T 7.2.2.1

Pulse Code Modulation 564 002

Technology

T 7.2.2.1
Pulse Code Modulation

Version: 22AN11PME13W10

564 002

Notes

Use of the CASSY example files

- Copy the files included in the CD (text as *.pdf and measurement files as *.labx) in the same subdirectory.
- Install the CASSY Lab 2 Software on your PC.
- By clicking CASSY Lab 2 example, a special measurement file is loaded, customized for the experiment. If the automatic start doesn't work on your computer, you can start the file manually.
- Loading the measurement files requires the CASSY Lab Version 1.6.1 or higher.

EMC

The sensitive electronics of the equipment that is used for experimentation in this manual can be affected by ESD (electrostatic discharge). Therefore, static electricity must be avoided or eliminated by discharge. If necessary, move the experiment set-up to a desk, which is less sensitive to interference.

Experiments

This manual might contain additional experiments with devices which are not included in the scope of delivery. In this case only those experiments can be performed, for which LD delivered the required material. Thus customers are not entitled to claim for compensation such as free-of-charge deliveries of supplementary apparatus. Additional experiments deviating from the procedures described herein are possible, if carried out by qualified personnel taking into consideration prevailing security standards. The sample solutions given in the results are only approximate. Actual results can differ in principle for the following reasons from the figures given here:

- Setting of the operating points (potentiometers, encoders)
- Component and measurement tolerances
- Fluctuations in the supply voltages etc.

All measurement results were recorded with the equipment from the material list. Because Sensor-CASSY2 and the software CASSY Lab2 are very versatile tools, they are preferred for experimentation.

Manuals

Instruction sheets or additional information (software) of third-party devices and must be read prior to the tests.

Further claims from this manual are excluded!

Table of contents

Equipment overview

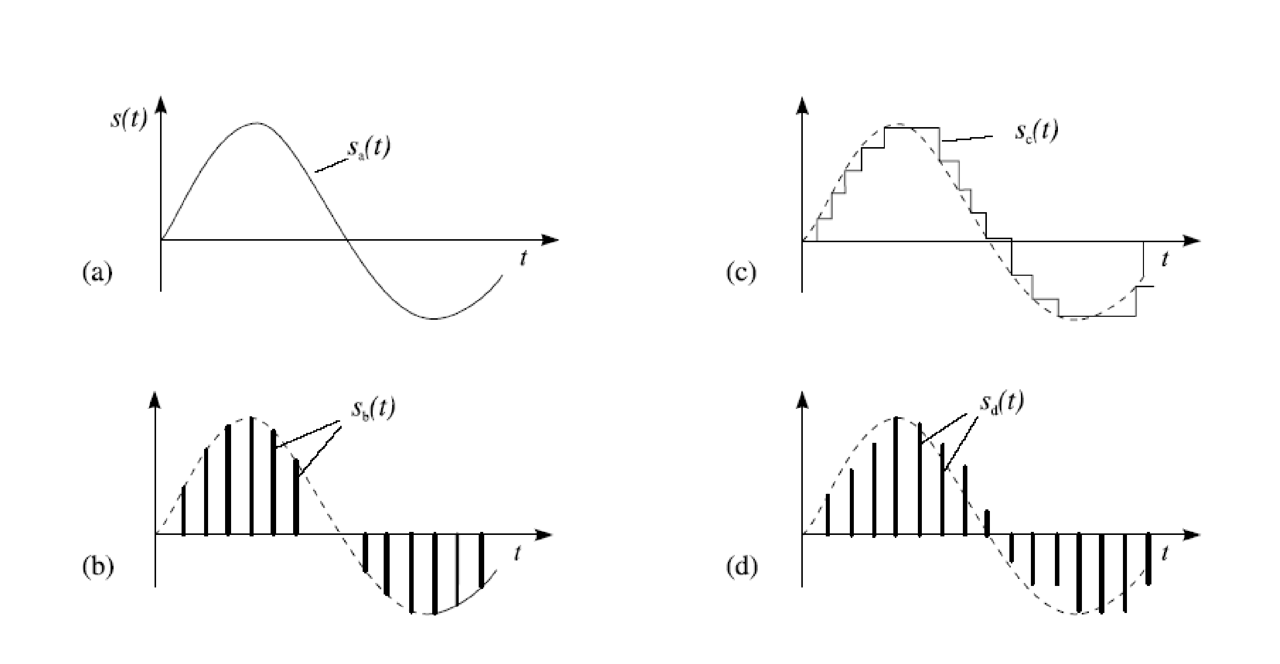
Qty.	Cat.-no.	Designation
1	736061	PAM Modulator
1	736071	PAM Demodulator
1	736101	PCM Modulator
1	736111	PCM Demodulator
		Accessories
1	72609	Panel frame T130, Two Level
1	72686	DC-Power Supply ± 15 V/3 A
2	726961	Function Generator 200 kHz, 230 V
1	524013S	Sensor-CASSY 2 Starter
3	50059	Set of 10 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

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:

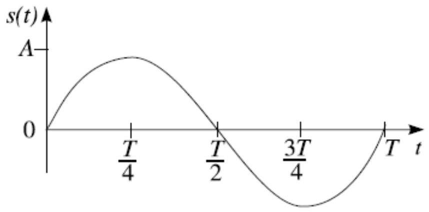
- **Analog signals**
  A signal is called analog if it is both time as well as value-continuous.
- **Digital signals**
  A signal is called digital, if it is both time as well as value-discrete.



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

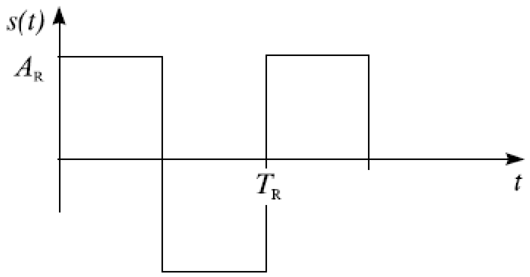
Time and spectral domain

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.

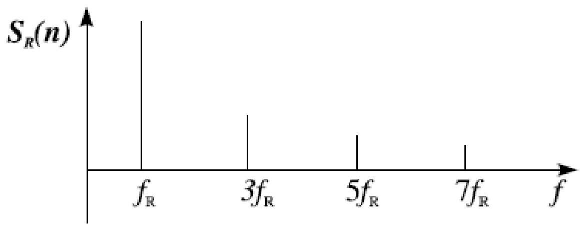


Harmonic function, time domain

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:

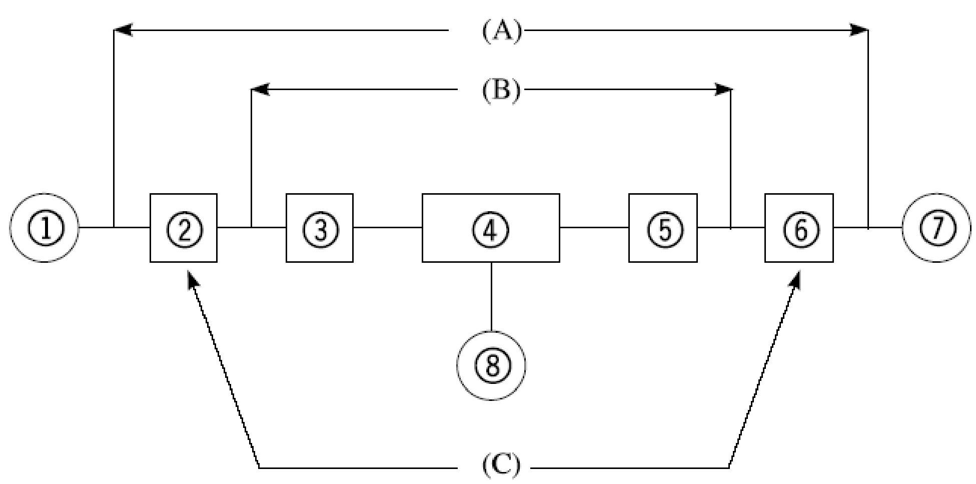
- Modulation enables the matching of the modulating signal to the characteristics of the transmission channel. (radio links e.g. are only possible for a certain frequency band.)
- Existing transmission channels can be multiply exploited using modulation, (frequency or time division multiplex systems).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

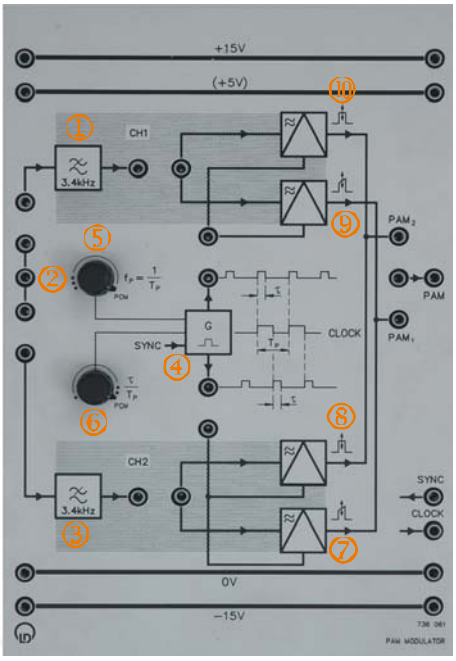
If a single transmission channel is considered, (i.e. no telephone technology), then we can concentrate on the remaining functions illustrated by the following scheme.

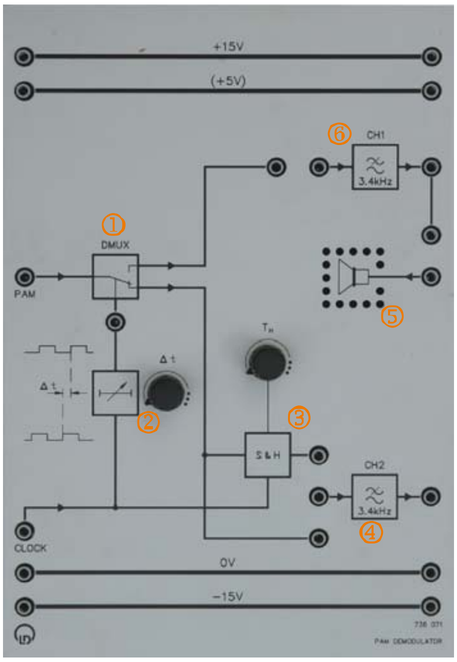


(A) The telecommunications system
(B) Message transmission
(C) Message processing
1 Message source (human being, measurement sensor etc.)
2 Converter (microphone, television camera, strain gauges, thermo sensor etc.)
3 Transmitter
4 Transmission channel (radio link, transmission cable, data storage system)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

Training system



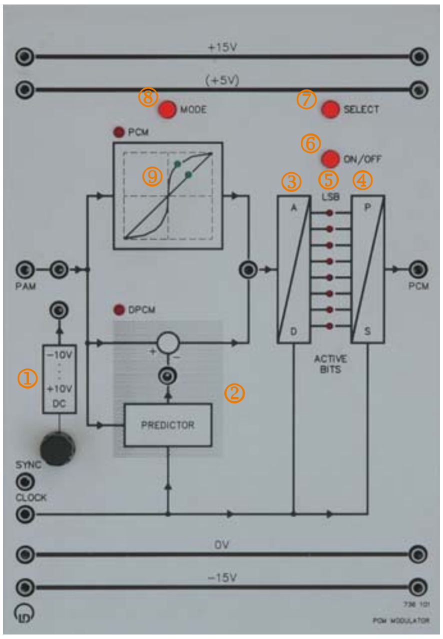


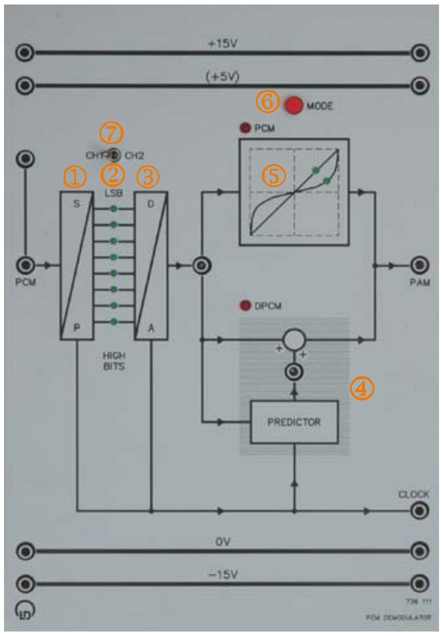
PAM Modulator

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





PCM-Modulator

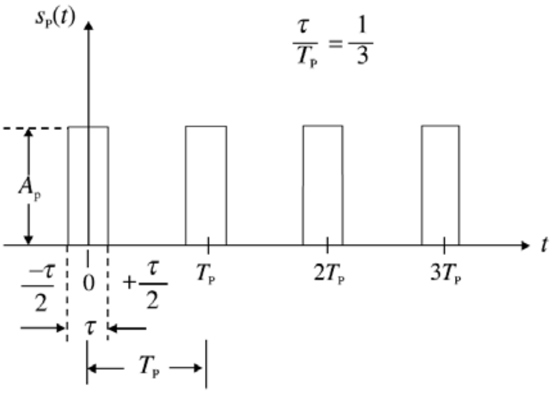
1. DC voltage source (10-turn potentiometer) for static modulation experiments
2. Predictor module to form prediction value in DPCM operating mode
3. A/D converter (8 bit, coding is carried out according to magnitude and polarity)
4. Parallel/serial converter. The data stream at the output of the PCM 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

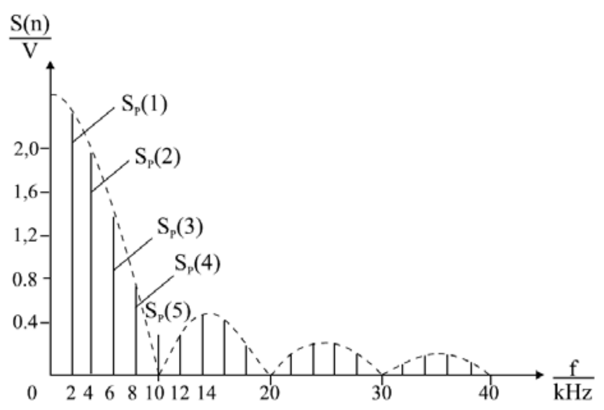
Instead of the pulse period T_P, its reciprocal value, the pulse frequency $f_P = 1/T_P$ can also be equally employed.

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

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

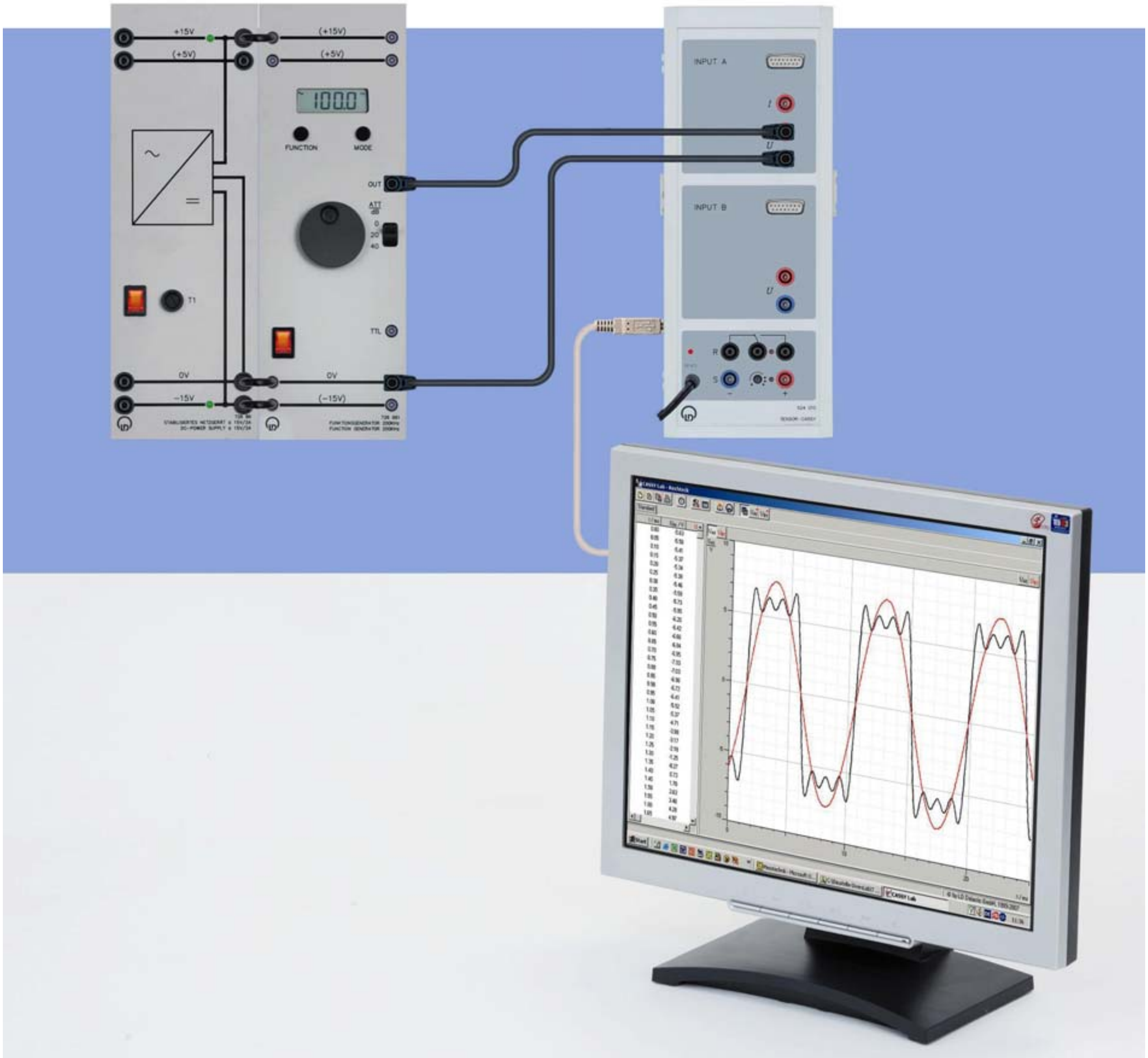
A_P=6V, τ/T_P=0.2, f_P=2kHz

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

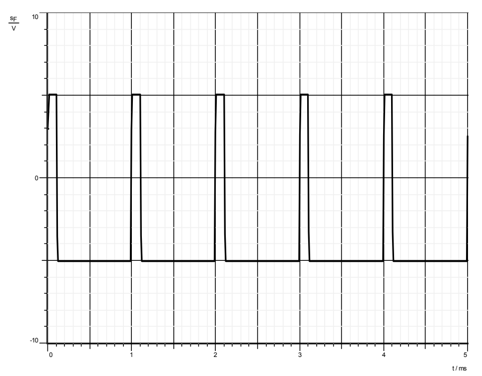
1	524 013S	Sensor CASSY 2 Starter
1	726 961	Function generator 200 kHz
1	726 86	Stabilized power supply ±15 V, 3 A
1	726 09	Panel frame T130, two level
1	501 461	Pair of cables 100 cm, black
1	501 511	Set of bridging plugs, black
1	564 002	Book: Pulse Code Modulation
1		PC

Carrying out the experiment

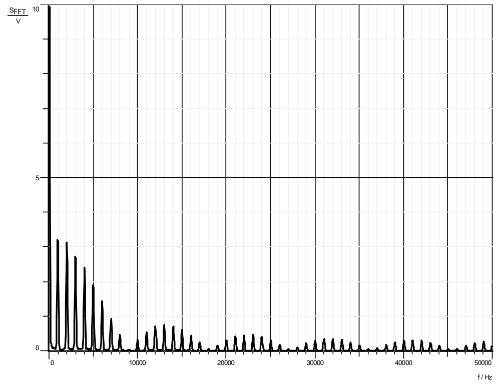


- Set up the shown experiment.
- Select a pulse train at the function generator with fp = 1 kHz, pulse amplitude A_P = 5 V (10 V_{pp}) and duty cycle τ_1/T_P = 1/10.
- Load the CASSY Lab 2 example PulseTime.labx.
- Start the measurement by pressing *F9*
- Determine the time characteristic of the pulse train.
- Determine the spectrum of the pulse train. Load the CASSY Lab 2 example PulseFFT.labx.
- 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_2/T = 2/10, τ_3/T_P = 3/10, τ_4/T_P = 4/10, τ_5/T_P = 5/10 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?

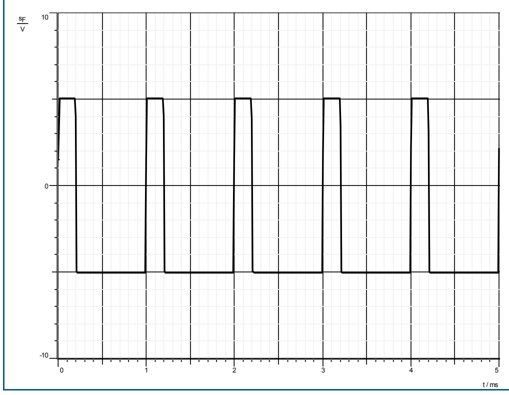
Results



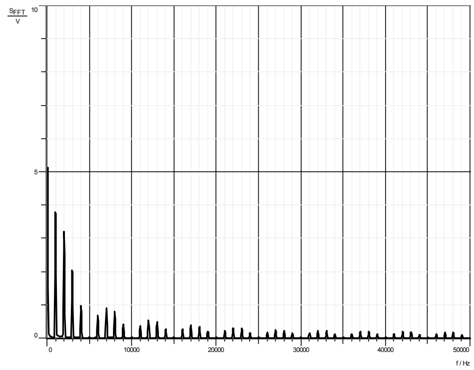
Time characteristic of the pulse train
A_P = 5 V, f_P = 1 kHz, τ_1 = 1/10



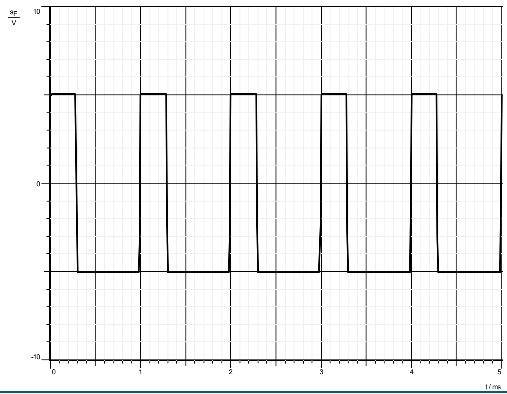
FFT spectrum of the pulse train
A_P = 5 V, f_P = 1 kHz, τ_1 = 1/10
Number of lines in each sub spectrum: l = 9
1. zero crossing of the sync-function: 10 kHz



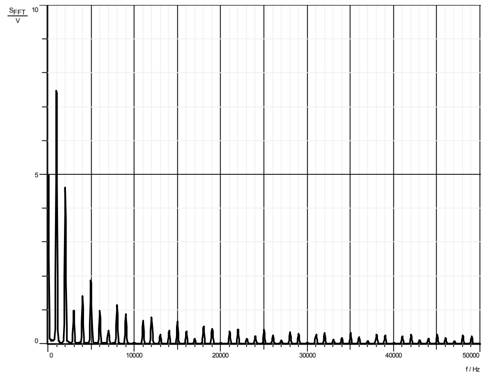
Time characteristic of the pulse train
A_P = 5 V, f_P = 1 kHz, τ_2/T_P = 2/10



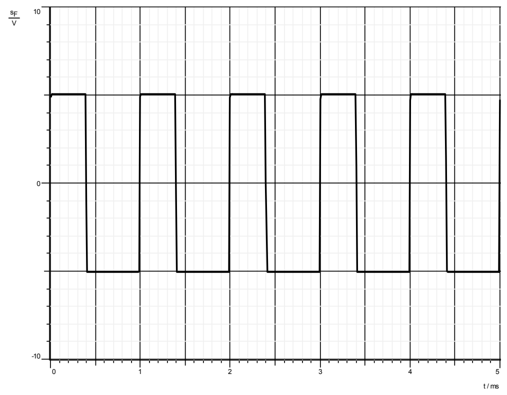
FFT spectrum of the pulse train
A_P = 5 V, f_P = 1 kHz, τ_2/T_P = 2/10
Number of lines in each sub spectrum: l = 4
1. zero crossing of the sync-function: 5 kHz



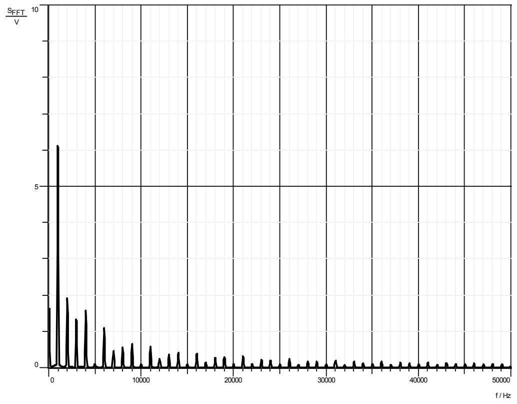
Time characteristic of the pulse train
A_P = 5 V, f_P = 1 kHz, τ_3/T_P = 3/10



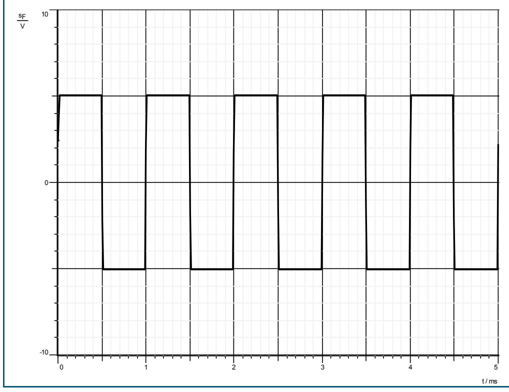
FFT spectrum of the pulse train
A_P = 5 V, f_P = 1 kHz, τ_3/T_P = 3/10
Number of lines in each sub spectrum: l = 3
1. zero crossing of the sync-function: 3.3 kHz



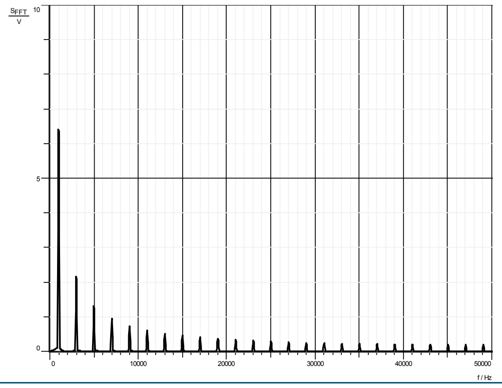
Time characteristic of the pulse train
A_P = 5 V, f_P = 1 kHz, τ_4/T_P = 4/10



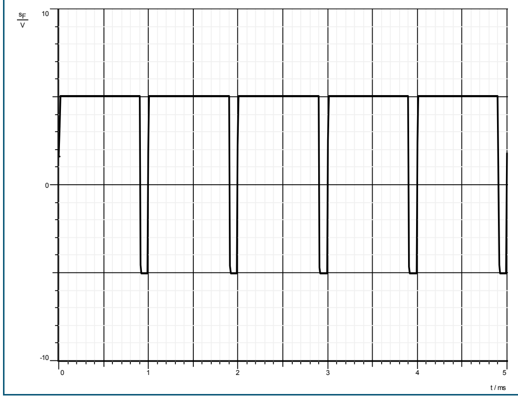
FFT spectrum of the pulse train
A_P = 5 V, f_P = 1 kHz, τ_4/T_P = 4/10
Number of lines in each sub spectrum: l = 2
1. zero crossing of the sync-function: 2.5 kHz



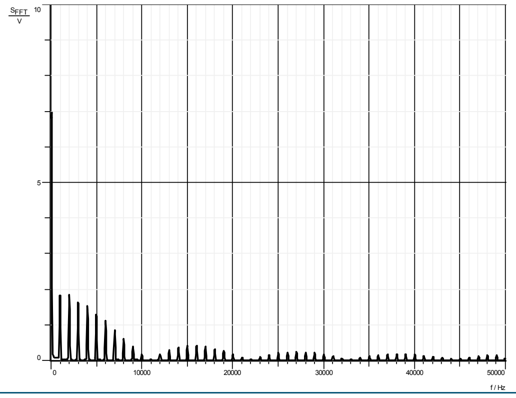
Time characteristic of the pulse train
A_P = 5 V, f_P = 1 kHz, τ_5/T_P = 5/10



FFT spectrum of the pulse train
A_P = 5 V, f_P = 1 kHz, τ_5 = 5/10
Number of lines in each sub spectrum: l = 1
1. zero crossing of the sync-function: 2 kHz



Time characteristic of the pulse train
A_P = 5 V, f_P = 1 kHz, τ_6/T_P = 9/10



FFT spectrum of the pulse train
A_P = 5 V, f_P = 1 kHz, τ_6/T_P = 9/10
Number of lines in each sub spectrum: l = 9
1. zero crossing of the sync-function: 10 kHz

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	10.00	20.00	30.00
2/10	5.00	10.00	15.00
3/10	3.33	6.66	10.00
4/10	2.50	5.00	7.50
5/10	2.00	4.00	6.00

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/ττ. 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.

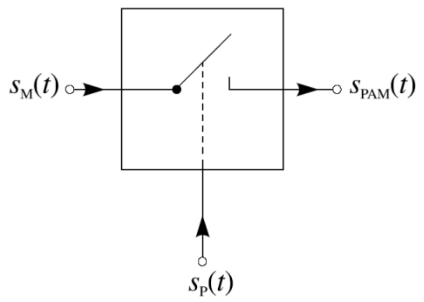
Reducing the duty cycle means:

- 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



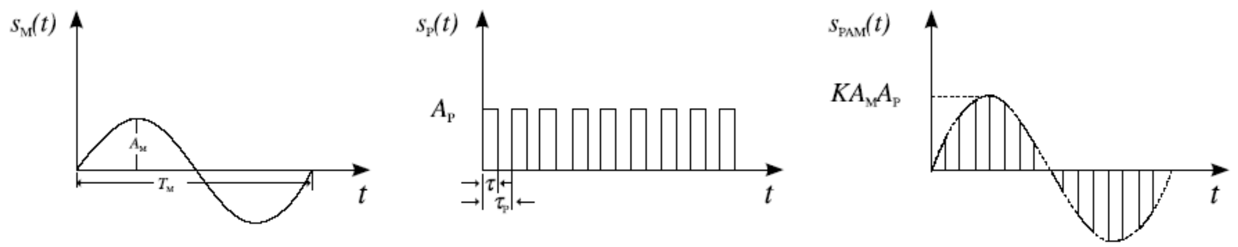
PAM means modulation by a switch. The conversion of the analog signal into a digital signal requires three steps:

- 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 $s_M(t)$.



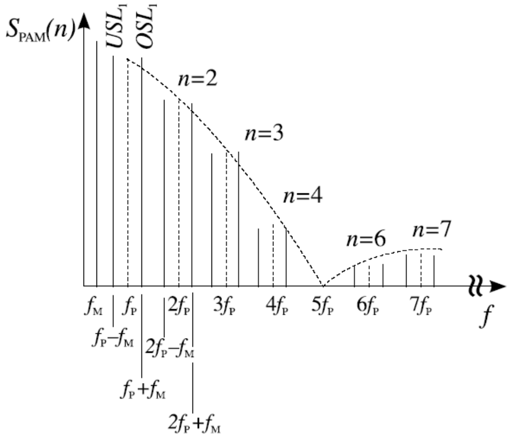
Generation of PAM

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 sM(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:

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

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**:

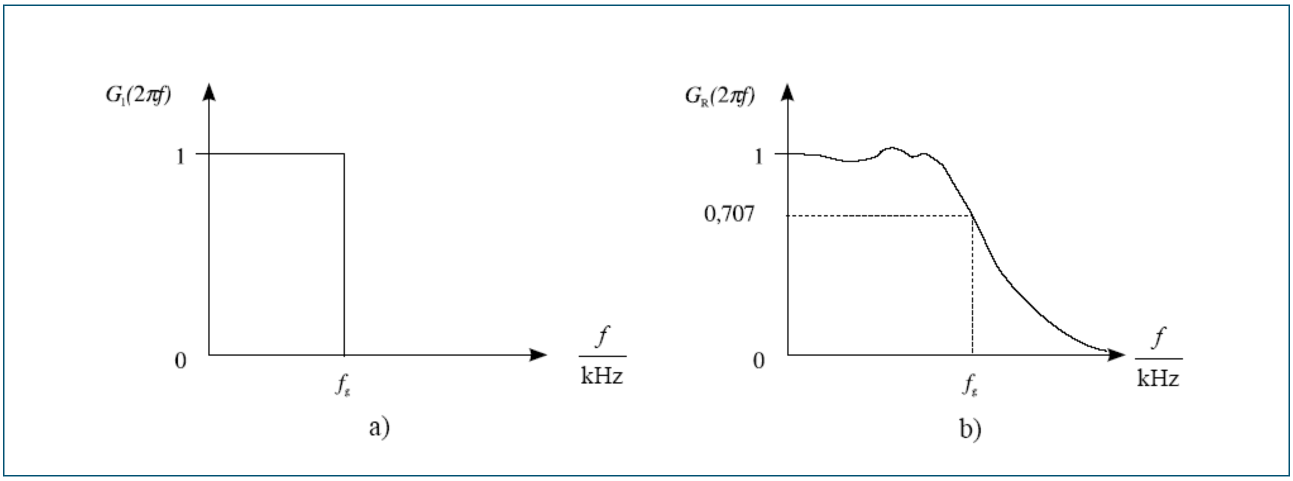
- Each time function limited to a particular bandwidth is specifically determined by its discrete sampling values if the sampling process supplies at least two samples per oscillation at the maximum occurring signal frequency.

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.

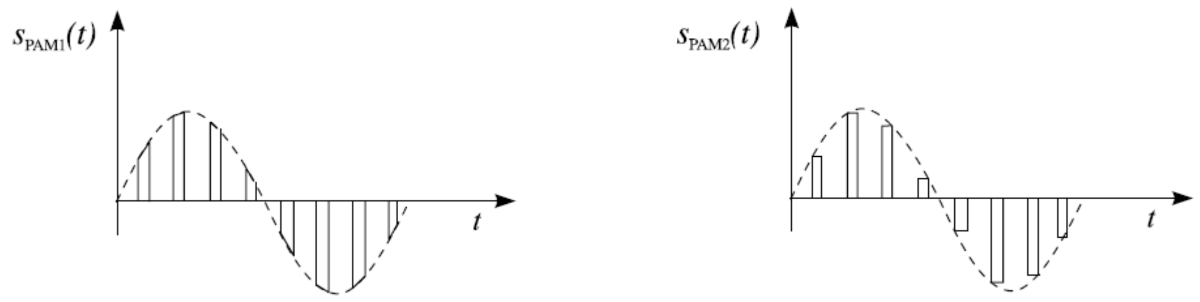


Amplitude response of low pass filters: a = ideal, b = real

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

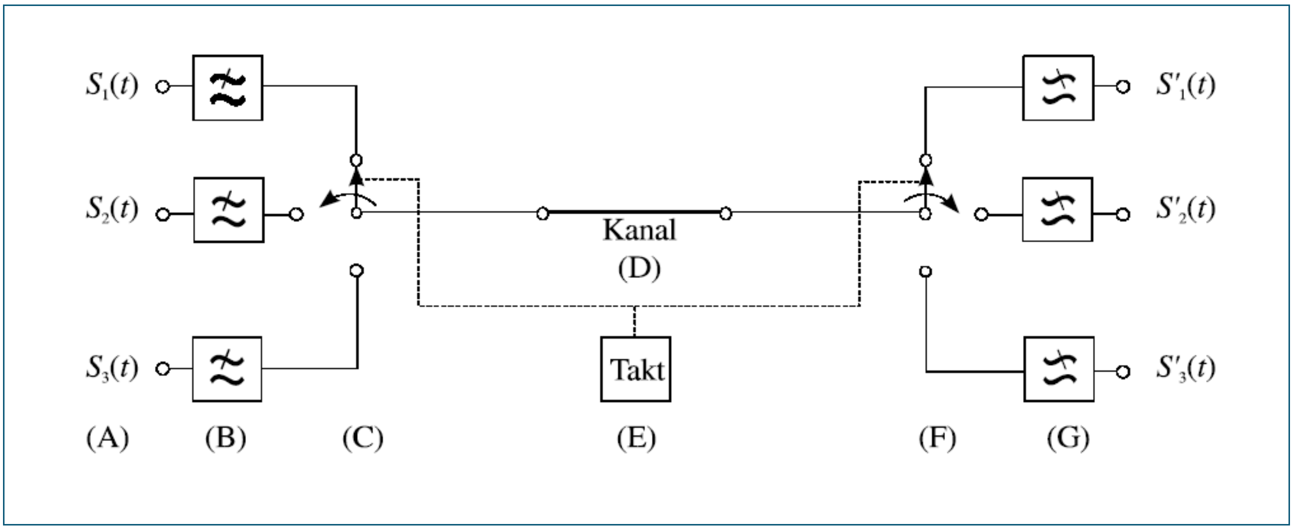
- Both PAM types contain the modulating signal $s_M(t) = A_M \cos(2\pi f_M t$. In PAM_1 this spectral component is only evaluated with the constant factor τ/T_P P. In the case of PAM_2 there is an additional evaluation factor $si(\pi f_M T_0)$. This brings about an additional, signal frequency dependent 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.



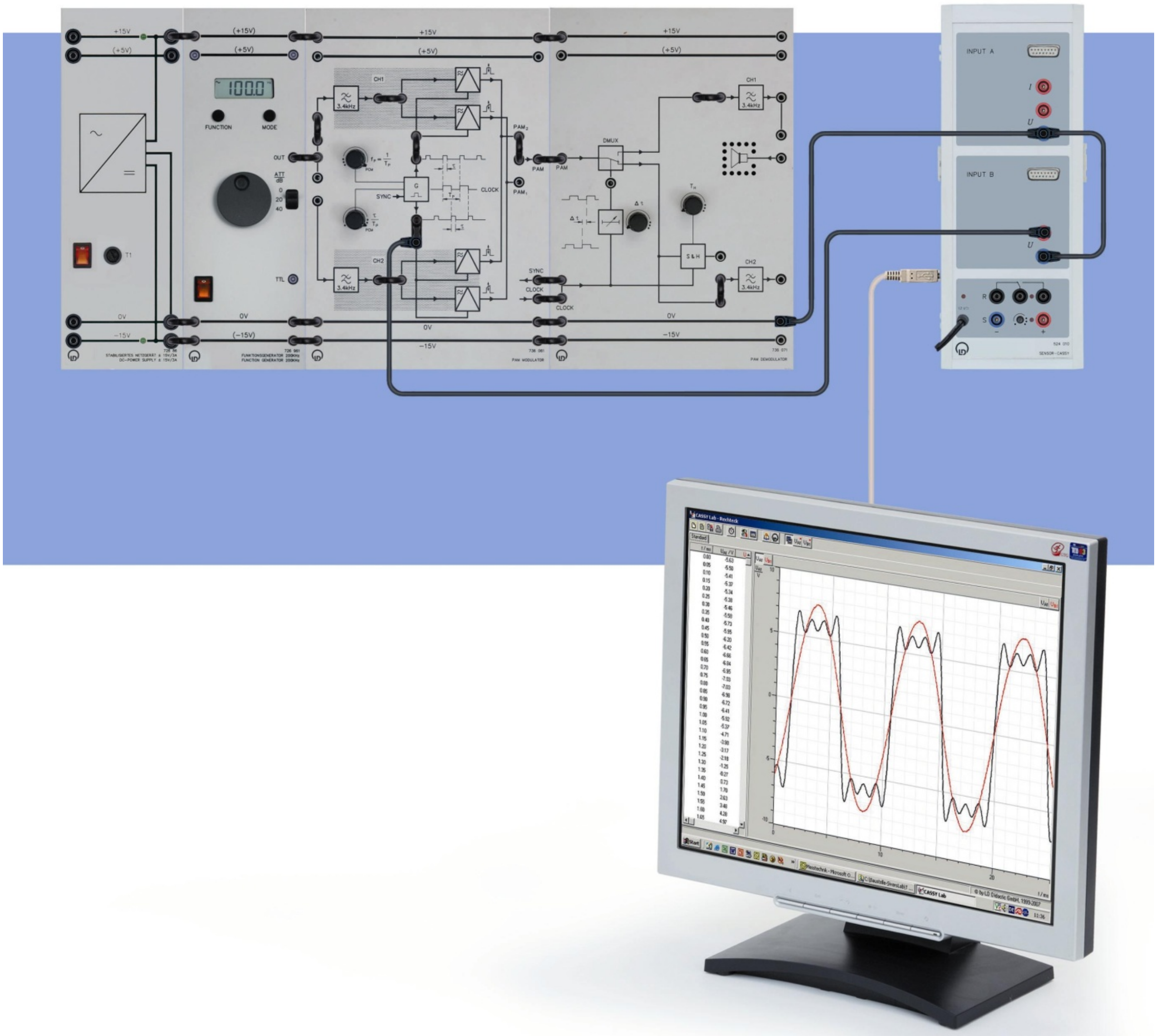
Principle of TDM with PAM
A: Signal sources
B: Band limiting filter
C: Multiplexer
D: Transmission channel
E: Synchronization clock
F: Demultiplexer
G: Low pass demodulator

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*.
- Load the CASSY Lab 2 example pulse frequency5000.labx.
- 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.
- Load the CASSY Lab 2 example PAMTimeInOut.labx.
- 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.
- Load the CASSY Lab 2 example PAMTime.labx.
- 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:
  CASSY UB1 $\rightarrow$ Clock generator *G*.
  Load the CASSY Lab 2 example DutyCycle.labx.
  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.
- Load the CASSY Lab 2 example PAMModDem.labx.
- Start the measurement by pressing *F9.*
- Repeat the measurement for τ/T_P = 30 % and τ/T_P = 10 %.
- Sketch your results.

Variants

- Measure the input- and output signal of the channel filter CH1 for different frequencies and signal forms.
- Display the PAM signal for different duty cycles.
- Investigate the function of the hold stage at the PAM demodulator (T_H). Measure the pulse width as a function of T_H.

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.
- Load the CASSY Lab 2 example PAMFFT.labx.
- Start the measurement by pressing *F9.*
- Repeat the measurement for f_M = 1 kHz and f_M = 2 kHz.
- Sketch your results.

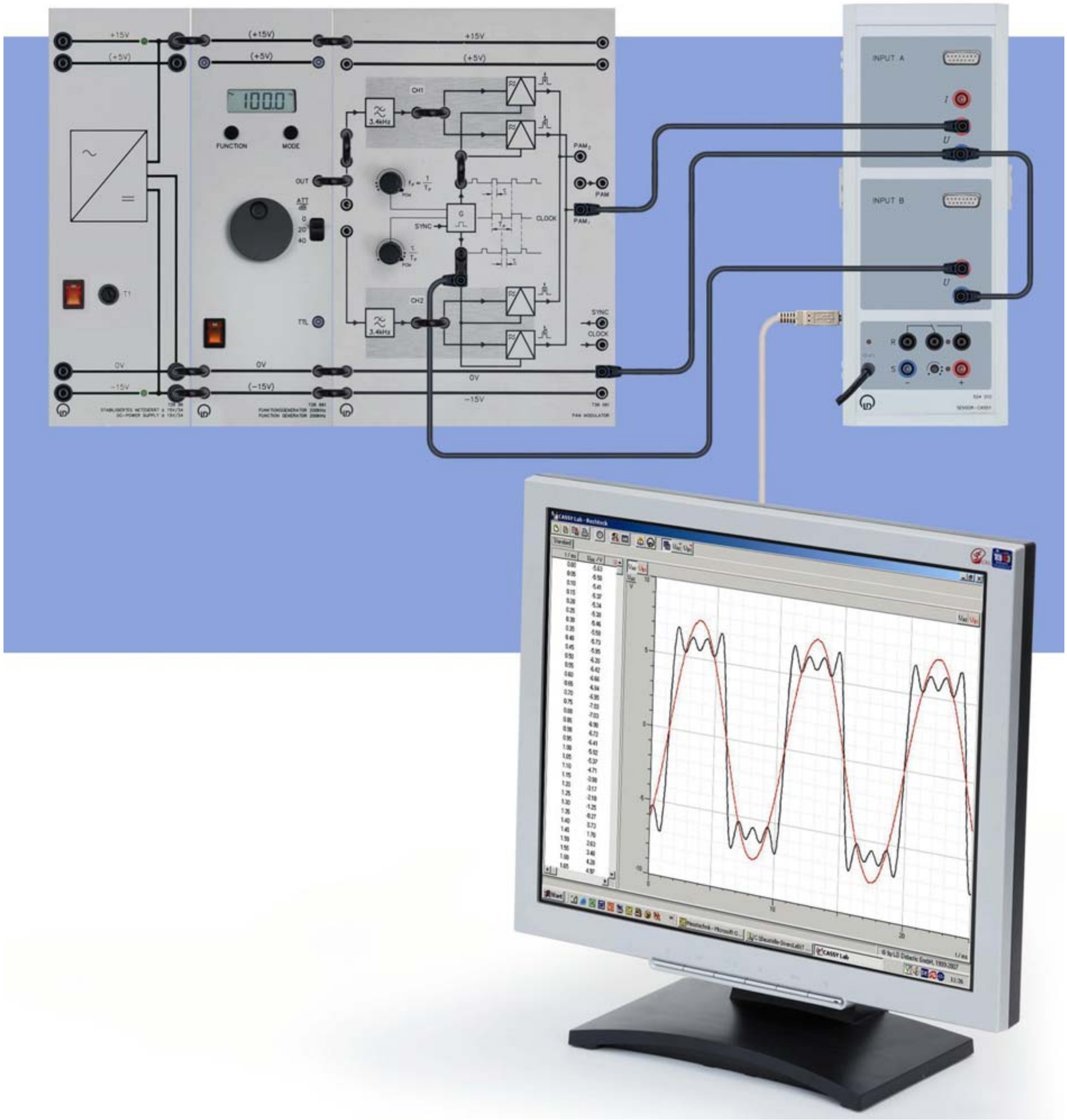
Das PAM_1-Spectrum as a function of the duty cycle.

- 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:
  Load the CASSY Lab 2 example DutyCycle.labx.
  Start the measurement by pressing *F9.*
  Set the duty cycle to τ/T = 30 %.
- Load the CASSY Lab 2 example PAMFFT.labx.
- 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.
- Load the CASSY Lab 2 example PAMFFT.labx.
- 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.

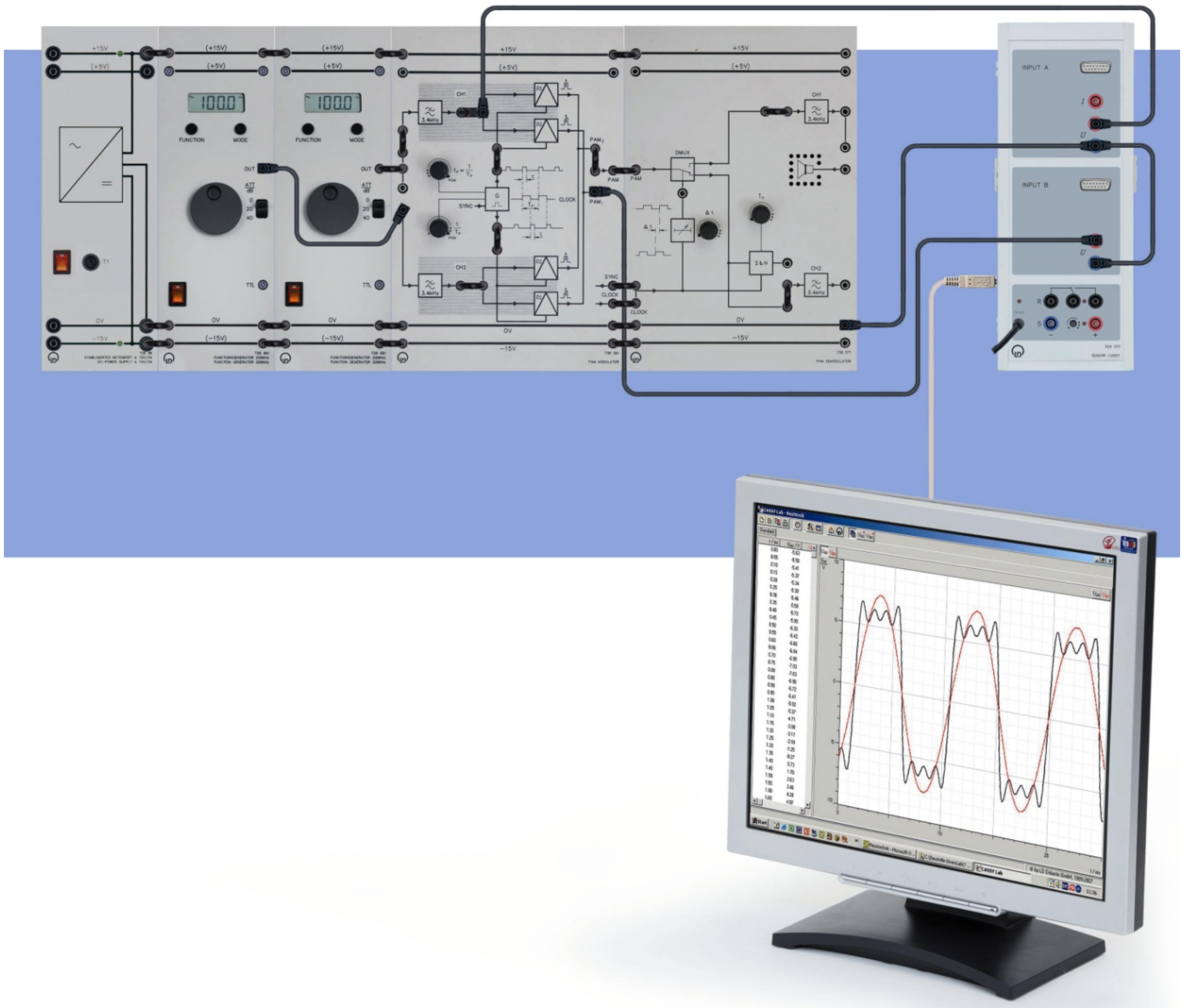
Subsampling in the frequency domain

- 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: Load the CASSY Lab 2 example pulse frequency5000.labx.
- For the setting of the duty cycle τ/T_P= 20 %: Load the CASSY Lab 2 example DutyCycle.labx.
- For the spectrum: Load the CASSY Lab 2 example PAMFFT.labx.
- 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
- Load the CASSY Lab 2 example PAMTime.labx.
- 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.

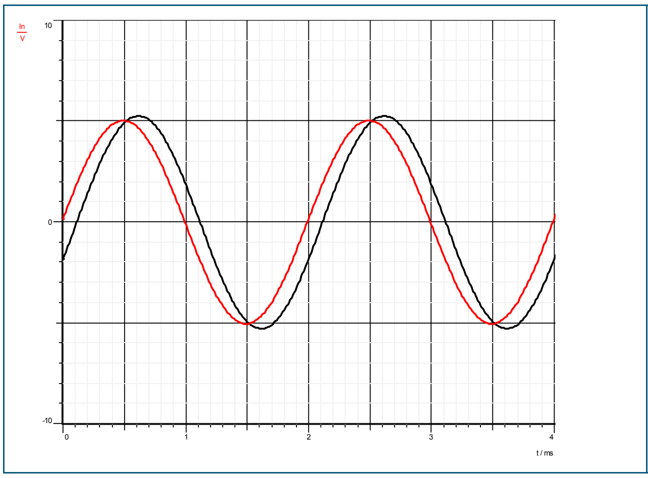
- Sampling frequency f_P = 5000 Hz, duty cycle maximal.
- Function generator 1: Triangle, f_{M1} = 200 Hz, A = 5 Vpp.
- Function generator 2: Sine, f_{M2} = 300 Hz, A = 10 Vpp.
- CASSY UA1 → Input PAM modulator channel CH1.
- CASSY UB1 → Output PAM modulator PAM_1.
- Load the CASSY Lab 2 example PAMTDMInput.labx.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

PAM demodulator time shift Δt → left/middle

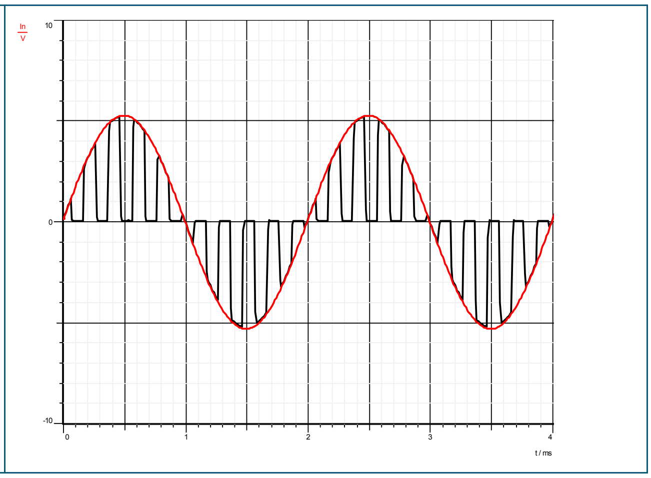
- CASSY UA1 → Output PAM demodulator channel CH1.
- CASSY UB1 → Output PAM demodulator channel CH2.
- Load the CASSY Lab 2 example PAMTDMOutput1.labx.
- Start the measurement by pressing *F9.*

Results

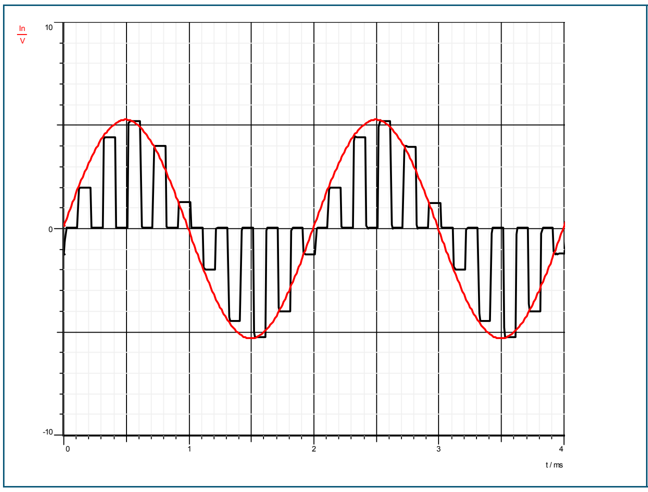
Time characteristics of the PAM



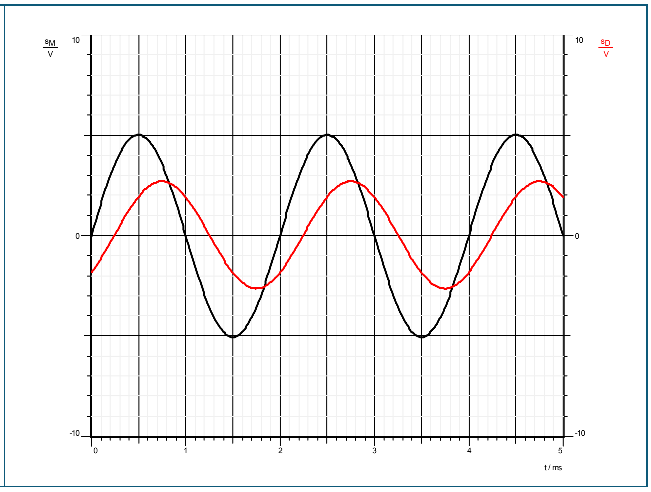
Input signal (red)- and output signal (black) of the input filter of *CH1*.



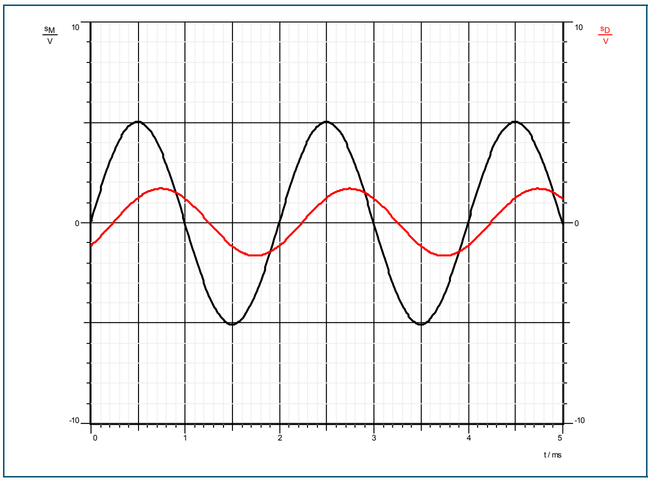
Modulating signal and *PAM₁* signal A_M = 10 Vpp, f_M= 500 Hz f_P = 5000 Hz, τ/T_P = 50%



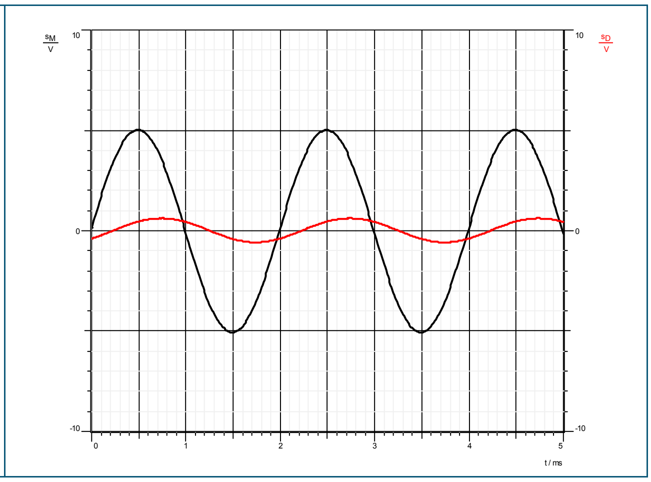
Modulating signal and *PAM₂* signal A_M = 10 Vpp, f_M= 500 Hz f_P = 5000 Hz, τ/T_P = 50%



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



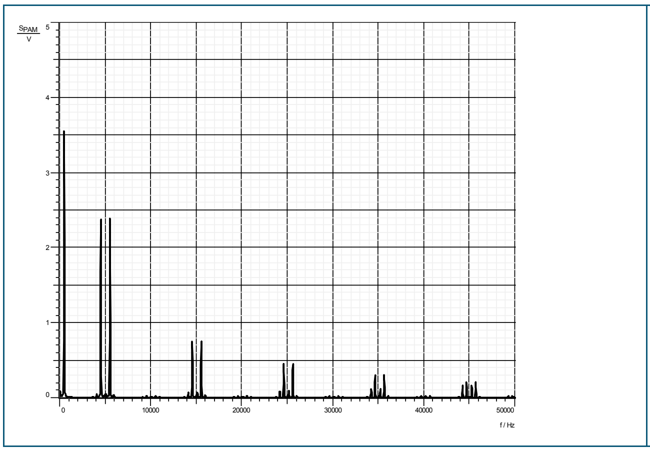
Modulating signal $s_M(t)$ and demodulated signal $s_D(t)$ for τ/T_P = 30%



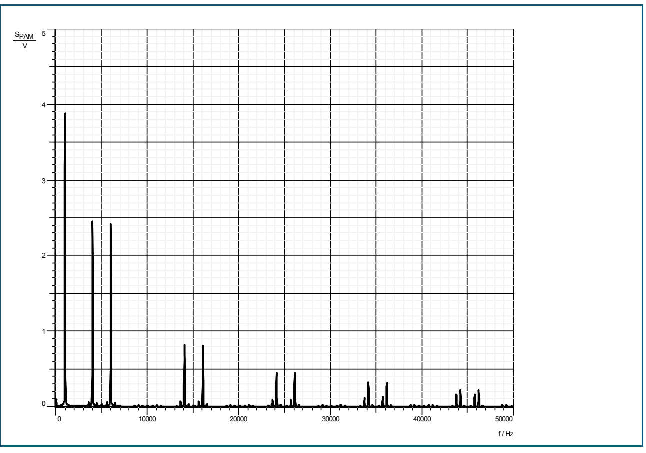
Modulating signal $s_M(t)$ and demodulated signal $s_D(t)$ for τ/T_P = 10%.

Spectra of the PAM

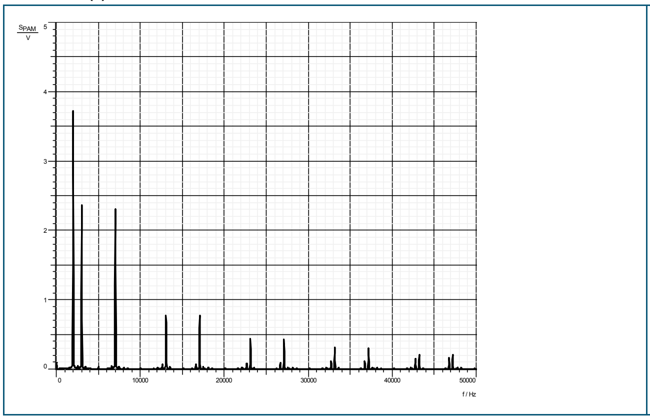
PAM_1



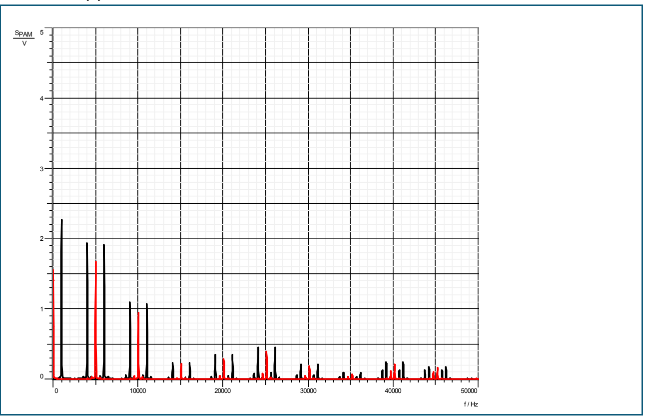
$s_P(t)$: A_P = 5 V τ/T_P = 50% f_P = 5000 Hz
$s_M(t)$: A_M = 5 V f_M = 500 Hz



$s_P(t)$: 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

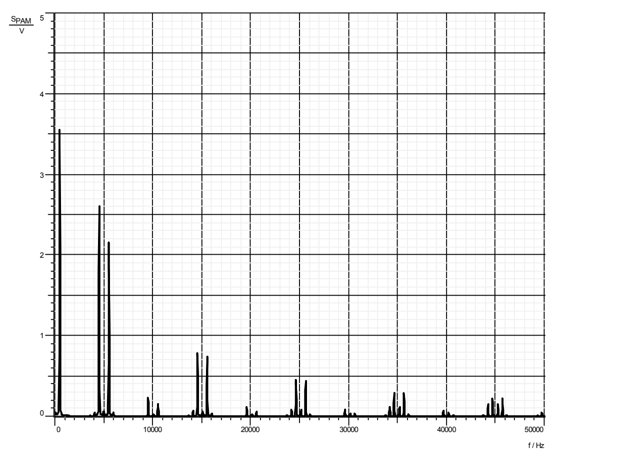
Summary

- The calculation of the PAM spectra makes use of:

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

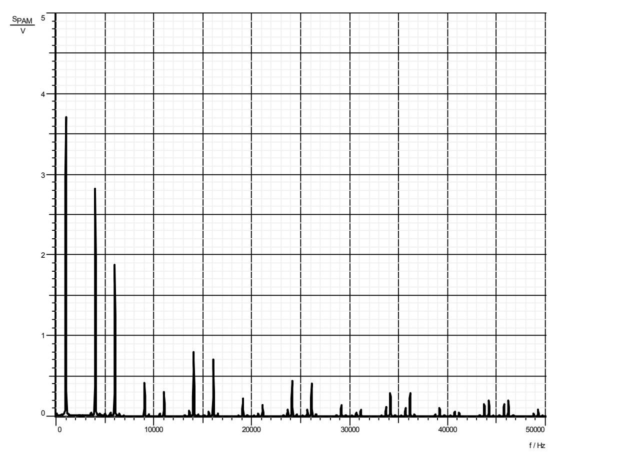
- The spectral amplitudes determined apply respectively for the **upper** and **lower sidelines** (*USL*n and *LSL*n).
- A double line appears with bipolar PAM instead of an individual carrier line. The frequency interval of the sidelines of the suppressed carrier is equal to the signal frequency f_M.
- 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**

PAM₂



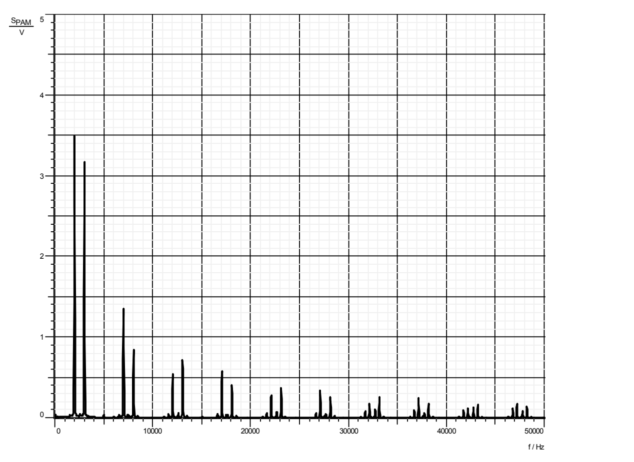
$s_P(t)$: A_P = 5 V τ/T_P = 50% f_P = 5000 Hz

$s_M(t)$: 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

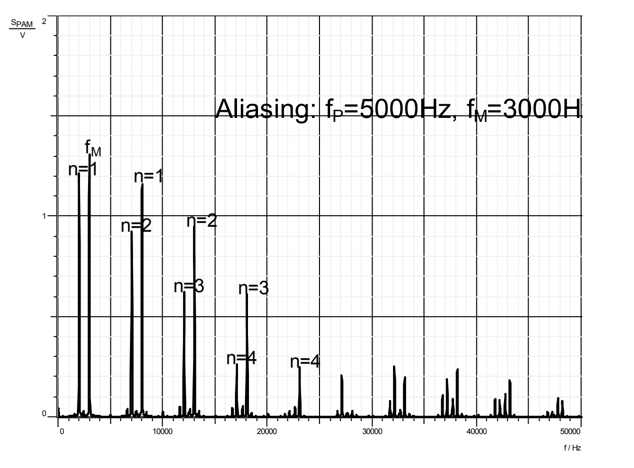
$s_M(t)$: A_M = 5 V f_M = 2000 Hz

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.

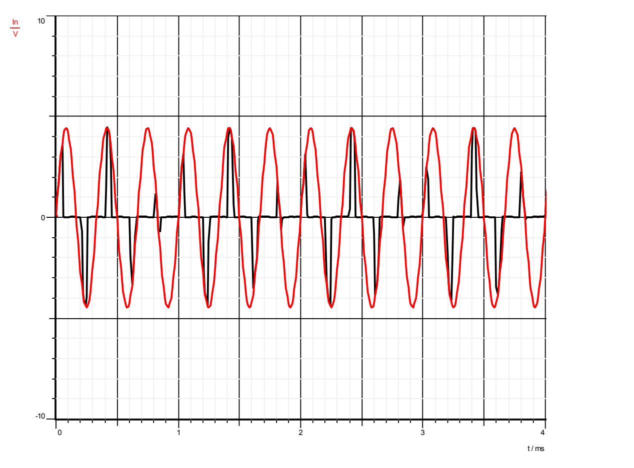
Displaying aliasing



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

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

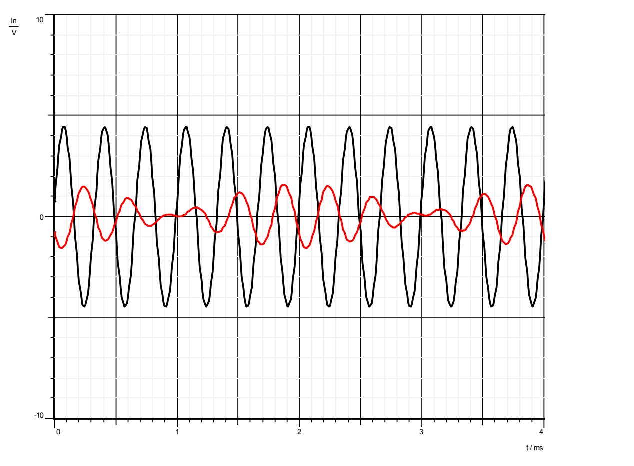
PAM spectrum for subsampling



$s_P(t)$: 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



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

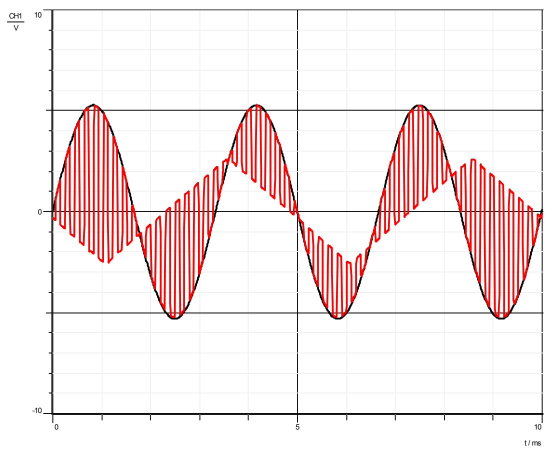
$s_M(t)$: A_M = 5 V f_M = 3000 Hz

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

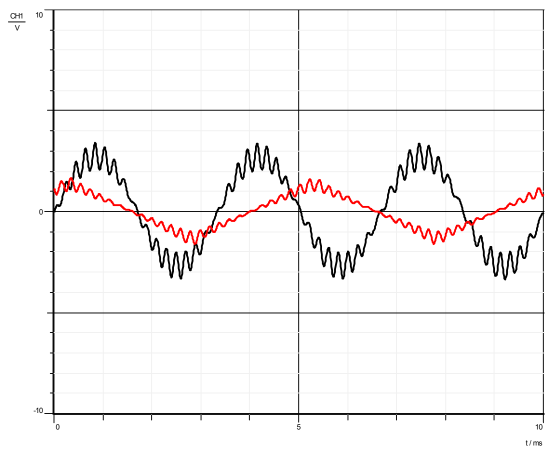


$s_P(t)$: A_P = 5 V τ/T_P = 20% f_P = 5000 Hz

$s_M(t)$: A_M = 5 V f_{M1} = 200 Hz f_{M2} = 300 Hz

PAM time multiplex input

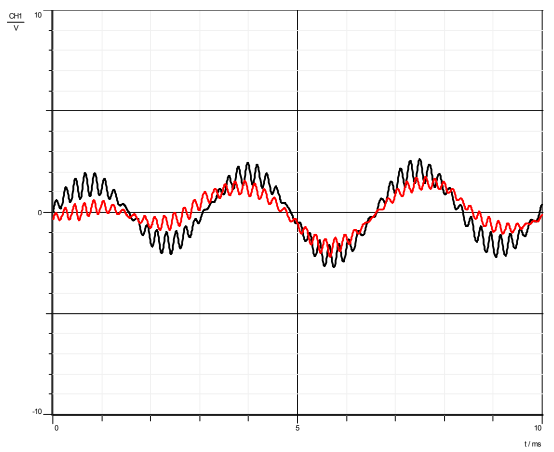
The envelopes of both channels are limiting the time multiplex signal



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

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

Demod. signals at the output of the demultiplexer
time shift $\Delta t \rightarrow$ left



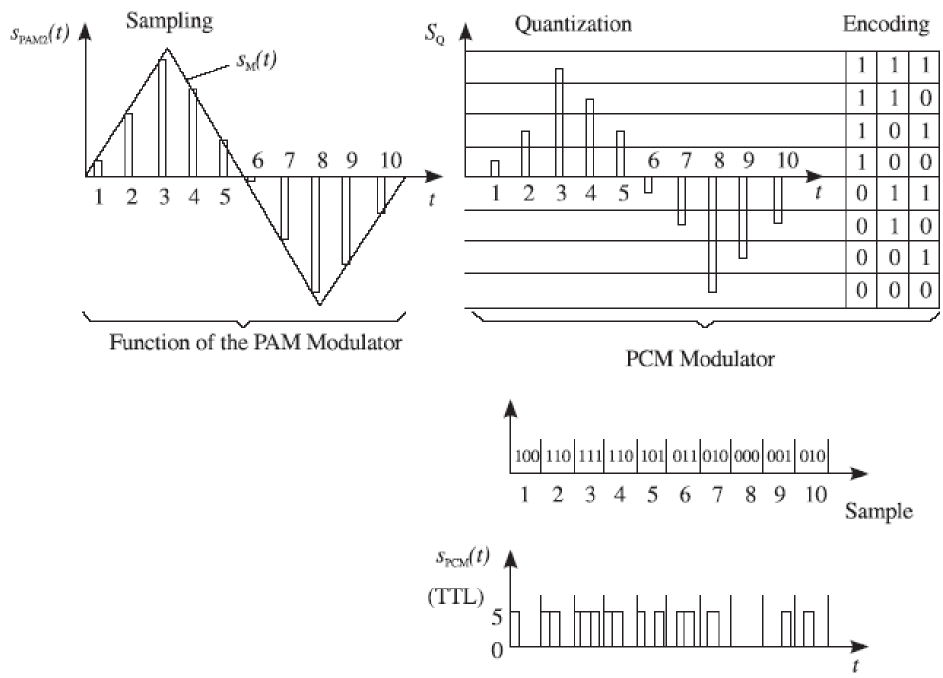
$s_{M1}(t)$: A_{M1} = 5 V f_{M1} = 200 Hz

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

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

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

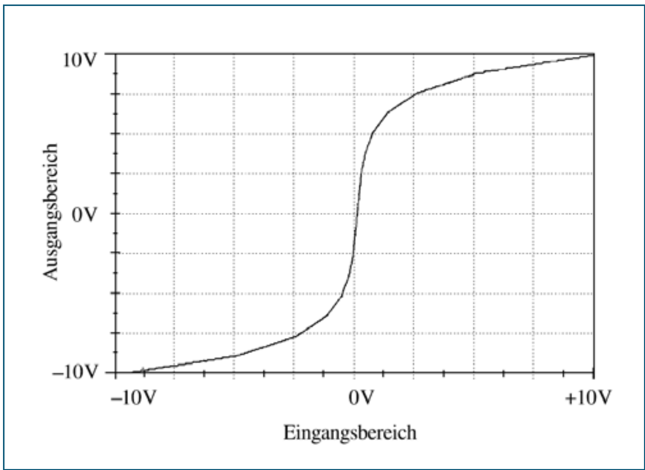
Evidently a sampled, discrete value signal is unsuitable for transmission via a noisy transmission channel. The digital signal at the output of the quantifier requires a shape better suited to the relationships prevailing on the channel. The conversion process required for this is called encoding. Coding 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

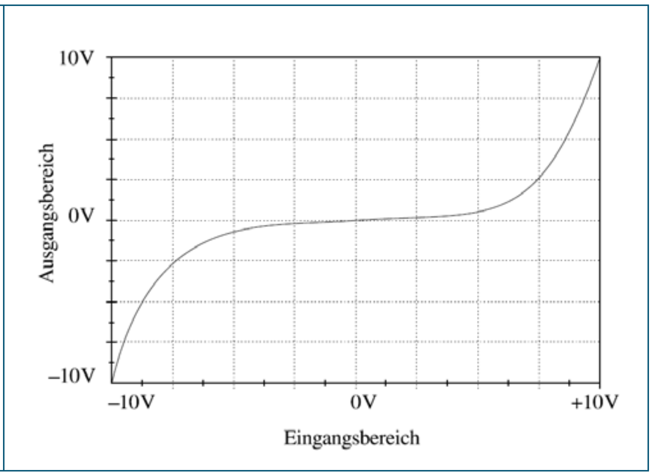
The dual code is called multi-stepped, because in the transition from one binary character to the next, it is possible that several bits have to be converted at the same time (see for example the transition from 3 to 4) in the decimal system. The sampling of the modulating signal $s_M(t)$ in the PAM 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

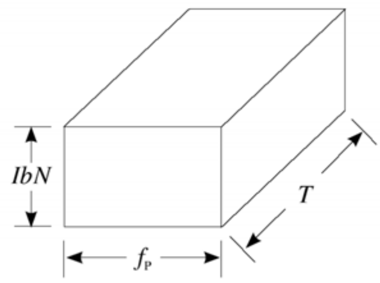


Expander characteristic

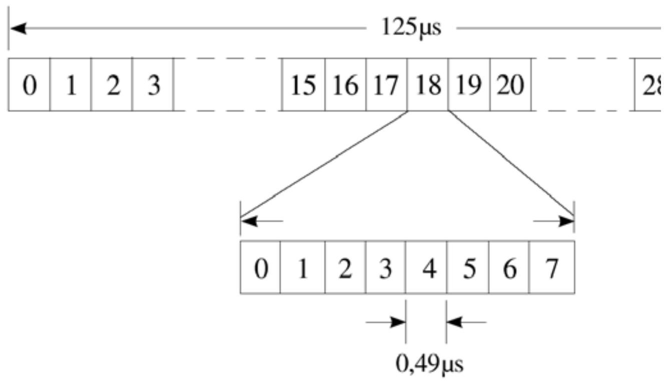
Time division multiplex with PCM

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.



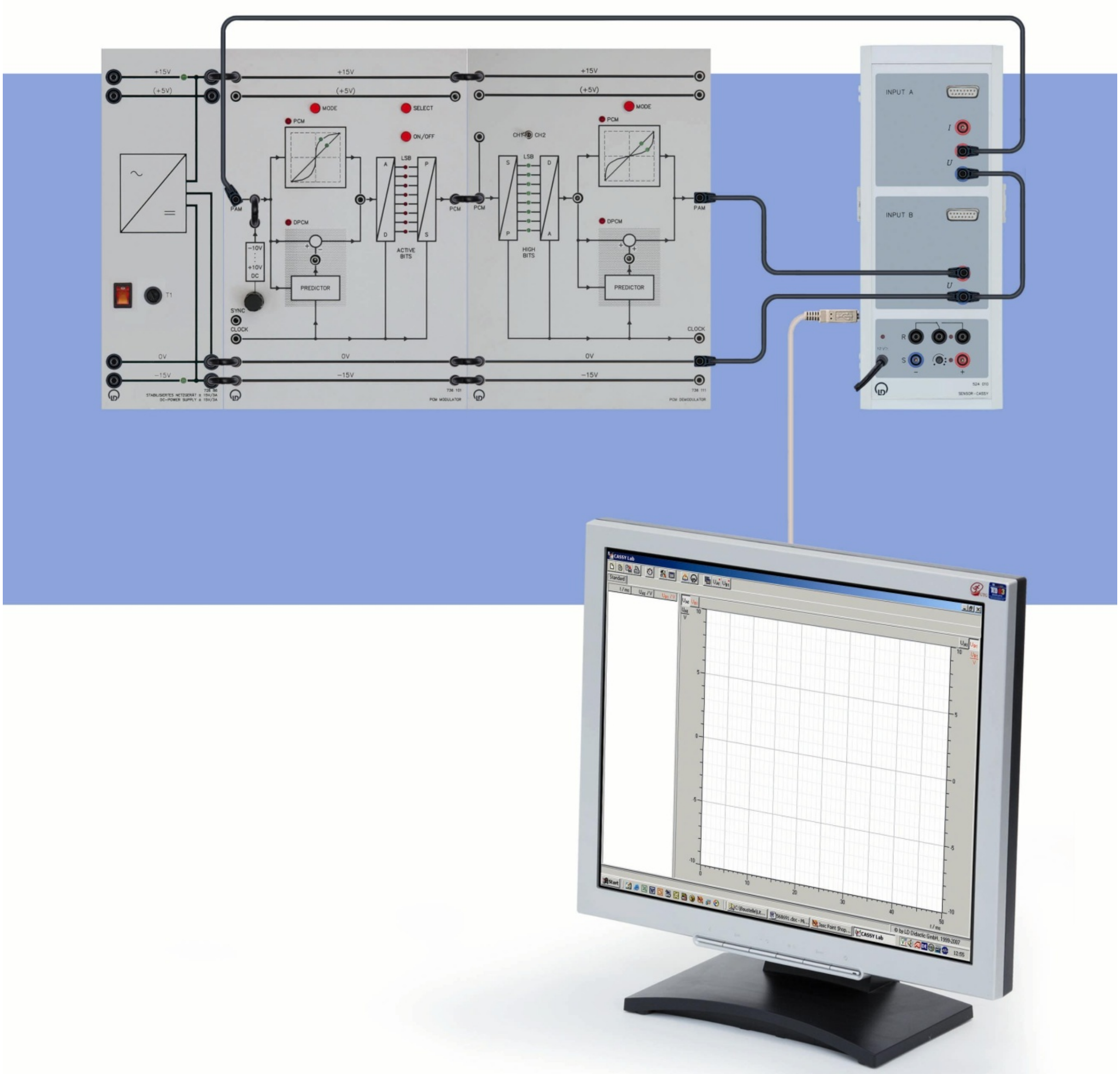
The pulse frame of the PCM 30/32

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.

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.
- Load the CASSY Lab 2 example Quant.labx.
- 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.

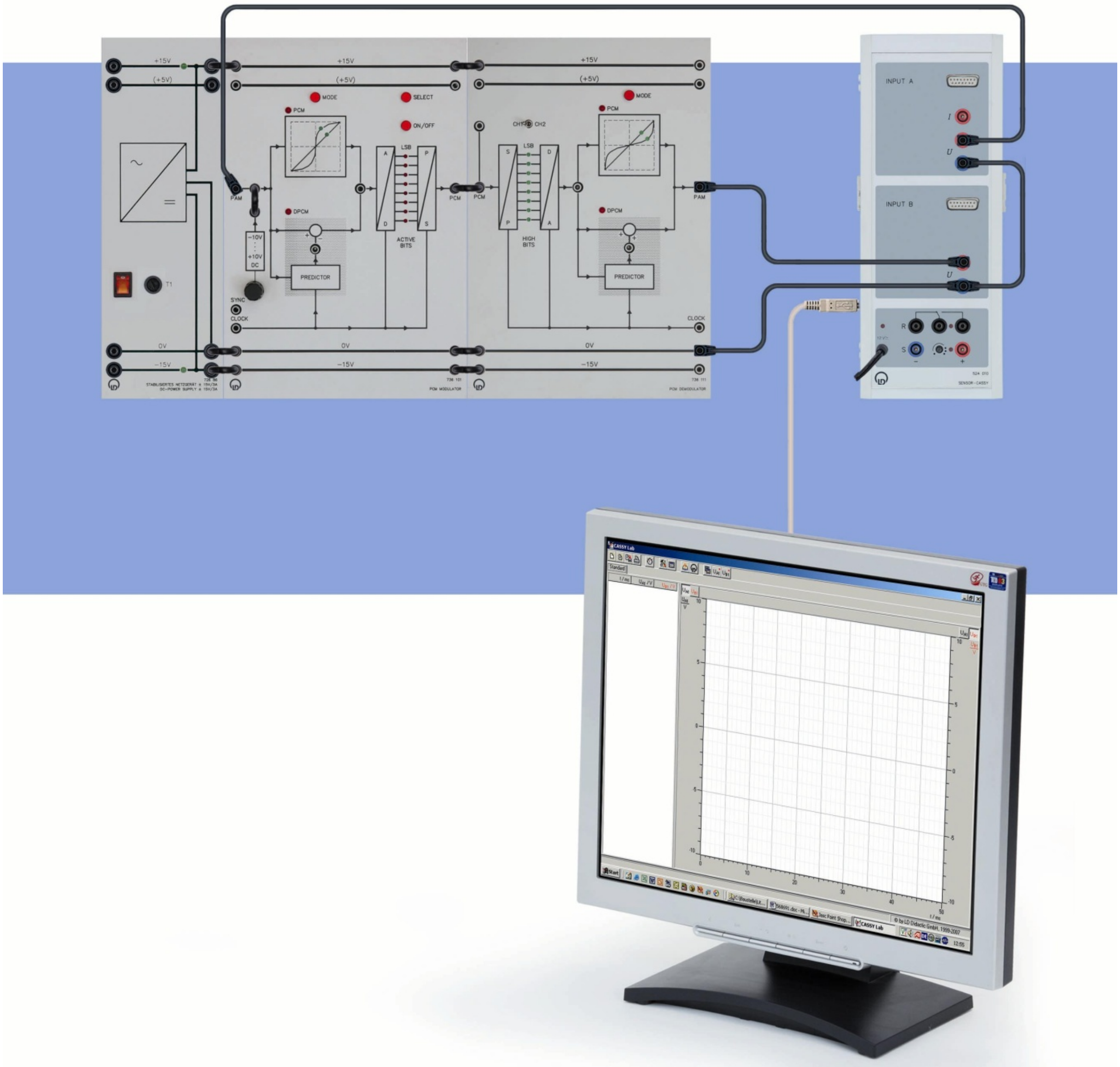
Compressor / Expander characteristic

- 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.

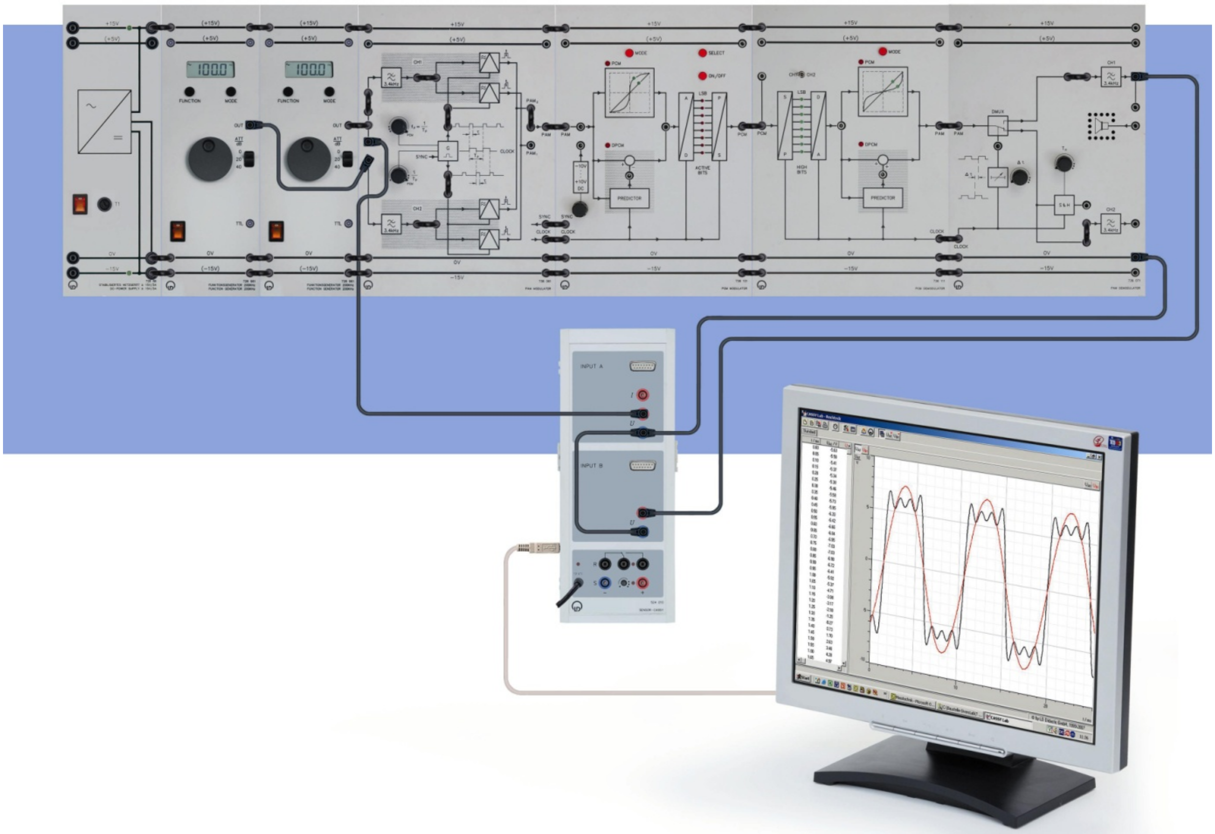
Encoding



- 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.
- Load the CASSY Lab 2 example Code.labx.
- 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	LSB MSB
-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		

PCM transmission

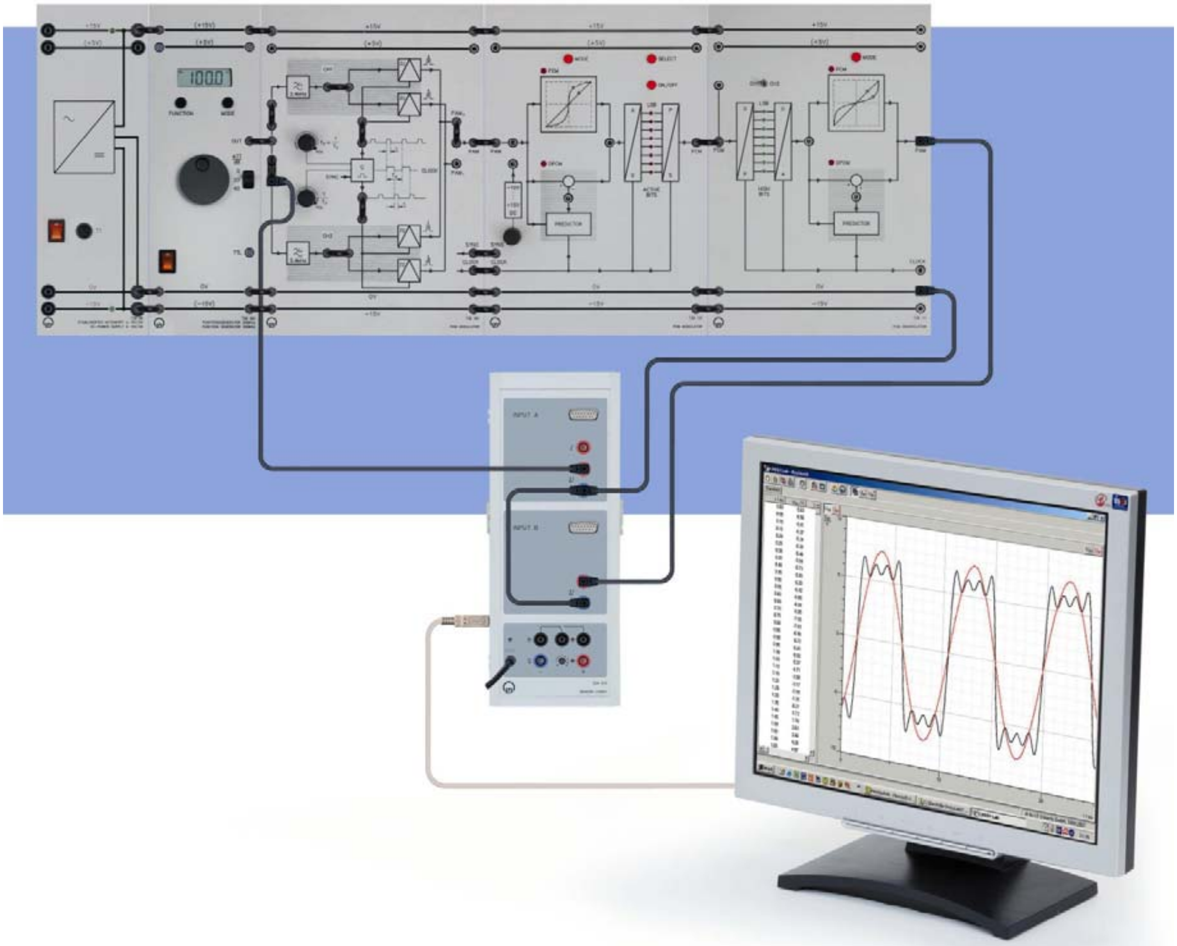


- Set up the shown experiment and switch on the power supply.
- By pressing several times MODE set the PCM modulator and PCM demodulator to: PCM, linear quantization (watch the allocated LEDs).
- Activate all bits. For that press the button SELECT, until all (red) LEDs in the PCM modulator indicate ACTIVE.
- 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
- Load the CASSY Lab 2 example PCMTrans.labx.
- Start the measurement by pressing *F9.*

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

Quantization noise



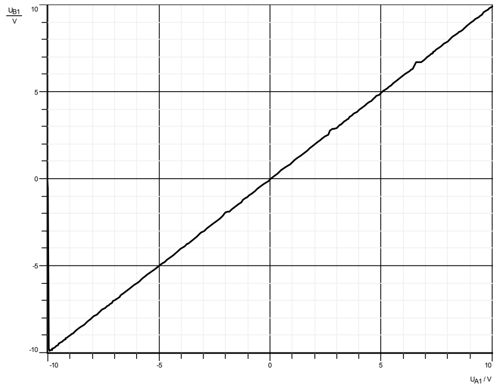
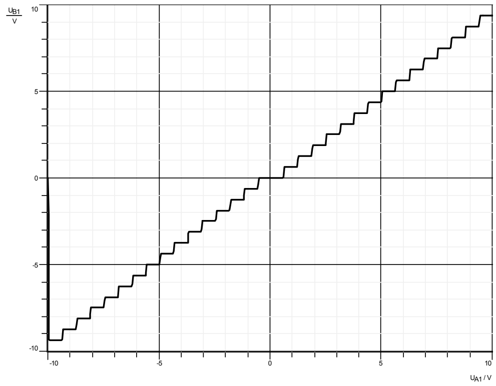
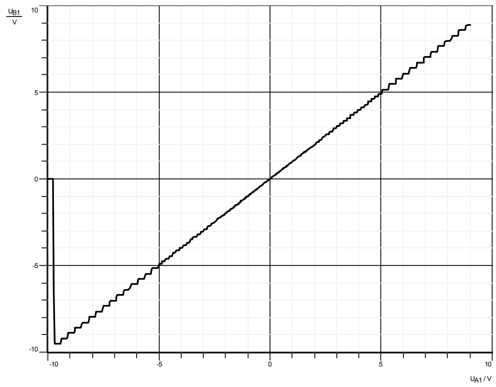
- Set up the shown experiment and switch on the power supply. Connect both channels (*CH1* and *CH2*) of the PAM Modulator (736 061) with the function generator. This avoids time gaps at the output of the PCM demodulator.
- Function generator: Triangle, 30 Hz, 12 V_{pp}
- By pressing several times MODE 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.
- Load the CASSY Lab 2 example QNoise.labx.
- 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.

Variants

- Use a sinusoidal modulating signal. This results in a more complicated structure of the quantization noise.
- Record the quantization noise even for non-linear quantization.

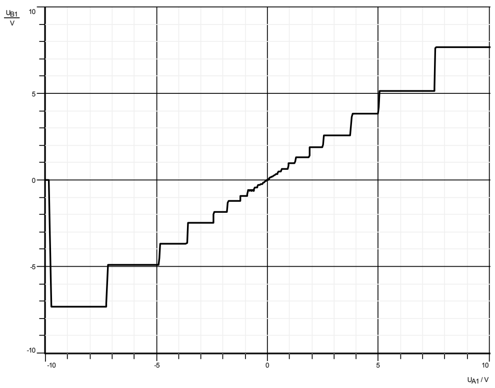
Results

Quantization

Resolution: 8 Bit Linear quantization **Interpretation:** with 8 bit 2^8 = 256 amplitudes can be coded. Assumed a linear quantization the dynamic range is: 256∗80 mV = 20,4 V approx ±10 V.	
Resolution: 5 bit Linear quantization **Interpretation:** 5 bit clearly result in 32 steps of equal height.	
Resolution: 8 bit Non-linear quantization	

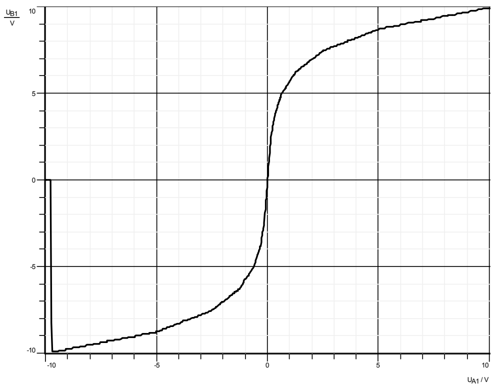
Resolution: 5 bit
Non-linear quantization

Interpretation:
Non-linear quantization makes use of quantization steps of different height. With 5 bit resolution the quantization is fine near the zero point. It becomes more coarse for increasing amplitudes.

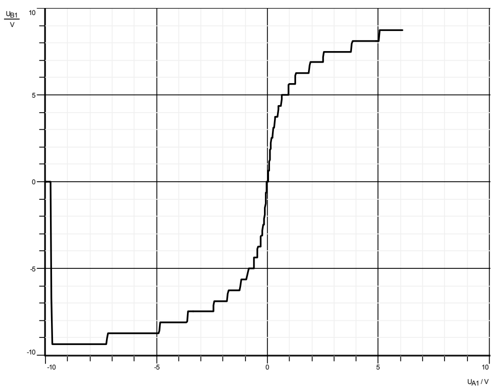


Resolution: 8 bit
Compressor characteristic

Interpretation:
The slope of the characteristic is maximum near the zero point. This gives an increase in amplification for small amplitudes. For strong In put voltages the slope is reduced resulting in an attenuation of the amplitude values.

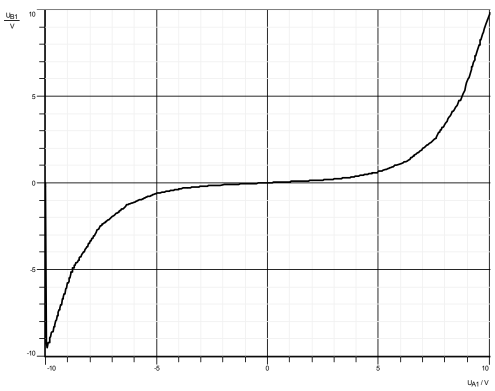


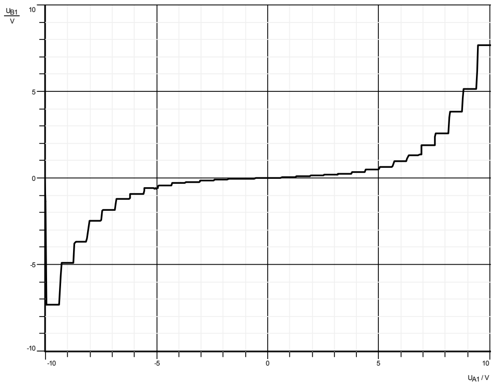
Resolution: 5 bit
Compressor characteristic



Resolution: 8 bit
Expander characteristic

Interpretation:
The expander characteristic shows the inverse curve. Here the slope of the function is minimal around the zero point, the values increased in the compressor are lowered again. In the range of the higher amplitudes the slope is considerable, here the values reduced in the compressor are increased again.



Resolution: 5 bit Expander characteristic	

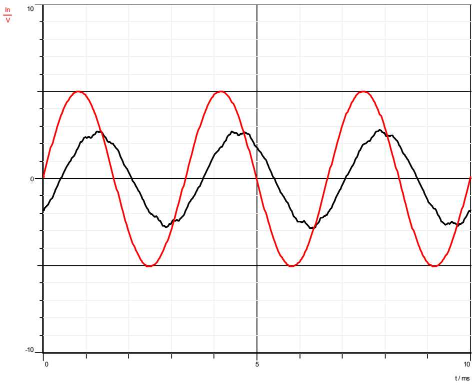
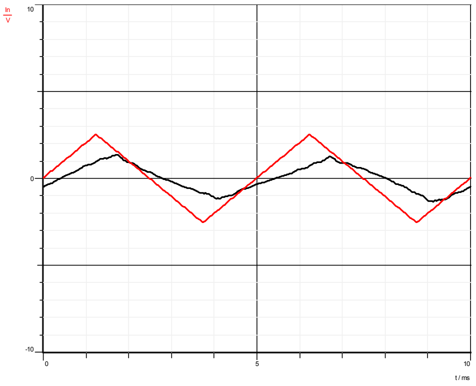
Encoding

Coding protects particularly small amplitude values. These are represented by the less significant bits in the PCM word. However, they are just as safe from disturbance as the more significant bits as representatives of the larger signal values. Known binary codes are:

- Dual code
- Symmetrical binary code
- Gray code.

UA1/V	UB1/V	LSB							MSB
-9.96	-9.64	1	1	1	1	1	1	1	0
-9.00	-8.98	1	1	0	0	1	1	1	0
-8.00	-7.96	0	1	1	0	0	1	1	0
-7.00	-7.03	0	1	0	1	1	0	1	0
-6.00	-6.01	1	0	1	1	0	0	1	0
-5.00	-5.00	0	0	0	0	0	0	1	0
-4.00	-4.03	1	1	1	0	1	1	0	0
-3.00	-3.04	1	1	1	0	0	1	0	0
-2.00	-2.03	0	1	0	1	1	0	0	0
-1.00	-1.01	1	0	1	1	0	0	0	0
0.00	-0.07	1	0	0	0	0	0	0	0
1.00	0.94	0	0	1	1	0	0	0	1
2.00	1.96	1	0	0	1	1	0	0	1
3.00	2.94	1	1	1	0	0	1	0	1
4.00	3.91	0	1	0	0	1	1	0	1
5.00	4.92	1	1	1	1	1	1	0	1
6.00	5.93	0	0	1	1	0	0	1	1
7.00	6.95	1	0	0	1	1	0	1	1
8.00	7.88	1	0	1	0	0	1	1	1
9.00	8.90	0	1	0	0	1	1	1	1
10.00	9.91	1	1	1	1	1	1	1	1

PCM transmission

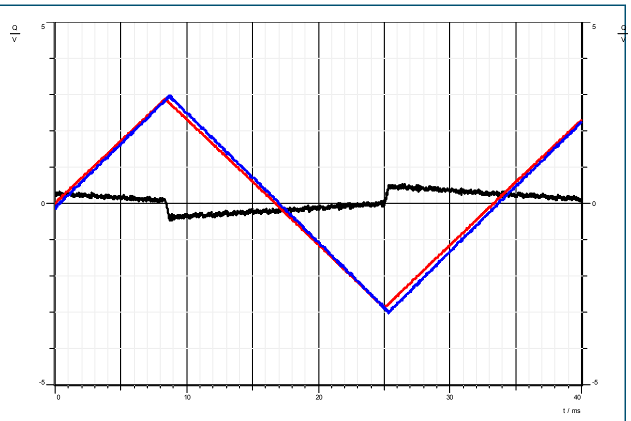
Resolution: 8 bit PCM transmission Function generator 1: Sine, f_{M1} = 300 Hz, A = 10 Vpp.	
Resolution: 8 bit PCM transmission Function generator 2: Triangle, f_{M2} = 200 Hz, A = 5 Vpp.	

Quantization noise

Resolution 8 bit
Triangle 12 V_{pp} → 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

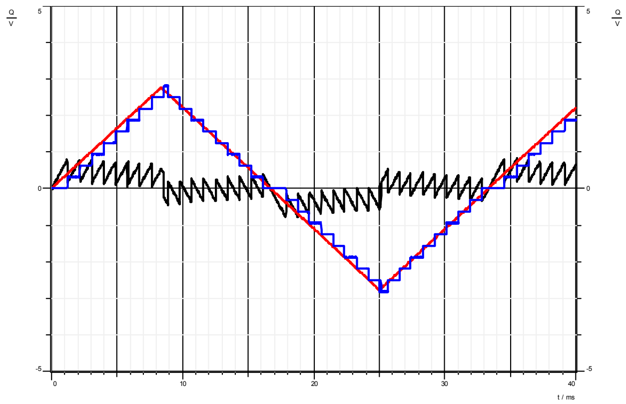
Interpretation
Only a small rectangular quantization noise is visible. Positive during ascending slope. Negative during descending slope.



Resolution 5 bit
Triangle: 12 V_{pp} → 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

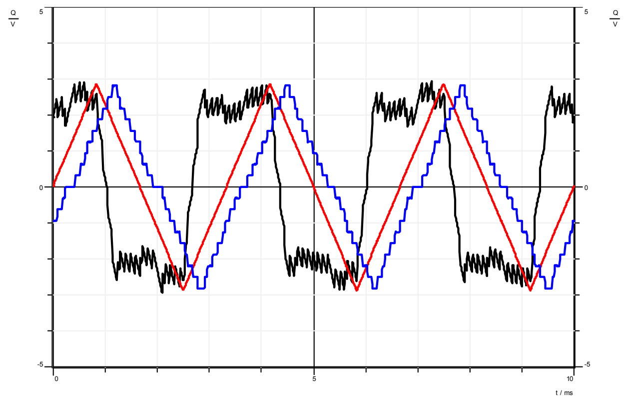
Interpretation
More coarse quantization For f_M = 30 Hz the phase shift between the signal from the function generator and the output *PAM* of the PCM demodulators can be neglected.



Influence of phase shift
Resolution: 5 bit
Triangle: 12 V_{pp} → 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

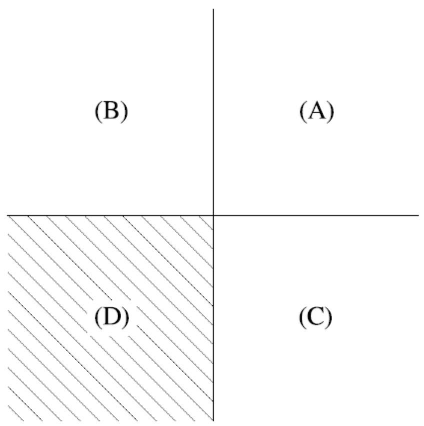
Interpretation
The phase shift and the quantization result in a strong quantization noise.



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.

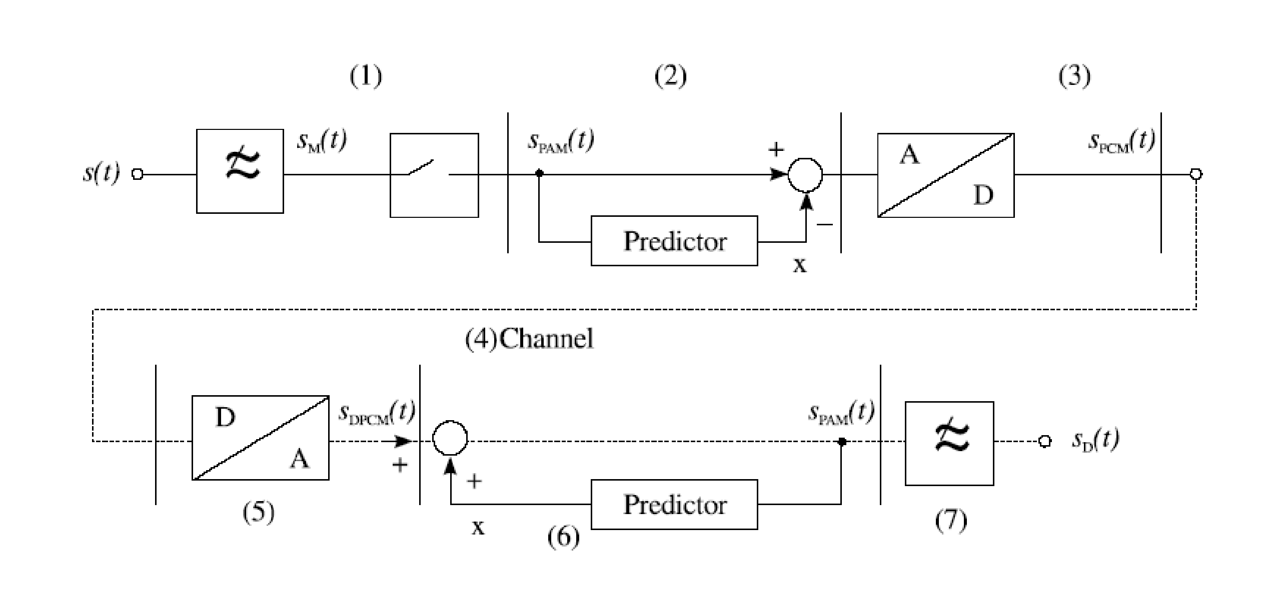


The message plane

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.

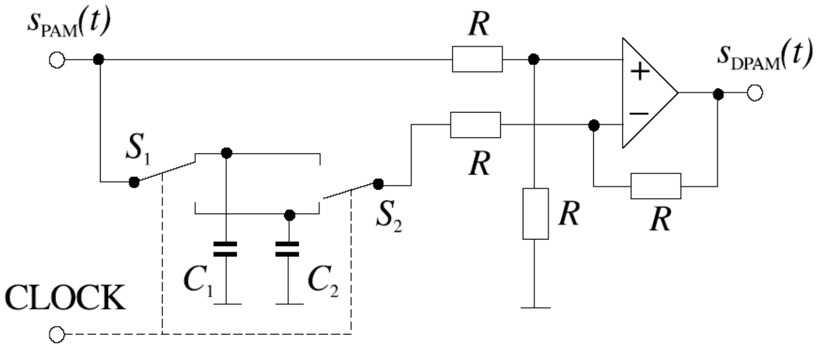
The DPCM constitutes a method of reducing the redundancy of the information signal. The figure demonstrates one possible operating principle for a DPCM link. The quantization is no longer carried out for each individual PAM value. Only the difference between the current PAM value and a predicted value *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.

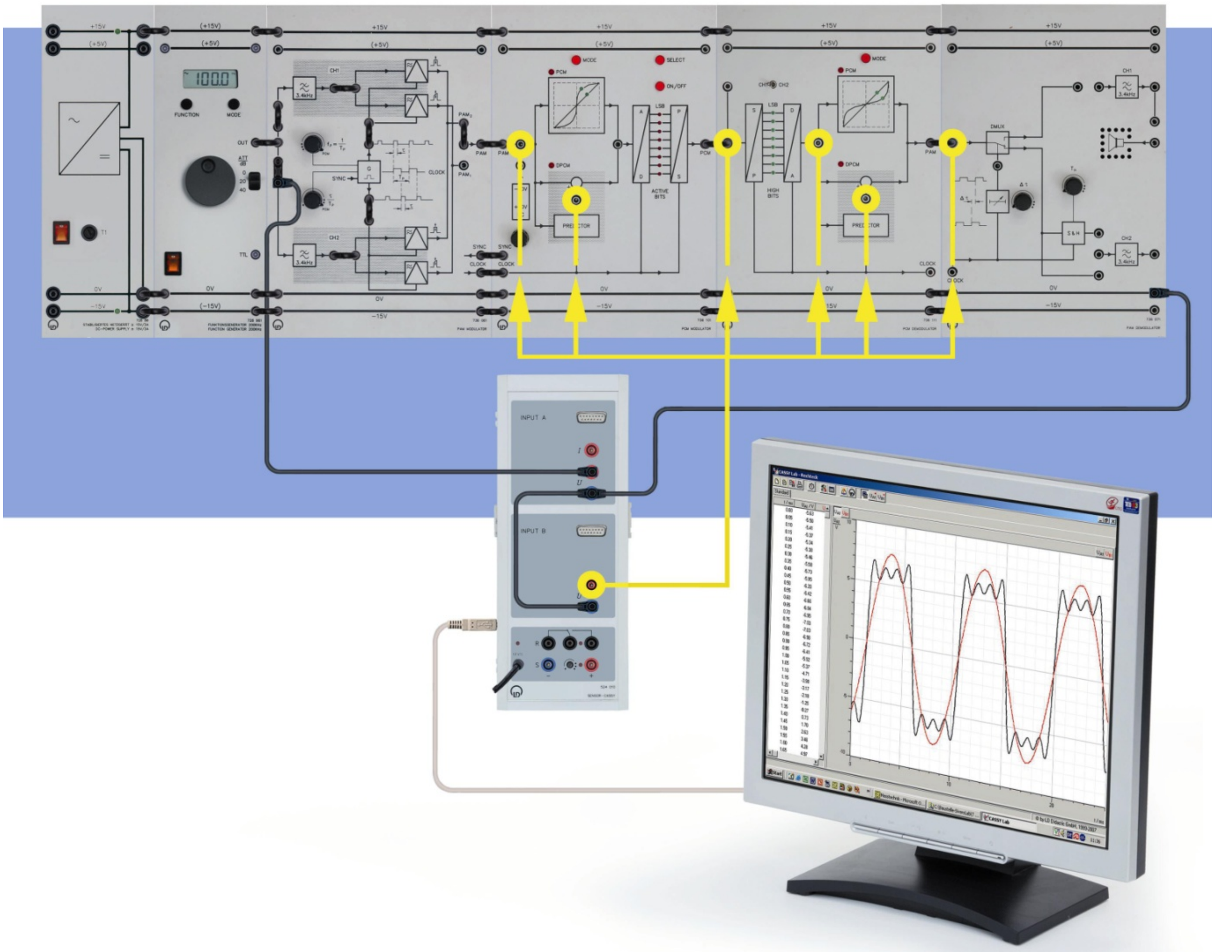


The DPCM modulator contains a differential amplifier and an analog memory. The switches S_1 and S_2 synchronized at the sampling frequency f_P operate in counter sense. Together with storage capacitors C_1 and C_2 they form the PREDICTOR. The formation of the prediction value is simple. The PAM pulses 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

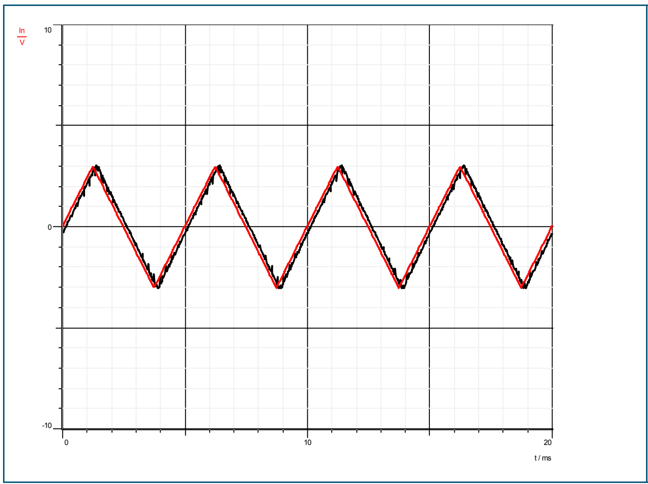
- Load the CASSY Lab 2 example DPCM.labx.
- Start the measurement by pressing *F9.*
- Connect the channel UA1 of the CASSY with the input signal of the PAM modulator. With channel UB1 of the CASSY record successively the following signals:
  PAM input
  Predictor of the DPCM modulator
  Output of the 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.
- Deactivate the following bits

LSB							MSB
ON	ON	ON	--	--	--	--	ON

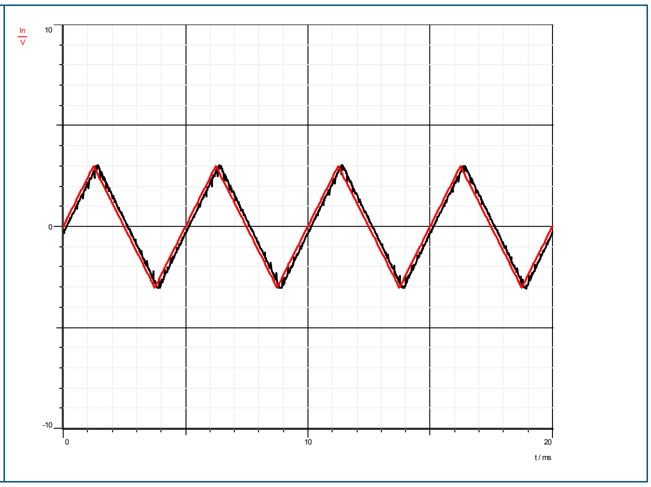
- Repeat the experiment for a resolution of 4 bit.

Results

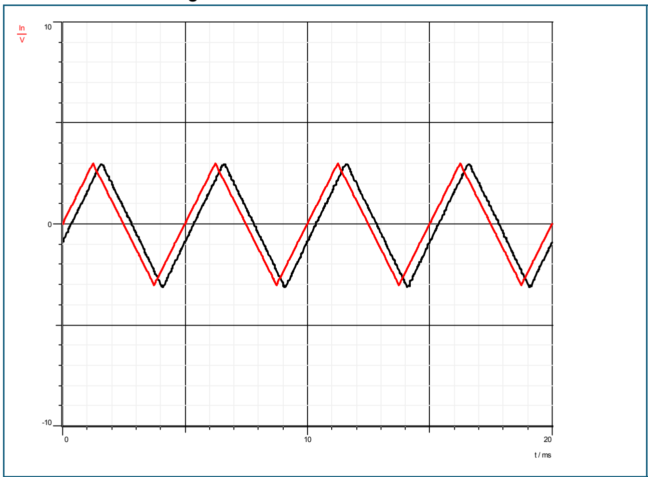
DPCM-Modulator



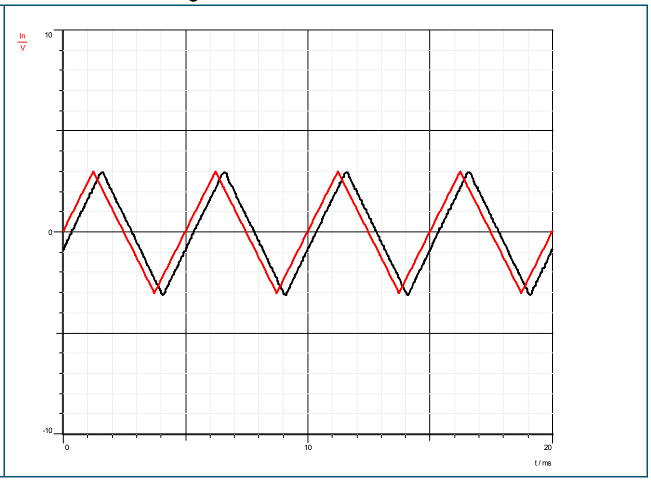
Red: Triangular signal from function generator
Black: PAM-Signal 8 bit



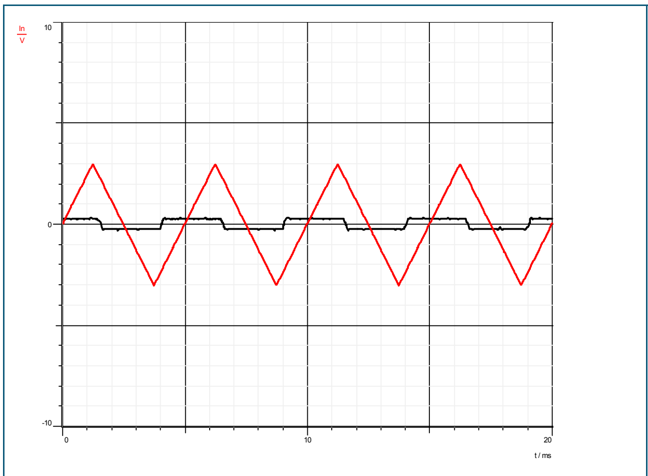
Red: Triangular signal from function generator
Black: PAM-Signal 4 bit



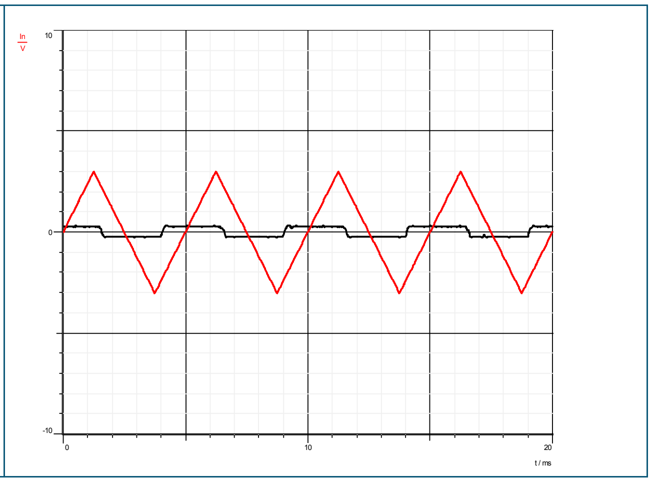
Red: Triangular signal from function generator
Black: Predictor signal 8 bit



Red: Triangular signal from function generator
Black: Predictor signal 4 bit

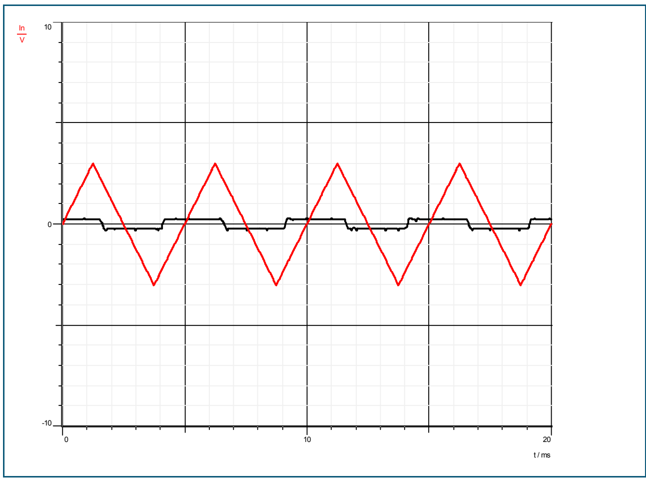


Red: Triangular signal from function generator
Black: DPCM output 8 bit

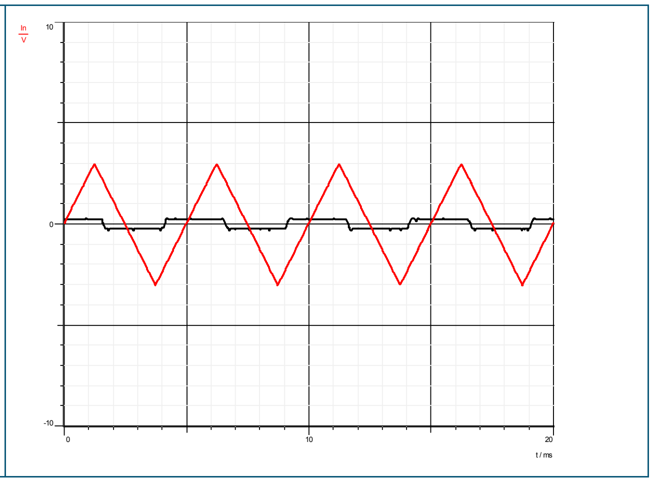


Red: Triangular signal from function generator
Black: DPCM output 4 bit

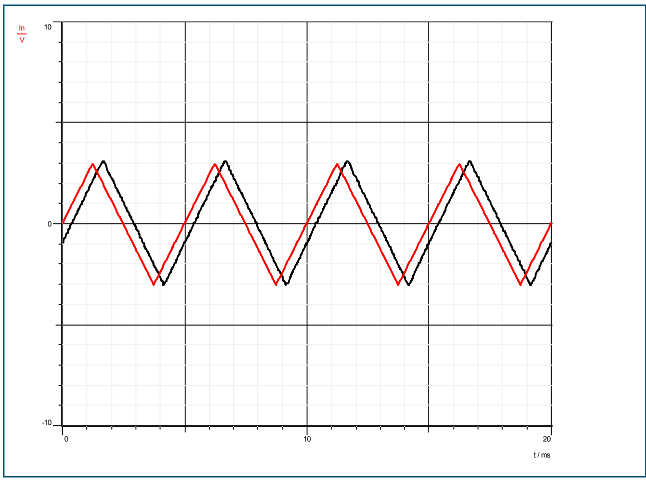
DPCM-Demodulator



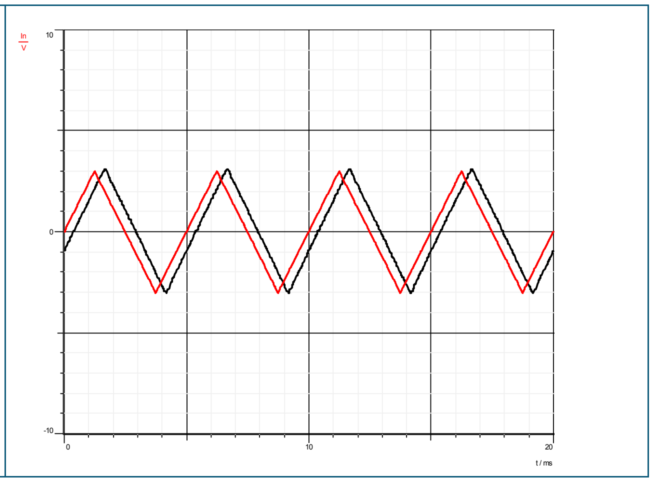
Red: Triangular signal from function generator
Black: Input DPCM demodulator 8 bit



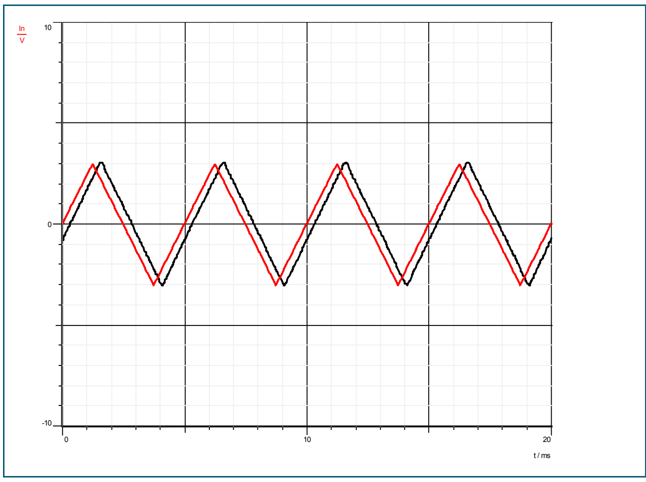
Red: Triangular signal from function generator
Black: Input DPCM demodulator 4 bit



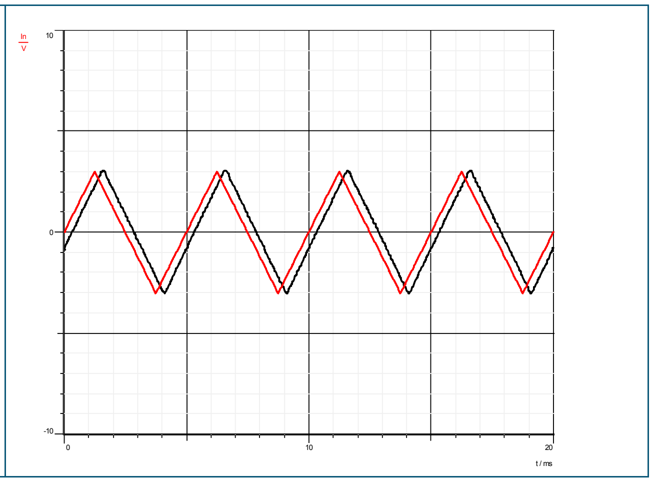
Red: Triangular signal from function generator
Black: Predictor signal, demodulator 8 bit



Red: Triangular signal from function generator
Black: Predictor signal, demodulator 4 bit



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



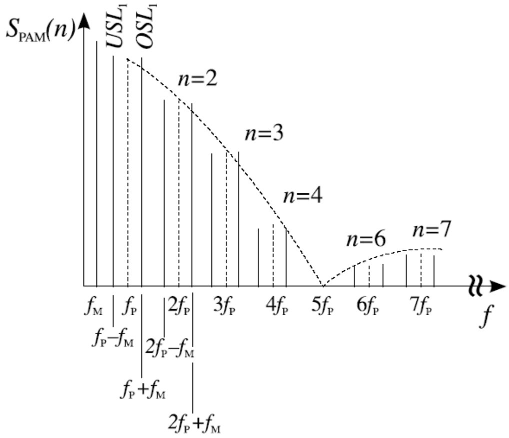
Red: Triangular signal from function generator
Black: PAM output DPCM dem. 4 bit

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.

Worksheets

PAM spectrum



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

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

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**:

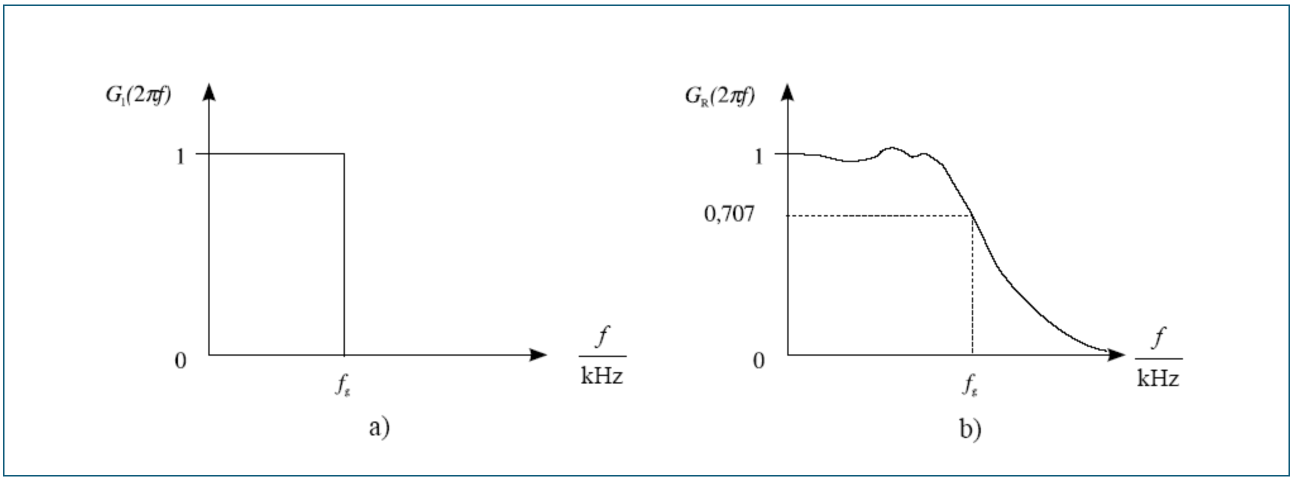
- Each time function limited to a particular bandwidth is specifically determined by its discrete sampling values if the sampling process supplies at least two samples per oscillation at the maximum occurring signal frequency.

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.

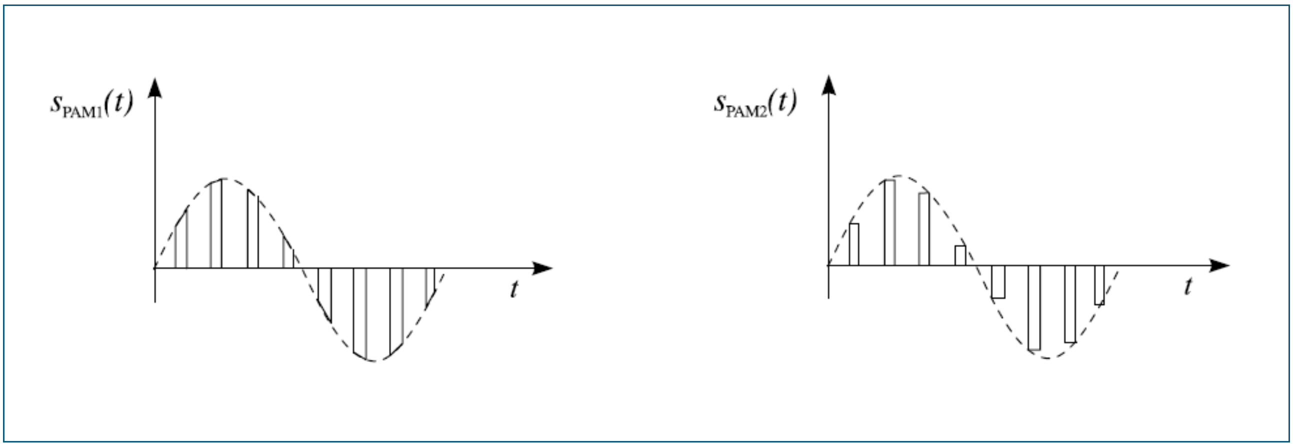


Amplitude response of low pass filters: a = ideal, b = real

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

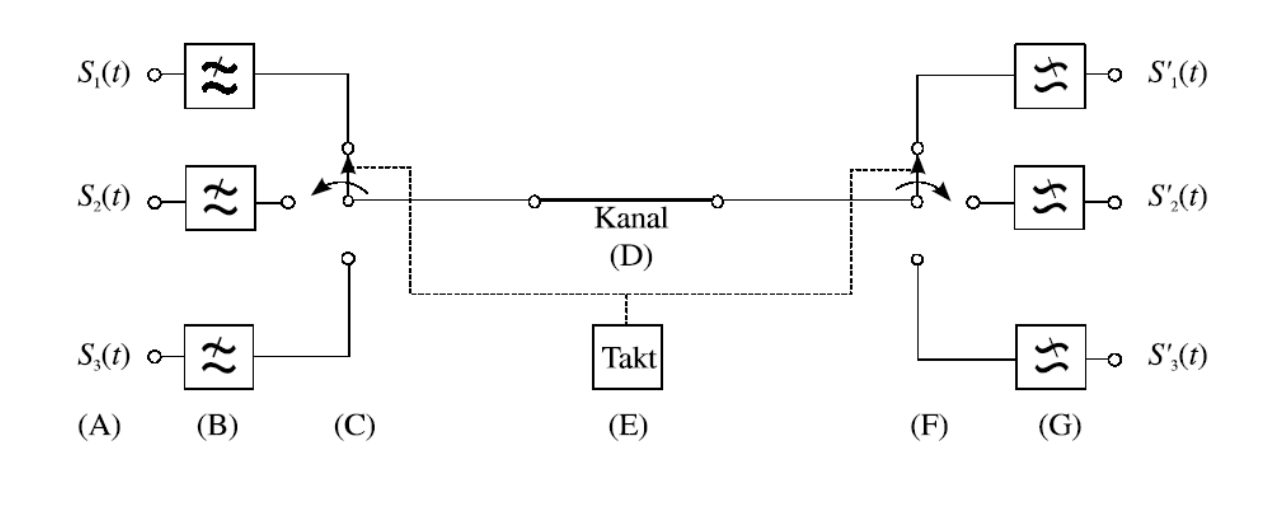
- Both PAM types contain the modulating signal $s_M(t) = A_M \cos(2\pi f_M t$. In PAM_1 this spectral component is only evaluated with the constant factor τ/T_P P. In the case of PAM_2 there is an additional evaluation factor $si(\pi f_M T_0)$. This brings about an additional, signal frequency dependent 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.



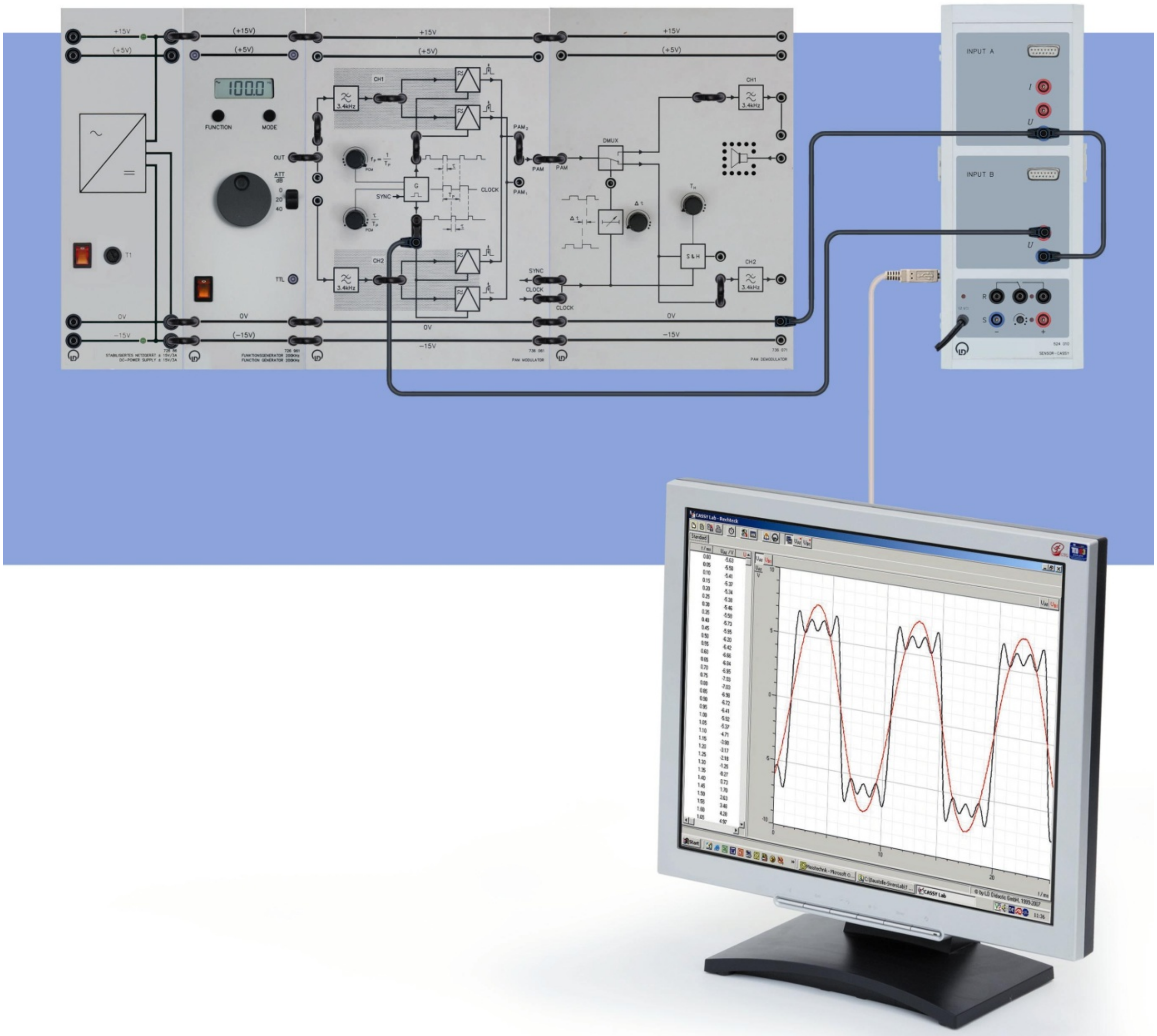
Principle of TDM with PAM
A: Signal sources
B: Band limiting filter
C: Multiplexer
D: Transmission channel
E: Synchronization clock
F: Demultiplexer
G: Low pass demodulator

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 → Clock generator *G*.
- Load the CASSY Lab 2 example pulse frequency5000.labs.
- 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 → Input of channel filter CH1.
- CASSY UB1 → Output of channel filter CH1.
- Load the CASSY Lab 2 example PAMTimeInOut.labs.
- Start the measurement by pressing *F9.*

Display the time characteristic of the PAM.

- Function generator: Sine, 500 Hz, A = 10 Vpp.
- CASSY UA1 → Input PAM Modulator channel CH1.
- CASSY UB1 → Output PAM_1.
- Load the CASSY Lab 2 example PAMTime.labs.
- 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:
  CASSY UB1 → Clock generator *G*.
  Load the CASSY Lab 2 example DutyCycle.labs.
  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 → Input of channel filter CH1 at PAM modulator.
- CASSY UB1 → Output of channel filter CH1 at PAM demodulator.
- Load the CASSY Lab 2 example PAMModDem.labs.
- Start the measurement by pressing *F9.*
- Repeat the measurement for τ/T_P = 30 % and τ/T_P = 10 %.
- Sketch your results.

Variants

- Measure the input- and output signal of the channel filter CH1 for different frequencies and signal forms.
- Display the PAM signal for different duty cycles.
- Investigate the function of the hold stage at the PAM demodulator (T_H). Measure the pulse width as a function of T_H.

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.
- Load the CASSY Lab 2 example PAMFFT.labs.
- Start the measurement by pressing *F9.*
- Repeat the measurement for f_M = 1 kHz and f_M = 2 kHz.
- Sketch your results.

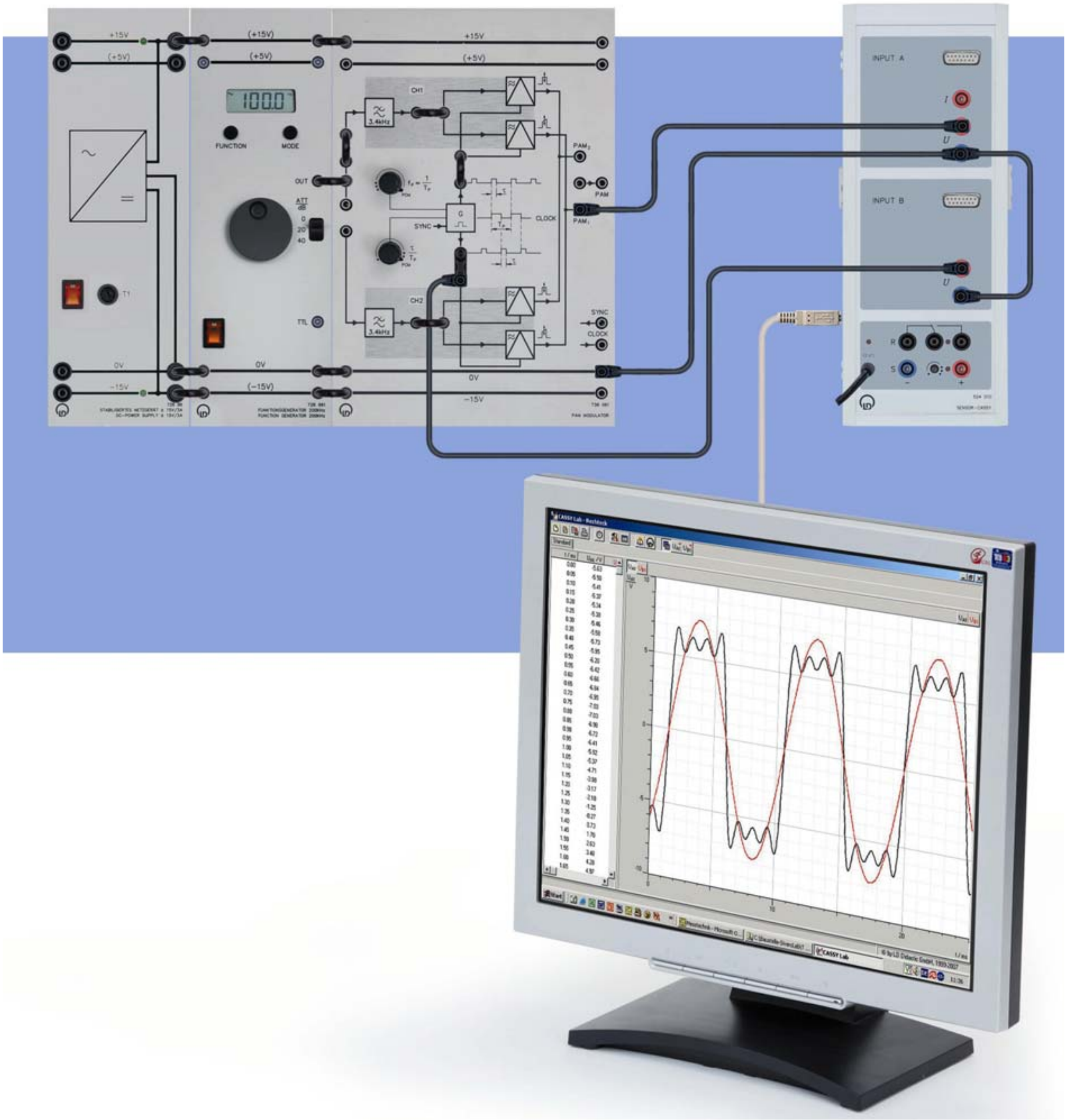
Das PAM_1-Spectrum as a function of the duty cycle.

- 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:
  Load the CASSY Lab 2 example DutyCycle.labs.
  Start the measurement by pressing *F9.*
  Set the duty cycle to τ/T = 30 %.
- Load the CASSY Lab 2 example PAMFFT.labs.
- 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.
- Load the CASSY Lab 2 example PAMFFT.labs.
- 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.

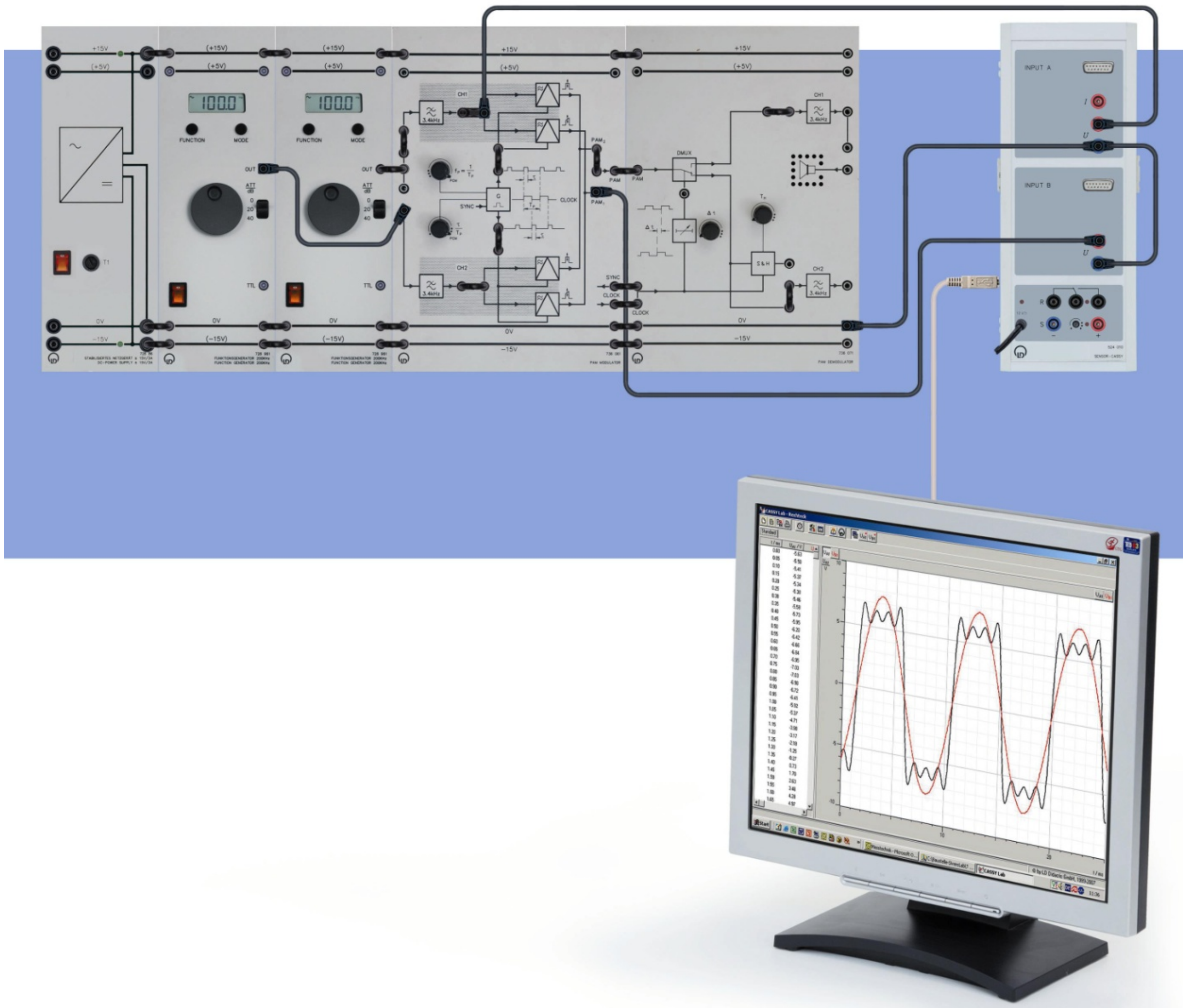
Subsampling in the frequency domain

- Function generator: Sine, 3000 Hz, A = 5 Vpp.
- CASSY UA1 $\rightarrow$ Output PAM_1 at the PAM modulator.
- CASSY UB1 $\rightarrow$ Clock generator *G*.
- For the setting of the sampling frequency f_P = 5000 Hz: Load the CASSY Lab 2 example pulse frequency5000.labs.
- For the setting of the duty cycle τ/T_P= 20 %: Load the CASSY Lab 2 example DutyCycle.labs.
- For the spectrum: Load the CASSY Lab 2 example PAMFFT.labs.
- Start the measurement by pressing *F9.*
- Sketch the results.

Subsampling in the time domain

- Function generator: Sine, 3000 Hz, A = 5 Vpp.
- CASSY UA1 $\rightarrow$ Input of the PAM modulators
- CASSY UB1 $\rightarrow$ Output PAM_1 at the PAM modulator
- Load the CASSY Lab 2 example PAMTime.labs.
- 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.
- Load the CASSY Lab 2 example PAMTDMInput.labs.
- Start the measurement by pressing *F9.*

PAM demodulator time shift Δt → left/middle

- CASSY UA1 → Output PAM demodulator channel CH1.
- CASSY UB1 → Output PAM demodulator channel CH2.
- Load the CASSY Lab 2 example PAMTDMOutput1.labs.
- 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

Interpretation

PAM_2

$s_P(t)$:	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

Measurements are taken at the PAM_2 output.

Interpretation

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

$s_P(t)$: 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

Interpretation

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

Interpretation: PAM time multiplex input

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

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

Interpretation:

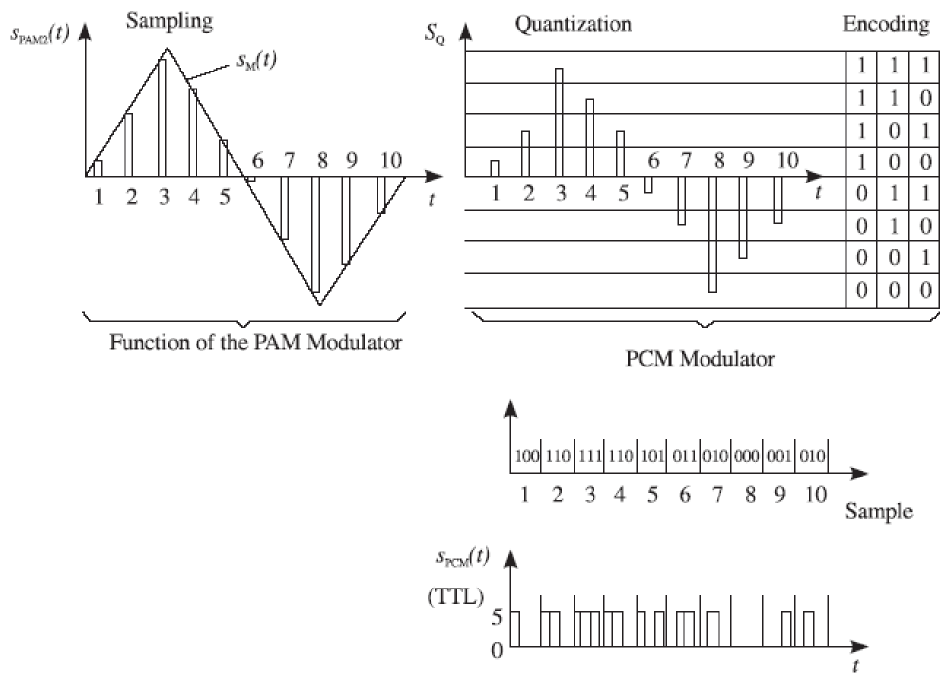
$s_{M1}(t)$: A_{M1} = 5 V f_{M1} = 200 Hz

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

Interpretation:

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

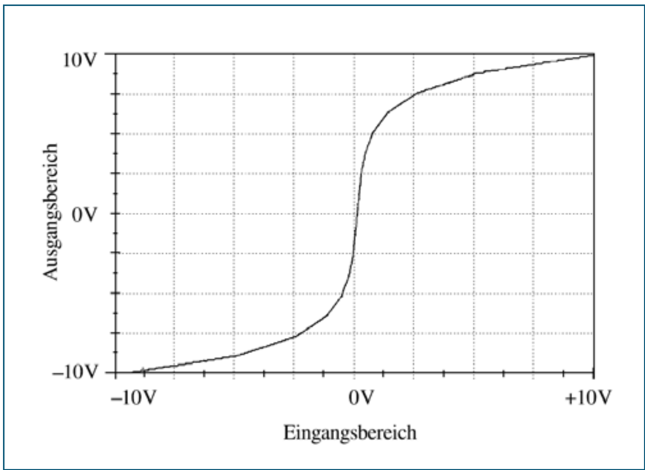
Evidently a sampled, discrete value signal is unsuitable for transmission via a noisy transmission channel. The digital signal at the output of the quantifier requires a shape better suited to the relationships prevailing on the channel. The conversion process required for this is called encoding. Coding 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

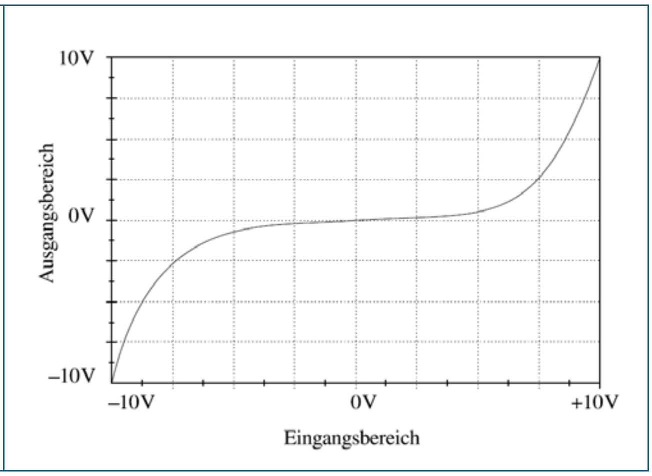
The dual code is called multi-stepped, because in the transition from one binary character to the next, it is possible that several bits have to be converted at the same time (see for example the transition from 3 to 4) in the decimal system. The sampling of the modulating signal $s_M(t)$ in the PAM 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

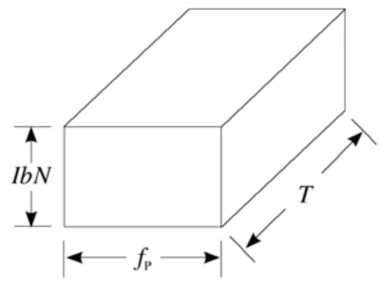


Expander characteristic

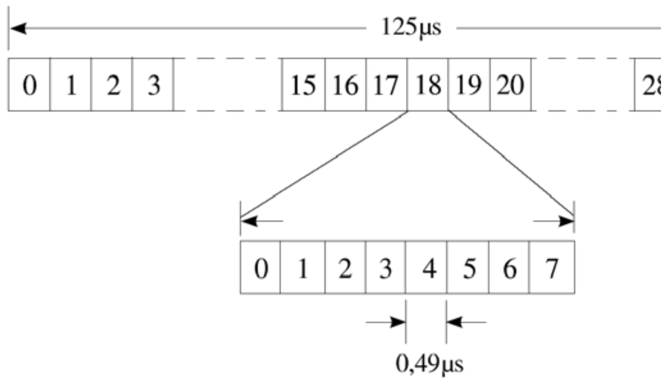
Time division multiplex with PCM

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.



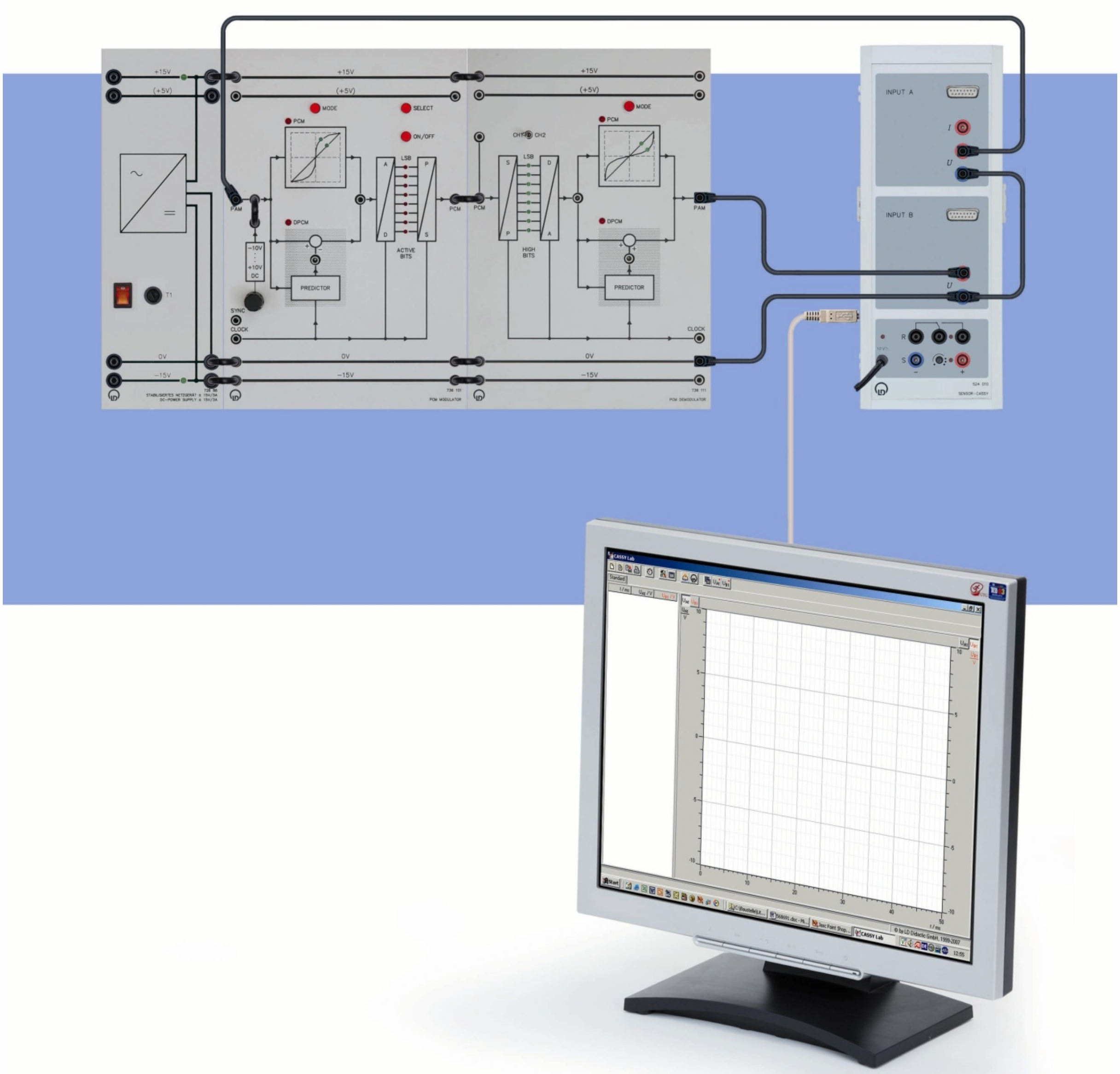
The pulse frame of the PCM 30/32

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.

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.
- Load the CASSY Lab 2 example Quant.labs.
- 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.

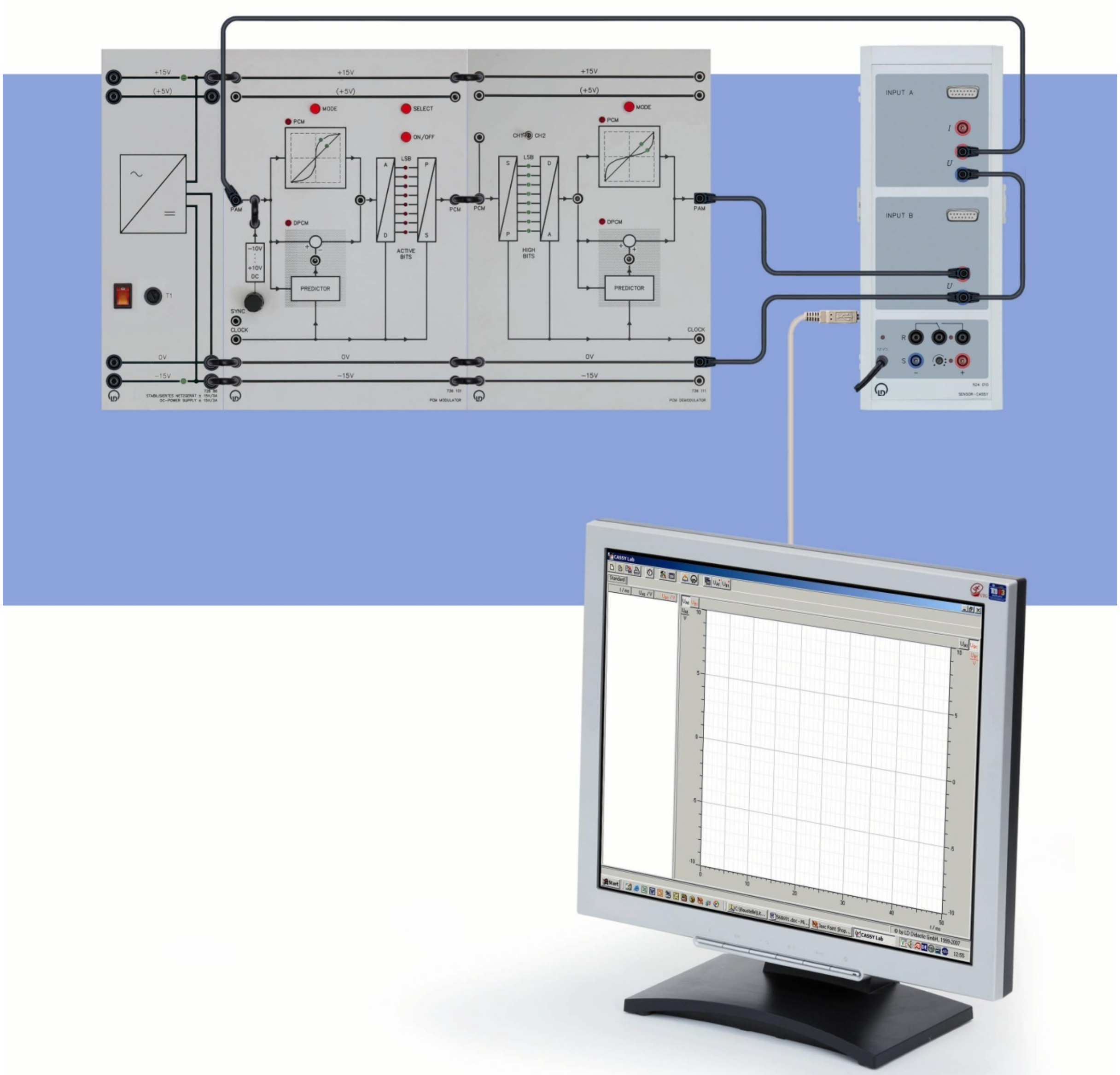
Compressor / Expander characteristic

- 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.

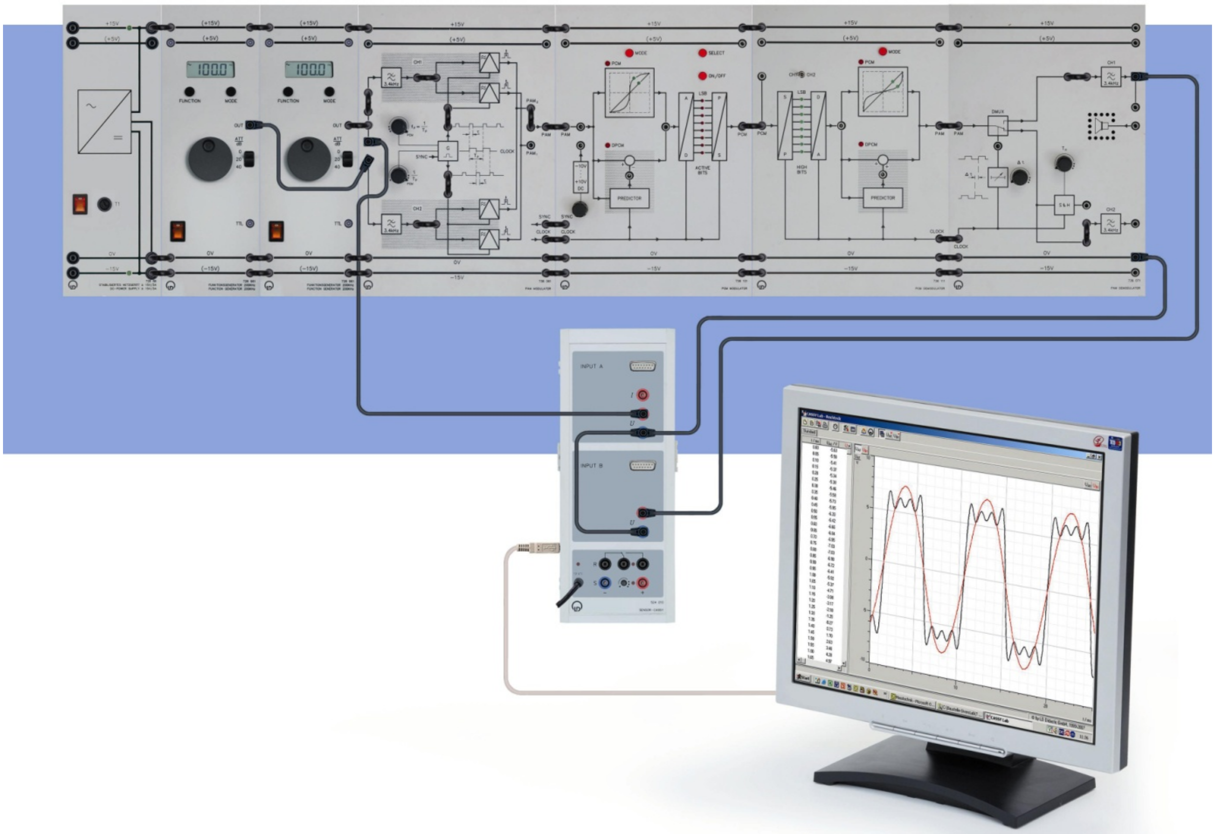
Encoding



- 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.
- Load the CASSY Lab 2 example Code.labs.
- 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	LSB MSB
-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		

PCM transmission

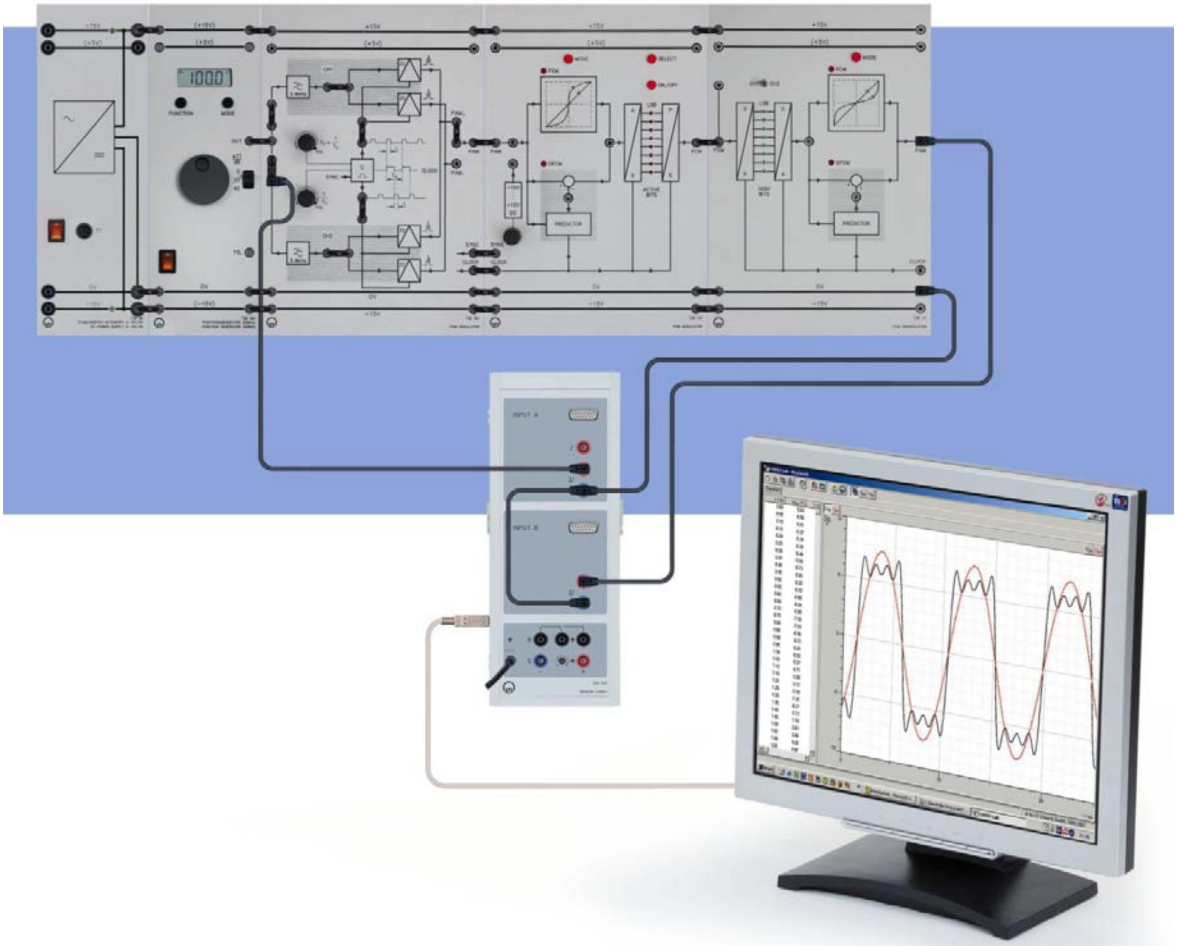


- Set up the shown experiment and switch on the power supply.
- By pressing several times MODE set the PCM modulator and PCM demodulator to: PCM, linear quantization (watch the allocated LEDs).
- Activate all bits. For that press the button SELECT, until all (red) LEDs in the PCM modulator indicate ACTIVE.
- 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
- Load the CASSY Lab 2 example PCMTrans.labs.
- 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.

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.
- Load the CASSY Lab 2 example QNoise.labs.
- 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.

Variants

- Use a sinusoidal modulating signal. This results in a more complicated structure of the quantization noise.
- Record the quantization noise even for non-linear quantization.

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

Coding protects particularly small amplitude values. These are represented by the less significant bits in the PCM word. However, they are just as safe from disturbance as the more significant bits as representatives of the larger signal values. Known binary codes are:

- Dual code
- Symmetrical binary code
- Gray code.

UA1/V	UB1/V	LSB MSB
-9.96		
-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		

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.	

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

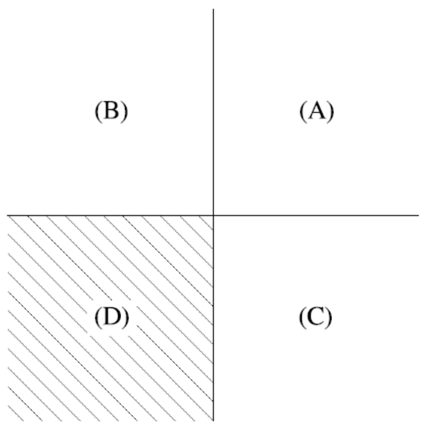
Curve	Color	Signal
U_{A1}	red	Input at PAM-Mod
U_{B1}	blue	*PAM* at PCM-Dem
$Q = U_{A1}-U_{B1}$	black	Quantization noise

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.

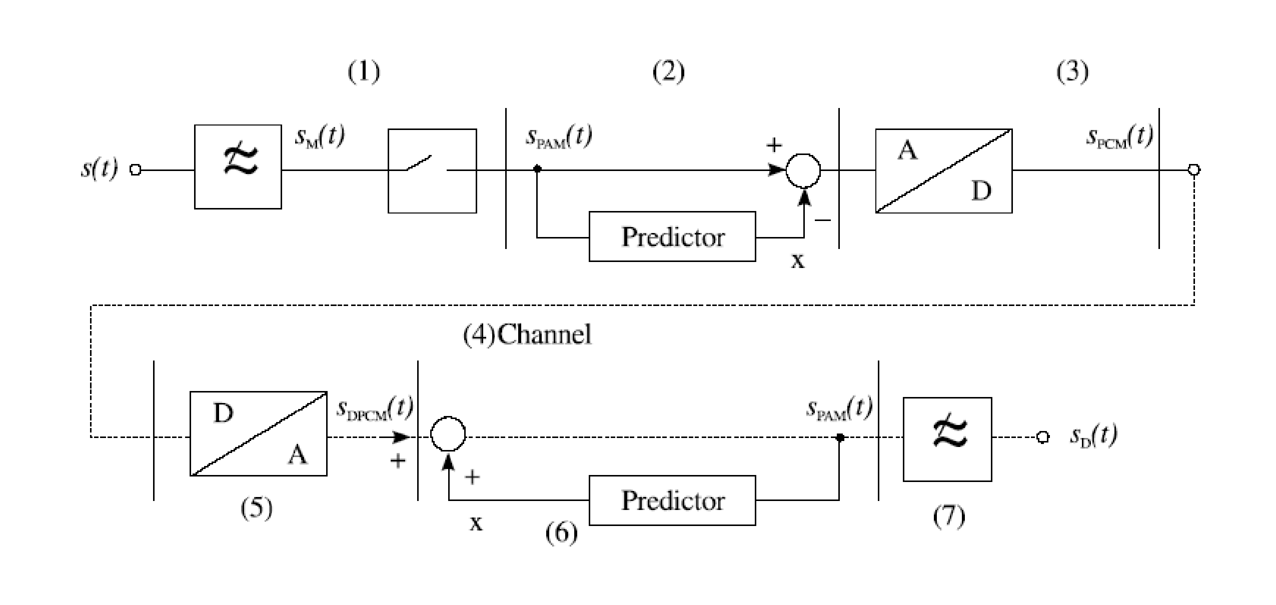


The message plane

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.

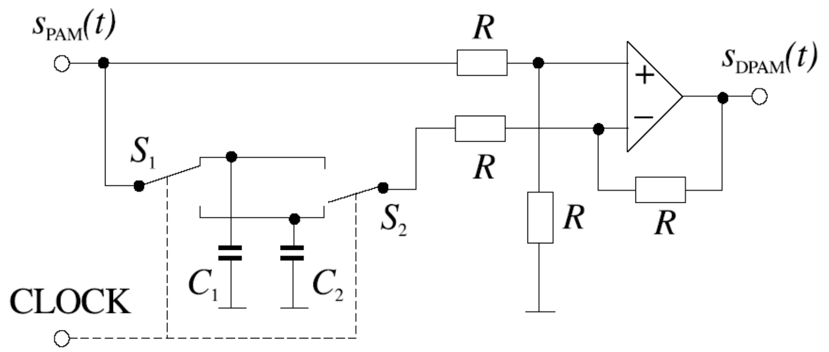
The DPCM constitutes a method of reducing the redundancy of the information signal. The figure demonstrates one possible operating principle for a DPCM link. The quantization is no longer carried out for each individual PAM value. Only the difference between the current PAM value and a predicted value *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.

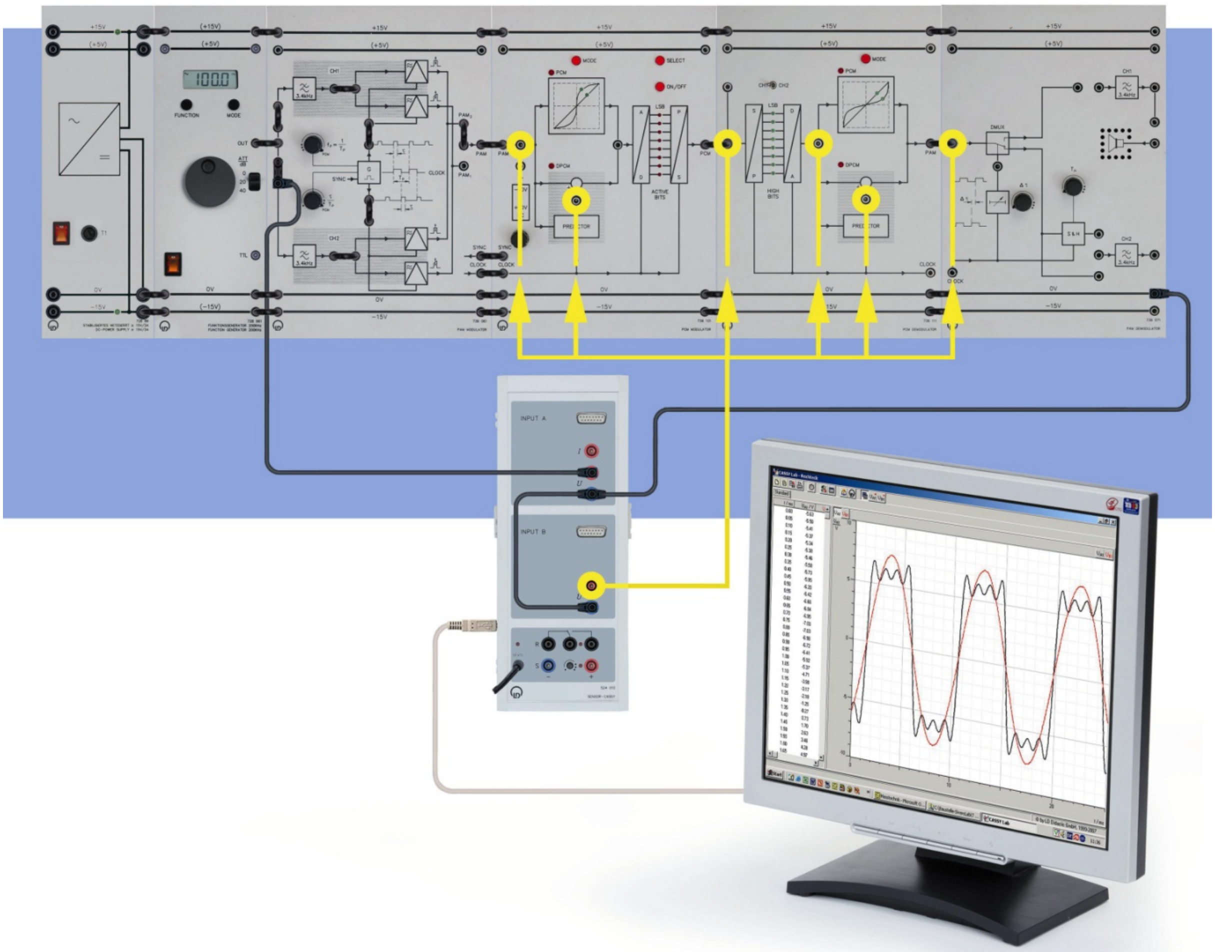


The DPCM modulator contains a differential amplifier and an analog memory. The switches S_1 and S_2 synchronized at the sampling frequency f_P operate in counter sense. Together with storage capacitors C_1 and C_2 they form the PREDICTOR. The formation of the prediction value is simple. The PAM pulses 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

- Load the CASSY Lab 2 example DPCM.labs.
- 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.
- Deactivate the following bits

LSB							MSB
ON	ON	ON	--	--	--	--	ON

- Repeat the experiment for a resolution of 4 bit.

Results

DPCM-Modulator

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

Red: Triangular signal from function generator
Black: PAM-Signal 4 bit

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

Red: Triangular signal from function generator
Black: Predictor signal 4 bit

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

Red: Triangular signal from function generator
Black: DPCM output 4 bit

DPCM-Demodulator

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

Red: Triangular signal from function generator
Black: Input DPCM demodulator 4 bit

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

Red: Triangular signal from function generator
Black: Predictor signal, demodulator 4 bit

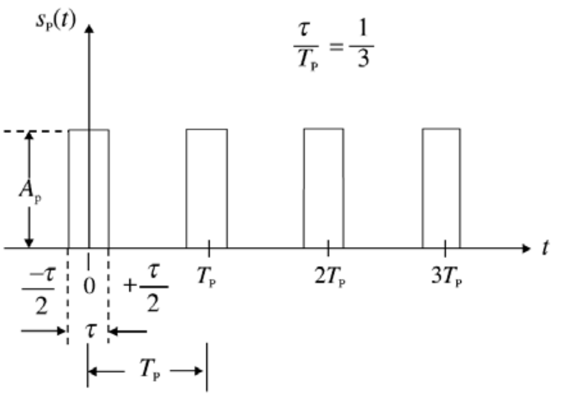
Red: Triangular signal from function generator
Black: PAM output DPCM dem. 8 bit

Red: Triangular signal from function generator
Black: PAM output DPCM dem. 4 bit

Interpretation

Pulse train

Theory



Time domain representation of a pulse train.

A_P	pulse amplitude
τ	Pulse duration
T_P	Pulse period

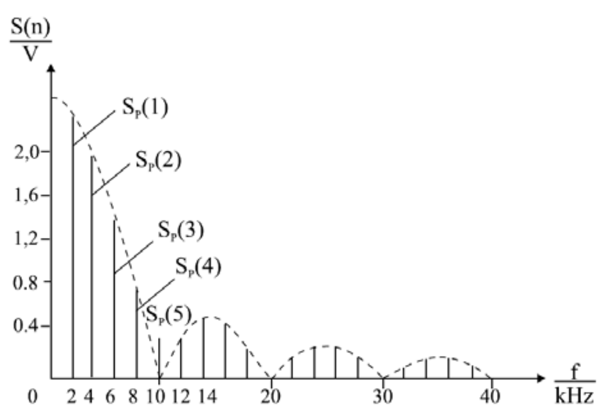
Instead of the pulse period T_P, its reciprocal value, the pulse frequency $f_P = 1/T_P$ can also be equally employed.

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

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 an 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

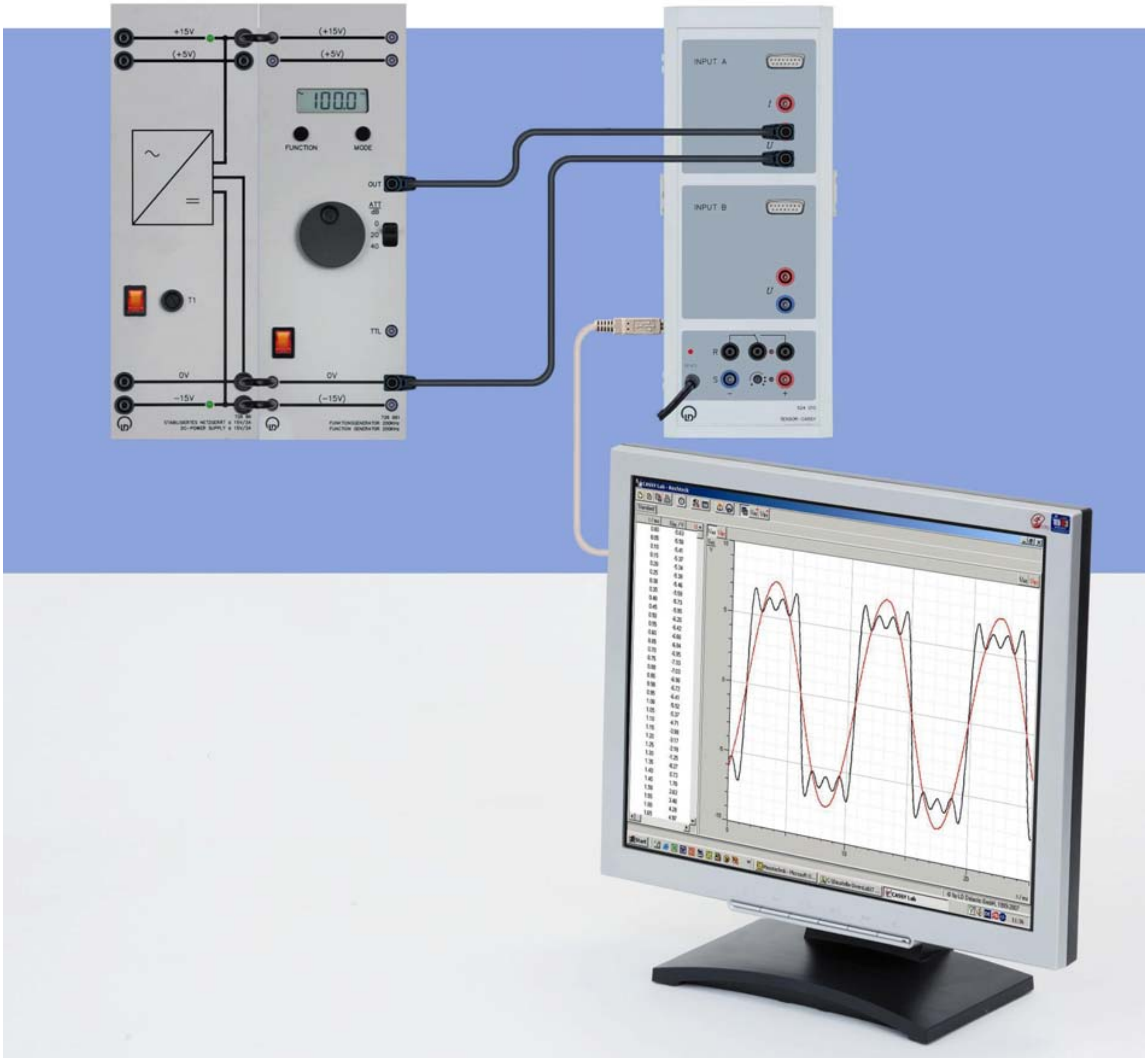
A_P=6V, τ/T_P=0.2, f_P=2kHz

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 fp = 1 kHz, pulse amplitude A_P = 5 V (10 V_{pp}) and duty cycle τ_1/T_P = 1/10.
- Load the CASSY Lab 2 example PulseTime.labs.
- Start the measurement by pressing *F9*
- Determine the time characteristic of the pulse train.
- Determine the spectrum of the pulse train. Load the CASSY Lab 2 example PulseFFT.labs.
- 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_2/T = 2/10, τ_3/T_P = 3/10, τ_4/T_P = 4/10, τ_5/T_P = 5/10 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 = 9
1. zero crossing of the sync-function: 10 kHz

Time characteristic of the pulse train
A_P = 5 V, f_P = 1 kHz, τ_2/T_P = 2/10

FFT spectrum of the pulse train
A_P = 5 V, f_P = 1 kHz, τ_2/T_P = 2/10
Number of lines in each sub spectrum: l = 4
1. zero crossing of the sync-function: 5 kHz

Time characteristic of the pulse train
A_P = 5 V, f_P = 1 kHz, τ_3/T_P = 3/10

FFT spectrum of the pulse train
A_P = 5 V, f_P = 1 kHz, τ_3/T_P = 3/10
Number of lines in each sub spectrum: l = 3
1. zero crossing of the sync-function: 3.3 kHz

Time characteristic of the pulse train
A_P = 5 V, f_P = 1 kHz, τ_4/T_P = 4/10

FFT spectrum of the pulse train
A_P = 5 V, f_P = 1 kHz, τ_4/T_P = 4/10
Number of lines in each sub spectrum: l = 2
1. zero crossing of the sync-function: 2.5 kHz

Time characteristic of the pulse train
A_P = 5 V, f_P = 1 kHz, τ_5/T_P = 5/10

FFT spectrum of the pulse train
A_P = 5 V, f_P = 1 kHz, τ_5 = 5/10
Number of lines in each sub spectrum: l = 1
1. zero crossing of the sync-function: 2 kHz

Time characteristic of the pulse train
A_P = 5 V, f_P = 1 kHz, τ_6/T_P = 9/10

FFT spectrum of the pulse train
A_P = 5 V, f_P = 1 kHz, τ_6/T_P = 9/10
Number of lines in each sub spectrum: l = 9
1. zero crossing of the sync-function: 10 kHz

Summary

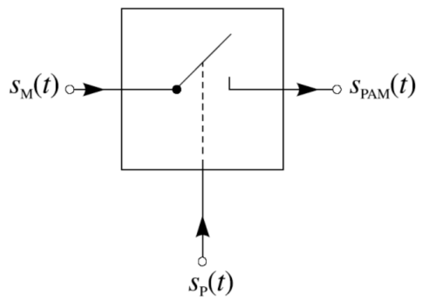
Expressed by τ/T_P and f_P, the zero crossings appear at:

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

	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			

Pulse amplitude modulation (PAM)

Theory



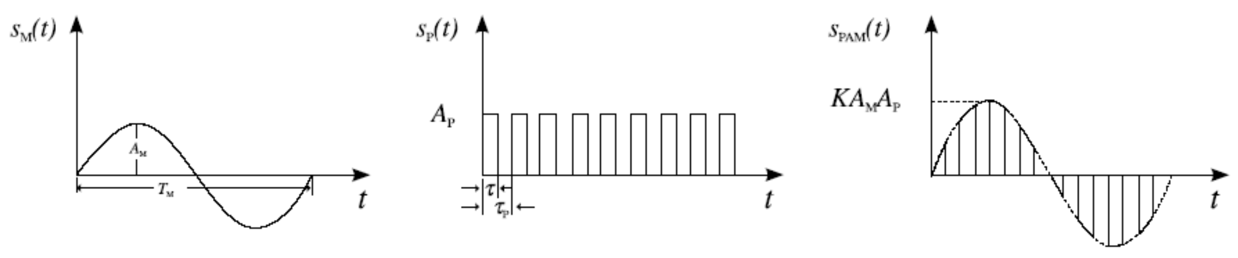
PAM means modulation by a switch. The conversion of the analog signal into a digital signal requires three steps:

- 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 $s_M(t)$.



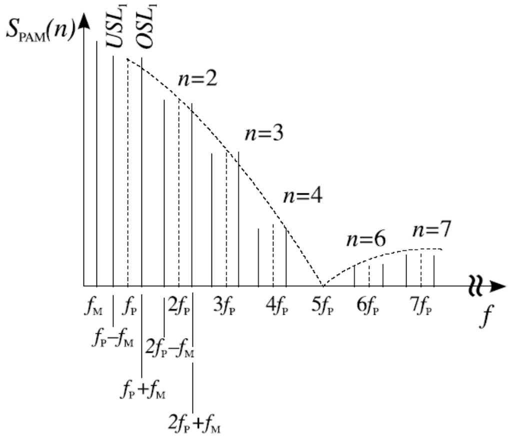
Generation of PAM

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:

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

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**:

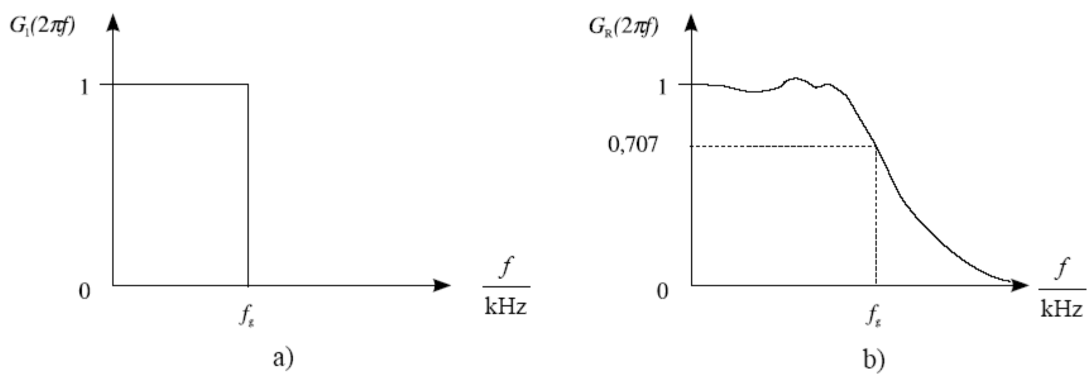
- Each time function limited to a particular bandwidth is specifically determined by its discrete sampling values if the sampling process supplies at least two samples per oscillation at the maximum occurring signal frequency.

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.

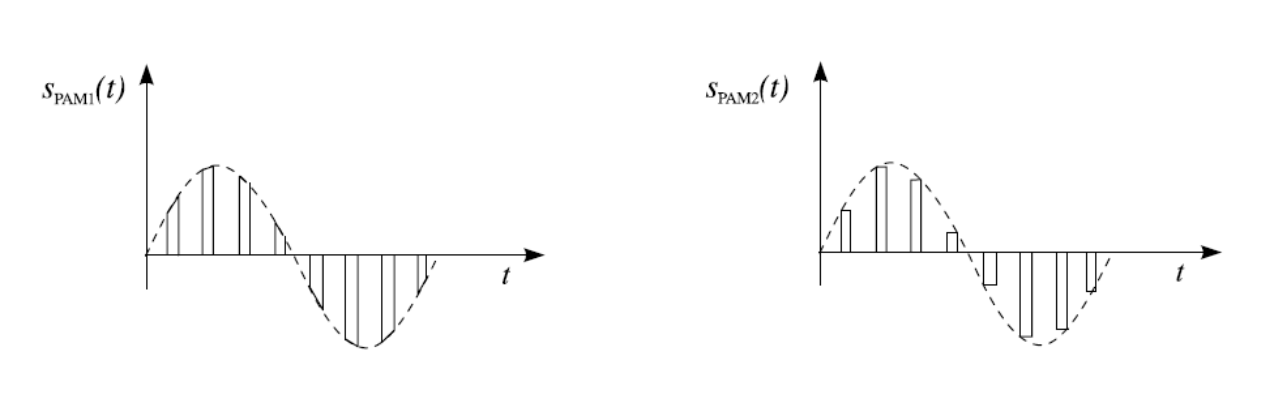


Amplitude response of low pass filters: a = ideal, b = real

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

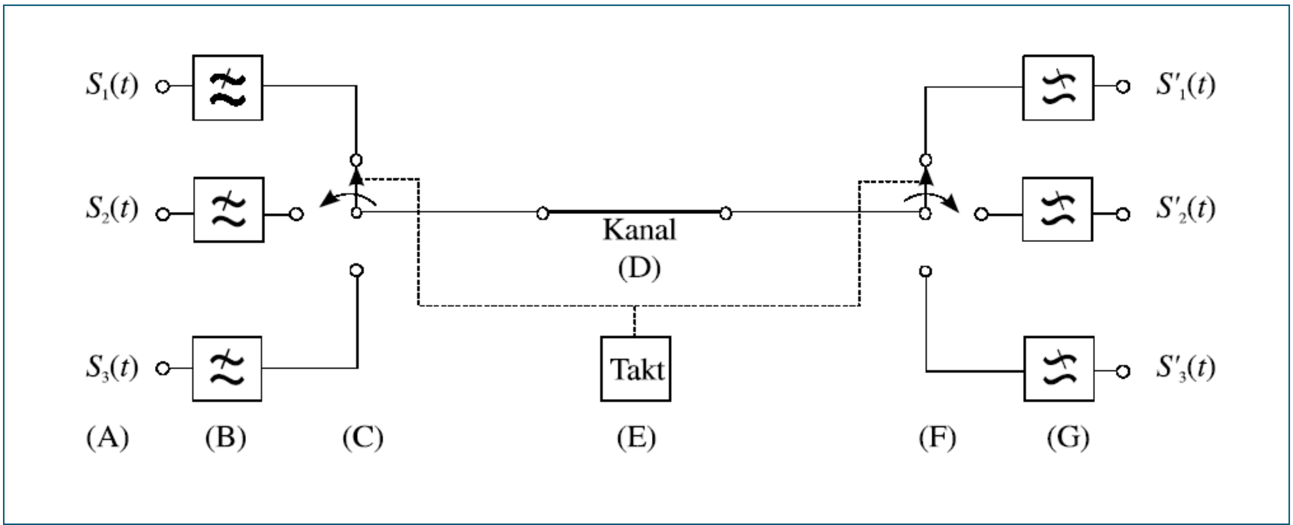
- Both PAM types contain the modulating signal $s_M(t) = A_M \cos(2\pi f_M t$. In PAM_1 this spectral component is only evaluated with the constant factor τ/T_P P. In the case of PAM_2 there is an additional evaluation factor $si(\pi f_M T_0)$. This brings about an additional, signal frequency dependent 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.



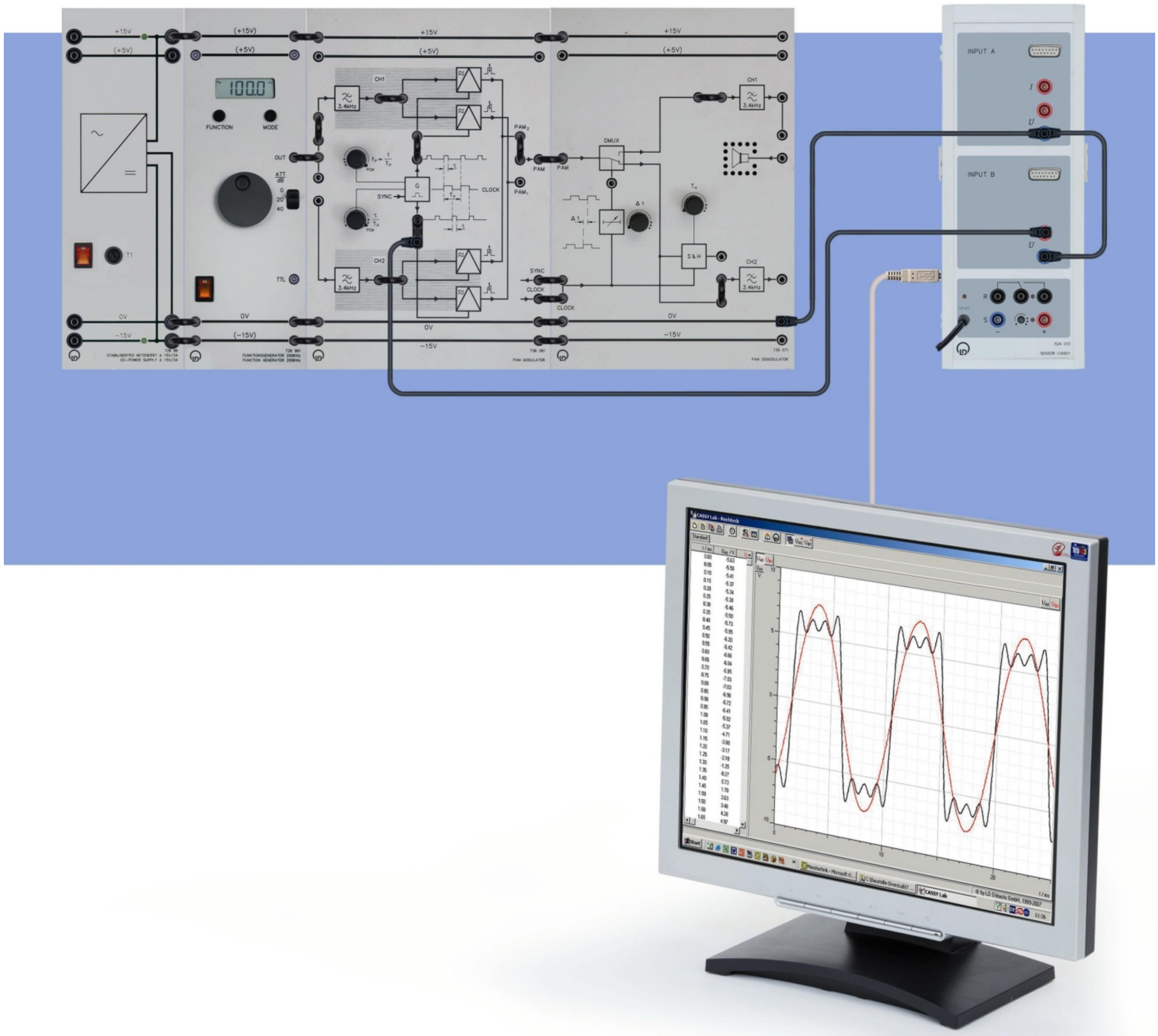
Principle of TDM with PAM
A: Signal sources
B: Band limiting filter
C: Multiplexer
D: Transmission channel
E: Synchronization clock
F: Demultiplexer
G: Low pass demodulator

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 → Clock generator *G*.
- Load the CASSY Lab 2 example pulse frequency5000.labs.
- 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 → Input of channel filter CH1.
- CASSY UB1 → Output of channel filter CH1.
- Load the CASSY Lab 2 example PAMTimeInOut.labs.
- Start the measurement by pressing *F9.*

Display the time characteristic of the PAM.

- Function generator: Sine, 500 Hz, A = 10 Vpp.
- CASSY UA1 → Input PAM Modulator channel CH1.
- CASSY UB1 → Output PAM_1.
- Load the CASSY Lab 2 example PAMTime.labs.
- 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:
  CASSY UB1 → Clock generator *G*.
  Load the CASSY Lab 2 example DutyCycle.labs.
  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 → Input of channel filter CH1 at PAM modulator.
- CASSY UB1 → Output of channel filter CH1 at PAM demodulator.
- Load the CASSY Lab 2 example PAMModDem.labs.
- Start the measurement by pressing *F9.*
- Repeat the measurement for τ/T_P = 30 % and τ/T_P = 10 %.
- Sketch your results.

Variants

- Measure the input- and output signal of the channel filter CH1 for different frequencies and signal forms.
- Display the PAM signal for different duty cycles.
- Investigate the function of the hold stage at the PAM demodulator (T_H). Measure the pulse width as a function of T_H.

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.
- Load the CASSY Lab 2 example PAMFFT.labs.
- Start the measurement by pressing *F9.*
- Repeat the measurement for f_M = 1 kHz and f_M = 2 kHz.
- Sketch your results.

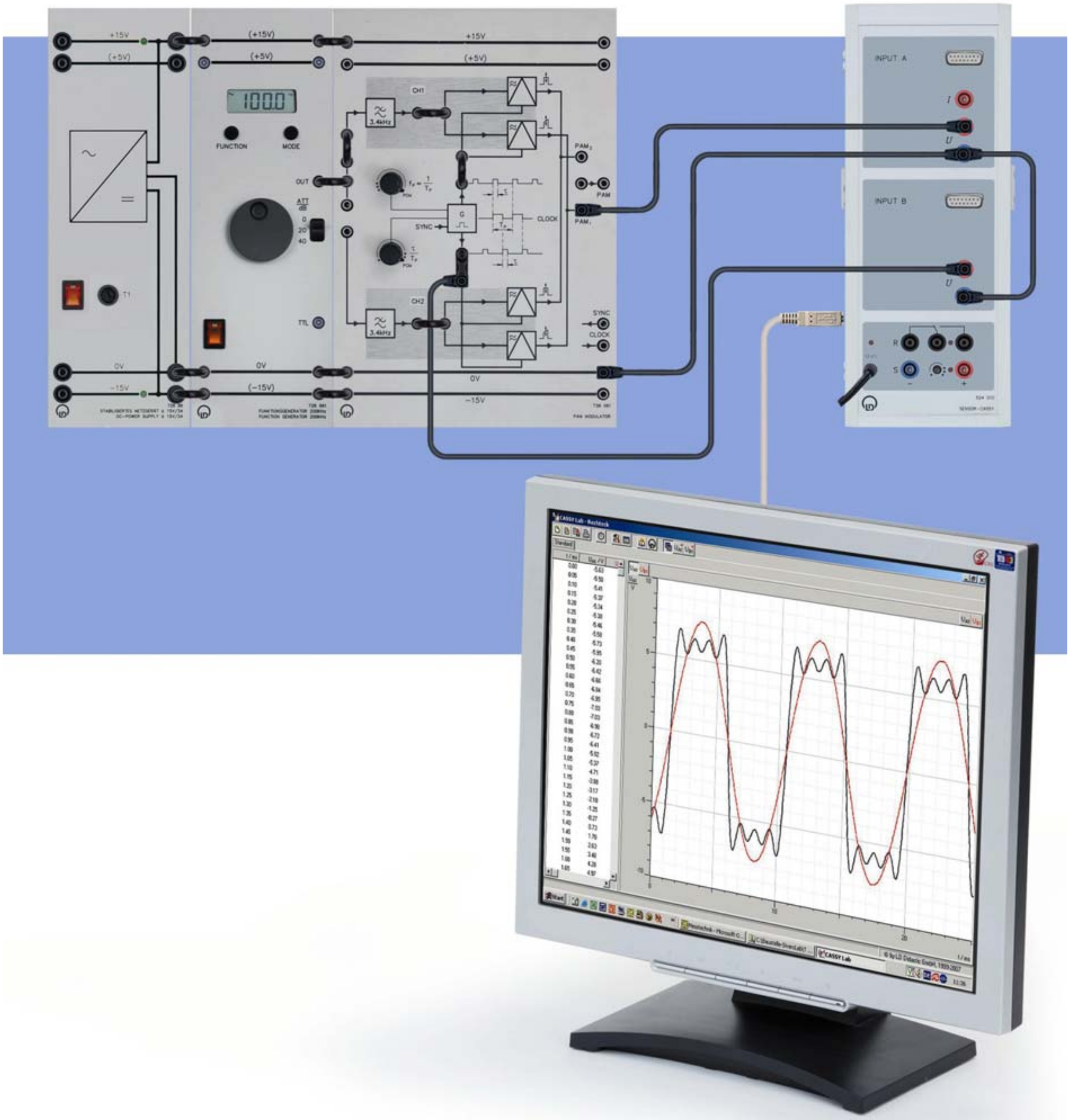
Das PAM_1-Spectrum as a function of the duty cycle.

- 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:
  Load the CASSY Lab 2 example DutyCycle.labs.
  Start the measurement by pressing *F9.*
  Set the duty cycle to τ/T = 30 %.
- Load the CASSY Lab 2 example PAMFFT.labs.
- 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.
- Load the CASSY Lab 2 example PAMFFT.labs.
- 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.

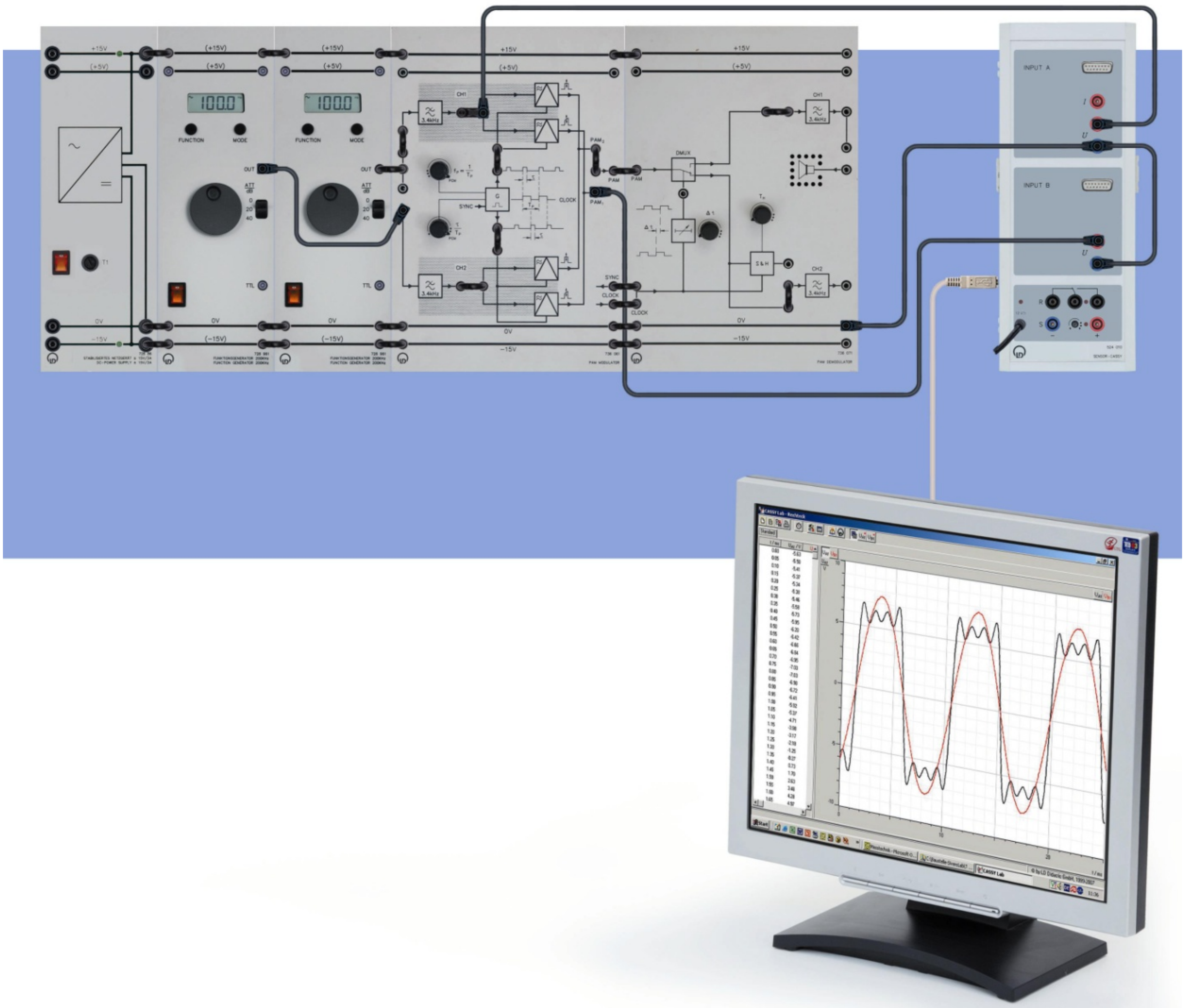
Subsampling in the frequency domain

- 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: Load the CASSY Lab 2 example pulse frequency5000.labs.
- For the setting of the duty cycle τ/T_P= 20 %: Load the CASSY Lab 2 example DutyCycle.labs.
- For the spectrum: Load the CASSY Lab 2 example PAMFFT.labs.
- 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
- Load the CASSY Lab 2 example PAMTime.labs.
- 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 $\rightarrow$ Input PAM modulator channel CH1.
- CASSY UB1 $\rightarrow$ Output PAM modulator PAM_1.
- Load the CASSY Lab 2 example PAMTDMInput.labs.
- Start the measurement by pressing *F9.*

PAM demodulator time shift $\Delta t \rightarrow$ left/middle

- CASSY UA1 $\rightarrow$ Output PAM demodulator channel CH1.
- CASSY UB1 $\rightarrow$ Output PAM demodulator channel CH2.
- Load the CASSY Lab 2 example PAMTDMOutput1.labs.
- 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 PAM_2 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	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

Measurements are taken at the PAM_2 output.

Interpretation

Displaying aliasing

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

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

PAM spectrum for subsampling

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

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

PAM signal for subsampling

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

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

Demodulated signal for subsampling

Interpretaion

PAM time multiplex

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

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

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

Demod. signals at the output of the demultiplexer time shift $\Delta t \rightarrow$ left

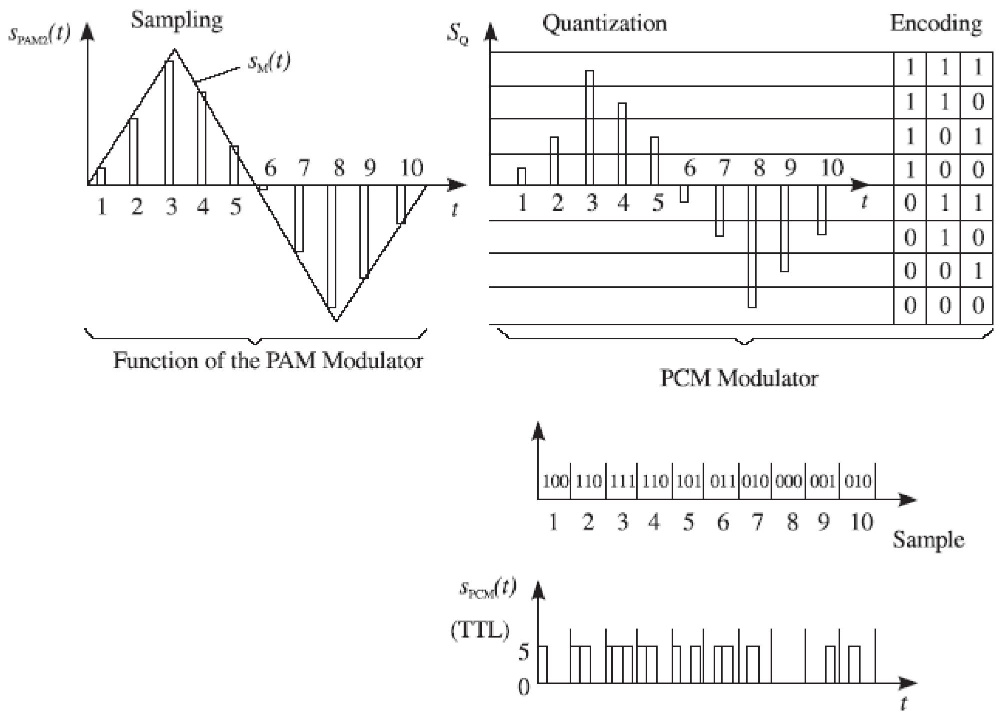
$s_{M1}(t)$: $A_{M1} = 5$ V $f_{M1} = 200$ Hz

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

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

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

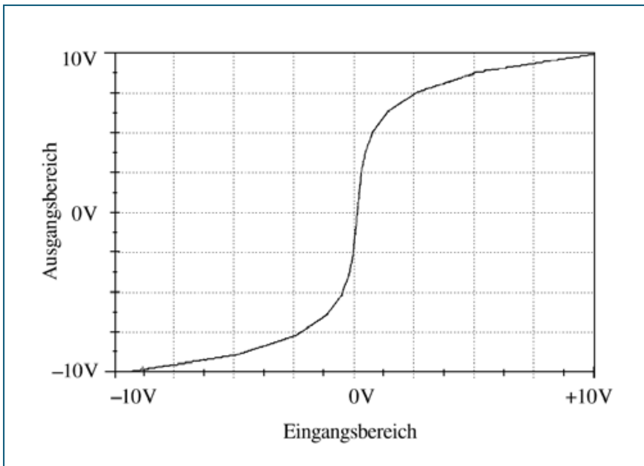
Evidently a sampled, discrete value signal is unsuitable for transmission via a noisy transmission channel. The digital signal at the output of the quantifier requires a shape better suited to the relationships prevailing on the channel. The conversion process required for this is called encoding. Coding 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

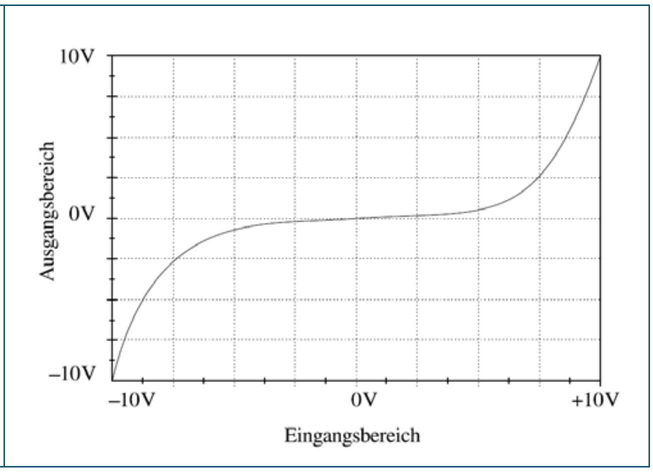
The dual code is called multi-stepped, because in the transition from one binary character to the next, it is possible that several bits have to be converted at the same time (see for example the transition from 3 to 4) in the decimal system. The sampling of the modulating signal $s_M(t)$ in the PAM 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

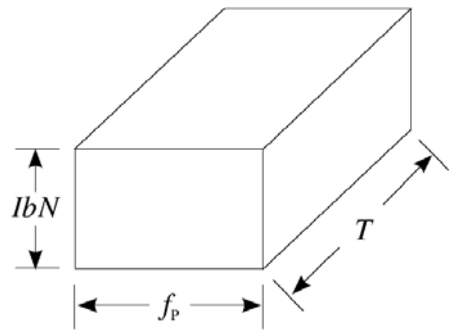


Expander characteristic

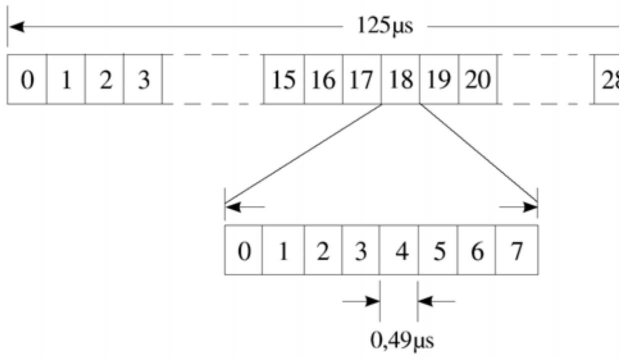
Time division multiplex with PCM

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.



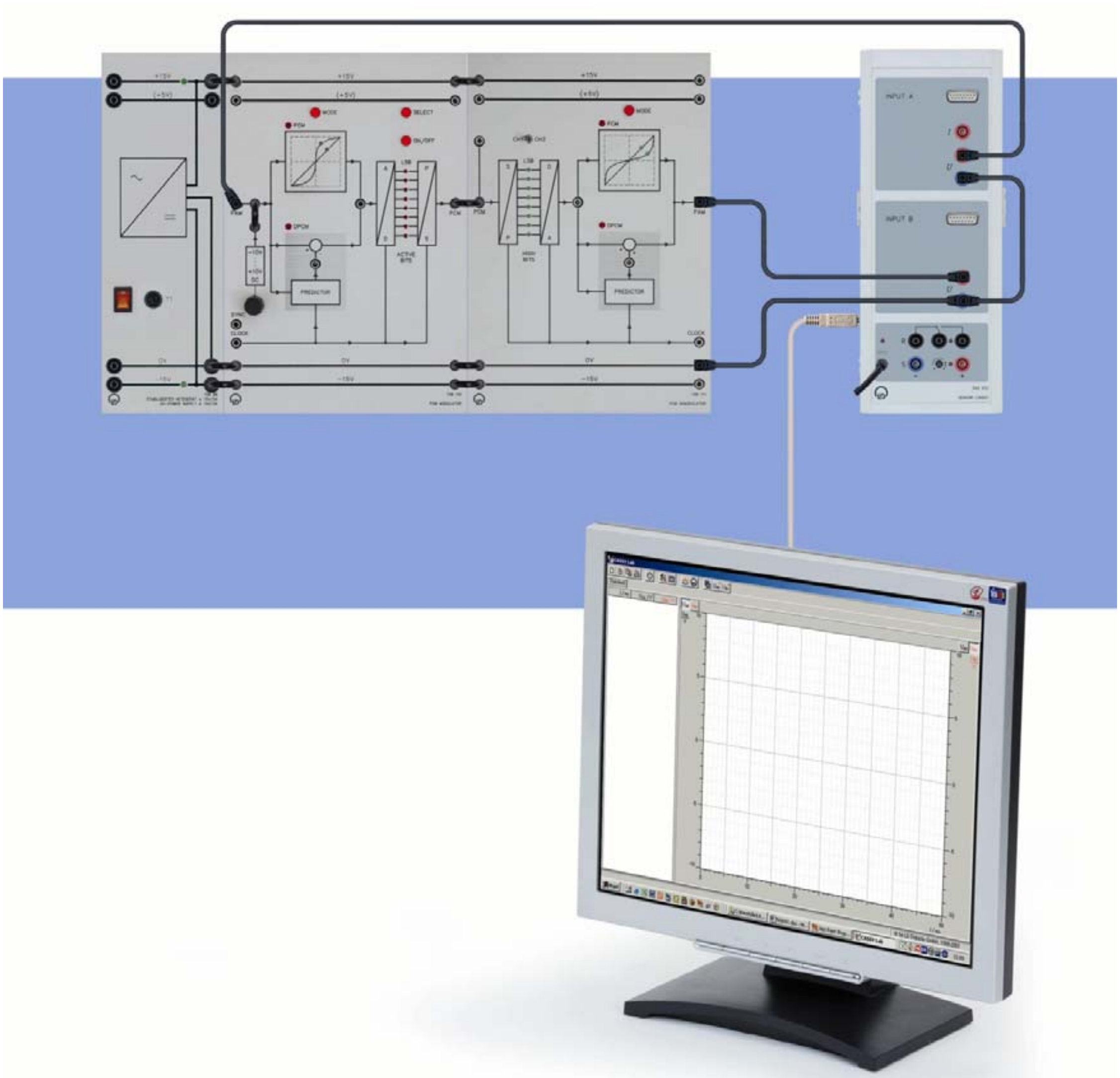
The pulse frame of the PCM 30/32

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.

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.
- Load the CASSY Lab 2 example Quant.labs.
- 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.

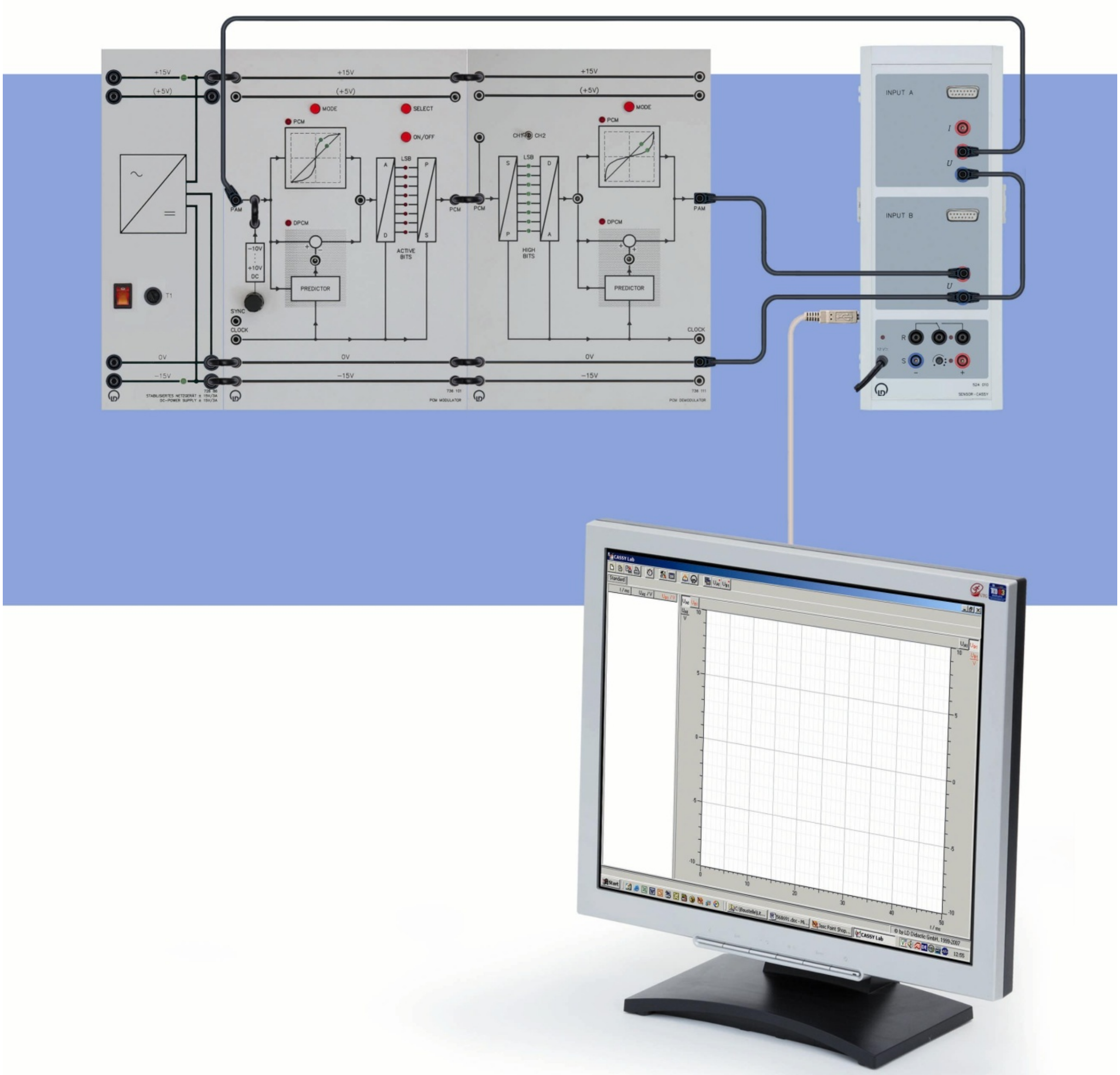
Compressor / Expander characteristic

- 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.

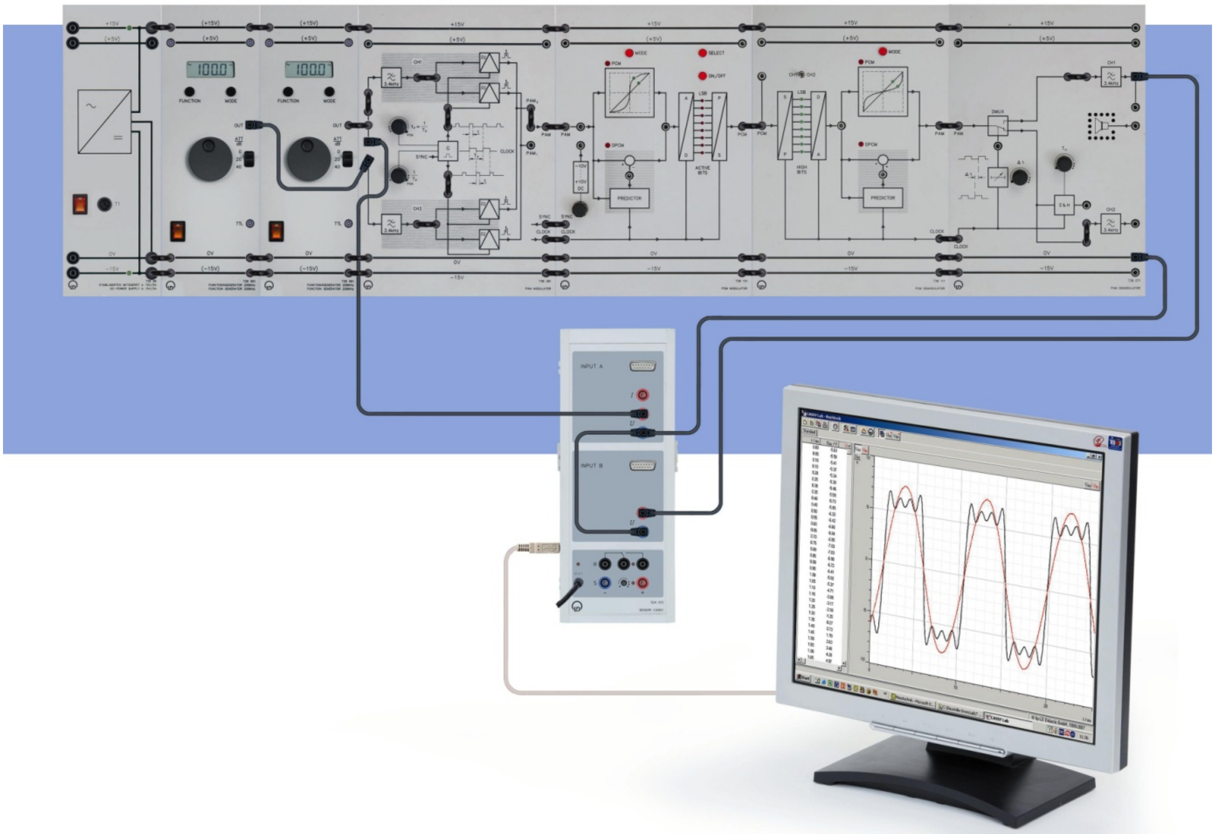
Encoding



- 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.
- Load the CASSY Lab 2 example Code.labs.
- 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	LSB MSB
-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		

PCM transmission

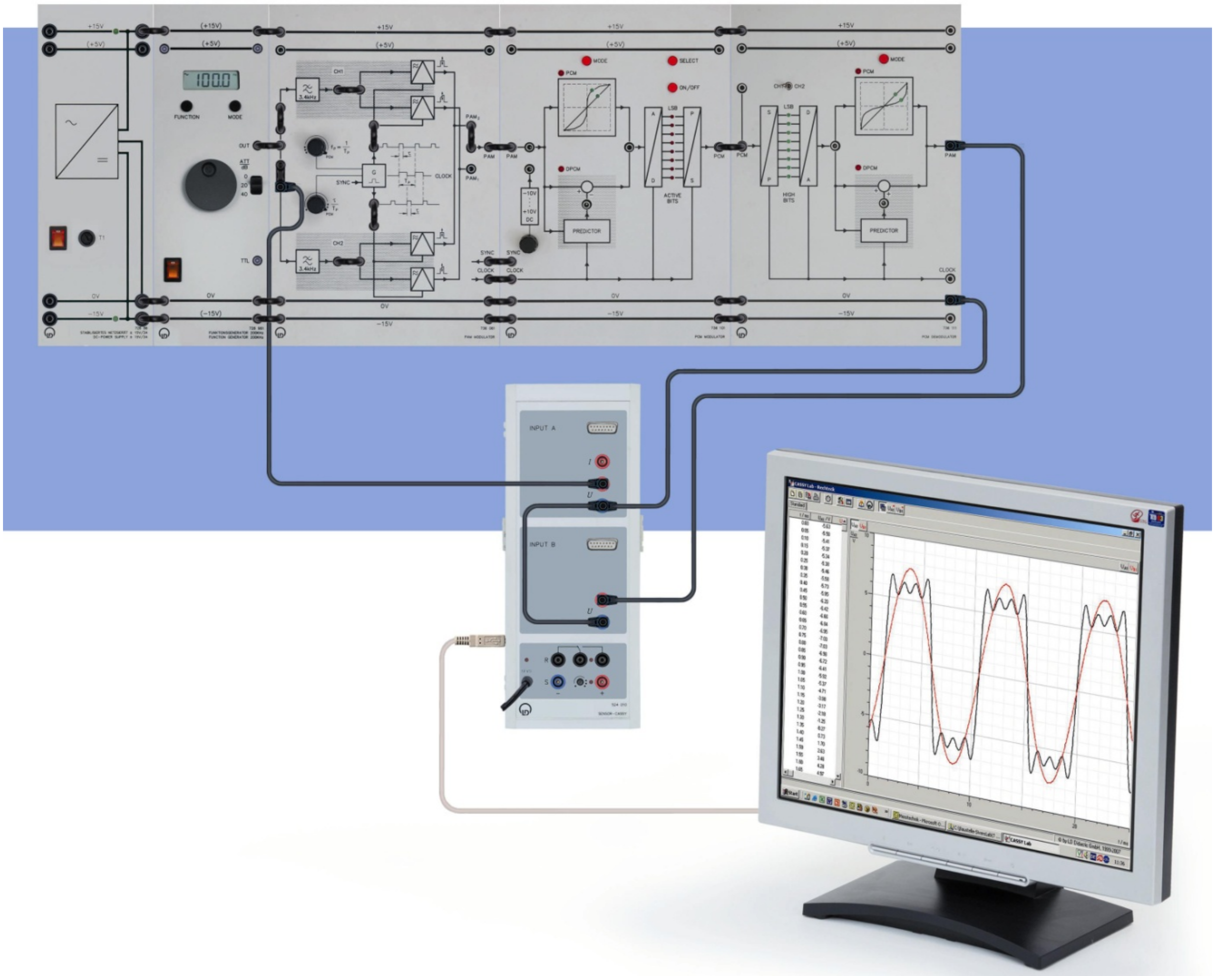


- Set up the shown experiment and switch on the power supply.
- By pressing several times MODE set the PCM modulator and PCM demodulator to: PCM, linear quantization (watch the allocated LEDs).
- Activate all bits. For that press the button SELECT, until all (red) LEDs in the PCM modulator indicate ACTIVE.
- 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
- Load the CASSY Lab 2 example PCMTrans.labs.
- Start the measurement by pressing *F9.*

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

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.
- Load the CASSY Lab 2 example QNoise.labs.
- 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.

Variants

- Use a sinusoidal modulating signal. This results in a more complicated structure of the quantization noise.
- Record the quantization noise even for non-linear quantization.

Results

Quantization

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

Resolution: 8 bit Compressor characteristic **Observation:**	
Resolution: 5 bit Compressor characteristic **Observation:**	
Resolution: 8 bit Expander characteristic **Observation:**	
Resolution: 5 bit Expander characteristic **Observation:**	

Encoding

Coding protects particularly small amplitude values. These are represented by the less significant bits in the PCM word. However, they are just as safe from disturbance as the more significant bits as representatives of the larger signal values. Known binary codes are:

- Dual code
- Symmetrical binary code
- Gray code.

UA1/V	UB1/V	LSB MSB
-9.96		
-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		

PCM transmission

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

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

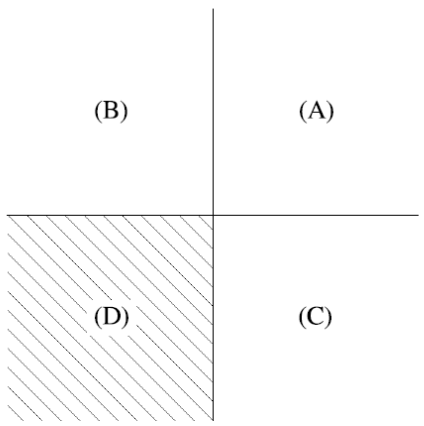
Curve	Color	Signal
U_{A1}	red	Input at PAM-Mod
U_{B1}	blue	*PAM* at PCM-Dem
$Q = U_{A1}-U_{B1}$	black	Quantization noise

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.

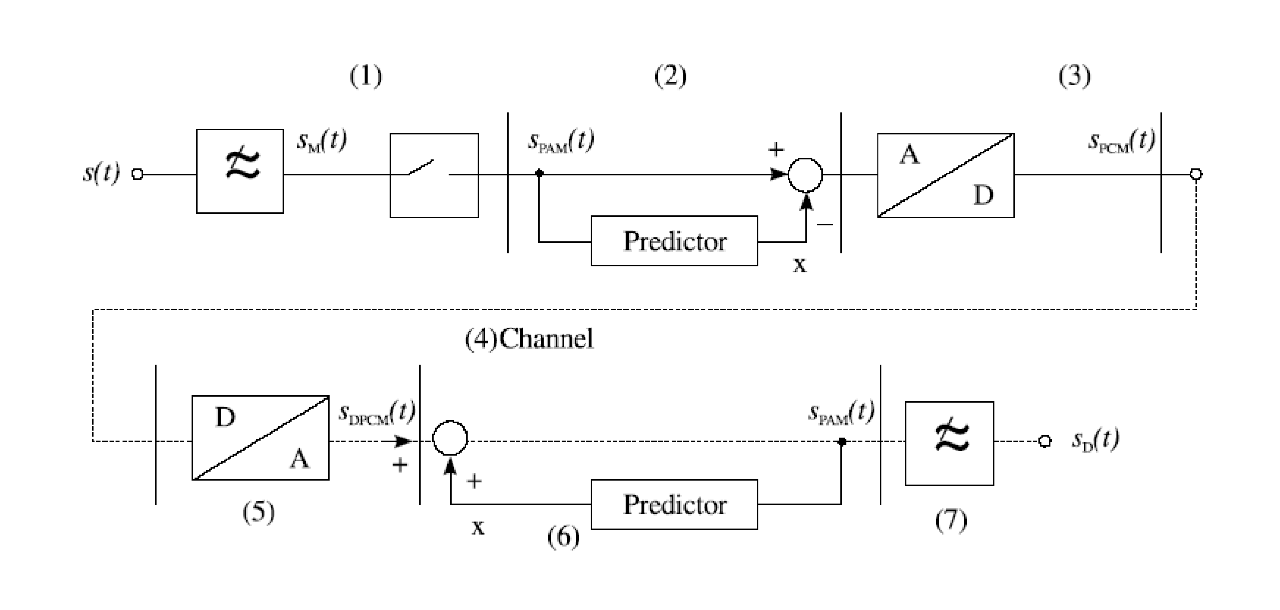


The message plane

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.

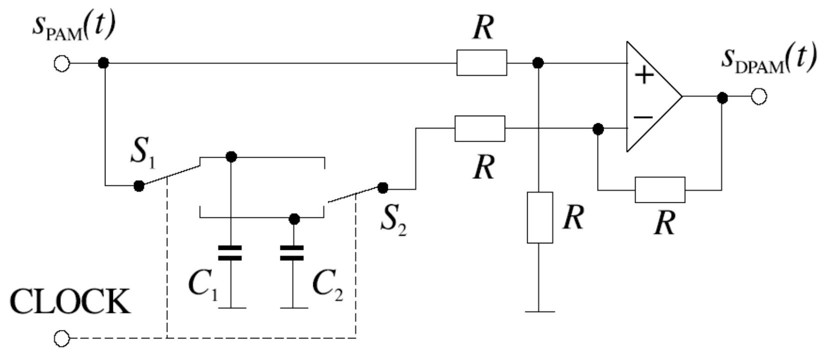


Basic setup of a DPCM system
1: Sampling (PAM modulator)
2: Decorrelation, formation of the predicted value (PCM modulator)
3: Quantization, coding (PCM modulator)
4: Channel
5: Decoding, D/A conversion (PCM demodulator)
6: Recovery of the redundancy (PCM demodulator)
7: Recovery of the time continuation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

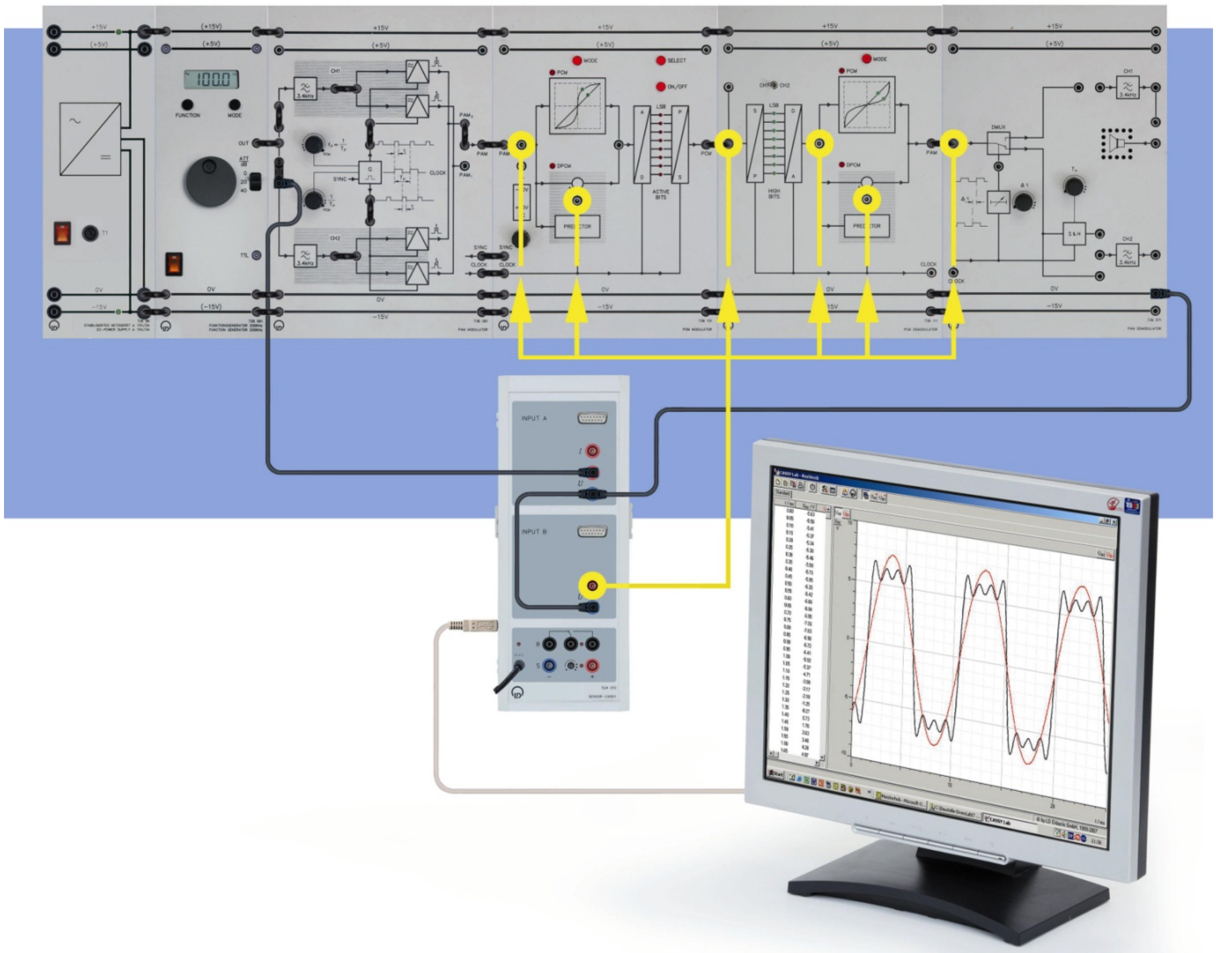


The DPCM modulator contains a differential amplifier and an analog memory. The switches S_1 and S_2 synchronized at the sampling frequency f_P operate in counter sense. Together with storage capacitors C_1 and C_2 they form the PREDICTOR. The formation of the prediction value is simple. The PAM pulses 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

- Load the CASSY Lab 2 example DPCM.labs.
- 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.
- Deactivate the following bits

LSB							MSB
ON	ON	ON	--	--	--	--	ON

- Repeat the experiment for a resolution of 4 bit.

Results

DPCM-Modulator

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

Red: Triangular signal from function generator
Black: PAM-Signal 4 bit

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

Red: Triangular signal from function generator
Black: Predictor signal 4 bit

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

Red: Triangular signal from function generator
Black: DPCM output 4 bit

DPCM-Demodulator

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

Red: Triangular signal from function generator
Black: Input DPCM demodulator 4 bit

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

Red: Triangular signal from function generator
Black: Predictor signal, demodulator 4 bit

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

Red: Triangular signal from function generator
Black: PAM output DPCM dem. 4 bit

Interpretation