



University of Tabriz, Faculty of Electrical and Computer Engineering
Department of Biomedical Engineering

Title

**Digital Feedforward Compressor Design and Implementation
(using Matlab Audio System Toolbox)**

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1. Theory

Dynamic range control is the adaptive adjustment of the dynamic range of a signal. The dynamic range of a signal is the logarithmic ratio of maximum to minimum signal amplitude specified in dB.

You can use dynamic range control to:

- Match an audio signal level to its environment
- Protect AD converters from overload
- Optimize information
- Suppress low-level noise

Types of dynamic range control include:

- Dynamic range compressor — Attenuates the volume of loud sounds that cross a given threshold. They are often used in recording systems to protect hardware and to increase overall loudness.
- Dynamic range limiter — A type of compressor that brickwalls sound above a given threshold.
- Dynamic range expander — Attenuates the volume of quiet sounds below a given threshold. They are often used to make quiet sounds even quieter.
- Noise gate — A type of expander that brickwalls sound below a given threshold.

This tutorial shows how to implement dynamic range control systems using the compressor, expander, limiter, and noiseGate System objects from Audio Toolbox™. The tutorial also provides an illustrated example of dynamic range limiting at various stages of a dynamic range limiting system.

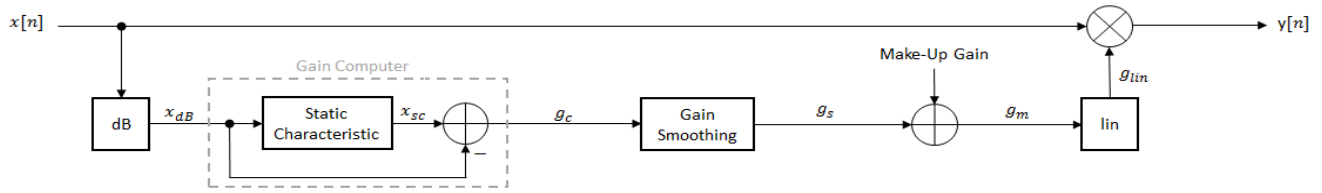


Figure 1: The diagram depicts a general dynamic range control system.

In a dynamic range control system, a gain signal is calculated in a sidechain and then applied to the input audio signal. The sidechain consists of:

- Linear to dB conversion: $x \rightarrow x_{dB}$
- Gain computation, by passing the dB signal through a static characteristic equation, and then taking the difference: $g_c = x_{sc} - x_{dB}$
- Gain smoothing over time: $g_c \rightarrow g_s$
- Addition of make-up gain (for compressors and limiters only): $g_s \rightarrow g_m$
- dB to linear conversion: $g_m \rightarrow g_{lin}$
- Application of the calculated gain signal to the original audio signal: $y = g_{lin} \times x$

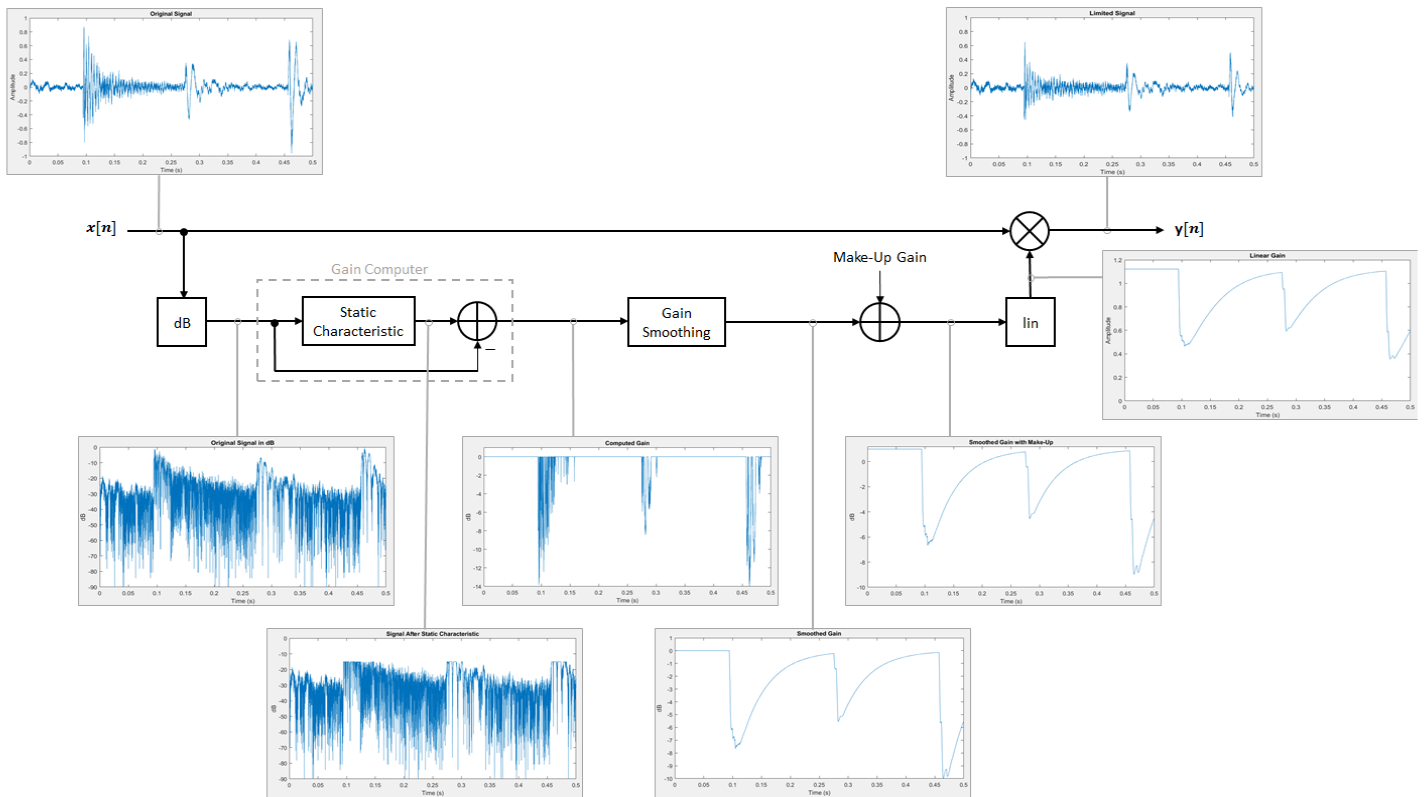


Figure 2: The diagram depicts a detailed block diagram of dynamic range control system.

✓ More detailed information can be found in:

<https://www.mathworks.com/help/audio/ug/dynamic-range-control.html>

2. The Structure of Programs

In this project Matlab audio system toolbox compressor function was used.

Main Function:

	Name	Type	Size
B.Sc. Simulations			
Baseline Methods			
Implementing Different Parts of DRC			
Using Matlab Audio System Toolbox			
Documentation			
Matlab Files			
References			
Databases			
	Results	File folder	
	Template	File folder	
	ConfigPath2.m	M File	2 KB
	DRCUsingAudioSystemTB_D.Giannoulis2012.mlx	MLX File	145 KB

- **DRCUsingAudioSystemTB_D.Giannoulis2012.mlx**

Description:

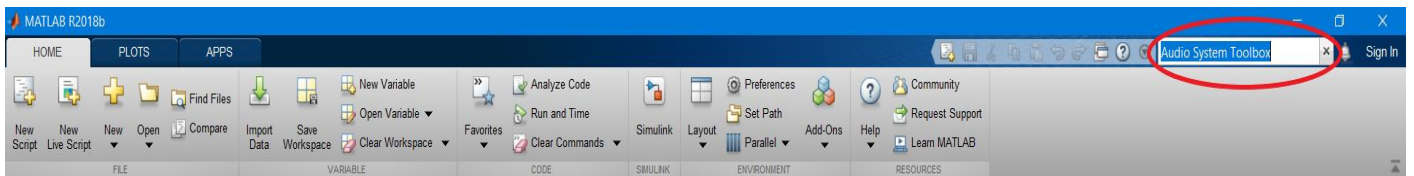
In this live script Matlab file, the compressor function from audio system toolbox was used to compress the input signal. It has three stages:

- At first the dsp.AudioFileReader and audioDeviceWriter System objects™ were set up to read the information of input signal.
- Then the ‘compressor’ function (from audio system toolbox) was called and compressor parameters were set up to have a threshold of -15 dB, a ratio of 7, and a knee width of 5 dB.
- At the end the processed audio will be visualized on the scope and played.

Matlab help was used to create this live script. Stages are described as:

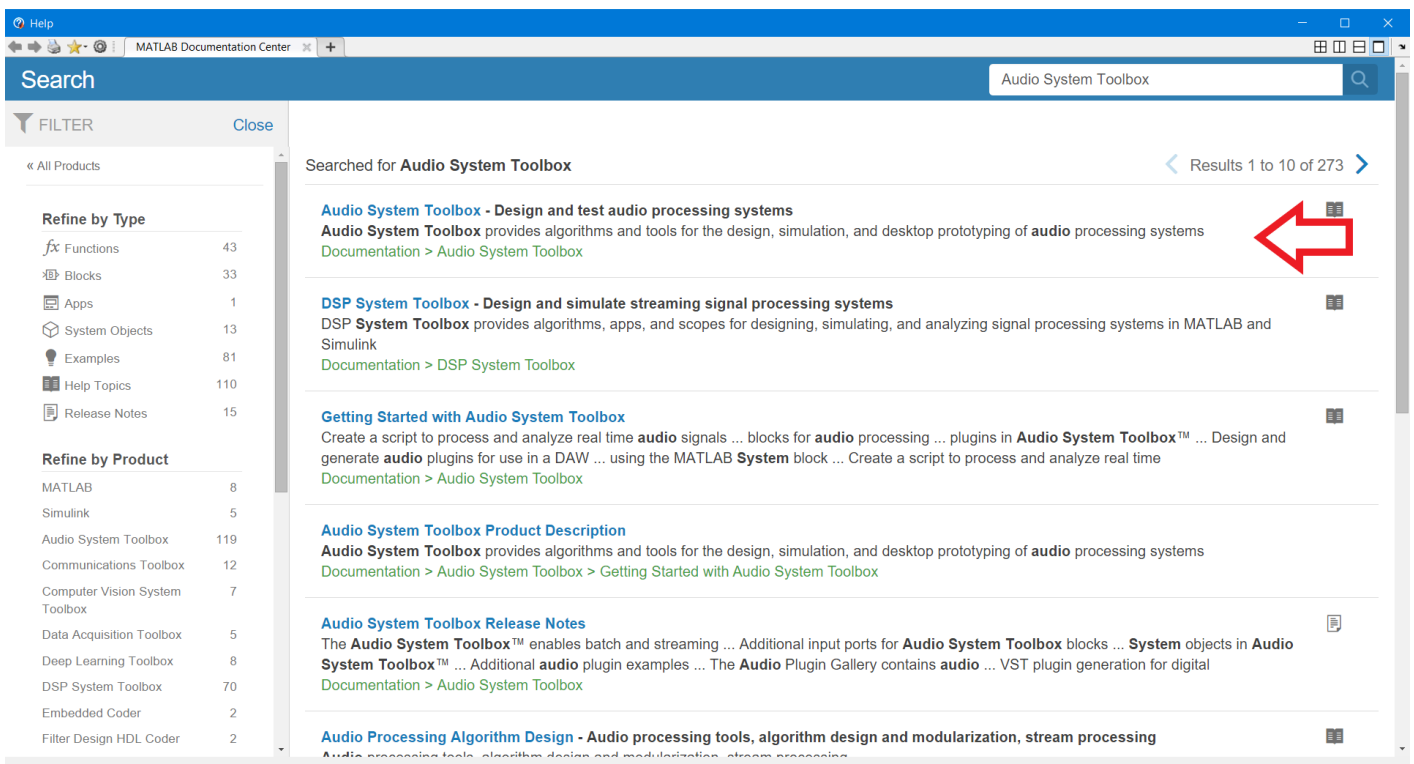
2.1 Stage 1: Search for 'Audio System Toolbox' in help

Search for 'Audio System Toolbox' in help as shown below.



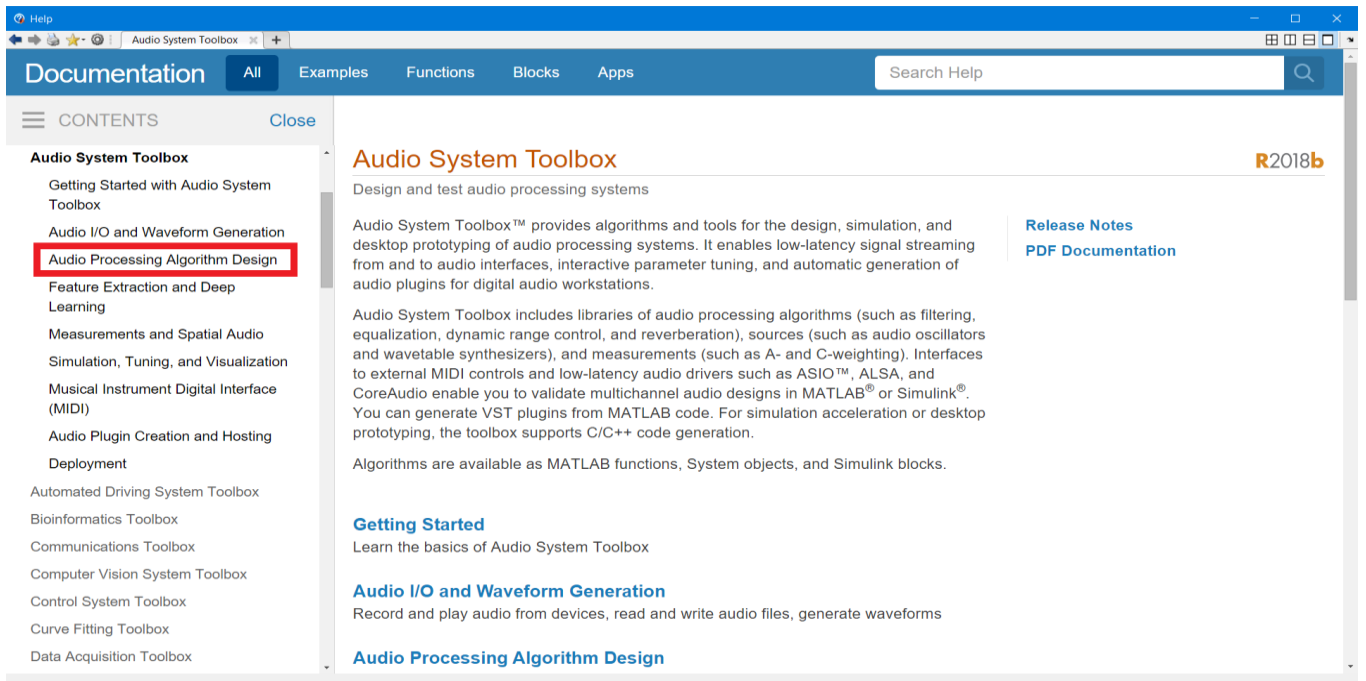
2.2 Stage 2: Click on 'Audio System Toolbox - Design and test audio processing systems'

From the results click on 'Audio System Toolbox - Design and test audio processing systems' as shown below.



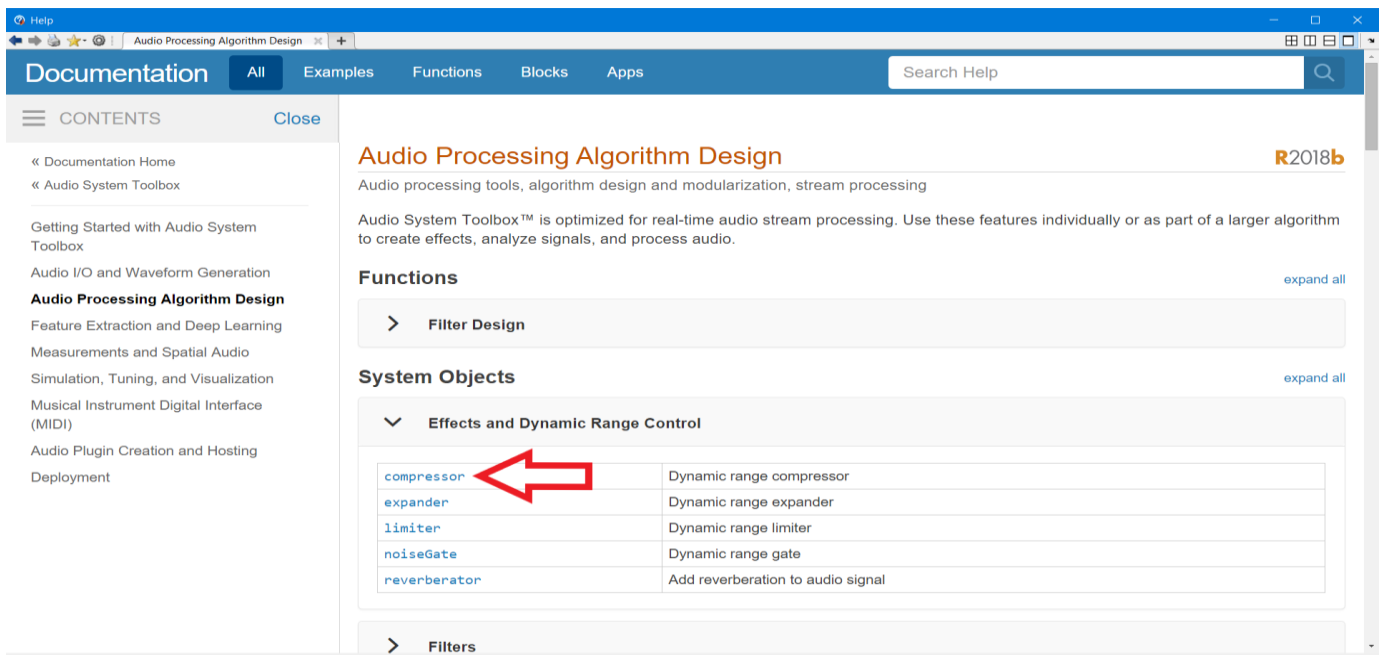
2.3 Stage 3: click on ‘Audio Processing Algorithm Design’

From the contents of Audio System Toolbox click on ‘Audio Processing Algorithm Design’.



2.4 Stage 4: click on ‘Compressor’

From ‘System Objects/Effects and Dynamic Range Control’ click on ‘Compressor’.



2.5 Stage 5: click on 'Examples'

Click on 'Examples' from contents.

The screenshot shows the MATLAB documentation page for the **compressor System object**. The left sidebar contains a 'CONTENTS' menu with 'Examples' highlighted. The main content area includes a description of the compressor and a block diagram. The diagram shows an 'Original Signal' entering a summing junction, where it is added to a 'Gain' signal. The output is a compressed signal. A 'Static Characteristic' graph is shown, plotting Output (dB) against Input (dB). The graph shows a curve that starts at -50 dB input and -50 dB output, rises to a plateau at -25 dB output for inputs above -30 dB. A 'Gain Smoothing and Make-Up' block is shown below the graph.

2.6 Stage 6: Expand 'Compress Audio Signal' and click on 'Open Live Script'

After opening the example live script file, it was modified according to the needs.

The screenshot shows the MATLAB documentation page for the **compressor System object**, specifically the 'Examples' section. The 'Compress Audio Signal' example is expanded, showing a list of steps and code snippets. A red arrow points to the 'Open Live Script' button. The code snippets include setting up the audio file reader and writer, creating the compressor object, visualizing the static characteristic, and setting up a scope to visualize the original audio signal, the compressed audio signal, and the applied compressor gain.

```
frameLength = 1024;
fileReader = dsp.AudioFileReader( ...
    'Filename','RockDrums-44p1-stereo-11secs.mp3', ...
    'SamplesPerFrame',frameLength);
deviceWriter = audioDeviceWriter( ...
    'SampleRate',fileReader.SampleRate);

dRC = compressor(-15,7,...
    'KneeWidth',5,...
    'SampleRate',fileReader.SampleRate);

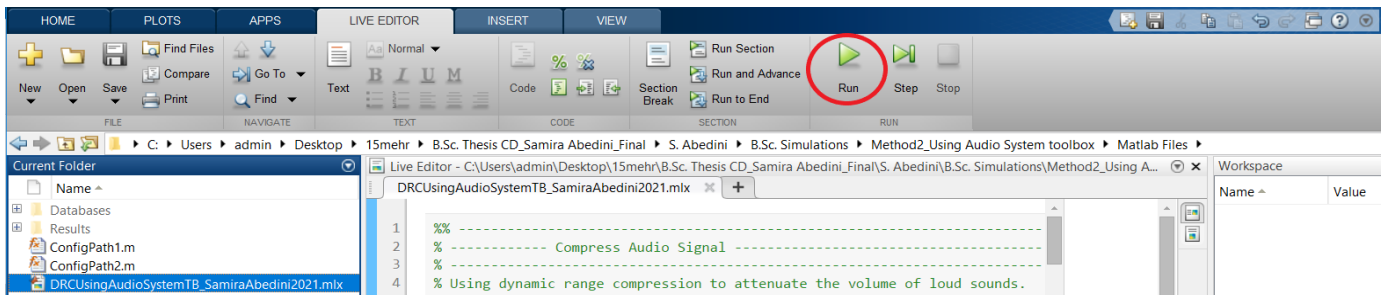
visualize(dRC)

scope = dsp.TimeScope( ...
    'SampleRate',fileReader.SampleRate, ...
    'TimeSpan',1
```


3. Simulations Results

3.1. Running the Programs

In order to run this program, just open the **DRCUsingAudioSystemTB_SamiraAbedini2021.mlx** live script file in Matlab Environment and click the Run button from live editor in Matlab toolstrip (as shown by red circle in below).



Note:

The input signal , ‘noisy.wav’ file , is located in ‘**B.Sc. Simulations\Method2_Using Audio System toolbox\Matlab Files\Databases**’ File.

The output signal , ‘CompressedSignal2.wav’ file , will be saved in ‘**B.Sc. Simulations\Method2_Using Audio System toolbox\Matlab Files\Results\Compressed Signal**’ file.

3.2. Experimental Results

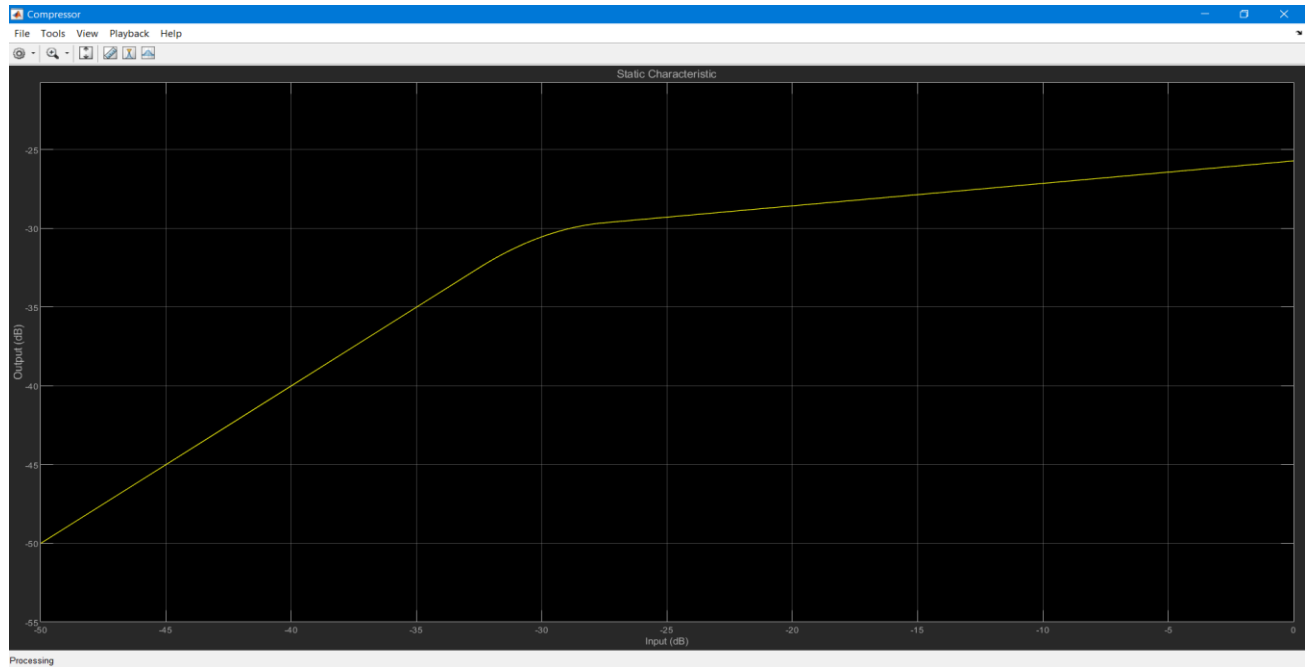


Figure 4: output (static characteristic)

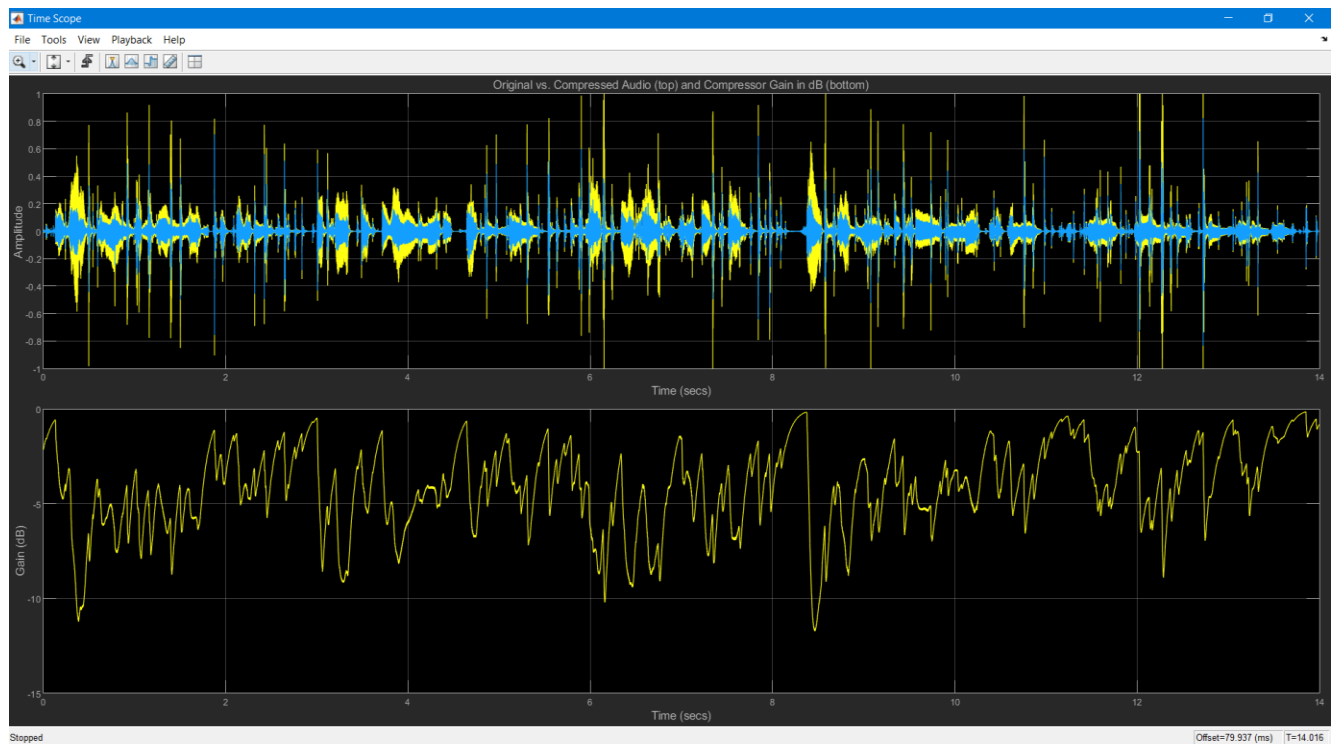


Figure 5: compressed signal and gain

References:

- [1] P. Dutilleux, et al., “Nonlinear Processing, Chap. 4,” in Dafx:Digital Audio Effects, U. Zoelzer, Ed. (2nd ed: Wiley, John & Sons, 2011), p. 554.
- [2] J. O. Smith, Introduction to Digital Filters with Audio Applications (Booksurge Llc, 2007)....
- [3] Giannoulis, Dimitrios & Massberg, Michael & Reiss, Joshua. (2012). Digital Dynamic Range Compressor Design—A Tutorial and Analysis. AES: Journal of the Audio Engineering Society. 60.
- [4] <https://www.mathworks.com/help/audio/ref/compressor-system-object.html>