General characterization of the mixed analog/digital circuits parameters

In fig. 1 is presented the general block schematic of a telephone connection in a digital telephone network – the schematic allows the identification of the parameters characteristic to different components of this network [Pa1]. The major components of this connection are:

- the analog channel which usually ensures the subscriber – local exchange connection.
  - the digital superchannel which ensures the connection between two exchanges – this digital superchannel includes the transmission of the information on different physical mediums (cable, optic fiber, radio), the switching and multiplexing operations.

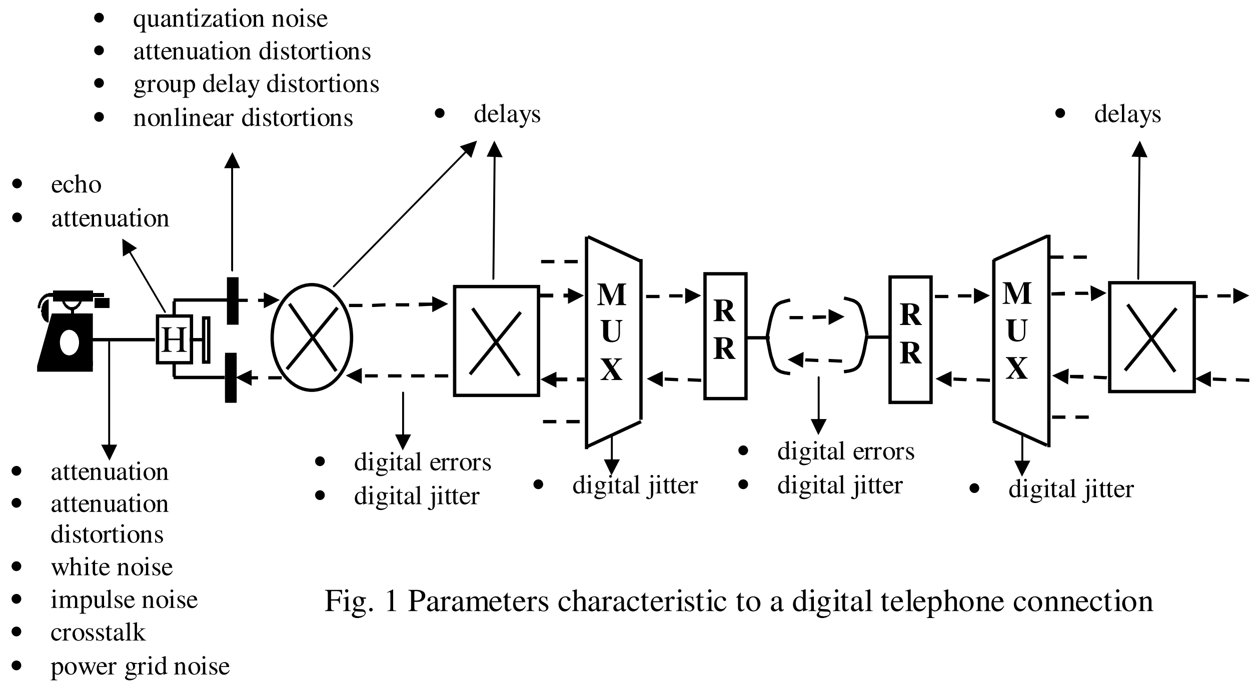

Fig. 1 Parameters characteristic to a digital telephone connection

- ♦ the subscriber – local exchange connection is characterized by attenuation, attenuation distortions and different noises (the crosstalk is considered a signal dependent noise);
- ♦ the hybrid transformer is responsible for the echo phenomenon and generates supplementary attenuations;
- ♦ the analog/digital and digital/analog conversion generate quantization noise, attenuation and group delay distortions due to the antialiasing and reconstruction filtering of the analog signal;
- ♦ the digital switching systems are responsible for the delays of the transmitted signal;
- ♦ the digital multiplexing/demultiplexing systems generate digital jitter;
- ♦ the transmission systems generate digital errors, delays and digital jitter;

The digital parameters specific to digital transmission systems are determined by the analog parameters of the physical circuits used to accomplish effectively the transmission – fig. 2 and 3 presents shortly the main parameters of digital and analog channels which can be found in a digital telephone network.

The parameters presented in fig. 2 characterize a digital super-channel which includes rhe processing necessary to transmission, multiplexing and switching.

- o Some parts of these parameters are determined by digital processing and other parts, especially the transmission processes, by analog processing - see fig. 3.
- o The parameters presented in fig. 3 characterize globally the analog channels, some of these parameters being characteristic only to channels which transmit modulated signals.

- The parameters of the analog channels determine partially the parameters of the digital channels like the bit error probability (BER), bloc error probability (BLER) and digital jitter.

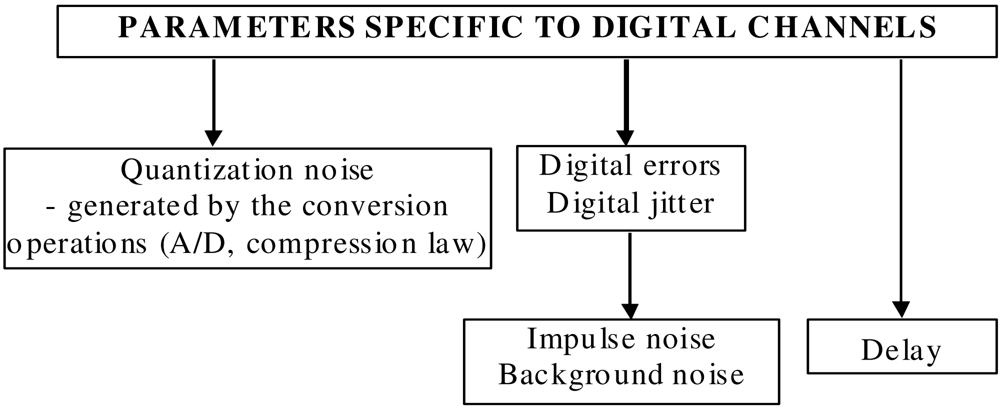

Fig. 2 Parameters specific to digital telephone channels

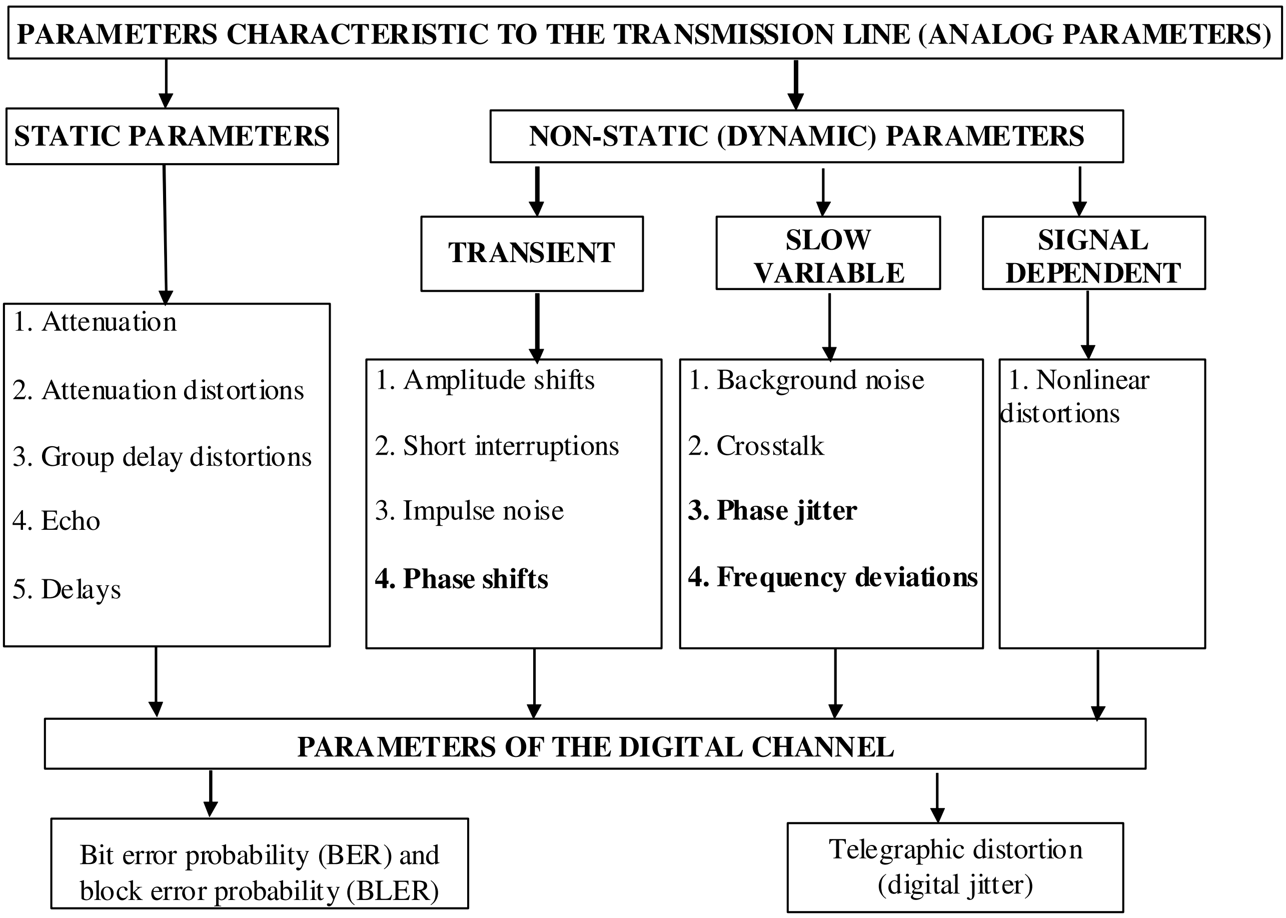

Fig. 3 The main parameters of a telephone channel which includes analog and digital circuits

1. Attenuation and group delay distortions

The global in-out characteristic of a channels according to the frequency is defined by the transfer function $H(\omega)$: $H(\omega) = A(\omega) \cdot e^{j\phi(\omega)}$ (1), where $A(\omega)$ is the amplitude (gain) characteristic, and $\phi(\omega)$ is the phase characteristic; these characteristics are independent of the amplitude applied at the input of the channel. The ideal channel characteristic id described by:

$A(\omega) = ct\,;\, \phi(\omega) = \tau_g \omega + \phi_0\,;\, \tau_g = \frac{d\phi(\omega)}{d\omega}$ (2), meaning constant gain with frequency and linear phase characteristic with frequency – constant group delay time with frequency.

The attenuation and phase distortions represents the modification according to frequency of the amplitude and phase of a sine signal transmitted in a linear system, without generating any harmonics of the transmitted signal. These distortions are characteristic to filtering operations. Will be considered only the attenuation and group delay distortions induced by the analog channel between the subscriber and the exchange, the distortions induced by the transmission systems connecting the switching points being included in the bit and block error probability and the digital jitter which affect the digital signal transmitted in these systems. The voice transmissions are sensitive only to the attenuation distortions, the group delay distortions (phase distortions) having practically no effect on these transmissions, but the data transmissions are sensitive both to attenuation and group delay distortions, the later one being even more important than the attenuation distortions.

The pairs of twisted wires from the subscriber loops represent one of the main sources of attenuation distortions. The characteristics of these lines are determined firstly by the distributed capacitive (C/km) and resistive (R/km) parameters – *see the cables laboratory*.

The group delay characteristic of the twisted wires has a slow decreasing variation with frequency, this type of distortion induced by twisted wires being negligible – for example a cable with length 10km and wires with diameter 0.6mm induces in the voice band a group delay distortion of 75μs [Pa1].

Another source of attenuation distortions at the inferior edge of the telephone band is represented by the power supply bridges of the exchange acting as high pass filters with 300-400Hz cut off frequencies – *see the analog access/telephone device laboratory*.

A major source of attenuation and group delay distortions in a telephone network is represented by the antialiasing and reconstruction filters. In figures 4 and 5 are presented the spectral masks of the attenuation and group delay characteristics of the mentioned filters [IT712] – it is considered that the antialiasing and reconstruction filters have identical characteristics. The characteristics presented in figures 4 and 5 specify the limits imposed to the transfer characteristics in discussion, not being possible to implement completely identical filtering characteristics.

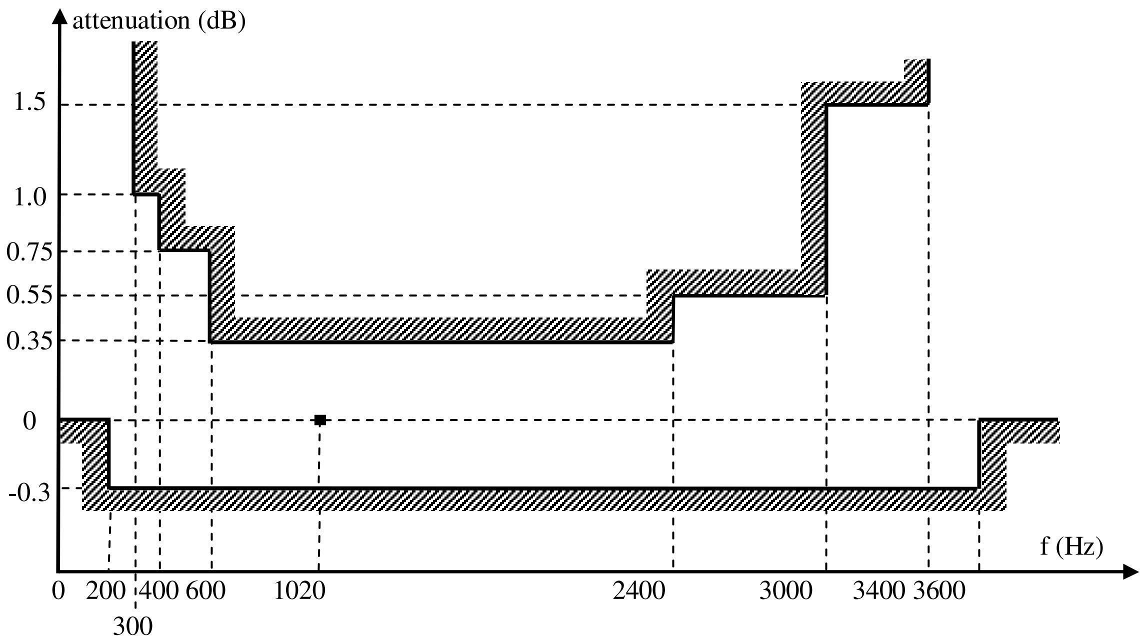

Fig. 4 Mask of the attenuation - frequency characteristic of the antialiasing and reconstruction filters used in PCM coders and decoders

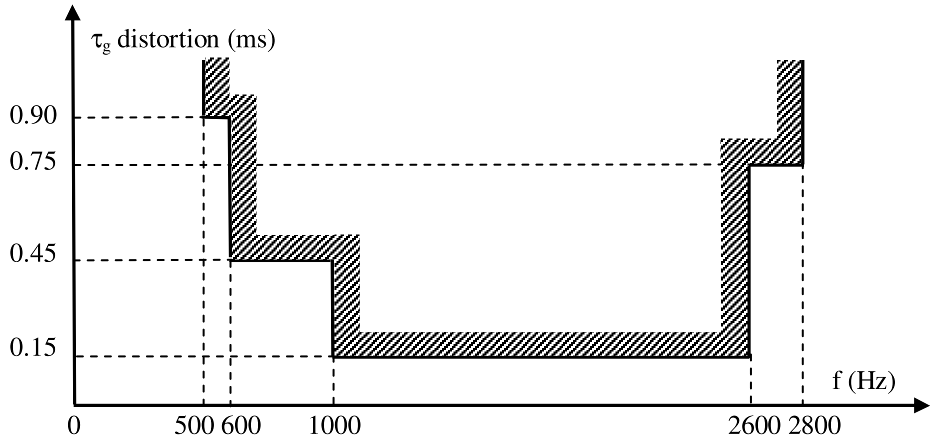

Fig. 5 Mask of the group delay - frequency characteristic of the antialiasing and reconstruction filters used in PCM coders and decoders.

The analysis of the attenuation characteristic of the antialising and reconstruction filters shows that in the 600Hz - 3000Hz frequency band the attenuation can vary between -0.3 and 0.55dB for a filter, and on a channel (includes two filters – one for antialiasing and one for reconstruction) between -0.6 and 1.1dB. These relatively small variations are imposed by the sensitivity of the voice to attenuation distortions and are appropriate for data transmissions in the telephone band.

In what concerns the group delay characteristics, 0.15ms variations of the group delay are allowed for central frequencies of the telephone channel (1000Hz – 2600Hz) and at the edges of the frequency band these variations could be as large as 0.45ms or 0.75ms. This group delay characteristic is appropriate for voice transmissions, but not for data transmissions. This can be demonstrated by a very simple calculation: a symbol rate of 1200Bd (Baud) means a symbol period of approximately 833μs, and a symbol rate of 2400Bd means a symbol period of approximately 416μs; the allowed variations of the spectral components delays in the central part of the telephone bands could be as large as 300μs – a maximum of 150μs inserted by the antialising filter and a maximum of 150μs inserted by the reconstruction filter, which means approximately 36% of the 1200Bd symbol rate period and 72% of the 2400Bd symbol rate period; it results a strong distortion of the transmitted symbols – the InterSymbol Interference phenomenon ISI. Especially in the case of 2400Bd symbol rate transmission, but possibly also at symbol rates larger than 1200Bd, will be necessary equalizer circuits which correct the group delay distortions of the telephone channel.

1.1 Echo

The echo represents another important source of attenuation and group delay distortions. The summing of sine signals having the same frequency but different phases due to the delay characteristic to the echo generates attenuation and group delay transfer characteristics of the channel with ripples. This has significant negative effects especially on the data transmissions.

The echo appears due to the impedance mismatches in the differential systems realizing the conversion from 2 wire transmissions to 4 wire transmissions and the opposite conversion from 4 wire transmissions to 2 wire transmissions [pană1] [zăhan] [feher1]. The echo is generated by the mismatches from the opposite (remote) end and two types of echo can be identified, namely:

- Transmitter echo
- Receiver echo

In figure 6 are presented the ways the two types of echo are generated.

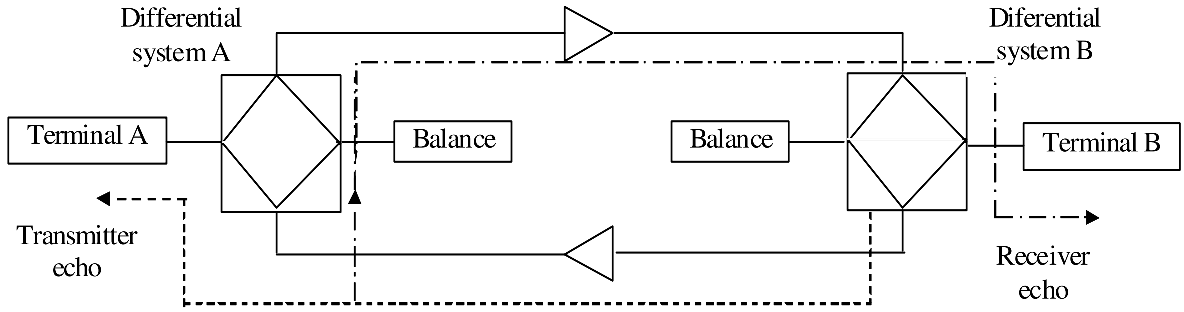

Fig. 6. The echo generation mechanisms

If on the channel are present several 2 wire – 4 wire and 4 wire – 2 wire conversions multiple echoes are generated both at the transmission and reception sides.

The effects of the echo on voice and data transmissions are completely different. In the case of voice transmissions the speaker is hearing a delayed replica of the transmitted or received voice signal.

In the case of data transmissions the receiver echo has a significant influence over the performances of the transmissions due to the fact that the transfer frequency characteristic of the channel presents ripples (both the attenuation and group delay transfer characteristics present ripples). This shape of the frequency transfer characteristic is due to the vector addition of the spectral components with their attenuated and different phase replicas. Even if the delay is constant with the frequency, the phase difference between the spectral components and their delayed replicas is changing with the frequency. If the attenuation and the delay inserted by the circuit on the receiver echo path are constant then the frequency transfer characteristic of the circuit affected by the receiver echo can be described by the filtering process implemented by the filter presented in figure 7, filter having the transfer function given by relation (3) [Fe1] [Zăh]. In figure 7 a_{ec} represents the gain of the echo path, meaning 1/(echo path loss), and τ_{ec} represents the delay of the echo path.

$$H_{filtru-ec}(\omega) = \frac{Y(\omega)}{X(\omega)} = \frac{1}{1 - a_{ec} \cdot e^{-j\omega\tau_{ec}}}$$

$$X(\omega) = \mathrm{Fourier}(x(t)) \quad ; \quad Y(\omega) = \mathrm{Fourier}(y(t)) \quad (3)$$

$$\left|H_{filtru-ec}(\omega)\right|^2 = \frac{1}{1 + a_{ec}^2 - 2 \cdot a_{ec} \cdot \cos(\omega \cdot \tau_{ec})}$$

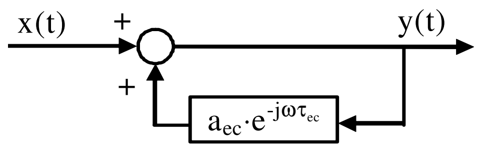

Fig. 7 The receiver echo path modeling

In figure 8 [Fe1] is presented as example the frequency transfer function of a circuit affected by echo, the delay of the echo being 1ms and the echo attenuation 5 respectively 2 (considered as ratio), meaning an echo path gain of 0.2 respectively 0.5.

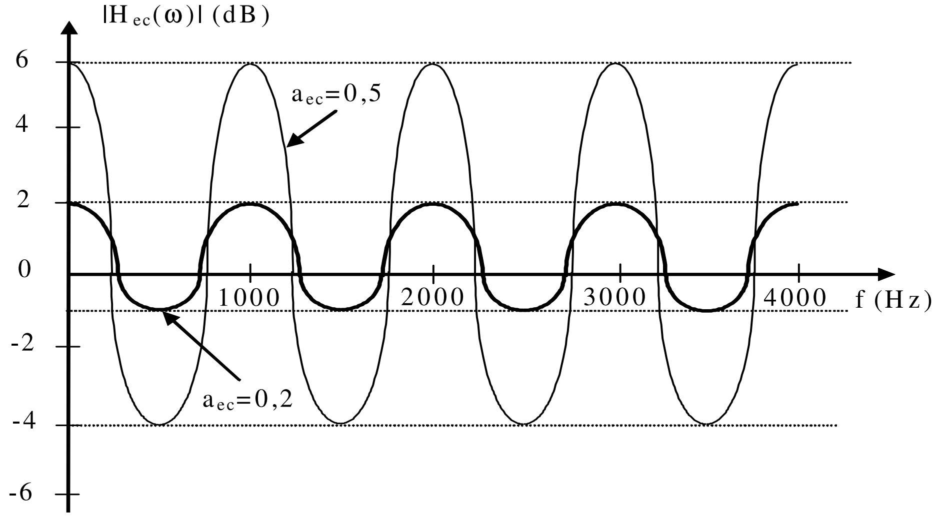

Fig. 8 The modulus of the frequency transfer function of a circuit afected by echo

2. Amplitude shifts and interruptions

The amplitude shift (amplitude hit) is a positive or negative sudden change of the amplitude with relatively moderate value, but higher than a given threshold (for ex. ±2dB) and having duration larger than 4ms (usually) – see fig. 9 [Pa1] [IT95. It has a multiplicative and transient character and can be modeled as an amplitude modulation process: $s(t) = A \cdot [1 + \Delta A(t)] \cdot \cos(\omega_p t)$ (4), where A is the signal amplitude with frequency ω_p and $\Delta A(t)$ is the amplitude variation. Amplitude shifts are random and seldom processes and are characteristic to analog transmissions.

Among the main sources of amplitude shifts can be mentioned: switching on standby equipments, connection in parallel with the circuit of equipments with impedance not high enough, intermittent faults of some components/equipments.

Amplitude shifts affect especially the data transmissions realized on telephone lines, being imposed a maximum number of 10 hits with a ±2dB minimum threshold in 15 min.

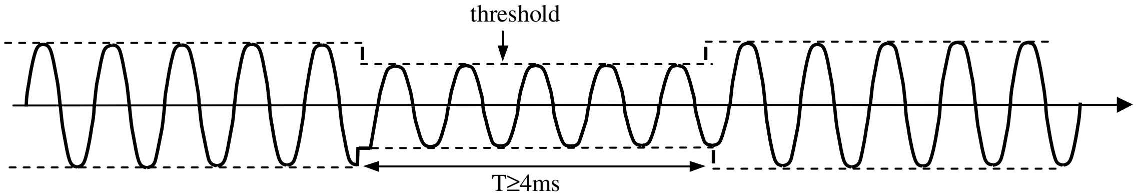

Fig. 9 Amplitude shifts

The short interruptions are deep decreases of the signal level, below an imposed threshold P and durations located between two values T_1 and T_2, imposed by the bandwidth of the channel. It is supposed that the signal level returns to the original value. The interruptions could be considered decreases of the signal level with minimum 10dB and having durations at least 3,5ms [Pa1] [IT61]. The interruptions represent a transient phenomenon and can be considered as an amplitude modulation process. Are generated especially by imperfect switching elements, maintenance operations a.s.o. and have a significant effect on data transmissions (it is imposed a maximum 1 interruption in every 30 min).

3. Delays

Delays are generated by the finite propagation time of the electromagnetic field on the channel. If the delays of the spectral components are uniform this parameter has a low importance for the physical transmission of the information, but has importance for the higher level protocols controlling the transmission.

4. Phase hits

The phase hits represent a sudden phase variation, forward or backward, larger than a given threshold and having duration of at least 4ms [pană1]. This last condition is required to make the separation between phase hits and noise impulses which affect also the phase of the transmitted signal. The origin of this phenomenon consists in commutation on standby generators, de-synchronization of clock generators in multiplexing systems. The maximum allowed limit is 10 hits in 15 minutes, at a 15° threshold.

5. Phase jitter

The phase jitter has different characteristics in digital and analog systems. In analog systems it represents a continuous parasitic phase modulation which modifies the zero crossing moments of a (test) sine signal. In digital systems it represents a modification of the significant moments (i.e. transitions) of the digital signal relatively to the ideal positions.

In analog systems the phase jitter is described by a complex relation consisting of a sum of components [pană1]:

$$\Phi_j(t) = \sum_{i=1}^{N} \Phi_{ji} \cdot \cos\left(2\pi f_{ji} t + \varphi_{ji}\right) \tag{5}$$

where Φ_{ji}, f_{ji} şi φ_{ji} represent the amplitude, frequency and phase of each jitter component.

The frequencies of the jitter components are located in the 4Hz – 300Hz interval, and the peak to peak voltages of these components are usually smaller than 20°.

Among the jitter generating sources in analog networks could be mentioned:

- The imperfect filtering of the power sources, which could affect the phase of the carrier signals. The frequency of such jitter components has the value 50Hz (the frequency of the power distribution grid) or the values of the harmonics of this frequency (100-300Hz).
- Slow variations of the load of the exchanges when the same sources power both the switching and multiplexing/transport systems. Usually it is about a low frequency jitter (2–10Hz).
- The ringback signal (20Hz).

In the case of digital systems the phase jitter appears as a modification of the transitions of the digital signal, modifications generated by the imperfect recovery of the clock at reception or in regenerators and by the intersymbol interference induced by the transmission channel [zăhan].

6. Frequency shifts

The frequency shits represent a positive or negative deviation of the transmitted signal frequency (of the spectral components of the transmitted signal) on the considered channel. This phenomenon appears in systems realizing frequency multiplexing due to the frequency differences and imperfect synchronization of the signal generators from the transmitter and receiver. The maximum allowed limit is ±5Hz. On national circuits this limit could be smaller, i.e. ±2Hz or ±1Hz.

7. Noise

The noise is one of the most important parameter of the transmission systems and represents any disturbance which affects the transmitted signal, being added with this signal. The effects of the noise on the quality of the transmissions differ radically according to the type of the service (telephone, data transmission, a.s.o.). In telephone transmissions the noise generates a "masking" effect of the audibility threshold, this threshold increasing with the noise level. In data transmissions the noise generates digital errors at the reception side. A classification of the noises of the telephone systems is presented in fig. 10 [Pa1].

A few useful comments related to the aspects presented in figure 10 are the following: the unwanted sine signals – signals used for ex. in the signaling process – are induced in the telephone channel by linear crosstalk, the only difference from other crosstalk signals being the fixed frequencies of the mentioned sine signals. This noise has also stationary character, a typical example is the fundamental of the power supply voltage – *see the cables laboratory*.

The A/D converter noise signal it is not the quantization noise but a noise due to the offset voltages of the converter, voltages which affect the conversion of low amplitude signals. The errors from the digital streams usually generate impulse noise, but could produce also a background noise.

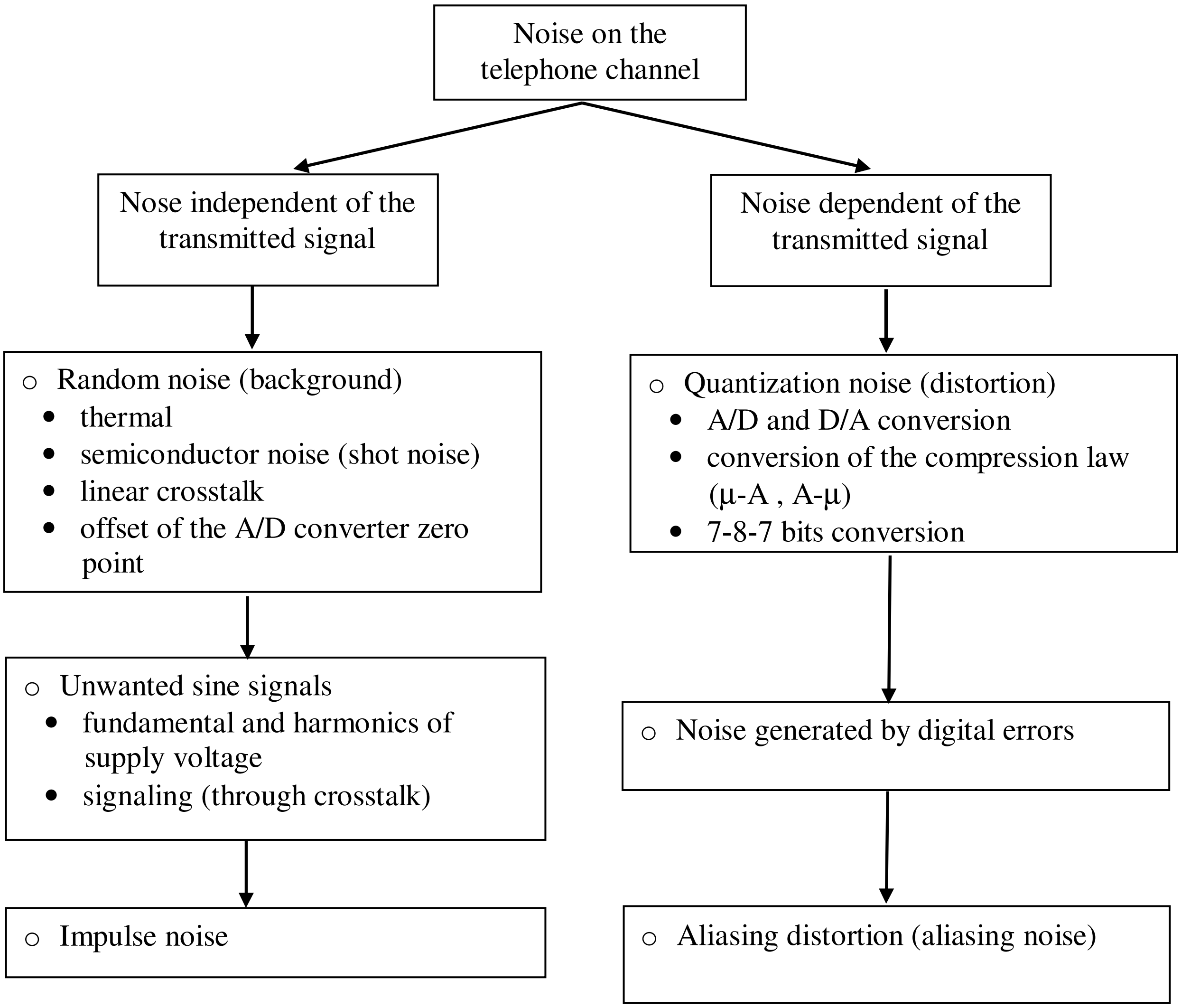

Fig. 10 Classification of noises characteristic to telephone channels

The random noise (or background noise) is the most important component of the signal independent noise. It has a stationary character and it is perceived like a continuous hiss. The most important component of this noise is the thermal noise or resistance noise, which is a noise with uniform power spectral distribution and is called also white noise. The amplitude values of this noise are distributed according to a Gaussian distribution, fact which gives the name of Gaussian noise. The value of the noise voltage on a resistance with value R in frequency band B, at a 20°C temperature it is given by the relation: $U_{ef}(\mu V) = 0.126 \cdot \sqrt{RB}$ (6). The effective value of the noise voltage and the amplitude distribution are given by the following relations:

$$U_{ef} = \sqrt{\frac{1}{T}\int_0^T u^2(t)dt} \qquad (7)$$

$$w(x) = \frac{1}{\sqrt{2\pi}} \cdot e^{-\frac{1}{2}x^2} \; ; x = \frac{U}{U_{ef}} \qquad (8)$$

Another important component of the background noise is the shot noise generated by active devices. It is also a temperature dependent noise.

The linear crosstalk noise is the phenomenon related to the appearance of a transmitted signal in a wrong (forbidden) place. If this transferred signal is an intelligible speech signal means that the confidentiality of the communication is lost.

The most important component of the signal dependent noise is the quantization noise induced by the PCM coders due to the A/D and D/A conversions and by the code converters realizing the conversion of the PCM code words obtained with the A and μ compression laws and of the PCM code words represented on 8 bits or 7 bits;

The quantization distortions generated by different conversion operations can be expressed as a function of the quantization distortion induced by an ideal PCM coder, distortion called qdu - *Quantization Distortion Unit.*

It is useful in many situations to equate the quantization noise with a white noise. In table 1 m represents the number of qdu units, S represents the level of voice signal (-17dBm), Z_g is the level of the equivalent white noise expressed in dBmp – (p – psophometric level – *see the levels and attenuations laboratory*), S/Z_g – signal to white noise ratio expressed in dB.

m (qdu)	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15
S/Z_g (dB)	59,8	49,6	44	39,7	36,6	33,8	31,5	29,5	27,8	26,4	25	23,9	22,7	21,6	20,5

Tab. 1 The equivalence of the quantization noise expressed in qdu units with white noise

The impulse noise is characterized by voltage shifts which exceed the usual peaks of the background noise existing on a transmission channel (the impulse noise exceeds usually the effective value of the background noise with at least 12dB [Pa1] [IT71]). The impulse noise is not very disruptive in telephony, due to the low sensitivity of the ear at this type of distortion – can be perceived as isolated shots when the level of impulse noise exceeds some threshold, but it has importance in data transmission realized on telephone circuits. The impulse noise is generated both by transmitted signal independent and dependent sources. Among the signal independent noise generating sources can be mentioned: dialing pulses, metering pulses, ringing signal, electric discharges, sparks, fluorescent lamps, while among the signal dependent noise generating sources can be mentioned: digital errors in the PCM digital signal or in the multiplexed signal transmitted at higher hierarchy levels.

The most important properties of the impulse noise are: time variability, normal distribution of the impulse levels, concentration in packets of the impulses – non-uniform distribution in time, variability according to the period of the day and of the week, limitation of the minimum and maximum duration in the telephone channel due to the limitations of the frequency band – the maximum frequency of 3400Hz limits the duration of the impulses to approximately 0,3ms while the minimum frequency of 300Hz limits the mentioned duration at approximately 3ms; the usual durations are situated between 0,5ms and 1ms.

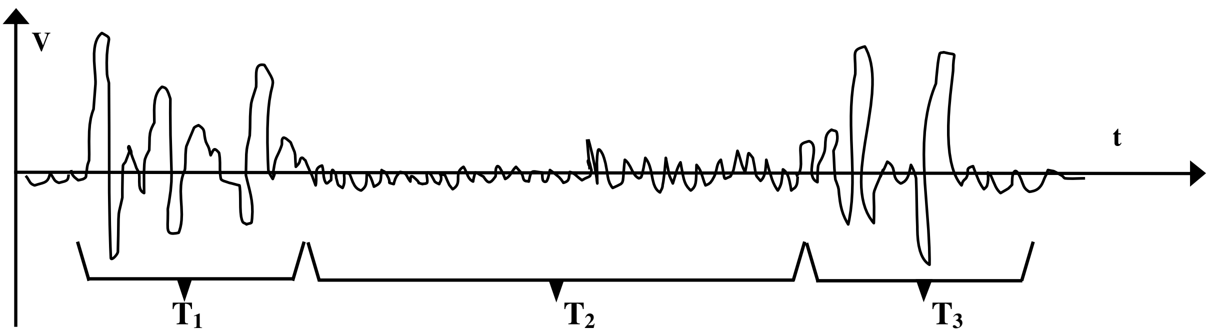

Fig. 11 Packets of noise impulses

The mathematical characterization of the impulse noise is very complex and involves four distribution functions, namely: distribution of the impulse amplitudes, distribution of the impulse durations, distribution of the time between two consecutive impulses and the spectral distribution of the impulses.

In the case of the noise generated by digital errors must be considered the time response to an isolated pulse of the reconstruction filter from the PCM decoder and the error distribution – errors in packets (having usually a Neyman distribution) generate a higher number of noise impulses then isolated errors (having usually a binomial distribution), especially if the average number of errors located in one error packet is between 5 – 10.

The isolated errors are generated usually by the higher hierarchy signals of digital multiplexing systems, the error packets of these streams being reflected in the primary PCM signals, after demultiplexing, like isolated errors. In the case of the most disadvantageous situation (error distribution in packets of 5-10 errors) the objective of 18 impulses / 15 minutes imposed by the ITU-T standards (M.1020 and M1025 standards [IT1020] [IT1025]) is fulfilled if the bit error rate (BER)<10^{-6} in the 2Mbps PCM digital stream.

Error distribution	BER	10^{-7}	10^{-6}	10^{-5}	10^{-4}	10^{-3}
Neyman distribution No. of errors in a packet	5 ÷ 10	-	6	60	480	1880
	200	-	1	8	40	520
Binomial distribution		-	-	-	8	172

Tab. 2 No. of noise impulses detected in 15 minutes at an impulse threshold of -21dBm for various bit error rates and error distributions.

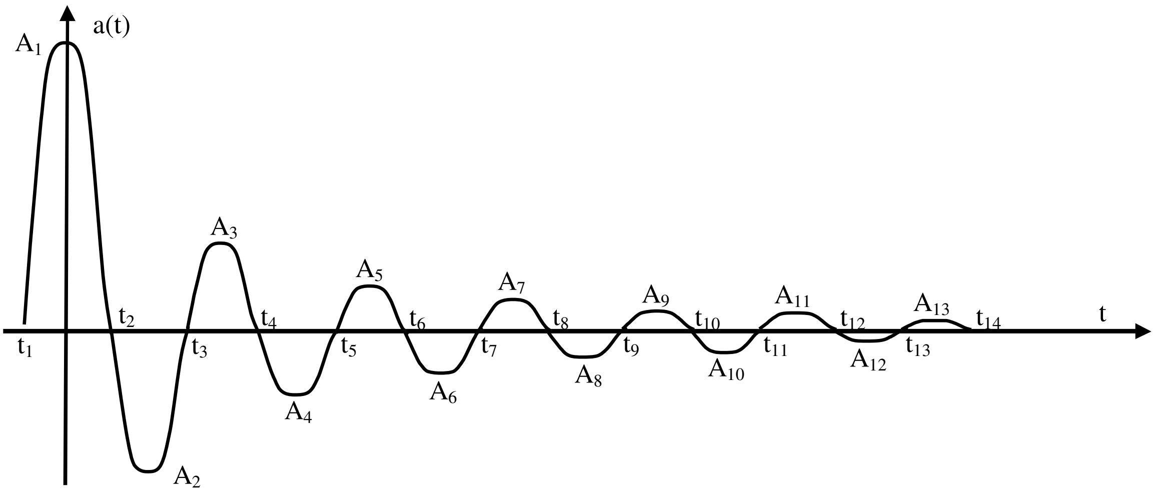

Fig. 12 Response to an isolated pulse of the PCM decoder reconstruction filter [Pa1]

The amplitudes A_1, A_2, ... and the durations of the impulses which compose the reconstruction filter response to a noise impulse generated by digital errors in the PCM signal (figure 12) depend on the order of the erroneous bits of the PCM codeword and on the number o consecutive PCM frames (frames of 125μs) with packet errors in the considered time slot.

Order of the erroneous bits	Amplitude A_1 (dBm)
2	-22,1
3	-34,1
2 and 3	-10,1
2, 3 and 4	-4,1

Tab. 2 Amplitude of the reconstruction filters response to a noise impulse generated by digital errors affecting different bits of the PCM word.

The sampling noise (distortion) or aliasing noise represents a component of the signal dependent noise which is generated by the aliasing phenomenon (meaning improper filtering).

The nonlinearity distortions induced by the transmission channel represent another component (an important component) of the signal dependent noise. It is characterized by appearance of new spectral components generated from the useful signal and added with this signal. The main source of this distortion is the nonlinearity of the input-output transfer characteristic of different elements of the telephone channels (see fig. 13) – in the case of a digital telephone network it is about mainly of the non-uniformities of the A/D and D/A conversion characteristics, meaning the gain variations in the telephone band – in fig. 14 it is presented the mask of the allowed gain variation of the PCM coders and decoders [IT712]. The human hearing has a high sensitivity to these distortions, which must be kept in tight range (-0.3dB ÷ 0.3dB) for the entire dynamic range of the channels input signal (-40dBm ÷ 3dBm).

The characterization of the harmonic (nonlinear) distortion can be made with the help of the following parameters:

- The harmonic distortions expressed in percents: these percents are expressed for each harmonic apart: $d_k = \frac{U_k}{U_1} \cdot 100\%$ (9) , where U_k is the k-th order harmonic, and U_1 is the fundamental.
- The total harmonic distortion:

$$d_t = \frac{\sqrt{U_2^2 + U_3^2 + \ldots}}{U_1} \cdot 100\% \quad \text{sau} \quad \frac{\sqrt{U_2^2 + U_3^2 + \ldots}}{\sqrt{U_1^2 + U_2^2 + U_3^2 + \ldots}} \cdot 100\% \tag{10}$$

- The harmonic distortion attenuation, one of the most used parameter in telecommunication:
  - ♦ The k-th order harmonic distortion attenuation: $$a_k = 20 \cdot \lg \frac{U_1}{U_k} [\mathrm{dB}] \tag{11}$$
  - ♦ The total harmonic distortion attenuation: $$a_t = 20 \cdot \lg \frac{U_1}{\sqrt{U_2^2 + U_3^2 + \ldots}} [\mathrm{dB}] \tag{12}$$

Note: the test signal used for the evaluation of the nonlinear distortion parameters is a sine signal having a single spectral component.

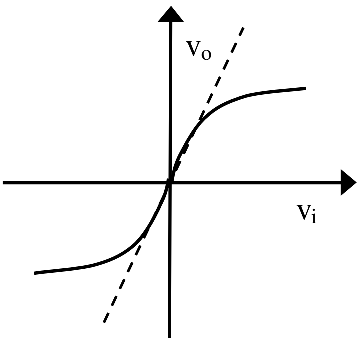

Fig. 13 Nonlinear voltage transfer characteristic.

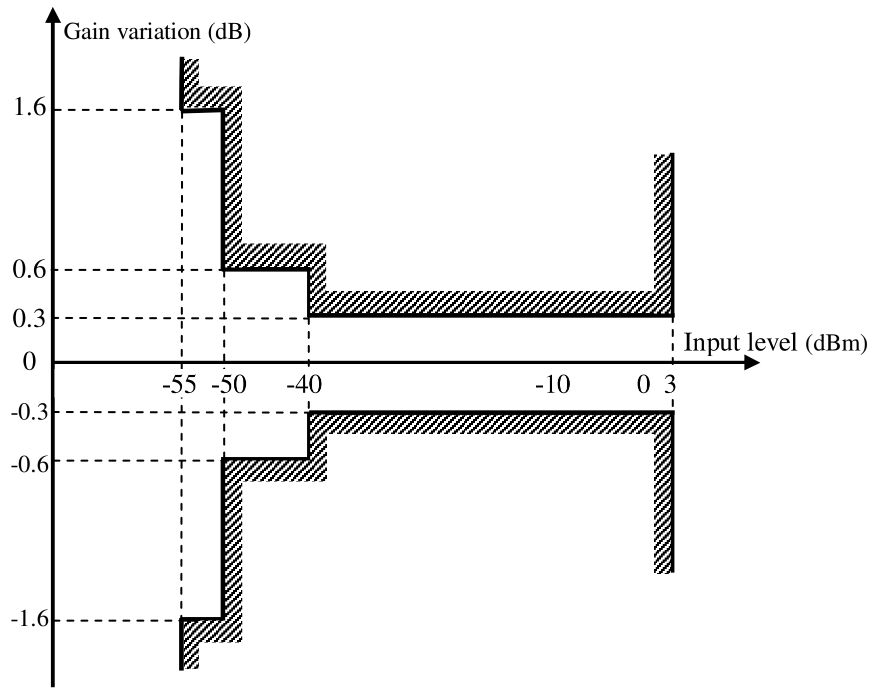

Fig. 14 Mask of the PCM coders/decoders allowed gain variation.