**Lab 5: Analog-to Digital and Digital-to-Analog Conversion**

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**2. Pre-lab Questions**

c) If we use larger number of bits used for quantization, then the quality of the digitized signal would more conform to the original signal.

**3. DAC using National Instruments DAQ Board and LabView Program**

2. a) The analog voltage of 0V corresponds to “11111111”, 3V corresponds to “11001100”, and 1V corresponds to “10011001”

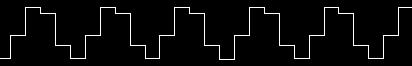
b) The “step size” is about 0.035 V.

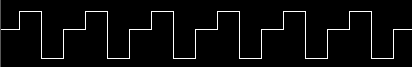
3. a) The digital values have fewer digits and are thus less precise.

b) The step size increases so it is less precise.

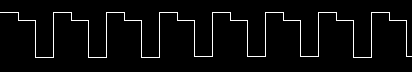
4. b) There are 5 samples taken over one period of the sine wave. This matches the predicted parameters because the sampling rate was 5000 Hz over 1000 samples read.

5. a) When the sampling rate is decreased, the digital signal becomes more and more imprecise. In the final case where the sampling rate is 1 kHz, there is in fact only one sample, resulting the flatline that is observed.

5.0 kHz: 

3.0 kHz: 

2.0 kHz: 

1.5 kHz: 

1.0 kHz: 

For the most part, the frequency of the sampled signal does not change with the sample rate changing. The exception, of course, occurs at 1.0 kHz, where the frequency is not equal. This rate is one-fifth the value of the original signal’s frequency.

6. As sampling rate increases, the “quality” of the playback sound increases. This is because there are more samples, so more of the analog signal is preserved, and the music becomes more coherent.

8. a) We can understand the lyrics of the song at a 6-bit resolution level.

**4. Simple Resistor-Based DAC**

3.

|  |  |
| --- | --- |
| **D5D4D3D2D1D0** | **Vout (V)** |
| 000000 | 0.000 |
| 000001 | 0.082 |
| 000010 | 0.167 |
| 000100 | 0.326 |
| 001000 | 0.644 |
| 010000 | 1.276 |
| 100000 | 2.535 |
| 111111 | 5.000 |

The measured voltages are approximately equal to the expected values, but most of the errors are positive.

4. Yes, the signal represents the AC signal as specified by the program.

5. The period of this signal is determined by the float c2.

6. **BONUS**: Verefied by Joy. We added a capacitor connected from one of the signals to ground. This created a low-pass filter that removed the high-frequency component spikes and the glitches that resulted.

7. When we remove D5, the function is severely distorted. Removing D4 results in a more jagged type of curve. Removing any of the other pins results in negligible effects compared to the original function. D5 has the largest impact on the signal, because it contributes the most to the overall Vout.

9. **BONUS**: Verified by Greg/Joy.

// this sketch generates digital numbers that correspond to sampled and quantized

// values of the function f=1/2\*(1+cos(2\*pi\*f\*t)

int M=6; // the size of the quantizer, in bits.

int N=100; // number of samples that function is calculated in

int f = 0; // quantized sample value

// the setup routine runs once when you press reset:

void setup() {

// initialize the digital pins as outputs.

pinMode(0, OUTPUT);

pinMode(1, OUTPUT);

pinMode(2, OUTPUT);

pinMode(3, OUTPUT);

pinMode(4, OUTPUT);

pinMode(5, OUTPUT);

}

// the loop routine runs over and over again forever:

void loop() {

for (int n=0; n<=N; n++){

f=round(n\*63/100);

digitalWrite(5,bitRead(f,5)); //5th bit of f goes to D5

digitalWrite(4,bitRead(f,4)); //4th bit of f goes to D4

digitalWrite(3,bitRead(f,3)); //3rd bit of f goes to D3

digitalWrite(2,bitRead(f,2)); //2nd bit of f goes to D2

digitalWrite(1,bitRead(f,1)); //1st bit of f goes to D1

digitalWrite(0,bitRead(f,0)); //0th bit of f goes to D0

}

}