Voice Activated Wireless Control of Devices in a Room Environment

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User Manual

Chapter 1

User Manual

1.1 Introduction

This device will enable you to wirelessly control the switching of various devices in the room environment, merely using the spoken command.It is aimed at the handicapped user who cannot push buttons on a remote but also has universal appeal as an easy and comfortable way to switch devices,without interrupting the work at hand.

This manual gives the information you would need to use the device or to service it. Please read the manual carefully before operating your unit.

1.2 Getting started

1.2.1 Initial check list

Your voice activated wireless controller should contain the following units —

- Word recognizer unit – with LED display and mic.
- Transmitter unit – with 4 wire connector to connect to word recognizer.
- Receiver – to be mounted on the switch board.
- 9V alkaline battery for receiver.

All the units are preconnected, so no assembly is required on your part. Please attach the two 9 volt batteries in the LED display unit and the switch-board unit in the space provided. You may use a multimeter to verify that each battery has a voltage of at 9V. Also, you may switch on the main switches and check if the power on LED turns on. Place the transmitter section at a suitable location (such as on the wheelchair where the LEDs are visible).

1.2.2 Specifications

Word Recognizer

The recognition system is capable of recognizing upto 10 isolated distinct words, spoken clearly and at normal speed. In the device, the number of words is set at

5 words. Thus 5 devices can be controlled using the system. The recognition is speaker dependent, and each user has to undergo a training sequence initially to store the templates of his/her speech. The system is restricted to single user as multiple user templates are not stored. There is a delay of about 5 secs between speaking of the word and its recognition or rejection.

FM Transceiver

The transceiver has been designed to work below the commercial FM frequency band. This avoids interference and minimizes the chances of jamming and hence can work in an electrically noisy environment. The transmission of device code is done using DTMF tones. It takes about a second to transmit the code, after the spoken word is recognized correctly. The receiver applies error-checks on the received code and in case of error, it cancels the operation, i.e switching is not performed. In such case, the command has to be given again. If correct device code is received, the corresponding device is toggled. For reliable reception, the distance between transmitter and receiver should not be more than 50 meters.

Front Panel

The front panel consists of two red power LEDs in the lower right corner. The main display LEDs include three green word LEDs and one red state LED. The green LEDs denote binary coded integer between 1 to 5 which indicates the index number corresponding to the recognized word. The red state LED denotes various states like detection of start of word, mismatch in recognition. There is also one red RESET LED.

The system also has array of 8 DIP switches which can be used for configuring the system for initial use or in case of some problems.

1.3 Initial tests

After switching on the device, the both the power LEDs should light up. It should be followed blinking of RESET LED indicating that the system has started execution. The four display LEDs should flash 3 times indicating that the system initializations are over and system is ready for input.

In a blank system, the training is required before it can be used. Thus the system automatically goes into training mode. This is indicated by flashing of state LED and binary code 1 on the word LEDs. This denotes that user should start training from word no–1. For detailed operation of the training sequence, refer to the next section.

To test whether the system is working smoothly, user can speak any word into the mic. The state LED should light up indicating start of word. But as the training is not done, the word spoken will be taken as the first word to be trained. Please refer to training section for interpreting the response of the system.

NOTE: *Speaking into the system in training mode might cause inadvertent storage of the word. Thus speaking frequently into the blank system should be avoided. The user must carry out proper training sequence.*[1]

[1] if incorrect template gets stored in the system, it can be erased completely using the DIP

The accuracy and range of the RF link can be tested using the *RF Link Test* which checks whether the codes are being transmitted properly over the link.

1.4 Operation

There are two basic modes of operation of the system –

- Training
- Recognition

The basic steps involved in the two operations are described below.

Training

To ensure accurate recognition, the training is essential for any new user intending to use the device. The training sequence involves the storage of the user's template in the system memory. The system stores template for only one user. Thus device cannot be used reliably by more than one users simultaneously. The template remains stored even after power-off.

On power-on, the system fetches the templates stored in it. If the number of words stored in the template is less than 5, the system goes into training mode from next word onwards. This is indicated by flashing of state LED and word number on the word LEDs. The system can be forced to go to training mode by switching on the DIP switch-1 before power-on. In this case, the system erases the previously stored template and starts with word no-1 same as in initial blank state. This is indicated by binary code 1 flashed along with the state LED.

The training sequence goes as follows —

1. Pronounce the device name to be switched clearly, in your normal voice at normal speed. The system will flash the current word number as an acknowledgment that it has captured the features of the spoken word.

2. Pronounce the same word again clearly. The system checks whether the two pronunciations of the same word are close enough. On close enough match, the system will store the template of the word and flash the word number twice indicating that the training for the word has been successful. If system doesn't find acceptable match betwen the two instances, it will indicate an error by flashing the state LED. The system will go back to the same word number and the steps 1 and 2 will have to be repeated for that word.

3. On success, the system goes into training of next word.

4. The steps 1-3 repeat until all the 5 words have been successfully recorded.

5. The system will automatically switch to recognition mode.

NOTE: *In case of training initiated using DIP switch, turn the switch off before next power on; otherwise the recorded template will be destroyed.*

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switches provided. (Refer to the section on training for using DIP switches)

Recognition

This mode is valid only when system has undergone full training. In this mode, the system continuously samples the audio signal and detects the start of the word. This is indicated by blinking of state LED. On capturing the word, it extracts the parameters and compares them with the previously stored template. On identifying the match, the word number flashed on the word LEDs and the transmission of corresponding device code begins. The word LEDs remain unchanged till the transmission continues. On detecting the mismatch, the red state LED is flashed.

The recognition mode continues till the system is forced into training mode using DIP switches.

Note: *This product must always be connected to the power supply. It is activated only when the command is spoken and is in the idle or waiting state at other times.*

1.5 Safety precautions

1. Do not speak into mike when you do not wish to operate the device because it might respond to words similar to the ones specified and lead to unwanted switching.
2. Do not touch the inside components on the printed circuit board.
3. If attached to the wheelchair, make sure that the assembly is firm.
4. This product has been designed for indoor use only.

1.6 Detailed Specifications

Word Recognizer

- Supply Voltage - 9V DC
- Supply current - 150mA when idle and 300mA when in operation.
- Optimum Mike distance - 2 inches
- Mike - electret
- Voice sampling rate - 8KHz
- Maximum word length - 0.5 seconds
- Analysis time - 5 seconds

FM Transceiver

- Max bit rate: DTMF chip can switch at a maximum rate of 20Hz resulting in a gap of about one second between code generation and reception.
- Range: Tested on a 50 feet link. The transmitter design is reckoned to have a range of several hundred meters, but the signal strength and signal - noise ratio may deteriorate over this range.

- Error rate: Depends on many factors: distance, proximity to other radio stations etc. the latter has been removed by going below the FM bandwidth range. Over 20 meters, the error rate was virtually zero.

Part II

Theory of Operation & System Design

Chapter 2

Theory of operation

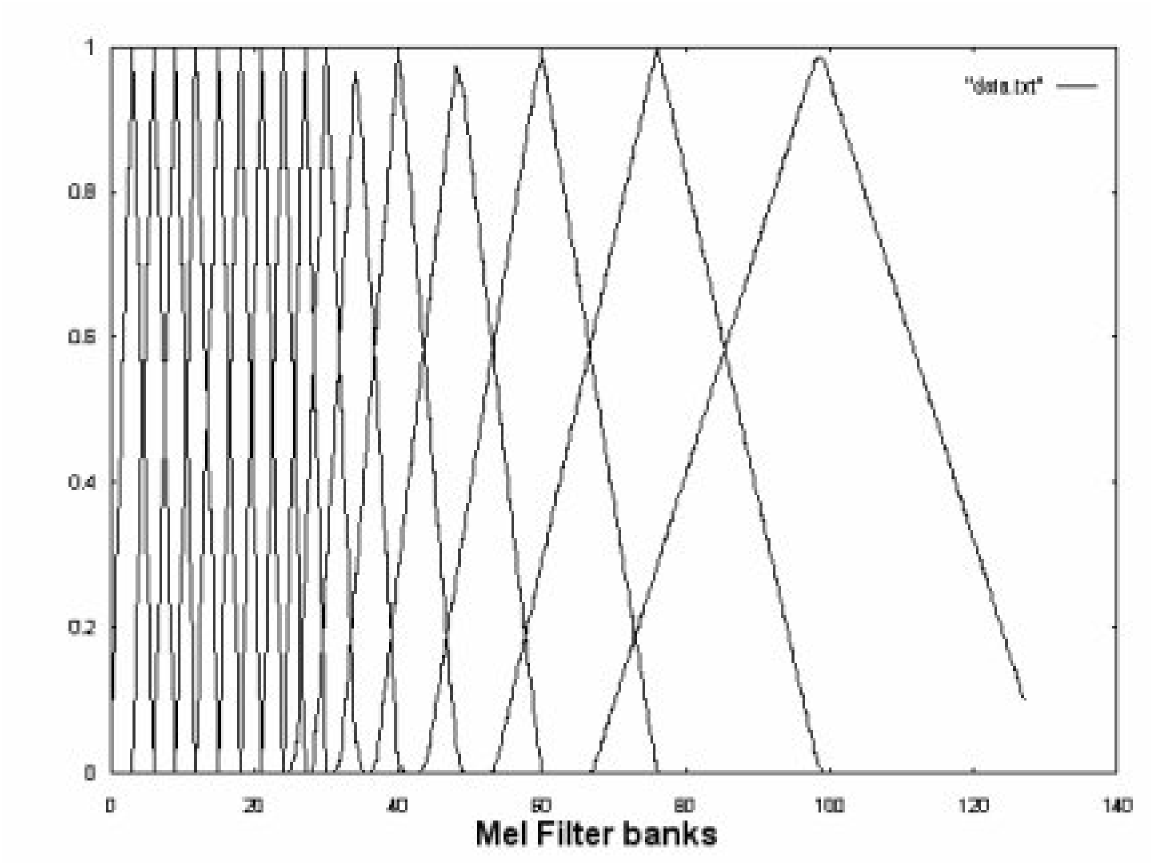
This chapter gives the detailed description of the operation of the system. The chapter is divided as follows. Section 2.1 gives the theoretical background and the block diagram of the system. Section 2.2 gives the details of the hardware utilized. Section 2.3 gives the details of the firmware that went into the running the hardware with required FSMs. Section 2.4 gives the details of the tests done during integration of the system.

2.1 System description

2.1.1 Word Recognition

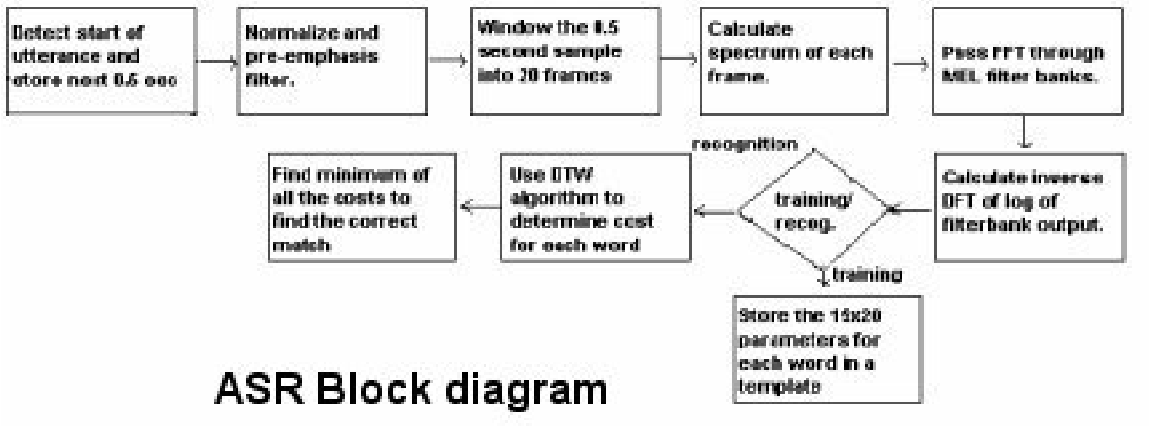
The word recognition algorithm is based on Mel Cepstral coefficients of the spoken isolated word. The first step is sampling the mic input and detecting the start of the word. The sampling is carried out at 8 ksps using 10-bit ADC. This sampling rate is sufficient for speech signal which has the bandwidth of about 4 KHz. The start detection is done by comparing the signal energy and zero-crossings with the background noise thresholds. The noise thresholds are calculated at system startup. The noise thresholds are checked for their correctness against the expected range of values. This ensures that the calculation faithfully represents only the background noise and not the actual signal. Once the start of word has been detected, the system starts storing the incoming signal as spoken word. The end of the word should also be detected. but this involves quite elaborate techniques as compared to the start detect and it often gives incorrect results e.g - if a word has a silent portion then end detect may detect it as end of word thus losing the rest of the word. On the other hand, a simplistic model stores the speech for fixed amount of time after start detect. We chose the latter model with word length fixed at 0.5 secs. This gives optimum combination of feature extraction distinction, system memory resources, and time for analysis.

The captured speech undergoes various pre-conditioning operations like preemphasis filter, framing, applying hamming windows. These operations are executed real time as the samples are being stored. After entire 0.5s duration word has been stored, it is divided into 20 frames of 256 samples each, with an overlap of 56 samples. The FFT algorithm is applied on each frame. Feature extraction is done using Mel Frequency Cepstral coefficients (MFCC). The



MFCC coefficients are those extracted from a triangular scaled filter bank. This filter bank tries to simulate the human ear frequency response and the process is called perceptual weighting. The Mel filter banks are applied to the FFT of the signal. Cepstral smoothening is done by taking Log of the output of the filter bank. The Mel Cepstrum coefficients are then extracted by taking IDCT (Inverse Discrete Cosine Transform). In our case, for real signals, and if we consider only the real part of the IDCT, the IDCT operation is equivalent to taking a IFFT. There are 15 coefficients per frame and thus 300 coefficients for a single word. During the training sequence, these coefficients form the stored templates. During recognition, the extracted coefficients are compared with each of the words in the stored template. The distance between words is calculated using the DTW (Dynamic Time Warping) algorithm. The minimum distance is compared against experimentally determined threshold value. Minimum distance less than threshold qualifies a match else mismatch is flagged. In training mode, the user has to speak the same word twice. The distance between coefficients of the two instances is calculated similar to recognition. The word is accepted only if the two instances match within the threshold. The average of the coefficients of the two instances is stored in the template.

The algorithms are implemented on a TI LF2407A DSP. The details of hardware and firmware are described in subsequent sections.



2.1.2 Transmission Protocol

The device code transmitted using the FM transceiver is 8 bit code having the following format:

3 bit switchboard id	3 bit device id	Reserved (on/off)	1 bit parity

The switch board id uniquely identifies one of possible 8 switch boards in the room. The device id identifies the particular device among possible 8 devices on the selected switchboard. Currently the switching takes place in toggle mode thus the on/off bit is kept reserved. The last parity bit makes number of 1's in the code odd. Thus it is used at the receiver end for error checking.

The codes we are currently using are:

Device	Switchboard ID	Device ID	on/parity	off/parity
Device 1:light	010	001	10	01
Device 2:fan	010	010	10	01
Device 3:tube	010	011	11	00
Device 4:TV	010	100	10	01
Device 5:AC	010	101	11	00

Note: *As switching is in toggle mode, the value of the on/off bit has no effect. Thus any one of the two 2-bit codes in the last two columns can be used.*

The 8 bit code is transmitted serially using DTMF tone signalling with a unique start and stop tones. Thus one communication cycle contains 10 tones interleaved with null tones.

2.1.3 Alternative designs considered

RF was preferred over IR due to the line of sight problems associated with the latter. This enables the user to use the product from any position in the room. Other modulation techniques such as QPSK were also considered but FM was finally chosen due to its ease of implementation and compatibility with the readily implementable tone signaling.

In word recognition, the features are extracted for the whole word. A sophisticated technique involves detection and recognition of individual phonemes of the word. This is essential for improving the resolution between similar sounding words. But this involves unrealizably high complexity in terms of time, space and coding effort.

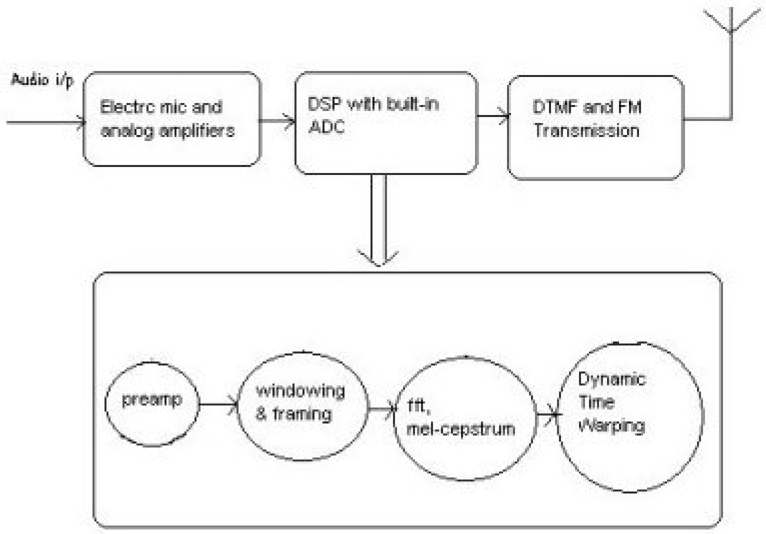


Figure 2.1: System Block Diagram

2.1.4 System block diagram

The system block diagram is shown in figure 2.1.

2.2 Hardware

2.2.1 DSP Board

The DSP board is designed around Texas Instruments (TI) TMS320LF2407A DSP. It is the member of the C2000 family of DSP controllers. The prominent features of LF2407A include:

1. 40 MIPS performance at clock speed of 40MHz.
2. Low power 3.3V operation.
3. Havard architecture with two separate internal busses for data and program for high speed operation.
4. 32K on chip FLASH memory for program, 2.5K on chip fash access data RAM.
5. External memory interface (EMIF) with separate program space, data space and IO space - each with 64K addressing capability.
6. 32 bit ALU
7. TI C2xx compatible Instruction set featuring several specialized instructions and addressing modes optimized for implementing standard DSP algorithms like filtering, FFT.

These specifications make LF2407A the most advanced processor among C24x family.

Power Supply

The DSP requires 3.3V for its operation and 5V for internal flash programming. The two voltages are generated from DC 9V input using TI TPS76833 and TPS76850 low drop-out voltage regulator chips respectively. The TI supervisor chip TPS3305-33 monitors both the supply lines and triggers the reset if the voltage drops below 98%of its normal value.

On board Memory

The onboard memory consists of 64K x 16 bit, 10ns Cypress CY7C1021CV33-10 SRAM and 64K x 16bit, 150ns Atmel AT29LV1024-15 parallel FLASH. The upper 32K words of the SRAM are mapped into external data space. The lower 32K of SRAM can be used in lower 32K program space if the DSP is in microprocessor mode. This is useful during program development and debugging as the program can be loaded and run from SRAM instead of burning it in internal FLASH. However in microcontroller mode, the lower 32K of SRAM is left unutilized.

The external flash is mapped into the upper 32K of external program space. Thus it can be used for storing program if the code size exceeds 32K. It can also be used to store permanantly any data in the form of tables which are not accessed very frequently. The problems in accessing flash are - it is quite slower (150ns) compared to CPU clock speed (25ns) and it is programmable sectorwise with sector size of 128 words. The required read, write algorithms are implemented as the part of firmware.

GPIO — General Purpose Input Output

The board has

1. 6 LEDs for indicating various states and jumper configurations.
2. 10 general output LEDs controlled by 10 GPIO pins (IOPF0-5, IOPA6-7, IOPC6-7) of the DSP.
3. 8 bit general purpose output port controlled by port B (IOPB0-7) of the DSP.
4. 8 bit general purpose input port at port D (IOPD0) and port E (IOPE1-7) of the DSP.

The LEDs are driven using ULN2803 Darlington pairs array. The input/output ports are 5V compatible. The TI Level Shifter chip SN74ALVC164245 is used to convert 3.3V level of DSP to 5V level and vice versa.

Audio Codec

The board has TI TLV320AIC320 DSP codec for the mic and speaker interface. The codec has 16 bit sigma delta ADC and DAC, mic preamplifier, antialiasing filter, decimation filter, interpolation filter. It supports glueless serial interface to the TI DSPs. For LF2407A, the interface is through SPI (Serial Peripheral Interface) module. The codec is suitable for audio applications and has maximum sampling rate of 22ksps.

Analog Part

Analog circuit consists of an microphone preamplifier. It is the cascade of two inverting amplifiers using OP27 opamps. OP27 is chosen for its better noise performance. The each of the stages gives gain of about 10 thus making the maximum total gain of 100. Gain can be adjusted by changing the potentiometer. The optimum gain is adjusted such that the normal speech signal gets amplified to full scale voltage but without clipping. The opamp circuit is designed for using unipolar 5V supply available on the board using 7805 voltage regulator. The output is obtained unipolar between 1.2-3.3V. It is biased at 2.5V. The output of the preamplifier is connected to the channel 0 of the internal ADC of LF2407A. Zenner protection diode is used at the output to avoid damage to the ADC in case of output rising above 3.3V.

JTAG

The board also has **JTAG** port for real time Scan-Based Emulation, IEEE Standard 1149.1 which can be used for in-system loading and debugging of programs. The board also has provision for RS232 interface using Maxim MAX3221 chip. It can also be used for program loading after enabling the boot ROM inside the DSP.

2.2.2 FM Transceiver

The wireless link of the product is designed modularly. The functional requirement of the wireless link is that it should be able to reliably transmit and receive the codes corresponding to the different devices to be switched, and switch the appropriate line high. This wireless transmitter takes in the device code of the device to be switched on, from the word recognizer (front end) and transmits it over the channel. The receiver then receives the code, decodes it and switches on the appropriate line. The modulation technique used is frequency modulation (FM).

The transmitter side is divided into two modules, one to obtain device codes and convert them into appropriate format for transmission over the link and the other to actually perform the transmission at high frequency.

Similarly the receiver side has two modules, one to receive and demodulate the high frequency signal and the other to identify the device from the signals obtained.

Transmitter side

Encoder:

This project uses tone signaling for transmitting bits. We use a DTMF (Dual Tone Multiple Frequency) tone generator for the purpose. Two tones are set aside for the two bits 0 and 1, and two tones for the start and stop bits. Some tones are used for checking the RF link. Once the spoken command has been given and the microcontroller at the front end has extracted the 8-bit binary device code, this device code is serially transmitted over the link. The codes are designated as follows:

Each time a code has to be transmitted, a 8-bit combination from the first 6 columns above and either of the last two columns above, as appropriate, is selected.

The 8 bit code is sent to the DTMF tone generator serially. The tones corresponding to each bit is then sent to the input stage of the transmitter. A start bit is sent at the beginning of each 8-bit code and a stop bit is sent at the end of it, bringing the total to 10 tones per ID. The DTMF tone generator produces a combination of two frequencies, one from a high range and one from the lower range, as a tone. This makes the transmission robust and minimizes interference from room noise.

The DTMF generator is also used for signaling in telephone lines. The input to the DTMF generator is a binary combination selecting the tone to be generated. There is a matrix of 4 rows and 3 columns to choose the tone from, and the tone is selected by grounding the appropriate row and column, as is on the telephone dial pad. The DTMF O/P is a tone combination at 2V and has to be stepped down to a few milli-volts for the transmitter to transmit it without distortion. This is performed simply, using a resistor bridge at the transmitter I/P.

FM Transmitter Blocks:

First amplification stage: This is a standard common emitter amplifier. A capacitor isolates the microphone from the base voltage of the transistor and only allows alternating current signals to pass.

Oscillator stage: The tank circuit, the transistor and the feedback capacitor are the oscillator circuit here. An input signal is not needed to sustain the oscillation. The feedback signal makes the base-emitter current of the transistor vary at the resonant frequency. This causes the emitter-collector current to vary at the same frequency. This signal fed to the aerial and radiated as radio waves.

Note that the tank circuit does not oscillate just by having a DC potential put across it. Positive feedback must be provided.

The green trim cap will enable you to tune the Tx. A lower value of the capacitor will move the frequency up towards the other end of the FM band.

Final Amplification Stage: This RF stage adds amplification to the RF signal. It needs an RF transistor to do this efficiently. We use a 1018. L2 (an RFC - radio frequency choke) and the capacitor in parallel with it are designed to reduce harmonics from the circuit.

A small coupling capacitor on the aerial is optional to minimize the effect of the aerial capacitance on the final stage LC circuit.

Receiver side

FM receiver:
The receiver has to perform the demodulation of the received high frequency signal. We used the TDA7000 chip for the same. The IC has an FLL (Frequency-Locked-Loop) system with an intermediate frequency of 70 kHz. The I.F selectivity is obtained by active RC filters. The only function which needs alignment is the resonant circuit for the oscillator, thus selecting the reception frequency. Spurious reception is avoided by means of a mute circuit, which also eliminates too noisy input signals. Special precautions are taken to meet the radiation requirements.

The TDA7000 includes the following functions:

1. R.F. input stage (Hand wound inductors; 7 turns of 22 gauge copper wire $\sim$ 130nH)
2. Mixer
3. Local oscillator (Hand wound inductors; 5 turns of 20 gauge copper wire $\sim$ 56nH)
4. I.F. amplifier/limiter (70 KHz)
5. Phase demodulator
6. Mute detector
7. Mute switch

2.2.3 Circuit diagrams

The circuit diagrams are given in the appendix at the end.

2.2.4 Block tests

Word recognition

1. Mike and amplifier were tested with speech i/p and waveforms as expected were observed.
2. The LF2407A DSP itself was tested for program loading and running through JTAG.
3. The memory interface and input/output ports were tested by writing and reading specific patterns.
4. The internal ADC of the LF2407A was tested firstly with waveforms using signal generator and then the actual speech. The sampled data was dumped into the memory and the graph was viewed using the Code Composer's graph utility.
5. The audio codec serial interface was tested by running the dummy communication cycles.

FM Transceiver

Bit rate The link was checked by transmitting different codes continuously. The time lag between the switching of different lines was measured. The time lag was found to be about 1 second, meeting specifications.

Range The link was tested for the range of approximately 50 meters. The link worked reliably.

Error rate Virtually error free when tuned properly.

2.3 Firmware

The firmware code is developed using the *Code Composer v4.1*. It is an Integrated Development Environment for programming C2000 series DSPs. It has a C compiler, an assembler and linker. It also provides the debugging features through the JTAG emulation interface. We used the XDS510PP-PLUS JTAG pod for downloading and running the codes on our board. The code is written mainly in C except for assembly code for some parts relating to the configuration of the DSP and the peripherals.

The basic support files were written for the register address definitions, memory map of the target processor, vector tables, initial configuration of various control and status registers, initialization of various peripheral modules like timers/counters, GPIO, ADC module. This involved writing mainly the MACROs and small functions (assembly/c).

2.3.1 Routines and FSMs

FSMs

The receiver FSM is shown in figure 2.2.

Routines

The routines involved are classified as follows —

1. Initialization routines - initDSP(), initBoard(), initTimersA(), initADC(). These configure the CPU and peripheral registers for desired operating mode of the hardware.

2. External FLASH access routines - flash_clear(), flash_BLKread(), flash_BLKwrite().

3. Interrupt Service Routines - bad_trap(), nmi_trap() and c_int1() which catches the ADC end of conversion interrupt and analyses the sample data for start detection.

4. FFT routines - four1() from Numerical Recipes which works with float data and intFFT() provided by TI for integer FFT on C2xx DSPs.

5. Core routines - processData() which forms the core of the speech recognition algorithm. It starts with stored word and calls FFT routine, extracts the coefficients and passes them to the DTW routine - recognitionTraining() and minTraining().

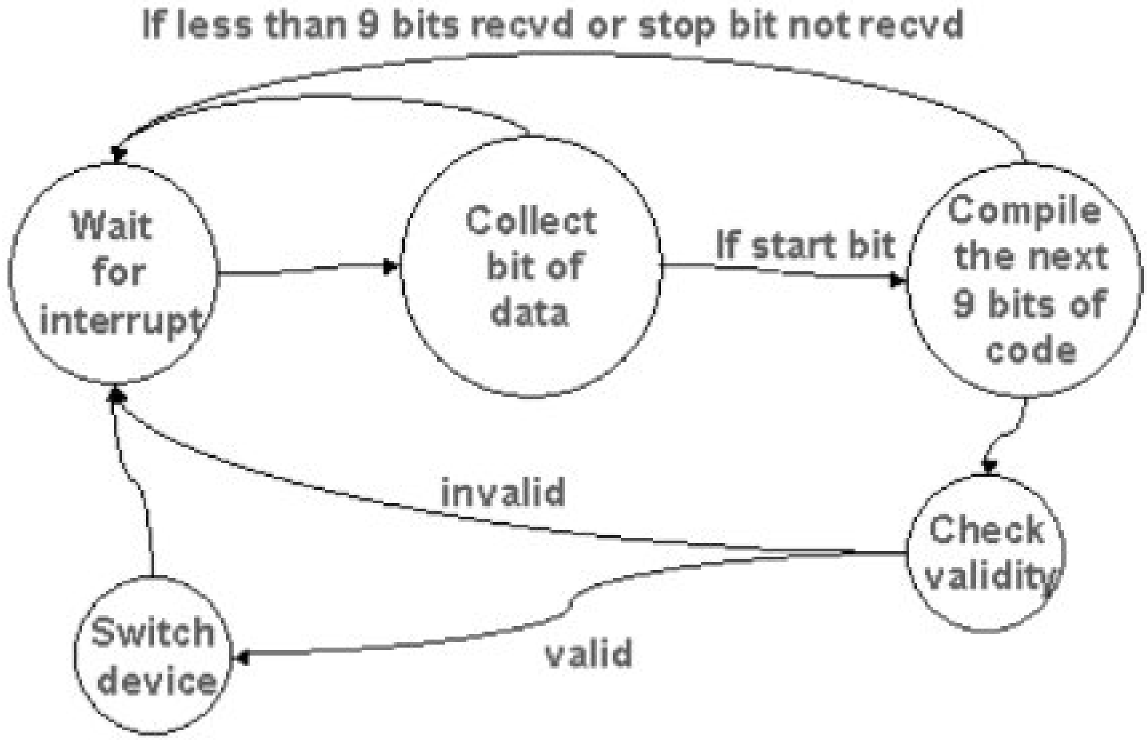


Figure 2.2: Receiver - FSM

6. RF link interface - the routine rf_link_test() carries out RF link test by transmitting all the tones that are used during normal transmission. The routine xmitbyte() takes a byte and transmits it serially with interleaving null tones with suitable delay.

7. main() - as always the starting point of the program execution.

2.3.2 Routine tests

Each of the routines was tested individually. The test procedure involved, applying the routine under test to standard input signal whose output is known or can be calculated. The observed results were compared to the expected ones. Then several routines were combined together and their behavior was tested using standard input as well as the realistic input. The test criteria include the accuracy, amount of memory space consumed, execution time.

2.4 System integration and test

DSP board

1. Firstly the voltage levels at various nodes on the DSP board were measured. The system clock (CLKOUT of DSP) was observed for its frequency. The SRAM was tested using pattern test. The program loading and smooth execution of the code was tested through the Code Composer.

2. Analog Part: The amplifier was tested first with regular input in the form of sinusoidal waveform and then the actual speech signal was applied. The output waveform was analyzed so as to adjust the gain of the amplifier such that there is near full scale amplification with no clipping.

3. The ADC module of the DSP was tested with regular sine wave at the input of the preamplifier. The mean level and maximum/minimum were observed. The mic was connected and the noise threshold levels were calculated. Then the actual speech signal was captured and plotted in the graph utility of the Code Composer. The time domain waveforms of the known words like 'one', 'two', etc were found to be satisfactory as compared to the waveforms obtained in simulations.

4. The start detect algorithm was tested for its correctness and reliability. The thresholds were adjusted so that only a well spoken word is captured.

5. The speech recognition algorithm was tested on the TI 6211 DSK (DSP Starter Kit). On our board, the testing was carried out by dividing the code into logically separate processing steps and the data after every processing step was observed in the graph utility.

6. The entire speech recognition algorithm was combined and tested for recognition accuracy and time for analysis.

FM TRANSCEIVER

1. Testing the encoder: Different tone codes were given as I/P to the DTMF generator, through its input pins. The O/P tone-pairs generated at pin 16 of the DTMF chip were observed and were as expected. Thereafter, different bit sequences were dumped onto the input pins of the 5089 and the O/P tone-pairs were confirmed.

2. The FM transmitter was built and its range and tenability tested. It worked reliably with clear reception (using normal FM radio) over range of 50 meters. Voice input was used to test the link. Also the settings of the trimcap was disturbed a few times to test how easy tuning was. It was found to be easy enough to tune the transmitter to the receiving station.

3. Thereafter the DTMF tone generator O/P was fed as an I/P to the transmitter. However the received tone, checked using a jack, was highly distorted. After a lot of testing it was identified that the transmitter takes I/P signals only in the range of a few milli volts whereas the O/P tone of the DTMF chip is of the order of 2V. This incompatibility was removed by using a simple resistor bridge.

4. Decoder: The DTMF decoder circuit was built and it was fed a direct I/P from the DTMF generator circuit. The circuit worked reliably, outputting the appropriate binary codes. The StD pin which signals the detection of valid tone-pair at the input was found to be high each time.

5. FM receiver: The FM receiver was built and tested by transmitting over our transmitter tuned to below the FM commercial band. After various changes to the capacitor values in the local oscillator, the right combination was hit upon. The inaccuracy in calculation of the capacitor values arose due to the usage of hand wound inductors. The receiver was able to clearly catch the words spoken into the transmitter, when in a close range. The link was thereafter also tested over a longer range and was found to be working reliably.

6. The link was then tested by transmitting different bit sequences from the DSP board. The Std pin at the DTMF decoder that signals the reception of a valid tone was checked to be high each time. The LEDs on the pcb displayed the tone number received. Thus the link worked reliably for transmission and reception.

7. The microcontroller at the receiver was coded to receive the 8 bit device code serially and then compile it together to regenerate the transmitted sequence. It would then write 1 to the output line corresponding to the appropriate device. This link was tested using all the blocks. The transmitter was coded to transmit all the codes in sequence, twice. Thus at the receiver, each device was switched on first in turn and then switched off. This was displayed by the LEDs at the O/P port of the microcontroller.

8. Thereafter the link was tested for speed of switching. Each time the delay introduced by the delay routine at the transmitter was reduced. It was found that the link worked reliably with switching delays of around

100ms. Thus allowing transfer of one byte within one second as per the specification.

Part III

Service Manual

Chapter 3

Service manual

3.1 Problem identification

1. Check whether both the Power-On LEDs are shining on the DSP board. If not then check whether the power supply wires are connected securely and polarities are correct. Check the voltage using multimeter.
2. Check whether the state LED glows everytime you speak into the mic. If not then decrease the mike distance from mouth and try again. If not still then press the reset switch and try again. If it is still not responding then adjust the potentiometer in the preamp circuit to increase the gain of the amplifier. If possible, the output waveforms can be checked at the test point provided.
3. Check whether the red LED at the receiver lights up during the transmission after the command is recognized correctly. It indicates correct reception of the codes, If it is not lighting up or is flickering then there is some problem in the transmission and the reception of the codes. The A first solution would be to align the antenna of the system more strategically. Also the distance between transmitter and receiver can be decreased to improve the reception. If still the transmission is unreliable, the transmitter has to be tuned correctly by adjusting the trimcap. Refer to the detailed description of the transmitter.

In case the receiver is still unable to receive the correct code, check the following points: Testing points: VCC and GRND of all circuits. The two tx modules should have a common ground and similarly for the two receiver modules.

ENCODER ALE pin of the microcontroller. Should give a waveform resembling a noisy square wave. Pin 16 of the DTMF chip. Should show different tone-pairs when in use.

FM TRANSMITTER Can only check the points in the pre-amplifier stage, since beyond that high frequency oscilloscope will be required. If that is available, the waveforms across the local oscillator can be measured. They should be at a frequency of about 70 MHz. Else the transistors might have become faulty. The rest are all passive devices and don't really get spoilt with usage.

FM RECEIVER Check the local oscillators (you could do this by checking the frequency across the inductor of the L.O.), else replace the capacitors or inductors appropriately. The final O/P of the receiver should be a tone-pair similar to the one generated at the transmitter. Else try replacing the active components.

DECODER If the tone is received properly at the input to the decoder but isn't being recognized, then check the DTMF detector. If the StD pin is high at each correct tone transmission, check the microcontroller. Either the microcontroller has malfunctioned or there is some bug in the code. The microcontroller can be checked by checking the ALE pin(31).

Part IV

Appendices

Appendix I

3.2 Circuit Diagrams

The circuit diagrams for mic preamplifier, DTMF encoder, FM transmitter, FM receiver and DTMF decoder are provided. The circuit diagram of the DSP board is too complex to be given in the report. Please refer to the schematic and board files - edl16dsp.sch and edl16dsp.brd - in Eagle 4.1.

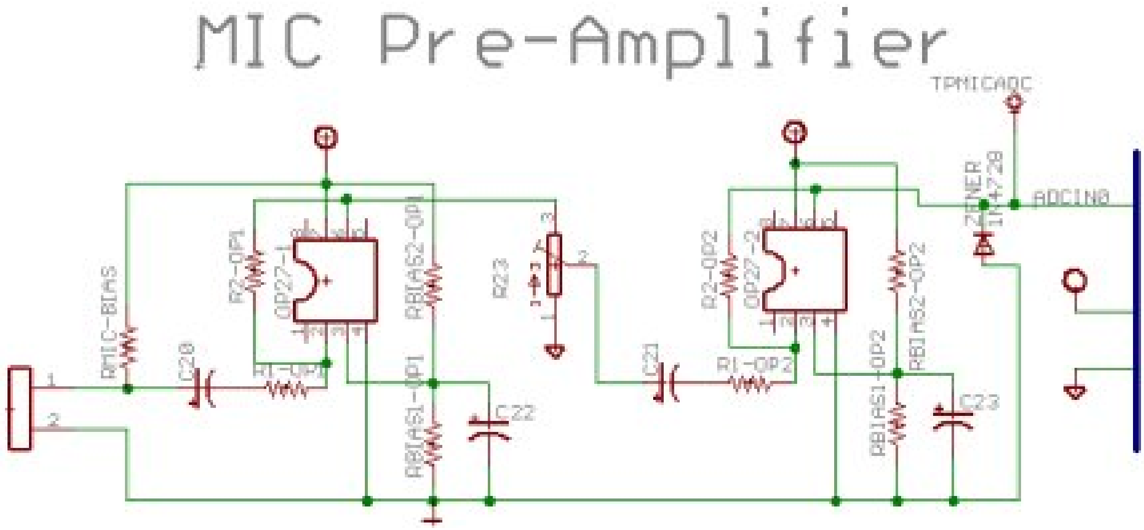


Figure 3.1: Mic Preamplifier

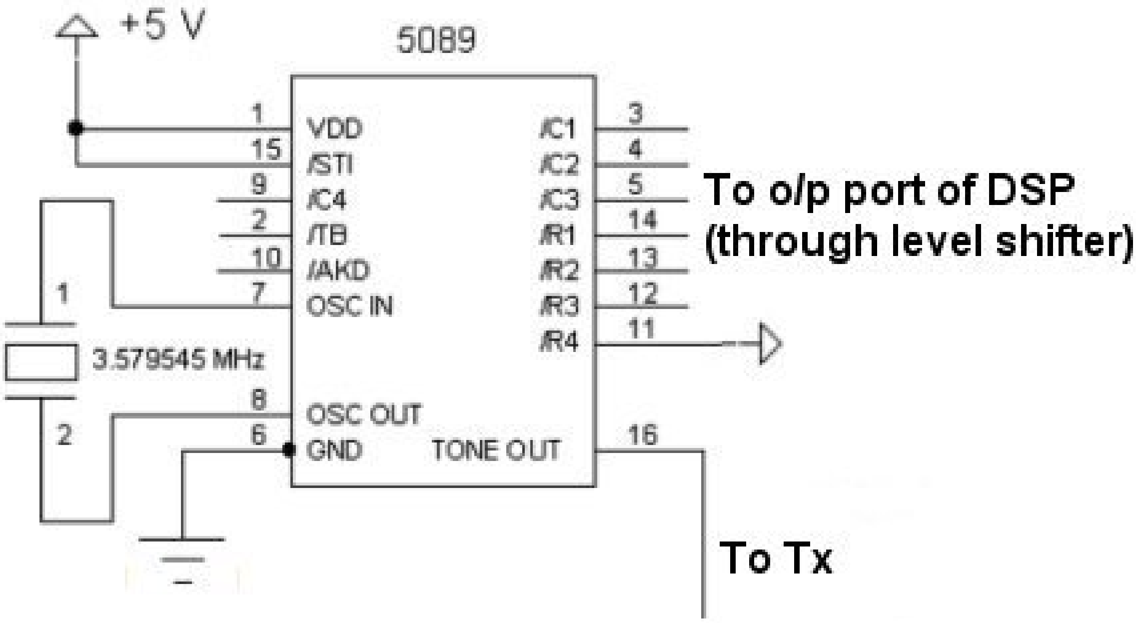


Figure 3.2: DTMF encoder

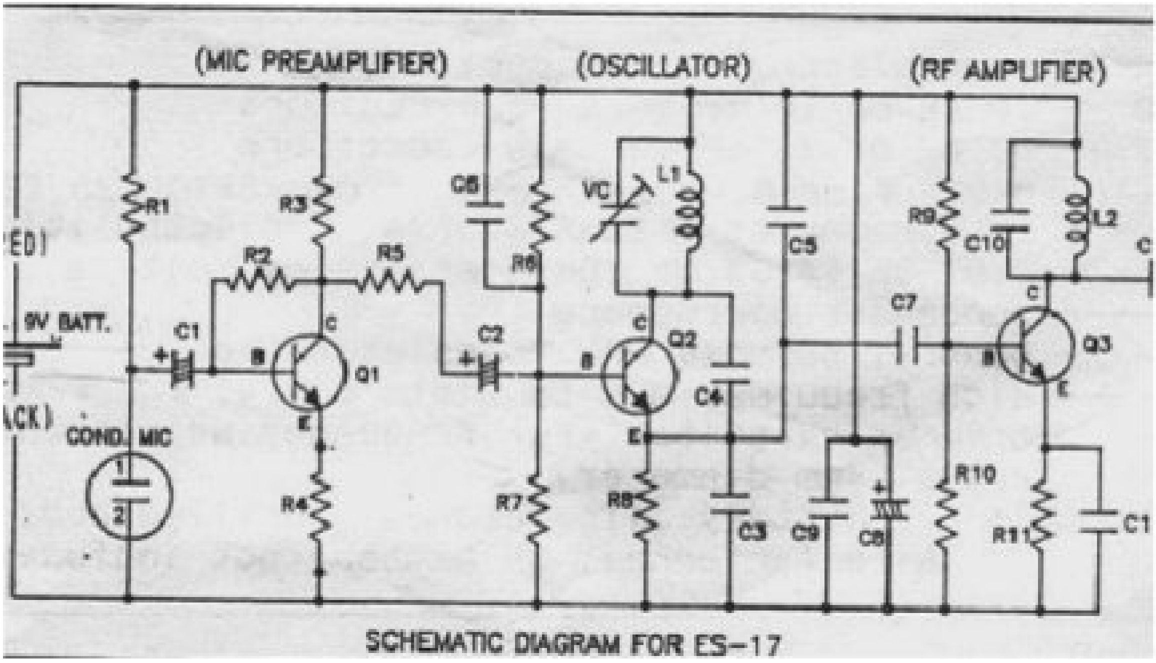


Figure 3.3: FM Transmitter

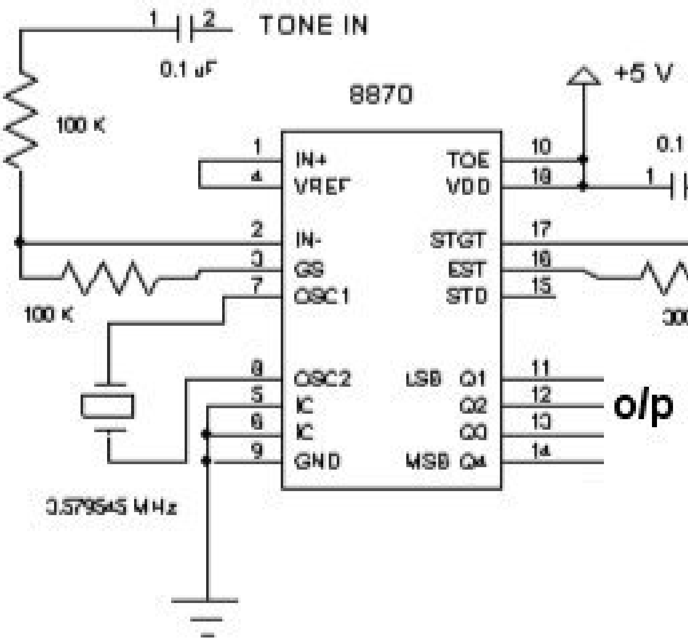


Figure 3.4: DTMF Decoder

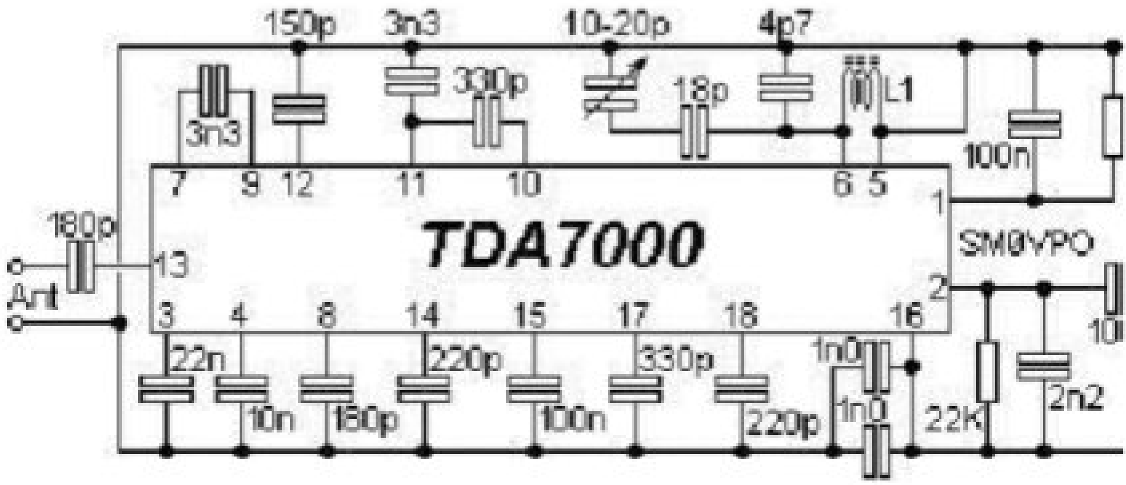


Figure 3.5: Receiver - TDA7000