

WaveGene help (translated from Japanese)

Table of contents

Introduction	3
Install / Uninstall	4
Example of operation	5
How to use	9
Format setting	12
Setting Waveform Parameters	13
Extension of frequency setting	23
Frequency setting bar	24
Band limit	28
Output filter	32
How to use user waveform	48
Measurement of frequency characteristics using cyclic sweep	56
Playback device setting dialog	62
Command line parameters	68
Operation with keyboard	69
Setup Examples	70
Hints	79
Others	85
History	87

Introduction

Multifunction high precision test signal generation software

WaveGene Help

V1.50 (2013/01/05)

Please see the [history](#) for changes from the previous version.

Introduction

This WaveGene is a tool that generates various test audio signals by calculation and outputs them to sound devices and files.

Up to 8 arbitrary combinations (mix) of nine kinds of waveforms of sine wave, rectangular wave, triangular wave, sawtooth wave, pulse train, pulse train (+ -), white noise, pink noise, M series noise (MLS) You can output.

(It corresponds to multichannel Wave file of up to 8ch (WAVEFORMATEXTENSIBLE format))

* With regard to periodic waveforms (sine wave, rectangular wave, triangular wave, sawtooth wave, pulse train, pulse train (+ -)), it is also possible to generate a signal without band limited alias.

* You can import and generate two types of arbitrary waveform (Wave format and original text format) as the user waveform.

* Sweep (frequency, amplitude, phase, combination thereof, continuous, step), modulation (AM, DSB, FM, PM, PWM), tone burst generation function.

Depending on the PWM function, it is also possible to change the duty cycle for periodic waveforms other than sine waves.

User waveforms can also be used as modulation signals. (Some restrictions apply)

* In addition, various settings such as phase and sample delay setting, sorting of output destination of L / R, offset, polarity, etc. are possible.

* Frequency setting can be set in cycles according to the number of samples in addition to Hz.

* Amplitude setting can be set with 3 types of dB, numerical value, and percentage. (Also, these units can easily be interconverted)

* Frequency and amplitude can be changed continuously smoothly by the mouse wheel, and it is operability like an analog oscillator.

* The frequency of the two waveform generators can be changed synchronously while holding an arbitrary ratio.

You can sweep the two waveform generators synchronously.

* Output can be filtered with arbitrary characteristics.

You can also oversample (x2, x4, x8, x16) to generate a signal and filter it.

The filter can change its characteristics even during signal generation.

* The Wave format is 8, 16, 24, 32, Float 32bit, 1 to 8ch, sampling frequency can be set arbitrarily (currently from 1000Hz to 99MHz).

* MME, DirectSound, ASIO, WASAPI (exclusive mode Windows Vista SP1 or later) driver compatible.

* When outputting files, you can output in text format in addition to Wave format.
(Since this file has the same format as the user waveform, it can also be read as user waveform again.)

* All settings can be exported as setting parameter file with comment, and you can reproduce complicated settings at any time by reading again later.

Especially, for convenience when using with spectrum display software like WaveSpectra like my work, FFT There is a convenient function that allows you to set frequencies with just one touch (no spectral error) for extracting data when doing.

For checking sound devices and various audio, CD-R You can use it as a test signal source for CD - DA creation by.

Operating environment

- Windows XP or later
(**Note:** V1.50 to 95/98 / Me no longer work)
CPU clock 500 MHz or more.
(However, some higher-speed CPUs may be required depending on the function to be used, such as band limitation and output filter)

*** MME, DirectSound, ASIO, WASAPI (Exclusive mode only) Driver support.**

(Accurate output is possible with ASIO and WASAPI)

[ASIO Interface Technology by Steinberg Media Technologies GmbH]



Created with the Personal Edition of HelpNDoc: [Create iPhone web-based documentation](#)

Install / Uninstall

Installation

Make a suitable folder and just insert WG.EXE and WX02DLL.DLL (and help file WG.CHM).

However, please be sure to put in **ordinary folder of the local disk in the PC (eg C:\WG**

or D:\WG_TEST etc) .

And if you make a suitable shortcut etc for starting, that is it.

Also, while using Windows It does not write to the system folder or the registry at all, so you can use it with confidence and testing only.

However, it is necessary to save the window position, display condition and others and reproduce it at the next startup Make the WG.INI file in the same folder as the started WG.EXE.

(It will be made automatically at the end, you do not need to be at the beginning)

If you want to use under different conditions, please make folders separately and put them in.

Also, at the initial startup, initial setting values of all parameters are set in the setting parameter file As INIT.WG, it automatically writes to the same folder as WG.EXE.

This INIT.WG can be used to restore all settings to the initial state.

(Please refer to " [Read configuration parameter file](#) ")

* By default, the setting parameter file (*.WG) It is saved in the same folder as WG.EXE.

Uninstallation

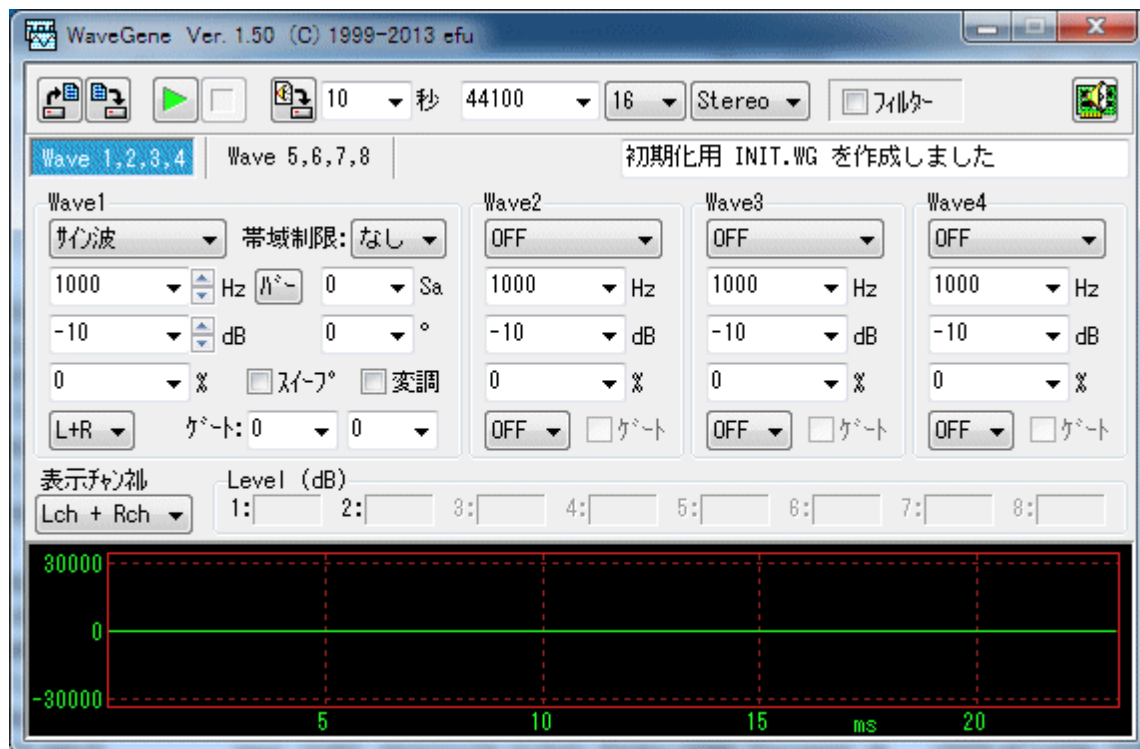
Since we do not use the registry, we just need to erase all the files for each folder, which is the end.

Created with the Personal Edition of HelpNDoc: [Easy EPub and documentation editor](#)

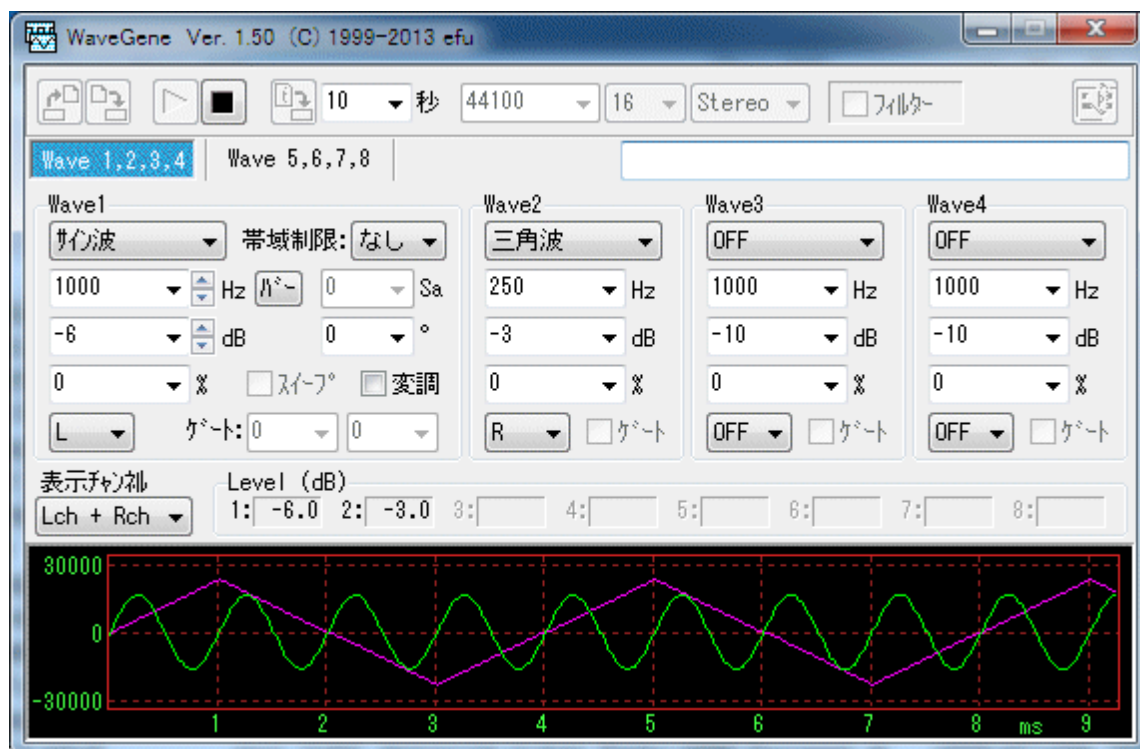
Example of operation

Operation example

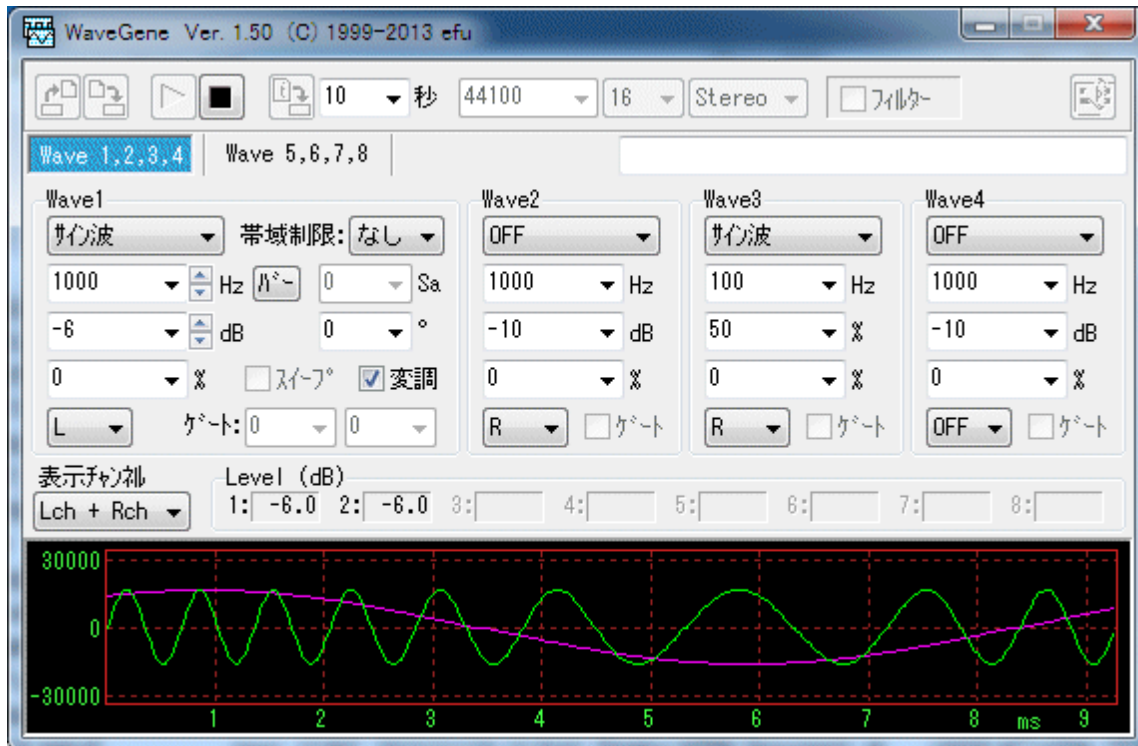
Immediately after the first start



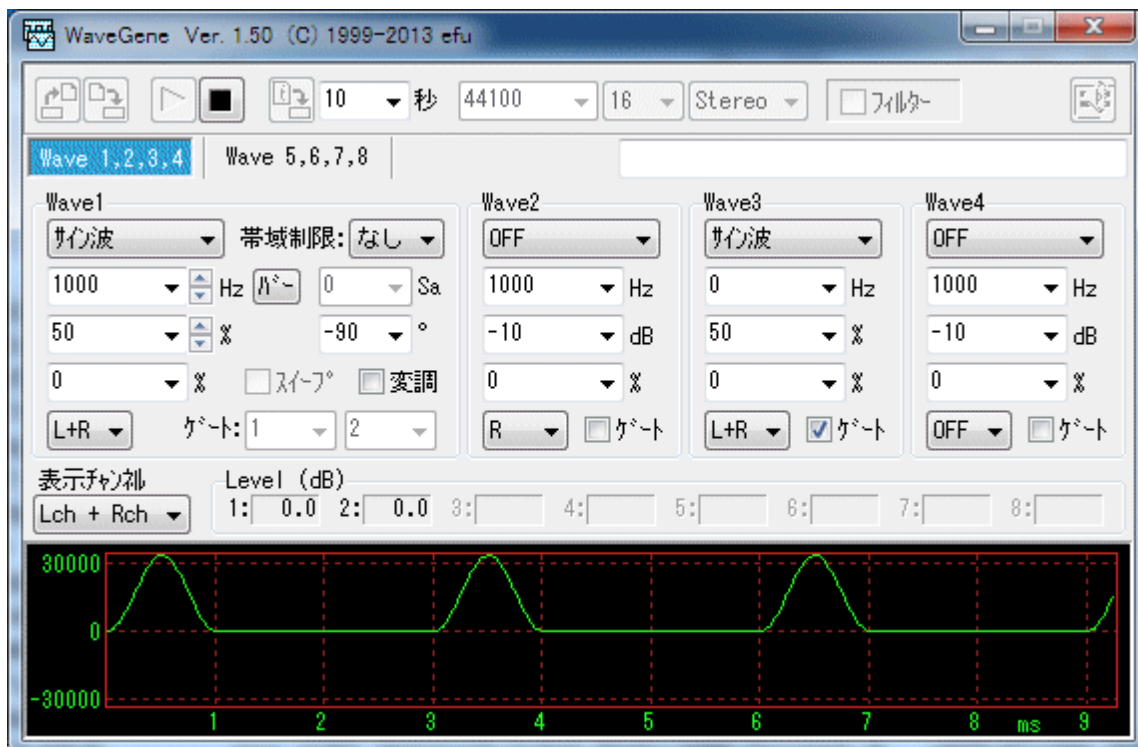
Sine wave and triangle wave



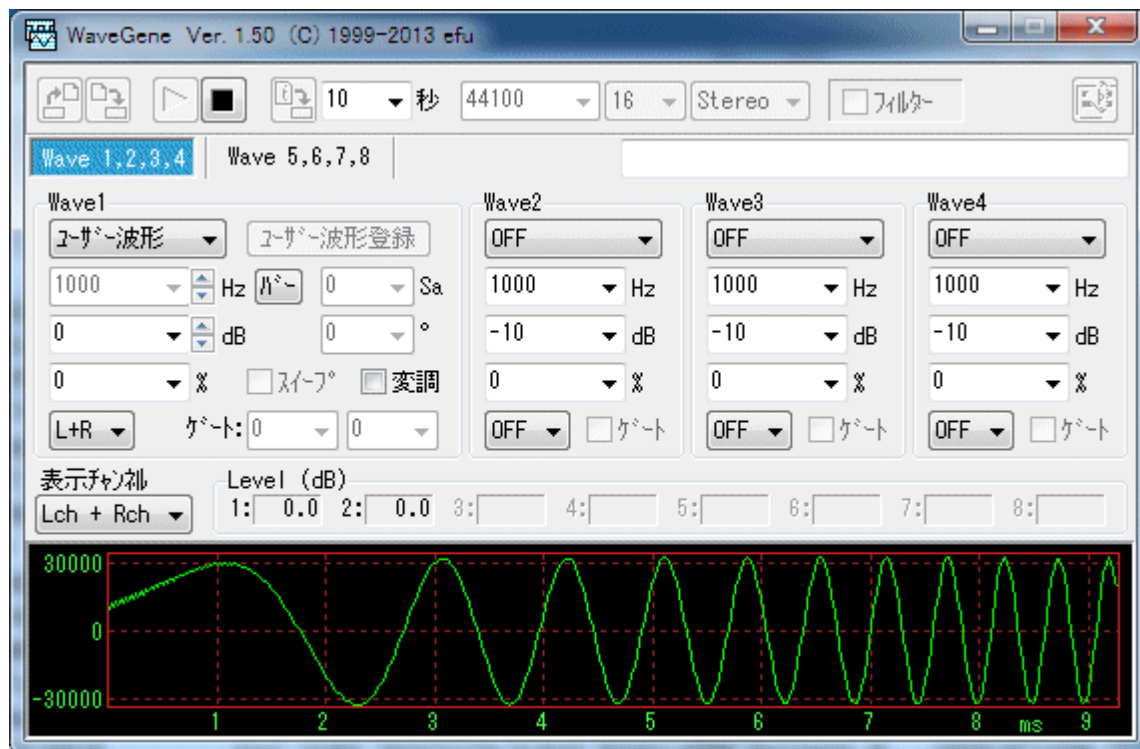
FM (frequency modulation)



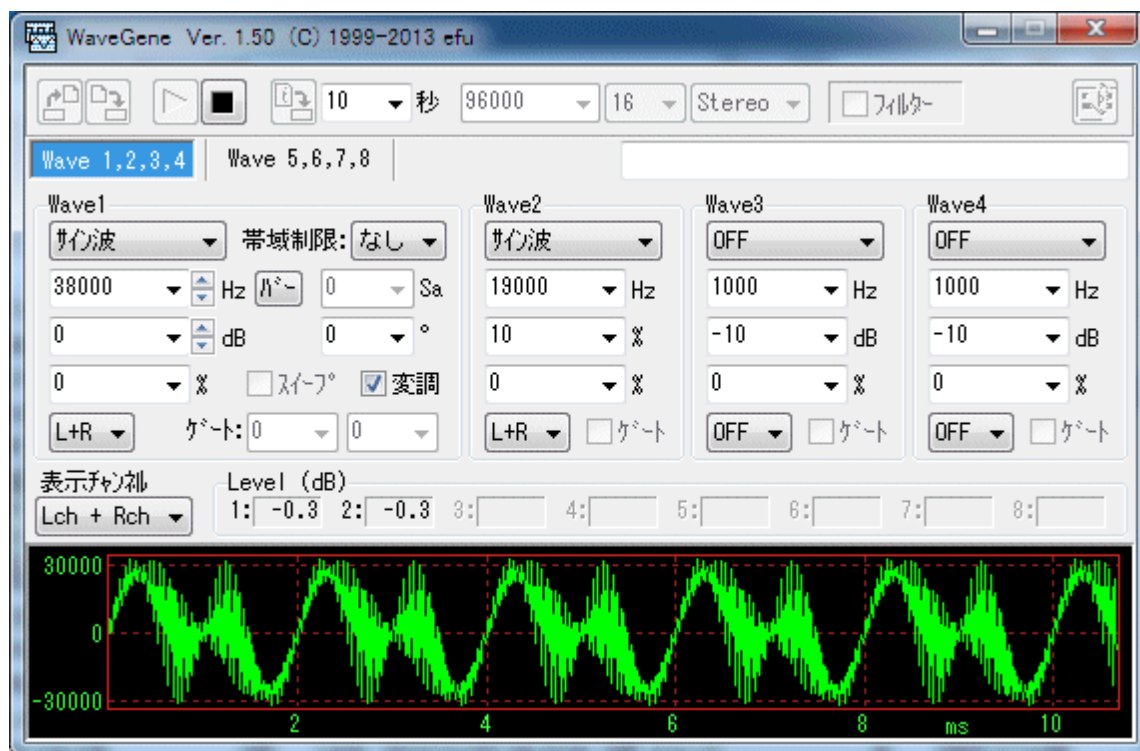
raised cosine burst



User waveform



FM stereo composite signal (L: 500 Hz R: 1000 Hz)



How to use

How to use

* In order to make the display screen compact as much as possible, explanatory characters on the screen are omitted as much as possible, so each button and the setting combo box The function will be displayed by stopping the mouse pointer for several seconds on each, so please check in advance.

In the explanation below, we will use the name of the button etc. displayed at that time.

Also, since you are using heavy popup menus with the right mouse button, please check by pressing the right button on the combo box or unit letters etc.

* Most numeric setting combo boxes except the combo box fixed exclusively for selection have direct values other than selecting the value of the previously set list
By writing You can enter " any value "

Reading / saving setting parameter file

-  **Read parameter button**

You can read the setting parameter file and recreate all saved parameters.

* By loading INIT.WG which was automatically created at the time of initial startup, you can restore all settings to the initial state.

* You can also open it by dragging and dropping a parameter file (*. WG) from Explorer etc.

-  **save parameter button**

All current parameters can be saved as setting parameter file.

You can reproduce it at any time by storing the current setting with a name and saving it.

When saving, it is convenient to write a comment for explanation in the edit box under the button.

(Comments and parameters are saved in the file)

Output to sound device

-  When you press the **output device button** to the sound device, Output the parameter signal directly to the sound device.

* The sound will not stop until  **stop button** .

Wave file / output to text file

- **Wave file length / sweep length combo box**  and the file length
Immediately after setting  **Immediately press the output button to Wave file**
By doing so, you can create a wave file.

Since a dialog box for specifying the file name opens, specify the file name and create the file.

Hold down the **Wave file output button** while pressing **Shift key** to output to a text file (*. WGT) can.

arbitrary value by selecting the preset value in the Wave file length / sweep length combo box or writing the direct value in the file length > .

* Wave File Length / Sweep Length You can also set the number of seconds in the combo box below the decimal point.

For example, when the sampling frequency is 48000 Hz If you want to create a Wave file of only 4096 samples, combo box It is possible to set it to 0.0853333. (4096 / 48000)

* The Wave file length / sweep length combo box is also used to specify the length of the sweep.

* An example of a text file format.

"WaveGene Text Format", 1.00 ← First line title and version

"Sampling Frequency", 48000 ← Sampling frequency

"Bits", 16 ← number of bits

"Channels", 2 ← number of channels

"Samples", 48 ← number of samples

"Max", 32768 ← Maximum value of data

"Data", 96 ← indicates that 96 amplitude data follow

0, 16422, 12539, 16422, 2317, 16422, 30273, 16422, 32767, -16422 ...

↑ Thereafter, 96 pieces of amplitude data (repeated every 10 rows per row)

In the case of Stereo, in order of Lch and Rch

Number of channels, number of samples, maximum value, amplitude data is the same as the text format of the user waveform.

[How to use user waveform](#) .

Select playback device

-  **Pressing the playback device button** opens a dialog box and uses it for playback You can select the sound device you want.
For details, please refer to [Playback Device Setting Dialog](#) .

End

- Termination, especially because there is no dedicated button, either the upper left system menu button or the upper right **Close button**  .

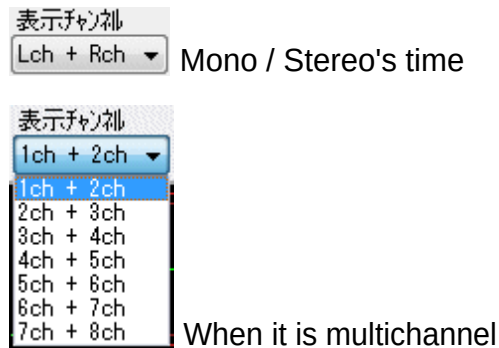
Waveform display

You can change the size of the waveform display by dragging the bottom edge of the window.

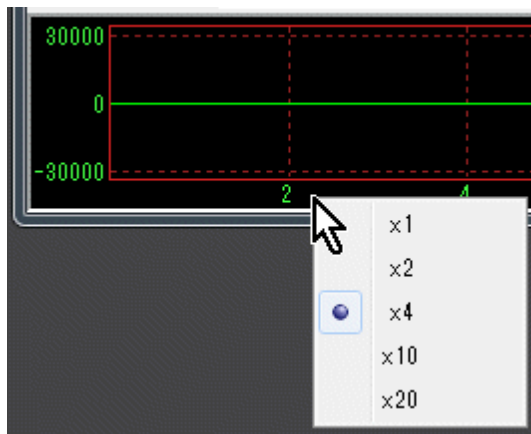
(Waveform display part can be expanded and contracted only to the lower side, it can not stretch sideways.)

The outputs of the left and right channels are displayed in different colors, respectively.

In multi-channel format, waveforms are displayed 2ch at a time, and can be switched by "display channel" combo box.



The display magnification of the horizontal axis (time) and the vertical axis (amplitude) can be changed in the popup menu displayed by clicking the right button of the mouse near the numeric display area of each scale.



Display magnification of horizontal axis



Display magnification of vertical axis

* The horizontal axis (time) range is the number of samples of the internal calculation buffer (filter length + 1) when the display magnification is x1.
For details, see [filter length](#).

* When the display magnification of the horizontal axis (time) is x2 or more, in order to make the flow of the waveform as much as possible and make it easy to see, the same thing as the trigger action of the oscilloscope is displayed.
(Trigger level: 0 Slope: +, the trigger source is Lch in the case of Stereo, the first channel of the display channel when it is 3 or more)

Also please be aware that it may not always stop depending on the waveform.

* The maximum value of the vertical axis is -127 to 128 when the bit length of the format is 8 bits, otherwise it is -32768 to 32767.

Even if it is 24 bits or more, please note that it will be converted into 16 bit value.

* In addition, the waveform display is only connecting the sample points of the PCM signal with a straight line for the sake of convenience.

Therefore, it is different from the analog waveform actually output from the D / A converter.

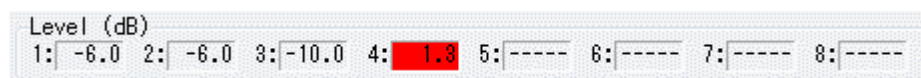
For example, in the case of a sine wave with a high frequency, there are few sample points, so it looks like a jaggy waveform, but the analog waveform actually output from the D / A converter is interpolated into a smooth sine wave.

In order to reproduce an actual analog waveform, it suffices to interpolate the waveform between sample points and draw it, but for now it does not support it.

* The output level is displayed in dB for each channel during signal output (including output to file).

At this time, when the output signal overrides the level, its background is displayed as " red ".

(If the value is zero, " ----- " will be displayed)



Created with the Personal Edition of HelpNDoc: [Free Qt Help documentation generator](#)

Format setting

Format setting

44100 ▼ 16 ▼ Stereo ▼

- **sampling frequency**

You can set **arbitrary value** by choosing a preset value in the combo box or writing a direct value .

* There is no particular limitation on the setting value if it is over 1000 Hz.

If you set a number less than 1000, it will be changed to 1000 after notifying with a warning tone.

The upper limit is up to 8 digits 99999999 (99.9 MHz) Hz from the top of the scale and input as it is now, apart from practicality.

* However, it depends on each environment whether it can play at the set frequency.
(It is optional when outputting to Wave file or text file)

At this time, if the setting frequency of the combo box of Wave 1 to Wave 8 exceeds 1/2 of the sampling frequency, the background of each combo box is " Red " and warn you.

Note: Input in this case may be changed to the lowest numerical value by checking the limit in real time when entering Please note that you will need some tips such as erasing unnecessary digits after entering again.

- **bit length**

Set 8, 16, 24, 32 or F32.

24 and 32 bits are 3 bytes / sample, 4 bytes / sample F 32 is an integer format of 24 bits of the mantissa part and 8 bits of the exponent part 32 bit normalized float (absolute value 1.0 or less) 4 bytes / sample Floating point format (IEEE_FLOAT).

- **Mono / Stereo / 3 ch ... 8 ch**

Set the number of channels.

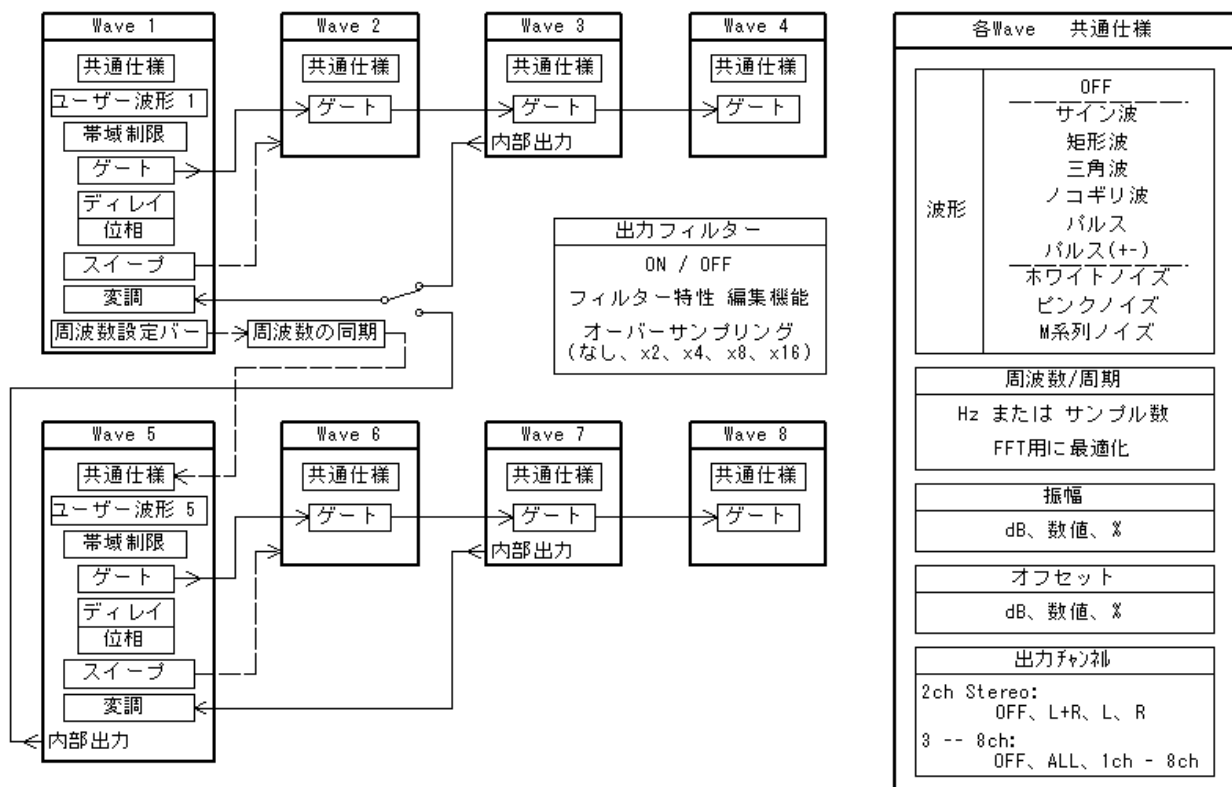
When it is 3ch or more, it becomes WAVE_FORMAT_EXTENSIBLE format which is an extended format supporting multi-channel multi-bit length.

Created with the Personal Edition of HelpNDoc: Qt Help documentation made easy

Setting Waveform Parameters

Setting Waveform Parameters

Waveform generator block diagram



*** Waveform generator (generator)**

- There are Wave 1 ~ Wave 8 waveform generators, and you can mix and output each

waveform output.

- I have another pair of Wave 1 ~ Wave 3 up to V1.40 plus Wave 4.
That is, Wave 1 and Wave 5, and Wave 2, 3, 4 and Wave 6, 7 and 8 are the same except for some.
For Wave 1 and Wave 5, a band limitation waveform generation function not included in V 1.40 has been added.

Wave 1 to 4 and Wave 5 to 8 are tabs Wave 1,2,3,4 Wave 5,6,7,8 | You can switch with.

The tab of the currently effective one is highlighted (the color changes), but the display of 1 to 4 and 5 to 8 are displayed in Wave 1 1/4~ Please be aware that it is the same except that there is a button.

- Besides this, oversampling and filtering with arbitrary characteristics can be multiplied and output.
Detail is [Output filter](#) Please refer to.

*** It is the setting group box of Wave 1 on behalf of**
If you click the position of a figure button, combo box, etc., it jumps to the explanation of that part.



==== Wave 1 ~ Wave 8 common specification ====

- Waveform** サイン波
If **OFF: OFF**, the function stops and becomes invalid regardless of other Wave settings. (From V1.50)
So it is a good idea to set **OFF** for unnecessary Wave. (Useless CPU load can be reduced)

periodic waveform: sine wave , square wave , triangle wave , sawtooth wave , pulse , pulse (+ -) .

Pulse (+ -) generates positive and negative alternating pulse trains. (From V1.50)

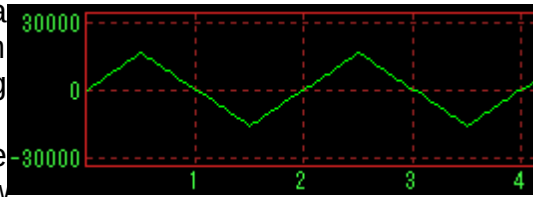
Noise: White Noise , Pink Noise , M Series Noise

* Note that the triangle wave and the sawtooth wave are consistent with the band limit waveform (see the function of only Wave 1 and Wave 5 - [band limitation](#)) Therefore, the starting point of the waveform has been changed as follows. (Start from the negative maximum value)

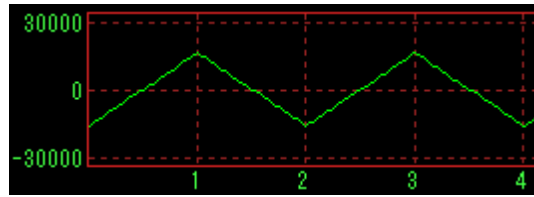
V1.40

V1.50

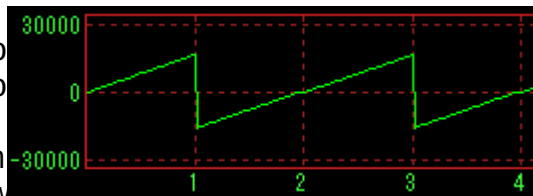
t
r
i
a
n
g
l
e
w
a
v
e
s
a
w
t
o
o
t
h
w
a
v
e



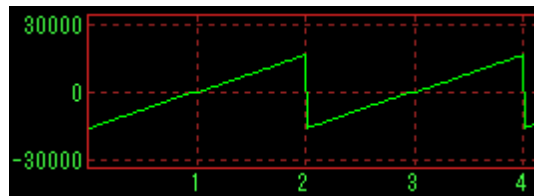
-> >



phase
difference
90°



-> >



phase
difference
180°

- frequency / period**

When the unit is **Hz**, the frequency is set as it is, and when the unit is **Sa**, it is set as the cycle according to the number of samples.

You can set it to an **arbitrary value** by selecting the preset value in the combo box or by directly writing the value.

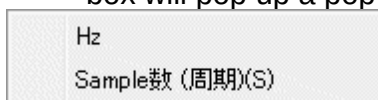
If the set frequency exceeds 1/2 of the sampling frequency, the background of each combo box becomes **"red"** and warns you.

In this case, the signal is generated without errors in particular. Although it does not become a correct signal with what is called "return", it can be used for purposes that make it on purpose after returning.

(No wrapping signal will be output if band limitation is set to anything other than "no" at Wave 1 or Wave 5)

* Unit setting / modification

Right clicking on the unit part such as **Hz** on the right of the frequency / cycle combo box will pop up a pop-up menu, so set it there.



Instead of right-clicking, you can also change the order by left-clicking on the unit part.

If you change the unit, the previous value will be converted and displayed in the new unit. (Even if you set it in any unit, it is the same internally.)

* Frequency setting extension

There is an extension function for special frequency setting.

See [Frequency setting extension](#) for details.

- **Amplitude**

When the unit is **dB**, when the maximum is 0 decibel value (negative) and the unit is **value**, it is directly up to 32767 (when 16 bits or more), or 127 (When 8 bit), and when the unit is **%**, it will be set with a maximum of 100 percentage value.

* However, this is the maximum value as a PCM signal, which is usually set as a reference value for clarity if you set it more than this, but the output filter It is also possible to set a value (positive) of 0 dB or more so that it can be amplified when it is known that the amplitude becomes obviously small, such as when multiplied. (V1.50 From Please check the level display and set it.

By selecting the preset value in the combo box or by directly writing the value **arbitrary value** can be set.

* Setting / changing units

Right clicking on the unit part such as **dB** on the right of the amplitude combo box will pop up a pop-up menu and set it there.



Instead of right-clicking, you can also change the order by left-clicking on the unit part.

If you change the unit, the previous value will be converted and displayed in the new unit. (Even if you set it in any unit, it is the same internally.)

When amplitude is input when the unit is **value**, you can also enter it in hexadecimal by adding "0x" before the numerical value.

Also, as you can see at the time of conversion from others, it is not necessary to be an integer.

(The same is true for hexadecimal numbers)

Note: When bit length of Wave format is switched, when the unit is dB and%, it is automatically converted to the maximum possible value internally, but when the unit setting is the value, the automatic conversion is It will not be done. Therefore, when changing from 8 bit to 16 bit, the sound becomes smaller, and there is a possibility of level over in 16 bit → 8 bit.

Note: For 24, 32 bit format

Actually outputted values are integers up to $2^{23} - 1$, $2^{31} - 1$ respectively, but for the sake of compatibility and ease of setup up to now, unit setting is **value** At the time of, the numerical value set in the combo box is 32767 in all cases as well as 16 bit.

(It's usually easier to set it with **dB** or **%**)

- **Offset**

Specifies the offset (DC component) to add to the waveform.

The specification method is exactly the same as the amplitude.

Note:

Even if the waveform is **OFF** (invalid) only when using this wave as a modulation signal, only this offset is effective as an internal signal and can be used as a modulation signal.

(Output is not done regardless of the output channel setting.)

- **output channel**

Set the channel to output the signal generated by this wave.

In Stereo (Mono), both channels are output by **L + R** , left channel by **L** , only to right channel by **R** I will.

In the case of multi channel with 3ch or more, **ALL** on all channels, **1 ch** , **2 ch** , **3 ch** ... It is output to each channel only.

OFF: **OFF** will not output anywhere.

==== Wave 1 and Wave 5 only features ====

- **User Waveform**

Wave 1 and Wave 5 can also generate and generate arbitrary waveforms (Wave format and original text format) as user waveforms.

For details, please refer to [How to use user waveform](#) .

- **Bandwidth limit**

In the periodic waveforms of Wave 1 and Wave 5, it is possible to generate a periodic waveform with [band limit](#) without alias distortion.

See [Bandwidth Limit](#) for details.

- **Smooth change in frequency / cycle and amplitude (volume) with mouse wheel**

Using the Sweep function of Wave 1 and Wave 5, it became possible to smoothly and continuously change the frequency and amplitude (volume) with the mouse wheel, and it approached the operability of the analog oscillator. (V1.50 From

With the mouse cursor on the frequency / period combo box  of Wave 1 or Wave 5 and turning the mouse wheel , Frequency, or period, and you can change the volume by rotating the mouse wheel over the amplitude combo box

 I will.

To be exact, the mouse wheel will operate on the up / down control on the right of each combo box as well.

* For these mouse wheel operations, press the up and down buttons of each up / down control  Please note that it is exactly the same.

The change amount and change time when turning the mouse wheel can be changed by each up / down control setting.

For details, refer to the following up / down control section.

- **Frequency / period Up Down control**

For Wave 1 and Wave 5, you can also change the value with the up / down control to the right of the frequency / period combo box.

You can freely change the amount of change and change time in the dialog displayed by right-clicking the mouse on the up / down control.



- Change amount
You can change it with **number** or **%** .
Set the change amount to **arbitrary value** by selecting the preset value in the combo box or by directly writing the value You can do.

For example, if the unit is frequency, % and setting the value to 5.94630943592953 (the second root of 2), you can change it semitone by semitone.

Note: If you change 1000 Hz by%, if it is up by 10% It becomes 110% of 1100 Hz, but in case of down 90% not 900 Hz, $1000 / 1.1 = 909.090909090909$ Hz Please be careful.

It will be calculated up / down and back.

- Change time
The change time is the time to change the change amount of the above, and it can set an arbitrary value (up to 100 seconds).

If the change time is set to something other than 0, it multiplies that time and makes a continuous change like a sweep.

If the change time is set to 0, continuous change does not occur and it becomes step operation.

Also check the **Reset buffer** checkbox, which will be enabled at that time, so that the buffer will be cleared every time it changes and will change after a moment of interruption. (It works the same as V 1.40.)

If you want to change while keeping the phase relation between each wave, it is good to set like this.

The way of change is Log (logarithmic) if the **Log** checkbox is checked when the change time is not 0, it changes linearly if it is not checked.

However, at the time of setting a general small change rate, neither should be particularly different.

It is effective when setting a bigger change as shown below.

For example, when the current oscillation frequency is 20 Hz, setting the change amount to 19980 and the change time to 60 seconds and pressing the up button once takes 60 seconds It will sweep from 20 to 20000 Hz.

Linear sweeping is also possible with Log / Linear setting at this time.

Sweep operation is possible even if it is not the original sweep setting (described later).

- **Frequency setting bar**

Frequency setting bar button to the right of Wave 1 only, frequency / period combo box to open the [frequency setting bar](#) window to change or sweep You can also do.

Furthermore, it is possible to change the frequency of Wave 5 synchronously at an arbitrary ratio.

See [frequency setting bar](#) for details.

- **Amplitude Control**

For Wave 1 and Wave 5, you can also change the value with the up / down control on the right of the amplitude combo box.

Change amount and change time can be freely changed in the dialog displayed by right clicking the mouse on the up / down control.



Usage is the same as above frequency / cycle.

- **Number of delay samples**

You can delay the occurrence of Wave 1 and Wave 5 from the others by the set number of samples.

(The waveform is shifted to the right by the number of samples, and the value of the waveform in the delayed part is all zero.)

You can set **arbitrary value** by choosing a preset value in the combo box or writing a direct value .

- **Phase**

Advance the phases of Wave 1 and Wave 5 from the others by the set frequency (0 to 360°).

(The waveform shifts to the left by that amount)

You can set **arbitrary value** by choosing a preset value in the combo box or writing a direct value .

These two are valid only with periodic waveforms (sine wave, rectangular wave, triangular wave, sawtooth wave, pulse, pulse (+ -)).

- **sweep** ☐

By checking this check box you can activate the sweep function.

Three of frequency, amplitude, and phase can be swept at the same time, but

For Wave 1: Wave 1 setting is initial value , Wave 2 setting is final value

Wave 5 setting is initial value, Wave 6 setting is final value for Wave 5:

For example, if the frequency of Wave 1 is 100 Hz amplitude -10 dB, Wave 2 If the frequency is 1000 Hz and the amplitude is 0 dB, the sound changes from 100 Hz to 1000 Hz as the sound increases.

(Therefore, please set the same settings for Wave 1 and Wave 2, or Wave 5 and Wave 6 for objects that you do not want to change in reverse)

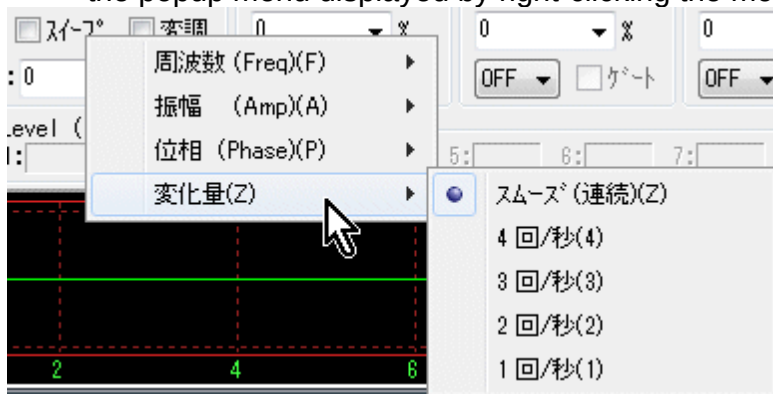
* Final Values For Wave 2 and Wave 6 settings, only the frequency and amplitude values are used, the rest are irrelevant to the sweep operation itself.

So if you do not need it, you can leave the waveform **OFF**.

Actually when you want to use the waveform of Wave 2, Wave 6 Please enable and set the output destination.

Sweep time (seconds), " Wave file length / Sweep length " Specify by combo box.
(Even when outputting to a sound device that normally does not stop until you press the stop button, the sound will stop after sweeping at the set time here.)

How to change the sweep (linear, Log, continuous change, step change) are set in the popup menu displayed by right-clicking the mouse on the check box.



* There is no Wave 2 or Wave 6 setting for only the phase, so the final value is also set in the popup menu.

(Value of Wave 1 -> 360° or Wave 1 value -> 0°)

(Wave 5 value -> 360° or Wave 5 value -> 0°)

Also, the phase change is linear only.

* It is also possible to reduce the value of Wave 2 (Wave 6) rather than Wave 1 (Wave 5) in the frequency and amplitude, and change it in the opposite direction.

* Even sweeping of the amplitude is possible even in the case of white noise, pink noise, M series noise.

Since the user's waveform is handled differently from others, it can not be swept.

- **Modulation** ☐ 変調

If you select this check box, you can activate the modulation function.

The modulation wave (modulation signal) is fixed at Wave 3 or Wave 5 in Wave 1, and Wave 7 in Wave 5.

(In V 1.40, the modulation wave of Wave 1 was Lch output of Wave 3, but it was changed.)

* When Wave 5 uses a user waveform as the modulation signal of Wave 1, it is possible to select a signal to be used as a modulation signal separately from the normal output destination selection.

See [user waveform](#) for details.

The type of modulation (AM, DSB, FM, PM, PWM) is set in the popup menu displayed by right-clicking the mouse on the check box. (PWM From V1.50)
In Wave 1, Wave 3 or Wave 5 can be selected with the lowest " modulation signal ".



AM (Amplitude Modulation Amplitude Modulation)

When the amplitude of the modulated wave (Wave 3/5 or Wave 7) is set to 100%, the degree of modulation of the carrier wave (carrier Wave 1 or Wave 5) is 100% .

For example, if the amplitude of the carrier wave Wave 1 is set to 50% and the amplitude of the modulation wave Wave 3 is set to 100%, the peak of the output becomes the maximum amplitude (0 dB).

DSB (Double Side Band Carrier Suppression Both Sideband Amplitude Modulation)

When the amplitude of the modulated wave (Wave 3/5 or Wave 7) is set to 100%, the degree of modulation of the carrier wave (carrier Wave 1 or Wave 5) is 100% .

For example, if the amplitude of the carrier wave Wave 1 is set to 100% and the amplitude of the modulation wave Wave 3 is set to 100%, the peak of the output becomes the maximum amplitude (0 dB).

* Mathematically it is a product of the carrier wave and the modulation wave, so when the amplitude of the modulated wave is 0%, the output also becomes zero.

FM (Frequency Modulation Frequency Modulation)

When the amplitude of the modulation wave (Wave 3/5 or Wave 7) is set to 100%, the frequency deviation of the carrier wave (carrier Wave 1 or Wave 5) is $\pm 100\%$.

For example, if setting the frequency of the carrier wave Wave 1 to 1000 Hz and setting the amplitude of the modulation wave Wave 3 to 100%, the output frequency changes from zero (direct current) to 2000 Hz.

PM (Phase Modulation Phase Modulation)

When the amplitude of the modulation wave (Wave 3/5 or Wave 7) is set to 100%, the phase shift of the carrier wave (carrier Wave 1 or Wave 5) is $\pm 360^\circ$.

PWM (Pulse Width Modulation Pulse Width Modulation)

When the amplitude of the modulated wave (Wave 3/5 or Wave 7) is changed from the positive maximum value to the negative maximum value, the duty cycle (duty ratio) of the carrier wave (carrier Wave 1 or Wave 5) It will be 100% ~ 0%.

Actually, however, it is limited to 99% to 1%.

When the modulation wave has zero DC level, the duty cycle of the carrier wave is 50%.

The relationship between the modulation wave amplitude (DC level) or offset and duty cycle is as follows.

amplitude	+100%	+98%	+96%	+90%	+80%	+70%	+60%	+50%	+40%	+30%	+20%	+10%	0
DUTY	(99%)	99%	98%	95%	90%	85%	80%	75%	70%	65%	60%	55%	50%

DUTY = amplitude / 2 + 50 [%]

amplitude	0	-10%	-20%	-30%	-40%	-50%	-60%	-70%	-80%	-90%	-96%	-98%	-100%
DUTY	50%	45%	40%	35%	30%	25%	20%	15%	10%	5%	2%	1%	(1%)

*The signal that can be used as a carrier wave is a periodic waveform (rectangular wave, triangular wave, sawtooth wave, pulse, pulse (+ -)) other than sine wave. (However, it is not possible to change the duty cycle of the band limit waveform by addition synthesis.)

* When the number of samples of the cycle of the carrier wave is small (frequency is high), the setting of the duty cycle becomes difficult to reflect, as a matter of course.
(To set it to 1 or 99, it will naturally not be reflected unless the period is 100 or more)

Carrier limit

There are no restrictions other than white noise, pink noise, M series noise, user waveform in the FM, PM and PWM carrier types can not be used for carrier type.

There is no limitation on the type of modulation wave (Wave 3/5 or Wave 7) at all.

* Therefore, if AM, DSB, it is also possible to modulate the user waveform of Wave 1 with the wave form of Wave 5.

Modulation wave

The signal used as a modulation wave is the internal output of Wave 3/5 or Wave 7, not related to each output setting.

Therefore, you can leave the output channel setting **OFF** when there is no need to output.

Conversely, if you want to use it, you can set the output.

(The output of Wave 3 was used as a modulated wave and was not mixed as output, but it was changed up to V 1.40 when the modulation function was enabled.)

- **Gate (for tone burst)**

Tone burst can be output by setting optional On period (wavenumber) and Off period.
(Naturally, only the periodic waveform is the target)

Both On cycle and Off cycle can be entered in units of 0.5.

For example, if the On period is 0.5 and the Off period is 9.5, only the first half period On, followed by zero for the 9.5 period, full period A tone burst of 10 is output.

It is effective only when you set a value other than zero for both On cycle and Off cycle.

By selecting the preset value in the combo box or by directly writing the value **arbitrary value** can be set.

- * You can control the output of Wave 2, Wave 3, Wave 4 by the gate operation of Wave 1.
 - * You can control the output of Wave 6, Wave 7, Wave 8 by the gate operation of Wave 5.
- See below.

==== Functions of Wave 2, 3, 4 or Wave 6, 7, 8 only ====

- **Gate (synchronized with the gate of Wave 1 or Wave 5)** ☐ ゲート

When this check box is checked, in synchronization with the gate operation in Wave 1 or Wave 5, Wave 2, 3, 4 or Wave 6, 7, 8 will also be turned on / off.

You can check only when the gate operation of Wave 1 or Wave 5 is enabled.

- * In addition to using common waveforms as Wave 2, 3, 4 or Wave 6, 7, 8, you can also make pulses of the gate signal itself by turning On / Off Offset I can do it.

- * All the setting functions can be changed in real time even during signal generation.

~~~~~  
Depending on the setting, however, it may become invalid and can not be changed during signal generation.

Also, when parameters such as frequency are changed during operation, there may be cases where the phase between each wave is shifted or lost due to the original operation, so especially when the phase relation is important, sweep, modulation In case of, please make a change after stopping once.

---

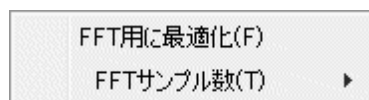
Created with the Personal Edition of HelpNDoc: [Free HTML Help documentation generator](#)

---

## Extension of frequency setting

### Extended function of frequency setting

Frequency / period combo box  Hz when setting the frequency (when the unit is **Hz** Clicking the right mouse button displays a pop-up menu for the extended frequency setting function.



#### Optimized for FFT

- By selecting this, sister software WaveSpectra For convenience when used with spectrum display software such as FFT It is possible to convert accurately to the value closest to the original setting value at the frequency where no error for data extraction is generated (only the spectrum becomes one) at the time of

That is, the minimum frequency resolution of the FFT "sampling frequency / Data length " and is converted to the closest one.

**Example:** Original setting: 1000 Hz, 44100 Hz sampling, 4096 point FFT  
 $44100/4096 = 93$  times of 10.76660 Hz ----> > 1001.294 Hz

**Note:** In order to obtain a spectrum with no cropping error as shown above with waveforms other than sine wave, this is not enough, Is required.  
For more information,

---

[Please refer to](#)

in order to eliminate cutting error in FFT with waveform other than sine wave.

### FFT sample number

- Set the sample data length of the FFT to be used for observation, for optimization above.  
(2048 to 131072)

\* Only one spectrum is observed when observing with FFT above this value.  
(In this case, the window function of FFT is " none " In other words, it becomes so if it is a rectangular window)

### Multiples of Wave 1 or Multiples of Wave 5

- This menu is added for Wave 2, Wave 3, Wave 4 and Wave 6, Wave 7, Wave 8.  
This allows the frequencies of Wave 2, 3, 4 to be set to an exact multiple (harmonic, harmonic) of the frequency of Wave 1, or the frequencies of Wave 6, 7, 8 to Wave 5 It can be set to an exact multiple (harmonic, overtone) of the frequency of.  
(X 2, x 3, x 4, x 5, x 6, x 8, x 10)

\* All setting functions can be changed in real time even during signal generation.

~~~~~

Created with the Personal Edition of HelpNDoc: [Benefits of a Help Authoring Tool](#)

Frequency setting bar

Frequency setting bar

Only for Wave 1, "Frequency setting bar" on the right side of the frequency / period combo box  by pressing the button , the following frequency setting bar window is displayed and the position of the knob of the track bar.

You can change the frequency and sweep by changing it.

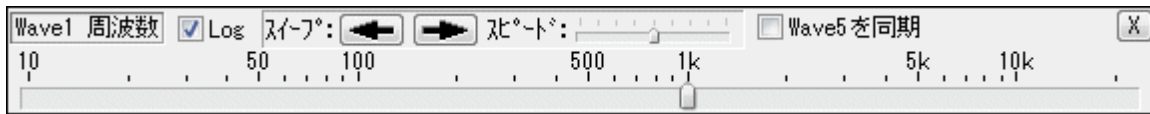
* You can also change the frequency of Wave 5 synchronously with the frequency of Wave 1 and sweep (to be described later)

Note:

"Frequency setting bar " button  will not take effect unless the frequency / period unit of

Wave 1 is frequency, that is, **Hz** .

Also, changing the unit from **Hz** to **Sa** (number of samples) when the frequency setting bar window is displayed will close the frequency setting bar window.



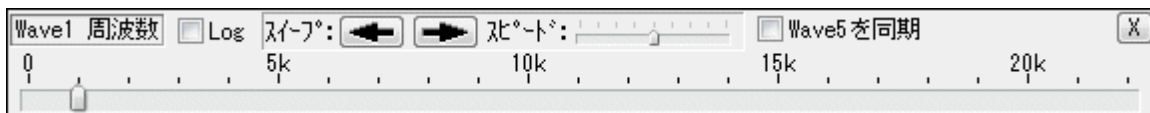
* You can also change the frequency by rotating the mouse wheel on (anywhere on) this window.

This is exactly the same as turning the mouse wheel over the Wave 1 frequency / period combo box Hz in the main window.

Similarly, the setting of the change amount and change time follows the setting of frequency / cycle up / down control of Wave 1.

The same is true when changing with the knob of the track bar of this frequency setting bar, and changes according to the setting of frequency / cycle up / down control of Wave 1.

If you uncheck the **Log** check box ☐ Log , the display on the frequency axis becomes a linear scale I will.



Frequency setting bar Since the width of the window can be changed freely, it can be used widely.



(Height can not be changed)

The track bar is synchronized with the display of the frequency / period combo box of the frequency of the main screen, and it will be the same regardless of the change.

* However, since there is a limit to the resolution of the position specification of the track bar and it can not be set lower than that, if you want to set the frequency to the exact frequency, Frequency / period combo box Hz must be set.
(The higher the sampling frequency, the lower the resolution, so the gap will increase.)

By clicking the left and right area of the knob of the track bar, you can change a small amount in that direction.

Sweep

By pressing the **Sweep** button in the horizontal direction



sweep operation can be done.

- Press the left arrow button to sweep to the lower frequency, and the right arrow sweep to the higher frequency.
Since the button is On / Off operation, the sweep operation will remain as it is on.
During the sweep operation, the color of the On button changes to  .

When the lower limit or the upper limit of the frequency scale is reached, the sweep stops automatically.

If you press a button in the direction opposite to the button that is turned on during sweep execution, the sweep direction is reversed.

- Log sweep when the frequency axis display is Log, linear sweep when it is linear.
- Move the position of the knob on the right **speed** track bar, you can change the speed of the sweep.
Moving to the right increases the sweep speed, moving it to the left slows the sweep speed.

When the knob is in the center, it is set to sweep the entire range from the lowest (left) to the highest (right) of the track bar in about 2 minutes.
(Depending on the PC's timer precision, some errors may be generated.)

Sweep speed is increased 1.5 times, 2 times, 2.5 times, 3 times each time the knob is moved to the right by one
Sweep speeds of 1 / 1.5, 1/2, 1 / 2.5, 1/3 every time you move the knob one position to the left
Become.

- **Note:** The sweep operation, that is, the movement of the knob on the track bar, moves regardless of the actual Wave 1 operation.
In other words, even if there is no sound, turning the sweep button On will move the knob.
- Only the sweep operation by this button, the setting of the frequency / cycle up / down control of the main window is irrelevant, and it always sweeps smoothly.
- By dragging the area other than the knob of the track bar to the left and right, the amount is small, but it becomes a sweep toward that direction.
(Faster sweeping will be achieved if you drag fast)

Sync Wave 5

Check the **Sync Wave 5** check box ☐ Wave5を同期 to turn it On, the frequency of Wave 5 is set to Wave 1 You can change it synchronously with the frequency of.

Note:

Synchronize Wave 5 checkbox ☐ Wave5を同期 means that the frequency / period unit of Wave 5 is frequency, that is, **Hz** Unless it is not valid.

- If you check the check box and set it to On, the following dialog box will be displayed

first.



Here, how to change the frequency of Wave 5 with respect to the frequency of Wave 1 is set.

* If you select the upper **Wave 1 + number** , the frequency of Wave 5 will always be the frequency of Wave 1 plus the number you set here. (Equality difference change)
If Wave 1 is 1000 Hz and the number is 100, Wave 5 will be 1100 Hz.
When Wave 1 changes to 2000 Hz, Wave 5 becomes 2100 Hz.
The number can be a negative number.

* If you select the lower **Wave 1 x number** , the frequency of Wave 5 will always be the frequency of Wave 1 multiplied by the number you set here. (Geometric change)
If Wave 1 is 1000 Hz and the number is 1.5, Wave 5 becomes 1500 Hz.
When Wave 1 changes to 2000 Hz, Wave 5 becomes 3000 Hz.

The numerical value can be set to **arbitrary value** by selecting the preset value in the combo box or directly writing the value can.

When closing the dialog with **Close** button, the frequency of Wave 5 is changed to the frequency set here.

- State checking the **Sync Wave 5** check box ☒ Wave5を同期 , the frequency of Wave 5 It will always change with the ratio set in the dialog box above for Wave 1 frequency.

Changes due to the position of the knob of the track bar, change due to rotation of the mouse wheel, sweep by the sweep button, furthermore the frequency / cycle of the main window Changes due to combo box 1000 Hz , changes due to rotation of the mouse wheel on the combo box, Following, the frequency of Wave 5 changes.

Note: Only in the main window ☐ スイッチ .

- When the **Sync Wave 5** check box is checked ☒ Wave5を同期 , Right clicking on it will display the setting dialog again and you can change the ratio of change.

Simply left click will clear the check and cancel the sync state.

- Explanation of the preset value of " Wave 1 x numerical value "
When changing by equal ratio, several musical ratios are set to the preset value of " Wave 1 x numerical value ".
When these numbers are set, they always change their ratios while maintaining their respective ratios.
Numbers with () are not preset. Please set it if necessary.

	12 Equal Temperature	Genuine law
--	----------------------	-------------

short two degrees	(1.059463094359295)	(1.066666666666667)
long twice	(1.122462048309372)	(1.125)
short three degrees	1.189207115002721	(1.2)
long three degrees	1.259921049894873	(1.25)
Completely four times	1.334839854170034	1.333333333333333
Four degrees increase	(1.414213562373095)	(1.40625)
Perfect five times	1.498307076876681	1.5
Short six degrees	(1.587401051968199)	(1.6)
Longitudinal six degrees	(1.681792830507429)	(1.666666666666667)
Short seventh	(1.781797436280678)	(1.777777777777778)
longitude season	(1.887748625363386)	(1.875)
Octave	2	2
short three degrees down	0.840896415253714	(0.833333333333333)
Long three degrees down	0.793700525984100	(0.8)
Completely four times under	0.749153538438341	0.75
Completely five times down	0.667419927085017	0.666666666666667
Octave below	0.5	0.5

* In the equal temperament it will be the power of 2 12 's root 1.05946

* According to the conditions of pure law it may differ from the above values.

* To close this window, press the  button in the upper right.

You can also close it with the ESC key.

You can also close it by pressing the " Frequency setting bar " button  in the main window.

Created with the Personal Edition of HelpNDoc: [Free help authoring environment](#)

Band limit

Band limit

V1.40 , Periodic waveforms other than sine waves generated simple graphical waveforms. This is useful for testing hardware and software, as a modulation signal, but it was not correct for use (listening) as a voice signal.

It generates so-called aliasing (folding noise, aliasing) because it contains components above the sampling frequency / 2 (Nyquist frequency) that should not be present as an audio signal.

Beginning with V1.50, band-limited periodic waveforms without them (**square wave** , **triangle wave** , **sawtooth wave** , **pulse** , **pulse (+ -)**).

When using it as an audio signal, please use this band-limited waveform.

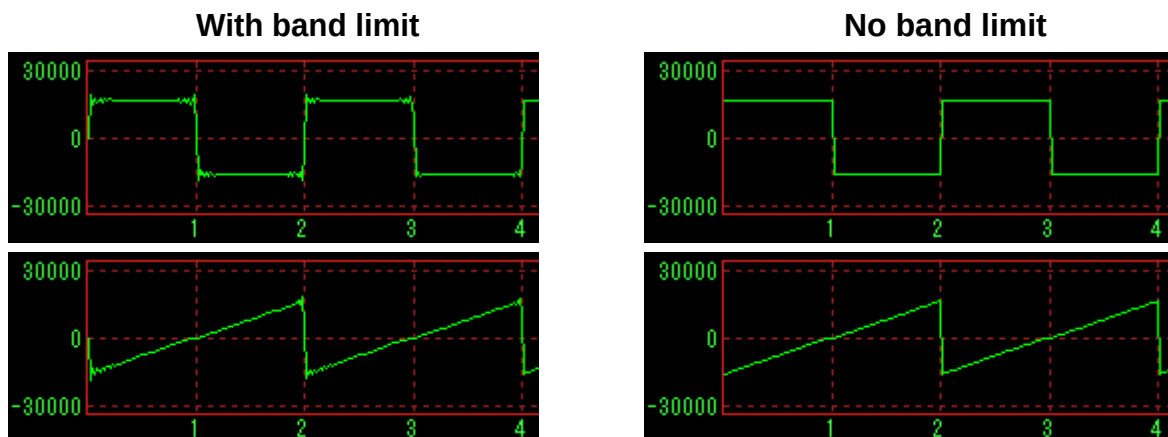
In order to generate band-limited periodic waveforms, **band limit** combo box of Wave 1 or Wave 5 **帯域制限:** なし ▼ **None** Set it other than.

Currently, it is possible to use two kinds of addition synthesis (Additive Synthesis) and BLIT synthesis (Band Limited Impulse Train Synthesis).



* Although it does not synthesize, even if you are generating a sine wave, if you set the band limit combo box to something other than **none**, no signal will be output if you set a frequency above the Nyquist frequency.

* Since the amplitude is increased by the vibration waveform due to the so-called Gibbs phenomenon compared to the signal which is not band-limited in the band-limited signal, when the amplitude setting is generated at 0 dB, the level is normally over. Therefore it is good to keep it to about -3 dB.



Especially when sweeping, it is necessary to lower it further for the reason of Note below.

* It is possible to sweep even band-limiting signals and change the duty cycle by PWM. (Changing duty cycle is only for BLIT synthesis)
However, the following cautions are necessary.

note:

When the frequency of the band limit signal is continuously changed, amplitude discontinuity occurs because the harmonic (harmonic) component suddenly disappears (increases).

Particularly in the case of BLIT synthesis, because the amplitude of the band limitation pulse becomes discontinuous and integrating it, it is possible that the waveform jumps greatly depending on the condition (shock sound may be generated depending on circumstances) depending on the condition please.

Addition synthesis

- Additive synthesis is a method of adding literally harmonic (harmonic) components up to the Nyquist frequency of that waveform and synthesizing them.

It approximates by Fourier series expansion of the waveform.

Accurate synthesis is possible, but if the fundamental frequency is low, the number of components to be synthesized will be large, so it will take much computation time.

- Combining formulas for each waveform with the amplitude correction value omitted is as follows.

Rectangular wave:

$$\sin(x) + 1/3 * \sin(3x) + 1/5 * \sin(5x) + 1/7 * \sin(7x) \dots$$

Triangular wave:

$$\cos(x) + 1/9 * \cos(3x) + 1/25 * \cos(5x) + 1/49 * \cos(7x) \dots$$

Sawtooth wave:

$$\sin(x) + 1/2 * \sin(2x) + 1/3 * \sin(3x) + 1/4 * \sin(4x) \dots$$

pulse:

$$\cos(x) + \cos(2x) + \cos(3x) + \cos(4x) \dots$$

Pulse (+ -):

$$\cos(x) + \cos(3x) + \cos(5x) \{ \cos(7x) \dots$$

note:

You can not change the duty cycle by PWM by adding synthesis.

Also, there are differences in starting point between other waveforms in square wave and sawtooth wave.

(Please see the bottom [difference in starting point of addition synthesis](#))

- In consideration of computation time and practical accuracy, we currently synthesize harmonics up to the 1000th order.
(It is up to the 1000th even with rectangular waves including only odd harmonics.)
Therefore, if the fundamental frequency is 20 Hz, harmonics up to 20 kHz will be synthesized.
If the sampling frequency is 48 kHz, there are bands up to 24 kHz, but only 20 kHz is synthesized.

In the case of sounding in real time, if the CPU is not powerful, it may take time to calculate if it is a very low frequency and sound skipping will occur.

In that case please write it to the wave file once and use it.

(Currently the program is not particularly speeded up, but it may be faster in the future)

BLIT synthesis

- BLIT synthesis is a method of creating a band-limited impulse train (Band Limited Impulse Train), integrating it and synthesizing the waveform.
Since the band-limited impulse is used, the synthesized waveform is also band-limited.

However, because there are fundamental errors due to the fact that the amplitude of

the pulse is constant and unchanged, and instability to integrate, although the harmonic number included in the band limitation pulse varies depending on the synthesized frequency, is inferior in terms of error and stability. It may become unstable when sweeping or when the duty cycle is not 50%.

However, since the calculation time is the same regardless of the fundamental frequency, it is different from addition synthesis. Even when the fundamental frequency is very low, it is possible to synthesize signals containing all the harmonics in the band.

- Pulse and pulse (+ -):
These are created directly as positive and negative BLIT.

Rectangular wave:

Integrating alternating + - impulses (and removing the DC component) produces a square wave.

(Actually, for sake of stability when changing the duty cycle, sawtooth wave with different phase from positive and negative pulses is created, and the difference is taken as a rectangular wave.)

Triangular wave:

A triangle wave is obtained by integrating the rectangular wave again.

Sawtooth wave:

Integrating a positive pulse train with a negative constant proportional to the cycle produces a sawtooth wave.

* The duty cycle can be changed by PWM for square wave, triangle wave, pulse (+ -).

- **note:**
As mentioned above, this method creates waveforms using integration, so it may become unstable depending on conditions.
(The waveform goes somewhere)
Although devised so as to make stability and waveform accuracy compatible by correction and incomplete integration by an appropriate time constant, depending on the correction of the difference in the amplitude of the pulse by the fundamental frequency, it is inevitably unstable depending on the initial value of the integration. It may become.

When sweeping, stability is improved by decreasing the time constant of incomplete integration, but the waveform slightly distorts by that amount.
(Tilt it)

Even when not sweeping, it is difficult to obtain an accurate initial value in principle when the duty cycle is set to something other than 50%, so the waveform may jump greatly at the start.

In that case just a few need be enough, please change the duty cycle or fundamental frequency.

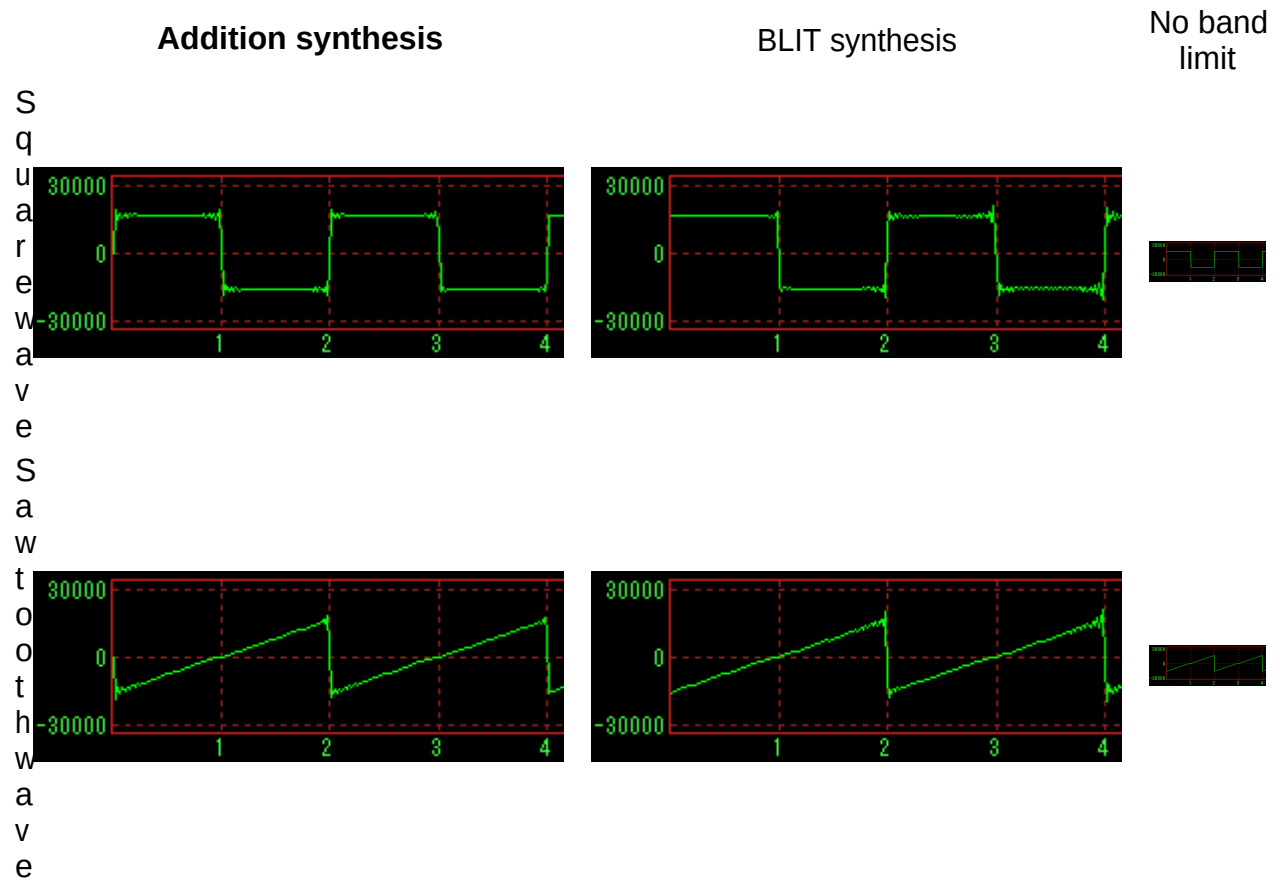
In most cases this should be solved.

- References:
Alias-Free Digital Synthesis of Classic Analog Waveforms

[Tim Stilson, Julius Smith, Music Department, Stanford University]
<https://ccrma.stanford.edu/~stilti/papers/blit.pdf>

Difference in starting point of addition synthesis

- In addition synthesis, the start point differs from other waveforms, "zero" It starts from the point of. (1 sample different)



Created with the Personal Edition of HelpNDoc: [Easily create iPhone documentation](#)

Output filter

Output filter

The created signal can be output after applying a filter (FIR filter) with arbitrary frequency characteristics.

You can also create a signal with an integral multiple of the specified format and then filter it. (Oversampling)

* You can also change the characteristics of the filter during signal generation.

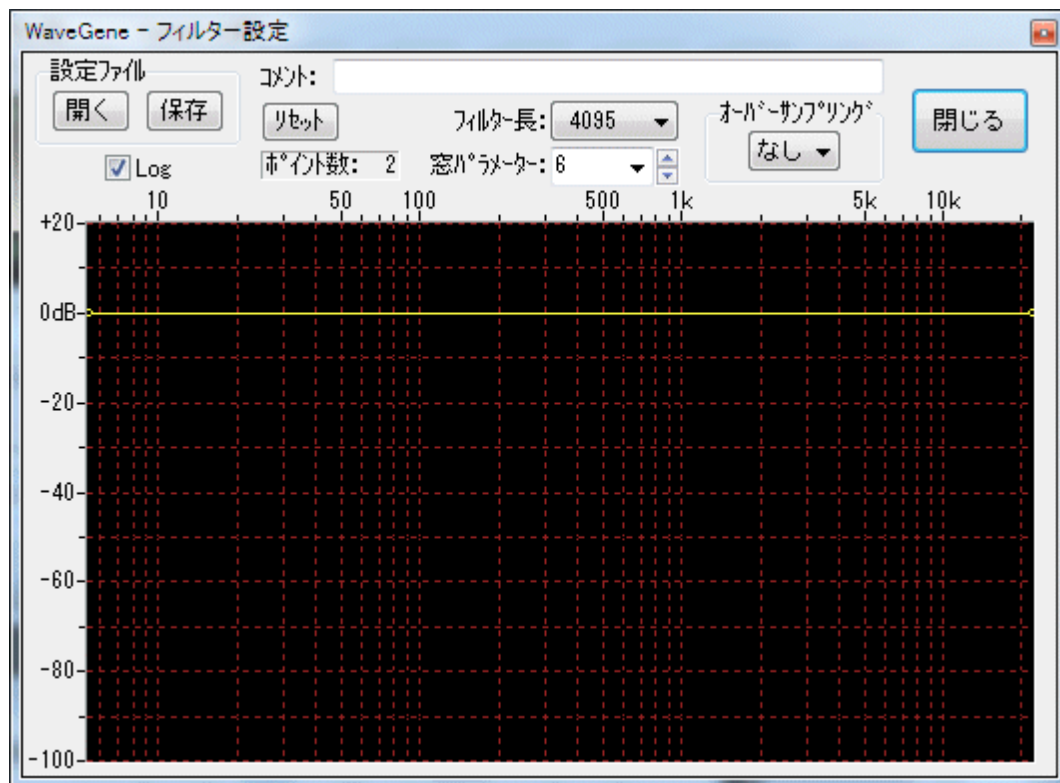
Check the **Filter** check box ☐ フィルター to activate the filter.

When checked, it changes to ☒ フィルター and the filter setting window is displayed.

The filter is enabled only when the filter setting window is displayed.

Filter Settings window

The frequency characteristics of the filter can be set / changed.
(The size of the window can be changed freely.)



- **Edit control point (points)**

It is a bending point that determines the curve of the frequency characteristic of the filter, and it has a frequency and the amplitude (gain) there.

(Hereafter abbreviated as Point)

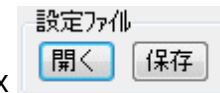
In the state in which the initial state filter is not applied, the gain is both 0 dB at the left end (0 Hz) and the right end (sample frequency / 2) as shown above. (Flat in all bands)

By editing the number of points and the frequency / gain, set a curve of arbitrary frequency characteristics.

- **Import and save configuration file (filter parameter file)**

The current filter setting is saved in the WG.INI file at the end of WG.EXE together with the waveform parameters etc. without doing anything, and it is reproduced on the next start.

When you want to save it separately, you can reproduce it at any time by saving it as a filter parameter file (*. WGF) with a name given to the current setting.



With the **Open** , **Save** button in the configuration file group box , You can read and save the filter parameter file.

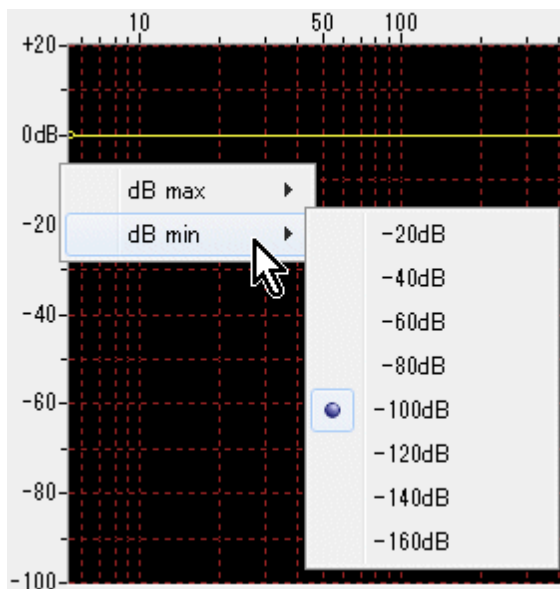
When saving, it is convenient to write a comment for explanation in the comment edit box.

(Comments and parameters are saved in the file)

* You can also load filter parameter file (*. WGF) by dropping it on this window.

- **Toggle vertical axis (amplitude / gain) display**

Change the setting of the scale **dB max** (+ 40 to 0 dB) of 0 dB or more and the scale **dB min** (-20 to -160 dB) of 0 dB or less in the pop-up menu displayed by right-clicking the mouse in the scale display area on the vertical axis can.



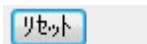
Point setting / movement by dragging the mouse is limited to the upper and lower limits of the display area.

Therefore, if you use only 0 dB or less, setting **dB max** to 0 dB always fixes the upper limit to 0 dB, making it easier to set up and convenient.

- **Toggle horizontal axis (frequency) display**

☒ Log If the check box is checked, logarithmic display will be displayed, if not checked linearly.

- **Reset properties and check the total number of points**



Reset button 示°イノ数: 2 you can reset the filter settings.

The total number of current points is displayed below the reset button.

When reset, the point for determining the characteristic (curve) of the filter is initialized to two ends of the band (0 Hz and sampling frequency / 2), and the flat (gain 0 dB) characteristic.

- **Filter Length**

In the **Filter Length** combo box, you can change the length (order) of the coefficients of the filter.



A larger sharper blocking becomes possible, but as a matter of course the load on the CPU becomes larger, it is better to reduce it when there is no need for a sharp change.

The length of the internal calculation buffer will be the length of this setting + 1, and the calculation of the filter will also be calculated in this length unit.

Actually, filter calculation (convolution) is done with FFT twice the length.

(This FFT is required twice for each channel)

The resolution of the filter's frequency axis is the sampling frequency / (filter length + 1).

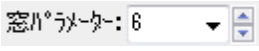
For example, at 48000 Hz sampling with filter length 4095, it becomes $48000/4096 = 11.72$ Hz, and point setting / movement is possible only at this multiple frequency position.

If the frequency axis display is logarithmic display (Log), changing the filter length will change the minimum frequency of display accordingly.

Note :

In the case of logarithmic display, in principle 0 Hz can not be displayed, but for the sake of convenience, the position which is half of the frequency of the resolution of the filter, which is the lowest frequency that can be set, is 0 Hz (Left end) is displayed.

- **Window parameters**

Window Parameters Combo box  to determine the shape of the Kaiser window used for filter design You can change the value of the variable. Values range from 0 to 10, you can directly set any value other than the preset value. (You can also change it with 0.1 steps with the right up / down control.)

The larger the value, the sharper the cutoff becomes possible, but the spread from the point becomes large, so it is good to set it while checking the actual frequency characteristics (see below).

* `_char_` may be expressed as `_char_`.

In that case, `_char_ = _char_ * _char_` 'and ' `_char_` is the value before multiplication by `_char_`.

The Kaiser window has a shape resembling the existing window depending on the value of `_char_`.

`_char_ = 0`: rectangular window (no window) `_char_ _char_ 5`: Hamming window

`_char_ _char_ 6`: Hanning window

`_char_ ? 9`: Blackman window

- **Oversampling**

If the **oversampling** combo box is set to **none** it is only filtered, but here **none** (x 2 < If you set it to, **x4** , **x8** , **x16**), you can create a signal with that multiple sampling

frequency and then filter I can do it.



For example, if you specify x16 when formatting with a sampling frequency of 48 kHz, create a signal with a sampling frequency of $48 \times 16 = 768$ kHz and set it here A band-limiting filter (decimation filter) of a frequency characteristic that passes only up to $48/2 = 24$ kHz is thinned out, and sampling frequency It is returned to the 48 kHz signal.

It is possible to suppress out-of-band components of unrestricted waveforms other than sine waves, and to create high-quality noise.

(If you generate M series noise at about x 8 and apply decimation filter, you can create high quality white noise

Also, at that time, filtering with a frequency characteristic of -3 dB / oct (-10 dB / dec) in the whole band results in pink noise.

Noise of any other characteristic can be created in the same way)

* As the multiplier is increased, the computation time increases, so if you try to play it in real time on a non-powerful CPU, you may skip the sound.
In that case please write it to the wave file once and use it.

(Currently the program is not particularly speeded up, but it may be faster in the future)

note:

Even if it is set to something other than **none** (x2, x4, x8, x16), the signal is re-created at that multiple sampling frequency and then filtered, so the frequency of the created signal is the same as when the filter is not applied Hmm. (It is only bandwidth limited)

Also, it does not mean that the sampling frequency is raised by inserting data of zero as much as the original data.

For example, if you create a 1 kHz sine wave with **none** at a sampling frequency of 48 kHz, **x 2** will create again at twice the sampling frequency 96 kHz and decimate it, It returns to 1 kHz of the same 48 kHz sampling as the original.

However, only the user waveform is handled specially.

As it is not possible to re-create the data with the user waveform newly, if you set the sampling frequency of **x 2** , the frequency will be doubled, the playback time will be halved fast? It will become.

In this case, it lacks consistency with other signals, so only overwhelming user waveforms are inserted at the oversampling by zero, so that the playback frequency is not changed. That is, it acts as an interpolation filter.

* oversampling tricks (file creation)

(It's not even a thing like that ...)

Normally, even if you create a wave file with x2, x4 ... as above, the format of the completed file will be restored and the sampling frequency will be It will not be x2, x4 ..., but if you do the following, the file with the sampling frequency in the state of x2, x4 ... (before decimation of data by multiplying only the decimation filter) You can create.

*  **Press the output button to Wave file** , and output to " Wave file " Hold down **Ctrl key** and press the " Save " button when you press the " Save & , You can create a wave file in the format of x2, x4

Note: Pressing the Ctrl key is not the time to press the **Output to Wave File** button. This is when you press the " Save " button in the subsequent dialog box for specifying the file name.

(You can release Ctrl key when file creation starts)

Also, only Wave files can be created.

You can not create a text file (*. WGT) in the x2, x4 ... format by pressing the **Output to Wave File** key while holding down the Shift key.

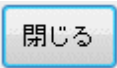
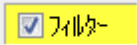
* Sampling frequency conversion (interpolation) of Wave file

Using this trick, it is possible to convert the sampling frequency of the wave file, although it is only x2, x4, x8, x16.

* After loading the wave file you want to convert to the user waveform, set the oversampling setting to x2, x4, etc. **Wave file length / sweep length combo box** by setting the length (sec) of the read file with  **Output to Wave file By clicking the button** , you can create a Wave file whose sampling frequency has been converted.

Always **Ctrl key** while pressing the " Save " button in the file name designation dialog.

Also, do not forget to set the sampling frequency of the WG creation format to be the same as the value of the loaded file, and reset the filter characteristics to flat.

- **Close button** 
Close the filter setting window.
It is also possible to close the setting window by unchecking the **filter** check box  in the main window I can do it.

Editing filter frequency characteristics

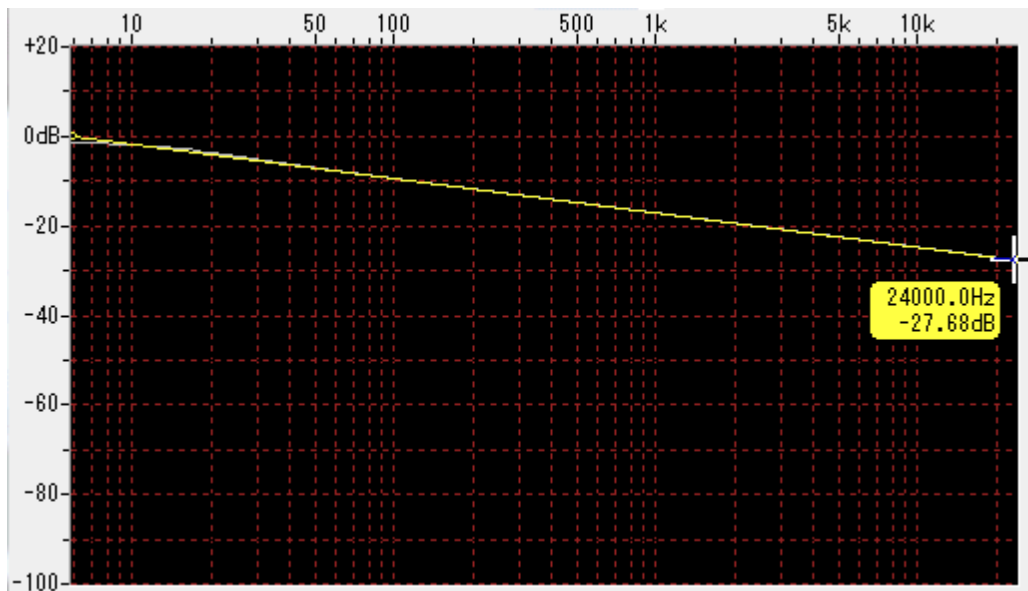
- Set curves of arbitrary frequency characteristics by editing the number and frequency / gain of edit control points (points).

* The point editing method is almost the same as other waveform editing software.

• Review and move points

When placing the mouse cursor over a small ○ indicating a point, the cursor changes to a crosshair cursor, and the frequency and gain of that point are displayed in a small floating window. (The color of the floating window also turns yellow.)

You can move the point by dragging the mouse as it is, and follow the display of frequency and gain.



In the above example, since it is in the initial state immediately after resetting, neither point can change the frequency only by the gain, the frequency can not be changed.

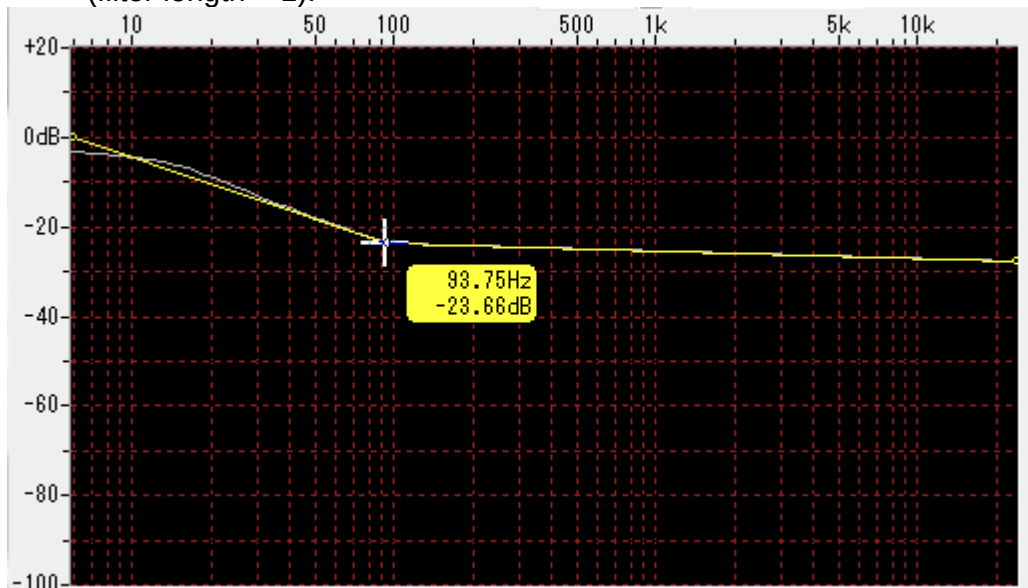
- **Create point**

You can create a new point there by clicking the left mouse button in a place without a point.

It is possible to move by dragging as it is, and the point of the newly created point can change the position of the frequency unlike points at both ends.

However, it is not possible to move to the position of the frequency exceeding the point of both neighbors only by the allowable range.

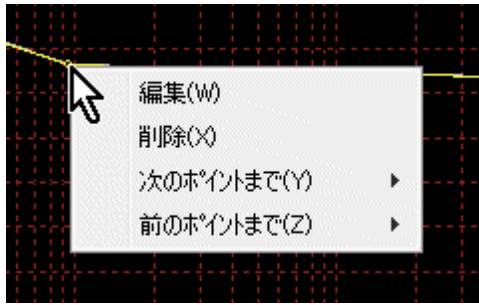
* The minimum spacing of points is the resolution of the filter, ie sampling frequency / (filter length + 1).



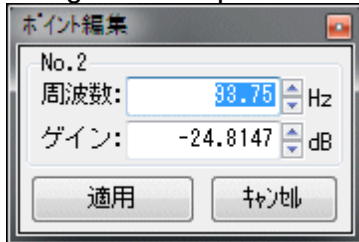
Up to 100 points can be created for points.

- **Edit point**

Right-clicking the mouse on the point will display a pop-up menu like the one below.



When you select **[1] edit**, the point editing window opens and you can enter the frequency and gain of the point numerically.



For convenience, the left end (0 Hz) is numbered as No.1, and the frequency increases to No.2, No.3 ... in the right direction.

* Frequencies can only be set in multiples of the resolution of the filter.

Even if you say it is difficult to calculate yourself, you can enter the frequency you want to set in the edit box appropriately.

It is automatically changed to a multiple of the resolution about 1 second after the key input end.

Moreover, it can not set to a value exceeding the frequencies of both neighboring points.

You can change it even with the up / down control  but in this case it is always a valid numeric value that is an integer multiple of the resolution.

* Regarding the gain, it is only from dB max to dB min set on the scale on the vertical axis of the graph, but if necessary it is possible to enter a wider range of values.

So, for example, although it is virtually meaningless in practice, if you want to set the gain to infinity - zero (zero signal) - you can set it to -200 dB or -300 dB.

However, on the graph display, ○ is displayed at the upper limit dB max or lower limit dB min of the range of the current display area.

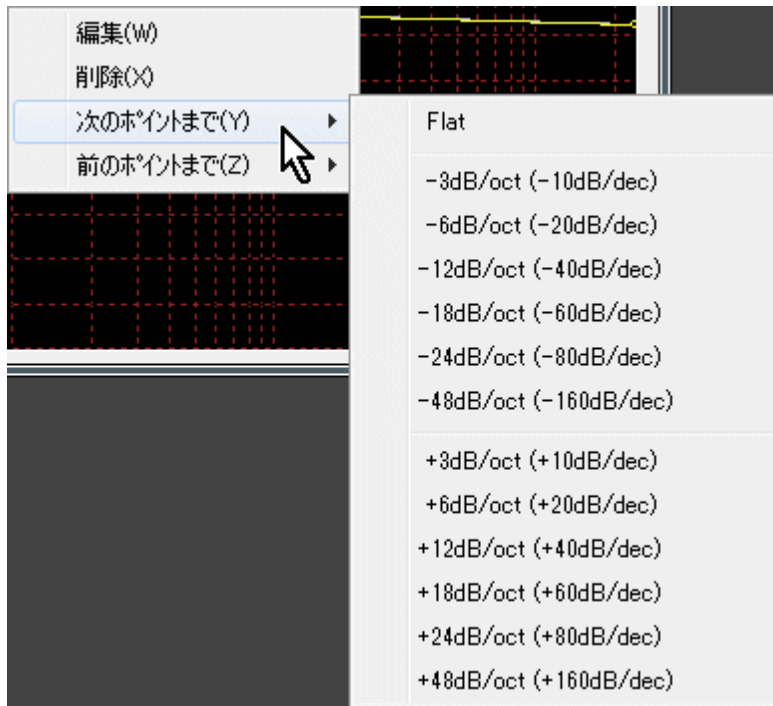
Therefore, although it may be displayed quite differently from the actual inclination, the curve of the internal filter is accurately calculated.

Furthermore, it is certain to check actual frequency characteristics (described later).

[2] Delete, the point will be deleted.

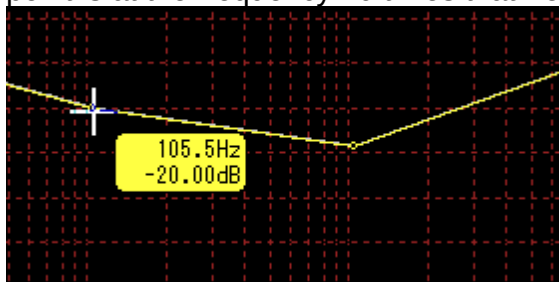
* Of course you can not delete points at both ends.

[3] Choosing to the next point will bring up a further pop-up menu.

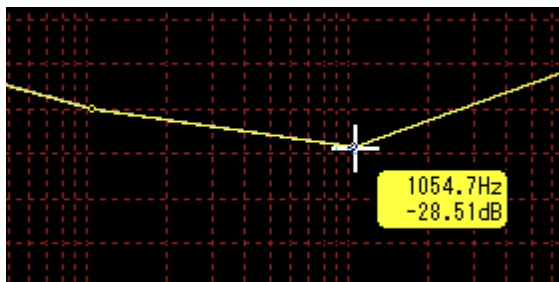


By selecting these menus, you can connect up to the point on the right of the current point (the one with the higher frequency) with the line of the specified slope [flat, attenuation, increase] The gain of the point is changed.

For example, as in the example below, when the left point is 105.5 Hz - 20 dB, and the right point is at the frequency 10 times that frequency

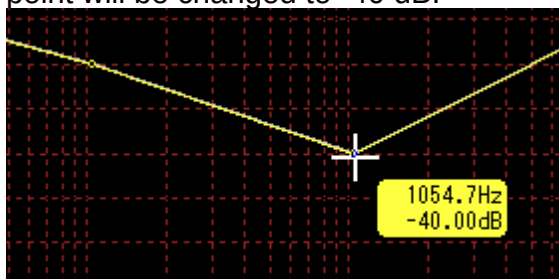


The left point is 105.5 Hz - 20 dB



The right point is 1054.7 Hz -28.51 dB

If you select **-6 dB / oct (-20 dB / dec)** to the next point at the left point, the gain of the right point will be changed to -40 dB.



The same is true when you want to make other slopes, and if you want to make both gains the same, you can choose **Flat**.

* For example, if you want to set the slope to -3 dB in all bands, it is possible to select **-3 dB / oct (-10 dB / dec)** to the next point at the 0 Hz point at the left end immediately after resetting.

If you filter the white noise with this filter, it will become pink noise.

If you apply a filter of **-6 dB / oct (-20 dB / dec)**, it will become Brown noise.

(Since the overall gain goes down, it is a good idea to raise the gain of 0 Hz to the extent that the peak of the output signal does not exceed 0 dB beforehand)

Note: When calculating the gain slope as in this case the point of 0 Hz is actually the lowest frequency (the frequency of the resolution) Half of the frequency point It will be treated as special. (Correction for Log scale)

[4] To the previous point changes to specifying the inclination to the point next to the left with the point on the right as the base point, otherwise it is exactly the same as above [3] is.

Actual frequency characteristics of the filter

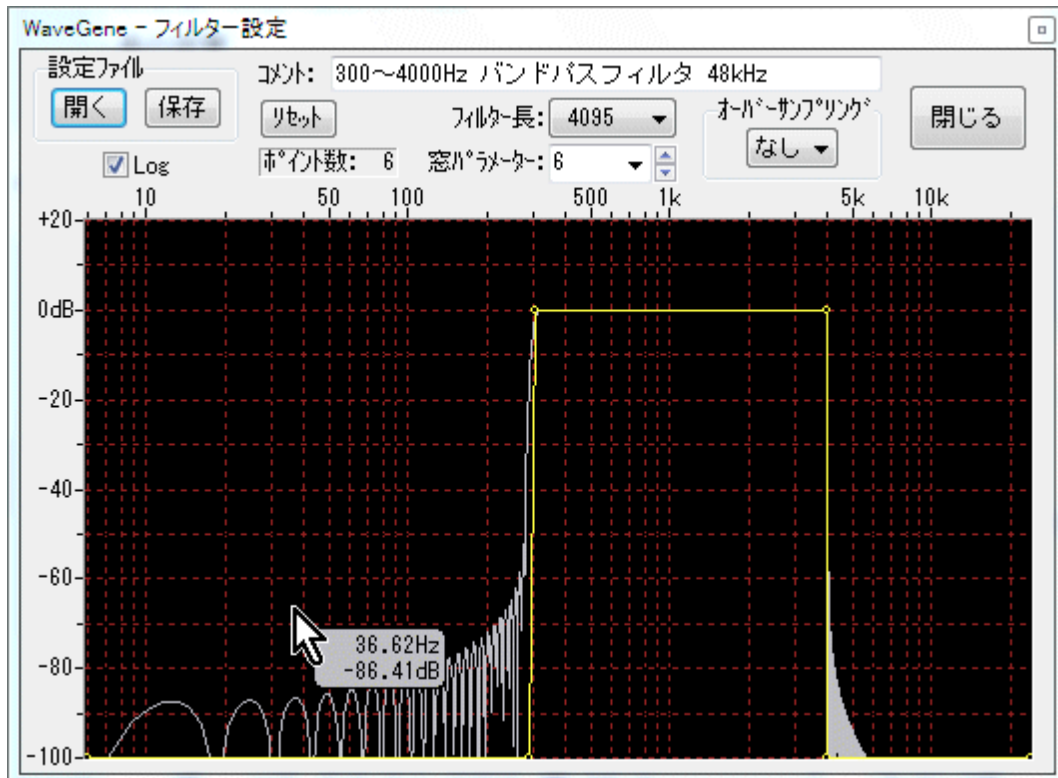
Even if you set a point and set the curve of the desired frequency characteristic, the frequency characteristics when actually operating as a filter will not be as per the curve. For that purpose, the filter coefficients (impulse response) require infinite length, but in reality naturally it is limited.

Especially if you give sharp changes to the setting curve, the actual characteristics deviate greatly from the set curve.

Actual frequency characteristics vary depending on the setting of filter length and window parameters, but it is convenient as it is designed to display in real time the frequency characteristics actually obtained to make it easier to design the desired characteristics.

Even if the position of the point is moved, the actual frequency characteristics will be displayed following the change in real time.

- It is an example of a band pass filter of 300 to 4000 Hz.
It is the curve set to be displayed in yellow, and the actual frequency characteristic is displayed in gray.

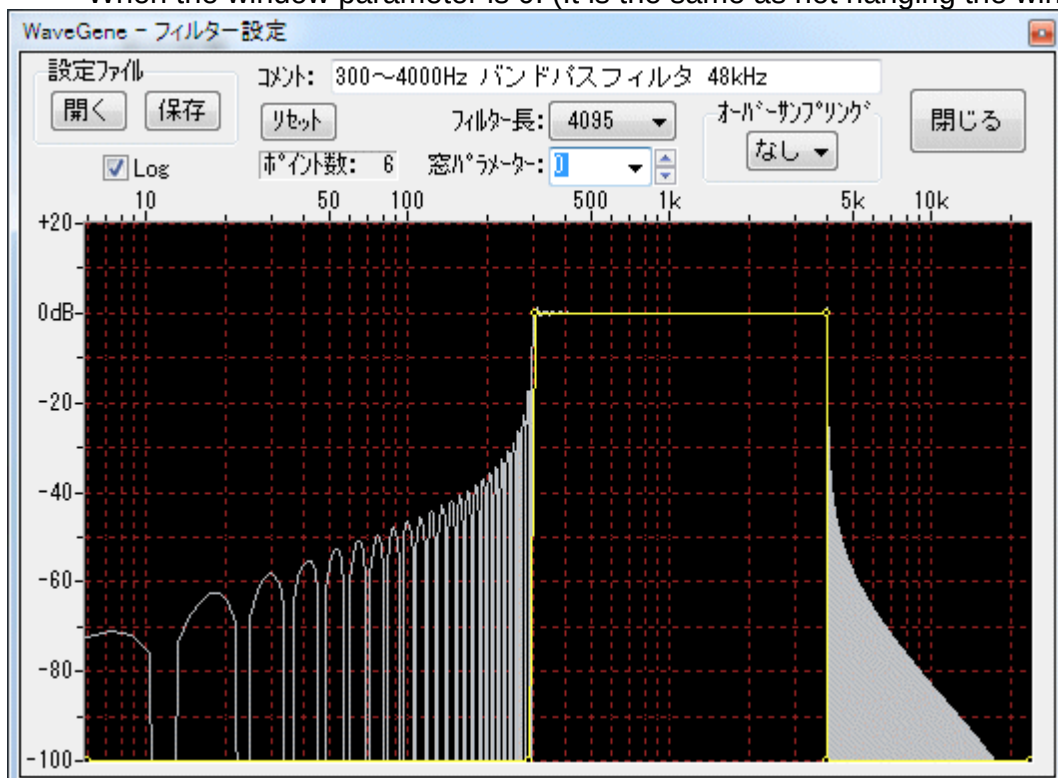


When you move the mouse cursor on the graph, the gain of the actual characteristic at that frequency position is displayed as a floating window (gray).

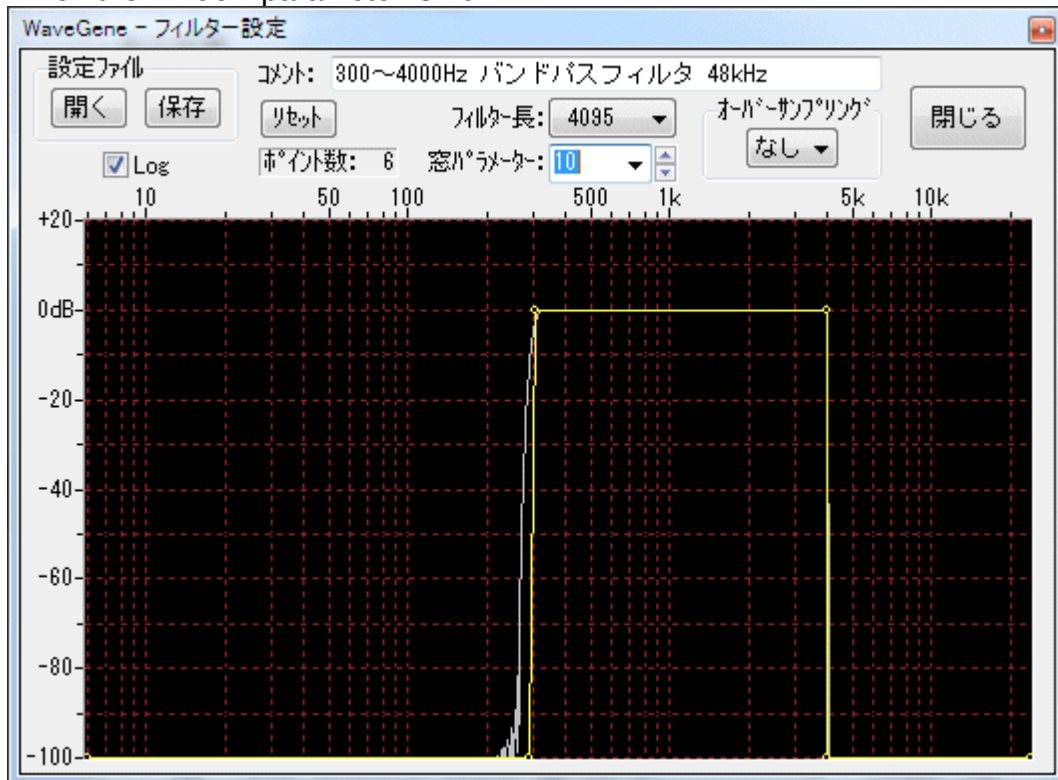
If you move the mouse cursor to a point, the cursor will change to a cross (the color of the floating window also turns yellow), and the point setting value will be displayed.

- It is when the window parameter is changed.

When the window parameter is 0. (It is the same as not hanging the window)

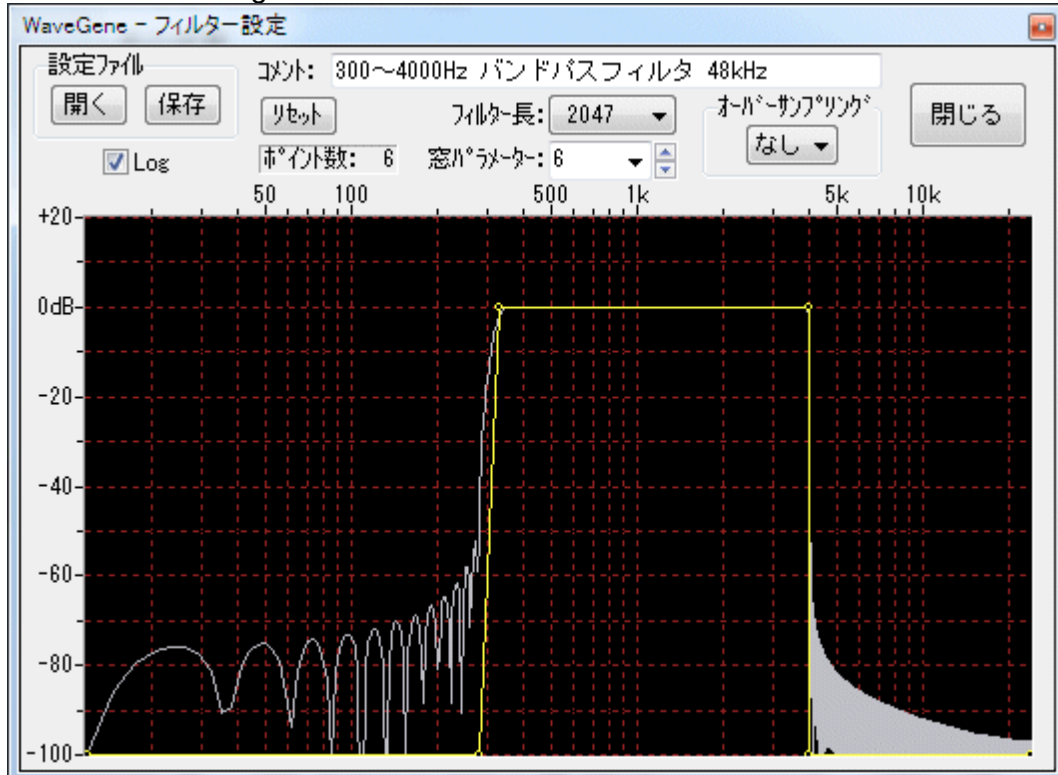


When the window parameter is 10.

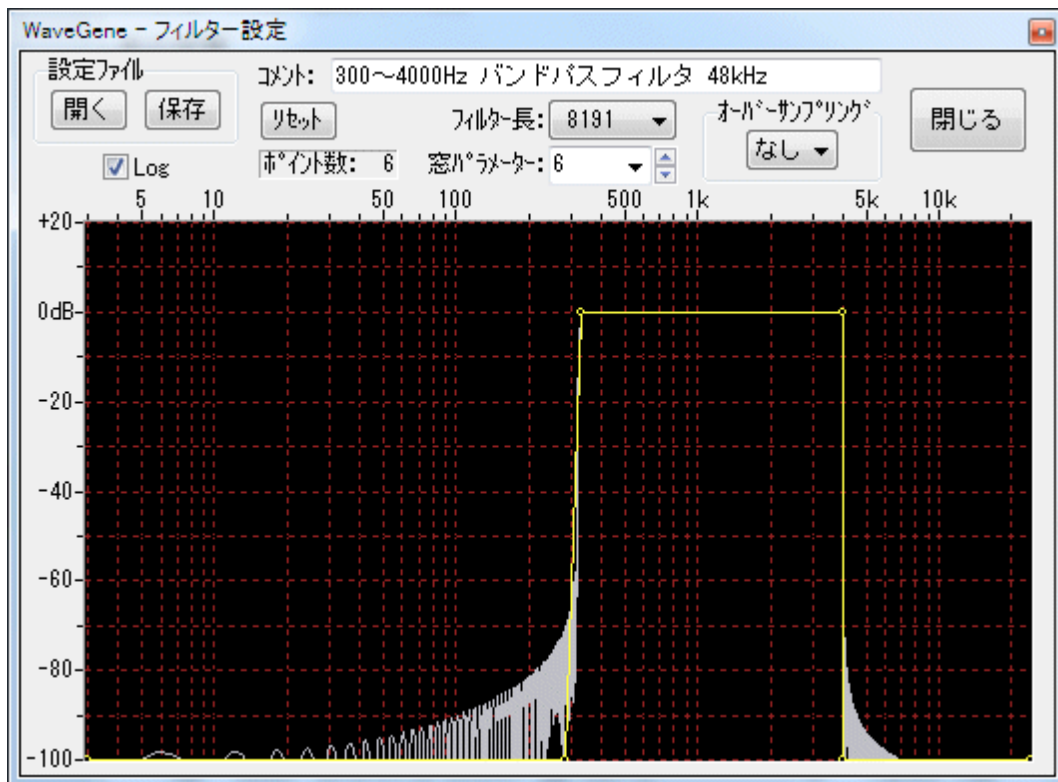


- This is when you change the filter length.

For filter length 2047.



In case of filter length 8191.



Changing filter characteristics during signaling

- You can change the filter characteristics when

button is pressed to generate a signal, that is, even during operation  I will. You can try filtering it while sounding such as white noise and move the peak of its frequency characteristics.

However, please be aware that the load on the CPU increases by that much, so skipping may occur depending on the environment.

In addition, since it changes only in units of length of filter length + 1 (corresponding time), if the filter length is increased to cause a very sharp temporal change, the amplitude becomes discontinuous depending on the change of the frequency characteristic. In that case, please be aware that you will naturally hear it as a petit noise at that position.

Other

- When editing a point, when the frequency axis is Log display, the point interval becomes narrower in the high frequency part, so it is better to remove the **Log** check and make a linear display. It becomes easy to do.
- When switching the frequency axis to linear display, the curve connecting each point is displayed as a straight line for convenience, but note that the gradient in the linear scale is not the slope of the actual filter characteristic. Stay here. The change in the actual frequency characteristic of the filter is calculated so as to be the slope of the line displayed at the Log scale only, so it will not be a straight line at the linear scale. (Exponential curve)

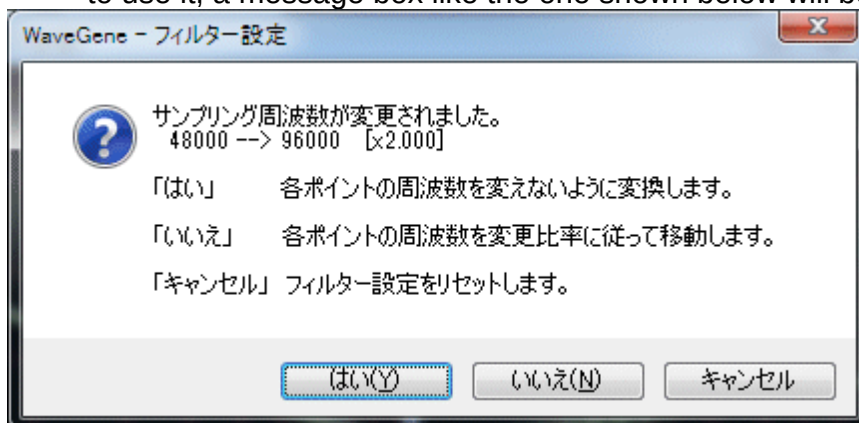
Please confirm with the actual frequency characteristic to be displayed.

* It is not possible to make such a filter characteristic that it becomes the inclination of a straight line as a linear scale.

- Filter frequency resolution is sampling frequency / (filter length + 1).
However, the characteristics of the window function are added, and the actual resolution slightly expands on both sides of the center frequency.
For example, if you only want to cut out components of a certain frequency, you should set not to cut only the point of that frequency but to cut it by slightly expanding both sides.
Adjust while confirming actual frequency characteristics with filter length and window parameter value.
- Characteristic setting of the filter, that is, the setting of the frequency position of each point, depends on the sampling frequency as described above.
Therefore, when setting a filter (for example, 10 kHz low pass filter) with an absolute fixed frequency, it is necessary to set the point of the filter for each sampling frequency.

When the sampling frequency is changed from the design time, a confirmation message is displayed as to how to convert the frequency position of each point.

When changing the filter designed at sampling frequency 48 kHz to 96 kHz and trying to use it, a message box like the one shown below will be displayed.



If you press **Yes**, only the internal parameters are converted so that the set frequency of each point does not change the original frequency value. (Gain does not change)

When you press **No**, the setting frequency of each point is moved relative, and in this case, everything is doubled. (Gain does not change)

Press **Cancel** to change to the reset state (flat) at 96 kHz.
Please be aware that the cancellation is a reset.

Even when * **yes**, since the resolution of the filter changes, the frequency may change slightly due to that limitation.

If you want to set it strictly, please confirm after conversion and reedit if necessary.

- **Format of filter parameter file (*. WGF)**
The filter parameter file has the following text format.

Example: 300 to 4000 Hz band pass filter (in the setting example below)

"WaveGene Filter Parameter", 1.00 ← 1st line title and version

"Comment", "300 to 4000 Hz band pass filter 48 kHz" ← Comment

"OverSampleRate", 1 ← oversampling rate

"Size 1", 4096 ← filter length + 1

"AmpScale_dB_Max", 20 ← Value of amplitude scale dB max

"AmpScale_dB_Min", -100 ← Value of amplitude scale dB min

"KaiserAlpha", 6 ← value of window parameter

"Sampling Frequency", 48000 ← Sampling frequency

"Points", 6 ← number of points

; No. | Indx | Frequency | Amp.dB ← Description Comment line

1, 0, 0, -300 ← or less, No, frequency index, frequency and gain dB value of each point are repeated.

2, 25, 292.96875, -300

3, 26, 304.6875, 0

4, 341, 3996.09375, 0

5, 342, 4007.8125, -300

6, 2048, 24000, -300

Since it is a text file, it can be edited with a text editor.

The frequency index indicates the position of each point at the sampling frequency / Size 1, which is 0 to Size 1/2.

- Drawing of the actual frequency characteristic is usually calculated at the frequency position of the fineness of 8 times the frequency resolution by the filter length. If you have enough CPU processing speed and want to display it in more detail, after terminating temporarily, add the following description to any position in the Filter section of WG.INI and reexecute it.

[Filter]

ResponseRate = 16

4, 8, 16 can be specified.

If it is set to 4, the display becomes rough but the CPU load decreases.

Even if it is 4, the intermediate peak and dip positions are displayed properly, so there is no problem in practical use.

Configuration example

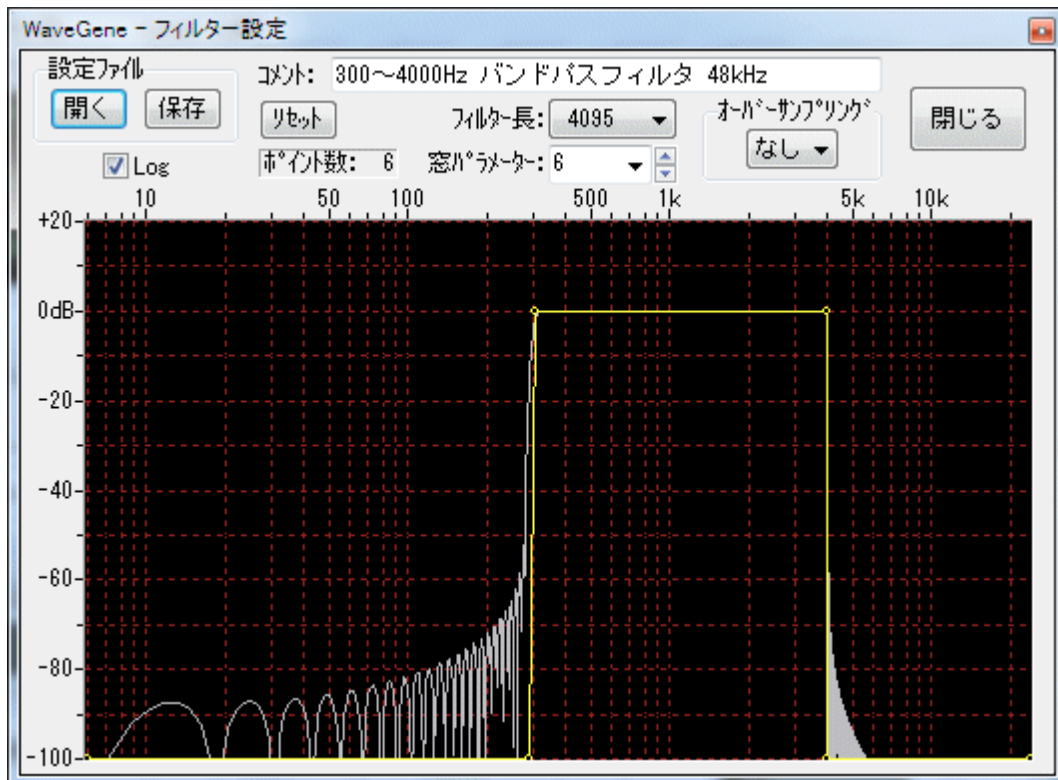
- **[Example 1] 300 to 4 kHz bandpass filter**

It is just a band pass filter.

48 kHz 6 points

No.	Frequency	Amp.dB
1	0	-300
2	292.96875	-300
3	304.6875	0
4	3996.09375	0

5	4007.8125	-300
6	24000	-300



Changing the filter length and window parameters will change the actual characteristics.

If you want to use a filter that is not steep like this, add more points and make it gentle.

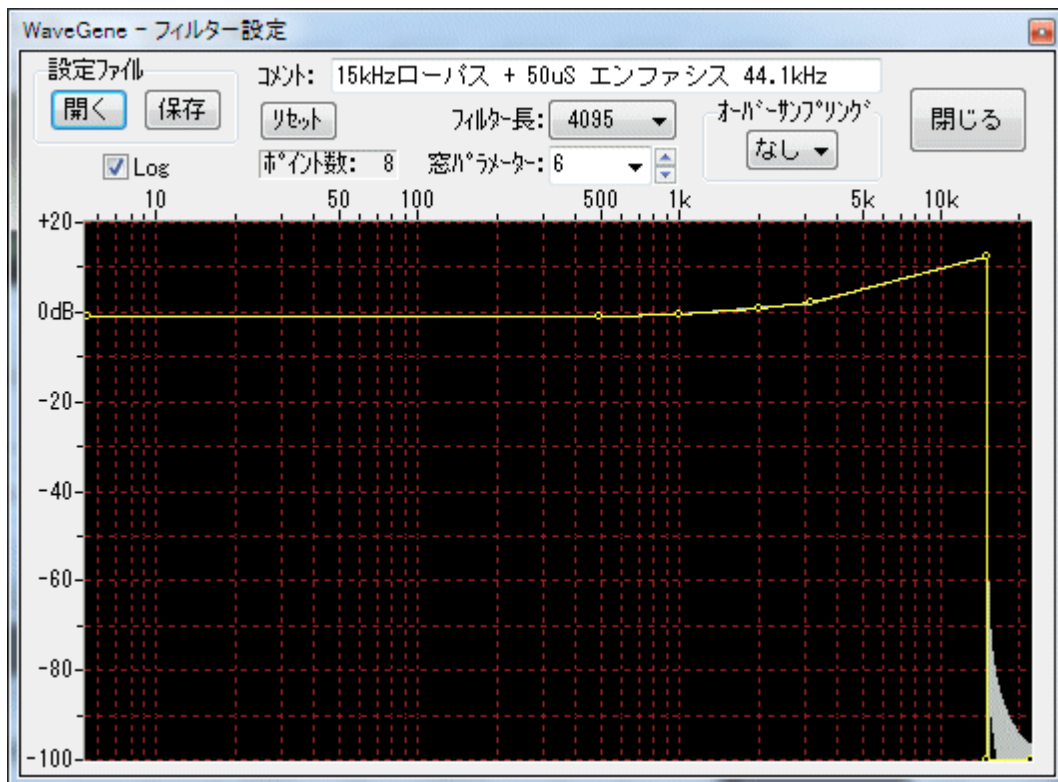
- **[Example 2] 15 kHz low pass + 50 u s emphasis**

FM broadcast This is a filter applied to a wave file for compositing signal.

50uS emphasis is a rough approximation. For the emphasis, the overall level has been lowered by 1 dB.

44.1 kHz 8 points

No.	Frequency	Amp.dB
1	0	-1
2	495.2636719	-1
3	1001.293945	-0.5
4	2002.587891	1
5	3176.147461	2
6	14997.87598	12.5
7	15008.64258	-300
8	22050	-300



Created with the Personal Edition of HelpNDoc: [Create iPhone web-based documentation](#)

How to use user waveform

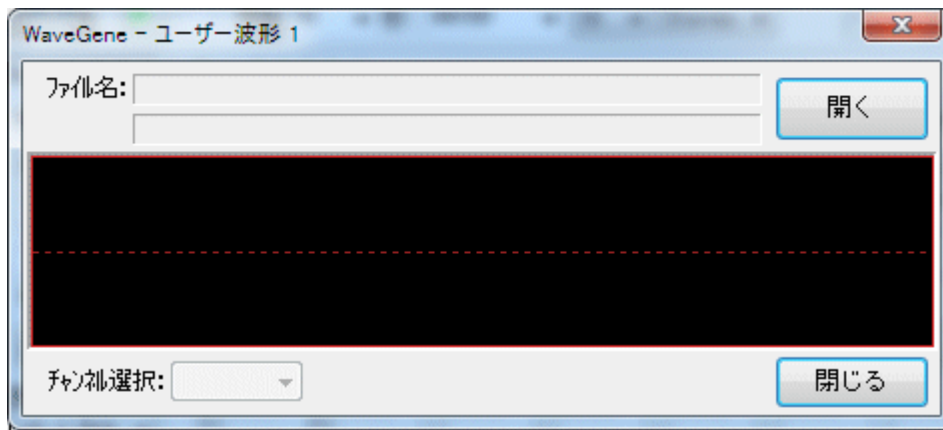
How to use user waveform

In the waveform combo box of Wave 1 and Wave 5, the user waveform ユーザー波形, you can generate arbitrary waveforms by reading a file in Wave format or your own text format.

Register user waveform

- In the waveform combo box of Wave 1 and Wave 5, the user waveform ユーザー波形 when you select the user waveform registration button ユーザー波形登録 is displayed.

* When not yet registered, when you stop the mouse pointer on the waveform combo box for a few seconds, it pops up with a popup message " Unregistered " will be displayed.



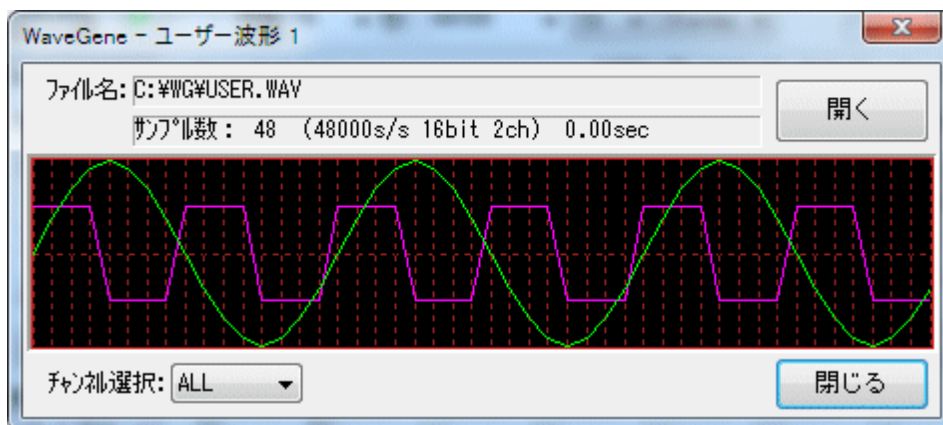
When you press the

- **Open** button, a dialog to select the user waveform appears, so select the file you want to register.

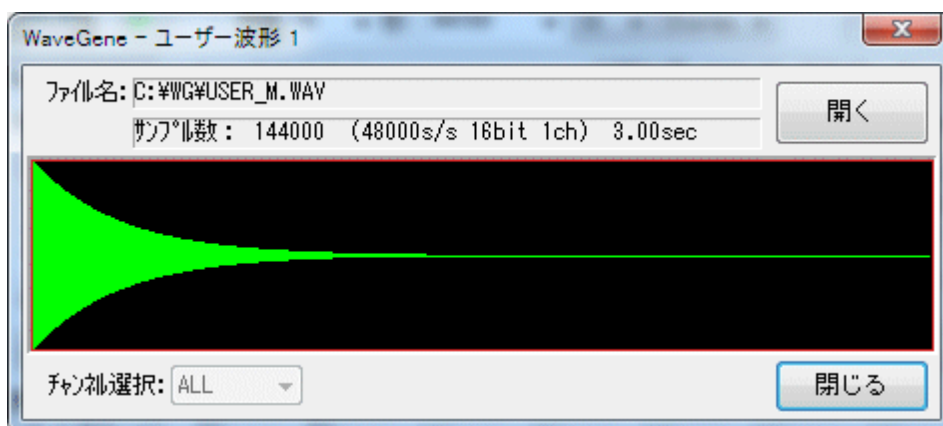
* Wave format and original text format (*.WGT) is to be chosen, depending on the "type of file" " Switch with.

* You can also register files by dragging and dropping files from Explorer etc. to this window.

(You can register similarly by dragging and dropping files to the main window.)



Example 1



Example 2

After reading, the waveform is displayed like this, so after confirmation **Close** button to close the dialog.

* The sample number, format, and time are displayed under the file name.

When text format file (*. WGT) is registered, sampling frequency, number of bits, and time are not displayed.

After * registration, when you hold the mouse pointer on the **waveform** combo box for a few seconds, the pop-up message displayed is the file name of the waveform.

User waveform registration button  if you want to confirm the registered waveform after registering * Please press" border="0"> to display the dialog.

* If you exit WaveGene with **user waveform** selected in the **waveform** combo box of Wave 1 or Wave 5, That user's waveform is automatically loaded.

For confirmation, the waveform is automatically displayed for about 2 seconds after startup.

If you want to use the same user waveform immediately at the next startup, please exit with this **user waveform** selected in this way.

Please note that if you exit with **user waveform** selected, the user waveform will be cleared and will return to the unregistered state without being automatically loaded at the next startup.

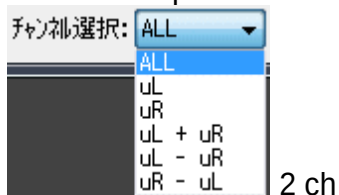
* The size of the registration file is not particularly limited.

As long as memory permits, even huge files can be read.

Output of user waveform

- Please be aware that the waveform output method after registration is slightly different from other waveforms.

In the **Channel Selection** combo box of the registration dialog, select the channel to be output.



In the case of 2ch above, in the case of **ALL**, the 2ch signals of the registered waveform are assigned directly to L and R as they are normally and are output target.

Actually it follows the output channel setting of the main screen, so it is possible to output only Lch or Rch etc.

When other than **ALL** is selected, the selected signal (the signal of a single channel, the signal of sum or difference) is treated as the generation signal (monaural) of Wave 1 or Wave 5, Output channel according to the output channel setting.

uL Lch of registered waveform

uR Register waveform if Rch

If **uL + uR** sum of Lch and Rch of registered waveform

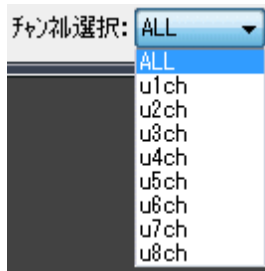
uL - uR difference of registered waveform between Lch and Rch

uR - uL If difference between Rch and Lch of registered waveform

Becomes the signal of Wave 1 or Wave 5.

When the registered waveform is monaural, the combo box becomes invalid, it becomes the signal of Wave 1 or Wave 5 as it is, and it is output to the destination according to the

output channel setting of the main screen.



Multichannel case

Likewise, when the registered waveform is multi-channel, the signal of the waveform channel becomes the output target of 1ch to Xch as it is when **ALL**, and the other **u1ch**, **u2ch** .. is selected, the selected single channel signal is treated as Wave 1 or Wave 5 generation signal and output to the destination according to the output channel setting of the main screen.

* When the number of channels of the current format setting on the main screen is smaller than the number of channels of the registered waveform, the number of channels of the main screen is obviously used only.

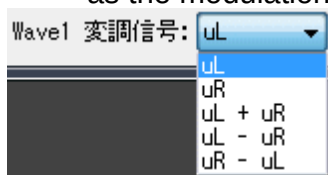
- When a user waveform is output, the registered data is repeated without pause time.
- **Note:** For user waveforms, only sample data is loaded and used even for Wave files. The sampling frequency of the original Wave file itself is not used directly. The frequency (pitch) of the user waveform actually output varies depending on the value of the sampling frequency of the format setting of the main window.

For example, even if the sampling frequency of the loaded Wave file is 48000 Hz and 1000 Hz data, the sampling frequency of the format setting of the main window is In the case of 96000 Hz, the actually output frequency is 2000 Hz which is twice the frequency. (Time will naturally be halved)

Therefore, when you want to output with the same frequency (pitch) as the original, please specify the sampling frequency of the format setting of the main window to the same value as the original Wave file.

* Select modulation signal (Wave 5 only)

- Wave 5, the **Wave 1 modulation signal** combo box like the one below is added to the registration dialog, and the channel to be used as the modulation signal of Wave 1 is selected.
It can be selected independently from the output target.
(It is the case when Wave 1 modulation function is enabled and Wave 5 is selected as the modulation signal and the user waveform is used as the modulation wave)



2 ch

The upper one is 2 ch, but the selected signal (signal of single channel, sum or difference signal) is used as modulation signal of Wave 1.

uL Lch of registered waveform

uR Register waveform if Rch

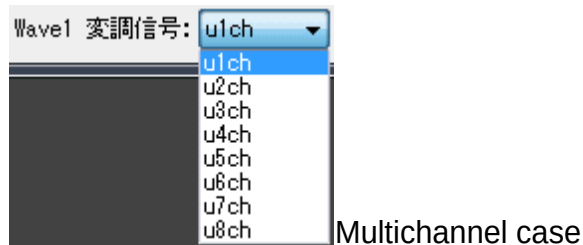
If **uL + uR** sum of Lch and Rch of registered waveform

uL - uR difference of registered waveform between Lch and Rch

uR - uL If difference between Rch and Lch of registered waveform

Becomes the modulation signal.

If the registered waveform is monaural, the combo box becomes invalid and it becomes a modulated signal as it is.



If the registered waveform is multi-channel, the selected **u 1 ch**, **u 2 ch** ... signals become the modulation signal.

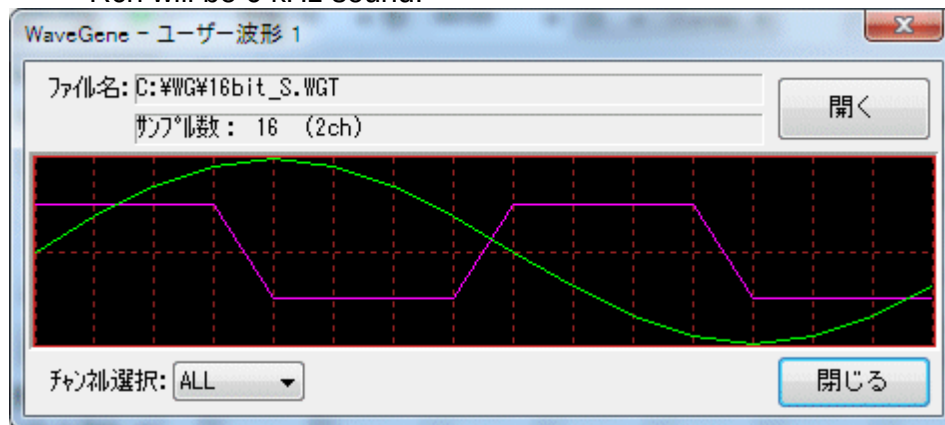
Original text format (*. WGT) file format

- You can create arbitrary waveforms and read them in the following way with a text editor or the like.

It is an example of sample number 16, Stereo, 16 bit integer.

Lch is a sine wave with an amplitude of 0 dB, and Rch is a rectangular wave of -6 dB (reverberation).

When this is played back at a sampling frequency of 48000 Hz, Lch Will be 3 kHz, Rch will be 6 kHz sound.



It is the contents of this file 16bit_S.WGT.

"Samples", 16 ← number of samples

"Channels", 2 ← number of channels

"Max", 32768 ← maximum value

"Data", indicating that 32 amplitude data are continued below 32 ←

0, 16422, 12539, 16422, 2317, 16422, 30273, 16422, 32767, -16422

30273, -16422, 23170, -16422, 12539, -16422, 0, 16422, - 12539, 16422

-23170, 16422, -30273, 16422, -32767, -16422, -30273, -16422, -23170, -16422
-12539, -16422

On the first three lines,
Number of samples
Number of channels (Mono: 1 Stereo: 2)
Maximum value of data
Specify.

* The order of these three lines does not have to be in the above order.

The maximum value indicates how many bits the amplitude data is represented.

For 8 bit integers, 128

For 16 bit integer, 32768

In the case of 24 bit integer, 8388608

In case of 32 bit integer, 2147483648

For 32 bit floating point, 1

You specify.

* The number of amplitude data and the amplitude data are always described at the end.

The number of amplitude data is the same as the number of samples for Mono, and twice the number of samples for Stereo.

Amplitude data is an integer equal to or less than the maximum value, or a real number (in the case of 32 bit floating point format) of 1 or less, in the case of Stereo Repeat Lch and Rch in this order.

There is no particular limitation on the number of lines arranged in one line.

In a general periodic waveform, it is enough to describe only one cycle of data as above.

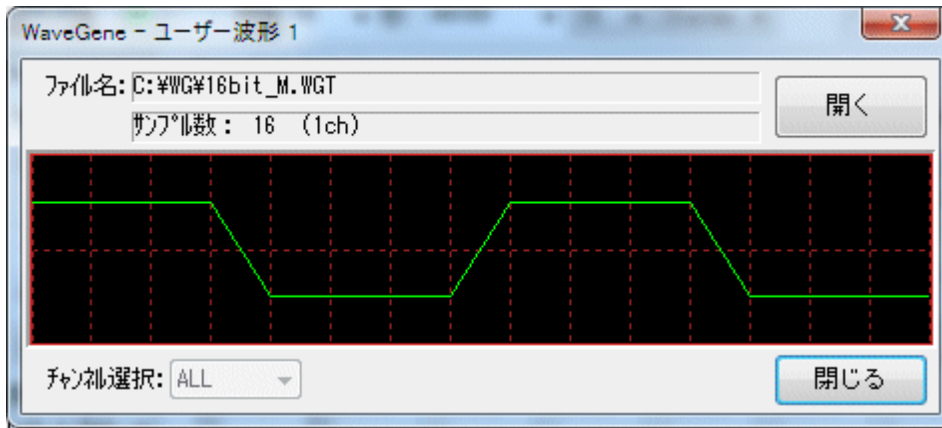
In the WG, amplitude data is all double (64 bit floating point format) It is converted and stored.

Note: Files (*. WGT) created by WaveGene's own text format output function You can also load it as it is.

(In this case there are extra lines, but all irrelevant lines are ignored.)

Since the user waveform itself can also be output in Wave format or text format, it is also possible to create complicated waveforms by repeating mixing and processing after processing, such as mixing with other waveforms.

- It is the case of Mono with the same waveform as above and only rectangular wave.



It is the contents of this file 16bit_M.WGT.

```
" Samples ", 16
" Channels ", 1
" Max ", 32768
" Data ", 16
16422, 16422, 16422, 16422, -16422, -16422, -16422, -16422
16422, 16422, 16422, 16422, -16422, -16422, -16422, -16422
```

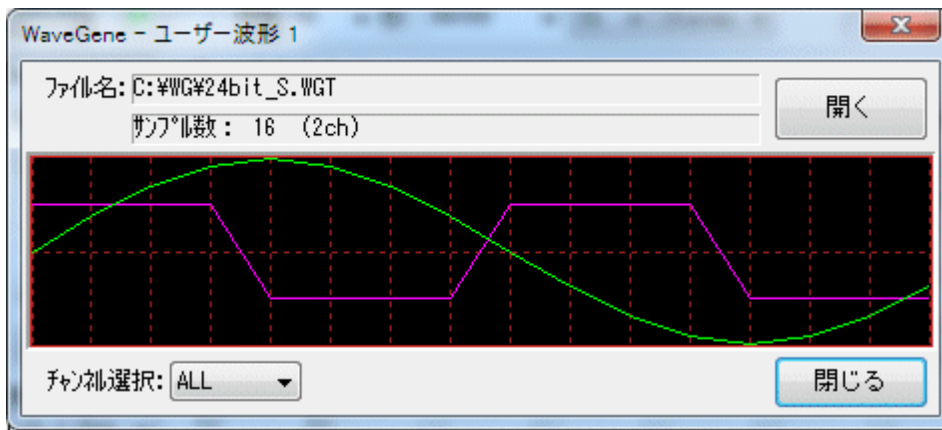
- This is an example of 8 bit integer.



The contents of this file 8bit_S.WGT.

```
" Samples ", 16
" Channels ", 2
" Max ", 128
" Data ", 32
0, 64, 49, 64, 90, 64, 117, 64, 127, -64
117, -64, 90, -64, 49, -64, 0, 64, -49, 64
-90, 64, -117, 64, -127, -64, -117, -64, -90, -64
-49, -64
```

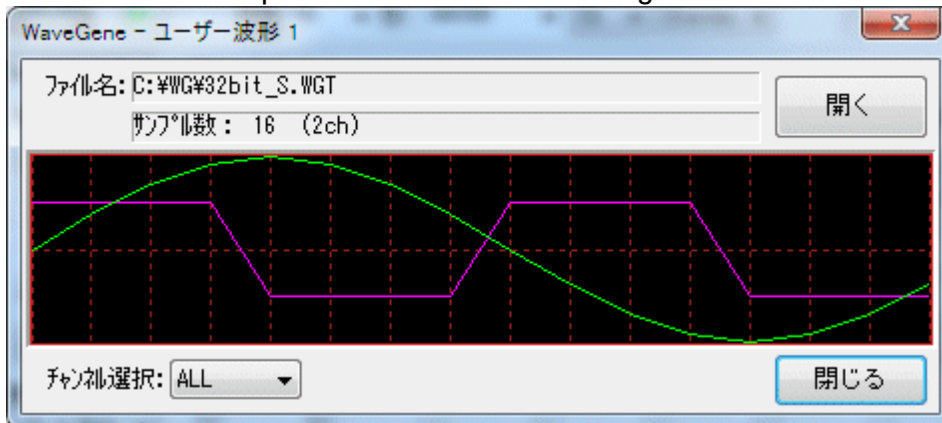
- It is an example in the case of 24 bit integer as well.



It is the contents of this file 24bit_S.WGT.

```
" Samples ", 16
" Channels ", 2
" Max ", 8388608
" Data ", 32
0, 4204263, 3210181, 4204263, 5931641, 4204263, 7750062, 4204263, 8388607, -
4204263
7750062, -4204263, 5931641, -4204263, 3210181, -4204263, 0, 4204263, -3210181,
4204263
-5931641, 4204263, -7750062, 4204263, -8388607, -4204263, -7750062, -4204263, -
5931641, -4204263
-3210181, -4204263
```

- It is an example in the case of 32 bit integer as well.

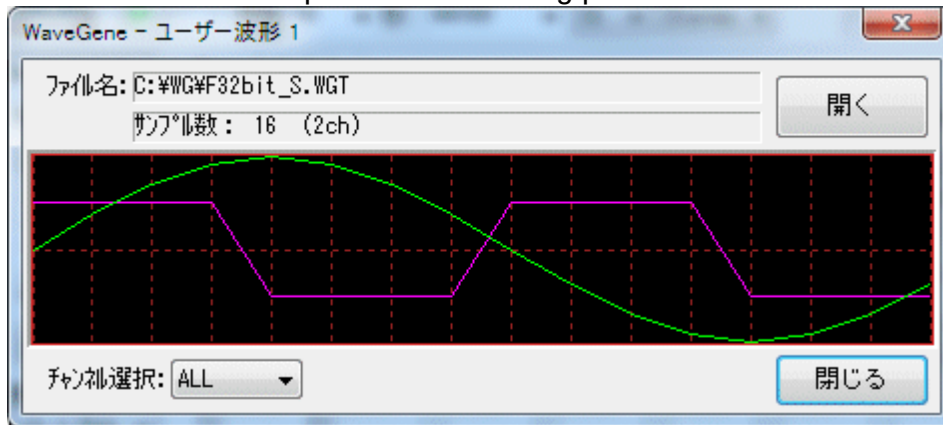


It is the contents of this file 32bit_S.WGT.

```
" Samples ", 16
" Channels ", 2
" Max ", 2147483648
" Data ", 32
0, 1076291388, 821806413, 1076291388, 1518500249, 1076291388, 1984016188,
1076291388, 2147483647, -1076291388
1984016188, -1076291388, 1518500249, -1076291388, 821806413, -1076291388, 0,
1076291388, -821806413, 1076291388
-1518500249, 1076291388, -1984016188, 1076291388, -2147483647, -1076291388, -
1984016188, -1076291388, -1518500249, -1076291388
```

-821806413, -1076291388,

- This is an example of 32 bit floating point.



The contents of this file F32bit_S.WGT.

```
"Samples",16
"Channels",2
"Max",1
"Data",32
0.0000000,0.5011872,0.3826834,0.5011872,0.7071067,0.5011872,0.9238794,0.501187
2,0.9999999,-0.5011872
0.9238794,-0.5011872,0.7071067,-0.5011872,0.3826834,-
0.5011872,0.0000000,0.5011872,-0.3826834,0.5011872
-0.7071067,0.5011872,-0.9238794,0.5011872,-0.9999999,-0.5011872,-0.9238794,-
0.5011872,-0.7071067,-0.5011872
-0.3826834,-0.5011872
```

Created with the Personal Edition of HelpNDoc: [Free CHM Help documentation generator](#)

Measurement of frequency characteristics using cyclic sweep

Measurement of frequency characteristics using cyclic sweep

WaveGene (hereinafter referred to as WG), and sister software WaveSpectra (WS) below.

There are the following methods for measuring frequency characteristics with WG + WS.

- (1) **A method of using cyclic sweep with WG's user waveform.**
- (2) **A method using a direct WG sweep signal. (Method using WS peak hold function)**

Normally, the accurate frequency characteristics can be easily obtained in a short time, so the method (1) shown below is recommended.

This method can be used in most cases, such as characteristics when the sound device itself is loop connected, external circuit such as an amplifier connected to the sound device, simple measurement of the speaker via the amplifier / microphone, etc. (When direct response can be obtained in real time)

(1) can not be used, for example, when the response is not returned in real time, such as the recording / playback characteristics of the sound recording device, the method using the peak hold function of (2) general sweep signal + WS will be used .

For points to note about (2), please see the help of WS.

1. Generation of cyclic sweep by WG

- Cyclic sweep to read as user waveform of WG Or download periodic noise signal separately.
<http://efu.jp.net/soft/wg/wg.html> UserWaveSample1.ZIP < BR> After downloading, unpack it and put it in the same folder as WG.EXE.

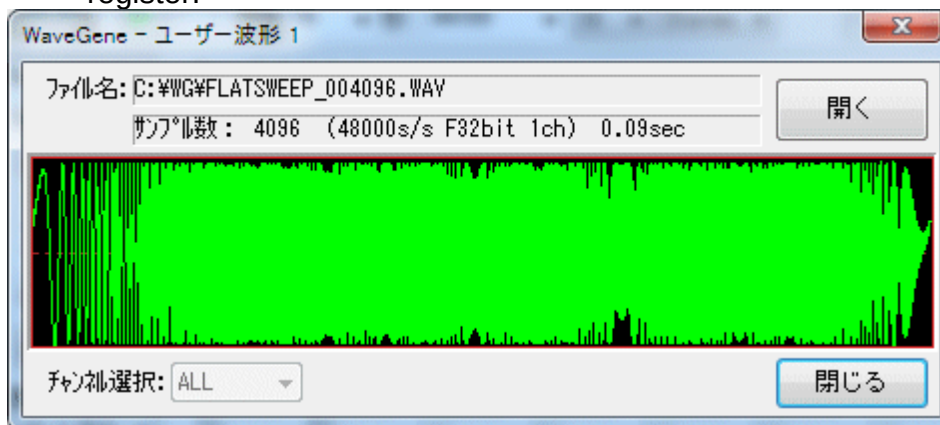
The number of the file name of each signal represents the length of one cycle (number of samples). (2048 - 131072)

Select the user waveform in

- Wave 1 **waveform** combo box, Register a periodic sweep or periodic noise signal. For how to register please see [How to use user waveform](#) .

* Both the cyclic sweep and the periodic noise can be used in the same way, but the cyclic sweep is better than the S / N and the periodic noise is extremely slight (+ - 0, 01 dB) flat As the sex is disturbed, it is usually better to use a cyclic sweep.

As an example, a cyclic sweep of 4096 samples FLATSWEEP_004096.WAV I will register.



- Set the format of the playback device, sampling frequency, etc. used for measurement as appropriate.

* The time of one cycle of the cyclic sweep signal is the number of samples / sampling frequency.

In the example, if the sampling frequency is 48000 Hz, it is 4096 samples, so 0.085 seconds .

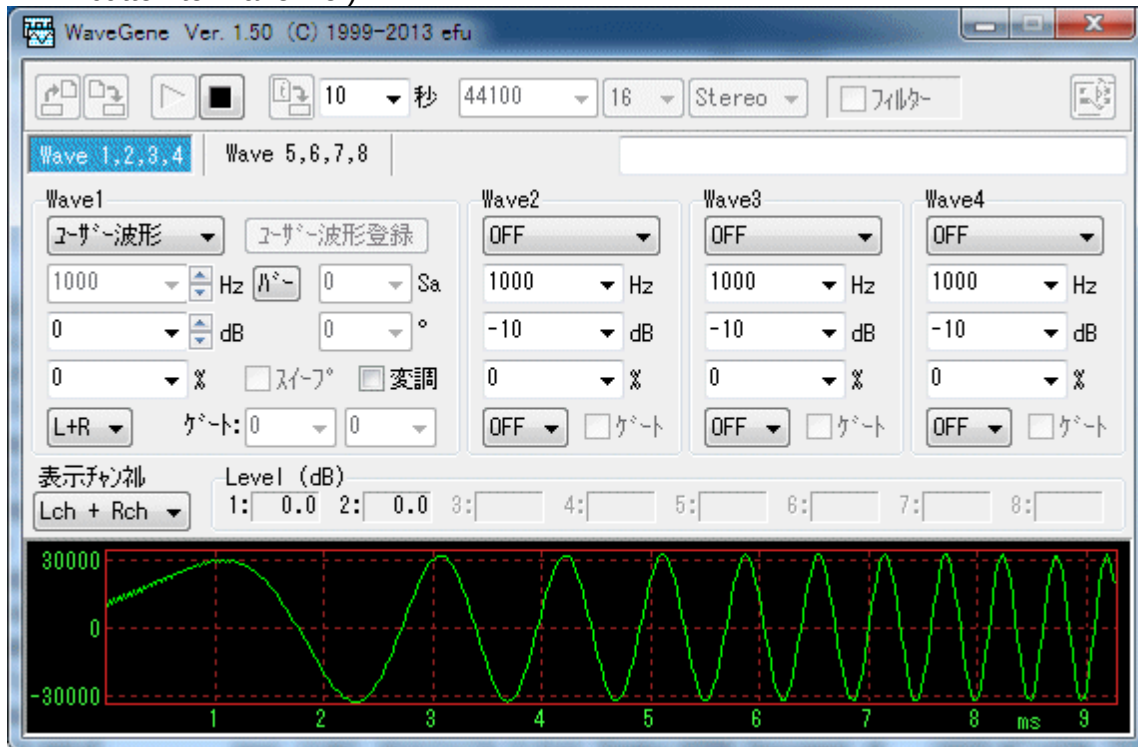
Set the output level using the

- **amplitude** combo box. (0 to -3 dB [Note](#) below)

Confirm that it can be generated continuously with the

- **output to the sound device button.**

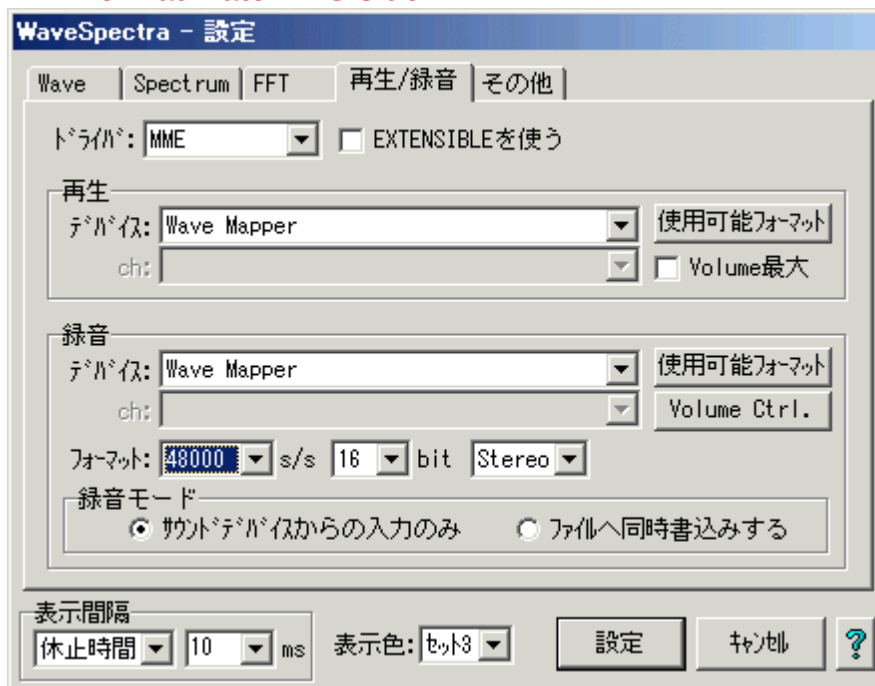
(It is the same as before to be able to create Wave file for specified time with output button to Wave file.)



2. Setting on the WS side

- In the WS setting dialog, set as follows.
(The example below is for WS version V 1.40, but the same for later versions.)

a. In the " Play / Record " tab, make the recording device and recording format match WG side.



Recording format same as WG side 48000 Hz 16 bit Set it to Stereo.

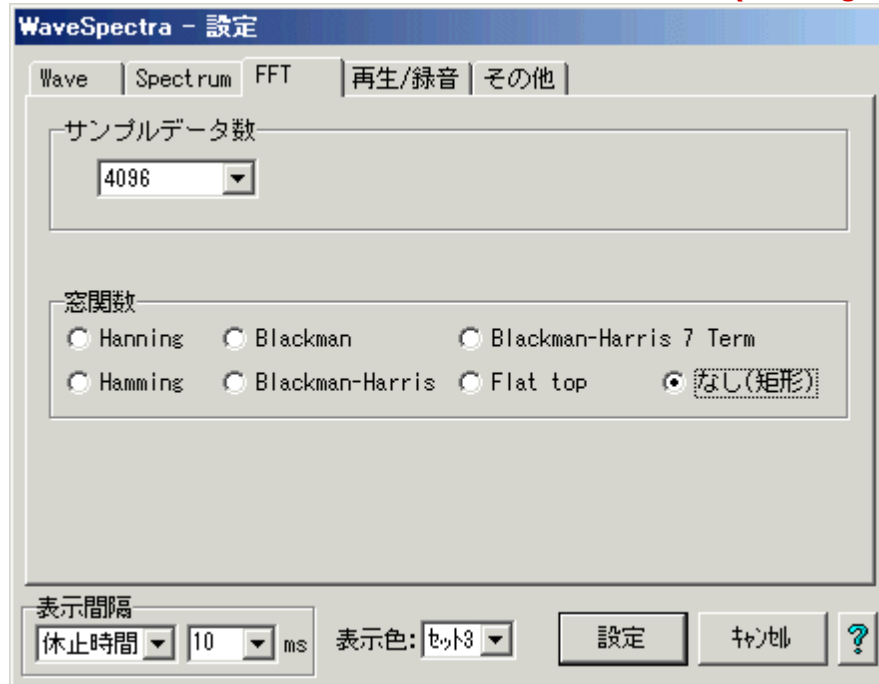
* Recording device and driver settings are here MME, Wave Mapper, but specify the same

device and driver as those actually set on the WG side.

b. In the " FFT " tab, With the length (number of samples) of the cyclic sweep registered by WG.

For example, set it to 4096.

c. Make sure that the window function is "none (rectangle) " .



3. Measurement of frequency characteristics

- When setting above is done, both WG and WS are operated. (The order is not specially specified)
(Output to " Sound Device " Button, " input / recording from sound device " Press the button)

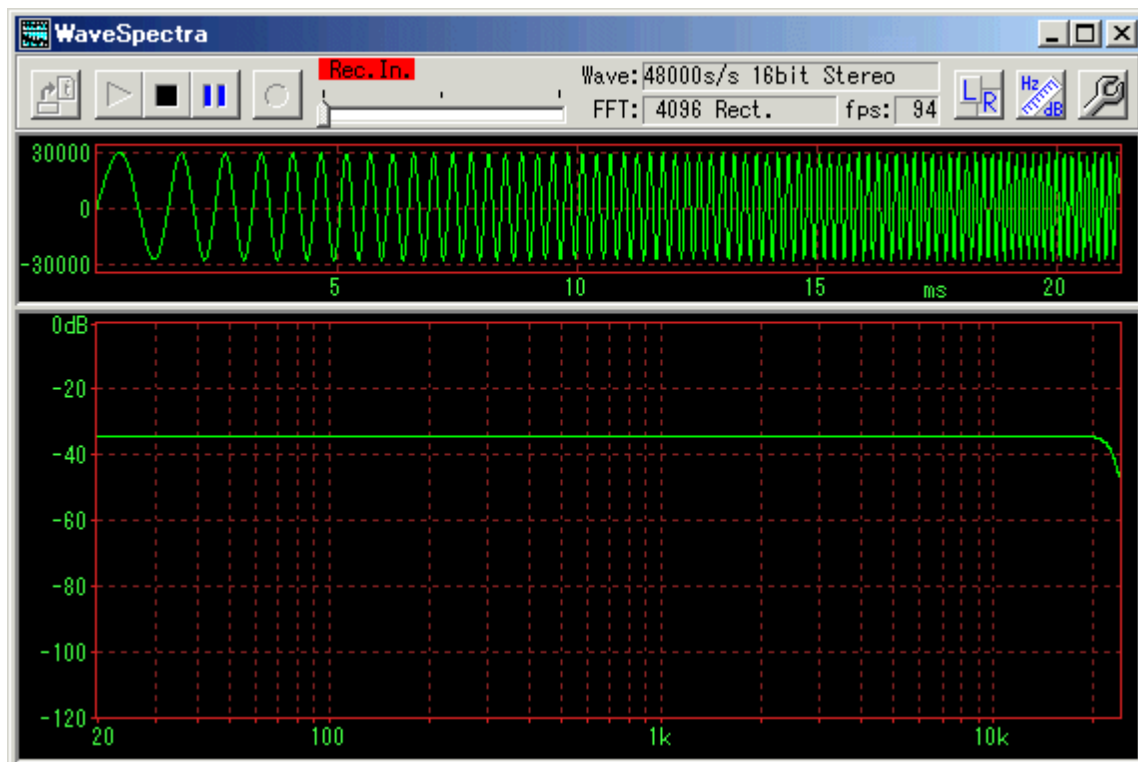
A graph of frequency characteristics is drawn every cycle of the cyclic sweep.

If the number of samples per cycle is long, it will not be an accurate graph until one cycle is over, so please wait for a while.

* First of all, connect the sound device input / output (loop back) and measure the frequency characteristics of the sound device itself.

I / O connections can be connected with code or specified on the mixer of each device.

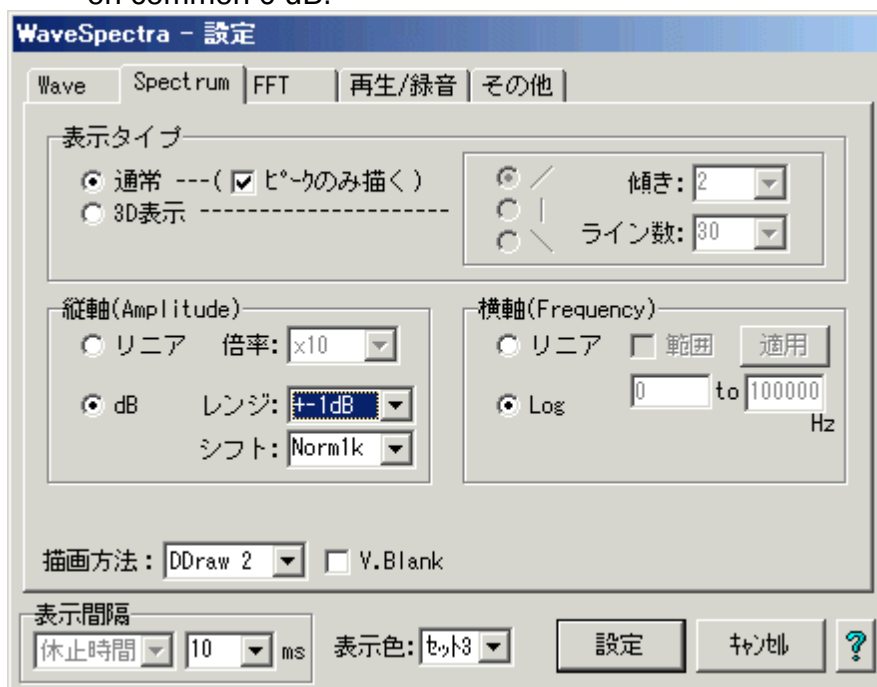
It is recommended to measure via the device under test (amplifier, speaker, etc.) after confirming that it operates stably.

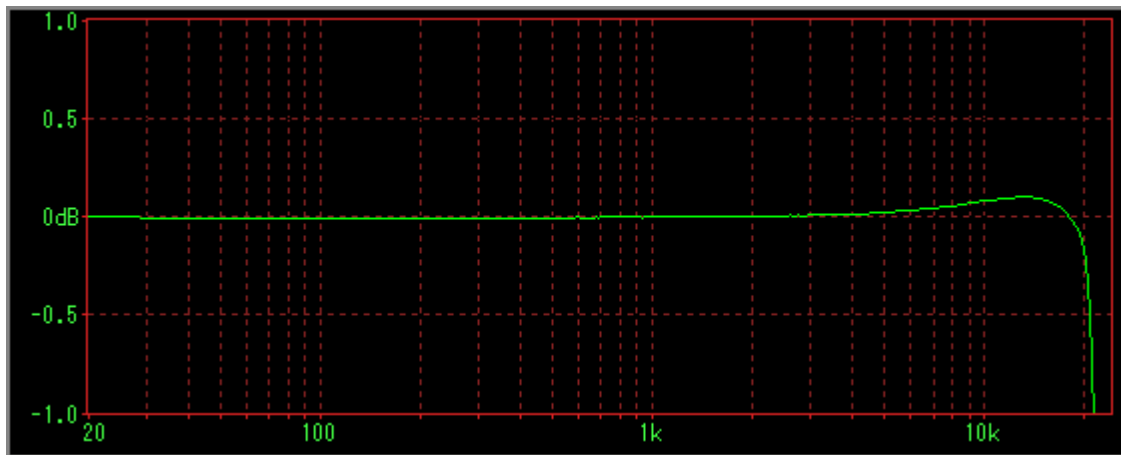


On the

- WS "Spectrum" tab, the vertical axis By changing " Range ", " Shift ", it is possible to make it easier to see the graph.

" Range " to +1 dB, etc., " shift " If you set Norm 1k etc., you can draw a graph based on common 0 dB.





" shift " of Norm 100, Norm 1 k, Norm 10 k Always have spectra of 100 Hz, 1 kHz, 10 kHz It is an option to fix to 0 dB.

It is useful to use it for graphs of frequency characteristics of filters and equalizers.

4. Attention, other

- Sound devices used for both WG and WS use the same device that can simultaneously record and reproduce.
Different devices can not be used on both sides.

Also, the clock for recording / playback must be synchronized.

(Some sound devices seem to have some clocks that can not be synchronized by recording / playing, but such things can not be used.)

Also, even when an internal sampling rate converter is included, it seems that the display of frequency characteristics may unstably shake and be unusable.
Even though the specifications are written as being supported, in reality there were cases in which it was unstable and not good at high sampling rates of 96 kHz or higher.

Both of them may be judged to be usable if the display graph of WS frequency characteristics is displayed in a stable manner.

- It is better that the input / output level is as close as possible to the maximum value (0 dB), but depending on the sound device, 0 dB There are also things that can not be input / output (distorted), so in that case please try to lower it by a few dB.
(Please check with loopback first as above)
- Since the length of the cyclic sweep is the FFT length as it is, the longer the period sweep length, the higher the frequency resolution.
(Frequency resolution = sampling frequency / number of FFT sample data)
However, it should be chosen so that the frequency characteristics of the measurement object do not change within the length of that length.

In addition, I think that it is almost unlikely to be related to normal measurement, but if the time of the impulse response of the measurement object is long, set it to a longer time.

(For example, if the time of the impulse response is 1 second, 65536 samples are required for 48000 s / s)

- Since it is necessary to operate both WG and WS stably, it is recommended to lower the load on the CPU as much as possible.

WS setting dialog box, " Wave " tab, and The " drawing method " of the " Spectrum " tab " Please set it appropriately and set the drawing method to decrease the CPU load.

For details, please see the help of WS. (Automatic setting is now possible from V1.50)

- When WG and WS can not be operated at the same time for some reason, once the WG file creation function of the WG has output the periodic sweep to the Wave file of the length necessary for the measurement, the second Please start WS and play it with it.

(At this time, if you set the second WS like the first one, you can observe the characteristics of the cyclic sweep itself.)

- Frequency characteristics can be obtained by directly applying the same periodic pulse, but the S / N is low.

During measurement by cyclic sweep, "**Waveform** combo box" Waveform "user waveform" When changing to "pulse" from "Impulse" directly, you can check the degree of improvement of S / N.

However, the cycle of " pulse " It must be the same as the cyclic sweep in the **frequency / period** combo box.

Created with the Personal Edition of HelpNDoc: [Free help authoring environment](#)

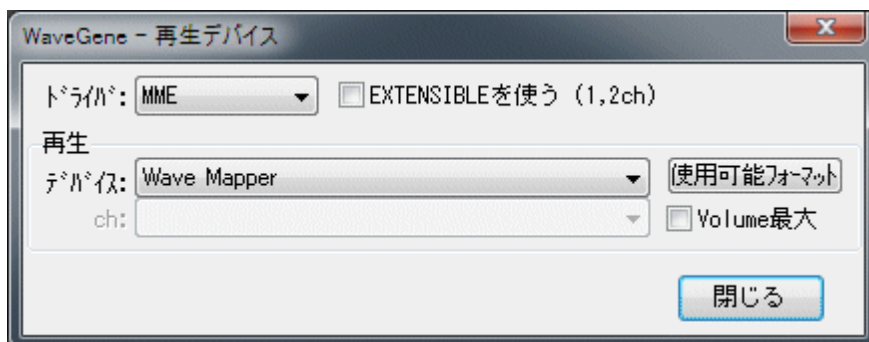
Playback device setting dialog

Playback device setting dialog

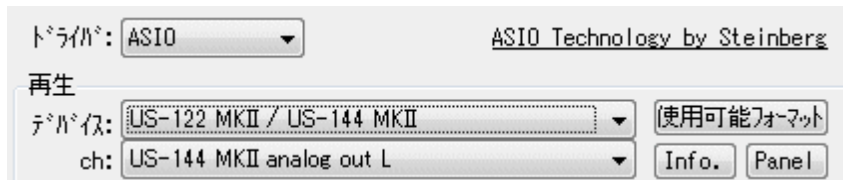


This is the setting dialog displayed when you press the **playback device button** .

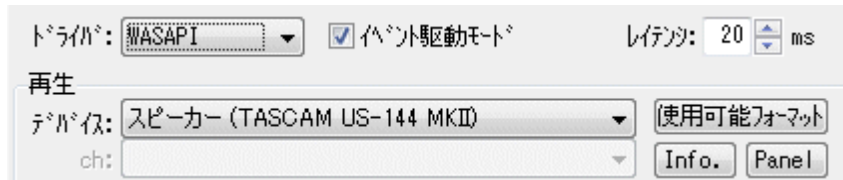
Perform various settings when using a sound device.



When ASIO



WASAPI time



=== Driver ===

- MME (Multimedia Extensions [standard WaveIn, WaveOut API])
DirectSound
ASIO (Steinberg Audio Stream I / O)
WASAPI (Windows Audio Session API) exclusive use only

We choose from four.

* WASAPI is supported on Windows Vista SP1 and later.
Other than that, it is not displayed in the list.

Using EXTENSIBLE (1, 2 ch)

- It becomes effective in case of MME, DirectSound.
If checked, it is an extended format corresponding to multi-channel multi-bit length
Open the device in the WAVE_FORMAT_EXTENSIBLE format.

* If the driver is WDM (Windows Driver Model) It can be used in the case of 24 bit, 32
bit length will be available.
(However, please note that the driver may convert merely but not actually supported
bit length.)
may not be meaningful at the moment.

Note that (1, 2 ch) As it is, it is effective only in the case of Mono, Stereo.
In the case of multi-channel of 3 channels or more, originally
WAVE_FORMAT_EXTENSIBLE It is a format.

Event Driven Mode

- It becomes effective in case of WASAPI.
There are also two types of exclusive mode of WASAPI, event drive mode and timer
drive mode, and if this check box is checked, event drive mode It works with.

In event drive mode, buffer read / write is performed by periodic event (interrupt), but
in timer mode, timing is measured with a timer.

Although event-driven people may feel accurate, there is no data missing in principle as long as they are working with Both-chan.

Normally you should put it in the event drive mode, but depending on the device there are formats that are only valid in the timer mode, so if you can not use the desired format, remove this check Please check with timer mode to see if the format is valid.

Latency

- It becomes effective in case of WASAPI.
WASAPI latency (buffer length) .
It will be a delay time of playback / recording, but if it is too short, sound will skip, and if it is too long, a problem may occur.
[Info. You can check the settable range and operation mode of the latency with the button](#) (described later).

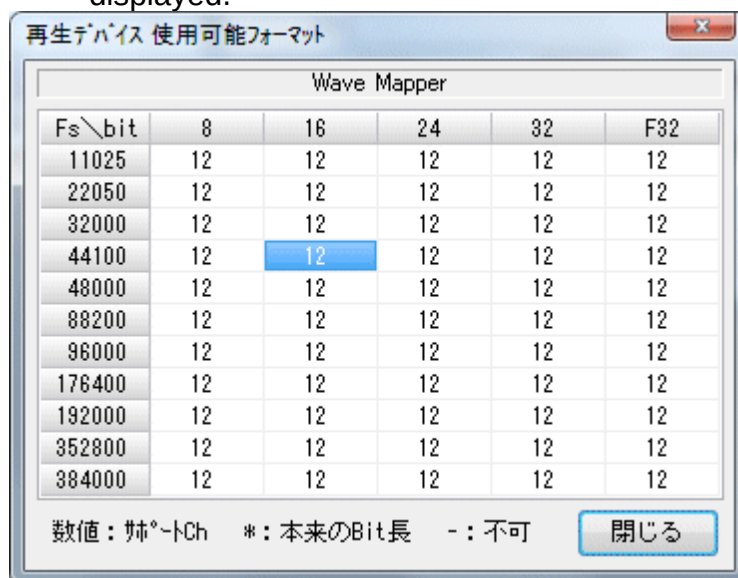
=== Reproduction ===

device

- Select the device to use for playback.
- * Wave Mapper when driver is MME and When DirectSound is selected, if the primary sound driver is selected, the device set in Windows will be used.

usable format

- When this button is pressed, a list of formats (sampling frequency, bit length, number of valid channels) that can be used with the currently selected playback device will be displayed.



- Number indicates the number of supported channels.
like "12", It means that 1ch (Mono), 2ch (Stereo), are supported.
"-" Indicates that it is a format that can not be used.
- The part of the format currently set is highlighted.

Just like setting it with [format setting](#) by setting " double click " with mouse with any of the available formats can.

(However, only the sampling frequency and the bit length can be set, please set by Stereo / Mono format setting)

- F32 of the top bit length represents 32 bit floating point format.
- In order to judge whether it can be used, the device is actually opened once in each format.
Therefore, depending on the driver it may take quite a while.

Although it seems to be able to understand the process of scanning, it takes time, so when you want to stop Please press **stop button**.

Note that while holding down the Shift key, the available format button By pressing, WAVE_FORMAT_QUERY It is possible to change to the operation just to inquire with, the time will be shortened, but some drivers report lies occasionally, so it is definitely better to actually open.

- Depending on the driver, even if it is displayed here as usable, it may not actually operate normally.
- Normally, checking is performed for 2 ch However, if multichannel data is open, that number of channels will also be checked.

The figure below shows 6ch (5.1ch) Open the data and press the usable format button.

"126" and 6ch are also checked, you can see that it is valid.

Fs\bit	8	16	24	32	F32
11025	126	126	126	126	126
22050	126	126	126	126	126
32000	126	126	126	126	126
44100	126	126	126	126	126
48000	126	126	126	126	126
88200	126	126	126	126	126
96000	126	126	126	126	126
176400	126	126	126	126	126
192000	126	126	126	126	126
352800	126	126	126	126	126
384000	126	126	126	126	126

数値: 対応Ch *: 本来のBit長 -: 不可 閉じる

- While holding down the Ctrl key, the available format button If you press, forcibly check all channels.
Even if you are not opening a multi-channel file, all channels are checked, so it is effective when you want to check all the channels.
However, it will take a lot of time.
Currently, the number of channels supported It is up to 8 channels.



- When the driver is ASIO or WASAPI, it becomes a special display.
ASIO and WASAPI, The operation bit length is fixed, and the application will convert the data accordingly.
The original operation bit length is "* 12". It is distinguished with an asterisk (*). A place without
* indicates that data is being converted.
* However, conversion to the direction in which the data length becomes shorter is not performed, so if there is validity, there is no loss of data.

In the figure below, since it is originally processed at 32 bits, we can see that all other bit length data is converted to 32 bits and output.

Here * is one place of 32 bits, but it may be attached to more than one place, of course.



Volume maximum

- It becomes valid when the driver is MME, DirectSound.
If checked, it will play with full volume.
(Some drivers do not become valid)
- Depending on the driver, it is sometimes configured to output at about -6 dB by default, especially with digital output, it can not output with the correct original value.

In that case, if you check here, you may be able to output with the correct value.

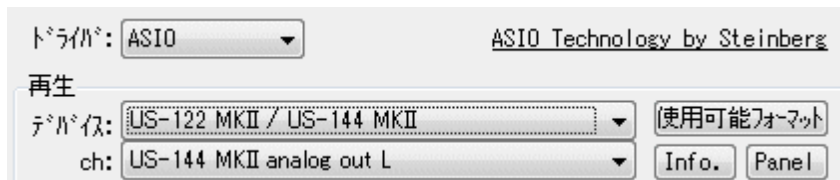
- In DirectSound, it may not be effective unless you set it on the MME side, so please set it up first in MME. (Play once)

ch (ASIOPlay ch)

- It becomes valid when the driver is ASIO.
Select the channel to use for playback.
The specified channel will be on the Lch side of the stereo.
- When multiple channels are available, normally 0 or 2 (channels are When it starts from 0), even-numbered channels are specified, but there is no distinction in the channel itself, so odd-numbered channels may be specified.
In that case, if 2 channels are considered as a normal stereo pair, it will be output to the original Rch side of one pair and the Lch side of the next pair.

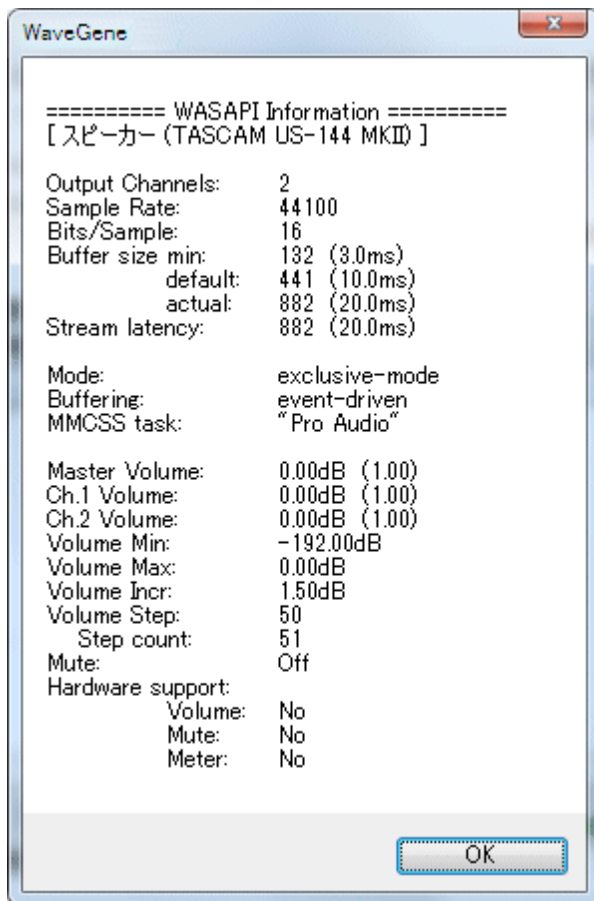
For example, when there are 4 channels in total including AnalogOut 1, AnalogOut 2, DigitalOut 1, DigitalOut 2 When ch is set to AnalogOut 2, data of Lch is output to AnalogOut 2 and data of Rch is output to DigitalOut 1.

Note that in the case of monaural file playback, it is output to only one channel.



Info.

- This button is enabled when the driver is ASIO and WASAPI.
If you press this button normally, the current ASIO or WASAPI The driver setting information is displayed, and you can check the number of channels, sample rate, bit length, latency and so on.
- * For WASAPI, hold down the Shift key and press this button to display more detailed information.



Panel

- This button is enabled when the driver is ASIO and WASAPI.
 Press this button, ASIO Control, ASIO Control You can open Panel.
 (Nothing happens with unsupported driver)

WASAPI If it is, open the sound control panel (playback).

Created with the Personal Edition of HelpNDoc: [Easy to use tool to create HTML Help files and Help web sites](#)

Command line parameters

Command line parameters

Launch icon property - shortcut Command line parameters By setting it, it is possible to read the setting parameter file specified at startup, start up with that setting, or play it as it is.

By creating shortcuts of multiple settings, it is convenient as it can generate various signals.

Command line and parameter setting example

- WG.EXE PARAMFILE.WG

PARAMFILE.WG Start up with the setting.
 If already activated, change the setting.

- WG.EXE PARAMFILE.WG /P

When / P (/ p is also possible) is attached, it automatically reproduces with that setting after changing parameters.

Wave file length / sweep length Play only for the length (seconds) set in the combo box, then stop.

- WG.EXE PARAMFILE.WG /P5000

Also play automatically.

However, the number after / P (ms here, thus 5 seconds in the example above) After playing only during, stop.

(Preferred length designation here)

About parameter file name

- * The extension .WG is absolutely necessary.
- * If the directory name is not added and the file name is only as shown above, it is assumed that it is in the same directory as WG.EXE.
When it is located elsewhere, please specify it with full path.
(In a shortcut "Work folder" Note that setting is irrelevant)
- * If the file (path) name contains spaces, double quotes("")Please pinch both ends with.

Created with the Personal Edition of HelpNDoc: [Easily create CHM Help documents](#)

Operation with keyboard

Operation with keyboard

Each button can also be operated with a keyboard.

- Ctrl + O (o): Read parameters
- Ctrl + S: Save parameters
- Ctrl + W: Output to Wave file
- Ctrl + D: Select playback device
- space: alternately output to sound device - Same as stop.

However, except when the focus is in an editable combo box or the like.

Therefore, to turn on / off output in space, please remove focus from an editable combo box.

(Select waveform combo box etc.)

- Alt + F4: End

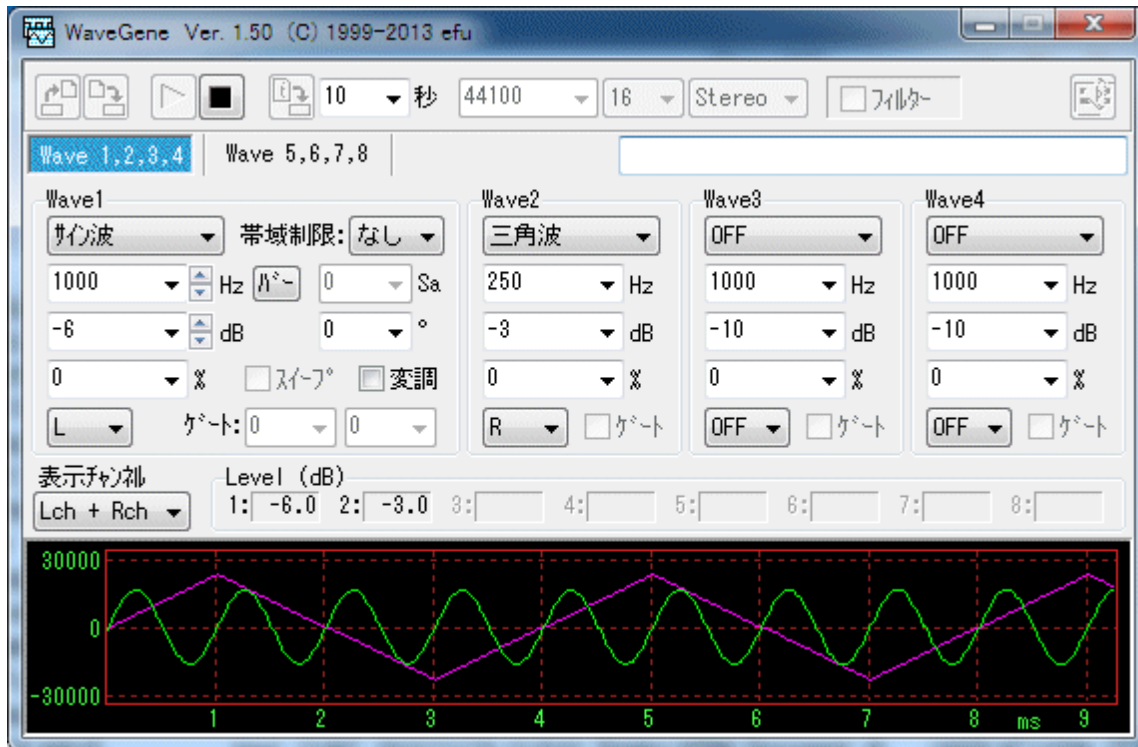
Created with the Personal Edition of HelpNDoc: [Easy CHM and documentation editor](#)

Setup Examples

Setting example

[Operation example](#) is shown as an example.

< 1 > sine and triangle wave

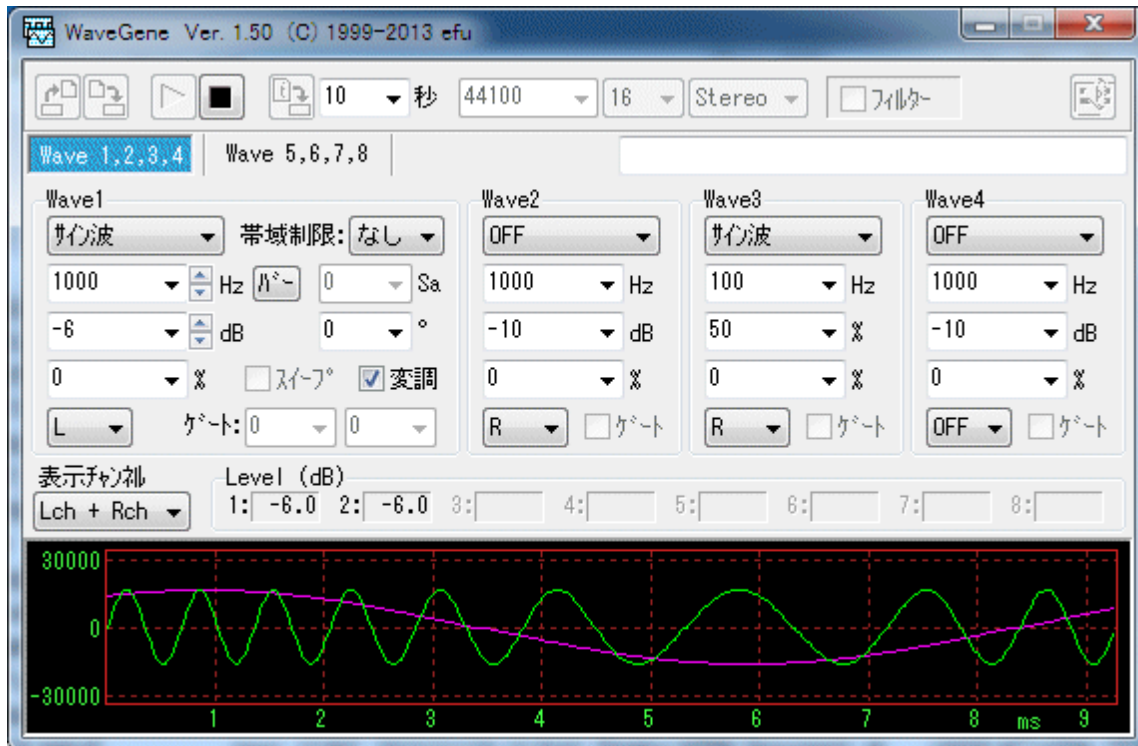


- Simply set Wave 1 and Wave 2 to sine wave and triangular wave respectively and output to L and R.

	Wave 1	Wave 2
Waveform	Sine wave	Triangular wave
Frequency	1000 Hz	250 Hz
Amplitude	-6 dB	-3 dB
Output channel	L	R

* Since the triangle wave here is Wave 2, the band is not limited. (Including aliases)
If necessary substitute Wave 1 or change to Wave 5 and use the [band limit](#) waveform.

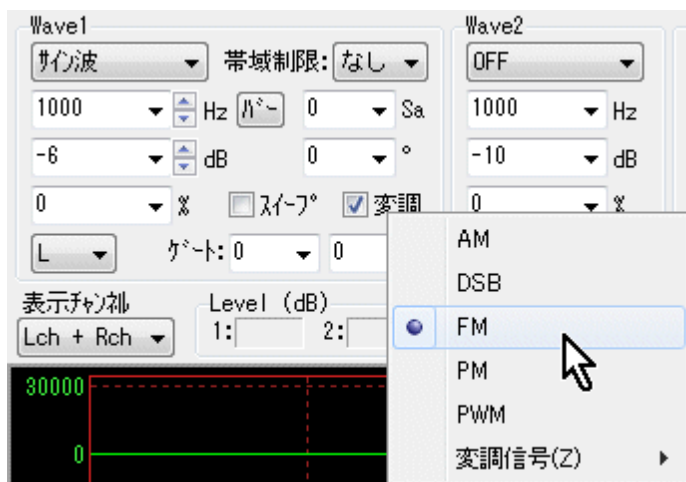
< 2 > FM (frequency modulation)



- [Frequency Modulation](#) at 1 Hz of Wave 1 at 100 Hz of Wave 3.
Since the degree of modulation is set to 50%, Wave 1 changes between 500 Hz and 1500 Hz.
Carrier (carrier, modulated wave) Wave 1 is output to Lch with amplitude of -6 dB and modulation wave Wave 5 is output to Rch as it is.

	Wave 1	Wave 3
Waveform	Sine wave	Sine wave
Frequency	1000 Hz	100 Hz
Amplitude	-6 dB	50%
Output channel	L	R

Then check the **modulation** of Wave 1, press the right button on it to display the pop-up menu and check **FM**.



< 3 > warble tone

- In the so-called wavble tone, the frequency modulation and setting of [2] are the same, but a specific setting expression for changing the range of arbitrary frequency is shown.

When changing the frequency from f_1 to f_2 (here, $f_1 < f_2$), the set frequency f of Wave 1 is obtained by $F = (f_2 + f_1) / 2$ [Hz]
I will.

The amplitude a of Wave 3 that determines the degree of modulation is set by%, and $A = (f_2 - f_1) / (f_2 + f_1) * 100$ [%]
Set to.

For example, if you want to change between 900 and 1100 Hz, $F = (1100 + 900) / 2 = 1000$ [Hz]
 $A = (1100 - 900) / (1100 + 900) * 100 = 10$ [%]
So it should be set as below.

	Wave 1	Wave 3
Waveform	Sine wave	Sine wave
Frequency	1000 Hz	10 Hz
Amplitude	-6 dB	10%
Output channel	L + R	OFF

Of course, the amplitude of Wave 1 and the frequency of Wave 3 which is the speed of change are arbitrary.

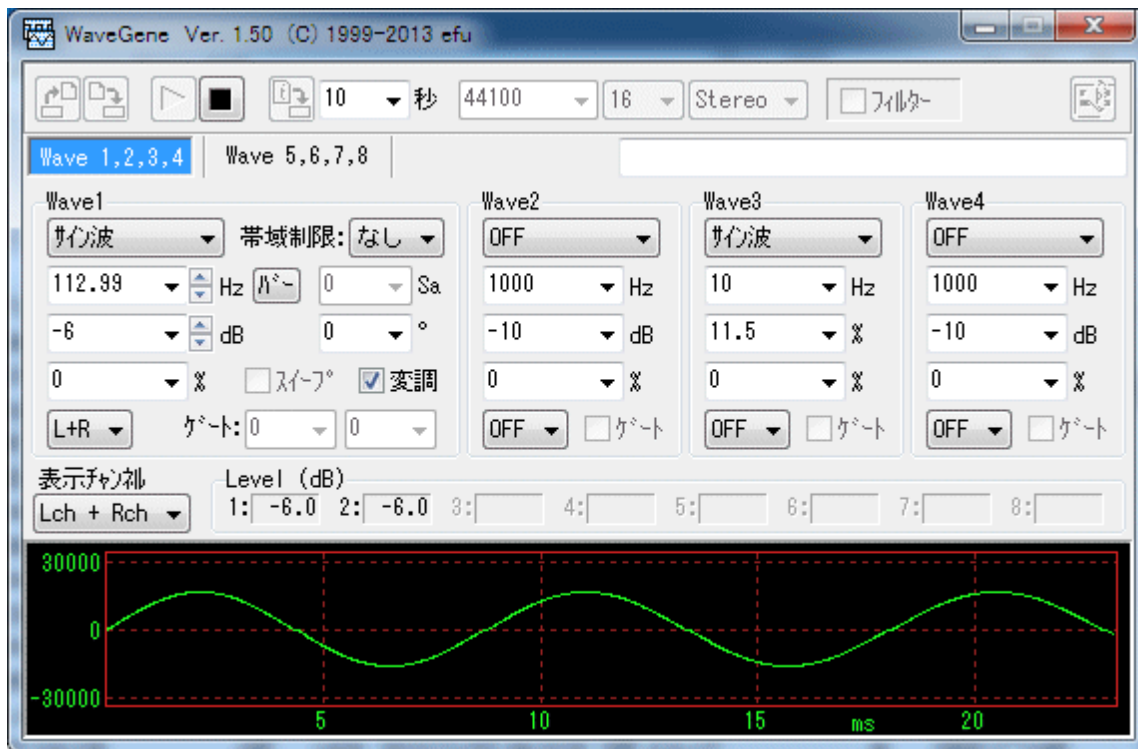
(Unlike [2], output of modulated wave is unnecessary, so it is set to **OFF**)

Furthermore, if you want to create a 1/3 octave wobble tone centered on an arbitrary frequency, Since the ratio of 1/3 octave is 2 cube root = 1.259921,
 $F_2 = 1.259921 * f_1$ and the amplitude a of Wave 3 which determines the degree of modulation is obtained by using $A = (1.259921 - 1) / (1.259921 + 1) * 100 = 11.50$ [%]
Therefore, set the desired center frequency f to Wave 1 and set the amplitude of Wave 3 to 11.5%.

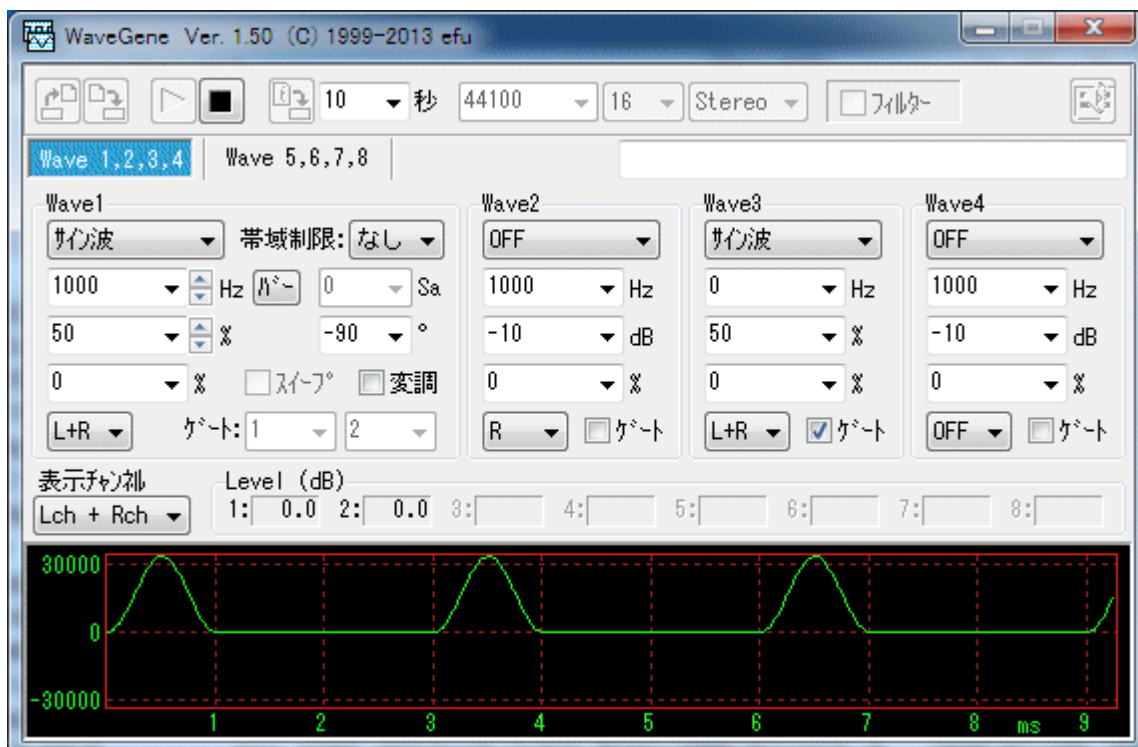
When it is desired to determine f_2 , f_1 , for example, between 100 and 200 Hz, the interval is 100, 125.99, 158.74, 200, so it is 100 to 125.99 Hz , The center frequency f is the same as above, $F = (125.99 + 100) / 2 = 112.99$ [Hz]

It should be set to Wave 1.

The amplitude setting of Wave 3 does not change at 11.5%.



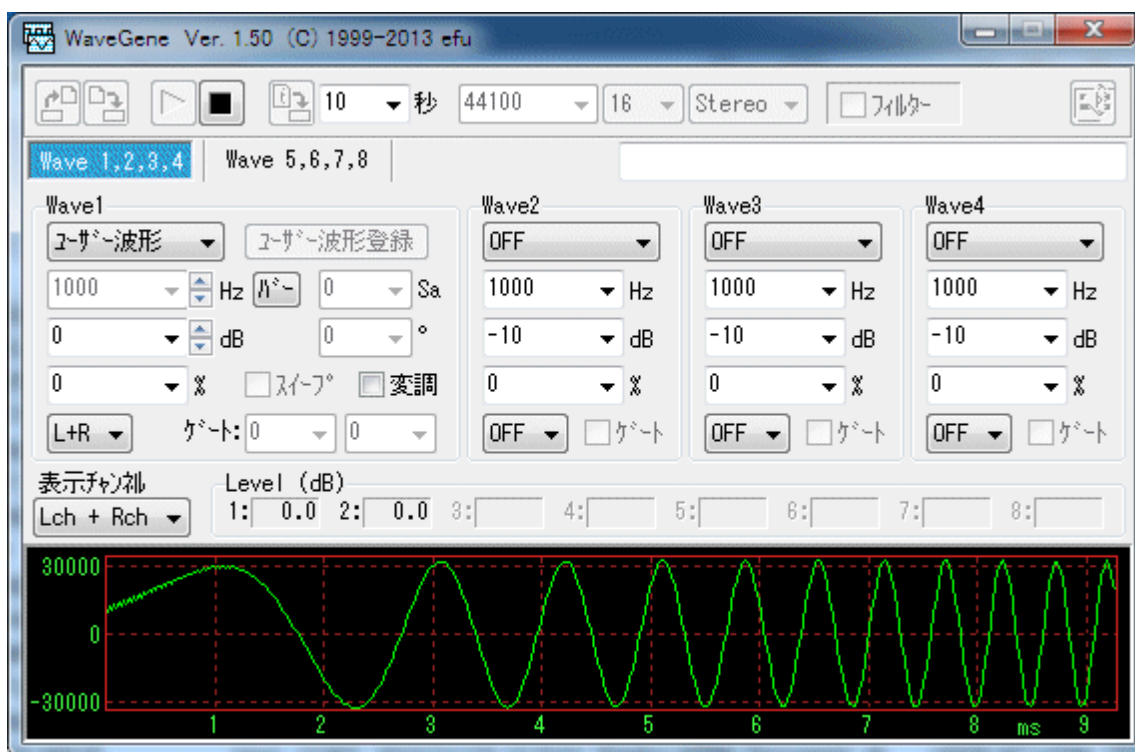
< 4 > raised cosine burst



- Gate the 1 kHz of Wave 1 at a 1: 2 cycle (1 cycle in 3 cycles On), synchronize with the same gate and DC (0 Hz) with amplitude of 50% of Wave 3 Let it raise it up by half with a pulse made by switching.
Set the phase of Wave 1 to -90 degrees and start from the negative maximum value of the sine wave.
The output is L + R and both channels mono output.

	Wave 1	Wave 3
Waveform	Sine wave	Sine wave
Frequency	1000 Hz	0 Hz (direct current)
Amplitude	50%	50%
Output channel	L+R	L+R
Gate	On: 1 Off: 2	On (synchronized to 1)

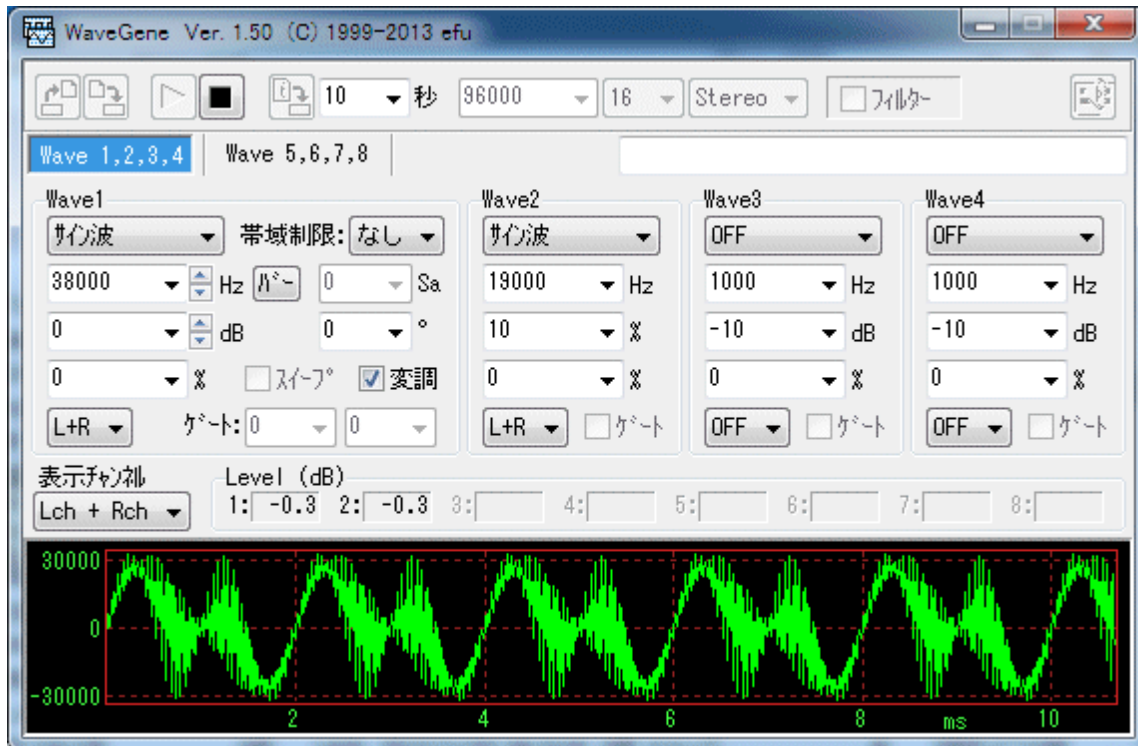
< 5 > user waveform



- This is simply generated by loading the user waveform to Wave 1.
Frequency, phase, sweep and gate settings are invalid in user waveforms.

	Wave 1
Waveform	User waveform
Frequency	invalid
Amplitude	0 dB (optional)
Output channel	L + R

< 6 > FM stereo composite signal (L: 500 Hz R: 1000 Hz)

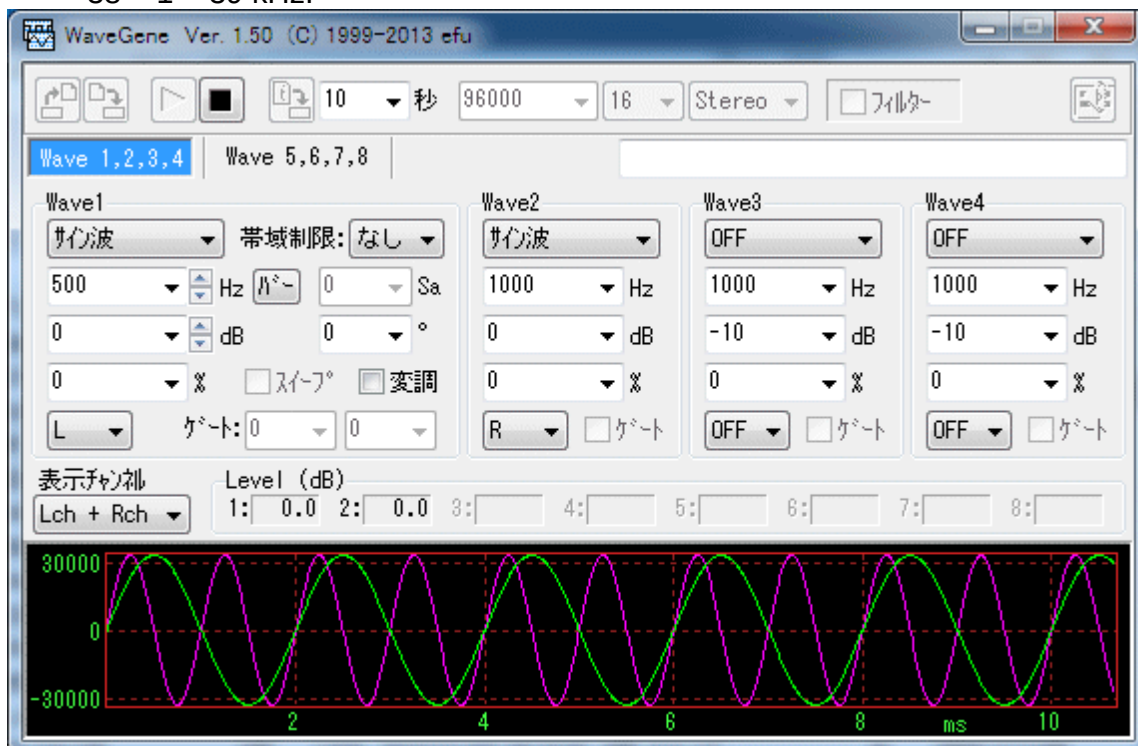


- Composite signal of FM stereo broadcasting, stereo signal of Lch 500 Hz, Rch 1000 Hz maximum volume.
When this is input to the FM transmitter, it can be received as a stereo broadcast. (I actually sent it and confirmed it)

The procedure is a bit complicated.

[1] First, create a Lch 500 Hz, Rch 1000 Hz Wave file (maximum amplitude) to be transmitted.

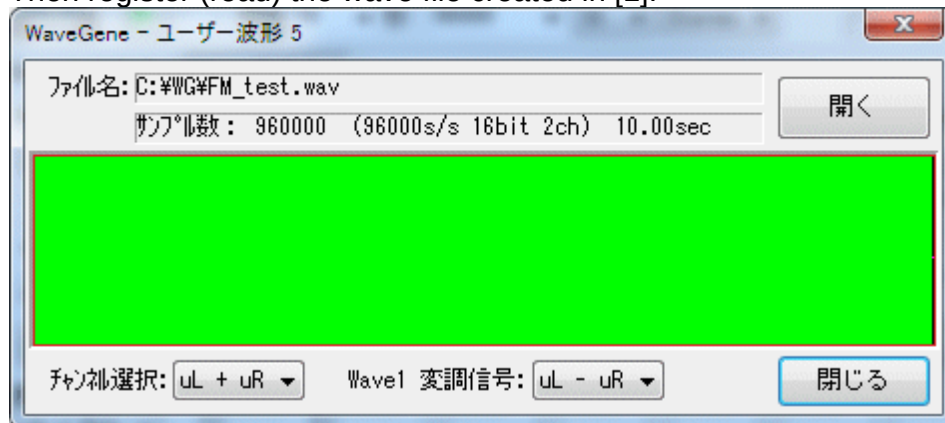
The sampling frequency is set to 96000 Hz since it requires a bandwidth of at least $38 + 1 = 39$ kHz.



	Wave 1	Wave 2
Waveform	Sine wave	Sine wave
Frequency	500 Hz	1000 Hz
Amplitude	0 dB	0 dB
Output channel	L	R

Now press the **Output to Wave File** button and create a 10-second Wave file (example: FM_test.wav) for the time being.

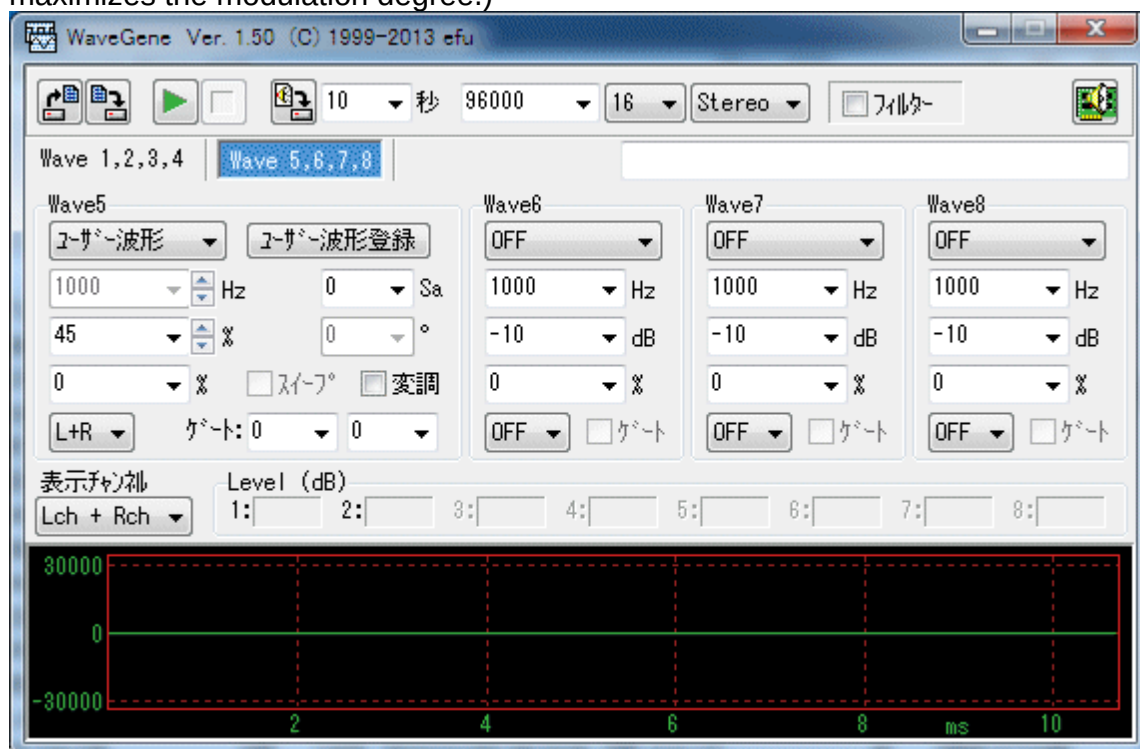
[2] Next, switch to Wave 5 - 8 on Wave 5, 6, 7, 8 tab and **Wave 5 User Waveform**. Then register (read) the wave file created in [1].



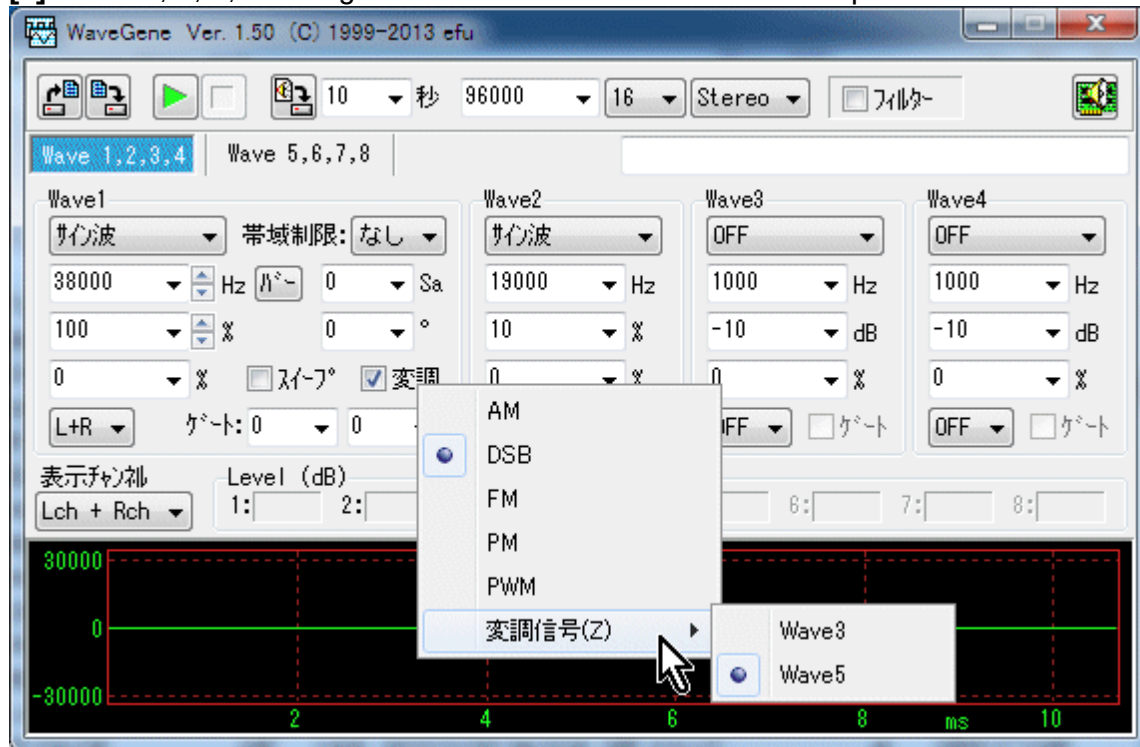
uL + uR , Wave 1 modulation signal uL Do not forget to set it to - uR .

The sum of the left and right of the transmission signal is used as it is as a monaural signal, and the difference between the left and right is used as the modulation signal of the 38 kHz subcarrier modulation (DSB).

[3] When loading the user waveform into Wave 5, change **amplitude** setting to **45%** and **output channel** Set it to **L + R** . (Since the pilot signal is 10%, 45% is the value that maximizes the modulation degree.)



[4] Wave 1, 2, 3, 4 tab again to switch to Wave 1 - 4 and set up Wave 1 and Wave 2.



	Wave 1	Wave 2
Waveform	Sine wave	Sine wave
Frequency	38000 Hz	19000 Hz
Amplitude	100%	10%
Output channel	L + R	L + R

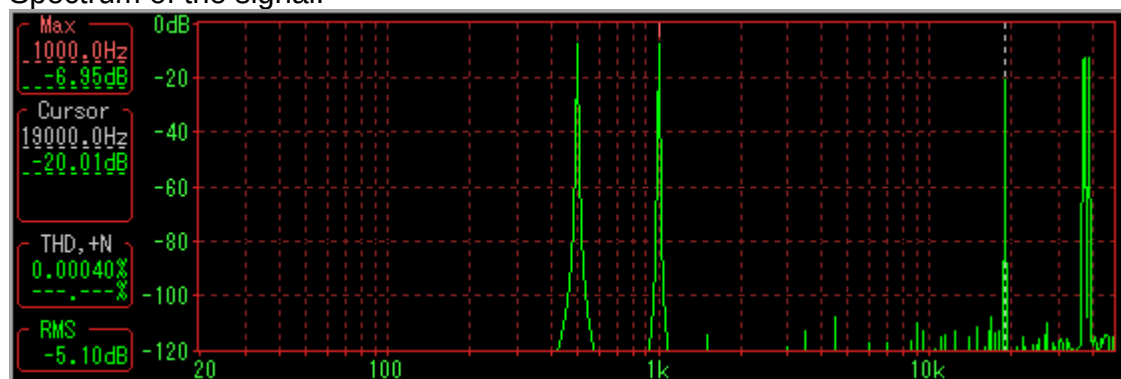
Wave 1 is the subcarrier, and Wave 2 is the pilot signal.

Also check the **modulation** of Wave 1 as shown above, **DSB** with the right button and **Wave 5** with **You have selected**.

The output is **L + R** with both channels mono output. (Actually, only one is enough)

You can create a composite signal by creating a playback button or a wave file.

Spectrum of the signal.



< 7 > FM stereo composite signal (music signal version)

- If you want to create with a music signal, not just a simple sine wave signal like the above <6>, instead of the Wave file created in the above [1], You can load Wave file of music signal as Wave 5 user file with [2].

However, in order to create with the sound quality of your choice, it is necessary to apply pre-emphasis to the signal Wave file in advance with a low pass filter up to 15 kHz.

In addition, since it handles signals up to 15 KHz, it is necessary to create the sampling frequency at 176.4 kHz or 192 kHz.

* Assuming that a general 44.1 kHz Wave file is used for music signals, the sampling frequency of the creation format is set to 176.4 kHz which is four times.

* Filter uses output filter [Example 2] [15 kHz lowpass + 50 uS emphasis](#).

[1] First of all, the sampling frequency of the created format is set to 44100 which is the same as the wave file.

[2] Import the original music wave file into the user waveform of Wave 1 or Wave 5. Keep other Wave **Waveform** settings **OFF**.

Check the ' [3] **output filter** to make it effective and set the characteristic curve of [15kHz lowpass + 50uS emphasis](#) in the filter setting And **oversampling** must be set to **x4**.

[4] **Wave file length** Combo box  to the same length as the wave file)
Output to Wave file button  to name another Wave file. (Example: FM_test.wav)

* At this time, when you press the " Save " button with the " Output to Wave File " , Be sure to hold down **Ctrl key** and press the " Save " button.

This creates a Wave file with a sampling frequency four times the filtered frequency, ie 176.4 kHz. ([Oversampling Background](#))

* Since pre-emphasis is applied with the output filter, files created may over-level and clip depending on the high frequency range of the original wave file.

While checking the level display, create a file, if it exceeds 0 dB (red display), reduce the amplitude setting from that by 0 dB and create it again.

[2] of the previous <6> (**L: 500 Hz: 1000 Hz**) after changing [5] creation format to 176400 Do the same in the Wave file that created the "strong" or later".

If you play this composite signal and send it on the FM transmitter, you can receive it properly as a stereo broadcast.

(Be careful with the input level of the transmitter so that it does not become overmodulation when actually sending it, it should be set to be about the same volume as real broadcasting.)

Hints

Hints collection

About waveforms other than sine waves

- Up to V1.40, periodic waveforms other than sine waves simply produced graphical waveforms (for example, binary waveforms in the case of rectangular waves) and output them as they were.

These are useful as tests and modulated signals for hardware and software, but because they contain components above the sampling frequency / 2 (Nyquist frequency), there is aliasing, and for correct signaling to be used as a voice signal (listening) There was no.

Since V1.50 it is now possible to generate [band limited](#) signal, so please use this when using as audio signal.

Also, since the [output filter](#), which generates a signal by oversampling, has also been added, as with the above waveforms where loopbacks fit in the band cut by decimation It is also possible to create a band limited to the band.
(Although it is faster to use the band-limited waveform itself from the beginning)

Supplement:

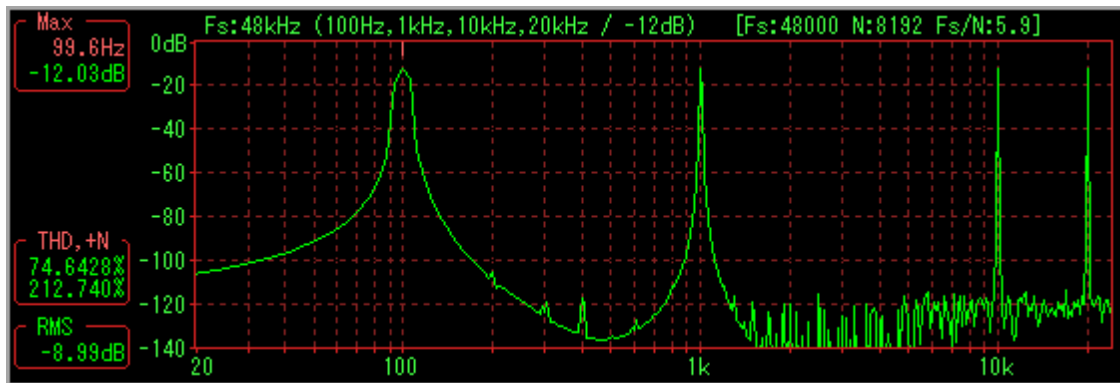
Even for graphical waveforms, if you set the number of samples with an even period and the number of samples to 2 N as described below, a spectrum without aliasing can be obtained for the time being, but this is also strictly a correct signal There is none.

For example, when a rectangular wave of 8 sample periods, that is, 6000 Hz is created at a sampling frequency of 48 kHz, the spectrum shows a fundamental wave of 6 kHz And the third harmonic 18 kHz, and there seems to be no alias.
However, at this time, the harmonic components are odd multiples of 18 kHz, 30 kHz, 42 kHz, 54 kHz However, 30 kHz is 18 kHz in return - 30 = 18), 42 kHz likewise becomes 6 kHz (48 - 42 = 6) all overlaps the frequency seen by the spectrometer.
Therefore, since the component of the third harmonic 18 kHz originally has an amplitude of 1/3 of the fundamental wave 6 kHz, it should not have become the level difference - 9.54 dB, but actually It is -0.68 dB and -8.33 dB when making a rectangular wave with an amplitude of -3 dB, which shows that the level difference is - 7.65 dB, which is not a correctly band-limited rectangular wave.

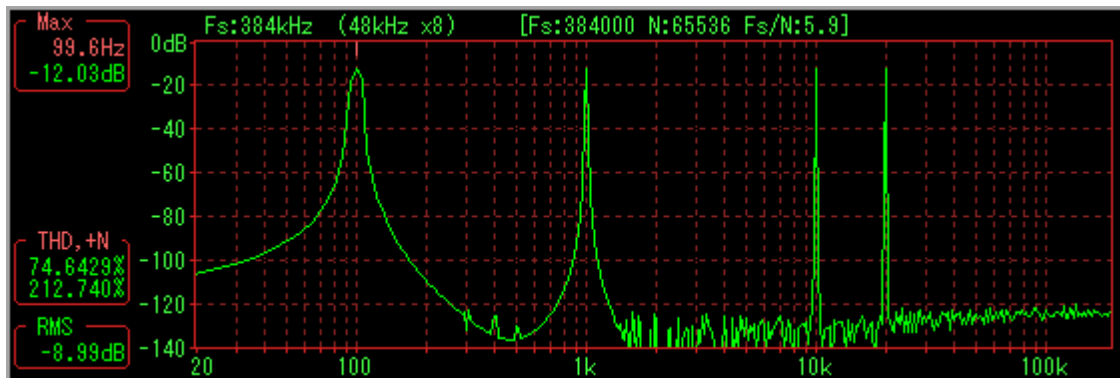
Sampling frequency conversion (interpolation) of Wave file using output filter oversampling function

- By using output filters [oversampling file creation function](#), it is only x2, x4, x8, x16, but sampling frequency conversion of the wave file (interpolation You can do it.
For details, please refer to the upper link destination.

48 kHz before conversion



384 kHz after x8 conversion



About pronunciation latency

- [filter length](#) +1 in the [filter settings window](#) becomes the basic buffer length for internal operation. It is getting.

For example, when filter length 4095, that is, buffer length 4096 at 48000 Hz sampling, one block operation of the filter is $4096/48000$. It will be updated every approximately 85 ms.

In other words, internal waveform updates are done at this time interval. (However, the actual waveform display is limited to about 10 fps at the maximum, and if it is faster than that, the display will be thinned out in order to reduce the drawing load)

When shortening the filter length and setting the buffer length to 1024, the response time becomes as fast as about 21 ms and response such as the characteristic display at the time of changing the filter characteristics becomes faster, but the actual pronunciation latency is different from this, it is fixed by taking multiple buffers so that it is usually at least about 0.5 seconds. (To prevent skipping)

Therefore, the response of the change in sound generation when turning the mouse wheel is always about 0.5 seconds, so even if the filter length is shortened, this response time does not become faster.

If you want to shorten the actual pronunciation latency, after ending it once, add the following description to any position in the Misc section in WG.INI and re-execute.

```
[Misc]
PlayBuffer_ms = 100
```

Set it to the extent that sound skipping does not occur in ms. (10 to 2000)
(Make sure to skip if you move the window or activate another software)

Note 1:

Conversely, if the sampling frequency is 22050 Hz and the filter length is set to 16384, the internal buffer will be 0.74 seconds in length, in which case the pronunciation buffer will also be taken as much, so the pronunciation latency will also be the lowest It will be 0.74 seconds.
(It will be 0.74 seconds even if it is set to less than this with PlayBuffer_ms.)

Note 2:

When the sampling frequency is high and the internal buffer length becomes short, it is adjusted not to be less than 20 ms.
For example, when the buffer length is 1024 at 96000 Hz sampling, the response time is $1024/96000 = 10.7$ ms, which is less than 20 ms. In this case It is doubled and changed to 2048.
(However, because the waveform display will remain 1024 samples to maintain consistency with others, there are no apparent changes in appearance in particular)

In order to eliminate cropping error during FFT with waveform other than sine wave

- In order to eliminate error due to data extraction at observation by a spectrum analyzer with a periodic waveform other than a sine wave such as a rectangular wave, [extended setting function of frequency setting](#), It is not enough to just set the frequency to an integral multiple of the minimum frequency resolution of the FFT. In order to do so, the number of samples in the positive part and the negative part of the waveform must be an integer and equal, but in most cases it will not.

In order to do so, the frequency is set to the minimum frequency resolution of FFT Set to 2 N times power.

That is, conversely, if the number of samples as a period is increased to "N" power of "2" It is good to set it. (It is preset in the cycle combo box.)

For example, the frequency (44100 / number of samples) in the case of sampling frequency 44100 Hz is

Periodical sample number frequency

1024: 43.07 Hz

256: 172.27 Hz

64: 689.06 Hz

16: 2756.25 Hz

And so on.

About noise

- Both white noise, pink noise, M series noise, Wave 1 ~ Wave 8 output the same timing signals (same time series).
For example, both Wave 1 and Wave 2 are set to the same white noise or pink noise, and Wave 1 When outputting to Lch and Wave 2 to Rch, both output signals are the same and become completely monaural signals.
- White noise is _lrnd (cycle 2^{64} multiplication congruential method) It is a normal random number generated by the Box-Muller method from the uniform random number of
As a measure to secure power to some extent and deviation It is normalized by

multiplying the limiter by + -3.5.
(The limit rate is applied is about 0.023%)

- Pink noise is -10 dB / dec for white noise (-3 dB / oct) filter.
In WaveGene, this filter is performed by FFT filtering, and in order to generate it in real time, it requires more CPU power than other signals.

Although it is not worried at this time in the CPU, if the sound is interrupted, please write it to a wave file once and then play it with another Wave file playback software.
(From V1.40, it can be generated by reading to user waveform)

If you really want to output from a sound device in real time, Stereo It may be able to improve by changing from Mono to Sampling Frequency.

(Depending on the operating environment, generally Pentium It seems that it will not sever if 133 ~ 166 MHz or more)

- M series noise is the cycle $2^{32} - 1$.
- * Dither
Set one wave to noise, for example in the case of 16 bit amplitude You can also use it as dither by setting it at around -0.5 dB or "quote; value " with the other wave.

About internal calculation accuracy

- All real operations including FFT are double (64 bit floating point) I am doing it.
Computation and mixing are calculated by double, and finally It is converted (rounded) to 32, 24, 16 bit or 8 bit.
Note that the integral parts are optimal integers respectively.

About DC component

- Since it became possible to add an offset independently from V1.10, it became very meaningless, but in a waveform that can set the frequency, By setting " zero ", it is possible to add direct current to the output by the amplitude value at that time.
(Values may be positive or negative)

Inversion (reverse phase) output method

- o Especially, there is no button etc for polarity reversal, but when the unit is other than dB, setting the "negative" value as the amplitude enables you to invert the Wave output.
In the case of dB, you can invert it by changing it to a value or% and then adding a '-' at the beginning.
Example: When you want to output white noise with opposite left and right phases
Both Wave 1 and Wave 2 are set to white noise, the output channel is set to Lch for Wave 1, Rch for Wave 2, and set the amplitude to the same value and make only one of them "negative".
Waveforms other than noise are apparently apparent if the phase setting of Wave 1 is set to 180°.
However, this method can be used to make exactly opposite phases.

About level over of output signal

- o In the case of level over, the over level value is displayed as it is with the background as " **red** " The actual output signal of the current format is the maximum value in the current format (at the time of 16 bits - 32768, 32767, It is designed to clip with 8-bit when -128, 127).

Of course the signal at that time is distorted, it is different if you intend, but in general you need to make each setting to avoid level over.

* Even if the level display is 0.0 dB, if the background is "red", it is still clipping.
(← 0.04 dB over)

About the name of the output file

- o Wave The name displayed in the dialog box when saving each of the output file and the setting parameter file is to inherit the name added at the time of saving first either to the other by changing only the file extension It is.

Sweep speed in frequency sweep

- o When you observe the frequency characteristics using a frequency sweep in combination with a spectrum analyzer like the sister software WaveSpectra, the faster the sweep speed, the lower the value than the original value due to the influence of the change speed.
Especially in the Log sweep, the higher the frequency, the faster the change speed, the lower the graph.
If using a linear sweep instead of a Log sweep, the drop will be constant.
For more information, please see the help of WaveSpectra.

About repetitive frequency sweep using frequency modulation

- o The original sweep by the frequency setting bar [sweep function](#),  [manual sweep with mouse wheel?](#)
However, it is also possible to output repeated sweeps using the frequency modulation function.
(From * V1.40, it is now possible to output repeated sweeps simply by outputting a single sweep to a user waveform after outputting it to a Wave file etc.)

By choosing FM in the modulation, the frequency f of Wave 1 and the modulation wave Wave 3 (Wave 1) are multiplied by the expression below the lower and upper frequencies f_1 and f_2 of the frequency range to be swept The sweep can be repeated in that range by obtaining the amplitude a of the waveband, selecting the sawtooth wave as the type of Wave 3, and setting the reciprocal of the repetition period as the frequency of Wave 3.

* When changing from frequency f_1 to f_2 (here, $f_1 < f_2$), the set frequency f of Wave 1 is calculated as follows:

$$f = (f_2 + f_1) / 2 \text{ [Hz]}$$

* The amplitude a of Wave 3, which determines the degree of modulation, is set by%, and
 $a = (f_2 - f_1) / (f_2 + f_1) * 100 \text{ [%]}$

(Example 1) When repeating sweep with 2 second period from 1000 Hz to 2000 Hz

As the frequency of Wave 1, $(1000 + 2000) / 2 = 1500 \text{ Hz}$ set

Set the amplitude of Wave 3 as $(2000 - 1000) / (1000 + 2000) * 100 = 33.333\%$

Set $1/2 = 0.5 \text{ Hz}$ as the frequency of Wave 3

Set the type of Wave 3 to sawtooth wave

(Example 2) Case of repeating sweep with 10 second cycle from 20 Hz to 20000 Hz

As the frequency of Wave 1, $(20 + 20000) / 2 = 10010 \text{ Hz}$ Set

Set the amplitude of Wave 3 as $(20000 - 20) / (20 + 20000) * 100 = 99.8\%$

Set $1/10 = 0.1 \text{ Hz}$ as the frequency of Wave 3

Set the type of Wave 3 to sawtooth wave

When choosing a sawtooth wave for Wave 3, it becomes a linear sweep.

About temporarily stopping waveform display and prohibiting clearing of waveform display at tone generation stop

- Usually, the waveform displayed on the waveform display section is cleared at the same time when you stop the sound by pressing the stop button, but if you stop while holding down the Ctrl key, the waveform is not cleared and the display is stopped. You can do it.
- Furthermore, from V1.50, it is now possible to pause the waveform display at any time while holding down the Ctrl key.
- Also, when you first start, press [Ctrl] **Output to Sound Device** button  Starting while holding down the key, you can display the waveform of the starting point. (Hold down the Ctrl key as needed)
It is useful when you want to temporarily stop the waveform for confirmation, for example when you make complex settings.

About customizing the waveform display color

- Customization by dialog etc. can not be done, but still it is possible. You can customize WG.INI to an arbitrary display color by directly editing it with an editor.
In Windows color notation (hexadecimal), " BGR " in this order.

Attention when using simultaneously with the sister software WaveSpectra

- In case of simultaneous use, the output of WaveGene is interrupted. WaveSpectra's display may be jerky.
Regarding sound interruption, if you try to activate WaveGene, you think that it will be uninterrupted for the time being, but for stable use, WaveSpectra In the setting on the side, "display interval", "quiet time" To 0 other than 0.
For more information, please see the help of WaveSpectra.

About multi-client support of ASIO driver

- When using with the sister software WaveSpectra at the same time, when using both ASIO drivers of the same device, it is necessary for the driver to support multi-client (and multi stream?).
For more information, please see the help of WaveSpectra.

About error when using MME driver

- Depending on the embedded audio codec, Wave There seems to be a case that it hangs up with an error when format listing is done with Mapper.

In that case, do not use that CODEC, or if you can not do so, WG.INI In the [Misc] section of the page, " Notepad " For example,

Use_WF_DIRECT = 1

Add the line and re-execute it, the device will be opened with the WAVE_FORMAT_DIRECT flag, so please try to avoid errors.
(Please be aware that MME can not be used on Windows Me, 98/95)

Others

Other

* This program is written with Borland C ++ Builder and MS VC ++ (DLL).

* It will operate on Windows XP or later.

[Acknowledgment]

To create this html help, I was allowed to use UNA 's free software " Help "
Thank you for the wonderful software.

Frogs

[Http://www3.to/kaeru-machine/](http://www3.to/kaeru-machine/)

I used WoodyBells' free software " WinShot " for screen capture for this help. Thank you for the wonderful software.

[Http://www.woodybells.com/winshot.html](http://www.woodybells.com/winshot.html)

[Acknowledgment]

Thanks to Mr. Ross Bencina

IASIOThiscallResolver (calliasio - thiscall Adapter for the IASIO interface) (public domain / open Source)

<http://www.rossbencina.com/code/iasio-thiscall-resolver>

[Disclaimer]

The author is not responsible for any damage caused by using this program.

* This program is free software.

Copyright is in efu.

* In case of non-profit, you can use it freely.

Also please feel free to post the web results of the measurement results with this program.

(Of course you will be happy if you let me know (^ ^ ;;)

You can use it even at companies etc. Please contact us first, please do not use without permission.

* If you can post to magazines etc., please be sure to contact us by e-mail beforehand.
Please do not publish without permission.

* Donation welcome

We are sorry to ask you for your donation from time to time, so we made a window for you

once.

I would like you to appreciate the amount that you received, and devote it to the purchase cost of hardware / software for development / verification. Thank you.

Freeware originally no use restrictions, but **Vector share registration** Please use remittance service with shareware treatment by.

It is sister software WaveSpectra's, but please inform here in common.

The share registration number is **SR 272800** is.

Remittance page,

http://shop.vector.co.jp/service/servlet/NCart.Add?ITEM_NO=SR272800

Price is **630 yen** (500 yen + 5% consumption tax + fee 105 yen) is.

I wanted to lower price, but the above price is Vector Shareareji Please note that it is the lower limit of setting.

Also, as the information on the remitter can not be told to us by the terms of the share registration service, even if you remit, we can not contact you and give thanks for your name.

Thank you for your understanding about this.

[Contact Information]

Report of disability, impressions, hope, other, thank you here.

E-Mail: efu@efu.jp.net (changed on 8/01/2014)

However, please be aware that HTML mails, e-mails with attachments are automatically deleted and can not be received at all.

*** Especially when you use Outlook (Express), please note that it is often HTML mail settings.**

In "Sending" tab of "Tools" "Options", please set "Mail transmission format" to "Text format".

Please also pay attention to the setting in the case of Web mail. (* Do not use HTML mail)

Also, please be sure to add the software name (WaveGene) to the subject of the mail. Otherwise, if it is a simple subject, it may be deleted with the spam filter.

*** Even after a few days with a question etc., if there is no reply, it may not have arrived, please check the setting again just in case.**

When reporting a problem, in addition to the operation procedure, please be sure to include as much detailed operating environment as possible including OS and sound device type.

If possible use WG.INI or the parameter file used Copy and paste (*. WG) contents to the body of the mail (please do not attach).

Since failure can not be reproduced it can not be handled, so please.

The latest version is

<http://efu.jp.net/> (changed on August 1, 1414)

It is in

Sister software

- High-speed real-time spectrum analyzer ----- > WaveSpectra
 - Wave file exclusive comparison software with auto cueing function ----> WaveCompare
 - High precision de-emphasis software for wave file ----- > DE.EXE
 - Software for DAT playback / recording by DDS drive ----- > WaveDAT (Shareware)
-

2013/01/05 efu (2014/08/01 URL / Change only email address)

Created with the Personal Edition of HelpNDoc: [Create cross-platform Qt Help files](#)

History

History

V1.50 (2013/01/05)

- Corresponds to [WASAPI \(Windows Audio Session API\)](#) . (Windows Vista SP1 or later)
Exclusive mode exclusive (Both event drive mode and timer drive mode)
- Corresponds to [multichannel up to 8ch](#) .
It corresponds to multi channel Wave file (WAVEFORMATEXTENSIBLE format).
- [Add Wave 4. Add 2 more pairs Wave 5 ~ 8.](#) (created up to 99 MHz)
- Add [pulse \(+ -\)](#) to the waveform. Stop the operation [OFF](#) addition.
- Add [band limitation \(addition synthesis, BLIT synthesis\)](#) to the periodic waveform of Wave 1, 5.
- [Change the sawtooth and triangle start position](#) according to the band limit waveform. (From negative maximum value)
(From the one sample of 0 for sawtooth wave only for addition synthesis)
- In Wave 1 and 5, you can now [rotate the mouse wheel to continuously change the frequency / cycle and amplitude \(volume\)](#) . To the operability like the analog oscillator.
(Change amount and change time can also be set)
- Added [pulse width modulation \(PWM\)](#) to the modulation of Wave 1,
Duty cycle can now be changed. (Other than sine wave, addition synthetic waveform))
- [Add Wave 5 to the modulation signal of Wave 1](#) . It became possible to select Wave 3 / Wave 5.
Therefore, the user waveform (Wave 5) can also be used as a modulation signal.
- Add [frequency setting bar](#) to Wave 1.
Smooth frequency change and sweep with the knob of the track bar became

possible. (Sweep speed can also be set)

[It is possible to change the frequency of Wave 5 synchronously](#) while keeping an arbitrary ratio to the frequency of Wave 1.

- Change registration of [user waveform](#) from right button to dedicated button.
- Change handling of registration window and user waveform of [user waveform](#).
Output channel and modulation signal of Wave 1 (Wave 5 only) can be selected separately.
- Add [output filter](#) function.
Filter setting window which can easily change various parameters.
Oversampling function.
Filter characteristics can be changed in real time even during sound generation operation.

Improvement

- Improve internal processing of playback.
- Improved [waveform display](#).
Change display sample number to same as internal buffer.
Improve accuracy of graduation. Improve display rate.
The waveform can be stopped and displayed when the display magnification is twice as wide as the horizontal magnification. (Trigger function)
- Change the display window when pressing the available format button in [Playback Device Setting Dialog](#).
The window is displayed first and the currently checked place is displayed in real time.
Also, from the display of S, M, etc., it changed to display with the number of the supported channel number.
Enlarge usable format to 384 kHz.
I separated the Info / Panel buttons respectively. (ASIO, WASAPI)
- We recorded the version number in WG.INI and displayed the initial message + help when the version changed.
- buffer change, FFT size change.
- ASIO IASIOThiscallResolver changed.
- The maximum 0 dB limit was eliminated by setting the amplitude value.
- Change preset value of combo box. It corresponded to theme.

[Bug fix]

- Fixed bug in waveform display when reading large user waveform.
- Fixed size limit processing when changing window size.
- Fixed line breaks of lines missing 10 data in text format file output.

V1.40 (Dec. 15, 2007)

- [Added user waveform generation function](#). (Modulation is AM, DSB only)
- M series noise (MLS) added.
- Add [text format output](#) (*. WGT).
- Added [up / down control](#) for changing frequency and amplitude.
- Optimized sample data length for FFT is supported up to 131072.
- When setting it to 1/2 Fs or more, the background of the frequency setting combo box is made to be " red ".
- Improvement of noise.
- Improvement of each response at playback. (Buffer reset, buffer configuration)

change, waveform display update speed)

- Improve the accuracy of the waveform when the frequency is high.
- Fixed a bug in the scale on the horizontal axis of the waveform display, x 17 when it is 176400 s / s or more.
- Fixed that sweep length could only be set in integer seconds. (3.5 seconds etc.)
- Fixed that the output frequency did not follow when changing the sampling frequency when the unit of frequency setting was periodic.
- Changed Help to HTML Help.
- Changed the archive file to * .ZIP.

V1.31 (February 22, 2004)

- Corresponds to drag & drop of parameter file (*. WG).
- By specifying the parameter file as the argument of the command line, it can be started with that setting (only setting change when activated), and automatic playback (with length specification) also enabled.
- Fixed a bug that could not be started due to an error at the next startup when setting driver to DirectSound and finishing.
- Fixed a bug that could not be used at all when Windows ME / 98/95 driver is MME. (I made WAVE_FORMAT_DIRECT possible to turn on / off with WG.INI)
- Fixed a bug that could not handle resizing of title bar.
- Fixed a bug that the free space became zero and writing was impossible when directly specifying the network drive at the time of file writing.

V1.30 (2003/10/04)

- ASIO, DirectSound, and WAVE_FORMAT_EXTENSIBLE (2ch only) format compatible.
(Significant remodeling of playback section)
- Supports IEEE_FLOAT (32 bit floating point) format playback and wave file creation.
- Conventional (at MME), WAVE_FORMAT_QUERY when displaying supported format list Changed to actually open once you were inquiring with.
(For drivers reporting lies?) By double clicking on
- compatible format list, we can change it to that format.
- Changed to show the waveform display from the first startup.
- Fixed a bug where the length of sweep was wrong length when changing sampling frequency.
(The length setting is an integer only and the same as before)
- Fixed a bug in waveform display and screen locking occurred.
- Depending on the environment, avoiding occurrence of zero divide error when selecting wave mapper is avoided.
- Other, minor modifications and changes.

V1.20 (2000/01/25)

- Supports 24 bit, 32 bit format.
- Added the playback device designation function and display format display function.
- Changed to be able to set the number of seconds as a decimal value.
- Corresponds to the number of samples for optimization for FFT up to 65536.
- Fixed a bug that caused ZeroDivide error or general protection error when blank or small value in combo box was set when setting arbitrary sampling frequency.

V1.10 (February 28, 1999)

- Addition of sweep function (frequency, amplitude, phase)

- Added modulation function (AM, DSB, FM, PM)
- Addition of gate function (tone burst)
- sawtooth wave addition.
- Add offset.
- Added waveform display.
- Added comment to parameter file.
- The level indication can be displayed up to 0.1 dB.
- Fixed that amplitude could only set integer part.
- Other, minor modifications and changes.

V1.00 (1999/02/11)
