Experiment 10

Sampling and Reconstruction

In this experiment we shall learn how an analog signal can be sampled in the time domain and then how the same samples can be used to reconstruct the original signal.

Figure 10.1 shows the experiment board you will use in the laboratory. The board consists of five internally connected card edge connectors to realize the block diagram shown in Figure 10.2. The input signal to the board is supplied from the function generator or from a music source. Stereo headphones or an audio speaker can be connected to the output port of the circuit.

We shall test the components of the system one by one and then put them all together to listen to the effects of different system parameters including the sampling rate and the bandwidth of the reconstruction filter.

The experiment board requires the three supply voltages of +5 V, +15 V and -15 V. Set each voltage individually, disable the output and connect the power supply to the experiment board. You will make four connections including the ground.

10.1 Antialiasing Filter

The first PC board shown in Figure 10.3 consists of two electronic circuits. The first circuit is used to convert the stereo signal from an audio source to a mono signal. This eliminates the need to duplicate the electronics for the two channels for this experiment. In practice, digital signal processing is performed on both channels using independent circuitry.

The second circuit is a 4th order Butterworth low-pass filter, which is used to reduce the signal power above a cut-off frequency. The function of this filter can be explained as follows: In the theory section, we have learned that for accurate reconstruction of the original signal from its samples, the sampling rate must be higher than twice the maximum frequency component of the original signal. As an example, let's suppose we wish to use a sampling rate of 20 kHz. We know that this

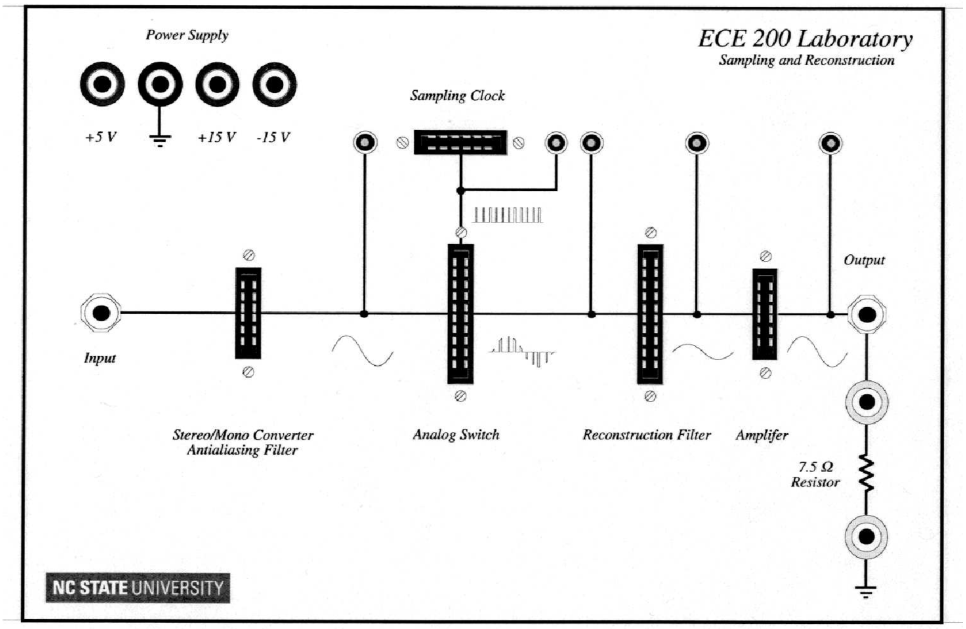

Figure 10.1: Sampling and Reconstruction Board used in this experiment

rate can be successfully used only if the original signal contains negligible power above 10 kHz. On the other hand, we know that the music signals can have considerable power at frequencies well above 10 kHz. Therefore, if we really have to use this low sampling rate of 20 kHz, the frequencies above 10 kHz must be filtered out by a low-pass, preconditioning filter prior to sampling.

We shall begin the experiment by measuring the cut-off frequency of the antialiasing low-pass filter. The cut-off frequency is commonly defined as the frequency where the signal power drops to one half of its value in the filter pass-band. This corresponds to a power gain of

$$G = 10 \log \frac{P_{out}}{P_{in}} = 10 \log \frac{P_{in}/2}{P_{in}} = 10 \log(0.5) \qquad (10.1.1)$$

which yields - 3dB. For this reason, the cut-off frequency is sometimes referred to as the 3 dB cut-off frequency.

Preparation:

- Disable the power supply output.
- Insert the antialiasing filter PC board into the card edge connector.
- Connect the antialiasing filter output to channel 2 of your oscilloscope.

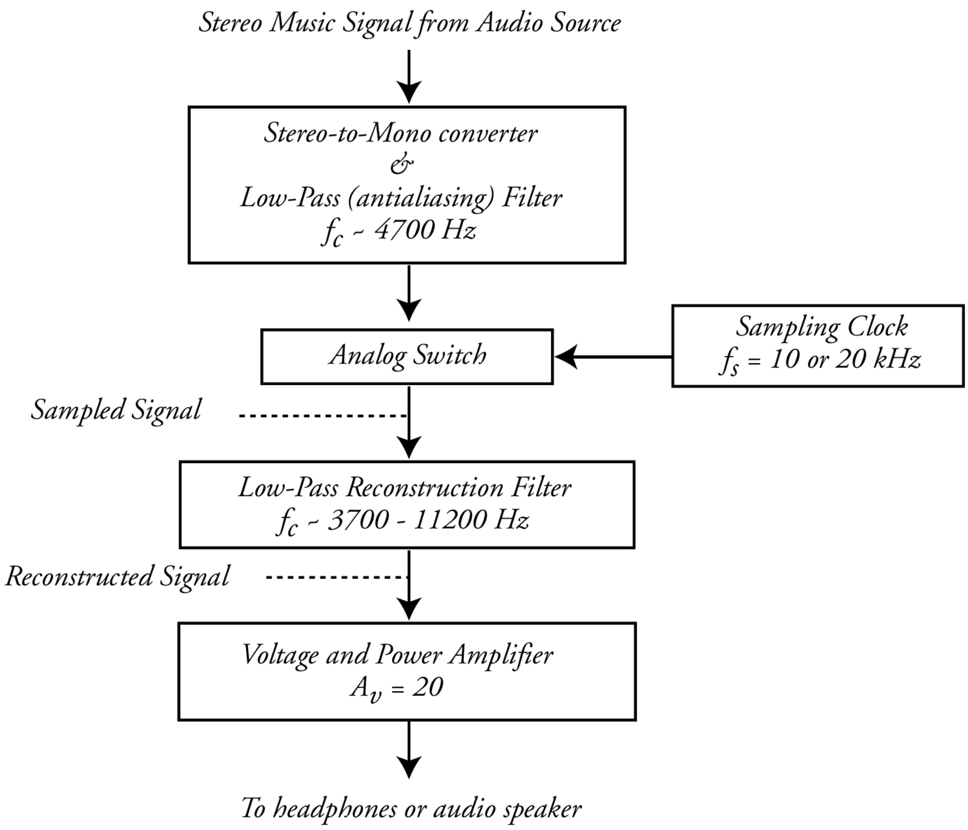

Figure 10.2: Building Blocks of the Sampling and Reconstruction System

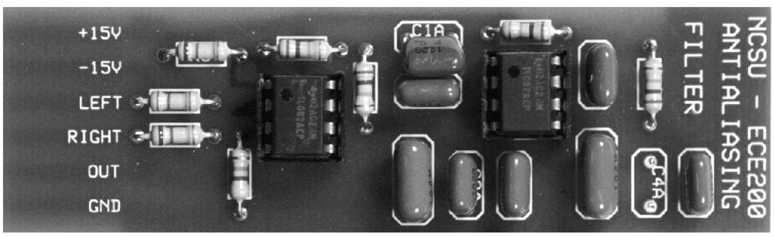

Figure 10.3: Antialiasing Filter

- Connect a BNC T-connector to the function generator's BNC output connector.
- Connect one end of the BNC T-connector to the experiment board using a BNC-to-1.4" Phono adapter cable.
- Connect the other end of the BNC T-connector to channel 1 of your oscilloscope.
- Enable the power supply output.
- Set the function generator to generate a 100 Hz sinusoid with a peak-to-peak value of less than 500 mV. Note that this value is not critical. You can choose any input voltage as long as the output signal is free of noise and it is not distorted. Adjust the oscilloscope controls to display the input and output signals to your satisfaction.
- Turn on the spectrum analyzer (FFT), Choose Channel 2. Set the frequency span to 50 kHz and the center frequency to 25 kHz. You should see a single spike at 100 Hz.

Measurements & Questions:

We will now measure the power gain of your filter at different frequencies to generate its frequency response.

1. Measure the input and output signal power at different frequencies ranging from 100 Hz to 50 kHz. Note the following:
   (a) As long as you do not change the peak-to-peak value of the sinusoid produced by the function generator, you do not need to measure the input signal power at every frequency. It should remain constant throughout the measurement range.
   (b) In plotting the frequency response of an amplifier or a filter, it is customary to use a logarithmic frequency axis. To do this, measure the signal power at 100 Hz intervals up to 1000 Hz and then every 1000 Hz up to 10 kHz. Depending on how quickly the output is dropping, you may increase the increment even more above 10 kHz.
2. Calculate the power gain, G at each frequency.
3. Tabulate your results.
4. Use your measurement results to plot the *frequency response* of your filter. Recall that the frequency response is just a plot of power gain versus frequency.

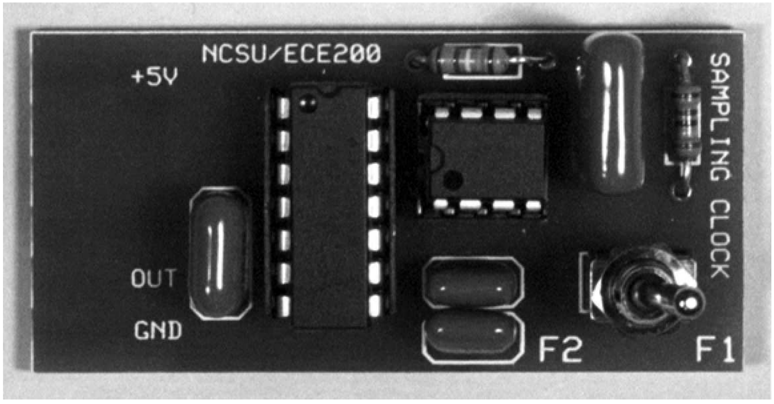

Figure 10.4: The Sampling clock circuit capable of generating two rectangular pulse trains with sampling rates of 10 and 20 kHz.

5. Use your plot to determine the 3 dB cut-off frequency of your filter.
6. Save the oscilloscope display at the 3 dB frequency to an image file to include in your report.
7. Plot the output signal power as a function of frequency. Use a logarithmic scale for the frequency axis.
8. Calculate the amplitude of the sinusoid in the pass-band and at the cut-off frequency using the measured signal power in Table **??**. Is the amplitude at the cut-off frequency equal to 0.707 of its value in the pass-band?

10.2 Sampling Clock

In this part of the experiment we shall test the sampling clock circuit shown in Figure 10.4. The circuit generates a square pulse train with a duty cycle of approximately 5 percent. There are two possible sampling frequencies of 10 and 20 kHz selectable by a toggle switch mounted on the PC board.

Preparation:

- Disable the power supply output.
- Insert the sampling clock into the card edge connector on the experiment board.
- Connect the board output to Channel 1 of your oscilloscope.

- Enable the power supply output.

Measurements & Questions:

1. Set the toggle switch on the PC board to F1. This will set the sampling frequency to a value around 10 kHz. The measured frequencies will be slightly different for each laboratory station due to variations in the actual values of the resistors and capacitors used in the circuit.
   (a) Save the oscilloscope display to an image file to include in your laboratory report.
   (b) Measure the frequency F_1, of the sampling clock
   (c) Measure the duty cycle of the sampling clock.
2. Set the toggle switch to F2, which will set the frequency to a value around 20 kHz.
   (a) Save the oscilloscope display to an image file to include in your laboratory report.
   (b) Measure the frequency, F_2, of the sampling clock.
   (c) Measure the duty cycle of the sampling clock.

10.3 Sampled Sinusoid in Time Domain

In this part of the experiment we shall use an electronic circuit, which operates similar to the triac in the Electric Power experiment. Figure 10.5 describes the operation of the circuit. The rectangular pulse train we have seen in the previous section decides on the state of the switch. When the voltage from the sampling clock is high, the switch closes and allows the analog signal to pass through. Otherwise, the switch is closed and the output voltage is zero. As such, the output signal consists of the samples of the original signal. The analog switch is a standard building block of all analog-to-digital converter integrated circuits. The electronic circuit used in this experiment is shown in Figure 10.6.

Preparation:

- Disable the power supply output.

- Insert the Analog Switch PC board in its card edge connector. Now you should have three boards on your experiment board: antialiasing filter, sampling clock and analog switch.
- Connect Channel 1 of your oscilloscope to the input of the analog switch.
- Connect channel 2 of your oscilloscope to the output of the analog switch.
- Set the sampling frequency to F1 by setting the toggle switch to F1.
- Enable the power supply output voltage.
- Verify that the function generator is still connected to the experiment board. If not make the connection.
- Adjust the horizontal and vertical scales of the oscilloscope so that the original and the sampled sinusoid are displayed simultaneously. For triggering, use the channel displaying the original sinusoid.

Measurements & Questions:

1. Set the function generator to produce a 100 Hz sinusoid with a peak-to-peak voltage of 1 V. Adjust the oscilloscope controls to display the input and output signals of the analog switch simultaneously. You should see a display of the sampled version of the original sinusoid. Use the sinusoid for oscilloscope triggering. If you are still having difficulty in obtaining a stable display, you can freeze the display by pressing the **Run/Stop** button.
   (a) Save the oscilloscope display to an image file to include in your laboratory report.
   (b) Measure the frequency and the pulse width of the samples.
2. Set the sampling frequency to F_2 by flipping the switch on the sampling clock PC board to F2.
   (a) Save the oscilloscope display to an image file to include in your laboratory report.
   (b) Measure the frequency and the pulse width of the samples.
   (c) Measure the RMS values of the analog signal and the sampled signal.
   (d) Calculate the normalized signal power of the original sinusoid and the sampled sinusoid in mW.

3. Compare your measured values for sampled signal frequency and pulse width with those of the sampling clock waveforms measured in the previous section.

4. Your measurements should indicate that the sampled signal power is considerably less than that of the original signal power. What percentage of the original power is retained in the sampled signal?

10.4 Reconstruction Filter

In this part of the experiment we shall test the reconstruction filter, which is a 2nd order Butterworth filter. The cut-off frequency of the filter is adjustable between 3700 to 11200 Hz. In practice, the cut-off frequency of this filter is fixed at the frequency corresponding to half the sampling rate. As mentioned, the filter used in this laboratory is far from ideal and we will adjust the pass-band to study a variety of effects. The PC board is shown in Figure 10.7.

Preparation:

- Disable the power supply output.
- Insert the 'reconstruction filter' into its card edge connector.
- Set the sampling frequency to F_1
- Connect the analog switch output to channel 1 of the oscilloscope and the output of the reconstruction filter to channel 2.
- Turn the potentiometer on the reconstruction filter all the way counter-clockwise. This sets the cut-off frequency of the filter to its lowest value.
- Enable the power supply output voltage.

Measurements & Questions:

1. Set the frequency of the sinusoid to 300 Hz and the peak-to-peak voltage to 1 V. You should now see the samples of the original sinusoid as well as the reconstructed sinusoid displayed on your oscilloscope. Save the display to an image file to include in your laboratory report. How is the quality of the reconstruction process? Do you have a pure sinusoid or a periodic waveform that resembles a sinusoid?

2. Increase the frequency of the input sinusoid as you continue to observe the reconstructed signal. How is the reconstructed signal changing as you increase the frequency of the input signal? Is the reconstruction getting better or worse? Save the display at a higher sinusoid frequency to help you explain your observations in your lab report.

3. Set the frequency of the sinusoid back to 300 Hz. Slowly turn the potentiometer knob on the reconstruction filter clockwise. You are increasing the cut-off frequency of the low-pass filter. Is the reconstruction becoming better or worse? How is the reconstructed signal changing as you do this?

4. Turn the potentiometer on the reconstruction filter all the way clockwise. Save the display to an image file to include in your report.

5. Change the sampling rate to $F_2 \sim 20kHz$ by flipping the switch on the sampling clock to F2. Save the oscilloscope display to an image file to include in your lab report.

6. Suppose the sampled waveform is applied to an RC circuit and we are measuring the voltage across the capacitor.

   (a) Speculate on the shape of the voltage waveform across the capacitor. Would the resulting waveform resemble what he have at the output of the reconstruction filter?

   (b) Is there a similarity between changing the RC time constant and the cut-off frequency of the reconstruction filter?

10.5 Audio Amplifier

The fifth and the last circuit is an audio amplifier, which you have used in previous experiments. It amplifies the reconstructed signal and serves as a power amplifier to drive a pair of headphones or an audio speaker.

Preparation:

- Turn off the spectrum analyzer.
- Disable the power supply output.
- Insert the 'audio amplifier' into its card edge connector.
- Connect the reconstruction filter output to channel 1 of the oscilloscope and the audio amplifier output to channel 2.

	RMS Voltage	Normalized Power (mW)
Antialiasing Filter Output		
Analog Switch Output		
Reconstruction Filter Output		
Audio Amplifier Output		

Table 10.1: RMS values and Signal Power of the original sinusoid and the reconstructed sinusoid before and after amplification

- Turn the potentiometer on the reconstruction filter all the way counter-clockwise to set the cut-off frequency of the filter to its lowest value.
- Set the sampling frequency to F_2.
- Set the frequency of the input sinusoid to 300 Hz.
- Enable the power supply output voltage.

Measurements & Questions:

1. Display the input and output signals of the audio amplifier and save the display to an image file to include in your report.
2. Determine the voltage gain of the amplifier.
3. To understand the function of the audio amplifier we shall first measure the RMS value of the signals at different points of the sampling and reconstruction block diagram. We shall then use these voltages to determine the signal power at those points. Use the **quick measurement** feature of your oscilloscope to measure the RMS value of the signals at the following points:
   (a) Output of the antialiasing filter switch.
   (b) Output of the analog switch.
   (c) Output of the reconstruction filter.
   (d) Output of the audio amplifier.
4. Enter the measurement results in Table 10.1. Calculate the normalized signal power for each voltage according to $P = V_{RMS}^2$ and enter the results in the same table.
5. Consider the following questions in your lab report:

(a) Why is the signal power of the sampled signal less than that of the original signal?

(b) How much of the original signal power is retained in the sampled signal? Could you relate this percentage to the duty cycle of the sampling clock?

(c) Compare the signal power of the original signal with that of the reconstructed signal before and after amplification. Write a short paragraph describing the function of the amplifier.

10.6 Sampled Signal in Frequency Domain

In this part of the experiment, we shall examine the power spectrum of the sampled signal.

Preparation:

- Connect the analog switch output to Channel 1 of your oscilloscope.
- Turn off channel 2.
- Set the frequency of the input sinusoid to 1 kHz.
- Turn on the spectrum analyzer (**FFT** feature under **Math**) and display the power spectrum of the sampled signal. Set the frequency span to 50 kHz and center frequency to 25 kHz. In order to get the frequency components to appear as narrow peaks, set the time scale to 1 ms/div.

Measurements & Questions:

1. Set the sampling frequency back to F_1. You should now have the sampled signal displayed in both time and frequency domains.

   (a) Save the oscilloscope display to an image file to include in your laboratory report.

   (b) Determine the frequencies of the harmonics within 50 kHz.

   (c) Determine the fundamental frequency of the sampled signal.

2. Change the sampling frequency to F_2 by flipping the switch on the sampling clock board to F2.

   (a) Save the oscilloscope display to an image file to include in your laboratory report.
   (b) Determine the frequencies of the harmonics within 50 kHz.
   (c) Determine the fundamental frequency of the sampled signal.

3. Slowly increase the frequency of the original sinusoid. Observe the changes in the spectrum to answer the following questions:
   (a) Is the fundamental frequency of the sampled sinusoid changing?
   (b) How are the harmonics moving on the frequency axis?

10.7 Reconstructed Signal in Frequency Domain

Preparation:

- Connect the audio amplifier output to Channel 1 of your oscilloscope.
- Connect the headphones to the experiment board.
- Set the frequency of the input sinusoid to 1 kHz.
- Set the cut-off frequency of the low-pass filter to its lowest value by turning the potentiometer all the way counterclockwise.
- Leave the settings of the spectrum analyzer unchanged.

Measurements & Questions:

1. Set the sampling rate to $F_1 \sim 10kHz$. You should now have a display of the reconstructed signal in both time and frequency domains for the lowest cut-off frequency of the reconstruction filter. Save the oscilloscope display to an image file to use in your report.
2. Slowly increase the cut-off frequency of the low-pass filter as you observe the changes in the output spectrum. Listen to the output signal. Can you hear any changes in the sound? If yes, how can you describe the changes in your own words?
3. Set the cut-off frequency of the reconstruction filter to its largest value by turning the potentiometer clockwise all the way. Save the oscilloscope to an image file to include in your report.

4. Set the cut-off frequency of the reconstruction filter back to its lowest value. Slowly increase the frequency of the original sinusoid as you observe the spectrum of the reconstructed signal. How is the spectrum changing with frequency? Can you relate your observation to the output spectrum of the sampled signal? Try flipping the sampling rate toggle switch on the sampling clock board back and forth as you change the frequency of the input signal. Can you hear any changes in the sound as a result of changing the sampling rate?

5. Can you determine the maximum frequency of the original sinusoid that can be reconstructed with the highest cut-off frequency of the reconstruction filter? Use information in both time and frequency domains. Can you reconstruct a 2 kHz sinusoid? How about 3 or 4 kHz ? Save at least one oscilloscope display to an image file to help you explain your observations in your lab report.

10.8 Sampling and Reconstruction of Music Signals

Preparation:

- Disable the power supply output.
- Disconnect the function generator from the input of the Sampling and Reconstruction board.
- Connect the output of your computer's sound card to the experiment board input using a 1/8" Mono cable and the 1/8" Mono to 1/4" Mono adapter.
- Plug the headphones/speaker to the output of the audio amplifier.
- Enable the power supply output.
- Set the cut-off frequency of the reconstruction filter to its lowest value.
- Start playing your favorite music. You should now observe the reconstructed music signal in both time and frequency domains.

Measurements & Questions:

1. Set the sampling rate to F_1. Save the display to an image file to include in your report. Use the **RUN/STOP**button to freeze the display.

2. Set the sampling rate to F_2. Save the display to an image file to include in your report.

3. Flip the toggle switch between F1 and F2 to determine the impact of the sampling rate on the quality of the sound Listen to the sound output carefully to answer the following questions:

   (a) Which one, F_1 or F_2, do you rate qualitatively better?

   (b) Given that the input signal is limited to about 5 kHz by the antialiasing filter, explain why the sampling rates of 10 kHz ($F_s/2 \sim 5kHz$) and 20 kHz ($F_s/2 \sim 10kHz$) should or should not give significantly different results.

4. Set the sampling rate to F_1. Slowly change the cut-off frequency of the reconstruction filter as you listen to the output sound carefully. How would you describe the change in sound quality in your own words?

5. Repeat the previous step after setting the sampling rate to F_2. Listen to the output sound. How did the impact of the filter cut-off frequency change as a result of sampling the original signal at a higher rate?

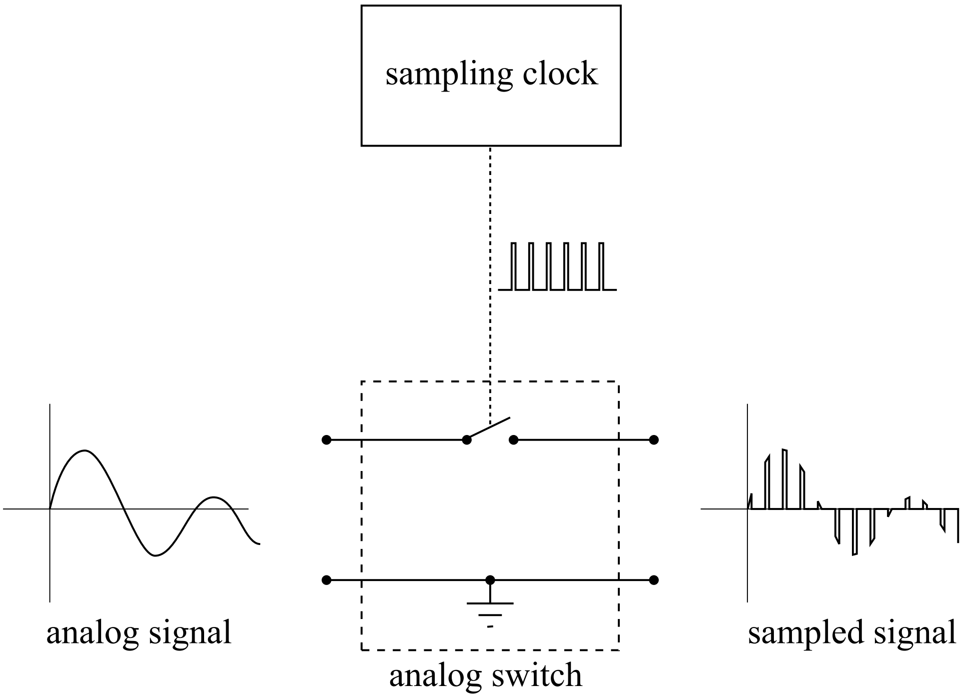

Figure 10.5: An electronic switch is used to sample the analog signal. The state of the switch is determined by a rectangular pulse train generated by the sampling clock.

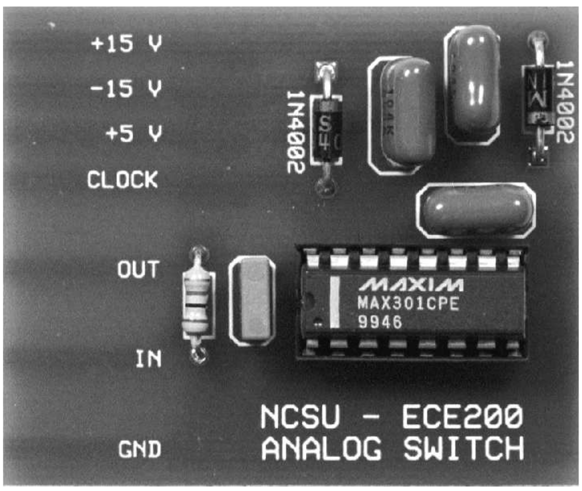

Figure 10.6: Analog Switch Circuit

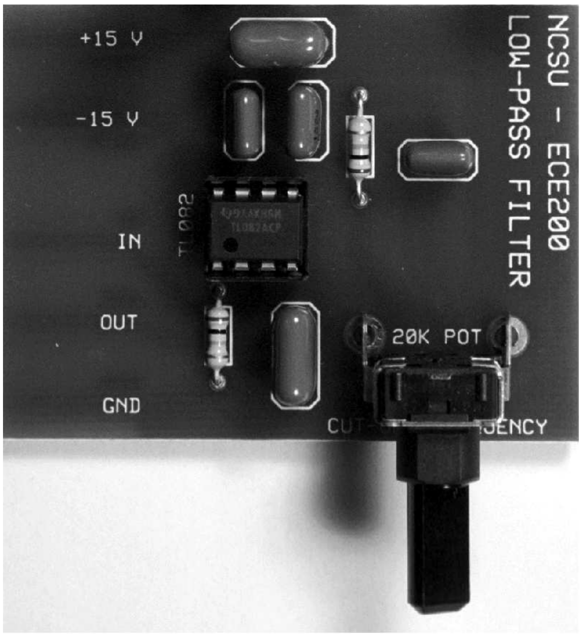

Figure 10.7: Low-pass filter used to reconstruct the original signal. The potentiometer is used to set the 3-dB cut-off frequency between 3700 Hz to 11200 Hz