

Electronic musical instruments

SAMPLING

Sampling

- **Sampling** generates musical sounds in EMIs by playing back recorded sounds - **samples**.
- The recorded sounds are processed with simple operations: transposition, looping, filtering, adding envelope, adding sound effects.
- With sampling, we can turn any sound source into a musical instrument.
- It's not a sound synthesis – the sound is not created from scratch, we start with an already existing sound.

Sampler

Sampler – an EMI that is able to:

- create instruments from sound samples,
- play back the samples, allowing a musician to play on it and use it as a musical instrument.

Sampler types:

- hardware (with keyboard or as a sound module),
- software (virtual).

Usually, we use computer software to create instruments from sound samples.

Sound samples

Sound sample:

- a digital signal – a digitized sound,
- a sample (an example) of sounds that a given source can produce (a sample of many possibilities), e.g. a musical instrument played with a specific articulation.

Do not confuse a sound sample (a sound recording) with a signal sample (a single value of a digital signal).

Creating sound samples

We can create instruments from samples that we recorded, or we can use existing instrument banks.

Recorded samples have to be pre-processed in a sound editor:

- trimming (silence and unwanted parts removal),
- loudness normalization (important!),
- adding sound effects,
- it is useful to determine the pitch.

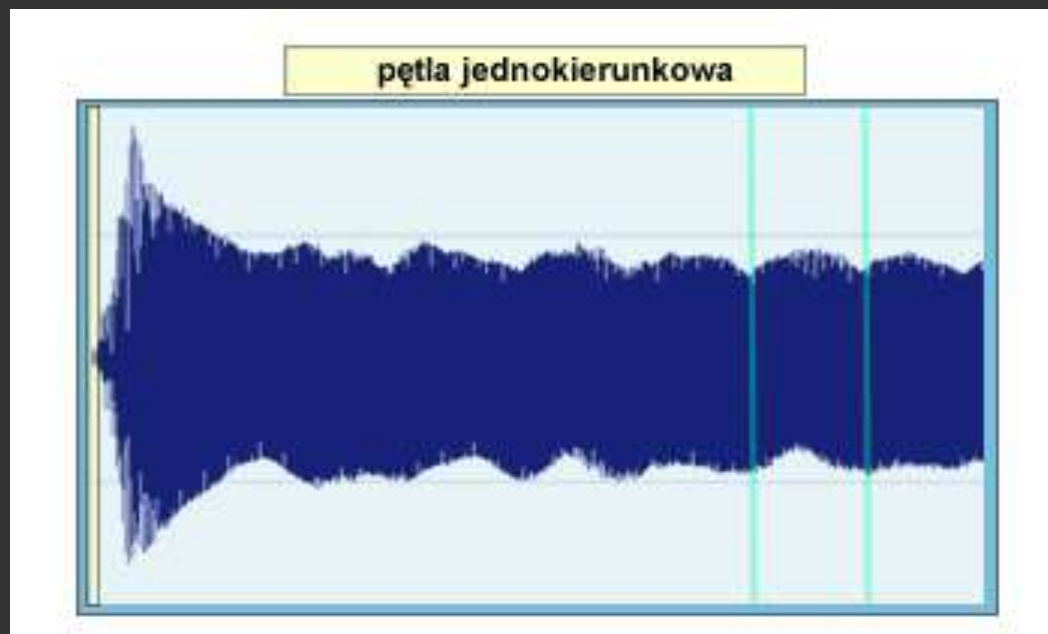
It is worth to record several samples of the same sound and select the best sounding one.

Looping

- We can play back the samples:
 - once, from the beginning to the end (e.g. percussive),
 - looped (repeating).
- The problem: we want the sound to continue playing until we release the key. But the recorded sample has a defined duration.
- Sample looping allows for repeating a selected sample part and to lengthen the sound as much as needed.
- We don't have to loop the whole sample, we can select a looped fragment, e.g. a sustain phase of a sound.

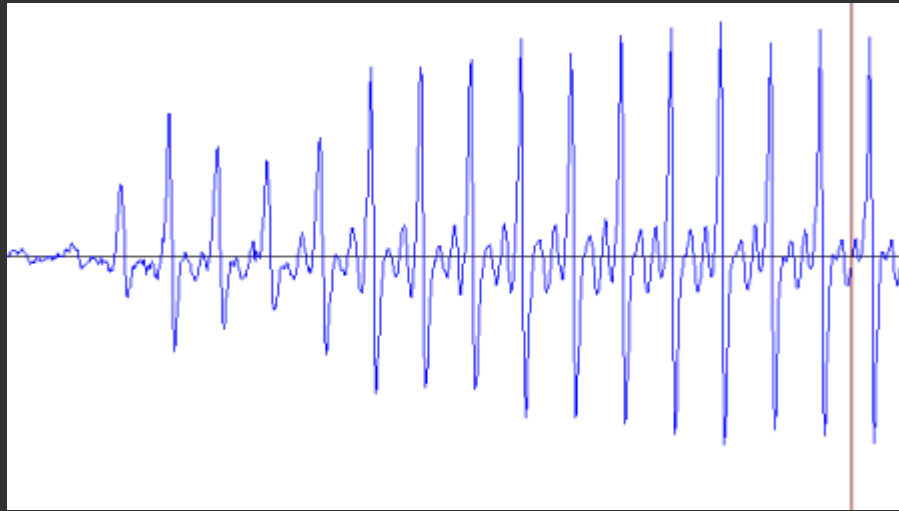
Looping

- In many cases, looping the whole sound is not what we need.
- We let the attack phase play once.
- We usually loop a single “pseudo period” of a sound, near the beginning of the sustain phase.

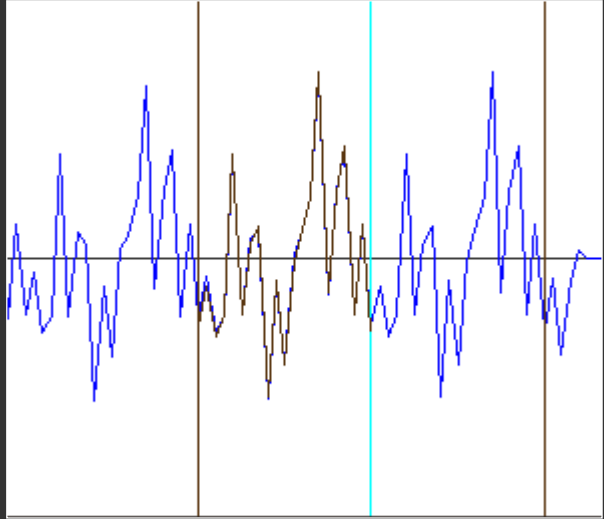
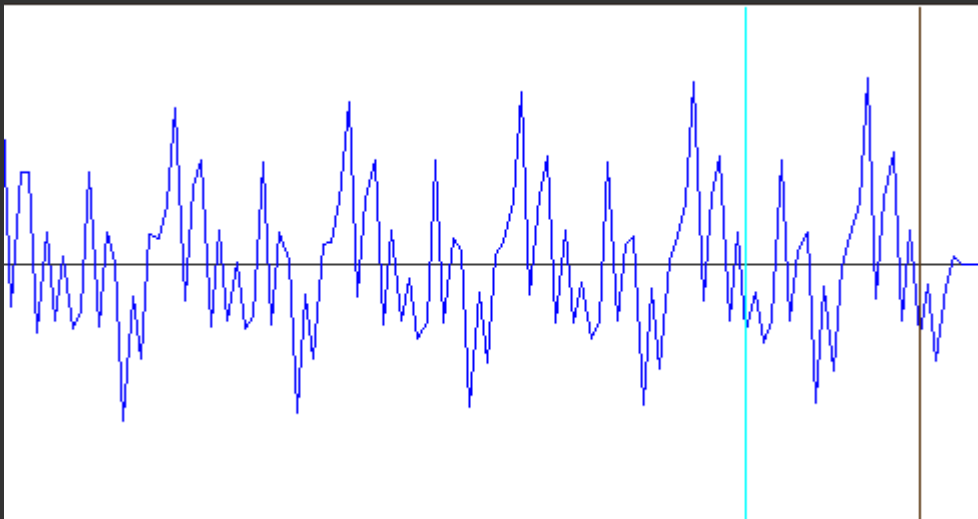
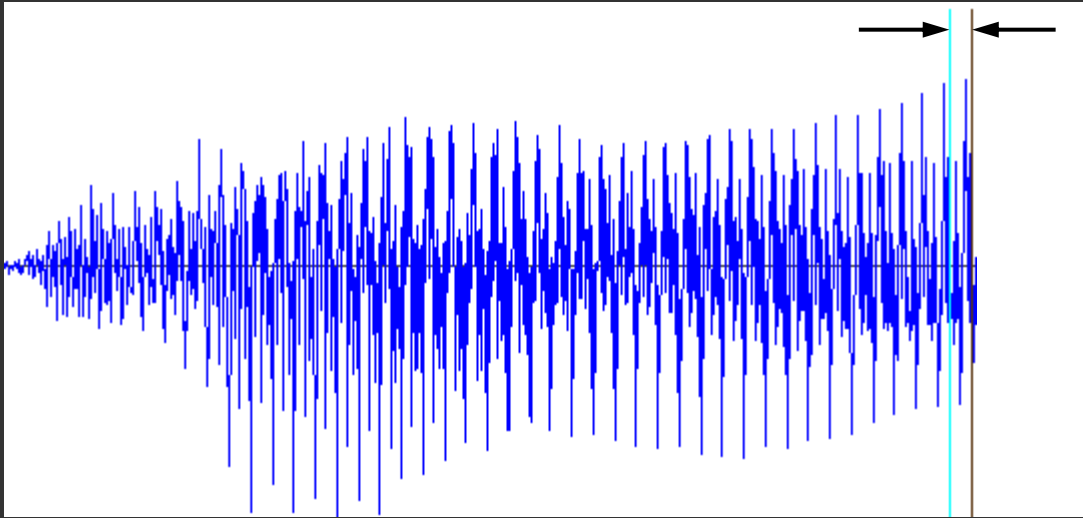


Looping

- Usually, we set the beginning and the end of the loop at the zero crossing, to avoid amplitude jumps.
- We try to select a looped fragment that is identical to a fragment before the loop (software helps with that by showing an overlapped fragment on a time plot).
- We can remove the sound part after the loop. In the release phase, we continue to play the loop, with added envelope that fades out the sound.



Looping examples



Root key

- A sample is described by its fundamental frequency:
 - we need to determine that frequency with a sound analyzer (e.g. SPEAR)
 - samplers often can determine the frequency automatically, but sometimes they fail, we have to verify the result.
- If we know the fundamental frequency, we can assign the sample to a key number that plays that frequency - this is a **root key**.

Sample transposition

- We determine a range of keys that play the sample.
- We want each key to play different pitch.
- A sampler automatically transposes the sample (changes the pitch) within a defined range.
- Transposition is done by **resampling**.
- Resampling causes **temporal distortions**:
 - the sound becomes shorter when transposed up,
 - the sound becomes longer when transposed down.
- It's not a big problem for looped sample sections, but sample parts played only once (e.g. the attack) become distorted.

Sample transposition

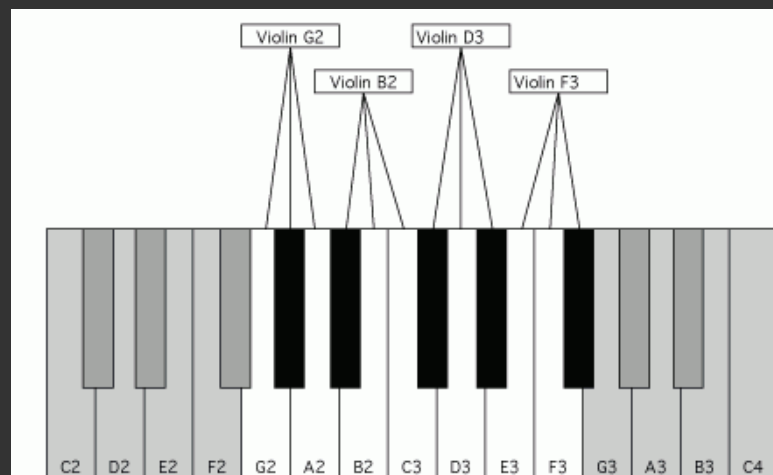
- A range of frequencies in which transposition works fine is usually not very wide (especially without looping).
- How do we cover the entire keyboard?
- An obvious solution: we record a separate sample for each key. But it's very difficult in practice:
 - we need to ensure the same timbre for all samples,
 - it is very difficult to tune the instrument (each sample must be carefully tuned),
 - it is time consuming,
 - it takes a lot of memory.

Multisampling

In practice, we use multisampling:

- we record several samples with different pitch,
- we assign each sample to a root key and we define the transposition range,
- we select samples and tune the instrument so that skipping between samples is not noticeable, so the timbre and articulation of recorded samples must be similar.

A set of samples of the same source and their assignment to key numbers is called a **split** or a **map**.



Multisampling

How many samples do we need to cover the entire keyboard? That depends on how good the transposed samples sound. In practice:

- we start with a sample in the middle of a keyboard,
- we transpose it and we determine the range of keys in which the sample sounds good,
- we add another sample outside that range, and we repeat the process until the whole keyboard is covered.

Usually, we put more samples in the middle range and less samples at the edges.

Samples should have the same articulation and timbre.



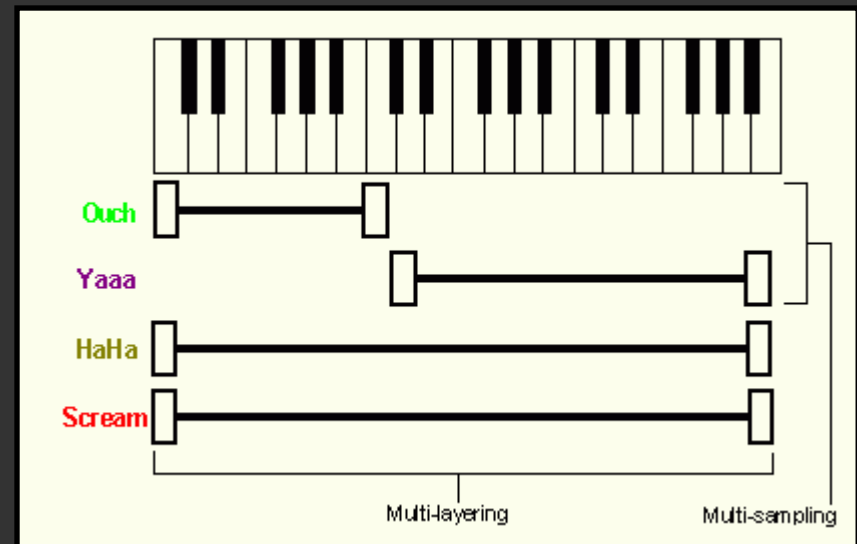
Multisampling

What can we do if we cannot obtain several samples with different pitch (only a single pitch can be recorded)?

- We can only process the sample offline, using a sound editor and transposition algorithms.
- We produce artificial samples with different pitch.
- We try to transpose the samples without altering their duration.
- Some examples of algorithms:
 - PSOLA (an algorithm for speech pitch shifting),
 - additive resynthesis (very useful here!).

Multilayering

- Multilayering: two or more samples are assigned to the same key, forming layers.
- When a key is pressed, multiple sounds may be produced.
- Multisampling and multilayering are often used together.



Multilayering

A practical example of multilayering in samplers:

- We record samples of an acoustic guitar, with different pitch and different articulation (normal, soft, strong pick).
- For each articulation, we create a separate layer.
- We use multilayering – we assign layers to the same keys.
- A layer is selected according to the velocity parameter (a strength of key pressing).
- We can also mix the layers with different proportions.
- This way, we can introduce some articulation into the instrument.

Instrument

Instrument in a sampler is a set of samples of the same sound source, with different pitch.

- **Sample** is assigned to a root key.
- **Split** defines the transposition range for each sample.
- All samples and their splits create an **instrument**.
- Sometimes, instruments are grouped into **presets**, allowing us to use multiple instruments at the same time (with multilayering or a keyboard split).

Bank

- Bank is a set of instruments, presets and samples that are loaded into the sampler memory.
- We can use any instrument that is present in the active bank.
- If we need an instrument from another bank, we need to unload the current bank and load another one.
- In a software sampler, a bank is usually contained in a single file loaded to the program.

Bank standards

- Banks are binary files. A format of storing data in the bank is often proprietary, it depends on the software publishers.
- There are some standard formats that allow for using the same bank in different instruments.
- Tools for conversion between formats are available.
- The most commonly used standard formats:
 - *SoundFont (SF2)*,
 - *DLS (Downloadable Sounds)*.

Melodic and percussive instruments

There are two types of instruments in a sampler:

- **melodic** – for sound samples with a defined pitch; different keys play samples with an appropriate pitch,
- **percussive** – for sound samples with undefined pitch (percussive sounds, e.g. a drum kit):
 - if we want a realistic instrument, we don't use transposition, so we don't need the whole keyboard,
 - a convention: different sounds (instruments) are assigned to each key.

Sound processing in samplers

Additional sound processing in a sampler includes:

- adding an envelope to shape the loudness changes, especially in the attack and release phases,
- filtering – timbre changes, may be controlled with an envelope generator or a velocity sensor,
- LFO – modulation of pitch, loudness or filter cutoff,
- modulating the sample with a velocity sensor, modulation wheel and other controllers,
- sound effects, such as delay, chorus, phasor, etc.

Articulation problem in samplers

- The main problem with samplers: instruments sound just like the recorded samples, always the same.
- We can introduce some articulation into the sound by using the velocity sensor (determining the strength of key pressing) to modify:
 - loudness and its envelope (standard),
 - filter cutoff (e.g. stronger key press – we pass more spectral components),
 - the selected layer (described before).
- Anyway, synthesizers are much better than samplers in terms of using the articulation.

A history of samplers

Fairlight CMI (1979)
(ca. 25 000\$)



E-mu Emulator II
(1984)
(ca. 10 000\$)



A history of samplers

- 1986: *Akai S900*
a first really affordable sampler;
12 bit, RAM 750 kB
- 1988: *Akai S1000*
the most popular sampler
in late 1980s,
many functions,
16-bit 44100 Hz stereo,
RAM 32 MB,
24-bit processing



Software samplers

Software samplers are computer programs that allow for instrument creation and playback. They are easier to use than hardware samplers, and they have more functions.

Selected software samplers:

- *Kontakt* (Native Instruments)
- *GigaStudio* (Tascam)
- *FL Studio (Fruity Loops)* (Image Line Soft.)
- TX16Wx (Cwitec, free version)

Kontakt - a software sampler

The image displays the Kontakt software interface, a digital audio workstation (DAW) sampler. The interface is divided into several sections:

- FILE BROWSER:** Located on the left, it shows a tree view of the Kontakt library. The current selection is 'TECKMATIC BEAT' under the 'ABSYNTH' folder.
- TECKMATIC BEAT:** The main interface for the selected sampler. It features a central waveform display with a red playhead. Above the waveform, there are controls for 'TEMPO' (131 BPM) and 'TAKT: 4/4'. Below the waveform, there are buttons for 'PLAY', 'LOOP', and 'MODE'.
- MODULATION:** This section includes a 'PITCH ENVELOPE' and an 'ADSR ENVELOPE' with various parameters like 'VEL', 'AHT', 'CRV', 'ATT', 'HLD', 'DEC', 'SUS', and 'REL'. It also features a 'RETRIG.' button and a 'DEST' dropdown.
- EFFECTS:** This section includes a 'FILTER' (with 'TYPE' and 'CUTOFF' controls), 'LO-PF' (with 'BIT RE.' and 'S.PATE' controls), 'DISTORTION' (with 'DRIVE' and 'DRIP' controls), 'DELAY' (with 'TIME' and 'FEED.' controls), and a 'MASTER FILTER' (with 'EQ' and 'CUTOFF' controls).
- ENV FOLLOWER:** This section includes a 'DEST' dropdown and a 'DEST' button.
- MASTER VOLUME:** Located on the right, it includes a volume knob and a 'SEN' button.

ROMpler

- There are EMIs that only play back the samples stored in a read-only memory, without possibility of adding custom samples.
- Such an instrument is informally called a **ROMpler**.
- They are neither samplers nor synthesizers.
- The easiest and cheapest digital instruments for musicians who need a “plug & play” device that plays sounds of many different instruments.

Wavetable synthesis vs. sampling

Wavetable synthesis:

- single periods of wave shapes are stored in memory,
- the whole signal is looped,
- waves can be changed and altered during playback,
- sound is generated in real time.

Sampling:

- longer sounds stored in memory,
- selected fragments are looped, or no looping at all,
- custom samples may be used,
- transposition is more problematic and introduces more audible distortion.

Modern EMIs

EMIs available in the market often combine functions:

- sampling – instruments created from recorded sounds,
- instrument samples stored in ROM,
- sound synthesis, usually subtractive,
- advanced sound processing (modulation, effects, combining multiple sounds).

If we want to create custom instrument banks, we use a software sampler, and the created banks are loaded into a hardware or software sampler.

Modern EMIs

Clavia Nord Stage 2 (2011)

- *Piano section* – samples of various pianos,
- *Organ section* – organ sound synthesis by physical modelling,
- *Synth section* – subtractive synthesis, FM (3-op) and sampling.



Summary

Pros of sampling:

- we can obtain realistic sounds of real instruments,
- we can create custom instruments from samples,
- any sound source can be turned into an instrument,
- many existing banks ready to use,
- novel sound creation capabilities in 1980s,
- easy to build (DSP + RAM),
- easy to use with a software implementation.

Summary

Cons of sampling:

- we are limited to existing sounds,
- not many options to alter the sound, lack of real articulation,
- large amounts of memory are needed,
- time-consuming instrument creation and tuning,
- transposition distortions,
- many musicians use ready-to-go templates,
- it's not possible to shape the sound as much as in synthesizers; creation of “new sounds” is greatly limited.

Bibliography

- Martin Russ: *Sound Synthesis and Sampling*. Focal Press, Oxford 1996.
- S. de Furia, J. Scacciaferro: *The Sampling Book*. Third Earth Publishing Inc., Pompton Lakes 1987.
- Wikipedia - sampling
- Creative Vienna – the help file
- *Viena*: www.synthfont.com