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SP Signal Processing

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Real-Time Automatic Gain Control for Singing Voice Applications

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Outline

1. Motivation
2. Algorithm
3. Optimization
4. Results
5. Future Implementations

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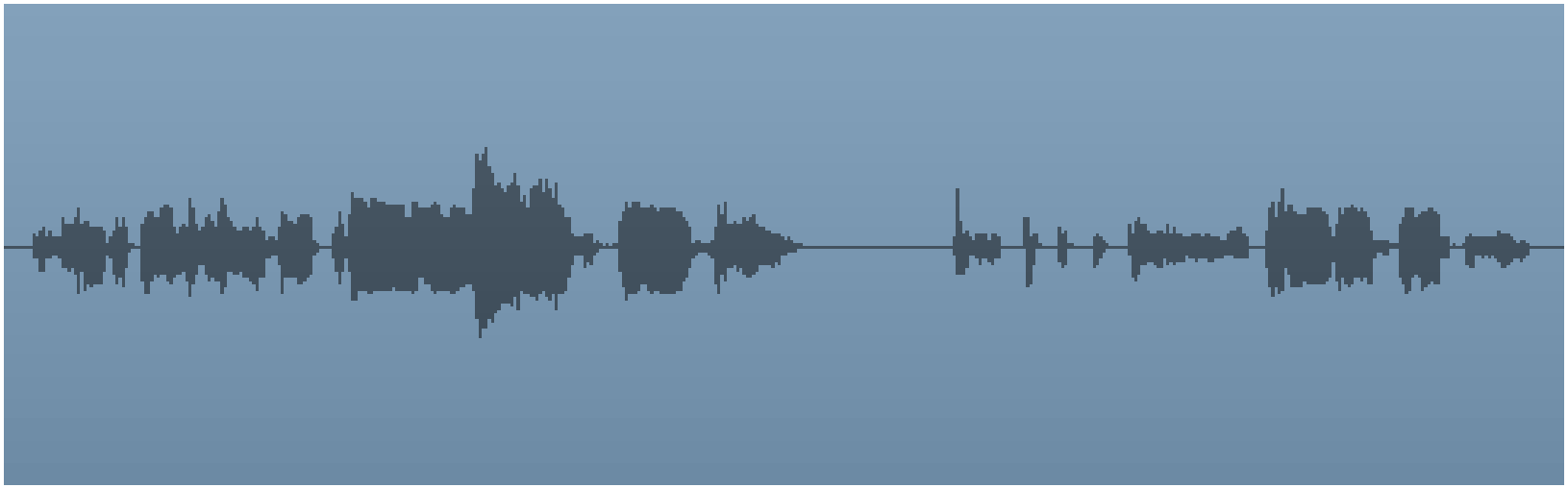
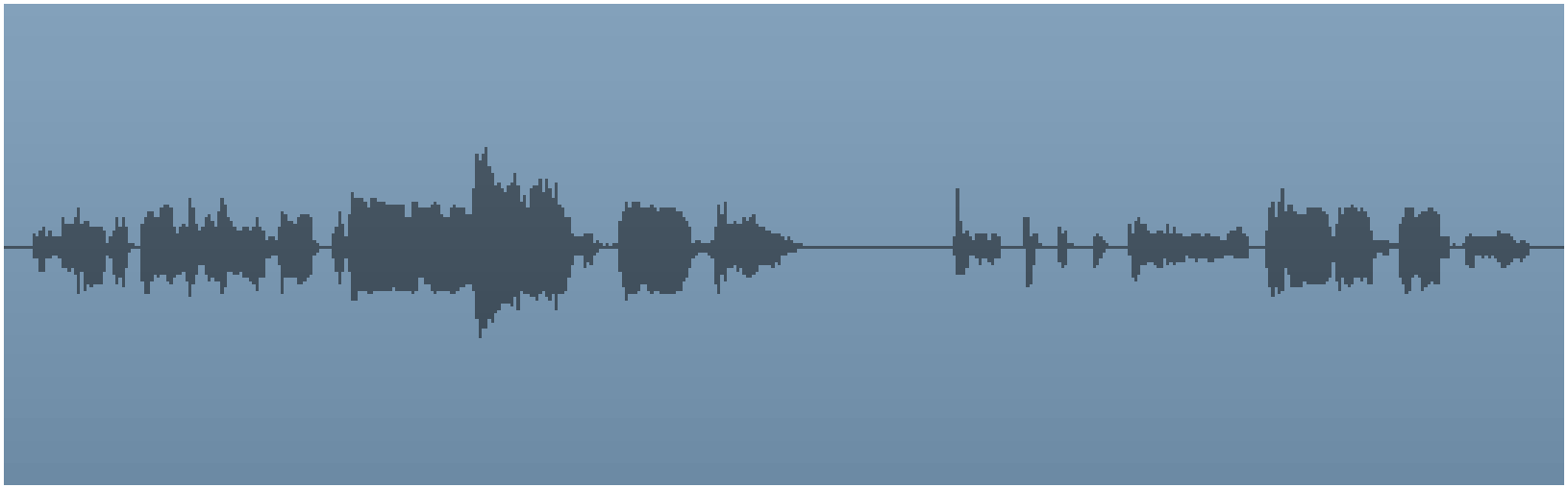
Motivation

Motivation

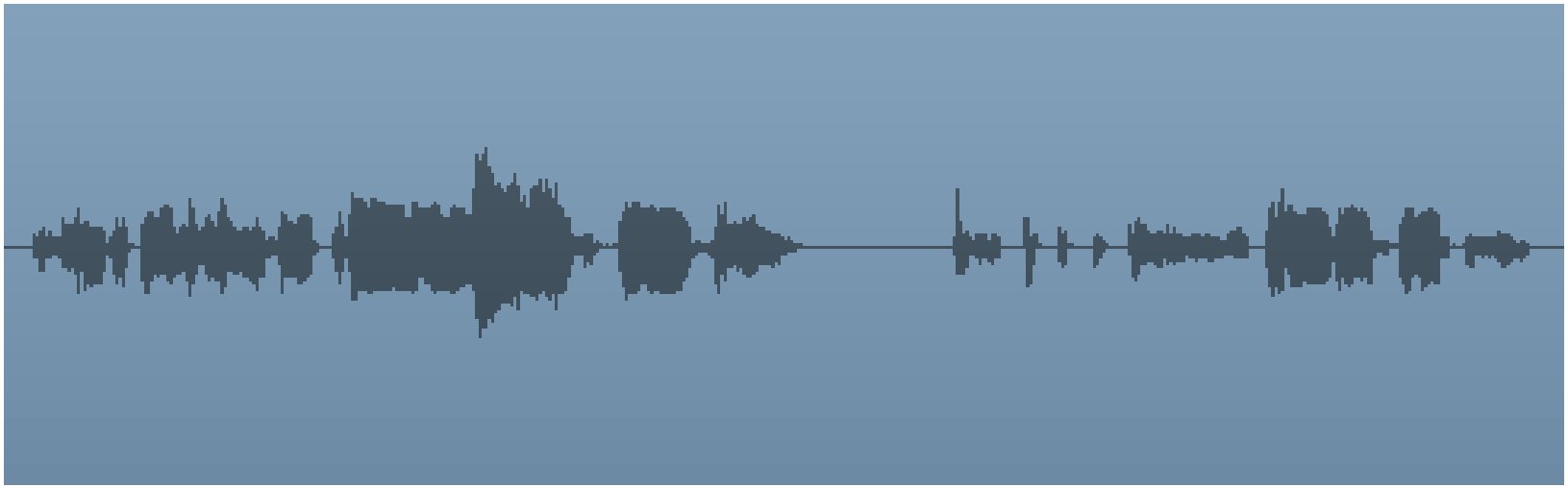


Motivation



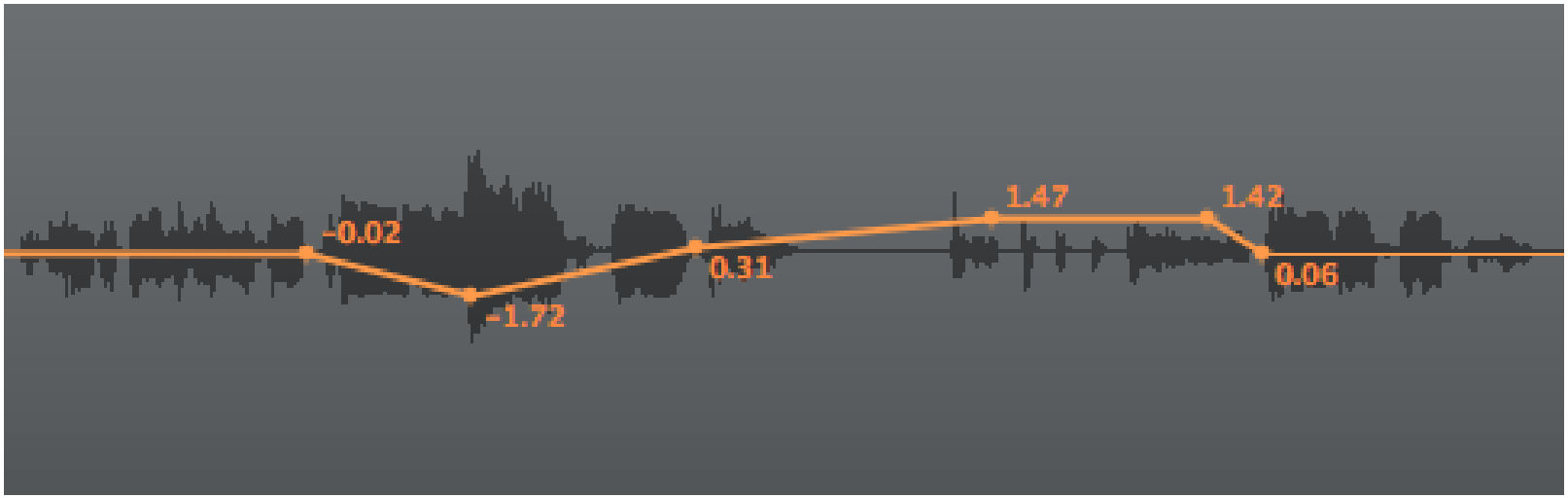
- compressor to fast

Motivation



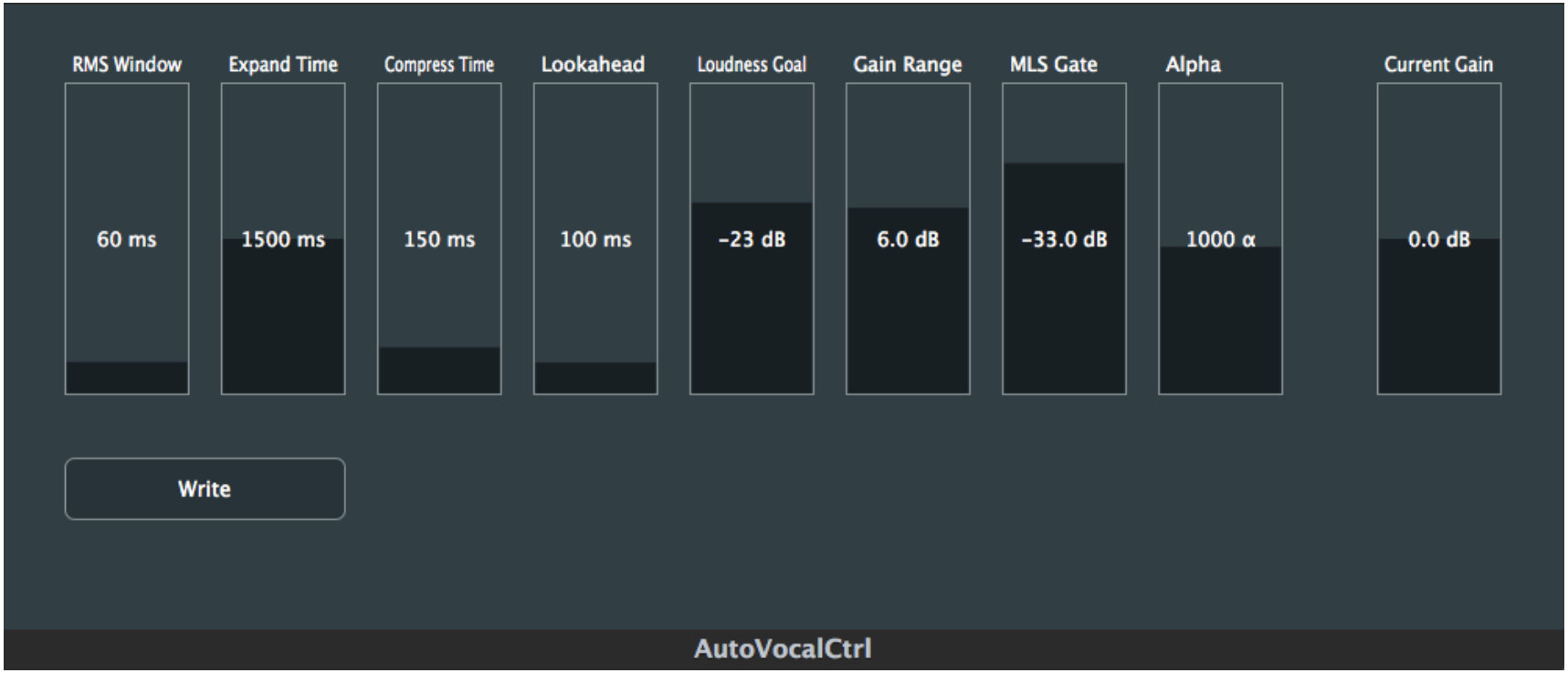
- compressor to fast
- factor in human perception

Motivation



- compressor to fast
- factor in human perception
- save time

Dummy UI


RMS Window
60 ms
Expand Time
1500 ms
Compress Time
150 ms
Lookahead
100 ms
Loudness Goal
-23 dB
Gain Range
6.0 dB
MLS Gate
-33.0 dB
Alpha
1000 α
Current Gain
0.0 dB
Write
AutoVocalCtrl

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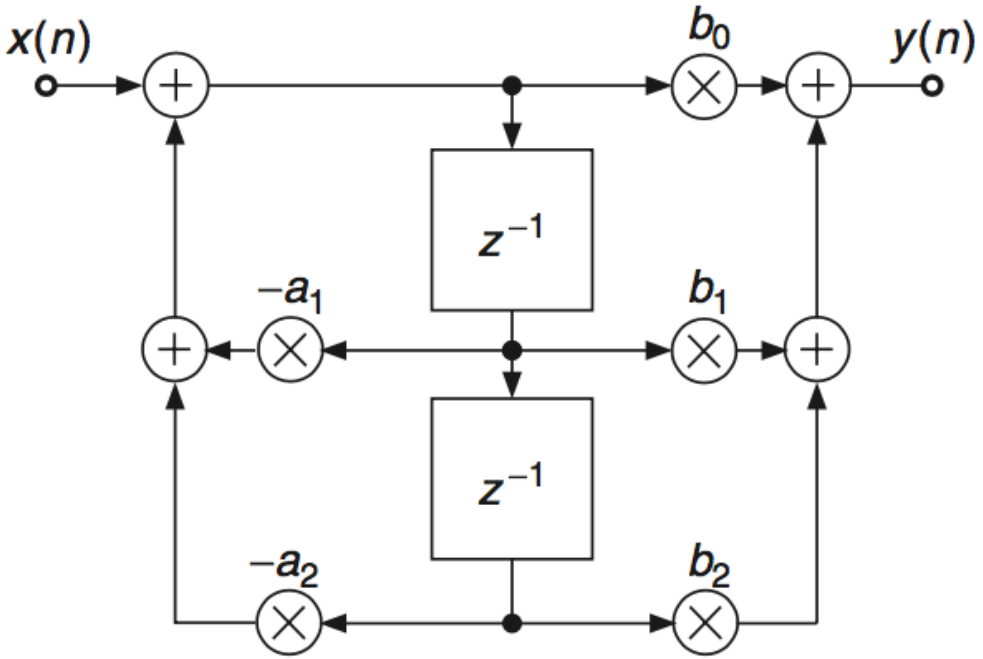
Algorithm

Algorithm

2. Algorithm
   2.1 Filter
   2.2 RMS
      2.2.1 Time Coefficients
   2.3 Gate
   2.4 Gain
      2.4.1 Loudness Goal Adaption
      2.4.2 Write Automation
   2.5 Lookahead

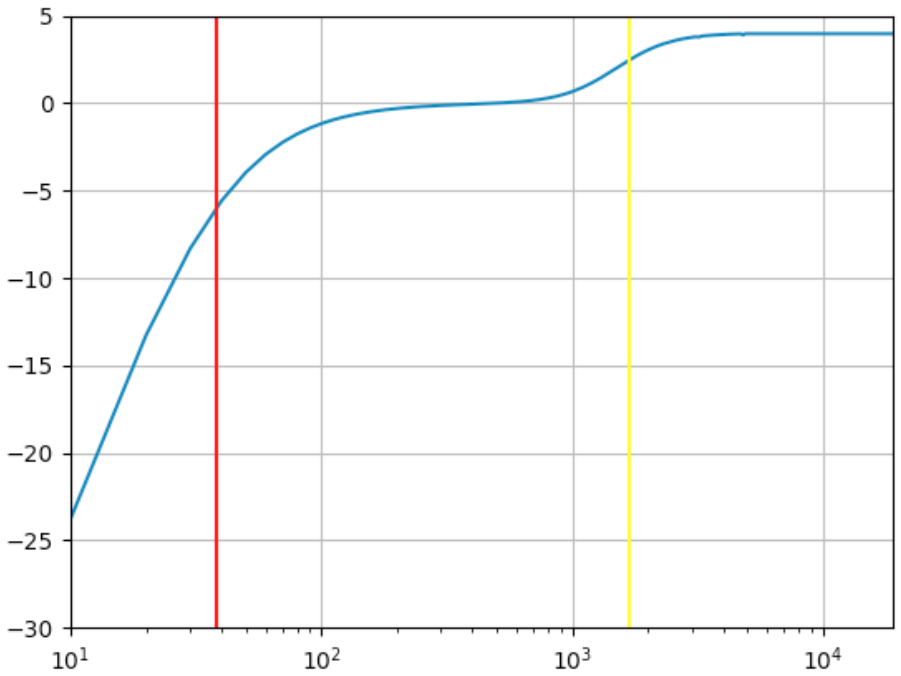
Algorithm: Filter



[1]

[1] **Figure from DAFX: Digital Audio Effects by Udo Zoelzer.**

Algorithm: Filter



- lowcut (38 Hz), highshelf (1681 Hz)[2]

[2] **from Recommendation ITU-R BS.1770-4.**

Algorithm: RMS

Root Mean Square (RMS):

```
void AutoVocalCtrlAudioProcessor::updateRMS(int channel)
{
    rms[channel] = (1. - rmsCo) * rms[channel] +
                   rmsCo * (filterSample[channel] * filterSample[channel]);
}
```

[3]

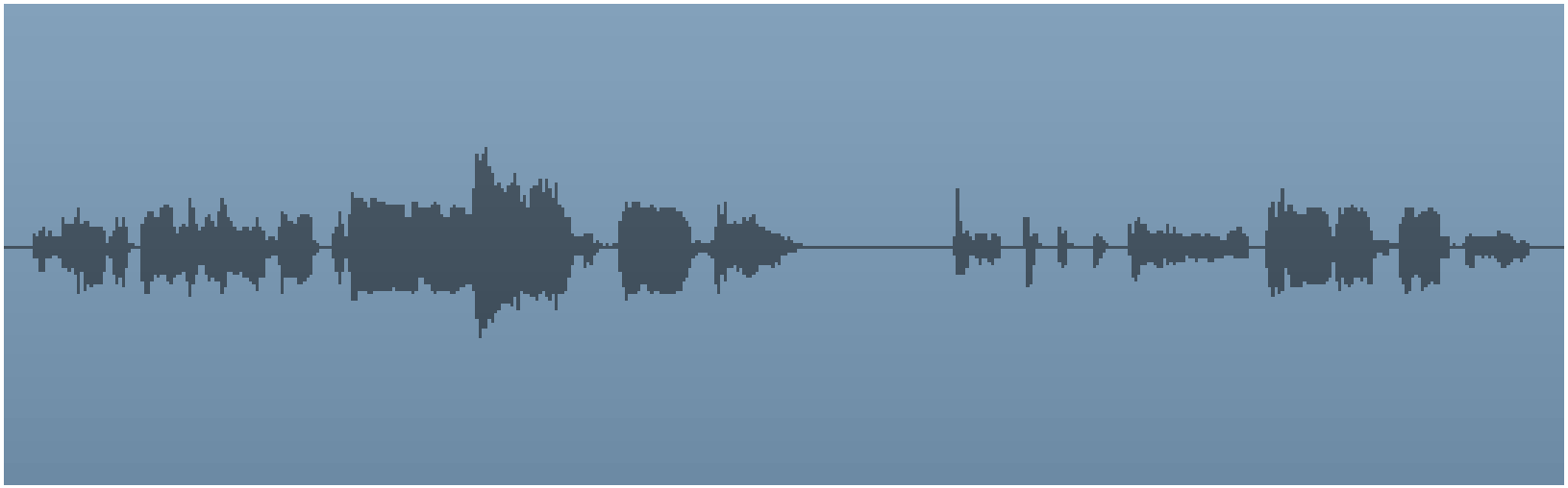
Time Constants:

```
float AutoVocalCtrlAudioProcessor::getTimeConstant(float ms)
{
    if (ms > 0.f)
        return 1.f - exp(-2.2*(1./currentSampleRate)/(ms/1000.));
    else
        return 1.f;
}
```

[2]

[3] **Based on Book: Digital Audio Signal Processing by Udo Zoelzer.**

Algorithm: Gate



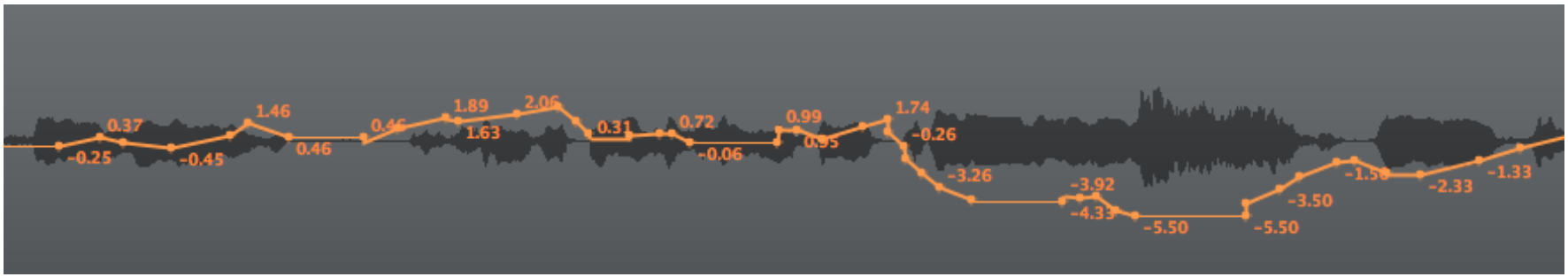
Algorithm: Gain

```
void AutoVocalCtrlAudioProcessor::updateGain(int channel)
{
    const double g = *loudnessGoal - mls[channel];
    const double co = g < gain[channel] ? compressTCo:expandTCo;
    gain[channel] = clipRange.clipValue((1 - co) * gain[channel] + co * g);
    updateAutomation();
    ...
}
```

Algorithm: Gain: Loudness Goal Adaption

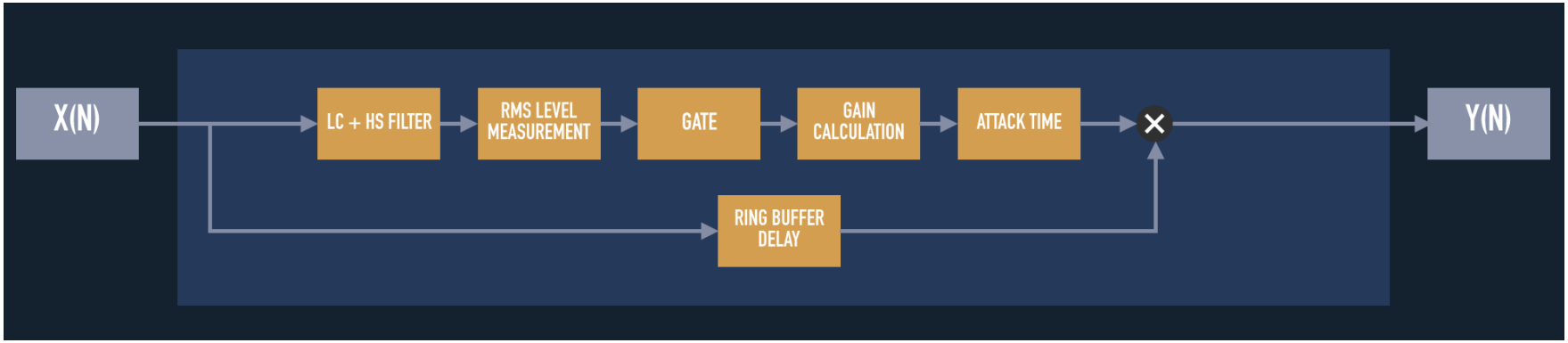
```
void AutoVocalCtrlAudioProcessor::updateGain(int channel)
{
    ...
    alphaGain[channel] = (1 - alphaCo) * alphaGain[channel] + alphaCo * gain[channel];
    if (channel == getTotalNumInputChannels() - 1)
    {
        ...
        updateLoudnessGoal();
    }
}
```

Algorithm: Gain: Write Automation



Algorithm: Lookahead

per ringbuffer

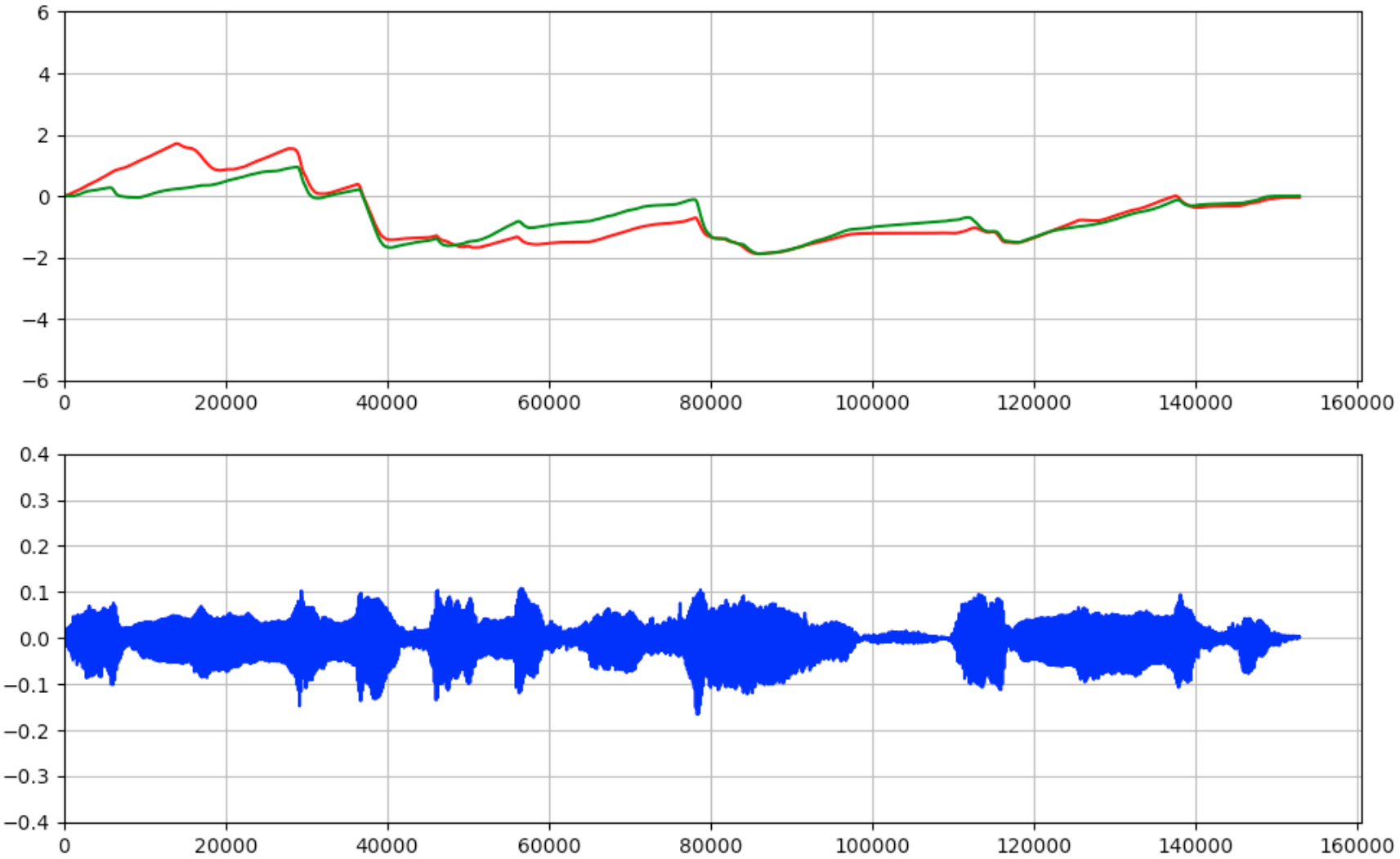


Optimization

Optimization

```
...
res = optmze.brute(compareGainCurve, bnds, full_output=True, finish=optmze.fmin)
...
res = optmze.minimize(compareGainCurve, x0, bounds=bnds, options={'disp': True, 'eps': 0.5})
...
x1 = np.array([-28.22, 5.65, 110.14, 2044.41, -33.94, 0.])
...
```

Optimization: Results



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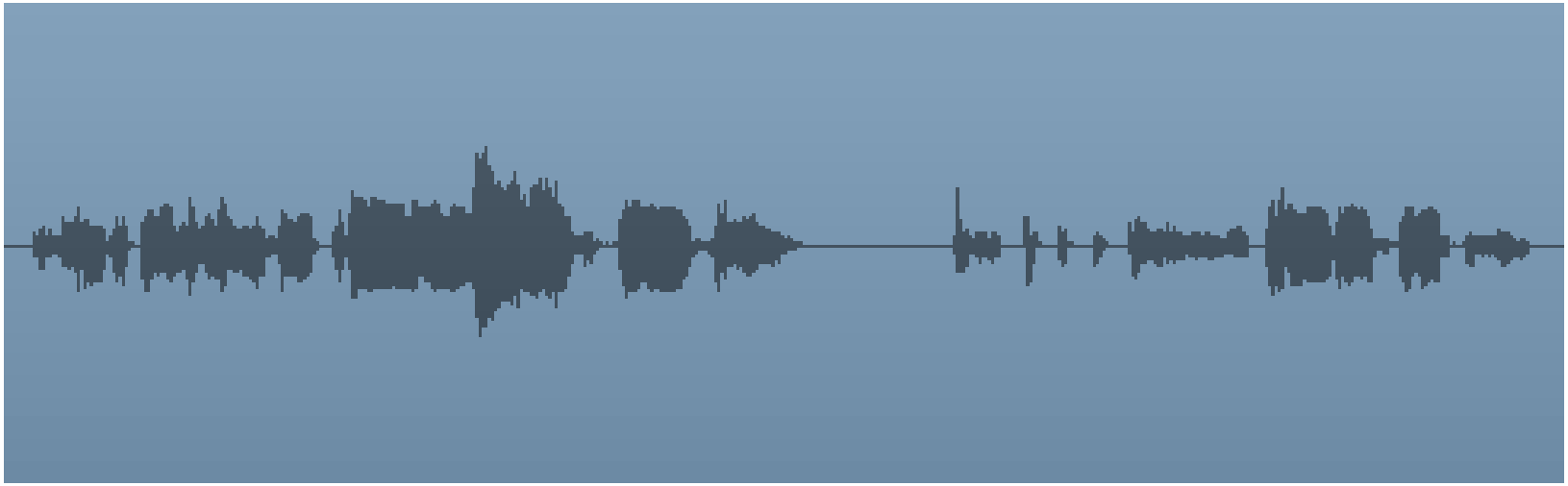
Results

Audio Examples



- Original:
- Gain Control:

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Future Implementations

Future Implementations

- side chain backtrack
- offline loudness goal calculation
- set parameters ➜ simplify UI
- improve writing of automation
- idle time?
- wet / dry?

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