• Using wavesurfer, ESPS get_f0 to obtain f₀ time-series

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▼ FIFTEEN3.wav [Configuration: Waveform]
Hz
300 -
250 -
200 -
50 -
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Using wavesurfer, ESPS get_f0 to obtain f₀ time-series



- wavesurfer uses the ESPS get_f0 command to obtain f_0 time-series
- syntax: get_f0 [options] input_file output_file

ESPS is a package of UNIX-like commands and programming libraries for speech signal processing.

You can download a recent .deb package for ESPS from http://www.phon.ox.ac.uk/releases

David Talkin's paper on get_f0 is here: http://www.ee.columbia.edu/~dpwe/papers/Talkin95-rapt.pdf



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- syntax: get_f0 [options] input_file output_file

```
for i in *.wav
> do get_f0 $i $i.f0
> pplain $i.f0 >$i.f0.csv
> done
```

On one line, that's:

for i in *.wav; do get_f0 \$i \$i.f0; pplain \$i.f0 >\$i.f0.csv; done



• These .csv files are simple ASCII text files like this: ------

• The first column is f_0 , the second is voicing; ignore the other two



0 0 0 0.272821 0 0 34.4253 0.551759 0 0 44.9999 0.641592 0 0 242.326 0.515299 176.894 1 401.017 0.553706 174.434 1 399.113 0.931412 167.352 1 378.998 0.951326 162.623 1 358.704 0.927735 160.734 1 356.843 0.931884 154.345 1 250.132 0.617554 170.107 1 159.65 0.843205 0 0 82.2662 0.494668 0 0 92 7429 0 730789 0 0 110 42 0 433576 0 0 71.1711 0.53332 0 0 59.6541 0.419894 0 0 62.9074 0.538716 0 0 53.2908 0.319767 0 0 47.034 0.288603 0 0 38.5281 0.346631 0 0 47.2128 0.452121 0 0 50.4091 0.43822 0 0 44.2441 0.568226 --More--(28%)

 You can open a .csv file in a spreadsheet programme, and plot the data (see, you don't have to have Praat or wavesurfer to draw speech parameters)





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How you could analyse intonation ...

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• With the discontinuity (voiceless portion) excised:





• Cf. a linear regression line; it may not look quite so good, but it's actually a better fit (to the *whole* line)





• A logarithmic regression curve fits better





Why single-point measurements of sampled data are not great

• What is the true minimum of this curve? - 9968 or -9969?







Functional Data Analysis

- Modelling sampled data using (continuous) functions
- General approach:
 - Possibly smooth the data a bit, to iron out irrelevant wiggles
 - Possibly normalize the data
 - Registration: some sort of time alignment of the individual tokens (not always necessary)



Functional Data Analysis

Choose a general kind of (basis) function that looks like your data

- For periodic data: Fourier series
- For nonperiodic data: B-splines sometimes: Orthogonal Polynomials (Example 1)
- others are possible
- Probability density functions
- E.g., for normally-distributed data: Gaussians (Exa

(Example 2)

Fit the function to the data

i.e. find the parameters of the function that minimizes the differences between the function and the data



- Put numeric data into Matlab's vector notation:
 f0 = load('FIFTEEN3.wav.f0.csv');
 y = f0(:,1);
 y = y(y>0);
 Normalize it: yn = y/mean(y)-1;
- (Better: ym = yn/max(abs(yn));

Normalize the time dimension to the interval [-1 1], and turn it into a column vector:

 x = ((1:length(yn))-length(yn)/2)/(length(yn)/2);
 x = x';



• Fit the normalized data to a polynomial (e.g. a cubic)

 $y = a_1 x^3 + a_2 x^2 + a_3 x + a_4$

[a,S] = polyfit(x,yn,3);

Output values: a = 0.19321 0.63340 -0.70280 -0.19866

•The fitted function is given by fit = getfield(S, 'yf'); and restored to the original units (e.g. Hz)

ysynth = mean(y) * (fit+1);



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•The fitted function is given by fit = getfield(S, 'yf');
and can be restored to the original units (e.g. Hz) by
vsynth = mean(v)*(fit+1);















Normalized f₀

Normalized time



Normalized time



Normalized time



Normalized time



Orthogonalisation

• Translate polynomial coeffients into orthogonal (Legendre) polynomial coeffs:

c = [0.4*a(1) 2*a(2)/3 a(3)+6*a(1)/5 a(4)]

%% a = 0.19321 0.63340 -0.70280 -0.19866 %% c = 0.077284 0.422269 -0.470948 -0.198659



Loop over all the "good" files

for i = 2:172

```
eval(['fid = fopen(''FIFTEEN', int2str(i), '.wav.f0.csv'');']);
```

```
if (fid ~= -1) %% checks file FIFTEEN i ... exists
```

```
eval(['f0 = load(''FIFTEEN', int2str(i), '.wav.f0.csv'');']);
```

y = f0(:, 1);

y = y(y>0);

```
yn = y/mean(y)-1;
```

```
x = ((1:length(yn))-length(yn)/2)/(length(yn)/2);
```

```
x = x';
```

```
[a,S] = polyfit(x,yn,3);
```

```
c = [0.4*a(1) 2*a(2)/3 a(3)+6*a(1)/5 a(4)];
```

```
C(i,:) = [i c];
```

end

```
end
```

save('coeffs.csv','C');



Now do your statistics

• Rather than applying statistical tests to the raw data, examine the means, variances etc of the coefficients of the functions you're using to model the data

