Wave Audio Player

Digital Fundamentals

Student Name: ____

Acknowledgements

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Purpose

This lab introduces the student to data contained in computer files and hardware that can make use of the data contained in those files. This knowledge helps the student have a better understanding of how the digital world of computers is a world of 1s and 0s and hardware.

System Rationale

The wave audio player circuit for this lab is a complete system. It is stand-alone except for a power source. The student will first view the complete circuit from a systems perspective, and then delve into each of the "building blocks" that comprise the system. The student will be able to measure the signals between each block and ensure that each block is performing as required.

The student should develop an understanding of the systems found in a simple digital audio player. The player found in this lab plays Wave files. From a block diagram viewpoint, the main difference between this audio player and an MP3 player is that the MP3 player has a MP3 decoder system.

System Concepts

This system covers the following system concepts (signified by an X):

_X_ S1. A system can be defined in terms of its functional blocks i.e., a "structured functional unit."
_X_ S2. A system has a purpose, transforms inputs into outputs to achieve a goal.
_X_ S3. A system is defined by the flow of materials, energy and information, between its functional units.
___ S4. A system may be open or closed. In an open system additional inputs are accepted from the environment.
_X_ S5. A system is more than the sum of its parts. Individual components can never constitute a system.
_X_ S6. A system provides feedback to the operator and services to the user. Some system functions may involve operator action.
_X_ S7. Systems have unique problems.

Student Learning Outcomes

For a full course SLOs, click the link and click SLO tab.
http://www.esyst.org/Courses/DC-AC/_delivery/index.php

1. Define the terms analog and digital and distinguish between digital and analog signals.
2. Represent quantities in binary codes, convert between the decimal and binary number systems. Convert between the hex and binary codes.
5. Explain how binary values are represented with electronic signals.
16. Explain the operation of a binary up counter, down counter, BCD counter, and frequency divider.

Prerequisite Knowledge & Skills

The student should have the fundamental understanding of the following circuits and components.

- Clock Sources
- Digital Counters
- Memory Devices
- Digital-to-Analog Conversion
- DC and AC Circuits

Learning Objectives

Relevant knowledge (K), skill (S), or attitude (A) student learning outcomes include:

K1. To have an understanding that sound and music are represented by 1's and 0's in a computer file system.

K2. To understand how a simple digital music player works and the different systems that are needed to make it work.

S1. Be able to identify the header portion of an audio Wave file.

S2. Be able to identify a clock signal and the importance of it.

S3. Be able to program a memory device such as an EPROM, EEPROM, or FLASH memory chip.

S4. Be able to identify an R2R ladder circuit.

S5. Be able to identify a Class A MOSFET amplifier.

Process Overview

The student will demonstrate lab completion by playing an audio clip via the audio player circuit to his/her instructor. This lab should be performed by each individual student.

Time Needed

Lab Performance:
It should take students approximately four hours to work through the entire lab. Each student should work alone.

Equipment & Supplies

Item	Quantity
Components	
27C020 or 29C020 Memory Chip (DIP version)	1
74HCT4040 Counter	2
74LS14 Schmitt Trigger Inverter	1
.1 uF Capacitor (disc)	2
10 uF Capacitor (electrolytic)	1
100 uF Capacitor (electrolytic)	1
5.6 Ohm Resistor (2 Watt rating or greater)	1
470 Ohm Resistor (1/4 Watt)	1
10 KOhm Resistor (1/4 Watt)	26
33 KOhm Resistor (1/4 Watt)	1
68 KOhm Resistor (1/4 Watt)	1
1 KOhm Potentiometer	1
50 KOhm Potentiometer	1
IRFZ44 MOSFET	1
Speaker (permanent magnet type)	1
5 Volt Power Supply	1
Equipment and Software	
Personal Computer (PC) or Laptop running Windows 2000 or XP	1
Universal Programmer capable of being ran from the PC or laptop stated in the above item. The Universal Programmer must also be capable of programming the 27C020 or the 29C020 Hex Editor software such as HHD Free Hex Editor found on the Internet.	1
Audio Wave Recording software such as Hotkey Sound Recorder found on the Internet.	1
Oscilloscope	1

Special Safety Requirements

There are no safety requirements for this lab.

Lab Preparation

The student should review the specifications sheets for the 74LS14, 74HCT4040, and the 27C020 or 29C020 depending on which memory chip is used. These “spec” sheets can be found on the Internet.

Introduction

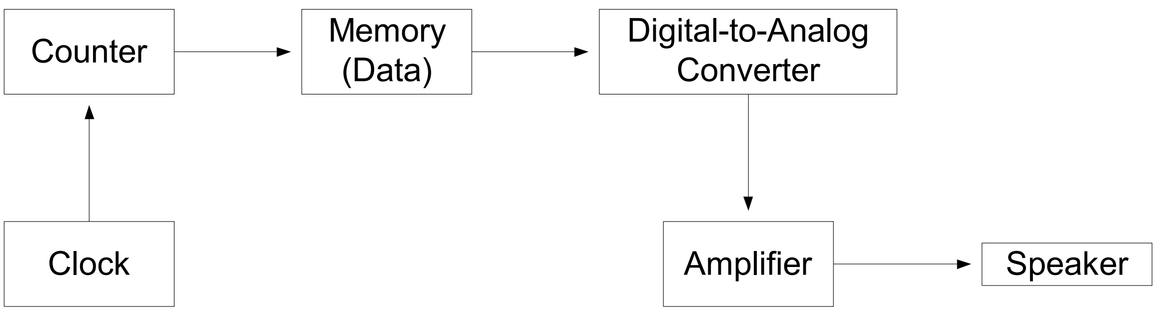
With the advent of computers with audio capabilities, sounds have been captured and digitally stored to be processed and played back later. The digital representation of the captured sound is stored as a computer file. The format of the file depends on the recording software used to capture the sound. There are many different file formats available such as MP3 or Wave. Some recording software allows manipulation of the recording frequency or the number of bits used to represent the sound data; or even if the data will represent a stereo or monophonic signal.

To play back the audio signal from an audio file, the playback software reads a portion of the file, called the header, and extracts the settings of how the file was recorded and also the length of the file. The playback software then configures itself to the same settings and plays back the audio sound. The sound data that is extracted from the file is digital and along the way is converted to an analog signal. The analog signal then is amplified to a sufficient level so that it can be heard.

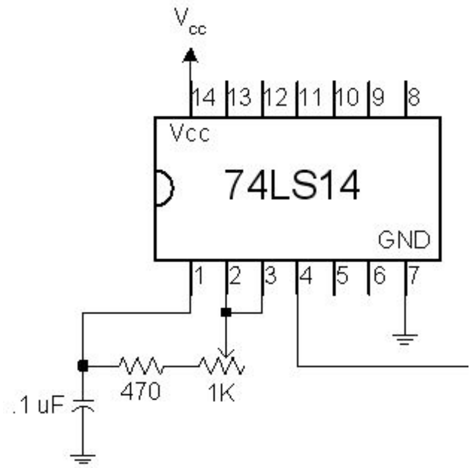
Depending on the file format, the digital data might directly represent the captured sound or it might be a compressed version of the captured sound. If the data is “compressed” then it must first be “processed” either by software or hardware to return it to an “uncompressed” form for playback.

For this lab, the sound recording software will be set to give us data that directly represents the captured sound. The audio data will then be transferred to a memory chip that will then be used in the audio player.

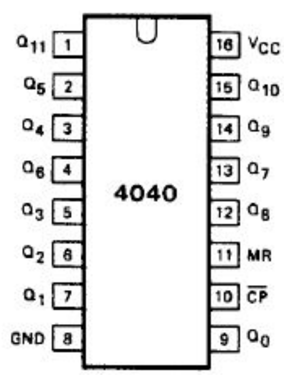
Block Diagram of the Audio Player Hardware



- Clock – The clock provides the TTL square wave to keep the count advancing on the counters. The clock is comprised of a Schmitt trigger inverter (74LS14) with an RC network. The charging and discharging of the capacitor keeps the inverter switching continuously. The rate of charge and discharge is controlled by the value of the capacitor, as well as the value of the feedback resistance. In the diagram below, the 1 KOhm potentiometer allows for the adjustment of the clock frequency. The initial clock signal can be taken off of Pin 2, but in this case is fed into another inverter to ensure smooth level transitions. The final clock output is from Pin 4.



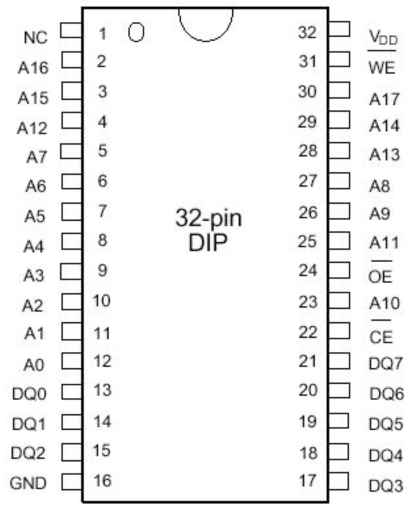
- Counter – The Counter is comprised to two 74HCT4040 12-bit counters in a cascaded fashion. The counters provide the 18 lines needed to fully address all the data bytes on the 27C020 or 29C020. The Counter counts from 00000 Hex to 3FFFF Hex and then repeats over. This is accomplished by connecting output Q6 of the second counter to the Master Resets (MR) of both 74HCT4040s. The Master Reset is an active "high" input pin, so when the Counter reaches 40000 Hex, Q6 of the second counter has just gone "high" causing the Counter to reset back to 00000 Hex. The following diagram shows the pinout for the 74HCT4040.



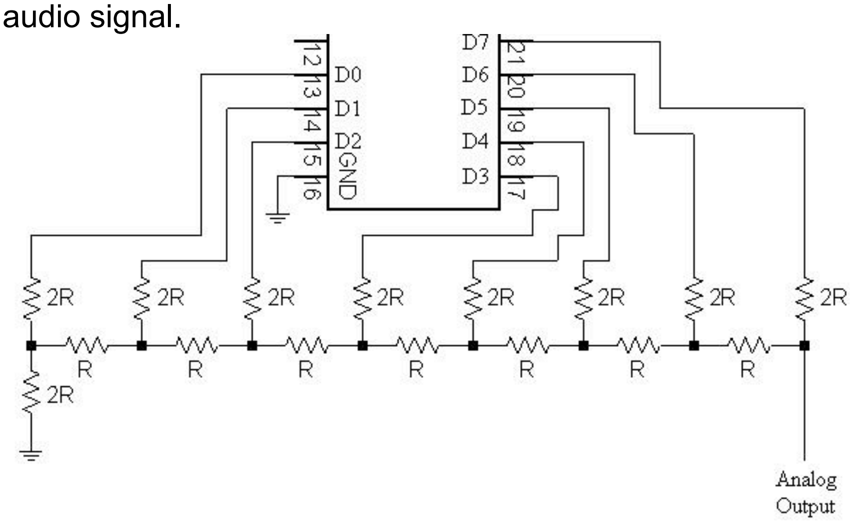
- The 27C020 or 29C020 memory chip stores the audio data to be retrieve through the data lines by the "cycling" of the address lines by the two 74HCT4040 counters. Both chips are 256 Kwords by 8 bits per word. The exact number of bytes contained in the memory chip is 262144. This number is arrived at by taking 2 to the 18th power, where 18 is the number of address lines the chip has.

The 27C020 is an EPROM memory chip while the 29C020 is a FLASH memory chip. The advantage of the EPROM is that they are very inexpensive since most new designs do not use them anymore. The disadvantage is that a Ultra-Violet (UV) Eraser is needed to erase the EPROM chip before it can be programmed with the Universal Programmer. The advantage of the FLASH memory chip is that it can be erased and programmed with the Universal Programmer but they normally run about twice the price of a comparable EPROM version.

The following pinout diagram can be used for both the 27C020 and the 29C020. When the chip is in the audio player circuit, the data outputs are always active due to the WE (active low) pin being tied "high" and the OE (active low) and CE (active low) pins being tied "low".

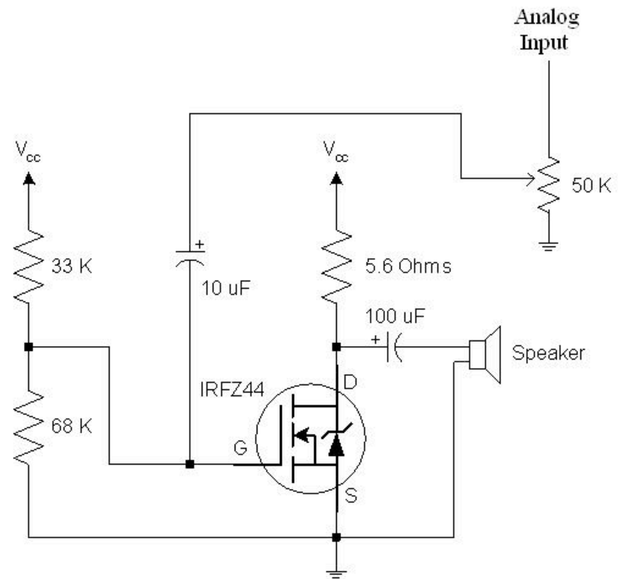


- Digital-to-Analog Converter – The 8-bit data bytes provided by the memory chip are then "fed" into an 8-bit R2R Digital-to-Analog Converter which provides an analog audio signal.



- Amplifier – The analog audio signal from the Digital-to-Analog Converter is amplified to a respectable level by a Class-A MOSFET amplifier. The active component in this circuit is an IRFZ44 MOSFET. Other MOSFETs could be used as well such as IRF510 or IRF511. Class A amplifiers are not power efficient but they are simple and work well. The power inefficiency comes from the transistor/s being biased halfway between fully on and fully off. Therefore, even if nothing is being amplified, there is a constant flow of current between the Drain and the Source of the MOSFET. The biasing of the transistor is controlled by the voltage divider formed by the 33 KOhm and 68 KOhm resistors. The AC signal to be amplified is introduced at the Gate terminal while the amplified AC signal is taken out at the Drain terminal. The 10 uF and 100 uF capacitors allow AC signals to pass while maintaining DC isolation.

The 50 KOhm potentiometer is used as a volume control for the amplifier.



- Speaker – The amplified AC signal is then reproduced into sound by the permanent magnet type speaker.

Audio Wave Recording Software Settings
The audio player circuit provides playback of 8-bit sound data per a single channel, therefore the recording settings must be set for 8-bit mono sound.

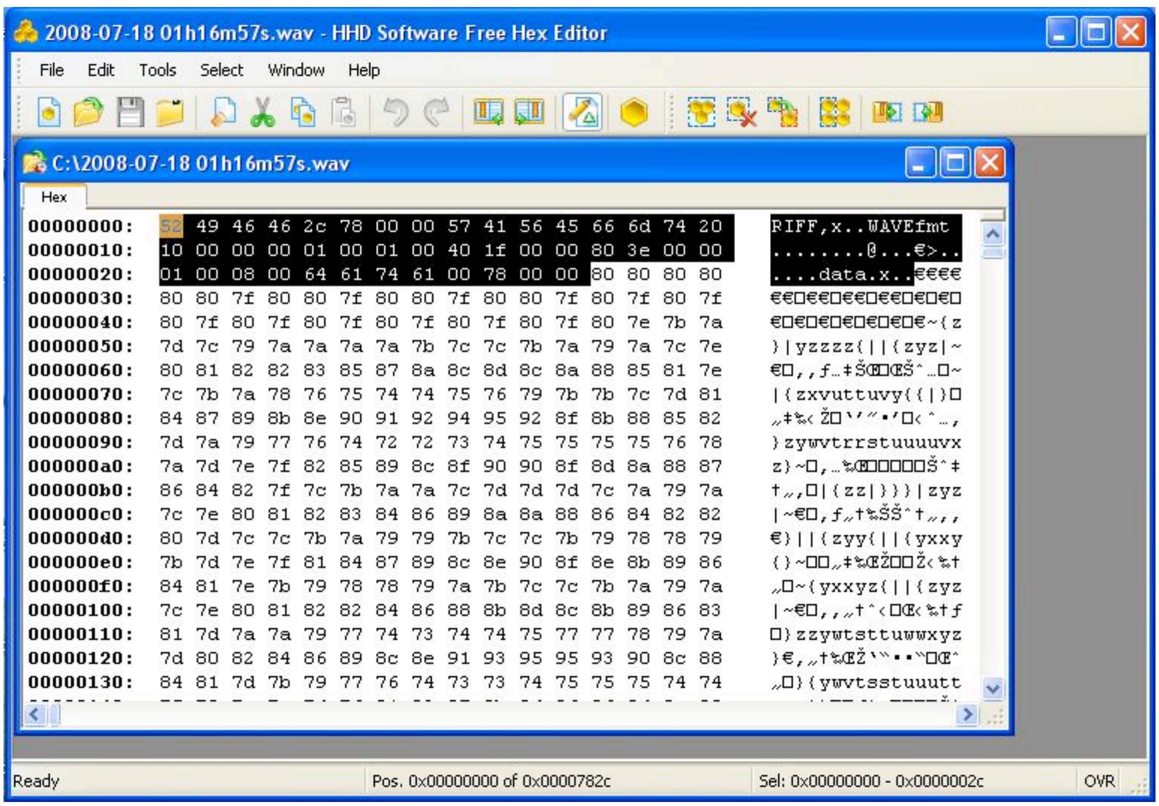
The sampling rate is also very important and will directly impact the quality of the recording. As the sample rate increases, so does the quality of representation of the original signal. Voice quality recordings are usually made at 8 KHz while CD quality recording are made at 44.1 KHZ.

A limiting factor of the audio player is the total number of samples that the memory chip can store which is 262144. If a recording was made at 8 KHz, which is 8000 samples per second, then the playback time would be about 32 seconds worth. If a recording was made at a higher sampling rate such as 32 KHz, which is 32000 samples per second, then the playback time would be about 8 seconds worth. The playback would sound better because a higher sample rate would represent the original audio signal more accurately, but the playback time would suffer. Keeping these constraints in mind, the audio player was designed to play audio recorded at 8 KHz and at 16 KHz, providing a playback time of approximately 32 seconds and 16 seconds, respectively.

Wave File Header

When a sound recording is made and saved as a Wave file, the file contains a "header" that holds information on how the recording was made: sampling rate, file length, stereo or mono, etc. This information is needed so that a player can set itself on the appropriate settings and play back the sound recording appropriately.

The audio player in this lab is not able to read the header and only requires the audio data found in a Wave file, therefore the header must be removed from the file. The hex values that make up the header and audio data of the file can be viewed and edited with a hex editor. The following picture shows an audio file being viewed with HHD Software Free Hex Editor. The header portion of the file is seen "high-lighted" and is found at address 0x0000 Hex thru 0x002D Hex. This portion of the Wave file must be removed. Once the file header is removed, the remaining data will automatically be renumbered in the hex editor viewing screen. The remaining data is the audio data.

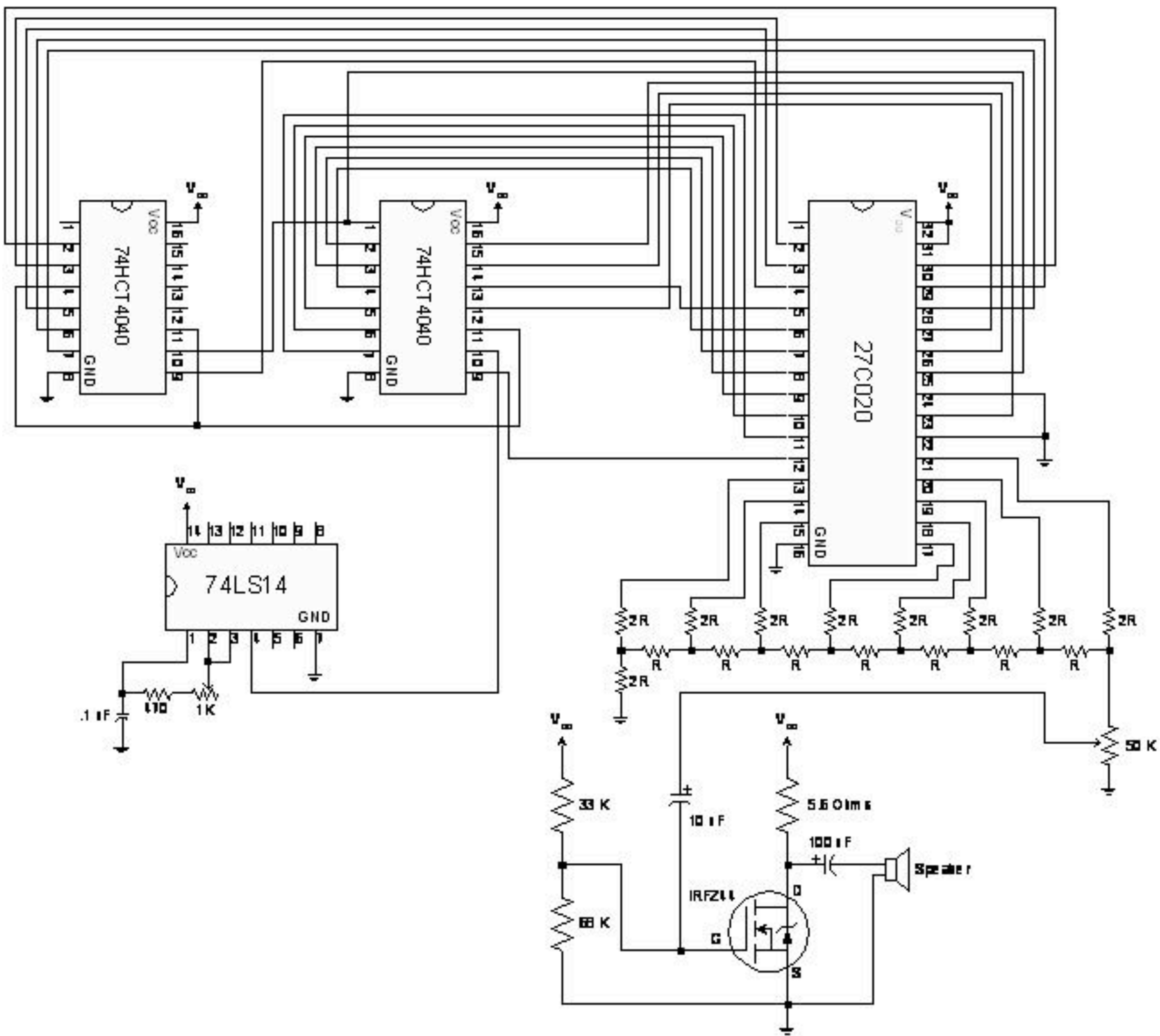


To completely fill the storage capacity of the 27C020 or 29C020, audio data should be found from 0x000000 Hex thru 0x03FFFF Hex. Any data past this range must be removed.

Task

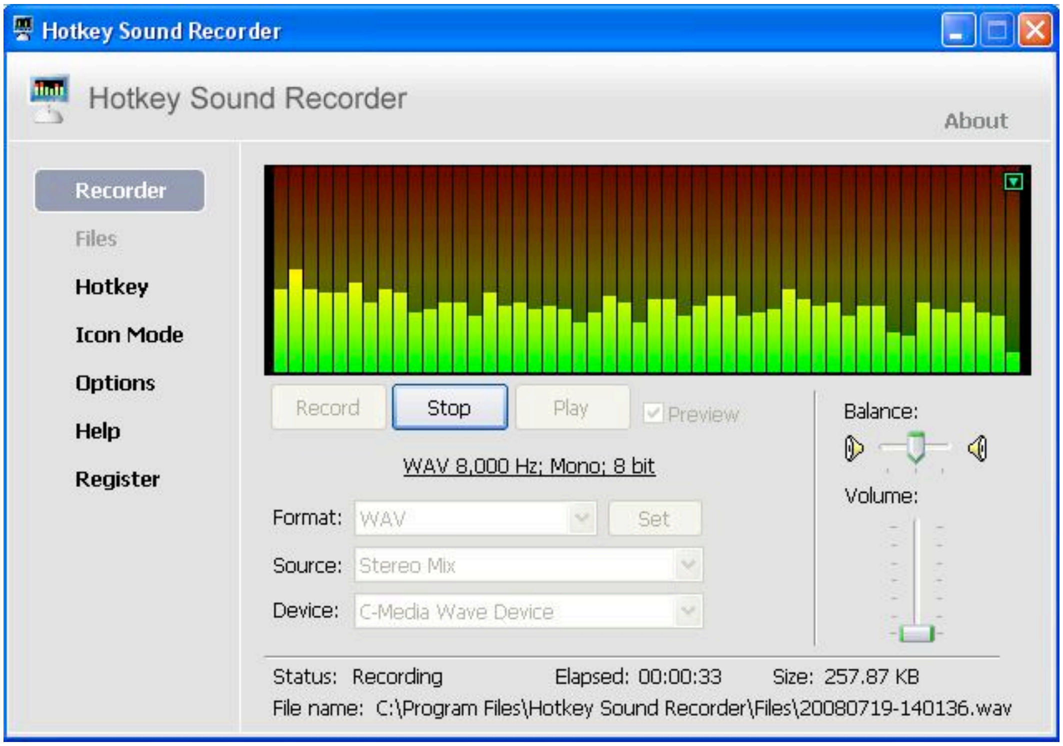
The task for this lab is first to program a memory chip with audio data and then to build or assemble the circuit shown in the following diagram (Audio Player Schematic). The construction of the circuit will be done on a block-by-block basis with the student verifying the output of each block. The lab is completed when the instructor and student are able to hear the wanted audio signal being played through the audio player.

Audio Player Schematic



Performance

1. Using recording software such as Hotkey Sound Recorder, record at least 33 seconds of music with the settings shown in the following picture. When the Stop button is pressed, a Wave file will be created and can be found at the location shown at the bottom of Hotkey Sound Recorder window.



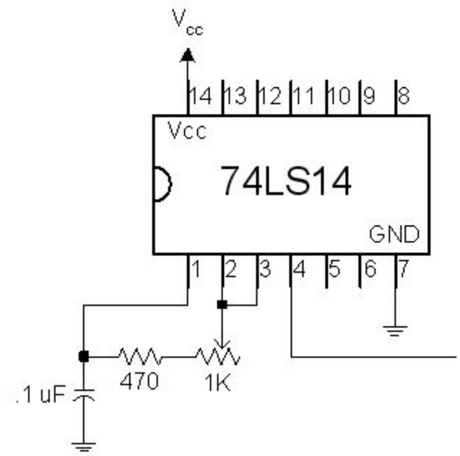
2. Use a hex editor such as HHD Software Free Hex Editor to remove the header of the audio file created in Step 1.

   The remaining data is the audio data. Ensure that the audio data ranges from 0x000000 Hex thru 0x03FFFF Hex. Remove any data beyond the 0x03FFFF address. If there is not enough data, then record another sound clip that exceeds 256KB.

   Save the audio data of the range 0x000000 thru 0x03FFFF as a file with the ".bin" extension (i.e. music.bin). The ".bin" extension is merely a reminder that it is a binary file. Naming it as such will not alter the contents or add to the file.

3. Use a Universal Programmer to program a blank 27C020 or 29C020 with the data file created from Step 2. There are many programmers available and software as well. Refer to the programmer's instruction manual for assistance if help is needed. The data file from Step 2 should be loaded as a Binary file, not Hex.

4. Assemble the following clock circuit. The supply voltage is 5 Volts for this circuit, as well as the entire audio player.

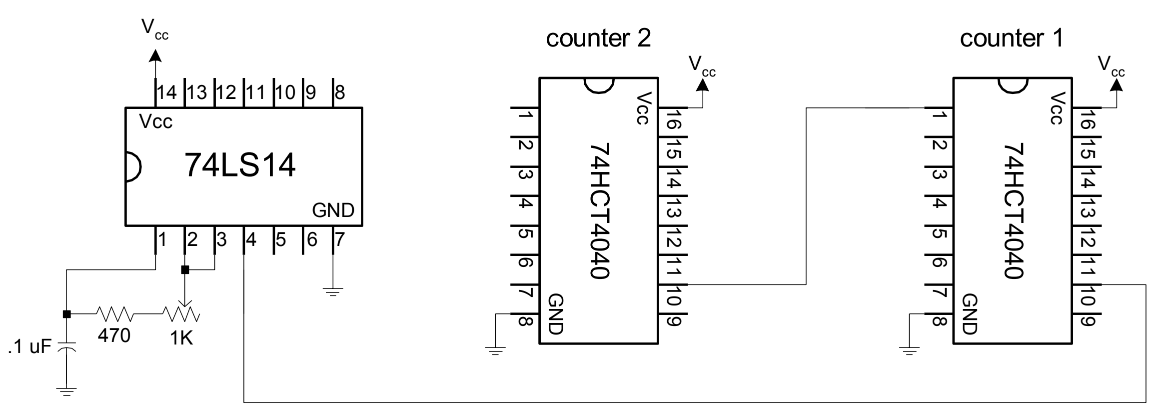


5. Use an oscilloscope to monitor the output signal at Pin 2. Is there a square wave present? _______________
   If "Yes", proceed to the next Step.
   If "No", check the circuit wiring and/or replace components until a square wave signal is seen at Pin 2.

6. Use an oscilloscope to monitor the output signal at Pin 4. Is there a square wave present? _______________
   If "Yes", this signal should be the inverse of the signal seen in Step 2. Proceed to the next Step.
   If "No", then check the circuit wiring and/or replace components until a square wave signal is seen at Pin 4.

7. Set the clock frequency to 8 KHz by adjusting the 1 KOhm potentiometers. Verify the frequency by using the oscilloscope and then power down the circuit.

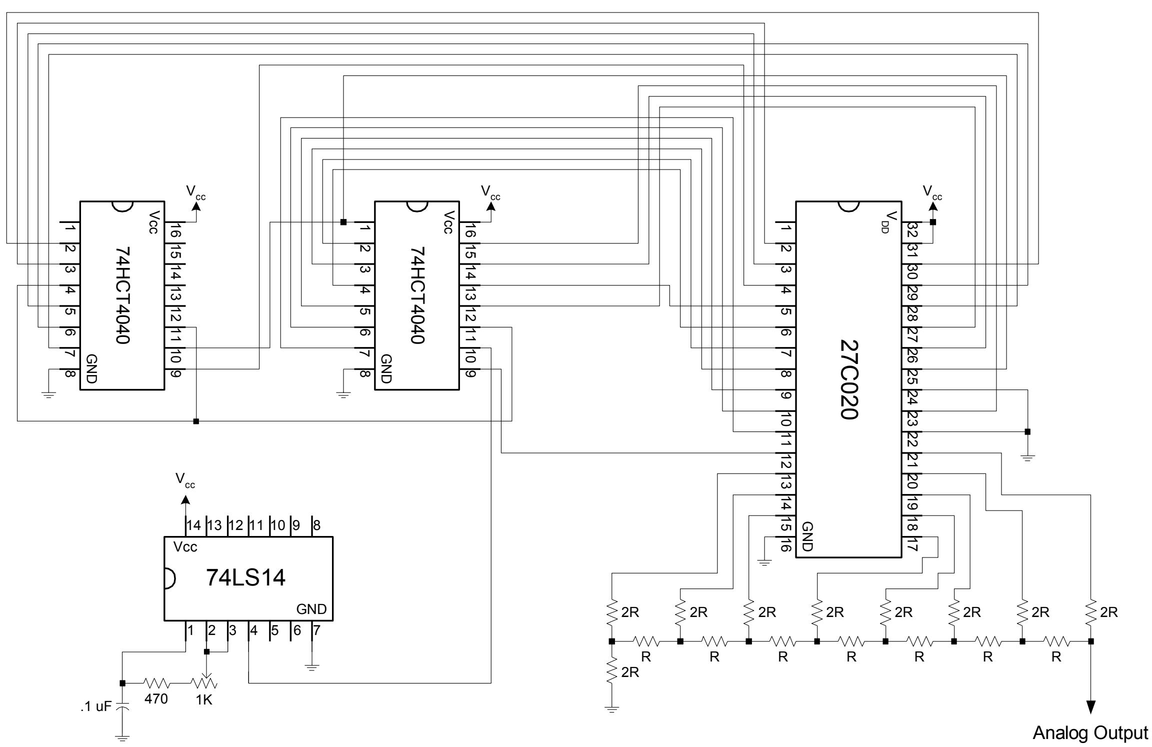
8. Add the Counter circuit to the clock circuit as shown in the following diagram.



9. Power up the circuit shown in Step 8 and use the oscilloscope to verify that the Counter is working properly. The following table indicates the frequencies that should be found at the corresponding pins to each of the counters, provided that the clock frequency is still 8 KHz.

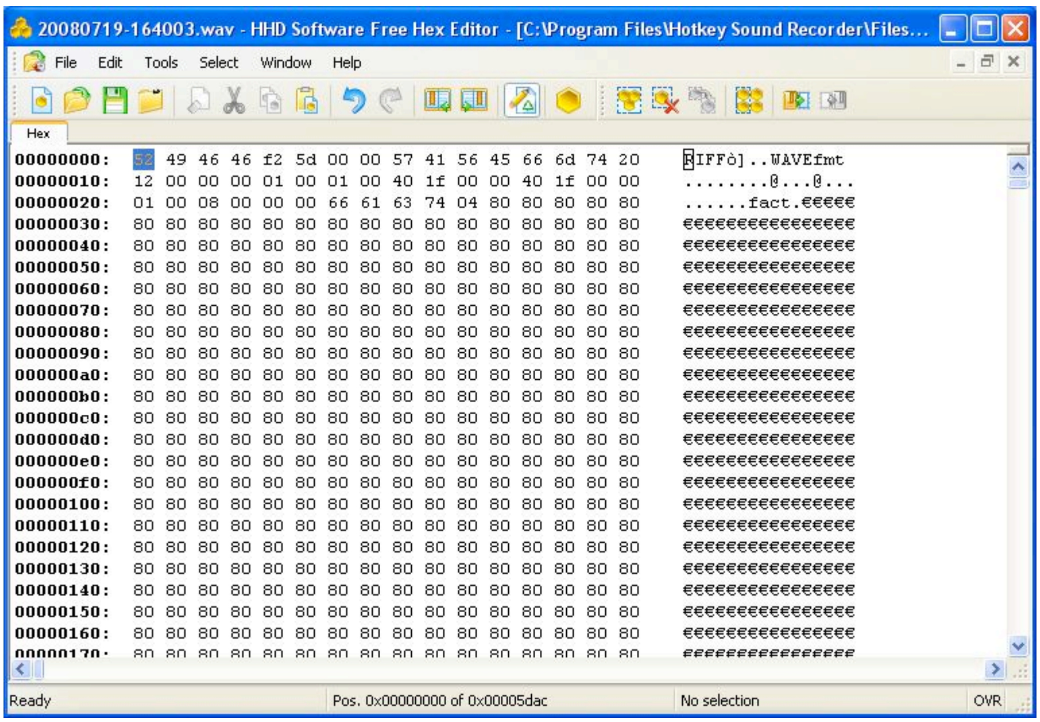
Pins	Counter 1 Frequencies in Hertz	Counter 2 Frequencies in Hertz
Q0	4000	.976
Q1	2000	.488
Q2	1000	.244
Q3	500	.122
Q4	250	.061
Q5	125	.030
Q6	62.5	.015
Q7	31.2	Output not needed
Q8	15.6	Output not needed
Q9	7.81	Output not needed
Q10	3.90	Output not needed
Q11	1.95	Output not needed

10. Power down the circuit and continue assembling the Audio Player by adding the memory chip and Digital-to-Analog Converter (R2R Ladder). Use the following diagram.



11. Power up the circuit shown in Step 10 and then visualize the analog output on an oscilloscope. The output should be seen “dancing” on the oscilloscope. What is the voltage that the “dancing” is centered on? _______________

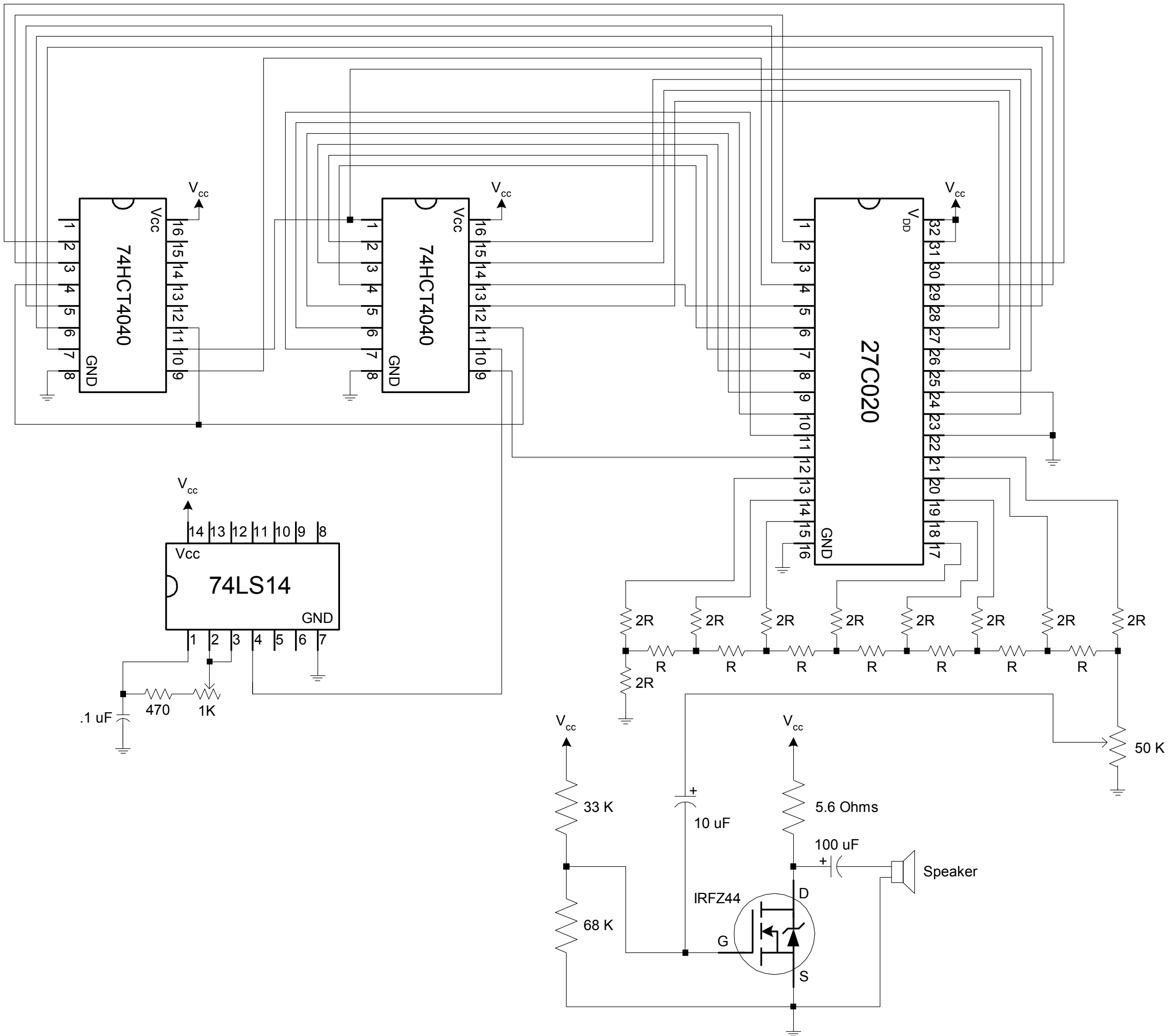
A point to make is that the analog audio signal is centered on 2.5 Volts. This is because the audio recording made was 8-bit, which can vary between 00 Hex and FF Hex. For the 5 Volt Audio Player, 00 Hex represents 0 Volts, and for FF Hex it represents 5 Volts. An audio signal must be free to swing up or down in amplitude. The best place to center the audio signal is halfway between the two extremes. Halfway between 00 Hex and FF Hex is 80 Hex which represents 2.5 Volts. The following picture is the Hex content of a blank 8-bit Wave file. Notice that after the header, the audio data is 80s in Hex.



The “dancing” viewed on the oscilloscope only proves that there is a varying analog, it does not mean that it is the desired signal. To verify that the wanted audio signal is present, a small permanent magnet speaker may be placed at the Analog Output and referenced to ground. If all is well, the wanted audio signal may be heard, though it will probably be at a very low volume.

Power the circuit down. If the wanted audio signal was heard, proceed to Step 12. If no sound was heard or if the sound heard was distorted and not recognizable check the wiring connections and retry. Do not proceed until the problem has been resolved.

12. Complete the Audio Player circuit by adding the amplifier stage and speaker. Use the following diagram.



13. Power up the Audio Player. Using the 50 KOhm potentiometer, adjust the volume. Do you recognize what is being played? _______________
If "Yes", then the circuit is working properly. Proceed to Step 14.
If "No", check the connections of the amplifier. Do not proceed until the problem has been resolved.

14. Use the oscilloscope to view the signal at the Gate of the MOSFET. You should see a DC bias voltage with an AC sound signal from the Analog-to-Digital converter riding the DC bias voltage. Adjust the volume if necessary to increase the AC sound signal. Do not adjust again.
What is the voltage of the DC bias? _______________
What is the overall voltage amplitude of the AC sound signal? _______________

15. Use the oscilloscope to view the signal at the speaker. An AC sound signal centered on 0 Volts should be seen. What is the overall voltage amplitude of the AC sound signal? _______________

16. What is the approximate AC voltage gain of the amplifier? _______________
Use the following equation.

$$\text{AC Voltage Gain} = \frac{\text{AC Voltage Out}}{\text{AC Voltage In}}$$

17. Based on the information found in Step 16, what type of amplifier is this? Is it most likely a voltage amplifier or a current amplifier? _______________
If needed, go back and view the amplitude of the signal at the Digital-to-Analog converter output.

Deliverable

The student is to demonstrate a working circuit to their instructor.

Grading

Your instructor will let you know how this lab will be graded.