University of Sheffield

COM3502-4502-6502
Speech Processing

Main Programming Assignment

Instructions

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November 1, 2018

Contents

1 Overview (please read carefully)

By now you should have completed the '`Pure Data`' (`Pd`) introductory programming exercise, and submitted your answers via MOLE. If not, then you are strongly advised to do so before proceeding to the main assignment described below. In any event, it may be advisable to review the introductory programming exercise, especially Sections 7 and 8 which cover working with real-time audio and speech.

Note: *This programming assignment is worth 45% of the overall course mark.*

You may work in pairs on this assignment.

Please team-up with someone else within your COM3502 or 4502-6502 cohort. It is important that you pair-up within these cohorts as COM4502-COM6502 students have additional assessments. A COM4502 student may pair with a COM6502 student. If you cannot find a partner, please contact the lab demonstrators as soon as possible.

You are permitted to re-use any of the `Pd` examples provided during the course.

1.1 Logistics

You are free to complete this assignment in your own time. However, 'drop-in' sessions are available in Regent Court room G22 Blue on Mondays from 12:00 to 12:50 (starting in week 7). During these times it will be possible to gain feedback, advice and guidance from a post-graduate demonstrator. If you require help outside of the timetabled arrangements, please contact one of the post-graduate demonstrators by e-mail (contact details are available on MOLE).

1.2 Provided Materials

In addition to these instructions, you have been provided with a `.zip` file containing a number of items[1] that you will need for this assignment. In particular, you have been supplied with a LaTeX[2] template - `YourNames.tex` - which will you use to compile your response sheet.

LaTeX (pronounced [lɑːtek] or [leɪtek]) is a *free* document preparation system for high-quality typesetting, and having a working knowledge of LaTeX is a key skill for any computer scientist[3]. LaTeX is very easy to learn[4], and there are many resources available on the web, e.g. `https://www.sharelatex.com/learn/Learn_LaTeX_in_30_minutes`.

[1] The provided `screenshot.jpg` is simply a placeholder which you will replace with your own images.

[2] LaTeX includes features designed for the production of technical and scientific documentation, so it is the de-facto standard for the publication of scientific/technical papers and reports. Unlike WYSIWYG ("*What You See Is What You Get*") word processors (such as *Microsoft Word* or *OpenOffice*), LaTeX is based on plain-text source files (containing html-style markup) which are compiled into a typeset document. This separation of 'content' from 'style' facilitates the production of very professional scientific documents in terms of their consistency, readability and design. You can find out more about LaTeX here: `https://www.latex-project.org`.

[3] The Department strongly recommends that you consider using LaTeX for your Dissertation; Prof. Moore can provide a template on request.

[4] Students who took **COM2005-3005 Bio-Inspired Computing** in 2017/18 should be familiar with LaTeX already.

LaTeX distributions are available for Linux, Mac OSX and Windows (see: `https://www.latex-project.org/get/`). However, for this assignment, you might like to use an online collaborative environment such as `Overleaf` (`https://www.overleaf.com`).

★ **Step 0**: Before you start working with LaTeX, edit the filename of the provided `.tex` file by replacing `YourNames` with your names.

Your edited `.tex` file should compile to produce a `.pdf` document with your names on the front cover, followed by your responses to a number of questions. You will be submitting the `.pdf` file when you have completed the assignment, *not* the `.tex` file.

1.3 Hand-In Procedure

Once you have completed the assignment, one member of each pair should submit a `.zip` file (via MOLE) containing your response sheet (in `.pdf` format) and the requested `Pd` source files. **Do not** submit your LaTeX source files. The `.zip` filename should be of the form `YourNames.zip`.

Please make sure that your names are shown correctly on the front page of your response sheet (as explained in the previous section).

Standard departmental penalties apply for late hand-in[5] and plagiarism[6].

Feedback (including provisional marks) will be provided via MOLE within three working weeks of the hand-in date.

The deadline for handing-in this assignment (via MOLE) is ...

Midnight Friday 7th December 2018 (week 11)

[5] `http://www.dcs.shef.ac.uk/intranet/teaching/public/assessment/latehandin.html`
[6] `http://www.dcs.shef.ac.uk/intranet/teaching/public/assessment/plagiarism.html`

2 Part-I: Voice Manipulation

As you saw in Lecture 1, speech processing has many applications across a wide variety of market sectors. One area of particular interest is the creation of appropriate voices for fictional characters in television, cinema, games and related areas of edutainment[7]. Such robot, alien or cartoon voices are often achieved by manipulating the speech of a voice actor (either in real-time or in post-production[8]) - a process that is sometimes referred to as 'Voice FX' or 'Vocal FX'.

The aim of this assignment is to implement a range of speech processing algorithms for *real-time* voice manipulation. The final outcome will be a 'Voice Changer' that can be configured to perform a variety of different manipulations on your own voice. Part-I involves creating the core components in `Pd`, and Part-II brings them together into a single `[VoiceChanger]` application.

The assignment has been structured such that the early stages are relatively straightforward, but the final steps are quite challenging. Please work through the following sections in the order given, providing responses the relevant questions as you come to them[9].

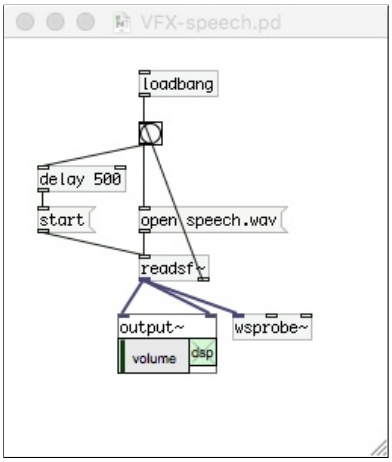
Note: *In Part-II you will reuse many of the components you create in Part-I. Therefore, you are strongly advised to adopt professional working practices, e.g. saving versioned copies of each piece of software that you develop.*

Note: *Some of the steps below require you to listen critically to sounds generated by your software. It is thus advisable for you to use headphones and/or work in a quiet environment. Take care not to annoy other people with your sound output.*

2.1 Continuous Speech Input

Your first task is to program a `Pd` patch that can provide a continuous stream of real-time speech which can be fed into the various manipulation techniques that follow.

★ **Step 1**: Create the following patch ...



[7] Wilson, S., & Moore, R. K. (2017). Robot, alien and cartoon voices: implications for speech-enabled systems. In *1st Int. Workshop on Vocal Interactivity in-and-between Humans, Animals and Robots (VIHAR-2017)*. Skovde, Sweden.

[8] Rose, J. (2012). *Audio Postproduction for Film and Video*. Taylor & Francis.

[9] by editing your `.tex` file

Note: *Recall from the introductory programming exercise that the audio output control will appear when you create* `[output~]` *object.*

★ **Step 2**: Click on the `bang` at the top of the patch, increase the volume slider on `[output~]` and verify that you can hear the speech[10] on a continuous loop.

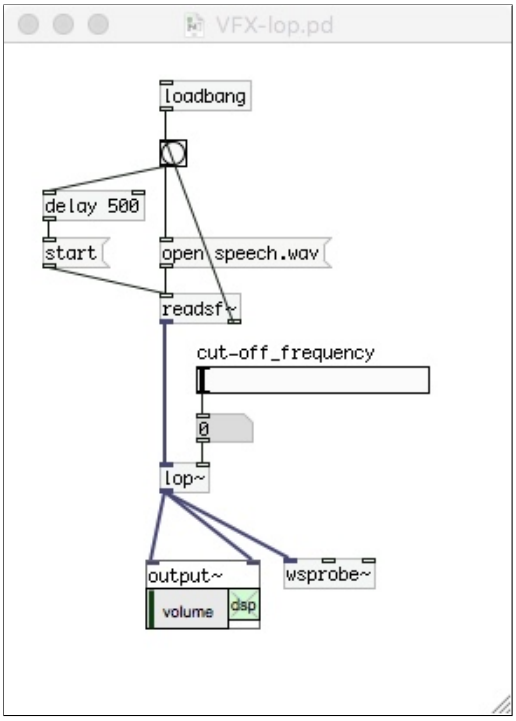
★ **Step 3**: Now click on `[wsprobe~]`. This should open a separate patch that displays the waveform and spectrum in real-time (as you have seen several times in the lectures). Experiment with the various options provided in the `[wsprobe~]` GUI, and take particular notice of the differences between voiced and voiceless sounds.

QUESTION 1 ***(worth up to 5 marks)***
Provide a screenshot of `[wsprobe~]` for a typical voiced sound, and explain the features in the waveform and spectrum that distinguish it from an unvoiced sound. *Hint: use the 'snapshot' feature in* `[wsprobe~]` *to obtain a static display.*

2.2 Filtering in the Time-Domain

One of the most basic manipulations in Voice FX is to 'filter' a signal in order to increase or decrease energy at particular frequencies. The simplest types of filter are 'low-pass', 'high-pass' and 'band-pass', and these may be implemented in `Pd` using the `[lop~]`, `[hip~]` and `[bp~]` objects respectively.

★ **Step 4**: Modify your patch to include a low-pass filter with a 'cut-off frequency' controlled by a horizontal slider covering the range 0 to 10,000 Hz as shown here[11] ...



[10] The provided `speech.wav` file is the same as the one used in the Lectures (and as provided in `com3502-4502-6502_pd-examples.zip` available on MOLE).

[11] Recall that the slider's range can be set by right-clicking and adjusting its 'properties'.

voiced 的频率分布的集中在低频，用lowpass几乎没有任何影响。
Vowels 都是 voiced 的。
consonants 中的摩擦音比如f等是分布在高频的，通过low-pass filter后就消失了。
通过过滤器之后，音量减小了

★ **Step 5**: Experiment with different values for the cut-off frequency, and observe the consequences for the output sound as well as the waveform and spectrum displays.

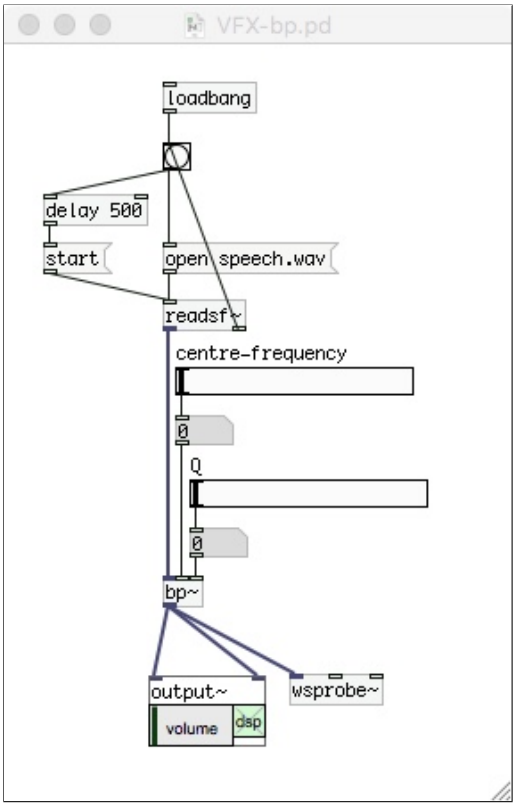
QUESTION 2 ***(worth up to 5 marks)***
Which sounds are most affected when the low-pass cut-off frequency is set to around 500 Hz - vowels or consonants - and why?

★ **Step 6**: Edit the `[lop~]` object to become a `[hip~]` object, and again experiment with different values for the cut-off frequency.

QUESTION 3 ***(worth up to 5 marks)***
How is it that the speech is still quite intelligible when the high-pass cut-off frequency is set to 10 kHz?

As you have seen, these simple low-pass and high-pass filters have only one control parameter - their cut-off frequency. A simple band-pass filter has two parameters - its 'centre frequency' and its 'Q factor'. The Q of a band-pass filter is defined as $Q = 1/B$, where B is the bandwidth (in Hz). Hence a high-Q filter has a narrow bandwidth and *vice versa*.

★ **Step 7**: Edit the `[hip~]` object to become a `[bp~]` object, and add another horizontal slider covering the range 0 to 100 as shown here ...



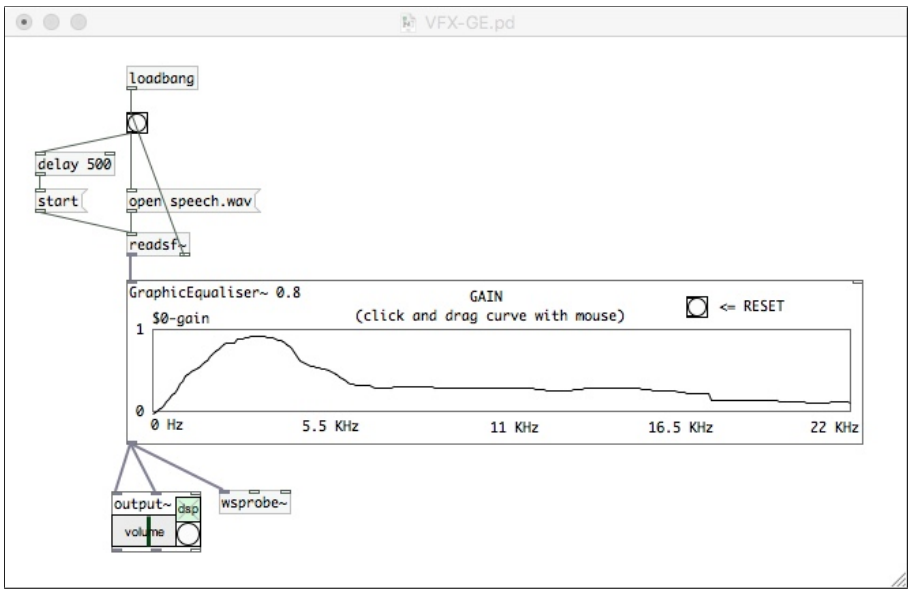
★ **Step 8**: Experiment with different values for the centre frequency and Q.

2.3 Filtering in the Frequency-Domain

The [lop~], [hip~] and [bp~] objects operate directly on the speech waveform. Hence, they are examples of 'time-domain' signal processing. An alternative to time-domain processing is to process signals in the 'frequency-domain', i.e. by modifying the spectrum rather than the waveform.

A classic example of filtering in the frequency-domain is a 'graphic equaliser'. This is a common component of a high-fidelity (HiFi) audio system, and it allows the spectrum to be shaped by setting gains across a range of frequencies. You have been provided with a Pd 'abstraction'[12] [GraphicEqualiser~] that can perform this function.

★ **Step 9**: Remove your [bp~] object and its slider controls, and insert the provided [GraphicEqualiser~]. Connect it between [readsf~] and [output~], and experiment with different filter profiles by clicking on the graph (in 'run mode') and dragging the curve into different shapes. Note how you can make low-pass, high-pass and band-pass filters (and many more), simply by drawing the appropriate shapes, as shown here ...



★ **Step 10**: Open the [GraphicEqualiser~] abstraction to see how it works (by right-clicking on the object, and selecting open). Take particular note of how the GUI objects (the graph and reset button) are made to appear in the parent patch using the '**graph-on-parent**' option in the sub-patch properties[13].

QUESTION 4 ***(worth up to 5 marks)***
COM3502-4502-6502: The [GraphicEqualiser~] object uses an FFT internally; what does FFT stand for and what does an FFT do?
COM4502-6502 ONLY: What is a DFT and how is it different from an FFT?

★ **Step 11**: Use [GraphicEqualiser~] to simulate the effect of a land-line telephone by eliminating all energy below 300 Hz and above 3,400 Hz.

[12] Refer back to the introductory programming exercise if you have forgotten what an abstraction is.
[13] Note that you can also look inside [output~] or [wsprobe~] to see the same 'graph-on-parent' functionality.

FFT: fast Fourier transform
converts a signal from its original domain (often time or space) to a representation in the frequency domain.

2.4 Low Frequency Oscillator

Many 'Voice FX' are achieved by modifying some characteristic of the speech using a low frequency oscillator or 'LFO'. LFOs typically have two controls: `speed` (which is specified by the frequency in Hertz) and `depth` (which specifies the magnitude of the effect). Your `[VoiceChanger]` will require several LFOs, so it makes sense to implement one as a `Pd` 'abstraction'.

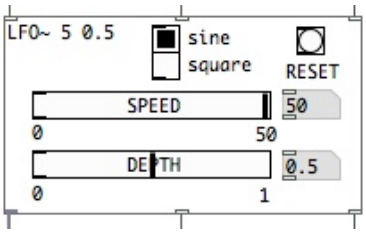
★ **Step 12**: Using the skills you acquired in the introductory programming exercise, create a `Pd` abstraction `[LFO~]` that is capable of generating an audio sine/square wave between 0 and 50 Hertz (e.g. using the `[osc~]` object). The `speed` and `depth` of the LFO should be controllable by suitable sliders, and they should appear in the object using the 'graph-on-parent' approach.

★ **Step 13**: Add an option to select either 'sine' or 'square' wave output using a `Vradio` button. *Hint: to obtain a square wave, multiply the output of the `[osc~]` object by a very large number and pass the result through `[clip~ -1 1]` before being multiplied by `depth`.*

Additional features that will be useful later in the assignment are:

- `[number]` GUI objects for `speed` and `depth`
- a `reset` function
- external inputs for `speed`, `depth` and `reset`
- external outputs for `speed` and `depth`
- an ability to handle creation arguments for `speed` and `depth` (also linked to `reset`)
- an ability to reset the phase of the oscillator when `speed` = 0 in order to ensure consistent output when initialising/resetting

Your resulting `[LFO~]` abstraction should look something like this (note the use of `speed` = 5 and `depth` = 0.5 as creation arguments) ...



Although your `[LFO~]` object outputs audio, you are unlikely to be able to hear it as the frequency is so low. However, you can check that it is functioning correctly by connecting it to `[wsprobe~]` and/or by using the appropriate `number` GUI object to wind the `speed` up to an audible frequency.

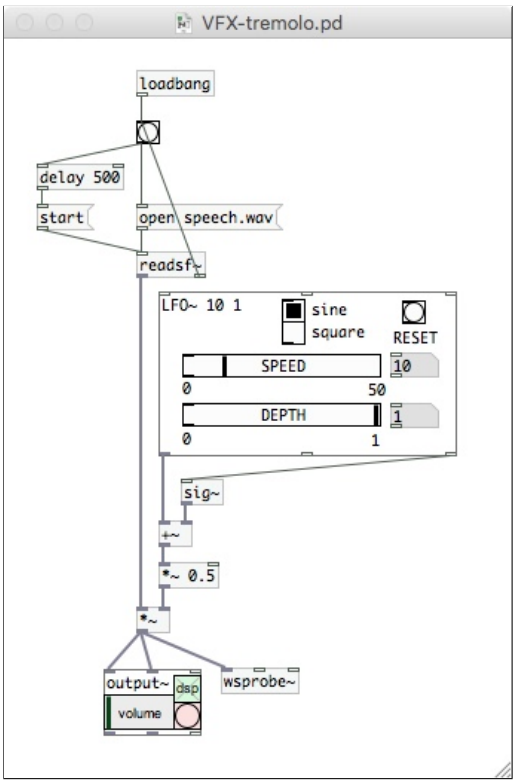
★ **Step 14**: Test your `[LFO~]` object by creating an `[LFO~-help]` object.

QUESTION 5 ***(worth up to 10 marks)***
With `speed = 50` and `depth = 0.5`, what are the minimum and maximum amplitudes of your LFO output, and how do they vary with changes in these two settings? Also, please provide two screenshots: (a) your `[LFO~-help]` object and (b) the internal structure of your `[LFO~]` object.

2.5 Amplitude Modulation: *Tremolo*

'Tremolo' is one of the most basic voice manipulations that makes use of an LFO. In this effect, the amplitude of a speech signal is modulated, i.e. the speech waveform is multiplied by a variable gain that ranges between 0 and 1. Your LFO outputs an audio signal between `-depth` and `+depth`. So, in order to modulate the amplitude of the speech correctly, the output of the LFO has to be scaled appropriately - in this case, by adding `depth` (e.g. using the audio math object `[+~]` connected to the `depth` output of your LFO) and dividing the result by `2` (e.g. using the audio math object `[*~ 0.5]`).

★ **Step 15**: Implement 'tremolo' using your `[LFO~]` object using the principles given above. The resulting program should look something like this (note the use of a `[sig~]` object to convert numerical data to audio data) ...



★ **Step 16**: Experiment with different settings for `speed` and `depth`. In particular, note that a square wave with `speed` between 3 and 4 Hz (and `depth` = 1) has a very destructive effect on the intelligibility of the output. This is because 3-4 Hz corresponds to the typical syllabic rate of speech.

2.6 Ring Modulation

Another basic effect is to *multiply* the speech signal by the output of an LFO. This is known as 'ring modulation'.

★ **Step 17**: Modify your implementation of 'tremolo' by removing the two scaling objects (`[+~]` and `[*~ 0.5]`) and connecting the audio output of your `[LFO~]` directly to the `[*~]` object.

★ **Step 18**: Experiment with different settings for `speed` and `depth`, and note how the timbre of the resulting sound is subtly different from 'tremolo'.

Note: *In the BBC TV series Dr. Who, the voices of the alien Daleks*[14] *are generated by a ring modulator with an LFO set to around 30 Hz. The voice actors also spoke in a stilted monotonic tone to enhance the effect. You can try this yourself by adding a 'live speech input' option to your code (i.e. using* `[adc~]` *and some means to toggle between the prerecorded and live input)*[15].

QUESTION 6 ***(worth up to 5 marks)***
In your own words[16], why is this effect known as 'ring modulation'?

2.7 Frequency Shifting

Many Vocal FX are the result of altering the frequencies present, e.g. changing the pitch of a voice. There are many algorithms for frequency shifting, but you have already implemented an approximate solution with your ring modulator.

★ **Step 19**: Use the `number` object in your `[LFO~]` GUI to wind up the `speed` of the LFO in your ring modulator to around 300 Hz. If all is working correctly, as you raise the modulation frequency, you will hear that the frequencies in the speech seem to be both increasing *and* decreasing. Indeed, if you listen carefully at a fixed `speed` of around 300 Hz, you may be able to detect two voices speaking simultaneously at different frequencies.

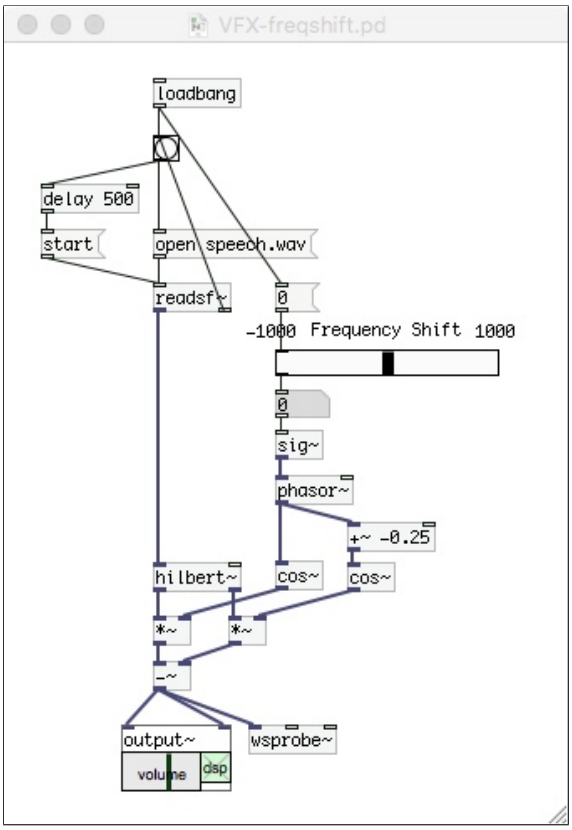
The reason for this result is that multiplying one signal by another - known as 'heterodyning' - produces an output which consists of the sums and differences of the frequencies of the two input signals. This means that the frequencies in the original speech are both raised and lowered by an amount corresponding to the `speed` of the LFO - hence the strange mixture of higher pitched and lower pitched voices in the output.

The sum and difference components are known as the upper and lower 'sidebands'. So, to avoid the distortion caused by having both sidebands present in the output, we need to remove one of them, i.e. we only want one sideband. This can be achieved by creating two versions of the original signal that differ in phase by 90°, then modulating the two versions with two heterodynes that also differ in phase by 90°. These phase differences are such that, when the two out-of-phase heterodyned signals are subtracted, the lower sideband is cancelled out. This process is known as 'single-sideband modulation' or 'SSB'[17].

QUESTION 7 ***(worth up to 5 marks)***
Why is SSB commonly used in long-distance radio voice communications?

Splitting a signal into two with a 90° phase difference between them can be achieved using the 'Hilbert Transform'. Hence the following patch shows the `Pd` object `[Hilbert~]` being used to create a frequency shifter based on SSB modulation ...

[14] `https://en.wikipedia.org/wiki/Dalek`
[15] You will need this feature later, in any case.
[16] I.e. do not plagiarise from Wikipedia.
[17] SSB is commonly used in long-distance radio voice communications, especially by amateur radio enthusiasts.



★ **Step 20**: Implement the above patch, and experiment with different values for the frequency shift. Note that the frequency-shifted output now sounds much cleaner that the output of the ring modulator (i.e. only one voice). Also note that the pitch of the voice can be shifted down as well as up.

QUESTION 8 (*worth up to 5 marks*)
COM3502-4502-6502: Why can the voice be shifted up in frequency much further than it can be shifted down in frequency before it becomes severely distorted? /emphHint: look at `[wsprobe~]`.
COM4502-6502 ONLY: Your frequency shifter changes all the frequencies present in an input signal. How might it be possible to change the pitch of a voice *without* altering the formant frequencies?

A classic 'robotic' voice can be achieved by simply adding frequency-shifted speech back to the unprocessed original. This effect is known as 'harmony'. However, rather than simply adding the signals in equal amounts, your final `[VoiceChanger]` will benefit from a more general purpose approach.

★ **Step 21**: Implement a 'mixer' that adds the original speech with the manipulated speech in different proportions. Use a slider that has 100% original at one end, 100% manipulated at the other end and 50-50 in the middle.

★ **Step 22**: With your mixer at the 50-50 setting, experiment with different frequency shifts in order to produce the best robotic sounding output.

2.8 Frequency Modulation: *Vibrato*

Now that you have the ability to shift the frequencies in a speech signal, it is very easy to implement another common voice manipulation technique - 'vibrato'. All that is required is for the frequency shifter to be controlled by the output of an LFO.

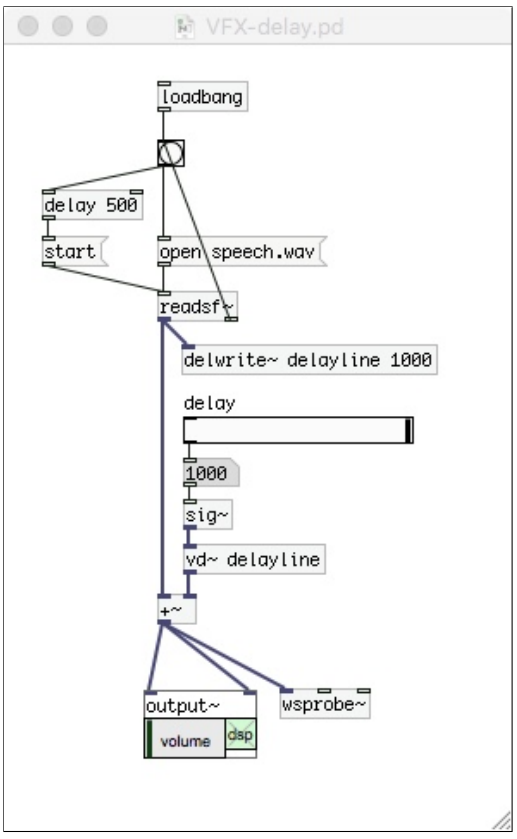
★ **Step 23**: Implement 'vibrato' by connecting an LFO to your frequency shifter, and experiment with different values for `speed` and `depth`. Note that the LFO output will need to be scaled to provide an appropriate frequency shift range and then added to the output of the frequency shift slider.

Note: *With* `speech.wav` *as input, it is possible to simulate the voice of Gollum from the Lord of the Rings by setting LFO = sine wave,* `speed = 50 Hz` *and* `depth = 350`*.*

2.9 Time Delay Effects: *Echo, Comb Filtering and Flanger*

Many interesting voice FX can be achieved by delaying the signal and recombining it with itself. `[Pd]` provides the possibility of writing to and reading from audio 'delay lines' using the objects `[delwrite~]` and `[delread~]`. However, for full flexibility in your `[VoiceChanger]` application, you need to be able to vary the time at which audio data is read out of the delay line. So, rather than using `[delread~]` it is better to use `[vd~]`.

★ **Step 24**: Implement the following patch[18] with the 'delay' slider's properties set to operate *on a log scale* from 1 to 1000 msecs ...



[18] Of course, you may choose to replace the `[+~]` with the mixer you implemented earlier.

★ **Step 25**: Experiment with various values for the delay, and note the different effects you can achieve with delays (a) below 20 msecs, (b) between 20 and 100 msecs, and (c) above 100 msecs.

You should observe that, with delays below 20 msec, the signals combine to create a subtle 'phasing' effect. This is known as 'comb filtering' as the signal is effectively interfering with itself, and frequency components corresponding to multiples of the delay time are enhanced or cancelled out (due to 'superposition'). Delays between 20 and 100 msecs give the effect of the voice being in a reverberant room. Delays above 100 msecs sound like straight echoes.

Finally, in this section, it is possible to use an LFO to vary the delay. The resulting effect is known as a 'flanger'.

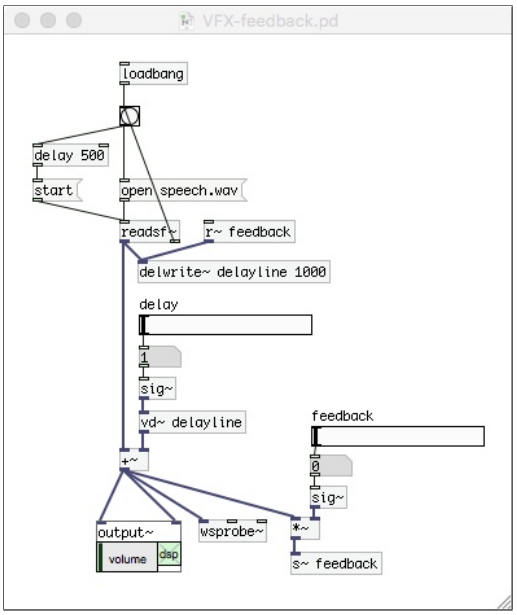
★ **Step 26**: Add an LFO to your 'delay' patch to create a 'flanger', and experiment with different settings. Note that you will need to scale the output of the LFO, and you will get different effects depending on whether the delayed signal is mixed with the original or not.

Note: *Using 'flanger' it is possible to achieve an underwater effect by setting LFO = sine wave,* `speed` *= 10 Hz,* `depth` *= 0.4.* `delay` *= 5 msecs and* `mix` *= 100% processed.*

2.10 Feedback: *Reverberation*

The previous section noted that an echo occurring between 20 and 100 msecs after the original sounds like reverberation. However, in a real environment there are many echoes caused by multiple reflections from the different surfaces present, and a signal might bounce around a very reverberant environment many times. A simple method for achieving such an effect is to feed a proportion of the output signal back to the input. This is known as 'feedback'.

★ **Step 27**: Modify your delay patch as follows, and experiment with different values for `delay` and `feedback` ...



QUESTION 9 ***(worth up to 5 marks)***

In a practical system, why is it important to keep the feedback gain less than 1?

Note: *A short delay with a lot of feedback can give a metallic sound to a voice, e.g. C3PO from Star Wars. Try* `delay` *= 10 msecs,* `feedback` *= 0.9 and* `mix` *= 90% processed.*

Clearly feedback could be applied to any/all of the effects that you have implemented so far, thereby increasing the range of possible voice effects considerably - as you will discover in Part-II.

3 Part-II: Real-Time Voice Changer

3.1 General Requirements

The final part of this assignment is to compile all of the different components you developed in Part-I into a single `[VoiceChanger]` application. The application should allow 'live' speech input in addition to the ability to select a particular prerecorded file (i.e. no longer hard-wired for `speech.wav`). The GUI should be well thought out, easy to use and attractive, and it should not only allow the different effects to be controlled, but also allow them to be connected to each other in a logical sequence. You should also provide innovations, for example preset effects (using `Vradio` buttons to select particular combinations of settings).

3.2 COM4502-6502 Only

It is possible to convert a single voice into multiple voices by combining the outputs of parallel manipulations, especially different frequency shifts. This effect is known as 'chorus'. Your objective is to add 'chorus' to your `[VoiceChanger]` application by careful use of abstractions and a method for selecting one or more output voices.

3.3 Final Steps

★ **Step 28**: Implement `[VoiceChanger]` according to the requirements stated above.

QUESTION 10 ***(worth up to 50 marks[19])***
Please provide a short[20] description of the operation of your `[VoiceChanger]` application, together with a screenshot of your final GUI.

Congratulations, you have almost completed this programming assignment. All that is left now is to package up your code and submit it for marking.

★ **Step 29**: Create a `.zip` file of the form '`YourNames.zip`' containing the following items ...

- Your response sheet in `[.pdf]` format with your names on the front cover and containing your responses to all of the questions.
- Your final `[VoiceChanger]` code, together with any necessary abstractions (such as `[LFO~]`, `[LFO~-help]` etc.).

Important: For marking, we expect your code to work 'out of the box'.

Do **NOT** include ...

- Your `.tex` file and associated image files.
- Your work-in-progress code examples.

★ **Step 30**: Hand-in your `.zip` file in accordance with the instructions at the beginning of this document.

That's it - you're done!

[19] 25 for functionality, 15 for design/layout, 5 for `Pd` features, 5 for innovations
[20] no more than 200 words