Midterm — November 20, 2002

Write and sign the honor pledge.

1. (2 pts) Print your name on *every* sheet.

2. (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:

3. (10 pts)

   The transfer function of a bi-quadratic filter section is

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

   In lab we implemented lowpass filters and, somewhat arbitrarily, normalized the gain of each section to unit magnitude at 0 Hz. The *multiplicative* normalization factor that we used was:

   $K_{\text{lowpass}} =$

If we were to implement a highpass filter in the same manner and normalize the biquad sections to have unit magnitude at their highest frequency the *multiplicative* normalization factor would be:

$K_{\text{highpass}} =$

4. (10 pts) For the filter section in Figure 1 write the equation for the transfer function $W_3(z)/X(z)$.

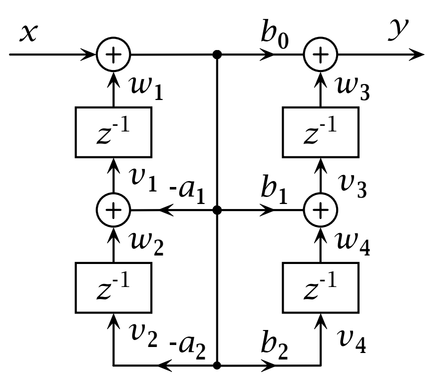

Figure 1: Filter section.

5. (10 pts) The matrix state equations

$$\begin{aligned} \mathbf{w}(n+1) &= \mathbf{A}\mathbf{w}(n) + \mathbf{B}\mathbf{x}(n) \\ \mathbf{y}(n) &= \mathbf{C}\mathbf{w}(n) + \mathbf{D}\mathbf{x}(n) \end{aligned}$$

can be used as the basis of a formal procedure to determine filter impulse responses in the time domain and transfer functions in the frequency domain.

Write the **A**, **B**, **C** and **D** matrices for the filter section shown in Figure 2.

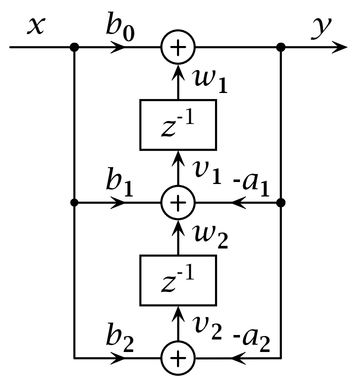

Figure 2: A biquad filter section

6. (10 pts) In an early lab exercise we investigate the implementation and operation of a direct digital synthesizer (DDS) such as shown in Figure 3. The

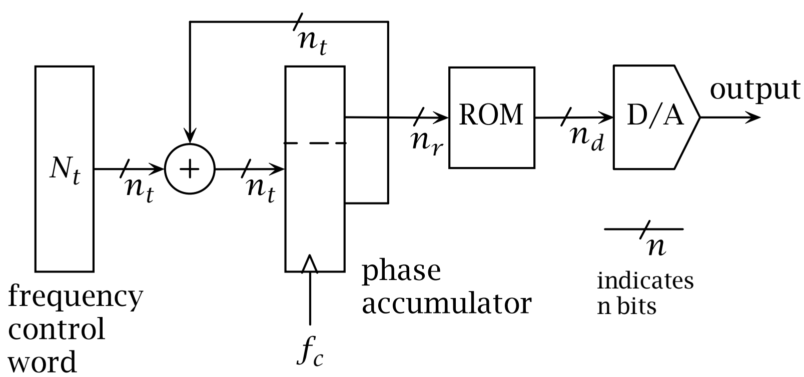

Figure 3: Basic direct digital synthesizer block diagram

lower case n's indicate number of bits, the upper case N_t represents an unsigned integer value representable using n_t bits. DDS devices are increasingly finding application in radio applications. One such device is the Analog Devices AD9857 quadrature upconverter. In this device the clock frequency, f_c, can be as high as 200 MHz. This device uses a frequency control word and phase accumulator having 32 bits.

(a) What decimal frequency control word value would most closely (i.e., most accurately) in frequency generate a sine output at 800 kHz using a clock frequency of 150 MHz.?

(b) What is the smallest increment in frequency that can be generated using this clock frequency and at what frequency?

7. (10 pts) The Motorola DSP56303 has a number of addressing modes. Given the contents of R1 = 123, N1 = 5 and M1 = $FFFF what are the effective source value memory addresses for the following instructions?

```
move  x:(R1),x0

move  x:(R1+N1),x0

move  x:(R1+2),x0

move  x:-(R1),x0

move  x:(R1)+,x0
```

8. (10 pts) For the TI C5402 what is the purpose or function the following status bits:

   (a) `OVM`

   (b) `SXM`

   (c) `SST`

   (d) `FRCT`

9. (12 pts) Using hexadecimal express the following numbers:

   (a) 1.75 using Q2.5 notation

   (b) -1.75 using Q2.5 notation

(c) 23 using Q7.0 notation

(d) -23 using Q7.0 notation

(e) 0.5 using Q0.15 notation

(f) -1.0 using Q0.15 notation

The Qm.n notation used here represents n bits to the right of the binary point and m bits to the left not including the sign bit. There are a total of n+m+1 bits in a word.

10. (10 pts) The MATLAB `kron` function is extremely useful in simulations involving upsampling. What is the result of

```
kron([1 0 -1],[1 2 3]);
```

11. (10 pts) Given a data set of 10 samples taken at a rate of F_s Hz. I append 7 zero values to each sample obtaining a set of 80 sample values. I then form the DFT of the resulting set of values.

    (a) What is the frequency spacing between consecutive values in the DFT?

    (b) What is the effective Nyquist frequency of the modified set of samples?

12. (10 pts) Given a data set of 10 samples taken at a rate of F_s Hz. I append 70 zero values to the end of the set obtaining a set of 80 sample values. I then form the DFT of the resulting set of values.

    (a) What is the frequency spacing between consecutive values in the DFT?

(b) What is the effective Nyquist frequency of the modified set of samples?

13. (2 pts each) Which processor and/or associated EVM (NEITHER or TI or MOT or BOTH)

(a) has a 16-bit word size? ______

(b) has the CODEC with the highest supported sample rate? ______

(c) uses three registers to form an accumulator? ______

(d) has two banks of dual access random access memory? ______

(e) supports convergent rounding? ______

(f) has two accumulators named A and B? ______

(g) supports circular addressing? ______

(h) supports only fractional multiplication? ______

(i) has a one instruction full divide instruction? ______

(j) has a protected execution pipeline? ______

(k) has a multiply and accumulate instruction? ______

(l) has its stack growth toward increasing addresses? ______

(m) has its stack growth toward the low addresses? ______

(n) maps processor registers into memory address space? ______

(o) supports memory to memory data transfers? ______

(p) has a pre-increment indirect addressing mode? ______

(q) has a pre-decrement indirect addressing mode? ______

(r) has two arithmetic-logic units that can operate in parallel? ______

(s) supports bit-reverse addressing? ______
(useful when implementing radix-2 FFT algorithms)

14. (9 pts) We discussed three methods that can be used to determine polynomial function approximations. These were: Taylor series, least mean squares and Chebyshev approximation. Which

    (a) produces an equiripple approximation error?

    (b) generally places its approximation error near the end of the interval being approximated over?

    (c) for a given error tolerance generally requires the most terms?

15. (10 pts) We made use of several MATLAB functions in our homework. What do to the following functions do?

    (a) `freqz`

    (b) `fftshift`

(c) `kron`

(d) `tf2sos`

16. (12 pts) For the IIR lab exercise we considered the use of four filter types offered us by MATLAB.

    (a) Which filter type (name) did we not use because it required an excessive number of poles?

    (b) Which filter type (name) has equiripple only in the passband?

    (c) Which filter type (name) has equiripple only in the stop band?

    (d) Which filter type (name) has equiripple in both the passband and the stopband?

    (e) Which filter type (name) has the smallest transition band for a given order?

    (f) Which filter type (name) has the maximally flat passband?

17. (10 pts) Figure 4 shows the pole and zero locations for the Elliptic filter design we used for the IIR lab exercise. The letters `a` through `d` identify the upper half pole pair positions and the letters `r` through `u` identify the upper half plane zero pair positions.

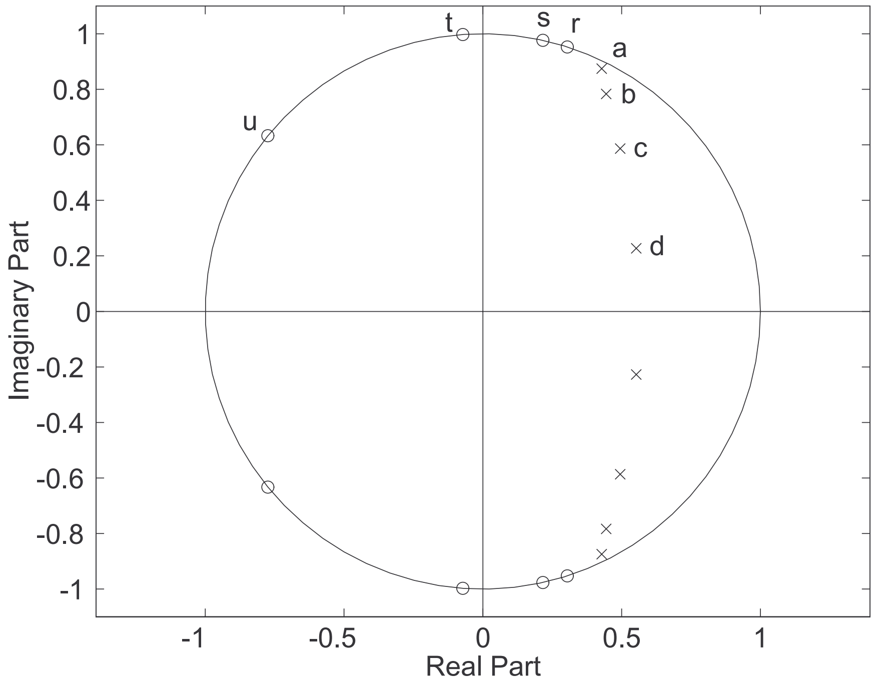

Figure 4: Elliptic filter pole-zero locations in the z-plane.

To implement our cascade of biquad sections we followed some commonly accepted guidelines regarding matching poles and zeros and ordering the resulting sections.

(a) Which zero pairs should be paired with which pole pairs to form biquad sections?

(b) When cascading biquad sections how should the sections be ordered going from the filter input to output? Identify the sections using the letters identifying the poles.

18. (20 pts) During the semester we gained experience in calculating base 2 logarithms of fractional values. These were used in generating logarithmic (dB) displays.

(a) What is the smallest positive non-zero value that can be represented using a 24-bit fractional value (Q0.23 or Q23 format)?

(b) What is the base 2 logarithm of this value?

(c) What is the largest positive value that can be represented using a 24-bit fractional value (Q0.23)?

(d) Approximately, what is the base 2 logarithm of this value?

(e) If we were to use a 24-bit value to represent the range of log values obtained above what would be the values of n and m in the $Qm.n$ format that would preserve as much accuracy as possible in the fractional part of the representation?

19. (12 pts) Emphasis was placed on *prior* planning when making phase measurements to be used in estimating group delay. The TI programs used early in the semester used DMA support to move data in blocks of 256 samples. It is estimated that data flowing through the TI processing encounters four block delays plus the delay from a symmetric 176 coefficient FIR filter. The FIR filter is a lowpass filter with a cutoff frequency of 3.3 kHz. The sample rate is 16000 samples per second. In order to minimize the measurement time you choose to configure the SigLab analyzer to produce a 100 value data set. For this exercise we will ignore any delays in the A/D converter.

(a) What would be the pre-measurement estimated group delay in seconds?

(b) What is the equation that you would use to relate the transfer function phase with group delay? Specify your units.

(c) How many cycles would the phase be expected to change if the measurement is made over the range from 0 Hz to 3200 Hz?

(d) Over what frequency range starting at 800 Hz would the phase be expected to change by exactly 12 cycles?

20. (10 pts) A surprisingly useful FIR filter is one that has all coefficient values equal to one. The equation describing this filter in the time domain is

$$y[n] = x[n] + x[n-1] + \cdots + x[n-(M-1)].$$

This filter forms the basis for the cascaded integrator comb filter.

The transfer function for this FIR filter can be written in closed form as the ratio of two simple polynomials in z. Please do so.

21. (10 pts) The cascaded integrator comb (CIC) filter implements the unit coefficient FIR filter as a cascade of two separate filters by implementing the transfer function's numerator and the denominator separately. Sketch the block diagram for a $M = 3$ CIC filter clearly indicating which part of the diagram corresponds to the integrator and which corresponds to the comb.

22. (10 pts) Two data structure types that we made of use this semester are the first-in-first-out (FIFO) and the last-in-first-out (LIFO).

    (a) a stack is a ______

    (b) a circular buffer is a ______

23. (10 pts) Analog to digital converters quantize their continuous time inputs into a sequence of discrete values represented using $B + 1$ bit words. Given an A/D converter that relates it's most negative integer value to an analog input of $-V_m$ Volts and its zero integer output value with zero volts, what is the voltage change, Δ, that will produce a count change of precisely one in the A/D output values?

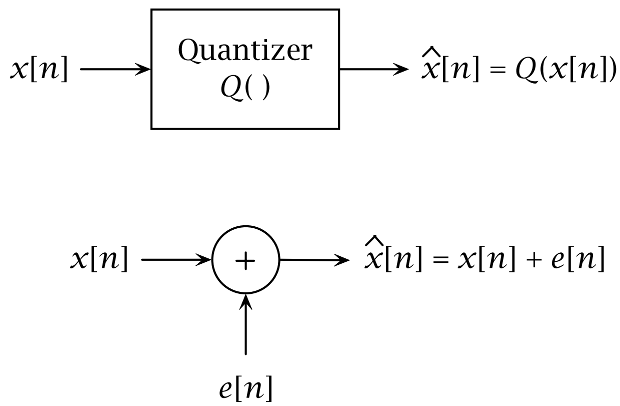

Figure 5: Quantizer modelled as adding a noise source to the original exact waveform.

24. (12 pts) Experience has shown that in most cases quantized values can be *modelled* using the original unquantized value to which an amount of noise is added. This model is shown in Figure 5.

    In lecture we made three assumptions about the properties of the unquantized waveform and the noise when modelling the quantization noise process. These were:

    (a)

    (b)

    (c)

25. (10 pts) Using the assumptions asked for in the above question various theoretical analyses pretty much give the same estimate of the noise variance, σ_n^2. What is the expression for σ_n^2 in terms of Δ as defined above?

    $\sigma_n^2 =$

26. (10 pts)

    (a) When oversampling we sample a waveform at a rate of F_a using a $B+1$ bit A/D converter. If we had sampled this waveform at rate F_b an integer multiple of F_a, digitally filtered the samples to a bandwidth of $F_a/2$ and then decimated the result back down to rate F_a we would have reduced the variance of the noise on the samples. This, in effect, increases the effective number of bits in the A/D converter.

What should the integer multiple be in order to make an improvement of 1 bit?

(b) When doing a similar processing using a first order sigma-delta A/D converter we found that the improvement in signal-to-noise ratio corresponds, in effect, to an increase of how many bits if we had used a "traditional" A/D converter for each doubling of the sigma-delta input sample rate:

27. (6 pts) In lecture and in the real-time FFT lab exercise we looked into the concept of leakage and how to mitigate it's effects. We found that using "window" functions reduced leakage but at the expense of decreased frequency resolution. In particular we used three window function. These were the boxcar, the Hamming, and the Chebyshev windows.

(a) Which had the most leakage?

(b) Which had the greatest effect in reducing the leakage?

28. (10 pts) The C language is highly dependent upon the concept of a stack. Because we were working both in C and assembler on the C5402 we needed to be aware about how the stack was implemented and used.

In lecture we identified five uses of the stack. One of these was the use of the stack by an interrupt support routine to hold the address to control to once the interrupt is serviced. List the remaining four uses and using Figure 6 indicate what portion of the stack they would use.

(a)

(b)

(c)

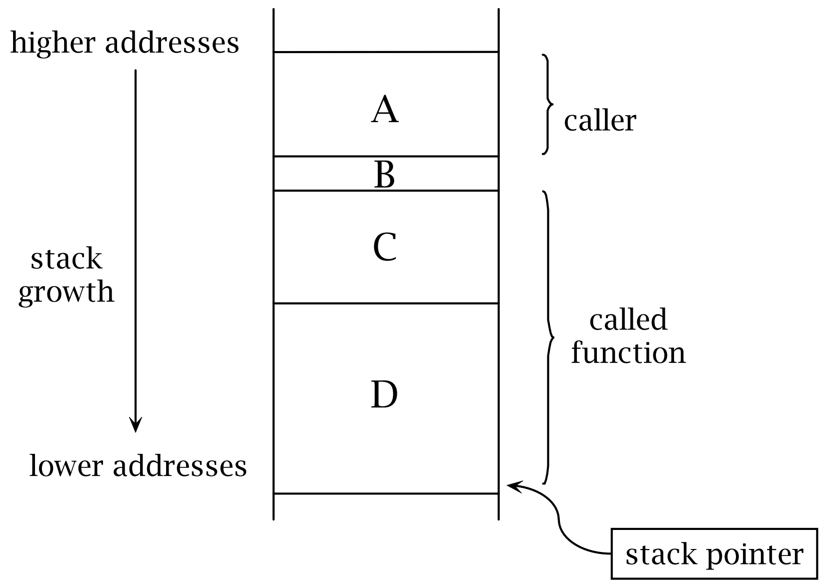

Figure 6: Stack usage in the TI C5402 C environment.

(d)

29. (10 pts) Draw and label the block diagram of the IIR direct form 2 biquad section.

30. (10 pts) Figure 7 shows the block diagram of a delta-sigma A/D converter that uses a first order loop. What are the expressions that describe the transfer functions $Y(z)/X(z)$ and $Y(z)/E(z)$?

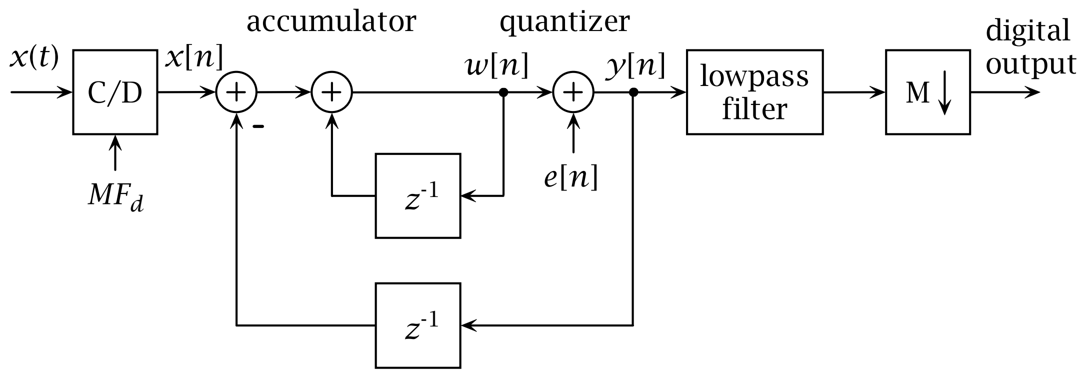

Figure 7: Delta sigma A/D converter using a first order loop.

31. (10 pts) Which registers fulfill the role of pointers

    (a) in the C5402?

    (b) in the DSP56303?

32. (10 pts)

    For each C5402 instruction listed below tell what the instruction does,

    - `mac`
    - `mas`
    - `mvdk`

- `mvmd`
- `mvmm`

33. (10 pts) How many words of internal memory is present in the DSP56303 and how is it organized?

34. (10 pts) How many words of internal memory is present in the TI C5402 and how is it organized?