

Audio Signal Processing

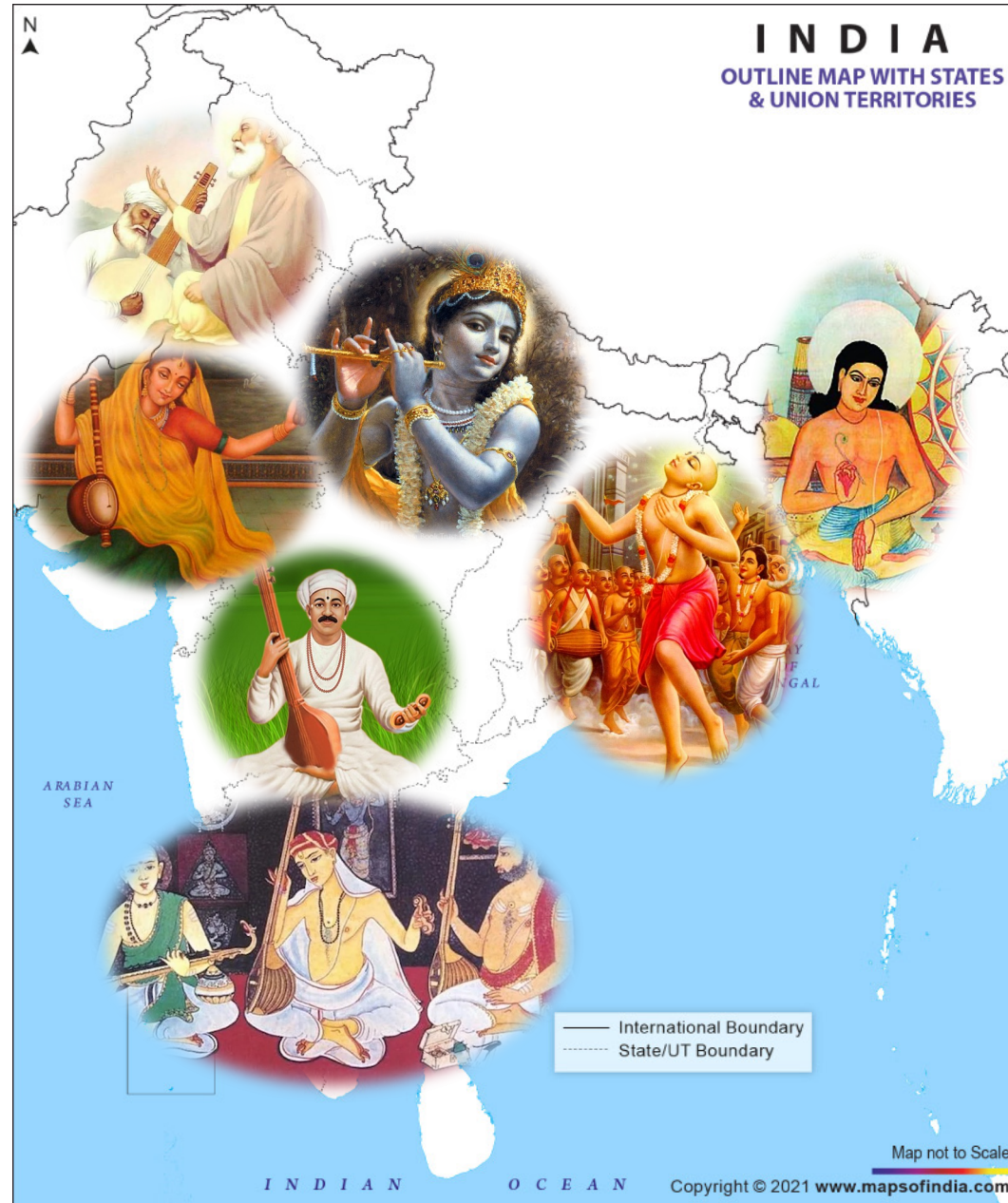
Vipul Arora
Department of EE, IIT Kanpur



MADHAV
Machine Analysis of Data
for Human Audition and Vision *lab*

WiSSAP Class Rules

1. Ask questions
2. Ask questions
3. Ask questions
4. Ask questions
5. ...



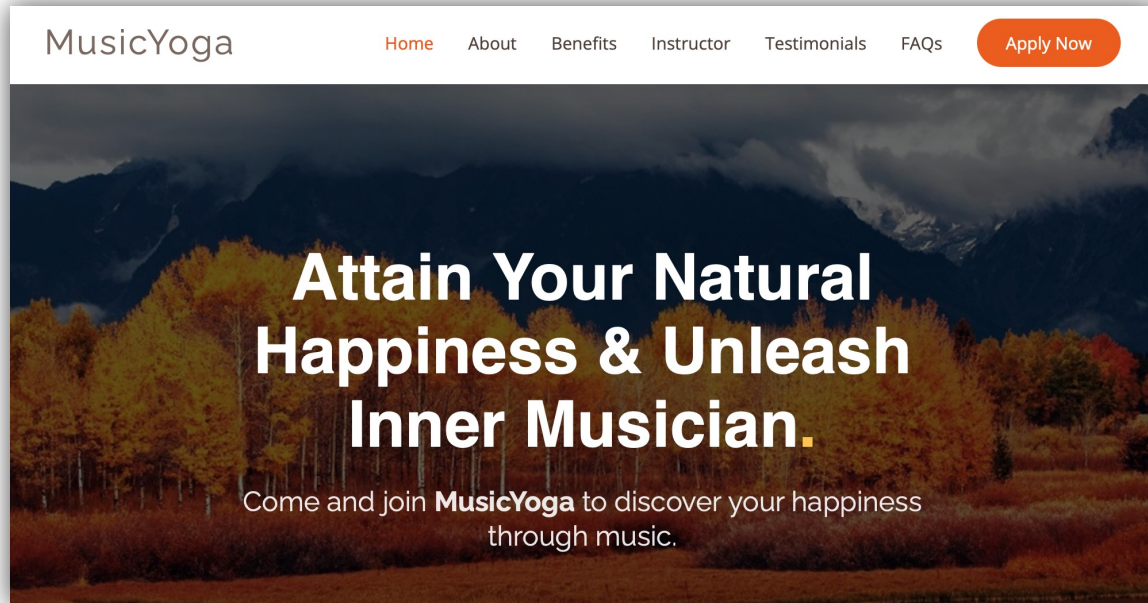
Source: internet

Music in Society



Source: internet

Music and Health



The screenshot shows the homepage of the MusicYoga website. At the top left is the logo "MusicYoga". To its right is a navigation menu with links for "Home", "About", "Benefits", "Instructor", "Testimonials", and "FAQs". Further right is an orange button labeled "Apply Now". The main content area features a background image of a forest with mountains in the distance. Overlaid on this image is the text: "Attain Your Natural Happiness & Unleash Inner Musician." Below this, in smaller text, it says: "Come and join **MusicYoga** to discover your happiness through music."



Source: internet

Music and Development



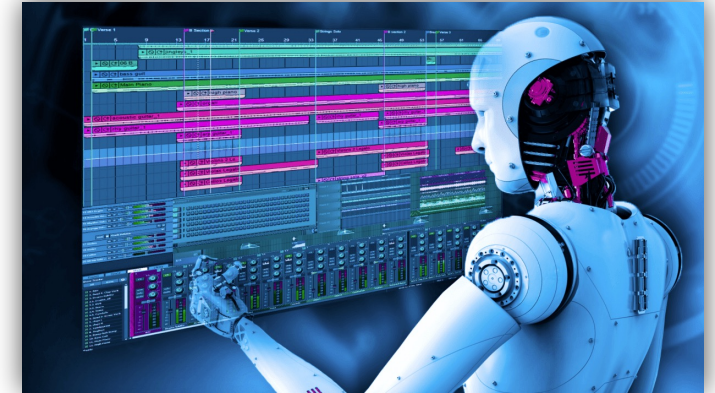
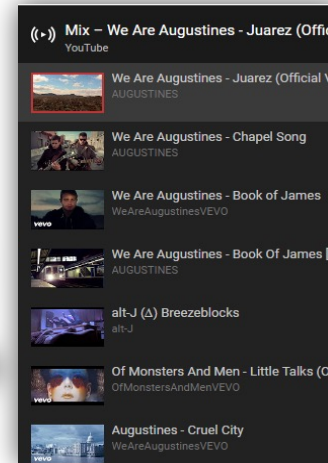
Audio Signal Processing and AI



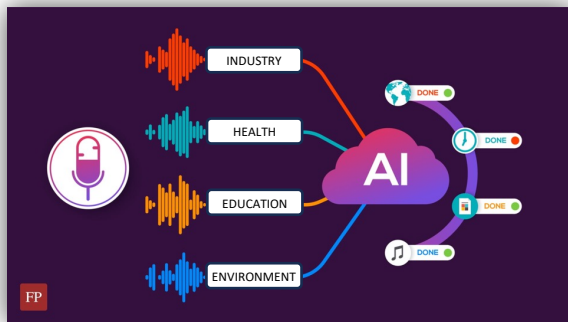
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Listen

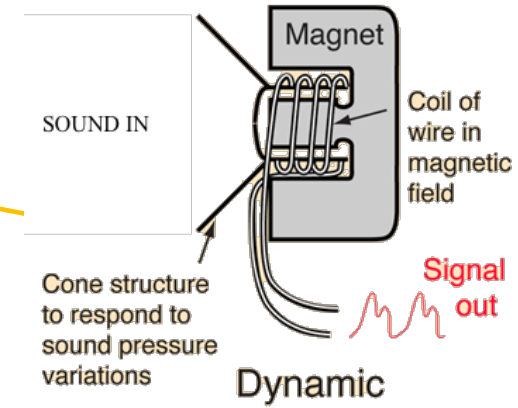
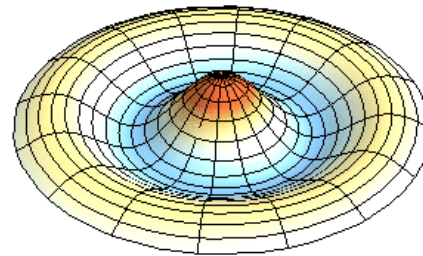


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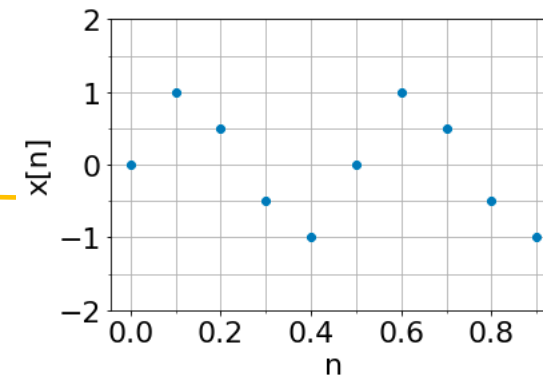


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Digital Audio



<http://hyperphysics.phy-astr.gsu.edu/hbase/Audio/mic.html>



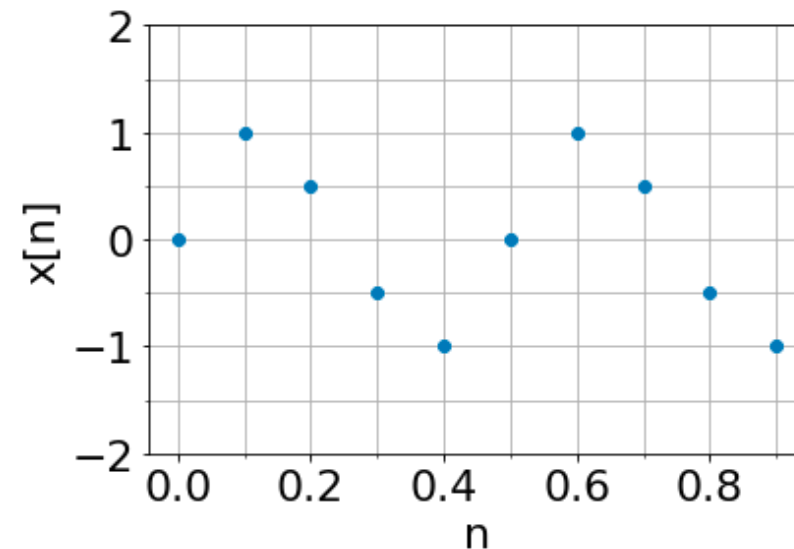
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Sampling

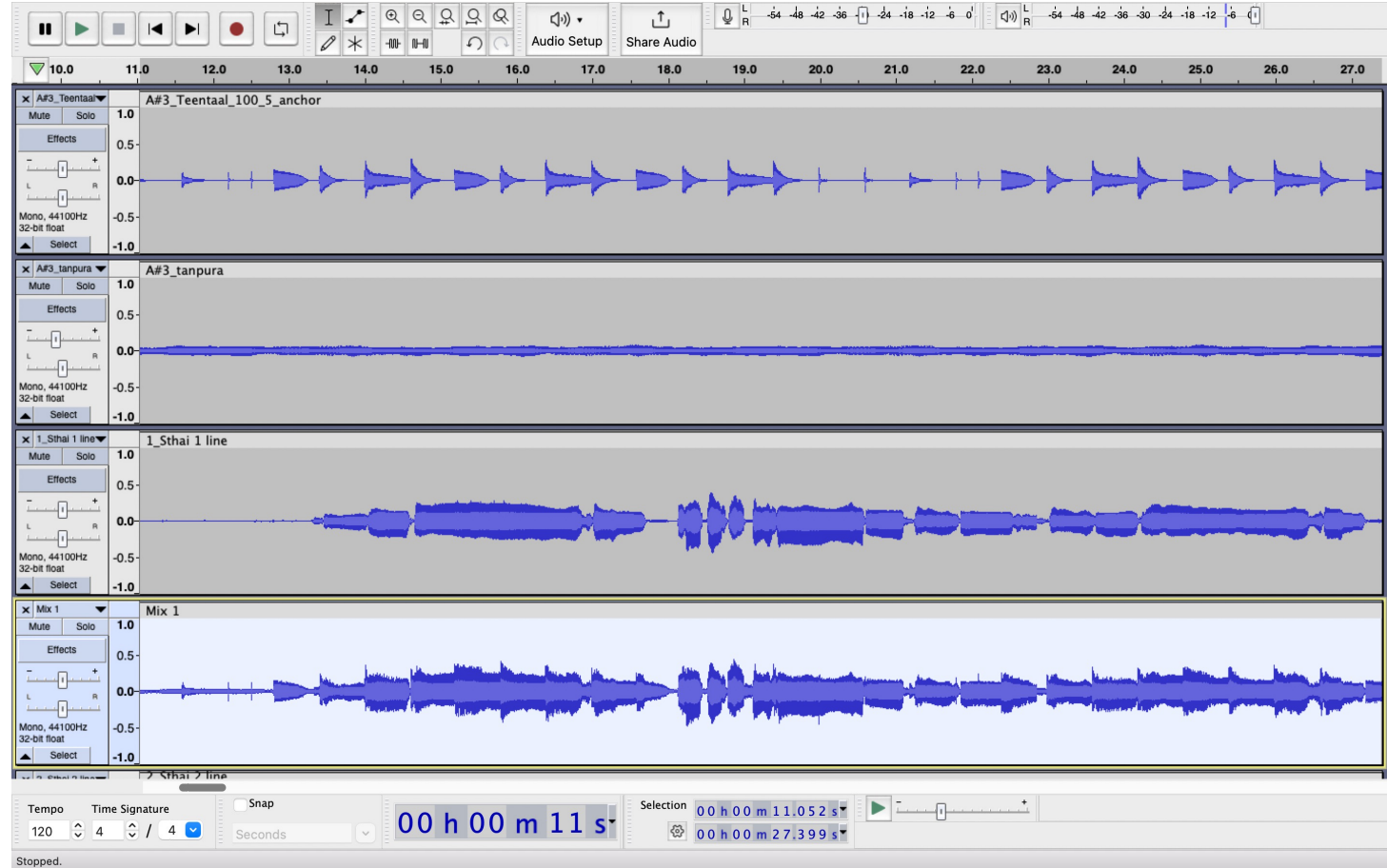
- Nyquist Sampling theorem
- Humans can hear in the range 20Hz to 20kHz
- Popular: 44.1kHz for CD recordings

Quantization

- Converting $x \in \mathbb{R}$ to a digital number
- Q bits per sample $\Rightarrow 2^Q$ possible integer values per sample
- Popular: 16 bits per sample for CD recordings

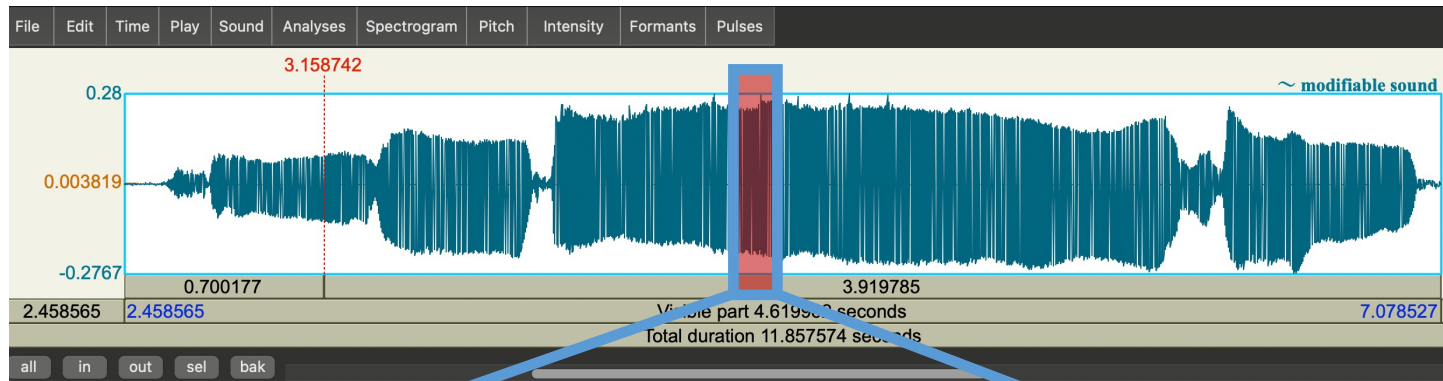


Waveforms

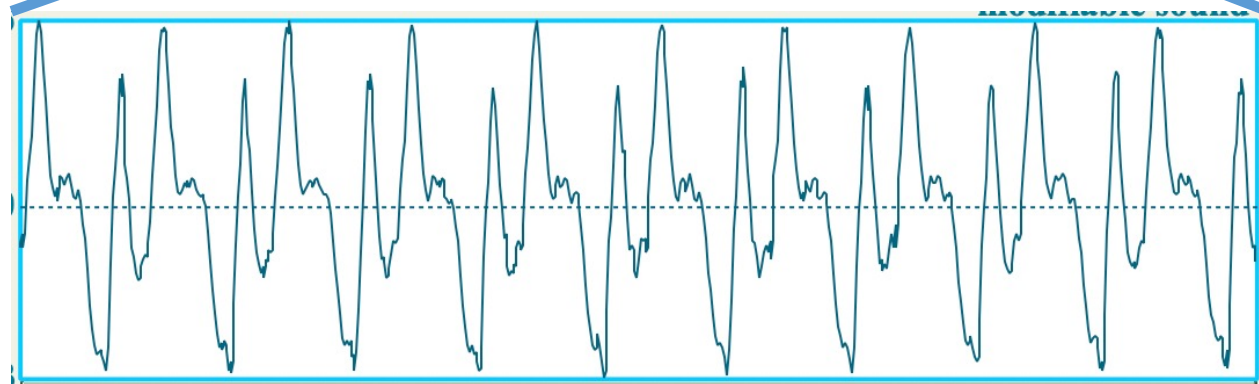


Audacity

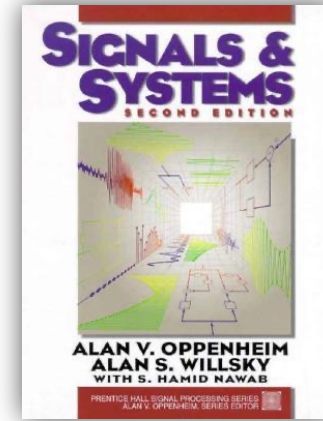
Processing ... we need mathematical model



Praat

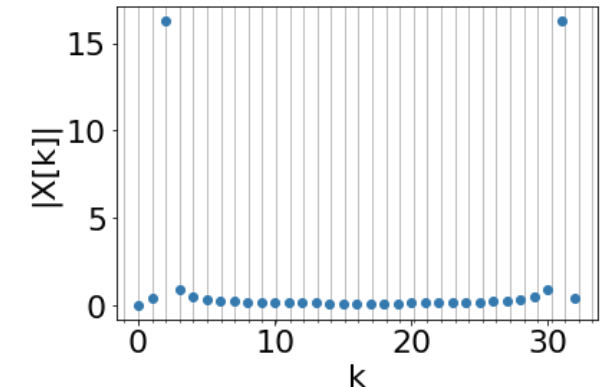
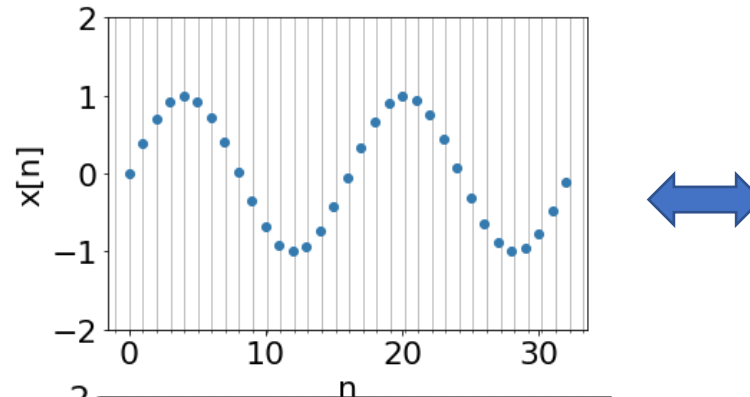


Fourier Transform

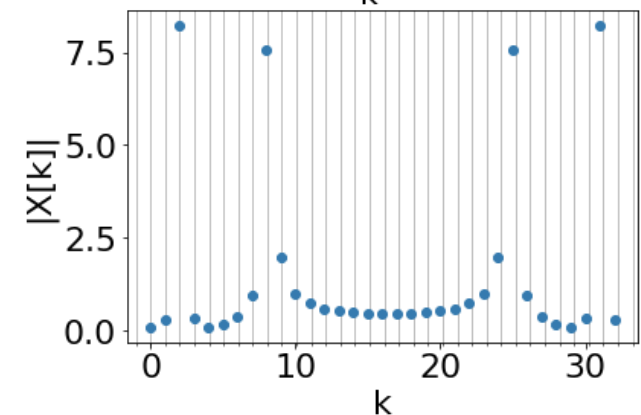
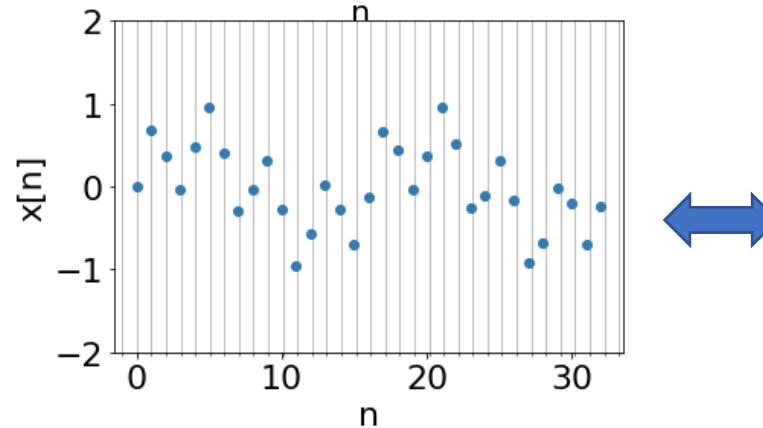


$$X[k] = \sum_{n=0}^{N-1} x[n] e^{-j\frac{2\pi}{N}kn} ; n = 0, \dots, N - 1$$

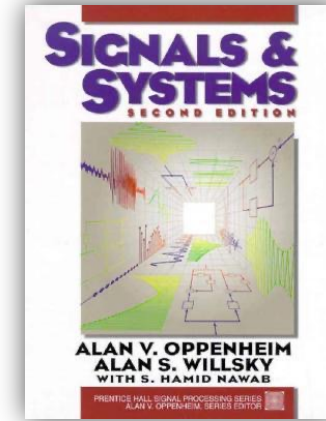
$$x[n] = \sin(2\pi * 2/32 * n)$$



$$x[n] = 0.5 * \sin(2\pi * 2/32 * n) + 0.5 * \sin(2\pi * 8/32 * n)$$

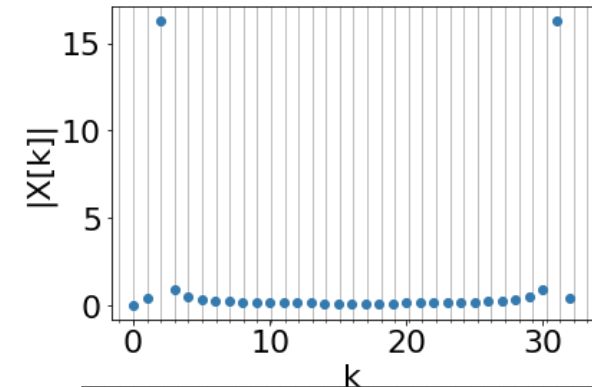
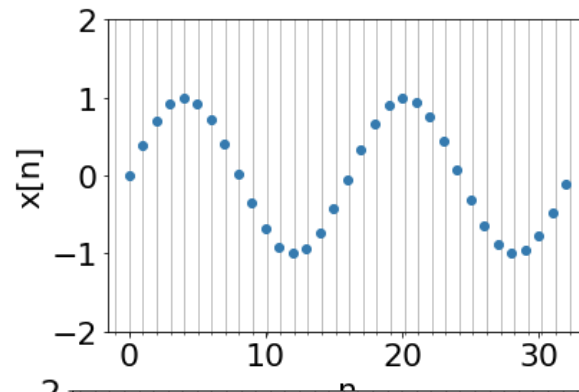


Fourier Transform

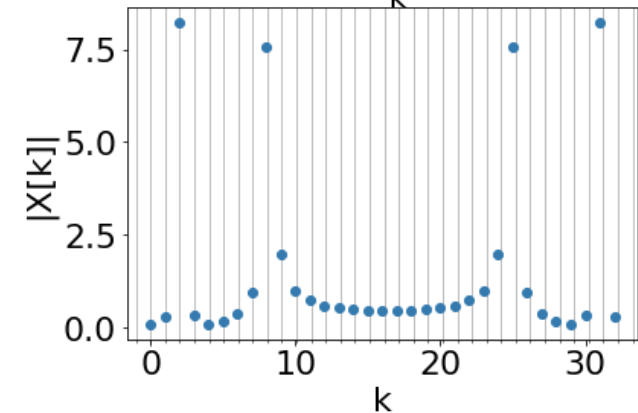
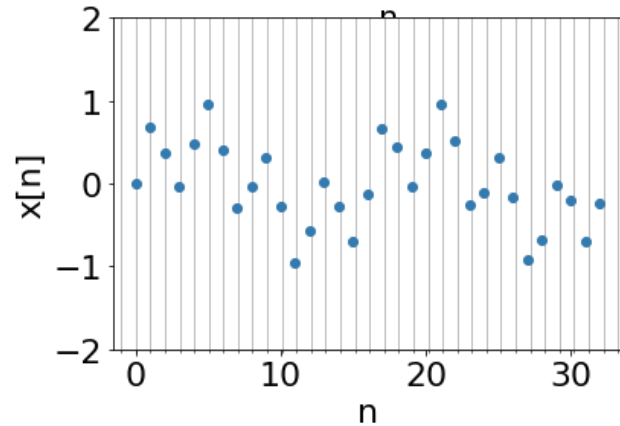


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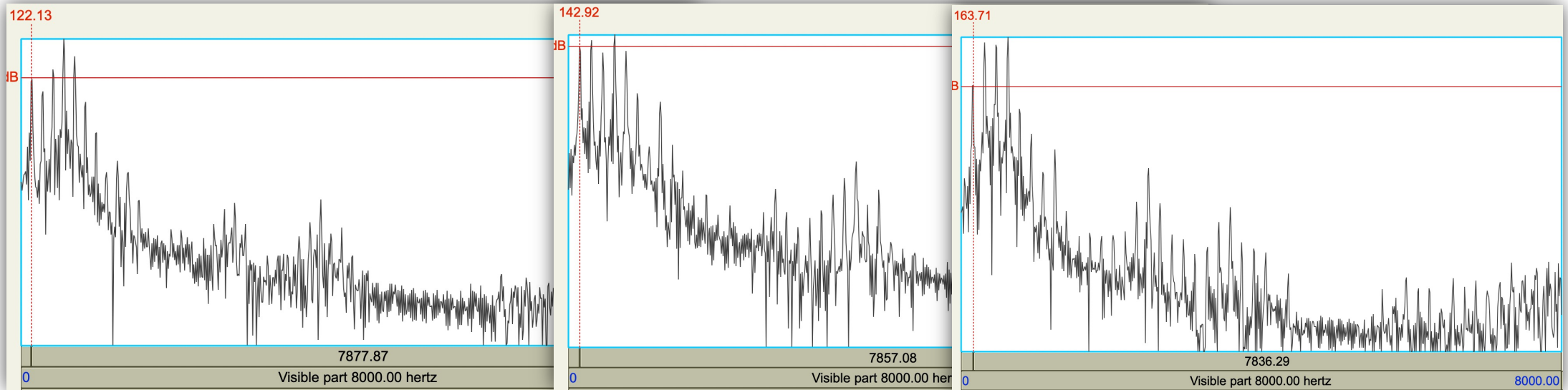
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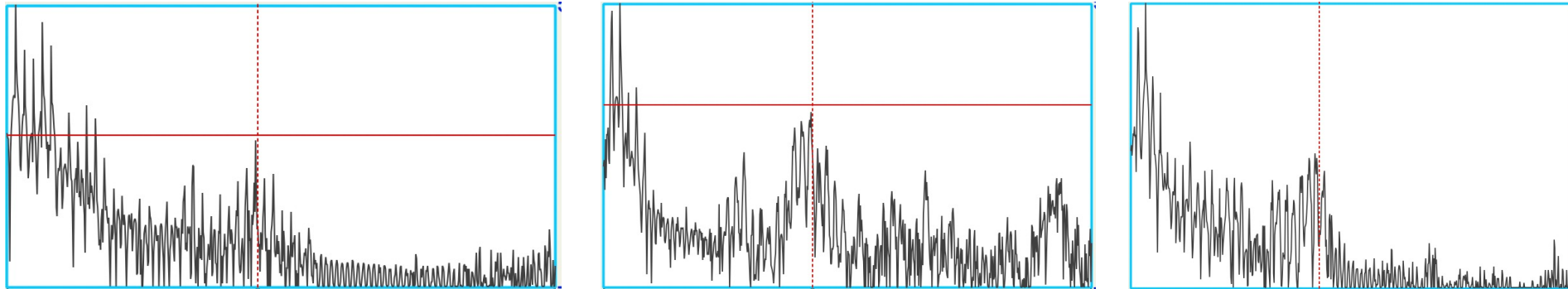


Varying the Pitch

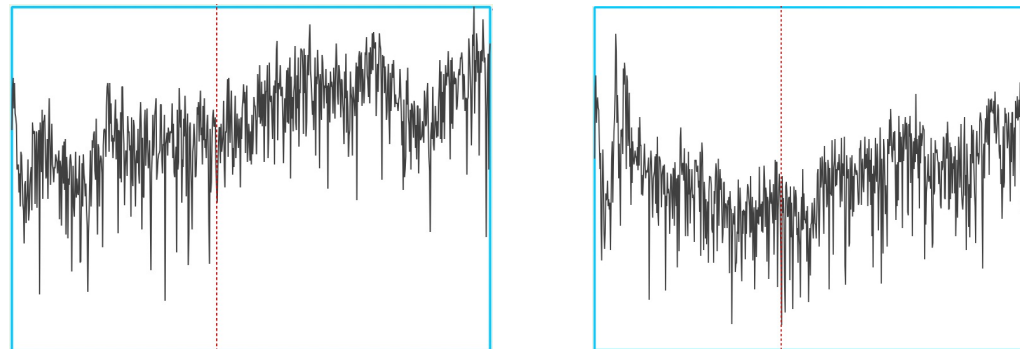


Spectra of speech

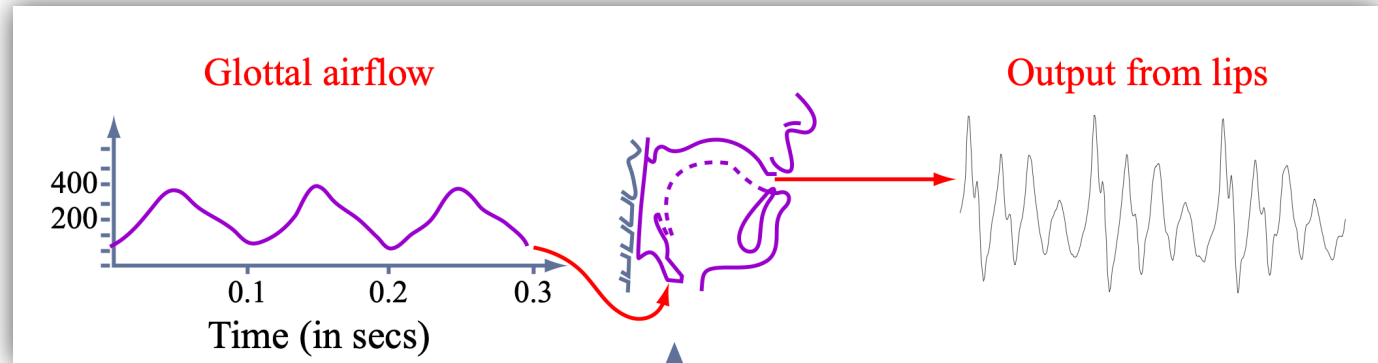
- Sounds with periodic waveforms: /a/, /i/, /m/



- Sounds with aperiodic waveforms: /s/, /f/



Filter Theory

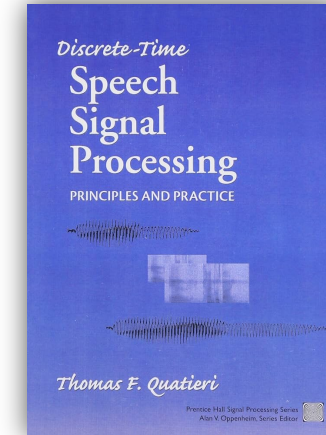
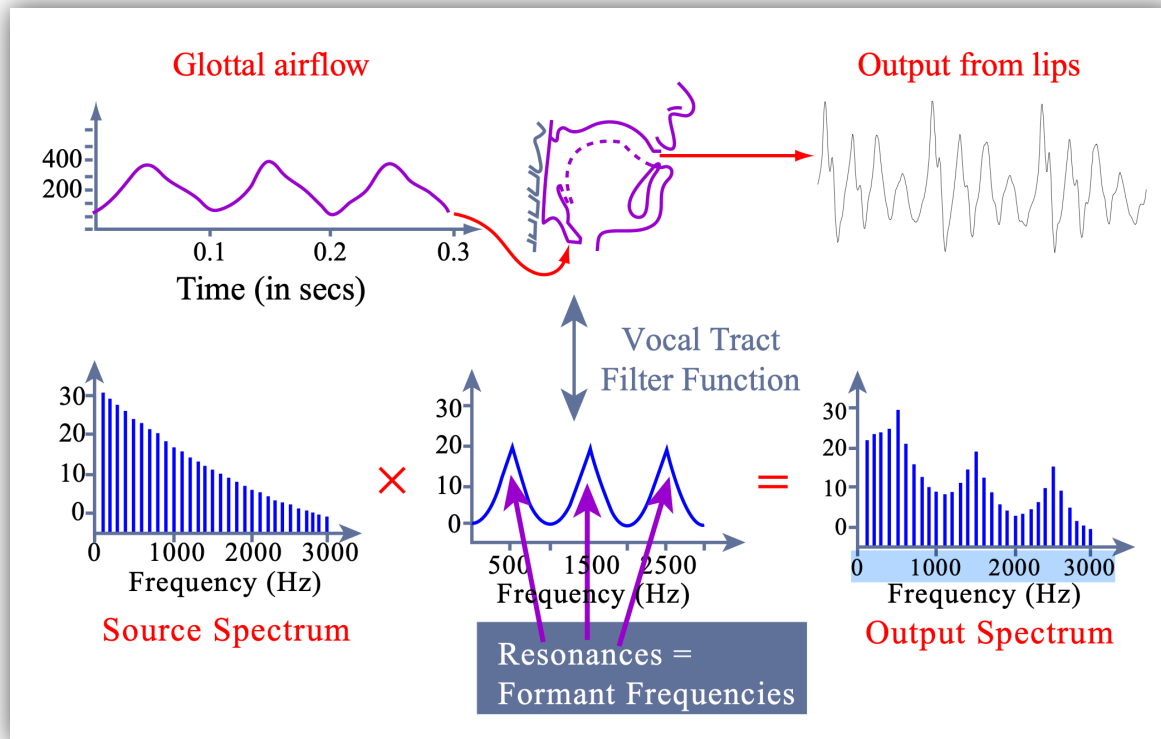


$$y[n] = x[n] * h[n] = \sum_{k=-\infty}^{\infty} x[k]h[n - k]$$

$$Y[k] = X[k]H[k]$$

Source Filter Model

- Linear Time-Invariant (LTI) filters



How do I make real applications
with this?

Designing Representations

Representations should be

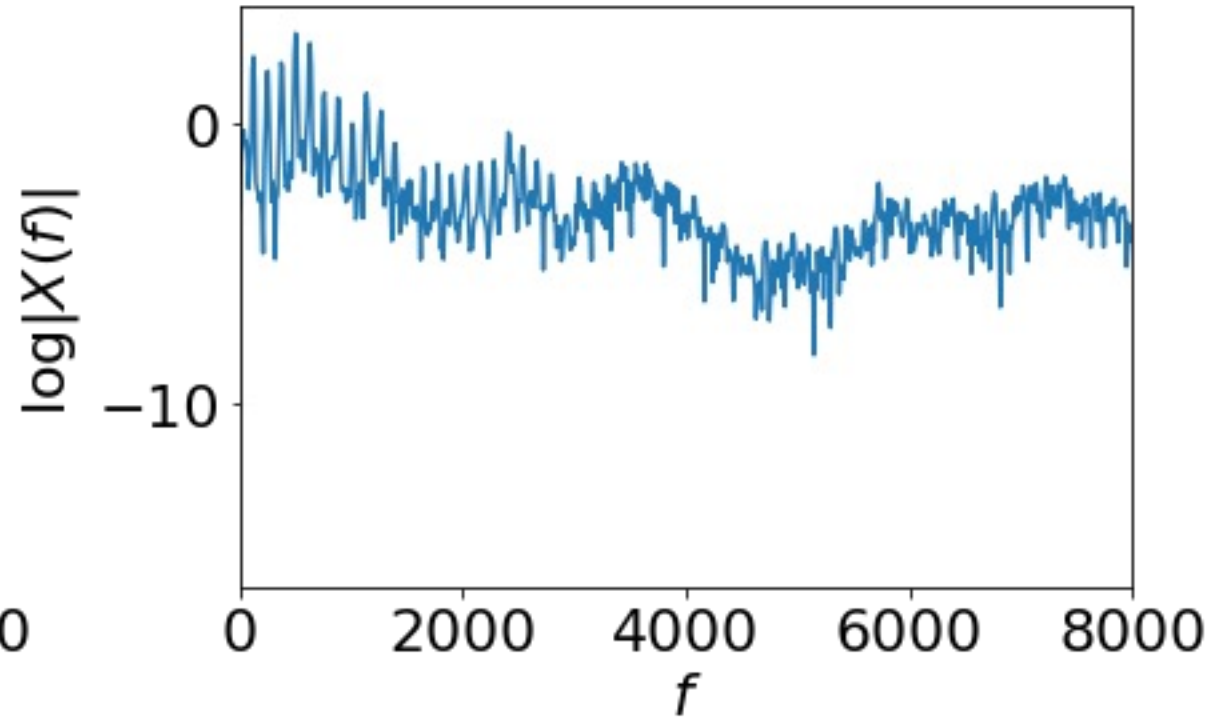
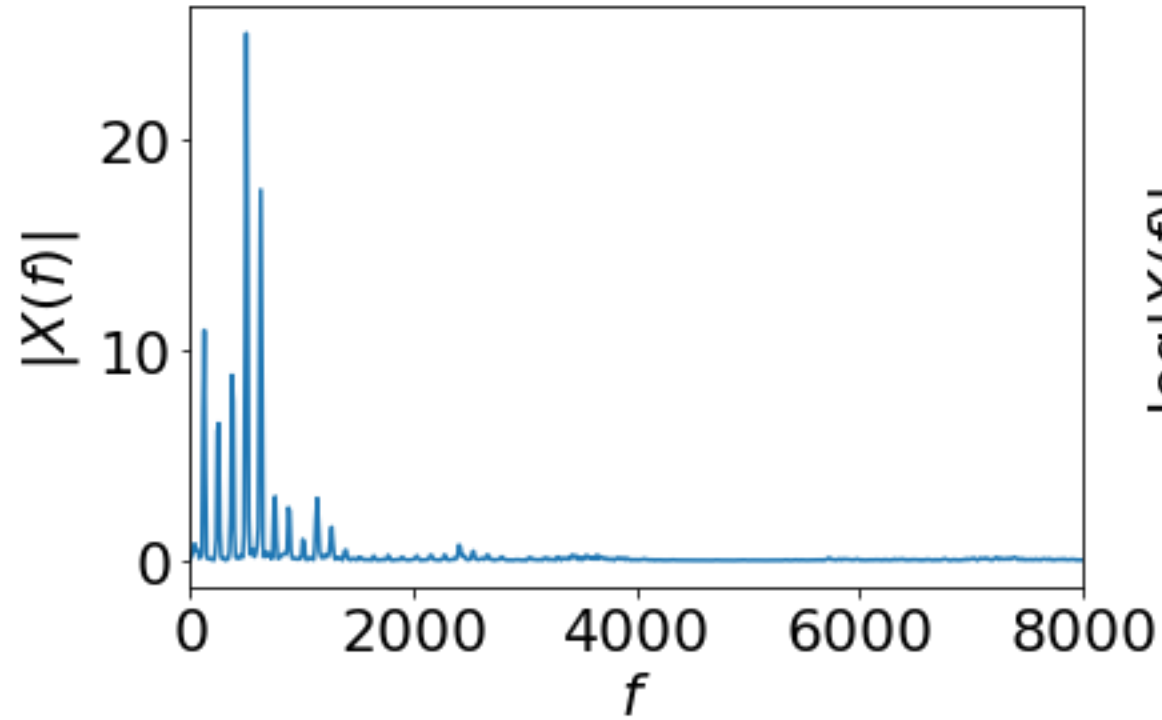
- **minimal** in size
- **distinguishing** for what we are interested in
- **invariant** to what we are not interested in

- Design the space so it may have **uniform sensitivity** (more in Audio Retrieval hands-on by Anup)

Designing Representations

- for Pitch
 - Look at the peaks of spectrum
- for instrument/phoneme
 - Look at the spectral envelope

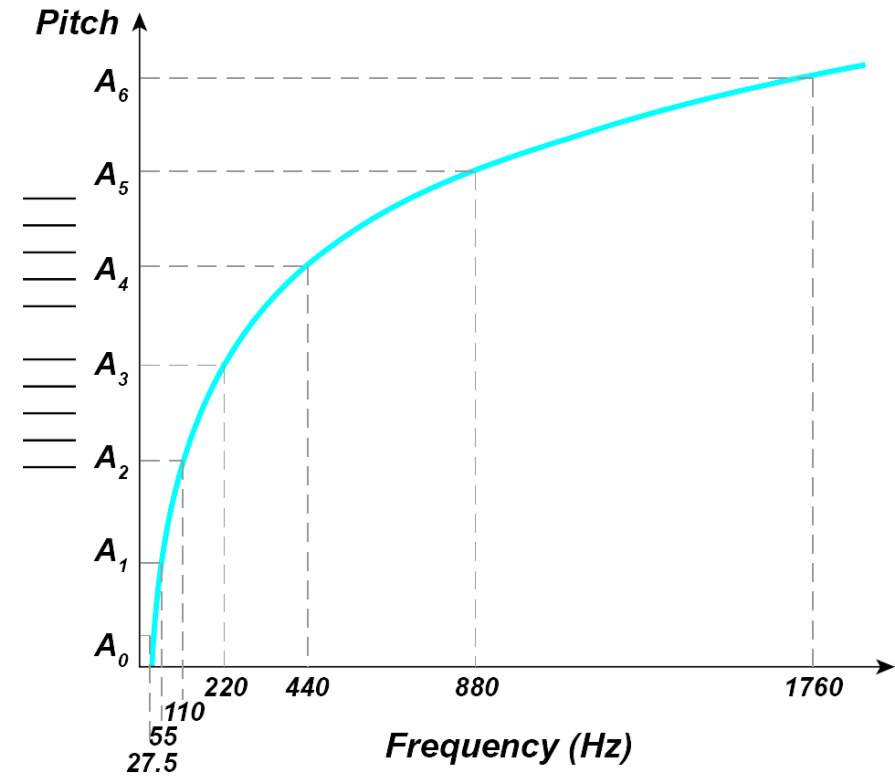
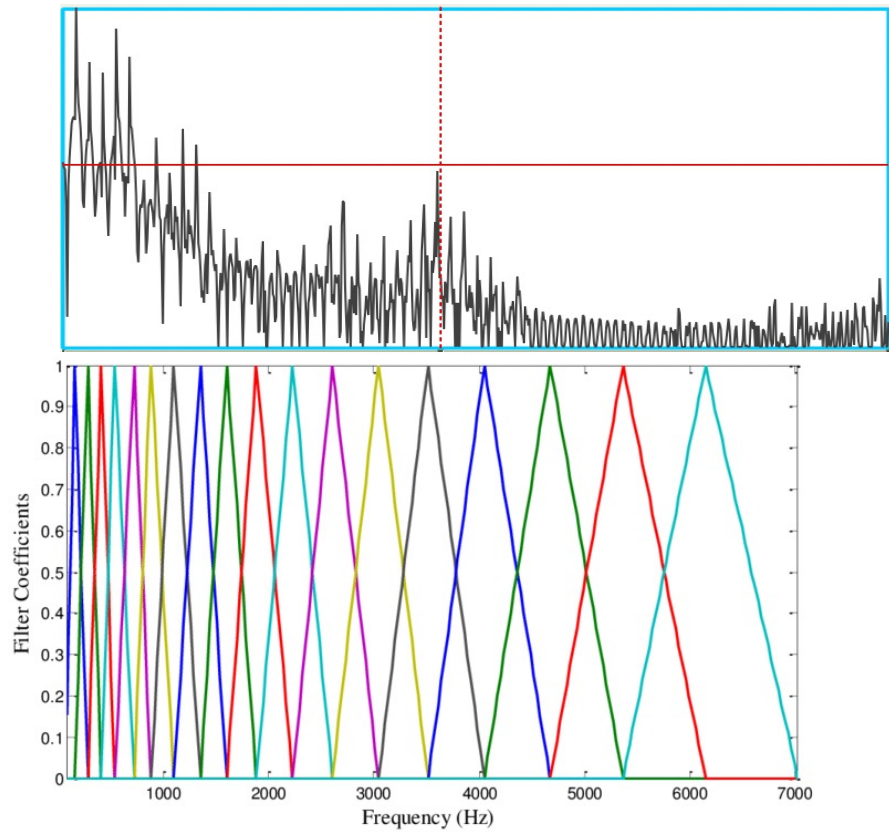
Amplitude



$$A_{dB} = 20 \log_{10} A$$

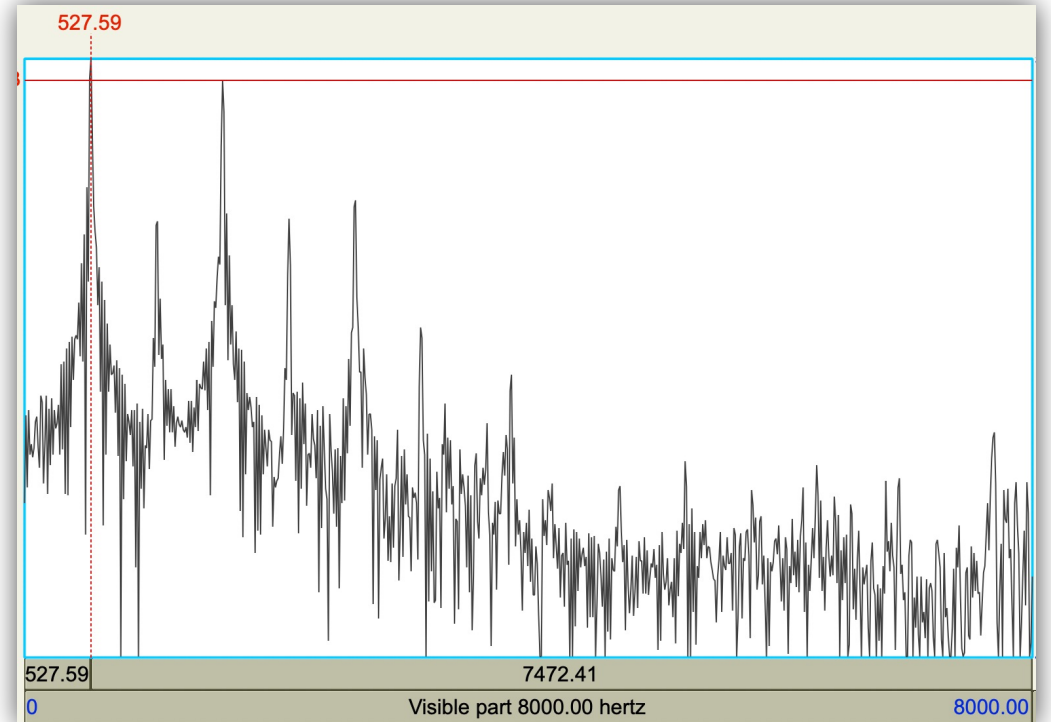
Frequency

- $\tilde{f} \propto \log f$



Spectral Envelope

- $|X[\tilde{f}]|_{dB}$
- Mel-frequency, dB amplitude
- Take low frequency components of Fourier transform (DCT) of $|X[\tilde{f}]|_{dB}$



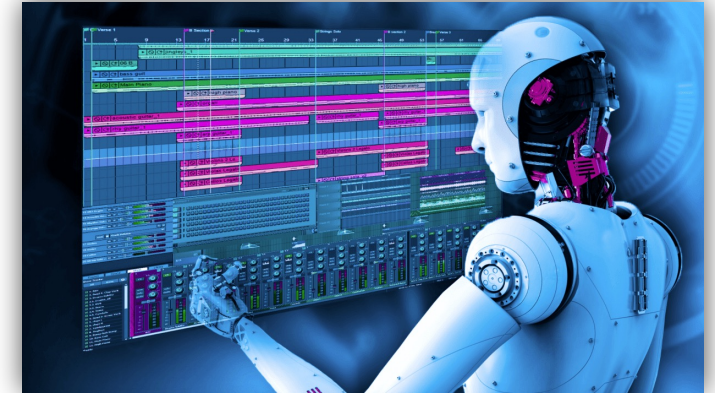
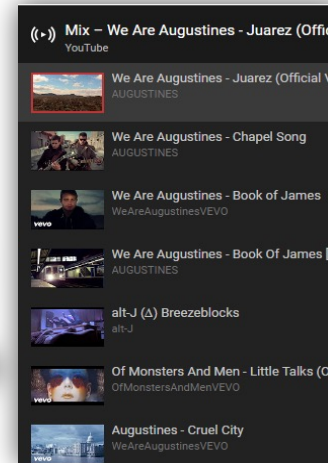
You are ready!!!



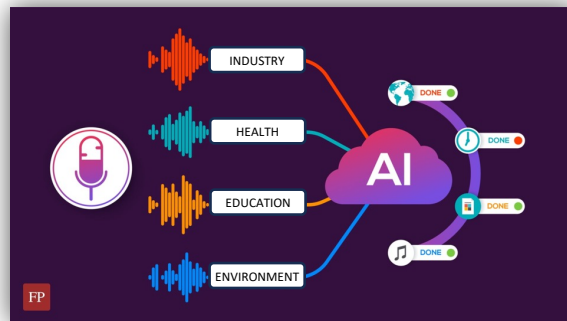
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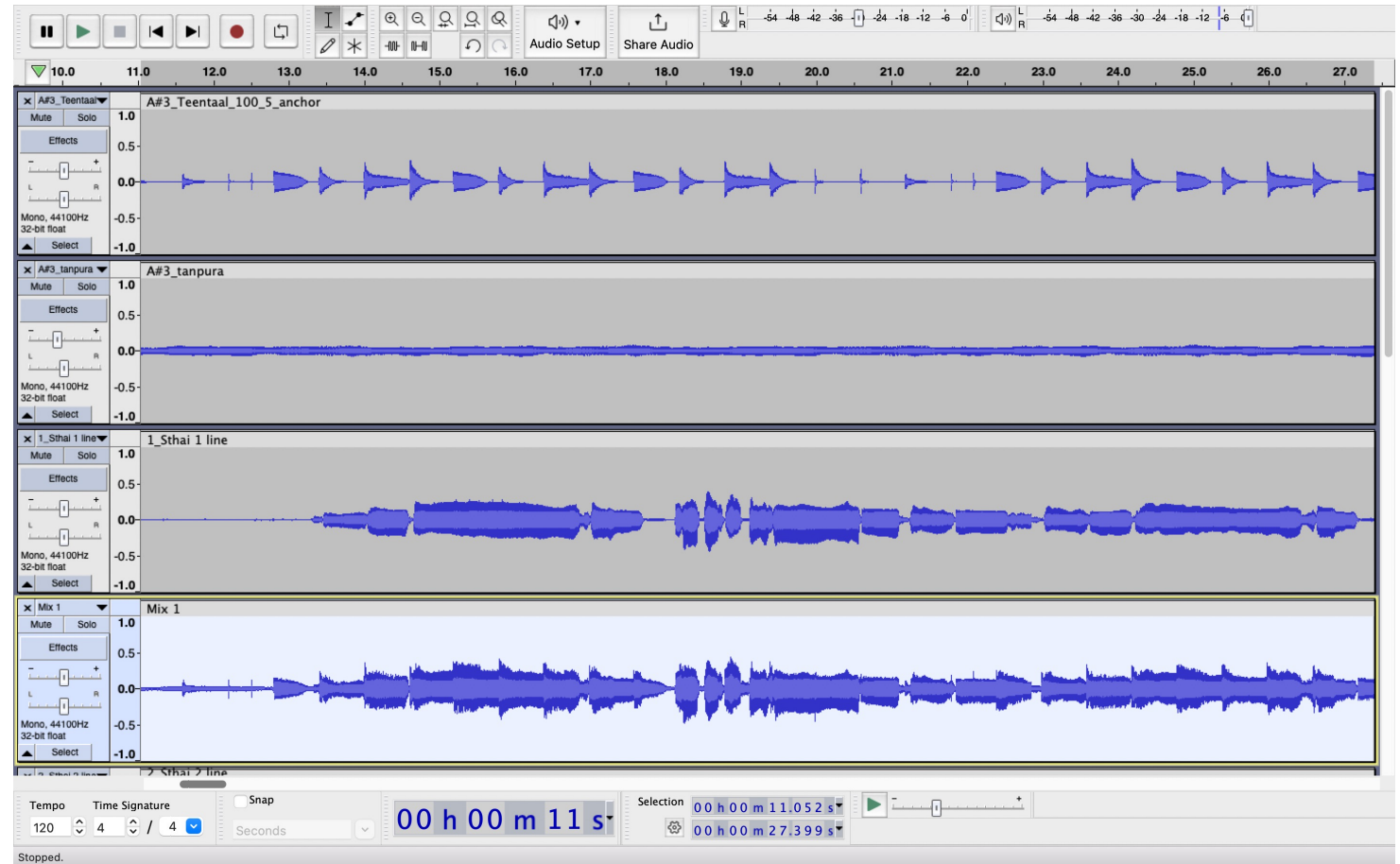


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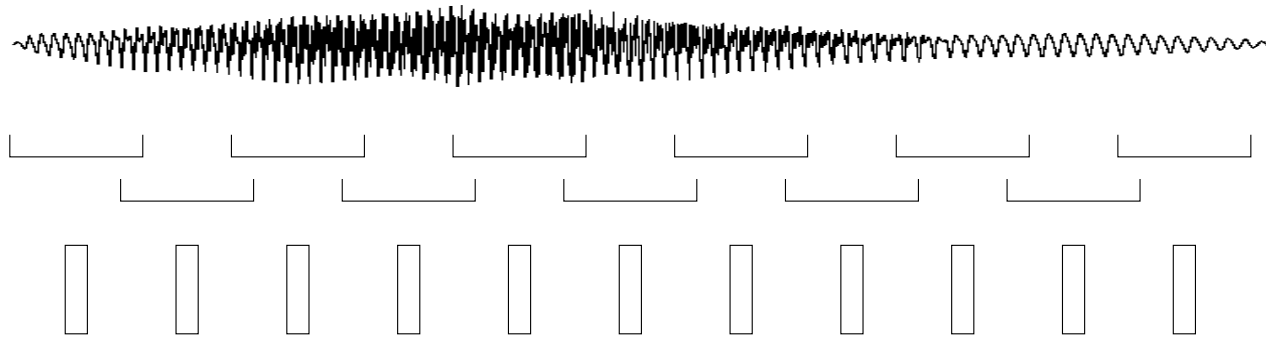


Not yet

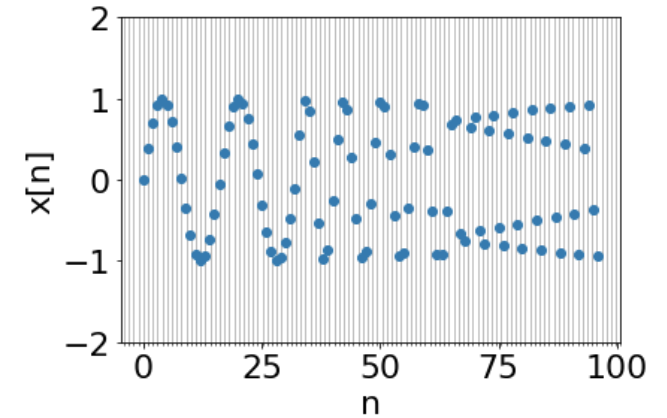
- Dynamic behavior
- Time Series Analysis



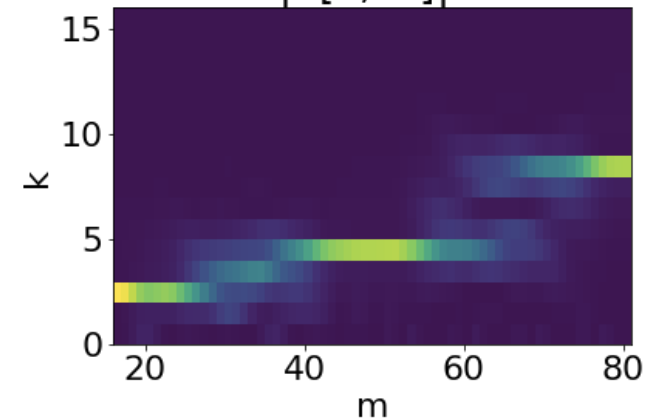
Short Time Fourier Transform



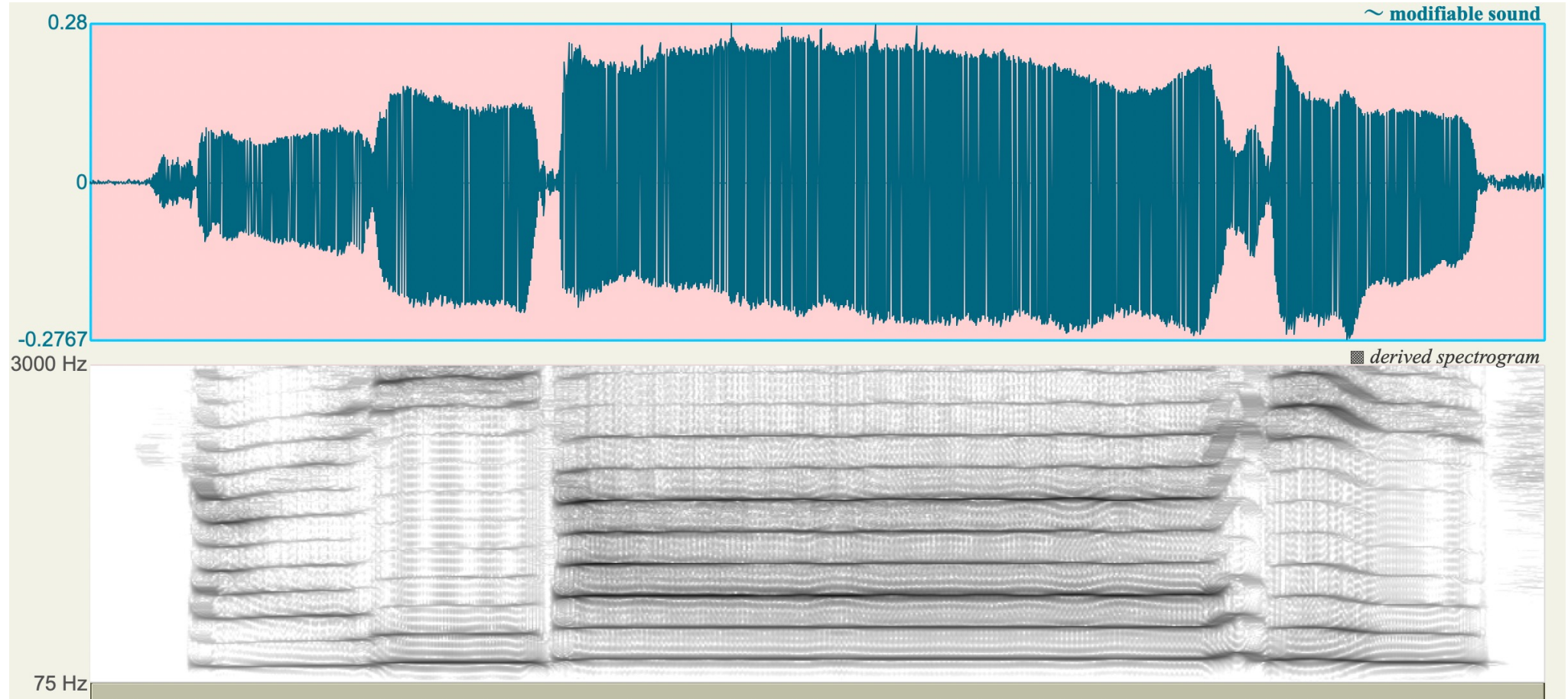
$$X[k, m] = \sum_{n=0}^{N-1} x[n]w[n - mH]e^{-j\frac{2\pi}{N}kn}; \quad k = 0, 1, \dots, N - 1$$



$|X[k, m]|^2$

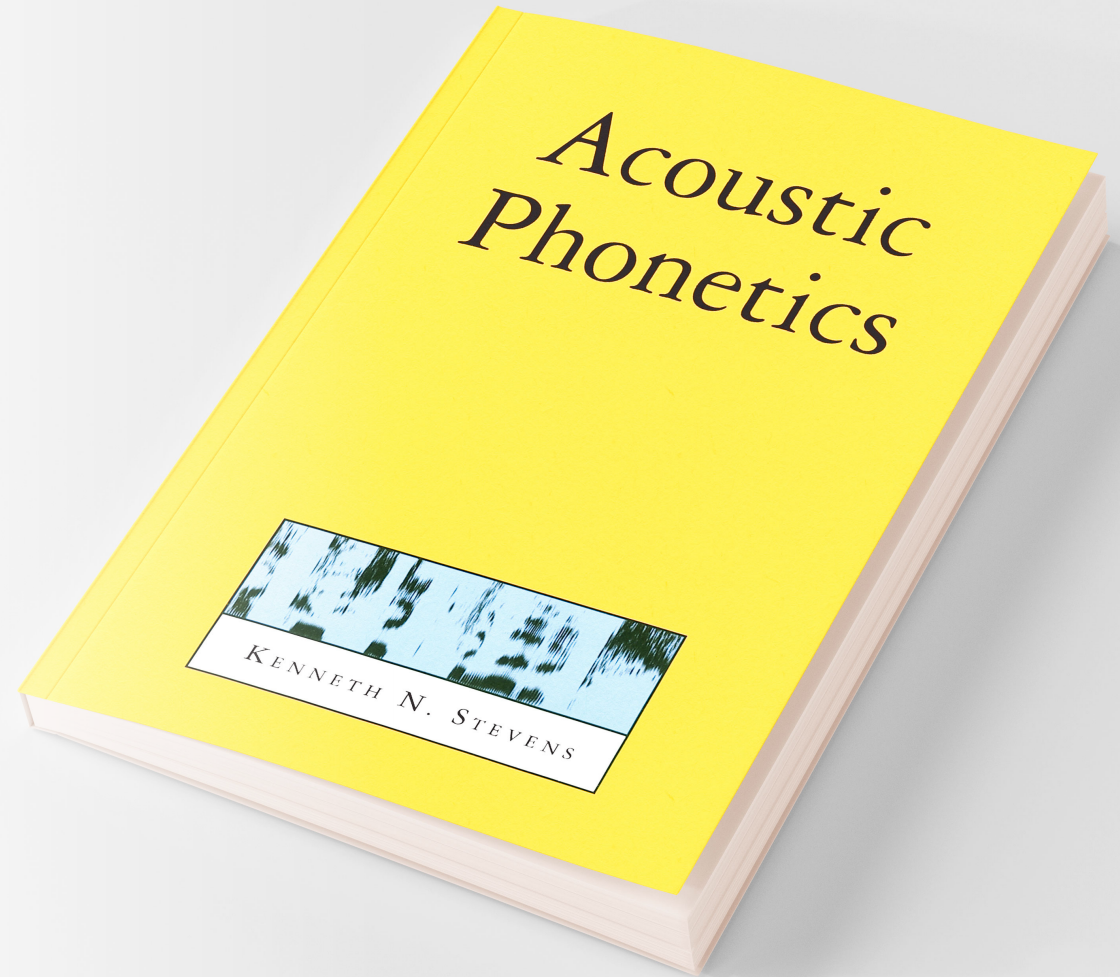


Short Time Fourier Transform



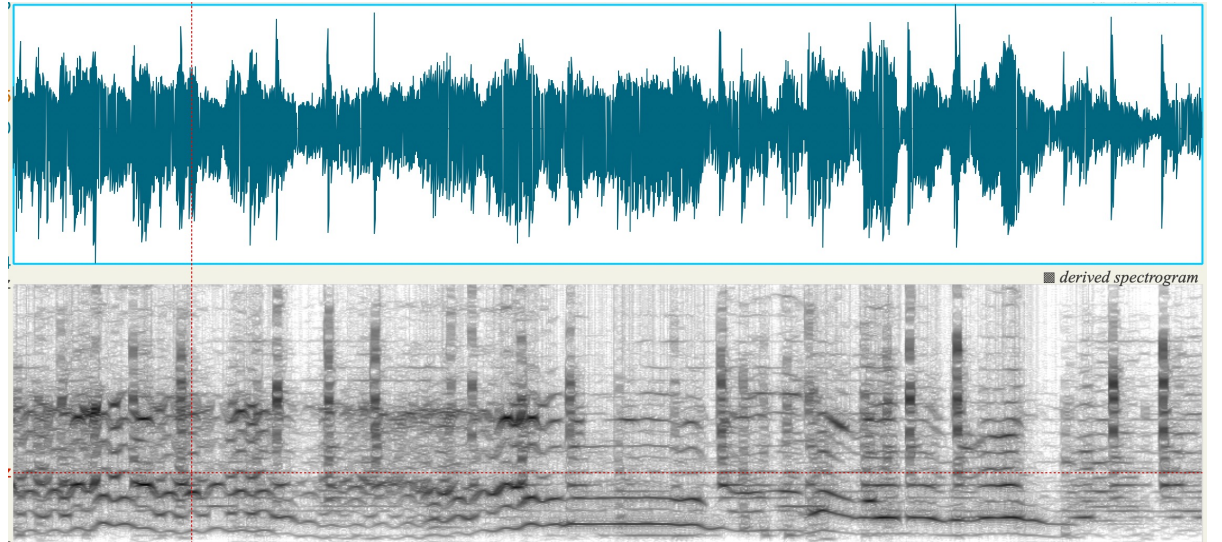
It is possible

But only in ideal situations



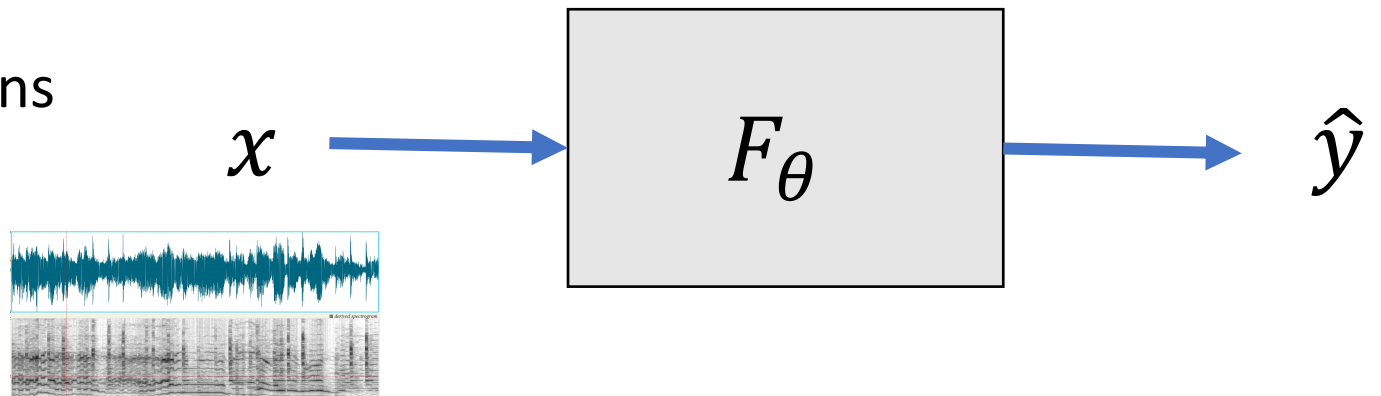
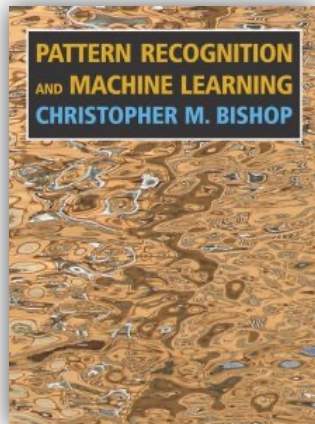
Real-world Variations

- Context (co-articulation)
- Running speech
- Speakers, instruments
- Languages
- Recording equipment
- Acoustic conditions



Machine Learning

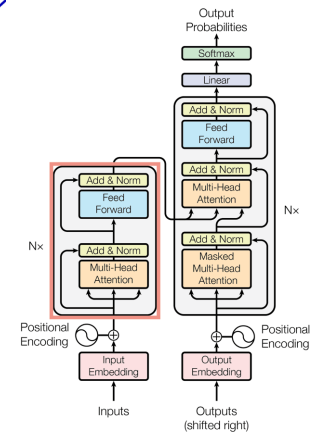
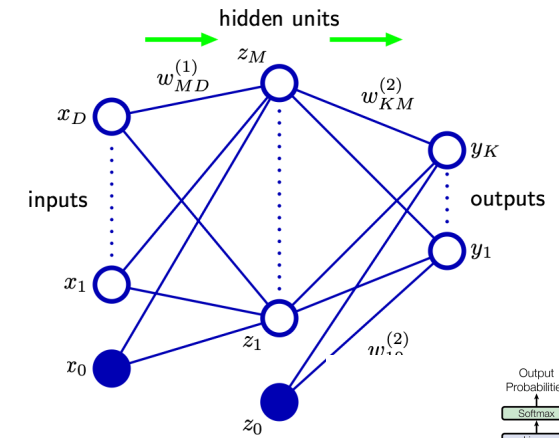
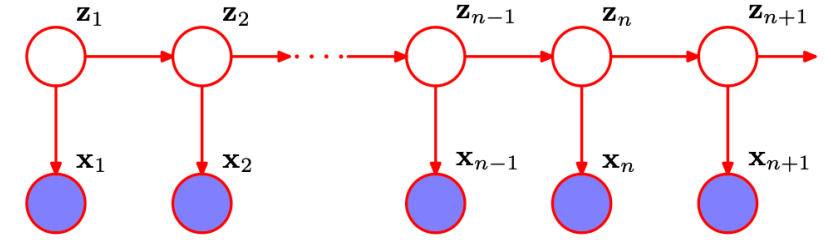
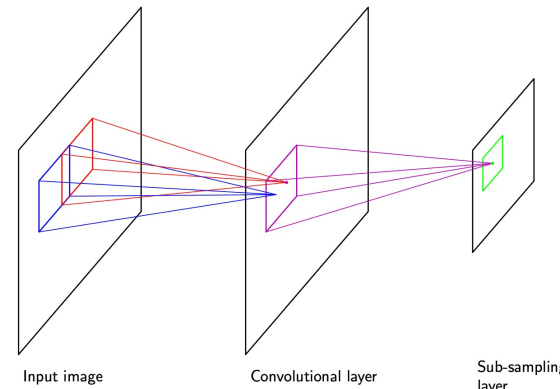
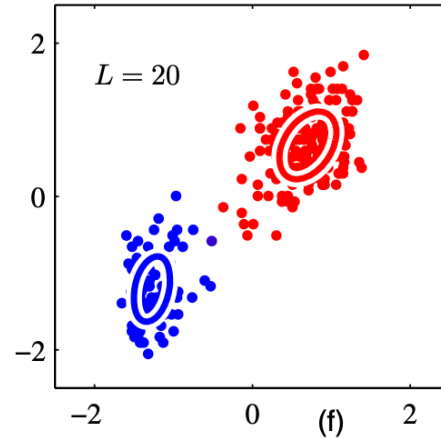
- Parametric models to learn the feature transformations
- Learn the mappings from
 - speech to text
 - audio to audio
 - audio to labels/classes
 - audio to recommendations



$$\theta = \operatorname{argmin}_\theta \mathcal{L}(y, \hat{y}; \theta)$$

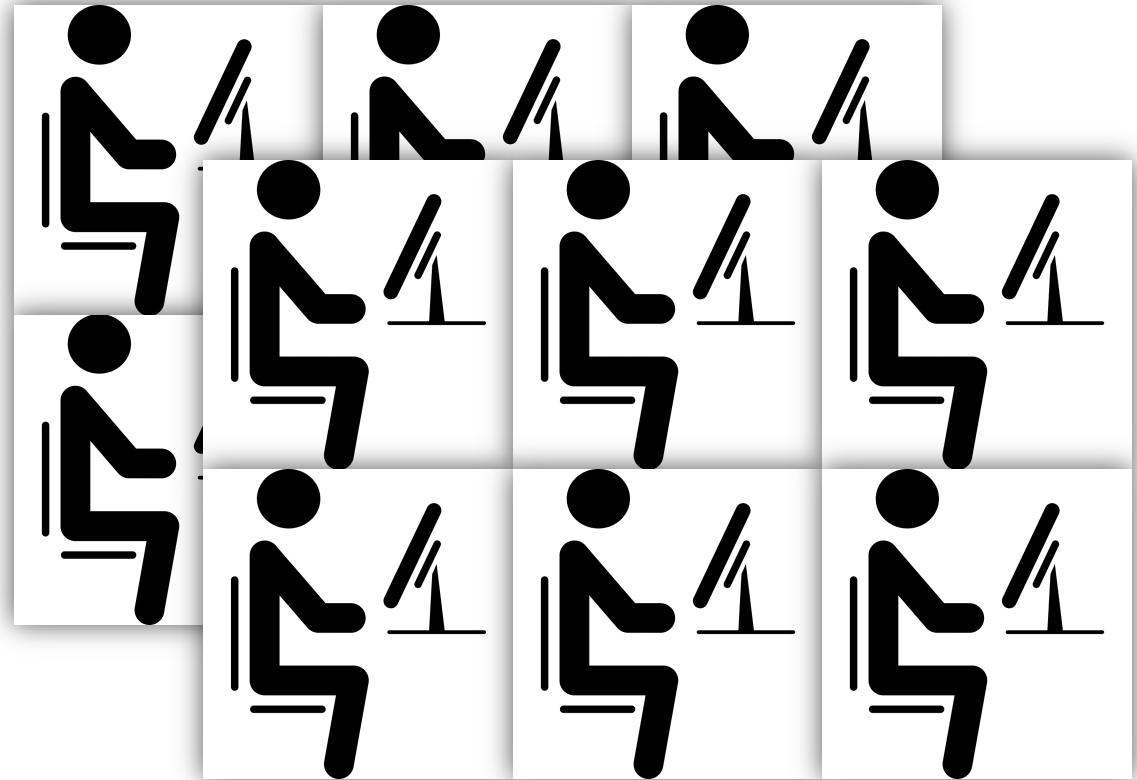
Supervised Learning

- Gaussian Mixture Model
- Hidden Markov Model
- Multi-Layer Perceptron
- Support Vector Machine
- Convolutional Neural Network
- Recurrent Neural Network
- Transformers

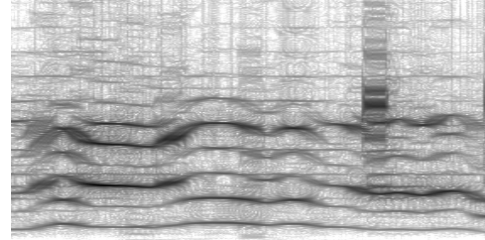


Source: PRML Bishop and
<https://proceedings.neurips.cc/paper/2017/file/3f5ee243547dee91fbd053c1c4a845aa-Paper.pdf>

Bottleneck (x, y)



Self-supervised Learning



Use F_θ instead
of hand
designed
representations!

