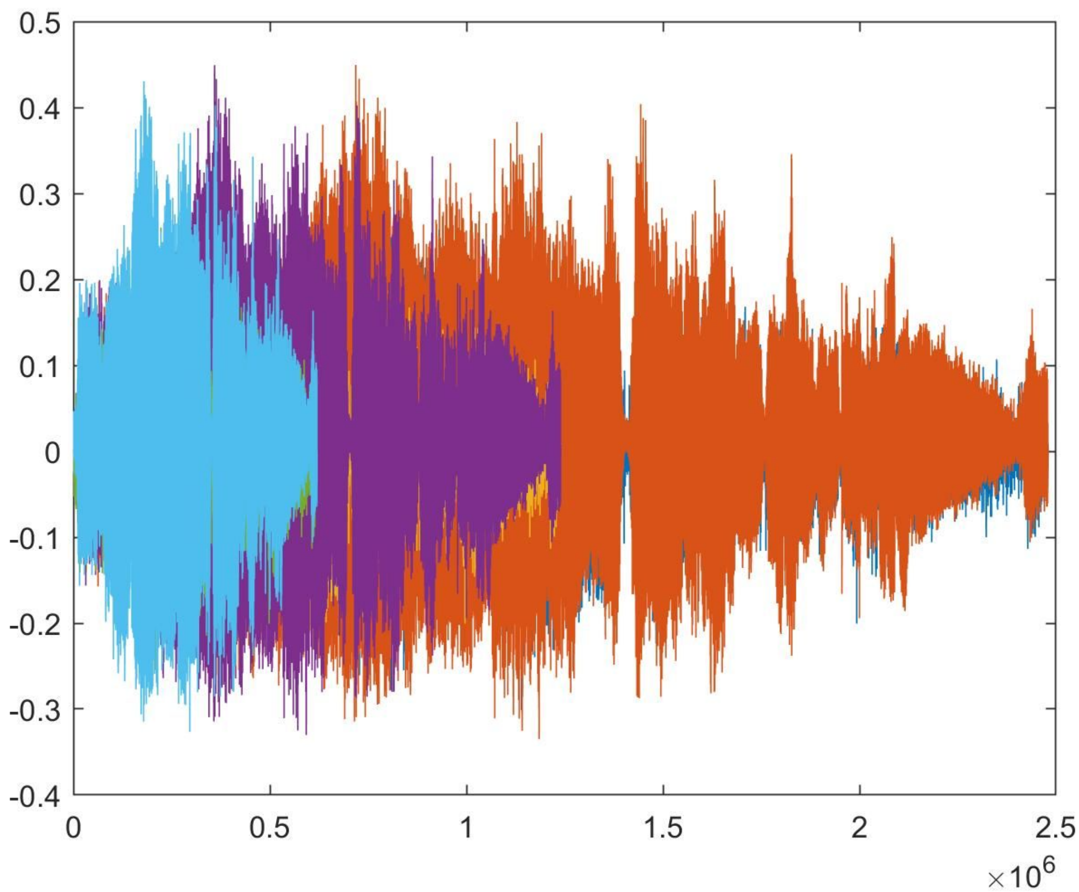
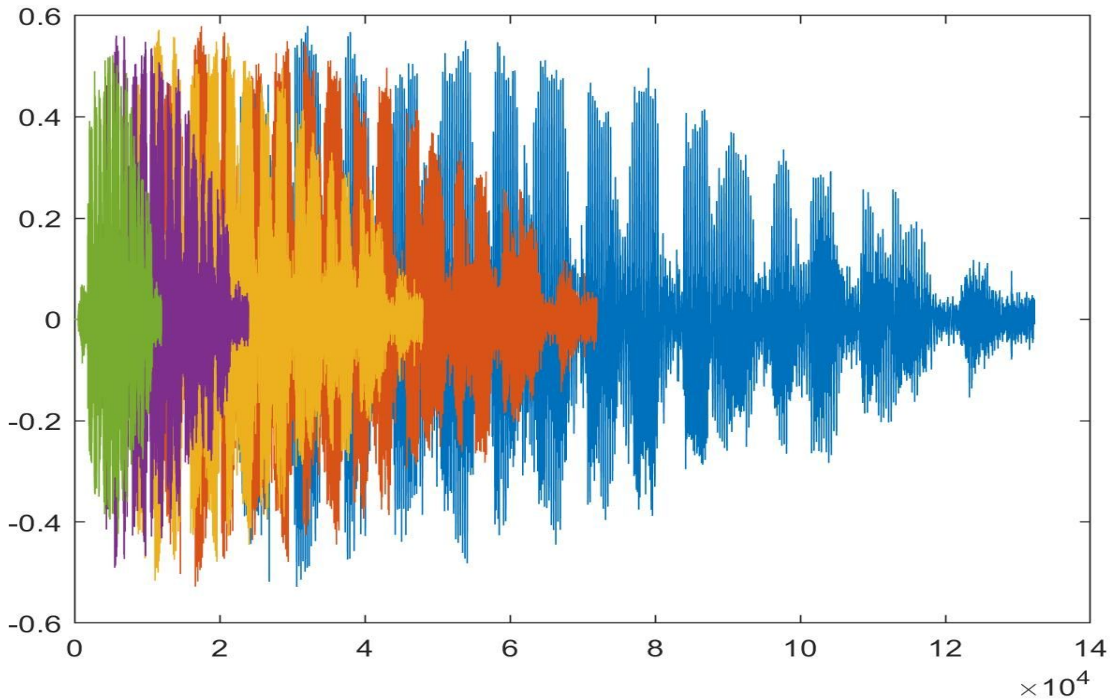
Assignment-1 Report 20171037

1. The audio file is read. It is then resampled to half and double frequencies, and then plotted.



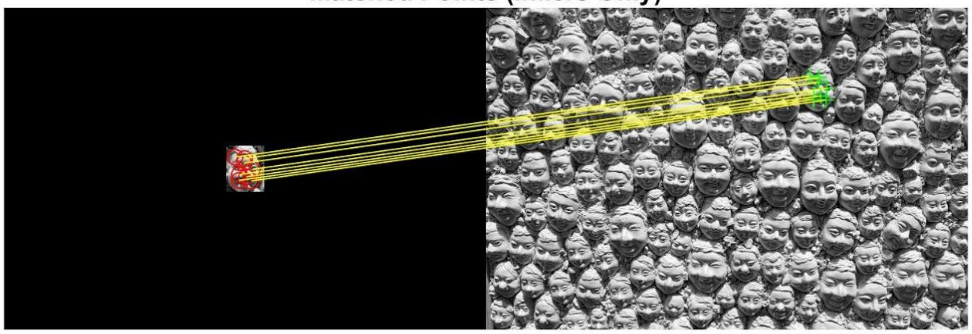
2. An audiorecorder object is created and then a voice is recorded. It is then resampled at various frequencies and then plotted along with the original signal. The various audio samples from different environments are read, and then they I do their convolution with the 8KHz sample.



3. The reference image and the target report are read. Then the feature points in both the images are detected. Then the strongest feature points found in the reference and target images are visualised. Then the feature descriptors at the interest points in both images are extracted, and then their features are matched using their descriptors. Then the matching point pairs with the outliers removed are displayed.

Output -->

Matched Points (Inliers Only)



Since the F2.jpg image is noisy, the strongest feature points matching may be less due to distortion due to noise. Thus, for greater accuracy, we can determine more number of feature points in the images and match the correlations. Other way to improve accuracy is to remove noise from the image by using functions like imnoise or through fft. Certain filters like gaussian and averaging filters are appropriate for this purpose. Since each pixel gets set to the average of the pixels of its neighbourhood, local variations caused by the grain noise are reduced.

4. Nearest Neighbour Interpolation :

First, to get the new row and column sizes the old row and column sizes are multiplied by the scale factor. This result is rounded down to the nearest integer with ‘floor’. If the scale factor is less than 1 we could end up with a weird case of one of the size values being 0, which is why the call to ‘max’ is there to replace anything less than 1 with 1.

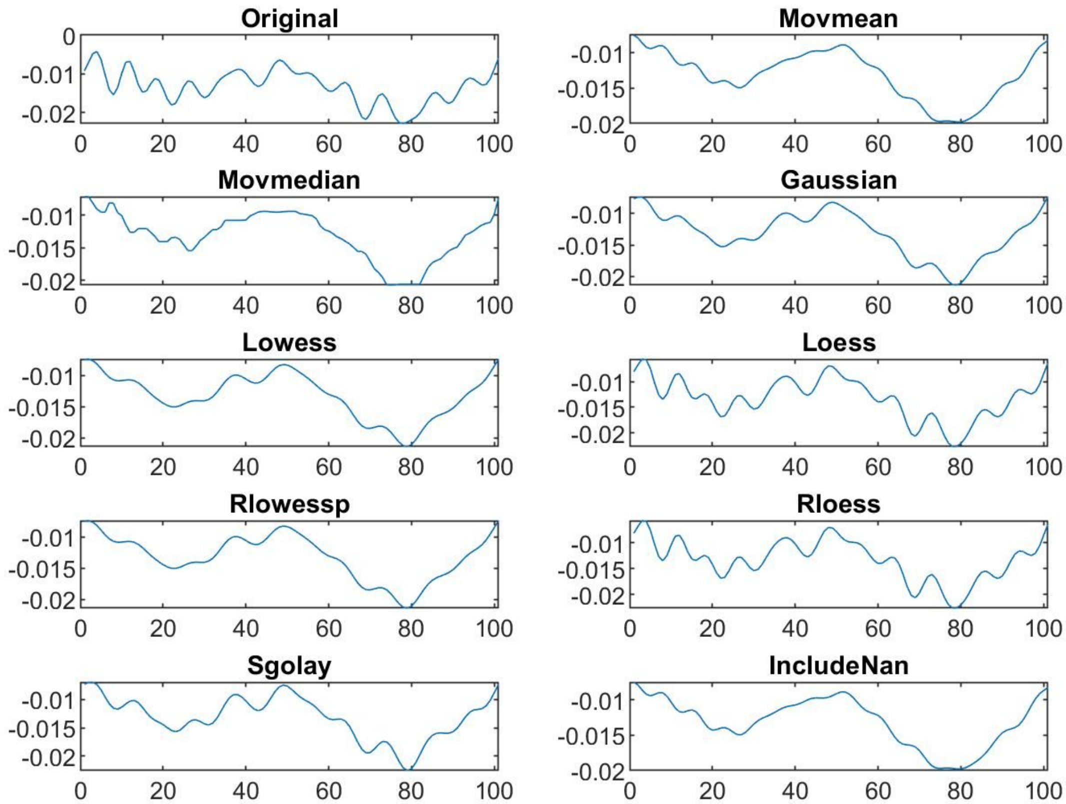
Next, a new set of indices is computed for both the rows and columns. First, a set of indices for the upsampled image is computed. Each image pixel is considered as having a given width, such that pixel 1 spans from 0 to 1, pixel 2 spans from 1 to 2, etc.. The "coordinate" of the pixel is thus treated as the center, which is why 0.5 is subtracted from the indices. These coordinates are then divided by the scale factor to give a set of pixel-center coordinates for the original image, which then have 0.5 added to them and are rounded off to get a set of integer indices for the original image. The call to ‘min’ ensures that none of these indices are larger than the original image size.

Finally, the new upsampled image is created by simply indexing into the original image.

Bilinear Interpolation:

Bilinear interpolation uses a first-order, 2-D polynomial – a plane – to estimate a subpixel value.

5. The given audio file is read, and then the ‘smoothdata’ function is applied on it along with various filter parameters. I have plotted then output signal from 100 to 200.
Among the various filters, ‘Sgolay’ works the best, because it works best for quadratic-like signals, and the given signal is quadratic-like. The output is smooth and free from noise.


Original
Movmean
Movmedian
Gaussian
Lowess
Loess
Rlowessp
Rloess
Sgolay
IncludeNan