EE 337: SPI DAC Interface to Pt-51
Lab 8

Objectives

After completing this lab, you will be able to:

- use SPI protocol for interfacing an DAC (TLV5616) with Pt-51.
- use DAC (TLV5616) for waveform generation.
- use DAC (TLV5616) for basic waveform synthesis to generate attack, decay, sustain, release (ADSR) envelops to simulate tonal output from real instruments.

Background

We have used SPI (Serial Peripheral Interface) for interfacing ADC TLV1543 to Pt-51. In this week's experiment we shall use SPI to connect TLV5616 to Pt-51. The TLV5616 is a 12-bit DAC with 4-wire serial interface. The TLV5616 is programmed with a 16-bit serial string containing 4 control and 12 data bits.

In this experiment, we shall interface the DAC with Pt-51 to generate basic waveforms (ramp and sinusoidal) and observe them on DSO. We shall then use DAC and Pt-51 for waveform synthesis to generate ADSR envelops.

ADSR

In timer experiment, we generated musical notes and played using a speaker with driver using L293 as staccato, with each note sharply detached or separated from the others.

However, there are no real instruments that instantly start producing sound at some volume level and continue at exactly the same level until the sound stops. Usually, the volume ramps up (sometimes very fast, sometimes more gradually) when a note is played, and falls off slowly when the note is released. The period of time during which a musical tone is building up to some amplitude (volume) is called the **"Attack time", A** and the time required for the tone to partially die away is called its **"Decay time", D** . The time for final attenuation is called the **"Release time", R**. Many instruments also allow to hold

We would like to thank Texas Instruments for providing the DAC (TLV5616) for this experiment!

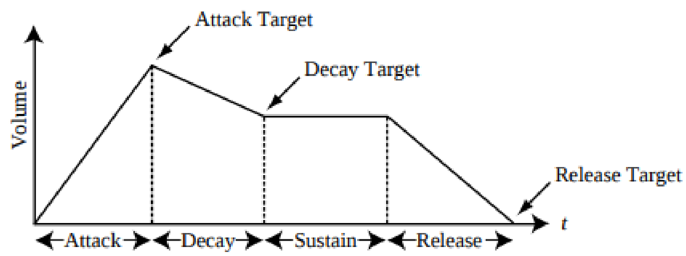

Figure 1: ADSR envelop

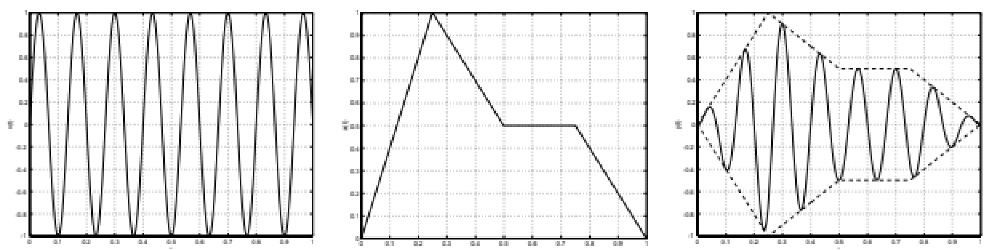

Figure 2: Amplitude modulated ADSR envelope

the tone for a period of time, which is known as the **"Sustain time", S**. This is illustrated in Fig. 1. The durations and slopes (can be linear or exponential) of A, D, S, and R can be different for different instruments.

The envelop can be generated by point-by-point multiplication of the tone and the ADSR waveform resulting in ADSR envelope as shown in Fig. 2, that closely resembles the sound of a real instrument. Please refer the paper titled "Computer Music in Undergraduate Digital Signal Processing" uploaded with this document for more details.

Details regarding the DAC interfacing

The TLV1516 uses 4 wire lines for serial interfacing. The SCLK is used for clocking the chip and synchronizing it. DIN is the pin through which the 4 bit control signals along with the 12 bit data signal is received serially. The Frame Sync signal, FS, should be issued as per the timing diagram requirement to get the chip running. The chip select signal is used to select the chip when we connect and communicate with multiple chips using SPI protocol.

Note that here the data sampling happens at the falling edge of the SCLK signal. Therefore while making the configurations, also make the necessary changes to meet this requirement.

In the power down mode, all the amplifiers within the TLV5616 are disabled.

The output voltage (full scale determined by external reference) is given by:

$$Vout = 2 * V_{\text{ref}} * \frac{V_{digital\_in}}{2^n} \tag{1}$$

where V_{ref} is the reference voltage and $V_{digital\_in}$ is the digital input value within the range of 0 to $2^{(n-1)}$, where n = 12 (bits).

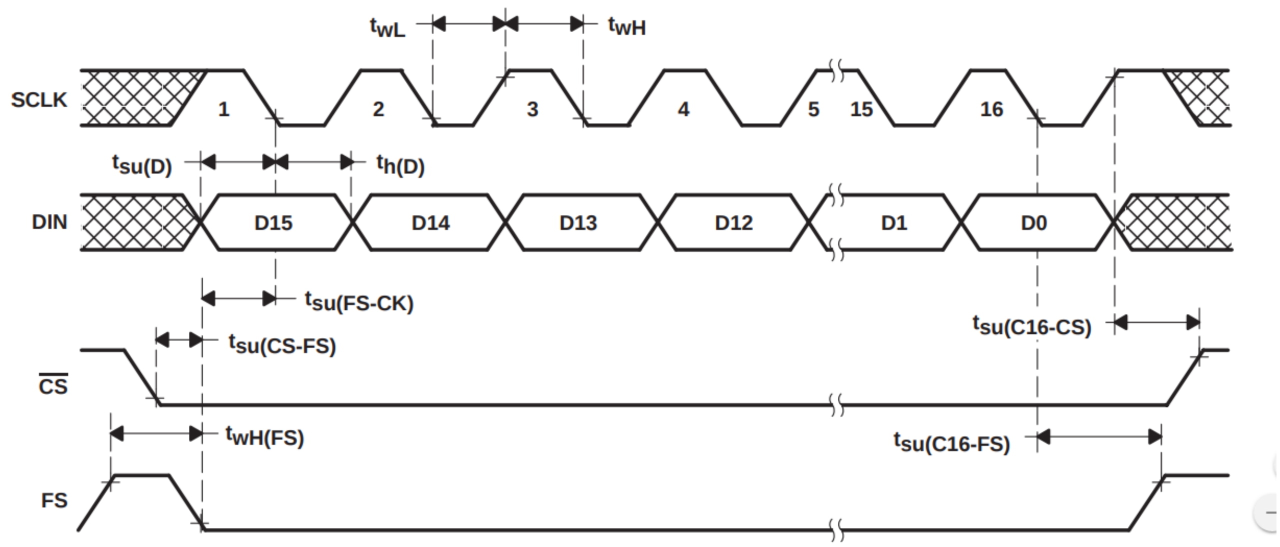

Figure 3: Timing diagram.

D15	D14	D13	D12	D11	D10	D9	D8	D7	D6	D5	D4	D3	D2	D1	D0
X	SPD	PWR	X	New DAC value (12 bits)											

X: don't care
SPD: Speed control bit. 1 → fast mode 0 → slow mode
PWR: Power control bit. 1 → power down 0 → normal operation

Figure 4: Data format.

Homework

1. Read the datasheet of TLV 5616 for details from `http://www.ti.com/product/TLV5616/technicaldocuments`,the paper "Computer Music in Undergraduate Digital Signal Processing" uploaded with this document and also this document.

2. Most common and simplest way of generating sine wave is to use a look up table (LUT). Using MATLAB or equivalent software, compute the 32 samples of the sinusoid. Since the DAC is 12-bit, the LUT values should have 12 bit resolution. Note that the sine wave to be generated is **unipolar**.

3. Calculate the theoretical analog output values that can be obtained from the DAC for the following digital inputs(Vref = 2.5V and VCC = 5V):

Digital input	Expected output voltage
0x000	0
0x400	01.25
0x800	2.5
0xFFF	5

4. Understand the timing diagram requirements and find the values of the following registers for interfacing the TLV5616 DAC.

Registers	Value
IEN1	0x04
SPCON	0x75

Lab Work

1. **Interfacing TLV5616 to Pt-51**

   (a) Write a program for interfacing DAC TLV5616 with Pt-51.

   (b) Testing your code: Make the connections as shown in Fig. 5. The serial DAC input data and the control signals are sent through I/O port 3 of the controller. The serial data is sent on the MOSI line from Pt-51, with the serial clock output on the SCLK line. P3.4 and P3.5 are configured as outputs to provide the chip select ($\overline{CS}$) and Frame Sync(FS) signals for the TLV5616.

   (c) Test the analog output for the digital values corresponding to the test cases mentioned in the homework part 3.

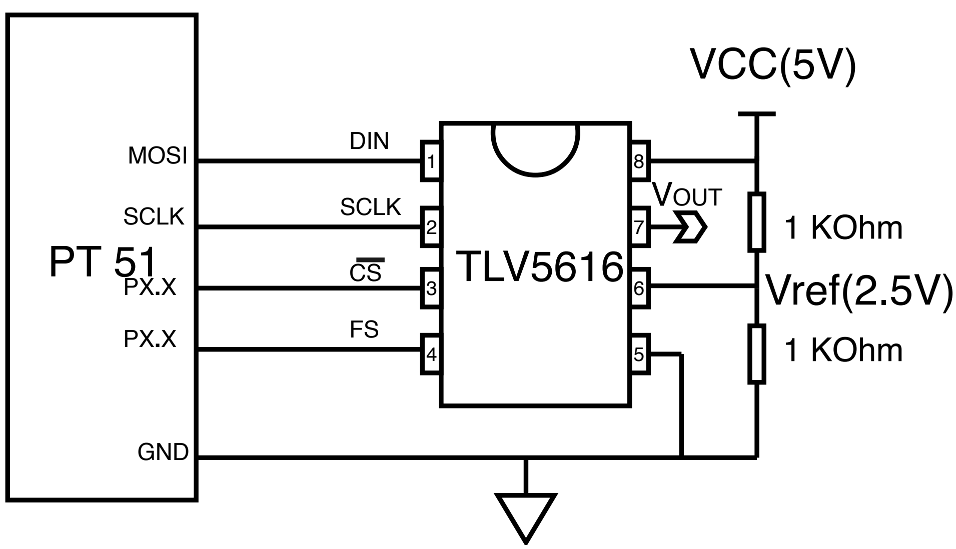

Figure 5: DAC connection diagram.

2. **Generating triangular and sine wave signals**

   (a) Write a program to generate a symmetrical triangular signal that varies from 0 to 5V with a period of 12sec (6sec+6sec).

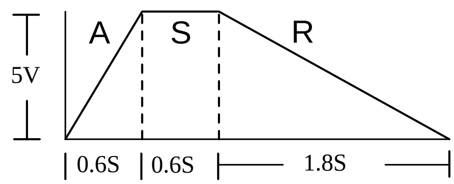

Figure 6: ADSR Waveform to be generated

(b) Observe the waveform on the DSO. Check if the period and amplitude of the triangular waveform is as expected.

(c) Modify the program to generate the waveform shown in Fig. 6.

(d) Use the look up table that you generated in homework problem and generate the sine wave tone with frequency 240Hz. Observe on DSO. Play the tone on the PC speaker or your earphone.

(e) Use the given `notes.c` code to generate "Sa Re Ga Ma " square wave tones (You have done this using assembly in Timer experiment). Play all the notes in ascending and then descending order through PC speaker or your earphone.

3. **Playing ADSR modulated envelops**

To keep it simple, we shall use square wave tone to generate ADSR envelopes as multiplication with square wave is easier and we had already generated. Moreover, the sound with square wave tones is more pleasant than single sine wave tone that you would have noticed in earlier part.

(a) Generate the ADSR envelops with square-wave tones using the waveform shown in Fig.3a. Use the `notes.c` program provided to generate the musical notes.

(b) Generate the envelopes by multiplying the ADSR waveform generated in (2b) with the square wave tones generated above. You will have to use appropriate scaling factor so that the envelops' amplitude is well within the 0-5V limits. Observe them on DSO.

(c) The output envelope will be the amplitude modulated envelope with the square tone as carrier signal and ADSR waveform as modulating signal. **Note that the DAC is unipolar and hence the envelop will not be centered around 0V unlike the one shown in Fig. 2.**

(d) Now play the notes (envelopes). Do you feel any difference between the square wave tones and ADSR tones musically ?

(e) Change the attack and decay time and see the effect.

(f) You may also try to obtain the waveforms shown in Fig.2 with sine wave tones generating full envelopes. [Optional!]