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A Complete Design of Advanced embedded chatter box for physically challenging persons

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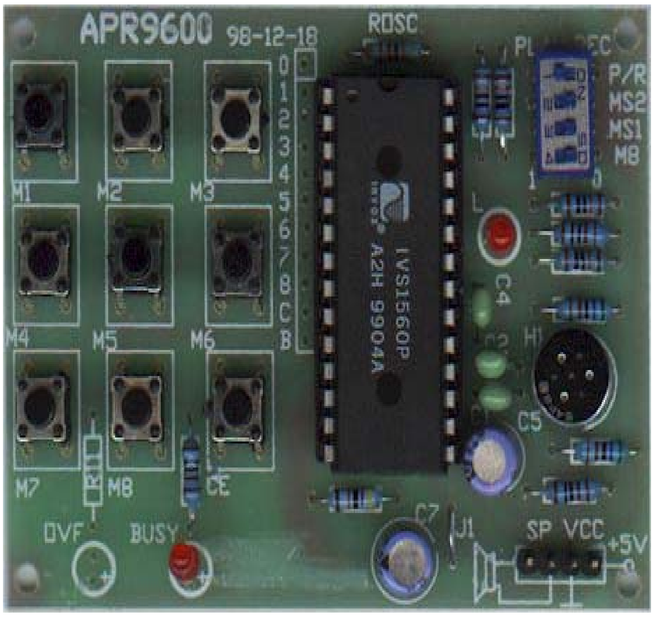
Abstract- In this world, deaf and dumb people are present thought that can't we help physically challenging persons(Deaf, Dumb) by any simple device by using our technology with low cost and easy usage to give a alternative way to talk and hear for them brought me to do this paper. . In this paper we are going to propose a system which shows, how to build a speaking device for physically challenging persons. A novel concept of this device is how far useful for the people who are challenged with speaking and/or Hearing. Its purpose is to aid communication for the physically challenged persons. This will include basic 16 x 2 characters LCD – black on green 3.3V. Nonspeaking Individuals with problems with their fine motor skills can also use text-to speech to aid their abilities which include people who suffer from:

- **. ALS (Lou Gehrig's disease**
- **Traumatic Brain injury**
- **Laryngectomy**
- **First stroke patients. Also**

1. INTRODUCTION:

After seeing many deaf and dumb people around me a thought that can't we help physically challenging persons(Deaf, Dumb) by any simple device by using our technology with low cost and easy usage to give a alternative way to talk and hear for them brought me to do this paper. This paper details how to build a device which can speak out the text given to it by a speaker and helpful for dumb people as a mouth to them and its LCD display acts as a ear to the deaf people by making them to understand what the other person saying them. In this proposed device APR9600IC plays the lead role for the functionality of the device and here we describes of this IC, APR9600 is a low-cost high performance sound record/replay IC incorporating flash analogue storage technique. Recorded sound is retained even after power supply is removed from the module. The replayed sound exhibits high quality with a low noise level. Sampling rate for a 60 sec recording period is 4.2 kHz that gives a sound record/replay bandwidth of 20Hz to 2.1 kHz. However, by changing an oscillation resistor, a sampling rate as high as 8.0 kHz can be achieved. This shortens the total length of sound recording to 32 seconds. Total sound recording time can be varied from 32 seconds to 60 seconds by changing the value of a single resistor. The IC can operate in one of two modes: serial mode and parallel mode. In serial access mode, sound can be recorded in 256 sections. In parallel access mode, sound can be recorded in 2, 4 or 8 sections. The IC can be controlled simply using push button keys. It is also possible to control the IC using external digital circuitry such as micro-controllers and computers. The APR9600 has a 28 pin DIP package. Supply voltage is between 4.5V to 6.5V. During recording and replaying, current consumption is 25 mA. In idle mode, the current drops to 1 mA. The APR9600 experimental board is an assembled PCB board consisting of an APR9600 IC, an electrets microphone, support components and necessary switches to allow users to explore all functions of the APR9600 chip. The oscillation resistor is chosen so that the total recording period is 60 sec's with a sampling rate of 4.2 kHz. The board measures 80mm by 55mm.



APR9600 board

Here paper is classified into 3 folds, where the first part describes about APR9600 operation and device construction , second fold describer's how the program operation done, Finally we draw to a conclusion how this embedded device is helpful for Physically challenged persons.

2. SYSTEM DESCRIPTION:

Pin-out of the APR9600 is given in Figure 1. A typical connection of the chip is given in Figure 2 (This is the circuit diagram of the module). Pin functions of the IC are given in Table 1. During sound recording, sound is picked up by the microphone. A microphone pre-amplifier amplifies the voltage signal from the microphone. An AGC circuit is included in the pre-amplifier, the extent of which is controlled by an external capacitor and resistor. If the voltage level of a sound signal is around 100 mV peak to-peak, the signal can be fed directly into the IC through ANA IN pin (pin 20). The sound signal passes through a filter and a sampling and hold circuit. The analogue voltage is then written into non-volatile flash analogue RAMs.

It has a 28 pin DIP package. Supply voltage is between 4.5V to 6.5V. During recording and replaying, current consumption is 25 mA. In idle mode, the current drops to 1 mA

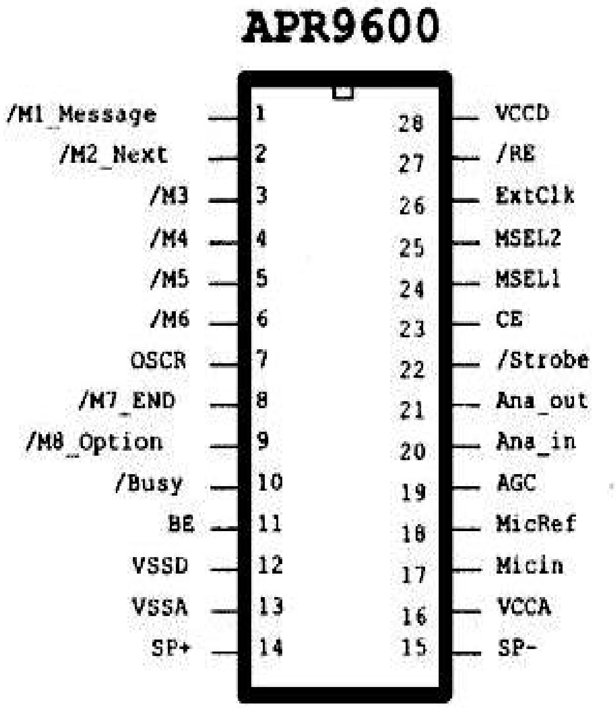


Figure 1 Pin-out of APR9600

Pin	Name	Functions	Pin	Mane	Functions
1	-M1	Select 1st section of sound or serial mode recording and replaying control (low active)	15	SP-	Speaker, negative end
2	-M2	Select 2nd section or fast forward control in serial mode (low active)	16	VCCA	Analogue circuit power supply
3	-M3	Select 3rd section of sound	17	MICIN	Microphone input (electret type microphone)
4	-M4	Select 4th section of sound	18	MICREF	Microphone reference input
5	-M5	Select 5th section of sound	19	AGC	AGC control
6	-M6	Select 6th section of sound	20	ANA-IN	Audio input (accept a signal of 100 mV p-to-p)
7	OSCR	Resistor to set clock frequency. See Table 3 for details	21	ANA-OUT	Audio output from the microphone amplifier
8	-M7	Select 7th section of sound or IC overflow indication	22	STROBE	During recording and replaying, it produces a strobe signal
9	-M8	Select 8th section of sound or select mode (see Table 2)	23	CE	Reset sound track counter to zero/ Stop or Start / Stop
10	-BUSY	Busy (low active)	24	MSEL1	Mode selection 1 (see Table 2)
11	BE	=1, beep when a key is pressed =0, do not beep	25	MSEL2	Mode selection 2 (see Table 2)
12	VSSD	Digital circuit ground	26	EXTCLK	External clock input
13	VSSA	Analogue circuit ground	27	-RE	=0 to record, =1 to replay
14	SP+	Speaker, positive end	28	VCCD	Digital circuit power supply

Table 1 Pin-functions of the APR9600

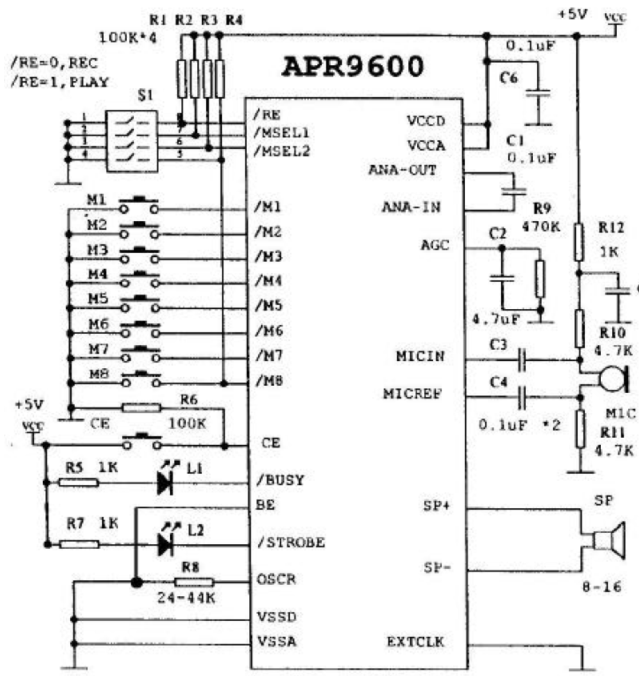
During sound replaying, the IC's control circuit reads analogue data from flash RAMs. The signal then passes through a low-pass filter, a power amplifier and output to an 8 to 16 Ohm speaker. There are different sounds recording and replaying modes (see Table 2). These modes are selected using MSEL1 (Pin 24), MSEL2 (Pin 25) and –M8 (Pin 9). –M1 to –M7 keys have different functions in different modes.

MSEL1	MSEL2	-M8	Function Keys	Functions
0	1	0 or 1	-M1, -M2 to select 1st and 2nd sound tracks. CE to stop	Parallel mode, 2 sections, 30 seconds for each
1	0	0 or 1	-M1 to –M4 to select a sound track, CE to stop	Parallel mode, 4 sections, 15 seconds for each
1	1	1	-M1 to –M8 to select a sound track, CE to stop	Parallel mode, 8 sections, 7.5 seconds for each
1	1	1	-M1 to –M8 to select a sound track, CE to stop	Pressing and hold down a key from –M1 to M8 to play the selected sound track repeatedly
0	0	1	-M1 and CE	Serial mode, allow up to 256 sound tracks to be recorded and played. Sound tracks are played from 1st to N in order after –M1 is toggled. Press CE to play from the 1st sound track.
0	0	0	-M1,-M2 and CE	Serial mode, Press –M1 to replay one sound track. Toggle –M2 once to move to the next sound track. Press CE to play sound from the 1st sound track

Table 2 Modes and selection of modes

Notes:

- RE=0 to record sound. RE=1 to replay sound
- Press -M1 to -M8 once to replay a sound track. Press the key again to stop replaying the track
- Press and hold –M1 to -M8 continuously, the corresponding track will be replayed repeatedly
- During recording, -M1 to M8 should be pressed while the sound is being recorded. Releasing the key terminates recording.



2.1 APR9600 module

The circuit diagram of the module is shown in Figure 2. The module consists of an APR9600 chip, an electrets microphone, support components, a mode selection switch (-RE, MSEL1, MSEL2 and – M8) and 9 keys (-M1 to –M8 and CE). The oscillation resistor is chosen so that the total recording period is 60 seconds with a sampling rate of 4.2 kHz. Users can change the value of the ROSC to obtain other sampling frequencies. It should be noted that if the sampling rate is increased, the length of recording time is decreased. Table 3 gives the details. An 8-16 Ohm speaker is to be used with the module. Users can select different modes using the mode selection switch. The module is measured 80mm´55mm. Connection points (0-8, C and B) can connect to other switches or external digital circuits. In this cased, on-board keys M1 to M8 and CE are by-passed.

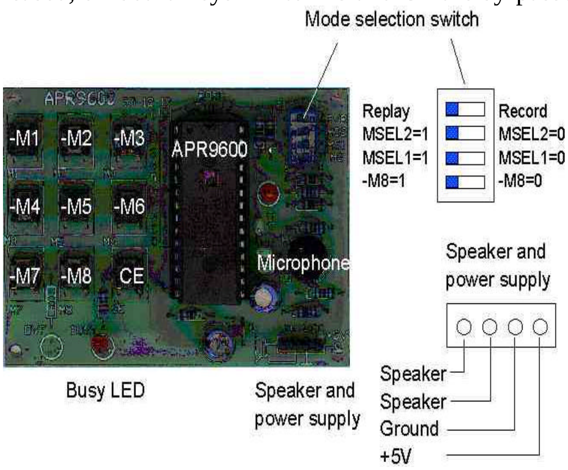


Figure 3 APR9600 module with connector details (to record in parallel mode, the switch setting should be same as displayed above. To record, the top switch should be on the right-hand side. To replay, the top switch should be on the left)

3. USING THE APR9600 MODULE

3.1 Parallel mode recording and replaying

Record sound tracks

This is an example of recording 8 sound tracks. The mode switch should have the following pattern:

MSEL1=1(switched to left-hand side of the mode selection switch), MSEL2=1 (left-hand side). –M8=1 (left-hand side). RE=0 (right-hand side). The maximum length of the 8 tracks is 7.5 seconds. Press –M1 continuously and you will see BUZY LED illuminates. You can now speak to the microphone. Recording will terminate if –M1 is released or if the recording time exceeds 7.5 seconds. Similarly, press –M2 to -M8 to record other sound tracks.

Replay sound tracks

Now make RE=1 (switched to Left-hand side of the mode selection switch) while keep other switches at the same location. Toggle –M1 to –M8 (press key and release) causes a particular sound track to replay once. While the sound is playing, press the same key again or press CE key will terminate the current sound track. Press other key while a sound is being played causes a new sound track to be played. If a key from –M1 to -M8 is pressed continuously, the particular sound track will be played continuously. Press CE to stop playing the sound track.

3.2 Serial mode recording and replaying

Record sound tracks sequentially

This is an example of recording sequential sound tracks. The mode switch should have the following pattern:

MSEL1=0(switched to right-hand side of the mode selection switch), MSEL2=0 (right-hand side). M8=1 (left-hand side). RE=0 (right-hand side). Press CE first to reset the sound track counter to zero. Press and hold –M1 down and you will see BUZY LED illuminates. You can now speak to the microphone. Recording will terminate if –M1 is released or if the recording time exceeds 60 seconds (in this case you will run out the memory for your next sound track). Press –M1 again and again to record 2nd, 3rd , 4th and other consecutive sound tracks. Each sound track may have different lengths, but the accumulated length of all sound tracks will not exceed 60 seconds.

Replay sound tracks sequentially Now make RE=1 (switched to Left-hand side of the mode selection switch) while keep other switches at the same location. Toggle –M1 (press key and release) causes the 1st sound track to be Played once. Toggle –M1 again and again will play the 2nd, 3rd, 4th and other consecutive sound tracks. Press CE to reset the sound track counter to zero.

Record sound tracks with forward control

This is an example of recording sound tracks with forward control. The mode switch should have the following pattern: MSEL1=0(switched to right-hand side of the mode selection switch), MSEL2=0 (right-hand side). –M8=0 (right-hand side). RE=0 (right-hand side). Press CE first to reset the sound track counter to zero. This mode is rather similar to the above sequential sound recording. The only difference is that after –M1 is pressed and released, the sound track counter does not increment itself to the next sound track location. To move to the next sound track, –M2 should be toggled. So if –M1 is not toggled again and again without toggling –M2, sound will be recorded at the same sound track location.

Replay sound tracks with forward control

Now make RE=1 (switched to Left-hand side of the mode selection switch) while keep other switches at the same location. Toggle –M1 (press key and release) causes the 1st sound track to be played once. Toggle –M1 again and again will still play the 1st sound track. Once –M2 is toggled, the sound track counter is incremented and the next sound can be played. Press CE to reset the sound track counter to zero.

3.3 Sampling rates

The sampling rate is determined by the value of the OSC resistor (R8 in the circuit diagram). It can be adjusted by users to suit their specific requirements. The relationships amongst the resistance, sampling rate, bandwidth and recording period are shown in Table 3.

OSCR resistance [kOhm]	Sampling rate [kHz]	Bandwidth [kHz]	Recording time [Second]
44	4.2	2.1	60
38	6.4	3.2	40
24	8.0	4.0	32

Table 3 OSC resistances

3.4 Application tips

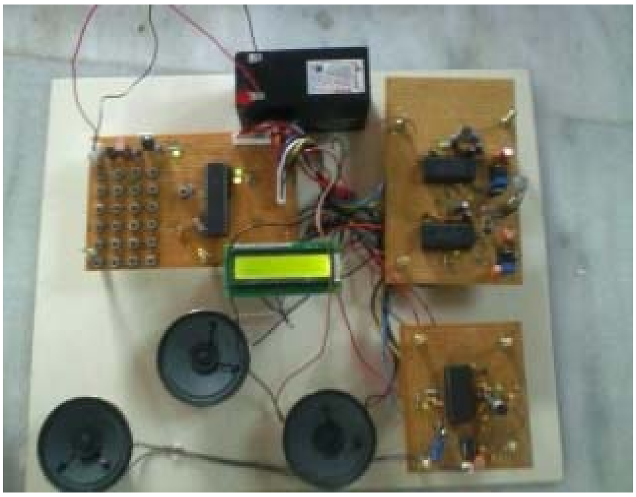
Tips for better sound replay quality:

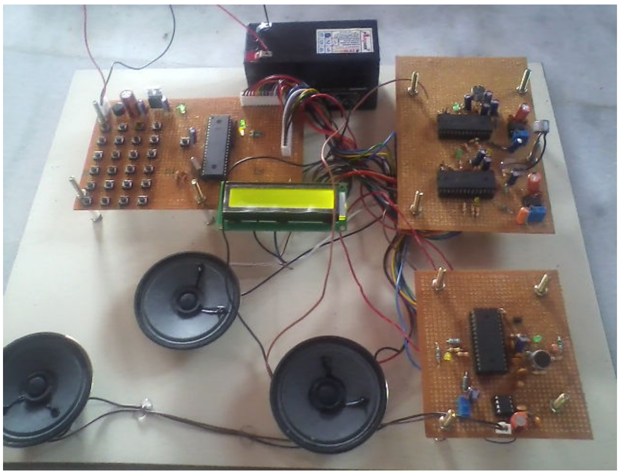
1. Use a good quality 8 Ohm speaker with a cavity such as speakers for computer sound systems. Do not use a bare speaker which gives you degraded sound.
2. For better sound replay quality, speak with a distance to the on-board microphone and speak clearly. Also keep the background noise as low as possible.
3. For even better sound replay quality, use microphone input or Audio Line In input. If Audio Line In is used; the amplitude of input signal should be < 100 mV p-p.

4.SOFTWARE CODEING:

```
#include "stdio.h"
#include "AT89x52.h"

#define RS P0_0
#define EN P0_1

sfr  DATA =       0x80;

void Delay(unsigned int time)
{
        while(time--);
}

void LCD_DAT(unsigned char dat)
{
        DATA=(dat&0xF0);
        RS=1;
        EN=1;
        Delay(0xFF);
        EN=0;

        DATA=(dat&0x0F)<<4;
        RS=1;
        EN=1;
        Delay(0xFF);
        EN=0;

        Delay(0xFF);
}

void LCD_CMD(unsigned char cmd)
{
        DATA=(cmd&0xF0);
        RS=0;
        EN=1;
        Delay(0xFF);
        EN=0;

        DATA=(cmd&0x0F)<<4;
        RS=0;
        EN=1;
        Delay(0xFF);
        EN=0;

        Delay(0xFF);
}

void LCD_Goto(unsigned char row,unsigned char col)
{
        if(row==1) LCD_CMD(0x80+col);
        else if(row==2) LCD_CMD(0xc0+col);
        Delay(0x550);
}

void LCD_CLR(void)
{
        LCD_CMD(0X01);
        Delay(0xFFF);
}

void LCD_INIT(void)
{
        LCD_CMD(0X28);
        Delay(0xFFF);
        LCD_CMD(0X28);
        Delay(0xFFF);
        LCD_CMD(0X0E);
        Delay(0xFFF);
        LCD_CMD(0X06);
        Delay(0xFFF);
        LCD_CLR();
        Delay(0xFFF);
}

void LCD_Display(unsigned char *ptr,unsigned char
row,unsigned char col)
{
        if(row==1) LCD_CMD(0x01);
        while(*ptr!='\0')
        {
                LCD_Goto(row,col);
                LCD_DAT(*ptr);
                ptr++; col++;
        }
}


void main(void)
{

        unsigned char i = 0;
        P0=P1=P3=P2=0xFF;

        LCD_INIT();
        Delay(0xFFFF);
        LCD_Display("ENTER THE TEXT",1,1);
        Delay(0xFFFF);
        while(1)
        {
          if(i>15) i=0;
                if(P1_0==0)
                {
                        LCD_Goto(2,i);
                LCD_DAT('A');
                        i++;
                        Delay(0xFFFF);
                }
                if(P1_1==0)
                {
                        LCD_Goto(2,i);

        |||||Similary rest of the alphabets|||||

        LCD_DAT('Y');
                        i++;
                        Delay(0xFFFF);
                }

        }
}
```

5. Finally how the device looks:





Conclusions & Further Recommendations

Thus advanced embedded chatter box for physically challenging persons has been done and their data sheets corresponding to the chatter box, i.e. mainly controller, and related software Kiel is been verified with a demo program with interfacing modules as led's, and manly speak jet ic which acts like main interfacing component for the physically challenged to text to speech converter, hardware implementation has been done for verifying its functional properties. Finally it may be one of the useful devices for deaf people to understand what the others are saying by LCD display and dumb people can speak with the help of this device. In future an Ic which may be inbuilt with all speaking features can be used instead of APR9600 ic and may be more useful for physically challenging persons.

Acknowledgements

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Short Biography

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