# Design and Implementation of Speech Recognition Systems

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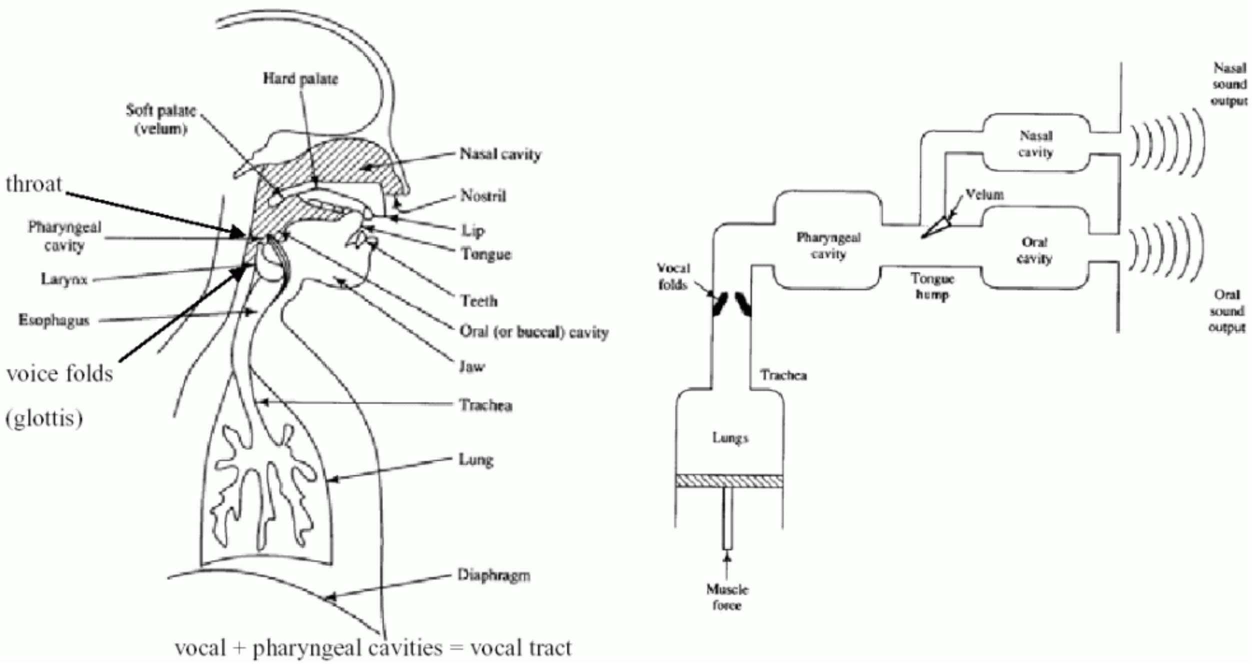
## Class 2: Data Capture

22 Jan 2014

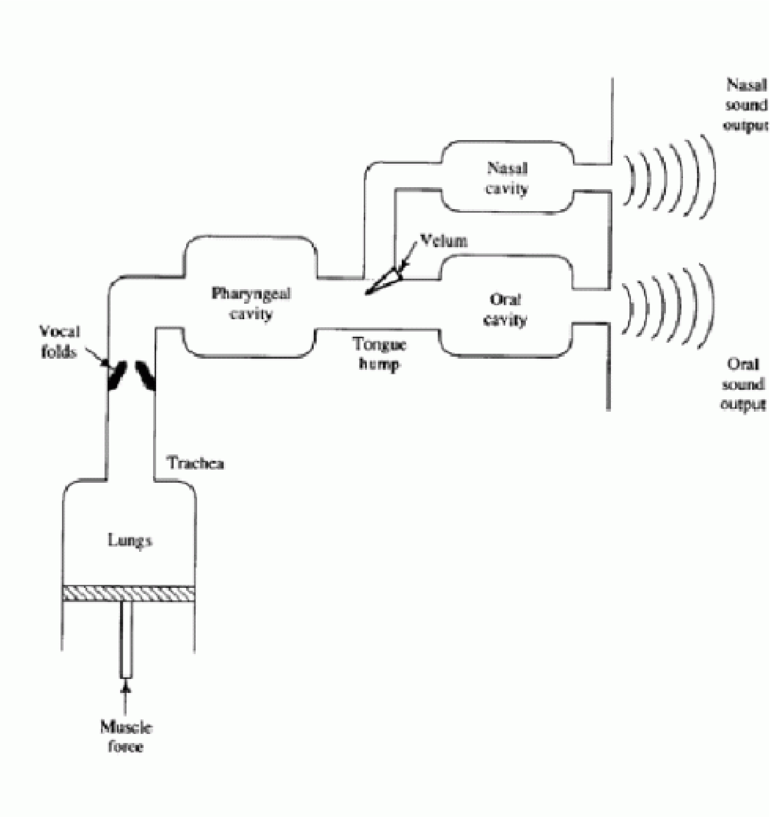
# Producing the Speech Signal



vocal + pharyngeal cavities = vocal tract

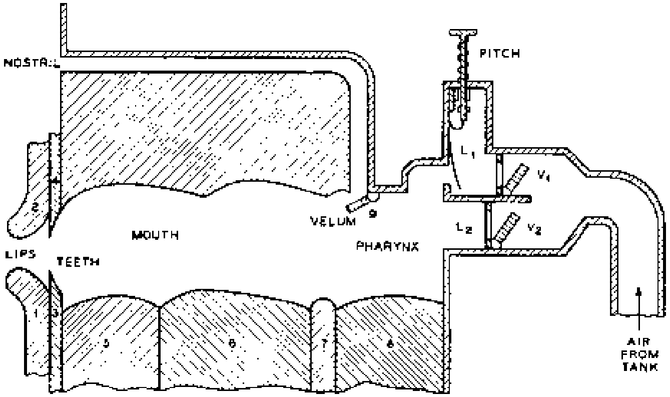
- All sounds are actually pressure waves
- The speech producing mechanism is resonant chamber that produces pressure waves
  - Left: Cross section of human vocal tract
  - Right: A physical “model”

# The speech production mechanism



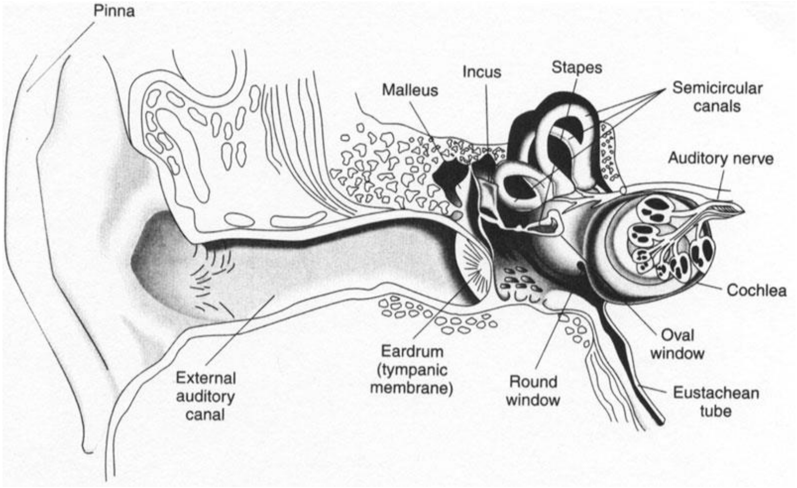
- The lungs are like bellows: they pump air out to produce speech
  - You only speak while breathing *out*
- The vocal folds may vibrate to add a basic (fundamental) frequency
  - To get semi-periodic puffs of air going up the vocal tract
- The muscles shape the vocal tract to add other resonances (and anti-resonances) in the air traveling up the tract
- The lips affect the final nature of the air puffs coming out of the mouth.

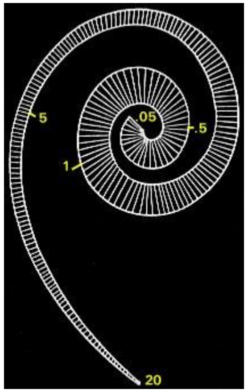
# Some Models



- Left: Wolfgang von Kemeplen’s bellows: 1773
  - – Imitated the bellows and tube nature of the vocal tract
    - Included a bellows, a reed (larynx) and tubes
- Right: Robert Reisz’s 1937 machine
- Could produce simple vowel- and consonant-like sounds

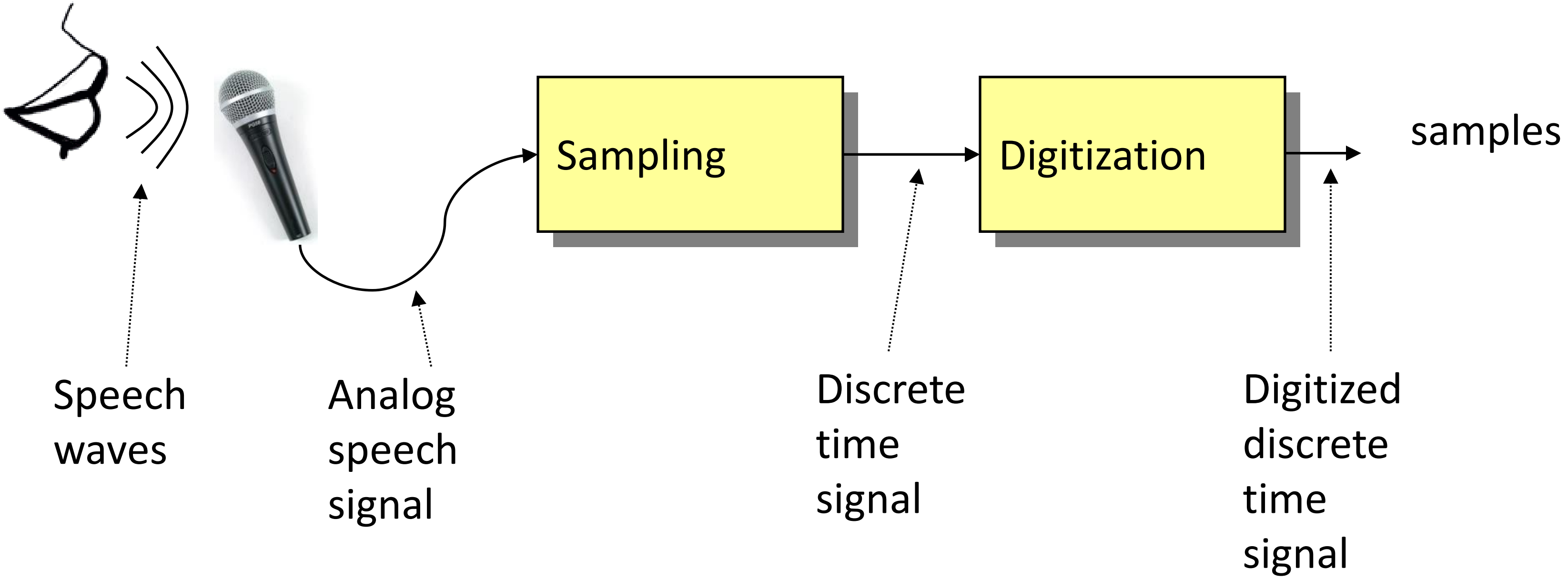
# The *Hearing* Apparatus





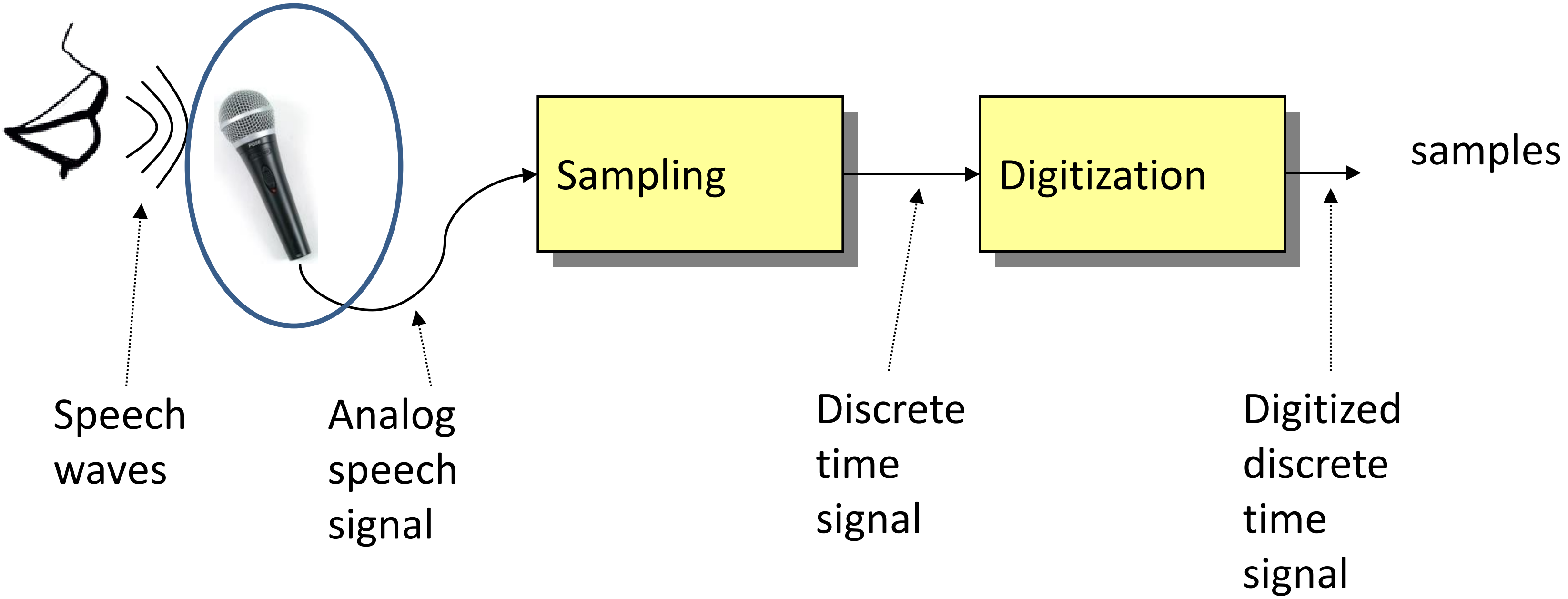
- The human auditory system consists of
  - A membrane that is moved back and forth by the pressure waves
  - A complex mechanism that converts the movement of this membrane to the motion of fluid
  - A collection of tuning forks that resonate to the fluid motion
  - Our brain interprets the movement of the turning forks
    - Actually hair fibres in the cochlea
- The key component is the moving membrane that translates pressure waves to the movement of fluid

# Signal Capture



- The goal of signal capture is to convert pressure waves to a series of numbers

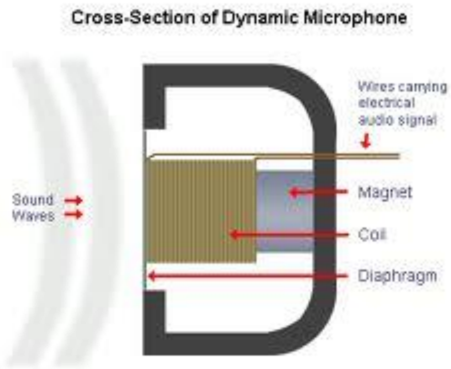
# First Stage: Microphones

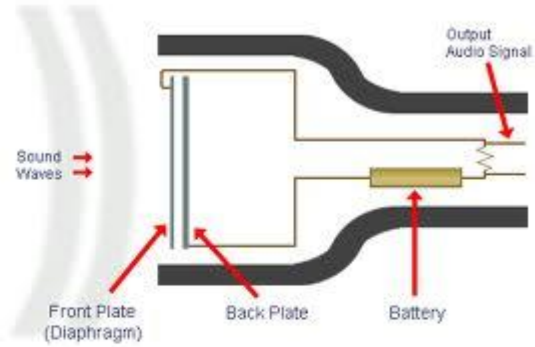


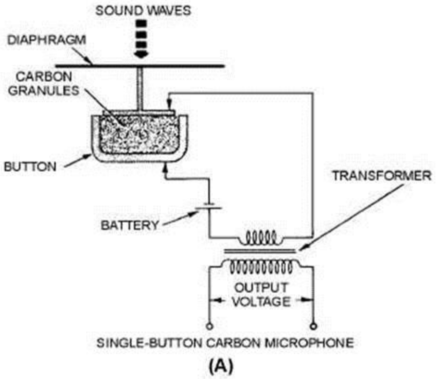
- The goal of signal capture is to convert pressure waves to a series of numbers

- To do so, we imitate the auditory system
  - A membrane that is moved by pressure waves
  - A transduction mechanism that converts the movement of the membrane to a meaningful signal

# Microphones


Cross-Section of Dynamic Microphone
Sound Waves
Wires carrying electrical audio signal
Magnet
Coil
Diaphragm

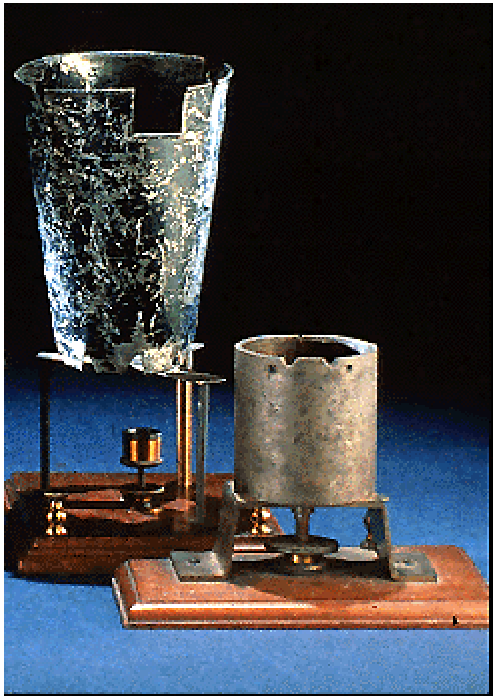




- Comprise
  - a membrane that is moved by the pressure wave
  - Some mechanism for converting this motion to a time-varying electrical signal
- The time varying signal is what we record.

# The First Microphone

- Graham Bell’s “liquid” microphone
  - Metal cup filled with water, mixed with Sulphuric acid
  - Needle suspended in water (stuck to a diaphragm)
  - Sound waves move the needle up and down
    - Changing the resistance of the cup
  - A fixed voltage induces a time-varying current



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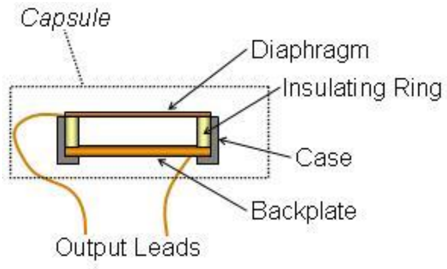
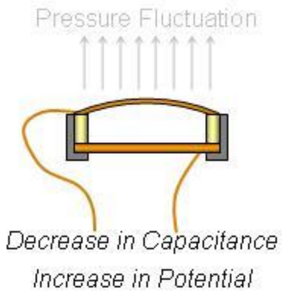
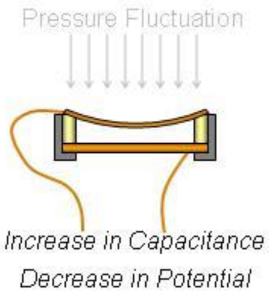
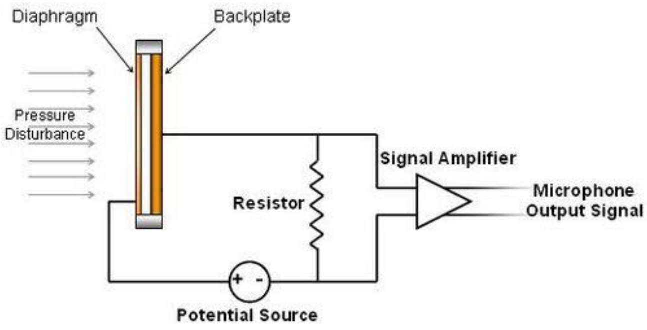
- The design was actually “copied” from Elisha Gray
  - Apparently from a drawing Bell saw in the patent office
  - In later incarnations he discarded the liquid microphone

# Condenser Mic: Capacitance based



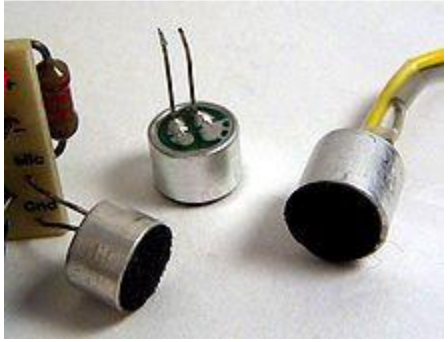


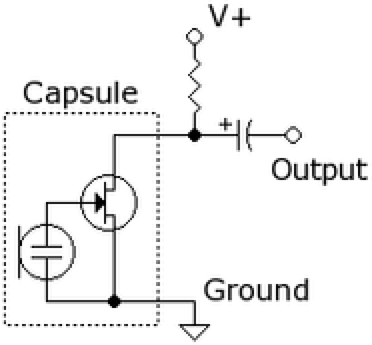




- Diaphragm and back plate maintained at different charge levels
- Pressure waves change the thickness of intervening dielectric (air)
  - Changing the capacitance of the capsule
  - Thereby changing the potential difference across the plates
    - Q = C x V
- The changing potential difference is measured across the large resistor

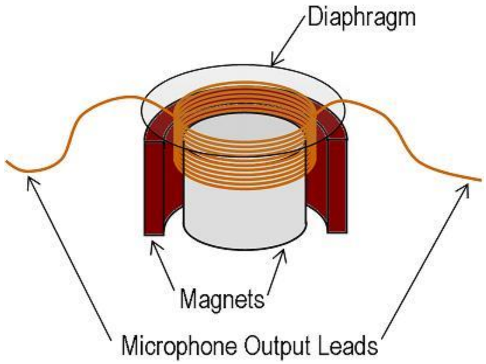
# Electret Microphones: Capacitive

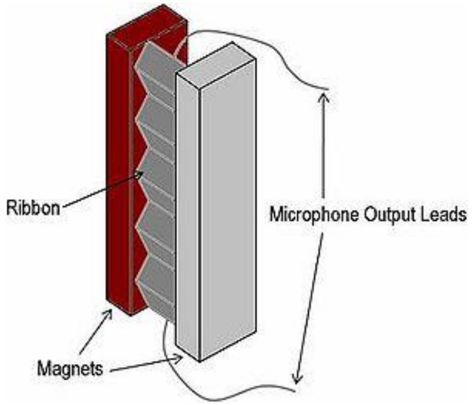




- Electret microphones are Condenser mics with fixed charge level
  - Either the diaphragm or the backplate carry a fixed charge
  - No external voltage source needed to maintain charge
    - But do require power for preamp
    - Usually from the DC
  - Used to be low quality, no longer
- Most computer microphones are electrets

# Dynamic Mics: Inductive

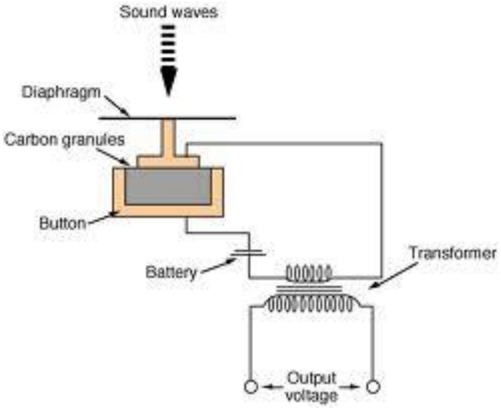




- Based on inductance, rather than capacitance
- Time-varying voltage levels created by moving an element in an electromagnetic field
  - Moving coil mic:  Diaphragm connected to a coil, suspended between magnets
    - Movement of diaphragms moves the coil and generates a voltage
  - Ribbon mic: A metal ribbon suspended between magnets

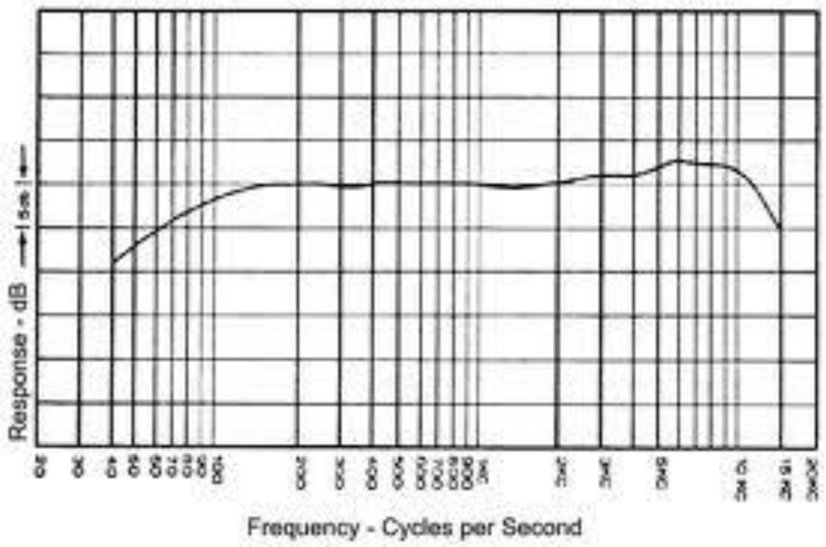
- Does not require phantom power
- Frequency response generally not flat
  - Also, small mics may be noise prone

# Carbon Button Microphones: Reisistive



- Based on *resistance*
- Carbon granules between a diaphragm and a backplate
- Motion of the diaphragm changes the resistance of the microphone
- This changes the current in the circuit
  - Typically transformed to a voltage as shown
- Typical in older telephone handsets
  - Cellphones and recent handsets use electrets

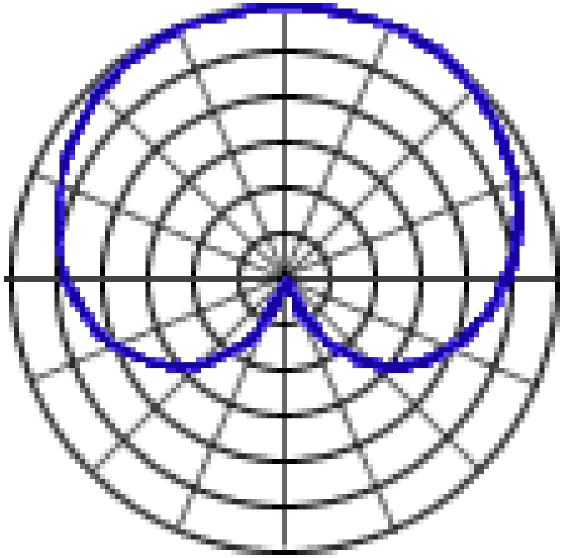
# Choice of Microphones



- Characteristics to look for:
  - Frequency response
    - Must be flat in the frequencies of interest
  - Directivity
    - More on this later
  - Noise level
    - Usually stated as dB(A) SPL (Indicates the noise floor of the mic)
    - Lower is better
      - Good microphones: 20dB SPL, Ultra very good mics: 0dB
- Frequency response: Condenser microphones are usually used for high-quality recordings
  - Even cheap electret microphones can be surprisingly good

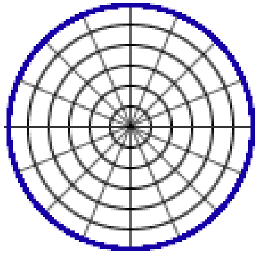
# The "Directionality" of a Microphone

- The response of a microphone depends on the direction of the sound source
- This depends on the construction of the microphone, as well as the shape of its chassis
- The response is typically represented as a polar plot
  - The distance of the curve from the "center" of the plot in any direction shows the "gain" of the mic to sounds from that direction

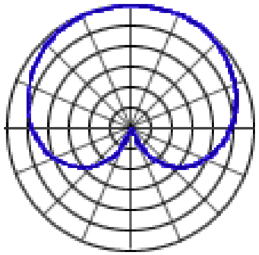


# Typical Directivity Patterns

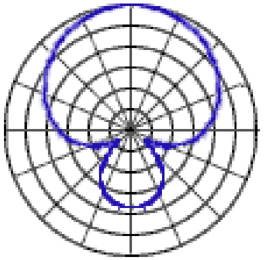
- Omnidirectional
  - Picks up sound uniformly from all directions



- Cardioid
  - Picks up sound from a relatively wide angle of directions from the front and some sound  from the side



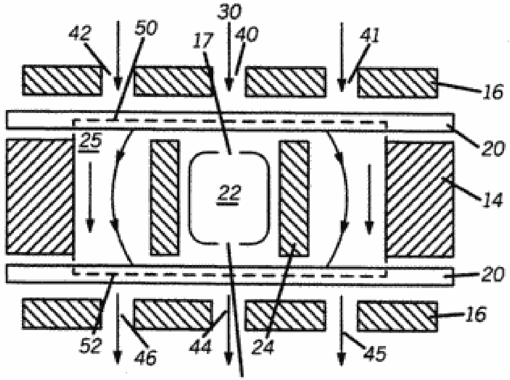
- Hyper cardioid
  - Picks up sound from a relatively narrow angle to the front, but also some noise from behind



# Directional Patterns

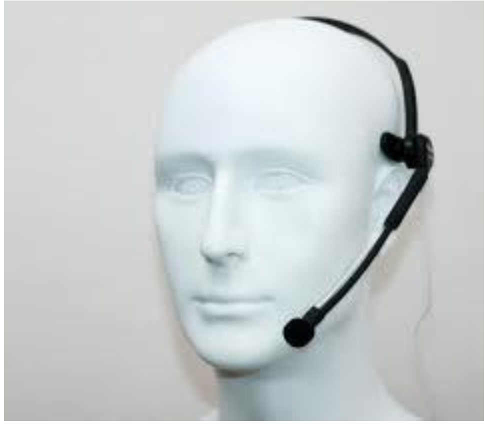
- Omnidirectional microphones are useful when we wish to pick up sounds from all directions
  - For speech applications, however, this will result in picking up a lot of noise from non-speech directions
- Where the speaker location can be somewhat localized, Cardioid microphones are more precise
  - Pick up less non-speech noise
  - Still susceptible to noise
- Hypercardioids are a better choice when the speaker location can be well localized
  - Risk of picking up noise from behind
- In general, utilizing microphone directionality is the best way of minimizing extraneous noise from recordings

# Noise Cancelling Mics



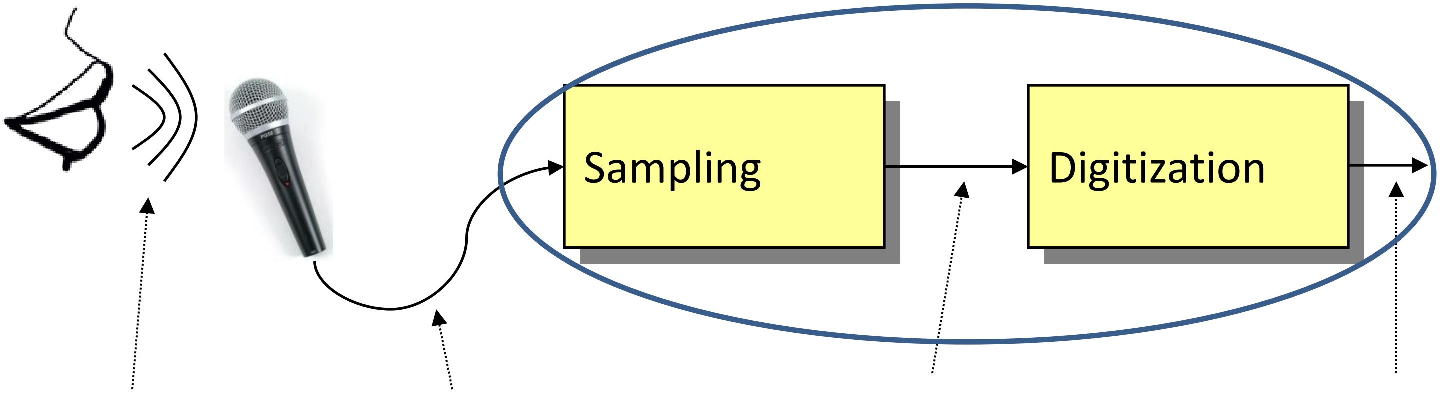
- Noise-cancelling microphones are designed to reduce ambient noise in close-talking scenarios
  - Headset mics for example
  - The noise field in the front and back are roughly the same
  - The speech is much more energetic in the front than the back
    - Because of $1/r^2$ fall off in energy
  - Subtracting the signal from the back to the signal from the front results in selective reduction of noise
    - Also results in a natural high-pass filtering effect that attenuates very-low frequencies
  - Similar results can be obtained by using two microphones

# Wearing/Positioning the Microphone



- Microphone must ideally be close enough for high speech level
  - Speech level significantly greater than background noise
- Positioning it too close can result in puffs of air from mouth saturating the mic
  - E.g. sounds such as “P”, “F”, etc.
- To avoid this, desktop mics frequently use a wind sock
  - A foam/sponge covering
- Head-mounted mics must be placed slightly to the side
  - Away from the direct path of puffs

# Sampling and Digitization



Sampling

Digitization

samples

Speech waves

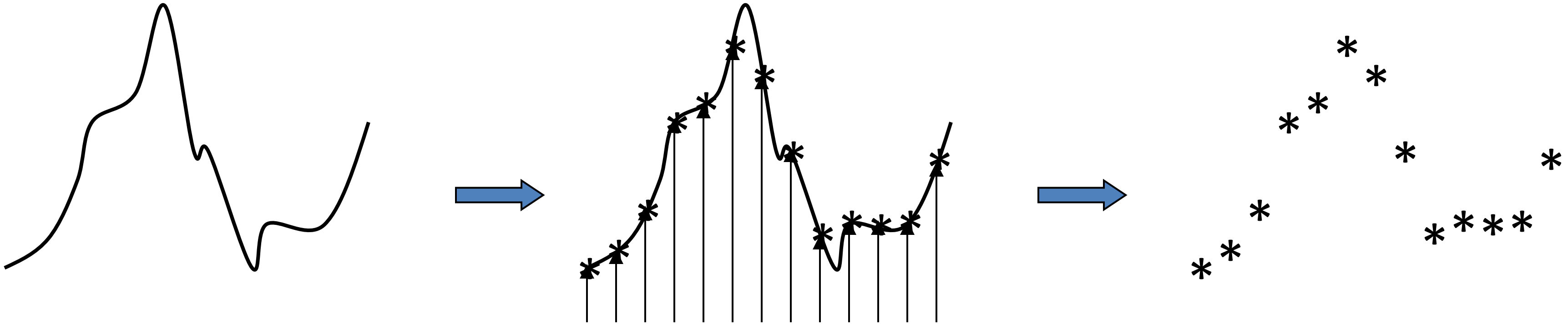
Analog speech signal

Discrete time signal

Digitized discrete time signal

- The signal from the microphone goes through a pre-amp
  - The gain of which can be adjusted
  - The output of the pre-amp is a continuous-time electrical signal
    - Usually a voltage signal
- The signal is digitized by an analog to digital converter
  - The signal value is sensed at regular, periodic intervals of time
    - Sampling
  - The value at each instant is quantized to one of a number of fixed values
    - Quantization

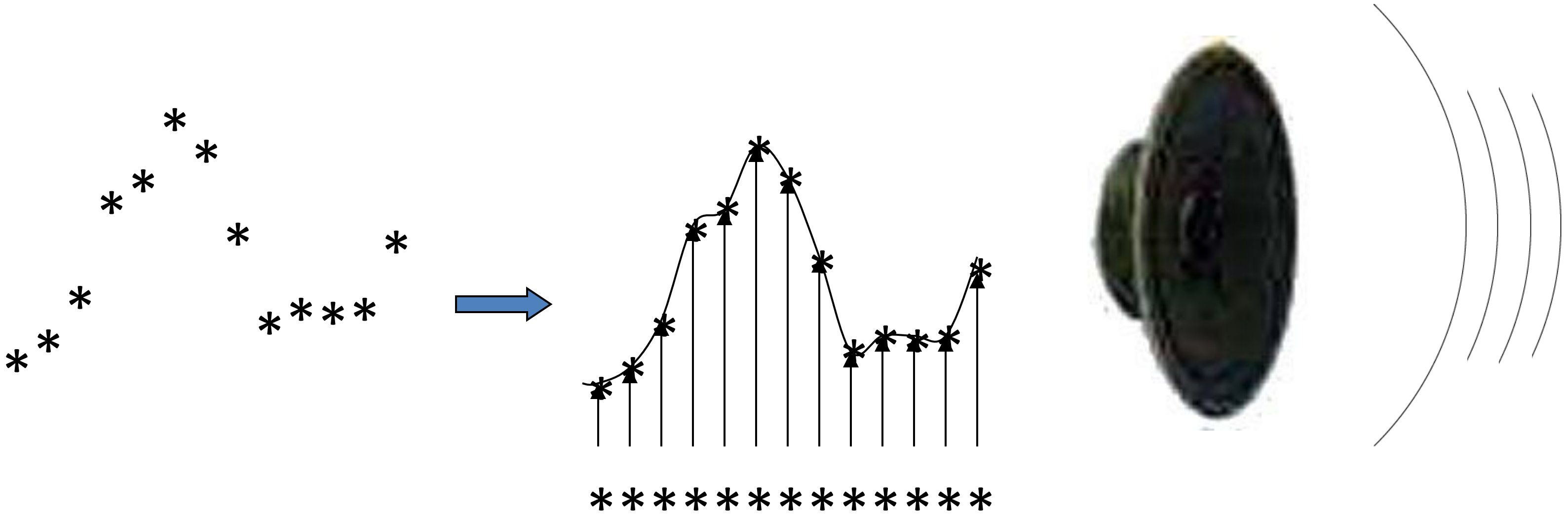
# Sampling



- Sampling is the process of capturing snapshots of the signal at discrete instants of time
  - For speech recognition, these samples will be uniformly spaced in time
- Requirement: The sequence of numbers must be sufficient to reconstruct the original signal perfectly
  - To retain all the information in the signal

# Sufficiency of Samples

- How to determine if the captured samples are enough?
  - Recreate the sense of sound
- The numbers are used to control the levels of an electrical signal
- The electrical signal moves a diaphragm back and forth to produce a pressure wave
  - That we sense as sound
- The recreated sound must be identical to the original

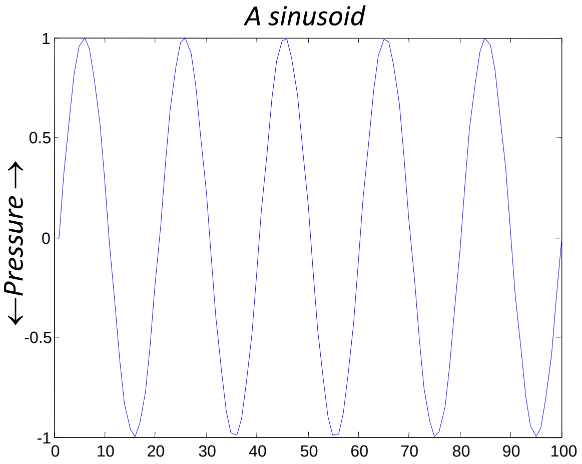


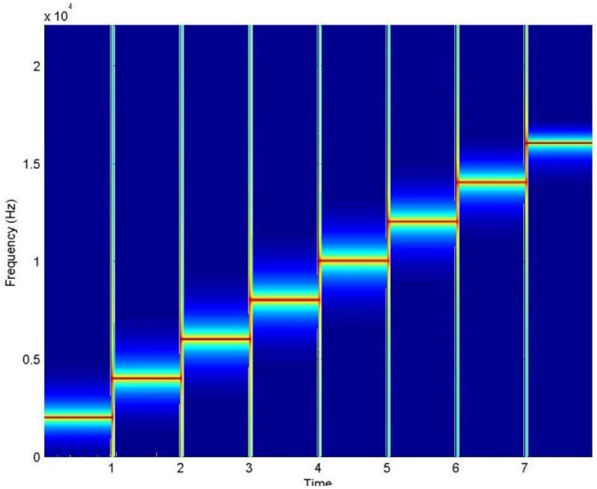
# How many samples a second

- Convenient to think of sound in terms of sinusoids with frequency

- Sounds may be modeled as the sum of many sinusoids of different frequencies
  - Frequency is a physically motivated unit
  - Each hair cell in our inner ear is tuned to specific frequency

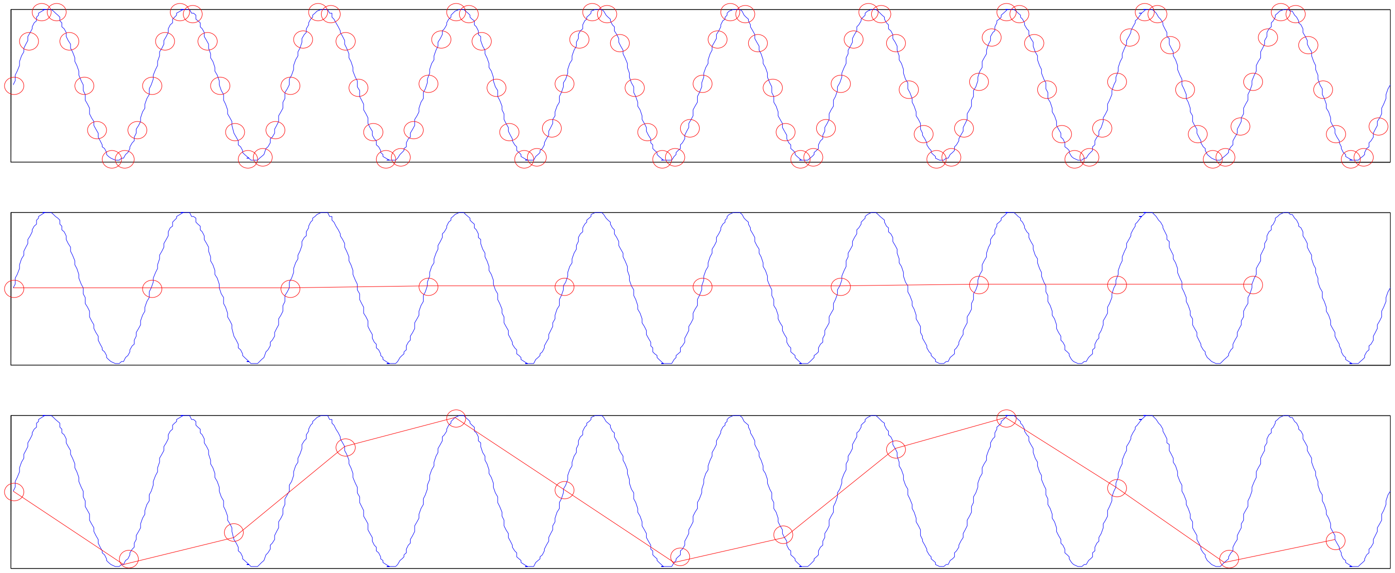
- Any sound has many frequency components
  - We can hear frequencies up to 16000Hz
    - Frequency components above 16000Hz can be heard by children and some young adults
    - Nearly nobody can hear over 20000Hz.





# Aliasing

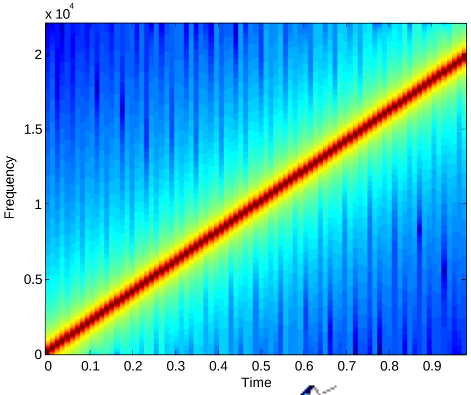
- *Sampling frequency* is the number of samples taken per second
  - Usually measured in hertz (Hz) : 1 Hz = 1 sample per second
- Low sample rates result in *aliasing*
  - High frequencies are misrepresented
  - Frequency $f_1$ will become (sample rate – $f_1$ )
- **NYQUIST THEOREM: To represent any frequency *f* accurately, you need at least 2*f* samples per second**



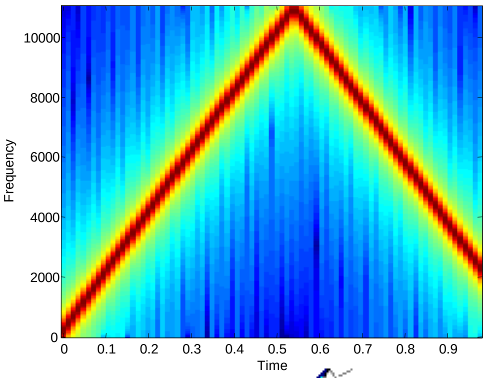
# Aliasing examples

## Sinusoid sweeping from 0Hz to 20kHz

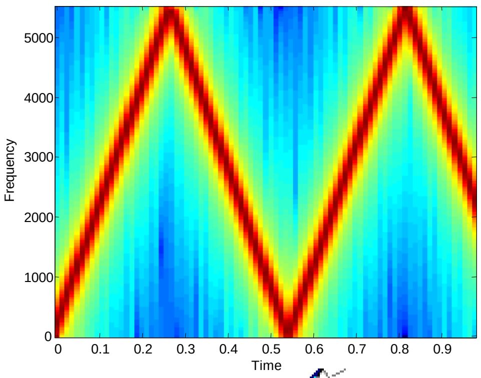
44kHz SR, is ok



22kHz SR, aliasing!



11kHz SR, double aliasing!

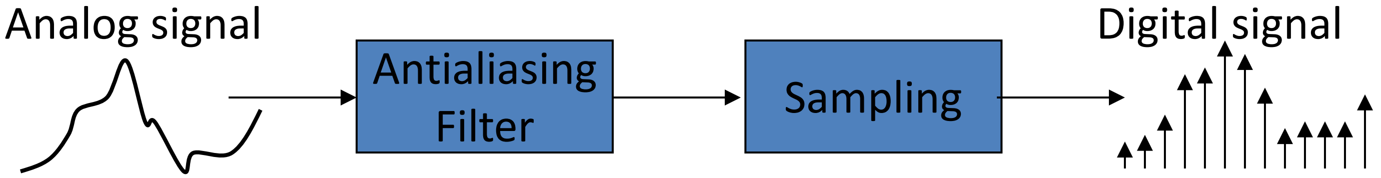


## On real sounds

*at 44kHz* *at 11kHz* *at 4kHz*

*at 22kHz* *at 5kHz* *at 3kHz*

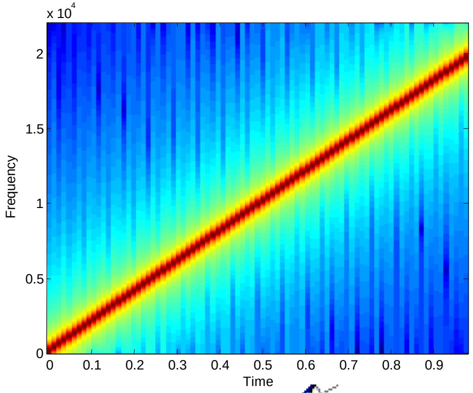
# Avoiding Aliasing



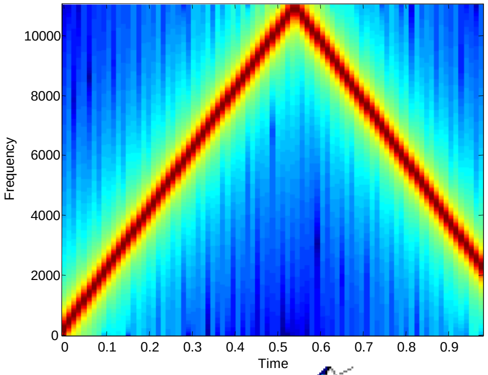
- Sound naturally has all frequencies
  - And then some
  - Cannot control the rate of variation of pressure waves in nature
- Sampling at *any* rate *will* result in aliasing
- Solution: *Filter the electrical signal* before sampling it
  - Cut off all frequencies above samplingfrequency/2
  - E.g., to sample at 44.1Khz, filter the signal to eliminate all frequencies above 22050 Hz

# Aliasing examples: Sweep 0-20khz

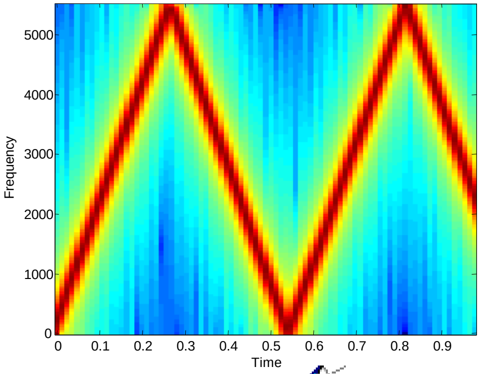
44kHz SR, is ok



22kHz SR, aliasing!

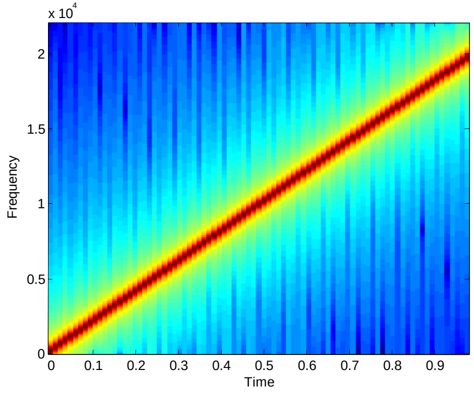


11kHz SR, double aliasing!

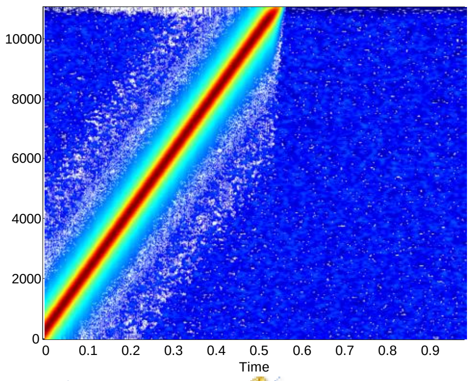


# Anti Aliasing examples: Sweep 0-20khz

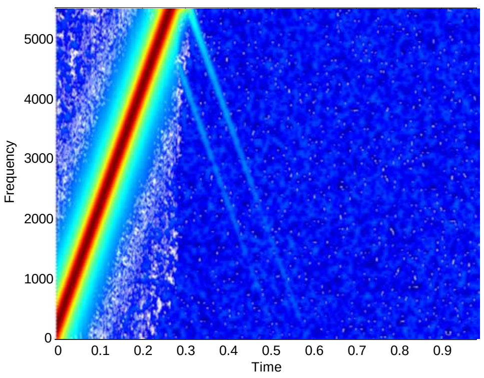
44kHz SR, is ok



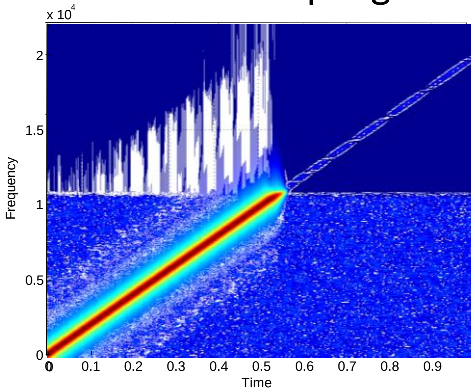
22kHz SR, antialiased!



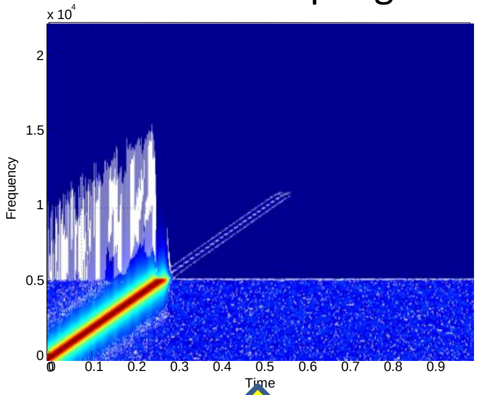
11kHz SR, antialiased!



Prior to 22kHz downsampling



Prior to 11kHz downsampling



# Anti Aliasing Examples: Real sounds

## Aliased

🔊 *at 44kHz* 🔊 *at 11kHz* 🔊 *at 4kHz*

🔊 *at 22kHz* 🔊 *at 5kHz* 🔊 *at 3kHz*

## Anti aliased

🔊 *at 44kHz* 🔊 *at 11kHz* 🔊 *at 4kHz*

🔊 *at 22kHz* 🔊 *at 5kHz* 🔊 *at 3kHz*

# Anti Aliasing

- Speech signal frequencies are audible up to 20 kHz
  - Which means a sampling frequency of 40 kHz should capture all relevant information
- However, the signal has frequencies going up to over 100kHz
  - Sampling at 40 kHz causes frequencies > 20kHz to be aliased
  - These frequency components will “fold over” and distort lower frequencies
- The analog signal must first be filtered to eliminate frequencies above 20 kHz to sample at 40 kHz

# Typical Sampling Frequencies

- Speech remains intelligible if we filter out high frequencies
  - Telephone speech only has 300Hz – 3.3kHz

- Practical systems use a variety of sampling rates
  - For speech 8kHz, 11.5kHz and 16kHz and 44.1 kHz are all used
  - Sampling hardware will usually employ appropriate anti-aliasing
    - Not always – be careful

- The best speech recognition performance is obtained when the antialiasing filter retains frequencies till at least 8kHz
  - With sampling rates 16kHz or higher

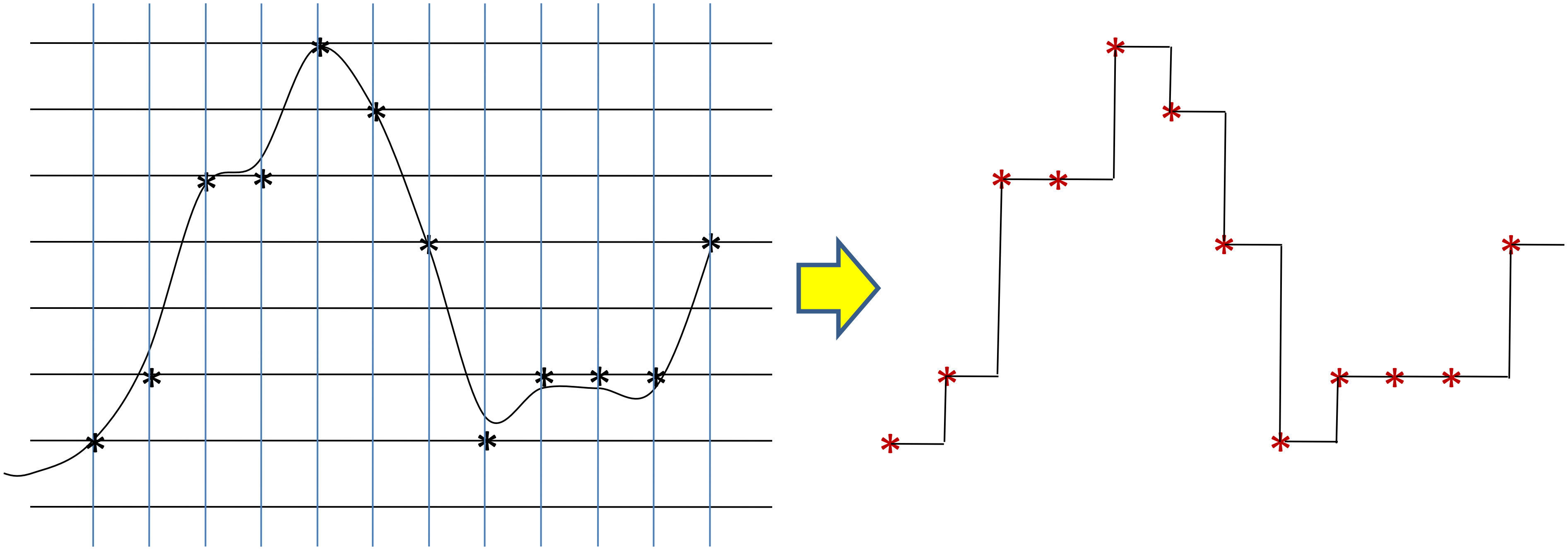
- Lower frequency cutoffs result in poorer performance
  - E.g. recognition with telephone speech is worse than that with wideband speech (going up to 8kHz)
  - *Oversampling* does not increase frequency content
    - *I.e.* there is no benefit from sampling telephone speech at 16kHz
    - We normally sample telephone speech at 8kHz

# The Speech Signal: Sampling

- Audio hardware typically supports several standard rates
  - *E.g.*: 8, 16, 11.025, or 44.1 KHz (*n* Hz = *n* samples/sec)
  - CD recording employs 44.1 KHz per channel – high enough to represent most signals most faithfully

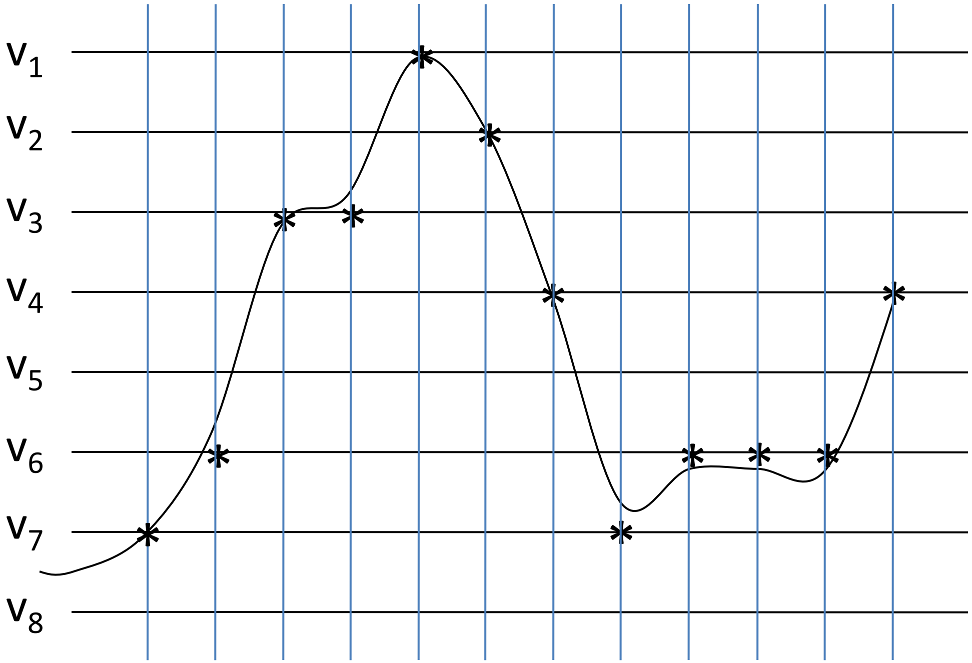
- Speech recognition typically uses 8KHz sampling rate for telephone speech and 16KHz for wideband speech
  - Telephone data is *narrowband* and has frequencies only up to 4 KHz
  - Good microphones provide a *wideband* speech signal
    - 16KHz sampling can represent audio frequencies up to 8 KHz
    - This is considered sufficient for speech recognition

# Sample Resolution



- Samples cannot take any value
- They are restricted to a finite set of values
  - – Typically one of $2^N$-1, for N=8,16,32
  - – Can be represented using integers

# Process of Quantization



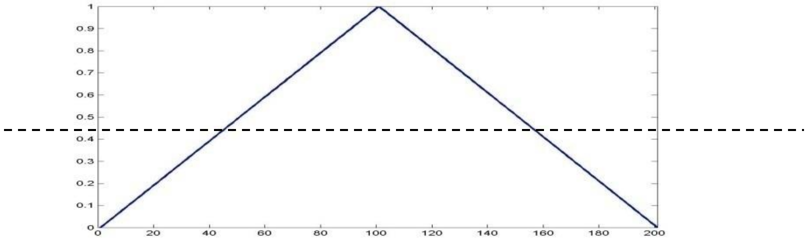
| index | Value |
|---|---|
| 0 | $v_1$ |
| 1 | $v_2$ |
| 2 | $v_3$ |
| 3 | $v_4$ |
| 4 | $v_5$ |
| 5 | $v_6$ |
| 6 | $v_7$ |
| 7 | $v_8$ |

- Determine the values the signal will be rounded to
- Maintain a table of values
- Store only the indices into the table
  - The actual values are read from the table when recomposing speech
    - The table may be implicit

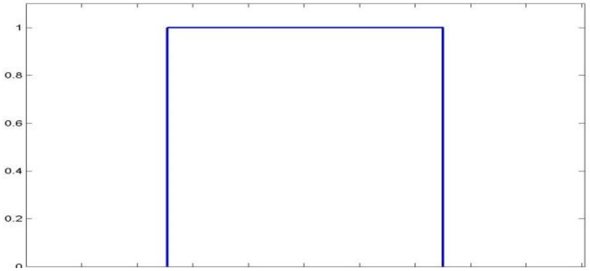
# Mapping signals into bits

- Example of 1-bit *Uniform* sampling table: Only 2 levels

| Signal Value | Bit sequence | Mapped to |
|---|---|---|
| S > 2.5v | 1 | 1 * const |
| S <=2.5v | 0 | 0 |



Original Signal

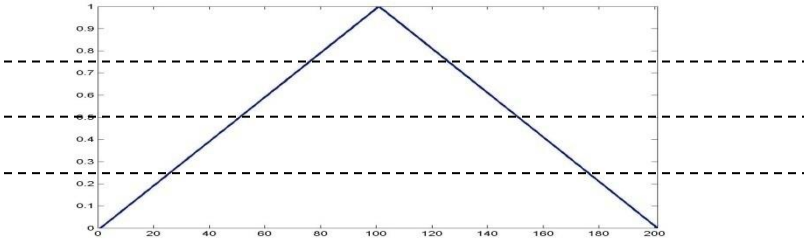


Quantized approximation

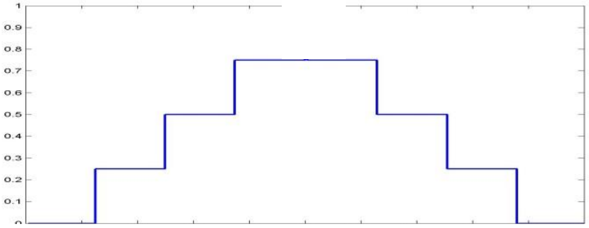
# Mapping signals into bits

- Example of 2-bit *Uniform* sampling table: 4 equally-spaced levels

| Signal Value | Bit sequence | Mapped to |
|---|---|---|
| S >= 3.75v | 11 | 3 * const |
| 3.75v > S >= 2.5v | 10 | 2 * const |
| 2.5v > S >= 1.25v | 01 | 1 * const |
| 1.25v > S >= 0v | 0 | 0 |



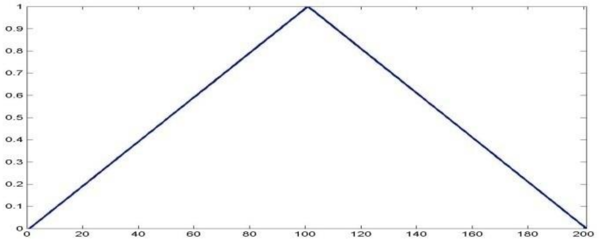
Original Signal



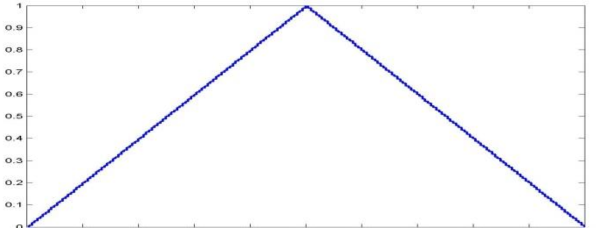
Quantized approximation

# *Uniform* Sampling, different levels

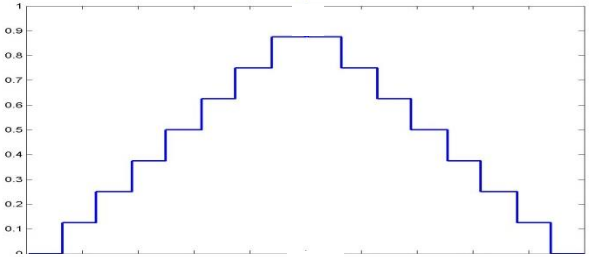
- The original signal



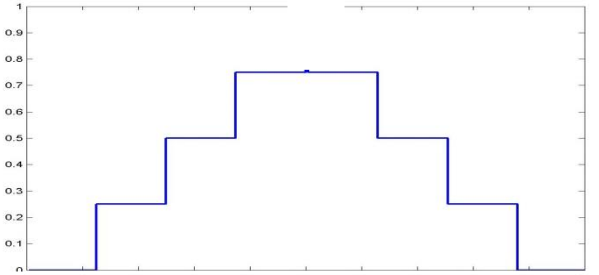
- 8 bit quantization



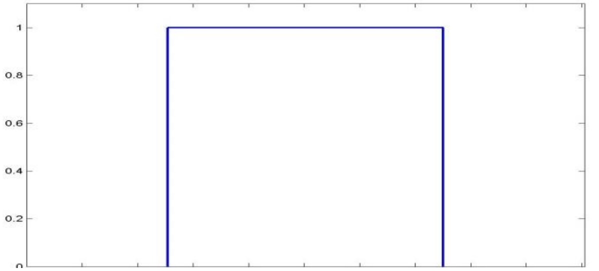
- 3 bit quantization



- 2 bit quantization

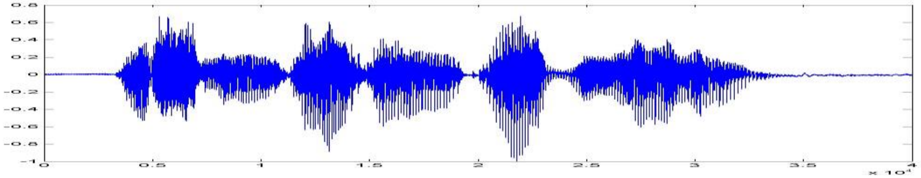


- 1 bit quantization

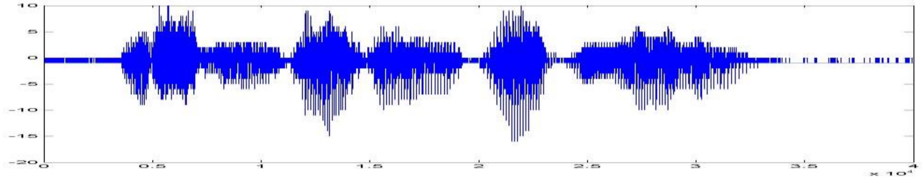


# Tom Sullivan Says his Name

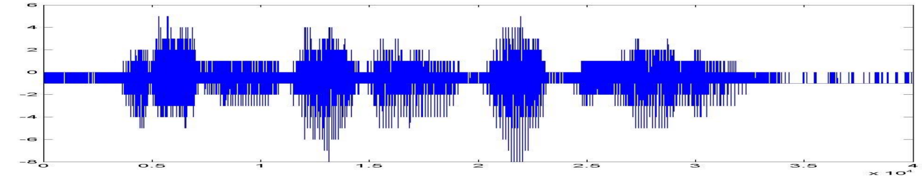
- 16 bit sampling



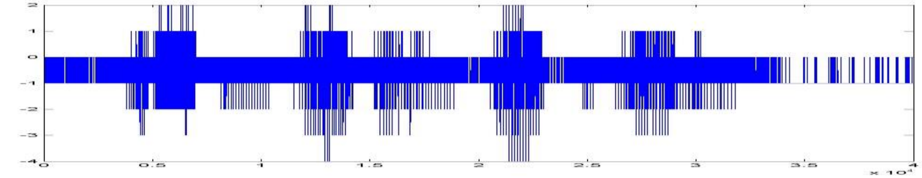
- 5 bit sampling



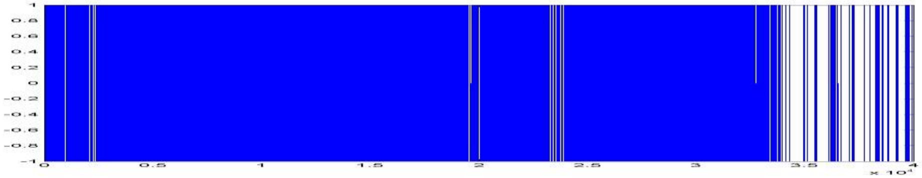
- 4 bit sampling



- 3 bit sampling

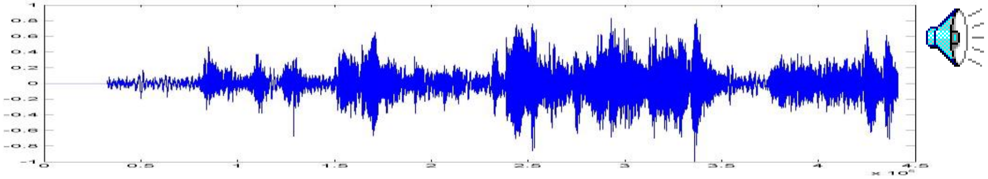


- 1 bit sampling

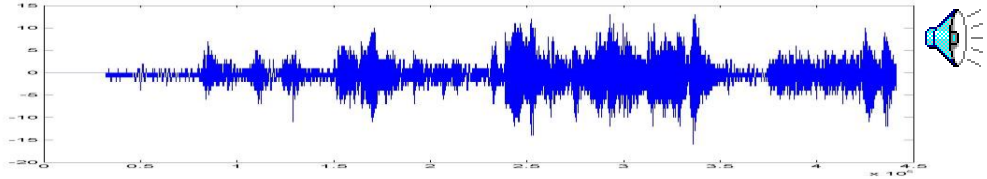


# A Schubert Piece

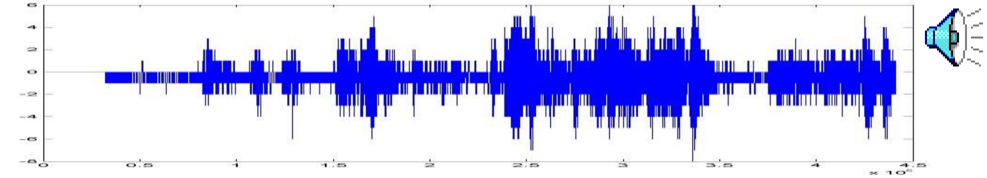
- 16 bit sampling



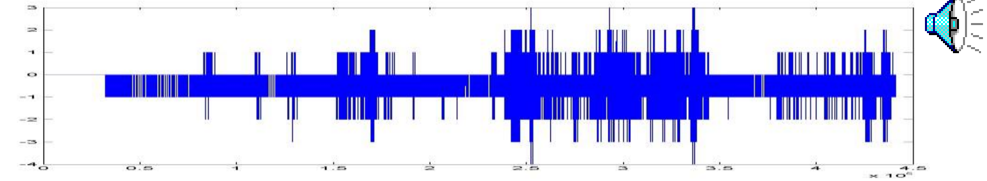
- 5 bit sampling



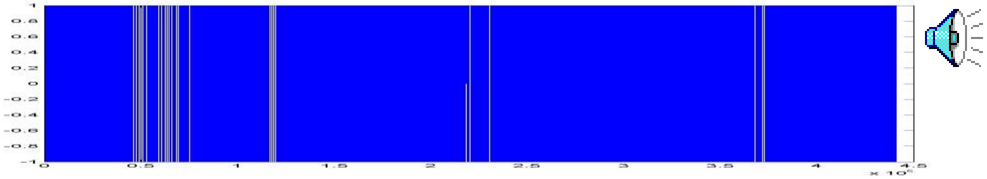
- 4 bit sampling



- 3 bit sampling



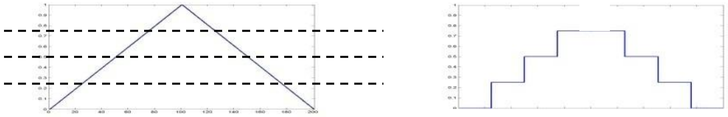
- 1 bit sampling



# Sampling Formats

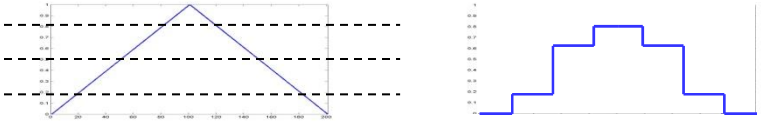
- *Uniform* sampling: Sample values equally spaced out

| Signal Value | Bits | Mapped to |
|---|---|---|
| S >= 3.75v | 11 | 3 * const |
| 3.75v > S >= 2.5v | 10 | 2 * const |
| 2.5v > S >= 1.25v | 01 | 1 * const |
| 1.25v > S >= 0v | 0 | 0 |

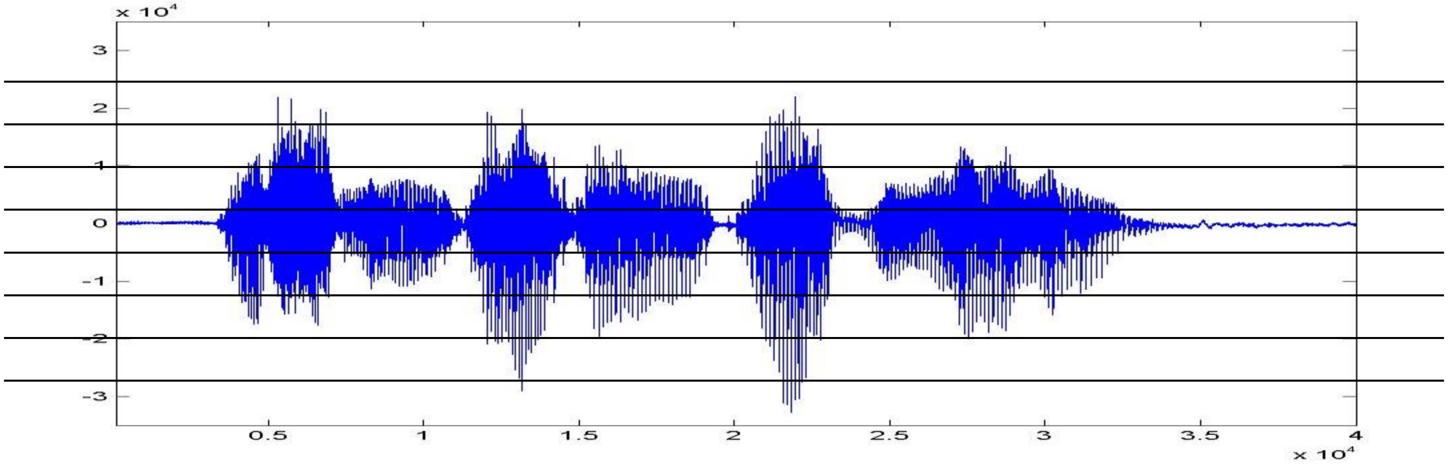


- Sampling need not be uniform: nonuniform sampling

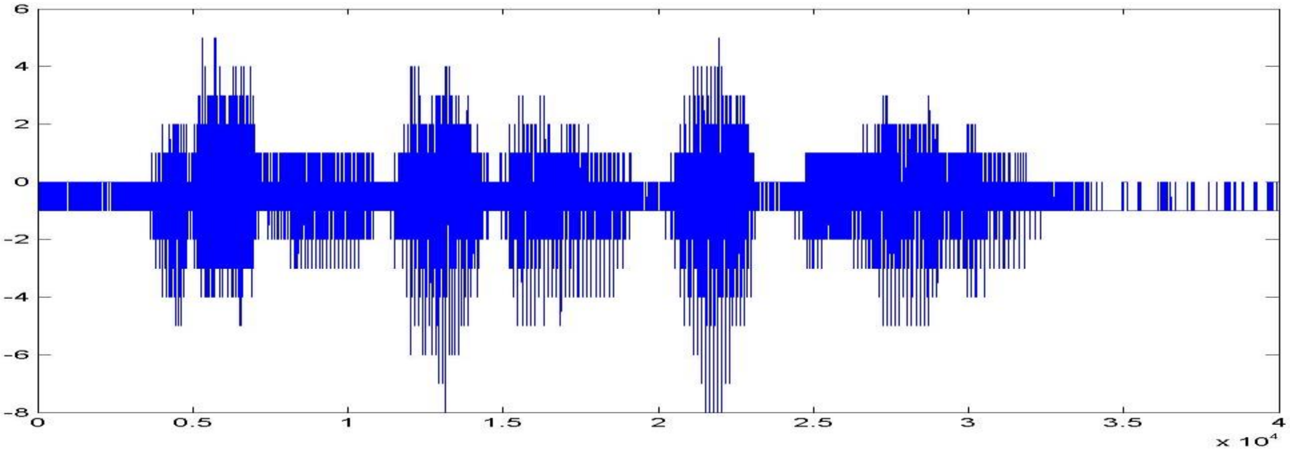
| Signal Value | Bits | Mapped to |
|---|---|---|
| S >= 4v | 11 | 4.5 * const |
| 4v > S >= 2.5v | 10 | 3.25 * const |
| 2.5v > S >= 1v | 01 | 1.25 * const |
| 1.0v > S >= 0v | 0 | 0.5 * const |



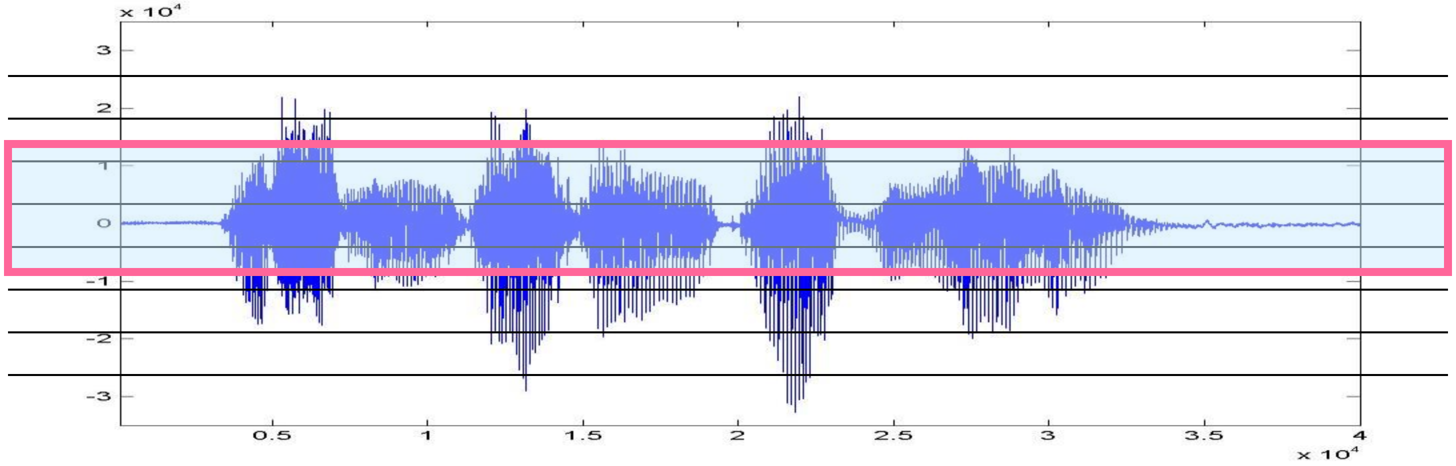
# Uniform Sampling



UPON BEING SAMPLED AT ONLY 3 BITS (8 LEVELS)

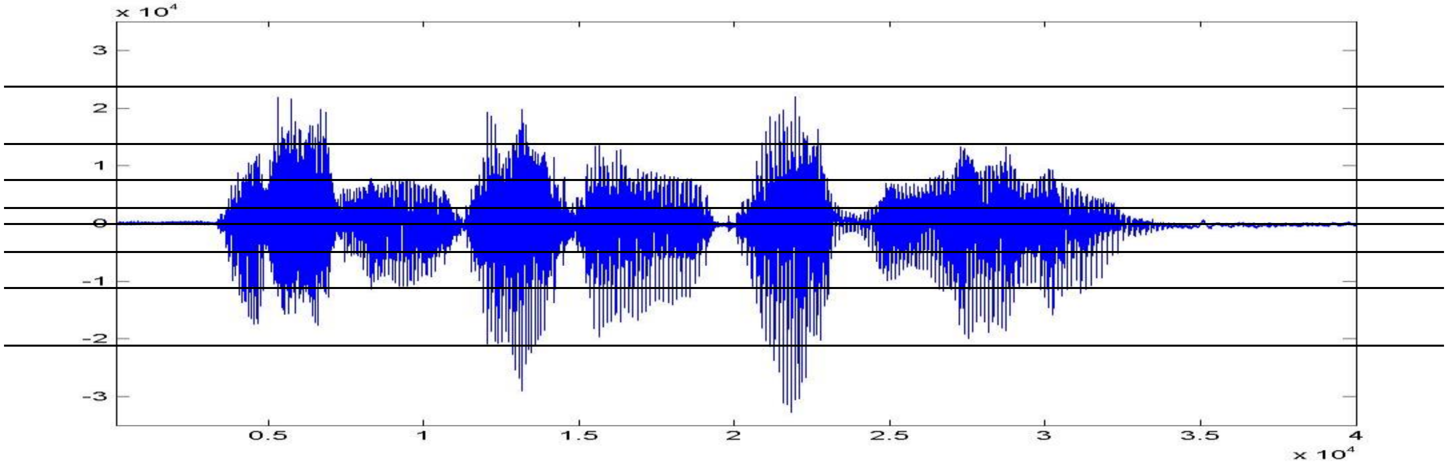


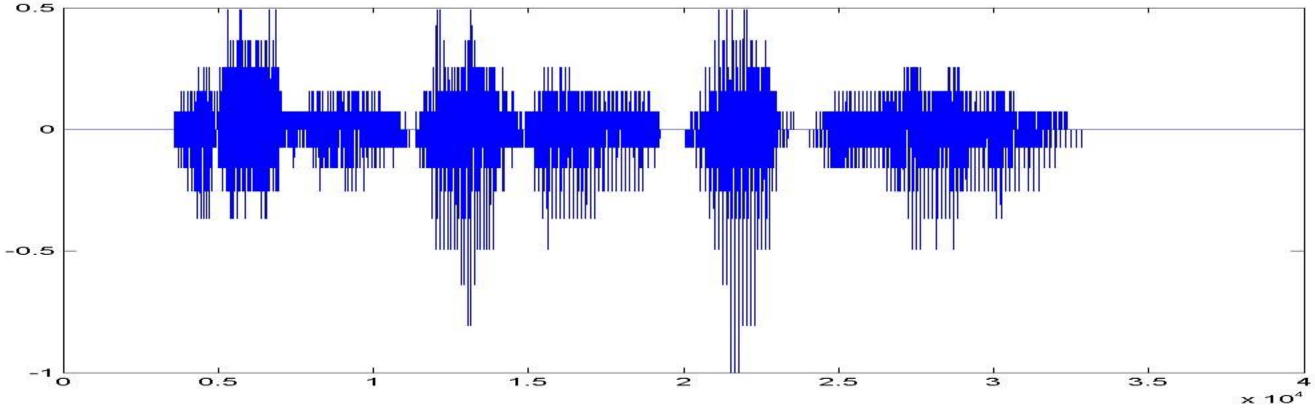
# Uniform Sampling



- There is a lot more action in the central region than outside.
- Assigning only four levels to the busy central region and four entire levels to the sparse outer region is inefficient
- Assigning more levels to the central region and less to the outer region can give better fidelity
  - for the same overall number of levels

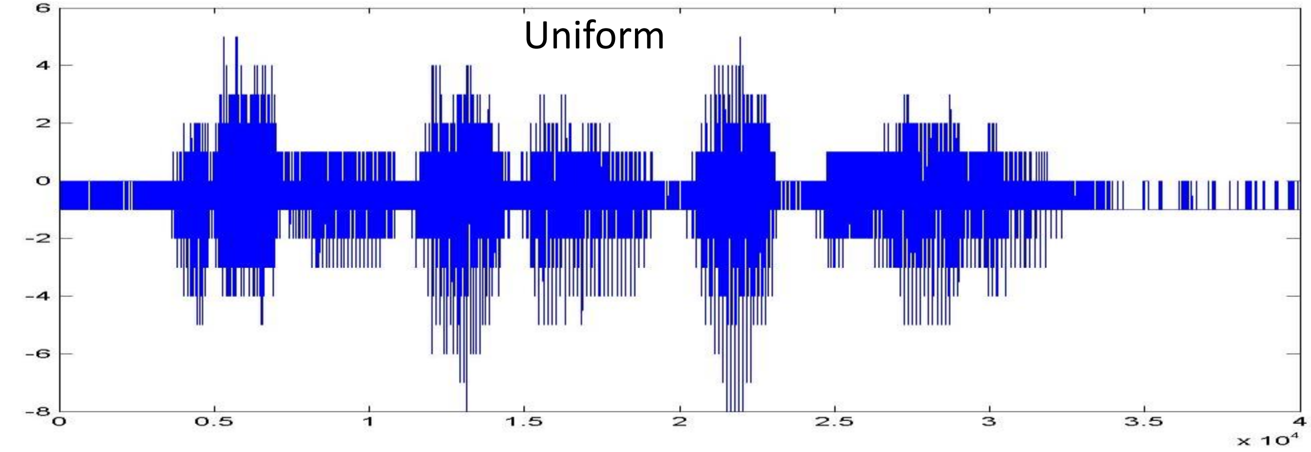
# Non-uniform Sampling

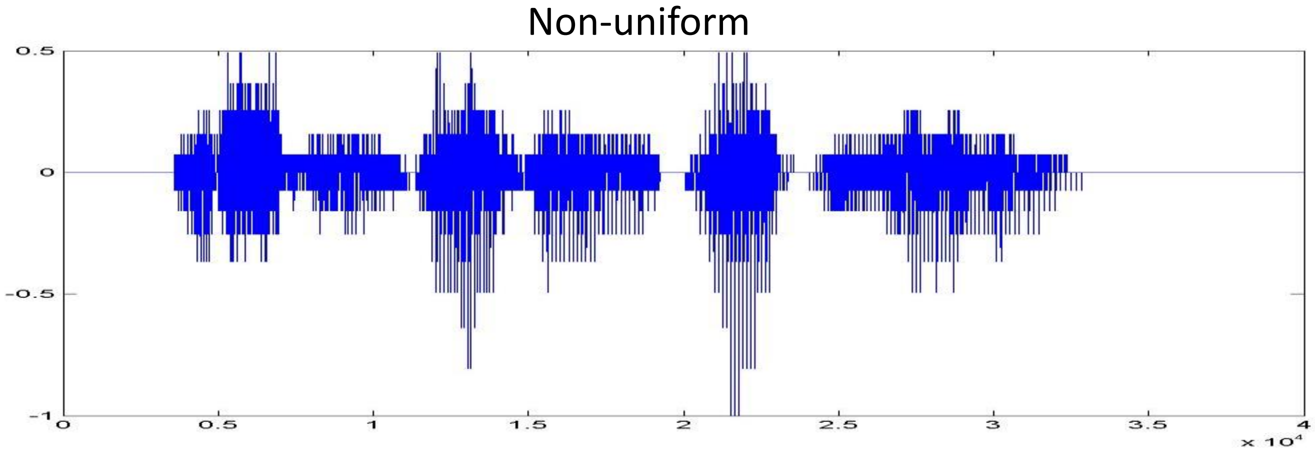




- Assigning more levels to the central region and less to the outer region can give better fidelity for the same storage

# Non-uniform Sampling





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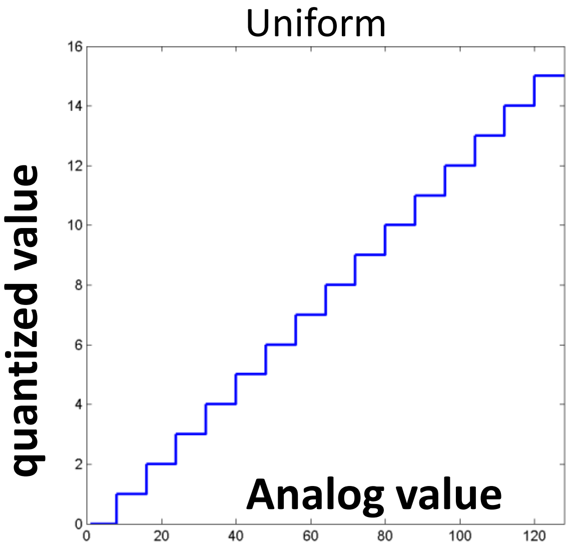
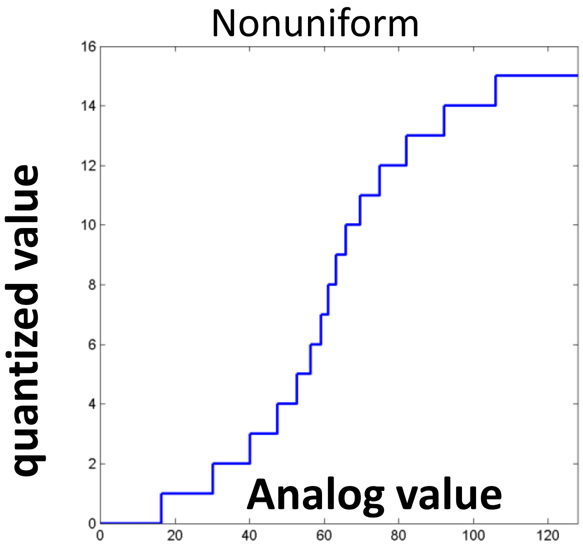
# Non-uniform Sampling





- Uniform sampling maps uniform widths of the analog signal to units steps of the quantized signal
- In non-uniform sampling the step sizes are smaller near 0 and wider farther away
  - The curve that the steps are drawn on follow a logarithmic law:
    - Mu Law: Y = C. log(1 + μX/C)/(1+mu)
    - A Law: Y = C. (1 + log(a.X)/C)/(1+a)
- One can get the same perceptual effect with 8 bits of non-linear sampling as 12 bits of linear sampling

# Capturing / Reading Audio

- The numbers returned by an ADC or read from a file are *indices* into the table
- The indices must be converted back to actual values
- For uniformly sampled data the indices are proportional to the value and can be used directly
  - Called "Linear PCM" or "PCM" encoding
- For non-uniform sampling, table lookup (or function inversion) is required
  - Often performed automatically by the audio data read / capture functions in most audio libraries
  - If we write the read / capture functions ourselves, we must perform appropriate table lookup

# Audio capture: Conversion summary

| Signal Value | Bits | Mapped to |
|---|---|---|
| S >= 3.75v | 11 | 3 |
| 3.75v > S >= 2.5v | 10 | 2 |
| 2.5v > S >= 1.25v | 01 | 1 |
| 1.25v > S >= 0v | 0 | 0 |

| Signal Value | Bits | Mapped to |
|---|---|---|
| S >= 4v | 11 | 4.5 |
| 4v > S >= 2.5v | 10 | 3.25 |
| 2.5v > S >= 1v | 01 | 1.25 |
| 1.0v > S >= 0v | 0 | 0.5 |

- Capture / read audio in the format provided by the file or hardware
  - Linear PCM, Mu-law, A-law, Coded
- Convert to 16-bit PCM value
  - I.e. map the bits onto the number on the right column
  - This mapping is typically provided by a table computed from the sample compression function
  - No lookup for data stored in PCM
- Conversion from Mu law:
  - http://www.speech.cs.cmu.edu/comp.speech/Section2/Q2.7.html

# The Effect of Signal Quality

- The quality of the digitized signal depends critically on many factors:
  - The electronics performing sampling and digitization
    - Poor quality electronics can severely degrade signal quality
      - *E.g.* Disk or memory bus activity can inject noise into the analog circuitry
  - Improper Antialiasing
    - Not using an antialiasing filter is a cause for many problems
  - Insufficient quantization levels
    - Minimally 16 bit PCM or 8 bit Mu / A law is needed
  - Proper setting of the recording level
    - Too low a level underutilizes available signal range, increasing susceptibility to noise
    - Too high a level can cause *clipping*
  - The microphone quality
  - Ambient noise in recording environment
- Suboptimal signal quality can affect recognition accuracy to the point of being completely useless

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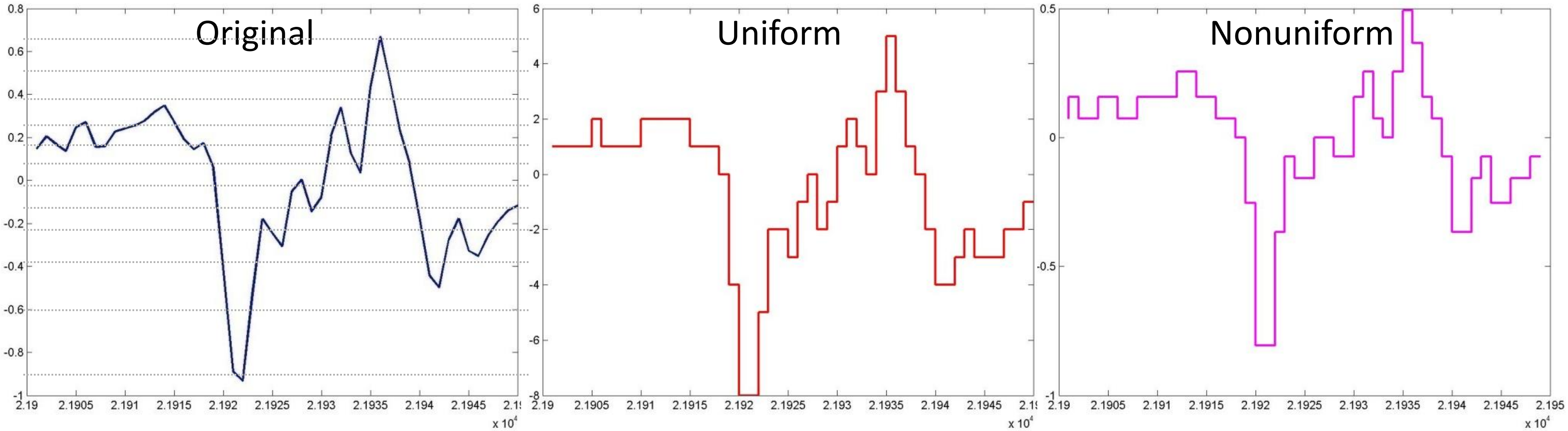
  - Ambient noise in recording environment
- Suboptimal signal quality can affect recognition accuracy to the point of being completely useless

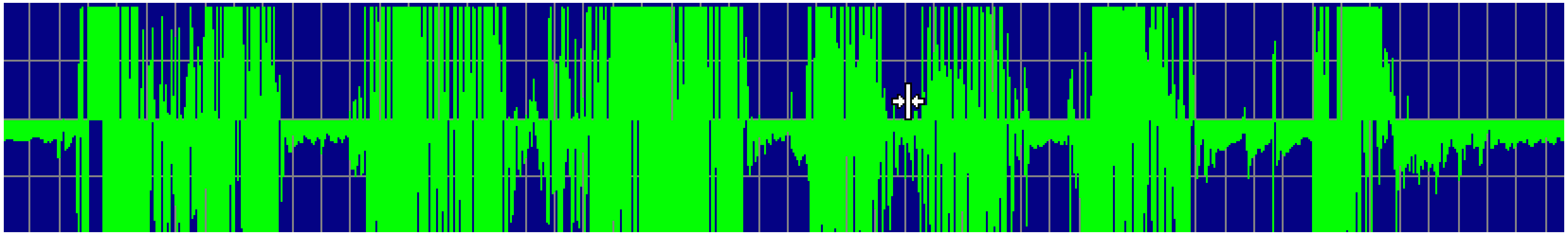
# Uniform and Non-uniform Sampling

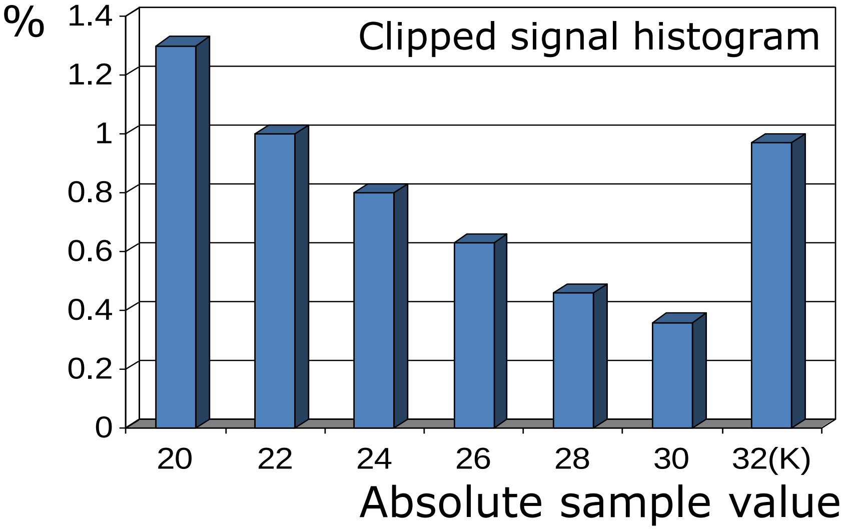


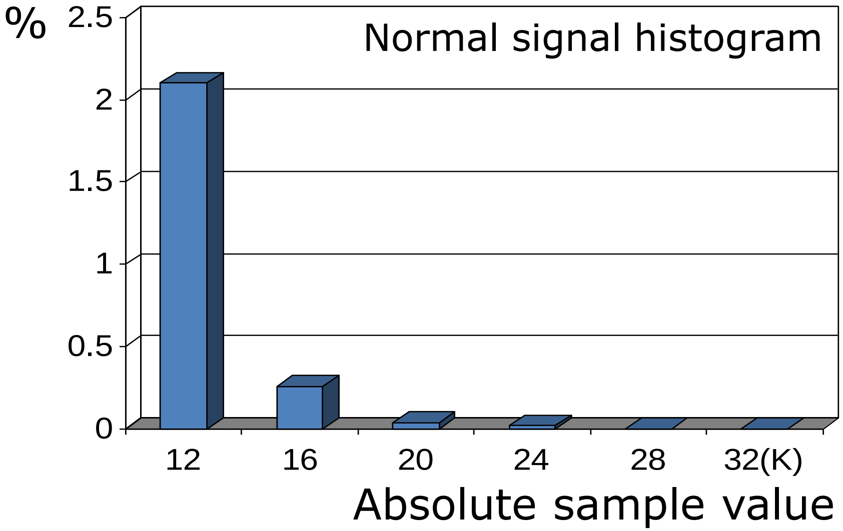
- At the sampling instant, the actual value of the waveform is rounded off to the nearest level permitted by the quantization
- Values entirely outside the range are quantized to either the highest or lowest values

# Clipping in Speech Signals

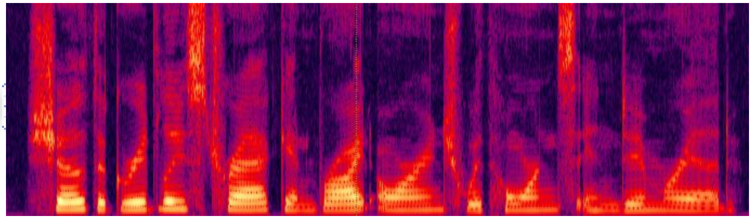
- Clipping and non-linear distortion are the most common and most easily fixed problems in audio recording
  - Simply reduce the signal gain (but AGC is not good)

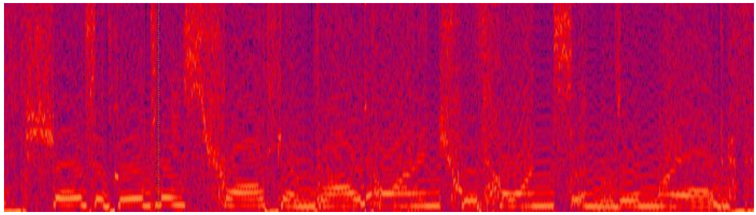






# The Effect of Noise



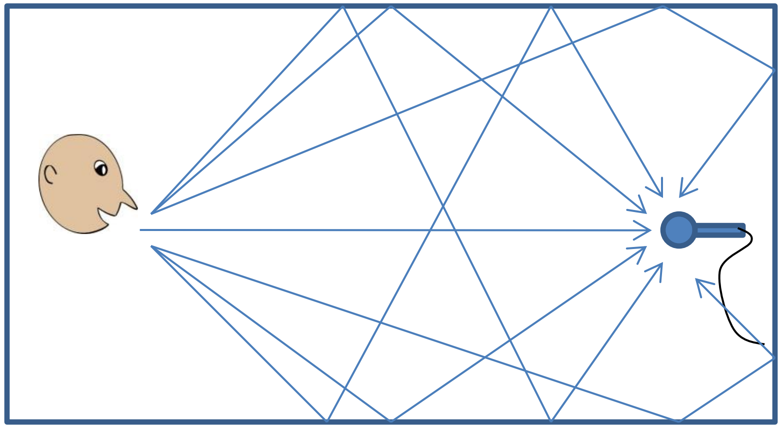


- Non-speech signals in the background can corrupt the recording
  - Diffuse background noise, e.g. in an automobile
  - Localized non-speech sounds, e.g. air conditioner
  - Background talkers, music..
- These result in degraded recognition accuracy
- Solution
  - Denoising algorithms for cancelling noise
  - Or, much (much) better – use directional microphones that will not pick up the noise

# The Effect of Recording Channel

- Recording channels/media can result in signal distortions
  - Telephone channels reduce the bandwidth of the signal
    - 300-3300Hz
    - Loss of information results in poorer recognition accuracy
  - Cellphone / VOIP channels introduce *coding* distortion
    - Coding and decoding speech introduces distortions
  - They can also result in *discontinuities* from dropped packets
  - Poor microphones can result in spectral distortions
  - Computer recordings may be affected by memory and disk activity
- Distortions cannot usually be recovered from
  - Recognizer must learn to perform well even on distorted data

# Reverberation



- Reflective walls etc. can introduce *reverberation* into a recording
  - Due to repeated addition of a signal with delayed reflections of itself
- Reverberation can be viewed as the combination of a linear-channel distortion and additive noise
  - Linear channel – represents effect of immediate past on current signal
  - Additive noise – represents effect of extended past on current signal
- Small levels of reverberation is usually beneficial to human perception, but even small levels of reverberation degrade automatic recognition
- Best option: Select recording environments with minimal reverberation

# Reading/Capturing the signal

- Audio can be obtained by:
  - Read from files
    - for offline/batch processing
  - Captured directly from hardware
    - for online processing

- **Reading from a prerecorded file**: Various library routines/calls
  - Matlab: [s,fs,c,…] = wavread('filename', args)
  - Libraries in C, Java, python, fortran..
- **Offline capture into a file**: You can use tools available for your favorite OS
  - Windows provides a "Windows recorder"
  - Several audio capture tools are also available for windows
  - Linux and most Unix machines provide "arecord" and "aplay"
    - If these are not already on your machine, you can download them from the web
  - Other tools are also available for linux

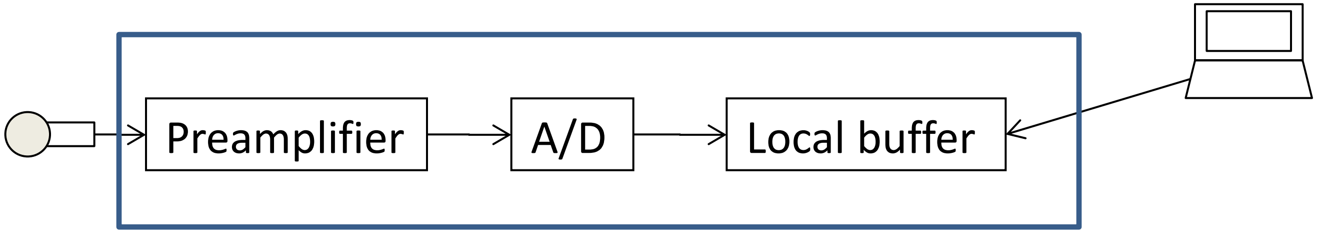
# Storing Audio/Speech Files

- There are many storage formats in use. Important ones:
  - PCM raw data (*.raw)
  - NIST (*.sph)
  - Microsoft PCM (*.wav)
  - Microsoft ADPCM (*.wav)
  - SUN (*.au, *.snd) etc.

- The data are typically written in binary, but many of these formats have headers that can be read as ascii text.
  - Headers store critical information such as byte order, no. of samples, coding type, bits per sample, sampling rate etc.

- Speech files must be converted from stored format to linear PCM format for further processing
  - Audio I/O library routines will usually process the headers to obtain the necessary information and perform the appropriate conversion
  - If we write the I/O, we must do this ourselves

# ***Capturing*** speech signals



- Your computer must have a sound card or an external A/D converter, and audio input devices such as a microphone, line input etc.

- Capture
  - Signal is captured by a microphone
  - Preamplified
  - Digitized
  - Stored in a buffer on the sound card

- Processor
  - Reads from buffer
  - At some prespecified frequency
    - Too frequent: can use up all available CPU cycles
    - Too infrequent: High latency

# Capturing Audio

- Capturing audio from your audio device
  - Open the audio device
    - Syntax is OS dependent
  - Set audio device parameters
    - Sampling rate,  quantization type/level, no. of channels.
  - Record blocks of audio
  - Close audio device

- Recorded audio can be stored in a file or used for live decoding
- Two modes of audio capture for live-mode decoding
  - **Blocking**: Application/decoder requests audio from audio device when required
    - The program waits for the capture to be complete, after a request

  - **Callback**:  An audio program monitors the audio device and captures data. When it has sufficient data it calls the application or decoder

# Capturing speech signals

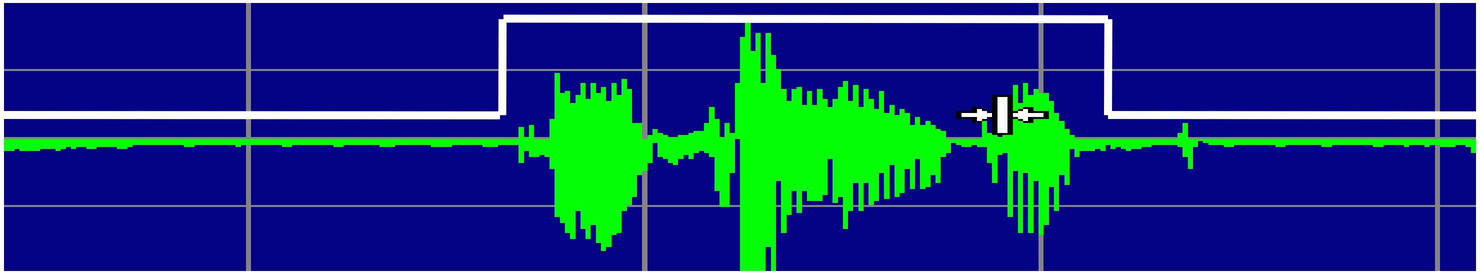
- Example linux pseudocode for capturing audio on an HP IPaq (for single-channel 16khz 16bit PCM sampling):

```
fd = open(“/dev/dsp”, O_RDONLY);
ioctl(fd, SOUND_PCM_WRITE_BITS, 16);
ioctl(fd, SOUND_PCM_WRITE_CHANNELS, 1);
ioctl(fd, SOUND_PCM_WRITE_RATE, 16000);
while (1) {
    read(fd, buffer, Nsamples*sizeof(short));
    process(buffer);
}
close(fd);
```

# Speech Capture Models

- *Push* Model: The application captures the speech signal
  - And "pushes" the audio into the speech recognition system for recognition
    - And requests a full or partial result when required
  - The speech recognizer does not need to know about speech capture
- *Pull* Model: The application requests the speech recognition for recognition output
  - The recognizer is responsible for capturing the audio
- The former provides the application more flexibility, but also imposes greater responsibility

# Endpointing



- A very key component of automatic speech recognition is to know the end points of an utterance
  - *I.e.* Endpointing
- Must avoid attempting to recognize speech in non-speech regions
  - Unnecessary computation load
  - May spuriously recognize words where there are none
- Accurate endpointing is very important in both online and offline (batch) recognition scenarios

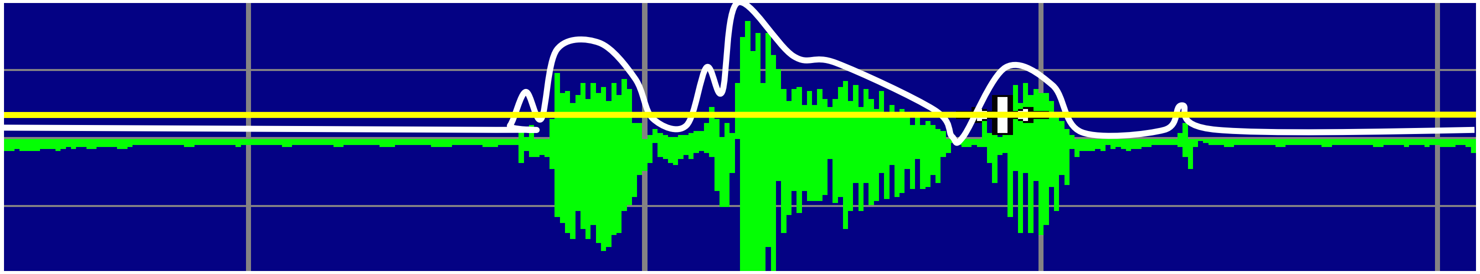
# Online Endpointing Formats

- Push to talk:
  - Manually identify speech regions
    - E.g. hitting a button at the beginning and end of an utterance
    - E.g. maintaining a button pushed while speaking
- Hit to talk:
  - Manually identify the *beginning* of an utterance
    - E.g. hit a button/key and start speaking
  - System automatically determines when the utterance ends
- Continuous listening:
  - The system "listens" to incoming audio continuously
  - It automatically determines which regions of captured audio are speech and must be recognized

# Batch Endpointing

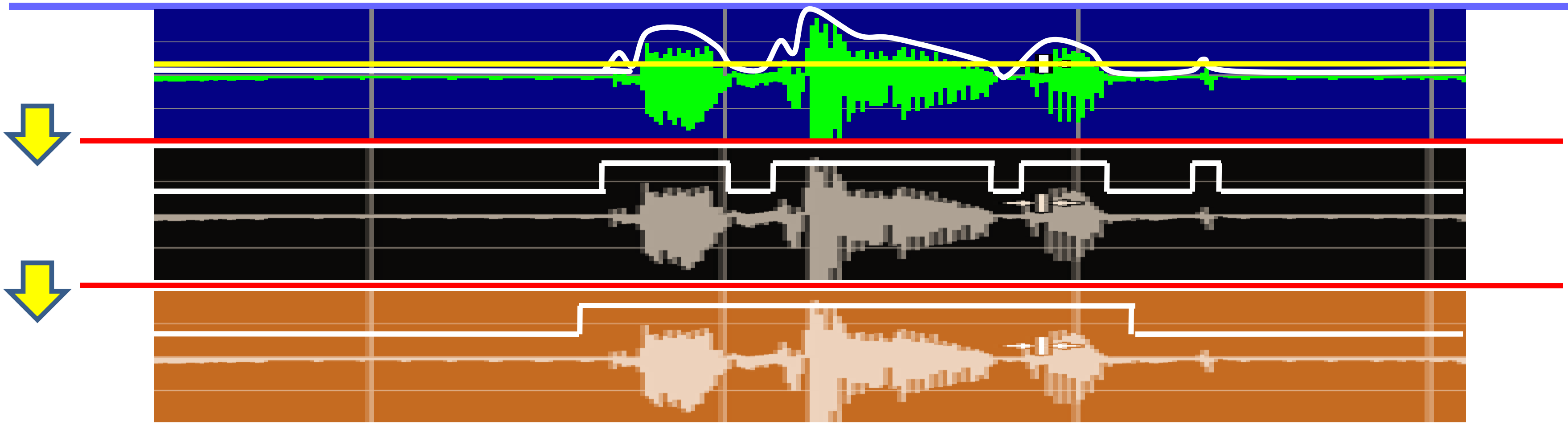
- Similar to the mechanism used for continuous listening, with a couple of options
  - Perform endpointing based on acoustic properties of the signal
    - The same mechanism used by continuous listening systems
  - Use the speech recognizer itself to determine where speech is present
    - M*ultiple passes* of processing are allowed
  - Combination of the above
    - A first pass of conservative acoustic-based endpointing
    - Subsequent refinement by recognition

# A Simple Endpointing Scheme



- Energy in speech regions is typically higher than the energy in non-speech regions
- Simple algorithm: apply a threshold on the current energy level of the signal
  - – If energy exceeds the threshold, it is probably speech
- *Energy-based endpointing*

# Energy Based Endpointing



- Compute the energy at each time t:
  - Energy at time=t is computed as the sum of squared sample values in a short “frame” of samples around *t*
  - Typically expressed in decibels (dB): 10 log($\Sigma$ $x_i^2$), where $x_i$ are the sample values in the frame
- Compare the energy to a pre-defined threshold
  - If the energy is above the threshold, we label the signal at time *t* as speech
- Smooth the labels:
  - Minimum silence and speech segment length limits may be imposed
  - Segments of silence that are shorter than (e.g.) 100 ms are relabeled as speech, provided the resulting speech segment is longer than (e.g.) 250 ms
  - Extend the boundaries of the speech segment by 250ms at the beginning and end
- This scheme works reasonably well under quiet background conditions

# An Adaptive Endpointing Algorithm

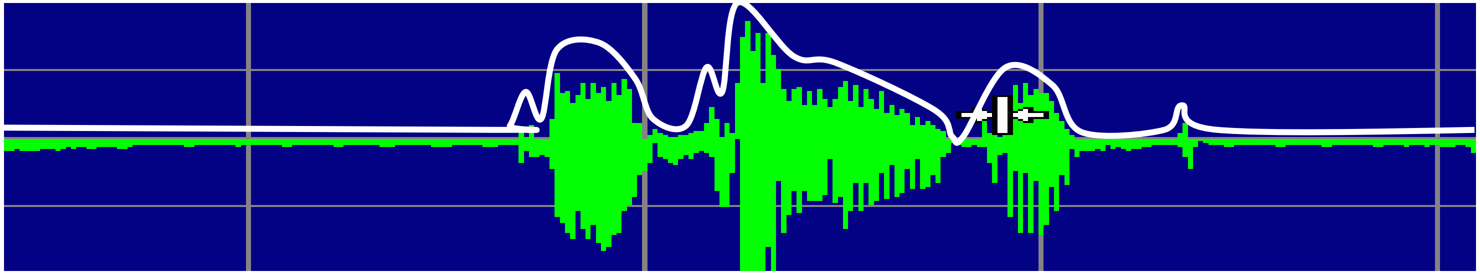
```
Function classifyFrame(audioframe):
    current = EnergyPerSampleInDecibel(audioframe)
    isSpeech = False
    level = ((level * forgetfactor) + current) / (forgetfactor+ 1)
    if (current < background):
        background = current
    else:
        background += (current - background) * adjustment
    if (level < background):   level = background
    if (level - background > threshold): isSpeech = True
    return isSpeech
```

- “forgetfactor” = forgetting factor;   typical value >= 1
- “level” = smoothed signal level; initial value typically set to energy of first frame
- “threshold” = upper (onset) threshold
- “background” = background signal level; initially set to avg energy of first 10 frames
- “adjustment” = adaptive update factor for background; higher values assume faster varying background.  Typical value 0.05

# Adaptive Endpointing

- The adaptive endpointing
  - Compares a smoothed signal level to a fast-varying estimate of background
  - *Adaptation* achieved by adapting background level
- Smoothing of speech/non-speech labels is still required

# A More Complex Algorithm



- The *onset* of speech is indicated by a sudden large increase in speech energy
  - As compared to the “background” energy level
  - This is apparent even in high noise
- The *termination* of speech is indicated a smaller fall in energy level
  - Which may not be apparent in noise
- The more complete endpointing algorithm uses *two* thresholds to represent these phenomena
  - The thresholds are also made adaptive to account for changing noise conditions

# A Two-Threshold Formula

```
OnsetThreshold = X
OffsetThreshold = X – dynamic_range
If (inspeech):
      if (energy-background < OffsetThreshold):   inspeech = 0
Else:
      if (energy-background > OnsetThreshold):   inspeech = 1
 If (inspeech):
       background += (energy-background)*trackingfactor
 Else:
      background = alpha*background + (1-alpha)*energy
```

- As before, the speech energy may also be smoothed
- The actual label sequence obtained must be smoothed to prevent unnecessary chopping up of audio
  - But oversmoothing can cause a lot of non-speech segments to be labelled as speech

# End pointing: Hit to talk

- In Hit-to-Talk the *start* of speech is indicated
  - However actual speech may begin shortly after the button is clicked
  - You may also record the button click
    - Undesired signal

- You only need to determine the end of speech
- Some adjustments are needed in the algorithm to determine background level from the early segment of the recording
  - Alternately, we may listen continuously in order to update background estimate, but only indicate speech when the button is hit

# Assignment

- Write a program for data capture
- Must include:
  - Endpointing with Hit-to-talk
  - Data capture
    - 16kHz, 16bit PCM data
  - Endpointed segment must be written to file

- You can use portaudio for the audio capture