Opportunities via IATL

International Conference of Undergraduate Research (ICUR)
The call for abstracts is now open for the fourth annual International Conference of Undergraduate Research (ICUR), which will be held on 27-28 September 2016. Please visit www.icurportal.com to submit your 250 word abstract and for more information. The deadline to apply is *Tuesday 15 May 2016.*

IATL student funding opportunities - deadline approaching
The next IATL student funding deadline is fast approaching; don't miss out! Make sure your application is submitted by midnight on *Wednesday 16 March 2016*. Or if you are a member of staff at the University please encourage your students to find out more about the funding opportunities that IATL offer. Further information and an application form can be found here.

Session 8

Electronic Music

Gavin Bell

Science of Music

Electronic music

Musical sound can be generated electronically.
How?

Is recorded music always “electronic”?

Is live, acoustic music never “electronic”?

Discuss!

Musical sound generation 1

https://www.youtube.com/watch?v=uSm5IQFaTZA

https://www.youtube.com/watch?v=w5qf9O6c20o

Musical sound generation 2

Recorded music

Edison 'Home' model B phonograph, 1906.

Images and sound file from www.tinfoil.com

Electronic music

Musical sound can be generated electronically.

Analogue or digital sound creation. Analogue could be purely electrical, or electro-mechanical / electro-acoustic.

Is recorded music always "electronic"?

For all practical purposes, yes, but again could be *digital* or *analogue*.

Is live, acoustic music never "electronic"?

Sound reinforcement very often used (e.g. module concert in the Arts Centre Studio Theatre: acoustic instruments, all amplified).

Measured Tones

- Chapter 6
  - A slightly random history of the physics of electromagnetism linked somewhat tenuously to the classical → romantic shift.
  - Followed by discussion on acoustic impedance.
  - Then a useful section on air pipes (cf. *acoustic* instruments session).
- Interlude 7
  - Nice overview of electronic instrument development

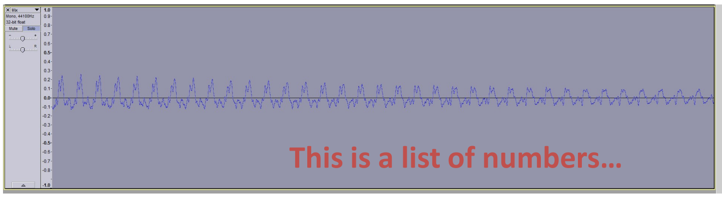
Today's session

- Digital and analogue
- Pioneers
- Electronic instruments
  - Theremin – demo
  - Analogue synthesis – demo
  - Digital synthesis
  - Types of synthesis
    - Additive, subtractive, physical modelling, FM…
- Not covering many possible topics!
  - Electro-acoustic / electro-mechanical instruments
  - Recording / "studio-as-an-instrument"
  - Etc.
- Views of the role of electronics in music

"Digital" and "analogue"

Analogue signal $A(t)$
t = time, A = amplitude

Digital signal… near end of session 7 clip in Audacity



Sampling 1

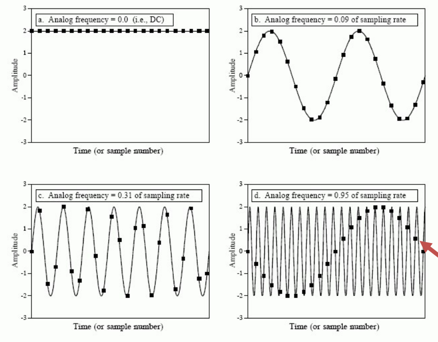


FIGURE 3-3
Illustration of proper and improper sampling. A continuous signal is sampled *properly* if the samples contain all the information needed to recreate the original waveform. Figures (a), (b), and (c) illustrate *proper sampling* of three sinusoidal waves. This is certainly not obvious, since the samples in (c) do not even appear to capture the shape of the waveform. Nevertheless, each of these continuous signals forms a unique one-to-one pair with its pattern of samples. This guarantees that reconstruction can take place. In (d), the frequency of the analog sine wave is greater than the Nyquist frequency (one-half of the sampling rate). This results in *aliasing*, where the frequency of the sampled data is different from the frequency of the continuous signal. Since aliasing has corrupted the information, the original signal cannot be reconstructed from the samples.

A-C
"Proper" sampling
Dots = list of numbers
Regular intervals
Interval = (sample rate)$^{-1}$

D
"Improper" sampling

Sampled signal has a different frequency: aliasing.

From S.W. Smith, The Scientist and Engineer's Guide to Digital Signal Processing

Sampling 2

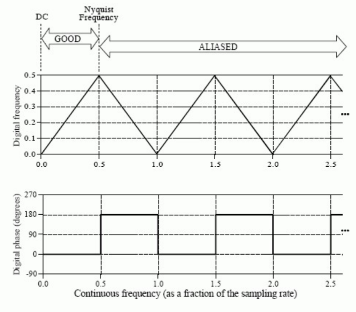


FIGURE 3-4
Conversion of analog frequency into digital frequency during sampling. Continuous signals with a frequency less than one-half of the sampling rate are directly converted into the corresponding digital frequency. Above one-half of the sampling rate, aliasing takes place, resulting in the frequency being misrepresented in the digital data. Aliasing always changes a higher frequency into a lower frequency between 0 and 0.5. In addition, aliasing may also change the phase of the signal by 180 degrees.

Nyquist frequency f_N

Normally (not always) defined as half the sample rate.

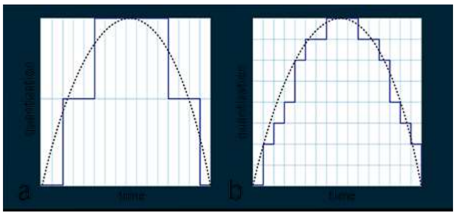
Proper sampling (usually) achieved for frequencies below f_N.

Above the Nyquist frequency sampling introduces aliasing and possible phase inversion. Yuk!

CD sample rate: 44.1 kHz
DAWs often use 48 kHz or 96 kHz.
Upper limit of human hearing about 20 kHz.

Sampling 3

Commodore 64 – classic “8-bit” sound
SID chip: “sound interface device”



Bit depth

Each amplitude level sampled must be given a number.

IEEE double precision format: uses 64 bits per number, 53 “significand” bits

16 bits: 2^{16} = 65536 possible values
24 bits: 2^{24} = 16777216 possible values
CD: 16 bit   Most DAWs: 24 bit

Summary: for modern digital audio, do not worry about sampling.

Pioneers

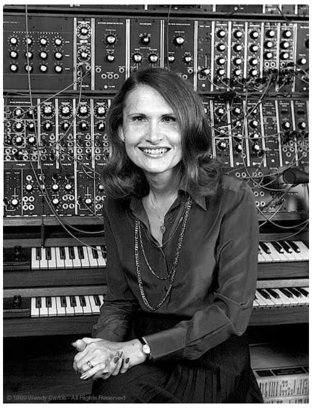
冨田 勲

Isao Tomita

Delia Derbyshire

Pioneers

Terry Riley



Wendy Carlos

Oscillators – remember these?

- *Deformation* leads to a *restoring force*
- *Restoring force* plus *inertia* leads to an *oscillation*

Theremin

Léon Theremin

Physicist, worked with Ioffe and Pavlov.

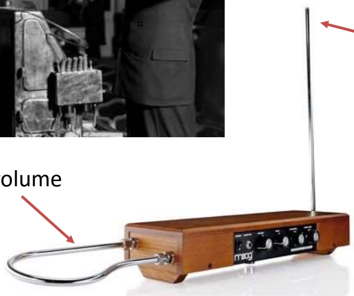
Using RF oscillators to measure dielectric constants ε of gases.

Electrical capacitance C is affected by ε, and oscillator frequency (pitch) is affected by C.

Objects near antennae detected by changing capacitance around an antenna. Changes frequency of oscillator.

Theremin uses oscillators ~500 kHz with frequency divider circuits (*heterodyning*) to bring tones below 20 kHz.

volume



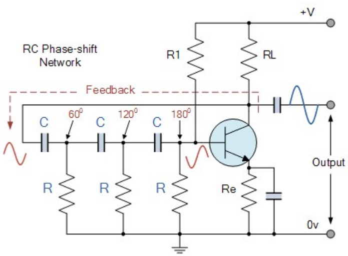
Electronic oscillators

- *Inductor (L)* and *capacitor (C)*

$$f = \frac{1}{2\pi\sqrt{LC}}$$

- Resistors (R) and capacitors

$$f = \frac{1}{2\pi RC\sqrt{2N}}$$



N stages in phase-shift network

Quite easy to build electronic analogue oscillators… BUT
Their resonant frequency depends on temperature.
Warm up → go out of tune!

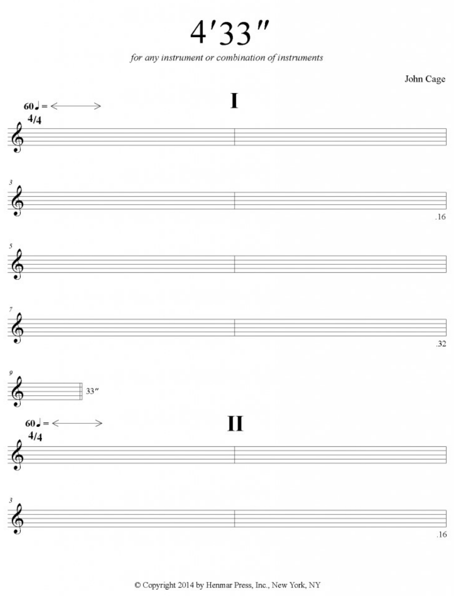
Hammond organ – mechanical rotator provides master frequency control.
Magnet and coil, metallic "tone wheel" – similar to electric guitar.
Use heterodyning to divide tone wheel pitches over several octaves.

Need more than oscillators…

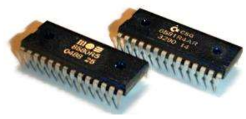
Recall simple tones built in MATLAB back in session 2 – not very musical! How to get timbre and dynamics?

John Cage: Pitch, Duration, Loudness, Timbre, "Morphology"

Real sounds are very complex – harmonic content varies in time. *Combination of time domain and frequency domain description.*



C64 SID chip



Partly digital, partly analogue.
8-bit digital.

- 3 oscillators
- Each oscillator can make sawtooth, square, triangle or noise waveforms.
- Each oscillator has an ADSR envelope.
- One filter (high-pass, low-pass, band-pass, combo)
- 3 ring modulators

Recall: ADSR = attack-decay-sustain-release volume envelope

Additive synthesis

- Add sine waves with time-dependent amplitudes.
- Build complex timbres from these harmonics (or inharmonics).
- Implement digitally or analogue.

- Hammond Organ effectively an additive synth.
- Less common than subtractive synthesis in development of synthesised sounds.

Can modulate the *frequency* of waveforms: “FM synthesis”.

Subtractive synthesis

- Take a starting harmonically-rich waveform, e.g. sawtooth.
- Apply a filter to remove or attenuate some harmonics.
- Filter can vary in time (e.g. driven by low-frequency oscillator, LFO)

Examples: many early analogue synthesisers (e.g. Moog, Korg, ARP), now commonly implemented digitally.

Natural example: human voice!
- Vocal folds produce harmonically rich sound.
- Mouth and throat filter the sound (choice of formants).

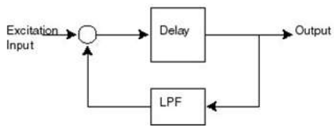
Physical modelling synthesis

- Try to make a mathematical model of sound waves in an actual instrument.
- Implemented digitally.
- 1D example: oscillating air column in organ or flute.
- 2D example: vibrating drum skin.

Use the physical properties of the medium to model how sound waves behave.
Can be computationally complex though there are efficient approximations.

Digital waveguide synthesis

- Standing wave is sum of “forwards” and “backwards” travelling waves, e.g. on a string.
- Sum a wave and a delayed copy to get the standing wave at a particular point and time.
- Each time the travelling wave hits the end point of the string, some energy is lost.
- This loss is e.g. frequency –dependent, so use a **filter**.
- Can include complicated effects such as non-linearity.



Example for a string from
http://www.cim.mcgill.ca/~clark/nordmodularbook/nm_physical.html

Decoding scientific titles

“Towards complete physical modelling synthesis of performance characteristics in the violin”

This paper discusses the need for total performance capabilities of a playable physically modeled violin by means of precise left-hand finger interaction within the system. The paper describes the design of a violin physical model, in which the computational need for adjacent string fingering performance of playing is minimized. The main focus of this paper is on the principles of developing a convincing physical model for the complete tonal range of sound synthesis of the violin. The realistic tonal beaviour of modelled strings compared to adjacent strings of a real instrument, and multiple-stopping performance techniques (e.g. when a 4-pitch chord is played using "quadruple-stop"). This enables the interaction of the string with the left-hand fingering. The artistic performance using physical modelling synthesis is shown to be feasible by using only two string models. The system also considers symphatetic coupling between different parts of modelled instruments in connection to expressive performance of the violin.

Cynical view: “Towards” = ABSOLUTELY NOWHERE NEAR

Your views

Musicians playing traditional instruments still have jobs. Why?

Does "electronic music" mix well with acoustic instruments or do you consider it a separate genre?

Discuss!

A personal view

- Synthesis and electronic music are excellent tools.
  - Combining traditional and electronic sounds is fun.
- Ubiquitous, easy recording is an excellent tool.
  - Digital recording is best.
  - I used to use cassette 4-track – eternal battle against tape hiss.
  - "Analogue warmth" and so on is a bit of a myth.
  - I can't do music purely on the computer – needs instruments.
- It is just easier (for me) to pick up a guitar and try to play something expressive than do it electronically.
  - Cables are a big old faff!
  - Software is a big old faff!
  - Guitars just work!
- I like electronic drums – acoustic drum kits are the biggest faff of all.