WORKREADY ELECTRONICS

lab activity

Digital Signal Processing

Acknowledgements: Developed by JD Neglia, P.E., Electronics Program Director at Mesa Community College, Mesa, Arizona.

Lab Summary: This laboratory experiment introduces practical concepts associated with digital signal processing (DSP) and DSP hardware. Measurements will be made and calculations will be performed to illustrate the capabilities of a state-of-the-art DSP chip and to provide real hands-on experience with these devices.

Lab Goal: The goal of this lab is to build a DSP circuit, observe its operation and limitations, and perform measurements of its performance in filtering audio signals.

Learning Objectives

1. Assemble a DSP demonstration circuit while following accepted ESD practices.
2. Thoroughly test the circuit to verify basic functionality and ADC and DAC linearity.
3. Observe the circuit limitations for dynamic range and aliasing.
4. Apply audio frequencies to the DSP circuit, and measure and plot the circuit frequency response over the entire audio spectrum for each of the following filters: all-pass, low-pass, high-pass, and band-pass.
5. Apply music signals from a CD or MP3-player to the DSP circuit, and observe the audible effects of the circuit in the all-pass, low-pass, high-pass, and band-pass modes.

Grading Criteria: Your grade will be determined by your performance in the following aspects of this exercise:

1. Circuit construction (neatness and ESD practices)
2. Troubleshooting
3. Measurements and calculations
4. Graphing results
5. Answering lab questions

Time Required: 6 - 7 hours. This exercise should be performed in three parts: 1) Circuit Construction, 2) Troubleshooting and preliminary measurements, and 3) Measurements, graphing, and analysis.

Special Safety Requirements

Static electricity can damage the DSP device used in this lab. Use appropriate ESD methods to protect the devices. A grounded wrist-strap is provided in the parts kit for this circuit. Be sure to wear it at all times while handling the electronic components in this circuit. The wrist strap need not be worn after the circuit construction is complete.

No serious hazards are involved in this laboratory experiment, but be careful to connect the components with the proper polarity to avoid damage.

Lab Preparation

- Read the WRE DSP Narrative Module.
- Read this document completely before you start on this experiment.
- Acquire required test equipment and appropriate test leads.
- Gather all circuit components and the three panel solderless breadboard.
- Print out the laboratory experiment procedure that follows.

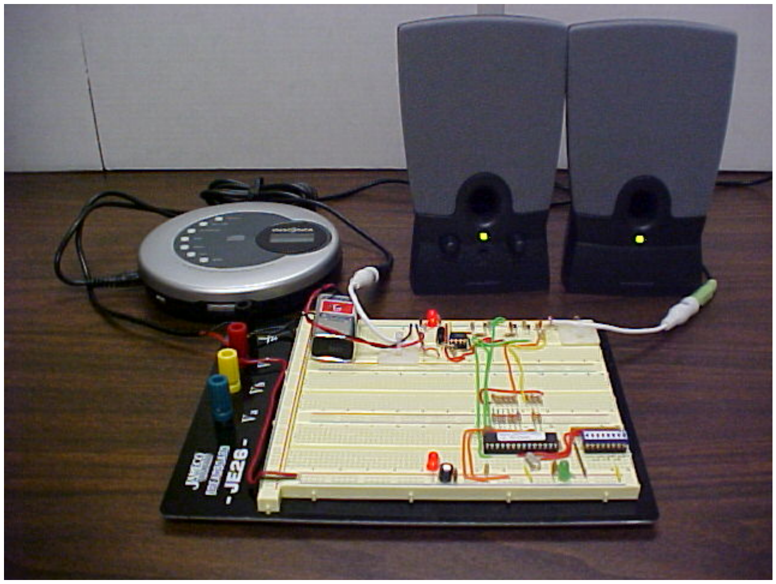
Equipment and Materials

Each team of students will need the test equipment, tools, and parts specified below. Students should work in teams of two.

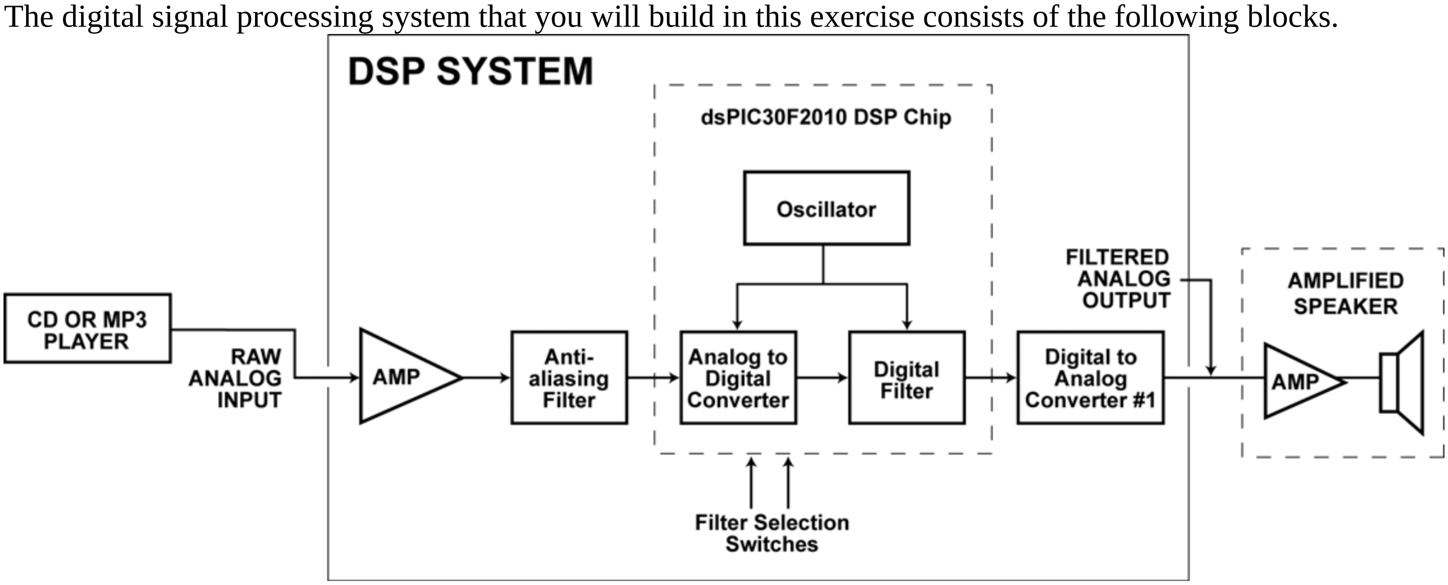
Test Equipment and Power Supplies	Quantity
Digital Multimeter	1
Fixed (not variable) 5 Volt, 500 mA Regulated DC Power Supply	1
Oscilloscope (Dual trace, 20 MHz or greater)	1
Function Generator (50 kHz or greater)	1
Frequency Counter	1
CD or MP3 player with headphone output	1
Audio Cable (standard 3.5 mm mini-plug to 3.5 mm mini plug), 12” or longer	1
Amplified Computer Speaker (with standard 3.5 mm mini-plug)	1
Battery, 9V	As needed
Clip, 9V Battery	As needed
Tools	**Quantity**
Needle-nose Pliers	1
Wire Strippers	1
Wire Cutters	1
ESD Wrist Strap	2

Parts and Hardware	Quantity
Digital Signal Processor: dsPIC30F201030ISP	1
Socket: ED3328 300 DIP, 28-Pin, Mach--	1
Crystal: X076. 7.3728 MHz, Parallel	1
OpAmp: 1 LM358	1
LED, Red: LH3330	2
LED, Green: 1 LG3330	1
Diode, Zener, 5.1V: 1N751	1
Diode, Rectifier: 1N4001	1
Resistor, 1/4 W, 1 kOhm, 10% R: RA1.0K	2
Resistor, 1/4 W, 100 Ohm, 10%: RA100	4
Resistor, 1/4 W, 470 Ohm, 10% R: RA470	1
Resistor, 1/4 W, 820 Ohm, 10%: RA820	2
Resistor, 1/4 W, 4.7 kOhm, 10%R: RA4.7K	1
Resistor, 1/4 W, 8.2 kOhm, 10%: RA8.2K	2
Resistor, 1/4 W, 10 kOhm, 10%: RA10K	5
Resistor, 1/4 W, 22 kOhm, 10%: RA22K	1
Resistor, 1/4 W, 1.1 kOhm, 5%	7
Resistor, 1/4 W, 2.2 kOhm, 5%	9
Potentiometer, 5 kOhm: 3299Y-502	1
Capacitor, 0.001uF: DC.001	2
Capacitor, 0.01uFDC.01	2
Capacitor, 0.1uF, 25 V: DC.1/25	6
Capacitor, 10uF, 25 V: A10J/25	5
Capacitor, 100uF, 16V: A100J/16	1
Breadboard, Solderless, 3 panel: JE26	1
Foam, Conductive	As Needed
Box, Storage (12x9x2): 072750	1
Cable Ties	As Needed
Cable Tie Mounts: 2 HC-101	2
Mono F Jack, 3.5mm: PJ0354	2
Wire, Solid, 22AWG, Red: 122R/C	As Needed
Wire, Solid, 22AWG, Blk: 122BK/C	As Needed
Wire, Solid, 22AWG, Yel:122Y/C	As Needed
Adhesive Mounting Pad, 1" square	As Needed
DIP Switch, 8-position	As Needed

Introduction



The digital signal processing system that you will build in this exercise consists of the following blocks.



As shown above, a raw analog audio signal is first amplified and then filtered with an anti-aliasing filter. The resulting signal is applied to the DSP chip. The DSP chip in this system is a Microchip dsPIC30F2010. This chip includes an analog-to-digital converter which samples and converts this input signal to digital at a rate determined by the internal oscillator. This digitized input signal is then processed with one of several digital filter algorithms that have been pre-programmed into your chip. You may select which filtering algorithm is used by connecting two pins on the chip to the high or low supply rails (+5.0V or 0.0V) according to the following table:

Filter Type	Pin 11	Pin 12
Band Pass	0 V	0 V
High Pass	5 V	0 V
Low Pass	0 V	5 V
All Pass	5 V	5 V

The result of the filtering operation is then converted back to analog with a digital-to-analog converter (DAC). The DAC is an R-2R ladder connected directly to the pins of the DSP chip.

The filtered analog output signal is then sent to an amplified speaker.

In this exercise, before we filter audio signals, we will first measure and graph the ADC and DAC transfer function, and the frequency response of each of the filters (BPF, LPF, HPF, and APF) to verify proper operation. Then, we will filter music from a CD- or MP3-player with this system and observe the filtering effects on the music.

<u>Lab Procedure</u>

CONSTRUCT THE CIRCUIT

Note: The schematic and photos showing the layout are at the end of this procedure.

Construct the circuit shown in the schematic diagram at the end of this procedure. Note that this is a substantial task and will take some time to do correctly. Pay special attention to ESD procedures, and be sure to wear a grounded wrist strap. You may find the following suggestions helpful:

1. Use a highlighting marker to mark off each component and circuit node after it is connected.
2. To help protect the DSP chip, install the SOCKET into the solderless breadboard while wiring the circuit. Leave the DSP chip in its conductive foam until the rest of the circuit is built and checked.
3. Because this circuit contains both analog and digital signals, it is important to isolate the two as much as possible. The DSP chip is operating at an internal frequency of almost 120 MHz. This generates radio frequency interference which can add substantial noise to the analog portion of your circuit. Therefore, build the analog portion of the circuit on one panel of the solderless breadboard, and the digital/DSP section on a separate panel. The dashed lines in the schematic diagram show how to separate the circuits. See the circuit photograph to see a suggested layout.
4. Note that the analog circuits and the digital circuits do not share the same power and ground signals. The digital/DSP portion of the circuit is powered with the fixed 5 volt regulator, and the analog portion is powered with a 9 volt battery. This is necessary, as the high speed logic in the DSP induces noise on the power supply. This noise is difficult to remove and will corrupt the analog signal. The 9 V battery provides pure DC which is relatively noise-free. The two circuits, however, must share a common reference "ground" point. Thus, the two grounds are connected together at one point only: at the binding post where the 5.0 V power is brought onto the board.

5. Keep the signal wires and component leads as short as possible as you build this circuit. Neatness counts. Long leads in the digital circuit act as transmitting antennae, and long leads in the analog circuit act as receiving antennae, adding noise to your signals.
6. Due to their geometry, solderless breadboards inherently add capacitance which couples all the nodes of your circuit together. This also acts as a corrupting influence on your analog signals. Therefore, be sure to separate your analog and digital circuits with as much distance as is practically possible.
7. Be sure to provide strain-relief for all the wires that go off the board. Tie them down with mounting pads and cable-ties. Also, affix the 9V battery to the solderless breadboard with an adhesive mounting pad. Leave enough space to attach and remove the 9V battery connector after the battery is attached to the board.
8. The pins of the DSP chip are usually flared out by the manufacturer for automatic insertion equipment. These pins will need to be bent inward slightly to allow the chip to fit into its socket. Do this by holding the ends of the IC (one in each hand), and pressing a row of pins down against a conductive surface, such as a sheet of aluminum foil on a desk. (Do not forget to wear your wrist strap.) When inserting the chip, be sure that no pins are bent underneath the package. “Bent under” pins can be easily overlooked, so check carefully. Finally, be certain of the pinout of the chip before inserting it.
9. The crystal oscillator must be located adjacent to the pins 9 and 10 of the DSP chip.
10. After the circuit is completely assembled, have an unbiased third party check the wiring against the schematic diagram. It will be helpful here to make a new copy of the schematic that can be marked up with a highlighting marker as each connection is checked.

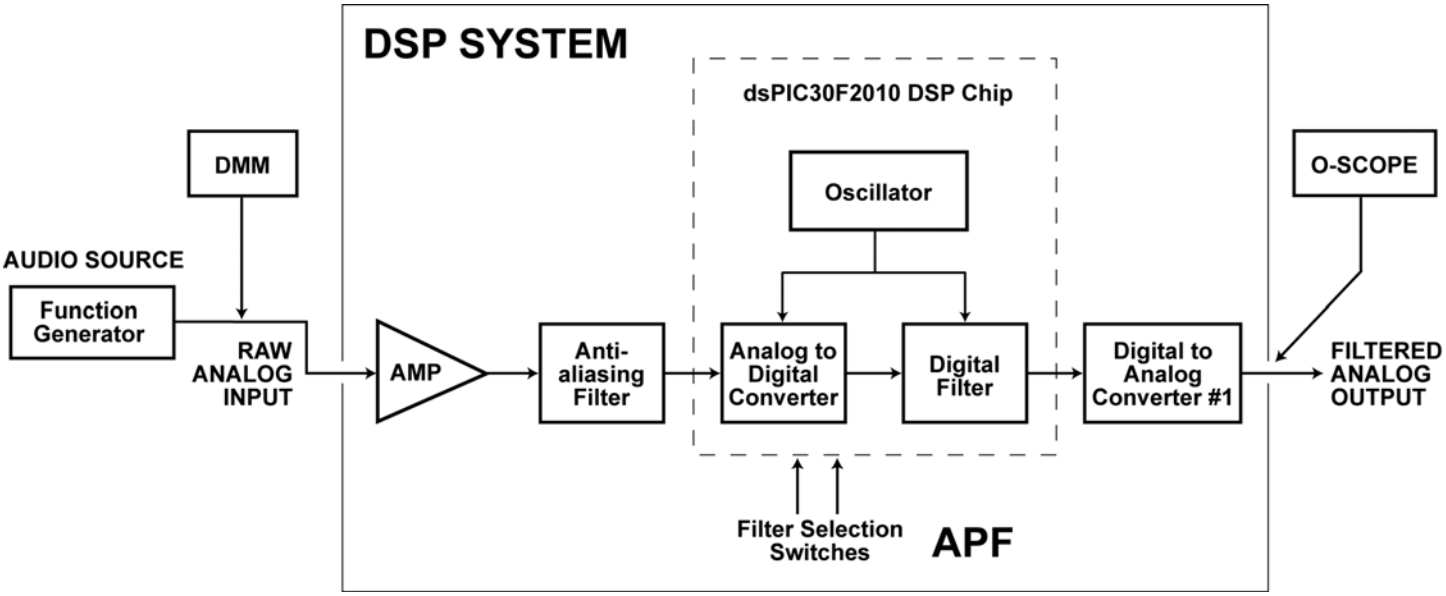
STATIC TEST

1. First, predict the voltages that you would expect to be present at the following nodes if the circuit is operating correctly: (Assume no external equipment is connected.) Each of these nodes should have pure DC signals.

	Predicted (V)	**Measured (V)**
Node B		
Node C		
Node D		
Node E		

2. Without any equipment connected, apply power to the circuit by turning on the 5V regulated power supply and connect the 9V battery. The two red power indicator LEDs should illuminate. If they do not, check the power connections to your circuit. The green "alive" LED should blink at a few Hertz. If it doesn't, then check the following DSP pins:
   a. Pin 1: should be +5 V.
   b. Pin 9: should have a 7.37 MHz sine wave at about 1 V peak to peak. (Measure this pin with a 10X oscilloscope probe only.)
   c. Pins 8, 19, and 27 should be at 0.0V.
   d. Pins 13, 20, and 28 should be at 5.0 V.
3. Now that power is applied, measure the voltage at Nodes B, C, D, and E and complete the table above. Be sure your measurements match reasonably well with your predictions before proceeding.

ADC/DAC STATIC TEST:



1. First, be sure the DSP is in “All Pass Filter” mode.
2. Using an oscilloscope or a frequency counter, measure the frequency on DSP chip Pin 16. This is the sampling frequency of this system. Record below.

   Sampling Frequency: ___________________

3. Take the voltage at Node E (measured and recorded above), and convert it to digital as follows:

$$N = \frac{V_E}{5.0V} \bullet 255$$

4. Express N in decimal: __________________
5. Express N in binary: __________________
6. Now, measure each of the following pins of the DSP. Record the state of each pin as “1” or “0” in the space below.

	MSB							LSB
DSP Pin	21	22	23	24	25	26	17	18
Level								

7. Measure and record the DAC output voltage at Node F.

8. Measured Vout = ______________
9. Temporarily jumper Node E to ground, and measure V_{out}. Verify that it is approximately equal to 0.0 V.
10. Remove the temporary jumper to ground, and then jumper Node E to +5V. Verify that V_{out} is now approximately equal to + 5V.
11. Remove the temporary jumper.

ADC/DAC DYNAMIC TEST

1. With no power applied to your DSP system, temporarily remove the LM358.
2. Install a temporary jumper between Node A and Node E. This jumper bypasses the AC coupling capacitor and input amplifier and allows you to apply DC signals to the ADC.
3. Connect a function generator to Node A. Set the amplitude to zero, and enable the DC offset function. (This is usually accomplished by pulling on a knob.) This allows the function generator to output DC signals.
4. Apply power to the system.
5. Set the DIP switches to ALL PASS FILTER mode.
6. Using a DMM, set the function generator to output a 2.50 VDC signal.
7. Using an oscilloscope, observe the output voltage at Node G. You should observe a DC signal of 2.50 VDC +/- 20%. If not, check the wiring of the R-2R ladder. If the wiring is correct, observe the output of each DSP pin and verify that each switches between hi and low as you vary the input voltage from the function generator. When V_{in} = 2.50V, the bit pattern should be as follows:

	MSB							LSB
DSP Pin	21	22	23	24	25	26	17	18
Level	1	0	0	0	X	X	X	X

or

	MSB							LSB
DSP Pin	21	22	23	24	25	26	17	18
Level	0	1	1	1	X	X	X	X

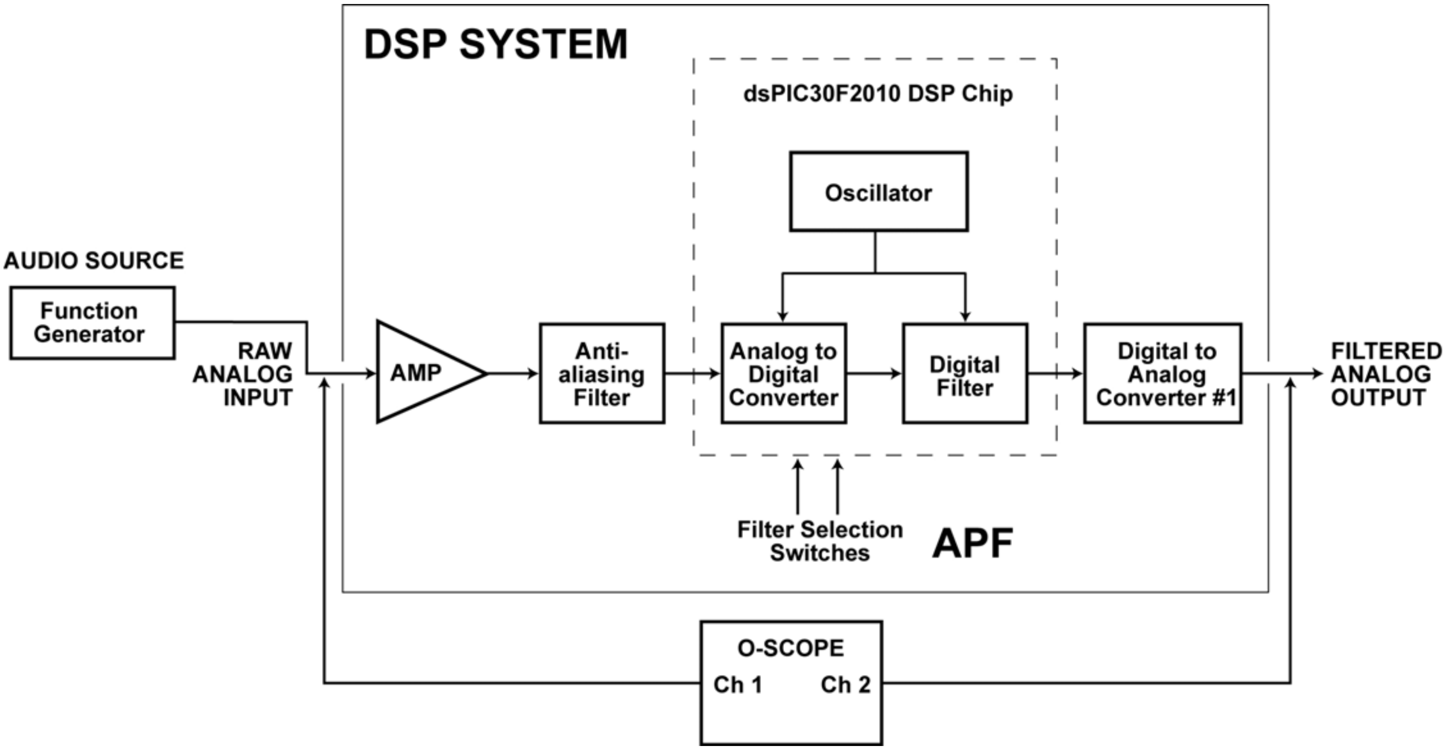
where "X" bits may be either high, low, or switching due to noise.

8. Once the ADC and DAC are working, graph the input/output characteristic curve by plotting the output voltage (Node G) as a function of the input voltage (Node A).

V_{in}	0.0	0.5	1.0	1.5	2.0	2.5	3.0	3.5	4.0	4.5	5.0
V_{out} ± 20%											

9. Verify that the relationship is linear, and that Vout varies at least between 1V to 4V. (In other words, the output voltage should reach to at least within 1 volt of the supply rails.
10. Remove power from the system.
11. Remove the temporary jumper between Node A and Node E.
12. Replace the LM358.

ALIASING TEST



1. Remove power from the system.
2. Use an oscilloscope to display both the input signal (Node A) on Channel 1, and the output signal (Node G) on Channel 2.
3. Enable "ALL PASS FILTER" mode on the DIP switches.
4. Using a function generator, apply a 1 kHz sine wave to Node A. Adjust the amplitude of the sine wave such that the signal is as large as possible without noticeable distortion.
5. Observe the output signal; it should be a reasonably good replica of the input signal.
6. Increase the frequency gradually. Note how the circuit output follows. As you approach 20 kHz, note the quantization effects and briefly describe in the space below.

7. Increase F_{in} beyond 22.05 kHz. Note that the output amplitude drops dramatically. This is mainly due to a low pass filter ("LPF") which limits the input frequency of this system.

8. Set F_{in} to 30 kHz. Except for any residual noise, V_{out} should be approximately zero.
9. Remove power and carefully make the following temporary modifications to your circuit
   a. Place a jumper across the 820 Ohm resistor at Node E.
   b. Place a jumper across the 8.2 kOhm resistor at Node G.
   c. Remove the 0.001 uF capacitor at node G.
10. What happens to the output signal? Record your observations below.

11. The sampling frequency of this system is 44.1 kHz. The maximum allowable input frequency is therefore 22.05 kHz. Vary F_{in} over several kHz above 22.05 kHz. What is the relationship between F_{in} and F_{out} in this region? Record the relationship below.

12. Remove the jumper across the 820 Ohm resistor and the 8.2 kOhm resistor. Replace the 0.001 uF capacitor.

DYNAMIC RANGE TEST

(Refer to the figure in the Aliasing Test, above.)

Noise in this system will limit the dynamic range that is achievable. The dynamic range of commercial CD and MP3 players is typically 90 dB or more. In this exercise, you will measure the dynamic range of your DSP system.

1. Set the filter mode to APF by setting the DIP according to the chart located elsewhere in this lab exercise.
2. Set the function generator to Sine, 1 kHz output.
3. Using an oscilloscope, view the circuit output (Node G).
4. Adjust the function generator such that the sine wave output at Node G is as large as possible *without introducing noticeable distortion.* Look especially for clipping of the sine wave. Increase the amplitude until you notice clipping, then back off slightly. This is the maximum output signal. Record its peak-to-peak amplitude.

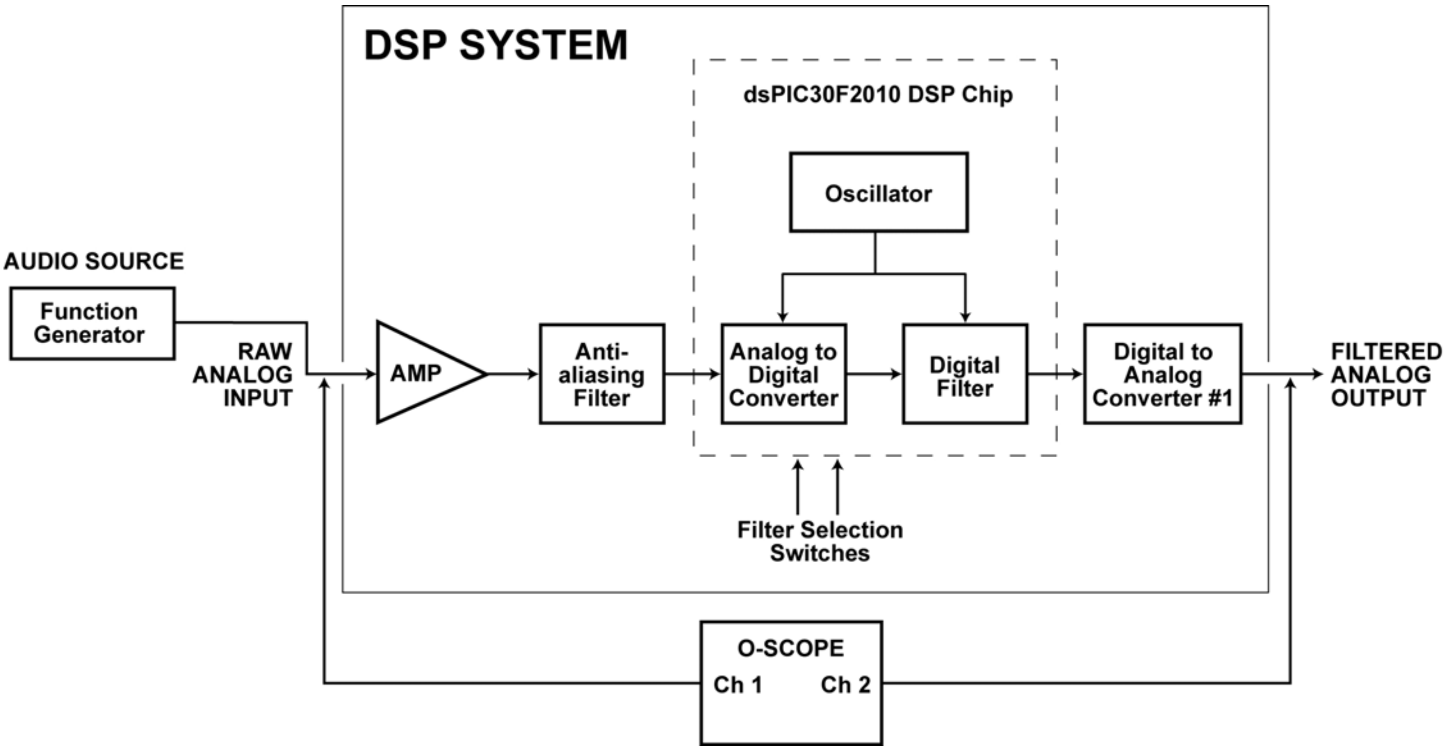
5. Now reduce the amplitude of the sine input to zero. (If the function generator output cannot go down to zero, disconnect it and short Node A to ground.)
6. Carefully measure the output amplitude (Node G) again. Note that ideally, since $V_{in} = 0$, that Vout should be zero also. Unfortunately, however, there will be some nose present. Crank up the vertical gain on the oscilloscope, and record the noise amplitude (peak-to-peak).

7. Calculate and record the system dynamic range using the following formula:

$$DynamicRange = 20\log_{10}\left(\frac{V_{OUT,MAX}}{V_{OUT,MIN}}\right)$$

8. Compare your result to a commercial CD player with DR = 90 dB. Is your DR higher or lower? Explain this result below.

FILTER TRANSFER FUNCTIONS



In the following four sections, you will evaluate the performance of the digital filters in this DSP system. There are four separate filters pre-programmed into the DSP chip in this lab.

All-Pass: This filter simply passes the input (ADC) signal directly to the output (DAC). This is useful as a test mode for the DSP system. It also provides a way of measuring the baseline system performance before any of the following filters are enabled. Graphing the frequency response of this mode will yield the bandwidth of the system that can be compared with the next three graphs. This enables you to determine the performance of the digital filter algorithm itself, independent of the supporting hardware.

Low Pass (LPF), High Pass (HPF), and Band Pass (BPF): These three filters are implemented with a 4-tap IIR filter, with coefficients calculated to yield an elliptic ("Cauer") response. You will determine the critical frequency of each of these filters when you measure and graph its frequency response below.

1. Use a function generator and oscilloscope as shown in the figure above. If your function generator does not contain an integrated frequency counter, then connect a separate frequency counter to the input signal as well. This will allow you to set the frequency more quickly and more precisely.
2. Display the input signal (Node A) on channel 1 and the output signal (Node G) on channel 2 of the oscilloscope.

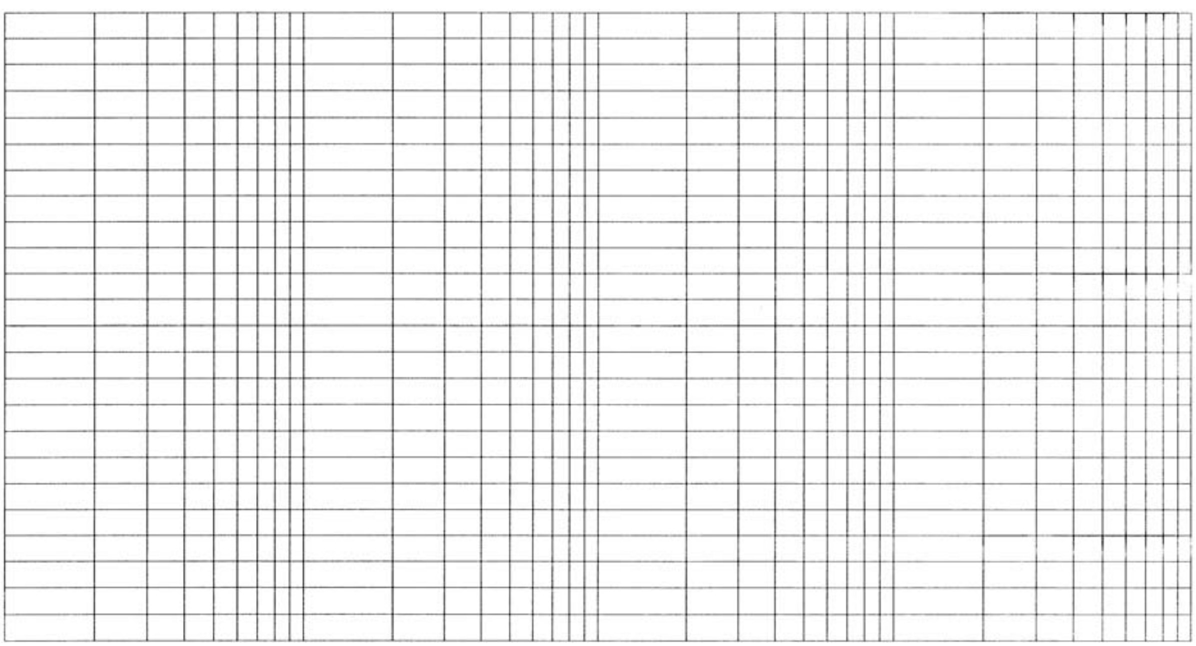
3. Select the desired filter using the DIP switches on your circuit. Use the following table to select the filter:

FILTER	SW11	SW12
Low-Pass	ON	OFF
High-Pass	OFF	ON
Band-Pass	ON	ON
All-Pass	OFF	OFF

   a. SW11 is the switch that is connected to Pin 11 of the DSP chip.
   b. SW12 is the one that is connected to Pin 12.
4. Briefly scan through the audio frequency range to verify that the correct filter is selected.
5. Measure and record the circuit gain at each frequency listed in the Tables below. Repeat Steps A, B, C, D, and E below for each frequency.
6. Adjust the function generator to output a sine wave of the appropriate frequency.
7. Adjust the sine wave amplitude to set the input signal to be as large as possible without causing noticeable clipping or distortion of the output waveform.
8. At frequencies where the gain of the filter is very low, it may be difficult to obtain a meaningful measurement of the output signal, as it will be very small and buried in noise. In these situations, you may raise the input amplitude. This will increase the amplitude of the output signal, yielding a better measurement. (Be sure to record the input amplitude as well.) As always, do not increase the input amplitude so much as to create a distorted signal.
9. Record the input and output amplitudes in peak-to-peak volts.
10. Allow your lab partner to copy these two measurements and convert them to Gain (both Volts/Volt and decibels) and plot them on the graph while you take the next data point.
11. Take data at **additional** points where there is a dramatic change in gain between points. This is necessary to yield an accurate plot of the filter response in the transition region. Extra space is provided in the data collection tables below for this purpose.
12. Repeat the steps above for each of the four filters.

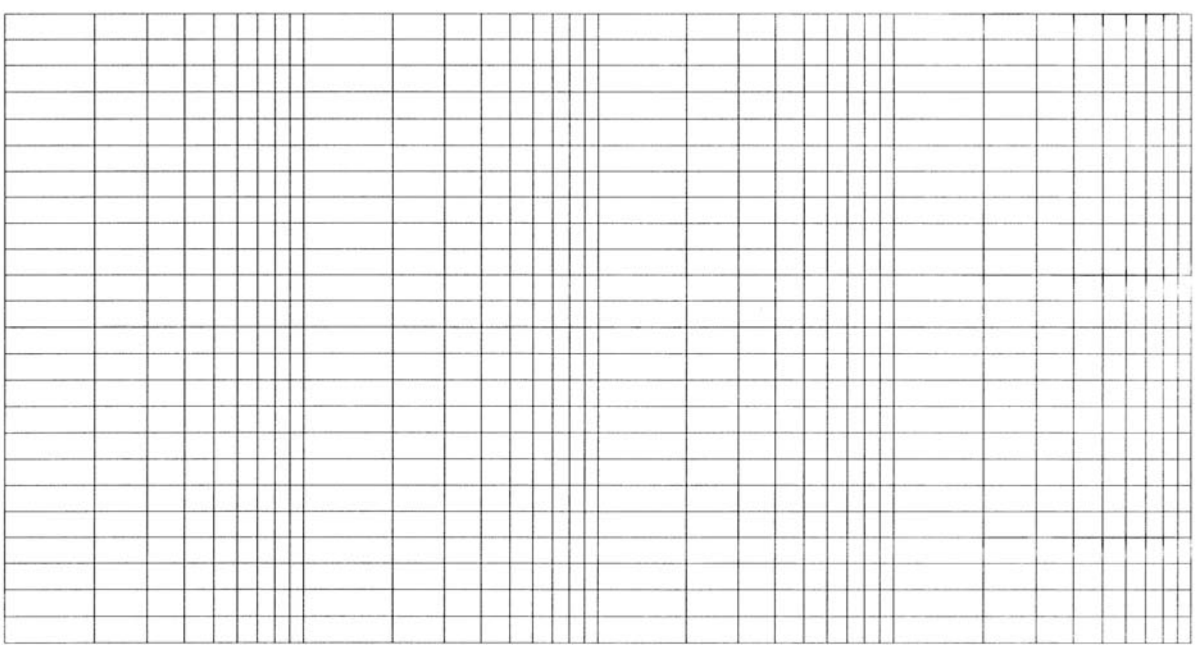
DATA COLLECTION TABLE & GRAPH: **LOW PASS FILTER**

Freq (Hz)	Input Ampl. (Vpp)	Output Ampl. (Vpp)	Gain (V/V)	Gain (dB)	Freq (Hz)	Input Ampl. (Vpp)	Output Ampl. (Vpp)	Gain (V/V)	Gain (dB)
20					4000				
30					5000				
40					6000				
50					7000				
60					8000				
70					9000				
80					10000				
90					11000				
100					12000				
200					13000				
300					14000				
400					15000				
500					16000				
600					17000				
700					18000				
800					19000				
900					20000				
1000									
2000									
3000									



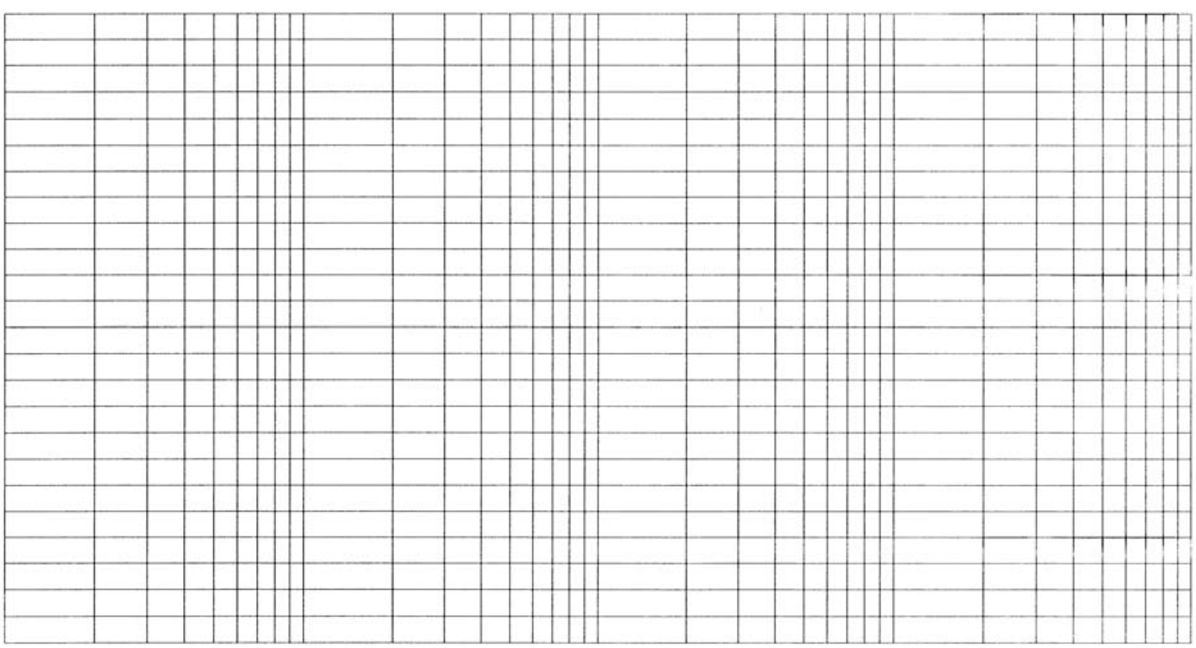
DATA COLLECTION TABLE & GRAPH: **HIGH PASS FILTER**

Freq (Hz)	Input Ampl. (Vpp)	Output Ampl. (Vpp)	Gain (V/V)	Gain (dB)	Freq (Hz)	Input Ampl. (Vpp)	Output Ampl. (Vpp)	Gain (V/V)	Gain (dB)
20					4000				
30					5000				
40					6000				
50					7000				
60					8000				
70					9000				
80					10000				
90					11000				
100					12000				
200					13000				
300					14000				
400					15000				
500					16000				
600					17000				
700					18000				
800					19000				
900					20000				
1000									
2000									
3000									



DATA COLLECTION TABLE & GRAPH: **BAND PASS FILTER**

Freq (Hz)	Input Ampl. (Vpp)	Output Ampl. (Vpp)	Gain (V/V)	Gain (dB)	Freq (Hz)	Input Ampl. (Vpp)	Output Ampl. (Vpp)	Gain (V/V)	Gain (dB)
20					4000				
30					5000				
40					6000				
50					7000				
60					8000				
70					9000				
80					10000				
90					11000				
100					12000				
200					13000				
300					14000				
400					15000				
500					16000				
600					17000				
700					18000				
800					19000				
900					20000				
1000									
2000									
3000									



DATA COLLECTION TABLE & GRAPH: **ALL PASS FILTER**

Freq (Hz)	Input Ampl. (Vpp)	Output Ampl. (Vpp)	Gain (V/V)	Gain (dB)	Freq (Hz)	Input Ampl. (Vpp)	Output Ampl. (Vpp)	Gain (V/V)	Gain (dB)
20					4000				
30					5000				
40					6000				
50					7000				
60					8000				
70					9000				
80					10000				
90					11000				
100					12000				
200					13000				
300					14000				
400					15000				
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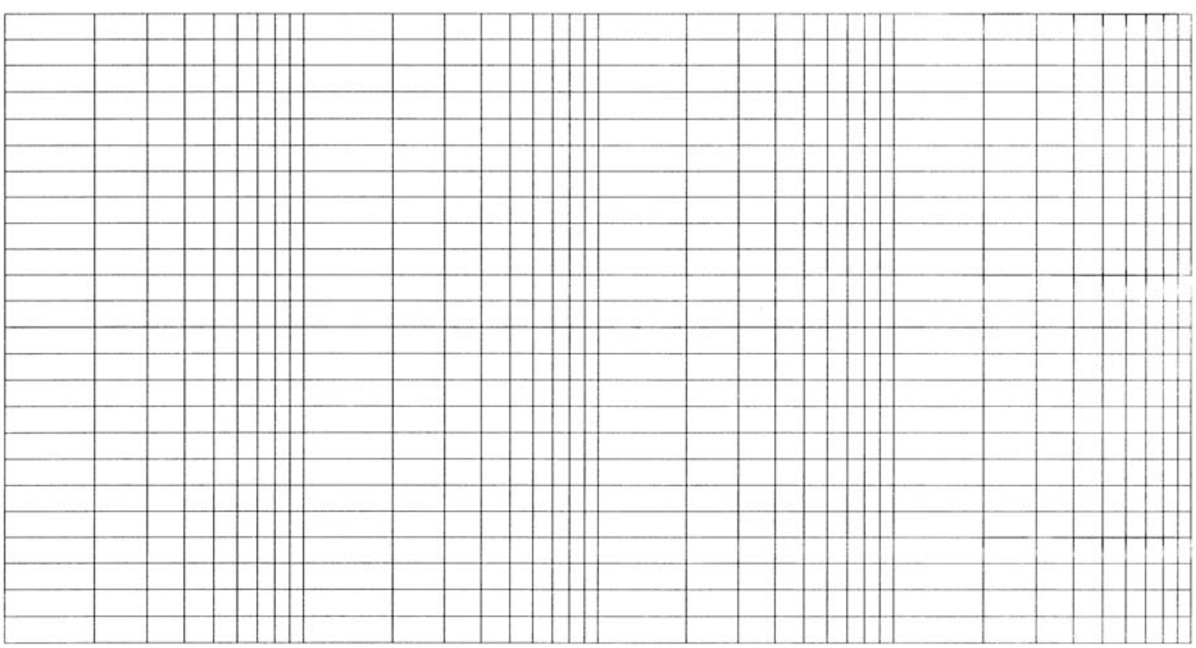
900					20000				
1000									
2000									
3000									



ANALYSIS

1. If you have not yet done so, calculate the filter gain in volts per volt and in decibels for each frequency for which you have data. Recall that for signals,

$$dB = 20\log_{10}\left(\frac{V_{OUT}}{V_{IN}}\right)$$

2. Plot the dB gains on the frequency response graph.
3. After plotting the frequency response curve for each of the four filters, determine the following filter parameters from the graphs:

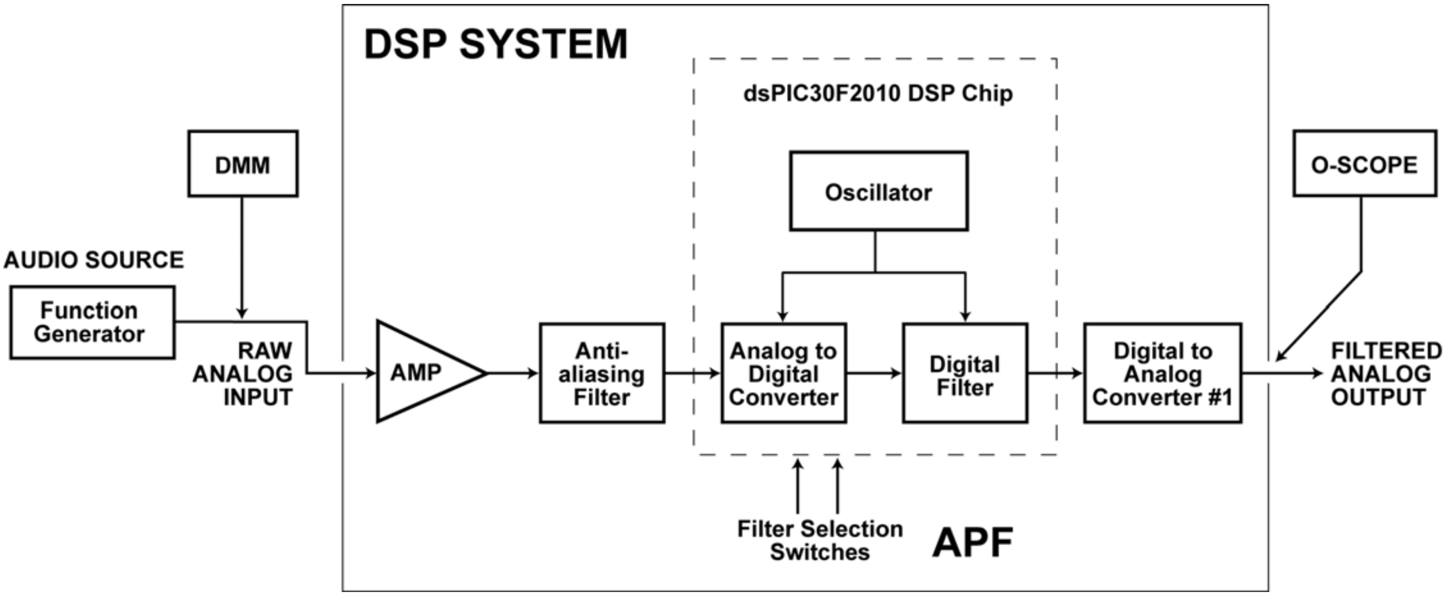
Filter Parameter	LPF	HPF	BPF
Critical Frequency			
Passband Ripple			
Stopband Attenuation			

QUANTIZATION EFFECTS

It is now time to evaluate the subjective effects that digital filtering has on actual sounds. In this section, you will play music through your digital filter and note the effects that you hear.

Commercial CD and MP3 players usually output audio signals at 16-bits resolution or greater. Our system outputs signals at 8-bits resolution. To evaluate how this affects the sound quality, follow the following steps.

1. Connect your CD or MP3 player directly to the amplified speakers with a standard audio patch cord.
2. Listen to a song carefully, evaluating the sound quality.
3. Now interpose the DSP system between the audio source and the amplified speakers as shown below. Use an audio patch cable to connect the headphone output of your CD or MP3 player to the input of your DSP system.



4. Set the filter to All-Pass mode with the DIP switches.
5. Listen to the same song you listened to in Step 2 above. Be sure the volume is set to a similar level. (To assure that the test is “fair”, set the CD or MP3 player output volume as high as possible, and set the

amplified speaker volume relatively lower. This will minimize the quantization and noise effects introduced by the DSP system.)

6. Record your impressions briefly in the space provided.

Audio Filtering

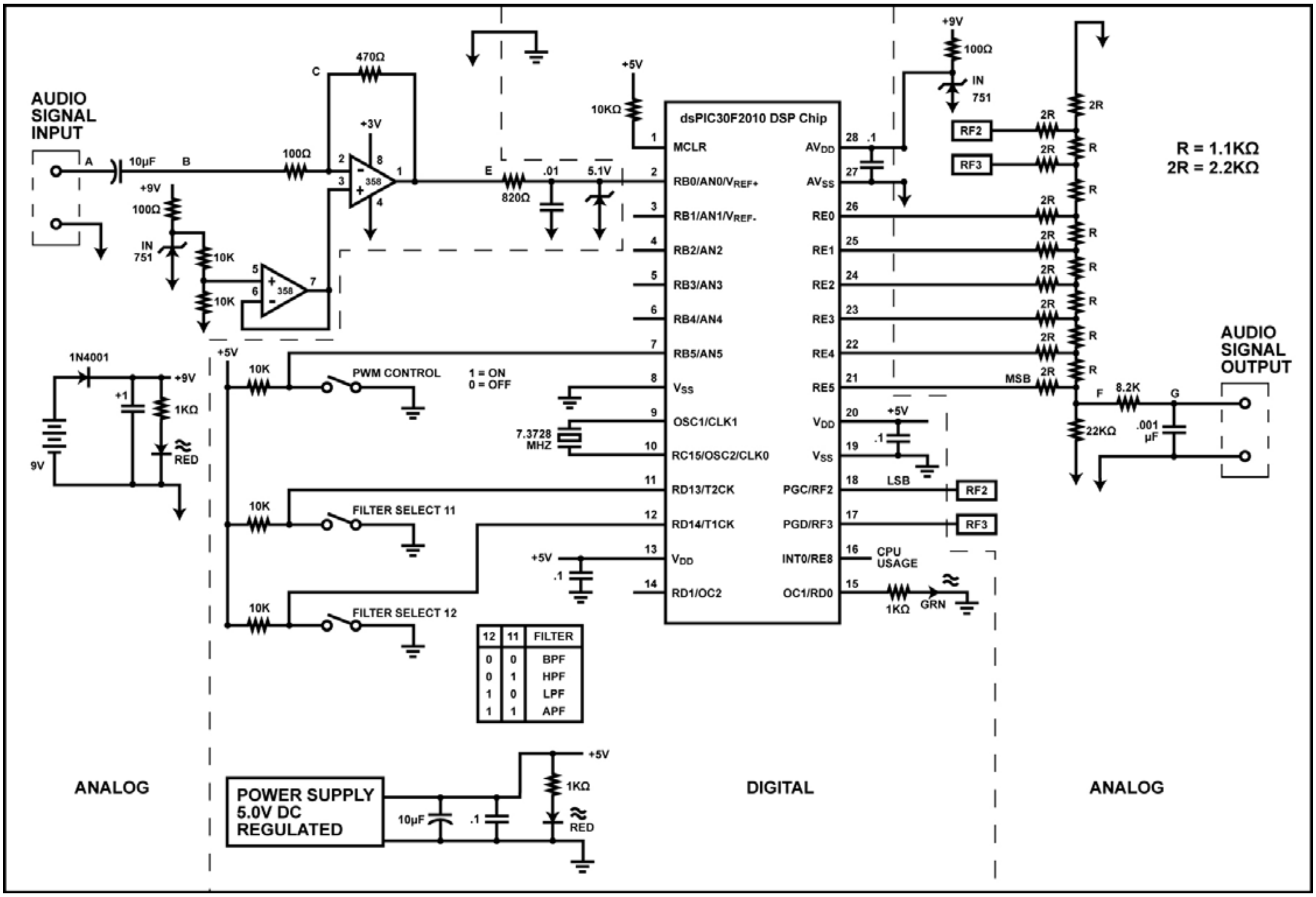
To determine the effects of these digital filters on actual sounds, leave your system connected to the CD player and speakers as above. Start a disk playing, then, follow these steps:

1. Activate the LPF by setting the DIP switches as shown in the table in the Frequency Response section above.
2. Note the difference in sound. Switch the LPF on and off a few times to evaluate the difference.
3. Record your observations below.

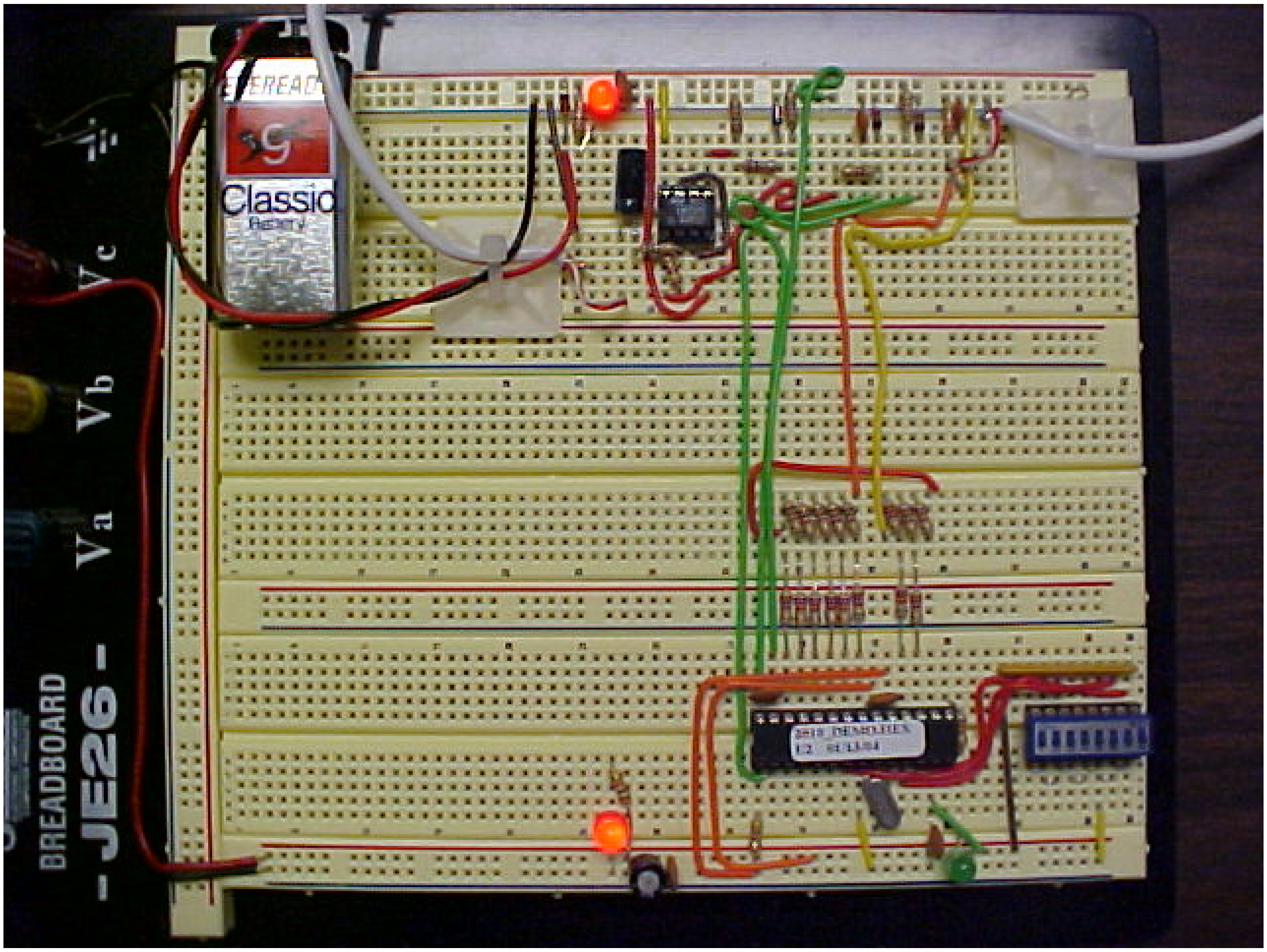
4. Switch off the LPF.
5. Switch on the HPF. Note the difference in sound. Record your observations below.

6. Finally, switch off the HPF, and switch on the BPF. As before, note the difference in sound and record your observations below.

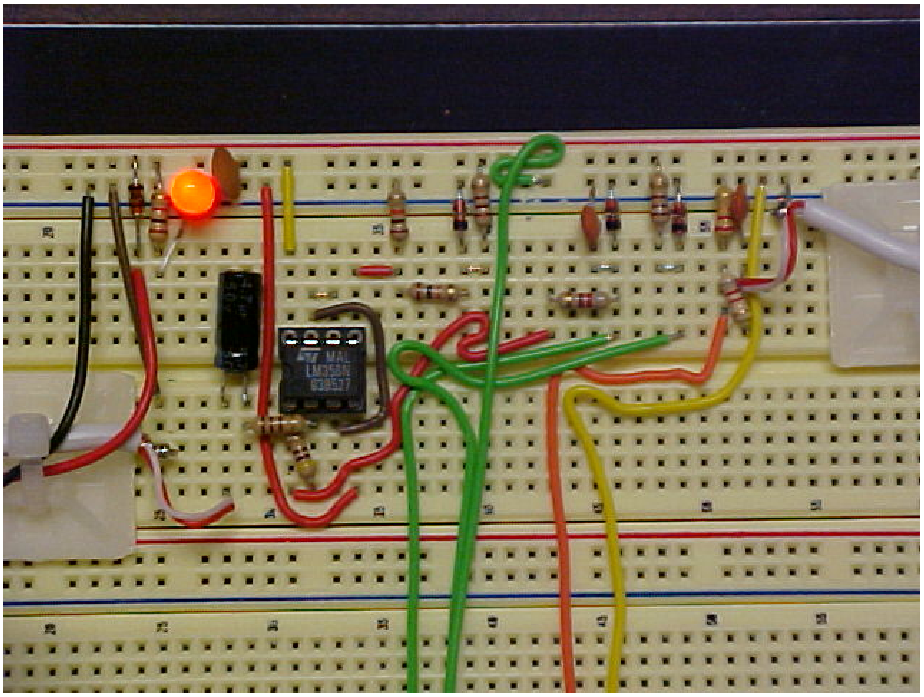
Schematic: Circuit Layout



Suggested DSP Circuit Layout



Analog Section



Digital Section