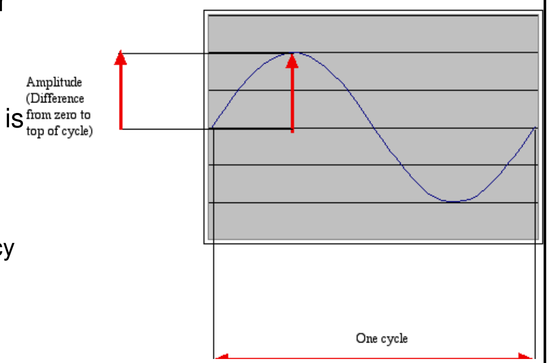
Sound Encoding and Manipulation

What is sound?

- Waves of air (or water if underwater) pressure

How sound works: Acoustics, the physics of sound

- Sounds are waves of air pressure
  - Sound comes in cycles
  - The *frequency* of a wave is the number of cycles per second (cps), or *Hertz*
    - (Complex sounds have more than one frequency in them.)
  - The amplitude is the maximum height of the wave

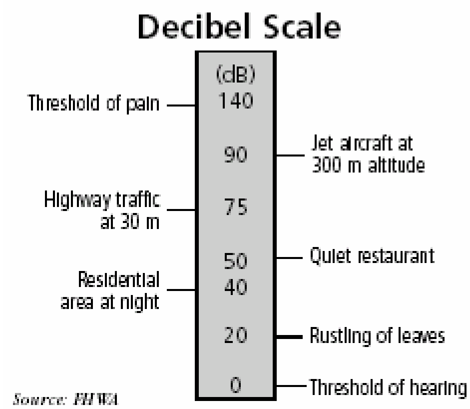


Decibel is a logarithmic measure

- A *decibel* is a ratio between two intensities: $10 * \log_{10}(I_1/I_2)$
  - As an absolute measure, it's in comparison to threshold of audibility
  - 0 dB can't be heard.
  - Normal speech is 60 dB.
  - A shout is about 80 dB

Intensity of Sound

- Decibel Scale

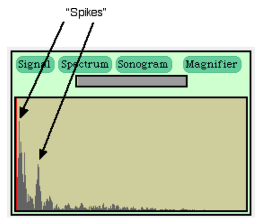


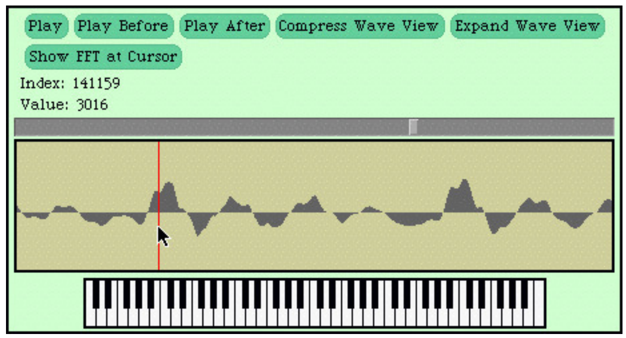
Volume and pitch: Psychoacoustics, the psychology of sound

- Our perception of volume is related (logarithmically) to changes in amplitude
  - If the amplitude doubles, it's about a 3 decibel (dB) change
- Our perception of pitch is related (logarithmically) to changes in frequency
  - Higher frequencies are perceived as higher pitches
  - We can hear between 5 Hz and 20,000 Hz (20 kHz)
  - Middle C is 262 Hz
- It's strange, but our hearing works on *ratios* not *differences*, e.g., for pitch.
  - We hear the difference between 200 Hz and 400 Hz, as the same as 500 Hz and 1000 Hz
  - Similarly, 200 Hz to 600 Hz, and 1000 Hz to 3000 Hz

Demonstrating Sound MediaTools

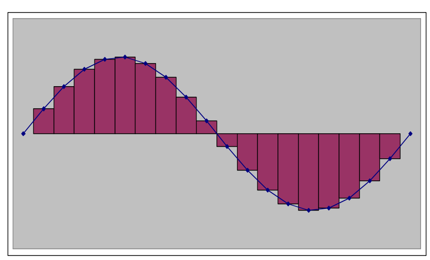
Fourier transform (FFT)





Digitizing Sound: How do we get that into numbers?

- Remember in calculus, estimating the curve by creating rectangles?
- We can do the same to estimate the sound curve
  - Analog-to-digital conversion (ADC) will give us the amplitude at an instant as a number: a *sample*
  - How many samples do we need?



Nyquist Theorem

- We need twice as many samples as the maximum frequency in order to represent (and recreate, later) the original sound.
- The number of samples recorded per second is the *sampling rate*
  – If we capture 8000 samples per second, the highest frequency we can capture is 4000 Hz
    - That's how phones work
  – If we capture more than 44,000 samples per second, we capture everything that we can hear (max 22,000 Hz)
    - CD quality is 44,100 samples per second

Digitizing sound in the computer

- Each sample is stored as a number (two bytes)
- What's the range of available combinations?
  – 16 bits, 2^{16} = 65,536
  – But we want both positive and negative values
    - To indicate compressions and rarefactions.
  – What if we use one bit to indicate positive (0) or negative (1)?
  – That leaves us with 15 bits
  – 15 bits, 2^{15} = 32,768
  – One of those combinations will stand for zero
    - We'll use a "positive" one, so that's one less pattern for positives

+/- 32K

- Each sample can be between -32,768 and 32,767

Why such a bizarre number?

Because 32,768 + 32,767 + 1 = 2^{16}

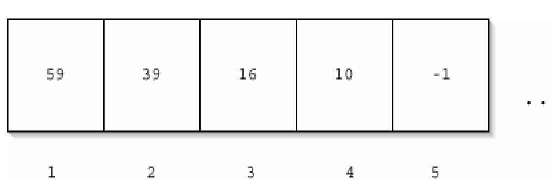
< 0 **> 0** **0** **i.e. 16 bits, or 2 bytes**

Compare this to 0..255 for light intensity

(i.e. 8 bits or 1 byte)

Sounds as arrays

- Samples are just stored one right after the other in the computer's memory
- That's called an *array* **(Like pixels in a picture)**
  - It's an especially efficient (quickly accessed) memory structure



Working with sounds

- We'll use **pickAFile** and **makeSound**.
  - We want .wav files
- We'll use **getSamples** to get all the *sample objects* out of a sound
- We can also get the value at any index with **getSampleValueAt**
- Sounds also know their length (**getLength**) and their sampling rate (**getSamplingRate**)
- Can save sounds with **writeSoundTo(sound,"file.wav")**

Demonstrating Working with Sound in JES

```
>>> filename = pickAFile()
>>> print filename
c:\preamble.wav
>>> sound = makeSound(filename)
>>> print sound
Sound of length 421109
>>> samples = getSamples(sound)
>>> print samples
Samples, length 421109
>>> print getSampleValueAt(sound, 1)
36
>>> print getSampleValueAt(sound, 2)
29
```

Demonstrating working with samples

```
>>> print getLength(sound)
220568
>>> print getSamplingRate(sound)
22050.0
>>> print getSampleValueAt(sound, 220568)
68
>>> print getSampleValueAt(sound, 220570)
I wasn't able to do what you wanted.
The error java.lang.ArrayIndexOutOfBoundsException has occured
Please check line 0 of
>>> print getSampleValueAt(sound, 1)
36
>>> setSampleValueAt(sound,1, 12)
>>> print getSampleValueAt(sound, 1)
12
```

Working with Samples

- We can get sample objects out of a sound with **getSamples(sound)** or **getSampleObjectAt(sound, index)**
- A sample object remembers its sound, so if you change the sample object, the sound gets changed.
- Sample objects understand **getSample(sample)** and **setSample(sample, value)**

Example: Manipulating Samples

```
>>> soundfile=pickAFile()
>>> sound=makeSound(soundfile)
>>> sample=getSampleObjectAt(sound, 1)
>>> print sample
Sample at 1 value at 59
>>> print sound
Sound of length 387573
>>> print getSound(sample)
Sound of length 387573
>>> print getSample(sample)
59
>>> setSample(sample, 29)
>>> print getSample(sample)
29
```

Can you hear the difference?

"But there are thousands of these samples!"

- How do we do something to these samples to manipulate them, when there are thousands of them per second?
- We use a *loop* and get the computer to *iterate* in order to do something to each sample.
- An example loop:

```
for sample in getSamples(sound):
   value = getSample(sample)
   setSample(sample, value)
```

Recipe to Increase the Volume

```
def increaseVolume(sound):
  for sample in getSamples(sound):
    value = getSample(sample)
    setSample(sample, value * 2)
```

Using it:

```
>>> setMediaPath()
>>> s = makeSound(“gettysburg10.wav")
>>> increaseVolume(s)
>>> play(s)
>>> writeSoundTo(s, "louder-g10.wav")
```

Starting the loop

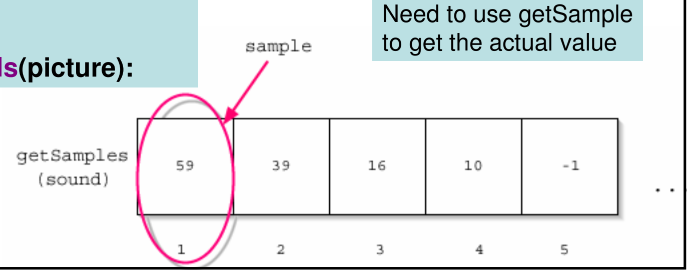
- **getSamples(sound)** returns a sequence of all the sample objects in the **sound**.
- The **for** loop makes **sample** be the first sample as the block is started.

```
def increaseVolume(sound):
  for sample in getSamples(sound):
    value = getSample(sample)
    setSample(sample, value * 2)
```

Compare:

for pixel in getPixels(picture):

Need to use getSample to get the actual value



Executing the block

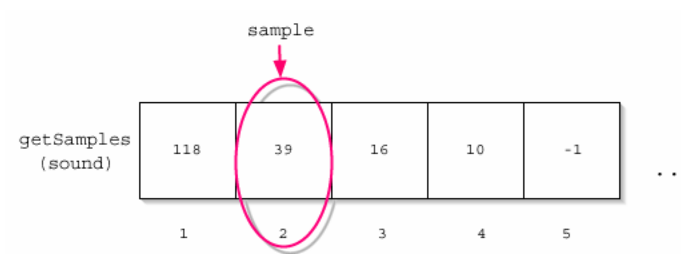
- We get the value of the sample named **sample.**
- We set the value of the sample to be the current value (variable **value**) times 2

```
def increaseVolume(sound):
  for sample in getSamples(sound):
    value = getSample(sample)
    setSample(sample, value * 2)
```

Next sample

- Back to the top of the loop, and **sample** will now be the second sample in the sequence.

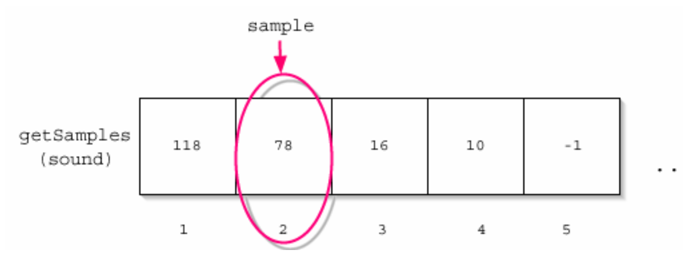
```
def increaseVolume(sound):
  for sample in getSamples(sound):
    value = getSample(sample)
    setSample(sample, value * 2)
```



And increase that next sample

- We set the value of *this* sample to be the current value (variable **value**) times 2.

```
def increaseVolume(sound):
  for sample in getSamples(sound):
    value = getSample(sample)
    setSample(sample, value * 2)
```



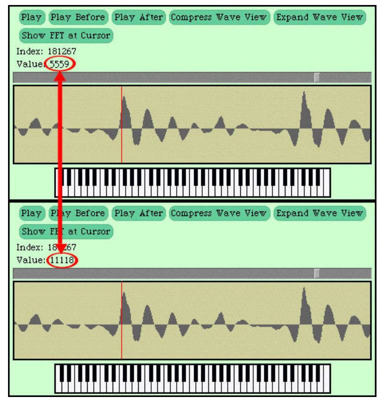
And on through the sequence

- The loop keeps repeating until *all* the samples are doubled

```
def increaseVolume(sound):
  for sample in getSamples(sound):
    value = getSample(sample)
    setSample(sample, value * 2)
```

How are we *sure* that that worked?

```
>>> print s
Sound of length 220567
>>> print f
C:\mediasources\gettysburg10.wav
>>> soriginal = makeSound(f)
>>> print getSampleValueAt(s, 1)
118
>>> print getSampleValueAt(soriginal, 1)
59
>>> print getSampleValueAt(s, 2)
78
>>> print getSampleValueAt(soriginal, 2)
39
>>> print getSampleValueAt(s, 1000)
-80
>>> print getSampleValueAt(soriginal, 1000)
-40
```



Decreasing the volume

```
def decreaseVolume(sound):
  for sample in getSamples(sound):
    value = getSample(sample)
    setSample(sample, value * 0.5)
```

This works *just* like **increaseVolume**, but we're *lowering* each sample by 50% instead of doubling it.

Recognize some similarities?

```
def increaseVolume(sound):
  for sample in getSamples(sound):
    value = getSample(sample)
    setSample(sample, value * 2)
```

↔

```
def increaseRed(picture):
  for p in getPixels(picture):
    value = getRed(p)
    setRed(p, value*1.2)
```

```
def decreaseVolume(sound):
  for sample in getSamples(sound):
    value = getSample(sample)
    setSample(sample, value * 0.5)
```

↔

```
def decreaseRed(picture):
  for p in getPixels(picture):
    value = getRed(p)
    setRed(p, value*0.5)
```

Does increasing the volume change the volume setting?

- No
  - The physical volume setting indicates an *upper bound*, the potential loudest sound.
  - Within that potential, sounds can be louder or softer
    - They *can* fill that space, but might not
    - E.g. TV commercial volume louder than show
- What happens if we keep calling increaseVolume over and over again?

Avoiding clipping

- Why are we being so careful to stay within range? What if we just multiplied all the samples by some big number and let some of them go over 32,767?
- The result then is *clipping*
  – Clipping: The awful, buzzing noise whenever the sound volume is beyond the maximum that your sound system can handle.

Maximizing volume

- How, then, do we get maximal volume?
  – (e.g. automatic recording level)
- It's a three-step process:
  1. find the current loudest value (largest sample).
  2. find how much we can increase/decrease that value to fill the available space
     - We want to find the amplification factor *amp,* where *amp * loudest = 32767*
     - In other words: *amp = 32767/loudest*
  3. amplify each sample by multiplying it by *amp*

Maxing (*normalizing)* the sound

```
def normalize(sound):
   largest = 0
   for s in getSamples(sound):
      largest = max(largest, getSample(s) )
   amplification = 32767.0 / largest

   print "Largest sample value in original sound was",  largest
   print ”Amplification multiplier is", amplification

   for s in getSamples(sound):
      louder =  amplification * getSample(s)
      setSample(s, louder)
```

This loop finds the loudest value between the two #'s

This loop actually amplifies the sound

max()

- **max()** is a function that takes *any* number of parameters, and returns the largest.
- There is also a function **min()** which works similarly but returns the minimum
- We could also write these ourselves, like we did on homework #2

```
>>> print max(1, 2, 3)
3
>>> print max(4, 67, 98, -1)
98
```

Or: use if instead of max

```
def normalize(sound):
  largest = 0
  for s in getSamples(sound):
    if getSample(s) > largest:
      largest = getSample(s)
  amplification = 32767.0 / largest
  print "Largest sample value in original sound was",  largest
  print "Amplification factor is", amplification
  for s in getSamples(sound):
    louder =  amplification * getSample(s)
    setSample(s, louder)
```

Check each in turn to see if it's the largest so far

Aside: positive and negative extremes assumed to be equal

- We're making an assumption here that the maximum positive value is also the maximum negative value.
  - That should be true for the sounds we deal with, but isn't necessarily true
- Try adding a constant to every sample.
  - That makes it *non*-cyclic
    - I.e. the compressions and rarefactions in the sound wave are not equal
  - But it's fairly subtle what's happening to the sound.

Processing only *part* of the sound

- What if we wanted to increase or decrease the volume of only *part* of the sound?
- Q: How would we do it?
- A: We'd have to use a **range**() function with our **for** loop
  - Just like when we manipulated only part of a picture by using range() in conjunction with getPixels()
  - More about this next time….