations on the theory behind the individual topic areas. You can also listen to this text.

This field contains images, animations, videos, circuit diagrams and interactions. The message field tells you what excise you will be doing.

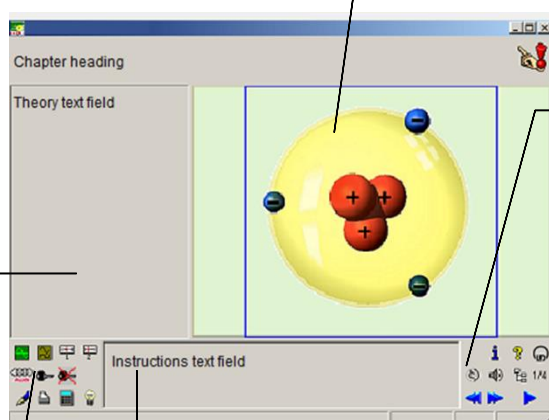


Fig1

Menu

This is the operating field. Here are the explanations for the most important function:

This is the most often-used button, used to take you step-by-step through the course. Clicking on this button also checks the exercise you have carried out. If the exercise is not correctly completed, you will not be advanced.

This button takes you page-by-page through the course .Be careful however, as this will skip the individual steps.

In the Help file you will find detailed explanations for further operation of the system.

Tool Box

This box contains all the virtual instruments you will need during this course.

The instruments are explained in more detail when you use them for the first time.

Message Field

This field contains the instructions for your activities (e.g. how to set up an experiment, how to approach a task, etc..)

Course structure

The COM3LAB course “Modem Technology” is divided into two main topics:

Modulation procedures (here those with a harmonic carrier and digital modulating signals as they are typically used in modems) on one hand, and characteristic values and operating modes of modems on the other. The adjacent graphic shows a detailed overview of the topics covered in the individual chapters.

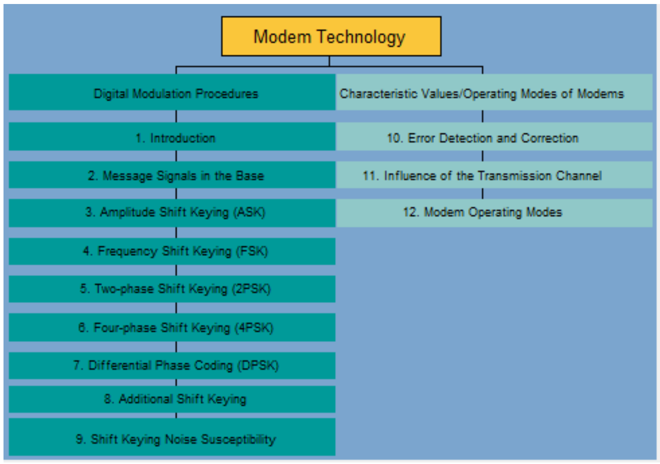


Fig2

1) Introduction: chapter 1 provides a brief overview of the principle of modulation and introduces the COM3LAB board 700 74 as well as the associated Control Panel.
2) Message Signals in the Base: in chapter 2 first the message signals in the base (I > e > without modulation) are considered. Various coding types (NRZ, differential and Manchester format) are introduced.
3) Amplitude Shift Keying (ASK): Chapter 3 deals with the principle and properties of Amplitude Shift Keying (ASK) in the time and frequency area as well as corresponding demodulation procedures.
4) Frequency Shift Keying (FSK): The emphasis of Chapter 4 is on the principle and properties of frequency shift keying (FSK) in the time and frequency area as well as corresponding demodulation procedures.
5) Two- phase Shift Keying (2PSK): chapter 5 covers the principle and properties of two-phase shift keying (2PSK) in the time and frequency area as well as corresponding demodulation procedures including the problem of carrier recovery.
6) Four-Phase Shift Keying (4PSK): chapter 6 covers the principle and properties of four-phase shift keying (4PSK) in the time and frequency area as well as corresponding demodulation procedures including the problem of carrier recovery .
7) Differential Phase Coding (DPSK): Chapter 7 is devoted to differential phase coding and focuses especially on the advantages as compared with conventional phase shift keying.
8) Additional Shift Keying: Chapter 8 covers additional, especially bandwidth-saving shift keying such as MSK and GMSK, which are of particular interest for the mobile arena.
9) Shift Keying Noise Susceptibility: Chapter 9 compares the noise susceptibility of the previously discussed shift keying methods with each other.
10) Error Detection and correction: chapter 10 focuses on the topics of error detection and correction using parity bits. Highlighted are the concepts of Hamming distance, redundancy and payload data rate.
11) Influence of the transmission channel: chapter 11 discusses the influence of the transmission channel in message transmission. This is investigated in a series of experiments with various transmission media (line types).
12) Modem operating Modes: The emphasis of chapter 12 is on the technical construction of modems and the various operating modes (simplex, half-duplex, full-duplex).Other topics include the commands for modem control (AT commands) and modulation standards.

History

In data communication –whose beginnings go back to Morse telegraphy – digital data are transmitted over direct cable connections, a telephone line or via radio frequency. At the end of the transmission channel are tow communication terminal devices, such as computers. To be transmitted, the data must be adapted to the communication channel by means of modulation.

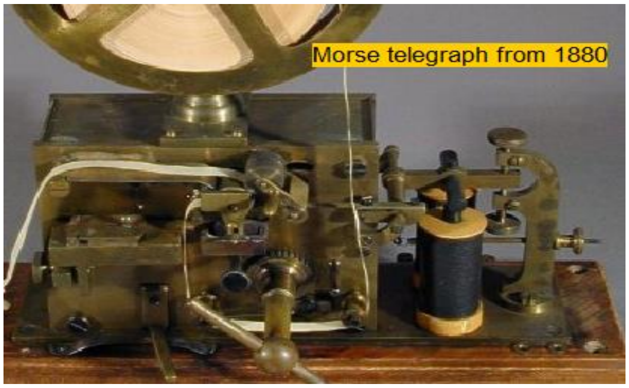


Fig3

Principle of Modulation

The principle of Modulation is used to adapt to the transmission channel and to use the channel simultaneously for a variety of signals. The payload signal is thereby modulated in the modulator on to a (for example sinusoidal or pulse-shaped) high-frequency carrier and then transmitted. On the receiver side a demodulator then regenerates the original payload signal from the modulated signal. There are various types of modulation. (Note: Output D on the Modem Technology Board is located between units (3) and (4), and Output D is between units (5) and (6)).

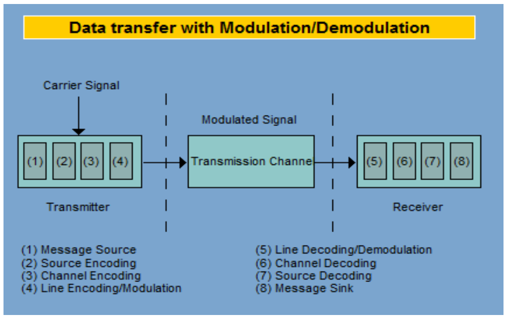


Fig4

From the picture above:-

1) The massage source is where the electrical signal to be transmitted is generated.
2) The source encoding ensure that the massage is represented in a form suitable for the transmission (for example by means of digitizing with the help of an A/D converter); usually with a reduction in the transmission effort (for example by means of entropy encoding using Huffman code) or by omitting

signal components irrelevant to the receiver by mans of band restriction in telephony or television technology.

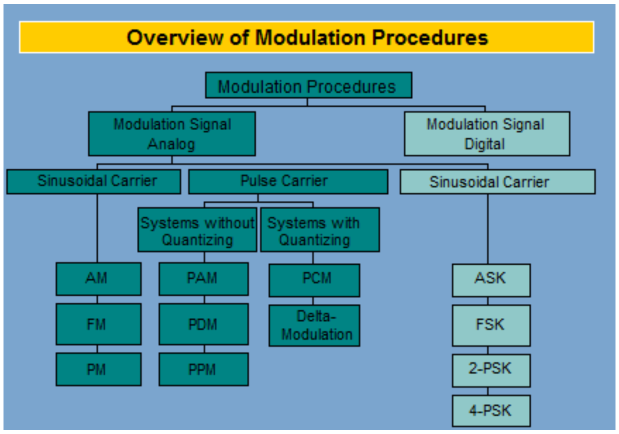
3) Channel encoding protects the message from transmission errors (for example using the additional check characters of the Hamming code).
4) Line encoding / modulation adapt the signal to the physical channel (for example by means of pulse shaping or carrier modulation).

Transmission Channel: the signal is generally distorted in the transmission channel (for example by physical filter or thermal noise as an additive noise signal).

5) Line decoding/demodulation provide demodulation of the transmitted signal or message using special procedures for suppressing the noise (such as signal-adapted filters or echo equalization).
6) Channel coding represents the inverse of channel encoding and is used for recognizing and / or correcting certain error patterns .If necessary in the case of non-correctable errors, a repetition of the message can be requested.
7) Source decoding and provides the appropriate signal form for the following message sink.
8) As the last link in the transmission chain, the message sink prepares the signal provided by the source decoding.

Modulation Procedures

Modulation procedures are characterized as those having a sinusoidal carrier signal and those with a pulse-shaped carrier signal. On the other hand, the modulation signal representing the massage can be analog or digital. This course deals with those modulation signal is modulated on to a sinusoidal carrier (Shown lighter in the adjacent overview). The basics of the various modulation procedures are provided by the corresponding instructional systems.



- Which modulation procedure is used for traditional radio?
  - _ Sinusoidal carrier / Analog modulation signal.
  - _ Pulse carrier / Analog modulation signal.
  - _ Pulse carrier / Digital modulation signal.
  - _ Pulse carrier / Digital modulation signal.

The COM3LAB-Board 700 74

Modem Technology contains all the components needed for an introduction to digital modulation.

This includes especially modulators for amplitude, frequency and two- and four-phase shift keying as well as the corresponding demodulators.

Data transfer can be performed over various channels which can be noise induced using an integrated noise generator.

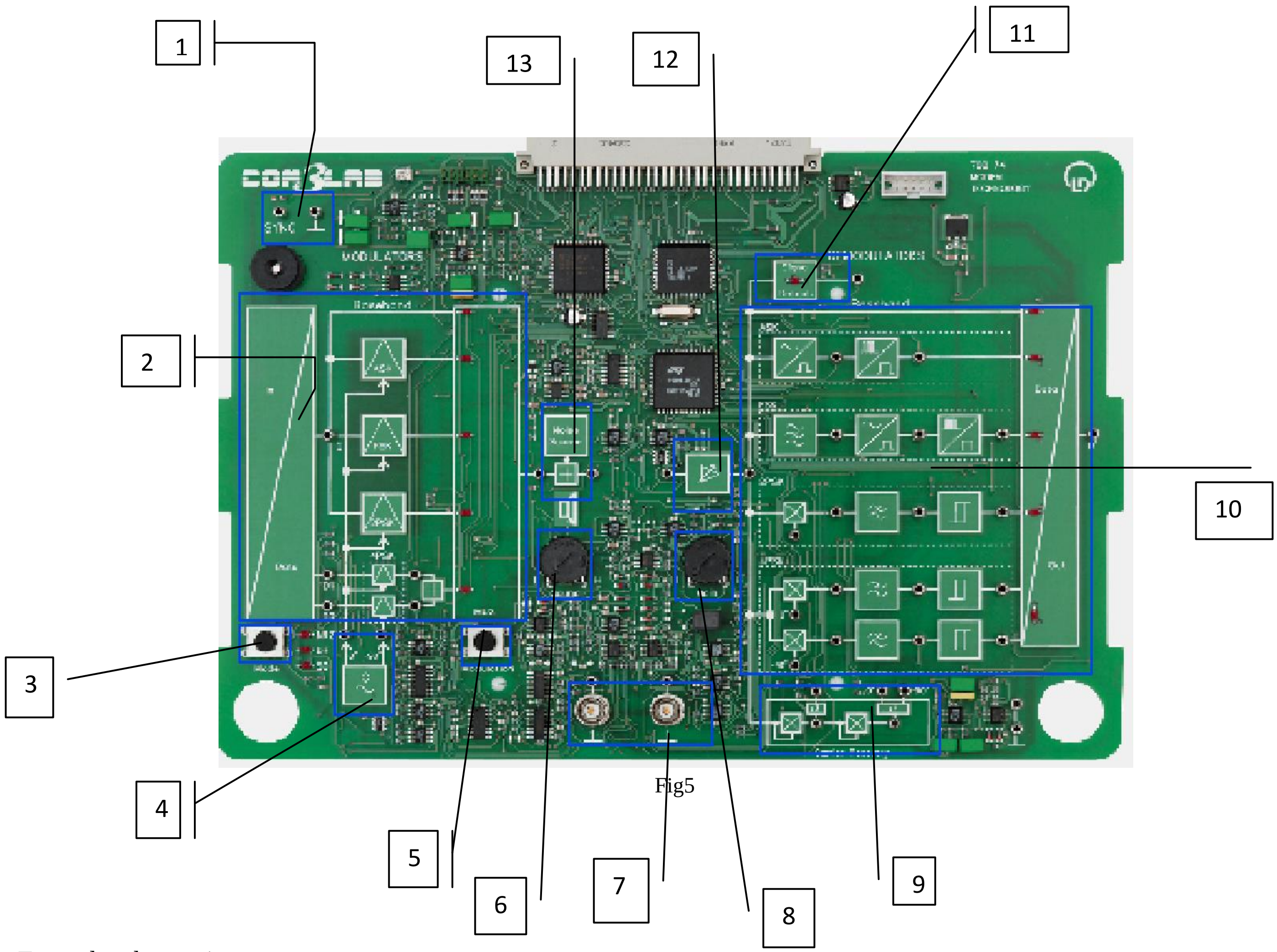


Fig5

From the above picture:-

1) Adjustment for amplifying the modulated signal after transmission.
   Synchronization signal (can for example be used for triggering the oscilloscope).
2) Modulators: Base transmission, Amplitude shift keying, Frequency shift keying, 2-Phase shift keying, 4-Phase shift keying.
3) Button for changing data signal encoding.
4) Carrier signal with null phase or 90 degree phase shifted.
5) Button for selecting the modulator over the multiplexer.
6) Adjustment for loudness of the modulated signal.
7) BNC sockets for connecting the modulator and demodulator using coax cable..
8) Adjustment for amplifying the modulated signal after transmission.
   Should be in the full left position unless otherwise required.
9) Circuit for carrier recovery.

10) Demodulators: Base transmission, Amplitude shift keying, Frequency shift keying, 2-Phase shift keying, 4-Phase shift keying.
11) Unit for signal detection .A red LED indicates whether a signal is recognized.
12) Input amplifier for receiver side.
13) Noise generator with adjustable amplitude.

The Modem control Panel

Technology course allows you to control all the important board function such as the data encoding type.

Modulation / demodulation type as well as the amplitude of the noise source .It also allows you to manually enter the characters for transmitting and enable automatic periodic sending of a settable data byte. The control panel is opened by clicking on the button.

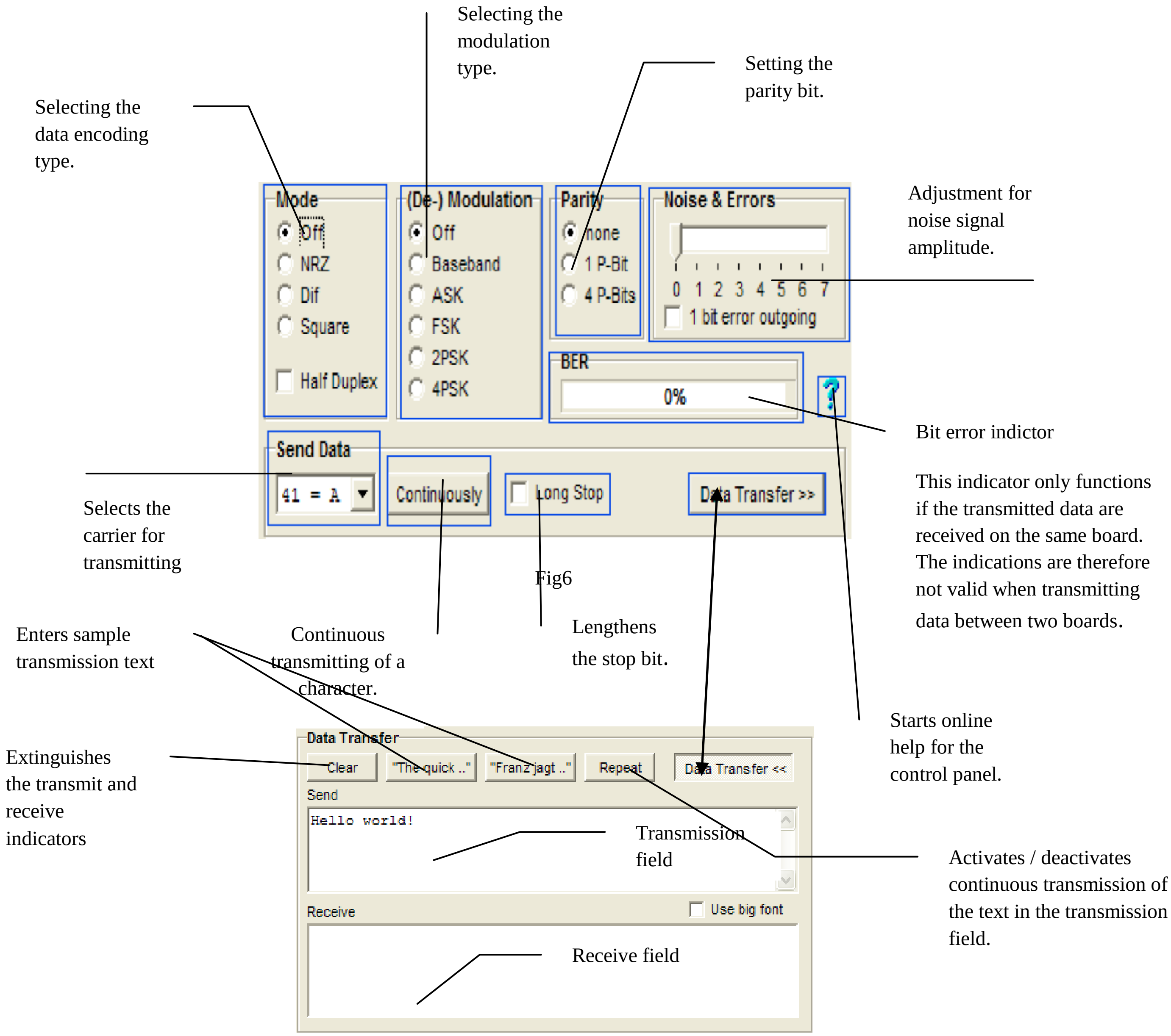


Fig6

Message signals in the base band

NRZ Format:-

A general distinction can be made between synchronous and asynchronous transmission; only the latter will be considered in context of this course.

In NRZ (nonreturn to zero) format the signal amplitude is constant during the entire duration T of a data bit. Each data byte is introduction by a start bit (logical 0) and ends with a stop bit (logical 1). A data bit of 0 is represented by a LOW level, and a data bit of 1 by a HIGH level.

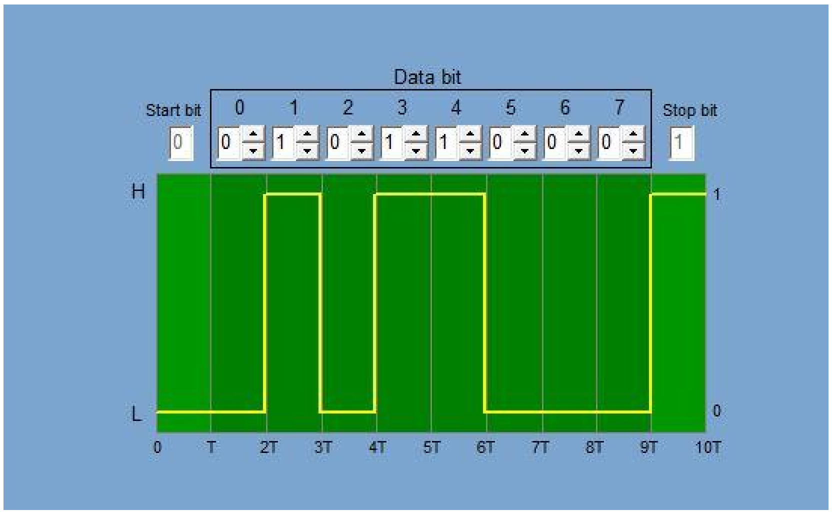


Fig7

Experiment: Time curve of the NRZ signal

In the following experiment you will study the NRZ data format in greater detail .Periodically different data bytes will be transmitted and the curve of the encoded data signal recorded with the digital analyzer.

From the recorded curves the bit rate of the data signal will then be determined.

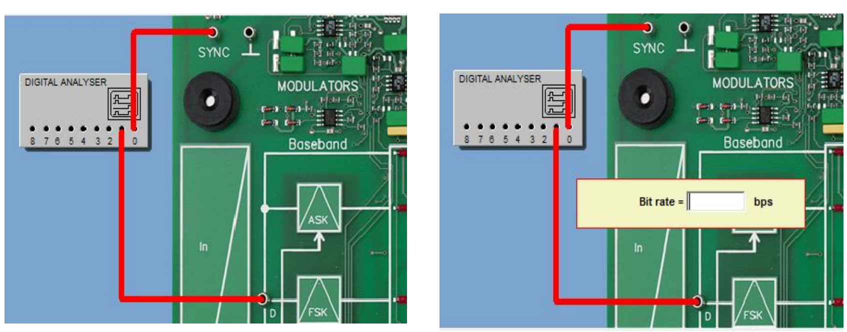


Fig8

Note :

The curves blow show the expected structure of the signal consisting of the start bit, data bits (between the blue and red cursor) and the stop bit (logical 1, to the right of the red cursor). After the start bit follows first the least significant bit (LSB), then the higher value bits.

So , Bit Rate =

Differential Format

In differential format the signal amplitude is likewise constant during the entire duration T of a data a start bit. Each data byte is introduced by a start bit and ended by a stop bit.

The signal level is however dependent on the preceding signal level: If a bit (start, stop or data bit) is at logical 1 , the signal level does not change ; if it is at logical 0 , it is inverted . As will be shown at a later time, differential format is especially suited for use with 2PSK/4PSK (DPSK).

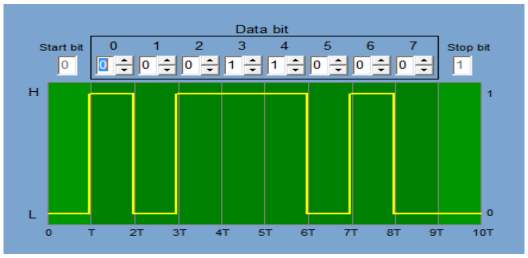


Fig9

Experiment: Time course of the differential data signal

In the following experiment you will study the differential data format more closely. Periodically differing data bytes will be transmitted and the course of the encoded data signal will be recorded with the digital analyzer. The obtained signals will be compared with those of the NRZ encoding.

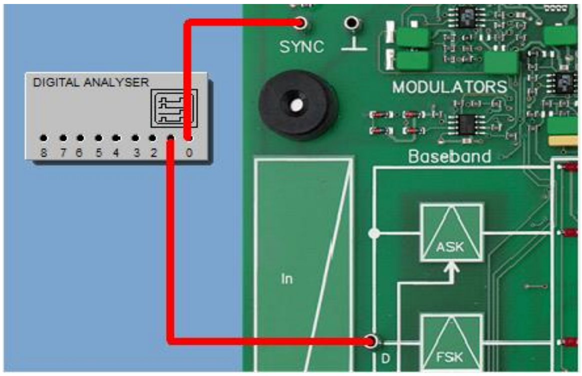


Fig10

- For which data bytes is there a periodic square wave signal?
  _ None
  _All
  _00 and 55
  _55 and FF

Knowledge Check

In the fig.11 you will see a summary of the material covered in this chapter.

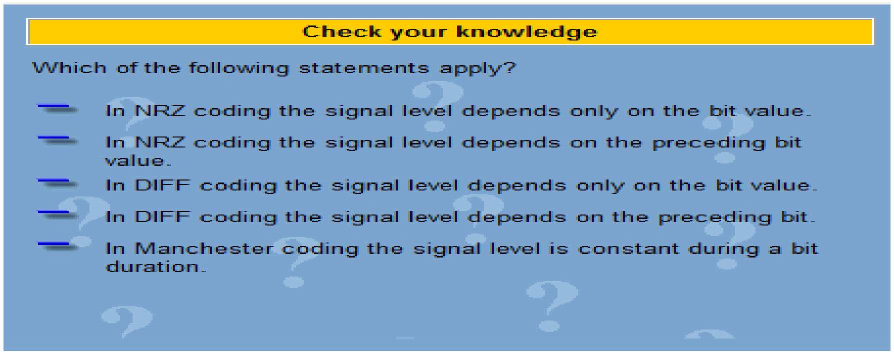


Fig11

Summary

In the fig.12 you will see a summary of the material covered in this chapter.

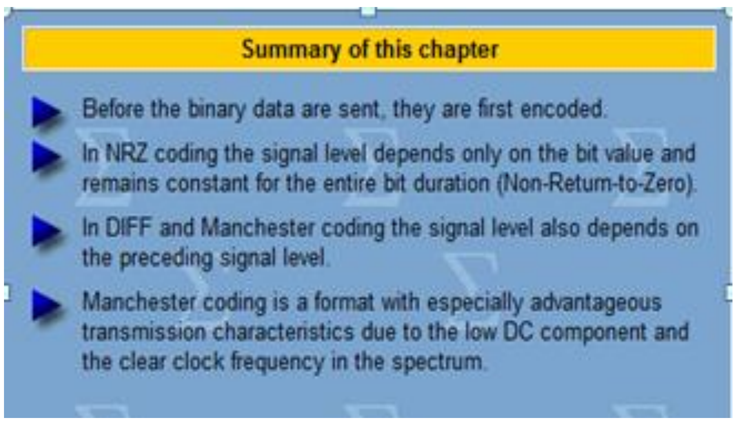


Fig12

Amplitude Shift Keying (ASK)

Time function

In Amplitude Shift Keying (Amplitude Shift Keying ASK) a sinusoidal carrier signal having frequency f_0 is turned on and off by the data signal (which is why this procedure is also known as ON/OFF keying). If the data signal has the value logical 1 (HIGH), the carrier signal is turned on, and a data signal of logical 0 (LOW) turns it off. ASK can therefore be implemented by means of a switch.

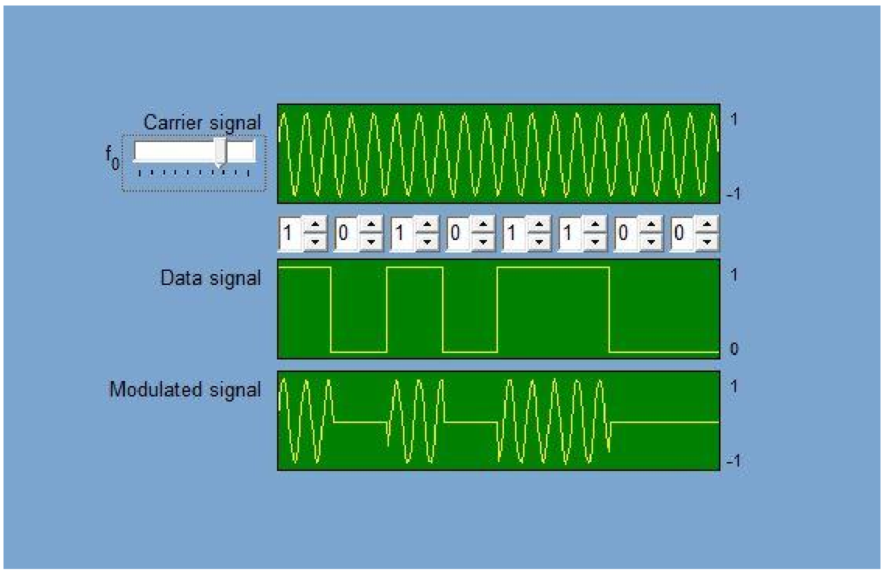


Fig13

Experiment: Time course for ASK

In the following experiment you will first study the modulated signal in amplitude shift keying. Periodically a constant data byte will be sent as a data signal modulated on to sinusoidal carrier signal. Both signals will be compared using the oscilloscope.

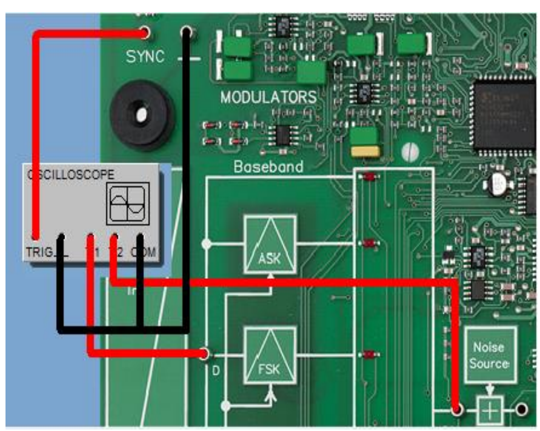


Fig 14

- How many oscillations of the carrier signal are there for one bit width?
  _ 2
  _ 8
  _ 10
  _20

ASK in the frequency range

In the frequency range amplitude shift keying causes a shift of the data signal spectrum by the frequency of F_0 the carrier signal. The graphic at the right shows this using the example of a periodic square-wave signal having frequency F_D. This signal has a line spectrum consisting of spectral line with odd multiples of the base frequency F_D, in other words for the frequencies F_D, 3 F_D, 5 F_D … The form of the spectrum itself is not changed by ASK.

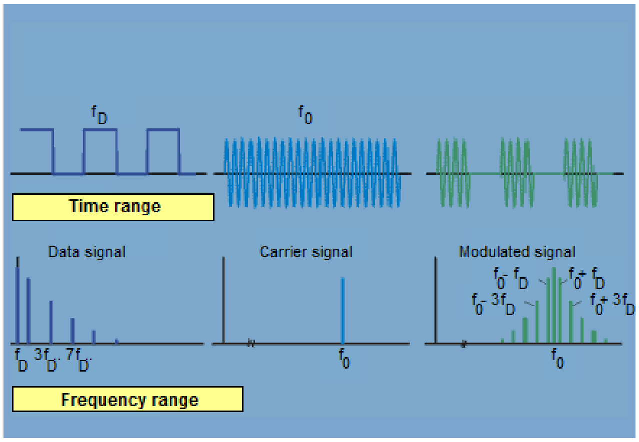


Fig15

Bandwidth requirements for ASK

For unambiguous detecting of the signal state on the receiver side, it is sufficient if the spectrum is transmitted up to the first pair of side-bands, in other words in the range F_{0-} F_D … F_{0+} F_D. The theoretical minimum required bandwidth is therefore

$$B = 2.F_D$$

In practice a value 1.4X is generally used.

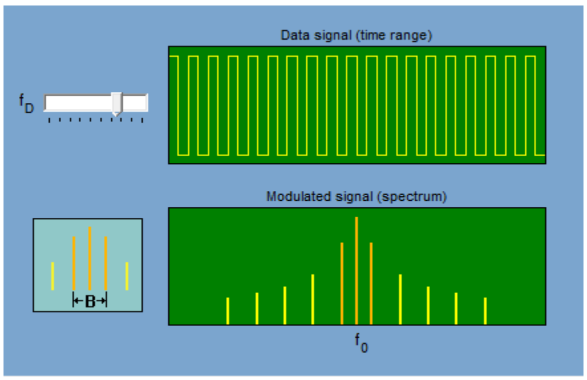


Fig16

Experiment: Measuring the ASK spectrum (SQ Format)

In the following experiment you will study the amplitude spectrum of the modulated signal in amplitude shift keying. A periodic square-wave signal having a DC component and generated by the SQ format will be used as the data signal. The COM3LAB Spectrum analyzer will be used to obtain the spectrum; open the analyzer by clicking on the button .

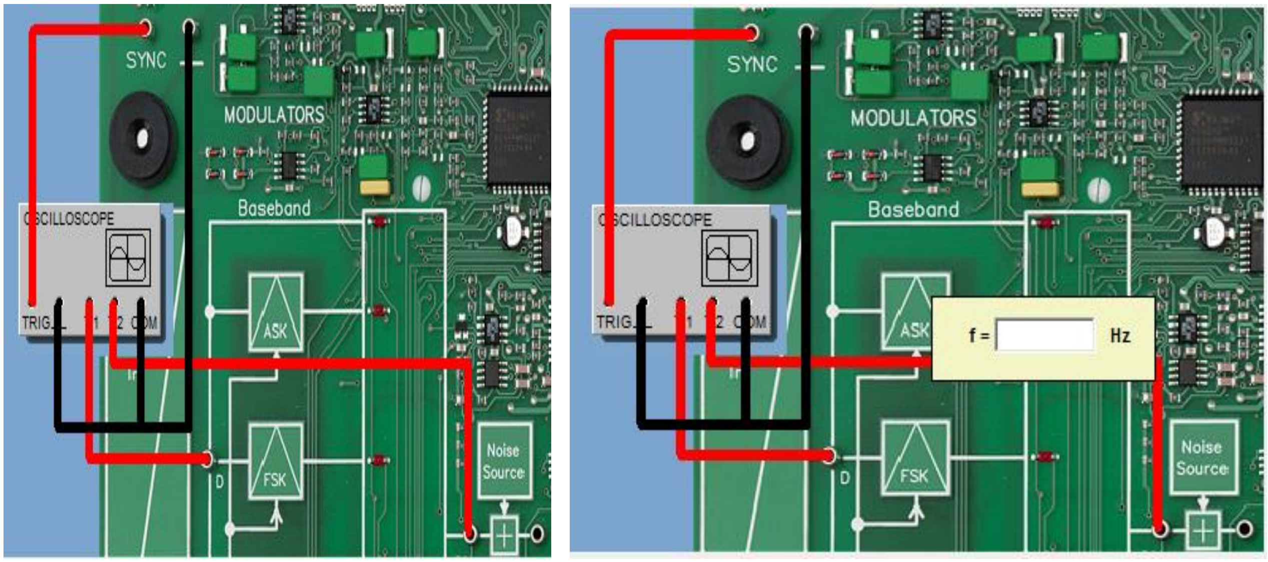


Fig17

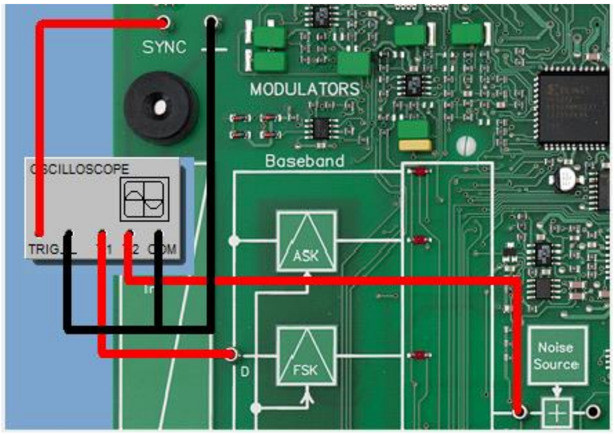
Note:

The spectrum of the data signal is shifted by the carrier frequency F_0 ; the individual spectral lines appear mirrored left and right of the carrier. At $f_o = 10$ KHz the original data DC component thus appears, whereas the side-bands occur at the frequencies

Find F_0 *and* F_D

Experiment: Measuring the ASK spectrum (NRZ/DIF-Format)

In the following experiment you will again study the amplitude spectrum of the modulated signal with amplitude shift keying, but now continuously different data in NRZ or DIFF format will be sent as the data signal. The COM3LAB Spectrum Analyzer will again be used to determine the spectrum.



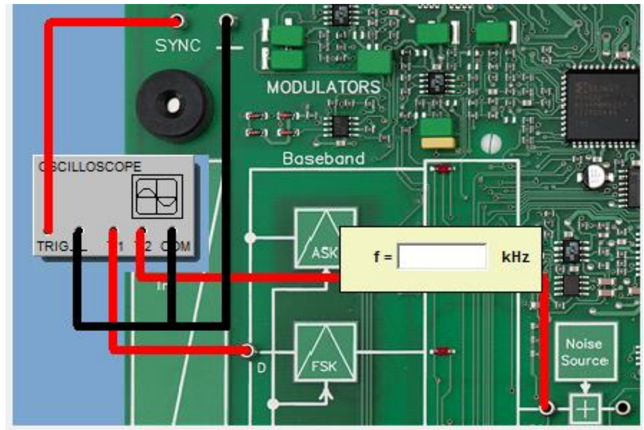


Fig18

Demodulation

For demodulation the ASK signal, envelope curve demodulation is used. In the first step acomparator is used to hide the negative half-waves of the sinus oscillations and their amplitude is restricted to TTL level. In a second step a re-triggerable monoflop generates a signal total pulse from the individual pulses belonging to a bit at logical 1; this total pulse then has the original length of the bit duration.

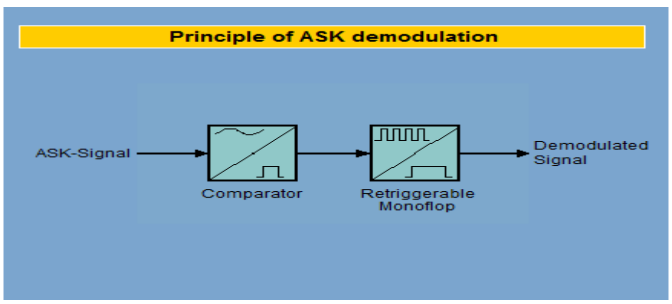


Fig 19

Experiment: Investigate the envelope curve demodulator with ASK

In the following experiment the demodulation of an ASK-modulated signal with NRZ formatting will be studied more closely. Periodically a data byte of 55 will be sent and the course of the signal behind the comparator and monoflop of the ASK demodulator recorded. Note: Be sure that the ‚Gain' adjustment is at full left for this and all following experiments.

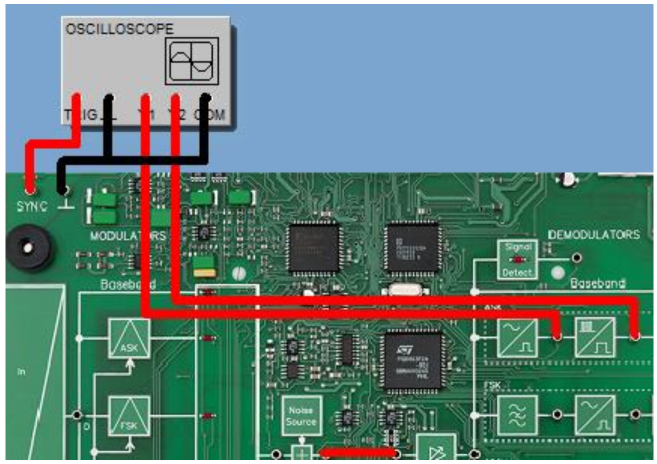


Fig 20

- How many pulses occur behind the comparator for a data bit of logical 1?
  _ 1
  _ 2
  _ 4
  _ 8

Summary

In the fig.21 side you will see a summary of the material covered in this chapter.

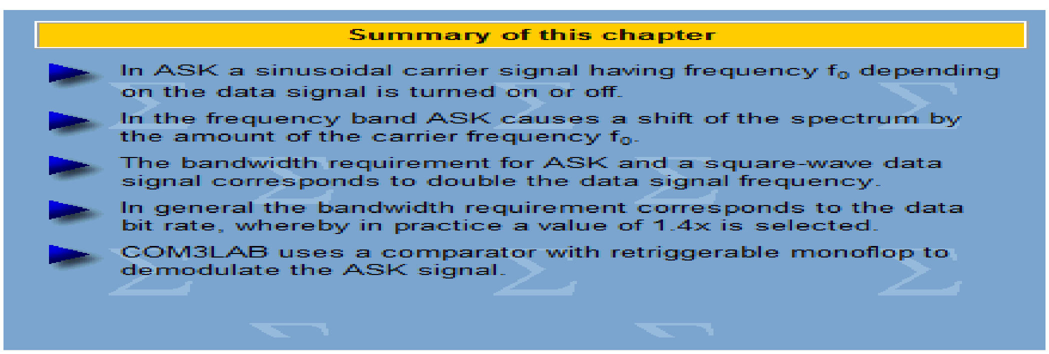


Fig 21

Frequency Shift Keying (FSK)

Time function

In frequency Shift keying (FSK) depending on the data signal the shift is made between two frequencies $f_{0\,-}\,\Delta f$ (data bit is logical 0) and $f_{0\,+}\,\Delta f$ (data bit is logical 1) The frequency f_0 is termed the center frequency, the frequency Δf the frequency deviation. Compared with ASK, FSK is significantly less noise-sensitive.

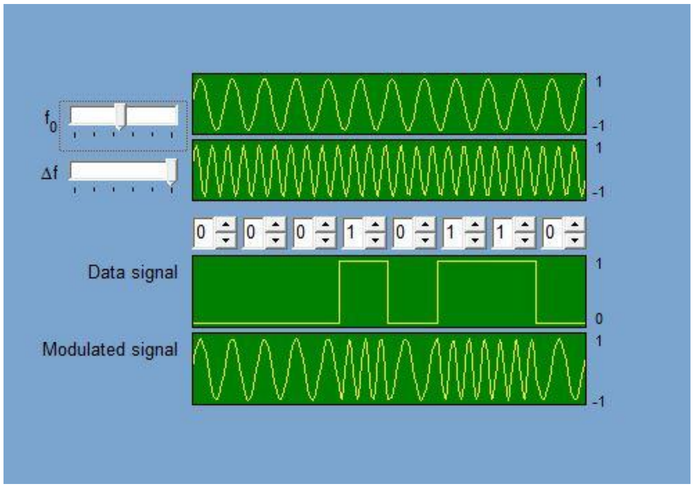


Fig 22

Experiment: Observing FSK on the oscilloscope

In the following experiment we will first study the modulated signal in frequency shift keying. Periodically a constant data byte will be sent as the data signal and modulated on to the sinusoidal carrier signal. The center frequency and frequency deviation can then be determined from the signals.

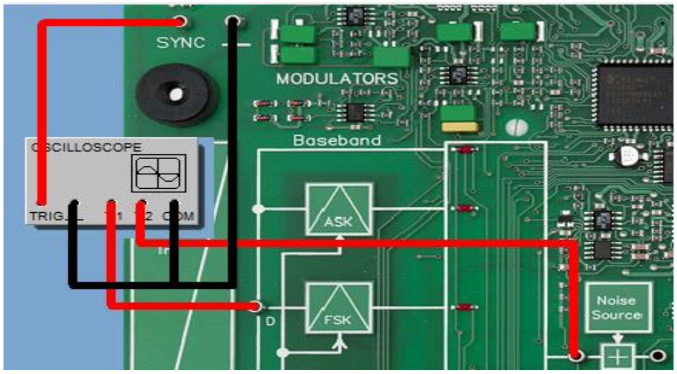


Fig 23

- What approximate values are obtained for the center frequency and frequency deviation?
  _ 5 KHz and 2.5 KHz
  _ 7.5 KHz and 2.5 KHz
  _ 7.5 KHz and 5 KHz
  _ 10 KHz and 5 KHz

FSK in the frequency range

The frequency-shifted signal can be interpreted as the combination of two amplitude-shifted signals having frequencies

$$f_1 = f_o + \Delta f \text{ and } f_2 = f_o - \Delta f.$$

The required bandwidth is therefore significantly greater than in the case of Amplitude shift keying. To reduce the bandwidth the data signal in therefore usually low-pass filtered first. The graphic at right shows the spectrum of the FSK signal with a square-wave data signal having frequency FD.

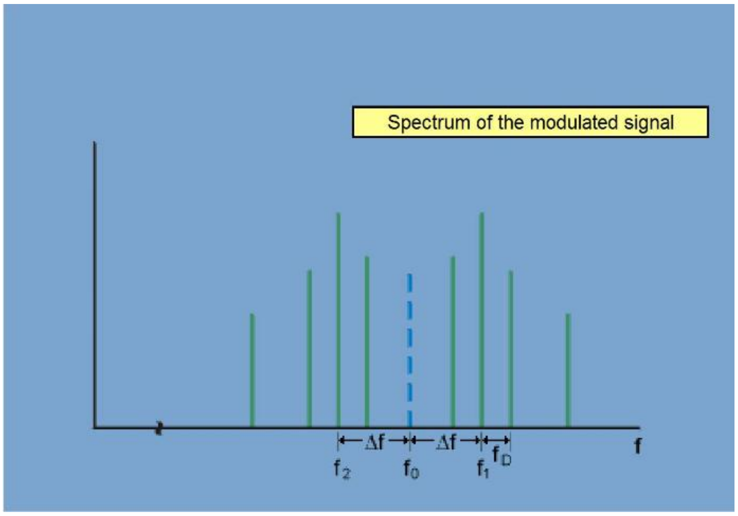


Fig 24

Experiment: Measuring the FSK spectrum (SQ Formal)

In the following experiment we will investigate the amplitude spectrum of the modulated signal with Frequency Shift Keying. A periodic square-wave signal with a DC component and frequency f_D =600 Hz will again serve as the data signal.

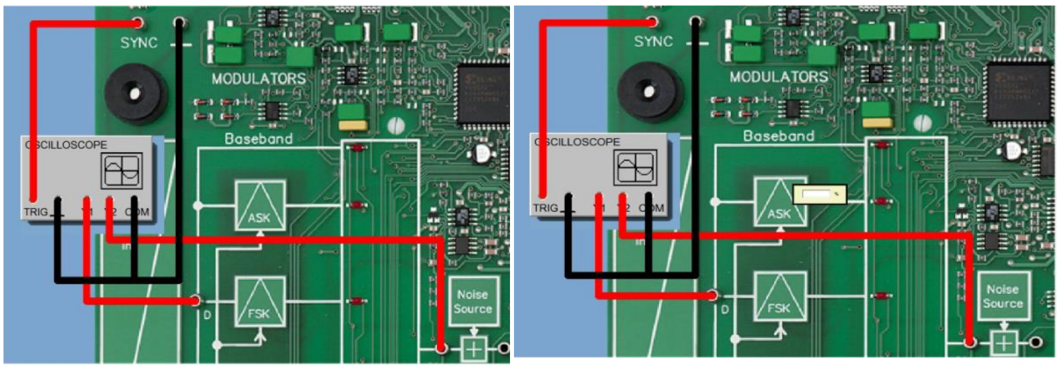


Fig 25

Note :

An amplitude spetrum symmetrical with respet to the center frequency of f0 =7.5 kHz. To the left and right of the characteristic frequencies are side-bands whose distances from the characteristic frequencies are each odd multiples of the data signal freuency FD. Both "partial spectra" left and right of the center frequency are non-symmetrical with respect to the amplitudes of the side-bands.

Experiment: Measuring the FSK spectrum (NRZ/DIF-Format)

In the following experiment the amplitude spectrum of the modulated signal in Frequency Shift Keying will be studied, but now continuously varying data will be sent as the data signal in NRZ and DIFF format. The COM3LAB spectrum Analyzer is again used to determine the spectrum.

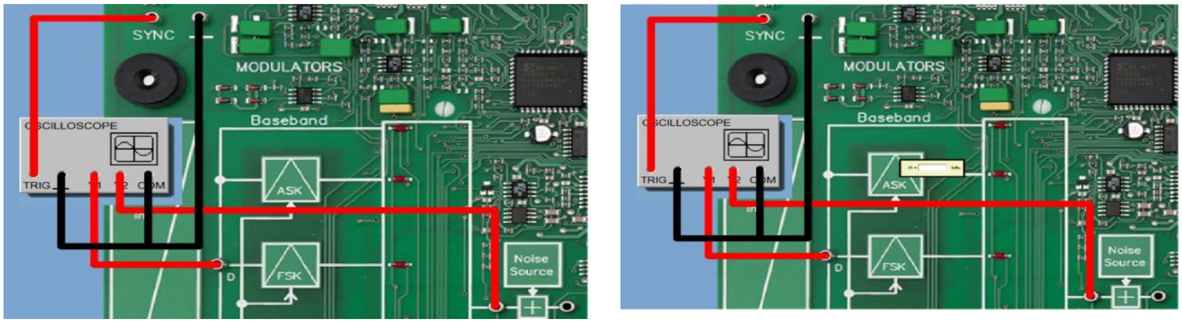


Fig 26

Note :

As in the case of square-wave modulation, a line spectrum (in this case due to the "Max Hold" function as an overlay of several individual spectra) is formed around the two center frequencies F_1 and F_2. The resolution of the spectrum analyzer is in sufficient however to separate the individual lines. The structure of the spectrum (particularly the envelope) corresponds rather exactly with that of the spectrum for square-wave modulation, so that the results obtained there can be qualitatively assumed for more general data signals.

Demodulation

To demodulate the FSK signal it is first converted into an ASK signal. This is done using an absorption circuit (edge discriminator), which acts as a band-stop filter and allows frequency(bit logical 1) to pass through virtually unchanged, but which strongly damps the lower frequency (bit logical 0). The ASK signal thus created is then demodulated in the follow-on ASK demodulator (comparator + mono flop). This is referred to as non-synchronous demodulation. Synchronous demodulators are in practice also often used.

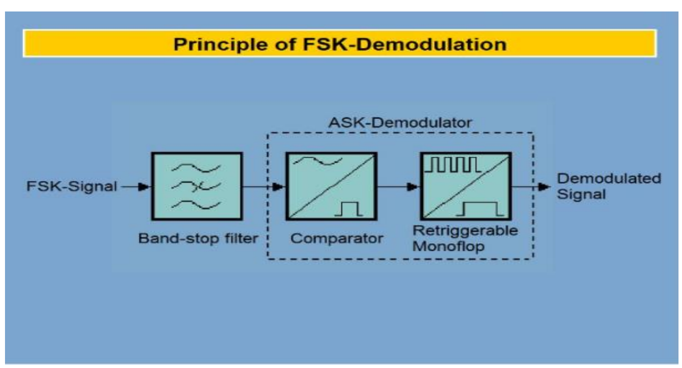


Fig 27

Experiment: Measurements on the frequency discriminator

In the following experiment we will study the demodulation of an FSK-modulated signal in NRZ formatting more closely. A periodic square-wave signal (SQ format) will be sent an the signal trace recorded in the various stages of the FSK demodulator. The original data signal and demodulated data signal will then be compared.

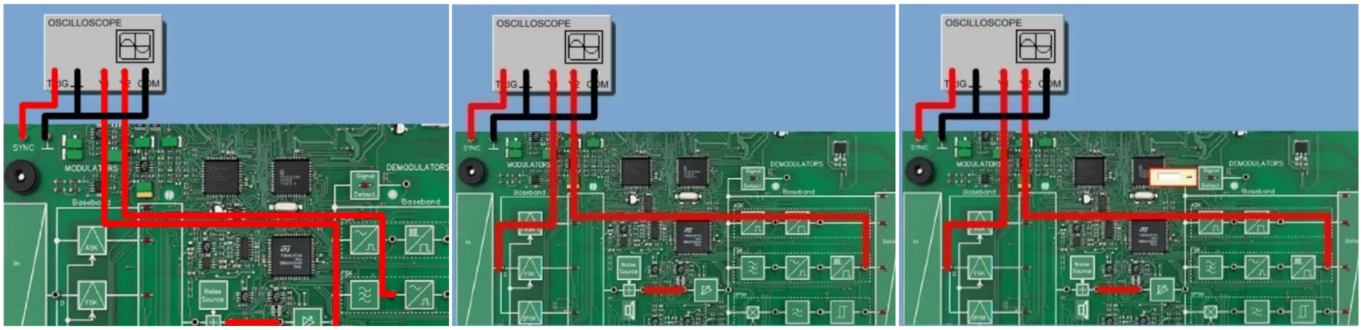


Fig 28

Note :

You can clearly see the effect of the band-stop filter: Whereas the upper frequency is only weakly damped, the amplitude of the lower frequency drops considerably. The band-stop filter also causes a phase shift in the signal however. Therefore there is a time delay of approx. 100 µs between the original data signal and the demodulated signal at the output of the FSK demodulator.

Summary

In the fig 29. blow you will see a summary of the material covered in this chapter.

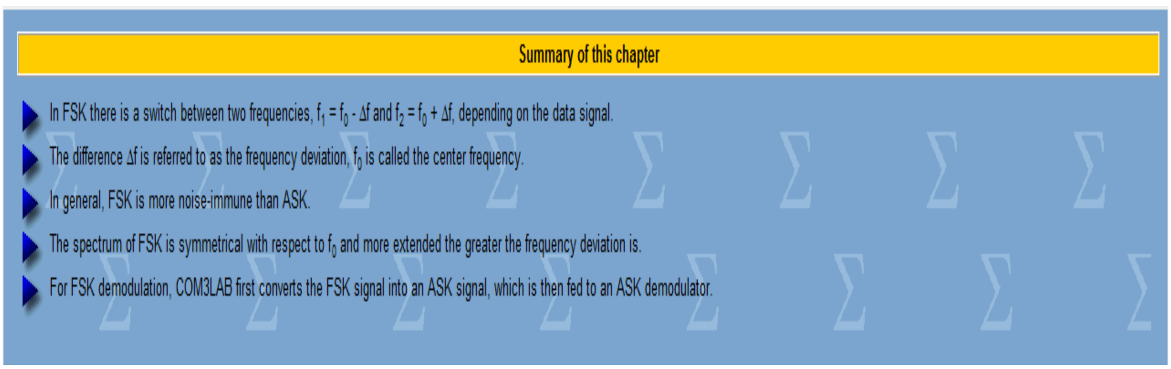


Fig 29

Two – phase Shift keying (2PSK)

Time function

In two –phase shift keying (2PSK) a sinusoidal carrier signal with frequency f_0 is toggled between two different phase positions depending on the data signal. Since disturbances generally affect only the amplitude of the signal and not its phase position, Phase Shift Keying is very noise-immune.

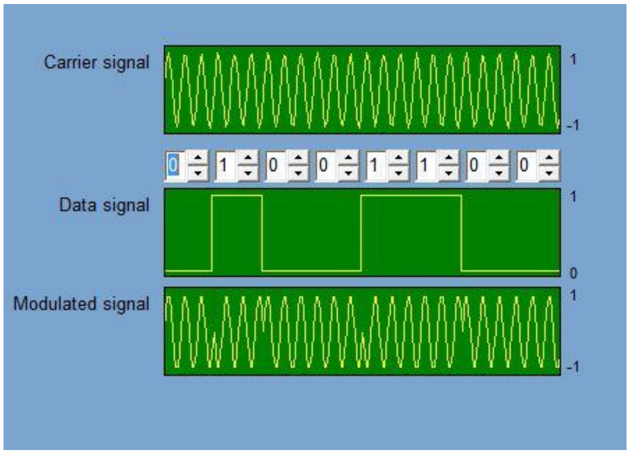


Fig 1

Experiment: Displaying 2PSK on the oscilloscope

In the following experiment we will first study the modulated signal in two –phase shift keying. A periodic square-wave signal (SQ format) will be sent as a data signal and modulated on to the sinusoidal carrier signal. Both signals will be compared on the oscilloscope. Both the phase positions for logical 0 and logical 1 will be determined from the signals.

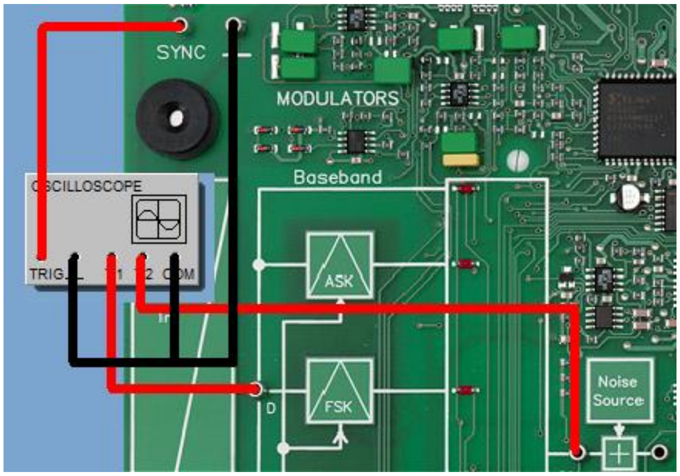


Fig 2

➢ Which phase position of the sinusoidal carrier results for logical 0 and logical 1?

_ 0^0 and 180^0

_0^0 and 90^0

_-180^0 and 180^0

_ 0^0 and 45^0

Note:

With the selected square-wave data signal alternating LOW and HIGH pulses are sent; each has a duration of 800 µs (upper curve).

The frequency of the carrier is constant at a value of 10 kHz. For a data bit of logical 0 the carrier has a phase shift of 0^0 , and for a data bit of logical 1 a phase position of 180^0.

2PSK in the frequency range

The spectrum of the 2PSK signal corresponds to amplitude modulation with suppressed carrier.

The adjacent graphic shows the spectrum for modulation with a square-wave data signal having frequency f_D . As in the case of ASK, the minimum required transmission bandwidth is therefore

$$B = 2.f_D$$

In practice a value of 1.4 x is generally used.

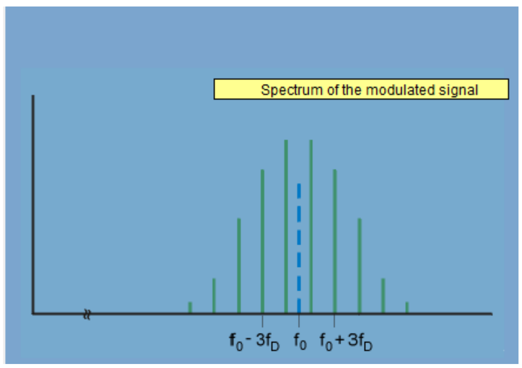


Fig 3

Experiment: Measuring the 2PSK spectrum (SQ -Formal)

In the following experiment we will study the amplitude spectrum of the modulated signal in two –phase shift keying. As the data signal we will again use a periodic square-wave signal having a DC component (SQ -Formal) and frequency f_D =600Hz.

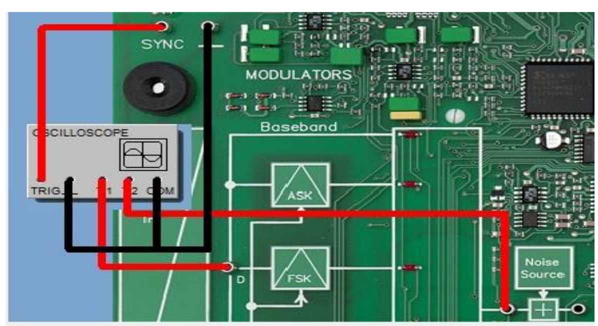


Fig 4

- How large is the amplitude of the spectral line for the carrier frequency f_D ?
  _ 0
  _ 0.2
  _ 1.5
  _ 10

Note:

The result is an amplitude spectrum which is symmetrical with respect to the carrier frequency of f_0 = 10 kHz, whereby the carrier itself no longer appears in the spectrum. To the left and right of the carrier frequency are side-bands whose separations are each an odd multiple of the data signal frequency f_D and whose amplitude decreases with increasing separation.

Experiment: Measuring the 2PSK spectrum (NRZ/DIF-Format)

In the following experiment we will again study the modulated signal in two- phase shift keying, whereby the data signal will now consist of continuously varying data in NRZ and DIF format. The COM3LAB Spectrum Analyzer is again used to determine the spectrum.

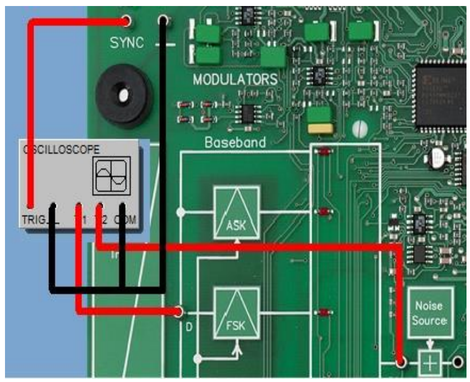


Fig 5

Note :

As in the case of square-wave modulation , a line spectrum (in this case due to the 'MaxHold' function as an overlay of several individual spectra) is formed around the carrier frequancy $f0$ for NRZ and DIFF format .The resolution of the spectrum analyzer is insufficient however to separate the individual lines .The structure of the spectrum (particurlarly the envelope) corresponds rather exactly with that of the spectrum for squre-wave modulation ,so that the results obtained there can be qualitatively assumed for more general data signals.

Demodulation

For synchronous demodulation of the 2PSK signal the carrier signal with the reference phase position is required. By multiplying with the 2PSK signal , the result is first a signal with double the carrier frequency whose center value over a bit width varies with the actual data bit .A downstream low-pass filter with pulse former (Schmitt trigger) finally regenerate the original data signal .

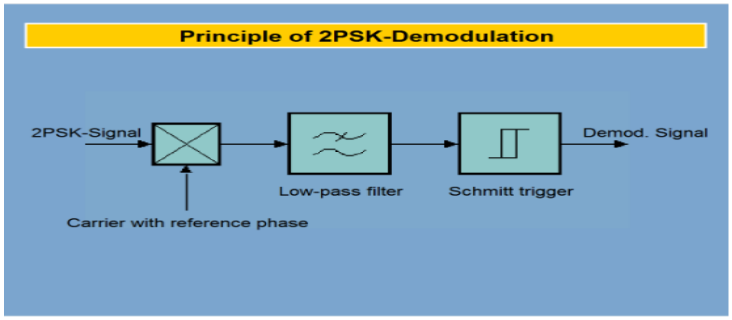
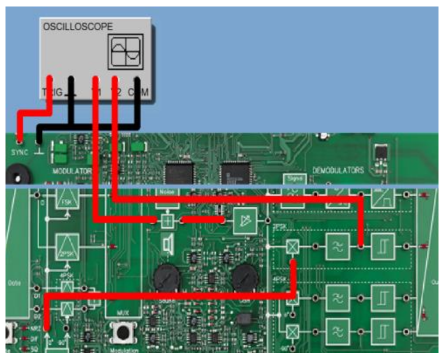


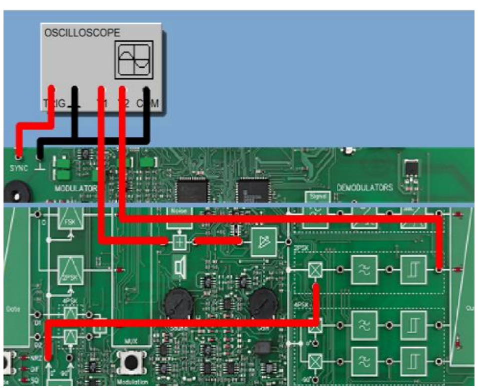
Fig 6

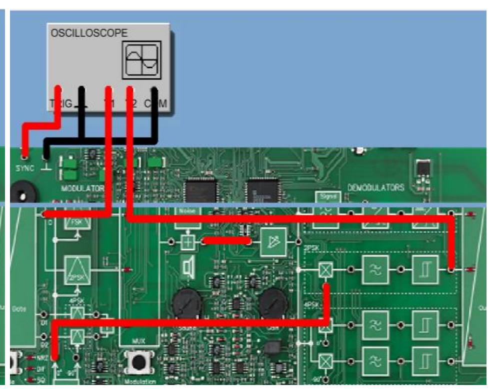
Experiment: Synchronous demodulation with 2PSK

In the following experiment we will investigate in greater detail the demodulation of a 2PSK modulated signal with NRZ formatting. A periodic square-wave signal (SQ format) will be sent and course of the signal recorded in the various stages of the PSK demodulator. The original data signal and demodulated data signal will then be compared.



(1)





(2) (3)

Fig 7

Note :

The first thing you notice is the effect of the multiplication by the frequency- and phase- identical carrier signal. The low-pass filter results in a signal with an e-shaped, rising (bit logical 1) or falling (bit logical 0) edge. The Schmitt trigger generates the actual data signal from that. Which due to the effect of the low-pass and Schmitt trigger is shifted by around 120 µs from the original signal.

Carrier recovery

Since the carrier itself is not sent in 2PSK, it must be recovered in correct phase position for coherent demodulation at the receiving site. For this purpose the modulated signal can be squared; the result is a (DC component –containing) oscillation with double the carrier frequency, from which the carrier can be recovered by means of frequency dividing-though with a phase uncertainty of 180°.

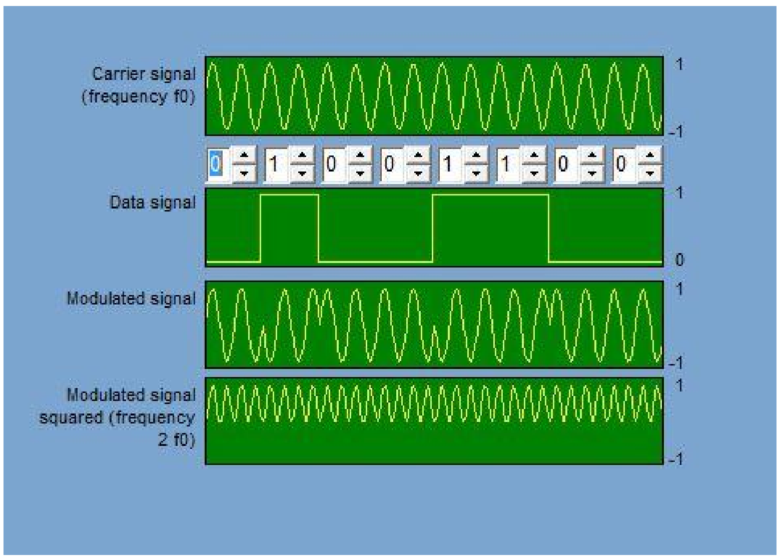


Fig 8

Experiment: Measurements on carrier recovery with 2PSK

In the following experiment we will study carrier recovery with two- phase shift keying by squaring the modulated signal. the analysis will be made **both in the time and frequency range** the data signal will again be a periodic square-wave signal with a DC component and a frequency of f0 = 600 Hz .

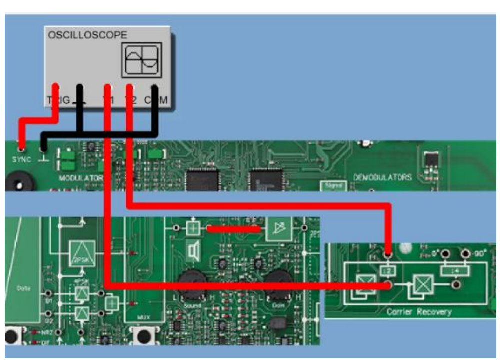


Fig 9

➢ Which spectral lines dose the output signal for the frequency divider show?
_ Just one line at 10kHz
_ lines at 10, 20, 30….kHz
_ lines at 10, 30, 50….kHz
_ lines at 10, 50, 100….kHz

Note:

Multiplying the 2PSK signal by itself results in purely sinusoidal signal with a frequency of 2 f0 = 20 KHZ. Whose Dc component is filtered out on the COM3LAB board. The frequency divider forms this signal into a square-wave signal with frequency f which due to square shape also contains harmonics at odd multiples of the carrier frequency .

Summary

In the fig.10 you will see summary of the material covered in this chapter .

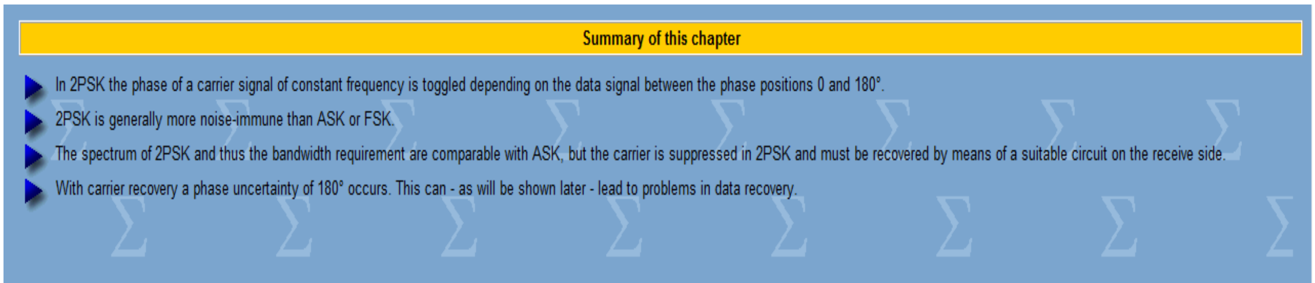


Fig 10

Four – phase Shift keying (4PSK)

4PSK time function

In four-phase shift keying (4PSK) a sinusoidal carrier signal having frequency f_0 is toggled depending on the data signal among four phase positions (45°, 135°, 225° and 315°). Since two bits are required for distinguishing four phase positions, in 4PSK each phase position is specified by a pair of successive bits (Dibit) and therefore is retained for each doubled bit duration. If the transmission channel has the same properties, 4PSK allows double the bit rate compared with 2PSK, or the same bit rate with half the bandwidth requirement.

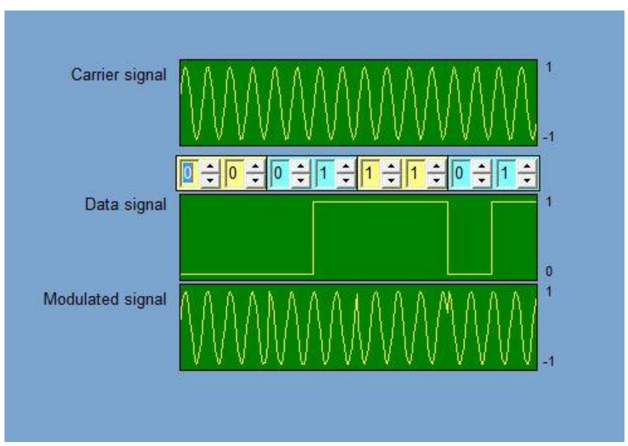


Fig 11

4PSK generation

Four-phase Shift Keying is implemented with double, parallel Two-phase Shift Keying of the Dibit signals $D_1(t)$ and $D_2(t)$ with orthogonal carrier signals (phase position 0° and -90°) and summing of both signals. The 4PSK signal obtained in this way then has, depending on the bit combination present, phase positions of 45°, 135°,225° and 315°.

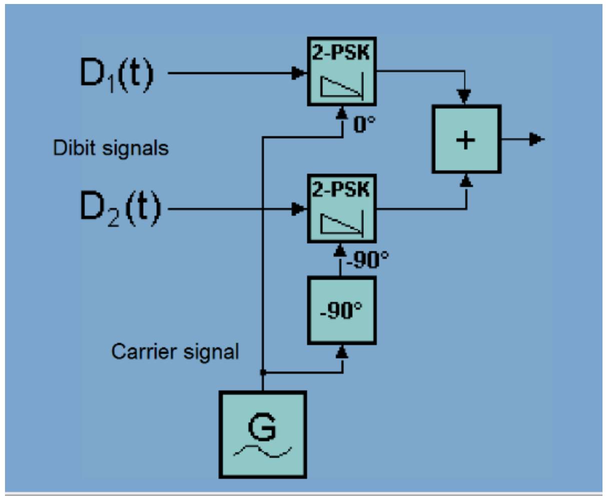


Fig 12

Experiment: Carrier signals in 4PSK

In the following experiment we will first study the courses of the two orthogonal carrier signals used in 4PSK. The courses will be captured by the oscilloscope and recorded both over time and in XY coordinates.

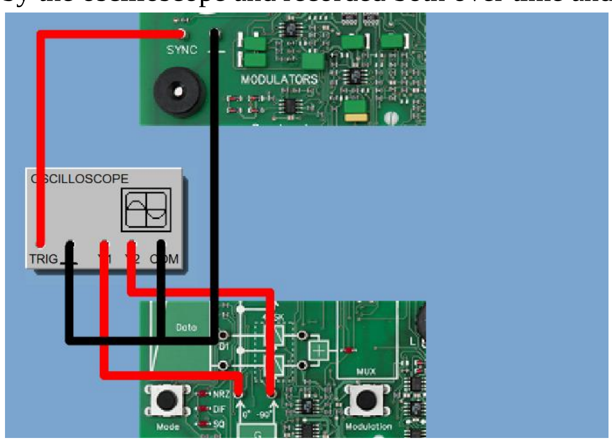


Fig 13

- Which curve form do you see in the XY representation?
  _ Square or rectangular
  _ Ellipse or circle
  _ Vertical line
  _Horizontal line
  _ Spiral

Note:

The oscillograms show clearly in the time representation the phase shift of -90° between the two carrier signals in the upper and lower 2PSK branch of the 4PSK modulator. In the XY representation there is (depending on the scaling) an ellipse or (if the axes have the identical scale) a circle, since the carrier signals each represent a sinusoidal resp. co sinusoidal function.

Experiment: Elements of the 4PSk signal

In the following experiment we will analyze the 4PSK signal made up of the two 2PSK signals. A data byte of 03 will be sent periodically. The courses of the two 2PSK signals will be captured with the oscilloscope.

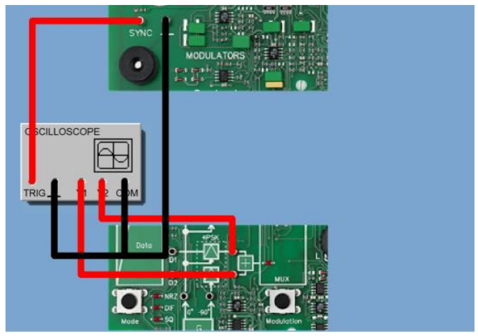


Fig 14

- How large is the phase jump for both 2PSK signals?
  _ 90° resp. 180°
  _ 45° resp. 90°
  _Both times 90° each
  _Both times 180° each

Note:

The oscillograms show clearly the phase shift of -90° between the two carrier signals in the upper and lower 2PSK branch of the 4PSK modulator. Since both branches perform “normal” 2PSK modulation, the result for 0-1- and 1-0- edges of the data signal are jumps of 180° each.

Dibit –generation

The digit signals $D_1(t)$ and $D_2(t)$ obtained by means of a serial-parallel conversion from the original bit sequence D(t). In principle this is an extension of the bit to the duration $T_{Dibit} = 2T_{Bit}$ and a division into two separate signal sequences. The first bit pair is delayed by the time duration of one bit in the order to bring both dibit signals to the same clock start.

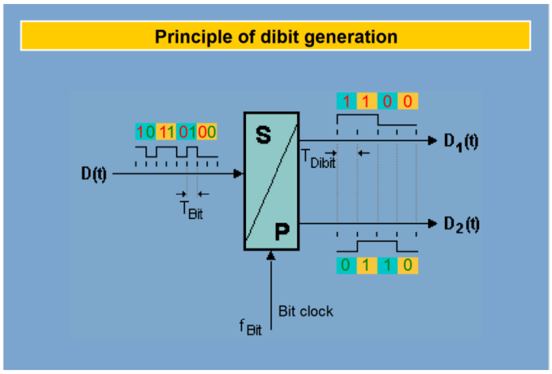


Fig 15

Experiment: Measuring dibit signals D1 and D2

In the following experiment we will investigate the principle of dibit generation through serial-to-parallel conversion. A suitable data byte will be periodically generated and the resulting dibit signals $D_1(t)$ and $D_2(t)$ recorded with the help of the digital analyzer. Form the courses we will then determine the transmission rate for four-phase shift keying.

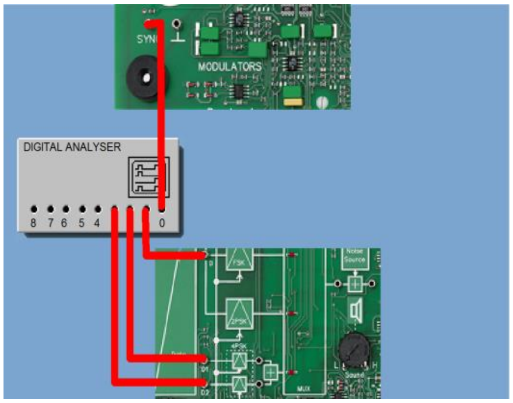


Fig 16

Question :

From results determine the new bit rate ?

4PSK in the frequency range

If four-phase shift keying is operated at the same bit rate as two-phase shift keying, a signal step has twice the duration. This reduces the spectrum as compared with 2PSK to half the width, as shown in the adjacent graphic for a square-wave data signal having frequency f_D. The minimum required transmission bandwidth is thus only $B = f_D$.

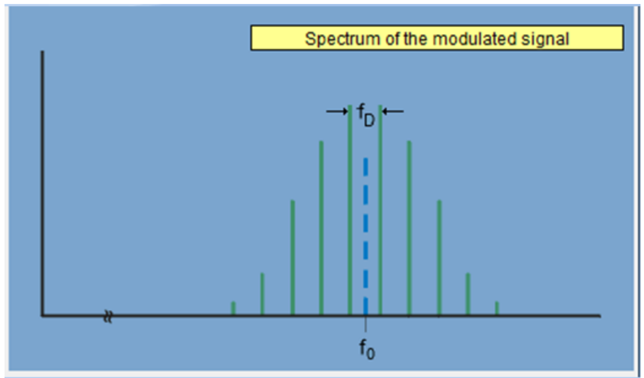


Fig 17

Experiment: Measuring the 4PSK spectrum

In the following experiment we will investigate the amplitude spectrum of the modulated signal in Four-phase Shift Keying. The data signal will again be a periodic square-wave signal with a DC component and a frequency F_D =600 Hz.

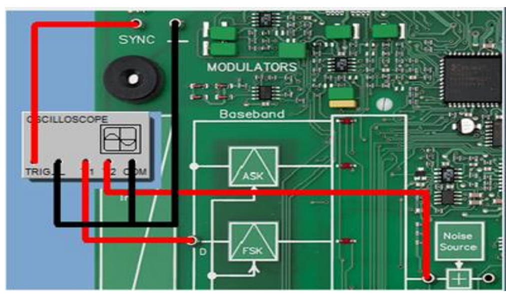


Fig 18

- ➢ What can you say about the spread of the 4PSK spectrum as compared with the 2PSK spectrum?
  _ It is spread by just about half
  _ it has about twice the spread
  _ it has about the same spread
  _ it has about Four times the spread

Note:

Four-phase shift keying works with double the bit rate as 2PSK, but the spectrum has about the same spread, in other words with the same bandwidth requirement 4PSK lets you send at double the bit rate. The carrier itself does not appear in the spectrum in 2PSK.

Experiment: Composition of the 4PSK spectrum

In the following experiment we will deepen our understanding of the 4PSK spectrum with square-wave modulation by investigating the composition of the spectrum made up of the two 2PSK spectra, in other words the spectra of the carrier signals D1 and D2 and shifted by 90° (in-phase modulator and quadrature phase modulator). A square-wave signal (SQ format) will again be used as the modulating signal.

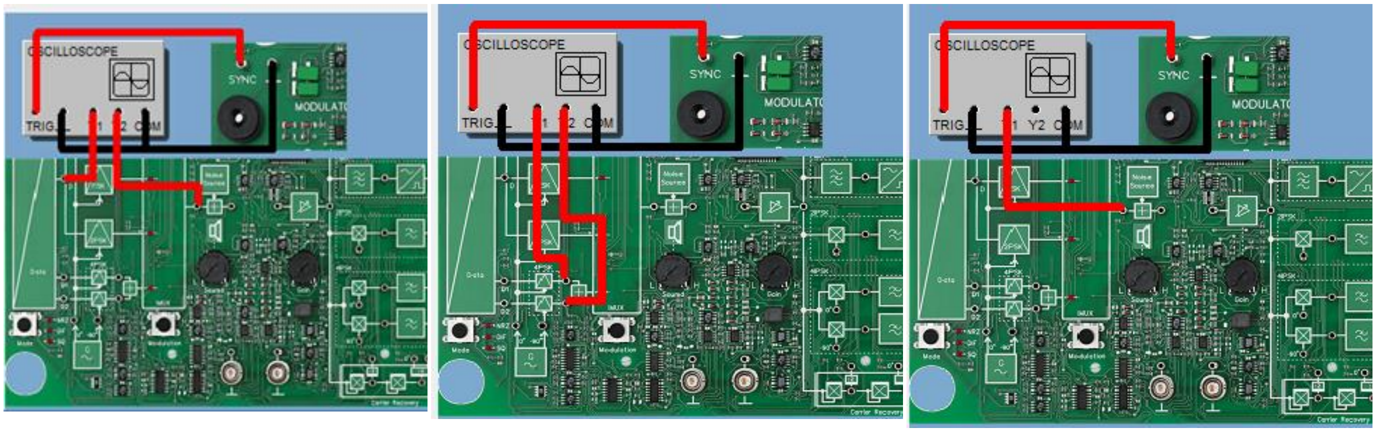


Fig 19

- Question :
  Compare the spectrum of square wave data, 2PSK, in-phase 4PSK, quadrature phase 4PSK and 4PSK?

- Which of the following applies to the line separations of the spectra for the in-phase and quadrature phase modulator?
  _ They are both about the same
  _ The spectrum of the in-phase modulator has about half the line separation
  _ The spectrum of the in-phase modulator has about double the line separation
  _ The spectrum of the in-phase modulator has about four times the line separation

Note :

Because the frequency of the D_1-square-wave signal is twice as high compared with the D_2-square-wave signal, the output of the in-phase modulator shows a spectrum with about twice the line separation (namely 1200 Hz). The 4PSK spectrum is then created by additive overlay of both spectra and has twice the spread of the 2PSK spectrum in spite of the doubled data transfer speed. The measurement of the 4PSK spectrum on the multiplexer output is however incorrect with respect to amplitudes due to the amplitudes due to the phase uncertainly of the FFT.

Demodulation

For synchronous demodulation of 4PSK signal, two parallel 2PSK synchronous demodulators are used which work with -90° shifted carrier signals. The resulting dibit signals D_1(t) and D_2(t) are restored to the total data signal D(t) in a parallel/serial converter.

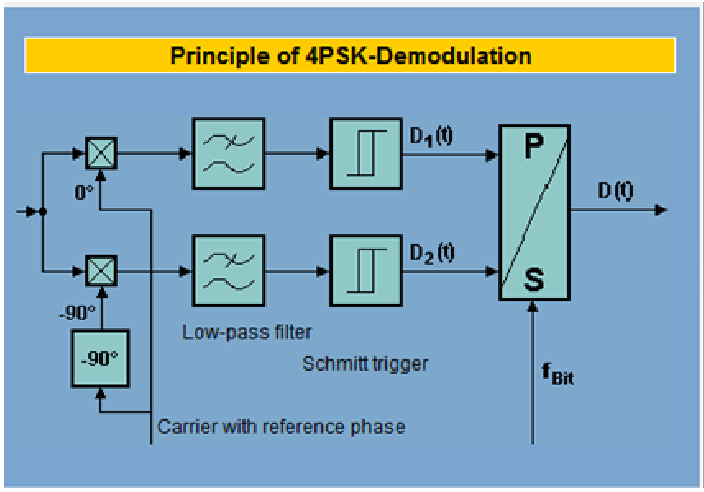


Fig 20

Experiment: investigating 4PSK demodulation with the oscilloscope

In the following experiment we will see that the 4PSK demodulator consist essentially of two separate 2PSK demodulators which each work with 90° shifted carriers. We will also investigate the demodulation of a 4PSK modulated signal in NRZ format, whereby periodically a data byte of 33 is sent. The original data signal and demodulates data signal will then be compared.

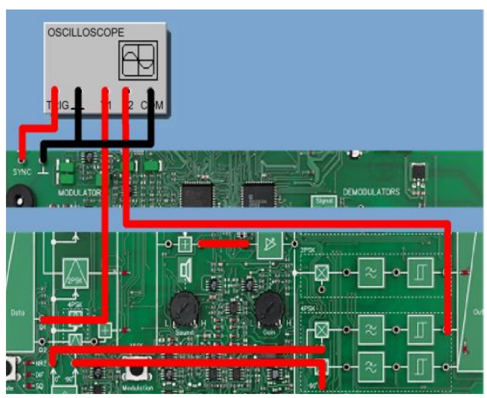


Fig 21

Note:

Result Since 4PSK demodulation consists essentially of two separate 2PSK synchronous demodulators, you get for the individual debits identical ratios and signal courses as for 2PSK demodulation. The original and demodulated debit are each shifted due to the effect of the low-pass filter and Schmitt trigger**.**

Carrier recovery

Also with four-phase shift keying the carrier needs to be recovered at the receiving location. For this purpose the modulated signal is squared twice; the result is an oscillation with a DC-component and frequency components at 2 f_o (caused by the first quadrupling) and 4f_o (caused by the second quadrupling). The carrier can be recovered from this by filtering out the fourth harmonic (in other words the frequency component at 4 f_o)and then dividing the frequency by four-though with a phase *uncertainty of I 90°.The 2x squared modulated signal does not change when the data signal changes.*

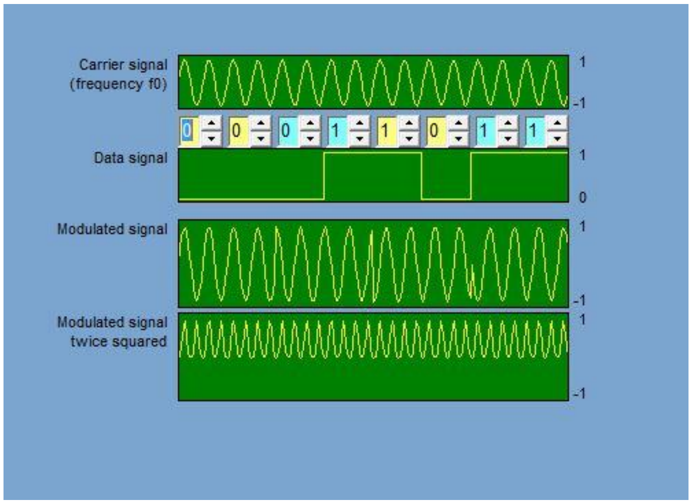


Fig 22

Experiment: Measuring carrier recovery with 4PSK

In the following experiment we will investigate carrier recovery with 4PSK whereby the modulated signal is squared twice. The analysis will be carried out **in both the time and frequency range**. Of particular interest are the signal spectra in front of the behind the as the "data signal". frequency divider. To make the connections especially clear, the original carrier in 90º. Position will be used as the "data signal".

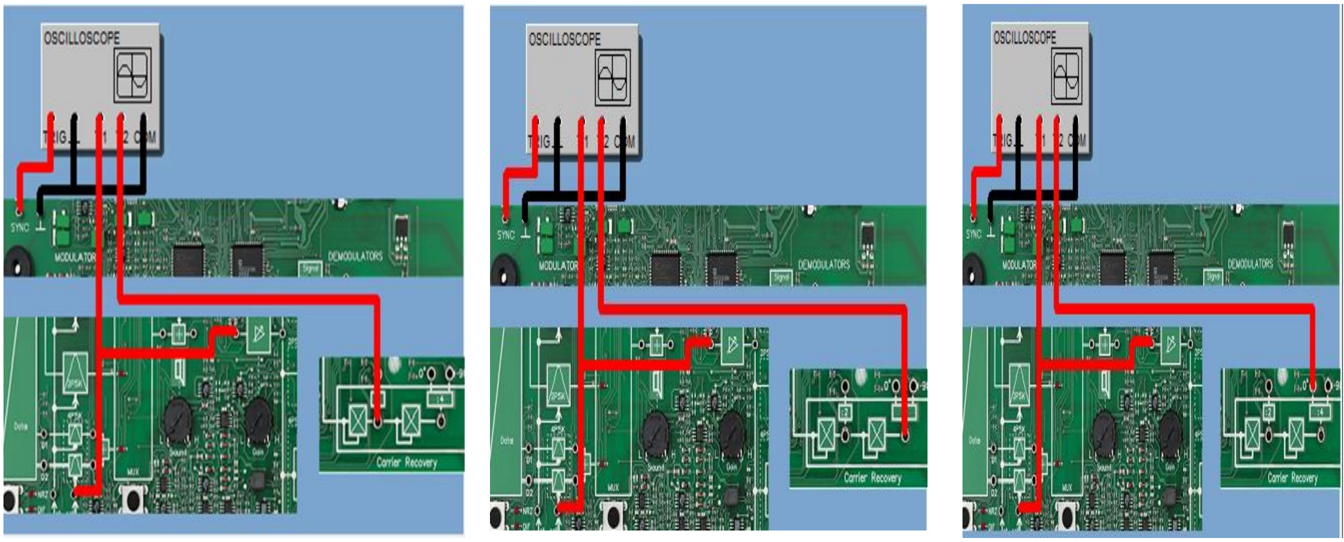


Fig 23

- Which spectral lines does the signal behind the signal behind the frequency divider show?
  _ Just one line at 10kHz
  _ Just one line at 20 kHz
  _ lines at 10, 20 and 40 kHz
  _ lines at 10 and 30 kHz

Note:

The first quadrupling results in a sinusoidal signal having frequency $2\,f_o = 20\,\mathrm{kHz}$, the second quadrupling in a signal with a frequency of $4\,f_o = 40\,\mathrm{kHz}$, the corresponding spectra thus consist only of a signal line. The frequency divider reconstructs the signal into a (square-wave) signal having frequency f_o, which also has low-amplitude harmonics at odd multiples of the carrier frequency.

Summary

In the fig.26 you will see a summary of the material covered in this chapter.

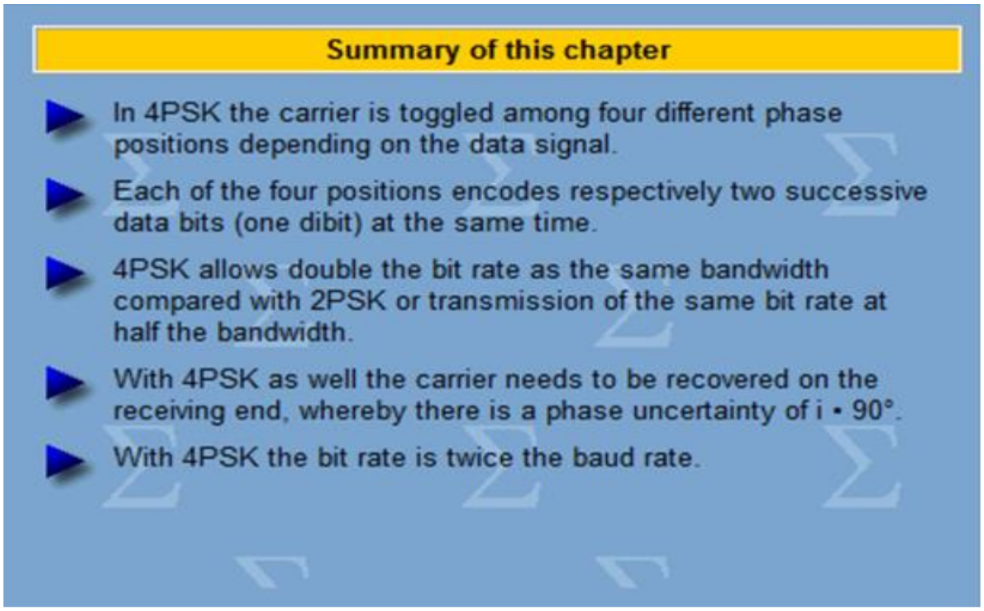


Fig 26