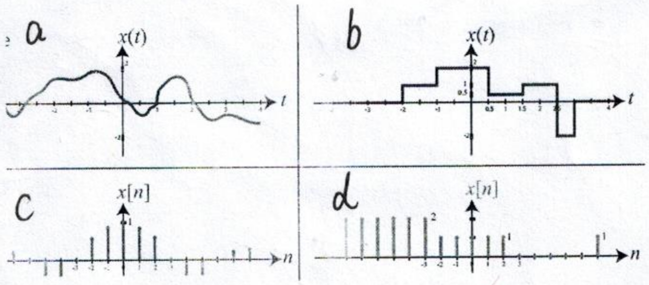
Quiz 1

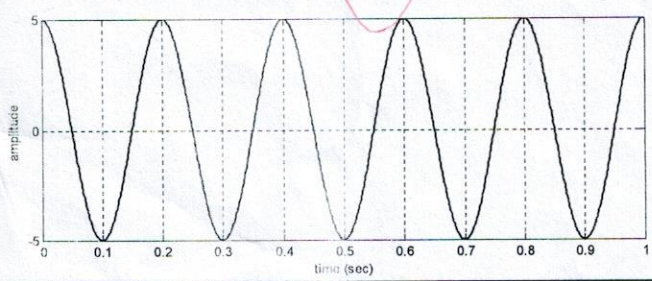
姓名: 姜春香

1. Which one is analog signal? <u>a</u>



2. Find the parameters for the following sinusoid:

A=<u>5</u> f=<u>5Hz</u> T=<u>0.2S</u> φ=<u>$\frac{\pi}{2}$</u>



Review Card 3

N位 $\frac{V_{ref}}{2^N} = \frac{V_{input}}{X}$

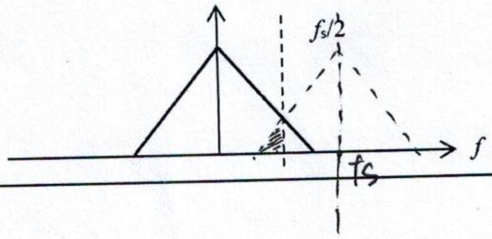
Name: 姜春香

1. If an ADC has a resolution of 12 bit and a reference voltage of 5 volts, the digital output for an analog input of 1.56 volts will be =<u>1278</u> (in decimal form)

$\frac{5}{2^{12}} = \frac{1.56}{x}$ $\frac{V_{ref}}{2^N}$

2. If we want to digitize an analog signal with frequency of interest not higher than 8 kHz, the minimal sampling frequency should be <u>16KHz</u>. The anti-aliasing filter prior to digitizing is a <u>analog</u>(analog or digital) <u>low pass</u>(low- or high- pass) filter with a cut-off frequency of <u>8 KHz</u>

3. Draw the side effect if an analog signal is sampled as follows:



Review Card 2

Name: 姜春香

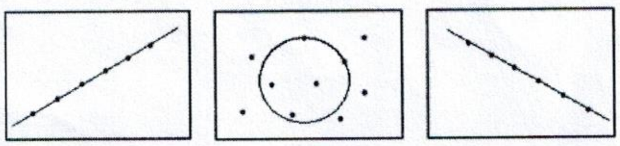
1. For a random signal {2 3 1 5 3 4 3 1}, calculate the following descriptive statistics:

Mean=<u>2.75</u> Median=<u>3</u>, Mode=<u>3</u>

Standard Deviation=<u>~~1.509~~ 1.388</u>

$\sqrt{\frac{\sum(x-\bar{x})^2}{8-1}}$

2. Determine the following correlation coefficients (*r*):



r=<u>1</u> *r*=<u>0</u> *r*=<u>-1</u>

Review Card 4

Name: 姜春香

Read the datasheet of a DAC chip from Texas Instruments below and answer the following questions:

DESCRIPTION

The DAC161S997 is a very low power 16-bit ΣΔ digital-to-analog converter (DAC) for transmitting an analog output current over an industry standard 4-20mA current loop. The DAC161S997 has a simple 4-wire SPI for data transfer and configuration of the DAC functions. To reduce power and component count in compact loop-powered applications, the DAC161S997 contains an internal ultra-low power voltage reference and an internal oscillator. The low power consumption of the DAC161S997 results in additional current being available for the remaining portion of the system. The loop drive of the DAC161S997 interfaces to a Highway Addressable Remote Transducer (HART) modulator, allowing injection of FSK modulated digital data into the 4-20mA current loop. This combination of specifications and features makes the DAC161S997 ideal for 2- and 4-wire industrial transmitters. The DAC161S997 is available in a 16-pin 4 mm × 4 mm WQFN package and is specified over the extended industrial temperature range of –40°C to +105°C.

1. The resolution of the DAC chip is: 16 bit

2. The way it transfers data between the chip and the computer is serial (parallel 并行 or serial 串行), because it has a 4-wire SPI

3. For a reference voltage of 3.3v, the voltage that 1 LSB represents is 1×10^{-5} V, the output analog voltage will be 0.006V for a digital input of 0x78.

Review Card 5

Name: 姜春香

1. Complete the number conversion between different bases:

$(1101.0111)_2$=(13.4375)$_{10}$

$(4F.6E)_{16}$ =(79.4296875)$_{10}$

$(87.35)_{10}$ =(~~11001101.01101011~~ 1010111.010110……)$_2$

$(CD.6A)_{16}$=(11001101.01101011)$_2$

$(99.025)_{10}$=(63.0666……)$_{16}$

$(11100110.01111001)_2$=(E6.79)$_{16}$

$(011100110.011110010)_2$=(346.362)$_8$

3 4 6 3 6 2

2. The range of a signed char (8-bit) is: −128 ~ +127

3. Represent the negative number $(-13)_{10}$ in different ways:

Sign-and-Magnitude: 10001101

1s Complement: 11110010

2s Complement: 11110011

Review Card 7

Name: 姜春香

1. A linear time-invariant system should have the following three properties: 齐次性 (homogeneity), 相加性 (additivity) and 时不变性 (time-invariant)

2. Decide whether the system described by the following formula is a linear time-invariant system:

$$y(n) = nx(n)$$

✧ Linear? ~~Abb~~ Yes

✧ Time-invariant? NO

Review Card 6

Name: 姜春香

1. For an 8-bit binary number 1001 1101,

(A) If it represents an **unsigned** number, the decimal value is 157;

(B) If it represents an **signed** number, the decimal value is:

a) −29 if it is in sign-and-magnitude form;

b) −98 if it is in 1s complement form;

c) −99 if it is in 2s complement form;

2. The three parts of a float number is: sign, mantissa and esponent. The advantage of float number is: 表示范围较大. In IEEE standard, it uses 32 bits to represent a number of **single precision**, and 64 bits for **double precision**.

3. Rewrite the following number in **base-2** normalized form and represent it by a 10 digit float number (4 digit for exponent part):

(A) $(13.25)_{10}$

a) normalized form: ~~$(1101.01)_2$~~ 0.110101×2^4

b) float-number representation:

0	1	1	0	1	0	0	1	0	0

(B) $(-0.1325)_{10}$

a) normalized form: -0.10000×2^{-2}

b) float-number representation (2s complement):

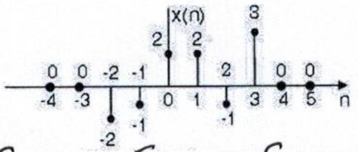
1	1	0	0	0	0	1	1	1	0

Review Card 8

Name: 姜春香

1. There are usually three ways to uniquely characterize a LTI system: ~~homogeneity~~ h(n), ~~additivity~~ H(f) and ~~time-invariant~~ H(z)

2. Please rewrite $x(n)$ by the form of impulse decomposition:



$x(n)= -2\delta(n+2)-\delta(n+1)+2\delta(n)+2\delta(n-1)-\delta(n-2)+3\delta(n-3)$

3. If the length of an input $x(n)$ is 8 points and the length of the system impulse response $h(n)$ is 7 points, then the convolution of $x(n)$ and $h(n)$ would be 14 points.

4. Please calculate the convolution of the following two discrete signals through both the input and output side algorithms:

$x(n)=\{1,3,4,2,5\}$

$h(n)=\{6,5,7\}$

(A) The input side algorithm:

$y_0(n)=\{6, 5, 7\}$

$y_1(n)=\{0, 18, 15, 21\}$

$y_2(n)=\{0, 0, 24, 20, 28\}$

$y_3(n)=\{0, 0, 0, 12, 10, 14\}$

$y_4(n)=\{0, 0, 0, 0, 30, 25, 35\}$

Convolution=$\{6, 23, 46, 53, 68, 39, 35\}$

(B) Draw the steps as $h(n)$ moved along $x(n)$, as well as the corresponding element of $y(n)$ for each step:

1 3 4 2 5
7 5 6 ⇒ $y[0]=1\times6=6$

1 3 4 2 5
7 5 6 ⇒ $y[1]=1\times5+3\times6=23$

1 3 4 2 5
7 5 6 ⇒ $y[2]=1\times7+3\times5+4\times6=46$

1 3 4 2 5
7 5 6 ⇒ $y[3]=3\times7+4\times5+2\times6=53$

1 3 4 2 5
7 5 6 ⇒ $y[4]=4\times7+2\times5+5\times6=68$

1 3 4 2 5
7 5 6 ⇒ $y[5]=2\times7+5\times5=39$

1 3 4 2 5
7 5 6 ⇒ $y[6]=5\times7=35$

Review Card 9

Name: 姜春香

1. For a continuous signal $f(t)$, the formula of its Fourier transform is: $F(f)=\int_{-\infty}^{+\infty} f(t)\cdot e^{-j\omega t}\,dt$; For a discrete signal $x(n)$, the formula of its Fourier transform is: $X[k]=\sum_{n=0} x(n)\cdot e^{-j2\pi kn/N}$

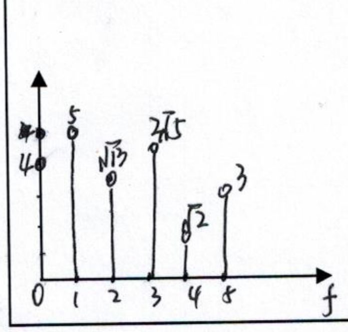
2. If the sampling frequency is 20Hz and a signal $x(n)$ has 10 points, then $x(n)$ can be decomposed into (6) sinusoid components. The frequencies of these sinusoids are: 0, 2, 4, 6, 8, 10.

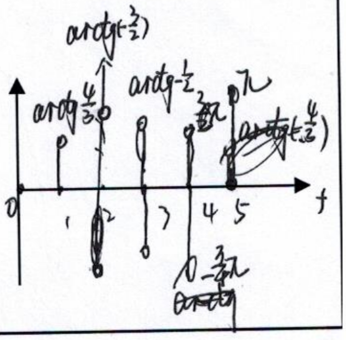
3. If the sampling frequency is 10Hz and a signal $x(n)$ has 10 points, then frequency resolution of the spectrum = 1 Hz.
Assuming that the first 6 points of $X(k)$ [the Fourier transform of $x(n)$] are: 4, 3+4i, -2+3i, 4-2i, -1-i, -3.

(A) How many points are missing for $X(k)$? 4
The missing points are: $-1+i$, $4+2i$, $-2-3i$, $3-4i$

(B) Draw the amplitude and phase spectrum:

$[-\pi, \pi]$





Review Card 10

Name: 姜春香

1. If the sampling frequency is 1000Hz and the length of x(n) is 500, then frequency resolution △f of the Fourier transform of x(n) is: 2Hz.

Is it possible to separate 80Hz and 82Hz? Yes

Is it possible to separate 80Hz and 81Hz? NO

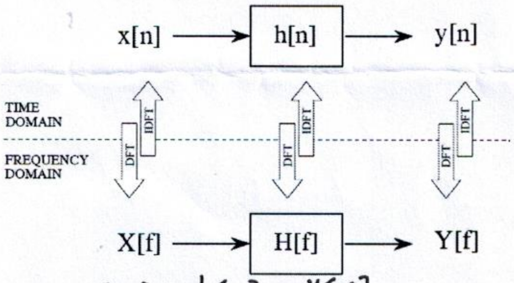
Is it possible to separate 79Hz and 81Hz? NO

Is it possible to separate 1000Hz and 1002Hz? NO

2. For frequencies of 10Hz, 30Hz, 40Hz and 50Hz, what is the fundamental frequency? 10Hz The third harmonic is 30Hz The second harmonic of the power line frequency in China is ~~二次谐波~~. 100Hz

3. If the length of h(n) is 500 and the sampling frequency is 1000Hz, what is the frequency resolution of H[f]? 2Hz

4. For a system below, list two methods to calculate y[n]



Method (1): x[n] * h[n] = y[n]

Method (2): $X[f] \times H[f] = Y[f] \xrightarrow{IDFT} y[n]$

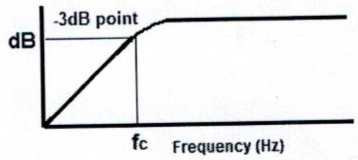
If both x[n] and h[n] have 100 points, y[n] will have 199 points and Y[f] will have 100 points. In this case, will the y[n] be the same for the two methods? NO

If no, what can you do to make the y[n] results the same? 时域补零. x[n] h[n] 补零至199个点.

Review Card 11

Name: 姜春香

If the amplitude spectrum of a system is as flows:

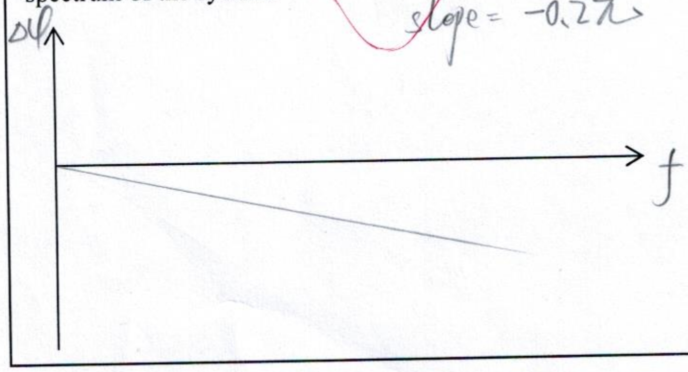


(1) Describe the system:

High pass filter. 截止频率 fc

(2) If the sampling frequency is 10Hz and the system always delays the input signal by one point, please draw the phase spectrum of the system:

slope = −0.2π



Review Card 12

Name: 姜春香

1. Which type of filter is better in performance, analog or digital? digital

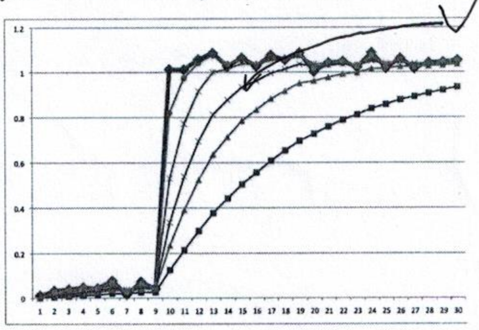
2. There are two ways to evaluate the performance of a digital filter: in time domain and in frequency domain.

✧ Time Domain

(A) What type of system response is used? 阶跃响应

(B) What parameters are used? risetime. overshoot. linear phase

(C) The responses of several filters are as follows, which filter do you think is best? (Mark the curve)

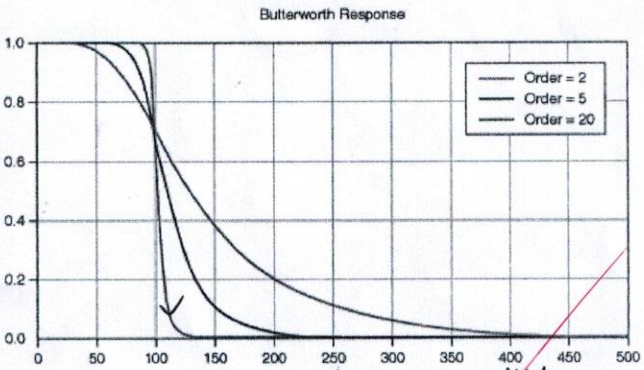


✧ Frequency Domain

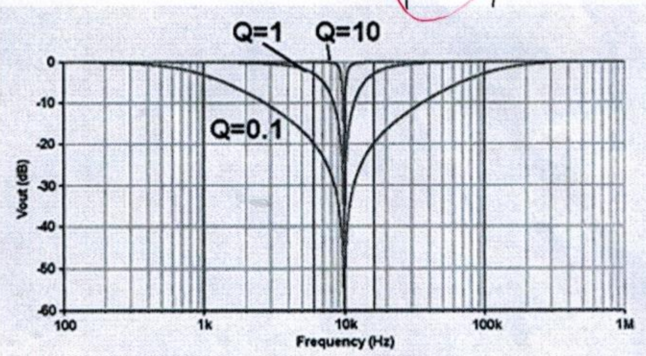
(A) What type of system response is used? frequency response

(B) What parameters are used? fast roll-off. No passband ripple. good. stopband attenuattion

(C) The responses of several filters are as follows, which filter do you think is best? (Mark the curve)



What type of filter is above? Low-pass filter



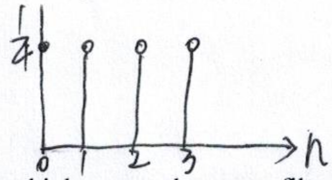
What type of filter is above? band-reject filter

Review Card 14

Name: 姜春香

The Moving Average Filter (MAF)

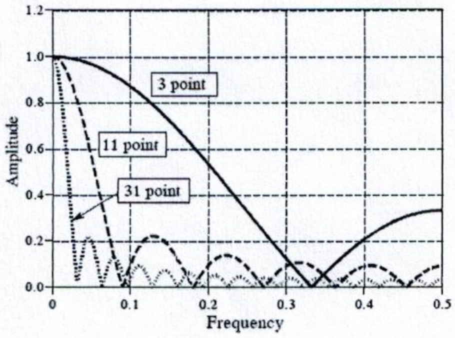
1. For a four-point MAF, draw the impulse response h(n):



2. Is it a high-pass or low-pass filter? Low-pass.

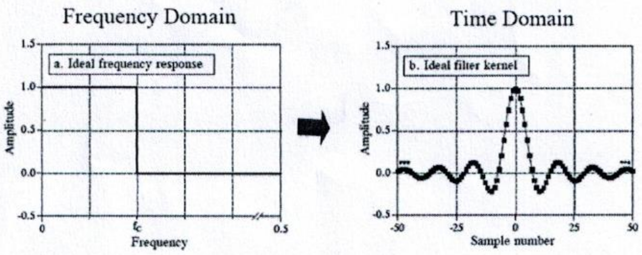
3. The aim we use the MAF is to 去除高频噪音.

4. See figure below of MAF's with different averaging points. The advantage of using more points (such as 31 points) is 去噪效果更好, and the disadvantage is: 带宽过窄,衰减部分有用信号.

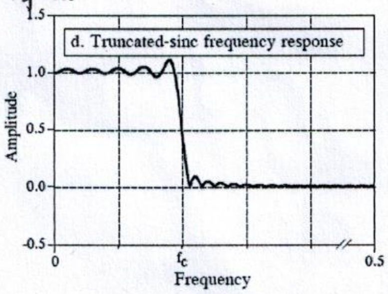


The Windowed-Sinc Filter (WSF)

1. How do we get the filter kernel of WSF from its frequency response (see below)? iFFT



2. The actual frequency response is different from its ideal response (see below), why? 截断导致信号跳变



3. We see that the response above is poor, how to solve the problem? 加窗

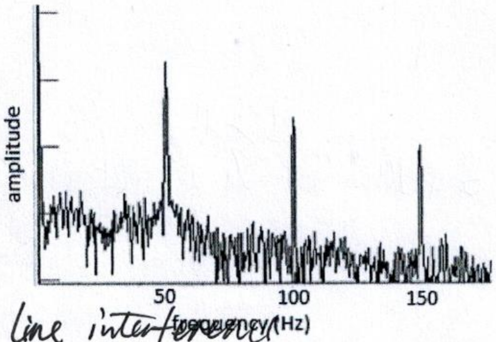
4. What's the two parameters when designing WSF? cut off frequency M (number of points used in moving average.)

fc.
M: 加窗长度

Review Card 15

Name: 姜春香

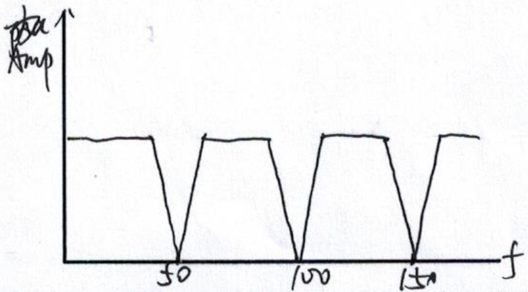
When measuring ECG signals, Xiaoming found that the recorded signal was not as what he expected. After performing a FFT, he found that there were large spikes in the spectrum, which were so large that it was impossible for him to observe the actual ECG signals. Please help him to find a solution to extract the ECG signals from such a noisy recording.



power line interference

1. Where did the interferences come from? 工频干扰

2. Please draw the frequency response of the filter that can remove the interferences.



3. What type of filter is the one you draw above? (Moving average, Windowed-Sinc, or Custom filter) Custom filter

4. Please describe the steps to get the $h(n)$ of this filter?

(1). Given a frequency response needed by you.

(2) Took the IDFT of the frequency response to get the impulse response.

(3) truncated the impulse response into M+1 samples and shifted into the right by M/2 samples.

(4) Multiplied the truncated impulse response by a Hamming or Blackman window to get the filter kernel.

(5) Took the DFT of the kernel to get the actual frequency response

Review Card 16

Name: 姜春香

1. Compare FIR and IIR filter:

	FIR	IIR
冲击响应（是/否）有限长度	是	否
负反馈（有/无）	无	有
输出 y(n)与（输入/输出）有关	输入.	输入. 输出.
需要计算的参数	h(n)	A. B.
计算量	大	小
线性相位（是/否）	是	否 (信号变形)

2. Assume that the input and output of a filter satisfies the following equation:

二阶 filter

$$2y(n)+4y(n-1)+6y(n-2)=7x(n)+8x(n-2)$$

(1). Please calculate the H(z) of the system:

$$H(z)=\frac{3.5+4z^{-2}}{1+2z^{-1}+3z^{-2}}$$

(2). The recursive coefficients of the filter is:

A= {3.5, 0, 4}

B= {1, 2, 3}

(3) What Matlab function can be used to plot the frequency response of a filter, given coefficients A and B? freqz

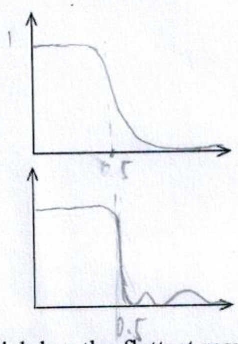
Review Card 17

IIR

Name: 姜春香

相位非线性 → 信号变形

Draw the typical frequency responses of the following low-pass IIR filters: Butterworth, Chebyshev Type I, Checbyshev Type II and Elliptic filters (assuming a cut-off of 0.5)



A. Which has the flattest response? Butterworth

B. Which has the fastest roll-off? Elliptic

C. Which has ripples in pass-band? Cheby I Elliptic

D. What are the Matlab functions for the four types of filters, respectively? butter, cheby1, cheby2, ellip

E. After getting the A and B coefficients by using one of the above functions, Xiaming used them to filter input signal x immediately, do you think it is OK? NO

F. If no, what step do you think Xiaoming missed?

用freqz看一下.