

Function Description of Digital Telemetry

1. Digital Telemetry Systems

1.1 Telemetry Systems with PCM-Technology

For the wireless transmission of several information channels, several different RF transmission frequencies are necessary and as many selective RF receivers, which would increase the cost/benefit ratio disproportionately. The solution of this problem provides the digital transmission technique. The real task of multi-channel telemetry systems is to interlace and bundle the different channels timewise with each other in that way, until they are available as a "2-wire line" to be transferred with one single RF transmitter. This type of transmission is realized by digitizing, multiplexing and PCM encoding of all signal channels. This kind of digital PCM transmission (pulse-code modulation) has been used for many years in the communications technology, eg. worldwide transmission of telephone-channels.

The advantages of PCM transmission speak for themselves:

- constant signal / noise ratio by digitization of the signal
- multiple use of one transmission-channel by multiplexing
- interference free signal transmission, low susceptibility to crosstalk
- direct acquisition and processing of the received digital PCM signal to a PC

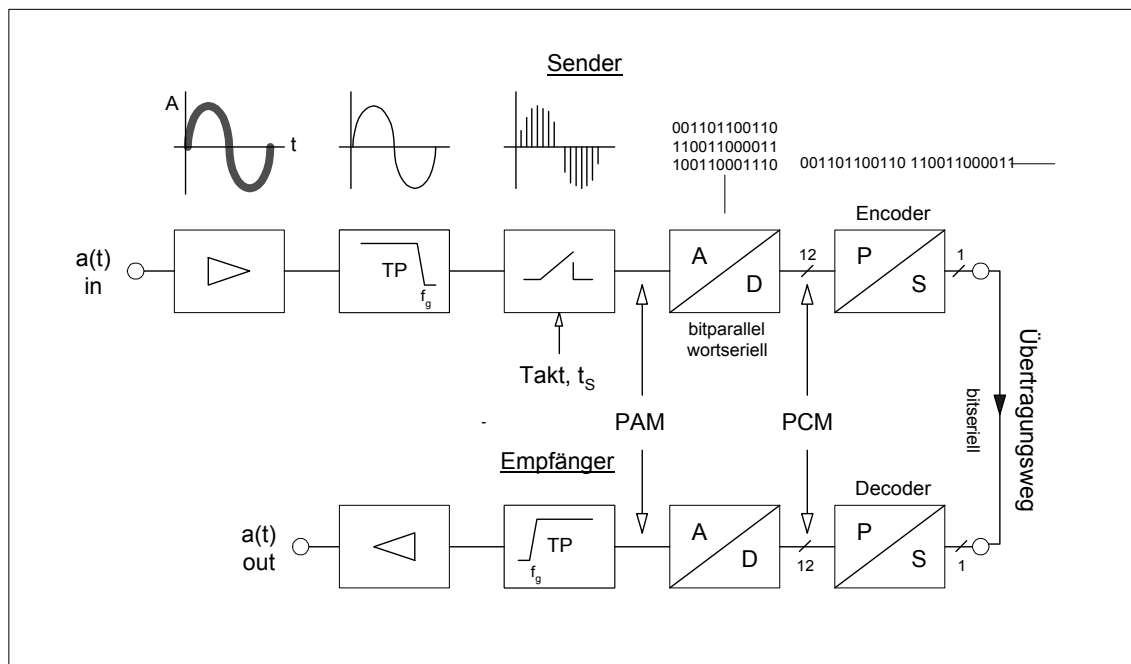


Figure 1: Block diagram of a 1-channel PCM transmission

2. Generation of a PCM signal

The Pulsecode modulation (PCM) plays an ever increasing role in the collection, transmission and analysis of measured values. In the following the production of a digital PCM signal and the process of sampling, quantization and coding are described in more details.

The analog signal $a(t)$ is initially amplified (conditioned), and in its bandwidth filtered (Fig. 1). This is followed by an essential step, the discretization of the continuous measuring signal. An electronic switch (Sample & Hold) - controlled by a clock generator - takes individual samples from the signal, whereas the pulse amplitude in time corresponds to the momentary value of the analog input voltage. The output of the electronic switch is a pulse amplitude modulated signal, the PAM signal.

The sampling theorem specifies the minimum frequency rate an analog signal has to be scanned, so that the original signal can be extracted without loss of information from the re-sampling. The sampling frequency (f_s) must be greater than twice the highest containing frequency in the analog signal (f_g):

$$f_s > 2 \times f_g$$

In practice there are 4 to 5 samples per Hz bandwidth taken. The effect of pulse-amplitude-modulation becomes more clearly by viewing the signals in the time - and frequency domain (Fig. 2). The sampling process produces a sequence of pulses which contains – according to the Fourier Analysis – a DC component and a sum of sinusoidal voltages, which are integer multiples of the fundamental frequency.

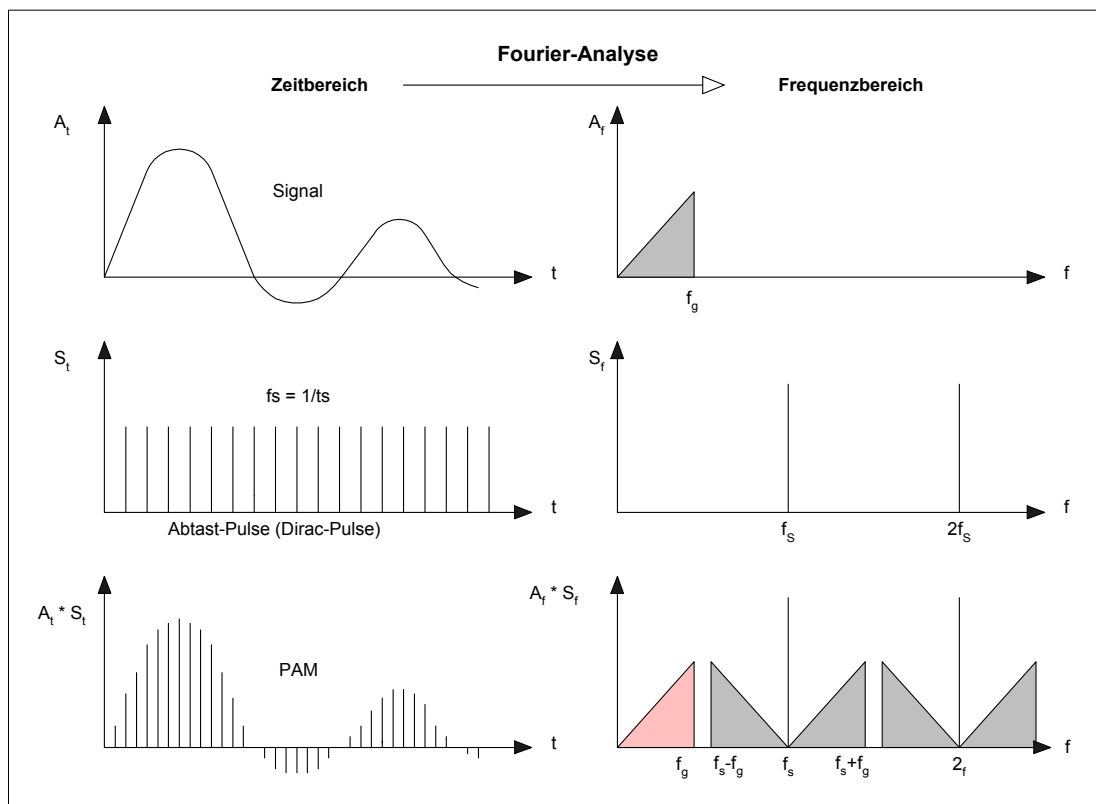


Figure 2: Scanning in time- and frequency domain

In the frequency domain the pulse-samples create systematically spectral-lines at intervals of f_s . At the right and left sides of these carriers arise modulation sidebands (similar to amplitude modulation) with upper and lower sidebands at $f_s - f_g$, $f_s + f_g$, $2f_s - f_g$, $2f_s + f_g$, etc. The signal information is in each side band. For the signal transmission is, however, only the "red" coloured baseband in Figure 2 is used.

In the frequency range it's also evident that by a magnification of the signal-cutoff frequency the modulation sidebands expand and would fall into each other. In this moment the so called "aliasing" appears, which only can be prevented by a higher sampling rate. In practice this problem does not occur because the signalbandwidth is already low pass filtered (by anti-aliasing filter) at the frontend.

The puls amplitude modulated signal in Figure 1 is still an analog frame of the input signal. But the samples can be much better processed in digital form. In order to finalize the quantization and coding, the PAM signal is fed to a 12-bit-A/D-converter. The A/D-converter converts (quantized) each PAM pulse according to their current amplitude into 12-bit words, a digital resolution of 1024 steps.

Therefore a PAM pulse amplitude of 1 volt is digitised with a resolution of <1mV. The digitized PAM signal is called a PCM signal. The 12-Bit-A/D-Wandler follows a parallel/serial converter, which converts the 12-bit words into a bitserial data stream and can be transferred on a data line or via a fiber optic RF line. To enable the receiver to synchronize on to the serial data stream, synchron bits in front of each data word are transferred as well.

On the receiving side the same is done thing in reverse. After the series / parallel conversion, the 12-bit words are converted with the help of D/A-converter in PAM signals and then low pass filtered to smoothed amplitude continuous signals. Each signal value is equal to the average of the corresponding quantization interval. After signal amplification according to the level adjustment, the original measuring signal $a(t)$ is available again.

2.1 Multiplexing and Demultiplexing

Multiplexing allows multichannel, synchronous transmission of several PCM channels. As shown in Figure 1 and Figure 2 the sample pulses take only a very limited amount of time and between the samples are relatively large time gaps. According the time division multiplexing, the 12-bit code words of several transmitted signals can be timed to stagger, so that they do not affect each other, but fill in the free time gaps. This results in a PCM multiplex signal. Figure 3. illustrates the basic principle of the temporal interlacing of several messages (code words) and the transmission over a common line. The time multiplexing process is done complete electronic.

Figure 3 shows four input signals, which are cyclically scanned by a rotating switch A. Synchronous with the consequence of the incoming code words, the switch A is driven on the next input. At the output of the switch A the PCM time multiplex signal is then available. The period in which a code word is transmitted is called "time slot".

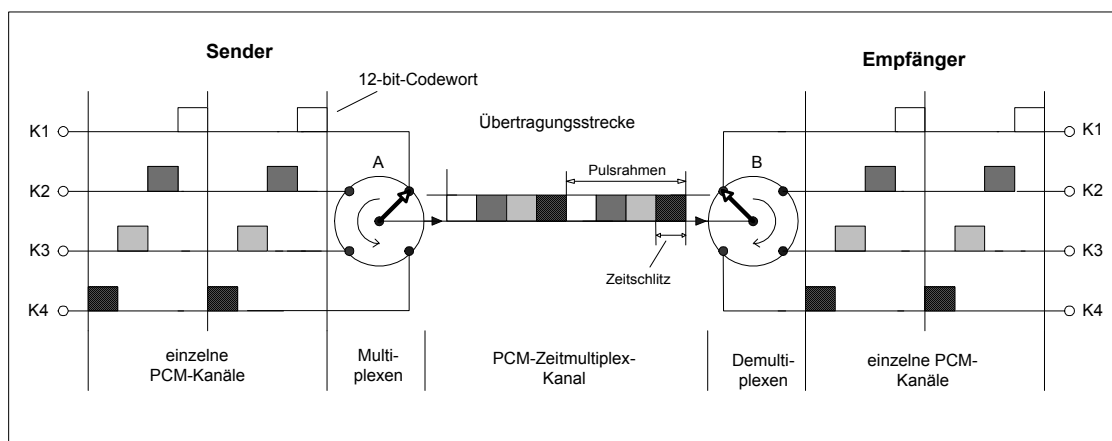


Figure 3: Schematic representation of time multiplex and time demultiplex

A bit sequence, which contains a codeword from each input signal, is called a puls-frame. The example listed in Figure 3 shows a puls frame consisting of four code words from the input signals K1 ... K4. The required sampling frequency of the multiplexer for the complete transfer of digital information is

$$f_s > 2 \times f_g \times \text{numbers of channels}$$

On the receiving side, the individual PCM signals are recovered from the time division multiplex signal, i.e. the 12-bit code words are applied to the corresponding outputs. The rotating switch B distributes synchronous the code words to the four outputs. Like the time multiplexing at the transmitter side, all operations are executed complete electronic.

3. Multichannel telemetry systems

Figure 4 and 5 show the basic construction of a multi-channel telemetry system, as an example with 4 transmission channels. The PCM transmission systems consists of two units, a PCM encoder for recording and coding of the signal values on the transmit side and a PCM decoder to decode and output the measured values on the receiving side.

Functions of the encoder:

- Signal processing of the analog input signal (sensor signal)
- Bandwidth limitation by Low Pass-Filter
- Simultaneous sampling by a Sample & Hold Amplifier
- A/D conversion (ADC) of the sampled signal
- Parallel-serial conversion of the 12-bit words
- Insertion of synchronization bits (marks)
- Conversion into a PCM code
- FSK modulation of RF transmitter

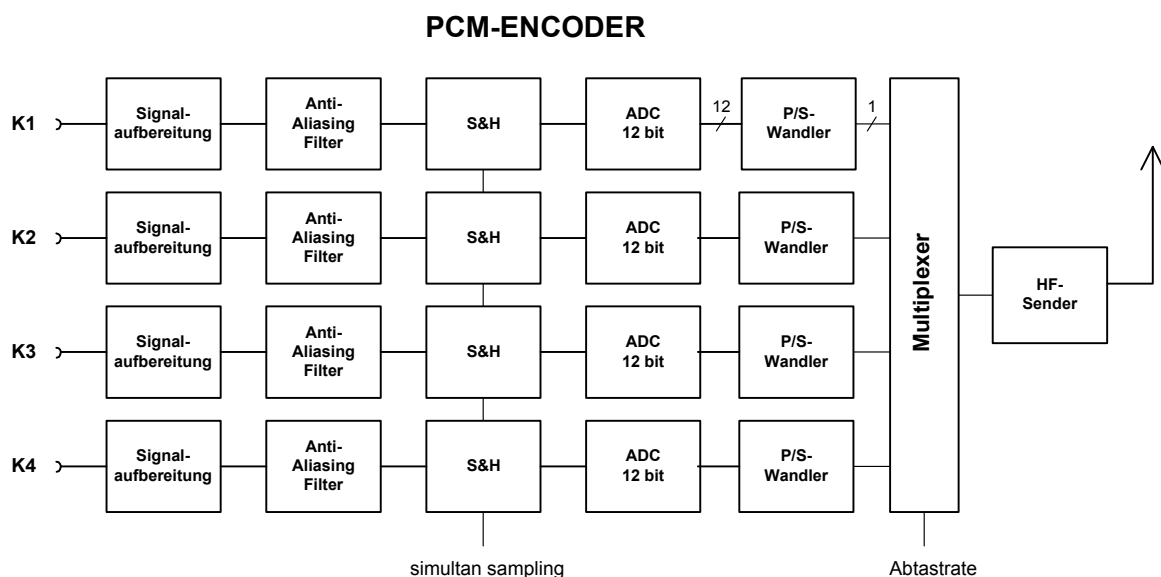


Figure 4: Block diagram from a multichannel PCM encoder

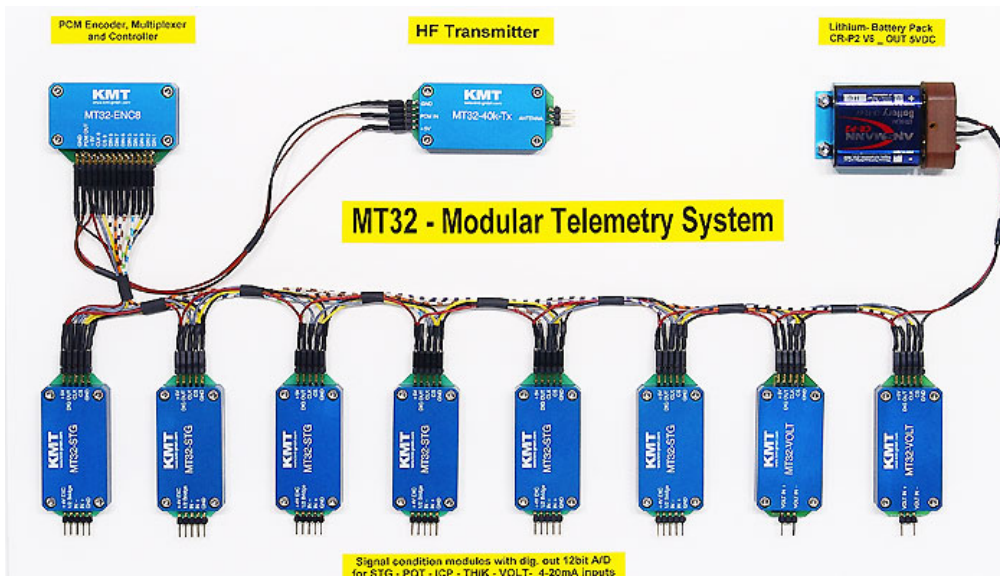


Figure 4.1: MT32, 8-channel modular telemetry system with signal conditioning, analog to digital converters, encoder and RF-Transmitter (433/866MHz)

The decoder at the receiver side performs the following tasks:

- selective amplification and demodulation of the RF signal
- Regeneration of the incoming serial PCM signal
- Generation of an input signal to the synchronous clock
- Detection of synchronization bits and generate the corresponding signal addresses
- Output data in bit-parallel, word-serial formate to a PC interface card (IF16)

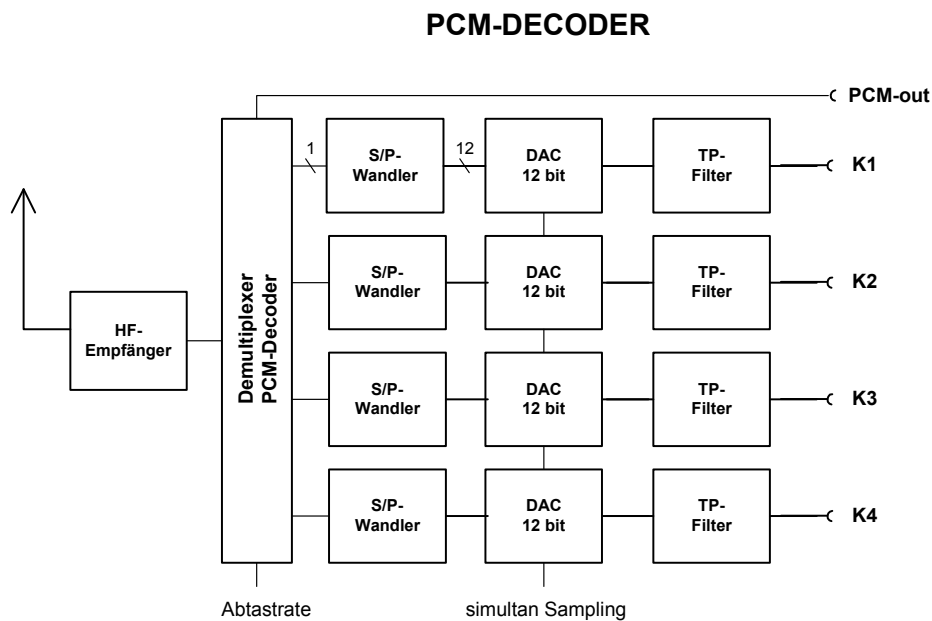


Figure 5: Block diagram of a PCM multi-channel decoder



Figure 5.1: MT32, 8-channel telemetry cecoder with 8 analog signal outputs (+/-5V)

3.1 Synchronization of the encoder and decoder

Thus the decoder is able to recognize the timing of the digitized measured values, a so-called synchronous word is inserted. This sync word consists of a fixed length of 4 bits and is inserted at the beginning of each PCM Pulsframe. The decoder is synchronized to this sync word and is always in exact synchronism with the corresponding transmitter. In addition, the synchronous word provides about its encoding other useful information, such as the battery capacity of the transmitter.

Figure 6 shows the structure of a single pulsframe, consisting of sync word and 4 channels. The length of the serial PCM Pulsrahmens is $4 \times 12 \text{ bits} + 4 \text{ bits} = 52 \text{ bits}$.

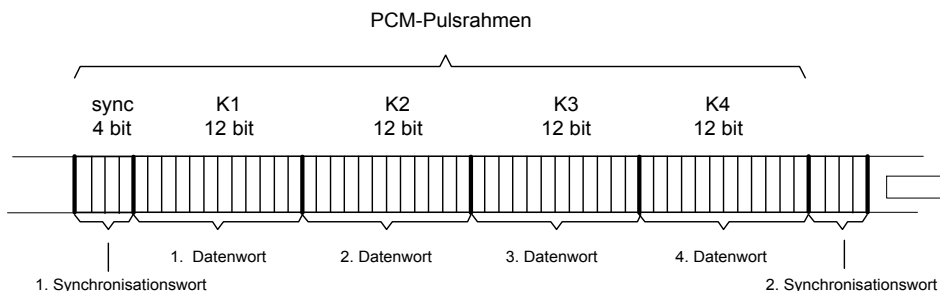


Figure 6: PCM frame with sync word, example: 4 channels

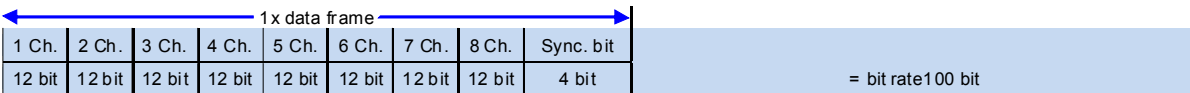
The max. transmitted signal bandwidth of each channel is directly related to the multiplexer scanning speed. Table 1 gives an overview of the achievable signal bandwidth (S_b) as a function of sampling rate (A_{bt}) of the PCM system. The actual frame length (see Figure 6) is calculated from the number of transmitted channels. Thus, the transmission rate of a telemetry system is the product of sampling rate and frame length.

$$\text{transmission rate (bit/s)} = \text{sample rate (Hz)} \times \text{frame length (bit)}$$

Cut off frequency from anti-aliasing filter (-3dB) and scanning rate (red)					
Bit rate	2 Channels	4 Channels	8 Channels	16 Channels	32 Channels
2560 kbit/s	24000Hz (91428 Hz)	12000 Hz (49231 Hz)	6000 Hz (25600 Hz)	3000 Hz (13061Hz)	1500 Hz (6598Hz)
1280 kbit/s	12000 Hz (45714 Hz)	6000 Hz (24615 Hz)	3000 Hz (12800 Hz)	1500 Hz (6530 Hz)	750 Hz (3298 Hz)
640 kbit/s	6000 Hz (22857Hz)	3000 Hz (12308 Hz)	1500 Hz (6400 Hz)	750 Hz (3265 Hz)	375 Hz (1649 Hz)
320 kbit/s	3000 Hz (11428 Hz)	1500 Hz (6154 Hz)	750 Hz (3200 Hz)	375 Hz (1632 Hz)	190 Hz (824 Hz)
40 kbit/s	375 Hz (1428 Hz)	190 Hz (770 Hz)	95 Hz (400 Hz)	47 Hz (204 Hz)	23 Hz (103 Hz)

Scanning rate, signal bandwidth and frame length depending on bit rate and number of channels

Frame example with 8 channels as following: 8Ch x12 bit = 96 bit + 4 bit sync. = 100 bit



- 32 Ch. x 12 bit = 384 bit + 4 bit sync. = 388 bit
- 16 Ch. x 12 bit = 192 bit + 4 bit sync. = 196 bit
- 8 Ch. x 12 bit = 96 bit + 4 bit sync. = 100 bit
- 4 Ch. x 12 bit = 48 bit + 4 bit sync. = 52 bit
- 2 Ch. x 12 bit = 24 bit + 4 bit sync. = 28 bit

Scanning you can calculate e.g.: 40kbit transfer rate, 8 Ch. = 40000 : 100bit = 400Hz per Ch.

Table 1: Relationship between transmission rate, numbers of channels and PCM-frame length

$$\text{Transmission rate (bit/s)} = \text{Sample Rate (Hz)} \times \text{frame length (bit)}$$

Transfer rate calculations, e.g. for 8 channels:

$$8 \times 12 \text{ bit} = 96 \text{ bit} + 4 \text{ bit sync.} = 100 \text{ bits}$$

$$\text{Bitrate} = 6400\text{Hz} \times 100 \text{ bit} = 640 \text{ kbit/}$$

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July 2009

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