EE442/EE592 Real-Time Digital Signal Processing
Laboratory 2: Real-Time Programs

1 Opening

1.1 Documentation

Documentation for Domain Technologies' Debug-56K, *Debug-56K User's Manual*, is available in laboratory.

2 PASS.ASM

2.1 Executing PASS

STEP 1: Download all files comprising the **Modified Pass Pack** for the DSP56302EVM from the course website (Freescale DSP56K Code tab).

STEP 2: Assemble the code and download PASS.CLD to the DSP board. In Debug56K, make sure the baud rate for downloading onto the DSP board is set to 56000bps and memory verify is disabled.

STEP 3: Attach one audio cable from the sound card's audio output to the DSP board's audio input (line in). Attach another audio cable from the DSP board's audio output (line out) to the multimedia speakers.

STEP 4: Play an audio file out the sound card and see if the DSP board passes the audio to the speakers.

2.2 Changing the Sample Rate

Control words for the various sample rates that the codec is capable of sampling at are defined in **`ADA_EQU.ASM`**.

```
SAMP_RATE_48   equu   $003000        ; 48   kHz sample rate
SAMP_RATE_32   equ   $001800         ; 32   kHz sample rate
SAMP_RATE_27   equ   $001000         ; 27   kHz sample rate
SAMP_RATE_16   equ   $000800         ; 16   kHz sample rate
SAMP_RATE_9    equ   $003800         ;  9.6 kHz sample rate
SAMP_RATE_8    equ   $000000         ;  8   kHz sample rate
```

In order to change the sample rate using one of the defined control words, you need to edit the line that reads:

```
CTRL_WD_12  equ   NO_PREAMP+HI_PASS_FILT+SAMP_RATE_48+STEREO+DATA_16   ;CLB=0
```

In the line above, replace `SAMP_RATE_48` with `SAMP_RATE_8`. Assemble, load, and execute. Get familiar with the sounds as you alter code. Clearly lowering the sampling rate, lowers the bandwidth of the signal (according to Nyquist theory). One of the most important lessons you will learn from this course is recognizing program bugs from the sounds they produce.

3 Simple Delay / Echo

Complete Program 4 (Simple Delay) in Chapter 4 of the textbook. Be sure to also complete the simple digital echo.

4 FIR.ASM (Single Channel / Mono)

Complete Program 5 (mono FIR filter) in Chapter 4 of the textbook. The digital filter is a simple Moving Average (MA) filter which acts as a lowpass filter. Can you verify the lowpass nature of the filter? Try designing another FIR filter using MATLAB (see fir1) and use those filter coefficients in this program.

5 FIR.ASM (Dual Channel / Stereo)

Complete Program 6 (stereo FIR filtering) in Chapter 4 of the textbook.