The phone device – basic schematic

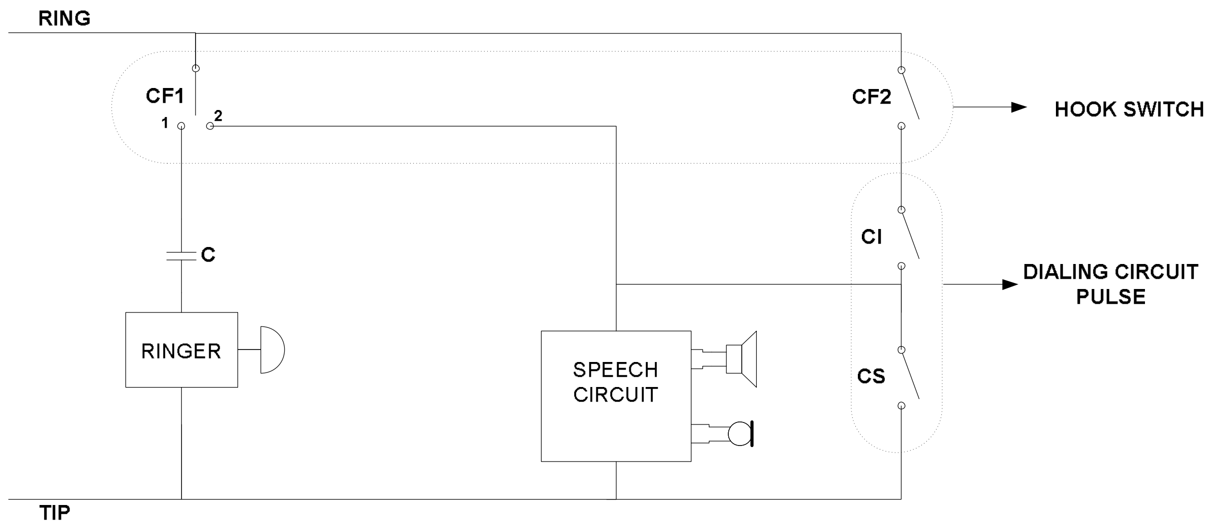


Figure 1 - Basic schematic of the telephone device

Active state (OFF - HOOK) : CF1 – 2, CF2 – ON
Inactive state (ON - HOOK) : CF1 – 1, CF2 – OFF

CS – short-circuits the speech circuit during the dialing process to avoid distorting the dialing impulses due to the complex impedance of the speech circuit and to suppress the noise generated in the receiver by opening/closing of the DC loop.
CI – opens the DC loop to signalize to the local exchange the dialed number.
C – decouples the ringer from the line in DC.

The supply voltage could be generated locally (batteries, etc.) or remotely (from the exchange). The DC supply voltage $\mathbf{V_{sup} = -48\ V}$, and the loop current is $\mathbf{I_{loop}=20\text{-}50mA}$.

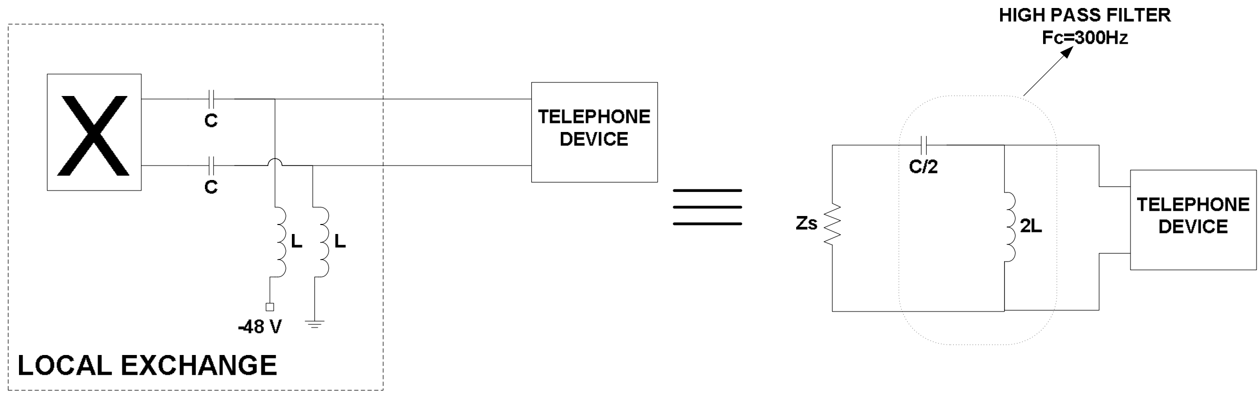


Figure 2 – The telephone device power supply bridge located in the local exchange

In Figure 2 can be noticed the capacitances, C=2µF, having the role to separate the circuits of the exchange from the DC supply voltage. The coils, L=0.5H, separates the power supply equipments from the (voice) signal transmitted on the subscriber loop. The telephone device can be considered as a signal source and the power supply bridge can be represented by a high pass filter with a cut-off frequency F_c = 300Hz. The cut-off frequency of this filter imposes the inferior limit of the telephone band (Figure 3).

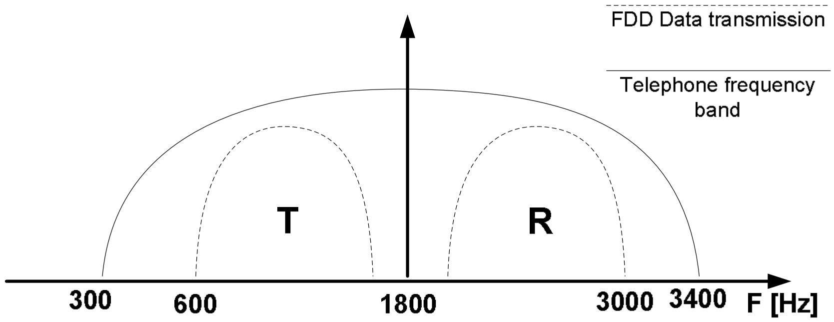


Figure 3 – The telephone band

The superior limit of the telephone band was selected according to the characteristics of the voice signal. (Even if the approximate frequency band of the human hearing is located between 20Hz and 16kHz, the largest sensitivity of the hearing is obtained in the 200Hz - 1kHz band). The value selected for the upper limit of the telephone band has imposed the sampling frequency employed by digital networks, $\mathbf{F_s = 8kHz}$.

The ringer

The electromagnetic ringer was used in the old phone devices; the characteristics of the ringing signal are strongly related to this ringer. New electronic phones have as ringer polyphonic tone/signal generators, but the parameters of the ringing signal were kept unchanged. The main parameters of the ringing signal are the following: $\mathbf{V_{ringer}=40\text{-}150V}$, $\mathbf{F_{ringer}=20\text{-}40Hz}$, **Timing:** $\mathbf{T_{ON}=2s, T_{OFF}=4s}$.

Disadvantage: due to the large voltage of the ringing signal electrocution danger exists and large impulse noise can be induced in the neighbor lines, affecting the data transmissions performed on that lines (xDSL transmissions).

Transmission of the called subscriber phone number to the local exchange

Transmission of the called subscriber number to the exchange can be realized in two ways: by using PULSE dialing or by using DTMF (Dual Tone Multi Frequency) dialing.

PULSE dialing

Each digit is encoded by a number of impulses (Figure 4). These impulses are generated by interrupting the subscriber loop employing the switch CI (Figure 1).

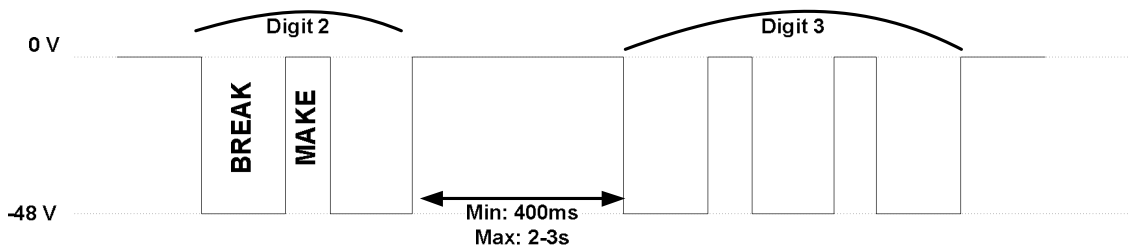


Figure 4 – Pulse dialing

When dialing a digit: CF1 – position 2, CF2 - CLOSED, CS - OPEN
When transmitting an impulse: BREAK: CI - OPEN, MAKE: CI - CLOSED
T_{BREAK} = 63ms, T_{MAKE} = 37ms, Nr. impulses/sec = 10

Disadvantages: **-** The large voltage hit could generate transient signals with amplitudes up to 300V;

- Can be induced impulse noise in the neighbor lines;
- There are affected the data transmissions (DSL – Digital Subscriber Line) due to crosstalk;
- Impulses can not be transmitted by the transport network; pulse dialing can be used only for subscriber loop signaling;
- Low dialing speed – large duration of the dialing process;

Advantages: it is a simple method which can be implemented in simple terminals – electromagnetic phones with dialing disk;

Dual Tone Multi Frequency (DTMF) dialing

The dialed number is transmitted to the exchange by using a combination of two frequencies ($Finf_j$,$Fsup_i$) i, j ∈ {1,2,3,4} according to table 1.

Fsup [Hz] / Finf [Hz]	1209	1339	1447	1633
697	1	2	3	A
770	4	5	6	B
852	7	8	9	C
941	*	0	#	D

Table 1 – DTMF frequencies and digit encoding

From Table 1 can be noticed that all the DTMF frequencies are located in the inferior part of the telephone band – this band coresponds to the transmission band of the dial-up modems employing FDD (Frequency Division Duplexing) and originating the transmission (this modem dials the number) – see Figure 3.
Table 2 gives the voltage levels of the DTMF signals

	Min.	Nom.	Max.
F_{sup}	160 mV	190 mV	200 mV
F_{inf}	130 mV	150 mV	160 mv

Table 2 – DTMF voltage levels

Problem related to DTMF dialing : the nonlinear distortions of the line. These distortions could create unwanted harmonics of the DTMF signals.

Solution: Avoid harmonic relations between the inferior and superior DTMF frequencies, meaning $Fsup_i \neq 2*Finf_j$, i,j = {1,2,3,4}.

Selection of different amplitudes (Table 2) for the inferior and superior frequencies (lower amplitudes for the inferior frequencies) represents another protection mechanism against the nonlinear distortions – lower amplitudes mean lower nonlinear distortions.

Problem related to DTMF dialing: simulation of the DTMF tones by the voice signal or telephone signaling.

Solution: Employment of a pair of frequencies for each digit decreases the simulation probability by the voice signal and signaling. By using groups of 3 frequencies for each digit the simulation probability could be decreased more but the complexity of the system increases. Another protection mechanism against simulation of the DTMF signals is represented by the minimum duration of the DTMF tones, $\mathbf{T_{DTMF\text{-}min}=100ms}$.

Advantages:
- Does not involve large voltage levels;
- DTMF signaling takes place in the telephone frequency band;
- Does not affect the data transmissions – DSL transmissions;
- Dialing speed is larger – duration of the dialing process is shorter;
- DTMF tones can be transmitted by the transport network;

Example

We consider a DTMF system having the following pair of frequencies assigned to keys 1 and 2:

KEY 1 : (600 Hz, 1200 Hz)

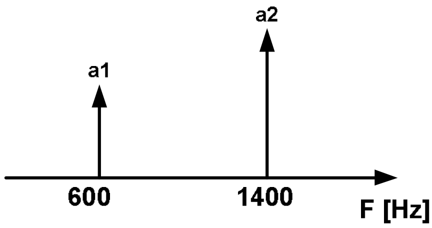
KEY 2 : (600 Hz, 1400 Hz)

- between the inferior and the superior frequency of key 1 we have a harmonic relation

The transmission line is represented by a nonlinear characteristic: $F(x) = x^2+x$.

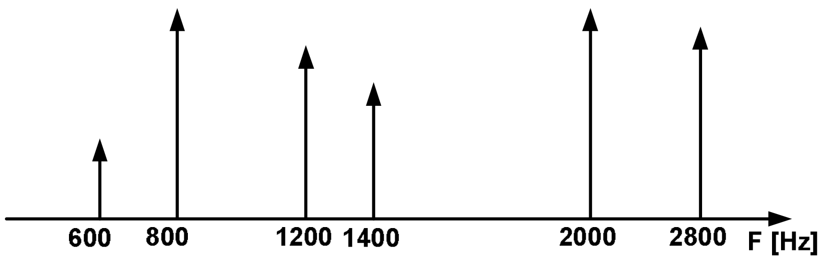
1. When key 2 is pushed the DTMF signal transmitted on the line is the following:

$$S_{DTMF}(t) = a_1 \cdot \sin(2\cdot\pi\cdot 600\cdot t) + a_2 \cdot \sin(2\cdot\pi\cdot 1400\cdot t)$$



2. At the input of the DTMF receiver, the signal affected by the nonlinear characteristic of the line is the following:

$$s_r = a_1\cdot\sin(2\cdot\pi\cdot 600\cdot t) + a_2\cdot\sin(2\cdot\pi\cdot 1400\cdot t) + a_1\cdot a_2\cdot\cos(2\cdot\pi\cdot 800\cdot t) - \frac{a_1^2}{2}\cdot\cos(2\cdot\pi\cdot 1200\cdot t) -$$
$$- a_1\cdot a_2\cdot\cos(2\cdot\pi\cdot 2000\cdot t) - \frac{a_2^2}{2}\cdot\cos(2\cdot\pi\cdot 2800\cdot t) + \frac{a_1^2}{2} + \frac{a_2^2}{2}$$



Neglecting the 800Hz, 2000Hz and 2800Hz frequency components (these tones are not part of the DTMF system) we can notice that the DTMF receiver can not decide between key 2 (600Hz, 1400Hz) and key 1 (600Hz, 1200Hz), in the condition when the signal levels are not considered. It results that it is necessary to avoid harmonic relations between the inferior and superior frequencies.

The electromagnetic phone device

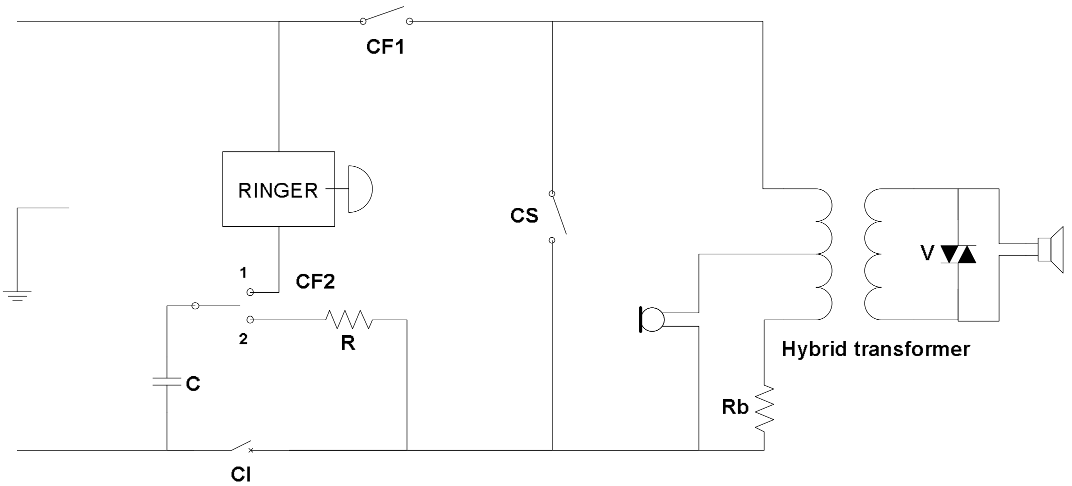


Figure 5 – The electromagnetic phone device

- Serial R,C circuit in parallel with CI – it is a protection circuit against sparks
- Varistor V – it is a voltage limiting device for protection against overvoltage

The hybrid transformer (differential transformer)

In the case of telephone devices the hybrid transformer has the role to separate the microphone and speaker circuit, suppressing the so called **side effect.** This effect consists in passing the voice from the microphone circuit to the speaker circuit, the subscriber hearing own voice. To suppress the side effect the attenuation between ports 3 and 4 (Figure 6) has to be large.

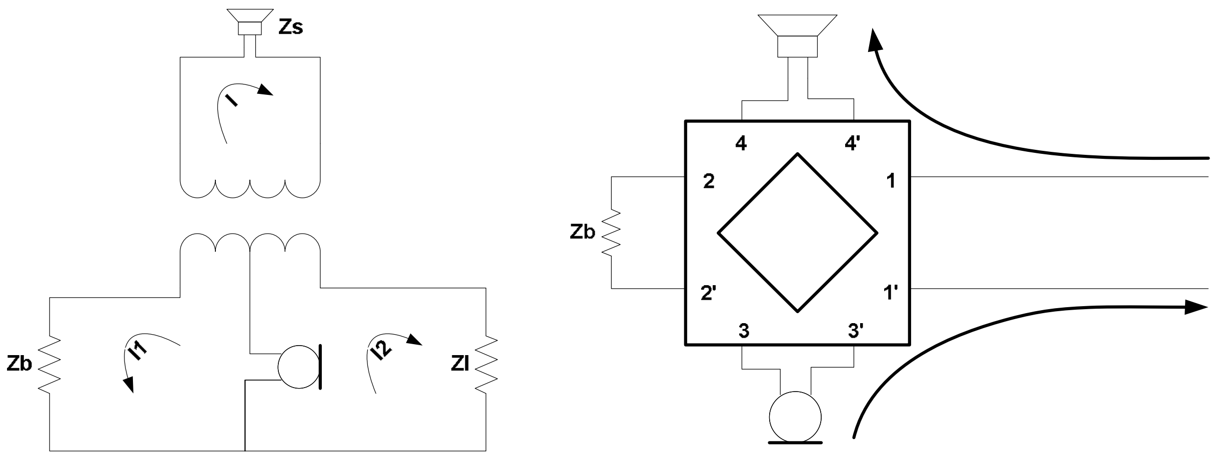


Figure 6 – The hybrid transformer

We propose to compute the balance condition, when separation is ensured between the microphone and the speaker:

- we apply the Kirchhoff law in the secondary circuit

$$Zs*I + j*w*Ls*I + j*w*Lc*I1 - j*w*Lc*I2 = 0$$

Ls – inductance of the secondary winding
Lc – coupling inductance between the primary and secondary winding

$$I = (I2-I1)*K(w)$$

- we impose the condition I=0 (we do not have signal in the speaker circuit) => I1=I2 => **Zl = Zb**

- the balance condition: Zl = Zb

Zl – line impedance
Zb – balance impedance
This hybrid ensures galvanic separation between the line and the speaker circuit.

The electronic hybrid

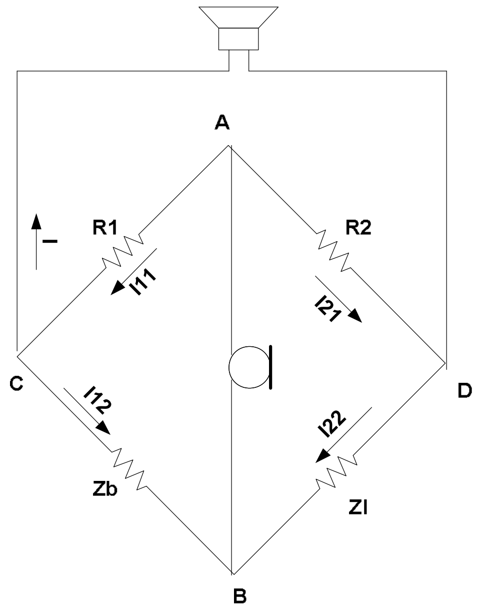


Figure 7 – The electronic hybrid

We impose the condition: I=0 => I11=I12=I1, I21=I22=I2

V_{AB} = I1*(R1+Zb) = I2*(R2+Zl) => I1/I2 = (R2+Zl)/(R1+Zb)   (1)
V_{CD} = I1*R1-I2*R2 = 0 => I1/I2 = R2/R1   (2)
From (1) and (2) => **balance condition: R2/(R2+Zl) = R1/(R1+Zb)**
This hybrid does not ensure galvanic separation.

Supplementary information about the telephone device:
http://www.tkk.fi/Misc/Electronics/circuits/teleinterface.html