Course 2 - 3
Analog access in the telephone network. Fundamental notions of digital telephony. Primary PCM multiplex

1. Analog access. Basic characteristics

- Analog access – the most simple access method in the telephone network; it is characteristic for analog telephone networks POTS (Plain Old Telephone Service); used, due to his simplicity, in some IDN networks.
- Main characteristics of the analog access:
  - o Bandwidth 300Hz – 3400Hz.
  - o Two wire access and remote power supply from the exchange at 48Vcc.
    - ▪ The telephone device works on 4 wire but the transmission to the switching takes place on two wire;
    - ▪ Analog switching takes place on 2 wire but digital switching and long distant and international switching (both analog and digital) takes place on 4 wire (see fig.1 and fig.2). The notion of 4 wire refers to two channels with opposite direction of transmission on different physical medium (wire for examples);
  - ▪ There are necessary 2 wire – 4 wire transition points ensured by a differential system named hybrid (H).

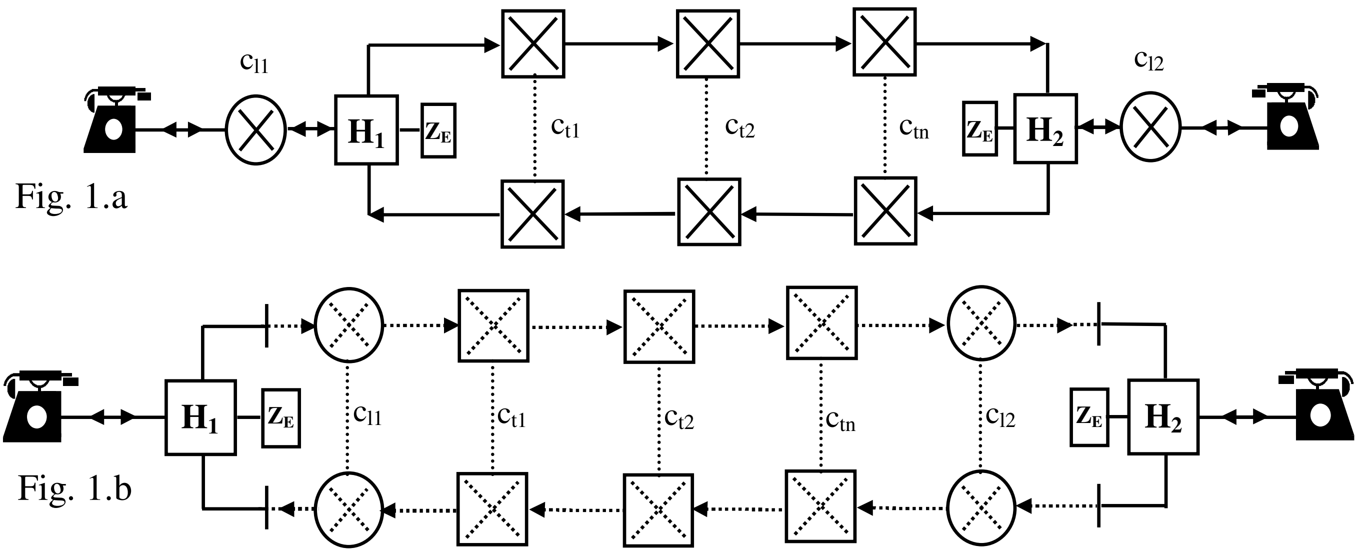


Fig. 1 The schematic of a distant analog (fig. a) and digital (fig. b) telephone connection

- ▪ Functions of the hybrid: - transfers the signals generated by every terminal (T_1, T2) on transmission branch (a-b ,b'-a') of the 4-wire circuit a4nd from there on the 2-wire circuit of the destination subscriber.

  - attenuates the signals that pass from the reception branch on the transmission branch.

- ▪ The differential system is a bridge whose balance is ensured by the relation $Z_l = Z_b$ (1), where Z_l is the line impedance and Z_b balance impedance.
- ▪ condition (1) can't be fulfilled exactly in the whole frequency bandwidth and for all length of subscriber lines; it is not ensured a perfect balance – impedance adaptation is not ensured – a fraction of the signal received in the 4-wire loop is transmitted in the opposite direction as an echo.

2. Fundamental notions of digital telephony

- The transmission technique used in fixed digital telephone networks is the PCM (Pulse Coded Modulation) – practically 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 obtained rate for a telephone channel is 64kbps; more advanced voice coding techniques can ensure a significant reduction of the necessary bit rate; among these techniques are ADPCM (Adaptive Differential PCM) and parametric coding techniques – take into account the characteristics of the vocal signal – it can be used just for the coding of the vocal signal – it is not possible the data transmission by modem on a network that uses such coding; table 1 presents some standardized voice coding techniques.

ITU-T standard	Coding technique	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

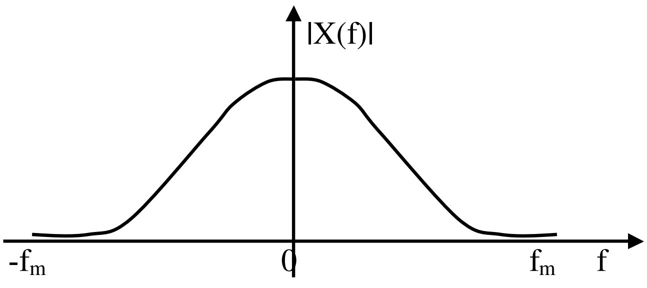
Tab. 1 Standardized voice coding techniques and associated bit rates

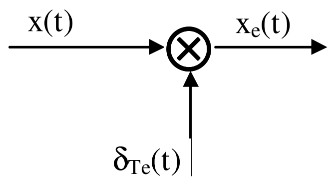
- Processing required by PCM: sampling, quantization and coding
  - The sampling theorem – basic relations, the aliasing phenomenon

$$x_e(t)=x(t)\cdot\delta_{T_e}(t)=x(t)\cdot\sum_{n=-\infty}^{\infty}\delta(t-nT_e)=\sum_{n=-\infty}^{\infty}x(nT_e)\cdot\delta(t-nT_e) \tag{2}$$

$$X_e(\omega)=\frac{1}{T_e}\cdot\sum_{k=-\infty}^{\infty}X(\omega-k\cdot\omega_e) \tag{3}$$

    - The sampling theorem (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 his samples $\{x(nT)\}$, if $T=(\pi/\omega_M)$





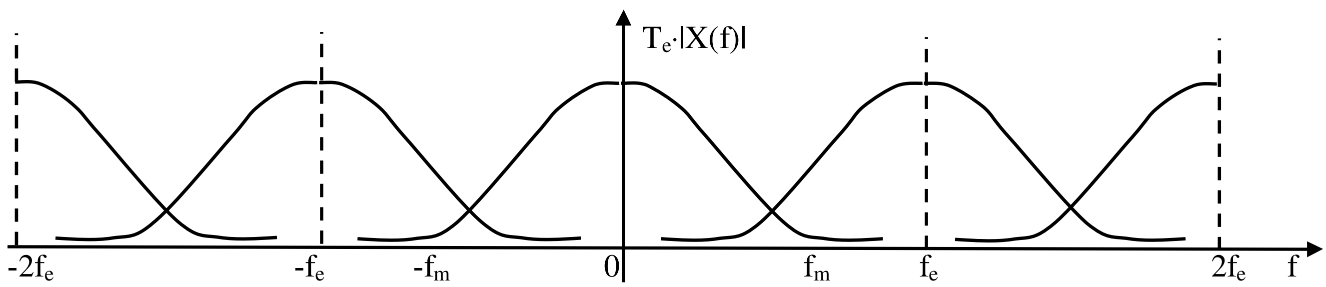


Fig. 2.a Spectral properties of the sampled signals and the aliasing phenomenon

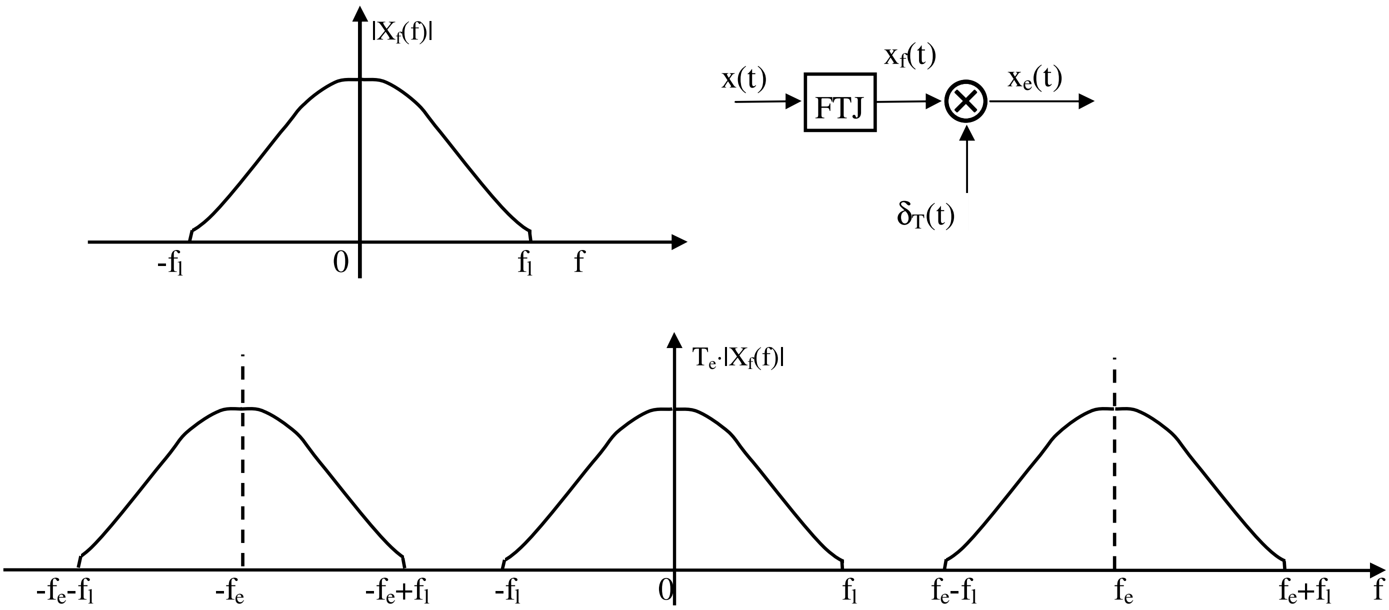


Fig. 2.b Spectral properties of the sampled signals and spectral aliasing avoidance

- Reconstruction of the sampled signals using LP filtering; base relations

$$H(\omega) = \begin{cases} T_e = \dfrac{1}{2f_M}, \forall|\omega| < \omega_M \\ 0, \forall|\omega| > \omega_M \end{cases} \quad ; \quad h(t) = \sin c(\omega_M t) \tag{4}$$

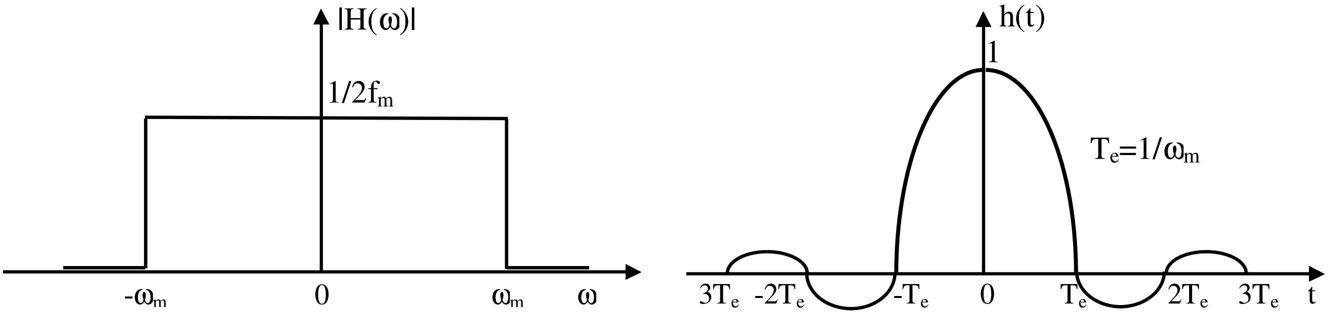


Fig. 3 Frequency characteristic and impulse response of an ideal reconstruction filter

- Uniform and non-uniform quantization

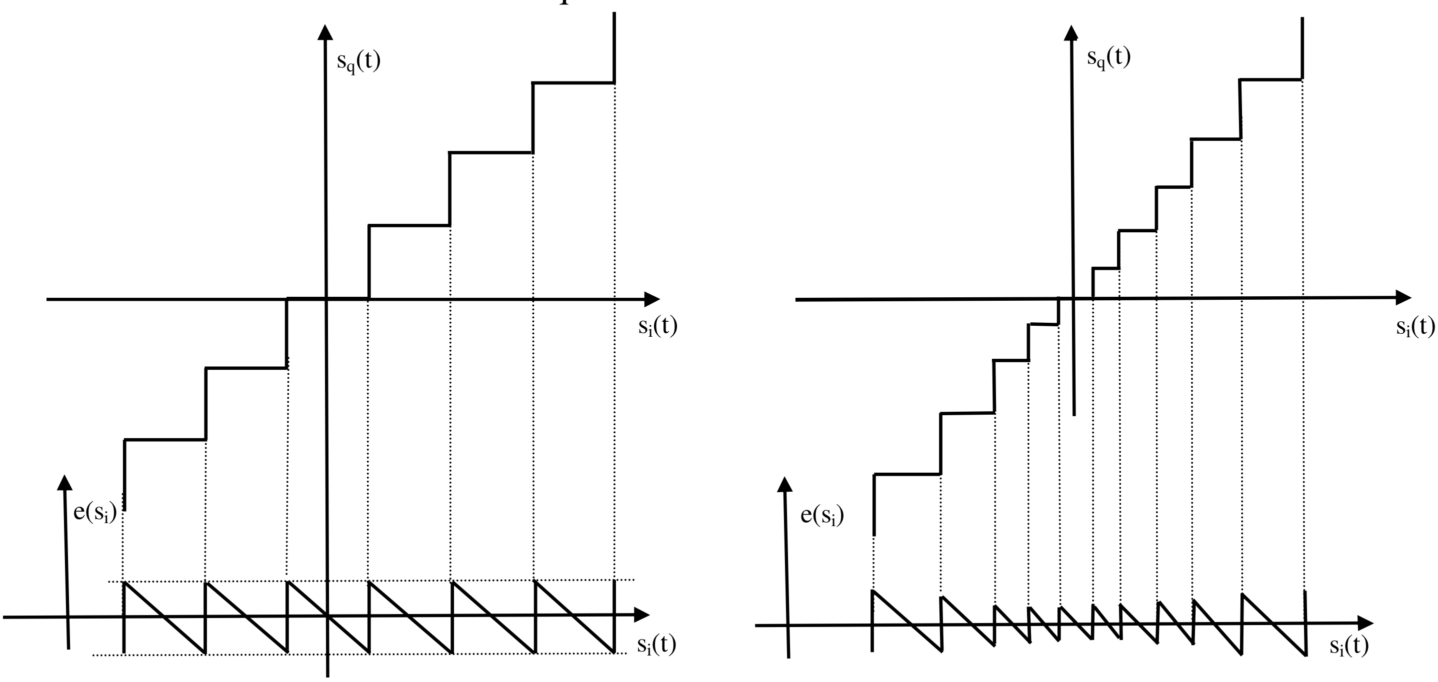


Fig. 4 Quantization techniques a) uniform quantization b) non-uniform

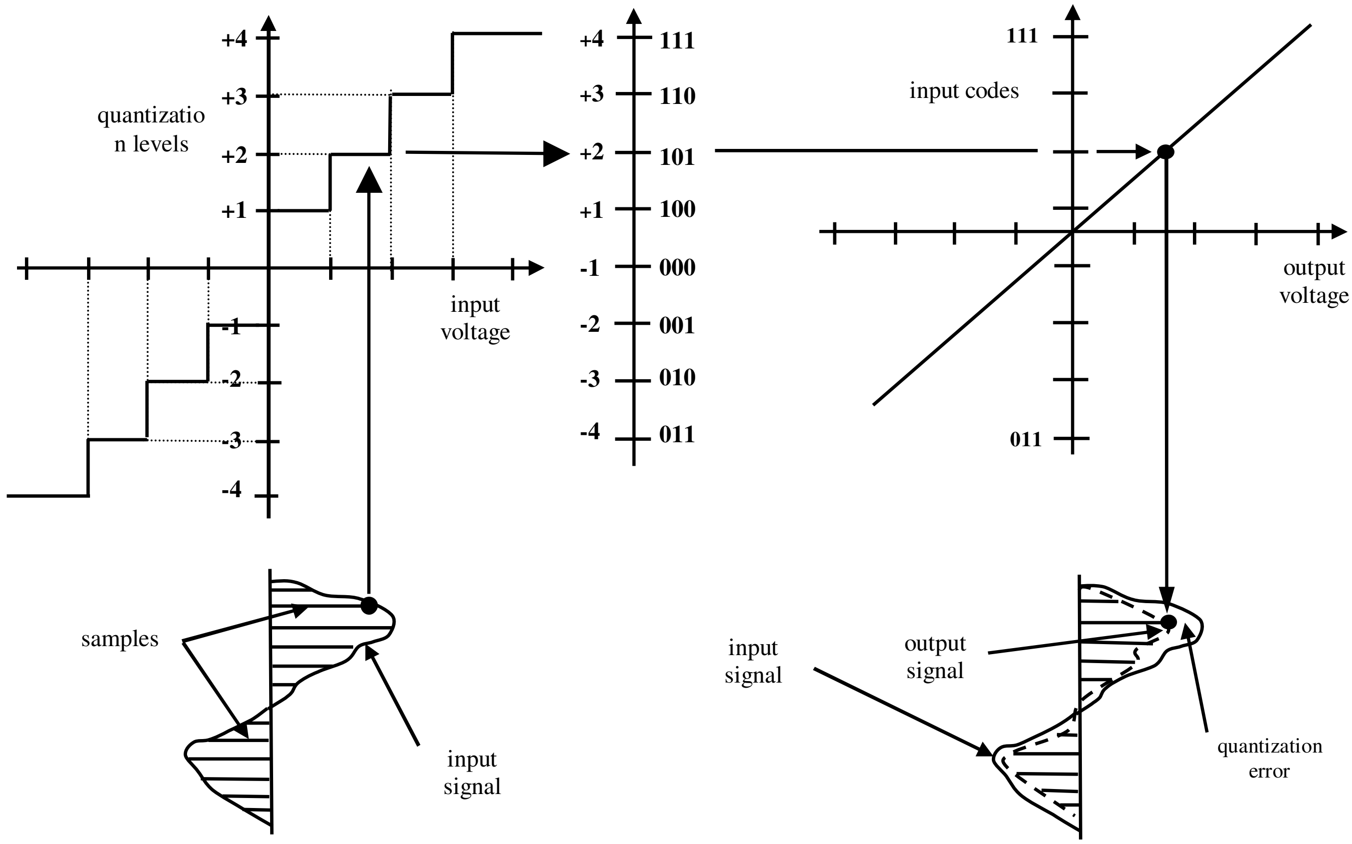


Fig. 5 Explanation of PCM coding and decoding process in the case of uniform quantization with 3 bits/sample

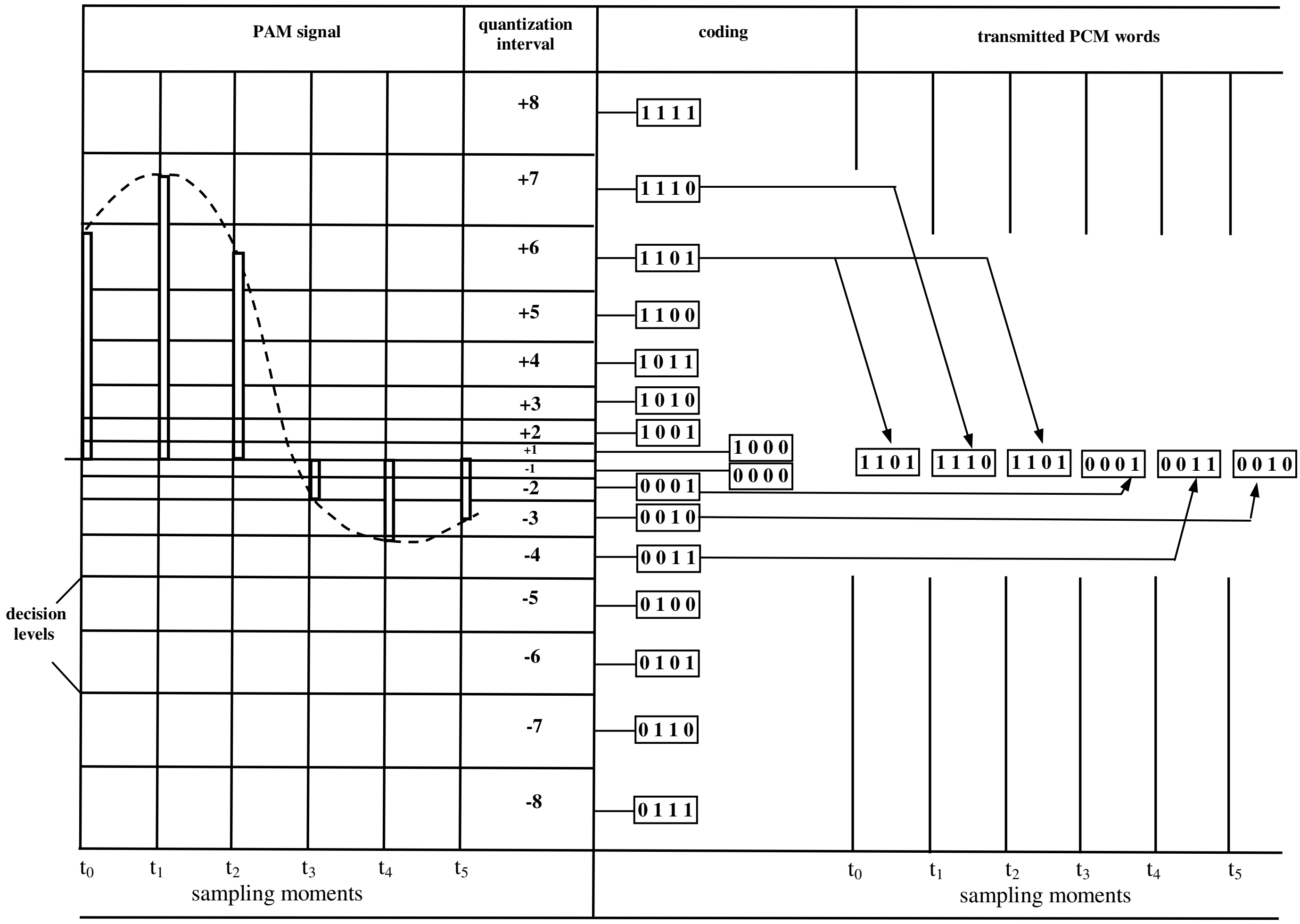


Fig. 6 Explanation of the PCM coding and decoding process in the case of nonuniform quantization with 3bits/sample

- An important parameter which characterizes the PCM modulation is the signal/ quantization noise ratio, the quantization error being considered a noise signal. A general expression of the mean quantization noise power is given by:

$$P_q = \sum_{i=1}^{N} p_i \cdot P_{qi} \qquad (4)$$

where N is the no. of quantization intervals, p_i is the probability that the transmitted signal is located in the i-th quantization interval, P_{qi} is the power of the quantization noise in interval i.

  - If the dynamic range of the transmitted signal is 2V and width of the quantization intervals are Δ_i, then the p_i probabilities are given by: $p_i = \frac{\Delta_i}{2V}$ (5)
  - Considering the uniform distribution of the quantization error inside a quantization interval, the power of the quantization noise in interval i is given by:

$$P_{qi} = \int_{-\frac{\Delta_i}{2}}^{+\frac{\Delta_i}{2}} \frac{1}{\Delta_i} \cdot e_r^2 \cdot de_r = \frac{\Delta_i^2}{12} \qquad (6)$$

  - The signal/quantization noise ratio, RSZ_q, is defined as: $RSZ_q = \frac{P_s}{P_q}$ (7)

- The implementation of the non-uniform quantization can be achieved use in analog or digital compression – see fig. 7.a and 7.b

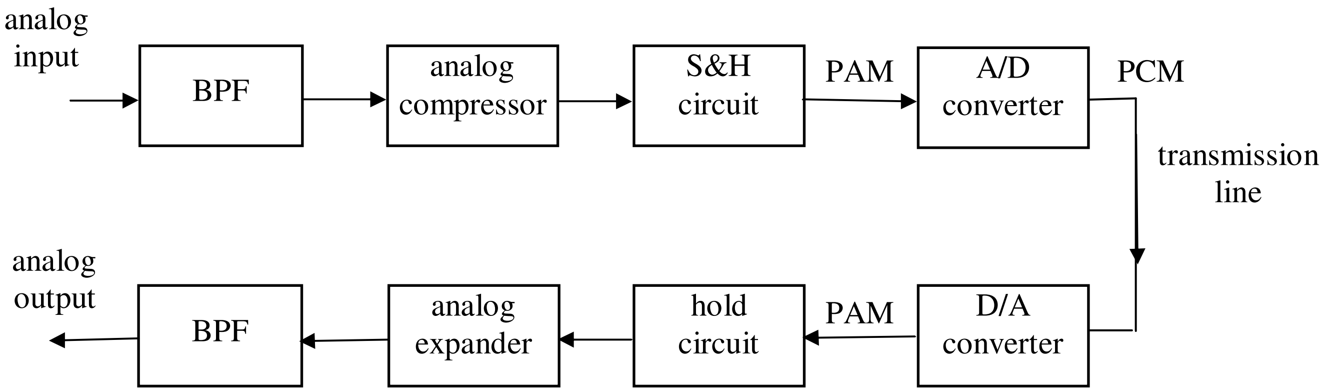


Fig. 7.a Processing sequence required by the PCM coding-decoding in the case of analog compression

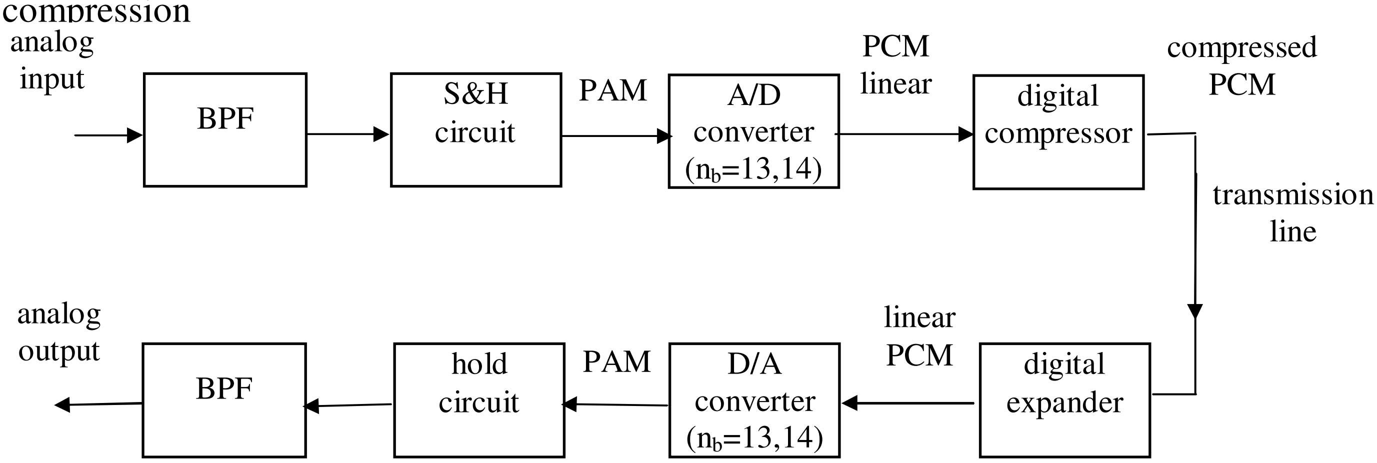
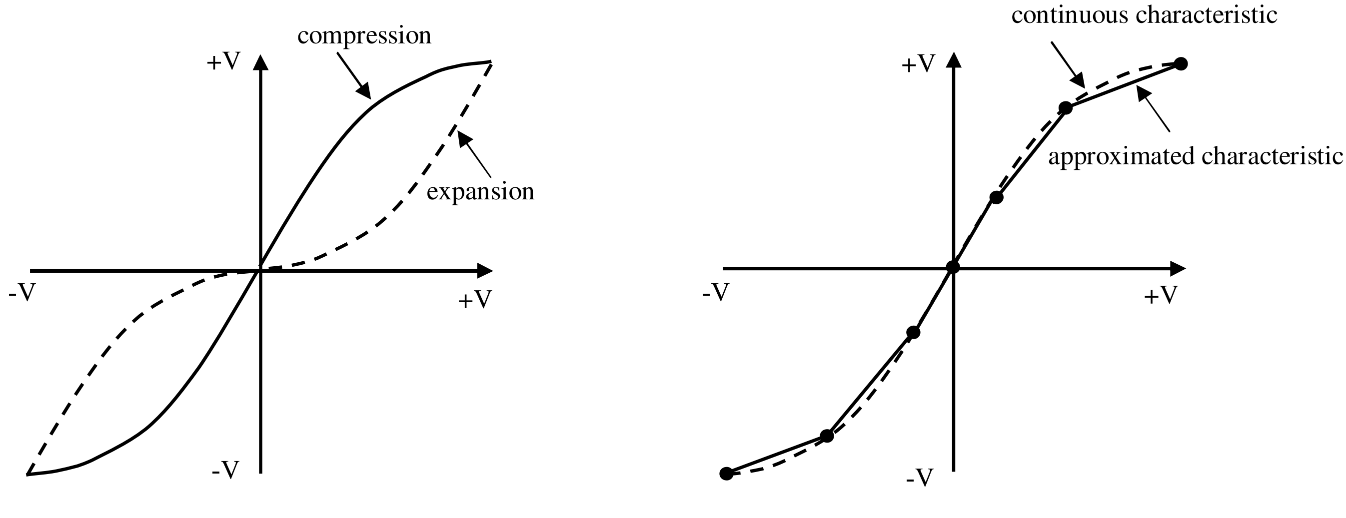


Fig. 7.b Processing sequence required by the PCM coding-decoding in the case of digital compression

- Continuous and approximated characteristics



a) Continuous characteristic b) Segmented characteristic

Fig. 8 Continuous and segmented compressed characteristics

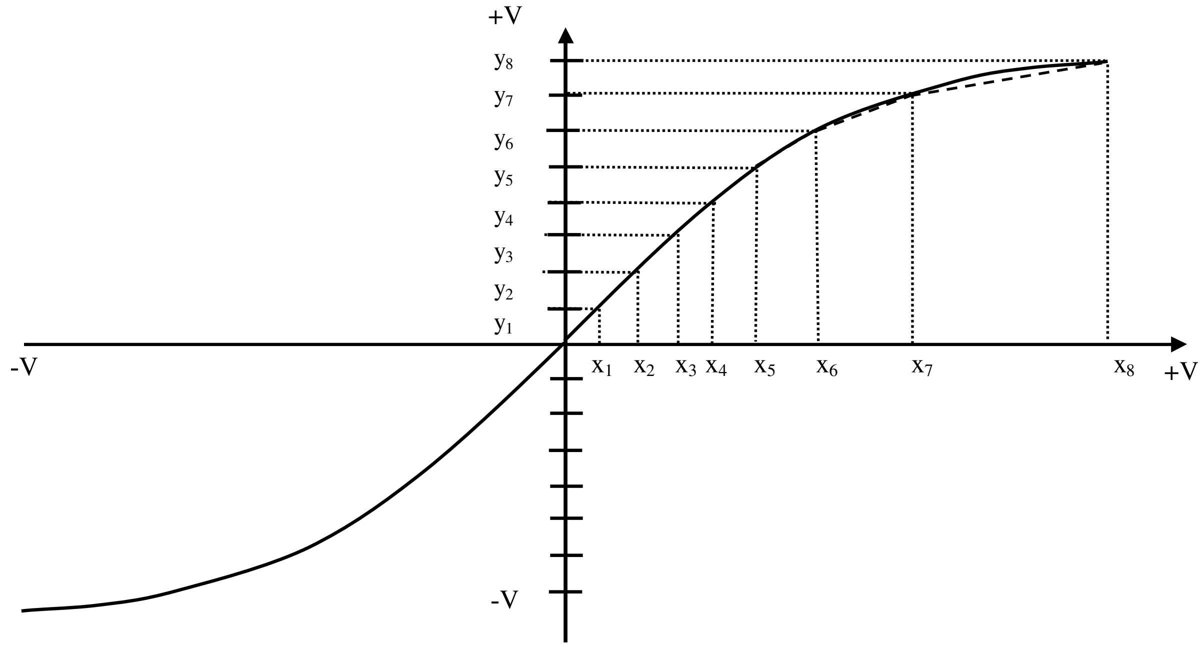


Fig. 9 Construction of the segmented compression characteristic

- **Compression laws used in digital telephone systems**
  - The μ compression law is given by:

$$y = \frac{\operatorname{sgn}(x)\cdot \ln(1+\mu\cdot|x|)}{\ln(1+\mu)} \quad ; \quad -1 \le x \le 1 \tag{8}$$

$$x = \operatorname{sgn}(y)\cdot\frac{1}{\mu}\cdot\left[(1+\mu)^{|y|}-1\right] \quad ; \quad -1 \le y \le 1 \tag{9}$$

- The A compression law is given by the following relations for input values x≥0

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

$$y = \frac{Ax}{1+\ln(A)}\text{; pentru } 0 < x < \frac{1}{A} \qquad\qquad x = \frac{\exp(y\cdot(1+\ln(A))-1)}{A} \quad ; \quad \frac{1}{1+\ln A} \le y \le 1$$

(10) (11)

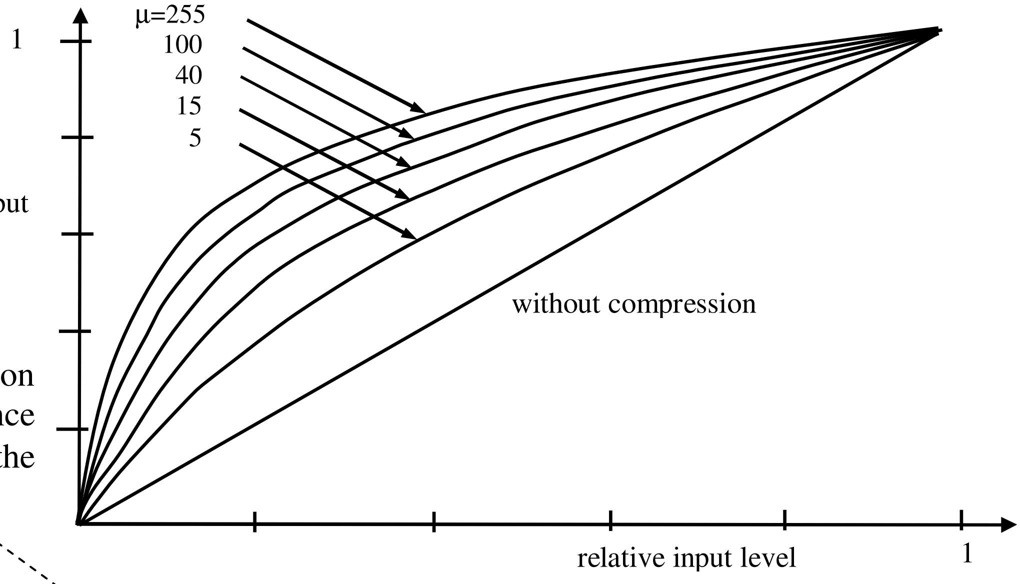


Fig. 10 The μ compression characteristic. The influence of parameter μ on the compression characteristic

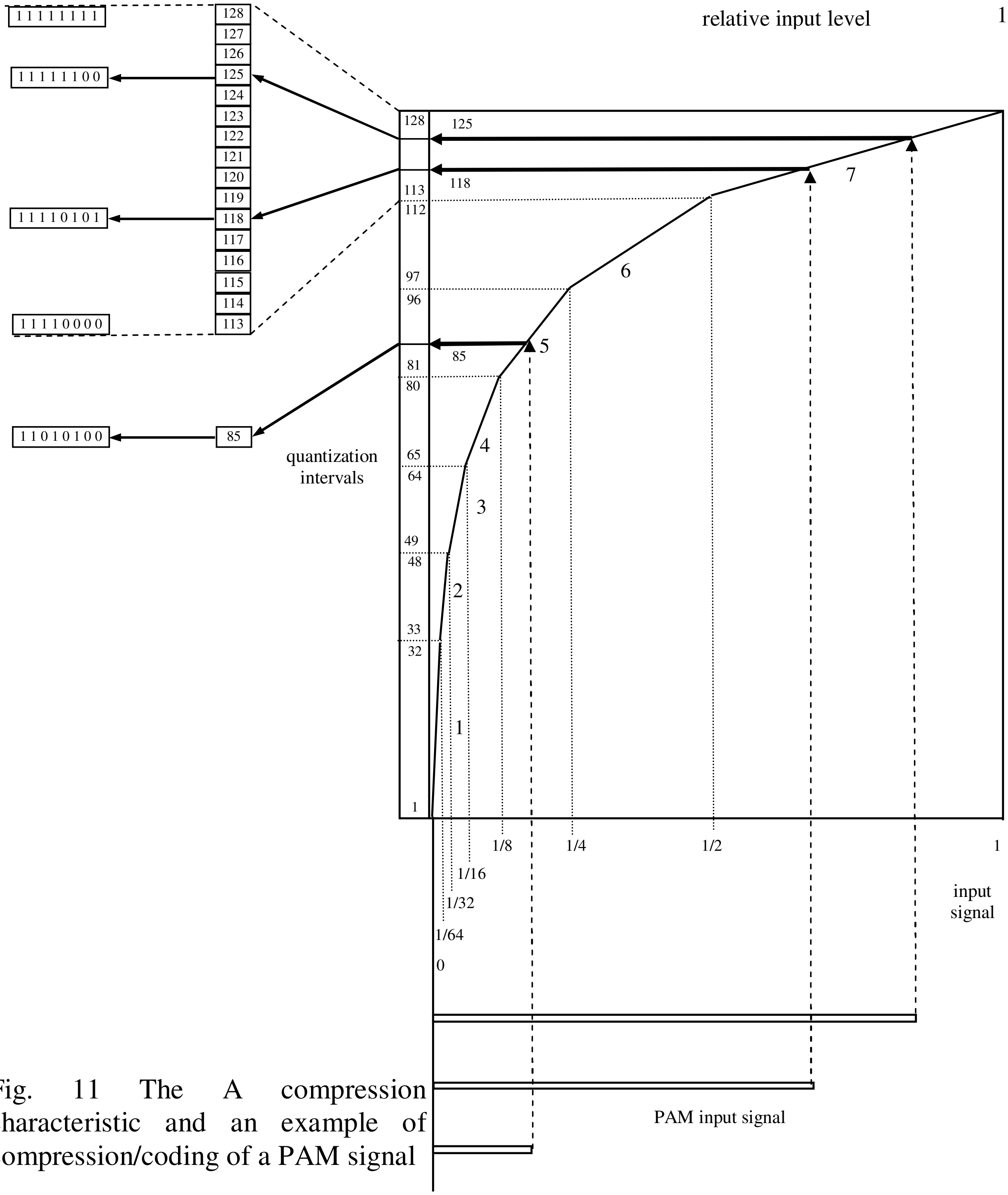


Fig. 11 The A compression characteristic and an example of compression/coding of a PAM signal

- 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 if the diferential PCM modulation;
- The DPCM modulation:
  - The next sample is predicted based on the previous samples and it si coded (quantized) only the difference between the current sample, $x(kT_e)=x_k$, and the predicted sample, $x\hat{}_k\ (kT_e)=x\hat{}_k$;
  - If the diffrence 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;

$$d_k = x_k \quad \hat{x}_k$$

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

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

$$C = \frac{\overline{x_k \cdot \hat{x}_k}}{\overline{x_k^2}} \Rightarrow \overline{d_k^2} = 2\overline{x_k^2} \cdot (1 - C)$$

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

    - 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;
  - Can not be used with the same parameters for voice and date;
  - If there are errors on the line there are affected several samples;

Block schematic of the DPCM coder

- The predictiction circuit works with N samples;

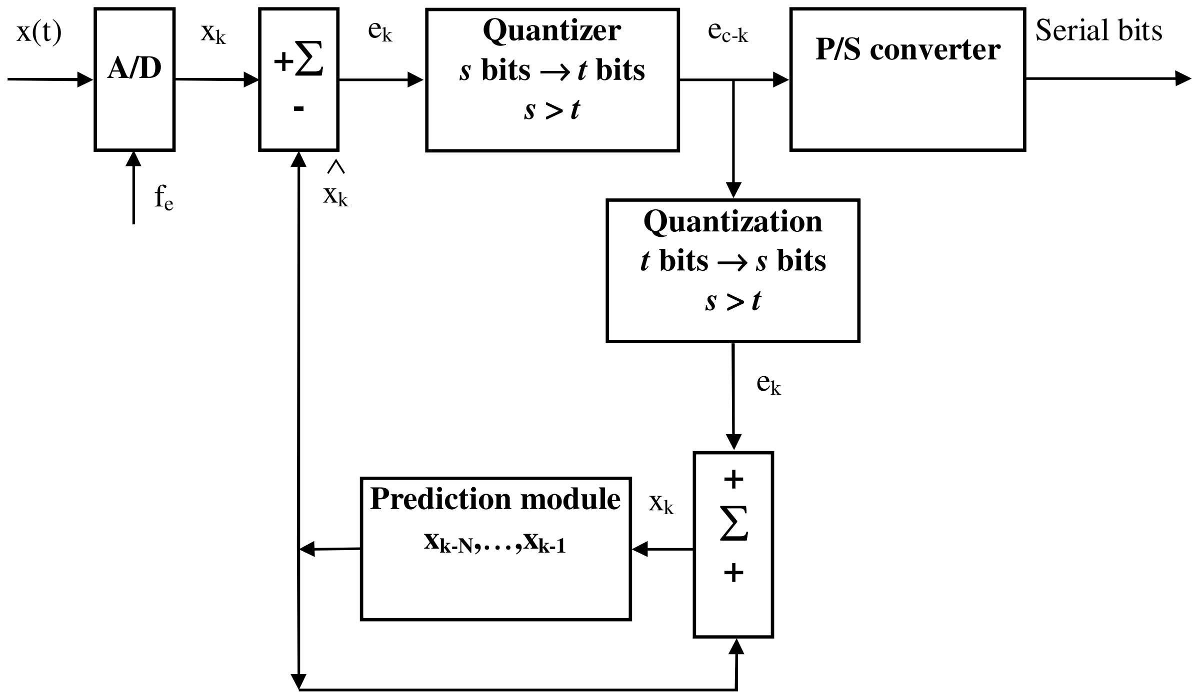


Fig. 12.a Block schematic of the DPCM coder

Block schematic of the decoder

- The predictor works with N samples;
- q_k – quantization errors;

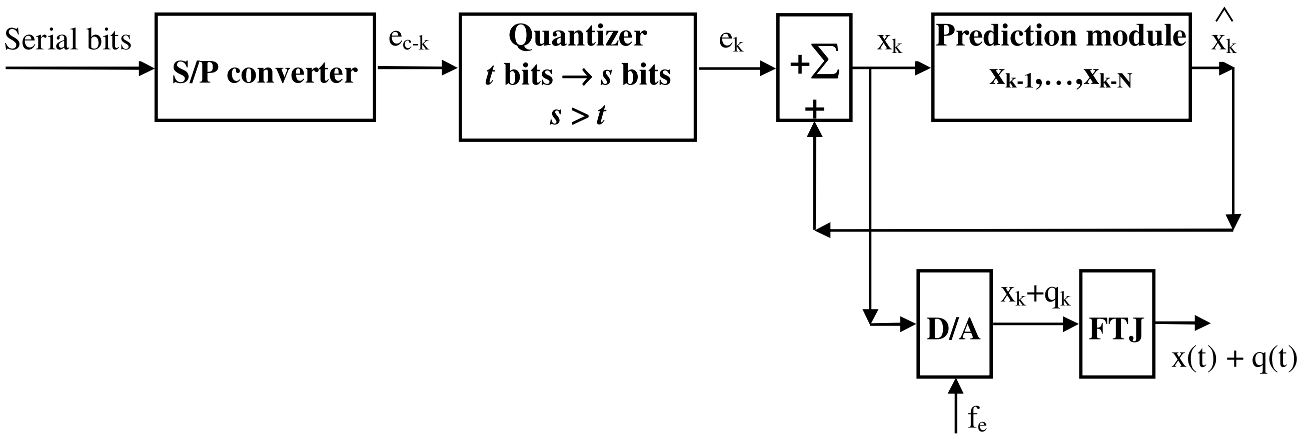


Fig. 12.b Block schematic of the DPCM decoder

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:

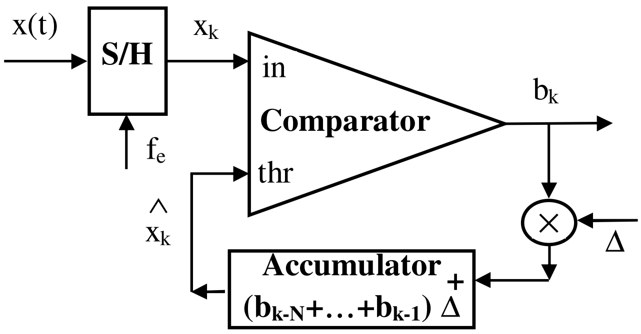


Fig. 13.a Block schematic of the non-adaptive Delta encoder

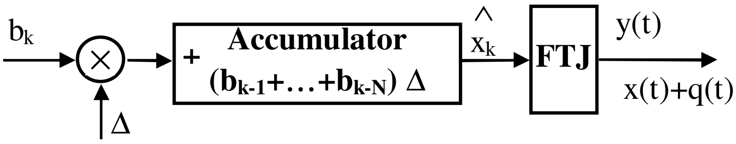


Fig. 13.b Block schematic of the Delta decoder

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:

$$\begin{cases} x_k \ge \bar{x}_k \Rightarrow b_k = '1' \ (+1) \\ x_k < \bar{x}_k \Rightarrow b_k = '0' \ (-1) \end{cases}$$

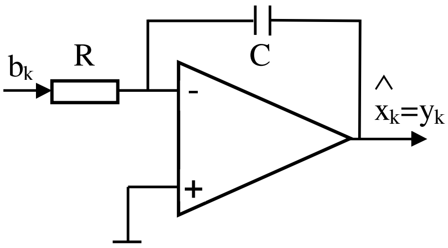
  - Computation of the predicted signal (the equation of the accumulator):

$$\bar{x}_k = \bar{x}_{k-1} + b_{k-1} \cdot \Delta_{k-1}$$

  - Computation of the quantization step:

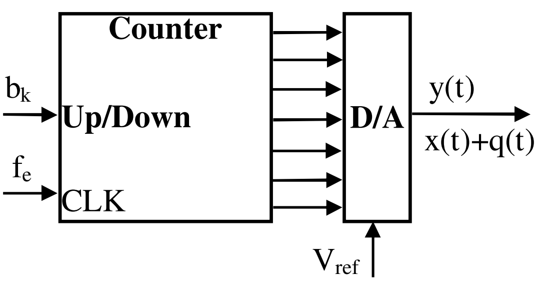
$$\Delta_k = f\left(\Delta_{k-1}, b_k, b_{k-1}, \cdots, b_{k-N}\right)$$

  - In the case of non-adaptive Delta modulation the quantization step is constant = Δ;
- Implementation methods of the accumulator:
  - Analog and digital implementation of the accumulator:



$$|\Delta V_c| = \frac{1}{R \cdot C} \int_0^{T_e} b_k\, dt = \frac{b_k \cdot T_e}{R \cdot C} = \Delta \ ; if\ b_k = 1 \Rightarrow \frac{T_e}{R \cdot C} = \Delta$$

Fig. 14.a Analogue implementation of the accumulator using an integrator



- N – number of bits of the D/A converter;

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

Fig. 14.b Digital implementation of the accumulator using a counter

Distorsions characteristic to Delta modulations:

- The slope overload distortions:
  - Appears if the slope of the source signal is larger than that of the predicted signal;

$$\begin{cases} source\ signal\ slope = \left|\dfrac{dx(t)}{dt}\right| \\ Delta\ signal\ slope = \dfrac{\Delta}{T_e} \end{cases}$$

- The granular distorsion (granular noise)
  - Represents a quantization noise;
  - Apperas if the slope of the source signal is smaller than that of the tredicted signal;

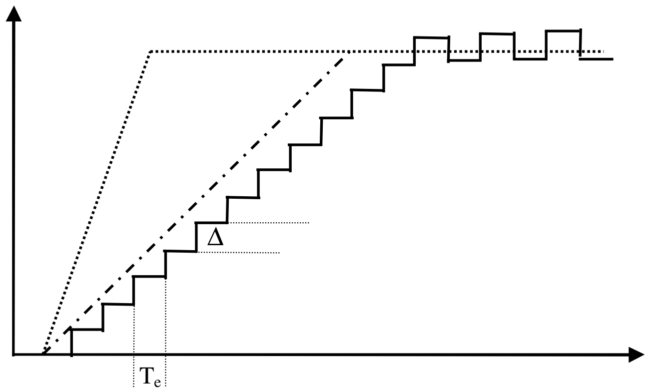


Fig. 15 Distortions affecting the Delta modulated signal

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

- It is considered f_e=2f_m (f_m – the maximum frequncy of the spetrum);
- 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;

$$\begin{cases} \left|\dfrac{dx(t)}{dt}\right| = \Delta \cdot f_e \\ \left|\dfrac{dx(t)}{dt}\right| \approx K \cdot \sqrt{P} \end{cases}$$

- The quantization noise is computed in the following way:

$$\begin{cases} P = \dfrac{\Delta^2 \cdot f_e^2}{K^2} \\ P_{zg-q} = \dfrac{\Delta^2}{12} \end{cases}$$

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

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

Adative Delta modulation

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

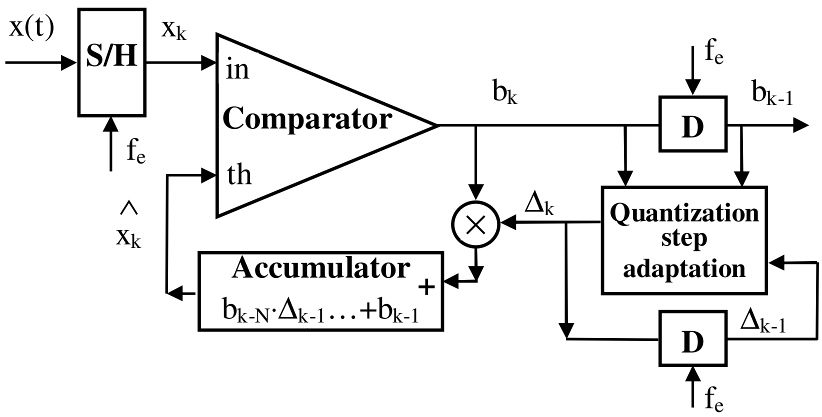


Fig. 16.a Block schematic of the adaptive Delta encoder

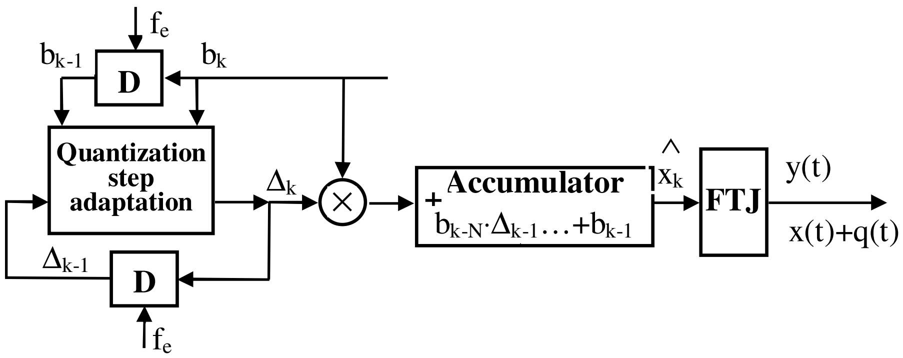


Fig. 16.b Block schematic of the adaptive Delta decoder

Types of adaptive Delta modulations – according to the quantization step modification rules:

- The Song modulation;
  - The quantization step modification rule:

$$\begin{cases} \Delta_k = \Delta_{k-1} + \Delta_0 \ \ if \ \ b_k = b_{k-1} \\ \Delta_k = \Delta_{k-1} - \Delta_0 \ \ if \ \ b_k \neq b_{k-1} \\ \quad if \ \Delta_k < \Delta_0 \ \Rightarrow \ \Delta_k = \Delta_0 \end{cases}$$

- The Jayant modulation;
  - The quantization step modification rule:

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

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

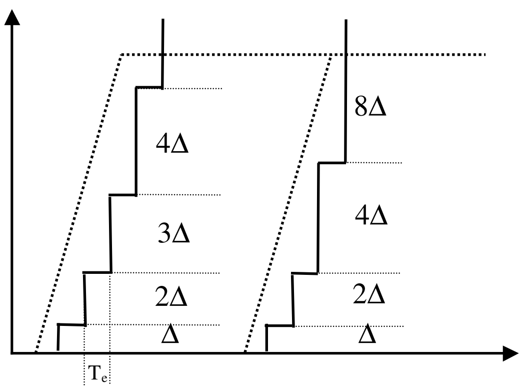


Fig. 17 Modification of the quantization step in the case of adaptive Delta modulations

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

$$epm = \frac{\sum_{k=1}^{M}(x_k - \bar{x}_k)^2}{M}$$

    - The predicted signal represents the demodulated signal;

3. PCM primary multiplex

- The PCM multiplexing is the first level of multiplexing
  - uses a time division multiplexing of the telephone channels being strongly connected with the switching process.

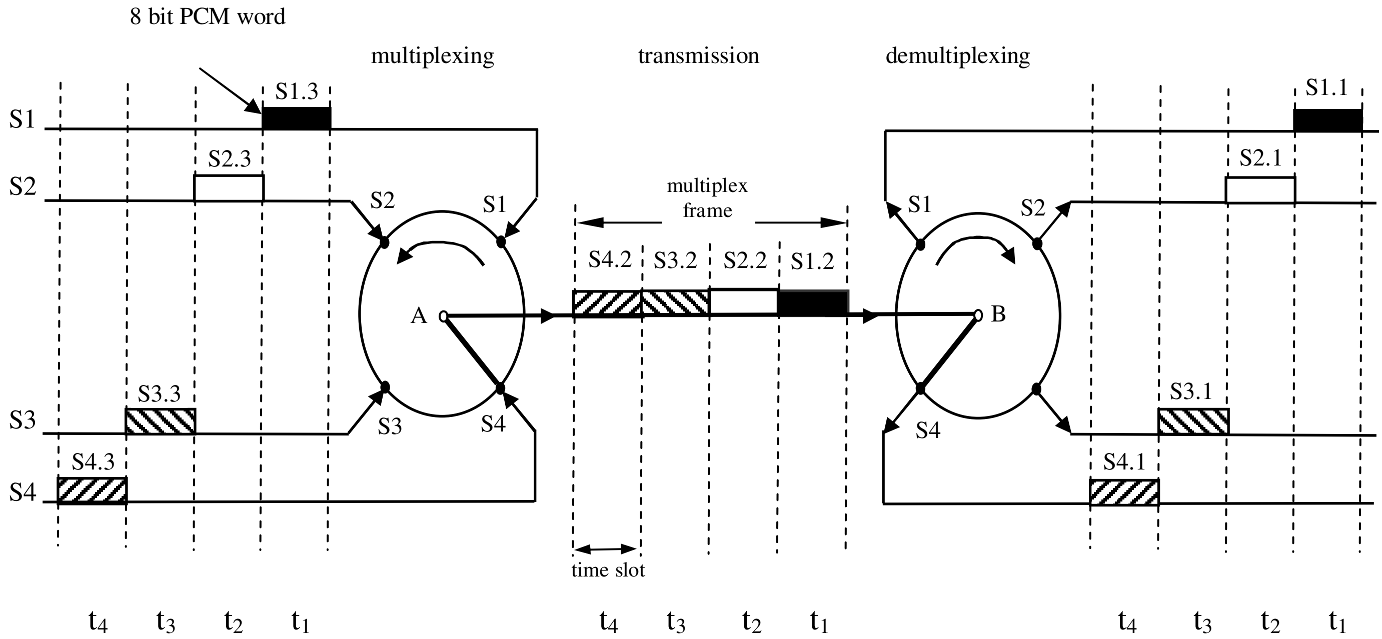


Fig. 12 Principle of the PCM multiplexing

  - 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, to each word corresponding a separate time slot.
  - The bit assigned to the multiplex frame must be N times larger then 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 intervals assigned to different channels and the transmission of the words extracted from the time slots to the destination using a bit rate characteristic to this equipments.

3.1 Types of PCM multiplex frames

3.1.1 The E1 PCM frame used in Europe

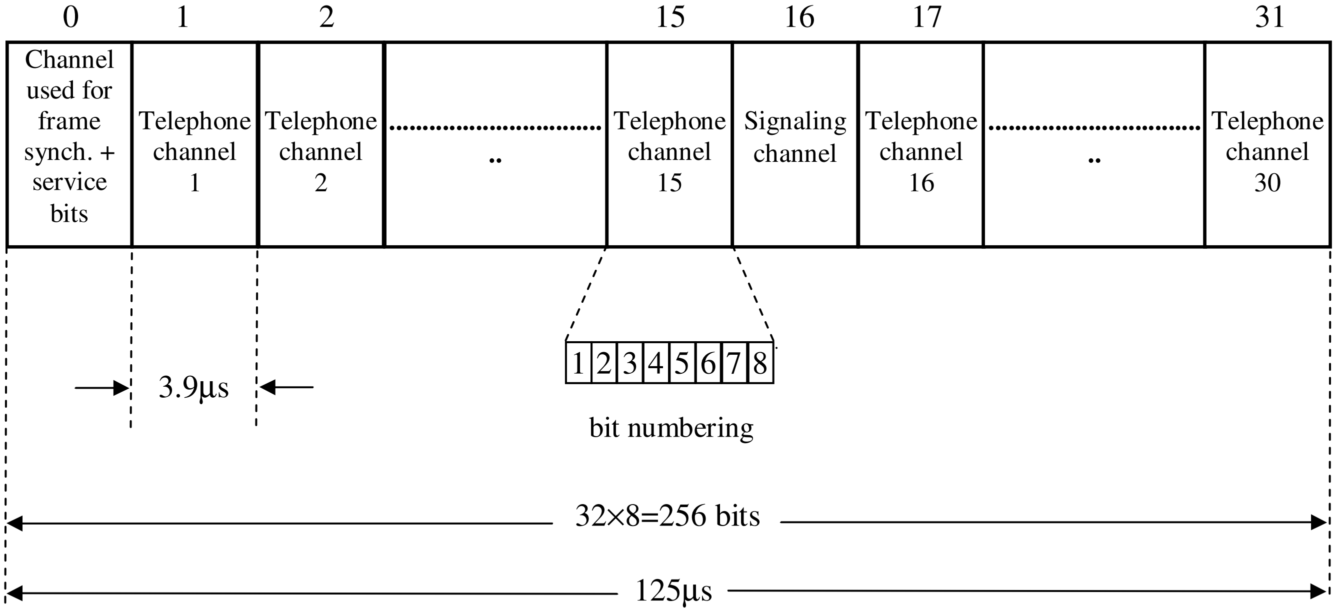


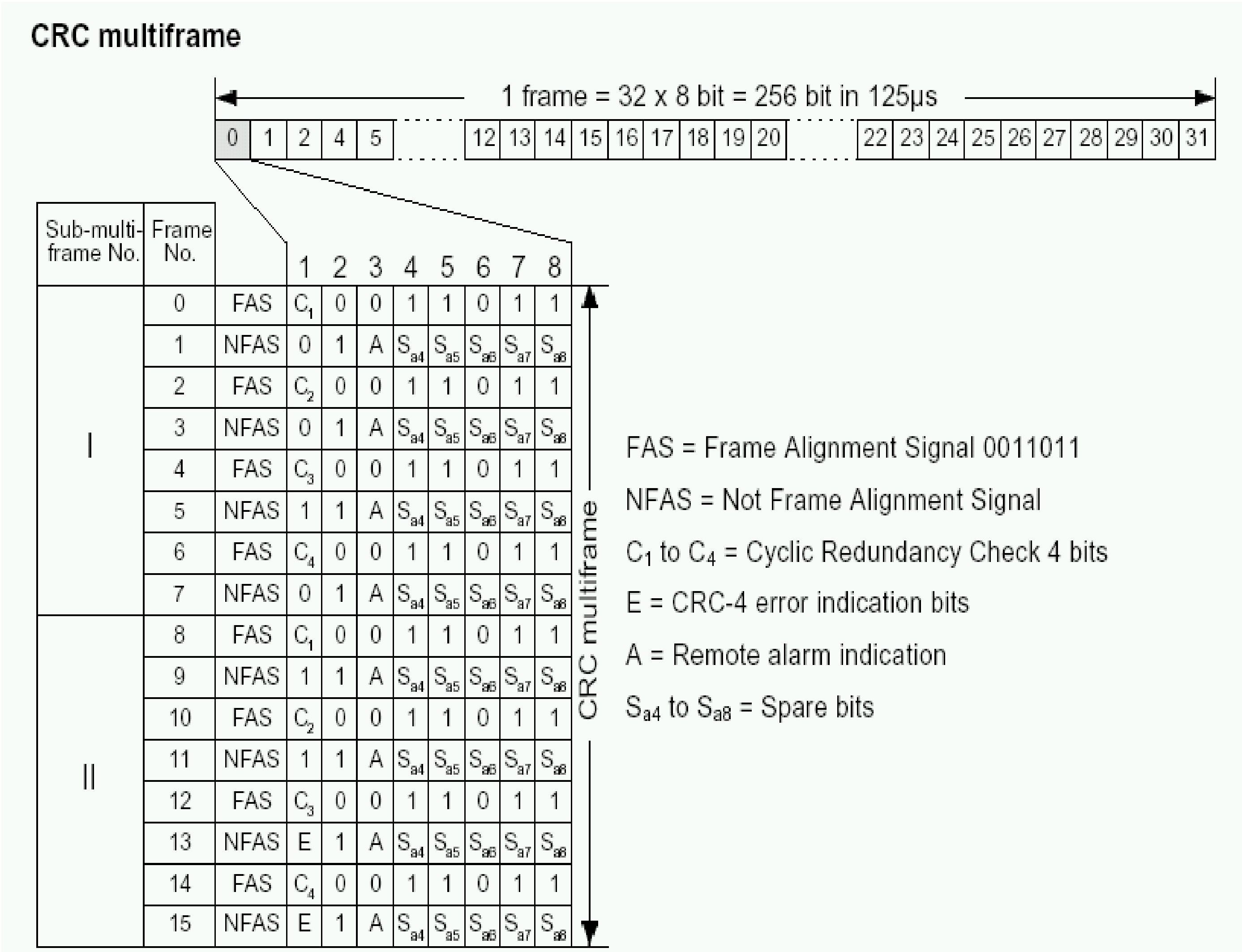
Fig. 13 The structure of the E1 multiplex frame

- The E1 frame includes 32 elementary channels of 64kbps, the bit rate associated to this frame being 2,048Mbps, precision: ±50ppm.
- 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 and channel 16 is used for multiframe synchronizations, service bits and signaling operations, this channel 16 being dedicated especially to signaling operations.
- There are two operation modes on channel 16, namely: channel associated signaling – CAS and common channel signaling CCS; for the management of the CAS signaling a multiframe is composed using 16 PCM frames.
- There are two operation modes on channel 0, namely: normal mode without CRC (*Cyclic Redundancy Check*) and CRC-4 mode, which uses CRC error control.

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			

Tab. 2 Structure of the E1 PCM multiframe.
Normal operation on slot 0 and CAS signaling on slot 16

- TS0 in even frames: Y0011011 – frame synchronization word ; 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 word ; Z – multiframe synch. loss alarm bit ; X – not used (national bits)



Sub-multi-frame No.	Frame No.		1	2	3	4	5	6	7	8
I	0	FAS	C_1	0	0	1	1	0	1	1
	1	NFAS	0	1	A	S_{a4}	S_{a5}	S_{a6}	S_{a7}	S_{a8}
	2	FAS	C_2	0	0	1	1	0	1	1
	3	NFAS	0	1	A	S_{a4}	S_{a5}	S_{a6}	S_{a7}	S_{a8}
	4	FAS	C_3	0	0	1	1	0	1	1
	5	NFAS	1	1	A	S_{a4}	S_{a5}	S_{a6}	S_{a7}	S_{a8}
	6	FAS	C_4	0	0	1	1	0	1	1
	7	NFAS	0	1	A	S_{a4}	S_{a5}	S_{a6}	S_{a7}	S_{a8}
II	8	FAS	C_1	0	0	1	1	0	1	1
	9	NFAS	1	1	A	S_{a4}	S_{a5}	S_{a6}	S_{a7}	S_{a8}
	10	FAS	C_2	0	0	1	1	0	1	1
	11	NFAS	1	1	A	S_{a4}	S_{a5}	S_{a6}	S_{a7}	S_{a8}
	12	FAS	C_3	0	0	1	1	0	1	1
	13	NFAS	E	1	A	S_{a4}	S_{a5}	S_{a6}	S_{a7}	S_{a8}
	14	FAS	C_4	0	0	1	1	0	1	1
	15	NFAS	E	1	A	S_{a4}	S_{a5}	S_{a6}	S_{a7}	S_{a8}

Tab. 3 The structure of the E1 PCM multiframe. CRC-4 operation mode on slot 0

- In CRC-4 mode on slotul 0 the Y bits from frames with even number are used to transmit CRC sequences on 4 bits.
  - In 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 and in 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:

$$p(x) = x^4 + x + 1 \qquad (13)$$

  - The nondetection probability of bit error packets with more then 4 error is 6.25%; detection probability of these error packets is 93.75%; all error packets with at most 4 errors are detected.
  - In the normal frame, without CRC can be monitored only 7 bits (the frame synchronization bits) in each group of 505 bits.

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

- **Loss of multiframe synchronization in the CAS case:** two consecutive MFAS signals with errors or two multiframes with all zero bits in slot 16.
- **Frame and multiframe synchronization in the considered operation modes**
  - **Normal frame synchronization:** correctly received FAS, bit two in NFAS 1, next FAS received correctly.
  - **CAS multiframe synchronization:** correctly received MFAS and slot 16 of the previous frame is not zero.
  - **CRC multiframe synchronization:** bit 1 of NFAS frames generates the sequence: 0 0 1 0 1 1; 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.

- Alarms associated to frame E1. Terms associated to alarm events
  - 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 then 10^{-3} – any of these event generates a red alarm at the end where they take place; the equipment which receives bit Z=1, declares yelow alarm;
  - multiframe alarm (remote alarm) bit: bit Z of slot 16 frame 0 (called also bit Y); value 0 – normal operation, value 1 – loss of MFAS signal (yellow alarm transmitted to the opposite end);
    - using these bits is signaled the remote alarm – RAI – „Remote Alarm Indication”
  - AIS – „Alarm Indication Signal” – called also keep alive signal
    - 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 equipments, or just in slot 16 it is transmitted a continuous 1 (MFAS error); the terminal equipment detects the AIS signal and declares AIS state called also blue alarm;
    - generated by a multiplexer when it receives a yellow alarm from the opposite end – it is continuous signal – can be detected by the equipment at the opposite end (if we do not have LOS or LOF) and this equipment declares AIS state;
    - generated by a multiplexer toward a terminal equipment when receives a yellow alarm;
    - the AIS represents at least 509 one bits in a bloc of 512 bits or less than 3 zero bits in 2 frames (in the case of slot 16 less than 3 zero in this slot during two consecutive multiframes);

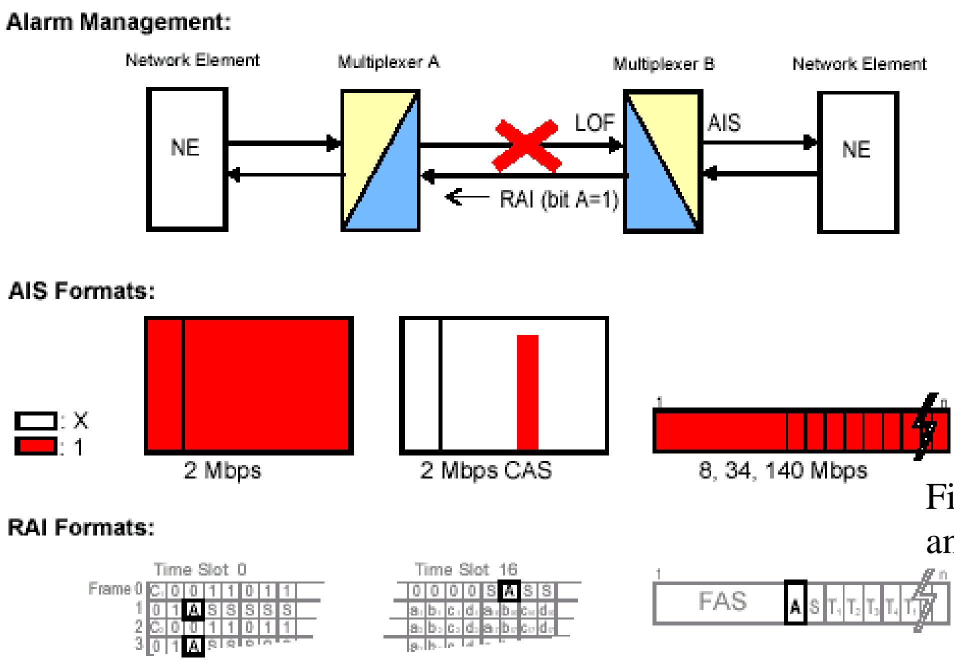


Fig. 14 Alarm management and AIS formats

- The LOS and LOF events generate a red alarm
- Transmission of E1 frame
  - 4 wire full duplex.
  - AMI (*Alternate Mark Inversion*) coding – the logical zero bit is coded with a 0V level and the logical one bits are alternatively coded with ±A impulses – this code has no DC component (avoiding saturation of the separation transformers core), has relatively narrow bandwidth, simple decoding but 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.

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 -

Tab. 3 HDB3 coding rule

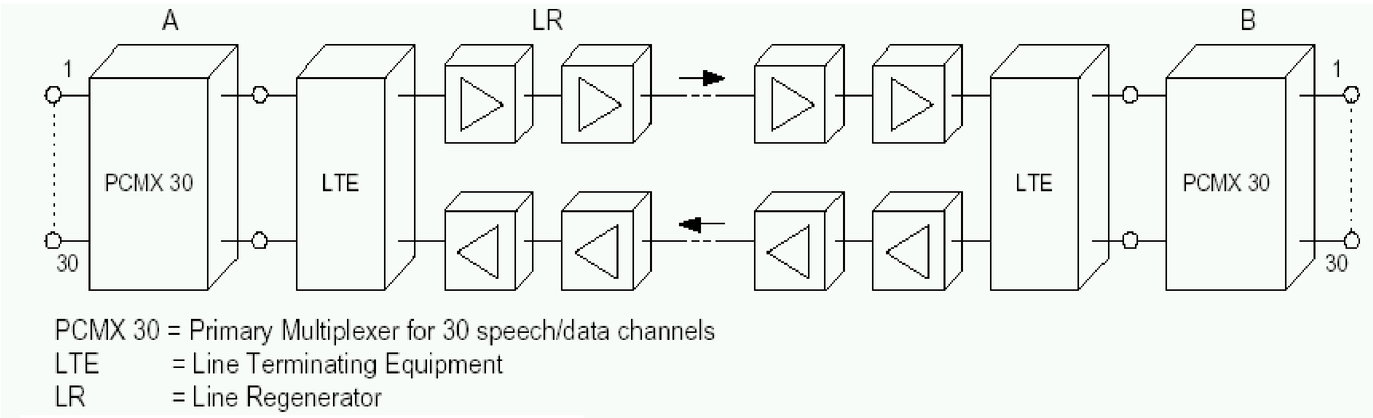


Fig. 15 PCM transmission system

  - E1 interface characteristics

ITU-T G.703 RECOMMENDATION

Pulse shape (nominally rectangular)	All marks of a valid signal must conform with the mask (see figure 15/G.703) irrespective of the sign. The value V corresponds to the nominal peak value.	
Pair(s) in each direction	One coaxial pair	One symmetrical pair
Test load impedance	75 ohms resistive	120 ohms resistive
Nominal peak voltage of a mark (pulse)	2.37V	3V
Peak voltage of a space (no pulse)	0 ± 0.237V	0 ± 0.3V
Nominal pulse width	244ns	
Ratio of the amplitudes of positive and negative at the centre of the pulse interval	0.95 to 1.05	
Ratio of the widths of positive and negative pulse the nominal half amplitude	0.95 to 1.05	
Maximum peak-to-peak jitter at an output port	Refer to Section 2 of Recommendation G.823	

Tab. 4 Main characteristics of E1 interface

- It is specified a mask of the coded impulse and a frequency characteristic of the jitter

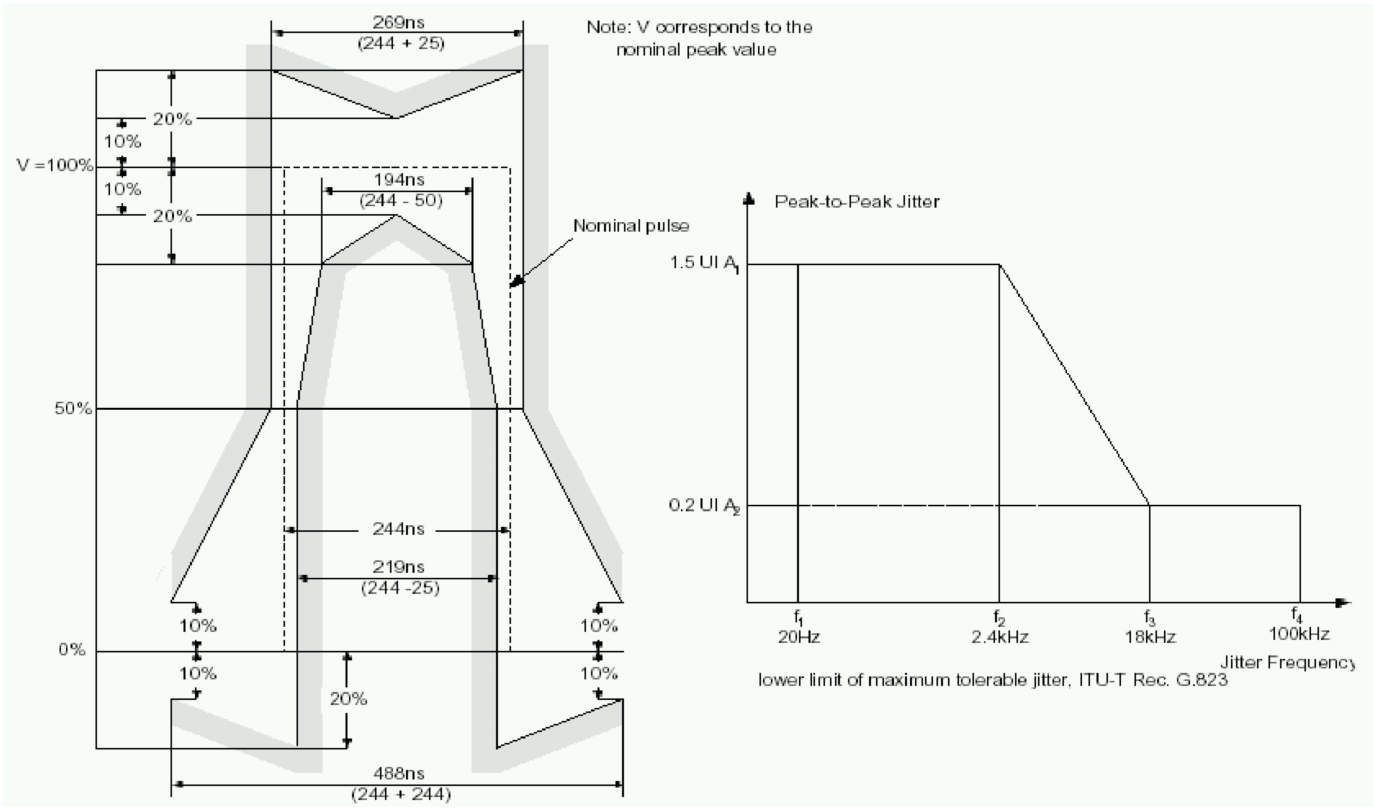


Fig. 16 Coded impulse mask and jitter freq. characteristic

3.1.2 The T1 (DS1) PCM multiplex frame used in USA

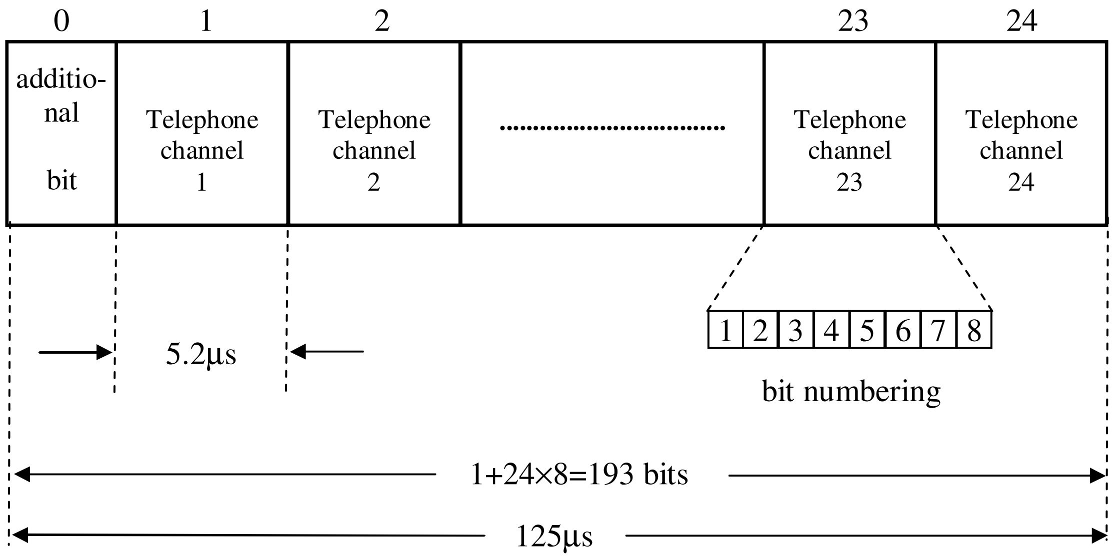


Fig. 17 Structure of the T1 PCM multiplex frame

- The T1 multiplex frame includes 24 telephone channels + 1 bit additional, the F bit, which is used for synchronization or for implementation of a special data channel.
- There are two multiframe formats, namely: SF – *Super Frame* composed of 12 frames and ESF – *Extended Super Frame* composed of 24 frames.
- The SF multiframe has no separate time slot for synchronization or signaling.
  - The frame and multiframe synchronization is accomplished with the help of the supplementary F bit and 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.

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

Tab. 5 Structure of the T1 SF multiframe with CAS signaling

- In the case of EFS multiframe composed of 24 frames, the F is used to frame and multiframe synchronization – special sequence, having the structure 0 0 1 0 1 1, located in frames with even number multiple of 4, implements a 4kbps bit rate data channel, the M channel (management, control, alarms), in odd frames, and transmits a CRC-6 control sequence in even frames which are not multiple of 4.
    - The transmission of the signaling is accomplished in a similar way as in the case of SF multiframe, the 8-th of each channel of each channel of every sixth frame is used for signaling - 4 bit CAS signaling for each channel, bits A B C and D.
- The transmission of T1 frames is similar with those of frames E1 – 4 wire full with repeaters from 1.5 in 1.5km
- The used coding is B8ZS *(Bipolar with 8 Zero Substitution)*, AMI type cod which replaces the groups of 8 zero consecutive bits with a coded sequence having the structure:0 0 0 0 V 1 0 V 1, meaning 4 zero bits, a violation of the AMI coding rule, followed by 1 0 normally coded and after that a new violation of the AMI coding rule and finally a 1 normally coded.
- the used CRC mechanism detects all error packets with at most 6 errors and detects 98.4% of error packets with more than 6 errors.
- On the M data channel two type of signals can be transmitted:
    - bit oriented signals which are unscheduled messages; they begin with a one byte followed by a zero bit, a command/message identifier on 6 bits and finally a zero bit; the 6 bit identifier codes alarms and different messages: protection switching activation, loop-back activation, a.s.o. The yellow alarm is coded: 1111111 0000000;
        - 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 (error level, CRC errors, synch. errors, coding rule violations) and are controlled by a communication protocol; can be interrupted by bit oriented signals.
- In the case of SF frame the yellow alarm is transmitted by setting the bit no. 2 of each slot to 0.

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

Tab. 6 Structure of the ESF T1 multiframe with CAS signaling

- **T1 alarms**
  - 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)

Parameter	Specification
Nominal line rate	1544 kbit/s
Line rate accuracy	In a self-timed, free running mode, the line rate accuracy shall be ±50 bits/s (±32 ppm) or better.
Line code	Either (1) AMI with no more than 15 consecutive zeros, and at least N ones in each and every time window of 8(N + 1) digit time slots (where N can range from 1 to 23), or (2) B8ZS (Note 1).
Frame structure	No frame structure is required for 1544 kbit/s transmission or higher level multiplexing to higher level DSN signals.
Medium	One balanced twisted pair shall be used for each direction of transmission.
Test load impedance	A resistive test load of 100 ohms± 5% shall be used at the interface for the evaluation of pulse shape and the electrical parameters specified below.
Pulse amplitude Pulse shape	The amplitude (Note 2) of an isolated pulse shall be between 2.4 V and 3.6 V. The shape of every pulse that approximates an isolated pulse (is preceded by four zeros and followed by one or more zeros) shall conform to the mask.
Power level Pulse imbalance	For an all-one signal, the power in a 3 kHz ± 1 kHz band centered at 772 kHz shall be between 12.6 dBm and 17.9 dBm. The power in a 3 kHz± 1 kHz band centered at 1544 kHz shall be at least 29 dB below that at 772 kHz. In any window of seventeen consecutive bits, the maximum variation in pulse amplitudes shall be less than 200 mV, and the maximum variation in pulse widths (half amplitude) shall be less than 20 ns.
DC power	There shall be no DC power applied at the interface.
Verification access	Access to the signal at the interface shall be provided for verification of these signal specifications.
NOTE 1 – B8ZS is one method of providing bit sequence independence. Bit sequence independence in turn allows unconstrained clear channel capability. Zero Byte Time Slot Interchange (ZBTSI) is another method of providing clear channel transmission. NOTE 2 – While both voltage and power requirements are given to assist in qualification of signals at the interface, the values are not equivalent. Voltage specifications are given for isolated pulses, while power levels are specified for all-ones signal.	

Tab. 7 Main characteristics of T1 interface

4. Transmission of data and of synchronization signals between data terminal equipments and multiplexers (local transmission between the equipments of a local switching/multiplexing point)

- There are two types of interfaces between the local equipments corresponding to two transmission strategy of data and synchronization signals
    - Codirectional interfaces – correspond to the case when each equipment transmits the data together with his own synchronization signal – all equipments must have the same clock synchronized from an external source.
    - Contradirectional interfaces – the multiplexer transmit the synchronization information for both transmission directions.

- **Codirectional interfaces**
  - A complex signal, combining both the information and the synchronization signals (bit clock and byte clock) is transmitted between the connected equipments; it is necessary a single channel composed of a pair of wire 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.

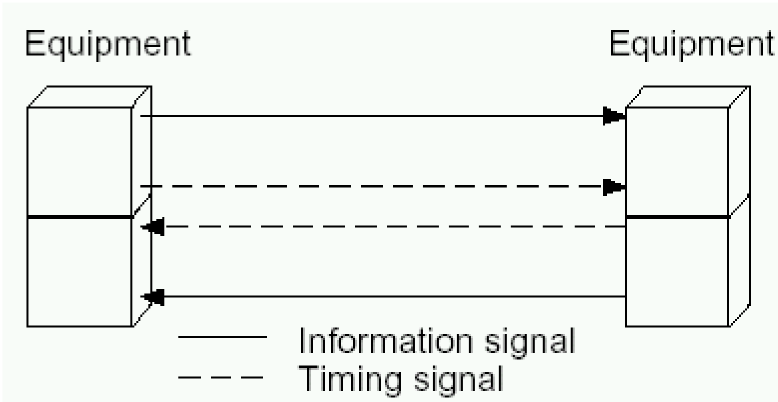


Fig. 18 Codirectional interfaces. Basic schematic.

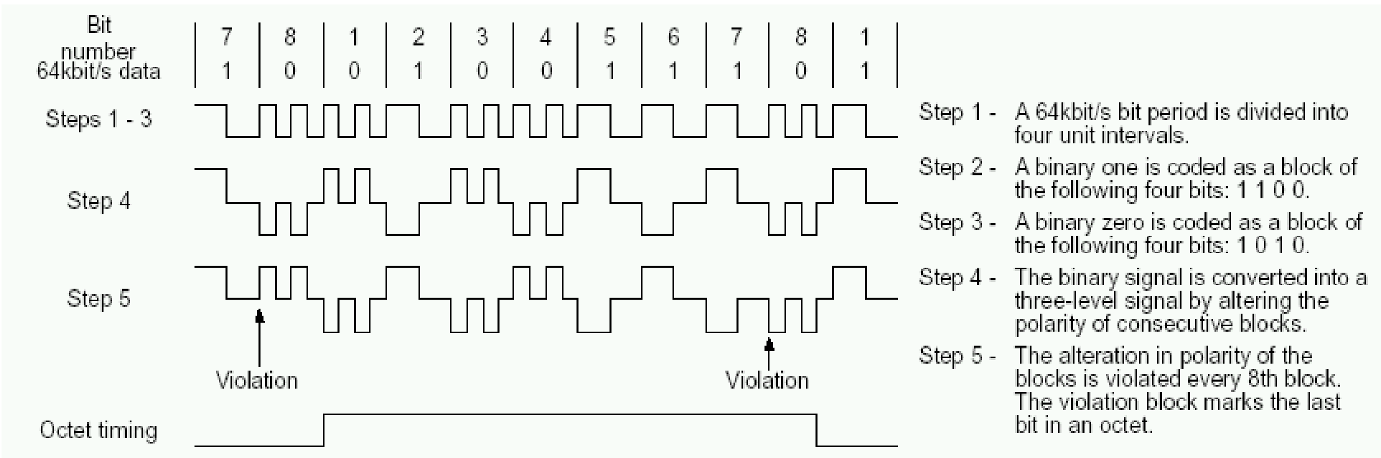


Fig. 19 Generation of the signals transmitted on the interface

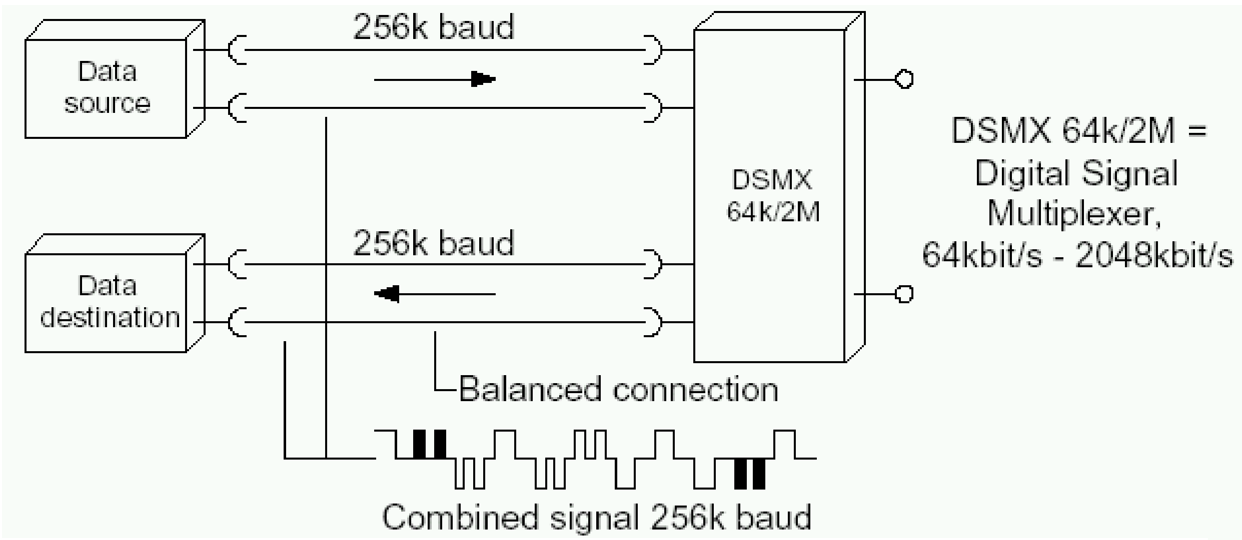


Fig. 20 Codirectional interface - details and signals transmitted on this interface

- **Contradirectional interfaces**
  - Both the data signal and the synchronization signal is transmitted between equipments; the synchronization signal is transmitted from multiplexer to terminal equipments; there are necessary to 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

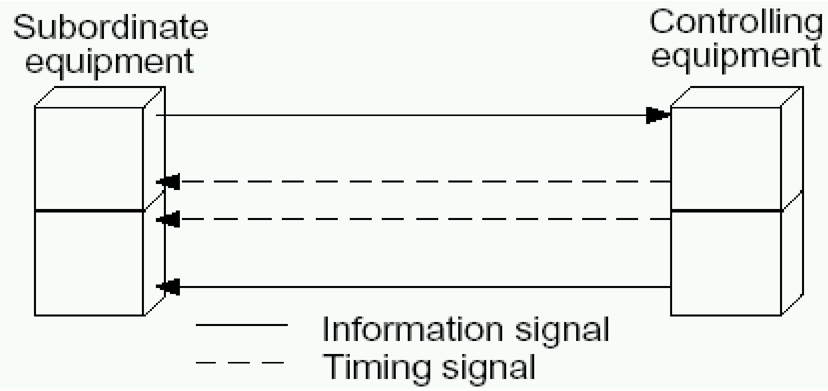


Fig. 21 Contradirectional interfaces. Basic schematic.

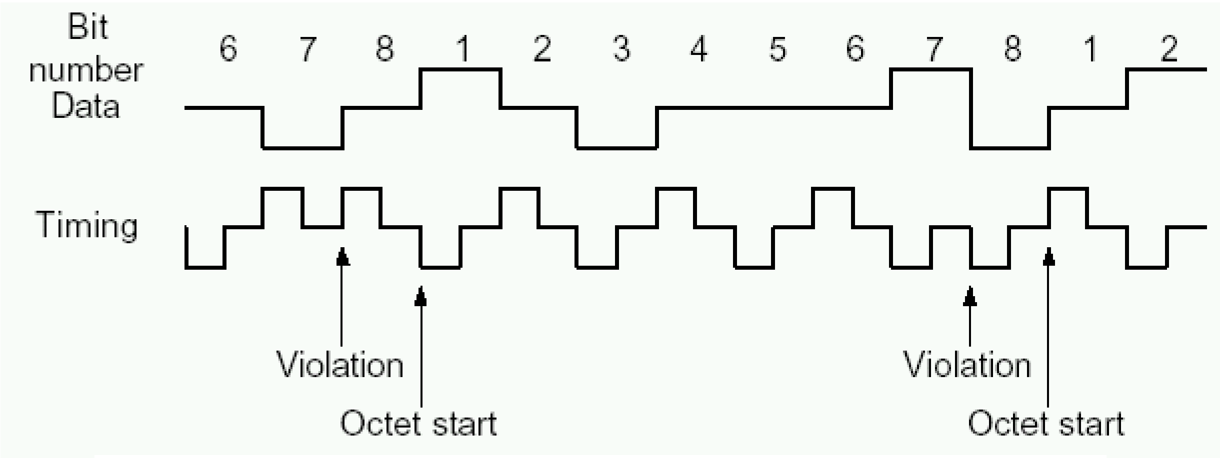


Fig. 22 Generation of the signals transmitted on the interface.

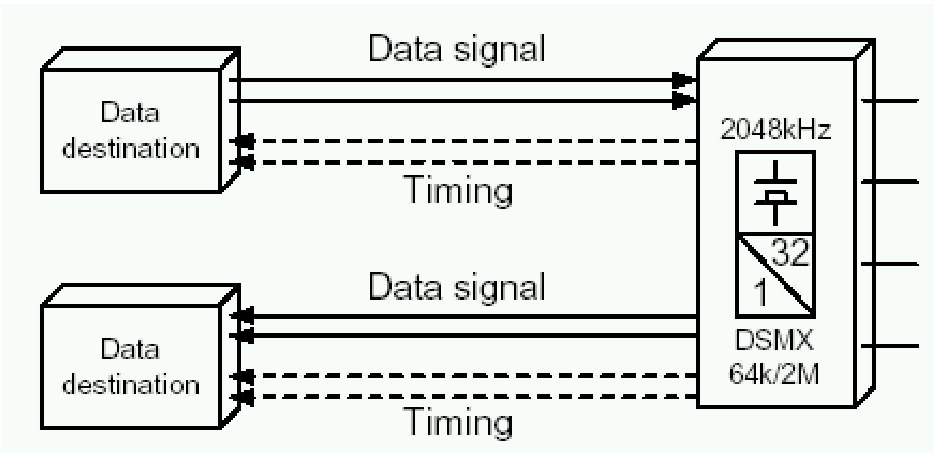


Fig. 23 Contradirectional interface - detail