TELEPHONY
Theory and applications

Zsuzsanna Ilona Kiss Zsolt Alfréd Polgár

May 19, 2020

"*Before anything else, preparation is the key to success.*"

- Alexander Graham Bell -

Foreword

This book deals with some important issues concerning digital telephone and data networks. The approach is more from a practical perspective, without neglecting the theoretical analysis of the considered aspects. Specifically, the book proposes a combination of theory, exercises and practical applications that are especially useful for students undergoing telephony or data communications courses, but the book can be useful to all those interested in the field of telecommunication. The practical applications, discussions and exercises presented are designed in a way that will help to understand and deepen the theoretical aspects.

The issues covered are grouped into 8 chapters, each chapter consisting of a brief introduction, which briefly explains the aims pursued, a theoretical part, the presentation of some equipment and/or software, by which the theoretical aspects considered can be tested/experimented, and the description of a practical application followed by a selection of questions and exercises that can help to better understand the theoretical and practical aspects dealt with. A brief description of these chapters is as follows:

Chapter 1 deals with a number of issues related to subscriber access to a telephone network, being considered issues related to terminal remote power feeding, duplex transmission, and signaling on the subscriber loop.

Chapter 2 presents a number of issues regarding the telephone subscriber terminals. Schematic block diagrams, simplified electrical diagrams and configuration methods for more complex electronic telephone terminals are presented and discussed.

Chapter 3 is dedicated to the mathematical modeling of telecommunication cables, of crosstalk affecting the transmissions on these type of channels, and the presentation of more relevant practical aspects related to the internal structure of telecommunication cables.

Chapter 4 deals with practical aspects as well as some theoretical aspects related to the wiring of telephone and data networks. Connection and protection equipment are briefly described, as well as the rules underlying the installation of cables in the aforesaid connection equipment.

Chapter 5 shows how to define important concepts for telephone and data networks, such as signal levels, attenuation/signal loss inserted by circuits, transmission chain reference points, noise level assessment, etc.

Chapter 6 is dedicated to voice coding techniques used in commercial and in special telephone networks. More specifically, Pulse Coded Modulation and Delta Voice Coding techniques are considered, the focus being on the digital implementation methods and evaluation of the quality indicators of the transmissions that uses the considered voice coding techniques.

Chapter 7 deals with a number of theoretical and practical aspects of data transmissions in telephone band, namely FAX transmissions and dial-up data transmissions, being also considered some aspects of telephone answering machines. The chapter also presents some concrete examples of equipment and software that can be used to achieve these transmissions and step-by-step configuration of these equipment. Even though the configuration issues are specific to some specific equipment, the presentation is trying to identify the "essential" issues related to the configuration of FAX and modem equipment, resulting in some degree of generality, which is also useful for configuring other equipment.

Chapter 8 is dedicated to digital broadband access techniques on the subscriber loop, specifically ADSL access techniques. The basic theoretical aspects regarding ADSL and ADSL2+ transmissions are addressed and some ADSL equipment from the user side and the local exchange are presented. The chapter briefly presents the configuration steps of these equipment, but at the same time touches the essential aspects of these operations.

Just as in the chapter dedicated to data transmissions in the telephone band, the result is also a certain degree of generality of configuration operations, which is very useful for configuring other ADSL devices.

The authors would like to thank colleagues and all those who contributed directly or indirectly to the appearance of this book.

Chapter 1

Analogue access in the telephone network

1.1 Introduction

The chapter aims at presenting the essential aspects related to the access in the telephone network, considering various elements related to this issue, such as: remote power feeding of the subscriber terminals, duplexing techniques, block diagram of a telephone device as well as a local telephone exchange (the access point to the telephone service), signaling processes taking place between the telephone device and the local telephone exchange. Specifically this chapter proposes the study of analogue access techniques, still widely used in telephone networks. This choice is also motivated by "learning" aspects, making it easier for the students to understand a signaling process based on analog signals that can be viewed on an oscilloscope than a more abstract process based on digital symbol sequences (bit sequences). The issues related to remote power feeding of subscriber terminals, the implementation of duplex transmissions, the logic diagram of the signaling process between the telephone device and the telephone exchange are also common to the digital access techniques.

1.2 Transmission on the subscriber loop

The use of an analog transmission on the subscriber loop (the channel between the subscriber and the local exchange on twisted wires) is the simplest way to access the telephone network. This mode of access is characteristic to Plain Old Telephone Service (POTS) analogue telephone networks, but is also widespread in digital telephone networks due to its simplicity. The main characteristics of the analogue access are the following: [1] [2] [3]:

- frequency band: $300Hz - 3400Hz$;
- 2 wire access - 2-wire full duplex transmission;
- remote power feeding from the local exchange at a voltage of $V_{supl} = -48Vcc$;
- DC current through the subscriber loop $I_{loop} = 20 - 50mA$.

The phone's power supply voltage drops to $10 - 20V$ (or to a lower value if the length of the subscriber loop is great) when the phone becomes active and the subscriber loop is closed, the DC resistance of a telephone device being hundreds of Ω (a disc phone has a DC resistance of $\approx 850\Omega$).

Note: the power supply voltage on the subscriber loop may be less than $48V$ for PBXs (Private Branch Exchanges) due to significantly shorter subscriber loops (lengths of subscriber loops of tens of meters, possibly a few hundred meters in some cases).

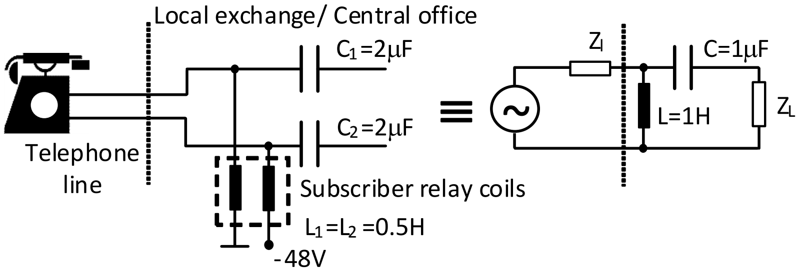

Figure 1.1: The power feeding circuit of the telephone device and its equivalent diagram.

Note: the analog phone can also be powered from a local power source (battery), but this is not common for public telephone networks. Local power (with a power adapter) is used for more complex phone devices, such as digital phones, faxes, auto-answer machines or cordless phones, or for data transmission equipment, such as dial-up modems, which require a greater power than that provided on the subscriber loop.

Note: the bandwidth of the channel implemented on twisted wires is much larger than the telephone band, limiting the voice transmission to this band is usually done in the local exchange or even in the telephone device. For this reason, broadband digital transmissions (like xDSL transmissions), that require a significantly wider bandwidth than the bandwidth of the telephone channel, can be performed on the subscriber loop.

In order to ensure the supply voltage on the subscriber loop, there is a need for a power feeding circuit (located in the telephone exchange) [1] [2] [3], the simplified diagram of this circuit being presented in Fig. 1.1. In addition to supplying the voltage to the telephone device, this circuit must provide the following signal separations: the DC supply voltage must be separated from the telephone exchange circuits, which is ensured by the capacitors C_1 și C_2 having the capacity $C = 2\mu F$; the separation of the transmitted signal on the subscriber loop from the power supply in the telephone exchange must also be achieved as the DC voltage source short-circuits the signal at the input of the telephone exchange. In addition, the separation from the signal ground of the exchange must be achieved as well in order to ensure on the subscriber loop a differential transmission, necessary for rejection of common mode noise. Separation from the voltage source and the signal ground is achieved by means of coils L_1 and L_2 with inductance $L = 0.5H$. These coils are also part of the subscriber relay, which has the role of sensing the closing or the interruption of the subscriber loop. If we see the telephone device as a signal source and the exchange circuits as a load, the power feeding circuit can be equated with a high pass filter (HPF) with cutting frequency $f_t = 300 - 400Hz$ (see Fig. 1.1). This explains the lower limitation of the telephone band.

The upper limit of the telephone frequency band (much lower than the upper limit of the voice signal band $\approx 20kHz$) was chosen based on the characteristics of the frequency spectrum of the voice signal and the bandwidth limitations. The bandwidth of the telephone channel is particularly important in carrier systems (systems which connect switching nodes) where high spectral efficiency is required. A value of the upper limit of the telephone frequency band $\approx 4kHz$ provides a reduced bandwidth in terms of an acceptable quality of the telephone connection (to ensure the intelligibility of the conversation and the speaker identification).

The two-wire duplex transmission on the subscriber loop also raises several issues, namely [1] [2] [3]:

- the telephone device is working inside on four wires, that is, there is a separate microphone circuit and a separate speakerphone/headphone circuit (see Fig. 1.2), which means that there is a need for a circuit that achieves the transition from 4-wire full-duplex transmission to 2-wire full-duplex transmission. The 2-wire transmission on the subscriber loop is explained by economic reasons, the total length of the subscriber loops consisting of copper wires from a telephone network being very large.

- digital switching and digital coded voice transmission between the telephone exchanges is done on 4 wires, which means that the transition from the 2-wire full-duplex transmission to the 4-wire full-duplex transmission must be done in the local telephone exchange to which the subscriber is connected - see Fig. 1.3 (E_l - local exchange; E_t - transit exchange) [4] [5].

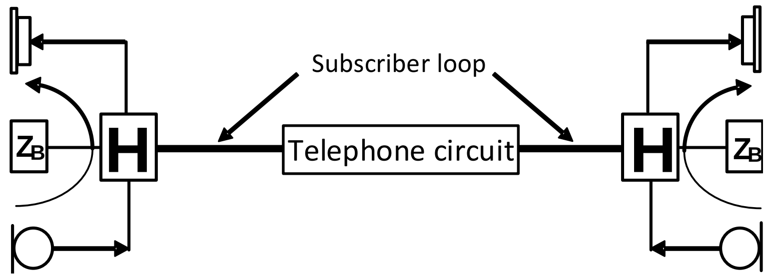

Figure 1.2: Connecting the voice circuits of a telephone device to the line.

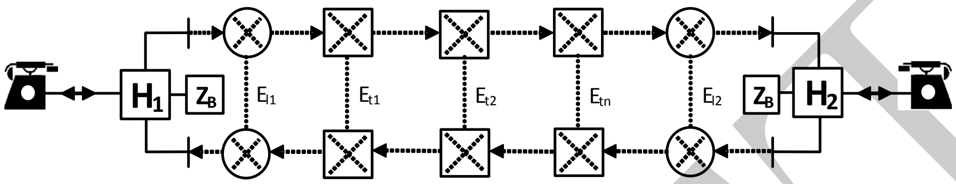

Figure 1.3: Telephone connection established in a digital network.

The devices that perform the transition from 2-wire transmissions to 4-wire transmission are the so-called hybrid transformers, which have to provide a high (theoretically infinite) attenuation between the transmission paths of the 4-wire circuits.

The transition from 2-wire transmission to 4-wire transmission can not be achieved ideally, i.e. the separation of the paths of the 4-wire circuits (i.e. reception and transmission paths) is not ideal, which causes the phenomenon of echo, a very annoying phenomenon for both voice and data transmissions in the telephone network.

Note: "more special" analog phones connected to PBX exchanges can work full-duplex on 4 wires on the subscriber loop, the reduced length of the subscriber loop in this particular case permitting this mode of operation.

Note: digital phones (e.g. ISDN phones) usually work full-duplex on 4 wires. These phones connect to a PBX or a network terminator located in the proximity of the phone. But there are also digital phones that work full-duplex on 2 wires.

1.3 The hybrid transformer

It is one of the most important components of a telephone network, being largely responsible for echo and circuit stability. The role of this circuit is to ensure the interfacing between two different transmission systems, namely a 2-wire full-duplex system and a 4-wire full-duplex system [1] [2] [4] [5]. The hybrid transformer, also called differential or duplex system, is a circuit characteristic not only to telephone systems but also to other systems or equipment where separation of transmission and reception paths is required. For example, a radio transmitting equipment that uses the same antenna for both transmission and reception will require such a hybrid transformer to separate the output of the transmitter from the receiver amplifier input. Considered as a quadriport, the hybrid transformer has a bidirectional port (1 - 1 '), which provides the connection to the 2-wire line, a balancing port (2 - 2 '), where the balancing impedance is connected, Z_B, and two unidirectional ports that connect to 4-wire circuits, (3-3') - reception port and (4-4') - transmission port. The circuit can be characterized as any quadriport, but in the case of a transmission system the most important parameters will be [2] [4]:

- attenuation between ports (3 - 3 ') - (4 - 4'), i.e. the attenuation between the reception and transmission paths in the 4-wire loop. This attenuation must be as high as possible (ideally infinite) and in real circuits it has values of $20 - 30dB$. This attenuation represents the return loss of the hybrid transformer that

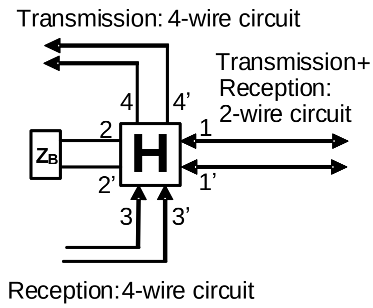

Figure 1.4: Hybrid transformer seen as a cuadriport.

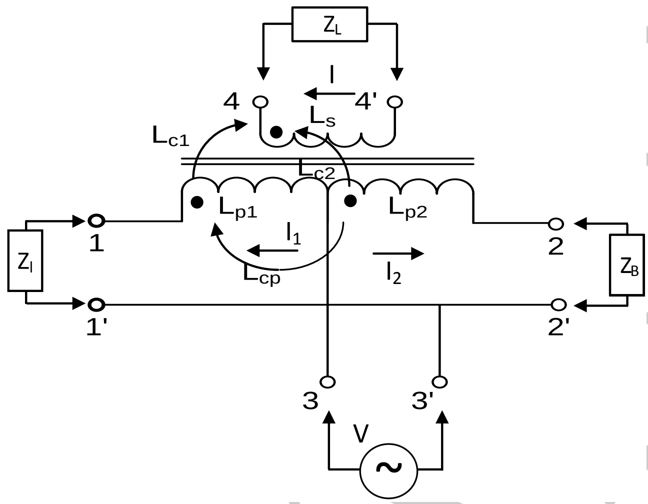

Figure 1.5: Hybrid transformer used in a rotary dial phone.

depends on the transformer's parameters and its balance. By reciprocity, the same attenuation must be ensured between the ports (1 - 1 ') - (2 - 2'), but this attenuation is less important.

- attenuations between ports (3 - 3) - (1 - 1 ') and (1 - 1) - (4 - 4'), i.e. the attenuation between the reception path of the 4-wire circuit and the 2-wire circuit and the attenuation between the 2-wire circuit and the transmission path of the 4-wire circuit. These attenuations have to be as small as possible, being usually 3dB in the case of a hybrid transformer with a symmetrical structure due to the equal division of power received at a port of the hybrid transformer at the adjacent ports. By reciprocity the attenuation between ports (2 - 2 ') - (4 - 4') is equal to the attenuation between the ports (1 - 1') - (4 - 4') and the attenuation between ports (3 - 3') - (2 - 2') is equal to the attenuation between ports (3 - 3') - (1 - 1').

- input/output impedances at ports (1 - 1 '), (3 - 3'), (4 - 4 '), important in terms of impedance matching between the hybrid transformer and the 2-wire and 4-wire circuits.

In the case of a digital telephone network with analogue access, the hybrid transformer can be found in the telephone devices and in the subscriber interfaces of the local telephone exchanges. In the case of telephone devices, the hybrid transformer has the function of separating the microphone circuit from the speaker circuit, eliminating (or reducing) the side effect (the coupling between the microphone and the speaker circuit). In Fig. 1.5 the schematic diagram of a hybrid transformer used in a rotary dial telephone is presented, and Fig. 1.6 shows the equivalence between the ports of this hybrid transformer and those of a general hybrid and how it connects the microphone and diffuser circuits to this general hybrid transformer [2] [4] [6].

The hybrid transformer shown in Fig. 1.5 is built using a transformer with center tap. This transformer is characterized by the following parameters: inductances of primary windings L_{p1} and L_{p2}, inductances identical and equal to L_p, coupling inductance between primary windings L_{cp}, inductance of secondary winding L_s,

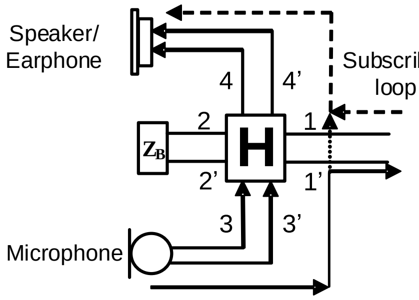

Figure 1.6: The separation of the microphone and the speaker circuit using a hybrid transformer.

coupling inductances between primary windings and secondary winding, L_{c1} and L_{c2}, identical inductances equal to L_c.

It is of interest establish the balancing condition and calculate the return loss of this hybrid transformer, which is why it is considered a voltage source V connected in place of the microphone, the load impedance Z_L connected in place of the speaker, impedance Z_l connected in place of the subscriber loop and the Z_B balance impedance connected to the corresponding terminals. The primary windings are traversed by currents I_1 and I_2, and the secondary winding by current I, as shown in Fig. 1.5. Considering for the simplification of relations a zero resistance of all windings, the following equations are obtained:

$$V = (Z_B + j\omega L_p) \cdot I_2 - j\omega L_{cp} \cdot I_1 - j\omega L_c \cdot I = (Z_l + j\omega L_p) \cdot I_1 - j\omega L_{cp} \cdot I_2 + j\omega L_c \cdot I \tag{1.1}$$

$$(j\omega L_s + Z_s) \cdot I + j\omega L_c \cdot (I_1 - I_2) = 0 \Rightarrow I = (I_1 - I_2) \cdot \frac{j\omega L_c}{j\omega L_s + Z_s} = (I_1 - I_2) \cdot K(\omega) \tag{1.2}$$

Based on previous equations, the relationship between the currents I_1 and I_2 can be found:

$$\frac{I_2}{I_1} = \frac{Z_l + j\omega L_p + j\omega L_{cp} - j2\omega L_c \cdot K(\omega)}{Z_B + j\omega L_p + j\omega L_{cp} - j2\omega L_c \cdot K(\omega)} \tag{1.3}$$

The hybrid transformer is in balance if the current I is zero, that is, we do not have signal in the secondary winding, condition fulfilled if $I_1 = I_2$. From the above relation results that the equality of currents I_1 and I_2 draws the equality of $Z_l = Z_B$, which is the balance condition of the hybrid transformer in question.

The hybrid transformer return loss is given by the relation $a_{rloss} = \frac{V}{V_L}$, where V_L is the voltage on the impedance Z_L in the secondary winding. Based on previous relations and under simplifying conditions $L_{cp} = L_p = L_c = L_s = L$ the following equation is obtained:

$$a_{rloss} = \frac{j\omega L \cdot (Z_B + Z_l) + Z_B \cdot Z_l \cdot (j\omega L + Z_s)}{j\omega L \cdot Z_s \cdot (Z_B - Z_l)} \tag{1.4}$$

If we have the balancing condition $Z_l = Z_B$ (in the entire frequency band of the signal) then $a_{rloss} \rightarrow \infty$ and we have an ideal separation of the microphone and speaker circuit, that is, we have a total suppression of the side effect.

The hybrid transformer presented is a solution for separating transmission and reception circuits in a telephone device where is no need for a very strong separation of the mentioned circuits (i.e., no very precise hybrid balancing is required), where common paths between different ports are admitted and where is no need for galvanic separation between the telephone line and the internal circuits of the equipment.

Taking into account the relatively light conditions imposed on the hybrid transformer in the phone device, regarding the separation of the different circuits connected to it, it is possible to design a simpler hybrid that does not include coils or transformers. Such a hybrid is usable in electronic telephones, where the use of coils and transformers (coils and transformers that work at low frequencies have a large physical size) is not a good solution because of the reduced physical size of these phones. Such a hybrid diagram is shown in Fig. 1.7

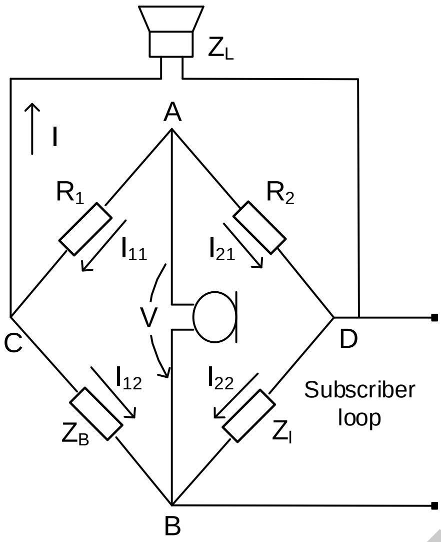

Figure 1.7: Hybrid transformer without coils used in an electronic phone.

[2] [4]. This hybrid is actually a Wheatstone bridge, which in balance ensures the separation between the circuits connected to its diagonals. The Z_B balancing impedance consists of an R-C group, corresponding to the capacitive impedance of the telephone line in the telephone band.

In balance conditions no current has to pass through the receiver ($I = 0$), which entails the following relations between the currents of the branches of the bridge: $I_{11} = I_{12} = I_1$; $I_{21} = I_{22} = I_2$. A zero current through the receiver means a zero potential difference, that is, $V_{C-D} = 0 \Rightarrow V_{A-C} = V_{A-D}$. From the above results the following equations, including the condition of the bridge balance.

$$V_{A-B} = I_1 \cdot (R_1 + Z_B) = I_2 \cdot (R_2 + Z_l) \ ; \ V_{A-C} = I_1 \cdot R_1 = V_{A-D} = I_2 \cdot R_2 \tag{1.5}$$

From the previous equation results the balance condition:

$$\frac{I_1}{I_2} = \frac{R_2}{R_1} = \frac{R_2 + Z_l}{R_1 + Z_B} \Rightarrow \frac{R_1}{R_1 + Z_B} = \frac{R_2}{R_2 + Z_l} \tag{1.6}$$

In the subscriber interfaces of local telephone exchanges it is necessary to use more complex hybrid transformers, which besides the interfacing between the 4-wire circuit and the 2-wire circuit must also ensure the galvanic separation of the circuits. An electronic hybrid transformer diagram usable in the subscriber interfaces of telephone exchanges is shown in 1.8 [7]. Transformers T_{4r}, T_{4t} and T_2 provide galvanic separation of the hybrid between 4-wire and 2-wire circuits. In the case of a digital telephone exchange, where the hybrid transformer connects the subscriber line and the PCM encoder, respectively PCM decoder, transformers T_{4t} and T_{4r} may be missing, not being necessary a galvanic isolation between the hybrid transformer and the PCM coder/decoder. The effective balance bridge is made up of resistors R_1, R_2, impedance Z_B and impedance Z_{r-l}, that is the subscriber line impedance reflected into the T_2 transformer primary winding, impedance given by the equation:

$$Z_{r-l} = \left(\frac{n_1}{n_2}\right)^2 \cdot Z_l \tag{1.7}$$

The balancing bridge is identical to that used in electronic phones (see Fig. 1.7) and the balancing condition of the bridge will be:

$$\frac{R_1}{R_1 + Z_B} = \frac{R_2}{R_2 + Z_{r-l}} \tag{1.8}$$

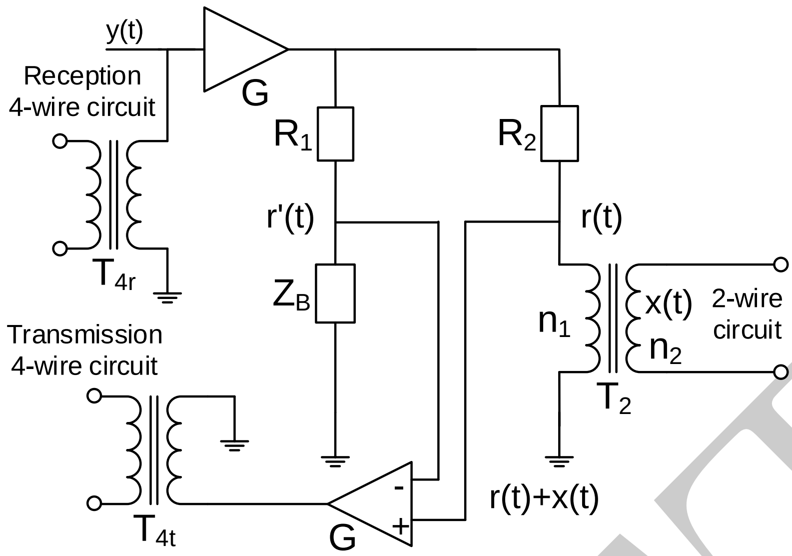

Figure 1.8: Electronic hybrid transformer with galvanic separation.

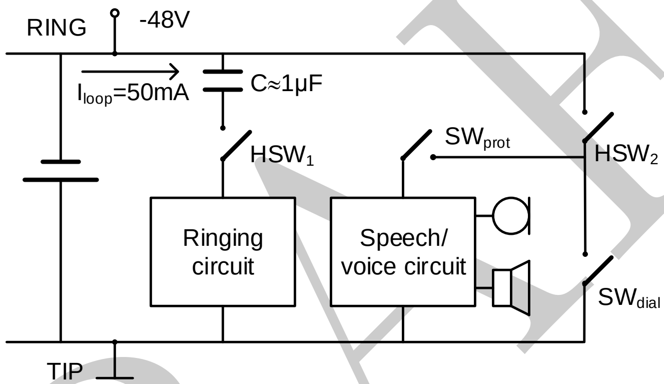

Figure 1.9: Schematic block diagram of an analog telephone.

1.3.1 Schematic block diagram of an analog telephone

An analog telephone is made up of four basic blocks [2] [3] [4]:

- the hook switch, which establishes the state of the device, which may be active (OFF-HOOK) or inactive (ON-HOOK).
- the dialing circuit that encodes and sends the called number to the PBX. The number can be coded using two techniques, namely pulse dialing or DTMF tone dialing (Dual Tone Multiple Frequency).
- the ringer circuit, which may be electromagnetic or electronic. This circuit alerts the user when a phone call is received.
- the voice/speech circuit, including the microphone, headphone/speaker, possibly microphone and speaker amplifiers and a hybrid transformer that separates the two circuits. The voice circuit ensures the acoustic signal - electric signal conversion and vice versa, the electric signal - acoustic signal conversion.

The schematic block diagram of the telephone in question is shown in Fig. 1.9, where we can also see the names of the wires of the pair representing the subscriber loop as well as the remote power feeding of the device [2].

In Fig. 1.9 the hook switch is represented by the switches HSW_1 and HSW_2 and the dialing circuit is represented by the switch C_{dial} (pulse dialing is supposed) and SW_{prot}. The hook switch is driven by a mechanical system (or optical system in some newer phones) that detects whether the handset is lifted or not.

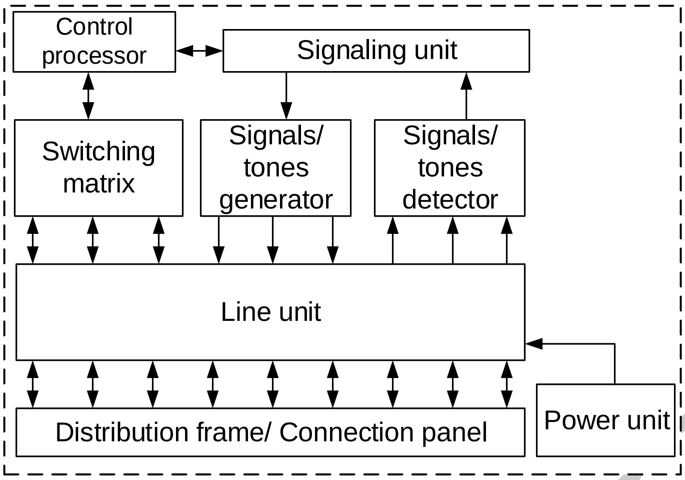

Figure 1.10: Schematic block diagram of a local exchange.

In inactive state, the HSW_1 switch is closed and connects the ringer to the telephone line through capacitor C, which blocks the DC voltage from the line to the ringer. The HSW_2 switch is open and disconnects the voice circuit and the dialing circuit from the telephone line. In the active state, the HSW_1 switch is open and disconnects the ringer from the line, and the HSW_2 switch is closed and connects the voice circuit and the dialing circuit to the telephone line. Dialing pulses are generated by switch SW_{dial}, controlled by a mechanical timer (rotary dial) or an electronic one (in electronic phones), and during dialing switch SW_{prot} is open and protects the voice circuit against the pulses generated during the dialing process. During the call, the switch SW_{dial} is open and the SW_{prot} switch is closed and feeds power to the voice circuit.

1.3.2 Schematic block diagram of a local exchange

The block diagram of Fig. 1.10 shows the main blocks/modules of a local telephone exchange (CO - Central Office) or PABX (Private Automatic Branch Exchange) to which subscribers are connected [8].

From the point of view of network access of the subscribers, the following modules should be mentioned:

- the main distribution frame, which represents a connection panel where the twisted wires of the subscriber loops are connected. This panel allows connecting/disconnecting the wires (pairs), isolating them in case of failure or during maintenance work and for performing measurements.

- the line unit, which includes overvoltage/overcurrent protection equipment, hybrid transformers, power supply circuits, filters, amplifiers, PCM encoders/decoders.

- the power supply block, which includes DC voltage generators that provide remote power feeding of the telephone terminals.

- signal generator that generates signals/tones used in access and supervision signaling on the subscriber loop.

- signal/tone detectors, which detect the loop current and the signals/tones generated by the telephone device in the access signaling and address signaling processes that take place on the subscriber loop.

- the signaling block, which controls the signaling processes that take place on the subscriber loop. This block selects the signals that are sent to the telephone device in different phases of the signaling processes, respectively interprets the signals (or signal changes) generated by the telephone device during the different signaling processes.

1.4 Signaling on the subscriber loop

The signaling that takes place on the subscriber loop (between the telephone and the local exchange or PBX) can be divided into four major categories. [2] [3] [4]:

- access signaling: informs the local exchange (Central Office -CO)/PBX exchange about the status of the telephone terminal. In the case of analog access, the signaling used is of loop start type and consists of the detection of the current I_{loop} on the subscriber loop. If the phone is in active state, the circuit formed by the voltage source in the exchange, the telephone line and the telephone device is closed and a current detector in the exchange detects the loop current I_{loop}. If the phone is in the inactive state then the current does not pass through it (see the block diagram of the telephone device - Fig. 1.9) and we have no loop current. If the active state of the phone is detected (transition to the active state can be interpreted as a request for access in the network) then the local exchange responds with the dial tone, if there are resources available in the network to continue the call (i.e. the access request is accepted), or with the busy tone, if there are no resources to continue the call or if the phone is locked for some reason (i.e. the access request is rejected). The dial tone usually is a continuous sinusoidal signal with frequency $f = 400 - 420Hz$, and the busy tone usually is a discontinuous sinusoidal signal with frequency $f = 400 - 420Hz$ and the timing $0.5s$ ON - $0.5s$ OFF. The amplitude of the mentioned tones is $\approx 0.5V$. In some networks other signals can be used to "encode" the states of acceptance or rejection of the access request of the telephone terminal in the network.
- address signaling: consists of sending the called number to the local exchange - see details below.
- alert signaling: informs the telephone terminal about the reception of a call, that is to say, it is the signaling that causes the phone to ring. In the case of analogue access, the local exchange sends to the telephone terminal a signal that is applied to the ring circuit (see the schematic block diagram of the telephone device - Fig. 1.9). The ring signal is usually a sinusoidal signal or an amplitude modulated sinusoidal signal. The signal frequency is $f = 20 - 40Hz$, and the amplitude is $A = 80 - 150V$, which can generate interference (noise pulses) in neighboring pairs. The ring signal timing is usually $2s$ ON and $4s$ OFF, but it depends on the local exchange.
- supervisory signaling: informs the subscriber about the status of the ongoing call. If the called subscriber is busy or the call does not pass due to network problems (congestion, faulty circuits, etc.)then the calling terminal receives a busy signal. If the call goes through and the called terminal receives the call then the calling terminal receives the ring back tone, which is an intermittent sinusoidal signal (or a modulated sinusoidal signal), but which differs from the busy tone. The ring back tone is generated by the local exchange where the called subscriber is connected and depends on the exchange. The timing of the ring back tone is, usually, identical to the timing of the ring signal. Special tones (e.g. faulty line, congested circuits, non-existent number, etc.) are part of the supervision signaling, the subscriber being informed about the reason for not being able to make a phone call. In some networks, the special tones are replaced by the busy tone.

1.4.1 Address signaling

The address signaling consists of encoding in a certain format and transmitting to the local exchange the number of the called subscriber. In the case of analogue access, the numbers/symbols of the telephone number (or the identifier of a terminal, equipment or service) can be encoded by a sequence of pulses (pulse method) or by a combination of tones (DTMF - Dual-Tone Multi-Frequency method).

Pulse signaling

Each digit in the dialed number is encoded by a number of pulses [2] [4]. The pulses are generated by interrupting the subscriber loop for a short period ($I_{loop} = 0$) - see Fig. 1.11. For each digit, the number of impulses is equal

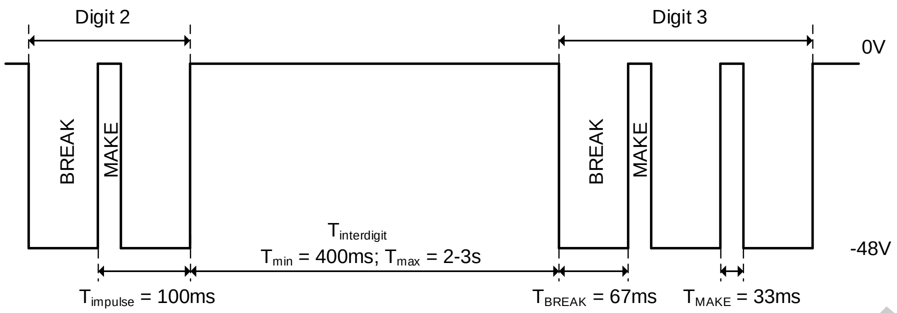

Figure 1.11: Pulse dialing.

to the value of the digit, except for the number 0 to which 10 pulses correspond.

Note: in some networks other encoding of the digits are used, e.g. digit 0 is encoded by a single pulse, digit 1 by 2 pulses and so on.

Note: short interruptions (tens - hundreds of ms) of the loop current do not cause the telephone device to be disconnected. To make this happen, the loop current interruption must be several seconds.

The BREAK period, when the loop is interrupted, is $T_{BREAK} \approx .67ms$, and the MAKE period, when the loop is closed between two consecutive interruptions, is $T_{MAKE}\ approx.33ms$, the nominal impulse transmission rate being $R = no.\ imp/s = 10imp/s$. As a rule, the telephone exchanges allow certain deviations from the nominal value of the pulse rate, the minimum value being $R_{min} = 8 - 9imp/s$, and the maximum one $R_{max} = 11 - 12imp/s$. The time period separating two digits (interdigit time) is between several hundred of ms and several seconds.

In the block diagram of the telephone device presented in Fig. 1.9 the dialing pulses are generated by the switch C_{call}, which is open during BREAK period and closed during MAKE period.

Note: there is not specified a telephone number terminator symbol. The local exchange considers that the dialing is completed if no pulses are generated after a period of several seconds (period > "maximum interdigit time")

The pulse dialing has several of disadvantages, among which can be mentioned: the slow transfer rate of the telephone numbers, which implies a relatively long duration of the dialing process; distortion of impulses transmitted on the subscriber loop, which may cause wrong calls; transient signals of large amplitudes (even $300V$ in certain situations), which may cause interface equipment failure in the local exchange; induction of impulse noise in neighboring pairs, which may affect the transmissions that occur on these pairs (especially data transmissions). The advantage is that this solution is simple and can be easily implemented, especially in rotary dial phones.

Tone (DTMF) signaling

For each digit/symbol a set of two different frequencies is used, namely [2] [3] [4] (F_{inf_i}, F_{sup_j}), $i, j\ in\ 1, 2, 3, 4$ - see Tab. 1.1. DTMF tones last (approximately) $T_{DTMF} = 100$ms. Unlike the pulse dialing besides the 10 digits, several symbols are available, most often the * and # symbols are used. These symbols are available on all electronic phones.

The frequencies used for the DTMF dialing are located in the lower half of the telephone band, between $700 - 1700Hz$, which is the transmission band (by convention) of a calling modem/data equipment in the case of full-duplex transmission on the telephone line implemented using the FDD (Frequency Division Duplexing) technique. In the case of an automatic call made by the modem, the transmission of the telephone number will be performed,of course, in the transmission band of the modem. The frequencies used for the DTMF dialing are located in the central part of the modem transmission band (lower half of the voice band) to reduce the distortions caused by the transmission filters - see Fig. 1.12.

Nonlinear distortions of the line can create unwanted harmonic components of the DTMF signals, which can negatively affect the decoding of DTMF tones - see the example below. The solution to this problem is the

Table 1.1: DTMF tones frequencies.

$F_{sup}[Hz]$ / $F_{inf}[Hz]$	1209	1339	1477	1633
697	1	2	3	A
770	4	5	6	B
852	7	8	9	C
941	$*$	0	#	D

Table 1.2: DTMF tones amplitudes.

	$A_{min}[mV]$	$A_{nominal}[mV]$	$A_{max}[mV]$
F_{sup}	160	190	200
F_{inf}	130	150	160

selection of DTMF frequencies so as to avoid the harmonic relationships between the frequencies F_{inf} and F_{sup} (Tab. 1.1). Also the DTMF tones in the lower set (which can generate harmonics located in the upper set) have smaller amplitudes (Tab. 1.2), thus being better protected against nonlinear distortions.

The tones sent on the subscriber loop can be simulated by the voice signals or other signals captured by the phone's microphone (ambient noises), which can generate erroneous decoding of the DTMF symbols transmitted to the telephone exchange during the address signaling process. Using a pair of frequencies to encode a symbol significantly reduces the probability of simulation by the signals picked up by the microphone, not being necessary to use more tones (e.g. 3 tones) to encode the numbers/symbols on the phone's keypad.

The advantages of the DTMF dialing consist of the following: higher speed of transmission of digits/symbols, which reduces the time required to dial the telephone number; no impulse noise is generated in the neighboring pairs and thus the transmissions that take place on these pairs are not affected (this is especially the case of the data transmissions); the distortions induced by the channel have smaller effects on the process of decoding the digits/symbols; it is possible to define additional symbols besides the 10 digits, symbols that can be used to access additional services offered by modern switching nodes; DTMF signaling is performed in the telephone band, thus it is possible to transmit the DTMF symbols through the entire telephone network and use DTMF tones to access services offered by different nodes in the network (nodes different from the local exchange). The impulses used to encode the numbers (pulse dialing) can only be transmitted to the local exchange, not being possible the digital conversion respectively the transmission through the network of these impulses.

Example: It is considered a DTMF system that encodes Key 1 (600Hz, 1200Hz) and Key 2 (600Hz, 1400Hz). The transmission line has a nonlinear transfer characteristic: $F(x) = x^2 + x$.

When the Key 2 is pressed, the transmitted DTMF signal is: $s_{DTMF} = a_1\ sin(2\ pi600t) + a_2\ sin(2\ pi1400t)$, the spectrum of this signal being shown in Fig. 1.13.

At the input of the DTMF receiver is obtained the signal: $s_r(t) = s^2_{DTMF}(t) + s_{DTMF}(t)$, meaning: $s_r(t) = a_1 \sin(2\pi 600t) + a_2 \sin(2\pi 1400t) + \frac{a_1^2}{2} + \frac{a_2^2}{2} - \frac{a_1^2}{2}\cos(2\pi 1200t) - \frac{a_2^2}{2}\cos(2\pi 2800t) + a_1 a_2 \cos(2\pi 800t) -$

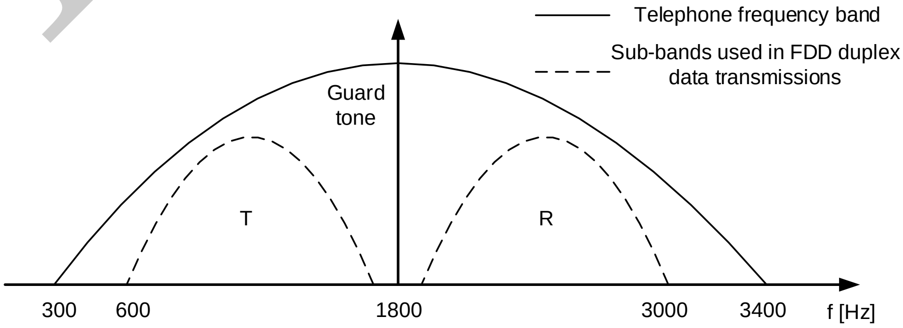

Figure 1.12: Telephone frequency band and sub-bands used in FDD duplexing.

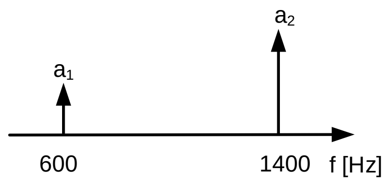

Figure 1.13: Spectrum of the transmitted DTMF signal.

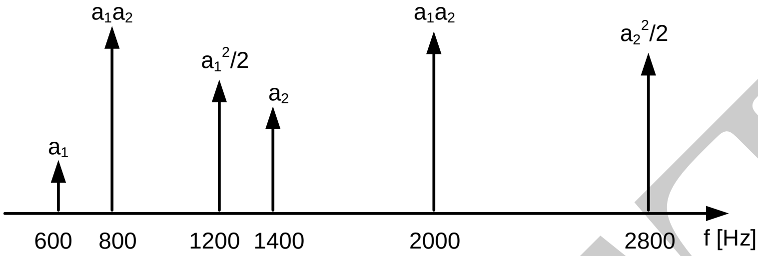

Figure 1.14: Spectrum of the received DTMF signal.

$a_1 a_2 \cos(2\pi 2000t)$. The spectrum of the received signal is presented in Fig. 1.14.

If we neglect the components on the 800Hz, 2000Hz and 2800Hz frequencies (which are not part of the DTMF system considered), we can see that the receiver cannot decide between the two keys. Hence the need to avoid the harmonic relationship between the frequencies F_{inf} and F_{sup}.

Other aspects related to signaling on the subscriber loop

At least two particular aspects must be mentioned in relation to the signaling on the subscriber loop in case of analogue access. The first aspect is related to the signaling between the telephone device and the PABX exchanges, which can offer additional and more special functions such as: redirecting the call, blocking a telephone terminal, transferring the call, teleconferencing, defining and transferring the service class, placing in hold one or several calls, etc. Accessing these functions, when using analog terminals, can be achieved with the FLASH signal/pulse (also called HOOK FLASH). This signal is represented by a short interruption of the subscriber loop, typically lasting between $90ms$ and $600ms$ (in the case of some phones between $80ms$ and $1s$), the usual values being $90ms$, $120ms$, $270ms$, $375ms$ and $600ms$. The FLASH impulse announces the PBX exchange that the terminal wishes to access the special functions of the exchange, to which the exchange will respond in a specific way. The FLASH pulse duration is longer than the BREAK pulse duration, which should eliminate unwanted interactions with pulse address signaling. However, it should be noted that modern PABX exchanges require the use of DTMF addressing to access special functions, especially since the special function selection codes include the symbols * and #.

The second aspect is related to the Calling Line Identification Presentation (CLIP) signaling, which allows the identification of the calling terminal and is a component of the supervision signaling. In the case of analogue access, the number of the calling terminal can be transmitted to the called terminal (by the local exchange where it is connected) using DTMF symbols or FSK (Frequency Shift Keying) modulation, a transmission that takes place before or during the alert signal (i.e. before the ring signal or in parallel with this signal). If the telephone terminal is able to decode this code and has a display, then the number of the calling terminal will be displayed during the call, possibly even after the call is terminated if the called terminal has not answered.

1.4.2 Signaling sequence characteristic of a local telephone connection

As a conclusion to those discussed in the previous paragraphs, in Fig. 1.15 is presented the full signaling sequence associated with a local telephone connection, i.e. a telephone connection made between two terminals connected to the same local telephone exchange or PBX. With italic font are identified the signals of a generic

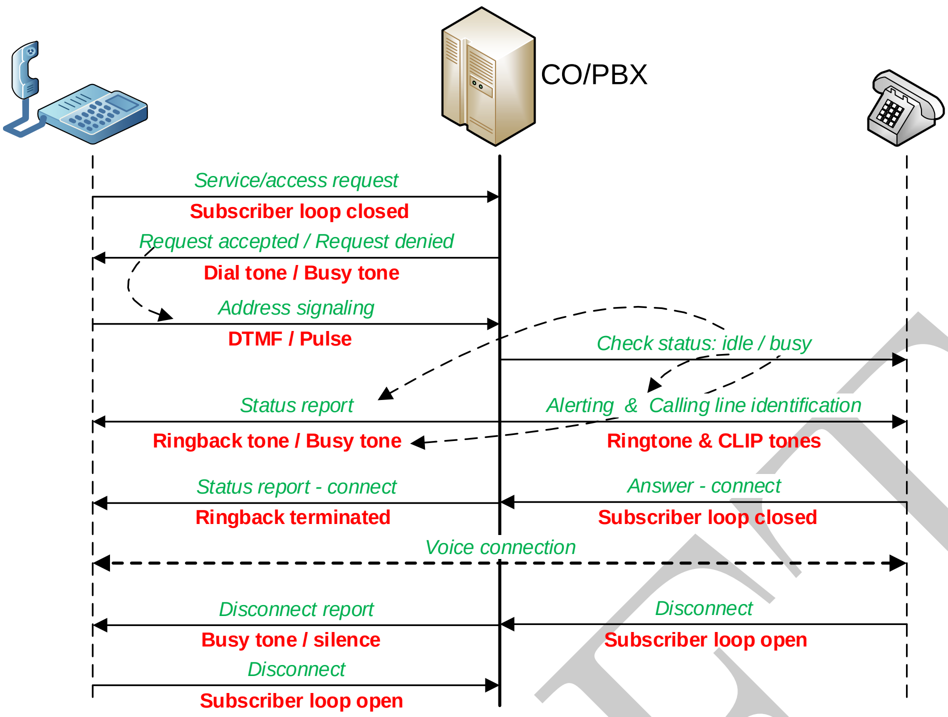

Figure 1.15: The signaling sequence associated with a local telephone connection.

signaling process, independent of the type of access on the subscriber loop (analog or digital), and with bold font are identified the signals of the same signaling process, but in the case of an analog access on the subscriber loop.

1.5 Application

1. Analyze the topology and identify the components of the access network characteristic to a private telephone network, respectively a public telephone network. How can full-duplex transmission be achieved on the subscriber loop? Different possible methods are analyzed and compared.

2. Analyze the general block diagram and the functioning of an analog telephone.

3. Analyze the internal schematic of a small capacity PABX exchange and identify the main functional blocks.

4. Analyze the schematic of the power feeding bridge in the exchange. Why is this circuit necessary and what effects does it have?

5. Analyze the schematic the hybrid transformer (implemented using transformers) and determine the balance condition and the return loss for the particular situation $L_{cp} = 0$. Consider the cases in which $Z_L = 0$ respectively $Z_L \neq 0$.

6. Analyze the schematic of the electronic hybrid and determine the balance condition for the following particular situations: $R_1 = R_2$, $R_1 = 2R_2$. Consider the cases in which $Z_L = 0$ respectively $Z_L \neq 0$.

7. Analyze the pulse dialing technique. What are the advantages and disadvantages of this technique?

8. Analyze the DTMF dialing technique. What are the advantages and disadvantages of this technique? Why are two frequencies used to encode a symbol? What conditions apply to DTMF frequencies?

9. How long does it take to send the numbers 0264 999 999 and 0262 111 111 to the telephone exchange using the pulse dialing technique? What if the DTMF technique is used? Consider separately the case of the

manual dialing, performed by the user, respectively of the automatic dialing, performed by the telephone terminal.

10. There are considered two analog telephone terminals connected to a PBX. The probes of a digital oscilloscope/spectrum analyzer with two channels are connected in parallel on the two subscriber loops. WARNING! the ground wire of each probe is connected to the TIP (GND) wire on both subscriber loops. The following operations are performed:

    (a) Measure the supply voltage on the subscriber loops with the phones in the ON HOOK (inactive) state. Explain the measured values.

    (b) Measure the supply voltage on the subscriber loops with the phones in the OFF HOOK (active) state. Explain the measured values.

    (c) Measure the frequency and the amplitude of the dial tone.

    (d) Call an active terminal to generate a busy signal on the subscriber loop of the calling terminal. Measure the frequency, timing and amplitude of the busy tone.

    (e) Call an idle terminal to generate a ring signal on the subscriber loop of the called terminal and a ring back tone on the subscriber loop of the calling terminal. Measure the amplitude, frequency and timing of the two signals. Explain the values of the ring signal amplitude. What is the time relationship between the ring signal and the ring back signal? What is the relationship between the timing of the two signals?

    (f) Consider the situation from the previous point. On the subscriber loop of the called terminal, the caller identification signal (CLIP) is displayed. Where is it inserted and when is the CLIP signal transmitted? What is the duration of this signal?

    (g) Establish the voice path between the two terminals. The voice signal is visualized and the amplitude and frequency band of this signal are measured on the subscriber loop of the calling telephone (source) and of the called telephone (destination), i.e. at the entrance and exit of the telephone exchange.

    (h) Configure a telephone terminal for impulse dialing and dial a digit (larger than 2). The generated impulses are visualized and measure the following parameters: the amplitude of the impulses, the amplitude of the transient signals generated on the line and the BREAK, MAKE and Interdigit time periods.

    (i) Configure a telephone terminal for DTMF dialing. Dial a few digits and using the spectral analyzer and the oscilloscope measure the amplitude, frequency and duration of the DTMF tones.

    (j) Make a call from a terminal to another one and answer the call after 1-2 ring signals. Measure and store on the digital oscilloscope the signals on the two subscriber loops. Analyze the stages of the signaling process between the telephone exchange and the telephone terminals. Identify the main phases of the signaling processes under discussion.

    (k) Generate a FLASH pulse with a telephone terminal. Store the pulse on the digital oscilloscope and measure the amplitude and duration of this pulse. Can FLASH pulses be generated if we have pulse dialing? What is the interaction between the two processes (FLASH pulse generation and pulse dialing), if FLASH pulse generation would be possible?

Chapter 2

Telephone devices for analog subscriber loops

2.1 Introduction

The chapter proposes the study of the following issues:

- analysis of the electrical diagrams of some electronic analog phones with pulse dialing, DTMF dialing and combined pulse/DTMF dialing.
- a brief overview of cordless telephones.
- analysis of the internal diagrams of some rotary dial and electronic analog telephones.
- analysis of how to set up/configure simpler electronic phones and more complex ones, including cordless phones.

This chapter proposes basically a "little exercise" of electronics in which the diagrams of relatively simple telecommunications equipment will be analyzed, but such an exercise is useful for acquiring the skills needed to analyze more complex telecommunications equipment and terminals. There is also an exercise in setting up/configuring more complex electronic telephone terminals, an exercise that can be useful for acquiring the skills needed to configure telecommunications terminals and equipment.

2.2 Electrical diagrams of analog phones

2.2.1 Electrical diagram of a rotary dial telephone

The electrical diagram of Fig. 2.1 includes the following components [6] [9]:

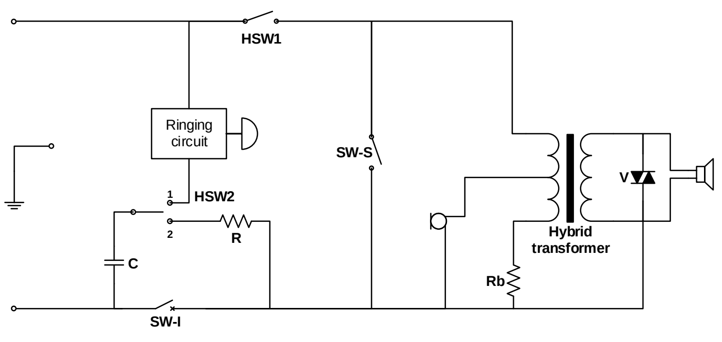

Figure 2.1: Electrical diagram of a rotary dial telephone with pulse dialing.

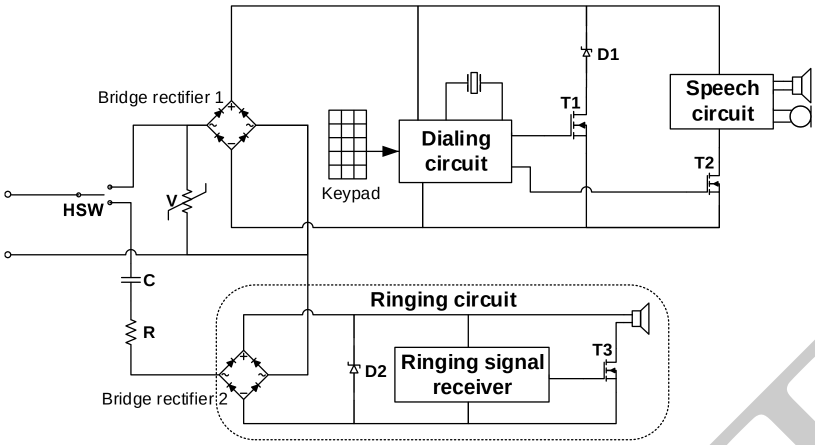

Figure 2.2: Electrical diagram of an electronic telephone with pulse dialing.

- the hook switches CF1 and CF2; they are made up of mechanical switches operated by a lever that support the handset of the telephone in idle state. CF1 connects the voice circuit to the telephone line in the OFF HOOK state, respectively disconnects this circuit in the ON HOOK state. CF2 connects the ringer to the telephone line in the ON HOOK state, respectively disconnects it in the OFF HOOK state. In the OFF HOOK state CF2 closes the R (100Ω) - C ($1\mu F$) circuit, connected in parallel with the CI switch of the dial disc, R-C circuit which has the role of preventing sparks from occurring between the CI contacts, when these contacts are closed and open.

- the ringer; it is usually an electromagnetic ringer (buzzer), but it can also be an electronic ringer. The ringer is connected to the telephone line through the capacitor C.

- the dial disc; it is basically a mechanical timer acting on switch CI that generates the dialing pulses on the line. During the dialing, the switch CS is closed and short circuits the voice circuit. This prevents sending of disc impulses to the receiver, these impulses being disturbing.

- the voice circuit; this circuit includes: microphone, headset, hybrid transformer, balancing resistor Rb (680Ω) of hybrid transformer as well as varistor V, which acts as a voltage limiter and ensures overvoltage protection of the telephone speaker. The handset, which includes the microphone and the speaker, connects to the telephone device on three wires, with a common wire used for both the microphone and the headphone.

The telephone is connected to the line on two wires (RING and TIP), the typical case, but also can be connected on three wires (a grounding wire is used) for party line subscribers.

2.2.2 Electrical diagrams of electronic analog phones

Electrical diagram of an electronic telephone with pulse dialing

The diagram is shown in Fig. 2.2 and is composed of three main blocks: the line connection circuits, the ringer circuit and the voice and dialing circuit [10] [11].

The CF switch is the phone hook switch. This switch connects to the line the ringer circuit in the ON HOOK state, respectively the voice and dialing circuits in the OFF HOOK state. The capacitor C separates the ringer circuit from the DC voltage on the subscriber loop, and the V varistor (or other voltage limiting device) protects the voice and dialing circuits against accidental overvoltages on the subscriber loop.

The two bridge rectifiers in the diagram have the following roles:

- Bridge rectifier 1 protects the telephone device circuits from changing the polarity of the DC voltage on the subscriber loop due to the reverse connection of the two wires of the line to the telephone device. Due

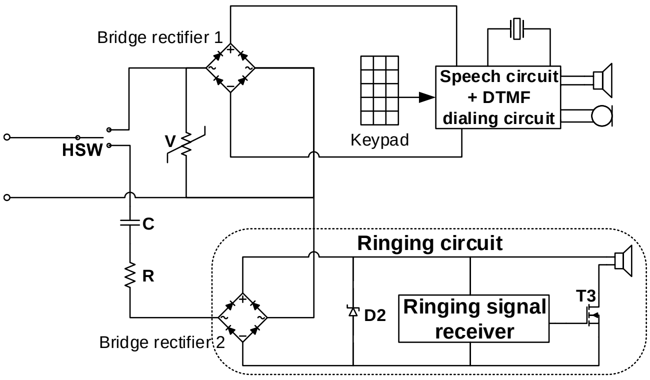

Figure 2.3: Electrical diagram of an electronic telephone with DTMF dialing.

to the low level of the voice signal, compared to the DC voltage on the line, it is not rectified (the bridge diodes cannot be opened/closed by the voice signal), but the diodes can induce nonlinear distortions.

- Bridge rectifier 2 rectifies the ring voltage (AC voltage with high amplitude) and ensures the DC supply voltage for the ringer circuit. The ring signal has an ON-OFF repetition scheme. When there is a ring signal we have a DC supply voltage of the ringer, which will ring, and when we do not have a ring signal we have no DC supply voltage of the ringer, which does not ring.

The zener diode D2 limits the rectified ring signal, that is, it limits the DC supply voltage of the ringer. The D2 diode can be replaced with a DC voltage stabilizer block.

The dialing circuit disconnects the voice circuit during the dialing by blocking the T2 transistor and generates the disc pulses by interrupting the subscriber loop using the T1 transistor. So, during dialing transistor T2 is open (blocked), thus preventing the sending of the disc pulses to the speaker, and transistor T1 is closed/opened for generating the MAKE and BREAK pulses, which characterize the pulse dialing. The dialing circuit has a quartz crystal or RC oscillator that generates the clock signal necessary for the timers that give the timing of different stages of the dialing process. A keypad attached to the dialing circuit allows to input the numbers and other symbols used in the dialing process.

The zener diode D1 acts as a protection device of the dialing circuit against short-circuiting the line during the MAKE pulse. During this impulse, the transistor T1 is closed and there is a very low voltage on this element, which would practically short-circuit the line and the dialing circuit. The zener diode provides a high enough voltage for the correct functioning of the dialing circuit during the MAKE pulse.

The voice circuit includes the microphone amplifier and the speaker amplifier as well as an electronic hybrid that separates the microphone circuit from that of the headphone/speaker circuit and thus reduces the side effect.

Electrical diagram of an electronic telephone with DTMF dialing

The diagram is shown in Fig. 2.3 and is composed of the same major blocks as the diagram in Fig. 2.2. Due to the fact that it is no longer necessary to generate pulses and to interrupt the subscriber loop, the electrical diagram becomes simpler, as regards the functional blocks. The voice and dialing circuits can be combined into a single functional block, possibly implemented as a dedicated integrated circuit. For other functional aspects see the explanations from the previous diagram.

Electrical diagram of an electronic telephone with pulse and DTMF dialing

The diagram is shown in Fig. 2.4 and is composed of the same three major blocks as the diagrams presented in Fig. 2.2 and Fig. 2.3.

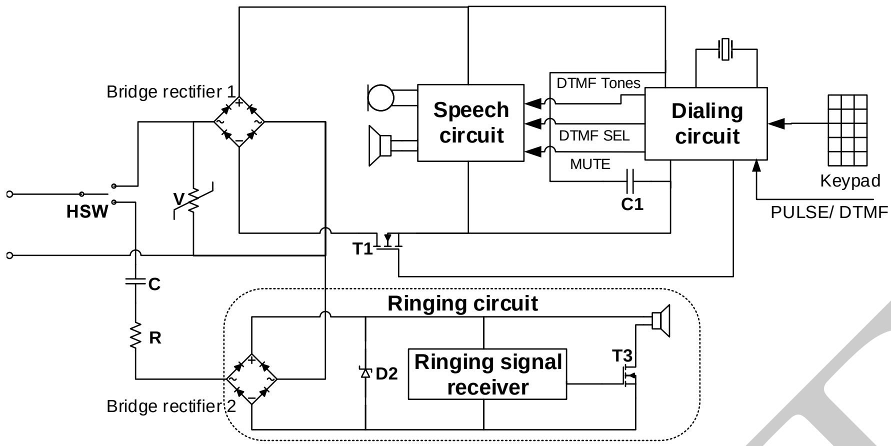

Figure 2.4: Electrical diagram of an electronic telephone with pulse and DTMF dialing.

The dialing circuit can work both in pulse and DTMF modes, selecting the working being performed by the user with the aid of a switch or, in the case of more complex telephones, by entering a selection code specific to the telephone device.

In the case of pulse working mode during the dialing process, a MUTE signal is generated to the voice circuit to prevent the sending of pulses to the headphone or speaker. The BREAK and MAKE pulses are generated by the T1 transistor which operates in switching mode and opens and closes the subscriber loop. During the BREAK impulse, when the subscriber loop is open, the dialing circuit has no power supply and will not be able to operate. To solve this problem, the capacitor C1, with a capacity of several μF or several tens of μF, was placed in the electrical diagram. This capacitor maintains the supply voltage of the dialing circuit during the interruption of the subscriber loop. The low power consumption of the dialing circuit (low supply current required) allows providing the supply voltage through this simple solution during BREAK pulses, the capacitor being charged during MAKE pulses, when the loop is closed and we have loop current

In the case of the DTMF dialing mode, the T1 transistor is ON continuously, and the DTMF tones are sent to the voice circuit, being heard in the headphone and amplified by the microphone amplifier (or another dedicated amplifier). The SEL signal allows the microphone to be switched off and its amplifier used to amplify DTMF signals. The T1 transistor can generate the FLASH pulse, if it receives an appropriate command.

2.3 Internal diagrams of analog telephones

2.3.1 Internal diagram of a rotary dial telephone

This type of telephone device has, in principle, a motherboard with the electrical components necessary for the operation, the board to which the following modules are connected (see Fig. 2.5):

- the handset, which contains the headphone and the microphone. The connection to the motherboard can be made on three or four wires. In the first case the microphone and headphone have a common wire, while in the second case the two circuits are completely separate.
- the dialing disc, which contains a mechanical timer that activates the electrical contacts that actually generate the dialing pulses, respectively short-circuits the voice circuit during the call. The disc connection is usually made on three wires - see the electrical diagram in Fig. 2.1.
- electromagnetic or electronic ringer (buzzer), connected on two wires to the motherboard.

At the motherboard is connected the telephone line on two or three wires, the third wire being the ground used in the past to connect the party line subscribers on the same telephone line.

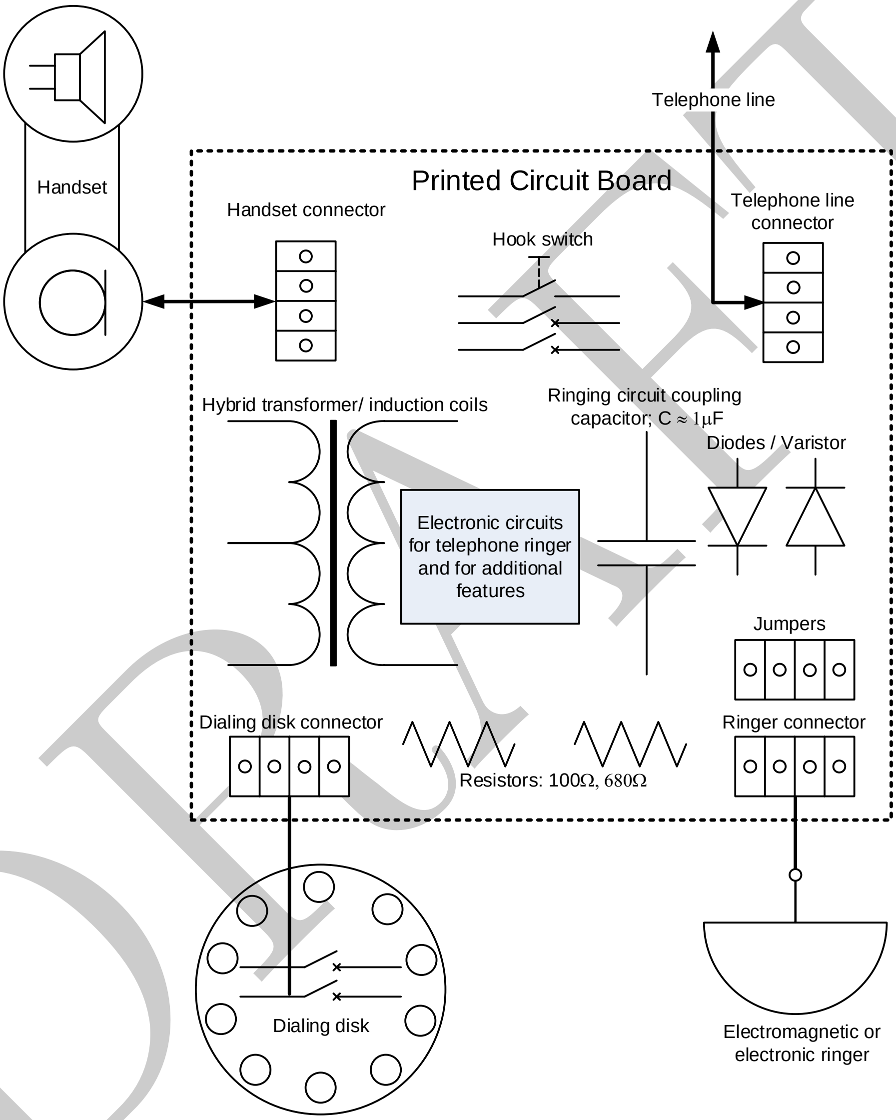

Figure 2.5: Internal schematic diagram a rotary dial telephone.

Note: in the past, due to the poor development of the telecommunication infrastructure, it was common practice to connect two subscribers on the same subscriber loop (party line or shared service line). The two subscribers had different phone numbers, and their separation was achieved using the ground wire. At one subscriber the ringer was connected between the TIP wire and the ground, and at the other subscriber between the RING wire and the ground, making it possible to call each subscriber individually, but not at the same time. The billing of the calls and the identification of the subscriber who made a call was also done with the help of the ground wire by measuring the potentials on the TIP and RING wires relative to the ground. This method does not ensure confidentiality of the call and can create problems of incorrect billing of the subscribers.

The motherboard contains, besides the connectors used to connect the line and modules mentioned above, the following components:

- hook switch - mechanical switch actuated by the handset support system.
- the ringer coupling capacitor(s).
- the hybrid transformer made with a low frequency transformer with center tap.
- voltage limiters (varistors or diodes) for protection of the headphone/speaker.
- resistors, components of the hybrid transformer and the dialing circuit.
- possible, electronic circuits for electronic ringer or for other features: e.g. the use of a keypad instead of the dialing disc or the use of the phone as an intercom or secretarial telephone.
- jumpers for changing the working mode, e.g. connecting party line subscribers or adding additional modules, such as the transfer button used in secretarial phones.

Note: it is possible to replace the dialing disc with an electronic circuit that includes a keypad, keypad control circuits, timers, electronic switches and control circuits of the electronic switches. The b circuit can be connected to the dialing disc connector, the supply power being taken from the phone board.

2.4 Internal diagram of an electronic telephone

This type of telephone device has a motherboard with the electrical and electronic components necessary for the operation, board to which the following modules are connected (see Fig. 2.6):

- the handset, which contains the headphone and the microphone. The connection to the motherboard is made on four wires, the two circuits being completely separated. The handset is connected using a RJ9 4P4C modular connector.
- the keypad, with which the dialing is performed, is connected to the motherboard with a multi-wire ribbon cable. The keypad controller is located on the motherboard.
- the electronic ringer, more precisely the speaker of this ringer, connected on two or three wires to the motherboard.

  Note: usually, the ringer speaker is not a speaker itself but a piezoelectric element capable of generating specific sounds for a telephone ringer.
- liquid crystal display. The more complex telephone devices use a display to display the time, the numbers typed, the number of the caller, the duration of the call, the missed call numbers, information about the operation mode, these telephone devices being programmable within certain limits.
- switches with which the dialing mode can be set, i.e. DTMF or pulse, respectively the ringer signal level. This is, as a rule, the case of simple electronic phones, on the more complex devices the aforementioned parameters being programmed from the keypad or from special keys. On some phones the ringer level can be adjusted using a potentiometer.

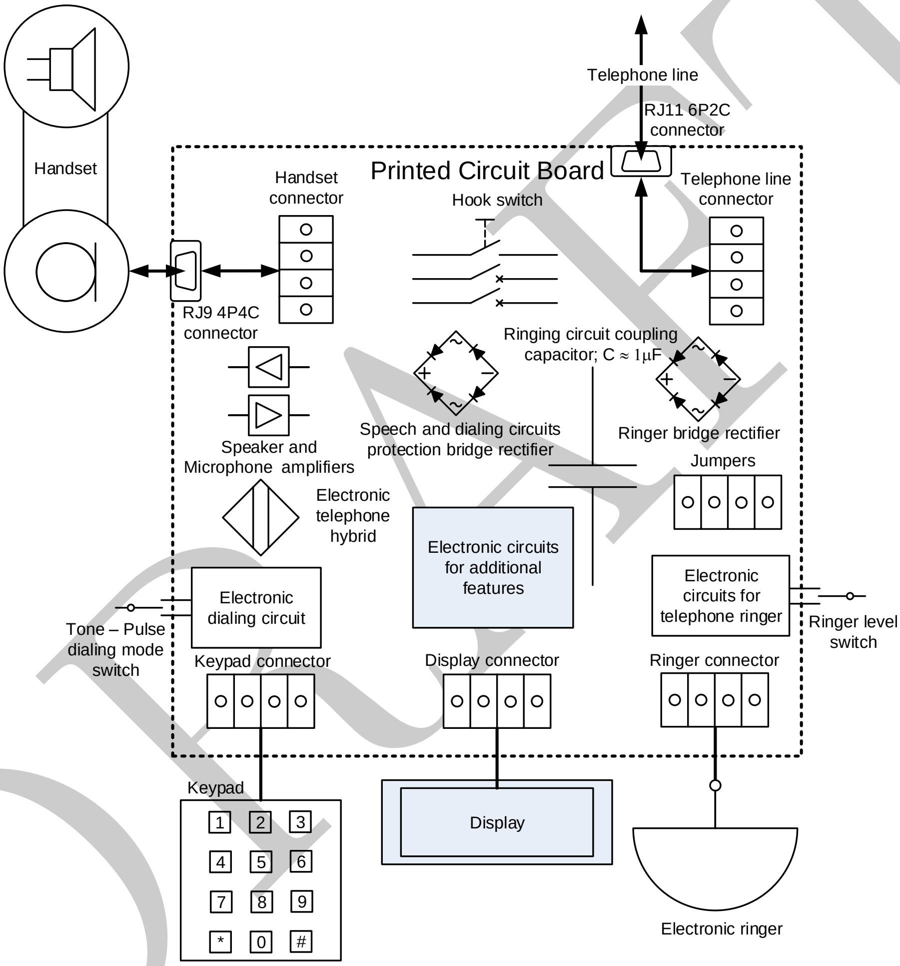

Figure 2.6: Internal schematic diagram of an electronic telephone device.

- keys for programming the phone's operating mode. This is the case of more complex phones that offer additional features.
- keys for selecting numbers stored in the phonebook/directory and for storing numbers in the phonebook.

The telephone line is connected to the motherboard on two wires using an RJ11 6P2C modular connector.

The motherboard contains, besides the connectors used to connect the telephone line and the modules mentioned above, the following components:

- hook switch - mechanical or optical switch operated by the handset support system.
- the ring coupling capacitor(s).
- bridge rectifier for the ringer circuit, respectively for the coupling of the voice and dialing circuits.
- microphone and speaker/headphone amplifiers.
- hybrid transformer made with an R-C bridge.
- electronic circuits for telephone ringer.
- the control circuit of the keypad and of the programming keys.
- jumpers to configure the phone or add additional modules.
- microphone and speaker for hands-free operations.

  Note: some more complex phones allow such operations and have a separate microphone and speaker for hands-free operations.

Note: some small compact phones, which are usually installed on the wall, have all the components, i.e. the motherboard and the rest of the peripherals (keyboard, ringer, display, etc.) integrated in the handset, the holder with which is installed on the wall including only the telephone line connectors. In some cases the hook switch and the ringer can be included in the holder.

The diagrams presented above should be considered in a general sense, since it is impossible to integrate all the functionalities and characteristics of electronic phones into a limited number of diagrams of relatively low complexity. In the following paragraphs will be exemplified some electronic phones and will be mentioned the functionalities they offer and the way in which these functionalities can be programmed/activated.

2.5 Cordless phones. A short presentation

2.5.1 Basic characteristics of cordless telephones

A cordless phone replaces the cable connection of the receiver (handset) with a radio connection [12]. Such a device is composed of two units (see Fig. 2.10): a base station (or simple base) that is connected to the subscriber loop (like any other corded telephone) and the receiver connected to the base station via a radio connection. The coverage range depends on the carrier frequency on the radio link between the base and the receiver. The base station must be powered from a separate power supply, as it is not possible to feed directly from the loop current, due to limitation of this current to several tens of mA. The handset is powered by a battery that is usually charged from the base.

The key difference between a mobile phone and a cordless one is that the base station is at the user, but in some specialized networks the base stations can be owned by a telecommunications operator. The new standards of digital cordless phones allow handover operations between the base stations and data transfer, besides voice communications and connection of several terminals to a base station, as well as direct communication between terminals connected to the same base station. Advanced applications, such as teleconferencing calls, are also possible.

The first cordless phones appeared in the mid-60s, early 70s. The first standards used analog modulations on the radio link. The first cordless phone solutions even used an acoustic connection between the handset and the base. The first models of cordless phones used the 1.7MHz band (AM modulation, 5 channels available). Later (1984) the 43-50MHz band and FM modulation were used, then the 900MHz unlicensed band (1994) was used. The 1.9MHz band was allocated with the advent of the Digital Enhanced Cordless Telecommunications (DECT)standard. Subsequently, standards appeared that work in the unlicensed bands of 2.4GHz (1998) and 5.8GHz (2003). Unlicensed bands (900MHz, 2.4GHz, 5.8GHz) are also used by other wireless systems (Bluetooth, WLAN, baby monitor) that can create interference. The 1.9GHz band (1880 - 1900MHz) is reserved only for cordless phones using the DECT standard, which guarantees a low level of interference. In 1995, the DSS (Digital Spread Spectrum) technique was introduced on the radio connection of cordless phones, a technique that spreads the data to be transmitted on several frequencies, which brings a substantial improvement of the security, making it more difficult to intercept these transmissions.

2.5.2 The DECT standard in short

DECT is a standard of digital communication developed especially for cordless phones, but not only [13] [14]. It was designed to replace the older standards working in the 900MHz band. It has been adopted practically all over the world, in the USA being required modifications of the frequencies used due to the specific allocation of frequencies in the 1900MHz band (1920 - 1930MHz - DECT 6.0). The DECT standard also can be used for other applications such as "baby monitor", data transfer, remote controls, industrial applications. An interesting application is the implementation of radio subscriber loops. Even though it is used especially in small offices and houses/apartments it is also integrated in PABX systems used by medium and large companies. The standard is capable of providing public access by using a large number of base stations and handover operations.

Cordless telephones network connections are made usually using an analog subscriber loops, but there are techniques which use VoIP interfaces. The average transmission power in the DECT standard is $10mW$, with a peak power of $250mW$. By using directional antennas and limiting the traffic capacity the coverage can be extended up to over $10km$, an important fact in wireless subscriber loop applications. The average transmission power is limited to only $4mW$ (peak power $100mW$) in the SUA. In 2011, the DECT ULE - "Ultra Low Energy" standard was developed, which allows, besides lowering the radiation level, the use of the DECT interface in security systems, home automation systems, smartphones, etc. The radio connection is characterized by a combined FDMA/TDMA access. 10 carriers with a separation of 1,728 MHz (only 5 carriers in DECT 6.0) and 2x12 time slots (for uplink and downlink) are defined. Security is ensured by 64-bit encryption. The audio encoder can be G726 (ADPCM - 32kbps), G711 (PCM), G722 (ADPCM with 7kHz band), G.729.1 or MPEG-4 (the last two with voice compression).

2.6 Examples of electronic phones, features and configuration

Concorde C-915 telephone

It is a simple electronic phone that offers some basic features found in all electronic phones. The selection of the dialing mode can be achieved with a switch, as well as the selection of the ringer signal level. Besides these switches the phone has three functional keys, namely (see Fig. 2.7):

- Hold - is basically a MUTE key that allows the microphone to be deactivated during the call.
- Flash - generates a fixed duration FLASH pulse, a pulse used to access the special functions of the PABX exchanges.
- Redial - allows to redial the last dialed number, which is stored in the phone memory.

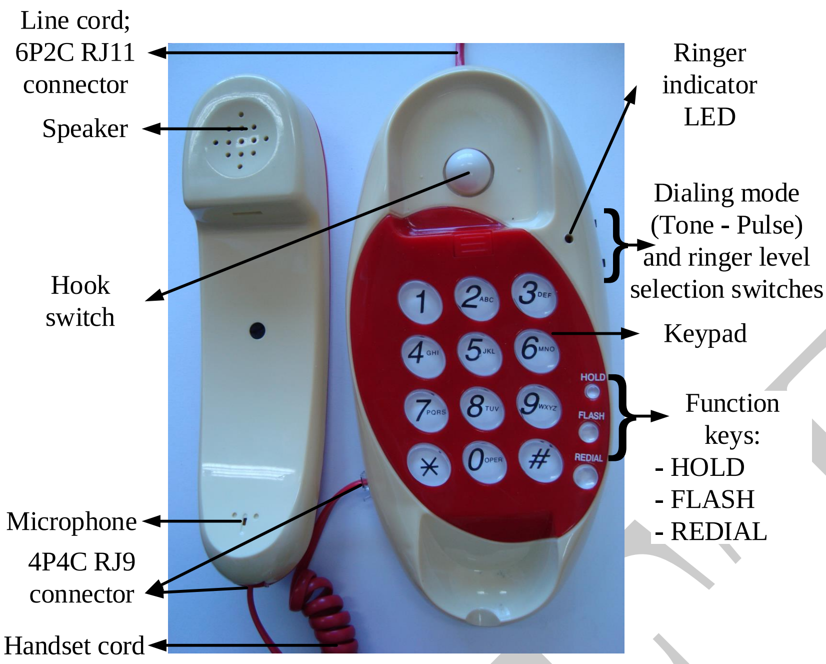

Figure 2.7: Concorde C-915 electronic phone.

Siemens Euroset 5010 telephone

It is an electronic phone that offers a number of additional features [15]. The operating mode is set using a set of 6 function keys. There are no switches used to set the dialing mode and ringer level, all phone functions are programmed using the function keys and numeric combinations introduced from the keyboard. The phone has a phonebook with 20 numbers on a maximum of 32 digits.

The features offered by this phone are, in short, the following:

- redialing the last dialed number.
- dialing numbers stored in the phonebook, respectively storing the numbers in the phonebook.
- "Baby call" or "Direct call" - the phone is locked and a number programmed by the user is called when pressing any key, except the programming key and the MUTE key. It is not possible to call another number until the function is deactivated.
- defining an emergency number, which can be called even if the phone is locked.
- adjusting the ringer level.
- setting the ringer frequency.
- setting the reception level (speaker volume).
- activating a music signal when the microphone is deactivated.

  Note: in this case, deactivation of the microphone also leads to deactivation of the receiver, during which the phone can send a music signal on the line, if set (music on hold).
- locking some groups of numbers. You can check the first three digits of the number.

  Note: this feature is useful for blocking long distance or international calls.
- locking the phone, not being possible to call any number except the emergency number.
- insertion of a prefix for external lines (in PABX networks), prefix established by the user.

Figure 2.8: Siemens Euroset 5010 electronic telephone.

- changing the dialing mode, tone or pulse mode.
- FLASH pulse generation.
- setting the duration of the FLASH pulse. This duration can be set at the values: $90ms$, $120ms$, $270ms$, $375ms$ and $600ms$.

The use of the programming keys is briefly the following (see Fig. 2.8):

- all special functions can be activated/deactivated with the programming key (function key 1).
- deactivation/activation of the microphone is performed with the function key 2.
- the activation/deactivation of the "Baby call" function is performed with the programming key (function key 1) and function key 6.
- the FLASH pulse is generated with function key 3 (R key).
- the redial of the last number as well as the insertion of breaks between the digits is performed with the function key 4. The key is also used in other operations, e.g. setting the prefix.
- function key 5 (Shift key) is used for selecting and saving the phone numbers in the phonebook; each number key can be associated with two telephone numbers, the selection being made with the Shift key.

  Note: Using the Shift key can be reduced the number of keys associated with the phonebook.

Note: the exact details related to the activation and configuration of the different features can be found in the user manual of the phone.

Siemens Euroset 5020 telephone

Is a digital telephone with multiple features, actually being an improvement of Siemens Euroset 5010 phone(see Fig. 2.9) [16]. The phone is equipped with an LCD display, an additional microphone and speaker for hands free operations. The phone also has a phonebook with 20 numbers on a maximum of 32 digits.

The additional features that this phone has, compared to the Siemens Euroset 5010, are the following:

Figure 2.9: Siemens Euroset 5020 electronic telephone.

- displaying the time, respectively the duration of the call.
- hands free call, a very useful functionality in teleconferencing.
- setting the ringtone beside setting the ringtone level.
- the functionality of Baby call/Direct call is activated using the programming key (function key 5) and a specific key combination; there is no separate function key to activate this function.
- display of the typed digits/symbols.
- display of the caller's number (Calling Line Identification Presentation - CLIP).

  Note: the CLIP functionality is conditioned by the support provided by the network to which the terminal is connected.
- call list with maximum 50 entries; the caller's number is stored as well as the date and time of the call - this information can be displayed using the Shift key (function key 1). Scroll through the list of calls can be made with the arrow keys (function keys 6 and 7). The automatic dialing of a number from the list can be done with the OK key (function key 8); the numbers in the call list can be deleted. If there are new entries in the list, a specific symbol is displayed.
- in some situations, the network operator may provide the user with additional features, such as immediate or conditioned call forwarding, call waiting, anonymous call, etc. These functions can be activated by the user through key combinations specified by the operator. For easier activation/deactivation of these functions the key combinations mentioned above can be associated with directory (phonebook) keys. Thus it is necessary to use a single key instead of a combination of 5-6 keys. The operations for assigning functions to the directory keys are described in the user manual. These operations are performed with the programming key and the Shift key (function keys 5 and 1).

The use of the function keys is similar to that of the keys on the Siemens Euroset 5010. Some of the most important differences have been mentioned above. Adjusting the volume of the receiver and the level of the ringer is done with the volume keys, and the activation of the hands free functionality is done with the "speaker" key.

Figure 2.10: Gigaset AP180 cordless telephone.

Gigaset AP180 telephone

It is a cordless phone with displays and multipple features [17]. The receiver displays the called number, the caller's number (if CLIP functionality is active), time, date, battery charge level. The menu control can be done with the display keys (menu entry, selection of options, exit from the menu), and scrolling through the menus is done with the control key (see Fig. 2.10). There is a separate key for the phonebook, i.e. saving a new number, respectively selecting a number. The selection is done together with the control key. There is a separate key for messages and access to the call list. The scroll in this list is also done with the control key, which also allows the level of the receiver and the ringer to be changed.

The lifting of the receiver respectively the dialing of the number entered and displayed on the screen is done using the call key, and placing the receiver in the fork is done with the call end key. There is also a RECALL key and a MUTE key. The phone allows a number of additional features such as setting an alarm, setting the date and time, setting the language in which the information is displayed, changing dialing mode (tone/pulse), setting the FLASH pulse duration, setting the pause duration after accessing the line, etc. - see the phone manual for using these features.

The receiver can be placed in the base fork, when the battery begins to charge and the receiver is registered at the base, if it is not yet registered. The base has a button with which it is possible to register a receiver that is not placed on the fork (press the button until the registering takes place) or it can send a localization signal to the receiver, i.e. the receiver starts to ring and so it can be located.

2.7 Application

1. The following operations are performed with rotary dial phones:

   (a) The devices are opened and the major functional blocks are identified as well as the interconnection of these blocks.

   (b) The mode of operation of the hook switch and the way of connecting the telephone to the line are analyzed.

   (c) The mode of operation of the dialing disc and its connection to the telephone board are analyzed.

   (d) The mode of operation of the electromagnetic or electronic bell and the connection of the bell to the telephone board are analyzed.
   (e) It is analyzed the way of connecting the microphone and the headphone to the device main board.
   (f) The hybrid transformer and balancing impedance are identified.

2. Connect the rotary dial phones to the PABX exchange in the laboratory and check the operation.
3. The following operations are performed with electronic phones:
   (a) The devices are opened and the major functional blocks are identified as well as the interconnection of these blocks.
   (b) The mode of operation of the hook switch is analyzed.
   (c) The keypad of the dialing circuit is opened and the component elements are identified.
   (d) It is analyzed the way of connecting the microphone and the headphone to the device main board.
4. The following operations are executed with the programmable and cordless electronic phones:
   (a) The devices are connected to the PABX exchange in the laboratory.
   (b) Based on the user manuals, the parameters that can be set are identified and the corresponding settings are performed. The effect of these settings is checked.
   (c) A call involving a cordless phone is made. The receiver is removed from the base until the connection is broken or degraded. The indoor coverage radius is estimated.

Chapter 3

Telecommunication cables

3.1 Introduction

The chapter proposes the study of several theoretical and practical aspects related to telecommunication cables (cables made of twisted pairs) used in the telephone access network and in the local data networks. This chapter proposes the study of the mathematical modeling of the cable's transfer function, the cable being a circuit with distributed parameters. The theoretical determination and measurement of the attenuation characteristic of the pairs and the coupling characteristic between the neighboring pairs of the cable will be the main issues considered. The chapter also focuses on the internal structure of the cables made of twisted pairs, cables used in the subscriber loop, in the transport network between the exchanges, respectively in the local data networks and a number of important parameters are discussed. Finally, are consider some practical aspects related to the installation of the telecommunication cables made of twisted pairs.

3.2 Modeling of the telecommunication cables

Cables represent transmission media with distributed parameters, the diagram of a section of cable, with a certain length, being presented in Fig. 3.1 [18] [19] [20]. Such a transmission medium with a linear characteristic is characterized by a distributed resistance R, expressed in Ω/km, a distributed capacitance C, expressed in nF/km, a distributed inductance L, expressed in $\mu H/km$ and a distributed conductance G, expressed in $\mu S/km$. The unit of length of $1km$ is used in telephone access and transport networks due to the large lengths of cables in these networks. In other networks, such as Local Area Networks (LAN), the unit length can be $100m$ or even $1m$.

Some important observations regarding the distributed parameters of the cables, called also primary or line parameters, are the following: the primary parameters are frequency dependent - the distributed resistance and conductance increase with the frequency, the distributed capacitance is constant with the frequency, and the distributed inductance decreases with frequency; in the telephone band only the distributed resistance and capacitance matter; at frequencies higher than $30kHz$ the distributed inductance begins to intervene, and at

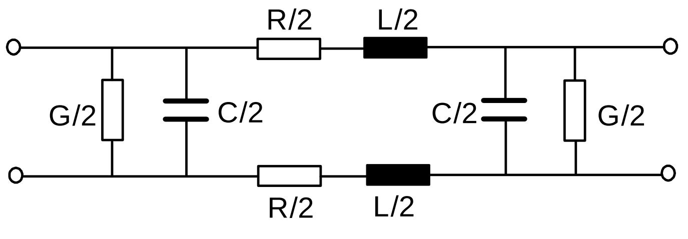

Figure 3.1: Equivalent diagram of a telecommunication cable section.

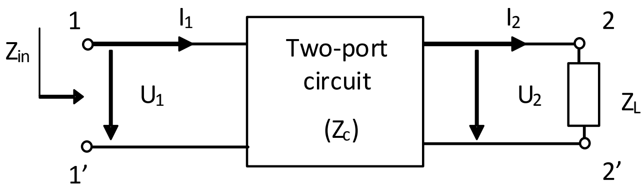

Figure 3.2: Impedance matching between a cable and the impedances connected at the cable ends.

frequencies higher than hundreds of kHz the distributed conductance also begins to matter.

Based on the primary parameters, other parameters characteristic to cables (secondary parameters) can be defined, parameters which are related to the cable transfer function and the impedance matching between the cable and external load impedance. Two very important parameters are the propagation index γ and the characteristic impedance Z_c, parameters given by the following relations [18]:

$$\gamma(\omega) = \alpha(\omega) + j\beta(\omega) = \sqrt{(R + j\omega L)(G + j\omega C)} \tag{3.1}$$

$$Z_c(\omega) = \sqrt{\frac{R + j\omega L}{G + j\omega C}} \tag{3.2}$$

where α is the cable attenuation constant, expressed in Np/km, and β is the phase constant, expressed in rad/km. From relation 3.1 results the double variation of the propagation constant and characteristic impedance with frequency, through the frequency parameter and through the variation of the primary parameters with frequency.

Note: the transformation from Np to dB can be done according to the relation: $1Np = 8.68dB$.

The importance of the characteristic impedance is the following: if at one end of a cable with characteristic impedance Z_c a load impedance with the value Z_c is connected, then the input impedance seen at the other end of the cable also will be Z_c, i.e. there is a "move" of the load impedance from one end of the cable to the other end of the cable, which is very important if an impedance matching is desired throughout the circuit path (see Fig. 3.2).

Based on the attenuation constant and the phase constant, frequency attenuation function $a(\omega)$ can be determined, as follows:

$$a(\omega) = e^{\alpha(\omega)l} e^{j\beta(\omega)l} = \frac{\overline{U_1}}{\overline{U_2}} \tag{3.3}$$

where l is the length of the cable. If a sinusoidal signal with angular frequency ω_s, amplitude A_i and phase φ_i is applied to the cable input, then at the cable output a sinusoidal signal of the same angular frequency will be obtained, with amplitude A_o and phase φ_o, given by the relations:

$$A_o = A_i e^{-\alpha(\omega_s)l}; \varphi_o = \varphi_i - \beta(\omega_s)l \tag{3.4}$$

A telecommunication cable can be modeled as a two-port circuit (see Fig. 3.2), the parameters of this two-port circuit being given by the secondary parameters γ and Z_c. For example, a two-port circuit can be characterized by the transmission parameters A, as follows [21]:

$$U_1 = A_{11}U_2 + A_{12}I_2;\ I_1 = A_{21}U_2 + A_{22}I_2 \tag{3.5}$$

$$\begin{pmatrix} U_1 \\ I_1 \end{pmatrix} = \begin{pmatrix} A \end{pmatrix} \begin{pmatrix} U_2 \\ I_2 \end{pmatrix}; \begin{pmatrix} A \end{pmatrix} = \begin{pmatrix} A_{11} & A_{12} \\ A_{21} & A_{22} \end{pmatrix} \tag{3.6}$$

If the cable is made up of several sections with different parameters then the global matrix (with global

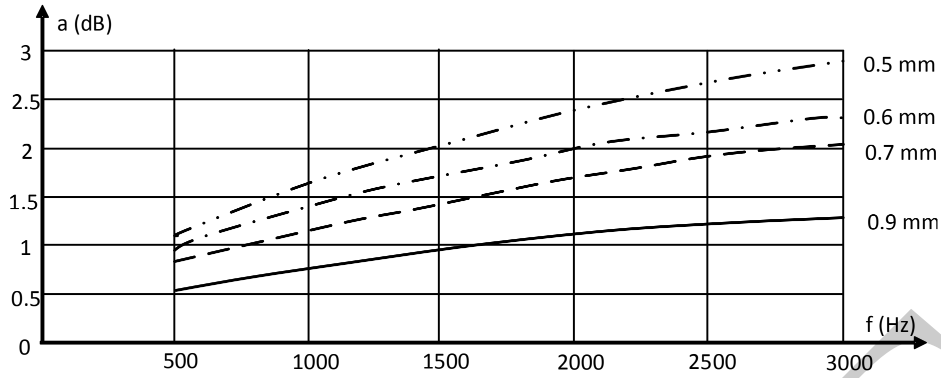

Figure 3.3: Variation of attenuation in the telephone band over a length of $1km$ of twisted pairs of wires with different diameters.

transmission parameters) that characterizes the whole cable is the product of the matrices with the transmission parameters of each component section:

$$(A) = \prod_{i=1}^{n}(A_i) \tag{3.7}$$

The transmission parameters can be determined according to the length of the cable l, the propagation constant α and the characteristic impedance Z_c as follows [1]:

$$(A_i) = \begin{pmatrix} cosh(\gamma_i l_i) & Z_{ci} sinh(\gamma_i l_i) \\ \frac{1}{Z_{ci}} sinh(\gamma_i l_i) & cosh(\gamma_i l_i) \end{pmatrix} \tag{3.8}$$

where i is the index of the cable section.

3.2.1 Aspects related to the attenuation transfer function of twisted pairs cables

The attenuation of the cable at a certain frequency can be calculated as $a(f) = V_i^2(f)/V_o^2(f)$, where V_i is the voltage applied to the cable input, and V_o is the voltage measured at the cable output. The general law of variation of the attenuation of twisted pairs with frequency is given by the relation: [18]:

$$a(f)_{dB} = \begin{cases} k_1 \cdot f^{1/2}; & f \leq 10kHz \\ k_2 \cdot f^{1/4}; & 10kHZ < f \leq 100kHz \\ k_3 \cdot f^{1/2}; & f > 100kHz \end{cases} \tag{3.9}$$

where the constants k_x depend on the geometry and length of the cable and the temperature.

In Fig. 3.3 it is presented the variation of the attenuation in the telephone band for a length of $1km$ of the pairs of twisted wires with different diameters [1].

Note: the characteristics of the twisted pairs cables in the subscriber loop are mainly determined by the distributed resistive (R/km) and capacitive (C/km) parameters.

Other empirical equations for calculating cable attenuation depending on the frequency and length (general and specific relations) at a temperature of $20°C$ are as follows [22]:

- general relation:

$$L(f)_{dB} = (a\sqrt{f} + b \cdot f)d \tag{3.10}$$

  where the distance d (cable length) is given in km, the frequency f is given in MHz, the parameter a depends on the diameter of the wires, and the parameter b depends on the insulation of the wires.

- BKMA cable for aerial telephone installations:

$$L(f)_{dB} = (2.1\sqrt{f} + 0.3f)d \tag{3.11}$$

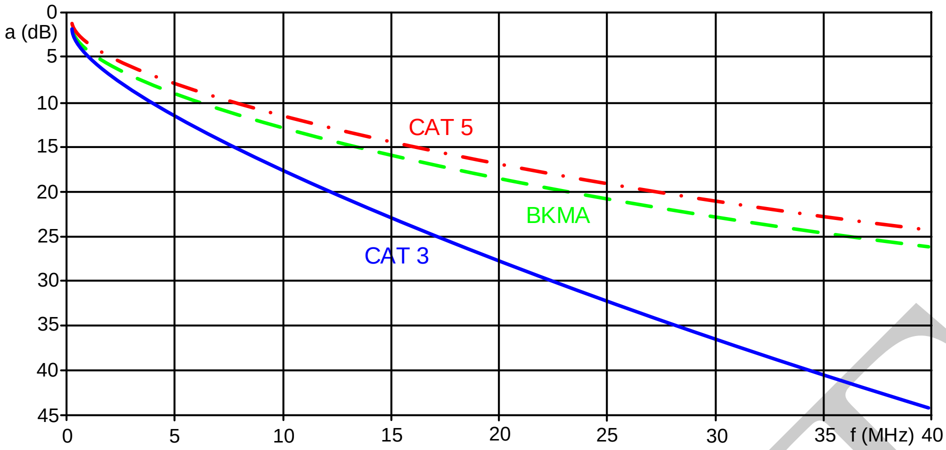

Figure 3.4: Variation of the attenuation with of BKMA, CAT.3 and CAT.5 cables at room temperature for a length of $180m$.

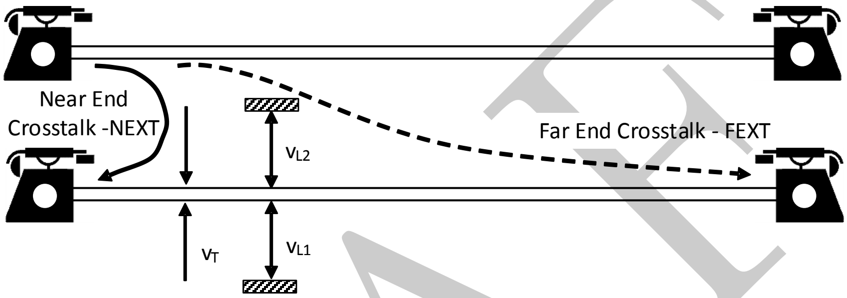

Figure 3.5: The ways the crosstalk phenomenon occurs.

- UTP CAT.3 cable, usually used in the subscriber loop, worst case function, $d = 100m$:

$$L(f)_{dB} = (2.32\sqrt{f} + 0.238f) \tag{3.12}$$

- UTP CAT.5 cable worst case function, $d = 100m$:

$$L(f)_{dB} = (1.967\sqrt{f} + 0.023f) \tag{3.13}$$

The variation of the attenuation with frequency of the BKMA, CAT.3 and CAT.5 cables for a length of $180m$ (i.e. "$600feet$") and a temperature of $20°C$ is depicted in Fig. 3.4.

3.3 Linear crosstalk noise

The linear crosstalk noise is the phenomenon by which a useful signal occurs in the wrong place. In telephony this noise signal can be intelligible and can lead to loss of confidentiality [1]. The single frequency noise is also generated by the crosstalk phenomenon and consists of unwanted signals of fixed frequencies induced in the telephone channel (e.g. signaling tones between the telephone terminal and the local exchange). This noise has a stationary character, a typical example being the network hum.

The crosstalk phenomenon occurs between wire pairs in the same cable in two fundamental ways: Near End Crosstalk (NEXT) and Far End Crosstalk (FEXT) - see Fig. 3.5.

The main causes that generate the crosstalk phenomenon on the subscriber loop are the parasitic capacitive couplings between the neighboring pairs and the imperfect balancing to earth of the circuits. Twisting of wires reduces significantly the effect of inductive couplings between neighboring pairs, but without completely eliminating this coupling.

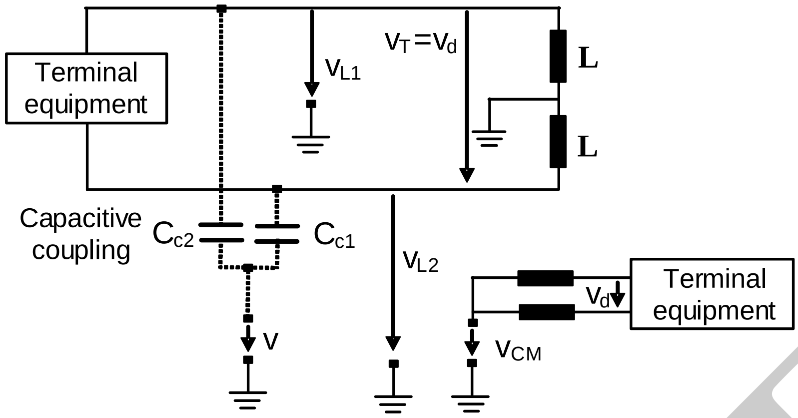

Figure 3.6: Longitudinal - transverse capacitive coupling.

On a cable made of twisted wires there are two types of voltages (see Fig. 3.5): a transverse (or differential) voltage, v_T, between the two wires, voltage representing the useful signal, and longitudinal voltages, v_{L1} and v_{L2}, between each wire and ground (the common point). Longitudinal voltages are usually parasitic signals induced by capacitive couplings with different signal sources. Ideally, the difference between the longitudinal voltages should be zero, in which case no disturbing transverse voltage is generated. [1] [4].

In Fig. 3.6 it is presented the situation in which a signal source v (e.g. the mains power supply) is coupled through the parasitic capacitance C_{c1} and C_{c2} with the two wires of the twisted pair, which causes the occurrence of two longitudinal voltages v_{L1} and v_{L2} between wires and the common ground, voltages that can be written as:

$$v_{L1} = v_{MC} + \frac{v_d}{2};\ v_{L2} = v_{MC} - \frac{v_d}{2}$$
$$v_{MC} = \frac{v_{L1} + v_{L2}}{2};\ v_d = v_{L1} - v_{L2} \tag{3.14}$$

where v_{MC} is the common mode voltage, and v_d is the generated differential disturbing voltage.

The situation described in Fig. 3.6 is called longitudinal-transverse coupling, and the ratio in decibels between the longitudinal (common mode) voltage and the differential (transverse) voltage is called the Common Mode Rejection Ratio, $CMMR$ [1]:

$$RRMC = 20lg\frac{v_{MC}}{v_d}[dB] \tag{3.15}$$

If the capacitances C_{c1} and C_{c2} are equal and the inductances L of the telephone power feeding bridge are equal, then the longitudinal voltages v_{L1} and v_{L2} are equal and the disruptive transverse (or differential) voltage is zero.

The situation presented in Fig. 3.6 is characteristic to the coupling between the mains power supply and the pairs of twisted wires, in which case network hum is induced in the telephone channel.

Fig. 3.7 shows the situation corresponding to the capacitive coupling between the neighboring pairs of twisted wires, coupling that generates the phenomenon of crosstalk and induces signaling tones and parasitic voice signals in the affected pair.

In the case of the situation presented in Fig. 3.7 in the first circuit, the differential signal v_1, a useful signal, is transmitted, and in this circuit there is an imbalance generated by the parasitic capacitance C_d. In the absence of this capacitance the longitudinal voltages v_{L1} and v_{L2} are identical, but in the presence of the capacitance C_d these voltages are no longer identical. A transverse-longitudinal coupling appears between the signal source that generates v_1 and the wires of circuit 1. Due to the parasitic capacitances C_{c1} and C_{c2} a longitudinal-transverse coupling appears between circuit 1 and circuit 2, that is, between the signal sources represented by v_{L1} and v_{L2} and the wires in circuit 2. In circuit 2 is generated a disruptive transverse (differential) voltage v_2, even if the

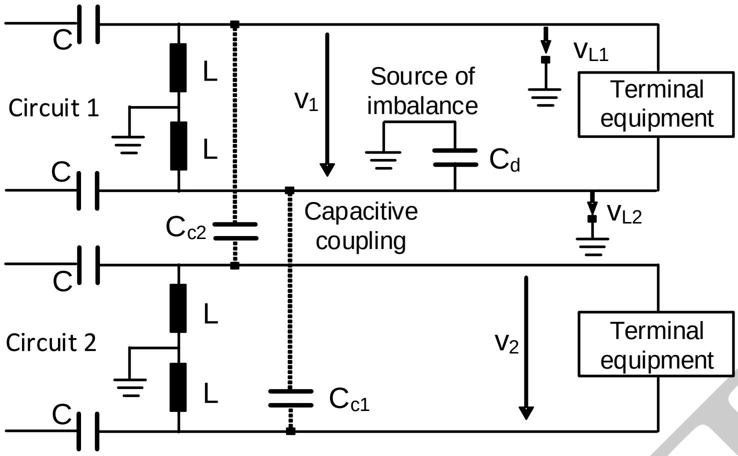

Figure 3.7: Transverse - longitudinal coupling and longitudinal - transverse coupling.

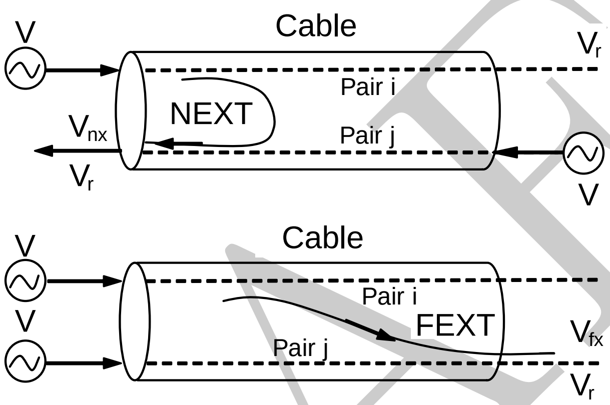

Figure 3.8: NEXT and FEXT between the pairs of a cable.

C_{c1} and C_{c2} capacitances are identical. The transverse-longitudinal coupling is characterized by the crosstalk loss [1]:

$$a_{cross} = 20lg\frac{v_1}{v_2}[dB] \tag{3.16}$$

Other causes which lead to the phenomenon of crosstalk: inductive couplings between pairs (coupling reduced by twisting), parasitic couplings between electronic sub-assemblies, capacitive and inductive couplings between the tracks of the printed circuit boards and between electronic components, crosstalk induced by PCM coding modules, excessive gain of the voice repeaters used on cable pairs.

3.3.1 Mathematical modeling of the crosstalk phenomenon

The two types of crosstalk phenomena (NEXT - Near End Crosstalk and FEXT - Far End Crosstalk) that occur between the pairs of a cable (see Fig. 3.8) can be characterized by frequency dependent crosstalk attenuations or losses, $A_{nx}(f)$ and $A_{fx}(f)$. The mentioned parameters can be computed according to the following relations: [22]:

$$A_{nx}(f) = \frac{V^2(f)}{V_{nx}^2}; A_{fx}(f) = \frac{V^2(f)}{V_{fx}^2} \tag{3.17}$$

If several pairs pair interfere with pair j, the sum of NEXT and FEXT interference signals is given by:

$$V_{nx}^2 = \sum_{i \neq j} V_{nx,j}^2; V_{fx}^2 = \sum_{i \neq j} V_{fx,j}^2 \tag{3.18}$$

If it is considered a cable with k pairs of which n pairs interfere (with a given pair), at high frequencies ($> 500kHz$), the crosstalk losses expressed in dB have the expressions [22] [23]:

$$\begin{aligned} A_{nx,dB}(f) &= K_{nx} - 15 \cdot ln(f) + 6 \cdot ln(m/n) \\ A_{el-fx,dB}(f) &= K_{fx} - 20 \cdot ln(f) - 10 \cdot ln(d) + 6 \cdot ln(m/n) \\ m &= k - 1 \end{aligned} \tag{3.19}$$

where K_{nx} and K_{fx} are constants of crosstalk coupling and depend on the parameters of the cable, and m is the possible number of sources that generate interference. The frequency is expressed in MHz, and the distance (cable length) in km. The relation changes for different types of cable as follows:

- BKMA cable for aerial installations:

$$\begin{aligned} A_{nx,dB}(f) &= 40.3 - 15 \cdot ln(f) + 6 \cdot ln(m/n) \\ A_{el-fx,dB}(f) &= 35.6 - 20 \cdot ln(f) - 10 \cdot ln(d) + 6 \cdot ln(m/n) \end{aligned} \tag{3.20}$$

- UTP CAT.3 cable, worst case function, $d = 100m$:

$$\begin{aligned} A_{nx,dB}(f) &= 41.3 - 15 \cdot ln(f) \\ A_{el-fx,dB}(f) &= 51 - 20 \cdot lg(f/0.772) \end{aligned} \tag{3.21}$$

- UTP CAT.5 cable, worst case function, $d = 100m$:

$$\begin{aligned} A_{nx,dB}(f) &= 62.3 - 15 \cdot ln(f) \\ A_{el-fx,dB}(f) &= 63 - 20 \cdot lg(f/0.772) \end{aligned} \tag{3.22}$$

Note: the level of the far end crosstalk signal induced in a circuit (i.e. a pair) depends both on the FEXT loss and on the loss of the signal that generates the FEXT, loss suffered on the circuit on which this signal is transmitted. For this reason, the EL-FEXT (Equal Level Far End Crosstalk) parameter is defined, which represents the difference between FEXT loss and the loss inserted by the circuit on which the crosstalk generator signal is transmitted. The previous mathematical relations, (3.20), (3.21), (3.22), give the EL-FEXT parameter (A_{el-fx}), which must be considered in practice for evaluating the effect of the far end crosstalk phenomenon [22] [24].

Note: theoretically, the NEXT loss and implicitly the level of the disturbing signal induced by NEXT does not depend on the length of the cable, the coupling being located at the near ends of the two channels. In reality, however, the NEXT coupling is not located in a single point and occurs on a certain length of cable. For this reason, the ACR (Attenuation to Crosstalk Ratio) parameter is defined, which represents the difference between the NEXT loss and the loss of a finite-length cable section, usually $100m$ [25].

In Fig. 3.9 it is presented the variation with the frequency of the near end crosstalk loss for the three types of cables mentioned above, and in Fig. 3.10 it is presented the variation with the frequency of the EL-FEXT parameter for the cable types considered, for cable sections with a length of $180m$ (i.e. $600feet$).

Other relationships for pair-pair crosstalk according to the spectral density of the signals that generate interference (crosstalk) are as follows [26]:

$$\begin{aligned} Next(f) &= S(f) \cdot K_N \cdot f^{1.5} \\ Fext(f) &= S(f) \cdot K_F \cdot f^2 \cdot l \cdot |H(f)|^2 \end{aligned} \tag{3.23}$$

where f is the frequency in Hz, l is the length in feet, K_N is a constant equal to $8.536 \cdot 10^{-15}$, K_F is a constant

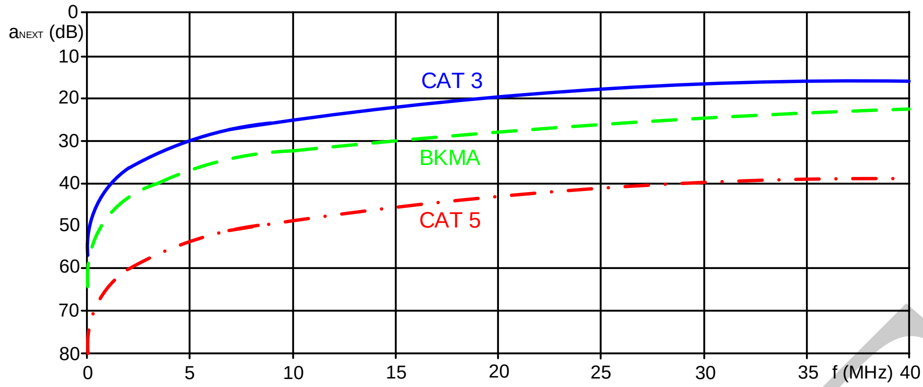

Figure 3.9: Variation of the NEXT loss with the frequency for BKMA, CAT.3 and CAT.5 cables.

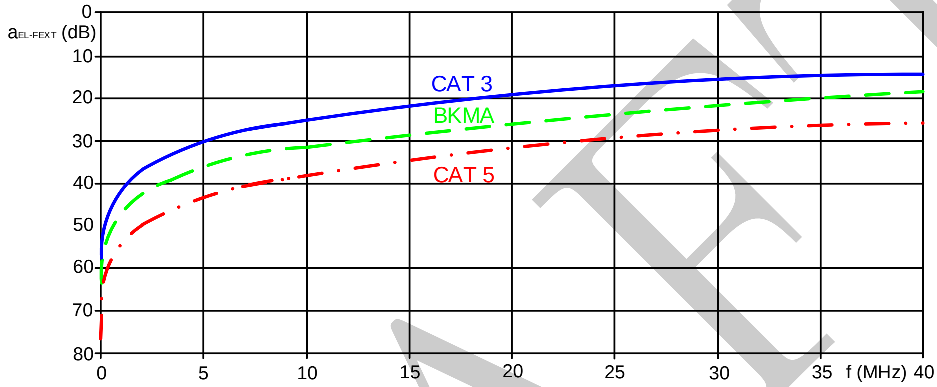

Figure 3.10: Variation of the EL-FEXT parameter with the frequency for BKMA, CAT.3 and CAT.5 cables for a length of $180m$.

equal to $7.74 \cdot 10^{-21}$, $S(f)$ is the spectral density of the interfering signal, and $H(f)$ is the frequency transfer function of the channel.

If there are n identical disruptive (interfering) sources, the summation of the effects can be done as follows:

$$Next\,(f,n) = S\,(f) \cdot K_N \cdot f^{1.5} \cdot n^{0.6}$$
$$Fext\,(f,n) = S\,(f) \cdot K_F \cdot f^2 \cdot l \cdot |H(f)|^2 \cdot n^{0.6} \qquad (3.24)$$

The 0.6 value of the exponent is determined empirically and is especially valid for cables use in USA. Cables in Europe have an exponent of 0.7 or 0.8. The formula has higher accuracy for a large number of interfering signals.

3.4 Internal structure and parameters of telecommunication cables

3.4.1 Telecommunication cables used in the subscriber loop

The telecommunication cables used in the subscriber loops as well as for connecting private PBXs or other equipment (e.g. concentrators) to the public telephone network are made up of several twisted pairs (between tens and thousands of pairs) and are intended to operate at a maximum voltage of $150V$ [27] [28].

From the point of view of the positioning/installation mode, the telecommunication cables used in the subscriber loop can be grouped as follows:

- aerial/overhead telecommunication cables mounted on walls or on poles;
- underground telecommunication cables, positioned directly in the ground or in ducts.

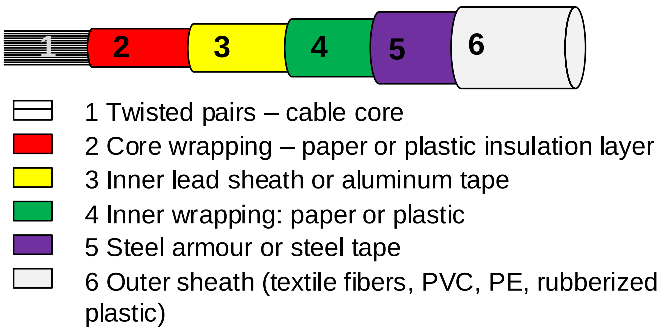

Figure 3.11: General structure of telecommunication cables.

The general structure of such a cable is shown in Fig. 3.11.

The cable is made up of a number of twisted pairs, which can be four pairs in the case of cables used in local data networks, tens of pairs in the case of overhead (aerial) lines or hundreds or thousands of pairs in the case of buried cables. The wires are insulated with paper (in the old cable standards; the paper is a special one with dielectric properties conforming to the requirements of the telephone networks) or with a special plastic (with appropriate dielectric properties). The pairs form the core of the cable and the interstices between the pairs can be filled with air (air core cables) or with a non-hygroscopic material (e.g. gel) (filled core cables) that prevents the infiltration of moisture inside the cable. The pairs are also twisted into concentric layers according to certain rules that depend on the number of pairs and the structure of the cable, to reduce crosstalk [29] [30].

The pairs in the core of the cable are wrapped with several protective layers (see Fig. 3.11) which provides protection against moisture (plastic in new cables or paper layers in old cables), sheaths or strips/foils (lead sheath in old cables, aluminum strips/foil in new cables) that provide electromagnetic shielding and can also provide protection against moisture in certain situations (e.g. if using metallic sheath), steel armor (in the case of old cable) or a steel strip (in the case of new cables) to ensure physical resistance. Over the steel sheath/strip a protective coating is applied, coating made of plastic, rubberized plastic or bitumen-coated textiles in the case of older cables.

The main parameters of the cables used in the subscriber loop are:

- the electrical resistance measured on the physical circuit (pair) in DC at a temperature of $20°C$. It has values between $57\Omega/km$ $(2 \times 0.9mm)$ and $284\Omega/km$ $(2 \times 0.4mm)$.

- the insulation resistance of each wire to all other wires connected together to the metal sheath $(5000M\Omega/km)$.

- the working capacity measured at $800Hz$ between one pair and the other pairs connected together to the metal sheath $(40 - 60nF/km)$.

- the attenuation (at $800Hz$) computed according to the relation:

$$\alpha = \sqrt{\frac{\omega RC}{2}}[Np/km] \tag{3.25}$$

  where R is the electrical resistance of the pair of wires in Ω/km, C is the nominal or maximum capacitance of the pair in F/km, and ω is the angular frequency in rad/s.

- the breakdown voltage: finite length cable must support these voltages without having breakdown for a minute:

  – a voltage of $1500V$ applied between the metal sheath connected to the ground and all wires connected together;

  – a voltage of $500V$ applied between all the first wires of each pair connected together and all the other wires (the second wire of the pair) connected together.

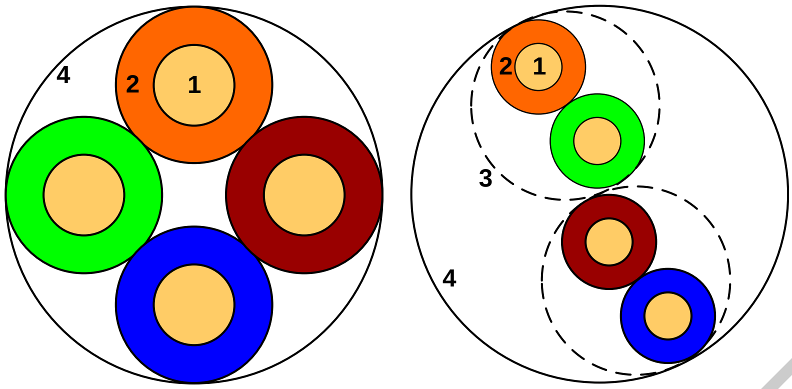

Figure 3.12: Transmission circuits organized in quads.

3.4.2 Telecommunication cables used in the transport network

The transport network provides connections between the local and transit switching nodes of the telephone network [27] [28]. In modern telephone systems these connections are made on optical fibers and only in certain cases are used telecommunication cables, e.g. when connecting PCM concentrators/multiplexers to local exchanges, connecting small capacity exchanges to transit exchanges, etc. These cables are designed to operate at a maximum voltage of $250V$ and, unlike the cables in the subscriber loop, a duplex communication channel is made of two pairs of wires, i.e. 4-wire duplex transmission. The four wires form a quad and there are two main types of quads (see Fig. 3.12):

- double pair or DM (Diesel Horst - Martin) quad - two twisted pairs are formed with a certain twisting step, then the two pairs are twisted together with another step.
- star quad - the 4 wires are twisted together using a certain step.

The pairs of the quad, respectively the quads, can be covered with a plastic film and possibly with an aluminum foil, the latter ensuring the shielding of the pairs respectively of the quad.

In digital networks such cables can be used for transmissions of primary PCM frames or PDH frames.

The main parameters of the cables used in the transport network are:

- the electrical resistance of a pair measured in DC at a temperature of $20°C$, must have values between $57\Omega/km$ $(2 \times 0.9mm)$ and $32\Omega/km$ $(2 \times 1.2mm)$.
- the insulation resistance measured between a wire and all other wires connected together to the metal sheath of the cable and must be at least $10000M\Omega/km$.
- the nominal capacity, measured at the frequency of $800Hz$ between the wires of one pair and all other wires connected together to the cable sheath; should be at most $26.5nF/km$ for high frequency circuits and $38.5nF/km$ for low frequency circuits.
- the attenuation constant which is the real part of the propagation constant and must have the following values:
  - $75mNp/km$ la $f = 800Hz$ for low frequency circuits;
  - $240mNp/km$ la $f = 120kHz$ for high frequency circuits;
  - $350mNp/km$ la $f = 240kHz$ for high frequency circuits.
- the test voltage: each finite length of cable must withstand, without breakdown, for one minute the following voltages:

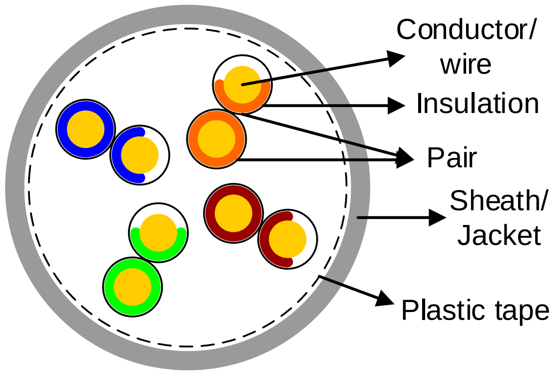

Figure 3.13: 38/5000 The structure of a 4-pair UTP cable.

– a voltage of $2000V$ ($50Hz$) applied between the metal sheath connected to the ground and all the wires of the circuits connected together;

– a voltage of $500V$ ($50Hz$) applied between any of the wires of a quad (one wire) and all other wires of the quad connected together.

3.4.3 Other aspects related to the structure of telecommunication cables

Depending on how the electromagnetic shielding of the twisted pairs of the cable core is made, shielding that reduces the crosstalk between the pairs and the external interferences, the following types of cables can be identified [31] [32]:

- UTP (Unshielded Twisted Pair) cable: there is no pair shielding nor cable shielding against external interference. It is the cable with the lowest price, but it is also the easiest to install, being more flexible than the shielded cables. UTP cables are typically used for indoor cabling, an environment in which electromagnetic interference is usually lower. It is widely used for the implementation of local data networks (Local Area Network - LAN). Fig. 3.13 shows the internal structure of a 4-pair UTP cable (used in data networks), and Fig. 3.14 shows a picture with the internal structure of such a cable.

- S-UTP (Screened Unshielded Twisted Pair) cable: also called Foiled Twisted Pair (FTP) cable, the pairs are not shielded, but there is a screen of the cable (metal foil wrapped over the core of the cable) that protects the pairs against external interference. The structure is similar to the cables used in the subscriber loop. Fig. 3.15 shows the internal structure of a 4-pair S-UTP cable (used in data networks), and Fig. 3.16 shows an image with the internal structure of such a cable.

- STP cable (Shielded Twisted Pair): the cable pairs are shielded (each pair is wrapped with a metal foil), but there is no cable shield. This type of cable substantially reduces the crosstalk between the pairs and at the same time protects the pairs against external interference. The cable is more expensive and at the same time more difficult to install due to the low flexibility of the internal structure. This type of cable is worth using in high speed data transmission systems, where a high noise signal ratio is required to ensure high transmission capabilities. Fig. 3.17 shows the internal structure of a 4-pair STP cable.

- S-STP (Screened Shielded Twisted Pair) cable: also called S-FTP (Screened Foiled Twisted Pair), the individual pairs are shielded by wrapping a metal foil around each pair and the cable also has a separate shield. This cable ensures a significant reduction of the crosstalk between pairs and at the same time greatly reduces the effect of external interference. The cable is expensive, difficult to install and is intended for high-speed data transmission systems. Fig. 3.18 shows the internal structure of a 4-pair S-STP cable and Fig. 3.19 shows a picture with the internal structure of such a cable.

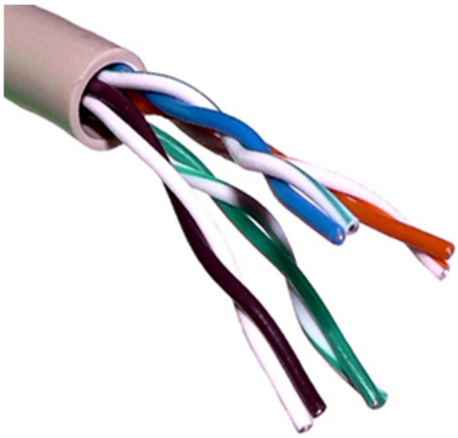

Figure 3.14: Twisted pairs of a UTP cable.

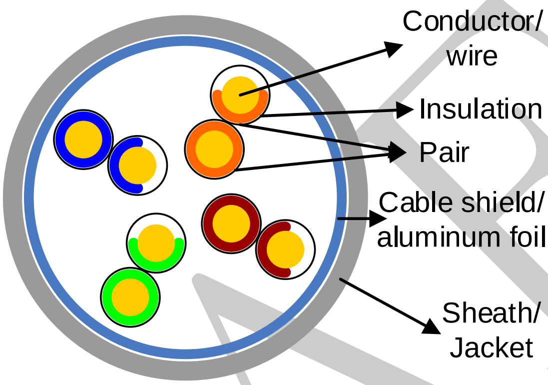

Figure 3.15: The structure of a 4-pair S-UTP cable.

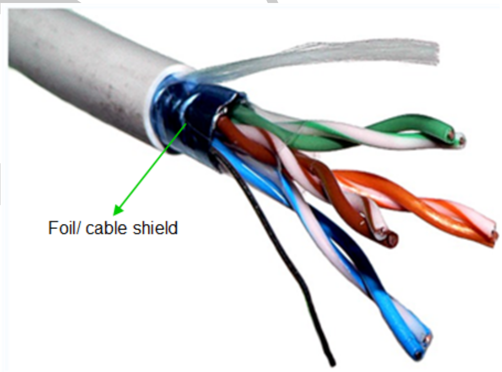

Figure 3.16: Twisted pairs of S-UTP cable.

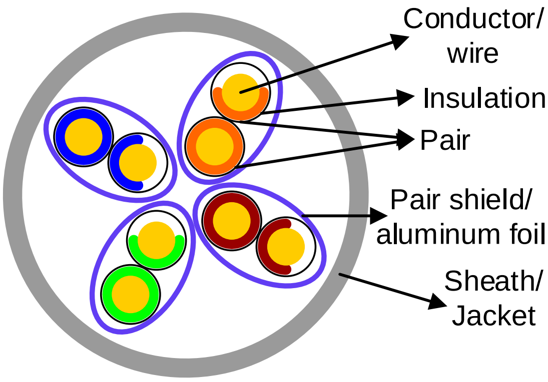

Figure 3.17: The structure of a 4 pair STP cable.

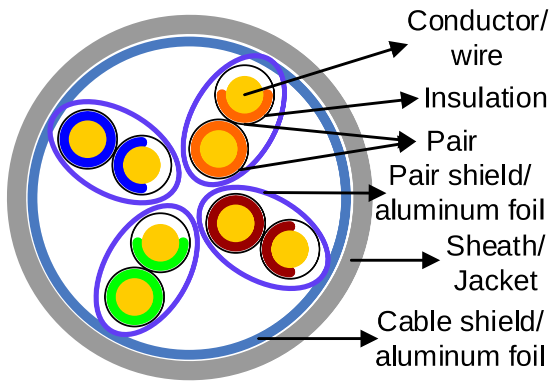

Figure 3.18: Structure of a 4-pair S-STP cable.

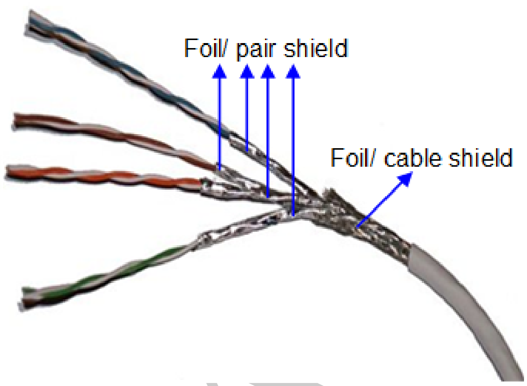

Figure 3.19: Twisted pairs of S-STP cable.

Other issues to be considered refers to the installation of cables (i.e. wiring of access networks or local telephone or data networks) [33] [34]. An ideally twisted pair, meaning all turns are identical and the pair shows perfect symmetry, performs better in terms of bandwidth and attenuation distortions than a coaxial cable. Unfortunately, such a cable can not be manufactured, being impossible to ensure a perfect symmetry of the twisted pair. The symmetry of the pair is also affected by the installation process. Due to the bending of the cable in the installation process, certain pairs loosen (they change the geometry) and alter the symmetry of the cable. The process is illustrated in Fig. 3.20 [30] [34]. The figure also shows the asymmetry of the twisted pairs, that is the variable distance between the wires of the twisted pairs. A solution to the problems mentioned above is the so-called bonded pair cables. In this case the two wires of the pair have bonded/joined insulation (see Fig. 3.21), which greatly improves the symmetry of the twisted pair, i.e. the wires in the pair are equally spaced in each turn. In addition, as can be seen in Fig. 3.21, if the cable is bent in the installation process the wires do not loosen and do not alter the symmetry of the cable [34]. The disadvantage of such a cable is the higher price compared to the "simple" twisted cables.

Regarding the internal structure of a cable, the concept of category of a cable should be mentioned [29]

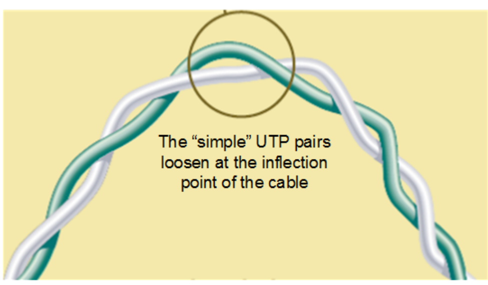

Figure 3.20: "Simple" twisted pairs.

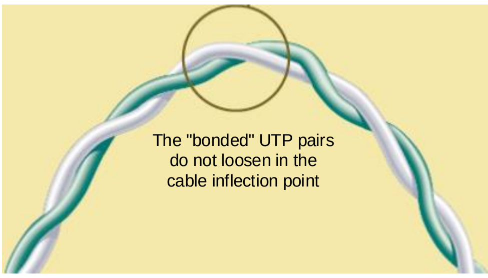

Figure 3.21: "Bonded" twisted pairs.

[30] [35]. This concept refers to the performance of a cable in terms of bandwidth, shielding against external interference and reduction of interference between pairs. The category of a cable depends on the internal symmetry (that is, the accuracy of the twisting step) and the shielding used (that is, the shielding of the whole cable and/or the shielding of the individual pairs). In telephone networks can be used category 3 cable (CAT.3), which also allows data transmission at medium rates (several Mbps). Only for voice transmissions can be used even lower category cable. For local data networks (data rates of 100Mbps or 1Gpbs) category 5 cable (CAT.5 or CAT.5E - improved category 5) or category 6 cable (CAT.6 and CAT.6E) can be used. Category 7 cables (CAT.7 and CAT.7A) are S-STP cables with pair and cable shielding and are used in very high data rate (10Gbps) local data networks.

Another aspect, closely related to the process of installing cables in buildings, concerns the insulating material used in the insulation of the pairs and in the plastic coats/sheaths of the cables [36] [37] [38]. The plastic insulation (usually polyethylene) of most cables is flammable and generates a lot of heat and toxic smoke upon ignition. All these are serious problems in the event of a fire. There are, of course, fire resistant plastic materials that can be used to insulate cables (e.g. flame retardant polyethylene, polyvinyl chloride, ethylene fluoride propylene, etc.), but cables made with these insulation are more expensive. In an indoor wired network, two types of cables can be identified, namely (see Fig. 3.22):

- vertical cables or "Riser" cables: these are the cables that connect the floors of a building, i.e. the connection panels on different floors. The volume of these cables is lower and thus have a small effect in the event of a fire, although they can be a way of spreading fire between floors. As a rule, a less fireproof insulation of these cables is allowed.

- horizontal cables or "Plenum" cables: these are the cables that connect the terminals to the connection panels. The volume of these cables is large or even very large (if there are a large number of terminals on the floor of the building) and thus have a great effect in the event of a fire. These cables can be fire propagating pathways and in addition can maintain fire and produce a high volume of toxic smoke. Usually these cables must have a better quality fireproof insulation.

The selection of the type of cable used in indoor wiring (more precisely the type of cable insulation) must take into account the mentioned aspects, aspects as important as the electrical characteristics of the cables.

Note: connection panels from different floors are not necessarily simple panels with connectors but may include switches, repeaters, multiplexers, hubs etc.

3.5 Application

1. The equivalent diagram of a cable section is analyzed. Analyze the variation with the frequency of the primary parameters in some concrete cases (tables with the variation of the primary parameters with frequency are available in the laboratory).

2. Using an RLC-meter, the primary parameters of a CAT.5 UTP cable segment of $305m$ in length are measured. Compare the measured values with the theoretical values. Multiple CAT.5 UTP cable segments

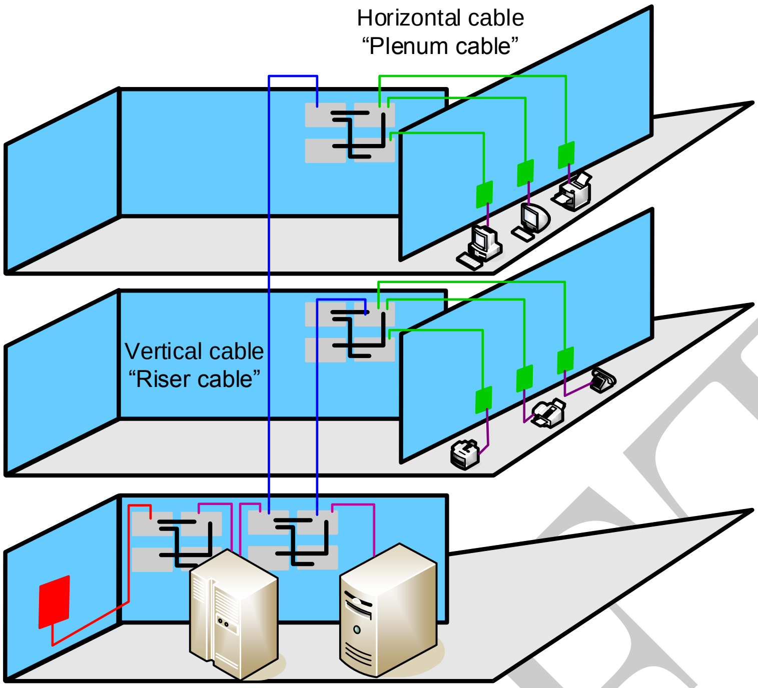

Figure 3.22: Types of cables used in wiring inside a building.

are concatenated (2-3 segments) and the primary parameters are measured. Compare the measured values with the theoretical values.

3. The importance of the characteristic impedance is identified. The mode of calculation of the characteristic impedance is analyzed in the whole frequency band, respectively in the telephone band. Analyze the variation of the characteristic impedance with frequency in some particular cases.

4. Analyze the calculation of the attenuation constant and the phase constant in the entire frequency band, respectively in the telephone band. Analyze the variation with the frequency of these parameters in some particular cases.

5. The empirical attenuation calculation formulas for different types of cable are analyzed. The attenuation of a 305m long CAT.5 UTP cable segment is measured. In order to ensure the symmetry of the transmission medium, the test signal generator and the measuring device (e.g. oscilloscope or spectral analyzer) will be separated from the cable segment by transformers; the impedance matching (at least partially) at the two ends of the cable will be ensured. Compare the measured values with the theoretical values. Repeat the experiment described for a longer segment of CAT.5 UTP cable (2-3 segments of 305m are concatenated).

6. The process of generating the crosstalk between the pairs of a cable is analyzed and the next experiment, described in Fig. 3.23 is performed:

   - the Tf_1A phone connects to a PBX exchange on a pair of a CAT.5 UTP cable segments.
   - the Tf_2A phone connects to the PBX exchange on another pair of a CAT.5 UTP cable segments. This phone will be a rotary dial phone, to which it is easier to connect a measuring instrument/oscilloscope in the reception circuit.
   - phones Tf_1B and Tf_2B connect directly to the PBX ports. The Tf_2B phone will be a rotary dial phone for the reasons stated in the previous point.
   - phones Tf_3A and Tf_3B connect directly, without going through the PBX, using another pair of the CAT.5 UTP cable. These phones will be rotary dial phones that do not require power for the receiving circuit and can be easily connected to a measuring instrument/oscilloscope.

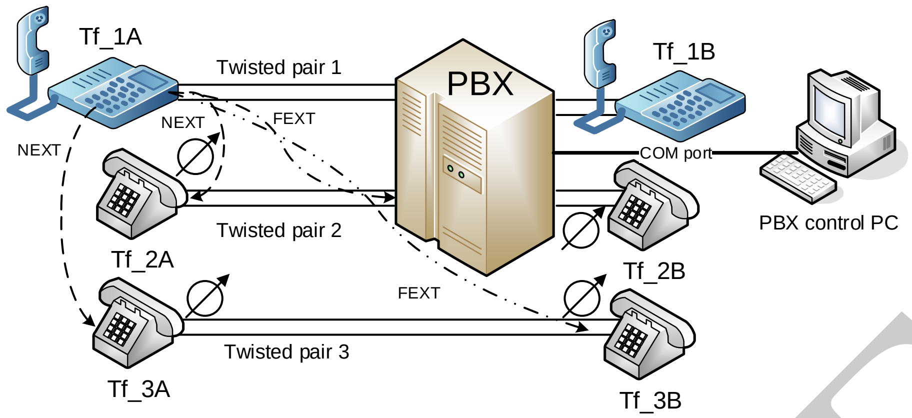

Figure 3.23: Experimental measurement of the crosstalk generated by the unbalance of the twisted pairs to ground.

- the PBX exchange can be connected on a serial communication port (COM port) to a PC computer, from which the PBX can be controlled/configured. If this connection is made the PBX and the computer will have a common ground signal.

In the conditions described in the figure mentioned above, the impedance matching is ensured and the transmission channels unbalance to ground due to the measuring instruments used is avoided. The experiment has two phases, namely:

- the PBX exchange is disconnected from the computer. A telephone connection is made between phones Tf_1A and Tf_1B. The Tf_1A phone will generate the source signal for the crosstalk: voice signal or other signal applied to the microphone or DTMF signaling tones. A telephone connection is made between phones Tf_2A and Tf_2B. The amplitude of the NEXT signals is measured in the reception circuits of the Tf_2A and Tf_3A phones. The amplitude of the FEXT signals is measured in the reception circuits of the Tf_2B and Tf_3B phones.
- the PBX exchange is connected to the computer and the two equipments will have a common ground, which will lead to unbalance of the pairs to ground. The amplitude of the NEXT signals on the Tf_2A and Tf_3A phones is measured and the amplitude of the FEXT signals is measured on the Tf_2B and Tf_3B phones. Compare the measured values with those obtained at the previous point.

7. The empirical calculation formulas of the crosstalk loss for different types of cable are analyzed. The crosstalk (NEXT and FEXT) loss between the pairs of a 305m long CAT.5 UTP cable segment is measured. The experiment, described in Fig. 3.24, consists of applying a test signal on one pair and measuring the signal level at the end of another pair. In order to ensure the symmetry of the transmission medium, the test signal generator and the measuring device (e.g. oscilloscope or spectral analyzer) will be separated from the cable segments by transformers; the impedance matching (at least partially) at the ends of the pairs involved in the test will be ensured. Compare the measured values with the theoretical values. Repeat the experiment described for a longer segment of CAT.5 UTP cable (2-3 segments of 305m are concatenated).

8. The internal structure of some cables used in telephone networks as well as of a CAT.5 UTP cable is analyzed. The importance of the parameters that characterize the telecommunication cables is identified and the way of defining these parameters is analyzed.

9. Analyze the (indoor) wiring of a multi-storey building. Identify the challenges raised by this process and propose possible solutions.

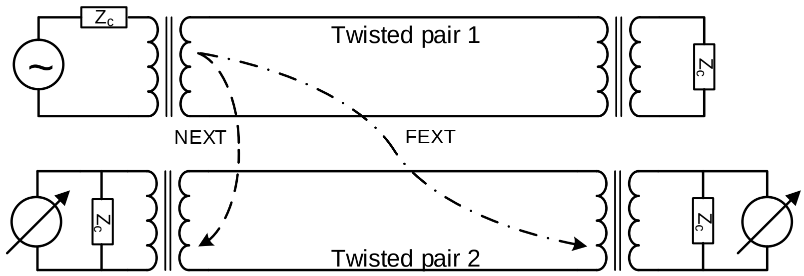

Figure 3.24: Experimental measurement of crosstalk loss between twisted pairs.

3.5.1 Exercises

1. The primary parameters of a cable at $1591.549Hz$ ($\omega = 10000rad/s$) frequency are $68\Omega/km$ and $68nF/km$. What load impedance must be connected to the cable end in order to have at the opposite end an input impedance equal to the load impedance? Does this impedance depends on the cable length?

2. The primary parameters of a cable at $1591.549Hz$ ($\omega = 10000rad/s$) frequency are $100\Omega/km$ and $40nF/km$. If a sinusoidal signal with amplitude 1V and phase 0 rad is applied at the input of the cable with length $2400m$, calculate the amplitude and phase of the signal obtained at the output of the cable.

3. The transmission parameters of a cable segment are: $A_{11} = A_{22} = 2$, $A_{12} = 3$, $A_{21} = 2$. If the load impedance connected to the cable output is set to $Z_s = 2 - j3$, calculate the cable input impedance.

4. It is considered a BKMA cable with the length of $3km$ which has 50 pairs of which 10 pairs interfere (with a given pair). Calculate the NEXT loss at frequencies: $4kHz$, $80kHz$, $120kHz$ and $240kHz$. Calculate the EL-FEXT parameter at the above mentioned frequencies.

5. It is considered a 25-pair BKMA cable segment with a length of $400m$ on which an ADSL data signal with a power spectral density of $-40dBm/Hz$ is transmitted. Calculate the NEXT and FEXT crosstalk signal at frequencies $500kHz$, $1MHz$, $1.5MHz$ and $2MHz$, if 4 pairs, 12 pairs, respectively all pairs interfere (with a given pair). Redo the calculations for a cable length of $1200m$. Analyze the results obtained. Calculate the signal/crosstalk ratio (separately for NEXT and FEXT) for the cable lengths and frequencies considered.

3.6 Annex

In Fig. 3.25 it is shown the patch panel to which 4 segments of CAT.5 UTP cable are connected in a certain way. These cable segments are used for measuring the parameters of the cables (attenuation, crosstalk loss, primary parameters, etc.) as well as for emulating subscriber loops. In order to be able to connect in parallel with the twisted pairs terminal equipment, signal generators and different measuring instruments and to be able to concatenate the segments, each end of the cable is connected in parallel to three sockets, as shown in Fig 3.25.

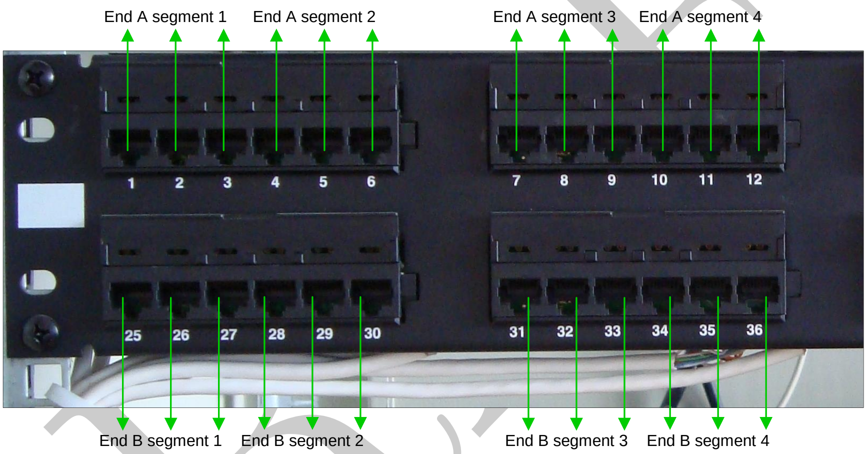

Figure 3.25: Connection panel for test cables.

Chapter 4

Wiring a telephone network

4.1 Introduction

The chapter proposes to present some basic aspects related to the wiring of a telephone network, but without making an exhaustive presentation of this problem, which is otherwise impossible due to the limited time devoted to studying the issues in question. To a certain extent, aspects related to the wiring of a data network are reached, the full separation of the telephone and data networks not being possible. The chapter presents the connectors used to connect voice and data terminals, the connection equipment used on the subscriber loop and some of the principles used to install/wire these connection equipment. As a practical application it is proposed to create a telephone network that connects several telephone terminals to a PABX telephone exchange through subscriber loops that include communication cables and connection devices and equipment.

4.2 Indoor wiring of telephone and data networks

In Fig. 4.1 it is presented the schematic diagram of an indoor network, which contains a connection/junction box (or panel) where the transmission medium coming from outside is connected. This transmission medium may be represented by a twisted pair cable or a fiber optic cable [27] [28]. In the case of an optical transmission between the access point to the network (that is, the local exchange in the case of a telephone network) and the user or in the case of multiplexing several individual channels on a broadband channel, the connection box will also include the optical-electrical conversion equipment and/or the necessary multiplexing equipment, in addition to the connection equipment.

From the connection box come out individual cables to the sockets to which the terminal equipment can be connected. For telephone sockets, a flat cable, without metallic shield, with 2, 4, 6 or 8 non-twisted stranded wires, with a diameter of $0.15mm$, is usually used. The most commonly used cable is with 2 or 4 wires. The short length of these cables, the low crosstalk (a separate cable is used for each terminal), and the low level of interference in an indoor environment (at least in most situations) allows the simple construction described

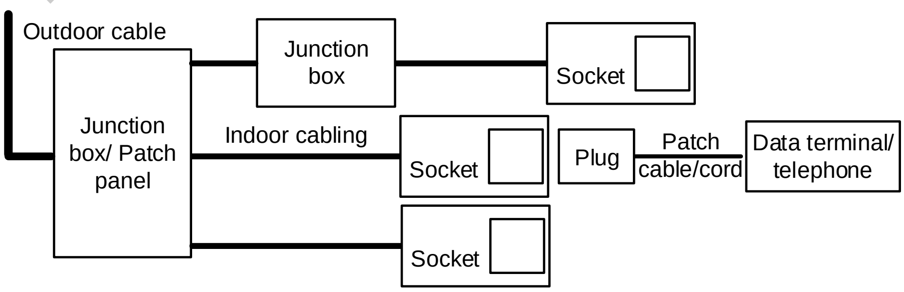

Figure 4.1: Wiring an indoor network.

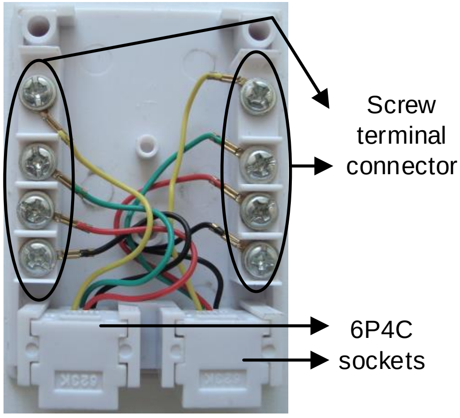

Figure 4.2: 6P4C dual telephone socket.

above. Only two wires are required to connect an analog telephone, and 2 or 4 wires are required to connect a digital telephone, but in some cases additional pairs are required, e.g. for remote power feeding (some pairs are used only for power feeding, thus providing a higher power supply current), for additional ringer or other signaling operations. The cord with plug that connects the terminal to the socket is also made up of a flat cable with several wires, usually 2 or 4 wires.

Note: the non-twisted flat cable can also be used to connect (to the distribution panel) some low or medium bit rate data equipment.

Over time, different types/standards of connectors used in telephone networks have been developed, but in current networks, so-called Registered Jack (RJ) connectors are used. These standards refer both to the structure of the socket and the plug/jack as well as to the connection of wires (wiring/crimping) to these connecting equipment.

RJ modular connectors have four different sizes, i.e. they can have 4, 6, 8 or 10 locations for pins (RJ50 connector with limited use in some proprietary data systems has 10 locations), and sometimes not all contacts/pins are installed. For example, the jacks used for telephone equipment have 6 locations for pins, but usually there are only 4 pins in the jack. For modular RJ connectors, the notation $xPyC$ is used, where x is the total number of positions/locations for the pins, and y is the number of pins or existing contacts (installed).

For indoor wiring of telephone networks, the connectors $4P4C$ (RJ9), $6P2C$, $6P4C$ and $6P6C$ are usually used [39]. The $4P4C$ connector is typically used to connect the handset to the phone, 2 wires are used for the microphone (outer wires) and 2 for the headphone/speaker (inner wires). The $6P2C$ (RJ11), $6P4C$ (RJ14) and $6P6C$ (RJ25) connectors are used to connect the telephone devices to indoor cables, the difference between these connectors consisting only of the number of pins/contacts. In the usual technical language these connectors are usually "confused" and are used as RJ11.

Note: the $8P8C$ connectors (RJ33X, RJ34X, RJ35X) [39] are used to deploy telephone networks with small number of private extension (terminals), networks in which a terminal can have direct connections to multiple terminals (i.e. the so-called "key telephone systems" or KTS). The importance of these types of networks has decreased due to the spread of PABX exchanges.

The connection of the wires to the sockets of the modular connectors mentioned above is usually done with a screw system (see Fig. 4.2 and Fig. 4.3), which requires the separation of the pairs and stripping of the wires, operations that alter the transfer characteristic of the twisted pairs, that is, increases the crosstalk and impedance mismatches appear at the connection between the pair and the socket. The problem of the impedance mismatch also occurs in the case of cables with non-twisted wires.

Note: separation of the pairs for the connection to the socket must be done on a section as small as possible in order to limit the effects mentioned above. The aforementioned effects are lower (they may even be "neglected") in telephone networks, but they are very important in data networks, especially in high-speed networks.

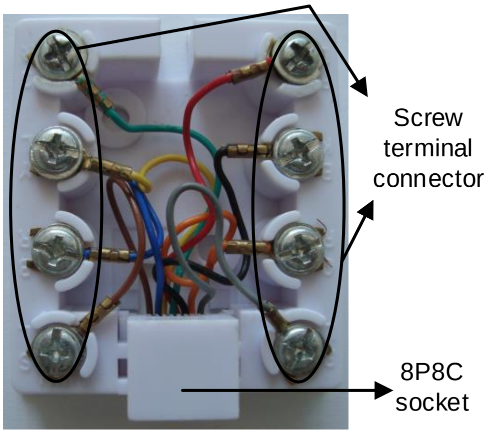

Figure 4.3: 8P8C telephone socket.

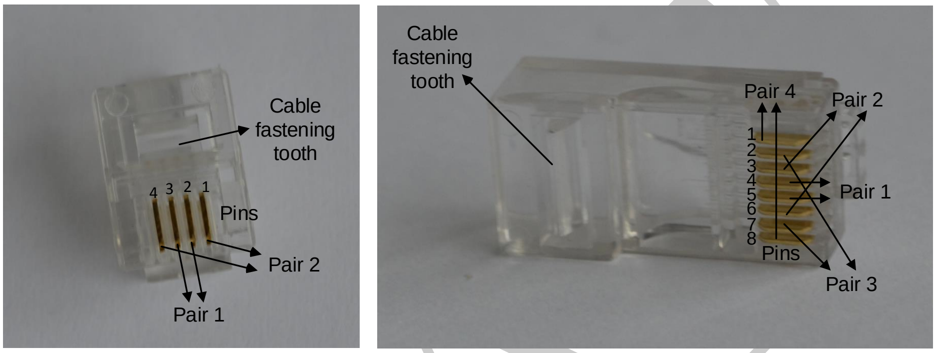

Figure 4.4: $6P4C$ and $8P8C$ plugs.

The system for fastening the wires in the plugs of the aforementioned connectors (see Fig. 4.4 and Fig. 4.5) is of IDC (Insulation Displacement Contact) type, also called as IPC (Insulation Piercing Contact). When the plug is tightened with the crimping pliers the pins cut into the insulation of the wires and make contact with the conductors and at the same time secures the wires. For the mechanical fastening of the wires in the plug there is also provided a special "tooth" in the plug (see Fig. 4.4), which presses on the cable cover and ensures better attachment of the plug to the end of the cable.

The mode of assigning the pairs to the pins of the $6P4C$ and $8P8C$ connectors in the telephone networks and the numbering of the pins of these jacks/plugs is presented in Fig. 4.4 [39] [40] [41]. It can be seen the "concentric" assignment of the pairs, the first pair is connected to the pins in the middle, the next pair to the pins located near the middle and so on. The assignment of pairs to the pins of the jacks used in telephone networks, RJ11, RJ14 and RJ25, and the color code used to identify the pairs in the cables used for indoor wiring are presented in the table Tab. 4.1.

Note: the T/R abbreviations in 4.1 refer to the wires in the telephone pair, namely T - Tip (positive polarity)

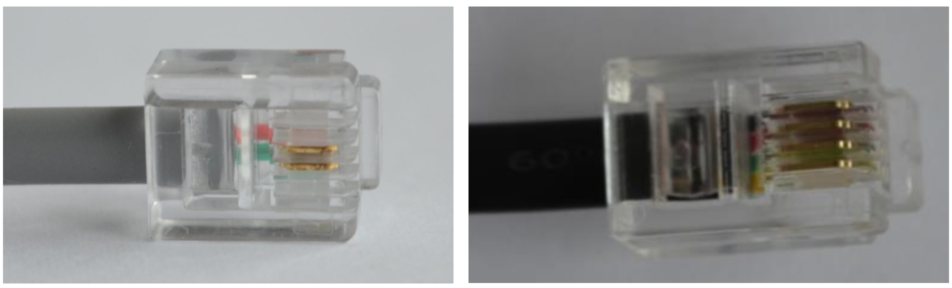

Figure 4.5: $6P2C$ and $6P4C$ plugs installed on the cable.

Table 4.1: Assignment of the pairs to the RJ11, RJ14, RJ25 jacks and the color code used to identify the wires.

Position	Pair	T/R	RJ11	RJ14	RJ25	Old colors	Cat5e/6 colors
1	3	T			T3	orange	white-green
2	2	T		T2	T2	black	white-orange
3	1	R	R1	R1	R1	red	blue
4	1	T	T1	T1	T1	green	white-blue
5	2	R		R2	R2	yellow	orange
6	3	R			R3	blue	green

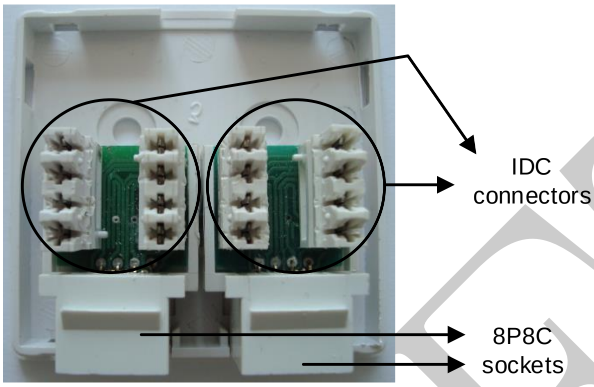

Figure 4.6: Double network socket.

and R - Ring (negative polarity).

Modular connectors of 8*P*8*C* type are also used for wiring data networks. Local area networks (LANs) using Ethernet technology (one of the most used LAN technologies) use the 8*P*8*C* RJ45 connector, with the plugs wired according to the TIA/EIA standard [39] [42]. The cables used to connect the equipment from the Ethernet network are UTP cables with 4 twisted pairs of category CAT.5e or CAT.6. The high transmission rates (10Mbps - Ethernet, 100Mbps - Fast Ehernet or 1Gbps - Gb Ethernet) require the twisting of the pairs and the use of higher category cables.

The sockets used in the Ethernet data networks have an IDC type wire fastening system, which requires loosening the pairs on a shorter length and makes it possible to install the cable more securely. In Fig. 4.6 a double network socket is shown where the system used for fastening the wires can be observed. It should be remembered that in the case of data networks the wiring of sockets and jacks should be done more carefully so as to be affected as little as possible the homogeneity and symmetry of the pairs, in order to avoid as much as possible the decrease of the crosstalk loss between the pairs and the generation of impedance mismatch at the connections between pairs and the connection equipment.

Note: there are special sockets for S-UTP (or FTP) cables, sockets where the 8*P*8*C* connector has a metal casing and there is a mounting system specifically designed for the cable shield. The 8*P*8*C* plugs for S-STP cables also have a metal casing.

Note: there is compatibility between the *xPyC* modular plugs and sockets of different types, which simplifies the wiring of indoor networks and the use of common connection equipment for telephone and data networks. For example, a 4*P*4*C* jack can be plugged into a 6*P*6*C* socket, and a 6*P*6*C* jack can be plugged into a 8*P*8*C* socket. The pins of the smaller jack will "step" on the central pins of the larger socket. Inserting/removing a smaller jack into/from a larger socket must be done carefully to avoid the damage/deformation of the marginal pins in the socket.

The wiring of the pairs of UTP cables used in data networks in the 8*P*8*C* RJ45 jack is specified by the TIA/EIA-568 standard (see Tab. 4.2 and Fig. 4.7) so that there is compatibility with the telephone networks. There are two modes of positioning the wires, namely according to standard T568A and T568B respectively [39] [42]. A cable that is wired according to the T568A standard at one end and the T568B standard at the other end is a crossover cable. These cables were used for interconnecting data equipment of the same type (e.g.

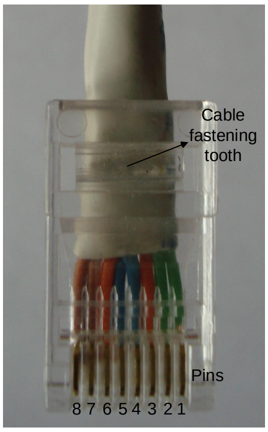

Figure 4.7: $8P8C$ jack wired according to T568A.

Table 4.2: Wiring of UTP cable pairs in RJ45 jacks.

Position	T/R	568A pair	T568A colors	568B pair	T568B colors
1	T	3	white-green	2	white-orange
2	R	3	green	2	orange
3	T	2	white-orange	3	white-green
4	R	1	blue	1	blue
5	T	1	white-blue	1	white-blue
6	R	2	orange	3	green
7	T	4	white-brown	4	white-brown
8	R	4	brown	4	brown

two data terminals/PC), before using the auto-MIDX capabilities (i.e. automatic detection of the pairs used for transmission/reception). The current network cards implement auto-MIDX. A cable that is wired identically at both ends is called a patch cable. In the case of the T568A plug, the green pair is interchanged with the orange pair.

Regarding the wiring of pairs in the RJ45 jack, the following aspects can be mentioned:

- the concentric wiring of the pairs according to the rules used in the telephone networks is not usable in the data networks because the pair that would connect to the marginal pins would have to loosen too much, which would negatively influence the crosstalk loss between the pairs.

- Ethernet transmissions at 10Mbps and 100Mbps rates use 4-wire duplexing, meaning one pair is used for transmission and one for reception, and thus remain two unused pairs (brown and blue pair) of the four pairs of the UTP cable. In case of transmission at the 1Gbps rate all four pairs of the UTP cable must be used.

- the unused brown pair, connected to pins 7-8 of the RJ45 socket respectively jack may be used for the remote power feeding of the network card from a switch or router.

- the unused blue pair, connected to pins 4-5, the central pins of the RJ45 jack and socket respectively, can be used for an analog telephone connection. A telephone cord with a $6P2C$ RJ11 jack will step over pins 4-5 of the $8P8C$ RJ45 network socket. Thus, a data terminal or an analog telephone can be connected to a network socket. If a dual network socket or a combined RJ45 - RJ11 socket is used and the blue pair is connected to one of the sockets, and the orange and green pairs to the other socket, a UTP cable with four pairs can provide at the same time a telephone and a data connection - see Fig. 4.8.

- the blue pair connected to pins 4-5 of an RJ45 socket and the orange pair (T568A) or the green pair (T568B) connected to pins 3-6 of the RJ45 socket (that is, positioned outside the blue pair) can be used to connect a digital telephone that works on 4 wires. In this case, only the digital telephone can be

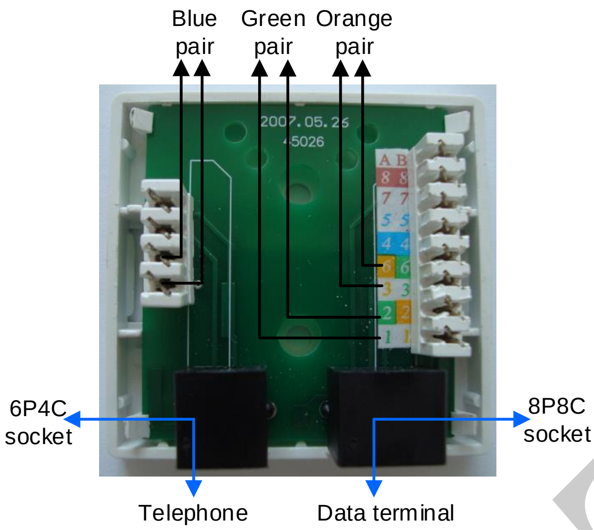

Figure 4.8: Combined network and telephone socket and pinouts for the telephone and data sockets.

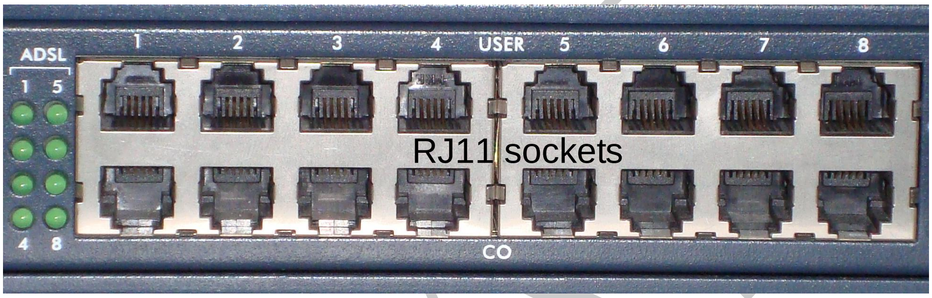

Figure 4.9: Connection panel of an 8-user ADSL access multiplexer made of RJ11 sockets.

connected to the UTP cable with four pairs and it is not possible to connect a data terminal because the 3-6 pair is used for the telephone.

4.3 Connection panels and boxes

Connection panels installed on telephone network equipment serving a small number of users can be made of RJ11 sockets. This is the case of small capacity PBX echanges, multiplexers, xDSL equipment with small number of ports, etc. In Fig. 4.9 can be seen the connection panel of an 8-port (users) ADSL access multiplexer. To each user is assigned one port for the subscriber loop and one for the connection with the telephone switching equipment, so 2 pairs for each user, i.e. two $6P2C$ RJ11 sockets for each user. A similar situation occurs in the case of the Evolio 308 A-Call PABX exchange with 8 extensions and 3 external lines, each port being accessible via a $6P2C$ RJ11 socket.

This solution for making the front panel is not suitable for telephone network equipment that serve a larger number of users, even if this number is not very large, because the RJ11 sockets take up a lot of space. For example, the HiPath 1120 small capacity PABX exchange (8 analog extensions, 4 extensions for system telephones, 2 external lines) uses Dinkle connector blocks with spring based wire attachment system (see Fig. 4.10) [43]. Using such connector blocks, can be made compact panels with a large number of ports, which can be used in the case of medium or large capacity telephone equipment.

Modular connectors with a larger number of pairs, such as the RJ21 connector, have been standardized to make compact telephony equipment panels. [44]. This connector has 25 pairs (50 wires) and can provide 25 standard telephone ports. In Fig. 4.11 the RJ21 socket is shown, and in Fig. 4.12 it is presented the connection panel of an ADSL2+ access multiplexer made of an RJ21 socket. The ADSL2+ module serves 12 users, each user having associated two ports (i.e. two pairs), one to which the subscriber loop is connected and one to

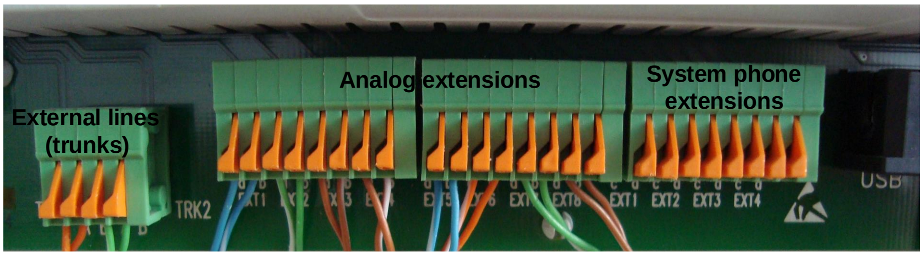

Figure 4.10: Connection panel of a low capacity PABX made of Dinkle connector blocks.

which the port of the telephone switching equipment is connected. So, in total, 24 pairs are required to connect the ADSL2+ access multiplexer to the network and if a RJ21 socket is used a pair (out of the 25) will not be used. The assignment of the pairs to the pins of the RJ21 socket and the color code used to identify the pairs from the cable to which they are connected are presented in table Tab. 4.3.

Table 4.3: Wiring of the RJ21 jack and the color code used to identify the pairs.

Color	Pin (Tip)	Pin (Ring)	Color
white-blue	26	1	blue-white
white-orange	27	2	orange-white
white-green	28	3	green-white
white-brown	29	4	brown-white
white-gray	30	5	gray-white
red-blue	31	6	blue-red
red-orange	32	7	orange-red
red-green	33	8	green-red
red-brown	34	9	brown-red
red-gray	35	10	gray-red
black-blue	36	11	blue-black
black-orange	37	12	orange-black
black-green	38	13	green-black
black-brown	39	14	brown-black
black-gray	40	15	gray-black
yellow-blue	41	16	blue-yellow
yellow-orange	42	17	orange-yellow
yellow-green	43	18	green-yellow
yellow-brown	44	19	brown-yellow
yellow-gray	45	20	gray-yellow
violet-blue	46	21	blue-violet
violet-orange	47	22	orange-violet
violet-green	48	23	green-violet
violet-brown	49	24	brown-violet
violet-gray	50	25	gray-violet

A larger indoor telephone network (e.g. covering several floors of a building and several rooms on each floor) includes several connection boxes (or terminal boxes), the connection equipment in these boxes allowing interconnection of cable segments that connect between different parts of the network (e.g. between the floors of a building) as well as the interconnection of the main cable segments with the terminal cable segments (at the end of which the sockets are installed) [27] [28]. In Fig. 4.13 it is shown the internal structure of a 50-pair connection box used in indoor telephone networks. In the figure you can see the connector blocks (or terminal strips) (5 blocks/terminal strips of 10 pairs), the cable fastening system, the grounding (earthing) socket and the pairs assignment table on the inside of the box lid. The box shown in Fig. 4.13 is made of plastic (being installed indoors it does not have to provide protection against moisture and other elements), but there are also metal indoor boxes.

The 10-pair connector block (or 10-pair terminal strip), a component of the connection boxes used in both

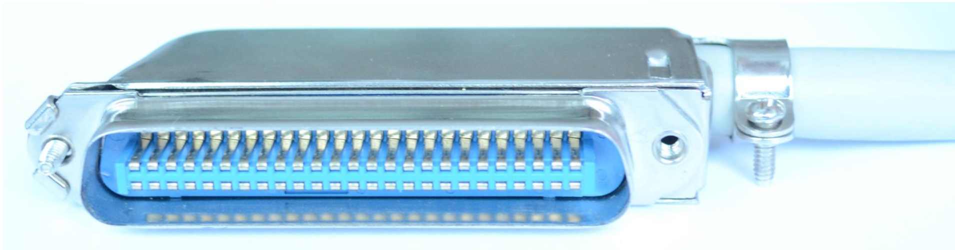

Figure 4.11: RJ21 modular connector jack with 25 pairs.

Figure 4.12: Connection panel of an ADSL2+ access multiplexer with 12 users made of an RJ21 connector.

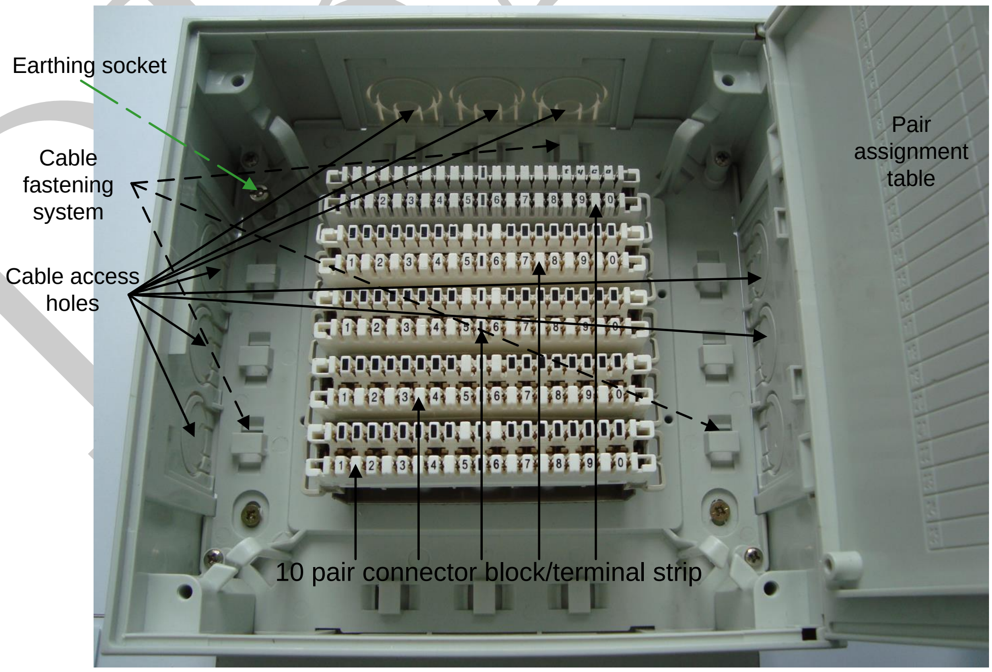

Figure 4.13: Connection box with 50 pairs for indoor telephone networks.

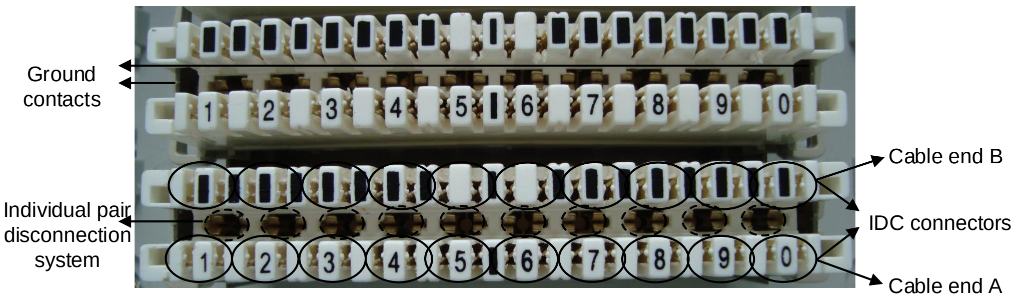

Figure 4.14: Krone type terminal strip with 10 pairs.

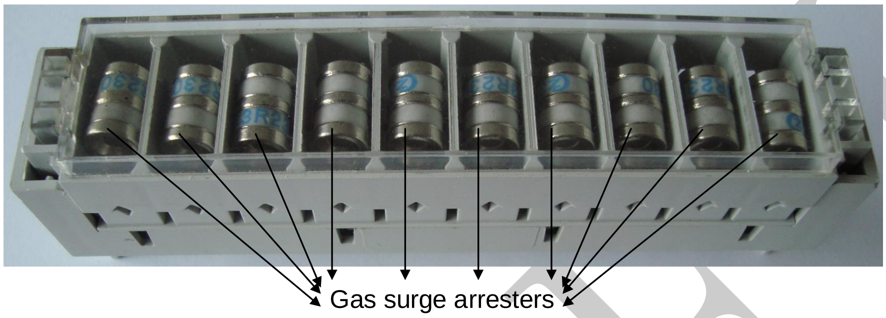

Figure 4.15: Block of gas surge arresters.

indoor and outdoor networks, is shown in Fig. 4.14. The strip uses an IDC type wire clamping system, which ensures an easy and safe installation of the wires. In the case of some types of connectors, such as Krone terminals [45], the contacts associated with the wire clamping system can be interrupted separately for each pair with the help of plastic wedges and thus it is possible to isolate the individual segments that make up the subscriber loop, which is very useful in network maintenance operations [27].

Note: connecting the wires to the connector blocks mentioned above requires loosening of the pairs on a larger length, but the negative effects of this procedure on the crosstalk loss and impedance matching are acceptable in a telephone network.

In the case of outdoor aerial telephone lines (installed on poles or building walls) it is necessary to protect the telephone equipment from the voltage surges that can occur mainly due to electrical discharges in the atmosphere. There are different types of surge arresters (surge protector/surge arrestor), of which can be mentioned mention those made of metal oxide varistors (MOV) or gas surge arresters (gas discharge arrestor) [27] [46] [47]. In Fig. 4.15 it is shown a block of gas dischargers (block of 10 units) that can be installed on the Krone connector blocks mentioned above.

Note: the protective elements of the "surge arrestor" channel the overvoltage to the ground and for this reason two arresters (each connected to one wire of the pair) are used with a common pin between them, pin that connects to the ground. The terminal strips used in the connection boxes have a separate earthing circuit to which the common pin mentioned above is connected. Of course, the earthing circuit of the terminal block is connected to the earthing socket of the connection box.

Data equipment (switch, router, hub, etc.) has the connection panel usually made of RJ45 sockets. The use of 4 pairs for each individual connection and the much tougher conditions imposed for the crosstalk loss justify such a solution, even if it takes up more space. Switching/routing equipment with a large number of ports can also use RJ21 type connectors in the connection panel. The RJ21 connection usually will only be used between the switching equipment and a connection panel (patch panel).

The connection panels used in data networks (patch panels) are also made of RJ45 socket blocks, for the reasons stated above. In Fig. 4.16 it is presented a patch panel with 48 RJ45 sockets, organized in groups of 6 sockets. The wiring is done with an IDC system identical to the one shown in Fig. 4.6. Connection panels and switching and routing equipment are mounted in cabinets of different sizes. Due to the compatibility between

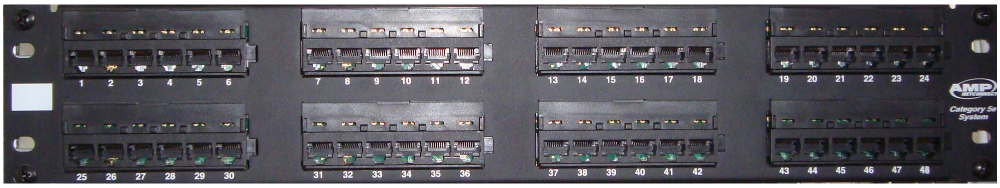

Figure 4.16: Patch panel (connection panel) with cu 48 sockets.

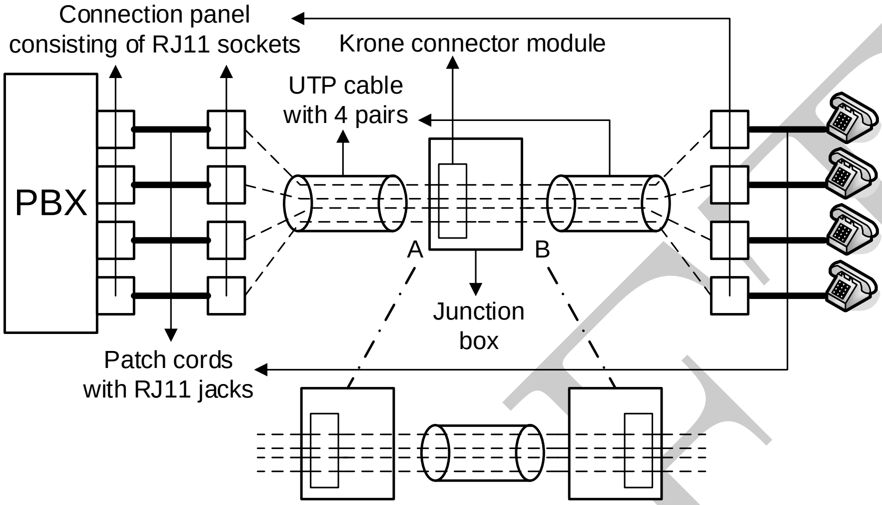

Figure 4.17: Wiring diagram of an indoor telephone network.

the RJ45 sockets and the RJ11 sockets, the connection panels for the telephone equipment also can be made of patch panel units, ensuring a convergence of the telephone and data networks.

4.4 Other aspects

Wiring of outdoor telephone networks comply with the principles that have been discussed in the paragraphs above, existing in principle the following differences:

- the connection panels of the large-capacity telephone exchanges (called also distribution frames) are composed of a large number of connector blocks; in modern networks, connector blocks with IDC type wiring system are usually used.

- the subscriber loop is composed of multiple cable segments interconnected using terminal boxes (terminal cabinets). These boxes/cabinets must be robust to prevent unauthorized access and must provide protection against atmospheric weather. The pairs interconnection equipment are made up connector blocks of different types, the IDC blocks being used in modern systems.

- outdoor telecommunication cables have a large number of pairs and specific color codes are used to identify the pairs.

- the connections between various equipment (multiplexers, switches, concentrators, xDSL access multiplexers) are done in modern networks using optical cables.

Wiring of outdoor data networks, except for a few particular cases (e.g. xDSL data network using the twisted pairs of the telephone access network) is done with optical cables.

4.5 Application

1. Consider the RJ11, RJ45, RJ21 modular connectors used in the wiring of telephone and data networks. The constructive elements of the sockets and plugs/jacks of the modular connectors considered are identified. Analyze how to connect the wires to the pins of the RJ11 and RJ45 jacks and how to connect the wires to the RJ11 and RJ45 sockets.

2. Analyze how to use the crimping tools used for wiring of the RJ11 and RJ45 jacks.

3. Analyze how to use the tools used for wiring of the IDC type RJ45 sockets.

4. Consider a double RJ45 socket or a RJ45-RJ11 combined socket. Install at one end of a 4-pair UTP cable an RJ45 jack, and connect the opposite end of the cable to the double or combined socket so that an analog telephone can be connected at one of the sockets and a data terminal at the other socket. Find a solution for connecting the cable to a PBX port and an Ethernet switch port (at the same time).

5. Modify the connection of the UTP cable to the double socket so that it is possible to connect a digital telephone that works on 4 wires, to one of the sockets of the double socket. Can a digital telephone and a data terminal be connected to the dual socket at the same time?

6. Consider the 10-pair block connectors (terminal strips) used in the connection boxes. Identify the constructive elements and analyze how to mount the block connectors in the connection boxes.

7. Analyze how to use the crimping tools used for wiring the connector blocks (terminal strips) used in the connection boxes.

8. Build the telephone access network presented in Fig. 4.17 which includes one or two connection boxes (the section between points A and B can be replaced by two connection boxes connected by a cable segment), as well as all the elements necessary for connecting analog telephone terminals to a PBX exchange. Practice the installing of the pairs (wiring) in the telephone sockets and in the connector blocks used in the telephone connection boxes. Analyze how the cable segments may be isolated and the way in which the connector blocks may be fitted with surge protection elements.

Chapter 5

Definition of level and attenuation in telephone networks

5.1 Introduction

Defining the signal levels and the attenuations (or losses) of the circuits and transmission lines, respectively of the units of measure used for these parameters is very important in telecommunications networks. It is also important to define the reference points relatively to which certain levels or attenuations (or losses) are defined, as well as the character of the load impedances.

The chapter aims at understanding of the concepts mentioned above as well as how to determine or measure the levels and attenuations (losses) in a telephone network. It is important to specify that the definition of levels and attenuations (losses) is generally valid in any telecommunications network, even if certain particularities may arise in the case of certain networks.

5.2 Signal levels

The level by definition represents the expression in logarithmic form of the ratio between two quantities of the same nature, of which one is considered the reference. Two categories of levels can be identified, namely absolute and relative levels, both expressed in decibels. In the case of the absolute levels an absolute reference value is imposed, identical for the whole network, and in the case of the relative levels the reference value is the value of the level from a certain point of the network.

5.2.1 Absolute power level

The absolute power level of a single frequency signal represents the apparent power of a sinusoidal signal, relative to the apparent power of $1mVA$, expressed in dBm [1]:

$$L_a = 10lg\frac{P}{1mVA} = 10lg\left(\frac{\frac{V^2(f)}{|Z_n(f_0)|}}{1mVA}\right)[dBm] \tag{5.1}$$

In analog networks the reference power of $1mW$ is common, but on the line interfaces (two-wire interfaces) of the digital telephone exchanges complex impedances are usually used, which means that the notion of apparent power must be used [1].

For systems with resistive impedances the active power in the band of interest (f_1, f_2) is calculated as: $P = \int_{f_1}^{f_2} \frac{V^2(f)}{R} df[mW]$, where R is the load resistance in Ω, $V^2(f)/R$ is the power spectral density in mW/Hz, and f_1 and f_2 represent the frequency band boundaries in Hz. The absolute power level in this case is:

$$L_a = 10lg\frac{P}{1mW}[dBm] \tag{5.2}$$

For systems with complex impedances the apparent power in the band of interest (f_1, f_2) is computed as: $P = \int_{f_1}^{f_2} \frac{V^2(f)}{|Z(f_0)|}df[mVA]$. f_0 represents the reference frequency at which the load impedance is measured. Typical values in telephone networks are 800Hz, 1000Hz, 1020Hz. The absolute power level in this case is:

$$L_a = 10lg\frac{P}{1mVA}[dBm] \tag{5.3}$$

The variation of the level with the frequency is dependent only on the variation of the voltage with the frequency, the impedance being determined only at the reference frequency.

In the case of a signal specified in time (more precisely the signal voltage is a function of time $v(t)$) the power can be computed according to the relation:

$$P = \frac{1}{T \cdot |Z(f_0)|}\int_{-T/2}^{T/2} v^2(t)\,dt \tag{5.4}$$

In the case of a periodic signal T is the period of the signal, and in the case of a non-periodic signal (that is, in the case of an impulse) $T \to \infty$.

Calculation of the power of a random (or stochastic) signal, x, which is characterized by the probability density function of the signal values, $pdf\,(x)$, is made according to the relation:

$$P = \frac{1}{|Z(f_0)|}\int_{X_i}^{X_s} pdf\,(x) \cdot x^2 dx \tag{5.5}$$

where X_i and X_s represents the lower and the upper limit of the random (or stochastic) signal x.

Examples of random signals are voice signal, white noise, quantization noise, etc.

5.2.2 Absolute psofometric power level

The psofometric power level (expressed in $dBmp$) takes into account the physiological characteristics of the ear, characteristics that cause the perception of sounds to depend on the frequency according to a weighting function $W(f)$, called the psofometric weighting function (see Fig. 5.8) [1] [48]. Low-frequency spectral components (e.g. under $300Hz$) or the higher-frequency components (e.g. more than $3000Hz$) are less noticeable to the human ear than frequency components contained in the band $800Hz - 2000Hz$, where hearing acuity is the greatest. If this weighting function is taken into account, the expressions that give the power and the absolute power level become:

$$P_p = \int_{f_1}^{f_2} \frac{V^2(f)}{|Z(f_0)|}10^{W(f)/10}df[mVA] \tag{5.6}$$

$$L_P = 10lg\frac{P_p}{1mVA}[dBmp] \tag{5.7}$$

In the above relation the weighting function $W(f)$ is given in dB.

5.2.3 Absolute noise power level

The absolute power level of noise is defined for noise signals, which usually have much lower levels than the useful signals. The absolute level of noise power is expressed in $dBrn$ [1]. The reference power for complex impedance systems is $1pVA$, and for resistive impedance systems is $1pW$. From this results the relation between the absolute power levels expressed in dBm and the absolute noise power levels expressed in $dBrn$: $0dBrn = -90dBm$, meaning $L[dBrn] = L[dBm] + 90dB$.

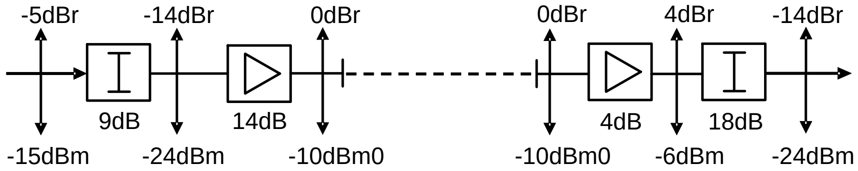

Figure 5.1: Defining absolute and relative power levels on a transmission chain.

5.2.4 Absolute voltage level

Regarding the definition of the absolute voltage level it should be kept in mind that the reference value can also be a predefined voltage. By definition the absolute voltage level expressed in dBu is given by the following relation[1]:

$$L_u = 20lg\frac{U_x}{775mV}[dBu] \tag{5.8}$$

where U_x is the effective value of the measured voltage, and $775mV$ is the effective voltage that produces a power of $1mW$ on a 600Ω impedance. The voltage level is different from the power level if the load impedance value is different from 600Ω, in which case we have the relation:

$$L_a = L_u + 10lg\frac{600}{|Z|}[dB] \tag{5.9}$$

5.2.5 Relative power level

The relative power level at a given point in the transmission chain is given by the following ratio expressed in dBr [1]:

$$L_r = 10lg\frac{P(1020Hz)}{P_0(1020Hz)}[dBr] \tag{5.10}$$

where P and P_0 represents the apparent powers of the signal at the considered point and at the reference point.

For the evaluation of relative levels, single frequency signals are usually used, the frequency value being equal to the reference frequency value. The relative level is numerically equal to the attenuation (if on the transmission chain the measurement point is located before the reference point) or the gain (if on the transmission chain the measurement point is located after the reference point) at the reference frequency between the considered point (the measurement point) and the reference point, where the relative level is $0dBr$. The measurement signal applied at the transmission reference point ($0dBr$) usually has a value of $-10dBm$ to avoid the risk of occurring non-linearity distortions. The relative level allows the characterization of the gain or the attenuation (loss) between different interfaces. The level expressed in $dBm0$ represents the absolute power level measured at the reference frequency ($1020Hz$) in the transmission reference point ($0dBr$). In some measurement point having relative level L_r, a reference level of L $dBm0$ generates an absolute power level $L_a = (L + L_r)$ $[dBm]$.

An illustrative example for defining absolute and relative power levels on a mixed analog-digital transmission chain is shown in Fig. 5.1.

5.2.6 Choice of reference point

The choice of the reference point in a transmission chain is an important issue, the evaluation of the attenuations/losses and the gains being performed relative to this point. In the case of a digital telephone channel, the choice of the reference point on or near the digital path is indicated, due to the fact that the digital paths (channels) have attenuation of $0dB$. There are attenuation distortions on the physical paths (channels) on which the digital transmission is actually achieved, attenuation distortions that contribute to a certain extent to bit errors, but the values of the PCM coded samples transmitted on these digital

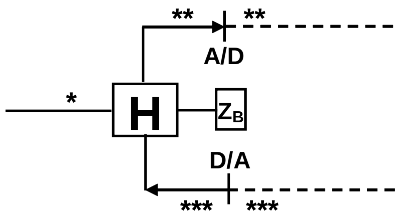

Figure 5.2: Choice of transmission reference points in digital telephone networks.

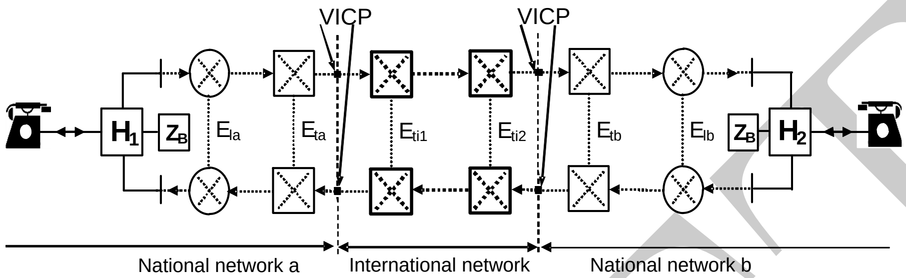

Figure 5.3: Defining virtual international connecting points.

paths/channels do not change (the effect of bit errors can not be considered a change of the attenuation/gain) [4].

If a common reference point is desired for both transmission directions, then the reference point can be set at the input of the hybrid transformer (point $*$ in Fig. 5.2), and if different reference points are desired for the two transmission directions, then on the transmission path the reference point can be chosen before or after the A/D converter (points $**$ in Fig. 5.2), and on the reception path the reference point can be chosen before or after the D/A converter (points $***$ in Fig. 5.2).

Regarding the choice of the reference point, the virtual international connecting points must also be mentioned - VICP (Virtual International Connecting Point) [1]. Virtual international connecting points define the boundaries between the national and international parts of a connection (see Fig. 5.3). These points are also used as global reference points for national and international network segments.

By definition, at these points the relative nominal levels are as follows: $0dBr$ transmission, $0dBr$ reception for digital circuits and $0dBr$ transmission, $-0.5dBr$ reception for analog and mixed circuits [1].

5.3 Defining attenuation

The attenuation or loss, a, is generally defined as a ratio expressed in decibels (or in nepers) between real or apparent powers or between effective voltage values according to the following relations [49]:

$$a = 10lg\frac{P_i}{P_o}[dB] \tag{5.11}$$

$$a = \frac{1}{2}ln\frac{P_i}{P_o}[Np] \tag{5.12}$$

$$a = 20lg\frac{|V_i|}{|V_o|}[dB] \tag{5.13}$$

$$a = ln\frac{|V_i|}{|V_o|}[Np] \tag{5.14}$$

Note: Np units are less used in practice than dB units.

If the ratio is defined inversely, that is, between the power (voltage) of the output signal and that of the input signal, we are dealing with gain.

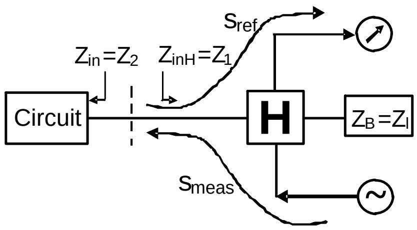

Figure 5.4: Definition and measurement of return loss.

Defining the attenuation/loss as a voltage ratio is theoretically valid only if the impedances on which the two voltages are measured are identical or if the voltage measurement points are separated by operational amplifiers, but in practice it is accepted (at least in some cases) this definition of attenuation, even if the above conditions are not met.

5.3.1 Return loss

Return loss (a_a) is a parameter, specific to transmission systems, that reflects the degree of mismatch between two impedances Z_1 and Z_2 and is given by the following relation [1]:

$$a_a = 20lg\left|\frac{Z_1 + Z_2}{Z_1 - Z_2}\right| = -20lgr[dB] \tag{5.15}$$

where $r = \left|\frac{Z_1 - Z_2}{Z_1 + Z_2}\right|$ is the reflexion coefficient.

This loss (attenuation) represents the ratio expressed in dB between an incident signal and the reflected signal in an impedance discontinuity point characterized by the impedances Z_1 and Z_2. How to define and how to measure the return loss are shown in Fig. 5.4.

5.3.2 Apparent power loss

The apparent power loss characterizes a circuit from the point of view of the power transfer [1]. In Fig. 5.5 it is presented the simplified case of a circuit that has a voltage source with value E and internal impedance Z_i connected to the input and a load impedance Z_l is connected to the output. Also in this figure is presented the structure of the reference circuit used to define this attenuation. The impedance Z_x in the reference circuit is equal, in this case, with Z_i.

The apparent input power P_i is defined as the apparent power generated by the source on a load impedance equal with the impedance Z_i of the generator, that is:

$$P_i = \frac{E^2(f)}{4|Z_i(f_0)|} \tag{5.16}$$

The power P_0 is the apparent output power on the load impedance Z_l, that is:

$$P_o = \frac{V_o^2(f)}{|Z_l(f_0)|} \tag{5.17}$$

The apparent power loss is given by the relation:

$$a_a = 10lg\frac{P_i}{P_o} = 20lg\left(\frac{E(f)}{2V_o(f)}\sqrt{\frac{|Z_l(f_0)|}{|Z_i(f_0)|}}\right)[dB] \tag{5.18}$$

and represents the ratio in dB between the maximum power that the source connected to the input can generate and the power that is generated by the source on the load impedance connected at the output of the circuit. The above general formula is applicable in the case of passive circuits, such as filters or reciprocal passive two-port networks (circuits), in which case the concept of attenuation is based on the ratio of apparent

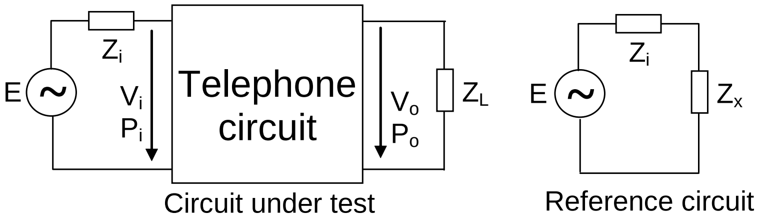

Figure 5.5: Definition of apparent power loss

power levels between input and output (identical definition in both transmission directions). If we have in the transmission chain amplifiers (for example amplifiers built with operational amplifiers), whose transfer function is not influenced by the Z_i and Z_l impedances, the condition $20lgE/V = ct$ depending on the frequency is met. In this situation the apparent power loss is expressed at the reference frequency $f_0 = 1020Hz$ in the following way [1]:

$$a_{a0} = 20lg\left(\frac{E(f_0)}{2V_o(f_0)}\sqrt{\frac{|Z_l(f_0)|}{|Z_i(f_0)|}}\right)[dB] \tag{5.19}$$

The apparent power loss is expressed according to the frequency as follows:

$$a_a(f) = 20lg\left(\frac{E(f)}{2V_o(f)}\sqrt{\frac{|Z_l(f_0)|}{|Z_i(f_0)|}}\right)[dB] \tag{5.20}$$

Since the impedance ratio is expressed at the reference frequency f_0, the variation of loss (attenuation) with frequency depends exclusively on the voltage ratio E/V. The use of impedances at the reference frequency value is indicated in practice due to the fact that it is more difficult to determine/measure the impedance values according to the frequency and the values of the impedances used in a transmission network are usually standardized at certain frequencies. So it is simpler to use the impedance values at certain frequencies to define attenuations/losses.

5.3.3 Insertion loss

The insertion loss characterizes a circuit also from the point of view of the power transfer. The situation is similar to that shown in Fig. 5.5, the impedance Z_x in the reference circuit being equal in this case to the load impedance Z_l [49] [50].

The apparent input power P_i is defined as the apparent power generated by the source on a load equal with the load impedance Z_l, that is:

$$P_i = \frac{E^2(f)|Z_l(f_0)|^2}{|Z_i(f_0) + Z_l(f_0)|^2|Z_l(f_0)|} \tag{5.21}$$

P_o is the apparent output power on load impedance Z_l, that is:

$$P_o = \frac{V_o^2(f)}{|Z_l(f_0)|} \tag{5.22}$$

The insertion loss is given by the relation:

$$a_i = 10lg\frac{P_i}{P_o} = 20lg\left(\frac{E(f)}{V_o(f)}\frac{|Z_l(f_0)|}{|Z_i(f_0) + Z_l(f_0)|}\right)[dB] \tag{5.23}$$

and represents the ratio between the power that a source can generate on a load impedance directly connected to the source and the power that is generated by the source on the same load impedance connected at the output of the circuit. Like in the case of apparent power loss the general formula is applicable in the case of passive circuits [1].

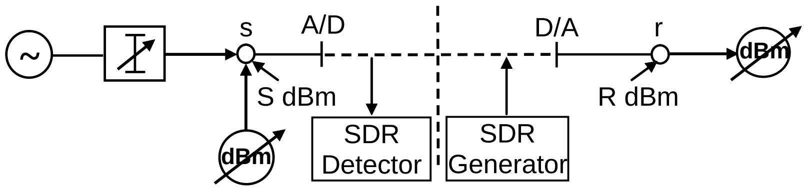

Figure 5.6: Determining the relative levels at the points located before the PCM encoders and after the PCM decoders.

The insertion attenuation is expressed at the reference frequency $f_0 = 1020Hz$ as follows:

$$a_{i0} = 20lg\left(\frac{E(f_0)}{V_o(f_0)}\frac{|Z_l(f_0)|}{|Z_i(f_0)+Z_l(f_0)|}\right)[dB] \tag{5.24}$$

The formula is especially useful for circuits that include operational amplifiers.

The insertion attenuation is expressed according to the frequency as follows:

$$a_i(f) = 20lg\left(\frac{E(f)}{V_o(f)}\frac{|Z_l(f_0)|}{|Z_i(f_0)+Z_l(f_0)|}\right)[dB] \tag{5.25}$$

5.3.4 Measurement of levels in circuits with PCM coders and decoders

For measuring the coders and decoders of the PCM systems from the point of view of the inserted losses, the concept of Digital Reference Sequence (DRS) was defined. DRS is a possible sequence of PCM codes that by decoding, with the help of an ideal decoder, provides a signal with a level of $0dBm$. Conversely, an analog signal with the $0dBm$ level applied to the input of an ideal encoder will generate a digital reference sequence [1].

The frequency of the analog measurement signal (sinus more precisely) is $1kHz$ and this signal can be represented as a sequence of 8 samples. Another possible choice of the DRS signal frequency is that it is not a sub-harmonic of the sampling frequency of $8kHz$, with a suitable value of $1020Hz$, but in this case the generation of the DRS signal will be more complicated. The DRS samples obtained from the reference sinusoidal signal have a peak value of 118, the maximum coding level of 127 corresponding, in the case of a PCM coder using the A law, to a sinusoidal signal with the level of $3.14dBm$.

The use of DRS sequences allows the determination of relative levels at points s and r located before the A/D converter (i.e. PCM encoder) and after the D/A converter (i.e. PCM decoder). If the level at point s is adjusted until the DRS detector placed after the A/D converter identifies this sequence corresponding to the $0dBm$ level, then the absolute level S measured at point s represents the relative level in this point expressed in dBr.

Similarly, in the case of the PCM decoder, if a DRS sequence is applied to the input of the D/A converter and an absolute level of $RdBm$ is measured at point r, then the relative level in this point is $RdBr$. See Fig. reffig:nivele-fig6 related to what was discussed.

5.4 Loudness rating

A telephone communication implies the existence at the two ends of two defining elements: one emits the sound message (mouth), and the other receives the sound message (ear). Such communication involves an acoustic pressure at both ends of the connection and an electrical signal for remote transmission of the sound message. Proper planning of a telephone network requires the expression of the electroacoustic loss between the acoustic source and the acoustic receiver in dB , loss called Loudness Rating (LR). If the considered circuit is subdivided into smaller units, then the global loudness rating is the sum of the individual LR values [1].

If the VICP point is taken as a reference for the national system, LR can be defined by the following two components:

- loudness rating at transmission (LRT): it represents the attenuation of the sound between the talking subscriber's mouth and the VICP point - in this case the attenuation of the sound represents the weighted average (in dB) of the acoustic pressure of excitation related to the measured voltage.

- loudness rating at reception (LRR): it represents the attenuation between the VICP point and the listening subscriber's ear - in this case the attenuation of the sound is defined as the weighted average (in dB) of electric excitation force relative to the measured sound pressure.

Remark: the loudness rating at transmission (LRT) and at reception (LRR) can be determined in principle at any interface of the telephone network.

Note:related to the weights mentioned above, see equations 5.27 and 5.28.

In Fig. 5.7 is given the whole diagram of a telephone connection, from the sound source to the sound receiver, consisting of several parts in chain.

We have the following relations for the loudness rating at the transmission and the reception:

$$
\begin{aligned}
LRT &= LRT(\text{phone device}) + \sum_{i=1}^{n} LRC_i \\
LRR &= LRR(\text{phone device}) + \sum_{i=1}^{n} LRC_i \\
LRG &= LRT + LRR
\end{aligned} \tag{5.26}
$$

Determining the loudness rating can be done subjectively - methods that are not reliable, or objectively - a psycho-acoustic model is needed that simulates how the brain interprets sound impressions. The ear is represented by a set of band filters regularly distributed on a logarithmic frequency scale. If the signal in a band exceeds the audibility threshold the corresponding filter emits an output signal and all signals from the filter outputs are composed, the composition rule depending on the sound level [1]:

- at low sound levels the composition is made in power.

- at normal sound levels the composition is made according to the relation:

$$
LR = L_0 - \frac{10}{m} lg \left(\sum_{i=1}^{N} K_i 10^{-0.1 m L_i} \right) \tag{5.27}
$$

  where L_0 is a constant (in the case of the LRC is 0), N is the number of band filters, i is the frequency index f_i, L_i is the attenuation at frequency f_i, m is a constant that depends on the sound level, namely: $m = 0.2$ for normal levels, $m = 0.5$ for small levels, $m = 1$ for very low levels ($m = 0.2$ applies for LRG, LRT, LRR and LRC - see Fig. 5.7), K_i is the weighting coefficient at the frequency f_i and these weighting coefficients have the property $\sum_{i=1}^{N} K_i = 1$. If $m = 0.2$ (normal levels) and the difference between the L_i values does not exceed $10 - 15dB$, the simplified relation can be applied:

$$
LR = L_0 + \sum_{i=1}^{N} K_i L_i \tag{5.28}
$$

The loudness rating of the telephone terminals is determined with the help of special equipment - a very exact determination is not required, the actual LR depending on the subscriber. The loudness rating of the circuit is equal to the attenuation at the frequency of $1020Hz$, if the attenuation is constant in the band, respectively with the average attenuation in the band $300 - 3400Hz$, if there are significant attenuation distortions. The loudness rating of the subscriber circuits can also be determined with the relation: $LRC = KL\sqrt{RC}$, where R is the distributed resistance of the cable in Ω/km, C is the distributed capacity of the cable in nF/km, L is the length in km, K is a constant that depends on the cable's terminating impedance Z_0 ($K = 0.014$ for

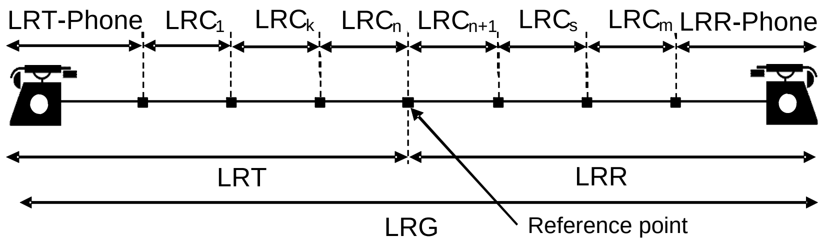

Figure 5.7: Loudness ratings defined for a telephone connection.

$Z_0 = 600\Omega$, $K = 0.015$ for $Z_0 = 900\Omega$, $K = 0.016$ for complex Z_0). Under normal operating conditions a value of $LRG = 10dB$ is considered acceptable.

5.5 Evaluation of noise level

To evaluate the noise level, different weighting filters (also called weighting functions) are used for different performance evaluation situations, as follows [1] [48]:

- the unweighted frequency (flat) function is used to evaluate the performance of telecommunications equipment and to evaluate the performance of data transmissions performed on telephone channels (or on other channels). The unweighted function is a low pass type transfer function with a bandwidth of $3kHz$ and which is flat up to the frequency of $50Hz$ in order to be able to evaluate the effects of the frequency of the power supply network and its harmonics induced in the telecommunication lines.
- the narrow band frequency weighting function is necessary for the identification and evaluation of single frequency disturbances (noise represented by sinusoidal signals, e.g. the signaling tones).
- the psophometric weighting function is a special weighting (filtering) function for evaluating the effects of noise on voice signal transmission. The frequency sensitivity function of the ear is taken into account. The standardized weighting function is shown in Fig. 5.8.
  - the noise spectral components located in the band $800 - 2500Hz$ are the most distressing, the ear sensitivity being the highest in this band.
  - ear sensitivity decreases at frequencies below $300Hz$ and over $4000Hz$, the noise spectral components below and above these frequencies having a relatively small effect.
  - the absolute level of weighted (psophometric) power is expressed in dBmp units.

5.6 Application

1. The ways of defining the signal levels will be analyzed. The importance of the level concept in transmissions systems will be identified.
2. Using a multifunctional measuring instrument available in the laboratory, the level of periodic signals (sinus, rectangle, triangle) and of a white noise signal will be measured. The amplitude and frequency of the periodic signals respectively the power and the bandwidth of the noise signal will be changed.
3. The modes of defining the apparent power loss and the insertion loss will be analyzed. The advantages of these methods of defining the attenuation will be identified.
4. The ways of defining the reference points will be analyzed.
5. The importance of return loss in a cable access network will be identified.

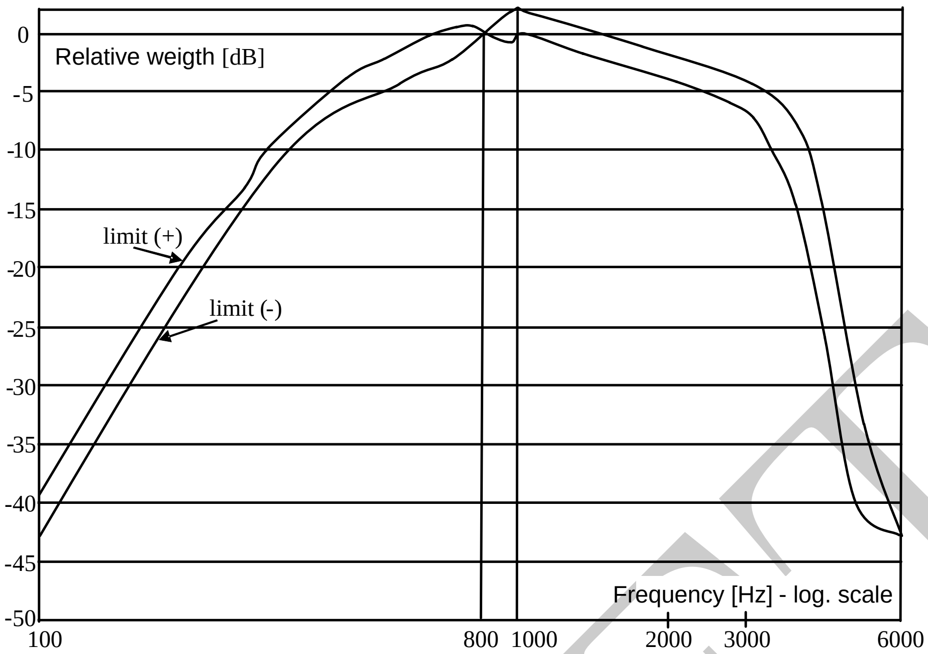

Figure 5.8: Psophofometric weighting function.

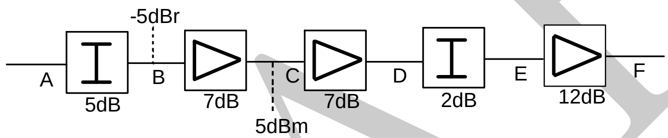

Figure 5.9: Transmission chain. Example 1.

6. Will be analyzed the way of defining and generating the DRS signal as well as the utility of this signal.

7. Will be analyzed how to define the loudness rating and will be identified the importance of this parameter.

8. The methods of evaluating the noise level will be analyzed. The cases in which each of the methods in discussion are used will be identified.

5.6.1 Exercises

1. On a resistance of 600Ω an absolute power level of -6dBm is measured. What is the amplitude of the sinusoidal signal corresponding to this level? What is the amplitude of the bipolar rectangular signal with a duty factor of 25 %, 50 %, respectively 75 % corresponding to this level? Redo the calculations for a unipolar rectangular signal.

2. Give the formulas for calculating the absolute power level for a tone respectively for a signal with frequency band $f_1 - f_2$.

3. At the input of a cable with length 3km and an attenuation constant of 0.4Np/km a sinusoidal signal with a level of -5dBm is applied. What is the absolute power level obtained at the output of the cable and what is the amplitude of the sinusoidal signal if the load impedance is 500Ω. Redo the calculations for a bipolar rectangular signal with a duty factor of 50%.

4. Calculate the absolute power level of a stochastic signal with the dynamic range $[1; 2.71828]$ and the probability density function 1/x. Redo the calculations for a constant probability density of 1/1.71828. What conditions must meet the probability density function?

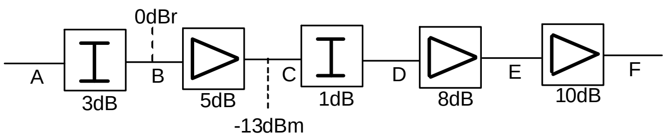

Figure 5.10: Transmission chain. Example 2.

5. Calculate the absolute voltage level of a 2V amplitude sinusoidal signal. The load impedance is 500Ω. Redo the calculation for a bipolar symmetrical triangular signal with the same amplitude.

6. Calculate the absolute power and voltage level of a unipolar rectangular signal with amplitude 1V and a duty factor of 25%. The complex load impedance module is 400Ω. What is the reference power used in this case? Redo the calculations for the 50% and 75% duty factors.

7. A cable has a load impedance of 600Ω, and the input impedance is 500Ω. Calculate the return loss? What is the maximum (possible) and minimum (possible) values (in dB and ratio, respectively) of the return loss?

8. A certain signal has the amplitude spectral density (effective voltage) $X(f)$ defined by the following function:

   $X(f) = 0.002V/Hz,\ f \in (200, 800)\,Hz;\ X(f) = 0.004V/Hz,\ f \in (800, 3800)\,Hz$. Calculate the absolute power level of this signal on a unit impedance.

9. It is given the transmission chain in Fig. 5.9. Calculate the relative power levels expressed in dBr at each point. Calculate the absolute power levels expressed in dBm at each point. Where is the reference point located? In what unit of measure is the absolute level of power expressed at this point?

10. Repeat the previous point for the transmission chain given in Fig. 5.10.

Chapter 6

Voice signal encoding techniques used in telephone networks

6.1 Introduction

The chapter proposes the study of some voice coding techniques (i.e. analog to digital conversion techniques for voice signal) used in digital communications networks. More specifically, it is about Pulse Coded Modulation (PCM) and Delta Modulation. The PCM technique is used in public digital telephone networks while Delta modulation is used in dedicated/special digital telephone networks, such as military networks.

The chapter specifically proposes to address the following issues:

- the study of the theoretical principles of PCM modulation with nonuniform quantization. The focus will be on understanding the digital implementation of the nonuniform quantization process.
- computation of some parameters that characterize PCM modulation with nonuniform quantization, parameters necessary for designing such a modulator.
- experimental evaluation of the performances of this voice coding technique (PCM). More precisely, the effect of the quantization noise level on the quality of the reception of test voice signals will be monitored.
- the study of the theoretical principles of linear Delta modulation as well as those of adaptive Delta modulation. The study of particular adaptive Delta modulations with simple adaptation of the quantization step.
- computation of parameters that characterize the linear Delta modulation and the adaptive Delta modulation, parameters necessary for designing linear or adaptive Delta modulators.
- experimental evaluation of the performances of linear Delta modulation and adaptive Delta modulations. Specifically, the effect of the quantization noise level and the effect of slope overload distortion on the quality of the reception of test voice signals will be monitored.

6.2 PCM modulation

PCM (Pulse Coded Modulation) is a voice signal encoding technique defined by ITU-T G.711 standard [51] and is the technique used in fixed digital telephone networks to encode the voice signal. PCM modulation is actually an analog to digital conversion technique that involves a series of steps, as presented below [4] [52].

The first step in converting the analog voice signal into a digital signal is filtering the analog signal, that is, limiting the frequency band to the telephone band $[300 - 3400Hz]$. The next step is sampling at a frequency

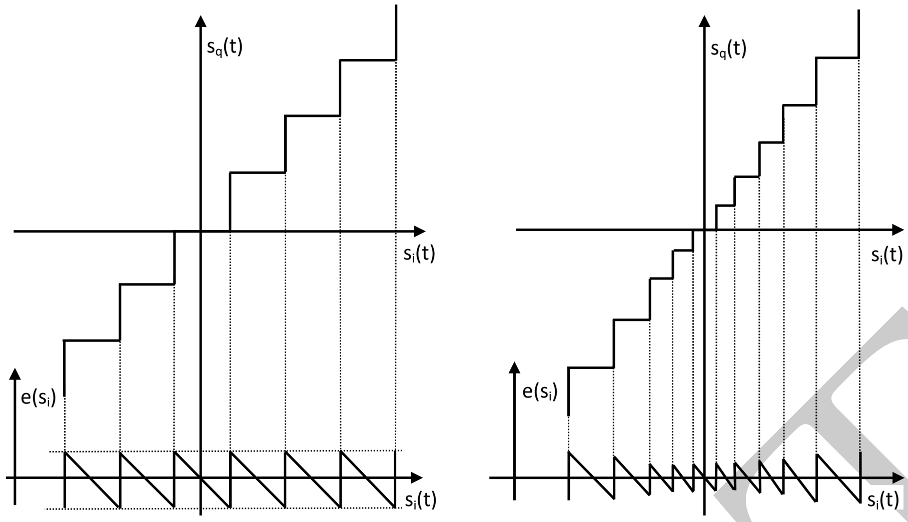

Figure 6.1: Uniform and nonuniform quantization process.

that complies with the sampling theorem, i.e. $f_e > 2 \cdot f_m$, so the sampling frequency was chosen $f_e = 8kHz$. The filtering has the role of preventing the occurrence of the aliasing phenomenon.

Note: due to the choice of $8kHz$ for the sampling frequency the cutoff frequency of the low-pass or band-pass antialiasing filter can be set around $4kHz$, which means that the telephone band in fixed digital systems can be a bit larger than the standard telephone band.

The next step is to quantize the values of the samples and then assign binary codes on a certain number of bits to the quantized values. In the quantization process appears a quantization error called quantization noise [53]. Due to the logarithmic shape of human acoustic sensitivity function it is necessary to reduce the quantization noise at low levels of the voice/acoustic signal and it is possible to allow a higher level of the quantization noise at higher levels of the voice/acoustic signal, where the hearing acuity is more reduced. In Fig. 6.1 it is presented the process of uniform quantization and that of non-uniform quantization [4] [53]. The figure also shows the variation of the quantization error within a quantization interval/step. The steps in the quantized signal, $s_q(t)$, are equal to the median value (that is, the middle value) of the quantization intervals. The steps in the signal $s_q(t)$ represent the reconstruction (or restoration) levels of the samples at the reception.

Each quantization interval is identified by a binary code and the sample falling within a certain interval is associated with the binary code of the interval. In digital telephone systems each sample is encoded on 8-bit, i.e. 256 quantization intervals are used, 128 negative and 128 positive intervals. The bit rate of a digital telephone channel will be: $D_0 = f_e \cdot nbit/eșant = 8kHz \cdot 8biți = 64kbps$

Note: ensuring acceptable level of the quantization noise in the entire dynamic range of the voice signal also can be achieved by using uniform quantization, but in this case a small quantization step should be used, which ensures a low quantization noise at low signal levels. This small quantization noise does not need to be provided at high signal levels. A small quantization step, however, means a large number of quantization intervals and thus a large number of bits per sample, i.e. a large bit rate.

In order to carry out the nonuniform quantization process, as a rule, the procedure is as follows [4]:

- at the transmission side, a nonlinear processing of the source signal, called compression, is performed, followed by a uniform quantization process. The compression process amplifies more the low signal levels and less the high signal levels - see Fig. 6.2. For simplicity of implementation of the compression process, the compression characteristic is approximated with line segments - see Fig. 6.2.

- at the reception side, a digital to analogue conversion is performed according to a uniform quantization process, followed by a nonlinear processing of the obtained samples, called expansion. This process amplifies less the low signal levels and more the high signal levels - see Fig. 6.2. The expansion function represents the inverse of the compression function and thus overall the source signal samples are not

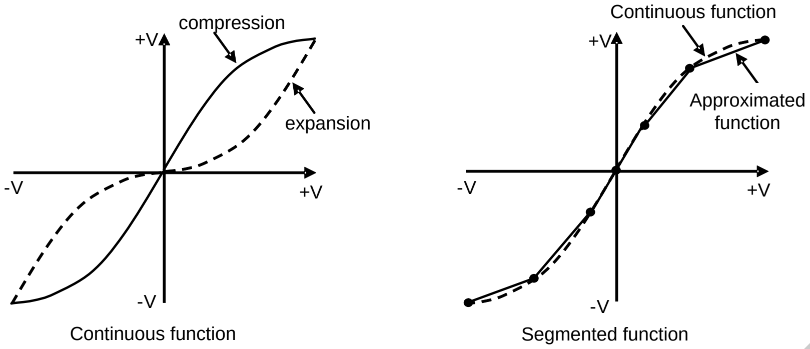

Figure 6.2: Continuous and segmented compression and expansion functions.

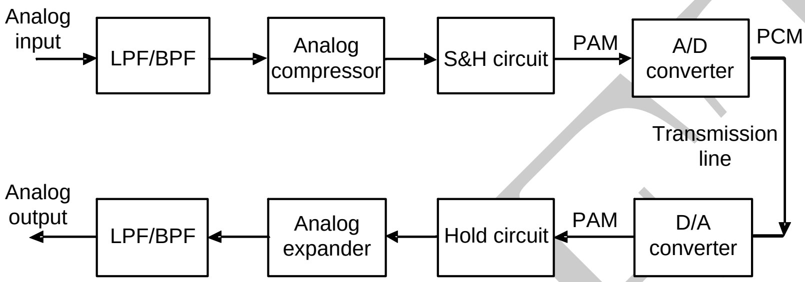

Figure 6.3: The processing chain characteristic to a digital transmission using PCM modulation with companding in the analog domain.

affected/distorted. Also, for simplicity of implementation, the expansion characteristic is approximated with line segments - see Fig. 6.2.

Note: the combined process of compression and expansion is called the companding [52].

In Fig. 6.3 it is presented the processing chain involved in the digital transmission of the voice signal using PCM modulation with companding. The processes tacking place both at the transmission and reception ends are highlighted.

Note: the Sample and Hold (S&H) circuit samples the analog signal and holds the sample value during a sampling period.

In digital telephone networks the compression operation is performed according to so-called compression laws, namely the μ law (6.1) and the A law (6.2) [4] [52] [54].

The μ law, used in the United States and Japan, is defined according to the equation::

$$f(x) = sgn(x)\frac{ln(1+\mu|x|)}{ln(1+\mu)}, 0 \leq x < 1 \tag{6.1}$$

The value of μ parameter is 255.

The A law, used in Europe, is defined according to the equation:

$$f(x) = \begin{cases} \frac{Ax}{1+ln(A)}, & 0 \leq |x| < 1/A \\ sgn(x)\frac{1+ln(A|x|)}{1+ln(A)}, & 1/A \leq |x| < 1 \end{cases} \tag{6.2}$$

The value of parameter A is 87.6, value for which the continuity of the compression function is obtained.

The implementation of the compression and expansion functions in the analog domain raises a number of issues, even if the functions in question are approximated with line segments. A more precise implementation, immune to the tolerances of electronic components and to the variability with temperature and time of these components, can be done in the digital domain. In Fig. 6.4 it is presented the processing chain characteristic to a PCM transmission with companding implemented in the digital domain. Both the processing from the transmission side and from the reception side are highlighted. Essentially it is about the following processing:

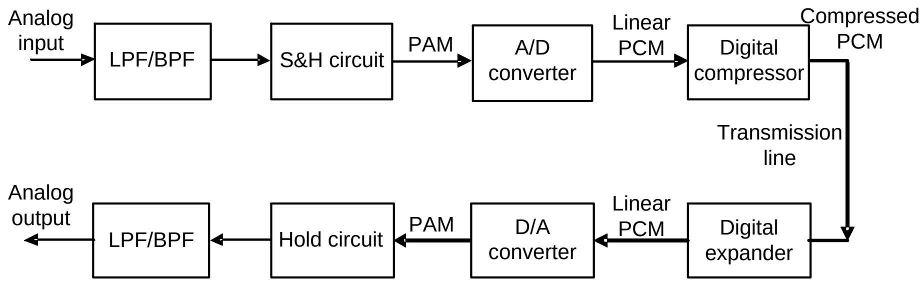

Figure 6.4: The processing chain characteristic to a digital transmission using PCM modulation with companding in the digital domain.

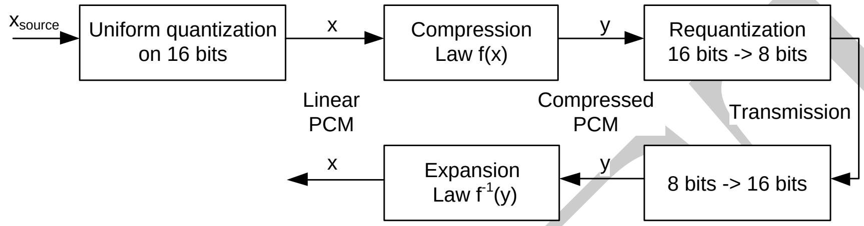

Figure 6.5: Digital implementation of the compression and expansion processes.

- at the transmission side a linear PCM modulation is performed, using a uniform quantization, with a large number of bits per sample (13 bits in case of A compression law and 14 bits in case of mu compression law) followed then by a compression process in the digital domain. Finally, a PCM modulated signal with nonuniform quantization and 8 bits per sample is obtained.
- at the reception side, an expansion process is performed in the digital domain, a process that receives the PCM codewords with 8 bits per sample and generates linear PCM codewords (uniformly quantized) with a large number of bits per sample (13 bits in case of A compression law and 14 bits in case of mu compression law). It follows the process of demodulating the linear PCM modulated signal resulting the signal samples and then it is generated the analog demodulated signal by applying the samples and hold circuit followed by a low pass or band pass filtering.

In Fig. 6.5 it is presented a possible solution (not the one specified by the ITU-T G.711 standard) for digital implementation of the compression and expansion process, that is to say a PCM modulation with companding. The solution provided allows a better understanding of the of digital implementation technique of the companding (compression and expansion) process.

The source signal is modulated using linear PCM with 16 bits per sample (i.e. 16 bits PCM codeword), then the PCM coded samples are applied to a digital module that implements the compression function, obtaining 16 bits PCM codewords with companding. Then it follows a process of reducing the number of bits per PCM codeword from 16 to 8 bits, which is achieved by suppressing the last 8 bits (see Fig. 6.6). This last process is practically a quantization process.

At the reception, the number of bits per sample, i.e. per PCM codeword, is first extended from 8 to 16 bits by adding to the 8 bit codeword a bit of 1 followed by 7 bits of 0 (see Fig. Ref fig PCM-fig6). This is also a quantization process. The 16 bit PCM codewords are applied to a digital module that implements the expansion function, resulting the 16 bit linear PCM codewords, which are then decoded.

In conclusion, the digital implementation of the compression process involves applying the compression function to linear/uniform PCM encoded samples represented on a large number of bits followed by a quantization process, that is, decreasing the number of bits per sample/PCM codeword. The digital implementation of the expansion process also involves a quantification process, i.e. increasing the number of bits per sample/PCM codeword, followed by applying the expansion function to PCM codewords with compression.

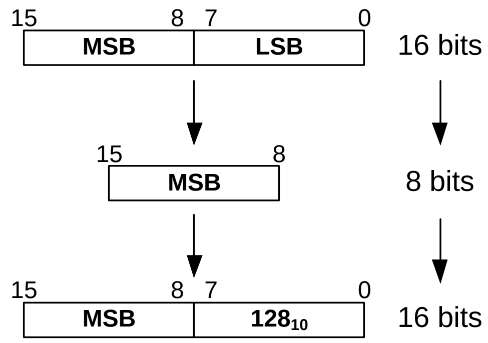

Figure 6.6: Reducing respectively increasing the number of bits per PCM codeword.

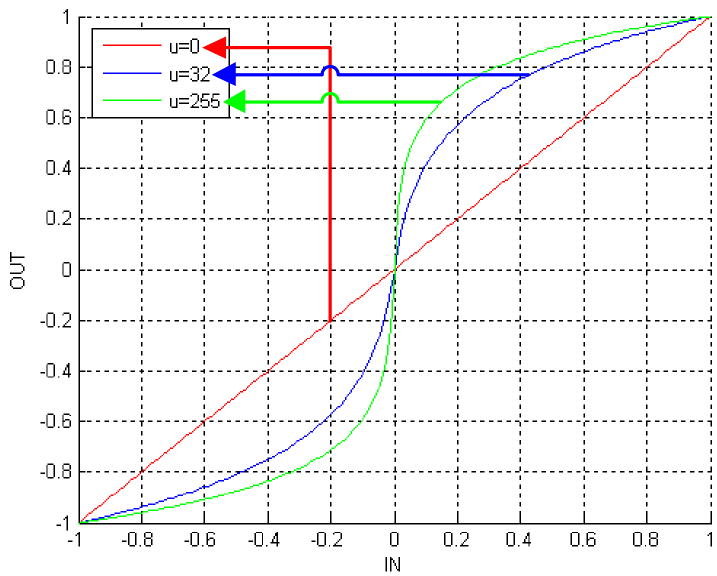

Figure 6.7: The normalized μ compression function for different values of the μ parameter.

Note: at the destination, switching from 8 to 16 bits per sample is accomplished by replacing the 8 LSB bits eliminated at transmission with the value 10000000_2 (128_{10}), which halves the maximum value that can take the quantization error associated with the quantification process that took place at transmission.

6.2.1 μ compression law

As mentioned above the μ compression law (6.1) is used in digital telephone systems in the United States of America and Japan, the compression law parameter having the value $\mu = 255$.

In Fig. 6.7 it is is presented the normalized μ compression function for various values of parameter μ, and in Fig. 6.8 it is presented the segmented normalized μ compression function, that is approximated by segments (only the positive side of the function is presented). The parameter $\mu = 0$ corresponds to the situation without compression.

For implementation reasons the compression function ($\mu = 255$) is approximated with 16 segments, 8 on the positive side and 8 on the negative one (Fig. 6.8), and each segment is divided into 16 subsegments - see Fig. 6.9 which shows the first positive segment of the normalized compression function. On the y axis of the compression function all the segments have the same width and an uniform quantization on 8 bits is achieved. Within each segment on the x axis, a 4 bit uniform quantization is performed. The quantization step is constant within a segment, but varies from segment to segment, i.e. it increases with the index of the segment - thus a nonuniform quantization is achieved.

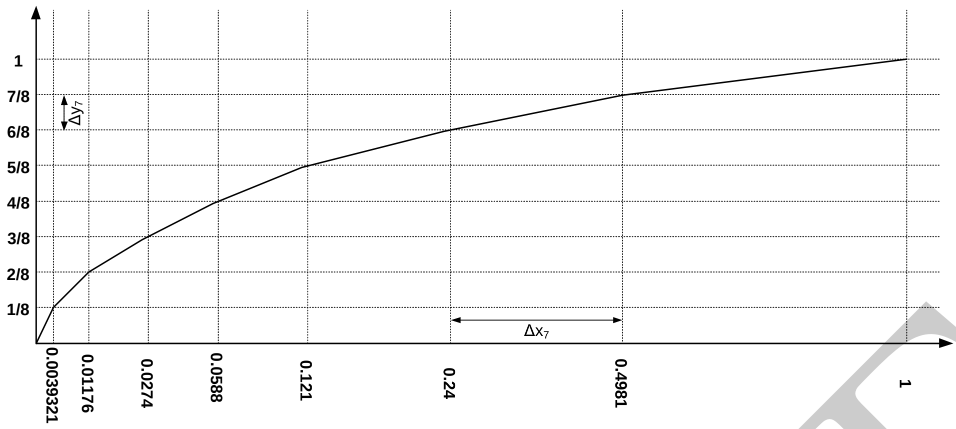

Figure 6.8: Segmented mu compression function - the positive side.

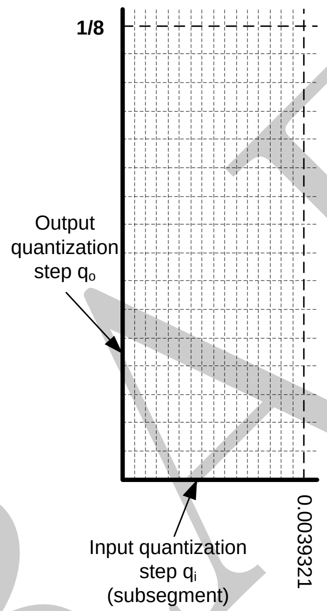

Figure 6.9: The first positive segment of the segmented μ compression function. Division into subsegments.

Note: for a more suggestive and easier to follow graphical representation, the segments on the x axis of Fig. 6.8 are not on the scale.

Based on the previous observations, the coding of the quantized samples of the input signal is done as follows (see Fig. 6.10):

- the most significant bit, b_7, indicates the polarity/sign of the sample;
- the next 3 bits, b_6, b_5, b_4 indicate the segment in which the sample falls;
- the last 4 bits, b_3, b_2, b_1, b_0, indicate the subsegment in which the sample falls.

μ compression function parameters

In the following are given the relations for calculation of the μ compression law parameters. It should be emphasized that the parameters defined below characterize any compression function, only the values of these

b_7	b_6	b_5	b_4	b_3	b_2	b_1	b_0
Sign	Segment			Subsegment			

Figure 6.10: The coding rule of the quantized samples.

parameters differ.

Compression rate on segment i:

$$Rc_i = \frac{\Delta x_i}{\Delta y_i} \tag{6.3}$$

where Δx_i represents the size of the input segment i, and Δy_i the size of the output segment i (see Fig. 6.8), which equals 0.125 for any output segment. When the compression ratio is less than 1 means that expansion is done, and when it is greater than 1 compression is performed.

The input quantization step on segment i represents the size of a subsegment of the input segment i and can be computed using the relation:

$$q_i = \frac{\Delta x_i}{\text{No. sub-segments}} = \frac{\Delta x_i}{16} \tag{6.4}$$

The output quantization step represents the size of a subsegment of an output segment and is computed by the relation:

$$q_o = \frac{\Delta y_i}{\text{No. sub-segments}} = \frac{0.125}{16} = 0.0078125 \tag{6.5}$$

The elementary quantization step is the step with which the source signal should be uniformly quantized in order to achieve the same performance (reception quality) as in the case of nonuniform quantization (14 bits in the case of μ law).

$$q_e = \frac{2}{2^b} = \frac{2}{2^{14}} = 0.00012207 \tag{6.6}$$

The number of elementary quantization steps of an input subsegment:

$$n_i = \frac{q_i}{q_e} \tag{6.7}$$

The power of quantization noise in segment i [53]:

$$Pzq_i = \frac{q_i^2}{12} \tag{6.8}$$

The total power of the quantization noise [4]:

$$Pzq = \sum_{i=-8 i\neq 0}^{8} p_i \cdot Pzq_i \tag{6.9}$$

where p_i represents the probability that the sample amplitude falls into segment i. If we have the same probability of occurrence of any sample (uniform distribution of sample amplitudes) then the probability that the sample is located in segment i is equal to the size of segment i, in the case of the normalized function, that is $p_i = \Delta x_i/2$. In the case of the non-normalized compression function, with a dynamic range of the signal $-V \div V$, we have $p_i = \Delta x_i/2V$.

Taking into account the antisymmetry of the compression function the equation of the quantization noise power (6.9) can be transformed into:

$$Pzq = \sum_{i=1}^{8} 2p_i \cdot Pzq_i = \sum_{i=1}^{8} \Delta x_i \cdot Pzq_i \tag{6.10}$$

Note: if we calculate the total power of the quantization noise for the μ compression law we obtain $Pzq = 47.39 \cdot 10^{-6} W$, and if we calculate the power of the quantization noise in the case of uniform quantization with the same number of bits, 8 bits, we get $Pzq_{uniform} = 5 \cdot 10^{-6} W$. Although the power of the quantization noise in the case of uniform quantization is lower than the power of the total quantization noise in the case of the nonuniform quantization, it can be observed, in Tab. 6.1, that the power of the quantization noise in the case

Table 6.1: The values of the parameters that characterize the μ compression law.

Segment i	Δx_i	Rc_i	q_i	n_i	Pzq_i
1	0.003921	0.031368	0.000245063	2.007552	5.00464E-09
2	0.007839	0.062712	0.000489938	4.013568	2.00032E-08
3	0.01564	0.12512	0.0009775	8.00768	7.96255E-08
4	0.0314	0.2512	0.0019625	16.0768	3.20951E-07
5	0.0622	0.4976	0.0038875	31.8464	1.25939E-06
6	0.119	0.952	0.0074375	60.928	4.6097E-06
7	0.2581	2.0648	0.01613125	132.1472	2.16848E-05
8	0.5019	4.0152	0.03136875	256.9728	8.19999E-05

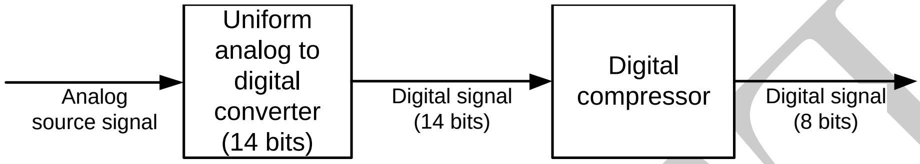

Figure 6.11: The PCM modulation process with compression according to the μ law.

of uniform quantization is much greater than the power of the quantization noise in the case of the nonuniform quantization for the first segments, that is for the small levels of the voice signal, where the ear is more sensitive to quantization error. The quantization error for nonuniform quantization is higher at high levels of the voice signal, where the noise is not so noticeable, but it affects the average/total quantization signal-to-noise ratio (SNR) that is important for data transmissions.

μ law coding process

In the previous sections it was explained how the companding (compression and expansion) can be done in the digital domain (see Fig. 6.5). In the following will be presented and analyzed a concrete solution for the implementation of the companding in the digital domain, solution described also in the ITU-T G.711 standard [51]. In contrast to the solution presented in Fig. 6.5 the compression and the quantization process are grouped together in a single process optimized in terms of required processing effort.

As shown in Fig. 6.11, the voice signal is applied at the input of a 14-bit linear/uniform analog to digital converter, at the output of which 14-bit codewords are obtained which encode the samples of the input signal - linear/uniform PCM signal. The digital compressor generates, according to the compression function, the 8-bit code that will be transmitted on the line (PCM signal with compression), corresponding to the 14-bit code from the digital compressor input.

The digital compressor in Fig. 6.11 performs a quantization process, i.e. grouping 14-bit codes generated by the uniform A/D converter and replacing these groups with 8-bit binary codes. The process is illustrated in the table presented in Fig. 6.12, table taken from ITU-T G.711 standard [51]. The process of grouping the 14-bit codes is based on the values of parameter n_i in Tab. 6.1, parameter that has the value 2 for the first segment, the value 4 for the second segment and doubles in the following segments, reaching the value 256 for the last segment (the eighth segment). This means that in the first segment the 14-bit codes generated by the uniform A/D converter are grouped 2 by 2, in the second segment are grouped 4 by 4 and so on, and in the last segment are grouped 256 by 256. The grouping process described can be seen in Fig. 6.12 in column 2, and in column 3 are given the 14-bit codes (i.e. their decimal values) lower delimiting the positive segments. Of course, it can be easily determined the values on 14-bit (represented in decimal) that delimit each segment and subsegment. For each subsegment represented by a group of 14-bit codes (the subsegments in the i segment are represented by a group of n_i consecutive 14-bit codes), an 8-bit code is assigned, the codes given in column 6 of the table in Fig. 6.12. These codes are obtained according to the coding rule presented in Fig. 6.10, after which an inversion of the bits of the 8-bit PCM code occurs.

Note: there is defined in ITU-T G.711 standard a separate table for coding negative signal levels.

TABLEAU 2a / G.711

μ-law, positive input values

1	2	3	4	5	6	7	8
Segment number	Number of intervals × interval size	Value at segment end points	Decision value number n	Decision value x_n (see Note 1)	Character signal Bit number 1 2 3 4 5 6 7 8	**Quantized value (value at decoder output) y_n**	Decoder output value number
		8159	(128)	(8159)			
			127	7903	1 0 0 0 0 0 0 0	8031	127
8	16 × 256				(see Note 2)		
			113	4319			
					1 0 0 0 1 1 1 1	4191	112
		4063	112	4063			
7	16 × 128				(see Note 2)		
			97	2143			
					1 0 0 1 1 1 1 1	2079	96
		2015	96	2015			
6	16 × 64				(see Note 2)		
			81	1055			
					1 0 1 0 1 1 1 1	1023	80
		991	80	991			
5	16 × 32				(see Note 2)		
			65	511			
					1 0 1 1 1 1 1 1	495	64
		479	64	479			
4	16 × 16				(see Note 2)		
			49	239			
					1 1 0 0 1 1 1 1	231	48
		223	48	223			
3	16 × 8				(see Note 2)		
			33	103			
					1 1 0 1 1 1 1 1	99	32
		95	32	95			
2	16 × 4				(see Note 2)		
			17	35			
					1 1 1 0 1 1 1 1	33	16
		31	16	31			
	15 × 2				(see Note 2)		
			2	3			
1					1 1 1 1 1 1 1 0	2	1
			1	1			
↓	1 × 1				1 1 1 1 1 1 1 1	0	0
			0	0			

Note 1 – 8159 normalized value units correspond to T_{max} = 3.17 dBm0.

Note 2 – The character signal corresponding to positive input values between two successive decision values numbered n and $n + 1$ (see column 4) is $(255 - n)$ expressed as a binary number.

Note 3 – The value at the decoder output is $y_0 = x_0 = 0$ for $n = 0$, and $y_n = \frac{x_n + x_{n+1}}{2}$ for $n = 1, 2, \ldots, 127$.

Note 4 – x_{128} is a virtual decision value.

***Note 5* – In Tables 1/G.711 and 2/G.711 the values of the uniform code are given in columns 3, 5 and 7.**

Figure 6.12: The tabular coding rule for μ compression law according to ITU-T G.711 standard.

Note: the 8-bit PCM code in column 6 of the table in Fig. 6.12 is the one after the bit inversion. The sign bit (before bit inversion) for positive samples has the value 0, and for negative samples it has the value 1.

The inversion of the bits of the companded PCM code is necessary because the voice signal level does not have a uniform distribution, the probability of occurrence of the small levels being much higher than that of the high levels of the signal. From this it follows that there is a greater probability that a PCM code will contain more zero bits (the code of the segments corresponding to the small levels contains more zero bits). The line code used in the case of the μ compression law is B8ZS (Bipolar with 8 Zeros Substitution), which is a variant of the AMI (Alternate Mark Inversion) code and which replaces the sequences of 8 consecutive zero bits with violations of the coding rule. Because the AMI code encodes 0 bits with zero level, long sequences of 0 affect the synchronization of the bit/symbol clock. Although the B8ZS code replaces the sequences of 8 consecutive 0 bits, the possibility of sequences of maximum 7 consecutive 0 bits remains. Because of this and considering that the probability of occurrence of codes containing more zero bits is higher, the 8 bits obtained at the output of the digital compressor are reversed before transmission on the line.

It can be noticed (see Fig. 6.12) that after the bit inversion the code obtained for the first positive subsegment (sign: 0, seg: 000, sub-seg: 0000) is $11111111b$ (FFh) and in the silence periods during a call, a sequence of continuous one bits will be transmitted, sequence which can be confused with the blue alarm indicator. For this reason this code ($11111111b$) is not transmitted on the line and instead the code of the first negative subsegment (sign: 1, seg: 000, subseg: 0000) is transmitted, whose code obtained after bit inversion is $01111111b$ ($7Fh$). The latter is the code transmitted during the silence periods. Due to this aspect the coding of the first segment is done separately, i.e. we have a separate subsegment for the zero signal level, a subsegment consisting of a single 14-bit code (0 bits), the next 15 subsegments being formed by grouping two 14-bit codes. (see Fig. 6.12, column 2).

The expansion process is also based on the table presented in Fig. 6.12. Based on the 8-bit PCM code received on the line, it is possible to identify the segment and the subsegment as well as the threshold values (on 14 bits) that delimit the subsegment. The 14-bit code generated by the expansion circuit represents the middle of the subsegment, that is, the 14-bit code located in the middle of the group of consecutive 14-bit codes that correspond to the subsegment, or in other words it represents the arithmetic mean of the 14-bit values that delimit the subsegment (see Fig. 6.12, column 7). The 14-bit code obtained applies to a uniform D/A converter.

The implementation of the digital compressor requires a series of operations to compare the value of the 14-bit code, generated by the uniform A/D converter, with the threshold values that identify the segments and subsegments. Based on simple mathematical operations, a coding table is generated Tab. 6.2 [54] which allows an easy conversion of the 14-bit code generated by the uniform A/D converter into the 8-bit PCM code sent on the line. The table can also be used in the decoding operation of the PCM code received on the line. A first simplification of the operations of converting the 14-bit code into the 8-bit code consists in adding to the 14-bit code the value 33 decimal. This operation is meant to make the ends of the segments be powers of 2 (see Fig. 6.12 column 3: 31+33=64, 95+33=128, 223+33=256,...). Following this operation, it can be observed in the coding table (Tab. 6.2) that the identification of the segment in which the sample is located, sample which is applied to the input of the digital compressor, can be made based on the position of the first 1 bit (from MSB to LSB) after the sign bit has been extracted. The subsegment code can be obtained from the bits that follow after the first 1 MSB bit.

Note: adding the 33 decimal value to the 14-bit code has the undesirable effect of reducing the useful dynamic range of the uniform A/D converter from 8192 to 8159.

We observe in Tab. 6.2 how:

- the position of the first bit of 1 indicates the segment's code:

  COD SEGMENT$_{10}$=7-no. zeros in front of the first bit of 1 = -5 + the position of the first bit of 1;

- and the subsegment code represents the following 4 bits, after the first bit of 1.

Table 6.2: Coding table for the μ law .

Digital compressor input (after adding the decimal value 33)														Code with compression							
															Segment			Subsegment			
bit	12	11	10	9	8	7	6	5	4	3	2	1	0	bit	6	5	4	3	2	1	0
	0	0	0	0	0	0	0	1	a	b	c	d	x		0	0	0	a	b	c	d
	0	0	0	0	0	0	1	a	b	c	d	x	x		0	0	1	a	b	c	d
	0	0	0	0	0	1	a	b	c	d	x	x	x		0	1	0	a	b	c	d
	0	0	0	0	1	a	b	c	d	x	x	x	x		0	1	1	a	b	c	d
	0	0	0	1	a	b	c	d	x	x	x	x	x		1	0	0	a	b	c	d
	0	0	1	a	b	c	d	x	x	x	x	x	x		1	0	1	a	b	c	d
	0	1	a	b	c	d	x	x	x	x	x	x	x		1	1	0	a	b	c	d
	1	a	b	c	d	x	x	x	x	x	x	x	x		1	1	1	a	b	c	d

For example, if at the input of the compressor, after adding the decimal value 33, we have the binary code 10000111011100 we can determine the segment code as follows: SEGMENT $\text{CODE}_{10} = 7 - 4 = 3 \Rightarrow$ SEGMENT $\text{CODE}_2 = 011$, and the subsegment code is represented by the following 4 bits: SUBSEGMENT $\text{CODE}_2 = 1101$. The sign bit remains unchanged: CODE $\text{SIGN}_2 = 1$. Thus the code obtained is 10111101_2. The cod is inverted and transmitted on the line. At the destination the code taken from the line is again inverted and the code 10111101_2 is obtained. Based on the segment code we determine the position of the first bit of 1, and we obtain $00001xxxxxxxx$, after which we fill the following 4 bits with the subsegment code, $000011101xxxx$. The last 4 bits remain to be filled, bits for which we have no information. Thus, to halve the maximum quantization error, we choose the middle value of the domain that could be represented on the last bits (more precisely the middle of the subsegment - see the previous discussion on expansion using the table in Fig. 6.12, respectively the discussion regarding the operations presented in Fig. 6.6) and we obtain the code 0000111011000_2, to which is added the sign bit, 10000111011000_2.

6.2.2 A law coding process

As mentioned above, the A compression law (6.2) is used in digital telephone systems in Europe. The compression characteristic parameter has the value $A = 87.6$, the value for which the continuity of the compression and expansion functions is obtained.

In Fig. ref fig: pcm-fig13 it is presented the segmented normalized A compression function, that is, approximated by segments (only the positive side of the function is presented) [51]. It can be observed that in case of A law the dimensioning of the segments is changed, the segments of the normalized characteristic are delimited by values in the form of $1/2^n, n = 6, ..., 0$, that is 1/64, 1/32,...,1/4, 1/2, 1, and the first and second segments are located on the same line, which leads to a total of only 7 segments, both on the positive and negative sides. The first segment of the compression function is composed of 32 subsegments, and the remaining segments are composed of 16 subsegments, but the first segment can be divided into two identical segments, each of 16 subsegments and thus the structure of the PCM code generated by the compressor that works according to the A law remains identical to that of the μ law.

Note: the segments on the x axis of Fig. 6.13 are represented to scale.

The PCM coding process with compression is also done in this case according to the processing chain shown in Fig. 6.11, but the uniform quantization is only on 13 bits. The compression process, i.e. the conversion from the 13-bit code generated by the uniform A/D converter to the 8-bit PCM code with compression is done according to the table presented in Fig. 6.14, table taken from ITU-T standard G.711 [51]. The operations performed are similar to those presented in the implementation of the μ compression law, the differences consisting of the following: the number of segments is reduced to 7 (Fig. Ref fig: pcm-fig14 column 1), the segments are identified (segment ends) by values which are power of two (column 3), i.e. 64, 128, ..., 2048, 4096, the 8-bit codes generated by the compressor are inverted only at the even positions before the transmission on the channel

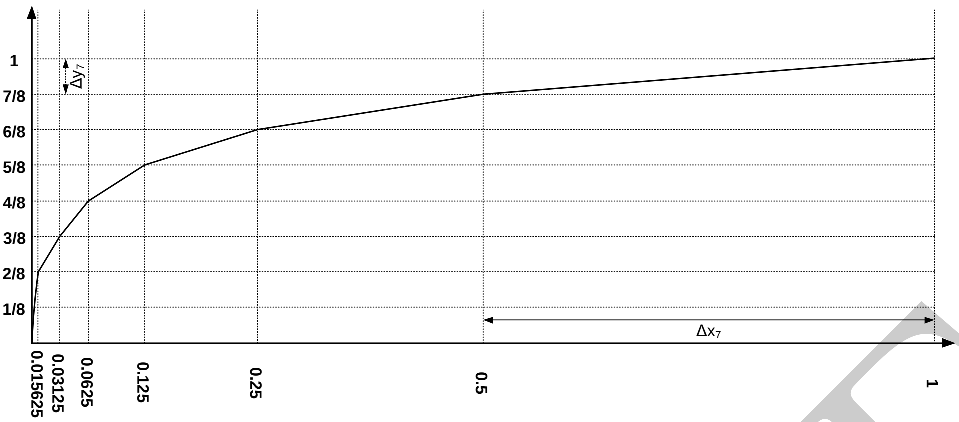

Figure 6.13: Segmented A compression function - the positive side.

Table 6.3: Coding table for the A law.

Digital compressor input													Code with compression							
														Segment			Subsegment			
bit	11	10	9	8	7	6	5	4	3	2	1	0	bit	6	5	4	3	2	1	0
	0	0	0	0	0	0	0	a	b	c	d	x		0	0	0	a	b	c	d
	0	0	0	0	0	0	1	a	b	c	d	x		0	0	1	a	b	c	d
	0	0	0	0	0	1	a	b	c	d	x	x		0	1	0	a	b	c	d
	0	0	0	0	1	a	b	c	d	x	x	x		0	1	1	a	b	c	d
	0	0	0	1	a	b	c	d	x	x	x	x		1	0	0	a	b	c	d
	0	0	1	a	b	c	d	x	x	x	x	x		1	0	1	a	b	c	d
	0	1	a	b	c	d	x	x	x	x	x	x		1	1	0	a	b	c	d
	1	a	b	c	d	x	x	x	x	x	x	x		1	1	1	a	b	c	d

(there are not inverted all bits as in the case of the μ law).

Note: the 8-bit PCM code from column 6 of the table in Fig. 6.14 is the one before the bit inversion on the even positions. The sign bit for positive samples has the value 1, and for negative samples it has the value 0.

The optimization of the compression and expansion process implementation can be achieved with the help of the coding table Tab. 6.3 [54]. Adding the decimal value 33 or any other value is no longer necessary because the segment ends are powers of 2 (column 3, Fig. 6.14). The coding is done according to the coding table as in the case of the μ compression law. The line code that is used is HDB3 (High Density Bipolar 3 Zeros), which is also a modified AMI code, which replaces sequences of 4 consecutive zero bits with violations of the coding rule.

It can be seen that the only differences that appears in the coding table compared to the μ law are in the first two segments. At the first segment cannot be identified a first bit of 1 MSB, but only 7 bits of 0 MSB. Starting with segment two, a first bit of 1 MSB can be identified, preceded by a number of bits of 0 MSB. The rule for determining the segment's code remains the one presented in the case of the coding table corresponding to the μ law. The subsegment code is determined in the same way as in the case of the μ law, meaning the subsegment code is given by the four bits that follow after the first bit of 1 MSB. A special situation is also at the first segment (where we do not have a first bit of 1 MSB) where the subsegment code is given by the four bits that follow after the first 7 bits of 0 MSB.

6.3 Delta modulation

Delta modulation is a particular case of differential PCM (DPCM) modulation. DPCM modulation quantifies the prediction error of the current sample instead of the value of the current sample, which has the consequence

TABLE 1a/G.711

A-law, positive input values

1	2	3	4	5	6	7	8
Segment number	Number of intervals × interval size	Value at segment end points	Decision value number n	Decision value x_n (see Note 1)	Character signal before inversion of the even bits Bit number 1 2 3 4 5 6 7 8	Quantized value (value at decoder output) y_n	Decoder output value number
		4096	(128)	(4096)			
					1 1 1 1 1 1 1 1	4032	128
7	16 × 128		127	3968	(see Note 2)		
			113	2176			
					1 1 1 1 0 0 0 0	2112	113
		2048	112	2048			
6	16 × 64				(see Note 2)		
			97	1088			
					1 1 1 0 0 0 0 0	1056	97
		1024	96	1024			
5	16 × 32				(see Note 2)		
			81	544			
					1 1 0 1 0 0 0 0	528	81
		512	80	512			
4	16 × 16				(see Note 2)		
			65	272			
					1 1 0 0 0 0 0 0	264	65
		256	64	256			
3	16 × 8				(see Note 2)		
			49	136			
					1 0 1 1 0 0 0 0	132	49
		128	48	128			
2	16 × 4				(see Note 2)		
			33	68			
					1 0 1 0 0 0 0 0	66	33
		64	32	64			
1 ↓	32 × 2				(see Note 2)		
			1	2			
					1 0 0 0 0 0 0 0	1	1
			0	0			

Note 1 – 4096 normalized value units correspond to T_{max} = 3.14 dBm0.

Note 2 – The character signals are obtained by inverting the even bits of the signals of column 6. Before this inversion, the character signal corresponding to positive input values between two successive decision values numbered n and $n + 1$ (see column 4) is $(128 + n)$ expressed as a binary number

Note 3 – The value at the decoder output is $y_n = \frac{x_{n-1} + x_n}{2}$ for n = 1, ..., 127, 128.

Note 4 – x_{128} is a virtual decision value.

Note 5 – **In Tables 1/G.711 and 2/G.711 the values of the uniform code are given in columns 3, 5 and 7.**

Figure 6.14: The tabular coding rule for A compression law according to ITU-T G.711 standard.

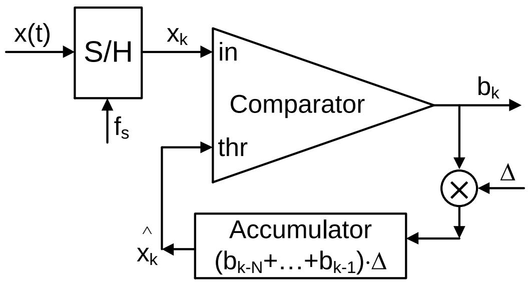

Figure 6.15: Block diagram of the linear Delta modulator.

of reducing the number of bits per sample and implicitly reducing the transmission rate. Delta modulation quantizes the value of the prediction error of the current sample on a single bit, i.e. only information related to the prediction error sign is generated and transmitted [52].

Basically the Delta modulation is an analog to digital, respectively digital to analog, conversion technique of the voice signal with a single bit per voice sample. The reduced number of bits/sample leads to a degradation of the quality of the reconstructed analog voice signal (high quantization noise, as well as the distortion of the reconstructed analog signal). For this reason, Delta modulation can only be used in voice transmission/processing systems where the signal quality is not the most important criterion, but where the transmission rate should be as low as possible. For Delta modulation to be usable, a strong correlation between consecutive samples of the signal is required.

If the value of the quantization step used is constant we are talking about a linear or fixed Delta modulation, and if the value of the quantization step changes depending on the signal dynamics we are talking about adaptive Delta modulation.

6.3.1 Linear Delta modulation

As mentioned above, the DPCM type modulations quantify the error (difference) between the predicted value of the current sample and the actual value of this sample, and therefore it is necessary to generate a predicted signal based on the previous samples/values of the signal. In the case of linear Delta modulation, the updating of the predicted signal (more precisely of the predicted signal samples) is done on the basis of a mechanism independent of the law of variation of the signal samples (more precisely of the previous samples of the signal) [52]. The block diagram of a linear Delta modulator/encoder is shown in Fig. 6.15.

The analog signal $x(t)$ is transformed by the linear Delta modulator into a string of binary symbols that are transmitted on the communication channel. The S/H (Sample&Hold) module the diagram is the sampling and hold circuit. The sampling frequency f_e obeys the relation $f_e \gg B$, where B is the frequency band of the analog signal. The comparator continuously compares the samples of the analog signal $x(t)$ with those of the predicted signal $\hat{x}_k$ and on the basis of this comparison generates at the sampling moment the bit b_k. The accumulator has the role of generating the predicted signal that tracks the analog signal $x(t)$. It can be implemented analogically using an integrator or as a digital circuit using an up/down counter followed by a D/A converter.

At each sampling moment, the sign of the difference between the value of the input signal $x(t)$ and the value of the predicted signal $\hat{x(t)}$ is calculated. So, the current bit, b_k, is calculated according to the following relation:

$$\begin{cases} x_k \geq \hat{x}_k \Rightarrow b_k =' 1'(+1) \\ x_k < \hat{x}_k \Rightarrow b_k =' 0'(-1) \end{cases} \tag{6.11}$$

Depending on the sign of the difference between the samples x_k and $\hat{x(t)}$, the command of increasing or decreasing the following predicted sample with a value Δ, called the approximation quantum or quantization

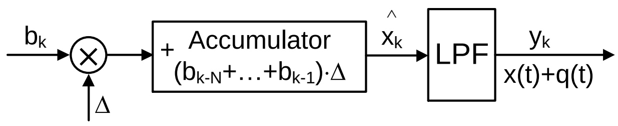

Figure 6.16: Block diagram of the linear Delta demodulator.

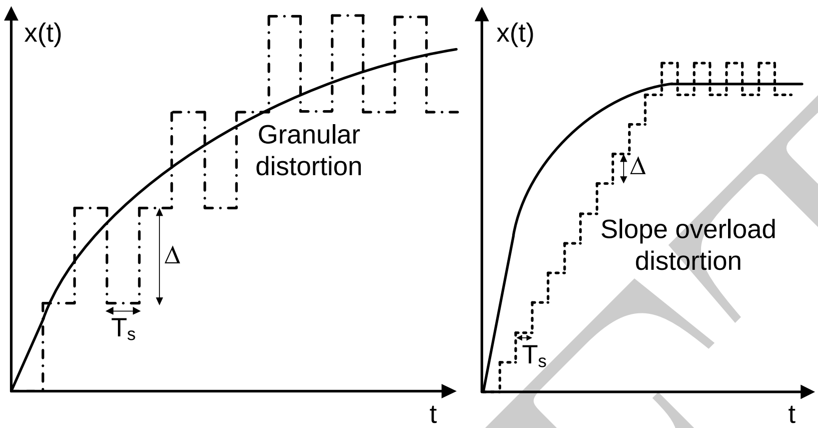

Figure 6.17: Distortions characteristic to Delta modulation.

step, is issued. The calculation of the predicted signal (the equation of the accumulator) is performed according to the relation:

$$\hat{x}_k = \hat{x}_{k-1} + b_{k-1} \cdot \Delta \tag{6.12}$$

In the case of the linear Delta modulation, the quantization step is constant.

The block diagram of the linear Delta demodulator/decoder is shown in Fig. 6.16. It can be seen that the Delta demodulator includes the prediction circuit, which generates the predicted signal samples based on the bits received from the channel, and a low pass filter that filters the predicted signal samples. The filter ensures the conversion from the sampled signal domain to the continuous analog signal domain.

6.3.2 Distortions characteristic to Delta modulation

Delta modulation (whether we talk about fixed or adaptive Delta modulation) is affected by two types of quantization errors, namely [52]:

- slope overload distortion, which occurs when the analog source signal ($x(t)$) variation rate (i.e. the slope of the signal) is higher than the predicted signal (the $\hat{x}(t)$ signal in steps) variation rate .

$$\begin{cases} \text{slope source signal} = |dx(t)/dt| \\ \text{slope Delta signal } = |\Delta/T_s| \end{cases} \tag{6.13}$$

  where T_s is the sampling period.

- the granular distortion/noise, which occurs when the variation rate of the analog source signal is lower than the variation rate of the predicted signal.

In Fig. 6.17 it is presented graphically how the two distortions affect the Delta modulation.

In order to avoid the occurrence of slope overload distortions, the slope of the two signals (the source signal and the predicted signal) should be equal, which is impossible to achieve in the case of the linear Delta modulation that works with a constant quantization step.

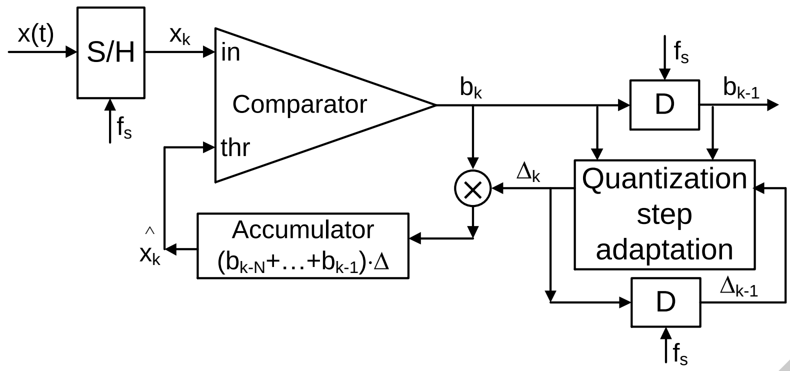

Figure 6.18: Block diagram of the adaptive Delta modulator.

If the quantization step is large, the slope overload distortions may be smaller (depending on the signal dynamics), but the granular distortions will be larger. If the quantization step is small, the granular distortions will be smaller, but the slope overload distortions will be larger (depending also in this case on the signal dynamics).

The power of quantization noise in the case of linear Delta modulation, if it is considered that the slope overload distortion is corrected, is $P_{zq} = \Delta^2/13$, and the signal-to-noise ratio is P_s/P_{zq}, where P_s is the power of the analog source signal.

6.3.3 Adaptive Delta modulation

Improving the quality of the Delta transmission, i.e. reducing the quantization noise as well as the distortions affecting the analog signal reconstructed at the reception, can be achieved, without increasing the transmission rate, by using the adaptive Delta modulation. In this case the quantization step is no longer fixed, but changes adaptively depending on the dynamics of the source signal. On the slow variable portions of the source signal small quantization steps are used, while on the rapidly variable portions of the source signal large quantization steps are used. Thus, it is possible to ensure both the better tracking of the slope of the source signal and the reduction of the quantization noise. The measurement (more correctly the evaluation) of the slope of the source signal can be done by analyzing the sequence of modulated bits generated by the Delta modulator. Specifically, the quantization step at one sampling time (instantaneous step) is calculated based on the quantization step from the previous sampling moment, the bit generated at the current moment and a number of bits generated at the previous sampling moments, according to the following relation:

$$\Delta_k = f(\Delta_{k-1}, b_k, b_{k-1}, \ldots, b_{k-N}) \tag{6.14}$$

In Fig. 6.18 it is presented the block diagram of an adaptive Delta modulator, for the particular case in which the evaluation of the source signal slope is based on the current and previous bit. The difference to the linear Delta modulator diagram, see Fig. 6.15, consists of the quantization step calculation block, the prediction of the current sample being achieved also with the help of a accumulator that contains the previous samples.

In Fig. 6.19 it is presented the block diagram of an adaptive Delta demodulator, also for the particular case in which the evaluation of the source signal slope is based on the current bit and the previous bit. As in the case of the linear Delta demodulator, see Fig. 6.16, we have the accumulator that calculates the predicted sample from the current moment, based on the previous samples, and then the predicted samples are low pass filtered. The current quantization step (which weighs the current received bit) is computed as in the case of the modulator, based on the quantization step from the previous sampling moment and a number of consecutive received bits.

The adaptive Delta modulation algorithms differ according to the adaptation (calculation) mode of the quantization step. In the following two simple examples of quantization step adaptation/modification

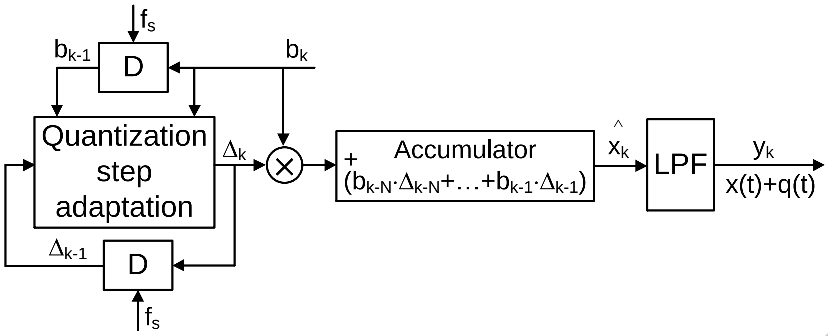

Figure 6.19: Block diagram of the adaptive Delta demodulator.

algorithms, namely an additive modification (Song algorithm), respectively a multiplicative modification (Jayant algorithm) of the quantization step are presented [55] [56].

Song algorithm

The rule for modifying the quantization step is the following:

$$\begin{cases} \Delta_k = \Delta_{k-1} + \Delta_e \text{ if } b_k = b_{k-1} \\ \Delta_k = \Delta_{k-1} - \Delta_e \text{ if } b_k \neq b_{k-1} \\ \text{if } \Delta_k < \Delta_e \Rightarrow \Delta_k = \Delta_e \end{cases} \tag{6.15}$$

where Δ_e is the elementary quantization step.

Compared to the linear Delta modulation, the Song modulation decreases both types of distortions, but the granular distortion cannot be completely canceled, as the slope tracking is not perfect either.

Jayant algorithm

The rule for modifying the quantization step is: $\Delta_k = \Delta_{k-1} p^{sgn(b_k b_{k-1})}$, meaning:

$$\begin{cases} \Delta_k = \Delta_{k-1} p \text{ dacă } b_k = b_{k-1} \\ \Delta_k = \Delta_{k-1}/p \text{ dacă } b_k \neq b_{k-1} \end{cases} \tag{6.16}$$

The variation of the quantization step is dictated by the p factor. In order to obtain a good approximation of the source signal, for a large class of analog signals, it is required that the p factor to be chosen as follows: $1 \leq p \leq 2$. If necessary, limits can be imposed also in this case for the minimum or maximum value of the quantization step.

The algorithm can ensure better tracking of the steep slope signals, but as in the case of the Song algorithm the granular distortion cannot be completely canceled.

Note: for the significant reduction of the distortions affecting the Delta modulation, is necessary a quantization step adaptation algorithm which is a combination of the Song and Jayant algorithms, mentioned above, and the specific law of variation of the source signal, such as the voice signal in the case of telephone systems, must be taken into account.

6.4 Analysis and simulation software

In order to understand the principles of the PCM and Delta modulation techniques, respectively to evaluate the performances of these voice coding mechanisms, several evaluation and simulation software have been developed. The performance evaluation software use as test signals voice signals (voice files) and allow to listen

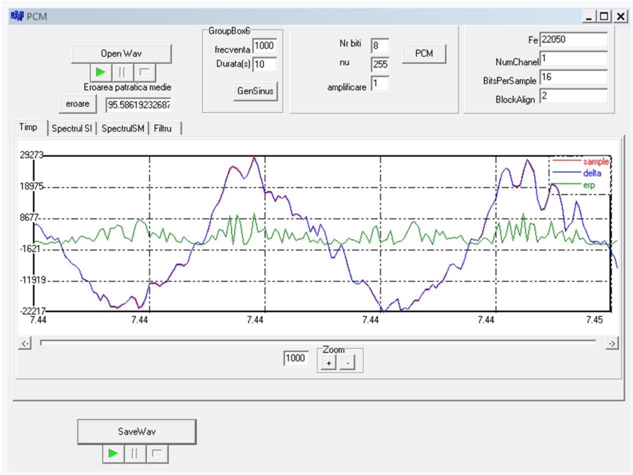

Figure 6.20: The main window of the *PCM* simulation software.

to the modulated and then demodulated voice signals, being possible the subjective evaluation of the effects of the quantization noise and other distortions (slope overload, distortions due to filtering, etc.). The simulation software use simple test signals (basic periodic signals: sine, triangle, trapezoid) and allow the visualization of the signals that appear in different stages of the modulation and demodulation processes. It is also possible the spectral analysis of the mentioned signals as well as the evaluation of the distortions introduced by the modulation/demodulation processes.

PCM evaluation and simulation software

The *PCM* software allows the simulation of the uniform PCM modulation respectively of the PCM modulation with companding according to the μ law using voice test signals or a sine test signal with adjustable frequency and duration. The software allows at the same time to evaluate the performances of the uniform PCM modulation respectively of the PCM modulation with companding in conditions close to reality. The parameters of the PCM modulation that can be set are: the number of bits per sample, the value of μ parameter (if set $\mu = 0$ the uniform PCM modulation is obtained) and the gain/attenuation of the source signal.

The software allows the loading of a $*.wav$ type sound/voice file and as a result of this process the samples of the test signal are displayed, as well as its parameters, such as: sampling frequency, signal type (mono or stereo), number of bits/sample. The uploaded file can be played with the player application integrated in the simulation software. The obtained PCM signal can also be saved in $*.wav$ format in a separate file and then can be played using the in-app player. The application also allows the use of a sinusoidal source signal, especially useful for visualizing PCM modulated signals with different parameters. The main window of the *PCM* software is shown in Fig. 6.20.

The *PCM* software allows to display the error between the source signal and the signal obtained after the PCM modulation and then the PCM demodulation, that is to say the quantization noise that affects the demodulated signal. The software displays the mean squared value of the error, that is, the value of the power of the quantization noise. The software also has a zoom feature for a better visualization of the demodulated PCM signal and of the quantization error/noise.

The *PCM* allows to visualize the spectrum of the source signal as well as of the modulated and then demodulated PCM signal (see the *Spectrul SI* and *Spectrul SM* buttons in the main window of the software - Fig. 6.14). The software also allows the bandpass filtering of the demodulated PCM signal, the cutoff frequencies and the order of the bandpass filter being adjustable. The demodulated and filtered PCM signal can be saved in a separate $*.wav$ file and can be played. The filter window is shown in Fig. 6.21. The window is accessible by pressing the *Filtru* button. The filtered demodulated PCM signal can be analyzed both in the time domain and in the frequency domain.

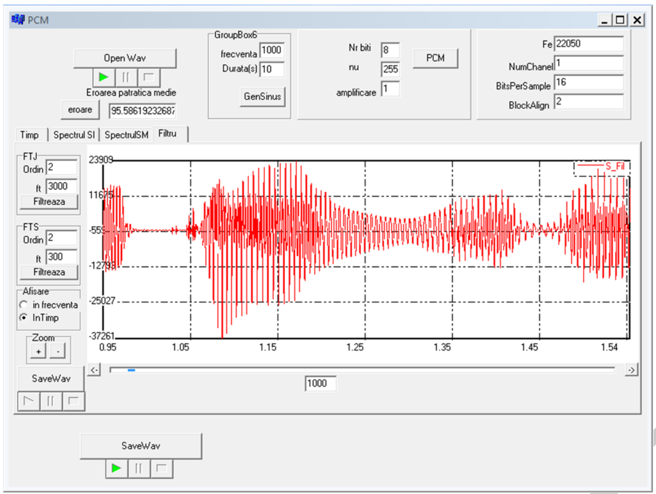

Figure 6.21: The *Filter* window of the *PCM* simulation software.

Note: the bandpass filter used to filter the modulated and then demodulated PCM signal is composed of a low pass filter followed by a high pass filter. The cutoff frequencies and the orders of the two filters can be changed independently.

Delta modulation simulation software

The *Delta_sim* software allows the simulation of the linear Delta modulation and the adaptive Delta and Song and Jayant modulations. The test signals are: sine, triangle and trapezoid. The amplitude and the value of the DC component of all test signals can be modified, and in the case of the triangle and trapezoid signals the slope of the signal also can be modified and thus different triangular (from symmetrical triangle to saw tooth) and trapezoidal (including rectangular signal) signals can be generated. The number of displayed periods can be changed and the source signal spectrum can be visualized.

In the case of the linear Delta modulation, the quantization step and the initial value of the predicted signal can be modified (also called the Delta modulated signal). In the case of adaptive Delta modulations, the elementary quantization step (Song modulation) can be set, respectively the parameter p (Jayant modulation), also the initial value of the quantization step, the initial value of the predicted signal and the value of the first bit, that is the bit generated at the zero moment when the modulation process begins can be set. For all Delta modulations implemented, can be changed the sampling frequency (also called the modulation frequency), more precisely the number of samples per period of the test signal. After pressing the *Modulare* button, the Delta signal and the sequence of modulated bits are generated and displayed. Can be visualized the spectrum of the Delta modulated signal and can be displayed the error (difference) between the source signal and the Delta signal and can be computed the mean square value of this error. After pressing the *Demodulare* button the Delta signal is displayed, that can then be filtered with a low pass filter, with adjustable cutoff frequency and order (slope of the filtering characteristic). The filtered signal is displayed and the error between the source and the demodulated (and filtered) signal can be displayed and the mean square value of this error can be computed (the mean square value of the error can be considered a ”total” noise, after the demodulation, which affects the source signal transmitted). In Fig. 6.22 and Fig. 6.23 displayed two screenshots showing the modulation and demodulation windows of the *Delta_sim* simulation program.

Delta modulation evaluation software

The software *Delta_voice* allows the simulation of the linear Delta modulation and the adaptive Delta modulations Song and Jayant using voice test signals, but at the same time it allows the evaluation of the Delta modulation performance in conditions close to reality. The application implements the linear Delta modulation and the adaptive Delta modulations Song and Jayant. In the case of linear Delta modulation, the

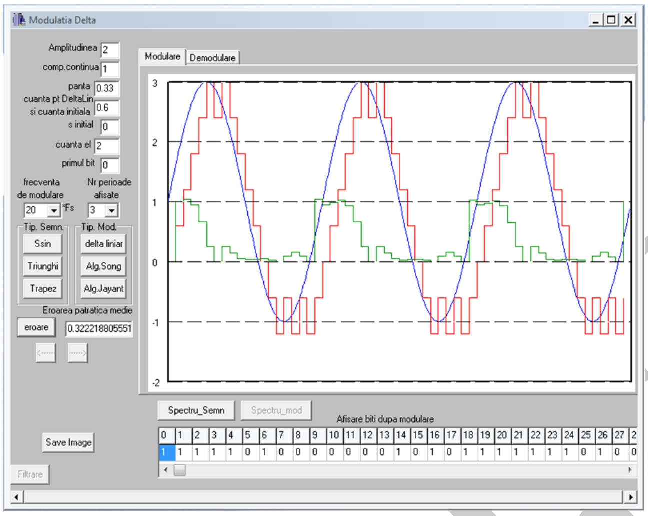

Figure 6.22: The *Modulation* window of the *Delta_sim* software.

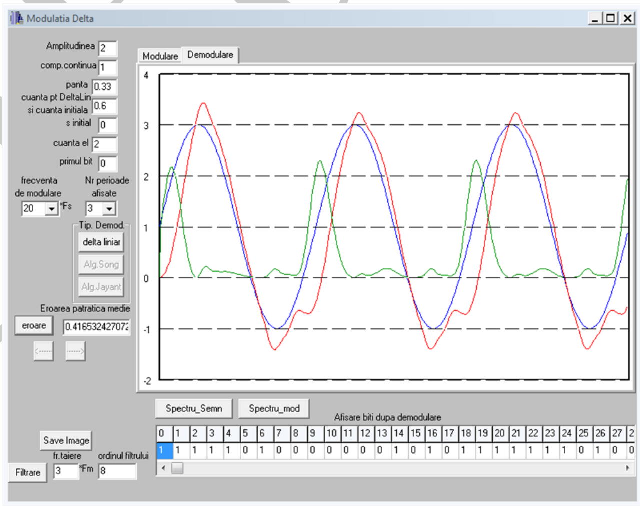

Figure 6.23: The *Demodulation* window of the *Delta_sim* software.

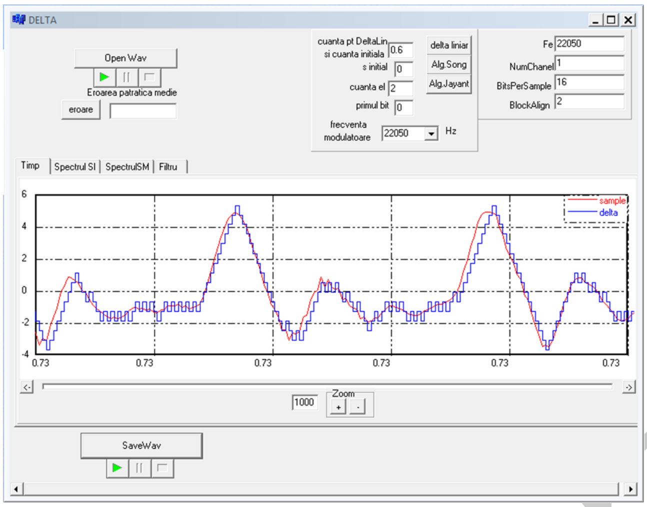

Figure 6.24: The main window of the *Delta_voice* software.

quantization step, the initial value of the Delta modulated signal and the sampling frequency can be changed. In the case of Song modulation, the elementary quantization step, the initial value of the quantization step and of the Delta signal, the first modulated bit, that is the bit generated at time zero (starting moment) of the modulation process and the sampling frequency, can be modified. The parameters that can be controlled are similar for the Jayant modulation, but in this case instead of the elementary quantization step we have the p parameter.

The software allows the uploading of a $*.wav$ type sound/voice file and after this process the samples of the test signal are displayed, as well as its parameters, such as: sampling frequency, signal type (mono or stereo), number of bits/sample. The uploaded file can be played with the player integrated in the application. The obtained Delta signal can also be saved in separate $*.wav$ type file and then can be listened to using the in-app player software. The main window of the *Delta_voice* software is presented in Fig. 6.24.

Note: the Delta modulation frequency, that is, the sampling frequency with which the Delta modulator works cannot be increased above the value of the source signal sampling frequency. It can only be reduced.

Note: Delta demodulation involves generating the Delta signal and then the low pass filtering of this signal. Even if the Delta signal generated by the application and stored in a $*.wav$ file is not previously filtered, the filtering will be provided by the filters on the sound card of the computer running the application.

The *Delta_voice* software allows the display of the error between the source signal and the modulated Delta signal (linear or adaptive), that is, the quantization noise that ultimately affects the demodulated Delta signal. The application displays the mean square value of the error, that is, the value of the power of the quantization noise. The software also has a zoom feature for a better visualization of the Delta modulated signal and the quantization error/noise.

The *Delta_voice* software allows to visualize the spectrum of the source signal as well as of the Delta signal (see the *Spectrul SI* and *Spectrul SM* buttons in the main window of the software - Fig. 6.24). The software also allows low pass filtering of the Delta signal, the cutoff frequency and filter order being changeable. The filtered Delta signal can be saved in a separate $*.wav$ type file and can be listened to. The filter window is shown in Fig. 6.25. The window is accessible by pressing the *Filtru* button. The filtered Delta signal can be analyzed both in the time domain and in the frequency domain.

6.5 Application

1. Analyze the steps involved in PCM modulation. Identify the need for nonuniform quantization and analyze the methods of implementing PCM modulation with nonuniform quantization.

2. Analyze the parameters of the μ compression law. Calculate and analyze the parameters of the A

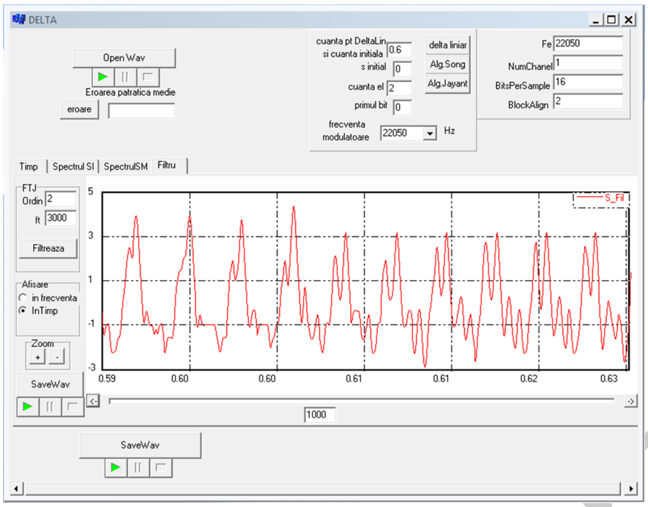

Figure 6.25: The *Filter* window of the *Delta_voice* software.

compression law.

3. Analyze he coding/decoding tables (i.e. compression/expansion tables) for the μ and A compression laws.

4. Using the *PCM* software the following operations are performed:

   (a) Upload a test file and visualize and listen to the test voice signal.

   (b) The test signal is modulated using linear PCM modulation with 8 bits per codeword. The demodulated PCM signal is visualized and listened to and the quantization error is displayed. The effect of the quantization noise is subjectively identified.

   (c) Attenuate the source signal (at least half) and the previous step is repeated. How is the quantization noise perceived in this case? Explain the differences (if any) from the situation in the previous point.

   (d) The test signal is modulated using PCM modulation with compression according to the μ law with 8 bits per codeword. The demodulated signal is visualized and listened to and the quantization error is displayed. The effect of the quantization noise is subjectively identified. How is the quantization noise perceived in this case compared to the situation without compression? Explain the differences (if any) with respect to the case without compression.

   (e) Attenuate the source signal (at least half) and the previous step is repeated, i.e. the source signal is modulated PCM with compression according to the μ law. How is the quantization noise perceived in this case? Explain the differences (if any) from the situation with linear PCM modulation.

   (f) Repeat the previous steps (the 4 previous steps) for a smaller number of bits per PCM codeword, for example 5 bits per PCM code word. The PCM modulation without and with compression is performed for the situation in which the test signal is not attenuated, respectively is attenuated. In each case, the quantization noise is displayed and the effect of this noise is evaluated subjectively.

   (g) Consider the linear PCM modulated signal with 16 bits per codeword. The PCM signal is filtered to the telephone band using the band pass filter integrated in the application. The effect of limiting the signal band to the telephone band is subjectively identified. The filtering of the source signal to narrower bands is repeated and the effects of these filtering operations are subjectively identified.

5. Analyze the steps involved in the linear and adaptive Delta modulation. Analyze the methods of implementing the linear Delta modulation, respectively the adaptive Delta modulations with simple adaptation of the quantization step.

6. Using the *Delta_sim* software the following operations are performed:

(a) The linear Delta modulation is considered and the following steps are taken:

- consider a test signal from those that can be generated and select a low value for the sampling frequency.
- calculate/estimate the value of the quantization step that minimizes the slope overload distortion.
- generate the Delta modulated signal and evaluate the slope overload distortion (the error displayed by the application can be used for this purpose).
- by successive tests determine the "optimal" value of the quantization step, value that minimizes the slope overload distortion and compare it with the previously calculated value.
- repeat the operations described above for another 2-3 (larger) values of the sampling frequency.
- repeat the operations described above for other test signals (for at least one more test signal).
- it is considered one of the modulation processes from the previous steps (test signal and linear Delta modulation parameters) and the demodulation process is carried out.
- calculate/estimate the "optimal" values of the low pass filter parameters which ensure the minimization of the error (i.e. of the mean squared value of the error) after demodulation.
- determine through successive tests the "optimal" values of the low pass filter parameters which ensure the minimization of the error and compare it with the calculated values.
- repeat the demodulation operations described above for at least one more Delta modulation process.

(b) The Song adaptive Delta modulation is considered and the following steps are taken:

- consider a test signal from those that can be generated and select a low value for the sampling frequency.
- generate the Delta modulated signal and evaluate the slope overload distortion (the error displayed by the application can be used for this purpose).
- by successive tests determine the "optimal" value of the elementary quantization step, value that minimizes the slope overload distortion.
- by successive tests determine the "optimal" value of the elementary quantization step, value that minimizes the mean square error. Compare the two "optimal" values.
- repeat the operations described above for another 2-3 (larger) values of the sampling frequency.
- repeat the operations described above for other test signals (for at least one more test signal).

(c) The Jayant adaptive Delta modulation is considered and the following steps are taken:

- consider a test signal from those that can be generated and select a low value for the sampling frequency.
- generate the Delta modulated signal and evaluate the slope overload distortion (the error displayed by the application can be used for this purpose).
- by successive tests determine the "optimal" value of the p parameter, value that minimizes the slope overload distortion.
- by successive tests determine the "optimal" value of the p parameter, value that minimizes the mean square error. Compare the two "optimal" values.
- repeat the operations described above for another 2-3 (larger) values of the sampling frequency.
- repeat the operations described above for other test signals (for at least one more test signal).

7. Using the *Delta_voice* software the following operations are performed:

(a) Upload a test file and visualize and listen to the test voice signal.

(b) The test signal is modulated using linear Delta modulation. The modulated signal (i.e. the predicted signal) is visualized and listened to. The distortions introduced by the linear Delta modulation are subjectively identified.

(c) Repeat the previous step (i.e. linear Delta modulation) for different values of the quantization step and find an "optimal" value of the quantization step.

(d) The test signal is modulated using Song adaptive Delta modulation. The modulated signal (i.e. the predicted signal) is visualized and listened to. The distortions introduced by the Song adaptive Delta modulation are subjectively identified.

(e) Repeat the previous step (i.e. Song adaptive Delta modulation) for different values of the elementary quantization step and find an "optimal" value of the elementary quantization step.

(f) The test signal is modulated using Jayant adaptive Delta modulation. The modulated signal (i.e. the predicted signal) is visualized and listened to. The distortions introduced by the Jayant adaptive Delta modulation are subjectively identified.

(g) Repeat the previous step (i.e. Jayant adaptive Delta modulation) for different values of the p parameter and find an "optimal" value of the p parameter.

(h) Considered the linear Delta modulated signal. Calculate/estimates the optimal parameters of the low pass filter from the demodulator. The Delta signal is filtered using the built-in low pass filter. Adjust the cutoff frequency and the filter order and repeat the experiment several times and determine experimentally the "optimal" values of the filter's parameters. Compare the "optimal" values of the filter parameters obtained theoretically with those obtained experimentally.

6.6 Solved PCM modulation problems

Consider the normalized A compression law (see Fig. 6.13) and a source signal with dynamic range $-1.5V \div 1.5V$. Determine the code obtained at the output of the PCM coder working according to the A compression law for a sample having the voltage $V_{in} = -0.457790V$. It is considered the 8-bit binary code obtained as a result of the PCM encoding of the sample with voltage V_{in}. Determine the voltage V_{out} of the sample obtained at the output of the PCM decoder that works according to the A compression law.

6.6.1 PCM coding

Method a. - using the compression characteristic:

Since the compression characteristics are normalized, the first step is to normalize the input voltage:

$V_{in_{norm}} = -0.457790/1.5 = -0.305193$

The value of the sign bit is set $b_{semn} = 0$ and further on the value of the modulus of the normalized input voltage is used $|V_{in_{norm}}| = 0.305193$.

The next step is to determine the segment where this input voltage falls. For this we consider the axis x of the compression function (the A law) and it is observed that this value falls in segment 6, delimited by the values $0.25 - 0.5$. Thus the segment code obtained is: 110_2.

The last step is to determine the subsegment in which the value of the input voltage falls. To do this, it is determined the size of a subsegment of segment 6, $q_6 = (0.5 - 0.25)/16 = 0.015625$. Using this value can be determined the subsegment in which the input voltage is found: $subsegment = (0.305193 - 0.25)/0.014625 = 3.3$ $\Rightarrow$ subsegment code $= 0011_2$.

The code obtained at the output of the PCM encoder is: 11100011.

Method b. - using the coding table:

In a first phase, the code from the input of the digital compressor is determined. For this it is calculated the elementary quantization step of the uniform A/D converter working on 13 bits: $q_e = 2/2^{13} = 1/2^{12}$, and then it is determined the decimal code returned by the mentioned A/D converter: $val = |V_{in_{norm}}|/q_e = 0.305193 \cdot 2^{12} = 1250.070528 \Rightarrow val = 1250$, value written in binary as 010011100010_2.

The sign bit has already been determined, $b_{semn} = 0$, and from Tab. 6.3 can be determined the segment code: 110_2, and the subsegment code: 0011_2.

The code obtained at the output of the PCM encoder is the same as the code obtained with method a.: 11100011.

6.6.2 PCM decoding

Method a. - using the compression characteristic:

At the destination the value of the output voltage can be determined using the relation:

$V_{out} = \textit{Limită inferioară}_{seg_i} + \textit{cod subsegment} \cdot q_i + 1/2 q_i$

and it is obtained:

$|V_{out}| = (0.25 + 3 \cdot 0.015625 + 0.5 \cdot 0.015625) \cdot 1.5 = 0.3046875 \cdot 1.5 = 0.45703125,$

having a quantization error of: $er_q = (0.305193 - 0.3046875) \cdot 1.5 = 0.0005055 \cdot 1.5 = 0.00075825.$

The multiplication with 1.5 achieves the de-normalization of the normalized output voltage, respectively of the quantization error, which result from the normalized compression characteristic.

Method b. - using the coding table:

From the code received from the line and based on Tab. 6.3 can be determined the sign, the position of the first bit of 1 and the next 4 bits $0010011xxxxxx$. Fill in the last 6 bits with a bit of 1 followed by 0 bits and we get 0010011100000_2. Transform the binary code into a decimal number and the value 1248 is obtained. The output voltage is computed as:

$|V_{out}| = 1248 \cdot q_e \cdot 1.5 = 1248 \cdot 2^{-12} \cdot 1.5 = 0.3046875 \cdot 1.5 = 0.45793125.$

Multiplying by 1.5 it is achieved the the de-normalization of the normalized output voltage that results from using a uniform D/A converter with normalized transfer function.

6.7 PCM modulation questions and problems

1. A compression function is approximated with 4 segments, each segment being divided into 8 subsegments. How many bits is needed for PCM coding?

2. It is given the approximated compression function: $(0,0)-(0.1,0.25)-(0.3,0.5)-(0.6,0.75)-(1,1)$. Give the expansion function approximated by line segments.

3. Determine the mathematical expressions corresponding to the expansion function for the μ and A compression laws (variable x according to the variable y).

4. Explain the advantages/disadvantages of nonuniform quantization.

5. Compute the parameters that characterize the A compression law (see the parameters in Tab. 6.1).

6. A compression function is approximated with 4 segments: $(0,0)-(1/8,0.25)-(1/4,0.5)-(1/2,0.75)-(1,1)$, and each segment is divided into 8 subsegments. What code is obtained at the output of the PCM coder if the input sample's voltage is $0.755V$, and the dynamic range of the signal is $-1V \div 1V$? (The compression function coordinates are given in the format (x, y)). Compute the voltage obtained at the output of the compressor block. Repeat the exercise for a dynamic range of the input signal of $-2.5V \div 2.5V$.

7. Consider the PCM binary codes obtained in the previous exercise. Determine the voltages of the samples obtained at the output of the PCM decoder.

8. Repeat the solved problem for other values of the input sample's voltage and the dynamic range of the source signal.

6.8 Solved Delta modulation problems

It is given the following signal: $x(t) = \sin(2\pi 100t)$. The sampling frequency is $f_e = 10f_m$, where f_m is the maximum frequency in the source signal spectrum (sinusoidal signal frequency in this case).

a. Determine the quantization step for linear Delta modulation.

b. Using the linear Delta modulation determine the bit sequence obtained at the output of the modulator.

c. Plot the modulated signal obtained by using the linear Delta modulation.

d. Compute the signal-to-noise ratio for linear Delta modulation.

e. Using the Song algorithm determine the bit sequence obtained at the output of the modulator. The initial quantization step is $\Delta_0 = 0.2$, and the elementary quantization step is $\Delta_e = 0.2$.

f. Plot the modulated signal obtained by using the Song adaptive Delta modulation.

g. Using the Jayant algorithm determine the bit sequence obtained at the output of the modulator. The initial quantization step is $\Delta_0 = 0.2$, and the p parameter is 1.2.

h. Plot the modulated signal obtained by using the Jayant adaptive Delta modulation.

Solution

a. The quantization step of the linear Delta modulation must be chosen so to eliminate (practically minimize) the slope overload distortion. It follows that the optimum quantization step can be computed as: $\Delta = |dx(t)/dt| \cdot T_e$

   $T_e = 1/f_e = 1/(10 \cdot 100) = 1/1000 = 1ms$

   $\Delta = 10^{-3} \cdot 200\pi cos(2\pi 100t)$.

   From the previous relation it is observed that the optimal value of the quantization step is not constant with time and should be modified according to the value of the source signal, which is impossible in the case of linear Delta modulation. In this case, at time $t = 0$ the value of the quantization step should be $\Delta = 10^{-3} \cdot 200\pi = 0.628$, this being the maximum value of the optimal quantization step. At time $t = T_m/4 = 1/4f_m$ the value of the quantization step should be $\Delta = 0$, this being the minimum value of the optimal quantization step. The quantization step of the linear Delta modulation must be chosen, in this case, between the two extreme values corresponding to the moments of time $t = 0$ and $t = T_m/4$. A possible solution (not necessarily the best one) is to use the optimal quantization step from time $t = T_m/8$, i.e. the value of the quantization step will be:

   $\Delta = 10^{-3} \cdot 200\pi cos(\pi/4) = 0.2\pi \cdot 1/\sqrt{2} = 0.444288$.

   To simplify the calculations it will be used the value $\Delta = 0.44$.

b.c. The values of the source signal samples, samples considered over a period of the signal, are the following:

   $x(0) = sin(0) = 0$

   $x(10^{-3}) = sin(2\pi 0.1) = 0.58$

   $x(2 \cdot 10^{-3}) = sin(2\pi 0.2) = 0.95$

   $x(3 \cdot 10^{-3}) = sin(2\pi 0.3) = 0.95$

   $x(4 \cdot 10^{-3}) = sin(2\pi 0.4) = 0.58$

   $x(5 \cdot 10^{-3}) = sin(2\pi 0.5) = 0$

   $x(6 \cdot 10^{-3}) = sin(2\pi 0.6) = -0.58$

   $x(7 \cdot 10^{-3}) = sin(2\pi 0.7) = -0.95$

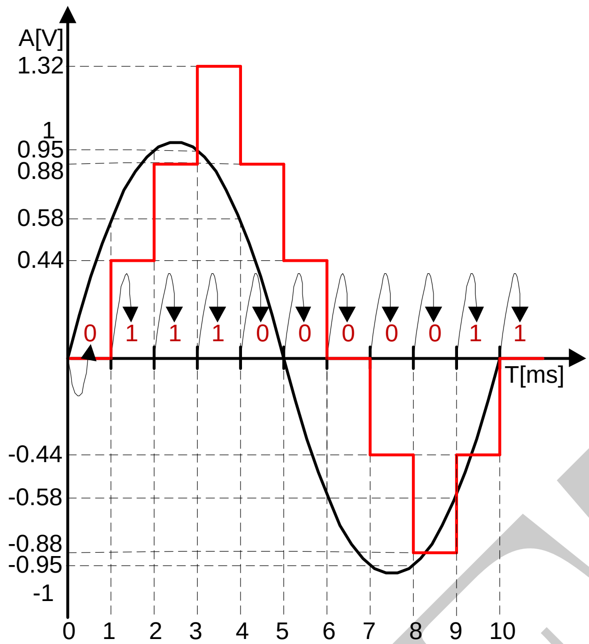

Figure 6.26: The sinusoidal source signal and the Delta signal obtained with the linear Delta modulation.

$x(8 \cdot 10^{-3}) = sin(2\pi 0.8) = -0.95$

$x(9 \cdot 10^{-3}) = sin(2\pi 0.9) = -0.58$

$x(10 \cdot 10^{-3}) = sin(2\pi 1) = 0$

The initial value of the predicted signal is considered $\hat{x}(t) = 0$.

In Fig. 6.26 it is presented the source signal, the predicted signal (i.e. the linear Delta modulated signal) and the bit sequence generated by the modulator. The moments of comparison between the predicted signal and the source signal are also identified in the figure (see the curved arrows).

d. $P_s = A^2/2 = 0.5$

$P_{zq} = \Delta^2/12 = 0.44^2/12 = 0.016133$

$SNR = P_s/P_{zq} = 0.5/0.016133 = 30.99$

$SNR_{dB} = 10lg(30.99) = 14.91dB$

e.f. Considering the source signal sample values, the value of the elementary quantization step $\Delta_e = 0.2$, as well as the initial value of the quantization step $\Delta_0 = 0.2$, the bit generated by the modulator at time 0 $b_0 = 0$ and the initial value of the predicted signal $\hat{x}(t) = 0$, the quantization step values in the case of Song adaptive Delta modulation will be, at the sampling time moments, the following:

$b_1 \neg b_0 \Rightarrow \Delta_1 = \Delta_0 - \Delta_e = 0 \Rightarrow \Delta_1 = \Delta_e = 0.2$

$b_2 = b_1 \Rightarrow \Delta_2 = \Delta_1 + \Delta_e = 0.4$

$b_3 = b_2 \Rightarrow \Delta_3 = \Delta_2 + \Delta_e = 0.6$

$b_4 \neg b_3 \Rightarrow \Delta_4 = \Delta_3 - \Delta_e = 0.4$

$b_5 = b_4 \Rightarrow \Delta_5 = \Delta_4 + \Delta_e = 0.6$

$b_6 = b_5 \Rightarrow \Delta_6 = \Delta_5 + \Delta_e = 0.8$

$b_7 = b_6 \Rightarrow \Delta_7 = \Delta_6 + \Delta_e = 1$

$b_8 \neg b_7 \Rightarrow \Delta_8 = \Delta_7 - \Delta_e = 0.8$

$b_9 = b_8 \Rightarrow \Delta_9 = \Delta_8 + \Delta_e = 1$

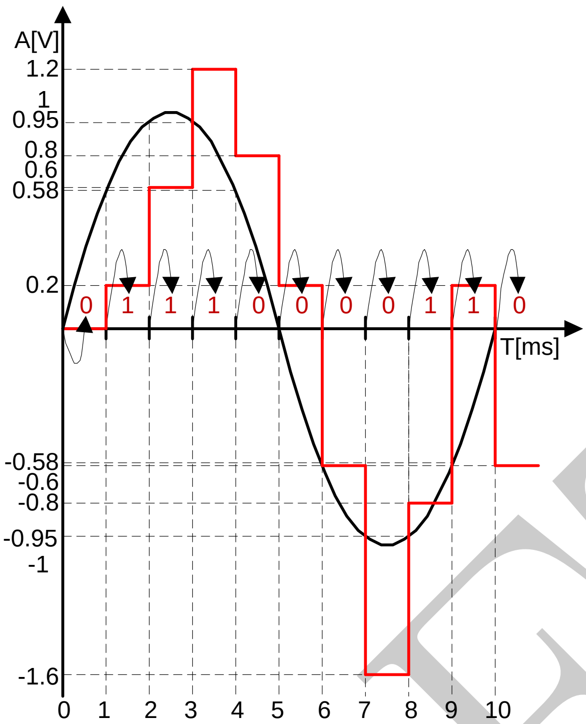

Figure 6.27: The sinusoidal source signal and the Delta signal obtained with the Song adaptive Delta modulation.

$b_{10} \neg b_9 \Rightarrow \Delta_{10} = \Delta_9 - \Delta_e = 0.8$

In Fig. 6.27 it is presented the source signal, the predicted signal (i.e. the Song adaptive Delta modulated signal) and the bit sequence generated by the modulator. The moments of comparison between the predicted signal and the source signal are also identified in the figure (see curved arrows).

g.h. Considering the source signal sample values, the value of parameter $p = 1.2$, as well as the initial value of the quantization step $\Delta_0 = 0.2$, the bit generated by the modulator at time zero $b_0 = 0$ and the initial value the predicted signal $\hat{x}(t) = 0$, the values of the quantization step in the case of Jayant adaptive Delta modulation will be, at the sampling time moments, the following:

$\Delta_0 = 0.2$, $p = 1.2$, $b_0 = 1$

$b_1 \neg b_0 \Rightarrow \Delta_1 = \Delta_0 / p \Rightarrow \Delta_1 = \Delta_0$

$b_2 = b_1 \Rightarrow \Delta_2 = \Delta_1 \cdot p = 0.24$

$b_3 = b_2 \Rightarrow \Delta_3 = \Delta_2 \cdot p = 0.288$

$b_4 \neg b_3 \Rightarrow \Delta_4 = \Delta_3 / p = 0.24$

$b_5 = b_4 \Rightarrow \Delta_5 = \Delta_4 \cdot p = 0.288$

In Fig. 6.28 it is presented the source signal, the predicted signal (i.e. the Jayant adaptive Delta modulated signal) and the bit sequence generated by the modulator. The moments of comparison between the predicted signal and the source signal are also identified in the figure (see curved arrows).

Note: due to the multiple values that can be taken by the quantization step and implicitly by the Delta signal, which generates certain "problems" of graphical representation (i.e. to avoid overloading the figure), the Jayant Delta signal is represented only for the first 5 sampling moments.

6.9 Delta modulation questions and problems

1. It is given the signal in Fig.6.29. The sampling frequency is $f_e = 10 f_m$.

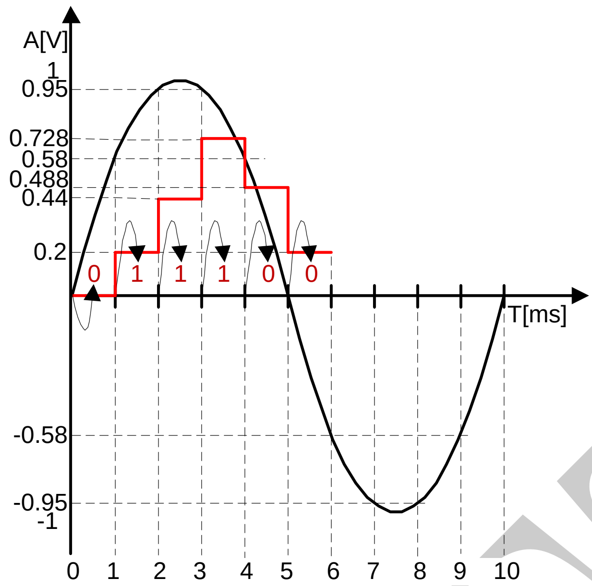

Figure 6.28: The sinusoidal source signal and the Delta signal obtained with the Jayant adaptive Delta modulation.

a. Determine the quantization step for linear Delta modulation.

b. Using the linear Delta modulation determine the bit sequence obtained at the output of the modulator.

c. Plot the modulated signal obtained by using the linear Delta modulation.

d. Compute the signal-to-noise ratio for linear Delta modulation.

e. Using the Song algorithm determine the bit sequence obtained at the output of the modulator. The initial quantization step is $\Delta_0 = 0.2$, the initial bit generated by the modulator is $b_0 = 0$ and the elementary quantization step is $\Delta_e = 0.2$.

f. Plot the modulated signal obtained by using the Song adaptive Delta modulation.

g. Using the Jayant algorithm determine the bit sequence obtained at the output of the modulator.The initial quantization step is $\Delta_0 = 0.2$, the initial bit generated by the modulator is $b_0 = 0$ and the p parameter is 1.2.

h. Plot the modulated signal obtained by using the Jayant adaptive Delta modulation.

2. What are the advantages and disadvantages of Delta modulation compared to PCM modulation?

3. What the prediction circuit of the current sample for Delta modulation is made of? Discuss some implementation solutions.

4. What the demodulation circuit of the linear Delta respectively the adaptive Delta modulation is made of?

5. What is the rule of quantization step adaptation in the case of the Jayant algorithm? But in the case of Song algorithm? What would be an "optimal" adaptation rule for the quantization step in the case of adaptive Delta modulation?

6. Ideally how should the value of the quantization step be chosen in the case of linear Delta modulation?

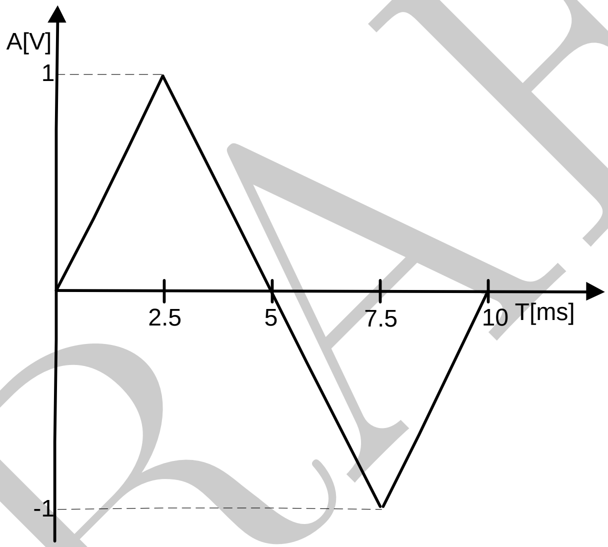

Figure 6.29: Symmetrical triangle source signal.

Chapter 7

Data transmissions in the telephone band

7.1 Introduction

This chapter aims to familiarize the reader with the basic issues related to data transmissions performed in the telephone band. There are considered fax transmissions and data transmissions with dial-up modems. In addition to presenting the basic aspects of the transmissions mentioned above, the chapter proposes a practical exercise for configuring fax and data transmission equipment.

The main aspects that are specifically considered in this chapter are the following:

- setting up a fax machine and performing fax transmissions using such equipment.
- configuring a fax modem computer card with automatic answering equipment (telephone answering machine) capabilities.
- testing the fax modem card mentioned above in automatic and manual answering modes and performing fax transmissions using different software interfaces.
- configuring a dial-up modem using AT commands and performing file transfers using specific communication protocols.
- spectral analysis of the modulated signal generated by a dial-up modem.

7.2 Fax transmissions

Fax protocols ensure the transfer of text and images over the telephone line. These transmission techniques replaced the old Telex techniques, which only provided text transmissions at reduced rates, and dedicated networks were used for these transmissions. Due to the reduced bandwidth of the telephone channel the fax transmission rates and thus the resolution of the transmitted images are reduced. For this reason and due to the emergence of other alternatives (e-mail services, chat, etc.), following the development of broadband digital networks, the importance of fax transmissions has decreased significantly, although still are used on a fairly large scale [58] [59].

Digital standards for fax transmission over the telephone line, so-called Group 3 standards, allow the following resolutions:

- horizontal: 100 lines, vertical: 100 lines - basic resolution.
- horizontal: 200 lines, vertical: 100 lines - standard resolution.

- horizontal: 200 lines, vertical: 200 lines - fine resolution.
- horizontal: 200 lines, vertical: 400 lines - superfine resolution.
- horizontal: 300 lines, vertical: 300 lines.
- horizontal: 400 lines, vertical: 400 lines - ultra-fine resolution.

Note: the first fax transmission standards, so-called Group 1 and Group 2 standards, were based on analogue transmissions and had a working principle similar to analog TV transmission. These standards are no longer used.

Group 4 standards are intended to operate on 64kbps ISDN digital lines and allow higher resolutions of the transmitted images. There are also dedicated standards for fax transmissions over IP networks (FoIP - Fax over IP). The resolution of the transmitted images varies from small values (150DPI) to large values (9600DPI or higher values).

A fax machine that works on the telephone line is composed of a scanner, a printer (the scanner and printer having the resolutions mentioned above), a dial-up modem, which ensures the digital transmission, and the control block that performs the synchronization between the fax equipment and implements the required signaling operations. The fax transmission rates over the telephone line are identical to the rates that are provided by the dial-up modems included in the equipment. Depending on the dial-up modem standard used, fax transmissions over the telephone line can be performed at the following bit rates [59] [60]:

- 2400bps, 4800bps - standard ITU-T V.27.
- 4800bps, 7200bps, 9600bps - standard ITU-T V.29.
- 7200bps, 9600bps, 12000bps, 14400bps - standard ITU-T V.17.
- 28800bps - standard ITU-T V.34.
- 33600bps - standard ITU-T V.34bis.

The signaling between the fax equipment takes place in the telephone band (in band signaling) after the set up of the voice connection. The detection of the equipment (handshaking) takes place using specific fax tones (there is a tone generated by the calling fax equipment and one generated by the called fax equipment), after which the dial-up modems are synchronized and the capabilities detection takes place, meaning the fax standards which the equipment are using are detected. The signaling sequence corresponding to a fax connection on the telephone line is shown in Fig. 7.1 [58].

The transmission fax signal is continuously generated after dialing the number to inform the called terminal as soon as possible that a fax transmission will be performed. There is also the possibility that the fax transmission and, implicitly, the generation of the transmission fax tone to be triggered manually after the telephone connection is established. The called fax machine can answer automatically after a preset number of ring signals, or the fax reception can be triggered manually, at which point a specific tone is generated. The manual mode is used when the fax machine is also used as a telephone, with the possibility of a person answering a call in which a fax transmission is desired.

7.3 Dial-up transmissions

Data transmissions performed in telephone networks, also called dial-up transmissions, were the first solutions for the implementation of WAN (Wide Area Network) computer networks [61]. Later, at the beginning of the Internet network, they represented the main user access technique to the Internet. With the development of broadband networks and access technologies the importance and usefulness of dial-up transmissions has fallen sharply, currently being used as backup solutions or in certain special applications. Dial-up data transmissions

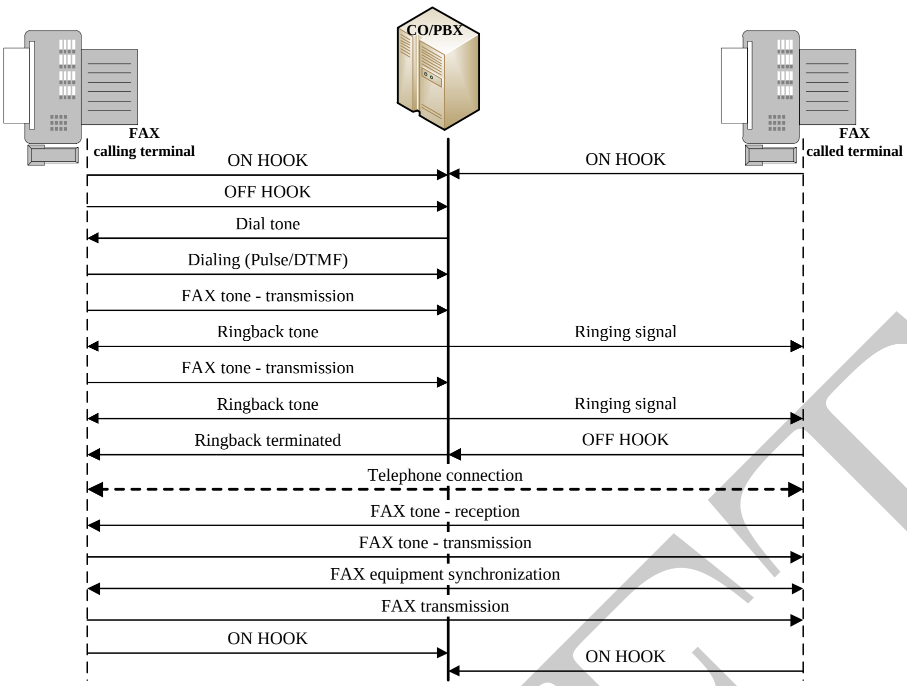

Figure 7.1: The signaling sequence corresponding to a fax connection on the telephone line.

take place in the telephone band (300 - 3400Hz) or in the extended telephone band (0 - 4000Hz), in the case of newer standards, but these transmissions can only take place in digital telephone networks (with digital switching and transport).

Over time, several dial-up modem standards have been developed, standards that are briefly mentioned below [60]:

- Bell 101: 110bps; 2-wire duplex transmission; switched telephone lines; FSK modulation.
- Bell 103 / ITU-T V.21: 300bps; 2-wire duplex transmission; switched telephone lines; FSK modulation.
- Bell 202: 1200bps; 2-wire simplex transmission; switched telephone lines; FSK modulation.
- Bell 212A / ITU-T V.22: 1200bps; 2-wire duplex transmission; switched and leased telephone lines; FDD (Frequency Division Duplexing) duplexing; QPSK modulation.
- ITU-T V.22bis: 2400bps; 2-wire duplex transmission; switched and leased telephone lines; FDD duplexing; 16QAM modulation.
- ITU-T V.23: 1200/600bps; simplex transmission; switched telephone lines; FSK modulation; 75bps backward channel for ARQ operations, FSK modulation, positioning at the lower edge of the telephone band.
- ITU-T V.26: 2400bps; 4-wire duplex transmission; leased telephone lines; QPSK modulation.
- ITU-T V.26bis: 2400/1200bps; simplex transmission; switched telephone lines; BPSK/QPSK modulation; 75bps backward channel, FSK modulation, positioning at the lower edge of the telephone band.
- ITU-T 26ter: 2400bps; 2-wire duplex transmission; switched telephone lines; duplexing by echo compensation; QPSK modulation.
- ITU-T V.27: 4800bps; simplex transmission; leased telephone lines; manual channel equalizer; 8PSK modulation; 75bps backward channel, FSK modulation, positioning at the lower edge of the telephone band.

- ITU-T V.27bis: 4800/2400bps; 4-wire duplex transmission; leased telephone lines; automatic channel equalizer; 4PSK/8PSK modulation; optional backward channel.

- ITU-T V.27ter: 4800/2400bps; simplex transmission; switched telephone lines; automatic channel equalizer; 4PSK/8PSK modulation; optional backward channel.

- ITU-T V.29: 9600bps; 4-wire duplex transmission; leased telephone lines; automatic channel equalizer; 16QAM modulation (circular constellation - double 8PSK).

- ITU-T V.32: 9600/7200/2400bps; 2-wire duplex transmission; switched and leased telephone lines; duplexing by echo compensation; 16QAM/32QAM modulation; trellis coding option.

- ITU-T V.32bis: 14400/12000/9600/7200/4800bps; 2-wire duplex transmission; switched and leased telephone lines; duplexing by echo compensation; 128QAM modulation (the largest modulation constellation); trellis coding option.

- ITU-T V.33: 14400bps; 4-wire duplex transmission; leased telephone lines; 128QAM modulation; trellis coding option.

- ITU-T V.34: 33600/31200/28800/26400/24000/21600/19200/6800/14400/12000/ 9600/7200/4800/2400bps; 2-wire duplex transmission; switched and leased telephone lines; duplexing by echo compensation; 1664QAM modulation (superconstellation), 4 dimensional trellis coding.

- ITU-T V.90: 56.0/33.6 kbps; pair of digital and analog modems on switched lines; downstream rates 28800 - 56000bps with an increment of 8000/6 bps (the digital modem is connected to a digital switched network, on a digital BRI/PRI interface); downstream modulation PAM/PCM A/μ law ; upstream maximum speed 33600bps (V.34); switched telephone lines; duplexing by echo compensation.

- ITU-T V.91: 64kbps; digital modem; 4-wire duplex transmission; switched and leased telephone lines; rates between 28000 and 64000bps with an increment of 8000/6bps; PAM/PCM modulation A/μ law .

- ITU-T V.92: 56.0/48.0 kbps; pair of digital and analog modems on switched lines; downstream rates 28000 - 56000kbps with an increment of 8000/6 bps (the digital modem is connected to a digital switched network, on a digital BRI/PRI interface); downstream modulation PAM/PCM A/μ law; upstream rates 24000 - 48000bps with 8000/6 bps increment, PAM/PCM modulation A/μ law; switched telephone lines; duplexing by echo compensation.

7.4 Modem - data terminal interface

The connection of a dial-up modem (Data Communication Equipment - DCE) to a data terminal/computer (Data Terminal Equipment - DTE) is usually realized on a serial communication port (COM port) of Universal Asynchronous Receiver/Transmitter UART type [62]. A physical UART port has the following "encoding" of the logical levels: 0 logic, active state, assigned voltages: +3 ÷ +15V; 1 logic, inactive state, assigned voltages: -3 ÷ -15V; usual voltage values on the UART port pins: ±5V, ±10V, ±12V and ±15V. The connectors used are DB25 (25 pins) and DB9 (9 pins). The DB9 connector replaced the 25-pin connector on the newer data terminals. The diagram of the modem - data terminal connection and the signals on the RS232 UART port are shown in Fig. 7.2. The signals on this serial interface are the following:

- GND - signal ground; there is also a chassis ground (the metal frame of the connector); for the usual data/PC computer equipment the two ground wires are connected together.

- DTR - Data Terminal Ready (DTE → DCE); data terminal ready for transmission.

- DSR - Data Set Ready (DCE → DTE); communication equipment ready for transmission.

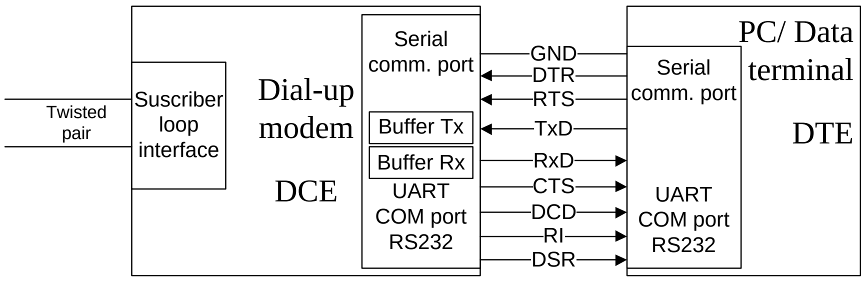

Figure 7.2: Connecting the dial-up modem to the data terminal on a UART RS232 port.

- RTS - Request To Send (DTE $\rightarrow$ DCE); transmission request from the data terminal.
- CTS - Clear To Send (DCE $\rightarrow$ DTE); communication equipment ready for transmission.
- TxD - Transmitted Data (DTE $\rightarrow$ DCE); data transmission.
- RxD - Received Data (DCE $\rightarrow$ DTE); data reception.
- DCD - Data Carrier Detected (DCE $\rightarrow$ DTE); carrier detection.
- RI - Ring Indicator (DCE $\rightarrow$ DTE); ring detection.

Due to the fact that the port is asynchronous we have no clock signals between the modem and the data terminal, and the transmission takes place at character level, each character having its own synchronization information. More specifically it is the so called Start - Stop transmission, which consists of the following: 5 - 8 bit characters are transferred (usually 7 - 8 bits), each character begins with a Start bit (0 logic) and ends with 1 - 2 Stop bits (1 logic); between characters can be pauses, while the date line is in 1 logic; there is the possibility of using a parity bit (even or odd parity) with which a bit error can be detected on the character (more precisely an odd number of errors can be detected; if we have an even number of errors, their effect compensates each other).

The parity bit is computed according to relation: $\left(\sum_{i=1}^{N} b_i + P\right)_{mod2} = I$, where b_i represents the bits of the character, P represents the parity bit, and I is 0 for even parity and 1 for odd parity.

The following bit rates can be programmed on a UART RS232 interface [63]: 110, 150, 300, 600, 1200, 2400, 4800, 9600, 19200, 38400, 56000, 57600 and 115200bps. There are also UART interfaces that can be programmed with other rates, respectively the maximum rate can be increased up to 230400bps, but usually the maximum rate is 115200bps.

If a modem card is used that connects to the internal data bus of the data terminal, or if the external modem connects to a USB communication port, the COM UART interface between the modem and the data terminal is a virtual one that emulates the functioning of the physical port.

7.5 Control of data transfer in dial-up modem transmissions

A flow control algorithm must be implemented on the UART interface, because due to the bit rate difference on the modem - data terminal interface and the modem - telephone line interface and the burst transmissions the modem transmission buffer can be filled up, at which point the transmission of the data terminal must be stopped. After clearing the buffer the transmission can resume. This process of controlling the transmission rate on the UART interface can be performed hardware, using the RTS/CTS signals, or software using special ASCII characters to stop or start the transmission, Xon/Xoff mechanism (Xon: DC1 ASCII / 11 hex / CTRL + Q; Xoff: DC3 ASCII / 13 hex / CTRL + S). In this case data sequence manipulations are required to avoid the effect of Xon/Xoff characters that may appear in the data sequence [64].

For controlling the data transfer between modems, several protocols for text and data transfer have been developed over time (it usually about files). Some of the most common protocols are:

- ASCII protocol - ensures the transfer of ASCII characters/text on the UART interface to and from the modem [65].
- Xmodem/Ymodem - simple File Transfer Protocol (FTP) that integrates basic file transfer functionality: packet generation, error detection, simple Stop&Wait type Automatic Retransmission ReQuest (ARQ); several variants have been developed, not all compatible with each other [66].
- Kermit - represents an improved version of the Xmodem/Ymodem FTP protocols that supports text and file transfer; implements a Sliding Window ARQ mechanism; it has better performance than previous versions and has evolved into a "de facto" standard (officially non-standardized mechanism) for FTP applications on dial-up modems [67].
- Zmodem - is one of the most advanced FTP protocols for dial-up applications; provides significantly higher performance compared to other previous protocols; implements a Sliding Window type ARQ mechanism, the error detection being performed with a CRC-32 control sequence; the data transfer can be initiated by the source (it was not possible in the previous protocols) and is restartable, that is, an interrupted file transfer can continue from the point where the transfer was interrupted without the need to restart the transfer from the beginning of the file, as it happens in the case of the previous protocols [68].

The standardization bodies have also developed several standards that refer to the control of errors in dial-up data transmissions respectively to the data compression, in order to increase the effective transfer rate. Of these standards the following can be briefly mentioned [60]:

- ITU-T V.41: code independent error control mechanism; the error correction is based on an ARQ process performed by an intermediate equipment, inserted between DTE and DCE (can be included in one of these equipment); for error detection the $x^{16} + x^{12} + x^5 + 1$ generator polynomial is used; the retransmission requests are transmitted on the low rate backward channel; the standard is intended for low-speed simplex modems.
- ITU-T V.42: it is a standard specifying an error correction protocol based on the ARQ technique; is intended for DCE duplex equipment and allows the receiver to request immediate retransmission of any lost data packet, but does not guarantee a minimum retransmission time; it defines a generator polynomial of order 32 (standard ITU-T CRC-32 polynomial): $x^{32} + x^{26} + x^{23} + x^{22} + x^{16} + x^{12} + x^{11} + x^{10} + x^8 + x^7 + x^5 + x^4 + x^2 + x + 1$; the standard also ensures the conversion of asynchronous data received on the UART interface into synchronous data according to standard ITU-T V.14 - it is essentially a positive or negative justification mechanism that inserts or deletes stop bits.
- ITU-T V.42bis: it is an adaptive data compression standard based on the Lempel Ziv dynamic dictionary algorithm; it can also work in a transparent way, i.e. without data compression. The concrete algorithm used is BTLZ (British Telecom Lempel Ziv).
- ITU-T V.44: is an adaptive data compression standard based on LZJH (Lempel-Ziv-Jeff-Heath) algorithm incorporated in V.92 dial-up modem standard; V.44 offers a better compression rate than V.42bis, at least for certain types of data sequences and provides on average a 15% higher throughput.

7.6 Control and configuration of dial-up modems using AT commands

AT (Attention) commands, also called Hayes commands by the name of the inventor (Denis Hayes), were developed to allow automatic dialing of the dial-up modems and software configuration of these modems [69] [70]. AT commands were initially developed (in 1981) for 300bps modems and most dial-up modems recognize these commands. The AT commands consist of a series of short text sequences that can be combined/concatenated (no special characters for concatenation are defined) to command different sequences of operations (e.g. line connection, number dialing, transmission/connection parameters change/adaptation, transmission termination detection, line disconnect, etc.). With the development of dial-up modem technologies the configuration of these equipment has become more complex, newer modems having a number of additional functions. To configure these functions new commands have been developed, identified by the & prefix character (ampersand commands), respectively by the % character (percent commands), while the commands in the base set are still supported. The % commands usually refers to proprietary commands (or at least related to proprietary solutions implemented in the modem) and can vary significantly from one type of modem to another.

The main innovation of the AT/Hayes commands is that the modem configuration is performed without defining physical signals or new pins on the modem - data terminal serial interface. Commands are sent to the modem using the data transmission line, TxD, and the answers generated by the modem are received on the data reception line, RxD. From the point of view of the transmission and execution of the AT commands, the modem can be in two distinct states:

- data transmission mode (data mode): the modem considers all characters received on the UART port as data and transmits them on the line.
- command mode: characters received on the UART interface are considered as part of the AT text commands generated by the data terminal.

Switching from the data mode to the command mode is performed with the "+++" (escape sequence) followed by a pause of approximately 1s. It should be mentioned that after the connection is established, the modems automatically switch to data transmission mode. Manual switching from the command mode to the data mode can be done with the command O (online). In addition to the types of AT commands mentioned above there is also a separate set, so called register commands. A register represents a specific physical location in the internal memory of the modem and the AT register commands serve to establish and modify the values in these registers, respectively to read the values contained in the registers. The registers are identified by the character S followed by a number, i.e. the number of the register.

Due to the fact that AT commands are not standardized by any standardization institute (they represent a "de facto" standard) there are numerous incompatibility problems between different manufacturers. These incompatibilities refer both to the differences regarding the change of the modem state to different commands (the timing of these state changes) and the management of the errors that have occurred, as well as to the existence of specific commands for certain types of modems, dedicated commands for certain operations. For these reasons, certain communication protocols offer options that allow the user to configure the modem using commands specific to that transmission equipment.

There are AT commands (maybe more correctly AT type commands or AT similar commands) developed for GSM/3G modems [71]. There are ETSI standards that specify these commands [72], but in this case also there are differences between different types of modems regarding the commands actually implemented. There are also, in many cases, proprietary extensions of the AT command set. The transmission/reception of the AT commands to/from GSM/3G modems is based on a Point to Point Protocol (PPP) connection between the data terminal and the modem (only between these two devices). In the case of higher-speed modems (e.g. 4G modems) it is common to define a virtual Ethernet interface, which involves the use, instead of AT commands, of specific commands of the equipment manufacturer.

7.7 FAX equipment

Samsung SF-360 inkjet telephone fax

It is a fax machine with inkjet printer that can also be used as a telephone [73]. The connection to the line is made using an RJ11 socket. There is also an RJ11 socket for connection of another telephone device or

Figure 7.3: Samsung SF-360 inkjet fax equipment.

automatic answering/telephone answering machine (TAM - Telephone Answering Machine) for taking (and storing) telephone messages. Some fax machines also have built-in answering machines such as the Samsung SF-365TP fax machine. The SF-360 allows four working modes, namely: telephone (TEL mode), fax (FAX mode), automatic (AUTO mode) and telephone answering (TAM mode).

- in TEL mode, the equipment works like a normal phone, meaning it cannot answer a call automatically, but it is possible to send or receive a fax manually.
- in FAX mode, the machine automatically responds to a call after a certain number of received ring signals and enters the fax reception mode (it generates the reception fax tone), without waiting for the detection of the transmission fax tone.
- in AUTO mode, the machine automatically answers a call, but does not start the fax reception unless it detects a transmission fax tone. If this tone is not detected it generates/imitates a ring tone, announcing the user that it is a phone call. If the user does not answer, after a while, the equipment enters the FAX mode and waits for a fax transmission tone. If it does not receive a fax tone within a certain time, it will be automatically disconnected.
- in TAM mode, the equipment waits for several ring signals, which means that it will allow the answering machine time to answer the call. If this does not happen, it switches to FAX mode, answers the call and generates the reception fax tone.

In Fig. 7.3 it is presented the SF-360 fax machine as well as the front panel containing the display and the function keys.

The configuration of certain parameters such as working mode, resolution, gray level (the scanner and the printer are black and white) is done from the function keys or from a menu (most settings), accessible with the Menu key. The selection in the menu is done with the scroll keys, and the validation of an option is done with the OK key. Exiting a menu is done with the Back key. The Start Fax key allows fax transmission or reception to be initiated manually, and the Stop/Clear key allows any transmission or configuration operations to be aborted. The Start Copy key allows to scan and print a document, i.e. a xerox working mode. The specific functions of the telephone, redial, microphone lock (Mute), hands free call (On Hook Dial) are activated with specific keys. The phone book is managed using the Phone Book key and a specific menu.

Configuring the equipment, i.e. setting the working mode, setting the printer's parameters, managing the phone book, setting the number of automatic answer rings, setting the date and time, setting the ringer level

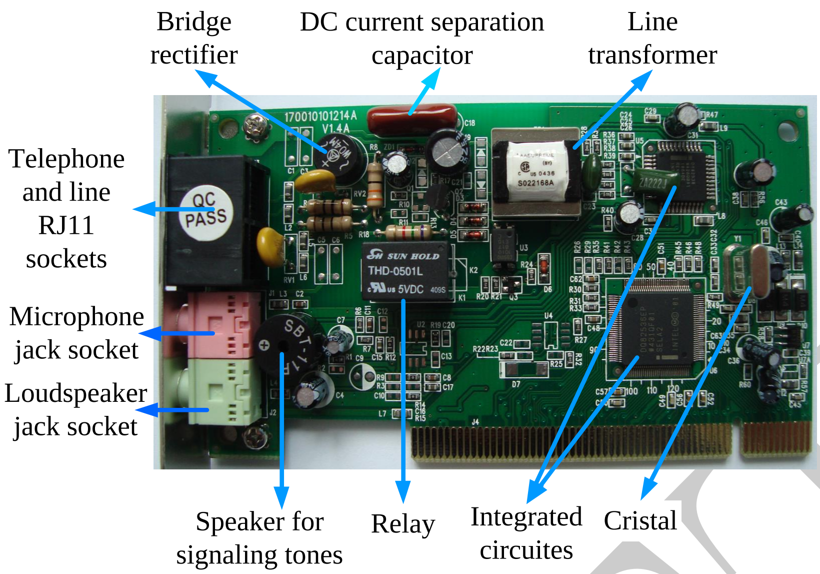

Figure 7.4: Wayjet fax modem expansion card.

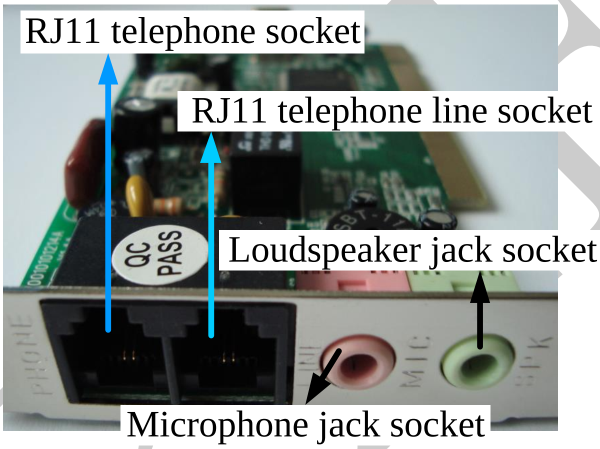

Figure 7.5: The connectors of the Wayjet fax modem card.

and the reception level, setting the level of the tones associated with the keys, setting the system's identifier, setting the resolution of the scanner, establishing the working mode of the printer, testing the scanner, etc., will be performed based on the user manual.

Note: in the case of some fax equipment, such as SF-365TP, it is possible to make a connection with a computer on a USB port, and use the equipment as a printer.

Wayjet fax modem card

It is a 56kbps fax modem that implements the following ITU-T communications standards: V.90 (reception at 56kbps), V.92 (transmission at 48kbps), V.34+ (transmission/reception at 33.6kbps), V.17 , V.29 and V.27ter (Fax Group III - transmission at 14.4kbps), H.324 (teleconferencing support), V.42bis/MNP5 (data compression), V.42/MNP2-4 (error correction). The fax modem also offers support for voice functions and integrates an automatic answering machine/telephone answering machine (TAM) equipment. The Wayjet fax modem is an internal modem (expansion board on the PCI bus of the computer - see Fig. 7.4) with Intel 536EP chipset. On the modem board can be seen (Fig. 7.4) the following components:

- jack sockets for microphone and speaker - see also Fig. 7.5.
- RJ11 sockets for connection to the line of the modem, respectively for connection to the modem of an analog telephone - see also Fig. 7.5.

- a mini speaker for signaling tones used on the subscriber loop.
- a bridge rectifier for rectifying the ring signal.
- a capacitor that ensures the connection to the line of the ringer circuit.
- line transformer.
- modem/telephone switching relay.
- the ICs that implement the modem.
- the quartz crystal resonator used by the board's clock generator.

With the help of a microphone and an external speaker (or a headset + microphone), can be made telephone calls (voice calls), without having to use an analog telephone connected to the modem.

The FaxTalk Communicator software

The Wayjet fax modem can be controlled/configured using a software tool dedicated to voicemail and fax communications called FaxTalk Communicator [74]. It should be mentioned that the fax modem control can also be done with other software tools, for example those integrated in the Windows operating systems. In Fig. 7.6 it is presented the main window of the application in question as well as the Inbox and Outbox windows.

The modem has fax and TAM functionality, meaning that it can automatically receive faxes and record telephone messages. Faxes and messages are stored in the Inbox along with information regarding the phone number from which the call was received, user identifier, message type, call date and time, voice message duration, etc. The faxes can be viewed by clicking on the selected fax entry, and the messages can be listened to using the integrated player application, the control keys of this application being available in the Inbox window. The main window displays the number of faxes and messages that have not been viewed or listened to. Of course the faxes and messages from Inbox can be deleted. The Outbox window displays faxes (phone number, date, status, recipient name) that could not be sent for different reasons, respectively those that are scheduled to be sent at a certain date and time.

The main window of the application includes a keyboard with the typical symbols of a common telephone keypad. With this keypad can be dialed phone numbers. We have separate keys for: "Dial", "Mute" - the microphone is switched off, "Flash" - a flash pulse is generated, "Hold" - the current call is placed on hold, "Speaker" - the speaker is switched on/off, respectively it is used for manual answer of an incoming call. We have keys for speed dialing of several phone numbers - "Speed Dial", and we have the possibility to define a phone book "Phonebook". There are defined two adjustment bars for the volume of the microphone and the speaker.

To send a fax, select the "Send a Fax" option from the "File" menu and then open several windows in which the phone number, the recipient's name/identifier, the fax header, possible files attached to the fax, the date/time at which the fax will be sent, etc. have to be specified - see the windows shown in the following figures (Fig. 7.7 - Fig. 7.10), each window corresponding to a separate step for sending a fax.

There is also the possibility of the manual reception of a fax, in the case of manual answer of an incoming call, for example using the phone connected to the modem. If the specific fax tone (transmission fax tone) is heard then from the File menu the Manual Receive option is selected. When the answering fax tone (reception fax tone) is heard, the phone is hanged up and the fax equipment are allowed to communicate. Configuring the parameters of the fax transmission and other parameters that characterize the process of automatic response/recording of messages, speed dial, etc., can be done from the Configure menu. In the following figures are presented some of the configuration operations.

In Fig. 11 shows the two configuration windows that can be opened from the "Configure - Mailboxes" menu. It is about configuring the way in which the received messages are received and reported (i.e. to an administrator who remotely connects to the equipment) - the "General" window. In the "Greetings window of the same menu

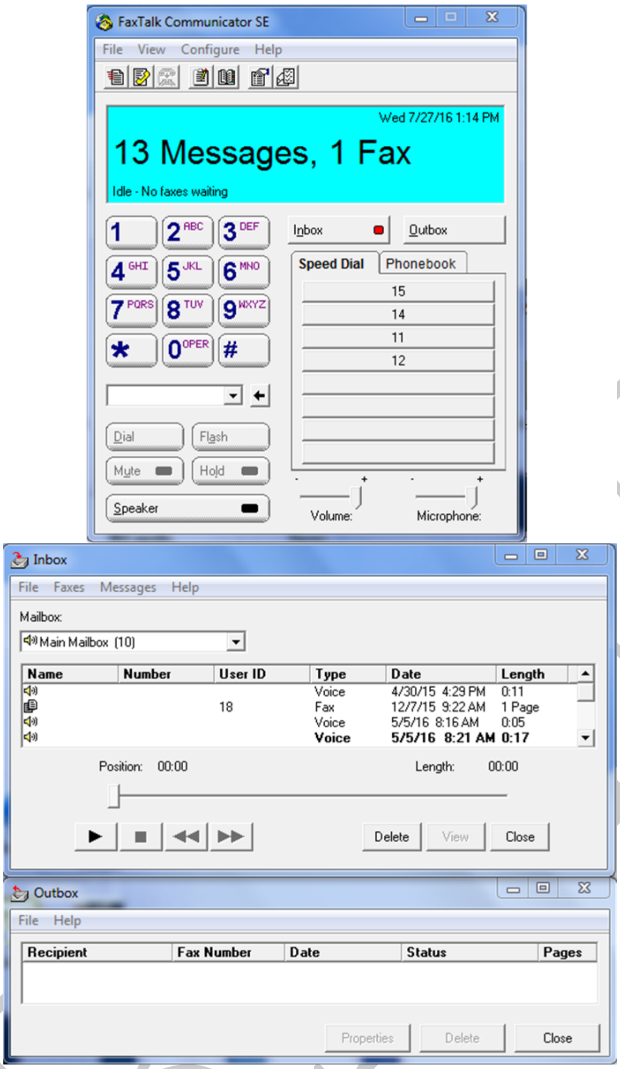

Figure 7.6: The main window of the FaxTalk Communicator application. The Inbox and Outbox windows.

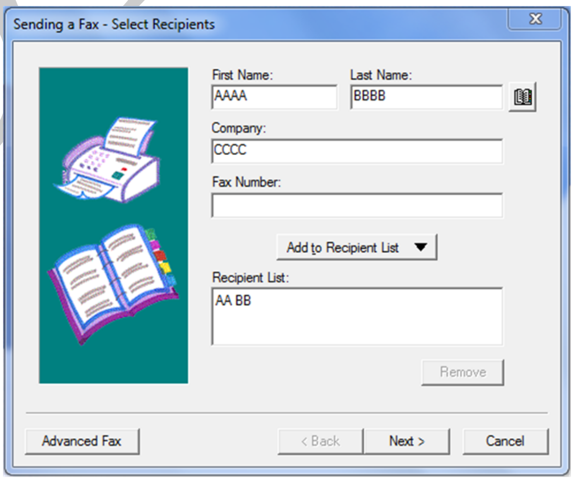

Figure 7.7: Sending a fax with the FaxTalk Communicator tool. Step 1.

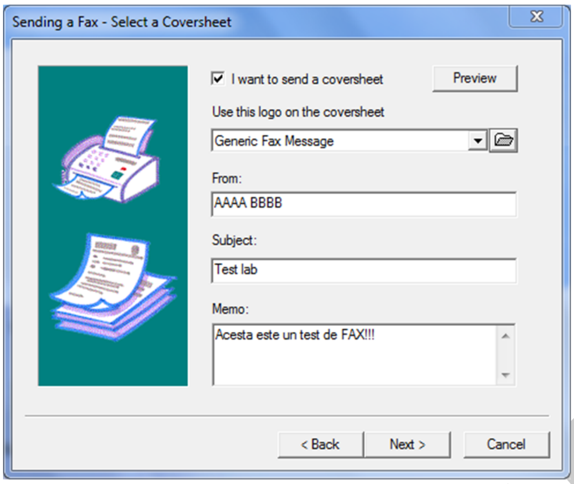

Figure 7.8: Sending a fax with the FaxTalk Communicator tool. Step 2.

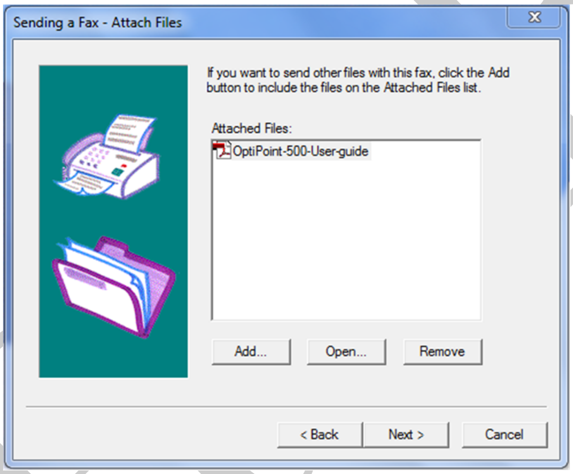

Figure 7.9: Sending a fax with the FaxTalk Communicator tool. Step 3.

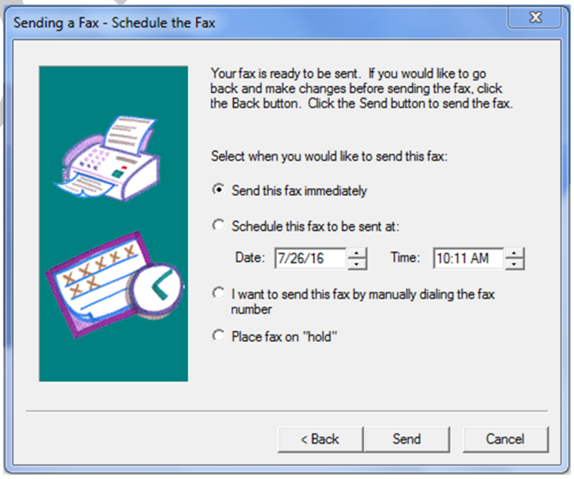

Figure 7.10: Sending a fax with the FaxTalk Communicator tool. Step 4.

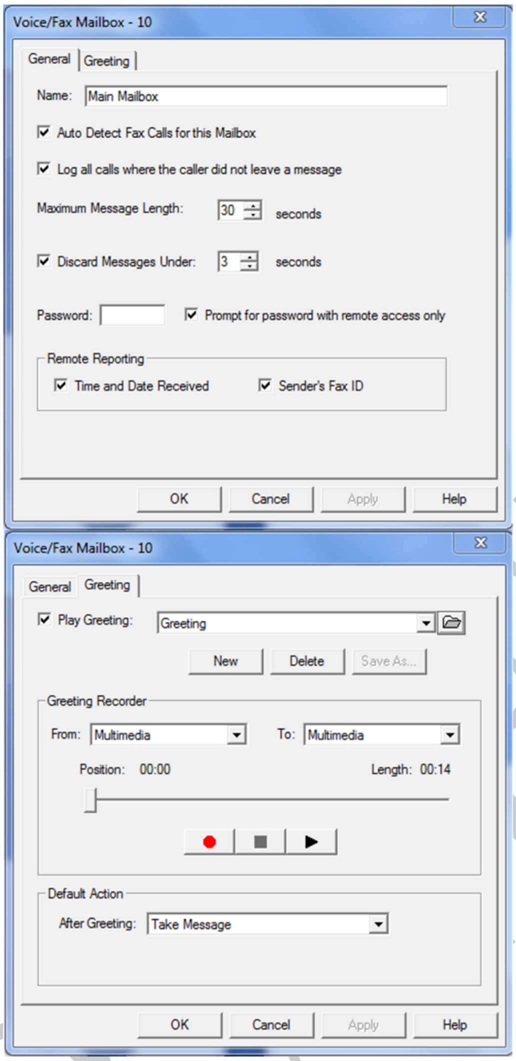

Figure 7.11: The windows of the Configure - Mailboxes menu.

the greeting message generated by the telephone answering machine can be selected, this message can be listened to and the action that the equipment executes after the greeting message can be specified: receive message, receive fax or disconnect.

From the Configure - Program menu can be configured the way the FAX equipment works, more precisely:

- activation of the automatic answer and setup of the number of rings after which the message or the fax is received; actions that are performed when a call is received (When phone rings) - see the General window in Fig. 7.12.
- how to announce the reception of a message or fax (generating sounds, displaying a changing icon - simulating a blinking LED), display of status information when sending or receiving a fax, automatic printing of the received faxes - see the Announce window in Fig. 7.13.
- definition of the fax user identifier, fax number, logo, information included in the fax header, etc. - see the User window in Fig. 7.14.
- setting of the fax transmission parameters: resolution (200×100 dpi or 200×200 dpi), page format, error correction, data compression, number of retries if the fax sending operation fails, etc. - see the Fax window in Fig. 7.15.
- configuration of modem parameters: modem type, transmission/reception speed, driver, flow control on the modem - data terminal interface, initialization commands, sensitivity level of the signaling tones

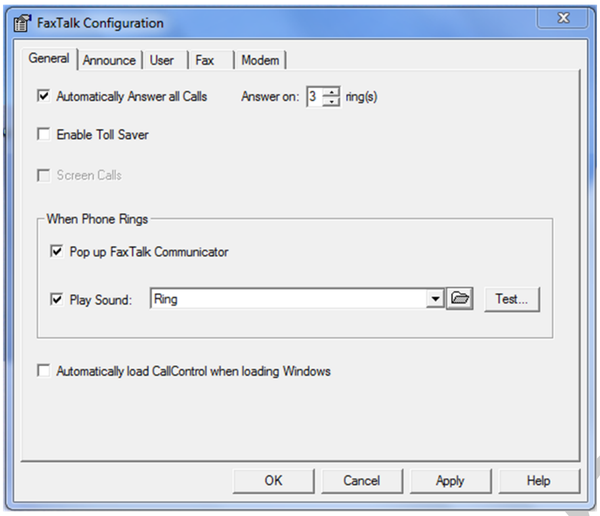

Figure 7.12: The General window of the Configure - Program menu.

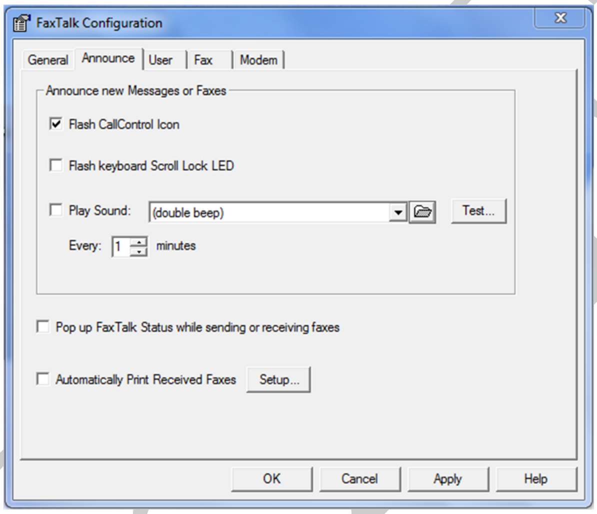

Figure 7.13: The Announce window of the Configure - Program menu.

detector, etc. - see the Modem window in Fig. 7.16.

Note: the FaxTalk Communicator software tool works only under the Windows XP operating system. Under the Windows 7 operating system cannot be activated all the features of the FaxTalk Communicator, e.g. sending faxes, existing some problems with the Wayjet card drivers.

Tools for fax applications integrated into the Windows operating systems

Tools for fax applications integrated into the Windows XP operating system

Click on the Start button, and in the window that is open select Printers and Faxes or Start - Control Panel and in the window that is open select Printers and Faxes. Following this action the window shown in Fig. 7.17 is open. Select the fax equipment to be configured, click on the respective icon and perform the configuration steps.

Note: under the Windows XP operating system, the Wayjet fax modem configuration will be carried out with the FaxTalk Communicator tool.

Tools for fax applications integrated into the Windows 7 operating system

Click on the Start button and select Devices and Printers or Start - Control Panel and in the pop-up window select Devices and Printers and the window shown in Fig. 7.18 appears.

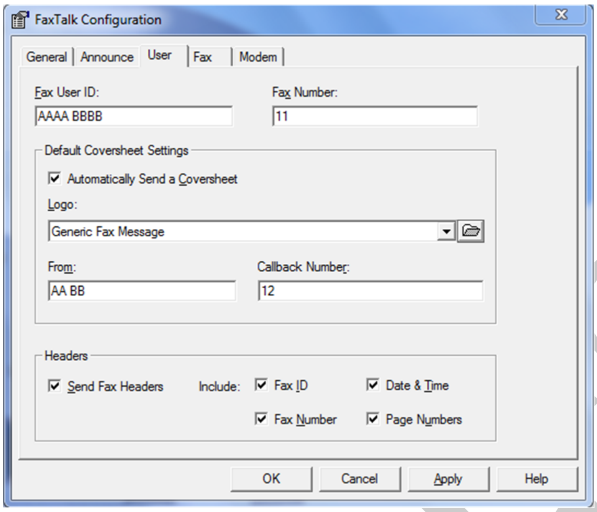

Figure 7.14: The User window of the Configure - Program menu.

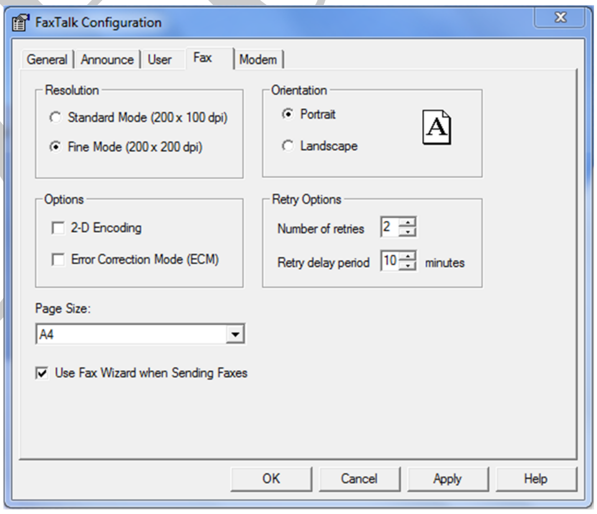

Figure 7.15: The Fax window of the Configure - Program menu.

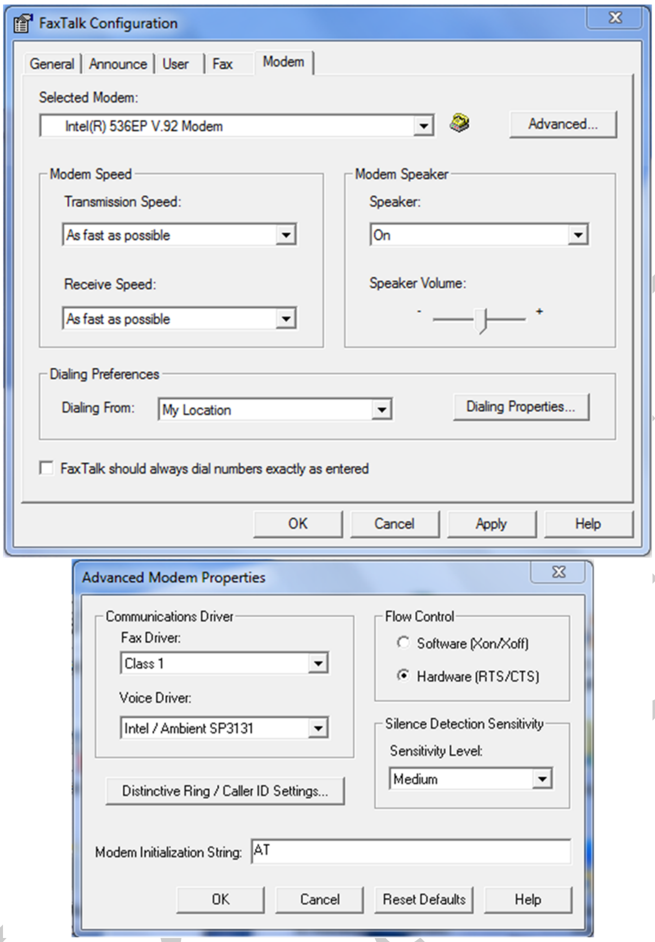

Figure 7.16: The Modem window of the Configure - Program menu.

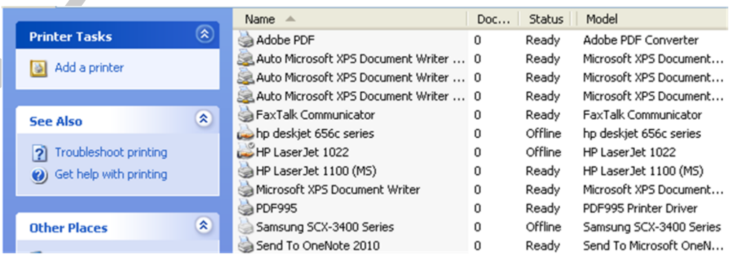

Figure 7.17: The window for selecting the fax and printer equipment under the Windows XP operating system.

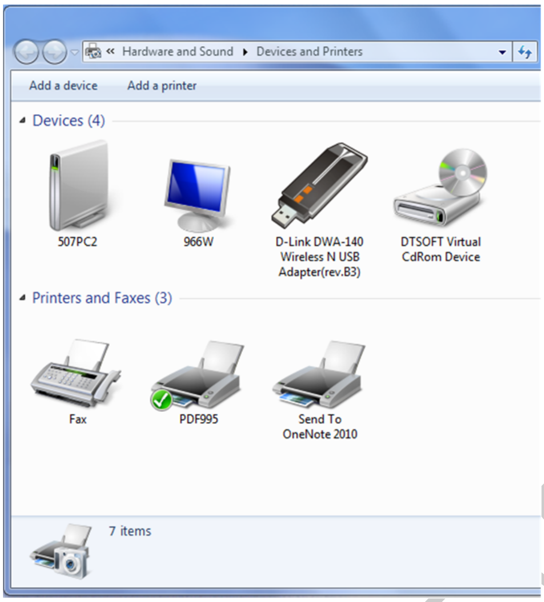

Figure 7.18: The window for selecting the fax and printer equipment under the Windows 7 operating system.

Select the fax equipment that is configured and click on the corresponding icon. Following this action, the main configuration window of the selected fax machine opens - see Fig. 7.19. In the Inbox, also visible in Fig. 7.19, all received faxes are stored together with other information such as the date of reception of the fax, the size in kB, the number of pages, the identifier of the user who sent the fax, etc.

From the File menu you can select the operation that will be performed with the fax equipment, i.e. send a fax, scan a document in a file or scan a document and send it immediately by fax - see Fig. 7.20 (partially can be seen the Inbox window in this figure). If the Fax option is selected, a window appears (see Fig. 7.21) that allows the generation of a fax message or the initiation of the fax transmission operation. In this window can be set the degree of urgency of the fax, the telephone number of the recipient, the subject, notes in the cover page, the text of the fax, etc. Pressing the Send button initiates the recipient's call process and then the sending of the fax. In Fig. 7.21 also can be seen the Sent Items window, which includes all the faxes that have been sent successfully.

The configuration/setting of some parameters/information related to the fax transmission, respectively to the fax user, can be done using the "Tools" menu - see Fig. 7.22. By clicking Tools in the main window, a menu with different options appears - see Fig. 7.22.

With the Sender Information option you can set information about the fax user, as shown in Fig. 7.23.

With the Fax Settings option, can be configured the fax equipment, as seen in the windows shown in Fig. 7.24 and Fig. 7.25. Can be selected the fax equipment, can be set the working mode (source/destination fax or both), the answer mode (automatic/manual), the number of rings for the automatic answer, the number of attempts if the fax transmission operation fails, the time intervals when the fax transmission is resumed, etc.

If the manual reception mode is set, then the reception of a fax and implicitly the generation of the reception fax tone takes place when from the Tools menu (see Fig. 7.22) the Receive a Fax Now option is selected. Also from the "Tools" menu can be open a window for monitoring the fax transmission/reception process - the Fax Status Monitor option - see Fig. 7.26. The monitoring window displays information about the fax call (when calling, if called terminal answers, etc.), whether or not the fax transmission is successful, the order in which the fax pages are sent, etc.

In Fig. 7.26 also can be seen the Outbox window, which contains the faxes that are sent (only temporarily until they are sent), the faxes ready for subsequent transmission, and the faxes that could not be sent.

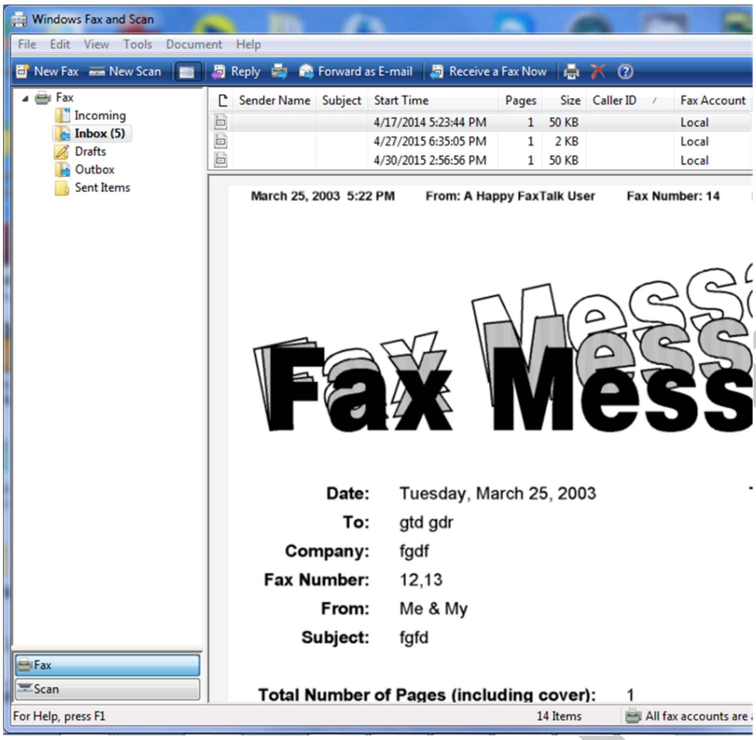

Figure 7.19: The main configuration window of a fax machine.

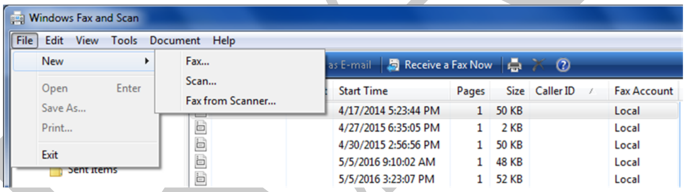

Figure 7.20: The File menu of the main configuration window of a fax machine.

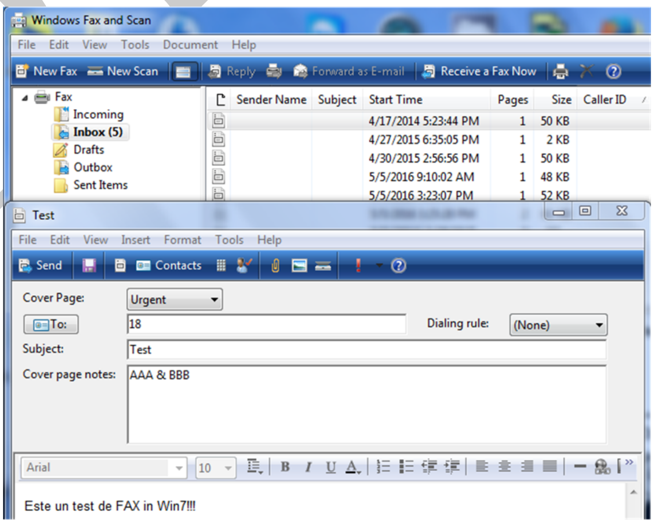

Figure 7.21: The procedure of sending a fax.

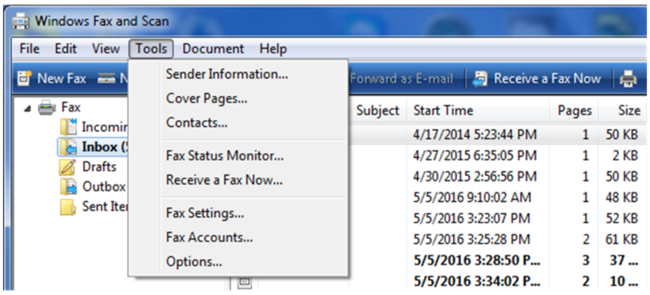

Figure 7.22: The Tools menu of the main configuration window of a fax machine.

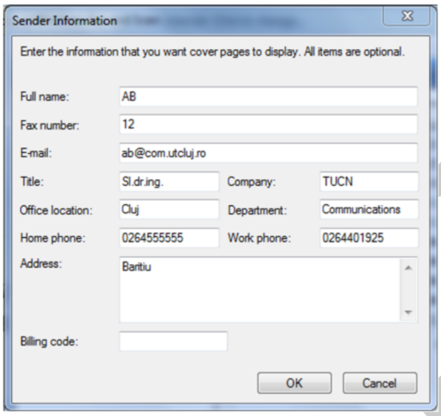

Figure 7.23: The Sender Information window of the Tools menu.

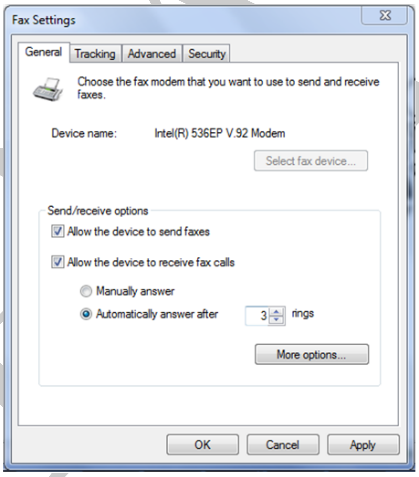

Figure 7.24: The General window of the Fax Settings menu.

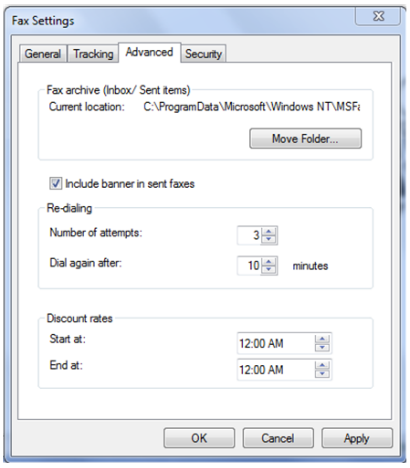

Figure 7.25: The Advanced window of the Fax Settings menu.

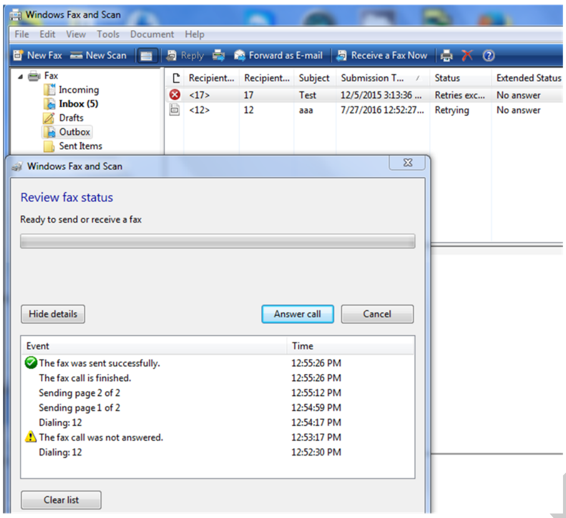

Figure 7.26: Fax transmission/reception monitoring window.

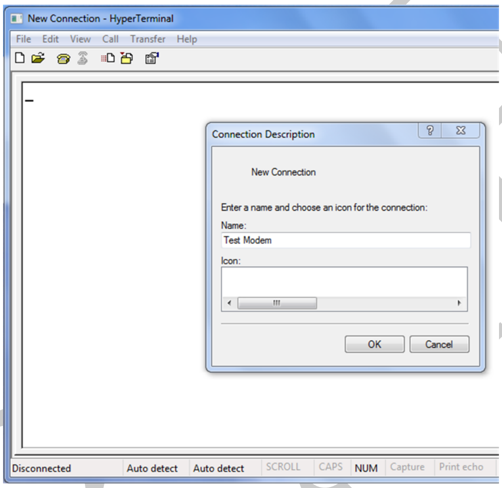

Figure 7.27: The main window of the HyperTerminal application and the Connection Description window.

The HyperTerminal application

This application is a software tool for emulating a data terminal that can make data connections through communications ports (COM ports), dial-up modems and computer networks using the TCP/IP protocol. The tool is integrated into the Windows XP operating system and older operating systems, but in newer operating systems, such as Windows 7, it must be installed separately, not being integrated. The installation basically consists in copying the *hypertrm.exe* and *hypertrm.dll* files to a separate directory, for example the HyperTerminal directory on the Desktop. Starting the application, in this case, is done by starting the *hypertrm.exe* application. In the case of the Windows XP operating system the HyperTerminal application starts as follows: click on the Start button and select Accessories - Communications and from this menu select HyperTerminal. After starting the application the window shown in Fig. 7.27 appears, and in this window is required to be specified a name/descriptor for the connection to be made (serial port connection or using TCP/IP sockets).

Specify an identifier (e.g. Test Modem) (see Fig. 7.27), then click OK and the window shown in Fig. 7.28 appears, window which requires to set a number of parameters such as: region, telephone area code, telephone number of the modem to be called and the modem or COM port or TCP/IP communication socket used for set up of the connection. Can be seen the connection status at the bottom left part of the window, which is currently Disconnected. Also can be seen the phone icon (located next to the Folder icon) that represents a phone in the ON HOOK state.

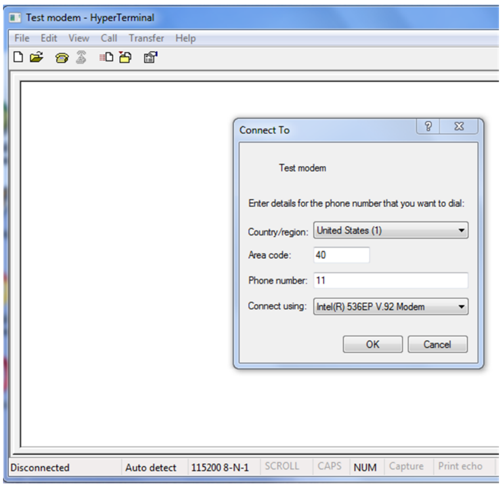

Figure 7.28: The Connect To window of the HyperTerminal application.

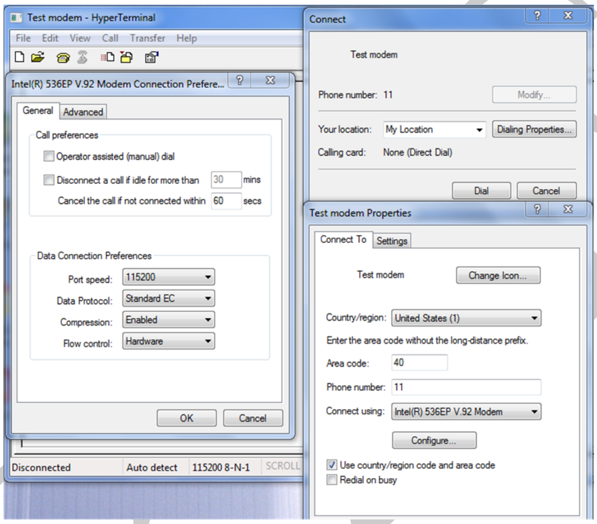

Figure 7.29: Connection parameter configuration windows - General option.

Note: the settings related to the country and the country prefix are not important in the test considered and any values can be specified in these fields.

Click on the OK button (in Fig. 7.28) and the Connect window appears, which includes the Dial button, which starts the call, and the Modify button, which allows to set the connection parameters - see Fig. 7.29. If you click on the Modify button, the Properties - Connect To window appears, window in which can be set again the communication port used from the drop down list in the Connect Using option. It can be set again the area, the area code and the telephone number to be called. If you click Configure button, the connection configuration windows appear. The General window allows you to specify some parameters regarding the call (how to call, when to disconnect, etc.), respectively the parameters that characterize the data connection (COM port bit rate, error correction and data compression on the modem connection, flow control on the COM port) - see Fig. 7.29. In the Advanced window can be set the parameters of the Start-Stop transmission that takes place on the UART interface, respectively the way of displaying the terminal configuration window before and after the call - see Fig. 7.30.

In the Properties - Settings window select ASCII Setup, and in the window that appears select the options: "Send line ends with line feeds" and "Echo typed characters locally", to generate echo in the text type command window, that is, the commands sent to the modem are displayed on the screen - see Fig. 7.31.

Note: it is not always necessary to select the options mentioned above to generate the echo at the terminal, especially under the Windows 7 operating system. If the echo is automatically generated and the options

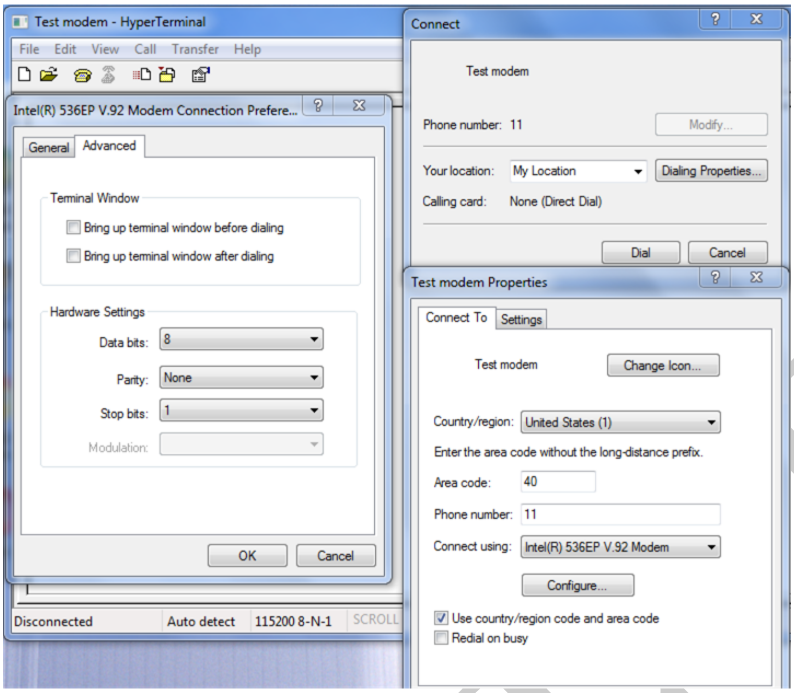

Figure 7.30: Connection parameter configuration windows - Advanced option.

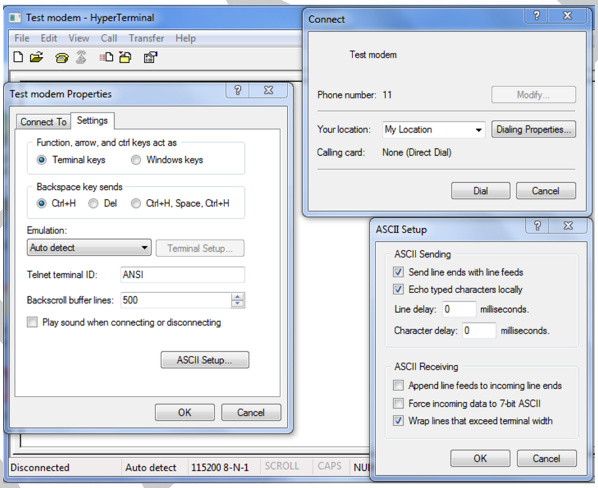

Figure 7.31: The ASCII Setup window of the HyperTerminal application.

mentioned are selected, then the typed characters are displayed twicw. So, it must be checked whether it is necessary to activate the options under discussion.

Note: the Properties configuration window can also be open from the File menu where the Properties option is selected.

After completing the configuration operations commands can be sent to dial-up modem. After the first command, e.g. command "at", the modem connects to the line (goes in the OFF HOOK state) and the communication between the modem and the computer begins on the UART interface - see Fig. 7.32. On the bottom left of the main window can be seen the Connected status and the time since switching to this state. The phone icon under the menu at the top also shows the transition in the OFF HOOK state. At the bottom of the window can be seen the parameters of the Start-Stop communication on the UART interface.

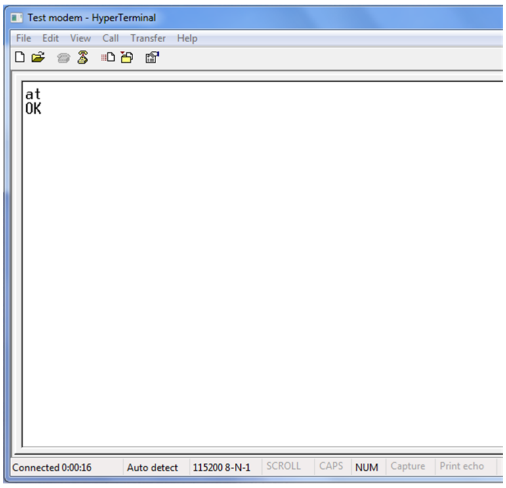

Figure 7.32: Generating text commands in the main window of the HyperTerminal application.

7.8 Application

1. The SF-360 fax machine is considered. It is connected to a BPX exchange and an analog telephone is connected to the fax machine. The following operations are performed:

   (a) The following basic equipment settings are performed:

   - the working mode is set, the time and date are set.
   - set the ring signal level and the receiver level.
   - the sound level associated with the keys is set.
   - the resolution and the gray level are set.
   - set the number of ring signals after which the call is answered.
   - one/two numbers are stored in the phone book.
   - test the speed dialing mode.
   - test how the scanner and printer work (use the Start Copy button).
   - check how to change the printer toner.

   (b) The fax equipment is set by turn in the TEL, FAX, AUTO and TAM modes. A call is generated to the fax equipment from a telephone and the operation is tested in each of the selected working modes.

   (c) A fax transmission from the SF-360 to another fax equipment is performed. The phases of transmission and the associated signaling are identified.

   (d) A fax transmission from another equipment to the SF-360 (FAX or AUTO working mode) is performed. The phases of transmission and the associated signaling are identified.

2. Connect the Wayjet fax modem card to the telephone line, connect an analog telephone to the modem, respectively connect the microphone and speaker to the audio jack inputs. The FaxTalk Communicator application is started. The following operations are performed:

   (a) A telephone call is made between the fax modem board and another telephone terminal connected to a PBX exchange. The functionality of the keys on the FaxTalk Communicator interface is checked (Fig. 7.6).

   (b) Configure the telephone answering machine (follow the steps presented in Fig. 7.11 - Fig. 7.13) and test this functioning mode. Listen to the greeting messages and make two - three telephone calls to the fax modem and record the telephone messages. Monitor the automatic answer mode and the contents of the Inbox and listen to the stored messages.

(c) The parameters of the fax user as well as those of the fax modem are configured (follow the steps shown in Fig. 7.14 - Fig. 7.16).

(d) Two - three fax transmissions with other fax terminals connected to the PBX are performed. The operations described in Fig. 7.7 - Fig. 7.10 for generating and sending a fax message are performed. Fax reception is performed automatically. The fax tones generated by the transmitting and receiving equipment are listened to. To visualize the fax tones, an oscilloscope will be connected to the telephone line, in parallel with the fax equipment.

(e) Deactivate the automatic answer mode and send/receive faxes manually.

3. On computers running Windows 7 operating system, two - three fax transmissions using the Wayjet fax modem card, controlled by the tools integrated into this operating system, are performed. More precisely, the following steps are performed:

   (a) The parameters of the fax user as well as those of the fax modem are configured (see the steps shown in Fig. 7.18 - Fig. 7.19 and Fig. 7.22 - Fig. 7.25).

   (b) Two - three fax transmissions with other fax terminals connected to the PBX are performed. The operations described in Fig. 7.21 for generating and sending a fax message are performed. Fax reception is performed in the automatic answer mode. The fax tones generated by the transmitting and receiving equipment are listened to. To visualize the fax tones, an oscilloscope will be connected to the telephone line, in parallel with the fax equipment. The status of the fax sending/receiving operation is tracked in the monitoring window shown in Fig. 7.26.

   (c) Deactivate the automatic answer and send/receive faxes manually.

4. Set the SF-360 fax equipment in TAM mode and interconnect with the Wayjet modem. Experiment the telephone answering machine functionality in this configuration.

5. The HyperTerminal application is run and the connection configuration steps specified in this chapter are followed (perform the steps presented in Fig. 7.27 - Fig. 7.31). After completing the configuration steps, close the configuration windows (see the configuration windows shown in Fig. 7.27 - Fig. 7.31)) and return to the Connect window and click on the Cancel button - we do not want to make the call at this time. From the main window give the AT command and check if the echo is activated and if the modem is connected to the line. The signaling tones on the telephone line will be listened to in the speaker connected to the audio jack interface of the modem.

6. Some of the basic AT commands are tested:

   - connect/disconnects the modem to/from the line and listen to the signaling tones in the speaker.
   - change (increase/decrease) the level of the speaker.
   - change the tone/pulse dialing mode.
   - make a call to a phone connected to the PBX.
   - a call is made from a telephone to the modem and the automatic answer process is observed; the R0 register must be configured with the number of calls after the modem answers.
   - read the values from various registers of the modem.

7. The data connection is established between two modems and the following operations are performed:

   (a) Send some text sequences and then switch the modem from the data mode to the command mode and execute some simple commands (read the registers, change the speaker level, etc.).

   (b) Switch the modem back into the data mode and check entering this mode by sending some text messages.

(c) Transfer short files using the Xmodem, Ymodem and Zmodem protocols and the functioning of the mentioned protocols is observed (the messages displayed by the modems are observed).

(d) One of the modems is disconnected from the line and observe the disconnect at the other end of the connection (to disconnect the modem it must be switched in command mode).

8. Using the AT commands from the Wayjet dial-up modem user's manual, the display of the transmission information is configured on the UART port. A oscilloscope/spectral analyzer is connected in parallel with the modem and the following operations are performed:

   (a) The modem is configured (by turn) in the ITU-T standards: V.21, V.22, V.23, V.26, V.27, V.32, V.34.

   (b) For each selected standard, the data connection between two modems is realized and the transmitted signals and the spectrum of the modulated signals are observed. The modulation used is identified and the duplex transmission is observed.

   (c) For each selected standard, the synchronization process between the modems is observed on the oscilloscope (and listened to in the speaker).

7.9 Annex

Note: Tab. 7.1 [75] does not include all AT commands, the purpose being not a comprehensive presentation of these commands; there are specific commands for voice modems (modems with telephone answering machine functionality) and there are proprietary AT commands characteristic to certain types of dial-up modems. Specific AT commands for the Wayjet dial-up modem with Intel 536EP chipset can be found in the modem's user manual.

Table 7.1: Dial-up modems AT command set.

Command	Function
AT	Attention - this precedes all commands except A/ and +++
A/	Execute previous command - does not require a $< CR >$.
A	Causes the modem to go off hook. If a call is coming in, the modem will try to answer it. The procedure for answering a call is a short silence and then an answer tone. Sending a character to the modem during this procedure will abort the answer sequence. The amount of time the modem will wait for a carrier is programmable by modifying the S7 register.
B0	Select CCITT V.22 (1200 bps).
B1	Select Bell 212A (1200 bps).
B2	Select CCITT V23 - Originate mode will transmit data at 75 bps and receive data at 1200 bps. Answer mode will transmit data at 1200 bps and receive data at 75 bps. The command N0 (disable auto mode) must be selected.
D	D alone will take the modem off-hook and wait for a dial tone. (see X command for exceptions). The length of time to wait for a dial tone before dialing is programmable in register S6.
Dmn	ATDmn will dial a phone number where m is the modifier: L, W, ,, ;, @, or S. It will dial the telephone number n.
L	Dial last number.
W	Wait for dial tone. If you have selected X0 or X1 (disable dial tone detection), then you can use this modifier to override that setting.
,	Pause during dial. The amount of time to pause is specified in register S8.
;	Return to command mode after dialing. It does not wait for carrier or hang up.
@	Wait for 5 seconds of silence. This is used to access systems that do not provide a dial tone.
!	Hook flash. Causes the modem to go on-hook for 0.5 seconds. This is used in PBX systems and for voice functions like call waiting.
S=(0-9)	Dial a stored number. Up to ten numbers can be stored, and the addresses are from 0 to 9. To store a number into one of these addresses, use the &Z command
E0	Commands issued to the modem are not echoed to the local terminal. This only matters in the command mode. It does not affect the modem's ability to send response codes.

E1	Commands are echoed to the local terminal.
H0	Force modem on-hook (hang-up).
H1	Force modem off-hook (to answer or dial).
I0	Return numeric product code.
I1	Return hardware variation code.
I2	Report internal code.
I3	Report software revision number.
I4	Report product features listing.
L0	Speaker volume zero.
L1	Speaker volume low.
L2	Speaker volume medium.
L3	Speaker volume high (Hardware currently limits volume adjustment to on/off).
M0	Speaker always off.
M1	Speaker on until carrier detected.
M2	Speaker always on.
M3	Speaker on only during answering.
N0	Disable auto-mode. This forces the modem to connect at the speed specified in register S37.
N1	Enable auto-mode. The modem will answer at the highest available line speed and ignore any ATBn command.
O0	Return to data mode. If it was entered the command mode using the time independent escape sequence, this will switch the modem back in data mode without going on-hook.
O1	Retrain the modem. If the line condition has changed since the original connection, retraining the modem will cause it to reconnect at the most efficient speed for the current line condition.
P	Pulse dialing allows the modem to work on telephone networks where DTMF is not supported. Pulse and tone dialing cannot be mixed on the same command line.
Q0	Enable response to DTE.
Q1	Disable response to DTE. The modem does not respond to the terminal. Issuing a command will not produce a response (unless the command is something like ATZ, which will restore this setting to default).
Sn	Set default S-register. Any subsequent = or ? commands will modify the default S register.
Sn=m	Set register n to value m.
Sn?	Return the value of register n.
T	Tone dialing (DTMF)- Pulse and tone dialing cannot be mixed on the same command line.
V0	Result codes will be sent in numeric form. (See the result code table).
V1	Result codes will be sent in working form. (See the result code table).

W0	Report DTE speed only. After connection, there will be no message about what Error Correction or Data Compression protocol is in use.
W1	Report DCE speed, Error Correction/Data Compression protocol, and DTE speed.
W2	Report DCE speed only.
X0	Send OK, CONNECT, RING, NO CARRIER, ERROR and NO ANSWER. Busy and dial tone detection are disabled.
X1	Send X0 messages and CONNECT speed.
X2	Send X1 messages and NO DIAL TONE.
X3	Send X2 messages and BUSY and RING BACK. Dial tone detection is disabled.
X4	Send all responses.
Y0	Disable long space disconnect.
Y1	Enable long space disconnect; with error correction, hang up after sending 1.6 second long space; without error correction, hang up after 4 second long space.
Z0	Reset modem to profile 0.
Z1	Reset modem to profile 1.
+++	This is the default escape sequence. Switches the modem from data mode to command mode. Must be preceded by at least 1 second of no characters and followed by 1 second of no characters. O0 (ATO0 or ATO) returns the modem to data mode.
=n	Sets the value of the default S register.
&	Ampersand commands.
&C0	Force data carrier detect (DCD) on.
&C1	DCD follows remote carrier.
&D0	DTR is assumed on.
&D1	DTR drop causes modem to go back to command mode without disconnecting.
&D2	DTR drop causes modem to hang up.
&D3	DTR drop causes modem to initialize; &Y determines which profile is loaded.
&F	Load factory profile.
&K0	Disable flow control.
&K3	Enable RTS/CTS flow control.
&K4	Enable XON/XOFF flow control.
&K5	Enable transparent software flow control.
&K6	Enable both RTS/CTS and XON/XOFF flow control.
&P0	Selects 33%-67% make/break ratio at 10 pulses per second.
&P1	Selects 33%-67% make/break ratio at 20 pulses per second.

&P2	Selects 39%-61% make/break ratio at 10 pulses per second.
&P3	Selects 39%-61% make/break ratio at 20 pulses per second.
&S0	Force DSR on.
&S1	DSR on at the start of handshaking and off after carrier loss.
&T0	Terminate test.
&V0	Display active profile.
&V1	Display stored profiles.
&V2	Display stored telephone numbers.
&W0	Save active profile to profile 0.
&W1	Save active profile to profile 1.
&Y0	Use profile 0 on power-up.
&Y1	Use profile 1 on power-up.
&Zn=m	Save telephone number (up to 36 digits) into memory location n (0-9).
%	Percent commands.
%A	Default is set to each country encoding law. For example, for USA %A is 0, for Germany %A is 1.
%A0	Mu-law encoding.
%A1	A-law encoding.
%C0	Disable data compression.
%C1	Enable MNP5 compression.
%C2	Enable V.42bis compression.
%C3	Enable both V.42bis and MNP5.
%E0	Disable auto-retrain.
%E1	Enable auto-retrain.
%E2	Enable auto-retrain and fallback to smaller rates.
%E3	Enable auto-retrain and fast hang up.
%L	Report received signal level in -dBm.
%N0	Dynamic CPU loading disabled.
%N1	Dynamic CPU loading not to exceed 10%.
%N2	Dynamic CPU loading not to exceed 20%.
%N3	Dynamic CPU loading not to exceed 30%.
%N4	Dynamic CPU loading not to exceed 40%.
%N5	Dynamic CPU loading not to exceed 50%.
%N6	Dynamic CPU loading not to exceed 60%.
%N7	Dynamic CPU loading not to exceed 70%.
%N8	Dynamic CPU loading not to exceed 80%.
%N9	Dynamic CPU loading not to exceed 90%.
%Q	Report line signal quality.
\	Backslash commands.
\A0	64-character max. MNP block size.
\A1	128-character max. MNP block size.

\A2	192-character max. MNP block size.
\A3	256-character max. MNP block size.
\Bn	In non-error correction mode, transmit break in 100ms units (1-9 with default 3).
\G0	Disable XON/XOFF flow control (modem to modem).
\G1	Enable XON/XOFF flow control (modem to modem).
\Kn	Define break type.
\L0	Use stream mode for MNP.
\L1	Use interactive block mode for MNP.
\N0	Normal mode; speed control without error correction.
\N1	Plain mode; no speed control and no error correction.
\N2	Reliable mode.
\N3	Auto-reliable mode.
\N4	LAPM error correction only.
\N5	MNP error correction only.
*	Asterisk commands.
*Q0	Send the "CONNECT xxxx" result codes to the DTE when an invalid TIES escape sequence is detected after the "OK" response has already been sent.
*Q1	Does NOT send the "CONNECT xxxx" result codes to the DTE when an invalid TIES escape sequence is detected after the "OK" response has already been sent.
Reg. S	S Registers.
Reg. 0	Rings to auto-answer. Sets the number of rings required before the modem answers. 0 setting disable auto-answer. Range is 0-255 rings. Default is 0 for auto-answer disabled.
Reg. 1	Ring counter. Counts the number of rings before the modem answers. Range is 0-255 rings. Default is 0.
Reg. 2	Escape character. Defines the character used for the three-character escape code sequence. 0 setting disables the escape code character. Range is 0-127. Default is 43 (+).
Reg. 3	Carriage return character. Defines the character for carriage return. Range is 0-127. Default is 13 (carriage return).
Reg. 4	Line feed character. Defines the character for the line feed. Range is 0-127. Default is 10 (line feed).
Reg. 5	Backspace character. Defines the character for the backspace. Range is 0-127. Default is 8 (backspace).
Reg. 6	Wait before dialing. Sets the length of time to pause after off-hook before dial. Range is 2-255 seconds. Default is 2 seconds.

Reg. 7	Wait for carrier after dialing. Sets the length of time that the modem waits for a carrier from the remote modem before hanging up. Range is 1-255 seconds. Default is 50 seconds.
Reg. 8	Pause time for dial delay. Sets the length of time to pause for the pause dial modifier ",". Range is 0-255 seconds. Default is 2 seconds.
Reg. 9	Carrier detect response time. Defines the length of time a signal is detected and qualified as a carrier. Range is 1-255 tenths of a second. Default is 6 (0.6 seconds).
Reg. 10	Lost carrier hang up delay. Sets the length of time the modem waits before hanging up for a carrier loss. Range is 1-255 tenths of a seconds. Default is 14 (1.4 seconds).
Reg. 11	DTMF speed control. Sets the length of tone and the time between tones for the tone dialing. Range is 50-255 milliseconds. Default is 95 milliseconds.
Reg. 12	Escape Prompt Delay (EPD) timer. Sets the time from detection of the last character of the three character escape sequence until the "OK" is returned to the DTE. Range is 0-255 fiftieths of a second. Default is 50 (1 second).
Reg. 13-17	Reserved.
Reg. 18	Test timer. Sets the length of loopback test. Range is 0-255 seconds. Default is 0 (disable timer).
Reg. 19-24	Reserved.
Reg. 25	Delay to DTR. Sets the length of time the modem ignores DTR before hanging up. Range is 0-255 hundredths of a seconds. Default is 5 (0.05 seconds).
Reg. 26-28	Reserved.
Reg. 30	Disconnect inactivity timer. Sets the length of time allowed for inactivity before the connection is hung up. Range is 0-255 in minutes. Default is 0 (disabled).
Reg. 32	XON character. Sets the value of XON character. Range is 0-255. Default is 17.
Reg. 33	XOFF character. Sets the value of XOFF character. Range is 0-255. Default is 19.
Reg. 34	56k data rate (bit-rate). Sets the maximum bit rate for 56k modem. Range is 0-32. Bit rate = 32000bps + S34*2000bps. V.34 data rate (bit-rate). Sets the maximum bit rate for V.34 modem. Range is 0-8 (2400 baud), 1-10 (3000 baud), 1-11 (3200 baud), 1-13 (3429 baud). Bit rate = ((S34)+1)*2400bps. Default is 13 (33600 bps)

Reg. 36	Reserved.
Reg. 37	Line connection speed. 0-Attempt to connect at the highest speed. 3-Attempt to connect at 300 bps. 4-Attempt to connect at 1200 bps. 6-Attempt to connect at 2400 bps. 7-Attempt to connect at 4800 bps. 8-Attempt to connect at 7200 bps. 9-Attempt to connect at 9600 bps. 10-Attempt to connect at 12000 bps. 11-Attempt to connect at 14400 bps. 12-Attempt to connect at V.34. 13-Attempt to connect at 56K, Default is 0.
Reg. 38	Delay before forced hang up. Sets the delay to hang up after the disconnecting command is received. Range is 0-255 seconds. Default is 20 seconds.
Reg. 39-48	Reserved.
Reg. 82	Reserved.
Reg. 86	Call failure reason code. 0-Normal disconnect (no error), 4-Loss of carrier, 5-V.42 negotiation failed to detect an error correction modem at remote end, 6-No response to complete negotiation, 9-No common protocol, 12-Remote modem initiated a normal disconnect, 13-Remote modem did not respond after 10 message retransmissions, 14-Protocol violation, 15-Compression failure, 20-Hang up by inactivity time out.
Reg. 91	Transmit level. Set the transmit level in -dBm. Range is 9-15 (-dBm). Default is 11 (-11 dBm)

Chapter 8

Digital ADSL access techniques on the subscriber loop

8.1 Introduction

This chapter proposes the study of the following aspects related to ADSL (Asymmetric Digital Subscriber Line) access techniques:

- basic theoretical aspects and principles of ADSL transmissions.
- basic theoretical aspects regarding the ADSL protocol stack.
- the architecture of the ADSL digital access network and the entities of this network.
- interfacing and interconnecting the entities in the ADSL access network.
- configuration of some basic parameters of ADSL equipment.

The chapter focuses in particular on the practical aspects related to connection in the telephone network of the ADSL equipment, both at the user side (ADSL modem) and at the exchange side (DSLAM - DSL Access Multiplexer), and on the basic configuration of these equipment. The aims are to understand the architecture of an ADSL access network, to explain the configuration of DSL equipment for users access to the Internet, and to identify and analyze the parameters that characterize the ADSL digital connection.

8.2 Theoretical notions

XDSL Access Techniques. Characteristics and allocation of frequency bands

XDSL digital access techniques refer to technologies and equipment used in the telephone access network to ensure the connection on twisted copper wires to a high speed digital network. It represents the cheapest solution for accessing digital communications services [76] [77]. The access network comprises both the transmission lines and the equipment located between the subscriber's terminal equipment (the subscriber's premises) and the access node in the network. Depending on the services offered, the network access node can be a telephone exchange, a cable TV terminal (CATV headend), a data server, etc. In telecommunication networks with integration of the services (ISDN networks) the terminals can be telephones, videophones, personal computers, TVs or various control systems of some equipment located at the subscriber's premises. The broadband services at the subscriber level can be: video telephony, trading, video conferencing, distance learning or telesupervision, video on demand, work at home, teleshopping or telemedicine. There are two basic categories of xDSL access techniques, namely [76] [77]:

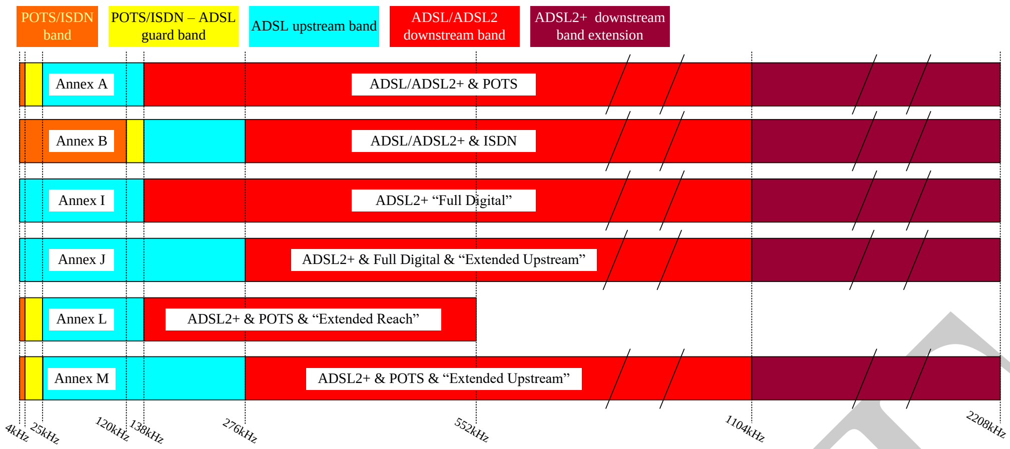

Figure 8.1: Frequency band allocations characteristic to ADSL, ADSL2 and ADSL2+ standards.

- SDSL - Symmetric DSL. This technique ensures the same transfer rate in both directions of transmission:
  - upstream (US): subscriber - central equipment.
  - downstream (DS): central equipment - subscriber.

  Due to the attenuation and crosstalk these systems can only work at medium frequencies. SDSL is ideal for connecting to the transport network of local networks (LAN - Local Area Network), bidirectional video terminals or web servers.

- ADSL - "Asymmetric DSL". This technique provides in the downstream a broadband channel, located at high frequencies, and in the upstream a narrower-band channel located at low frequencies.

The allocation of ADSL frequency bands has two reasons: the amount of information transmitted in downstream is usually higher in the case of typical subscribers (domestic subscribers, small offices, etc.) and in addition it reduces the NEXT, which is higher at high frequencies. ADSL can provide rates between 8 and 16 Mbps in downstream (depending on the configuration of the transport channels implemented over the physical ADSL layer) and up to 1Mbps in upstream [50].

The ADSL technique was initially developed (in the late 1980s) for video on demand transmissions on twisted wires. With the development of the Internet the ADSL technology was (and it is) used as a user access technique to the Internet.

The ADSL technique uses a single pair of twisted wires and allows the coexistence with the data transmission of the standard telephone service (POTS) or the ISDN digital access service [50].

ADSL2 and ADSL2+ techniques (an extension of the ADSL2 standard) are asymmetrical access techniques that bring a series of improvements to the ADSL standard [78]. The ADSL2+ technique ensures downstream rates between 24 and 32Mbps and upstream rates between 1 and 4Mbps.

The frequency band allocation of ADSL, ADSL2 and ADSL2+ standards is shown in Fig. 8.1. The ADSL/ADSL2 downstream band extends up to 1.1MHz, and ADSL2+ extends the downstream band up to 2.2MHz. The two downstream sub-bands, i.e. 0.138MHz - 1.1MHz and 1.1MHz - 2.2MHz, can be allocated together or separately, either of them. ADSL standards include several annexes with specifications for specific situations. Appendix A refers to the situation where the standard telephone service is provided on the subscriber loop, and Annex B to the situation where the basic ISDN service (ISDN BA - Basic Access) is provided on the subscriber loop. These annexes are characteristic for all ADSL standards. The other annexes are characteristic only for ADSL2+ standards and are the following:

- Annex I describes the full digital mode, where no other services are available on the subscriber loop and the POTS or ISDN spectrum is allocated to the ADSL upstream band.

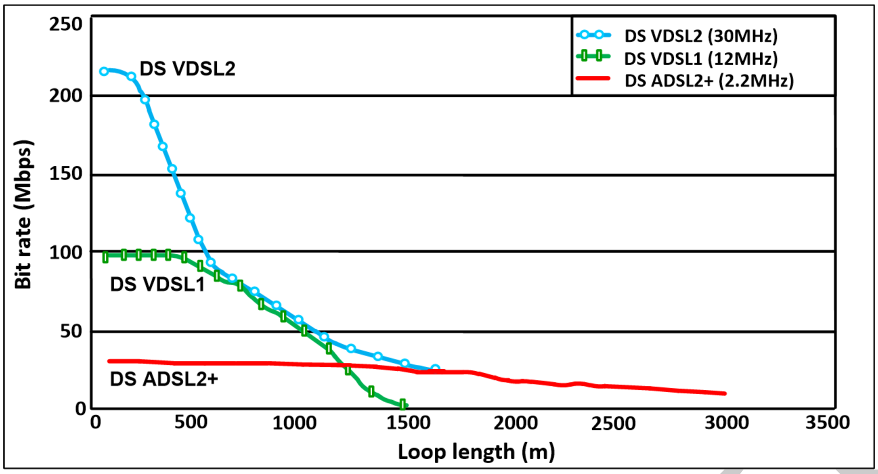

Figure 8.2: Changing the VDSL2 working mode depending on the length of the subscriber loop.

- Annex J describes the fully digital working mode with upstream bandwidth extension, i.e. the upstream band doubled.

- Annex L describes a working mode which provides extended coverage (longer cables) in the situation where the POTS service is offered. The coverage extension leads to a reduction of the bandwidth allocated for downstream, due to the poor quality of the channel on long twisted wires at high frequencies.

- Annex M describes the working mode with upstream bandwidth extension and POTS service on the subscriber loop.

The ADSL2/ADSL2+ standards additionally bring a number of other supplementary features, such as: power management, continuous rate adaptation, fully digital mode, sub-channel definition for various services, crosstalk reduction techniques, impulse noise protection, use of several twisted pairs (Bonded ADSL) [78].

VDSL (Very high bit rate Digital Subscriber Line) is an xDSL technique that allows both asymmetrical and symmetrical operation mode. The coverage area extends from tens of meters to hundreds of meters. In the asymmetrical mode it allows downstream transfer rates up to 50Mbps and upstream rates up to 6Mbps. In symmetrical working mode the VDSL technique allows (symmetrical) transfer rates up to 34Mbps. The VDSL technique works on a single twisted pair and allows the analog telephone service. The maximum bandwidth is 12MHz [4].

The VDSL2 standard brings a number of improvements such as higher throughput, better adaptation to the channel characteristic, adaptive working mode (if the cable length increases, it switches to VDSL or ADSL2 working mode, see Fig. 8.2), better control of interference (crosstalk), fully digital working mode, impulse noise protection, power management, use of multiple twisted pairs, etc. The VDSL2 technique can provide downstream speeds of up to 250Mbps. The maximum bandwidth is 30MHz [79].

ADSL duplexing methods and line connection of equipment

In the case of ADSL transmissions, there are two methods for separation of the upstream and downstream transmission directions, namely: separation by frequency division and separation by echo compensation [4], the two situations being presented in Fig. 8.3 and Fig. 8.4. The figures also show the filtering functions that are used on the ATU-C (ADSL Transceiver Unit-Central) equipment, that is the ADSL access multiplexer, and the ATU-R (ADSL Transceiver Unit-Remote), that is the ADSL modem, as well as the spectral distribution of the signals transmitted/received by the two equipment mentioned above.

The separation of the transmission directions by frequency division is more inefficient, being allocated separated frequency bands for upstream and downstream, but it ensures a better separation of the two transmission directions. The separation of the transmission directions by the echo compensation technique

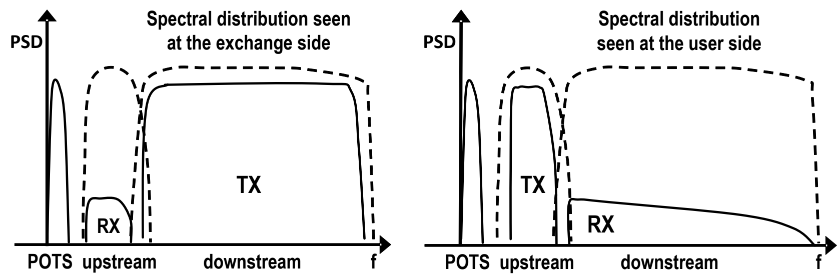

Figure 8.3: Separation of the upstream and downstream transmission directions by frequency division. Filtering functions and power spectral distributions.

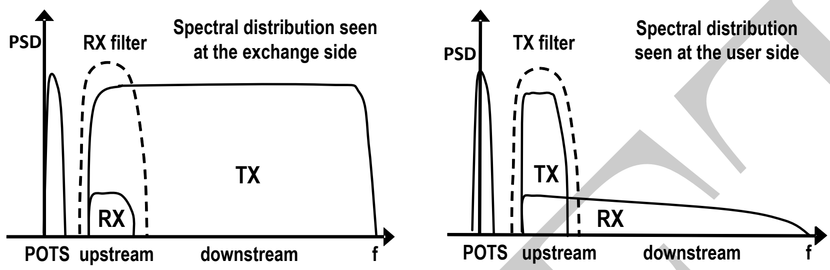

Figure 8.4: Separation of the upstream and downstream transmission directions by the echo compensation technique. Filtering functions and power spectral distributions.

ensures a more efficient use of the frequency band, more precisely the downstream band and implicitly the downstream rate, will be greater, but the separation of the upstream and downstream transmission is weaker, especially under crosstalk and limited dynamic range conditions of the echo compensator.

The overlap of digital (DS and US) signals and telephone (or ISDN) signals on the subscriber loop requires the use of an ADSL splitter [4] [50]. The effective way of connecting the ADSL equipment and the telephone equipment to the twisted pair of wires, when using the ADSL splitter, is illustrated in Fig. 8.5, and the internal schematic of an ADSL splitter is shown in Fig. 8.6.

The ADSL splitter includes a two way low pass filter (LPF) with a cutoff frequency of approximately 4kHz and may include optional capacitors ($C \approx 20 - 30nF$) that block the DC voltage on the subscriber loop to the ADSL modem. The LPF in the splitter separates the voice band in which the classic telephony works, and the high pass filter (HPF), included in the ADSL modem, separates the frequency band in which the ADSL transmission is performed. If the ADSL splitter is not used, the telephone band separation LPF is missing, which results in a decrease of the ADSL transmission performance for two reasons:

- impedance matching issues may occur between the ADSL modem and the telephone line, the LPF also having the role of separating the input impedance of the telephone devices from the telephone line (the impedance of the telephone devices is not the same during the idle, talk and dialing period).

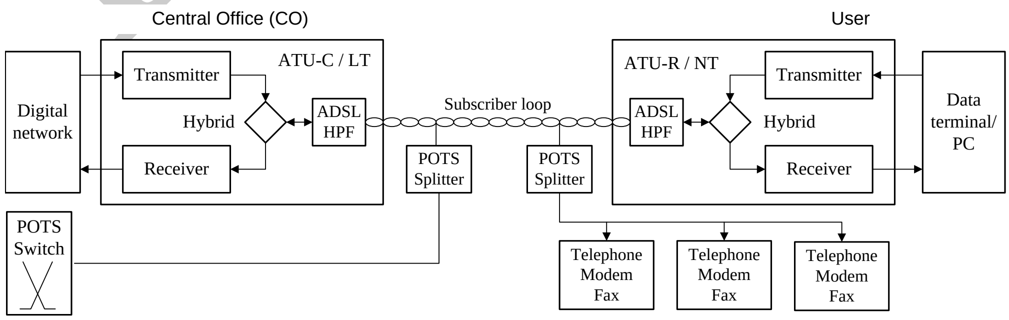

Figure 8.5: Connection to the subscriber loop of ADSL equipment and telephone equipment when using the ADSL splitter.

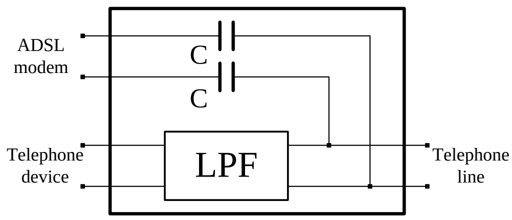

Figure 8.6: Internal schematic of an ADSL splitter.

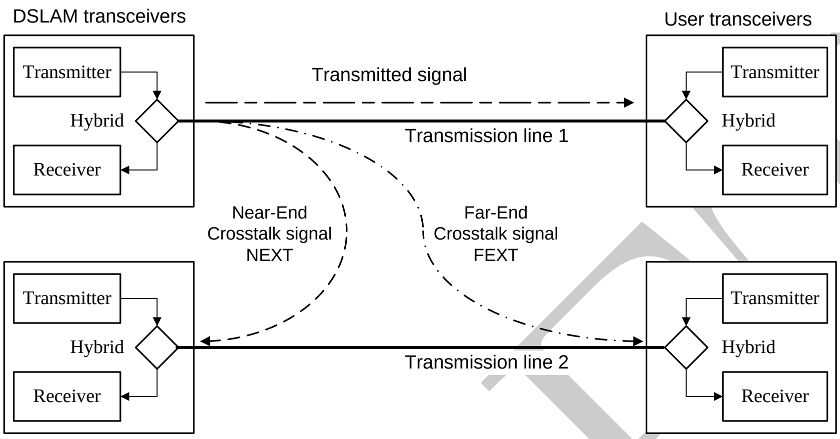

Figure 8.7: NEXT and FEXT between twisted pairs used in ADSL transmissions.

- telephone equipment can generate frequency components (harmonics) that fall outside the telephone band and are located in the ADSL frequency band. These components are mainly generated by the call signals (ring signal, disc pulses). This explains the lower rates provided by the G.Lite ADSL access technique, a technique that does not use ADSL splitter.

Distortions affecting the ADSL transmission. The modulation technique used

ADSL transmissions on the subscriber loop are affected by the following distortions [4] [18]:

- nonuniform attenuation - frequency function of the twisted wires. The distortion is more pronounced at large lengths of the subscriber loop (of the order of km).

- impedance mismatches generated by the use in the subscriber loop of cable sections with different diameters and by unterminated bridge taps, i.e unused twisted pairs. These bridge taps must be removed from the subscriber loops on which ADSL transmissions are used. The reflected signals generated by the impedance mismatches can create spectral zeros in the ADSL frequency band that are difficult to equalize.

- noise (white noise, shot noise, impulse noise) and electromagnetic interference from the environment.

- nonlinear distortions generated by the amplifiers in the ADSL modems as well as by the amplifiers on the transmission line.

- crosstalk between the pairs of the same cable.

Crosstalk has two components, namely, Near End Crosstalk - NEXT, which occurs at the close ends of the pairs, and Far End Crosstalk - FEXT, which appears at the far ends of the pairs (see Fig. 8.7) [1]. Due to the allocation of the upstream and downstream bands the NEXT has a smaller effect (the attenuation of the crosstalk is high at low frequencies) or has no effect if duplexing is realized by frequency division. FEXT is a noise component that dependents on the transmitted signal with great importance on the performance of digital transmissions on the subscriber loop [80].

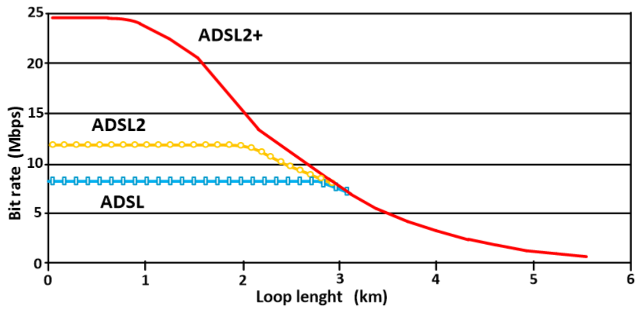

Figure 8.8: Downstream ADSL, ADSL2 and ADSL2+ bit rates function of the length of the subscriber loop in white noise conditions.

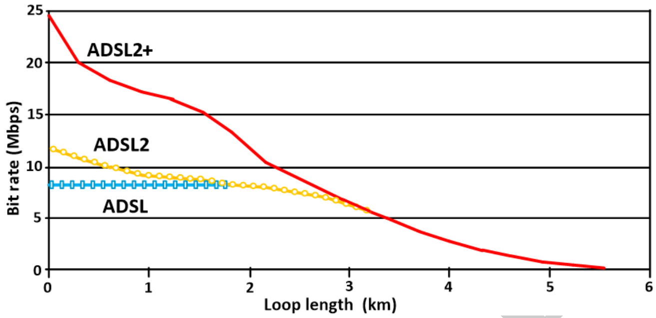

Figure 8.9: Downstream ADSL, ADSL2 and ADSL2+ bit rates function of the length of the subscriber loop in crosstalk conditions.

In Fig. 8.8 and Fig. 8.9 are presented the performance of some ADSL, ADSL2 and ADSL2+ transmissions, more precisely the maximum downstream bit rate depending on the length of the subscriber loop, in the condition of white noise on the channel (without crosstalk) and in the condition in which the transmission is affected by crosstalk generated by other identical digital transmissions (self crosstalk) on 12 neighboring pairs of the cable. The cable used has a diameter of 0.4mm (26awg - American Wire Gauge), typical for subscriber loops, and the spectral density of white noise is -140dBm/Hz, a typical value for cable transmissions. The figures clearly show the significant effect of the crosstalk on the high speed ADSL transmissions (ADSL2/ADSL2+ transmissions), as well as the effect of the subscriber loop length on these transmissions.

Ensuring high bit rates under severe attenuation distortions, impulse noise and interference (crosstalk and external interference) requires more "special" modulation techniques that can cope with the aforementioned distortions. The modulation technique used by ADSL transmissions is DMT (Discrete Multitone), a multicarrier modulation technique [4] [18]. The basic principle, presented in Fig. 8.10, is to separate the data stream into a number of N substreams, each being transmitted on a separate carrier (subcarrier or tone) on a separate subchannel. The transmission on each subcarrier is independent of the other transmissions and the QAM modulation constellation on each subcarrier is chosen according to the subchannel characteristic and the noise/interference level in the subchannel.

The separation of the subcarriers is done under orthogonality conditions and not by filtering, which allows adjacent subchannels without the need for separation guard bands. If f_s is the bandwidth of a subchannel, then the complex subcarriers are $e^{jk\omega_s t}$, where k is the index of the subchannel. The orthogonality condition imposed on subcarriers is as follows: $\frac{1}{T_s}\int_0^{T_s} e^{jm\omega_s t}\cdot e^{jn\omega_s t}dt = \begin{cases} 0\ if\ m \neq n \\ 1\ if\ m = n \end{cases}$. The symbol frequency of DMT modulation is also f_s. The digital sequence applied to a subcarrier is not filtered, as a rule, and the filters with impulse response g(t) in Fig. 8.10 may be missing. The value of f_s is 4.3125kHz for ADSL transmissions.

In the case of an ADSL transmission, 255 subcarriers are defined, and in the case of an ADSL2+ transmission, 511 subcarriers are defined. Not all subcarriers are allocated, due to the bandwidth reserved for the telephone transmission and the guard bands between the telephone band and the ADSL band, respectively between the upstream and the downstream bands (if we have frequency division duplexing). One group of subcarriers is

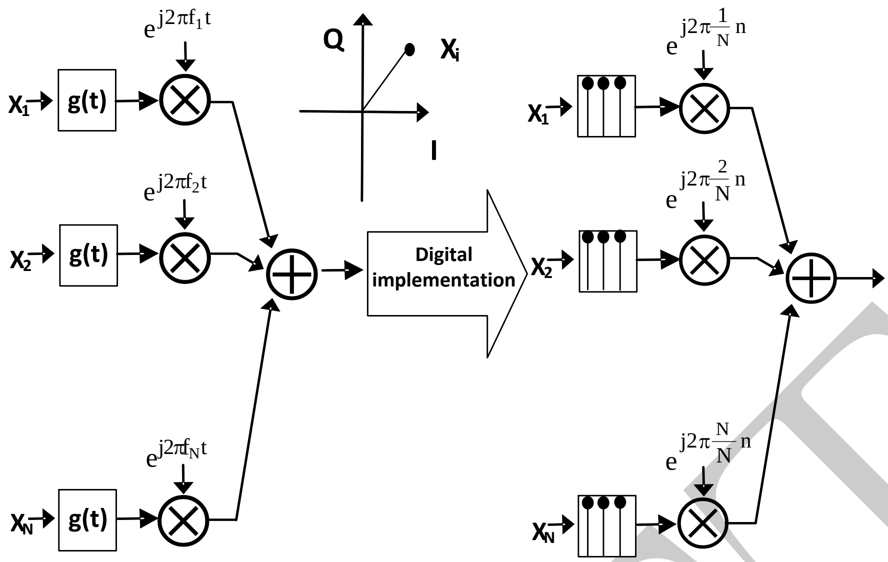

Figure 8.10: The principle of DMT modulation.

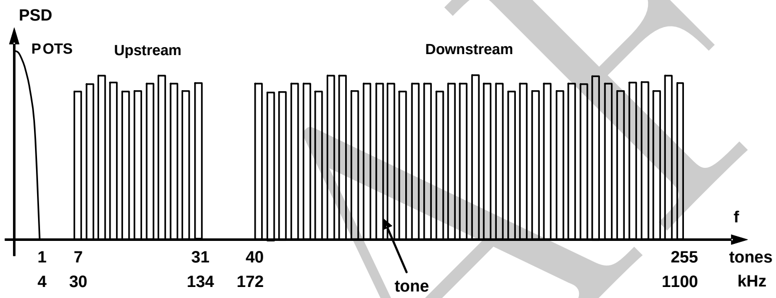

Figure 8.11: Allocation of frequencies and tones for ADSL transmissions using DMT modulation and frequency division for separation of transmission directions.

allocated for upstream and another group for downstream. The two groups are not disjoint in the case of duplexing by echo compensation technique. Fig. 8.11 shows the allocation of subcarriers in the case of ADSL transmissions with frequency division duplexing.

8.2.1 ADSL transmission's protocol stack

The ADSL transmission's protocol stack [50] [81] [82] is shown in Fig. 8.12. The routing in the Internet network is performed by the Internet Protocol (IP Protocol) using IPv4 or IPv6 addresses. Most xDSL devices work with IPv4 addresses.

In the second layer of the OSI (Open Systems Interconnection) protocol stack, that is, the Data Link layer we can have STM (Synchronous Transfer Mode) or ATM (Asynchronous Transfer Mode) protocols depending on the type of data stream being transmitted at this level, namely STM or ATM flows. Most xDSL equipment transmit ATM flows.

ATM protocols

The ATM protocols provide addressing of the nodes in the operator's core network (which uses such technology), multiplexing of the data flows and managing the resources in the core network to ensure the QoS (Quality of Service) requirements for different services [83] [84]. The transmissions made in the ATM network as well as the addressing (that is, identifying the connections/channels) are based on the concepts of Virtual Path (VP) and Virtual Channel (VC). The virtual path may include multiple virtual channels established between two nodes.

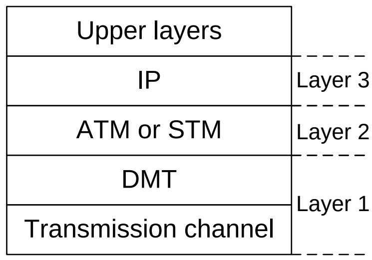

Figure 8.12: The ADSL protocol stack.

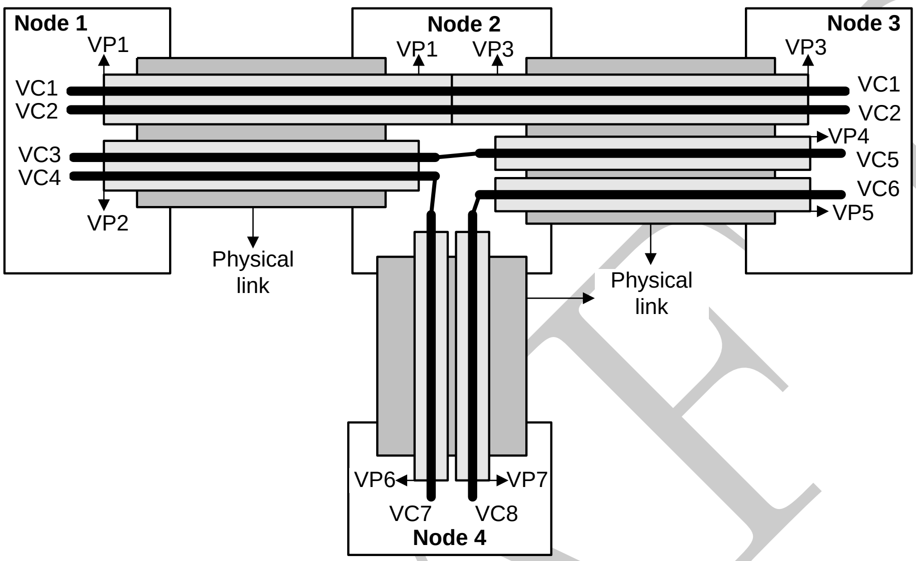

Figure 8.13: Virtual paths and virtual channels established in an ATM network.

The concepts of VP and VC as well as the interconnection of these paths and channels are illustrated in Fig. 8.13.

The ATM transmission is based on its own frame structure, called ATM cell, small size frame, of only 53 bytes, of which the first 5 bytes represent the header of the ATM cell. The header contains the VP and VC numbers, the data type (payload), the flow control class, the cell priority, and an 8-bit Cyclic Redundancy Check (CRC) for header error control (header only) [83]. The stack of ATM protocols, i.e. the sublayers of the data link layer characteristic for ATM transmissions, is shown in Fig. 8.14 [84].

The ATM layer controls the flow of cells, establishes and interrupts virtual circuits and paths. The ATM adaptation layer ensures the identification and classification of data flows arriving from the upper layers (Convergence Substrate) and sets the parameters of the ATM transmission according to the QoS requirements of the services that generate the flows in question. The segmentation/reassembly substrate ensures the division of the packets of data of larger length, coming from the upper layers, in the ATM cells, respectively the reassembly of these packets at the remote end of the connection.

As shown in Fig. 8.14 ATM protocols classify services into four classes, namely: constant bit rate services, variable bit rate real time audio/video services, and connection oriented or connectionless data services. The different classes of services are characterized by the parameters of the ATM cells such as:

- the maximum cell rate (Peak Cell Rate - PCR) - is defined for services with constant rate.
- the Sustainable Cell Rate (SCR) - is defined for burst rate services.
- the Maximum Burst Size (MBS) - is defined for burst rate services.
- the Minimum Cell Rate (MCR) - is defined for services with guaranteed rate.
- the burst tolerance (Burst Tolerance - BT) - is defined for burst rate services.

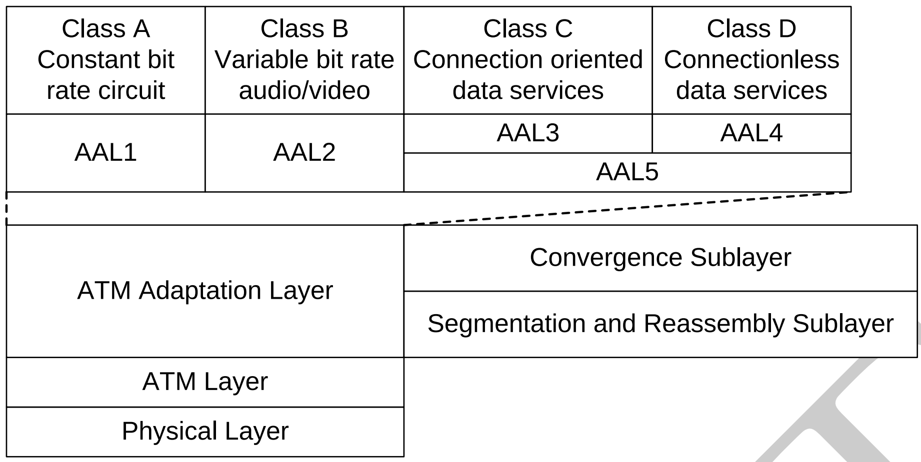

Figure 8.14: The ATM protocol stack.

- tolerance to cell jitter (Cell Delay Variation Tolerance - CDVT) - is defined for burst rate services.

Regarding the multiplexing of data flows generated by different applications running in parallel we have two possibilities:

- LLC multiplexing - the flows generated by different applications are transmitted on the same virtual channel.
- VC multiplexing - each application uses a separate virtual channel (VC), channels that are then multiplexed on the virtual paths (VP).

ADSL access network architecture

The architecture of the ADSL access network is shown in Fig. 8.15. Besides the "classic" elements, i.e. ADSL modem, splitter, subscriber loop, etc. can be noticed the existence of several distinct networks, namely LANs (Local Area Network), on the user side, and a WAN (Wide Area Network).

The user can connect to the ADSL modem several data terminals (personal computers, mobile terminals, etc.) through an Ethernet switch or a WiFi access point. Some ADSL modems have integrated Ethernet switch and/or WiFi access point. ADSL modems also integrate an IP router, as does the DSL access multiplexer. The router in the ADSL modem has one or more LAN interfaces and a WAN interface. This router ensures the routing of the IP packets received on the LAN interface/interfaces through the WAN network to the DSLAM router. The control of the users access to the Internet is usually ensured by a dedicated server and a Point-to-Point Protocol (PPP) [82] [85]. With the help of this protocol the users authentications are performed and the IP address of the WAN routers of the ADSL modems are assigned.

The DSLAM module has a built-in splitter for each DSL port. The connection of the DSLAM module to a transport network is usually done on an Ethernet interface.

8.3 ADSL equipment

8.3.1 ADSL modems and ADSL splitters

Thompson SpeedTouch 516 (ST516) modem

It is an ADSL/ADSL2+ modem with built-in IP router and Ethernet network interface [86]. The front of the modem, where several indicator LEDs are placed, is shown in Fig. 8.16, and the back, where the LAN/WAN network sockets, the power button, etc. are placed, is shown in Fig.8.17.

There are 4 LEDs on the front of the modem. The Power LED shows that the modem is turned on when it is green. During the initialization of the modem, after power on, the Power LED is red. The Ethernet

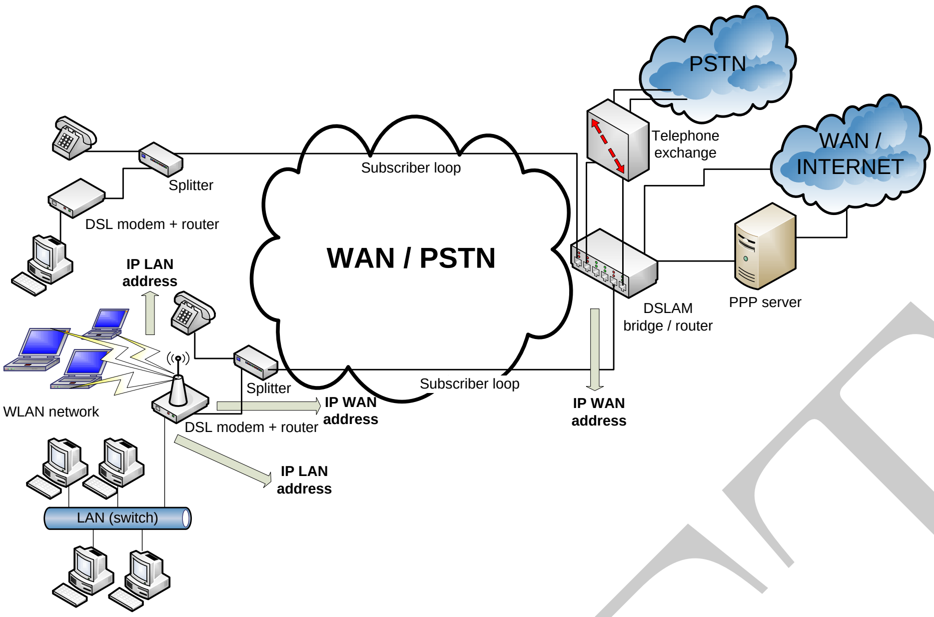

Figure 8.15: The architecture of the ADSL access network.

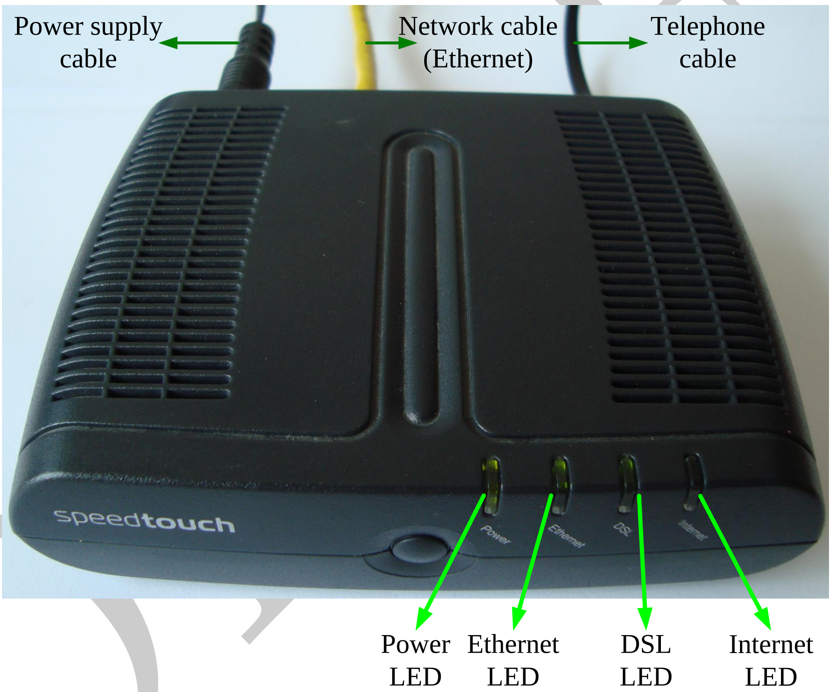

Figure 8.16: The front of the ADSL/ADSL2+ ST516 modem.

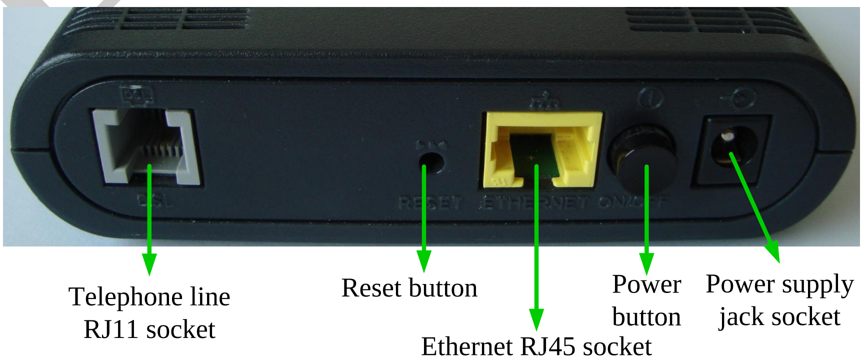

Figure 8.17: The back of the ADSL/ADSL2+ ST516 modem.

LED shows that the modem is connected to an Ethernet network card of a data terminal and that the data connection between the modem and the computer is established. During the initialization of this connection, i.e. the transmission rate is established (10Mbps or 100Mbps) and the IP address of the data terminal interface is assigned by the modem, this LED blinks. The DSL LED indicates that the data connection between the modem and the DSLAM equipment is established, i.e. the modem and DSLAM are synchronized and the physical DSL connection as well as the ATM data channels are established. During the modem and DSLAM synchronization process the LED blinks. In the initial phase of the synchronization process it blinks more slowly, and in the final phase it blinks faster, and after the synchronization is achieved, it lights continuously. The Internet LED lights up if Internet access is possible, i.e. the modem authentication on the network, respectively the assignment the WAN IP address to the modem router have been successfully completed.

On the back the ST516 modem has a RJ11 socket for the telephone line, an RJ45 Ethernet socket for connecting the data terminal, a jack socket for connecting the external power supply, the power button, respectively the reset button, which must be held down for a few seconds to reset the modem. Connecting multiple data terminals to the ADSL modem requires the use of an external Ethernet switch.

ZyXEL P-660-T1 modem

It is also an ADSL/ADSL2+ modem with built-in IP router and Ethernet network interface [87]. The front of the modem, where several indicator LEDs are placed, is shown in Fig. 8.18, and the back, where the LAN/WAN network sockets, the power button, etc. are placed, is shown in Fig. 8.19. The modem has 5 indicator LEDs namely: Power LED, lights when the modem is powered, Ethernet LED, lights when the modem - data terminal (computer) connection is established, DSL LEDs, lights when the modem - DSLAM connection is established and the two equipment are synchronized. During synchronization, the DSL LED blinks, slower in the first phase and then faster in the second phase of the synchronization. The Internet LED lights when the Internet connection is established (modem authenticated, WAN IP addresses assigned to the modem), and the USB LED lights if the connection between the modem and the data terminal is made over a USB port and this connection is established.

On the back the P-660 modem has a RJ11 socket for the telephone line, an RJ45 Ethernet network socket for connecting the data terminal, a USB socket used as an alternative for connecting the data terminal to the modem. We also have a jack for connecting the external power supply, power button and reset button.

Huawei HG655b modem

It is an ADSL/ADSL2+/VDSL2 modem with IP router, Ethernet switch and built-in WiFi access point [88]. The front of the modem, where several indicator LEDs are placed, is shown in Fig. 8.20, and the back, where the LAN WAN network sockets, the power button, etc. are placed, is shown Fig. 8.21.

On the front of the modem can be found the indicator LEDs presented in the case of the other ADSL modems, i.e. the Power LED, the DSL LED, the Internet LED. In this case we have several LAN/Ethernet LEDs because the modem includes an Ethernet switch. We also have a WLAN/WiFi LED that indicates the activation of the wireless interface. We also have additional LEDs, namely WPS (WiFi Protected Setup) which indicates a certain way of securing the WiFi network, a VoIP (Voice over IP) LED that lights if VoIP functionality over the DSL line is activated and a USB LED that shows the use of the USB port available on the modem.

On the back the HG655b modem has a RJ11 socket for the telephone line and two RJ11 sockets for connecting analog phones. These sockets are used if the voice connection is provided by VoIP technology over the DSL connection, in which case the phones connect directly to the modem. If the telephone service is traditionally offered on the subscriber loop, these sockets are not used. We have four RJ45 Ethernet sockets to which 4 data terminals (computers) can be connected and we have a jack socket for connecting the external power supply. The power button, the reset button, the WLAN activation button, and a USB socket are located on the side of the modem.

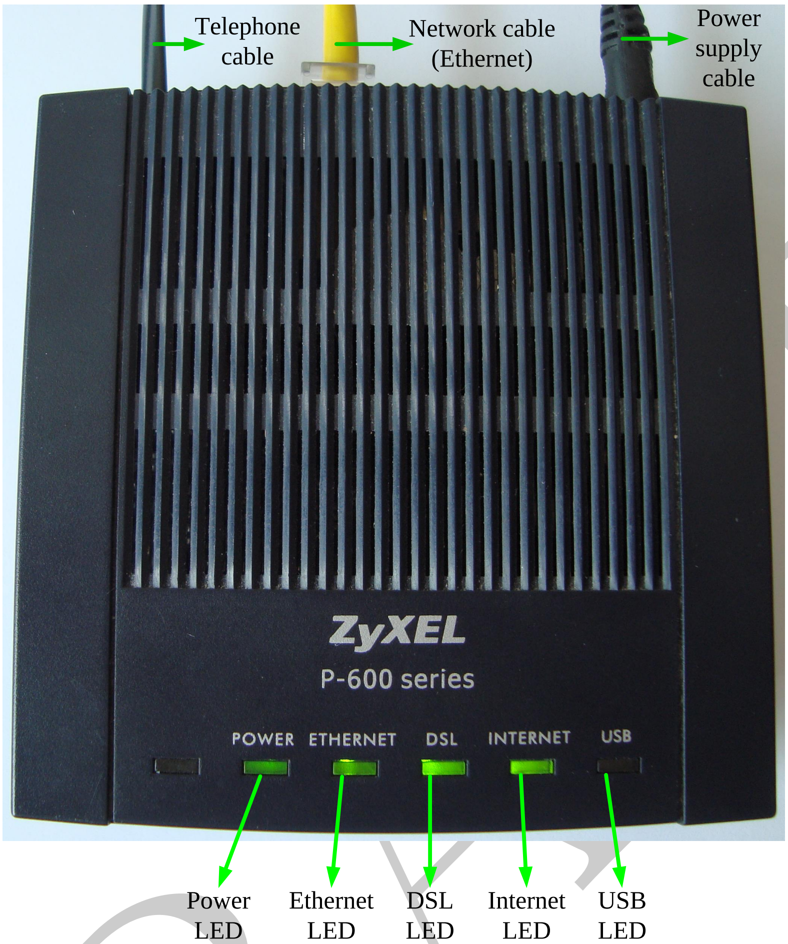

Figure 8.18: The front of the ADSL/ADSL2+ P-660 modem.

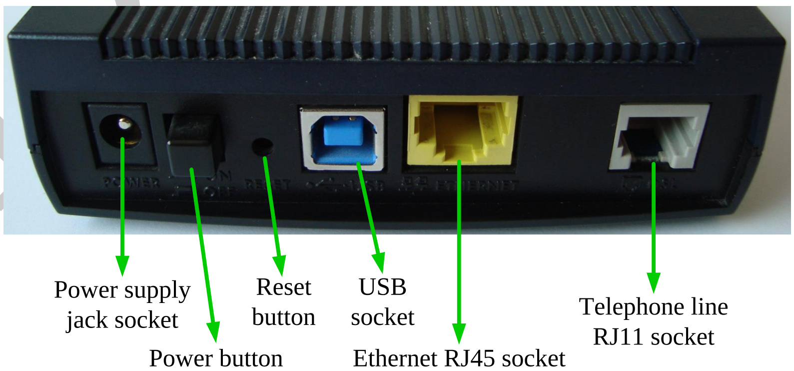

Figure 8.19: The back of the ADSL/ADSL2+ P-660 modem.

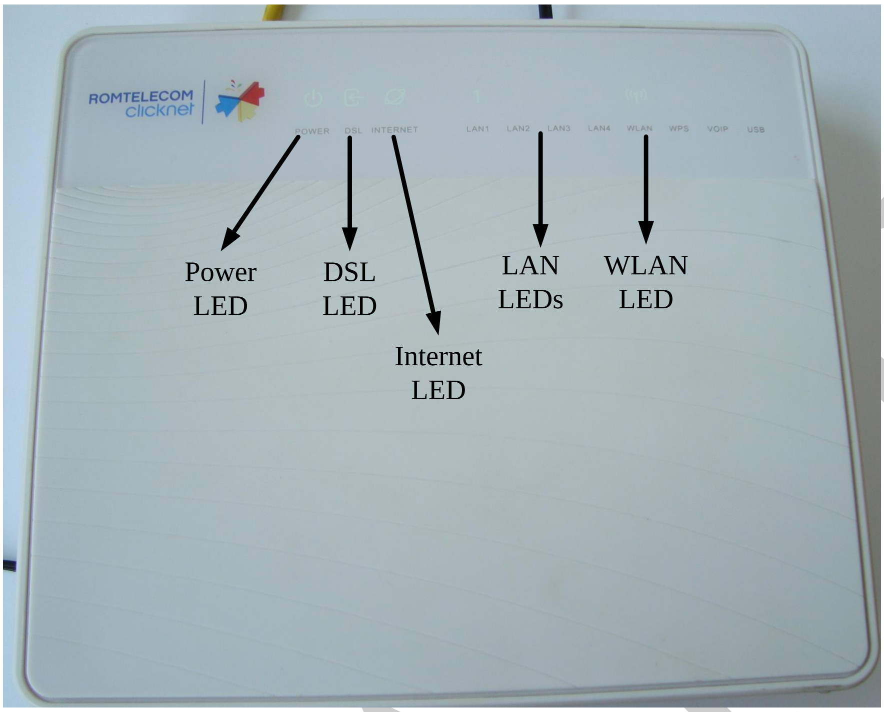

Figure 8.20: The front of the ADSL/ADSL2+/VDSL2 HG655b modem.

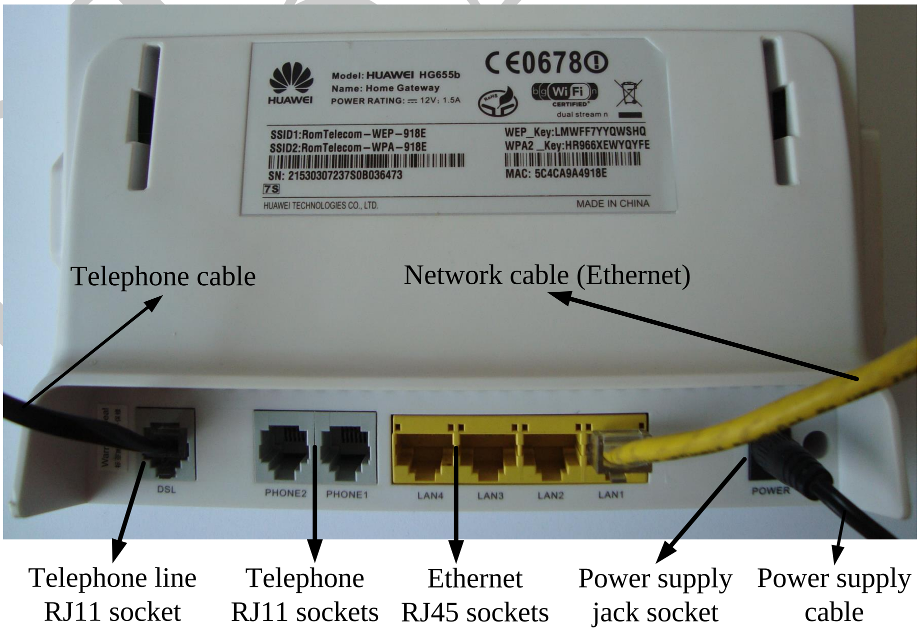

Figure 8.21: The back of the ADSL/ADSL2+/VDSL2 HG655b modem.

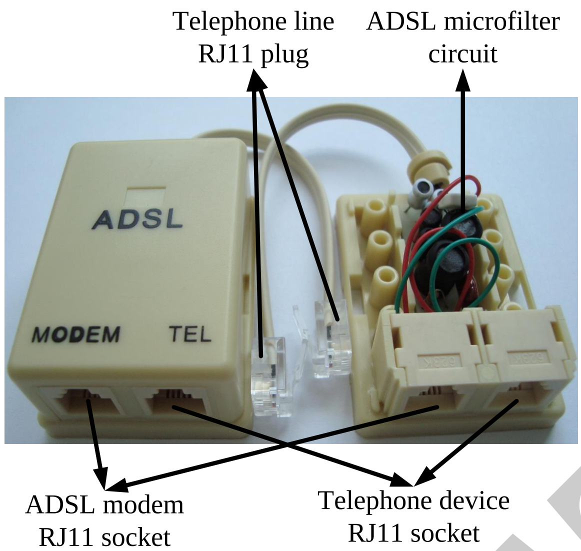

Figure 8.22: ADSL Splitter. Internal structure.

ADSL splitters and microfilters

Connection to the subscriber loop of the ADSL modem and of different POTS terminals (telephone, fax, modem) requires an ADSL splitter. If several telephone terminals are connected to the subscriber loop then they are separated by an ADSL microfilter. In Fig. 8.22 and Fig. 8.23 are presented several types of ADSL splitters, and in Fig. 8.24 are presented several types of ADSL microfilters. In Fig. 8.22 can be seen the internal structure of an ADSL splitter, more precisely can be seen the microfilter, respectively the RJ11 sockets of the distributor in the ADSL splitter.

8.3.2 Configuring ADSL modems

Accessing ADSL modems

Configuration of an ADSL modem requires first to access it, for which it is necessary to determine the IP LAN address of the modem. Usually the modem's router runs a Dynamic Host Configuration Protocol (DHCP) server that assigns IP addresses to all data terminals connected to the modem. If, on the computer, on which the ADSL modem configuration is to be performed, the Windows XP operating system runs, the following procedure will be followed to determine the IP LAN address of the ADSL modem (see Fig. 8.25):

- click on the Start button and select Control Panel.
- from the Control Panel window select Network Connections.
- in the window that appears all network interfaces of the computer are displayed; click on Local Area Connection.
- in the window that appears, select Support and Details.
- in the window that appears are displayed the MAC and IP address of the computer, the Default Gateway address, the address/addresses of the DNS server/servers and other information. The modem's LAN IP address is the IP Default Gateway address.

If, on the computer on which the ADSL modem configuration is to be performed, the Windows 7 operating system runs, the following procedure will be followed to determine the LAN IP address of the ADSL modem (see Fig. 8.26):

- click on the network icon on the right side of the Taskbar or click on the Start button, then click on the Control Panel and select Network and Internet.

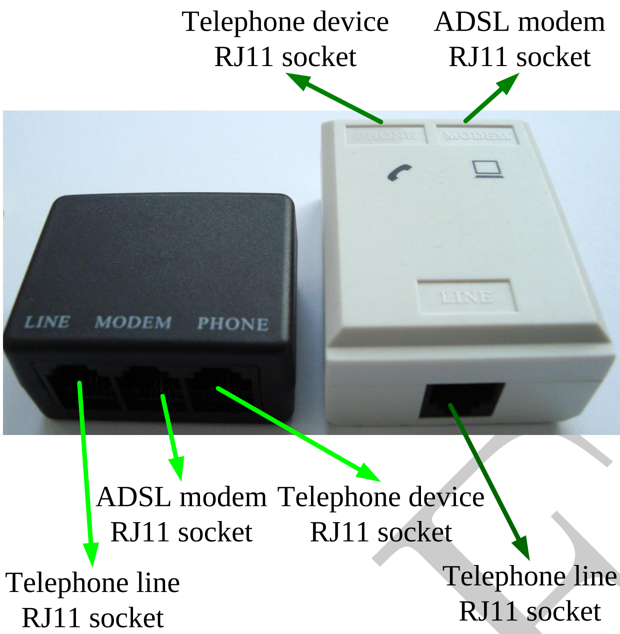

Figure 8.23: ADSL splitters of various types.

Figure 8.24: ADSL microfilters of various types.

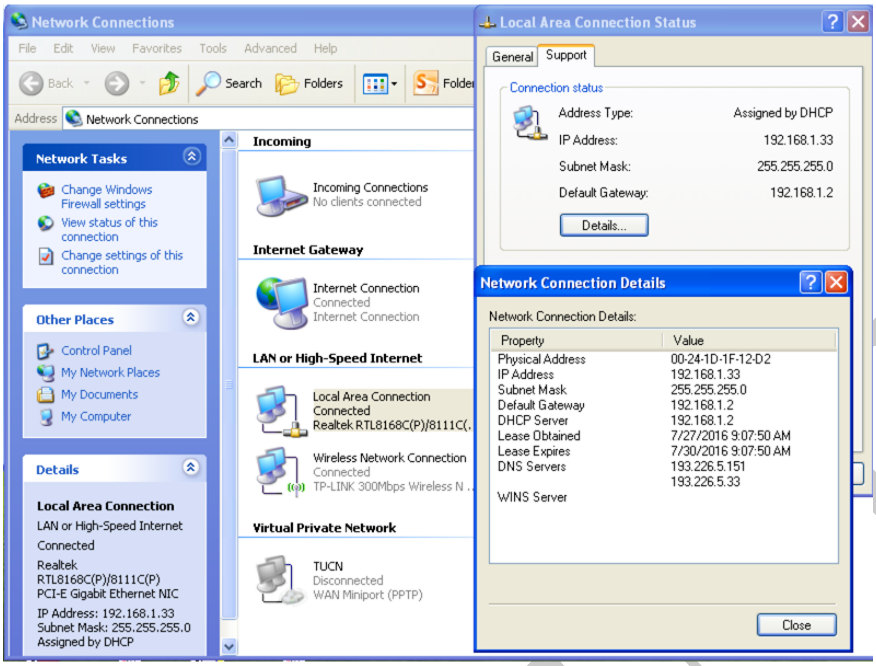

Figure 8.25: Determining the IP LAN address of the ADSL modem when using the Windows XP operating system.

- in the window that appears, select Network and Sharing Center.
- in the window that appears, select Change Adapter Settings.
- in the window that appears all network interfaces of the computer are displayed; click on Local Area Connection.
- in the new window select Details and a window appears with the computer's MAC and IP addresses, Default Gateway address, DNS server/servers address(s) and other information. The IPv4 Default Gateway address is the LAN IP address of the modem.

Note: the Windows 7 operating system allows the use of IPv6 addresses, but most ADSL modems use IPv4 addresses.

Configuration of the ST516 modem

Enter the IP address of the modem, obtained following the steps described above, in a web browser (Internet Explorer, Chrome, Mozilla Firefox, etc.). The main configuration window, shown in Fig. 8.27 opens.

Note: there is the possibility to configure a relatively large number of parameters of the ADSL modems, but this application focuses only on the configuration of some parameters necessary for achieving connectivity. Configuration of other parameters are outside the scope of this book.

From the Broadband Connection menu select DSL Connection. In the window that appears (see Fig. 8.28) a series of ADSL link parameters, such as upstream and downstream flow rates, ADSL standard used, upstream and downstream transmission powers, cable attenuation, various events that appear on the line (errors, synchronization, loss of signal, etc.) are displayed.

The SpeedTouch window (see Fig. 8.29) displays the access protocol which can be used, i.e. PPPoA (Point to Point Protocol over ATM) or PPPoE (Point to Point Protocol over Ethernet), and from the Set Up menu can be set the type of access protocol as well as the VPI and VCI parameters of the ATM connection.

Note: the ST516 modem can only access the Internet by using a PPP access protocol, which requires a dedicated access server. The WAN IP addresses of the ADSL modem are provided through this protocol. Modem authentication is also performed by this protocol. For this reason with this modem we cannot access directly the Internet.

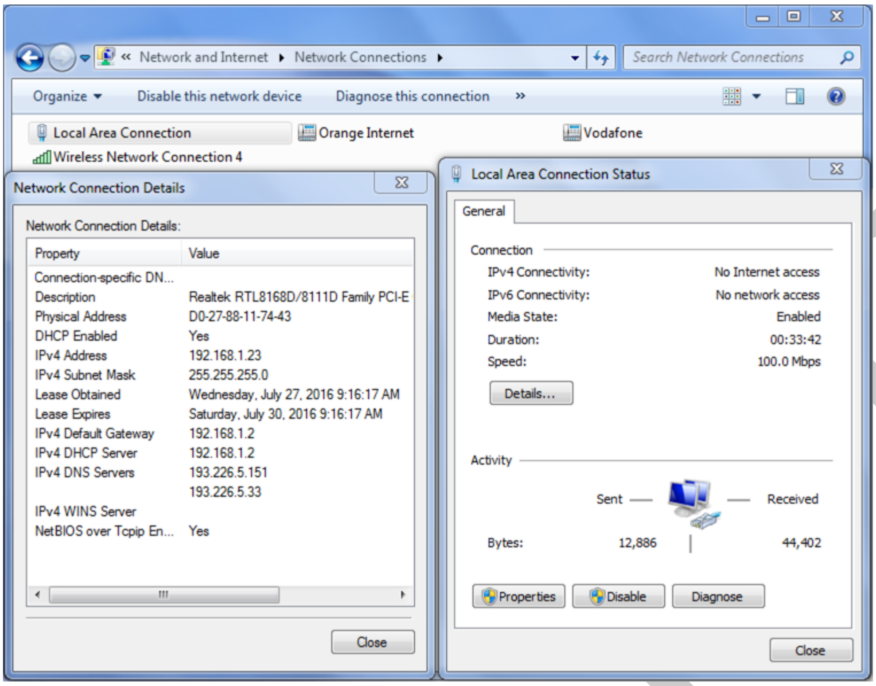

Figure 8.26: Determining the IP LAN address of the ADSL modem when using the Windows 7 operating system.

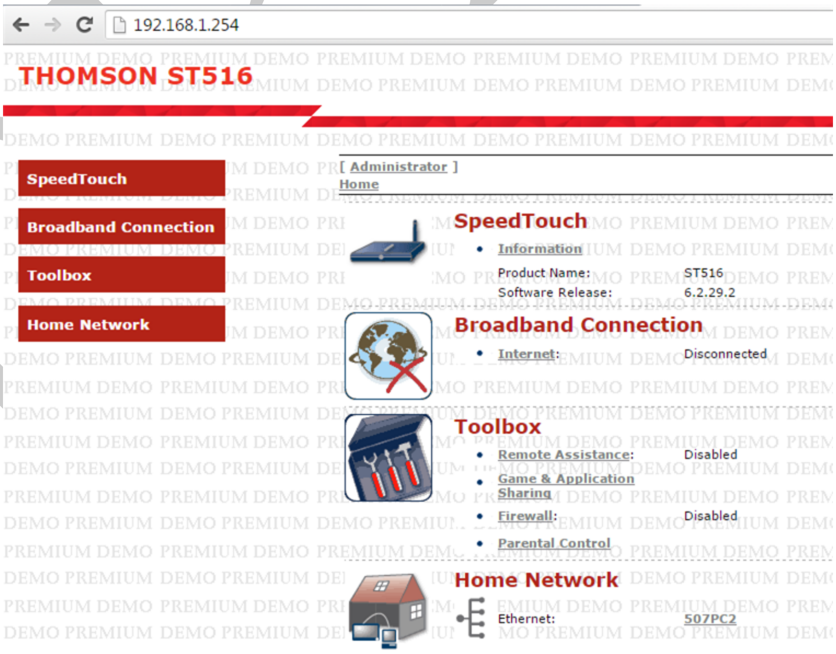

Figure 8.27: The main configuration window of the ADSL/ADSL2+ ST516 modem.

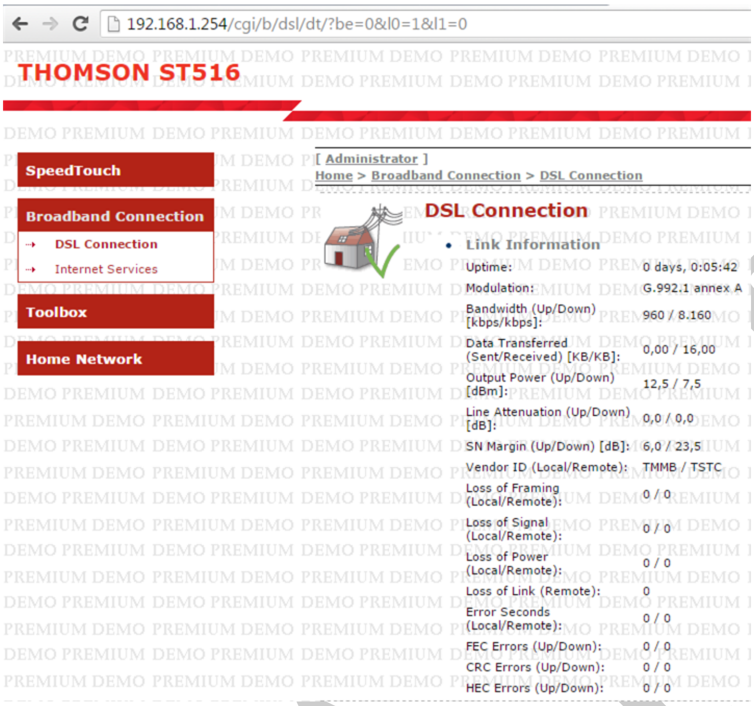

Figure 8.28: The monitoring window of the DSL connection parameters of the ADSL/ADSL2+ ST516 modem.

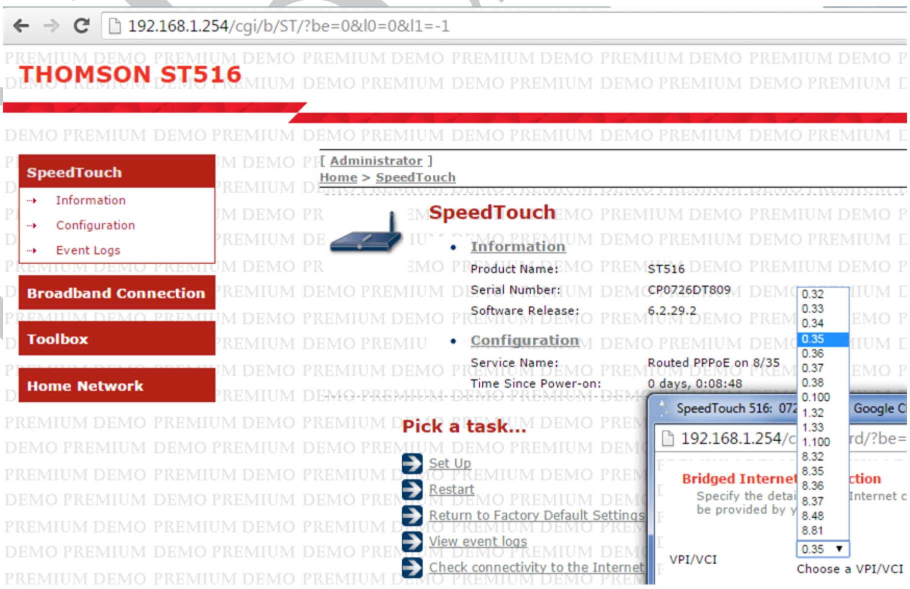

Figure 8.29: The configuration window of the access protocol and the ATM parameters for the ADSL/ADSL2+ ST516 modem.

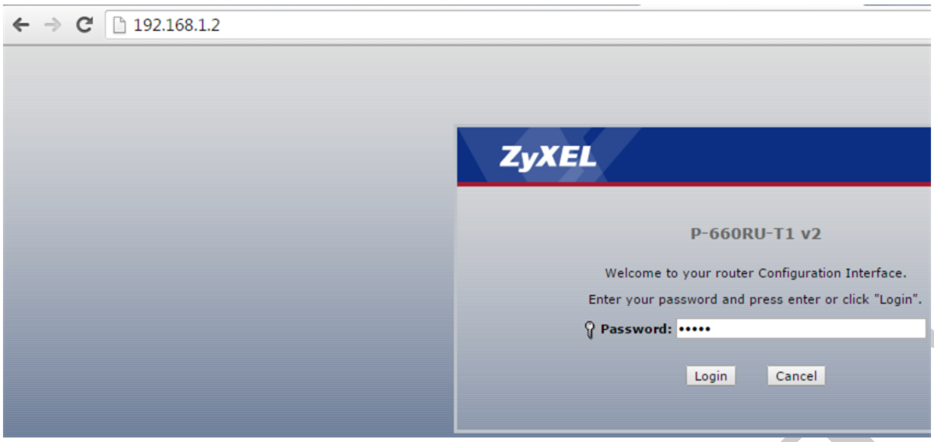

Figure 8.30: The authentication window of the ADSL/ADSL2+ P-660 modem.

Configuration of the P-660 modem

Enter the IP address of the modem, obtained by the steps mentioned above, in a web browser. The authentication window, shown in Fig. 8.30 is open. Enter the access password and the main configuration window shown in Fig. 8.31 is open.

The main configuration window also displays a set of information related to the equipment, the LAN and WAN IP addresses as well as the status (Up/Down) and the bit rates on the DSL and the LAN/Ethernet ports. If the password is not entered in the authentication window (or a wrong password is entered), the configuration window will be displayed, which will display all the mentioned status information, but it will not be possible to change any parameter.

The configuration of the IP LAN addresses of the modem is done in the LAN configuration window, shown in Fig. 8.32.

The P-660 modem allows the configuration of the modem's WAN IP addresses, so a dedicated server for these operations is no longer necessary. The configuration is performed in the WAN configuration window shown in Fig. 8.33. Also in this window can be configured the parameters of the ATM connection, as well as the working mode of the router in the modem, i.e. router or bridge. Regarding the configuration of the ATM parameters, it is possible to establish the encapsulation, and implicitly the access protocol used (PPPoA, PPPoE or ENET ENCAP), the multiplexing mode on the ATM channels (LLC - Logical Link Control or VC - Virtual Circuit multiplexing) of the data from various applications, respectively the VPI (Virtual Path ID) and VCI (Virtual Circuit ID) parameters. The use of the ENET ENCAPP access protocol allows direct access to the Internet without the need for a dedicated access server. The Gateway address represents the address of the next node in the WAN network where the IP packets are routed.

The P-660 modem allows the diagnosis of the physical ADSL connection (see Fig. 8.34), the ATM connection and the IP connection. There is the possibility of displaying physical link parameters, such as subcarrier loading, upstream and downstream transmission powers, signal to noise ratios, as well as some ATM connection parameters, such as number of transmitted/received ATM cells, number of lost ATM cells, etc. There is also the possibility to perform some loop tests by sending "pings" to various IP addresses.

Configuration of the HG665b modem

The ADSL2+/VDSL2 HG665b modem is a more complex modem that allows multiple functionalities and whose configuration is more complex. In the following will be presented only a few essential steps in the configuration of this equipment, without going into all details. As in previous cases, the modem IP LAN address is entered in a web browser and the authentication window is accessed (see Fig. 8.35), which asks for user and password. The modem cannot be accessed without authentication.

The Status menu allows the monitoring both of the LAN (modem - data terminal) and the WAN (modem

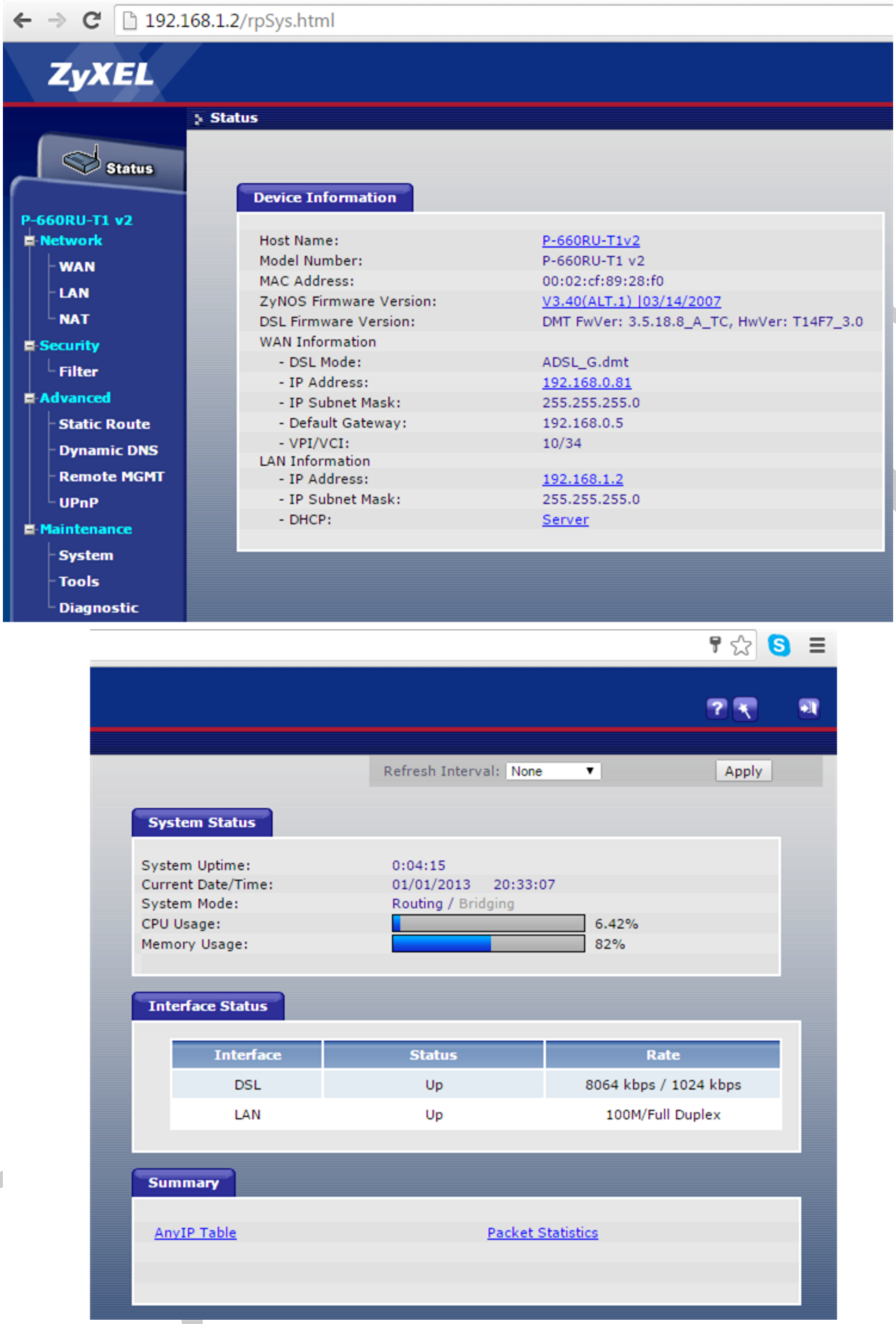

Figure 8.31: The main configuration window of the ADSL/ADSL2+ P-660 modem.

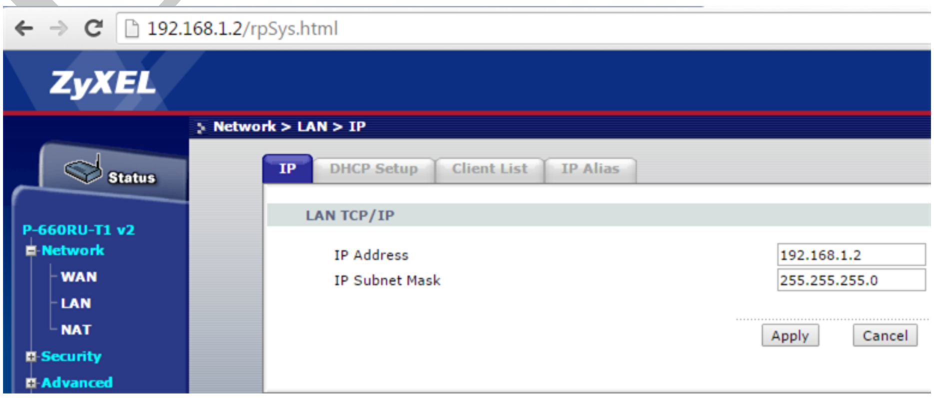

Figure 8.32: The LAN configuration window of the ADSL/ADSL2+ P-660 modem.

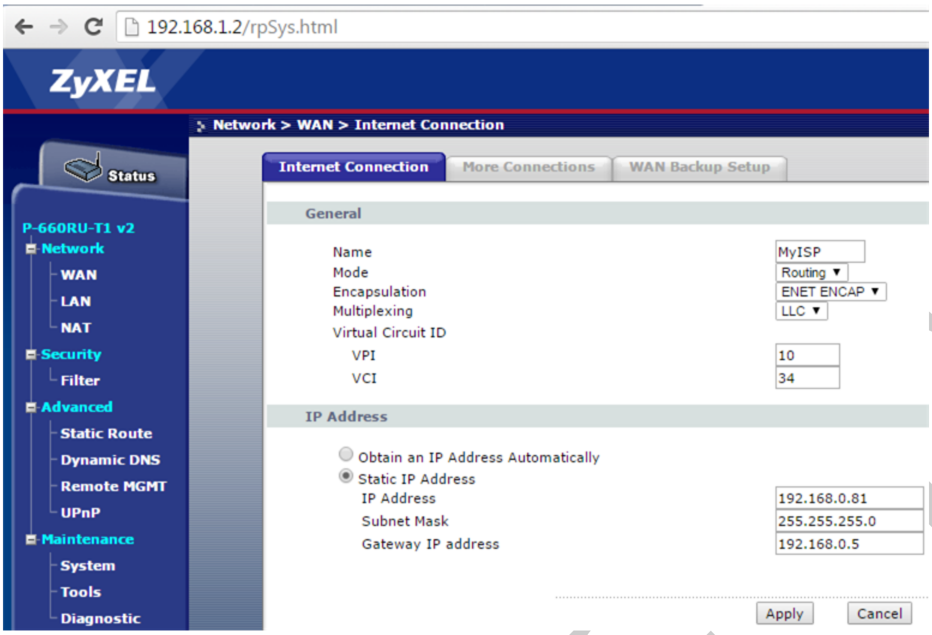

Figure 8.33: The WAN configuration window of the ADSL/ADSL2+ P-660 modem.

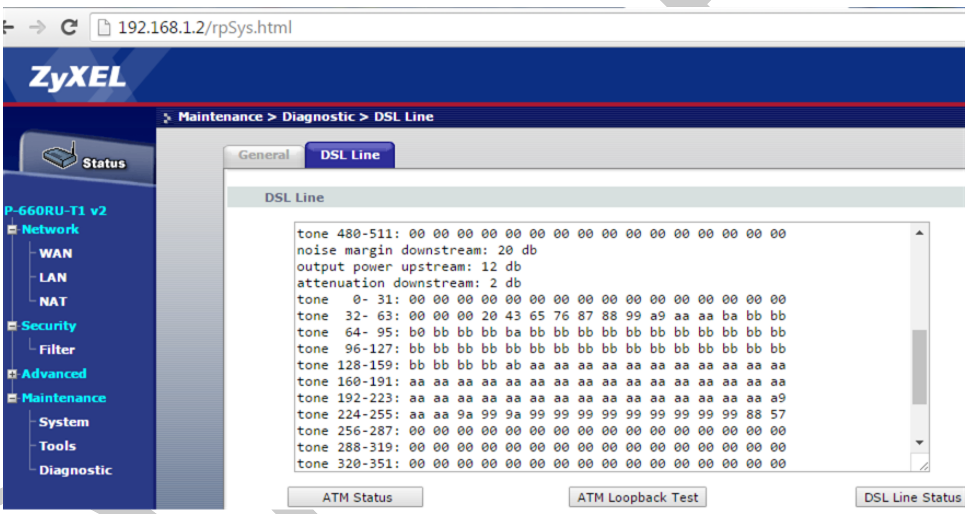

Figure 8.34: The diagnostic window of the DSL connection status of the ADSL/ADSL2+ P-660 modem.

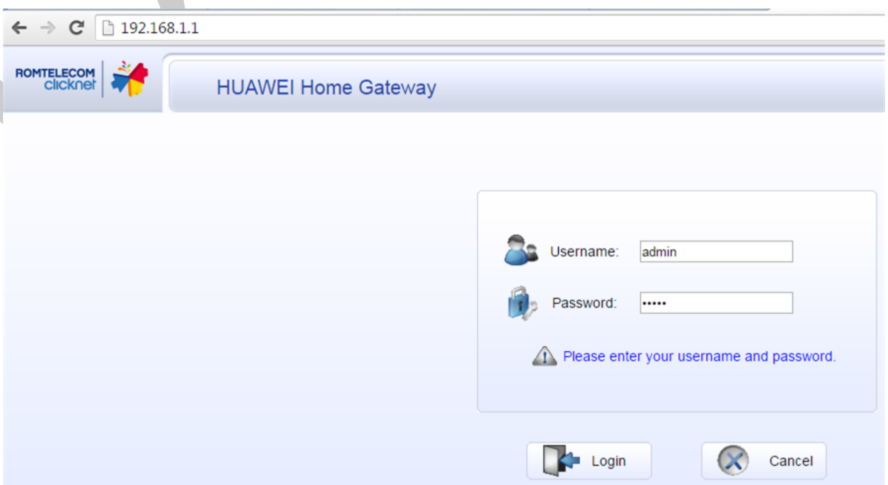

Figure 8.35: The authentication window of the ADSL2+/VDSL2 HG655b modem.

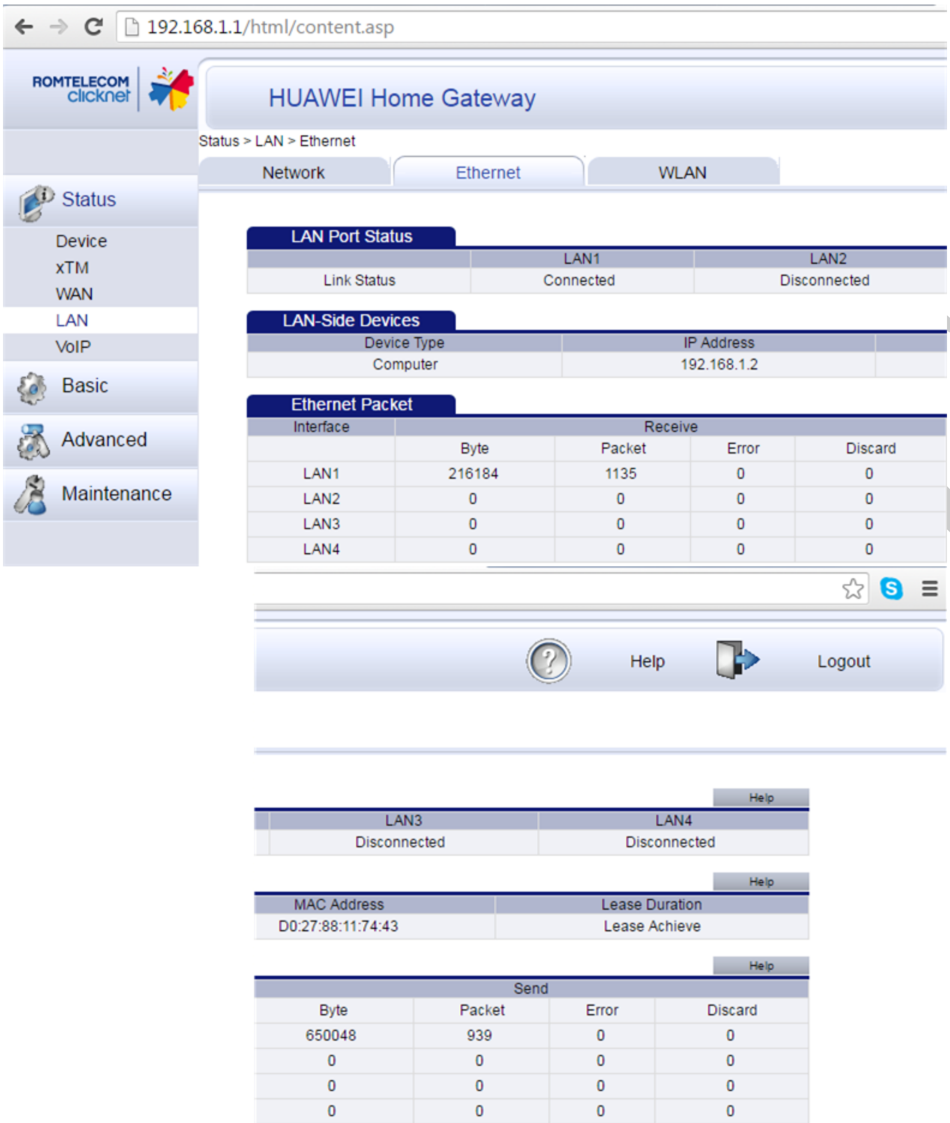

Figure 8.36: The Ethernet link status monitoring window of the ADSL2+/VDSL2 HG655b modem.

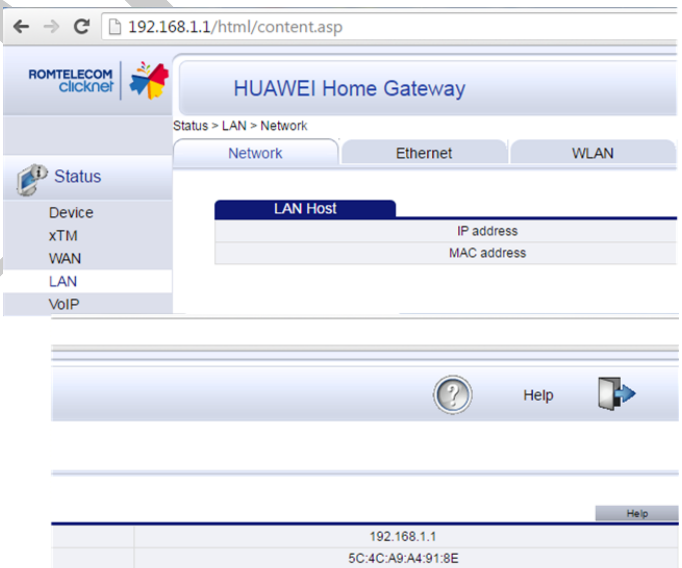

Figure 8.37: The window that displays the MAC and IP LAN addresses of the ADSL2+/VDSL2 HG655b modem.

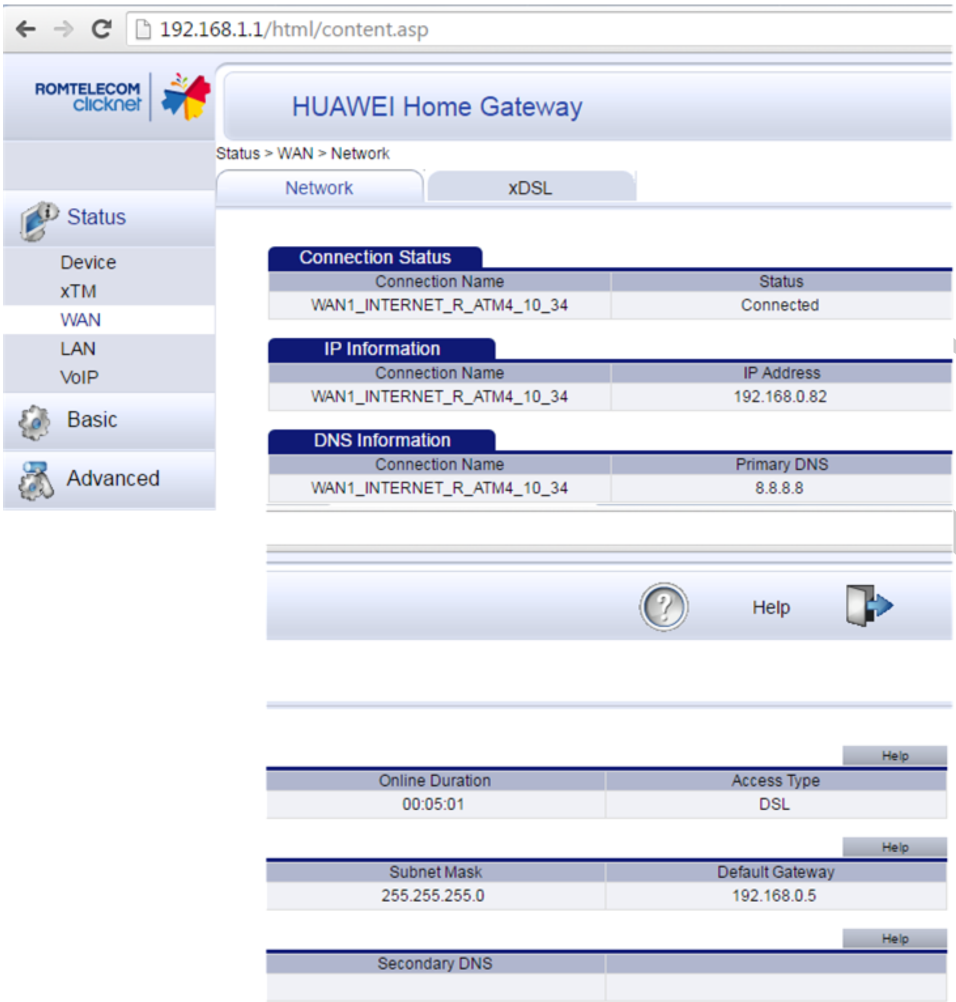

Figure 8.38: The window that displays the WAN IP addresses of the ADSL2+/VDSL2 HG655b modem.

- DSLAM) connections. In Fig. 8.36 is presented the window for monitoring the Ethernet connections between the ADSL modem and the data terminals (maximum 4 terminals) connected to the modem. Can be monitored the status of the Ethernet connections, the number of bytes/Ethernet packets transmitted/received as well as the packets with errors or the rejected packets. This window also displays the IP and MAC addresses of the data terminals connected to the modem. In Fig. 8.37 it is shown the window displaying the IP and MAC LAN addresses of the modem, and in Fig. 8.38 are displayed of the modem's WAN IP addresses (WAN IP address, DNS server IP address). Also are displayed the values of the VPI and VCI identifiers, the status of the WAN link (connected/disconnected) and the type of access technology (DSL in our case).

In Fig. 8.39 it is presented the window for monitoring the parameters of the physical DSL connection between the modem and DSLAM. The standard used, the upstream/downstream bit rates, the upstream/downstream transmission powers and the signal to noise ratios in the two transmission directions are displayed. Also are displayed the losses inserted by the cable in both directions as well as the quality parameters, such as the number of wrong Cyclic Redundancy Check (CRC) sequences and the number of error correction events performed by the FEC (Forward Error Correction) codes.

The Basic menu allows the basic configuration of the modem, i.e. the configuration of the ATM connection, the IP LAN and WAN addresses, as well as the configuration of the WiFi access point. In Fig. 8.40 is presented the configuration window of the ATM connection, which implies setting the VPI and VCI parameters, that characterize the ATM transmission, the Internet access technique (EoA - Ethernet over ATM, PPPoA -Point to Point over ATM, IpoA - IP over ATM), the encapsulation/multiplexing mode (LLC or VC) and the ATM service class. The DSL delay path (Path0 or Path1) can also be set. This setting is related to the error protection

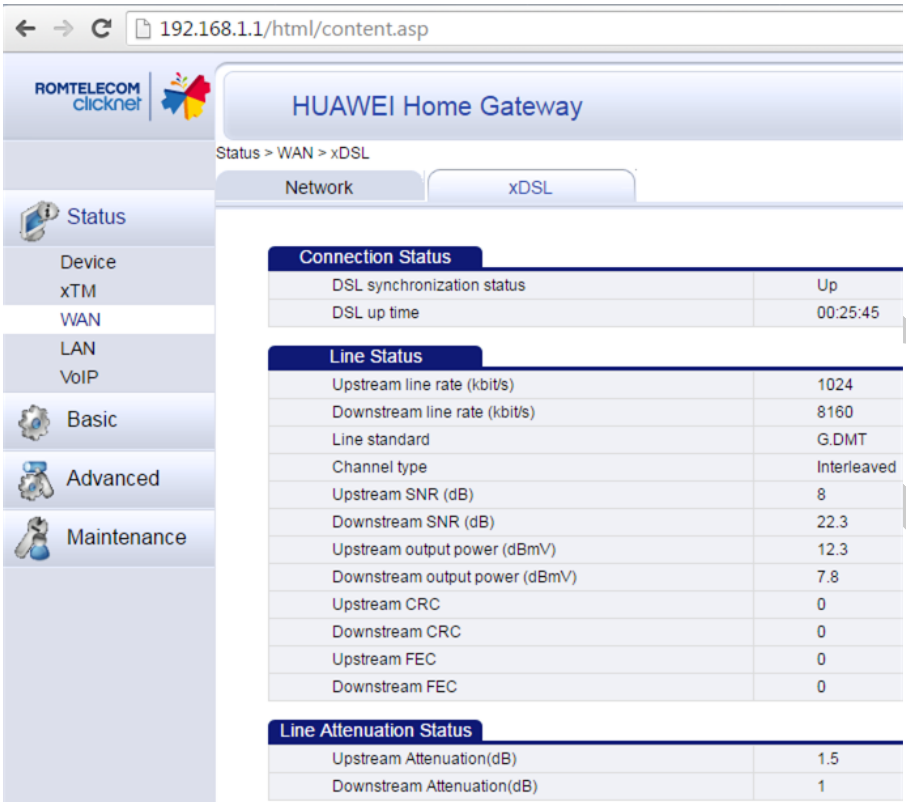

Figure 8.39: The window used for monitoring the DSL link parameters of the ADSL2+/VDSL2 modem HG655b.

mechanism, especially for burst errors. Using the EoA access mode allows direct access to the Internet without the need for a dedicated access server.

In Fig. 8.41 it is presented the configuration of the modem's LAN IP address as well as of the DHCP server that assigns IP addresses to the connected data terminals. It is possible to set the IP addresses of the DNS servers used by the terminals connected to the modem. These addresses can also be set in the WAN IP address configuration window.

In Fig. 8.42 it is shown the WAN IP address configuration window. The Default Gateway address represents the next node in the WAN network where the IP packets are routed. Finally in Fig. 8.43 it is shown the configuration window of the WLAN/WiFi access point integrated in the HG655b modem. Can be configured the WiFi standard (802.11b/g/n) used, the channel selection mode (automatic or manual), the transmission rate, the transmission power, the network identifier (SSID - Service Set Identifier) and the encryption protocol.

There is also the possibility to perform certain operations for diagnosing the ATM and the IP connections using the Maintenance menu.

8.4 ZyXEL AAM-1008 DSLAM access multiplexer

The AAM-1008 DSLAM equipment is an ADSL access multiplexer with 8 DSL ports and equipped with 100Mbps Fast Ethernet network interfaces [89]. The front panel of the equipment, shown in Fig. 8.44, contains a number of sockets and indicator LEDs. The module does not have its own power supply, being in fact a circuit board that is inserted into a rack that ensures the power supply and the cooling of the equipment. In Fig. 8.45 it is shown the AAM-1008 DSLAM module installed in the IES1000 rack [89] [90], also manufactured by the ZyXEL company.

The AAM-1008 module is equipped with 8 RJ11 sockets to which the subscriber loops are connected (they represent the DSL ports) and other 8 RJ11 sockets where the telephone lines coming from the telephone exchange are connected (they represent the CO ports). For each DSL port there is an integrated ADSL splitter

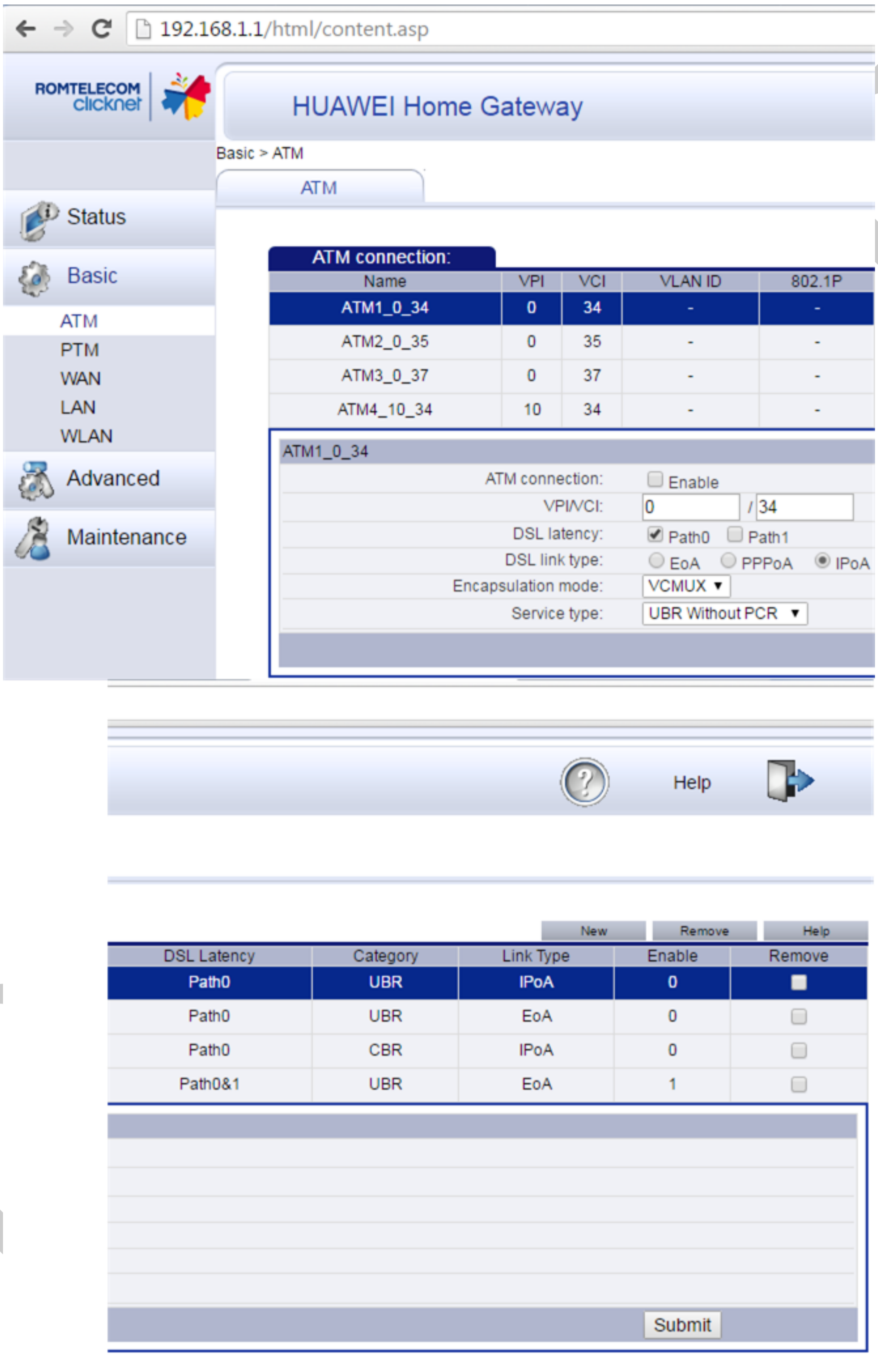

Figure 8.40: The configuration window of the ATM connection parameters of the ADSL2+/VDSL2 modem HG655b.

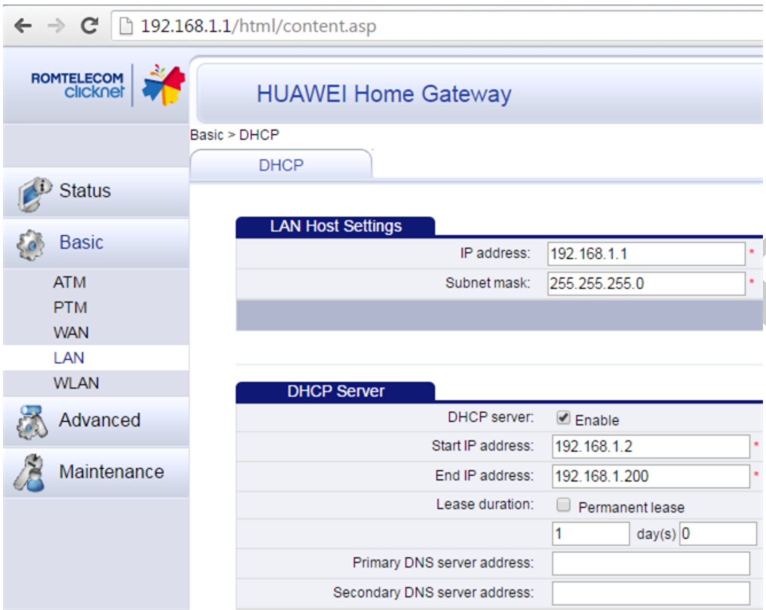

Figure 8.41: The configuration window of the LAN IP address of the ADSL2+/VDSL2 HG655b modem.

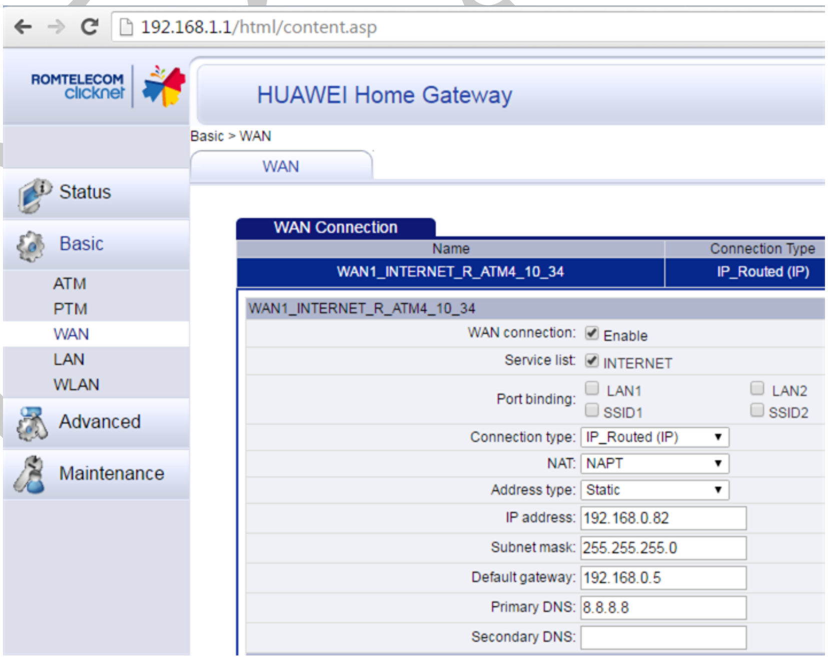

Figure 8.42: The configuration window of the WAN IP address of the ADSL2+/VDSL2 HG655b modem.

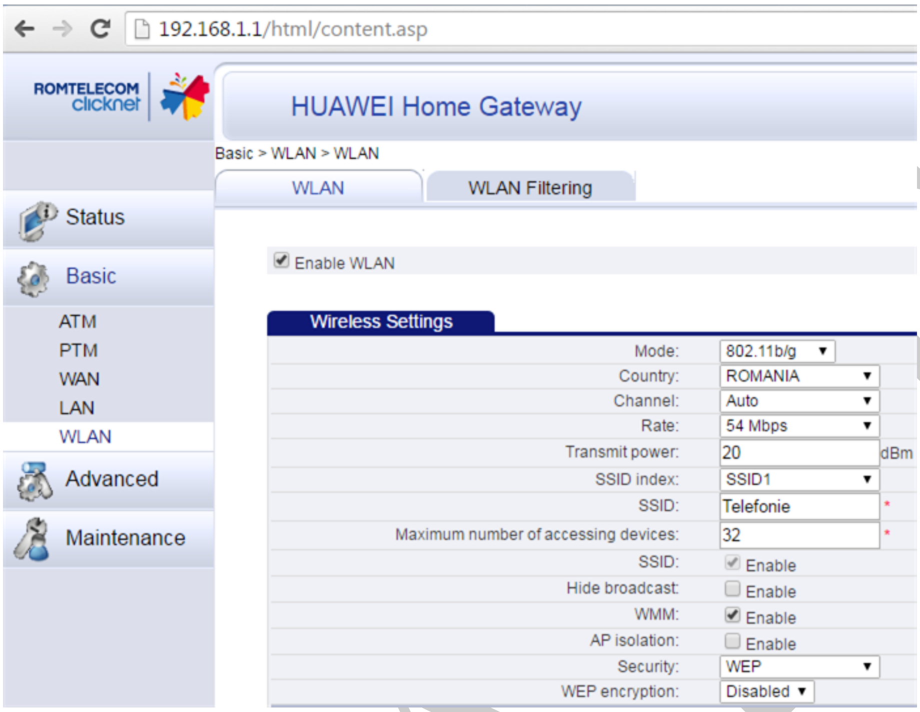

Figure 8.43: The WLAN configuration window of the ADSL2+/VDSL2 HG655b modem.

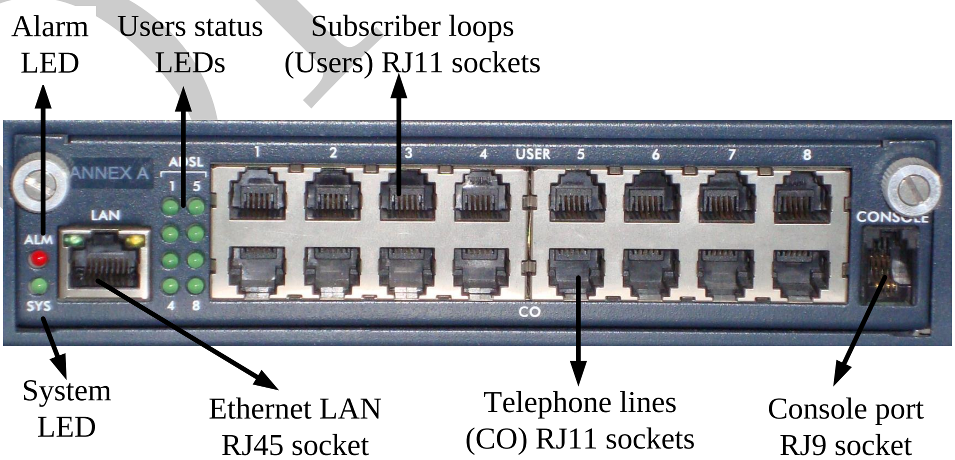

Figure 8.44: The front panel of the AAM-1008 ADSL access multiplexer

Figure 8.45: AAM-1008 DSLAM module installed in an IES-1000 rack.

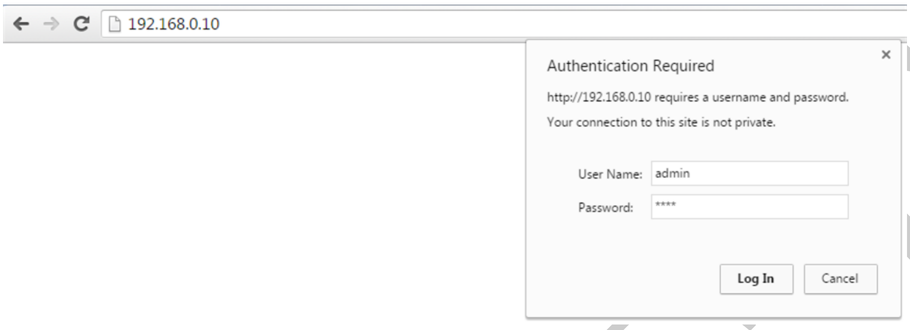

Figure 8.46: The authentication window of the AAM-1008 ADSL access multiplexer.

that separates the DSL port from the CO port. The module has an RJ45 Ethernet socket that ensures the connection of the DSLAM module to a WAN/transport network. An RJ9 socket provides connection to the DSLAM module of a configuration console. The console port basically is a UART serial port to which a computer can be connected, computer that can be used to configure the DSLAM module. Configuration is also possible remotely through the Ethernet connection, using a web application.

Each DSL port has an indicator LED that lights when the modem connected to that port is synchronized with the DSLAM. The module is also equipped with a system indicator LED (green), which blinks when the system is initialized and lights when the system is working properly, and with an alarm LED (red), which lights if the supply voltage changes or the overheating of the equipment is detected.

8.4.1 Configuration of the AAM-1008 ADSL access multiplexer

The configuration of the AAM-1008 access module can be done locally from a computer connected to the console port or remotely through the Ethernet connection, using the Telnet protocol or a web application that uses the HTML protocol. The latter will be presented below.

To access the DSLAM configuration application it is necessary to know the WAN IP address of this module. Enter this address in a web browser and access the authentication window (see Fig. 8.46), which requires a user name and a password. The main configuration window is shown in Fig. 8.47. The DSLAM is a complex equipment and therefore it exists the possibility to configure a large number of protocol parameters in the different layers of the OSI stack. In this chapter we will focus only on the configuration of some main parameters of the protocols that control the modem - DSLAM communication.

Note: the DSLAM module acts as a master entity and for this reason establishing the parameters of the protocols that control the modem - DSLAM communication is performed in the DSLAM.

A very important first step is to configure the DSL ports, that is, the transmission parameters at the physical layer. Clicking on the Port Setup menu opens a window (see Fig. 8.48) showing all the communication ports of the module, including the Ethernet port - port 0. For the Ethernet port the status (active/inactive), the working mode (with or without self-negotiation), the flow rate and the duplex working mode (half duplex or full duplex) are displayed.

For each DSL port the status (active/inactive), the port name, the name of the transmission profile associated with the port, the working mode, i.e. the ADSL standard configured on the port (Auto mode, means that the DSLAM automatically detects the ADSL standard with which it works the modem connected to that port),

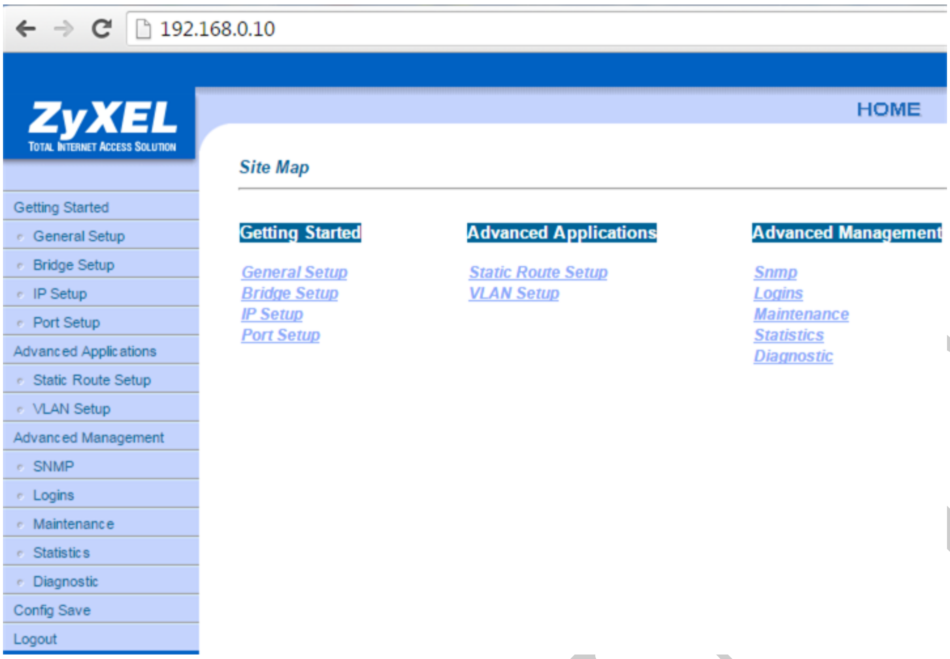

Figure 8.47: The main configuration window of the AAM-1008 ADSL access multiplexer.

the maximum upstream and downstream rates and the number of ATM channels instantiated on the physical ADSL connection are displayed.

Clicking on the port number opens another window that allows the configuration of that port. As shown in Fig. 8.49 can be set the port name, can be enabled/disabled this port, can be set the transmission profile, the working mode, the priority of the data packets on this port and the identifier of the virtual LAN (VLAN) network that the DSL port is part of (the last two settings are beyond the scope of this book).

The transmission profile allows the setting of the transmission parameters in the physical layer, the same settings, i.e. the same profile, being possible to be assigned to several ADSL ports. This is the reason for defining these profiles, i.e. avoiding the separate set up of the transmission parameters for each individual ADSL port. Clicking on Profile Setup in the window shown in Fig. 8.47 it is displayed another window (see Fig. 8.50) with all defined profiles. Clicking the "Ad" button opens a window (see Fig. 8.51) that allows the definition of a new transmission profile. With the Delete button can be deleted the selected profiles.

In Fig. 8.51 it can be observed that a transmission profile is characterized by the following parameters: name, ADSL path (fast/interleave), maximum and minimum bit rates in upstream and downstream, maximum, minimum and target noise margins for upstream and downstream. The noise margin is a parameter that indicates the quality of the physical connection and represents the value in dB with which the signal to noise ratio (SNR) can decrease so that the transmission performance (e.g. BER or BLER) does not fall outside some imposed limits. Usually the target noise margin is 6dB.

After setting the parameters of the physical layer the parameters of the protocols in the upper layers must be configured. The DSLAM IP WAN address and the ATM channel parameters must be set. Clicking on IP Setup from the main configuration window opens a window where can be set the WLAN IP addresses of the DSLAM - see the window in Fig. 8.52.

For each ADSL port, can be configured the ATM channels (one or more) instantiated over the physical connection. In the IP Setup - Edit Port Setup window (see Fig. 8.49) click on the Channel Setup and a window showing the instantaneous ATM channels appears - see Fig. 8.53. Also from this window new channels can be added, using the Add button, or delete the selected channels, using the Delete button. Each ATM channel is characterized by the VPI/VCI numbers, status (active/passive), the PVID virtual network identifier (Port VLAN ID) and an ATM channel specific profile, which characterizes the management of the transmission resources performed by the ATM protocols on this channel. Clicking on the VPI/VCI numbers is open a new

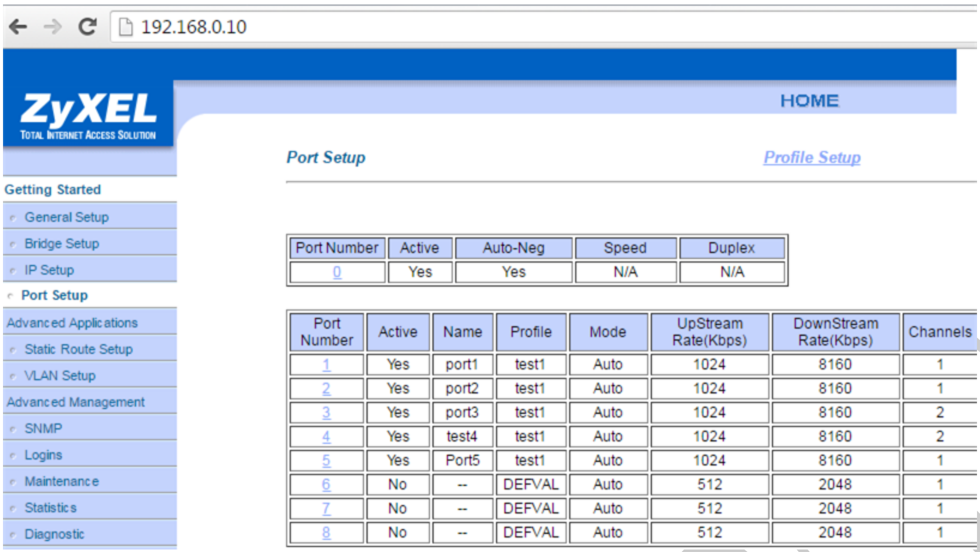

Figure 8.48: The window of the Port Setup menu of the AAM-1008 ADSL access multiplexer.

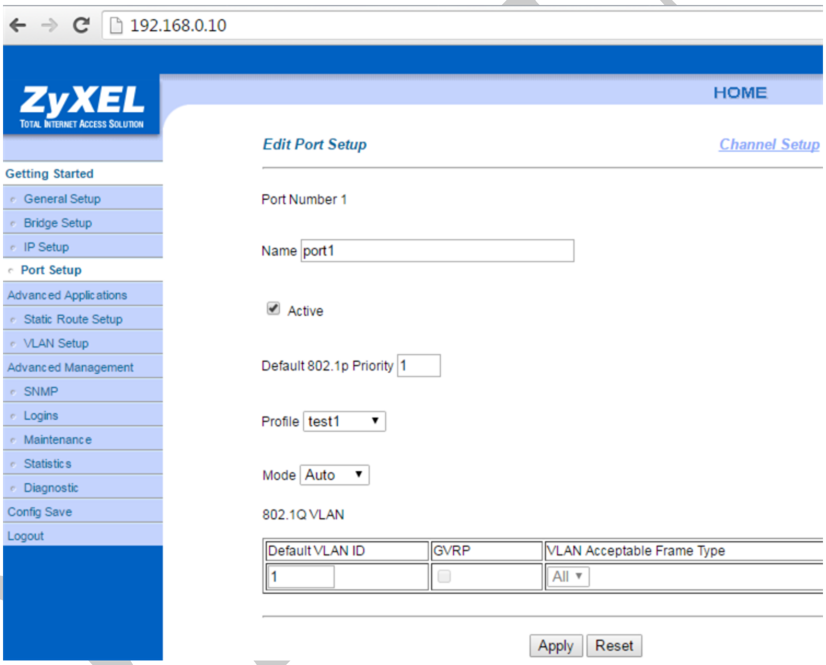

Figure 8.49: The Edit Port Setup menu window of the AAM-1008 ADSL access multiplexer.

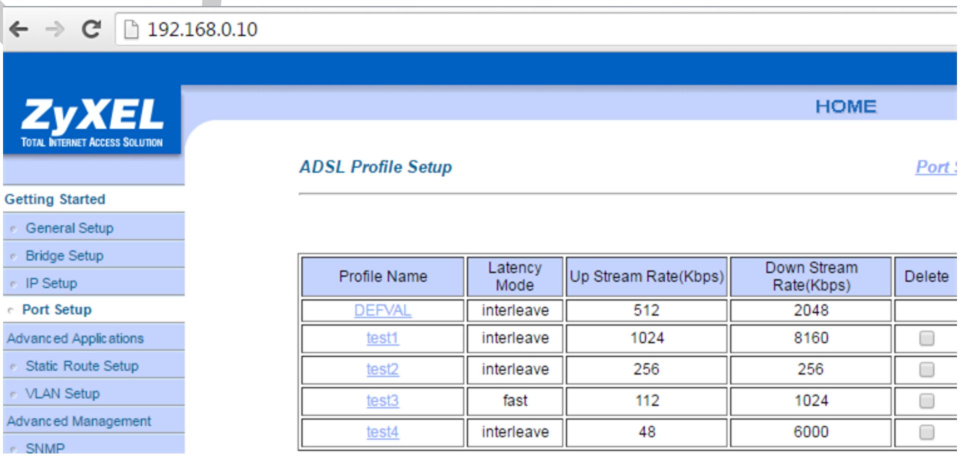

Figure 8.50: The ADSL Profile Setup menu window of the AAM-1008 ADSL access multiplexer.

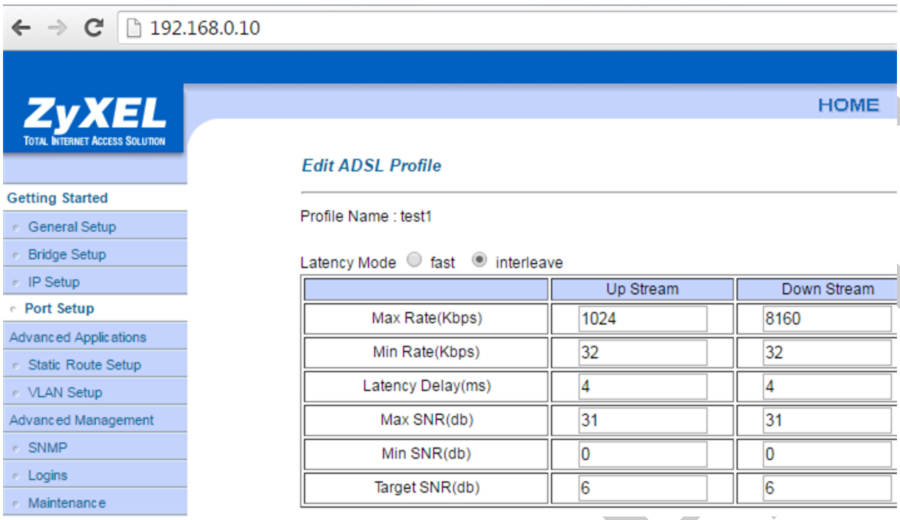

Figure 8.51: The Edit ADSL Profile menu window of the AAM-1008 ADSL access multiplexer.

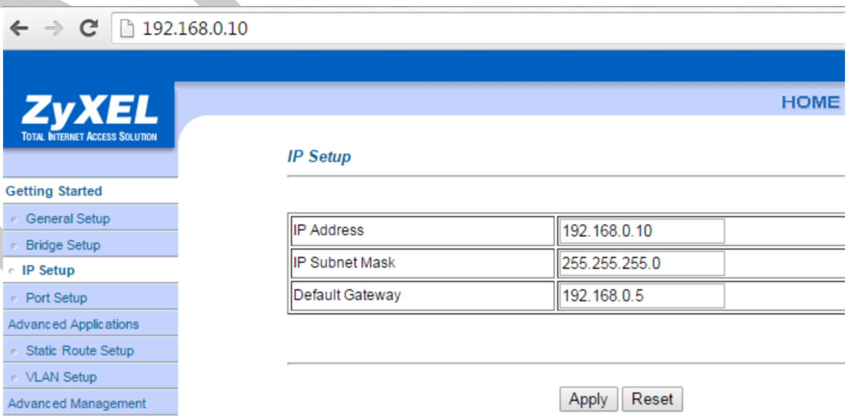

Figure 8.52: The IP Setup menu window of the AAM-1008 ADSL access multiplexer.

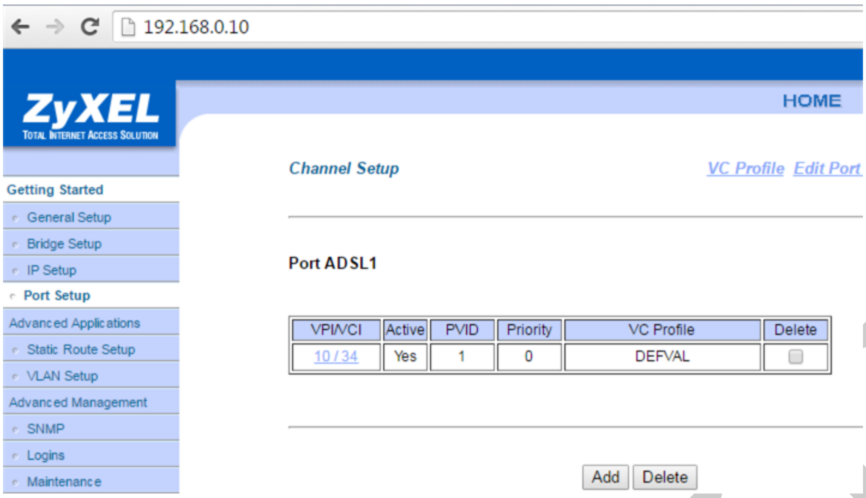

Figure 8.53: The window of the Channel Setup menu of the AAM-1008 ADSL access multiplexer.

window that allows the configuration of the aforementioned parameters of the ATM channel - see Fig. 8.54. A similar window opens for the Add operation, the Add Channel Setup window.

In the Port Setup - Channel Setup window (Fig. 8.53) click on the VC Profile and a window appears showing all the defined ATM channel profiles - see Fig. 8.55. With the Add button can be defined new channel profiles, and with the Delete button can be deleted the selected profiles. For example, clicking on the Add button the window shown in Fig. 8.56 opens, window which allows setting the ATM channel profile. The profile is characterized by a name, the type of encapsulation/multiplexing used (LLC or VC), the AAL protocol (ATM Adaptation Layer), the class of services for which the channel is defined and the parameters of that class of services - see the ATM Protocols section.

Note: the details related to the parameters of the ATM protocols are outside the scope of this book.

The AAM-1008 access multiplexer has integrated DSL line diagnostics functionalities, see the window shown in Fig. 8.57, which allows performing test loops at the physical, link or the ATM channel level (layer). There are also functionalities for monitoring the DSLAM hardware, DSL ports, ATM channels and VLAN virtual networks. At the port level, the number of Ethernet packets transmitted/received, the number of bytes transmitted/received, the number of errors and the transmission time are displayed - see Fig. 8.58. At the ATM channel level, the number of Ethernet packets transmitted/received, the number of ATM cells transmitted/received and the number of bytes transmitted/received are displayed.

8.5 Application

1. The ADSL modems described above are considered and for each modem the following operations are performed:

    (a) Power on the modem and connect to the computer on the Ethernet interface.

    (b) The IP address of the computer and the modem is determined by following the steps described in the previous sections.

    (c) The web configuration interface is accessed and the state parameters are identified.

    (d) Follow the modem configuration steps, i.e. configuration of IP LAN address, DHCP server, ATM channel parameters, access protocol, WAN IP address.

2. Connect the ADSL modems to the telephone sockets. In parallel with each modem connect an analog telephone using an ADSL splitter.

3. The ADSL access multiplexer (DSLAM) described above is considered and the following operations are performed:

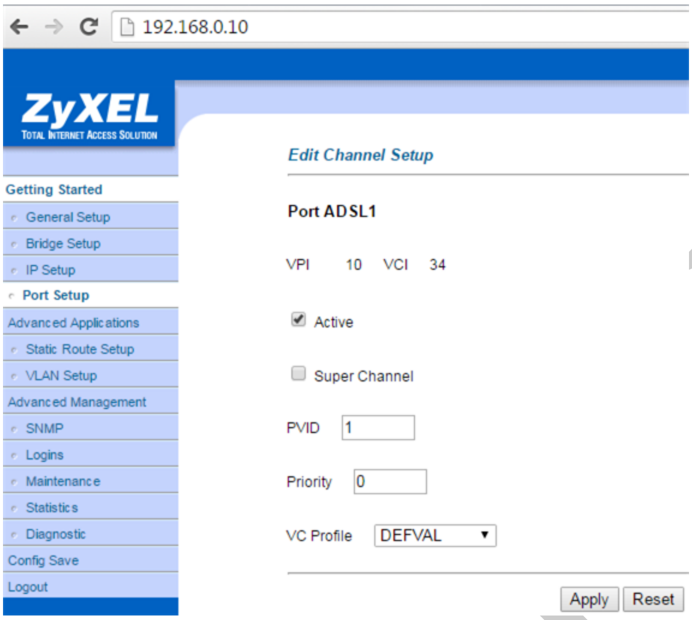

Figure 8.54: The window of the Edit Channel Setup menu of the AAM-1008 ADSL access multiplexer.

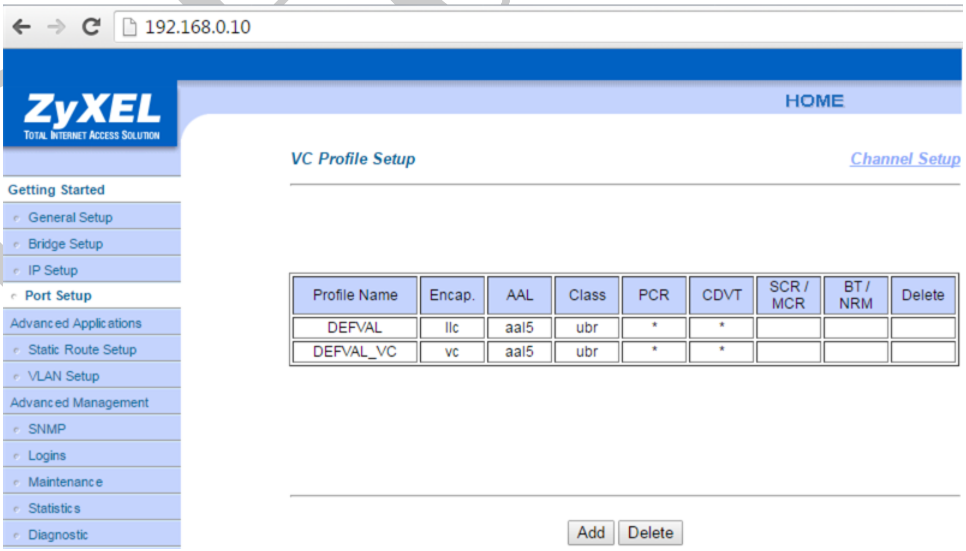

Figure 8.55: The window of the VC Profile Setup menu of the AAM-1008 ADSL access multiplexer.

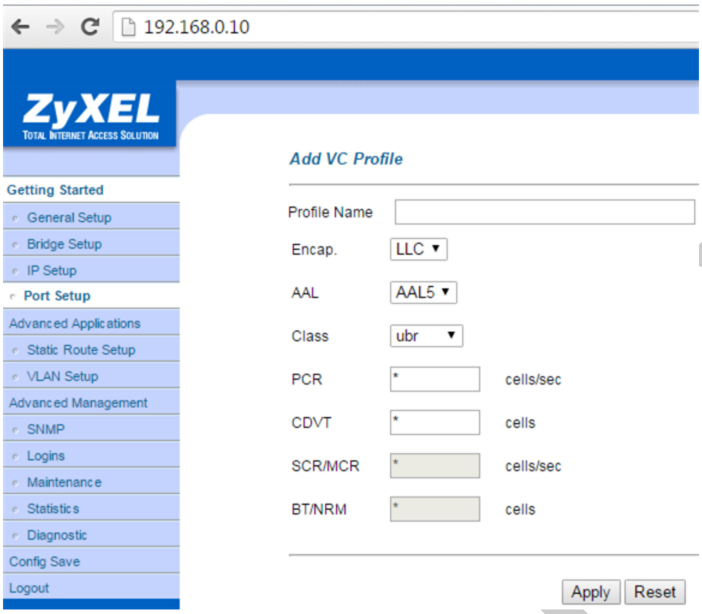

Figure 8.56: The window of the Add VC Profile menu of the AAM-1008 ADSL access multiplexer.

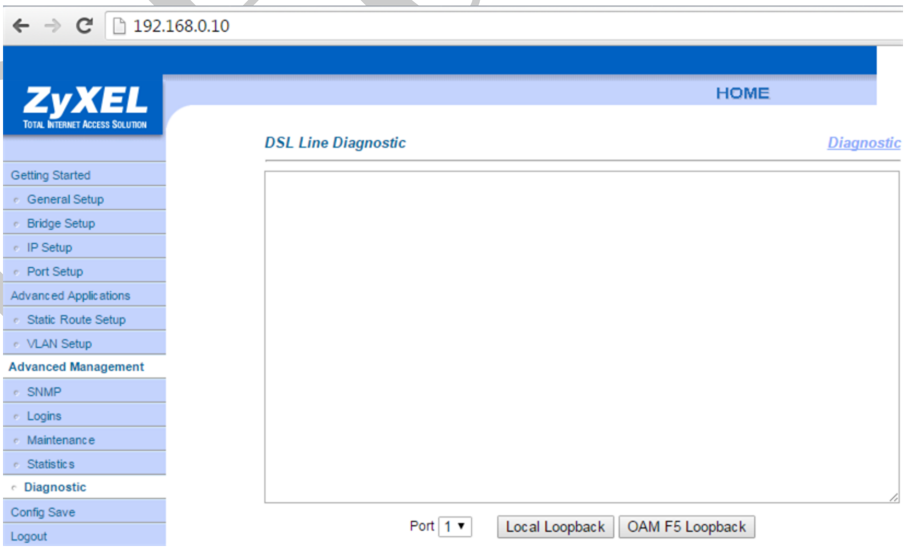

Figure 8.57: The diagnostic window of the DSL connection of the AAM-1008 ADSL access multiplexer.

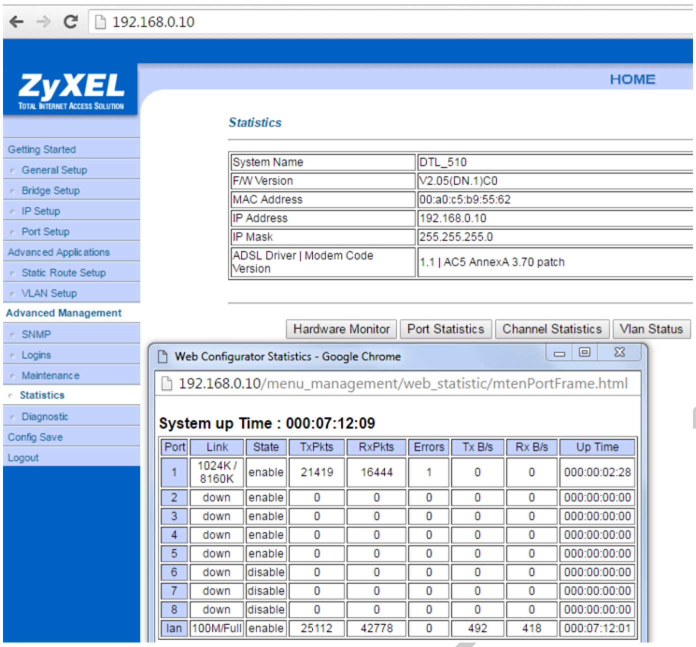

Figure 8.58: The DSL connection monitoring window of the AAM-1008 ADSL access multiplexer.

(a) Power on the DSLAM module and connect to a LAN using the Ethernet interface. The LAN network replaces in the present application the WAN transport network - see Fig. 8.44 - Fig. 8.45.

(b) The web interface for configuring the DSLAM module is accessed - see Fig. 8.46 - Fig. 8.47.

(c) The configuration of the ADSL ports (Fig. 8.48 - Fig. 8.49), of the transmission profiles (Fig. 8.50 - Fig. 8.51), of the WAN IP address (Fig. 8.52), of the ATM channels (Fig. 8.53 - Fig. 8.54) and of the profiles that characterize the ATM channels (Fig. 8.55 - Fig. 8.56) is performed/observed. Identify the main parameters that characterize the physical ADSL transmission and the transmission on the ATM channels (Fig. 8.57 Fig. 8.58).

4. The subscriber lines and ports of a PBX exchange are connected to the DSLAM. Subscriber loops are implemented using CAT5 cables. For each subscriber loop 300m of cable is used.

5. Check if the ADSL modems are synchronized with the DSLAM module. Checks if the analog telephone connections work in parallel with the DSL digital connections. Check if the telephone calls have effect on the ADSL digital connections.

6. Access the web interfaces for configuring the ADSL modems and follow the steps for configuring these equipment described in the previous sections.

7. Access on the web interfaces for configuring the ADSL modems the windows indicating the connection status and the connection diagnostic windows. Identify and analyze the displayed parameters that characterize the ADSL connection.

8. Checks which ADSL modems can access the Internet and which cannot. Identify the causes that allow or do not allow to access the Internet.

9. In the case of modems that can access the Internet, run applications for measuring the transfer rates (e.g. http://www.speedtest.net/) and measure the upstream and downstream bit rates. Compare the measured transfer rate values with those indicated in the windows displaying the link status. Explain the differences.

10. Access the ADSL connection diagnostic windows from the DSLAM web interface (Fig. 8.57 - Fig. 8.58) and run the diagnostic operations. Analyze the results.

11. For some ADSL ports, the parameters of the physical connection are changed, respectively 1-2 transmission profiles are defined (the steps shown in Fig. 8.46 - Fig. 8.51 are followed).
12. For the ADSL ports mentioned in the previous point, the ATM channels, respectively the profiles of the ATM channels are defined (follow the steps shown in Fig.8.53 - Fig. 8.56).
13. Follow the steps necessary to define the WAN IP address of the ADSL access multiplexer - see Fig. 8.52.
14. Change the length of the subscriber loop (reduce it to a few meters and then increase it to $600m$) and identify the effects of this operation on the parameters of ADSL transmission.
15. Connect an oscilloscope/spectral analyzer in parallel with one or more subscriber lines (one at a time). The ADSL + POTS composite signal is visualized over time and the spectral distribution of this signal is analyzed. The POTS, upstream ADSL, downstream ADSL, and the guard frequency bands are identified.
16. Using ADSL microfilters an additional analog telephone is connected in parallel with each ADSL modem. The effect of this additional terminal on the digital ADSL connection is checked, i.e. the effect of the telephone call when two phones are connected in parallel with the ADSL modem.
17. Replace the ADSL splitters with simple splitters and check the effect of simple connection in parallel with the ADSL modem of an analog telephone device - the parameters displayed in the ADSL modem diagnostics windows are observed. Telephone calls are made and the effect on the ADSL digital connection is checked. Explain the effects of the telephone connection on the ADSL digital connection if the ADSL splitter is not used.

8.6 Questions and exercises

1. What are the major advantages of DMT modulation?
2. The implementation of DMT modulation for ADSL transmissions uses 256 tones (from 0 to 255), and the symbol frequency per tone is 4.3125kHz. If the transmission directions are separated by frequency division duplexing, to the upstream channel the tones 7 - 31 are assigned, and to the downstream channel the tones 40 - 255 are assigned. Compute the frequency bands allocated to the upstream and downstream channels.
3. Explain why tones 1 - 6 and tones 31 - 40 are not used in ADSL transmissions with frequency separation of transmission directions.
4. If the transmission directions are separated by echo compensation, to the upstream channel the tones 7 - 31 are assigned, and to the downstream channel the tones 7 - 255 are assigned. Compute the frequency bands allocated to the upstream and downstream channels.
5. The implementation of the DMT modulation for ADSL transmissions uses 256 tones (from 0 to 255), the separation between two tones being 4.3125kHz. If to the upstream channel are assigned tones 7 to 31, and to the downstream channel are assigned tones 40 to 255, what are the minimum and maximum bit rates in the upstream and downstream? The modulation constellations can be selected from 4PSK to 32768QAM.
6. The implementation of DMT modulation for ADSL transmissions uses 256 tones (from 0 to 255), the separation between two tones being 4.3125kHz. If to the upstream channel are assigned tones 7 to 31, and to the downstream channel are assigned tones 7 to 255, what are the minimum and maximum bit rates in the upstream and downstream? The modulation constellations can be selected from 4PSK to 32768QAM.
7. What are the orders of magnitude of the upstream and downstream bit rates of ADSL/ADSL2+ transmissions?

8. What are the modulations that can be used in ADSL transmissions? What are the advantages/disadvantages of these modulations? Consider also the case of SDSL transmissions.
9. How can the quality of services be ensured in ADSL transmissions?
10. What are the distortions affecting the ADSL transmissions?
11. How is ADSL transmission adapted to the channel's characteristics?
12. What are the roles of ADSL splitter?
13. How can the ADSL modem be connected to the line? Analyze the possible solutions.
14. What are the similarities and differences between ADSL and SDSL transmissions?
15. How is the frequency band allocation and the duplexing performed in the case of ADSL transmissions?
16. What conditions must be imposed on the transmission line (twisted pairs) in order to allow ADSL transmission?
17. What are the techniques that can be used to reduce crosstalk in ADSL/ADSL2+ transmissions?
18. Is it possible to directly connect two ADSL modems? But what about two SDSL modems?
19. What are the roles of the DSLAM access multiplexer?

Contents

Bibliography

[1] Liviu Pană, *Parametrii de transmisie ai circuitelor telefonice mixte analog/digitale utilizate în transmisiuni de date în banda vocală*, CNSCC, București, 1995.

[2] Tomi Engdahl: Telephone line audio interface circuits,
`http://www.epanorama.net/circuits/teleinterface.html#general`

[3] Julian Macassey, *Understanding telephones*, Ham Radio Magazine, September 1985.
`http://massis.lcs.mit.edu/archives/technical/how.phones.work`

[4] Zsolt Alfred Polgar, *Telefonie digitală*, Editura Risoprint, Cluj Napoca, 2006.

[5] John C. Bellamy, *Digital telephony*, Third Edition, John Wiley & Sons Inc., New York, 2000.

[6] ' ': Schema electrică a telefonului EM-72.

[7] Kamilo Feher, *Comunicații digitale avansate*, Vol. 1, Editura Tehnică, București, 1993.

[8] Virgil Dobrotă, *Rețele digitale în telecomunicații*, Vol. 1, Ediția a III-a, Editura Mediamira, Cluj Napoca, 2002.

[9] ' ': Aparatul telefonic electromagnetic cu disc,
`http://www.scritub.com/tehnica-mecanica/APARATUL-TELEFONIC-ELECTROMAGN2152413123.php`

[10] ' ': Lucrare de specialitate aparate telefonice,
`http://www.scritub.com/tehnica-mecanica/LUCRARE-DE-SPECIALITATE-APARAT35316.php`

[11] ' ': Phone circuits,
`http://www.next.gr/telephone/Telephone-Circuits/`

[12] Craig Freudenrich: How Cordless Telephones Work,
`http://electronics.howstuffworks.com/cordless-telephone.htm`

[13] ETSI: DECT,
`http://www.etsi.org/technologies-clusters/technologies/dect`

[14] Ian Poole: DECT Specification/Standard,
`http://www.radio-electronics.com/info/wireless/dect/dect_specification_summary.php`

[15] ' ': User manual Euroset 5010,
`http://www.ascon.ro/UTILE%20site/download/siemens/telefoane%20analogice/SB_EUROSET_5010_E.pdf`

[16] ' ': User manual Euroset 5020,
`http://lastmanuals.com/manual/SIEMENS/EUROSET%205020`

[17] ' ': User manual Gigaset AP180,
http://gse.gigaset.com/fileadmin/legacy-assets/A31008-M2003-N131-1-7719_fr_FRK.pdf

[18] Liviu Pană, *Tehnologii de acces și sisteme de transmisiuni digitale pe linii bifilare din rețelele locale*, Vol. 1, INSCC, București, 1998.

[19] Vasile Bota, *Transmisiuni de date*, Ediția a II-a, Editura Risoprint, Cluj Napoca, 2003.

[20] Richard Lao, *The Twisted-Pair Telephone Transmission Line*, High Frequency Electronics, Nov. 2002.

[21] Adelaida Mateescu, *Semnale, circuite și sisteme*, Editura Didactică și Pedagogică, București, 1984.

[22] Gang Huang, and J. J. Werner, *Cable Characteristics*, Contribution to T1 Standards Project, T1E1.4/97-169, May 12-16, 1997 Clearwater.

[23] E. Karipidis, N. Sidiropoulos, A. Leshem, Li Youming, R. Tarafi, and M. Ouzzif, *Crosstalk Models for Short VDSL2 Lines from Measured 30MHz Data*, EURASIP Journal on Applied Signal Processing, Vol. 2006, Article ID 85859.

[24] Fanny Mlinarsky: Understanding FEXT and ELFEXT, White paper Agilent Technologies, 2000
http://www.octoscope.com/English/Collaterals/Whitepapers/octoScope_WP_UnderstandingELFEXT_200009.pdf

[25] SearchNetworking: Attenuation-to-crosstalk ratio (ACR) or headroom,
http://searchnetworking.techtarget.com/definition/attenuation-to-crosstalk-ratio

[26] Stefano Galli, and Kenneth J. Kerpez, *Methods of Summing Crosstalk From Mixed Sources-Part I: Theoretical Analysis*, IEEE Transactions on Communications, Vol. 50, No. 3, March 2002.

[27] Marina Dumitrescu, *Tehnologia rețelelor de comunicații*, Material de predare, 2009.
http://cndiptfsetic.tvet.ro/materiale/Materiale_de_predare/RC/06_Tehnologia%20retelelor%20de%20telecomunicatii.doc

[28] ' ': Rețele de comunicații prin cablu,
http://www.scritub.com/stiinta/informatica/REELE-DE-COMUNICAII-PRIN-CABLU241121191.php

[29] General Cable: Telecommunications Cable,
http://www.generalcable.com/na/us-can/products-solutions/communications/telecommunications-cable

[30] Belden: Data Cables,
https://www.yumpu.com/xx/document/view/40248722/belden-data-cables-kassex

[31] Nexans Cabling Solutions: Why Screening Screened Cabling vs Unscreened, White Paper, May 2013,
http://www.nexans.co.uk/eservice/UK-en_GB/fileLibrary/Download_540239149/UK/files/FTP%20vs%20UTP_V5_GBA_HFO%20DWI2%20NR3_PBE_V2_1.pdf

[32] Kristen Poulos: STP, UTP, FTP Cable & More: 7 Types & When to Use Them!,
http://www.belden.com/blog/datacenters/STP-UTP-FTP-Cable-More-7-Types-When-to-Use-Them.cfm

[33] Stephen H. Lampen, Martin J. Van Der Burgt, and Carl W. Dole *High-Definition Cabling and Return Loss*, Proceedings of 34th SMPTE Advanced Motion Imaging Conference, February 3-5 2000, San Francisco, CA.

[34] Belden: The Impact of Installation Stresses On Cable Performance, Technical Bulletin,
https://www.belden.com/docs/upload/TB66.pdf

[35] Firewall.cx: Unshielded Twisted Pair (UTP) - CAT1 to CAT5, 5E, CAT6 & CAT7,
http://www.firewall.cx/networking-topics/cabling-utp-fibre/
112-network-cabling-utp.html?

[36] Robin Suttel, *Liability Issues Loom Over Abandoned Cable*, Buildings, July 2006,
https://www.chemours.com/Cabling_Solutions/en_US/assets/
downloads/buildings_article.pdf

[37] David B. Kiddoo, *Cable Component Material Innovations for Stringent Fire Safety and Environmental Compliance Requirements*, International Wire and Cable Symposium, November 2007.

[38] Daniel Kennefick, *Limited Combustible: The New Standard for Fire Safety*, White Paper Nexans,
https://www.anixter.com/content/dam/Suppliers/Berk-tek/
Tek%20Briefs/LC_WhitePaper3_05.pdf

[39] AllPinouts: Archive of cable and connectors pin-outs,
https://allpinouts.org/

[40] LANshack: How to Wire a Phone Jack,
http://www.lanshack.com/wire_phone_jack.aspx

[41] CableSupply: Telephone Connection and RJ11 Wiring,
http://cablesupply.com/content/
47-telephone-connections-and-rj11

[42] LANshack: Category 5/5E & Cat 6 Cabling Tutorial and FAQ's,
http://www.lanshack.com/cat5e-tutorial.aspx

[43] Dinkle: Dinkle products,
http://www.dinkle.com/

[44] John Highhouse, *A Guide for Telecommunications Cable Splicing*, Delmar Publishers, 1997.
https://books.google.ro/books?id=dqoDQmrU2_gC
&printsec=frontcover#v=onepage&q&f=false

[45] Canford: Krone wiring modules,
http://www.canford.co.uk/KRONE-WIRING-MODULES

[46] Littelfuse: Transient Suppression Devices and Principles, Application note, 1998.
http://www.littelfuse.com/data/en/application_notes/an9768.pdf

[47] Hager: Protecție la supratensiune,
http://www.hager.ro/files/download/0/866_1/0/
Descarcatoare_supratensiune.pdf

[48] ITU-T: Recommendation O.41 - Psophometer for use on telephone-type circuits,
https://www.itu.int/rec/T-REC-O.41/en

[49] Adelaida Mateescu, și Neculai Dumitru, *Semnale și circuite de telecomunicații*, Editura Didactică și Pedagogică, București, 1979.

[50] ITU-T: Recommendation G.992.1 - Asymmetric digital subscriber line (ADSL) transceivers, https://www.itu.int/rec/T-REC-G.992.1/en

[51] ITU-T: Recommendation G.711 - Pulse code modulation (PCM) of voice frequencies, https://www.itu.int/rec/T-REC-G.711

[52] Mihai Radu, Sorin Stoica, Viorica Mărîi, și Adrian Barcaru, *Telefonie numerică*, Editura Militară, București, 1988.

[53] Phil Schniter: EE-597 Notes - Quantization, Course Notes June 2004, http://www2.ece.ohio-state.edu/s̃chniter/ee597/handouts/quantization.pdf

[54] Texas Instruments: A-Law and mu-Law Companding Implementations Using the TMS320C54x, Application Note 1997, http://www.ti.com/lit/an/spra163a/spra163a.pdf

[55] Das A., *Digital Communication, Signals and Communication Technology*, Springer-Verlag, Berlin Heidelberg, 2010. http://www.ti.com/lit/an/spra163a/spra163a.pdf

[56] Hyun-Jik Kim, *A new adaptive delta modulation system*, Iowa State University Digital Repository, 1975, http://lib.dr.iastate.edu/cgi/viewcontent.cgi?article=6485&context=rtd

[57] Chris Woodford: Fax machines, http://www.explainthatstuff.com/faxmachines.html

[58] Dialogic: Dialogic NaturalAccess NaturalFax API Developer's Manual, https://www.dialogic.com/webhelp/NaturalAccess/Release9.0/NaturalFax_API_Dev_Manual/default.htm#group_3_protocol.htm

[59] Gao Research: Group 3 Fax Modem Software and Protocols, http://www.gaoresearch.com/products/faxsoftware/other/group3fax.php

[60] ITU-T: ITU-T Recommendations V series: Data communications over the telephone network, http://www.itu.int/itu-t/recommendations/index.aspx?ser=V

[61] Cisco: Internetworking Technology Handbook, http://docwiki.cisco.com/wiki/Internetworking_Technology_Handbook#Dial-up_Technology

[62] TechnologyUK: The RS-232 Interface, http://www.technologyuk.net/telecommunications/communication_technologies/rs-232_interface.shtml

[63] Sparkfun: Serial Communication, https://learn.sparkfun.com/tutorials/serial-communication/ rules-of-serial

[64] Windmill Software Ltd: When to use RS232 Handshaking or Flow Control?, http://www.windmill.co.uk/handshaking.html

[65] Weschler Instruments: Tricolor Serial Communication ASCII Protocol, August 1997, http://www.weschler.com/_upload/sitepdfs/techref/manuals/BGTC_serial_manual02.pdf

[66] Chuck Forsberg: Xmodem/Ymodem Protocol Reference,
`http://cristal.inria.fr/ doligez/zmodem/ymodem.txt`

[67] Frank da Cruz, *Kermit, A File Transfer Protocol*, Digital Press, Bedford MA, 1987
`http://www.kermitproject.org/kproto.pdf`

[68] Chuck Forsberg: the ZMODEM Inter Application File Transfer Protocol,
`http://cristal.inria.fr/ doligez/zmodem/zmodem.txt`

[69] ' ': Serial Programming/Modems and AT Commands,
`https://en.wikibooks.org/wiki/Serial_Programming/`
`Modems_and_AT_Commands`

[70] ' ': Basic Hayes Modem AT string,
`http://www.computerhope.com/atcom.htm`

[71] Syeda Anila Nusrat: Introduction to AT commands and its uses,
`http://www.codeproject.com/Articles/85636/`
`Introduction-to-AT-commands-and-its-uses`

[72] Ubinetics: GSM AT Command Set, Application Note 010,
`http://www.zeeman.de/wp-content/uploads/2007/09/`
`ubinetics-at-command-set.pdf`

[73] ManualGuru: Samsung SF-360 Series User's Manual,
`http://www.manualguru.com/samsung/sf-360-series/users-manual`

[74] Thought Communications: FaxTalk Communicator SE for Windows, User's Guide, Version 4.7, August 2007,
`http://www.vshopu.com/item/2111-0901/cd-rom/Conexant_D400/FAXTALK/`
`FaxTalk%20Communicator%20SE%204.7%20User%20Guide.pdf`

[75] : Modem AT Command Set,
`http://michaelgellis.tripod.com/modem.html`

[76] WhatIs: Fast Guide to DSL (Digital Subscriber Line),
`http://whatis.techtarget.com/reference/`
`Fast-Guide-to-DSL-Digital-Subscriber-Line`

[77] EXFO: Local-Loop and DSL Reference Guide,
`http://www.equicom.hu/wp-content/uploads/`
`EXFO_Reference-Guide-Local-Loop-DSL-v1_en.pdf`

[78] ITU-T: Recommendation G.992.5 - Asymmetric digital subscriber line 2 transceivers (ADSL2) - Extended bandwidth ADSL2 (ADSL2plus),
`http://www.itu.int/ITU-T/recommendations/rec.aspx?rec=G.992.5`

[79] Broadband Forum: DSL Technology Evolution,
`https://www.broadband-forum.org/downloads/About_DSL.pdf`

[80] Aware: ADSL2 and ADSL2+. The New ADSL Standards, Whitepaper 2003,
`http://nikportal.cickany.hu/view/BMF/2008-2009-1/`
`Halozatok/PDF/adsl2.pdf`

[81] DSL Forum: ADSL2 and ADSL2+ - The new ADSL standards, Whitepaper 2003,
`https://www.broadband-forum.org/marketing/download/`
`mktgdocs/ADSL2_wp.pdf`

[82] Dirk Van Aken, and Sascha Peckelbeen: Encapsulation Overhead(s) in ADSL Access Networks, Whitepaper 2003,
http://wand.net.nz/∼sim513/2006/readings/adsl-2.pdf

[83] Vasile Dadarlat: ATM Asynchronous Transfer Mode, notițe de curs UTCN,
http://ftp.utcluj.ro/pub/users/dadarlat/master-anVII/curs3-retATM.pdf

[84] ' ': Layers of the ATM model,
http://meseec.ce.rit.edu/eecc694-spring2000/694-4-25-2000.pdf

[85] H3C: PPPoE Technology White Paper,
http://www.h3c.com.hk/products___technology/technology/wan/technology_white_paper/200911/654415_57_0.htm

[86] ' ': SpeedTouch 516 User manual,
https://help.teksavvy.com/hc/en-us/articles/200049899-SpeedTouch-516-User-Manual

[87] ' ': Quick Start Guide P-660RU-T v2,
ftp://ftp.zyxel.com/P-660RU-T1_v2/quick_start_guide/P-660RU-T1%20v2_1.pdf

[88] Telekom: Manual de utilizare Router HG665b,
https://www.telekom.ro/images/docs/Legal_docs/Arhiva_Romtelecom/Manual_de_utilizare_router_HG655b.pdf

[89] ZyXEL: IES-1000 Integrated Ethernet Switch User's Guide, version 2.05, Nov. 2005,
https://www.zyxel.fr/support/download/57804_1

[90] ' ': IES-1000 Multi-Service IP DSLAM,
http://www.ral.ro/template/pdfprod/produs_966.pdf