An Introduction to

Building an audio effect plug-in in JUCE

An Introduction to REAL-TIME AUDIO PROCESSING

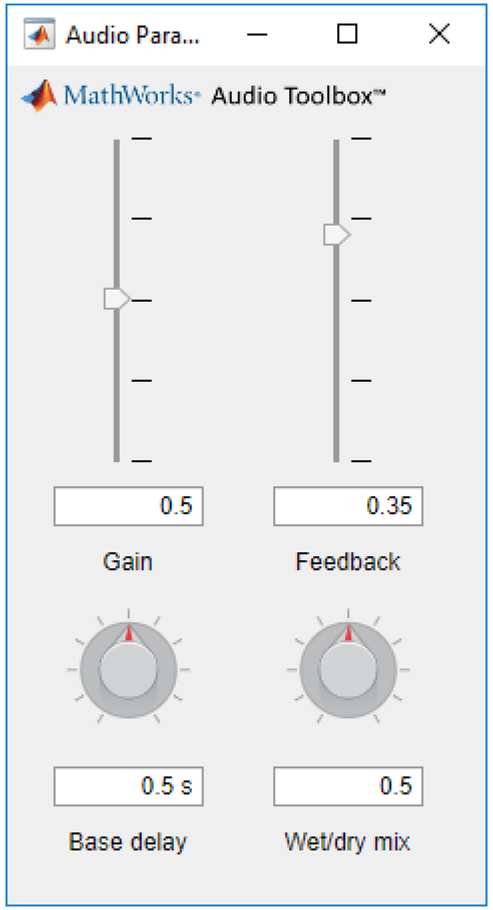
Non-REAL-TIME

```
Fs = 48000;              % Sampling Frequency
T = 1 / Fs;              % Sampling Step Size
duration  = 0.01;        % Signal duration
f0 = 100;                % Signal Frequency
t = T:T:duration;
u = sin(2*pi*f0*t);      % Signal
%% Signal processing stage
out = 2*u;

figure(1)
plot(t,out)
hold on
plot(t,u)
```

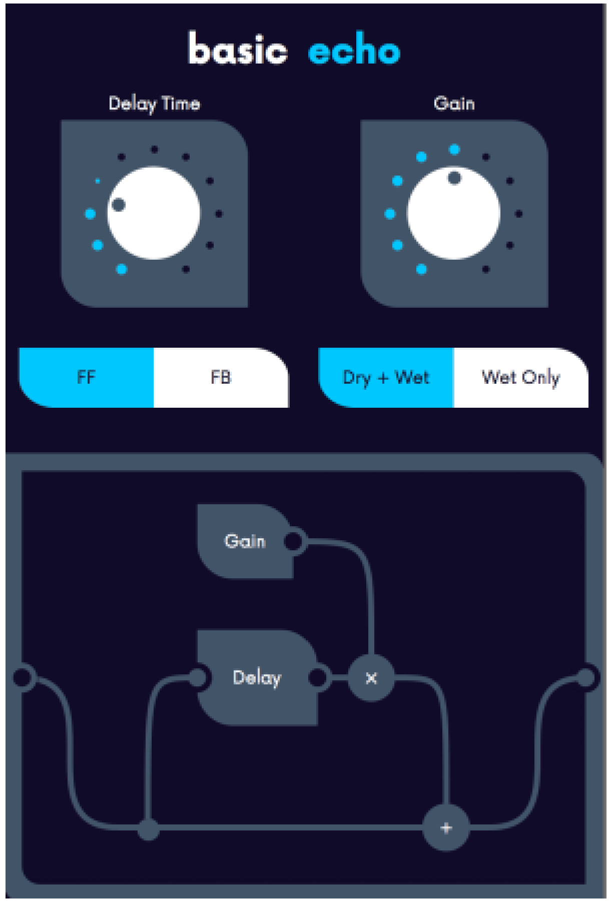
What if the whole signal is not available?

Process data in chunks using the buffer

1. Load data to the buffer.
2. Process the buffered data.
3. Output the buffer.



Gain: 0.5
Feedback: 0.35
Base delay: 0.5 s
Wet/dry mix: 0.5



An Introduction to JUCE

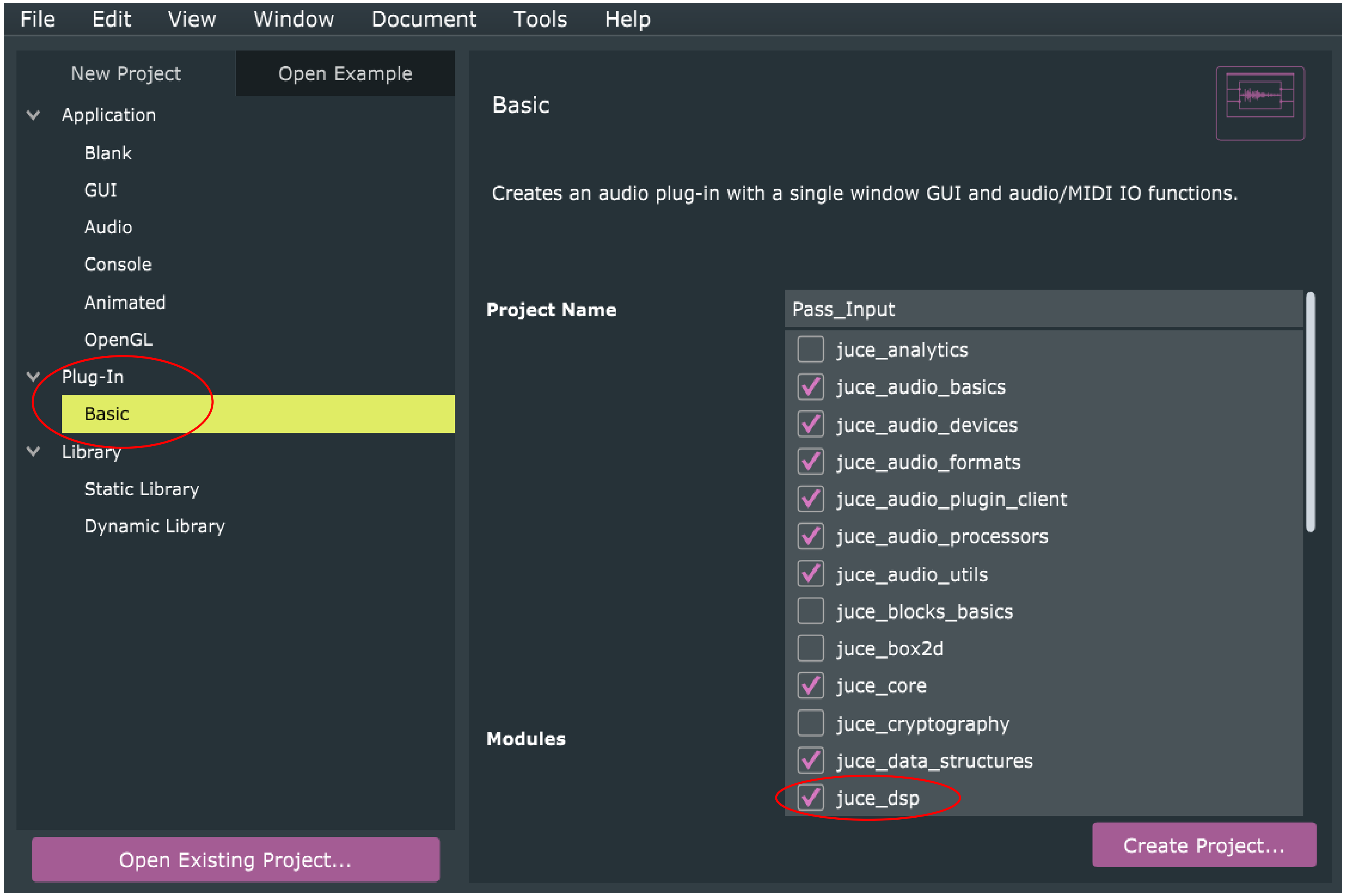
What is JUCE

Partially open-source framework for C++ application and plugin development.

- It is mainly used for or its GUI and plug-in libraries.
- Has useful dsp libraries and the learning curve is gentle.

JUCE was used to build Max.

Building a plug-in for audio effect in JUCE



File Edit View Window Document Tools Help

New Project | Open Example

- Application
  - Blank
  - GUI
  - Audio
  - Console
  - Animated
  - OpenGL
- Plug-In
  - Basic
- Library
  - Static Library
  - Dynamic Library

Open Existing Project...

Basic

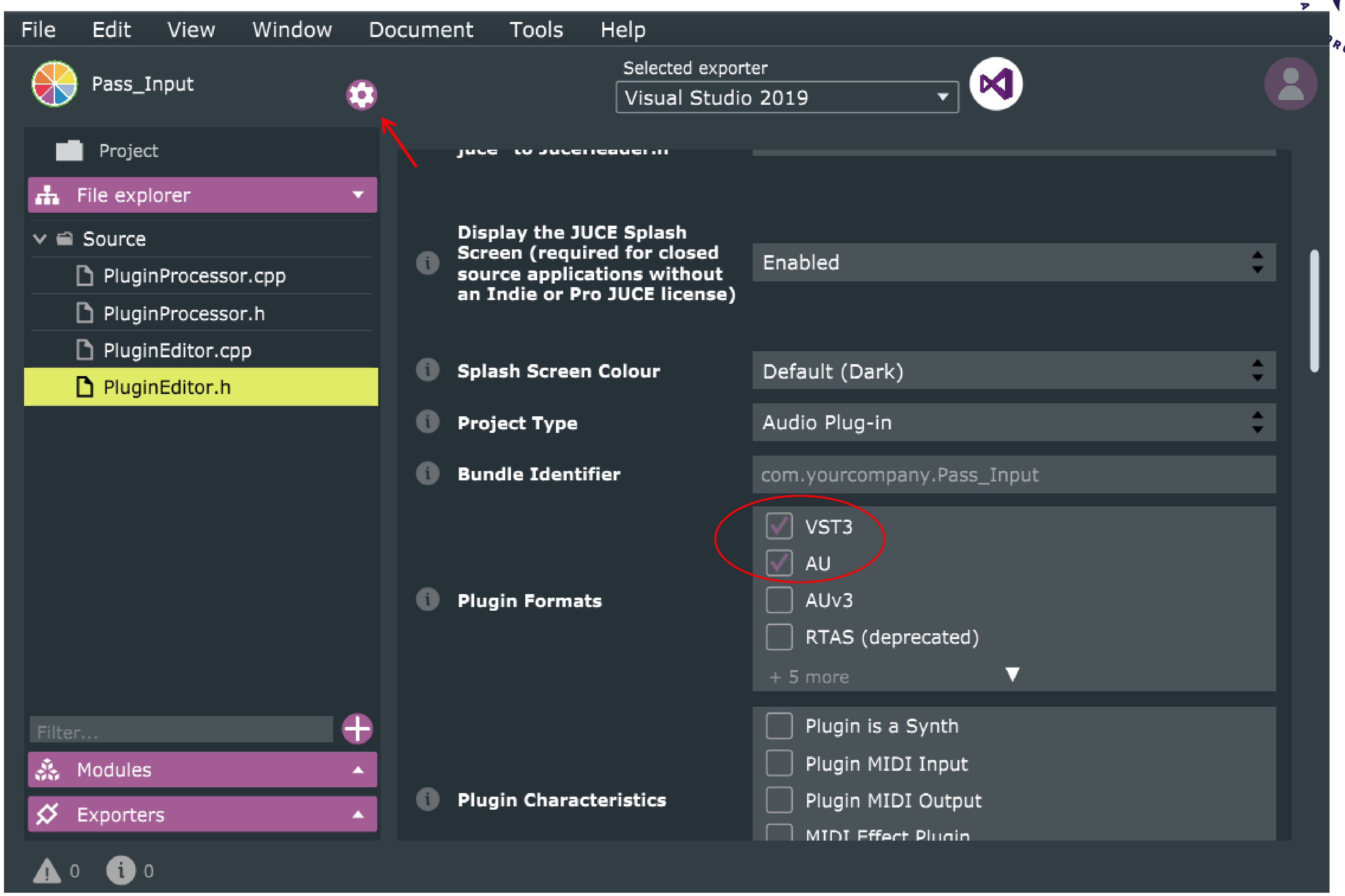
Creates an audio plug-in with a single window GUI and audio/MIDI IO functions.

Project Name Pass_Input

Modules

- [ ] juce_analytics
- [✓] juce_audio_basics
- [✓] juce_audio_devices
- [✓] juce_audio_formats
- [✓] juce_audio_plugin_client
- [✓] juce_audio_processors
- [✓] juce_audio_utils
- [ ] juce_blocks_basics
- [ ] juce_box2d
- [✓] juce_core
- [ ] juce_cryptography
- [✓] juce_data_structures
- [✓] juce_dsp

Create Project...



Display the JUCE Splash Screen (required for closed source applications without an Indie or Pro JUCE license)	Enabled
Splash Screen Colour	Default (Dark)
Project Type	Audio Plug-in
Bundle Identifier	com.yourcompany.Pass_Input
Plugin Formats	✓ VST3 ✓ AU AUv3 RTAS (deprecated) + 5 more
Plugin Characteristics	Plugin is a Synth Plugin MIDI Input Plugin MIDI Output MIDI Effect Plugin

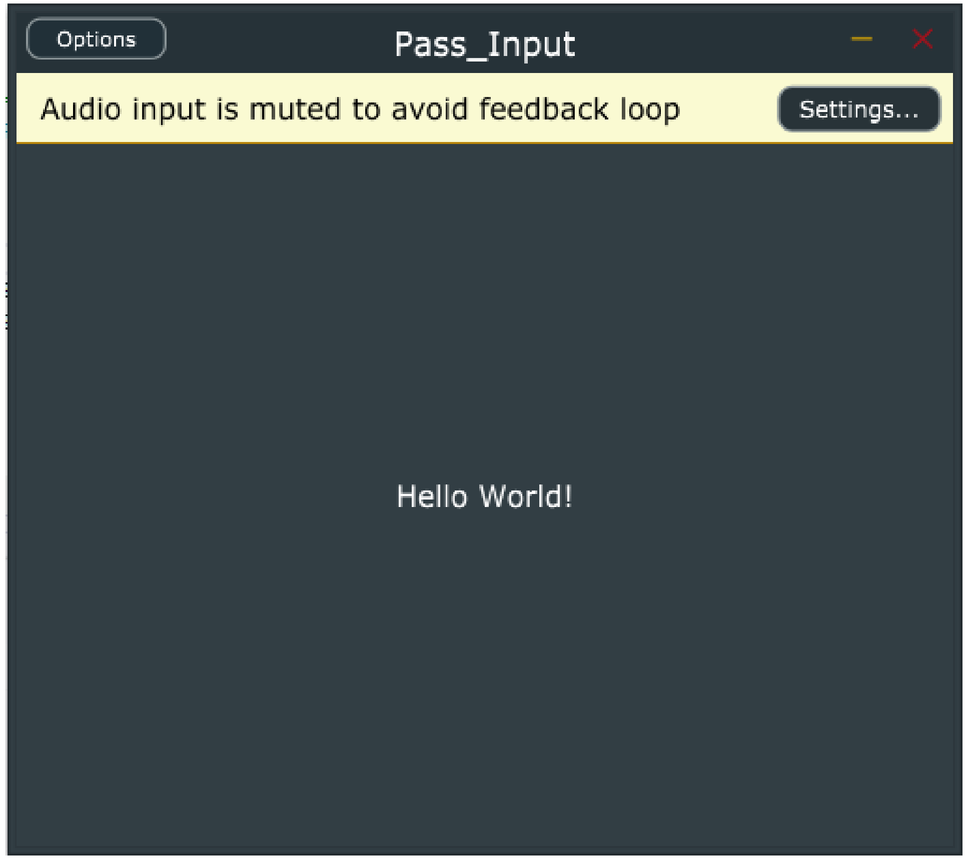
We don’t need to change anything in these

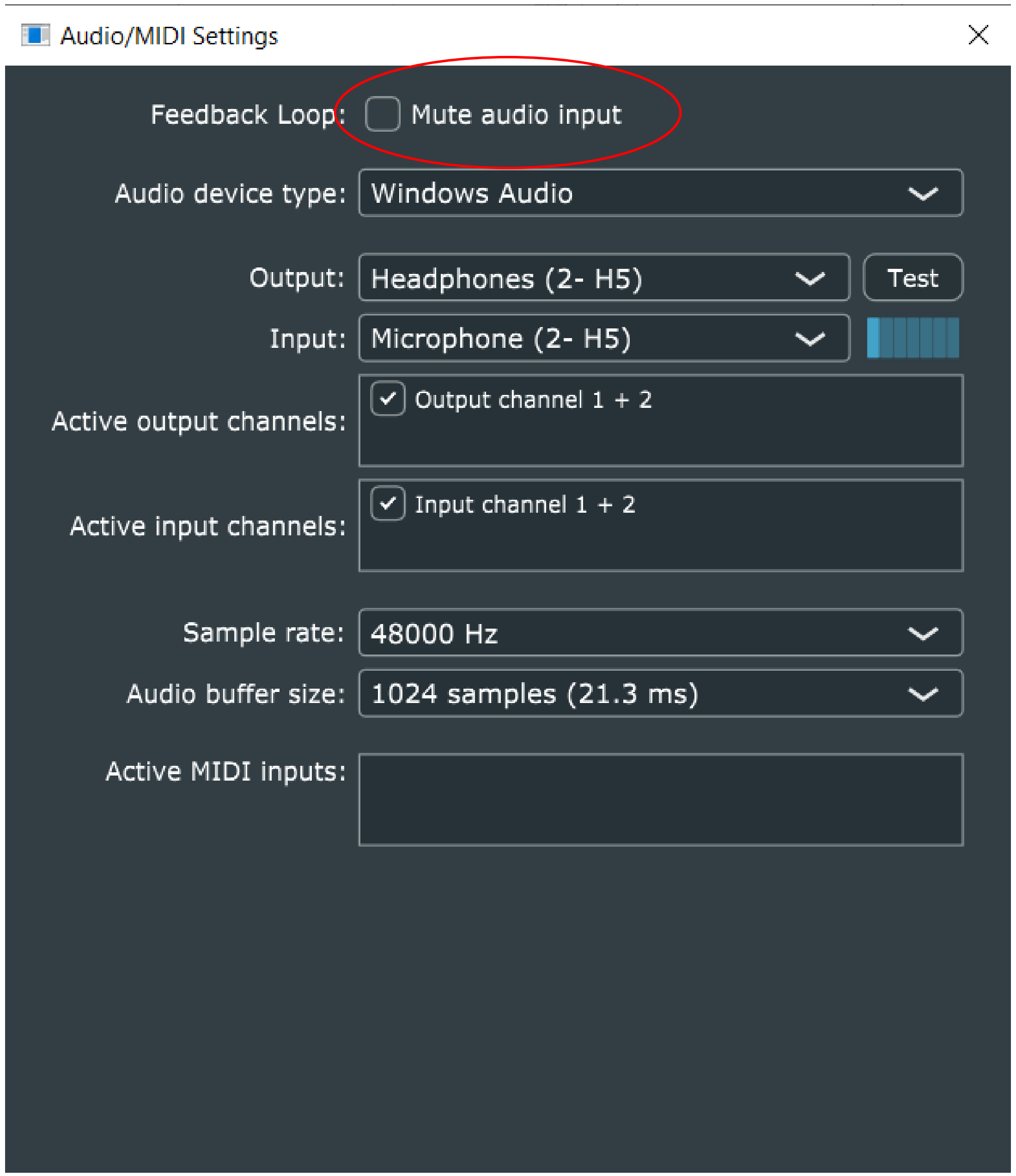
```
Pass_InputAudioProcessor::Pass_InputAudioProcessor()
Pass_InputAudioProcessor::~Pass_InputAudioProcessor()
const juce::String Pass_InputAudioProcessor::getName() const
bool Pass_InputAudioProcessor::acceptsMidi() const
bool Pass_InputAudioProcessor::producesMidi() const
bool Pass_InputAudioProcessor::isMidiEffect() const
double Pass_InputAudioProcessor::getTailLengthSeconds() const
int Pass_InputAudioProcessor::getNumPrograms()
int Pass_InputAudioProcessor::getCurrentProgram()
void Pass_InputAudioProcessor::setCurrentProgram (int index)
const juce::String Pass_InputAudioProcessor::getProgramName (int index)
void Pass_InputAudioProcessor::changeProgramName (int index, const juce::String& newName)
void Pass_InputAudioProcessor::releaseResources()
bool Pass_InputAudioProcessor::isBusesLayoutSupported (const BusesLayout& layouts) const
bool Pass_InputAudioProcessor::hasEditor() const
juce::AudioProcessorEditor* Pass_InputAudioProcessor::createEditor()
void Pass_InputAudioProcessor::updateFilter(bool realFreq)
void Pass_InputAudioProcessor::getStateInformation (juce::MemoryBlock& destData)
void Pass_InputAudioProcessor::setStateInformation (const void* data, int sizeInBytes)
juce::AudioProcessor* JUCE_CALLTYPE createPluginFilter()
```

processBlock

```
void Pass_InputAudioProcessor::processBlock (juce::AudioBuffer<float>& buffer, juce::MidiBuffer& midiMessages)
{
    auto totalNumInputChannels = getTotalNumInputChannels();
    auto totalNumOutputChannels = getTotalNumOutputChannels();
    /***************************************************************************************/
    for (int i = totalNumInputChannels; i < totalNumOutputChannels; ++i) // clear the buffer
        buffer.clear(i, 0, buffer.getNumSamples());
    /***************************************************************************************/
    for (int channel = 0; channel < buffer.getNumChannels(); ++channel){
        for (int sample = 0; sample < buffer.getNumSamples(); ++sample){
            vin = buffer.getSample(channel, sample);    // get the input sample
            vout = vin;
            buffer.setSample(channel, sample, vout);    // set the output sample value
        }
    }
}
```

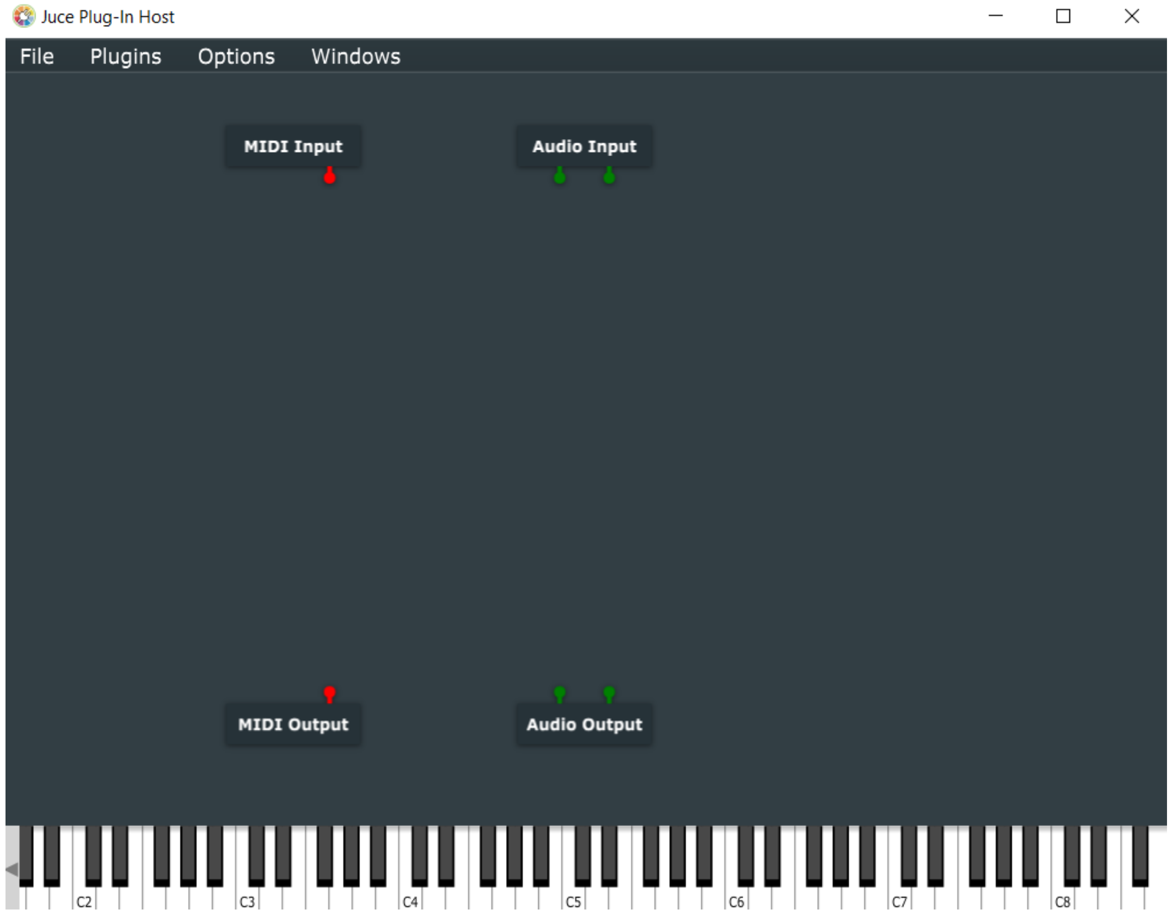
That’s it!!

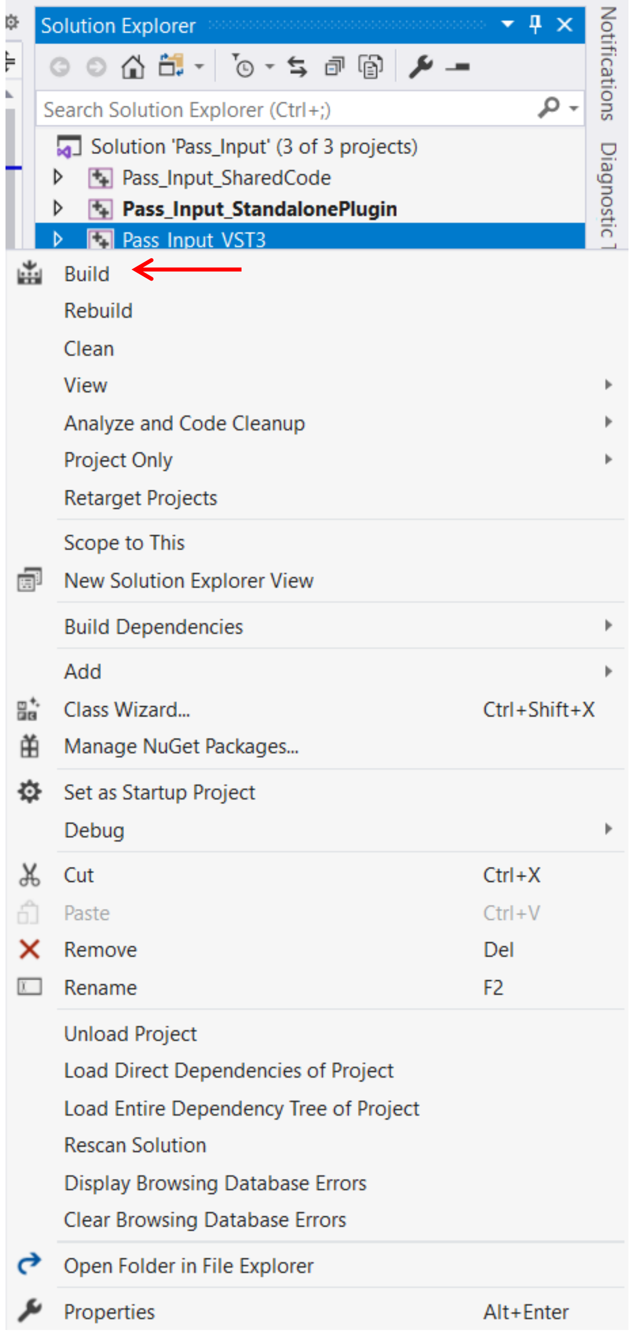

Options
Pass_Input
Audio input is muted to avoid feedback loop
Settings...
Hello World!


Audio/MIDI Settings
Feedback Loop: ☐ Mute audio input
Audio device type: Windows Audio
Output: Headphones (2- H5) Test
Input: Microphone (2- H5)
Active output channels: ✓ Output channel 1 + 2
Active input channels: ✓ Input channel 1 + 2
Sample rate: 48000 Hz
Audio buffer size: 1024 samples (21.3 ms)
Active MIDI inputs:

Build a VST3 or (AU) plugin and test it

- Audio Plugin Host
  - JUCE\extras\AudioPluginHost\Builds\VisualStudio2019\x64\Release\App





Let’s control the gain

```
Pass_InputAudioProcessorEditor:: Pass_InputAudioProcessorEditor(Pass_InputAudioProcessor& p,
juce::AudioProcessorValueTreeState& vts)
    : AudioProcessorEditor(&p), audioProcessor(p), audioTree(vts)
{
    setSize(450, 250);
    controlGain.setColour(0x1001400, juce::Colour::fromRGBA(0x00, 0x40, 0x00, 0x80));   *
    controlGain.setColour(0x1001700, juce::Colour::fromRGBA(0x00, 0x00, 0x00, 0x00));

    controlGain.setSliderStyle(juce::Slider::LinearHorizontal);
    controlGain.setTextBoxStyle(juce::Slider::TextBoxLeft, false, 40, 20);
    addAndMakeVisible(controlGain);
    sliderAttachGain.reset(new juce::AudioProcessorValueTreeState::SliderAttachment(audioTree, "controlGain_ID",
controlGain));
    labelGain.setText(("Input Gain"), juce::dontSendNotification);
    labelGain.setFont(juce::Font("Slope Opera", 16, 0));
    labelGain.setColour(juce::Label::textColourId, juce::Colour::fromRGBA(0x40, 0x40, 0x80, 0xff));
    addAndMakeVisible(labelGain);
}
```

- Add `juce::AudioProcessorValueTreeState audioTree;` to the header file under `public` class definitions

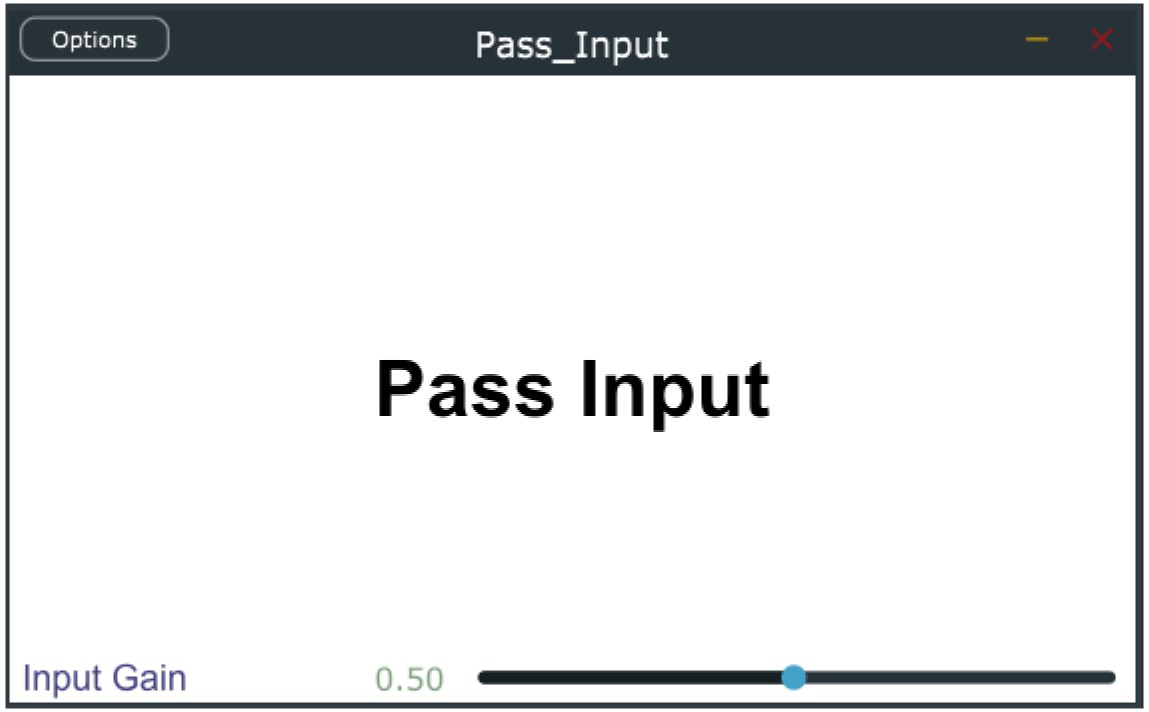
```
Pass_InputAudioProcessorEditor::~Pass_InputAudioProcessorEditor()
{
    sliderAttachR.reset();
}
void Pass_InputAudioProcessorEditor::paint(juce::Graphics& g)
{
    // (Our component is opaque, so we must completely fill the background with a solid colour)
    g.fillAll(juce::Colours::white);
    // set the current drawing colour to black
    g.setColour(juce::Colours::black);

    // set the font size and draw text to the screen
    g.setFont(15.0f);
    g.setFont(juce::Font("Slope Opera", 35.0f, 1));
    g.drawFittedText(“Pass Input", getLocalBounds(), juce::Justification::centred, 1);
} void Pass_InputAudioProcessorEditor::resized()
{
    controlGain.setBounds(140, getHeight() - 20, getWidth() - 140, 20);
    labelGain.setBounds(0, getHeight() - 20, 140, 20);
}
```

```
#include <JuceHeader.h>
#include "PluginProcessor.h"
class Pass_InputAudioProcessorEditor  : public juce::AudioProcessorEditor
{
public:
    Pass_InputAudioProcessorEditor (Pass_InputAudioProcessor&, juce::AudioProcessorValueTreeState&);
    ~Pass_InputAudioProcessorEditor() override;

    //==============================================================================
    void paint (juce::Graphics&) override;
    void resized() override;

private:
    // This reference is provided as a quick way for your editor to
    // access the processor object that created it.
    Pass_InputAudioProcessor& audioProcessor;
    juce::AudioProcessorValueTreeState& audioTree;
    juce::Slider controlGain;
    juce::Label labelGain;
    std::unique_ptr <juce::AudioProcessorValueTreeState::SliderAttachment> sliderAttachGain;
    JUCE_DECLARE_NON_COPYABLE_WITH_LEAK_DETECTOR (Pass_InputAudioProcessorEditor)
};
```

```
Pass_InputAudioProcessor::Pass_InputAudioProcessor()
#ifndef JucePlugin_PreferredChannelConfigurations
     : AudioProcessor (BusesProperties()
                     #if ! JucePlugin_IsMidiEffect
                      #if ! JucePlugin_IsSynth
                       .withInput  ("Input",  juce::AudioChannelSet::stereo(), true)
                      #endif
                       .withOutput ("Output", juce::AudioChannelSet::stereo(), true)
                     #endif
                       ),
                    audioTree(*this, nullptr, juce::Identifier("PARAMETERS"),
                        {
std::make_unique<juce::AudioParameterFloat>("controlGain_ID","ControlGain",juce::NormalisableRange<float
>(0.01, 1.0, 0.01),0.5)
                        })
#endif
{
    audioTree.addParameterListener("controlGain_ID", this);
    controlledGain = 0.5;
}
void Pass_InputAudioProcessor::prepareToPlay (double sampleRate, int samplesPerBlock)
{ controlledGain = 0.5;}
```

```
void Pass_InputAudioProcessor::parameterChanged(const juce::String& parameterID, float newValue)
{
    if (parameterID == "controlGain_ID") {
        controlledGain = newValue; // controlledGain is defined in the header file as a private variable
    }
}
void Pass_InputAudioProcessor::processBlock (juce::AudioBuffer<float>& buffer, juce::MidiBuffer& midiMessages)
{
    auto totalNumInputChannels = getTotalNumInputChannels();
    auto totalNumOutputChannels = getTotalNumOutputChannels();
    /*******************************************************************************/
    for (int i = totalNumInputChannels; i < totalNumOutputChannels; ++i) // clear the buffer
        buffer.clear(i, 0, buffer.getNumSamples());
    /*******************************************************************************/
    for (int channel = 0; channel < buffer.getNumChannels(); ++channel){
        for (int sample = 0; sample < buffer.getNumSamples(); ++sample){
            vin = buffer.getSample(channel, sample);    // get the input sample
            vout = vin * controlledGain;
            buffer.setSample(channel, sample, vout);    // set the output sample value
        }
    }
}
```

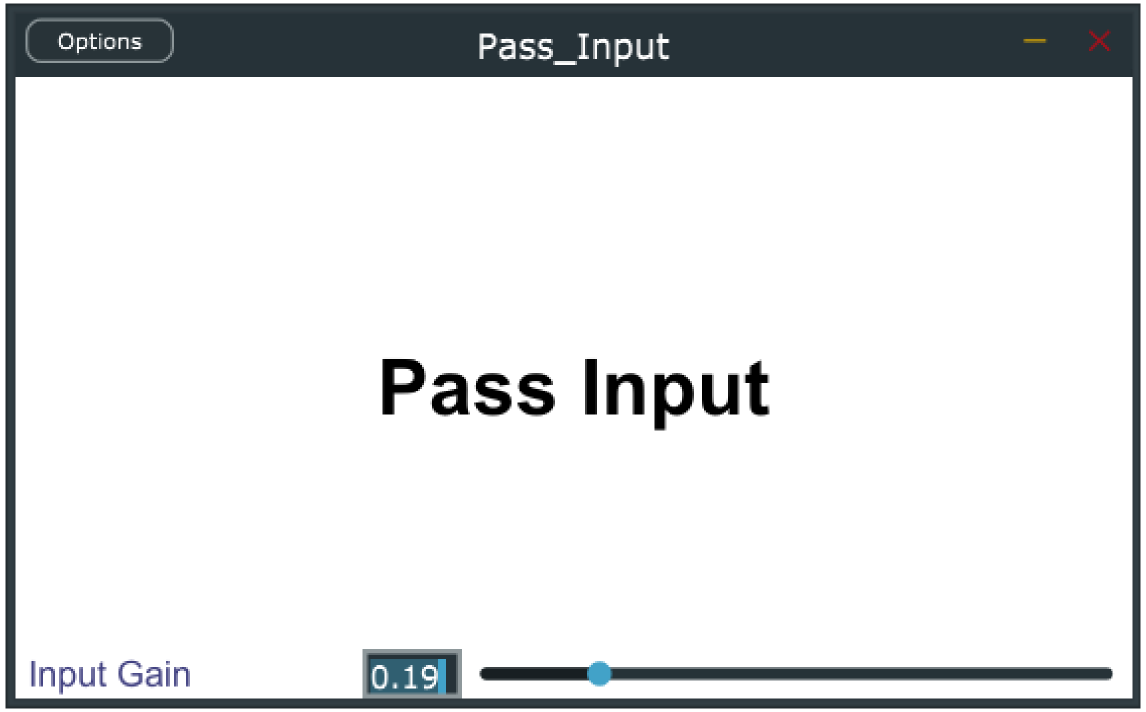
Wow


Options

Pass_Input

Pass Input

Input Gain 0.50


Options

Pass_Input

Pass Input

Input Gain 0.19

Oversampling simplified

1. Up-sample the input
2. Apply steep low pass at Nyquist of original signal
3. Process the signal
4. Apply low pass again
5. Down-sample the output

```
#include "PluginProcessor.h"
#include "PluginEditor.h"
//==============================================================================
Pass_InputAudioProcessor::Pass_InputAudioProcessor()
#ifndef JucePlugin_PreferredChannelConfigurations
     : AudioProcessor (BusesProperties()
                     #if ! JucePlugin_IsMidiEffect
                      #if ! JucePlugin_IsSynth
                       .withInput  ("Input",  juce::AudioChannelSet::stereo(), true)
                      #endif
                       .withOutput ("Output", juce::AudioChannelSet::stereo(), true)
                     #endif
                       ),
                        audioTree(*this, nullptr, juce::Identifier("PARAMETERS"),
{std::make_unique<juce::AudioParameterFloat>("controlGain_ID","ControlGain",juce::NormalisableRange<float>(0.01, 1.0, 0.01),0.5)
                        }),
               →       lowPassFilter(juce::dsp::IIR::Coefficients< float >::makeLowPass((48000.0 * 4.0), 20000.0))
#endif
{
  →   oversampling.reset(new juce::dsp::Oversampling<float>(2, 2, juce::dsp::Oversampling<float>::filterHalfBandPolyphaseIIR,
true));
}
Pass_InputAudioProcessor::~Pass_InputAudioProcessor()
{
  →   oversampling.reset();
}
```

prepareToPlay

```
void Pass_InputAudioProcessor::prepareToPlay (double sampleRate, int samplesPerBlock)
{
    oversampling->reset();
    oversampling->initProcessing(static_cast<size_t> (samplesPerBlock));

    juce::dsp::ProcessSpec spec;
    spec.sampleRate = sampleRate * 4;
    spec.maximumBlockSize = samplesPerBlock * 3;
    spec.numChannels = getTotalNumOutputChannels();

    lowPassFilter.prepare(spec);
    lowPassFilter.reset();

    controlledGain = 0.5;
}
```

```cpp
void Pass_InputAudioProcessor::processBlock (juce::AudioBuffer<float>& buffer, juce::MidiBuffer& midiMessages)
{
    auto totalNumInputChannels = getTotalNumInputChannels();
    auto totalNumOutputChannels = getTotalNumOutputChannels();
    /*****************************************************************************/
    for (int i = totalNumInputChannels; i < totalNumOutputChannels; ++i) // clear the buffer
        buffer.clear(i, 0, buffer.getNumSamples());
    juce::dsp::AudioBlock<float> blockInput(buffer);
    juce::dsp::AudioBlock<float> blockOutput = oversampling->processSamplesUp(blockInput);
    updateFilter(0);
    lowPassFilter.process(juce::dsp::ProcessContextReplacing<float>(blockOutput));
    /*****************************************************************************/
    for (int channel = 0; channel < blockOutput.getNumChannels(); ++channel){
        for (int sample = 0; sample < blockOutput.getNumSamples(); ++sample){
            vin = blockOutput.getSample(channel, sample);   // get the input sample
            vout = vin * controlledGain;
            blockOutput.setSample(channel, sample, vout);   // set the output sample value
        }
    }
    updateFilter(0);
    lowPassFilter.process(juce::dsp::ProcessContextReplacing<float>(blockOutput));
    oversampling->processSamplesDown(blockInput);

    updateFilter(1);
    lowPassFilter.process(juce::dsp::ProcessContextReplacing<float>(blockInput));
}
```

```
void Pass_InputAudioProcessor::updateFilter(bool realFreq)
{
    double frequency;

    if (realFreq) frequency = 48e3;
    else frequency = 48e3 * 4;

    *lowPassFilter.state =
*juce::dsp::IIR::Coefficients<float>::makeLowPass(frequency, 20000.0);
}
```

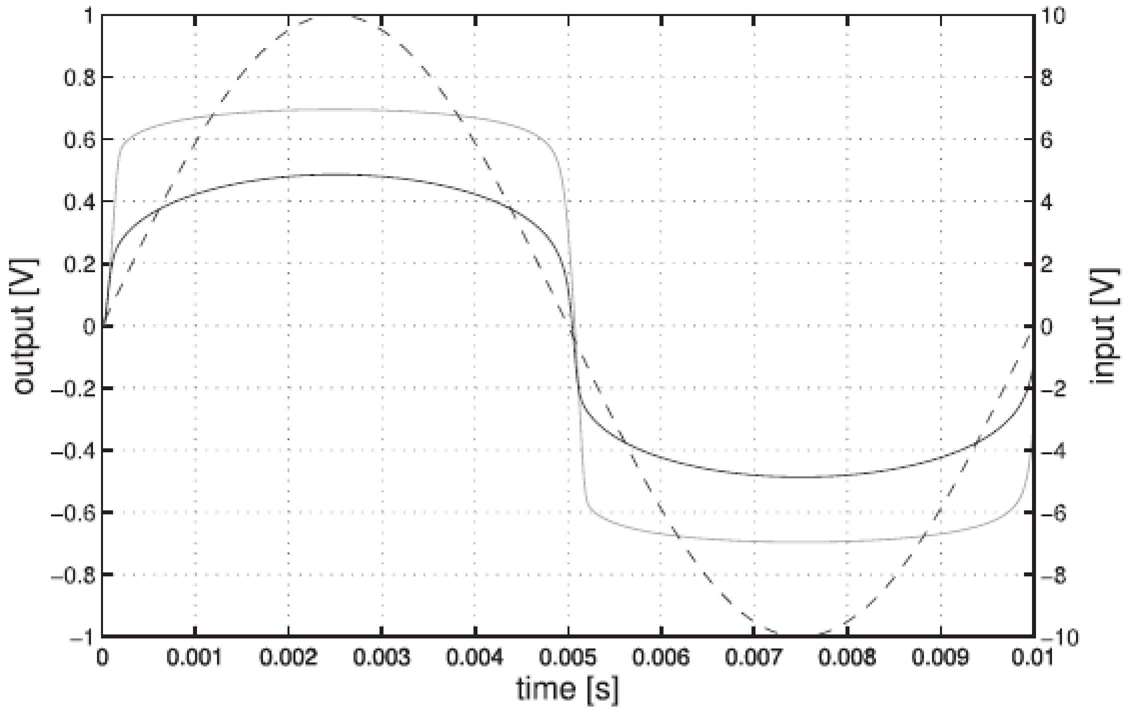
Newton Raphson Diode Clipper

Fixed Point

```
v(n) = R*T/(R*C+T)*(gdExp(-vTemp) - gdExp(vTemp))
+ T/(R*C+T)*newU(n) + R*C/(R*C+T)*v(n-1);
```

Newton Raphson

```
vNom = T*vTemp*R*gdExpDiff(-vTemp) + T*R*gdExp(-vTemp) +
v(n - 1)*R*C + T*(gdExpDiff(vTemp)*R*vTemp -
R*gdExp(vTemp) + newU(n));

vDenom = T*R*gdExpDiff(vTemp) + T*R*gdExpDiff(-vTemp) +
R*C + T;

v(n) = vNom/vDenom;
```



Using exponential diode characteristic

$$g_D(v) = I_D(e^{\frac{v}{2V_E}} - 1)$$

Pass_InputAudioProcessor ➔ DiodeClipperNRAudioProcessor

PluginProcessor.h

```
public:
…
    double gdExp(double vc);
    double gdExpDiff(double vc);
    double limiter(double val);
private:
    double controlledGain;
    double Id, C, Ve, Vp, R, err;
    double Fs, T;
    double vNom, vDenom;
    float vin;
    float vout, voutTemp, voutOld;
…
```

PluginProcessor.cpp

```
…
double DiodeClipperNRAudioProcessor::gdExp(double vc)
{
    return Id * (std::exp(vc / (2 * Ve)) - 1);
}
double DiodeClipperNRAudioProcessor::gdExpDiff(double vc)
{
    return (Id * std::exp(vc / (2 * Ve))) / (2 * Ve);
}
double DiodeClipperNRAudioProcessor::limiter(double val)
{
    if (val < -1){
        val = -1;
        return val;
    } else if (val > 1){
        val = 1;
        return val;
    } else return val;
}
…
```

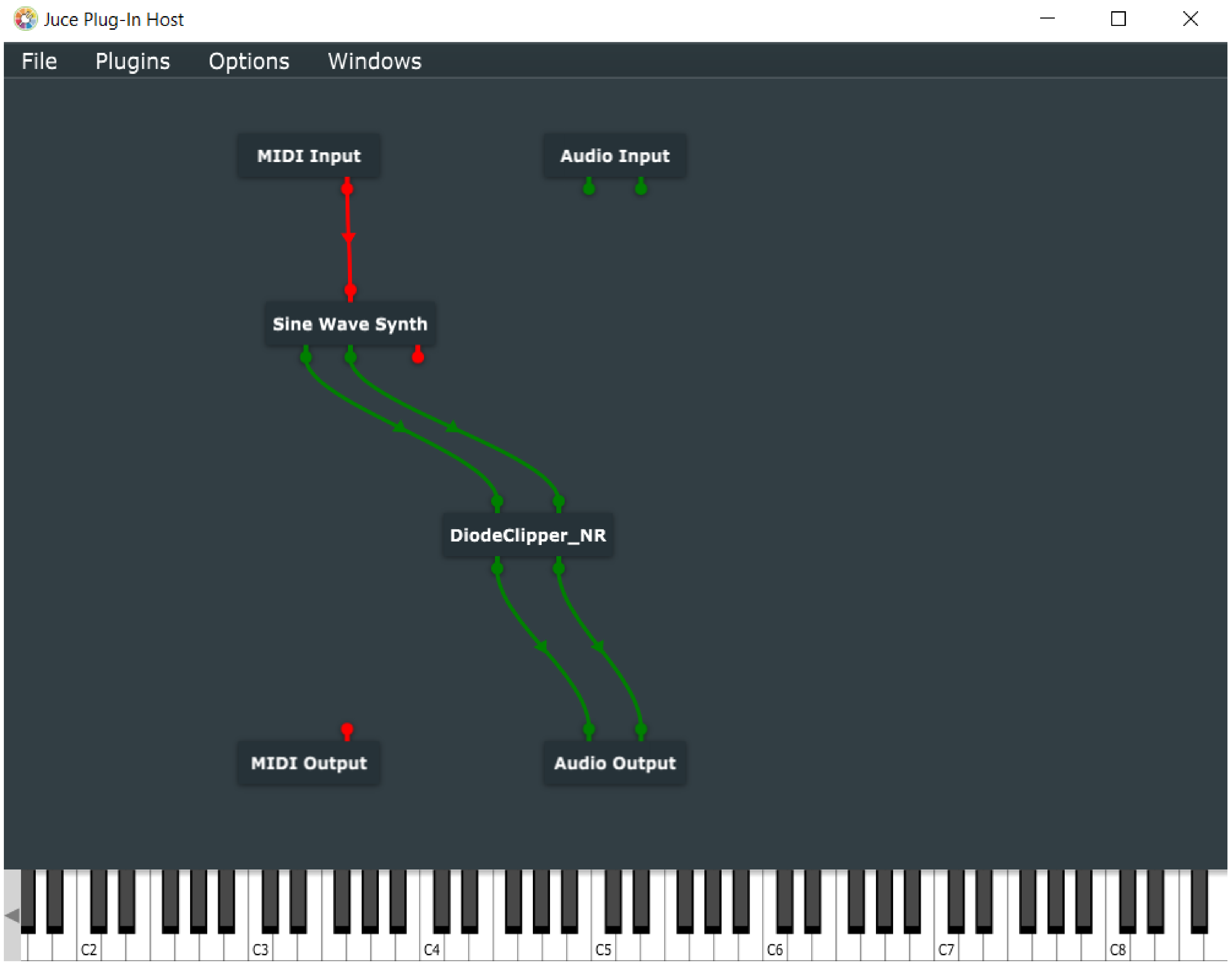
```
DiodeClipperNRAudioProcessor::DiodeClipperNRAudioProcessor()
#ifndef JucePlugin_PreferredChannelConfigurations
    : AudioProcessor(BusesProperties()
#if ! JucePlugin_IsMidiEffect
#if ! JucePlugin_IsSynth
        .withInput("Input", juce::AudioChannelSet::stereo(), true)
#endif
        .withOutput("Output", juce::AudioChannelSet::stereo(), true)
#endif
    ),
    audioTree(*this, nullptr, juce::Identifier("PARAMETERS"),
        {
std::make_unique<juce::AudioParameterFloat>("controlGain_ID","ControlGain",juce::NormalisableRange
<float>(0.0, 130.0, 0.1),1.0)
        }),
    lowPassFilter(juce::dsp::IIR::Coefficients< float >::makeLowPass((48000.0 * 16.0), 20000.0))
#endif
{
    oversampling.reset(new juce::dsp::Oversampling<float>(2, 4,
juce::dsp::Oversampling<float>::filterHalfBandPolyphaseIIR, false));
    audioTree.addParameterListener("controlGain_ID", this);
    controlledGain = 1.0;
}
```

```
void DiodeClipperNRAudioProcessor::prepareToPlay(double sampleRate, int samplesPerBlock)
{
    …
    spec.sampleRate = sampleRate * 16;
    spec.maximumBlockSize = samplesPerBlock * 15;
    spec.numChannels = getTotalNumOutputChannels();
    lowPassFilter.prepare(spec);
    lowPassFilter.reset();
    Fs = sampleRate;
    err = 0.1e-3; // err for stopping iterations
    T = 1 / Fs;
    C = 100e-9;
    R = 1e3;
    Vp = 0.17;
    Ve = 0.023;
    Id = 2.52e-9;
    controlledGain = 1.0;
    voutOld = 0;
}
```

```
void DiodeClipperNRAudioProcessor::updateFilter()
{
    double frequency;
    if (realFreq) frequency = 48e3;
    else frequency = 48e3 * 16;
    *lowPassFilter.state =
*juce::dsp::IIR::Coefficients<float>::makeLowPass(frequency, 20000.0);
}
```

PluginProcessor.cpp `processBlock`

```
…
for (int channel = 0; channel < blockOutput.getNumChannels(); ++channel){
        for (int sample = 0; sample < blockOutput.getNumSamples(); ++sample){
            voutTemp = 1;
            vout = 0;
            vin = controlledGain *blockOutput.getSample(channel, sample);
            while (std::abs(voutTemp - vout) > err) {
                voutTemp = vout;
                vNom = T * voutTemp * R * gdExpDiff(-voutTemp) + T * R * gdExp(-voutTemp) + voutOld * R * C + T * (gdExpDiff(voutTemp) * R * 
voutTemp - R * gdExp(voutTemp) + vin);
                vDenom = T * R * gdExpDiff(voutTemp) + T * R * gdExpDiff(-voutTemp) + R * C + T;
                vout = vNom / vDenom;
            }
            voutOld = vout;
            vout = limiter(vout);
            blockOutput.setSample(channel, sample, vout);
        }
    }
…
```


Juce Plug-In Host
File
Plugins
Options
Windows
MIDI Input
Audio Input
Sine Wave Synth
DiodeClipper_NR
MIDI Output
Audio Output
C2
C3
C4
C5
C6
C7
C8

Useful links

- Audio Processor Tree https://docs.juce.com/master/classAudioProcessorValueTreeState.html
- Oversampling class https://docs.juce.com/master/classdsp_1_1Oversampling.html
- Sliders (Colour IDs) https://docs.juce.com/master/classSlider.html
- Filters https://docs.juce.com/master/structdsp_1_1FilterDesign.html
- Github link with Pass_Input JUCE files https://github.com/titas2001/JUCE_Audio_Effect_Core