CM3106: Multimedia
Tutorial/Lab Class 2 (Week 3)
Coursework Handout
Time-Frequency Analysis
(Short Time Fourier Transform)

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All Lab Materials available at:
http://users.cs.cf.ac.uk/Dave.Marshall/Multimedia/PDF/tutorial.html

Izotope **Iris**

You should develop an **Interactive Fourier-based Synthesiser** with **granular synthesis** capabilities in MATLAB:

The inspiration for the work is a piece of audio software called *Iris* by Izotope Inc.

What is Iris?

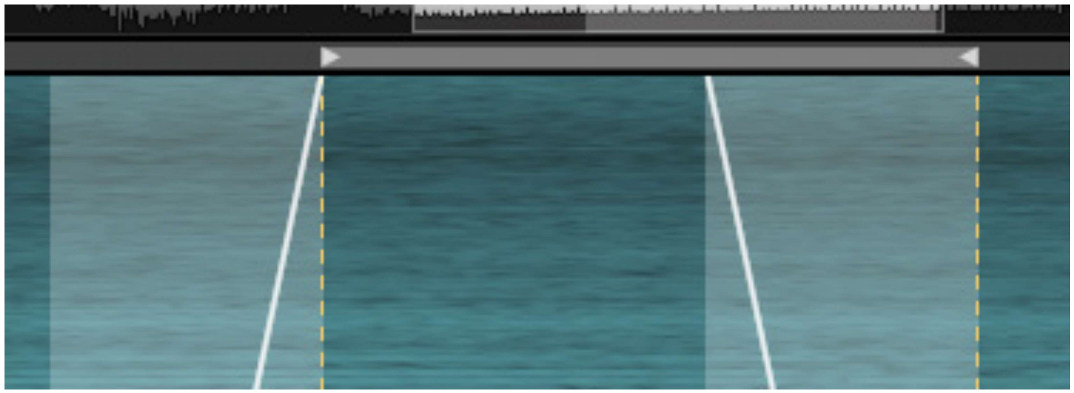
Izotope **Iris**

Iris is an innovative sampling **Fourier-based** re-synthesiser:

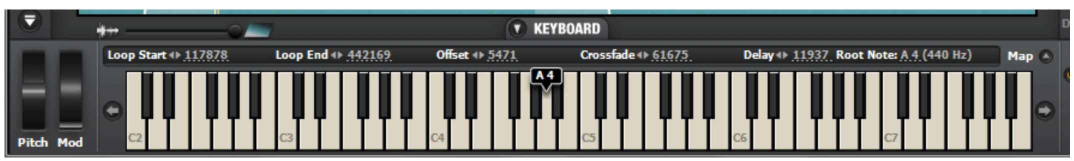
- You can input up to 3 waveforms and dissect and process them in many ways.
- Using *Iris*'s *spectrogram display* and easy drawing/selection tools to spotlight the most interesting spectral characteristics you can blend and layer your modified samples with some otherwise unrealisable filters.
- The sounds can then subsequently processed with other audio effects.

Key Iris Features

- Looping of samples:



- Keyboard Control:



See Izotope Iris manual: http://help.izotope.com/docs/izotope-iris-help.pdf and download the demo code to understand Iris's full potential.

- Layers, Audio Effects and ADSR and LFO control:

Granular Convolution

Granular Convolver

Source: youtube video

You are required to create a MATLAB program that implements a **Granular Convolver** enhanced with the **basic spectrogram editing** and playback functionality of **Iris** with some **additional additional audio processing**.

Granular Convolution and a simple version of this!

Basic Coursework Requirements

What do I need to do to pass the coursework?

- Input audio file, compute time-frequency **short-term Fourier transform** and **spectrogram**.
- Provide an **interactive means of editing** this time-frequency form via its displayed spectrogram.
- The resulting edited audio then needs to be **played back**: **inverse short-term Fourier transform**
- The resulting edited audio then needs to be played back: Playback need only be **monophonic**, *i.e.* it need only be able to play one note of audio at a time. Playback should be able to control:
  - the **pitch** of the audio — the audio pitch should be able to be playback at a given new pitch but at the same tempo.
  - the **tempo** of the audio — the audio tempo should be able to be playback at a given new tempo but preserving the same pitch.

Basic Coursework Requirements (Cont.)

Some more . . .

- Implement a basic form of **Granular Convolver**:
  - If the short-term Fourier transform on a very small sample window then aspects of granular synthesis can be implemented.
  - A basic description of short-term granular synthesis is discussed in the lectures. However, you expected to read around an develop a more in-depth and thorough understanding of granular synthesis as part of this coursework exercise.
  - The lecture notes reference some basic Granular Synthesis code (which you could build upon and enhance) and also demo some Granular synthesisers (publicly and commercially available)

Example commercial Granular Synthesiser:
http://www.newsonicarts.com/html/granite.php

Basic Coursework Requirements (Cont.)

Some more . . .

- Some **additional audio processing**:
  - You should implement some form of **volume shaping or envelope shaping** to control or modulate the basic sounds synthesised.
  - **The playback of audio DOES NOT have to operate in real time** – you can read the data in, generate/process the audio and then play the output, much like the demo MATLAB programs in the lectures/tutorials.
  - Your basic system need not have an advanced GUI (nothing as elaborate as *Iris* is expected!). You could satisfy all the requirements with a **minimal GUI** to display the audio/spectrogram and allow basic interactive editing.
    - You may use any GUI development environment to create your basic GUI.
  - Stay tuned for future lectures/labs and MATLAB code!

Additional Requirements

What do I need to do to get a High Mark:

Add **two** novel extensions or additional features.

Suggestions but feel free to be innovative!:

- Advanced granular convolution:
  - Applying multiple grains as convolvers in series or, even, parallel?
  - Looping through a short audio sample selecting (in series or randonly) grain convolvers.
  - Selection of a set number of grains as in the original granular convolver hardware.

Additional Requirements

What do I need to do to get a High Mark:

Add **two** novel extensions or additional features.

Suggestions but feel free to be innovative!:

- **Multi-layer sample playback** using more than one audio source, like IRIS.
- Advanced playback functionality to allow for **looping** of sections of audio, **reversing** sections of audio *etc.*.
- Some **further** digital audio effects.
- Provide a **user-friendly editor** for the audio and/or to enter musical data.
  - GUI elements to control the synthesis, filtering/modulation, effects and sound output may be provided.
- Provide support for **polyphonic output**.
- Provide **MIDI support** for data input.
- A **modular synthesis/effects pipeline**.

Demonstrating your system

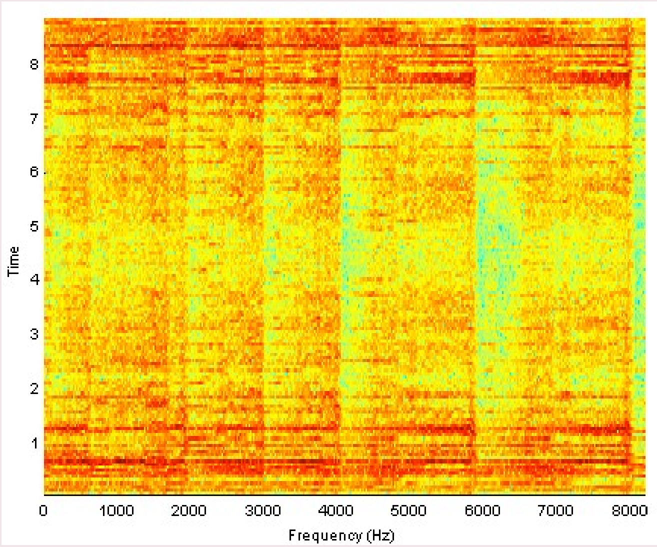
You must prove it all works (and makes a sound)

- You will be required to demonstrate your final system to the lab tutor in order to **verify** the extent to which the programs work according to specification. The tutor is only **guaranteed** to be available to sign at Multimedia Laboratory Sessions.
- For the demo you need only play a short number of sounds/notes. This will be enough to demonstrate that you can make interesting atmospheric and/or musical sounds!
- If you have any additional features in your system, it will be appropriate to demonstrate how they work and that they function accordingly.

Time-Frequency Analysis

Short-Time Fourier Transform (STFT):

STFTs as well as standard **Fourier transforms** (and other tools) are frequently used to analyse audio.



Visual information about an audio sample, **for example**:

- to locate the frequencies of specific noises (especially when used with greater frequency resolution)
- to find frequencies which may be more or less resonant in the space where the signal was recorded.

This representation can be used for **equalisation**, **tuning temporal shifting** and other audio effects.

Short-Time Fourier Transform (STFT)

Forward Short-Time Fourier Transform (STFT)

$$X(\tau, u) = \int_{-\infty}^{\infty} f(t) w(t - \tau) \mathbf{e}^{-\mathbf{2\pi itu}} \, dt.$$

where $w(t)$ is the window function, commonly a **Hann window** or **Gaussian window** **bell centered** around **zero**, and $f(t)$ is the signal to be transformed.

- $X(\tau, u)$ is essentially the Fourier Transform of $t(t)w(t - \tau)$, a complex function representing the phase and magnitude of the signal over time and frequency.

Inverse defined similarly

Short-Time Fourier Transform MATLAB Code

There are many implementations of the STFT.

Short-Time Fourier Transform MATLAB Code

- stft.m: Forward STFT
- istft.m: Inverse STFT

Part of a simple Phase Vocoder Toolbox (Useful: **More soon**)

```
function D = stft(x, f, w, h, sr)
% D = stft(X, F, W, H, SR)     Short-time Fourier transform.
% Returns some frames of short-term Fourier transform of x.  Each
% column of the result is one F-point fft (default 256); each
% successive frame is offset by H points (W/2) until X is exhausted.
% Data is hann-windowed at W pts (F), or rectangular if W=0, or
% with W if it is a vector.
```

See also MATLAB Central implementation with Pitch detection

Using stft.m and istft.m

Simple Example

```
load handel; % Get some audio

% stft parameters (can vary)
n = 512;
nhop = n/4;
Y = stft(y,n,n,nhop);

yback = istft(Y,n,n, nhop);
%should be same as y!
```

Spectrogram

Cooking your own Spectrogram, stft_spectrogram.m:

```
load handel; % Get some audio

% stft parameters (can vary)
n = 512;
nhop = n/4;
Y = stft(y,n,n,nhop);

% Make Spectrogram
specy = abs(Y)/n;

% set left-hand coordinate origin
imshow(flipud(255*specy));
colormap(hsv); %color display
```

Phase Vocoder

Phase Vocoder

An algorithm for timescale modification of audio.

- Basically we can stretch or compress the time-base of a spectrogram to change the temporal characteristics of a sound while retaining its short-time spectral characteristics;
  - **Narrowband spectrogram** — analysis window longer than a pitch cycle — preserving the pitch but change speed/tempo.
  - **Wideband spectrogram** — change pitch in a controlled way.

Phase Vocoder MATLAB Code

A Basic Phase Vocoder in MATLAB

- pvoc.m — the top-level routine
- pvsample.m — interpolate/reconstuct the new STFT on the modified timebase
  - `pvsample()` routine could also support arbitrary timebase variation (freezing, reversal, modulation) with simple modification — **useful for Coursework!**.

Requires: stft.m and istft.m

Original Code here: Phase Vocoder Toolbox

Phase Vocoder Tempo Change

Phase Vocoder Tempo Change Code, pvoc_speed.m

```
% Get some audio
load handel;

% Half Speed
yslow =pvoc(y,.5,1024);
% Compare original and resynthesis
sound(y,Fs);
sound(yslow,Fs);


% Twice as Fast
yfast =pvoc(y,2,1024);
% Compare original and resynthesis
sound(y,Fs);
sound(yfast,Fs);
```

Phase Vocoder Pitch Change

Phase Vocoder Pitch Change Code, pvoc_pitch.m

```
% Get some audio
load handel;

% Pitch up a Fifth
ypvoc =pvoc(y, 0.66666);
ypitch = resample(ypvoc,2,3); % NB: 0.666 = 2/3
sound(y,Fs);
sound(ypitch, Fs);
sound(y(1:length(ypitch)) + ypitch, Fs);

% Pitch up an octave
ypvoc =pvoc(y, 0.5);
ypitch = resample(ypvoc,1,2);
...


% Pitch down an octave
ypvoc =pvoc(y, 2);
ypitch = resample(ypvoc,2,1);
...
```

Phase Vocoder Pitch Change Explanation

Pitch Change Code Fragment, pvoc_pitch.m

```
% Pitch up a Fifth
ypvoc =pvoc(y, 0.66666);
ypitch = resample(ypvoc,2,3); % NB: 0.666 = 2/3
......
```

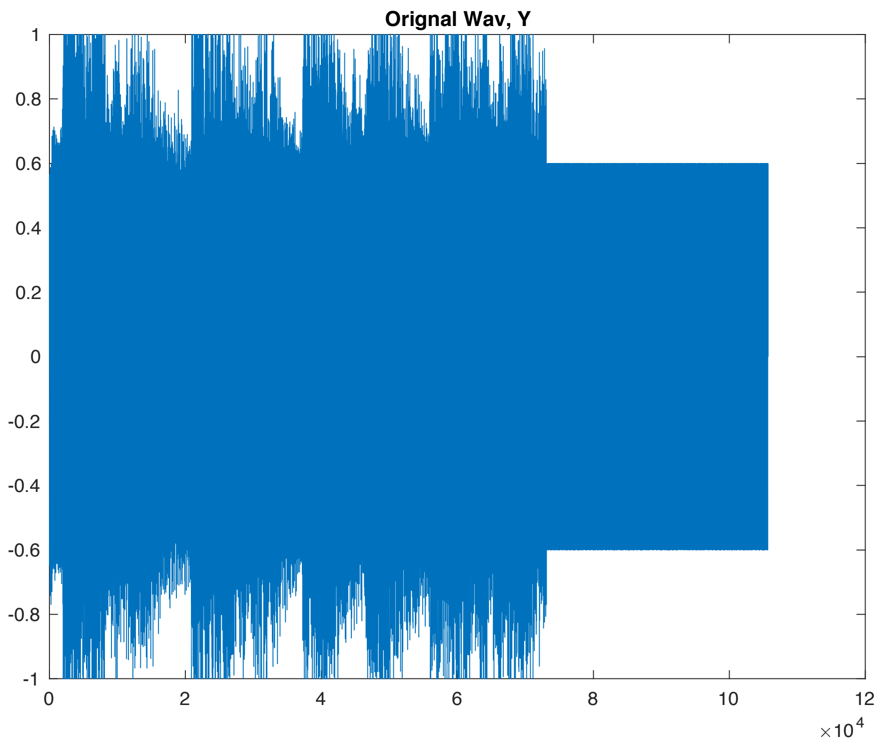
Pitch Change Explanation:

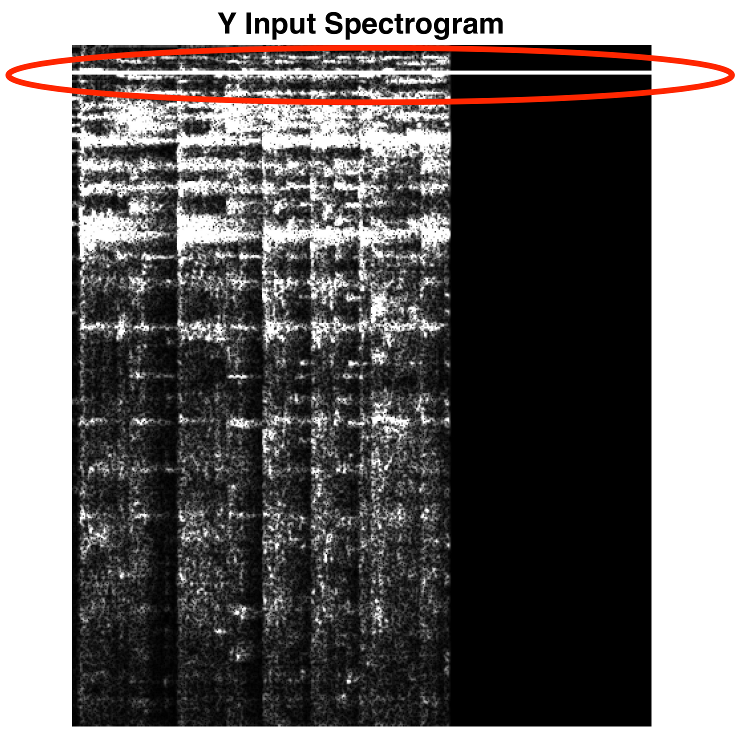
- Extending/compress duration with the phase vocoder:
  - `pvoc(y, 0.66666)` : Need to know appropriate fraction of pitch shift
  - Look up here: *Meantone intervals*. (**Octave shifts are obvious**.)
- Resampling to the original length
  - `resample(ypvoc,2,3)`:
  - **Note**: New sample length won't be same as sample of original pitch.

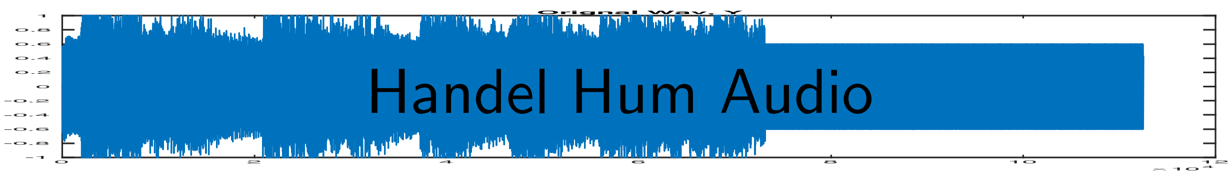
Practical Example (C/W Hint): Hum Removal

We have a piece of audio with some low frequency hum:

- Compute its spectrogram (via STFT):



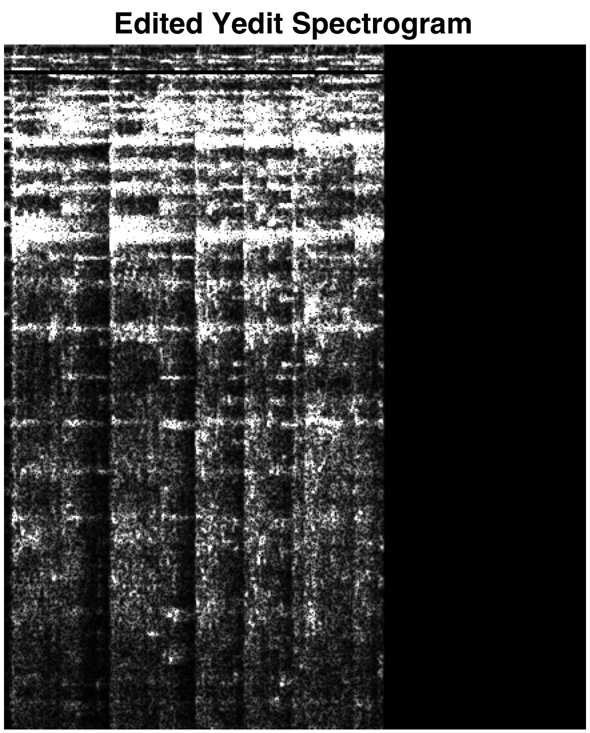


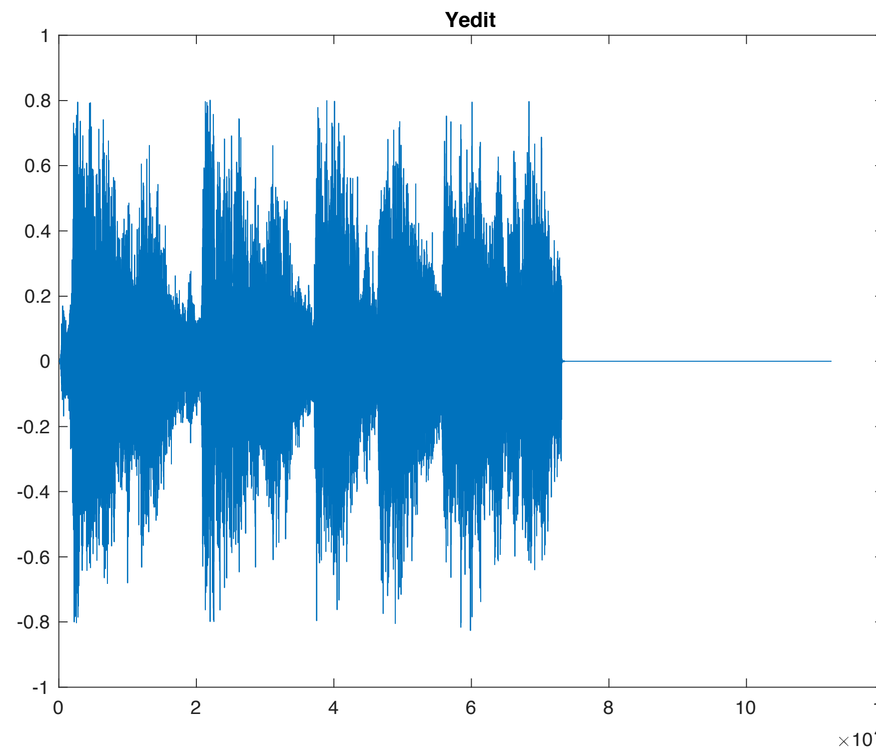


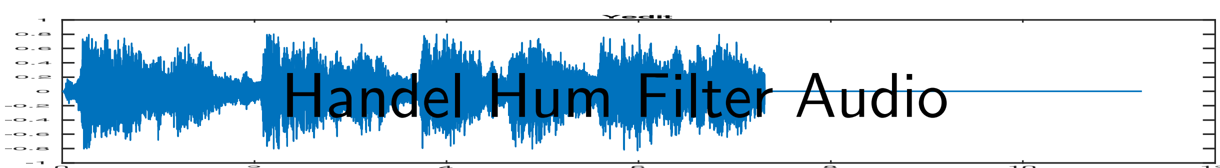
Hum Removal (Cont.)

Isolate frequency range of the hum and remove it (band pass filter)

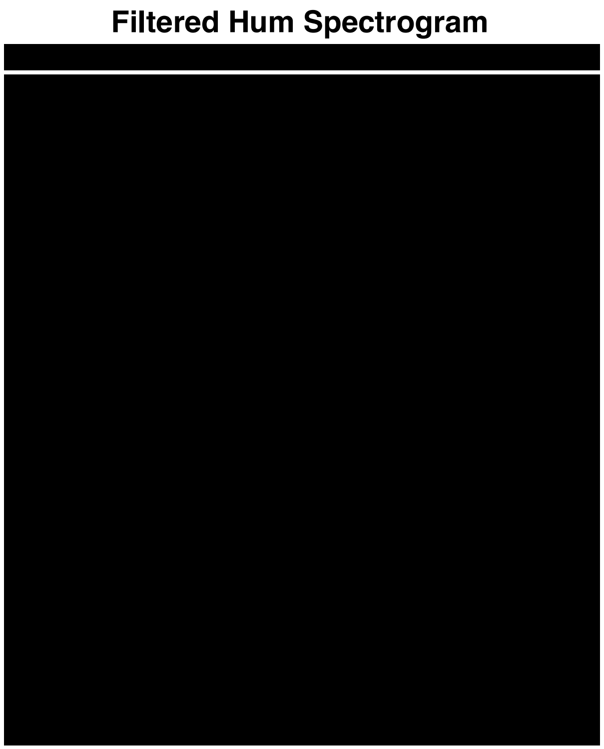
- Scrub out values in Spectrogram.
- Inverse the Spectrogram (ISTFT)

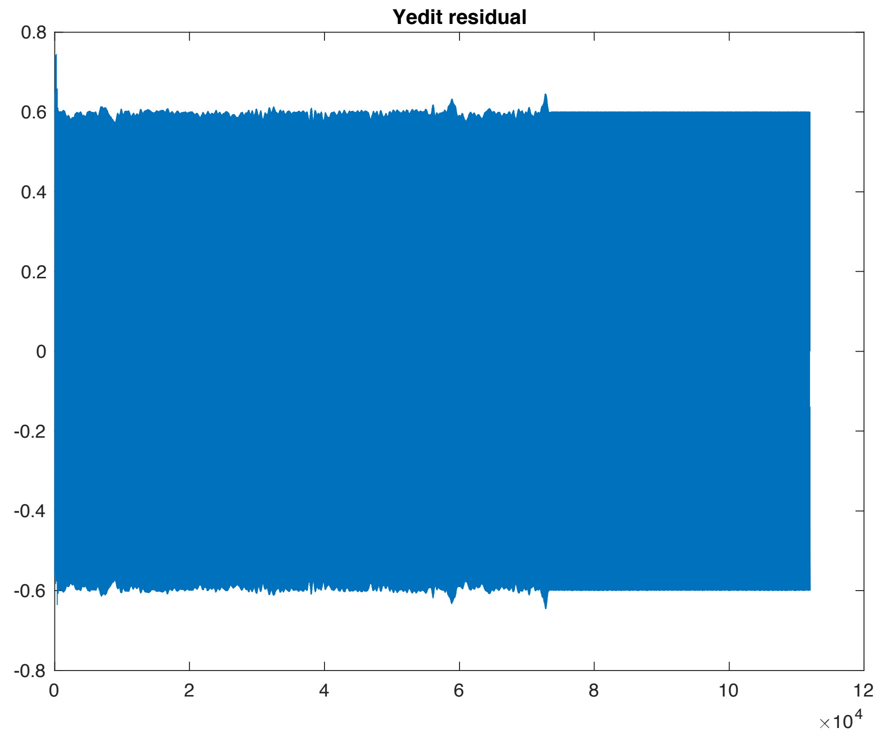


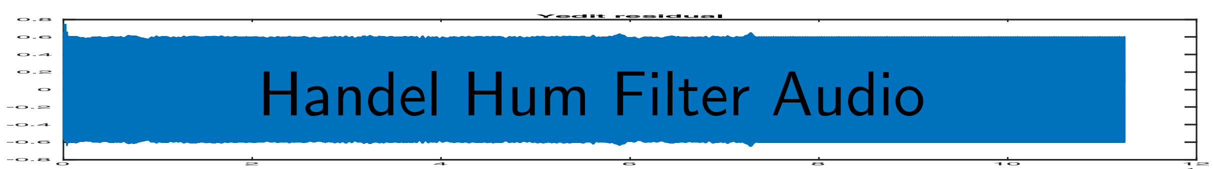




The Isolated Hum:







Hum Removal: MATLAB Code Example

stft_spectra_edit_eg.m:

```
[y,Fs] = audioread('handel_hum.wav'); % Get some audio

reply = input('Press return to play sound? ');
figure(1)
plot(y);
title('Orignal Wav, Y');

sound(y, Fs)

% stft parameters (can vary)
n = 1024;
nhop = n/4;
Y = stft(y,n,n,nhop);

[yn, ym] = size(Y);
```

Hum Removal: MATLAB Code Example (Cont.)

stft_spectra_edit_eg.m (Cont.):

```
figure(2)
imshow(255*abs(Y)/n)
colormap('hsv')
title('Y Input Spectrogram')

% Edit example cut a chunk out of Y Hum in
% Rows 20:22 around 160HZ
%                  (= 20*Fs (= 8192)/ n (=1024))

Yedit = Y;
Yresidual = zeros(yn,ym);

Yresidual(20:22,:) = Y(20:22,:)
Yedit(20:22,:) = 0;
```

Hum Removal: MATLAB Code Example (Cont.)

stft_spectra_edit_eg.m (Cont.):

```
reply = input('Press return to play edited sound? ');

figure(3)
imshow(255*abs(Yedit)/n)
colormap('hsv')
title('Edited Yedit Spectrogram')


figure(4)
imshow(255*abs(Yresidual)/n)
colormap('hsv')
title('Filtered Hum Spectrogram')
```

Hum Removal: MATLAB Code Example (Cont.)

stft_spectra_edit_eg.m (Cont.):

```
% Remake sound
yedit = istft(Yedit,n,n, nhop);
sound(yedit,Fs);

reply = input('Press return residual sound?');
figure(5)
plot(yedit)
title('Yedit')

figure(6)
plot(yedit(1:length(yedit)) - y(1:length(yedit))')
title('Yedit residual')

% Play residual
sound(yedit(1:length(yedit))-y(1:length(yedit))',Fs);
```

References:

Useful web links:

- Granular Convolution: youtube video
- Izotope Iris main web page: *https://www.izotope.com/en/products/create-and-design/iris.html*
- Izotope Iris help: *http://help.izotope.com/docs/izotope-iris-help.pdf*
- Izotope Tutorials: *Izotope Video Tutorials*
- A **demo** version of Iris is freely available: *Iris trial Version*
- *http://www.jyu.fi/musica/miditoolbox/* — MATLAB MidiToolbox. Reads/Writes Midi files, converts midi between note number, musical notes/pitches and frequencies.
  See also *http://users.cs.cf.ac.uk/Dave.Marshall/Multimedia/exercises_BSC/* for local copy of the MidiToolbox.
- *http://labrosa.ee.columbia.edu/matlab/* — MATLAB Audio Processing Examples
- http://www.phy.mtu.edu/ suits/notefreqs.html — musical pitches frequency relationship.
- *http://en.wikipedia.org/wiki/List_of_meantone_intervals*: List of pitch/tone intervals as ratios. (Useful for the Phase Vocoder)