Seminar Akustik. Übungen mit Bandpass-Filtern (Praat)

Erste Schritte

Auf dem Account `matlab` einloggen (Passwort wird bekannt gegeben).

Dann eintippen:

```
cd akustikfort
cd bandpassuebung
praat
```

Die Übung ist auch auf einem eigenen Account möglich (oder auf dem eigenen Notebook, wenn praat installiert ist). Dann müssen aber die 5 Wav-Dateien tec_schumann.wav, harmonics100.wav, whitenoise.wav, asa.wav und asha.wav hier (unter 4. Sitzung) runtergeladen werden:

http://www.phonetik.uni-muenchen.de/~hoole/kurse/akustik_ba/index.html

As Praat's commands are in English it seems to make more sense to continue in English ....

(1)

Load the file tec_schumann.wav

Filter it to sound like telephone speech using

`Filter > Filter > Pass Hann Band`

Choose appropriate values for '`From frequency`' and '`To frequency`'

Set the value of '`smoothing`' to 100 (if it is not already set to this value), and leave it at this value for exercises 1, 2 and 3.

> To see how '`From frequency`', '`To frequency`' and '`smoothing`' determine the shape of the filter, press the help button (in the command box entitled '`Sound: Filter (pass Hann band)`' that is currently open), and then click
> `Spectrum: Filter (pass Hann band)`

After pressing `OK` in the command box to carry out the filtering, a new object will appear in Praat's object list. Rename it in some appropriate way if you want to experiment with different filter settings. Listen to the signal, and also use the spectrogram display to check the filter worked as you intended.

(2)

Load the file harmonics100.wav.

This contains harmonics of 100Hz up to about 4kHz

Try and define 3 bandpass filters that will give rough approximations to [i], [u] and [a], respectively.

(Have a look at the sonagramm of tec_schumann for a hint, if necessary)

Rename each new object appropriately.

Which vowel is most difficult to approximate if we are only using a single band-pass filter to shape the spectrum?

In exercise 4 below we will try and do a better job.

(3)

Load the file whitenoise.wav

This has a flat spectrum up to 10kHz

Try and define 2 bandpass filters that will give rough approximations to [**s**] and [**ʃ**]

Load the two sound files asa.wav and asha.wav to give yourself a hint, if necessary.

To view the sonagram of these two files use the following settings:

In Spectrum>Spectrogram settings choose view range from 0 to 10000Hz, window length 0.005s and dynamic range 40dB.

In Spectrum>Advanced spectrogram settings set Pre-emphasis to 0dB (normally it is set to 6dB).

(We will discuss the reason for these settings in the course)

(4)

We will now try to approximate the vowels [i], [u] and [a] using a combination of two band-pass filters each (and three, if time).

(This also introduces one way of combining sounds in Praat that is used extensively in exercise 6 below.)

Choose a vowel (perhaps the one that was least successful in exercise 2) and decide on appropriate frequency ranges for two separate band-pass filters.

Mark the harmonics100 sound in the object list and filter it separately with each of these band-pass filters.

Rename the new sounds appropriately, e.g i_band_1, i_band_2, etc.

Mark these two new sounds in Praat’s object list

Then choose `Combine Sounds > Combine to stereo`

This will create a new object. Select this object (if it is not already selected) and choose `Convert > Convert to mono`

Give the new mono object a more convenient name and remove the stereo object.

(5)

If time, try out ‘whispered’ versions of [i], [u] and [a] using at least two band-pass filters each, as in exercise 4

Additional exercise

Normally there will not be time for Exercise 6 below.

As a slightly quicker alternative try and determine what bandpass filter with a width of 500Hz will sound most intelligible when applied to tec_schumann.wav.

In other words, if you are only able to keep a range of 500Hz from the original signal is it better to keep e.g. 0-500Hz or 2000-2500Hz, or whatever ......?

(6)

More complicated bandpass filtering based on an experiment by Greenberg et al. "*Speech intelligibility derived from exceedingly sparse spectral information*"

(More information, and useful praat scripts can be found here:

http://www.phonetik.uni-muenchen.de/~hoole/kurse/akustikfort/index_wise0708.html)

Before starting, remove all sounds from the above exercises except tec_schumann, which will form the basis for subsequent filtering.

First of all we will filter the speech with four different bandpass filters spread over the frequency range up to 6 kHz. Each bandpass is very narrow, so on its own contains very little information.

Use the following values for 'From frequency', 'To frequency' and 'smoothing' and rename each new bandpassed signal to tec_schumann_band1, tec_schumann_band2 etc.

1. 298 375 10
2. 750 945 20
3. 1890 2381 50
4. 4762 6000 100

Check the results with the spectrogram display, and listen to each new signal. Is any of these narrow spectral 'slits' at all intelligible?

Now we will examine the effect of combining these slits. We will look at the following combinations:

Band 1 + Band 2

Band 3 + Band 4

Band 2 + Band 3

To make each combination proceed as follows:

Mark the two bands in Praat's object list

Then choose Combine Sounds > Combine to stereo

This will create a new object. Select this object (if it is not already selected) and choose Convert > Convert to mono

Give the new mono object a more convenient name like tec_schumann_band_1_2, and remove the stereo object.

Compare the intelligibility of these 3 combinations.

Finally, combine the Band 1+2 signal with the Band 3+4 signal to give a new signal consisting of all 4 spectral slits.

This should now sound reasonably intelligible even though large parts of the spectrum are missing.

An interesting additional manipulation is to shift the spectral slits slightly in time so that they are no longer exactly synchronous.

Example for combination of Band2 and Band3, with Band3 shifted.

First of all define a short (e.g 75ms) new signal consisting of silence:

New > Sound > Create sound from formula

Name: pause75

End time: 0.075

Sampling frequency: 16000

Formula 0*x

This should result in a new object named pause75 appearing in the list

Select the band3 signal and press the copy button so that a copy of this signal appears below pause75 in the object list.

Select both these signals and choose

Combine sounds > Concatenate

A new object called 'chain' will be created.

Rename it to something like tec_schumann_band3_shift75

In order to combine this signal with the band2 signal we need to make sure they are exactly the same length. To do this:

Mark the shifted band3 signal and the band2 signal, and choose

Convert > Extract part

and specify time range from 0 to 4.25, and rectangular window

A new version of these two signals with _part added to the names will appear in the object list.

Combine these signals as above using Combine > Combine to stereo, and Convert > Convert to mono. Rename the final signal to something like tec_schumann_band_2_3shift75 (and remove some of the intermediate steps from the object list).

Compare this signal to the original band_2_3 combination.

If time, experiment with further shift values, and/or create a version with all four slits, but with two of the four slits shifted in time