CS489/W17 Lecture 06 Thu 19 Jan 2017

Office hours: Friday
Monday

Dynamic Range Compression
- what it is and why we (may) want to do it
- Demo #1: RNC1773 ("Real Nice Compressor")
- theory for "hard knee" compressor

- Demo #2: Synth sounds (from Audacity) put through RNC1773

A1.
Demo included

The aim of compression is to increase the level of the signal, sometimes by
a very large margin.
Quiet parts are amplified, while loud parts are "flattened out".
In the extreme case, the signal is simply clipped.
Compressors are sometimes called "limiters".

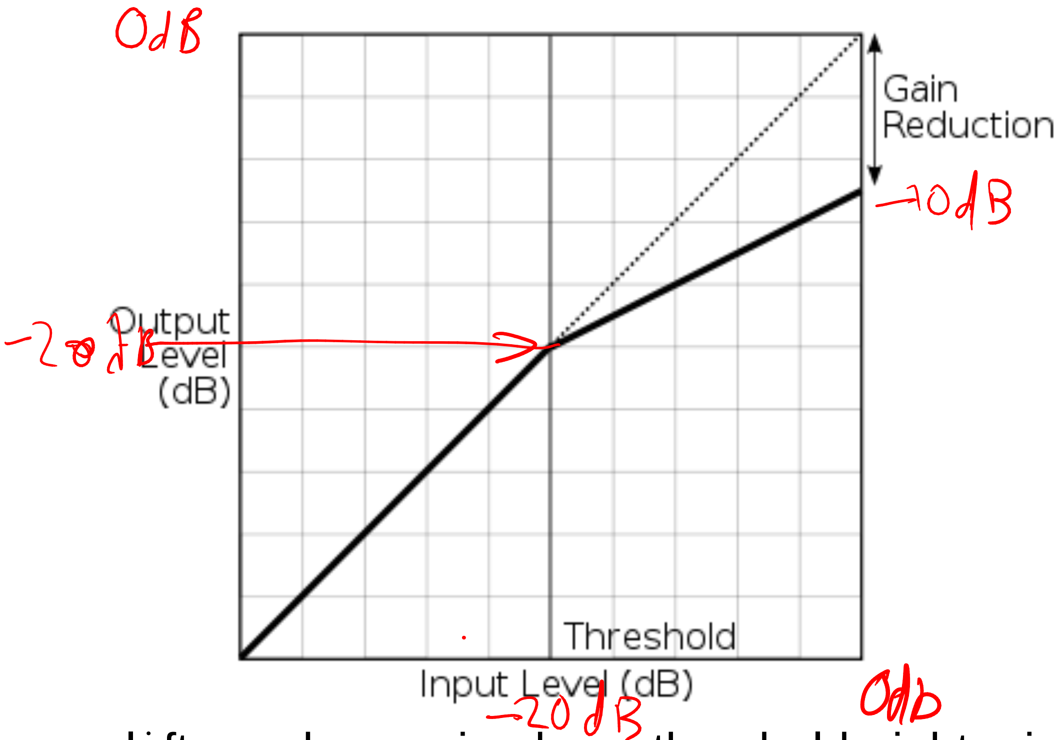
Why compress?
- Live sound, to limit output level, even if it introduces some distortion

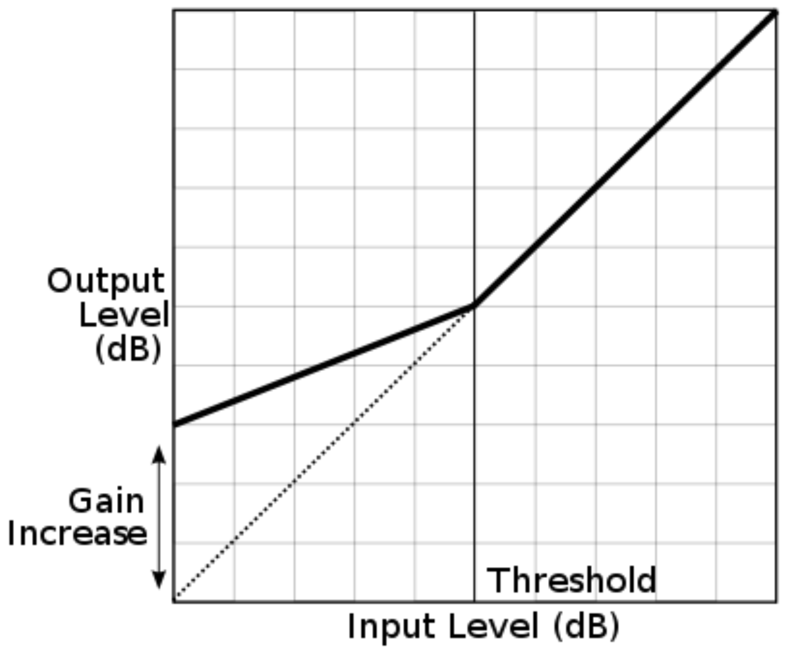
- To make music (or individual instruments) sound "louder".
    This has been abused, to make commercials and some music sound "louder" than others.

- If you have a noisy environment (eg., moving vehicle), you can make the sound "louder"

- hearing aids use multi band compresion. Each frequency band
is compressed separately. This allow compensation for hearing loss at
different frequencies. The compressor is "tuned" to the listener...

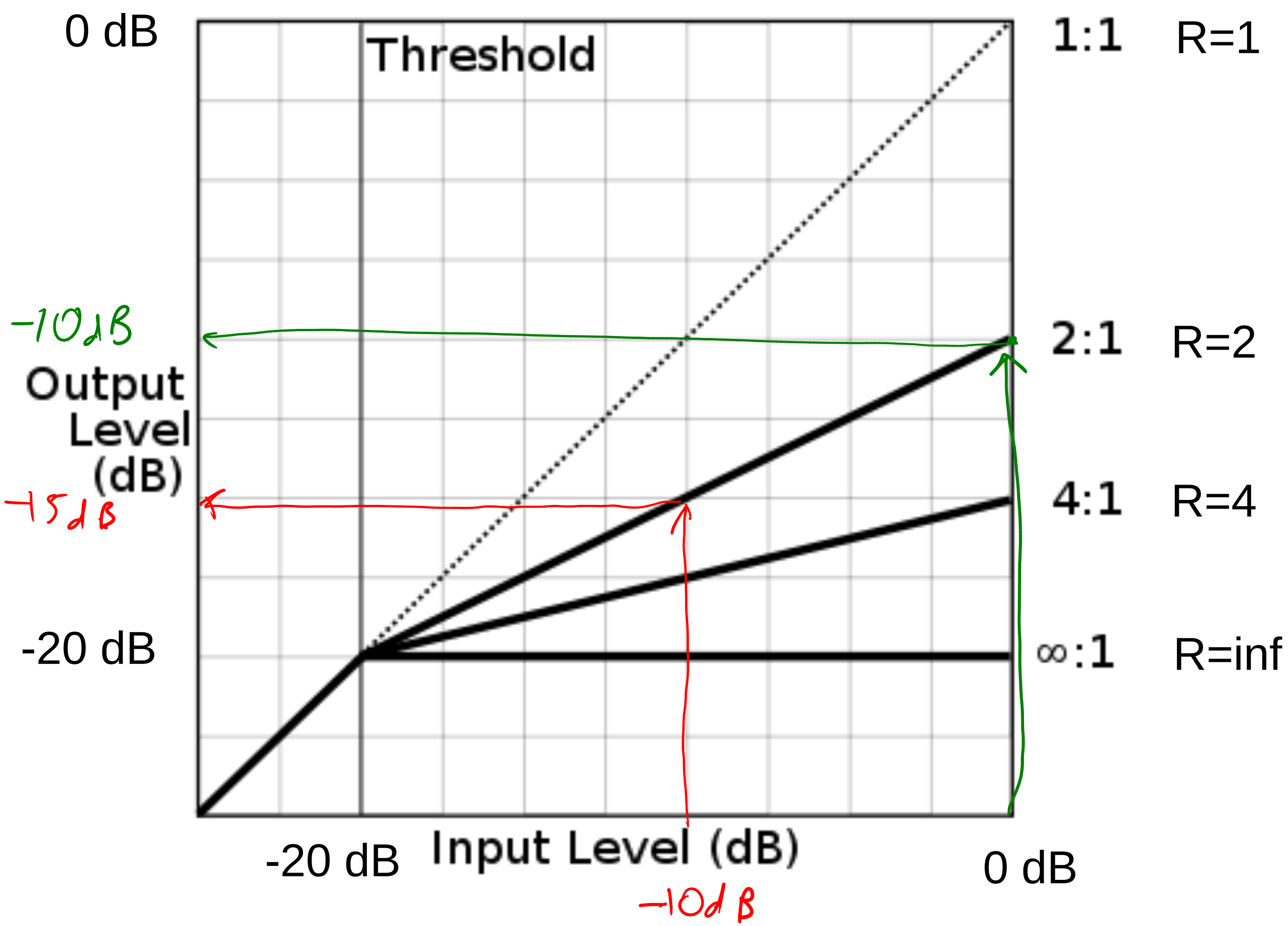
Compressor
(https://en.wikipedia.org/wiki/Dynamic_range_compression)





left = reduce gain above threshold, right = increase gain below threshold
we consider left case here.

Compressor Ratio (R:1)
(https://en.wikipedia.org/wiki/Dynamic_range_compression)



Example:
T: -20 dB
R: 2

Input. I = -10 dB

Output = T+ (I - T)/R
= -20 + (-10 -(-20))/2
= -20 + 10/2
= -15 dB

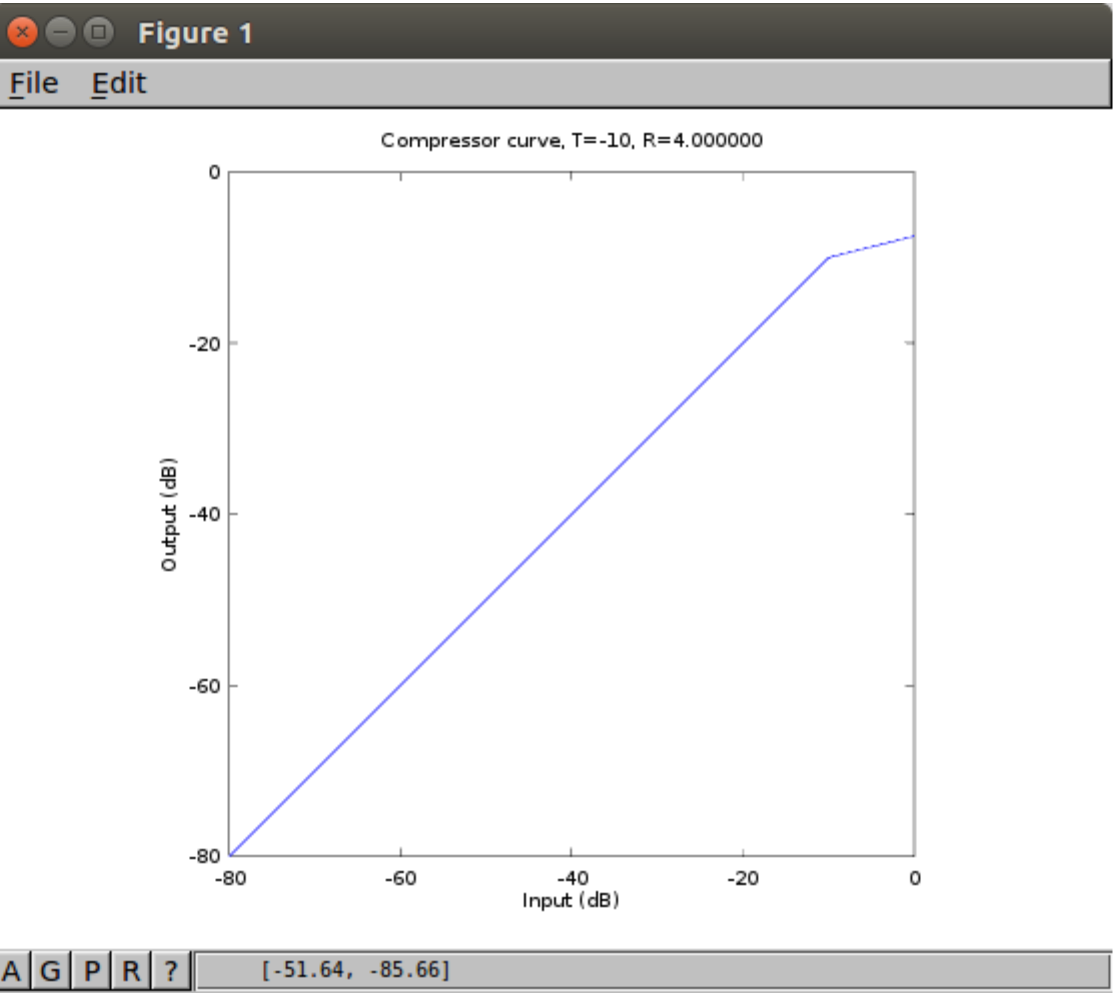
In other words, for every decibel input signal is above T, we get gain 1/2 decibels out.

When I = 0 dB (maximum),
Output = T -T/R = -20 + 20/2
= -10 dB

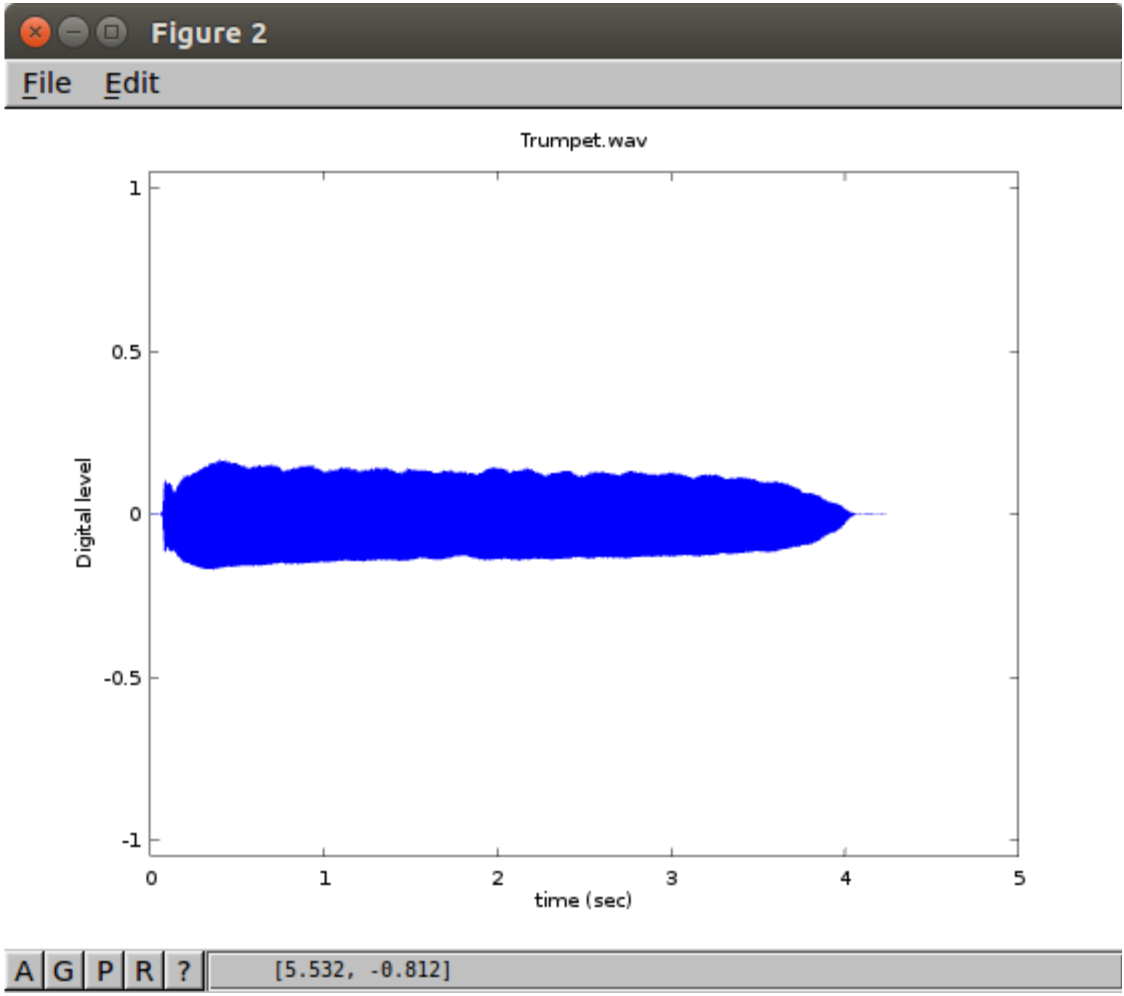
Demonstration.
This is part of A1.

	Trumpet	Oboe
"Light" compression T = -10dB R = 4 Gout = 0dB		
"Mid" compression T = -10dB R = 10; Gout = 0dB		
"Heavy" compression T = -10dB R = 60; Gout = 0dB		

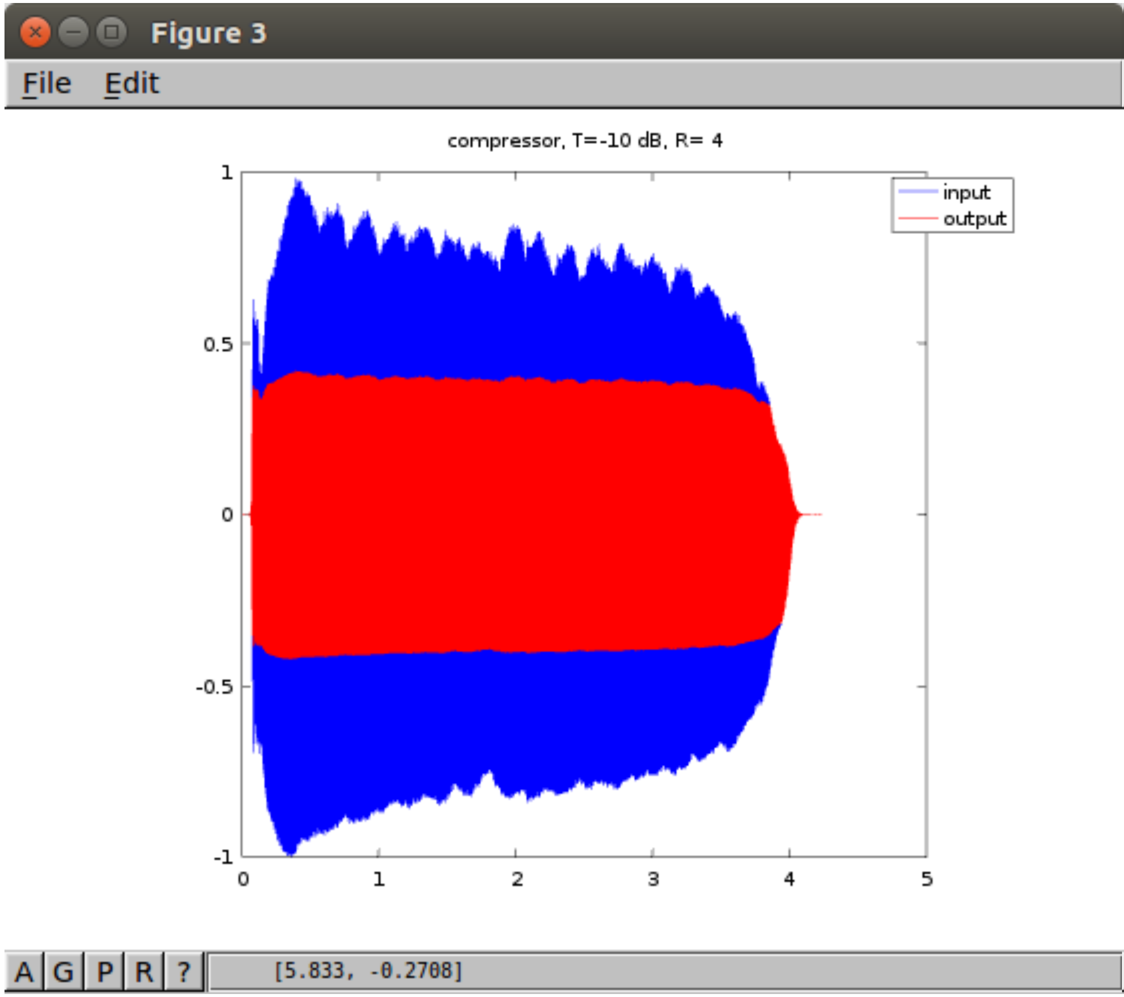
Compressor curve (T=-10dB, R=4)



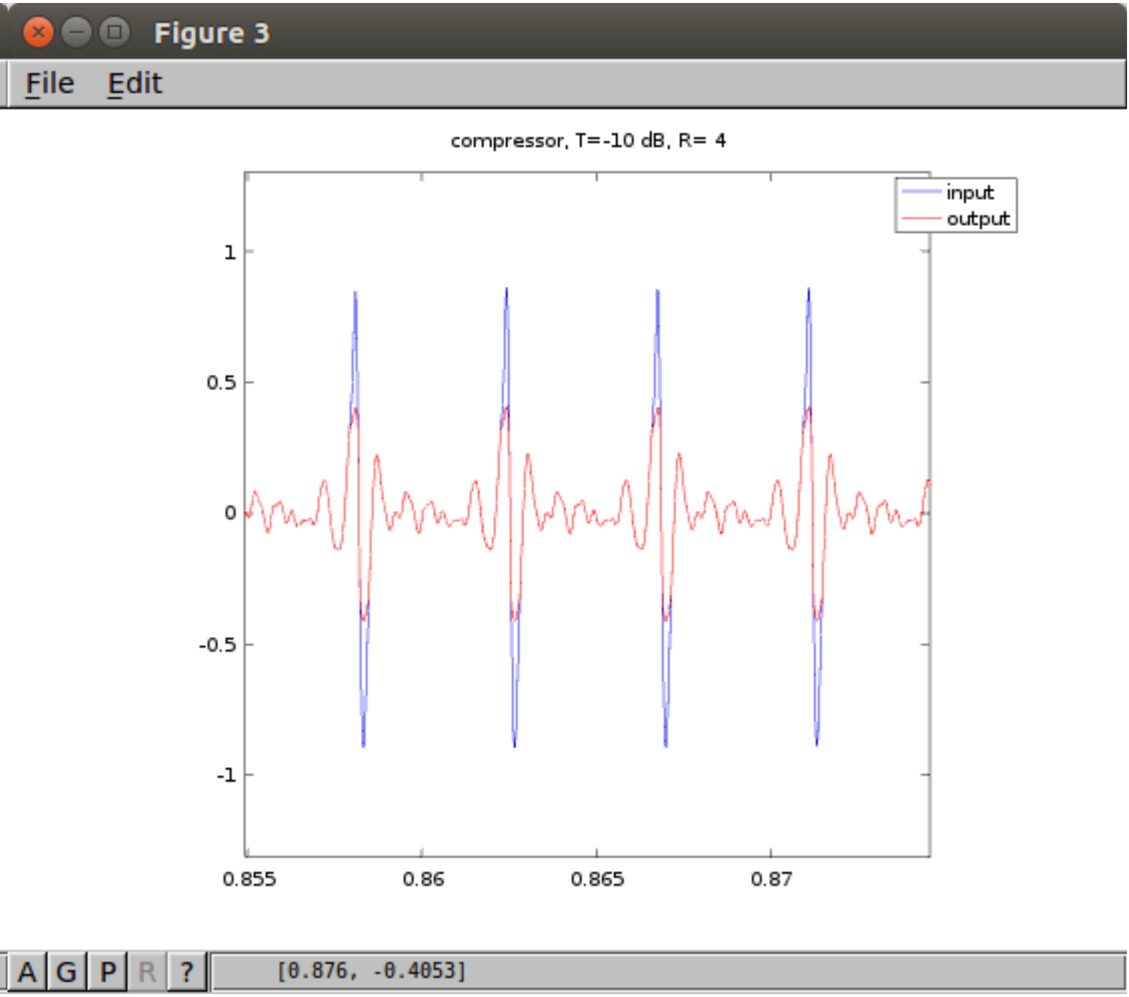
Input WAV file (Trumpet) unnormalized



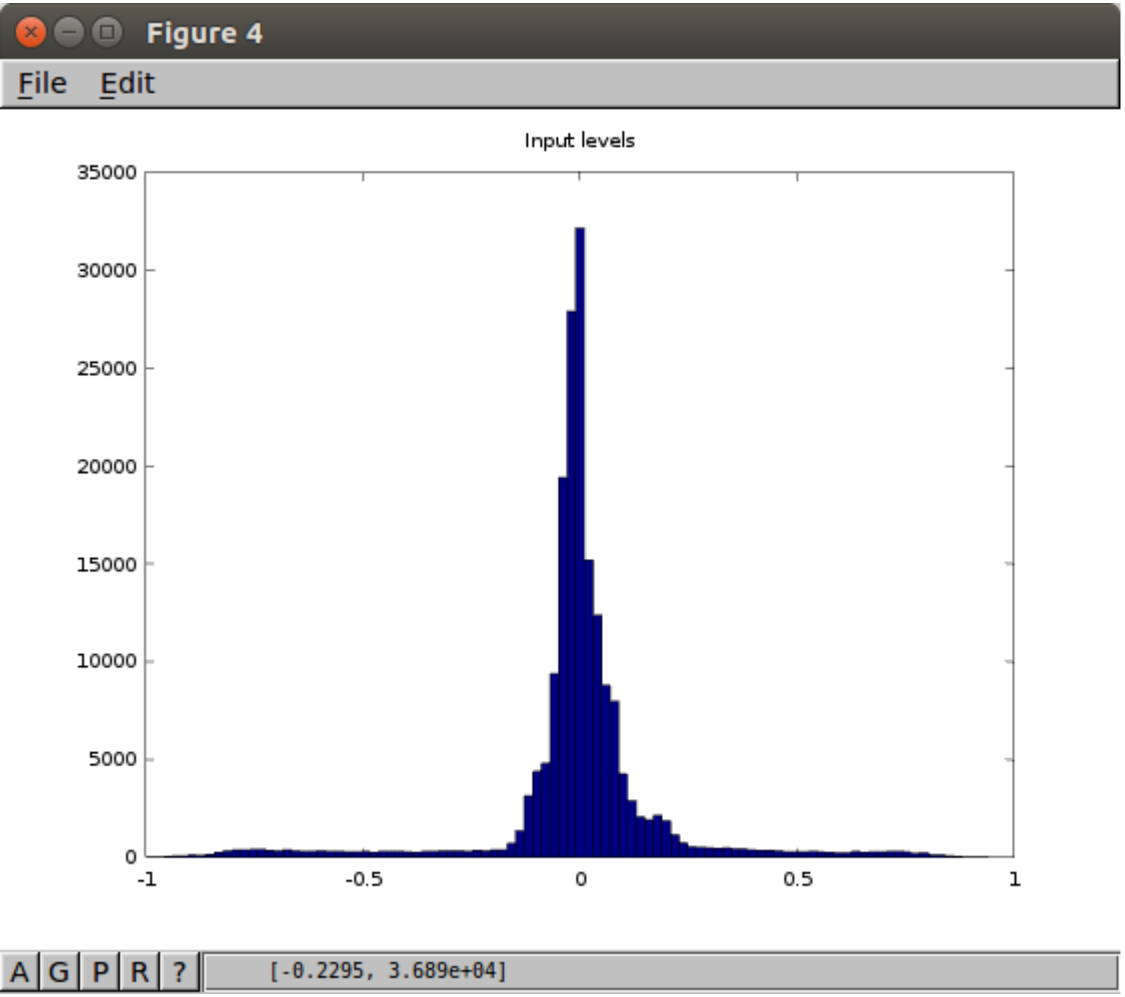
Blue: normalized input, Red, compressed

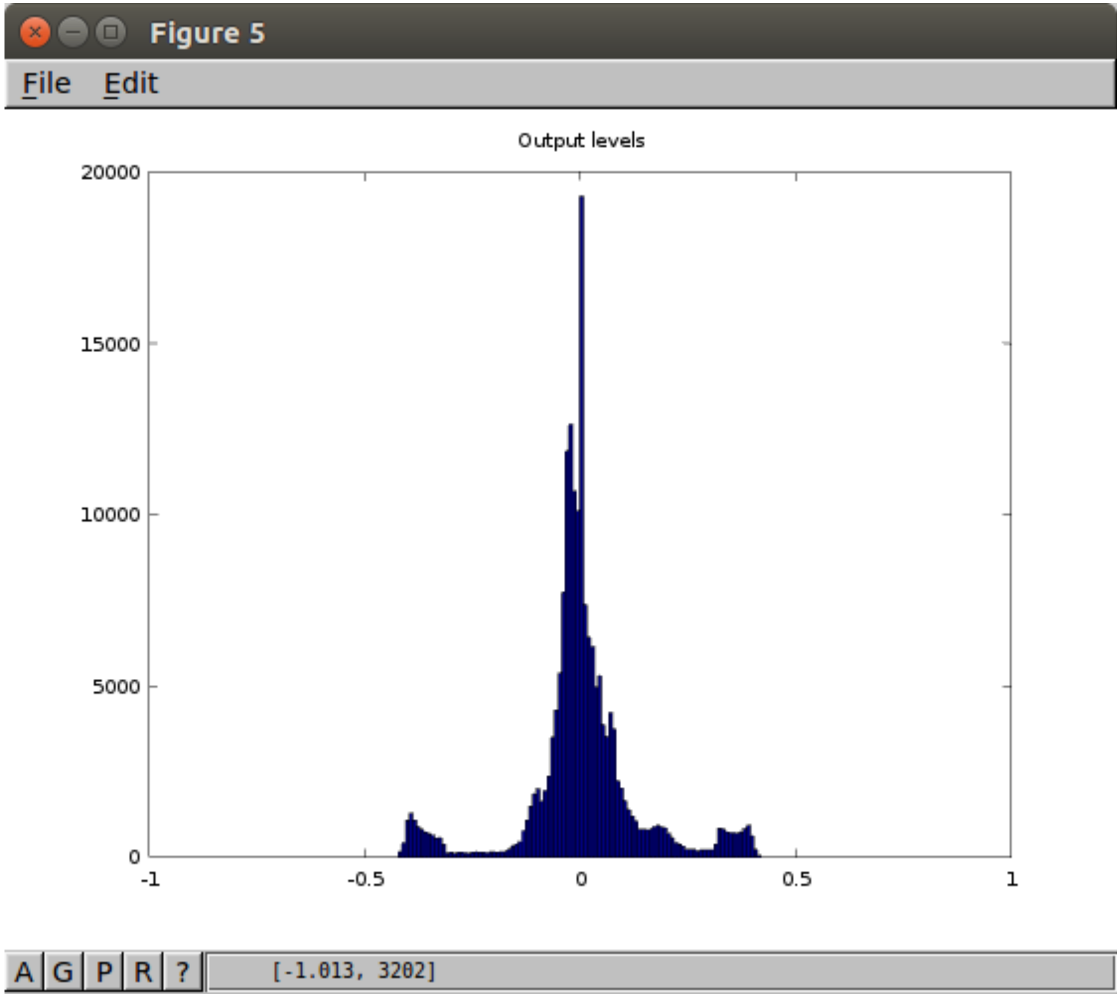


Blow up showing compression.
Note: below threshold, output=input

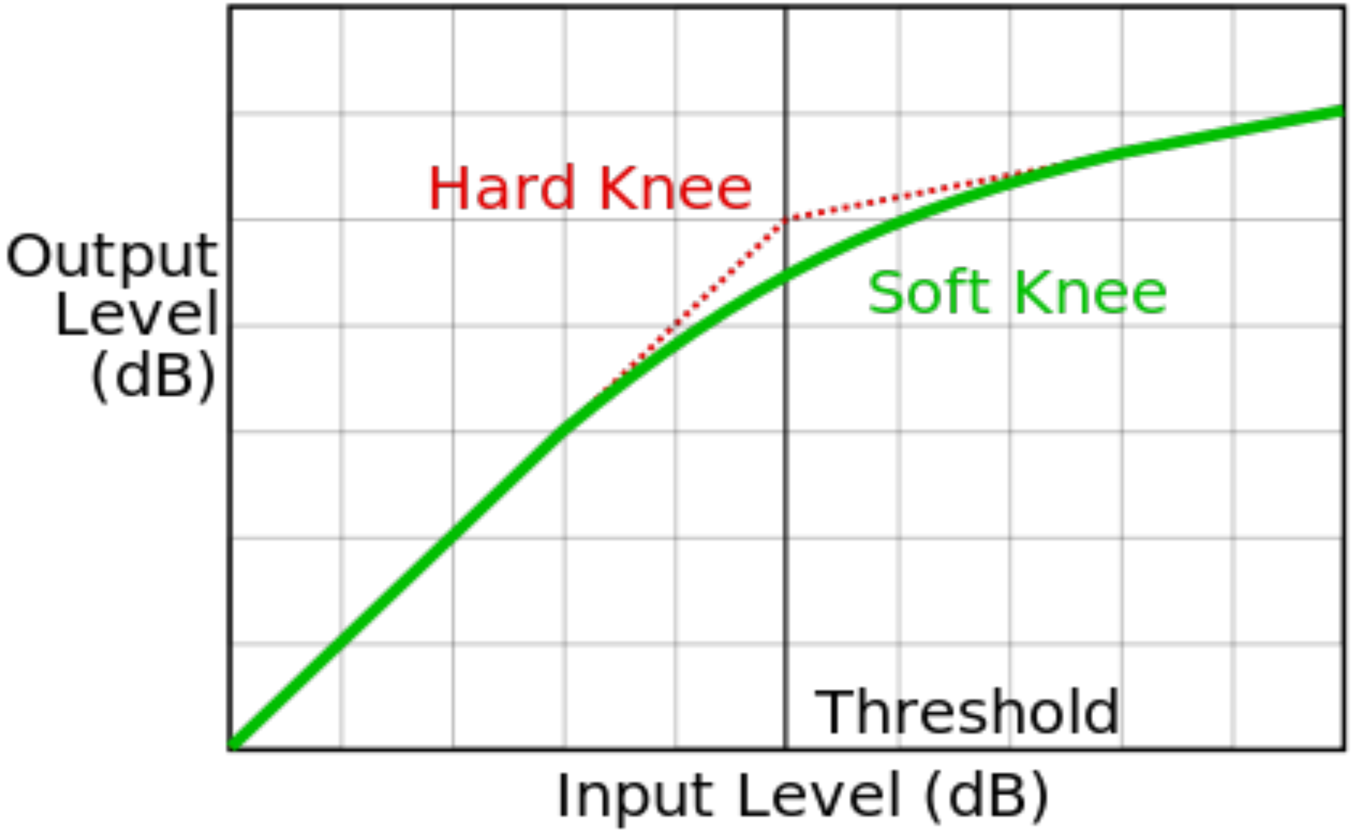


Histogram of audio samples, left uncompressed signal, right, compressed signal.
Note: compression pushes all samples from the extremes to the middle.
These often show up as extra peaks, at the extremes of the compressed file.

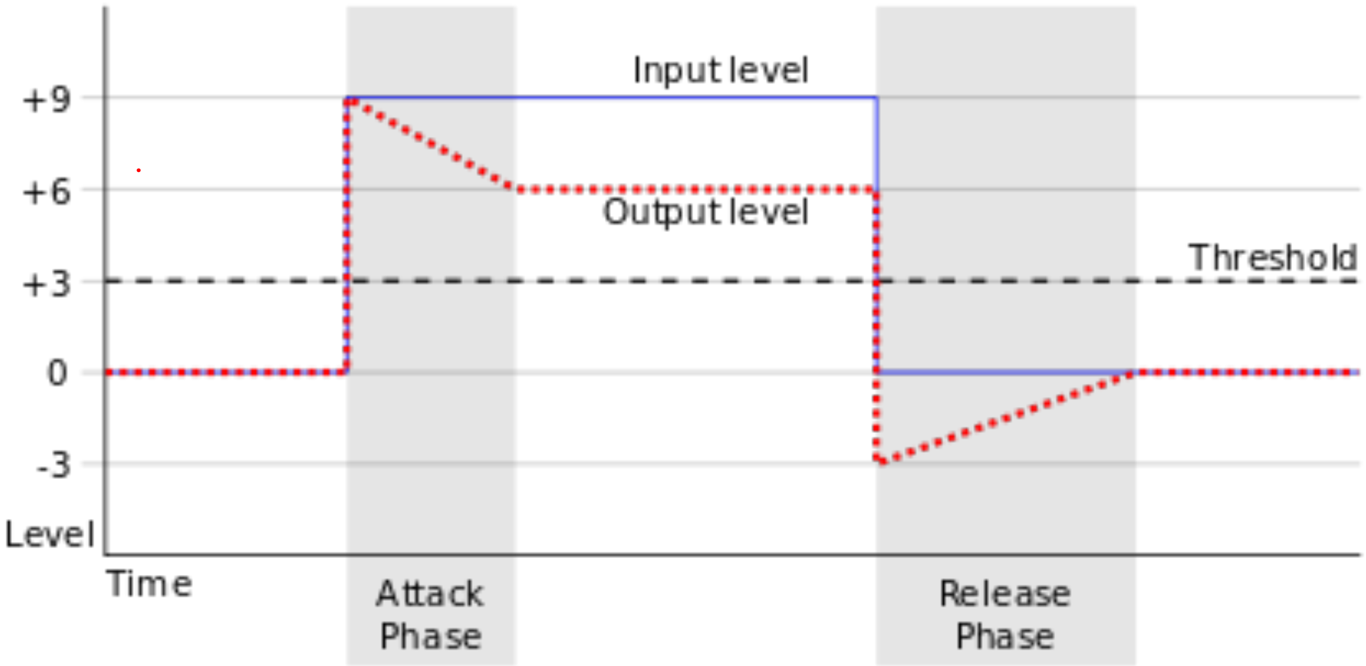




Compressor. "hard vs soft"
(https://en.wikipedia.org/wiki/Dynamic_range_compression)



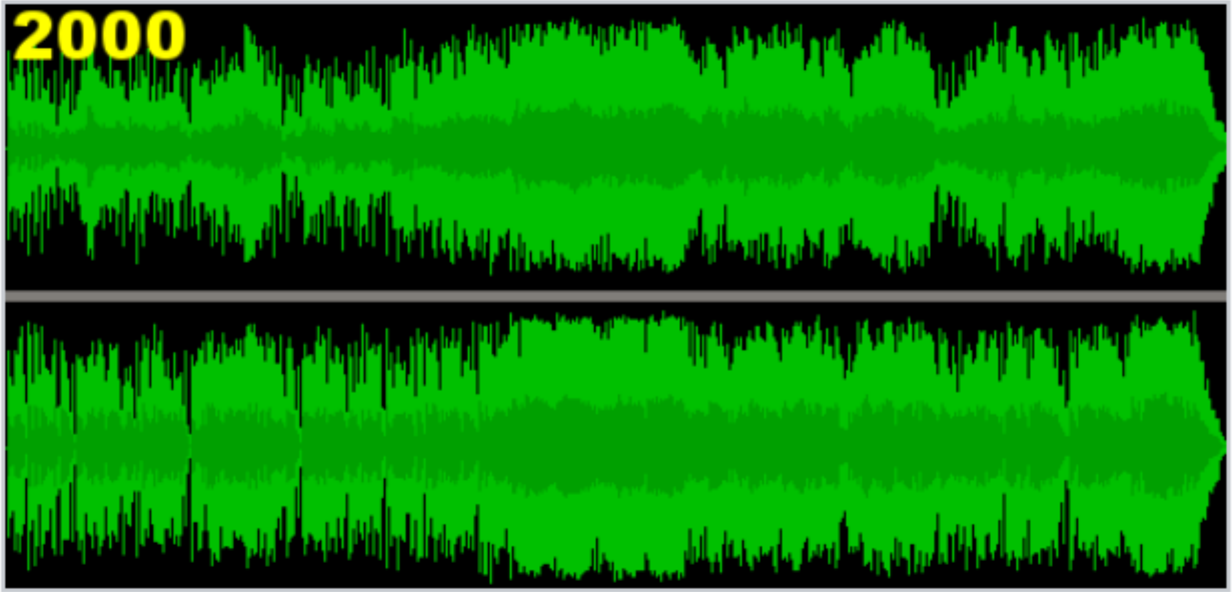
Compressor dynamics (attack, decay)
(https://en.wikipedia.org/wiki/Dynamic_range_compression)



n.wikipedia.org/wiki/Dynamic_range_compression

See also: Loudness war

Record



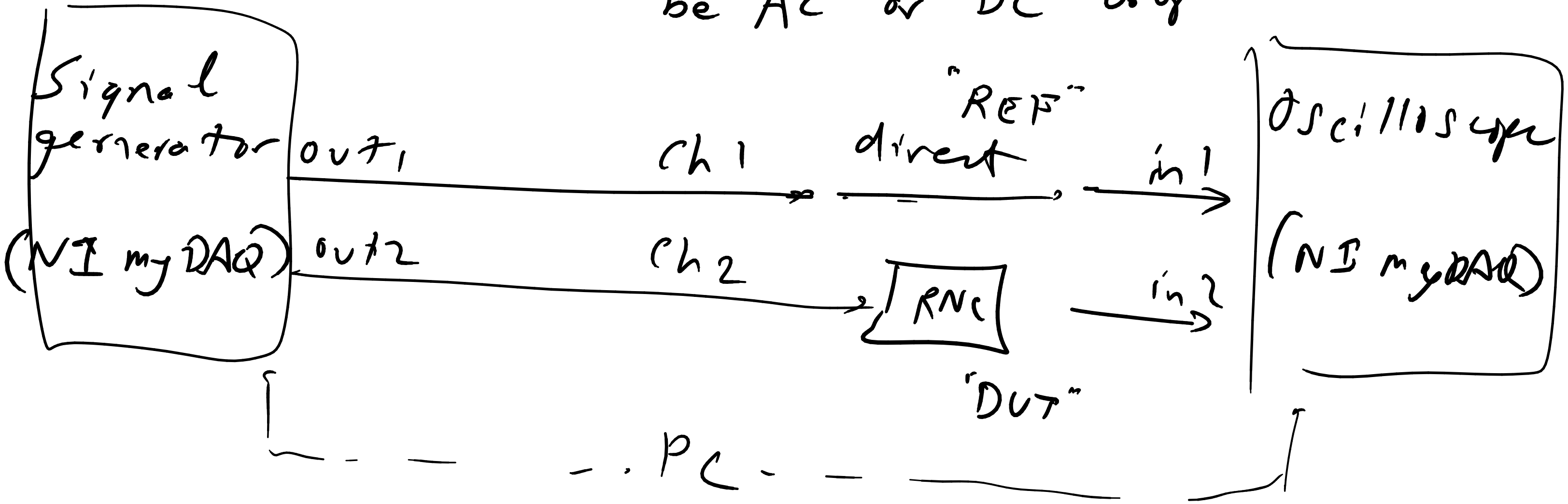
The trend of increasing loudness as shown by waveform images of the same song mastered on CD four times since 1983.

PROJECT IDEA: Review psychophysical theories of "Loudness". Apply some measures to music data files.

Compressor Demo:

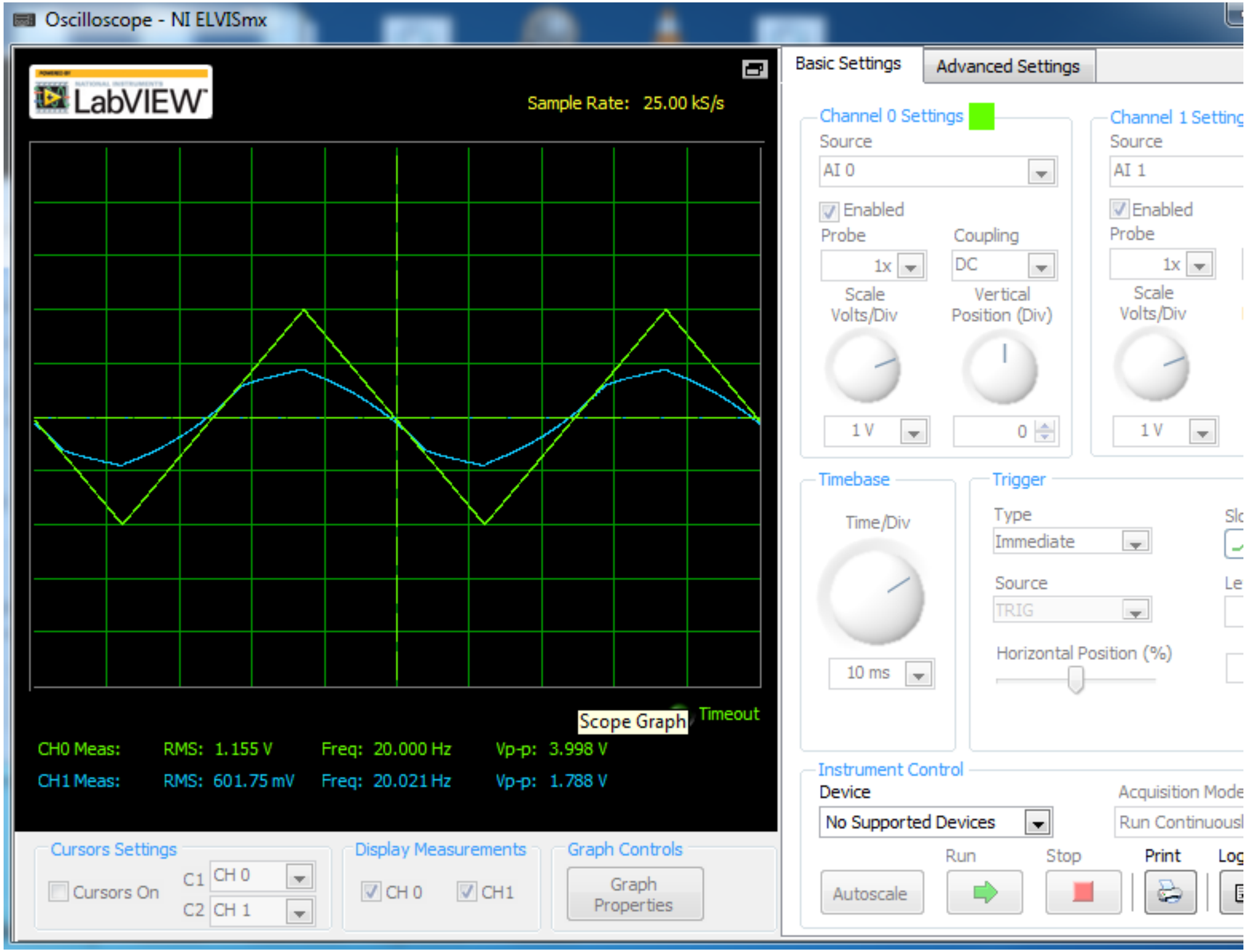
RNC (Really Nice Compressor) model RNC1773

Note: On myDAQ IO can be AC or DC coupled

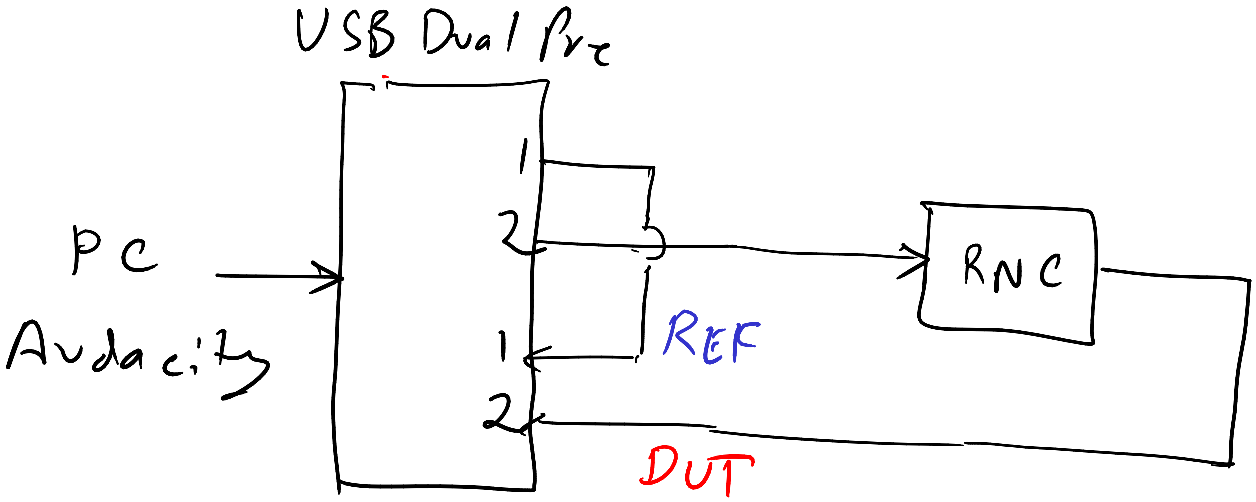


Test with: Sine wave, 233.8 Hz - Very level
Square wave ?

National Instruments myDAQ.
ELVIS "Virtual Instruments", Oscilloscope.
A Function Generator (not shown) applies a 20Hz Triangular wav.
Green signal is input to compressor,
Cyan signal is output of the compressor.
The threshold effect is clearly visible in the response.



Demo#2: Use Audacity to play a musical sample through the compressor.



REF: reference

DUT: "device under test"

Software: Audacity
- Create Stereo wav file (44100, 16bit)
- Generate drum and plucked sounds into this file (optionally loop samples)
- Record from Sound card. This will play the output and record simultaneously.
- Compare input (L channel) to output (R channel)

Note: There is a small delay, approx. 50 ms.
(check this on your system, it may be higher under Windows, for example)

PROJECT IDEA: Use this apparatus to test your own Guitar, effect or distortion pedals.

Compressor Algorithm (pseudo-code, not real Octave)

```
Let x[n] be input signal
    y[n] is output signal
    T threshold (decibels)
    R ratio (R>1)
    Gout = output gain

x[n] = x[n]/max(abs(x));          % normalize input (optional)
xdB[n] = 20*log10(abs(x[n]));     % convert levels of decibels.  Note abs()

if xdB[n] < T,
  ydB[n] = xdB[n]  % below threshold
else
  ydB[n] = T + (ydB[n] - T)/R     % above threshold
endif

ydB[n] = ydB[n] + Gout;  % add optional output gain

y[n] = sign(x[n]) * 10.^(ydB[n]/20);  % convert dB levels to amplitude
```

CS489/W17 Assignment 1.
Out: Thurs 19 Jan.
Due: Fri 27 Jan, 23:59.

Part 1. Measurement of ADC noise level.

The goal is to measure the "noise level" of your sound card input.

Measure the lowest possible input noise on your device.
To do this you should turn down the level to the minimum.
If you have a line input, leave it disconnected.
If you have a microphone input, try to get in the quietest place you can.

Note: Some sound cards will output exactly zero if the levels are turned down to their minimum level.
Make sure you have some signal, just the smallest value possible.

Record a small amount of audio, five seconds or less, and save it to a WAV file.
Input can be any of: a laptop microphone input, a laptop line level input, or an external (USB) sound card
You may wish to measure several devices and compare, but you should record at least one device.

Once you record, open the WAV file (Audacity or a similar program).
If the file is Stereo, split and save just one channel as a mono 16 bit wav file (sample at 44100 or 48000).
To see the noise, zoom in the WAV file on the vertical axis.
NOTE: Some sound cards may have a transient at the beginning.
If this is the case, prune off the beginning of the file and save the "steady state".

Next, you should estimate the mean and variance of the signal using the following Octave commands.
If input is a vector y, mean(y) gives the mean and std(y) gives the Standard deviation.

Fill in the following values below:

File name: _____ (name of the WAV file, to be submitted with your work)
File length (seconds): ___

Mean value: ___ (this should be close to zero, but may not be exactly zero)
Std deviation: ___
Relative noise level: ____ (this is \sigma divided q. q is the quantization size, q=2^-15 ~= 30.5 e-6)

Part 2. Simple compressor.

Implement a two simple compressors with the following characteristic:

2A. T=-10dB, R=6.

2B. T=-20dB, R=20

Given the sample data files,
Trumpet.wav
Oboe.wav.

In both cases the input files should be normalized (peak value set to -1 or +1) both before and after compression is applied.
No output gain is appied (Gout = 0 dB).

Produce four processed WAV files,
Trumpet_CompA.wav
Trumpet_CompB.wav
Oboe_CompA.wav
Oboe_CompB.wav