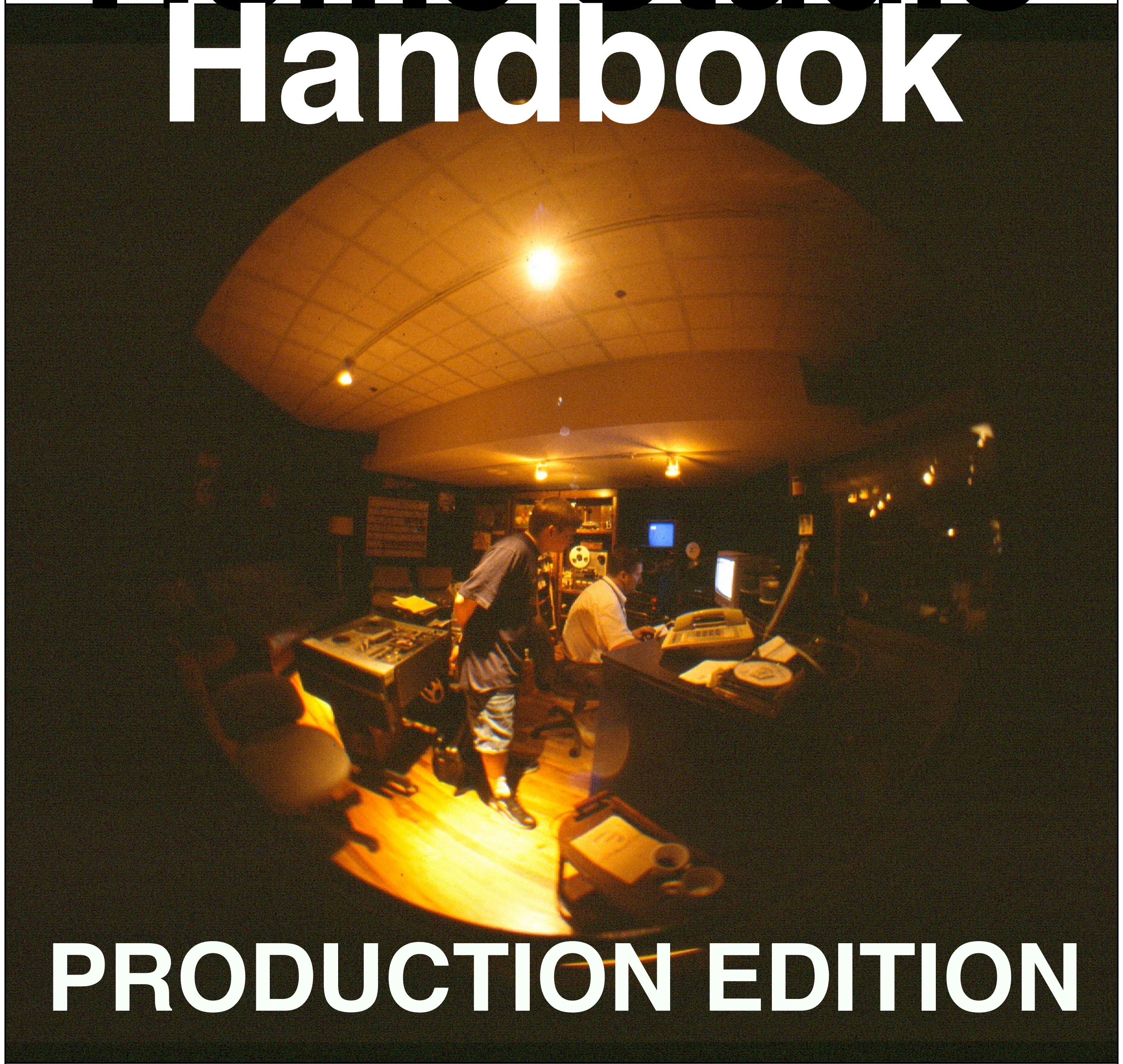
A PRODUCTION GUIDE FOR HOME STUDIOS

Home Studio Handbook



PRODUCTION EDITION

By Nick Sanabria

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Acknowledgements, Special Thanks
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The suggestions in this guide are the express opinions of the author based on years of actual in-the-trenches experience and are in no way to be confused with scientific or otherwise factual information.

There were no paid endorsements in this guide, although in hindsight, it might have been a good idea.

The drawings were done using the Sketchup program, all screen shots are courtesy of the manufacturers noted: Avid, Trillium Lane (Avid) McDsp, Izotope RX, Slate Digital, that were purchased and used in the making of this book.

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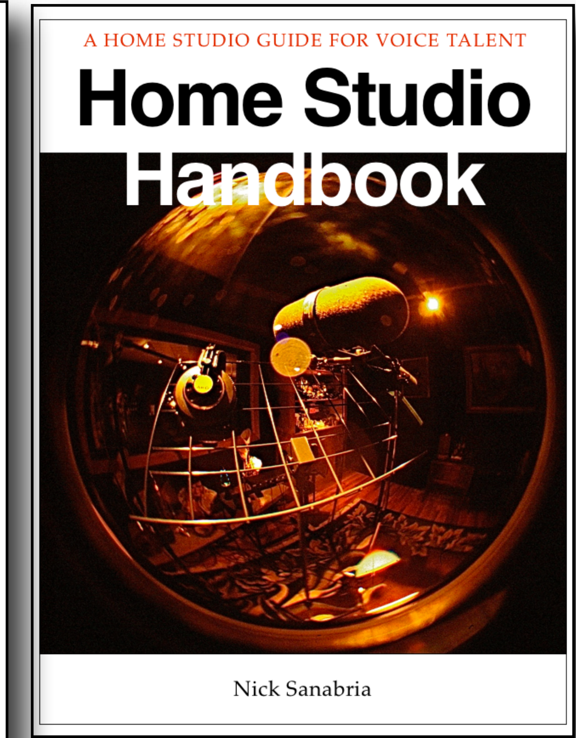

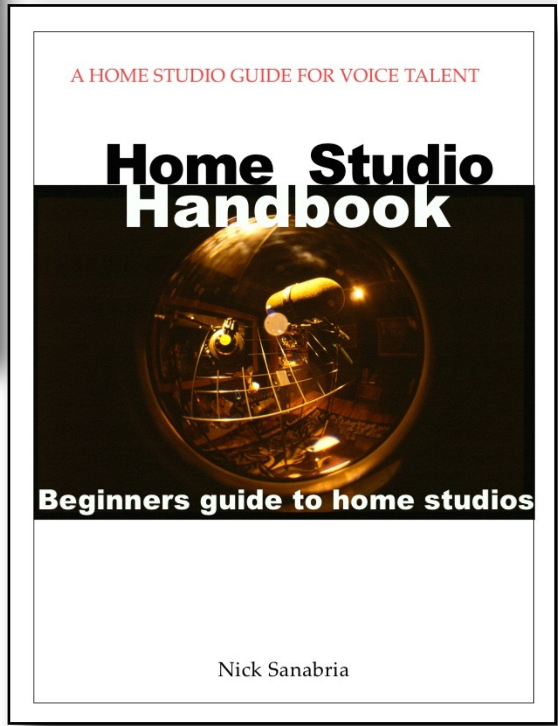

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PURPOSE:

This guide is meant for voice talent setting up home studios, new guys, studio interns, and engineers trying to expand their knowledge of techniques to improve spoken voice recording and editing. It is not a primer on acting techniques, wiring a patch bay, or how to use a particular program or plug in, although some of these topics are sideswiped. And opinions offered.
These are basic, real world solutions and get-arounds to common problems experienced in home studio set-up, recording, and editing voice.   Some basic production advice is also offered since many of your clients will want you to produce finished work.

The goal: the voice tracks you send out will sound great.
If you pick up one trick that either streamlines your work flow, corrects problems you couldn't fix in the past, or helps you save a client's butt, then this guide has paid for itself, and you should raise your rates.

And on the subject of rates:

Rule #17 taught at every business school: ***"They will never pay you more than they are paying you now."***

Many of you are voice talent, and are throwing in the recording/studio work gratis.
Your clients had been paying $175.00 (or more) per hour at a recording studio for the same work you are graciously doing for free.
And you are giving them respectable quality.
They are very happy with this system.

But you may want to consider this:
Your time is valuable and you should be compensated for your knowledge and ability to record and produce a quality product. Even in today's home studio environment, you should be fairly compensated for your knowledge and use of these tools to improve your client's finished product.

You are creating sound that will reflect on their business.

You have spent money on studio spaces and equipment, have taken the time to learn them, and have developed skills to fix problems so that you can produce sound as close to a pro studio as possible.   If this guide gives you enough chops and/or confidence in your abilities to call yourself a recordist, recording engineer, or audio editing pro, so that you feel justified in charging studio time in addition to your voice work............
Well then, I have done my job.

A WORD FROM THE AUTHOR

This book is about multi-track assembly, mixing and final mastering techniques, with a bunch of opinions on everything from music choices to creating sound effects thrown in.

Most of you are voice-talent that have been thrown production duties. You already have "ears" as far as voice, now it's time to perfect your production ears, learn a few new tricks, and create finished tracks ready to air.

In "Home Studio Handbook, a home studio guide for Talent", I included a bunch of production pages, that in hindsight, should have waited for this edition. If you already have that edition, you will see some chapters repeated here.

Most of the time you will just be doing music and voice, but that in itself has many problems that I can help you solve.

And perhaps sound effects added in? Or multiple voices?

For demonstration purposes, I will be using Protools, but any DAW will do and should work similarly.

An advantage of a DAW for multi-track work, is plug-ins. You can change the parameters of your plug-ins as you are mixing. Changing EQ, or compression amounts, swapping out plug-ins is easier using a DAW.
Audio Editors, like Audacity, are harder to use since your edits are destructive and are not easily changed BUT, you can use them with a few "get arounds".

There is a school of thought: Use Protools, it is the industry standard. There is a lot to be said for that, but there are equally elegant platforms that will work as well, for considerably less money.

I will be showing you some different channel set-ups, using aux sends, sub-mixes, and a variety of plug-ins to accomplish your goals.

All of these techniques were learned and used at my down-town studios. Some of the techniques were from the days of multi-track tape, but the theory and procedures still apply to digital; it's just easier now.

So take a deep breath.

It's not that hard.

With some of the tips I will be showing you, you should be able to develop enough chops to call yourself an engineer.

Let's get started.

CHAPTER 1

Monitoring

Monitoring your sound

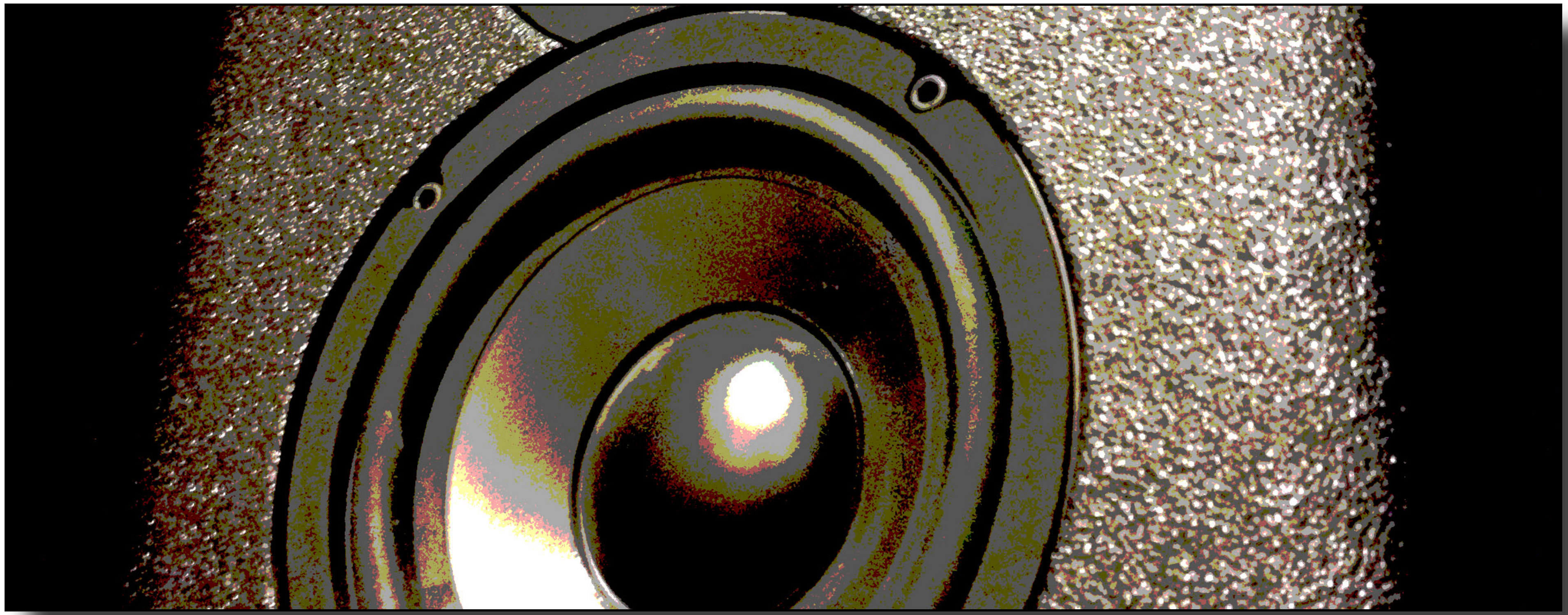
How you hear what you have recorded, and how you modify it to make it sound better, is greatly influenced by your monitors and monitoring environment.

If your monitors lack bass, you will over compensate and add too much bass.

If they have a midrange bump, you will take out midrange to compensate.

If your listening / mixing room creates the impression you need more treble, the result will be a bad sounding audio file that someone at the other end will have to fix.

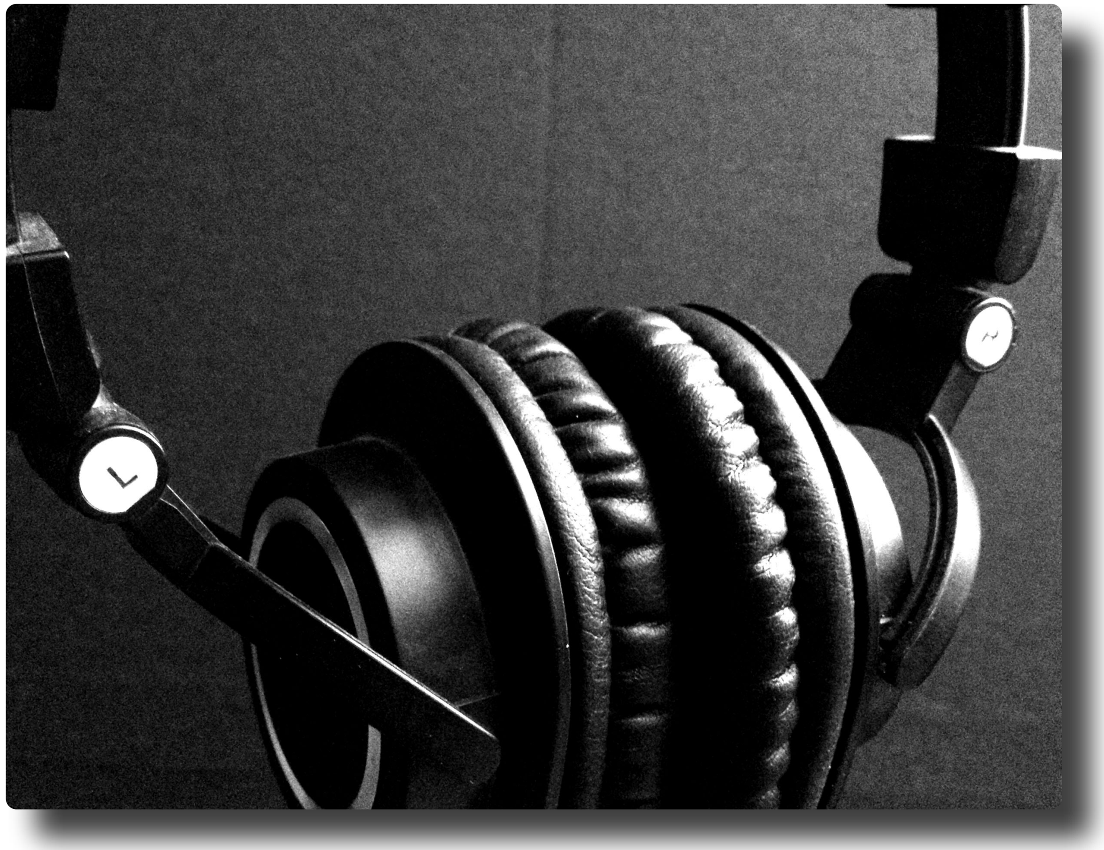
In this section I will discuss ways of finding a monitor solution that can work for you.



This is probably one of most important things to set up right, so let’s get it out of the way before the fun begins. If you don’t have accurate monitoring, the changes you make to the sound will be completely wrong.

It pays to get a good set of headphones, or if you are going to use monitors, make sure your room is treated and is accurate: make sure your monitors are positioned right, and are either on stands or pads to eliminate vibration. This part is tricky, but necessary before you can create accurate masters.

HEADPHONES.



Let's start with headphones:

Headphones are a good solution to home studios, and are even used in pro studios for close work. Mid-price headphones (around $150 to $200+) will usually give you a fairly accurate representation of what would be considered flat, or with a little coloration; each manufacturer creates a slight rise in certain pleasing frequencies to get consumers to buy their headphones "because they sound better".

If your headphones are "beat" headphones, they will accentuate bass frequencies, so you will change your mixes by lowering the bass, and then your mixes will be thin and lacking low end.

Same goes for headphones that accentuate highs. You will lower the high frequencies to sound right to your ear, when you should have left them where they were. Now your masters sound dull.

Headphones also isolate the audio from room sounds, computer noise, and other sound sources that might mask problems, and let you zero in on the areas such as mouth noises, background noises, breaths, etc. so that you can clean up the recording. But wearing headphones is fatiguing. Plus when someone taps you on the shoulder, it is guaranteed you will jump like a startled cat.

Headphones are a good strategy when you cannot treat the listening room adequately, but don't use them as an excuse to avoid fixing it down the road.

For most of you, I would recommend a mid-priced set of headphones:

(these are from the Sweetwater website)

Beyerdynamic DT 770 PRO Closed-back Studio Mixing Headphones $199.99

Audio-Technica ATH-M50x Closed-back Studio Monitoring Headphones $149.00

Audio-Technica ATH-M70x Closed-back Monitoring Headphones $299.00

There are many more out there; these are ones I have used an have found them to be quite good at hearing detail, but like all headphones, you need to check your mixes through another source: take them to a studio and listen to them through big speakers, listen through your car stereo, smart phone, computer, etc.....and go back and tweak.

You will learn your headphones shortfalls, and compensate. After you are familiar with their sound, you will get more and more accurate final mixes.

Cheaper headphones may not be accurate for mastering, might be good for listening to your voice while you are recording, just not for mixing.

DESKTOP SPEAKERS (computer/consumer)

There are a number of desktop speaker systems that go from ok to relatively good, and are generally not expensive. Some desktop computer speaker systems sold at computer stores consist of small 2 way powered satellite speakers (mid and high frequency) with a larger single sub-woofer for bass, and these can give you decent sound reproduction for a very low price. Computer 3 way systems are in the $80 to $200 range.

But note, these are for general monitoring overall sound and should not be trusted implicitly for accurate mastering. If you equalize to these, you may be compensating for the sound that the speaker manufacturer has designed in to make them sound more pleasing (such as heavy bass rise) to a non professional buyer. These should be used when headphones become fatiguing but never for mixing or mastering. They can, however, be used to check the accuracy of a mix to see how it will translate to non pro speakers.

NEAR-FIELD MONITORS

Music/pro-sound stores sell 2–way powered speaker systems that are relatively inexpensive and are far more accurate than computer/desk top speakers. Most have 5″ to 6″ low frequency drivers (woofers), and a variety of high frequency drivers. (tweeters). Larger systems have 8″ drivers and may have a midrange driver as well as a high frequency driver. 10″ and 12″ subs are available, but you are now going into pro level/music production type monitoring. Most powered speakers have level adjustments to balance the low and high frequency levels. Powered speakers have built in amplifiers that are designed specifically for their components, and are simpler to install. Passive speakers require a separate power amp and are generally used in pro level studio installations.

2 way powered near field speakers start at about $100 each and go up from there.

FINDING THE RIGHT SPEAKER

A good strategy for finding speakers you can live with, is to take a recording you are very familiar with, and play it through the various speakers at the pro sound/music retailer. Most have listening rooms that have all the speakers connected to switching panel so you can toggle back and forth between speakers. Just like at the eye doctor: “Which is better? A? or B? A? or B?” When you have narrowed it down to a few possible choices, do some homework. Go on line, see reviews, check blogs and forums to see what others think about them. Talk to your recording studio buddies. Consider price points also.

And make a deal with your retailer: that you can bring them back and exchange them for another brand if they don’t sound good in your studio.

PLACING THE SPEAKERS

With near field monitors, you should either place on speaker stands behind the desk (to isolate them from the resonance of the desk) or on isolation pads on the desk that do the same. Speakers will vibrate the surface of the desk and this will interfere with the imaging of the speakers, (left/right sound placement) or will create a perceived eq bump as the table top vibrates at its resonant frequency.

If possible, position the desk and speakers away from the front wall, and centered between the side walls.

You should place the speakers at ear level pointed toward your head as if your head is at the apex of a triangle, the speakers being at the base of the triangle. Keep the computer screen out of the direct line with the output of the speakers, and remove any objects that are in the "path" of the speaker. The sound should be direct to your ears with nothing between your ears and the speakers. Subwoofers, however, can be placed nearly anywhere, since the bass frequencies are generally omni directional.

CHECKING YOUR SYSTEM

A good policy once you have your monitoring system set up, is to record what you think is a good sounding sound file (after you have eq'ed it to taste), burn it to a CD and listen to it in your car, or on the stereo, or through the crappy computer monitor speakers, and listen through good headphones. Then send it to your audio buds at studios to get their opinion. If they all come back saying "way too much bass" or "too much 1K", then you need to compensate by either adjusting the monitors, fixing the room, trying a different set of headphones, and understanding what you thought sounded flat through your monitors, is not the case. You may have to trial and error it several times until you get a thumbs up from everyone.

You will learn to compensate for your systems' shortcomings, and fix what you can.

TREATING THE LISTENING ENVIRONMENT.

Many of the leading pro-sound retailers/music shops carry ready to hang audio panels for a typically sized room, that will diminish many of the problems associated with listening in a live room with parallel walls/ floor/ceiling. These range from several hundred to several thousand dollars, depending on the type of problems your room may have. Many of the companies have design programs that can help you pinpoint problem areas, or you can send the specs of your room to them via a web form and they will suggest treatments.

But, there are cheaper alternatives. You may need diffusers to scatter the sound, a typical diffusion grid is usually suspended directly over the engineers head or on the back wall. You probably will need bass traps, especially in corners. But every room is different and needs to be tweaked with tricks outlined in the ***Absorption/Diffusion section in Chapter 8, section 3***

There are even non-permanent sound panels that can be hung on hooks like paintings, or propped up on stands, or placed around the listening area that will greatly increase the accuracy of what you hear.

Even rearranging furniture will make a difference. Or changing the position of a large painting or mirror. Or changing the angle of a book case. Or using a throw rug, or not. Or hanging all the family photos on the back wall as a diffuser. Or heavy drapes over the windows. You do not want the listening/mixing room to be too dead, which will lead to fatigue and inaccurate equalization. Just as with the booth, tweaking a room is an on-going process.

CHAPTER 2

Basic Multi-track Production

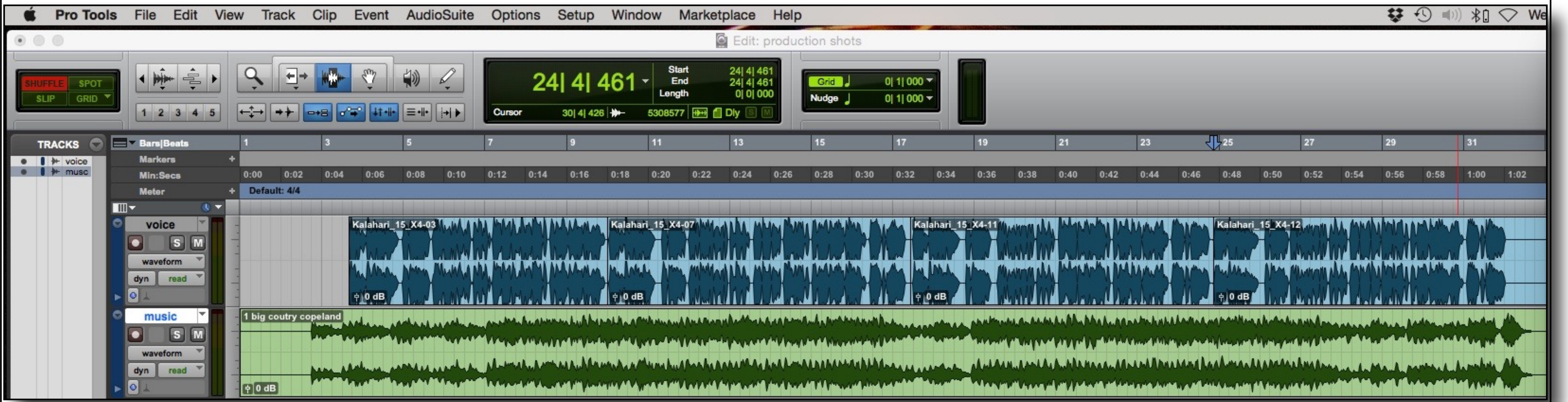


Photo courtesy of Avid

In this chapter I will cover the basics of multi-channel production using voice, music and sound effects. I will cover typical set ups that will speed your workflow, and show ways to make your mixes sound more professional. I will also cover Master channels, and typical plug-ins and plug-in combinations. Later sections will deal with Aux sends for effects like echo and reverb, and how to use those plug-ins to create the illusion of depth and space.

Basic set up #1 (and finer points.)

Multi track set ups

1. VOICE TRACK
2. MUSIC TRACK

Basic 2 channel set up: voice track, stereo music.

(voice track in these examples was imported from a stereo master, normally your voice track would be mono....play along with me here)

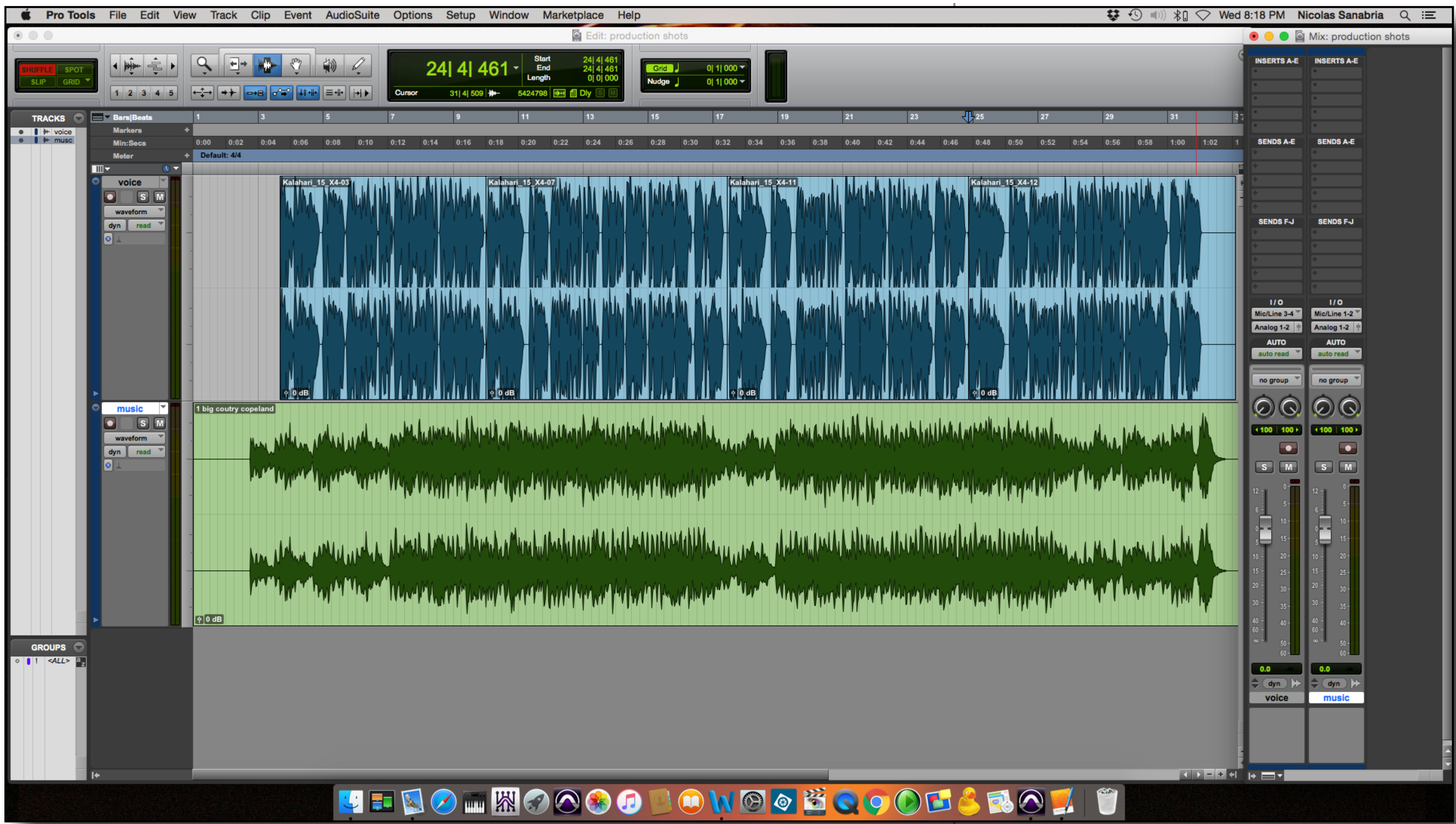
Photo courtesy of Avid

Let's start with your edited voice track, and a needle drop music track that you have imported to channel 2.

You have moved the voice track to the approximate start position, just after the music intro.

At this point, a rough arrangement is fine.

You can fine tune the placement as you go, and I will discuss the "musicality" of a mix a bit later in this chapter.



Note: this is a protools screen, but most other DAW's are similar in look, and operate pretty much the same.

With an audio editor, like Audacity, you may not have a console view, since audio editors operate differently.

Photo courtesy of Avid

Make things easier for editing by increasing the size of the edit windows:

This makes things easier to see when lining up audio elements, doing any additional edits to the voice track (if you need to take out or add time) or to line up elements with music changes.

Note the console view on the right. Scale the edit window so you can add the console window on the same screen. I like to keep both on the same screen so that you don't have to waste time toggling between them when making changes.

Start by focusing on the voice first. Mute the music track for now. (or Solo the voice)

Add an EQ as your first plug-in.

This is a parametric EQ, you can not only dial the exact frequency you want to adjust, you can also adjust the slope of the curve using the "Q" or width.

This is a very precise and flexible tool and most DAWs come with several, some come with a channel strip that includes a gate, compressor and EQ.

I like to adjust the EQ as the first step before adding additional plug-ins. Get things sounding good before you move on to other plug-ins.

Most small booths have a mid-bass rise, so I have lowered the mid bass around 400hz. I have boosted the low end in the 100hz range, and added some high end in the 4khz range. This is for my mic, in my booth. Yours may vary greatly.

Photo courtesy of Avid/McDsp

Next add a **compressor.** This will compress or lower the volume of the louder sections. You can then raise the overall level, making the softer parts louder.

A compressor is like an automatic gain rider, as a peak approaches the threshold, it automatically lowers the volume at the ratio you select. 2:1 is very light, 10:1 is very hard, or a limiter.

For voice, I usually use a 3:1 ratio. Most compressors have a make-up gain feature, so as you lower the peaks, you can raise the overall volume from the compressor itself, or you can raise it on the console channel. (6 of one; 1/2 dozen of the other)

You may notice you have to go back and tweak your EQ, that the compressor is exaggerating some of the higher frequencies. Or not.

Be careful not to be too aggressive with the amount: I usually like to stick around a -3db average to -5db on peaks, any more and you may hear a pumping sound, and it will greatly exaggerate breaths and softer parts.

Photo courtesy of Avid

Now add an **EQ** to the music track, and set a rough balance with the fader so the music track sounds balanced to the voice.

COMPLIMENTARY EQ
(covered in Chapter 6, section 4)

Here is a trick used to get the voice on top without having to dip the music track so low that it sounds like some background noise.

Complimentary EQ is boosting some frequencies in the voice, and removing corresponding frequencies in the music track. Look at the EQ on P12. There is a boost in the 4K; the music track has a corresponding dip in the 4K. (Try 1K or 3K.)

This allows you to mix the music hotter, without affecting the intelligibility of the voice. (I also boosted the bass on the music since it is not in the intelligibility range of speech.)

By lowering competing frequencies in the range of speech, you make the voice more intelligible.

Nothing is worse than having a client say “the music is too loud, you can’t hear the announcer”

Photo courtesy of Avid/McDsp

And you turn it down....and down....and down..until the client is happy. Then you hear it on the radio and it sounds like some music being played on a radio in the next room.

Start by putting a dip at around 1K in the music track, you will have to experiment with how much you lower it and how wide a Q, and which frequency works best. This creates a space for the voice with less frequencies interfering; the voice rides on the music track instead of fighting it.

Lead instruments tend to be in the range of speech: guitars, brass, piano, etc. If you can find a place in the track that is primarily bass and drums, you can usually pump the track and still understand the voice.

Also, if you have an aural exciter in your plug-in arsenal, this is a good time to use it on the voice. It creates harmonics that "fatten-up" the voice in the frequencies that are associated with intelligibility. Or you can add a touch of eq on the voice that corresponds with what you took out of the music track. And a touch more in the 5K range. (experiment: these settings are just starter suggestions) But listen. If the voice sounds too brittle, or abrasive, back off and just lower the track. A harsh voice-over defeats the purpose.

And if you are using reverb on the voice: eliminate it, or pull it way back. Reverb tends to push the voice back into the music.

You can mix with just these two channels, but there is a way to sum both channels and add a limiter.

And if you have multiple channels with sound effects, other voices, etc., this will make mastering much easier.

The master channel.

Photo courtesy of Avid

MASTER CHANNEL:

Create a master channel, (stereo).

Both your mix channels will be summed through a master channel. This will be very helpful, especially when you start using multiple channels.

I have added a Limiter, which you can set to keep any peaks from going into the red. Or if your client wants it to be mastered at -3db, (for audio books, for instance) you can set the ceiling at that and no audio will go above that. The threshold setting is where limiting begins. As with all processing, you can overdo it and create distortion.

And digital distortion is an ugly thing.

This is a good tool that will catch any “overs”; without one, you would have to lower the overall volume of the master to avoid a few peaks that would have pushed you into the red.

Limiters will make your master sound "louder", but you will notice, it takes away some of the dynamic range between the loud and soft parts.

FADES:

Finer point: Music intro, music out.

I have heard many spots with music tracks, jingles included, that do not dip the music under the announcer.

Normally, you would have the music up, dip under the announcer, then back up for the exciting finish/button. Otherwise, the music has no impact, it just sort of lays there, like a hammock under the voice. Yawn. (I exaggerate.)

The example below is an automated fade under the voice, and an ending fade up/finish.

You can either arm your channel in the automation menu and record the fader movement, or draw in the fades as I have done here in the "volume" window in Protools.

Photo courtesy of Avid

Your DAW may have a different system, but most let you draw or size in the edit screen. This is a bit more accurate than using the fader with your mouse. (But there is something so "studio" doing it that way!) I digress.

In the previous example, notice the fade starts down just before the voice comes in, and is at the "bed volume" after the first word or two. The mid point of the fade down should be into the first word....or so that you don't hear the music "duck" before the voice comes in.

Same with the fade up at the end. But you can be a bit more generous with the fade up and telegraph that this is the big finish.

The point is to make the music level change without anyone noticing. Your fades may vary.

But how do you know if the level and balance between the voice and music is right?

One way is to play it on your computer speakers, in your car, on your parent's hi-fi, (you'll need an 8-track transfer for that) on your smartphone, or at a friend's studio who has really expensive equipment....OR: **L.I.A.R.**

L.I.A.R TECHNIQUE *(also in chapter 6, section 4)*

Remember, the music and sfx are there to support the voice. But how do you know if they are mixed too loud or too soft? It sounds fine on the monitors and even in the headphones, but you suspect your listening area is not as accurate as you would like.

My friend and audio guru, Ken Goerres came up with a strategy that works even in the best studios: L.I.A.R or **Listen. In. Another. Room.** When you leave the room and listen from the hall, or down the hall, you get a more accurate reading of how everything sits in the mix. The mix you hear in another room is about as close to the balance you will hear when broadcast.

CUTTING VOICE/CUTTING MUSIC

Finer point

Musicality of voice with music.

Most announcers/voice talent have a cadence, and if you can find a piece of music that matches their internal click track, it just works magic.

The voice just drops right in and it sounds fantastic......It sounds like music.

Most of the time you don't have that luxury, so you have to make it happen. More work, better outcome.

You may find a piece of music, just what you had in mind (client approved, of course) and find that the beat is just a bit tad slower than the announcer's pace. No problem

Time compress the music. (You may have to cut in a bar or two to make up the difference)

You are close, but some of it is not lining up.

So cut the voice track so that it matches the music.

Add spaces in between paragraphs, so that the copy hits with the beat of the music. Most music has verses, choruses, and bridges. If you can cut the dialogue to hit with the music change, you are golden, as our former Governor used to say.

Just make sure you are not putting in pauses where they don't belong, breaking up the flow of the copy points. And you don't want every word to hit on the beat, just the beginning of the paragraph.

Or cut the music to match the copy. Add a few bars up front so that the music change corresponds with the copy change in the middle (when the hero product saves the day) and the tone of the read changes to upbeat.

More on music editing on page 132

Basic Set Up #2 Multiple Channels

Speaking of sound effects:

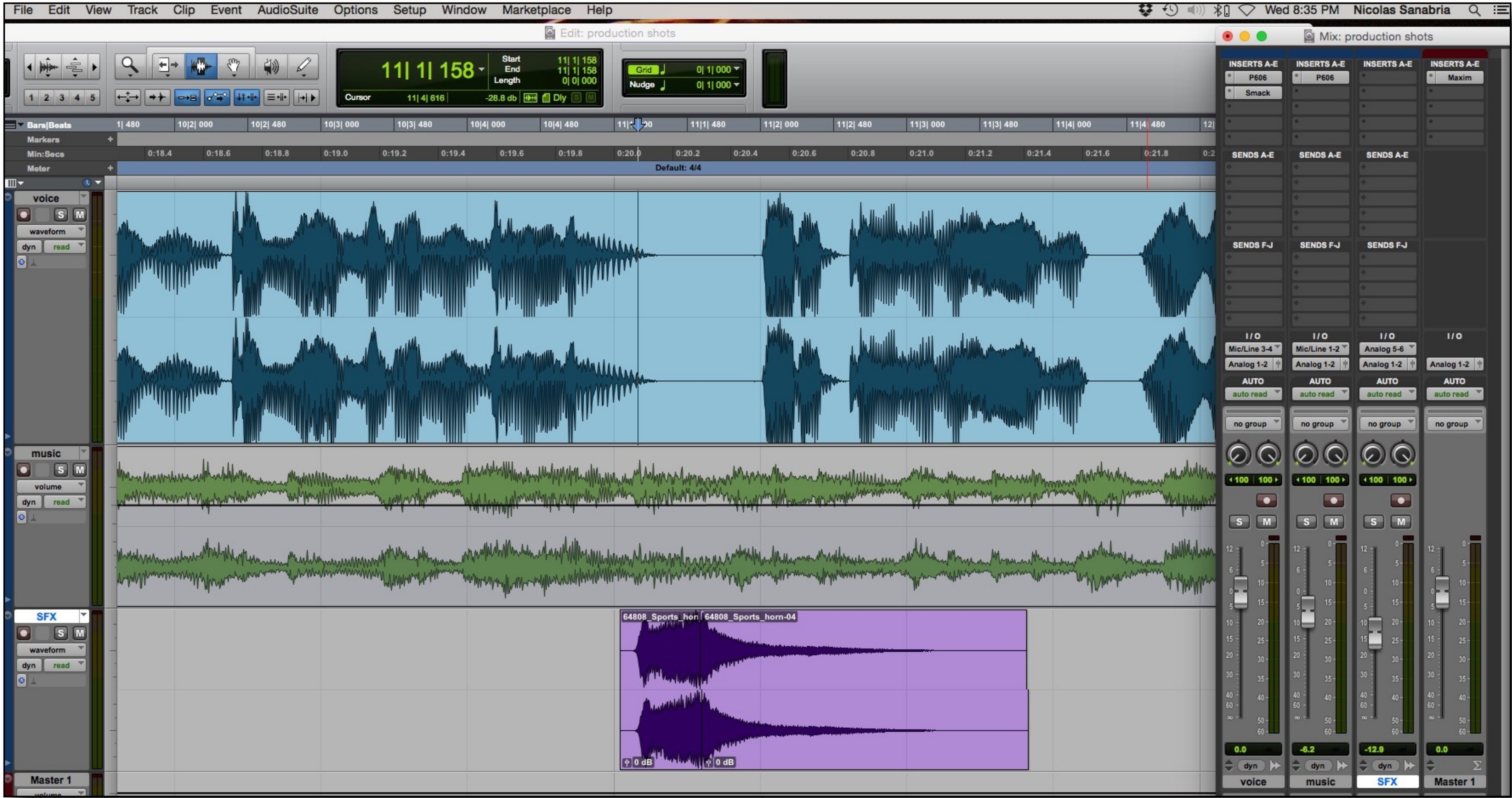


Photo courtesy of Avid

I have imported a sound effect and moved it into position.
Note it is in between words, and the hit lines up with a snare hit in the music track.

The trail off is under the next word. but that is fine, you normally want the “hit” of the sound effect in the clear as to not obscure the voice.

There will be more detailed placement thoughts in the SFX chapter. This is just the layout.

Note the level of the SFX has been brought down to an approximate level. No EQ is necessary.

Now you have 3 elements going, having a master channel is beginning to make more sense, especially with the limiter.

I always recommend a separate channel for each different sound effect.

If you have a repeating sound effect, like an explosion...(so popular with car commercials) you can copy and paste the same sound effect onto the same channel since the level of the effect will usually stay the same.

Since you will be mixing various elements, it is easier to adjust them if they are on separate channels.

Sometimes you will want to create a new sound effect by layering several separate sound effects:

The next example is a car race SFX, a separate race car pass SFX, and a crowd cheer SFX..

Note how the peak of the sound effects line up with the hole in the voice track.

You can edit more space in if necessary to create a larger hole for the SFX, and shorten

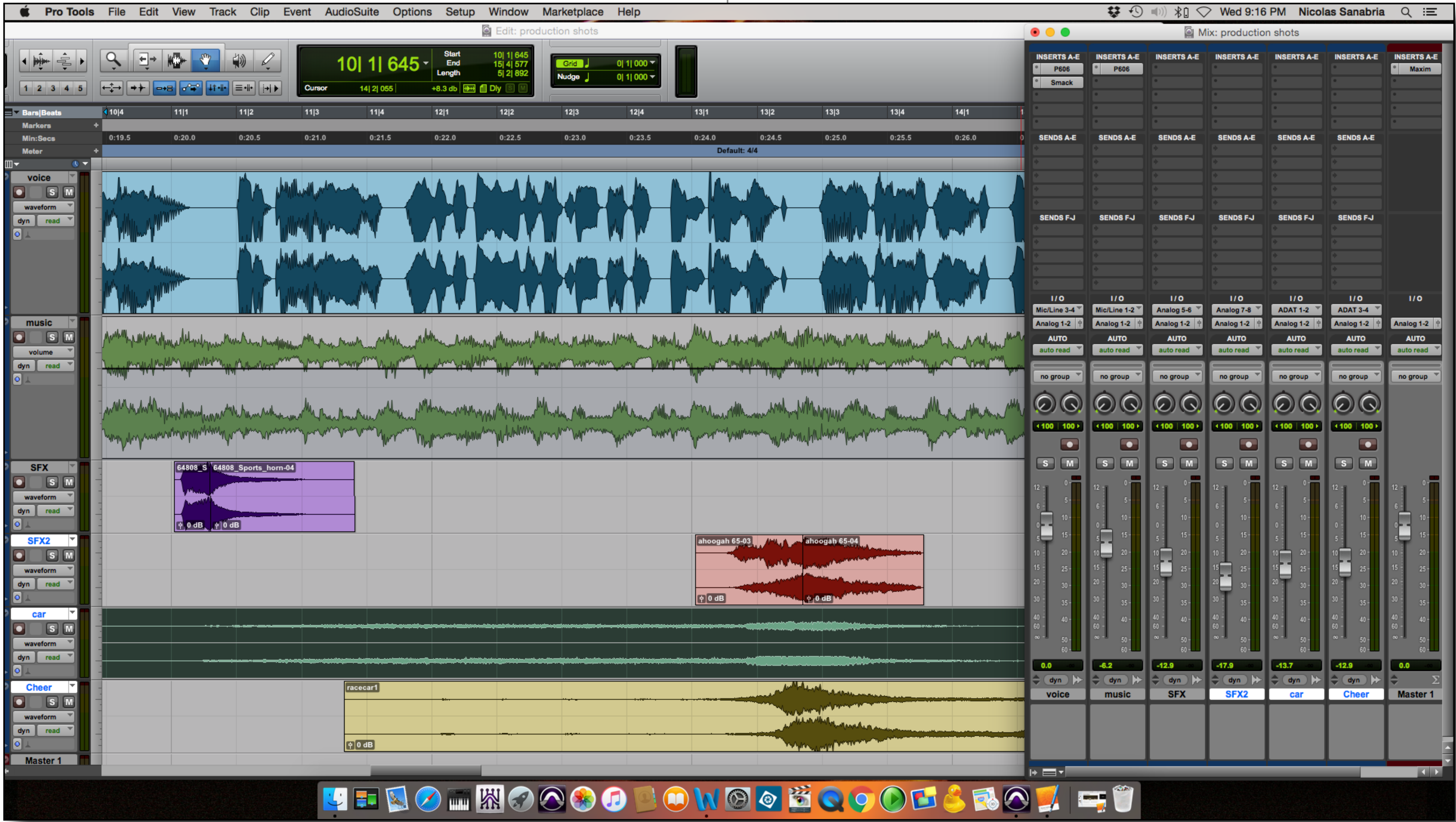


Photo courtesy of Avid

some spaces down the line to make up the difference.

If you have a sound effect, like the car coming, (highlighted track) that has a build and fade down, you can slide the sound effect so that the peak is in the hole.

And having the sound effect build and fade under the voice is perfectly acceptable.

Now the master channel is earning it's keep.

Without one, you can't know if you are running into the red: you don't have a master output meter.

Your individual channels may not be peaking, but the sum of all your channels could be going into the red.

The master channel is the sum of all the channels, and it's output meter will let you know what is your final output level.

And chances are you will need to tweak your Limiter.

Aux channel/ Buss (effects)

Photo courtesy of Avid

AUX CHANNEL/ BUSS for effects (reverb, echo, flange, etc.)

Ok. You have a new session, for a car dealer.

He wants reverb on the voice. You can either put a reverb on the channel after your compressor, and dial in the amount you want, OR you can put the reverb on an Aux channel where you will have more control.....especially if you are adding more voices (more voices coming up....this IS a car commercial, after all)

Setting up an aux channel for effects is much easier than dealing with individual reverb settings for each channel. If you are using more than one voice, they can be routed to the aux channel buss and will go through that reverb. It is also less CPU taxing on your computer.

Start with adding a STEREO Aux channel from your track menu.

Photo courtesy of Avid

Then on your voice channel, assign an output **send** to the Aux channel/ buss. Bus 1-2 (stereo)

This sends the same signal going to your master channel, to the bus.

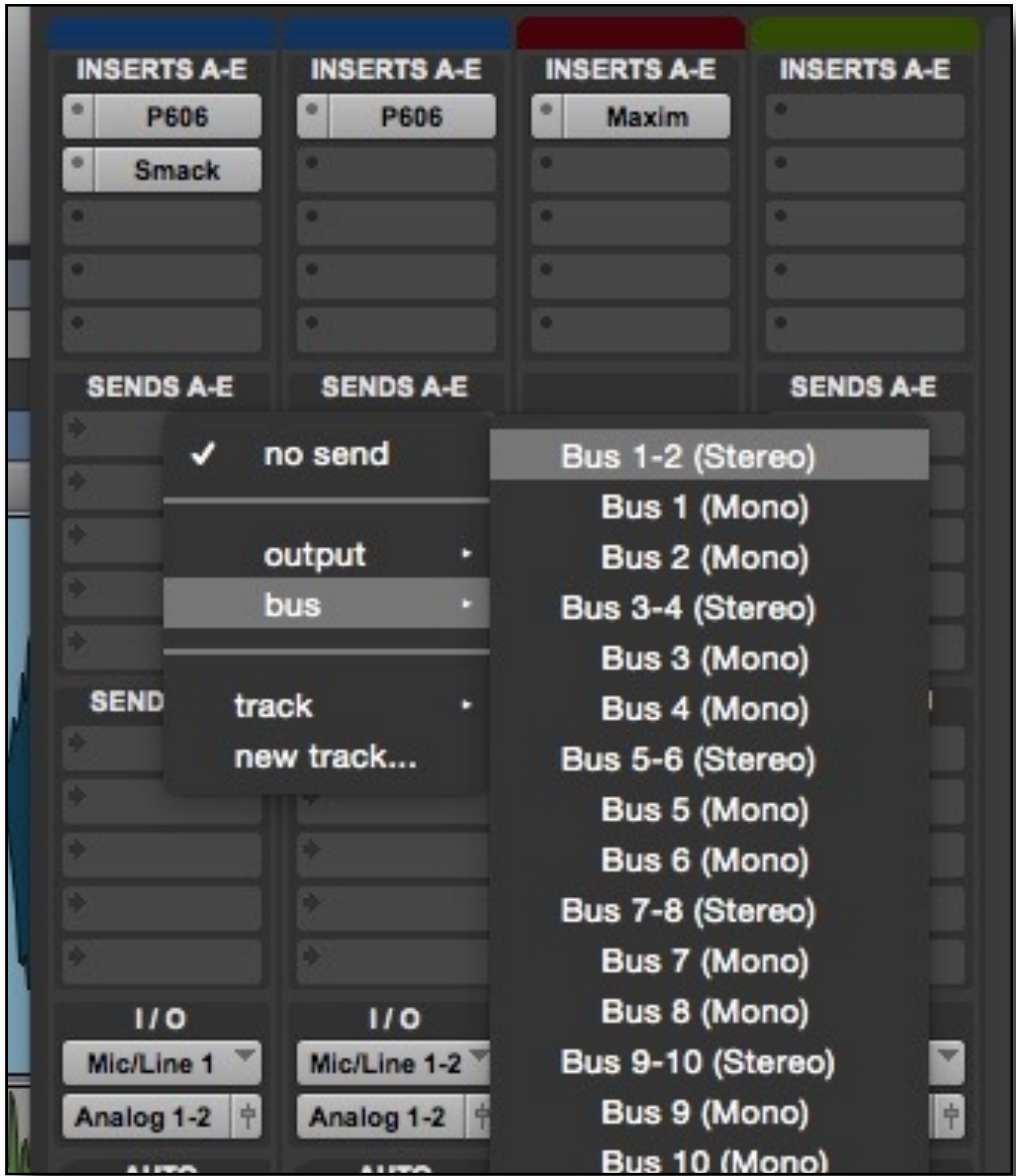


Photo courtesy of Avid

Another fader will pop up on your desktop.

For convenience, move it next to your console.

Set the SEND output to "0"

as on the example to the right.

Photo courtesy of Avid

Now you have to assign the input on the Aux channel/ buss. to receive the signal from the send.

Pull down the “no input” menu and click on BUS 1-2 (stereo)

Now your channel is “talking” to the bus.

You can have as many sends and busses as your DAW allows. And you can send multiple channels to a single effect bus, (coming up in another example) saving you time.

As before, your DAW may operate differently, so check with your manual to see how to open and assign busses.

Photo courtesy of Avid

Now add a reverb from the plug-in menu to the Aux channel/ buss.

And I have moved the channel to the left of the Master channel, (personal preference)

Most engineers have channel set up preferences; some like the master before the busses, totally up to you.

I have used a non-linear (gated) reverb that has a short tail, which does not interfere with the intelligibility of the voice as much as much as a reverb with a long tail; too much reverb can muddy the voice.   If the reverb is too “dark” you can add an EQ after it from the plug-in menu and add some high end.

Adjust the volume slider to get just the right amount of reverb.

Photo courtesy of Avid

MULTIPLE VOICE EFFECT:

Most car dealers believe "If you ain't yellin', You ain't sellin'." So the "US 30 Dragstrip" multiple voice effect is a must-have in your bag of production tricks.

Start by adding two (or more) mono channels and position them under your main voice track.

Copy and paste the audio phrase you want goosed up (tech term) onto the open channels and delay each one slightly.

Put a dip in the EQ in the lower frequencies of the echo voices. You can also lower the pitch on one or both to give it that real schmaltz.(another tech term)

Photo courtesy of Avid/McDSP

Then pan one to 9 o'clock, the other to 3 o'clock to give it some spread.

And push the aux sends on the echo voices higher than the main voice.

If you want to add an effect (like repeat) to just the echo voices, add another Aux channel.

set it to bus 3/4 and your sends to 3/4

Put a delay on that Aux channel. Adjust.

Now you can have delay on the echo voices only without having it on the main voice.

There are many other ways to do this, but this is my preference.

Photo courtesy of Avid/Slate Digital

You can also use an Aux channel as a bus or submaster:

If you have a bunch of sound effects, you can route the output of the sound effect channels to one Aux channel: that way you have all the sound effects controllable by one fader. You have control over the master level of all the sound effects, but can still change individual volumes on the individual sfx channel.

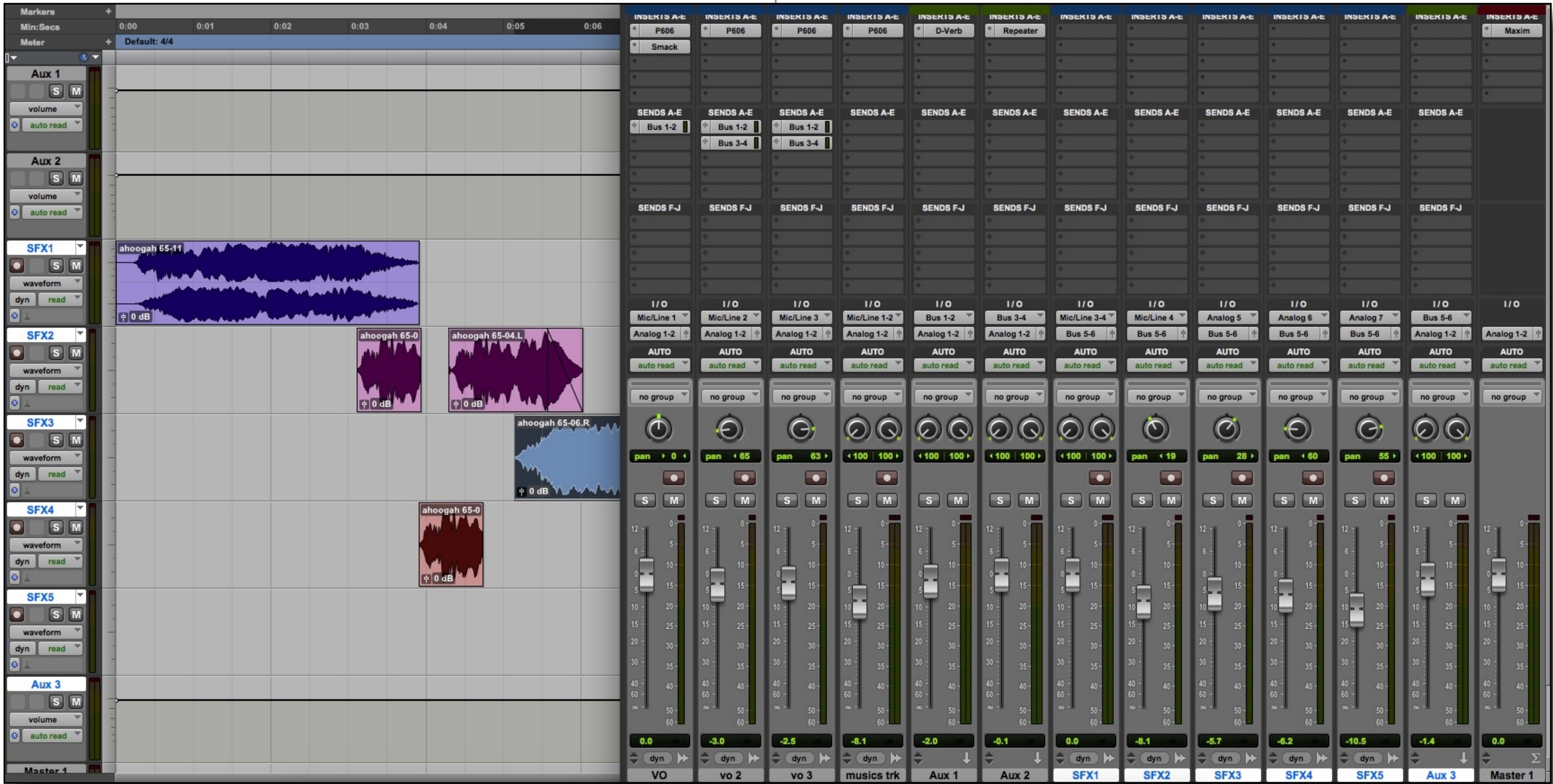


Photo courtesy of Avid

First thing is to open another Aux channel, and assign the input to bus 5/6 (since we already are using bus 1/2 and 3/4)

Now, change the **output** of all your SFX channels from Analog 1-2, to bus 5/6

All your SFX channels are now being summed through bus 5/6 before going to your master channel.

If you want an effect on the SFX submix, open another Aux 4 (7/8) and put a send on Aux 3, with the effect on Aux 4, just like you did with Aux 1 or 2.

Or you can just put the sound effect on Aux 3 and dial it in.

I know. It;s getting complicated, but trust me, after you do it once, you will go: "what's the big deal?"

Photo courtesy of Avid

As can see by the figure on page 30, you are starting to get a lot of things moving. A submaster can take the 5 SFX elements down to one fader.

Some engineers put their submasters to the right of the master output, and have the voice, music, and sound effects on 3 submasters; which allows you a quicker way to balance the elements. You only have 3 faders to balance vs the 9 channels on p.30.

If you are doing a complex mix for television sweetening, or a fully produced audio book, or even a multiple voice commercial, this trick might speed up your session.

And you still can change the individual channels feeding into the sub...for instance, if sound effect 4 is too soft, you can bring that up on it's channel.

This just gives you a bit more control. But this is just another way of getting to the same spot: a great mix.

Photo courtesy of Avid

EFFECTS: REVERB/DELAY

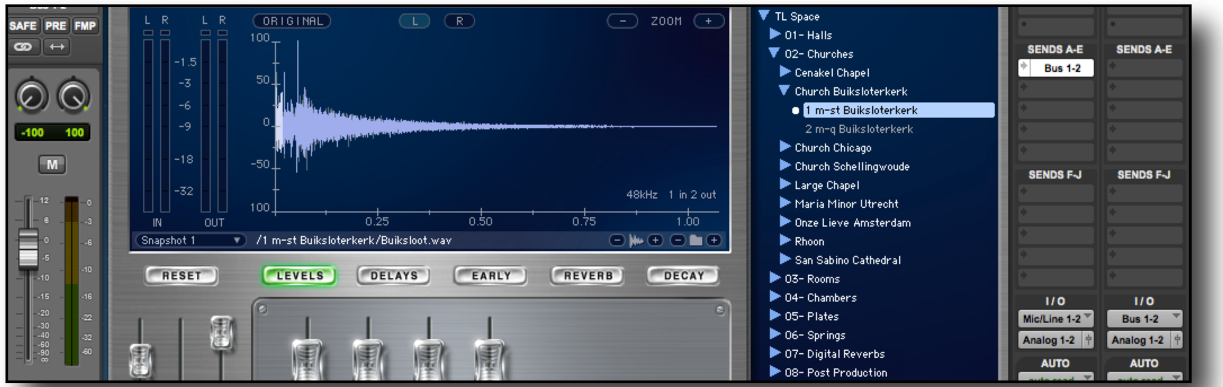


Photo courtesy of Avid

Reverb and delay add depth to the sound and simulate “spaces” to give your spot the illusion of being somewhere. Use it sparingly, as you want to create an illusion, not draw attention to it. (Unless, your really want to do Surf Music.)

Try some of the presets and tweak from there.

Experiment with the parameters, like pre-delay, which is the time from when the sound initiates, until it hits the first wall, or first reflection.

Reverb is the multiple reflections after the first reflection. You can set the amount of decay to simulate anything from a small room, (usually labeled “club” in the preset menu) to stadiums or caves on the other extreme.

And there are usually oddball presets like “vacuum cleaner hose” to “telephone”, “bullhorn” and other effects that can be used to create more authentic audio theater of the mind.

Delay is distinct “echoes” (HELLO, ...HELLO,.... HELLO) that you can control in the stereo field, the length of time between echoes, length of the decays. and how soon you hear them. These are different than reverberation, but can overlap a bit depending on settings

More of the finer points on Effects are in Chapter 5 Plug-ins; section 5 Reverb/Echo/ Delay

CHAPTER 3

Advanced sessions

Photo courtesy of Avid

Re-recording strategies

Re-recording at another studio

Dialogue replacement

Directing talent

Remote sessions

Re-Recording 2 channel

Photo courtesy of Avid

RE-RECORDING

Just about every talent has been called in to do a pick-up, correction or replacement line/ paragraph at a later date. This chapter will outline some techniques to match audio for a seamless insert.

Try to recreate the same set up, same mic, same location in the booth.

2 TRACKS, MULTIPLE TAKES IN A ROW:

Usually, it is quickest just to completely record the entire paragraph that needs changing.

An entire paragraph is a new thought, and even if the audio sounds just a tiny bit different, the listener will probably not notice. Also, it is sometimes easier to match an entire paragraph to the same volume/EQ of the original, for instance, than to try to match a sentence or single word.

Photo courtesy of Avid

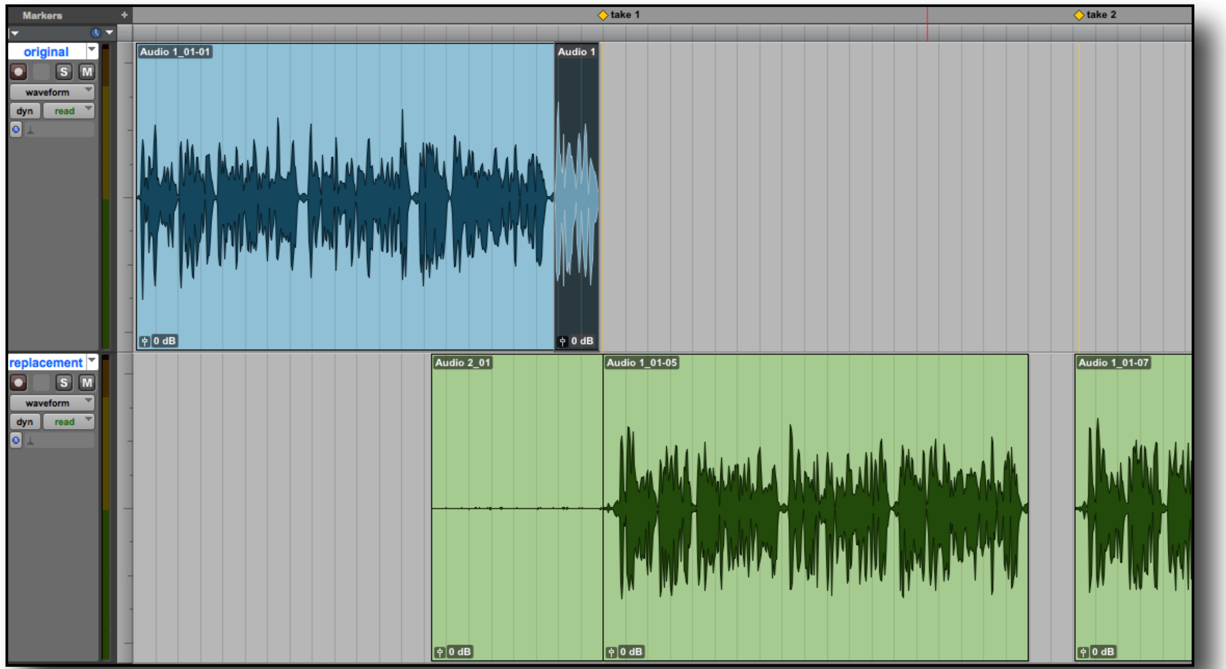


Fig b: cutting and pasting in a replacement sentence.

SESSION SET UP

Set channel one as the playback channel for the original section, channel 2 is the record channel for the new section. Let the talent hear the original section several times to get a feel for how they read it the first time. Let the talent do 3 takes in a row. After a few "3 in a row" takes, play the original again to make sure the talent is still in the same range and pace as the original.

Depending on how close the talent was to the original take in pace, inflection, etc, you may chose just to cut in a replacement sentence from one of the "3 in a row" takes. (fig b) By having the entire paragraph recorded, you have more options: to completely replace the entire section, just a sentence, or even a word that matches the original.

Always cut in the new section to check that the match works before releasing the talent.

Multiple re-record tracks

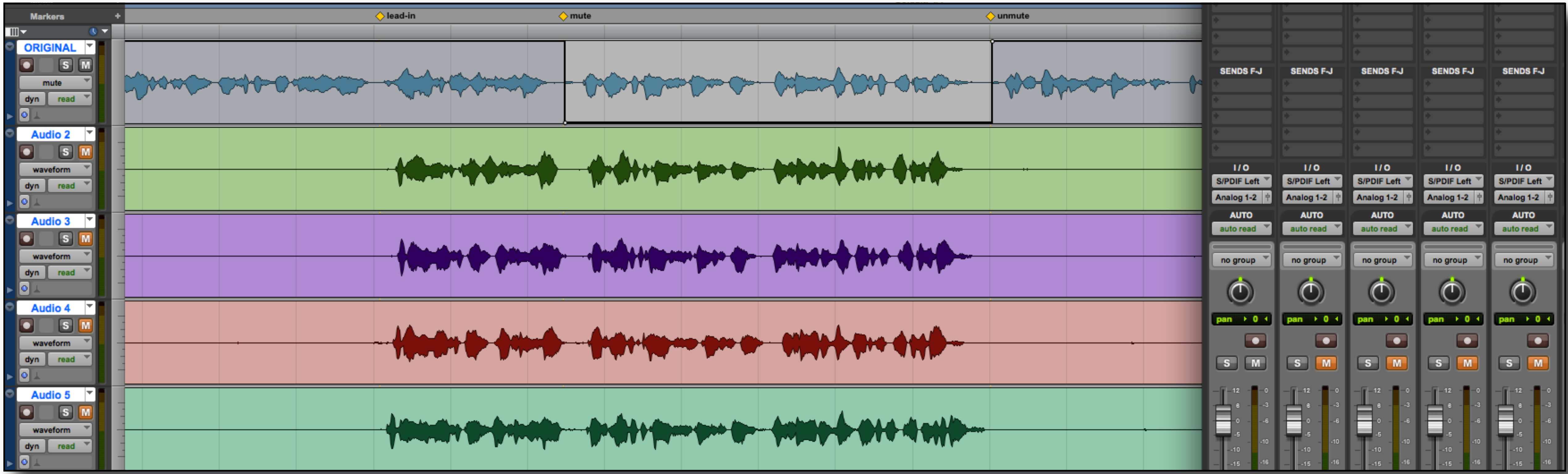


Photo courtesy of Avid

MULTIPLE TRACKS WITH ORIGINAL TRACK LEAD-IN

Set up a new session with the original track as a guide track, and multiple tracks for several takes of inserts. Use the same plug-ins with the same settings as the original track.

The guide track can be played back in both sides, or just one side of the headphones, it is up to the talent's preference. They get "a running start" and begin talking along with it, at some point, for pace, inflection, and energy ahead of where the new section will begin.

Where the new/insert section begins, the original track is either muted, or the audio has been cut beyond that point.

The talent continues into the new section. Some talent prefer to listen and not talk along with the original, and just read the new section in the clear. If that is the case, mute the track earlier so the talent reads into the insert section; you may find a more natural cut-in point before the actual insert.

SESSION SET UP

Be sure to match the insert track(s) plug ins and settings to the original track.

Create multiple new record tracks just for the insert takes. Record the talent reading along with the lead in too.

As you finish a take, mute it, and take another pass on the next channel.

This will help the talent maintain a rhythm and give better and better takes.

If you have several that seem right, listen to the original track and and then un-mute the track you think best at the insert point (this can be automated on most DAW's.) Keep the audio prior to the insert as you may find a more natural edit point.

At this point you can either cut and paste the best take into the original. Or you can just leave it where it is, using mute and un-mute automation to make the switch.

MATCHING

CUT AND PASTE/PROCESSING

You can cut and paste the insert into the original track. If however, you hear a difference, there are two ways to match. If there is a noticeable difference in volume, for instance, you can process the audio by increasing gain settings using channel automation (volume) or by processing to match the original, but this takes some trial and error, undoing, and trying again until you have a seamless match. Processing is a destructive process, in that you permanently affect the recorded take, which cannot be undone once you have saved the session.

SECOND TRACK/PLUG-INS

Or simply keep the new audio on the second track. The flexibility of having the new audio on a second track allows you to tweak volume levels eq settings using plug-ins to more closely match the original. Then, if you wish, you can process (print) changes to the audio and copy and paste it into the original track, or just leave it on the second track. If after mastering (bouncing) you decide you still need to tweak, it is an easy matter to fix if it is still on the second track. Plug-ins are non-destructive, in that you can change parameters without affecting the recorded take. The settings are remembered when you save.

You will probably have to re-space your original track after the insert to accommodate the new section. You may even be able cut in just the portion of the word to be replaced, (in the case of a mispronunciation, for example) where a cut and paste is usually seamless. More on complex editing in the EDITING chapter.

Re-recording at another studio

On occasion, a new section cannot be recorded at the studio where the original was cut. Here are a few tips to make the sound match more closely to the original, lessening the amount of tweaking needed to make it seamless.

Give the other studio as much information as possible: type of mic, type of pre-amp/interface, program, recorded flat or with pre-compression/what ratio/range, sample rate (44.1khz), etc. The more information you give them, the closer you can get to a match. Chances are they will not have the same equipment you do, but their engineers should be able to find a pretty close sonic match.

Send them an mp3 of the read so they can play it for the talent. In this case, have them record the entire paragraph several times. Have them send you the file in a professional format: .WAV or .AIFF. Most programs will let you import a file and will convert it to your format.

Avoid an mp3 (although it is easiest to email) just because this is a lossy conversion that is not as high quality as the file you originally recorded. It may sound different after importing and conversion into your session. (you can do it, but it takes a bit of eq-ing to get close)

However the large files above will need to be sent to and downloaded from an FTP site, or sent via a large file transfer service like Largefileasap, Drop-Box, YouSendit, etc. Check the web for which one would make the most sense for your needs.

Import the audio into your session on a second track and use plug-ins to match the sound to the original. Start out with the same plug-in settings as on the original track. Adjust volume to match, play with eq and compression settings to get as close a match as you can get.

Dialogue replacement

If your DAW supports video import and playback, great. This will be much easier. But even if it does not, you can do this with just audio.

VIDEO/FILM: LOOPING/DIALOGUE REPLACEMENT

On occasion you will have to replace a section of audio from a lip sync video or film. This can be difficult, but there are some strategies to accomplish this. First you need to import the audio track to a session. Import audio before and after the target section so you can get ambience from the track in case you need to add it under the dialogue to match the original.

If you have some ambience in the clear that you can copy and paste, all the better. Most programs will also import video, and may be used by talent who want to see the section they are looping. (It was originally called "looping" because they created a loop of film that repeated over and over while the talent recorded take after take to the tape recorder.)

Some talent will memorize the line and will want to look at the video to lip sync. They may go line by line.

Some just want a guide track in the headphones that they can mult to, just as someone would sing a harmony part with a lead vocalist. Some want both.

Either way, you should put a countdown on another track "5, 4, 3, 2, 1," or roll the video from the same point every time: you just need a cue point for the talent to start at the same point every time.

Copy and paste the countdown and video multiple times so the talent can fall into a rhythm.

Or you can have multiple tracks stacked under the original track and video, (see *example below*)

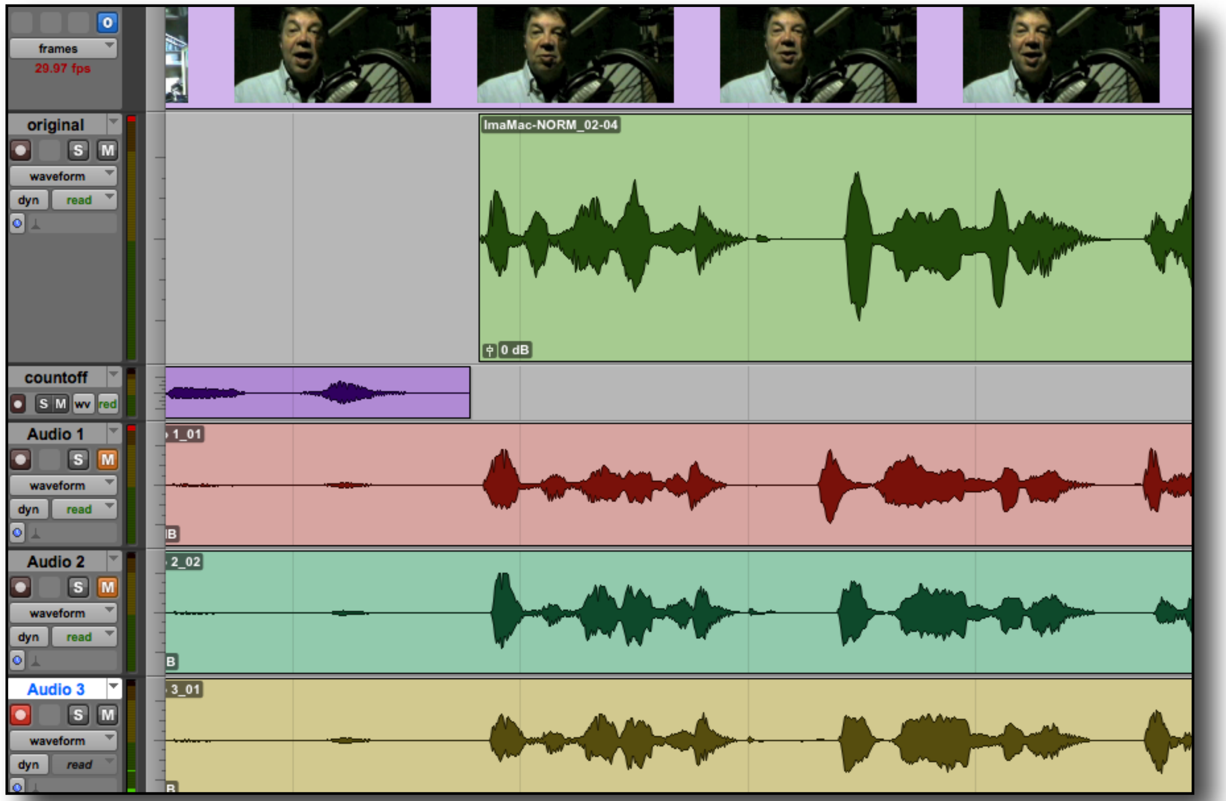


Photo courtesy of Avid

however, this set up is not as conducive to the talent falling into a rhythm since you have to stop, mute, arm the next track and record.

When you have 3 or 4 that are very close and you are all happy with the read, now comes fine tuning.

EDITING

Move the waveforms so they line up with the guide track. Starts are most important at this point. You will notice some do not end at the same point as the original.

Get as close as you can by using time compression/expansion, editing, or using a word from another take, so you get as close to the original as possible. Starts and endings of a word are important to maintain lip sync, as are some obvious vowels and consonants within the word.

You can tighten up or expand a portion of a word using time compression or expansion so that the word syncs perfectly to the original. But you may have to experiment, if it sounds funny when you play it in context of the sentence, you may have to find another take that is closer, or compromise a bit.

ADR (Automatic or Automated Dialogue Replacement)

There are programs, such a Vocalign, that will do the fine tuning for you by automatically compressing and expanding each word to sync with the guide track. This is much faster than the above editing strategy. It is best to get as close as possible lining up the words first which will lessen the chance of it sounding glitchy. If you are doing a lot of this type of work, or if you frequently add harmony parts with singers, this is a worthwhile program to have. There are several levels from LE (limited edition) to Pro, with corresponding feature/price differences.

AMBIENCE

Once you have a track that is synced to everyone's amazement and satisfaction, you need to match the ambience. Take the ambience and put it on a separate channel, you can raise or lower the level so it closely matches the original. Once you are satisfied, double check on headphones.

On occasion, there may be an echo or room sound that is suddenly missing from the replacement portion.

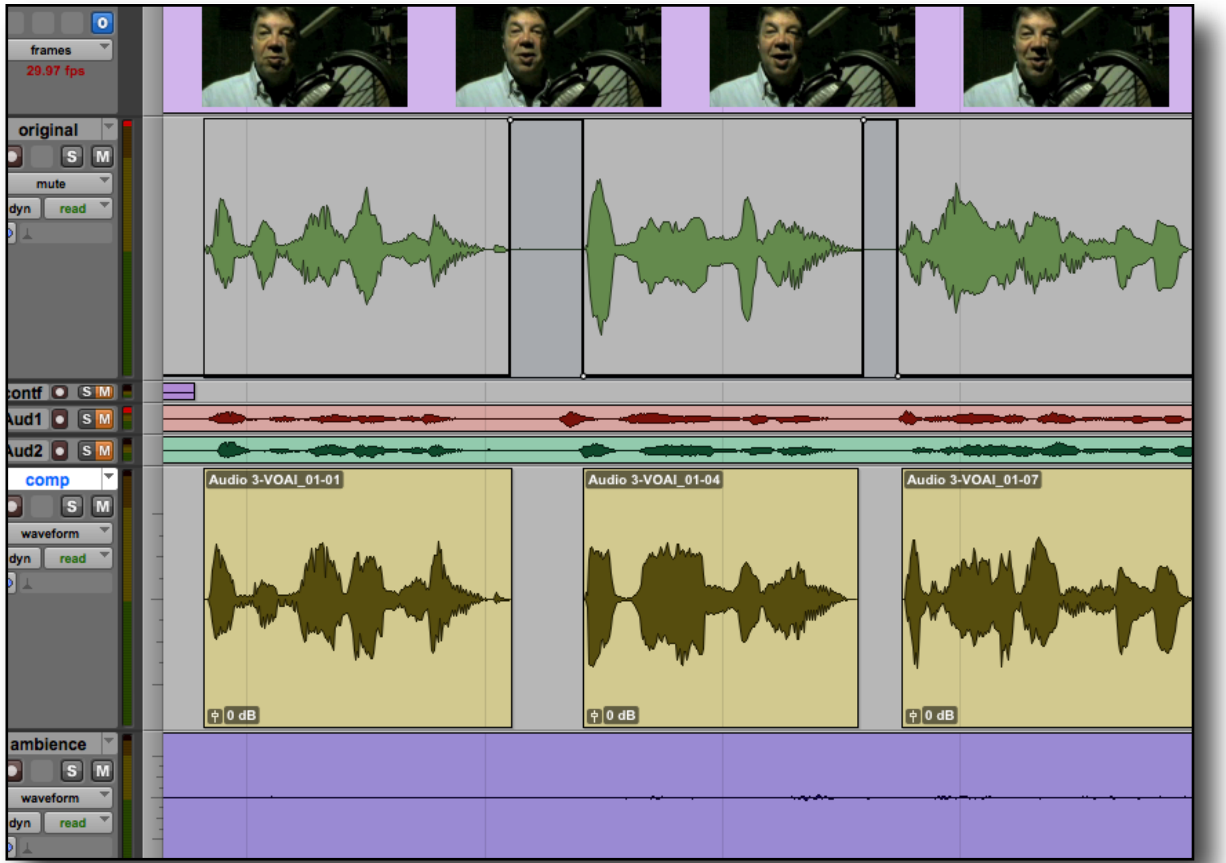
Strategy one: (fig. a) un-mute the original track in the spaces between words only, you will hear the echo from the original take, mix it until it sounds natural.

Strategy two: copy and paste the spaces immediately after the words from the original into the replacement track, use cross fades if necessary.

Or try copying and pasting ambience from a clear section elsewhere on the track.

Strategy three: find a reverb, room simulation, or echo plug-in and experiment until you have a match. This can be horribly time consuming, and frustrating, (you have to tweak multiple parameters) but once you have a close enough match, no one will notice.

You may have 3 things going on simultaneously: the dialogue replacement, the ambience, and either the room sound/echo from the original track, or the recreated room/echo.



(fig. a) Strategy one: un-muting the original track (top/green) in spaces after words. *Photo courtesy of Avid*

MASTERING

It is best to create the replacement audio track by starting before the insert using the original track as a lead in. This will allow the video editor to slide the track in sync with the original and either cut in or crossfade at the replacement points. Some will request/supply a beep as a sync point. Discuss how your editor wants to sync, since it varies from shop to shop. For most film/television work, 48k .AIFF or .WAV are standard

If it is a short video, mastering the entire track is advised.

Also, if you have Final Cut, or similar video software, you can do layback yourself using an .AIFF or .WAV file. Line up the new track under the old track, and mute the old version, and export the video with the new soundtrack.

This is an example of a quick and dirty video with layback soundtrack. (This was a parody making fun of the Vista commercials) Although these were the original voice takes recorded both in the camera and in protools, they were mixed, processed and flown back into Final Cut. The in-the-camera mic track was used for sync, then muted.

https://www.youtube.com/watch?v=RfL4Z0pPELw

Directing Talent

This section goes into the finer points of getting the read you need from someone else. You move from talent, to director. How-to's are offered. What to listen for, and what to pay attention to. Many of the suggestions and examples can be used even if you are self-directed talent. And there is quite a bit of personal opinion offered in this chapter on everything from grammar to pronunciation issues.

This section is mostly opinion based on years of experience.

The way we work changes, but the final result still has to work for the listener.

Communication is the goal.

All the components need to work together to create an artistic audio track that the listener accepts.

If there is an ear trip, or one element that distracts the listener, you have lost their attention.

It can be a technical glitch, such as a mouth noise, the music track is too hot making the listener strain to hear the announcer, a mispronounced word, or just a plain, bad reading by the announcer, which loses your audience.

Following are some things that move you into being a producer/director.

Oh, and take to heart some of these suggestions for your own reads.

CRITICAL LISTENING:

There have been one million sessions where after the talent has left, someone realizes a word was mispronounced in every take. No one was listening. The engineer was looking at levels, the producer was on the phone, the writer was rewriting the script that was too long and waiting approval from the client, the intern was thinking about dinner, and the talent didn't have a clue since no one corrected them and they were concentrating on getting a :66 into a :60.

In a case like this, everyone shares the blame. You just have to assume everyone else is asleep at the wheel and pay attention to every nuance of the read.

Something as slight as the incorrect emphasis on a name can slide by, but after hearing it over and over in post and then on the air, it begins to irritate like fiberglass in your underwear.

PICKY, PICKY, PICKY.

Language is fluid, and evolves. You may find a particular pronunciation or phrase, or slang grating, but it may be used in context to address a different demographic, or it may be appropriate in another market, such as a regionality: kitty-corner/catty-corner.

But if something is out of whack in the read, it becomes an ear-trip, a speed bump that causes you to lose attention. At that point in the script, you stop hearing the message and are distracted by something in the read that makes you pay more attention to it.

It may be a common, flat out mispronunciation that has become acceptable by its frequent misuse, such as “eXspecially, eXcetera,” and words that do not exist, but are heard frequently such as “irregardless”. All of these can be pretty easily fixed in post, but will take some editing time that should not have been necessary if anyone had been paying attention.

People tend to repeat bad pronunciation habits they have heard in common conversation. “I’m going to get some beers” is incorrect and the person hearing/saying it knows it is incorrect, but with repetition it becomes habit. Slang is a repeated word/phrase that becomes acceptable though constant use and reinforcement. And incorrect pronunciation of words and phrases, or even misquotes can gain acceptance by default.

“You’ve buttered your bread, and now you have to sleep in it”

ONE EXAMPLE:

A pronunciation anomaly that is becoming more prevalent is the emphasis of the modifier instead of the subject in a name, like “McCormick PLACE”, or “Bob’s House of FORD”, or in a common phrase like “garage-door OPENER”. Even “open HOUSE”.

There is also an added annoying emphasis in tone, with an octave higher pitch on the modifier.

A well known news personality constantly says “eye OPENER”

In these cases, the part with the primary importance, or focus should be emphasized as in “OPEN house”. Or all parts given equal emphasis, as in “OPEN-HOUSE” One trick in a name like “JIMMY’S House of toast”, should almost be a stair-step down with JIMMY’S being the primary emphasis. His name is primary, toast should be the lowest. Or JIMMY given the most weight, and “house of toast” given less, but equal emphasis unto itself.

That was one teensy, tiny example of speed bumps in a read. You have to assume that everyone else in the room is not paying attention, and that you have to listen critically to the performance. If you have to play second-chair to the producer, then address them with your concerns: “Not to be picky, but shouldn’t that be pronounced, WARrantee, and not warranTEE?” If they say, no that is the way it should be pronounced, and later they have to re-record, that is on them. But chances are they will say, “Good catch, thanks” (Sneaking into studio etiquette, which was not my intent and off topic)

If you get stuck on a pronunciation of a particular word, always check the dictionary or go to a pronunciation website for guidance.

HEAVY HANDED READS:

You don’t have to beat people over the head with your read. In fact, a more one on one relaxed read has been the norm for the last 20 years in advertising.   No one wants to be yelled at or sold to. Unless you are doing a drill sergeant spot and you are supposed to be that character, then it’s ok.

THE BIG "YOU":

When "YOU" in a sentence is over emphasized to make sure the listener knows the spot is addressing them. "get the job YOU want" or "We'll make sure YOU get the money YOU deserve!" Mindless direction invented by a Psych 101 drop out.

THE BIG "AND": It is a bad habit repeated by announcers throughout history. I don't know who started it, but it continues to this day: "You'll get power steering, power brakes, power windows AAAAANNNDDDDD a great deal more!"

All you ever remember is that Don Pardo-ish "AAAAANNNNNDDDDD."

"Percen-TOFF"

I hear this more often than not. Soften the "T" so it doesn't come out sounding like TOFF. Most of the time this is being read fast, so even if you have to cheat, percenOFF sounds better. Or edit: lower the gain of the "T" to match the level of the "percen" and tighten the space in post. If you are voice talent, practice this over and over so I never have to hear it again, thank you.

OTHER ANNOUNCER TRIPS:

"First Street" is impossible to enunciate at typical spot pace. You just have to say Fir Street and keep moving. Cellular is another word invented to humble announcers. Every announcer has a word combination-trip nemesis. But there are ways around most trips: First just repeat the phrase 4 times in a row. This usually does it. If not: Alter the pitch sequence, or "notes" that you are reading. If you always start on a high note on a problematic phrase and read down, try altering the notes/pitch to go up in the middle word. Come at it from a different angle: alter the rhythm/cadence by putting the emphasis on a different word in the combination. Beyond this, have them change the script, which sometimes works the best.

And a tip from the late Don Lafontaine, "If you are having trouble with a word or phrase at speed, exaggerate your mouth movements and it will come out sounding fine." And it works.

GRAMMAR:

Always check the script before the session and flag any possible trips. We had a script that had been approved by legal that said "All the employees in the Blah deBlah Ford Service Department is there for your every need." The talent suggested changing the "is" to "are" or start with "Each employee" to make it grammatically correct. They said it was approved by legal and had to be recorded verbatim. Just as a safety we recorded the sentence in question several ways. They called back the next day and said "Change it ... ….and thanks." Most clients will change copy once they realize the mistake, but you have to point it out, or at least pose the question "Are that right?"

But there are times you need to record it both ways until you get a definitive answer on which is correct. It takes a few seconds to do an alternate take, and an hour or so to set up another session for a re-record.

If you have a friend who is a grammarian, keep them on speed dial. If you did not pay attention in high school and college English classes, be sure to hire a voice talent who did.

ALTERNATE TAKES:

Sometimes the talent says it one way, you think it should be said another way, and the client does not have a clue as to which is right. Alternate takes. Perhaps you feel the emphasis should be on the client's product, the talent thinks it should be on the feeling you get from the product, and both have their merits: do both. Pick later in post.

AND YOUR POINT IS????

And sometimes the talent just doesn't get the purpose of the spot as a whole. They are doing a brilliant job of milking the "feel" words and it still isn't working. It doesn't seem focused. They are just reading words and doesn't ring true. Some blame the writing, but usually that's not it. The wrong things are being emphasized. Sometimes the talent is just reading the script without understanding where it's going.

Take a step back and ask what is the point of the entire spot in a few words. "This shampoo makes me feel like a movie star" or "We have to sell all of these Azteks" or "This is how the HEL- 9000 works" Talent should be asking this every time they pick up a script, but sometimes you may have to intervene.

With that in mind, the read may take on an entirely different tone, usually does, and is more focused since the talent gets the "mission statement".

If you are self directed, you have to be keenly aware of any pronunciation questions and have them addressed before the session. It may be a name, a regionality, (INsurance, vs. inSUREance) or just something you always pronounced a certain way that is incorrect.

If you are not working on phone patch or being directed real time, go over the script and flag anything that may be a problem, and ask them how they want it prior to recording.

Or record several versions of the line in question, and edit in the one they approve.

MOUTH NOISES:

You may be listening to the pace, the tone, if the actor is pronouncing the words correctly and hitting all their marks in the script, while watching levels, but you missed the mouth noises. The clicks, the smacks, the breaths. You can fix those in post, but you just cost yourself an extra hour of editing that a bottle of water or slice of green apple would have avoided.

NOISES:

The talent was waving their arms and brushed the script, hit the music stand, or the corduroy jacket they were wearing sounds like someone using a washboard in a jug band.

Stop them and do another take. Tell them to lose the jacket. Don't tell them to stop waving unless you can still pick it up, since most talent use gestures to help with hitting their marks in the script. Same goes for clanking jewelry and watches, tell them to put their items on the music stand and remind them to take them when they leave. We had a collection of earrings left by talent who took them off to use the headphones. And never a good pair, either.

WORKING WITH NON PROFESSIONAL "TALENT"

We all have or will have to deal with a client whose friends tell him he has a great voice and should do his own commercials. Or is so deluded that he thinks he can do a better job than a professional. Your job is to make him sound as good as possible. It may require hours of editing, but it can be done. And there is also a certain "suspension of belief " when working with non-pros, especially sports figures and athletes.

PUT THEM AT EASE:

Most non pros may become nervous the minute you say .... "RECORD".

If this happens, hit record and don't tell them. Say you are just going to run through it a few times until everyone is comfortable with levels, etc. Keep the talkback open on another channel and mark takes with a track marker. When you have a couple of good ones, then try some actual "takes". Sometimes they surprise you and you get something better. If not, you have some good takes to edit, and the client/voice is usually relieved it is over.

We had 2 famous athletes (new to recording) hired to do a spot who were complete stiffs. They couldn't read two words without pausing and picking up the line and pausing and stammering. We finally told them to start each sentence with, "Hey, **** you, Bob" and we'd cut that out in post.

Next read was like 2 old pros. Totally relaxed them and they were able to have fun with it. We cut the expletives/but hung on to them for an out-takes comedy reel.

Coaches, and ex sports figures are the most fun. They don't have to read the script perfectly, everyone knows how they really talk and wouldn't expect a polished read. So just have fun, it doesn't have to be perfect, but it does have to be "them". And generally, they are pros at this, and are in and out in a few takes.

WARNING:

Do not use libation to "relax" non pros: their enunciation will go right in the toilet.

If they are nervous, tell them to take 3 VERY deep breaths in a row, all the way down to their toes, and exhale slowly. This is another trick that stops the short breathing that is associated with anxiety. If the non pro is unable to complete a sentence in one breath, this can help. And noting to take a big breath, if necessary, before a long sentence will also help to make the read smoother. And this will also lower their pitch, which also goes up with anxiety.

COAXING:

Use every trick you can to coax a great read out of every talent. Maybe a joke to relax them, a light hearted, enthusiastic tone for the whole session, positive props to the talent, etc.

This is all studio etiquette, but keeping everyone in a positive mood is what makes for better results. In most sessions, when everyone is upbeat, it is fun from start to finish.

LINE READS

On occasion, the talent, even pro level, may not get the interpretation right. If giving them suggestions on how to read it does not get you any closer, you may have to step in and give them a line reading so they get how to read it. This is a last resort, but if all else fails, it can work as long as they can mimic what you have said.
Or understand what you are getting at.

EDITING A NON PRO TO SOUND NATURAL

Your comp track from a non pro may wind up looking like Frankenstein with even pieces of words cut together. A good place to start, especially if the talent was unable to do a complete take, is to go in the booth after the talent has left, and record a guide track yourself, reading it the way it should have been read.

Now you have a guide track and can assemble the best takes from the non pro to closely match your guide track.

You can use time compression to shorten words that were drawn out, or use it to extend words that were short. Try to match the comp track to the guide track. Generally the waveforms of each word should start at about the same spot for the guide and the comp tracks. Then listen and tweak. It may not match perfectly to the guide track once you are done, but it will be so much closer to an acceptable read, that you may be done at this point. This may take hours, but you are stitching together a read from raw materials. You may get a very good read done, only to find the whole thing is now 1:02. Time compress it. It will sound surprisingly better. Most non pros tend to talk a bit deliberately, and this makes it sound a bit more natural, and energetic. In fact, you may want to try compressing it even if it is to time, just to see if you can get it to sound more natural.

PROGRAMS:

There are programs like Vocalign, that will align your comp track to your guide track digitally. You get the pieces as close as possible, import the guide track into the program and then import the comp track into "dub". Hit align, then process, and all the time compression/expansion and alignment of words is done in seconds. There is an LE version that is quite affordable, (the Pro version has more features and a higher tab); if you do music and record back up singers/mults, etc, this will snap them into perfect sync., and is a great 2nd party program to have.

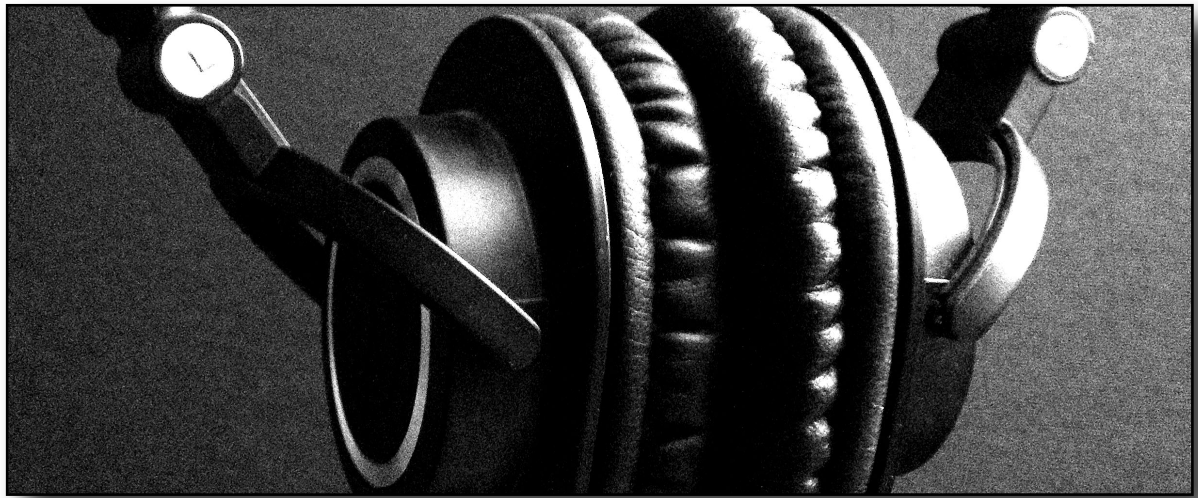
CORRECT TIMING OF SCRIPTS:

Most writers hit the stopwatch and read along with a script in their head. This is completely wrong. You have to read it out loud. And even that is not accurate. Most writers are not talent and will slur words in a way that talent would not, and may not put in pauses, etc. so their "timing" can be way off. Plus the natural pace of the talent you pick may be different than what you are expecting. But there are some general guidelines.

WORD COUNT:

This is a pretty easy one: 140 words for a :60; 70 words for a :30. This gives the talent enough time to act and breathe. After all, you hired an actor to read life into the script, don't make them try to set a record for most words spoken in :60.

If it is a high energy read, 160 words for the :60; 80 for the :30. More or less depending on your talent. This is a good jumping off spot, and you may find that a particular talent can sound relaxed reading 180. If you are not sure, send an audition to the talent you are considering. Most will be happy to work with you on this.



YOU ARE OVER: TIME COMPRESSION

If your talent cannot get the read to time, you can compress without noticing if it is 1 to 2 seconds over on a :60 . More than that, you should cut copy.

Or have the talent take a huge breath before a paragraph, and read as much as possible with one breath. You can edit these out later. This does not yield the best interpretation, but will cut seconds off the read. (Also see EDITING: Tightening a take without compression.)

Or have the talent lower his/her pitch and read with a "flatter" more relaxed read. You can pick up several seconds without sounding as rushed and this will time compress better without sounding unnatural. An enthusiastic read time compressed aggressively sounds frantic, and auto spot disclaimer-ish.

Or get them to cut copy, but that is too simple a solution to consider, forgive me.

Remote Sessions

You have to direct talent at their home studio. Assume you have some sort of phone patch with them: if worse comes to worse, have them put one earphone of their smart phone in one ear so they can hear you, and you listen to the read from their phone. You will not know what the actual recording sounds like until they send it to you, but you will have a good idea of the read.

A good strategy is to do a run through ahead of time.

They may have a professional phone patch, or Direct Connect, or even ISDN, but don't count on it. Phone patch is cheap, even for a landline unit, so figure that is what will be available, if you are lucky.

ISDN is on it's way out, expensive, and a pain in the ass; Direct Connect is primarily for Protools users, sometimes glitchy, not cheap, but phone patch works in most cases.

And you might not even have that.   If they put their smart phone on the music stand, behind the mic, you can hear the read, not optimal, but you can hear pacing, inflection, etc. After a few takes, they pick up the phone and hear your comments.

As I said, not optimal, but better than having them send you a bunch of takes where they mispronounce the client's name.

SESSION PREP/ INSTRUCTIONS

The first thing is to tell the talent you want them to send you unprocessed .wav or .aiff files. since you have no idea how accurate their monitors are.

I have received files that were so over compressed that I could not fix them, and called the talent to send me clean files. No EQ either.

Have them send via dropbox, or some other large file transfer protocol. Don't do mp3, it is just too lossy a format. (if you do, master at 320kbps, which is the highest quality mp3....but.....I would still go with .wav/.aiff.)

If you already did a run through, you know that they were recording the files too low, you had them bring up the recording volume, and then you noticed the hard drive whrrr, since their mic was on the same tabletop as their computer.

Now you have to minimze their studios problems.

Tell them to put the computer on some heating pads, same with the hard drives, and move the mic as far from the computer as possible, with the back of the mic (null point) pointing towards the computer noise source.

They can still see the computer, but the noise floor should be lessened.

Start the session as though this is now a pro studio. Direct away.

NOISE FLOOR

Once you receive the files, you notice there is still a noise floor, a hum, a whrrr, and some unidentifiable noises.

STRATEGIES:

Most DAWs and audio editors come with a noise reduction program, an expander/gate, and EQ filters; some are good, some you will need upgrade to 2nd party plug-ins/processors.

NOISE REDUCTION PROGRAMS

I have used Izotope's RX noise reduction programs; there are 5 or 6 modules in the suite. There is a basic program, usually on sale for $99, which is considerably better than the programs that come with DAWs. They have a full program, around $300, that has pro features, and an $800 advanced that can do everything except make you a sandwich.

I have previewed other programs, and my opinion is that this one is the most bang for the buck. (Which is a poor choice of words when talking about a noise reduction program)

Waves makes a respectable program, Soft Soap, Cedar ($$$$$$) and others, with new ones coming on line all the time, so before you plunk down your money, investigate: most give you a 30 day free trial, and it will take you that long to get the hang of it, unless you watch the video instructions.

GATE/EXPANDERS:

Most programs come with these. A gate is an automatic mute button: when program dips below the threshold you set on the gate, it automatically mutes the channel.

Or think of it as an automatic gain rider, that fades quickly to "off" after the word stops, and back up to volume when the next word starts.

You can set the sensitivity so that it doesn't cut off the ends of words that end in an "F" or cut off the beginning of words that start with a soft vowel or consonant. You can also set the slope, or speed of the gate closing and opening on most plug-ins.

Gates can be very useful if you have multiple mics and there is bleed between the mics. (There is another way to address this as shown in the edit chapter)

But there is a problem if you have a noticeable noise floor. If you listen on headphones, you can hear a "hole" since you are cutting the ambience completely out, which becomes jarring if not downright annoying. Or you hear a grainy sound as the gate is closing. This is because your noise floor is still just too loud.

You may have to reduce the noise prior to using a gate.

LOWERING AMBIENCE/ NOISE FLOOR WITHOUT A NOISE REDUCTION PROGRAM.

This is tedious, but it works. You have to lower the volume of all the spaces in between words by -8 db. using the gain plug-in. I know, I know, like working a punch press, but it works. Listen on headphones. You may find -4db or -7db, or even -11db works best for your noise.

Be sure to highlight at the null points after the word ends and before the next word begins . If you process in the middle of a waveform, you may hear an audible click. "Bad, very bad. " (sorry, can't help myself)

Or, try just cutting the space out (or insert silence) and see if that works.

If you are putting the voice over music, lowering the noise floor to zero may not be necessary, but should be attempted. A clean voice track makes for a clearer sounding spot. You may not be able to hear the ambient sound per se, but it will muddy up the sound a bit.

LOWERING HUM AND RUMBLE WITH EQ

Most DAWs come with a high pass filter. This may be confusing terminology, in that it only lets high frequencies pass. Also known as a low cut filter.

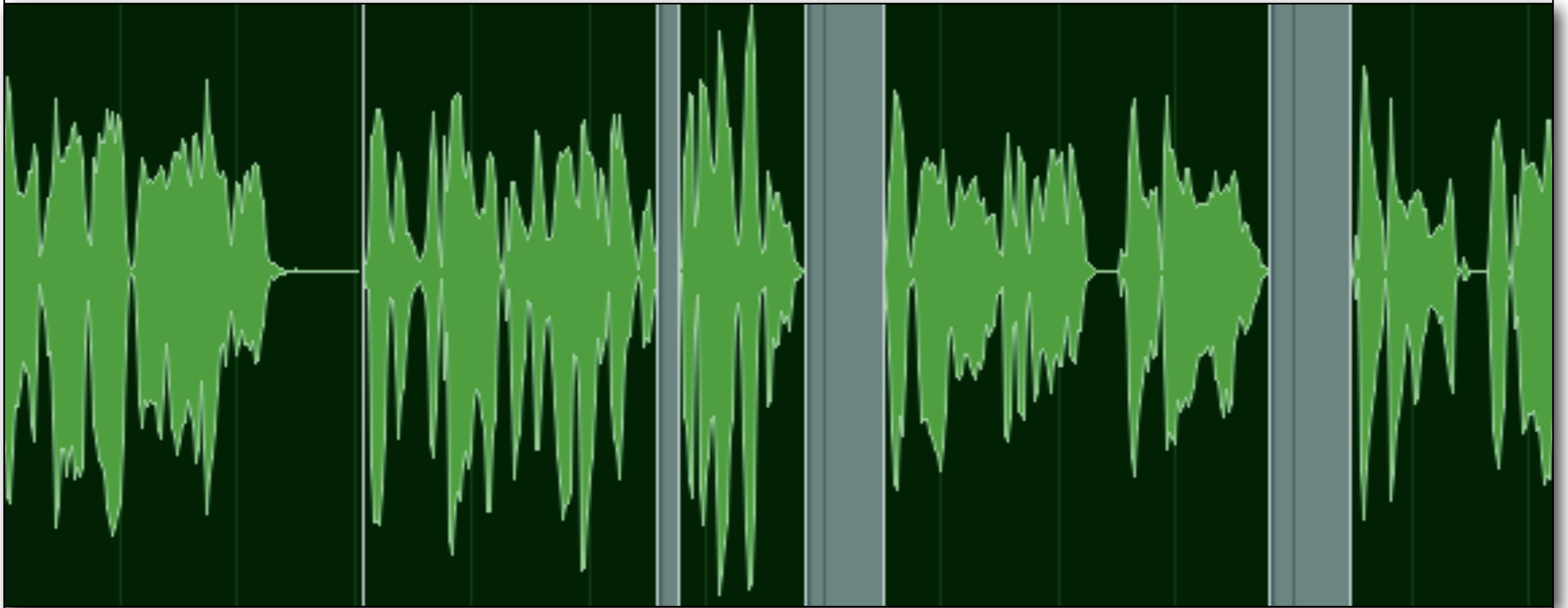
Most hum is in the 60 cycle range, the frequency of AC current, which is where most of the hum comes from. You can set the cut off on your high pass to 90, even 100hz, and it will eliminate a lot of rumble without affecting the human voice, but again, experiment.

There are "de-hum" modules in most noise reduction suites also.

And you can frequently cut the hum from your talent's studio by telling them to turn off the damn florescent light, which is making the hum in the first place. Incandescents, people, or LED.

CHAPTER 4

Editing



This chapter shows editing techniques for fixing commonly found problems in voice recordings, such as how to eliminate mouth clicks/ noises, plosives, editing to time, cutting good takes together, and misc. fixes.

Photo courtesy of Avid

Editing Breaths

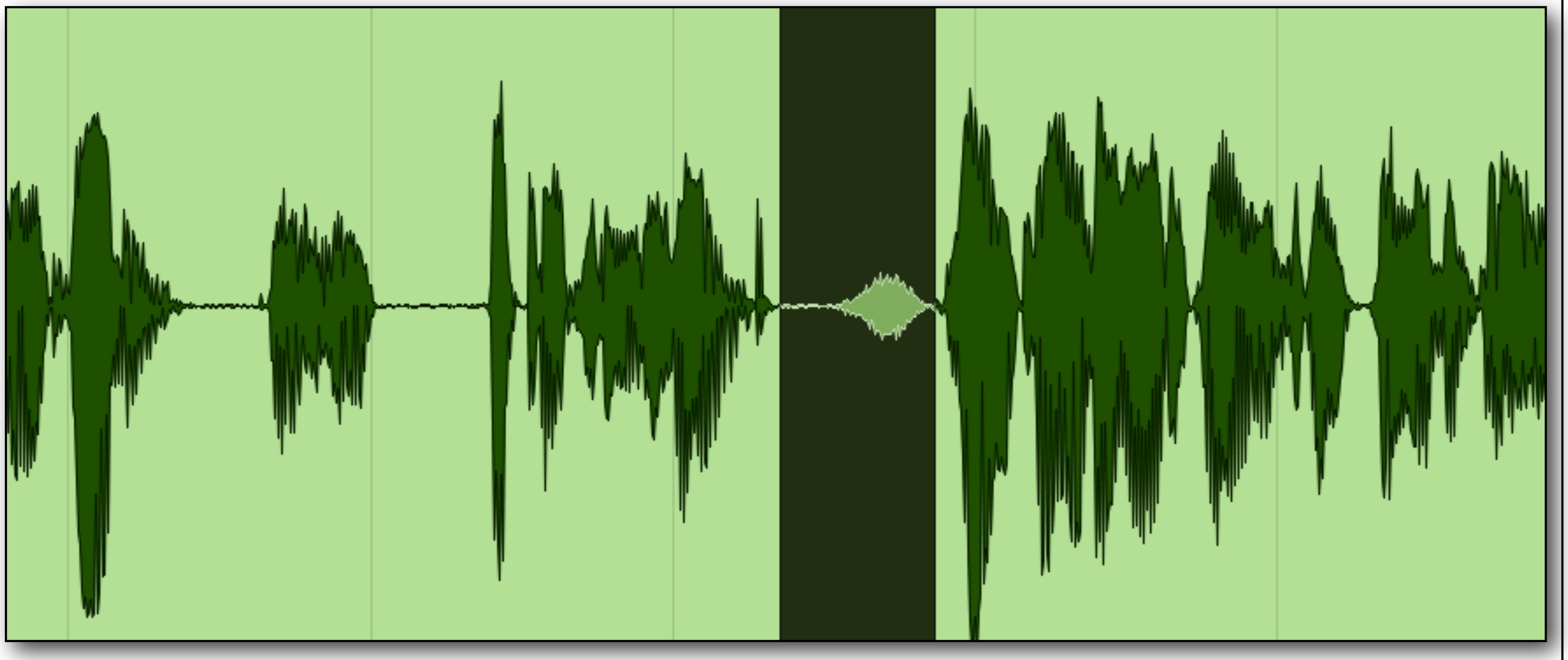
Breaths

Nothing is more annoying than hearing a talent gasping for breath in between words or sentences.

Jokes are still made about the Brenda Vaccaro commercial from the 80's where her gasping for breath was all you remembered.

Sometimes, signal processing, such as compressors/expanders set too aggressively, may push up the quiet sections, and exaggerate breaths and mouth noises. Television and radio broadcast chain compression/limiting will also exaggerate breaths.

As your ears become more attuned to finding and eliminating breath sounds, you will become increasingly enraged at the number of national commercials and programs that release this bush-league slop.



REMOVING BREATHS

Not only is it easy to remove breaths, but you can also tighten up the sentence in the process, making it sound like it had been read in one breath.

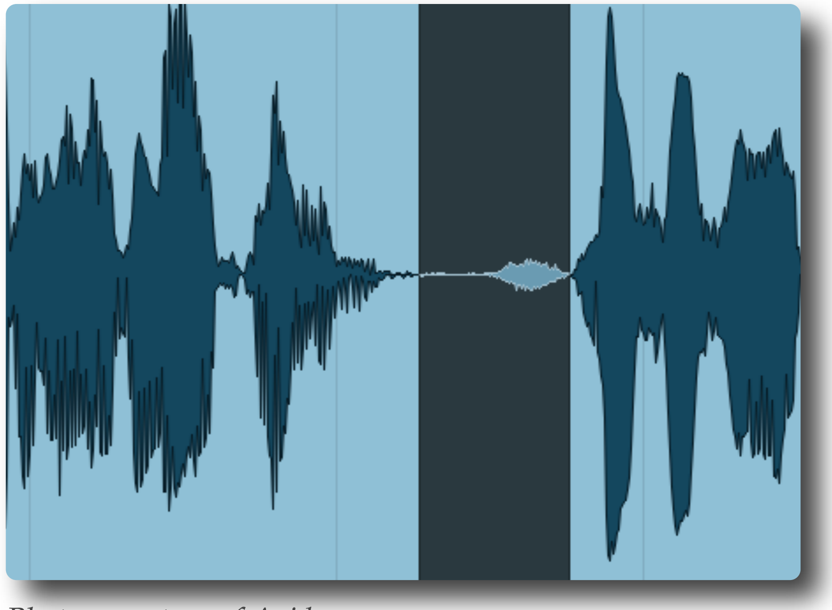
But be careful not to over do it, the point is to make the voice track sound natural.

A breath is an easy thing to spot on a wave form, they generally look like a football.

Photo courtesy of Avid

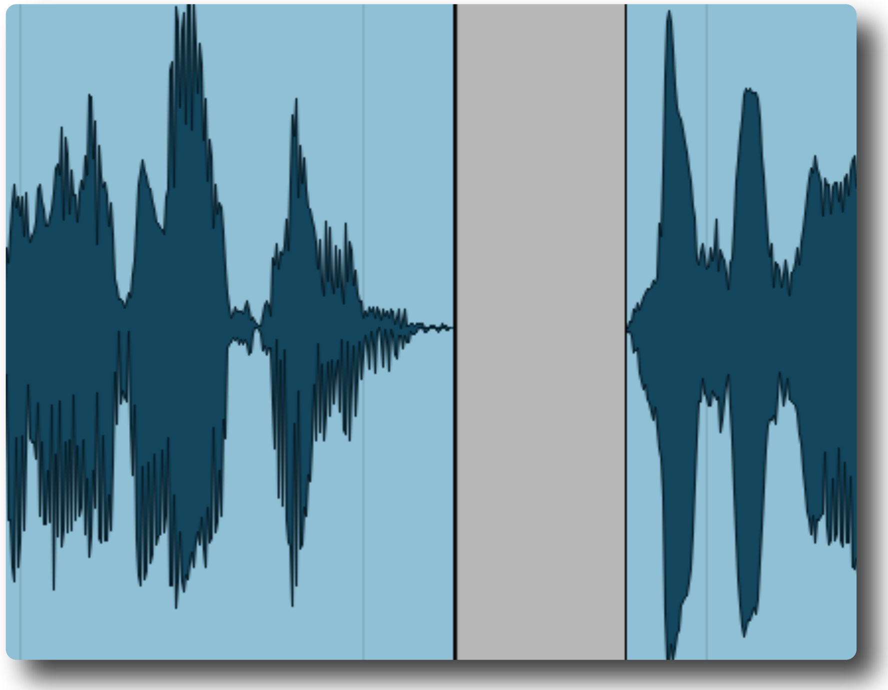
There was a very lengthy thread on a Linkedin discussion group regarding just this subject: to remove breaths, or to leave them in. The opinions were varied, but generally, broadcast and commercials: take them out since the broadcast chains tend to accentuate and exaggerate them; audio books, narration programs, videos, web, long form, etc: leave them in, or lessen them if desired.

CUT/SLIDING

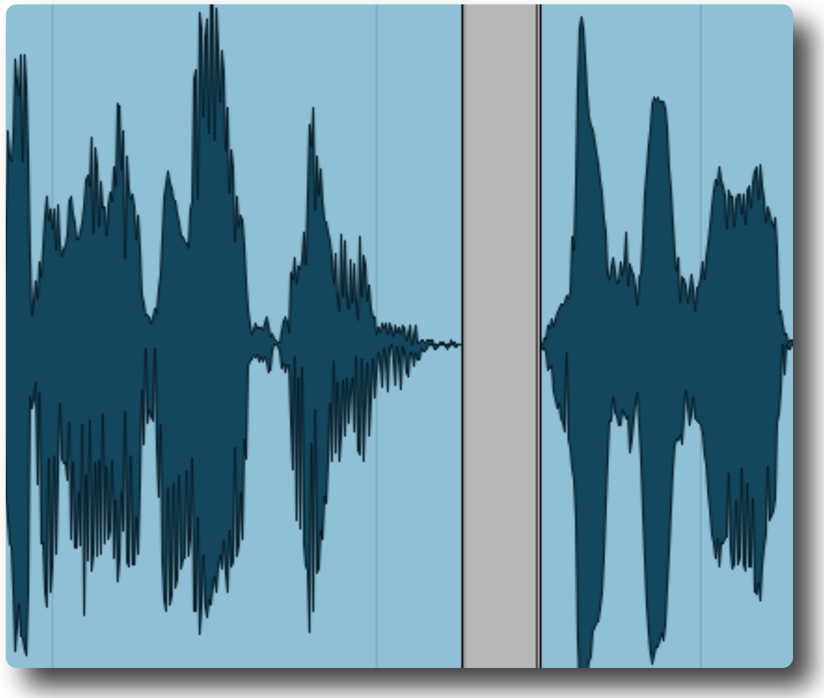


Photos courtesy of Avid

Highlight from the end of the word to the beginning of the next word, cutting on the null point. (where the waveform crosses the line)



Cut the highlighted area out, listen to make sure there is no audible click at the end or beginning of the area. If you hear a click, zoom in and cut on the null point.



Remove 50% of the space between words (approximate). This will give the impression there was no breath taken at that point.

If you have a quiet enough booth and no room sound, you can simply cut the breath out.

Cut right after the waveform before the breath, and right before the waveform after the breath.

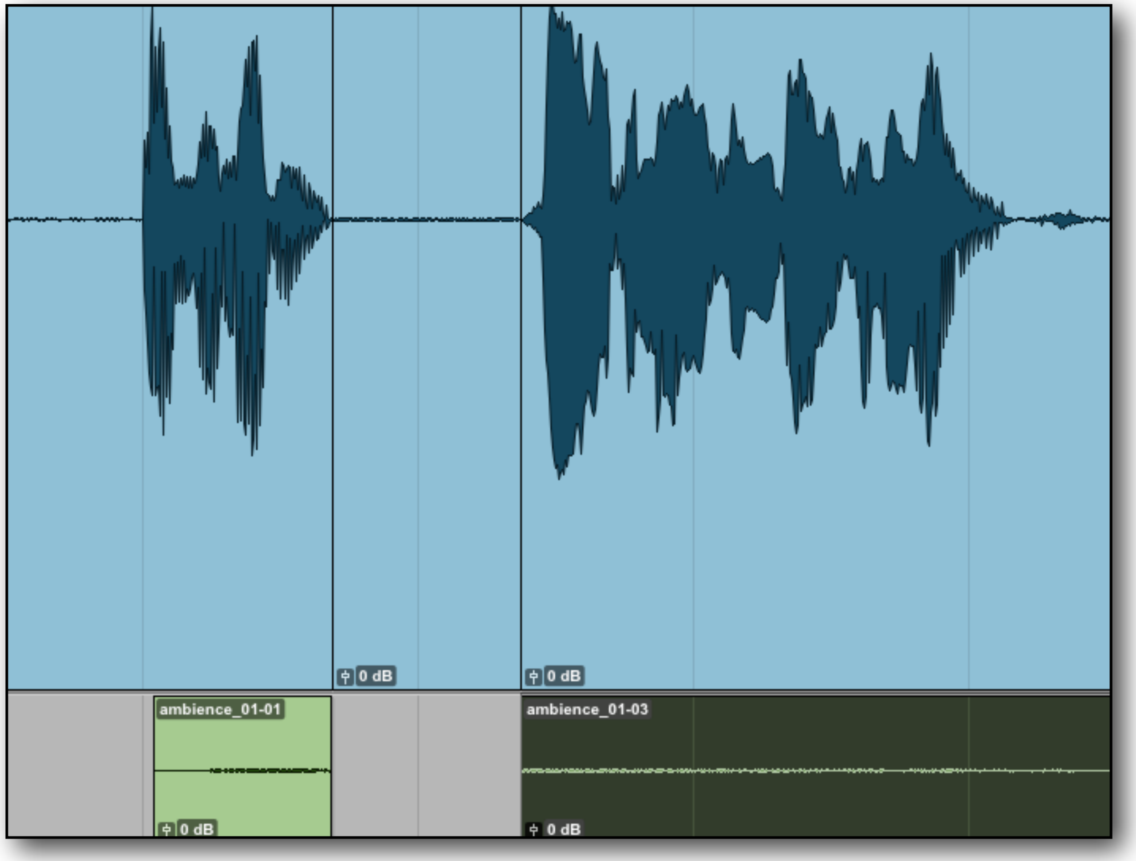
Be sure to cut on the null point of the wave form (where it crosses the line) to avoid a click or obvious cut out.

Then eliminate 50% of that space (cutting/shuffle edit/destructive edit, etc or sliding, depending on your program) and it will sound as though no breath was even taken at that point. (50% is just an average, with practice you can eliminate more or less to make it sound completely natural)

AMBIENCE PASTE/SLIDING

If you have a bit of room sound: copy and paste a section of ambience recorded in the clear and put it on a second track. Repeat the steps above, but cut or copy/paste a section of ambience in the sections you cut out.

AMBIENCE PASTE/SLIDING



You can either do this before eliminating the 50% or after. By having the ambience on a track directly below your edit track, you can easily judge how big a section of ambience you need to fill the void. This is time consuming, but sounds better than sudden gaps of silence.

Photo courtesy of Avid

GAIN REDUCTION: PROCESSED

If you have a strong room sound where even pasting ambience sounds unnatural, (such as an echo that suddenly disappears) you can highlight the area, and lower the breath using gain reduction. Experiment with various settings. This will not eliminate the breaths, but it will make them less apparent, and some feel, more natural a read.

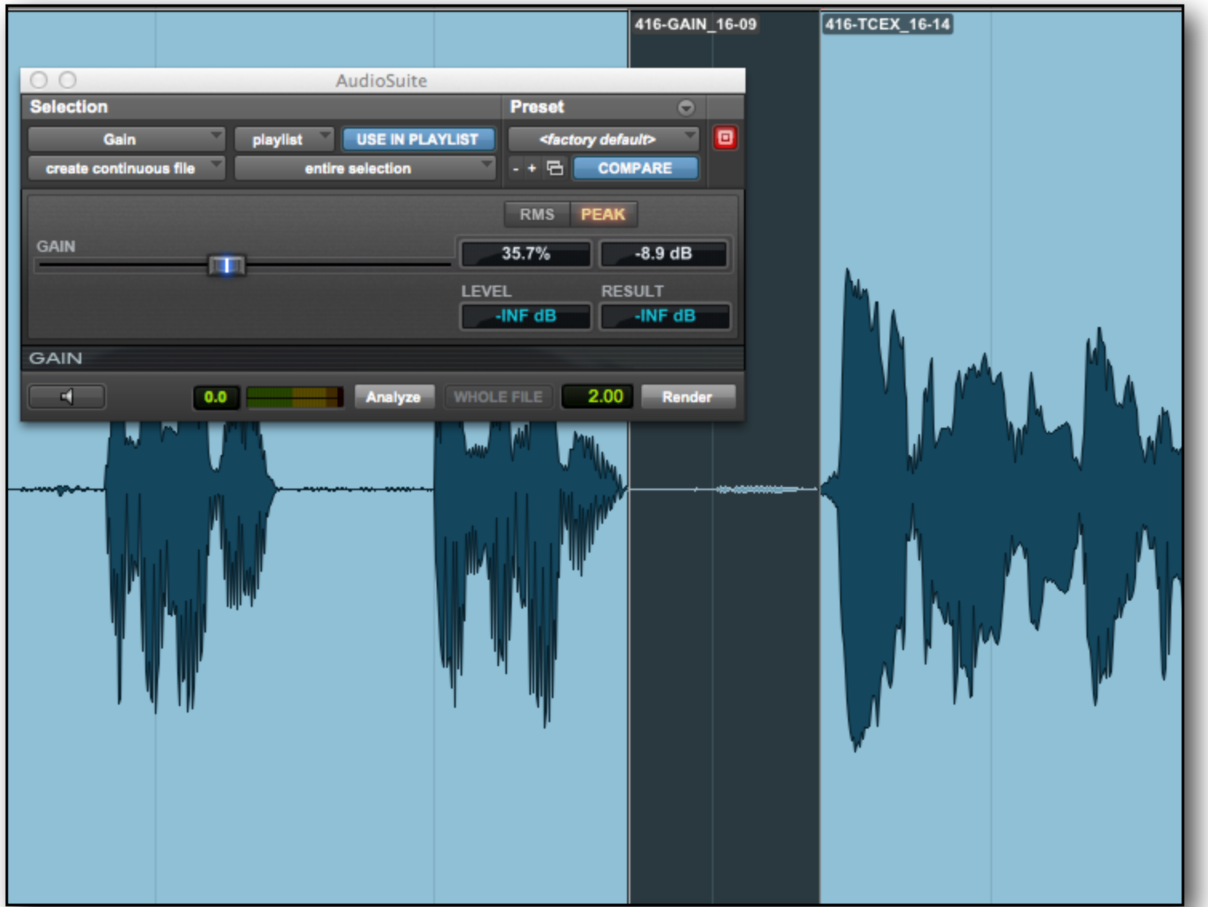


Photo courtesy of Avid

GAIN REDUCTION: VOLUME AUTOMATION

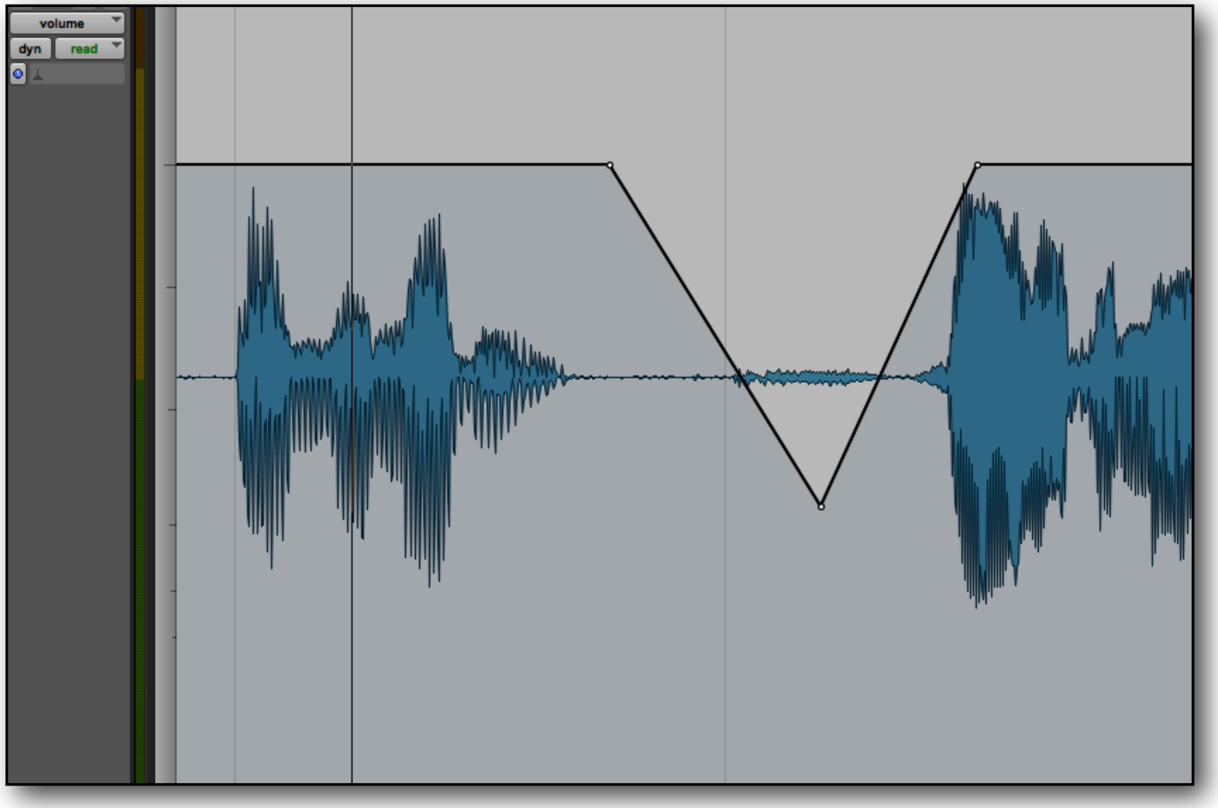


Photo courtesy of Avid

You can also "draw in" gain reduction, lowering the volume of breaths using the channel automation to lower the gain on the console. This may be time consuming, but can help solve the problem caused by strong room sound. You draw a "V" with the lowest point being at the apex of the breath.

You will still hear a breath, but not nearly as pronounced, and it will not affect the ambience as much as a straight processed gain reduction.

DE-BREATH PLUG INS

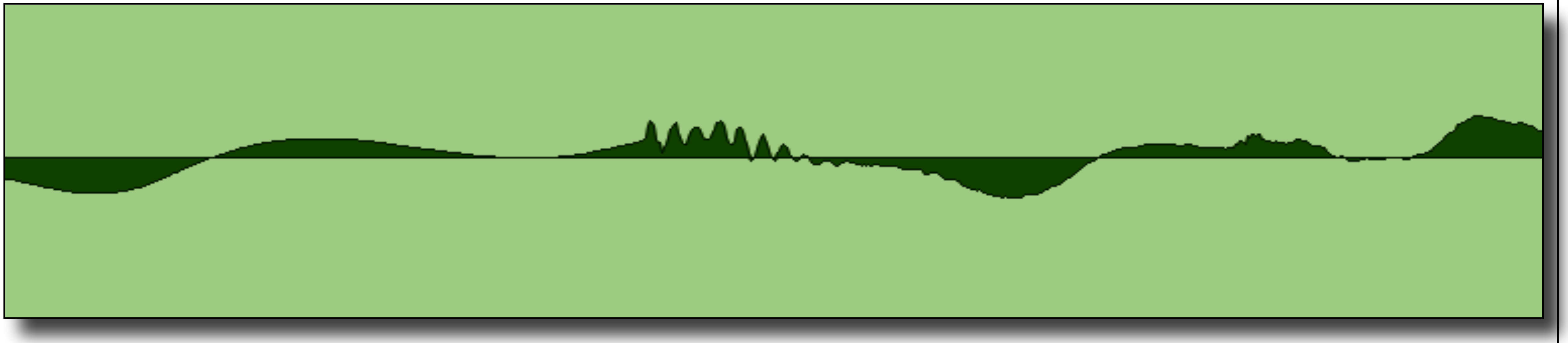
There are also plug-ins/programs that remove or lessen breaths. Most automated plug-ins can catch most of the breaths and minimize them, but you may have to manually delete one or two that get by. The best way to set up a plug in like this is to start with the average breath sounds and eliminate those, increasing the settings on the more obvious breaths, until it sounds unnatural, and then back off a bit. Some of these come with an ambience replacement feature. As with all plug-ins, you have to play with them until you find a happy medium between aggressive processing and transparent processing.

NOISE GATES

Most DAW programs come with a noise gate plug in. It is essentially a ducker, in that it automatically lowers the volume when it senses the voiceover/program/waveform has stopped, You can set the depth, the threshold, the attack, and release, which can be adjusted to sound natural. Always use headphones to check the fine settings. Use the average breaths as your setting point; you may have to manually remove the more obvious, or just lower them using gain reduction so they are in the same range as the ones handled by the noise-gate. Noise gates and de-breath plug-ins are invaluable when doing long form programs, such as audio books.

NOTE: If you set a gate too aggressively, it will clip off the ends of words that end with soft sounds, and will clip the softer sounds (like "H") at the beginning of words. If there is ambience, it will abruptly go to silence after pauses, which can be jarring for the listener. Monitor on headphones to get a good setting

Editing mouth noises, clicks



MOUTH NOISES/CLICKS

Every talent will produce mouth noises. It may be a click in between words, a tongue smack, a loud inhale, etc. which are easy to remove when they are in the clear using any of the techniques listed previously for breath removal. However, when they are in the middle of a word, it gets trickier.

Photo courtesy of Avid

REPLACEMENT

Find the same word, either on another take, or elsewhere in the program, copy it and paste it over the word with the flaw. If you are lucky, it was read with the same inflection and you are done. You are the one-click hero. The client was unaware of the flaw, so don't get too excited. This happens one in maybe 5 times.

If not, zoom in on the waveform and find the part of the word that is flawed then copy and paste just that part from a good take. Always highlight starting points and ending points at the null point, or where the waveform crosses the line. (this will produce a clean edit)

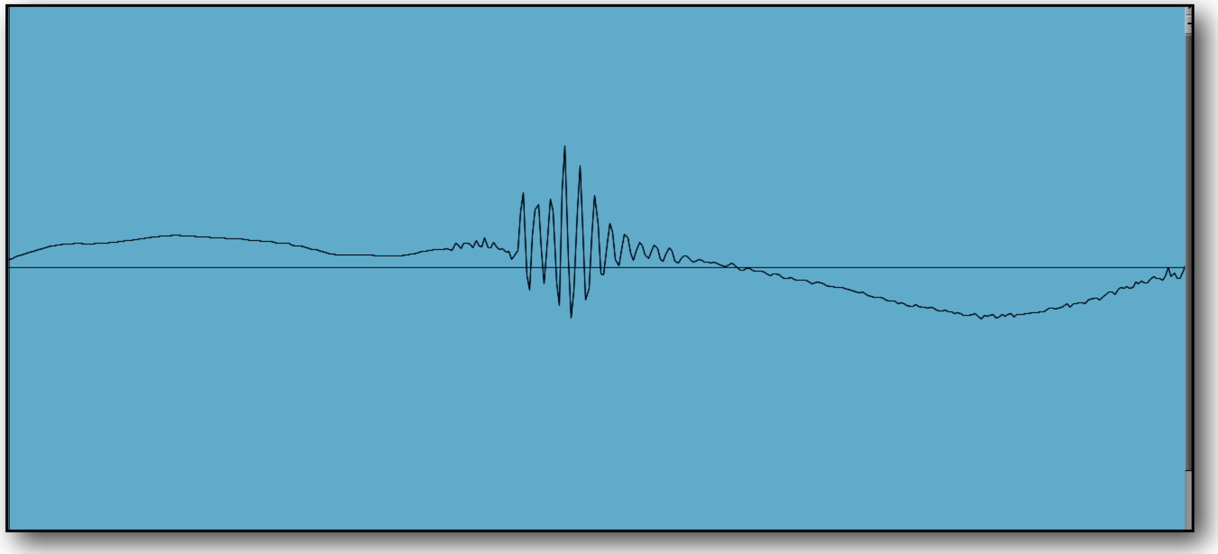
There are a few very expensive programs that will find and replace a word or section automatically. In general, it would be cheaper to hire a full time intern to do it, plus he or she would also get coffee and pizza.

ELIMINATION

DRAWING IT OUT

If you have a pencil tool and you can zero in on a mouth click at the waveform level, it will look like a saw blade in the middle of a smooth curve. Take the pencil tool and draw a line following how you think the curve should be.

Photo courtesy of Avid

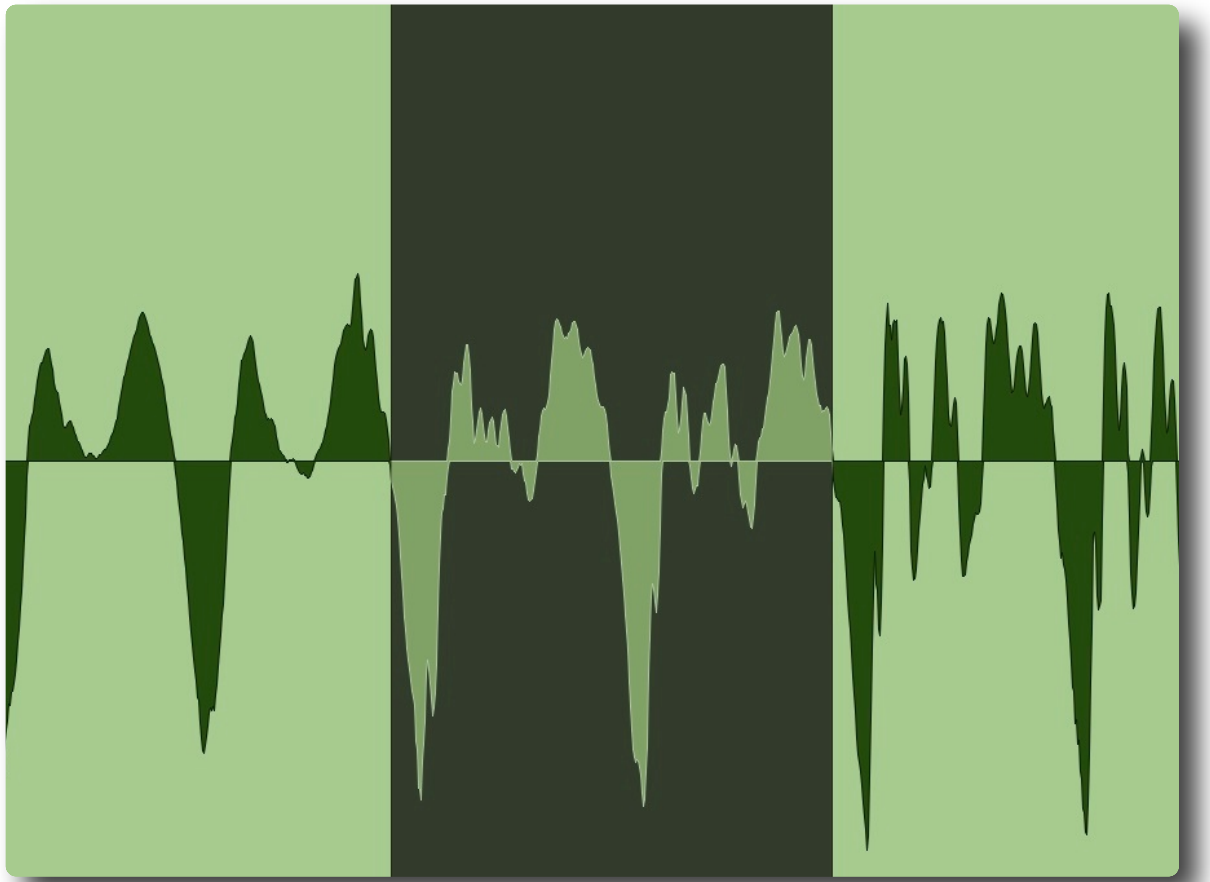


Start before the flaw and follow through after. If you play it back an it is gone, you are permitted, and even encouraged, to spin 360° on your chair, or raise both hands and give a victory whoop.

SPECTRAL

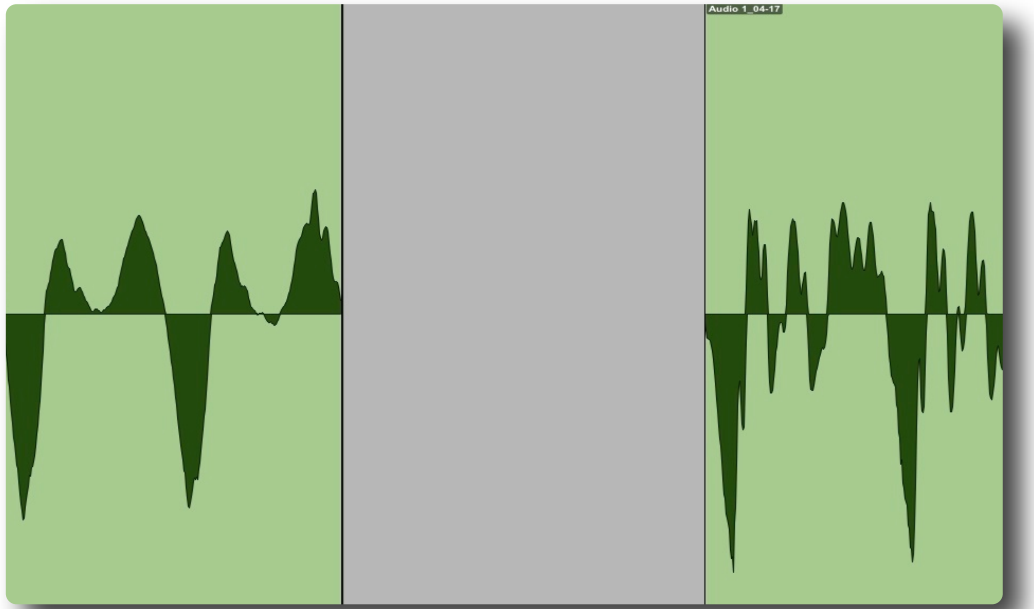
Some programs have a spectral analysis display where you can see areas that have broadband noise in red, and program material in yellow and white, and will also show areas that contain flaws like clicks and plosives. The manual will show you how to identify problem areas, and how to eliminate them. Many talent swear by this interface, but it is only available on a few programs.

CUTTING

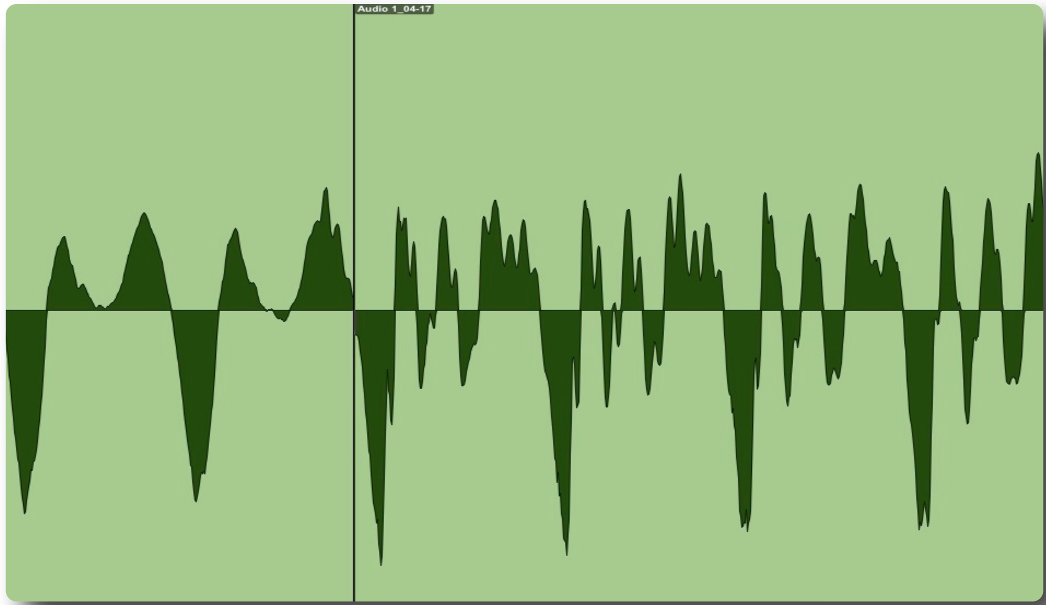


Determine the area that is flawed. Highlight at the null point of a downward waveform before the flawed section, and at the null point of a downward waveform after the flaw.

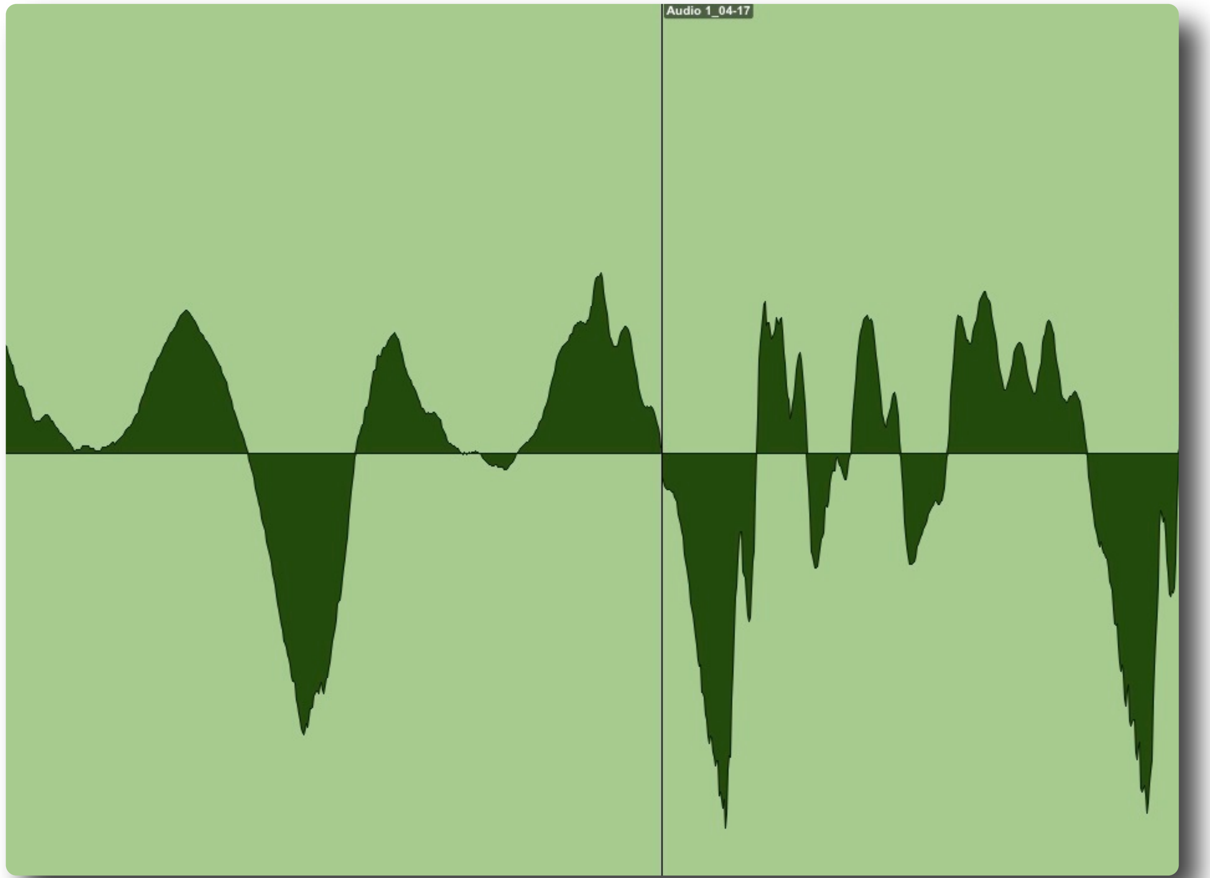
Photos courtesy of Avid



This is a cut using a non destructive edit for example. In some cases, if the slice is small enough, you may not need to shuffle edit to create a continuous file, the space will be virtually inaudible.



But not so with this edit.   The waveform needed to be combined using destructive/shuffle edit. Note how the pattern appears consistent at the edit point.



A closer detail of how the waveform crosses at the null point. Cutting on a null will usually eliminate any clicking associated with a jump in the waveform.   If you are a real hot shot, you can cut anywhere on the waveform as long as you match cut the sections, or redraw the waveform at the edit point. Show off.

With all programs, this is the most basic tool to eliminate flaws. By successively zooming in on the waveform as you play it, you can identify the problem area. At times there is no clear cut part of the wave form that jumps out as an obvious flaw. In cases like these, or with programs that have limited editing capabilities, cutting out a slice is your only option. This may be a trial and error situation.

Scrub the audio, or play up to a certain point. If you do not hear the noise, advance the highlighted area until you hear the noise at the end. Highlight after the noise and back the highlighted area until you hear the noise at the start. Zoom in and see if there is any obvious wave form anomaly that you can fix.

You will see a repeating pattern of wave forms that may look like stalagmites (pointing up) and stalactites (pointing down) (lame cave analogy, but now, once and for all, you know the difference.....you're welcome.)

Mark a similar "stalagmite" to the right of the noise, that matches a similar one to the left of the noise.

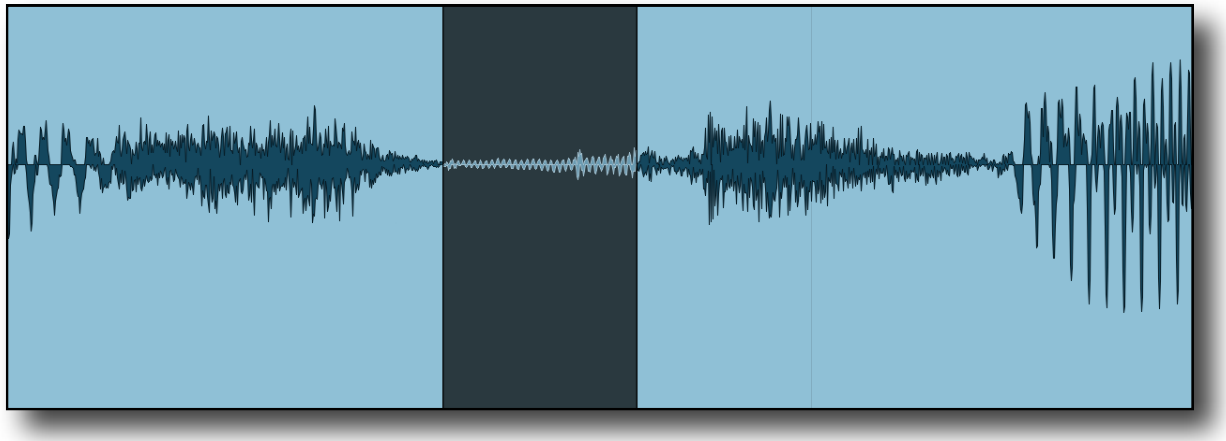
Cutting on the null point at the beginning of the lower waveform on the left and the beginning null point on the lower waveform on the right, cut out the space in between using a destructive edit/shuffle/etc. so that the upper waveform on the left now flows into the lower waveform from the right joined at the null point.

This should eliminate the noise. If not, undo and try again in a slightly different area. If you are really having problems isolating it, take out a much larger section and zero in from there. Listen to the word and if it sounds fine: done.

In some cases you can just cut a slice out without sliding the audio and it is imperceptible. Always check on headphones to play it safe.

Photo courtesy of Avid

NOSE NOISES, WHEEZES, STOMACH GROWLS



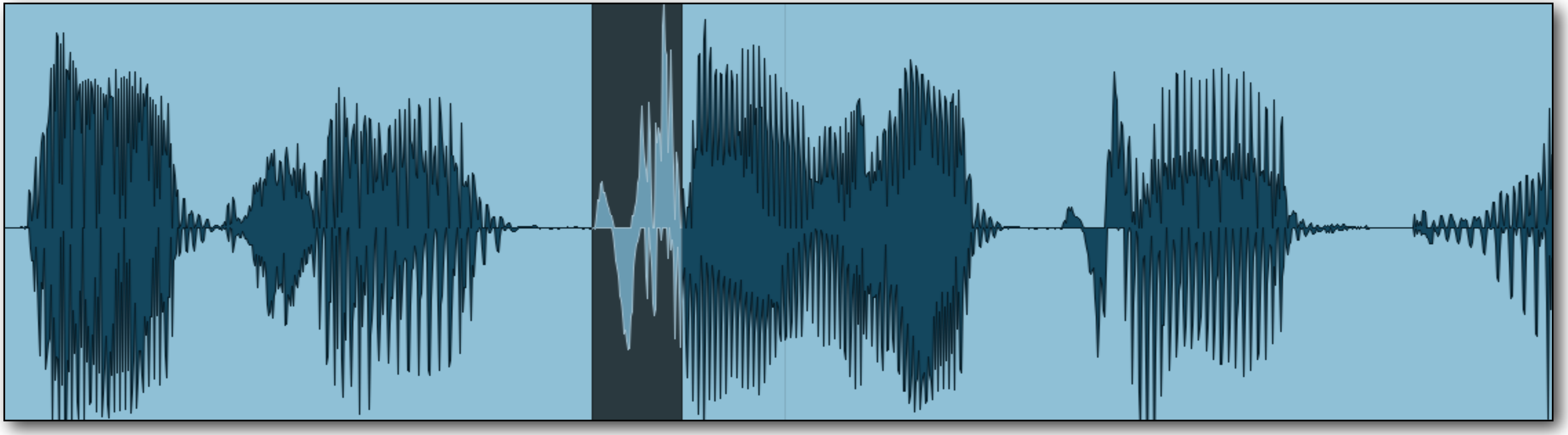
Sometimes you hear an odd sound that can only be chalked up to nose noise. It may be a tone, a wheeze, or snore-like, just something that makes you wince. Or perhaps a stomach growl. You always hear it after the talent has left. Usually it appears as a small wave form between words that looks like a bridge, or sort of breath-like.

Just cut it out. Literally. Do not use a destructive/shuffle/slide edit, just cut the space out and leave it blank. If you do hear a bit of a click, make sure you are cutting on the null point. That should do it. (You may have to add a bit of ambience as outlined in the EDITING BREATHS)

CLOTHING NOISES

Same thing here. A clothing noise generally is covered up by the voice, which is much louder. You hear it in between words. Cut out the space between the words (leaving them blank) and it will be gone, or lowered dramatically.   And have them lose the corduroy jacket before the next take.

Editing Plosives, Sibilance



PLOSIVE/POPS

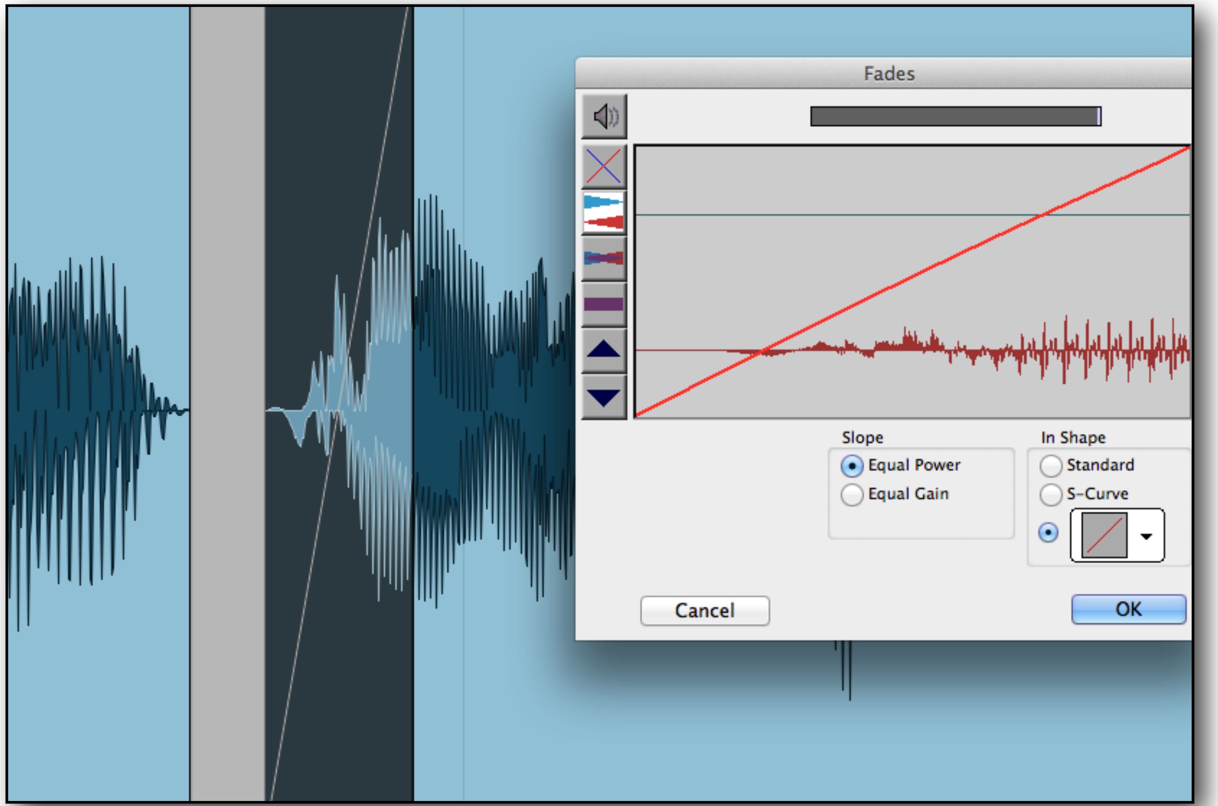
A "P" pop appears in the wave form as an "elephant trunk" or a more solid waveform at the beginning of the word. It can be replaced with another take by cutting and pasting just the beginning of the P, or it can be lessened by lowering just the plosive (highlighted above) by processing the gain by -8db or more.

Photo courtesy of Avid

You need to try different settings until it is diminished to the point of sounding like there was no pop in the first place.

You can also draw in a volume change starting 10db low before the pop going up to normal volume after, or create a fade up over the pop. (example a)

The same holds for other plosives such as “B’s” (as in “Buh” words) “D” “J” and “T” and there are others, depending on the way the talent says them.



Example a: fade-in lessening the plosive from the previous page

Photo courtesy of Avid

SIBILANCE

Sibilance is the spitty, high frequency sound that is often confused with distortion.

Some talent are just sibilant, and there are strategies to fix that.

Compressors will increase sibilance, since an “s” is a low volume sound, the compressor will raise the gain on it automatically.

MICROPHONE CHOICE.

A microphone that is not high frequency sensitive is a good choice for a talent with sibilance problems. This stops it at the recording, and little else need be done.

FIXING SIBILANCE

DE-ESSERS

Most programs come with a de-esser which can be set to lower just the high frequencies that are offensive. You can adjust the frequency and depth to remove enough to make it pleasing. But care must be taken as not to remove too much high frequency or it will sound dull and lose intelligibility.   A de-esser set too aggressively will “pump, and you can hear it working.

PROCESSING

If the program is short, processing “S’s” by lowering the gain by -3 to -5db is may fix the problem. An “S” looks similar to a breath and are dense waveforms. (example b) And you can process just the “S” in the middle of words without it becoming apparent.

CUTTING

Sometimes a talent may draw out an “S” which is one of the easiest sounds to fix. You can do a destructive edit/shuffle/slide edit and cut a section out of the middle, cutting the “S” to any length that sounds natural without hearing an edit.

THE “F”

“F’s” are another less offensive sibilant sibling. They can sound spitty and can usually be tamed with a fade up, by shortening the lead in (either by cutting and doing a fade in, or even with time compression); or simply by lowering the gain by 3db. If there is a plosive “F” in the mic, the same technique for taming “P-pop’s” should be employed.

****SIDEBAR: CUTTING ON AN "S" or an "F"

If you ever have to cut in the middle of a word, an "S"(or "F") is an easy choice as long as the waveforms match in amplitude. If one "S" is louder than the other, you can process either one to match using gain, and you will likely not hear the edit even with headphones.

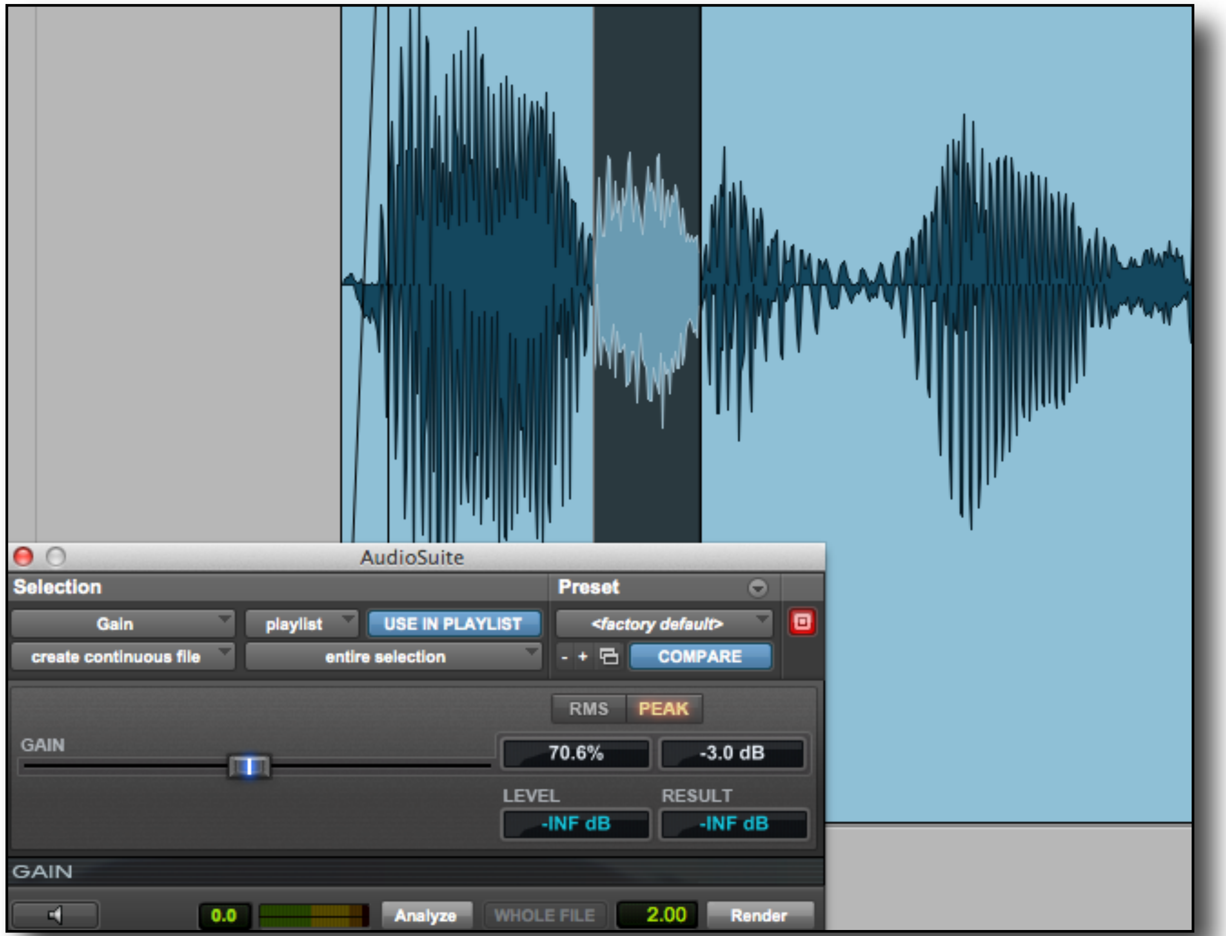


Photo courtesy of Avid

Example b: of a fixed "P' plosive (fade up technique) and an "S" highlighted ready to be processed -3db)

Editing to time

You have just edited a perfect 1:01 radio spot. Your client does not want to use time compression for some reason known only unto himself.

This will take more time, which you should charge him for, so don't say anything other than: "this will take more time"

Photo courtesy of Avid

TIGHTENING A TAKE WITHOUT TIME COMPRESSION

What you will be doing is essentially speeding up the talent's pace by removing spaces between words.

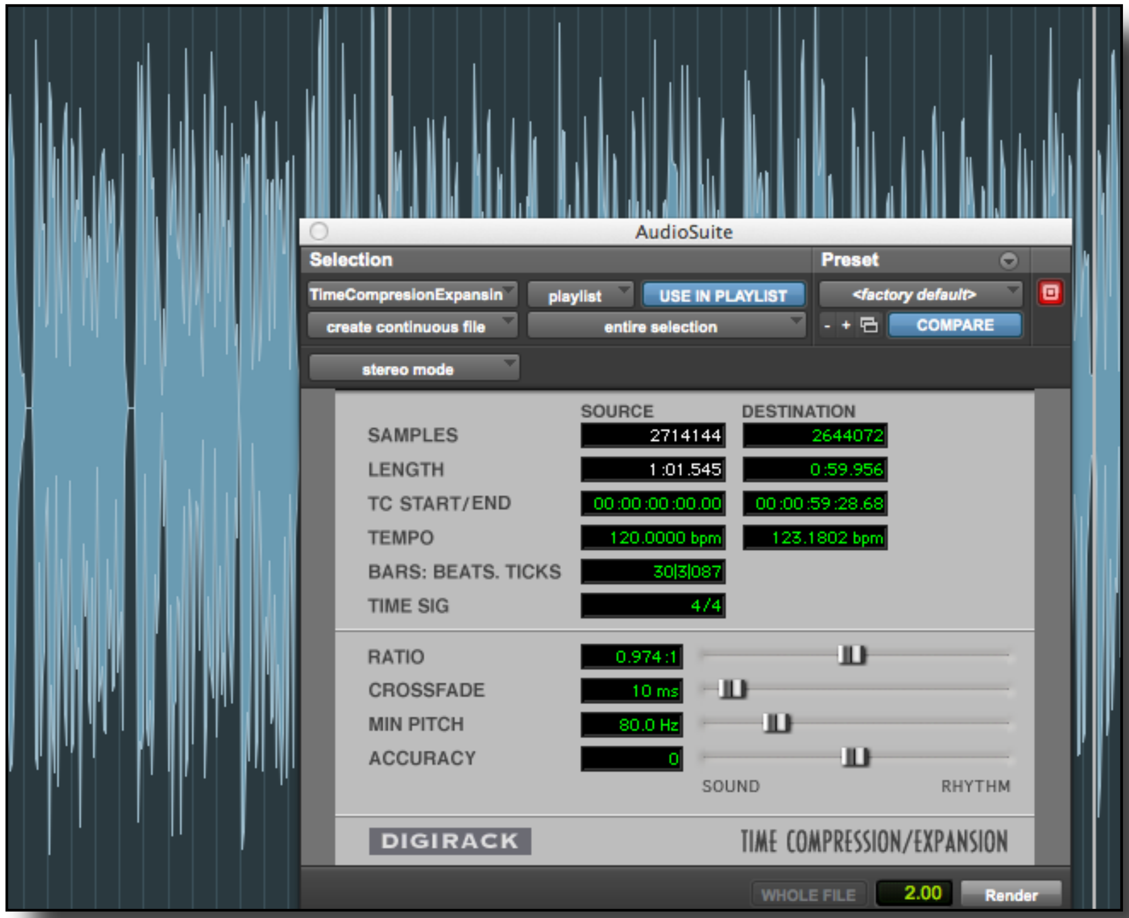
Using destructive edit/shuffle/slide edit, etc, take small pieces out between words, where you took out breaths, between sections that are already blank, etc. Usually a ¼ of the space will be sufficient and will not be noticeable. But listen to a sentence at a time. There may be pieces you have to put back in to make it sound natural.

There may be sentences you can combine, which will pick up even more time. Sentences that start with "And" can usually be combined with the preceding sentence to create one continuous thought; you have eliminated nearly all of that space and picked up a good chunk of time.

TIGHTENING WITH TIME COMPRESSION.

(I have no idea why the client above did not go with this option in the first place.)

Photo courtesy of Avid



Most programs have a time compression, time twist, or whatever they call it, and most work pretty well, to a point.

Most time compression schemes works by taking little slices out of the wave-file and putting little cross-fades in to smooth the transitions. They do not "speed up" like tape, which would raise the pitch also. Some take slices out of the spaces between words, as outlined above, but most just cut like a hard boiled egg-slicer. And you can only go so far before you hear it, or it sounds like the disclaimer on a car commercial. Some work better than others and you can be a bit more aggressive before you start hearing words being clipped off.

By experimenting, you will be able to find how far you can push your time compression before it sounds awful. 3 seconds on a :60 is pushing it with one program, 5 seconds is pushing it on another. Both can make a car spot disclaimer unintelligibly fast.

But in the case of the 1:01 mentioned above, you can time compress that and it will sound fine. You will be the only person who knows. But if you have a 1:04 that just sounds too rushed when you compress it to a :60, you may have to shorten it to a 1:03 using the above technique first. If that does not work, you may have to cut out a few precious words.

If you know all the takes are long when recording and the client does not want to remove any of his darling copy, and the best the talent can do is a 1:03, have the talent ***read in a more relaxed tone***, more laid back, perhaps even a slightly lower pitch. Number one, you can actually read faster in a relaxed tone (so you may actually get the copy recorded to time), but more importantly, (and number two) you can compress a relaxed read more aggressively than a high energy read.

When you do an excited read and you compress aggressively, it sounds frantic. That is why most legal on car ads are read in a monotone and then compressed to absurd speeds. If the talent read them with the same inflection as the body copy, after compression it would sound like hail hitting a dumpster.

Also, depending on the music select, you can get away with aggressive compression if you are mixing it with an aggressive music track. Yes. U.S.30 Raceway is gone, but not forgotten.

CHECKERBOARDING

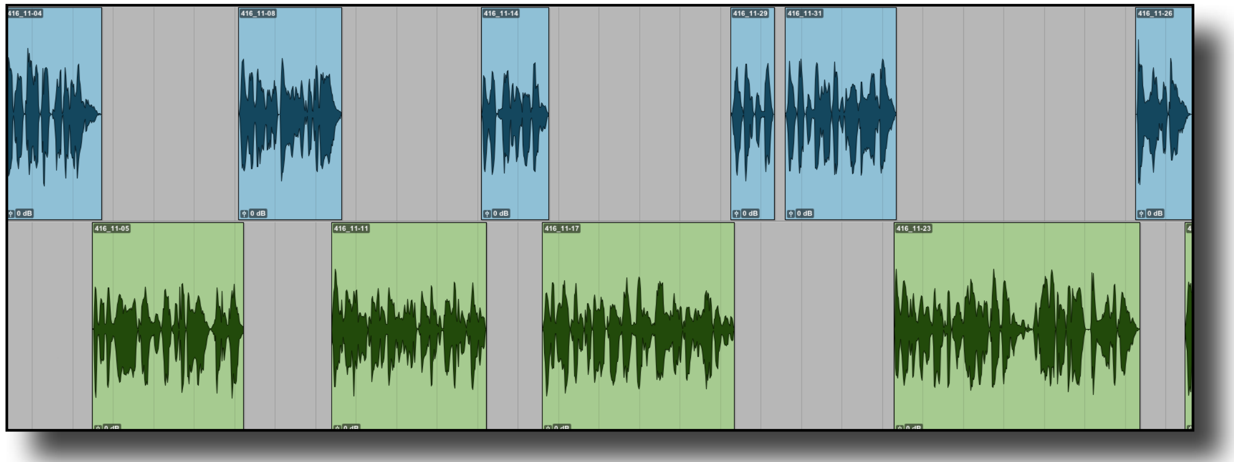


Photo courtesy of Avid

We have all heard this technique in the above mentioned U.S.30 spots: "dueling voices". One voice is stepping on the other voice, and one of the voices has a filter to give it a different sound.

If you are stuck with trying to save a 1:07 , this might do it.

Instead of one comp track, you have two. One is set up with an eq/ filter/effect to give it a different sound than the "normal" voice. (Usually the filtered voice is #2, sort of an "answer" of the main read.)

In track 1 put the first piece of copy. In track 2 copy/paste or drag the next section of copy and slide it so it is just stepping on the audio in track 1, and repeat this process.

It will look like a checkerboard, hence the clever name.

If you can, double up on the same word or phrase at some point for emphasis, ("SUNDAY!!!!") .....which is always funny for some reason.

Even if it is just one talent reading the spot, this technique works. For the double up, take the same word from a different take so it sounds like a second person.

You will find natural break points in the copy where you can take one sentence and make it the filtered/answer voice.

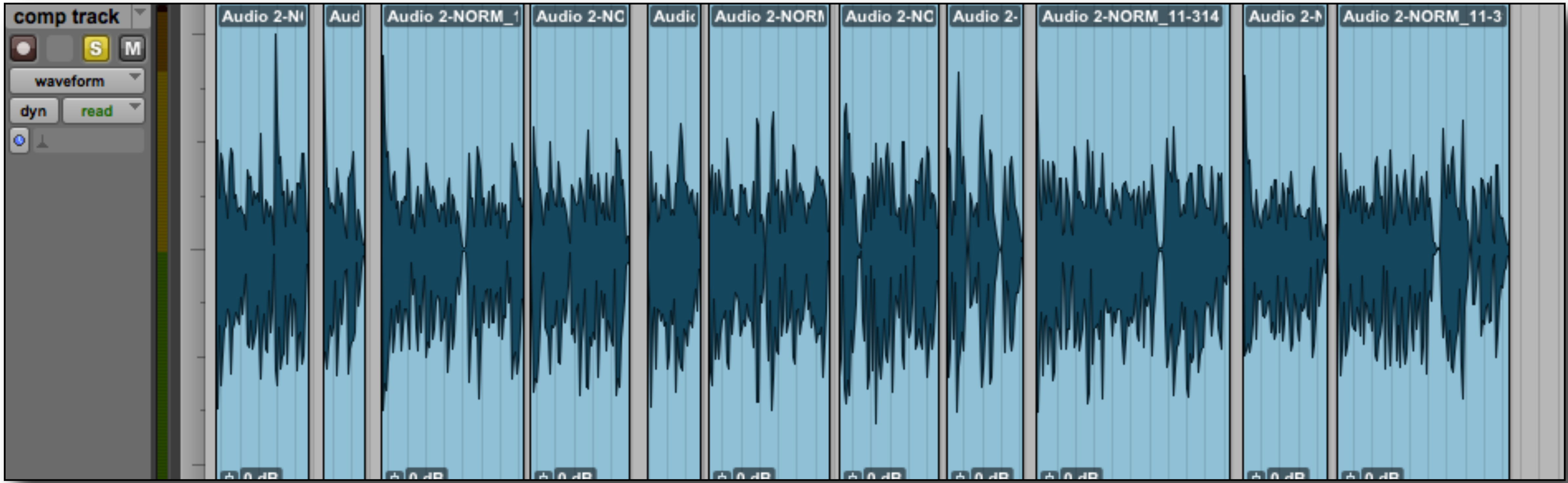
You may be able to cut :03/:04 out using this technique; ***then highlight BOTH tracks*** and time compress them together (so they stay in sync) to get them down to :60.

But you have to find the most obnoxious piece of music in your library to compliment this fine technique.

And believe it or not, they will all work together. And it will sound even better with a few explosions and gated reverb on the voices.

You can’t beat that.

Cutting good takes together



Photos/screen shots courtesy of AVID

CUTTING GOOD TAKES TOGETHER

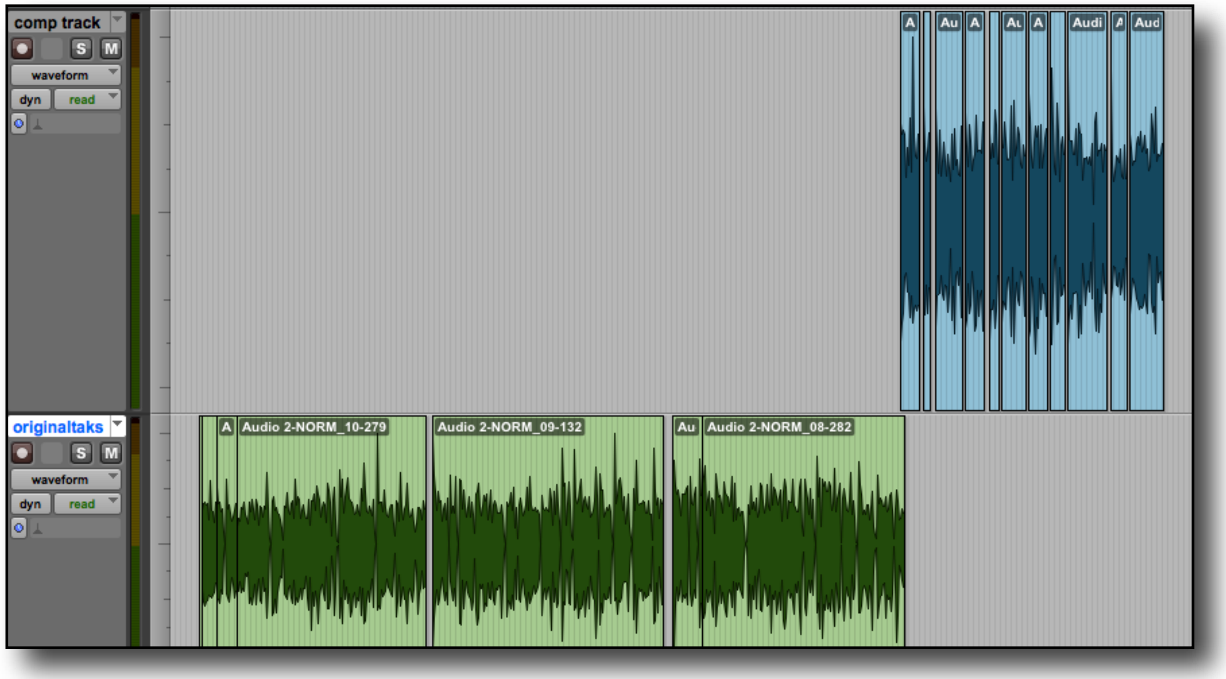
What you want to accomplish through editing is to create a single take that is a better read than the voice talent is capable of doing. (Although, sometimes the talent makes it very easy for you)

Rule of thumb is record the run through, ….you never know when a great take will happen. If they don't think it is being recorded, they may be more relaxed and give a take that is on the money.

Once the talent is familiar with the read and where they are going with it, record several takes (marking each one either with an audible slate, or by putting a track marker on the edit window) put marks/ make note of which parts of each take you like. When you hear (or do) a great take and everyone says: "that's it", always take one more as a "safety". There may be one thing in the safety that outshines the others, once again, the pressure is off.

There are several ways of editing the best takes together. If you are doing a destructive edit, on the same track, first copy and paste the entire recording after itself: work on that part. In case you decide that you need to replace a part after you have edited the entire piece, you don't have to undo all your work, you still have the intact original to borrow pieces from.

MULTI TRACK COMPING



This is a quick solution that is more flexible.

Create a blank "comp track" with the recorded tracks below it. Copy and paste the best parts to the comp track, tighten them up, and you now have a new track with the original recorded track untouched. You can also put several takes of a particular part next to each other and see which one flows the best, and then delete the others.

Photo courtesy of Avid

CREATING A NATURAL SOUNDING TAKE

As you are editing the best sections together to create one take, be aware of spaces between sections, words and phrases. Do they sound like a natural pause, or do they sound rushed? Or............. uncomfortably long? It may only be a small gap, but if it is out of the rhythm the announcer has established, it can be an "ear trip" ….where you are suddenly distracted by it and stop listening to the message. Slide the section until it sounds as though the talent read it perfectly. You may have to do this within sections also, assuming you have removed breaths outlined above.

Most talent have a cadence that is practically a click track during a read, so be aware of their pace. There may be a section where they intentionally speed up, slow down, or leave a dramatic pause; if so, put an appropriate space ahead of, or after that section to make it stand out. I like to call this "leaving just enough space so the listener can see you turn your head." Most of the time the talent can handle this just fine, but if you are cutting together a comp track, consider the "white space".

GUIDE TRACK

If there is a take that everyone liked, but there were sections that were flawed, create a "guide track" above the comp track. Line up the sections of the comp track to the guide track, which should give you a close pacing, and do fine adjustments from there.

MULTIPLE VOICE EDITING

When editing 2 or 3 tracks, create a comp track for each talent.

You will want individual channels for each so you can adjust volume, eq, pan, remove any mouth noises etc., and be able to edit pieces within that channel.

You need to eliminate microphone bleed; if you replace a section in channel 2, it will not match up with the bleed from the original take in channel 1.

You want each section of audio in the clear. Assemble the takes using the original tracks below as a rough guide track. It will look like a checkerboard. You can slide the sections around until it sounds natural.

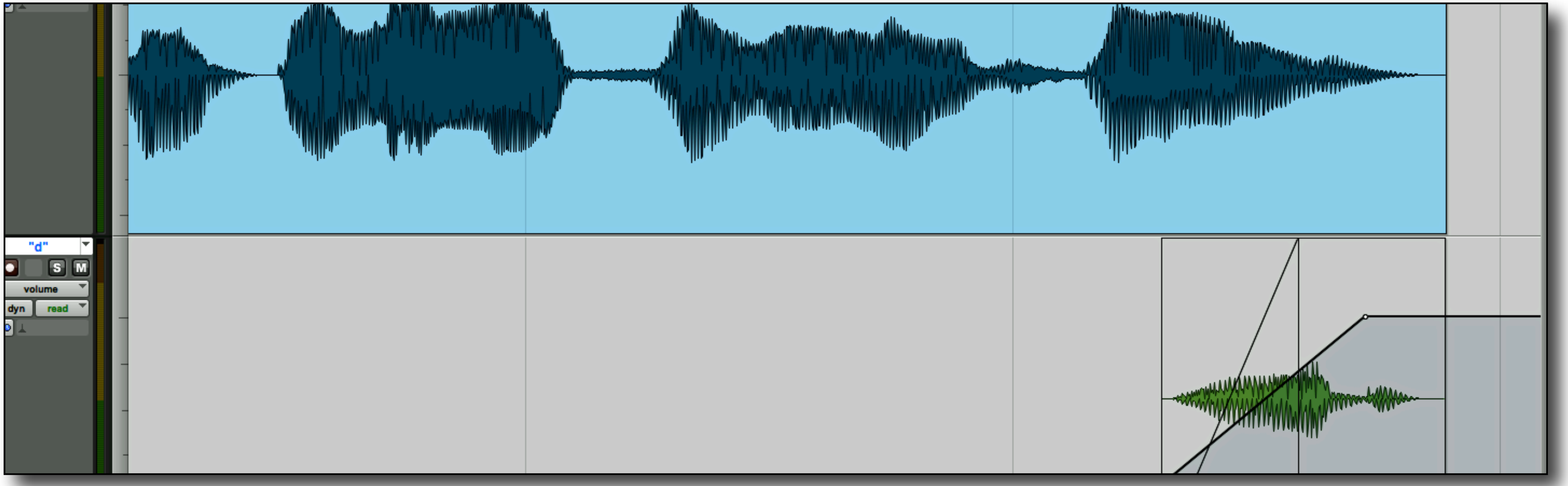
If you are lucky and have a complete take that sounds great, you should still remove the bleed between sections (checkerboard again) which will eliminate room sound and make each voice clearer.

If you like, set the pans for each channel one slightly to the left, one slightly to the right to give the voices a place in the sound field. Do not pan too far left or right: sometimes a station that does not broadcast in stereo will just take one side of the program instead of summing both sides.

You may need to insert an ambience track to make it sound more natural. Always record a minute or so of room sound. You can mix this in as needed, or cut small sections in as outlined in EDITING: cutting out breaths.

Photo courtesy of Avid

Misc. Fixes



Photos/screen shots courtesy of AVID

FIXING MISCELLANEOUS PROBLEMS:

This is an example of coming up with a solution and trying anything: a jingle singer who did not sing the "D" in Ford (the track was sent to us from another studio and the talent had to return to the half-way house) We tried to increase the gain on the "d" but it was barely audible. We looked in other parts to see if he had sung a 'd" that we could copy and paste. There was one, but it was in another key. We changed the pitch, but it was too great a change and was horribly obvious. Finally, everyone in the room took turns going into the booth and sang "Ford" along with his track as a guide. We grabbed the "d" and a little bit of the "r" from the engineer who sounded close enough. We put it on another track so we could eq, fade in, and compress it to match. Done.

CONSTRUCTING WORDS

In the case of a long form program, you can usually find enough pieces of other words to construct a word such as "cannot" (obviously can and not) or even more complex words depending on what pieces you have. If you are lucky, they are all read with the same inflection so they can cut together easily. Always cut on a null point, cut on a consonant, or in the middle of an "s" and try adding a crossfade at the edit point to smooth it out. Sometimes you are lucky and it works even without the crossfade.

PLURALS FROM SINGULARS

On occasion, a talent will read copy that is incorrect, or will read a typo that no one catches until, as always, the talent is long gone. In the case of a singular that should be plural, copying and pasting an "s" on is easy.

Find one at the end of a word elsewhere, copy it and paste it. Look at the waveform from the word you borrowed it from, and try to match the transition point on the new word. You may have to slide it slightly over the trailing waveform on the singular and add a crossfade. You may have to adjust the gain of the new "s", also.

SINGULAR FROM PLURALS.

Just cut the "s" which looks like a ball, off the end of the word. You may need to nibble a bit more off the waveform until it sounds right. You can try a short fade out also.

FIXING "IRREGARDLESS"

For some reason, everyone starts a sentence with this non-word, but you can fix it. Cut off the "EAr" so it starts with the "R". It will be clipped even if you cut it on the null (like you always should) because it is starting at volume. Draw in or create a fade. "Goldielocks-it": not to long, not too short…..If it sounds like "regardless" you are done. If the "R" is too long, undo the fade, nibble a bit more off and try again until you get it to sound right.

You can do pretty much the same thing saving "Ex-specially" and "Ex-cetera"

RAISING/LOWERING CONSONANTS

Most audio is mixed with other sounds/music; listened to in a car with road noise, on a kitchen radio, computer speakers, etc, so you want it to be intelligible even in the worst case scenario. Consonants need to be clear for understanding speech, especially with other intrusive sounds. If the audience has to strain to understand a voice, chances are you will be tuned out.

On occasion, talent will trail off at the end of a sentence, making the last consonant low in volume. If the voice is in the clear, the audience will get it, in fact, in may sound more natural. However, if there is a music track behind it, it will need to come up.

In some cases just highlighting it and raising the gain 3db is enough. Or 4. Other times, cutting in the last word or part of the word from another take is necessary.

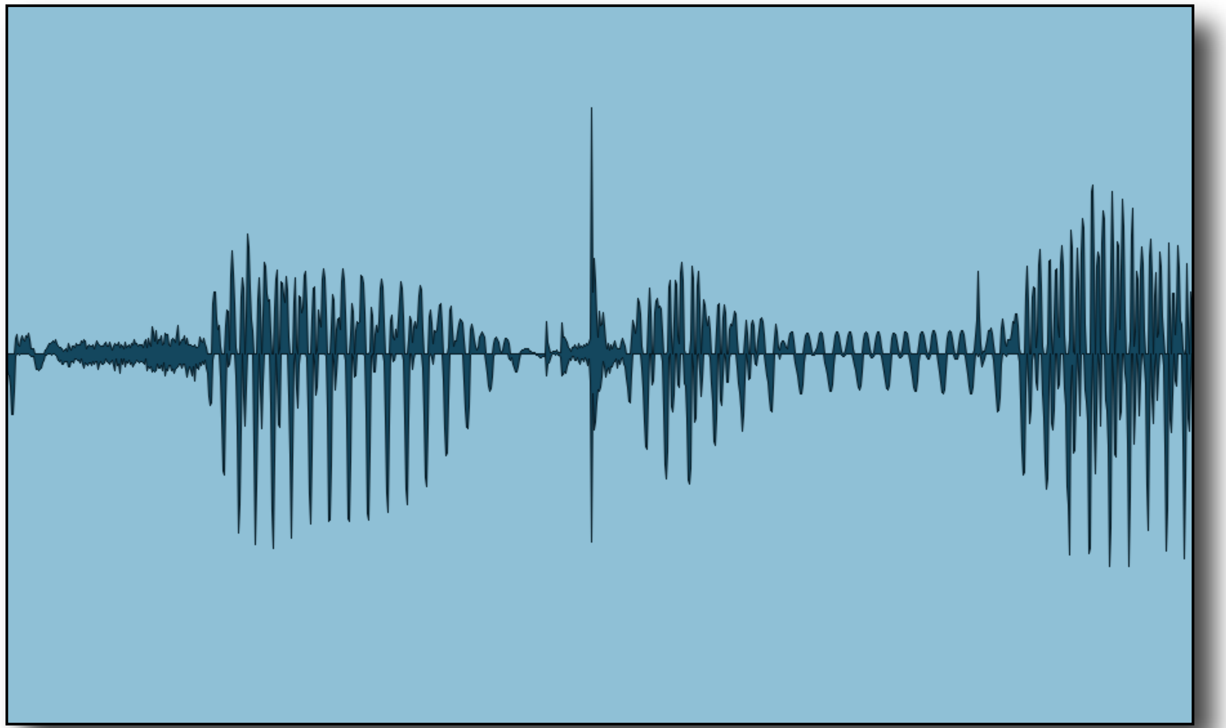


Photo courtesy of Avid

Sometimes a hard consonant like a "C" or "K" is much hotter (louder) than the rest of the program, (especially at the beginning of a word) you can soften it by lowering it 3/4db (or more). Most of the time problem areas are easy to spot in the wave form. (see example above) One part will be much larger (louder) that the rest on average. You can spot check and lower these 3/4db to smooth the program out a bit. This also permits you to raise the overall level of the program since you no longer have these peaks that would cause overshoot.

EXAGGERATED PRONUNCIATION

Every talent has a trip. It might be First Street (say it fast and it sounds like FurStreet) but if pronounced, it might come out FirsT-Ssstreet. You can lower the volume of the T, cut the Sss in half and push them closer together to make it sound more natural.

Or another frequently heard trip is "20 Percen-TOFF": Lower the volume of the "T" 3 to 6db and move it closer to the "n", you may need to cut the "n" in length or time compress it to shorten its duration.

Sometimes a name, such as McCormick Place is read with the emphasis on the last word: McCormick PLACE. You can lessen that exaggeration by pushing the McCormick up 1.8db, and lowering PLACE by 3db. Won't completely fix it, but it will

make it less obvious.
Or “garage door OPENER” same fix minimizes the ear trip.

These are things that should be addressed while recording since they fall under interpretation, and are much easier to fix with direction than in post. More on this in Chapter DIRECTING TALENT

THROWING IN THE TOWEL

Some things are unfixable, but you should try everything. We had a spot that called for a voice talent who sounded like a teen-ager…the talent ended all her sentences with an uptalk: “oKaaayyy??” We thought it was very funny, she sounded *totally* Valley Girl “Chipmunk Chatter”-voice and was the perfect character for the spot. The client loved it. After the spot was mixed, the client’s boss hated that particular speech affectation and wanted it changed. We drew in a pitch change, we processed parts of the word so it would pitch down instead of up, we slowed it down with time expansion. Nothing worked.

Finally we punted and called both clients back giving them a quote for what the revision would cost in terms of bringing back talent and studio time. Suddenly, they loved the original we had presented. 1hr. of waveform “playing around” time: un-billable. So the next time someone asks, “Can you change uptalk?” ……Say “Yes,…… this is what it will cost….oKaaaayyy?”

OUT OF PHASE AUDIO

Photo courtesy of Avid

You get a stereo file from a voice talent that sounds weird. The sound seems to be coming from your left shoulder and not from the speaker. Or when you sum to mono, the signal practically disappears. And when you look at the waveform in your edit window, it looks like a mirror image: on the top panel, the waveforms go up, on the bottom panel they go down at the same point.

Classic out of phase audio.

If you have a phase reversal processor, how do you reverse just one side in a stereo file?

Simple:

Copy and paste the stereo track to two mono tracks. Process one with the phase reversal plug/tool. This works for music and sound effect tracks that are out of phase, too.

Or in the case of voice, just mute or erase one track and use just one mono file. (Pan centered)]

Editing final thoughts:

Editing is like practicing piano: the more you do it, the better and faster you will become. And your ears will become more educated to flaws that you didn't notice, or consider, before. After awhile, you will be able to do complex edits without giving them a second thought.

CHAPTER 5

Plug-ins/ Processors

Photo courtesy of Avid

This chapter is an introduction to commonly used plug-ins, what they do, general use, the difference between plug-ins and processors, and some finer points and advanced features that can be used to enhance voice recordings or fix problems that typically pop up in a session.

Photo courtesy of Avid

Plug-ins/ EQ

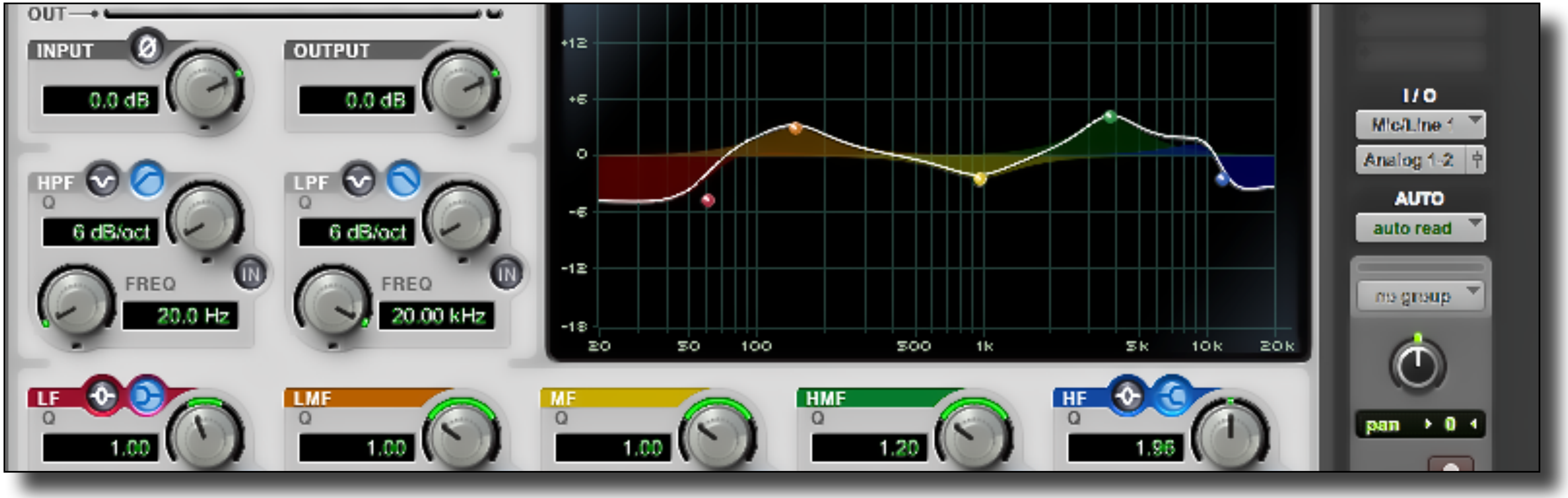


Photo courtesy of Avid

USING PLUG-INS/PROCESSING

PLUG-INS are processors inserted on an individual channel that change the signal without affecting the original signal. The parameters can be changed and saved and the original audio is the same. Most can be automated, but in most cases this is unnecessary. The advantage of plug-ins is that you can change parameters, go back to the session at a later date and change them again very easily.

Most plug-ins are signal chain devices such as compressors, limiters, equalization, and other devices such as reverb, echo, auto-tune, etc. This section will give you the basics of each type of plug-in/processor and how best to use them. Plug-ins have the advantage of being able to have your favorite settings saved and can be brought up ready to go for new session set ups.

PROCESSING is a destructive process: that is, you are permanently modifying the original signal. Once the session is saved, it cannot be undone. In some cases you can only process a soundfile, such as with Normalization (raising the peak volume of the file to unity) or Time Compression, or Gain. These are not available as plug-ins. If you decide after time compression that you need it longer again, you can re-process the file using time expansion; you can add gain if you subtracted too much, etc.

EQUALIZATION

This is the most basic and used plug-in/processor in your toolbox. It can change and improve the sound of your voice by modifying characteristics of your mic that may not be optimum for your voice. If your room/mic suppresses some high frequency, you can bring that up with eq. If your mic has a mid-bump that is not pleasing, you can pull that down. Or if your recording lacks low end, you can bring that up. You can modify the sound of your mic to mimic much more expensive sounding mics; and there are even "mic modeler" programs that have set eq characteristics found in high end mics that you can try. Mixed reviews on this, however; most people see it as more of a toy than a serious tool.

EQ WITH PROCESSING ONLY PROGRAMS

If your program is a process only, you will need to experiment via trial and error to find eq settings you like. If your eq lets you set multiple bands, start with 3 bands. Set the frequencies at 100hz, 1k, and 4k. Raise and lower the levels in those ranges until you get a curve that sounds good using "preview" If, however, your program only allows you to do one at a time with no preview, try this: Try boosting the bass: start around 100hz, add 3 db process, and see if you like it. Undo. Try boosting the 1k (1000hz) by 3 db. Process. Undo, 1k, cut by -3db. Process. Undo. Boost 4.5k by 3 db. Process. Undo. And try to the left or right of these frequencies until you have what you like. Then process one at a time, or all 3 at the same time if you can.

Remember, by processing you are making a permanent change to the signal once you save it. It cannot be undone. Keep playing until you have an eq curve that you like. Since you need to process every time, keep a note with your parameters handy. If your program allows you to save favorite eq curves by name, start a folder.

****IMPORTANT****

You have to check your monitoring, either speakers or headphones to make sure they are accurate before you start eq'ing your signal. If your headphones, for instance, lack high end, you will boost the eq in that range until it sounds good to you. Which to everyone else, sounds like you added way too much. You can overcompensate for your system's shortcomings.

You need to do legwork. Mix a file with it just the way you like it on your headphones and/or monitors. Send it to yourself on an mp3 and listen in your ipod. Burn a CD and listen to it in your car, on your home stereo (if you still have one of those) in your computer on the crappy speakers. If it still sounds ok, send it to your buddies at a recording studio and ask what they think. If they all say "way too much 4K" then you have over compensated for your monitors. Try another run with the high end flat. If everyone comes back with "could use a little more 4K", split the difference and send one more time. Buy them pizza&beer for their advice. Once you have the settings, you will get used to how your monitors should sound, and you will compensate correspondingly.

PARAMETRIC EQ

Photo courtesy of McDSP

A basic parametric eq has a control for frequency, and a control for gain. You can select what frequency range you want to modify and by how much. Some parametrics have a "Q" which is the width of the frequency range you want to affect. You can create a very tight notch to lower an offending frequency, or make it wider to affect a broader range of frequencies. Most parametrics have several bands, so you can boost 100hz by 3db with a narrow Q, cut 1000k by -2db with a narrow Q, and boost 4.5K by 4db with a medium Q. Most parametrics have a visual display so you can see where and by how much you are boosting/cutting. Some have "grab" points so you can grab a point with the mouse and move the db up and then move it to higher or lower frequency ranges, for instance.

Parametrics are the most flexible in that you have unlimited variables.

GRAPHIC EQ

We are all familiar with the classic graphic eq's where you can boost or cut in pre-selected frequency ranges. You have fewer options, but graphics can be faster to use.

HIGH PASS AND LOW PASS (OR BASS CUT/ HIGH CUT) FILTERS.

(I know....confusing: high pass lets higher frequencies pass i.e. bass cut; low pass lets lower frequencies pass = high cut.)

These are essentially one band parametrics that allow you to select the frequency where audio is cut off. If you set the Low Pass (High Cut) filter to 16kh, then it cuts all frequencies past that point. Or the High Pass (Low Cut) to 80hz, then no frequencies below that point will be heard.

These can be useful in reducing noise: if there is a very low frequency rumble or hum, or a very high frequency hiss. And if you are just recording voice, these frequencies are out of the range of speech, so you can clean up the signal a bit without affecting your voice.

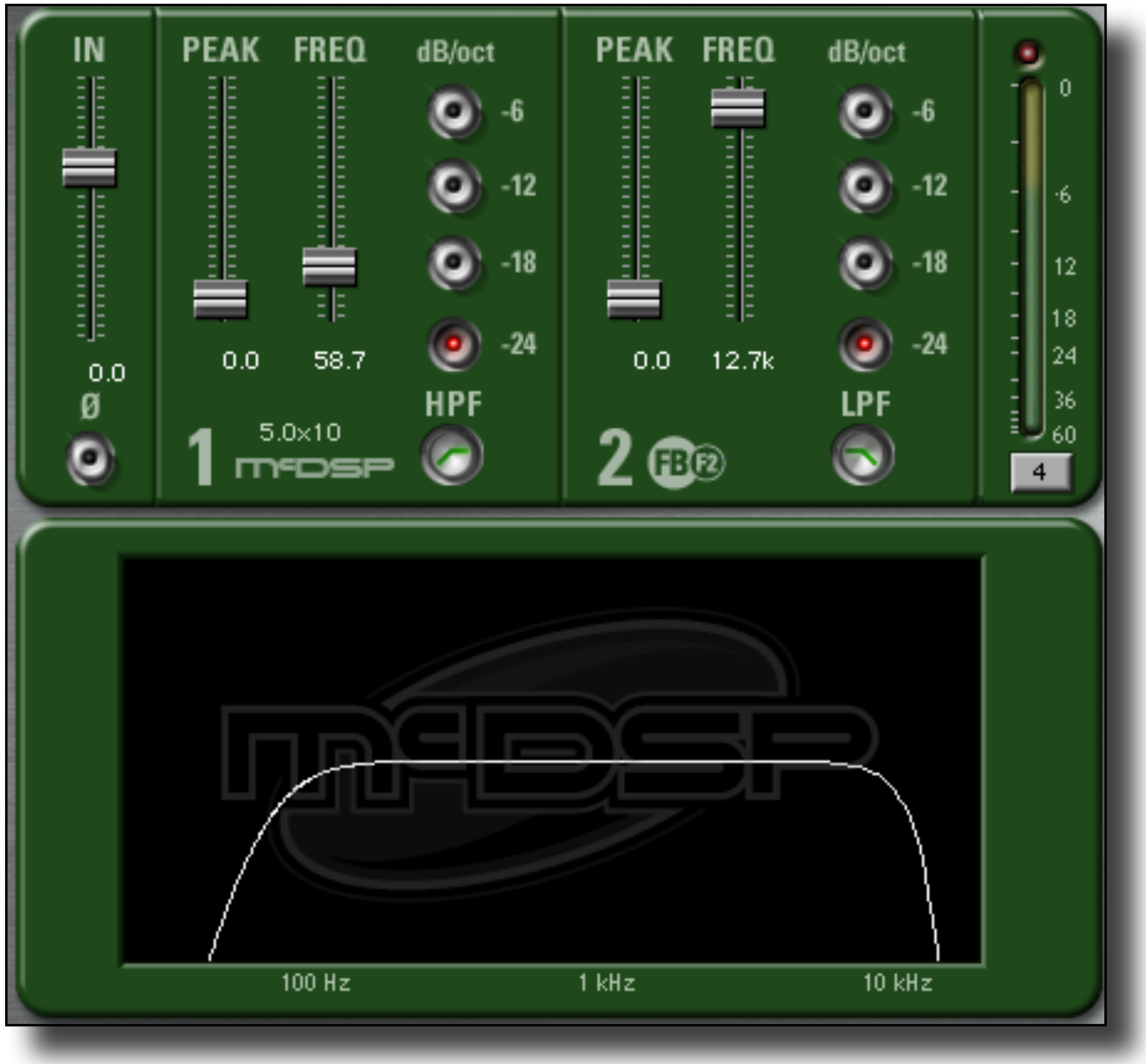


Photo courtesy of McDSP

A dual band low-cut, high-cut filter. In this example low frequencies are cut under 58.7 Hz, and also cut above 12.7khz These are useful in getting rid of hiss or low frequency rumble. These are usually separate filters for high or low frequencies.

Compression / De-essers / Limiting

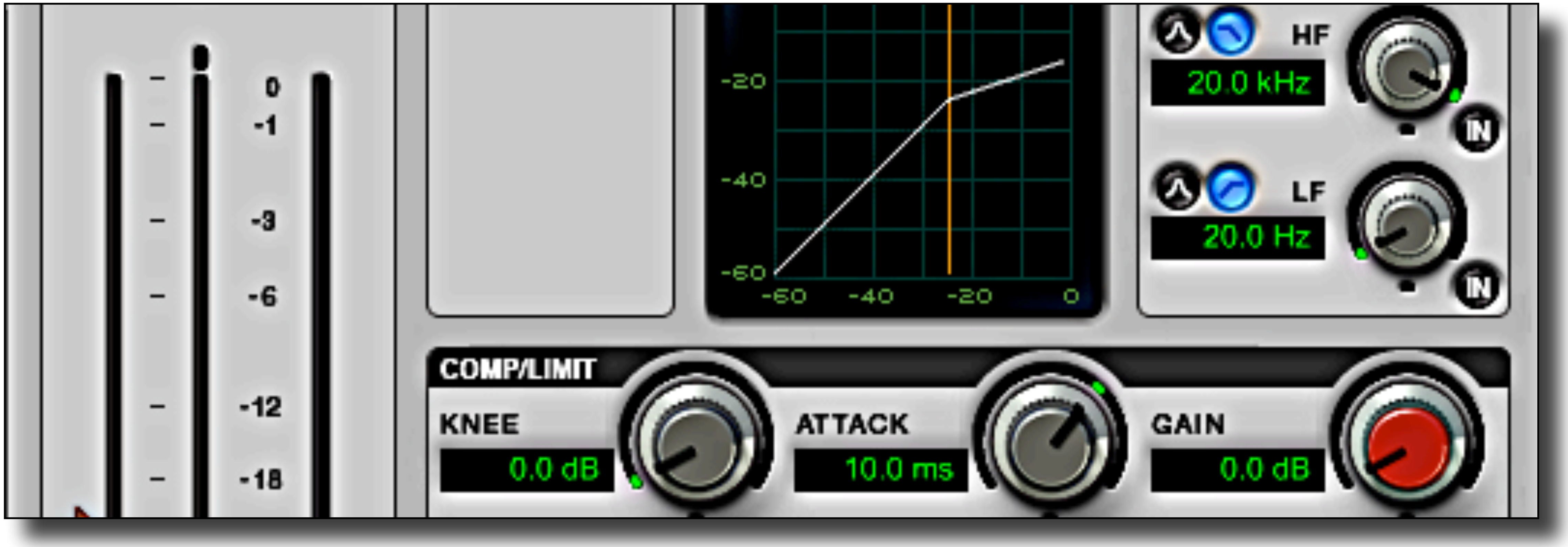


Photo courtesy of Avid

COMPRESSION

A compressor smoothes out the sound: it lowers the loud sounds so the soft sounds can be pushed up. Compression is an automatic gain rider, an electronic way of lowering the attack/volume of loud sounds. By increasing the compression, you lower the overall volume of the track. You can then increase the output volume of the compressor, thereby raising the overall level of the track before distortion. In essence, you squash the loud parts and then you can boost the soft parts. For most voice work, a 3:1 or 4:1 ratio sounds the most natural. The highest ratios become limiting, which we will get into next.

A compressor has attack and release controls, these allow you to select how quickly you want these parameters to work. Since most compressors are set to a default for voice (4:1) it is just a matter of setting the depth and   output volume.

Compressors are used in recording musical instruments with loud attacks, such as brass, piano, and even bass, hence the need for variables such as ratio, attack and release.

If the compressor is set to aggressively, you will hear a “pumping” sound, as if someone is turning the volume up and down very quickly. Back off on the input setting and it should sound fine.

Many compressor plug-ins have a wide range of pre-sets for voice that you can try and tailor from there.

DE-ESSERS

Photo courtesy of Avid

A de-esser is a compressor that operates in the higher frequency ranges to lessen sibilance. It will squash any frequencies in the range you set and by how much you select. If it is set too aggressively, it will pump or kill all other high frequency sounds, so a bit of playing around is essential to getting a good compromise. If there are only one or two “S’s” that pop out after that, process them first by lowering the gain -3db.

Also fiddle with the frequency range, the sibilance produced by   different voices varies.

Just like compressors, there are usually pre-sets that you can load to get in the ballpark.

LIMITING

A limiter is a compressor set with a "brick wall", that is, no level of gain will get past that point.

A limiter will aggressively compress a signal, and if set too high, can create distortion, or remove any dynamic range making the track sound unnaturally squashed.

Limiters can be used to keep the apparent volume level of a track up even with loud sound effects and music. U.S.30 Raceway spots come to mind. They are in your face with no subtlety whatsoever.

But if you are experiencing going into the red on just a few points in your read, you can put a limiter on your Output or Master channel and set it to prevent the overshoots on those words, while still maintaining the apparent volume of your read.

Generally, set it with minimal settings and increase them until you get the limiting you desire without coloring the rest of your read.

Limiter Photo courtesy of Avid

A compressor, in general, pushes a signal down, but releases it back up at a speed you can set with the release control. You can set a compressor to act like a limiter with some tweaking of attack and release times, and by selecting the 10:1 ratio; but most programs come with a limiter, so why bother.

But if you want to fiddle, by all means….it couldn't hurt and you will get a better understanding of how your tools work.

Channel Strips

Most of the major manufacturers offer Channel Strips, which are combination Eq/Compressors in one plug-in. They emulate features available in some high-end studio mix console channels.   And most produce the sonic characteristics of the console they are patterned after.

Izotope, Avid, Waves, McDsp, UA, IK, and many other manufacturers all offer channel strips, or multi-combination processing, (including gate, de-esser, limiting, and effects) like Izotope's Nectar plug in suite.

Photo courtesy of Avid

Time Compression/ Pitch Shift

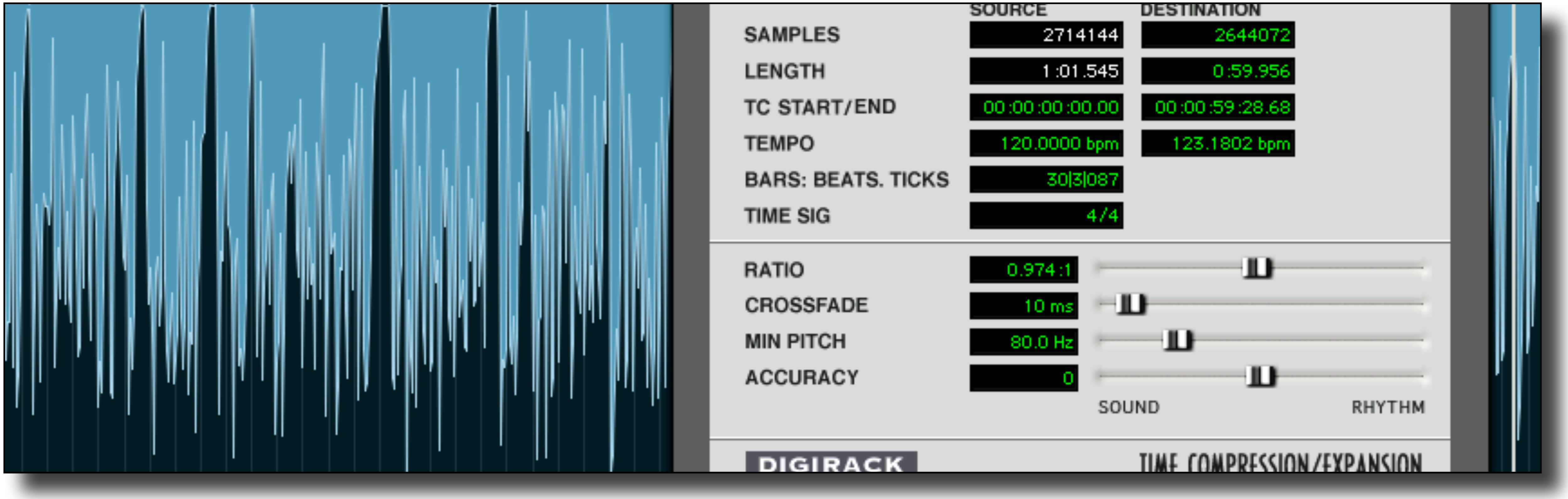


Photo courtesy of Avid

TIME COMPRESSION

You generally just highlight the audio to be compressed and set a target time and process it. Time compression is not a plug-in, it is in your processor drop-down menu, or audio suite, etc.

Remember, this is a destructive process, so once you save, this is the new file. (You can always go back and use the time-compressor to expand at a later date)

As stated before, most time compression schemes work by taking minute slices out of the waveform and putting in minute crossfades to smooth the transitions. The more you compress, the larger the slices until you begin to lose necessary samples for intelligibility. Congratulations. You have reached the point of gobbledygook.

Play with your program to see how far you can push it until you are aware that it sounds compressed. You should have an idea, for instance, that you can only compress a :30 by :02 without it sounding awful. This will let you know how far you can go.

Most programs have a ratio window that you make note of for future reference. Or just do it by ear. Some performances can be compressed more than others.

A more relaxed read can be compressed more aggressively, where as an energetic read will sound frantic at the same ratio.

There may be other controls on your time compression processor, some have controls for pitch, crossfade time, accuracy, etc. Experiment with these to see how they affect the sound.   And if all else fails, read the PDF manual that came with it.

You can use a slight bit of time compression to a program that seems to drag at certain parts, which can give a slight boost in apparent energy.   Or use it to match the pace of other speakers in the program to create a consistent pace overall. This should be used very subtly.

PITCH SHIFT/ AUTO-TUNE

These are programs that allow you to change the pitch of a read manually (in the case of a pitch shift processor: figure b) or automatically correct pitch with a program like Auto-Tune.(figure a)

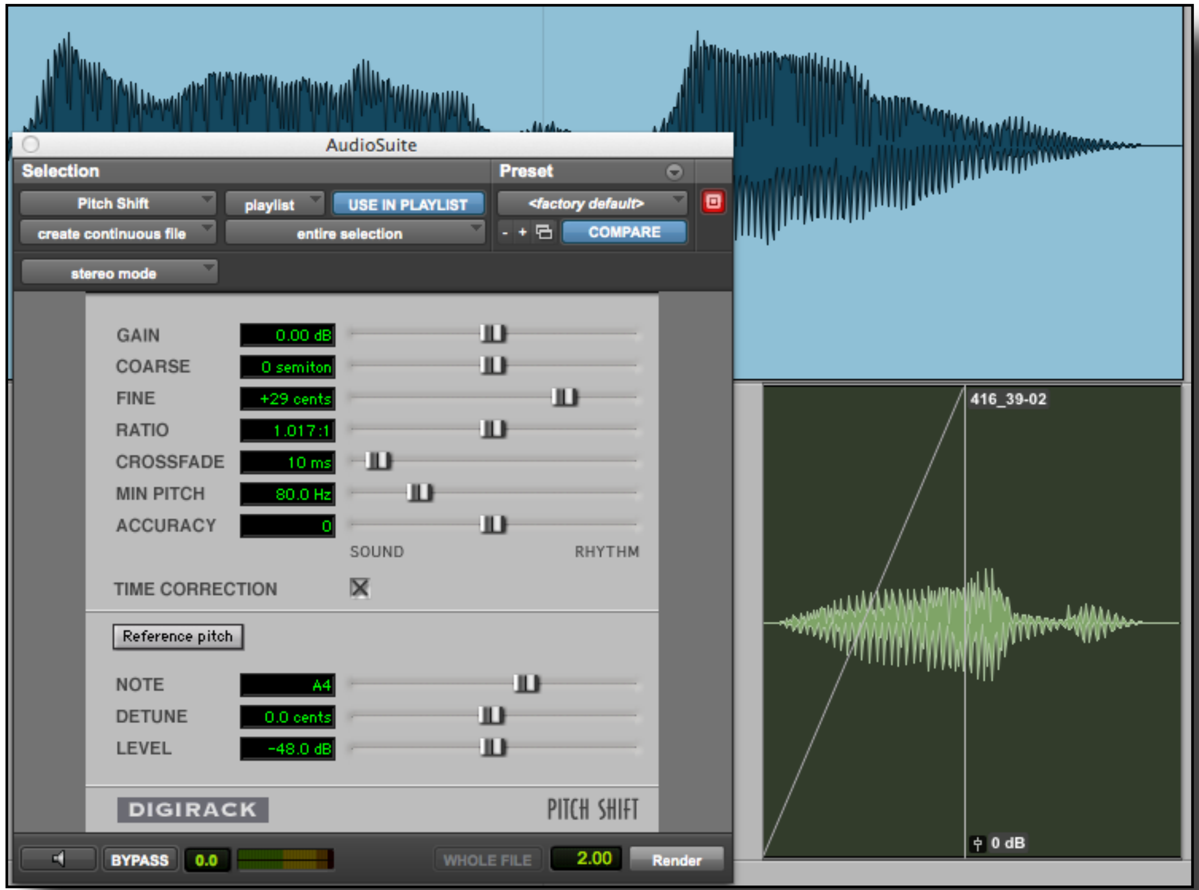
You can manually set how much pitch shift up or down you wish on a note per note basis (if you only have a few imperfections to correct) or if the singer is generally "A little pitchy, dog" use Auto-tune as either a plug in, or process the track.

Auto-tune and similar programs are pretty involved with many features and ways of correcting pitch, but they are a good tool for music/ jingle production and worth the time to explore and learn.

Photo/screen shot courtesy of Avid/Anteres
Figure a: Auto-tune is both a plug-in and a processor and is used mostly for correcting pitch variances on singers.

But for voice modification, the pitch shift program that comes with most DAW programs will work well enough to make elf or monster voices by changing pitch up or down. You will need to experiment to see how far you go before it sounds awful.

There are some 2nd party time/pitch programs that work better, and if you do a lot of this kind of modification they might be worth the expense.



Photo/screen shot courtesy of AVID
figure b: pitch shift processor

Most have a "preview" function that allows you to hear the changes before you commit to processing.

These can be of great help when doing multiple characters in an audio book with only one or two actors handling all the voices. The trick is to change them slightly without making them obvious.

ed note: I am not a big fan of yearly subscription models, but Slate digital has a suite of excellent plug-ins for $175 a year. This bundle has numerous pre-amps, compressors, EQ's, aural exciter/harmonic enhancers, mix bus compressors, mastering modules, reverb, delay, and tape recorder simulation. I have spent A TON on plug-ins from various manufacturers, that this bundle replaces.

My plug-in picks: Izotope RX6 noise reduction.
Slate Digital everything bundle.
time compression/pitch: Pitch N Time Pro ($$)
dialogue sync: VocAlign (project/pro)

There are tools out there for whatever you need, so troll Sweetwater, Mix, and download the 30 day trials. You'll be glad you did.

Noise Reduction / Noise Gate

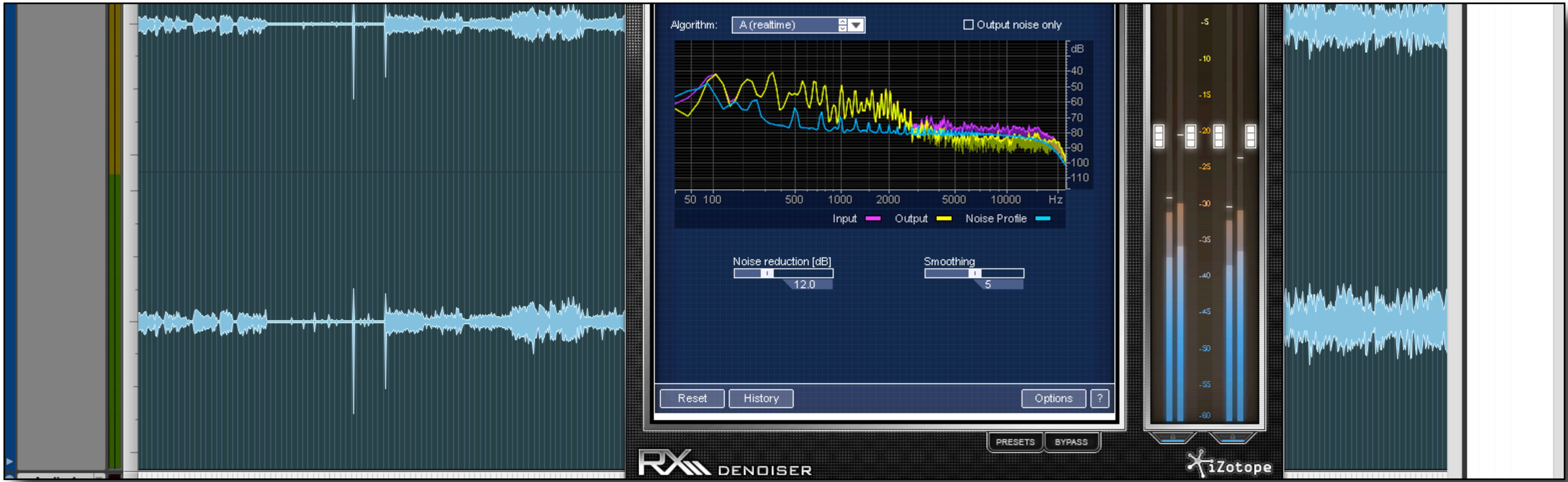


Photo courtesy of Avid/Izotope RX

NOISE REDUCTION

There are noise reduction programs ranging from free (included with your DAW program) and going from $100 to several thousand from 2nd party manufacturers. They all work basically the same: they take a sample of the broadband noise spectrum, and then eliminate those frequencies from the audio spectrum, hopefully leaving enough program material that sounds much cleaner and devoid of the offending sounds. Sometimes you have to compromise deferring to the overall sound quality of the voice, leaving in some noise vs. making the voice sound too thin, for instance.

You can always add eq to the voice to bring up frequencies that were removed after processing.

These programs range from highlight and process simplicity, to programs with a difficulty and complexity level of 10 with an inordinate amount of time needed to manually enter data points.

They all can reduce hiss and rumble, some can reduce or eliminate crackle and pops associated with vinyl (record) transfers, and most work fairly well eliminating 60 cycle hum without affecting the voice.

Each one has a range of features, and controls, and some work much better than others in addressing certain noise problems. Since most of these are 2nd party plug-ins, try the 15 day free trial download before you buy, to see which one suits your needs and is easiest to work with.

ed.note: I just got the new Izotope RX6: the plosive eliminator works better than a manual repair, the mouth noise module eliminates almost almost every click...I would recommend dropping the 300 bucks on this one.

NOISE GATE/EXPANDER

These work well in lessening breaths in long form programs, and lessening distracting ambient sounds.

These plug-ins detect the end of normal program material, and can be set to automatically lower or completely shut off the spaces in between words and sentences.

You can set the ratios to react immediately or gradually.

They can be set to lower the background sound or even diminish echo in a space, but care must be taken as to make it sound natural. If set too aggressively, they will cut off, or clip, the ends of words, and will cut off the beginning of words

starting with softer sounds, like “F”.
If the ambient noise is too loud, completely eliminating the sound between sentences will sound jarring. Sometimes you have to find a happy medium of just minimizing the background.

If it sounds like someone is noticeably adjusting the volume between sentences, you have gone too far.

And you may also notice a sound as the ambience disappears. Tweak the attack/release, range and threshold controls until this is not as noticeable.

Also, if there ambient sound under the voice and it disappears at the end of each sentence, that can get distracting to downright annoying, unless you are intentionally using that effect to end a thought.

If there is going to be music under the voice, you may be able to be more aggressive with your settings.

As with most plug-ins, there are a range of presets you can try out.

“DE-VERB” PROGRAMS

New on the market are programs that promise to remove some reverb and room sound from a sound file. This could be a great help to those who don’t wish to spend a lot of time or money on perfecting a great sounding studio/booth. But as with all short cuts, they only perform well to a certain point.

Check out the Youtube demos; download a free trial and see if it works for you. If it is a vast improvement, it may be worth the $250 or so. This is a business of sounding good and anything that can improve your sound is a good investment. And as is the case with all processors, you can only push them so far before they become a noticeable effect. (*ed note: de-verb is now included in Izotope RX 6 standard/advanced suite*)

Effects: Reverb /Echo/Delay

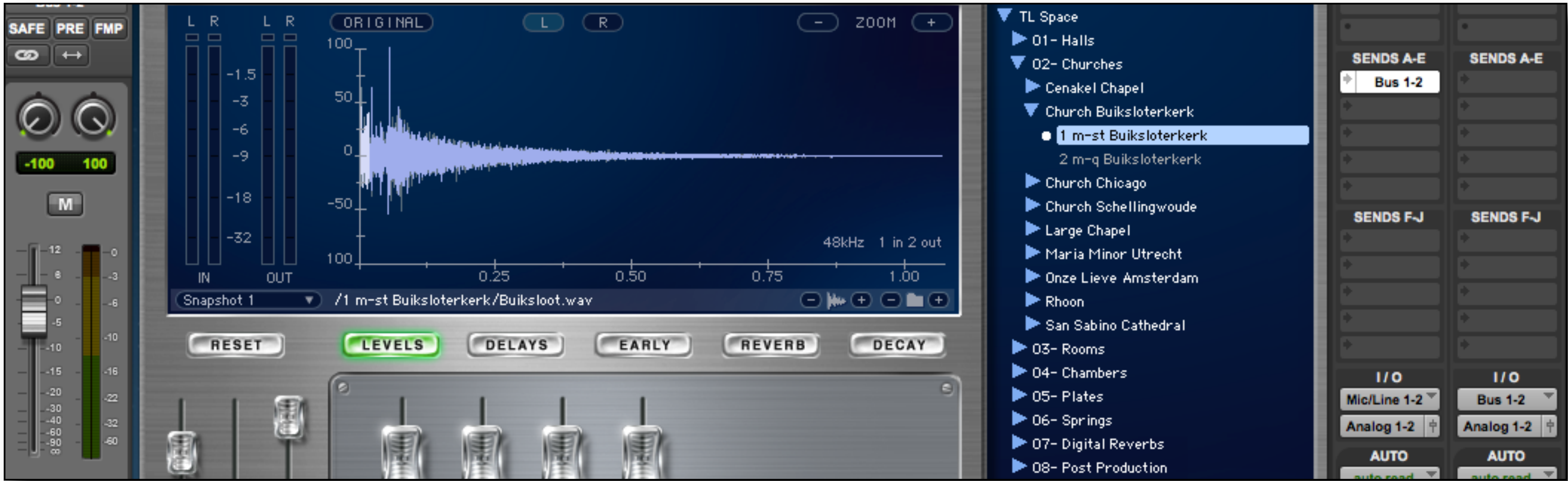


Photo courtesy of AVID/Trillium Lane Labs

REVERB; ECHO:

Most programs come with reverb and/or echo (delay). Both are used to simulate spaces from inside garbage cans, to concert halls, to inside massive caves. Most come with a library of spaces to get you started.

Convolution reverb programs are computer recordings of how sound behaves in an actual space and duplicated in the reverb program.

Others are mathematically created through synthesis, and most are very good at creating the illusion of space intended. They allow you to put your voice, your guitar, your music, in virtually any environment.

There are expensive, high-end reverbs produced by many famous name companies to replicate the sounds of their very expensive hardware units, (which are on the wish list of most audio geeks) but there are many alternatives that are very reasonably priced that will get the job done.

I was disappointed when my favorite go-to reverb was no longer available after I upgraded computers, but found an adequate replacement for $50.00 that filled in the gaps I needed. There are many available, and most will give you a 10 to 30 day free trial.

Most come with presets that are very descriptive, and will get you into the ballpark even before tweaking. Some have pictures of the spaces for reference.

Scroll through some presets before you decide on one, you may find one that sends you in a different direction than you had in mind.

For instance, you thought a concert hall might be good for the “string quartet” but the “nightclub” preset gave it a more intimate, and “real” sound.

When you first plug in a reverb, you will have to adjust the mix control, since most come up 100% “wet” (full reverb) You will need to turn the mix more to the “Dry” side before it sounds real, and tweak from there.

Or if you are using an Aux channel, leave it at 100%.

TASTE

And always remember: reverb is a seasoning: too much can overpower the voice, (to mix metaphors) and **the more reverb you use, the farther**

back in the mix it pushes the original signal. This works very well when you want to push the music track back a bit so that the voices stand out. I have heard mixes where the band is washed in a smooth reverb and the lead vocal is completely dry. It is almost jarring, but works astoundingly well.

The band appears to be 25 feet behind the singer, who appears to be 6 inches from your nose.

And if you listen critically to music from different eras, you will hear a different style, and sound to the reverb used. Not only can you create a space, you can create the sound of a recording from a particular era. If you are doing a period piece narration, this can give your recording more "authenticity".

Most reverb programs come with a "first reflection" parameter that allows you to set the time before you actually hear the echo/reverb, which works well in large spaces, but can help simulate a realistic sounding "room".

Play around with the parameters and see what you find intriguing. You can name and save changes you like into the presets folder and add them to your reverb library. When experimenting, use headphones since it will give you a more accurate monitoring of the space you have created. You will also be better able to hear early reflections, damping, etc., and how each one affects the perceived space.

ECHO/ DELAY

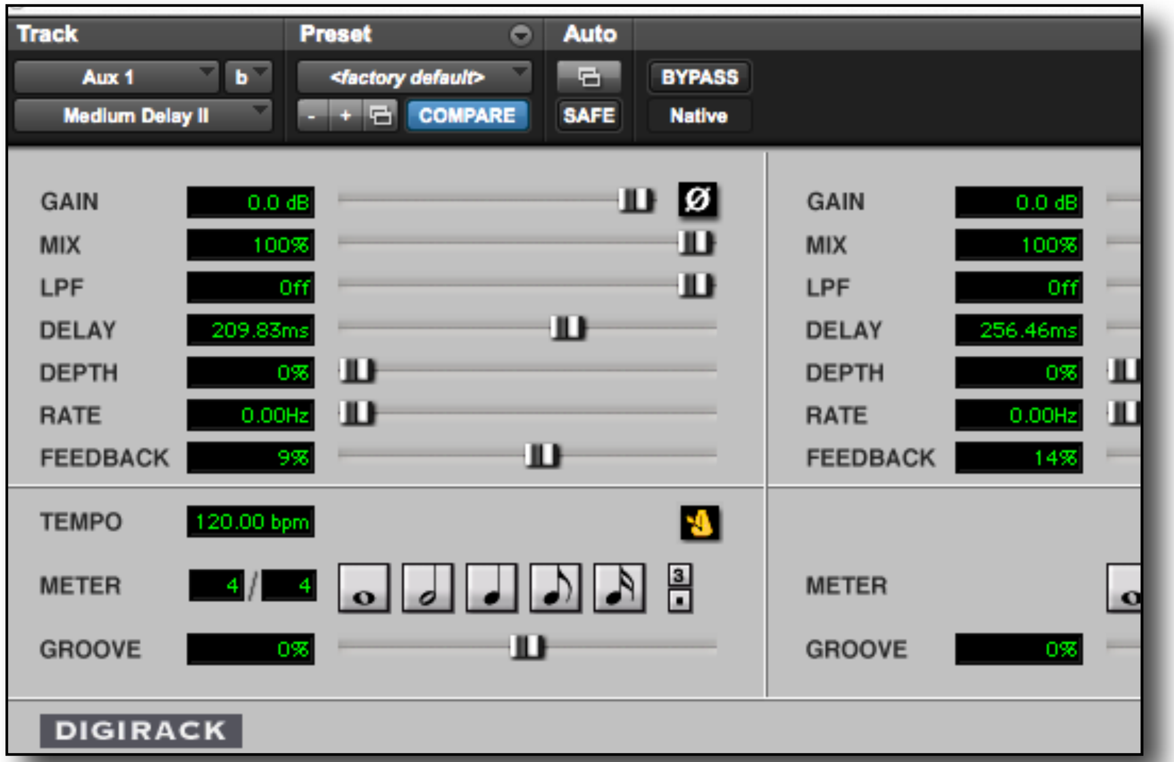


Photo courtesy of Avid

Echo is usually distinct "reflections" (or echo, echo, echo) with parameters for how soon you hear the first echo, then the second and so on. Most have presets like "slap echo", the sound used heavily in 50's recordings, to large spaces like "stadium" and most have parameters to control how long the echoes regenerate. Each echo is usually a few db less than the one before it, until they fade completely out. Most have left and right echo controls with individual time parameters you can vary, along with decay time. If you want to simulate a stadium P.A. announcer, you would probably want to use echo instead of reverb. (Although I am presupposing there is not a great "stadium" preset in your reverb program.)

PRE-ECHO

You can modify your reverb programs by putting echo first in the plug-in chain, and then adding reverb under it. But be aware, the distinct echoes will be somewhat washed out by the reverb. Play with the mix on each until you get a sound you like.

If you want to keep the echoes distinct, but want to use reverb too: use sends from the channel to 2 "aux" busses and have echo in one, reverb in the other.

DELIVERY:

Some clients want the flexibility of adding their own special simulations, and will want you to also send a "dry" version of your read. Other times, they appreciate the magic you created and will print the version "with". I usually send both. In some very rare instances, they want the voice dry and just the reverb printed so they can mix at their end.

EFFECTS

Flangers, phasers, ring modulators, etc.

These are simulations of hardware effects: ***Flanging*** was when you physically touched the flange on a moving reel of tape (the flange is the metal part of the reel) on an analog tape recorder causing the tape to move out of alignment with the playback head giving a swirling, whooshing type sound that moved spatially.

Phasers allow you to give the signal a swirling effect popular with guitarists. And so on.

All of these and hundreds of other effects are available as plug-ins to modify and transform a track. Used judiciously (and with taste) they can add an interesting twist to music or voice, or when used to enhance sound effects.

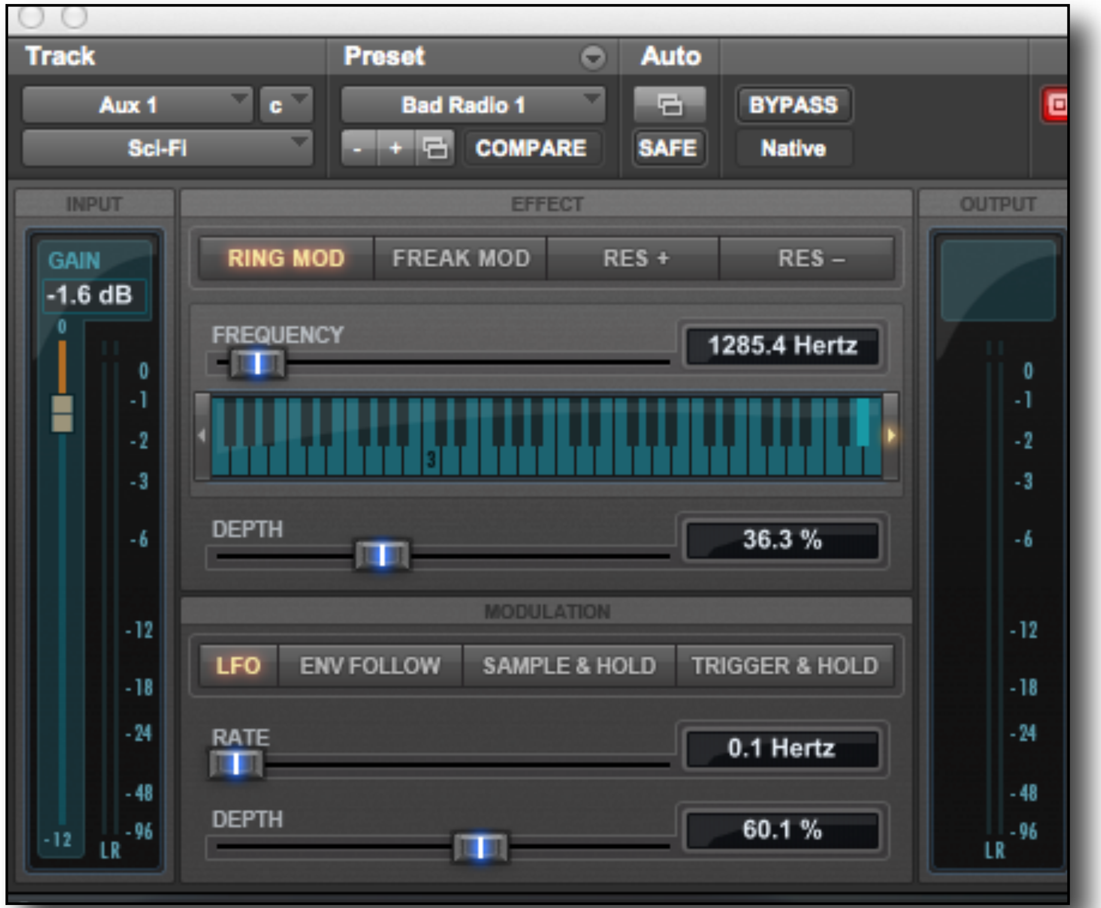


Photo courtesy of Avid

CHAPTER 6

Music/Sound effects

This chapter is primarily about music and sound effect libraries, royalty free vs. pay per play, music search, legalities, parody versions; picking and editing an appropriate music track, editing pop songs; creative use of sound effects, Theater of the Mind; and mixing, "L.I.A.R" technique; And plenty of editorial comments.

Music Libraries

Legal Disclaimer

I am not a lawyer, paralegal, or expert; these are just some basic rules I have picked up, and I would highly recommend discussing any specific questions with a copyright, entertainment, or contract attorney.

Out of ignorance, at one point or another, I have violated every one of these and was lucky enough to have only gotten a stern lecture.

Pay attention to contracts, details, terms and conditions.

The penalties for a mistake can be very costly.

MUSIC/SFX LIBRARIES:

Your clients will inevitably ask you to produce finished spots for them, since you have generously been doing your studio work gratis. They will expect more of the same, thereby cutting their audio studio out of the loop and pocketing the $175 an hour they used to pay. Library music is a good profit center;

even if you don't want to charge studio time. (which you should.) At least up charge for drop fees. Remember, they will never pay you more than they pay you now. Business principle #17.

There are a bunch of things you need to know before just slapping music over a voice.

There are two kinds of libraries: Pay per play/ Contract, and Buy-out or royalty free.

MAJOR LIBRARIES

All music you purchase or download is copyrighted and owned by the library. They are charging you for its use only in a specified program or production. You cannot take it and sell it to someone else. Most stipulate you cannot send an unmixed version to anyone else. (such as, you did the radio version, and now the video editing house wants the music in the clear to do their own mix for the television spot: verboten.) Unless it is royalty free, you cannot use it in another production without paying them again for each use.

Radio and television are separate charges. Some consider a word change another version. Most have different fees for radio, television, web, non-broadcast, different markets, etc. Some of the major libraries, like Killer Tracks, Non-Stop, Extreme, DeWolfe, etc, have blanket licenses that for a yearly contract, allow you to download as much music as your contract stipulates, allowing you to charge your client for each use, which will hopefully offset the contract fee. Most have a report form they request you fill out for every use; this is used to track and distribute ascap/ bmi payments to the composers. Be forewarned that willful copyright infringement is expensive. Even if you do win in court, the legal fees you had to cough up for a lawyer will bankrupt most people.

Play by the rules in this game.

The advantage of playing with the big boys is that you have a better selection of music with higher production values (real strings vs. synth) and most of the selections have been cleared by musicologists as not to have stepped on any other copyright holder's toes. (There are many sound-alike productions that just skirt copyright infringement rules….the big guys have already cleared these) Prices average $125 to $225 per download per use for television or radio in a single major market, but that is dependent upon the library. Each one has a very detailed price list. But a contract is a contract and they will hold your feet to the fire if you violate any of the terms.

MUSIC SEARCH

Most of the major libraries have an on-line search engine where you can preview the type of music you want, using search word categories, such as "50's- upbeat-guitar-doowop" or "sounds like" and you enter the name of a specific tune, and they will assemble a page of possibilities for you to preview. You then pay via credit card or enter your contract number and download the piece you have chosen. You can request either an mp3 or higher quality file.

Most of the smaller royalty free libraries also have ways of finding the type of music you are looking for either by category, feel, etc. And do consider your time searching for the right music tracks as billable time: you are doing work for your client, and should be compensated. Usually search time fee is 50% of your regular studio time, but should never fall below Big Box Greeter wage.

ROYALTY FREE

A less costly solution is buy-out or royalty free music. There are tons of sites on the net that sell very respectable music tracks with some sounding as good as those from the major libraries. You pay once for it and can use it in perpetuity in any and all productions you may produce in the future. (charging your client every time) There are usually no stipulations other than you cannot repackage it and/or sell it. They still hold the copyright, they are just allowing you unlimited use of the piece you buy. Most are downloads, using credit card or paypal; some sites still sell CD disks.

Prices range, on average, $35 to$95 per single track. Always keep a screen shot of payment transaction in the event a dispute arises. You do have to be careful with some of the smaller players, however: if you hear a track that sounds JUST LIKE a Beatles track, you could be on the hot seat if someone from the Beatles' legal team determines that the commercial you put on the air with this track constitutes infringement. You will get off the hook as far as willful infringement, however, you will still be living in a van after paying legal fees. Another rule of thumb, if it sounds too close and you feel like it could be a problem, find another track. Also, if you have a track that sounds very close to a certain Stones' tune and in the copy you say "If YOU can't get no satisfaction….." gotcha.

MORE LEGAL: PARODY VERSIONS

Don't think that you can just grab a karaoke version off the internet and sing new lyrics to it and you will be protected because "you bought it from the internet and the guy said it was cool to use it in a commercial."

The rule of law covering parody does not protect you if it is a commercial venture. (there are many interpretations of what is permitted and what is considered a commercial venture: best to contact an expert when you get into this arena.) Even Weird Al Yankovic has to get permission/ clearance and pay a portion of the royalties to the copyright holders for his parody songs, since they were for sale and therefore considered a commercial venture; even if you don't actually make a profit.

To use a pop song in a commercial, it first must be licensed by the copyright holder/ publisher. But be prepared to pay anywhere from $35,000 per quarter for the rights only, plus you then have to pay to produce a sound-alike music track with parody lyrics. You will pay much more if you want to use the actual recording, (synchronization rights) since the original musicians, or their heirs, will demand much more than studio musicians would.

There are quite a few companies that specialize in securing rights for commercial use of pop songs. If you are going this route, use one: they have the inside track on all of the legalities.

In the past, if you stepped on toes, you would get a Cease and Desist order, forcing you to take it down immediately or face a suit. That is no longer the case. A lawsuit may be filed immediately, and the first you hear of it is when you are served.

RULE OF THUMB: If someone else owns it, chances are they have taken steps to protect it, and always assume they have a hyena for a lawyer who will not leave a scrap of meat on your bones.

******SIDEBAR: OTHER LEGAL POTHOLES**********

Using a line from a movie and sound-alike voices: You may have the perfect kicker for a spot with a dead-on Jack Nicholson impersonator repeating the "truth" line, which makes everyone who hears it laugh out loud. But you can't use it. You could be sued by the studio that owns the rights to every character and line from that movie, and by the actor who owns the rights to the sound of his voice, and can sue for whatever fee he chooses. The "celebrity voice impersonated" disclaimer is no longer a "gimme". Rodney Dangerfield successfully sued for 99% of his very high voice-over fee, on the grounds that most people thought it was actually him endorsing the product, (the impersonator was dead-on) so for :58 seconds, in most peoples minds, he was the voice and was entitled to compensation. He won, and the client had to cough up the dough. A very famous Bette Midler sound-alike suit made Ford pay up more than her original asking fee. This is one reason why you rarely hear celebrity impersonations any more.

ANOTHER RULE OF THUMB: If you are intentionally trying to confuse the public that this is the actual band, actor, singer, or section of a movie, no matter how far fetched a parody, you can be contacted by their legal team to cease and desist.

Case 1: We got yelled at for a radio spot that sounded like a well known cartoon franchise, complete with cartoon voices, music, sound effects, and was even filtered to sound like it was from a 50's cartoon….and even though these were not the specific characters owned by the studio, (the music was original but in the "style" of theirs, and the sound effects were from their commercially available SFX library),……we were cautioned because in totality, it WAS close enough to confuse the general listening audience that this was their property, likeness and brand.

Case 2: We were approached to "go ahead and do some parody music without clearance, who's going to catch us, it's only playing in this one small market" by a local retailer. We said no. He got someone else to do it. A few weeks into the run it was heard by someone who knew the composer, dropped a dime and they got a cease and desist. They had spent a lot of money on studio time, musicians, singers, announcers, and it only had a two week run. And the radio and television stations will yank it immediately. They do not want to be named as a party in a lawsuit.

Case 3: Some dear friends had a terrific band that did a song about the Stooges. They used lines from Stooge movies and very good impersonations. It was very cute and in no way damaging to the reputation of the actors. But they did not have permission to use them. It was a hit; they were climbing the charts until the band was virtually sued out of existence by the heirs.

Enough legal.

SIGNATURE MUSIC

This may be an original piece of music or even a library piece that everyone loves that you use again and again behind a particular client's spot. After a while, it becomes a signature piece, almost like a jingle, in that people associate your client with that piece of music. It becomes a part of their campaign and actually determines the way spots are written around it. If you can find this piece, you are the hero. There are numerous examples on TV, from the Christmas Lexus campaign theme, to the Jimmy Dean "Sun" campaign music, and so on.

Picking, Editing Music

PICKING APPROPRIATE MUSIC

Generally, a music track should provide a platform that supports the tone of spot, and helps to enhance the voice-over. A hot rap/house-track behind a spot for glucose metering supplies does not hit the target demo. Nor does a classical violin behind a spot for a video game work either. (Ok, if you are clever, you can use a music track as an element of irony, or as a very funny contrast, but that usually has been written into the copy well ahead of your music search.)

Remember, it doesn't matter what kind of music you like, you are charged with finding music that will work: to push or support the voice and work to propel the message.

There are several things to consider when picking a music track.

Taste and thinking outside what is safe/expected is a good start. But here are some more too:

What is the pace of the announcer? Most voice talent have a pace, an inner click track. If you can find a piece that is close to the pace of the announcer, their voice just seems to fall into the pocket and it is almost like a rap, only not as annoying.

Look for tracks that fall into general category of spot. You don't want blues if it is a retail spot, unless again, the spot revolves around the "I got the I can't find a good prom dress blues"

You don't want a track with minor chords for a spot that's supposed to be cute. And you don't want an accordion track unless it is supposed to be funny.

Look for tracks that are appropriate to the type business, the types of music commonly used. Classical for "quality" furniture stores, real estate, etc; heartfelt "caring" tracks for hospitals, medical; upbeat, "call to action" tracks for sales, retailers, auto, etc; are good starting points, but still take your cues from the tone and pace of the read.

But these are safe, middle of the road choices, and the music is nothing more than background. You can find good pieces in these categories, but it takes a bit more work to find something that isn't completely expected, ordinary and forgettable.

And you may stumble upon a piece that is out of your initial search ideas that works incredibly well. A solo guitar that has just enough quirk to work perfectly with the wry read of the announcer. Or a silly music track underscores the comedy of a spot.

Or a dirge that makes the announcer's tale of gloom and doom seem even worse. (you would probably want to switch to an upbeat piece of music when he provides the perfect solution to the problem when he switches to a positive tone)

In other words, be open to something that will work outside of your initial search options.

Talent may even want to try to read along with a piece of music, it may pick up their pace, or energy, and give them a new slant on their read.

EDITING A POP SONG BEHIND THE ANNOUNCER.

This really needs to be said: there are a bunch of ignorant, lazy, (or to be kind) unaware producers who run a vocal section of a pop song behind the announcer, without cutting in an instrumental version of the song in the sections where the announcer is speaking.

Having the singer compete with the announcer is the most common bush-league practice that is now practically industry standard, but it still stinks. You cannot listen to a singer who is singing a lyric (that usually has nothing to do with the product) and pay attention to what the announcer is saying at the same time.

It is distracting.

And most think just ducking the volume of the track when the singers come in will do the trick, but most songs feature the singer up front in the mix, so you have to lower the overall track volume too low (and you still strain to hear the singer if you know the song.)

INSTRUMENTAL VERSION

Old school is best practice here. Get an instrumental version of the song and cut in instrumental sections where the announcer will be speaking. This way you can still keep the level of the music up higher, which moves the spot better. Or find instrumental sections, lead-ins, etc. and cut together a version from pieces.

CUTTING THAT MAKES SENSE

Just make sure it makes sense musically: you will need to cut on 2 bars or 4 bars and keep in mind the chord progression and arrangement of the song. It should sound as though it was played that way when you are done cutting. Cut on the null point, or use cross fades to smooth the edits. If you are good, you won't need the cross fades.

Cutting on a snare hit is usually an easy choice since they have a distinctive waveform and they are usually in a corresponding spot in the track (usually on the "2" and "4" beat in each measure) You may need to use a bit of time compression to get it to :29.5 or :60.

VOCAL ELIMINATION PROGRAMS

There are "vocal elimination" programs that can lower the level of the vocalist, but still leave residual vocals present.   Some of the newer programs are better at this, but some also remove frequencies of other instruments within the vocal range.   In most cases, getting an instrumental version of the track and cutting in the pieces you need still sounds best.

Since your client will be paying a healthy sum for a pop song usage in the first place, securing the instrumental version should be no big whoop.

EDITORIAL:

If and when your client/agency asks if you have any ideas for pop songs, please suggest one that has a direct connection with the product, and not just a hook line that they can shoe-horn in to try to make it fit with some vague connection.

I can point to some very well thought out uses of pop songs to underscore the brand message smartly and effectively, but an equal number of horribly inept uses: “Could someone explain to me why they used *that* song? What were they thinking???”

Keep your thinking cap on next time you watch a bunch of spots and try to figure out why they picked the track they used. And if you don’t have a clue, and you can’t even remember which credit card the spot was for, it was a bad spot. (and no, it was the other one.)

Plus a little thought into the song itself might not be bad: I have heard several songs from the 60’s (rife with sexual imagery) that have been used recently behind children’s products. If they had listened to the entire song and not just the one cutesy line, they would have recognized the innuendo. The client and the agency people didn’t get the double entendre.

Embarrassing.

So just find music that makes sense, underscores the product, and PLEASE: don’t suggest James Brown’s “I Feel Good”.

It’s been done. (HEY!!!!)

EDITING MUSIC (advanced stuff)

There will come a time when a library piece is too long, or too short, or doesn't line up with the change in copy.

Here are some editing tips:

Always cut on the beat, usually snare hits are easy to find in the waveform, and they are usually on the 2 and 4 beat. I always like to cut in the null right before the beat, that way the decay on the snare will fall after the cut.

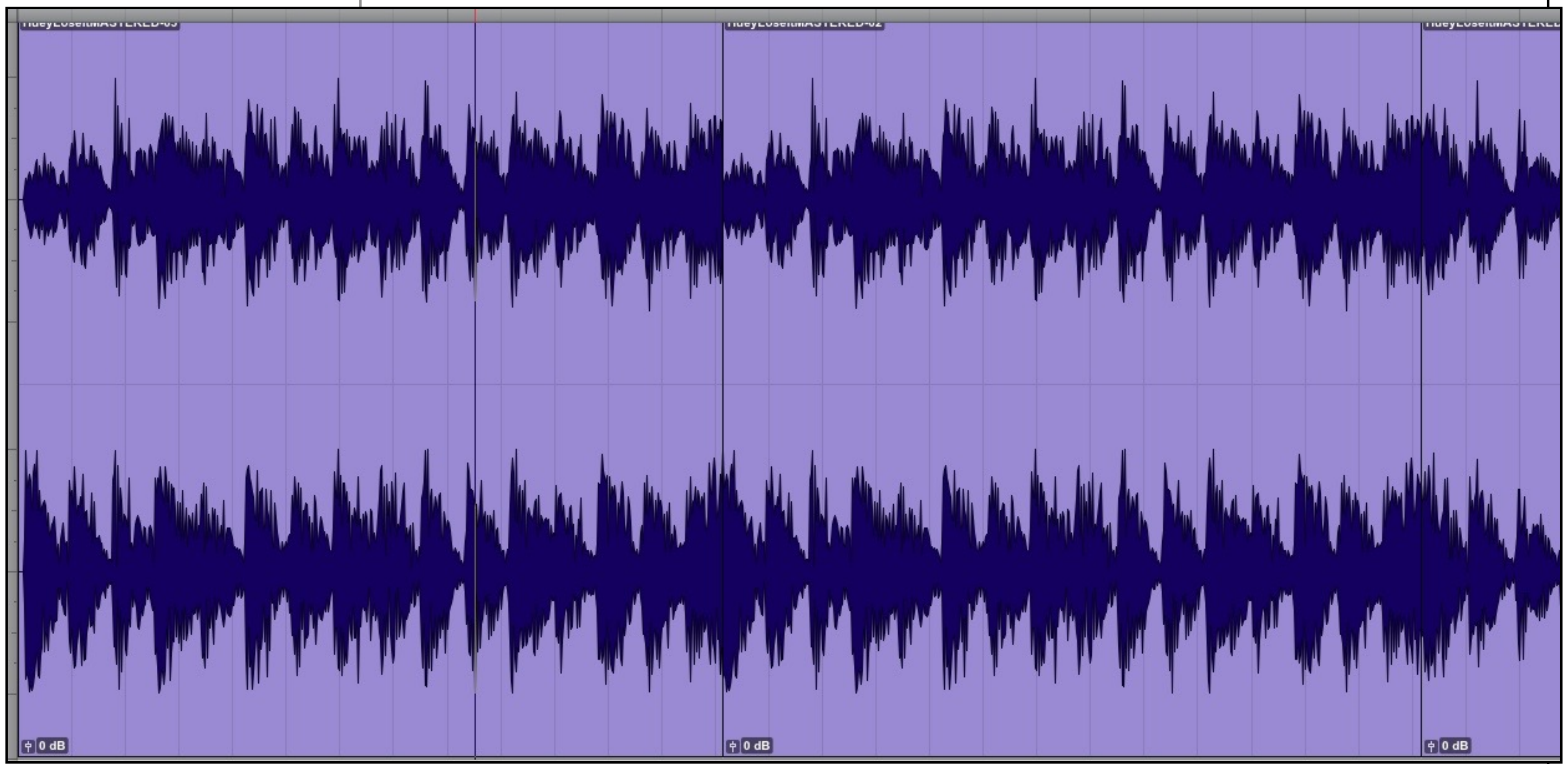
Always try to keep the bars the same, if you are running 4 beats to a bar, try to maintain that. You can add or subtract bars as long as it makes sense.

You can cut beats 2 out of a bar, but unless it is under the announcer, you will hear it since we tend to keep time in our head.

Use crossfades where you cut, unless you get good, or cocky, which will come with practice.

ADDING 4 BARS (16 beats)

Here is an example of extending the opening by 16 beats: Starting at the drum intro, Copy the section up to the drum fill leading into the next section. Using a slide edit, paste it at the point where you want it to begin at the drum fill.



You can see the two vertical lines where edit was pasted. (ignore the first line, that was the marker)

Use your trim tool to adjust the boundaries, sometimes dragging a little bit of the previous section into the new one blends better. Or you can use a crossfade at the sections.

VOLUME DISCREPANCIES

Sometimes, as you can see in the previous example, the drum intro in the upper channel is lower in volume.

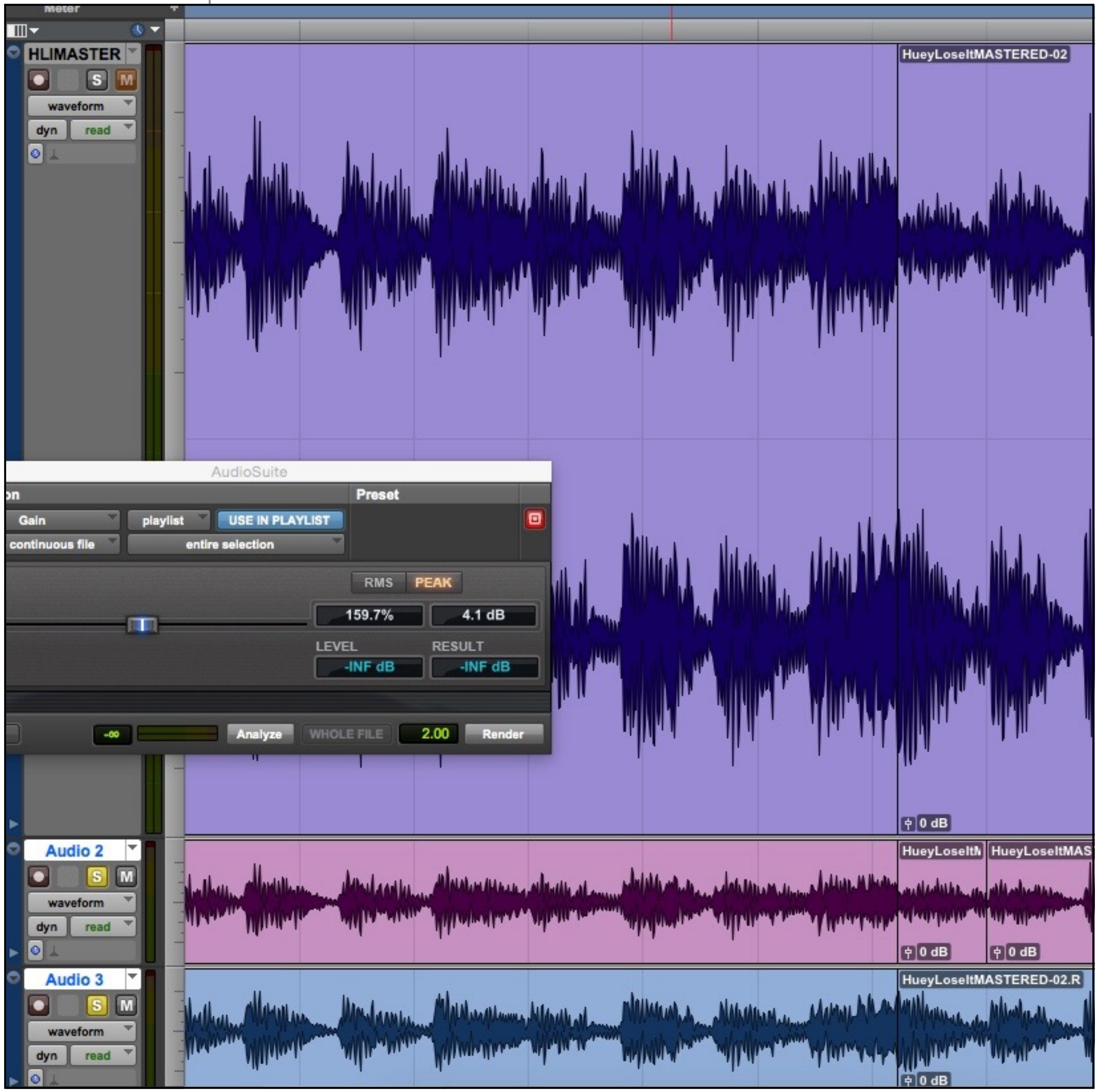
HOW TO FIX:

Add two mono channels.

Copy and paste the stereo channel to the two mono channels.

Adjust the gain in the section you want to change.

Either leave it as two mono channels, or copy paste back to the stereo channel.



2 CHANNEL EDITS.

You copied a section from a different spot in the track, but the sax is trailing off past the edit point you want to use. It suddenly cuts out and is jarring.

The easy way is to use long crossfades at the edit points, but sometimes this does not work.

Here is a fix, and this can work for many other things, like adding a drum fill, or a music hit, or even crossfading into a different piece of music.

Start by cutting in the section you want to add.

Add a second channel below.

Drag the section to the 2nd channel

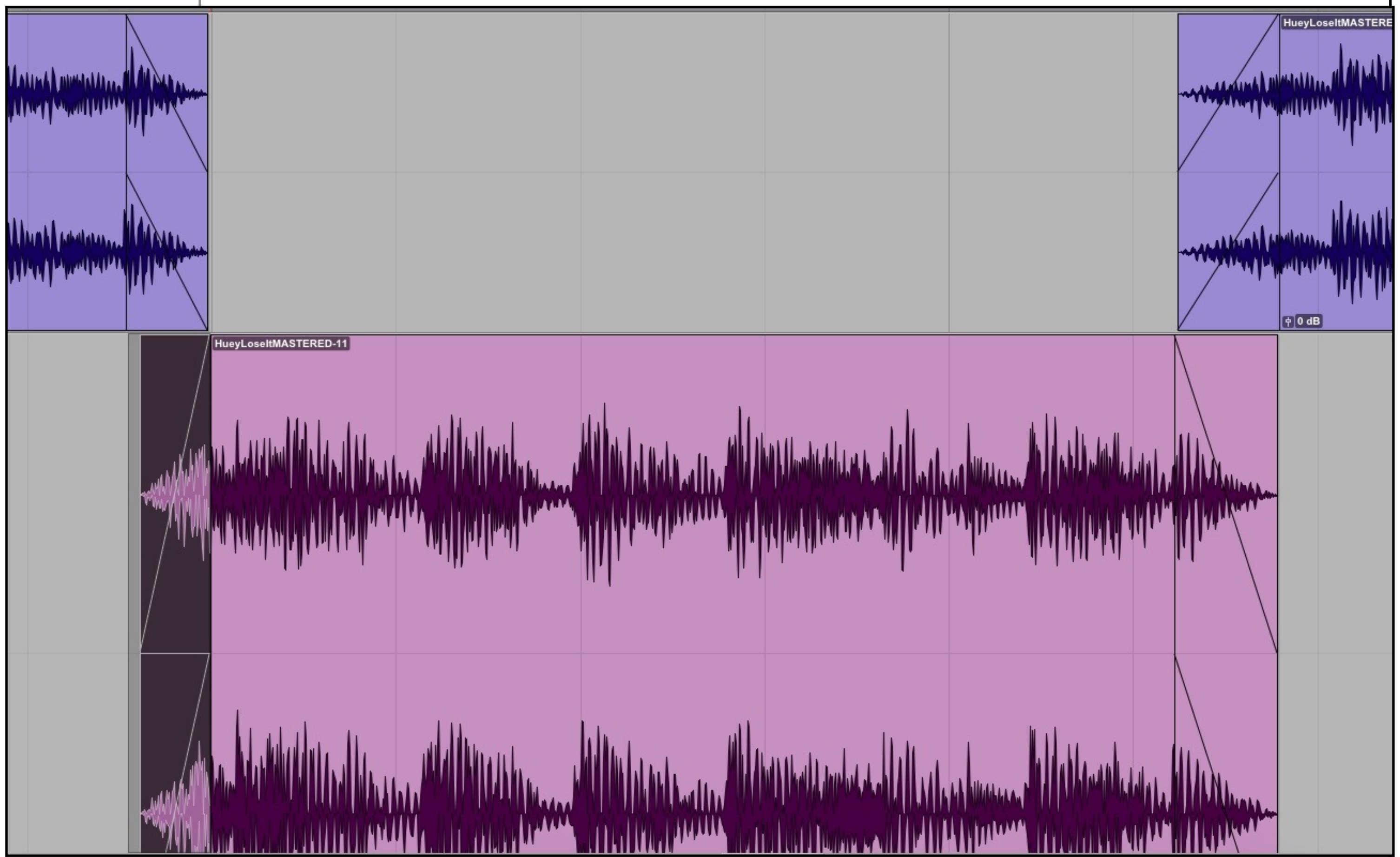
Using you drag tool, expand the edit points.

If you do not have this feature, start by adding space at either end of your edit points before you copy, and paste directly to the channel below. (You do not need to be accurate at this point)

Cut the top channel at the edit point and slide to create the space for the new section below.

Slide to align the edit points, then move and trim the sections so you have "overhang"

Now create fade outs and fade ins on both channels.



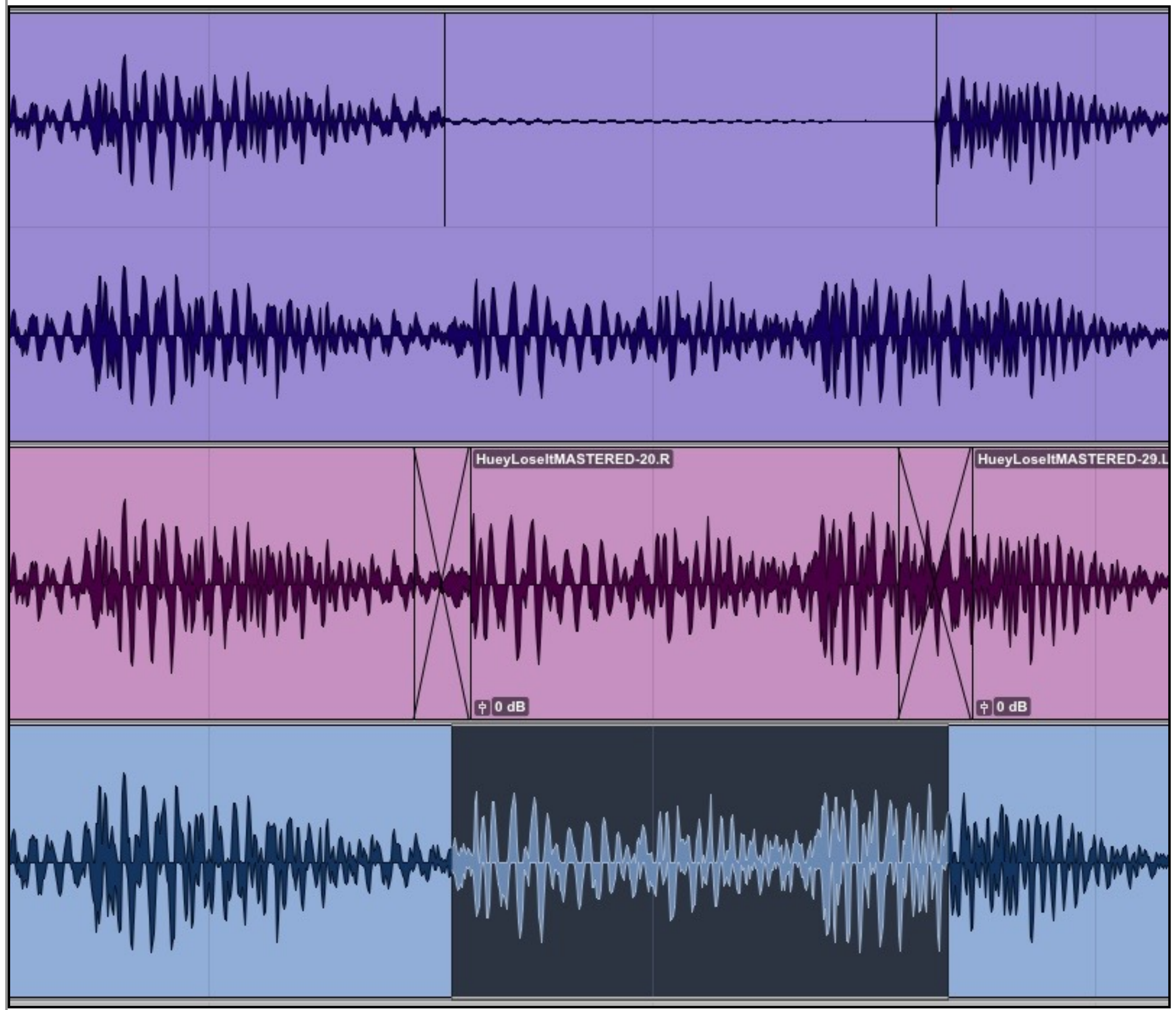
(This is also in the ***Fixing/Mixing*** section, but I thought it should be included here)

FIXING A PROBLEM IN A STEREO FILE:

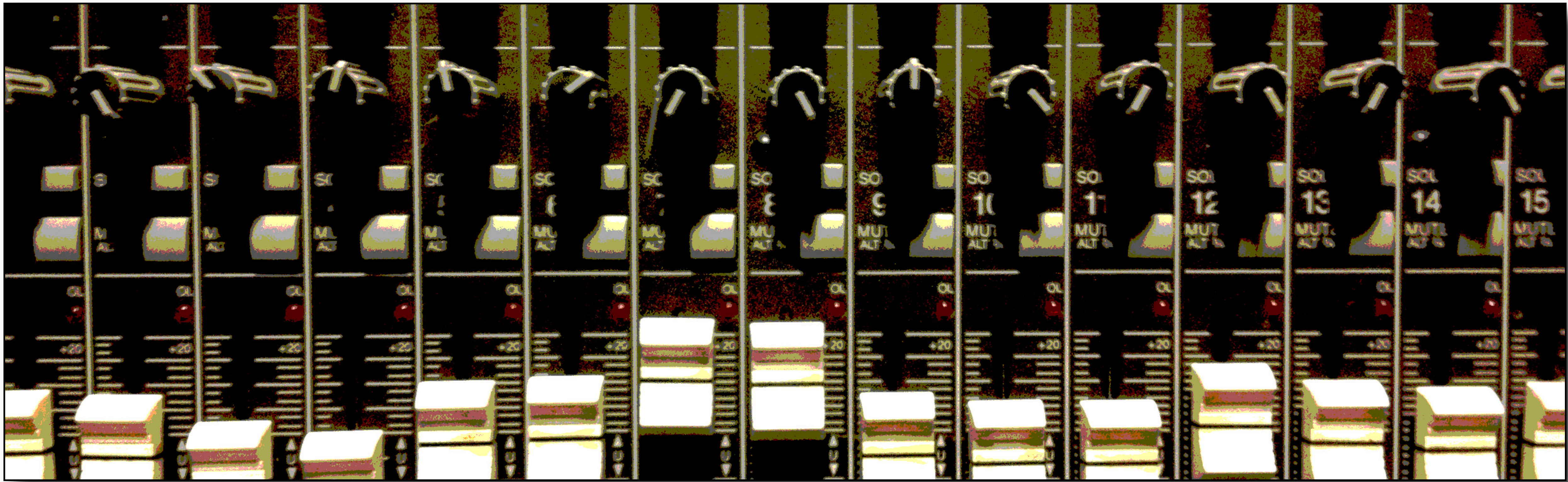
Suppose you have a stereo music file that has a drop out in one channel, or a microphone buzz for one second in one channel but not the other. What can you do?

A: Find a piece earlier in the music that is the same, (most music repeats: verse, verse, chorus, verse, chorus, etc.) Copy the waveform from an identical musical passage, (cutting on the nulls) and paste it over the broken piece. Put in short crossfades to smooth the transition. If you are good, and you will get there with practice, you won't need the crossfades.

B: Copy both channels to two mono channels. Copy the good portion of the audio from the bottom track, (in this example) and paste it over the broken section on the top track. Crossfades if needed. You can either copy and paste these back to one stereo track, or leave them as mono tracks. Up to you. Be sure your pans are full left and right if staying mono.



Sound Effects



In this section I have included links to my Youtube channel so that you can hear audio examples.

If you cannot link by clicking on the URL, Copy the link and paste it in your browser.

SOUND EFFECTS:

Sound effects can put your radio commercial somewhere other than in your dashboard, kitchen counter or earbuds. ***Theater of the mind.*** You can create a movie in your head with sound. And you don't have to spend a lot of money on CG.

Radio production used to be imaginative. Listen to Firesign Theater. Listen to Stan Freeberg. Or Ken Nordine's "Word Jazz".

And listen to some of the old radio shows from the 40's. There is a wealth of material that will expand your thinking into the use of sound effects, to put your commercial "somewhere"; to create a landscape, tell a story, and actually build a visual using only sounds.

Copy and paste the link, and enjoy.

LINK TO YOUTUBE "VIDEO":

"Theater of the Mind"

https://www.youtube.com/watch?v=ino2DCfOu38

SOUND EFFECT LIBRARIES.

Most sound effects and SFX libraries are sold on a perpetuity use license: but you cannot recopy or resell them. There are as many downloadable libraries as there are libraries that sell CD compilations, and some larger libraries sell external hard disks with the collection loaded. Each has their own search scheme, be it by heading (cartoon), category (animals, automobiles, etc) or alphabetical, these will allow you to find the sounds you are looking for quickly. Most SFX libraries have you register for downloads, and you can pay by credit card per sfx drop. Prices range from free to $5 per drop and more, depending on the library. Large collections can average in the $400 to $500 range. Most are stereo, but we will get into that in MIXING SFX.

You will gradually build a pretty good library, will make your own sounds by combining library sounds to create a new sound, and may even get a small digital recorder to gather your own sounds and ambiences.

CREATING A SCENE

PANNING SOUND EFFECTS:

You can create a bigger sound field by placing sound effect elements in different speakers.

Panning allows you to take a dog barking and place it in the distance in the right speaker, a distant lawnmower in the left, general outdoor ambience with birds chirping on both sides, and a distant motorcycle passing from left to right. You have created a scene where you can now focus on the two actors talking about firing up the brand new grill.

Or the general stadium sound with the organ panned right, the PA announcer just to the left, and the beer vendor center: Now you can have the guy explaining to the girl the fine points of the game until she suddenly heckles one of the players by name and proceeds to tell the guy why this player's stats will never be as good as another player and what was management thinking when they traded for him. Announcer finishes the spot, whatever it is for: beer, stadium, local team, contest, etc.

And you can move sound effects using automated pans, as in the motorcycle in the first example. Many programs will remember automation moves but only if they are armed to do so. Consult your manual.

But what if all the sound effects are stereo…how do you pan them?

A: move the right panner to the left so both panners are on the left.

Arm your automation. If you can grab both panners, move them both from the left to the right together to create the movement you want. If you can only do one at a time, do one then do another pass and try to track the other to match.

B: copy and paste the stereo channel to two mono channels. Erase one channel and pan the mono channel.

One thing to remember: some stations that broadcast in mono will sum both sides to create a mono signal; but some will take only one side of the stereo mix, so you will lose the sound effects that are in the other side. To avoid that, do not pan full hard left or right.

Here are some examples of spots we did using sound effects to create "visual" scenes:

Copy and paste the link in your browser.

https://www.youtube.com/watch?v=RyqgLI2ko5E

MAKING YOUR OWN SOUND EFFECTS:

Creating sound effects from stuff, may be the most fun you can have, non sexually, I mean.

In the old days of radio, they did sound effects live using any variety of items to create the illusion of what it was supposed to be. Foot-steps, creaky doors, thunder, horse hooves.....all were done using things that "sounded like" the suggested sound. A sheet of metal rattled does not sound like real thunder if you put them side by side, but as a suggested element of theater of the mind, you accept it without a second thought.

In that spirit, we had a "box o' junk" used for any number of creative sound effects: I cut a hole in the bottom of a Hinkley Schmidt water bottle, (yes I lost my deposit) spoke in the small end, and it made the perfect "bear roar" ...and a pretty good Darth Vader voice sound, too

(Disney used a large hurricane lamp glass to alter the pitch of the actor's voice and make it sound like a bear or lion. You spoke in the small end and recorded the sound coming out of the large end)

Grab a book or two on the old days of sound effect creation. Some great tips that can come in handy when you have a client who wants a squeeky bicycle wheel sound, or the sound of a porch collapsing: but it has to be "funny."

Here is another Youtube link to some sound effects we made up from the "junk box".

https://www.youtube.com/watch?v=S5m0n74h7Wk

CHANGING THE TONE OF THE ENTIRE SPOT WITH MUSIC OR SFX

Here is an example of how you can change the tone of the read with sound effects or music. You can make a straight read into something else. Same read in all 4 takes, just different sound effects and music changing the listener's perception of the announcer.

Here is a youtube link to "Changing the Read"

https://www.youtube.com/watch?v=qyKAvMlDP5E

And here is a nonsensical bit we did making fun of the hysteria surrounding the Mayan Calendar ending in 2012.

The premise is we send a fly with a wireless mic back in time to record what really happened in history.

Note the original music take on Flight of the Bumblebee using a "buzzy" synth lead, the sound effects and ambient sound to create the scene, and the music sting and "time travel sound" to wrap up the bit.

https://youtu.be/_4qvz-eOQsg

Oh, and while you're on the page, enjoy the "Bark Talk" PSA we did. Note the background sfx: dogs barking, and wind noise, geese. Wind noise adds that "golf audio" authenticity to outdoor scenes.

RECORDING YOUR OWN SOUND EFFECTS

Thanks to smart phones, and iPads, you can record pretty high quality sounds in the field if you use a higher quality mic plugged into your phone. There are also stand alone recorders, like the Zoom, that will record high quality files.

You can get mics for iOS phones for around $100, $150+, or you can get an interface so you can use studio grade mics....but remember, these sounds are background players, so unless you are doing sound effect/ sound design for a feature film, no need to go nuts here.

We still use ambiences (longer sound effects, like outdoor scenes) that were recorded on cassette.....so don't sweat needing 24bit/96khz. quality for your radio commercial.

Always record way more sound than you need, since all sorts of happy accidents happen. You are recording suburban outdoor ambience, getting the lawn mowers and the birds and then you hear some kids playing street baseball and you get that aluminum bat hitting the pavement. Gold.

Or a car comes by. Or a dog barks, or you hear a motorcycle in the distance. All sounds you can use parts of in a spot to paint a picture.

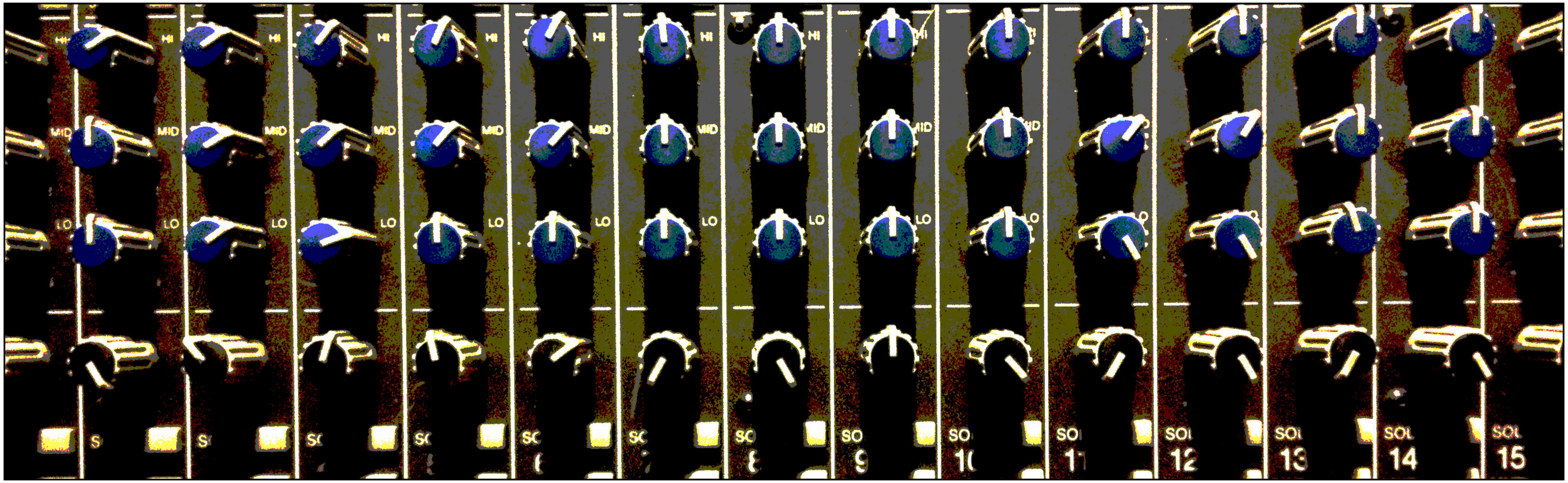
And record daily stuff: on the commuter train, in the DMV, the checkout aisle at the supermarket, an interior perspective in your car with starts, stops, turn signals, etc. We used these over and over in programs to put the listener somewhere instead of just voices coming from the void.

I had to hire a tow truck because I needed the sound of those huge chains rolling off of the deck and hitting the pavement. Cost $100 for the call because this particular sound effect did not exist in any library we checked. Worth it. Used it 6 times charging $25 each per sound effect drop.

And again: Charge your clients for your work!

They are getting a bargain over their old studio, so don't shortchange yourself!

Fixing and Mixing



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L.I.A.R TECHNIQUE

Remember, the music and sfx are there to support the voice. But how do you know if they are mixed too loud or too soft? It sounds fine on the monitors and even in the headphones, but you suspect your listening area is not as accurate as you would like.

My friend and audio guru, Ken Goerres came up with a strategy that works even in the best studios: L.I.A.R or Listen In Another Room. When you leave the room and listen from the hall, or down the hall, you get a more accurate reading of how everything sits in the mix. The mix you hear in another room is about as close to the balance you will hear when broadcast.

Oh, and one other piece of advice that was invaluable: "Never do a final mix right after you have tracked the string section". It will be string heavy in the mix. You just spent the last few hours critically listening to every nuance of the strings. You are finally in love with the best take/comp track. Chances are the first mix may feature the strings and obliterate the singer they were meant to support. Same goes for sound effects. Remember, they are supporting cast members, not the star.

Complimentary EQ

And after you have a mix you deem perfect, your client will inevitably say that the music is way too loud. If you lower it to what he thinks is right, it sounds like some annoying static in the background.

There is a way to preserve some of the level of the track and still have intelligibility of the voice: complimentary EQ. By lowering competing frequencies in the range of speech, you make the voice more intelligible.
Start by putting a dip at around 1K in the music track, you will have to experiment with how much you lower it and how wide a Q, and which frequency works best. This creates a space for the voice with less frequencies interfering; the voice rides on the music track instead of fighting it.

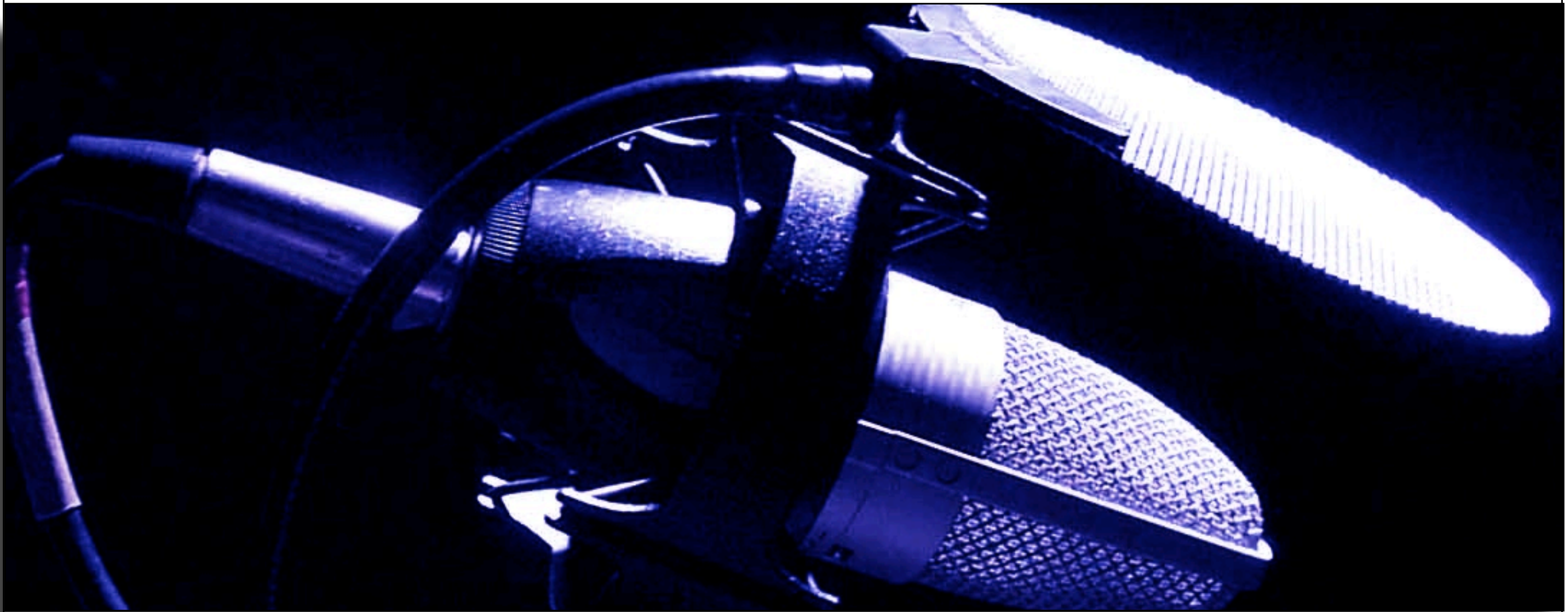
Lead instruments tend to be in the range of speech: guitars, brass, piano, etc. If you can find a place in the track that is primarily bass and drums, you can usually pump the track and still understand the voice.

Also, if you have an aural exciter in your plug-in arsenal, this is a good time to use it on the voice. It creates harmonics that "fatten-up" the voice in the frequencies that are associated with intelligibility. Or you can add a touch of eq on the voice that corresponds with what you took out of the music track. And a touch more in the 5K range. (experiment: these settings are just starter suggestions) But listen. If the voice sounds too brittle, or abrasive, back off and just lower the track. A harsh voice-over defeats the purpose.

And if you are using reverb on the voice: eliminate it, or pull it way back. Reverb tends to push the voice back into the music.

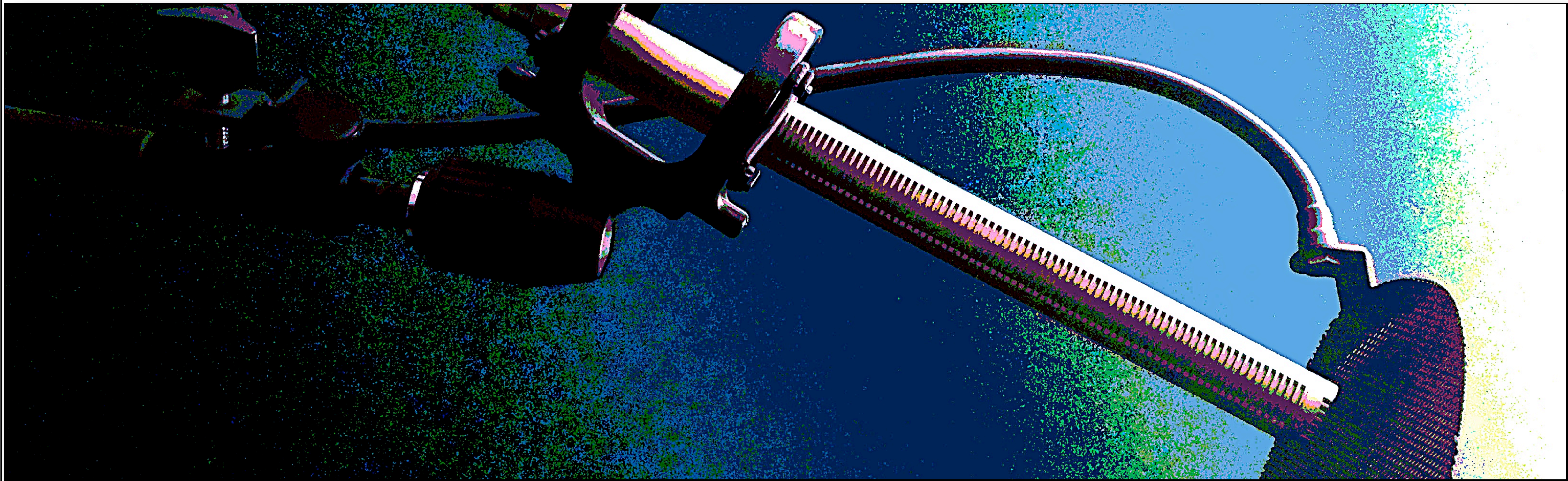
CHAPTER 7

Connecting



Connecting is a nuts and bolts chapter on phone patch, ISDN, source connect; to which file form is the best choice for mastering for specific end use, and delivery of files.

Connecting; Mastering; Delivery



PHONE PATCH

A phone patch allows the client/agency to hear you in the booth and allows you to listen to their direction/comments. There are commercial phone patch interfaces that allow you to hook into your landline and connect to your mixer/interface box so that they hear what is coming through the mic and you hear them through the headphones. They all work a bit differently depending on your set up. $40 to $250 +for radio station grade (which you don't need)

Or if your speakerphone is close enough to your monitor speakers so they can hear you, you can put a mic over the speakerphone running into channel 2 to record what they are saying on another channel. You can pan channel 2 to the right or left side, so you hear them in only one ear, but yourself in both. Recording their direction is also a good idea in case of a "misunderstanding" down the road.

Or use your cell: use one ear bud to hear them and hold the phone close enough so they hear you. Make sure you are not picking up bleed in the mic. If you have Bluetooth, you can try that too. But do a test run to see that you are not picking up extraneous bleed or interference.

Some talent are now using iPad and iPhone video via Skype in the booth using the mic on the device and headphones to hear clients. Experiment.

DIRECT CONNECTION:

ISDN was used to interconnect recording studios via high speed phone lines and codec boxes at both ends that compressed and uncompressed the signal to studio quality. It worked well in its day, you could have a voice over in Denver doing a two person spot with a talent in Richmond and it almost sounded like they were in the same room. I said almost. It is still used in some recording studios, but mostly it is used to connect promo/topical voices with television stations.

ISDN is rapidly becoming an obsolete protocol but is still being used by a majority of television stations because of Newton's law of motion: "A body using an antiquated technology will stay with that technology until affected by an irresistible force, ie, you can't get parts anymore."

It is expensive to use, many service areas are being eliminated, and it still sounded like it was coming through a phone line, which it was.

SOURCE CONNECT, ETC. There are programs that allow you to connect and lock Protools programs together and pass information in real time over DSL: you can record into your Protools and a connected studio is able to record into theirs at the same time, when it works. Most people swear by it, a few swear at it. There are levels of these programs from basic to deluxe with price points to match. And there are beta versions of new programs coming out as well. These need to be fiddled with to get them free of glitches. Check user groups to get tips and reviews before you chose a platform, or check with the studio you are likely to connect with to get their recommendation.

MASTERING/FILE FORMATS:

SHORT FORM:

Use .WAV or .AIFF for professional format high quality audio.

44.1 khz as default, but use 48khz for tv/video formats.

24bit depth if possible, 16bit as default. (standard CD bit rate)

mp3 is convenient, is not as high quality, but emails easily, and if it is just voice, the quality loss is insignificant. Use highest bitrate: 320kbps found somewhere on your mp3 menu.

PROFESSIONAL/ STANDARD FILE FORMATS.

Most masters, or bounces, are done in .WAV or .AIFF formats. These are large files that are full bandwidth audio with no compression of data. 44.1khz sampling rate is the standard for CD audio, 48khz is higher quality and standard television/ audio for video format. (most video is clocked to 48khz) There is a school of thought that if you track and mix at a higher sampling rate (88.2khz or 96khz) and then convert down to 44.1/48, that you have better sound since you started with a denser and more robust sample. This is an argument for those with better ears than mine. I have discerned the difference in only a few demonstrations in optimum listening conditions, however the difference was apparent. If you are mixing for theatrical release, highest sampling rate without question.

Local cable tv? 48khz.

There are other files formats that may be included as output/mastering/exporting options, such as Quicktime, AV, Raw, Sound Designer II, Paris, etc. These may work better for some applications, but .WAV and .AIFF are industry standards.

BIT DEPTH

Most systems record at 24bit depth rate, but some only export at 16 (standard CD bit rate). Rule of thumb is if you record at 24bit, you can output/ master/bounce at 24 bit, then by all means use it unless specified by your client. Most systems will convert if they only accept 16bit rate. 8bit is used only for non professional equipment like automated telephone prompt systems.

MP3

Even though it is a lossy format, most files, especially for radio, are now mastered as mp3 for their ease of delivery, primarily through email. You can either use the standard mp3 default of 128kbps, or master at a higher bit rate, which increases quality, but also increases file size. Most radio and television stations could care less. Be aware that an mp3 is a compression scheme that takes sonic information out of the file in tiny little slices.

It is amazing that you can take so much out and still have a decent sounding master. However, there are many who can hear the difference and rail at the acceptance of low quality audio. Usually a studio will boost a bit in the low and high frequencies to compensate for loss. With all of signal modification inherent in broadcast, the quality difference argument is merely an academic point, at best.

DELIVERY:

Large files need to be uploaded to an FTP site, or sent via a large file delivery service, like dropbox, etc.

Mp3's can be emailed.

FILE TRANSFER

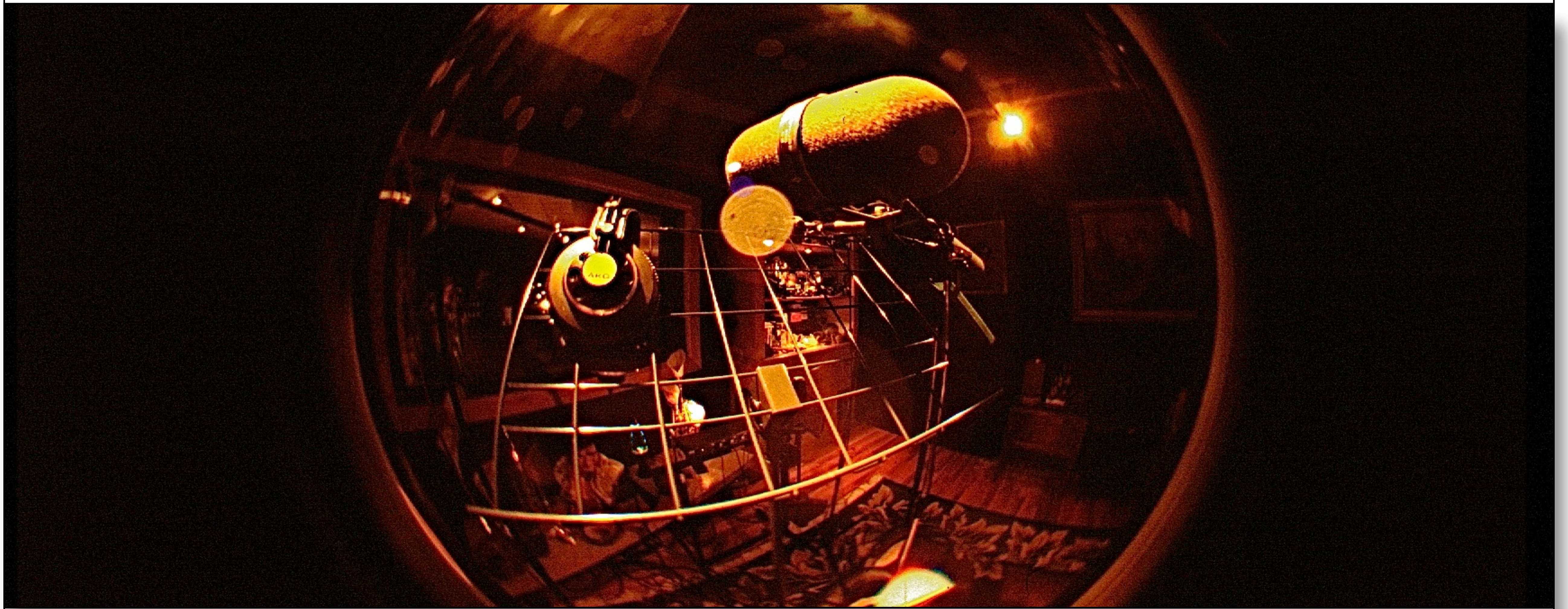
If you master to mp3, chances are you can just email the files, but if you are sending higher quality .WAV or .AIFF, the files are too big to email, so you will need to upload to an FTP folder either on your client's site, or on your website. First, you will need to get an "FTP client", a program that allows you to connect with FTP sites (free to just under $100 for ultra deluxe…but most free versions to $20 versions are all you will ever need) You will have to set up an FTP folder for your clients, and most hosting sites tech help will set this up for you in a few minutes. You will need to give your clients the FTP info and password to download, and likewise, they will need to give you that info if you are uploading to theirs.

Or you can use a large file transfer company, all have various plans depending on the size and amount of files you transfer per month. Range from free to $15 a month for average user, to $100 a month for Business/Pro users.

There are other protocols coming on line, so check with the client/studio/edit house to see what they prefer using.

CHAPTER 8

Sound Behavior



This is a reprint from the original Home Studio Handbook, hence the different type face. I have made some updates to the original text. This chapter will give you some background on the basics of how sound works in a room, some tips on eliminating or lowering common noises, and ways to improve the sound of your room by correcting problem areas to make your space sound better.

Sound Behavior/ Dynamics

Editorial

Your recording space is as important as your computer, your microphone, your monitors, and your recording program.

Some of the strategies in this and the following chapter will go from improving your sound with a very small budget, to creating a space that is close to what you would find in most pro studios. (And a few inexpensive tricks that will work in a pinch.)

But be prepared to spend time, if not money, tuning your space to make it sound as good as possible. It is trial and error, and you should always upgrade as funds allow.

And as your ears become more educated, you will find new things to fix and improve.

The room you will be recording in has inherent problems that you will need to fix to make a better recording environment.

Professional studios employ audio physicist/ designers using state of the art tools to create a space that conforms to the needs of the studio.

Some are live rooms (with pleasing echoes or natural reverberation) some have complete damping (dead rooms), some have the ability to change between the two, and some split the difference or have zones that are both dead and live.

For voice recording, you generally want a recording space that is free of reflections and any ambient sound.

Room sound:

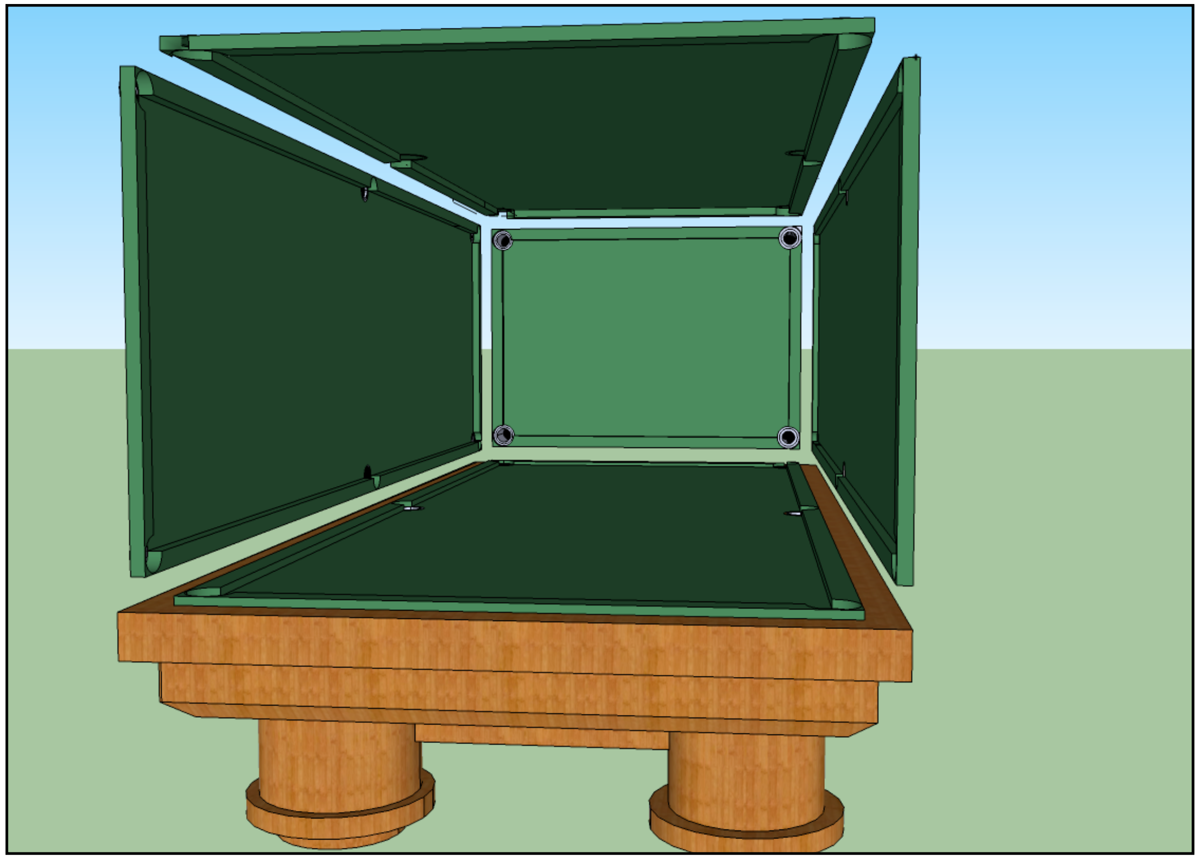
Picture a room as a pool table with the walls being the cushions. As a sound hits a wall, it bounces off just as a ball would off a pool table cushion, and hits another wall and so on.

And just like a pool ball, the more it bounces, the less energy it has. Only this pool table has 3 dimensions including the walls, the floor and the ceiling.

Sound emanates from a point and scatters in all directions. It bounces off hard surfaces (walls and windows) and is absorbed by soft surfaces such as carpeting and padded furniture.

Irregularly shaped objects scatter the sound and reduce the pressure of the sound wave (volume) such as bookshelves lined with a variety of objects.

Corners build up low frequency energy or bass rise, so where walls meet, where walls meet ceilings, these are problem areas. And parallel walls make for a perfect "pool table". Where the sound flected, you get echoes, or an unpleasant "room sound"



There are plenty of ways of treating a room from professional to make-shift that will yield results to make the recording environment more pleasing to the ear.

The first step is to find the problem areas. By clapping your hands, you create a loud sound wave that can help you pinpoint areas that need treatment. If you hear a flutter echo (a quick series of echoes that sound like someone rolling their "r's" and rises in pitch and speed) you are standing in a problem area.

Or record your voice in several areas of the room at least 3 feet from the microphone. This will also give you an idea of how much treatment your room needs. There are plenty of ways of treating a room from professional to make-shift that will yield results to make the recording environment more pleasing to the ear.

Lowering the noise floor.

As you lower the amount of room sound, you will become aware of other noises that you hadn't heard before, or were masked by the primary noise.

For instance, when you turn off the air conditioning/furnace fan, you may become aware of the wall clock ticking, when you eliminate that, you hear the computer fan; when you lower that, you may hear the rumble of traffic noise outside, and so on.

A professional studio will spend tens of thousands of dollars eliminating these noises permanently so that they are never an issue. Most home studios or location recordings can do work-arounds to create a good recording environment.

Identifying Noises.

The first step in making a quiet place to record is to lower the amount of ambient noise in the room. This section outlines some of the typical noises you find in an untreated room, and how to minimize them.

Trouble spots:

Near vents, especially vents that have grills: a grill creates turbulence, which creates noise, either a hissing or whooshing sound. The easiest fix is to remove the grill, which will lower the noise significantly. Try to put a mic as far away from a ceiling vent as possible. If there is not much choice, put the back of the mic pointing towards the vent, since most mics have a rear-rejection pattern that does not pick up sound from behind. If you have control over the HVAC, turn it off during recording.

Near windows. If there are blinds and draperies, close both. If you have a heavy, old-style moving blanket, comforter, or other thick blanket and can put it over the curtain rod, that will attenuate mid to high frequency noises. If you can put a bit of an airspace between the moving blanket and the windows, all the better. Low frequency road rumble can be attenuated after recording by rolling off any frequencies below 100k on the equalizer, or if the microphone has a high pass filter, (bass cut filter) that will eliminate some of the low frequencies at the microphone.

Floor: If you are actually getting vibration through the floor, putting the mic stand on foam padding will decouple it from the source of vibration. You should always use a suspension-type mic holder, also. If you have a hard surface floor, a large throw rug will eliminate slap echo between the floor and ceiling.

If you are in a multi-unit building, sound can be transmitted through the floor slab from several units, even several floors away. Thick carpeting can attenuate most sounds, but may not eliminate them completely. If you hear a noise during recording, always take a "safety" take (usually re-record the entire sentence) in case you need to edit that in to replace the version flawed by the noise.

Lighting: Florescent lighting, and lights on dimmers are to be avoided if possible. They produce either an audible buzz, or create an electrical interference that is picked up by the microphone cables, or through the AC to the computer/recorder, (if the AC is on the same circuit as dimmers) always make sure the dimmers are not engaged.

Power cables: Try to avoid placing AC power cables next to microphone cables. If the shielding in a mic cable is broken, it can pick up a 60cycle hum from an AC line. Try not to have them lay across each other.

Isolating a source of noise:

A good strategy to find the source of a background noise is to put on headphones, turn up the gain, and point the mic around the room until the source of noise is discovered. Sometimes there can be electrical interference, such as the interface box being placed too close to the hard drive. Or sometimes gear interacts poorly when placed next another piece. (internal power supplies may create interference with each other if placed in close proximity) Or equipment vibration, such as the computer hard drive is producing a vibration that is transmitting through the desktop and being picked up by the preamp or mic.

Or RF (radio frequency) that may be getting picked up by a mic cable with a bad shield.

Try moving pieces of equipment 90 degrees to see if that makes a difference. Sometimes the orientation of different pieces of equipment to another other can make a difference in how they interact.

And you could be getting interference from your desktop speakers: they have magnets which move the transducers, and that could be reacting with your mic or other gear, so try moving them away to see if that makes a difference. (Many manufacturers make shielded speakers that are supposed to suppress any interference.)

Grounding:

Grounding may be another consideration. Always use 3 pin power cords. Safety first. Most home studios do not contain racks of hardware gear like pro studios, so "star grounding" (accepted studio protocol) is generally not an issue. However, if you are getting a 60 cycle hum, try isolating your gear by separating the chassis from one another. If you have lots of hardware rack gear, check into pro-studio grounding solutions for different strategies.

Power Strips:

I would recommend getting a pro-sound "power conditioner", which is a pro-level power strip with 60 cycle/RF isolation, and surge protection. The strips you get at the hardware store do nothing to cancel noise, and are of questionable surge protection. These generally run around $100 or so, have multiple outlets, and usually have a master power switch.

Isolation/ Absorption/ Diffusion

Isolation

Isolation is blocking all sounds coming into a space, and the only way to block all sounds is with mass. Most professional studios are “rooms within rooms” with a room completely decoupled from the room it is within, with all air handling, control room windows, and doors “air lock” style with double doors, double/triple pane windows, and insulated duct work to shut out any outside noises. Once constructed, the only concern is with recording the performance, since outside noises are never an issue. However, this is extremely expensive and labor intensive. And in today’s marketplace, although still desirable, may not be completely necessary for voice recording.

Basements are frequently used for studios since the foundation is surrounded by ground providing the mass needed to block sound. However, the mechanicals of a house are in that same basement: furnaces, ductwork, water pipes, hot water heater, plumbing, and sump pumps are all major noise makers. Also, another source of noise is the floor above: washing machines, dryers and dishwashers transmit noise, as will people walking, since most floors squeak, especially when you are recording.

If you need to isolate an area, building walls and ceilings out of several layers of drywall is a good start. One of the best ways to build a wall is a 2X4 frame with drywall on the outer surface, fiberglass inside, and 1 or 2 layers of drywall on the inner surface. Plan ahead for electrical and ventilation. Put HVAC ductwork near the floor behind the mic. Electrical has to be put in before the inner drywall is hung. Tape and mud all seams and stagger the seams on the second layer of drywall. Seal it as if you were making an aquarium. If you have an air leak, you have a sound leak. Do the same for the ceiling, outside and front wall. Door should be sold core, with weather stripping put around the door jamb to create an air seal when it is shut.

These treatments apply equally to iso booths and control rooms. If your listening room is inaccurate, your mix will be inaccurate. But there are differences in how much absorption/diffusion you need for each type of room. Generally, you want more diffusion in the mix/control room which tends to be larger than a typical iso booth.

Absorption

Removing reflections can be easily done with either a fiberglass absorber, commercially available wedge tiles, (acoustic foam) or absorber panels. Please don’t use egg cartons: they do nothing to absorb sound, are a fire hazard, and look bush. They do however, work very well in the recycling bin.

Some commercially available sound panels have patterns, are visually pleasing and relatively affordable. Wedge tiles generally absorb only a narrow frequency band width (higher frequency) and are more costly. In most cases, you will also need bass traps (low frequency) for corners.

Diffusion:

Diffusion is scattering the sound and reducing its energy to create a pleasing environment. Diffusion panels are available at most pro sound stores and work very well. However, they are costly. But there are options for doing it yourself.

An open backed bookcase (against an absorbent wall treatment) with lots of objects scatters sound.

An irregularly surfaced wall (lots of pictures, plaques, shelves etc.) can also keep a room from becoming too dead. And space some of the plaques away from the wall using a 1" or 2" spacer.

Or a ceiling with commercially made or homemade diffuser panels can also scatter the sound and prevent slap echoes. (Diffuser panels are available from a variety of manufacturers, but are rather expensive, are not that difficult to approximate, and the results will be sonically very close.)

Floor lamps, end tables, hard surface furnishings, table lamps, etc all can diffuse sound energy in a room. And once again, placement is trail by error.

Old wooden pop carriers, shadow boxes, model airplanes hanging from fishing line, or other reflective objects can have the same diffusion effect. How you scatter and diffuse the sound is up to your decorating sensibilities. One studio downstate had a back wall of old hubcaps and antique auto parts. Looked cool, worked great.

All of these things fall under the "tweaks" category and make only a small percentage difference in the final recording quality. However, it does make a room more pleasing to work in for long periods, makes for more accurate monitoring, and is less fatiguing

An example of absorption and diffusion: the walls are covered with fiberglass absorbers (outlined in the next section) as well as the carpet, the objects on the table and book case (in the background) scatter the sound keeping the room from becoming too dead.

Also, the wood surfaces and the glass also diffuse sound, however, with a large, flat surface, like the window, it can act to reflect sound; in this studio, the glass was angled down to prevent direct reflected sound back to the mic position.

In a home studio, computer screens, and music stands can reflect sound back to the mic, so angle your screens, and cover the music stands with absorbent material.

CHAPTER 9

Making a Booth

This chapter shows how to construct a booth from basic to professional; how to build inexpensive absorber panels, sound blocking flats, to building a permanent installation pro-level booth.   And encourages you to learn how to use tools to do-it-yourself.

Audio Booth

AUDIO BOOTH

The purpose of any booth is to isolate the announcer from any outside sound sources; to create a neutral sound with no ambient room sound.

When there is a room sound, reverberation, flutter echo, or ambient noise, (like air conditioning noise) it masks the clarity of the voice, and makes the listener strain to understand the speaker.

The ideal is to find a space to record that eliminates the ambient room sound altogether. This is not always possible, but this section outlines some strategies to minimize these problems.

Isolation and sound treatment are the sum of all the parts. Everything you do to lower the amount of background noise and room sound improves the recording environment, and quality of your recording.

The booth you already have:

A walk-in closet is a pretty good make-shift recording area.
The clothes absorb high frequencies, if there is a carpet on the floor that usually stops a flutter echo between the floor and the ceiling, and outside of a lower mid-range frequency bump, (which can be eq'd out or minimized) a walk in closet is a quick solution for a quiet recording area. Once again, florescent lights should be eliminated, and wear headphones so you can find a spot where the mic placement sounds best.

Quickbooth: basement basic:

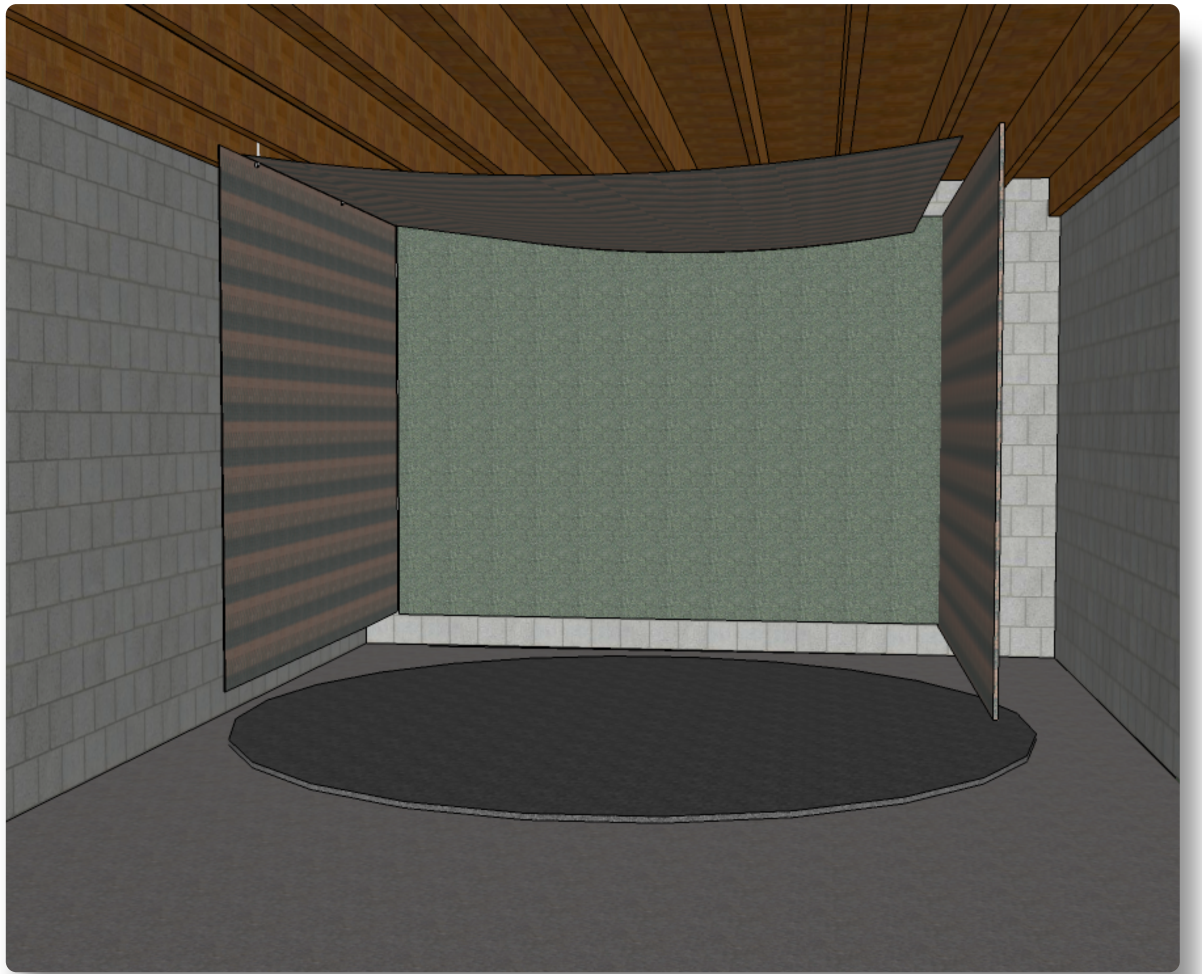
Heavy moving blankets, comforters, carpeting, cushions, and overstuffed furniture can be assembled into a make-shift booth. A local semi-pro studio had an area in its studio that looked like a children's "fort" with old style heavy moving blankets hanging from the rafters by light duty chains creating a "room". It worked. Not visually the most pleasing, but sonically very workable. They had hooks on the chains and eyelets on the blankets so that they could easily take down the blankets if needed.

If you make a setup like this, use industrial sound barrier blankets, (available through acoustic treatment companies) which are better than today's light duty moving blankets which have little sound absorption qualities. You can double them up for even more absorption.

You can also use sale carpeting or carpet remnants that are usually more than enough for a quickbooth, and very cheap. Be sure to hang carpeting about a foot from the wall, the airspace helps deaden the sound. Hang carpet/moving blanket above, and carpeting on the floor.

This is very rudimentary, but inexpensive and gets you on the learning curve. If you have an overstuffed chair and like to record sitting down, this will also absorb more of the reflections. Put in a nice reading lamp, a coffee table for the computer/tablet/recording program, and an end table, (so your martini is close at hand) and you are ready to podcast.

Many old recording studios employed heavy draperies on tracks that they could pull back to create different sized recording spaces. You can construct a similar "quiet zone" by mounting two parallel tracks into the ceiling (approximately 1 foot apart to create an airspace) and use heavy drapes to create a zone that will greatly eliminate the room from the recording. A heavy throw rug will also help eliminate reflection between the floor and the ceiling.
If you create a circular booth, you will find it a very flexible configuration, especially for voice. (Seemed to work for the Wizard of Oz who incidentally, had a pretty good studio for his day.)



Hang carpeting/moving blankets from chains screwed into the rafters. Use hooks and eyelets so you can take them down easily. Leave an air-space between the wall and the absorbers.

(Front wall left off for illustration purposes.)

HOW TO: 4'X4' sound panels:

Double-knit/frame/fiberglass treatment:

This treatment has a much lower cost factor than commercial tiles and panels, absorbs more frequencies than wedge tiles, and is tunable as to the amount of absorption. It is also visually pleasing, creating a flat panel like a picture frame, or a flat wall as though there was no treatment.

A 4 ft.x 4ft frame is constructed using 2"X2"s with the corners joined using 3" drywall or deck screws through pre-drilled pilot holes. (Be sure to get 2X2's that are straight.) This makes for a solid frame. 4'X4' panels are an easy size to handle and arrange on a wall. (Or hang on the wall with picture wire, or as a "cloud" from a ceiling joist/stud) Double-knit polyester material is available at virtually every fabric store and available in a variety of colors. Be sure to get the material that stretches.

Staple the fabric to the backside of the frame, stretching it around and taught to the opposite side and staple it to the back of the frame.

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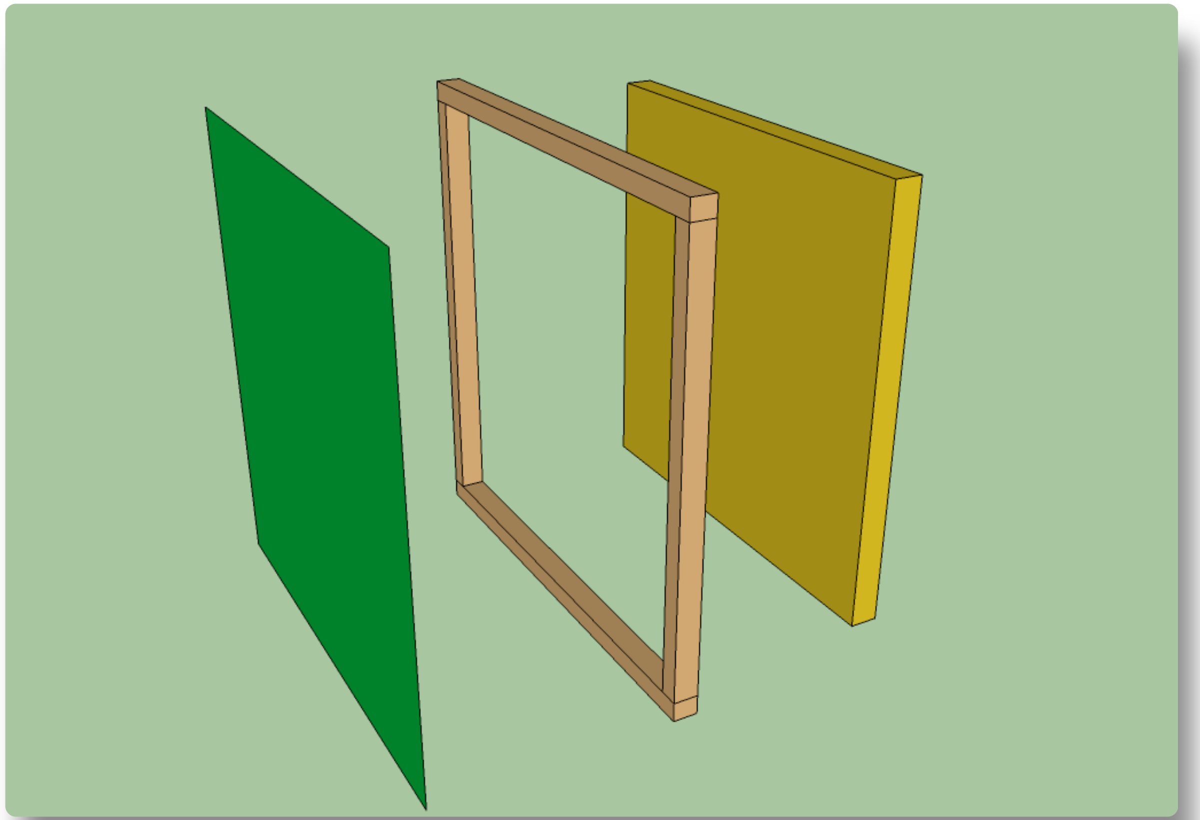


Do it yourself sound panels

LEGEND: in all drawings,
GREEN: polyester/ fabric covering
BROWN: wood frame
YELLOW: rigid fiberglass (Owens Corning 703 or equivalent)
PINK: (not shown in this drawing) high R-Value "big box" fiberglass

Tools needed: saw, drill with counter sink bit, driver bit (to screw in deck screws using the drill), stapler.

Fiberglass:

Owens Corning 703 2" fiberglass (or equivalent: there are several manufacturers) is the sound absorbing material. It is a sold panel of pressed fiberglass that absorbs a broad band of frequencies. It comes in 2 x 4 sheets. Wear long sleeves and gloves when cutting it, a box cutter works just fine. Place the frame face down putting the fiberglass in.

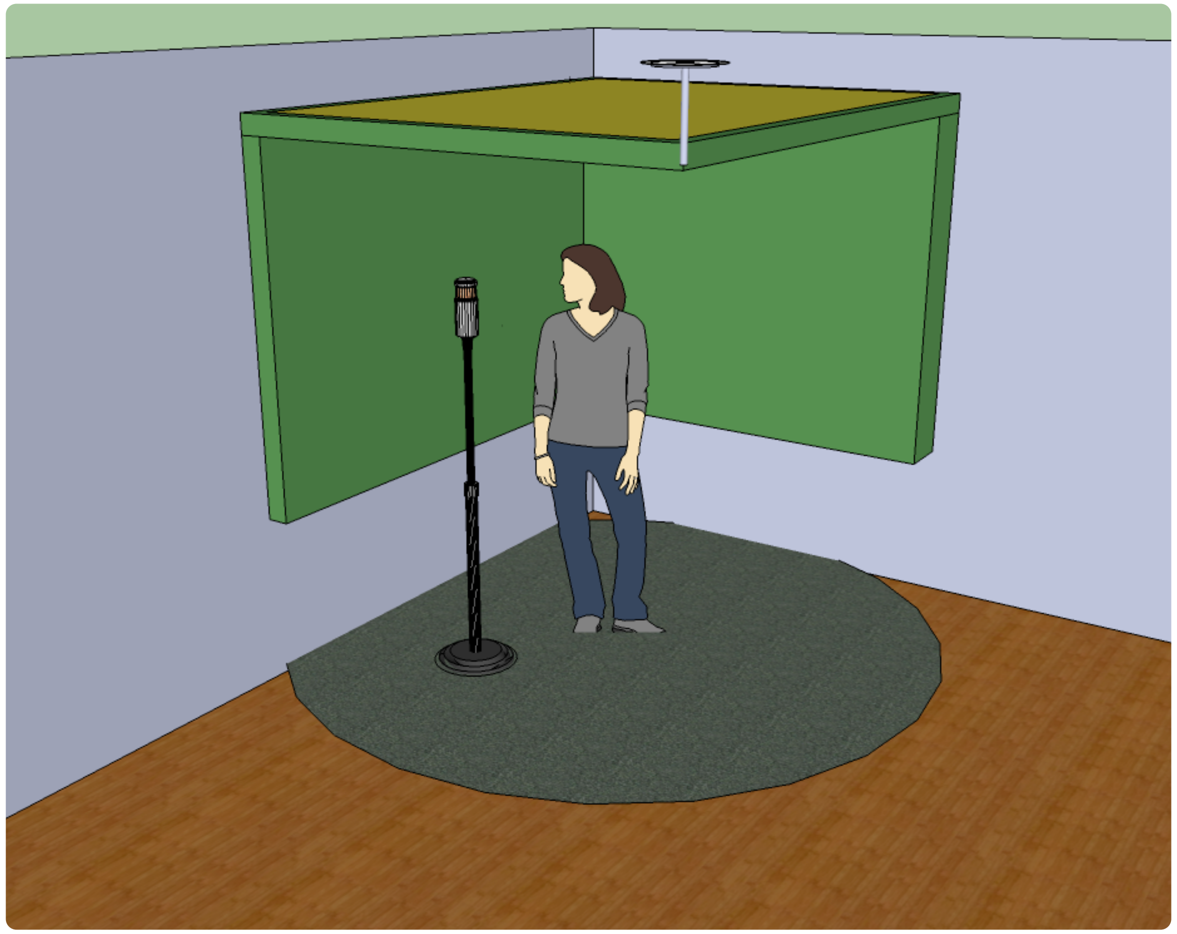
Use duct tape or equivalent to hold the fiberglass in the frame. (This makes it easier while mounting on the wall) Use the same drywall screws to attach to the wall. Use a stud finder for a secure mount.

Or put a piece of fabric, or cardboard over the back side to hold in the fiberglass if you are using picture wire to hang the panels. With picture wire, you can easily change the positioning with little damage to the walls until you find the optimal placement.

You can place them covering a wall leaving the opposite wall untreated, stagger them, or treat all the walls.

TIP: Look at on line photos of local recording studios to get some ideas. Note placement of absorbers, diffusors, furniture, mix position, booth orientation/location. The equipment may be far more exotic than yours, but the basic principles of absorbers, diffusors, and room arrangement apply.   And compare the setup to your room dimensions.

Or pay a visit to a local studio, ask questions, and make note how they built their booth/studios. Many of the construction details can be scaled down and utilized in the design of your studio.



Quickbooth 2: CORNER BOOTH

Wedge tiles, or absorbers mounted in a corner with the mic pointing into the treated area can remove some of the room sound.

Hang an absorber "cloud" over the area.   If you just need an occasional voice over/blog post, this is an adequate solution.

The treated area should be a minimum of 4ft, high and 4 ft wide on each wall, approx. The larger the area, the more useable recording space you create.

There are commercially available wall mounted cabinets with cupboard type doors with absorbers on the back of the cabinet and inside of the doors. And a reflexion filter (mic stand mounted baffle that blocks sound coming from the back of the mic) can take a bit more room sound out as well.

Rest the cloud on the side panels and use a chain to support the front corner. If you hang the side panels via picture wire, you will need to hang the cloud independently.

PVC stucture

If you are in an apartment and can't hang anything, you can build a free-standing frame out of 2" PVC pipe, (using "T's" 90º bends, etc. and PVC glue to assemble) and drape industrial sound blankets/ old moving blankets, even carpeting over it. Push it in a corner for even more isolation. Or use the frame to hang 4X4 absorber panels.

If you are setting up a studio in an unfinished basement, hanging carpet/absorbers from the rafters, building a PVC fort in a quiet corner may work out very well.

Quickbooth 4- mini isolator

There are several commercially available mic boxes that are portable enclosures with sound proofing inside. (generally sound wedge tiles) There are boxes with one end open, and some with 3 sides and a top that sit on a desk over a computer monitor and microphone. These isolate the area around the mic and lower room sound; if the mic is worked very closely, can further attenuate the room.

Care has to be taken to avoid a "boxy" sound, and to avoid computer noise/interference if working on a laptop. These generally run in the $100 to $400 range, and are perfect for demo/auditions/pod casting/etc. applications; and have been used by experienced voice talent on promos and spots when the talent does not have access to a professional recording environment.

The "eyeball" is ok for quiet spaces, but will not cut ambient noise that much; it will lower "room sound".

Quickbooth 5- the car in your garage.

I know of one station promo talent who has a booth in her mini-van that works very well for late breaking news topicals; the editors at the station claim they can't tell the difference between her car and her regular booth, although I suspect her regular booth sounds like a van. (On another note, I have always been leery of vans that have soundproofing.)

Other talent report very good results recording in their car in the garage (for isolation) with a few blankets and pillows to cut down on the boxy sound you get in a tight space like a car. This is probably the least acceptable option, however, if you are stuck and need a space to record during a family reunion, this is workable.

Quickbooth 6- in your hotel:

You are on vacation and get a call for an audition that you know you will win: build a fort out of the mattress, cushions from the chairs, blankets etc. as far away from any noise sources as possible.

Turn off the HVAC, and wait until no one is showering in the room next door. You may wind up backed into a closet working with a flashlight, but if it works, you get the gig and no one will be the wiser.

You can create a very usable space using the previous suggestions, or even in combination, that will limit and lessen the amount of room sound in your recordings. But if you are still frustrated with your sound, here are some workable booth plans. Many of the designs that follow use increasingly more aggressive sound isolation techniques. You may need to use some or all of the isolation techniques to improve the sound of your room.

CONTROL ROOM CONFIGURATION

This is a standard set up found in many control rooms; absorbers hung in a staggered configuration; bass traps in front corners, and a diffuser hung on the ceiling over the listening position.

The back wall will usually have absorbers and diffusion panels. Where the door to the room is located will dictate exact placement. Windows may also change configuration. Furniture, carpet, equipment, bookcases, etc, will also change the sound of the room.

Your control room acoustics are as important as your booth when you are mixing program material. If your control room is inaccurate, then your mixes will be inaccurate. If your room is too absorbent, your monitors will tell you that you need to add high end, making your mixes sound too bright; if your room accentuates low end, you will remove it making your mixes sound thin.

There are a number of companies that will give you a template for your room; you supply them with all the dimensions, they will send back a suggested blueprint of which panels to buy, and where to hang them.

There are also many MANY instructional videos on youtube regarding configurations and suggesting ways of treating a room.

This, of course, is when you are getting more serious about creating a truly professional control room/ booth combination.

Semi permanent small booth: Assembly:

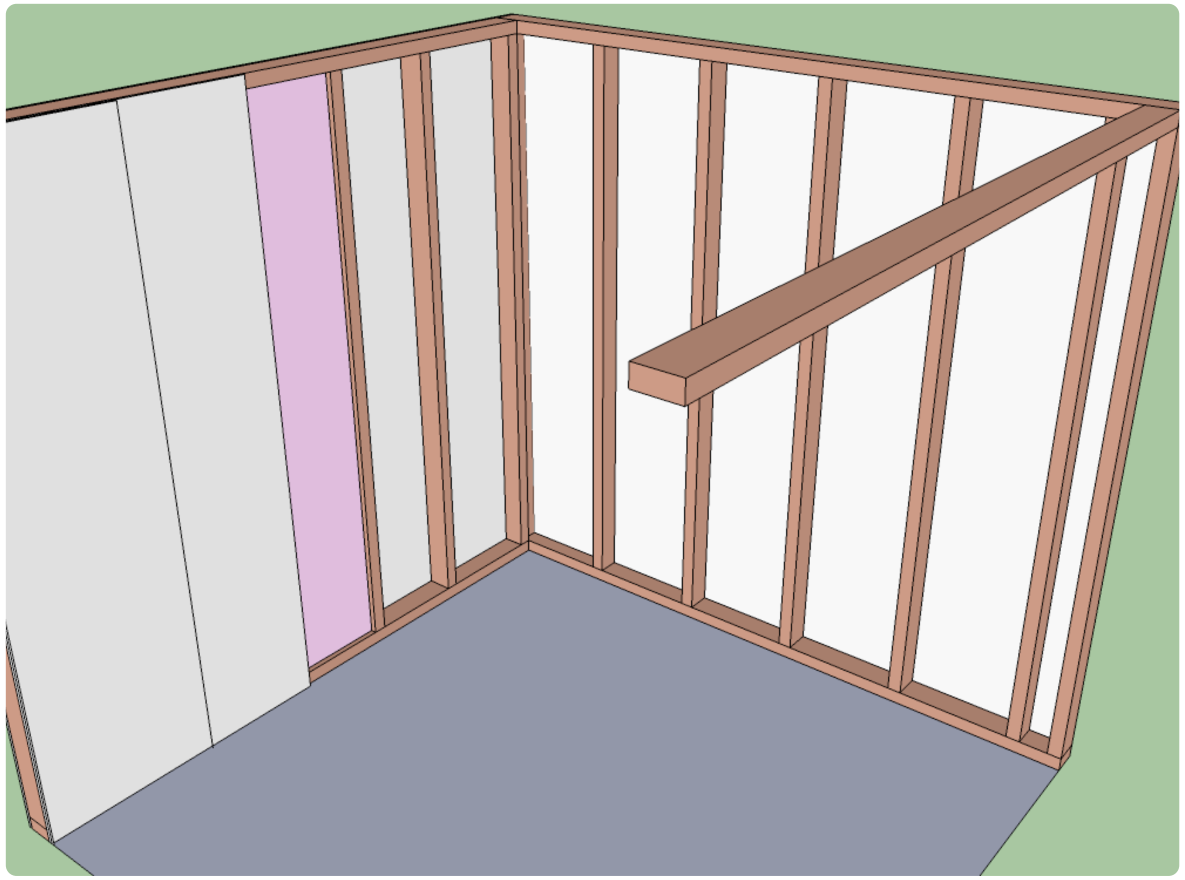
These designs assume you have adequate knowledge of power tool use and construction techniques; if you have any doubts, hire someone who does, they may even have some better, more efficient construction techniques.

Or ask your Dad, or your Uncle Al, (who will probably be loaning you tools anyhow.) And heed the advice they give you, that all power tools are like snakes, just waiting for a moment of inattention to bite you.   A trip to the emergency room could cost you more than hiring a professional to build it for you while you watched reruns of Castle.

Rough Framing a semi permanent booth

Build the frame to the exact dimensions you want for the booth, and assemble the rough frame using deck screws. This adds structural strength and makes for an easier build. Just as with the permanent structure booths in the following examples, rough frame first, outer drywall, then fiberglass, then inner drywall; create a roof frame, (like the wall structures) attach to the wall frames, fiberglass and drywall. then absorber panels. This will still be a semi permanent free standing structure that will not be permanently attached to the room.

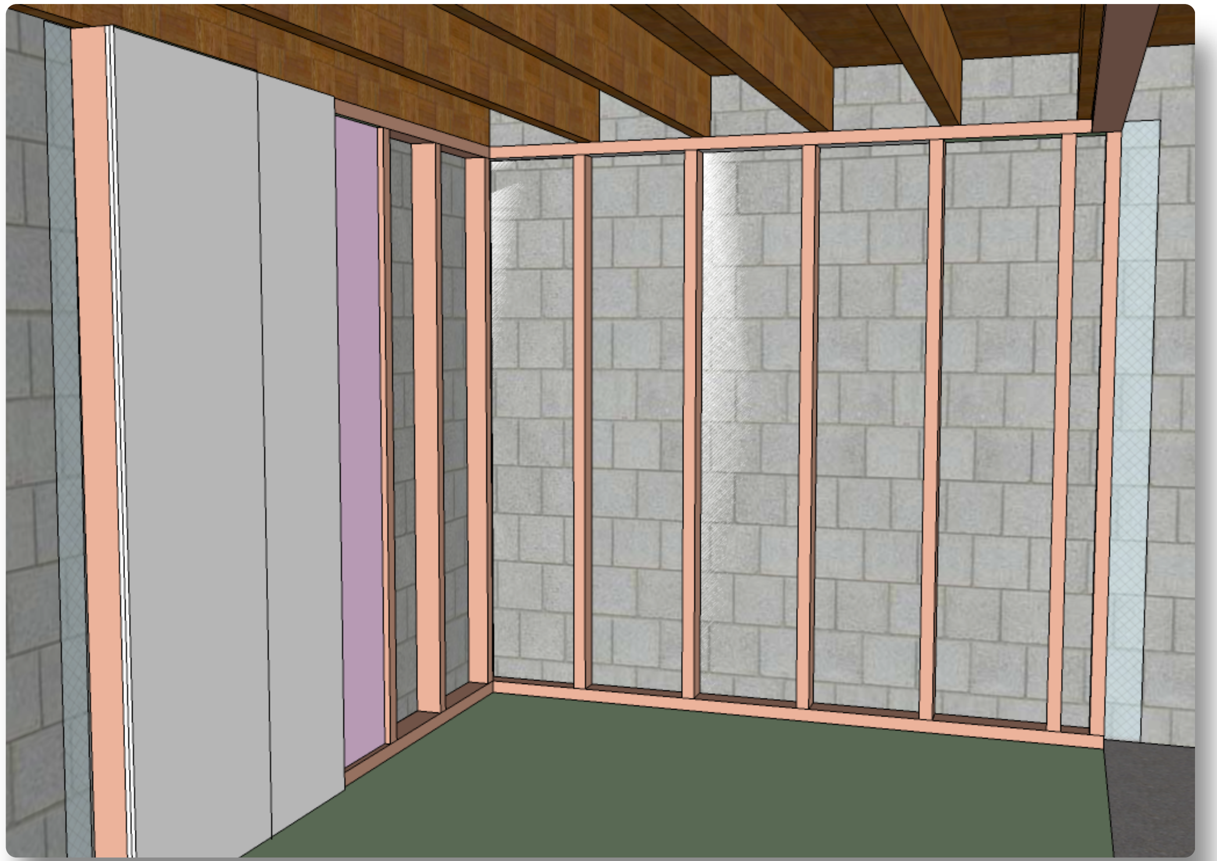
Rough Framed free standing



(studs are standard 16" on center)

Permanent construction booth:

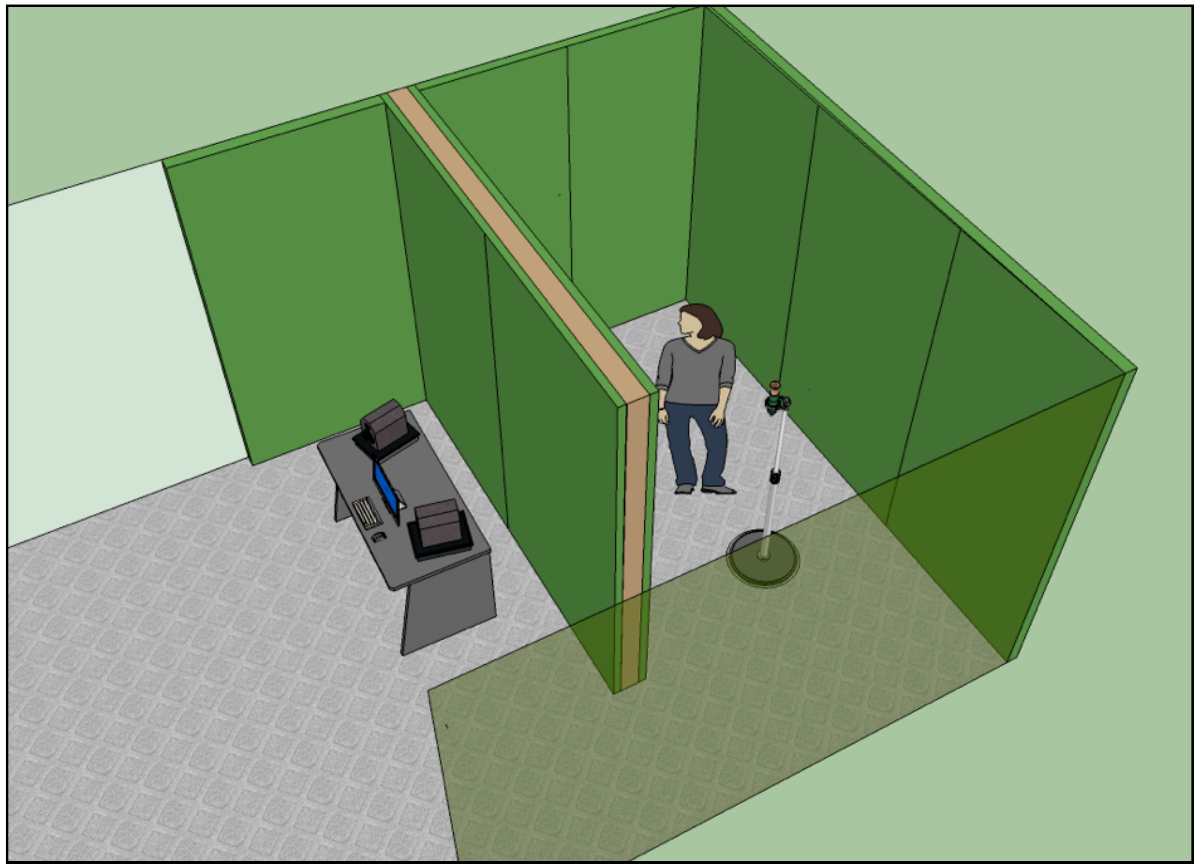
Permanent Booth framing: unfinished basement



If you are installing in an unfinished basement, use vapor barrier against concrete. Then fiberglass between studs, drywall and tape. Stuff High R-value fiberglass between the joists, and screw drywall ceiling to the joists. Seal all joints. If you need access to the mechanicals, use a dropped ceiling, add more fiberglass between the panels (tiles) and the joists; use a rough surface finish ceiling tile, (to help diffuse reflections) and also use diffusors panels/grids.

Be sure to check building codes/ordinances regarding steel stud vs. 2 X 4 construction. For illustration purposes I used standard 2 X 4's. Assume pink fiberglass between studs, 2 layers of drywall on inner surface, absorber panels over drywall.

A "switchback" configuration does not need a door

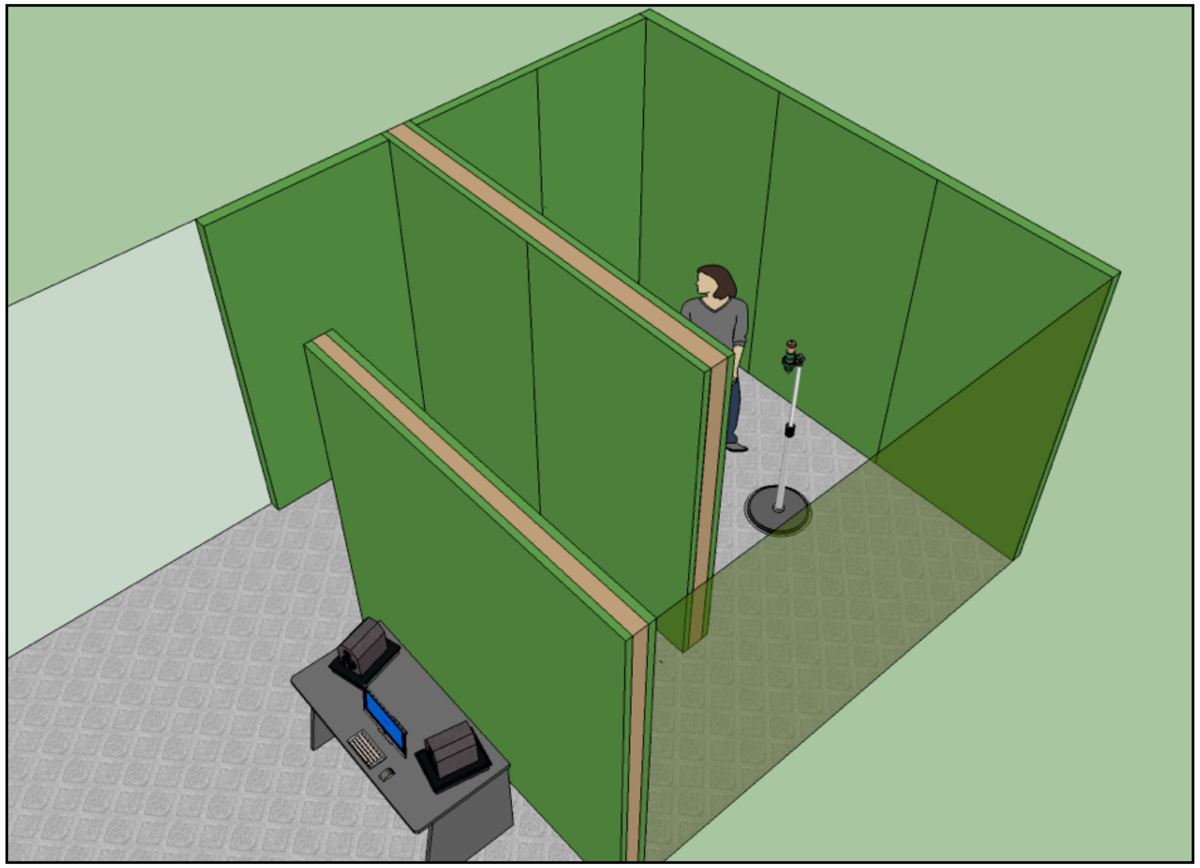


This design also does not need an in-booth ventilation system (dry wall, studs implied; transparent wall: absorbers)

SWITCHBACK DESIGN

You can make a booth without a door using a switchback, which will not require dealing with ventilation issues. A double switchback is nearly as sound attenuating as a studio door system.

Double switchback design



When designing, be sure to make allowances for electrical and microphone/headphone wiring. 2 PVC pipes (separate the AC from the mic cables) running though the wall make good cable runs, and then stuff with fiberglass for sound proofing. If you are going to mount an AC outlet box inside the booth, be sure to follow local electrical code requirements.

Doors

If you are going to put a door on your booth, a solid core door with a weather stripping seal is best. Or a pre-hung exterior door (with glass) if you are claustrophobic.

Example of flexible sound isolating ductwork

This was a return between the control room and the booth. Note airspace between walls; front wall on floor awaiting inner skin of drywall. Walls are usually constructed on floor, then tilted up into place and anchored

If you put in a door, you will have to consider putting in ventilation. Insulate the outside of metal ductwork using pink insulation, or use flexible insulated ductwork with an “S” bend to eliminate sound transmission. Put vents at floor level as far away from the mic position as possible, and putting them behind the mic position is best. Do not put grilles on the vents as a grill creates turbulence, which creates a sound.

A good strategy when dealing with HVAC is to use ductwork as large in diameter as the feed you are tapping into. Smaller diameter ductwork than the supply line can create a higher CFM pressure which would increase airflow pressure, making it audible. If you can step up to larger diameter ductwork than the feed using an adapter, that will lower the CFM going into your booth and will lower the airflow sound even more.

TUNING: bass traps

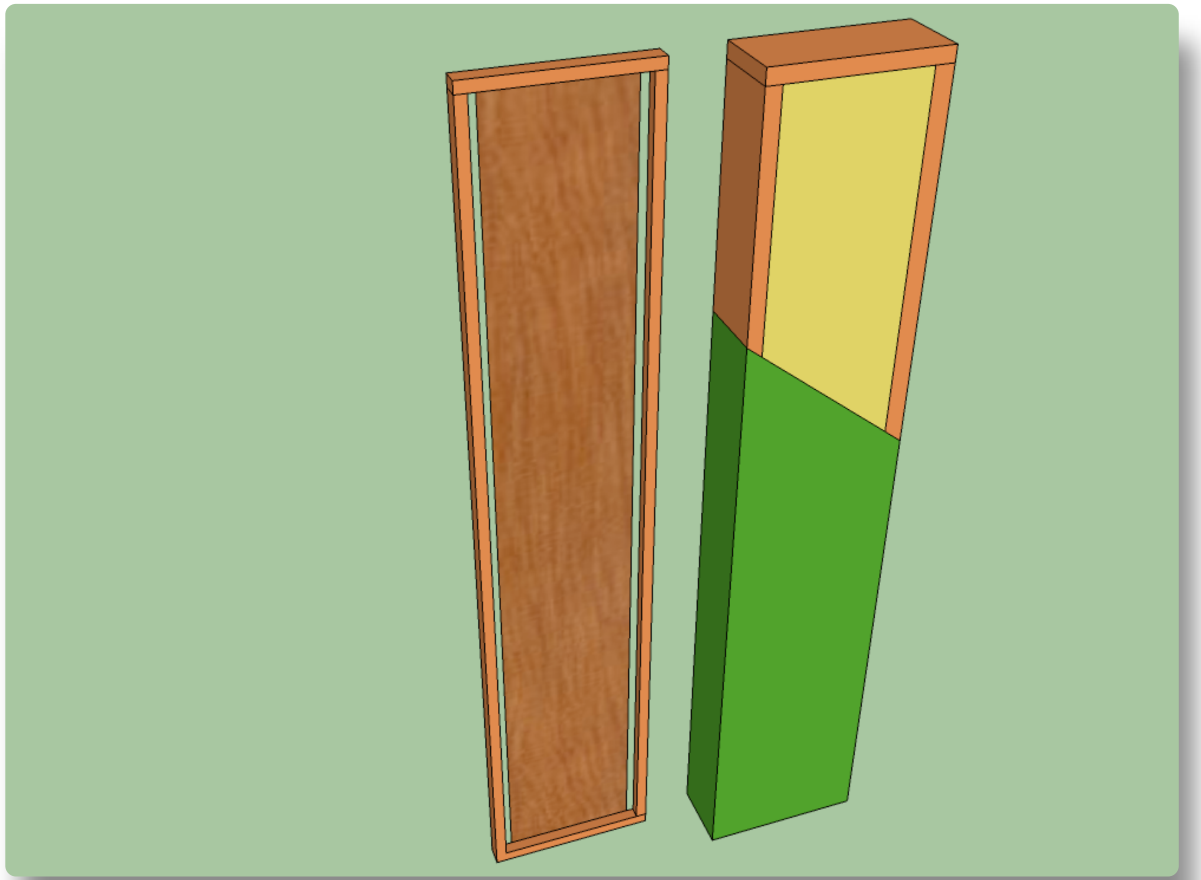
You may need to construct a bass trap, or purchase commercially available units for the corner(s)

A simple trap can be constructed by making a 2X2 frame 18” wide X the height of the booth but instead of putting fiberglass in, attach a 15”wide panel of Luanne wood secured at the top and bottom only and place it in the corner at a 45 degree angle (truncating the corner) and make sure the Luanne can move freely.

The Luanne will vibrate and absorb the bass energy that normally builds up in the corner. See the units on absorption and diffusion for more information on tuning the booth.

Or build using a 1”x4” frame (same dimensions as above) and put 4” of 703 rigid fiberglass inside, and secure at a 45º angle in the corner. Cover either with double knit polyester for aesthetics.

Examples of homemade bass traps



Mount in corner at 45º angle

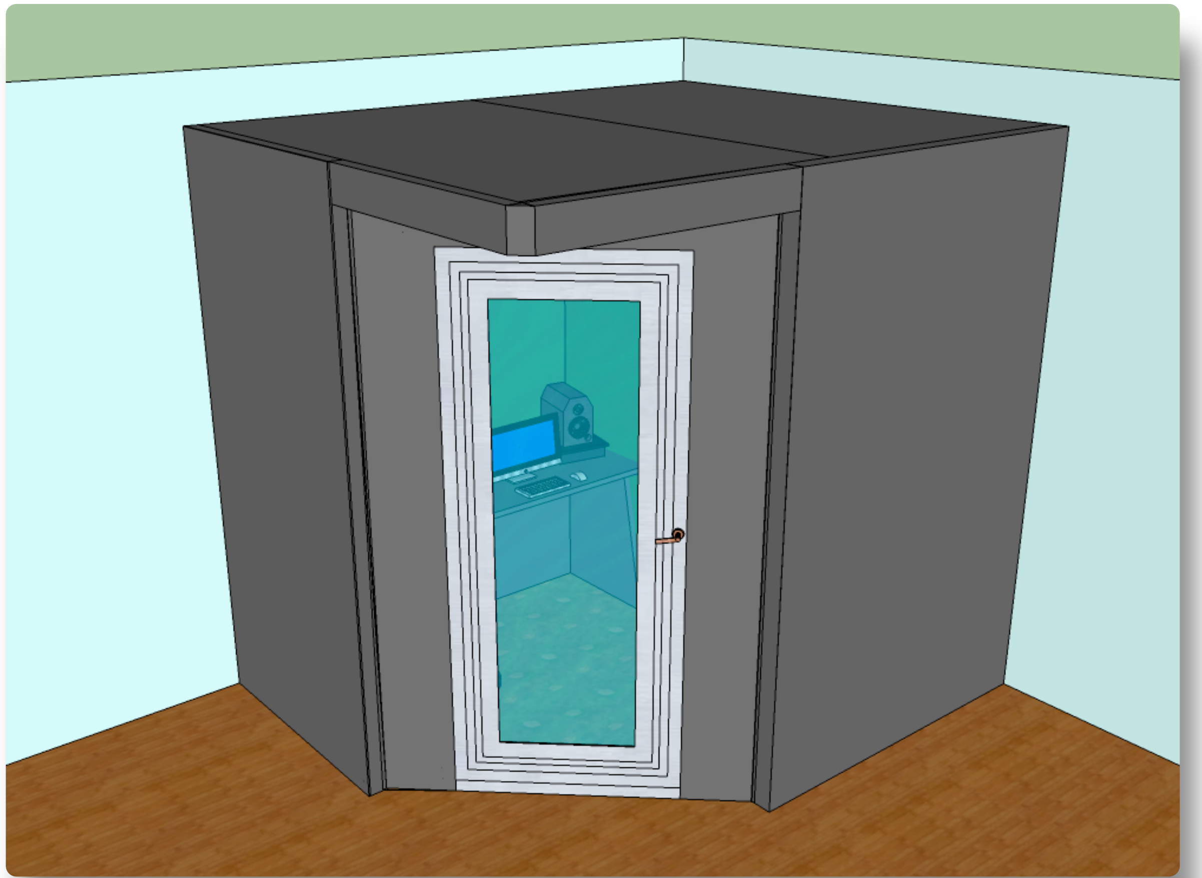
BOOTH DESIGN IDEA: Everything inside:

Many of the promo/station guys make the booth so that their computer screen is inside the booth, the computer outside to avoid fan noise. You can run all the connections through the wall. You can have a tablet with the script on it, a paper script to one side, a split screen on the monitor, or a two monitor set up, so that you have script on one side, recording program on the other. This way you can monitor levels, and record while sitting at your desk without needing to move to another area.   The only drawback is that you have to leave the “booth” to access the computer.
Big deal.

Many of the new computers are silent, so this would not be necessary. And many of the tablets now have recording apps. But be aware that computer screens reflect sound and should be cheated at an angle to avoid direct reflection of sound back into the mic. If you are a "self recorder", this may be an ideal set up for you.

Heat build up is an issue, and ventilation is a consideration. Many talent just open the door after recording while processing and editing. If you go this route, consider building a larger booth.

Desk in booth



(After a few times, you will always remember to turn the speakers off before you hit record.)

TOP OF THE LINE:

PROFESSIONAL GRADE STUDIO BUILD-OUT

A room within a room is still the most accepted form of sound isolation.

If you have the funds available, employing a reputable studio designer/builder will eliminate a variety of unforeseen problems. Most cities have contractors who are experienced studio designer/builders.

Tour some of studios they have built and ask the engineers what they think of the room, any problems, trouble points, cost over runs, etc., and get several bids. Even get bids from some top name designers. But if you are going to tackle the job yourself, planning is essential. From building permits, to zoning, to local codes for electrical, fire, to insurance, etc., you are the contractor and responsible for the job. Plus making the best design, construction, soundproofing purchasing decisions to create a great sounding room, and hopefully saving a tremendous amount of money in the process.

And consider taking several courses in acoustic/studio design before tackling a studio build out.

A pro-studio booth with double glass window

Here is an example of how the double knit absorbers give a flat, wall-like look.   Wedge-tile bass traps in corner.   This was overkill for voice. If you record bands, this kind of isolation is necessary.

If you are going to build a professional grade permanent booth, it is best to do some research into new products and techniques that are currently available.   Read up on studio construction techniques being used in the design of today's top facilities, and network with studio designers and acoustic engineers for tips and suggestions. And employ what ideas you feel will best suit your specific needs and budget.   Consider and design for future modifications. Cable and electrical, air handling, absorption/ diffusion, lighting, and the ability to tune the room are all factors to consider when designing your booth.

There are many commercial sound proofing companies/suppliers that have everything from isolators to create floating floors, sound blocking membranes, commercially finished sound absorbing panels that look like paintings, bass traps, diffusers, sound proof doors and windows, and a wide range of other products that are professionally finished and are seen in most studios.

They are easily found with on-line searches, and are carried by many of the major music store retailers or sound proofing supplers. The costs range from reasonable to very expensive.

Some of the items may have a profound affect on your sound proofing goals, some may have only a minor effect on the overall sound quality of the room.   And there are work-arounds on nearly every option.

fig a.

Such as a two door soundlock (fig a.) using solid core doors, vs. a commercially produced soundproof door system. The difference in isolation is nearly the same, the cost, however, is drastically different.

COMMERCIALLY AVAILABLE BOOTHS:

The most expensive option is commercially available sound booths that can run anywhere from $3,000 (just big enough for one person standing) to $30,000 plus for larger format booth plus upgrades. Some assembly required.

And even at that price, they usually need some tweaking: one of our regular pro voice talent bought one (used, around $7,000) and the first files he sent sounded like he was talking in a cardboard box. He needed to add some traps, diffusion and absorption to get it to sound flat. And do some mic placement adjustment.

Advantage: if you are in an apartment, a busy household, in an area with high ambient noise, or have no place to construct a professional booth, then this may be the answer. If you are doing more than a couple of hours of recording a day, the cost may be worth it.

They are semi permanent, modular, so they can be broken down and moved to another location if necessary. They qualify as the "room within a room" and are more expedient than building a booth.

You can find used isolation booths at a substantial discount in the electronic classifieds.

Be aware that ventilation is an "optional extra" that has to be considered; they are soundproof, which means airproof, and they get stuffy very quickly.

Some have called them dual duty booth/saunas.

But you still have to treat the control room outside if you don't have an "in box" set up. (You have to buy a large one for that) Having a tight booth and a poor listening/monitoring room is no bargain either.

If you learn to use power tools and do it yourself, you can build a booth that performs as well as these, and pocket enough to buy a couple of expensive mics, a top flight preamp, full blown computer, a bunch of expensive plug-ins (that will help you work faster and sound better) plus enough left over for that trip to Branson you always wanted to take.

The do-it-yourself rough frame booth, (either free standing or permanent) would cost under $1,500 in materials (for an 8 X 10.....estimating) vs. $20,000+ for a commercial sound booth roughly the same size.

The main power tools you need: a variable speed drill, either 18v or AC; a circular saw, mitre saw, and if you really get serious about building your own bookcase diffusers: a router. An electric stapler is a good idea too, unless you really want to develop that Kung-Fu grip by using a manual stapler.

Or if you have a friend who is a power tool user, hire him/her to build it and pay them generously. You will still come out WAY ahead. Besides, you know he/she needs the work, anyhow.

Go on line and check out the various manufacturers and prices.

A NOTE ABOUT WINDOWS:

I did not include windows in any of my booth designs for a reason.

It is just industry habit that everyone expects a window to watch the talent perform.

The way business is done today, ie: phone patch, ISDN, etc; it is a relic from the bygone days of big studios when most recording sessions were live music and the control room had to be separated from the studio to avoid bleed affecting the final mix. The monitors were loud, the equipment was noisy and everyone in the control room was talking. Plus they wanted to watch the band perform to see what was happening in the studio. The double glass windows were set in a non parallel configuration to avoid transmission of vibrations from the studio pane to the control room pane.

But with computer recording, there is no noisy equipment, so a window no longer needs to be built to studio specs for isolation. And producers probably don't come to your studio.

Since most of you will be doing self record, there is no reason to put in a window at all. But if you feel compelled to do so, (for natural light, or aesthetics) just pick up a double glazed pre-hung vinyl window at the big box hardware store. Get one that has a nice seal and feels solid. One of these should provide adequate sound attenuation. But don't forget, a window is a reflective surface.

Advantage: you can open it for ventilation after recording, and you can get one that gives your booth a nice cozy "home-y" feel; hang curtains, wood blinds, (good for diffusion) put on a flower box, etc.

Go on line to get rough framing tips; seal all gaps with GreatStuff, (or the like) or stuff fiberglass in any and all gaps before drywalling.

And if you decide against putting in a window and your client insists on watching you record, do facetime, or skype, etc. They will be amazed that you are so forward thinking to use technology this advanced.

Non-parallel walls:

If possible, try to make the walls inside the booth non-parallel. This will help eliminate some of the reflections you get when sound bounces back and forth between walls/ceiling/floor. (Remember the pool table analogy) Most of the better studios have many different elevations that help to diffuse reflections. Even a simple room within a room, semi-permanent, or permanent, can benefit from cheating walls away from parallel wall construction.

ACKNOWLEDGEMENTS: SPECIAL THANKS.

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(You should look up his work; his creative use of words and sound will open your ears to what Theater of the Mind can be)

John Metzger: Studio Designer / Builder. Chicago Media Works. John did the heavy construction on one of my studio build-outs.
It was as solid as a bank vault, and wanted to thank him for his terrific work. Many of the construction techniques in this book were from observing his work.

Linda and John Kelly, who helped renovate and soundproof my first downtown studio. (learned a lot of construction tips from them, too.)

And a big thank you to Danny Gustafson and all of our clients who were willing to take a chance on some of our hair-brained ideas (which all worked out great.....except one) Who supported us (literally) and kept coming back with incredibly fun projects. (well, except that one)

And all the talented voices who came through my studios, who made every day of my career a joy. (and provided some great stories, too.)
Rose Abdoo, Kate Burns, Mike Bacarella, Jim Barton, Kristie Berger, Sherri Berger, Sue Berg, Jerry Bloom, Paul Bolger, Linn Burton, Margaurite Bynum, David Bryson, Shelly Carlson, Bryon Carmody, Adam Conway, Tom Cramer, Michael Freeman, Charles Fuller, Shirley Hayes, Tracy Johnson, Rich Koz, Patricia Martinez, Kevin McAllen, Jonathan Menchin, Ralph Metz, Nina Montelione, Paul D. Morgan, Lisa Murray, Rose Nadolsky, Jamie Newell, Mike Noonan, Gary Price, Tony Pesce, Christine Rosencrans, Allen Rubin, Lisa Sesma, Margaret Scott, Cappy Silver, Emily Simer, Kim Spelman, Lesley Spencer, Alan Stagg, Darren Stephens, Laura Stigler, Tom Test, Lisa Taylor, Nancy Veselica, Ellie Weingardt, Fred Young, Tim Walkoe, Darryl Warren, Teri Wilder, Jeff Zimmerman, and Mike Ditka, Steve Carell, Alice Ghostly, Orion Samuelson, Emilio Estevez, and many more talented voices whom I may have overlooked unintentionally.

The orchestra is starting to play; thank you all again.

LINKS (YOU WILL PROBABLY HAVE TO COPY AND PASTE THESE INTO YOUR BROWSER)

ADK mics: www.adkmic.com Great sounding line of mics for VO and pro studio applications. Top notch sound; good company.

Ken Goerres: EXAKTE: Ken is an audio genius and Grammy nominated producer/engineer; I use his near-field monitors, some of his high-end wiring, and he showed me the double-knit/703 /frame absorber design (and Luanne bass trap design) which has made all of my studios sound and look fantastic. www.exakte.com and www.audioconductors.com

Sweetwater: This is a great resource for gear, software, and they have terrific customer service. If you have a problem, they have tech support. Can source an entire studio from computer to microphones, monitors, to sound attenuation products.
www.sweetwater.com

ATS: acoustic products: good source for Owens Corning 703 rigid fiberglass, or generic equivalent. Other useful sound attenuation products. www.atsacoustics.com

Guitar Center: Many locations, good source for mics, monitors, headphones, plug-ins, sound attenuation. www.guitarcenter.com

Vogue Fabrics: Great source for double knit polyester: low prices, large selection of colors. www.voguefabricsstore.com

The Copy WorkShop. This is a very smart group of people who publish books on advertising. Several are practically industry bibles. Get one or two. Or at least go to their website and listen to some classic radio commercials. Pick up some copywriting chops too, so you can save your client's half-baked script. www.adbuzz.com

Jordan Reynolds: Colorado Voice Talent/Musician/home studio owner who has a very entertaining and informative video tour of his studio; and a nice little breakdown on building absorber panels (slightly different than the ones outlined in the "BOOTHS" chapter.)
Good info, someone you should know. www.jordanreynolds.com (click on blog, scroll to down to video tour.)

cont.

LINKS, CONT. AND SOME SUGGESTIONS, TOO.

Greg Bennett: www.ethanhartguitars.com For those of you who are musician/guitar players, Greg has a beautiful line of guitars you should check out. These things could put Viagra out of business. He also invented the glider capo, which may be one of the cleverest capo designs in history. www.youtube.com/watch?v=Bo__FVbZSww

Joy Tillis: WJOY music clearance/licensing: www.wjoymusic.com If your client wants to license a pop song, Joy is always my first call.

Kate Bacon: WellDunne entertainment marketing: TV, promo, cable.   blog.welldunnetalent.com This newsletter has a ton of information on who is moving to what position at what station, job openings at stations; newly hired station voice talent (promo/topical/image)

Linkedin groups: You should get an account and sign up for Small Recording Studio Networking (for studio tips) and Voice Over Professionals. Many smart opinions offered and debated on a variety of subjects. Check into other groups too, and check into some of the Facebook group discussions on studio procedures, voice-over tips, etc.

Youtube: there are a ton of videos on constructing bass traps, sound panels, making a cheapo booth to a pro booth with a floating floor. Watch all of them. You will get some great ideas, construction tips, and know what to expect when you build your own.

Mix Magazine, Electronic Musician, Pro Audio Review, etc. Industry magazines that will alert you to new gear, new programs, new plug-ins, that may help you work faster and sound better. Get a subscription. These magazines are like the Chadwick's catalog for gear sluts.

AES, NAB, Namm, etc.: Trade shows. Get a pass from someone and go. The latest breaking technology is shown at these shows. You may be able to score a beta version either free, or at a greatly reduced price. Sometimes. (always ask)

Visit as many recording studios as you can. You will pick up some great tips on design, placement of mix position, absorber/diffusion arrays, even decorating/interior design elements. As they always say in Hollywood, "Steal from the best". (I'm just talking ideas here.)

One last piece of advice that may get you as far as any tips in this handbook: NETWORK!!!!!!!! You will get more work from people you have a relationship with than from any other source. There are a ton of people in this business and personal connections are the only way to separate yourself from the herd. (ok, "heard"). BUT you still have to *sound* as good as the best. And good luck to you; this career can be as much fun as you can have legally.   Stop reading. Go make a new contact. GO! (I heard a reflection, you still need to tweak your room.)

ABOUT THE AUTHOR:

Normally these are written in third person because it is a bit embarrassing to toot one's own horn, but times change. BFD.

I started in the business doing home recording in the analog era, when you had to make crappy sounding equipment sound as good as equipment you couldn't afford. You learned a lot of tricks, swapped ideas with other guys doing recording, and watched closely when you were actually in a real recording studio.

I am a guitar player; backed up Tanya Tucker on one of her early national tours. That led to a job at Leo Burnett as a copywriter. (It's too complicated to get into) Was a second chair voice-over fill-in at Burnett, and got interested in the voice-over business. Took some improv classes at Second City. Built a home studio in my apartment. Got married, moved to a house, built a second home studio and then gained enough clients to move to an historic downtown Chicago studio that had been vacant and needed major renovation. Ken Goerres (Exakte) taught me a bunch of guerrilla studio techniques that allowed me to improve the sound and the visual impact of the space for a very reasonable price. I hired an engineer, Jim Doherty, who is a skilled director, recording engineer, and was a creative partner in the business of making sound. We moved to a second studio after the first building was turned into a Hard Rock Hotel; we were at the second location for 12 years until the business shifted to home based studios. Built a home studio from scratch using most of the techniques outlined in the BOOTH chapter. And it sounds as good as the downtown spaces.

I have picked up a tremendous amount of information from friends; engineers, studio designers, construction guys, and Ed at Harold's True Value Hardware (who knows tools like nobody else)........and hope that some of what I have learned will be of benefit to you.

If you must know more, go to my linkedin page, www.linkedin.com/pub/nick-sanabria/0/403/758

www.nicksanabria.com (voice over)

www.radiocityrecording.com (studio/music)

www.yourpersonalannouncer.com (300+ silly announcer lines app for iPhone/iPad)