EE 230
Spring 2010
Experiment 11

Since this is last lab,you do not need to submit the lab report . Instead write your observations for the lab and submit along with the releavant figures. A few basic questions related to the lab are to be answered along with your lab report.

Data Converters – Some more properties

Equipment:
Obtain ADC0804 and DAC0808 chips from your TA. These are 8-bit ADC and DAC respectively.
The data sheets can be found at
http://www.national.com/mpf/DA/DAC0808.html#Order --> DAC
http://cache.national.com/ds/DC/ADC0804.pdf --> ADC

A typical circuit for ADC and DAC is shown below. This circuit should be sufficient for the lab requirements for this experiment.

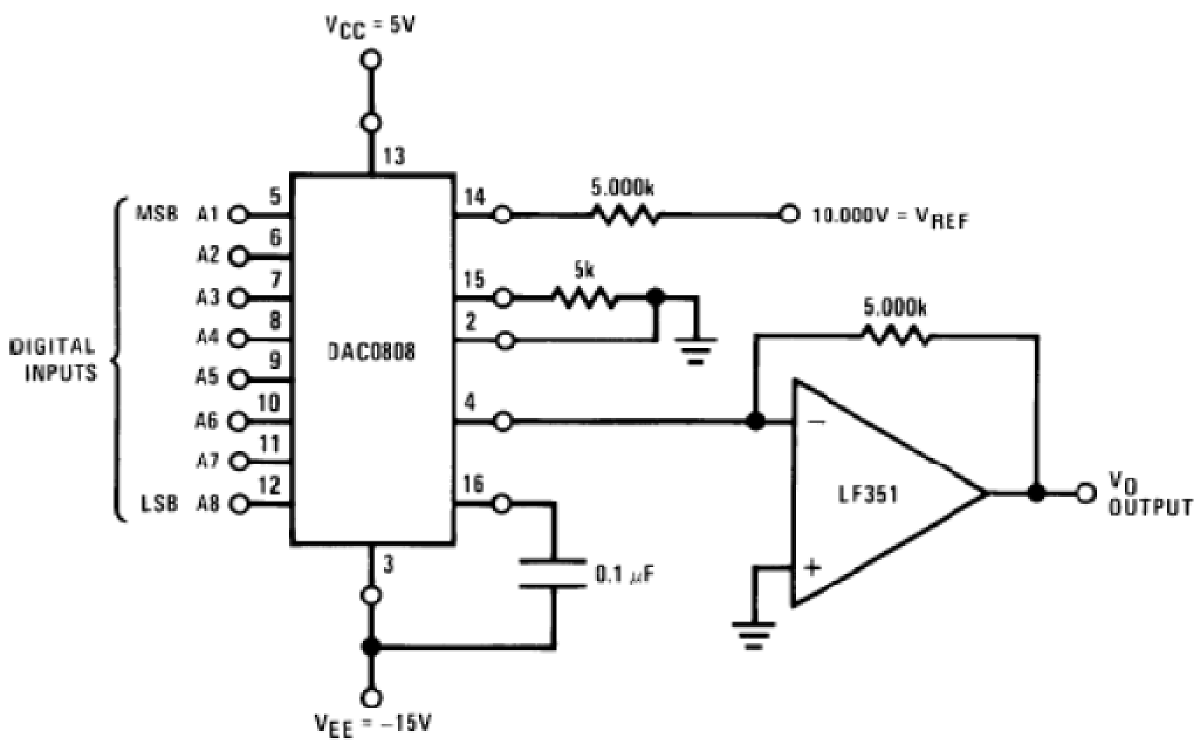

Figure 1: Typical DAC0808 Application.

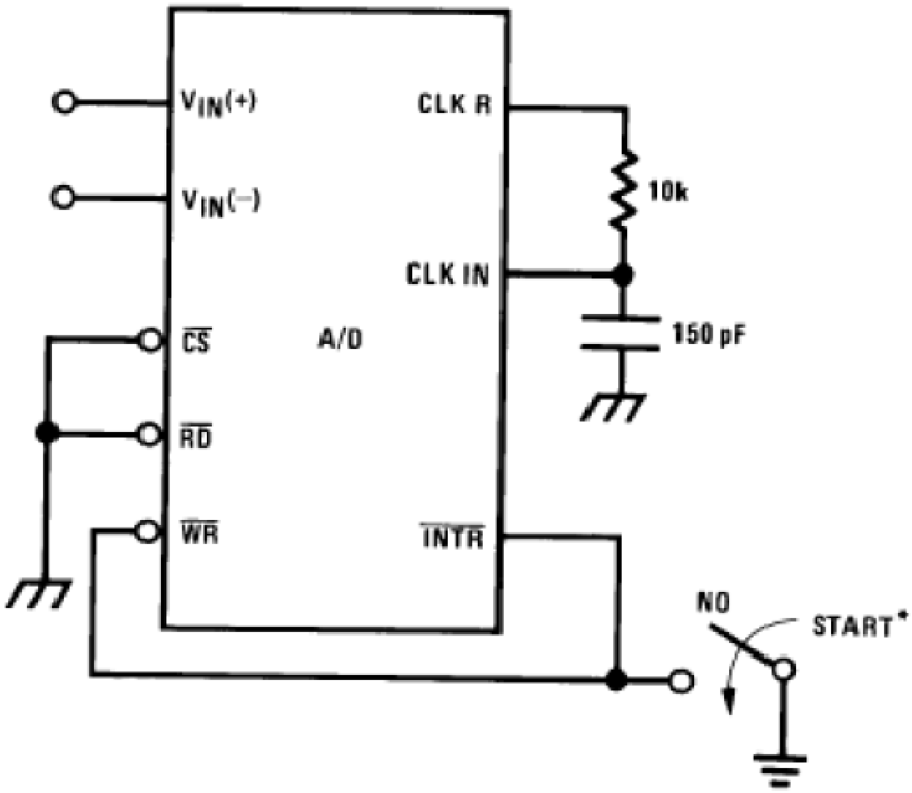

Figure 2: Typical ADC application.

Please note that ADC is WR’ and INTR’ are initially grounded and then opened for chip to start.
Also a voltage divider can be used at the Vin+ to prevent excess input cureent ..

Spectral Characterization

Any nonlinearities in the transfer characteristics of a data converter will affect the spectral performance of the data converter. Specifically, if a sinusoidal input is applied to the data converter, harmonic components will be present in the output.

For example, if the input to a data converter is

$$X_{IN}=X_M \sin(\omega t+\theta) \tag{13}$$

then the interpreted output will be of the form

$$X_{OUT}=A_0+A_1\sin(\omega t+\theta+\gamma_1)+\sum_{k=2}^{\infty} A_k \sin(k\omega t+\theta+\gamma_k) \tag{14}$$

where Ak is the magnitude of the kth harmonic component of the ouput. The terms in the right summand represent spectral distortion and is comprised of frequency components that are not present in the input signal. The THD is generally defined to be the total power in the second and higher harmonic terms relative to the power in the fundamental. That is,

$$THD=\frac{\sum_{k=2}^{\infty} A_k^2}{A_1^2} \tag{15}$$

This is often expressed in decibels as

$$THD_{dB}=10\log_{10}\left(\frac{\sum_{k=2}^{\infty} A_k^2}{A_1^2}\right) \tag{16}$$

Generally the contributions by the higher-order terms is negligible and the sum can be made over only the first few terms.

A second metric is often used to characterize the spectral performance and that is the spurious free dynamic range (SFDR). The SFDR is defined to be

$$SFDR=\frac{|A_1|}{\max_{1<k}\{|A_k|\}} \tag{17}$$

Usually the SFDR is expressed in dB as given by the expression

$$SFDR_{dB}=20\log_{10}\left(\frac{|A_1|}{\max_{1<k}\{|A_k|\}}\right) \tag{18}$$

The THD and the SFDR are generally measured by applying a sinusoidal excitation of near full-scale and then taking a large number of samples of the output waveform. From these samples, a Fourier Series representation of the output can be obtained and this Fourier Series representation is essentially that given in (14). The following theorem provides a practical method for obtaining the Fourier Series representation of a signal x(t) from samples of the signal.

Theorem: If a periodic signal x(t) with period T=1/f is band-limited to frequency hf and if the signal is sampled N times over an integral number of periods, N_P, then

$$|A_m| = \frac{2}{N}|X(mN_p+1)| \qquad \textbf{for } 0 \le m \le h\text{-}1$$

where $\langle X(k) \rangle_{k=1}^{N-1}$ **is the DFT of the sampled sequence** $\langle x(kT_s) \rangle_{k=1}^{N-1}$ **where T_S is the sampling period.**

The sampling period is thus given by $T_s = \dfrac{T \bullet N_p}{N}$ Thus, if x(t) is band-limited to hf, the magnitude of the coefficients of the Fourier Series Representation A_0, A_1, A_2, A_3, … are the magnitudes of the DFT elements X(0), X(N_P+1), X(2N_P+1), X(3N_P+1), … The Fast Fourier Transform (FFT) is a computationally efficient way for calculating the DFT, particularly when the number of samples is a power of 2. The FFT is a routine that is available in MATLAB.

Generally a prime number of periods of the input signal are sampled. It is critical that the hypothesis of the theorem be satisfied, that is, that the signal is sampled precisely over an integral number of periods of the excitation. Even a very small skew in the sampling requirements will cause major problems with using the DFT to obtain the Fourier Series coefficients.

Part 1 Time Quantization and Aliasing

Using the ADC -DAC configuration
a) Sample a sinusoidal signal with an amplitude of 5V pp at 5 times the Nyquist rate. Plot the resultant samples and compare with the original signal.You would need to measure the internal clock frequency for this part(which should be around 300k but may be slightly variable)
b) Repeat Part a) if sampled at only 2/3 the Nyquist rate .
c) Determine the aliased signal generated in part b) in the time domain.
d) Plot the input signal and the quantized signal and determine the quantization noise. Express the quantization noise in LSB.
d) Using the data acquisition card given to you sample the output of “ADC” and calculate the THD and SFDR for the wave.

Q1. Fill in these absolute maximum ratings for the DAC0808.Look for the values in your data
sheet.
Power Supply Voltage :
VCC :
VEE :
Digital Input Voltage, V5–V12 :

Applied Output Voltage, VO −11 :
Reference Current :
Reference Amplifier Inputs :

Q2. What should the relation between Vcc and Vref in ADC 0804 for proper functioning.

Q3.Fill in these absolute maximum ratings for the ADC0804.Look for the values in your data sheet.
Supply Voltage (VCC)
Voltage
Logic Control Inputs
At Other Input and Outputs