- Previous Lecture:
- Recursion
- Today's Lecture:
- Working with sound files
- Review vector, graphics, struct array, cell array
- Announcements:
- P5 due Saturday $4 / 14$ at 11 pm
- Review session Sunday 2:30-4pm, location TBA
- Prelim 3 Tuesday $4 / 17$ at $7: 30 \mathrm{pm}$

```
Reading and playing .wav files
    [y,rate,nBits] = wavread('austin.wav')
sound(y,rate)
```

A wav file is for the computer to processsoftware is required to play the sound.

Computing with sound in Matlab requires that we first convert the wav format data into simple numeric data-the job of wavread.

Computing with sound requires digitization

- Sound is continuous; capture its essence by sampling
- Digitized sound is a vector of numbers


Lecture 24

| Sampling Rate |  |
| :--- | :--- |
| Given human perception, 20000 samples/second <br> is pretty good <br> $(20000 \mathrm{~Hz}$ or 20 kHz$)$ |  |
| $8,000 \mathrm{~Hz}$ | required for speech over the <br> telephone |
| $44,100 \mathrm{~Hz}$ | required for audio CD |
| $192,400 \mathrm{~Hz}$ | required for HD-DVD <br> audio tracks |

Sampling rate affects the quality

If sampling not frequent enough, then the discretized sound will not capture the essence of the continuous sound...


Lecture 24

Resolution also affects the quality
Typically, each sampled value is encoded as an 8bit integer in the .wav file.

Possible values: -I28, -I27,...,-I, 0, I, ...,I27
Loud: -I20, 90, I22, etc.
Quiet: 3, $10,-5$


16-bit used when very high quality is required.
wavread converts the 8-bit values to floating point values between -I and I
[y, rate, nBits]= wavread('austin.wav')
0.4609
0.3516
0.2734
0.2891
0.2500
$0.1484 \longleftarrow y(50000: 50012)$
0.1094
0.1641
0.1484
0.0000
-0.1641
$-0.2734$
-0. 3281

| wavread |  |
| :---: | :---: |
| $\begin{aligned} & \text { [y, rate, nBits]= } \\ & n=\text { length }(y) \end{aligned}$ | wavread('austin.wav'); |
| $\begin{aligned} & n= \\ & \text { rate }=\begin{array}{l} 54453 \\ 11025 \end{array} \end{aligned}$ | austin.wav <br> encoded the sound with 54,453 8-bit numbers that were taken at 11025 samples per second |
| $\text { nBits }=8$ | A: rate*n |
| What is the play duration? | B: rate/n C: $\mathrm{n} /$ rate |
|  | D: none of the above |
|  | Leture 24 |



Example: playlist
Suppose we have a set of .wav files, e.g.,

> austin.wav
> sp_beam. wav
> sp_oz6.wav
and wish to play them in succession.

```
Possible solution
playList = {'austin',...
    'sp_beam',...
    'sp_oz6'};
```

for $k=1$ : length(playList)
[y,rate] = wavread(playList\{k\});
sound( $y$, rate)
end

| function playSegments(SA) |  |
| :---: | :---: |
| \% Play sound data stored in struct array$\%$ SA(k).data stores the k -th segment of |  |
|  |  |
| \% sound data | (from wavread) |
| \% SA(k).rate is sampling | rate of k-th seg. |
| for $k=1$ length(SA) |  |
| theData= SA(k).data; theRate= SA(k).rate; sound(theData, theRate) | Next call to sound will not begin until after the previous call is complete. |
| end | Not true in older versions! Calculate and add your own pause in that case. |
| Leetre $24 \times 18$ |  |

My emergency alarm clock!
The command
clock

returns a length 6 vector of these values
[year month day hour minute seconds]

Write this function:
function alarmclock(h,m,filename)
\% Play wav file at h:m


