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Design and analysis of a standalone localized telecommunication network for undeveloped areas

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Abstract - Due to the commercial nature of upcoming technologies, often the rural or remote inaccessible areas continue to remain undeveloped in terms of telecommunication facilities. This project develops a ***stand alone 'two wire' localized telecommunication*** *network for such places. The microcontroller ATmega32 is used, with inbuilt Analogue to Digital Converter and inbuilt Two wire Serial Interface module.*

Normal telephone like functionality is designed for calling purpose, hence the system is user-friendly. The ***total cost of each 'handset' is just 450 Rs, best suited for implementation by NGO's and charitable trusts. The*** *hardware is robust, with negligible circuitry, hence no maintenance problem/cost. There is no recurring calling cost, hence suitable for poor people. The current requirement is low, hence design is very favorable even with solar power. This makes the network to function as a complete stand alone system.*

In this system, we can connect maximum 128 different ***'handsets'. However, at a time maximum 10*** *bidirectional calls can be made (that is 20 users can be active simultaneously) due to limited data rates. This limitation is however not a disadvantage, owing to the fact that this system is designed for remote inaccessible* ***places with 'low population density' and 'low cellular traffic'.***

Key Words: Low cost telecommunication network, intercom, solar powered voice connectivity.

1. BASIC BLOCK DIAGRAM

The basic functionality of the system can be described by the following block diagram in Fig 1. Each device is further elaborated by the block diagram in Fig 2.

All the devices are connected in parallel across the 'SDA' and 'SCA' buses. Synchronous clock pulse is transmitted over the SCA and the data is transmitted serially over the SDA. The algorithm used for proper bus allocation is discussed later.

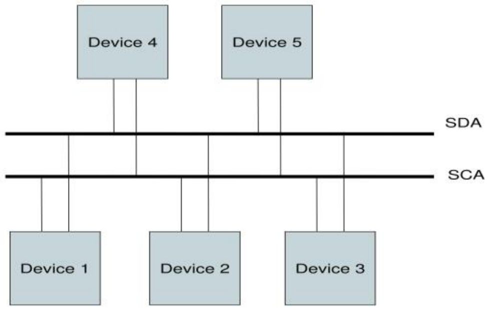


Fig 1.

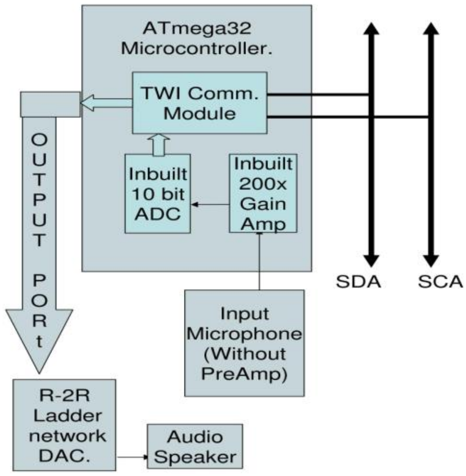


Fig 2.

Figure 2 shows the internal blocks of each device. Functionality just like an ordinary telephone is achieved as follows.

- The functionality of taking audio input is accomplished in the most optimized and innovative way, without the need of an 'pre-amplifier' that is usually needed with a microphone. Detailed description is provided under the title calculations.
- The functionality of generating audio output is made by using a R-2R ladder network Digital to Analogue Converter. The need of amplifier was eliminated hear by making use 'low values of R' so that the current of the microcontrollers output port is itself sufficient to source a small audio speaker.

The TWI module is an addressable Two Wire bidirectional data communication module which is used in multi-master mode to accomplish the transmission-reception of voice-call data.

2. DETAILED MATHEMATICAL CALCULATIONS FOR IMPLEMENTATION OF EACH BLOCK

A. Implementation of the audio input block (data acquisition from the microphone):

- The microphone generates output in the range of 0-800 μV, (and hence usually requires a pre-amp).
- But we have not used this external pre-amp, rather this 0-800 μV signal is given to the microcontroller's inbuilt 200x gain amplifier which amplifies the signal upto (200 x 800 μV) = (0.16 V)
- Now, this voltage level is still pretty low for being the input of the ADC. However it is made possible as follows:
  - The Refrence voltage for the ADC is set to its minimum value i.e. 2 V.
  - The resolution of the ADC is set to 10 bit, (hence there are 2^{10} distinct quantized levels, i.e. 1024 different levels)
  - Size of each step is (VRef/1024) = (2/1024) = 0.001953 V.
  - So, our microphones amplified input voltage range of 0.16 V can be sampled into (0.16 / 0.01953) = 82 distinct values.
  - Since 2^6<82 < 2^7, hence the sound quality is intermediate between 6-bit audio and 7-bit audio.
- These sound signal received is quantized in the range of 0-82 discrete steps. This low quality is compensated by implementing a predictor algorithm which estimates and regenerates this data into the full 8-bit range of 0-255 discrete steps.

B. Implementation of audio output block. (generating audible sound from R-2R ladder network at the output port without any amplifier)

- ATmega32 data sheet specifies the maximum output driving capacity of each output pin as 50 mA. Our circuit operates at 3.7 V, and hence we used the value of 2R = (3.7 V/50 mA) = 74 Ω.
- The peak value of current thus generated will be (50+25+12.5+6.25+3.13+1.56+0.0.78+0.4) mA = 100mA which is sufficient to drive an audio speaker, typically used in mobile phones.

Figure 3 shows the schematic of the R-2R ladder network as implemented in microcontroller output port.

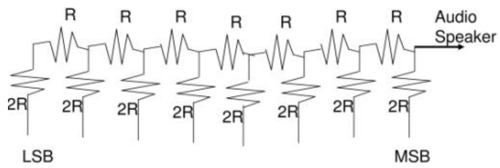


Fig 3

C. Implementation of TWI communication module.

All the microcontrollers are inter connected on the common TWI bus. Human voice is band limited by 4 kHz, hence by Nyquist principle the sampling rate of 8 kHz is sufficient. Each Sample is 7 bit in size, we added one parity bit, so data is now generated at 8k bits per second

Besides this, there is 7 bit address with each data frame, and approximate add on is of 10 bits, so 10k bits per second is further to be transferred with every data frame. Effective bandwidth requirement per device is thus 18kbps, and since communication is bidirectional, hence 36 kbps data transfer per call.

The maximum speed of data transfer in the TWI module is 400 kbps. Thus, ideally there should be (400/36) = 11 bidirectional calls possible. However, practically we found that the maximum number of calls connected was limited to 10. This was probably because each data transfer consumes some time in initialization processes.

3. PREDICTOR ALGORITHM FOR GENERATING 0-255 QUANTIZED LEVELS FROM 0-82 LEVELS OF RECEIVED DATA

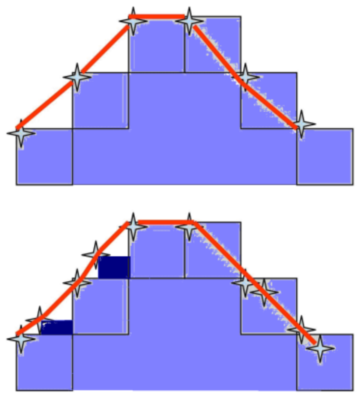


Fig 4 (a) & (b) (note the greater number of estimated samples in (b))

The following steps were implemented in this algorithm:

- Multiply the received value (0-82) by 3 so that the range now becomes 0-246. However, this is mere amplification of the signal and the required smoothening effect is still not achieved in the output waveform. For example, the sequence (...12,13,14,...) will merely convert to (...36,39,42,...) after multiplication.

- To achieve the required smoothening, we use the previous values to estimate the rise/fall or positive/negative slope. Then we add/subtract one value (ex : 39 changes to 39,40 since +ve slope)

Recvd Data Seq	Mult By 3	Slope of signal.	Incremented next/previous value	Effective estimated sequence
12	36			39(Recived) 40(Estimated) 42(Recived) 43(Estimated)
13	39	+ve	39+1=40	
14	42	+ve	42+1=43	
. . .				42(Recived) 41(Estimated) 39(Recived) 38(Estimate d
15	45			
14	42	-ve	42-1=41	
13	39	-ve	39-1=38	

4. TESTING METHODOLOGY

Following setup was implemented to test the reliability analysis of the system, especially to judge the performance of the predictor algorithm.

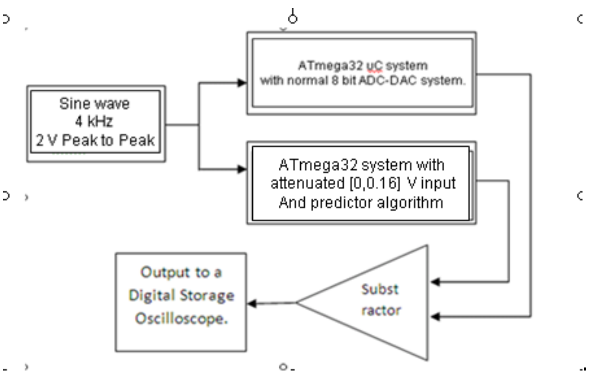


Fig 5

A 4 kHz 2 V peak-peak sine wave was taken as the test signal, representing the human voice signal. This signal was simultaneously given to two setups. The first setup consisting 8 bit ADC and simply followed by 8 bit DAC (suitable delay added equal to the computation time of second system). The second system consisted of attenuated input where [0,2] V was scaled down to [0,0.16] V using potential divider, and this scaled down version was given as the input to the microcontroller system with the previously explained predictor algorithm.

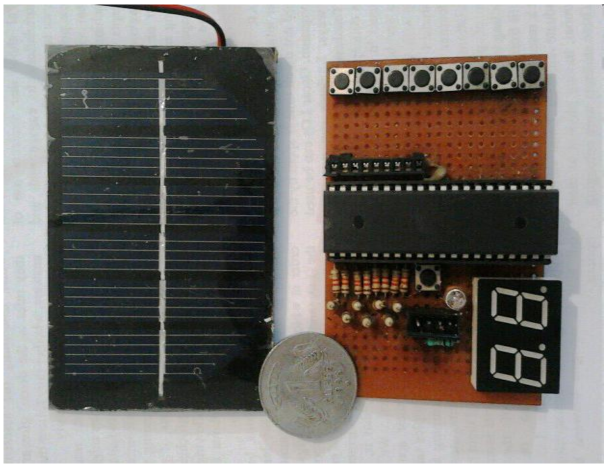
The output of both these systems were given to a opamp subtractor circuit, and the difference of the two signals was seen on the DSO. It was observed that the average error voltage on DSO was 0.13 V, the peak value was 0.21 V. These are 6.5 % and 10.5% respectively, which is not a huge effort.

5. POWER SUPPLY OF THE CIRCUIT

The circuit works at 8 mA in stand by mode, when no call is being made. During a voice call, the current requirement goes up to a peak value of 120 mA, however it is seen that in ordinary conversation this value remains around an average of 36 mA. The current consumption of the circuit is mainly due to the voice output, all other functionalities consume not more than a few mA current taken together. These low values strongly favor the use of solar powered circuit if needed

6. RESULT

The required functionality of the circuit is accomplished with very tight constrains, which is the biggest accomplishment. The telecommunication network is surely a stand alone system with advantages surpassing the limitations.



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