iJRASET
International Journal For Research in
Applied Science and Engineering Technology

INTERNATIONAL JOURNAL FOR RESEARCH

IN APPLIED SCIENCE & ENGINEERING TECHNOLOGY

Volume: 8 **Issue:** VII **Month of publication:** July 2020

DOI: https://doi.org/10.22214/ijraset.2020.30431

International Journal for Research in Applied Science & Engineering Technology (IJRASET)
ISSN: 2321-9653; IC Value: 45.98; SJ Impact Factor: 7.429
Volume 8 Issue VII July 2020- Available at www.ijraset.com

A Novel Approach in Audio Compression using Vedic Concepts

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Abstract: It is needless to say that the popular seven notes called as SAPTA SWARAS drawn from Shrutis are Shadja, Rishabha, Gaandhara, Madyama, Panchama, Daivata and Nishada which are written in short as Sa-Ri-Ga-Ma-Pa-Da-Ni. Natural sounds are produced from these basic notes. Chanting of Veda mantras and any way of expressing emotions is possible through these musical notes. Each note has a frequency, amplitude and time. A sound depends on these parameters. Set of such sounds give a meaning when vector addressed from a text file containing the audio code.
Keywords: ADC, Frequency to Voltage converter, Memory, Microcontroller, Recording, Multiplexing.

I. INTRODUCTION

This is an attempt to encode Sa-Ri-Ga-Ma-Pa-Da-Ni into a microcontroller to reproduce meaningful and clear sounds occupying less memory space [1]. This method records sound in 8-bit text format. This is stored into a small memory capacity of the range in KBs. These text data are the code to evoke the musical notes stored in the dedicated microcontroller. The text carries a sequence 8-bit data which is understood by the programmed microcontroller. Each expression of the 8-bit sequence carrying 3 bite. First 8 bits define a musical note of Sa Ri Ga…. Second 8 bit defines duration of the musical note. Third 8 bits define amplitude of the musical note. Since time component is not stored in the memory space is saved. Time is mentioned as 8-bit data in one location.

Sample code is as follows:

1) b'10110111' ; B7H
2) b'10101010' ; AAH
3) b'01001000' ; 48H

These go with B7H, AAH, 48H in hex values. 'B7H' stands for the note defined under it. 'AAH' stands for duration of the play which goes in milliseconds. '48H' stands for amplitude of the musical note in the strength of volts.
Pingala Chandassastra is the origination for these sequences with the help of which this research is carried.

II. DIGITIZED SIGNAL CAPTURE

Each of Sa-Ri-Ga-Ma-Pa-Da-Ni are a frequency. Their amplitude varies for producing them as a meaningful music [2]. This is described to capture an audio signal which can be enhanced to deliver stereo capture and an HTS of 5.1 or higher to deliver natural sounds with 3 dimensions in audio system.
Audio signal is always in analogue. To capture into a digital media a coded data is required. This is an attempt to capture analogue signal in the form of its frequency and amplitude at every division of time. Here time is 1m Sec.

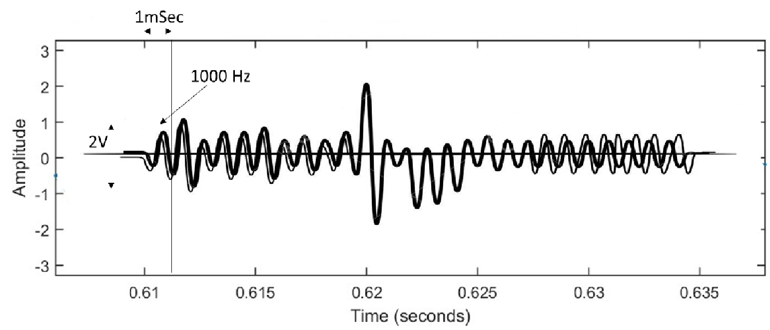

Fig.1 Audio signal with Amplitude, Frequency and Time

ISSN: 2321-9653; IC Value: 45.98; SJ Impact Factor: 7.429
Volume 8 Issue VII July 2020- Available at www.ijraset.com

Fig.1 indicates audio signal with amplitude, frequency and time. From the total time of 25mSec of the audio signal components of 1mSec is concentrated to take the parameters. From that it clearly indicates that the amplitude is 2V and Frequency is 1000Hz. These components are digitized to code it to hex data standing between 00H and FFH for amplitude and frequency. At every 1mSec such data is captured and stored into a temporary memory were the data is consolidated to the repeat of these parameters and are saved as time. This condenses the repeat data to single 3-byte data.

Each parameter carries a weightage for every byte. These are listed below:

Frequency 20Hz to 22KHz is bandwidth of 22020Hz and each of its 255 parts make it to 86.35Hz.

Amplitude 0V to 5V is divided into 255 parts, so each part is 0.0196V.

Time 1mSec to 255mSec is divided into 255 parts. Each part is 1mSec.

Taking an example from Fig.1 data saved into a memory is indicated below.

Memory Location 1 : 0BH (86.35 X 11 is nearly 1000)

Memory Location 2 : 66H (0.0196 X 102 is nearly 2)

Memory Location 3: 04H (considering the parameters are repeated 4 times)

This is repeated and saved into the memory locations which acts as a vector to the microcontroller bearing frequency notes of Sa-Ri-Ga-Ma-Pa-Da-Ni. The capture of codes to form a vector address to the microcontroller, a dedicated hardware is proposed.

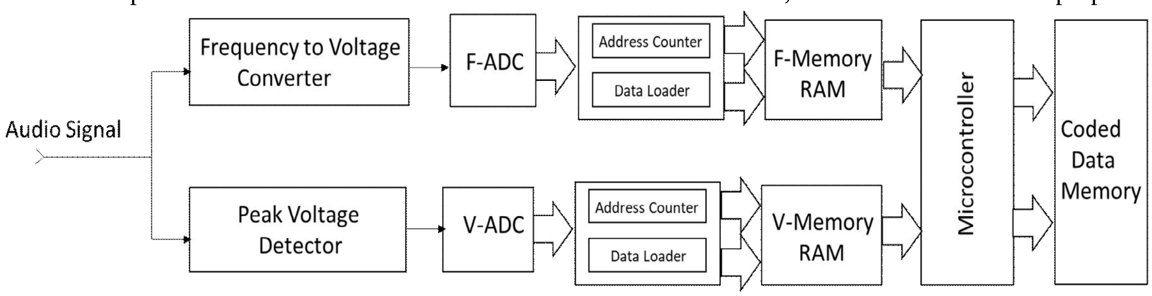

Fig.2 Conversion of Audio signal and storing in Memory

Fig.2 indicates the block diagram of the process of capturing the audio signal as frequency and voltage whose time period of the parameters is determined and encoded to the format by the microcontroller. Analogue signal is divided to two parts for sensing. One is to pic frequency and convert to its corresponding voltage [3][4]. This is done by Frequency to voltage converter [5]. The voltage developed likewise is converted to Digital data by F-ADC. Stream of Digital data is ported out in parallel. A counter to address the memory ports each of the digital weighted frequency into each of locations of memory at an interval of 1mSec. Similarly, Peak voltage detector picks up maximum voltage of the signal and is converted to digital data by V-ADC [6]. This also is picked up by the counter to address the memory location and each data with 1mSec space is placed in memory [7]. Each of the dedicated memories F-Memory and V-Memory are RAM location. This is done till the Audio signal is available in the line. Once this streaming stops the microcontroller takes charge to log all the data to compress and store into the Code data memory which is a ROM. Microcontroller first picks up data from the two RAMs one by one till both the parameters data weight repeats. Once the repeat stops number of data picks are counted and all the data extract as F-Data, V-Data is written into each of locations and repeat times is written into a location. This consolidated 24 bits are stored into the ROM. The process is repeated till the end of collected data in the RAMs.

III. PORTING OUT SA-RI-GA-MA-PA-DA-NI

Code Data stored in the ROM is logged into the microcontroller in the format as first Byte, Second Bite and Third Byte. Each of these three bytes carry a weight and definition. First byte is Frequency, second byte is Amplitude and third byte is time period of these two parameters to be played. The Controller has dedicated modules for delivering Frequency. Delivered frequency is tuned with the weighted amplitude by defining gain of the amplifier. This is timed to operate for the time period picked up from the third byte.

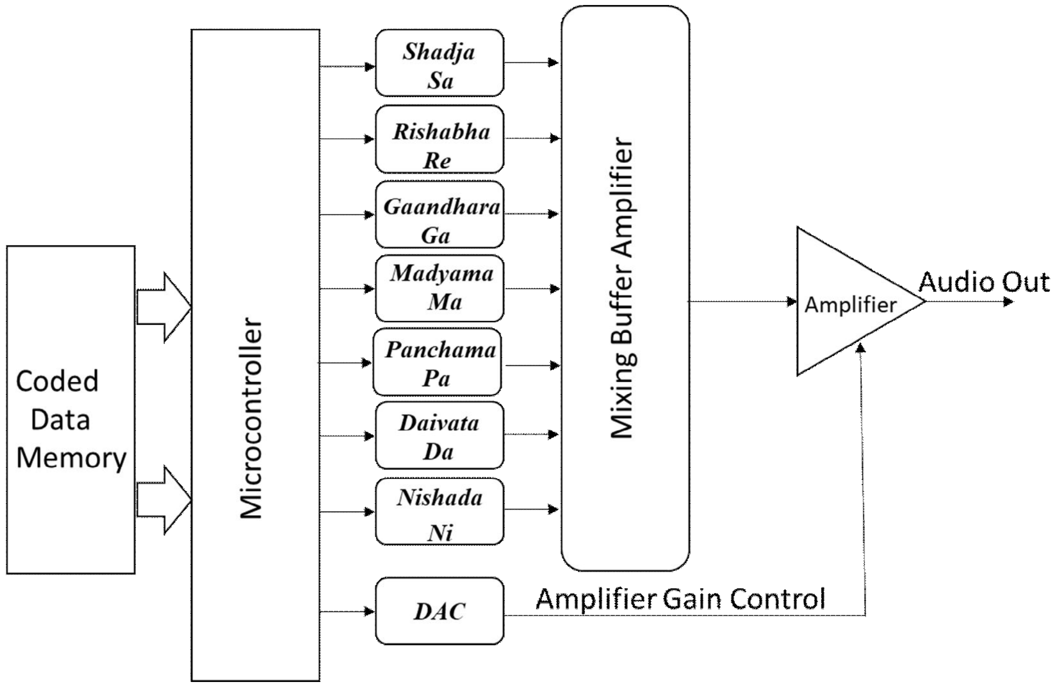

Fig.3 Conversion of Coded Data to Audio signal

Fig.3 is a conversion method to extract coded data in the ROM to an audio signal. Coded data stored in Rom is accessed by the Microcontroller in 3 bytes at a time. First byte defines frequency. The code vectors the range of frequency off the seven notes Sa-Ri-Ga-Ma-Pa-Da-Ni. Dedicated hardware modules are programmed to reproduce that frequency in a bandwidth of 3142Hz in each. 7 such modules have a defined bandwidth of frequency which corresponds to one of Sa-Ri-Ga-Ma-Pa-Da-Ni. The microcontroller vector addresses one of the 7 frequency modules with a code [8]. This produces frequency to the port. All the 7 modules are ported on to a Mixer Buffer Amplifier. Mixer Buffer Amplifier delivers one off the module frequencies to the Amplifier. A DAC is employed to deliver amplitude from the second byte of the code [9]. This tune defines the gain of the Amplifier to deliver code equivalent amplitude of the frequency selected from the modules. This runs till the microcontroller delivers the frequency for the period defined in the third byte of the code data. The flow of code from the ROM to amplifier continues till the end of the file. Since time related to storing all the signal weighted data into the memory is avoided by repeating the same code data for a defined time period. Thus, this occupies very less space in the memory device.

IV. STEREO AND HTS EFFECTS POSSIBILITY

The model is very futuristic and can be coded with Stereo data and HTS of 5.1 as used in present market or higher than 7.1 even. For any of these a single memory can be used but storage style changes. Here L and R are treated with FM stereo system and finally modulated with high frequency.

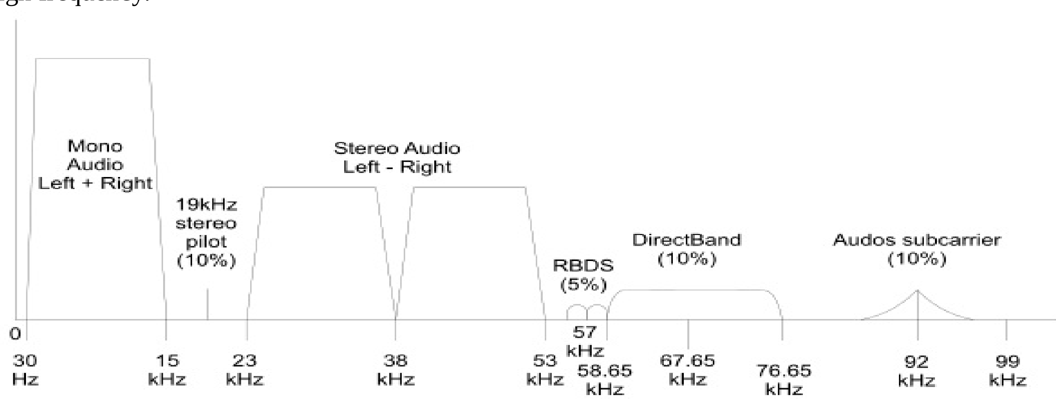

Fig.3 Stereo FM modulation method

Fig.3 Indicates a signal formation of Stereo (L+R) audio signal modulation. This modulated signal is further modulated with Frequency forming FM signal. Generally, FM carrier frequency is from 88MHz to 108MHz. That frequency bandwidth is to size the antenna to small and compact. In this approach we are not intending to transmit audio signal through air. We opt to go with a carrier frequency just 12 times higher than the modulating frequency. So, we go with 264000Hz (2.64MHz). This signal is captured and converted to its weighted digital data into a memory and goes as done with mono audio signal. Later while deriving to audio signal an FM demodulator is used to unpack the stereo channels and drive into two channel amplifiers after gaining tuned amplifier.

V. CONCLUSION

This concept needs extensive testings for its optimum performance and in competition to the present-day audio compression systems. This would give a promising result with respect to its sound quality and lesser memory space.

Vedas do have solutions for any need of man. It needs an exclusive understanding of the slokas which are grass root algorithms for techniques. Scientific face of the slokas are covered under the modern technology for the man has left strings of it to the level that man needs deeper research in the Vedic technologies. Nature speaks a lot close to the Vedic slokas resulting logical solutions for the human search.

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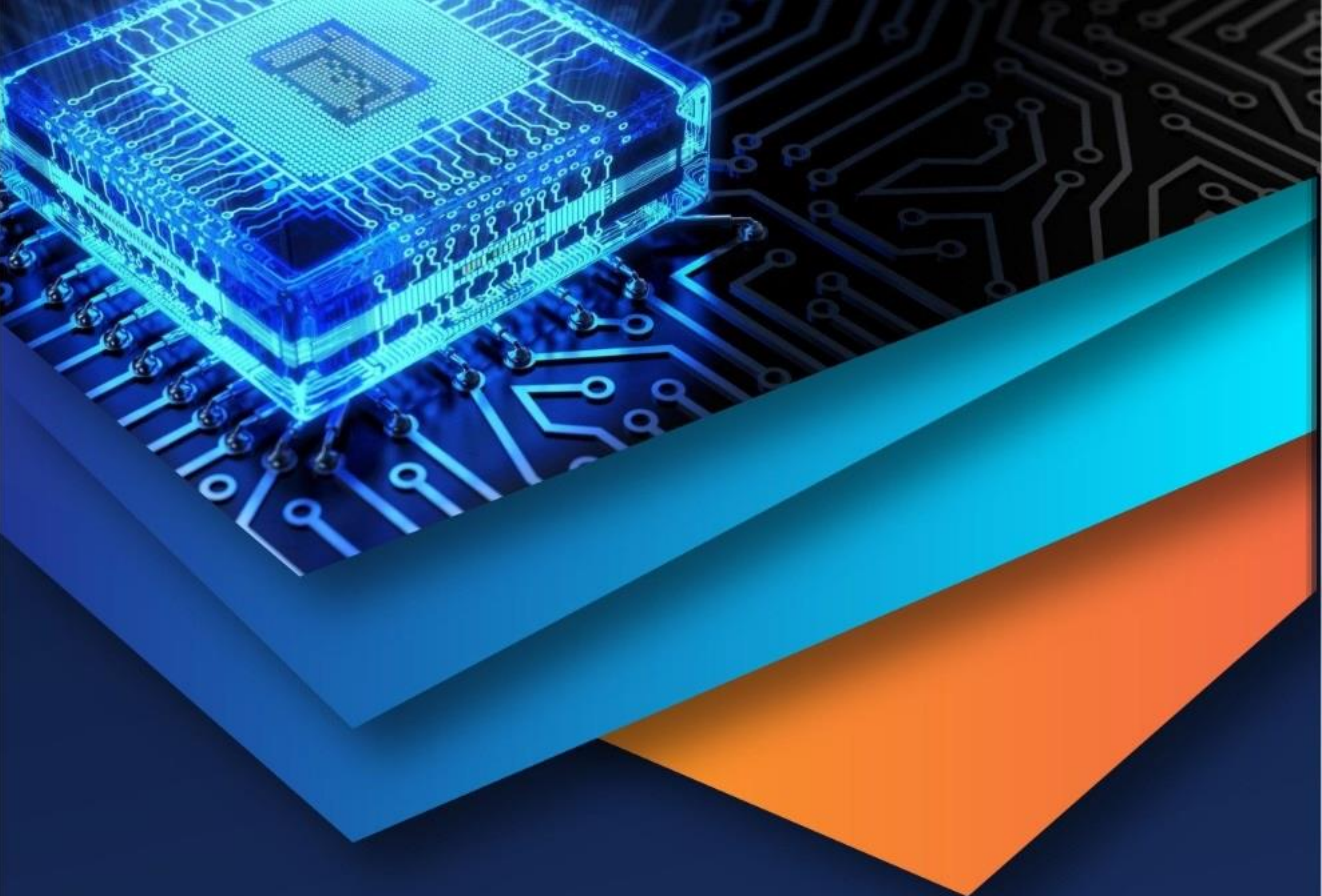

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