EASY STEPS TO PROGRAM OTP VOICE CHIP APLUS

Sound File Format

Starting from a sound source file, there are three steps to prepare a voice chip samples. They are:

- Sound Editing
- Sound Compile
- Voice Chip Programming

Suppose the sound comes from CD. The sound source will have format, 44100Hz sampling rate, stereo and 16Bit per sample. We must convert the sound from CD into the following format:

- Sound file in WAV or VOC format
- 8 Bit per sample
- MONO channel
- Re-sample rate to maximum value to get highest voice quality. See below table to calculate sample rate.

Therefore we need the help of sound editing software to convert the sound source captured from CD into the required format.

The table below shows how to calculate the target sampling rate with different encoding to fit into different OTP voice chip.

OTP Chip	Memory Size ADPCM (4BIT) Encoding	Memory Size PCM (8BIT) Encoding	Calculation of Sampling Rate (Memory Size / Sec)
10 sec	64K	32K	If ex 5sec voice S.R. = 64K / 5sec ≒ 12.8K HZ (ADPCM) If ex 5sec voice S.R. = 32K / 5sec ≒ 6.4K HZ (PCM)
21 sec	128K	64K	If ex 8sec voice S.R. = 128K / 8sec ≒ 16K HZ (ADPCM) If ex 8sec voice S.R. = 64K / 8sec ≒ 8K HZ (PCM)
42 sec	256K	128K	If ex 12.8sec voice S.R. = 256K / 12.8sec ≒ 20K HZ (ADPCM) If ex 12.8sec voice S.R. = 128K / 12.8sec ≒ 10K HZ (PCM)
85 sec	512K	256K	If ex 23sec voice S.R. = 512K / 23sec ≒ 22K HZ (ADPCM) If ex 23sec voice S.R. = 256K / 23sec ≒ 11K HZ (PCM)
170 sec	1024K	512K	If ex 42sec voice S.R. = 1024K / 42sec ≒ 24K HZ (ADPCM) If ex 42sec voice S.R. = 512K / 42sec ≒ 12K HZ (PCM)
341 sec	2048K	1024K	If ex 64sec voice S.R. = 1024K / 64sec ≒ 32K HZ (ADPCM) If ex 64sec voice S.R. = 2048K / 64sec ≒ 16K HZ (PCM)

Sound Editing

There are many different kinds of sound editing software available. We recommend GOLDWAVE, COOLEDIT AND SOUND FORGE.

The following example is based on the GOLDWAVE software. Use the software to open a sound file, then followed by CUT, Paste and other commands to editing the sound. Resample the sound using the sampling rate calculated from the table above. Depends on different voice and speaker, try to find a suitable sampling rate to get best quality. When finished editing, use the SAVE AS command to convert the sound into WAV, MONO 8 BIT format.

Normal, noise will be created when a sound file is sampling down into a smaller sampling rate. We usually use the following technique to reduce the noise or distortion:

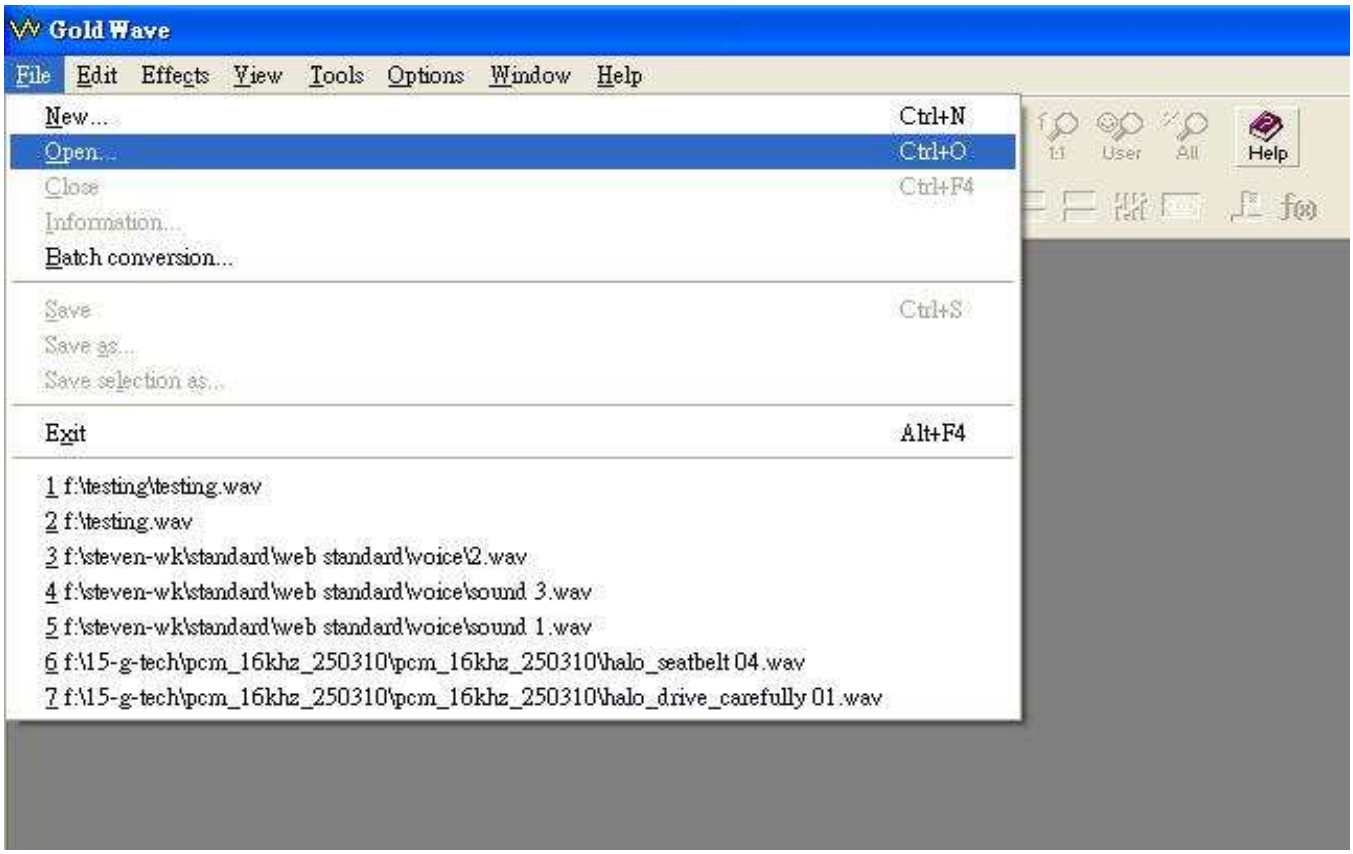
- EQ, Noise Reduction, Fade in, Fade out, Volume, etc

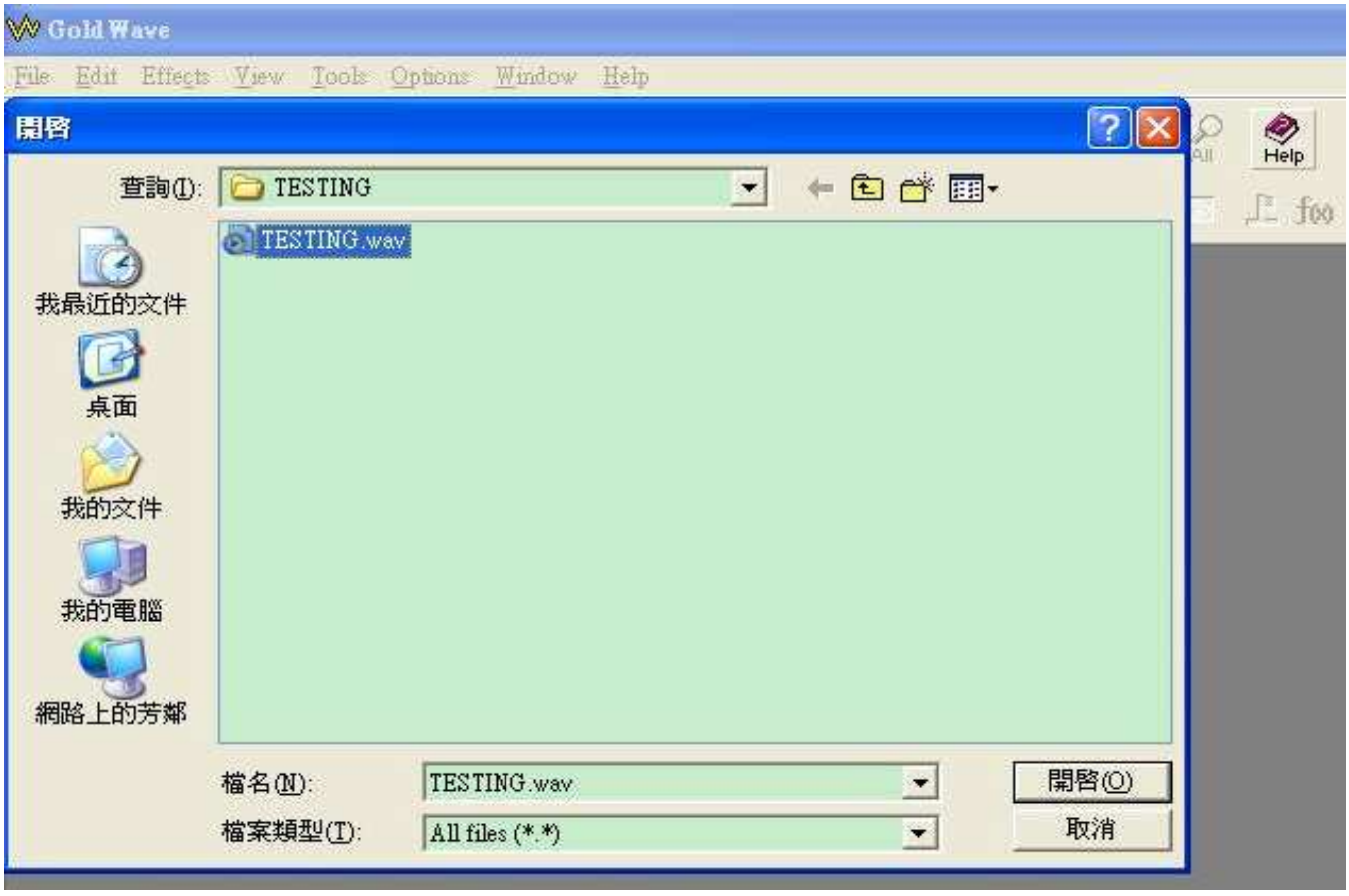
The value setting for the above commands are not fixed. User should choose the suitable value for his/her particular sound file.

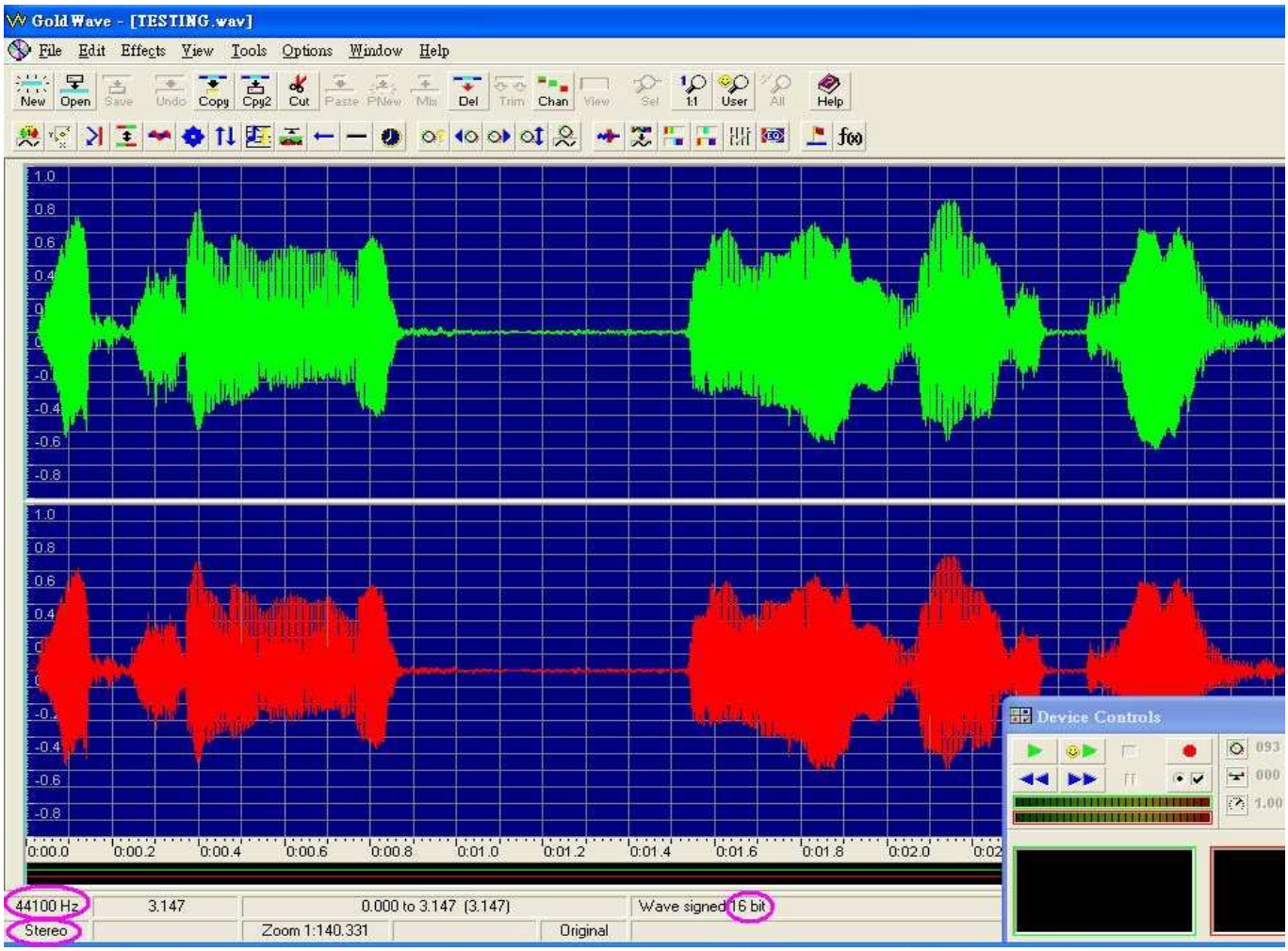
EQ setting example:

60 HZ or below	Depend on situation, sometimes noise will be in this frequency rang, therefore, we need to decrease the gain of this frequency range
150 HZ	Dog bark, Drum
400 HZ	Male speech
1000 HZ	High pitch male, female speech
2400 HZ	High pitch female
6000 HZ	Children
15K HZ or above	Usually, we will reduce the gain of this frequency range into minimum because our voice chip cannot response to this.

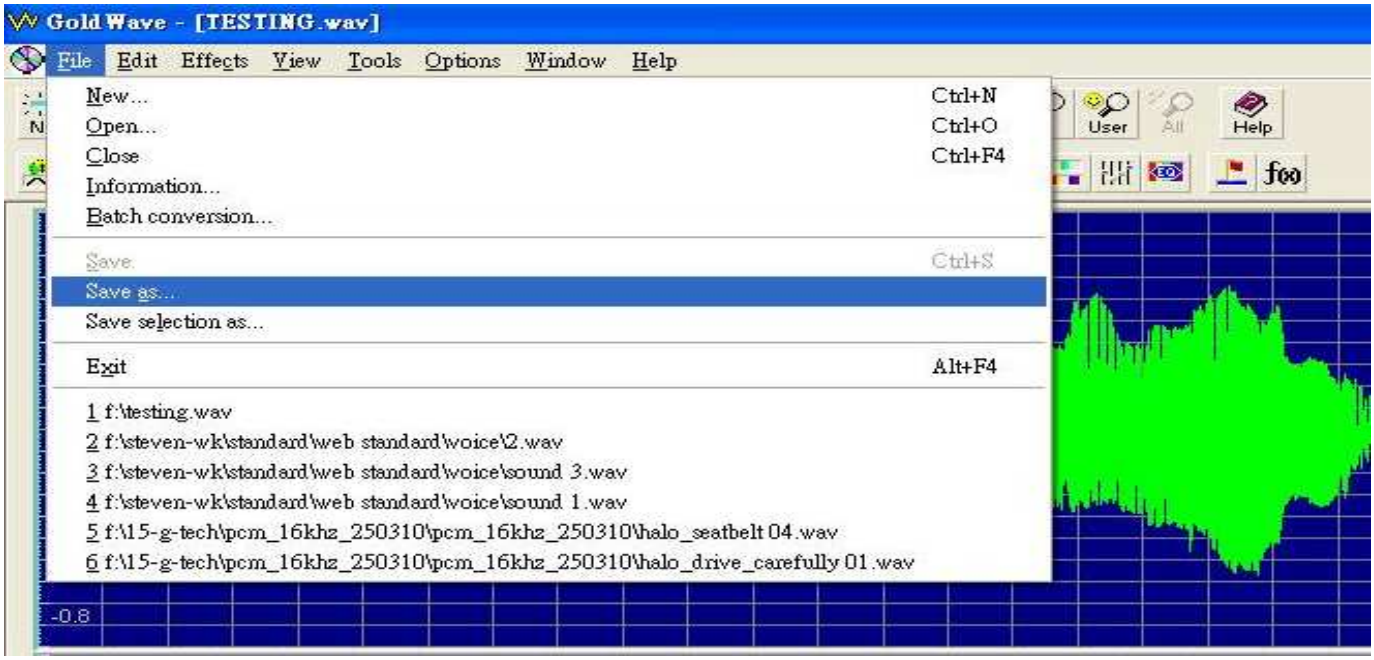
1. Let's use GOLD WAVE as an example to demonstrate how to edit the sound sources. Open the original sound file as below:



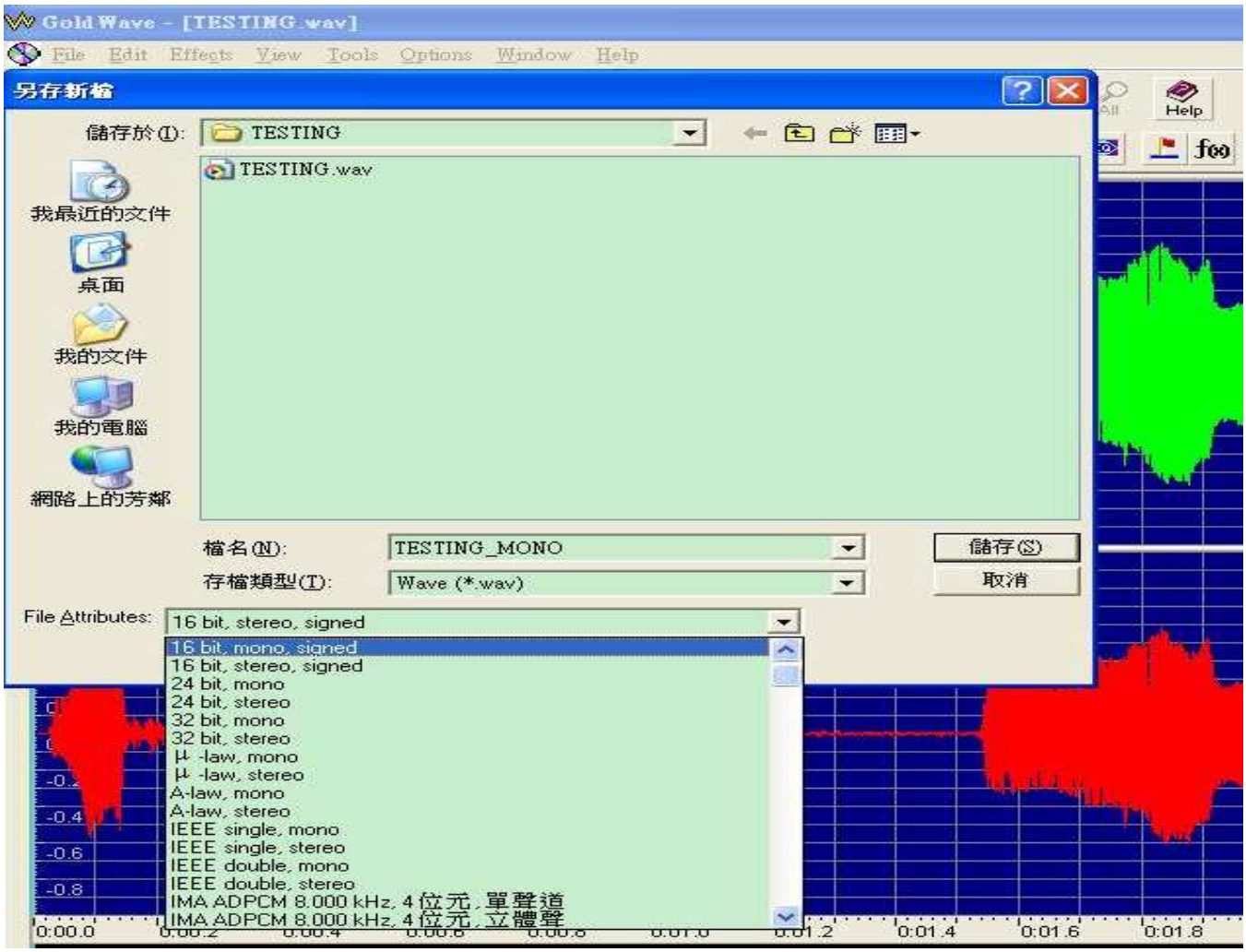




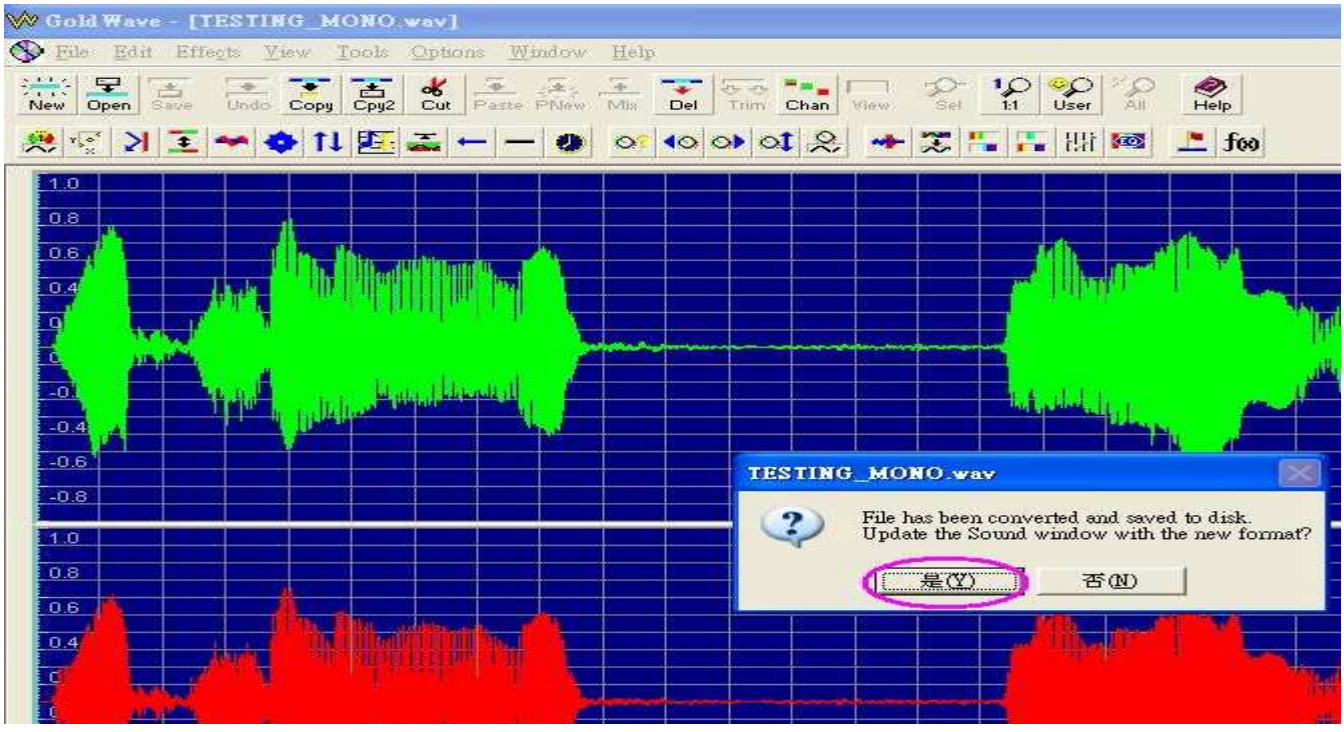
2. Convert the sound file to the format that we can accept (WAV and MONO)

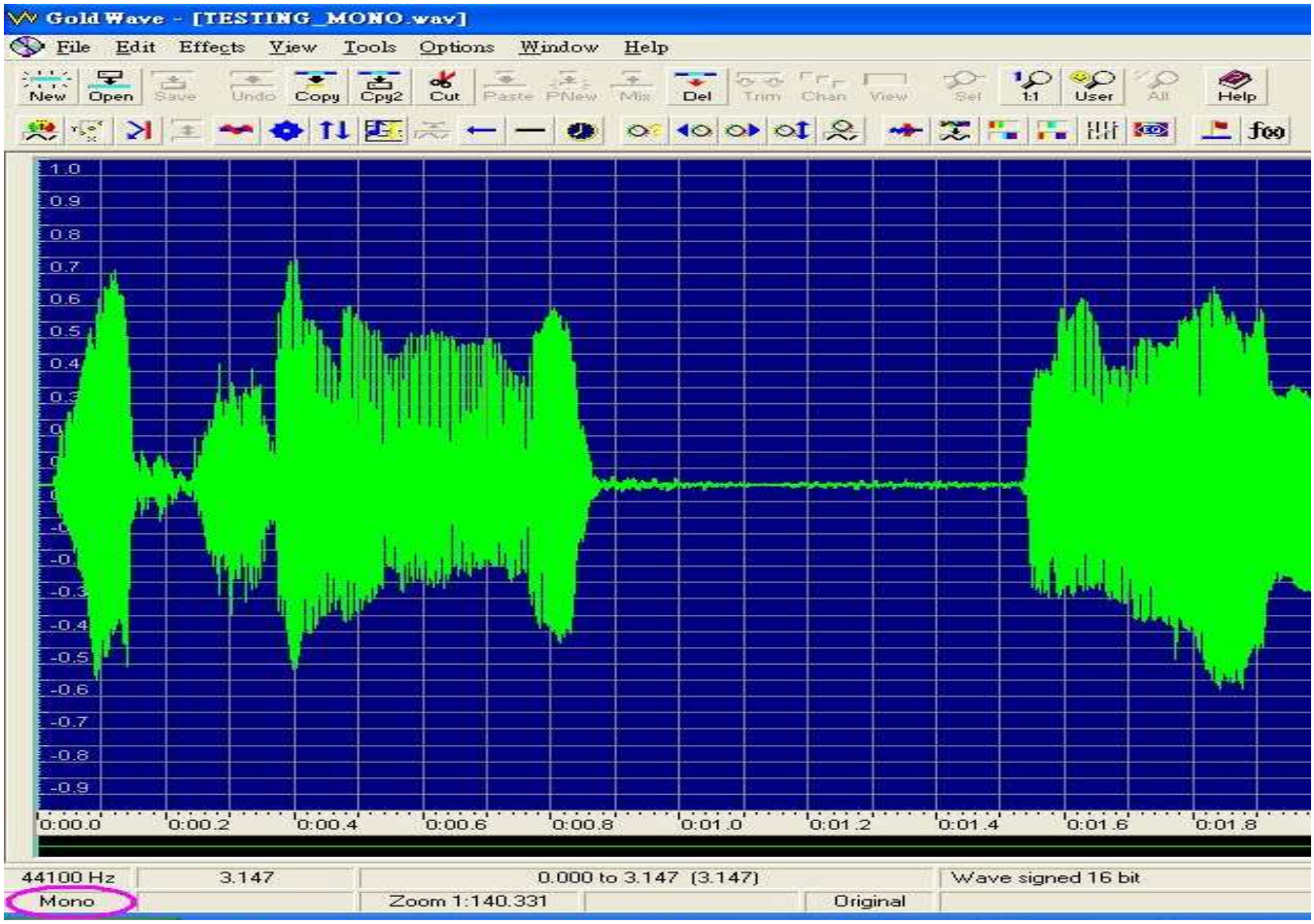


3. Save as another file name after choosing the correct format.



4. Click “OK” to confirm to accept the change of file format.





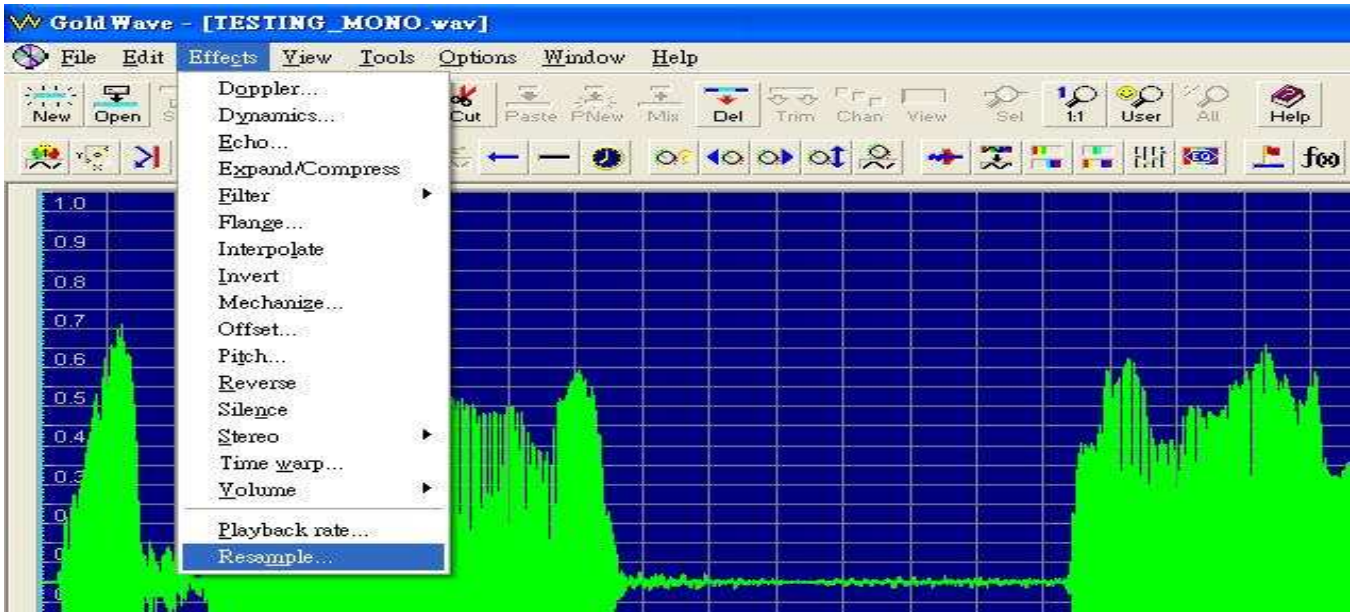
5. The sample rate is dependent on which IC body is used. We have the following IC bodies:

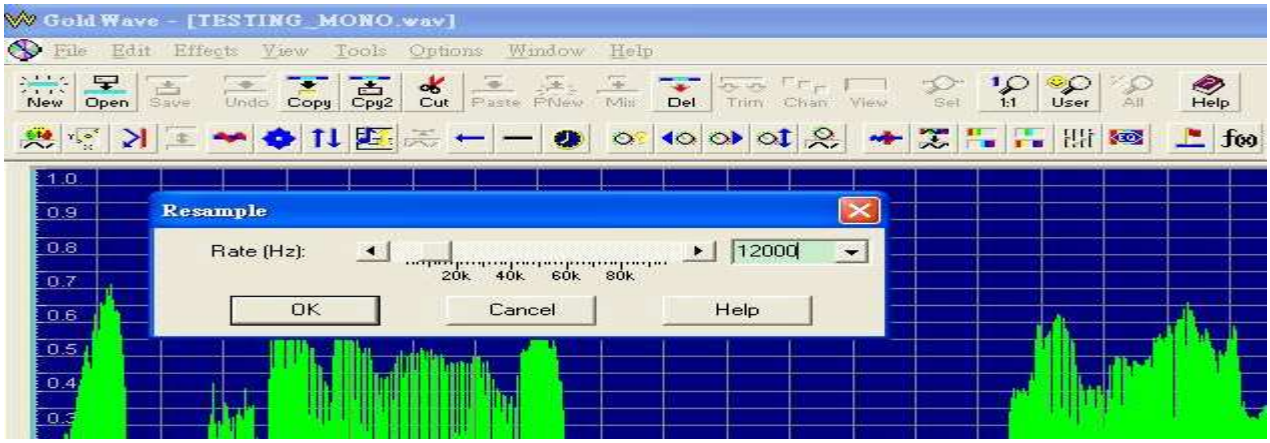
OTP CHIP	10 SEC	21 SEC	42 SEC	85 SEC	170 SEC	341 SEC
Memory Size ADPCM encoding	64K	128K	256K	2512K	1024K	2048K

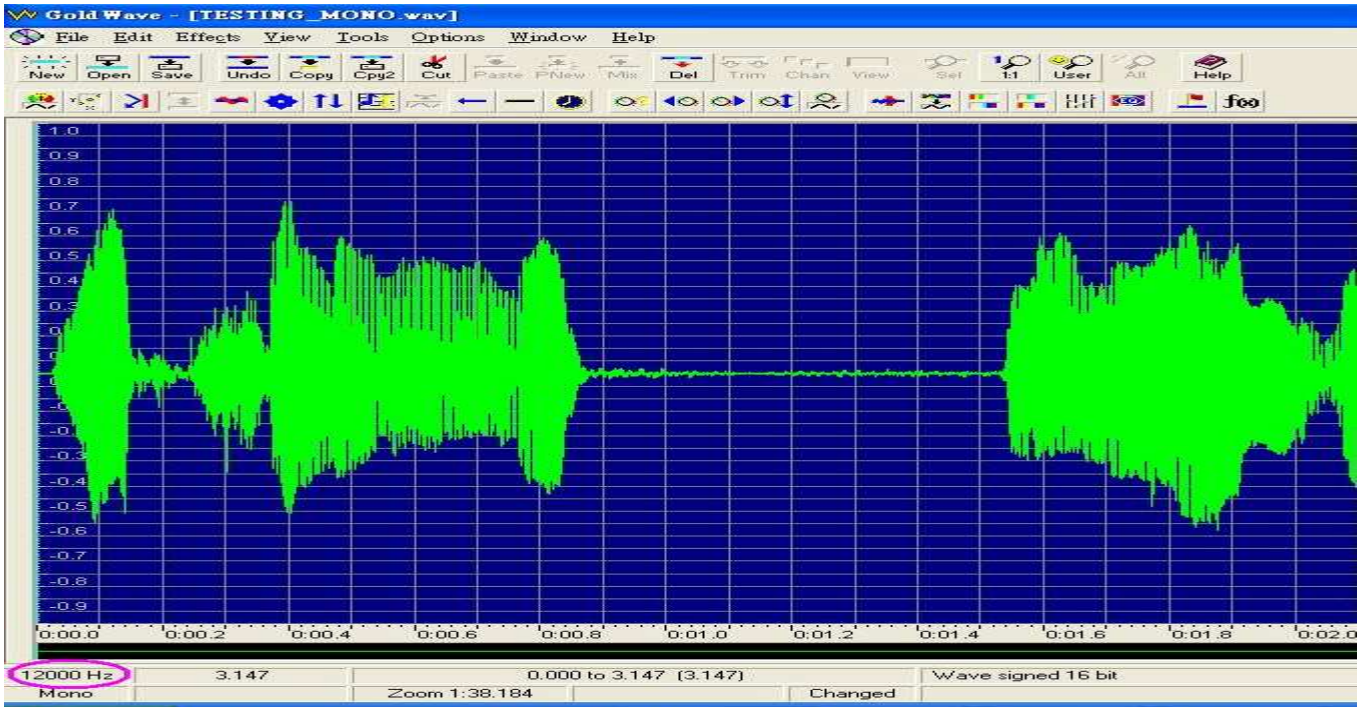
To determine the SAMPLE RATE, we use the following formula:
SAMPLE RATE = Memory Size / number of SEC

e.g. If we are going to use the 10SEC IC, the memory size is 64K, therefore the sample rate is:
64K / 10 = 6.4KHz
The maximum sampling rate should be a little smaller than the above value. Therefore, 6.0KHz rather than 6.4KHz should be used.

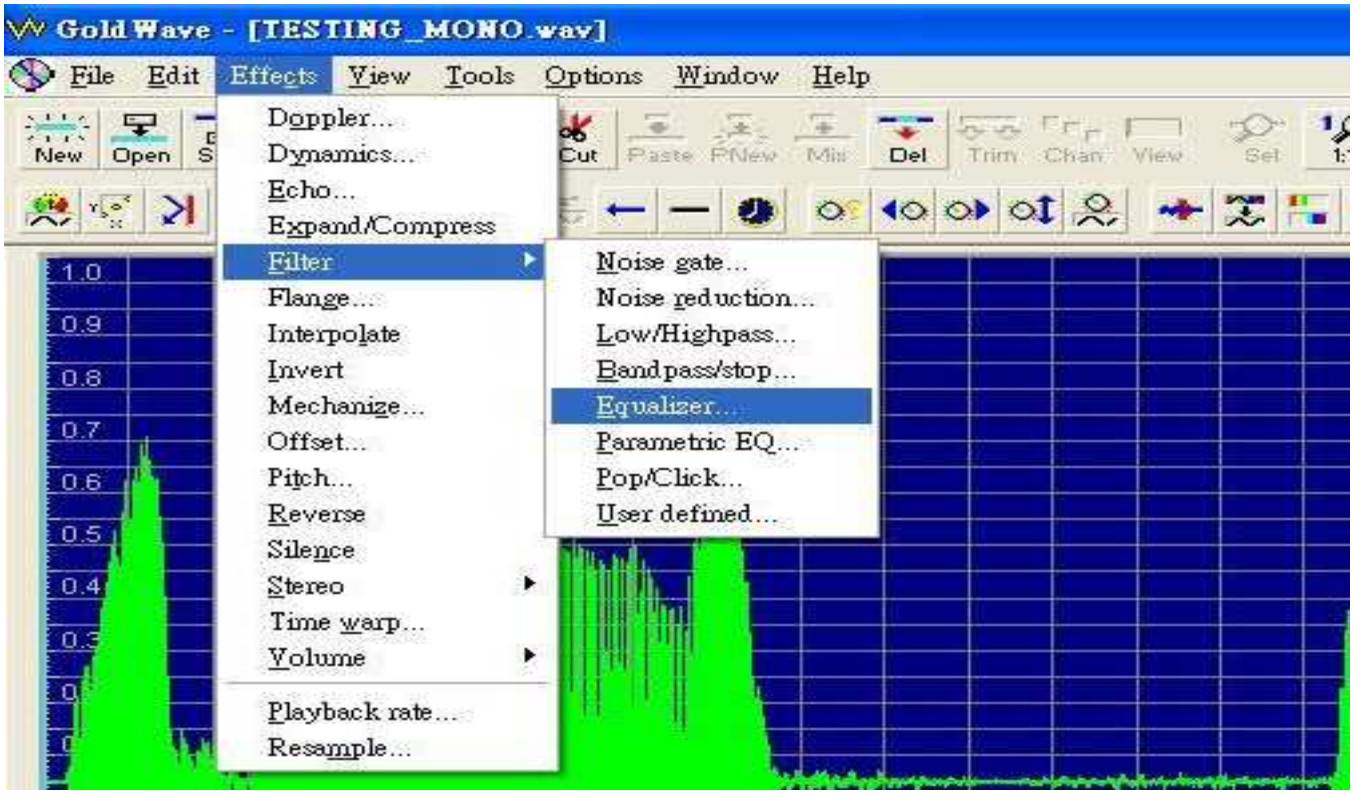
After obtaining the sample rate, we should resample the original sound. If GOLDWAVE is used, and assume the new sample rate is 12KHz, then, we can resample the sound file using 'Effect' and 'Resample' to modify the sample rate as shown in the figures bellows:

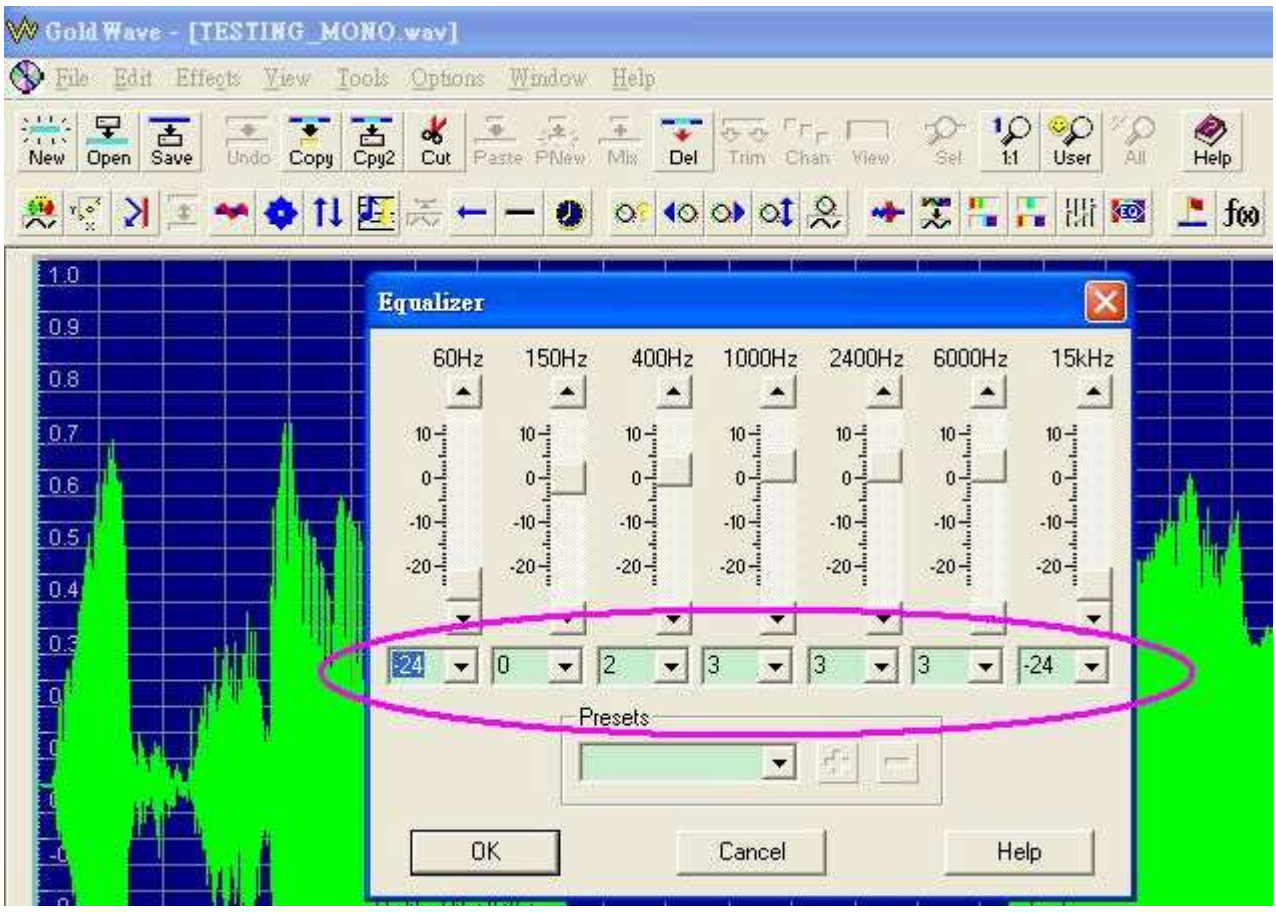

GoldWave - [TESTING_MONO.wav]
File Edit Effects View Tools Options Window Help
Doppler...
Dynamics...
Echo...
Expand/Compress
Filter
Flange...
Interpolate
Invert
Mechanize...
Offset...
Pitch...
Reverse
Silence
Stereo
Time warp...
Volume
Playback rate...
Resample...

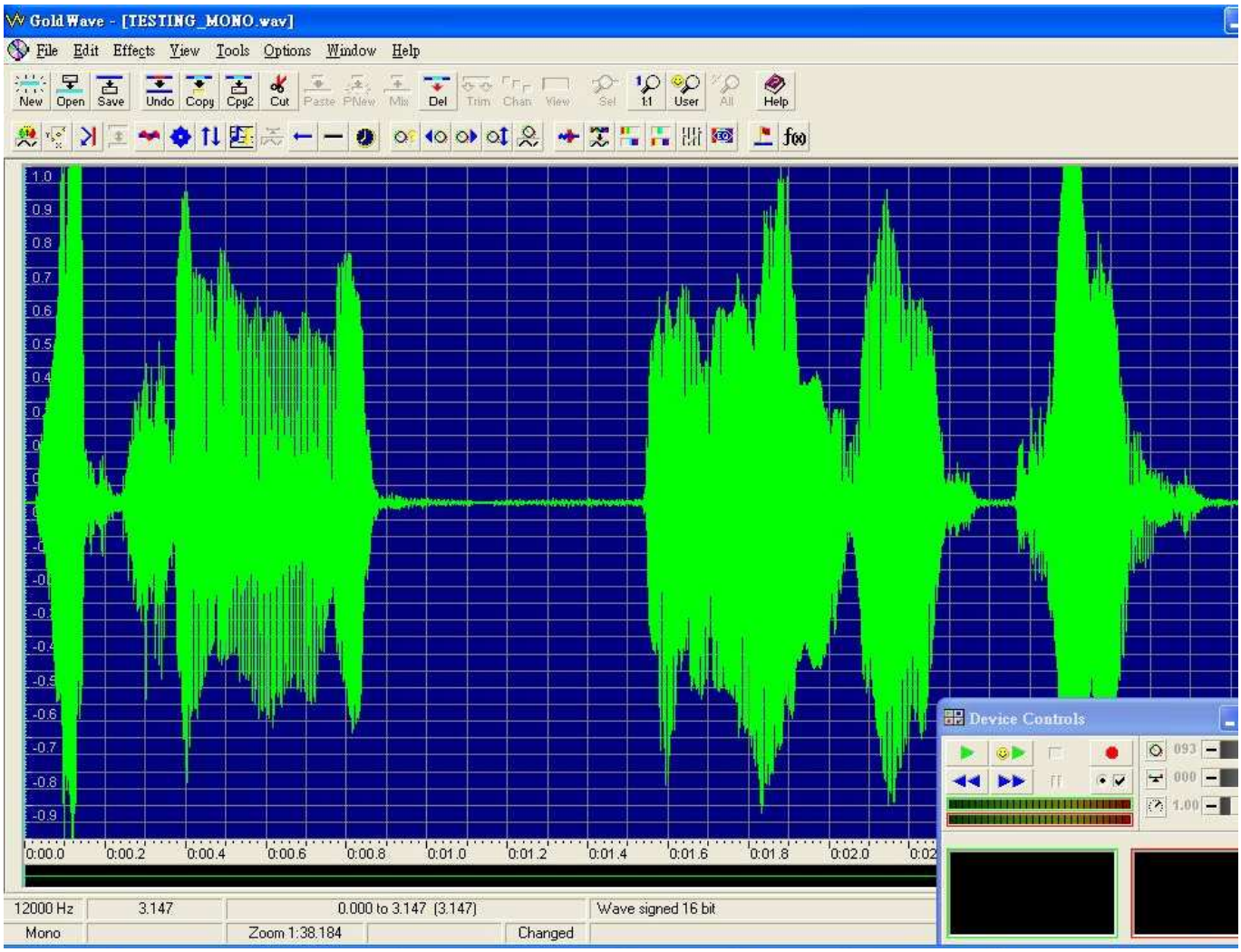

GoldWave - [TESTING_MONO.wav]
Resample
Rate (Hz):
12000
20k 40k 60k 80k
OK
Cancel
Help


GoldWave - [TESTING_MONO.wav]
12000 Hz
3.147
0.000 to 3.147 (3.147)
Wave signed 16 bit
Mono
Zoom 1:38.184
Changed

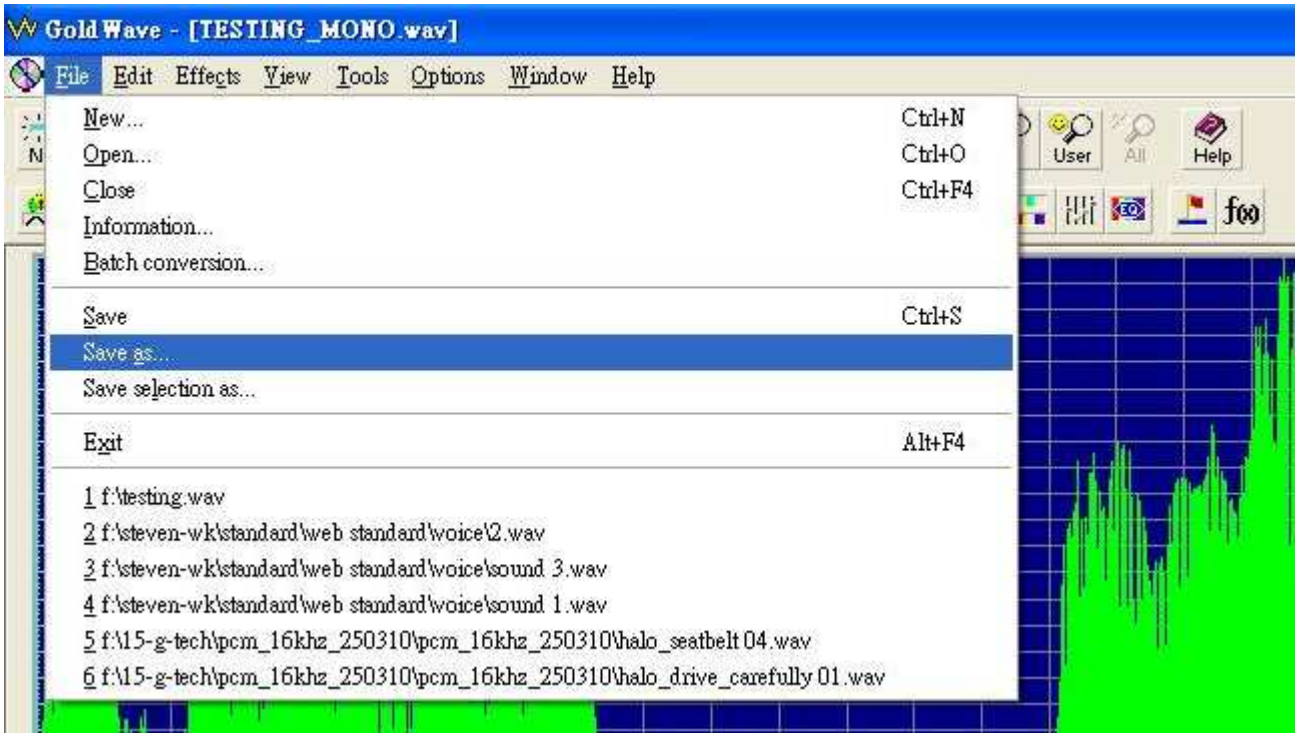
6. Normally, we will have high frequency noise introduced when we resample the original sound file. This high frequency noise can be reduced by using the Equalizer function as shown below:

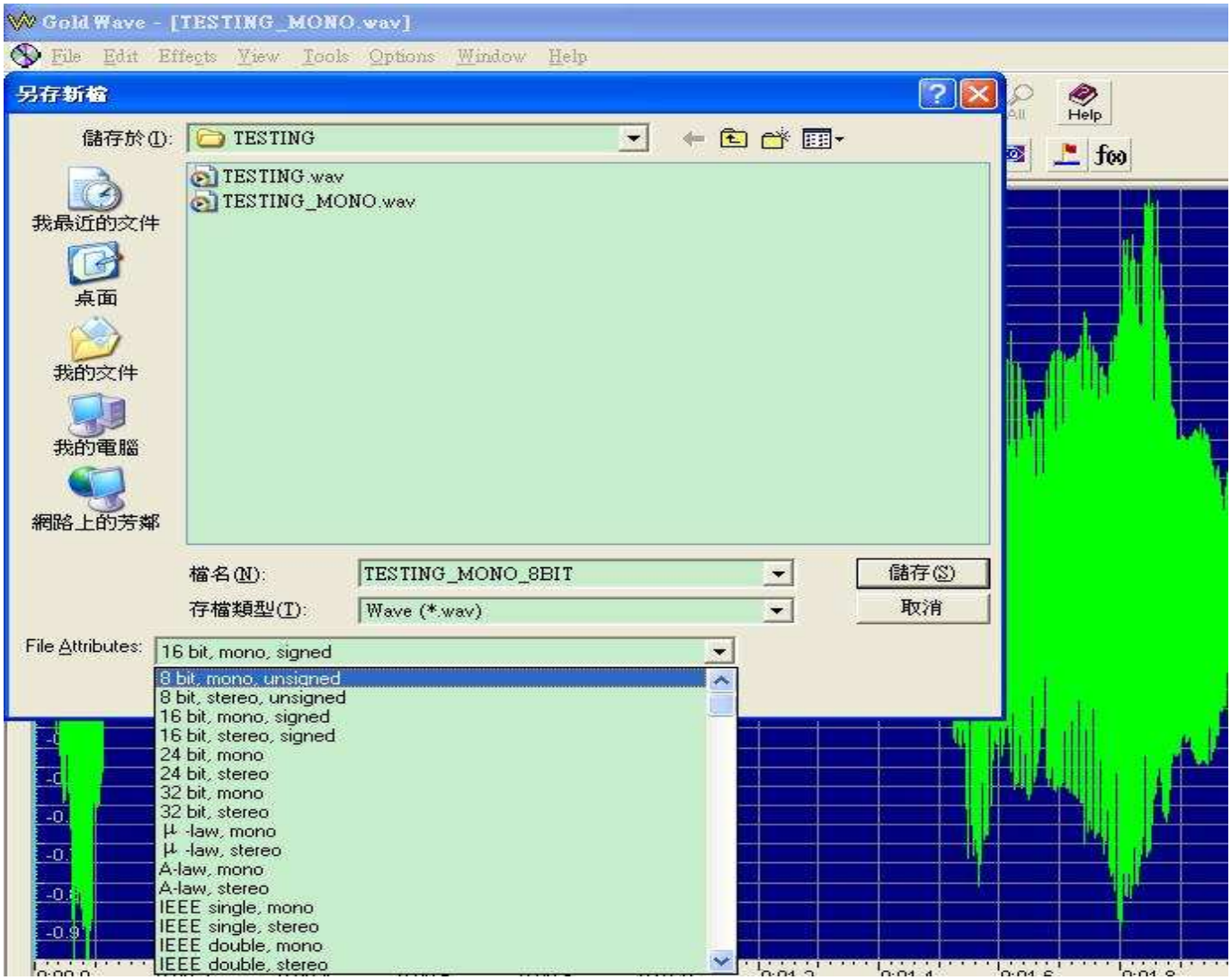




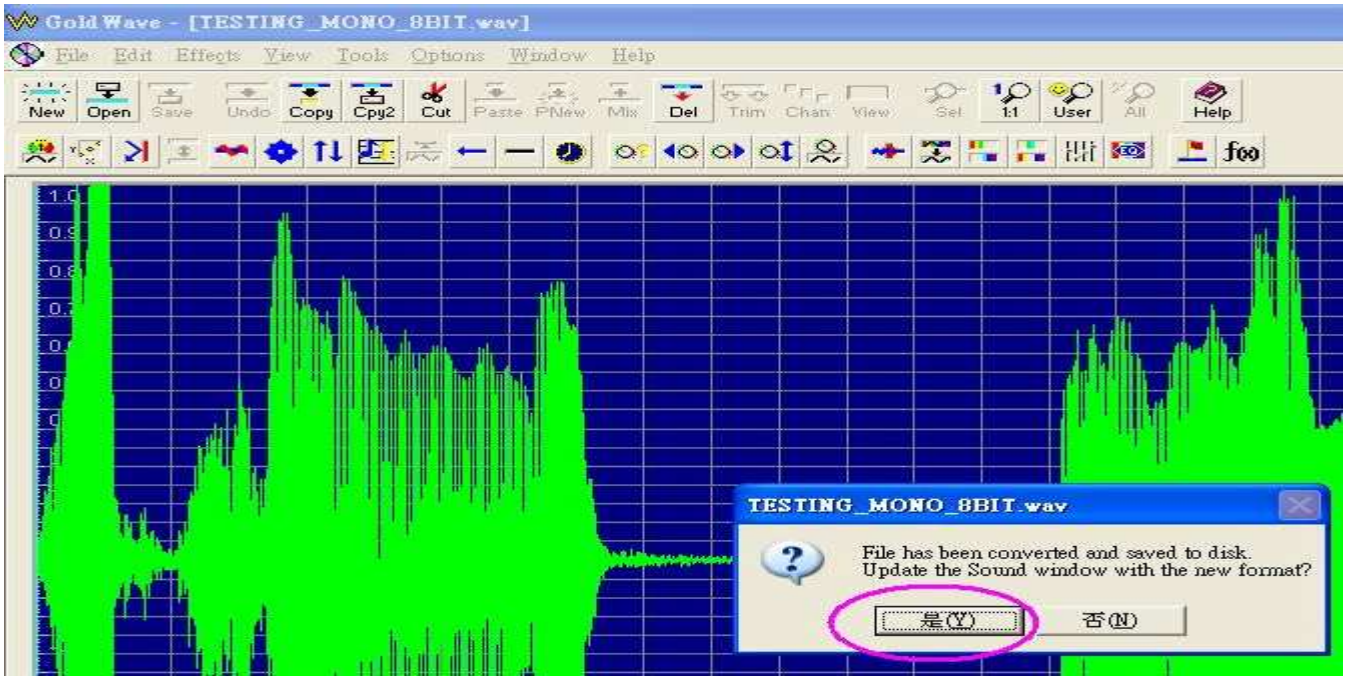


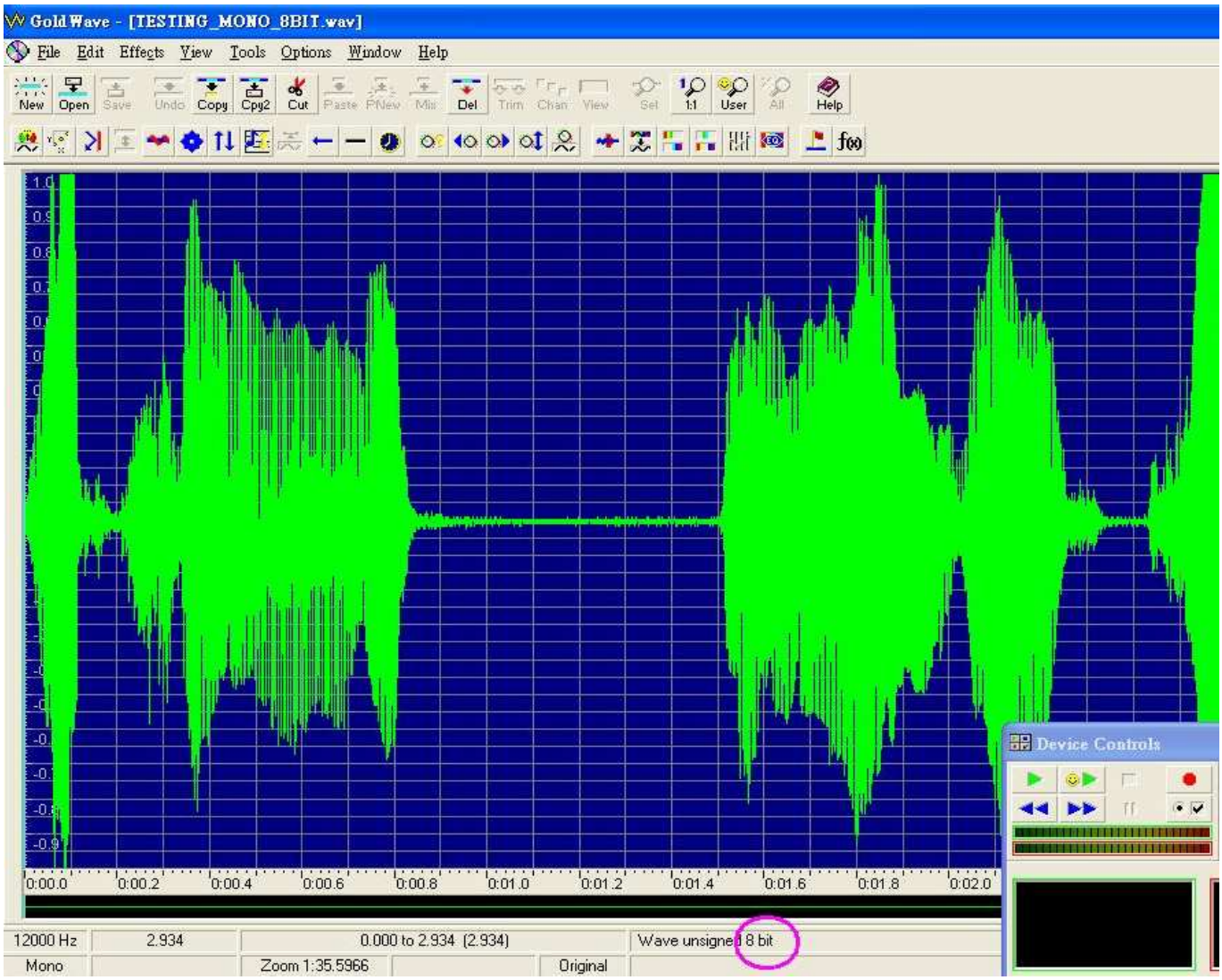
7. Convert the sound file to the format that we can accept (8-bit)



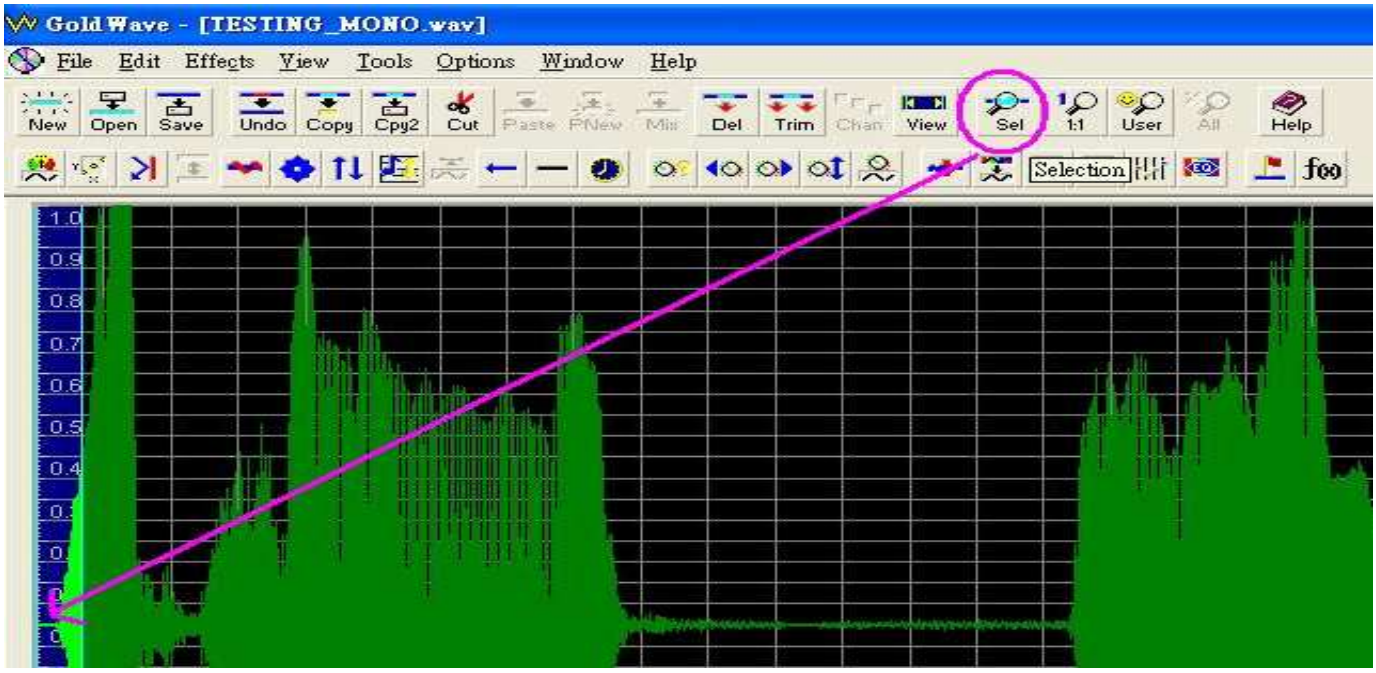


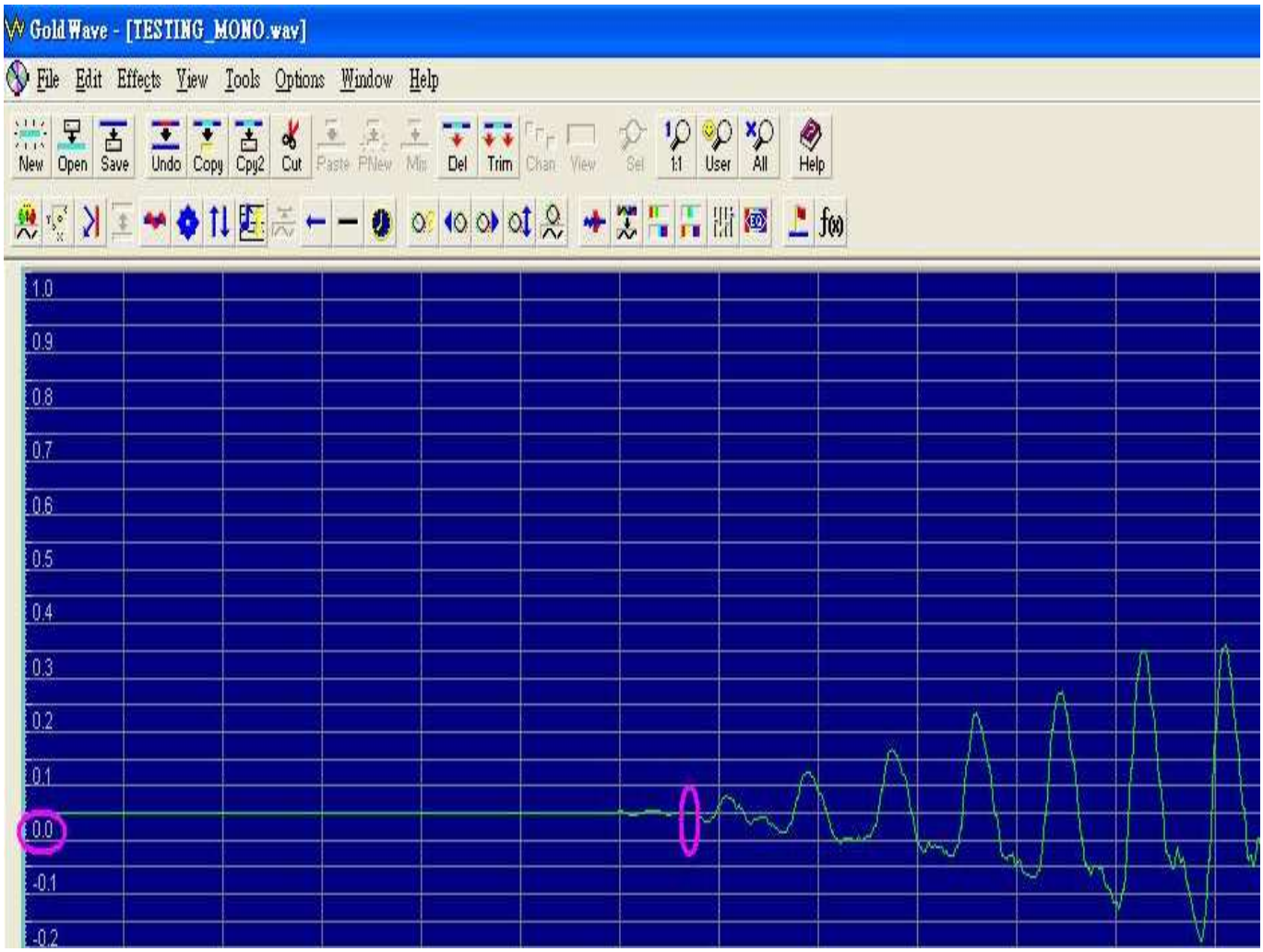
8. Click “OK” to confirm to accept the change of file format.

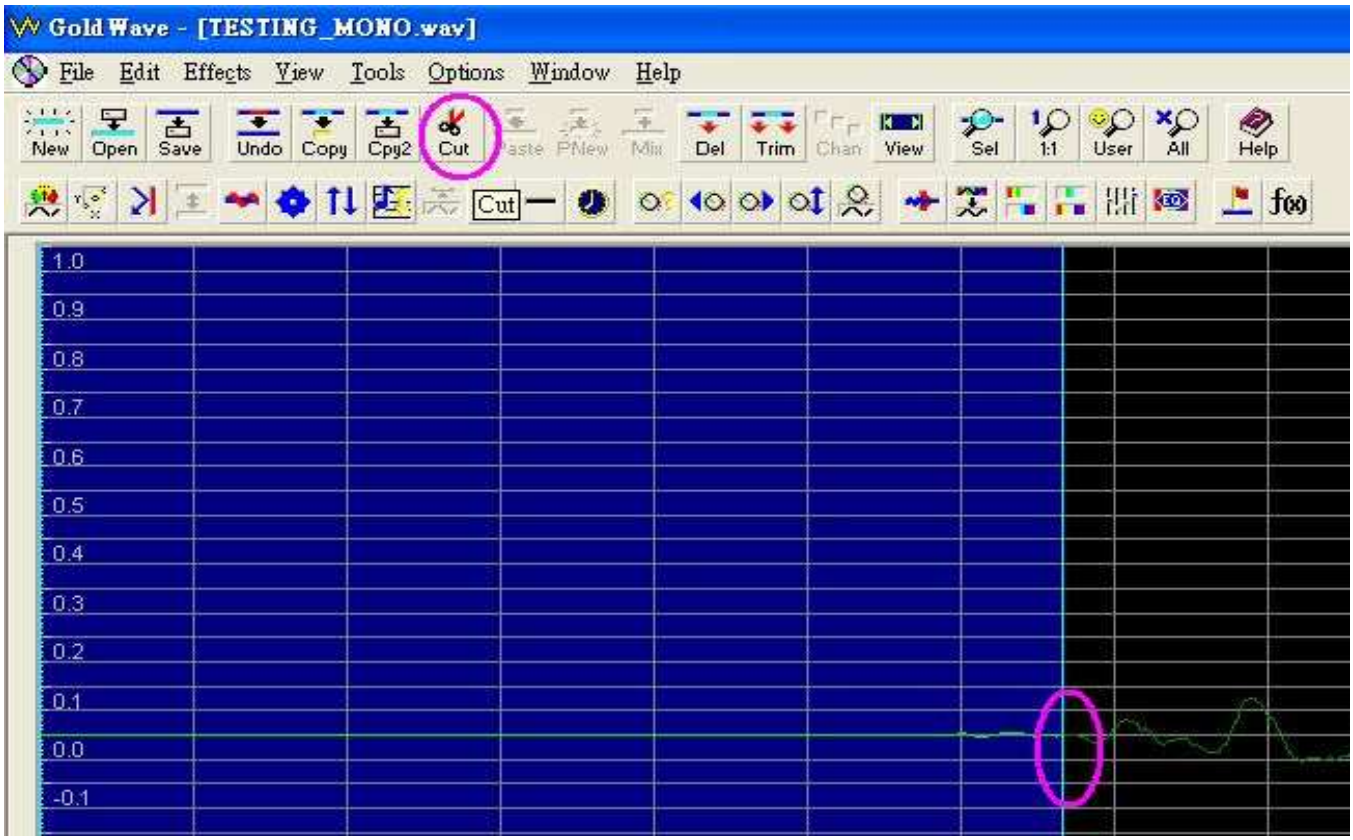


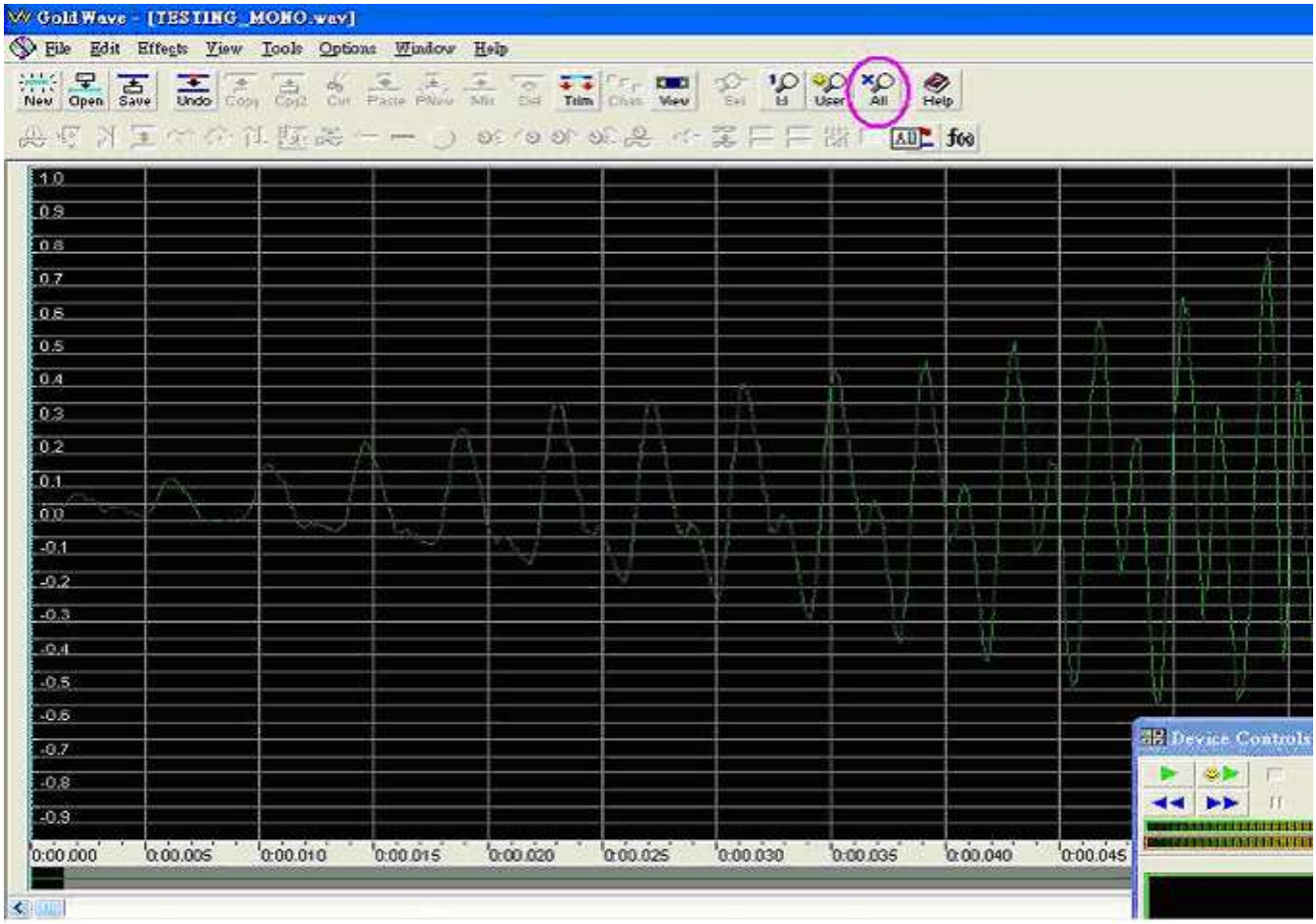


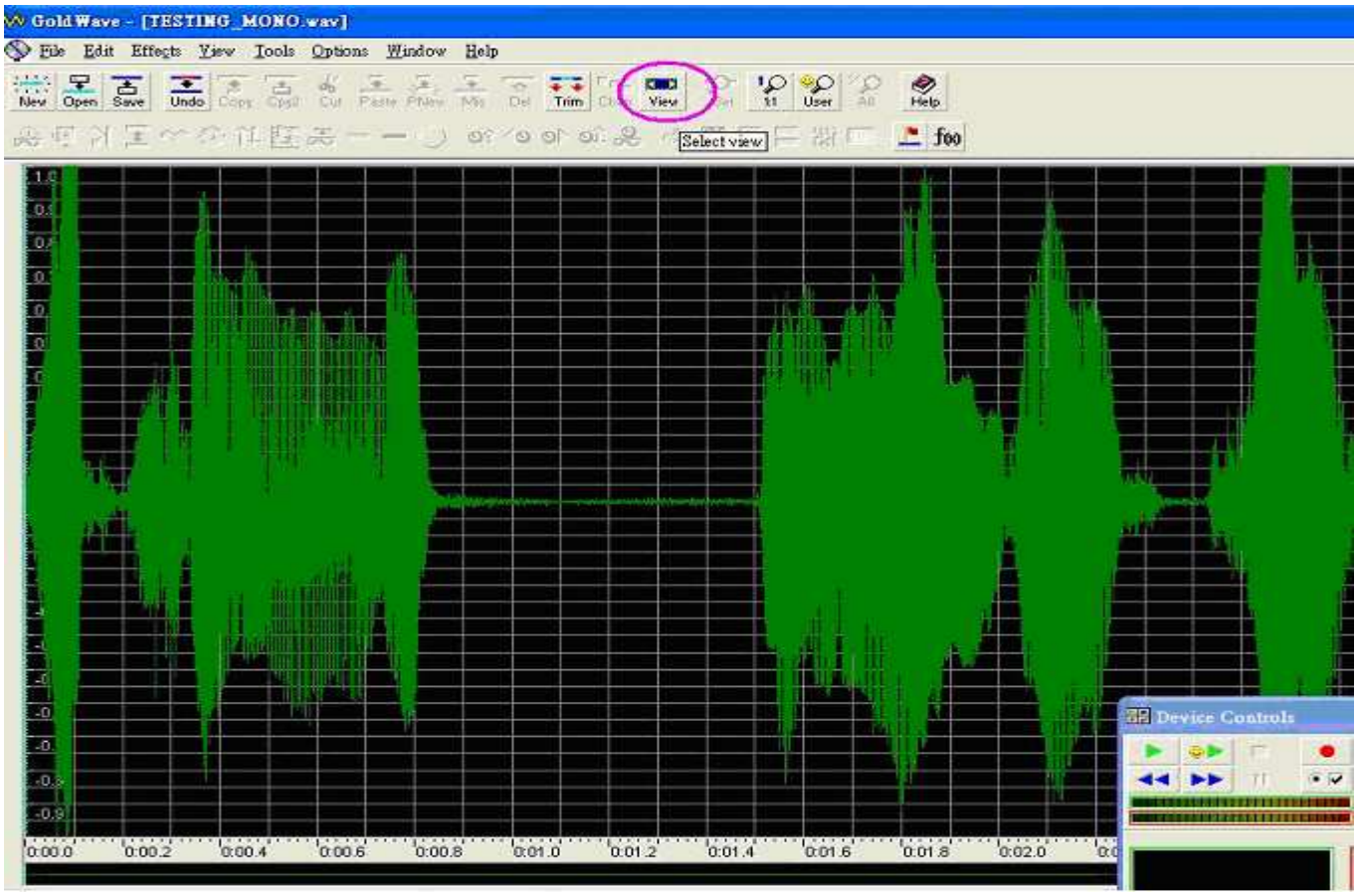
9. Remove the silence portion at the begin and end of the sound can reduce the chance of "POP" sound to happen.

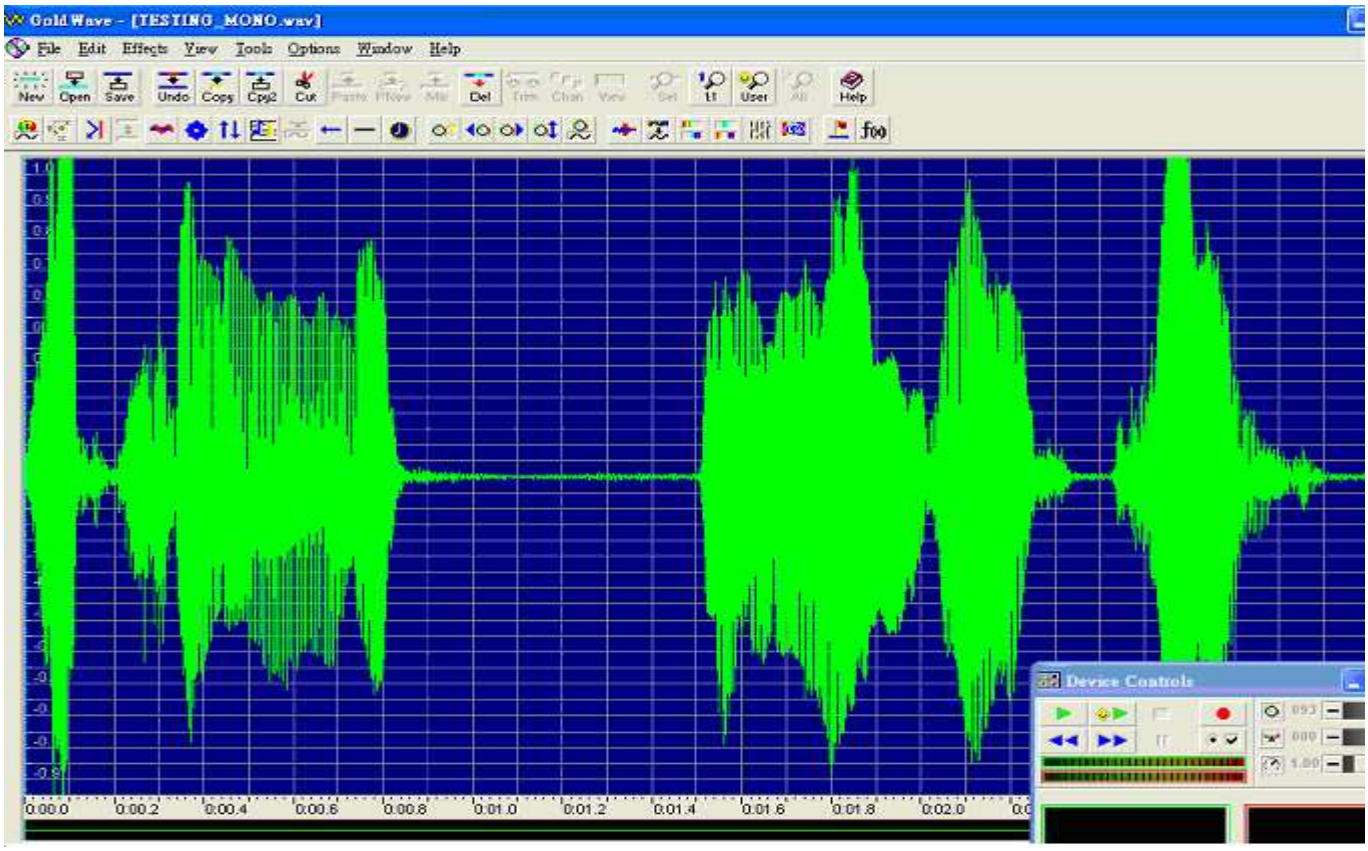



Gold Wave - [TESTING_MONO.wav]
File Edit Effects View Tools Options Window Help
New Open Save Undo Copy Cpy2 Cut Paste PNew Mix Del Trim Chan View Sel 1:1 User All Help
1.0
0.9
0.8
0.7
0.6
0.5
0.4
0.3
0.2
0.1
0.0
-0.1
-0.2

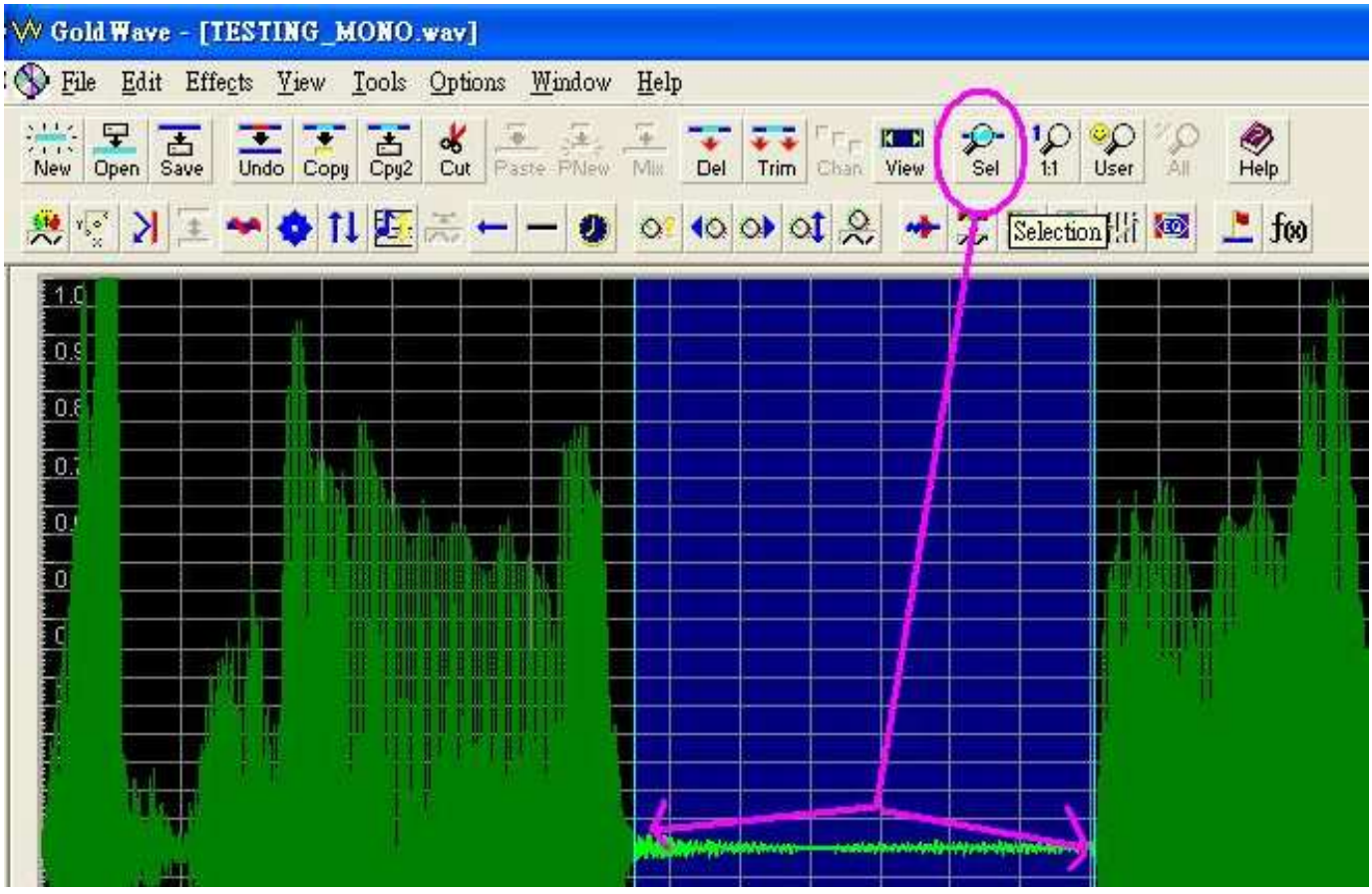

Gold Wave - [TESTING_MONO.wav]
File Edit Effects View Tools Options Window Help
New Open Save Undo Copy Cpy2 Cut Paste PNew Mix Del Trim Chan View Sel 1:1 User All Help
Cut
1.0
0.9
0.8
0.7
0.6
0.5
0.4
0.3
0.2
0.1
0.0
-0.1


GoldWave - [TESTING_MONO.wav]
File Edit Effects View Tools Options Window Help
New Open Save Undo Trim View All Help
Device Controls

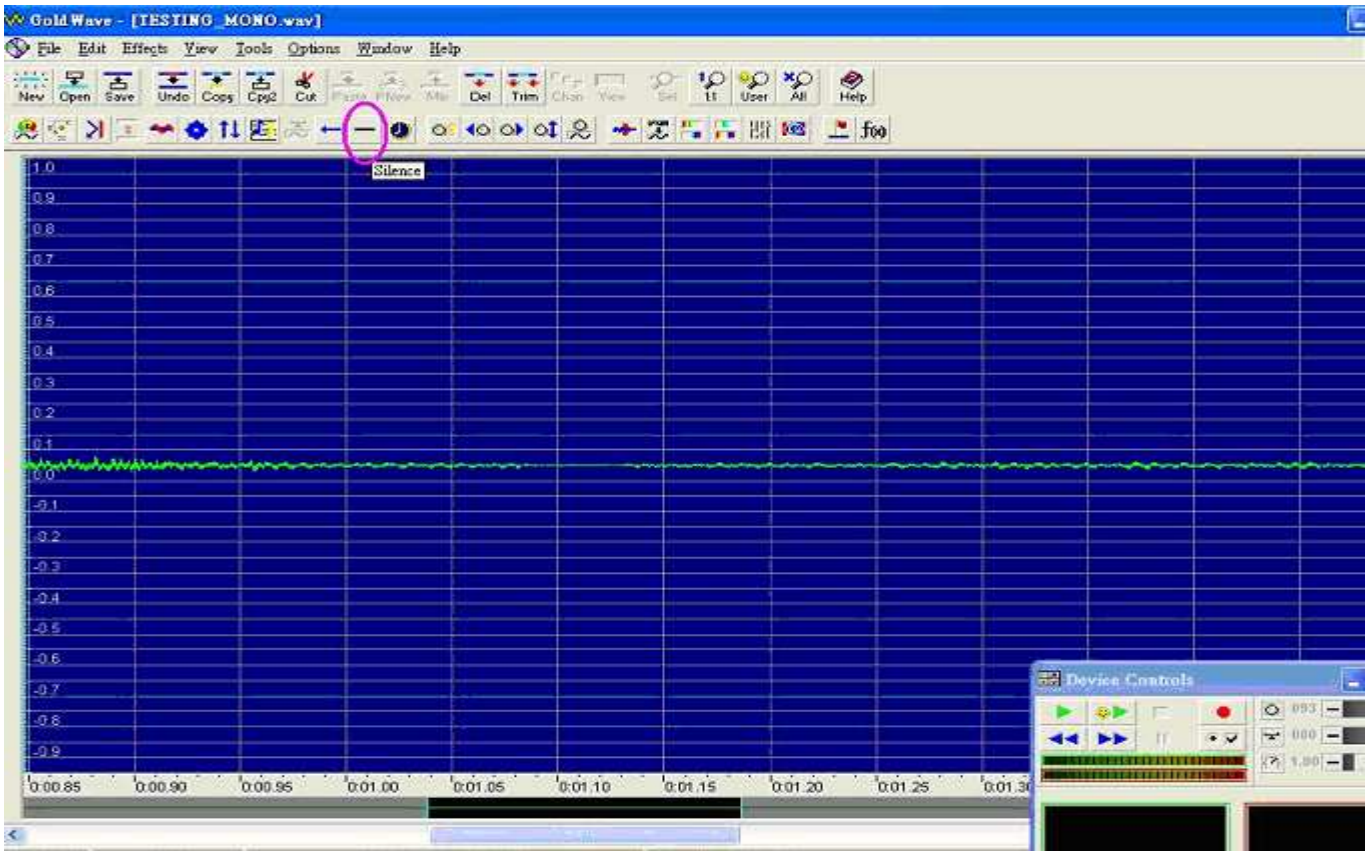

GoldWave - [TESTING_MONO.wav]
File Edit Effects View Tools Options Window Help
New Open Save Undo Trim View User Help
Select view
Device Controls

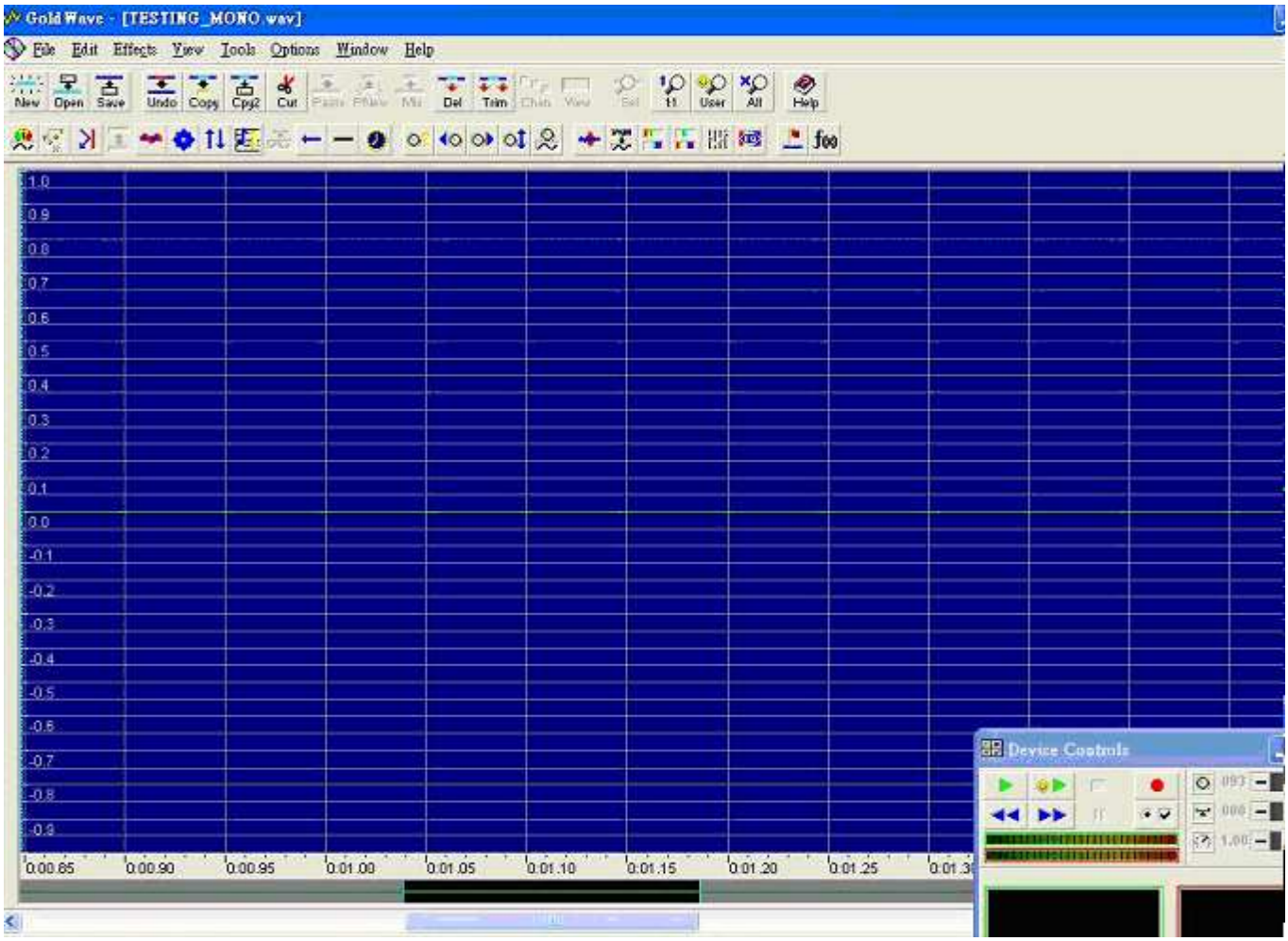


10. Finally, If there are very low volume part in the middle of the sound, it should be converted to total silence otherwise, the low volume sound maybe become noise sound in the final IC.



EASY STEPS TO PROGRAM OTP VOICE CHIP





Compile of Sound File

When the sound file is prepared, the **aP89W24USB** software is used to compile and convert the sound file into .DPM format data file to be programmed into the voice chip.

Procedures to Compile:
1. Select the suitable IC chip to be used.
2. Select the encoding method to be used, ADPCM or PCM.
3. Add the edited WAV or VOC files.
4. Select the speaker output to be used, VOUT <PWM> or COUT <DAC>.
5. Select the trigger mode to be used, CPU or KEY mode.
6. Select the number of voice groups.
7. In each voice group, select the trigger mode, LEVEL or EDGE, HOLD or UNHOLD, Retrigger or Non-retrigger.
8. Select OUTPUT OPTIONS. (busy/stop/led)
9. Double click the already added voice files into each voice group.
10. Click the COMPILE button to compile and save the above selected options and voice file into .DPM file.

Function：

Edge and Level Trigger:

For both Edge and Level trigger, the chip starts to play when the trigger button is pressed. For Level Trigger, if the trigger key is pressed and keep holding at the pressed position, the chip will continue to play the same sound again and again until the trigger key is released. However, if the chip is programmed to Edge Trigger, the chip will play the sound only for one time even the trigger key is kept pressing.

Hold and Unhold Trigger:

For Holdable Trigger, the chip will play only when he trigger key is kept pressing. Sound playback will stop immediatly once the key press is released. For Un-holdable Trigger, the sound section will be played to finished once the trigger is pressed no matter the trigger key is kept pressing or released.

Non-retrigger and Retrigger:

For retrigger option, the currently playing sound will be stoped and new sound section will be played when another trigger key is pressed. For non-retrigger option, new trigger action will not be granted until the current sound section is finished playing.

PWM or COUT:

PWM output is used to drive a small speaker (< 1/2 W speaker) directly. That means user can connect a small speaker to the VOUT1 and VOUT2 directly without adding extra circuit component. However, the sound volume will be limited. On the other hand, COUT is a current mode analog output in which at least a NPN transistor is needed as a amplifier. More high power amplifier can also be used with this COUT output to obtain larger power output.

OTP Chip Programming

Go the Writer page, press the LOAD button to load the .DPM file into the software. Note the CHECK SUM value. It should be the same as the value after Compile. If it is different, that means the .DPM file is corrupted. Voice files must be compiled again. Insert the suitable chip into the PC programmer socket and press the WRITE OTP button to start the programming. Upon finished, a 100% match should be display. That means the voice chip is programmed successfully.