Denis Sbragion's **DRC** **(Step by Step Guide v1.0)**

> *"**DRC** is a program used to generate correction filters for acoustic compensation of HiFi audio systems, including listening rooms compensation. **DRC** generates FIR filters which can be used with a real time or offline convolver to provide real time correction."*

The Software You'll need:

1. Cool Edit Pro 2.1.
2. Aurora Plug-ins.
3. Winamp's Realreverb plug-in.
4. Foobar2000's Convolution Plug-in.
5. Sbragion's DRC.
6. Patience and Obedience ☺

Notes regarding software:

* Please download Aurora version 3.2 (Aurora32Beta8.zip). I know that there are newer versions available, but I couldn't overcome several bugs in them.

* After you install Cool Edit Pro 2.0/2.1, drop all the Aurora plugins into its main directory.

* Pay attention that every time you use one of Aurora's features inside Cool Edit, you have a ¼ chance that Cool Edit will crash. Aurora implemented this annoying "Russian roulette", in order to remind you that you're using an unpaid version. If indeed Cool Edit crashed on you, next time you'll enter Cool Edit, it should ask you if you want to continue your last session, and if you'll agree, most chances are that you're going to get to the state just before the crash (so you'll lose nothing).

* When the Demo version of Cool Edit Pro is opened at the first time, it will contain a lot of Show Off material, which will make things hard for us. Close Cool Edit Pro. When you'll open it the next time, the Show Off material will disappear.

*You should drop Foobar2000's convolution plug-in (foo_convolve.zip), inside Foobar2000's main directory.

The hardware you'll need:

1) Omnidirectional Microphone.
2) Microphone Preamp.
3) Decent quality sound card.
4) SPL meter. (Radio Shack's is fine).
5) Multimeter (with a low ACV range, ~ 10V will due).

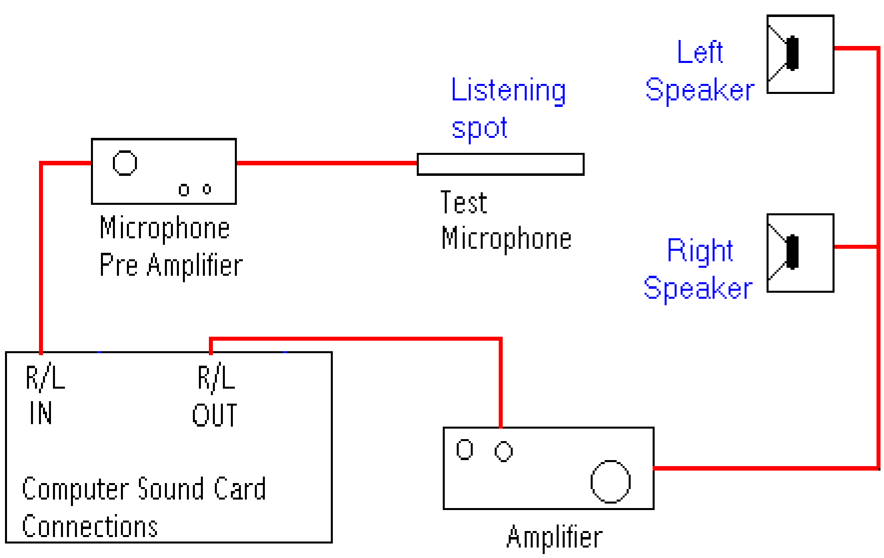
Notes regarding hardware:

* As for the Microphone, I use the Behringer ECM8000. It is pretty accurate for the job, and it cost pretty low (about $40 USD)

* As for the Microphone pre-amp, I really think that it is necessary to use one, since most Sound Card's mic inputs are awful. I bought the Behringer UB802 Mixer/Pre amp (and no!, I have No affiliation with Behringer, they just happen to sell some pretty decent products, for dirt-cheap prices). The UB802's frequency response proved to be excellent, so I can recommend it. Can be found for $50 USD.

* As for the Sound card, I use the M-Audio Revolution 7.1 ($99), which according to RMAA, provide pretty outstanding performance, and definitely won't be the weak link at the recording chain. Anyway, the guide expects your card to play 32 bit samples from Cool Edit, so I'm not sure a 16bit card will work well here.

Hardware hookup:



Notes regarding Hardware setup:

* Put the Microphone exactly where your head is going to be, at the sweet spot.

* Take notice that the Microphone Pre Amplifier is outputting a mono signal, which goes into the Sound Card’s stereo input. You should find out if the mono signal is received by the Left or Right input channels of the Sound Card. This info becomes handy later.

* **DRC** doesn't compensate for time misalignments caused by different distances between the speakers and the listening position (though it compensate for time misalignment and phase errors between the different speakers in a single loudspeaker, for example between the tweeter and the woofer). Having the speakers at a different distance from the listening position usually cause weird "phasyness" artifacts that are clearly audible. Usually less than 10 cm are enough to cause clearly audible problems. Of course also the microphone need to be exactly at the listening position, i.e. at the same distance from both loudspeakers if the speakers are properly positioned.

* **WARNING** – During the filter creation, you’ll be asked to play a pretty long, log Sine Sweep test tone. This test tone can be a real hazard to your speakers, mainly to your tweeters, when played at high SPL levels. Furthermore, in order to achieve good results with DRC, you’ll have to record this test tone with a pretty decent S/N factor, somewhere around 80-90dB. In order to achieve it, three things need to be happen :

1) Your recording hardware need to be of good quality.

2) The log Sine Sweep test tone needs to be long.

3) The SPL reading, at the sweet spot, while playing the log Sine Sweep, needs to be ~ 85-90 dB SPL.

Requirement #3, is, of course, the reason for this warning. At the beginning of the Step by Step guide, stages “A” to “L” will guide you of how to play a log sine sweep through your speakers, while keeping the speakers safe. Please, follow these steps, even if you think they are unnecessary in your situation. You will only need to follow them once. After this one time, you’ll be able to always start *after* the safety procedure.

Final note before we start: In this guide, I tried to make your life as easy as I could. As a consequence, this guide expects total obedience. You should follow the steps with 100% precision. For example, If you’ll decide, that it is better not to fully close a program, when the guide say so, since we’re going back to this program later, you might discover that it will be harder to follow the guide, later. Be creative only at the 2nd time you use this guide.

The Safety procedure:

A.

Find out your tweeter's **continuous power handling** rating. It can be usually found at your speaker's website, speaker booklet, etc.

If the rating value is **not** somewhere in between 1-20 Watts (for example, if it's 50W, or more), then this is probably not the true rating value we're looking for. In this case, it is better to assume that we do not know the true **continuous power handling rating** of the tweeter, and because of this it will be safer to play the log sine sweep, using 1Watt, max,

Important – if you do not know your tweeter's **continuous power handling rating,** going over 1 Watt with the sine sweep, can destroy the tweeter, so please, think twice if you want to take this risk.

In case the tweeter's rating value you found, DOES give the impression to be the right one, you should still use only HALF of this rating, just to be on the safe side.

B.

Now we're going to find out how high you will be able to set the amplifier's volume level, when playing the log sine sweep, without going over the wattage rating which is the max for your tweeter.

For this test, you'll need the Multimeter.

Set your Multimeter to a low "ACV" range (if you have a range value between "4" and "20", use it. If you have more than one range value in between 4-20, pick the lowest.) By picking such a low range value, we'll be pretty much immune to the Multimeter's reading inaccuracies.

C.

Connect the Multimeter to the left (or right) speaker wire ,which comes from the amp.

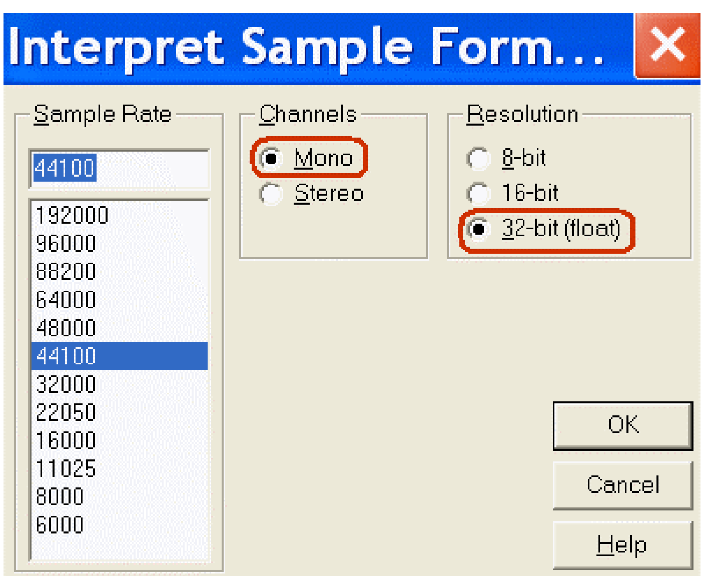
D.

Go into Cool Edit pro, and follow the images:

(If you can't find the feature "File", then you're probably at the "Multitrack" mode. press "F12", now you shoud be able to find it.)

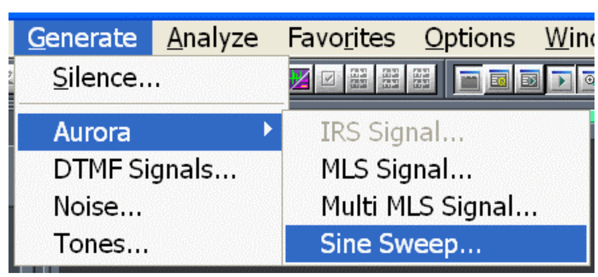
E.

Follow the image and press "ok".



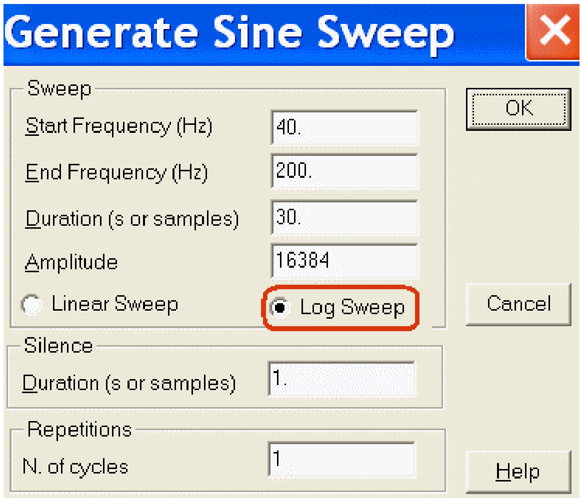
F.

Follow the image.



G.

Follow the image and press “ok”.



H.

Now you should have a 30 seconds log sine sweep from 40hz to 200hz, in front of you.

Turn your volume knob (amp or sound card) to a low position, and press “play”.

I.

Throughout the entire log sine sweep, the Multimeter should pretty much show a constant voltage value (+/- 5%). If it doesn’t, **<u>don’t</u>** continue any further before you figure out why, and how to correct it (one cause for it could be that you have a “loudness” feature on) .

While the test tone is playing, move the volume control knob gradually higher, in order to set the voltage limit, which, of course, should correspond to the Wattage limit for your tweeter. You should figure out the voltage limit, according to the following formula:

V = sqrt(P * R)

Where “V” = voltage, “P” = Power (ie.Watts), “R” = Resistance (which in this case, will be the “Impedance” of your speakers, which should provide a close enough approximation, at least for this task, as I’ve been told).

J.

Let’s take an example for the last section: If your speaker’s Impedance is 8 Ohm, and you want 2 Watts to be your output limit (which is ONLY if you found out that your tweeter’s continuous power handling is AT LEAST 4 Watts), then:

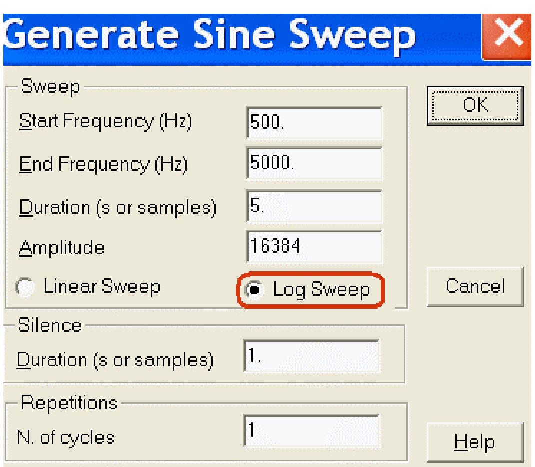
V = sqrt(2 * 8) = 4.0V (@ 8 Ohm)

So, in this case, you should be setting the limit of your volume control, only when the Multimeter shows 4 Volts.

Now that we know what is the maximum volume setting we can use, let’s move on.

K.

Do steps “D” to F” again, then follow this image, and press “ok”.



By this time, it could be that Aurora already crashed on you. Stay Cool ☺.

L.

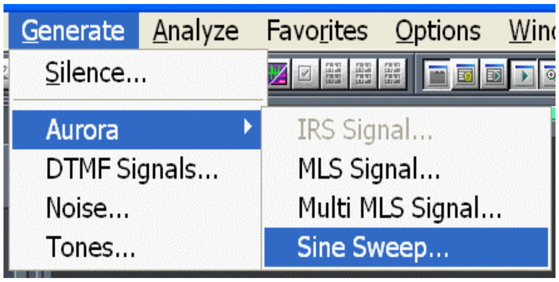
Now, go to the listening spot, and using the SPL meter, measure the SPL output you get, while one of the speakers is playing (doesn’t matter which) the new log sine sweep we’ve just created, when the volume control is set to the limit we figured out in section “J”. The SPL meter should be set to “Slow” reading.

The SPL reading you’ll get won’t be too consistent (heck, that’s why you’re using DRC in the first place), but, if it is around 85 dB SPL, or more, you should be just fine. Less than that, and your results might be less than optimum.

That’s it, close Cool Edit.

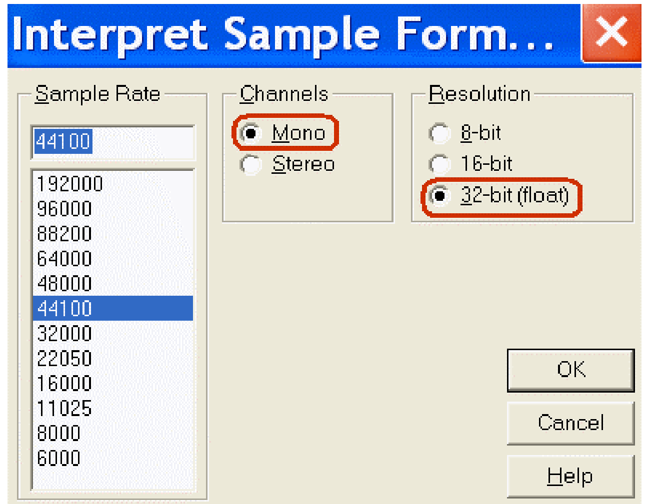
<u>The DRC Guide:</u>

1.
Enter Cool Edit Pro, and start following the images.



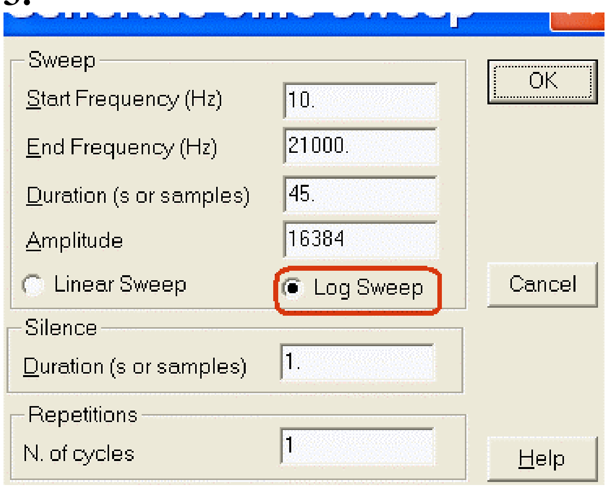
(If you can't find the feature "Generate", then you're probably at the "Multitrack" mode. press "F12", now you shoud be able to find it.)

2.



Press "ok".

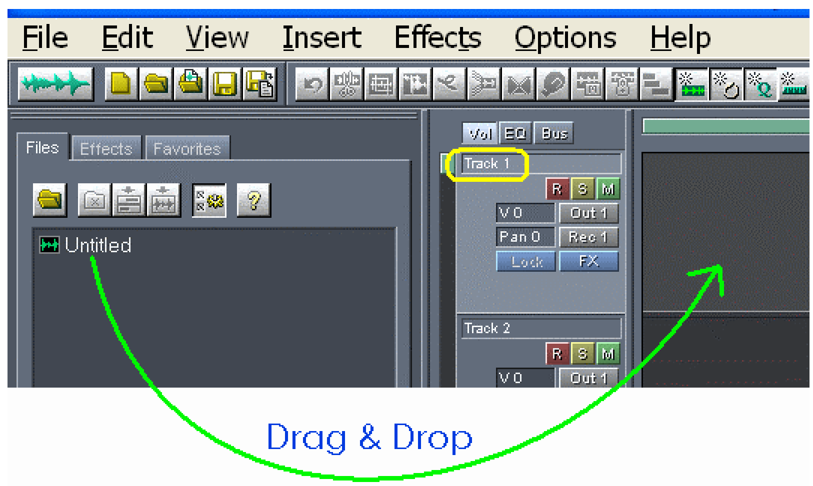
3.



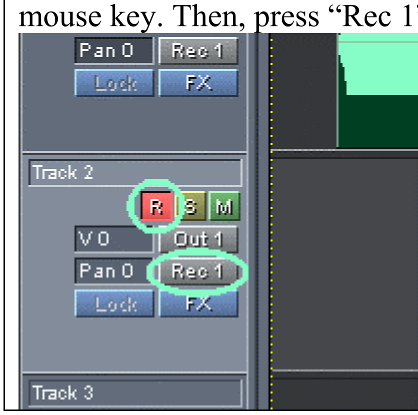
Press "ok", and then press "F12", to go into the "Multitrack" mode of Cool Edit Pro.

* Be aware that an ***inverse*** of the *log sine sweep*, has been automatically created and copied to your ***windows clipboard***, right after the log sweep was generated. We'll use it later.

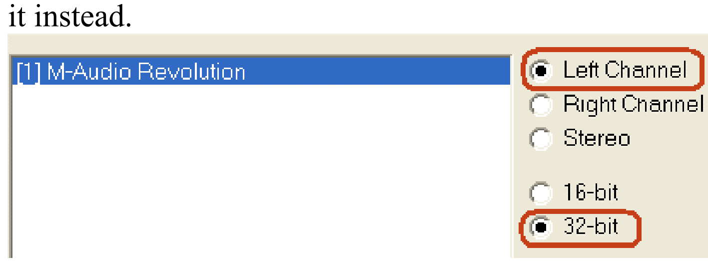
4.
You should Drag & Drop "Untitled", into the "Track 1" strip, using the left mouse key, just like shown in the image below.



5.
Press the red colored "R" of "Track 2", with the left mouse key. Then, press "Rec 1", using left mouse key.

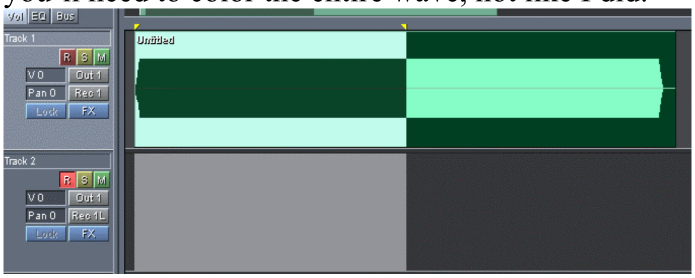


6.
Chose here "Left Channel", if your microphone pre-amp is outputting to the Sound Card's left channel line in. If yours is connected to the "Right Channel", chose it instead.



7.

Now you need to mark the entire green wave (no more, no less). In order to do this, come with the mouse key, the closest you can to the edge of the wave (left or right edge, doesn't matter), then press the left mouse key and hold. If you were close enough to the edge of the wave, Cool Edit will automatically help you start your marking EXACTLY at the beginning of the wave. You should finish marking using this same "trick". Please notice than in the picture below, I only did half job, you'll need to color the entire wave, not like I did.



8.

Now you'll need to go to your sound card's panel control, and mute one of the channels, since we want to measure one channel at a time. For this test, let's measure the right channel first, so mute the left channel.

We already set the hardware **<u>output</u>** levels, at the ***safety procedure.*** What is left (no pan) now is to set the hardware **<u>input</u>** levels.

What I do, is to set my sound card's input levels to minimum, and work only with the Mic's Pre Amp input gain control. This process yields the best S/N ratio for me. The problem with this process is, that if you don't have a VU meter on the Mic's Pre Amp (like I have with the UB802), you can't tell if the Mic's Pre Amp is clipping, which is a VERY BAD THING.

So, I suggest you use common sense, and if you see that in order to get a recording, which has about –5 dB peak amplitude (see next section), you'll need to set the Mic's Pre Amp gain control too high, then I suggest you better go **<u>up</u>** with the sound card's input gain, and **<u>down</u>** with the Pre Amp input gain.

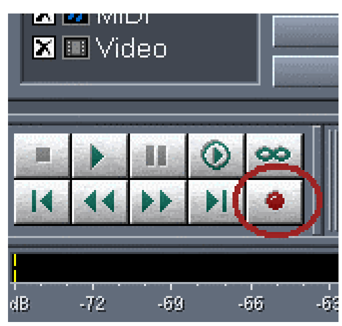
9.

This is an image of Cool Edit's VU meters. During the recording, the highest level that this VU meter should read (ie, peak amplitude), is ~ -3 dB (again, always on the safe side), don't go higher than ~ -3dB, but also, try not to go for much less (with your peak amplitude, that is).

If during the recording, the sound card input will clip, you will see a clipping indication, by a red box, which will show at the right side of the VU meter, inside the yellow circle I made. In that case, reduce the recording levels, and record again. (about *recording again*, see section 11)

10.

Now, make sure the output volume is not higher than the limit we set at section "J", and press the red recording button (It is important to keep the environment as quiet as possible, during the recording procedure.)

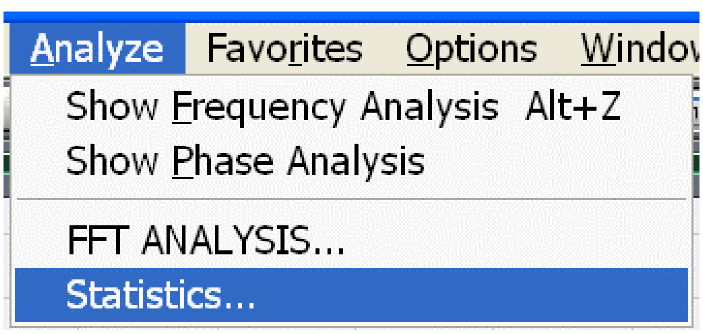


11.

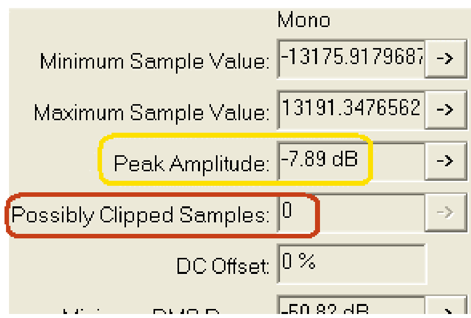
After the test signal recording is finished, "Track 2" should look something like in the image below. If it doesn't, double-check your last actions. If you need to make some changes to the input gain, and record it again, all you need is to do press the "rec" button again, and the new recording will overwrite the current "Track 2".
When everything looks fine, double-click on "Track 2", and this should bring you to a normal editing screen, and "Track 2" will be in front of you.

12.

Now that "Track 2" is in front of you, do the following:



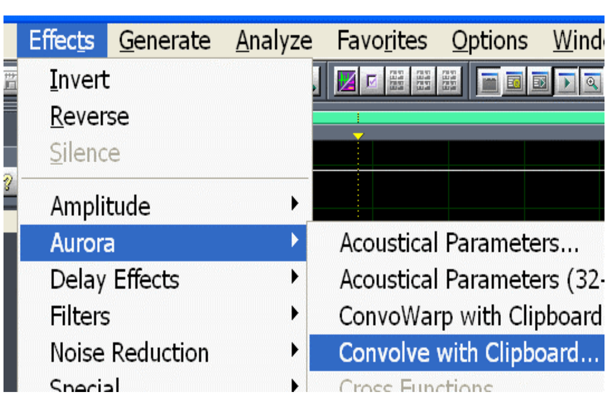
13.



Look at the yellow and red boxes. If the ***Peak amplitude*** is far from – 3 dB, by several dBs, I suggest you record again, using a higher recording gain. On the other hand, if the ***Possibly Clipped Samples*** box, shows anything else apart "0", do the recording again, this time with lower input levels. Only after everything is perfectly fine, move to the next section.

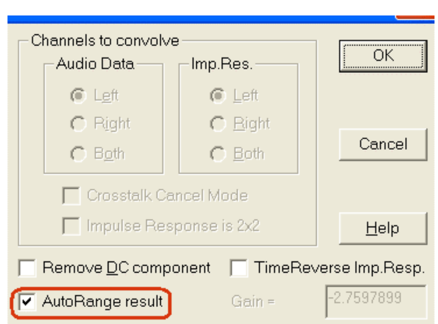
14.

Press "CTRL+6", and then follow the image.



15.

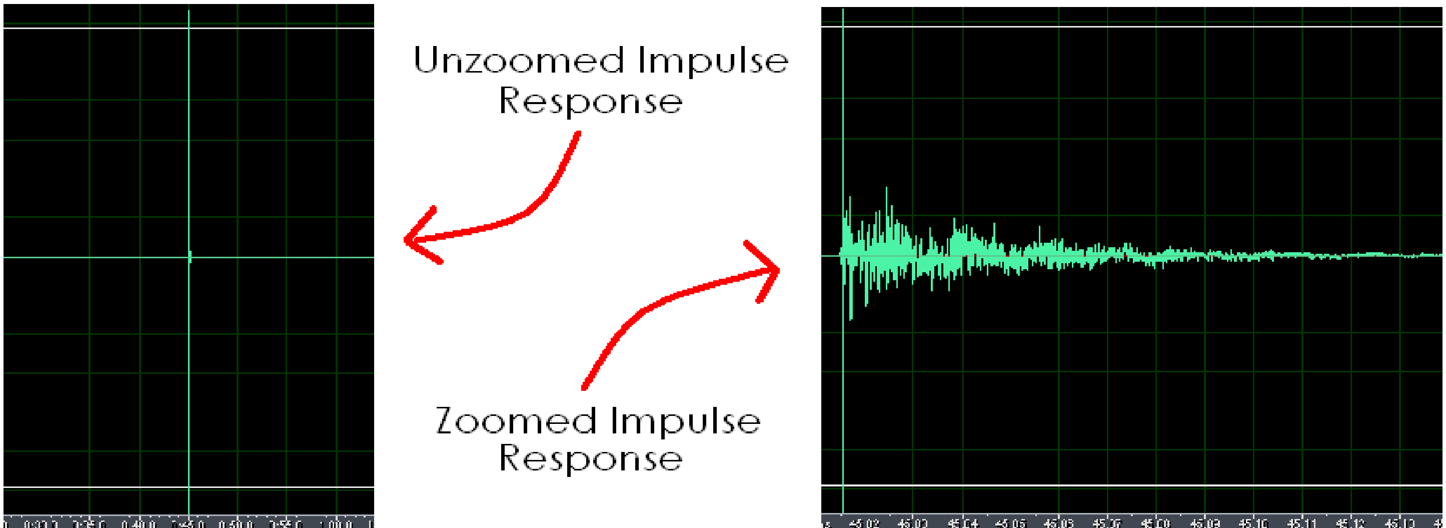
Press "ok".



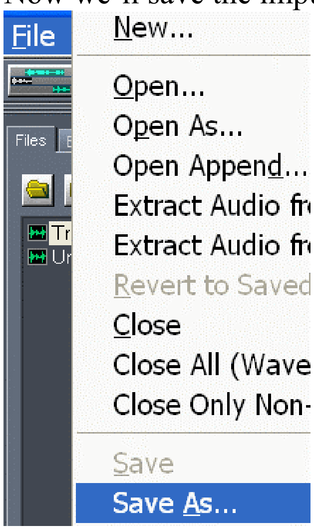
16.

Press "ok".

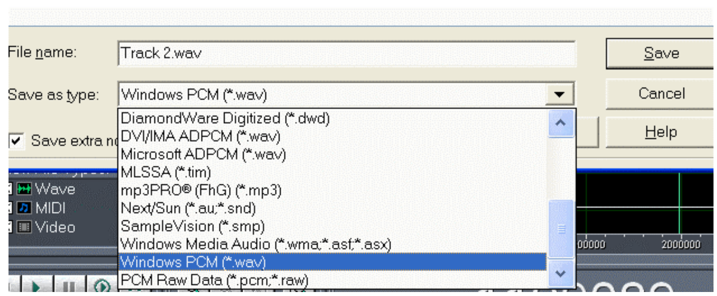
17.
On the left image, you can see how the non zoomed impulse response, should look. On the right image, you can see the same impulse response, only with a zoom on the main spike (which can only be seen as a tall line, on the left image). Your results should be similar to those. If they are not, recheck your process.



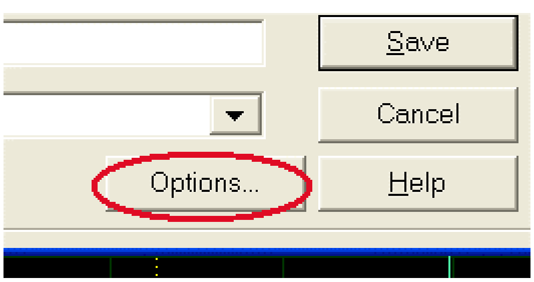
18.
Now we'll save the impulse response to a file.



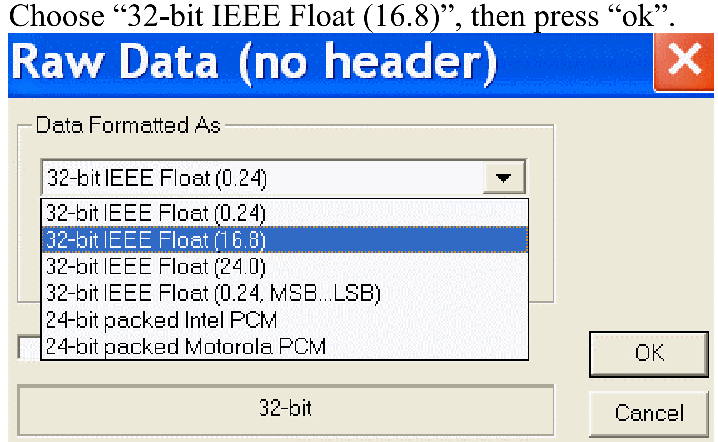
19.
Choose "PCM Raw Data".



20.
Now press "Options…".



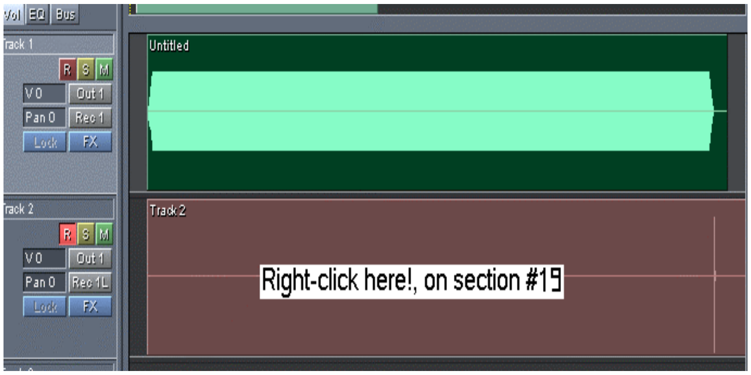
21.
Choose "32-bit IEEE Float (16.8)", then press "ok".



Now pick a name for the file, "Right Speaker" is a good choice, and now you can finally press "save".

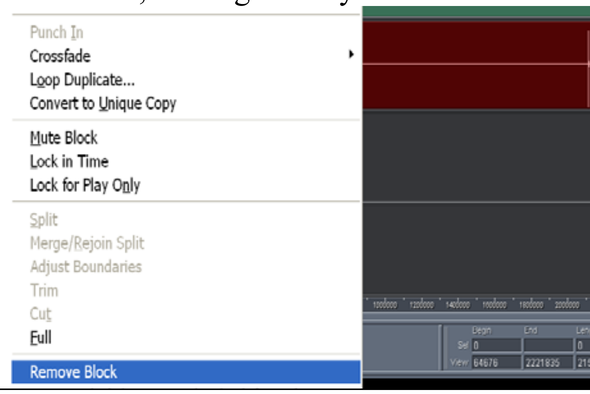
22.

Press “F12” to go to the “Multitrack” screen again. This is what you should see.



23.

Now right-click on “Track 2” (the red track), you’ll see the following menu appear. Press “Remove Block”. This will clear the wave from “Track 2”, making it ready for the next recording.



24.

Do steps 7 to 21 again, for the left speaker this time. Ouch!, I know.

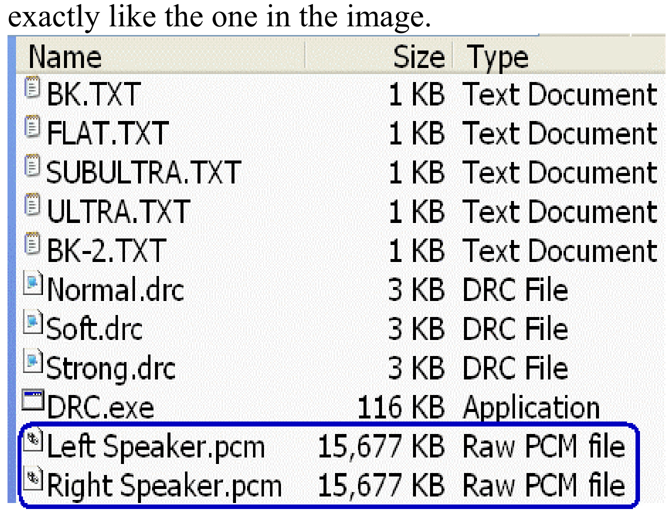
When you’ll finish, you will have two files:
1) “Right Speaker.pcm”
2) “Left Speaker.pcm”

These are the files that DRC is going to work with.

Now, close Cool Edit Pro.

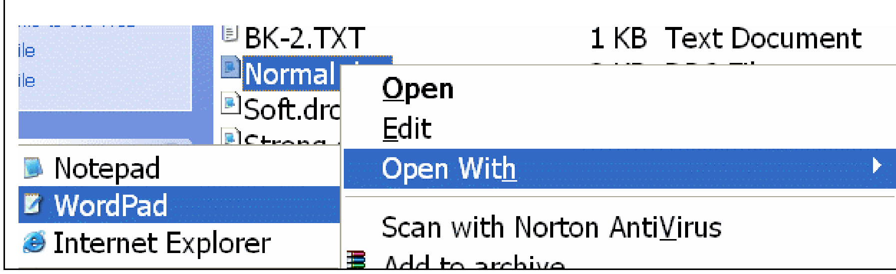
25.

Create a new directory, let’s say “C:\DRC”, and copy all the files from the “Sample” directory, of the program “DRC”, into it. To this directory you should add the last two files, which you can see at the image below, which are the files we’ve just created with Cool Edit. The directory should look exactly like the one in the image.



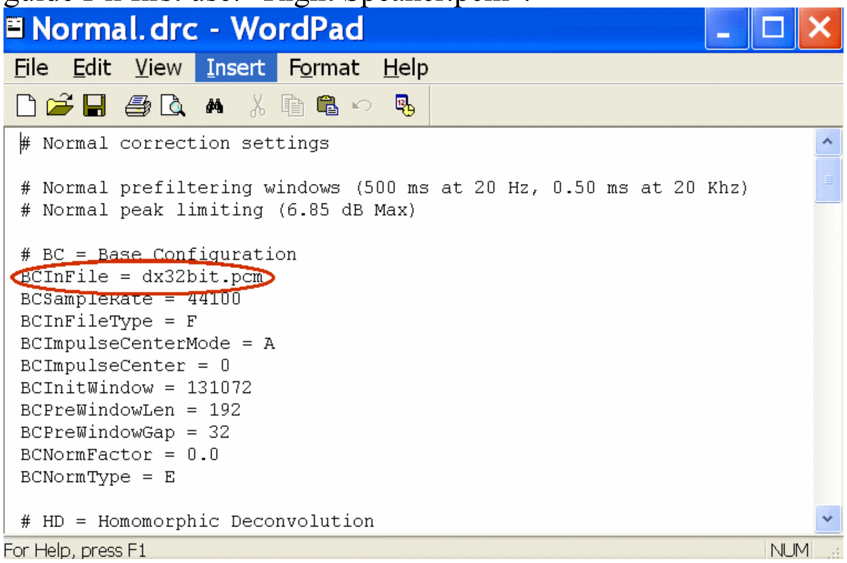
26.

Now you have to decide how accurate you want DRC to perform. You have 3 default options: Soft/Normal/Strong. For this guide, let’s choose Normal. Since we chose “Normal” we’ll need to edit the file “Normal.drc”, which contains all the instructions to make the filter. You can use WordPad for this.



27.

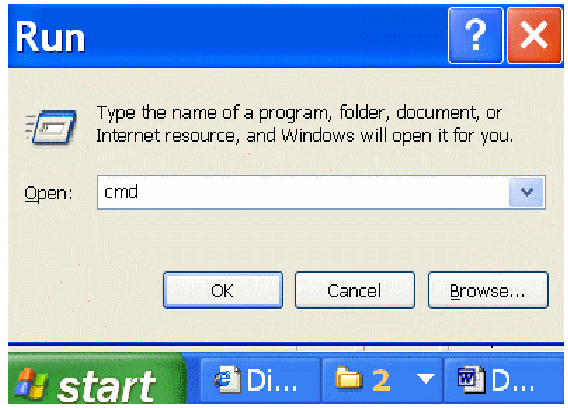
Locate the entry “BCInFile = dx32bit.pcm”, and instead of “dx32bit.pcm”, write the name of one of our pcm files. In this guide I’ll first use: “Right Speaker.pcm”.



Save and exit.

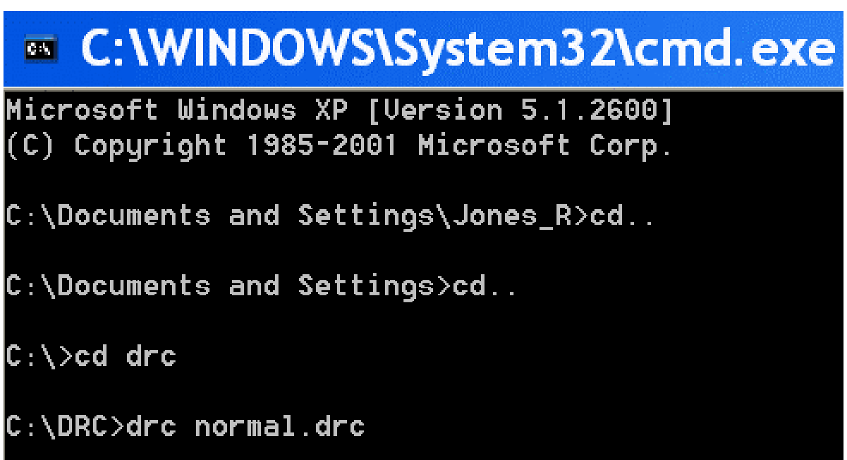
28.

Open a command prompt window. You can do it through “Start -> Run”, then write “cmd”, and press “ok”.



29.

In the command prompt window, go to “C:\DRC”. When you’re there, write “drc normal.drc”, press “enter”, and wait several minutes, while DRC calculates the filter for the impulse file you’ve inserted in section #27.



30.

When step 29 is over, you’ll discover two new files inside the directory “C:\DRC” :

1) “dxf.pcm”
2) “dxtc.pcm”

Rename these files to:

1) “Right Speaker–dxf.pcm”
2) “Right Speaker–dxtc.pcm”

*Be careful not to mix up between “dxf” and “dxtc”.

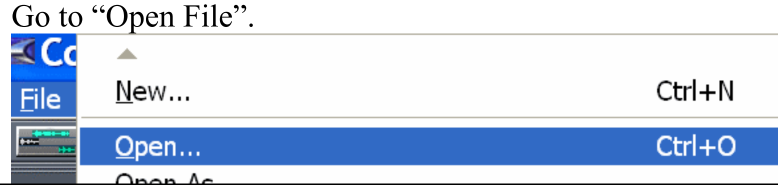
31.

Do steps 26 to 30 for the other pcm file, “Left Speaker.pcm”. (Ouch II!, the sequel.)

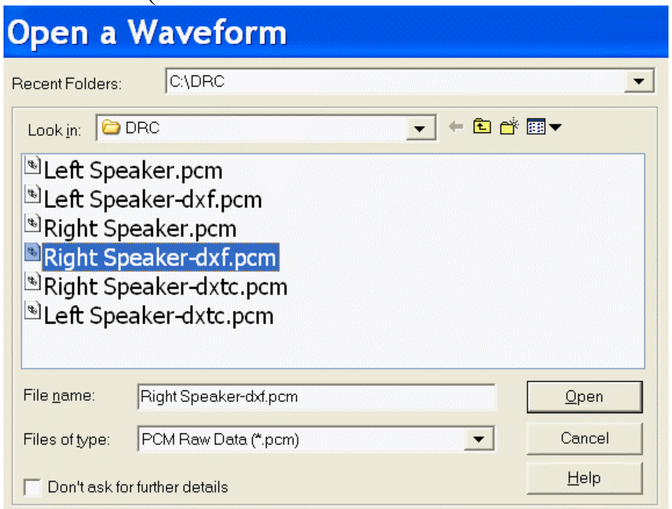
When you’re done, thank DRC, you won’t need it anymore.

32.

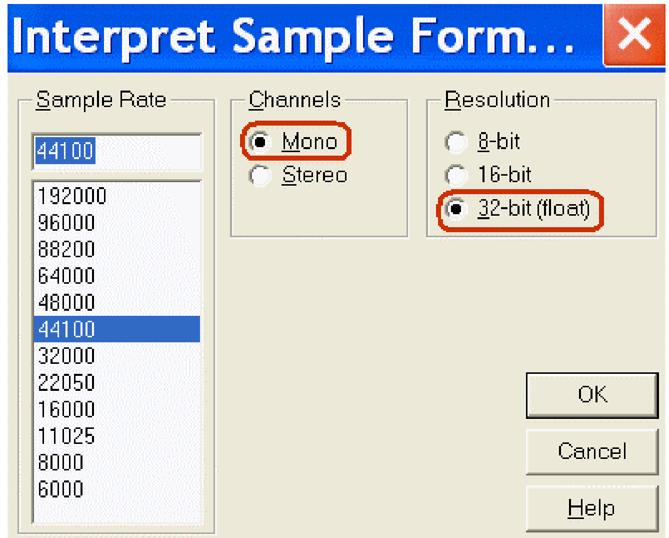
Go back to Cool Edit Pro.
Go to “Open File”.



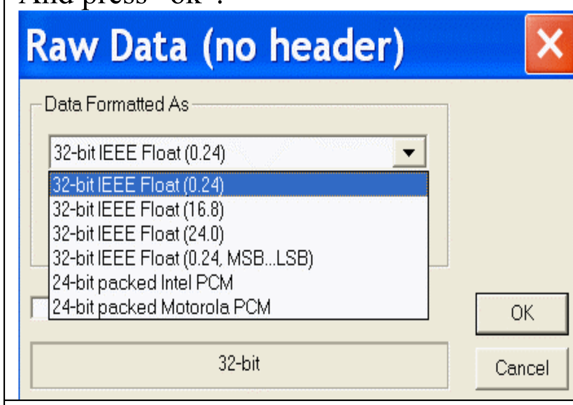
33.
Open the file “Right Speaker-dxf.pcm”, inside “C:\DRC” (“dxtc” is not needed for the filter creation)



34.
Choose the following and press “ok”.

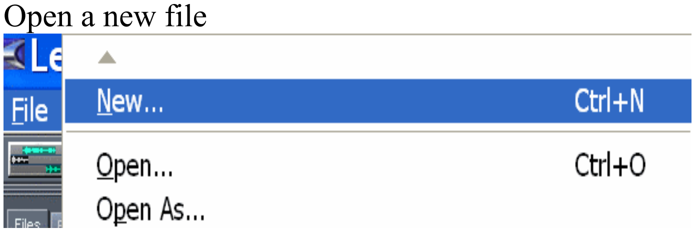


35.
Choose “32bit IEEE Float (0.24)”,
And press “ok”.

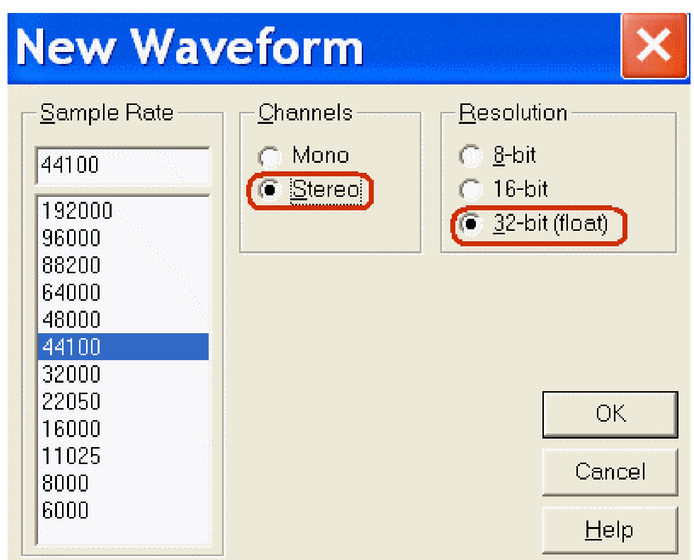


36.
Open “Left Speaker-dxf.pcm” in the same way (steps 32 to 35).

37.
Open a new file

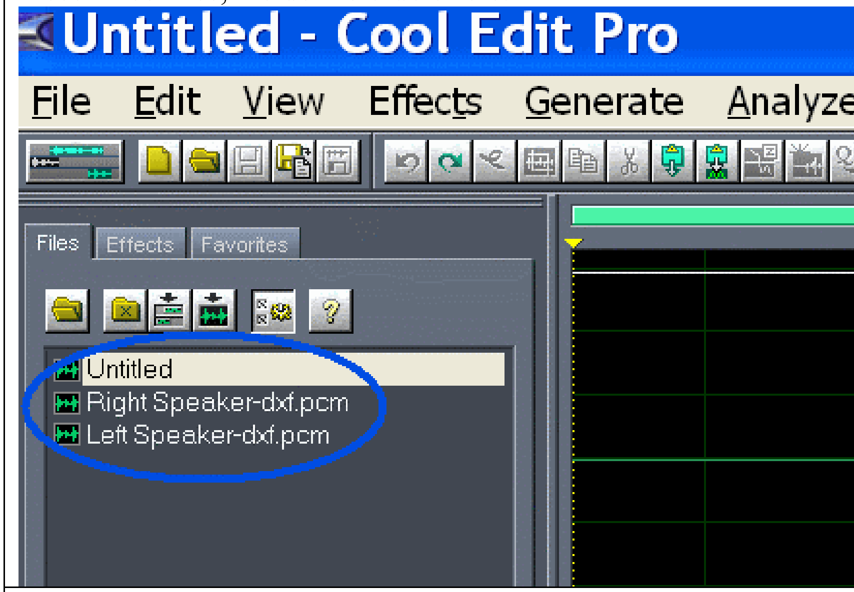


38.
This time notice we will be using a stereo file. Press “ok”.



39.

These are the three files that you should see now in the left screen of Cool Edit Pro. In section #40, I will ask you to click on these three files, inside the blue circle.



40.

Double click on “Right Speaker-dxf.pcm”.
Press “CTRL+C”.

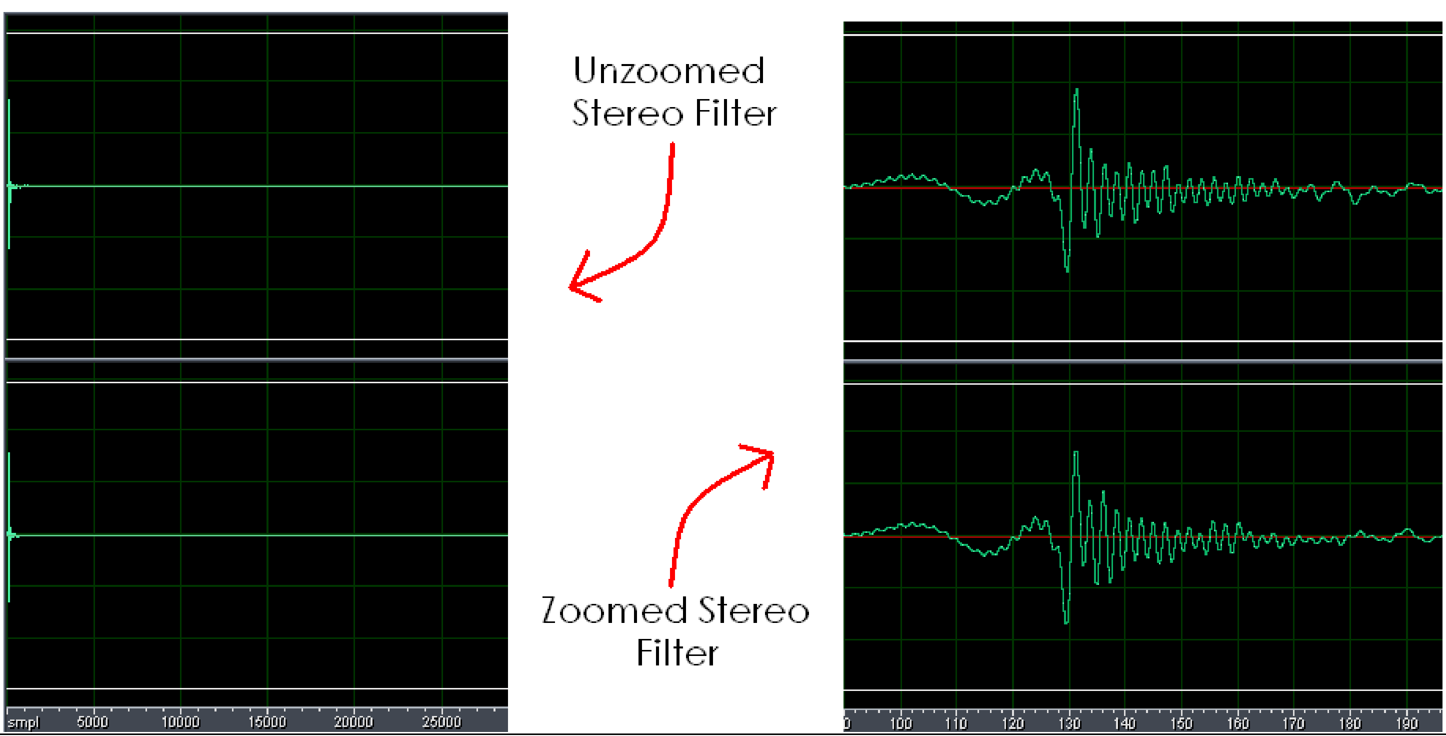
Double click on “Untitled”.
Press “CTRL+R”.
Press “CTRL+V”.

Double click on “Left Speaker-dxf.pcm”
Press “CTRL+C”

Double Click on “Untitled”.
Press “CTRL+L”.
Press “CTRL+V”.

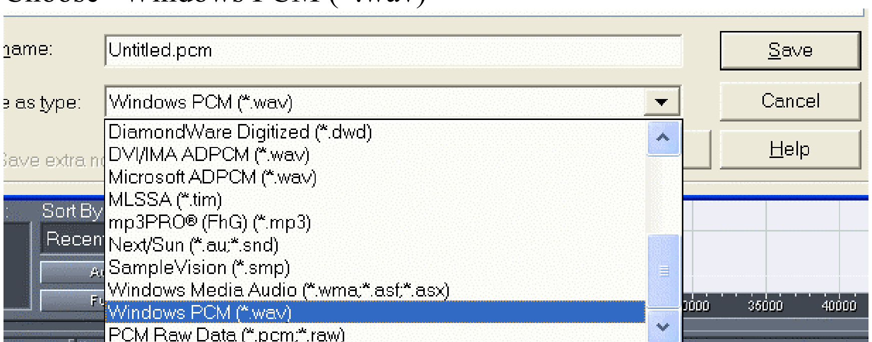
41.

That’s it, if you did everything right up to now, the stereo correction filter should be done, and should look similar to the images below. On the left side, you have an un zoomed image of the stereo filter (one impulse response for each channel, right at the beginning of the wave file). On the right side, you have another image of the same filter, only now I did a zoom on the first few milliseconds of the filter, so you’ll have a general idea of how the actual impulse response should look up close. If your filter vary by a very large degree from the images here, you should double check your process. If things looks pretty much the same, you can follow up to the next section, where we’ll save the filter.



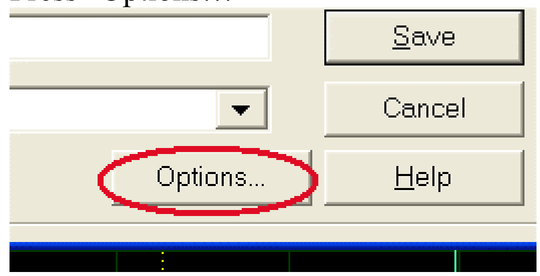
42.

Press “CTRL+S”.
Choose “Windows PCM (*.wav)”



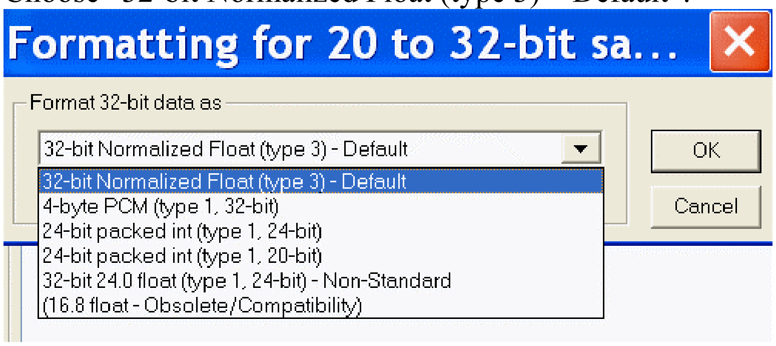
43.

Press “Options…”



44.

Choose “32-bit Normalized Float (type 3) – Default”.



Press “ok”.

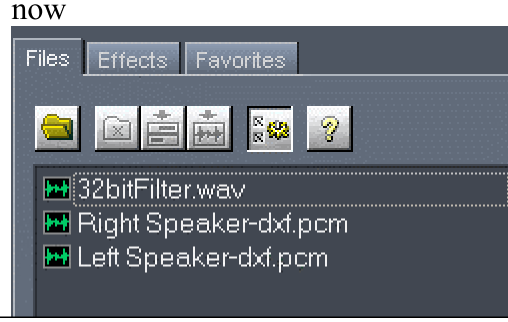
Change the name of the file (which currently is “untitled”) to something you wish. For this guide, let’s choose “32bitFilter.wav”. Press “save”.

That’s it, you just saved your new DRC filter.

45.

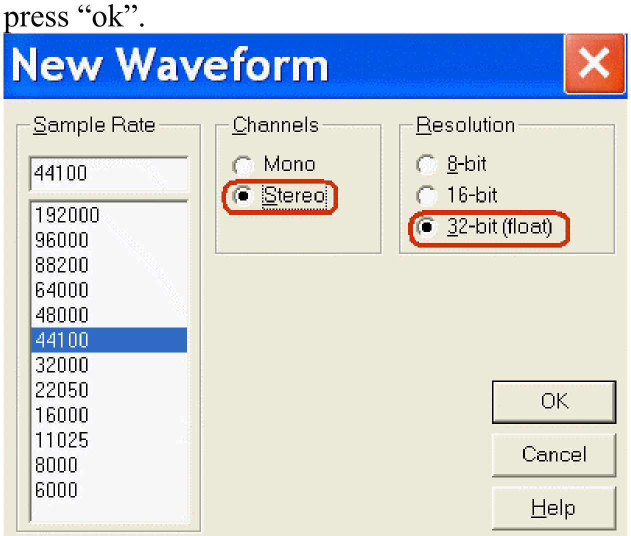
Now we’ll want to know how much we’ll need to attenuate the volume level of our filter, in order to avoid clipping at the convolution stage. The actual attenuation will be done inside the real time convolution plug-in. At the next sections, we are only going to find out the attenuation value to use with the plug-in.

The left screen of Cool Edit should look like this now

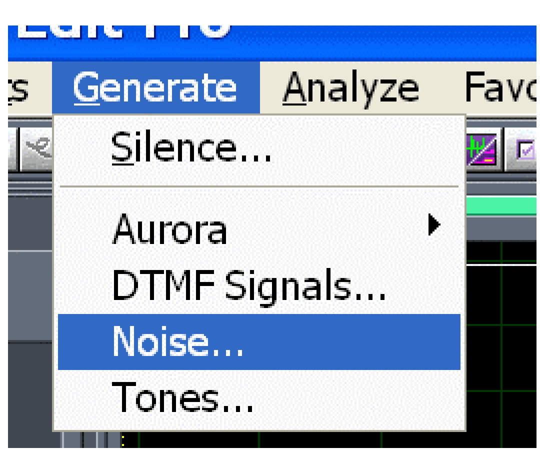


46.

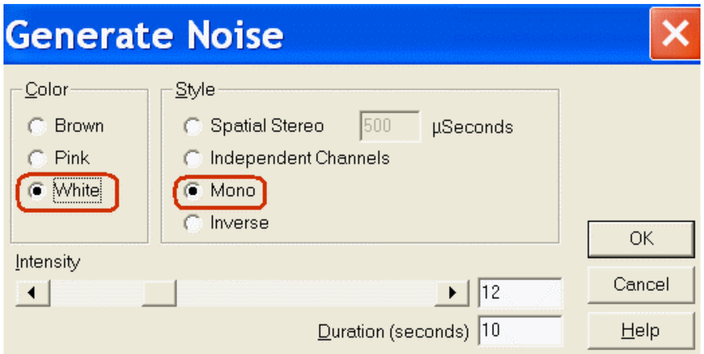
Press “CTRL+N”. Then follow the image below and press “ok”.



47.

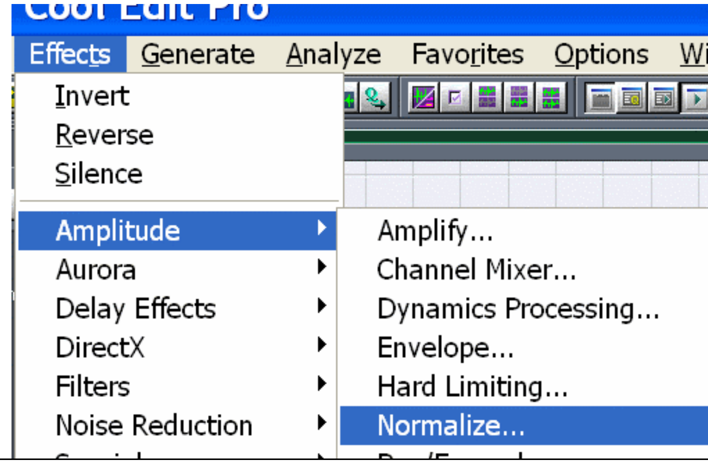


48.

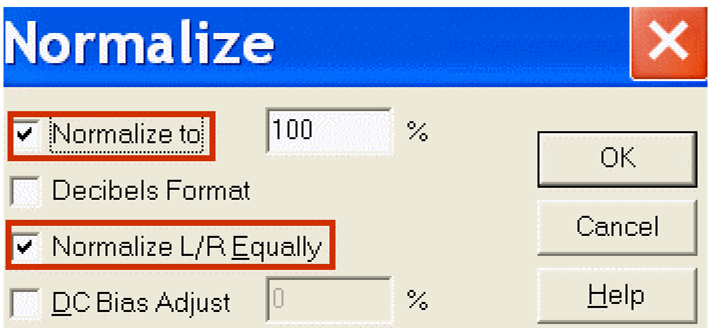


Press “ok”.

49.



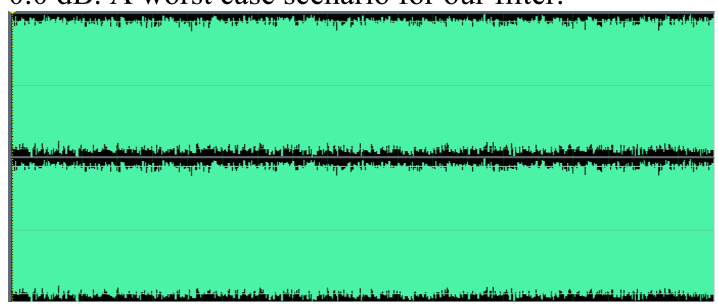
50.



Press “ok”

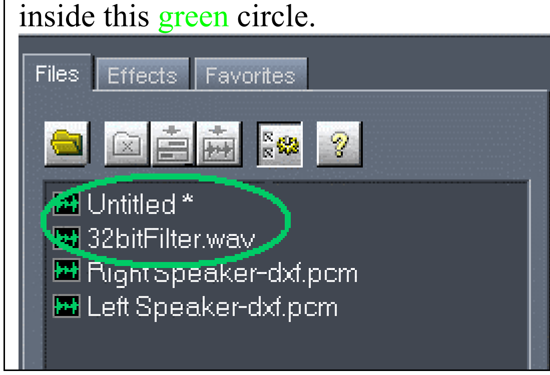
51.

Now you should have 10 seconds of white noise at -0.0 dB. A worst case scenario for our filter.



52.

In section #53 I’ll ask you to click on the two files inside this green circle.

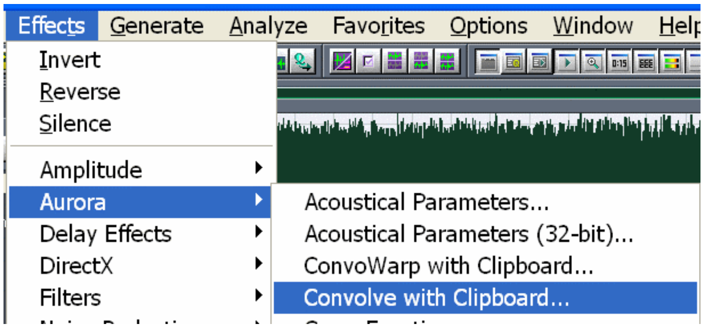


53.

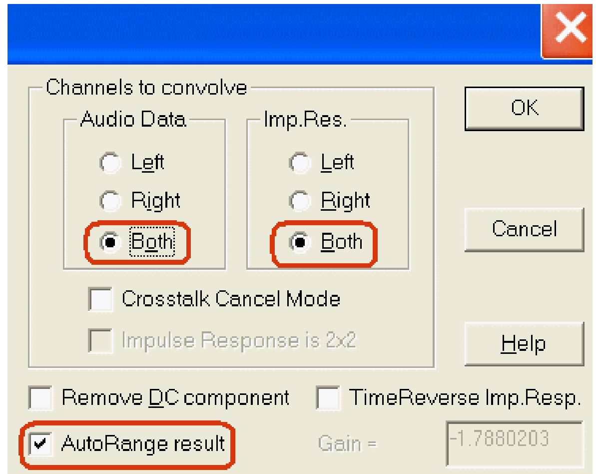
Double-click on “32bitFilter.wav”.
Press “CTRL+6”.
Press “CTRL+C”.

Double-click on “Untitled”.

54.



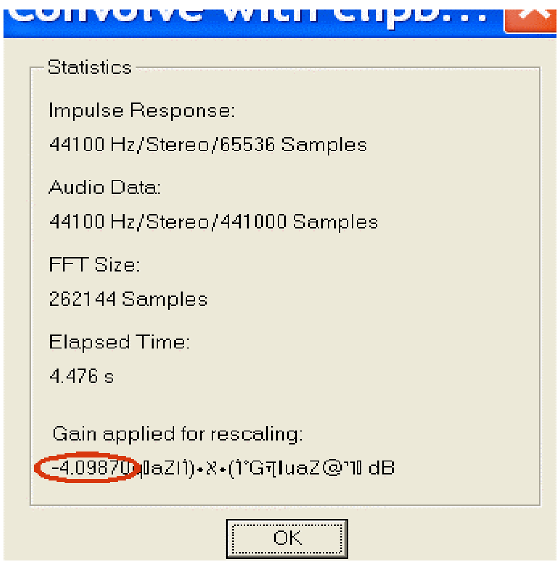
55.



Press "ok".

56.

The value inside the red circle, is the value you'll need to attenuate the filter (inside Foobar2000, or whatever player you'll be using), in order to avoid clipping.



57.

Unless you have the dynamic range to spare on the (hefty) attenuation value received in the last section, I suggest you do the same test (sections 52 to 56) but instead of convolving with –0.0 dB of white noise, you can try convolving using one of your most dynamically range compressed CDs (at close to –0.0 dB peak amplitude).

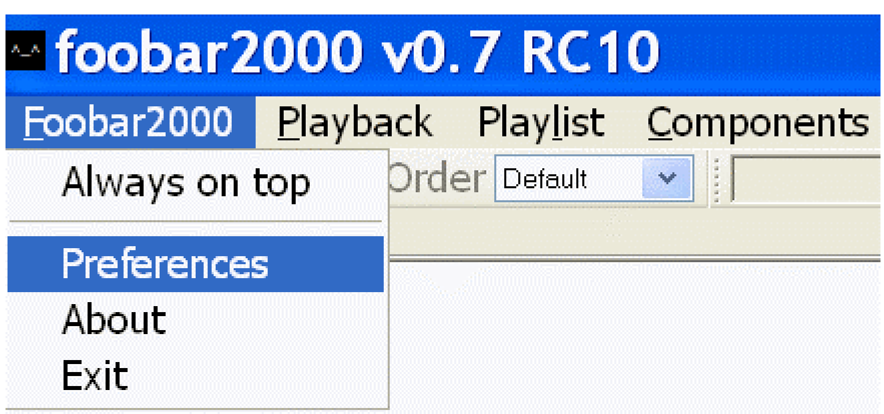
Doing this with such a song from Craig David's last CD, I got an attenuation value of –2.6 dB, which seems to work fine, producing zero clipping, with any other CDs I tested as of yet.

58.

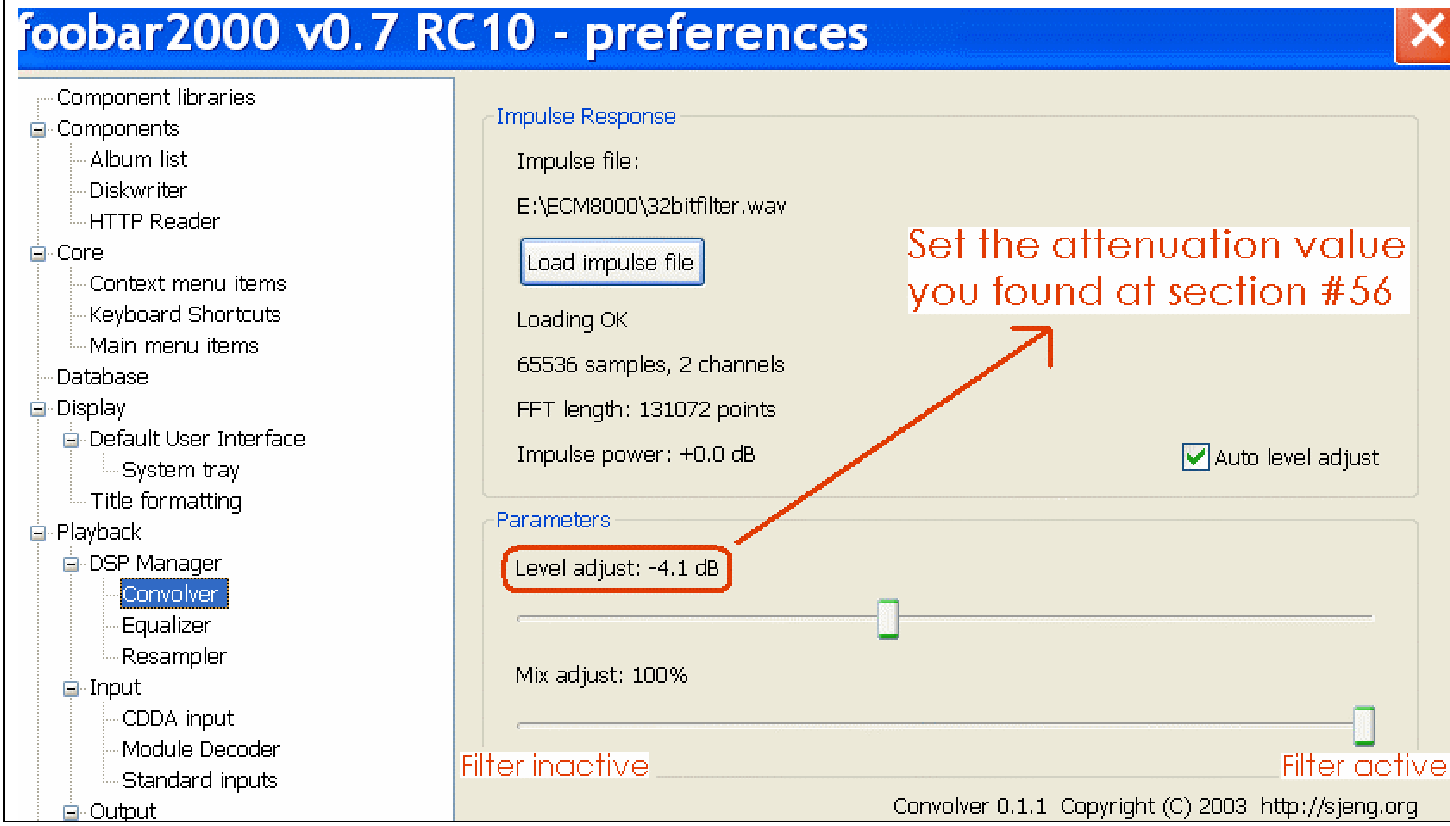
Foobar2000 users should follow the next two sections.

Enter Foobar2000.

59.



60.



Epilogue

This Guide was meant to give you first time experience with DRC and its capabilities. Now that you have this experience, if you are satisfied and want to continue using this freeware and create more filters, please, register Aurora/CoolEdit, since unlike DRC, they weren't created for the benefit of their users only.

Created by Jones Rush