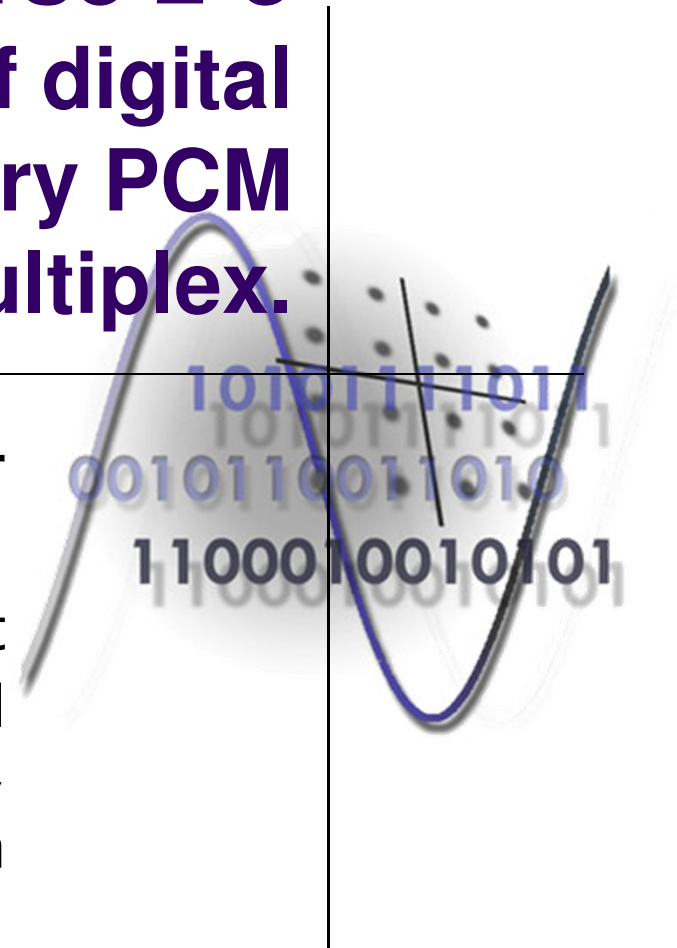


Course 2-3

Fundamental notions of digital telephony. The primary PCM multiplex.

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Cotent of the course



- Fundamental notions of digital telephony;
 - PCM modulation;
 - Delta modulation;
- The primary PCM multiplex;
 - The primary E1 multiplex;
 - The primary T1 multiplex;
 - Frame synchronization;
 - Alarms;
 - Line interfaces;
- Data terminal – multiplexer interfaces;

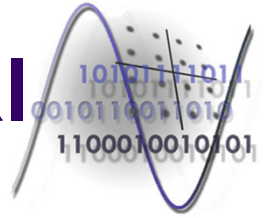
A/D conversion of the voice signal



- The transmission technique used in digital telephone networks:
 - PCM (Pulse Coded Modulation);
 - represents a non-uniform A/D conversion with 8 bits/sample followed by the transmission on the channel of the bits associated to the code words;
 - The bit rate obtained for a telephone channel is 64kbps;
 - More advanced voice coding techniques can ensure a significant reduction of the necessary bit rate;
 - ADPCM (Adaptive Differential PCM) and parametric coding techniques – take into account the characteristics of the voice signal;
 - it can be used just for the coding of the vocal signal – it is not possible the data transmission by using a modem on a network that uses such coding;
 - voice coding techniques:

ITU-T standard	Coding method	Coded signal bit rate (kbps)
G.711	PCM	64
G.721	ADPCM	32, 16, 24, 40
G.728	LD-CELP	16
G.729	CS-ACELP	8
G.723.1	Multirate CELP	6.3, 5.3

A/D conversion of the voice signal



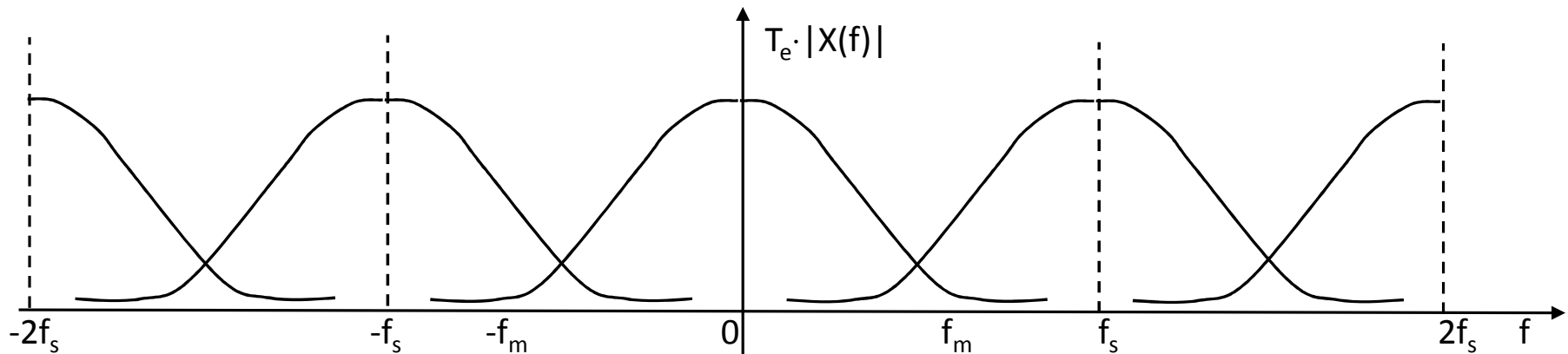
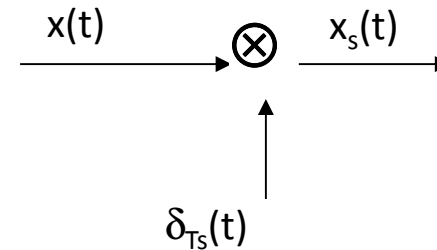
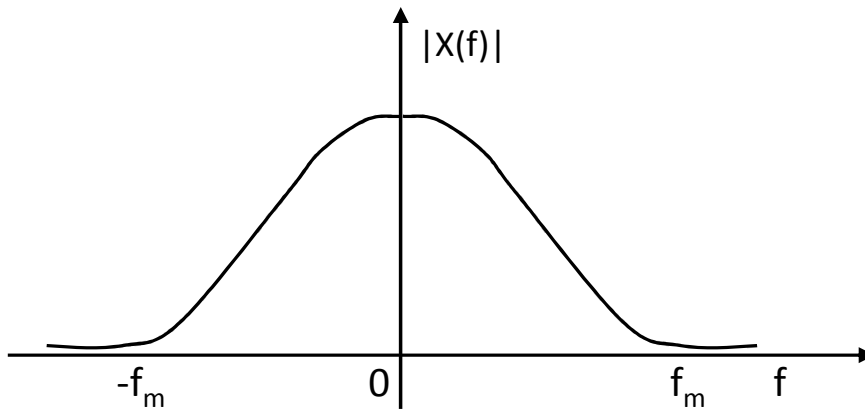
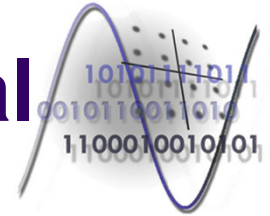
- Processing required by PCM:
 - Sampling;
 - Quantization;
 - Coding;
- The sampling theorem – basic relations, the aliasing phenomenon:

$$x_s(t) = x(t) \cdot \delta_{T_s}(t) = x(t) \cdot \sum_{n=-\infty}^{\infty} \delta(t - nT_s) = \sum_{n=-\infty}^{\infty} x(nT_s) \cdot \delta(t - nT_s)$$

$$X_s(\omega) = \frac{1}{T_s} \cdot \sum_{k=-\infty}^{\infty} X(\omega - k \cdot \omega_s)$$

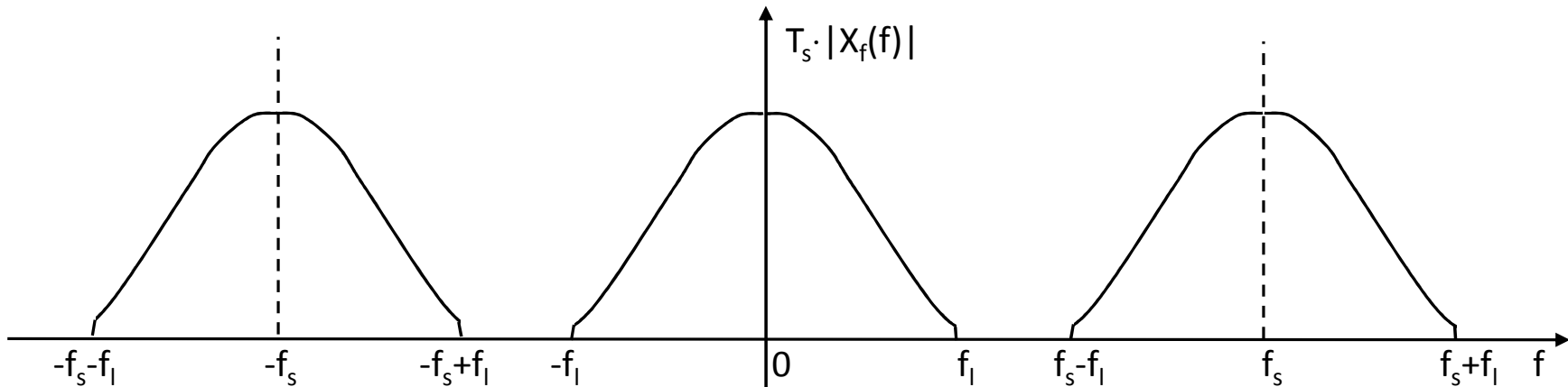
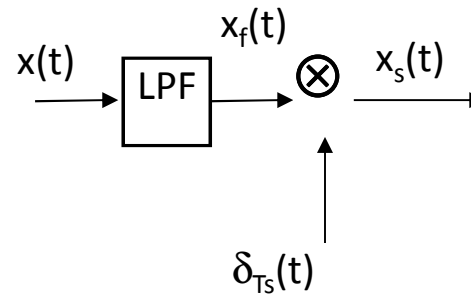
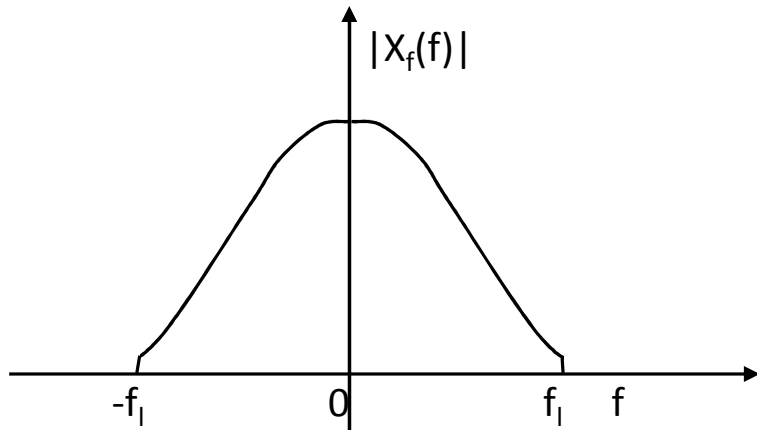
- Shannon's theorem:
 - Any signal $x(t)$ with finite input domain spectral density function $X(\omega)$ ($X(\omega)=0, \forall |\omega| > \omega_M$) is completely defined by its samples $\{x(nT)\}$, if $T=(\pi/\omega_M)$;

A/D conversion of the voice signal



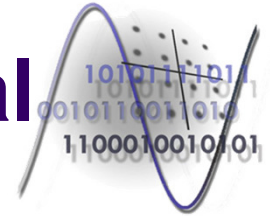
Spectral properties of the sampled signals and the aliasing effect.

A/D conversion of the voice signal



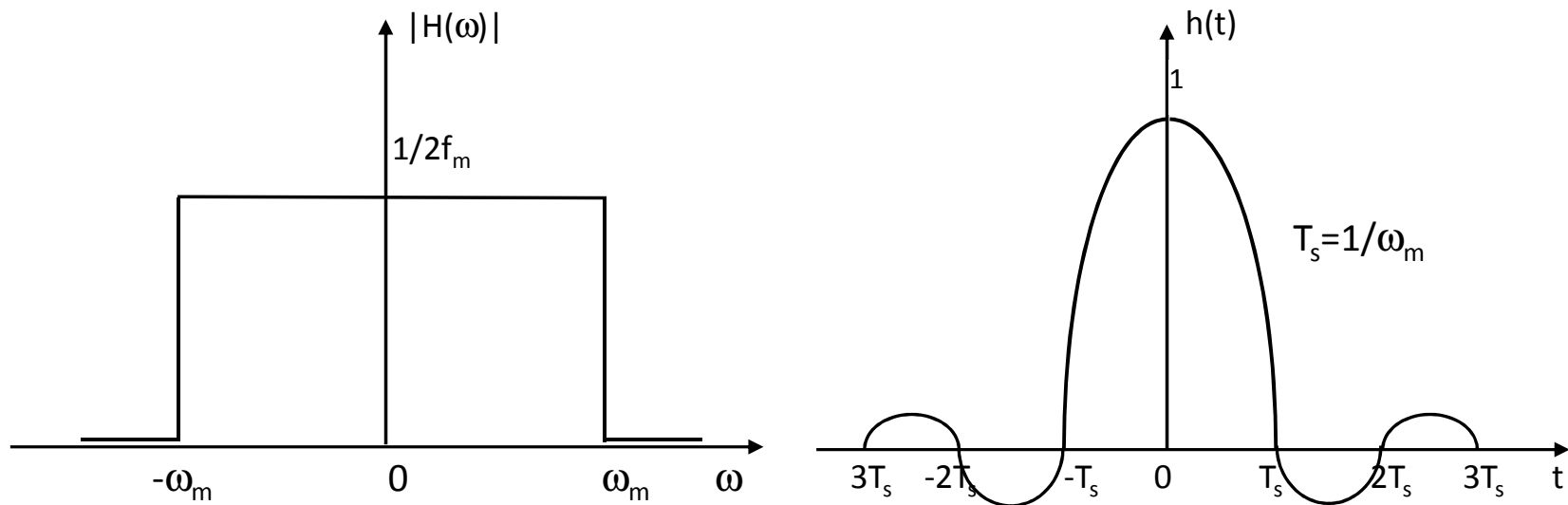
Spectral properties of the sampled signals and the suppression of the aliasing effect.

A/D conversion of the voice signal



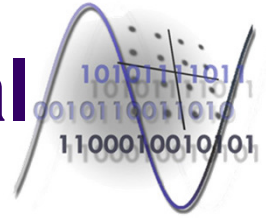
- Reconstruction of the sampled signals using LP filtering;
 - basic relation for ideal filtering:

$$H(\omega) = \begin{cases} T_s = \frac{1}{2f_M}, \forall |\omega| < \omega_M \\ 0, \forall |\omega| > \omega_M \end{cases} ; h(t) = \text{sinc}(\omega_M t)$$

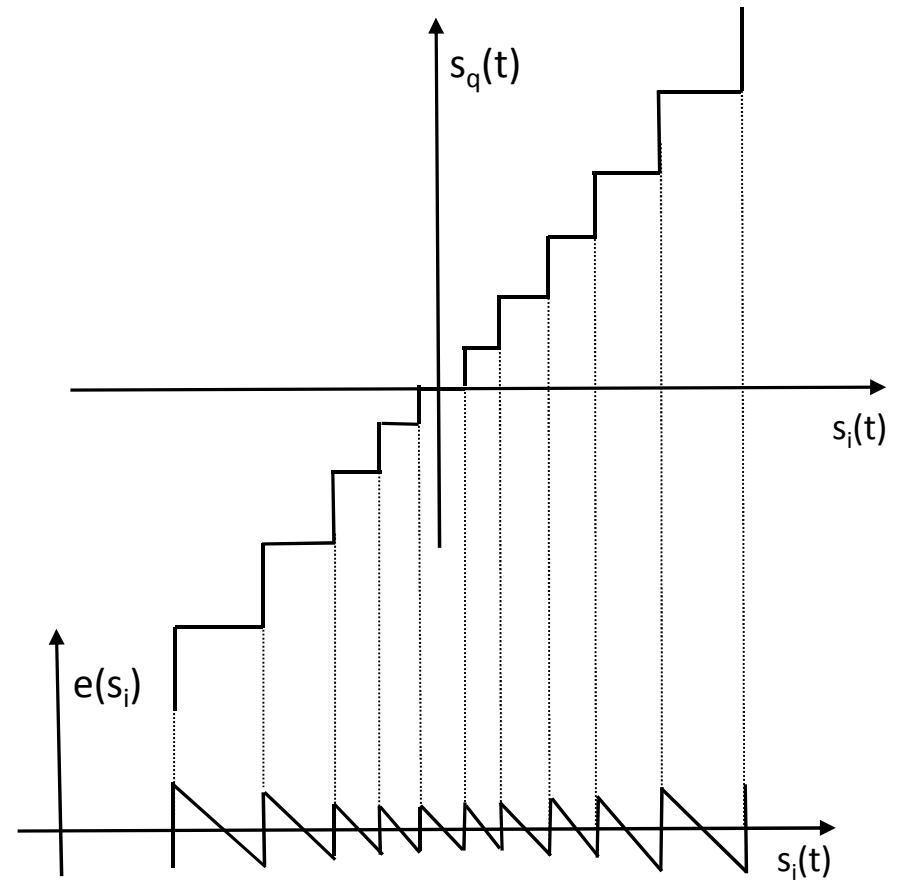
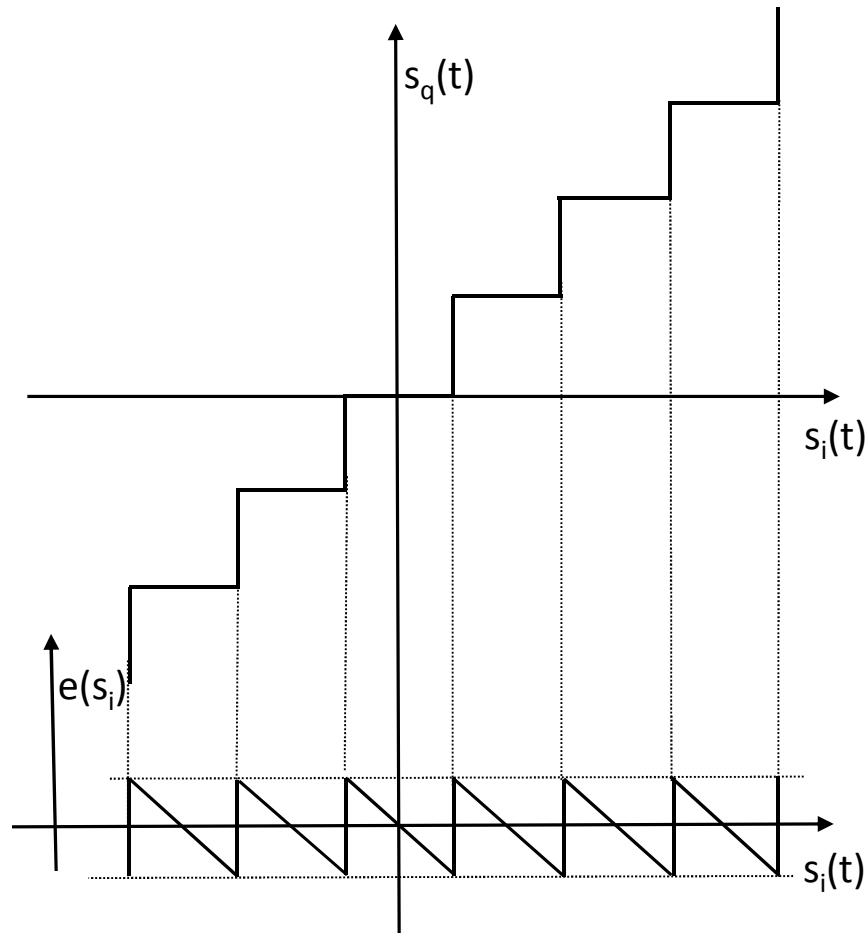


Frequency characteristic and impulse response of an ideal reconstruction filter.

A/D conversion of the voice signal



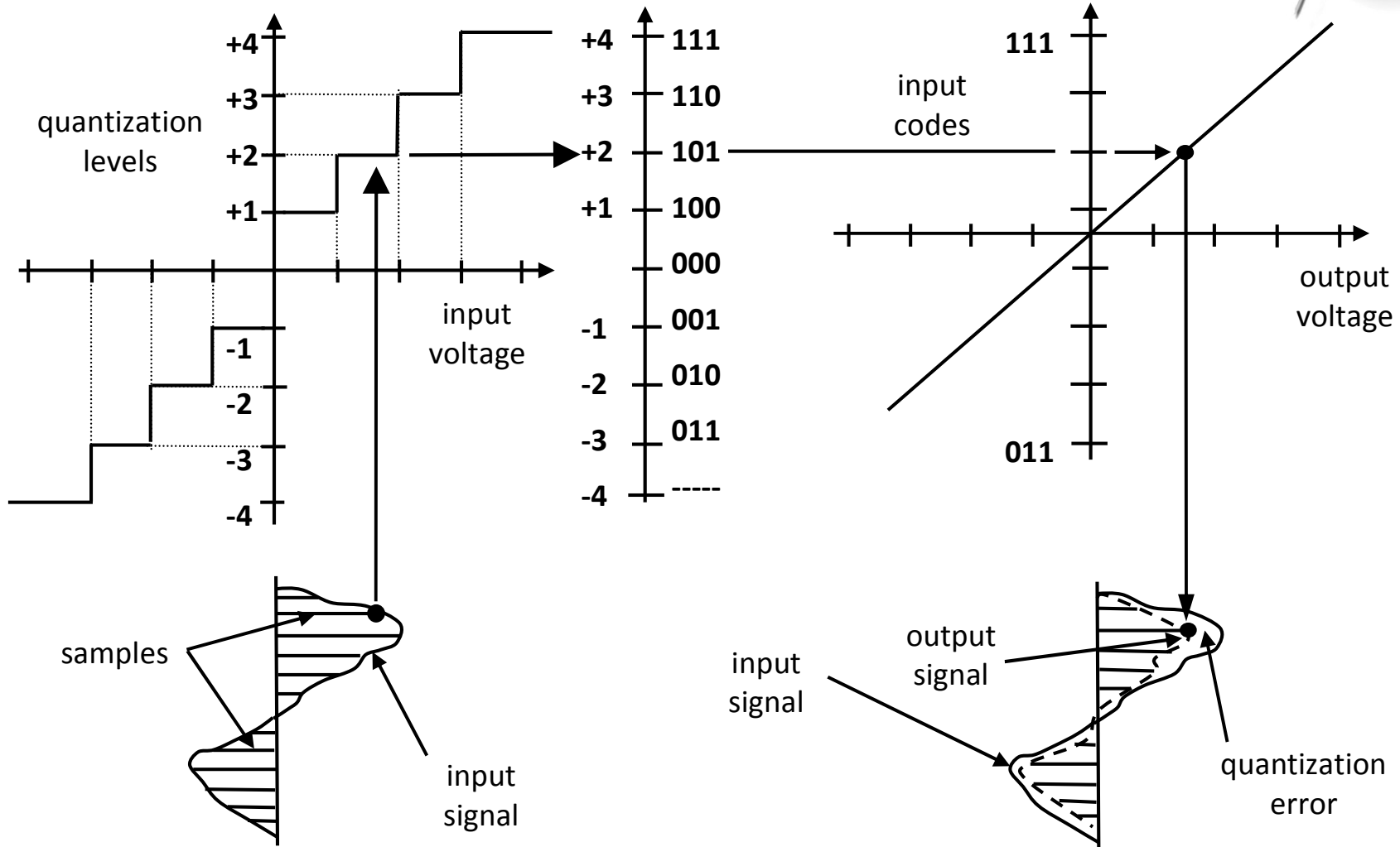
- The uniform and non-uniform quantization techniques;



Quantization techniques: a) uniform quantization

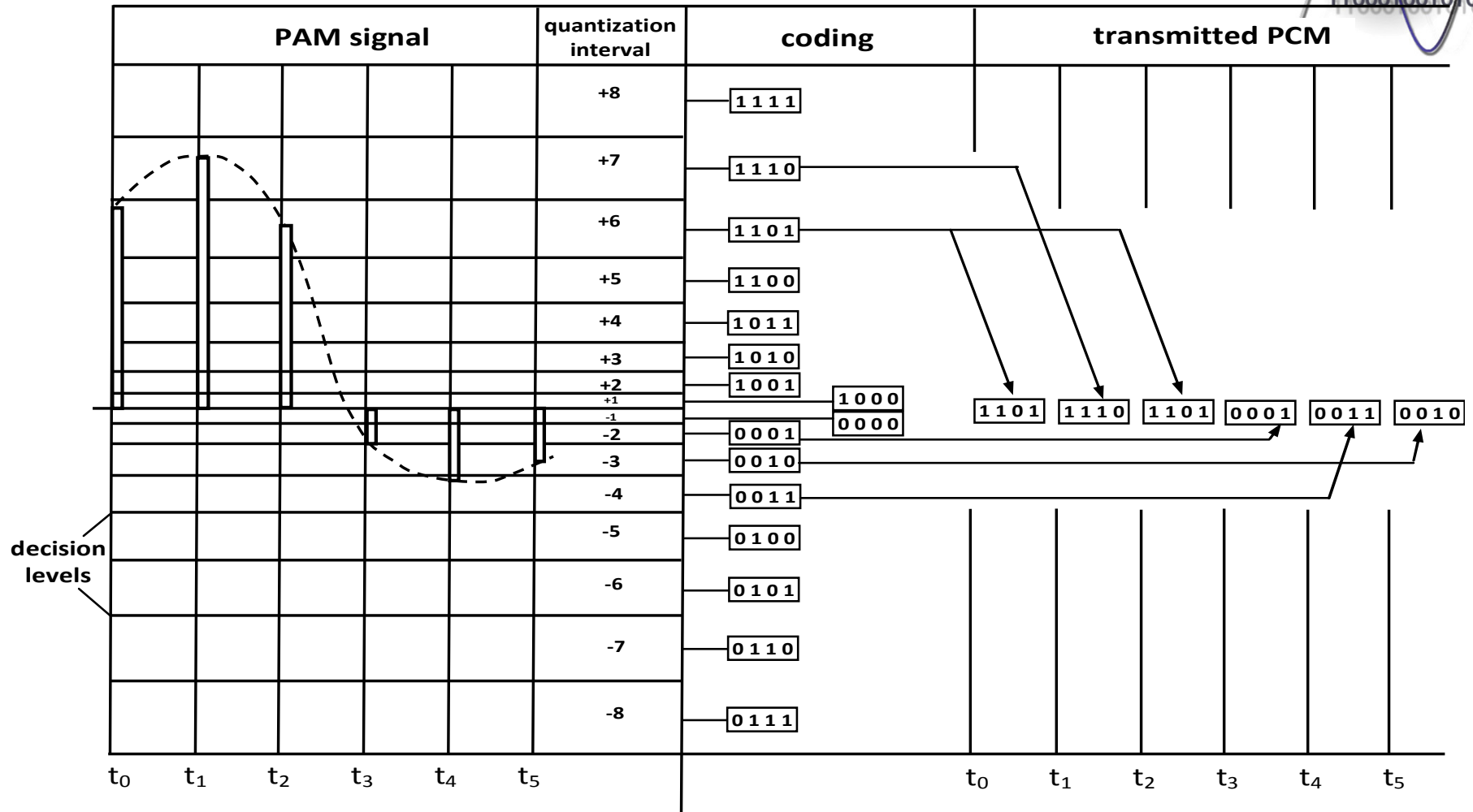
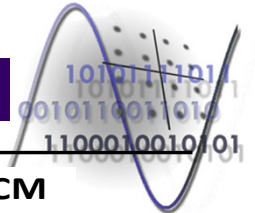
b) non-uniform quantization

A/D conversion of the voice signal



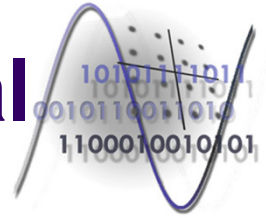
Explanation of PCM coding and decoding process in the case of uniform quantization with 3 bits/sample

A/D conversion of the voice signal



Explanation of the PCM coding and decoding process in the case of non-uniform quantization with 3bits/sample

A/D conversion of the voice signal



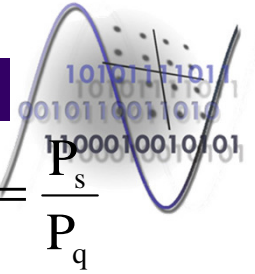
- The quantization signal to noise ratio;
 - The quantization error is considered a noise signal;
 - A general expression of the mean quantization noise power is given by:

$$P_q = \sum_{i=1}^N p_i \cdot P_{qi}$$

- N is the number of quantization intervals;
- p_i is the probability that the transmitted signal is located in the i^{th} interval;
- P_{qi} is the power of the quantization noise in interval i .
- If the dynamic range of the transmitted signal is $2V$ and the width of the quantization intervals are Δ_i , then the p_i probabilities are given by: $p_i = \frac{\Delta_i}{2V}$
- Considering the uniform distribution of the quantization error inside a quantization interval, the power of the quantization noise in interval i is given by:

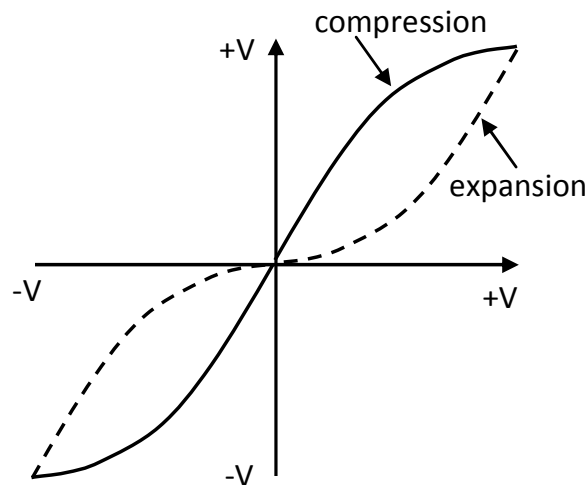
$$P_{qi} = \int_{-\frac{\Delta_i}{2}}^{+\frac{\Delta_i}{2}} \frac{1}{\Delta_i} \cdot e_r^2 \cdot de_r = \frac{\Delta_i^2}{12}$$

A/D conversion of the voice signal

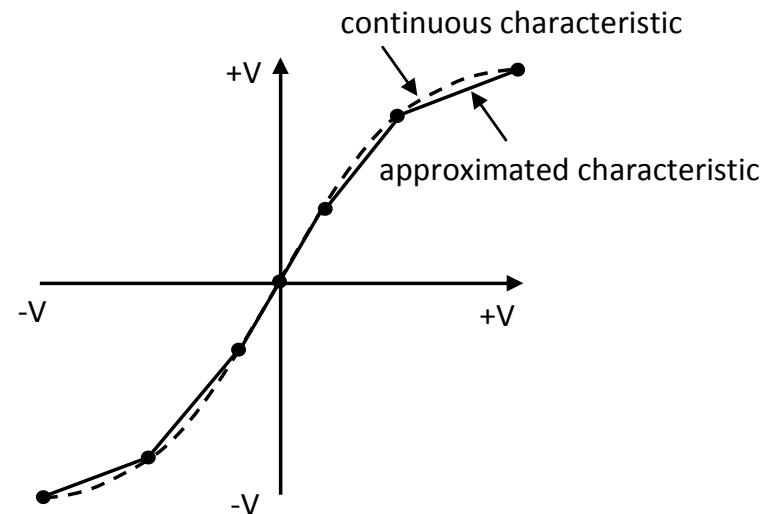


$$\text{SNR}_q = \frac{P_s}{P_q}$$

- The signal/quantization noise ratio, SNR_q , is defined as: $\text{SNR}_q = \frac{P_s}{P_q}$
- Implementation of the non-uniform quantization:
 - Usage of converters with non-uniform quantization;
 - Usage of converters with uniform quantization combined with compression / expansion circuits;
 - the compression/expansion characteristics could be continuous or segmented;



a) Continuous characteristic



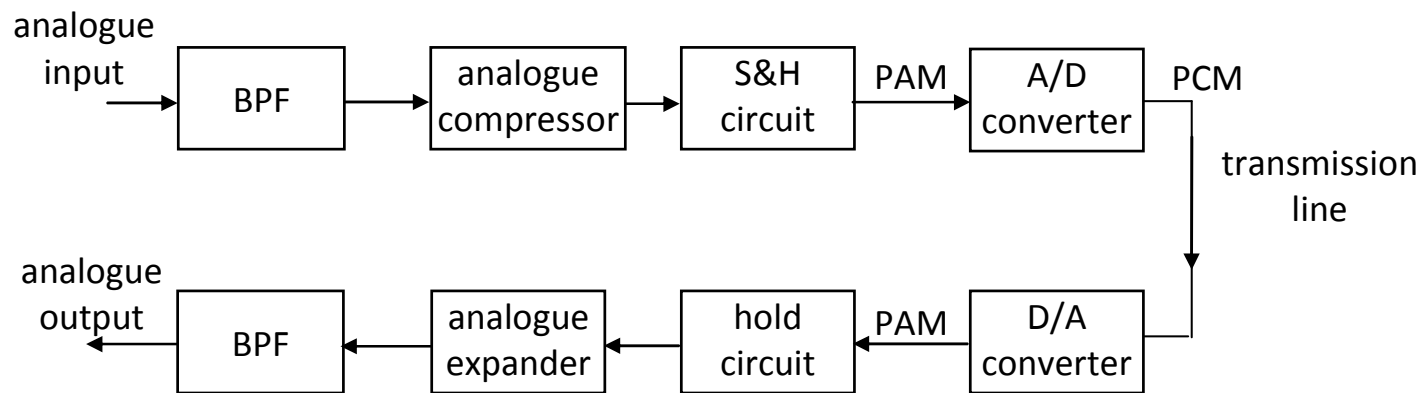
b) Segmented characteristic

Continuous and segmented compressed characteristics

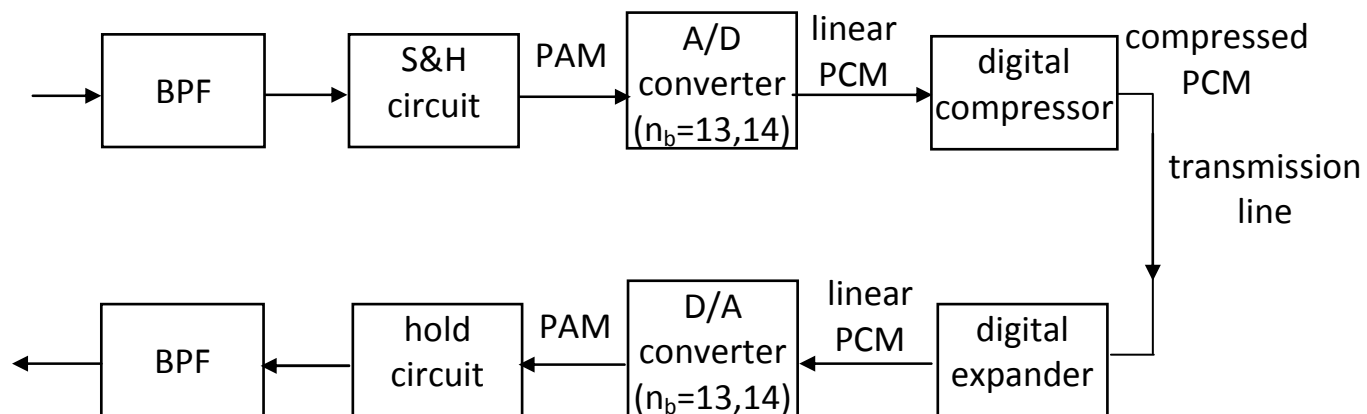
A/D conversion of the voice signal



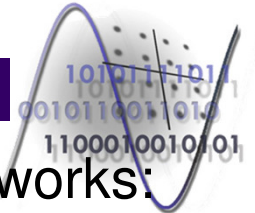
- The implementation of the non-uniform quantization can be achieved by using analogue or digital compression;
 - Processing chain of the PCM coding-decoding for analogue compression;



- Processing chain of the PCM coding-decoding for digital compression;



A/D conversion of the voice signal



- Compression and expansion laws used in digital telephone networks:

- the μ compression law is described by the following relation:

$$y = \frac{\text{sgn}(x) \cdot \ln(1 + \mu \cdot |x|)}{\ln(1 + \mu)} \quad ; \quad -1 \leq x \leq 1 \quad \quad x = \text{sgn}(y) \cdot \frac{1}{\mu} \cdot \left[(1 + \mu)^{|y|} - 1 \right] \quad ; \quad -1 \leq y \leq 1$$

- the A compression law is described by the following relation for input values $x \geq 0$:

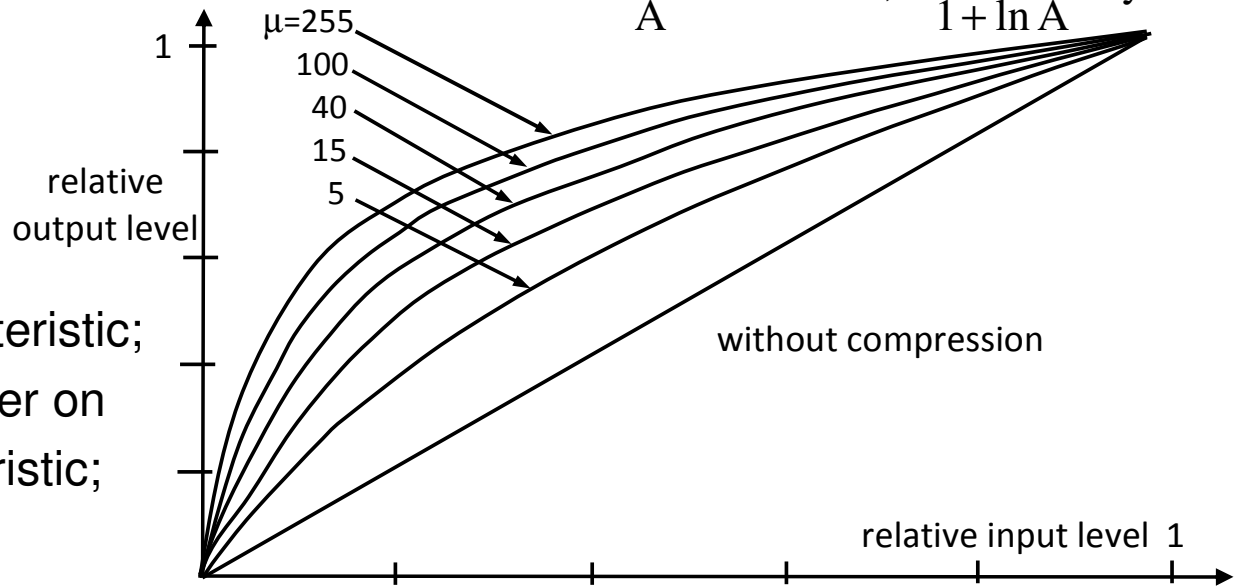
$$x = \frac{y \cdot (1 + \ln(A))}{A} \quad ; \quad 0 \leq y \leq \frac{1}{1 + \ln A}$$

$$x = \frac{\exp(y \cdot (1 + \ln(A))) - 1}{A} \quad ; \quad \frac{1}{1 + \ln A} \leq y \leq 1$$

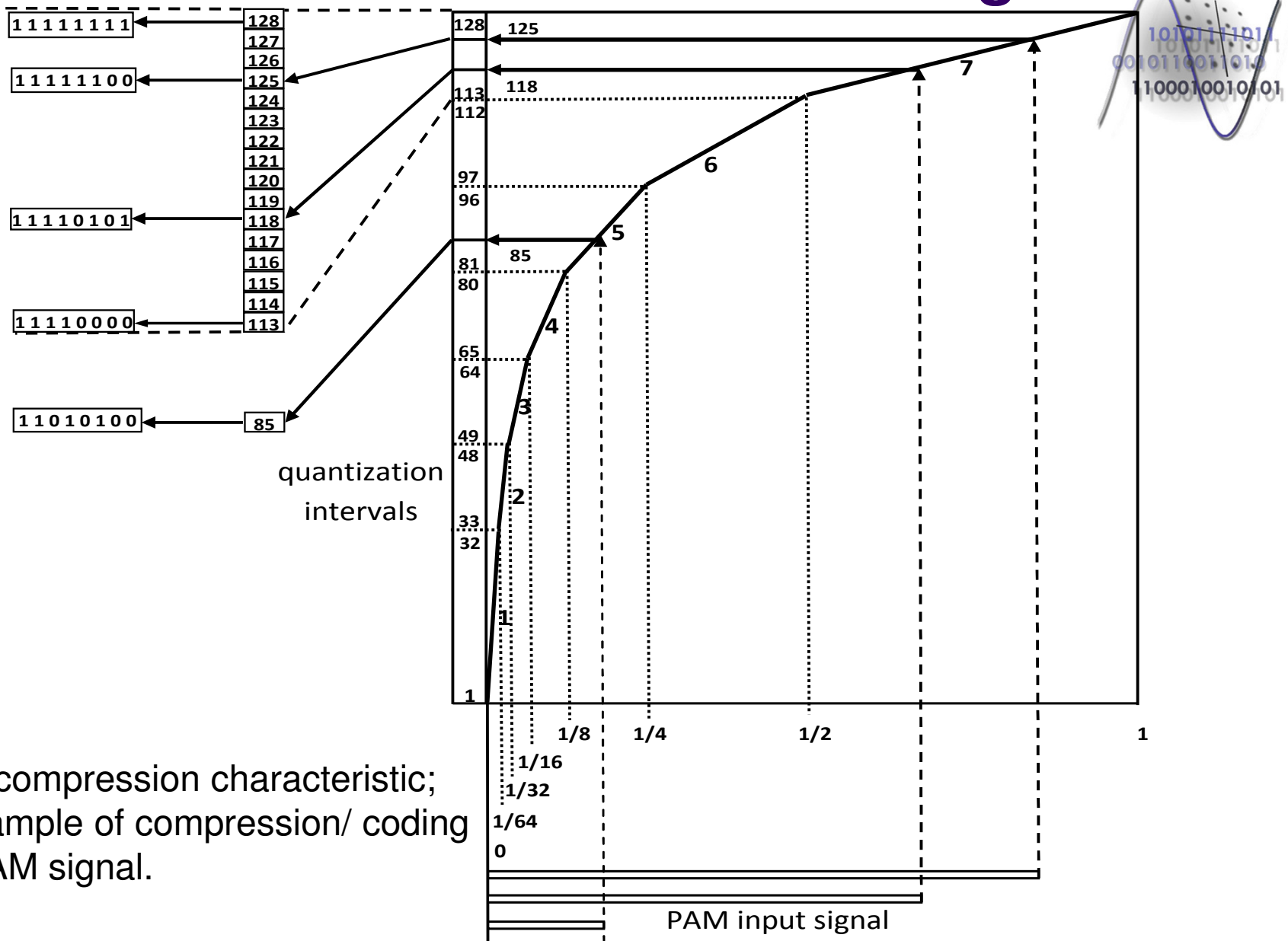
$$y = \frac{1 + \ln(Ax)}{1 + \ln(A)} \quad ; \quad \text{for } \frac{1}{A} < x < 1$$

$$y = \frac{Ax}{1 + \ln(A)} \quad ; \quad \text{for } 0 < x < \frac{1}{A}$$

- The μ compression characteristic;
- The influence of μ parameter on the compression characteristic;

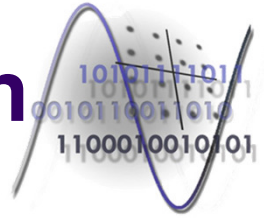


A/D conversion of the voice signal



- The A compression characteristic;
- An example of compression/ coding of a PAM signal.

DPCM modulation

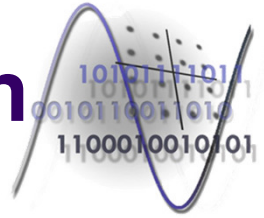


- Disadvantages of the PCM modulation:
 - Large transmission bandwidth – low spectral efficiency;
 - The decrease of transmission bandwidth can be realized by exploiting the correlation between the samples of the transmitted signal;
 - the use of the correlation between the samples of the signal represents the basic idea of the differential PCM modulation;
- DPCM modulation:
 - The next sample is predicted based on the previous samples and it is coded (quantized) only the difference between the current sample, $x(kT_e)=x_k$, and the predicted sample, $x^{\wedge}_k(kT_s)=x^{\wedge}_k$;
 - If the difference signal has a smaller dynamic range than that of the source signal the quantization can be performed on a smaller number of bits;
 - the transmission rate can be reduced;

$$d_k = x_k - x_{k-1}$$

$$\overline{d_k^2} = \overline{(x_k - x_{k-1})^2} = \overline{x_k^2 + x_{k-1}^2 - 2x_k \cdot x_{k-1}} = \overline{2x_k^2} - \overline{2x_k \cdot x_{k-1}}$$

DPCM modulation



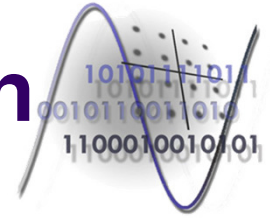
- It is defined a correlation coefficient (or correlation factor) C :

$$C = \frac{\overline{x_k \cdot x_{k-1}}}{\overline{x_k^2}} \Rightarrow \overline{d_k^2} = 2\overline{x_k^2} \cdot (1 - C)$$

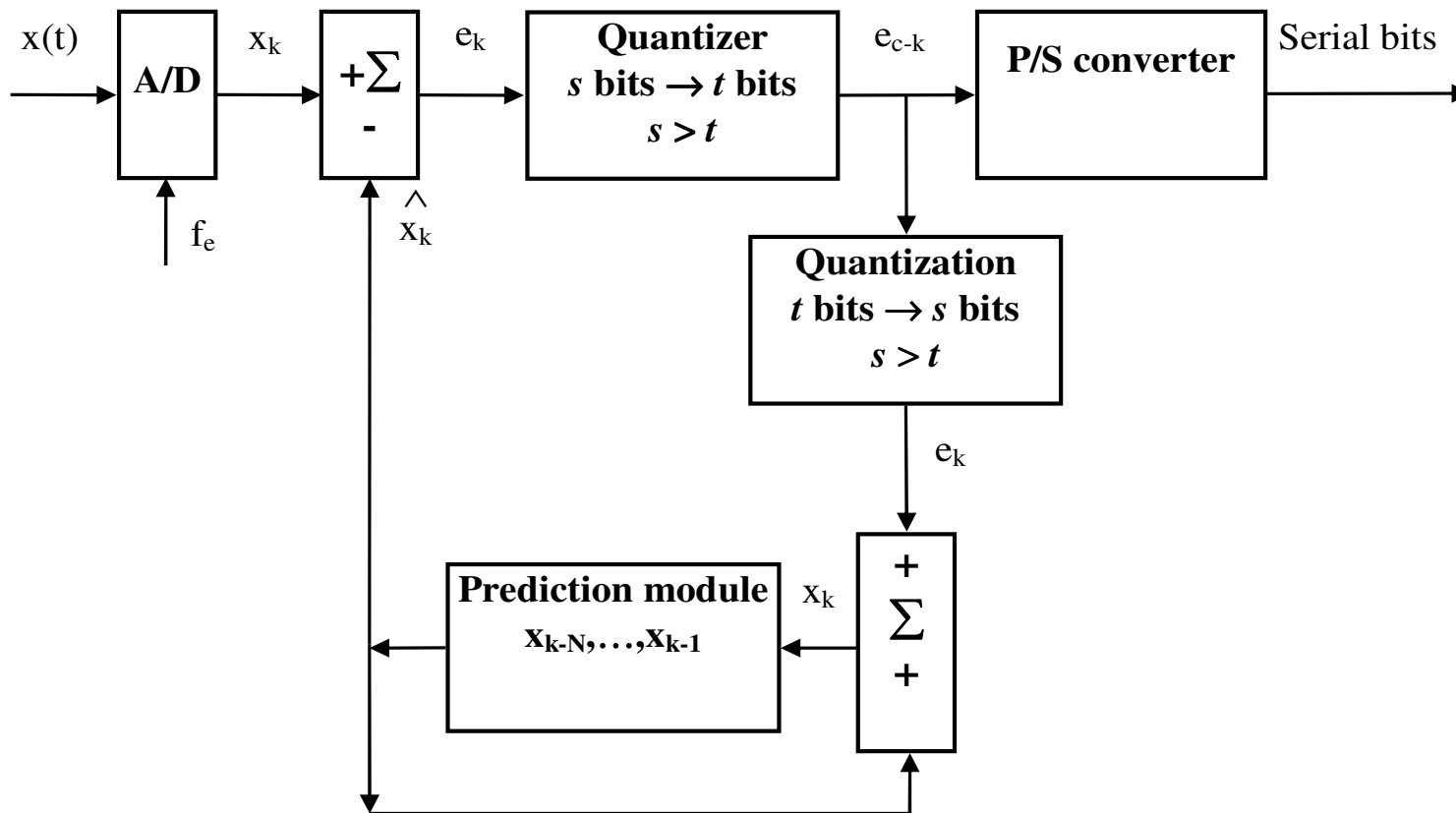
$$\begin{cases} 1. \text{ if } C > 0.5 \Rightarrow \overline{d_k^2} < \overline{x_k^2} \\ 2. \text{ if } C < 0.5 \Rightarrow \overline{d_k^2} > \overline{x_k^2} \end{cases}$$

- If $C < 0.5$ it does not worth to use DPCM;
 - the samples are decorelated and the bit rate decrease is small;
- If $C > 0.5$ it does worth to use DPCM;
 - the samples are correlated and the bit rate decrease is significant;
- Disadvantages of DPCM relatively to PCM:
 - it is more complex – it is required a prediction circuit of the current sample based on the previoius ones;
 - it can not be used with the same parameters for voice and data;
 - if there are errors on the line there are affected several samples.

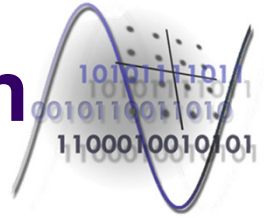
DPCM modulation



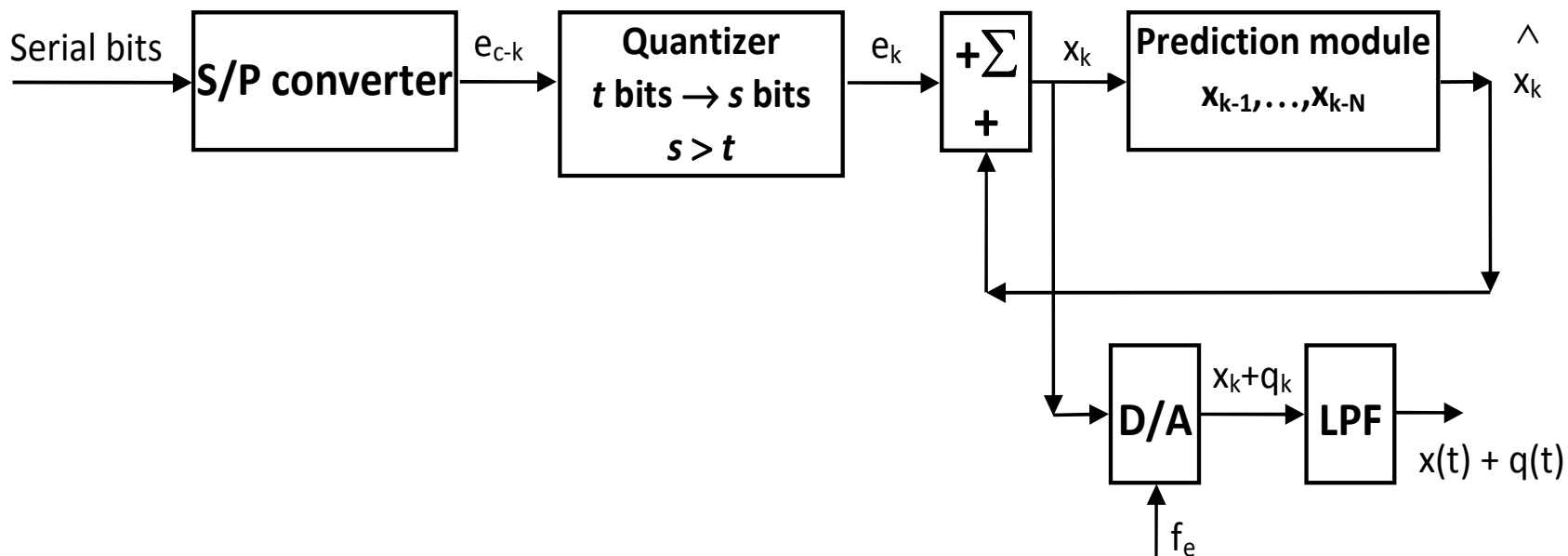
- Block schematic of the DPCM coder;
 - The prediction circuit works with N samples;



DPCM modulation



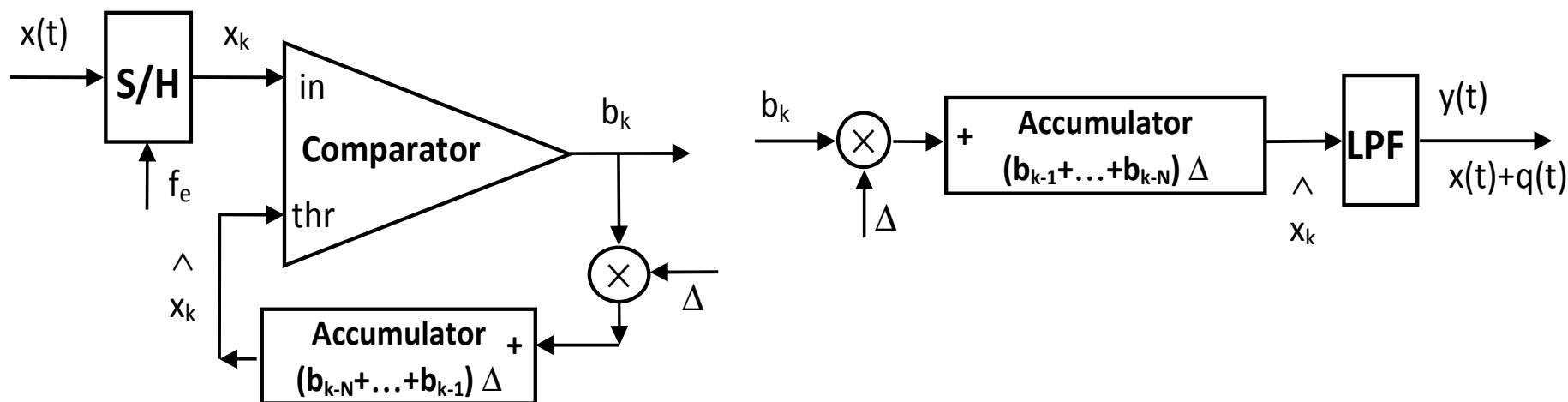
- Block schematic of the decoder;
 - The predictor works with N samples;
 - q_k – quantization errors.



Delta modulation



- The (non-adaptive) Delta modulation;
 - Particular case of DPCM modulation;
 - The signal quantization is performed on a single bit;
 - It is necessary a strong correlation between the consecutive samples;
 - The computation of the predicted signal is realized based on some fixed methods independent of the variation law of the previous samples;
 - Block schematic of the modulator and demodulator:



Delta modulation



- Basic relations;

- Describe the computation of the current bit, of the predicted signal and of the current quantization step:

- computation of the current transmitted bit:
$$\begin{cases} x_k \geq \hat{x}_k = y_k \Rightarrow b_k = '1' (+1) \\ x_k < \hat{x}_k = y_k \Rightarrow b_k = '0' (-1) \end{cases}$$

- computation of the predicted signal (the equation of the accumulator):

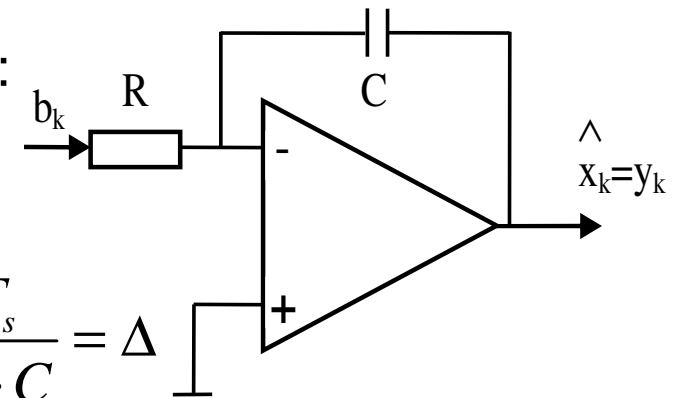
$$y_{k+1} = y_k + b_k \cdot \Delta_k$$

- computation of the quantization step: $\Delta_k = f(\Delta_{k-1}, b_k, b_{k-1}, \dots, b_{k-N})$

- In the case of non-adaptive Delta modulation the quantization step is constant = Δ ;

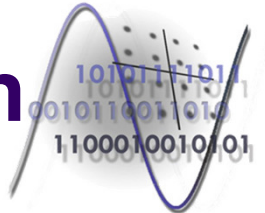
- Implementation methods of the accumulator:

- analog implementation of the accumulator:



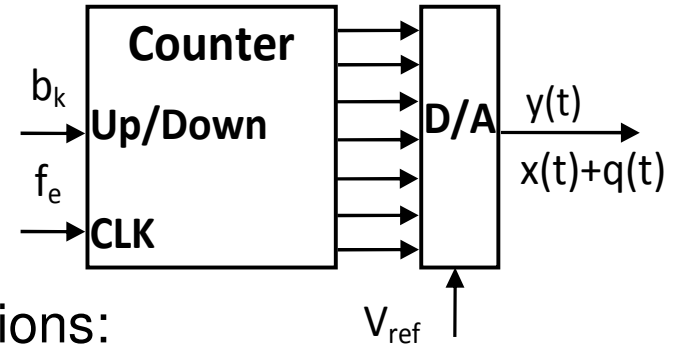
$$|\Delta V_c| = \frac{1}{R \cdot C} \int_0^{T_s} b_k dt = \frac{b_k \cdot T_s}{R \cdot C} = \Delta; \text{daca } b_k = 1 \Rightarrow \frac{T_s}{R \cdot C} = \Delta$$

Delta modulation



- digital implementation using a counter and a D/A converter:
 - N – the number of bits of the D/A converter;

$$\Delta_{Delta} = \Delta_{D/A\ converter} = \frac{V_{ref}}{2^N}$$



- Distortions which affect the Delta modulations:

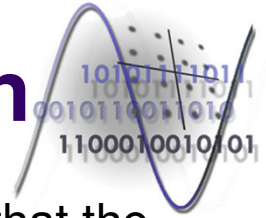
- the slope overload distortions:
 - appears if the slope of the source signal is larger than that of the predicted signal;
- the granular distortion (granular noise);
 - represents a quantization noise;
 - appears if the slope of the source signal is smaller than that of the predicted signal;

$$\left\{ \begin{array}{l} \text{source signal slope} = \left| \frac{dx(t)}{dt} \right| \\ \text{Delta signal slope} = \frac{\Delta}{T_s} \end{array} \right.$$

- Computation of the quantization noise power and of the quantization SNR:

- it is considered that $f_s=2f_m$ (f_m – the maximum frequency of the spectrum);

Delta modulation



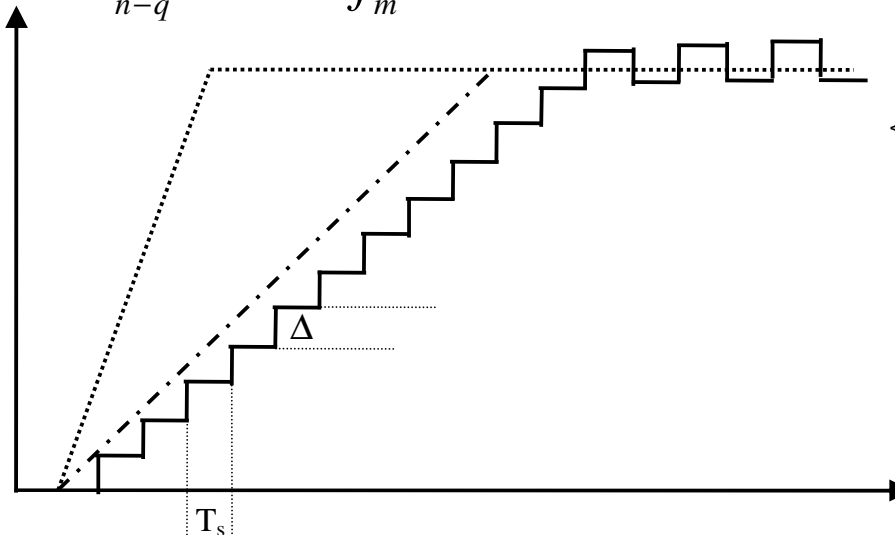
- it is considered that we do not have slope overload distortion and that the power of the signal, P , can be expressed according to the slope of the signal;
- the quantization noise is computed in the following way:

$$P_{n-q} = \frac{\Delta^2}{12} \cdot \frac{f_s}{f_s} = \frac{\Delta^2}{12} \cdot \frac{2f_m}{f_s} = \frac{\Delta^2 \cdot f_m}{6f_s}$$

$$SNR_{Delta} = \frac{P}{P_{n-q}} = \frac{6 \cdot f_s^3}{K^2 \cdot f_m}$$

$$\left\{ \begin{array}{l} \left| \frac{dx(t)}{dt} \right| = \Delta \cdot f_s \\ \left| \frac{dx(t)}{dt} \right| \approx K \cdot \sqrt{P} \end{array} \right.$$

- Distortions affecting the Delta modulated signal:

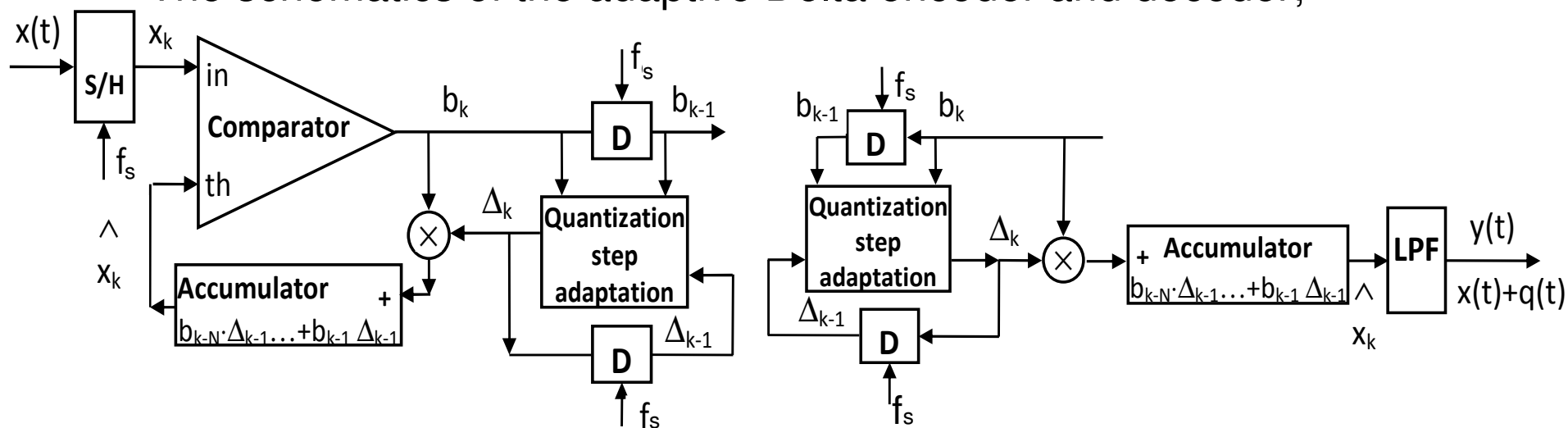


$$\left\{ \begin{array}{l} P = \frac{\Delta^2 \cdot f_s^2}{K^2} \\ P_{n-q} = \frac{\Delta^2}{12} \end{array} \right.$$

Delta modulation

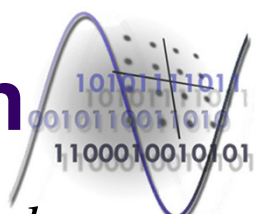


- Adaptive Delta modulation;
 - The quantization step is adjusted according to the slope of the source signal;
 - The measurement of the slope is based on the modulated bit sequence;
 - The schematics of the adaptive Delta encoder and decoder;

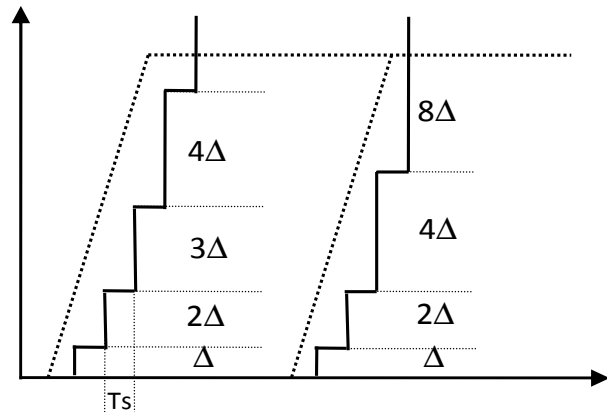


- Types of adaptive Delta modulations – according to the quantization step modification rules:
 - Song modulation;
 - Jayant modulation;

Delta modulation



- Song modulation;
 - Quantization step modification:
- Modulația Jayant;
 - Quantization step modification:



$$\begin{cases} \Delta_k = \Delta_{k-1} + \Delta & \text{if } b_k = b_{k-1} \\ \Delta_k = \Delta_{k-1} - \Delta & \text{if } b_k \neq b_{k-1} \\ \text{if } \Delta_k < \Delta \Rightarrow \Delta_k = \Delta \end{cases}$$

$$\Delta_k = \Delta_{k-1} \cdot p^{\text{sgn}(b_k \cdot b_{k-1})}$$

$$\begin{cases} \Delta_k = \Delta_{k-1} \cdot p & \text{if } b_k = b_{k-1} \\ \Delta_k = \Delta_{k-1} / p & \text{if } b_k \neq b_{k-1} \end{cases}$$

- Computation/measurement of the quantization noise power:
 - Can be used the mean square error (mse) between the source and the predicted signal:

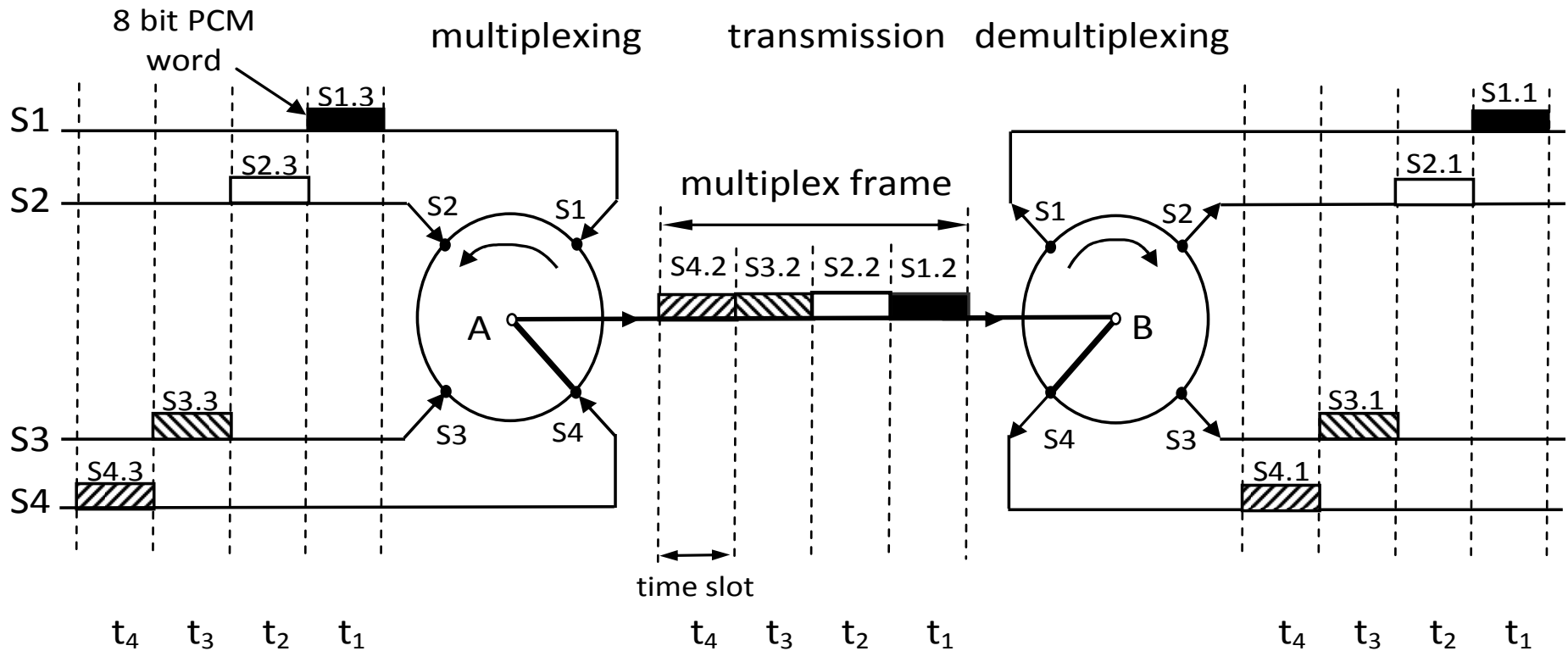
$$mse = \frac{\sum_{k=1}^M (x_k - \overline{x_k})^2}{M}$$

- the predicted signal represents the demodulated signal;

The PCM primary multiplex

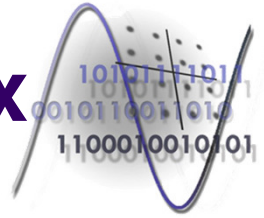


- The principle of PCM multiplexing:



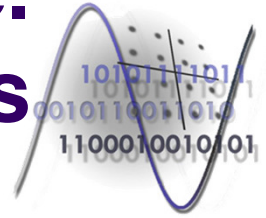
- The PCM multiplexing is the first level of multiplexing;
 - uses a time division multiplexing of the telephone channels, the multiplexing process being strongly related to the switching process;

The PCM primary multiplex

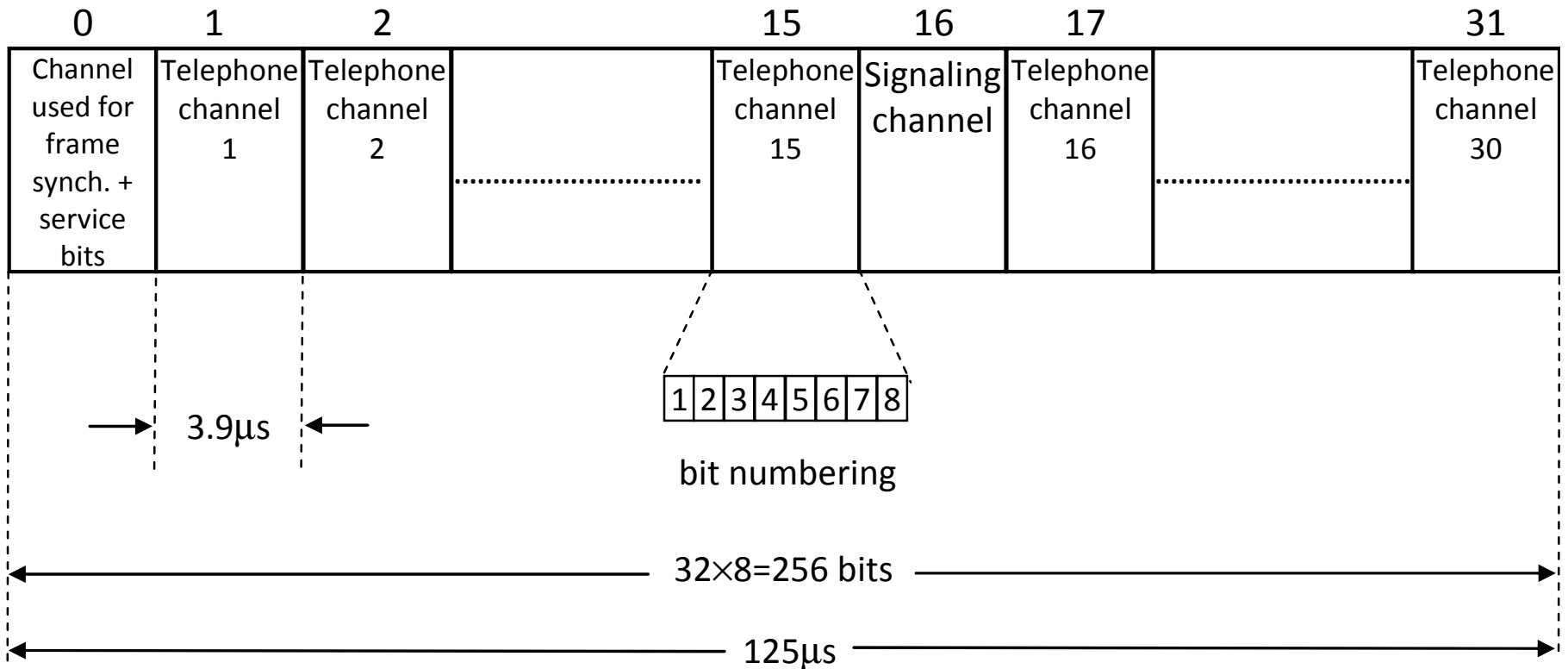


- to each 8 bit PCM word is allocated a time interval – *time slot*, interval in which the bits are transmitted;
- PCM words generated by different sources are interleaved, a separate time slot corresponding to each word;
- the bit rate assigned to the multiplex frame must be N times larger than the bit rate assigned to one of the multiplexed channels, N being the number of the multiplexed channels;
- the demultiplexing implies the identification of the time slots assigned to different channels and the transmission to the destination of the words extracted from the time slots, using a bit rate characteristic to the terminal equipment;

The E1 PCM frame. Structure & operations



- The structure of the E1 frame;



- The E1 frame includes 32 elementary channels of 64kbps;
 - the bit rate associated to this frame is 2,048Mbps; clock precision: ±50ppm;
 - can be defined fractional frames with a smaller no. of available channels;

The E1 PCM frame. Structure & operations



- 30 channels are used for voice transmissions, namely channels 1 ÷ 15 and 17 ÷ 31;
- Channel (slot) 0 is used for frame synchronization and service bits;
- Channel 16 is used for multiframe synchronizations, service bits and signaling operations;
 - it is a channel dedicated especially to signaling operations;
- There are two operation modes on channel 16, namely:
 - Channel Associated Signaling – CAS;
 - Common Channel Signaling – CCS;
 - for the management of the CAS signaling a multiframe is composed by grouping 16 PCM frames.
- If the E1 is used for data transmissions (for ex. to transport Ethernet frames) there are not defined channels/slots;
 - it is about a stream of bits with 2.048Mbps rate at the disposal of the user;

E1 PCM frame. Structure & operations

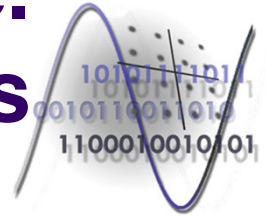


- There are two operation modes on channel 0, namely:
 - normal mode without CRC (*Cyclic Redundancy Check*);
 - CRC-4 mode, which uses CRC error control.
- Structure of the E1 PCM multiframe;
 - normal operation on slot 0 and CAS signaling on slot 16;

Frame number	Time slot 0 Bit number								Time slot 16 Bit number							
	1	2	3	4	5	6	7	8	1	2	3	4	5	6	7	8
0	Y	0	0	1	1	0	1	1	0	0	0	0	X	Z	X	X
1	Y	1	Z	X	X	X	X	X	Signaling ch. 1				Signaling ch. 16			
2	Y	0	0	1	1	0	1	1	Signaling ch. 2				Signaling ch. 17			
3	Y	1	Z	X	X	X	X	X	Signaling ch. 3				Signaling ch. 18			
4	Y	0	0	1	1	0	1	1	Signaling ch. 4				Signaling ch. 19			
5	Y	1	Z	X	X	X	X	X	Signaling ch. 5				Signaling ch. 20			
6	Y	0	0	1	1	0	1	1	Signaling ch. 6				Signaling ch. 21			
7	Y	1	Z	X	X	X	X	X	Signaling ch. 7				Signaling ch. 22			
8	Y	0	0	1	1	0	1	1	Signaling ch. 8				Signaling ch. 23			
9	Y	1	Z	X	X	X	X	X	Signaling ch. 9				Signaling ch. 24			
10	Y	0	0	1	1	0	1	1	Signaling ch. 10				Signaling ch. 25			
11	Y	1	Z	X	X	X	X	X	Signaling ch. 11				Signaling ch. 26			
12	Y	0	0	1	1	0	1	1	Signaling ch. 12				Signaling ch. 27			
13	Y	1	Z	X	X	X	X	X	Signaling ch. 13				Signaling ch. 28			
14	Y	0	0	1	1	0	1	1	Signaling ch. 14				Signaling ch. 29			
15	Y	1	Z	X	X	X	X	X	Signaling ch. 15				Signaling ch. 30			

- TS0 in even frames: Y0011011 – frame synchronization word ;
- TS0 in odd frames: Y1ZXXXX;
- Y international bit;
- Z frame synch. loss alarm bit;
- X - not used (national bits);
- TS16 in frame 0 : 0000XZXX ; in frames 1 – 15 : signaling for voice channels;
- 0000 – multiframe synchronization;
- Z – multiframe synch. loss alarm;
- X – not used (national bits);

The E1 PCM frame. Structure & operations



- The structure of the E1 PCM multiframe;
 - CRC-4 mode in slot 0;

Sub-multiframe number	Frame number	Frame type	Time slotd 0							
			Bit number							
			1	2	3	4	5	6	7	8
I	0	FAS	C ₁	0	0	1	1	0	1	1
	1	NFAS	0	1	Z	X	X	X	X	X
	2	FAS	C ₂	0	0	1	1	0	1	1
	3	NFAS	0	1	Z	X	X	X	X	X
	4	FAS	C ₃	0	0	1	1	0	1	1
	5	NFAS	1	1	Z	X	X	X	X	X
	6	FAS	C ₄	0	0	1	1	0	1	1
	7	NFAS	0	1	Z	X	X	X	X	X
II	8	FAS	C ₁	0	0	1	1	0	1	1
	9	NFAS	1	1	Z	X	X	X	X	X
	10	FAS	C ₂	0	0	1	1	0	1	1
	11	NFAS	1	1	Z	X	X	X	X	X
	12	FAS	C ₃	0	0	1	1	0	1	1
	13	NFAS	E	1	Z	X	X	X	X	X
	14	FAS	C ₄	0	0	1	1	0	1	1
	15	NFAS	E	1	Z	X	X	X	X	X

FAS – “Frame Alignment Signal” = 0011011;

NFAS – “Not Frame Alignment Signal”;

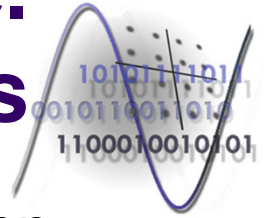
C₁ – C₄ – “Cyclic Redundancy Check-4” bits;

E – CRC-4 error indicator bits;

Z – alarm bit;

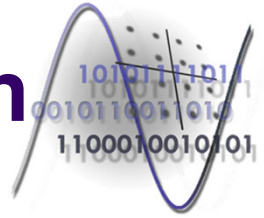
X – unused bits.

The E1 PCM frame. Structure & operations



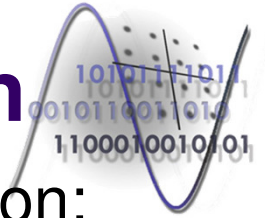
- In CRC-4 mode on slot 0 the Y bits from frames with even number are used to transmit CRC sequences on 4 bits;
 - in the Y bits from frames 0, 2, 4 and 6 is transmitted a $C_1 C_2 C_3 C_4$ sequence used for bit error detection in frames 0 – 7 of the previous multiframe ;
 - in the Y bits from frames 8, 10, 12 and 14 is transmitted a $C_1 C_2 C_3 C_4$ sequence used for bit error detection in frames 8 – 15 of the previous multiframe;
 - the generator polynomial used for the computation of the CRC-4 sequence is: $p(x) = x^4 + x + 1$
 - the non-detection probability of bit error packets with more than 4 error is 6.25%;
 - the detection probability of these error packets is 93.75%;
 - all error packets with at most 4 errors are detected.
 - Remark: in the normal frame, without CRC can be monitored only 7 bits (the frame synchronization bits) in each group of 505 bits;

The E1 PCM frame. Synchronization



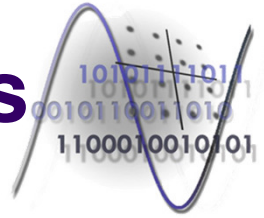
- Aspects related to frame synchronization;
 - Loss of frame synchronization detection:
 - three consecutive frames with FAS errors or,
 - three consecutive bit errors in position two in frames without FAS or,
 - bit error probability higher than 10^{-3} ;
 - the FAS signal is monitored for this error detection;
 - in CRC-4 working mode 1000 CRC comparison are performed in a second;
 - if the threshold of 914 bad comparisons (91.4%) is exceeded it is declared loss of the frame synchronization;
 - it ensures a better frame synchronization, being avoided the problem of frame alignment sequence (FAS) simulation.

The E1 PCM frame. Synchronization



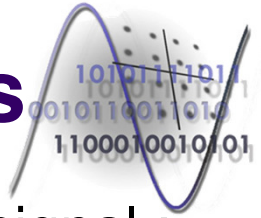
- Aspects related to frame and multiframe synchronization;
 - Loss of multiframe synchronization detection in the CAS case:
 - two consecutive MFAS signals with errors or,
 - two multiframes with all zero bits in slot 16;
 - Frame and multiframe synchronization detection:
 - **Normal frame synchronization:**
 - FAS received correctly, bit two in NFAS is “1”, next FAS received correctly;
 - **Multiframe synchronization with CAS:**
 - MFAS received correctly and slot 16 of the previous frame is not zero;
 - **Multiframe synchronization with CRC:**
 - bit in position one of NFAS frames generates the sequence: 0 0 1 0 1 1 ;
 - it is realized an initial frame and multiframe synchronization;
 - at least 2 CRC MFAS must be correctly received in a 8ms time interval (4 CRC-MF), between these MFAS detections being a time interval of 2ms or multiples of this value;
 - it is realized a check and validation of the synchronization based on the CRC control sequence.

The E1 PCM frame. Alarms



- Alarms associated to frame E1;
 - Frame alarm (remote alarm) bit:
 - bit Z of slot 0 (named also bit A);
 - (yellow alarm – transmitted to the opposite end);
 - value 0 – normal operation, value 1 – alarm event: power supply interruption, codec failure, loss of input signal, FAS error, bit error probability higher than 10^{-3} ;
 - any of these event generates a **red alarm** at the end where they take place (where are detected);
 - the equipment which receives bit Z=1, declares **yellow alarm**;
 - Multiframe alarm (remote alarm) bit:
 - bit Z of slot 16 frame 0 (called also bit A);
 - value 0 – normal operation, value 1 – loss of MFAS signal (yellow alarm transmitted to the opposite end);
 - The Z (or A) bit signals remote alarm – RAI : „Remote Alarm Indication”;

The E1 PCM frame. Alarms

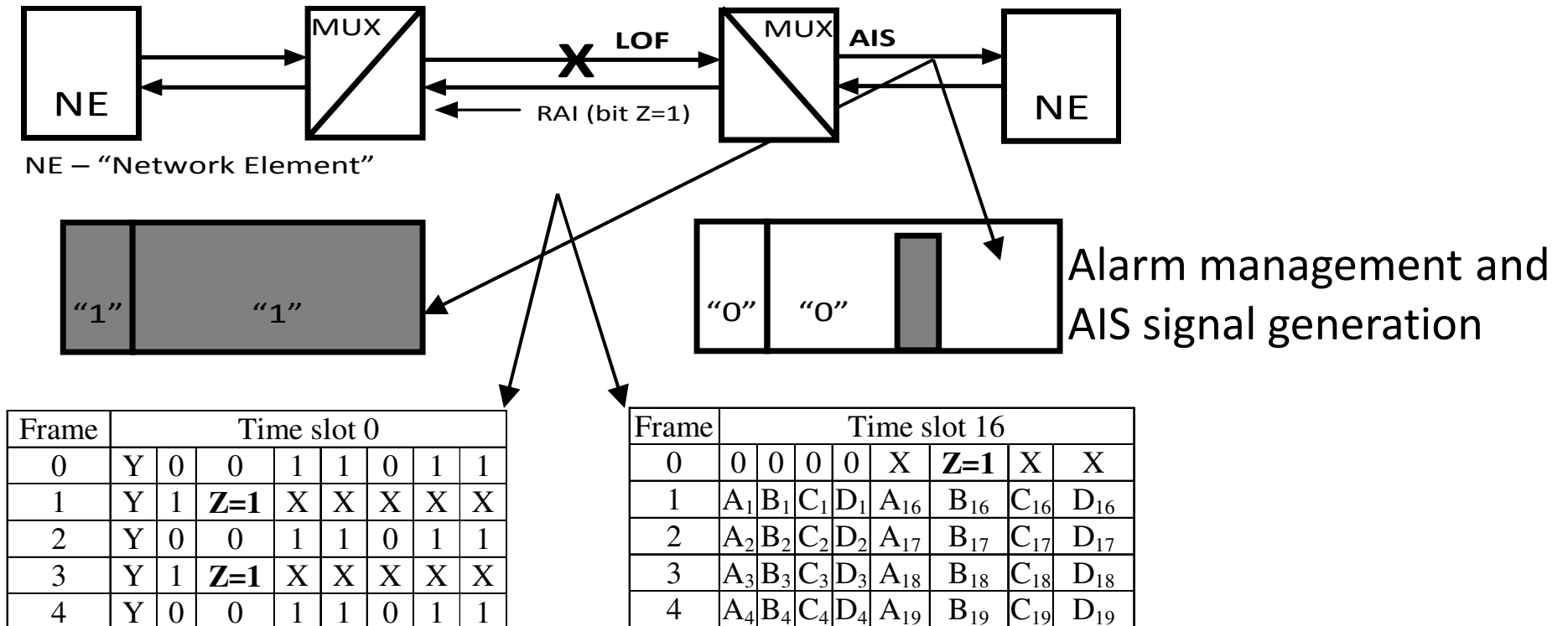


- AIS – „Alarm Indication Signal” – called also keep alive signal ;
 - the AIS represents at least 509 “1” bits in a block of 512 bits or less than 3 “0” bits in 2 frames (in the case of slot 16 less than 3 “0” in this slot during two consecutive multiframes);
- the terminal equipment which detects the AIS signal declares AIS state called also **blue alarm**;
 - AIS - generated by a multiplexer to the terminal equipment when it is detected a frame loss, signal loss or multiframe loss;
 - the output channels transmit continuous “1” – it is allowed to maintain the clock synchronization between the two equipment;
 - or, only in slot 16 it is transmitted a continuous “1” (MFAS error);
 - AIS - generated by a multiplexer when it receives a yellow alarm from the opposite end (multiplexer) - it is a continuous “1” signal;
 - it is permitted to maintain the clock synchronization between multiplexers;
 - AIS - generated by a multiplexer to the terminal equipment when it receives a yellow alarm;
 - can be detected by the terminal equipment (if we do not have LOS or LOF) and that equipment declares AIS state.

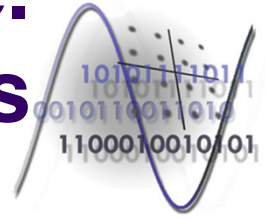
The E1 PCM frame. Alarms



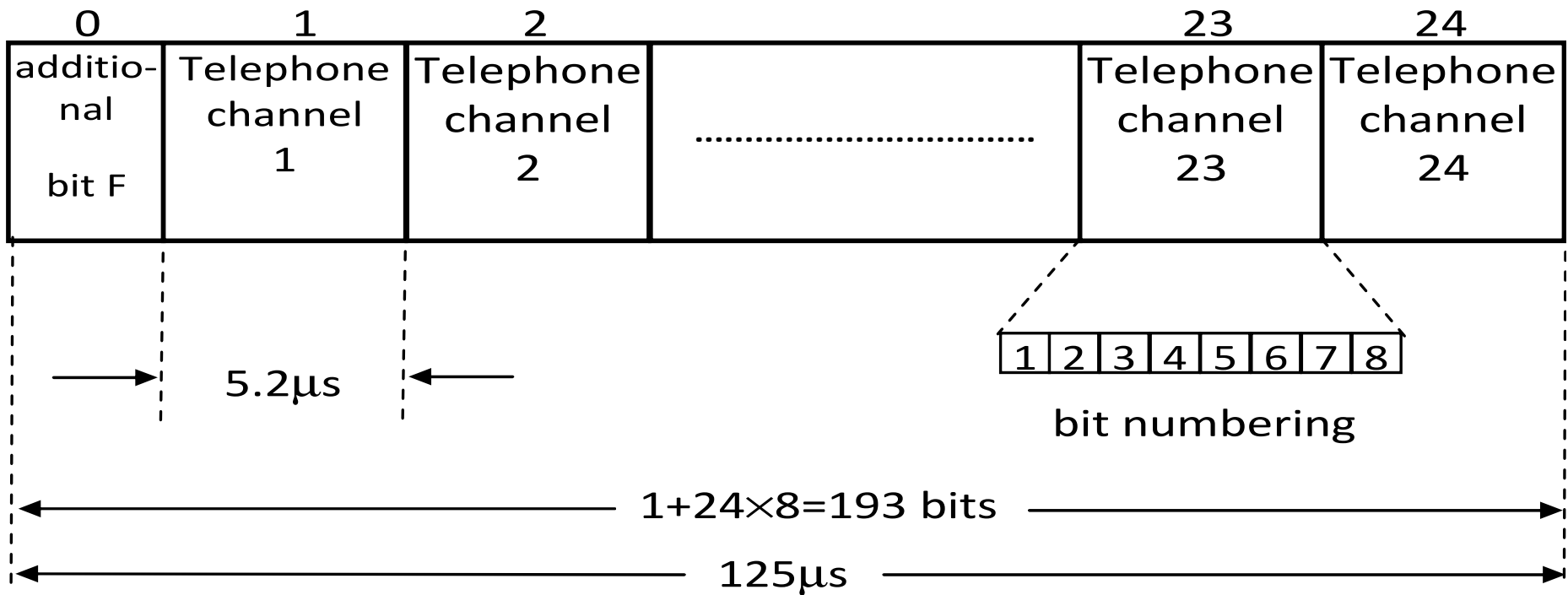
- The LOS and LOF events generate a red alarm;
 - in the case of the loss of multiframe synchronization a yellow alarm is transmitted to the opposite end using the dedicated Z bit;
 - the equipment detecting the loss of multiframe synchronization and the equipment detecting multiframe yellow alarm generate an AIS signal in slot 16;



The T1 PCM frame. Structure & operations

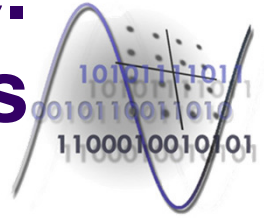


- The structure of the T1 frame;



- The T1 multiplex frame includes 24 telephone channels + 1 bit additional, the F bit;
 - bit F is used for synchronization or for implementation of a special data channel;

The T1 PCM frame. Structure & operations



- Types of T1 multiframes:
 - The Supper Frame (SF);
 - composed of 12 frames;
 - has no separate time slot for synchronization or signaling;
 - the frame and the multiframe synchronization is accomplished with the help of the supplementary F bit;
 - for channel assigned signaling it is used the last bit of each sixth frame : A – B type signaling;
 - this technique is called bit robbing;
 - for CCS signaling is used the slot 24 of the T1 frame;
 - in the case of SF frame the yellow alarm is transmitted by setting the bit no. 2 of each slot to 0;

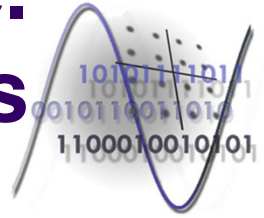
The T1 PCM frame. Structure & operations



- The structure of T1 PCM SF multiframe;

Frame number	Use of bit F		Number of info. bits per channel	Signaling bit position	Signaling channel
	Frame synch.	Multiframe synch.			
1	1	-	8	-	-
2	-	0	8	-	-
3	0	-	8	-	-
4	-	0	8	-	-
5	1	-	8	-	-
6	-	1	7	8	A
7	0	-	8	-	-
8	-	1	8	-	-
9	1	-	8	-	-
10	-	1	8	-	-
11	0	-	8	-	-
12	-	0	7	8	B

The T1 PCM frame. Structure & operations



- Types of T1 multiframes:
 - The Extended Supper Frame (ESF);
 - composed of 24 frames;
 - the F bit is used to frame and multiframe synchronization and other purposes;
 - a special sequence, having the structure 0 0 1 0 1 1, located in frames with frame number multiple of 4;
 - the frames having an odd number implement a 4kbps bit rate data channel, the M channel (management, control, alarms);
 - in even number frames whose number is not a multiple of 4 it is transmitted a CRC-6 control sequence;
 - the transmission of the signaling is accomplished in a similar way as in the case of SF multiframe:
 - the 8-th bit of each channel of every sixth frame is used for CAS signaling;
 - 4 bit for CAS signaling for each channel: bits A B C and D.

The T1 PCM frame. Structure & operations



- the CRC mechanism used detects all error packets with at most 6 errors and detects 98.4% of error packets with more than 6 errors;
- two type of signals can be transmitted on the M data channel :
 - bit oriented signals which are unscheduled messages;
 - they begin with a “1” byte followed by a “0” bit, a command/message identifier on 6 bits and finally a “0” bit follows;
 - the 6 bit identifier encodes alarms and different messages: protection switching activation, loop-back activation, a.s.o.
 - the yellow alarm is coded: 11111111 00000000;
 - the messages with high priority are transmitted continuously at least one second and the low priority messages are repeated ten times;
 - message oriented signals – consists of data packets composed of header, address field, control field, information field and error control field (CRC);
 - they are transmitted in each second and contain:
 - CRC errors, synch. errors, coding rule violations;
 - are controlled by a communication protocol;
 - can be interrupted by bit oriented signals.

The T1 PCM frame. Structure & operations



- The structure of the T1 PCM ESF multiframe with CAS signaling;

Frame number	Use of bit F			Number of info. bits per channel	Signaling bit position	Signaling channel
	Frame synch	Data link	CRC-6			
1	-	M	-	8	-	-
2	-	-	C ₁	8	-	-
3	-	M	-	8	-	-
4	0	-	-	8	-	-
5	-	M	-	8	-	-
6	-	-	C ₂	7	8	A
7	-	M	-	8	-	-
8	0	-	-	8	-	-
9	-	M	-	8	-	-
10	-	-	C ₃	8	-	-
11	-	M	-	8	-	-
12	1	-	-	7	8	B
13	-	M	-	8	-	-
14	-	-	C ₄	8	-	-
15	-	M	-	8	-	-
16	0	-	-	8	-	-
17	-	M	-	8	-	-
18	-	-	C ₅	7	8	C
19	-	M	-	8	-	-
20	1	-	-	8	-	-
21	-	M	-	8	-	-
22	-	-	C ₆	8	-	-
23	-	M	-	8	-	-
24	1	-	-	7	8	D

The T1 PCM frame. Alarms



- Can be defined fractional T1 frames with less used channels;
 - Some channels of the frame can are not available to the users;
 - the frame has assigned a smaller useful bit rate for transmission;
 - the transport network has to take a smaller rate from the users;
 - If the T1 link is used for data transmissions there are not defined channels/slots – the framing bit can be kept;
- The T1 alarms (shortly);
 - OOF (“Out Of Frame Condition”): 2 of 4, 2 of 5 or 3 of 5 synchronization bits are erroneous;
 - Red CFA (“Carrier Failure Alarm”): OOF for 2.5s; end of this state: no OOF for 1s;
 - Yellow CFA – yellow alarm transmitted to the opposite end;
 - LOS (“Los Of Signal”): no impulse detected in a window of 175+/-75 impulse periods (100 – 250 bits);

Line interface of the primary multiplex



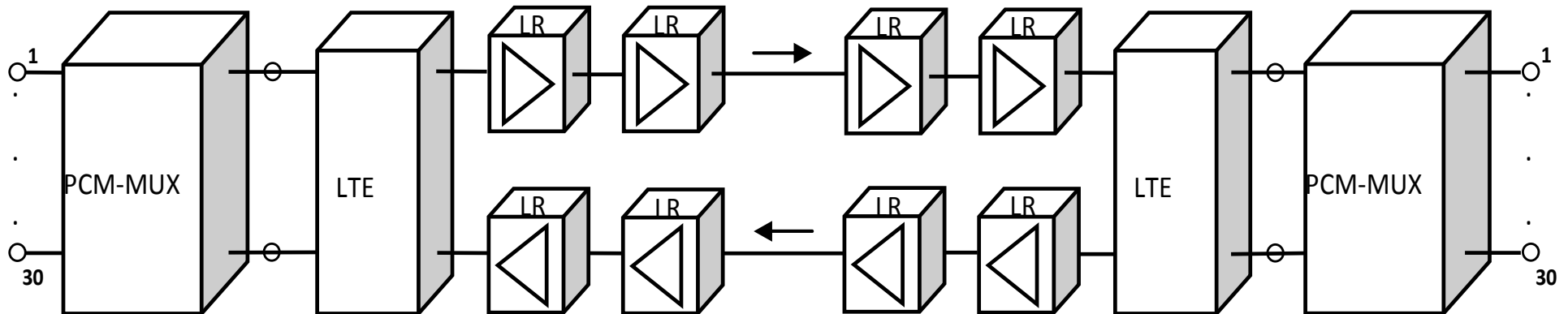
- Transmission of the E1 frame;
 - 4 wire full duplex transmission;
 - AMI (*Alternate Mark Inversion*) coding;
 - the “0” bit is coded with a 0V level and the “1” bits are alternatively coded with $\pm A$ impulses;
 - this code has no DC component (it is avoided the saturation of the separation transformer’s core);
 - has relatively narrow bandwidth;
 - simple decoding;
 - reduced synchronization capability;
 - it is replaced with a HDB3 (*High-Density-Bipolar-3 Zeros*) coding;
 - this code replaces groups of 4 zeros with violations of AMI coding rule – it is ensured also a reduced level of the DC component.

Line interface of the primary multiplex



The last impulse on line	Number of impulses from the last replace	
	Odd	Even
negative	0 0 0 -	+ 0 0 +
positive	0 0 0 +	- 0 0 -

- The HDB3 coding rule;



LTE – “Line Terminating Equipment”

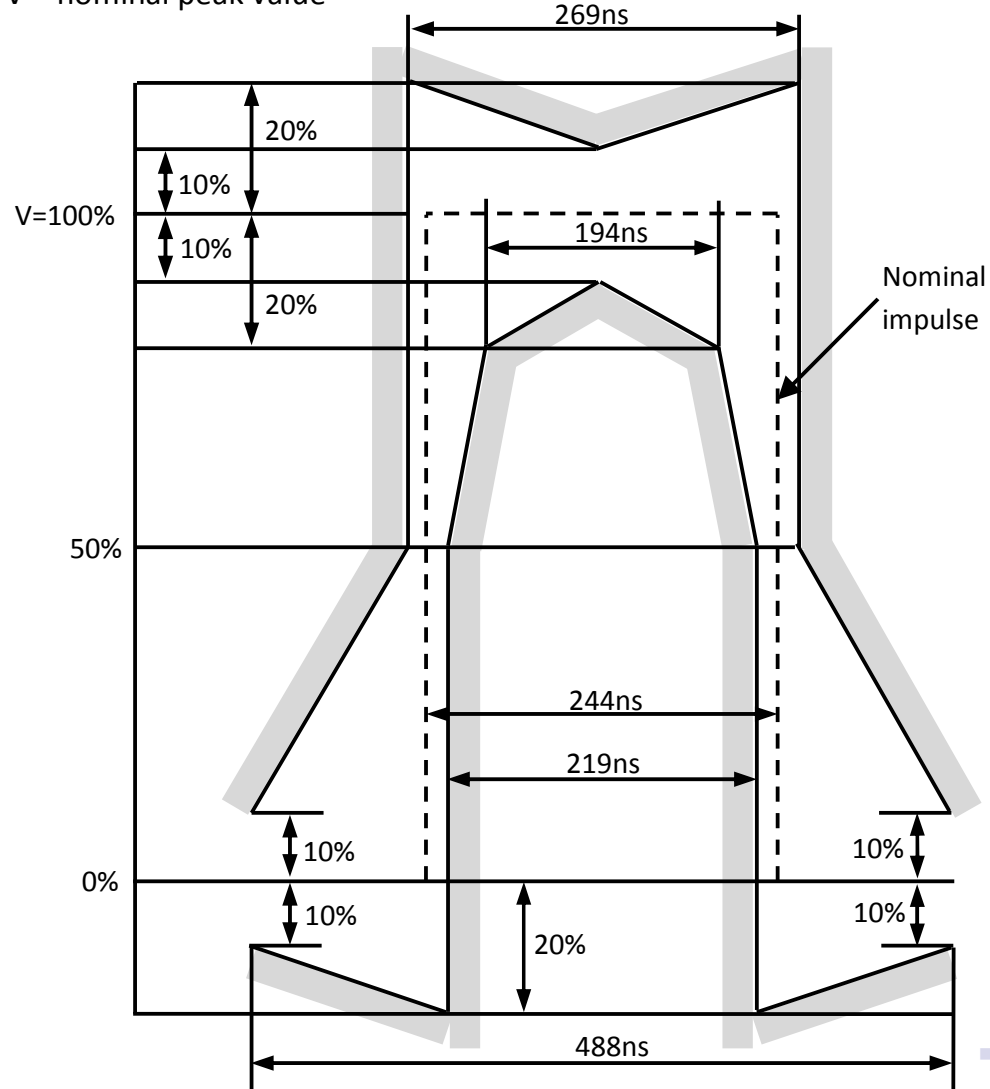
LR – “Line Regenerator”

- E1 transmission system;

Line interface of the primary multiplex

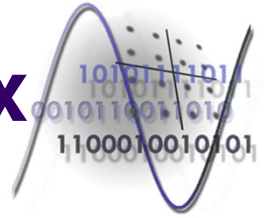


V – nominal peak value



■ Coded impulse mask;

Line interface of the primary multiplex



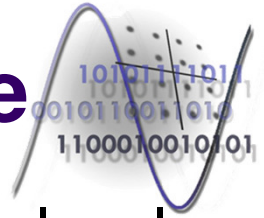
- (E1) line interface parameters:
 - Nominal bit rate: 2048kbps;
 - Precision of the nominal bit rate: at least ± 50 ppm;
 - Line code: HDB3;
 - Frame structure;
 - Transmission medium / Number of pairs in each direction;
 - Coaxial; Twisted pairs; 1 cable / 1 pair for each transmission direction;
 - Load impedance: 75Ω (coaxial), 120Ω (twisted pair);
 - Peak amplitude: 2.37V – 3V;
 - Power level and power spectral mask;
 - Impulse nominal duration: 244ns;
 - Ration of positive and negative amplitudes: 0.95 – 1.05;
 - Ratio of positive/negative pulse duration: 0.95 – 1.05;
 - Maximum peak to peak jitter;
 - DC power: has to be as low as possible;

Line interface of the primary multiplex



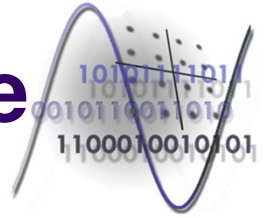
- T1 interface specific characteristics:
 - Transmission of T1 frames is similar with the transmission of E1 frames;
 - 4 wire full duplex with regenerators after each 1.5km cable length;
 - B8ZS (*Bipolar with 8 Zero Substitution*) line coding;
 - AMI type cod which replaces the groups of 8 consecutive zero bits with a coded sequence having the structure: 0 0 0 V B 0 V B;
 - 3 “0” symbols, a violation of the AMI coding rule, followed by a binary symbol encoded normally, a “0” symbol, after that a new violation of the AMI coding rule and finally a binary symbol encoded normally.

Terminal – multiplexer interface



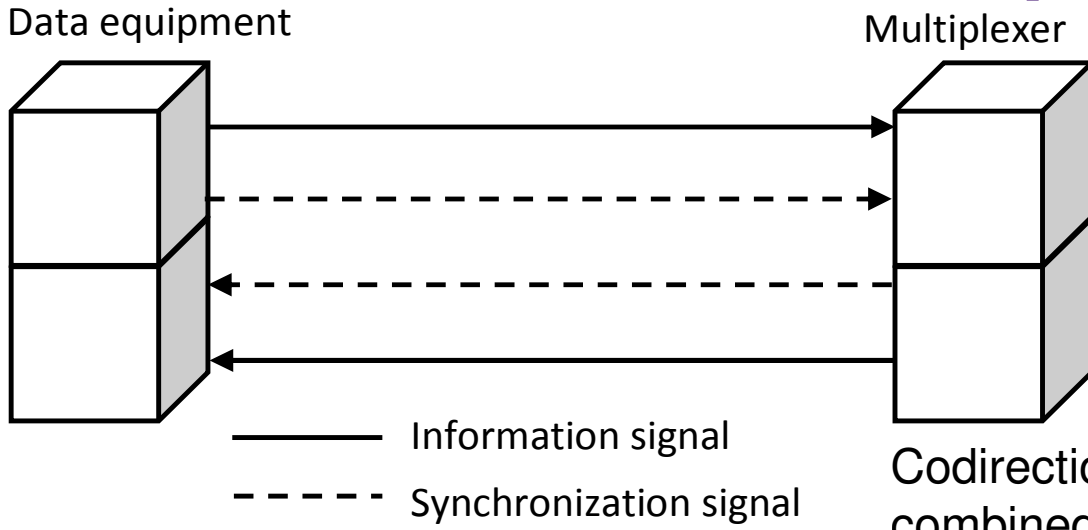
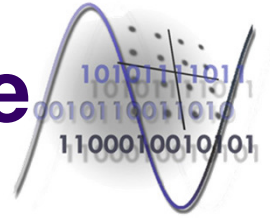
- There are defined two types of interfaces between the local equipment (terminal - multiplexer);
 - These correspond to two transmission strategy of data transmission and synchronization;
 - Codirectional interfaces;
 - correspond to the case when each equipment transmits the data together with his own synchronization signal;
 - all equipment must have the same clock synchronized from an external source;
 - Contradirectional interfaces;
 - the multiplexer transmit the synchronization information for both transmission directions;
 - it is not necessary an external synchronization source;

Terminal – multiplexer interface



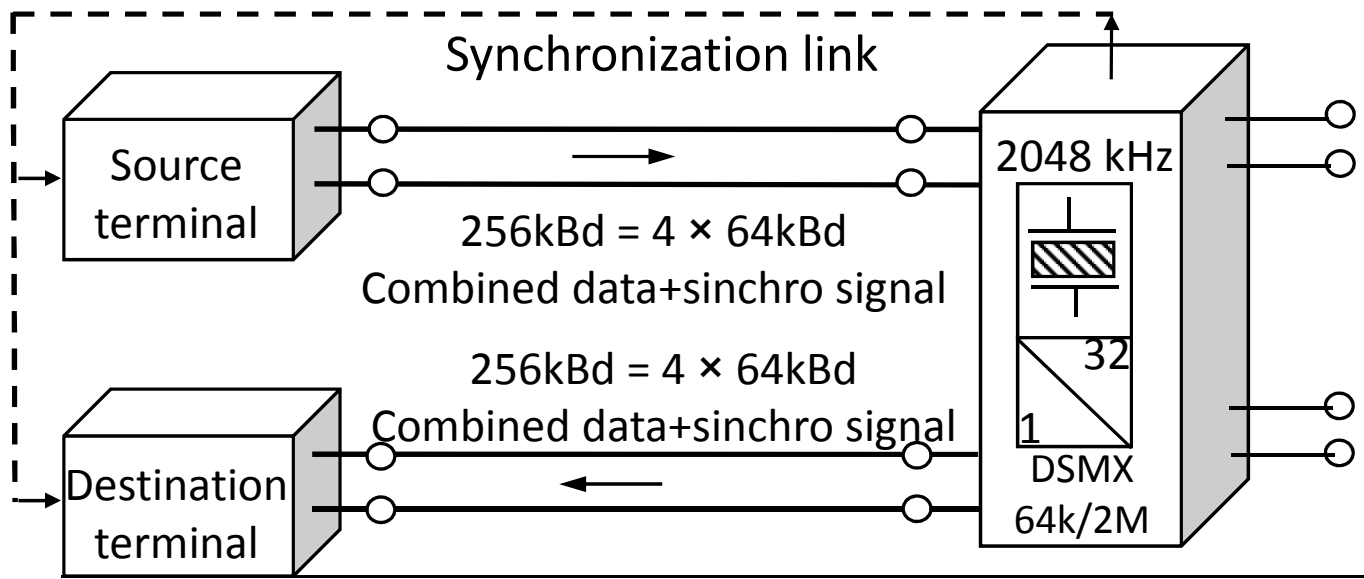
- Codirectional interfaces;
 - A complex signal, combining both the information and the synchronization signals (bit clock and byte clock) is transmitted between the connected equipment;
 - A single channel composed of a pair of wire is used in each directions;
 - separation transformers are usually used.
 - Precision of the clock signal: at least ± 100 ppm;
 - The clock generator of each equipment (multiplexer or terminal equipment) is synchronized with an external reference clock;

Terminal – multiplexer interface

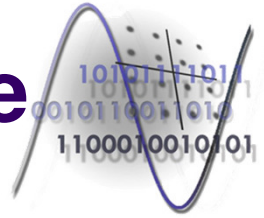


Codirectional interface – principle;
channels used;

Codirectional interface – technical details;
combined data and synchronization channels



Terminal – multiplexer interface

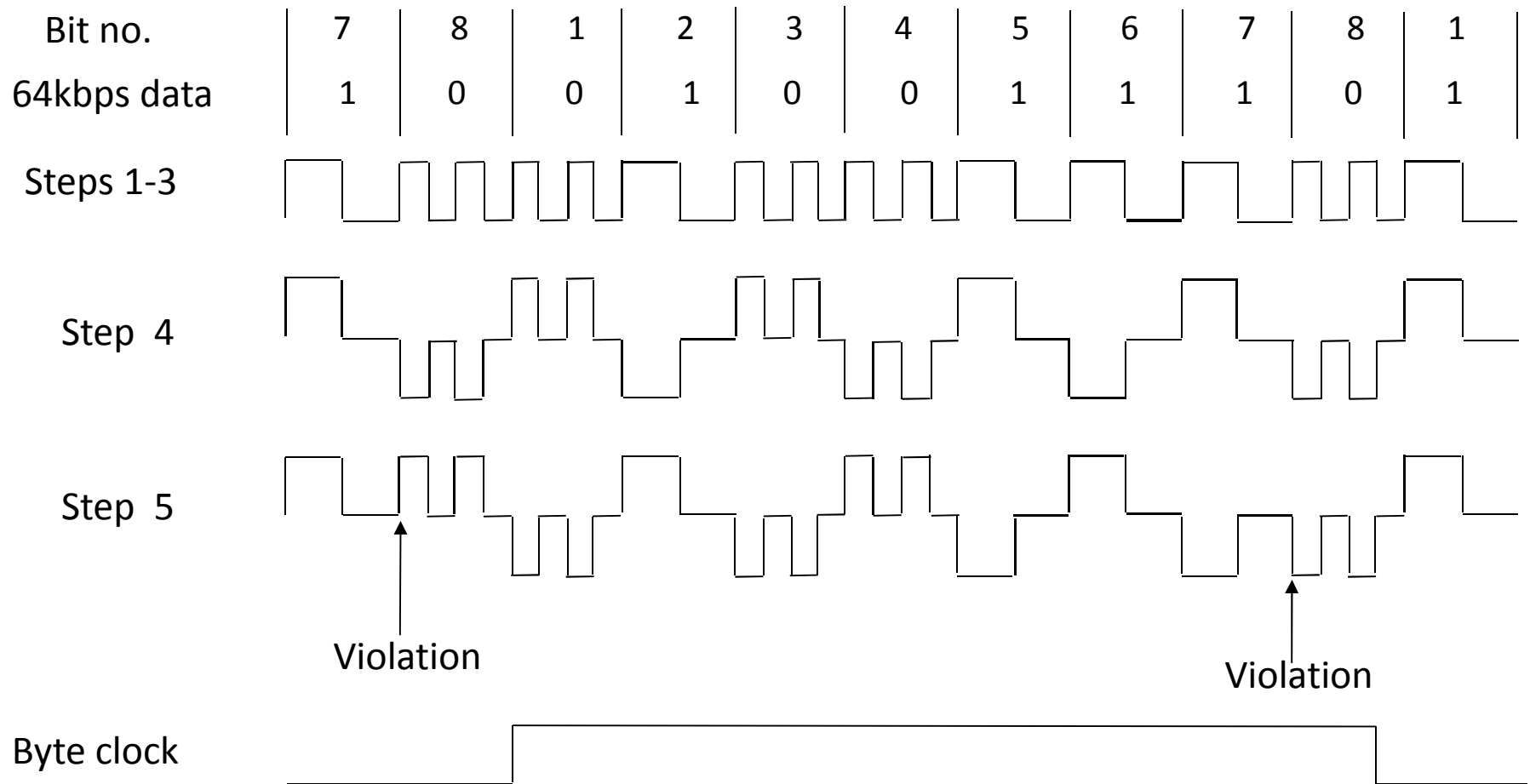


- Codirectional interface – the coding rule;
 - step 1: the period corresponding to the 64kbps rate is split in 4 unit intervals;
 - step 2: a binary “1” (64kbps rate) is coded as a block of 4 binary symbols (each having a period 4 times smaller): “1 1 0 0”; “0” represents 0V;
 - step 3: a binary “0” (64kbps rate) is coded as a block of 4 binary symbols (each having a period 4 times smaller): “1 0 1 0”;
 - step 4: the coded data signal is converted into a 3 level signal by alternating the polarity of consecutive blocks of 4 symbols;
 - step 5: the alternation of the polarity of the blocks is violated every 8th block, meaning the position corresponding to bit 8 in a byte;
 - in this way can be realized the byte level synchronization between the two equipment.

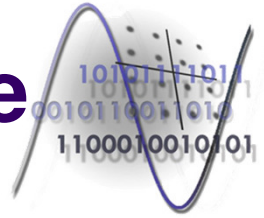
Terminal – multiplexer interface



- Codirectional interface – coding rule;

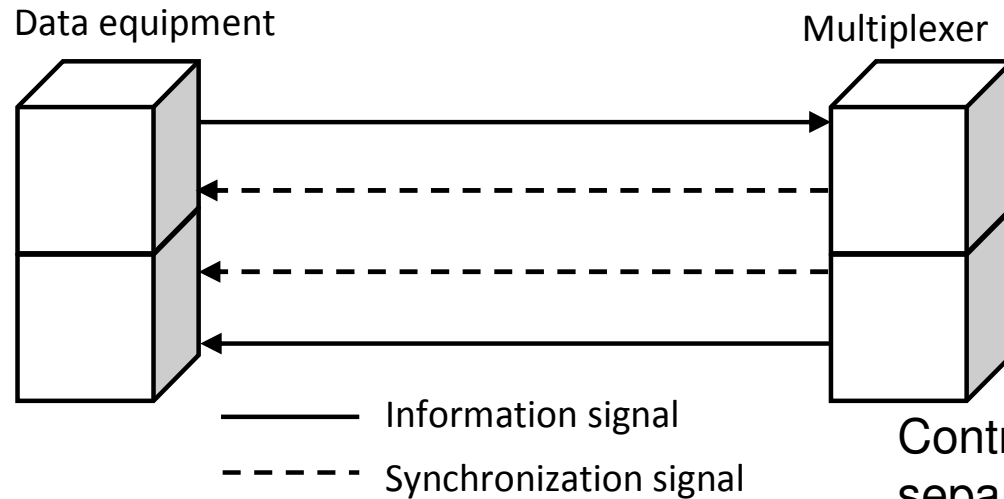


Terminal – multiplexer interface



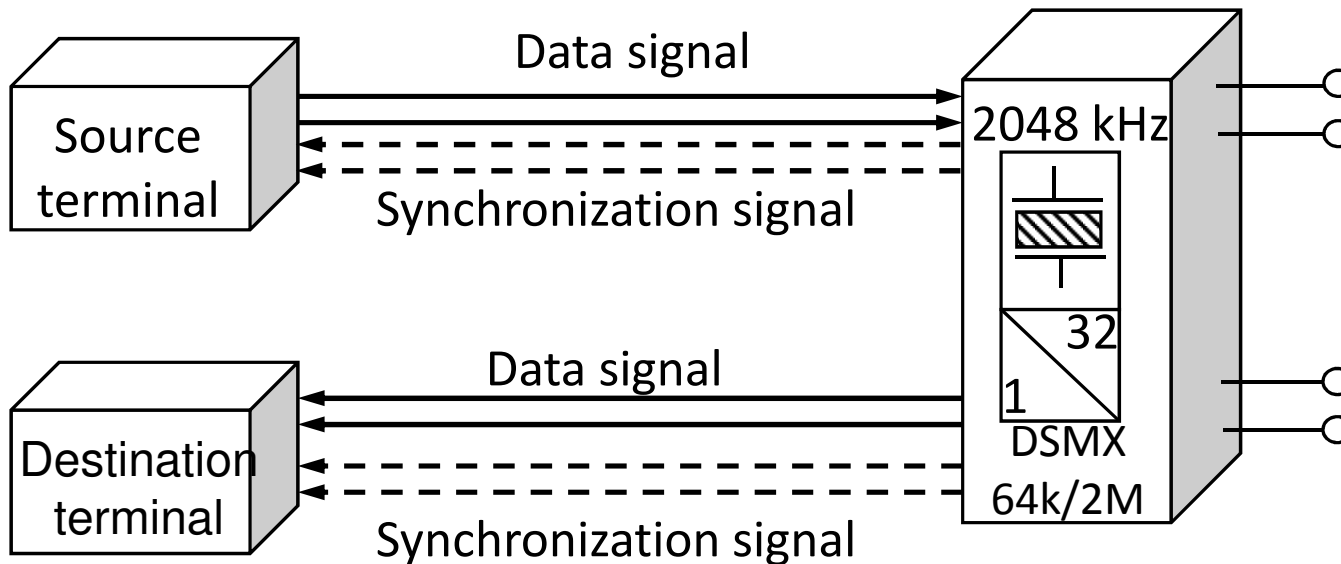
- Contradirectional interfaces;
 - Both the data signal and the synchronization signal is transmitted between equipment;
 - The synchronization signal is transmitted from multiplexer to terminal equipment;
 - There are necessary two channels, each on a pair of wire, in both directions: data and synchronization
 - bit clock and byte clock;
 - Precision of the clock signal: at least ± 100 ppm;
 - It is not necessary an external reference clock;

Terminal – multiplexer interface

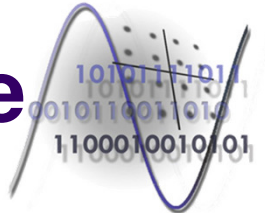


Contradirectional interface – principle; channels used.

Contradirectional interface – technical details; separate data and synchronization channels.



Terminal – multiplexer interface



- Contradirectional interface – coding rule;
 - the clock signal transferred between equipment is transformed into a three level signal by alternating the polarity of consecutive nonzero impulses;
 - at the end of each byte (bit on position 8) is inserted a violation in the polarity alternation of the impulses;
 - it is realized a byte level synchronization between equipment.

