INTRODUCTION TO COMPUTER MUSIC

ACOUSTICS AND PERCEPTION

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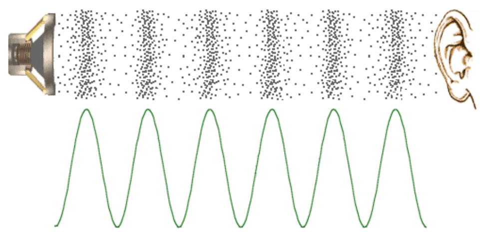
ACOUSTICS AND PERCEPTION

Acoustics and Perception

- Sound is vibration, air pressure fluctuations
  - About 0.001 lbs/in^2 for loud sound
  - Sea level: 14.7 psi
  - Cabin pressure: ~11.5 psi
  - Amplitude of deflection of eardrum = diameter of hydrogen atom for softest sounds
  - 1 psi ≅ 6895 Pascal (Pa)



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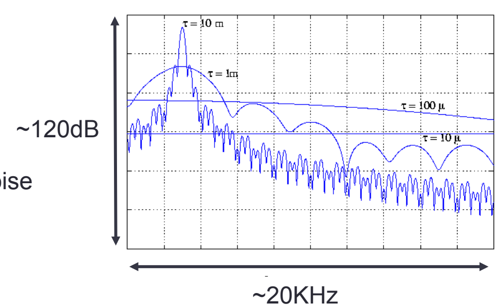
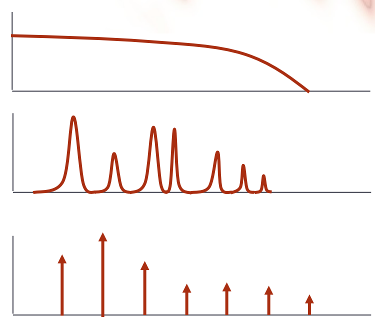
Acoustics and Perception

- What can we hear?
  - 20 to 20KHz frequency range (nominal)
  - ~120dB from threshold of hearing to threshold of pain
  - Practical range is often determined by background noise
- We are very sensitive to amplitude spectrum



Real World Sounds are Complex

- Noisy sounds ("shhhh") are
  - Broadband (all frequencies)
  - Random
- Percussion sounds
  - Thump, bell clang, ping, knock
  - exponentially decaying sinusoids
- Pitched sounds (tones)
  - Tend to have harmonically related sinusoids
  - Driven oscillation: harmonic and sustained
  - Non-driven (plucked, struck): less harmonic, exp. decay



PITCH PERCEPTION

What gives a sound "pitch"

More on “Real” Sounds

- Objects have modes of vibration
- Modes of vibration have characteristic frequencies
- In some cases, characteristics are approximately harmonic
- E.g. stretched strings
- Modes generally decay exponentially
- Non-linear coupling between modes can complicate things

More on “Real” Sounds

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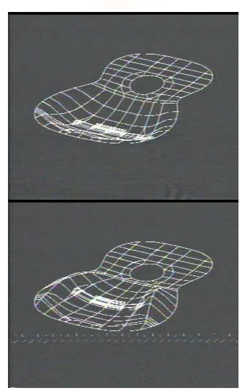


Dr. B. *Richardson, Cardiff University*

"Driven" Oscillators

- Something interesting happens when musical instruments are driven into oscillation
- The feedback mechanism that sustains oscillation results in periodic oscillation
- (Perfect) periodicity implies exactly harmonic partials
- Noise, jitter, non-uniform driving forces, etc., complicate the picture

A sarinda is a musical instrument from Afghanistan.

Pitch Perception

- Strongly related to frequency
- Sense of pitch enhanced by harmonics
- Pitch perception is approximately logarithmic
- Musical octave: 2 x frequency
- Divide octave into 12 log-spaced semitones (half-steps)
- Divide semitones into 100 log-spaced cents
- We are sensitive to about 5 cents

1 octave = double (fundamental) frequency

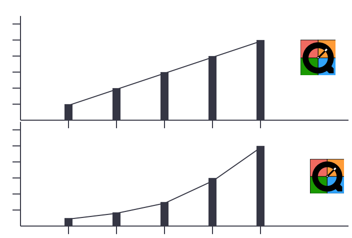
100 cents = 1 semitone= 1 half-step

10 cents

LOUDNESS AND LOCALIZATION

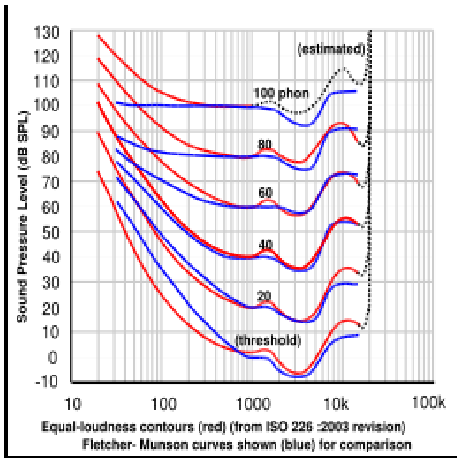
Perception Basics: Loudness

- Strongly related to amplitude
- Approximately logarithmic
- Double loudness: 10-fold increase in intensity
- Sensitive to about 1dB
- Fletcher-Munson curve: frequency dependent sensitivity



2dB change

Fletcher-Munson Curve



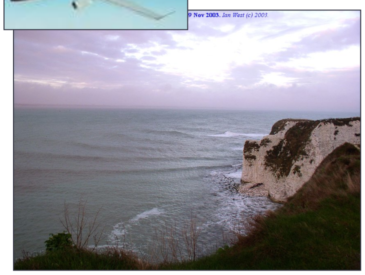
Perception: Localization

- Multiple cues
  - Relative amplitude
  - Relative phase
  - Spectral effects of pinnae (outer ears)
- Taken together, these effects are sometimes Head-Related Transfer Function
- Environmental cues are also important:
  - Reflection of sound
  - Reverberation and ratio of reverberation to direct sound
  - Knowledge of sound source (through vision, recognition, …)

More Acoustics

- Speed of sound: about 1 ft/ms
- Sound reflects from surfaces
  - >20ms: echo
  - Many fine reflections: reverberation
- Sound also refracts
  Wavelengths vary from 50 feet to fraction of an inch
- Linearity: (for pressure signals x and y):
  - F(ax) = aF(x)
  - F(x + y) = F(x) + F(y)

737: 575 mph (mach 0.82)
Sound: 701 mph
At sea level: 770 mph



Why does Linearity matter?

- Air, rooms, performance spaces are very linear
- What you hear is weighted sum of sound sources
- Filters weight or delay frequencies differently but independently
- Superposition principle

Summary

Perception	Phenomenon
Pitch	Frequency (20-20 KHz)
Loudness	Intensity (90 dB)
Timbre	Spectrum (and other)

- Perception is roughly logarithmic
- Struck objects typically exhibit characteristic frequencies with exponential decay rates
- Driven oscillators typically exhibit periodic signals, hence harmonic spectra
- Superposition principle applies to sounds

EFFECTS AND REVERBERATION

Audio Effects and Processing in Nyquist

- Delay
- Echo (feedback delay)
- Equalization
- Chorus
- Panning
- Compression/Limiting
- Sample Rate Conversion
- Sample Size Conversion
- Reverberation

Delay

- Creates echo effect
- Very short delays create spectral effects and/or chorus effect
- Implemented by delay buffer or simply by adding sounds using an offset:
  - **`cue`**(***sound***) **`@`** ***delay***
  - Note: **`cue`** coerces a sound into a behavior that can be shifted using the **`at`** transform.

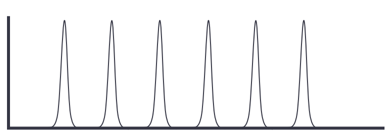
Echo

- What is it?
  - Copies of sound delayed and attenuated
- Algorithm:

- Code:
  - feedback-delay(*sound, delay, feedback*)
- Notes:
  - Append silence to source to avoid sudden ending:
    - feedback-delay(seq(*sound,* s-rest(10)), *delay, feedback)*
  - Delay is fixed (see delay, variable), but feedback can be sound

Comb Filter

- Essentially a very short feedback delay
- Frequencies that are multiples of the delay are reinforced, so you get boost at harmonics:



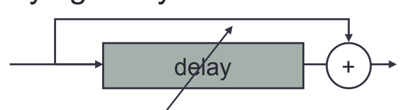
- See Nyquist comb function

Equalization

- What is it?
  - Filter used to adjust spectral balance
- Code:
  - load "grapheq"
  - nband(*input*, *gains*)
- Notes:
  - *Gains* is an array, evenly divided across 20-20kHz range
  - See filters in manual -- there are many
  - See eq editor in Nyquist IDE

Chorus

- What is it?
  - Very short, time varying delay
- Algorithm



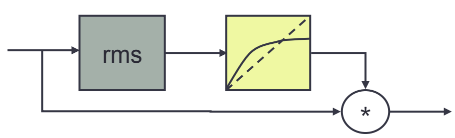
- Code
  - load "time-delay-fns"
  - chorus(*snd, maxdepth, depth, rate, saturation*)
- Notes:
  - chorus(*snd,* 0.05, 0.025, 0.5, 0.5)

Panning

- What is it?
  - Split mono signal to stereo outputs
- Code:
  - pan(*snd, where*)
- Notes:
  - *where* is 0 (left) to 1 (right)
  - simple linear panning

Compression/Limiting

- What is it?
  - Automatic gain control (not data compression)
- Algorithm



- Code
  - load "compress"
  - compress(*input, map, rise-time, fall-time, lookahead*)
  - agc(*input, range, rise-time, fall-time, lookahead*)
- Notes
  - See code and manual for *map*

Reverse

- What is it?
  - Play a sound backwards
- Algorithm:
  - read blocks of samples from file
  - Convert to array and reverse it
  - Convert back to sound and make a big sequence
- Code
  - See reverse.lsp

Sample Rate Conversion

- What is it?
  - High-quality interpolation to alternate sample rate
- Algorithm: digital low pass and reconstruction
- Code:
  - resample(*snd, rate*) ; fixed rate
  - sound-warp(*warp-fn, signal, wrate*) ; variable map
- Notes:
  - sound-warp applies *warp-fn* to *signal*
  - Nyquist must compute inverse of *warp-fn* at *wrate*

Sample Size Conversion

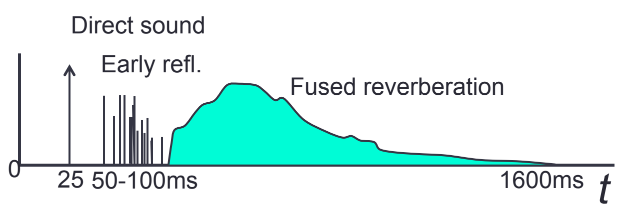
- What is it?
  - Simulate different sample formats by rounding
- Algorithm: conversion to integer and back
- Code:
  - quantize(*sound, steps*)
- Notes:
  - quantize(*sound, 255*) is 8-bit PCM

Reverberation

- Reflections in a concert hall are many
- Number increases exponentially with time
- Typically modeled in two parts:
  - Early reflections – discrete delays
  - Late reflections – generated by a network of all-pass and feedback delay filters
  - Often add low-pass filter to network because high frequencies are absorbed by air and room surfaces

Artificial Reverberation

- RT60 – time to decay from peak amplitude to -60dB (1/1000 amplitude)
- Typical RT60 is 1.5 to 3 seconds
- Impulse response:



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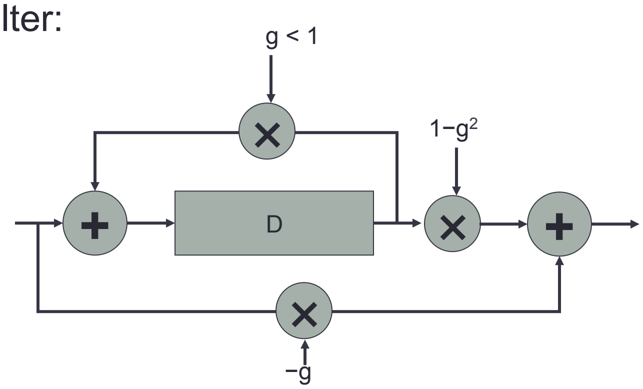
Reverberator Building Blocks

- Feedback delay (comb filter) provides decaying echo
- Allpass filter:



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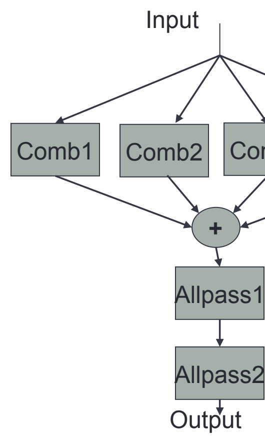
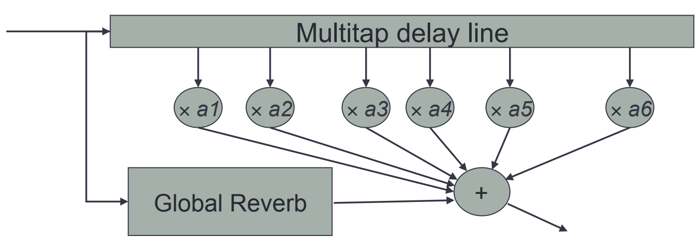
Implementations

- Schroeder



Improved Reverberators

- Add multitap delay for early reflections



- Low-pass in feedback loops
- More elaborate allpass filters
- Multichannel designs

Implementations in Nyquist

```
load “reverb.lsp”
reverb(sound, reverb-time)
```

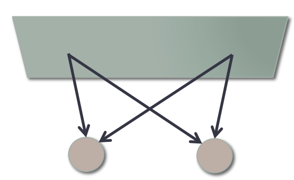
- Mix “dry” sound with reverb:

```
function reverb-mix(s, rt, wet)
  return s * (1 – wet) +
                reverb(s, rt) * wet
```

- Other reverb primitives in Nyquist:
  - nrev
  - jcrev
  - prcrev

Convolution-based Reverberators

- Reverberators can be seen as very big filters
- Long irregular impulse response
- Many modern reverbs
  - Measure impulse response of real rooms and concert halls
  - Four impulse responses for stereo
  - Apply impulse responses using fast convolution (multiplication in the frequency domain)



Summary

- Many audio effects are available
- Audio effects are crucial in modern music production
- Reverberation
  - The effect of millions of "echoes"
    - becoming denser with greater delay, and
    - generally decaying exponentially with delay