Audio Recording Guide

1 - Music Production & Business

2 - Music Recording & Mastering

3 - Recording Equipment

audiorecording.me: Audio Recording Tips

TABLE OF CONTENTS

Audio Recording Guide

TABLE OF CONTENTS

Audio Recording Guide

1 - Music Production & Business

Music Production Process: How does it take to produce a song?

Everyone has, at certain moments in his life, listened to commercial FM radio stations. You might hear a lot of great songs and some are still memorable today. Then you might ask: how did these songs come out to be played on the radio? Who created this one? Is this done by the song artist himself or is the recording company only involved?

Then you might see music videos of artists playing in bands, along with their instruments that might take place in a recording studio. Then again you might ask yourself: was the song recorded in that studio?
A lot of fascinating questions… Even I, before, didn't have a single idea of how the entire music production process worked.

Below are the steps and processes of music production:

Step 1: The moment the artist recording label (like Universal Records, Sony BMG, etc) decided to start an album recording project. The recording label assigns a record producer to the artist. In some cases, the artist will be the one to choose a record producer, but this is subject to recording label approval.

Step 2: The recording producers, along with what has been agreed with the artist and recording label, will finally decide on what type of sound they are planning to achieve. Some rough ideas on songs to be included on the album might be discussed in this stage (if the artist had some rough demos before).

Step 3: The recording producer drafts the recording plan and activities, and quotes the budget. The recording label will be the one to entirely finance the recording of the album songs.

Step 4: The recording producer with the coordination of artist and recording label will shop for songs, and contact music publishers they know or even songwriters they have partnered with. They will decide the final song list to be included on the album along with the possible singles (hit songs).

Step 5: The recording producer starts the recording session with the artist.
The recording producer is responsible of the song arrangement, instruments added to the song, sound and style of the music. And then the recording producer will be the one to rent a studio, hire an engineer, hire additional musicians and book additional vocalists to complete the recording. The budget is provided by the recording label.

Step 6: Once everything is recorded, the mixing session starts. The recording producer hires a mixing engineer appropriate for the sound of the album.

Step7: Once the mix has been finalized, the recording label and artist will approve it. Once approved, the mastering session will start.

Step8: The mastering session prepares all tracks for commercial production and replication. In this stage, the recording producer hires a mastering engineer. The song artist and recording label should also approve the master. The product of the mastering stage is the master CD ready for replication (that adheres to CD red book standard: *http://www.a1cds.co.uk/master.htm*). The recording producer work is now complete and he/she will provide the CD to the recording label.

Step9: The recording label will finalize the album art design, marketing plan and replication.

Step10: Singles or hit songs from the album (selected tracks) will be provided to the radio stations for massive airplay as a part of album marketing. Music videos will also be released to music video channels.
And the artist will start visiting radio and TV stations, and perform concerts as part of the promotion.

Step11: Listeners like us can now start to listen to the songs aired on the radio. The music production process has been completed.

Hiring a music producer: Basic tips and guidelines

If you're an artist, band or songwriter, are you about to hire a music producer/record producer? Think twice because you may or may not need a record producer at all.

Below are the conditions to 'really' need a record producer:

1) If you desperately need a high quality recording of your single. This assumes you wrote the song. Only a record producer can "professionally" transform your completely written song into a professional recording. This single may be used for promotional purposes, such as a demo submission, or to the radio, and so on…

2) If you drastically need to release an independent recording album which will compete with major label artists or other good independent recording label artists. Then you surely need to hire a recording producer to supervise the recording sessions, to create an artistic sonic vision of the album, and to produce a complete master CD.

What if you really needed a record producer, as your need falls into either of the two above conditions? Now, hiring a record producer is not an easy task because you need a "perfect" man to do the job which should pass in all these qualifications:

1 - **Attitude** – this is the most important. Treat a music producer like your dad: what attitudes do you like from your Dad? It is the same attitude requirement you require from your music producer. You need someone who is: supportive, honest, creative, strong and someone that has the potential to love and believe in your musical abilities.

2 - **Experience and mastery in your selected genre** – this is the second most important point. You might ask the potential candidate about his or her previous experiences and completed recording projects. Try to listen to it, and make sure that it is your selected genre. Of course, you should not hire a producer who has lots of experience in producing rock records, if what you need is to produce a hip hop record. It's important to like his/her work. If not, there is no point in hiring that producer.

3 - **Price** – the third most important point. Of course, you should hire a record producer who will offer a price you can afford. I'd suggest cutting the cost by hiring a producer who owns his/her own studio. This means that he/she will not be renting a studio, which could be interesting for both of you.

You really should plan in advance how you are going to pay the record producer back. One good suggestion is to offer him a share of the ownership in the sound recording or song copyrights. So instead of having only you as the songwriter of the song, you can add the music producer as one of the authors (of course the producer is contributing to the song). In that case, you won't need to pay anything to the record producer, other than the future music royalties that your music can earn if it is commercially successfully released.

4 - **Proximity to your location** – of course, it would be nice if both of you were living in the same location and close to each other. The farther the producer will be, the harder it will be for him/her to communicate with you, and the more expensive it will be to supervise the recording projects, since both of you will travel a lot just to get the recording done.

Technical steps in the recording production process

So you are new to music production and recording right? This short guide will explain the technical processes that are happening in the recording production process. This is written from a producer/engineer point of view and does not include the songwriters/artists perspective.

step 1 - **Tracking** is the process of recording tracks to a song. In this process, different instruments required for the song are being recorded; for example guitars, bass and keyboards.

The most complicated things to do in the tracking (the most delicate processes) are recording drums and vocals. Drums need lots of microphones, so the engineer or the recording producer would have to be sure that they are recording it right.

If you are an engineer and you are recording a rock song, I'd suggest you to start with the drums: the kick, snare and cymbals. You need to ensure that the beat and timing of the song are well laid. The drums are the foundation of the beats, on which the guitars, bass and other instruments (even the vocals) will strongly rely.
This can take time, especially if the drummer is often out of beat and timing, so re-recording is commonly done. After recording drums, you can easily record guitars, bass and finally the vocals.

The objective of the tracking stage is to produce as beautiful, clean and crystal recordings as possible. There is no excuse for any performers' mistake; it should be re-recorded.

step 2 - **Mixing** – once you have recorded all the tracks; you are now ready to mix all the recordings. While tracking is often the most time consuming stage in the recording production process, mixing is the most difficult. This is where a mixing engineer selected by a producer will be the one to execute the mixing job.

The overall goal of the mixing is to balance all the recording levels as much as possible, without mud. Instruments sound not be competing with each other. And then here comes the most important, the vocals. Vocals should be strong and clear, and the mix should have its own unique character. Clarity of the mix is of utmost importance.
Getting mixing professionally done is tough, that's why only certified and professional mixing engineers are paid to get this done. But home studios also can; however big producers hire hot shot mixers with solid track record.

The product of the mixing stage is a mix-down. It should be using the highest possible audio resolution; for example a 24 bit, 96 KHz mix-down. No compression or amplification is done yet to the mix down! You should leave this to the mastering engineer.

step 3 - **Mastering** – the mastering stage might not be the most difficult and time consuming one, but it surely is the most important. This is where the final sound of the music production is sculpted and comes to life. The mastering engineer deals with the following sub-steps:

a) *Equalization* – removing some noise, doing some filtering to remove rubble, too much bass, too many highs, adding presence; etc.
b) *Dithering* – this is the downgrade of the recorded bits from 24 bit (for example) to 16 bit, in preparation of production CD masters.
c) *Sample rate conversion* – this is the conversion of the sampling rate frequency of the audio wave (for example from 96 KHz to 44.1 KHz), in preparation of the CD master.
d) *Compression* – this is finally making the music as loud as it can get without getting distortion. This is often the trickiest stage, because some clients tend to abuse this, and to force the mastering engineer to make the recording as loud as possible, ignoring the distortions that will happen along the way.

The final product of the mastering stage is the CD master. It will then be sent to the CD replication plant for mass production, and then will be distributed by the recording label.

How to prepare and submit the mix to Audio CD Mastering Studio

The following are the important requirements that a mixing engineer should check before submitting any mix to mastering facilities for audio CD or track mastering. These are a sort of check list to make sure that your mix is ready for mastering.

Check #1: **Is your mix final? Has it already been approved by the concerned artists and recording producers?**

Purpose: If you create a mix-down, the final approval is done by the artist and the recording producers. You need to make sure that there is no other request for changes in the mix, because once the material will have reached the mastering studio, there'll be no turning back. If you turn back, you need to make a separate mix-down again with the artist/producer requested changes that can take time and destroy your schedule.

Check #2: **You SHOULD mix down to the highest possible audio resolution.**

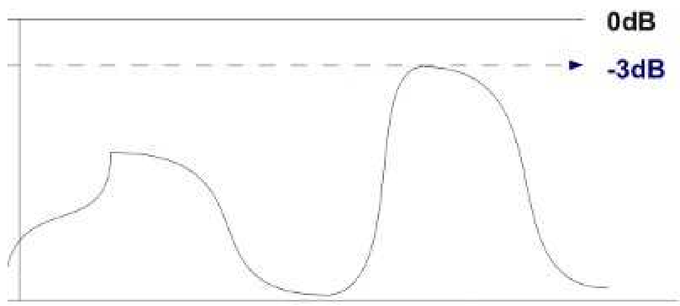
Purpose: You need to create a very high quality mix, don't you? If you mix-down to low resolution such as 16 bit 44.1Khz (by applying dithering for example), you are destroying your mix.
Do not apply any dithering during mix-down. Leave that to the mastering engineer. Also if your audio mixing software/DAW is able to mix-down at 24bit 96Khz (common in most professional studios), then do it. Eventually, that high resolution will be VERY favorable to the mastering engineers when they'll do the mastering work of the track.

The minimum bit depth should be 24 bits (you should never mix-down to 16 bits, ever!). The sampling frequency can be 44.1Khz, 88.2Khz or 96Khz. Using 192Khz is not recommended and isn't a beneficial contribution to the sound quality. But again, *24 bit 96Khz*, or *24 bit 44.1Khz* are the standards.

Check #3: **The maximum amplitude of any wave in your mix-down audio tracks SHOULD not exceed -3dB.**

Purpose: By allowing or setting a maximum amplitude of -3dB, you are giving the mastering engineer a lot of headroom, which is very important. Remember that a lot of headroom means less distortion after the audio mastering job, and a better sounding track.

You can use your audio mixing software to measure the maximum audio amplitude of the mix-down track. Bear in mind that maximum audio amplitude is DIFFERENT from average audio sound level (SPL). That's because maximum amplitude is the maximum recorded peak of the mix-down signal. See the screenshot below:



Check #4: **Never, ever compress your mix-down in an attempt to make it LOUD at the request of your clients.**

Purpose: By compressing to make the mix loud (in an attempt to praise clients or if it's a client request), you are destroying your mix, which is IMPOSSIBLE for mastering engineer to repair and restore dynamics in natural way.

Explain to your clients that it is the mastering engineer's job to make it VERY loud. If they need to hear what it sounds like if it is loud, simply turn the volume controls (of your studio console) to maximum without altering the wave or applying any compression to it.

Permission to record songs: Record producer tips

This guide is useful for anyone wanting to know how to get the permission to record a song. This guide can help you for any of the cases below:

a - You are planning to cover a song which was written by somebody else but was not released (or not popular).
b - You are planning to record a song which was written by your artist.
c - You're planning to re-record a popular song written by a popular writer.

Let's examine what music rights are exploited:

a. **Mechanical rights** – this is the right to reproduce a song in a physical media such as CD, DVD, Video Tape, and so on. If you want to exploit this, you need to contact the music publisher of the song, and to ask for a mechanical license.

b. **Printed sheet rights** – this is the right to reproduce the lyrics of the song.

Those are the only two possible rights exploited when you want to record a song. The "printed sheet" rights assume that you are going to include the song lyrics in your production. However, this can be skipped in most cases.
If you are a record producer, you are in-charge of producing an album that includes songs owned and NOT owned by your artist.

For songs owned by your artist, you simply need a mechanical license written by your artist.
The payment needs to be arranged and there is no standard rule.
In the USA, mechanical royalty rates are around 9.1 cents per song (http://www.songwriteruniverse.com/mechanical.html). If the artist wrote all 10 songs in the album, the mechanical royalty due is 91 cents or 0.91 US dollar. If there are about 1,000,000 albums reproduced, the total amount of royalty due is:
0.91 US dollar x 1,000,000 = 910,000 US dollars.

The question is: Are you, as a record producer, going to pay that amount of royalties?
The answer is NO. In fact, the recording label (not the record producer) will be the one to pay for royalties.
The record producer is only hired to produce recordings. Even the artist is hired by the label to perform. Your job as a record producer is to make sure the mechanical license is executed by the label and you have the permission to record the song.

Another popular question: If the artist is not financed by a major recording label, and if he's the one financing his own records but hire you as the producer, do you still need mechanical license?
There is no need for a mechanical license, because the artist himself is financing the recording session and selling the album, UNLESS you, as a record producer, hire the artist to perform, use his songs and sell the records under your own business name (such as an independent record label).

Finally, many producers might ask: "What is Harry Fox agency, and how are they going to help to secure permissions to record a song?"
The correct answer is that, you have to double check that the music publisher of the song is affiliated with Harry Fox agency. If not, you cannot get any help from them.
Harry Fox agency represents music publishers in issuing mechanical licenses. You can read more about Harry Fox here: http://www.harryfox.com/songfile/faq.html

Even though, as a record producer , you have to be technically good (to deal with sound and recording equipments). You also have to be good in administering legal documents such as licenses, because in the real world job, you'll have to deal with these.

How to broadcast your music online and earn money

OK let's say you have completed your songs and are planning to gain airplay. Cool, but is it really a simple process? ***No***.

It's a bit of a complex matter that needs some preparation. In fact, failing to observe certain required processes can prevent you getting airplays, or at worst you'd never earn a single penny from your promotions. Below are the important processes that every independent recording producer or artist should observe in order to broadcast or promote music online and earn some money:

First things first:

Step 1: Register your work in the government copyright office. If you are promoting your music in the USA, you can register it in the US Copyright office. You can do it online here: http://www.copyright.gov/eco/

The most important thing to do if you own the work, songs (lyrics and melodies), as well as the recording, is to file both PA and SR forms. PA stands for the copyright of the song, while the SR form is for the recording. If you find it expensive and would like to save money in the process, you can register them both using SR forms.

Step 2: As soon as the copyright office confirmed that they received your work, you can register with performing right societies, in order to receive performance royalties for your work. If you are US based, you can register with either of these 3 big societies: ASCAP, BMI or SESAC.

Step 3: Once you are a member with any of the mentioned societies, you need to register all of your songs/work (of course the songs that you need to broadcast or promote in the airplay) with the performing right society database. This will help them to monitor those performances of your songs and pay your money.

Step 4: After song registration, you also need to register your work with Nielsen Broadcast Data systems. The main purpose of this registration is for this system to capture performances of your songs, which are used by various radio stations.

In other words, Nielsen Broadcast Data systems make it possible for them to check which radio stations are playing your work. To register, send an email to: **clientservices@bdsonline.com**, then use the subject "Virtual encode", and then in the body area of the mail, include the following required information:

a. Your full name
b. Company name or your label name (if you do not have a label, use your full name)
c. Contact number (cellular phone number is fine as long as you include the country code).
d. Primary email address that you are using

Nielsen BDS representatives will then send you additional procedures to upload your work online.

Step 5: Once your work has been successfully encoded, you can proceed with actually submitting your songs to radio stations. For an online radio, you can submit it to Jango.com, which will expose your songs to a lot of listeners looking for fresh or established artists.

Offline promotions, especially commercial FM radio stations, may be a bit harder to get, because you would need a radio promoter to actually have your work submitted to those radio stations.

Performance Royalty System
The role of music consumers to cure the weakness

Every songwriter/publisher needs to be affiliated with performing right societies which will be the ones to collect performance royalties in behalf of the songwriters/publishers. It's trivial how these big performance right societies like ASCAP/BMI/SESAC will be able to track the performance of the song with the needle-tip accuracy.

Of course, it will be very easy to monitor the performance of the song if it is aired in commercial radio, big commercial/entertainment establishments, major college radio stations as well as the major television stations. The reason is that they have extensively wide and popular media coverage allowing ASCAP/BMI and SESAC monitoring systems to easily tap and get performance data.

The only problem I have seen is that the main players in this market are also the big players. This means it will be relatively easy for major recording labels and film/movie distributors to expose their records or work in this established, wide and popular media coverage because of advertising money.

Why this is such as problem, is because a majority of songwriters and even independent small publishers like me cannot even have access nor have slim chances of having the recorded material aired/performed in those stations. This is not a problem for me because I know a lot of ways to promote music but it can be a problem for the majority of the independent songwriters/publishers, who have song performances far beyond what ASCAP/BMI and SESAC can monitor. For example, small and independent publishers like me have small number of clients in a relatively remote environment being very far from the mainstream. It means that even if a song performance occurs, it may not be counted by ASCAP/BMI and SESAC. This situation is possible, for example if someone licenses the song to include it in a short indie movie, which will be aired or released in a small city or country. Or even producers who work themselves and do not even submit cue sheets. If a song performance cannot be monitored, it means the associated writers/publishers cannot be paid.

There are a lot of similar situations like the one above, not only applicable to indie films. Sometimes, when your song gets cut to be included in a workout video, which will be publicly daily performed in a city workout gym for 60 straight years. And gym owners might get money, but the performance societies might be completely clueless about these song performance activities. What's the role of the music consumer? The main role is to report performance records to the writer/publisher performing right societies.

People should take the initiative to submit cue sheets or file performance reports. This helps the writer/publisher to get paid, for example when a consumer is earning money using the music.

Luckily some will report, but some will not. For those that do not report and are earning money for the use of songs without paying public performance royalties, chances are that truth might someday come out, and they will have only two choices:

a. Pay the enormous amount of unpaid royalties in years of usage.
b. Close their business and go bankrupt.

This is how a cue sheet looks like:

Sample Music Cue Sheet

Series/Film Title: *Urban Skies*
Episode Title/Number: *Grape Soda (#12)*
Estimated Airdate: *1-12-99*
Program Length: *60 minutes*
Program Type: *Comedy series*

Company Name: *Urban Skies Productions*
Address: *7920 Sunset Blvd., L.A., CA 90027*
Phone: *1-800-662-4490*
Contact: *Chris Mak*
Network Station: *Showtime*

Cue #	Cue Title	Use*	Timing	Composer(s) Affiliation / %	Pulisher(s) Affiliation / %
1	*Urban Skies Theme*	*MT*	*0:16*	*Rhonda Sims (ASCAP) 100%*	*Urban Skies Music (ASCAP) 100%*
2	*Running Home*	*BI*	*0:08*	*Rhonda Sims (ASCAP) 100%*	*Urban Skies Music (ASCAP) 100%*
3	*Backwards Love*	*BI*	*0:13*	*Rhonda Sims (ASCAP) 100%*	*Urban Skies Music (ASCAP) 100%*
4	*Uptown*	*BI*	*0:09*	*Rhonda Sims (ASCAP) 100%*	*Urban Skies Music (ASCAP) 100%*
5	*Skies the Limit*	*BV*	*1:03*	*Terry Oakley (ASCAP) 33 1/3%* *Larry Joyce (PRS) 33 1/3%* *Ennio Blake (APRA) 33 1/3%*	*Terrycotta (ASCAP) 33 1/3%* *Larry Joyce Music (PRS/ASCAP) 33 1/3%* *Ennio B. Music (APRA/ASCAP) 33 1/3%*
[illegible]	[illegible]	[illegible]	[illegible]	[illegible]	[illegible]

You can read more about cue sheets here: http://www.ascap.com/playback/2005/winter/features/cuesheets.aspx

Basically, if you don't want to submit cue sheets, you've got to arrange with the song music publisher for some reasons. And if you are unable to contact the publisher, you need to inform performance right societies about the details.

How to sell your music or songs in iTunes

iTunes (*http://www.apple.com/itunes/*) is currently the leader in online music distribution in 2010 and may continue well in the future. Song downloading is currently one of the hottest and most modern ways to sell your songs online.

For an independent artist who writes his own songs and produces his own recordings, iTunes offers some ways for indie songs to be listed. I know it is almost impossible for an independent artist to get some kind of national or international distribution without a major recording label backup. If you need some kind of worldwide distribution, then iTunes is one of the best ways to distribute your music.

Below are the current procedures needed for you to submit your songs to iTunes. The process is made of two major steps. The first one deals with securing your songs and filing application; the second major step deals with submitting songs and application with iTunes partner sites like CD-Baby.
So if you are ready, here we go (the sub-steps):

Step 1: Write and produce broadcast quality song recordings.

Step 2: iTunes requires you to hold the sound recording copyrights. So it is essential to register your catalog (or the songs you are interested in submitting to iTunes) to US Copyright Office. You can file online here: *http://www.copyright.gov/eco/*

It is good to have this sorted out first, to avoid any delay with your songs submission, as well as with the legal processing done later.

Step 3: Once you have your copyright certificate or any evidence supporting your claim, it is time to apply it online (*https://itunesconnect.apple.com/WebObjects/iTunesConnect.woa/wa/apply*) as an Apple iTunes content provider.

Fill it up carefully. If you register some company or business entity, list it out. ITunes also asks for the number of songs in your catalog, and possibly other artists you are representing (if you have for example some sort of independent label).

Review all the data carefully and then submit it. It will take more than 3 weeks for them to give a reply because they will thoroughly review your application.

Step 4: If you received a positive reply, they will tell you to submit your music to independent artist partners (such as *www.cdbaby.com* for example, which partners with Apple Itunes). They might also be suggesting other sites. If you register your music with CD-Baby, they will be the ones to submit it to itunes.

Step 5: Wait until you have full confirmation from partner websites, some follow up, and filling up of forms on their partner websites. Now, you can upload your music. They will then process it and finally submit it to itunes.

Step 6: Depending on reporting methods, you can check your sales online.

Other important pages to check:
http://www.apple.com/itunes/content-providers/
http://members.cdbaby.com/

2 - Music Recording & Mastering

2.1 General Tips

Essential record producer and engineer skills

So you want to be a mixing engineer, mastering engineer or producer at home?
As the years go by, I have finally learned the technique to record, mix and master a song production properly using my own personal computer. It may require trials and errors on your part but I strongly suggest not giving up and keeping learning.

The pre-requisites for learning are the following:

a - Fast Windows PC, I recommend Pentium 4 or higher.
b - Lots of RAM (at least 2 GB).
c - PCI soundcard.
d - Lots of free disk space (at least 120 GB).
e - Mixer.
f - Headphones.
g - Digital audio workstation (also called digital audio recording software).
h - High end computer audio monitors (speakers), 2.1 recommended.
i - CD writer.

And below is the most important skills checklist, to develop or acquire as a recording professional (with assignments, whether it belongs to mixing, mastering engineer or record producer):

1. Skill to record sound in a computer. *-recording/mixing engineer*
2. Skill to remove noise from the recordings. *-recording/mixing engineer*
3. Skill to amplify and normalize an audio wave. *-recording/mixing engineer*
4. Ability to master the preferred digital audio workstation. *-recording/mixing engineer*
5. Skill to conduct a multi-track recording. *-recording/mixing engineer*
6. Skill to realistically pan instruments in the stereo field. *-recording/mixing engineer*
7. Skill to mix instruments in their optimal frequency location to prevent mud. *-recording/mixing engineer*
8. Skill to put proper reverbs or ambience in the recording. *-recording/mixing engineer*
9. Skill to properly use the compressor to optimize dynamics. *-recording/mixing engineer*
10. Skill to master the use of effects to get a professional recording. *-recording/mixing engineer*
11. Skill to optimize mixing volumes prior to mix-down. *-recording/mixing engineer*
12. Skill in dithering. *-mixing/mastering engineer*
13. Skill to detect out of tune instruments. *-mixing/master.eng./record prod.*
14. Ability to adjust a mixing console to get the targeted sound (by listening to some reference). *-recording/mixing engineer*
15. Ability to follow the commercial radio mixing trend. *-mixing/master.eng./record prod.*
16. Ability to detect the central frequency of the most common musical instruments. *-mixing engineer*
17. Ability to produce real drum sounds in the mix. *-mixing engineer*
18. Skill to initiate the mastering session. *-mastering engineer*
19. Skill to maximize track volumes without distortion. *-mastering engineer*
20. Skill in normalizing volumes. *-mixing/mastering engineer*
21. Skill to produce a CD master for replication. *-mastering engineer*
22. Skill in selecting the best songs for the album. *-record producer*
23. Ability to detect potential hit songs before releasing the songs. *-record producer*
24. Ability to manage the entire recording session (from mixing to mastering). *-record producer*
25. Ability to arrange the song parts to get the best results after song production. *-record producer*
26. Ability to produce successful recordings. *-record producer*
27. Ability to select the best crew for the recording session. *-record producer*
28. Ability to select the best artist to represent the material. *-record producer*
29. Ability to hear what is commercially good and what is crap. *-mixing/master.eng./record prod.*

So many skills, having them all is a blessing. Read this guide and I will share my experiences to develop all the necessary abilities and skills to successfully record, mix and produce a song.

13 important factors in computer audio recording

As a recording professional, I've been using and testing various gear. In my experience, the quality of your audio recording output will be affected by the following components:

1 - The quality of your soundcard.
2 - The quality of your audio mixer.
3 - The speed and size of the hard disk.
4 - The speed of your CPU.
5 - The amount and speed of RAM.
6 - The clarity and response of your studio monitors.
7 - The noise coming from the connections (wires, …).
8 - The quality of your DAW (*Recording software* or *Digital audio workstation*).
9 - The quality of your motherboard.
10 - The quality of your headphones.
11 - The optimization of your operating system.
12 - The acoustics of your room.
13 - The quality of your CD-drive.

Let's discuss these one by one:

1. **The quality of your soundcard** – I tried to use an onboard soundcard from ASUS-P4P800X. It was a Soundmax soundcard. Well, I can say it's a decent onboard soundcard, but the quality is far from being enough. Although it helped me produce 7 songs from my catalog, but it took me a very hard time to record because of the following weaknesses:
   a. Extreme noise
   b. No amplification of signals inputted to the computer
   c. No proper grounding, I got electrocuted every time I touched the ground surface and the bass/guitar strings
   d. Cannot reproduce bass frequencies very well

   'Extreme noise' can be handled by a great DAW, using the Noise Reduction feature. '*No proper grounding'* can be solved by placing your feet on an insulator.

   But the most serious problems are: '*No amplification of signals'* especially when recording bass, and *'cannot reproduce bass frequencies very well'*. And these seriously affect the bass tracks of the song. Still, I managed to reduce the problems that were appearing in the mixing process. But this type of approach is not a recommended technique for an audio recording/mixing beginner. I'm pretty sure it would produce crappy results.

   Until now, those problems are solved since I bought a Creative Audigy Value soundcard, a great upgrade for my Soundmax onboard soundcard. But some serious problems still can't be fixed by that soundcard alone, because the amplification feature of that soundcard still isn't very strong.

2. **The quality of your audio mixer** - Mixers are made to accept input signals for processing and output mixed signals to other audio components such as the soundcard of the personal computer.

   Well, the Creative Audigy amplification feature cannot handle passive bass guitars signal levels, because the amplitude is still too low. That's why I bought a Behringer Xenyx 502 mixer. It is very handy and branded. Results are wonderful, and I can now record great bass sounds, with strong signals and a fat sound.

3. **The speed and size of the hard disk** - In the audio recording process, the sounds coming from the guitar/any instrument are analog in nature. Hard disks cannot store analog data, and the job of the soundcard is to convert analog to digital signals, which are then stored in the hard disk.

The CD quality is 16 bit/44.1 KHz, although it is recommended to record at a much higher sampling rate. The sound file type is WAV, which is uncompressed and really heavy, and thus requires a lot of disk space. The sampling rate is also very important to consider when selecting the size of your hard disk. In my experience, a 160 GB hard disk is very efficient when I record at 16 bit/44.1 KHz, because file sizes still aren't big compared to the size of the disk. The quality is fine, but If you have a bigger hard disk, you should probably record at a much higher resolution, like 24 bit/96.0 KHz, and this would then give a bigger WAV file.

The speed of the hard disk is important during recording, because when the soundcard performs the analog to digital conversion, the disk will have to store the data. Its rotating speed is measured in revolutions per minute (RPM). Slow rotating disks can have problems with recording, so I suggest selecting a faster hard disk (such as 7200 RPM range).
I currently use 2 disk drives in my PC: Western Digital 160 GB 7200 RPM hard drives.

4. **The speed of your CPU** - this is very important because the CPU is the brain of the computer. The faster it is, the more information is processed.

The CPU commands the transfer of data from soundcard to hard disk, when using what we call PIO (Programmed input/output) mode. 'PIO mode' stresses the CPU so much, that it is not a recommended data transfer mode (the transfer of information coming from the soundcard (in digital form) has to pass through the CPU for checking, and then only the CPU will transfer data to the hard disk).
The most recommended mode is DMA (Direct memory access). All data transfers from the hard disk won't pass through the CPU for checking. Data will directly go from the soundcard to the hard disk. This is optimal during recording, where analog is being converted to digital.
The quickest way to check if PIO mode is activated is to listen to the track when audio is being played back. Even with a high processor, there will be distorted clicks on the audio. This is a sign that PIO is activated. That's why it isn't recommended for audio recording, because of problems such as latency and distortion.
Of course, it's highly recommended to choose a fast processor.

5. **The amount and speed of RAM** - RAM stands for Random-Access-Memory, one of the most important components in a computer system.

Its main role in computer audio recording is like a fridge. Without fridges at home, every time we would want to eat something, we'd have to go to the supermarket first. Going to the supermarket, then coming back to our home every time is a long run. Instead, having a fridge saves time because you just buy things once, and then store them in it. That way, the next time we're hungry, we will not need to go to the supermarket. We'll just open our fridge.

RAM works just like that in a computer system. In audio recording, chunks of digital audio data are being routed from hard disk to CPU. Instead of queering hard disk all the time, RAM stores the data temporarily, for faster data transfer. So in an audio recording computer, big amount of RAM is much needed.

6. **The quality and response of your studio monitors** - this is very important because studio monitors are gateways to your ears, they are your messengers. In a real life scenario, supposing you are in a battle and have a messenger giving you false information, you will surely not win the fight. If you have a studio monitor that cannot reproduce bass sounds (no sub-woofer for example), then you are in a big trouble. I highly recommend, whichever type of studio monitor you're using, to make sure:

a - It has wide and flat frequency response (around 20Hz to 20,000 Hz)
b - It is a stereo sound system (2.1 is recommended for audio mixing, more than that is just confusing)
c - It has a subwoofer dedicated to sub basses.
d - Medium to high power amplifier can be played loud without distortion or cracking.

One of the best affordable studio monitors for computer audio recording is the M-Audio Studiophile AV 40 powered speaker:

7. **The noise coming from the connections** - there is no perfect recording system, they all have noise. Noise is unwanted in recording, because it clearly affects your wanted signal. Noise can be further classified as “hum” (low frequency noise, such as 60Hz hum from power lines) or “hiss” (high frequency noise).

   Although all systems have noise, in computer audio recording it is highly suggested to keep it at minimum. Most home recording studios do not have expensive systems to control the noise. The best way to control noise is controlling the connections. Connections are forms of wires/transformers that are interconnected with each other. For maximum power transfer condition, these connectors must be matched. And it is highly suggested to use shorter wires, as signals will get lost as the wire gets longer. Buy a decent and clean rust free connector. Rust can increase the heat resistance, which adds some noise in audio.

8. **The quality of DAW (Digital Audio Workstation)** - this is important. A quality DAW can offer wide recording features to maximize the recording quality. Examples are noise reduction features, effects and plug-ins.

   Cheap DAW, especially freewares, cannot help to improve your recording and mixing. There are many high quality DAW, like Adobe Audition (the one I’ve been using for more than 6 years), Pro Tools from Digidesign (industry standard), Cubase (also a good one) , Sony Sound Forge (available with video editing features if needed). Those are the ones I’d recommend if you want to select a DAW. I highly suggest not shifting DAW all the time. It is good to master one of them, to maximize recording quality. I believe quality recordings are a function of one’s ability to handle his DAW. This means that a Cubase expert can produce better recordings than an amateur Pro Tools user. Mastery of your selected digital audio software is highly important in computer audio recording.

9. **The quality of your motherboard** - Powerful motherboards offer many RAM slots, PCI slots, faster connections (I mean busses). It’s like having your components in a very wide highway. You can drive a lot faster in a wide highway without traffic, than in a small alley. The same principle applies with expensive motherboards.

   At the time (2004), I was using a top of the class motherboard. It was an Asus P4P800-X.
   Far more expensive than any other class in 2004, but it was worth it. The P4P800-X had the best onboard soundcard that I’ve tried, and helped me to record up to 7 of my best songs, without even relying on a high-end soundcard. Now that I’ve switched to a Creative Audigy Value, I even missed the onboard soundcard sometimes. But I know that my new gear investment will be able to produce better results.

10. **The quality of your headphones** - although not very important, but you will use this to check the audio during a recording process, or to review the mix in headphones. It is not recommended to mix using headphones, but having quality headphones makes a difference in the quality of a recording.

11. **The optimization of your operating system** - this is very important, because the operating system of your computer controls the recording process. If it is not optimized, it will increase the latency of your recording, and drop-outs in your audio. In my system, I've been using Windows XP (latest 2010 release is Service Pack 3) for around 6 years, and it served me quite well since then. I made a few tweaks on my windows XP computer, and here are some tips to optimize it:

   a. Use DMA mode, not PIO mode.
      To check go to control-panel*, system*>*hardware*>*device-manager*>
      IDE/ATA controllers-Primary IDE channel-Advanced settings>Current transfer mode.
      Current transfer mode should be DMA.
   b. Defrag the computer once a month. Defragmentation process will make fragmented files closer to each other in a hard disk space, thus faster to be accessed.
   c. Optimize the background services. Audio drivers run in the background. To do this,
      'Control panel>*Device manager-system-advanced-performance-advance-processor-scheduling*>background services'.
   d. Optimize to get the best performance.
      To do this, go to 'Control panel>*device manager-system-advance-settings-visual effects*>adjust for best performance'.
      In this type of optimization. Windows XP will not display the flashy colors, animations thus helping to save it for audio usage.
   e. Optimize the virtual memory- set it to system managed.
   f. Clean files regularly to optimize hard disk space.

12. **The acoustics of your room** - of course it matters, although we cannot afford world class acoustics. But in my experience, I can learn the acoustics of my room by closely studying its frequency response. I'm doing this by listening to a professional mixed CD in my room. Listen to the bass levels, treble levels, mid frequencies. How it is responding to your ear is important. Then go to your mix and match to that level.

    By knowing the acoustics of your room, you'll be able to create a good mix. The most important is to take time to know the acoustics of your room, by listening to professionally mixed CDs and comparing them to your mix.

13. **The quality of your CD-drive** - many would say, this is not so important. Yes it is. But take note that, during a mastering process, when a mastered track is burned on an audio CD, this will use the CD drive. If your CD drive is not optimized for audio, then you'll encounter problems.

    To get the best results, for computer mastering/recording applications, you need a CD drive allowing a very slow burning speed, around 1x to 4x. That's because the slower your burning speed is, the higher it's accurate. The faster the burning speed is, the more it's prone to errors. That's why a master CD that is about to be submitted for replication, needs to be burned at a slow rate, to prevent digital errors during the CD replication.

2 - Music Recording & Mastering

2.2 Music Recording

2.2 Audio Recording

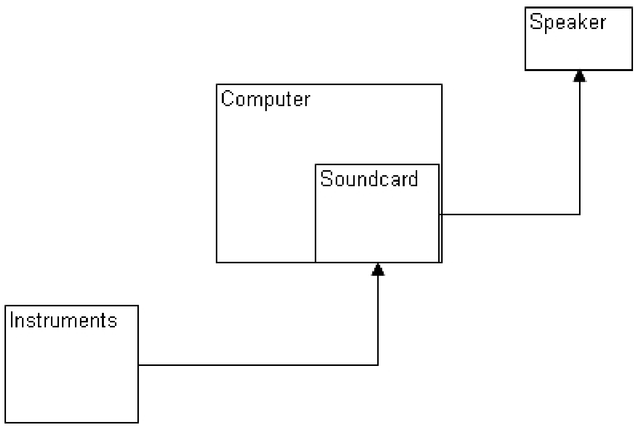
Ways to record sound using a personal computer

This is a basic guide. Before you can record sound using a personal computer, you need to have:

a - Soundcard (a decent one, at least Soundblaster Audigy)
b - Audio mixer
c - Studio speakers
d - Cables (at least 2)
e - Audio editing software

Take note there are lots of ways to record sound. I will show you different configurations and their strengths and weaknesses:

Method 1: The most basic configuration



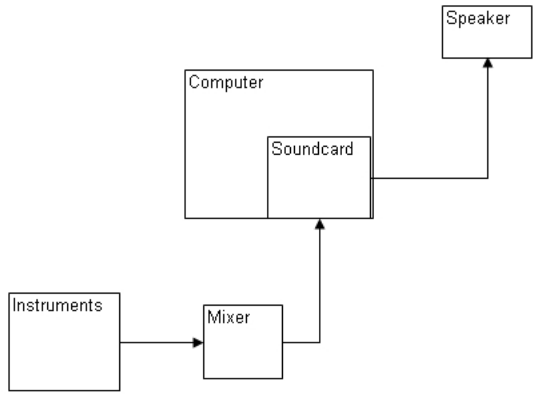
This is really basic, the instruments (could be anything) are plugged into the microphone/line-in input of the soundcard. The audio recording software in the personal computer takes the recording process in charge. And then the soundcard output is connected to the computer speakers for monitoring.

The advantage of this method is that it is very easy to setup. The disadvantages are:

a. Very noisy signal.
b. Low amplitude or really weak signal.
c. Poor frequency response.

This is only recommended to record voice signals such as an amateur voice tape recording, but this is not recommended for professional recording.

Method 2: Using a mixer on the instruments side:



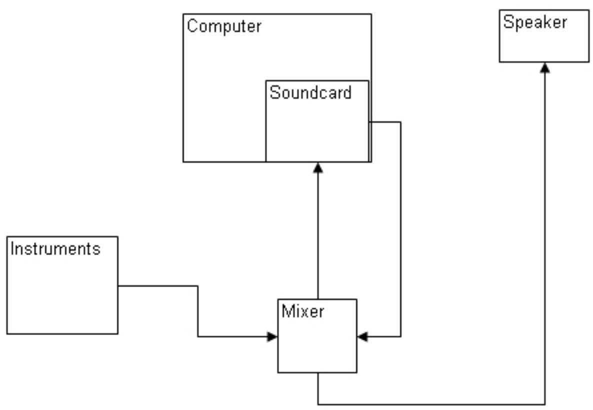
Instruments can be connected first to the studio mixer, allowing that way the signal to be amplified, and the noise to be removed.

This is recommended for studio recording because

a. signals are now strong enough to be recorded, and less noisy.
b. since an audio mixer is being used, signals will have an excellent frequency response, which is very important for professional recording.

This method is great but is not as efficient as the next one.

Method 3: Using a mixer to route all signals coming to and from the computer.



This is the most efficient one, because all volume controls are on the mixer. In this setup, the instruments are plugged to the mixer. The mixer then conditions the signals by using its pre-amp features and some EQ. Then it is finally going to the computer audio recording software. At the same time, the audio recording software feeds out a signal to the soundcard, which is then connected to another input in the mixer to control the volume. The mixer output is then connected to the studio monitors/speakers to hear the signals being played.

This is a great setup during multi-track recording, and this is the one I am using in my home studio.

Try to experiment other types of setups that fit your budget and your recording needs. Anyway, the third method is the most recommended one, if you are into professional audio CD production.

Recording bit rate quality in home recording studio

One of the best recording advices I could give to newbie in home audio recording and music production is to record at the highest bit rate as possible. If you are confused, ok this is an explanation in bulleted format:

1 - CD Audio standard is using 16 bits 44.1 KHz.
2 - Typically for better quality of recorded audio, it needs a high dynamic range for your music to flow.
3 - This high dynamic range can be obtained at a much higher bit rate (higher than 16 bits) which is 24 bit or 32 bits.
4 - Think of dynamic range as a size of your painting canvass. If a painting canvass size is limited; of course the resolution is limited also and you cannot clearly see the visual image of the painting at a farther distance.
5 - With big dynamic range available in higher recording bit rates, it is analogous to a large canvass painting which you can clearly see even the small details. This increased in resolution increases the ambiance, clarity and volume potential of the recorded audio. In music production, this is also offers a comfortable audio working environment since lots of audio details are available. This is recommended in mixing and mastering, where detailed tweaks to the audio signal are done.

Now understand the need to record at a much higher bit rate. So what is the standard?
It all depends on the following:

1 - **Sound card** – You need to ensure that your soundcard is capable of recording at a much higher than CD bit rate. For example, the industry standard has been 24 bits.
Quality inexpensive sound card can do this job, for example Audiophile 2496, which can record at 24 bits, 96 KHz. Of course, it is not possible to get this kind of recording bit rates using onboard sound card or other types of sound cards not designed for professional music production.

2 - **Recording software** – You need to make sure that your recording software can record at a much higher resolution than CD audio, to take advantage of increased audio resolution.
Pro tools for example is capable of processing 24 bit audio. Adobe Audition is using 32 bit float, but is essentially a 24 bit audio processing engine internally.

So how are you going to record 24 or 32 bits? If you are using Adobe Audition 1.5 (similar procedure can be found using other digital audio workstation), you can configure it as follows:

Step1: Install a high end soundcard capable of recording high resolution audio.
Step2: Launched Adobe Audition.
Step3: Go to multi-track view tab.
Step4: Go to "Options" – "Settings" – Multi-track
Step5: Make sure that the "Track Record" is set to 32 bit
Step6: It is important also to mix down to 32 or 24 bit, so make sure that mix down is set to 32 bit audio.

Actually, for best audio quality that you can get from your music production, you need to ensure that:
1 - You are recording either 24bit or 32 bit.
2 - Mixdown to 24 bit or 32 bit.
3 - Mastering at 24 bit or 32 bit.

Finally once your work has been completed, you can then dither 24bit or 32bit audio wav to 16 bit CD audio in preparation for your CD audio replication/duplication process. This is done in the mastering stage of your music production.

How to record acoustic guitar and electric guitar in your home studio

If you are looking for some great ways to record your acoustic and electric guitar tracks in your home studio (as part of your music production projects) then this tutorial will be useful. Below are the common technical problems when recording guitars at home using your personal computer as your DAW (Digital audio workstation):

"**High background noise level**" are often caused by using an onboard (motherboard) soundcard when recording as well not using a mixer/pre-amplifier that will clean up the signals.

If you directly plug your guitar to your onboard sound card and you are not using any audio mixing hardware then definitely you will get a very high background noise level. Even if you are using a high quality soundcard, it is still not advisable to plug your guitar directly to the soundcard input jacks because it can still get substantial noise which can affect guitar recording quality.

"**Out of tune guitars**" are also caused by not using a digital guitar tuner.

Most guitarists are not used to it especially the new ones and they tend to record their performance without double checking the guitar tuning. As a result they might hear a lot of complaints about out of tuned guitars.

You can buy a digital guitar tuner such as Matrix Digital Chromatic Tuner SR1060 for around $20 dollars at Amazon. So you need to re-confirm your guitar string tuning first before you do any recordings. Ear tuning is good as a start but you must re-confirm it with a digital tuner for best results.

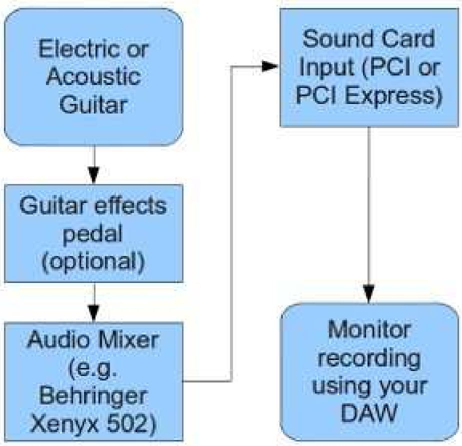
"**Very low recording level or volume**" is caused by not using a pre-amplifier/audio mixing hardware for your guitar signals.

If you will directly plug your guitar to the soundcard inputs, even if you max out the volume of the soundcard, you cannot still get the best recording volume because your guitar sound needs to be amplified before it will enter your sound card. This will get worse if your signal is already very noisy.

Therefore to get the best recording results of your guitar at your personal home studio, you must have the following:

a - Decent/high quality sound card (do not use onboard ones)
b - A digital guitar tuner
c - A personal audio mixing hardware (like Behringer Xenyx 502 will do).

Please read the guide about recommended mixing hardware and other components. So now you have all those requirements how you will be able to connect your guitar to your personal computer? Refer to the signal path below:



This is called a "direct line input type" of recording guitars. In big commercial studios, you can record guitars directly from a guitar amplifier using microphones, but you most likely not doing this in your personal home studio.

Recording distorted guitar: Tips and techniques

Distorted guitar is one of the most important recording production instruments for rock music. Wide genres will use distorted guitar in the production process, for example in pop, alternative, grunge, punk, etc... Because of this popularity, this short guide has been written to illustrate this important process.

This is written for someone who will use a personal computer (preferably Windows), a recording software (any software will do) and a decent recording soundcard plus mixer in recording guitar tracks. This will illustrate only direct line input recording process of recording distorted guitars. In this method, guitar is connected to a distortion pedal and then the pedal output is then connected to mixer. Finally the mixer output is then connected to the sound card line input. See illustration below:

The purpose of distortion pedal is to provide the desired distorted sound effect of the electric guitar sound. This may vary according to the recording producer preferences. The output of the distortion pedal is weak and the mixer will provide some amplification and cleansing of the distorted guitar sound signals. If a mixer is not used, the guitar sound output will be very weak and noisy, which is not recommended.

Below are the steps employed in order to successfully record a distorted guitar into your personal computer:

Step 1: Tune your guitar. It is highly recommended to use a digital tuner for accuracy.

Step 2: Once the guitar has been tune, consult with your recording producer to finalize the desired output of the distortion pedal. Different settings of the distortion pedal produce widely different distorted sounds. So it is useful if settings are finalized first before recording it to your computer in order to save time. Do not connect this yet to your computer, instead use your personal guitar amplifier to monitor the sound.
The guitarist should finalize the settings as per consultation with the producer.

Step 3: Once they have finally decided on the final sound of the distorted guitar. It can now be recorded to your computer. First switch on your mixer and connect the distortion pedal output to your mixer line input. Then connect the mixer output to your sound card input. Using your recording software, do a test whether the captured guitar sound is clean.

Step 4: Once the test recording has been setup, do a final take by hitting your software recording button.

Another way to record electric guitar is by using guitar amplifiers. In this method, a microphone is used to capture the sound output of guitar amplifier which is then connected to a mixer and finally to the sound card.

Recording bass guitar: Tips and techniques

Bass guitar is one of the most important pieces of instrument to establish some "heaviness" in rock, punk and alternative music. Without a bass, music can hardly be considered as "heavy rock".

This short guide is for beginners looking for ways to record their bass guitar in their personal home studio consisting of the following gears/equipments:

a - Personal Computer with Sound card
b - DAW (Cakewalk, Audition, Sonar, etc)
c - Audio Mixing hardware or Pre-amps
d - Passive or Active Bass guitar
e - Audio Cables
f - Bass guitar amplifiers
g - Microphones

There are basically two ways to record bass guitar:

a) *Using a direct line method*

In this method, you will not be using a bass guitar amplifier in the process but you will connect your bass guitar according to the following signal chain:

Bass guitar ===> Bass guitar pedal (optional) ==> Pre-amplifier or Mixer ===> Sound card line input ====> Personal Computer DAW to monitor recording

Discussion:

You need to connect your bass guitar first to your pedal but this is optional, if you need a clean bass guitar recording you need to plug it directly to a pre-amplifier or a mixer. The purpose is to boost the very weak bass guitar signals especially if you are using a passive bass guitar.
Then connect the main output of the mixer/pre-amplifier to the sound card line in input. Using your DAW, you will need to monitor the level of the bass guitar recording.

If the received signal is too weak; try adjusting the volume of your bass guitar, if this is still not enough, turn on the volume knob of your pre-amplifier or mixer. Sometimes turning the volume is not necessary because most mixers have gain control. So in this case, just use the gain control (also known as TRIM). Adjust it carefully, as increasing the gain also increases the overall noise.
Do these tweaks until you have a strong recorded, minimal noise and quality bass recorded as monitored by your DAW. You can then start recording at this stage by hitting the record button.

b) *Using bass guitar amplifier*

If you have a quality bass amplifier, you can use it for recording bass using the signal chain below:

Bass guitar ==> Bass guitar amplifier ====> Microphones ====> Mixer/Pre-amp ===> Sound card line input

As a result, you do not need to apply any effects after recording, since the recorded bass already includes some space/reverberation in it.

Feel free to experiment which sounds best between these two types. Personally I use the direct line method because I have limited room space. But if you have a well designed acoustics and to take advantage of the room ambience, you can use an amplifier to capture its benefits in your recording.

Home vocal recording tips

This is a short guide on how to do home vocal recording using the least amount of resources without compromising vocal recording quality.

What you need at a minimum:

1 - Personal Computer with Recording Software
2 - High End Sound card, read the tips on how to Select Soundcard for Home Recording Studio
3 - Medium quality microphone: Shure SM58 Dynamic Microphone will do.
4 - Medium home studio mixer (for example Behringer 502 5-Input Mixer).
5 - Microphone cables and Microphone stand (very important)
6 - Headphones

In order to minimize background noise when recording vocals, it is important to:

a) Record at night when everything is quite (midnight is good).

b) Turn off electric fans, air-conditioning unit, etc.

c) When you have a budget but not necessarily, install a vocal booth in your home studio. You can build your own $30 vocal booth. Vocal booth is a must if your studio is beside the road where it is impossible to have a quite environment. If you need to buy a professional vocal booth, you can have it around 999 bucks: Auralex Vocal Booth.

Configure your vocal recording gears to your computer according to this:

1 - Connect the microphone for example (SM 58) to the mixer.
2 - Connect mixer to the sound card line input.
3 - Connect headphone to the sound card line output.

Test your configuration by:

a) Adjusting mixer volume, sound card line input as well as the recording software volume to arrive at the lowest noise recording level and loud vocal recording signal.
b) Put some allowance, it is not always good to record at the maximum volume as it is prone to clipping which is bad.

Doing actual vocal recording session:

1 - Let the singer wear the headphones (dubbing/minor vocal recording like backup may not require headphones, but it is recommended)
2 - Let the singer be comfortable with the play volume. Too loud can be too distracting for the singer to sing. Too low headphone monitoring volume can reduce perception of background music which is not good for the singer.
3 - Do a test run; make sure you have a good recorded signal.
4 - Finally, hit the record button and do the final take.

2.2 Audio Correction

How to maximize volume in audio recording

There are lots of ways to increase volume in audio recording, take note of the following before you increase volume :

a. Maximizing volume is only done in the mastering process.
b. Maximizing volume in the mixing process is not advisable.
c. It doesn't only mean reaching maximum amplitude, but also having a well balance mix.

You need to make it loud? In my experience, before you make it loud, make it sound pleasant first. This can be done by properly mixing those tracks and keeping it well balanced.
As a rule, it is best to have maximum amplitude of around -3 dB to -0.5dB during your mix down. This means that after mixing, there should be no peak louder than -0.5dB. The basic reason is to give some space during the mastering process.

The mastering process involves the following important processes:

a. Track trimming and Noise reduction
b. Equalization
c. Compression

Those 3 major processes affect the quality of your overall tracks particularly equalization and compression process.

Compression is the process that sets the volume loud. Aside from compression, normalization also makes the sound loud. Normalization is different from compression in a sense that compression makes the track silent parts loud, while normalization does not.

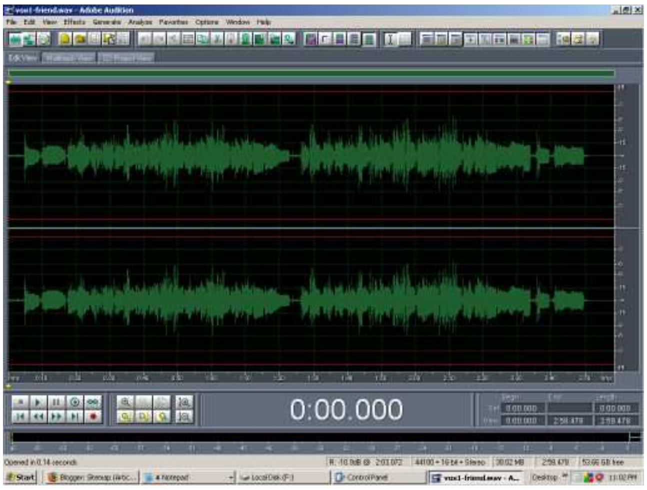
Normalization is a fairly easy process; the overall goal of normalization is to attain maximum track volume without compression. This preserves maximum sound quality compared to compression because of the absence of distortion. Compression can distort the tracks if overdone, and tends to lose track dynamics.
Normalization is a classic technique to make sounds loud and was commonly applied during 60's and 70's. Have you noticed that when you compare Led Zeppelin tracks and Green Day tracks? The Led Zeppelin tracks tend to sound less loud than the Green Day tracks, but sound cleaner and clearer. That's because Led Zeppelin mastering engineers used less compression and relied more on normalization.

The rest of the techniques illustrated below best apply to mastering process. Although you can individually boost each track during mixing, only normalization is recommended not compression. Compression is done better during mastering process.

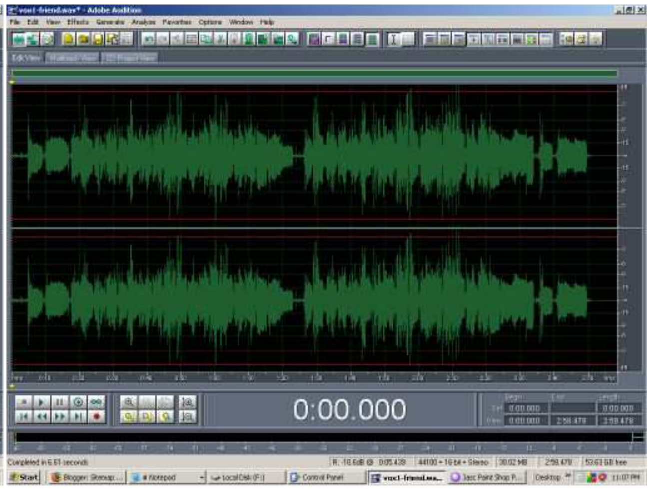
To do normalization in Adobe Audition 1.5:

1 - Using Edit view, highlight the parts of the audio wave you want to normalize.
2 - Go to effects—->amplitude
3 - Then in constant amplification tab, Find “*calculate normalization value*” then click “*calculate now*”
This will give the amount of boost in dB needed to reach maximum amplitude of 0dB.
4 - Then click OK. It will then normalize the tracks to 0dB.

See screenshot before normalization :

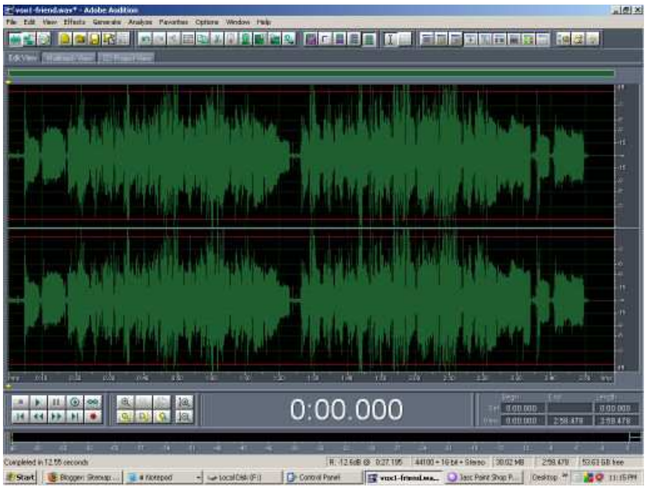


See screenshot after normalization:



See the difference??? Normalization only amplifies the whole wave in such a way no peak will be louder than 0dB.
In other words it will not sound as loud as using compression but it sounds absolutely clean.

Below is a screen shot of the same wave using compression :
Tools/Settings: Waves L2 Plug in, -8dB threshold, Out of ceiling= 0dB



By looking at the wave, this maybe the loud sound that you like,
but it may distorted at some point in the wave.

How to prevent phase cancellation in audio mixing & mastering

I received an inquiry from Ivan relating to "phase cancellation" problem in audio recording:

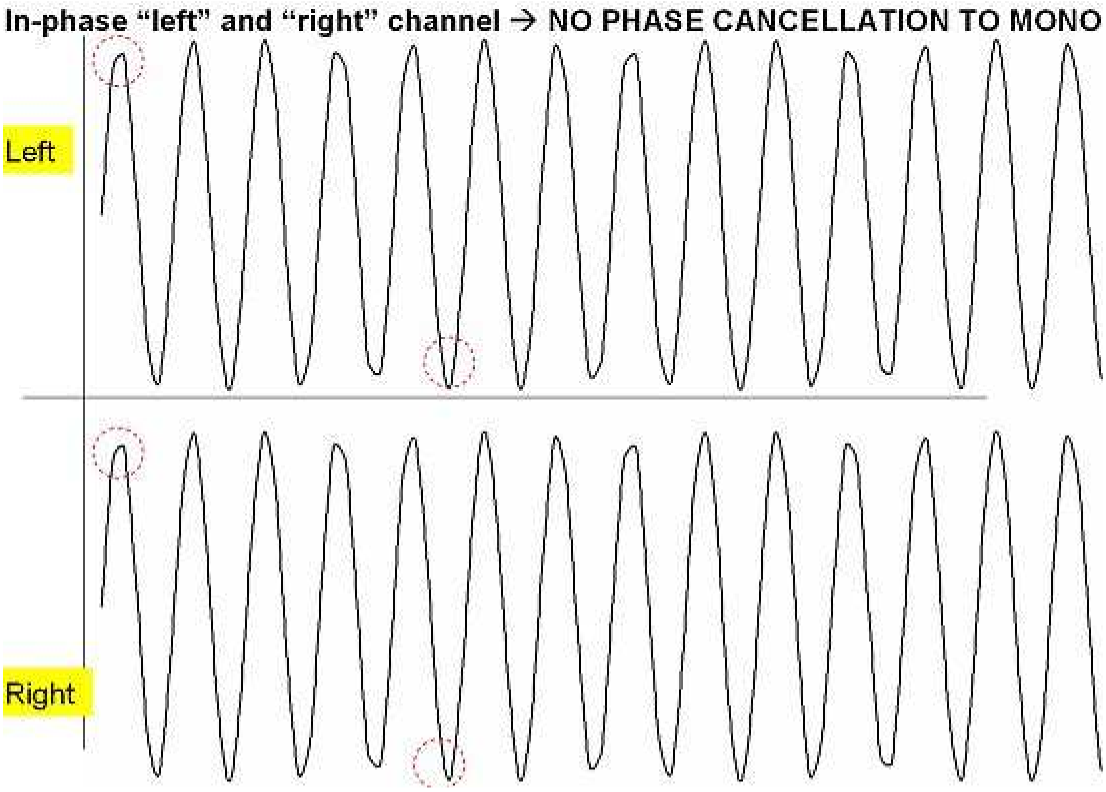
From Ivan:

Can you please explain what is a "phase cancellation" problem?
I think I have this problem in a song. When I check my mix in mono, my guitar tracks (left and right) disappear. Is this phase cancellation ?
If so, how can I fix that?
Thanks

Emerson reply:

Phase cancellation is a serious issue in recording/mixing/mastering. Your problem is indeed phase cancellation. It is because your left and right guitars are "out of phase" with each other (like 180 degrees). You will NOT notice this problem when played in stereo but the problem will seriously affect when your audio signal is converted to mono.

An out of phase can be best explained graphically, supposing you have a stereo sound wave below:

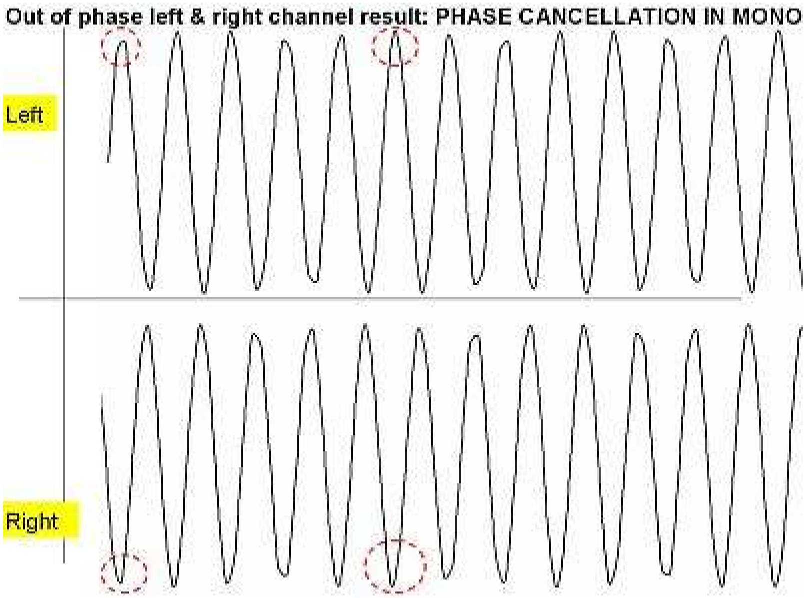


If the top peaks on the left channel correspond exactly or even "near" perfectly with the top peaks on the right channel, and then you can say that the audio stereo sound file is "in-phase".

If you convert a stereo to mono, the left and right channels are "sum up", so if they are in-phrase you can basically sum their amplitudes and there is no cancellation due to mismatch.

Example: (+ 2dB left channel) + (+2dB right channel) = 4dB summation, so there is no weakening of audio results.
A "+" symbol indicates the phase. Since both right and left channel are positive, then they are in-phase with each other such as shown in the above screenshot.

What would happen if the left and right channels of the stereo were “out of phase”?



The top peaks of the left channel corresponds to bottom on the right, they are now “out of phase”. Of course, this sounds normal in stereo. When the signal reverses polarity, the signs will change so a signal of -5dB will become 5dB and vice versa.

If out of phrase stereo is converted to mono: +2dB (left channel) + (-2dB right channel) ~ 0dB or almost no volume when played in mono because the wave are “cancelled”. This describes the “phase cancellation” problem. In actual scenario, they are not perfectly out of phase, so you can still hear mono signals but they are VERY weak unlike how they should sound in stereo.

How to prevent this?

1 - Do not use long delays between left and right channels. Chances are, these long delays can cause phase related issues. Be careful with the effects you use, make sure they are not introducing phase related problems.
2 - Check - with your audio recording software - the phase between the left and right signals. You can zoom in to make sure the left and right channels and matched / in phase.
3 - Check your speaker wiring. Make sure that left and right wires are in-phase with each other. Refer to your studio monitor manual.
4 - Always check your mix in mono. You can basically convert stereo to mono and then play it to spot some phase cancellation issues.
5 - Mike properly and monitor the phase of the resulting wave. You might be tempted to use 2 mikes simultaneously to record a guitar track (one using DI for the left channel and one using microphone to get sound from the guitar amplifier – recorded to right channel).

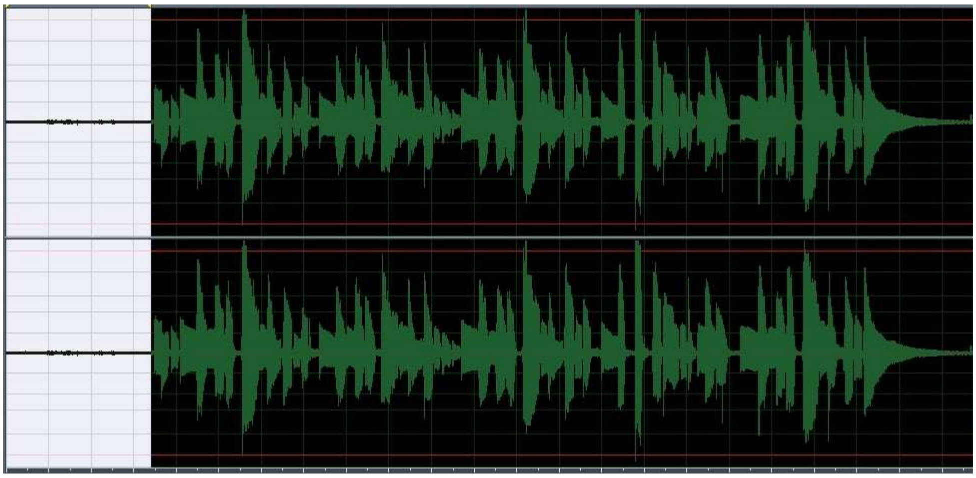
Delays might be good for a great stereo sound (delays are nice). But be careful, these delays can cause phase mismatch and introduce phase cancellation problems resulting to poor mono mix version of your audio track.

Remove noise in recordings using 4 easy steps

Noise is unwanted in recordings; in fact it does exist and should be removed. Noise in recordings can surely ruin the quality of the mix and decrease the professionalism of the work.
This guide covers the noise removal procedure using Adobe Audition 1.5 using Edit View (if you are doing multi-track, clicking those tracks in detail brings you to the edit view of the waveform per track)

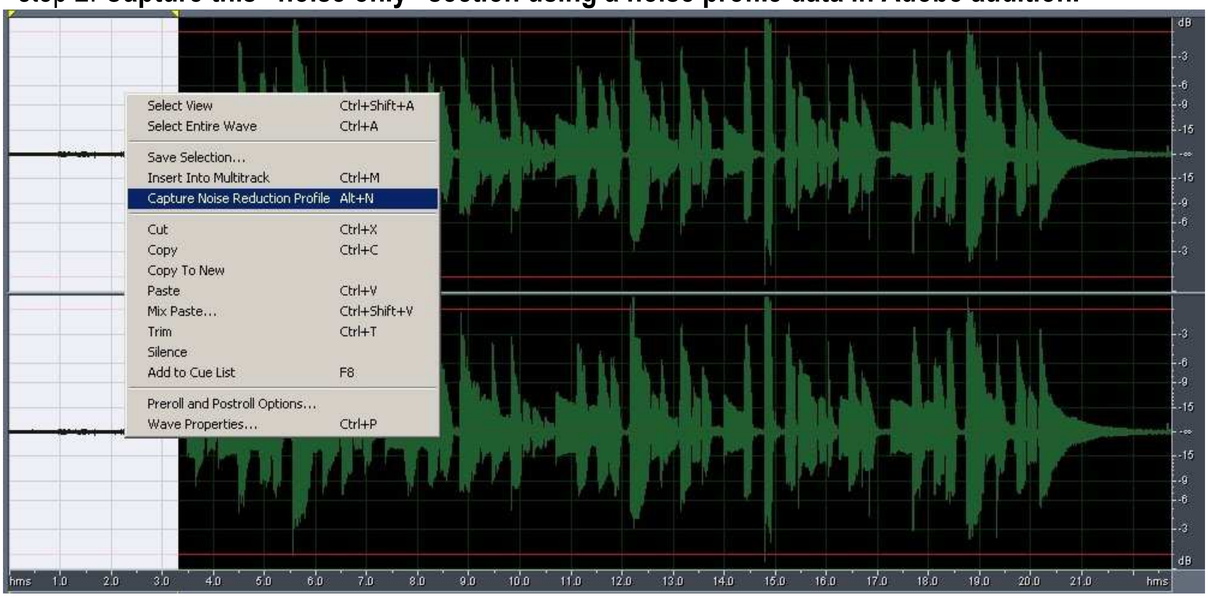
step 1: **Identify the portions of the recording where noise is the only signal.**

Looking below, I have planned to have at least 1 to 3 idle seconds before I start recording , the purpose is to have portions only for the noise. You can see below that the noise is being highlighted in the beginning portions.



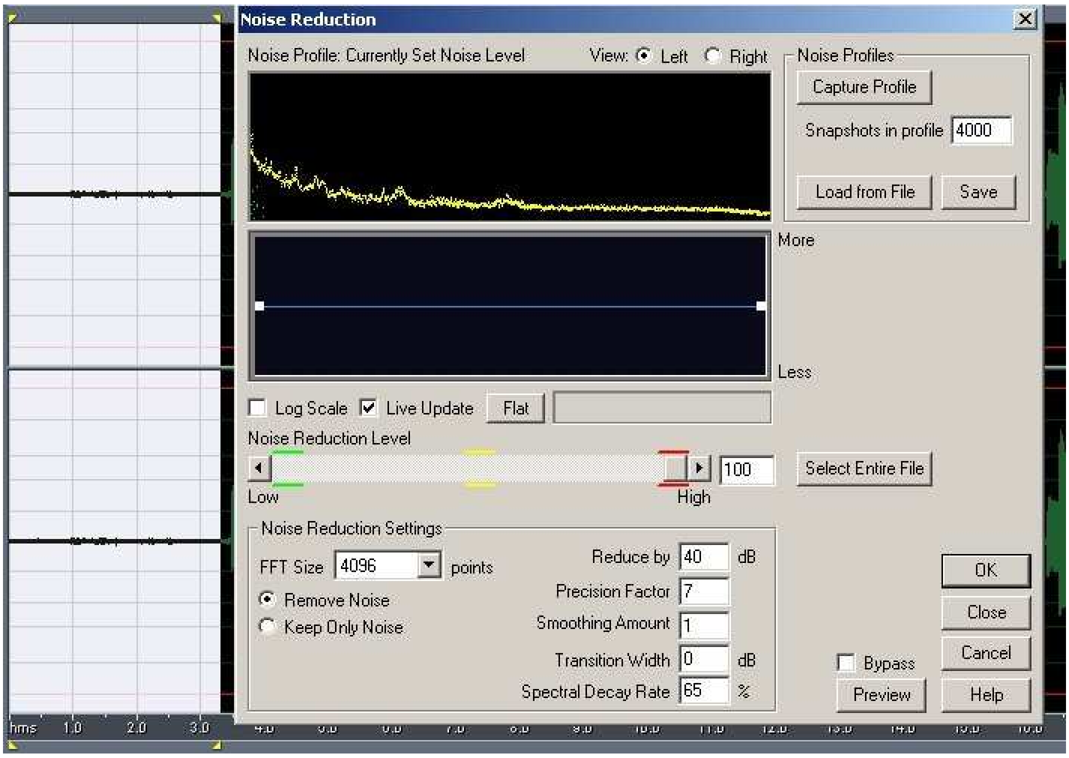
You can hear the mp3 equivalent of this “noise + recorded guitar track”:
***CLIP - With noise**: To listen to the audio clip, please go to “audiorecording.me” website, then >Recording Tips>”Remove Noise in Recordings Using 4 Easy Steps”*

step 2: **Capture this “noise only” section using a noise profile data in Adobe audition.**



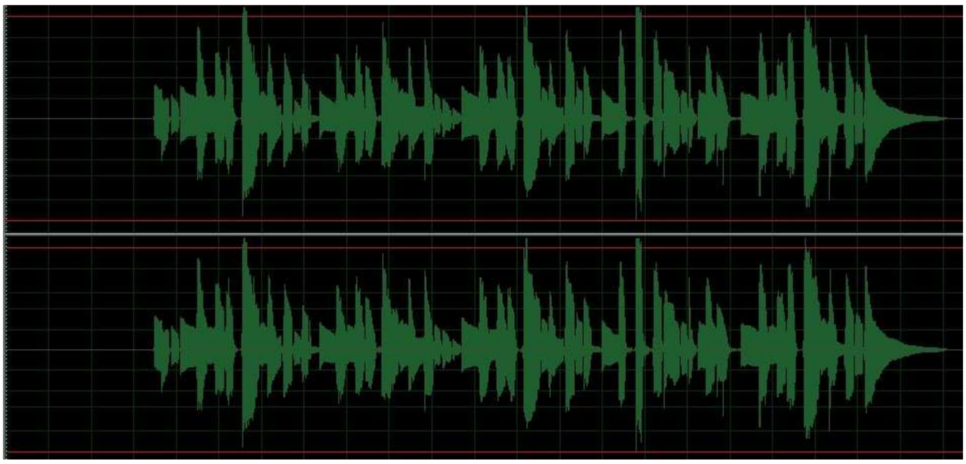
step 3: **Go to Noise Reduction feature in Adobe Audition 1.5**

You should see a very similar window as shown below. You can use my noise reduction settings, it works fine. Take note that in the graph, the green dotted below the yellow lines are the noise signals. It is very important to have a profile that consist of "noise only" signals, to avoid degradation of the true music signal that is being recorded.



step 4: **In the Noise reduction menu, click "Select Entire File", then click OK.**

This will apply all the settings and processing on the entire wave signals, including the true musical signals that is being recorded. After this noise reduction processing, you can now see that the noise on the beginning portions has been removed.



Listen the guitar track without noise, as you can hear, it sounds clean and pure, no hissing and humming.
***CLIP - Without noise**: "audiorecording.me" >Recording Tips>"Remove Noise in Recordings Using 4 Easy Steps"*

2 - Music Recording & Mastering

2.3 Audio Mixing

Common Tools

Audio compression: Tips for mixing
How to use a parametric equalizer
What a high pass filter does - Technical explanation
Low pass and high pass filters: Application in home studio

General Mixing Tips

How to improve home studio acoustic treatment
How to mix instrument frequencies for best sound
Easy live mixing tips: Getting the best out of a live performance

How to Mix Instruments

How to mix Guitar tracks: Beginners Guide
How to compress Lead guitar in the mix: Letters from readers
How to mix Electric guitars using "Double Tracking" technique
How to mix Baritone guitar: Tips and techniques
How to mix Bass guitar: Tips and techniques
How to mix Acoustic guitar and vocals: Simple guide
How to mix Vocals and put some professional effects
How to mix Vocals – EQ settings: Letters from readers
Rock Vocals and Guitar: Reverb settings
How to mix Piano Keyboard: Tips and techniques
How to mix Sitar: Tips and techniques

How to Mix Drums

How to mix Snare drums: Tips and techniques
How to mix Snare drums: EQ, compression and panning tips
How to make loud snare and kick drums in the mix
How to compress the snare drum and kick drum
How to mix Kick drum: EQ settings and compression
How to mix Cymbals and Hi Hats: Tips and techniques

General Panning Tips

Creating realistic stereo image with panning
Developing wide stereo ambiance sound in your recordings
How to create a sound as if it was coming from behind

How To Pan Instruments

How to pan acoustic or electric guitar in the mix
Pan Bass guitar in stereo track mixing
How to pan Drum instruments?

2.3.1 Common Tools

Audio compression tips for mixing

Professional audio mixing is basically two steps: EQ and Compression. Different instruments require different compression scenarios. To those who don't know what compression is, it's all about controlling the peaks of recorded signal.

The easiest way to understand how compression can be useful to a mix is recording vocals.
Vocalists, even professionals, have a tendency to sing very loud in some portions of the song. Now without compression, this vocal track could be very annoying. Compression sets balance by automatically compressing the signal when it reaches a certain level.

My major rule in compression is simple, compression takes away signals and sound quality. It's a fact that if it is overdone, it will drastically reduce the power of the sound. Now let me give you my tips on compressing common instruments in audio tracks.

Compressing vocals in the mix - I use the presets of Sony Wave Hammer - voice settings.
This plug-in is available in Sony Creative Sound Forge 10. If you have this installed in your computer, it can also be added to Adobe Audition or any other DAW that accepts Direct X plug-ins.
The characteristics of these compression settings are this:

Attack time- 5 ms
Release time- 50ms
Threshold- -10dB
Compression ratio – 5:1

Take note that the *attack time* is very short, because vocals are highly transient in nature. Notes of vocals are very short, so to capture effective compression, one must set it to short attack time.

The *release time* is a bit slow, because when a vocalist sings loud portions of the song, it will tend to last a longer time also. It would be odd to have very short loud portions of the vocals.

Threshold of the compressor is the level of the signal at which compression starts. For vocals, I need it to be set to -10dB, because more than -10 dB in vocals is already very dominative in mix.

The *compression ratio* is 5:1; which means a five times reduction of the signal when it reaches -10dB or above. Now that's a bit compression to lower the volume of the loud peaks.

Compressing guitar in the mix - For acoustic and clean electric guitars, I use Sony Wave Hammer guitar presets. The compression settings are:

Threshold: -20dB
Ratio: 5.0 is to 1
Attack time: 15ms
Release time: 15ms

The *threshold* is lower in volume compared to the vocals. The primary reason is that guitars need to sound a bit lower than vocals in the mix or else it would dominate the vocals. Compressing at -20dB ensures that any strong level above it will be suppressed five times to control the volume and not to be too loud.
Guitar sounds are not sharp transients in nature unlike vocals, kick and snare. They will have a sustaining and delaying sound. So medium set *attack* and *release times* are good.

Compressing bass guitar the mix - Bass guitar sounds need to be compressed to provide a steady beat backbone to the song. This is very important in modern rock and pop tracks.
That's why I use Sony Wave Hammer - bass guitar presets with the following settings:

Threshold= -20dB
Ratio: 6 is to 1
Attack time: 40ms
Release time: 80ms

The main concept of creating a big bass sound is slow *attack* and fast *release*. Since basses are not super fast transient, they first need to develop their level, then set compressor to attack the signal and release it immediately. The effect is a loud sounding bass. Compression settings are a bit higher than guitar and vocals, because basses need a more uniform sound to provide a steady beat.

Compressing kick drum in the mix - I do not compress kick drum in the mix, because I want to sound real, alive and not compressed. In my experience, compressing kick drums takes away their deep bass sound.

Compressing snare drums - Snare drums needs compression, so to compress snare I use these settings :

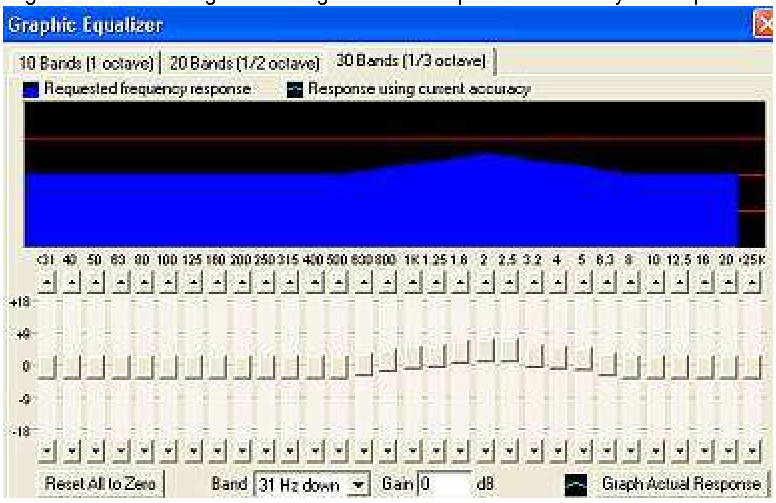
Attack: 20ms
Release: 40ms
Threshold: -12dB
Compression ratio: 5:1

Snares need to sound natural even though they are sharp transient in nature. I prefer to compress snares with a slower *attack time* and faster *release*. This will give a full snare and powerful sound. Compressing with too fast attack time can flatten a sound and will make it sound dull.

How to use a parametric equalizer

Parametric Equalizer is one of the most useful mixing tools available to any audio/recording/mixing engineer. Yet, amateurs do not fully understand the concept and operation of these parametric equalizers. As a result, they are not able to attain the sound quality they need.

Beginners in recording and mixing are often acquainted with only one equalizer which is called “Graphic Equalizer”.



While this is still an equalizer, it provides almost no use to mixing audio recording sessions, because of the following aspects:

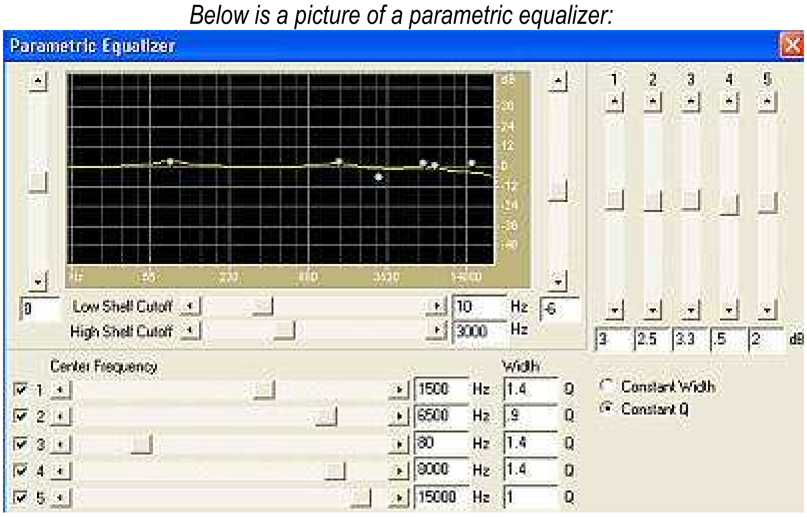
a. Graphic equalizers cannot provide Q control (in applying EQ settings to a particular range of frequencies).
b. Some graphic equalizers are NOT designed for mixing and just for hi-fi equipment monitoring purposes.

That means that the accuracy of the graphic equalizer is questionable.

Now how do we use the parametric equalizer? Before we illustrate the details, let me provide a good working definition of parametric equalizer.

Parametric equalizer is an audio frequency filtering tool that can let the engineer control the amplitude of certain range of frequencies. This control of amplitude can be done by boosting or cutting.

Below is a picture of a parametric equalizer:



While a graphic equalizer only lets the user adjust the amplitude, the parametric equalizer allows the engineer not only to control the amplitude but also to control certain frequencies as well.

The usual naming conventions used in parametric equalizers are stating the audio settings in terms of dB, center frequency Q. So for example:

> *Guitar (Left-Rhythm) = 3 KHz, Cut 6dB Q=1.4*
> It means that the engineer is adjusting the guitar left rhythm audio wave at 3000Hz and cutting 6dB with Q=1.4.

Q states how wide or narrow the cuts are (or boosted if it is a boosting settings). In audio mixing, the most common Q settings are the following:

Q=3.0 ~ Q=4.0	(very narrow cutting/boosting).
Q=1.4	(standard cutting or boosting)
Q=1.0	(medium wide cutting or boosting)
Q=0.7	(very wide cutting or boosting)

It is recommended in any mixing session to stick with the above Q settings, as extreme use of Q can severely affect the audio recorded signals.

Common applications:

In *Boosting*, I use a wider Q such as 1.0, and smaller amplitude (Q=1.0, 2dB Boost, 3 KHz), commonly used in boosting vocals, guitars, etc in their resonant frequencies.

In *Cutting*, I use a narrow Q such as 1.4 or 2.0 with higher cutting dB (Q=3.0, -6dB, 400Hz) for example in removing the cardboard sound of the kick drums.

Summary: The above settings are rough but a standard guide in using parametric equalizer. Feel free to experiment with your own audio recording software in shaping the sound you need (of course this is the main function of the parametric equalizer).

What a high pass filter does - Technical explanation

One of the most common but basic question people often ask is "What does a high pass filter do"? This is a technical term relating to audio mixing. This guide can help you understand what this filter does, and thus help you getting a better quality mix. As some problematic and troublesome frequencies in mixing can be corrected by doing high pass filter, let's start with its basic definition.

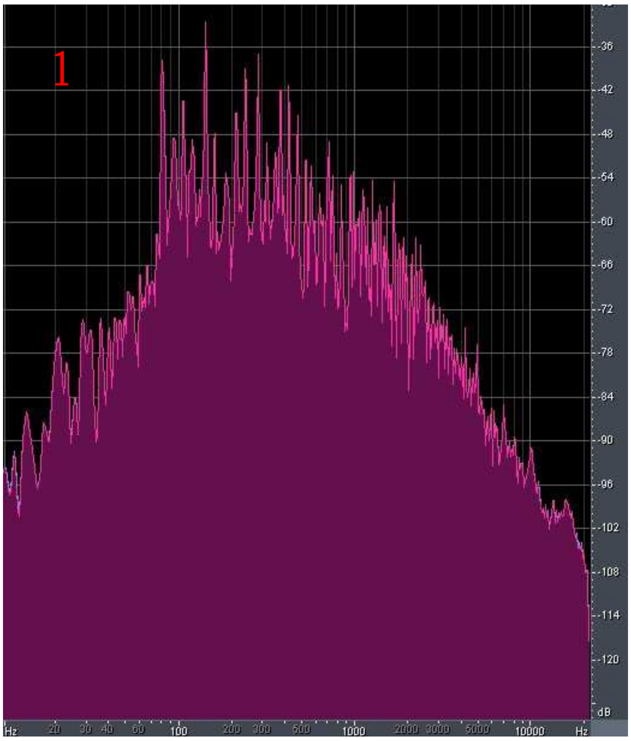
A high pass filter will "pass" high frequencies about a specified cut-off frequencies while attenuates "drastically" below.

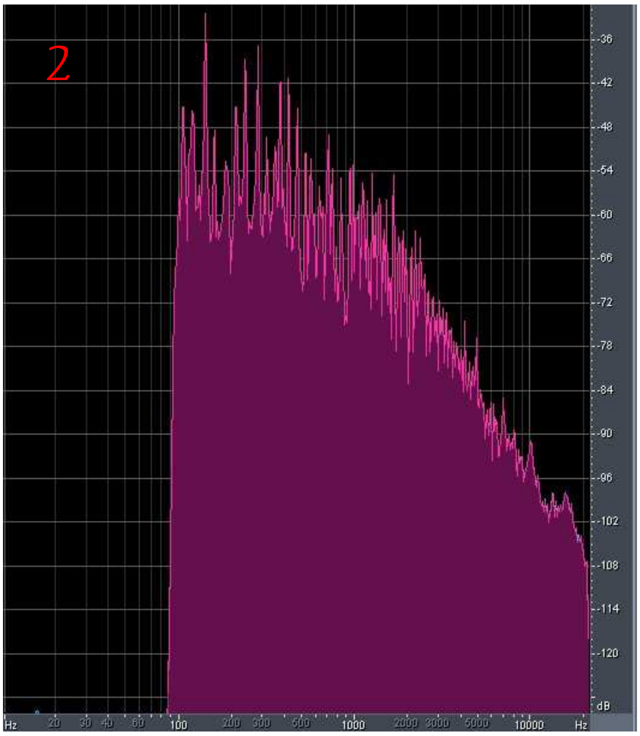
Let's illustrate an example to make easier for you. Supposing the cut-off frequency is 40Hz, if you are using a high pass filter on your audio wave sample, it will only pass 40Hz above (e.g., 50 Hz, 100Hz, 110Hz, 500Hz, 3Khz) while below 40Hz, the filter will attenuate or cut drastically in such a way it will not have "significant" amplitude.

Still confused? OK, let's use a frequency analyzer plot to illustrate the filter graphically.

A frequency analyzer is a tool that plots frequency vs. amplitude. The x-axis is frequency in Hertz while the y-axis is amplitude in decibels. *(See screenshot "1")*

Supposing you will apply a high pass filter to this audio with cutoff set at 100Hz. Using your audio recording software high pass filter function, your new frequency analyzer plot after filtering will be: *(See screenshot "2")*





You see that what stands below 100 Hz is drastically "cut" while allowing above 100Hz to pass.
Using your ear, you will notice that the basses are drastically cut off if the above filtered audio is played on your studio monitors.This is an example of what a high pass filter can do.

One useful application is to remove the rubble at 40Hz, as these frequencies are not in any way useful in audio. It will be removed to save the dynamics for more important frequencies.

Low pass and high pass filters: Application in home studio

This is a short guide on how to effectively use low pass filter and high pass filter in your home mixing and mastering sessions.

Let's start with the **properties of a filter**:

A filter is a signal processing unit that "attenuates" specific range of frequencies. Based on that definition, you might consider a parametric or a graphic equalizer as a "filter" also.

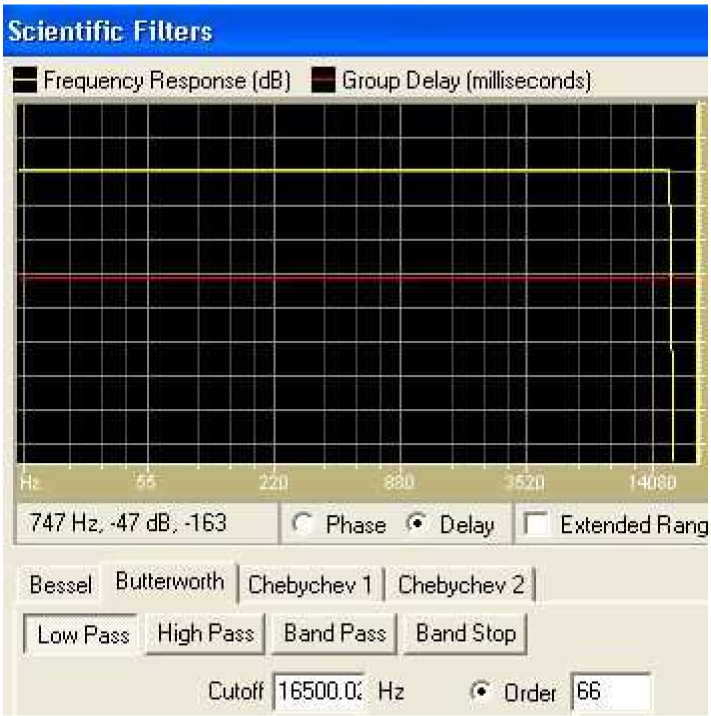
What makes a low pass filter and high pass filter special is the 'shelving' action on a wide limit of frequency ranges or even to infinity.
Here is a definition:
A "low pass filter" allows frequency lower than the cutoff frequency to pass while severely attenuating above it.

So for example, if the cutoff frequency of the low pass filter is 16Khz, then it is said to pass or allow 16Khz below while attenuate severely the frequencies above 16Khz.
This is opposite to high pass filter which allows high frequencies and block lower than the cutoff.

So how are you going to use these tools in a mixing and mastering session?

step 1 - Low pass filter is used to remove high frequency related "hiss" and cleaning of upper frequency spectrum, such as applying a cutoff at 16500 Hz (effective audio range is 20Hz to 16500Hz, the rest can be filtered). There are different kinds of low pass filter, I personally use Butterworth filter in Adobe Audition order 66.

Most commercial master recordings apply a low pass filter at around 16500Hz (this is done during the EQ stage of the mastering process). However, it is not popular to use a low pass filter in mixing process.



step 2 - High pass filter – I commonly use this filter during mixing but not so much in the mastering process. HPF is applied to vocals, guitars and other string instruments to avoid mud with the bass guitar or bass frequencies.

Example is the high pass filter applied to vocal tracks (cutoff frequency 200Hz).

One important distinction of high pass and low pass filters with the rest of the filters (like parametric equalizers) is the absence of Q in the settings. So if you want to use high pass/low pass filters, you need to set one important value, and that will just be the “cut off frequency”. Although in some scientific filters you need to set the “order”, which is the measure of steepness of the frequency attenuation (see screenshot above).

2.3.2 General Mixing Tips

How to improve home studio acoustic treatment

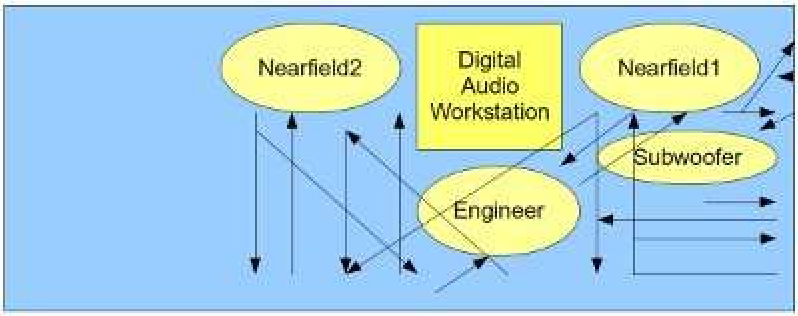
For beginners in home recording and mixing, it's of primary importance and priority that you acoustically treat your room for flattest frequency response. Flattest frequency response tells the truth, nothing more nothing less. So it means that what you really hear during mixing can translate "accurately" to a wide variety of audio reproduction systems (like iPod, CD discman, television, radio, or even hi-fidelity systems with subwoofers or not).

If the room is not acoustically treated to handle "frequency" biases, your mix might only sound good in your studio but sounds awful when reproduced in other audio monitoring systems. No matter how expensive your nearfield monitor is, you still need to acoustically treat your room.

If you are successful on this, it means that what you really hear can "accurately" translate to other audio systems. So if you mix it great, it will surely sound good in other audio systems. It is very important to have a mixing studio acoustically well treated.

What are the causes of this "frequency" biases problem that makes your mix lie to you (sound good in your studio but sound bad in other studios or audio monitoring/speaker systems)?

Consider the arrangement below (which is not correct and not optimal placement for mixing records at home):



Problems of this setup:

1 - Standing waves and reflected waves, distort the real level of the frequency being heard by the engineer.

Why? When you see the arrows in the above screen, these arrows are the sound waves leaving the speaker, they reflect on the walls without acoustic treatment. What happen is that , they reflect right away because the speaker sound waves needs only to travel in a shorter distance as shown in the above screenshot.

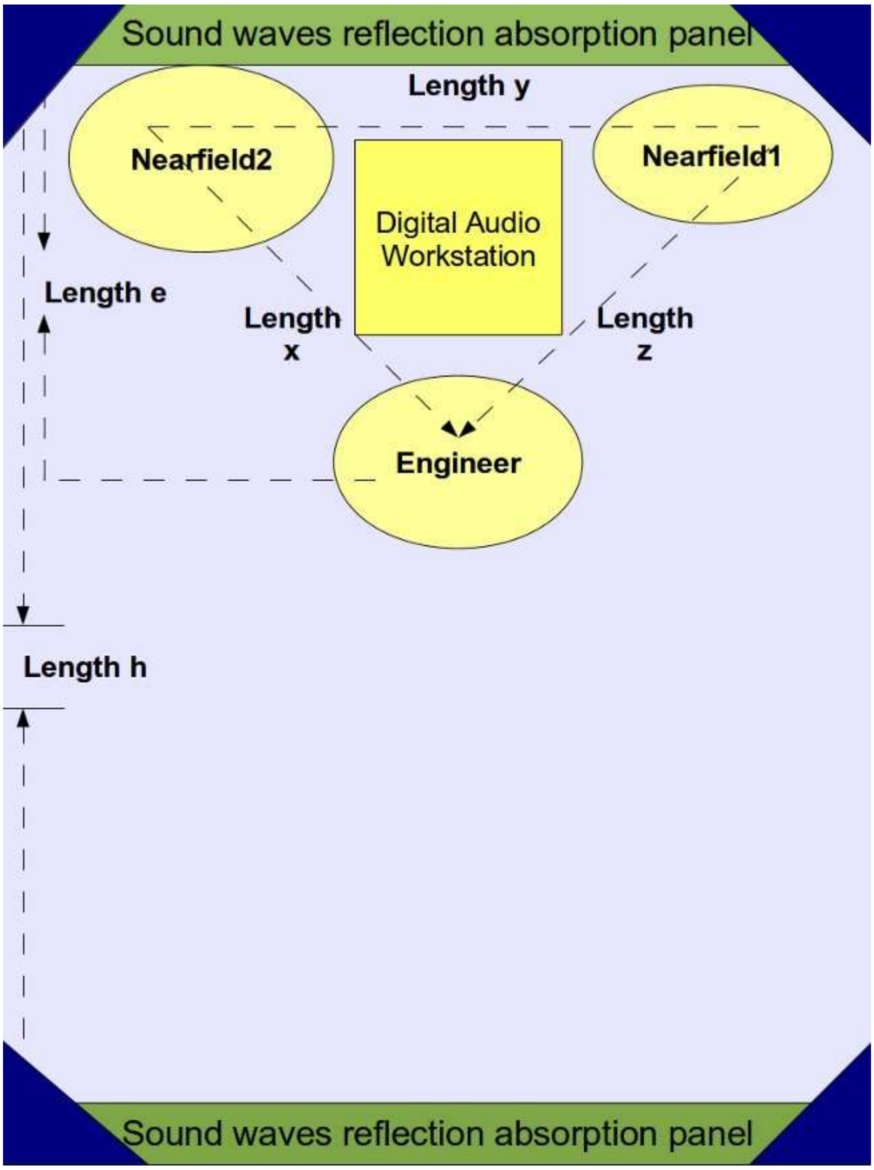
These reflective waves can magnify or distort the real sound level of the mixed audio. So for example if that frequencies belongs to a guitar track around 500 to 5000Hz, the engineer may think the guitar volume is too loud, so what will happen is that he will lower the volume of that track in the mix, which in reality is not really loud but is composed of the standing/reflected waves from the monitoring system.

2 - The studio monitors are not oriented well to the engineer, so he cannot visualize clearly the stereo image of the mix. Based on the screenshot, a lot of sound waves did not hit the engineers ear and instead reflect on the surfaces/walls of the studio. This again , can bias the reproduced sound.

Since mixing engineers depends on the reproduced sound to make judgment, erroneously reproduced sound (due to poor room acoustics) can produced an erroneous mix.

3 - The worst is the placement of subwoofer. Since it is placed near the room corners, it will reflect those bass frequencies too early which again amplifies the bass image which in reality is a low level bass signal.

The solution to the problem is not yet provided, below is the suggested room set up that can be applied to any home studio:



Some very important things you need to know:

a. length x = length y = length z , so it means that x , y and z formed an equilateral triangle.

The purpose why this should be an equilateral triangle is for optimal stereo listening. Of course, you should point the monitors to you in such a way that it forms a triangle like what is shown in the above screenshot.

b. The optimal length e can be calculated as: 38% x length h. length e is the distance from the engineer to the wall facing him. The reason for the 38% is discussed here thoroughly: *http://www.realtraps.com/art_room-setup.htm*

c. The blue colors on the corners are the "bass traps". The overall purpose is to absorb low frequencies and to prevent reflections which will distort the "real" bass sound level.

d. The green are the sound waves reflection absorption panel. Their main job is to absorb mid range to high frequencies.

If this room design is implemented, what will happen is that the undesirable standing waves or reflected waves will be drastically minimized. Thus the only sound in the room should be coming "exactly" from the monitors and does not include the reflected waves.

Finally, since what will come out of your studio monitors is exactly what you will hear, you can use this to create a fairly accurate mix that will translate to various hi-fi monitors or even normal speakers.

For placing subwoofers, you might refer to the following tutorials:

a - *http://www.soundonsound.com/sos/feb05/articles/studiosos.htm*
b - *http://www.gearslutz.com/board/rap-hip-hop-engineering-production/197461-where-do-i-put-subwoofer.html*

So in summary you need:

1 - Nearfield monitors that are properly placed in the room. (x=y=z).
2 - You will need to buy or create bass traps.
3 - You will a sound wave reflection absorption panel.
4 - Rearrange your room in such a way it reflects the studio design above.

Do more research on how to create bass traps and absorption panels by yourself without necessarily buying the expensive ones. You can do this by carefully examining the acoustic properties of the materials (carpet for example can be used as good bass traps).

How to mix instrument frequencies for best sound

Mixing is both an art and science. Why? An art because there are no limitations in being creative.
A science because there are methods to be followed.

Supposing you have completed the recording and panning processes, it is the proper time to start mixing the frequencies of the instruments. A song which is not mixed properly can result to a poor quality sounding record.

I made myself some educational recording and mixing of my own song “At the highway”, I played and recorded all the instruments (guitar, bass and drums) in multi-track and the vocals are performed by Jeanine Maningo.
There is no bass guitar involved. Listen to the sound clip below in which frequencies still aren’t mixed for clarity (although panning and recording processes are already done).
This mix does not include the bass guitar mixing, which will be discussed later.

MIX1 - At the highway (unmixed)*: “audiorecording.me” >Music Mixing>”How to Mix Instrument Frequencies for Best Sound”*

A quick critic to this mix is as follows:

1 - Guitars dominating the mix
2 - Muddy guitar sound
3 - Kick sounds so weak looks like punching a pail
4 - Vocal lower frequency range in conflict with lower guitar frequencies and kick

Though this is a demonstration of music production at its simplest form, it has been illustrated, that the muddiness of the mix can be corrected at the early part of the mixing process and should not be a part of mastering process.
Below is the corrected mix with proper mixing settings applied:

MIX2 - At the highway*: “audiorecording.me” >Music Mixing>”How to Mix Instrument Frequencies for Best Sound”*

What can you say? Clear isn’t? The kick and the rest of the instruments are not fighting with each other, the guitar sound is clear and not conflicting with the vocals anymore. Overall, the sound is not muddy anymore.

The secret in doing this, is very expensive in recording schools. But I will reveal it below:

Rule #1 - Each music instruments has its own center frequencies and range.
Use a parametric equalizer to adjust

Rule #2 - Cut and boost conservatively depending on the resulting sound.
Q setting of a parametric equalizer is important.

Here we introduce “Q” and parametric equalizer. What are those things?
Q is a measurement of how narrow or wide the frequency adjustment on a parametric equalizer. Parametric equalizer is a mixing tool that will enable you to manipulate frequencies of instruments and balance it in the mix, just like what a paint brush will do to a painter.

To simply understand Q:

a - a Q of 1.0 could be considered as medium wide.

b - a Q of less than 1.0 is considered to be wide frequency adjustment.

c - a Q of 1.4 is average adjustment.

d - a Q of greater than 2.0 is a narrow adjustment.

There is also something called “shelving”. It can be classified as low pass filter or high pass filter.

A low pass filter will preserve low frequencies and cut frequencies higher than the cut off. A high pass filter will preserve higher frequencies but cut frequencies lower the cutoff frequency.

A center frequency is a frequency where maximum amplitude occurs. It is a frequency most noticeable by the ear of that instrument.

Let’s say voice frequencies. It is around 300Hz to 3000Hz. It is a pretty wide range, but that is the bandwidth of a telephone line. That’s why telephone lines are optimized for voice transmission.
The center frequencies of that bandwidth are around 1650Hz~2000Hz. It’s where the voice frequency is the strongest. And during a mixing process, that certain spot in frequency range is reserved for the voice to avoid conflict with other instruments.

Also take note the amplitude adjustments in parametric equalizers are measured in dB (decibels).

Below is a list of common instruments I used to mix “At the highway” and the EQ adjustment :

Short Snare

Freq1: 100Hz, Boost 1dB, Q 1.0
Freq2: 2500Hz, Cut 3dB, Q 1.4
Freq3: 8000Hz, Boost 2dB, Q 1.0

Purpose: 100 Hz serves to fatten the snare sound; cutting at 2500 Hz will minimize conflict with vocal frequencies. Boosting at 8000Hz will add some crisp (audible snare strings)

Kick

Freq1: 75Hz,Boost 6dB,Q 1.0
Freq3: 400Hz, Cut 6dB, Q 1.0
Freq4: 4000Hz, Boost 1dB, Q 0.6

Purpose: Boosting at 75Hz creates super-punch for the kick., Cutting at 400Hz can prevent conflict guitar lower bass and improves kick sound by removing the cardboard like sound. Boosting at 4000Hz can make the kick sound to be heard on small speakers such as headphones.

Guitar

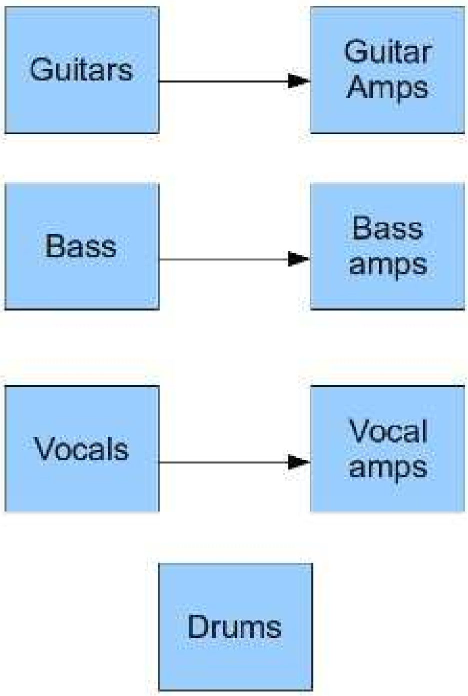
Freq1: 160Hz, high pass filter, Cut 6dB
Freq2: 3000Hz, Cut 9dB, Q 1.0 (Cutting this prevents direct mud sounds with vocals which occupies the same frequency range)

Easy live mixing tips: Getting the best out of a live performance

One of the most enjoyable musical experiences is watching live concerts. Often, great sound engineers hired by great artists are able to produce the best sound as possible when performing live in any concert venues.

This is not a problem. However, you might encounter live performances which sound dull and muddy. This is often associated with garage band like concerts or simply playing live band in your garage, home studios, friends party lounge, etc.

Below is the most common setup found in most indie garage live concerts or home party live band setups. It is because this is the most affordable setup with no expensive mixers required. However, by carefully crafting the sound, the band can produce a great sounding live performance.



As you have noticed, there is no mixer used in the setup. To get the best sound out of this setup, the following are recommended:

1 - **Most of the guitar amps** have low, mid and high EQ settings. Low corresponds to the bass frequency ranges, mid for medium frequency and high for treble frequencies.

a) To avoid guitar sounds to conflict with the bass frequencies (where kick drums and bass guitars dominate), turn the low(bass) setting to around 2 or 3 (assuming 10 is the loudest in the knob).
Setting this to 0 (minimum) is not recommended as it removes the entire guitar bass frequencies, which does not sound nice. Feel free to experiment; start with a setting of 3. If your guitar amp does not have 1 to 10 units, you can estimate it by turning the knob between the minimum and the average.

b) Next, the guitar sound tends to bury the vocals, especially if the guitar is using overdrive or a heavy strumming sound. To make the vocals as clear as possible, you need to turn the MID EQ knob of your guitar amplifier to around 3 to 4.

c) Set the treble (high) section to around 5.

2 - **For the setting of bass amplifiers**, to compensate the loss of guitar bass frequencies you need to beef up the bass guitar amplifiers to make the live performance as heavy as possible. Most bass amps like guitars also have EQ knobs (Bass, Mid and Treble). Set the bass knob to around 7 or 8.

The mid in bass guitar is not so important unless you are doing a bass solo targeting higher bass notes. You can trim down the mid knob to around 3 to 5 to give way for the vocals.
The treble section can be set to 5.

3 - **Finally the vocals**, assuming the vocal amps also have some kind of EQ function (low , mid and high). Set the following:

Low(bass) – 3 or 4
Mid – 7 or 8
High – 6 or 7

Set the echo or reverb setting to minimum. Do not overdo it, as extreme reverberation or echo can make the lyrics unclear. Put as little as reverb to make the vocal performance sounds fuller without getting drowned.

4 - **How about the drums**? Drums do not have EQ settings in live mixing. Unless you are miking each one of the drum components which is often not done (unlike when you are recording tracks in the studio, where drums need to be miked).

The above suggestions will work best for small, garage band like concerts and home party concerts. For big concerts like in a stadium or complex, drums need to be miked and all music instruments will need to be routed to a mixer where the engineer can make adjustments similar to the ones discussed above. The mixer output is then feed to power amplifer/monitors to generate as loud sound as possible.

2.3.3 How to Mix Instruments

How to mix Guitar tracks: Beginners Guide

One of my favourite instruments when mixing is the guitar. Mixing guitar tracks, especially in a home studio environment, can sometimes be a big problem especially for beginners, because guitars need special attention and tweaking to get a great sound while avoiding mud.

This is a beginner guide, so it means that if you are new to mixing, this guide can definitely help you to get started mixing guitar tracks to sound a bit more professional. It is assuming you have already done the following:

a - Recording of guitar tracks
b - No effects still applied to the guitar wave by the DAW.
c - You are using a PC based DAW such as Cakewalk or Adobe Audition

Bear in mind that having a well recorded guitar is as important as mixing the guitar. So make sure that you have the best recorded material possible (in terms of guitarist performance as well as recording quality).
Record at 24 bits 44.1Khz or 96Khz depending on the ability of your sound card.
It is highly recommended to avoid any noise removing process on the guitar signals as it degrades significantly the overall quality of the guitar sound. Instead, fix your recording hardware so that it will provide the least noise as possible.

Let's break down the tutorial into 4 categories. These are the *EQ*, *Panning*, *Compression* and *Effects*.

Guitar EQ settings during mixing:

The overall goal is to prevent muddiness with the vocal frequencies while still allowing the guitar to shine. You will also need to prevent muddiness of bass components (bass guitar and kick drums). The suggested settings are the following:

-8dB high pass filter at 200Hz
(this will roll off the bass frequencies below 200Hz of the guitar to prevent mud with the bass, while allowing above 200Hz to pass, you can use the shelving filter in your DAW to do this).
-9dB 3000Hz Q=1.4
(this will prevent muddiness with the vocals, thus improving the vocal clarity). The amount of cut in dB may be lower in heavy and alternative rock where vocal clarity is not extremely important unlike in pop and country.
+3dB 400Hz Q=1.0
(for overdriven guitars only, this will make them sound heavy).

+2dB 6000Hz Q=1.0
(for clean guitars only, improves the ambiance and clarity of high guitar frequencies).

Compression: Please refer to *page 34* (audio compression tips for mixing).

Panning: Recording producers sometimes record guitar tracks TWICE, one for the left and the other one for the right, panned at -80, +80 units. This will improve the guitar ambiance. This is called "double tracking".

Recording this twice introduces some delay that improves ambiance. If you have recorded guitar tracks only once, you can double track by pasting the original track to the second track (no need to re-record again), then applying "delay" effect (5ms -10ms). Do not overdo delay as this can introduce some undesirable effects in mono.

Effects: I rarely use reverb, chorus or phrasers.
If you're using reverb, I'd recommend to start with 500ms to 1000ms settings applied to only one channel (either left or right). If that doesn't sound good to you, you could apply it to both channels but decrease the overall reverb level.

Do not overdo reverb, as it will make the guitar sound weak and affect its role in the mix. For chorus and phraser, you might as well use external effects during recording, not directly applied to your DAW for best results.

Guitar EQ mixing for rock, country and pop

One of the most important instruments for producing rock, pop and country music are guitars.
In fact, a band alone can be formed by just having guitars, drums and bass (for example it can become "legendary" like Led Zeppelin). Because of this importance, it has been used frequently to produce songs; however some difficulties are encountered during the mixing process.

The common mixing problems with guitar frequencies are the conflicts with vocals and bass frequencies. If guitars are not properly mixed, it can result to a muddy sound. This means that guitars, vocals and bass won't be clearly heard in the mix. The most common problem with the guitar mud problem is "drowned vocals". Drowned vocals are surely not acceptable in pop and country genres, where the vocals should be clearly dominant in the mix.

The followings are the important mixing tips I used for producing rock, pop and country music.
The most important requirement before you can mix guitars is to be properly recorded, clean (noiseless) and of course "in tune". You should have a parametric equalizer to mix guitars. Remember that without EQ, you cannot properly mix guitar frequencies at the optimum level.

Quick Guide:

Mixing guitar frequencies for rock music, pop and country:

High pass filter with cut off frequency at 200 Hz (for all applicable genres)
This will roll off guitar low frequencies which can conflict with the bass guitar sound. Remember that in order to avoid mud and conflict with the bass guitar, low guitar frequencies must be attenuated.

Cut -9dB at 3Khz, Q = 1.4 (for pop and country)
by cutting 9 dB at 3000Hz, you are making a hole in these ranges of frequencies which vocals can sit in. -9dB is a strong cut off which is recommended for pop and country music. However you can extend cutting to a maximum of -15dB, remember to use your ear in judging the sound.
Too much cutting will tend to lose most of the guitar sound main character which can be bad for music production.
But rock music is more popular with drowning vocals. So instead, you can cut conservatively at -3dB only.

Boost 3dB at 400 Hz (Q =1.0) for rock music only
If you are producing rock guitars (with distortion or overdrive); boosting 3dB can make the guitar sound heavier than it seems. This works best when the bass guitar has been cut off at 400Hz also, if not then it may sound less effective.

Cut -6dB at 800Hz at Q=1.4 for all applicable genres
This will make the guitar sound clean and sound more professional. If this is not done, the "cheap" sound of the guitar will appear prominent which can be distracting to the ears.

Boost 2dB at 6000Hz at Q=1 for pop and country genres.
This is particularly important if you have a lot of acoustic guitar arrangements and it will tend to "shine" prominently with crisp at these frequencies. Also, this can be helpful to add more life and sparkle to guitar solos whether acoustic or overdriven guitars.

These are the only settings I care for when I mix guitar, the rest of the frequency ranges I leave it as it is.

How to compress Lead guitar in the mix: Letters from readers

I received a letter asking me for tips about compressing lead guitars in the mix.

From Tony:

I was reading your article with interest about compression settings, especially the guitar settings.
I've bought a behringer composer pro mdx2200 rack compressor to record some lead guitar onto backing tracks. Would it be better treating the lead guitar the same way as the vocal settings you have listed, or should I keep the settings you have for guitar?
Also on the attack dial it reads from 1 to 150 ms and the release dial reads from 0.05 to 5 secs. I know you may think this a silly question, but could you tell me how this is read. I know I've asked a lot of questions, but i am a real novice at this. Hope you don't mind me asking.
Regards, Tony

Emerson reply:

Compression techniques are actually an art, it depends on creativity and how you are going to set those settings in order to achieve the sound you want. So that means my settings are a little guide only. To answer your question, lead guitar frequencies are often high so compression techniques should be:

Lead guitar: *Fast attack*, *Slow release*

This is an opposite to bass guitar which is slow attack and fast release. The primary reason is that for bass are low frequencies which means that their period is long (Frequency = 1/Period), the higher the period , the lower will be the frequency, so to have the best compression effect we will wait until it near reaches its maximum amplitude in the wave then there we will start compression (start of attack).
To simplify, picture it as a wave:

Start of wave : amplitude =0
Middle of wave: (max) = 1
End of wave: back to zero since it is a cycle.

Slow attack is applying at the middle or towards the end of the wave, So it means that start of compression (called "*attack*") is high amplitude then we release it suddenly (*release*), the resulting sound is BIG. Lead guitar is fast attack and slow release, so same concept applies but in the opposite way. Since they have high frequencies, the period is small (milliseconds). In order to make it stronger, apply compression immediately and release it near the end. The primary reason is that the period is small and to have a stronger sound, longer period in the sound wave are to be applied with compression.

Again settings are references only. Try to experiment….
Remember that if you set compression ratio at 1:1, it is no compression at all while 2:1 is that the compression factor is twice. The higher the compression ratio is, the more it tends to equalize all volumes in the sound wave. "Over compression" is bad since you do not have silent areas anymore and all sound waves are now at the same level, resulting in unnatural sounding recordings.

How to mix Electric guitars using "Double Tracking" technique

One of the key elements in rock mix is thick and heavy guitar sound. One of the effective ways to accomplish this sound in the mixing process is through a technique called “Double Tracking”. In this post I will illustrate how to double track guitars in the mix with the objective of making it heavy and thick.

Bear in mind there is a lot of ways to thicken the guitar sound. Double tracking is one of the easier ways.
Alternatively you can do:

a. Compression on guitars to make it sound thick.
b. Applying effects such as maximizer to increase loudness.
c. Parallel compression.

If the guitar sounds thin and weak, it will tend to affect the commercial appeal of the song especially if it is being marketed as a pure rock or alternative music. It is highly essential to mix things right but…

The following are the important requirement before you can double track the guitar in the mix:

a - The recording of the guitar should be free of noise and normalize to the maximum volume.
b - If the guitar is recorded twice, it should also be clean and normalized. But it is not required to record it twice.
c - Record with the best distortion tone you need. Do not record it yet if you are not yet convinced of the distortion tone. It’s much better to experiment with a live band, before starting to record the guitar. The overall purpose is to have a clean and final recording ready for mixing. Remember it is not advisable to fix the distortion tone in the mix. It makes the mixing process get complicated.
d - Double check the tuning of the guitars. Even slightly out of tune guitars can be problematic, since if you double track, it will tend to worsen the out of tune guitars.

It is also highly important particularly in the recent pop rock music trend to achieve not only thick guitar sound but it is also a wide guitar sound. This will achieve the “airy” sound of the distorted guitars.

So how do we start the mix?

1 - Start with placing the 1st track in the Track one of the mixing session.

2 - Place the other guitar track in the Track two of the mixing session. If you are recording only once, just copy and paste the wav file in the Track one to Track two.
3 - Pan the Track one to -75 units (left). Depending on your recording software, this could be in %, for example if the maximum left pan setting is 100% so it will be 75/100 or 75%.
4 - Pan the Track two to 75 units (right).

5 - Now to get that wide thick sound, you can apply 5ms delay to one of the guitar (either left or right) (mix 100%)

6 - To even make it heavier, do not anymore apply reverb on any of the tracks (it is highly important that the reverb is from the room and amp based reverb that will be realized during the recording process). It is because if you start applying reverb on the guitar, it will tend to sound weak and far. Since you are mixing for rock, it is important to get the "in your face" guitar sound.
7 - EQ it properly, do not cut too much bass in the distorted guitar, it will help add the heaviness sound.

8 - Cut 1000Hz and 800 Hz on any guitar to make sound so clean and avoid the cracking sound.

9 - Adjust the track one and track two volume and stop when it is loud enough for the guitar tracks to be heard, not dominating the vocals.
10 - Cut 3000Hz with around -6dB and Q of 1.0 for both guitar tracks.

11 - If your effects are arrange serially below are the sequence of effects that will be placed in each guitar :
a. Parametric Equalizer
b. Compressor
c. Reverb (optional) necessary only if the guitar tracks is too dry.
d. Delay (only on one track)

It is highly important to rely on your ears to do the settings. Do not believe in holy grail settings of compressor, EQ, they are there to serve as a guide and it is important to stick with the basic principles in double tracked mixing such as above.

DEMO: Listen to mix below applying the principles above:
TRACK - At the highway: *"audiorecording.me" >Music Mixing>"Tips in Mixing Electric Guitars using 'Double Tracking' Technique"*
Song Title: *"At the highway"*
Author/Publisher: Emerson Roble Maningo
Artist: Jeanine Maningo
Producer: Emerson Roble Maningo

How to mix Baritone guitar: Tips and techniques

Baritone guitar frequencies sit between a standard guitar and a bass guitar. Baritone guitar looks exactly like a standard electric guitar, see photo below:

Except that the sound of the strings are of lower frequencies compared to standard electric guitar. One of the most important uses of baritone guitar is the production of rock and alternative music.
One of my favorite bands, Staind, uses baritone guitar. It is more appropriate in rock music because of its dark and moody sound. This means it will sound heavy.

Therefore it is important to mix baritone guitar correctly so that it is clearly distinguishable from standard pitched guitars. Below is a sample audio clip of the baritone guitar in clean mode:
CLIP - Clean Baritone*: "audiorecording.me" >Music Mixing>"How to Mix Baritone Guitar: Tips and Techniques"*
You have noticed that the guitar sounds at lower frequencies than what you can hear in ordinary guitars. Now try listening to this audio clip using distorted/riff baritone guitar:
CLIP - Dirty Baritone*: "audiorecording.me" >Music Mixing>"How to Mix Baritone Guitar: Tips and Techniques"*

To properly mix baritone guitar along with standard guitars, bass guitars and other instruments, below is the important guide:

You need to cut standard guitar lower mid range frequencies to give way for baritone guitars:

Standard guitar/
Cut 150 Hz -6dB (high pass filter)
Cut 500 Hz Q=1.0 level= -3dB to -6dB
Cut 800Hz Q= 1.4 level = -3dB

Baritone guitar
Cut 4000Hz (low pass filter) -3dB
Cut 150Hz (high pass filter) = -6dB

That's it; of course the bass guitar will occupy the lowest frequencies so baritone guitar and standard guitar are both cut to give way for bass guitars. Baritone guitars occupy lower mid range frequencies 200Hz to 1000Hz while electric standard guitars occupy higher frequencies (4000Hz and above).
1000 to 3000 Hz will be prioritized for vocals.

How to mix Bass guitar: Tips and techniques

Bass Guitar is very hard to mix. It is always the main reason why the mix sounds either dull, thin or mud.
The major problem is that all instruments have bass frequencies, but not so heavy as a bass guitar and a kick drum. In a mix, all instruments are played together, and the primary problem lies in the bass frequencies. That's why every time you hear tracks that haven't been mixed, it sounds mud.

I have been mixing for years and I love to present these two techniques I learned from experience in mixing. Basically you can only apply one technique per song. But you will have two choices in how to approach bass guitar mixing :

a - **The Rock Bass Guitar Sound Mix**

In this mix, the objective of the bass guitar is to sound heavy and partly dominant in the mix.
As a rock producer, I like the bass guitar to sound aggressive and up front in the mix. Did you notice that when you hear rock tracks today such as Trapt, Green day, Simple Plan, their bass guitar is very dominant? It is a secret of sound engineers in how to make the bass guitar loud while avoiding mud.

As a guide, we will designate 45 Hz to 250 Hz as the bass frequencies where kick drums and bass guitar mainly reside. The problem is how to blend those two together.

Since the bass guitar needs to sound heavy and dominant, it should occupy mainly the bottom 45 Hz to 250 Hz. But…
We will dip 100 Hz for the kick drum spikes to shine through. I usually dip the bass guitar around 100 Hz with Q settings of around 2.0 and -9dB reduction.

To balance, I will boost kick drum at around 100 Hz with Q settings of around 2.0 and 9dB~12 boost.
To sound better, I will apply high pass filter (so it will attenuate frequencies lower than 50Hz) on kick drum around 50 Hz -3dB reduction, for the deep bass guitar frequencies to dominate the sub woofer, making it sound heavy.

But I will not boost the bass guitar at any frequency between 45Hz and 200 Hz.
I finally boost 250 Hz for bass guitar to make those notes more audible. I use Q of 2.0, and boost at 3dB.

What about the kick drum?
As a rule the kick drum needs to be dip at around 250Hz to 400Hz with Q of 2~3, to remove those card board sound. This makes the bass guitar notes more audible as well as the distortion guitar.

What about other instruments?
All instruments are to be applied with high pass filter at around 250Hz -6dB reduction. This will make the bass frequencies 45Hz to 250Hz, a place just for bass guitar and kick drums.

What is the result? A heavy bass guitar sound typical for rock music.

b - **The Pop Bass Guitar Sound Mix**

This is very easy and simple to do. The principle is to avoid heavy bass sound to emphasize clarity, punch and elegance of vocals and guitar instruments. This is mostly applicable in pop music as well as country music.

The Principle:

The kick drum solely occupies the 45 to 150 Hz spectrum; this will make the kick drum sounds so fat and strong very catchy for pop music.

The bass guitar will rest at 200 Hz, it won't produce strong bass but the bass guitar notes are highly audible and it will be there to support the song "groove".

Specifically, the kick drum is boosted 6dB at 80Hz with Q of around 1.0. To prevent heavy muddiness which can affect clarity and airiness of pop music, both the kick and bass guitar are applied with high pass filter around 3dB reduction at 50Hz. Also the bass guitar is applied with high pass filter starting at 200 Hz, so it will attenuate frequencies below 200 Hz, making the sub woofer and the bass frequencies mainly composed of kick sound.

What about other instruments?

Again a simple high pass filter will be applied in all, as we do not need their bass frequencies to shine (such as electric guitar, acoustic guitar and vocals). I will set it at 250Hz, so below that frequencies, it will be attenuated.

The result? A very clear and defined mix for bass, ideally for pop and country music.

If you need to hear audio samples implementing these concepts, you can listen under "My Works" on '*audiorecording.me*'. I am using Adobe Audition 1.5 for mixing bass guitar with the built-in parametric equalizer (can be found at Effects –> Filters –> Parametric equalizer).

How to mix acoustic guitar and vocals: Simple guide

One of the most popular pieces of music in modern times is doing an "acoustic version" of a song. There are two ways in how to make an acoustic version of the song:

a - First method -> recording acoustic guitar and vocals together
b - Second method -> recording acoustic guitar and vocals in multi-track session

This short guide shows how to properly mix acoustic guitar with vocals using both recording techniques. Let's discuss the first method:

First method: Mixing acoustic guitar and vocals done together:
In this way, follow the simple steps:

step 1: Connect the guitar to a mixing console.

step 2: Connect the microphone for vocals in the mixing console, microphone section.

step 3: Now, test the vocals and guitars playing together.

step 4: Using Mixing console EQ or any equalizers. Cut -6dB at 3000Hz for guitars (Q=1.4). Boost 2dB at 3000Hz for vocals (Q=1.4). If you cannot find equalizers for each track, you may have trouble recording together.

step 5: Now, test again. You should clearly hear the vocalist lyrics played with the acoustic guitar together.

step 6: Commence the recording session. After recording it together, the acoustic version is cleanly recorded and mixed.

Second method: Mixing in multi-track – acoustic guitar and vocals

step 1: Connect the acoustic guitar first in the mixing console.

step 2: Record the acoustic guitar first (this is accompaniment) , the vocalist can sing but it is not recorded (only serve as a guide to the guitarist).

step 3: Record the vocals using recorded acoustic guitar in the multi-track view.

step 4: Now that the two are recorded together, Cut -6dB for acoustic guitar at 3000Hz (Q=1.4) and boost 2dB for vocals (Q=1.4 at 3000Hz).

step 5: Listen to the mix very carefully; you should hear the vocalist lyrics clearly played together with the acoustic guitar accompaniment.

step 6: Mix down the acoustic guitar and vocal tracks.

How to mix vocals and put some professional effects

Vocals are the most important instrument in any audio mixing work. In fact, if the vocals are not properly mixed, it will give a serious difference between good and poor audio production work.

Mixing vocals is therefore a big priority in the audio mixing session. In this article, I will share my personal techniques when it comes to mixing vocals. I will illustrate those essential principles I have in mind to come up with greatly produced vocals.

First Principle: Greatly recorded vocals will make it very easy to professionally produce vocals.

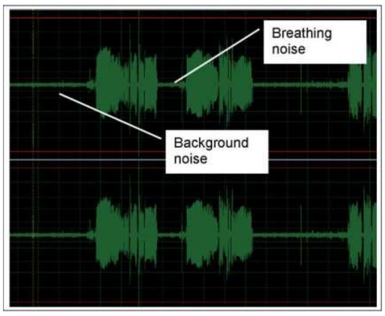
Explanation: Garbage in–> Garbage Out, never dream of having professionally mixed vocals if the recording sounds bad.
The problem now centers on your recording skills of vocals. It is why mixing is very different from recording when it comes to technical aspects of music production. When you say high quality recording it says, “It captures the best performance without any noise, interference or clipping”

How do to this?
I do this by testing the vocal level first before recording. The objective is to record the vocals at a maximum level *without clipping or distortion.*. Once you have set the levels, you can now proceed to recording.

It is highly important that recording should be completely raw and without ANY EFFECTS including reverb. Some engineers add reverb only feed to the vocalist headphones to ensure that she feels it LIKE A PRODUCED SOUND, so that she can sing with confidence.
A real recorded vocal from a professional studio is DRY. Without any effects I mean. It is highly important that if the recording is done digitally, it should be recorded above CD quality, this means going above the normal 16 bit 44.1 KHz standard. Popular sound cards such as M-Audio Audiophile 2496 PCI Digital Audio Card can get 24 bit 96KHz.

Second Principle: Clean up the audio first before using it in the mix

Now that the recording is done, open it up using your favorite audio editor. Then zoom it out and remove any background noise and unusual breathing related noise.
Noise normally occurs in the beginning and in the end. Breathing noise (should sound minor) occurs during the pause of the stanzas, or in the way to the chorus of the song.



You can refer to page 31 (tips to remove noise in recordings), regarding this in detail. After noise removal, since there are lost amplitudes, you can normalize the wave to bring up the signal to a maximum.

Third Principle: EQ first before Compress

Believe it or not, a vocal does not need serious effects like other instruments. The simpler your used effects are, the better the produced vocals will be.
It is highly important to EQ first using these settings :

Cut 200 Hz (high pass filter)= -6dB
Boost 3000Hz Q = 1 = 3dB
Boost 15000Hz Q = 1 = 3dB

Take note that if you take the summation of cut and boost, it is literally equal to zero, it cuts 6dB and I boost 6dB. So only EQ effect overall, the frequency response change, NOT the volume.
For compression, I share on page 34 some techniques that include vocals. Personally I like the Sony Wave hammer plug-in in Adobe audio, presets to voice. It produces some of my finest mixed vocals.

Fourth principle: Be very conservative with reverb

The mix vocals I used have a very low amount of reverb. This could be due to the fact that I am mixing for rock and pop, in which audible reverb is not very popular, unlike in other genres.

But not only that, having low amounts of reverb makes the vocals stood up and sounds very strong.
You can easily captivate the listener with strong vocals with low amounts of reverb. I use Sony ExpressFX Reverb, set to Plate reverbs. Plate reverbs is highly recommended for vocals and only set it:

Room size: 30
Liveliness: 10~15
%Original (dry mixing) = 85%~90%
*%Reverb = 15%~*10%

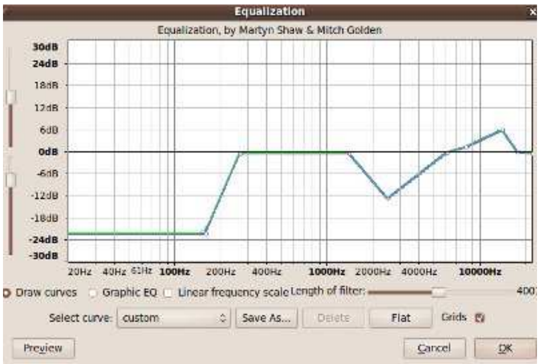
Then listen very carefully to the mix and avoid boosting any frequency of other instruments in the 3000 KHz range because it belongs to the vocals. Or else you will be facing muddy mix vocals that ruin the production. You can get the Sony Direct X plug-ins used in this tutorial from Sony Creative Sound Forge 10.

How to Mix Background Vocals: EQ, Compression, Reverb & Panning Settings

The use of background vocals is common in music production, particularly in the pop and country music. However, background vocals are often challenging to mix typically for a beginner in audio mixing. The aim is that it stands out of your mix while not affecting or drowning the lead vocals which of course are more important than the background vocals.

The most important audio parameters that can be adjusted to create great background vocal mix are the EQ , Reverb and Panning settings. Compression techniques are still useful but not as useful as the three based on my experience.
So lets start with **EQ Settings:**

Cut -9dB to -12dB, 3000Hz, Q=1.4
Boost +6dB, 15000, Q=1.0
Apply high pass filter starting at 250Hz



The 3Khz frequency range (Q=1.4) is cut to give way for lead vocals. The amount of dB to cut from that range is around -9dB to -12dB. It is highly important to use your ears during mixing, and these figures may not be applicable in all background mixing scenarios. For example, in some instances, -6dB cut is enough.

It is highly important to boost the hi-fi range (15Khz) to give smooth thin background vocals (with positive sibilance effect) which are desirable in pop and country music. Since bass vocal frequencies can often cause masking in the background/lead vocals, it is filtered using a high pass filter.

Panning Settings:

For panning, the lead vocal is always positionned in the center of the mix. But the background vocal can be either be panned slightly left or right.
The suggested settings: LEFT (-10 to -20) , RIGHT (+10 to +20).

Compression Settings:

Compressing background vocals is often desirable especially if it fairly occupies a substantial portion in the mix. I use the settings mentioned on page 34.

It is important to balance the overall volume particularly between the lead and background vocals. After you compress the background vocals; you need to adjust the overall volume of background vocal track in such a way it will not overpower the lead vocals but not too soft.

Reverb Settings:

Reverb settings in background vocals are very important.
Typically for lead vocal and background vocal mix, the lead vocal is left dry or with very little reverb (~300~500ms) but the background vocal reverb will have a higher reverb settings (around 1000ms ~2000ms).

Again use your ears to adjust the settings. If you use Sony Express FX reverb, background vocal settings are around 50% reverb and 50% dry. The most desirable reverb type for vocals are “plate” reverb types.

How to mix vocals: EQ Settings: letters from readers

I received an inquiry from a reader of this blog about vocal eq settings and tips:

> "I just wanted to thank you for *the tips* on mixing vocals, guitar, drums and bass. I was having real troubles with the bass and vocals getting lost in the mix, and I was amazed how your EQ and compression settings cured that.
> I noticed that the vocal really improves with the boosts you mentioned at 3khz and 15khz. I understand the 3k boost, because that is at the top of the vocal range, but how does the 15k boost work? It seems to help a lot.
> thanks again, Larry"

Emerson reply:

The inquiry *"but how does the 15k boost work?"* is also a question that came to my mind a few years ago. So how does the 15 Khz boost really work? I will answer this question based on the observation and experience:

> Professional vocal recordings, recorded using high quality microphones such as Neumann U87, can capture "throat" aspect of the voice. This "throat" voice frequency plays a strong role in the high frequency aspect of vocals (12Khz to around 15Khz). However, I also noticed that even cheaper modern microphones can capture these frequency ranges very well.
>
> If you listen to tracks done by modern recording artists, you will notice the "throaty" section plays a great role in the vocal quality, especially in the high frequencies. It is one of the reasons I will boost 15Khz to increase the "throatness" sound of the vocals. By being boosted, the vocal will not sound dull but it helps the high frequency vocal range to shine, beneficial for vocalist with throaty voice such as Bono/U2. The result is a professionally sounding voice.

It is amazing to know that you cannot hear this "throaty" section with a vocalist singing in front of you in acapella (without the aid of microphone). You will most likely hear only the 1KHz to 5Khz which are sensitive frequency ranges to human ears.

But with the use of high quality microphones - those that can faithfully capture the entire frequency range (20Hz to 20,000Hz) - a mixing/recording engineer can take this advantage to explore the nice aspect of the vocals which you cannot notice in an acapella performance (those singing without a microphone).

That's why it is common in commercial pop recordings to have this "throat" aspect clearly exposed. It adds hi-fidelity, ambience and clarity of the vocals. To maximize the benefits, the vocal is boosted at 15Khz while the cymbals and other hi hats are boosted lower than this (for example like in the 10Khz to 12.5Khz range), this is done to avoid "mud" in the high frequency range and lets the vocal hi-freq shine through as well as the drum hi-hats.

Rock vocals and guitar: Reverb settings

I have been mixing for quite a time already, and I admit reverb settings are the most abused effect in mixing. I can hear this quite often in amateur recordings where the instruments are played like in a football stadium or inside the church.
Worst will happen to rock music with improper use of reverb. It surely can ruin a mix and damage the recording quality of the song. Trends of recording in rock music drastically change over the last 20 years. In 1980's we often hear rock music with deeper reverb such as Outfield and the Police. But reverb settings came down to a minimum starting 1995 above until now.

Today, we almost can't hear reverb in rock music. But there is reverb in it; ordinary listeners just won't notice it. In this post I can give my reverb settings which I currently use in mixing rock music.
I will break it down according to the major multi-track components in mixing. These are: vocals, clean guitar and rock distorted guitar. I will make a separate post on the other instruments such as bass and drums.

The settings below are not the straightforward solution to all of your reverb mixing related problems. Always use your ear and remember to fit settings of reverb in the overall mix perspective. Use below settings as a starting point:

"**Vocal Mix Reverb Settings**"

I use plate type of reverb from Sony Express FX Reverb. To express it as a percentage with 100% original as zero reverb, I only use 23.6% reverb and the remaining 76.4% the original signal. The room size is also a bit conservative; I use 30% out of a maximum of 100%. The "liveliness" or echo is only 10 %.
Most engineers will measure milliseconds for reverb. Unfortunately, I am not using it for vocals and could not give you a conversion. Use your ear to convert.

From the technical point of view, you should put less reverb on vocals so that it will be "up front or on your face" and will enable vocals to dominate the mix. Although there are exceptions to this such as in "grunge", "metal" or "alternative" genre where vocals are buried in the mix.

"**Clean Guitar Reverb setting**"

Guitar is very important in the rock mix. It is the primary melody instrument, so it should also sound up front, but not to dominate the vocals for a pop-rock style mix.

I am using Adobe Audition Full Reverb effect and I set it to 1000 ms for *reverb length*, *attack time* is 20ms, *diffusion* is 500ms, *perception* is 70. And the mixing ratio among dry and reverb is 90 to 10. This means only 10% of the signal is applied with reverb. The remaining is the dry/original signal.

"**Rock distorted guitar**"

I came out with this type of setting after hearing "So far away" by Staind, although I could not perfectly match the reverb setting. But for a style of "slowish" not a "speedy" type of song, this setting applies. If your rock song is a speedy, just decrease the "*decay*" in ms by around 50% and your mix will be fine but use your ears!

I use Adobe Audition Studio Reverb Effect when putting reverb on distorted guitars,
Room size=20, *Decay*=3500 ms, *Early reflection*=0%, *Stereo Width*=9,
High Frequency Cut=20000Hz, *Low frequency cut*=500 Hz, *Damping*=0%, *Diffusion*=0%

And for the dry to wet ratio, I set dry signal to around 95% and only 5% for signals with reverb (wet).This is a good technique to give a good punch because of drier signals.

Audio sample: I mixed this one using the following reverb settings above:
CLIP - Here you are now *: "audiorecording.me" >Recording Tips>"Rock Vocals and Guitar Reverb Settings: Revealed!"*

How to mix piano keyboard: Tips and techniques

Piano is one of the most common instruments you can find in the pop and country mix. It is commonly found in ballads and adult contemporary music.

In this short guide, you will know how to properly mix a piano and apply panning/EQ techniques. The main objective is to let the piano tracks shine without causing any trouble with the rest of the instruments such as guitar and vocals.

To the recording producers and mixing engineers planning to use piano in the mix:

Rule1: If the piano is being used as an accompaniment, it is much better to use chords than high/medium note piano to play along with the vocals.

The primary reason is that it is much harder to isolate the mud between a high/medium note piano playing along with the vocals because they occupy a highly similar frequency range.

That's why in most commercial songs released using piano as one of the tracks, it is using piano chords to accompany with the vocals (for example "Let it be" by Beatles).

Rule2: EQ mixing/compression suggestions for piano:

a - Cut -6dB 3000Hz Q=1.4 (this will reduce the muddiness and frequency masking problems with the vocal and guitar frequencies)

b - Apply high pass filter at 200Hz (this roll-off /attenuate all the frequencies below 200Hz while passing all piano frequencies above 200Hz)
The overall objective of this high pass filter settings is to prevent the low piano/bass frequencies from masking bass guitar and kick drum frequencies.

c - Boost 2dB 6000Hz Q=1.0 – this EQ setting will add more gloss/shine to the piano tracks particularly in the intro and piano solo sections.

About compression you can apply the same settings than guitar. However, I recommended just enough compression to get a loud sound on the piano, as preserving dynamics could be very important for a piano.

Rule3: How to pan piano in the mix:

Like most instrument such as drums, vocals and guitars, you also need to properly pan piano in the mix to be more effective. Bear in mind that panning settings for piano in the mix is different for each intended use of piano, such that:

a - If the piano is used as an accompaniment with other instruments playing along it (like guitars), then it is much advisable to pan piano around

Left: *-45 to -55*
Right: *45 to 55 units*

Most guitars are panned much farther from the center (-60 to -100 or 60 to 100 units).
By panning it farther from the center, the piano will shine as an accompanying instrument and will not cause mud with other instruments near the center particularly the drums, bass guitar and vocals.

b - If the piano is used as a solo instrument with no other instruments playing along it, it is much better to pan the entire piano tracks in the center (center =0).

c - If the piano is used as a solo background instrument playing along with vocals, it is much better to pan the piano slightly off center (5 or units). This is applicable if the pianist wishes to play high notes along with vocals. If the piano background is purely chord based (no arpeggio playing), then it can be pan in the center.

How to mix sitar: Tips and techniques

Sitar is an Indian musical instrument.
I bet you probably have heard some sitars. You can hear them a lot in Indian and Middle East based movies or a Hollywood film with Middle Eastern theme. Sitar sounds similar to an over-bending "high notes" acoustic guitar.
This is how a sitar looks:

The mixing tips are still the same. In your multi-track editor, once you have recorded the Sitar very well, you should start EQ'ing the tracks then apply Compression.

The mixing objective really depends on what type of material you are mixing:

a - Is it a solo performance of sitar?
b - Is it an instrumental performance of Sitar with some little percussions? This is the common type of performance I've heard with sitar.
c - Is it a performance of sitar along with vocals?
d - Is it a performance of sitar along with different string instruments such as violin, electric guitar or bass guitar?

If it is a **solo performance** of Sitar, you really do not need any mixing or compression at all.
Just leave the performance as it is. Sitar sounds really unique and full when played alone.

However if it is an **instrumentation performance with percussive background**, you might want to position Sitar in the center of the mix (pan to the center) with percussions panned to either right or left.
When you see them live, percussions are on the right or left of the sitar. You do not need to cut or boost frequencies for sitar in this case, so no EQ settings needed. However you do need compression to make the softer notes of Sitar sound stronger. Use the same compression settings as guitar.

Things start to become critical when a sitar is **played along with the vocals**. Vocals will most likely be drowned with the sitar, because they fairly occupy the same frequency spectrum.
In that case, if vocals seemed to be a more important track in the mix, execute these EQ settings in your mix:

Cut -6dB 3000Hz Q=1.4 (for Sitar)
Boost +3dB 3000Hz Q=1.0 (for vocals)

Do the same for a performance with sitar along **with different instruments**.
You do not need to worry about the bottom, because sitars do not have much bass, when you listen to it live played. So you need to deal with high frequencies only. As a rule of thumb the sitar is occupying a higher frequency compared to acoustic guitars; but a little lower than a violin.

2.3.4 How to Mix Drums

How to mix snare drums: Tips and techniques

Snare drum is the loudest sounding part of the drums. The way it will be mixed drastically affects the genre of the sound.

For example, snare drums are mixed very differently in rock compared to jazz or blues music. In rock, we often consider snare drums as a loud sounding instrument, while it should sounds smooth and conservative in jazz. This little guide is applied for rock, alternative and pop music mixing of snare drums.

The key to have a good sounding snare is to have it recorded properly. Even though you are searching for ways to mix snare drums, you need to ensure that the recorded snare is good, which means:

1. No clipping or distortion.
2. No extreme or disturbing bleeding noises from other drum instruments.
3. At least recorded at a reasonable sounding level (not too loud, not too soft).

Mixing of the snare SHOULD basically comprise of a three-stage process.

Step 1 - Boosting for maximum possible volume:

For the first step, assuming you have recorded a good sounding snare that complies with the requirements discussed earlier, you then need to boost it to maximum sounding level.
Using your recording software, you can use normalization at 0dB. This will boost your snare sound wave at a maximum of -5 dB. The purpose of this process is to maximize volume which is essential for rock music but still has some room for adjustments.

Step 2 - Applying EQ:

The next step is applying some EQ. You will noticed that in rock music, snare sounds heavy and crispy, so you need to have this EQ settings done using a parametric equalizer:

8KHz Q=1.4, Boost at 3dB (adds the crispiness)
2Khz Q=1.4, Cut at 3dB (cuts to dullness sound of the snare, to make sound more heavy)
75hz Q=1.4, Boost at 3dB (boost the heaviness sound of the snare).

Arithmetically, you are adding a +3dB to the overall snare sound wave (+3dB-3dB+3dB = +3dB). It is the reason why you have it peaking at a not much too loud volume (for example = -5dB, to have room for EQ adjustments).

Step 3 - Appying further effects:

The last step is to apply some effects. My favourite is to add some echo and delay, however be careful not to overdo this effect. Make sure it extends a little of the snare hit sounds. You may not need to apply compression. The key is to apply as little effects as possible, while emphasizing the loudness of the snare (to be done by EQ and increasing volume). Doubling the snare in your multi-track can even help increase loudness.

How to mix snare drums: EQ, compression and panning tips

One of the most powerful elements in the rock drum mixing session is the snare drum. In fact, great rock songs are often associated with unique snare drum sound which we often remember throughout the years.

Snare drum mixing of music by Led Zeppelin, The Outfield, Spin Doctors and Nirvana are my personal favorites because they sound loud, crispy, heavy and well mixed. Modern pop beats sadly put less importance in drums, particularly the snare. But a well mixed snare drum can make the song worth remembering, so what are the EQ, compression and panning mixing tips for snare drums?
This guide puts more emphasis on rock music more than other genre like pop, jazz and country. The objective is to give the snare drums a great and loud sound.

step 1 - Let's start with the most important, the EQ.

It is important to know that EQ settings for snare drum mixing are treated differently between each song. There is no set of standard for EQ settings. However, you can categorize them as follows (you can select on the following settings depending on your mixing application):

Heavy snare with crispy snare string sound:

Boost +3dB 100Hz, Q= 1.0
Cut -6dB 2000Hz Q= 1.4
Boost +3dB 8000 Hz Q = 1.0

Dominant and crispy snare sound

Cut -3dB 200Hz Q= 1.0
Boost 3dB 8000Hz Q=1.0

Since the following settings use Q, you need a parametric equalizer to implement those settings. Bear in mind that the "dominant and crispy snare sound" is recommended only for instrumental rock music since 2000Hz – 3000Hz frequency ranges are not cut. If these are used in music with vocals, the snare drum may drown the vocal frequencies affecting vocal clarity.

step 2 - The compression settings which I already mentioned page 34 (audio compression tips).

Attack: 20ms
Release: 40ms
Threshold: -12dB
Compression ratio: 5:1

Feel free to experiment the attack and release settings.
The compression ratio can also be set to 4:1 depending on your flavor.

step 3 - Lastly the panning settings.

Most engineers pan the snare drums at the center. But in reality the snare drums are not perfectly located in the center of the mix. When you visualize the real drums, it is bit off-center. I recommend using 6.25 units to the right. In your mixing software, an instrument panned to the right will have a positive sign while panned to the left is negative, so it will be +6.25.
More details can be found on page 79 (How to pan drum instruments).

Final recommendation: The settings above are not meant to be the final solution. I recommend to highly rely on your ears to tweak the settings further to create a great snare sound in your mix starting with the settings provided above.

How to make loud snare and kick drums in the mix

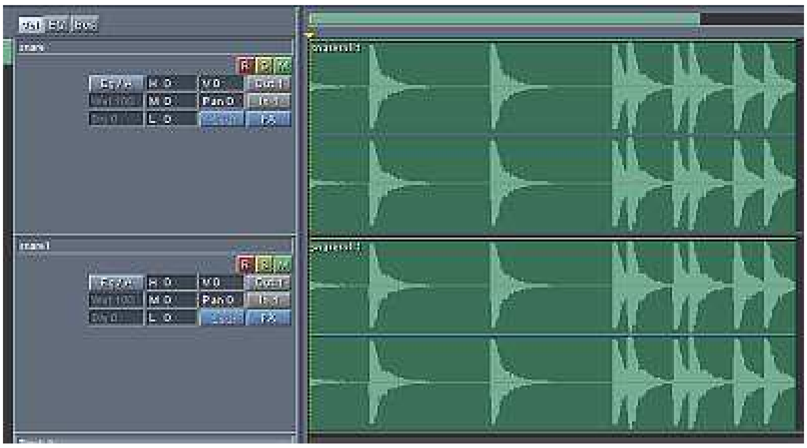
In every rock and pop record, it is important to attain a loud snare and kick drums in the mix. However a more precise requirement is not being loud at all, but mixing it with DYNAMICS.
If you are an engineer working on a mix, you should be able to mix it with dynamics. For example the 1st stanza is slow then, the intro drums to the chorus should be loud to bring in the dynamics and beat for the chorus. The purpose is to provide a sound hook to the listener in preparation for the chorus. This makes the song memorable especially if the sound hook is greatly done.

It can also be done before the 1st stanza (in the song intro section), a very good example is Nirvana's *Smells Like Teens Spirit*. The guitar intro sounds soft and medium. Yet a surprise snare and kick drum combination attached the listener to the main riff of the song. The result is a great and memorable drum hook as well as the associated guitar riff. This production technique is highly important in rock records. Yet some bands particularly those with poor drummers are not good in adding dynamics to their drums, making the sound of the drums too mechanical and predictable.

If you have this kind of drummer, there is a good way to fix it in the mixing stage instead of repetitive re-recording process which can be costly in terms of studio time. I call this technique as "doubling" process.

Doubling process is very simple; it is doubling the recorded wave in the same time duration. The result is the addition of resulting sound amplitudes which increases the volume. I use this technique only in some parts of the mix where I need the snare and kick drums to be very loud. For example below is result of the snare drum sound recording:

Now using doubling technique to naturally create loud snare is to simply duplicate the original snare drum and paste it in in the next track such as shown below AT THE EXACT TIME DURATION, INTERVAL AND ASSIGNMENT as the original wave (however creatively you can have them in "delay" but adding delay is more useful to create a "snare string sound like in a Cajon drum box" than making it purposely be loud):



The result is the increased loudness of snare in that particular section of the mix. This is a more natural approach of creating specific loud drum sections instead of drowning the track with effects.

How to compress the snare drum and kick drum

It has become part of the recording folklore to compress the snare and kick drum. But first, you have to know *why* you are doing it. If you do not know why, then you're never going to get a good result - the sound you will achieve will be no more than the work of random chance.

Compressing individual drums vs. the whole drum set

There are two ways you can approach compressing the drum set. One would be to compress individual drums, the other is to compress the drum set as a whole. These will produce entirely different results. You can do both if you wish, but here we shall concentrate on compressing individual drums, principally the snare and kick, but also the toms too.

The sound of drums without compression

A while ago an experiment was carried out where a snare drum was recorded and the recording played back through a PA system. The sound of both the drum itself and the PA were fed to an audio analyzer. Apparently, to reproduce the sound of the drum accurately and maintain the transient (the initial strike) properly, it took 1000 watts of amplifier power. The reason for this is that the transient, the very first few milliseconds, is VERY loud. The sound dies away quickly after that. So to reproduce the transient accurately, a lot of power is needed. In recording, then the level must be set so that the transient does not exceed 0 dBFS - the full scale level of the system before the red light comes on.

Why drums need compression

The problem now is that the transient is much louder than the 'body' of the sound, as the strike dies away. But the transient is short and does not fully register with the ear. So the drum is actually a lot louder than it sounds. Yes a drum played live sounds loud, but any other instrument played continuously at the level of the peak of the transient would be truly ear-splitting.

If the transient therefore can be made quieter than the body of the sound, overall the strike will sound subjectively louder. Actually, 'louder' is probably not quite the right word for the subjective experience. 'Fuller' or 'more powerful' would be better.

How to set the compressor to make the snare and kick sound fuller and more powerful

Every compressor - every decent one - has a control labeled 'attack'. This is confusing. Anyone new to compressors would think that more attack means a more attacking sound. In fact this control sets the speed at which the compressor responds to a sound. If you set a long attack time, say 100 milliseconds (a tenth of a second), then the transient of the drum would get through before the compressor had time to respond. So to lower the level of the transient, you should set a very short attack time, as low perhaps as just one millisecond.

When compressing individual drums, the attack time is the most important control. The compression ratio can be set to around 4:1 and the release time to 100 milliseconds. Naturally you should experiment with all of these settings.

Problems with a short attack time

One thing is very much for sure, you have to experiment with the attack time. Setting an attack time that is too short will result in a 'flattening' of the sound of the drum. It just doesn't sound natural any more. So you should pay a lot of attention to very small movements of the attack control because these small movements will make a lot of difference.

Differences between the snare drum and the kick drum

The main difference between the snaredrum and the kick drum is that the snare is always a very attacking sound with a sharp transient. The kick is always less attacking, but the degree of attack can vary. If a hard beater is used, then the sound will be attacking. Sometimes a piece of hard plastic is attached to the drum head to emphasize this. But if a soft beater is used, then the sound will not have such an aggressive transient. Either way, the sound can still benefit from compression. But you have to use your ears and fine-tune the settings to get the best results.

Compressing the toms

Toms can also benefit from this type of compression. However the body of the tom sound is louder compared to the transient than in the snare and kick drums. So effectively, the sound is already compressed in comparison to the snare and kick. Therefore, although this style of compression is certainly applicable, generally less compression will be used than for the snare and kick.

Summary and further considerations

What we have learned here is how to reduce the level of the transient compared to the body of the drum sound to make the overall effect fuller and subjectively louder. There are occasions, not covered here, where you might want to emphasize the transient. There is also a significant difference, not covered here, in the way you would approach compression of drums in digital and in analog recording.

How to mix kick drums: EQ settings and compression

Ever wondered how to mix kick drums? If you are thinking on how to approach kick drums in the mix, you should mostly be considering how to set the drum EQ settings and compression.

As a start, the kick drum equalizer settings and compression depends on the song genre. For example, your kick drum sounds different in rock versus jazz or country music. It is because of the way it has been approached in the audio mixing process. This short guide emphases the mixing of kick drum with strong perspective to modern pop and rock music. The suggestions below does not apply to country or jazz music, however you are free to experiment the settings in order to come up with the sound you need.

Suggested EQ Settings for Kick drums (targeting rock and pop music genre)

You need to have a parametric equalizer in order to use the following settings:

Cut -9dB Q=3 Center Frequency=400Hz
Boost 9dB Q=1.4 Center Frequency= 100Hz
Cut -6dB Q=1.4 Center Frequency = 50Hz

The purpose of having to cut -9dB on 400Hz is reduce the cardboard sound of the drum and to make the drum sounds more bass. Of course since you are mixing for rock and pop music, the heavy bass guitar sound should occupy the sub bass frequencies (less than 100Hz), so to make the bass guitar sound prominent (avoid muddy sounds with kick), the kick drum is cut at -6dB , Q=1.4 at a frequency of 50Hz.

Now to make the kick drum sound prominent among all instruments including the bass guitar, boost it at 100Hz.

Suggested Compression Settings for Kick

Personally in my own music production projects, I do not compress the kick in the mix, because compressing it can reduce its power during the mastering stage. However if you need compression settings for kick drums, below are the suggestions:

1 - Set compression ratio to around 4:1
2 - Set release time to 100ms.

If you need to read more details on compressing drums, read some suggestions here:
http://www.recordproducer.com/?a=13

How to mix cymbals and hi-hats: Tips and techniques

Some of the most wonderful sounds in drums are the hi-hats and cymbals. Without them, the song sounds dull and ugly. They define the brightness, air and level of professionalism of the produced track.

If you cannot hear the cymbals/hi-hats very clearly, or if it sounds too loud, it drastically affects the whole mix, not only the drums. Mixing them is not hard unlike bass guitars or kick drums. However, this underdog in the mix is sometimes what defines a good mix – an essential balance between bass and treble.

In recording production of drums, you need to know that there are actually 4 parts of cymbals and hi-hats used. They are:

1 - **Crash cymbal** – this is the cymbal that will be hit very hard by the drummer. See picture below:

Important mixing definition for crash cymbal:
Panning: Right, 9.375 units
Details: How to pan drum instruments, page 79

Frequency:
High pass filter at 500 Hz (means allowing only above 500 Hz to pass, cutting below).
Cut 3dB at 3000Hz, Q = 1.4
Boost 3dB at 12500Hz Q=1.0

Note: Some drum kits have only one cymbals, so in this case, it functions as both ride and crash cymbals. For mixing of this type of configuration, refer to ride cymbal tips below.

2 - **Ride cymbal** – this is the soft sound of the cymbal. As opposed to crash cymbal, this is located to the left of the drummer instead of the right.

Panning definitions: Left, 12.5
Frequency: Same with crash cymbal

3 - **Open hi hat –** this gives a crashy sound to the hi-hats, typically used for rock alternative choruses.

Panning definitions: Right, 12.5
Frequency: Same with crash cymbal

Picture of pedal and open hi hat:

4 - **Pedal hi hat-** this is the soft equivalent of open hi hat. Typically employed in song stanzas.

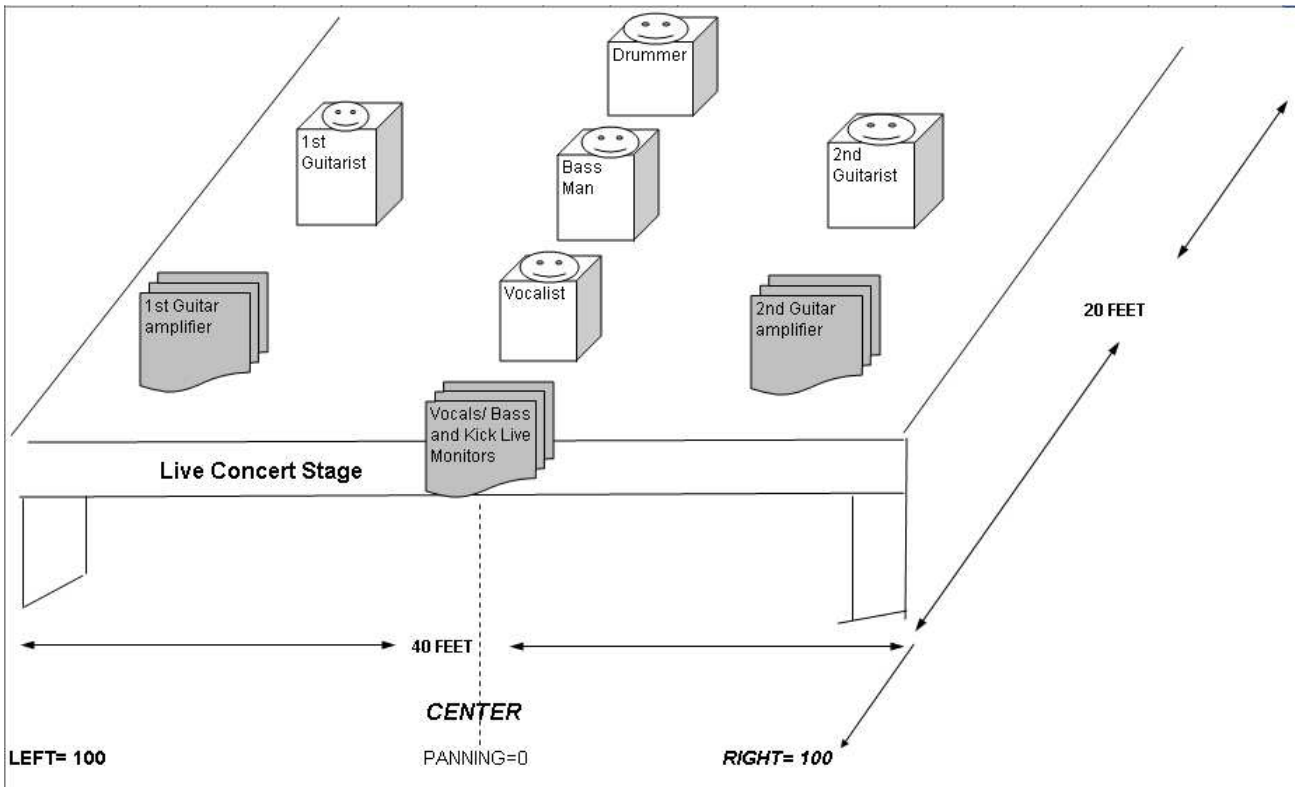
Panning definitions: Right, 12.5 (the same with open hi hat, since they are same structure, played in different style – open or closed)
Frequency: Same with crash cymbals

Overall note: **These are no strict rules, and this is more inclined to rock and pop mix. So I suggest to start with these settings and then tweak your sound to perfection.**

2.3.5 General Panning Tips

Creating Realistic Stereo Image with Panning

Panning is how the instruments are arranged within the stereo image. By properly using this mixing feature can create a very realistic mix. Mixing engineer should mix tracks with a live sound stage/concert stage perspective such as shown below:



Vocals always are in the center because the band lead singer is the star of the show. At the back of the vocals is the bass player. Typically at the back of the bass player is the drummer. The band's guitarists are on the left and the right. Bear in mind that different panning arrangement are possible for additional instruments (such as a band with piano or others). But this concept illustrates the basic things on how to do panning in audio mixing to create a very realistic stereo image.

Live sound monitors for vocals and bass are placed up front facing the audience, so that fans can get a great feel of the vocals and the bass groove. On the left and right loud sound monitors are the guitars.

In the **commercial audio production** and using a recording software, panning can be controlled between -100 and +100, where mostly -100 is the leftmost part of the stage and +100 is the rightmost part of the stage. Mixing from the real live stage perspective, a mixer can set:

a - Vocals to panning= 0 (center)
b - Kick drums = 0 (center)
c - Bass guitar= 0 (center)
d - 1st guitarist=75 (hard right)
e - 2nd guitarist= -75 (hard left)
f - Whole drum set (crash cymbals to ride cymbals)= -12.5 to 12.5

Using ratio and proportion, typical sound stage width is about 40 feet wide. Using recording software, this whole panning width is 200 panning units (length from -100 to +100). Therefore the ratio of panning units to feet is:

200 panning units/ 40 feet = 5 panning units/feet (For a 40 feet sound stage)

To check how realistic this conversion is, we will use the width of the real drum set.

Real drum set needs 5 feet width space when fully set-up.

Converting 5 feet to panning units in recording software is about : *5 feet x 5 panning units/feet = 25 panning units*, so our panning specifications are correct. This means that for a 40 feet sound stage, to create a real stereo image of drums, it should be panned between -12.5 to 12.5.

Guitarists are placed +75 to -75 respectively.

This means they are both located :

1st guitarist: -75 panning units/ 5 panning units= 15 feet from the left of the vocals.
2nd guitarist: +75 panning units/5 panning units= 15 feet from the right of the vocals.

One important thing to take note on panning is the energy level with respect to panning distance. Rule of thumb is that, the lowest frequencies should be panned on the center except for the vocals. And the higher the frequencies, the farther you can place them away from the center. Low frequencies such as bass occupy massive energies and need to be placed at the center for maximum volume.

The following are advantages of proper panning in mixing :

a - Create a real stereo image of an actual live sound stage.
b - Avoid battling the same frequencies in the same location of the stereo image.

Placing the vocals in the center of the mix means no conflict in low frequencies with guitars that occupy the same frequencies but are placed away from the center. Thus panning improves the clarity of the mix.

Developing wide stereo ambiance sound in your recordings

Professional sounding recordings are very easy to spot: they sound wide and with a great ambiance. Do you think this is a very difficult technique?
No, the technique is very easy. Think of live sound recording, or a band playing live in front of you. What do you see and feel?

Great and wide ambiance is caused by the following factors:

a - The stage is set wide compared to a human listener.
b - Stereo speakers are set wide apart (the left and right stereo I mean)
c - If there are two guitarists, having the two guitars playing will create some delays adding depth to the sound.
d - Reflections causing depths like echo or reverb.

These are the principles of creating a wide stereo ambiance in your recordings.
So how can we do this during the music production process?

The short answer is to do this right during the recording and mixing processes. You cannot make some miracles during the mastering process to create depth. Except widening the stereo, but this won't be as realistic as doing it in the mixing process.

Below are the techniques I used to widen stereo and ambiance :

a. **Double recording:**

Believe it or not, double recording is a very effective technique. To do this is to double record in left and right tracks in the stereo field.

For example: I record on the left (panned -50), then again record on the right (panned +50) same track. This will produce doubling effect and the little delay in the notes creates cool ambiance in your recording.

b. **Track doubling + Delay**

This technique is the artificial version of “Double recording”. But this artificial doubling creates reality like double recording. This is applicable if it is not possible to do double recording due to constraint in time and budget at the studio for example.

To do this, record only one track then first put it to the left (panned -50 for example). After that, duplicate that same track using your software (most recording softwares can do this), and move that duplicated track to the right (panned +50 for example). It now creates stereo (two mono sounds at -50, +50 stereo field, but no ambiance yet).

To add some ambiance, you can add delay to one mono track. The delay should be short enough just to add some space, not to create some obvious timing problems when heard by any listener.

c. **Reverb mono sources**

This is also a great effect to use. This simulates real listening, in which two mono sources are of different distances to the listener. By using some Doppler principles, it will create some delays in the ear creating ambiance and a wide stereo sound.

To do this, you need to have one completely dry track (no reverb effects), then put it to the left (for example panned -50). Then place a duplicated track to the right (for example panned +50), but put some reverb to it. The reverb must be natural, and around 500ms to 1500ms is enough.

d. **Chorus and Flanger on mono tracks**

This is similar to putting reverb on mono sources, but using some chorus or flanger effects instead.

Important: Since putting some reverb, chorus or flanger will to change the volume of the track, it is important to have both tracks at highly similar volume for this effect to realistically work, or else it will sound mono (one source stronger than the other).

How to create a sound as if it was coming from behind

The idea comes from *recording.org*, about how to create a realistic sound as if it was coming from behind. Supposing the sound is mixed with respect to a certain observer witnessing the event (missiles chasing an UFO), of course the observer (a person with two ears) is placed at the center of the event. That's why in movie theatres, the screen is always placed in the center with digital stereo system (left and right to simulate real situations).

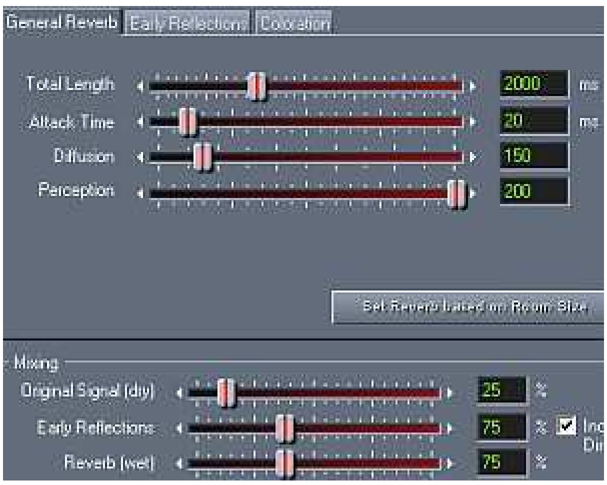
What if we have a scene with that a missile coming from left to right chasing the UFO then the missile explodes in the mountain far right? Then as it explodes, we can hear the sound of the falling rocks. To mix, assuming we use the samples of the 'recording.org' thread, we use three audio samples:

1 - The sound of the missile
2 - The sound of the explosion
3 - The sound of the falling rocks.

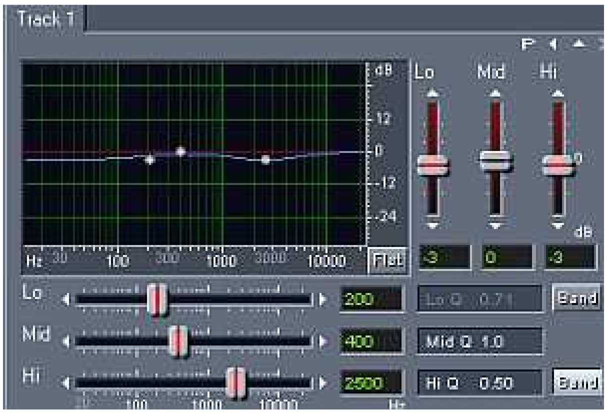
First, to simulate missiles flying from left to right, pan it from far left (-100) to right (+100) using your mixing software. Then as it flies away from the observer, the sound volume of the missile decreases. I use Adobe Audition to mix this one:

The green line is the volume line; it starts from the top and ends in the middle. The volume starts loudest then it decreases to -6dB (slanting green line from left to your right means a decreasing volume trend). Also, the pan line starts from the hard left all the way to the hard right (blue diagonal line) (to simulate a real time +100 to -100 adjustments in panning).I did not apply reverb on the missile track as I need to sound like it was very near to the observer.

Second is the explosion. To simulate that it is exploding far right, I pan 100 units right. Then to simulate like it was exploding behind. I apply reverb with the settings below:



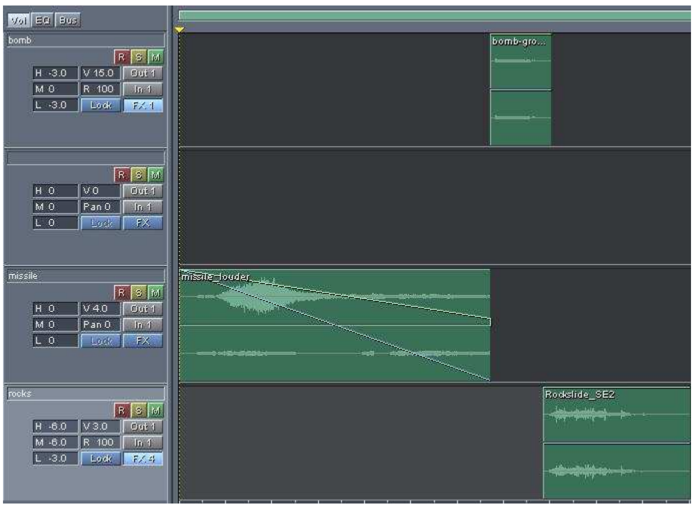
The objective of applying reverb is to add more depth to that event and to simulate it exploding from behind. I also apply some EQ to the bomb, see below for the settings:



The objective of adding EQ is to cut 2000Hz range which is not realistic to hear in actual scenarios; you hear some bass and cut mid range in actual sound.

Third, the rocks… I panned it also 100 units right because it will make sense since the location of missile explosion and rocks are the same. The reverb setting is the same with bombs since they are in the same perceptual location. I also apply some EQ to the rocks, with similar objective to missile explosion.

Here is an Adobe Audition multi-track view:



The final mix: ***CLIP - From Behind**: "audiorecording.me" >Music Mixing>"How to create a sound like it was coming from behind?"*

So what you can hear is that the missile is flying from left to right and explodes far right in the behind, with rocks falling from the mountains.

2.3.6 How to Pan Instruments

How to pan acoustic or electric guitar in the mix?

In mixing session, it is important to correctly pan the guitar in the mix. Inappropriate pannings can result to mud or unrealistic guitar sounds.

This is a short guide on how to properly pan guitar in the mix.

As a start, it is important to first learn the importance and concept of panning. Panning is the process of assigning a location of musical instrument in the stereo spectrum. A stereo is characterized by:

a. Left channel
b. Right channel
c. Center

The center is assigned as "0", while the rightmost part of the channel is assigned as +100 and leftmost part of the channel as -100. For example, if you assigned a certain musical instrument at a panning setting of "0", it means that you have assigned that instrument in the center of the stereo.

Panning is important because it is used by sound and mixing engineers to simulate live performance. These engineers vision stereo mixing as musicians playing on stage.
So it means that:

1 - The vocalist is at the center of the stage
2 - The drummer is at the center area also.
3 - The bass man is behind the vocalist and also at the center stage.
4 - At the right most is maybe the first guitarist of the band.
5 - While at the left most is the second guitarist.

The **standard panning** for acoustic or electric guitars must fall within:

Left channel: -50 to -100
Right channel: +50 to +100

And specifically:

a. If you need the guitar to be more punchy, strong and dominant, I will panned it at -50 , +50.

b. For background guitars which the sole purpose is for accompaniment with lots of instruments involved, you can pan it hard left and right (-100 and +100).

c. If there are 2 guitarist (lead and rhythm guitars), you can pan the lead guitar near to the center to give more "presence" in the mix (-50, +50) while the rhythm guitars will be placed at -100 and +100.

For more spacious, ambience and live feeling on the sound of the guitars, you can even put a delay between the left and right channel. So if the right channel is lag at 2ms from the left channel, this creates an illusion of wide stereo.

For best results with panning, you can record the guitar track twice. First place the 1st guitar on the left channel. Record the guitar track again and place it on the right channel. This creates a nice ambience very ideal for pop, acoustic and country recordings.

Pan bass guitar in stereo track mixing

One of the most confusing items to mix is the bass guitar. This short guide will teach you how to pan the bass guitar in the stereo track. The goal or objective is to have the loudest sounding bass guitar and to minimize interferences with other instruments.

As we all know, bass guitar frequencies are high energy in nature and occupies low and sub frequencies. If you need tips on mixing frequencies and avoiding mud with the kick drums, please read page 54.

New engineers who have to mix bass guitar are often not aware of the importance of correct panning. Aside from being "creative" when it comes to mixing bass, sometimes you need to know the optimal location of the bass guitar in the stereo.

OK, let's get started…

1 - First and foremost, the most important panning settings of bass guitar for rock and pop recording is in the center ("dead center") of the stereo mix. If you are using an audio mixing software, this setting corresponds to "0" in the pan settings.

2 - "0" means it is neither right nor left. It means it is centered.

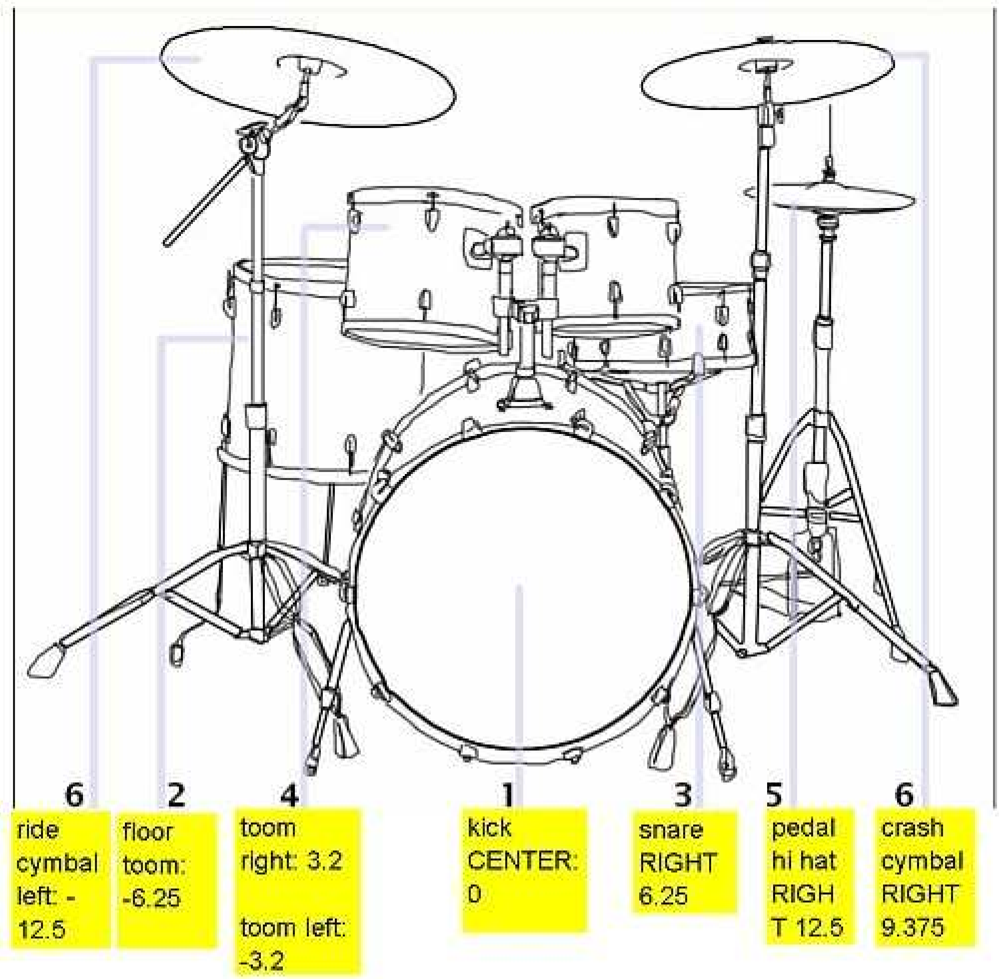
3 - Another good reason why you need to pan to the center is avoid problems in mastering for vinyl. Refer to this article for details: http://emusician.com/tutorials/mastering_vinyl/index1.html

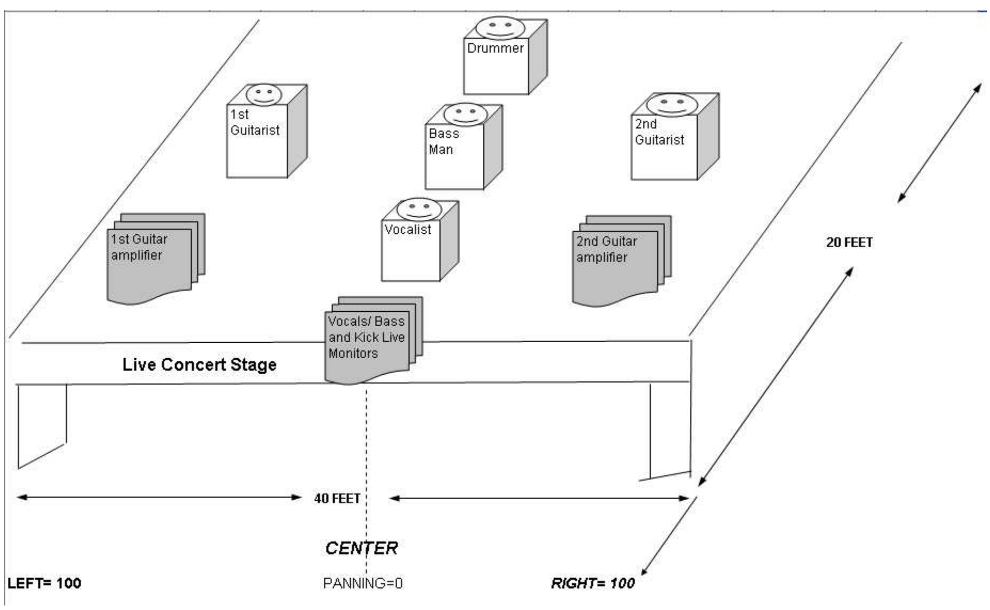
4 - If you are panning the bass either to the right or the left, consider that its energy might diminish, as the loudest sound and energy are located in the center. There are times for creative reasons that you will pan the bass to either right or left such as in jazz or experimental music.

How to pan drum instruments

On page 72, I have mentioned that the whole drum instruments can be panned somewhere between -12.5 and +12.5 settings. In this guide, I will give my approach on how to pan drum instruments and parts (snares, toms, cymbals and hi-hats).

Panning properly drums can create a very realistic drum sound in stereo.





Below are the most important components of a drum kit and it’s panning settings :

a. **Kick drum**- “*0*", this means it is panned to the center.
The reason for this is that the drum set is placed in the physical center of the stage and kick drum is in the center of the drum set. Below is how the band looks like playing live on the stage and take note the drum set is in the middle.

b. **Snare**- “*+6.25 right*”, this means it is panned very slightly to the right.
Looking at the real drum, it is located in the physical center between the hi-hats and the center of the kick drum. Some engineers pan the snare to the center, but I do not recommend this since a significant frequency response of snare can drown the vocals.

c. **Pedal hi hat**- “*+12.5 right*”, this the farthest right of the drums.

d. **Crash cymbal**- “*+9.375 right*”, located in the physical center between the snare drum and the hi-hat.

e. **Toom right**- “*+3.2 right*", again very slightly off center to the right.

f. **Toom left**- “*-3.2 left*", the pair of the other toom but this one panned to the left.

g. **Floor toom**- “*-6.25 left*", located in the physical center between the ride cymbal and the kick drum.

h. **Ride cymbal**- “*-12.5 left*", this is the leftmost part of the drum set.

So stereo drums need around 8 tracks in the multi-track recording, with each part in a different panning setting for a realistic stereo placement.

2 - Music Recording & Mastering

Mastering audio at home: General tips

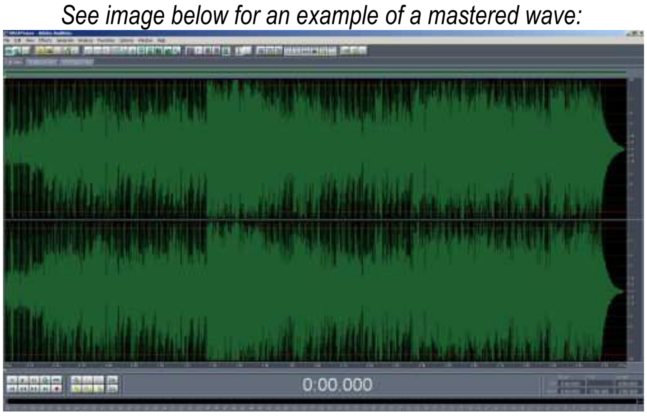
Mastering process is the last step in any music production process. It is highly recommended to master the tracks before using them for any commercial purpose. Here are the underlying reasons:

1. Consumers' audio players are different than yours, so it can sound different than what you expect it to be. Any deviation in audio will annoy normal or average listeners.

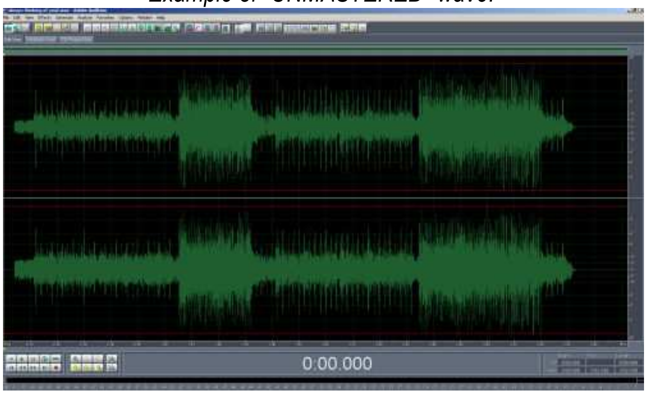
2. In the play list of any audio or mp3 player, 100% of those tracks are mastered, so if your song is on that list, it will sound very different and odd. Example is that, it will sound "weak", compared to other tracks.

3. Nobody likes to buy a CD or to download a song online if it sounds weak compared to what is "loud" for them. Listeners will have to turn the volume of the player up every time your track is played…

See image below for an example of a mastered wave:



Example of "UNMASTERED" wave:



Looking above, it is obvious; the mastered track will sound louder. But mastering is not making it loud, instead the primary aim is to make "loud" AND "beautiful". Making it "beautiful" can be achieved by EQ and compression processes.

Those are the reasons you should master tracks.
But before you can master tracks, what are the requisites? Of course, you cannot master a track if it is not mixed "properly". I emphasize this, because a poorly mixed track cannot be saved in the mastering process, so in this case it should be re-mixed again. Mixing process is completely different than mastering process.

Requisites for a good mastering:

1. Properly Mixed track (no peaks above 0dB digital), you can call this a “clipped” track indicated by red indicator in your digital audio workstation. This is not allowed to be mastered as this will give horrible results!

2. Master at highest resolution possible! Do not master MP3 version. Mastering is all about adjusting audio in digital domain so you should master at the highest resolution possible. In my case, I use the 32 bit resolution at 44.1 kHz, (this is higher than CD quality).

3. Good monitors with subwoofer, flat as possible. If you are in budget and looking for quality/flat frequency response monitor for home studio, you can have M-Audio Studiophile AV 40 Powered Speakers, this sounds great especially if you are using PC for mastering.

4. Good acoustics, know the response of your room acoustics very well.

Now for the actual show, how do you master the tracks???

1. Start with cleaning the tracks, you can remove noise, shortening gap in the beginning and ending. I recommend following CD audio red book standard for this.

2. EQ, add presence like +2dB in 2 kHz, +1dB in 1 KHz and +1dB in 4 kHz. Q is 1.

3. EQ part 2, remove mud sound, by cutting -3 dB in 200Hz (use Q of 1), -3dB (high pass filter action) at 35Hz.

4. EQ part 3, add bass and lower punch for more pop and rock sound (not applicable for jazz and country genre). If at mixing, kick is boosted at 100Hz, add +2dB in 100Hz, +2dB at 65Hz. Use Q of 1 for this.

5. Final EQ, add gloss and shiny sound, boost at 15kHz, Q =1.0, +2dB.

6. Apply low pass filter at 16500 Hz (this will pass everything below that , and severely attenuates above that mark). This is optional.

7. Compress, I use L2 Wave effects, compress so that the average RMS audio level is around -12dB for pop and rock volume average. The resulting wave should look like a mastered wave, such as shown above.

8. Save your work, and DO NOT DESTROY THE ORIGINAL MIX DOWN (unmastered track) from the mixing studio. Listen always. Use your EARS! This is the most precious studio equipment.

9. Finally criticize your work, open up your audio player, add commercial tracks similar to the genre you are mastering, and then add your mastered track…Does it sound different? Or sound competitively similar? Repeat the whole process if it fails the test. Use your ears to tweak those settings.

10. Save your work at 16bit 44.1kHz audio (CD audio quality) and use dithering.

Digital Audio CD mastering tips using personal computer

Welcome to the digital age, with all audio equipments, softwares and gears available for personal computers; any recording enthusiast might be able to do digital audio CD mastering work at home.
Mastering stage is the last stage of the music production process. The input is the completed audio mix down done by the mixing engineer and the output is the commercial (broadcast) quality version of the song, ready for marketing and album/single release.

An important hardware requirement is to have near field speakers such as M-Audio Studiophile AV 40 Powered Speakers. If you don't, you will have a hard time doing audio mastering work.

Below are the tips and procedures that will provide you a good head start in doing this type of post-production work (this is by no means a complete guide; your day to day experience might even teach you better):

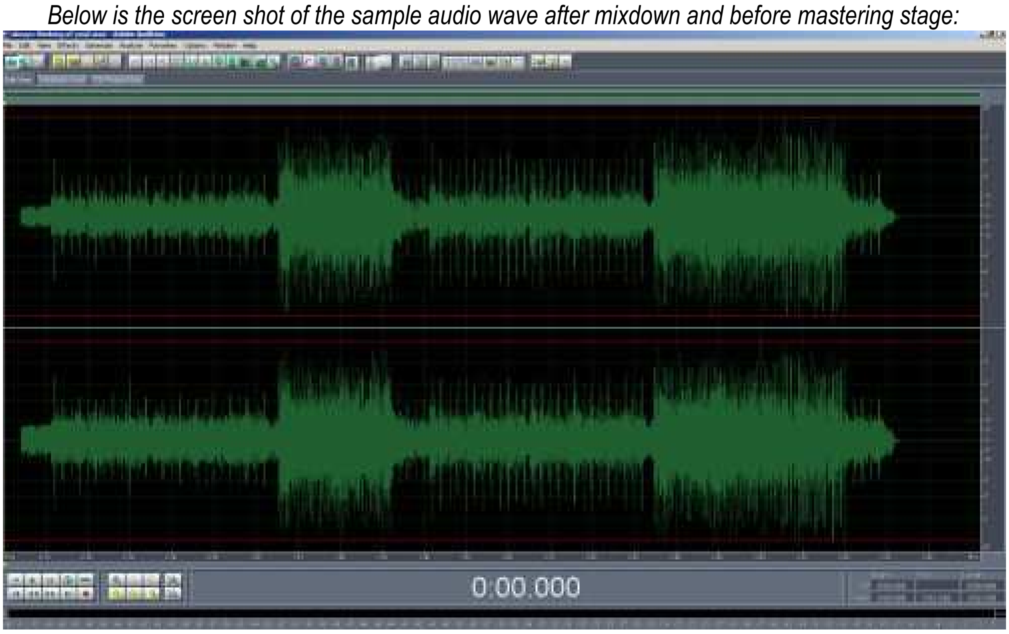
Step 1: Make sure you have a clean mix down in WAV format.

I suggest you to at least have 44.1 16 bit CD resolution where you need to work. Depending on your computer sound card, the best solution would be a higher resolution such as 44.1 Khz 32-bit or even 96 kHz 24 bit.
It depends on your mix down quality so make sure the one you have is the best possible version.

Do not master using MP3s. Instead, ask the mixing engineer to do a WAV mixdown using the above suggested resolution.

Another important thing to check if you have a clean mixdown is that the peak signals of the wave should not exceed -3dB to -1dB. This will provide some headroom or space for mastering work where you are compressing and boosting a lot of things.

Below is the screen shot of the sample audio wave after mixdown and before mastering stage:



Step 2: Start talking with the recording producers, artist or any A&R involved in the project.

Ask them for their inputs as to how the complete version should sound like. Be ready to research CD's of their reference artist. This is particularly important for commercial releases where the complete version needs to be consistent also with the targeted genre (rock, jazz or country). Write it down.

Step 3: Make a backup of the mix down, store it in your external hard drive. And load one to your digital audio workstation personal computer. Using your favourite audio software with audio editing capabilities, EQ effects, compression is enough for you to do mastering work.

A good software to use is Adobe Audition for Windows and Pro Tools for Mac. Another good software includes Cubase, Sonar, and Audacity for free audio recording software.

When it is loaded to your audio editing software it should look like the one in the above screenshot.

Step 4: Start with removing noise in the audio start and end only (silence section only not the entire audio mixdown wave). If possible cut it down to have a 0.5 seconds silence before the audio wave starts.

Step 5: Start with applying “presence”.
A good 1 dB boost (entire wave applied) at 2Khz using a Q of 0.5 is enough. Do not apply too much, it may sound “horn-like”. Presence makes it possible to have a clear vocals and instruments in poor hi-fi systems (or AM radio) or in too bassy systems common in most audio listening environments.

Step 6. Apply high pass filter at 20 Hz.
This will remove the rubble. This will attenuate starting 50 Hz all way down but passes above it. This will prevent mud in bass systems and focus energy to useful frequencies. However bear in mind that 50Hz is a very important sub woofer frequency for rock and pop music, you should not cut it drastically.

Step 7. Tailor the audio sound to something you like.
Use a parametric equalizer. Things like boosting at 80Hz for kick clarity, cutting at 200Hz for mud removal and applying boost at 15Khz is done in this stage. Again experiment and do not boost more than 3 dB in any channel or use Q less than 0.8 (recommended is 1).

Step 8. Use your compressor to maximize volume.
It should average -13dB to -12 dB rms after compression. Compare it with your reference audio. Is it too loud? Or too soft. Make some adjustments if necessary. Do not over compress. Your hearing ability and skill to distinguish what sounds good and bad is extremely important in your mastering process (this needs practice and may develop after many months and years of doing production work).

Step 9. Let your critics listen to your completed work (recording producers, artists etc) and ask them if they are happy with the results. Good luck.

Mastering audio at home: EQ Settings

Mastering is the last audio production process. It is the next process after audio mixing. In the mastering stage, the engineer does not concern with the mixing elements since he/she is only working on a single wave (the mix down).

Mastering is necessary to further enhance the mix down particularly to address these following audio quality issues:

1 - **Lack of presence**
Presence is an element that makes the music as well as the vocals clearly audible in sound reproduction equipments (like hi-fi audio equipment, your CD player, etc). If this is not addressed, other strong/dominating elements like the bass and the drums will drown the important musical elements like the vocals. Luckily, the mastering engineer can sort this issue using EQ.

2 - **Lack of punchy bass elements**
If you are producing songs in rock and pop genre, the bass elements are very important to push the song. Unluckily during the mixing process, the job is to simply avoid mud in such as way all instruments can be heard at their specified frequencies.

However, heavy boosting of bass elements is not recommended in the mixing stage and should be done in the mastering stage. The important musical instruments that are affected are the bass and kick drums.

3 - **Lack of bright hi frequency elements**
One of the common errors in independent music production is the lack of brightness with respect to high frequency elements. This makes the music too bassy and makes it impossible to capture hi frequency elements properly. So if your music includes drum hi-hats and cymbals, then you need to tweak these elements during the mastering stage using an EQ.

Now comes the time to formulate the EQ settings for mastering. You need to know how to use a parametric equalizer. The following are the settings I used (which are a good start, but it may depend on your mix down so you need to do minor tweaks):

High pass filter: 30Hz
Important: Do not apply this filter setting for rock, alternative, pop rock, pop genre as these frequencies are important). Apply this only when the song is too muddy on the sub frequency range.

70 Hz Q=0.8, 1.3dB boost	(this will boost the bass guitar frequencies)
100Hz Q=2.0, 1.3dB boost	(this will strengthen the kick drum)
200Hz Q=1.4, -0.5dB cut	(this will remove the muddiness of bass and guitar frequencies as well as vocals)
3000Hz Q=1.4, 1.0dB boost	(this will boost the vocals)
13500Hz Q=1.0, 1.2 dB boost	(this will boost the high frequencies to support hi hats, cymbals and vocals)

You might notice that the mastering EQ is more of a boost than a cutting of frequencies (frequencies were cut a lot during the mixing stage). This will balance the things up and shape the song sonic character.

Always remember to use your ear and compare your mastering to the standard produced records of the same genre (if you are mastering rock music, you can listen to rock records and compare whether it is comparable sonic quality or not).

Audio mastering tips: Doing setting screenshot

As we all know that "mastering" is the process of crafting the completed mix into a professional - commercial ready sound with its required loudness, punch, EQ, space and clarity. Mastering can be broken into the following general steps:

Step 1 - Signal conditioning
Removing of long silence before and after mix down. Also removing noise can be done in this step.

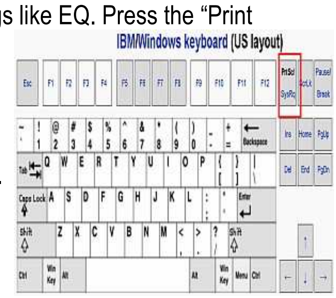
Step 2 - EQ using parametric equalizer
Purpose is to shape the sound to a certain standard, depending on your genre.

Step 3 - Compression
Purpose to make the audio wave as loud as possible without distortion.

If you are doing home mastering, particularly if you are using a digital audio software (like Adobe Audition, Cubase, Protools), things can be quite difficult and complicated. It differs a lot from real professional mastering where things can be adjusted in real time using a console.
Aside from having an incomplete set of equipments, acoustics, and gears which is common in home studio recording, it is very easy to get lost when doing mastering. This is particularly true when you are testing different settings of EQ and compression to get the "best sound". Mastering is more of less like a "home audio experiment", finding out the best settings for EQ and compression that will give the best results WITHOUT sacrificing the quality of the mix.

One of the tips I usually perform is doing a "screenshot" of the settings, an image of the settings.
The purpose of this screenshot is to SAVE the settings visually instead of memorizing or writing it which could take time and you can easily forget. In a mastering process using screenshot, it can be implemented by following the steps below:

1. Just BEFORE you click "OK" in your Digital Audio software to implement any audio settings like EQ. Press the "Print screen" button in your keyboard. See below (inside red box):



2. Open your favorite image editor. I recommend Irfanview as it is free, light and easy to use.

3. Using Irfanview/Photoshop/Paint, go to *EDIT* and click *PASTE*.

4. The screenshot will then be shown. The screenshot should clearly show the settings you have just made. You can even crop it or cut it depending on your taste.

5. Now that you have made the screenshot, save it to your desktop in a filename that can easily track your work. For example, I save it as *FIRST EQ.jpg*, to reflect my first batch of settings I have made on a parametric equalizer.

6. You can click OK on the parametric Equalizer to let your audio software implement the settings on the audio wave.

7. Now that you done with EQ, you can repeat the above procedure with your Compressor settings.

8. When everything has been completed (EQ and Compression adjustments) Listen to your mastered track and do take note carefully if you get the sound you want. You can even compare to professional released recordings at the comparable sound level to see if it pass a certain level of quality.

9. If you find the sound of the resulting adjustment to be unsatisfactory, you can open your screenshot again (*FIRST EQ.jpg* or *FIRST COMPRESS.jpg*) and see how many dB adjustments you need to make. If it seems that the sound has too much bass, I could reduce the current settings (see screenshot above) of 80Hz, 0.5 Q from 3dB to around 1dB. Then take screenshot again and listen. Repeat this process until you have perfected your audio.

Note: Remember to make a backup of your original mix down so that it won't get destroyed during audio EQ and compression adjustments.

Mastering with Cool Edit Pro / Adobe Audition

This is a short guide on how to do mastering using Cool Edit Pro (old name of Adobe Audition).
Particularly, this is written based on older Adobe Audition 1.5. But I do believe that techniques outlined here can also be applied to later Adobe Audition versions such as Adobe Audition 3.

Mastering is the last stage of the audio production process, next to mixing. So make sure that before you start mastering, your audio wave passes the following properties:

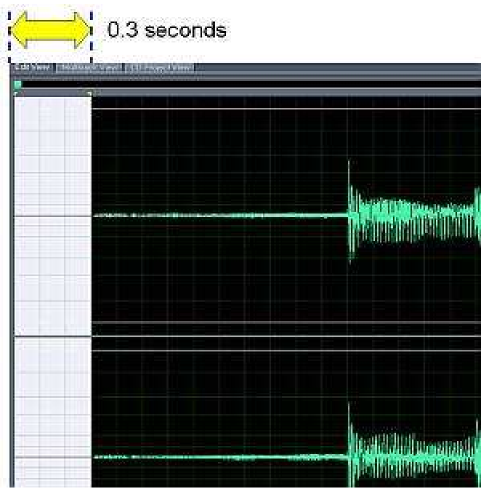
1 - **It is a mix down wave**.
This is a single waveform, which is the summation or the final result of the mixing process. It should only be one waveform but containing all musical instruments (vocals, drums, guitars, etc) mixed together.

2 - **No EQ or Compression still applied**.
It should be completely fresh, the single waveform is still NOT being adjusted with any mastering EQ and compression settings.

3 - **The maximum amplitude of the audio wave (Peak amplitude) should not exceed -1dB**.
An allowance is needed for EQ and Compression adjustments in mastering. If the wave is already peaking at the loudest level (0dB), then there is no room for EQ and Compression adjustments.
Having some room for EQ/mastering adjustments in the mixdown is a good habit to have. You can measure the peak amplitude by going to Edit View –> Analyze –> Statistics.

If your wave has been conforming, let's start the mastering process:

step1 - **Trimming of start and ending**.
I give a 0.3 second allowance before the start of the audio wave and 0.6 second allowance at the end of the wave. If the wave exceeds 0.3 second from the moment it started playing, cut it. Below is the sample screenshot of the trimmed wave at the start (it was exceeding 0.3 seconds before and I cut it to 0.3 seconds standard).



step2 - **EQ stage**.
The objective of EQ stage is to shape the final sound of the recording, apply presence, boost hi frequencies and lows as well as removing muddiness. Use the Adobe Audition Parametric Equalizer to adjust settings.

step3 - **Compression stage**.
The overall objective of the compression stage is to make the audio wave as loud as possible without creating distortion.
In this case, I am using the L2 Waves plug-in for Adobe Audition 1.5 (*Effects –> DirectX –> Waves –> L2*).
Under '*factory preset',* change to: *Hi Res CD Master,* then adjust the following: *Threshold: -7.5; Out of Ceiling: 0.2.*
This will maximize the volume of the recording to around -13dB (average RMS power) which is fine for master recordings.
You can check the loudness in terms of average RMS power by *Analyze –> Statistics*.

How to dither 24bit or 32bit audio wav to 16 bit CD audio

"Dithering audio" is the process of putting unrecognizable/white noise in a lower bit depth as a result of the conversion from a higher bit audio.

In the audio mixing process, the mixdown is mostly 24 bit audio or 32 bit which is higher than CD audio resolution (16 bit). If these are converted to 16 bit, there will be losses in the audio digital signals due to downgrading the audio resolution. These losses can be noticed in a 16 bit audio as minor distortion of the signal. Bear in mind that since dithering is adding "white noise", this strategy is more preferable than not applying dithering at all which results to "distortion" (or we can say that white noise is more preferable than distortion).

By applying "noise shaping" techniques you can further minimize the effects of "white noise" on the converted audio. Noise shaping will basically place "white noise" away from audible frequencies (those that can be heard easily by humans). This will be covered also in this guide (see below).

Below are some of the important steps in how to dither 24 bit or 32 bit audio to 16 bit CD audio.
The steps illustrated below are using Adobe Audition, although similar procedures and functionalities can be done in most audio recording and mixing softwares.

step 1 - Open the high bit depth audio in the Adobe Audition Editor view.

step 2 - Confirm the audio resolution by going into *View — Wave properties*.

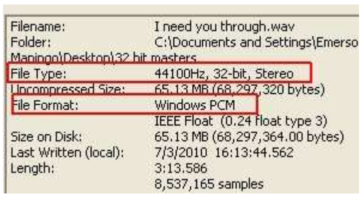
step 3 - Close the file in Adobe Audition Editor view.

step 4 - Go to *File — Batch processing* – Click "*Add Files*" — navigate to the audio file location in your hard drive and click "*Add*".

step 5 - Skip "R*un script*" but click "Resample" tab.

step 6 - Make sure "Conversion settings" has been checked.

step 7 - Click "Change destination format".



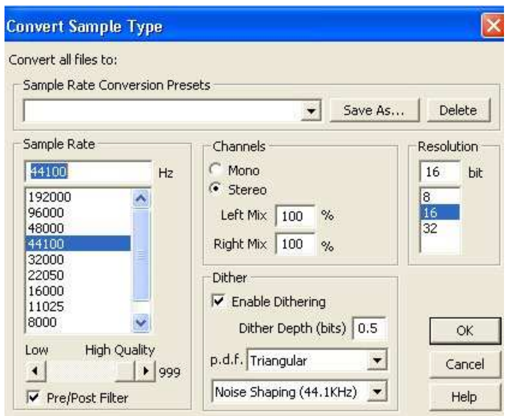
step 8 - Configure the following in the "Convert Sample type" menu.

Sample rate=44100
Channels = stereo
Resolution = 16
Pre/post filter = checked

step 9 - Check "*Enable Dithering*".

step10 - Dithering settings:

Dither depth = 0.5
p.d.f = Triangular
Noise shaping = 44.1 KHz



step11 - Under "Output format" select WAV file and 16 bit (44.1Khz).

step12 - Select where you would like to save your converted audio file.

Sample Rate Converter - Evaluating conversion quality

What is a sample rate converter in audio applications?

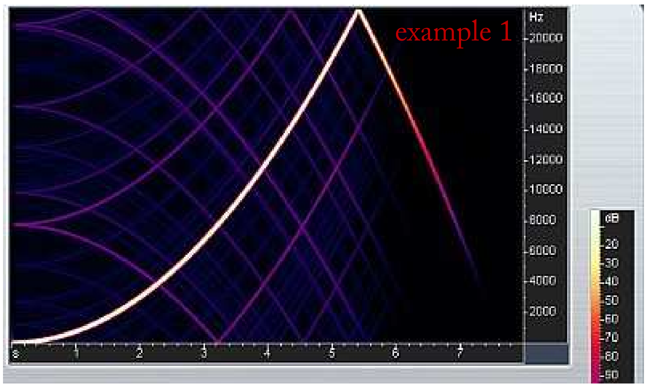
It is used when converting sampling rates in digital domain. For example, say you are recording audio in 24 bits 96Khz. 96Khz is the sampling frequency/rates. If you plan to down sample it to 24 bits 44.1 KHz then you need an audio sample rate converter. This is a very important mastering and mixing tool because sound engineers often work at a high fidelity environment and then down sample when finally exporting it for mass production (for example CD audio projects which is using 16 bit 44.1 KHz).

Audio sample rate converter is not a dithering tool, although most converters allow also reducing the bit depth of the recording. Bear in mind that if you down sample a 24 bits 96 KHz into a standard red book audio CD at 16 bit 44.1 KHz, you need both a dithering tool and an audio sample rate converter. The dithering tool is used to downgrade the 24 bits to 16 bit audio and then filling it with dithering noise such that normal listeners won't notice (while not changing the dynamic range, so it still sounds like a 24 bit audio). The sampling rate converter is used in changing the sampling rate frequency from 96 KHz to 44.1 KHz for example.

One of the best audio wav sample rate converter is Voxengo R8Brain. Not only it does perform very well but it's free and easy to use. You can download it here: *http://www.voxengo.com/product/r8brain/*

So how do you know that an audio sample rate converter is indeed performing well?

In the study of digital signal processing, an imperfect audio sample rate converter will add undesirable artifacts to the converted audio wave that affects the resulting audio quality. The good thing about this SRC (sample rate converters) is that they can be tested for quality purposes. If you visit this website: *http://src.infinitewave.ca/* , you will be able to read the frequency response curve of the converted wave using different sample rate converters and check if there is an artifact/aliasing introduced in the process.



For example, '*example 1*' is not a good sample rate converter frequency sweep response curve. There is too much aliasing frequency (artifacts generated) in the sweep signal from 0 to 48,000 Hz. The artifacts(aliasing signals) are the background cobweb like colors you can see. Some of these artifacts are around -60dB which can still be considered as a noise.

The sample rate converter of Voxengo R8Brain free (sweep frequency signal mode): *'example 2'*. As you have noticed, the only present signal is the original converted signal and there is no aliasing or artifacts introduced in the sample rate conversion process. You can see that the background is pure black (no cobwebs) indicating the entire absence of aliasing frequencies.

You may have noticed that the Voxengo R8brain low pass filter successfully attenuated the frequencies above 22000 Hz, because they cannot be represented in a 44.1 KHz format. Unlike the previous graph where you can still hear artifacts introduced after 22 KHz, that is one of the audio rate converter imperfections.
More of the test details and explanations are available here: *http://src.infinitewave.ca/help.html*

2 - Music Recording & Mastering

Test bass sound frequencies for your Subwoofer or headphone

One of the most difficult things to check is the bass frequencies of your studio related monitors (whether it is near field, subwoofer or headphones). If you are mixing or mastering audio in your home studio, it is highly recommended to be able to hear and feel the sub-woofer frequencies from 20Hz to 70Hz and then the rest of the bass frequencies from 70Hz all the way to 200Hz. Precisely the bass frequencies are commonly assigned from 20Hz to 200Hz range. These frequency ranges are audible by the human ear, as we know that the audio frequency range is from 20Hz to 20,000Hz.

If you have monitors in your home studio that do not have a sub-woofer set, then things can get very complicated because you will not be able to mix the sub-woofer bass frequencies properly (even impossible) which are highly critical in pop and rock genres. So if ever you have purchasing or acquiring a studio monitor but do not know how to check whether it can handle or play subwoofer frequencies, you can follow these procedures to know whether your monitor is enough, or if you need to buy a new sub-woofer system:

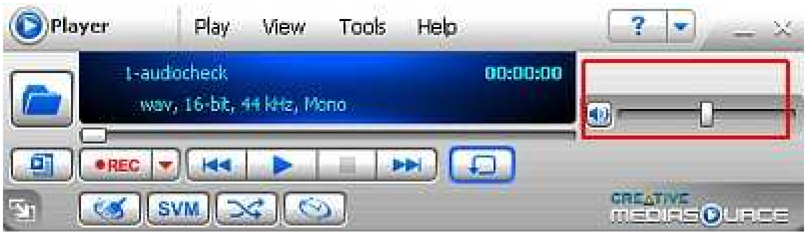
step 1 - Download the low frequency bass audio test file here:

http://www.audiorecording.me/audiosamples/audiocheck.net_frequencychecklow.wav
Source of file: http://www.audiocheck.net/audiotests_frequencychecklow.php

It is a wav file, and the file size of that test audio is around 3MB.

step 2 - Turn on your monitors and configure the following:

a. Set the audio player you used to average volume (for example if the volume scale is from 1 to 10, set it to 5).



b. Turn off any equalizers you are using. You should set everything "flat" or no EQ applied.
c. If the monitor has a volume control, set to average volume (turn the volume knob to its middle setting for example).
d. Load the test bass sound audio file that you just downloaded to your audio player.
e. Now start playing, a voice over of a person will mentioned the frequency to be tested. It starts with 10Hz and ends with 200Hz.

How to interpret the test results? You do not need a subwoofer if:

a. You start to feel 20Hz when being played even it sounds weak it is OK.

b. 45 to 65Hz should be STRONG enough to be considered loud in your monitor when being played. If during the playback of these frequencies, you hear it VERY weak, you need a subwoofer.

c. Normal monitors (like the hi-fi ones, no-subwoofer included) should play the bass frequencies of 80Hz and above really well.

If during the playback of 80Hz to 100Hz and it still sounds really weak, you even need a new monitor that can able to reproduce bass frequencies really well.

How to view the frequency spectrum of an Audio Wave

If you are dealing with frequency analysis of the audio wave “quantitatively”, one of the best tools I have learned in my signals/engineering class is FFT or Fast Fourier Transform. It is an algorithm to compute the Fourier transform equivalent of a time domain. This highly advanced technique is very simple to understand, it simply converts a time domain function into a frequency domain function.

After audio mix down (where all sounds of the instruments are cohesively combined into a single wave), the song is represented as a time domain function as we can see that the x – axis of the wave is using a time element in hours: minutes: seconds (only minutes and seconds are used realistically). See screenshot 1.

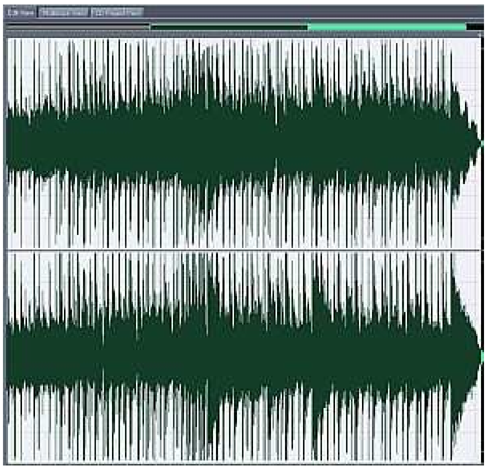
But time domain graph of the audio wave specially used during the mastering process of the track is simply a plot of amplitude (y-axis) versus time. Obviously you cannot see the frequencies of that wave. It is why we used FFT (Fast Fourier Transform) to convert this time domain representation into a frequency domain plot. With frequency domain, you can analyze the amplitude (y-axis) versus Frequencies.

Though, it is highly recommended to stick to your ear when working with commercial audio production however you can see a glimpse of the frequency response of the track. For example, if you like to check if the uppermost frequency ranges (20,000Hz) has been already filtered. It is very hard to detect it using a human ear. In this case, you need a spectrum plot.

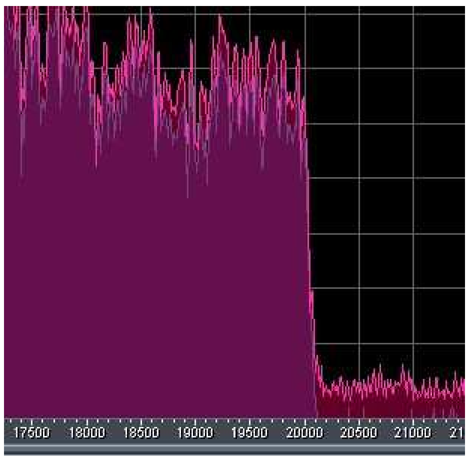
Using Adobe Audition, you can open your wave in edit view, and then click “Analyze” –> Show frequency analysis –> Set the FFT size to a maximum, as this increases the resolution and accuracy of the frequency plot. Set the type to default (Blackmann – Harris) and there you can see the frequency response graph. See screenshot 2.

As you can see, it is confirmed that that a sharp low pass filter has been implemented with cutoff at 20,000 Hz (see plot above). This low pass filter can be implemented using Butter worth.

1



2



This is just an example of the wonderful application of FFT algorithm to see the Frequency spectrum. I won’t go into details as it becomes technically difficult to be understood by common readers; because of the extreme use of mathematics.

For students, please check this out :
http://en.wikipedia.org/wiki/Fast_Fourier_transform
http://en.wikipedia.org/wiki/Discrete_Fourier_transform

Why you should not mix using headphones

Mixing is sometimes misleading newbies into using headphones to mix track. They always see pictures of studio professionals using headphones, and they might think these engineers are actually using headphones in mixing tracks. The truth is: engineers do not use headphones when doing a mix, instead they use some monitoring systems called “near field” studio monitors.

Below are the main disadvantages when using headphones and it is why engineers are avoiding it:

1 - Headphones will not substitute for the sub woofer which outputs very low/sub frequencies.
This means that if you are mixing using headphones, what in fact is very muddy in most monitors/speakers may sound less bassy.

2 - The frequency response of headphones is not the same as those studio monitors.
This means that there are times when you are working on a specific frequency and you may end up overestimating it, because your headphones are poor in replicating it.

3 - Flat frequency response is highly essential for mixing success.
Headphones do not substitute real studio monitors like near field which have “flat frequency response” such as shown below:

4 - Mixing using headphones causes problems in the interaction with the environment.
The studio environment plays a role in shaping the sound. So engineers make necessary adjustments in order to blend the complete mix with the surroundings. If you are using headphones, sound will be biased only to the environment inside your ear.

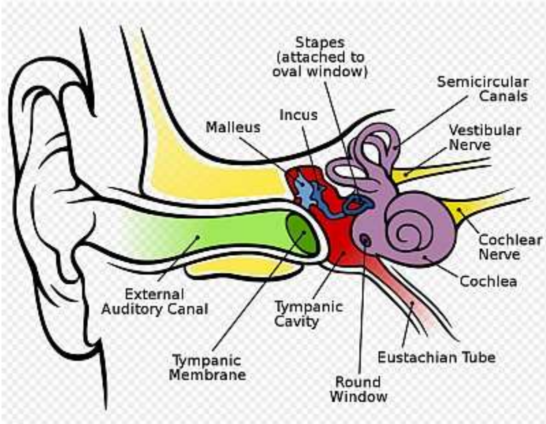
Recommendations:

If mixing is your hobby or might actually be your future career, I suggest buying near field monitors.

It’s a bit expensive but it’s worth the price. You can actually to mix like a pro. Stop using headphones because they are only used to check the results of the mix down and the mastering process to make sure the panning, ambiance and everything is cohesive for those headphone listeners. It is also used to check for small noise within your recording as they are sensitive.

How to take care of your "EAR" when mixing or mastering

Your ear is the most precious recording studio instrument. Without your ear, you cannot mix, master or finish your audio production projects. Sometimes when you are unaware, too excited to sit down and mix the track, you have a tendency to overuse your time which in turn stresses your ears too much.



A stressed ear cannot properly listen very well. This means impairing your judgment and hurting your ear. Ear is sensitive to certain frequencies, such as 20Hz to 20000Hz which are the only frequencies our ears can listen to. Again, ear is very nonlinear, it does not evenly hears it all. Instead, it is more sensitive to what is called "voice frequencies" which is around 300 Hz to 3000Hz. This is the bandwidth of the telephone also.

However in the high fidelity mix, you are expected to listen very well 50 Hz to around 15, 000Hz. Of course, the lowest frequencies (~50Hz) are the bass while the uppermost (15,000Hz) are treble, tweeter sound. These extremes in the hi-fi frequency range are the most sensitive when our ears are stressed. For example, if the ear is very tired, it cannot listen properly the hi-fi frequencies. Or even if you are always abusing your ear , like listening to a louder volume always, you will lose your high frequency reception earlier than what it is normal. It is why, old people cannot properly listen high frequencies because their ear cells are already old or damage.

So how can you take care of your EAR?

1. Listen to at most 3 to 5 hours. 5 hours can be stressful to others, but I consider this as my personal maximum limit.
2. Listen only at loud volume in mixing when necessary. Most people will listen to produced/commercial music at even low volumes, so you should not be mixing in a very loud volume. It looks unrealistic, at the same time , damaging your ear.
3. After doing mix, rest your ear by taking a nap or avoid loud sounds.
4. Clean your ear always with cotton buds.
5. Avoid ear infection.
6. When you are listening to music, limit or avoid the use of headphones.
7. Wear earplug in a loud environment. This may be either a public or working environment.
8. If you are fun of swimming in sea, avoid doing deep dives. Pressure can hurt your ear. Of course, you are a sound mixer not a scuba diver.
9. When taking a bath, avoid having some water to get inside your ear canal. This can lead to an infection.
10. When taking medicines, ask the doctor if this can have some side effects to your ear. Avoid taking medicines with such side effect to your ear.
11. Avoid someone blowing your ear. This is painful.
12. When you are contacted with cold infection, cure it as early as possible to avoid running nose and other complications to go to your ear parts.
13. If you are mixing daily, I suggest you will not expose your ear more than 3 hours per day. This will increase its life.

Producing drum tracks without a drummer

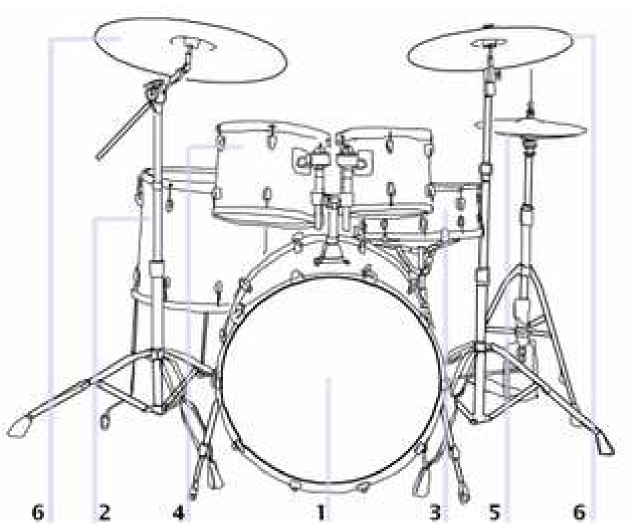
I have successfully produced drum tracks without a drummer. And the good thing is that the tracks are still playable live with a real drummer. It is just recorded without a drummer. With these type of set -up I get these following benefits :

a - No need to spend long hours in recording studio which translates to big recording budget.
b - No need to argue with a hard headed drummer.
c - No need to risk out of timing with the drummer.

Unlike other drum sequencers, I think this one produces an excellent sound. Try to hear one of my produced tracks using this type of drum sequencing method:

(" Sure" by Jeanine Maningo)
(Song Publisher info: www.musicforlicense.net)
TRACK - Sure*: "audiorecording.me" >Recording and Mixing Drums>"Producing drum tracks without a drummer"*

It is now hard to distinguish the difference in sound between a real drummer and this type of approach. Also, it will be 100% in time because the sequencing is software driven.



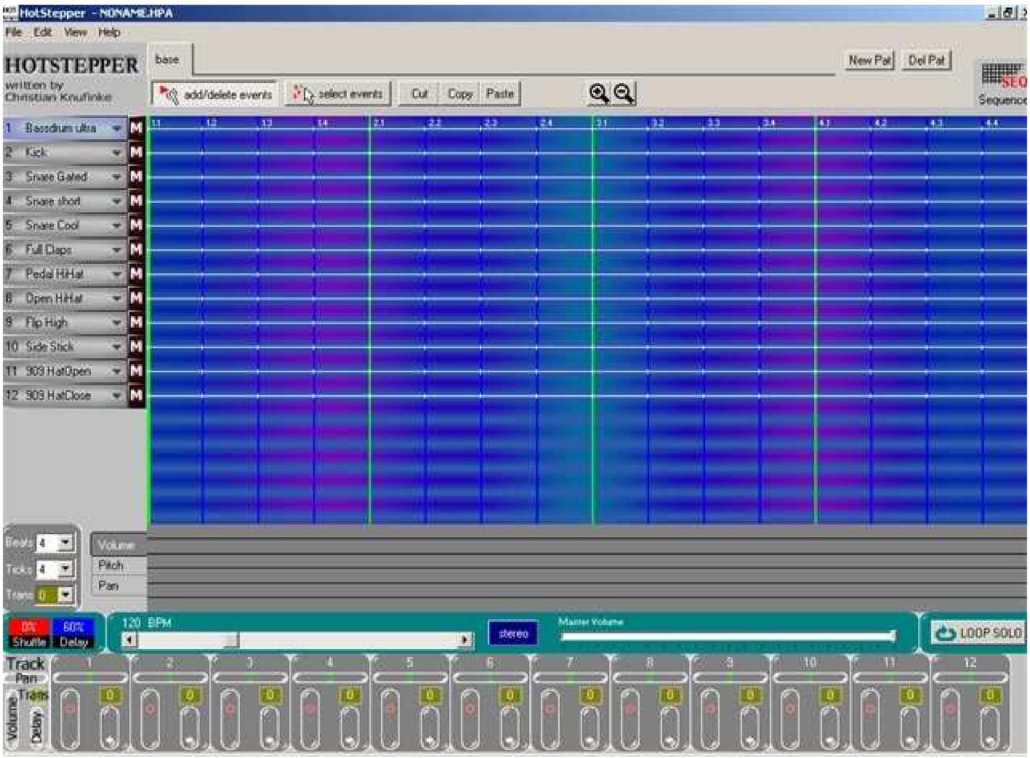
Above are the basic but complete parts of a real drum kit. The important parts are:

1. Kick drum
2. Floor toom
3. Snare
4. Left and Right Mid tom.
5. Open/pedal hi hats.
6. Right (Crash cymbal)
7. Left (Ride cymbal)

All of these are produceable without a drummer, thanks to the Hotstepper Drum Sequencer. This is a freeware-shareware courtesy of Christian Knufinke. I know he is not fully aware of the full potential regarding this open source solution.

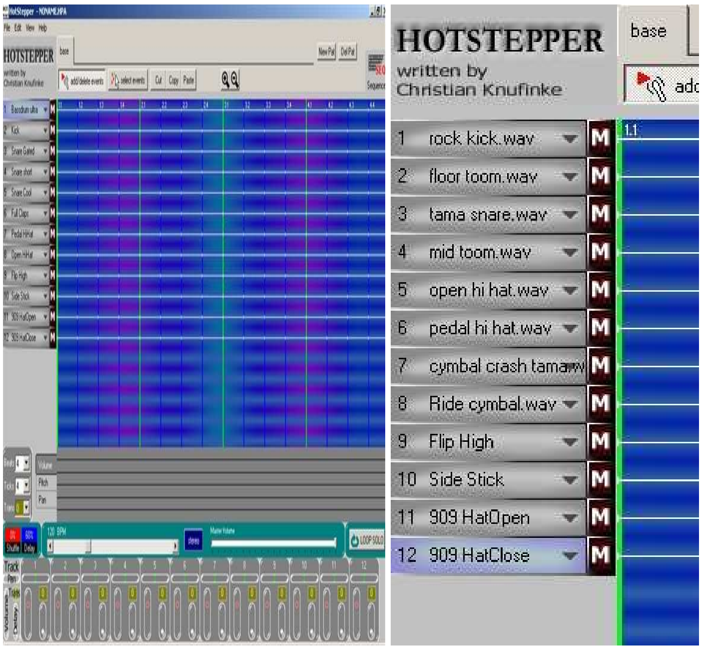
Please try to download Hotstepper, so you can try the demonstrations below. To open the archive, you need a RAR archiver. You can download the excellent freeware 7-zip.

Hotstepper Drum Sequencer :



Above is the view of a default hotstepper program. To open hotstepper, open the archive and click hotstepper.exe to run the program.

You can then see the default drum parts settings:



Hotstepper can play audio in a programmed sequence using its audio file library of real recorded drum tracks. You can record you own drum sample and name it using the following…

Take note these are real drum sounds beaten with real hands, not the midi sounds that will produce the common "demo" sound. The respective audio file names in the hotstepper library corresponding the real drum kit parts are:

a. Kick drum- "rock kick.wav"
b. Floor toom- "floor toom.wav"
c. Snare – "tama snare.wav"
d. Left and Right Mid tom."Mid toom.wav"
e. Open hi hat- "Open hi hat.wav"
f. pedal hi hats. – "Pedal hi hat.wav"
g. Right (Crash cymbal)- "Cymbal crash tama"
h. Left (Ride cymbal)- "Ride cymbal.wav"

Then change the following default settings to real drum sound mode (the default is midi):
Change default to:

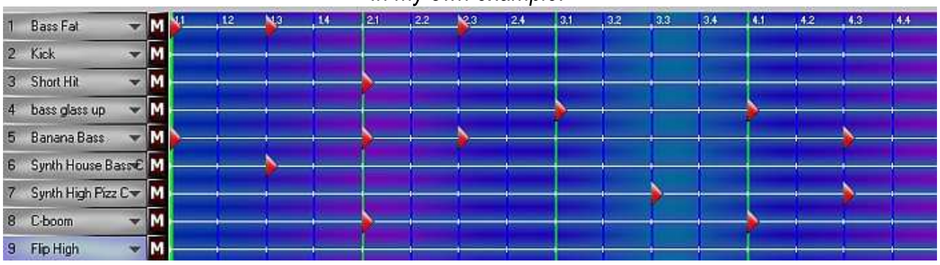
1. "Bass drum ultra" at No.1 to "rock kick.wav", you can change by selecting the arrow down besides the mute "m" symbol.(see screenshot below)
2. "Kick" at No.2 to "floor toom.wav"
3. "Snare gated" at No.3 to "tama snare.wav"
4. "Snare short" to "mid toom.wav"
5. "Snare cool" to "Open hi hat.wav"
6. "Full Claps" to " Pedal hi hat.wav"
7. "Pedal hi hat" to "Cymbal crash tama"
8. "Open Hi hat" to "Ride Cymbal.wav"

After changing the default to this, you should get a proper real drum sound.

Create hip hop drum loops with Hot stepper and Adobe Audition

Wanna produce hip hop tracks? I am not a hip hop producer but I can teach you the basics on how to create hip hop beats using Hot stepper and Adobe Audition. Launch the program, then experiment with your favorite drum loops right directly in the Hot stepper. You can use the mouse to create your own nice musical arrangements.

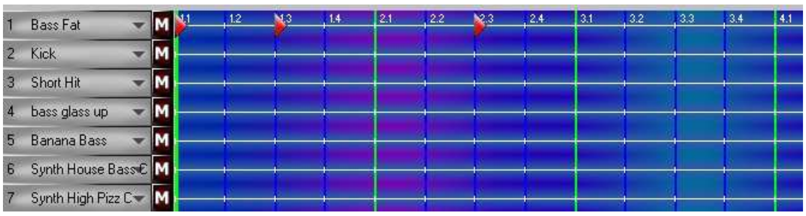
In my own example:



Do not forget to save your file. Click File – Save as –, then type the file name; in my example I name it as mylooops.hpa. The file extension should be there: *.hpa

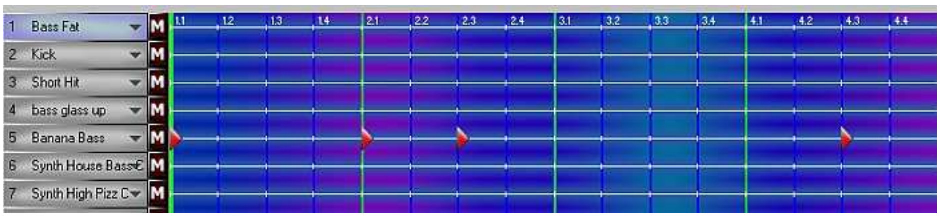
Secondly, export EACH track to WAV.

For example if I exported “Bass Fat” track, I would remove all of the tracks first (by right clicking on red buttons on the hot stepper sequencer) EXCEPT Bass fat. DO NOT SAVE THE FILE AFTER THIS or else your original arrangement will be overwritten. See screenshot (exporting Bass Fat):



To export, click File – Write pattern wav – {name of your wav file}

To export another track (example Banana Bass), go to File – Open, and in the warning: “Do you want to save before continuing?” select NO. Then find your original mix/arrangement that you have saved, and then open it again. Remove all red buttons except for Banana bass:



Do the same procedure as above to export the rest of the tracks to wave files.
Then open Adobe Audition, and load up each track in the multi-track session (one track for each wav file). You can then use your mixing software clip duplication feature to make several copies of your original loop.

Listen to this sample demo produced using the steps above:
***MIX - Drum Loops**: “audiorecording.me” >Recording and Mixing Drums>”Create hip hop drum loops with Hot stepper and Adobe Audition”*

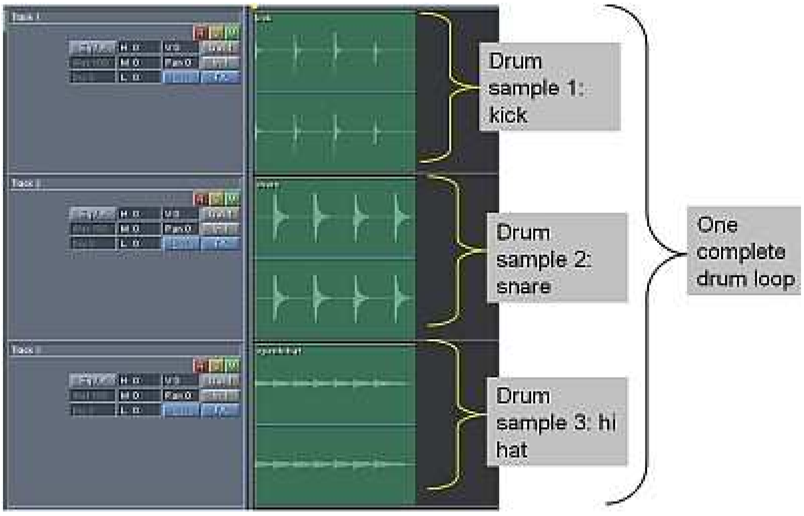
How to use drum samples in Adobe Audition multi-track

OK, say you are using drum samples to power the drum tracks of your produced song.

First, let us define **what a drum sample is**:

A drum sample is complete clip of a certain piece of drum parts. In most cases, a drum track can be define by the kick, snare and hi hats for most of the time, so we can say that there are 3 basic drum samples needed to define one complete drum loop.

Using Adobe Audition multi-track software, you can assign each of those drum samples to its own specific track so that you can mix them better. For example, assign track 1 to kick drum sample, track 2 to snare drum sample and finally track 3 for hi-hat drum sample.



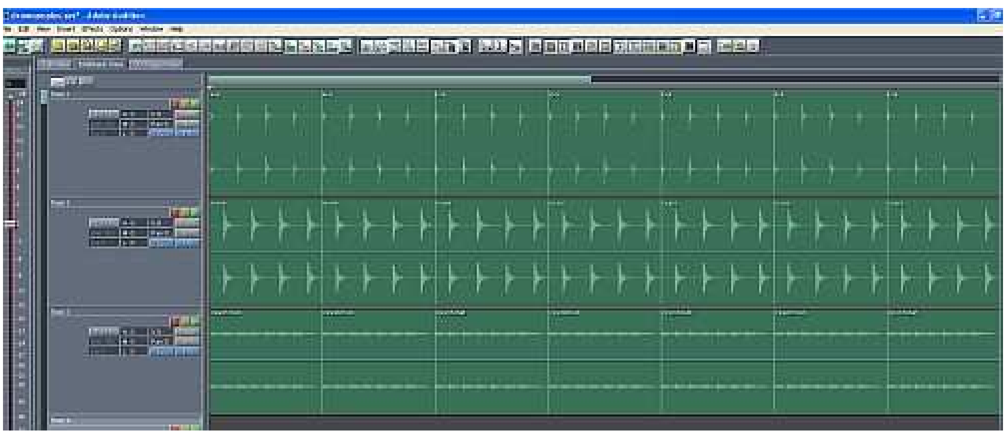
So when the time comes for mixing, it will be very easy to apply EQ/Compression/effects to each of those drum sample tracks.

What if you asked the question, **how do I create drum samples**?

There are lots of ways to do this and it depends on your drum sequencer/software. I am using Hot stepper, and you can read the procedure here to create drum samples:

The main objective of your drum sample is just to create one complete loop. You will have no problem once one loop has been loaded into the multi-track as shown in the screen shot above. You can easily use the Adobe audition clip duplication feature to replicate one loop to create many.

After clip duplication of the drum samples, you track will now look like this:



3 – Recording Equipment

3.1 General Tips

How to record music at home: Things you need to prepare

Do you seriously plan to record and produce music at home? Below are the details of what you need:

1 – Personal computer

First but the most important thing that you need is a very powerful personal computer. If you use Mac, consult them to provide a hardware specs suitable for audio recording. Then same thing for Windows PC. Invest in latest CPU, maximum RAM size for your motherboard and a lot of hard disk space. You'll find more details on page 15.

2 – Soundcard

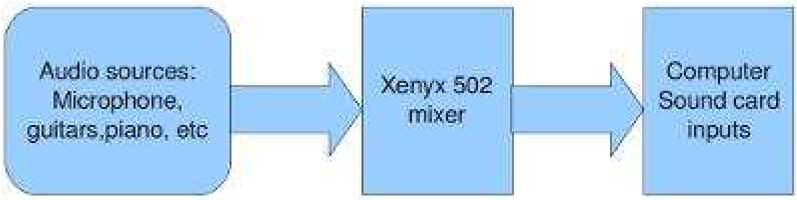
Secondly, do not forget the sound card. This is the most important piece of hardware in your computer for recording. A Creative Audigy sound card is enough but if you can afford a more expensive professional soundcard, it's better.

3 – Mixer

Buy a mixer, small is enough for home studio applications. I personally use a Xenyx 502 mixer.

4 – Connections

Configure both your mixer and computer together by connecting the audio cables. Read some very ways to record sound to your computer.
There are lots of options available, select those that are technically easy for you to setup. Currently, I find this configuration easy for beginners:



5 – Instruments

Secure additional hardware and music instruments. You should have your own instruments like guitars, keyboards, cables, effects, and so on. Then try to purchase a microphone stand. It is really helpful for vocals. Another thing you need is a decent microphone. It does not need to be super expensive. At home, I even prefer a Shure SM 58 or SM 57 because they are affordable yet great in performances.

6 – Software

Now that the hardware has been configured, it is time to focus on your software. Try to buy a decent recording software, master it, love it and do not ever switch to another software. This is my advice, mastering your software is a very important asset or skill to produce great recordings even at home. Switching to other software or even hardware can decrease your effectiveness in producing great sounding material. It is because you cannot attain mastery in a specific set of recording software unless you devote a lot of time using it. Bear in mind that these recording software have a lot of plug-ins and effects that you need to be familiar with, in order to tweak your recordings to sound great.

Advice for beginners: This is not about using hi-tech expensive recording gears, it is all about your "EARS" and years of experiences using your own original gears (it does not need to be expensive).

7 – Settings

Configure your software to accept recorded sound from your hardware. It all depends on your software. Try to read the manual as it provides tips in configuring it for best results.

8 – Seek more information

Visiting recording websites, you can learn a lot of ways about home recording , mixing and mastering. Start slow but learn well. Rushing too fast makes you less effective, because recording is all about "capturing the best performances"- more of trial and error strategy, professionals even do this. It takes patience.

Simple home studio setup: How to build your own

So you want to build a simple home studio? The simplest you could imagine involves a very little cost compared to professionally made studios. Of course if you are talking about being “simple”, it does not directly mean “poor quality”. In fact, you can even compete with those recordings produced by professional studios if you REALLY know what you are doing.

Let me outline what are needed in a simple home studio setup and the estimated costs:

1 - **Windows personal computer** (at least 320 GB hard disk, 4GB RAM) and using the latest processor – Cost: $289
Go for a computer with faster processor, higher RAM and disk drive space because you need it a lot during recording. A good example is using Dell computers: Dell 3.0 Ghz. Super Fast GX Computer, Gigantic 1 Terabyte (1,000 GB) Hard Drive, 4GB RAM

2 - **PCI soundcard**
One of the best soundcards, producing high quality recorded sounds for entry level production work, is the Creative Labs SoundBlaster Audigy 2 Value 7.1-Channel Sound Card. This is recommended if you are in tight budget – Cost: $150
As well as the M-Audio – Audiophile 2496 Audio Card (US41500C) which can cost around $90.
IMPORTANT: You need to ensure that your Windows PC supports these types of sound cards. Consult with their manual.

3 - **Behringer 502 5-input mixer**
This is the most recommended mixer to start with. Most of your recordings will be done in multi-track, so you need to record only one instrument at a time. So this mixer (with five inputs) can do this type of job. Cost- $45.83

4 - **Shure SM58S vocal microphone (with on/off switch)**
Of course, a dynamic microphone can do a professional job in recording vocals. Just make sure, there are no other distracting noises within your home studio and make sure that there no birds chipping and no dogs barking. Cost – $104

5 - **Music recording software**
Your software will be the one you will use to record, mix and master tracks. So it is very important. If you absolutely need a free recording software, you can try Audacity, but since it is open source, it does not have the full mixing, recording and mastering capabilities like other commercially made recording softwares.
For the best learning experience and a good way to start, use a professional/commercial recording software like Audition, Sonar, Cakewalk, Pro tools. But again, I am a fan of Adobe Audition which I use since 2004. Cost -$328

6 - **Guitars, bass and piano**
You absolutely need them if you need to be fully equipped. Again this will be a very long list. But hey, where is the drum? You can just go to a professional studio then record each of the drum sounds (snare, cymbals, kick etc) and just use a drum sequencer like Hotstepper: *http://www.threechords.com/hammerhead/hotstepper.shtml*
Load all the original sound drum samples in that sequencer and you have drums already for your own recording projects. Of course, you can buy a real drum with 3 additional mikes, but it won’t be “simple” anymore in my opinion, you are looking for a “complete home studio setup”.

7 - **Studio monitor system**
Some near field studio monitors/speakers are very expensive but for a start I recommend:
M-Audio Studiophile AV 40 Powered Speakers

All in all, assuming you already have guitars, amps, bass guitars as well as other musical instruments. The estimated total cost to build a simple home studio setup is around $1000.

At this price, your studio is capable of producing professionally made recordings. Once you have the gears needed, it is now time to fully master the art and science of recording, mixing and mastering sound. It will take long because your “ears” need to fully adopt professional sound. Always have your work be criticized by experts (whether as a song review or through a demo submission) because this is where you start to grow. This criticism is very important in order to train your “ear” which is undoubtedly the most important piece of recording equipment in your home studio.

3.2 Hardware

How to select a soundcard for home recording studio

If you are planning to create a home studio setup, then the computer that will be used for recording needs a decent sound card. So how are you going to select or buy a sound card for recording purposes? Below are the characteristics of a good sound card for recording studio purposes:

1 - **Can create high resolution recordings using analog to digital converter**
Take note that if you are recording, the signal is analog. However if you are planning to bounce it to your hard disks, then the analog will be converted to digital based recordings using a sound card.

Make sure that the sound card can record the analog signals at the highest possible recording resolution. Below is a short list of recording resolutions used by common sound cards and their remarks to quality:

a. *16 bit 44.1 Khz* –> this is considered to be a low resolution recording. This is not recommended for high quality recording purposes. Most onboard sound cards have this maximum recording resolution.

b. *32 bit 44.1 Khz* –> this is considered “OK” when it comes to recording resolution. Creative Audigy soundcards have this resolution.

c. *24 bit 96.0 Khz* –> this is the best recording resolution recommended for home studio purposes. The Audiophile 2496 has this resolution. Of course the higher resolution the better, but you will end up consuming a lot of hard disk space for this.

2 - **Cheap price**
I recommend not to buy a sound card more than $150, because you can buy home recording studio soundcards for less than $100.

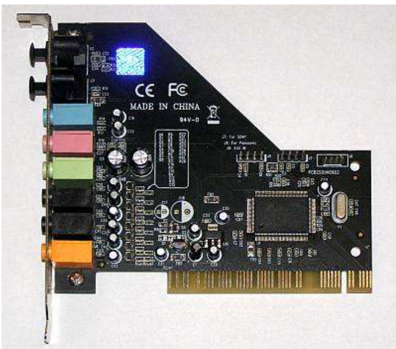
3 - **Flexible in different operating systems**
It would be nice if the sound card was supported not only in Windows but on other platforms as well, such as Mac, Linux, etc. This would offer more flexibility in the future if you chose to buy another computer (so you can reuse your soundcard).

4 - **Supported by many recording softwares**
Beware of buying sound cards that will only run in Protools or specific recording software. No one knows what software you will end up using. I recommend researching in advance for the different recording software that the sound card will support.

5 - **Running at least in 2.1 but supporting other audio formats as well**
If you are mixing audio for commercial CD production. Then a 2.1 output is fine. However it would be ice if you could get a sound card that can play 5.1 or more.

6 - **Highest SNR as possible**
The minimum acceptable level is 100dB. Anything less than 100dB is not good.

So there you are, put all choices in a spreadsheet and rate the soundcards one by one using the factors listed above.



How to select a mixer for home recording studio

Previously, I was recording without a mixer at home and it was just awful. It didn't have enough amplification of signals. The signals that were recorded into Adobe audition were very weak and noisy. But successfully I still managed to do it right, listen to one sample song below :

"Always thinking of you" by Jeanine Maningo"
Publisher info: www.musicforlicense.net
TRACK - Always thinking of you*: "audiorecording.me" >Recording Equipment>"Tips on buying a mixer for home recording"*

The problem that I was always facing is the difficulty of the recording process. To change instruments, I had to go at the back of the computer and unplug my cable. It is just too risky, sometimes I get electrocuted.

My New Year resolution was to buy a simple mixer at home. I surf the net and i found Behringer xenyx 502. I say it is the best mixer for home at the lowest budget while achieving the best quality of sound recording.

I am so contented of my purchase, that I made these tips :

1. Buy the smallest mixer based on your budget but check the brand. Always go for branded ones, those that already made some reputation in sound engineering industry. When we go to local store we see a lot of mixer, cheap ones but not branded. The sound quality is awful. Just be careful you would not buy a pirated/imitation one.

2. Buy in the city's best recording shop for gears. I buy my Behringer Xenyx 502 in the city's biggest mall and also sells some branded speakers/monitors like Bose and other stuff.

3. Quality is more important the price itself but not too much. I mean if you can afford and you get the quality you want, then go for it. Just do not exceed on buying the most expensive ones but the quality level is just the same.

4. Check for the number of inputs, Xenyx 502 got 5 inputs (mono), and it is enough for me. The higher the number of inputs the better, but buy only something you need. If you buy many inputs but only use one most of your time, it is wasted.

5. Check for warranty. Make sure your product is protected by warranty in case of early damage.

6. Check for wide applicability. This means, at the same I can use my mixer to:
a. Record sound
b. Record sound while I can monitor (by headphone maybe)
c. Can shift to headphone or to the loud monitor (there is a switch)
d. Has clip detector
e. Has trim function (gain, very important in recording if you are not using a direct box)
f. Stereo output

7. This means, you should fully understand the manual first before buying anything., to measure it's applicability.

Mixer tips: Behringer Xenyx 502

This is one of the smallest mixer available for home recording studios, yet even as powerful and sounds as good as a professional mixing console. Official documentation/manuals can be found on the Behringer Xenyx 502 website. You can buy a Behringer 502 5-Input Mixer at Amazon.

Tip #1: Pre-amplication of weak instrument signals

Honestly, I do not use this for analog mixing as I use this as an interface for recording sound from the instruments (vocals, guitars, piano). The diagram is shown below:

Instruments –> Behringer Xenyx 502 mixer –> Sound card
Note: I mixed digitally using digital recording software like Adobe Audition.

Currently since my Creative Audigy soundcard cannot handle full amplication of analog instruments. Xenyx 502 is there to get the job done. All you have to do is to connect your analog instruments to one of the line inputs. Do not disturb the EQ (set it flat) and adjust the main mix volume, enough to get a good recording level in your digital recording software.

Sometimes there have been occasions when the analog instrument input signal to the mixer is very weak; these are common in passive instruments such as bass guitar (not the active bass guitar type). However you can always boost/adjust the signal in Xenyx 502 mixer using TRIM knob. Adjust it until you get a good recording level.

Tip #2: Use Mic for microphone input and Line in for instruments

Xenyx 502 can accept XLR microphone inputs as well as the standard male plug type. Under MIC is where you will plug any vocal microphones only. Plug other instruments in line-in jack.

Tip #3: For long life of the mixer, unplugged the power supply immediately after use

Since these are made of solid state devices, unfortunately these are subject to various stresses that affect the reliability of the electronic components in the mixer. One of these frequent stresses is electricity. If you will not unplug the mixer from the AC source, then you are allowing some electrical energy to be stored or consumed by the mixer, which speeds up the degradation of the components. This will worsen if in any even the AC line gets shorted or there are spikes, and then you have no surge protector.

Tip #4: Always set the volume to minimum when plugging the mixer to soundcard if the computer is turned on.

This will avoid damaging your speakers, in case a transient loud sound will be accidentally transmitted due to faulty wiring settings or maximum volume settings. This extends to plugging analog instruments to the mixer.

Tip #5: Always rely on your soundcard digital mixer to work along with Xenyx 502 analog mixer.

Quality recording soundcards like Audigy has a pre-included “surround digital mixer” that governs how the sound will be inputted and outputted in your computer. You need to ensure that it has been set correctly to accept Xenyx 502 sound input. For example, if you are using Line-in then set in the “recording” button to line-in. If these are not set, the digital mixer will not pickup the sound as well as your recording software.

Best CD burner/writer & media for burning audio mastering CD

If you are mastering your own CD for replication or duplication, you might need to assess whether your CD writer (drive) and your media are the best ones for audio CD mastering applications.

Let's start with some theory and facts:

a - The higher your burning speed, the more likely jitter will happen. These jitters are undesirable errors that can affect audio quality. Read here more about the relationships between jitters and CD burning speed: *http://www.myce.com/article/CDR-Audio-Quality-Vs_-Burning-Speed-189/*

b - Old CD writers can write at very low speed; for example 1x to 4x.

c - Modern CD writers can write very fast (e.g. 52x), however optimum writing speed (lowest jitters as possible and best recording quality) can be estimated around 4x which most mastering engineers in the industry suggest.

d - Quality of the burn master CD is dependent on CD writing speed, quality of the media and the quality of writing software/red book CD software. Of course, the overall sound quality is dependent on the output of the mastering engineer.

Let's not discuss first about the CD writing software (red book CD mastering software) or how the mastering engineer does the mastering, but let's see how you will be able to select the best CD writer and media for home audio mastering use.

Most professional mastering engineers agree that the best CD burner/writer for mastering is Plextor. The recommended model is Plextor PX-716A – Disk drive IDE Internal CD Writer if your computer supports IDE based drives.

If you are looking for a SATA based CD burner/writer, you can have the Plextor PX-716SA DVD+\-RW Dual LayerSerial ATA Drive

As to the reason why most mastering engineers recommend Plextor it's because of its negligible jittering rate (errors) and high quality set of CD tools that you can used to check the errors present in the CD media. It is important that the master CD you will be submitting to the replication/glass master plant should be completely error free. Plextor has the set of tools that you can used to check if your CD master is error free or not.

These drives support burning of CD master at 4x which is the industry standard for burning CD masters. So if you are using these drives. It is highly recommended that you burn the CD audio master at a speed of 4x using Red book CD audio mastering software like Sony CD Architect 5.2.

Recommended CD Media for CD master: Again most professionals in audio industry appreciate the quality of Taiyo Yuden CD-R.

These CD-R media have the lowest BLER,E-12, and E-22 errors in the CD media industry. These errors are critical if you are creating a audio CD master, because these errors can seriously affect the audio quality output of your CD master. Read more about these errors here: *http://club.myce.com/f33/disc-grading-system-bler-70863/*

The maximum rated speed of these CD-R is 52X. However, you can create a CD master at a speed of 4X using Plextor drives.

How to select a microphone for home recording studio

Selecting a microphone for home recording can be a difficult decision to make. It all depends on the followings:

a. Budget
b. Your room
c. Your recording commitment and the depth of your project
d. Your existing recording gear

Basically, there are two types of microphones :

- Dynamic Microphones
- Condenser Microphones

During the selection process, below are the recommended guides.

Guide 1: If you have the budget and your home recording studio is using high end gear with good room acoustics (and for example a vocal booth), use a condenser microphone, it will produce the best results.

Guide 2: If you have the budget but not the vocal booth, then use high end dynamic microphones such as Shure. Dynamic microphones are not as sensitive compared to condenser microphones.

If you do not have a vocal booth, there could be noise during vocal recordings. And it can be isolated easily if you have dynamic microphones since they are not super-sensitive.
For those who don't know what a vocal booth is, it is a sound proof enclosure where the vocalist will record the vocals. The purpose is to avoid catching noise from other sources in the studio.

Guide 3: If you do not have the budget, dynamic microphones are cheap. Just make sure that you do not record during noisy hours, or else you will be disappointed.

If you have a mixer, consider the connectivity of the microphone you are going to purchase. For example, you should not be buying XLR based microphones if you do not have XLR inputs in your existing recording gear.

Also, if you are building a home recording studio to produce commercial projects, the recording commitment and the depth of your project is high and therefore, secure a more expensive microphone like the condenser ones. Once again, if your room does not have good acoustics, then settle for dynamic microphones since it is more flexible compared to condenser microphones.

Always buy from quality and reputable suppliers, do not buy from unknown stores, there are lots of stores nowadays selling fake microphones.

Best microphone for vocal recording: Tips and guides

OK, so you read this page in order to know the best microphone for vocal recording. Let me help you choose. Before you need to purchase microphones for vocal recording you need to know the extent of microphone applications in your projects.

Remember there are two types of microphones targeting a bit of different applications:

1 - **Dynamic microphones** – the most common type of microphones we see.
Used as a vocal microphone for most live performances especially. The most popular dynamic microphones for vocals is SM 58. The cost of dynamic microphones is affordable with respect to typical home studio applications.

You can purchase a very good dynamic microphone for vocal studio applications for as low as $30. If you need to know about typical technical specifications for SM-58, you can read it here:
http://www.shure.com/proaudio/products/wiredmicrophones/us_pro_sm58-cn_content

2 - **Condenser microphones** – this is an ultra-sensitive microphone commonly used for professional studio applications, commonly found in big and professional A class studios.
Typical condenser microphones cost a lot, a vintage Neumann U47 costs around $5500. You can read some important informations about this condenser microphone here: *http://www.recording-microphones.co.uk/Neumann-U47-tube.shtml*
Of course, they have superb reception and frequency response.

Now back to the question: What is the best microphone for vocal recording? The answer is using a condenser microphone, but… Is there really a substantial difference between dynamic and condenser microphones? The answer is no.

Listen to the following produced songs below:

Using condenser microphone:
“Dream” by Jeanine Maningo
***CLIP - Dream**: “audiorecording.me*
>Recording Equipment>”Best Microphone for Vocal Recording: Tips and Guides”

Using dynamic microphone:
“At the highway” by Jeanine Maningo
***CLIP - At the highway**: “audiorecording.me”*
>Recording Equipment>”Best Microphone for Vocal Recording: Tips and Guides”

The lesson here is that if you do have the budget and a long term plan building a professional recording studio aiming to get several clients, you need a condenser microphone. If you are just working in your home recording studio with a very limited budget while still aiming to produce quality recordings, you can just purchase a Shure dynamic microphone like SM-58 and this will get the job done pretty well.

Song mixing software recommended: Beginners Guide

If you are using a personal computer to produce music, then you should be using a song mixing software. This piece of software should be able to accomplish the following tasks in order:

Task #1: Recording tracks (guitar, bass, piano, drums, etc) to your computer hard drive.
Task #2: Removing noise (if necessary) on your song mixing software.
Task #3: Mixing recorded tracks to get the best sonic results as possible.
Task #4: Mix down all tracks into one; ready for audio mastering.

It is increasingly popular for independent musicians to create and produce their own music at home using computers. In fact, it is highly possible to create professionally made recordings from it. One of the most important factors for success is your "knowledge" and "skill" of your song mixing software. Before I list down all possible choices, the following are my friendly advice regarding the use for that piece of software:

1 - Never change your audio mixing software after months of practice or years of experience. Doing this can seriously affect the quality of your produced recordings. The "mastery" of your audio mixing software is a strong asset for you to create great recordings. And that includes manipulation of the EQ, compressors, plug-in and other important tools included in that software.

2 - The most expensive audio mixing software does not necessarily mean the best.

3 - No matter how great your audio mixing software, it won't still produce great results if you do not have a "trained" ear (that can recognize good and quality recordings), well acoustically designed studio and a near field monitor with a subwoofer.

With that in mind, below is my most recommended song mixing software depending on your budget, productivity and expected recording quality:

For beginners with budget ($200 above in software) and expecting to get the best recording productivity, produced recording quality and professional mixing environment:

Adobe Audition 3 = $340
Cakewalk Sonar 8.5 Producer Software = $350
Sony Creative Sound Forge 10 = $318
M-Audio Pro Tools M-Powered 8 = $203.24

For beginners with no budget and expecting to get a good recording productivity and quality (not offering the best professional mixing environment to get the best results in long term):

Audacity = free.

Final Recommendations:

- If you are a beginner and looking forward to have a life long career in recording production.
  Buy a great mixing software with budget $200 and above. Learn and stick with it through years. Upgrade it only when very necessary (e.g when the software manufacturer requires it for security reasons).

- If you are a beginner and not looking forward to have a life long career in recording.
  Installing Audacity will suffice everything. Of course, you will not expect it to have the best sets of recording and mixing features like those in paid mixing software listed above.

How to install, load and add a DirectX plug-in in Adobe Audition

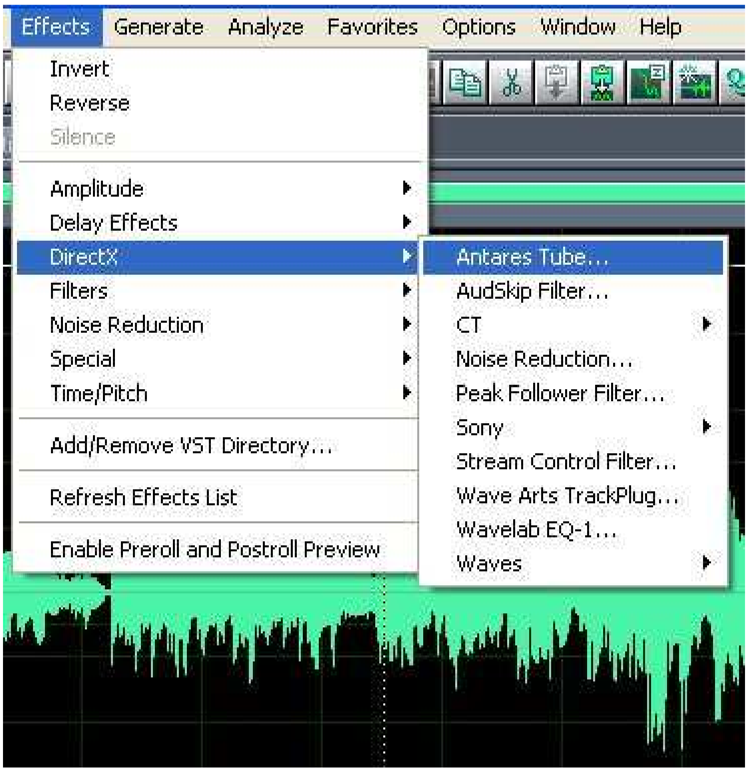
DirectX is a Windows based audio plug-in that you can use in Adobe Audition. This is different from VST which is owned by Steinberg, although they have the same purpose: adding more functionalities and features for your mixing and recording projects.

There are different types of DirectX plug-in that will be commonly used in your mixing and recording projects:

a - Reverb
b - Bass effect
c - Compressor
d - Delay
e - Guitar effects
f - Drum effects

DirectX plug-ins are third party plug-ins in Adobe Audition because if you don't install them, they won't be available in Adobe Audition.
If you are still new to DirectX plug-in, you can see if there are available DirectX plug-in by following the steps below:

1 - Launch Adobe Audition
2 - Load any sample wav track in editor view.
3 - Go to Effects – Direct X.
4 - You can then see the list of your available DirectX plug-ins.



To add a DirectX plug-in to your existing set of tools, follow those easy steps:

step 1 - Download the plug-in.

You can obtain your favorite DirectX plug-in from reputable software publisher. For example, you will download L2 wave's compressor plug-in here: *http://www.waves.com/Content.aspx?id=211*
You can even search Google for "free directx plug-in" although you should be careful you are NOT installing a virus or a malware. Double check if the software publisher can be trusted as well as scanning all files before installing in your Windows.

You can use this site to scan your plug-in files (only if the publisher seems unknown) for viruses: *http://virusscan.jotti.org/en* , if one of the results came out positive for virus; do not install the plug-in. You can also double check with the software publisher to make sure you are downloading a Microsoft compatible DirectX plug-in as well as a plug-in that will be compatible with Adobe Audition.

step 2 - After downloading, you need to install the plug-in on your Windows operating system.

step 3 - You then need to launch your Adobe audition.

step 4 - Go to Effects – refresh effects lists.

step 5 - After refreshing, you should be able to see your effects listed under DirectX.

Other easy ways of installing the plug-in is for you to install separate DAW, for example I got my Sony sets of DirectX plug-ins (wave hammer and Express FX) from an installation of Sony Sound forge in my Windows operating system.

After complete installation of Sound forge, I then refresh the effects lists in Adobe Audition and there; the Sony sets of plug-in are also added in Adobe Audition.

You can do the same with your other DAW; I do not know if other DAW offers it own sets of plug-in like the Sony wave hammer and Express FX plug-ins which are nice to have.

How to install Adobe Audition VST plug-in: Step by step guide

VST plug-ins offers additional sets of effects which are not otherwise in the default Adobe Audition list of effects. This short guide will teach you how to install VST plug-ins in Adobe Audition as well as how to search for freely available VST plug-ins. This guide only works with Adobe Audition/Windows.

Step1: Searching for free VST plug-ins.

You may want to visit *http://www.kvraudio.com/get.php* and search for available VST plug-ins. Follow the rest of the procedure until you will be able to download the VST exe file which will act as the installer.

This is the sample download page of one VST plug-in: *http://www.aodix.com/pageaodixv4.html* (the Aodix v4 plug-in).

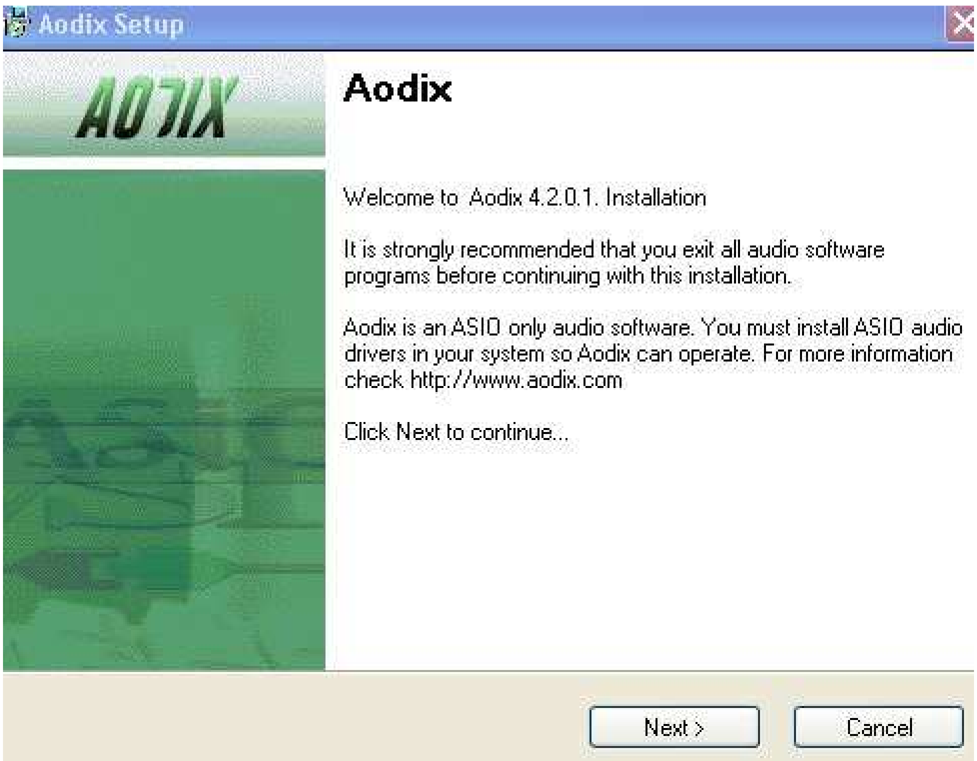
Step2: Double check the integrity of your file.

Make sure it is not containing any virus or malware. You can upload the file here to be checked by the scanners: *http://virusscan.jotti.org/en*

Step3: Once it is clean, double click the installer to start the installation process.

It is recommended to close all open programs to avoid conflict during installation.

You might see a warning before continuing the installation such as this:



It says that your system (soundcard and digital audio workstation/multi-editor/recorder) should support ASIO. You should pay attention to warnings to make sure the VST plug-in will work without problems.

Step4: Launch Adobe Audition – go to Effects – Add/Remove VST Directory.

Navigate to the folder where the VST plug-in has been installed. If you find many folders under the plug-in, make sure to select the general path, example: C:\Program Files\Aodix.

Based on the screenshot above, Adobe Audition will automatically scan the sub-directories under the main folder for VST plug-in related files.

Step5: Once added, go to Effects – and click "Refresh Effects Lists".

The VST plug-in should then be added to Adobe Audition.

This is just one example, there are countless of VST plug-ins available in the internet. I also recommend to read the plug-in reviews before proceeding with the installation to determine if it is great or not.

The good thing is that Adobe Audition supports both VST and DirectX plug-ins. So you're the one choosing what type of plug-ins you need to install. Although most of my Adobe Audition plug-ins are DirectX based, I am still using Steinberg VST plug-ins.

How to do a multi-track recording session

Recording multi-track is very efficient. It can save a lot of energy of re-performing tracks to record. Say the song has only one guitar track. Using multi-track techniques will enable you to continue recording tracks of the guitar without repeating the whole recording process again.

If you want to play the guitar track from intro to the ending of the song without a mistake, this is fine. But sometimes a mistake could occur when you are playing the guitar somewhere in the chorus. If you are not using multi-track recording techniques, that recording will be deleted and you will have to repeat the intro again. This will make you tired and will make the recording session very inefficient.

This is where multi-track recording sessions are very useful. It can save you a lot of time and energy. Also there are different multi-track recording softwares available. Frequently updating recording software or plug-ins is not recommended as a recording/mixing engineer as you need to fully master your gears (hence spending more time mastering it) to come up with the best quality. Where is the use of frequently updating gears if you forget to master it and keep producing crap records?

Below are the fundamental steps to do a multi-track recording sessions, this can applied in all types of recording softwares:

1. **Start recording the drums**. Make it right the first time. Be sure to complete all the drum recordings before proceeding to the next step.

   The drum is also the most difficult to record and arranged. The producer should have the idea what the song should sound like from start to ending (at this stage), and the drums should be providing good beats and timing of the song.

2. **Start recording the guitars.** Guitar recording needs creativity and time.

   A WORD OF CAUTION: Thoroughly check the tuning of your guitar before starting the recording.
   I heard a lot of noobies rush to recording the guitars only to find out after mastering stage the guitars are severely out of tune!

   I really love this stage, because I can be as creative as I need to be. This is the stage where the producer needs to hear great riffs, and guitar licking performance from the guitarist. Do not still do some panning at this stage, work out to get a clean recording of the guitar (of course noise free).
   Some tips: If you are playing the guitar and made a mistake. Shift to another track number and continue recording the rest. This is where multi-track recording is very useful.

3. **Record the bass**. Bass guitar recordings can be done in DI. Again as a word of caution do not record a bass guitar if it is grossly out of tune. Always check the tuning of the bass guitar before recording it. Same tips with the guitar, do not pan a bass and if you made a mistake, always select another track number next to it and continue recording the rest.

4. **Panning stage**- this means panning the instruments according to their location in the frequency spectrum. I have written some useful articles in the blog in how to approach panning. In this stage, you will make a duplicate copy of your guitar track and pan them on left and right. Always panned the kick and bass guitar in the center of the stereo image.

5. **Pre-mixing stage**:In this stage you will start applying EQ and compression to all recording tracks. Your aim is to sound them as clear and strong as possible.

6. **Finalize the minus-one**: After the pre-mixing stage, the producer will then invite the lead singer or artist to sing or practice the vocal tracks of the song. This is where the producer decides if all tracks matched or good enough to be paired with the vocals. If there are adjustments, the producer can still change the arrangements.

7. **Vocal recording stage**: This is the stage where the vocals will be recorded. Basic multi-tracking applies but a pro vocalist prefers to perfect the vocal recording in one session only, not cutting it in tracks.

8. **Final mixing stage**: This is the final mixing stage where all tracks are mixed for clarity in accordance to the producers' request. After the mixing stage, mix down will be performed and will be submitted for mastering.

Multi-track settings for recording and mixing

For those who are using Adobe Audition for recording and mixing audio, I am going to share my multi-track settings, as well as explaining the importance of each.
Correct multi-track settings ensure that you are recording at the best recording quality as possible, as well as working in the best mixing environment which will maximize the overall audio quality of your mix down.

If you are still new to Adobe Audition, you can see the multi-track and recording settings by:

Step1 - Launching Adobe Audition
Step2 - Going to “Options”
Step3 - Under “Options”, clicking “Settings”.
Step4 - Adjust the multi-track settings according to the following settings:

Explanation of each of those important parameters:

a - **Track Record**

Set to 32-bit, if you set this to 32 bit you are recording audio in 32 bit quality (originally a 24 bit + additional headers). This is the most recommended recording bit settings for optimum quality. Bear in mind that this is limited to your sound card; so make sure you install a sound card in your computer that is capable of recording 24 bits.

b - **Pre-Mixing**

32 bit, this will let you hear audio at 32 bits quality before mix down and after recording. This is primarily important for detailed monitoring of audio quality and lets you hear mistakes and other imperfections that would not otherwise be heard at lower audio quality.

c - **Correct in Drift in Recordings & Correct for Start Sync in Recordings**

If these options are checked, Adobe Audition will automatically correct any latencies and delays introduced in the multitrack recording sessions.
If these options are not checked, you might find out that the recording material is lagging behind other multi-track audio tracks which can be impossible or very difficult to correct after recording.

Other factors that will contribute to the delay or latencies are the speed and quality of your PC components. For best recording results while eliminating latency, you should be using a high end sound card, a powerful PC (at least 2 GB RAM and 2.4 GHz processor). Hard disk speed and size is also a factor so make sure your hard disk is not defective, at least 320GB and with a speed of 7200 rpm.

d - **Background Mixing Priority**

Set to 2, as you have noticed that lower this value the higher priority will be given to the background mixing which are important if you need to play multi-track recorded tracks quickly. This will allow more RAM to be allocated to the background mixing process.

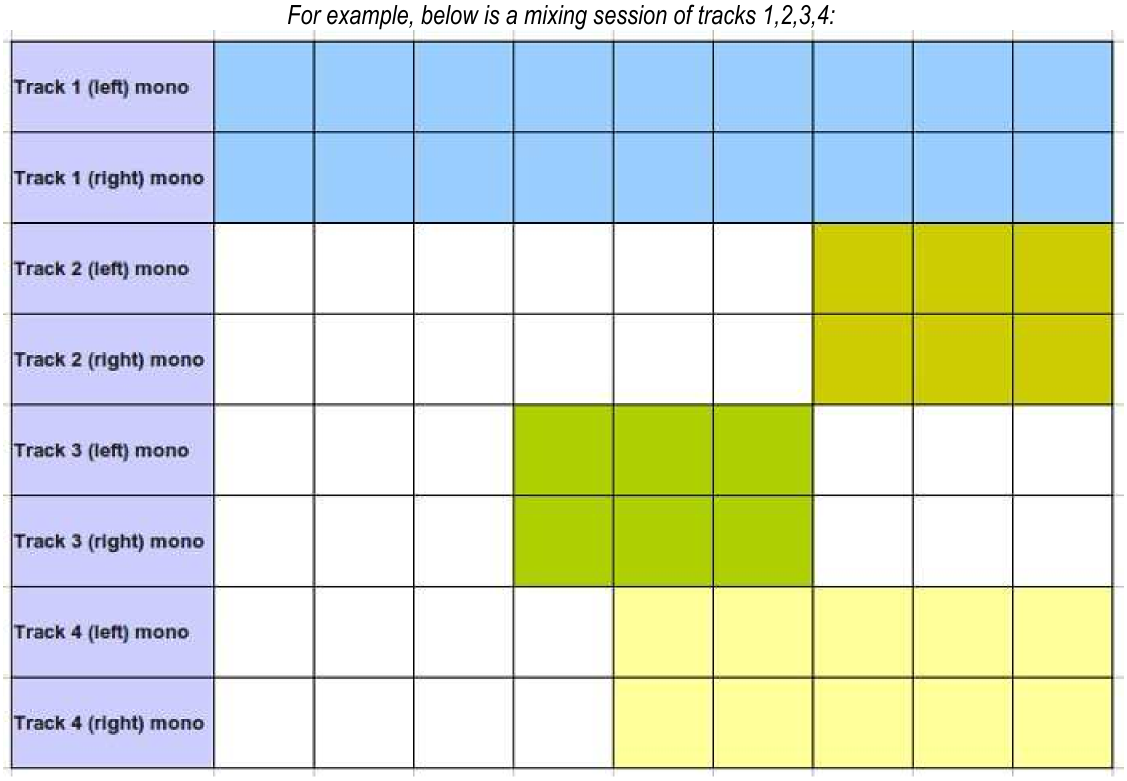
IMPORTANT: If you find your old settings working and you need to change your existing settings with the suggested settings above, do not forget to have a screenshot of the old settings so that you can revert to it when needed.

How to export a mixing session to any multi-track software

When you are mixing in your own computer/DAW, you might be using a mixing software like Adobe Audition. But sometimes, you might want others to remix your tracks and then submit those tracks to be mixed by another engineer.

Of course, you do not expect the other mixing engineer to be still using Adobe audition; so this is where this problem starts. This short guide will help you solve this classic problem in mixing and lets you export any mixing session you need which is to be done by another mixing engineer.

For example, below is a mixing session of tracks 1,2,3,4:



You may have noticed that each track is actually sub-composed of two sub-tracks (left and right mono waveforms). The left is panned to the left and right, panned to the right. Combining two mono waveforms results in stereo.

Track 1 started at 0 second. While Track 2, is started near the end. Track 3 is started somewhat in the middle but a rather short recording. Track 4 is also started somewhat in the midde until the end of the song.

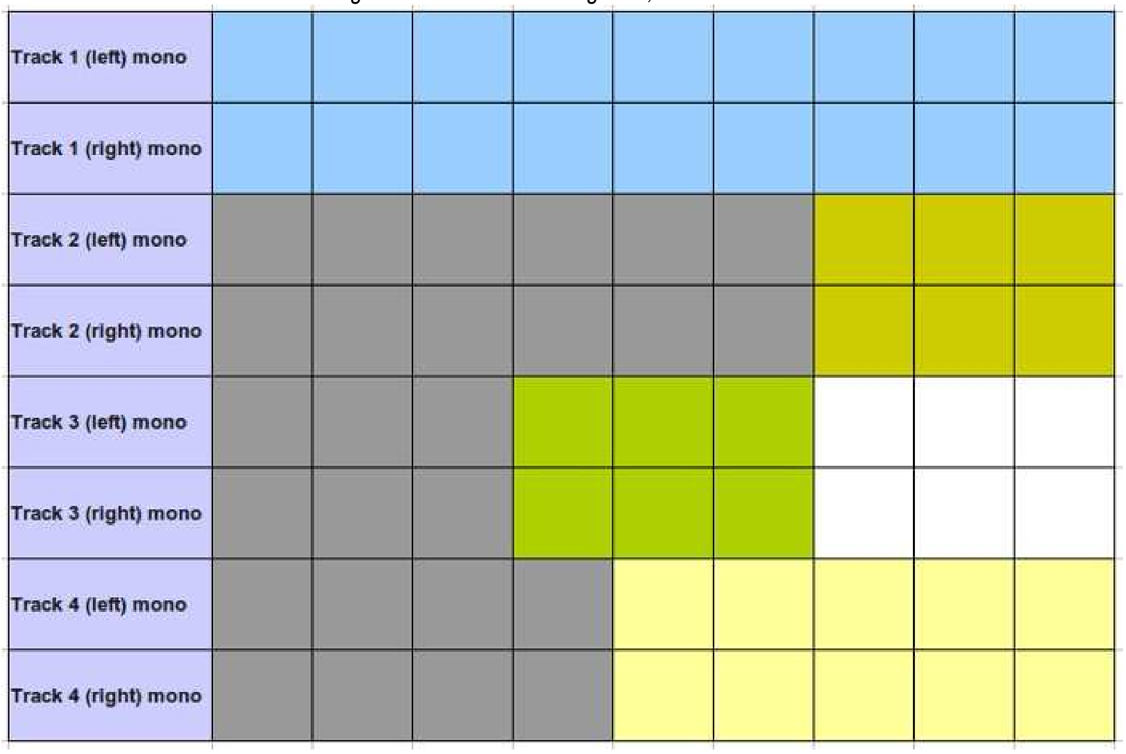
Supposing you are working on this using Cubase, Protools, Logic, Sonar or Adobe Audition; how are you going to export this to another mixing software, for example Sony Soundforge? Or even Audacity?

Of course, one approach is to save all audio files used in the mixing session and then give it to another mixer (using a different mixing software). But the new mixing engineer does not know the arrangement details of each tracks in the multi-track space such as shown in the screenshot above.
Chances are, if those tracks are inserted to his/her multi-track software, it will not sound synchronized and the timing will be destroyed.

The Solution: TRACK TO EVERYTHING TO ZERO BEFORE EXPORTING

The solution is to track everything to zero. So for example in the track 2 above ,since it starts near the end of the song, it will be “SILENCE” from 0 second up to the moment the track 2 starts playing. For example, track 3 is started in the middle of the song, but you can “stretch” and add silence to the track 3 waveform ALWAYS STARTING from 0 second up to the moment track 3 starts playing.

Visualizing this in a multi-track diagrams, it will look like below:



The gray colors are the silence ADDED to each wave which does not start at ZERO second. You can use your existing multitrack software to add this silence and then SAVE all affected files.

After all tracks are TRACKED to zero, you can safely save all files to a different media (DVD for example) and then give it to another mixing engineer. The engineer will simply load up all tracks with respect to zero second and then ALL tracks are still in timing and will be synchronized when gets played.

You cannot export effects done on the wave (on your current multitrack software), let the new mixing engineer decide what effects he/she needs to apply to your song. The export audio wave should entirely be a CLEAN recording only and not applied with ANY effects or whatsoever.

4 - Credits

Guides written on 'www.audiorecording.me', by Emerson R. Maningo
Check the website for more tutorials and updates.

CONTENTS SOURCES:

www.audiorecording.me
Emerson R. Maningo
www.musicforlicense.net
www.audiocheck.net
recording.org/pro-recording-forum

IMAGES OWNERS:

Emerson R. Maningo

Jeremy D. Silverman	Chidorian (flickr)
Stephan Czuratis	Eyeliam (flickr)
ZooFari	Kikisdad (flickr)
Alesis Products	M-Audio Products
Behringer Music	Musicman Company
Creative Products	Plextor Products
Fender & Squier Music	Shure Products
Ibanez Products	Taiyo Yuden
IXBT Labs	Yamaha Company
Adobe Software	Hotstepper Sequencer
Aodix Software	Voxengo Software

SOUND CLIPS OWNERS:

www.musicforlicense.net
Jeanine Maningo
Emerson Roble Maningo
Ruriko (recording.org)
King666nothing (youtube)