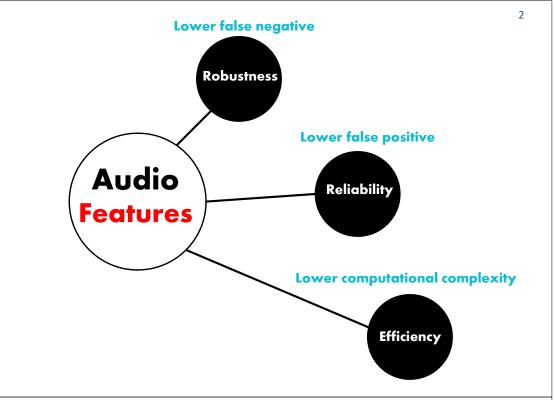
[4] Audio Features



Audio Spectral Envelope

Audio Spectrum Envelope $ASE[b] = \sum_{k \in Band[b]} P[k]$ $P[k] = |S[k]|^{2}$ S[k] = Spectrum $P(k) \sum_{31.25} \sum_{62.5} \sum_{125} \sum_{250} \sum_{500} \sum_{1k} \sum_{2k} \sum_{4k} \sum_{16k} \sum_{f(k)} \sum_{(\log_{2} - Hz)} \sum_{(\log_{2}$

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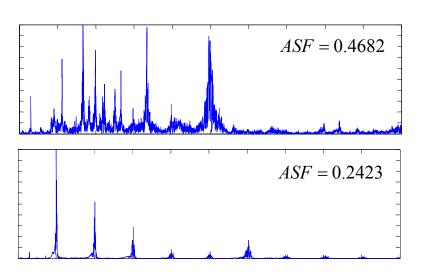
Audio Spectral Flatness

$$ASF[b] = \frac{\sqrt[N_b]{\prod_{k \in Band[b]} P[k]}}{\frac{1}{N_b} \sum_{k \in Band[b]} P[k]}$$
 Arithmetic Mean

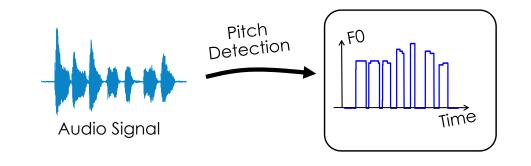
 $N_b = \text{No.of frequencies in each band}$

$$N_b \prod_{k \in Band[b]} P[k] = \exp \left(\frac{1}{N_b} \sum_{k \in Band[b]} \ln P[k] \right)$$

Audio Spectral Flatness



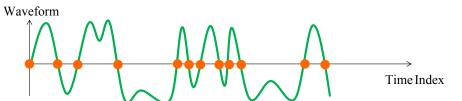
Fundamental Audio Frequency



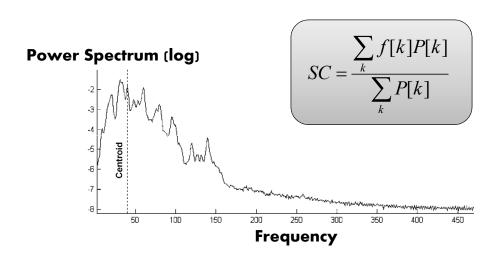
Zero Crossing Rate

$$ZCR = \frac{F_{S}}{2N} \sum_{n=1}^{N-1} |sign(s[n]) - sign(s[n-1])|$$

N = No.samples in s[i] $F_s = \text{Sampling Frequency}$



Spectral Centroid



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Harmonic Ratio

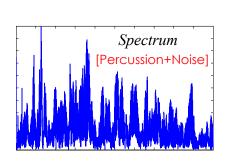
$$R[m] = \frac{\sum_{n \in Frame} s[n]s[n-m]}{\sqrt{\sum_{n \in Frame} s^{2}[n] \sum_{n \in Frame} s^{2}[n-m]}} \Rightarrow HR = \max_{m \ge M_{0}} R[m]$$

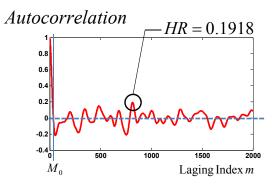
R = Autocorrel ation Function

m =Lagging index

 M_0 = Position of the first zero crossing of autocorrelation R

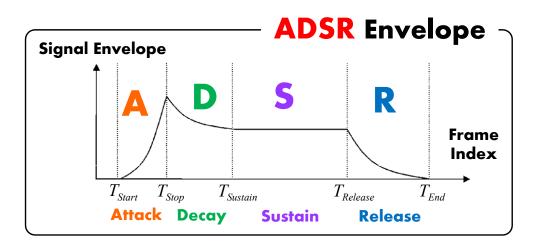
Harmonic Ratio





Harmonic Ratio Autocorrelation Autocorrelation HR = 0.9767 Spectrum [Harmonic Structure] Mo Laging Index m Flute Laughter Noise

Log Attack Time

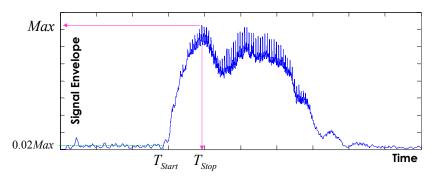


Log Attack Time

$$LAT = \log_{10} \left(T_{Stop} - T_{Start} \right)$$

 T_{Start} = The time the signal envelope exceeds 2% of its maximal value

 T_{Stop} = The time the signal evelope reaches its maximal value



Harmonic Spectral Centroid

LHSC = Local Harmonic Spectral Centroid of Audio Frame

$$LHSC = \frac{\sum_{h=1}^{N_H} f_h A_h}{\sum_{h=1}^{N_H} A_h}$$

 f_h = Frequency of the h^{th} harmonic

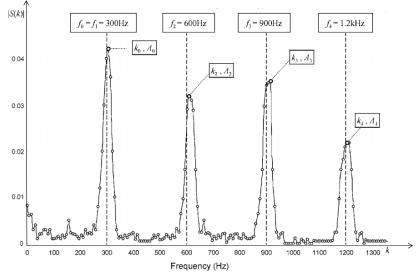
 $A_h = \text{Amplitude of the } h^{\text{th}} \text{ harmonic}$

 N_H = No. of harmonic peaks



L =No. of time frames

Harmonic Spectral Centroid



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Harmonic Spectral Deviation

Spectral Envelope

$$SE_{h} = \begin{cases} (A_{h} + A_{h+1})/2 & , h = 1\\ (A_{h-1} + A_{h} + A_{h+1})/2 & ; h \in [2, N_{H} - 1]\\ (A_{h-1} + A_{h})/2 & ; h = N_{H} \end{cases}$$

$$LHSD = \frac{\sum_{h=1}^{N_H} \left| \log_{10} A_h - \log_{10} SE_h \right|}{\sum_{h=1}^{N_H} \log_{10} A_h}$$

$$HSD = \frac{1}{L} \sum_{i=1}^{L} LHSD[i]$$

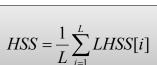
Harmonic Spectral Spread

$$LHSS = \sqrt{\frac{\sum_{h=1}^{N_{H}} (f_{h} - LHSC)^{2} A_{h}^{2}}{\sum_{h=1}^{N_{H}} A_{h}^{2}}}$$



Harmonic Spectral Variation

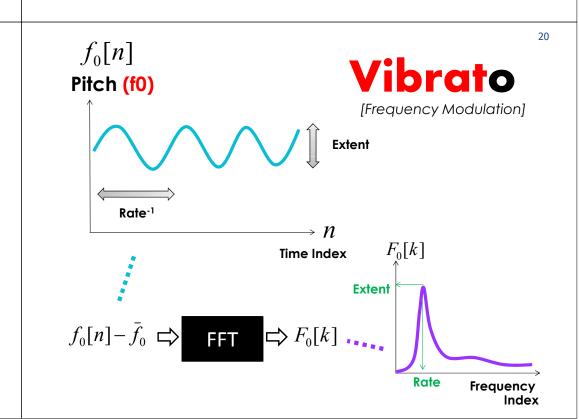
$$LHSV[i] = 1 - \frac{\sum_{h=1}^{N_H} A_h[i-1] A_h[i]}{\sqrt{\sum_{h=1}^{N_H} A_h^2[i-1]} \sqrt{\sum_{h=1}^{N_H} A_h^2[i]}}$$



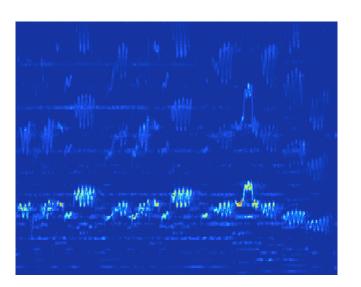
Vibrato

[Frequency Modulation]

- Variation of pitch.
- Extent of vibrato
 - = Amount of pitch variation
- Rate of vibrato
 - = Speed which the pitch is varied
- String instruments produce the FM dominant sounds

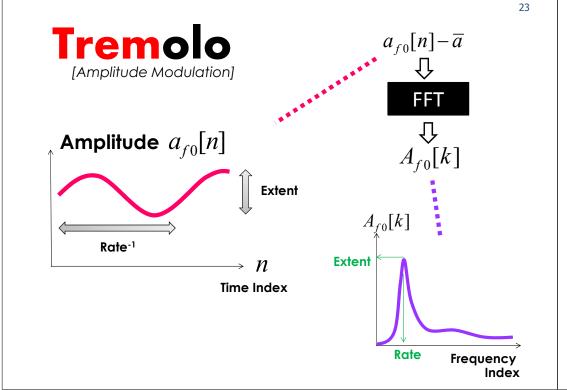




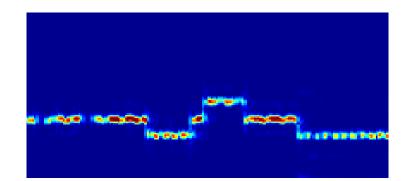




- Variation of sound intensity.
- Extent of vibrato
 - = Amount of intensity variation
- Rate of vibrato
 - = Speed which the intensity is varied
- Wind and brass instruments produce AM dominant sounds.



Tremolo [Amplitude Modulation]



Linear Predictive Coding

- A model used for predicting a value of current sample of signal from the previous samples
- Used in Speech Analysis, Speech Synthesis, Audio Classification, Audio Compression

LPC Coefficients

Current sample $S[n] = \sum_{k=1}^{P} a_k S[n-k] + \varepsilon[n]$ Prediction Error

Linear Predictive Coding

s[n]Training Signal $\{a_k\}$

Training LPC

- Find $\{a_k\}$
- To minimize $\sum_{n} \varepsilon^{2}[n]$

$$\varepsilon[n] = s[n] - \hat{s}[n]$$

$$\hat{s}[n] = \sum_{k=1}^{P} a_k s[n-k]$$

Linear Predictive Coding

$$E = \sum_{n} \varepsilon^{2}[n] = \sum_{n} \left(s[n] - \sum_{k=1}^{P} a_{k} s[n-k] \right)^{2}$$

$$\frac{\partial E}{\partial a_i} = \sum_{n} \left(-2s[n-i] \right) \left(s[n] - \sum_{k=1}^{P} a_k s[n-k] \right)$$

$$0 = \sum_{n} s[n]s[n-i] - \sum_{n} s[n-i] \sum_{k=1}^{P} a_k s[n-k]$$

$$0 = \sum_{n} s[n]s[n-i] - \sum_{k=1}^{p} a_k \sum_{n} s[n-k]s[n-i]$$

Linear Predictive Coding

$$0 = \sum_{n} s[n]s[n-i] - \sum_{k=1}^{P} a_k \sum_{n} s[n-k]s[n-i]$$

$$0 = R[i] - \sum_{k=1}^{P} a_k R[k-i]$$

$$\sum_{k=1}^{P} a_k R[k-i] = R[i]$$

Autocorrelation Function
$$R[i-j] = R[j-i] = \frac{\sum_{n} s[n-i]s[n-j]}{\sum_{n} s^{2}[n]}$$

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Linear Predictive Coding

$$\frac{\partial E}{\partial a_1} = 0 \Rightarrow \sum_{k=1}^{P} a_k R[k-1] = R[1]$$

$$\frac{\partial E}{\partial a_2} = 0 \Rightarrow \sum_{k=1}^{P} a_k R[k-2] = R[2]$$

$$\frac{\partial E}{\partial a_P} = 0 \Rightarrow \sum_{k=1}^{P} a_k R[k-P] = R[P]$$

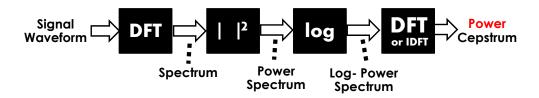
Linear Predictive Coding

$$\begin{bmatrix} R[0] & R[1] & R[2] & \cdots & R[P-1] \\ R[1] & R[0] & R[1] & \cdots & R[P-2] \\ R[2] & R[1] & R[0] & \cdots & R[P-3] \\ \vdots & \vdots & \vdots & \ddots & \vdots \\ R[P-1] & R[P-2] & R[P-3] & \cdots & R[0] \end{bmatrix} \begin{bmatrix} a_1 \\ a_2 \\ a_3 \\ \vdots \\ a_P \end{bmatrix} = \begin{bmatrix} R[1] \\ R[2] \\ R[3] \\ \vdots \\ R[P] \end{bmatrix}$$

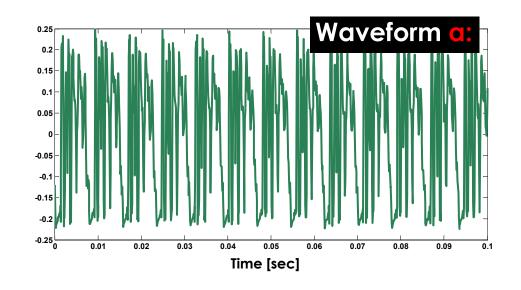
$$\begin{bmatrix} a_1 \\ a_2 \\ a_3 \\ \vdots \\ a_P \end{bmatrix} = \begin{bmatrix} R[0] & R[1] & R[2] & \cdots & R[P-1] \\ R[1] & R[0] & R[1] & \cdots & R[P-2] \\ R[2] & R[1] & R[0] & \cdots & R[P-3] \\ \vdots & \vdots & \vdots & \ddots & \vdots \\ R[P-1] & R[P-2] & R[P-3] & \cdots & R[0] \end{bmatrix}^{-1} \begin{bmatrix} R[1] \\ R[2] \\ R[3] \\ \vdots \\ R[P] \end{bmatrix}$$

Cepstrum

Fourier transform of the logarithm of the spectrum of signal



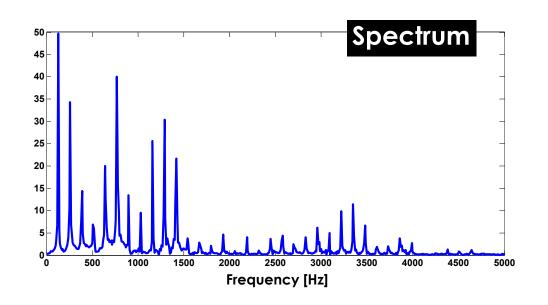
Cepstrum



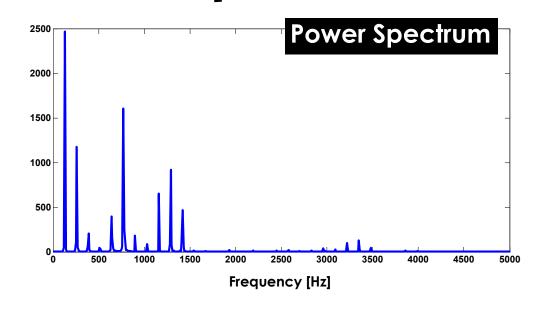
Cepstrum

33

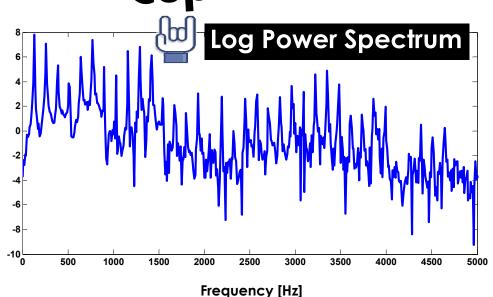
35



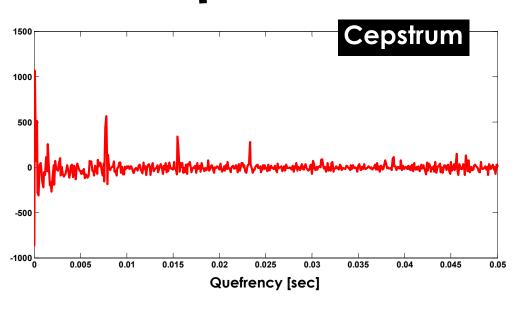
Cepstrum



Cepstrum

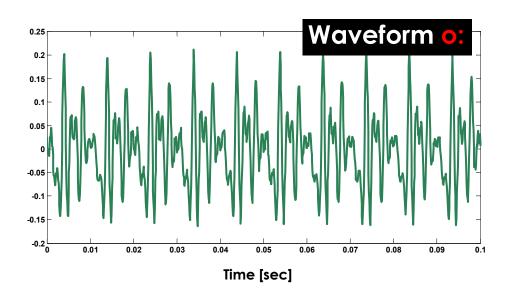


Cepstrum

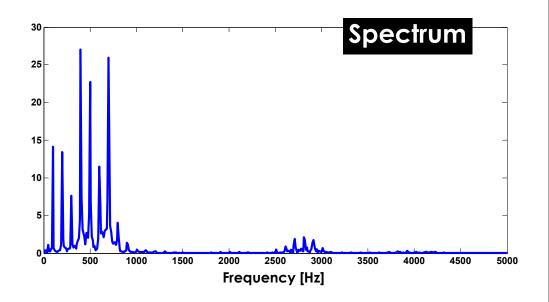


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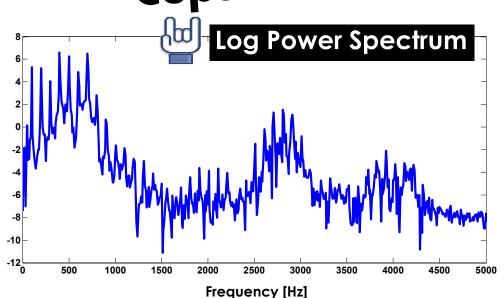
Cepstrum



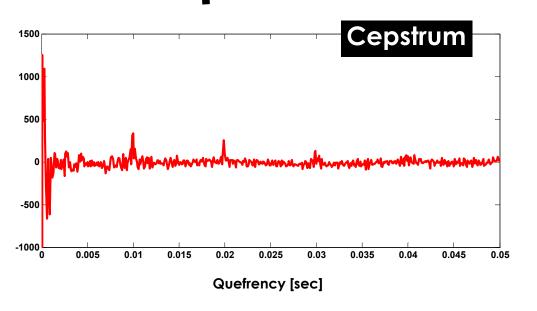
Cepstrum





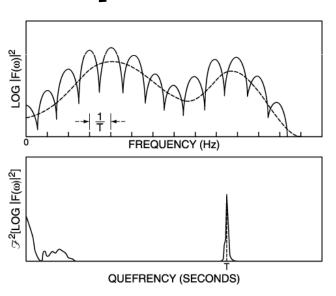


Cepstrum



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Cepstrum

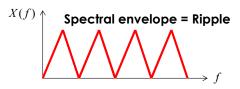


http://postgavindisorder.blogspot.com/2010_09_01_archive.html

Cepstrum

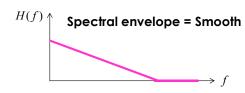


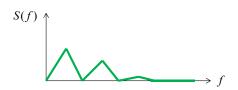
Impulse Response h(t)



s(t) = h(t) * x(t)

$$S(f) = H(f)X(f)$$





Cepstrum

 $Spectrum \qquad S(f) = H(f)X(f)$



Power spectrum $\big|S(f)\big|^2 = \big|H(f)\big|^2 \big|X(f)\big|^2$

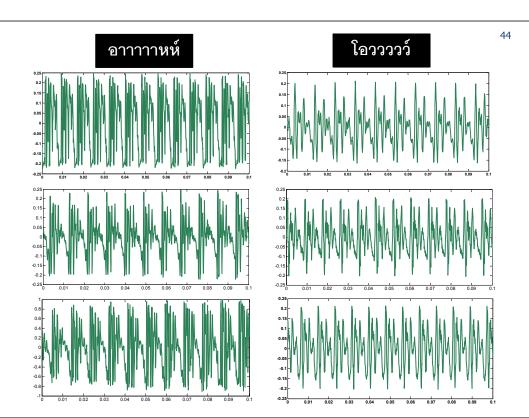


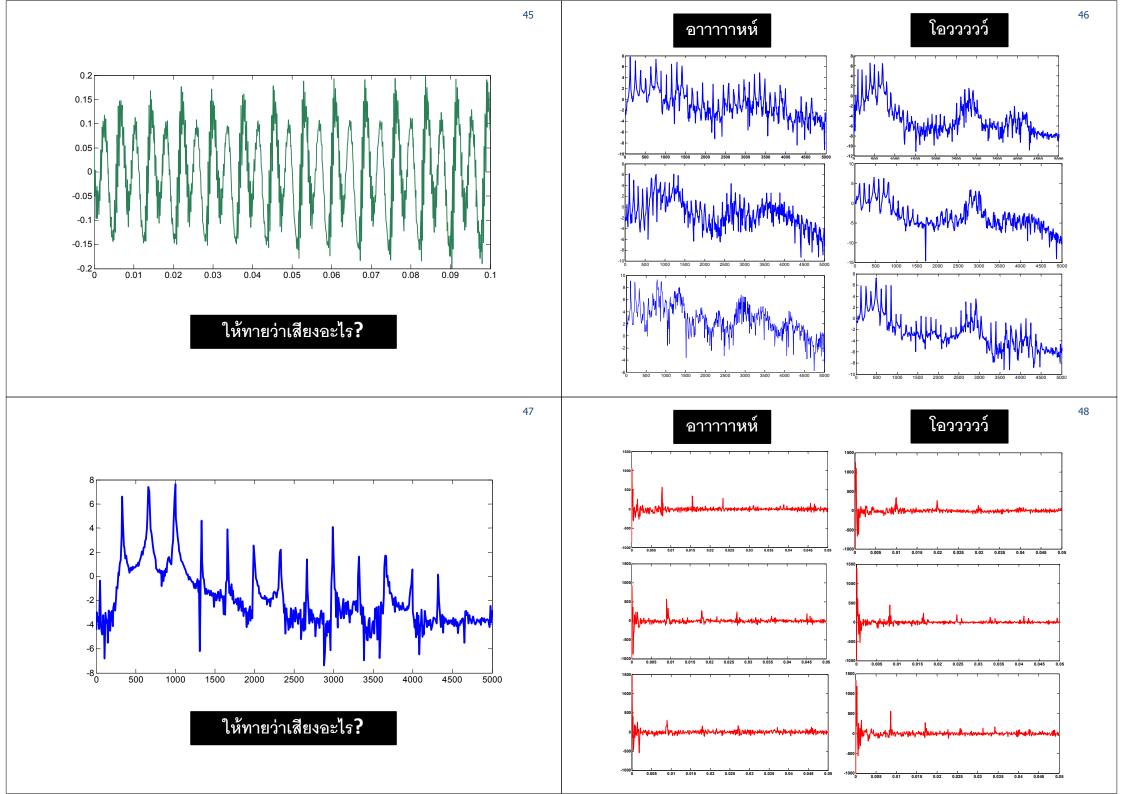
Log Power spectrum $\log |S(f)| = \log |H(f)| + \log |X(f)|$

Fourier Transform

Cepstrum $F\{\log |S(f)|\} = F\{\log |H(f)|\} + F\{\log |X(f)|\}$

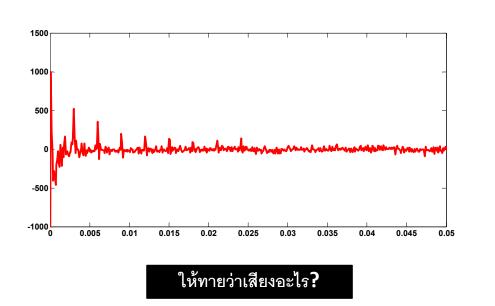
Low Quefrency Component High Quefrency Component



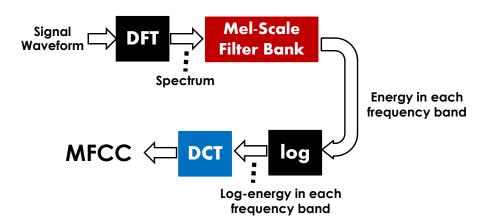




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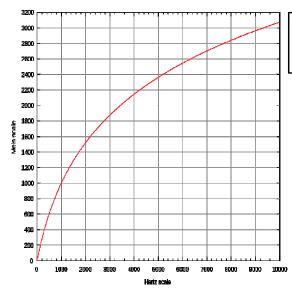


Mel-Frequency Cepstrum Coefficient [MFCC]



DCT = Discrete Cosine Transform

Mel-Scale



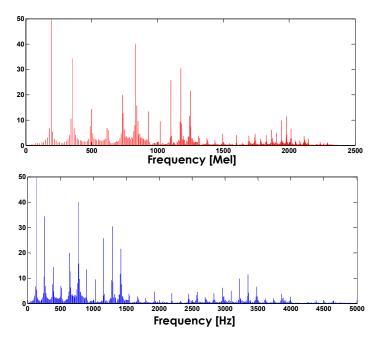


Perceptual scale of pitches

Hertz	Mel
174	250
391	500
662	750
1000	1000
1949	1500
3429	2000
5734	2500

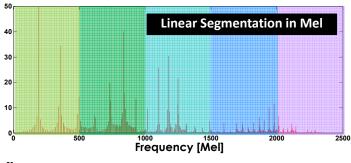
http://en.wikipedia.org/wiki/File:Mel-Hz plot.svg

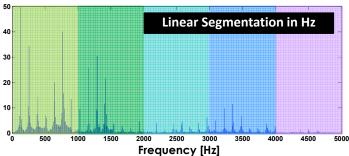
Mel-Scale



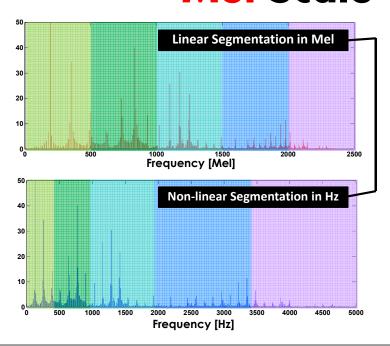
56

Mel-Scale





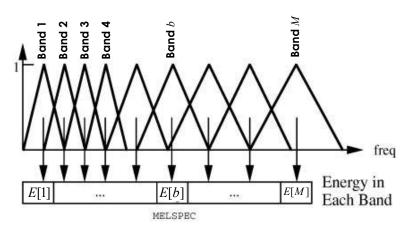
Mel-Scale



Mel-Scale Filter Bank

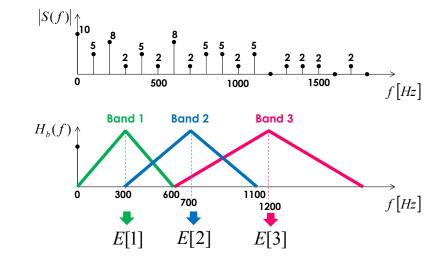
$$E[b] = \sum_{f \in Band[b]} |S(f)| H_b(f)$$

 $E[b] \colon \mathsf{Energy} \, \mathsf{in} \, \mathsf{Band} \, b$ $H_b(f) \colon \mathsf{Freq} . \, \mathsf{Response} \, \mathsf{of} \, \mathsf{Filter} \, \mathsf{for} \, \mathsf{Band} \, b$



Unsupervised speaker segmentation with residual phase and MFCC features S. Jothilakshmi, , V. Ramalingam , S. Palanivel

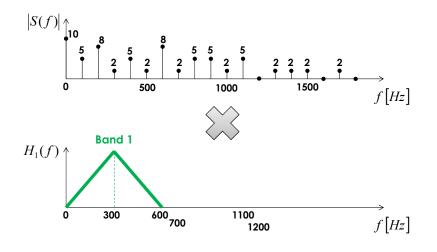
Mel-Scale Filter Bank



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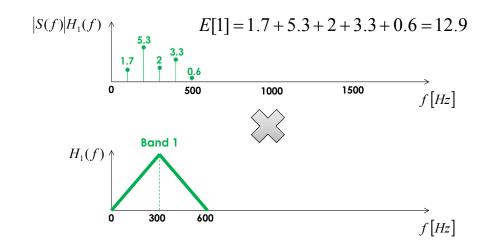
Mel-Scale Filter Bank

$$E[1] = \sum_{f \in Band[1]} |S(f)| H_1(f)$$



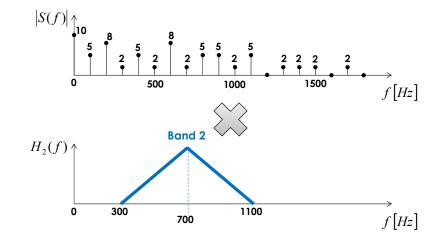
Mel-Scale Filter Bank

$$E[1] = \sum_{f \in Band[1]} |S(f)| H_1(f)$$



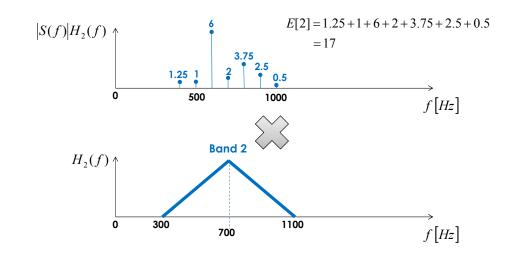
Mel-Scale Filter Bank

$$E[2] = \sum_{f \in Band[2]} |S(f)| H_2(f)$$



Mel-Scale Filter Bank

$$E[2] = \sum_{f \in Band[2]} |S(f)| H_2(f)$$

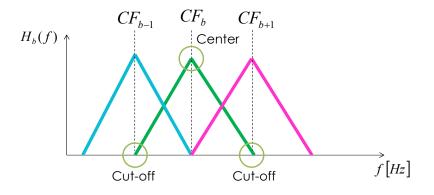


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Mel-Scale Filter Bank

• Cut-off frequency of filter in the current band determined by the center frequencies of the two adjacent filters.



Mel-Scale Filter Bank

• Triangular Filter

$$H_{b}(f) = \begin{cases} \frac{f - CF_{b-1}}{CF_{b} - CF_{b-1}}; & CF_{b-1} < f < CF_{b} \\ \frac{f - CF_{b+1}}{CF_{b} - CF_{b+1}}; & CF_{b} \le f < CF_{b+1} \\ 0; & ; otherwise \end{cases}$$

 CF_b = Center Frequency of Band b[Hz]

Mel-Scale Filter Bank

Center Frequency

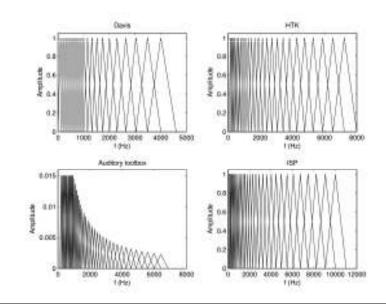
$$CF_b = F_{\min} + b \cdot \frac{F_{\max} - F_{\min}}{M+1} \qquad in [mel]$$

$$CF_b \quad in [Hz]$$

M =Number of Bands

 $F_{\min} \sim F_{\max} = Frequency Range$

Mel-Scale Filter Bank

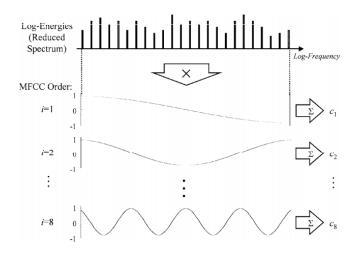


Discrete Cosine Transform [DCT]

$$c[i] = \sum_{b=1}^{M} \ln(E[b]) \cos\left(i\left(b - \frac{1}{2}\right)\frac{\pi}{M}\right)$$

c[i] = Mel Frequency Cepstrum Coefficient i = Order of MFCCM = Number of Frequency Bands

Discrete Cosine Transform [DCT]

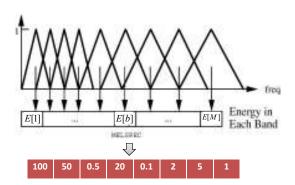


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Discrete Cosine Transform

[DCT]

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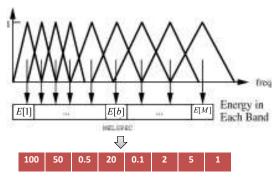


$$c[1] = \sum_{b=1}^{8} \ln(E[b]) \cos\left(\left(b - \frac{1}{2}\right) \frac{\pi}{8}\right)$$

$$= \ln(100) \cos\left(\frac{\pi}{16}\right) + \ln(50) \cos\left(\frac{3\pi}{16}\right) + \ln(0.5) \cos\left(\frac{5\pi}{16}\right) + \dots + \ln(1) \cos\left(\frac{15\pi}{16}\right)$$

$$= 6.6947$$

Discrete Cosine Transform [DCT]

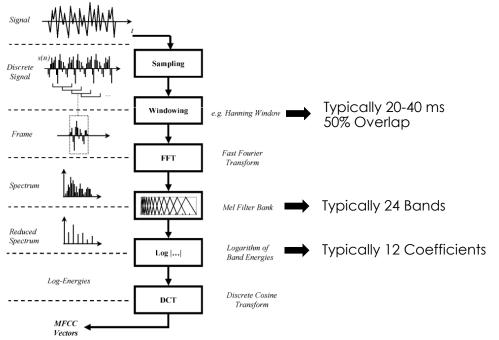


$$c[2] = \sum_{b=1}^{8} \ln(E[b]) \cos\left(2\left(b - \frac{1}{2}\right)\frac{\pi}{8}\right)$$

$$= \ln(100) \cos\left(\frac{\pi}{8}\right) + \ln(50) \cos\left(\frac{3\pi}{8}\right) + \ln(0.5) \cos\left(\frac{5\pi}{8}\right) + \dots + \ln(1) \cos\left(\frac{15\pi}{8}\right)$$

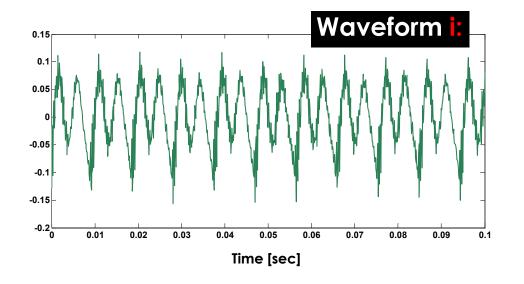
$$= 5.7272$$



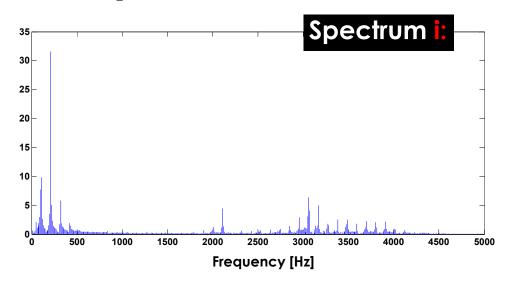


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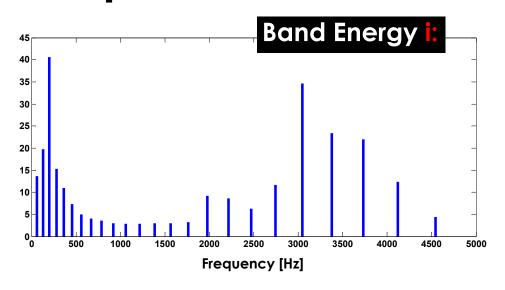
Mel-Frequency Cepstrum Coefficient



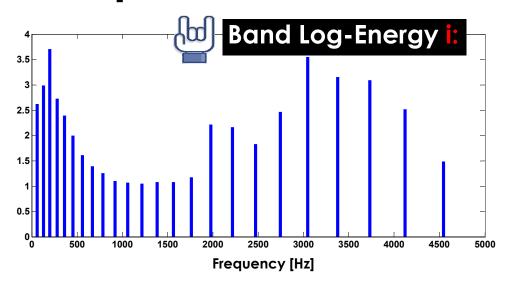
Mel-Frequency Cepstrum Coefficient



Mel-Frequency Cepstrum Coefficient



Mel-Frequency Cepstrum Coefficient



Mel-Frequency Cepstrum Coefficient

