## Intonation of FIFTEEN

- Using wavesurfer, ESPS get_f0 to obtain $f_{0}$ time-series


WaveBar - [0.000s 0.800 s$] 328.04 \mathrm{~mm} / \mathrm{s}$

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- 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

## Intonation of FIFTEEN

- Using wavesurfer, ESPS get_f0 to obtain $f_{0}$ time-series
- 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

```
0000.272821
```

0000.272821
0 34.4253 0.551759
0 34.4253 0.551759
0 44.99990.641592
0 44.99990.641592
0 242.3260.515299
0 242.3260.515299
176.8941401.0170.553706
176.8941401.0170.553706
174.4341399.1130.931412
174.4341399.1130.931412
167.3521378.9980.951326
167.3521378.9980.951326
162.6231358.7040.927735
162.6231358.7040.927735
160.7341356.843 0.931884
160.7341356.843 0.931884
154.3451250.1320.617554
154.3451250.1320.617554
170.1071159.650.843205
170.1071159.650.843205
0 82.26620.494668
0 82.26620.494668
0 92.7429 0.730789
0 92.7429 0.730789
0 110.420.433576
0 110.420.433576
0}71.17110.5333
0}71.17110.5333
0 59.65410.419894
0 59.65410.419894
0 62.90740.538716
0 62.90740.538716
0 53.29080.319767
0 53.29080.319767
0 47.0340.288603
0 47.0340.288603
0 38.5281 0.346631
0 38.5281 0.346631
0 47.2128 0.452121
0 47.2128 0.452121
0 50.40910.43822
0 50.40910.43822
0 44.24410.568226
0 44.24410.568226
--More--(28%)

```
--More--(28%)
```


## Intonation of FIFTEEN3

- 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)



## Intonation of FIFTEEN165

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## How you could analyse intonation ...

- Many approaches to phonetic analysis focus on particular points of interest in the time series, e.g.



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## Intonation of FIFTEEN3

- With the discontinuity (voiceless portion) excised:



## Intonation of FIFTEEN3

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



## Intonation of FIFTEEN3

- 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 ?



# Why single-point measurements of sampled data are not great 



| -8600 |
| :---: |
| -8800 |
| -9000 |
| -9200 |
| -9400 |
| -9600 |
| -9800 |
| -10200 |
| -1000 |

Answer: neither.
-9968 or -9969 ? The true value is -10000 ; it is a cosine function $\times 10000$

## 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 (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

## Orthogonal polynomials in Octave/Matlab

- Put numeric data into Matlab's vector notation:

```
f0 = load('FIFTEEN3.wav.f0.csv');
y = f0(:,1);
y = y(y>0);
```

```
y =
[176.894;
174.434;
167.352;
162.623;
160.734;
88.6733];
```

- Normalize it: $y n=y / m e a n(y)-1$;
- Normalize the time dimension to the interval [-1 1], and turn it into a column vector:

$$
\begin{aligned}
& x=((1: \text { length }(y n))-\text { length }(y n) / 2) /(\text { length }(y n) / 2) ; \\
& x=x ' ;
\end{aligned}
$$

## Orthogonal polynomials in Octave/Matlab

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

$$
y=a_{1} x^{3}+a_{2} x^{2}+a_{3} x+a_{4}
$$

$$
[\mathrm{a}, \mathrm{~S}]=\operatorname{polyfit}(\mathrm{x}, \mathrm{yn}, 3) ;
$$

$$
\text { Output values: } \mathrm{a}=0.19321 \quad 0.63340-0.70280-0.19866
$$

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

$$
\text { ysynth }=\operatorname{mean}(y) *(f i t+1) ;
$$

## Orthogonal polynomials in Octave/Matlab

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

$$
\begin{aligned}
& y=a_{1} x^{3}+a_{2} x^{2}+a_{3} x+a_{4} \\
& {[a, s]=\text { polyfit }(x, y n, 3) ;} \\
& \text { Output values: } a=0.19321 \quad 0.63340-0.70280-0.19866
\end{aligned}
$$

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## Orthogonal polynomials in Octave/Matlab

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

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

$$
\text { ysynth }=\operatorname{mean}(y)^{\star}(f i t+1) ;
$$










## 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)]
```

$$
\begin{array}{llll}
\% \% \mathrm{a}=0.19321 & 0.63340 & -0.70280 & -0.19866 \\
\% \% \mathrm{c}=0.077284 & 0.422269 & -0.470948 & -0.198659
\end{array}
$$

## Loop over all the "good" files

```
for i = 2:172
    eval(['fid = fopen(''FIFTEEN',int2str(i),'.wav.f0.csv'');']);
    if (fid ~= -1)
        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');

