Shhhhhhhhh,

Secret Rabbit Code

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or ..... Erik’s Very Handwavey Guide to the Mechanics of Sample Rate Converters and How to Evaluate their Performance.
What is it?

- Secret Rabbit Code is a sample rate converter.
- Name courtesy of Conrad Parker.
- Written in C, runs on *nix, win32 and Mac OSX.
- Released under the GPL.
- A commercial use license is also available.
- Capable of time varying conversion ratios.
- First unofficial release on October 6th, 2002 (mono only).

- First official release on November 28th, 2002.

- Now in Debian, Suse, Madrake/Mandriva, Fedora etc.

- Contains three different sample rate conversion algorithms, one of which doesn’t suck.

- I am currently working on developing a better algorithm. It will be released when it works. I won’t be discussing it in any detail today.
Sampled Audio

- Sound files like WAV and AIFF etc are basically a string of numbers.

- A common sample rate is 44100 Hz which means 44100 numbers for each second of audio.

- We can display these numbers with respect to time as follows where the height of the line represents the magnitude of the number:
• From a mathematical / engineering point of view real world signals are usually dealt with as a sum of sinusoidal components.

• An analogue signal can only be accurately represented as a sampled signal all of its sinusoidal components have frequencies of less than half the sample rate (ie the signal is band limited).

• Real world signals (eg audio from a CD) do have signal components with frequencies near half the sample rate.

• When converting from a higher to a lower sample rate, we need to ensure that frequency components above half of the destination sample rate are removed before resampling. This makes downsampling more difficult than upsampling.
Definitions

- Define the **critical frequency** as half of the minimum of the input and output sample rates.

- Define the **conversion ratio** as output sample rate divided by the input sample rate.
Sample Rate Conversion

- Original samples in blue.

- We want new samples at the new sample rate in red.

- How do we calculate the new sample values?
Zero Order Hold

- Hey, let's just use the last sample value.

- Something this simple is unlikely to be much good.
Linear Interpolation

- Idea is simple.

- Calculate the slope of a line which will pass through two adjacent samples.

- The new sample is calculated from the slope of the line and the sample on the left.
• Problem: In the general case, linear interpolation sounds like crap.

• It can be OK if the maximum frequency in the source signal is well below the minimum of the source and destination sample rate.

• For the case of upsampling from 44100Hz to 48000Hz, with the source signal consisting of a single sine wave.

<table>
<thead>
<tr>
<th>Sine Freq</th>
<th>Signal to Noise ratio</th>
</tr>
</thead>
<tbody>
<tr>
<td>333 Hz</td>
<td>146.0 dB</td>
</tr>
<tr>
<td>666 Hz</td>
<td>115.8 dB</td>
</tr>
<tr>
<td>1332 Hz</td>
<td>103.8 dB</td>
</tr>
<tr>
<td>2664 Hz</td>
<td>49.8 dB</td>
</tr>
<tr>
<td>5328 Hz</td>
<td>38.7 dB</td>
</tr>
<tr>
<td>10656 Hz</td>
<td>28.4 dB</td>
</tr>
<tr>
<td>21312 Hz</td>
<td>19.5 dB</td>
</tr>
</tbody>
</table>
Polynomial Interpolation

- **Idea**: use more than two points, fit a polynomial to the points and then evaluate the polynomial at the required point.

- For example, pick four points, fit a 3rd or higher order polynomial to the points and then evaluate. Repeat for each required output sample.
• This can work well if the highest source frequency component is well less than 1/n times the critical frequency (where n ≥ 4).

• Better than linear interpolation.

• Still not suitable for the general case of sample rate converting real world audio signals.

• Secret Rabbit Code does not currently include a polynomial interpolator.
Band Limited Sinc Interpolation

- This is one of the officially 'right' ways to do it.

- Technique was published by Julius O. Smith.

- Unemcumbered by patents.

- The Sinc converter in Secret Rabbit Code is a re-implementation of Smith’s technique.

- The technique is too complicated to explain here.

- Using the Sinc based converters in SRC results in a worst case signal to noise ratio of 97dB.
Evaluating a Sample Rate Converter

- What factors make a sample rate converter ‘good’:
  - Signal to Noise Ratio (more is better).
  - Flat passband frequency response (flatter is better).
  - Bandwidth (more is better).
  - Phase response (linear is best).
  - Speed of conversion (faster is better).
  - Dynamic range (more is better).
  - Transport delay (less is better).
Measuring SNR

- Generate a windowed sine wave, with sine frequency less than the critical frequency (defined earlier).

\[
\text{octave}\# \quad x = \text{hanning}(20000) .* \sin(0.04 * 2 * \pi * (1:20000)');
\]

Also need to make sure that the sine frequency is less the bandwidth of the converter (more later).
Pass the signal through the converter and retrieve the output.

Perform a Fast Fourier Transform (FFT) on the converter’s output.

```octave
# f = fft(y);
```

Calculate the magnitude spectrum from the FFT (also throw away the mirrored image half).

```octave
# spec = abs(f(1:length(f)/2));
```

Find the largest peak ($p_1$) and second largest peak ($p_2$) of the magnitude spectrum (ignoring side lobes of main peak) and calculate the SNR using:

$$SNR = 20 \times \log_{10}(\frac{p_1}{p_2})$$
The log magnitude spectrum can be viewed using:

```octave
# plot (20 * log10 (spec / max (spec))) ;
```

which looks something like this.
• Repeat the above steps for a number of sine frequencies and a number of conversion ratios.

• Be sure to include sine frequencies near the critical frequency.

• Be sure to chose conversion ratio greater and less than 1.

• The SNR is the worst (ie lowest) number recorded for any combination of signal frequency and conversion ratio.
Frequency Response and Bandwidth

- Remember that a sampled signal can only contain frequencies less than the critical frequency.

- In order to enforce this requirement, most converters start attenuating frequencies as they get closer to the critical frequency.

- To find the bandwidth, we look at the magnitude verses frequency response and find the frequency ($f_x$) where magnitude drops to 0.707 of the maximum value.
• Calculate the bandwidth of the converter as:

\[ BW = \left( \frac{f_x}{f_c} \right) \]

• The frequency response may also have ripple in the passband. This should be well less than 0.1dB.
Transport Delay

- All of the good sample rate conversion algorithms have a transport delay.

- Neglecting the fact that the input and output sample rates may be different, a transport delay implies that a sound that goes into the converter comes out some finite time later.

- For off line audio processing, the transport delay is not that important.

- For real time processing, it is undesirable for the transport delay to be much more than a couple of milliseconds.

- Measuring it is left as an exercise.
Things to Come

• What about this new converter then?

• Aiming for SNR of up to 150dB.

• The new algorithm should be significantly faster than the existing one.

• Reduced transport delay.

• Easy to parallelize using SSE/SSE2/Altivec instructions.

• Will probably include a integer arithmetic version.
The End

- Is anyone still awake?

- Any questions?