Creative Computing II

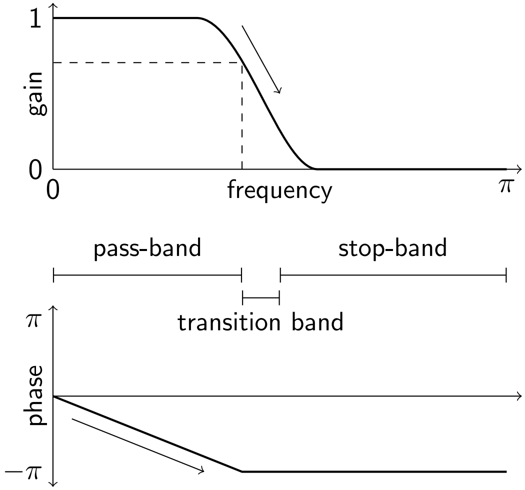
Christophe Rhodes
c.rhodes@gold.ac.uk

Autumn 2009, Tuesdays, 10:00–15:00
Winter 2010, Tuesdays, 13:30–17:30

Audio Filtering

Finite Impulse Response Filters

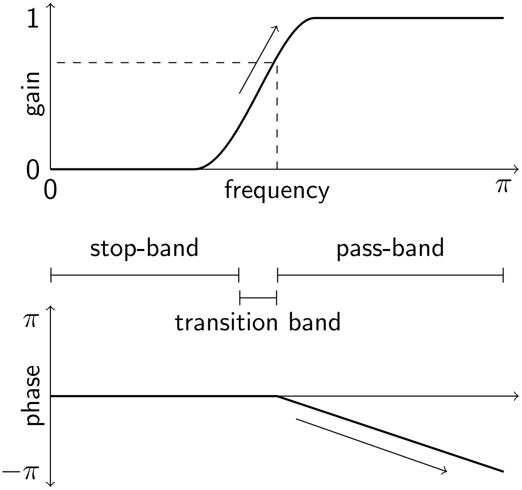
Practical **low-pass** filter frequency response:



Audio Filtering

Finite Impulse Response Filters

Practical **high-pass** filter frequency response:



Audio Filtering

Finite Impulse Response Filters

Practical **low-pass** filter frequency response:

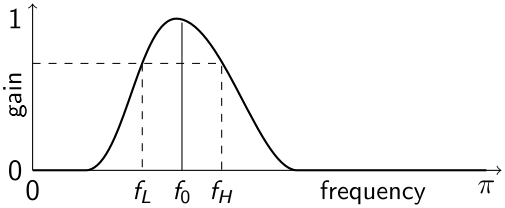
- gain of close to 1 for low frequencies;
- gain of $\frac{1}{\sqrt{2}}$ at cutoff frequency;
- rapid decline in gain at frequencies higher than cutoff;
- linear phase delay in pass-band region.

Practical **high-pass** filter frequency response:

- gain of close to 1 for high frequencies;
- gain of $\frac{1}{\sqrt{2}}$ at cutoff frequency;
- rapid decline in gain at frequencies lower than cutoff;
- linear phase delay in pass-band region.

Audio Filtering

Finite Impulse Response Filters



$$B = f_H - f_L = \Delta f$$

and

$$Q = \frac{f_0}{B}$$

Audio Filtering

Finite Impulse Response Filters

Octave:

- filter construction with `fir1` function:
  - `n`: **order** parameter;
  - `w`: band edges;
  - `type`, `window`, `scale` parameters;
- visualization with `freqz` function;
- application with `filter` function.
  - `y = filter(fir1(...), 1, x)`
  - or use `conv`

Audio Filtering

Subtractive Synthesis

e.g. Moog synthesizers

- Start with a rich waveform:

  ```
  f = zeros(1,44100);
  f(56:55:22050) = 1;
  f(44100:-1:22051) = f(1:22050);
  x = real(ifft(f));
  ```

- Apply a filter to the waveform:

  ```
  y = conv(h,x);
  ```

- Play the filtered waveform:

  ```
  sound(y, 44100)
  ```

(cf. **additive synthesis**: constructing waveform from explicit addition of partials)

Audio Filtering

Echo and Reverb

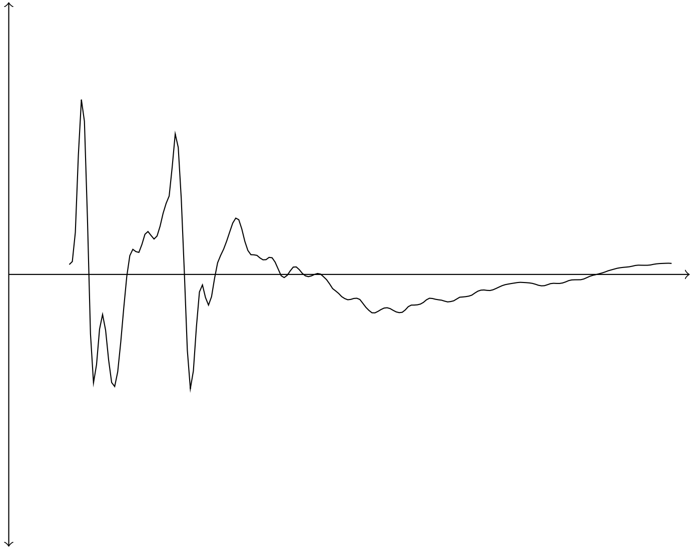
Rooms are systems too. Their impulse response can be categorized into two parts:

- echo:
  - few, discrete impulses at particular times;
  - caused by first reflections of sound off one or two surfaces;
  - typical timescale: $\sim 0.1s$.
- reverb:
  - noisy, decaying waveform;
  - caused by superimposed echoes of echoes of echoes (of echos...);
  - typical timescale: $\sim$ 1s–10s.

Audio Filtering

Echo and Reverb

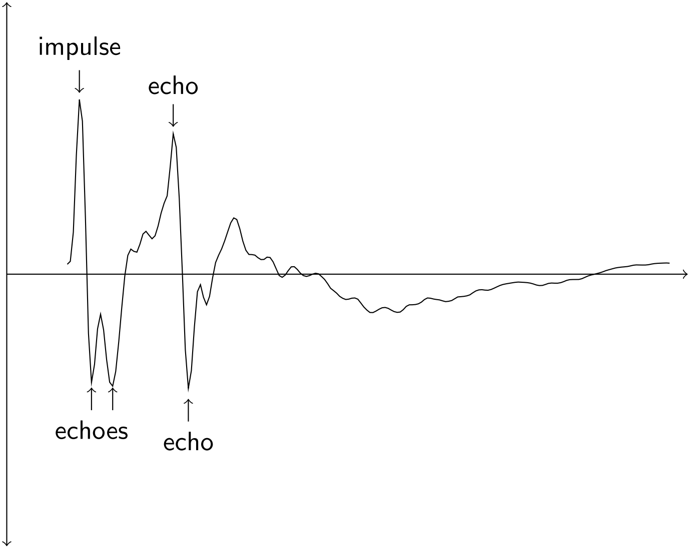
Echo:



Audio Filtering

Echo and Reverb

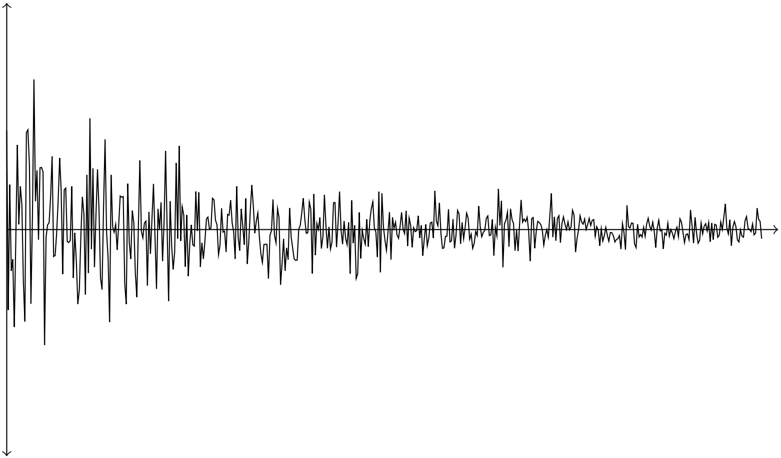
Echo:



Audio Filtering

Echo and Reverb

Reverb:



Audio Filtering

Resampling

Resampling: changing the sample rate of a discrete-time signal (while preserving its meaning).

Applications:

- Applying a filter to a signal with a different sample rate;
- Resampling synthesis: pitch shifting.

Octave: `resample` operator.

- `x`: signal to resample;
- `p`: interpolation factor;
- `q`: decimation factor.

Net effect is to transorm signal sampled at frequency f into the same signal sampled at frequency $\frac{p}{q}f$.