Midterm — March 27, 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:

$$K_{\text{lowpass}} = \frac{1 + a_1 + a_2}{b_0 + b_1 + b_2}.$$

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 normalization factor would be:

$K_{\text{highpass}} =$

4. (10 pts) For the filter section in Figure 1 write the equation for the transfer function $W_2(z)/X(z)$.

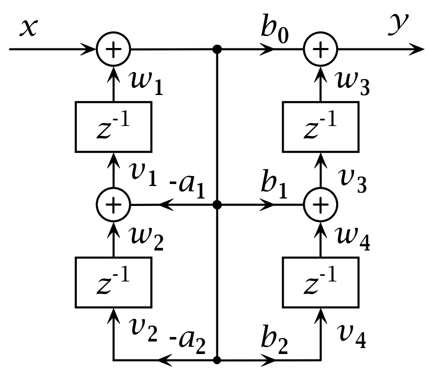

Figure 1: Filter section.

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 1.

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 2. DDS devices are increasingly finding application if radio appli-

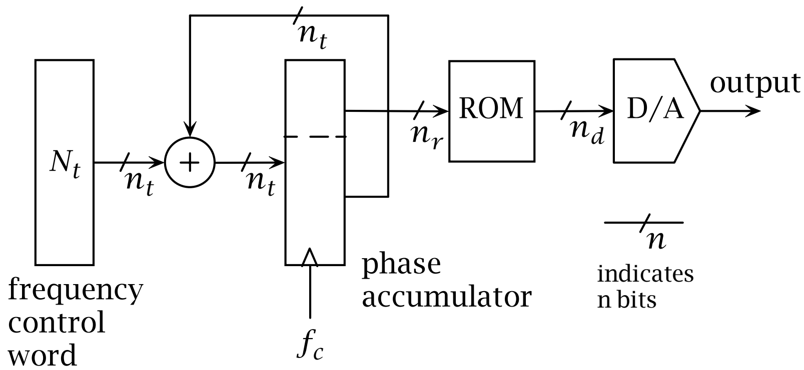

Figure 2: Basic direct digital synthesizer block diagram

cations. 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. The device uses a frequency control word and phase accumulator having 32 bits.

   (a) What decimal frequency control word value would best generate an output at 760 kHz?

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

7. (10 pts) For the TI C5402 what is the purpose or function the following status bits:

   (a) `OVM`

   (b) `SXM`

(c) SST

(d) FRCT

8. (10 pts) The MATLAB `kron` function is extremely useful in simulations involving upsampling. What is the result of

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

9. (10 pts) Given a data set of 10 samples taken at a rate of F_s Hz. I append 3 zero values to each sample obtaining a set of 40 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?

10. (10 pts) Given a data set of 10 samples taken at a rate of F_s Hz. I append 30 zero values to the end of the set obtaining a set of 40 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?

11. (2 pts each) Which processor (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) has a one instruction full divide instruction? ______

(h) has a protected execution pipeline? ______

(i) has a multiply and accumulate instruction? ______

(j) has an instruction specifically to evaluate polynomials? ______

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

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

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

(n) supports memory to memory data transfers? ______

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

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

(q) has a fixed dedicated stack pointer register? ______

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

12. (10 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 near the end of the interval being approximated over?

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

(d) for the cases we examined, minimizes the center approximation errors at the expense of the errors at the end points.

13. (10 pts) How would we group the multiply and additions in order to evaluate the following polynomial using Horner's method?

$$f(x) = a_0 + a_1 x + a_2 x^2 + a_3 x^3 + a_5 x^5$$

14. (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`

15. (10 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) did we use that has equiripple in both the passband and the stopband?

16. (10 pts) Figure 3 shows the pole and zero locations for the Elliptic filter design we used for the IIR lab exercise. The letters `a` through `d`

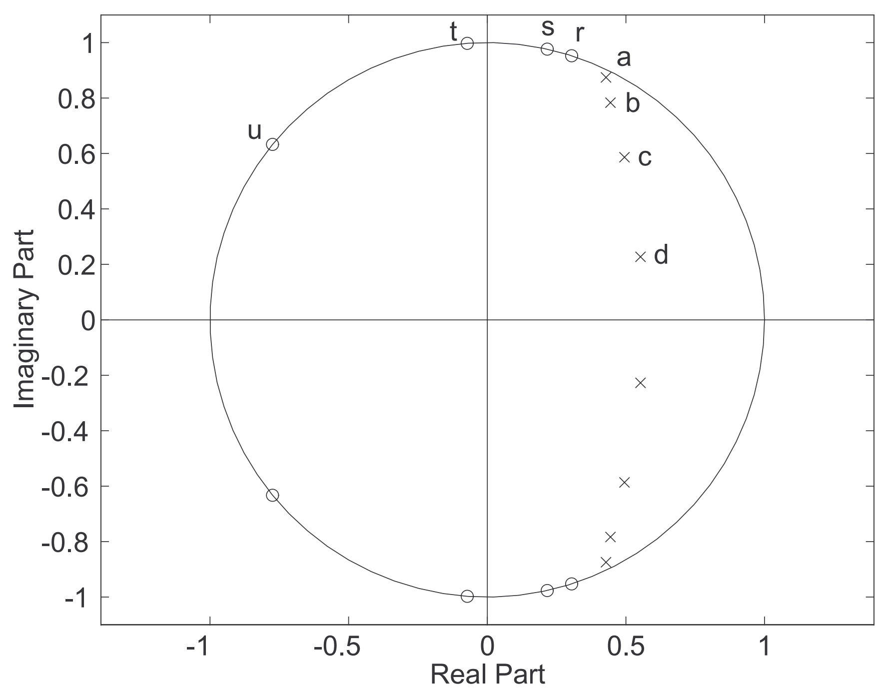

Figure 3: Elliptic filter pole-zero locations in the z-plane.

identify the upper half pole pair positions and the letters r through u identify the upper half plane zero pair positions.

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.

17. (10 pts) In the real-time DSP56303 FFT lab exercise we formed base-2 logarithms of fractional values limited to be less than one and greater than zero. The base-2 log values required that we place the binary point in the result so that integer values would be accommodated. Using Q notation, what is the largest value of n in Qn that will allow representing the log base-2 values?

18. (10 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 three block delays plus the delay from a 200 coefficient FIR filter. The FIR filter is a lowpass filter with a cutoff frequency of 3 kHz. The sample rate is 16000 samples per second. In order to minimize the measurement time you choose to configure the Tektronix analyzer to produce a 100 value data set.

(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) Over what frequency range would the phase go through exactly four cycles?

19. (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

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

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

20. (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 ______

21. (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, what is the largest voltage change, Δ, that will produce a count change of precisely one in the A/D output values?

22. (10 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 4. There are a number of assumptions made about the relationship between the unquantized waveform and the noise modelling the quantization noise process. Using these assumptions 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?

$\sigma_n^2 =$

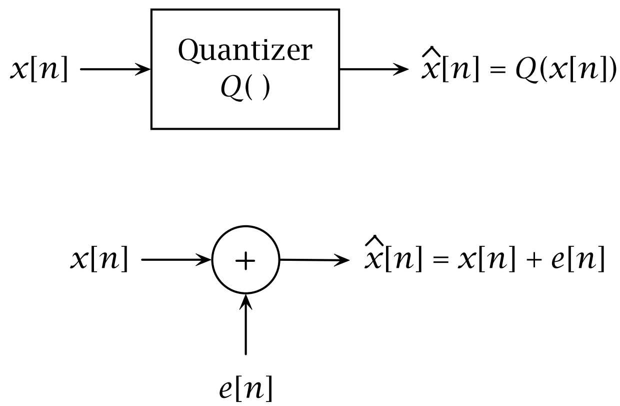

Figure 4: Quantizer modelled as adding a noise source to the original exact waveform.

23. (10 pts) Assume 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 this multiple be in order to make an improvement of 1 bit?

24. (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 5 indicate what portion of the stack they would use.

    (a)

    (b)

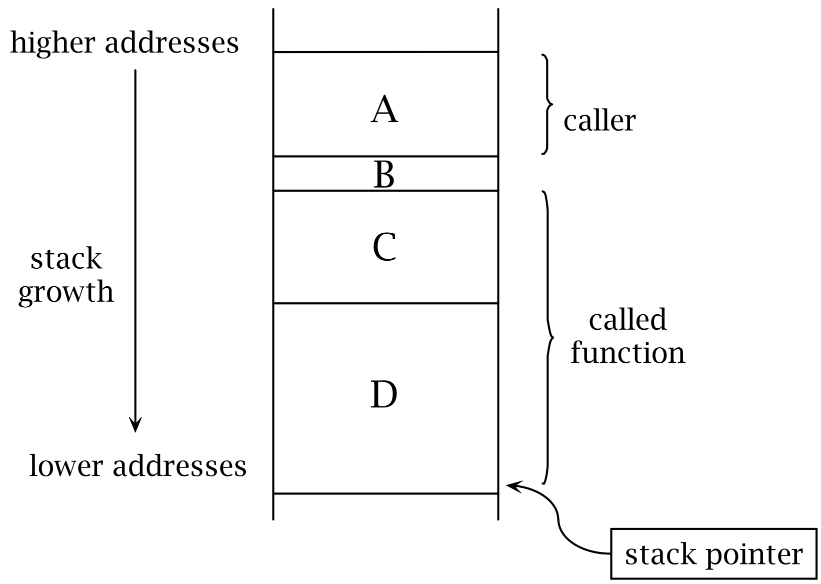

Figure 5: Stack usage in the TI C5402 C environment.

(c)

(d)

25. (10 pts) Draw and label the block diagram of the IIR transposed direct form 2 biquad section.

26. (10 pts) Figure 6 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 $Y(z)/E(z)$?

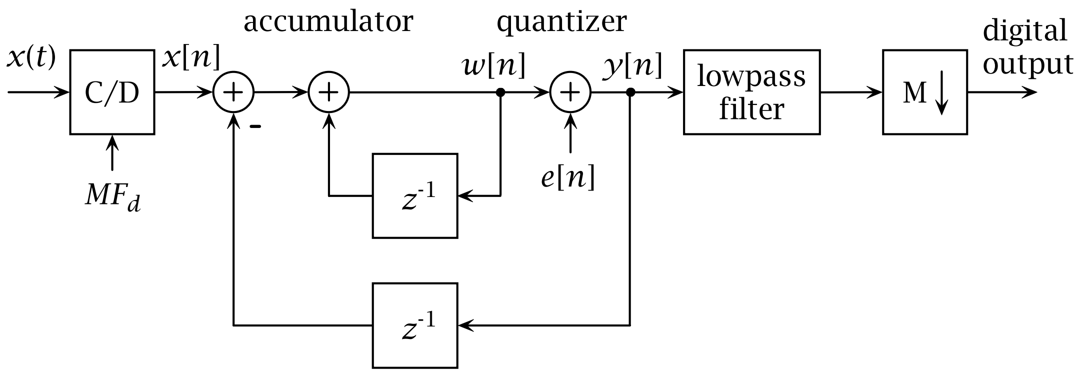

Figure 6: Delta sigma A/D converter using a first order loop.

27. (10 pts) Which registers fulfill the role of pointers

    (a) in the DSP56303?

    (b) in the C5402?

28. (20 pts) Using C and, as needed, intrinsic functions write the inner loop for FIR filter function listed below. Some of the TI supplied intrinsic functions are described in Figure 7.

Compiler Intrinsic	Assembly Instruction	Description
short _rnd(long src);	RND or ADD	Rounds src by adding 2^{15}. Produces a 16-bit saturated result. (OVM set)
short _sadd(short src1, short src2);	ADD	Adds two 16-bit integers, producing a saturated 16-bit result. (OVM set)
long _lsadd(long src1, long src2);	ADD	Adds two 32-bit integers, producing a saturated 32-bit result. (OVM set)
long _smac(long src, short op1, short op2);	MAC	Multiplies op1 and op2, shifts the result left by 1, and adds it to src. Produces a saturated 32-bit result. (OVM and FRCT set)
short _smacr(long src, short op1, short op2);	MACAR	Multiplies op1 and op2, shifts the result left by 1, adds the result to src, and then rounds the result by adding 2^{15}. (OVM and FRCT set)
long _smas(long src, short op1, short op2);	MAS	Multiplies op1 and op2, shifts the result left by 1, and subtracts it from src. Produces a 32-bit result. (OVM and FRCT set)
short _smasr(long src, short op1, short op2);	MASAR	Multiplies op1 and op2, shifts the result left by 1, subtracts the result from src, and then rounds the result by adding 2^{15}. (OVM and FRCT set)
short _smpy(short src1, short src2);	MPYA	Multiplies src1 and src2, and shifts the result left by 1. Produces a saturated 16-bit result. (OVM and FRCT set)
long _lsmpy(short src1, short src2);	MPY	Multiplies src1 and src2, and shifts the result left by 1. Produces a saturated 32-bit result. (OVM and FRCT set)
short _smpyr(short src1, short src2);	MPYR	Multiplies src1 and src2, shifts the result left by 1, and rounds by adding 2^{15} to the result. (OVM and FRCT set)

Figure 7: A few of the TI C compiler intrinsic functions. From the TI C manual.

```
/* firc.c */

extern int delaybuff[]; // delay buffer

void firc(int* input,   // points to input buffer
          int*  coeffs,  // points to FIR coefficients
          int*  output,  // points to output buffer
          int** delays,  // points to delay buffer pointer
          int   nh,      // number of FIR coefficients
          int   nx)      // number of values to process
{
    int  ctr_h;
    int* ptr_delay;
    int* ptr_coeff;
    int* delay_end;
    long sum;

    ptr_delay = *delays;
    delay_end = delaybuff+nh-1;

    while (nx--) {

      YOUR CODE GOES HERE

    }
    *delays = ptr_delay;
}
```

29. (10 pts) How much internal memory is present in the DSP56303 and how is it organized?

30. (10 pts) How much internal memory is present in the TI C5402 and how is it organized?

31. Sigma-delta D/A question.
32. addressing modes in DSP56303
33. addressing modes in C5402.