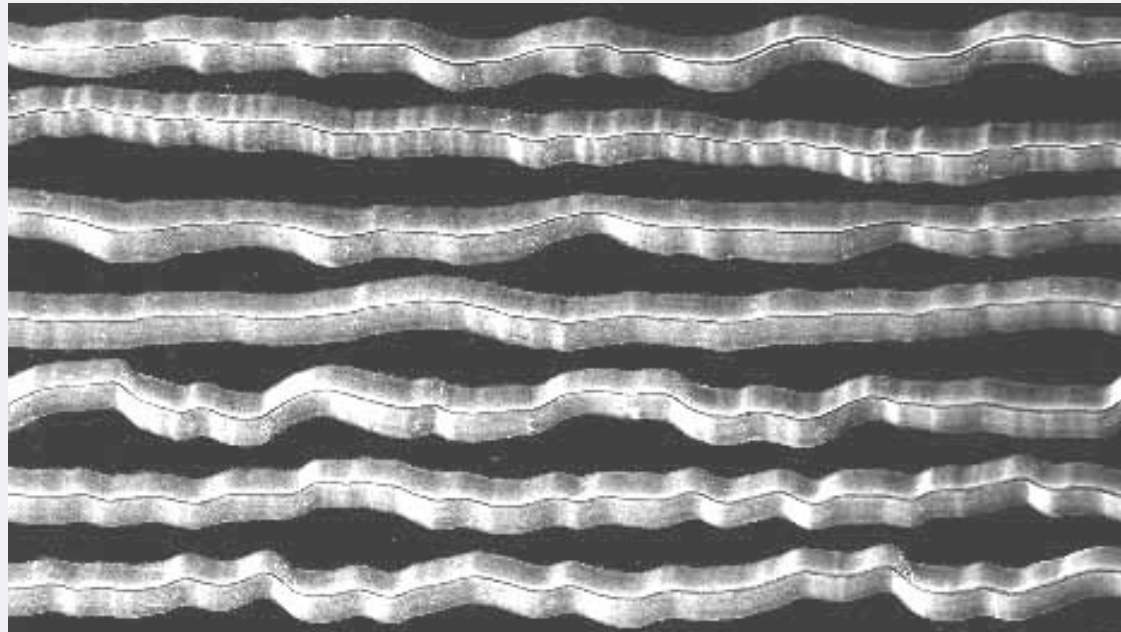


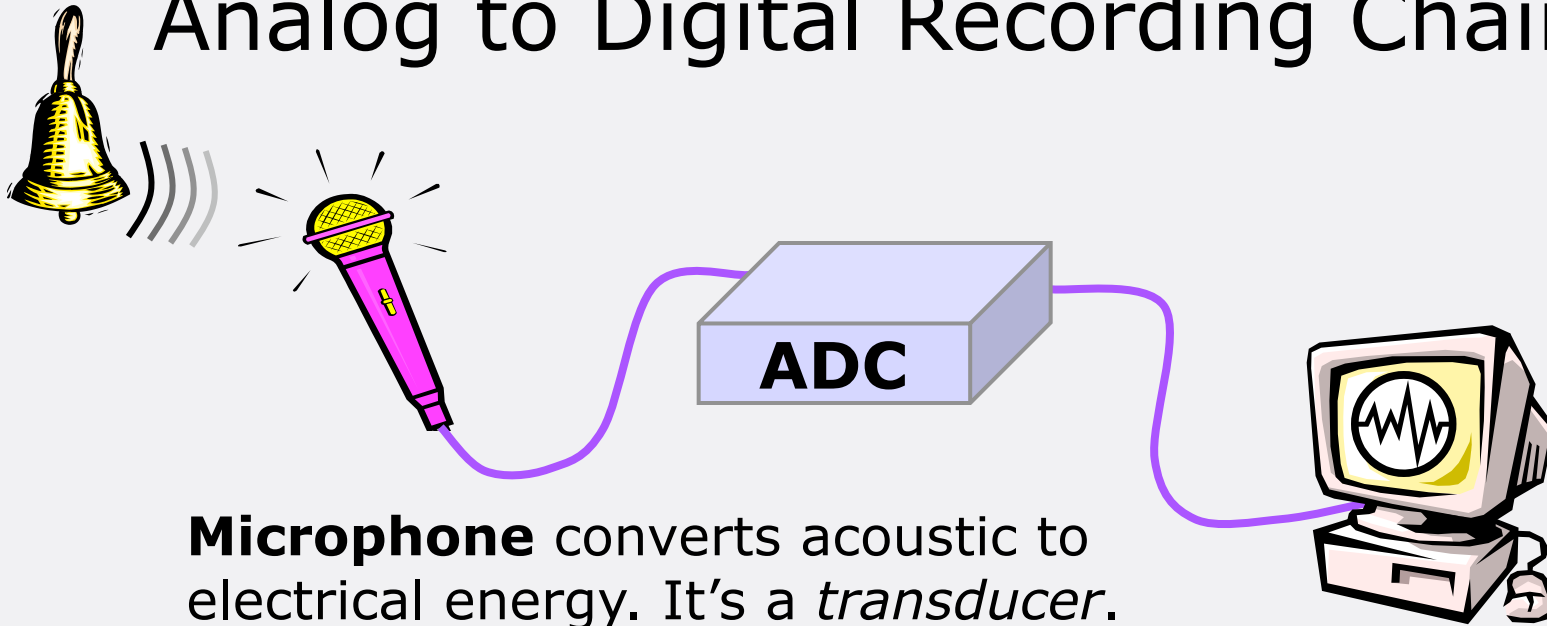
# Analog Representations of Sound

Magnified phonograph grooves, viewed from above:



The shape of the grooves encodes the continuously varying audio signal.

# Analog to Digital Recording Chain



**Microphone** converts acoustic to electrical energy. It's a *transducer*.

Continuously varying electrical energy is an **analog** of the sound pressure wave.

**ADC** (Analog to Digital Converter) converts analog to digital electrical signal. Digital signal transmits binary numbers.

**DAC** (Digital to Analog Converter) converts digital signal in computer to analog for your headphones.

# Analog versus Digital

## Analog

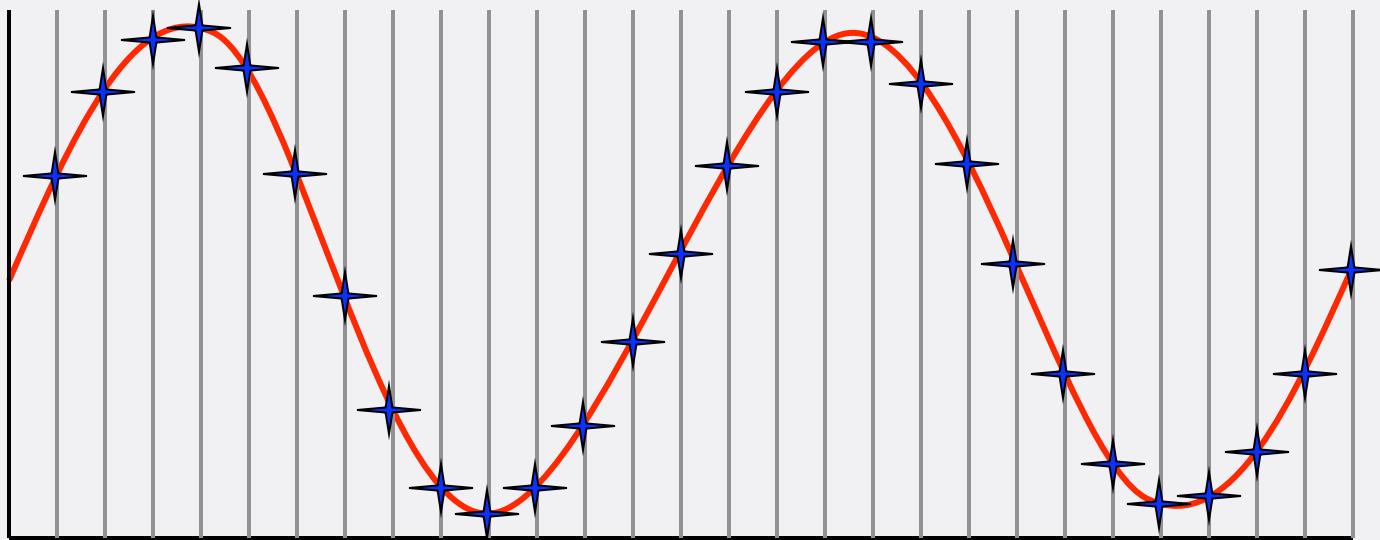
Continuous signal that mimics shape of acoustic sound pressure wave

## Digital

Stream of discrete numbers that represent instantaneous amplitudes of analog signal, measured at equally spaced points in time.

# Analog to Digital Conversion

Instantaneous amplitudes of continuous analog signal, measured at equally spaced points in time.



A series of "snapshots"

# Analog to Digital Overview

## Sampling Rate

How often analog signal is measured  
[samples per second, Hz]

Example: 44,100 Hz

## Sampling Resolution

[a.k.a. "sample word length," "bit depth"]  
Precision of numbers used for  
measurement: the more bits, the higher  
the resolution.

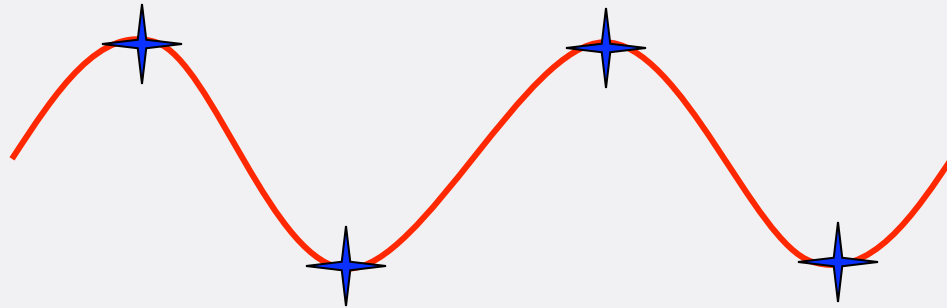
Example: 16 bit

# Sampling Rate

Determines the highest frequency that you can represent with a digital signal.

## **Nyquist Theorem:**

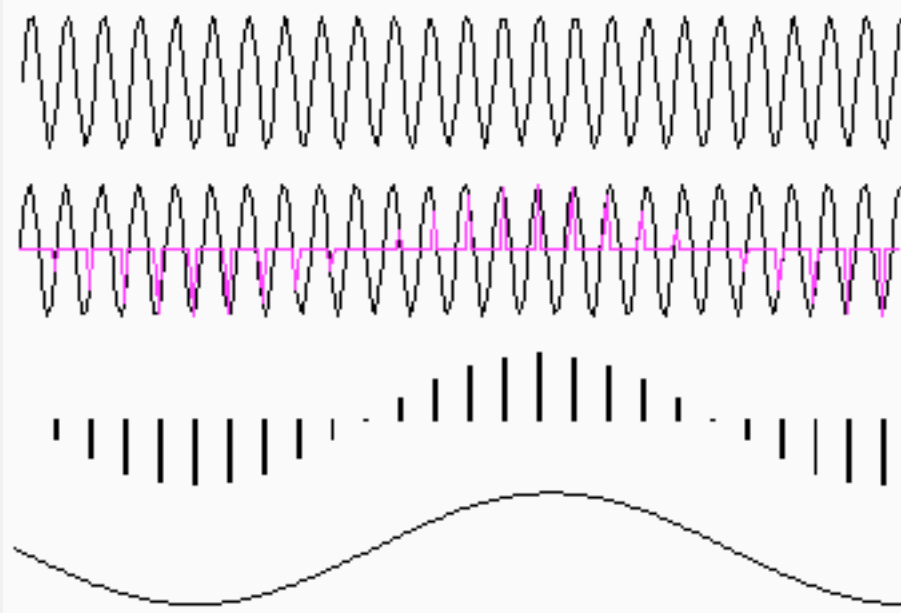
Sampling rate must be at least twice as high as the highest frequency you want to represent.



Capturing just the crest and trough of a sine wave will represent the wave exactly.

# Aliasing

What happens if sampling rate not high enough?



A high frequency signal

sampled at too low a rate

looks like ...

... a lower frequency signal.

That's called **aliasing** or **foldover**. An ADC has a low-pass **anti-aliasing filter** to prevent this.

Synthesis software can cause aliasing.

# Common Sampling Rates

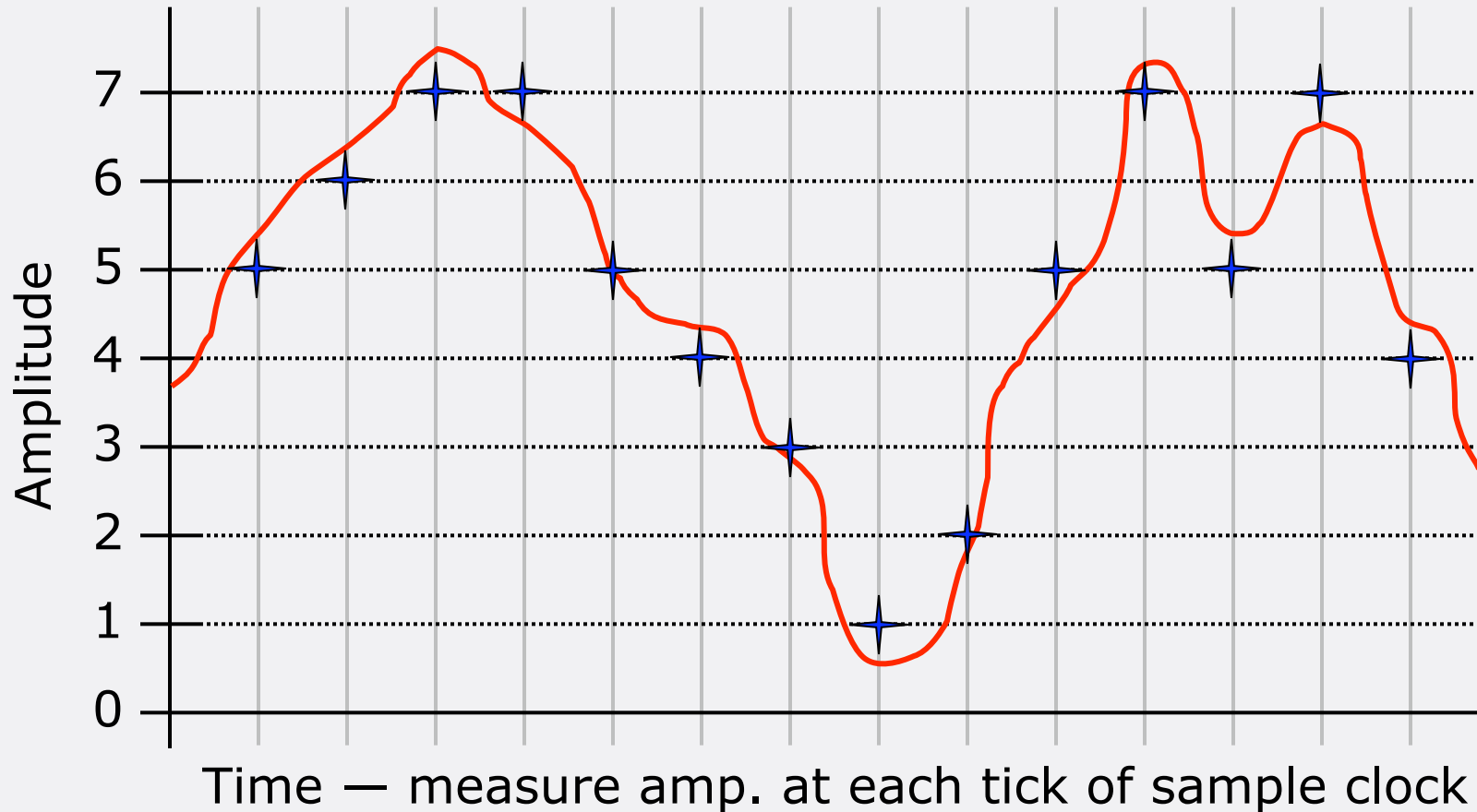
Which rates can represent the range of frequencies audible by (fresh) ears?

<b>Sampling Rate</b>	<b>Uses</b>
44.1 kHz (44100)	CD, DAT
48 kHz (48000)	DAT, DV, DVD-Video
96 kHz (96000)	DVD-Audio
22.05 kHz (22050)	Old samplers

Most software can handle all these rates.

# 3-bit Quantization

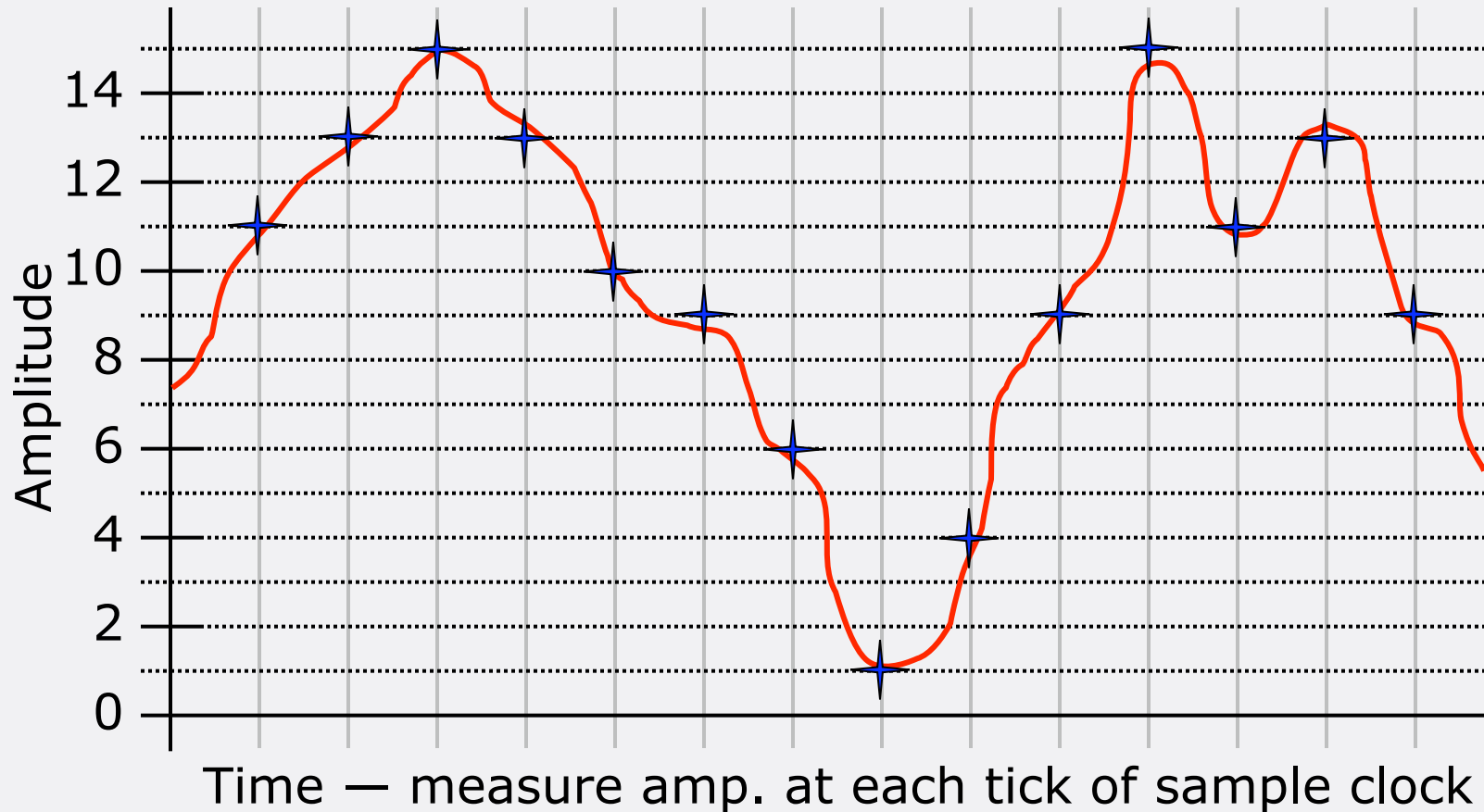
A 3-bit binary (base 2) number has  $2^3 = 8$  values.



A rough approximation

# 4-bit Quantization

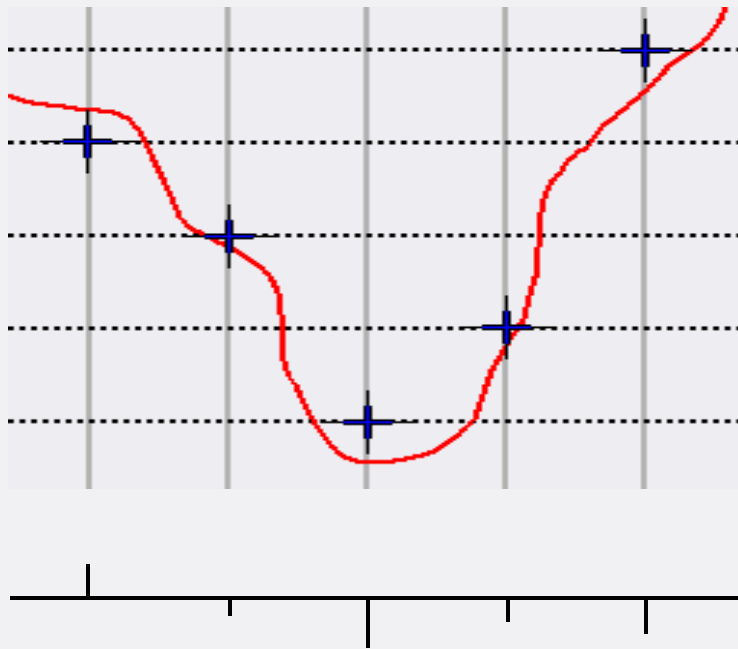
A 4-bit binary number has  $2^4 = 16$  values.



A better approximation

# Quantization Noise

Round-off error: difference between actual signal and quantization to integer values...



Random errors: sounds like low-amplitude noise

# The Digital Audio Stream

It's just a series of sample numbers, to be interpreted as instantaneous amplitudes: one for every tick of the sample clock.

Previous example:

```
11 13 15 13 10 9 6 1 4 9 15 11 13 9
```

This is what appears in a sound file, along with a **header** that indicates the sampling rate, bit depth and other things.

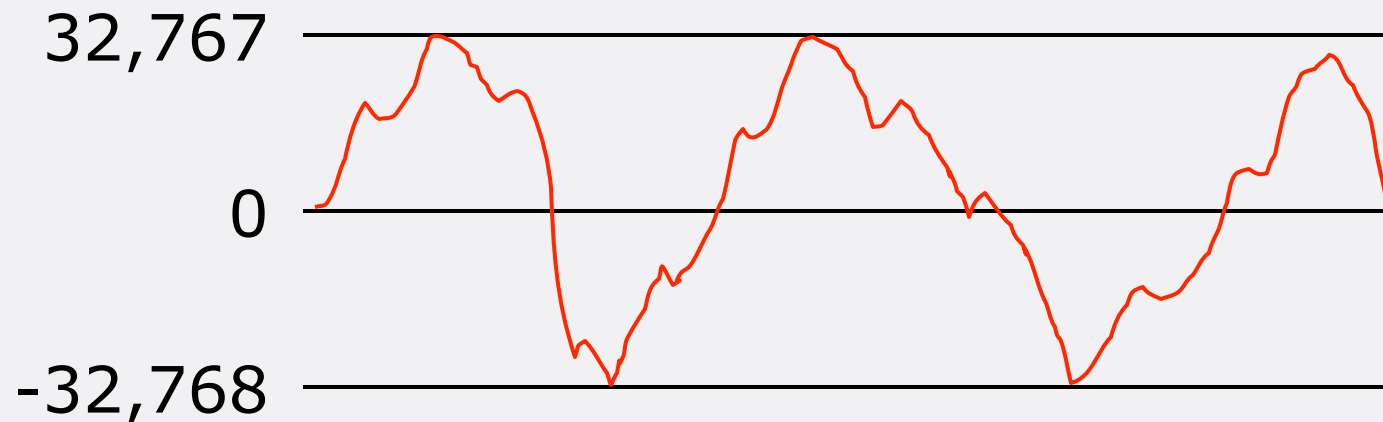
# Common Sampling Resolutions

<b>Word length</b>	<b>Uses</b>
8-bit integer	Low-res web audio
16-bit integer	CD, DAT, DV, sound files
24-bit integer	DVD-Video, DVD-Audio
32-bit floating point	Software (usually only for internal representation)

## 16-bit Sample Word Length

A 16-bit integer can represent  $2^{16}$ , or 65,536, values (amplitude points).

We typically use **signed** 16-bit integers, and center the 65,536 values around 0.



# Audio File Size

## CD characteristics...

- Sampling rate:  
44,100 samples per second (44.1 kHz)
- Sample word length:  
16 bits (i.e., 2 bytes) per sample
- Number of channels:  
2 (stereo)

How big is a 5-minute CD-quality sound file?

# Audio File Size

How big is a 5-minute CD-quality sound file?

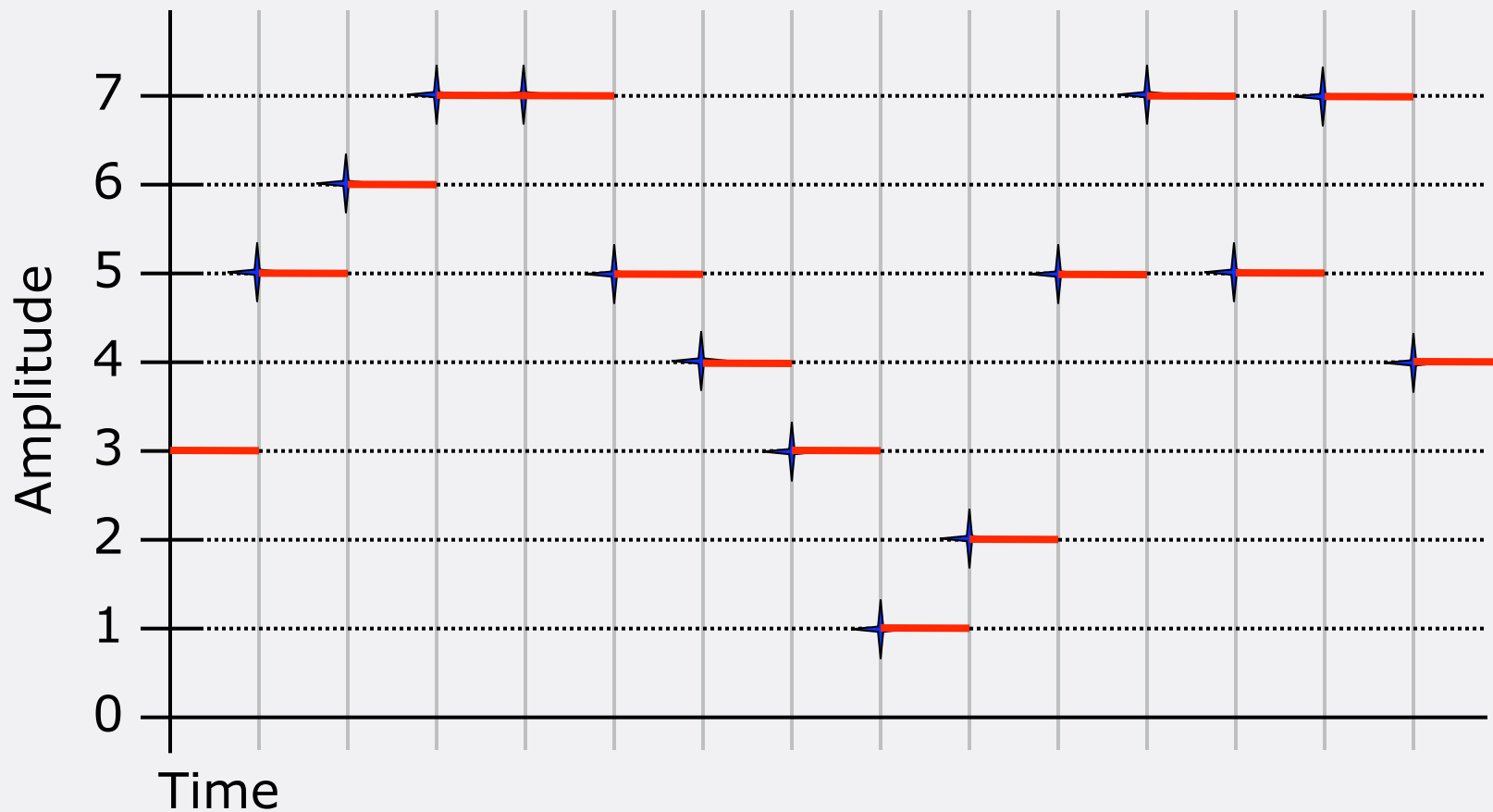
44,100 samples \* 2 bytes per sample \* 2 channels  
= 176,400 bytes per second

5 minutes \* 60 seconds per minute  
= 300 seconds

300 seconds \* 176,400 bytes per second  
= 52,920,000 bytes = c. 50.5 megabytes (MB)

# DAC: Sample and Hold

To reconstruct analog signal, hold each sample value for one clock tick; convert it to steady voltage.



# DAC: Smoothing Filter

Apply an analog low-pass filter to the output of the sample-and-hold unit: averages "stair steps" into a smooth curve.

