Summary for Written Test 1 **Sounds & Circuits** 2020 v1.0

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Creative Technology – M2: Smart Environments.

According to test information for Test 1 "Sounds".

Information taken from lectures, assignments, references and book.

Lecture 1

What is a signal? A physical quantity that: varies in time, carries information.

Examples: ECG (Electro Cardo Gram), Sunrise and Sunset, Hormone levels.

What is a sound? A mechanical wave (needs a medium/particles).

A pressure wave (longitudinal).

Speed of a sound Depends on temperature and the medium.

(In vacuum there will be no sound as there is no medium)

Electric signals Can be very fast.

Well suited for representing variations.

Easy and cheap to transduce from and to the electrical domain.

Transmission and realization of functions is easy and cheap.

Domains The world itself is one entity, a whole. But it is too big for us to gasp.

Domains divide this entity in different fields of specialists and disciplines.

Thermal	Chemical	Optical
	Mechanical	Acoustic
	Electrical	Others
	Fluidic	

Energy Energy is present in all domains and connects them.

Effects of energy can only be observed when it is being transferred from one place to another or from one form (domain) to another.

It cannot be destroyed or created; the total amount of energy never changes.

Transducers	Equipment that can transfer energy between domains. In every transfer energy is “lost” in heat.
Signals and math	Can be: Deterministic - predictable, describable via mathematical function Stochastic - unpredictable (at best statistical parameters), random
Periodic signals	Exactly repeats itself after a certain period of time. $s(t) = s(t+T)$ $T[s]$ is the period time of the signal. Periodic signals do not need to look “smooth” such as the sine wave.
The sine wave	$f(t) = A\sin\left(\frac{2\pi}{T}t\right) = A\sin(2ft)$ A [‘a physical unit’] is the amplitude (measure of magnitude) T [s] is the period f [Hertz or Hz = 1/s] is the frequency. The number of repetitions a second. $f = \frac{1}{T}$ $f = 440\ Hz \rightarrow T = 0.0023s$
Wall sockets	A is 325V in this case. f is 50 Hz so $T = \frac{1}{50} = 20\ ms$
Period and frequency	The longer the time period, the lower the frequency. The smaller the time period, the higher the frequency.
Fundamental Frequency	Describes the repetition rate of any periodic signal.
Pitch	The perceived frequency of a sound $\leftrightarrow$ The fundamental frequency
Tuning	Is done according to a certain standard, for example Western tuning, and make sure that all notes have the same fundamental frequency at every note and with that have the same pitch. Note A4 (440 Hz) is often used as a base for tuning (equal temperament).

Signal Manipulations

Shifting

Vertical (over y-axis)

$y = f(x) + k$

Has no frequency and will only add an offset.

Horizontal (over x-axis)

$y = f(x + h)$

$h > 0$: shift to left (earlier in time)

$h < 0$: shift to the right (later in time)

Called a phase shift, not audible.

Scaling:

Vertical (over y-axis)

$y = c \cdot f(x)$

$c > 1$: vertical stretching by factor of c

$c < 1$: vertical compressing by factor of c

Changes amplitude (audio → volume).

Horizontal (over x-axis)

$y = f(c \cdot x)$

$c > 1$: horizontal compressing by factor of c

Period becomes smaller, freq. increases.

$c < 1$: horizontal stretching by factor of c

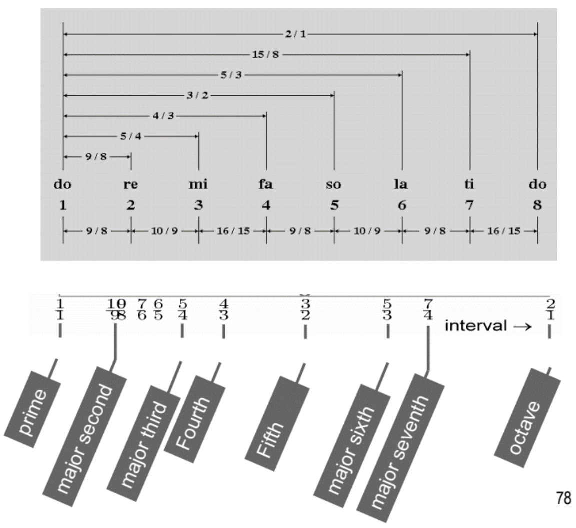
Period become larger, freq. decreases.

Frequency Intervals

If a sound consists of two frequencies, with f_0 and f_1 where $f_1 > f_0$

Then the interval is defined as: $\frac{f_1}{f_0}$

Name of intervals



Octave $f_1 = 2f_0$

Frequency ratio of 2:1

$f_1 = 2f_0$ is the first harmonic of f_0

Higher harmonics of f_1 are higher harmonics of f_0

$2f_0$ and f_0 are tones with large similarities, therefore they both have the note of f_0

Perfect Fifth $f_1 = 1.5f_0$

Frequency ratio of 3:2

Often sounds very harmonious.

Equal temperament Used in Western tuning standard, scales according to proportional tuning.

$f_n = f_0 2^{\left(\frac{n}{12}\right)}$ n is the number of semitones up

This makes that the same frequencies can be used for different scales (sext, quints, quarte, terts).

This sounds better than just intonation which uses natural steps.

Lecture 2

Music in real life Musicians need info on which note is played, or which notes played simultaneously (chords) and the duration of the notes.

Standards Universally applicable, can be learned by anyone, medium allows for easy reproduction, storage and transport etc. Essential for music.

Digital domain Human needs to be synchronised in order to play correctly.

For digital music this is no different, they too use a standard (MIDI).

Electronic format is easy to store, reproduce, transmit and distribute.

MIDI Musical Instrument Digital Interface (1983)

Addresses devices, tells them which note to play and for how long.

Many other parameters can be passed as well (loudness, pitch shift etc).

Playing notes Each note played on an instrument consists of a fundamental frequency (aka pitch) and higher harmonics (number varies)

Sinusoids

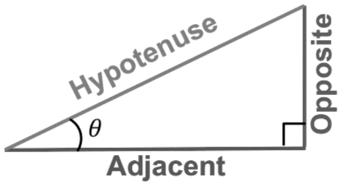
We use sinusoids as it is a well know function. $f(t) = Asin(2\pi f t)$

Many sounds and processes in nature can be described by them.

Sine functions with different frequencies can be used to construct *any* other periodic signal.

In other words: any period signal can be described as a sum of sine and cosine functions with each term having its own frequency.

Basic definition:

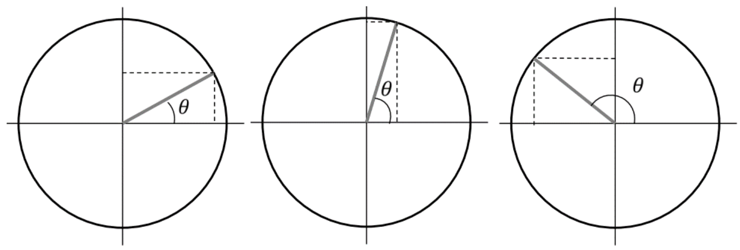


$$\text{Sin } \theta = \frac{\text{Opposite}}{\text{Hypotenuse}}$$

$$\text{Cos } \theta = \frac{\text{Adjacent}}{\text{Hypotenuse}}$$

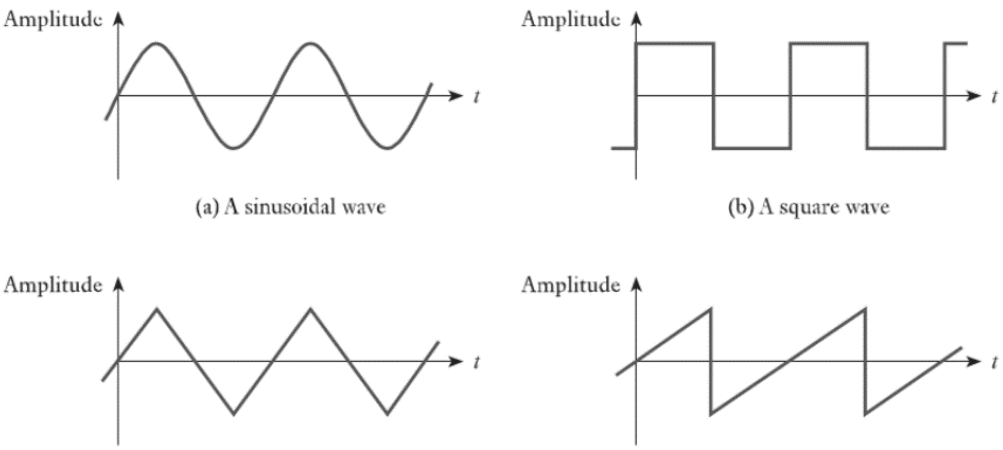
$$\text{Tan } \theta = \frac{\text{Opposite}}{\text{Adjacent}}$$

By using the unit circle (r=1) you can map the triangle to a coordinate system.



This links with periodic functions as they too repeat itself after one period T, in the circle this is achieved by a full rotation of $360°$ or 2π radians.

Basic periodic signals



(a) A sinusoidal wave (b) A square wave

(c) A triangular wave (d) A sawtooth wave

Timbre

What makes up the timbre of a note on different instruments?

It is fundamental frequency and the harmonics (which ones and their individual strength or amplitudes)

Fourier series

$s(t) = a_0 + \sum_{n=1}^{\infty} a_n \cos(2\pi n f_1 t) + \sum_{n=1}^{\infty} b_n \sin(2\pi n f_1 t)$

a_0	DC component (offset).	
$f_0 = \frac{1}{T}$	Fundamental frequency = pitch.	
$a_1 \cos(2\pi f_0 t) + b_1 \cos(2\pi f_0 t)$	First "harmonic" (fundamental)	
$a_n \cos(2\pi n f_0 t) + b_n \cos(2\pi n f_0 t)$	$n > 1$	Higher harmonics.
$n f_0$	$n > 1$	nth harmonic.

N-additions to the series are called frequency components of $s(t)$

Harmonics

n	Frequency	Harmonic	Overtone
1	f_1	1st harmonic (= fundamental frequency = pitch)	-
2	$2f_1$	2nd harmonic	1st overtone
3	$3f_1$	3rd harmonic	2nd overtone
4	$4f_1$	4th harmonic	3rd overtone
...			
n	nf_1	nth harmonic	(n-1)th overtone

All higher harmonics are INTEGER multiples of the fundamental frequency.

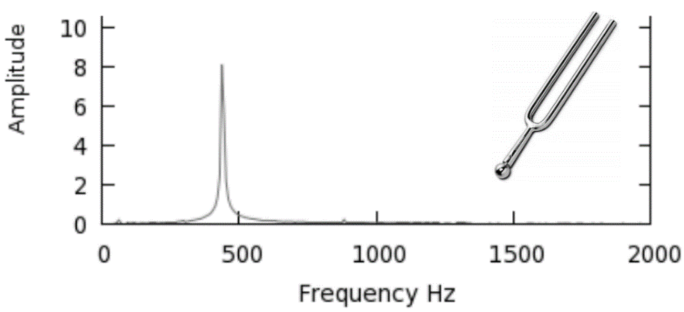
Analysis

<u>Frequency spectrum</u>:

Plots of amplitudes (M_n) of each frequency contribution vs frequency.

Analysis method in the digital domain is called: FTT (Fast Fourier Transform).

Spectrum of tuning fork ($f_0 = 440\ Hz$):



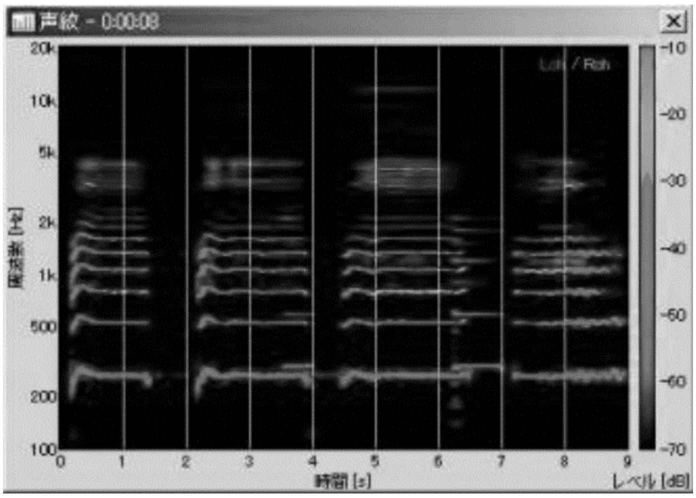
Spectrogram:

A frequency spectrum is a momentary recording (static). In case of a continuous spectral analysis, we use a spectrogram (which is dynamic).

Time [s] = x-axis

Frequency [Hz] = y-axis

Amplitude = colour of the f values



Usages of sines

We use sines to create or replicate sounds. The more accurate the approach should be the more harmonics are needed.

Chord

Multiple notes played at the same time. Mathematically you add all the different notes together. The same can be done by adding sines/cosines to each other.

Beating

Beats are cause by the interference of two wave at the same point in space.

At one spot there will be constructive interference (increasing the overall amplitude) and in other places destructive interference which will lower, almost mute, the amplitude.

This cause the sound to be alternatively be soft and loud.

$f_{beat} = | f_1 - f_2 |$

Lecture 3

Other Fourier form

$s(t) = a_0 + \sum_{n=1}^{\infty} M_n \cos(2\pi n f_1 t) + \varphi_n$

M_n is the real, physical amplitude of each component n

φ_n is a phase angle for each n

Sound synthesis

For sound synthesis of a signal $s(t)$, we need:

To create a note $p(t)$, with a fundamental frequency and proper mix of higher harmonics.

Describe its envelope $e(t)$

The total sound will then be $s(t) = e(t)p(t)$

If needed/desired, add sound effect such as filtering or an LFO.

<u>Sample based</u>:

Use sampled notes of real instruments as basic signal.

Known as wavetable/waveform synthesis.

Only a handful of notes are stored, the rest is created by pitch shifting (horizontal scaling).

$s_{new}(t) = s_{old}(c \cdot t) = s_{old}\left(\frac{T_{old}}{T_{new}} \cdot t\right)$

This method requires low processing power and low storage capacity, but it does not store all the details in the sound.

<u>Additive synthesis</u>:

Use elementary sine waves as basis signals. Also known as Fourier synthesis.

Any periodic signal can be approximated by a sum of sine waves.

$s(t) = a_0 + \sum_{n=1}^{\infty} a_n \cos(2\pi n f_1 t) + \sum_{n=1}^{\infty} b_n \sin(2\pi n f_1 t)$

f_1 = fundamental frequency

The other frequencies are multiples (higher harmonics) of the fundamental frequency.

Amplitudes a_n and b_n determine the signal shape.

This works for instrumental sounds too if we know the spectrum (which harmonics do we need to add).

<u>Physical modelling synthesis</u>:

Waveform to be generated is computed by using a mathematical model.

Based on simplified laws of physics that govern sound production.

This method can give extremely accurate sounds but requires very precise modelling. It also costs quite a lot of processing power but that is not much of an issue anymore.

Subtractive synthesis:

With this method parts of an audio signal (often one that has a lot of harmonics) are filtered to change the timbre of the sound and by that creating a new one.

Additional filters or synthesis methods can be added to alter the sound.

A common input is noise. For example:

White noise	Contains all frequencies equally strong
Pink noise	More balanced for our ears, still has all frequencies but drops off at higher ones.

Advantages with this method is that is much easier then additive (Fourier) synthesis, it can create a wide range of sounds, input waves are widely available and it can easily be linked to other (after) effects.

A downside is that it requires proper handling and precise usage to get the sound right with the filters.

Envelopes

In the real world, a sound decays over time. Usually, a signal does not.

By describing the amplitude as a function of time we can add an envelope.

$s(\mathrm{t}) = Asin(2\pi ft) \rightarrow s(t) = A(t)\sin(2\pi ft)$

In general: $s(t) = e(t) \cdot p(t)$

$p(t)$ is the periodic function

$e(t)$ is the envelope function

Envelopes differ significantly between instruments.

The sustain of an instrument: how long does a note keep ringing.

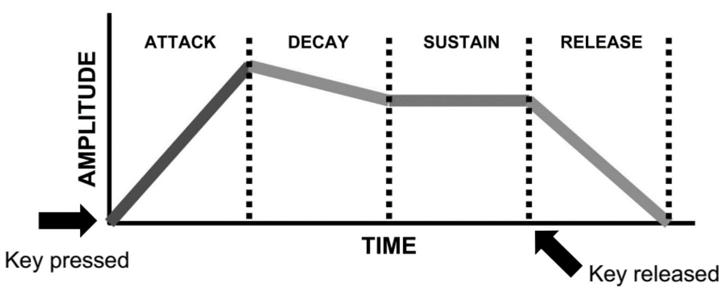
Are often described by exponential functions:

$e(t) = 10^{-at}$ or $e(t) = e^{-at}$

$a[s^{-1}]$ describes how fast or slow a process goes.

ADSR

To better match real life, we use 4 stages in envelopes:



A, D and R describe rising and decaying of signal and time is a key parameter.

S has two important parameters: constant amplitude (volume) and time.

The ADSR of many synthesisers can be tuned to match instruments or create new sounds.

Filtering:

Filter on specific frequencies, the filter has a certain frequency response.

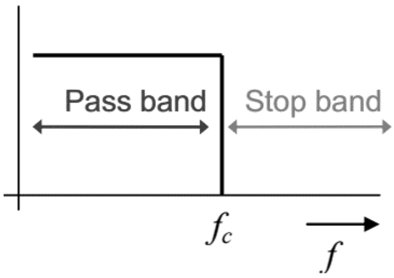
For a certain level of frequencies, it will block and others will pass.

We will only consider ideal filters, which stop or pass with abrupt transitions.

In reality these filters do not have abrupt transitions, there is a soft one which can be described as a fade to zero after reaching the cut off frequency (f_c) .

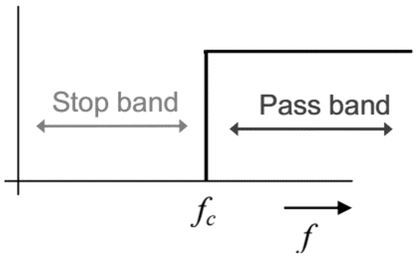
LPF, low pass filter (ideal):

Lower frequencies that fall in the pass band will be passed. Those in the stop band will be filtered out.



HPF, high pass filter (ideal):

Higher frequencies that fall in the pass band will be passed. Lower frequencies in the stop band will be filtered out.



Vibrato:

Audio effect consiting of a regular, pulsating change of pitch. Achieved with frequency modulation. Characterised by the amount and speed of the effect.

Tremelo:

Audio effect constising of a variations in amplitude. Achieved with amplitude modulation.

LFO, low frequency oscillators:

An electronic signal, usually below 20 Hz, used to create rhythmic pulses or sweeps.

Can be used to modulate other effects. Parameters to control are the waveform, the rate, the intensity and the "target" (the element it will modulate) of the LFO.