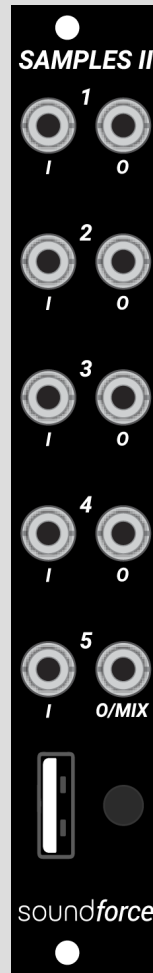


sound*force*



SAMPLES II

USER MANUAL

11 May 2024 – V1.5
Nicolas Toussaint

Thank you

Thank you so much for purchasing this module! SoundForce is a small business and I am grateful for everybody that supports us. I hope this module will bring you some fun and that it will complement your other modules well. It was a very challenging and fascinating module to build, I submerged myself in wave files, audio playback and conversions for weeks. I tried to leave no stone unturned and really pack as many features as this little 4HP module could. There are a lot to discover so make sure you read this manual carefully to use it to its fullest potential. I would like to thank Niels Moseley for his coaching and support concerning the sample rate conversion algorithm and Gert-Jan Bos for his suggestions especially for the multi-sample feature.

Overview

Samples II is SoundForce's next generation sample player module. It may look like the original Samples/S-909/S-707/S-LD modules, which became over the years some of the most popular SoundForce products, but it goes far further.

Samples II is a 5-channel USB Sample Player. It plays back and streams wave audio files directly from a USB memory stick giving users the flexibility to change samples and offering them virtually infinite memory space. Samples II is able to play back wave files of 1 hour long and more.

The latency performance was a priority during development which makes Samples II ideal for rhythmic sample playback. The decision was made to have trigger inputs instead of CV inputs to keep latency as low as possible. Thanks to an efficient triple memory buffering, the initial audio data is always available in RAM for very fast access, this combined with trigger inputs gives the module an excellent latency performance of only 0.15ms.

If the original Samples family of modules were designed specifically to be drum modules, Samples II can do more and new interesting playback modes have been implemented. The playback modes and other options can be changed in a settings text file that is automatically generated by the module and saved on the USB memory stick.

Even though the inputs are not CV inputs, it is still possible to modulate the playback speed for 2 channels. Using an LFO into input 2, you can modulate the playback speed of channel 1. The same applies for channel 3 and input 4. This is a setting that can be changed in the settings file.

During development a lot of effort was put in to avoid restrictions for the users as much as possible. A priority was the implementation of a wave file auto-converter, ideal to convert short wave files on the fly. Even though the DACs on the module run at 16bits/44.1khz, for the best audio quality vs amount of channels ratio, any wave file can be loaded by the module. The auto-converter will perform bit rate and sample rate conversion (FIR + linear interpolation) when a wave file is assigned to a channel for the first time. Stereo files can also be played back to a pair of outputs or only the left channel can be sent to a single output.

It was important for Samples II to stay compact and the 4HP format was kept from the original series. 1 tactile switch allows the user to scroll through different folders/directories and scan for wave files, and a basic attenuation/mixing system is available for set-and-forget levels.

If you have any feedback or requests: nicolas@sound-force.nl

Table of Contents

- 1. **INPUTS, OUTPUTS AND MIX 5**
- 2. **LED/BEEP INDICATOR 5**
- 3. **USB PORT USAGE AND CARE 6**
- 4. **USB MEMORY STICK REQUIREMENTS..... 6**
- 5. **ASSIGNING WAVE FILES TO CHANNELS..... 7**
- 6. **FILES AND FOLDERS RESTRICTIONS..... 8**
- 7. **STEREO ASSIGNEMENTS 8**
- 8. **CHANGING SAMPLES/NAVIGATION..... 9**
- 9. **LAST FOLDER MEMORY 9**
- 10. **WAVE FILES AUTO-CONVERTER..... 10**
- 11. **ATTENUATION/MIXING SYSTEM..... 11**
- 12. **SETTINGS FILE..... 12**
 - a. **STEREO OPTIONS 12**
 - b. **LOADING BEEP SOUND 13**
 - c. **PLAYBACK MODES 13**
 - d. **PLAYBACK SPEED MODULATION..... 16**
 - e. **MUTE SOURCES PER CHANNEL 17**
- 13. **SPECIFICATIONS..... 17**

1. *INPUTS, OUTPUTS AND MIX*

The module has 5 trigger/gate inputs labelled “ I “ which take typical Eurorack trigger/gate signals of 0 to 5V, but voltages up to 12V are tolerated.

For each channel of audio playback a single mono output is available and labelled “ O “. Outputs 1/2 and 3/4 can work in pairs and play back stereo wave files. For more about that, please read the section [stereo assignments](#).

Any output of channels 1 to 4 that is NOT patched will be summed together with output 5 and sent to the channel 5 output jack. The channels are mixed together in the analog domain using a simple opamp summer. This avoids using up a lot of mixer channels, this was an excellent suggestion of Alex/Mylar melodies, thank you for that mate!

When using the MIX out, it’s important that the mix levels between the different channels are right. To apply set-and-forget mix levels, please read the section [attenuation/mixing system](#).

Wave file files are often normalized, this means that the loudest audio peak is reaching the maximum value of the file bit range. This process is usually applied on commercial samples libraries and it is often a default option when bouncing/exporting in DAWs. Normalization is good in digital audio and helps make the most use of the DAC dynamic range and helps keeping the noise floor as low as possible.

Normalized files will output a signal with a peak of around 8vpp. This gives a good balance between sufficient amplitude/loudness for Eurorack and enough headroom for each channel when summing using the MIX out.

2. *LED/BEEP INDICATOR*

A red LED located under the front panel above the switch lights up around the switch cap to indicates that the module is busy loading or converting waves files before playback. Please note that the initial production batch doesn’t have this LED indicator and that the “beep” sound option can be used instead. [See the “settings file” section for more information](#).

3. *USB PORT USAGE AND CARE*

The USB port is available at the front of the module and is only for USB memory sticks! **Never connect another type of device to it, nor try to use it to charge a device.** Please note that the USB standard is powered with 5V and that the rest of the electronics inside of the module are working at 3.3V. The chips inside of the module cannot tolerate the large voltage swings available in Eurorack of +12v to -12V.

For this reason, I highly recommend **to keep a USB memory stick connected at all times in the module.** If you jam a jack cable with brute force into the USB connector, you may apply a voltage to the USB port that could destroy the module. Please be careful and keep the USB port covered.

4. *USB MEMORY STICK REQUIREMENTS*

Samples II has been tested successfully with a wide range of USB memory sticks old and new, even the most ancient sticks have been able to playback 5 mono files without dropping audio samples or ticking, with standard play back.

That being said, all sticks are not made equal and even within the same USB generation, some sticks have faster memory controllers than others. Relying on the speeds advertised by the manufacturer is not always providing the best results. In practical situations performance can differ from theoretical values.

The module is delivered with a compact USB memory stick that won't stick far out of your module and has been tested and proven to work for all possible applications of Samples II. The most demanding use-case in terms reading speed is the multi-sample playback mode ([see the settings file section](#)).

If the USB memory stick cannot deliver data fast enough to the module, audio ticks will be audible at the module's outputs. Usually, triggering 1 channel less helps solve this issue. But as explained above, any classic USB 2.0 or above USB memory stick will be able to deliver data for 5 channels of playback in the standard modes (non-multi-sample).

Using a modern and fast USB stick also shorten the conversion times.

5. ASSIGNING WAVE FILES TO CHANNELS

Due to its basic and straightforward 1-switch interface, a clear system is needed to help the users to assign specific wave files to specific channels. The alphabetical order system is used. When the module scans a folder/directory for wave files, all the files are internally listed and sorted alphabetically. The module then proceed to assign the first 5 files to the 5 playback channels.

For users to be certain of the wave files assignment, a prefix naming system is recommended. For this example sample set containing the following files:

- 909_snare.wav
- tom_h.wav
- ld_hats.wav
- Kick_808.wav
- toms_l.wav

simply rename the wave files on your PC/Mac by adding a small prefix:

- 1_Kick_808.wav -> goes to channel #1
- 2_909_snare.wav -> goes to channel #2
- 3_ld_hats.wav -> goes to channel #3
- 4_tom_h.wav -> goes to channel #4
- 5_toms_l.wav -> goes to channel #5

This will assign this sample set with a classic channel order of: kick, snare, hats, high tom, low tom.

The first character in the new file name will automatically adapt the alphabetical order of the sample sets and assign the wave file to your desired channel. Even though the “underscore” character is not specifically necessary, I recommend it as it adds clarity and structure.

Of course, you don't have to use prefixes and you can keep the original wave files names. In that case the module will assign them based on their original alphabetical order. You will then need to discover for yourself where every file has been assigned.

Please note, that you may not skip a channel. If you want to skip a certain channel, please add a placeholder wave file and omit triggering that channel.

Also note that stereo files will take up 2 channels if stereo is enabled ([see settings section](#)). For more detail about stereo assignments, see the [“stereo assignments” section](#)).

6. FILES AND FOLDERS RESTRICTIONS

The module is able to list and scan a maximum of 50 folders from the USB stick. In each folder, the module is able to list and sort a maximum of 50 wave files. Of course, you can only use 5 at the time, so I suggest you to create a "EXTRA" sub-folder in each sample folder, to store any extra samples from that specific sample set.

Each file path, that's the folder name plus the file name, can be a maximum of 50 characters long. It should be enough for most applications. If your samples have unusually long names, you will need to shorten them.

7. STEREO ASSIGNEMENTS

Both mono or stereo files can be played back by Samples II. Considering the total of 5 outputs, you can play back either [5 mono] files, [1 stereo + 3 mono] files or [2 stereo + 1 mono] files. Outputs 1/2 and 3/4 can be used to play back stereo files, depending on a setting in the settings file. The system will load and assign wave files as described in section 3, and for stereo files the stereo assignment works as follow:

If a stereo file is assigned to channel 1 and channel 2 is "stereo enabled", the file will be played back on output 1 + 2 when triggers come in on input 1. Channel 2 is in this case serving as right channel for the 1/2 stereo pair. Triggers on input 2 have no effect. Channel 2 will get no other wave file assigned so channel 3 will get the second alphabetically sorted wave file in the folder/directory.

The same behavior is also available for outputs 3/4.

The stereo playback is disabled by default to make sure that users always get 5 channels of playback out of the box. So stereo files are converted to mono by the auto-converter. You can enable the stereo for channels 1/2 and/or 3/4 in the settings file. [See the "stereo options" section of the settings file.](#)

Output 5 will of course only playback 1 channel. If a stereo file is assigned to channel 5, it will be converted to mono. [For more information about conversion, see the auto-converter section of this manual.](#)

Please note: stereo files need to be split to mono by the module, as the user is free to play them back either as mono channels or as stereo pairs. For long files, this result in a very long initial processing time. For that reason, long stereo ambient sounds or other fields recordings should preferably be split into mono pair of files in a DAW before loading them in the module and both left and right channels must be triggered at the same time. This will change in the next firmware revision to facilitate the playback of long stereo 16bits/44.1kHz files without conversion waiting time.

8. *CHANGING SAMPLES/NAVIGATION*

The key to change samples and switch sample sets is file/folders preparation on your PC/Mac. When a USB memory stick is plugged in for the first time, the root of the stick is scanned for wave files. If files are present, they will be assigned and potentially converted if necessary. If none are found, the module will proceed to scroll and scan alphabetically through the folders/directories on the stick until wave files can be found. When at least 1 wave file is found, the module will assign wave files to the channels. The files are then converted if necessary and primed for playback in the RAM buffers. When that's done, playback can start when inputs are triggered. This initialization process is very quick after the initial conversions took place.

You can scroll through different samples set on your memory stick using the tactile switch and switching takes less than 1 second (after the initial conversion is done). It is not instant as the old files need to be closed and new ones need to get opened and buffered, so there is a slight audio dropout when switching folders/directories on the fly.

When the last folder is reached the module will restart at the beginning and scan the root of the stick again.

9. *LAST FOLDER MEMORY*

The module automatically generates a "last_folder.txt" text file to save the last loaded folder/directory. This file is read at boot, so the module always reboots where you left it. If the folder/directory saved in the text file doesn't exist anymore, the module will restart scanning from the root again.

10. WAVE FILES AUTO-CONVERTER

The built-in wave file auto-converter accepts any combination of bitrate/sample rate and convert them to 16bits/44.1khz as needed by the module playback system and DACs. Converted .cvt files are created on the memory stick and saved into the folder/directory currently loaded. The original wave files still need to be present alongside the .cvt files, so don't delete them.

Wave files will potentially need conversion only the very first time the module scans and assigns them. When the module reboot in the same folder, all the previously generated .cvt files will be automatically loaded and start-up will be much faster.

Please note that if you edit a wave file and overwrite a wave file in need of conversion, you will need to manually delete its associated .cvt file. Otherwise the original .cvt will play back and will not include the changes from your edit. In that case, if you are editing wave files anyway on PC/Mac, it is highly recommended to export/bounce your work in 16bits/44.1khz, so no conversion is necessary.

Please note that the module is driven by a 216mhz single core Cortex-M micro-controller. It is an incredible chip with a very small size at an affordable price point, but it's nowhere near a 4 or 8-core desktop/laptop CPU in terms of computing power. The bottle neck is the simultaneous reading from one file and writing to a new one. If you want to use long audio files, or need to decode MP3/AAC files, use your PC/MAC to prepare the files. On Windows, Voxengo r8brain is practical for batch conversion of files. On MAC, Itunes has conversion options and they are paid software for batch conversion. For both systems, any DAW has plenty of export/bounce options.

Sample rate conversion is a deep subject of digital signal processing. In order to keep a good ratio between conversion times and sound quality, a FIR + linear interpolation solution has been implemented. They are far more advanced SRC strategies and algorithms. Samples II is not operating as Mastering or Hifi equipment and I am sure you will find the SRC quality to be more then suitable for Eurorack applications.

Please note: to avoid long waiting times that might make you believe that the module has crashed, there is a 30 seconds maximum file length for conversion. Any file longer then 30 seconds will not be converted and the channel will be inactive. There is also an option to play back a "beep" tone during the load and conversion process, so you know what's going. See the "settings file" section for more information.

11. ATTENUATION/MIXING SYSTEM

When using the MIX output, as un-patched channels will be summed together into output 5, the levels between each channel are important. Requesting the users to go back to their DAW, change mixer levels/gains and re-bounce the full sample set was not acceptable or user-friendly.

A basic set-and-forget attenuation system has been implemented using the single tactile switch. The procedure to use the attenuation system is as follow:

- Press and keep down the tactile switch at least 1 second
- The module will play a beep through every output, this notifies you that you entered attenuation mode
- Listen to the mix and balance between each channel as it is currently
- Mute all the triggers to the module, except the channel you'd like to attenuate
- Press the tactile switch to apply attenuation to that channel
- The module will scroll through the 8 different attenuation ratios (from 0db to -30db)
- When the last ratio is reached, the module will cycle back to 0db
- When you are happy with your level for a channel, mute the trigger to that channel and start trigger the next channel you'd like to attenuate and go through the same process
- When you are happy with the overall levels, press and keep down the tactile switch 1 second
- The module will play a beep to notify you that you left the attenuation mode

To clarify, when pressing the tactile switch in attenuation mode, the module will go through the different attenuation ratios for the last channel it received triggers on.

The different mix ratios for each channel are saved into a mix.txt file locally in each folders containing wave files. This file is not made to manually edited. The mix ratios are loaded at boot and when changing folders/directories, so you only have to set things right once.

12. SETTINGS FILE

A settings.txt file is generated automatically in the root of the USB stick the first time it gets plugged in (or when the file got deleted). The settings in root are global settings for every folder/directory on the USB stick, but you can copy the settings file inside a specific folder to override the global settings with new local settings. When switching folders/directories, the global settings are always re-loaded before looking for potential local settings.

Please note, you may not be able to edit this file on your PC/Mac when opening it directly from the USB stick. In this case, copy the settings.txt to your desktop, edit it with a text editor, save it and copy it back to your USB stick. You can find a tutorial at the end of the settings files that is a refresher of this manual.

Always carefully keep the structure of the settings.txt file. You MAY NOT add or remove empty lines. You MAY NOT add spaces, even at the end of the line. You MAY NOT have typos in the setting fields. Always respect the structure of the originally file. If you think you messed up the settings.txt file, simply delete it using your PC/Mac and plug it back again into the module to re-generate the default file.

a. STEREO OPTIONS

The stereo options define the playback behaviours for the 2 potential stereo pairs when stereo files are loaded in the module. See the section "[stereo assignments](#)" for more information. If you want to disable stereo playback for channels 1/2 or 3/4 change the default stereo option from **STEREO_ENABLE** to **STEREO_DISABLE**.

Here is an example of the stereo option settings section with 3/4 forced to mono:

```
//-----//  
/--Stereo options:--//  
//-----//
```

```
1/2 pair:  
STEREO_ENABLE
```

```
3/4 pair:  
STEREO_DISABLE
```

b. LOADING BEEP SOUND

The module will scan for wave files when booting up in the last opened folder and also when switching folders with the switch. Files that are not in the right format (bitrate not 16bits, sample rate not 44.1kHz) will be converted when scanned for the very first time. This conversion can take time, especially when files are long and bit/sample rates are high. To notify you that the conversion process is running, a beep sound can be generated during the load and conversion process. As this can be annoying during live performance, it is set to OFF by default. You can edit OFF to ON.

c. PLAYBACK MODES

They are different playback modes available to offer more flexibility and extend the possibilities of the module.

The different playback modes are:

rewind:

This mode is the standard playback for drum samples playback. When a trigger signal comes in, the rising edge of the trigger rewinds the playhead and starts the playback. The file will be played until the end or until a new trigger comes in.

rewind_fade:

This mode is similar to the rewind mode, but the playback starts fading when the trigger signal is going down (falling edge). It is ideal to shorten sounds using your sequencer gate/trigger length. It is interesting to create funky HH patterns for example. The file will be played until the end or until a new trigger comes in. The fade out envelope is tailored for HH sounds.

loop_reset: (firmware 1.3 and above)

In this mode, the audio file automatically start looping and the trigger signal will reset the file back to the start.

toggle_pause:

This mode is ideal for playing back and pausing long audio files, such as field recordings or ambience sounds. The playback will start when the first trigger comes in (rising edge). The next trigger coming in (rising edge) will pause the playback. At the 3rd trigger (rising edge), the playback will resume from where the play head previously had stopped. The file will auto-rewind when the end is reached and start playback again from the beginning.

gate_pause:

This mode is derivative of the mode toggle_pause. In gate_pause the playback is running when the gate/trigger is HIGH. So the playback is starting at the rising edge of a trigger/gate signal and it will pause at the falling edge of a trigger/gate. The file will auto-rewind when the end is reached and start playback again from the beginning.

multi_sample:

This mode allows you to assign multiple files to one channel and the system will play back a randomly picked file every time a trigger is going up (rising edge). It is ideal for glitchy beats, interesting HH patterns with different sounds or simple random playback.

The files for the said channel will need to be named with a prefix corresponding to the channel number. Additionally it should be followed by a letter for clarity. For example files for Channel 3 should be named: 3a_hh1.wav, 3b_hh606.wav, 3c_my_awesome_hh.wav, etc.. Those files can be stereo and if the right channel is stereo enabled, the playback will be stereo. Please note that because of this specific naming prefix system, stereo pairs will not skip wave files for assignment.

multi_sample_linear:

This mode is similar to multi_sample, but instead of randomly picking a new file at each rising edge of the trigger input, it scrolls through the files linearly in alphabetical order. Once the last files is reached, it rounds back to the first file. The file naming requirements and the channel limitations are the same as with the multi_sample mode.

Please note: Multi_sample and multi_sample_linear modes are demanding on the USB stick because the next file in line needs to be pre-buffered while the current file is already playing back. With the delivered USB memory stick you can run the following combinations:

- 1 mono channel on multi_sample/linear, 3 on other modes and 1 unused
- 2 mono channels on multi_sample/linear, 1 on other modes and 2 unused
- 1 stereo channel pair on multi_sample/linear, 1 on other modes and 2 left unused

This will guarantee no audio issues in all musical applications. But more than this might be possible when not sending too many triggers at the same time, but beware as glitches will appear when going too far. This limitation is necessary to keep the low latency performance on the 5 channels simultaneously.

An example of the playback modes settings section could as follow:

```
//-----//  
//--Playback modes:--//  
//-----//
```

```
CH1:  
rewind
```

```
CH2:  
toggle_pause
```

```
CH3:  
multi_sample
```

```
CH4:  
rewind_fade
```

```
CH5:  
gate_pause
```

d. PLAYBACK SPEED MODULATION

Using this setting you can modulate the playback speed of channel 1 and/or channel 3. This option is using the trigger input below the channel to track the frequency of an incoming LFO (or other signal generator). In the case of channel 1, input 1 is used to trigger the start of playback as usual and input 2 is used to modulate the playback speed of channel 1. The same also applies to channel 3 and input 4.

An incoming frequency of 100hz plays back the wave file at approximately normal speed. Increasing the frequency above 100hz plays back the file faster linearly, 200hz doubles the playback speed. Decreasing the frequency under 100hz plays back the file slower linearly as well, 50hz divides the playback speed by 2.

Please note that there is no firmware limitation of the playback speed. But of course, when increasing the playback speed, the demand on the USB stick memory controller increases. At some point, the system cannot keep up and you will hear artefacts, especially stuttering in the audio. This is a sign that the next audio buffers are not available quickly enough and that the old buffer keeps on playing back. This happens fast when using 2 of the available channels with playback speed modulation and high speeds and also when another channel is on multi_sample.

If you want to CV control the playback speed over time, you will need an LFO that is CV controllable. By changing the LFO frequency using CV over time, you will be able to modulate the playback speed.

A basic resampling of the audio data is applied using linear interpolation to remove most of the aliasing.

Of course, because an additional input is necessary, you will lose the playback on the channel needed for the playback speed LFO signal. For example, for channel 1 and input 2, you will not be able to trigger channel 2 anymore. But if channel 1 is playing back stereo files, the right channel will be played back as expected on output 2, including playback speed modulation.

To activate this setting on either channel 1 or channel 3, change OFF to ON:

```
//-----//  
/--Playback speed modulation:--//  
//-----//
```

CH1 modulation with Input 2: ON

CH3 modulation with Input 4: ON

e. MUTE SOURCES PER CHANNEL

Available from firmware version 1.3 and up. The playback of a channel can be muted by the incoming trigger on another channel. This allows to create hihat chokes and similar effects. For each channel, a mute source can be defined. After a channel has been muted, the playback will resume when its respective triggers comes in, unless a trigger on the mute source channel comes in simultaneously.

In this example, channel 2 will be muted when triggers come in channel 1:

```
//-----//  
/--Mute sources per channel:---//  
//-----//
```

CH1 mute source: OFF

CH2 mute source: CH1

CH3 mute source: OFF

CH4 mute source: OFF

13. SPECIFICATIONS

Width:

4HP

Depth:

38mm including power connector

Power:

+12V 85mA, -12V 11mA

Input signals:

Trigger/gate 0 to 5~12V.

Output signals:

8Vpp maximum for normalized wave files