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Serial Port Based Telephone Card

Group: D04

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Abstract

The aim of this project is to develop a telephone card, interfaced to the PC via serial port that would emulate a telephone set. Using this card, a user would be able to make and receive telephone calls using a software interface on the PC. In addition, this card would also allow a multitude of applications to be built on it in software. Telephone line sharing over a network, interactive voice response, voice mail/messaging are some of the applications we envisage.

1. Introduction

We propose to develop a telephone card, interfaced to the PC via serial port that would emulate a telephone set, as shown in Fig 1. When there is a ring on the telephone line, the user would be notified, and the card would allow the user to leave the system "on-hook" or to attend to the call. Similarly, the user would be able to enter a telephone number into a software interface on the computer and the card would dial that number. With the basic software control of the telephone line that this card provides, a multitude of applications can be built on it in software. Telephone line sharing over a network, interactive voice response, voice mail/messaging are some of the applications we envisage.

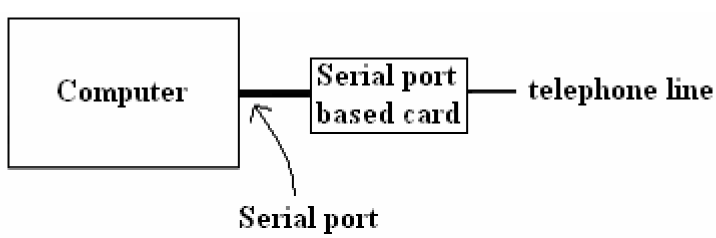

Fig 1: Target setup

2. Design approach

The entire system can be broken down into three basic blocks: the telephone line interfacing circuit, the codec circuit and the data transfer block. The telephone line interface and the codec circuit form the analog part of the system. The Data Transfer block is responsible for the transfer of the serial data streams to and from the codec to the computer. A block diagram of the system is shown in Fig 2. Each block is described in the following subsections.

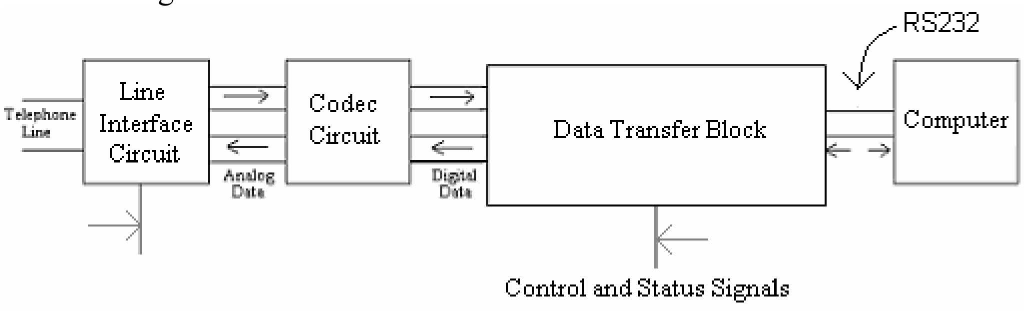

Fig 2: Block Diagram of the System

Telephone Line Interfacing Circuit

Since the telephone line carries the incoming and outgoing voice signal, we need a circuit that will separate them into two separate lines, to enable separate A/D and D/A conversion. This circuit plays the role of the hybrid transformer in early telephones, in addition to detecting rings, dialing and controlling the On-Hook/Off-Hook state of the system.

Codec circuit

The codec (coder-decoder) circuit digitizes the incoming analog voice signal from the telephone line and simultaneously converts to analog form the digital signal coming from the computer corresponding to the user’s speech. This duplex A/D and D/A conversion occurs at a sampling rate of 8 kHz.

The codec circuit block also includes the generation of the various clock signals required for the duplex A/D and D/A conversion.

Data Transfer Block

The digital input and output to most voice codecs is in the form of serial data streams. This block has to communicate these streams to the computer via Serial Port. Since the codec samples at 8 kHz, generating a byte per sample, the serial port needs to transfer data at 8kB/s. Adding the start and stop bits, we require a minimum Baud rate of 80 kb/s.

At 115.200 kb/s, the RS232 link will transfer voice both directions, with flow-control required for the transfer from the PC to the microcontroller.

A lower level circuit block diagram of the system is shown in Fig 3.

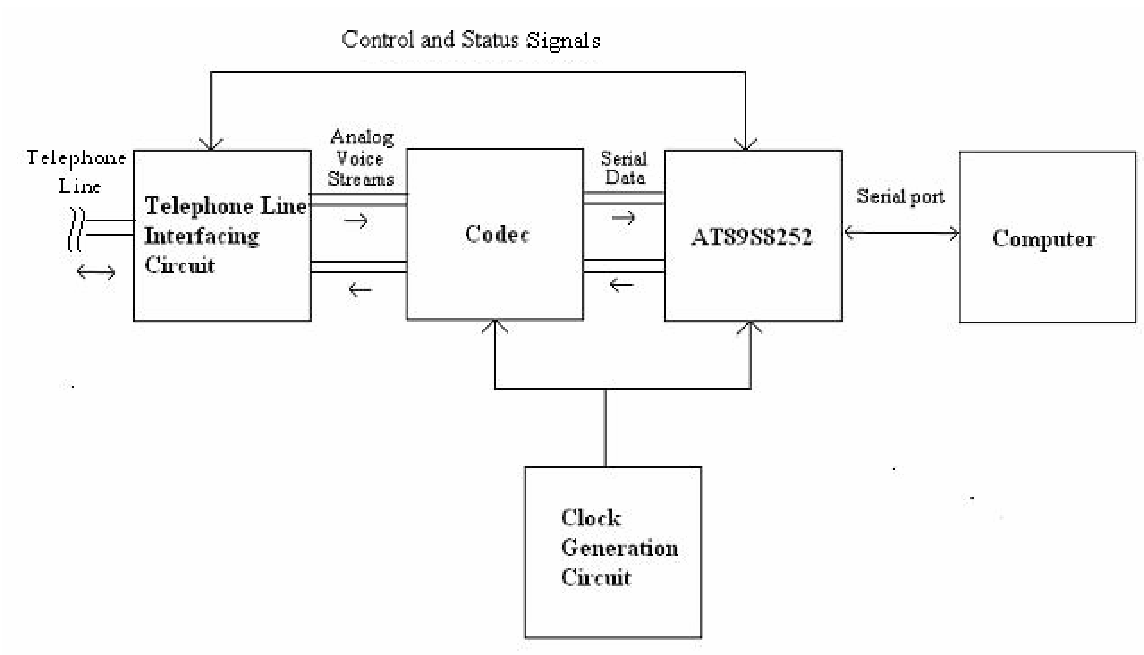

Fig 3: Circuit Block Diagram

3. Hardware

Block-by-block description of the hardware is given in the following subsections.

Telephone Line Interfacing Circuit

The telephone line interfacing block consists of a telephone line interfacing IC (TEA1062) and its peripheral circuitry, the ring detection circuitry, and the dialer circuitry as shown in Fig 4. The standard circuit has been modified to suit the present application. This IC and its peripheral circuitry separates the incoming and outgoing Analog voice streams on the telephone line enabling independent and duplex A/D and D/A conversion. The standard circuit has been modified to suit the present application.

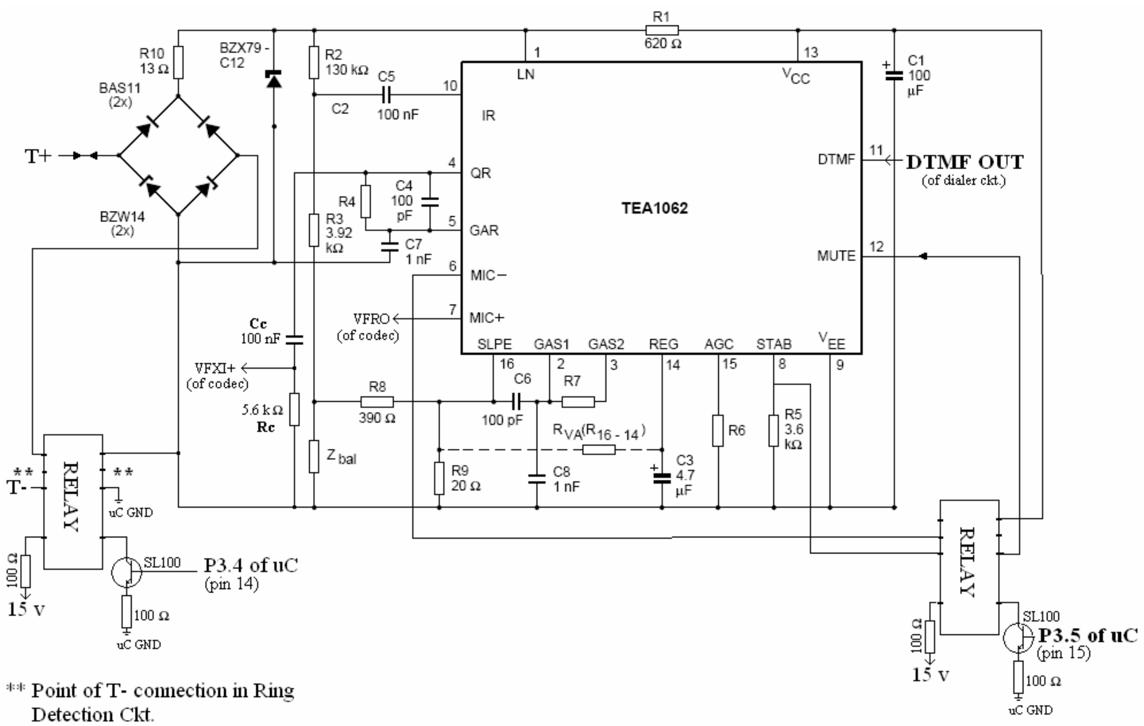

Fig 4: TEA1062 Circuit Diagram

The speaker output at GAS2 is fed to an RC filter. The resistance of the RC circuit is connected to the codec ground at the other end. The output across the resistor will be the AC part of the speech signal with respect to the codec ground.

The analog output of the codec is given as input to Mic+ (pin 7) and pin and Mic- (pin 6) is connected to ground.

The values of Rc and Cc selected for this purpose:
R=5.6 kΩ, C=100 nF

This series R-C combination gives us a high pass filter with a cut-off of approximately 300 Hz.

Ring Detection circuit

This circuit is responsible for notifying the microcontroller (AT89S8252) of a ring on the phone line.

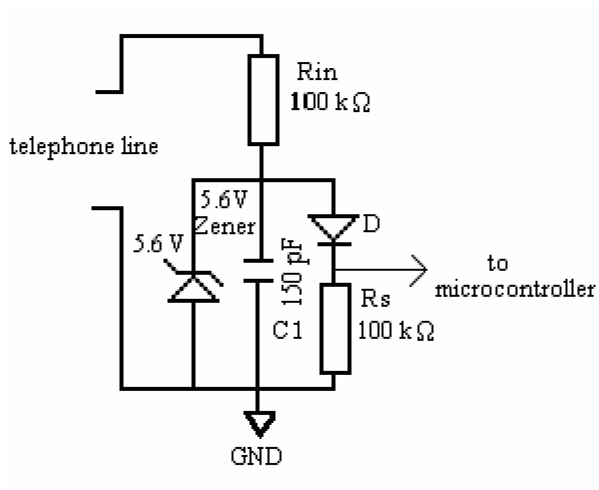

Fig 5: Ring Detection Circuit

In the on-hook state, when there is no ring, the telephone line provides a constant 48 volts DC voltage. In this state, the zener diode is reverse biased with 5.6 volts across its terminals. This voltage is, in turn, fed to the diode D and resistor Rs. The diode becomes forward biased and conducts. Approximately, 4.9 volts appears across the resistor Rs.

When there is a ring on the line, a 75 volts AC voltage is superimposed on the 48 volts DC. When the net voltage across the telephone line goes high, the circuit works in the same way as above. When the net voltage across the telephone line falls to a large negative value, the zener diode is forward biased. This makes the diode D reverse biased and hence the voltage across Rs remains close to 0V.

The voltage across Rs is fed to the microcontroller.

Dialer circuit

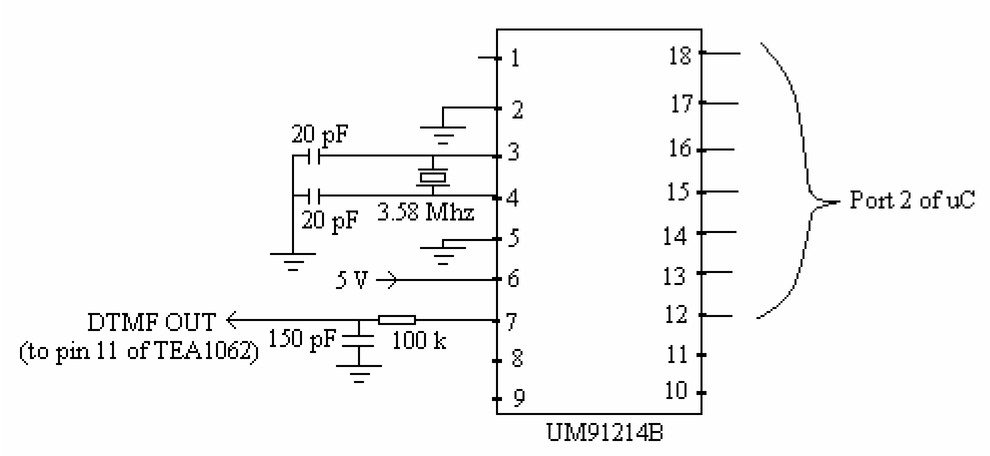

Fig 6: Dialer Circuit

We use the DTMF tone generator chip UM91214A to dial. Port 2 of the AT89S8252 is used to specify the digit to dial.

Clock generation circuit

The codec requires a 2 MHz master clock, provided by using a 2 MHz crystal and a CMOS inverter.

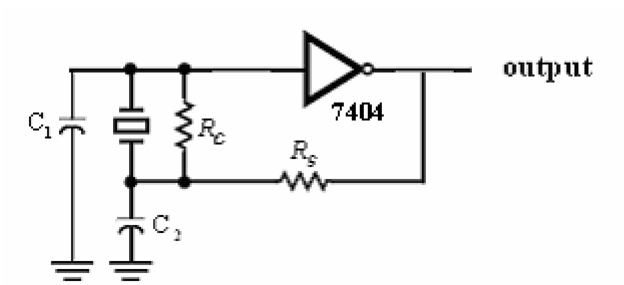

Fig 7: 2 MHz Master Clock generation circuit

The values of R_c and R_s used for this purpose are 560 kΩ and 2.2 kΩ respectively. C_1 and C_2 are 33pF each.

Also, 8 kHz (divide-by-256) and 125 kHz (divide-by-16) clocks **synchronous** with the Master clock are needed to drive the sampling (and reconstruction) process and to synchronize the serial transfer between the codec and the AT89S8252. For this purpose, we use CD4040, a 12-stage ripple carry binary counter.

Codec circuit

We use the voice codec TP3056B from TI. The connections between the codec and the telephone line circuit are depicted in the circuit below. The connections on the digital end are explained in the next section.

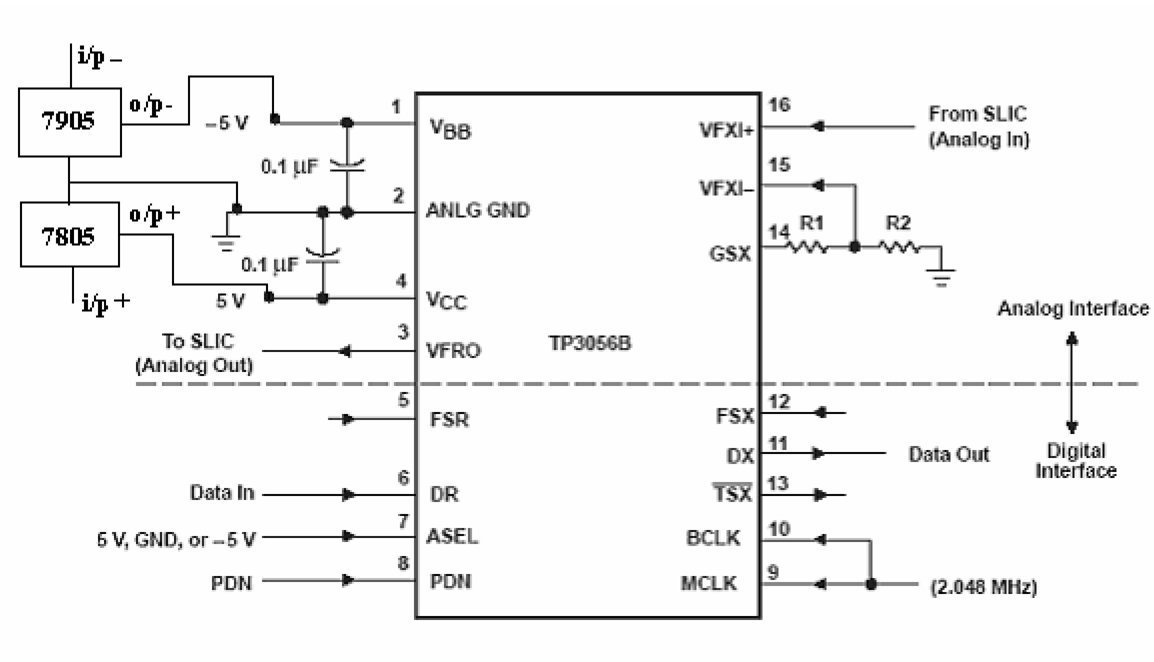

Fig 8: The codec and peripheral circuitry

Note: The codec requires the master clock (MCLK) to be 2.048 MHz. Due to non-availability of the crystal of the specified value, a 2 MHz MCLK is used. Correspondingly, the sampling frequency is not 8 kHz but 7.813 kHz. The output of the codec is not affected noticeably due to this variation.

AT89S8252

The codec serial format resembles that of an SPI slave device. Hence we chose the ATMEL AT89S8252, which has an SPI interface and a serial port too. For the SPI link both the codec and the AT89S8252 act as slave devices connected to each other, with the clock and slave select being generated by the clock generation circuit as shown in Fig. 9.

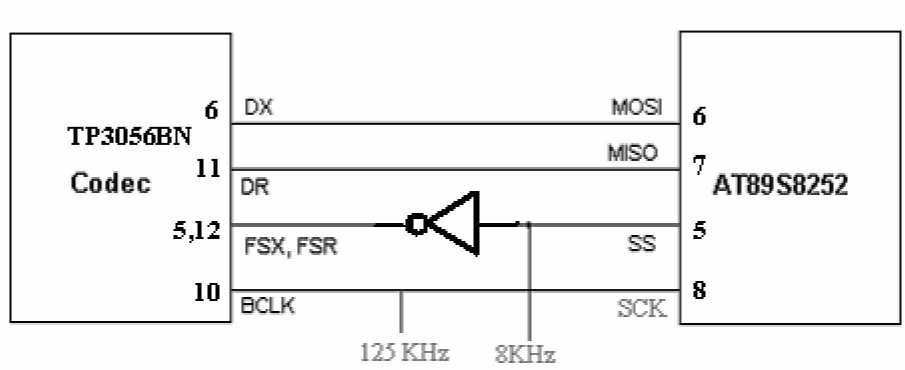

Fig 9: SPI interface between codec and AT89S8252

The AT89S8252 is connected to the PC COM port via MAX232 as shown in Fig. 10.

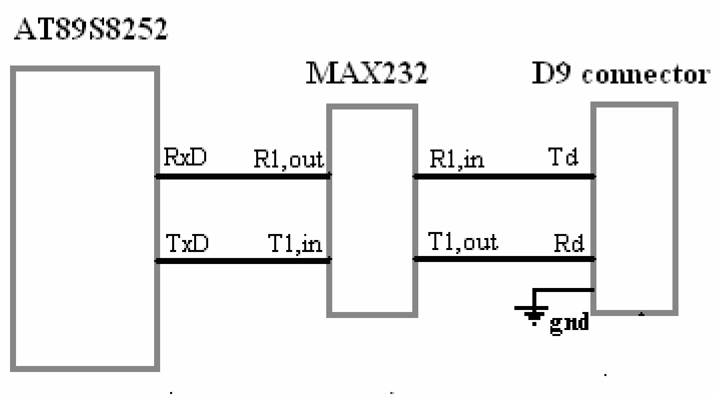

Fig 10: Serial connection between the PC and AT89S8252

There are three grounds in the circuit namely, the microcontroller ground from supply, the ground for the telephone line interfacing circuit and the telephone line T-. The appropriate switching is done by the use of relays as shown in Fig 4.

4. Test Procedures

Telephone Line Interfacing Circuit

We connect the circuit as shown in Figure 4 leaving the Mic+ and Mic- connections open. Across Mic+ and Mic- we connect a pure sine wave signal from a signal generator and listen at the other telephone connected to the Telephone tester. A continuous tone is heard. Changing the amplitude of the sine wave changes the loudness. Frequency variation changes the pitch of the tone that is heard. This is a crude test, however we have been unable to come up with a better test procedure for this block.

Codec Circuit

We applied a pure sine wave signal as input from a signal generator and shorted the digital output of the codec to the digital input. The analog output of the codec is a pure sine wave with a constant phase difference and gain with respect to the input, as long as the input frequency is within the pass-band of the codec (about 3.4 kHz).

The SPI link

We connect together only the codec and the AT89S8252, programmed to send back the byte it receives on the SPI link. On applying a pure sine wave to the codec analog input, a sine wave of the same frequency, with a constant phase difference with respect to the input signal, is obtained at the analog output of the codec, as long as the input frequency is within the pass-band of the codec (about 3.4 kHz).

The RS232 link between the PC and AT89S8252

By programming the 8252 to send back the byte it receives on the serial port, we were able to send a file down the serial link, and recover it at 115200 Baud.

Ring Detection

The output of ring detection circuit switches between 0 and 5 Volts when there is a ring on the telephone line and remains at a constant 5V when there is no ring on the the telephone line.

Dialing and Hook control

We have tested controlling the On/Off Hook condition of the line and the dialing of a number using the microcontroller.

5. Current status

All blocks of our system have been tested separately. The system was successfully implemented on bread-boards.

6. References

[1] "*TEA1062; TEA1062A Low Voltage Transmission Circuits with Dialer Interface*", Philips Semiconductors.

[2] *"TP3056B Monolithic Serial Interface Combined PCM Codec and Filter"*, Texas Instruments.

[3] *"AT89S8252 8 Bit Microcontroller with 8K Bytes Flash"*, Atmel.

[4] *"MAX232 +5 Volt Powered Multichannel RS-232 Drivers/Receiver"*, Maxim.

[5] *"UM91214 Tone/Pulse Dialer"*, UMC.