IAT 380: Sound Design

Fundamentals of Audio Processing

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Philippe Pasquier, September 2008

Outline of today's lecture

- Fundamentals of Acoustics:
  - Sound and Waveforms
  - Frequency
  - Amplitude
  - Phase and envelope
- Digitalisation of sound:
  - Sampling rate and bit depth
  - Aliasing and quantisation noise
  - The digital audio recording and playback chain

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What is sound?

- Did you hear that?
- For normally constituted humans, true silence does not exist
  - Surrounding sounds are always present: 15khz buzz of your monitor, birds, urban noise, ...
  - Otherwise, you start hearing your body (breath, heart, blood circulation, ...)
- Acoustics (part of classical physics) is the study and analysis of sound properties

⇨ It is the logical starting point to study how to effectively design sound for new media.

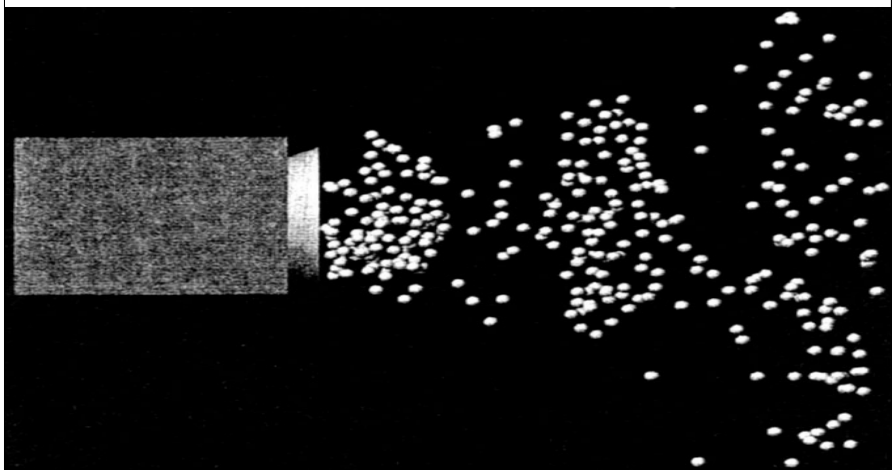
What is sound?

- **What is sound?**
  The aural perception of pressure vibrations in a given medium
  - **Production/Generation: an object vibrates and mechanical energy is transferred into acoustic energy. A good example is a speaker**
  - **Propagation: A sound has to travel from its source to its receptor. In this course, the medium will be air! Sounds are variations in atmospheric pressure which are propagating.**
  - **Perception: The sound is received and interpreted. The human ear is a good example of receiver.**

⇨ **Sound: Variations of atmospheric pressure in the sonic range.**

P&Propagation of sonic waves

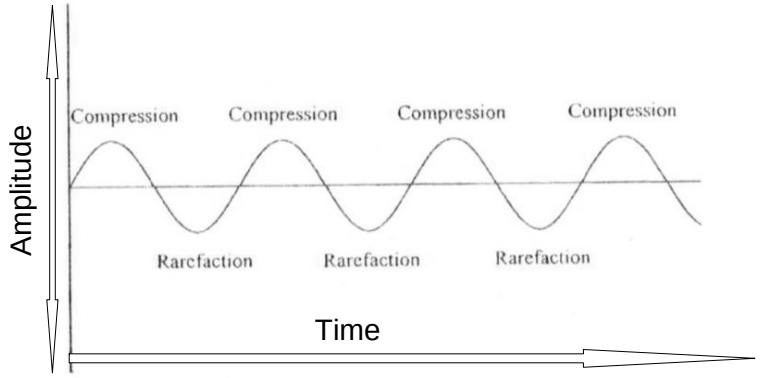
- **Compression and rarefaction: molecules are just slightly displaced only the wave is moving (propagating).**



P&Propagation of sonic waves

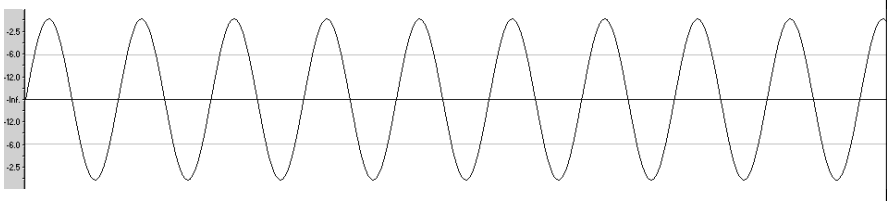
- **Another way of looking at the sound:**

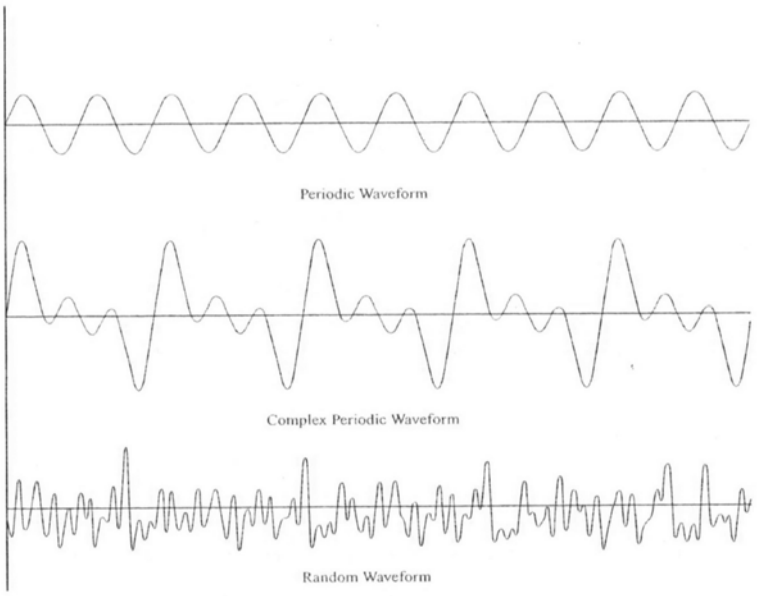
The waveform



Wavesforms

- **A waveform is a visual representation that helps to analyse the various characteristics of a given sound:**
  - **Frequency, amplitude, wavelength, envelope, harmonics**
- **Mostly, we will focus on sine waves**
- **Indeed, all waveforms can be thought of as a sum of sine waves (Fourier Theorem)**





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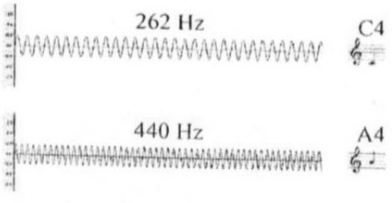
  - **Aliasing and quantisation noise**
  - **The digital audio recording and playback chain**

Sonic Waves Frequency

- Frequency: rate at which a cycle of compression and rarefaction occurs
- Expressed in Hertz (Hz): number of cycle per second.



- Frequency corresponds to perceived pitch: the higher the frequency, the higher the pitch.

Sonic Waves Frequency

- The frequency range of human hearing is approximately 20Hz to 20,000Hz
- Below, it is subsonic (or infrasonic) and above, it is ultrasonic
- Animals are not equal!

Table 1-1. Frequency range of common animals

Name	Lowest Approximate Frequency	Highest Approximate Frequency
Human	20 Hz	20,000 Hz
Cat	45 Hz	85,000 Hz
Dog	50 Hz	45,000 Hz
Elephant	5 Hz	10,000 Hz
Bat	10,000 Hz	120,000 Hz
Dolphin	1,000 Hz	120,000 Hz

Sonic Waves Frequency

- Conversely, objects can produce sonic waves in various frequency ranges:

Brass	Top
Soprano saxophone	225 Hz—1 kHz
Alto saxophone	125 Hz—900 Hz
Tenor Saxophone	110 Hz—630 Hz
Baritone saxophone	70 Hz—450 Hz
Bass saxophone	55 Hz—315 Hz
Trumpet (C)	170 Hz—1 kHz
Trumpet (F)	300 Hz—1 kHz
Alto trombone	110 Hz—630 Hz
Tenor Trombone	80Hz—600Hz
Bass Trombone	63Hz—400Hz
Tuba	45Hz—375Hz
Valve Horn	63Hz—700Hz
Strings	
Violin	200Hz—3.5K
Viola	125Hz—1K
Cello	63Hz—630Hz
Double Bass	40Hz—200Hz
Guitar	80Hz—630Hz
Keyboards	
Piano	28Hz—4.1K
Organ	20Hz—7K
Percussion	
Celeste	260Hz—3.5K
Timpani	90Hz—180Hz
Glockenspiel	63Hz—180Hz
Xylophone	700Hz—3.5K

Vocal	Approximate Frequency Range
Soprano	250 Hz—1 kHz
Contralto	200 Hz—700 Hz
Baritone	110 Hz—425 Hz
Bass	80 Hz—350 Hz
Woodwind	**Top**
Piccolo	630 Hz—5 kHz
Flute	250 Hz—2.5 kHz
Oboe	250 Hz—1.5 kHz
Clarinet (B flat or A)	125 Hz—2 kHz
Clarinet (E flat)	200 Hz—2 kHz
Bass clarinet	75 Hz—800 Hz
Basset horn	90 Hz—1 kHz
Cor Anglais	160 Hz—1 kHz
Bassoon	55 Hz—575 Hz
Double bassoon	25 Hz—200 Hz

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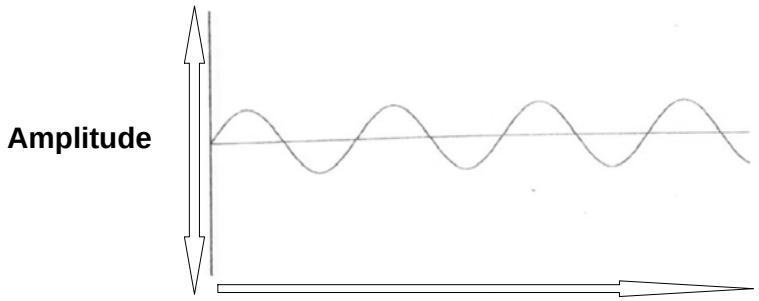
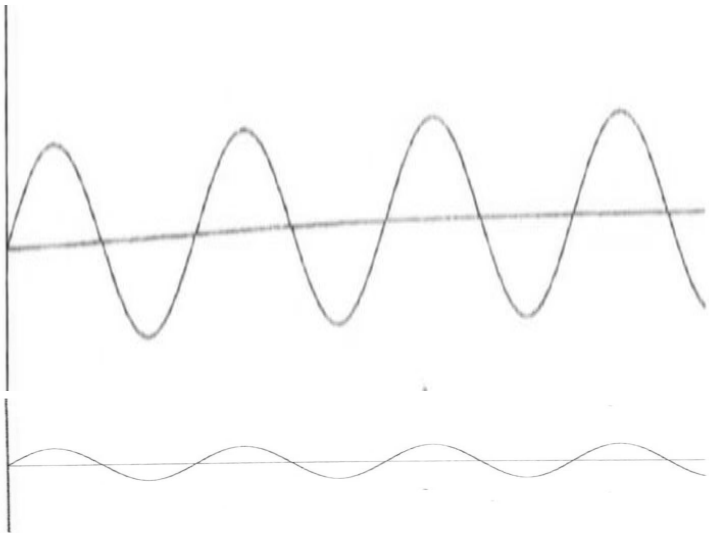
Amplitude of sonic Waves

- Amplitude measure the deviation from normal atmospheric conditions, that is how much compression or rarefaction
- Related to the energy behind the vibration



Amplitude of sonic Waves

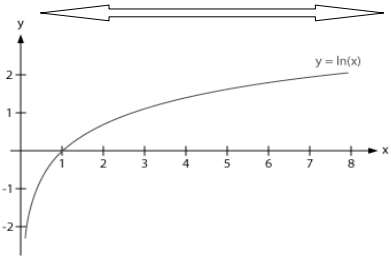


Amplitude of sonic Waves

- **Loudness (amplitude as perceived) is a complex averaging of peak-to-peak values.**
- **RMS (root mean square) is used for this average: RMS=peakvalues*0.707**
- **While intensity is usually expressed in Watt per meter square, sound level is measured using decibels (dB): a logarithmic unit to represent a ratio (in various domains):**
  - **This is adequate because humans perceive sound logarithmically**
  - **This solves the problem of the intensity scale**

Loudness coefficient

Difference in dB



Amplitude of sonic Waves

- **Literally, a decibel (dB) is one tenth of a Bel (from Alexander Graham Bell, more famous for telephone).**
- **Because dB is a generic unit, we use dB SPL (for Sound Pressure Level). Other varieties include dBm for electrical power and dBPWL for acoustic power**
- **Because it is expressing a ratio, dB measures are always compared to a reference level, usually set at 0dB. Ex. 0dB can be associated to silence (the threshold of hearing) or to intensity of a given input for a mixing board, ...**
- **Because dB measure a ration and is logarithmic: if a sound is 10dB louder then another, it means it is 10 times louder than a sound at 0dB while 5dB is only 3.1 louder. It is useful to remember that 3dB corresponds to a 2:1 ratio, 6dB to 4:1 and 12dB to a 16 one.**

Amplitude

The decibel scale

(is more reasonable than a linear intensity scale)

Sound	Decibel (dB)	Description
Threshold of hearing	0	The quietest audible sound
Quiet whisper	10	Just audible
Quiet recording studio		
Bedroom		
Public library	30	Room ambience, very quiet
Very, very soft music		
Normal suburban outdoors	40	Quiet but clearly audible
A small office	50	Low-level speech
Very soft music		
Low radio volume		
Noisy business office		
Conversational speech between two people 3 to 5 feet apart	60	Very noticeable, intrusive
Ringing telephones		
Normal practice piano		
Very noisy, large office	70	Hard to talk on the telephone
Road traffic		
Noisy restaurant		
Very loud singer 3'		
Loud busy traffic corner		
Average factory	80	Agitating and annoying
Heavy truck	90	Level at which hearing loss will occur if sustained for more than 8 hours a day
New York subway		
Inside large orchestra	100	Hearing loss will occur if sustained for more than 2 hours per day
Power tools	110	Hearing loss will occur if sustained for more than 15 minutes per day
Planes on a runway	120	Hearing loss will occur if sustained for more than a short period of time
Artillery 100 yards	130	Pain threshold
Pneumatic (jack) hammer		Brief exposure can cause hearing damage
Jet engines 25 meters	140	Danger levels
Physical pain	150	Extreme danger levels
Chest vibrates		
Choking		Permanent, instantaneous hearing damage
Close proximity explosion	160—180	Death of hearing tissue
		Perforation of ear drum

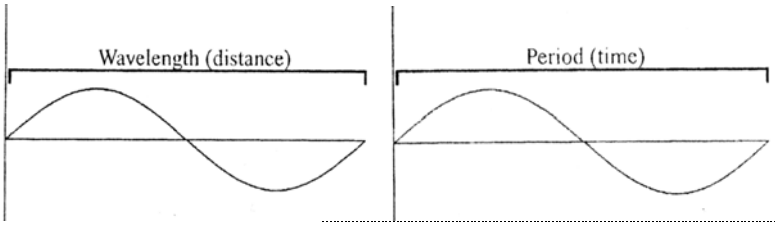
Sound Propagation and Amplitude

- **The velocity of sound: sound propagation occurs at 344m/s (+-1200km/hour) in air (at 20 degree Celsius, 70% humidity under normal air pressure). 350m/s at 30 degree.**
- **Sound intensity decreases with the distance: for every doubling of the distance, sound loses 6dB (factor 4) of intensity.**
- Example: if I am at 5 meters from the speaker and you are at 10 meters, you will hear the sound 6dB softer (signal is 4 times less intense) and someone at 20 meters 6dB (4 times) softer the you, that is a total of 12dB softer (16 times) then me.
- **This is known as the inverse square law.**

Sound propagation: Wave length

- The wavelength of a sound wave is the physical distance of a complete cycle
  - The longer the wavelength, the lower is the frequency.
  - Concretely, 20KHz as a 2cm Wavelength and 20Hz, 17meters
- The period is the time it takes to complete a cycle: on waveforms, we measure from one point of the waveform to the same point in the next cycle

Frequency(in Hz)=1/Period(in sec)



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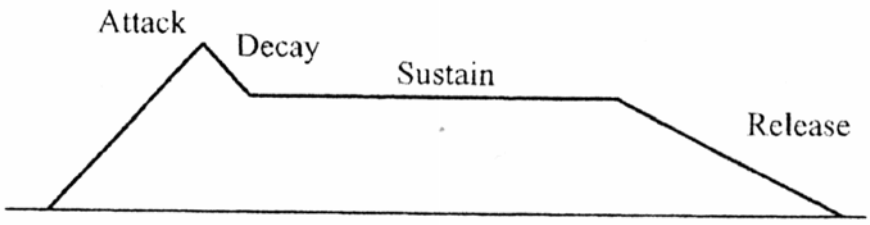
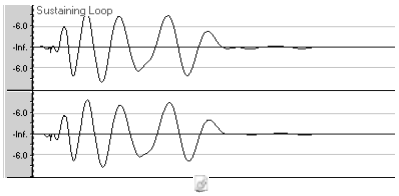
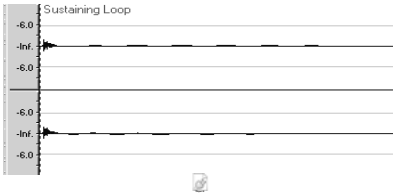
Envelope

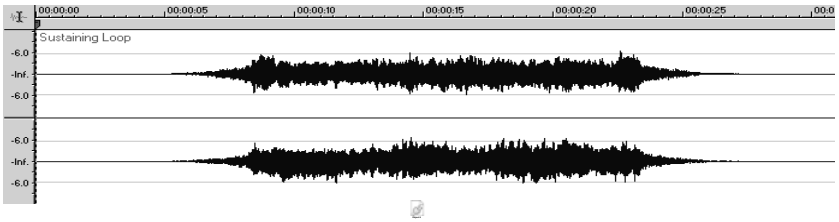
- **The envelope of a sound is the variation of intensity of this sound through time (on a human hearing scale).**
- **A popular type of envelope is the ADSR**



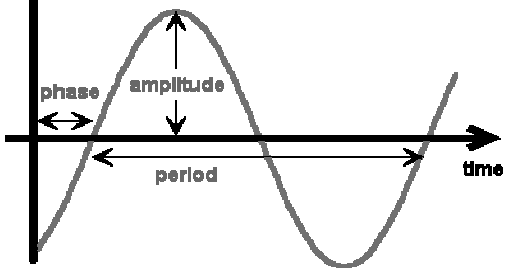
Envelope





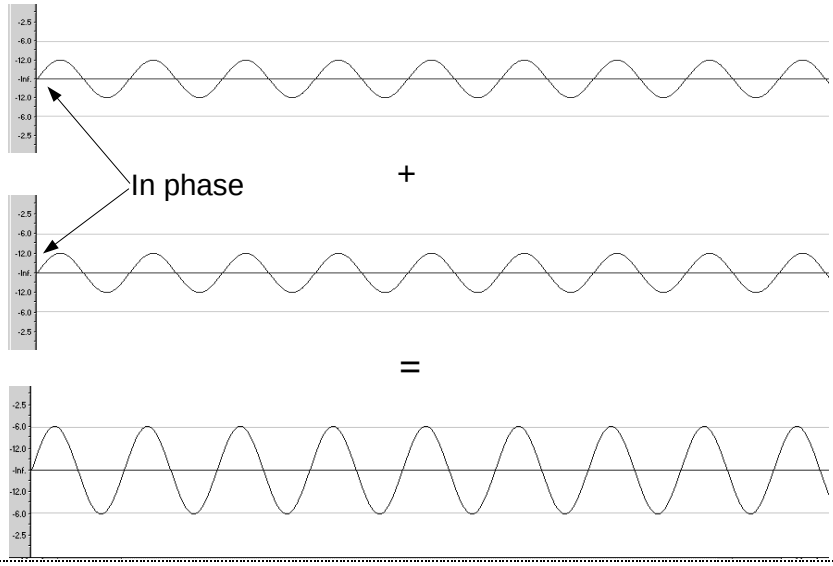


Phase

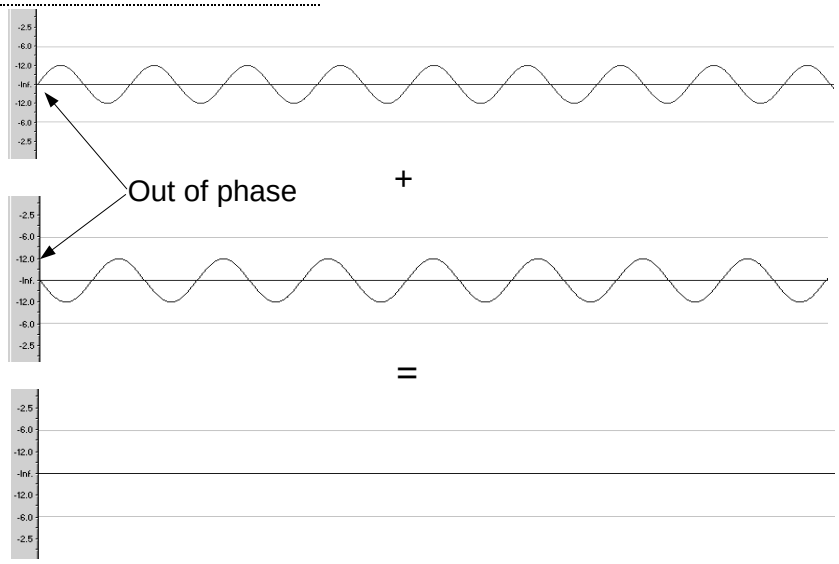


- **The Phase is the starting angle in a 360 degree cycle**
- **Acoustic is a linear phenomenon: the sound field created by several sources is the addition of the individual ones.**
- **The soundwave of two sounds being played at the same time is the addition of their soundwaves**

Phase: constructive interference



Phase: destructive interference



Sonic Waves Frequency Perception

- **We just saw the fundamentals of acoustics and how these are perceived.**
- **Psychoacoustics: The psychological study of human aural perception, i.e., the science of sound perception**
  - **Pitch: psychological perception of how low or high a given sound is (related to frequency)**
  - **Loudness: psychological perception of the volume, softness or loudness of a given sound (related to amplitude)**
  - **Timbre: the character or quality of a sound, i.e. its texture**

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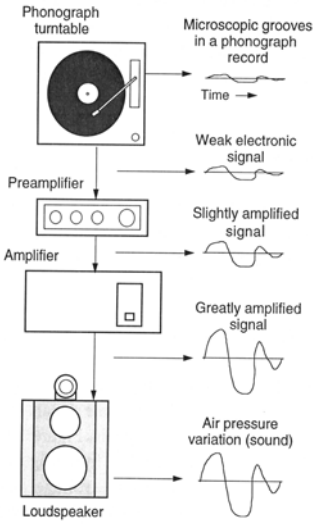
Representing sound

- **Sound is an analog phenomenon: that means the variations of pressure are continuous**

Without break or irregularities and dense

Between two levels of pressure, there is always an intermediate level.

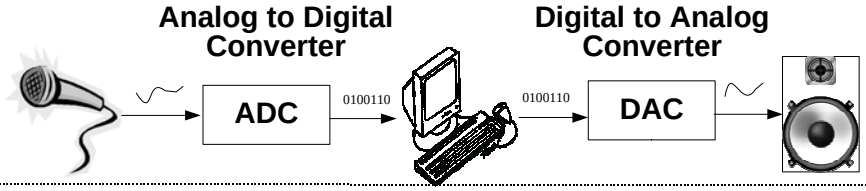


Representing sound

- **Computers are discrete machines using binary numbers.**

  Can only manipulate a finite or countable set of values

- **We need to digitalise sound, that is to reduce sound to a list of numbers (so that we can store and manipulate it with computers) and then reconstitute it.**

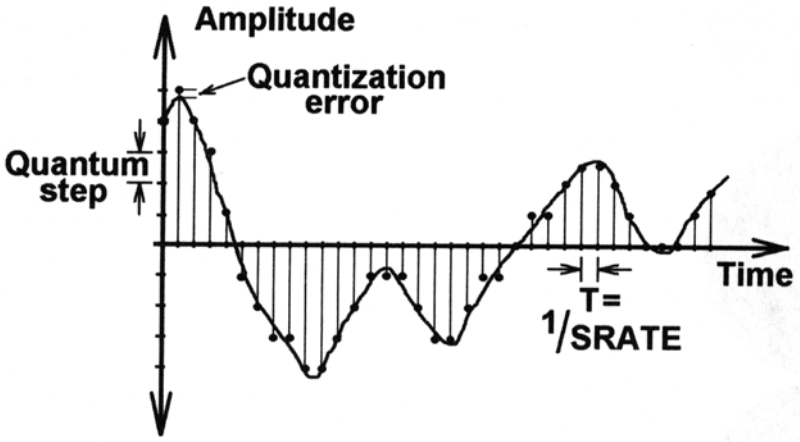


Digitalising sound

- **A sample is a numerical measure of the sound amplitude at a point in time.**
- **Sampling is the process of collecting the samples by taking snapshots of the sound wave (once transformed into electric signal)**
- **The sample rate is the frequency at which sampling is done. It is measured in Hz (number of samples per second).**
- **The bit depth is the format of the samples (expressed in the number of bits used to capture the range of possible amplitudes). Common bit depths are 8 bits, 16 bits, 24 bits, 32 bits.**

Digitalising sound

- **Sampling**



Digitalising sound

- **The bigger the sampling rate, the better the resolution. Where do we stop?**

2000Hz, 16bit

8000Hz, 16bit

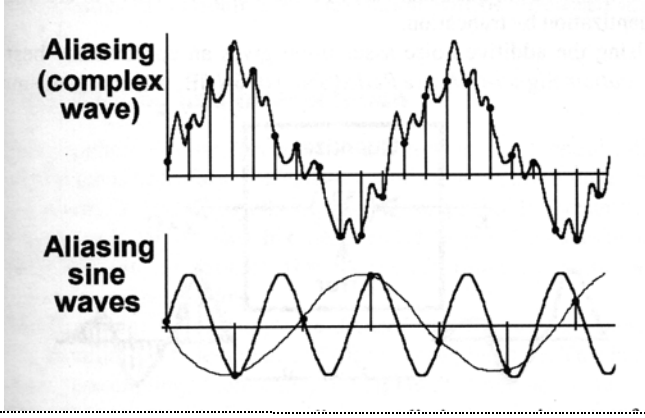
22050Hz, 16bit

44100Hz, 16bit

- **Nyquist showed that in order to be able to reproduce sound correctly, it is enough to sample at twice the highest frequency.**
- **Hence 40KHz, why 44.1Khz then?**

Digitalising sound

- The analog signal contains all the frequencies
- Those above the Nyquist frequency can create problems: aliasing (or foldover)



Digitalising sound

- Anti Aliasing low-pass filter at 20KHz (cutting frequency, but attenuated signal will go through up to 22050).
- If we double that number we do get 44.1KHz
- Similarly, the bit depth is very important

8000Hz, 8bit	44.1kHz, 8bit
8000Hz, 16bit	44.1kHz, 16bit

- 44.1Khz, 16bits forms the Redbook Standard
- Why then some equipment (solid state recorder, soundboards, DAT, ...) do more (which uses more memory)?

Digitalising sound

• Sampling rates commonly used:

- **8,000 Hz: telephone and encrypted walkie-talkie, wireless intercom and wireless microphone transmission; adequate for human speech but without sibilance; ess sounds like eff**
- **11,025 Hz: one quarter the sampling rate of audio CDs; used for lower-quality PCM, MPEG audio and for audio analysis of subwoofer bandpasses**
- **22,050 Hz: one half the sampling rate of audio CDs; used for lower-quality PCM and MPEG audio and for audio analysis of low frequency energy. Suitable for digitizing early 20th century audio formats such as 78s[7]**
- **32,000 Hz: miniDV digital video camcorder, video tapes with extra channels of audio (eg. DVCAM with 4 Channels of Audio), DAT (LP mode), Germany's Digitales Satellitenradio (German), NICAM digital audio, used alongside analogue television sound in some countries. High-quality digital wireless microphones.**
- **44,056 Hz: PCM adaptor using NTSC video tapes (245 lines by 3 samples by 59.94 frames per second), sometimes misused to play back audio streams sampled at 44,100 Hz (and vice versa)**
- **44,100 Hz: audio CD, also most commonly used with MPEG-1 audio (VCD, SVCD, MP3), adopted from the PCM adaptor using PAL video tapes (588 lines by 3 samples by 25 frames per second). Much pro audio gear uses (or is able to select) 44.1 kHz sampling, including mixers, EQs, compressors, reverb, crossovers, recording devices and CD-quality encrypted wireless microphones.**

Digitalising sound

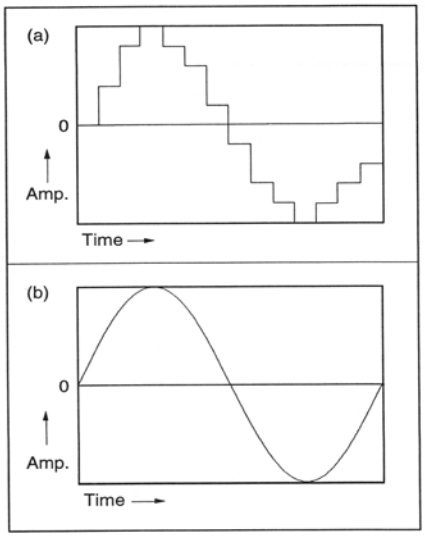
• Commonly used sample rates:

- **48,000 Hz: digital sound used for miniDV, digital TV, DVD, and films. Much pro audio gear uses (or is able to select) 48 kHz sampling, including mixers, EQs, compressors, reverb, crossovers and recording devices such as DAT.**
- **88,200 Hz: sampling rate used by professional recording equipment when the destination is CD (multiples of 44,100 Hz). Some pro audio gear uses (or is able to select) 88.2 kHz sampling, including mixers, EQs, compressors, reverb, crossovers and recording devices.**
- **96,000 Hz: DVD-Audio, some LPCM DVD tracks, BD-ROM (Blu-ray Disc) audio tracks, and HD DVD (High-Definition DVD) audio tracks. Some pro audio gear uses (or is able to select) 96 kHz sampling, including mixers, EQs, compressors, reverb, crossovers and recording devices.**
- **176,400 Hz: sampling rate used by professional recording equipment when the destination is CD (multiples of 44,100 Hz)**
- **192,000 Hz: DVD-Audio, some LPCM DVD tracks, BD-ROM (Blu-ray Disc) audio tracks, and HD DVD (High-Definition DVD) audio tracks, High-Definition audio recording devices and audio editing software**
- **2,822,400 Hz: SACD, 1-bit sigma-delta modulation process known as Direct Stream Digital, co-developed by Sony and Philips**

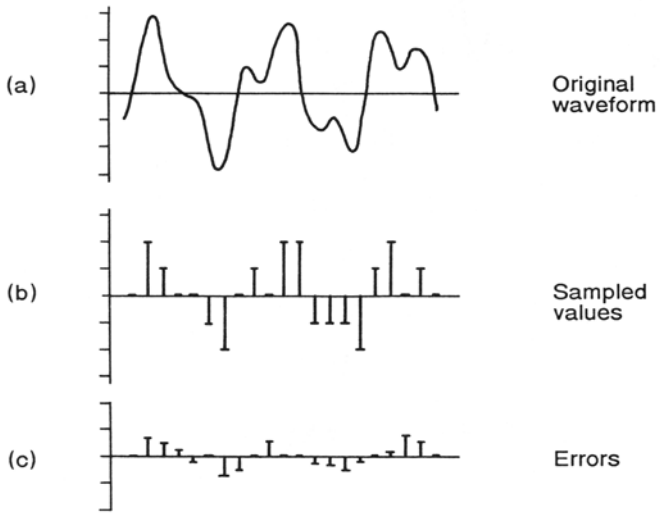
Digitalising sound: Quantisation

4-bit quantisation:
12 levels

16-bit quantisation:
65536 levels



Digitalising sound: Quantisation

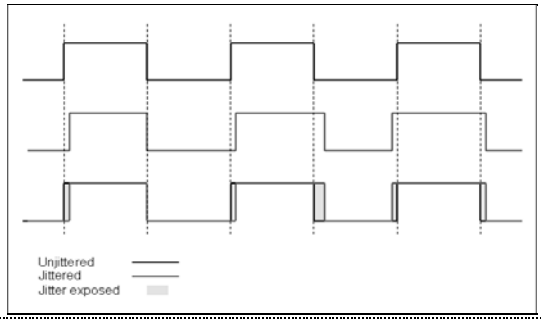


Digitalising sound: Quantisation

- The binary digits are approximations of the actual analog values (rounding to the nearest quantum or truncating): the amplitude difference between the original and quantized signal is lost forever.
- This the quantization error also called quantization noise
- It can be viewed as a noise signal upon reconstruction (uniform noise of ½ bit of amplitude).
- The quantization signal to noise ratio is roughly 6N dB where N is the number of bits used to represent the sound.

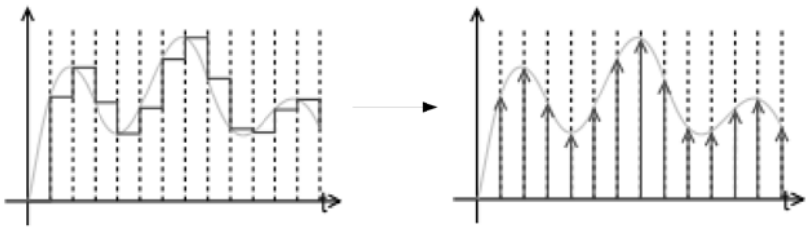
Digitalising sound

- This digital representation of an analog signal where the magnitude of the signal is sampled regularly at uniform interval is called: Pulse-code modulation (PCM)
- Another error that is more rare is jitter



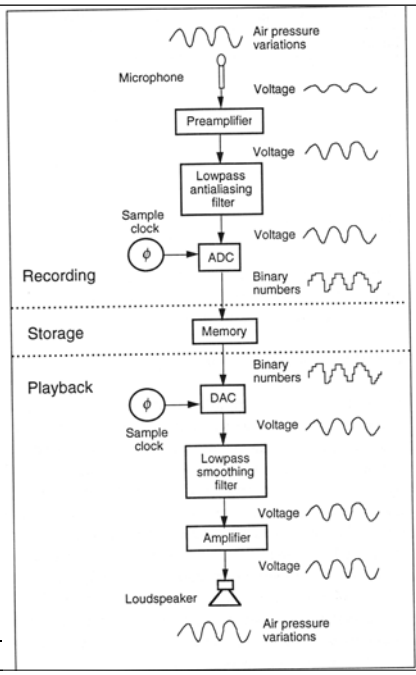
DAC: Digital to Analog Converter

- A digital to analog converter (DAC) is a device for converting a digital code (usually binary) to an analog signal (current, voltage or electric charge).
- Smoothing filters help to restitute a wave form that will sound like the original.



Digitalising sound

Overview of the audio digital recording and playback chain



Summary

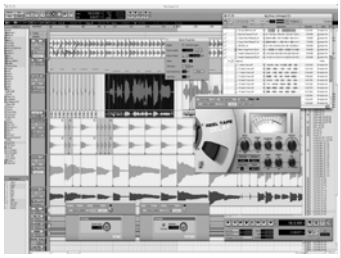
- **Today, we have seen most of the fundamentals of acoustics and digital audio processing:**
  - **Sound: its production, propagation and perception**
  - **Sonic waves representation and properties: waveform, frequency, amplitude, phase, envelope, ...**
  - **Sound digitalisation:**
    - **ADC and DAC**
    - **Sampling rate and bit depth**
    - **Aliasing, quantisation noise**

⇨ This technology continues to evolve

"Since light travels faster than sound, people appear bright until you hear them speak."

Chinese Proverb

Your lab this week



- **Andrew Hawryshkewich will drive you through the lab.**
- **We will start learning the basics of Pro Tools**
- **It is the standard sequencer**
- **Mac OS or Microsoft Windows**
- **Digidesign, a division of Avid Technology (line of products for editing of television shows, feature films, and commercials)**
- **Standard in the audio industries: music production, film scoring, television and post production.**
- **Pro Tools has three types of systems: HD, LE, and M-powered.**
- **We will use the M-powered system (Avid bought M-Audio in 2005)**

For next week

- **Readings:**
- Road, C. 'Chapter 1: Digital Audio Concepts'.*The Computer Music Tutorial.* US: MIT Press, 1996 ISBN: 0262680823, pp. 5 to 47, 43 of 1234 pages. Copyright: MIT Press / Access Copyright, 1996. **(if not already done)**
- Cancellaro, J. 'Chapter 2'. *Sound Design for Interactive Media.* US: Thomson Delmar Learning, 2005 ISBN: 1401881025, pp. 33 to 64, 32 of 258 pages. Copyright: Thomson Delmar Learning / Access Copyright, 2005
- Cancellaro, J. 'Chapter 3'. *Sound Design for Interactive Media.* US: Thomson Delmar Learning, 2005 ISBN: 1401881025, pp. 67 to 88, 22 of 258 pages. Copyright: Thomson Delmar Learning / Access Copyright, 2005
- That is a lot (97p) but that is to get you started and after there will be very few readings (sometime none).

"We have two ears and one mouth so that we can listen twice as much as we speak."

Epictetus (Greek philosopher)

Today's active listening session

- **Musique concrète (French; literally, "concrete music"), is a style of avant-garde music that relies on recorded sounds, including natural environmental sounds and other noises that are not inherently musical, to create music.**
- **Pierre Schaeffer (1910-1995) coined the term 'musique concrète' to describe a music made 'concretely' by working directly with sounds, as against music made 'abstractly' by working with symbols for sounds (as in a musical score).**
- **Other famous composers include:**
  - **Pierre Henry,**
  - **Luc Ferrari,**
  - **Pierre Boulez,**
  - **Karlheinz Stockhausen,**
  - **Edgar Varese**
  - **Iannis Xenakis**
  - **...**

Cinema for the ears

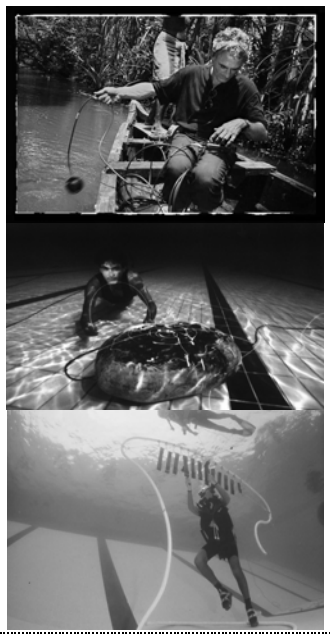
- **Pierre Shaeffer was a French composer:**
  - known to be the first composer to make music using magnetic tape
  - Etude aux Chemins de Fer (Railroad Study), composed by Pierre Schaeffer in 1948, is a recorded assemblage of steam engines, whistles, and other railroad sounds. It is the first recorded assemblage of sounds.
  - In 1951, he organized the Groupe de Recherche de Musique Concrète, the focus of which was working with tape recorders.
  - The photo shows Schaeffer in 1952 playing the phonogène à clavier, a tape recorder with its speed altered by playing any of twelve keys on a keyboard.

Cinema for the ears

- **Pierre Henry (1927-) started with Pierre Shaeffer**
- **Symphonie pour un homme seul (1950) (symphony for a lonely man)**
- **Variations pour une porte et un soupir (1963) (Variations for a door and a sigh)**
- **Pierre Henry worked for Maurice Bejart ballet.**
- **Psyche rock (1967): the piece is quite influential to pop culture as it became the theme song of the TV series Futurama (link to a video online)**

Cinema for the ears

- Iannis Xenakis (1922-2001):
  - was a Greek modernist composer and architect
  - He is known for using mathematical theories in composition
  - Diamorphose (1957)
- Michel Redolfi:
  - Desert Tracks is made of sounds recorded in the US deserts.
  - Mojave desert (1987)
  - You can ear insects, train, ...



Cinema for the ears

- Some are taking a more "naturalist" or "purist" approach
- Audio postal card, by Marc Pichelin and Jean Pallandre:
  - Playground (1995)
  - Fougere (1995)
- La Creuse, by Eric La Casa and Cedric Peyronnet:
  - Le bois de Parnac (2006)
  - Le confolent (2006)