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TRANSMITTING AND RECEIVING DIGITAL COMMANDS THROUGH TELEPHONE NETWORK USING DUAL TONE MULTI-FREQUENCY SIGNALS (II)

BY

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Abstract. The paper describes the hardware structure of a telephone line interface that, used in conjunction with an application system equipped with an ATMEL family microcontroller and with the interface to the user process, performs the transmission/reception of digital commands between remote processes using DTMF signaling through telephone networks. The line interface performs various telephone line operations, *i.e.* monitoring the line and the telephone set status, sending outgoing calls to the exchange, receiving incoming call signals, separating the directions of the DTMF signaling exchanged through the line, receiving and sending DTMF signals, receiving tones and transmitting information voice messages.

Key words: telephone line interface; subscriber line monitoring; receiving and transmitting DTMF signals; application system with microcontroller.

1. Introduction

In order to use telephone networks to transmit and receive digital commands that require a small amount of binary data to be exchanged between remote user processes, Dual Tone Multi-Frequency (DTMF) signaling can be

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used. This allows one to command processes using a simple telephone set, knowing the communications protocol and the system access key. The basic structure of a system for transmitting and receiving digital commands between user processes is shown in Fig. 1 (Borcoci, 1994; Rădulescu, 1994).

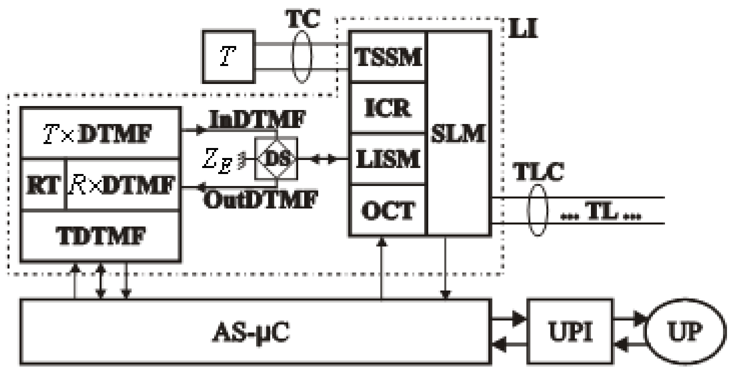

Fig. 1

The system consists of a line interface (LI) and a user process interface (UPI) both managed by an application system equipped with an ATMEL family microcontroller (AS-µC). The line interface performs the following tasks: subscriber line monitoring (SLM), direction separation for DTMF signaling using an active differential system (DS) and an impedance balancer (Z_E), DTMF signals transmission (*T*×DTMF) and reception (*R*×DTMF) through a DTMF transceiver (TDTMF).

The hardware structure of the line interface required for performing all the necessary monitoring functions and the structure of the differential system have been presented in a previous paper (Duma, 2012).

The present paper covers the command and control of the DTMF transceiver used for generating and receiving DTMF tones using a microcontroller-based application system. Further documentation is being prepared to cover the other topics related to the software implementation of: the DTMF signal transmission and reception processes, incoming call reception, telephone line monitoring, exchange-originated tones reception and, finally, the transmission and reception of the digital commands between remote user processes.

2. Transmitting and Receiving DTMF Signals

In order to transmit and receive DTMF signals, the integrated transceiver CM8880 is used. This transceiver includes a DTMF signal generator, an industry standard DTMF receiver and an interface to microcontroller-based systems. The circuit features an automatic tone burst mode with precise timing, a call progress mode, adjustable guard time characteristics, a low power consumption and provides a direct and simple interfacing compatible with microcontrollers from various families.

The DTMF generator consists of counters that automatically and precisely transmit the tone bursts and the pauses between them, digital–analogue converters and switched capacitor filters that insure low distortions and high accuracy DTMF signals. The call progress operation mode is used for detecting the tones transmitted by the telephone exchange on the subscriber line.

The DTMF receiver is a high performance one, due to the input operational amplifier with adjustable gain, the switched capacitor that separates the lower and the higher frequency groups, the digital algorithm and the code converter used for DTMF tones detection from the telephone line.

The internal structure of the CM8880 transceiver is shown in Fig. 2 (California Micro Devices, 2000). The notes used stand for: RC – row counter, CC – column counter, DAC – digital–analogue converter, SCF – switched capacitor filter, SA – summing amplifier, TBG – tone burst gate, GLCB – generator logical control block (for the DTMF generator); IA - input amplifier, TRNF – tone rejection notch filter, HGF – high frequency group filter, LGF – low frequency group filter, DAB – digital algorithm block, CCB – code converter block, LCB – logical command block, RLCB – receiver logical command block, Osc – internal oscillator, PCPB – power supply and circuit polarizing block (for the DTMF receiver); *R*×DR – receiver data register, CRA – control registers *A*, CRB – control registers *B*, SR – status register, *T*×DR – generator data register, DBB – data bus buffer, ILB – interrupt logical block, IOCB – input–output control block (for the microcontroller interface).

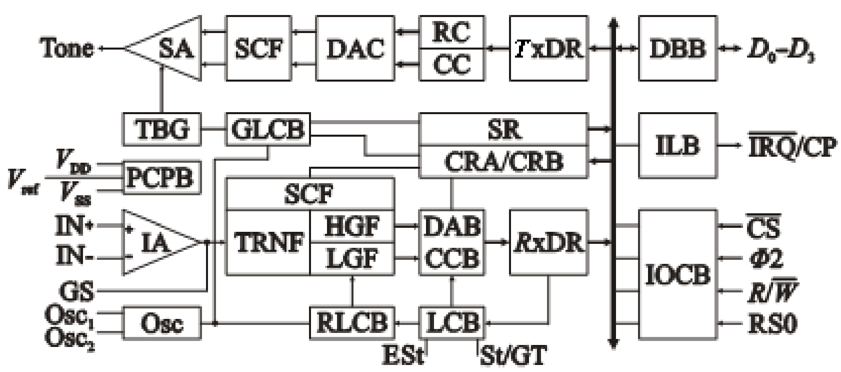

Fig. 2

The DTMF generator provides 16 tone pairs, each consisting of two sinusoid signals selected from two standard groups of four frequencies each. Each tone consists of the sum of a frequency from the lower group and one from the higher group. Table 1 includes the standard DTMF frequencies for digits/characters and the corresponding binary code. All the signals are obtained by dividing with adequate time constants the clock signal produced by the internal oscillator connected to an external quartz crystal. This method produces signals having a slightly different frequencies from the standard, but with a minimal error between –0.5% and +0.75%.

Table 1

f/F	F_1 = 1,209 Hz	F_2 = 1,336 Hz	F_3 = 1,477 Hz	F_4 = 1,633 Hz
f_1 = 697 Hz	**1** (0001)	**2** (0002)	**3** (0011)	**A** (1101)
f_2 = 770 Hz	**4** (0100)	**5** (0101)	**6** (0110)	**B** (1110)
f_3 = 852 Hz	**7** (0111)	**8** (1000)	**9** (1001)	**C** (1111)
f_4 = 941 Hz	***** (1011)	**0** (1010)	**#** (1100)	**D** (0000)

The generation method of a tone consists of delivering 32 samples at specific time periods. After writing the generator data register, the 4-bits data is stored and converted into two 8-bits code words. These code words are in fact the time constants for the programmable counters that deliver the time intervals between samples and set the signal frequencies. The tones are synthesized with row and column programmable counters, digital–analogue converters and switched capacitor filters. Two circuits are used to produce the row and column tones with low distortion and high accuracy that are in the end summed with a low noise amplifier.

The standard DTMF signal transmission duration is obtained in burst mode, when symmetrical tones and pauses are generated with 51±1 ms durations – a typical telephony application value. Different operating modes with different durations can be obtained by adequate command of the transceiver's control registers.

The operational amplifier at the input of the DTMF receiver requires for two resistors (R_{in}, R_F) to be connected to the inverting input in order to set the gain and the reference voltage (V_{ref}) to the non-inverting input (Fig. 3). The separation of low and high group frequencies is achieved by applying the DTMF signals to the inputs of two sixth-order switched capacitor band-pass filters. The high and low group filters include notch filters on 350 Hz and 440 Hz frequencies, for dial tone rejection. Each filter output is followed by a single-order switched capacitor filter section which smoothes the signals *prior* to limiting. Limiting is performed by high-gain comparators with hysteresis to prevent detection of unwanted low-level signals.

Following the filter section is a decoder which employs digital counting techniques to determine the frequencies of the incoming tones and to verify that the incoming tones correspond to standard DTMF frequencies. A complex averaging algorithm protects against tones simulation by extraneous signals (*e.g.* voice), while still providing tolerance to small deviations in frequency. When two valid tones are received (one from the low frequencies group and the other from high group) ESt output becomes active. Any subsequent loss of the DTMF signal characteristics will cause ESt to assume an inactive state. Before storing the binary code of a decoded pair of tones, the receiver checks for a valid DTMF signal duration. This check is performed based on an external time constant, *RC*, driven by ESt. The same circuit, in opposite sense, is also used for checking the pause duration between DTMF signals.

The interfacing of the CM8880 transceiver with a microcontroller from ATMEL family, for receiving and transmitting DTMF signals, is made as shown in Fig. 3 (California Micro Devices, 2000).

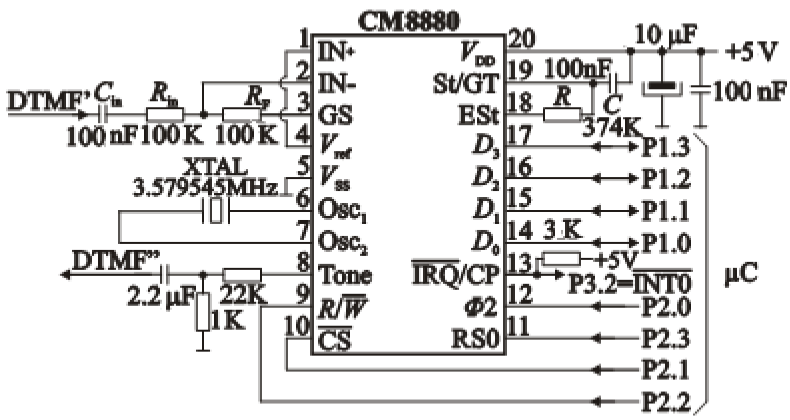

Fig. 3

This interfacing is simple and requires but a few external components: a capacitor (100 nF), that interrupts the continuous component of the DTMF signal from the input; two resistors (100 KΩ) for setting the gain of the input amplifier; a quartz crystal of 3.579545 MHz for the internal clock oscillator; a resistive divisor (22 KΩ in series with 1 KΩ) at the output of the generator for dividing the sent signal, followed by a capacitor (2.2 μF) for breaking the continuous component; a resistor to the open drain output IRQ/CP for interrupt requests; a capacitor (100 nF) and a resistor (374 KΩ) for the external time constant *RC*, that validates the DTMF signals and pauses duration; and two capacitor for decoupling the circuit (100 nF and, respectively, 10 μF).

The DTMF transceiver has two control registers (CRA and CRB), a status register, a receiver data register that contains the binary code of the last two valid tone pair and a generator data register where the microcontrollers writes the binary data that will determine the corresponding tone pair to be generated at the output.

The writing and reading operations into/from the transceiver's registers are made by the microcontroller by managing the signals $\overline{CS}$ (circuit select), RS0 (internal register select), $R/\overline{W}$ (read/write), $\Phi 2$ (clock) and D_3–D_0 (4-bits data bus); the basic functions implemented are detailed in Table 2.

Table 2

$\overline{CS}$	RS0	$R/\overline{W}$	$\Phi 2$	$D_3 - D_0$	Function
1	x	x	x	HZ	Invalid transceiver
0	0	0	⎍	In	Write data in DTMF generator
0	0	1	⎍	Out	Read data from DTMF receiver
0	1	0	⎍	In	Write the control register
0	1	1	⎍	Out	Read the status register

The structure of the data reading cycle *R*xDR after the microcontroller receives a tone or a status from SR is shown in Fig. 4. The command signals $\overline{CS}$, RSO and $R/\overline{W}$ must be active at least 23 ns (t_{AS}/t_{RWS}) before the rising edge of clock $\Phi 2$; after no more than 150 ns (t_{DDR}) from this edge of $\Phi 2$, the data becomes available on the data bus $D_3 - D_0$ for the microcontroller to process. After the falling edge of $\Phi 2$, the data is still available on the bus for at least 22 ns (t_{DHR}), while the command signals must be maintained for at least 10 ns (t_{AH}/t_{RWH}).

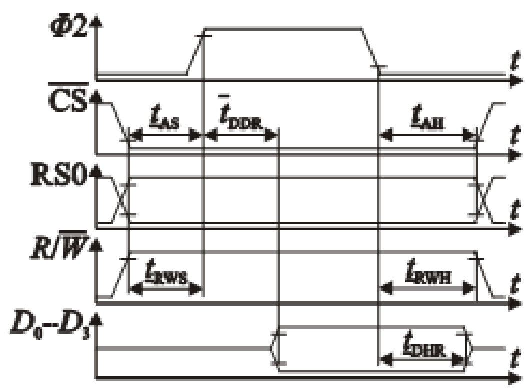

Fig. 4

The structure of the microcontroller write cycle, of a data into *T*×DR for generating a tone or of a control word in CRA and CRB, is shown in Fig. 5. The time restrictions for the command signals activation before the rising edge and their persistence after the falling edge of the $\Phi 2$ signal are the same as for the reading cycle. The binary data that is to be written must be stable for at least 45 ns (t_{DSW}) before the rising edge of the $\Phi 2$ clock and at least 10 ns (t_{DHW}) after its falling edge.

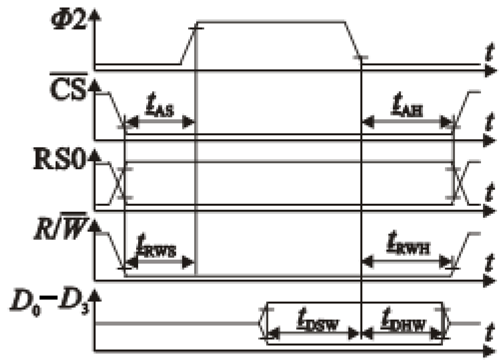

Fig. 5

The transceiver’s input and output DTMF signals are galvanically isolated from the external medium represented by the telephone network line, using linear isolation amplifiers (Fig. 6).

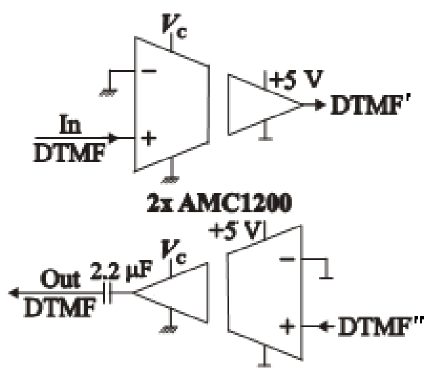

Fig. 6

Two AMC1200 precision isolation amplifiers are used, with the output separated from the input circuit by a silicon dioxide barrier that is highly resistant to interference (Texas Instruments, 2011). This barrier has been certified to provide galvanic isolation of up to 4 KV peak value. These isolation amplifiers feature very low nonlinearity (0.075% max. at 5 V), low offset error (1.5 mV max.), low noise (3.1 mV_{RMS} typically), a signal input band of minimum 60 KHz, fixed gain (8), high common mode rejection ratio (108 dB), etc.

The input section of the isolation amplifiers admits a power supply range voltage between 4.5 V and 5.5 V (typically 5 V) and a typical supply current of 5.4 mA (8 mA max.), while the output section allows a power supply voltage between 2.7 V and 5.5 V (typically 5 V) and a typical supply current between 3.8 mA and 4.4 mA, depending on the voltage (6…7 mA max.). One input section of an isolation amplifier and one output section of the other are powered from the 5 V supply of the microcontroller and of the DTMF transceiver, while the other two sections must be powered from the external medium (5 V voltage, 9.8 mA typical current, 15 mA max.). The TEA1062 circuit, used for telephone line interfacing, includes an internal stabilizer for power supply from the telephone line, which delivers on V_{CC} terminal a 5 V voltage for powering external peripheral circuit, but only up to 1.2 mA, which makes it unusable for this application (Philips Semiconductors, 1997).

In order to power the isolation amplifiers sections from the external medium, it is used a TEA1081 circuit in stabilized power output mode. The hardware structure is depicted in Fig. 7 (Philips Semiconductors, 1994).

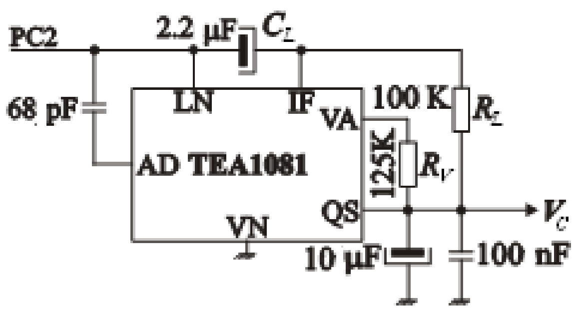

Fig. 7

This circuit provides power based on the current surplus from the telephone line that is normally shunted by the voltage regulator of the speech circuit used for sense separation for DTMF signals (TEA1062). The high input impedance of the circuit allows the direct connection to the telephone line through a diode bridge. An internal amplifier along with an internal resistor (R_S = 20 Ω) and an external low-pass filter, R_LC_L , perform the function of an inductance, $L = R_LC_LR_S$, of 4.4 H that achieve the required characteristics. Under the control of another internal amplifier a transistor supplies peripheral devices, while another transistor minimizes line signal distortion. All internal circuits are biased by a temperature and line voltage compensated reference current source.

In the stabilized output voltage operating mode it is necessary to connect a resistor, R_V , between terminals QS and VA. The output voltage is maintained constantly at $V_C = 2I_6R_V$, where I_6 is an internal control current that has the typical value of 20 µA. For a R_V value of 125 kΩ, a stabilized output voltage of 5 V is obtained, at a maximum output current of 30 mA, if this current can be drawn from the telephone line along with the necessary reserve for internal powering the circuits.

The output voltage is stabilized to value V_C, if the telephone line voltage is at least $V_C + I_1R_S + 2.5$, where I_1 is the current from the telephone line.

3. Conclusions

The hardware structure described for transmitting and receiving DTMF signals, based on transceiver CM8880, the line monitoring interface and the circuit for sense separation, was built in practice. Along with the application system equipped with microcontroller AT89S8253 and the interface of the user process, it performs the transmission/ reception of digital commands using DTMF signaling through the telephone network between remote processes. This structure is usable in applications that require low amount of command data to be transferred, when command delays are not an issue and only infrequent maintenance is required.

The CM8880 transceiver interfacing is simple and requires few external components. This microcontroller–driven circuit allows also the reception of the tones from the telephone line in order to determine the progress of the automated telephone connections performed by the remote user processes.

The line interface is galvanically insulated using photocouplers and relays for various digital signals, and, respectively, precision isolation amplifiers for the DTMF signals.

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TRANSMITEREA ŞI RECEPȚIONAREA COMENZILOR NUMERICE PRIN REȚEAUA TELEFONICĂ UTILIZÂND SEMNALE DTMF (II)

(Rezumat)

Se descrie structura hard a unei interfețe de linie telefonică care, împreună cu un sistem de aplicație echipat cu un microcontroler din familia ATMEL şi interfața procesului utilizator, realizează transmiterea/recepționarea de comenzi numerice cu semnale DTMF prin rețeaua telefonică între procese aflate la distanță. Interfața de linie monitorizează linia telefonică care constă din supravegherea liniei, transmiterea unui apel la centrală, recepționarea semnalului de apel şi supravegherea terminalului telefonic, separarea sensurilor semnalelor DTMF vehiculate prin linie, recepționarea şi transmiterea semnalelor DTMF şi transmiterea unor mesaje vocale de informare.