NAME ____ STUDENT # _____________

ELEC 484 Audio Signal Processing

Midterm Exam July 2008

CLOSED BOOK EXAM Time 1 hour

Listening test

Choose one of the digital audio effects for each sound example. Put only ONE mark in each column. 1 markseach, 6 marks total.

	Example 1	Example 2	Example 3	Example 4	Example 5	Example 6
Wah-wah						
flanger						
FIR comb (2tap)			1			
IIR comb	0.5					
reverberation	1					
Ring modulation				0.5		
Convolution						
Vibrato						
Tremolo						
Limiter						
Compressor						
Expandor		1				0.5
Noise gate		0.5				1
Variable speed replay						
Frequency shift				1		
Pitch shift						
Time stretch						
Distortion						
Fuzz						
Low pass filter					1	
High pass filter						
Bandpass filter						
Bandreject filter					0.5	

Marks are shown in brackets next to the question number

1. (4) Given a first order allpass filter. Write the transfer function *A(z)*. Sketch the pole and zero locations. How can we make a low pass filter *H(z)* using only *A(z)* and constant terms ? How can we make a shelving filter *H(z)* using only *A(z)* and constant terms ?

chapter 2 slide 5 and 25 and 30

$$A(z) = \frac{z^{-1} + c}{1 + cz^{-1}}$$

$$c = \frac{\tan(\pi f_c / f_s) - 1}{\tan(\pi f_c / f_s) + 1}.$$

•Lowpass

–H(z) = ½ [1 + A(z)]

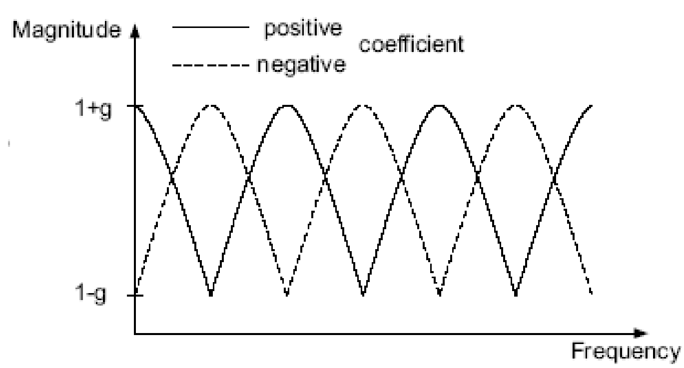
•Shelving

–H(z) = 1 + Ho/2 [1 + A(z)], where A(z) is first order (180 degree phase shift at high frequencies)

2. (2) Consider an FIR comb filter *y*[*n*] = *x*[*n*] + *gx*[*n-2*] operating on an audio file with sampling rate of 48 KHz. Assuming g > 0, sketch the frequency response and clearly label both axes with a correct scale.

Chapter 3 slide 3 and 4

$$\begin{aligned} y(n) &= x(n) + gx(n-M) \\ \text{with} \quad M &= \tau / f_s \\ H(z) &= 1 + gz^{-M}. \end{aligned}$$



Z^-2 = exp(-4jpif/fs) = +1 peaks at exp(j2npi) or f = nfs/2

3. (2) Describe how to implement fractional delays (i.e. delays by a non-integer number of samples). What do we use fractional delays for ?

chapter 3 section 3.2.4 slides 13 and 14

$$y(n) = x(n - [M + \mathrm{frac}])$$

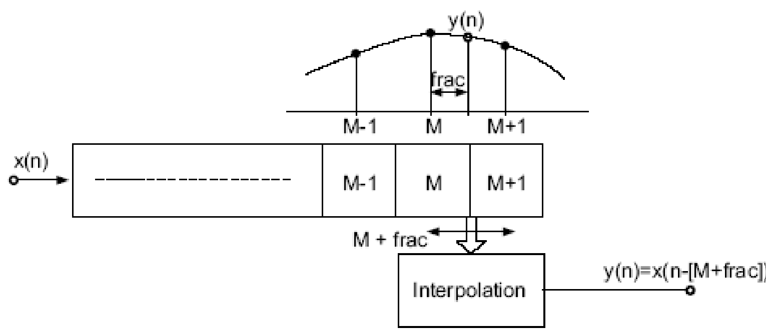


Figure 3.5 Fractional delay line with interpolation.

- linear interpolation [Dat97]

$$y(n) = x(n - [M + 1])\mathrm{frac} + x(n - M)(1 - \mathrm{frac})$$

- allpass interpolation [Dat97]

$$y(n) = x(n - [M + 1])\mathrm{frac} + x(n - M)(1 - \mathrm{frac}) - y(n - 1)(1 - \mathrm{frac})$$

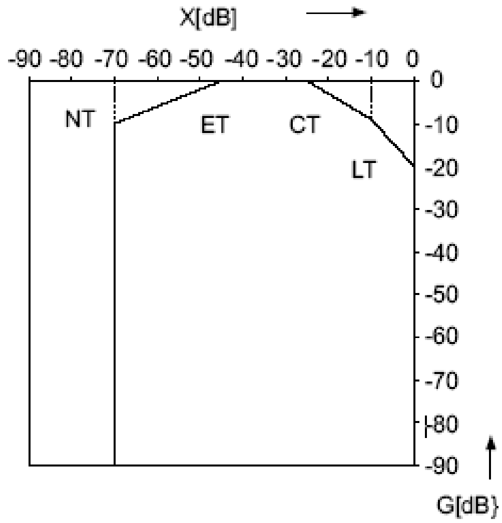
Use for wah-wah, vibrato, pitch shifting

4. (1) What is the difference between frequency shifting and pitch shifting?

Pitch shift preserves harmonic relationships, e.g. a signal with 440 and 880 Hz tones Pitch shifted will become 550 and 1100 Hz, whereas the same signal frequency shifted would become 550 and 990 Hz, and the octave 2/1 ratio is lost.

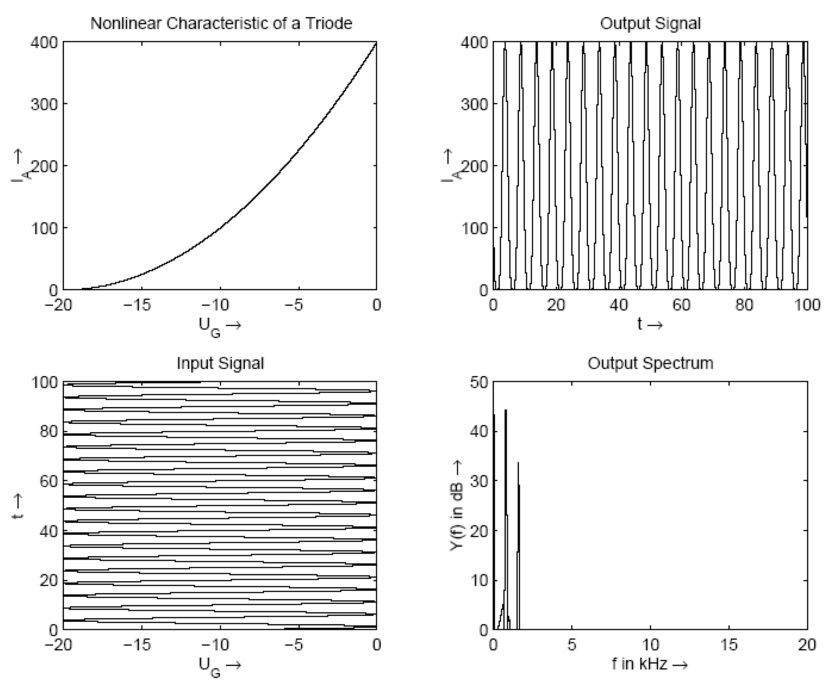
5. (4) Consider an audio signal consisting only of a 100 millisecond burst of a 1000 Hz sine wave with amplitude 1 volt p-p, followed by another 100 millisecond burst of the same sine wave with amplitude 2 volts p-p, the rest is silence (zero). This signal is used as input to an expander with a 1:3 slope. Sketch the waveform and both the static and dynamic gain as a function of time, assuming an attack time of about 10 msec and a decay time of about 50 msec. Put the 3 graphs underneath each other using a common time axis.

Signal increases 6 dB, with 3:1 expander will increase 18 dB.
Static gain will be 0 dB to start, then increase to 12 dB, but assuming max gain is 0 dB
We can start with –12 dB increasing to 0 dB.



6. (1) Sketch the characteristic of a non-linear device (output versus input) on a linear (not dB) scale that creates second and other order harmonics, but no odd-order harmonics.

Plot of x^2 vs x will do



7. (1) Sketch a graph of the perceived azimuth of a virtual sound source when a standard stereo layout (speakers at +/- 30 degrees from the listener) is driven with identical signals that only differ in level. Plot the perceived azimuth (left to centre to right) on the Y axis and the level difference in dB on the X axis. Put the correct numbers on the axes.

chapter 6 slide 10

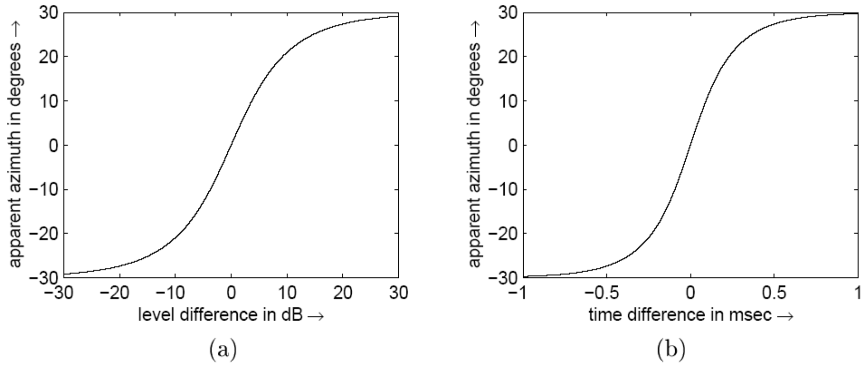


Figure 6.2 Perceived azimuth of a virtual sound source when a standard stereo layout is driven with signals that only differ in level (a) or in time delay (b).

8. (1) Which cue is used to decide if a sound source is directly above you ?

pinna

9. (4) Explain how to get a surround sound effect (sound from behind as well as in front and from the side) using only two speakers in front of the listener.

Chapter 6 HRTF to create surround with headphones, then use section 6.4.5 slide 47 interaural crosstalk cancellation for surround via 2 speakers

6.4.5 Transaural Stereo

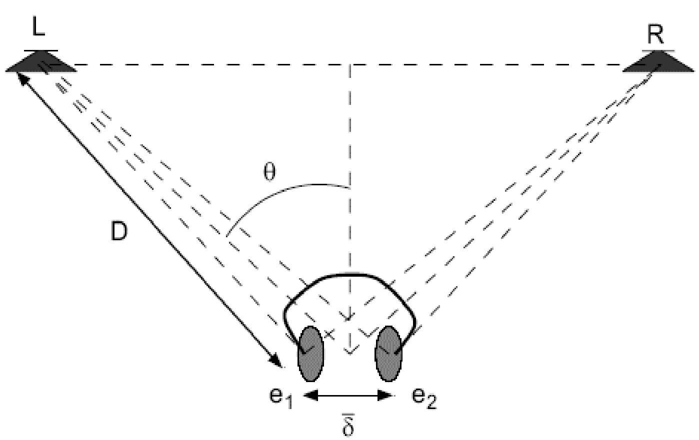
Signal at ear e1 e2

$$\begin{bmatrix} e_1(z) \\ e_2(z) \end{bmatrix} = \begin{bmatrix} 1 & gH(z)z^{-d} \\ gH(z)z^{-d} & 1 \end{bmatrix} \begin{bmatrix} L(z) \\ R(z) \end{bmatrix} \triangleq \mathbf{A}(z) \begin{bmatrix} L(z) \\ R(z) \end{bmatrix}$$

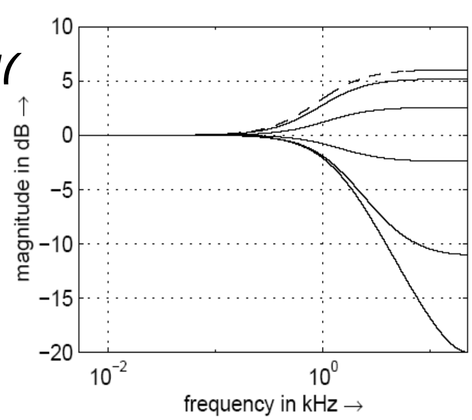
where d is the arrival time difference in samples of the signals at the ears.

$$g \cong \frac{D}{\bar{\delta}\sin\theta + D}$$

H(z) is head shadowing filter



H(



To play binaural signals e1,e2

solve for *L(z)*, *R(z)* given e1(z)

$$\mathbf{A}^{-1}(z) = \frac{1}{1 - g^2H^2(z)z^{-2d}} \begin{bmatrix} 1 & -gH(z)z^{-d} \\ -gH(z)z^{-d} & 1 \end{bmatrix}$$

10. (4) Explain one algorithm for pitch shifting with formant preservation, and another algorithm without formant preservation. Include appropriate and complete sketches, so that the method of Implementation Is clear.

without formant preservation chapter 7 section 7.4.2 slide 26

The order of pitch shifting and time scaling can be changed, as shown in Fig. 7.13. First, a time scaling algorithm expands the input signal from length N_1 to length N_2. Then a resampling operation with the inverse ratio N_1/N_2 performs pitch shifting and a reduction of length N_2 back to length N_1.

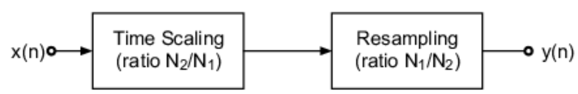


Figure 7.13 Pitch shifting by time scaling and resampling.

or

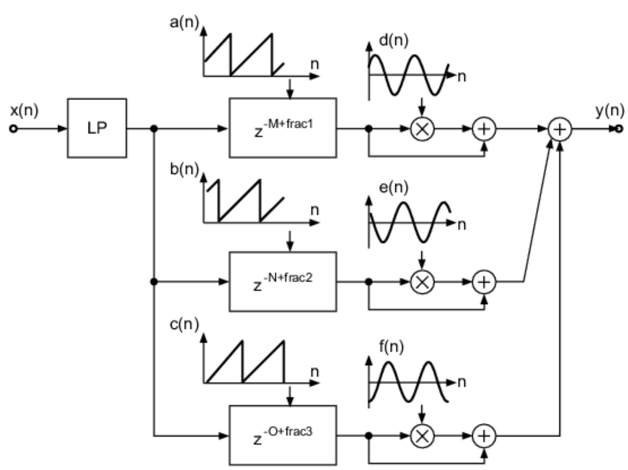


Figure 7.16 Pitch transposer: block diagram.

with formant preservation chapter 7 section 7.4.4 slides 34-41

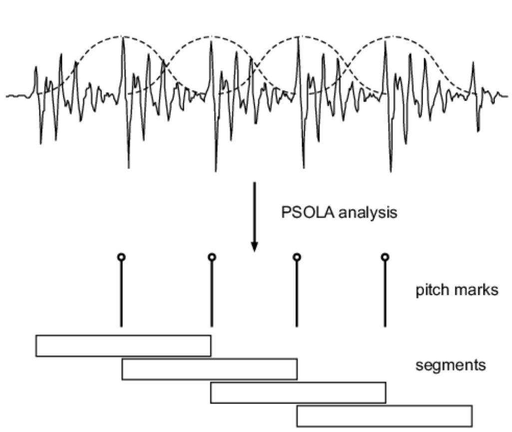
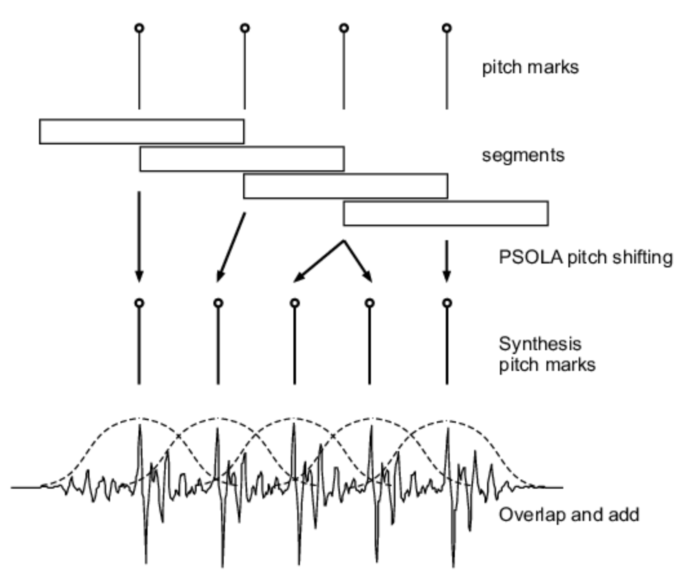
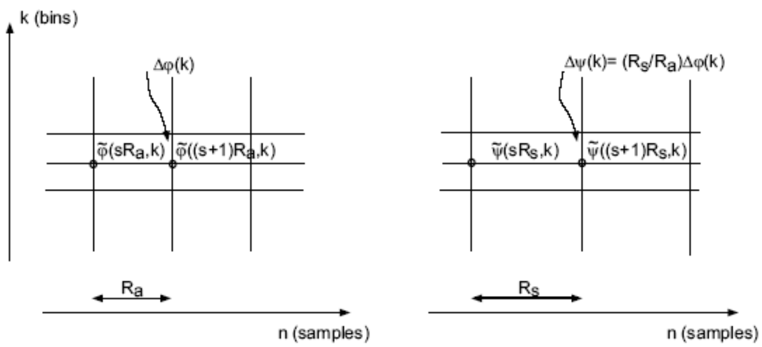


Figure 7.9 PSOLA pitch analysis.

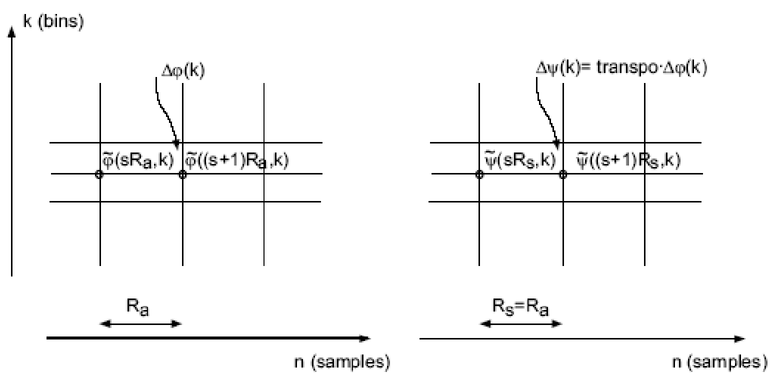


11. (4) Using diagrams but minimal mathematics, explain how the phase vocoder does pitch shifting and time stretching.

chapter 8 many slides
time stretch



Pitch shift



12. (4) Using diagrams, explain how a simple perceptual audio coder works. Make it clear what are the main ideas.

Hide quantization noise below masking threshold