Global Scientific JOURNALS

GSJ: Volume 8, Issue 6, June 2020, Online: ISSN 2320-9186
www.globalscientificjournal.com

Development of Playback ARS Using VRP Device

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ABSTRACT

The paper was carried out to develop a playback Audio and Recording System (ARS) using Voice and Record Playback (VRP) device. The recording and playback device is a device that makes use of digital storage devices using ISD17120 IC chip, which stores digitally, sound signals inputted from a microphone with impedance of 9kΩ. The processes carried out by the device are controlled by toggle switches to record, playback, forward, increase or decrease volume, erase and reset, in which each processes are indicated active by LED indicator with different display patterns. The audio output is achieved through 8Ω speaker via an inbuilt class D amplifier in the IC (ISD17120). The devices operate with a frequency of 8 kHz and a duration of 120seconds, the power supply to the device is 9V-DC, operating current of 20mA and standby current of 1µA, which is regulated using the 7805 regulator to produce a 5V to the device. This device is capable of recording up to 100 different messages with zero - power messages storage and eliminates the problem of loss of data on exposure to stray magnetic field and also reduce space occupied.

Index Term-- Recording, Playback, ISD17120.

1.0 INTRODUCTION

Voice and Record Playback (VRP) device is made up of Audio and Recording System (ARS). The VRP device is purely an electronic device which converts and analogue signal representing a voice to a digital signal and records the digital signal in a recording medium.

When the voice is recorded, it converts the digital signal to an analogue message when the voice is reproduced.

The device consists of several transducers, driver stages, regulated power supply unit and ISD17120, on which all data are process and stored. The input transducer is an Electret microphone.

It is a device that is capable of converting analogue speech sounds, in form of varying wave patterns, into slightly appreciable electrical signal (current). This signal is amplified by an in built amplifiers and decoded by decoder which converts the analogue signal generated into binary digits (bits) for further processing. [1, 2]

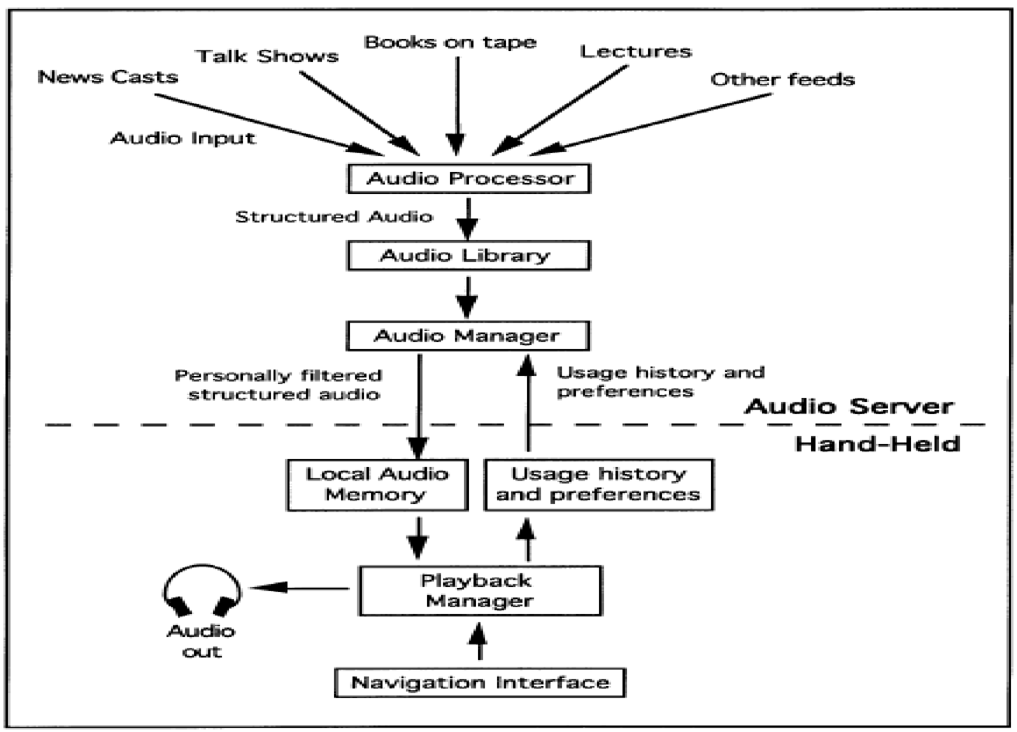

Fig. 1. An overview of digital audio playback device

The paper aims to make use of a single, flexible and versatile chip, ISD17120 IC, which can be used to record, store and playback analogue speech sound and microcontroller compatible for advanced usage.

Nowadays, most people need a means of recording voices, musical note, interviews, media presentation etc. and sometimes not only one, but a lot.

This brings trouble to the users because the means of recording with the previous devices, which are associated with the problem of low voice quality, memory loss due to magnetic influence, need for storage device and high battery consumption rate.

Besides that, we always realize that most of the people prefer to use the digital sound devices because of its smartness, better reproduction of the sound recorded and also its portability and flexibility.

Generally, there are four main scopes of work in this paper which includes:

1. System features design
2. Hardware development and prototyping
3. Hardware testing
4. System integration and implementation

System features design includes the initial idea of enhancing the current recording device and the features that can be added into it. Then, hardware development comes into places where various components are identified and purchased. Later on, all the components needed are connected according to the schematic and circuit designed. The circuit will be tested according to the schematic and circuit designed. The circuit will be tested on the connectivity and will be troubleshoot accordingly if the system fails. Training will be given to the system to test on its functions and modes. Further testing is required to make sure that the system is reliable and provides high quality output.

2.0 THEORETICAL BACKGROUND

Recording is a technique for storing information on a storage device. It serves as a means of permanently documenting, individual speech, telephone message, musical sounds and instructions. [3]

The technique usually involves converting the speech sounds to be recorded into small electrical currents by microphones. The alternating current generated are embedded on storage elements, which varies from days of gramophone recording to more robust magnetic tape storage principle. [4]

Playback can be only achieved by the use of special equipment that will extract the information from the storage and then reproduce it using several amplification stage and loud speaker. [5]

The quality of digital audio is measured by the following parameters:

1. **Sample Rate** is the number of samples or "snapshots" taken of the signal and is measured in hertz/second. The higher the sample rate (or the more samples per second) the better the digital representation will be. For recording music-quality audio the following PCM sampling rates are the most common: 44.1kHz, 48kHz, 88.2kHz, 96kHz, 176.4kHz, 192kHz.
2. **Bit Depth** refers to the number of bits used to represent a single sample, i.e. the number of bits used to represent a single audio wave (the word size) directly affects the achievable noise level of a signal recorded with added dither, or the distortion of an undithered signal. For example, 16-bit is a common sample size. While 8-bit samples take up less memory (and hard disk space), they are inherently noisier than 16-bit or 24-bit samples. The higher the bit depth, the better the recording; however, higher bit depths also lead to larger file sizes.
3. **Bit Rate** refers to a measurement of digital audio based on the following equation and is usually expressed in kilobits/second. Bit rate = (bit depth) x (sampling rate) x (number of channels).
4. **File Size** for recordings are calculated by combing bit depth, sample rate, channels, and recording time.

$fileSize = bitsPerSample * samplesPerSecond * channels * duration8$

e.g. for a stereo sound recording; 8∗44100∗2∗308= 2,646Kb/s

Manual Level Control

Manual level control involves the operator adjusting the levels by use of the input level or recording level controls. When recording with manual level control it is best to use a limiter to protect against clipping.

As with all digital audio recording, the ideal recording level is that which uses as much of device's digital bandwidth as possible without exceeding it. In practical terms, avoiding distortion caused by too loud a level, but also the excessive background noise and indistinctness of too quiet a recording.

1. Limiter: A limiter sets a threshold above which the signal will be gently pushed down in order to prevent clipping. This is preferred over ALC or AGC as it allows the operator to set optimal levels and minimizes noise while still protecting the recording from clipping. Limiters are not foolproof, however, and good levels must still be determined by the operator.

2. Automatic Level Control/Automatic Gain Control (ALC/ALG): ALC and AGC are circuits in a recorder that determine an average optimal level. Use of one of these will minimize the risk of clipping but typically not produce as high a quality of recording as manual level control, because they boost quiet moments in the recording up to record level and thus boost background noise.

3.0 SYSTEM DESIGN

The input stage consists of the input transducer (microphone) that converts the analogue speech sounds into electrical signal of small magnitude. Storing and processing of the input data is accomplished by a 120 seconds cell non-volatile Electrically Erasable Programmed Read Only Memory (EEPROM). The sampling frequency of the chip can also be adjusted from 4 kHz to 12 kHz with an external resistor, giving the user greater flexibility in duration versus recording quality for each application. Record and playback are initiated by a simple toggle switch while playback can be paused or altogether terminated by a simple push stop/reset button. The output (audio) is achieved through a 8Ω speaker from an inbuilt audio amplifier. Operating voltage spans a range from 2.4 V to 5.5 V to ensure that the ISD1700 devices are optimized for a wide range of battery or line-powered applications.

This device provides high-quality, single record/playback solutions for 120 seconds messaging application. The Integrated Circuit (IC) chip used "ISD17120" has an on-chip oscillator, microphone preamplifier, Automatic Gain Control (AGC), anti-aliasing filter, smoothing filter, speaker amplifier and high density multilevel storage array as shown below.

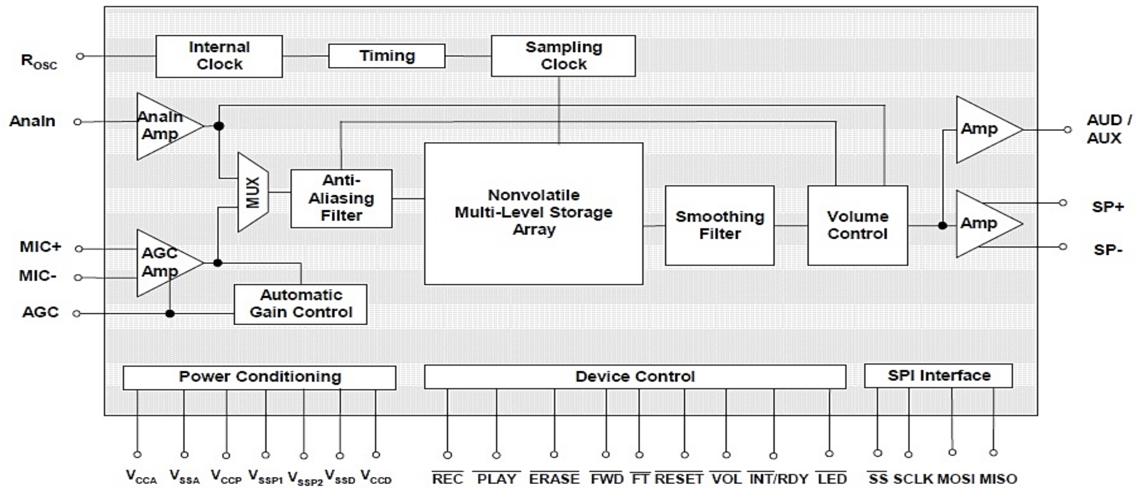

Fig. 2. Configuration of ISD17120 chip

The address and control pins can be connected to a microcontroller, manipulating different tasks such as message assembly, message concatenation, predefined fixed message segmentation and message management. [6]

Most voice recorder provides basic function such as record, stop, play and fast forward. To permit a user to selectively activate these functions, a number of manually operable switches are typically provided on the cosine of the recorder of which the Recording/Playback device also incorporates some. In addition, the device is microcontroller compatible allowing complex messaging and addressing to be achieved. Recordings are stored in on-chip non-volatile memory cells, providing zero-power message storage. This unique, single-chip solution is made possible through multilevel storage technology. [7]

Voice and audio signals are stored directly into memory in their natural form, providing high-quality voice reproduction.

Thus, this paper design provides single-chip record/playback solutions for 120 seconds messaging applications.

The input data from the microphone is stored in a cell, non-volatile multilevel storage arrays, after being amplified by microphone preamplifier, then to an automatic gain control (AGC) which provides constant signal level to the data processor no matter the incoming signal strength and subsequently filtered by 5-poles active anti - aliasing filter. The output (when selected from the address buffers, goes to decoder where the message is selected from the cell) also to 5 poles active smoothing filter to reduce the noise, then to a MUX (Multiplexer) which selects the desire message before amplification is done on the message again and finally pass the outgoing audio signal to the 8Ω loudspeaker. One LED is included to signify with different pattern of power supply state, recording and playback indication respectively. The circuit can be broadly divided into various stages, the input stage data processing and storage stage; and the output stage as shown in the block diagram below.

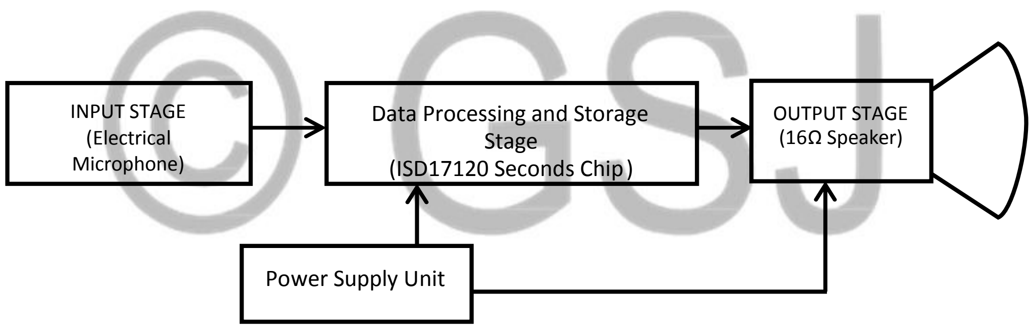

Fig. 3. Block Diagram of the Device

3.1 The Input Stage

This stage consists of the input transducer (the electrets microphone) which stands alone as a single component; it converts the analogue speech sounds into electrical signal of small magnitude. It also has a pre-amplifier and the main amplifier that raise the signal level to the desired level. There is also a provision for an Automation Gain Control (AGC) that ensures constant signal level into the data processor all of which are embedded in the IC.

The electrets microphone works with the principle in which current varies as the capacitance of a circuit varies. [2]

The advantages derived from using this type of transducer are as follows,

1. They are very sensitive if compared with others.

2. Surround sounds can be eliminated before the converted signal get to the in-built amplifiers and filter system of the IC.

3.2 Data Processing/Data Storage Stage

The major features of this stage consist of:

i. Filter Network.

ii. Memory address buffers.

iii. Signal decoder.

iv. Analog transceivers and

v. 480K cell non-volatile multilevel storage array.

All the above see the purpose of positioning the received signal (analog) in its appropriate bandwidth while eliminating unwanted signals, before the actual data storage. Nyquist's Criterion for sampling frequency *f(s)* must be considered, which stated that in order not to lose any signal information, then the criterion for sampling frequency f(s) is; $Fs \geq 2fmax$. Where: *f*max is the maximum frequency component of the signal being sampled. It is therefore vital analogue signals are subjected to anti-aliasing filtering prior to sampling in order to insure that no frequency components higher than Fs/2 are present. [8]

The message length of the ISD17120 chip is dependent on the sampling frequency used. Reducing the sampling frequency will increase the message length but with reduced audio quality.

3.3 Multilevel Storage Array (MSA)

MSA is a device on which the sampled data signal (digital format) is stored. It is classified as non-volatile, which means that it retains its stored data permanently, and removal of its power supply does not result in loss of data.

Any storage device must posse the following facilities [9];

1. A number of individual storage elements known as memory cells, each capable of temporary or permanent storage of a single binary digit.
2. A system of addressing which provides a means of selecting a specific memory cell (or group of cells) within the memory device.
3. A means of writing data into a specific memory location.
4. A means of reading data from a specific memory location.

3.4 Electrically Erasable Programmable Read-Only Memory (EEPROM)

Read —only Memory (ROM) is a type of memory that is used for non-volatile storage of data. Such storage is done by the hardware manufacturer according to bit patterns supplied by the user. In this project, the memory device used can be programmed by the user (i.e. it is field programmable), erased and re-programmed. These processes of erasure and reprogramming does not require that the device be removed from the circuit. [1]

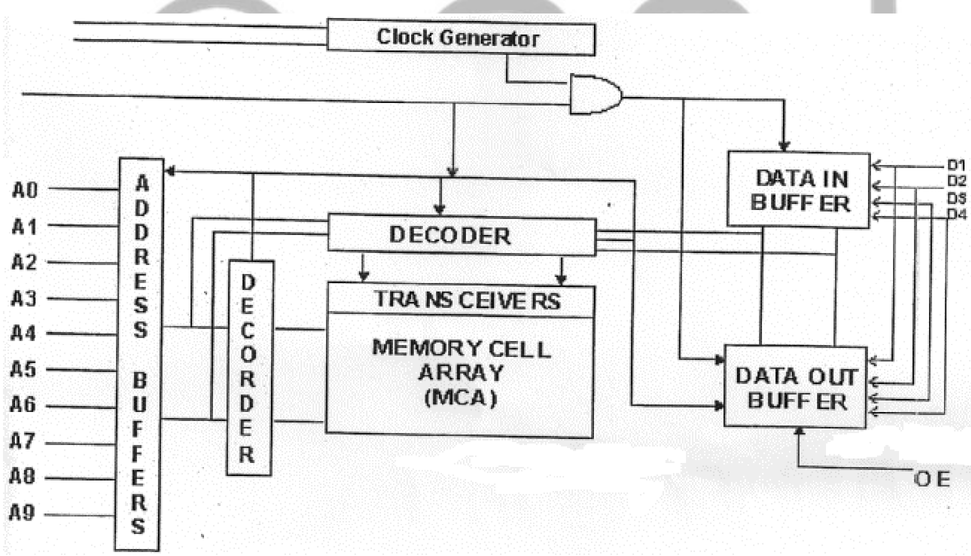

Fig. 4. Internal Structure of an EEPROM Device

3.5 Memory Address Buffers

A buffer is a device that is connected between two (2) parts of a system to prevent unwanted interaction. It serves as an Isolator.

It frequently consists of a current amplifier or a small memory. A conventional buffer used in a logic current would have its output at one of two different logic states; logic 0 (low) or logic 1 (high), but there is a third output state of high impedance (tri-state buffers). [10]

Data transfer are actually carried out when the address buffers have received appropriate control signals, applied to their chip enable (E) or chip select (Cs) input.

Controls signals require for the memory device used in this project design are as follows.

1. A chip select (Cs) signal to select one particular memory location or group of cells.
2. An (RD) Read signal to enable the output buffer to allow memory place data onto its output.

A write (WR) signal to enable the input buffer to allow data to be stored into the memory. [11].

3.6 Smoothing Filter

It is mandatory that the frequency bandwidth of the project circuit be restricted. This is necessary for the purpose of optimizing the signal-to-noise ratio.

Hence, it makes sure to limit the bandwidth of the system to the minimum, since noise power is usually directly proportional to bandwidth, which will allow the wanted audio signal to pass un-impaled.

On playback, analog transducers will convert the stored digital signal into a true replica of the original analog speech are removed by the smoothing filter, which is a low power active filter matched to the frequency spectrum of the analog speech sound. [10]

3.6 Output Stage

The output of the smoothing filter is further fed into an audio frequency (AF) power amplifier in order to raise the signal level to point suitable in driving the output transducer (Loud speaker).

Operational amplifier found diverse application in their regard. A pulse width modulation, class D operational amplifier was used in the design work for the sound output.

A loudspeaker function is to convert analog electrical signal to analogue speech/audio sounds.

Power Supply Unit.

All the stages described, will not operate unless suitable means of D.C power supply is available and applied appropriately to various required stages.

The most portable and available source of D.C power supply is dry cell battery. On the other hand, batteries are proving to leakages over the time, that is, their e.m.f is not usually constant overtime. Therefore, if battery are not to be used, the cheap alternating power supply (A.C 240, 50Hz) can be used with maximum of 12V –DC output. A 9V-DC battery is used to power this design. A D.C power supply which maintains the output voltage constant irrespective of mains fluctuations or load variation is known as voltage regulation. Most of the commonly used IC voltage regulator are three –terminal devices. Advantages of IC voltage regulator is that properties like thermal compensation, short circuit protection and surge protection can be built into the device.

The 7805 IC voltage regulator are characterized by Vout = 5V and Iout = 500mA, the maximum input voltage to the IC = 12V.

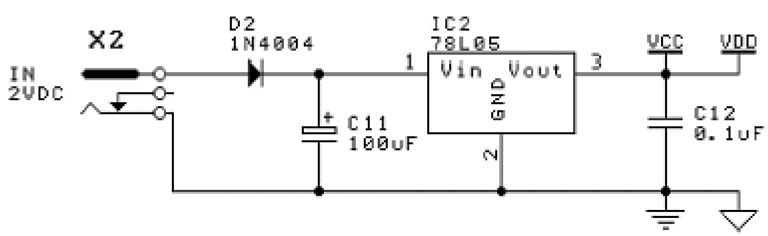

Fig. 5. Power Supply Unit

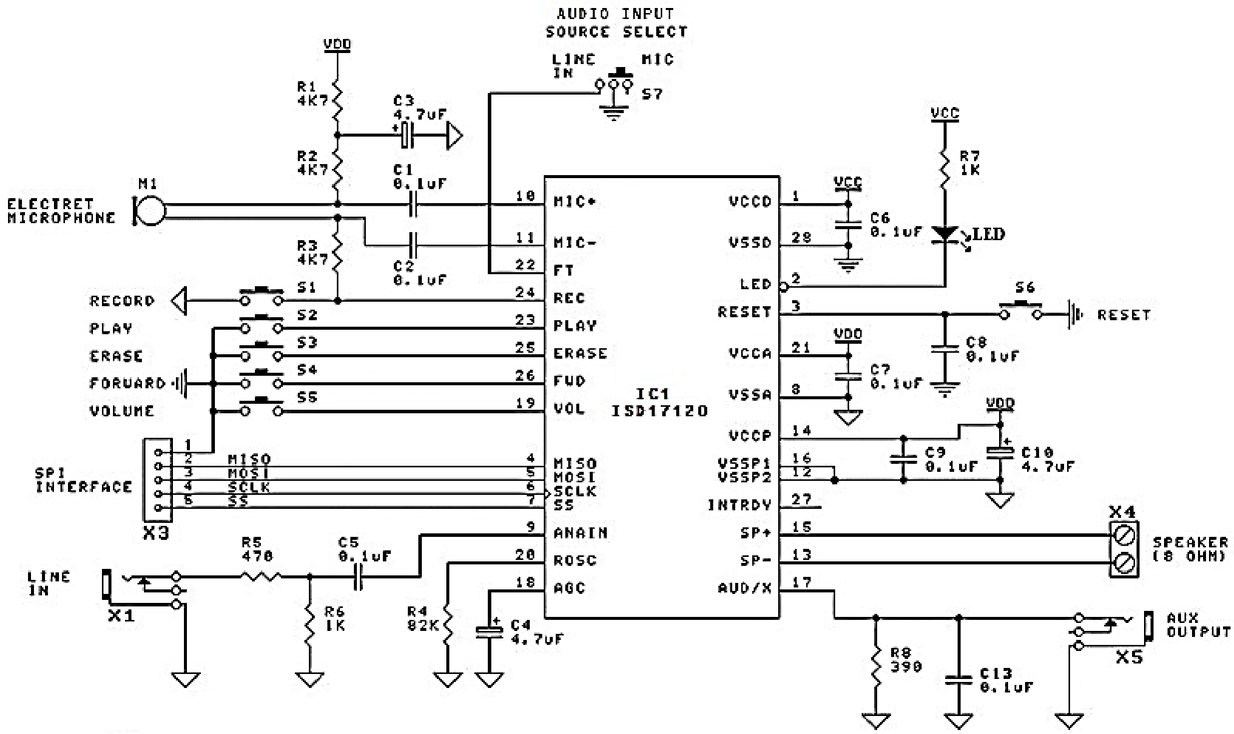

Fig. 6. Circuit Diagram of the Recording / Playback Device

From the fig 6 above, R4 act as the oscillating resistor which determine the sample frequency of the device which set duration. Variation in the impedance of R4, varies the frequency as shown in the table below;

Table I
Effect of oscillating resistor on message duration

Resistor Value (kΩ)	Sampling Frequency (kHz)	Minimum Storage Resolution (msec) ($f = \frac{1}{t}$)	Message length (secs)
53	12	83.3	80
80	8	125	120
100	6.4	156	150
120	5.3	187	181
160	4	250	240

4.0 RESULTS AND DISCUSSION

During the first stage of design, the project was mounted on the breadboard and some tests were carried out, the whole design was later transferred onto the printed board so far the aim and objective of the project has been achieved, then the final test was carried out. All the results recommended by the manufacture and the experimentally obtained ones discussed in a tabular and graphical form.

Table II
D.C Parameter Result

Parameters	Symbols	Manufacturer Recommendation	Values experimentally Obtained
Supply Voltage	V_{DD}	2.4V – 5.5V	5.0V
Input low voltage	V_{IL}	0.3V - 1.5V	0.8v
Input High voltage	V_{IH}	3.5 – 5V	4.6v
Output low voltage	V_{OL}	0.3V - 1.5V	0.8v
Output High Voltage	V_{OH}	3.5V – 5V	4.6v
Operating Current	Icc	20mA -30mA	20mA
Standby current	I_{SB}	(0.5 – 1)µA	1µA
Output head Impedance	R_{EXT}	16Ω	16Ω
Mic Preamp Input resistance	R_{MIC}	(4 – 15)kΩ	9kΩ
Aux Input Resistance	R_{AUX}	(5 — 20) kΩ	11kΩ
Gain from Mic to SP+	A_{MSP}	6 – 40dB	6.7dB*
Speaker Output Load	R_{SPK}	8 - 16Ω	8Ω

Note: DC battery was used in this paper with a 9V DC supply. The use of AC power supply is optional. Also, the Volume to the output speaker increase by 4dB per click.

GSJ: Volume 8, Issue 6, June 2020
ISSN 2320-9186

Timing Diagram Result

The timing diagrams obtained from oscilloscope for both record and playback with manufacturer accompanying timing diagram for LED operations are shown in figures below.

Record Operation

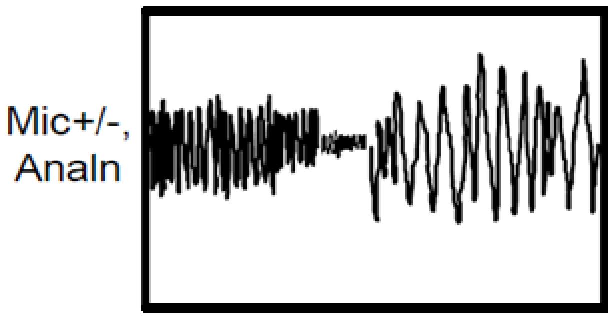

Fig. 7a. Recording timing diagram from oscilloscope

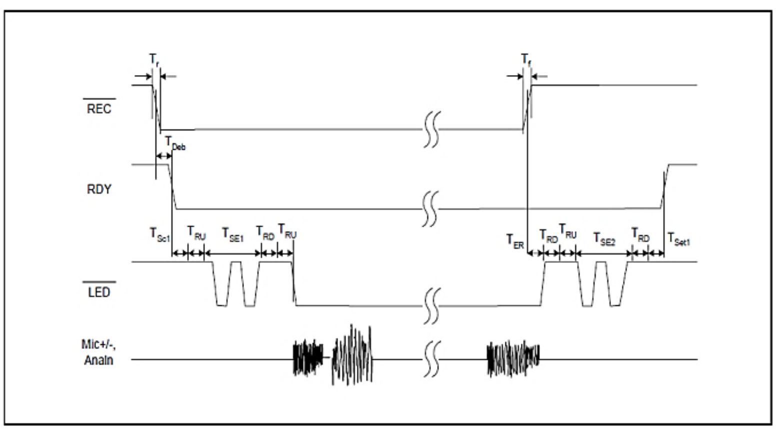

Fig. 7b. Record Operation of the manufacturer

Figure 7a shows the LED operation waveform on the oscilloscope screen when the recording operation is active. Comparing the waveform with that of the manufacturer's prescription (figure 7a and 7b), there is correlation between the two waveforms, this shows that the circuit configuration for the sound input is accurate.

4.4.2 Playback Operation

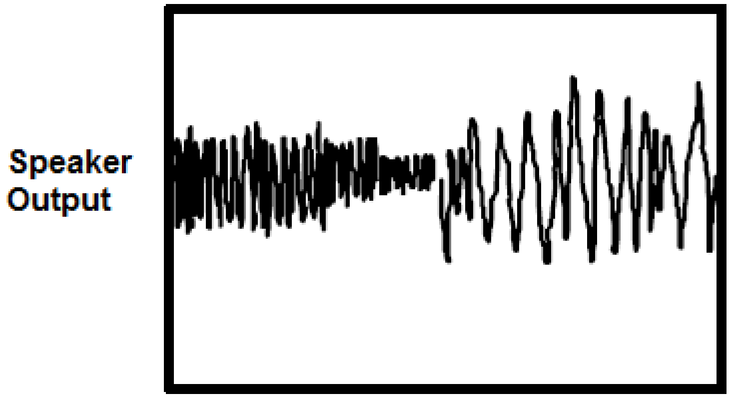

Fig. 8a. playback timing diagram from oscilloscope

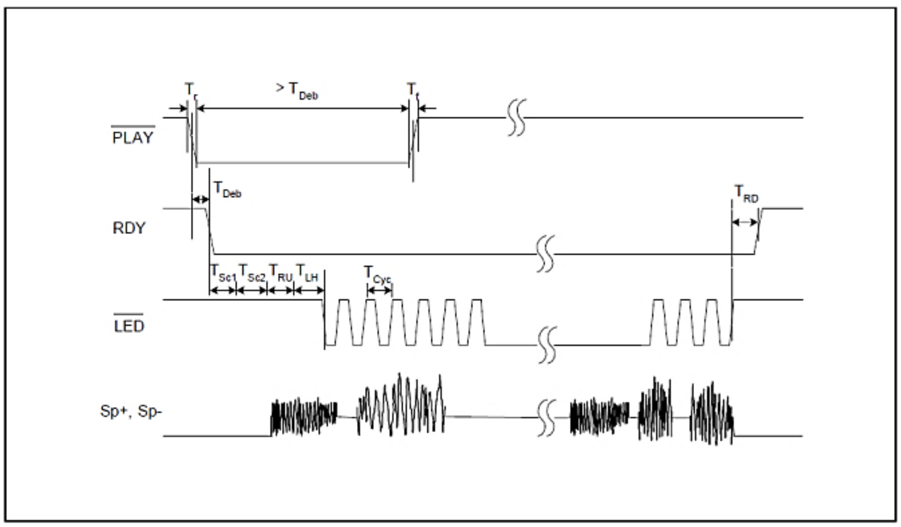

Fig. 8b. Playback Operation Record of the manufacturer

Figure 8a and 8b shows the LED operation waveform on the oscilloscope screen and the manufacture's view respectively, when the playback operation is active. Comparing the waveforms, there is similarities between the two waveforms, this shows that the circuit configuration for the sound output is accurate.

5.0 CONCLUSION

A recording/playback device makes use of the latest technology in digital storage devices using ISD17120 IC chip. It stores digitally, a sound signals inputted from a microphone (already converted to electrical signals) and after recording duration, playback the recorded message in an audible form.

The devices operate with a frequency of 8kHz and a duration of 120seconds, the power supply to the device is 9V-DC, which is regulated using the 7805 regulator to produce a 5V to the device. The users must shield the device from unnecessary environmental noise especially during operation. Also, all other precautions associated with electronic devices must be strongly observed.

Furthermore, there is no need for power supply in order to keep the message stored in the IC memory stored or not losing it, because the device provides zero-power message storage.

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