Signal Analysis

Young Won Lim 2/16/18 Copyright (c) 2016 – 2018 Young W. Lim.

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Young Won Lim 2/16/18 Signal Processing with Free Software : Practical Experiments F. Auger soxi

soxi s1.mp3 soxi s1.mp3 > s1_info.txt

Input File Channels Sample Rate Precision Duration File Siz Bit Rate Sample Encoding

sox -n s1.mp3 synth 3.5 sine 440 sox -n s2.wav synth 90000s sine 660:1000 sox -n s3.mp3 synth 1:20 triangle 440 sox -n s4.mp3 synth 1:20 trapezium 440 sox -V4 -n s5.mp3 synth 6 square 440 0 0 40 sox -n s6.mp3 synth 5 noise Sox s1.mp3 -n stat Sox s1.mp3 -n stat > s1_info_stat.txt

Samples read Length (seconds) Scaled by Maximum amplitude Minimum amplitude Midline amplitude Mean norm Mean amplitude **RMS** amplitude Maximum delta Minimum delta Mean delta **RMS** delta Rough frequency Volume adjustment

Sox s1.mp3 -n stats Sox s1.mp3 -n stats > s1_info_stat.txt

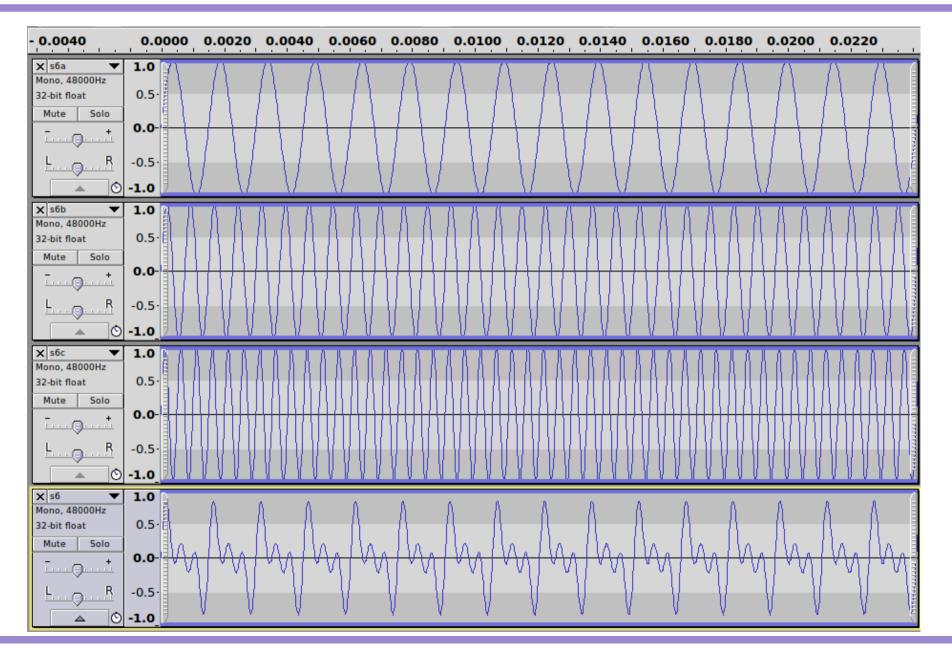
7

DC Offset Min level Max level Pk lev dB RMS lev dB RMS Pk dB RMS Tr dB Crest factor Flat factor Pk count Bit-depth Num samples Length s Scale max Window s

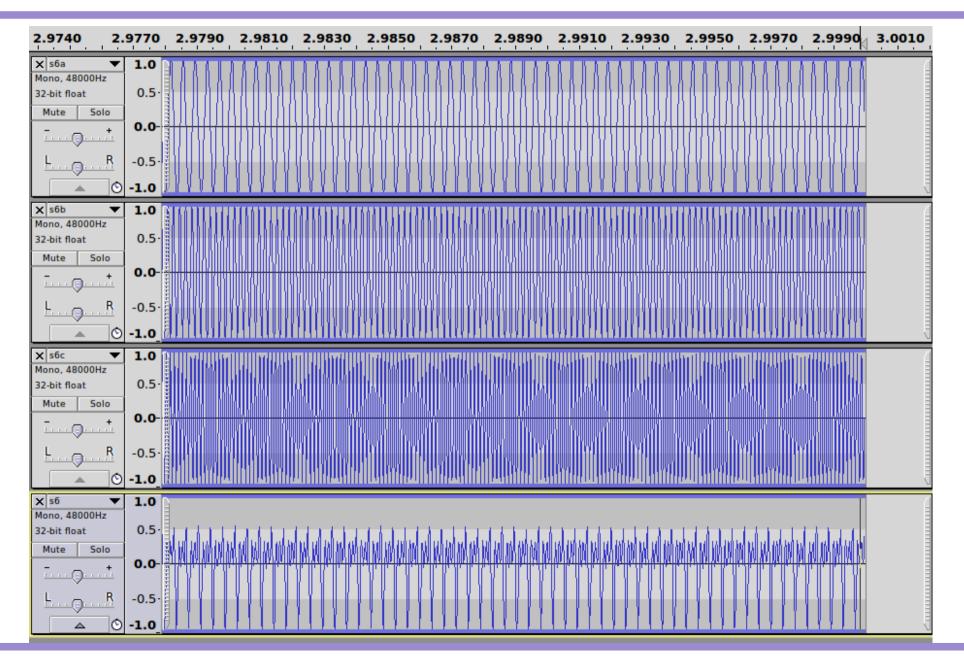
Sox -n s1a.wav synth 3 sine 660-2640 Sox -n s1b.wav synth 3 sine 1320-5280 Sox -n s1c.wav synth 3 sine 1980-7920 Sox -m s1a.wav s1b.wav s1c.wav s1.wav

sox s1.wav -n spectrogram -o s1_sp1.png sox s1.wav -n spectrogram -m -o s1_sp2.png sox s1.wav -n spectrogram -l -o s1_sp3.png sox s1.wav -n spectrogram -l -m -o s1_sp4.png sox s1.wav -n spectrogram -l -m -S 0.5 -d 1.3 \ -o s1_sp5.png

sox -m sla.wav slb.wav slc.wav sl.wav - beginning

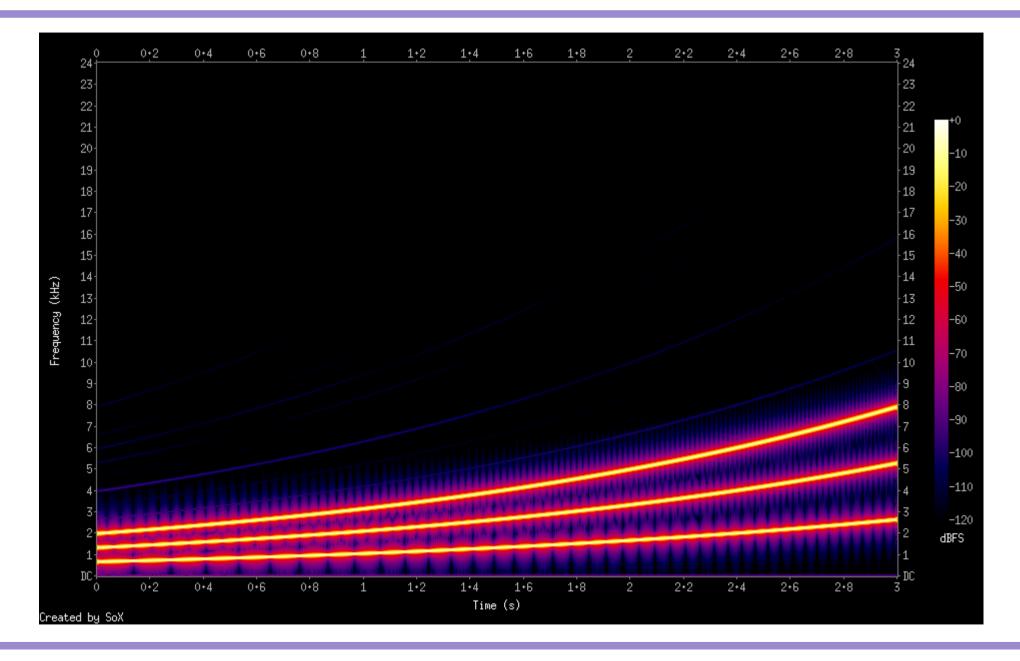


sox -m s1a.wav s1b.wav s1c.wav s1.wav - ending



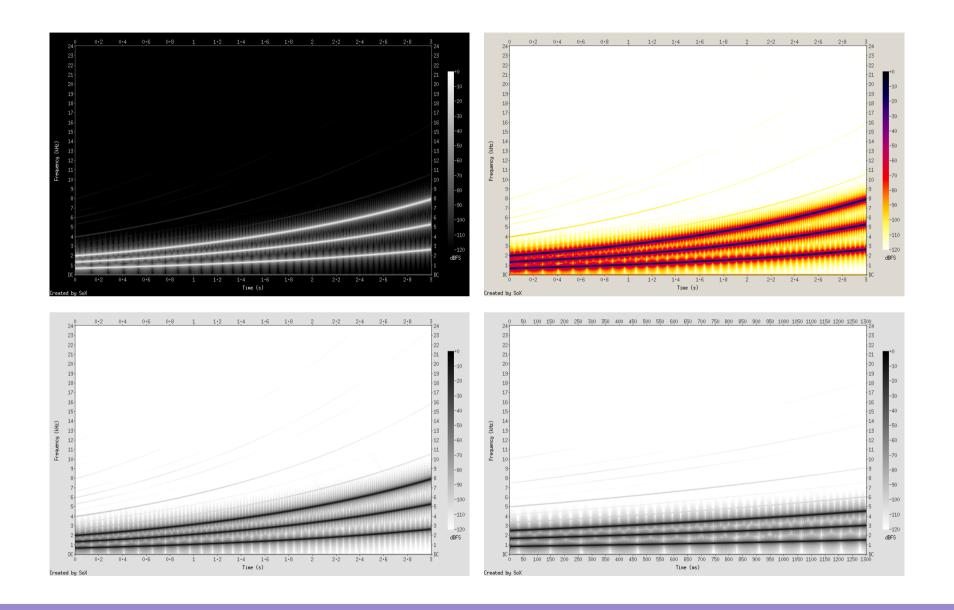
sox -n s1a.wav synth 3 sine 660-2640 sox -n s1b.wav synth 3 sine 1320-5280 sox -n s1c.wav synth 3 sine 1980-7920 sox -m s1a.wav s1b.wav s1c.wav s1.wav

sox s1.wav -n spectrogram -o s1_sp1.png sox s1.wav -n spectrogram -m -o s1_sp2.png sox s1.wav -n spectrogram -l -o s1_sp3.png sox s1.wav -n spectrogram -l -m -o s1_sp4.png sox s1.wav -n spectrogram -l -m -S 0.5 -d 1.3 \ -o s1_sp5.png soxi



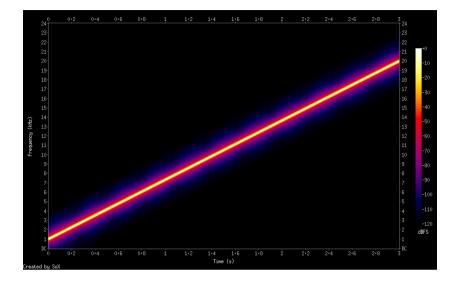
Audio Signal Analysis (1A)

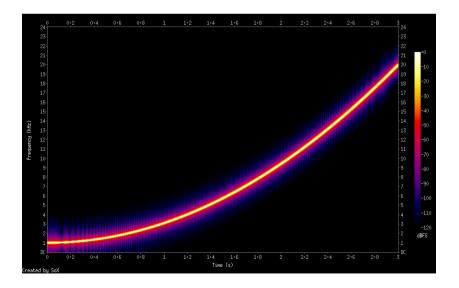
12

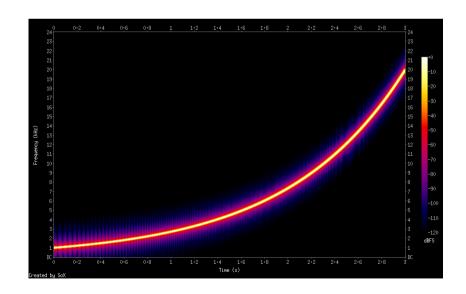


sox -n chirp1.wav synth 3 sine 1000:20000 sox -n chirp2.wav synth 3 sine 1000+20000 sox -n chirp3.wav synth 3 sine 1000/20000

sox chirp1.wav -n spectrogram -o chirp1_sp.png sox chirp2.wav -n spectrogram -o chirp2_sp.png sox chirp3.wav -n spectrogram -o chirp3_sp.png



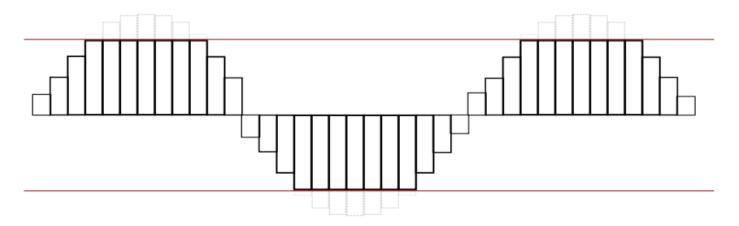




Decibels relative to full scale (dBFS) is a unit of measurement for amplitude levels in digital systems, which have a defined maximum peak level.

The level of 0 dBFS is assigned to the maximum possible digital level. a signal that reaches 50% of the maximum level has a level of -6 dBFS, which is 6 dB below full scale.

Conventions differ for root mean square (RMS) measurements, but all peak measurements smaller than the maximum are negative levels.



https://en.wikipedia.org/wiki/DBFS

Sox remix usage examples

sox input.wav output.wav remix 6 7 8 0

creates an output file with four channels where channels 1, 2, and 3 are copies of channels 6, 7, and 8 in the input file, and channel 4 is silent.

sox input.wav output.wav remix 1-3,73

the left channel is a mix-down of input channels 1, 2, 3, and 7 and the right channel is a copy of input channel 3. when a range of channels is specified the channel numbers to the left and right of the hyphen are optional and default to 1 and to the number of input channels respectively

sox input.wav output.wav remix -

performs a mix-down of all input channels to mono.

http://sox.sourceforge.net/sox.html

Sox trim usage examples

sox infile outfile trim 0 10

will copy the first ten seconds

play infile trim 12:34 =15:00 -2:00 play infile trim 12:34 2:26 -2:00

will both play from 12 minutes 34 seconds into the audio up to 15 minutes into the audio (i.e. 2 minutes and 26 seconds long), then resume playing two minutes before the end of audio.



| Audio Signal | |
|--------------|------|
| Analysis | (1A) |

For an <u>input</u> file, the most common use for this option is to inform SoX of the <u>number of bits per sample</u> in a 'raw' ('headerless') audio file.

sox -r 16k -e signed -b 8 input.raw output.wav

converts a particular 'raw' file to a self-describing 'WAV' file.

For an <u>output</u> file, this option can be used (perhaps along with -e) to set the output encoding size.

By default, the output encoding size will be set to the input encoding size. (providing it is supported by the output file type)

sox input.cdda –b 24 output.wav

converts raw CD digital audio (16-bit, signed-integer) to a 24-bit (signed-integer) 'WAV' file.

dither [-S|-s|-f filter] [-a] [-p precision]

Apply dithering to the audio. Dithering deliberately adds a small amount of noise to the signal in order to mask audible quantization effects that can occur if the output sample size is less than 24 bits.

With no options, this effect will add triangular (TPDF) white noise. <u>Noise-shaping</u> (only for certain sample rates) can be selected with -s.

With the **-f** option, it is possible to select a particular <u>noise-shaping filter</u> from the following list: lipshitz, f-weighted, modified-e-weighted, improved-e-weighted, gesemann, shibata, low-shibata, high-shibata.

Note that most filter types are available only with 44100Hz sample rate. The filter types are distinguished by the following properties: audibility of noise, level of (inaudible, but in some circumstances, otherwise problematic) shaped high frequency noise, and processing speed.

the following two commands are equivalent:

sox input.wav -r 48k output.wav bass -b 24 sox input.wav bass -b 24 rate 48k

the rate option is more flexible allows rate's other options and allows the effects to be ordered arbitrarily

sox input.wav -b 16 output.wav rate -s -a 44100 dither -s

sox input.wav -b 24 output.aiff rate -v -I -b 90 48k

http://sox.sourceforge.net/sox.html

sox input.wav -b 16 output.wav rate -s -a 44100 dither -s

default (high) quality resampling; overrides: steep filter, allow aliasing; to 44.1kHz sample rate; noise-shaped dither to 16-bit WAV file.

-s 'steep filter' changes resampling band-width from the default 95% (based on the 3dB point), to 99%.

a option is given, then aliasing/imaging above the pass-band is allowed.

http://sox.sourceforge.net/sox.html

sox input.wav -b 24 output.aiff rate -v -I -b 90 48k

very high quality resampling; overrides: intermediate phase, band-width 90%; to 48k sample rate; store output to 24-bit AIFF file.

M, -I, or -L option : Minimum, intermediate, or linear phase response
 p option : a custom phase response

Note that phase responses between 'linear' and 'maximum' (greater than 50) are rarely useful.

Delay one or more audio channels such that they start at the given position.

delay 1.5 +1 3000s

delays the <u>first channel</u> by 1.5 seconds,
the <u>second channel</u> by 2.5 seconds
(one second more than the previous channel),
the third channel by 3000 samples,
and leaves any other channels that may be present un-delayed.

The following (one long) command plays a **<u>chime</u>** sound:

```
play –n synth –j 3 sin %3 sin %–2 sin %–5 sin %–9 \
sin %–14 sin %–21 fade h .01 2 1.5 delay \
1.3 1 .76 .54 .27 remix – fade h 0 2.7 2.5 norm –1
```

and this plays a guitar chord:

play –n synth pl G2 pl B2 pl D3 pl G3 pl D4 pl G4 \ delay 0 .05 .1 .15 .2 .25 remix – fade 0 4 .1 norm –1

http://sox.sourceforge.net/sox.html

the audio is passed unmodified through the SoX processing chain to create a spectrogram of the audio

the spectrogram is rendered in a PNG file

- time in the X-axis
- frequency in the Y-axis
- magnitude in the Z-axis by the colour or the intensity

multiple <u>channels</u> are shown <u>from top to bottom</u> starting from channel 1 (the left channel for stereo audio).

http://sox.sourceforge.net/sox.html

Sox spectrogram usage examples (1)

'my.wav' is a stereo file
'spectrogram.png' is a spectrogram

sox my.wav -n spectrogram

sox my.wav -n remix 2 trim 20 30 spectrogram

sox my.wav – n rate 6k spectrogram

http://sox.sourceforge.net/sox.html

Sox spectrogram usage examples (2)

* to analyze a smaller portion of the audio

sox my.wav – n remix 2 trim 20 30 spectrogram

only from the <u>second</u> (right) channel <u>duration</u> of 30 seconds <u>starting</u> from 20 seconds in

* to analyze a small portion of the frequency domain the rate effect may be used

sox my.wav – n rate 6k spectrogram

detailed analysis of frequencies up to 3kHz (half the sampling rate) 2 20 30

6k

http://sox.sourceforge.net/sox.html

sox my.wav -n trim 0 10 spectrogram -x 600 -y 200 -z 100

controls the size of the spectrogram's X, Y, Z axes 600 by 200 <u>pixels</u> in size the Z-axis range will be 100 <u>dB</u>

sox -n -n synth 6 tri 10k:14k spectrogram -z 100 -w kaiser

an analysis 'window' with <u>high</u> <u>dynamic</u> <u>range</u> is selected to best display the spectrogram of a swept triangular wave.

http://sox.sourceforge.net/sox.html

append the following to the 'chime' command in the description of the delay effect (above):

rate 2k spectrogram -X 200 -Z -10 -w kaiser

X-axis 200 pixels/second

play –n synth –j 3 sin %3 sin %–2 sin %–5 sin %–9 \ sin %–14 sin %–21 fade h .01 2 1.5 **delay** \ **1.3 1 .76 .54 .27** remix – fade h 0 2.7 2.5 norm –1

http://sox.sourceforge.net/sox.html

to control the appearance (colour-set, brightness, contrast, etc.) and filename of the spectrogram; e.g. with

sox my.wav -n spectrogram -m -l -o print.png

a spectrogram is created suitable for printing on a 'black and white' printer.

m : Creates a <u>monochrome</u> spectrogram (the default is colour).
I : Creates a '<u>printer</u> friendly' spectrogram with a <u>light</u> <u>background</u> (the default has a dark background).

-d duration

sets the X-axis resolution such that audio with the given duration (a time specification) fits the selected (or default) X-axis width.

sox input.mp3 output.wav -n spectrogram -d 1:00 stats

creates a spectrogram showing the <u>first minute</u> of the audio, whilst the **stats** effect is applied to the entire audio signal.

See also -X for an alternative way of setting the X-axis resolution.

http://sox.sourceforge.net/sox.html

-x num

Sets the X-axis size in pixels

Change the (maximum) width (X-axis) of the spectrogram default value of 800 pixels [100 \sim 200000]. See also -X and -d.

–X num

X-axis **pixels/second**; the default is auto-calculated to fit the given or known audio duration to the X-axis size, or 100 otherwise. with -d, this option affects the width of the spectrogram; otherwise, it affects the duration of the spectrogram. [1 ~ 5000] need not be an integer. a slight adjustment for processing quantisation reasons; if so, SoX will report the actual number used (viewable when the SoX global option -V is in effect). See also -x and -d.

-y num

Sets the **Y-axis size** <u>in pixels</u> (*per channel*); this is the number of **frequency bins** used N.B. it can be slow to produce the spectrogram if this number is not one more than a power of two (e.g. 129). By default the Y-axis size is chosen automatically (depending on the number of channels). See –Y for alternative way of setting spectrogram height.

-Y num

Sets the target **total height** of the spectrogram(s). The default value is 550 pixels. Using this option (and by default), SoX will choose a height for individual spectrogram channels that is one more than a power of two, so the actual total height may fall short of the given number. However, there is also a minimum height per channel so if there are many channels, the number may be exceeded. See –y for alternative way of setting spectrogram height.

| Audio Sig | Inal |
|-----------|------|
| Analysis | (1A) |

−z num

Z-axis (colour) range in dB, default 120. This sets the dynamic-range of the spectrogram to be -num dBFS to 0 dBFS. Num may range from 20 to 180. Decreasing dynamic-range effectively increases the 'contrast' of the spectrogram display, and vice versa.

–Z num

Sets the upper limit of the Z-axis in dBFS. A negative num effectively increases the 'brightness' of the spectrogram display,

−q num

Sets the Z-axis quantisation,

i.e. the <u>number</u> of different <u>colours</u> (or <u>intensities</u>)

in which to render Z-axis values.

A small number (e.g. 4) will give a 'poster'-like effect

making it easier to discern magnitude bands of similar level.

Small numbers also usually result in small PNG files.

The number given specifies the number of colours to use inside the Z-axis range; two colours are reserved to represent out-of-range values.

-w name

Window: Hann (default), Hamming, Bartlett, Rectangular, Kaiser or Dolph.

By default, SoX uses the Hann window

which has good all-round frequency-resolution

and dynamic-range properties.

For better frequency resolution (but lower dynamic-range), select a Hamming window;

for higher dynamic-range (but poorer frequency-resolution), select a Dolph window.

Kaiser, Bartlett and Rectangular windows are also available.

–W num

Window adjustment parameter. This can be used to make small adjustments to the Kaiser or Dolph window shape.

A positive number (up to ten) increases its dynamic range,

a negative number decreases it.

- Hann (default)
- Hamming
- Bartlett
- Rectangular
- Kaiser
- Dolph

-s

Allow slack overlapping of DFT windows. This can, in some cases, increase image sharpness and give greater adherence to the -x value, but at the expense of a little spectral loss.

-m

Creates a monochrome spectrogram (the default is colour).

-h

Selects a high-colour palette-less visually pleasing than the default colour palette, but it may make it easier to differentiate different levels. If this option is used in conjunction with -m, the result will be a hybrid monochrome/colour palette.

http://sox.sourceforge.net/sox.html

-p num

Permute the colours in a colour or hybrid palette. The num parameter, from 1 (the default) to 6, selects the permutation.

-1

Creates a 'printer friendly' spectrogram with a light background (the default has a dark background).

-a

Suppress the display of the axis lines. This is sometimes useful in helping to discern artifacts at the spectrogram edges.

-r

Raw spectrogram: suppress the display of axes and legends.

http://sox.sourceforge.net/sox.html

-A

Selects an alternative, fixed colour-set. This is provided only for compatibility with spectrograms produced by another package. It should not normally be used as it has some problems, not least, a lack of differentiation at the bottom end which results in masking of low-level artifacts.

-t text

Set the image title - text to display above the spectrogram.

-c text

Set (or clear) the image comment - text to display below and to the left of the spectrogram.

–o file

Name of the spectrogram output PNG file, default 'spectrogram.png'.

If '-' is given, the spectrogram will be sent to standard output (stdout).

http://sox.sourceforge.net/sox.html

Sox stat usage examples (1)

stat [-s scale] [-rms] [-freq] [-v] [-d]

Display time and frequency domain statistical information about the audio. Audio is passed unmodified through the SoX processing chain.

The **-s** option can be used to scale the input data by a given factor. The default value of scale is 2147483647 (i.e. the maximum value of a 32-bit signed integer). Internal effects always work with signed long PCM data and so the value should relate to this fact.

http://sox.sourceforge.net/sox.html

Sox stat usage examples (2)

stat [-s scale] [-rms] [-freq] [-v] [-d]

The **-rms** option will convert all output average values to 'root mean square' format.

The $-\mathbf{v}$ option displays only the 'Volume Adjustment' value.

The **-freq** option calculates the input's power spectrum (4096 point DFT) instead of the statistics listed above.

This should only be used with a single channel audio file.

The **-d** option displays a hex dump of the 32-bit signed PCM data audio in SoX's internal buffer. This is mainly used to help track down endian problems that sometimes occur in cross-platform versions of SoX. *n* is the duration of the audio in samples, *c* is the number of audio channels, *r* is the audio sample rate, and x_k represents the PCM value (in the range -1 to +1 by default) of each successive sample in the audio Note that the delta measurements are not applicable for multi-channel audio.

 $\begin{array}{ll} \textit{mean amplitude} & \frac{1}{n} \sum x_{k} \\ \textit{RMS amplitude} & \sqrt{\frac{1}{n} \sum x_{k}} \\ \textit{max delta} & \textit{max} \left(|x_{k} - x_{k-1}| \right) \\ \textit{min delta} & \textit{min} \left(|x_{k} - x_{k-1}| \right) \\ \textit{mean delta} & \sqrt{\frac{1}{n-1} \sum \left(|x_{k} - x_{k-1}| \right)} \\ \textit{rough frequency} \\ \textit{volum adjustment} \end{array}$

http://sox.sourceforge.net/sox.html

Max amplitude : The maximum samplevalue in the audio; usually this will be a positive number. Min amplitude : The minimum sample value in the audio; usually this will be a negative number. Mean norm : The average of the absolute value of each sample in the audio. **Mean amplitude** : The average of each sample in the audio. If this figure is non-zero, then it indicates the presence of a D.C. offset (which could be removed using the dcshift effect). **RMS amplitude** : The level of a D.C. signal that would have the same power as the audio's average power. **Rough frequency** : in Hz Volume Adjustment : The parameter to the vol effect which would make the audio as loud as possible without clipping. Note: See the discussion on Clipping above for reasons why it is rarely a good idea actually to do this.

http://sox.sourceforge.net/sox.html

stats [-b bits|-x bits|-s scale] [-w window-time]

Display time domain statistical information about the audio channels; audio is passed unmodified through the SoX processing chain. Statistics are calculated and displayed for each audio channel and, where applicable, an overall figure is also given.

For example, for a typical well-mastered stereo music file:

| Audio Sig | nal |
|------------------|------|
| Analysis | (1A) |

http://sox.sourceforge.net/sox.html

DC offset, Min level, and Max level are shown, by default, in the range ±1.
-b (bits) to be scaled to a signed integer with the given number of bits;
for example, for 16 bits, the scale would be -32768 to +32767.
-x is similar to -b except in hexadecimal
-s scales by a given floating-point number.

Pk lev dB and **RMS lev dB** are <u>standard peak</u> and <u>RMS level</u> measured in dBFS. **RMS Pk dB** and **RMS Tr dB** are <u>peak</u> and <u>trough</u> values for <u>RMS level</u> measured over a short window (default 50ms).

Crest factor is the <u>standard</u> <u>ratio</u> of <u>peak</u> to <u>RMS</u> level (note: not in dB).

Flat factor is a measure of the <u>flatness</u>
(i.e. consecutive samples with the same value)
of the signal at its peak levels
(i.e. either Min level, or Max level).
Pk count is the <u>number</u> of <u>occasions</u>
(not the number of samples)
that the signal attained either <u>Min</u> level, or <u>Max</u> level.

http://sox.sourceforge.net/sox.html



The right-hand **Bit-depth** figure is the standard definition of bit-depth i.e. bits less significant than the given number are fixed at zero. the number of least significant bits that are fixed at zero

The left-hand **Bit-depth** figure is the number of most significant bits that are fixed at zero (or one for negative numbers) subtracted from the right-hand figure (the number subtracted is directly related to Pk lev dB).

http://sox.sourceforge.net/sox.html

For multi-channel audio, an **overall figure** for each of the above measurements is given and derived from the channel figures as follows:

- DC offset:
- maximum magnitude;
- Max level
- Pk lev dB
- RMS Pk dB
- Bit-depth
- maximum
- Min level
- RMS Tr dB
- minimum
- RMS lev dB
- Flat factor
- Pk count
- average
- Crest factor: not applicable.

| | Overall | Left | Right |
|--------------|-----------|-----------|-----------|
| DC offset | 0.000803 | -0.000391 | 0.000803 |
| Min level | -0.750977 | -0.750977 | -0.653412 |
| Max level | 0.708801 | 0.708801 | 0.653534 |
| Pk lev dB | -2.49 | -2.49 | -3.69 |
| RMS lev dB | -19.41 | -19.13 | -19.71 |
| RMS Pk dB | -13.82 | -13.82 | -14.38 |
| RMS Tr dB | -85.25 | -85.25 | -82.66 |
| Crest factor | _ | 6.79 | 6.32 |
| Flat factor | 0.00 | 0.00 | 0.00 |
| Pk count | 2 | 2 | 2 |
| Bit-depth | 16/16 | 16/16 | 16/16 |
| Num samples | 7.72M | | |
| Length s | 174.973 | | |
| Scale max | 1.000000 | | |
| Window s | 0.050 | | |

http://sox.sourceforge.net/sox.html

Length s is the duration in seconds of the audio, and
Num samples is equal to the sample-rate multiplied by Length.
Scale Max is the scaling applied to the first three measurements; specifically, it is the maximum value that could apply to Max level.
Window s is the length of the window used for the peak and trough RMS measurements.

http://sox.sourceforge.net/sox.html

sox -n chirp1.wav synth 3 sine 1000:20000 sox -n chirp2.wav synth 3 sine 1000+20000 sox -n chirp3.wav synth 3 sine 1000/20000

sox chirp1.wav -n spectrogram -o chirp1_sp.png sox chirp2.wav -n spectrogram -o chirp2_sp.png sox chirp3.wav -n spectrogram -o chirp3_sp.png sox -n chirp1.wav synth 3 sine 1000:20000 sox -n chirp2.wav synth 3 sine 1000+20000 sox -n chirp3.wav synth 3 sine 1000/20000

sox chirp1.wav -n spectrogram -o chirp1_sp.png sox chirp2.wav -n spectrogram -o chirp2_sp.png sox chirp3.wav -n spectrogram -o chirp3_sp.png

References

[1] F. Auger, Signal Processing with Free Software : Practical Experiments