Midterm — March 24, 2004

Write and sign the honor pledge:

1. (2 pts) Print your name on <u>every</u> sheet.

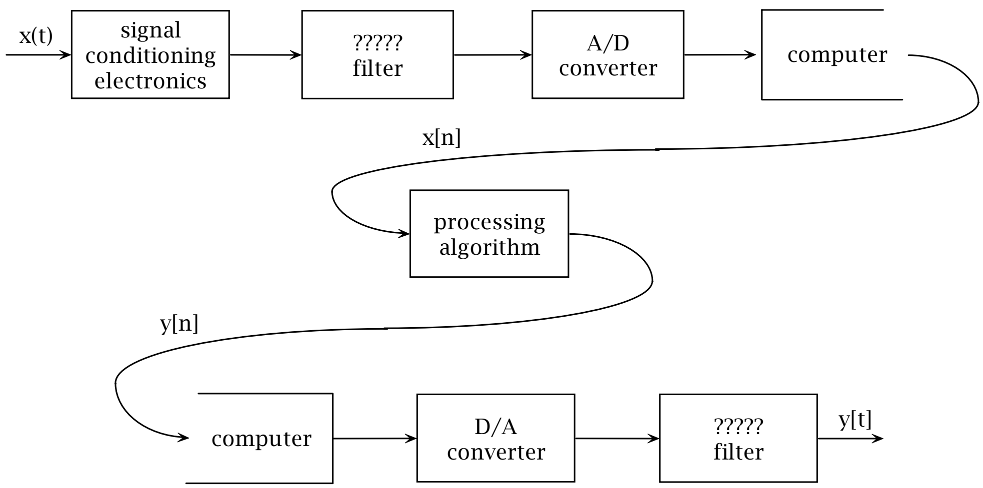


Figure 1: DSP paradigm.

2. (8 pts)

   Figure 1 shows the basic components that typify a DSP processing system.

   (a) What is the purpose of the filter at the input of the A/D converter?

   (b) What is the purpose of the filter at the output of the D/A converter?

3. (12 pts) Assume the use of a *B*-bit word size.

   (a) What is the value of the largest unsigned integer value that can be represented?

   (b) What is the value of the largest positive two's complement integer value that can be represented?

   (c) What is the value of the most negative two's complement integer value that can be represented?

   (d) What is the value of the largest unsigned Q(B-1) value that can be represented?

   (e) What is the value of the largest positive Q(B-1) value that can be represented?

   (f) What is the value of the most negative two's complement Q(B-1) value that can be represented?

4. (10 pts) Write the equations that define the discrete Fourier transform (DFT) and the inverse discrete Fourier transform (IDFT). Do not use the letter *W* in your answer.

   Forward transform:

   Inverse transform:

5. (8 pts) It is planned to sample a data stream at 8 kHz. A data set of size N values is desired in order to have a frequency spacing of 1/8 Hz when the DFT of the N samples is formed. What should the value of N be?

6. (8 pts) I have a data set of 500 sample values of a 1000 Hz sinewave. These were obtained using a sample rate of 8 kHz. Using MATLAB I take the DFT with the result being in the `X` array. I use `X=fftshift(X)` to reposition the data.

   (a) What is the frequency in Hz associated with the `X(1)` value ?

   (b) What is the frequency in Hz associated with the `X(500)` value?

7. (8 pts)

   Assume use of a 16-bit computer word. Given the binary bit pattern:

   0000 1111 0000 0000

   (a) what is the value if we interpret this an unsigned integer?

(b) what is the value if we interpret this as a two's complement integer?

(c) what is the value if we interpret this as a Q8 unsigned integer?

(d) what is the value if we interpret this as a Q15 two's complement value?

8. (8 pts)

   Assume use of a 16-bit computer word. Given the binary bit pattern:

   1111 0010 0000 0000

   (a) what is the value if we interpret this an unsigned integer?

   (b) what is the value if we interpret this as a two's complement integer?

   (c) what is the value if we interpret this as a Q8 unsigned integer?

   (d) what is the value if we interpret this as a Q15 two's complement value?

9. (9 pts) During the semester we designed IIR filters based on analog prototypes. To do this we used MATLAB's FDAtool. FDAtool offers a choice of four filter prototypes. These are: Butterworth, Chebyshev type 1, Chebyshev type 2, and elliptic. These prototypes have differing transfer function characteristics in terms of monotonicity and ripple.

   For lowpass use what are the distinguishing monotonicity and ripple characteristics of each prototype's transfer function?

Butterworth
- passband characteristic?
- stopband characteristic?

Chebyshev type 1
- passband characteristic?
- stopband characteristic?

Chebyshev type 2
- passband characteristic?
- stopband characteristic?

Elliptic
- passband characteristic?
- stopband characteristic?
- transition band characteristic?

10. (10 pts) We made extensive us of TI's C compiler in the lab exercises. Although this compiler meets the ANSI standard there are some special aspects that we need to keep in mind when using it.

    (a) How many bits per `char`? ________
    (b) How many bits per `int`? ________
    (c) How many bits per `long`? ________
    (d) How many bits per `float`? ________
    (e) How many bits per `double`? ________

11. (4 pts) The C compiler has two memory models that can be used. These are named the

    (a) _______________ memory model,

    (b) _______________ memory model.

12. (12 pts) When writing hand optimized functions for use by C we need know the calling conventions used by the C compiler. The IIR functions that were used in lab used a call of the form:

```
short iircas5(DATA* x, DATA* h, DATA* r, DATA* db, ushort nh, ushort nx);
```

    There are 6 parameter values that are passed. Specify the C5510 registers in which they are passed:

    (a) `*x` __________

    (b) `*h` __________

    (c) `*r` __________

    (d) `*db` __________

    (e) `nh` _________

    (f) `nx` _________

13. (4 pts) Given a function, $f(x)$ what condition must it satisfy if it is linear?

14. (16 pts) We have a symmetric FIR digital filter that theory indicates has a delay of 96 sample times. The sample rate is 48000 Hz. The nominal center frequency of the filter is 1500 Hz.

    (a) A transfer function measurement, magnitude and phase, is made over the range of frequencies from 750 Hz to 2250 Hz. Sketch the phase as it would be measured in cycles over this frequency range. Use a vertical axis going from -1/2 cycle to +1/2 cycle. Assume that the measured phase at 1500 Hz is 0 cycles.

(b) Indicate the values of phase that would be measured at 1125 Hz, 1500 Hz, and 1875 Hz. If only the measurements made at these three frequencies were used to form an estimate of the group delay what would be the estimated group delay?

15. (8 pts) I sample the waveform $127\cos(2\pi 800t + \pi/3)$ using a sample rate of 300 Hz. Next I take one second's worth of samples and form the DFT.

(a) At what frequency should I expect to see this waveform's negative frequency component?

(b) At what frequency should I expect to see this waveform's positive frequency component?

16. (8 pts) You are given a digital filter described using a biquadratic transfer function of the form

$$H(z) = \frac{b_0 + b_1 z^{-1} + b_2 z^{-2}}{1 + a_1 z^{-1} + a_2 z^{-2}}$$

and a sample rate of f_s Hz.

   (a) What is the value of $H(z)$ at 0 Hz?

   (b) What is the value of $H(z)$ at $f_s/2$ Hz?

17. (8 pts) In modeling quantization noise we made four assumptions that allowed us to simplify our model. These assumptions have been found to reasonably model reality when working with complex waveforms such as speech and music. The four assumptions were:

   (a)

   (b)

   (c)

(d)

18. (12 pts)

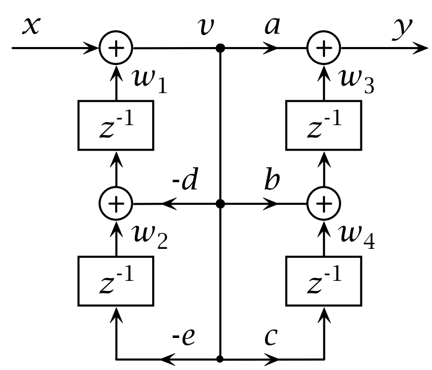


Figure 2: A filter section.

Draw the block diagram of the transposed version of the filter section shown in Figure 2.

19. (6 pts)

    (a) For "simple" oversampling A/D converter how many bits of equivalent performance are obtained per each doubling of the input sample rate? By equivalent performance I mean how many additional bits (perhaps fractional) would need to be added to a "normal" A/D converter in order to provide the same quantization noise to signal signal-to-noise ratio.

    (b) For the first order sigma-delta A/D converter how many bits of equivalent performance are obtained per each doubling of the input sample rate?

(c) For the second order sigma-delta A/D converter how many bits of equivalent performance are obtained per each doubling of the input sample rate?

20. (4 pts) Assume a B-bit A/D converter. The analog input voltages can go plus and minus. The converter generates digital output values using the two's complement number format.

    The most negative input analog input is $-V_m$ volts (mid-tread) and produces digital output value -2^{B-1}.

    What is the voltage change (Δ) at the input that produces precisely a one count change in the digital output?

21. (4 pts) The quantization error of the output of an amplitude quantizer was modelled as being uniformly distributed. Using this assumption an expression was derived that related the variance, σ_e^2, of the quantization error to the quantizer step size, Δ. What was this relation?

$$\sigma_e^2 =$$

22. (4 pts) An expression was derived in lecture that relates the signal-to-noise ratio of a signal waveform and the quantization noise to the number of the bits in an amplitude quantizer.

    Adding one bit to the size of the quantizer results in a change of how many dBs in the signal-to-noise ratio?

23. (6 pts) Several times this semester the equation form of Parseval's theorem for the DFT was encountered, both in homework and in lecture. Write down the equation that describes Parseval's theorem and indicate whether what you wrote corresponds to the standard definition of the DFT or to the alternate form that I claim to prefer.

24. (8 pts) Draw the block diagram of the transposed direct form two (TDF2) biquad section. Remember to label the coefficients.

25. (4 pts) Not counting DMA data transfers the C5510 can simultaneously perform up to

_______ memory reads and _______ memory writes.

26. (2 pts each)

(a) The amount of DARAM present on the C5510 is ________ Kwords.

(b) The amount of SARAM present on the C5510 is __________ Kwords.

(c) The C5510DSK CPU minimum cycle time is _______ ns.

(d) The AIC23 codec A/D and D/A word size is ________ bits.

(e) The C5510 has _______ auxiliary registers.

(f) The eXtended auxiliary registers each contain ________ bits.

(g) The C5510 has _______ temporary registers.

(h) The technical name for the norm that we used to scale for overflow in our IIR filter designs is the "__________ norm".

(i) The C5510 always fetches _______ bytes from program memory every clock interval.

(j) C5510 instructions vary in length and use from 1 to _______ bytes.

(k) There are _______ accumulators in the C5510.

(l) Each accumulator is made up of _______ registers.

(m) The total number of bits contained in the registers forming an accumulator is ________.

(n) There are _______ status registers in the C5510.

27. (6 pts) The C5510 supports three address spaces these are named

(a)

(b)

(c)

28. (4 pts) The transfer function of a biquad section is

$$H(z) = \frac{b_0 + b_1 z^{-1} + b_2 z^{-2}}{1 + a_1 z^{-1} + a_2 z^{-2}}.$$

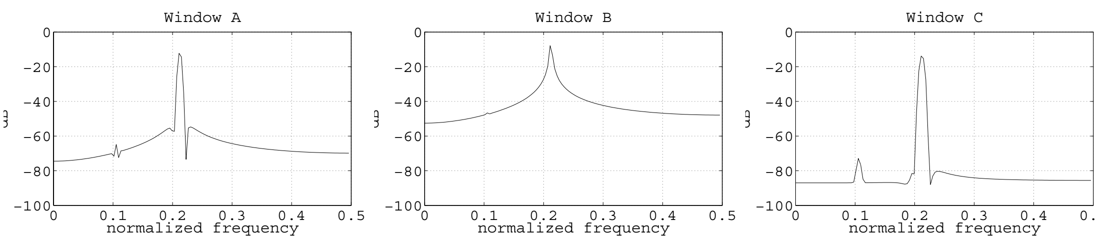
To have a stable filter section there are limits which the a_1 and a_2 values must meet (not necessarily simultaneously). These are:

$< a_1 <$

and

$$< a_2 <$$

29. (6 pts) In the realtime FFT lab exercise we used three window functions to demonstrate the use of windowing to reduce leakage. The above figure shows the effects that each window has on the same data set. The names of these three windows are:



(a) Window A:

(b) Window B:

(c) Window C:

30. (8 pts) What is the transfer function of the filter shown in Figure 3?

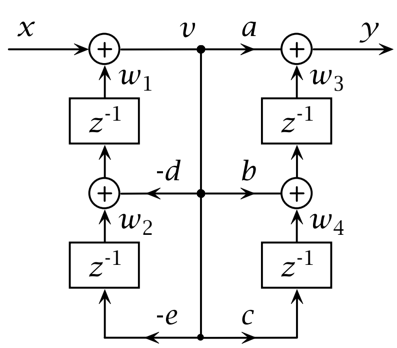


Figure 3: Filter block diagram.

31. (8 pts) In lab exercise 4 you worked with a direct digital frequency synthesizer that used a table containing 256 samples of a cosine wave. A 48 kHz clock was used to determine the rate at which samples were sent to the D/A converter. An unsigned 32-bit word was used as a phase accumulator. Every 1/48000th of a second a value was added into this accumulator. The top 8 bits of the phase accumulator were used to address a value in the cosine array. The value added to the accumulator was termed the frequency tuning value, FTV.

    (a) What is the frequency tuning value that will result in an output frequency as close as possible to 1000 kHz?

    (b) What is the smallest increment in Hz that can be made in the output frequency?

32. (8 pts) If we multiply a 16-bit Q14 format number by a 16-bit Q12 number on the TI C5510 with the FRCT bit set to 1.

(a) The result in the high-low portion of the accumulator is a 32-bit Qn number. What is the value n?

(b) The result is moved from the high word of the accumulator. What is the Qn value for the moved 16-bit value?

33. (20 pts) An addressing exercise.

The initial state of each instruction is shown here . . .

0x020105	0x21
x = 0x020106	0x30
0x020107	0x40
0x020108	0x50
0x020206	0x60

XAR1 0x020106

XDP 0x020106

T0 2

.dp x

Below, write down the state after each instruction . **Only those boxes whose contents have changed are to be filled in!.**

	AR1	AC0	T1	0x020106
MOV @(x+1),AC0				
MOV @(x+0x80),AC0				
MOV T0,*AR1+				
MOV *(#x),AC0				
MOV #4,@(x+128)				
MOV *(AR1+T0),T1				
MOV *AR1(T0),AC0				
MOV *AR1(#0x100),T1				
MOV @(x+129),AR1				
MOV *+AR1(#-1),AC0				

34. (8 pts) In laboratory exercise 7 we use a 32-bit FFT code from the TI DSPlib. This code does not scale its results (i.e., the 1/N is associated with the inverse transform). The sample rate is 48000 Hz and the FFT size is 1024 values. The display (positive frequencies) shows a single line in fft bin (at fft index counting starting at 0) 256 with amplitude 262144. A 16-bit A/D converter was used for the data source and was calibrated so that a value of 2^{15} corresponds to 1.0 Volts.

    (a) What was the frequency in Hz of the sine wave that was sampled?

    (b) What was the peak amplitude of the sinewave in volts?

35. (12 pts) Over a period of weeks we developed our own assembly language code for evaluating the arctangent of y/x where we were given both y and x rather than just their ratio. Having both x and y allows determination of angles over the full range going from -half cycle to +half cycle (or from -180 degrees to +180 degrees or from -pi radians to +pi radians depending on the units du jour).

    The programming task was divided into separate tasks.

    (a) The first task was to map the starting x and y values into values a and b such that both a and b were positive and $0 \le b/a < 1$. In order to do this mapping we divided the complex plane into _______ pie shaped wedges.

    (b) Because the C5510 does not possess a divide instruction we used the method named: ____________________________ to calculate the value of $1/a$.

    (c) In order to use this method effectively we jointly scaled a and b so that a was in the range between _________ and _______.

(d) In order to conserve bits we calculated $1/a$ using unsigned fractions.

The largest Q value that will hold $1/a$ as an unsigned integer is Q____

(e) The value of b/a was next formed, saturated and stored as a two's complement Q15 value. A polynomial approximation was then used to calculate the arctangent. The name of the approximation technique used to determine the polynomial coefficient values was ________________ approximation.

(f) A clever technique named _______________ method was used to evaluate the polynomial. Using this technique reorder the operations require to evaluate the polynomial shown below:

$$f(x) = a_5x^5 + a_4x^4 + a_3x^3 + a_2x^2 + a_1x^1 + a_0$$

$f(x) =$

36. The bit map of timer prescaler register, PRSCn, is shown in Figure 4.

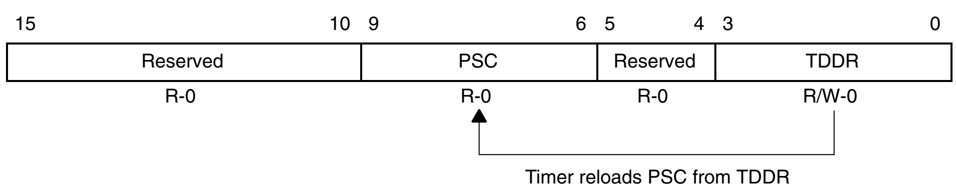


Figure 4: Timer n PRSC bit usage.

When running in a 200 MHz processor it may be necessary to slow up the counter. This can be done by placing by values into the prescaler. The prescaler can divide by factors of **one up through 16**. The following lines of code set use counter 0 to time the execution of the `test_func` function.

```
......

PRSC0=0x0082;
count0=(~TIM0);
test_func();
count1=(~TIM0);
```

```
PRSC0=0x0000;
count2=(~TIM0);
test_func();
count3=(~TIM0);

.....
```

Given that `test_func()` has an execution time of 240 cpu cycles,

(a) `count1 - count0 =` ________

(b) `count3 - count2 =` _________

37. The bit map of clock mode register, `CLKMD`, is shown in Figure 5. Assume that the input clock to the C5510 is 48MHz:

15	14	13	12	11–7
Rsvd	IAI	IOB	TEST (keep 0)	PLL MULT
	R/W – 0	R/W – 1		R/W – 00000

6–5	4	3–2	1	0
PLL DIV	PLL ENABLE	BYPASS DIV	BREAKLN	LOCK
R/W – 00	R/W – 0	R/W – pin	R – 1	R – 0

Legend:

R Read-only access
R/W Read/write access
– X X is the value after a DSP reset. X = pin indicates that the reset value depends on the signal level on the CLKMD pin.

Figure 5: Clock mode register bit usage.

$$\text{Ouput frequency} = \frac{\text{PLL MULT}}{(\text{PLL DIV} + 1)} \times \text{Input frequency}$$

(a) If `CLKMD=1 1 1 1 00110 01 1 00 0 1`, what is CPU clock rate? _________ MHz

(b) If we want to generate a CPU clock rate of 80 MHz, change the necessary bits in CLKMD. The new `CLKMD =` ______________________________

38. Consider the recursive oscillator shown in Figure 6.

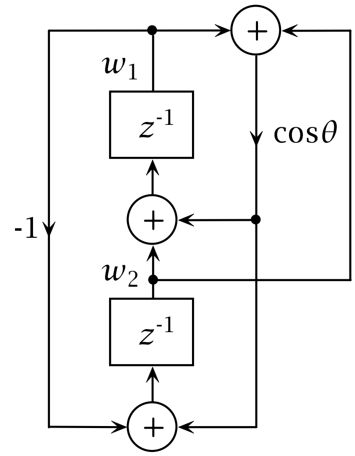


Figure 6: Second order recursive oscillator.

(a) We can use a matrix to represent the updating of the state of registers w_1 and w_2. What is it?

$$\begin{bmatrix} w_1(n+1) \\ w_2(n+1) \end{bmatrix} = \begin{bmatrix} \qquad\qquad\qquad\qquad\qquad\qquad \end{bmatrix} \begin{bmatrix} w_1(n) \\ w_2(n) \end{bmatrix}$$

(b) We can also use DDS to generate sinusoids. What is the main advantage of using this oscillator over DDS?

____

39. We use the same DSK to implement same filter with DF2 and TDF2 implementations.

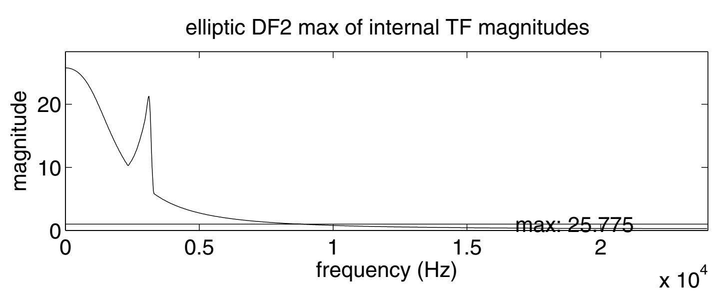


Figure 7: Across stage maxima for direct form 2 elliptic filter implementation.

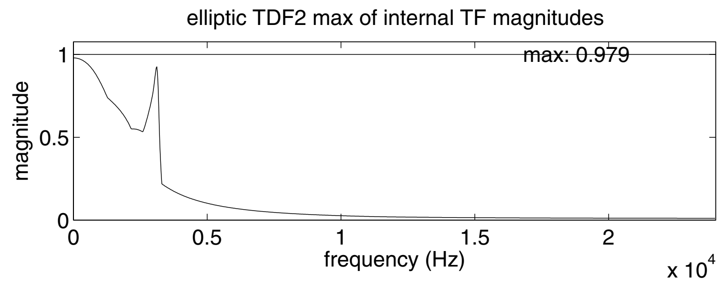


Figure 8: Across stage maxima for transposed direct form 2 elliptic filter implementation.

Suppose we ran the TDF2 implementation first, and we found that the maximum input peak-peak voltage we can avoid overflow problem at all frequencies is 4 Volts. Now we run DF2 implementation.

What is the maximum input peak-peak voltage that can avoid overflow at all frequencies?

The maximum safe (no overflow or saturating) input is _____________ Volts