ONDOLON
User manual
(Source ondolon-4.04)

Forewords

This software is a prototype, designed for experimental music . It is distributed as a free product, but any commercial use is forbidden. The author gives it as an open source software that may be changed or improved by users. It is based upon the Max/MSP/Jitter engine, primarily designed at IRCAM and currently distributed by Cycling 74.
The author does not garantee a perfect working, although this tool seems robust since several years of use , and he apologizes not to be as competent as necessary to help the users according to their software and audio environments.
However you can communicate about your feedback and ideas of improvement to cep@imagimuse.net.

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1. Introduction

1.1 Scope of the Ondolon software

The prime idea of Ondolon is to handle an audio excerpt like a video shot.

- to select the part which is interesting to operate,
- to crop it in height (pitch) and in width (speed),
- to zoom it to emphasize tiny sound details,
- to change its sound texture,
- to freeze it on a spot, or scan it at variable speed, including backward.

The second idea is to enable a polyphonic playing of the different voices in phase, in order to repect the articulation and discontinuities of the sound.

The purpose is definitely an instrumental approach : all the user's settings react in real time during the listening in order to provide the musician with a feedback. The technical principles are not new, but the goal is to operate them easily towards an aesthetical intention.

1.2 Overview

The Ondolon is globally similar to a polyphonic sampler controlled by MIDI, plus specific digital signal processing features :

1. The digital audio playing can be performed either from a wave table derived from the audio file, or from a phase vocoder.
   Actually, in a classical sampler (like in a magnetic tape player), to change the pitch, the playing speed is proportionally changed. For example, the playing speed must be multiplied by two to obtain an octave interval from the original sound, thus, the sound duration is divided by two. Thanks to the phase vocoder, speed and pitch may be separately defined, so all the polyphonic notes stay synchronized against the original sound morphological and spectral charasteristics.
2. The « sequence » mode : each new note is not played from the audio loop start, but after the previous one within the looping process. So the respect of the original sound, by following the *concrete* natural melody, prevail over the instrumental melodic sequence. Thus in polyphony, the voices play in *chord* this natural melody. The « legato » mode is similar to the « sequence » mode, but the ADSR enveloppe is applied to the global sequence rather than to every individual note.
3. The « canon » mode : new notes are automatically triggerred from a defined point in the loop.
4. Transients processing : just after having loaded a new audio file, a digital pre-processing is launched to anticipate on the further real-time audio processings. Particularly the audio areas containing transients, significant of the recorded sound articulation, are detected and displayed against the mono or stereo signal waveform. This allows interesting real-time processings :
   - the stationary sound areas being possibly strongly speeded down or up, these transient areas may be played wih a specific speed close to natural :
   - transient sounds may be accentuated or smoothed, thanks to the ADS envelope ;
   - the transient peaks may be accentuated using their « shadows », a sort of very next echo doubling the signal.

5. The AGC option may be used to boost the lower audio level audio areas.
6. The sound spectral texture may be modified, in separately fading the harmonic components (responsible of the sound pitch) and the anharmonic components (responsible of the sound matter), and in reducing any residual background noise.
7. Inversely, the *blurring* operator can shade the spectral contrasts through a light random jamming.
8. The spectral content of the output audio signal can be displayed and modified through a graphic equalizer.
9. For each played note, the *hold ctl* option is designed to freeze the volume and the panoramic MIDI controls at their values present at the note-on time, until the associated note-off, whatever the control changes on this channel.
10. A tuner enables to check the fundamental frequency absolute pitch in a representative point of the original sound, so allowing to compensate its bia against the desired MIDI pitch.
11. A low frequency binaural beating may be obtained by a slight shift of the stereo channels frequencies.

Ondolon processes the following MIDI data :

- semi-tone scaled pitches, from 0 to 127. The default value is 60 (C4).
- micro-pitch (*pitchbend*), from +1 to –1 half-tone by steps of about 1/64 tones. The default value is 64 x 128.
- velocity, from 0 to 127. The default value is 64.
- stereo localization (pan), from 0 (left) to 127 (right). The default value is 64.
- volume, from 0 to 127. The default value is 127.
- modulation wheel
- damper pedal
- MIDI channel, from 1 to 16.

2. Ondolon reference manual

2.1 Installation, starting up, set up and fast check-up

(Note: the previous versions Ondolon.app or Ondolon.exe don't work anymore on recent MacOS and Windows systems)

Mac OSX and Windows installation and starting up

Ondolon requires:
a) the Max/MSP software for MacOs or Windows .
Download and install, it's free for runtime use: https://cycling74.com/downloads

b) the Quicktime software, by default on Mac but to be downloaded on PC.

The Ondolon folder can be located anywhere.
Make sure that the files ondolon.mxf, odl_def.xml and silence1s.wav are located in the same folder.

Launch the application by running the file **ondolon.mxf** with Max. Max will open the Ondolon window. Close any other Max window or if Ondolon opens twice.

Fast check-up

a) When the Ondolon window is selected, the space bar on the alphanumeric keyboard will start the scanning of a red cursor in the ***Loop*** window on the first keypress, and stop it on the second keypress.

b) Click with the mouse on one of the white or black keys of the on-screen keyboard at the bottom of the Ondolon window. This will cause a red cursor in the ***Loop*** window to be scanned with the first click, and to stop with the second click on the same key. With the on-screen keyboard, several simultaneous keys trigger multiple slider scans.

c) Via the ***In*** menu at the top, select the external MIDI keyboard that may be connected. With a MIDI keyboard, several simultaneous notes trigger several slider scans.

At this point, no sound is emitted yet.

2.2 User interface

Overview

The Ondolon window:

- At startup, the Ondolon window is placed at the top left of the screen. It can be moved around the screen by clicking and dragging the top banner, or resized by moving the bottom right corner.
- The menu bar provides access to some useful functions of Max/MSP :
  – MAX / About Ondolon: displays the current version and the link to the application's website.
  – File / Maxmenus gives access to the complete Max menu.
  – Window / Zoom displays Ondolon over the entire width of the screen.

Types of settings:

- slider : horizontal, vertical or circular : continuous change on a value.
- selector : horizontal or vertical ruler, made of n squares or buttons: selection among n positions.
- menu : enables to select a choice ; the last done choice appears checked.
- toggle (check box): on/off selection, an X means the « on » position.
- key: trigger, flashing once.
- numeric box : input by mouse dragging.
- graphic pad to get mouse pointer xy coordinates.
- on-screen keyboard : to play MIDI notes by mouse clicks.

Types of displays:

- wave form, spectrum.
- player heads (red bars scanning the wave form)
- enveloppe drawings
- level indicators
- textual display box or area : values, pitches, messages
- key flash
- on-screen keyboard .

Color code:

- green : sample playing and looping modes, transients
- orange : settings for the time dimension,
- yellow : setting for the pitch dimension,
- blue : audio settings
- light blue : other sound parameters
- grey : real time displays
- purple : digital audio environment parameters, presets
- white: disabled setting, because without impact on the current configuration (therefore accessible in a configuration change perspective)

Layout

The different settings are grouped inside areas of the Ondolon window, called « panels ».

- ***In***: MIDI input
- ***Out*** : final equalizer filter and audio output
- ***Control*** : playing mode, in sequence or polyphony, presets management
- ***Audio***: input and output audio files
- ***ADSR*** : amplitude enveloppe drawing and output level control
- ***DSP***: digital audio procesing features
- ***Loop***: the audio loop programming
- ***Speed*** : audio loop speed settings
- ***Pitch*** : pitch settings

Help

When a setting is hovered by the mouse pointer, its function is displayed as bubble hint.

Use of the mouse and the alphanumerical keyboard

The sliders are taken into account in real time. The factory default setting (0, maximum or center, set up by design) may be fastly got by a double click.
Numeric input fields are set up by a vertical mouse dragging.
The drag and drop mechanism may be used to load audio or preset and playback files.

The space bar on the alphanumeric keyboard can be used :

- to generate a MIDI C4 note for a quick test (see §2.1 and §2.10)
- to "freeze" and resume a session replay (see §2.14).

Use of presets

The *preset* area is in the ***Control*** panel of the user interface. The textual box displays the last loaded *preset* .

At start up, the user interface is configured according to the *default* preset.
Click on ***default*** to get back to this configuration.
Click on ***save*** to save the current configuration into a XML file, joined with a .wav file containing the working audio file, the name of which is proposed as file name with the .xml suffix.
Click on ***load*** to get back a previously saved configuration. This file can also be dragged and dropped onto the title of the ***Contro***l panel with the mouse.

Click on ***sdef*** to save the current configuration as default configuration.
Click on ***factory*** to get back the factory default configuration, the working audio file being a 1 second silence sample.

2.3 Digital audio processing performed by Ondolon

The digital audio processing (DSP) is performed in three stages :

a) Pre-processing, just after an audio file is loaded or after some parameters were changed,

b) Real-time processing while playing notes, at every playing head position.

c) Real-time processing of the output signal resulting from the sum of the playing head signals.

Odolon can perform 8 function types :

1. Phase vocoder, enabling to control independantly the speed and the pitch. During the pre-processing, the audio signal is converted into a frequency spectrum. During the playing phase, when scanning the loop, the spectrum is reciprocally transformed into pitches that can be shifted according to the the MIDI pitch and the pitch settings. The vocoder function may be disabled and replaced by a wave table playing, more realistic but disabling several features relying on the vocoder.
2. Processing of transients : the audio signal area containing transients are detected during the spectral pre-processing and displayed as green areas under the left channel signal. During the real-time processing, when a playing head crosses a transient green area, several function may be implemented :
    - speeding up : when the global speed is lowered (time stretching) through the ***Speed*** panel settings, the down speeding in the transient area may be tempered, even annihilated in order to obtain a more realistic rendering ; the symetrical effect may be obtained for a global upspeeding ;
    - envelope : applying the ADS enveloppe from the transient area beginning, thus boosting the attack amplitude higher than the sustain one (except if the sustain is set up to maximum), and eventually with a smoothed attack curve ;
    - relief : producing a « sound shadow », a next brief echo to emphasize the attack dynamic.
3. Automatic gain control : during the pre-processing phase, an amplitude enveloppe is

computed along the whole audio file duration, smoothed on 100 ms and stored. In real-time, the user can apply an automatic gain computed from the playing head position against this stored envelope.

4. Spectral texture modification, by seperate fading of the harmonic components (the energy of which is focused in a few high amplitude frequencies) and the anharmonic sounds (the energy of which is spread in numerous low ampltude frequencies). The user may set up the amplitude level which splits the *pure* harmonic component and the *noise* anharmonic component. He also may decide to apply or not the pitch and MIDI shifts to the *noise* component.
5. Additionally a noise gate can eliminate any residual background noise.
6. Blur : smoothing the too brusque changes of the frequential content, which can lead the vocoder to produce undesirable phase warpings.
7. Control on the stereo effect and the binaural low frequency beating..
8. Graphic equalizer for the weighting of the output signal frequential spectrum.

2.4 Input interface (In panel)

- At right : menu to select the MIDI input port : external keyboard or internal bus.
- ***MIDI chan :*** selection of the MIDI input channel number : from 1 to 16.
- ***vol pan***: button to take into account the MIDI control 7 (volume) and 10 (panoramic).
- ***hold ctl:*** according to the MIDI standard, the received MIDI controls globally address the whole channel, thus samely for all the simultaneous current notes. When ***hold ctl*** is checked, the left-right *pan* (*control 10*), the *volume (control 7)* and the *pitch-bend* modulation are sampled at the beginning of each note and are individually holded until the note extinguishment after the *release* .

When a note enters through the MIDI port, the yellow key to the right of ***In*** flashes, the pitch is displayed digitally next to it and is indicated by the on-screen keyboard at the bottom of the window.

2.5 Output interface (Out panel)

The menu just right to the **Ou**t title selects the computer output audio port.
NB: if the menu doesn't give access to the list of available ports, Ondolon (actually Max/MSP) may have been disturbed during a previous session and is no longer connected to the audio driver. To restore :

1. *in the Ondolon menu, click File/Max Menus*
2. *then Options/Audio Status*
3. *In the Audio Status window select Core Audio (MacOs) or MME (Windows) or other if specific audio hardware.*

Graphic equalizer to modify the output signal spectrum :
The blue menu enables to select the filtering mode: ***display, lowpass, highpass, bandpass, bandstop, peaknotch, lowshelf, highshelf, resonant, allpass.***
The horizontal line may be moved up and down by the mouse to define a gain G in dB, according to the scaled rule at the right.
The two parallel vertical lines may be moved left and right and define the central frequency F and the filter quality factor Q. F, Q and G are displayed when the mouse pointer hovers this area.
The ***midi*** slider enables to shift the filter central frequency by the played MIDI note. The rate aries from 0 (no shifting) to 2.

Spectrum displaying : overlaying the graphic equalizer and using the same frequency logarithmic scaling from 4Hz to 22050 Hz.

Grey menu : ***no spectrum*** (no display), ***pre-spectrum*** (before the filter), ***post-spectrum*** (after the filter).

The final audio output volume is adjusted by the vertical ***vo***l slider in the ***ADSR*** panel.

2.6 Audio file loading (Audio panel)

Load opens the file browser to select an aiff, wav or mp3, mono or stereo file. A file can also be loaded through drag and drop over the ***Audio*** panel title.
(Note : the Load key and the drag and drop area may be also used to load a .txt session file. See §2.14)

The name, the duration, the sampling rate and the **0** (mono) or **8** (stereo) format of the file are displayed at the right of the ***load*** key.
If the file is mono, the ***spread*** round slider in the ***DSP*** panel will stay on *mono* during the time that this file is operated.

Amplitudes are normalized to - 3dB in the Ondolon internal storage.

The ***Loop*** panel window displays the two amplitude curves of the stereo sample (twice the same in the case of a mono signal), then it shows the signal pre-processing progress by using the loop size as a progression bar.
Ondolon tries to upspeed the pre-processing performance to the maximum cpu speed by operating the NonRealTime audio driver provided by Max/MSP. It requires that this one supports the same sampling rate as the current audiodriver. Elsewhere the preprocessing is performed in real time according to the audio file duration.

2.7 Envelope drawing (ADSR panel)

The ADSR enveloppe modulates the sound level in response to performing a MIDI key strike, that is, at the *start* point of the loop. If this *start* point is within the loop (between *loop* and *end*), the ADS enveloppe will be applied again at every crossing.

In the *volume* mode, the level is set by applying a gain envelope to the Attack, Decay and Release phases. In the l*owpass* or *hipass* mode, the level is set by applying an gain increasing slope to the Attack phase, then a sliding low pass or high pass filter to the Decay phase, and finally a gains decreasing slope to the Release phase.

- ***attack:*** slider for the duration of the attack slope (when receiiving the *note-on),* from 0 to 2 secondes,
- ***log***: the circular slider controls the enveloppe logarithmic curve ; the button selects the attack slope shapestraight or logarithmic.
- ***sustain***: vertical slider for the relative level (from 0 to 1) during the note sustaining and until the release.
- ***rel*** (r*elease*): slider for the duration, from 0 to 2 seconds, of the decreasing slope after the release (at the *note-off* reception).
- ***decay***: slider for the end of the decreasing slope from the attack level 1 to the sustain level; this slider is linear from 0 to 2 seconds, then quadratic beyond. Its value in seconds is displayed at its right extremity.
- ***agc*** : automatic gain control factor, from 1 (no change) to 48.
- ***true***: button to enhance the attack fidelity. Actually the vocoder operates a computing sliding window from 512 to 8192 digital samples. This computing smoothes the rendering of very short transients which are significant of the attack color. When the ***true*** button is

checked, the attack point at note-on will beneficiate of the digital processing applied to transients ;
- ***trans*** : reciprocally, application of the ADS enveloppe to transients ;
- ***curve*** : DSR enveloppe logarithmic shape.
- ***volume / lowpass / hipass***: menu replacing the *decay* (volume slope) by a frequency filter masking from treble to bass (lowpass) or from bass to treble (hipass).

2.8 Programming the audio loop (Loop panel)

To define the audio loop consists in selecting a musically significant segment of the audio file to establish a virtual instrument.

<u>1. To examine the audio file in details :</u>
click within the upper half of the *loop* window, drag down to up to zoom in, drag left or right to make slide the time range.

<u>2. To define the loop:</u>
Either directly move the start, loop, end sliders, or click within the lower half of the *loop* window : vertically drag to enlarge/reduce the length of the loop and laterally drag to move it along the audio signal.

- ***start***: slider to set up the start point when receiving thec *note-on* MIDI code
- ***end***: loop end slider
- ***loop***: looping point slider if the note duration is greater than the ***start*** -> ***end*** time
- ***lock***: 2 buttons to lock the loop :
  - above : coupling *start* and l*oop* sliders, if they have not to be distinguished,
  - below : locking the loop dimensions when sliding it along the audio signal.

<u>3. To program the loop path:</u>
The green selector offers three audio loop following ways:

- ***spot***: the played sound results from the spectrum frequencies around a fixed spot. The spot setting may be moved by the ***start*** slider, this allowing to manually scan the spectral content along the audio file (*freeze* effect).
- ***no loop***: unlooped simple playing, each *note-on* starting from the ***start*** point and ending either on the ***end*** point or after the *release* time after the *note-off.*
- ***loop:*** loop playing *start -> end* then *loop -> end* and so on until the *release* end after the *note-off.*

NOTE: if the ***start*** postion is after ***end***, the audio file is played backward.

<u>4. Additional options:</u>

- ***pingpong*** button: figure in palindrome :
  - unchecked: after ***end***, the playing immediately restarts from the ***loop*** point
  - checked: after ***end*** , the loop is played backward towards ***loop*** , then towards ***end*** again and so on.
- ***tail*** button ***:*** indicates to abort looping during the *release* time, after the *note-off,* so that the audio file can finish to be played beyond the ***end*** slider towards the the file end. This is ineffective if the mode is ***sequential*** or ***legato ,*** or during a pingpong backward path;
- ***declicker*** slider : in wave table mode, the jump from ***end*** to ***loop*** may produce a sound click; if needed, this slider sets up an signal overlap duration (from 0 to 100 ms) around the jump.

Up to 8 playing heads (red bars, for 1 to 8 polyphonic voices) display in real-time the audio loop progresses.

2.9 Operating modes (Control panel)

To control how the notes are managed in sequence and polyphony in relation to the sound sample.

1. The ***poly*** numeric box defines the possible number of voices, from 1 to 8.
Just right to this box, the red number displays the really played polyphony, taking into account the *release* time after each *note-off*, and the purple number displays the number of simultaneous key-downed notes from the MIDI input. If the number of simultaneous input notes is too big, momentarily the *poly* box color changes to dark green. In this case, the most recent note cuts the oldest (voice stealing). Thus, in monophony (***poly*** = 1) the playing can become *detached*. The number of red playing heads that follow the notes is related to the polyphony setting.

2. The green menu defines how strings of notes are sequenced:
- ***solo*** : each new note individually runs along the audio loop from the ***start*** point and is amplitude-modulated by the ASDR enveloppe.
- ***sequential*** : when a *note-on* happens before the end of the previous note *release*, the new note is played after it in the loop, not starting from the ***start*** point ; ***each note*** is amplitude-modulated by the ASDR enveloppe.
- ***legato***: sequential mode played in *legato*: when a *note-on* happens before the end of the previous note *release*, the new note is played after it in the loop, as in the *sequential* mode; however the ADSR enveloppe is applied to the whole sequence rather than to each individual note.
- ***canon*** : in the canon mode, each new note starts from the ***loop*** position, not from the ***start*** position ; If the ***start*** slider is set inside the loop (between ***loop*** and ***end***), everytime the current note crosses it a similar new note is automatically started from ***loop,*** thus creating a canon device, and so on with respect to the *poly* parameter limitation.

2.10 Pitch setting (Pitch panel)

The ***Pitch*** panel is designed to tune the instrument, all settings being possibly changed in real time. The frequency ratio between the played note and the C4 reference is displayed under the *Pitch* title.

touch : this black button allows you to choose the mode of use of the on-screen keyboard :
◦ polyphonic: one click-on to press the key, a second to pick it up.
◦ touch: the key is depressed by the click-off
The indicator above displays the velocity of the last note played,
- depends on the height position of the mouse on the on-screen keyboard key at the time of click-on,
- depends on the keystroke velocity on the external MIDI keyboard.

lock: by checking the blue lock button, this above slider remains locked on a single velocity value, by default 64. In this position it is also possible to use the space bar of the alphanumeric keyboard to trigger or stop a single C4 note. This function is disabled in plauback mode or when other notes are triggered from the on-screen keyboard or MIDI keyboard).
midi: button to take into account the MIDI input *pitch* . The neutral *pitch* is defined as 60 (C4).
transpo: semi-tone transposition selector.
oct: octave transposition selector.
trim: fine pitch setting slider, + or – one half-tone; reset to center by double-click.
nat : natural/tempered scale button, the natural (or Pythagorician) scale being based upon

harmonic frequencies used for the seven intervals of the the major scale (the white keys on a piano), after applying *transpo* and *tune*; the intermediate notes (black keys) keep their tempered scale pitches.
slide: glissando slider, between differently pitched notes.
beat : this slider, which can be coupled to the MIDI keyboard modulation wheel, enables to create a stereo binaural beating from 0 to 20 Hz, when the ***DSP*** mode is *pvoc* (phase vocoder mode, see § 2.11).

When the *noise-pures splitting* is operated in the ***DSP*** panel, the pitch setting adresses only the harmonic components. If the yellow ***midi*** box is checked, the setting also adresses the noisy components. See §2.11.

The post-setting desired pitch is displayed by the ***play*** field: note resulting from the MIDI input, from ***transpo*** and ***oct,*** and cents resulting from ***trim*** and the input *pitchbend* if any***.***

However the pitch actually played by the audio file depends on this of the actually sound signal that has been recorded in this audio file.
To tune Ondolon in order to finally play the desired MIDI note, the offset from the desired pitch and the resulting pitch must be counterbalanced by a ***bia*** shifting, measured through a tuner.
The procedure is as following:

1. playing preferably in ***spot*** mode, look for an area in the sample where the audio pitch is both stable and representative of the fundamental (lower) frequency ;
2. check the ***tuner*** button, so producing an additional pure C4 pitched tuner sound ; the tuner sound level is adapted to the ***start*** slider position to make easier to compare it to the level of the played sound;
3. modify ***transpo, oct*** and ***trim*** to, by listening, set the tuner in unison with the heared fundamental ; the ***bia*** number indicates the MIDI offset;
4. uncheck the ***tuner*** button, so turning the tuner sound off and resetting ***transpo, oct*** and ***trim*** to their default settings ; the ***audio*** indicators then display the pitch, cents and frequency related to the sound fundamental resulting from the previous process, while the ***play*** indicators display the pitch and cents resulting from the ***pitch*** settings and the MIDI input;
5. check the button at the left of ***bia*** to apply this shift so that the played pitch is the same as the produced audio pitch ; uncheck to eventually produce the sound without the correction.

The ***bia*** value may also be directly entered or modified through a vertical mouse drag.

2.11 DSP settings (Digital Signal Processing)

- ***internal s.r.*** (within the ***Audio*** panel) : defines the internal audio sampling rate (44,1kHz, 48kHz ou 96kHz (dependent on hardware), thus also the produced audio files sampling rate. The menu allows to modify it, thus causing to re-do the pre-processing.
- ***DSP*** : the menu under the *DSP* title defines the FFT computing window width (512, 1024, 2048, 4096 or 8192 samples). If this width is modified, the pre-processing is re-done.
- ***Pvoc, wtab, mix*** selector: to choose the playing mechanism, by vocoder (*pvoc*), by wave table (*wtab*) or adapted (*mix*) :
  - ***pvoc*** makes possible all the DSP features,
  - ***wtab*** works so that the ***speed*** is controlled by the ***pitch***, like in a classical sampler (or a tape transport) ; the other ***DSP*** and ***Speed*** settings thus are disabled and colored in white ; the *sequence* playing mode is also disabled here. The speed is multiplied by two for a pitch shifted by 12, etc..

- **mix** detects in real-time if the combination of *speed* and *pitc*h settings enables to use the wave table (speed ratio x pitch ratio = 1), and uses this one if yes and the vocoder if no. This feature is interesting to momentarily use the wave table, more realistic, in the place of the vocoder to render the transients when their speed is set to 1 through the ***transients*** slider (in the ***Speed*** panel) set fully at the right.

- ***noise-pure*** : a mixer, based upon the FFT based analysis of the harmonic/anharmonic sound texture. The harmonic component is generally assumed by a few high amplitude frequencies, while the (noised) anharmonic component is assumed by numerous low amplitude frequencies. The available settings allow the user to adjust the analysis according to the specificities of the sound to process and to modify it according to the characteristics that he wants to emphasize. These settings are disabled with the ***wtab*** playing mechanism.
  - Two horizontal logarithmic sliders split the frequencies into three packets according to their amplitudes :
    - ***gate*** : to set the threshold of noise elimination (noise gate), from 0 to 0.01
    - ***split*** : to set the amplitude threshold to split the frequencies considered as anharmonic components (*noise*) and harmonic (*pure*).
  - Two vertical sliders produce the mixing, ***noise*** at left (± 24 dB to ± 80 dB, according tothe ***split*** threshold), ***pure*** at right (± 24 dB)
  - The ***midi*** button enables to apply or not the pitch settings to the ***noise*** component.
  - The ***EF*** button is used to make the *split* threshold follow the the signal level (enveloppe follower), so that the noise range is shrinked when the level is low.
  - Click on ***reset*** to set the above settings to their neutral default values.
- ***trans.thresh*** : transient detection threshold : during the pre-processing, the spectrum content continuous variation from a FFT frame to the next one is stored as a continuous measure of the transient ratio along the audio, and the areas displayed in green show the areas where this ratio is above 10%. The ***trans.thresh*** slider rises this threshold up in order to display the more energetic transient ratios to be considered.
- ***shad***: to emphasize transients in juxtaposing their shadows in the form of a few milisecond late echo. The shadow of a left-channel transient is produce on thet right channel, and reciprocally. The slider sets the echo delay, from 0 to 50 ms, the duration of every shadow is twice this value and the button enables to apply the shadow or not.
- ***blur*** : the slider set the number of FFT frames that are concerned in the blurring ; the resulting blurring duration, displayed in ms, depends on the sampling rate and the FFT window width. The button enables to apply the blurring or not.

The ***spread*** circular slider controls the stereo channels mixing when the ***speed*** slow down is <1. Actually in this case the stereo phase bias are multiplied by the inverse of the speed and the resulting phase distorsions could be perceived as more disturbing than interesting. This slider sets the stereo spread between the speed decimal value and 1 for each of the 8 stereo polyphonic voices.

The ***cpu*** indicator displays the *cpu* load deriving from the signal processing : above 2/3, some glitches may happen in the produced sound. The polyphony multiplies the *cpu* load, depending on mono or stereo, the number of simultaneous notes, the FFT window width, the sampling rate and the operating DSP functions.

2.12 Setting the speed (Speed panel)

The ***Speed*** panel enables to change the sample playing speed, all parameter being modifiable in real-time:

- middle selector, speed multiplication or division, by powers of 2.
- ***freeze*** : button setting the speed to zero, so freezing the current playing position.
- ***adjust:*** fine adjusting slider, from 0,5 to 2; reset to 1 by double-click.
- ***smooth***: slider to smooth real-time speed changes, from 0 to 4s.
- ***transients*** : specific speed ratio applied to transients :
  - slider on the lef : same speed as the global speed
  - slider on the right : actual speed (ratio = 1) ;
  - intermediate position : the ratio is proportional to the transient ratio measured after the ***trans.thresh*** threshold***.*** When the slider is completely at the right, this ratio is set to 1 to improve the fluidity.
- ***midi***: button to work so that the ***speed*** is controlled by the ***pitch***, like in a classical sampler (or a tape transport) ; the ***Speed*** settings thus are disabled and colored in white ; the *sequence* playing mode also is unavailable here. The speed is multiplied by two for a pitch shifted by 12, etc..

The number under the ***Speed*** title displays the ratio resulting from the set of settings.

2.13 Audio recording (Audio panel)

To record the Ondolon audio output in a .wav 24 bits file:
Check the blue ***record*** button in the ***Audio*** panel to define an audio file name. The suggested and modifiable file name is the current preset name, preceeded by the *out_* prefix and ended by the *.wav* suffix, and displayed just at the right of the button.
For example, if the name of the preset is *toto.xml*, the proposed and modifiable name will be *out_toto.wav.*

It is displayed at the right of the button. The sampling rate will be the internal frequency *internal s.r.*.
To start the audio recording press the « enter » key on the alphanumerical keyboard, the ***record*** button becomes red and the counter starts counting seconds .
To stop the recording, click the ***record*** button again, becoming blue again.

The output volume ***vol*** setting is not taken into account for the audio recording.
NB: When the defined file name does not include the .wav extension, Ondolon automatically will add it.

2.14 Ondolon session recording and play-back

Simultaneously with the recording of the audio output in a .wav file, Ondolon systematically memorizes the state of all the settings at the beginning of the recording session, then the chronology of user actions until the end of the recording: MIDI events (notes, pitchbend) and settings modified during the session. When the session is over, a text file .txt with the same name as the .wav file is created and saved in the same folder. It contains the name of the current preset, the session duration and the data stored during the session.

Note: during the recording of a session it is not possible to modify the sampling rate (Audio panel) and the DSP calculation window width (DSP panel).

To play-back a session, Ondolon must have at its disposal :

1. the .XML file of the preset in force during the session (for example *toto.xml*)
2. the .WAV file associated with this preset (e.g. *toto.wav*)
3. the .TXT file of the session (e.g. out_*toto.txt*)

   NB: the audio file of the session (for example *out_toto.wav*) is not used here.

Immediately after this recording Ondolon still has the session log data internally. It can therefore be replayed immediately simply by clicking on the green ***playback*** button, which turns red.
During the replay the playback heads move according to the MIDI notes, which are replicated on the on-screen keyboard, and the settings are animated until the end of the session. The progress is followed by the slider at the bottom of the Audio panel, to the right of the playback button, and stops at the end of the replay.

Replay can be stopped at any time by clicking ***playback*** again. But it can only be resumed at the beginning of the play-back.

During replay, the user can "freeze" the position of the settings and read heads, either by clicking in the progression slider, or by pressing the space bar on the alphanumeric keyboard. In this state, for example, a new preset can be created to " snap" these settings. To resume the replay, click the cursor again or press the space bar.

One can interact on the settings being replayed, trigger new MIDI notes, but these actions are not added to the session data. Note also that clicking ***record*** during replay clears all previous data in order to record a new session. However, the file on disk is kept if the new record has a different name.

In the general case, the user may need to replay a previously saved session in a different preset context than the current one. To do so, proceed as follows:

1. load the session .TXT file by dragging and dropping it on the Audio panel or by clicking on ***load*** in the Audio panel to open a selection box;
2. launch the replay by clicking on ***playback*** ; Ondolon compares the name of the current preset with the name of the preset saved in the recording ;

   a) if it is identical, the session replay starts immediately;

   b) if it is different, Ondolon asks to load the corresponding XML file from the disk in order to benefit from the WAV audio file accompanying this preset;

   c) after loading and pre-processing, click again on ***playback*** to start the replay of the desired session.

Color indications of the playback button :

- *white*: Ondolon does not contain a replay file, or the replay file does not match the current preset;
- *green*: Wave contains a replay file that matches the current preset;
- *red*: Replay is running.

3. Ondolon technical data

Audio and text files

- Audio file input format .WAV, .AIFF, .MP3, mono or stereo, free sampling. Unlimited duration.
- Real-time audio output file and audio file joined to a preset file : stereo .WAV 24 bits.
- Output/input of preset file in .XML format
- Output/input session replay file in .TXT format.

MIDI inputs:

- 1 external or internal port , from which a channel number 1 to 16 is selected.
- Alphnumerical keyboard space bar
- Polyphony from 1 to 8.

Operated MIDI controls:

- Note (pitch/velocity)
- Pitchbend, Damper (control 64 : sustain pedal), Modulation (control 1)
- Volume (control 7) and Pan (control 10)
- Reset controls (control 121) and All notes off (control 123)

ADSR

- Attack (launched by note-on) : from 0 to 2 seconds, straight or logarithmic slope
- Decay (after note-on) : from 0 to 78 seconds
- Sustain : from 0 to 1
- Release (after note-off) : from 0 to 2 seconds.

WAVE FORM DISPLAYING

- Stereo amplitude
- Transients
- Real time spectrum

MODIFICATION OF THE SPEED

- Continuous, from /128 à X16, or freezed.

MODIFICATION OF THE PITCH

- continuous, from -4 to +4 octaves
- tempered or natural scale
- tuning bia correction.

DSP (digital signal processing)

- FFT from 512, 1024, 2048, 4096 or 8192 samples (related to the number of frequential *bins*). Overlap = 4.
- Evaluation of the transients amount by evaluating the spectral distance (= sum of the amplitude gaps of the composing frequency *bins*). The threshold of taking into account may be set up, from transient sensitive [distance > 10%] to not sensitive [distance = 100%].
- Spectral blurring: from 1 to 8 frames range (frame width = FFT size / overlap).
- Relief : echo from 0 to 50 ms.
- Noise threshold : elimination of *bins* the amplitude of which is under a [0.00 - 0.01] setting.
- *Noise* - *pure* splitting based upon a *bin* amplitude threshold, set up from 0 to 50, weighted by the enveloppe follower, (50 = empirical amplitude of an harmonic *bin*).
- *Noise v*olume : ± 24 dB to ± 80 dB, the limits being inverse proportional to the split value.
- *Pure* volume : ± 24 dB.