A

Project Based Lab Report

On

AUDIO COMPRESSION

USING FILTERS

Submitted in partial fulfilment of the
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DEPARTMENT

OF

ELECTRONICS AND COMMUNICATION ENGINEERING

(DEEMED TO BE UNIVERSITY)

CERTIFICATE

This is to certify that this project based lab report entitled “**AUDIO COMPRESSION USING FILTERS**” is a bona fide work done by **K.A.S.K.Bhargav (160040408), P.Jafarkhan (160040686), S.N.Sairam(160040800)** in partial fulfilment of the requirement for the award of degree in Bachelor of Technology in Electronics and Communication Engineering during the academic year 2018-2019.

We also declare that this project based lab report is of our own effort and it has not been submitted to any other university for the award of any degree.

Signature of the Project Guide **Head of Dept., ECE**

ACKNOWLEDGMENT

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ABSTRACT

A matlab-based implementation of an audio compressor was investigated. By varying the input parameters of the compressor, results were obtained which demonstrated the typical behavior and benefits of compressor use. The parameters however, did not correspond to those found on typical hardware or software compressors. Adjustment to the matlab code could be made to make the controls more understandable to audio practitioners.

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INTRODUCTION

DESCRIPTION:

In the evolution of communications systems, compression has been essential because it allows users to share the limited capacity of communications channels and storage spaces. Various types of compression coding for speech and audio have been developed, and these have found important applications in cellular phone systems, music delivery over networks, and portable players. However, most of the speech coding and audio coding standards in ISO/IEC MPEG* (International Organization for Standardization and International Electro Technical Commission Moving Picture Experts Group), such as MP3 (MPEG-1 audio layer 3) and AAC (advanced audio coder), achieve a high compression ratio at the sacrifice of minor waveform distortion and band limitation at the decoder.

In the near future, we can expect to enjoy a broadband network at a reasonable price, so greater bandwidth will be available. If we can make use of this rich information environment, we should be able to enhance perceptual quality dramatically. At present, we face the challenge of shifting from the need for efficient compression to the desire for excellent quality. To meet this challenge, we must find ways to exploit the rich bandwidth for higher quality, greater convenience, and more comfort in communications.

Audio compression is the process of reducing a signal's dynamic range. Dynamic range is the difference between the loudest and quietest parts of an audio signal. We need to reduce the dynamic range of most signals for them to sound natural on a recording. This is done by boosting the quieter signals and attenuating the louder signals.

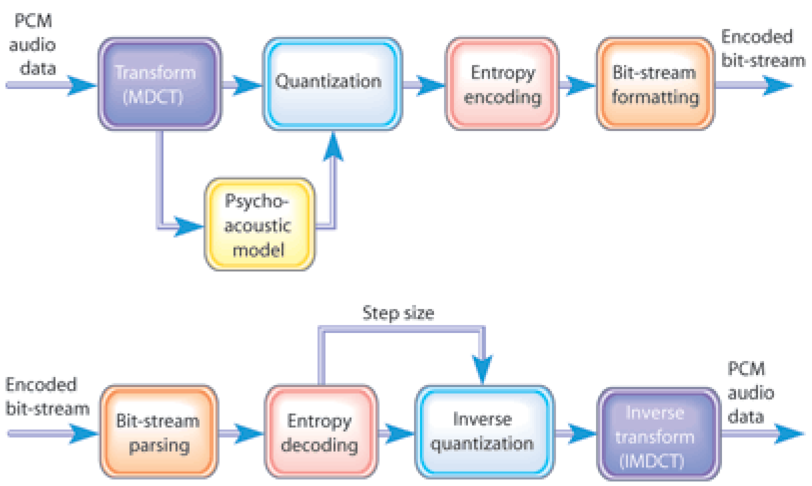
A compressor reduces the level of an audio signal if its amplitude exceeds a certain threshold. Threshold is commonly set in decibels dB, where a lower threshold (e.g. -60 dB) means a larger portion of the signal is treated.

The dynamic range is reduced, yet the apparent loudness of a signal is increased. Essentially, the loud bits are softened and the quiet bits are increased in level. Example uses may include; compressing a vocal, allowing it to sit on top of a mix; or compressing a mix to obtain the maximum loudness without overloading a transmission channel. By varying a compressor's parameters, its behavior can be changed to suit the application. A typical compressor has controls for the THRESHOLD above which sounds begin to be compressed, the RATIO of output to input above this threshold, and GAIN to be added to the compressed signal. Advanced controls include ATTACK and RELEASE. That is, how soon after a signal exceeds threshold should the compressor act, and how long after the signal drops below threshold until the compressor releases. These two controls can be used to affect the tone of a signal. E.g. a compressor on a snare drum may be set with fast attack, which would compress the transient, or a slow attack, allowing the transient to pass uncompressed

TECHNICAL DESCRIPTION

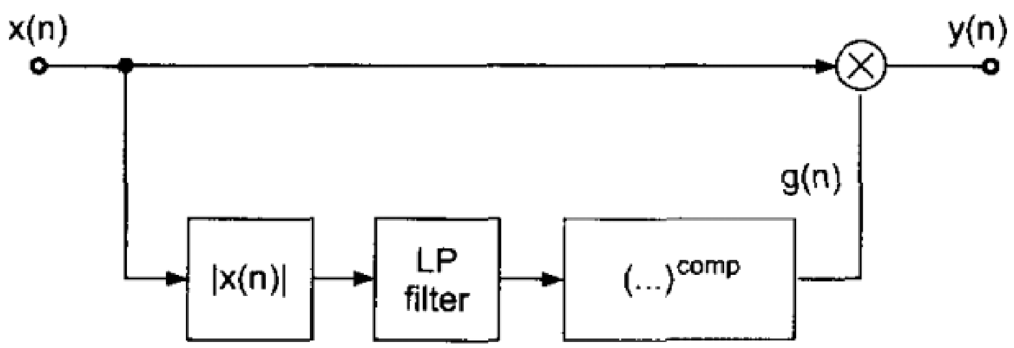
Basic Audio Compression:

First, the audio data is applied any transform like the DCT and is modelled by a psychoacoustic model and then quantized. These bits are entropy encoded and then formatted to generate an encoded bit stream. The quantization indices are entropy coded. Entropy coding is a lossless technique that uses fewer bits to represent more likely quantization indices. Huffman coding is the most commonly used technique in entropy coding. Finally, the resulting data is formatted based on the standard specifications and packetized for storage or streaming.



It's easy to see that the decoder can follow a precise reverse process to reconstruct the audio signal.

The basic block diagram of most standard audio codecs mimic closely the typical structure described above. However, each standard may add some extra blocks (also called tools) to support new features or improve compression performance.



Here, in our compressor, the function takes as its input an audio signal and two parameters; a compression parameter and a filter parameter. The function may operate equally as an expander (expanding the dynamic range of the signal) by choosing the compression parameter 0<comp<1. This functionality was verified but not investigated in depth. When -

1<comp<0, the function acts as a compressor, where more negative values of comp result in a greater degree of compression.

The function works by low-pass filtering the absolute value of the input signal, as a means of detecting the signals envelope. That is, it finds the gross variations in amplitude of the signal, ignoring the variations due to the oscillating nature of sound. The filter takes a parameter 0>a>1. Where larger values cause the filter to react more slowly to changes, in effect a slower attack and release time.

The compression is applied to this filtered signal, which when combined with the original signal, results in a compressed output. See equation (1) and (2).

$$g(n) = h(n)^{Comp} \qquad (1)$$

$$y(n) = x(n) * g(n) \qquad (2)$$

Where: x(n) = original signal

y(n) = compressed output signal

h(n) = envelope filtered original signal

g(n) = gain reduction to be applied

comp = compression parameter

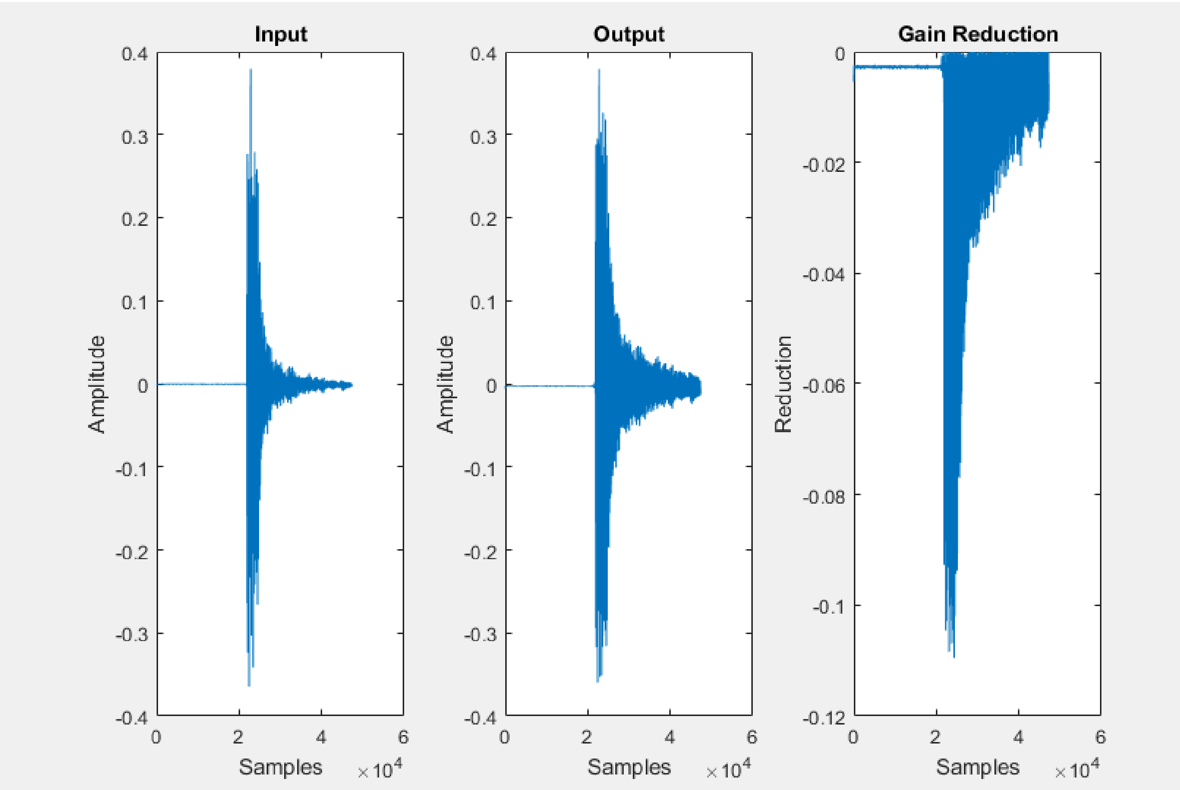
Finally, the signal is normalized, such that the maximum value of the new signal corresponds to that of the original signal. This is a simplification of the makeup gain as applied in a typical compressor.

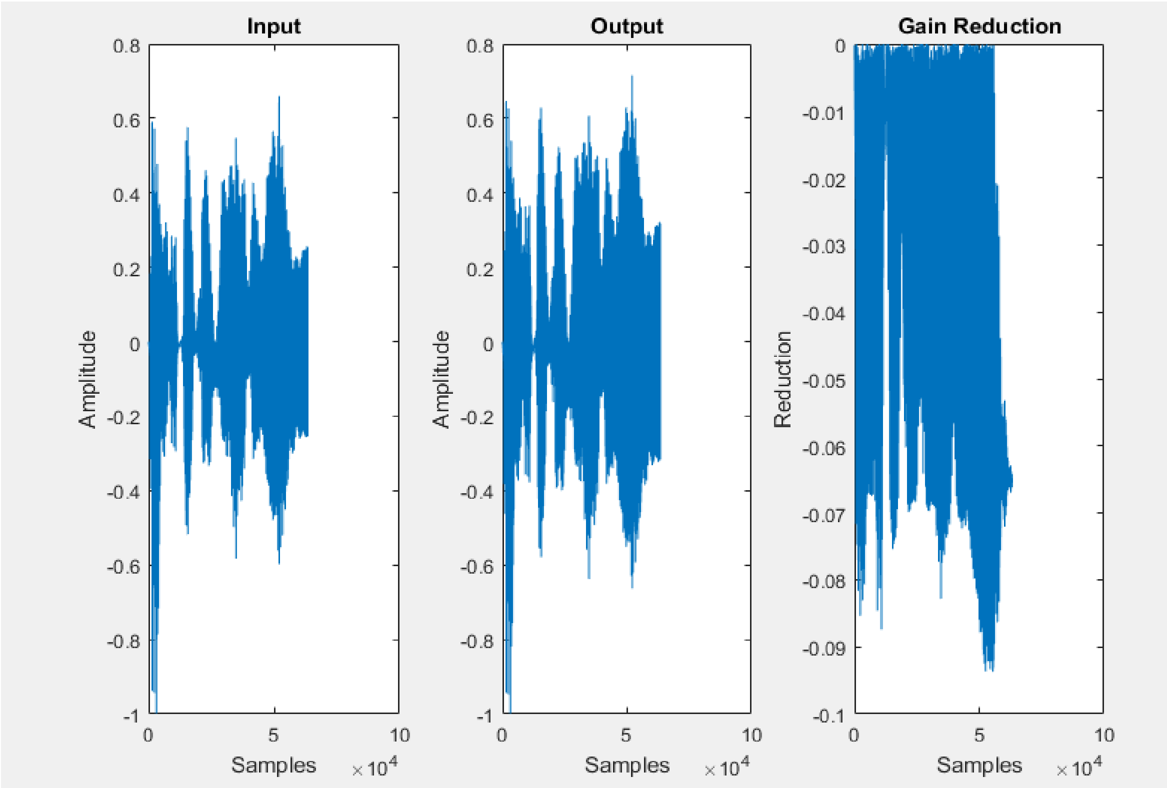
SOURCE CODE

```
[input,fs] = audioread('Snare1.wav'); %wave file input

%set parameters.
% comp    - compression: 0>comp>-1, expansion: 0<comp<1
% a - filter parameter: <1, related to attack time, how quickly the
filter
% responds to changes in input. Smaller numbers indicate faster
response.
comp = -0.3;
a = 0.6;

output = compexp(input, comp, a); %call compressor

% sound(input,fs); %play input and output sounds
% sound(output,fs);

sound(output)

audiowrite('CompressedSnare.wav',output, fs); %save output as wave
file

[input,fs] = audioread('anechoic_voice.wav'); %wave file input

%set parameters.
% comp    - compression: 0>comp>-1, expansion: 0<comp<1
% a       - filter parameter: <1, related to attack time, how
quickly the filter
% responds to changes in input. Smaller numbers indicate faster
response.
comp = -0.17;
a = 0.25;

output = compexp(input, comp, a); %call compressor

sound(input,fs); %play input and output sounds
sound(output,fs);

audiowrite( 'CompressedVoice.wav',output, fs); %save output as wave
file



function y=compexp(x,comp,a)

% Compressor/expander from DAFX, Zolzer
```

```
% comp  - compression: 0>comp>-1, expansion: 0<comp<1
% a         - filter parameter <1

h=filter([(1-a)^2],[1.0000 -2*a a^2],abs(x)); %envelope detector
filter
h=h/max(h); %normalise filter

h=h.^comp; %apply compression factor
y=x.*h; %apply compression curve to original signal

z=abs(x)-abs(y); % create gain reduction vector for display only

y=y*max(abs(x))/max(abs(y)); %normalise output signal to max of
input signal


figure ;
        subplot(1,3,1), plot(x),title('Input'),xlabel('Samples'),
ylabel('Amplitude');
        subplot(1,3,2), plot(y), title('Output'),xlabel('Samples'),
ylabel('Amplitude');
        subplot(1,3,3), plot(z), title('Gain
Reduction'),xlabel('Samples'), ylabel('Reduction');
```

SIMULATIONS & RESULTS





The compressor was applied to signals containing a single snare drum hit as well as a section of spoken words. With experimentation, suitable values of the input parameters were found which best demonstrate the expected functionality of a compressor. Figure 3 shows the input and output waveforms for the snare drum along with the corresponding gain reduction (pre-normalisation). As expected, the greatest amount of gain reduction is applied where the input signal amplitude is greatest. The effect this has on the output is to accentuate the decaying section of the waveform, which can be heard clearly as a more pronounced 'tail'.

The compressor as applied to a snare drum hit. The gain reduction is greatest when the input amplitude is greatest. The difference in the output between loud and soft is lessened, having the effect of making the quieter 'tail' appear louder.

The effect on the vocal is an increase to overall loudness. It was found that compression values <-0.2 caused the output to become distorted, an undesirable product for a compressor.

CONCLUSION AND FUTURE SCOPE

CONCLUSION:

A matlab function was used to simulate the behavior of an audio compressor. While the compressor lacked the control commonly found on traditional software or hardware compressors, it did demonstrate the effects of changing amount of compression and peak detection speed. Further work could be taken to improve the compressors functionality.

FUTURE SCOPE:

While this compressor function produces expected results, its main disadvantage is the lack of traditional controls such as threshold, ratio, attack and release. These controls can be included. Also, more emphasis has to be put on to change the way the compressor behaves on the go to achieve better results.

In spite of our ability to understand technological trends, it's difficult to predict market acceptance of codecs. Commercial success or failure depends not only on the technology but also on issues such as licensing terms, hardware compatibility, upgradeability, costs to new technology, support for digital rights management, and more.

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[4] https://en.wikipedia.org/wiki/Data_compression#Audio

[5] https://en.wikipedia.org/wiki/MP3

[6] https://en.wikipedia.org/wiki/Dynamic_range_compression