

A brief audio guide for stop-motion movies or else...

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*Kind spelling corrections done by Buxton.
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Introduction

Could we imagine the *Lord Of The Rings* or *Star Wars*, with the soundtrack quality of a wedding video ? Audio is a big part of the show, some directors even saying that it's 50% of it (David Lynch, for example).

This document is not intended to be software specific, or even hardware specific : it just explains some good techniques to use when working on an audio soundtrack : these are wise tricks, that can be used either for a vacation video or for *Blade Runner*, so it should fit perfectly with Lego® movies ;)

To me, audio production is as time-consuming as video editing. Yes, audio is also a frame by frame job, requiring many experiments, many tries and much time to find THE sound that fits to the pictures.

The great feedback you gave me after the release of the first version of this guide strongly encouraged me to go further in that way and to develop new topics, even if this guide is 85% identical to the first edition. New topics cover the monitoring, voice acting, portable recorders and so on.

That said, turn on your mics and software, and open your ears....

How to do a great voice / ambient recording

Voices in my head

Probably the most important thing in a movie : hard-to-understand voices really are a pain, and greatly contributes to the lack of interest.

And for animation movies, the problem is much more complicated because we don't have any substance – despite all my efforts, my minifigs don't speak to me... and recording the sound while I'm doing stop-motion animation would give a more shocking result than even *Pulp Fiction*'s dialogs !!!

So we will have to create everything from scratch : voices, foleys (i.e. sounds, fx, etc.), background ambiances and music.

Anyway the voice production requires a huge pre-production work, which is not really the plot of this document, even if the voice acting section is somewhat related to pre-production.

So let's assume your dialogs are ready : what shall we do before pressing that magic "Record" button ?

Spirits in a material world

Yes the material is highly important when recording anything : it won't do all the job, but believe me, a good microphone opens the Doors of the Creation.

Generally, a cheap mic is included with your computer, or on a headset... Well, bad news, these devices should be ignored when planning a serious job. Here's why : those mics are NOT sensitive at all, so you need to shout at it in order to have any audible sound, and if you raise the preamp level in your Windows™ console, you get a kind of Niagara falls shuffle and hum. The result is ugly : you've got MUCH noise, your voice is hardly audible and is often loud and distorted.

Now, let's enter the sound of silence : the top-notch configuration (which is mine, ah, ah, see photos.) : static (condenser) microphone with anti-pop grid, shockmounted on a heavy support, with XLR wiring to a tube preamp that output a balanced signal to a numeric console then to a \$600 audio-dedicated digital soundcard... But honestly, you **really** don't need that ;)



High-end static mic with shock-mount and anti-pop grid on a heavy mic stand. Top class...

Ok, then we have introduced the two "limits", so let's describe each element in the voice chain to see the improvement it will bring to your Frank Sinatra's voice.

FIRST : THE MIC

Here's a VERY important piece of advice : it's really worth spending some money on a decent microphone. Today, they are not so expensive, and you can find many of them on auction sites (beware of the quality : I would not buy Marilyn Manson's stage mic !). Try some SHURE SM58 or BETA or SENNHEISER or similar : they are rock solid, and sensitive enough for your movie. Plus, they don't need the additional power (technically called Phantom Power, +48V) that is required by condenser mics, plus they are cheaper, plus there are less sensitive to air, light and moisture.

If you want to spend more money, look into the static / condenser mics (RODE for example) : they are very sensitive, and therefore harder to use.



The ultra classic Shure SM58. Very good value for the money.

SECOND : THE WIRES

Often ignored, the wiring is the secret to avoid noises and hum.

The basic wire consist in 3,5mm non-shielded minijack : they are cheap, but not solid, and even worse, they capture any radiation : fan cooler, halogen lamp, computer screen...

More reliable are the shielded 6,35mm wires : the section is more important, allowing more current to pass and minimizing losses, and they are less sensitive to radiations. It's typically the musical jack that your favorite guitarist uses to connect his axe to the poweramp. Very good jacks can be very expensive (FENDER, not to mention MOGAMI, MONSTER CABLE), but you don't need those hyped cables.



A XLR plug with the 3 pin connector.

How to know the quality of a jack ? Here's a test : hit your cable on the ground; if you hear scratches in the speaker then your cable is crap. Sorry...

RCA cable is the twin of 6,35 jack except the plugs and the fact that it is more consumer oriented.

Next type of audio cable: the XLR or symmetric : it's used to connect hi-end mics and its plug has a 3-

pin contact. Because it's symmetric (I won't explain the electric details), it has very low sensitivity to radiation, and it can power the mic.



A shielded Fender jack : rock solid but expensive...

Anyway, don't forget : good cable = less distortion, less noise, more signal.

You can also buy adapters to convert XLR to 6,35 (losing symmetry), or 6,35 to 3,5 but beware, they are sometimes noisy.

THIRD : PREAMP or CONSOLE (optional)

A microphone, even with the best wires, delivers a low signal, so if you want to record it, you need to amplify it – it's called preamplification.

Notice that even if you plug your mic directly in the mic input of your soundcard, there is also a preamplification, but it's done by your soundcard. In fact, there is always preamplification.

Unless you own a good quality soundcard (pro or consumer), the preamp of your soundcard will be weak. I know this issue has been improved in the latest good soundcards (those SOUNDBLASTER Audigy / Extigy for example). But amplifying the mic signal in the computer is a little too late because the computer is a box of radiation, and before being amplified, your weak mic signal will be contaminated with noises and hum... and you will amplify both of them.

The issue is to amplify the signal away from the computer, in fact, as near as possible to the microphone.

Some soundcards have a front panel that fits in a 5/4" drive bay of your computer. Generally these are good soundcards, featuring a front 6,35 jack input allowing a mic or a low-impedance instrument to be connected (i.e. a guitar). This is the case of the Audigy Platinum or Emu APS...

If you just have a 3,5mm input at the rear of your PC, it's better to use an external device to amplify the signal. This can either be done by a preamp or a mixing console or both (if you can afford it). These items will amplify the low mic signal with good electronic devices, before sending it to the computer.

You can find very cheap consoles made by many manufacturers (EDIROL, BEHRINGER, PHONIC, MIDIMAN, YAMAHA, etc.), and you could buy a used one at minimum risk (and cost) without any difficulty (check some auction websites).

A console includes not only one or more preamps, but also useful features : EQ (2 or 3 bands – see below for further details about EQing), several inputs (so that you can plug another mic, allowing you to record a background noise or a foley in stereo or to do dubbing sessions with several partners), etc.

A little console is really worth the investment, and can be useful in many situations (incl. Karaoke sessions with your grandmother). Then, you plug the output of the mixer (line level signal) to the line input of your soundcard with GOOD cables: your recordings should be way more clear, and the obtained purity will almost make you cry...

These days, new devices have appeared on the musical market featuring a preamp, sometimes a multi-effect, and a direct connection to your PC via the USB port. This can be interesting, but they are not really cheap yet (EDIROL, YAMAHA, MIDIMAN) and are very music oriented.

I can hear you protesting : "A good mic, quality wiring and multichannel console ! Hey man, I'm not into bank robberies !!!". Well, eventually, you should seriously think about a bank robbery.

Seriously, if you only buy one thing, then go for a good mic, you'll thank me, but verify that the mic can be plugged into your soundcard, even if only with a little adapter. Ask the salesman (what a joke !).

But before to go on, let's examine another kind multi purpose solution.

Mini mini

Another alternative that could be very clever is to use some portable device to record audio.

I'm mainly thinking of a minidisk recorder, although there is now a large offer of portable mp3 recorders.

Associated with a good mic – again – you can have a quite affordable killer solution. Let's go into the advantages of such a solution.

First, it's portable, that means you can carry it anywhere thus use it in various situations. You can record your dialogs away from the noisy computer or reverberating room. You can record you own Foleys easily either in mono or in stereo (provided you've got a stereo microphone).

With the lowering price of these devices, it's worth the try. Note that the latest models can let you name each take you do, and can be directly connected to a computer via USB, allowing instant download. This function is called NetMD (at the time I write this article, upload to the computer is not yet allowed due to software limitation. But let's guess another soft will allow bi-directional exchanges between MD and computer and why not, transform your MD into a portable hard disk).

Many websites are dedicated to the MD, so it shouldn't be difficult to choose a model. Anyway, before ordering a MD recorder, be sure it has all the connections (optical, mic input, line input, line output / headphones). A front control of the recording volume is also very welcome.

Choosing the mic is quite simple as there is not so many references. Generally, those "mini" mics can be used on minidisk, but also on camcorders, mp3 recorder... They can be quite expensive (up to \$150 USD). Some hi-end stereo models will let you choose the directivity – e.g. the " width " of the recorded angle – 90°/120°, and there are also cannon mics that are very directive & ultra sensitive. However, you should not need those mics, there are more intended to be used in noisy environment (sports, windy place, wildlife recording). Many hi-end models are battery powered, and thus, preamplified. Good point for you !

Yes, you can use those mics on your soundcard, but remember that they are sensitive. So, if your computer fan is noisy or your soundcard is very cheap, you will not have the benefit you were waiting for.

The inverse is also true : you can use your regular mic (with an eventual 3,5 mm adapter) on a MD recorder.

But whatever the solution you choose, the following topic should help you to improve your takes.

Stop breathing

Where to place the microphone

Recording a good take directly depends on how you prepared it : I mean, you have to prepare both material and voice actors.

I know I've been annoying you with the need for a good microphone, the sensitivity and all things like that : but what's the use of that ?

Just because at the recording time, we will get slightly off the microphone !

When you speak into the mic at a distance of 15cm or less (this is called close-miking), the sound is not natural at all: there is a boost in the bass, the "p" or "t" cause a loud pop noise, the "s" or "z" are aggressive, the words are hardly audible especially with deep male voices. And you only record the mouth and the nose, not the throat.

To get rid of the proximity effect – this is the technical name – the best way is to stay between 60cm to 1m of the microphone.

If you have ever seen dubbing actors - it's a common bonus feature of the DVD version of most animation movies – you'll see that they stand behind a line on the ground, at about 1 m of the microphone (but they also record in a studio and have very good mics).

If you still hear pops on the “p”, you can either buy an anti-pop grid in a music shop, or try not to speak exactly in front of the mic (turn it a few degrees left or right), or do it yourself with rigid electrical wire and an old pair of stockings !

Now you can understand why you need a good mic : as you’re facing away from it, the incoming signal will be lower. If you want to capture something else than background noise and hum without shouting, you need a sensitive microphone (and an isolated room)!

There is another benefit of staying away : it adds the natural reverb of your recording room (in audio, we call it ambience) and make the take sound way more natural.

Hey, when you speak to someone, you don’t shout in its ears, don’t you ? It is exactly the same with your good old microphone !

Not to mention that you can record several actors with only one mic.

Another influence on the sound is the placement of the mic in your room.

First you need a very quiet room, away from your mom, fans, streets noises, cars, washing machines, mobile phones or neighbors making love. And please, shut off any halogen lamp : they radiate too much and generate intensive hum.

As you’re not close to the mic, any external noise will be recorded as well as your voice. In the next chapter we will see how to reduce the remaining background noises.

Also, don’t place the mic close to a blank wall. If your room is cubic or rectangular with blank walls, don’t place the mic at the center of the room : there would be some weird resonance. If you can, try to put some paintings or pieces of furniture or a screen or some hangings on the walls. The more “complex” your room is, the less resonance you’ll face.

For any voice recording, the actors need to be cool but they have to stand up. Don’t record actors when they are sitting or lying on the ground. So if you have a good mic, place it on a mic stand, in front of the head of your actors.

Checking the level

Whatever the device you record to, always monitor the input level. If it’s too low, you’ll hear too much background noise, if it’s too high, you’ll hear some clipping (i.e. digital distortion) on the voice. At highest signal, you should always be under the clipping limit (0 dB) but not too far. Hence, you will have the optimal recording level. This point is very important and is worth spending time to set it properly. Try to set your device so as to be at -3dB on the highest peak – i.e. highest recorded volume.

Speaking of digital format, always record in 44,1 kHz / 16 bits / mono or stereo or better. Don’t record in 22kHz or 8 bits.

In the audio world, we try to have the best source and the minimum treatment. Believe me, spending time on takes is 10 times worth than spending time on correcting a bad take. Repeat after me : “The source has to be as good as possible”.

Managing the voice

Well, now, it’s time to take care of you and your actors !

First, be zen !

Second, drink water after each take. You can eventually drink whisky if you want Joe Cocker’s voice in thir(s)ty years.

Third, do several takes, you will have more freedom to choose at editing time.

Fourth, speak clearly, articulate your words !!! As we are filming minifigs, don’t be afraid of being caricatured or funny. Try some wacky voices, just like those on Cartoon Network. I personally give \$10 to the one who can imitate Donald Duck ;)

How to reduce noises

The previous advice should have helped you to reduce the amount of background noise or hum in your takes, but some noise could remain, or you may want to correct old noisy takes.

Here are a few tricks to enhance the quality of the sound by removing the noise.

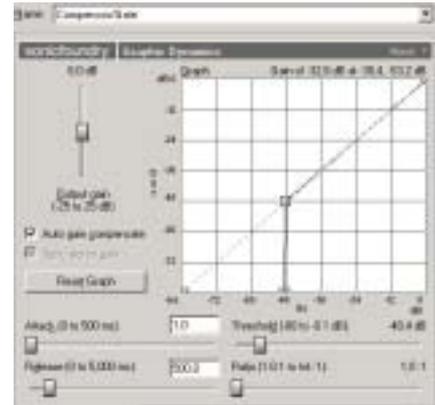
With your audio editor, you should cut the silent periods (between two takes for example).

Now if you feel that there is noise on your voice, concentrate on this : is there audible noise when you speak, or only in the short “silences” between two words or two sentences ?

Pipers at the Gate of Dawn

In the second case, a simple gate should help you.

A gate or noise gate is a type of processing that eliminates the signal below a certain threshold (that you set), and preserves the original signal path beyond this threshold. To set the threshold, take a short portion of your file where you only hear the background noise (the end of a sentence for example) and do an analysis to have the mean level of that noise and its variations. Then go to your gate effect and set the threshold a few dB higher than this noise level. The gate should now cut noise during the “silent” portions, but keep your voice unaffected. If you hear some brutal changes between silence and voice, adjust the attack and release parameter to allow smooth transitions.



A gate in action : nothing passes under -48 dB, but the signal is not processed beyond -48 dB.

If you don't have a gate in your toolbox, you can find free DirectX plugins on the web that your favorite audio editor can certainly host. I think there is a gate in Premiere (under the audio effects), and there's one in Vegas Video. You can also use music software to do that job, like Logic Audio or Cubase. In GoldWave, copy the following curve in your dynamics effect section (see photo).

Beware, if your background noise is really audible and you gate it “hard”, the result could sound very unnatural with moment of voice with noise alternating with total silence. Sometimes, it's even better to keep the noise (see further chapters).

The Sound of Silence

If a gate has failed in killing the noise, you will have to alter the spectral content of your take, by using EQ or Noise reduction. However this will alter your sound, so use it carefully.

Noise reduction effect is very dependent on the software you use. Some are dedicated to this task (Steinberg Clean, or SoFo Noise Reduction plugin for Sound Forge), and can be quite expensive. I don't know the quality of GoldWave's noise reduction algorithm.

In SoundForge, Noise Reduction allows you to capture a spectral “footprint” of your noise and remove it from your take. It requires many tries to correct the noise with limited damage to your voice ! That's why I previously wrote that it was generally easier to prepare a good take than correcting a bad one, with a better result. I'm not a big enthusiast of noise reduction !

More subtle is the EQ : if your noise is not widespread (i.e. it doesn't occur in the whole spectrum), then you can correct the noisy frequencies. Try to lower by a few dB the high-end of the spectrum (beyond 8kHz). If you've got an electric hum at 60Hz (or 50Hz in France), apply a -6 dB correction at this frequency with a strong Q factor as explained below, and a smoother correction at the double frequency (-3 dB at 120 Hz to kill the first harmonic). If you don't have Barry White or Cunnie William's voice, you can even cut the lower frequencies below 100 Hz.

If your takes are well done, you should not have to use EQ. However, if you record close to the microphone, you'll probably want to get rid of the proximity effect. Well, it's quite impossible to entirely remove it and sound natural. But, you can correct it to make your voice a little more pleasant to ear. First, use the EQ to remove the lower end of the voice by applying a -12 dB filter around 150Hz with a spread Q factor (0,5). Don't remove the low end entirely !

Try many settings until you are happy, you'll see that the result depends on the voice, and that there is no magic setting. The next thing to do is to apply a slight reverb, a latter chapter is dedicated to this topic.

Anyway, it's generally very hard to anticipate how annoying the noise is until you have done the whole mix with music, foleys,... So don't spend too much time on noise reduction, you can fix this at mixing time with a better idea of the general sound.

And, very important, always keep your original takes, don't apply hazardous treatments to your original material, always work on a copy of these takes. Now you can buy a 40GB hard disk with the change from a loaf of bread so don't be shy about making security copies.

How to find foleys and how to use them

A Strange Effect

Well, you can find many free foleys (sound fx) on the web, although too many of them sound lo-fi. Some are not free (SoundDogs.com, for example, has a huge library), and you can also buy CD or CD-ROMs of FX. Try www.univers-sons.fr or www.bestservice.de, these European online stores are dedicated to sampling CDs. Generally, professional foleys are very expensive (the Hollywood Edge collection or Hanna-Barbera FX are a hundred times the budget of a Lego movie!).

A clever way to find specific foleys is to try fan sites on the web : any *Star Wars* enthusiast has a collection of blaster and ships on its website . For my Lego project, I needed horse noises, I've found the site of a stable in the US that hosted various excellent sounds !!!

Also, a very good source is in the movies ! You need a medieval battle ? *Rob Roy*, *Highlander* or *The Lord Of The Rings* should bring you some wonderful foleys. Some gun sounds ? *Rambo*, *Pulp Fiction* or even *Doom*, *Unreal* or other video games... A quick rip, and the sound of your dreams is in your pocket.

Applying some effects to modify the sounds (EQ, chorus,...) can also be very creative. Anyway, if you plan to diffuse widely your movie, or plan to do some commercial use, I strongly suggest you not to use recognizable parts of movies ! You could be annoyed by someone who's picky on legal rights.

But probably, you'll enjoy doing your own foleys. Well this is a good idea and you will be allowed to add yourself as "sound designer" in the end credits ;)

Here, no surprises, you need a good mic (did I write something about this ?). If you want to do outdoor foleys (record your own harbor or crowd sounds), you'll need a portable recorder as explained before. This could be a DAT (more pro) or a minidisk or mp3 recorder. For backgrounds, it's better to use a stereo mic, and not to move it during the recording.

For foleys, you can record in either mono or stereo. Try to record the sounds as dry and isolated as possible (not in a reverb room), so that you can add any effect later.

If you record a single sound, do several takes, varying the action, the proximity, the angle.

If you record ambiances, record many more time than needed : you will be thankful if you need variations of this ambience, or if the director finally chooses to lengthen the scene.

If you're doing outdoor recording, take care of the weather ! Never place the mics in the wind : you'll just record an ugly woof. Try to protect your mic from any external aggression : wind, rain, moisture, very high level noises. As for the voices, record at the highest audio quality possible without saturation, and check the input level before pressing record (if your device allows that input monitoring).

There are many tips and tricks that sound designers use to imitate real sounds – think of coconuts for a gallop like in Monty Pythons' *Holy Grail*. I know several of them, although I never tried them, but unfortunately, my English vocabulary is too limited to describe the way to do it ☺.

A good advice is to always have your MD recorder ready for action, and to record every sound that you find interesting. This can even be a kind of game. And if you become good at recording Foleys, you'll save the hundreds of bucks of a professional Foley library.

Master of Reality

Now that you have collected all the sounds you needed, it's time to use them ! More details will be given in the mix section, but here are some generalities.

Background ambiances have to be used at a low level, except for specific use. If your background seems to interfere with your dialogs and damage the clarity, try to apply some reverb to the background sound. This will give a smoother feeling, and the voices should be more audible.

Take care of the mean level of your foleys and try to respect their level in real life. Then adjust the level and balance according to the distance and placement of the source on the screen.

For example, avoid loud footsteps in snow at the center when your character on the screen is 100m away of the cam on the right of the field of vision.

Adjust the level and placement in real time i.e. follow the movements of the sources on the screen. Fade in if it approaches, set a low level if it's out of the field of vision,...

And please, limit the level of those explosions and gunfire : in most brick movies they are way too loud.

You will also apply gentle harmonic corrections – i.e. EQ – on sounds : when the source of a sound is far away, the sound will tend to lose his bass / lo-medium register – remember the proximity effect : the more close a sound is, the more bass he has got. The inverse is also true. So try to apply these rules to your Foleys : if the action is far from the cam, cut the bass.

Believe me, this will add a very realistic touch to your soundtrack !

How to set the reverb to do more "living" voices



A typical reverb : you can see the predelay, the decay, the damping. In the upper right corner are the Dry/Wet sliders.

I think most voices in our Lego movies are recorded via headsets or built-in mics, resulting in a very claustrophobic sound that is not pleasant to hear. We've already seen how to choose a good mic and how to place it to avoid the proximity effect and make the voices sound natural and ambient.

In some cases, the reverb can also be a great improvement, and add a realistic touch to your movie.

First of all, setting a reverb is a really hard job, so be careful, never apply too much reverb, it can be very boring and ruin your efforts by making your movie sound like a tunnel or a cathedral.

Reverb is more a way to accentuate a place, an architecture, or to create an ambience – like in the dream sequence of many movies. Often, reverberation can't really be heard and has a more "subliminal" action.

The first point to do is to choose the right reverb. To me, there are two categories of reverbs, the technical ones and the realistic ones.

Technical reverbs are full of technical terms : decay, damping, pre-delay, early reflections and so on. However, some presets should help you to find a suitable setting.

- Decay is globally the time it takes for the reverb to fade out. Avoid long decays except for wide reverberating spaces (church).
- Damping is the attenuation of frequencies along the reverberation, it generally applies to the treble. On metallic wall, damping is low, but with regular wall with wood, cement, wallpaper or carpets, the damping is high.
- Predelay is the time that the sound takes to hit the first wall and go back to you. In a small room, it's very short, but in a cathedral, as the walls are far away, it can take several ms and be clearly audible.
- Early reflections is a special treatment for the first reverberations (heard after the predelay). A high value gives a tunnel effect.

Realistic reverb are more expensive and try to simulate a real room : you can draw a virtual room, place the virtual source and the virtual mic and even choose the material covering each wall. This fits very well to video footage and will give a really realistic impression, but they are much more expensive.

However very good results can be achieved with both products.

The secret of a realistic reverb is the Dry/Wet setting. Almost any reverb has got this setting, generally as one or two faders. If you can put keyframes in your audio/video editor (Premiere and Vegas can), you'll be able to achieve a very smooth and realistic balance.

Dry/Wet is the ratio between the original (dry) source and the processed (reverb) sound. This can be considered as the distance between the source and you in real life.

For example, let's go in a cathedral with a friend : if you are at both ends and keep talking, you'll have to shout and you will hear your friend's voice among a great amount of reverb. Here the Dry/Wet will be about 35%/65%.

If you stand near each other, you will not have to shout, and you'll clearly hear your friend's voice, the reverb just being in the background. So, Dry/Wet is about 80%/20%.

Synchronized to video, you will have to set this ratio for each cam movement or each cut.

It's a long irritating job, but it's worth it for the result !

How to blend voices and act to make a film more pleasant

This point is rather obvious but it's quite necessary to have various voices in your movie, I mean several voice actors.

Try to blend voices : if two characters are always together, give them two different voices. It will be more easy to understand.

A difficult thing is to find some voice actresses ☺. You can imitate a female's voice when you are a male if you're doing a remake of *Mrs. Doubtfire* but it might be more efficient to convince your girlfriend / wife / sister / friends to have a little role in your first class production.

Ok, this is pre-production work. You should also give precise indications of the tone on the script, so that your actors will know when they have to shout, cry, whisper and so on. This point is particularly important when collaborating via Internet. I'll be back to that point in some later chapters.

Voice acting is actually acting! This means not reading a text in front of a microphone, but make it live. So when you have to incarnate a minifig character, you are welcome to whisper, shout, do some pauses between words,...

Here is how I prepare myself for voice acting. Although I'm not an actor, I think those considerations may be helpful.

I ask the director to have all the lines, thus I see the interactions and relations between the different characters.

I make my sentences bolder or bigger, so as to read them at a certain distance (I don't put the text under my nose neither hold it my hands, I prefer to place it on a distant stand). I try to learn them by heart, so as to concentrate on the acting rather than the text. I do many tries before to record anything, with variations. I put some symbols in front of sentences, indicating that I should shout, scream, whisper, be sad, be ironic and so on.

Now that the text is "analysed", I concentrate on the recording, getting closer to the mic when whispering or far when shouting. When I say far, I mean I don't walk away, just put my head a little far from the microphone. Note that the volume is inverse-square related to the distance to the mic. If you increase the distance between your mouth and the mic by 3, you divide the volume by 9 or by more than -12dB (-3dB occurs each time the volume is divided by 2, the dB scale being logarithmic).

Another very smart addition to voice acting is to record also a pack of "voice FX", for example a couple of breathe sounds, grunts, gurps, burps, sniff, growls, ouch or whatever you want. This will give to the director the opportunity to fill the audio track with these sounds making the audio soundtrack more realistic.

How to use common effects

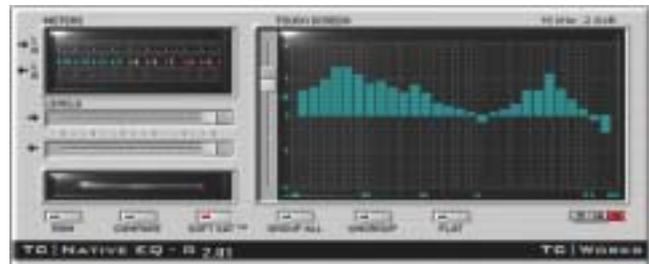
We've already seen how to use reverb to add more life to voices and how to use EQ to correct noises. There are many audio effects but not all are required in movie making. You can forget about phaser, flanger or autowah, they're pretty useless here.

In the audio world we distinguish two types of effects : insert effects & "parallel" effects. This refers to the way you should connect them to a soft or hard mixing console.

Insert effects are dedicated to dynamics and spectral content of the whole sound. These include EQ, Compressor, Limiter, Doppler, Time stretch, Pitch shift...

"Parallel" effects can be shared between channels using a channel effect send and a global send. Ok, this is not very obvious. So parallel effects can be applied to a part of the sound, so you will have to mix the original sound with the processed sound, in a word they feature a Dry/Wet setting. Sounds familiar (no pun!) ? Yes, the reverb is a kind of parallel effect. Modulation effects (chorus, phaser, vocoder) and time-based effects (reverb, delay) also belong to this wide family.

Equalization or EQ is a familiar effect that allows you to modify the spectral content of a sound. EQ is often used to correct a wrong repartition in the spectrum, or to attenuate or accentuate some frequencies. It can also be used as a creative tool : remove all frequencies of a voice except between 500Hz & 4kHz and you've got a typical phone voice. You can either work with a graphic EQ (fixed bands) or a parametric EQ (you draw a curve). I personally prefer parametric ones. My favorite are WAVES / TC WORKS / VINCENT BUREL DirectX plugins.



Graphic EQ : bands are fixed, you can only adjust the level of each band.

A parametric EQ generally has four parameters by band : frequency, level, Q and type. If there's no editable Q, it's called semi-parametric.

- Frequency : the center frequency (in Hz) that the treatment applies to.
- Level : the level of the correction in dB, the range can often be switched from +/-6 dB to +24dB/-infinite.
- Q aka quality factor : this does not refer to the quality of the sound, here the word "quality" is inherited from the electronic world. It describes the range of affected frequencies around the central frequency. Hi Q means that the range will be very limited, low Q (<1) will affect many frequencies around the central one.
- Type. There are several usual types of curves. Hi-pass will cut the lower frequencies, Lo-pass will the higher frequencies, normal will show a bell curve, notch (or band reject) will attenuate all frequencies except those around the central frequency point according to the Q setting. The following pictures should help you.



A parametric EQ. From left to right (yellow dots) : hi-pass, bell with high Q value, bell with low Q value, low-pass.

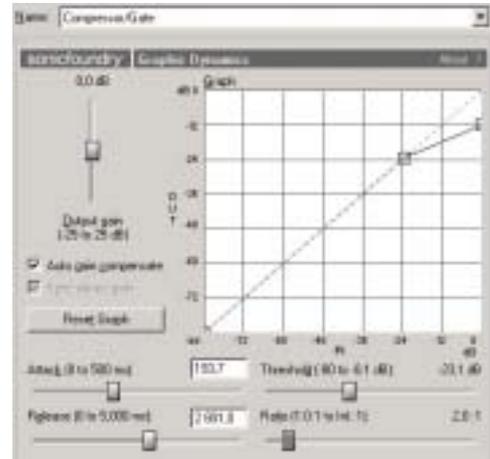
Compressor & Limiter are two twin effects that will affect the dynamics of a sound. Dynamics is an audio term that means the difference between the lowest and the highest level of a sound. It's mostly expressed in dB, but sometimes in bits, although the meaning is quite different.

A compressor will reduce the dynamic range of a sound by lowering the peaks. It can be used on the voices to homogenize the levels between a whisper and a shout : it will enhance the whispers, making them more audible, and lower the scream, making it less ear-breaking. Isn't it magic ?

As any audio effect, don't overuse it, or you might experience some weird horrible voice.

A compressor has at least the following parameters : Threshold, Ratio, Make-up gain, Attack/Release time.

- The threshold (in dB) represents the level where the compression starts. Below that point, the level is unaffected, beyond that point, it is attenuated depending on the ratio. It's sometimes called the knee.
- Ratio indicates the slope of the compression from 1:1 (no compression) to inf.:1 (every peak is cut down to the threshold, this is a limiter). For example, for a 2:1 ratio, if the input level is +3 dB beyond the threshold, the output level gains the half (+1,5 dB).
- Make-up gain (or output gain). As the peaks are attenuated, the overall volume of the sound decreases after a compression. We can compensate the loss by adding a post compression boost. By a clever setting, you can raise the volume of your sound when it's quiet, and lowers it when it's loud. Be careful with the make-up gain fader, a value that's too big can cause clipping and ruin your sound !
- Attack / Release. The time it takes for the compressor to react to the input signal. If it's too quick, the natural attack of a sound will be affected, sounding weird. If it's too slow, your compressor will always be late. Set a low value for hard attack noises (explosions, guns), and a 200ms value for the voice. Some compressors have got automatic attack / release calculation.



Typical compressor setting : below -24dB, the signal is not compressed, beyond -24dB, the signal is reduced with an 2:1 ratio. There is no make-up gain slider, it's replaced by the "Auto-gain compensate" checkbox. Notice the two attack/release sliders.

On some compressors, you can choose the kind of compression (tube, opto, transistor), the type of knee (i.e. how smooth is the curve around the knee), the frequencies of compression and so on. An expander is the opposite of a compressor (it boosts level below the threshold). Some good Direct X dynamics: TL Audio, Steinberg Magneto, Waves, TC Works, Vincent Burel...

Just to end with the term "compressor" : this has nothing to do with the file compression i.e. mp3 or else. Don't be confused !

Delay is just an echo generator. Depending on the stereo processing and the modulation possibilities, it can be called Mod Delay, Cross Delay, ... It's less useful for audio tracks, but you can use it to provide a strange feeling, or to reproduce specific ambiances : canyon, mountains, even magic forest, dream sequence or God's voice (!). Basically two parameters are required : the delay time (in ms) and the feedback (how many repetitions). If you set a short delay time, you've got a "slapback" effect that can be used on some FX. With short delay and hi feedback, you can transform a guitar into an Indian sitar.

Doppler is an effect that reproduces the changes in pitch & phase of an object moving around you. It's mostly used on vehicles : ambulances, jets, spaceships.

Pitch shift allows you to change the pitch of a sound without affecting the duration, while **Time Stretch** is the exact contrary. Can be interesting on sound design and for voices effects (monsters, dwarves, ...). Some voice processors can even turn a man's voice into a woman's and vice versa, by acting on the formants of a voice.



A vintage delay plugin.



A very complete Doppler simulation.

How to mix : the secret in the curves

There's no magic with mixing : a lot of experience is needed to get some good results. However there are a couple of things you should never forget when mixing.

In the video world, mixing applies to dialogs, sound fx, background and music.

Mixing is also a frame by frame work and requires much coffee.

Ghost in the Shell

First axis of mixing is the spectral content. The way to achieve a great mix is to ensure that the several audio layers are not sonically covering each other. The spectrum has to be balanced between low, middle and hi frequencies. If you notice some covering (in audio we call this masking), try to apply some EQ to cut annoying frequencies.

For example, a character is talking (the main spectrum of a voice is between 300 Hz and 4 kHz), and music is heard (just like in the typical 60's spy movies), making the voice harder to understand. Apply a gentle correction in the middle range to the music, and you should enhance the overall audio quality without lowering the music or boosting the voice. Magic !

Fade Away

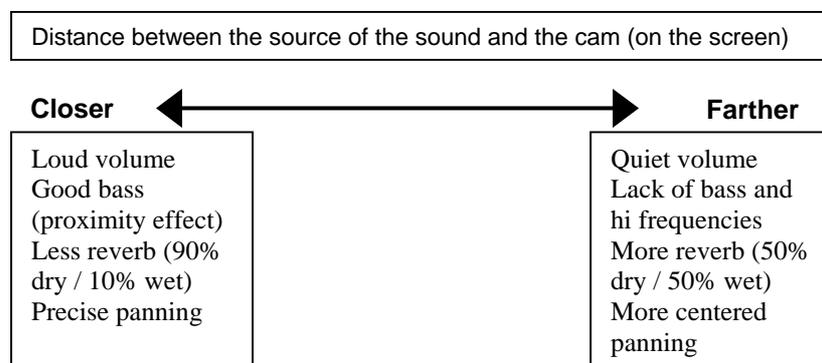
Acting on the volume sliders is rather obvious... When working on the volume, I concentrate on foleys and voices, and adjust the volumes to correspond to the pictures : louder volume on close-up, quieter on large angles... If your A/V software can handle keyframe, it's a piece of cake. Don't apply radical changes – it will irritate your audience. If you want to play with spaces, use the Dry/Wet parameter of your favorite reverb. Don't apply volume changes to music (except fade in and fade out), it will be done in the mastering stage.

Turn it up

Panoramic setting a.k.a. pan. This is the placement of a sound in the stereo field (L/C/R). A clever use of the pan can greatly improve the quality of your audio work. As always, do a keyframe editing if you can. It can be used to spatially distinguish two characters or to add foleys with no compromise on the voices' clarity. Here, you can put some radical changes for great effects : it's highly recommended on spectacular camera movements. However don't change the pan every 2 seconds or your audience will feel sick ! For stereo sounds like music, the term "balance" is also used.

With EQ, volume and pan, you have the control over 3 fundamental axes. Mixing is the sonic way to deal with spaces, in other words, it's the correlation between the sound (audio) and the space (video).

To resume the last topic, here is a little schema :



Note that this is just an indication and this is highly dependent on the mood of the movie you're working on. There are just generic rules, and rules are sometimes done to be broken...

How to do a mastering

Mastering is the last step : this is when we finalize the mix.

Once the mix is done, this is the way to homogenize the overall sound of your movie.

In A/V mastering, we usually use EQ and compressor.

Monitor

The mastering is a good topic to introduce a vast question : what's the best device to listen to the audio mix ?

Both at the mixing stage and the mastering stage, you need to listen to endless sound sessions so you'd better to choose a device that is both comfortable and accurate.

Even if pro studios are equipped with very good pairs – or walls for 5.1 mixing – of monitors, I think most of us use their computer loudspeakers or headphones to mix their movies. Well actually, this is not necessarily a bad thing !

Let's examine both solutions.

First, the headphones. Main drawbacks is that it's not really comfortable and it sounds unnatural. Advantages is that it's cheaper than loudspeakers, it's not depending on the acoustic of your room and it allows you to work without disturbing your neighborhood. I often use headphones to mix – even if it's not a pro solution. I recommend to buy a decent headphone however. You can find some AKG / Beyerdynamic or Sony ones at a reasonable price (\$100 USD). The more you spend, the more accurate it is and the more comfortable it is. Don't use in-ear headphones or cheap set – you know, the one that came with your computer featuring a headset microphone !

Here is the golden advice with headphones whatever the model you have : work at **quiet** volume ! Because of their proximity with the ear-drum, headphones can irreversibly damage your ears !

A more comfortable and less ear-damaging solution is the monitors i.e. loudspeakers. Well, choosing monitors is a vast topic with endless polemics but here are a few words of wisdom. Computer speakers are generally crap, unless you have bought something extra (like the Creative Labs or Altec Lansing sets). Don't be too trusty to your monitoring system : if some pro studios spends thousands of USD into a monitoring system, it is because it's worth ! Try to find the weaknesses of your actual system : does it lack bass ? Does it sound not wide ? So that you will be able to compensate it at the mixing / mastering stage.

If you've got a loudness or extra-loud or extra-bass setting, disable it. These systems are often outrageous and really distort the sound. A very strong influence on the sound is the placement of your monitors : try to place them 50cm (20 inches) away from a wall. Try to the corners in your room. Place the speakers at the height of your head and if you are the proud owner of a 2 or 3 ways system, place the tweeters (little speakers) at this height. Try to let 1,5 meter (5 feet) between each monitor for a wider sound, the ideal being to form an equilateral triangle between you and the two monitors. Try to place your monitor on a non resonant support (you can use some pads to isolate the speaker from the support). This mean you should add a sonically absorbent material between your monitors and your wooden or metallic desktop. This can be foam or textile (check Auralex website).

If you can't avoid a wall, lower the bass (provided you got a setting for it) by -2 or -3dB, if you can't avoid a corner, lower it by -4dB to -6dB. This will avoid a resonant woof effect because every close wall reflects the bass.

All this work should be done to let you work in an optimal way : you sound will be better and above all you will feel less tired.

While I'm on health considerations : do some pauses regularly, try to let some hours pass between each audio session, and don't do long audio mixing sessions. After 2 hours of work, your ears will be tired and you won't ear the same things ! Try to work with "fresh" ears. Don't be fooled : those advice are also used by professional musicians and audio workers.

Master of Puppets

Equalization is done if you want to apply slight changes in your final mix. You can, for example, apply a loudness so as to blow up your subwoofer.

Compressor is a more interesting tool when mastering : this will allow you to set the overall dynamic range of your whole movie, ensuring that low level sequences are still audible, and that action sequences are not too loud. A compressor will also automatically raise the music or background ambience when there are no dialogs. Isn't it great? The compressor is generally the last device in the audio chain, it is also used to maximize (or normalize) the main volume without saturation. This is very important because it will ensure the quality of the sound when you encode your video.

Some compressors are specifically dedicated to mastering – they are called “mastering compressor” or multi-band compressor ! The input signal is divided into several bands – low, middle, high – each band having a specific setting. Generally, those full-featured compressors include a limiter.

Some devices are also dedicated to mastering – EVENTIDE, AVALON,... in the hardware world, T-RACKS, SONICWORKS, PROSONIC,... in the software world. They include a mastering-oriented EQ, a multi-band compressor, a limiter, a stereo imager and sometimes vintage simulation.

Space Oddity

Another common effect used when mastering is the stereo image : it's used to make the mix sound wider and give an impression of space, just like a surround. Don't abuse on the wide spread for two reasons : it can quickly bother your audience and your mix will not be mono-compatible anymore.

These days, many hi end soundcards and speaker sets are 5.1 compatible, so some of us could think of doing 5.1 compatible movies. Remember that you also need 5.1 compatible mixing and authoring software ! For the moment, only a few top-notch programs include this feature. And if you have problems with a stereo mix, forget about 5.1, it's even harder to mix. Actually, I think 5.1 is more a commercial argument than a real need : I'll stick to the stereo world (but for how long ?). Mixing in 5.1 will also require a 5.1 monitoring system (ouch!) ! In fact 5.1 mixing is at these days more dedicated to pro or semi-pro studios that produce DVD. And obviously 5.1 audio will generate large files !

A few words about the encoding. Whatever the format you choose, the most important is to have the best sound possible before encoding, and to have the maximum volume. A good normalization of the volume will ensure that you use the bit resolution at its best. I won't go further because this could require another document about the digital conversion ! Please bear in mind that your overall mix should be as close as possible to 0 dB without exceeding it (look at the meters in you favorite editing software.)

Encoding is basically a bit reduction. If you have set a 16 bits digital audio file format, and your main volume is too low, you won't use all those disposable bits !!! On the encoding processing, you will spoil many of the available bits i.e. a good part of your audio quality.

How to work together

As we don't have a panel of voice actors within our studio and as we are living in different countries, we all have to work via the Internet, especially for voices.

But we've got different material, software and encoders. So when listening to the several sound files, there are chances that they aren't sonically compatible at all.

There are a couple of things to do to ensure that the quality of your sources is quite homogenic.

All Together Now !

First, if you're the director, give precise indications to the actors in the script : where the characters are located, the feeling they have, their relationships. You are the director, so it's your job to manage the actors, even voice actors ! These instructions should result in living voices, not flat at all. As our minifigs are not very expressive, only the actors have the possibilities to express feelings. This is pre-production work again.

If you're an actor and you feel confused when playing a character, ask for further information to your director.

Second, if you're the director, give technical specifications to your actors : what recording material they should use, how to use it. This is the purpose of this document, and I hope that it will help all of us.

Wave

Last thing, the encoding format to choose ! I assume that few of us have a cable/DSL modem and can handle uncompressed WAV files. Before sending files, always normalize them in your favorite audio editor. Don't apply effects : this is the sound engineer's job.

A pretty handy format is the MP3 file – with a bitrate superior to 96 kbps. More and more apps allow you to directly use mp3 files in your movie without conversion.

Less used is the Ogg Vorbis™ format. Encoded at the same quality as mp3, ogg files are smaller, but they need an ogg to wav conversion in order to be imported in your audio/video editor. However, this is my favorite format.

There are also numerous other compressed formats : RealAudio, Quicktime and so on.

Remember, whatever the format you choose, always normalize the original sound before encoding, and choose a decent bitrate when encoding.

Level Five

Despite all your precautions and instructions, you will certainly end with various sounding files. Here are the tricks to make them more coherent, by acting on the harmonic content and the volume.

First thing to check is the frequencies. Here you need to learn to separate the harmonic content of the tone of the voice. Obviously a male voice will have more bass than a female voice ! You have to check if certain files seem very sonically different : too much bass or treble, an audible hiss or hum or background noise.

If it is the case, isolate those files and apply correction to them using equalization so as to approach the global sounding of your files. You can eventually be helped by statistical analysis : many audio editors, and some audio players, offer the possibility to watch a spectrogram of the sound. This is a graphical representation of the harmonic content, allowing to visually detect the loudest and quietest frequencies. This is only an help, don't try to make all your files have the same spectrogram. As always, your ears will be your best indicators.

Second, you need to correct the volume of those files. Here, the tools can be more handy. Under many audio apps, you can see some statistics about the file you're hearing. Here, the most important indication is the RMS – root mean square – Power value. Serious statistics will show it in dB. Note this RMS power for every voice you've got. You should see the difference. You just have to lower – or increase – the volume of the files to make them quite close. Even if you have a precise tool, don't forget to confirm your settings with your ears. Depending on the sonic content, the RMS power will vary (that's why it's better to start with frequency checking, as stated in that paragraph).

Last but not least : process your voices globally ! If your actor has sent you a file for each sentences he says, merge them in one file using your audio editor before tweaking them. This way, you'll process every sentences at once, but

mainly, your sentences will remain coherent. Otherwise, you'll end with varying harmonic content / volume depending on each sentence. Just ugly !

However, and after hours of work, if you still end with very different voices, limit all your voices to the worst one.

If the worst has shuffle, add shuffle through all the movie !

If the worst is nose-sounding, EQ all your voices with nose-sounding.

If the worst has an audible reverb, add reverb on the whole movie.

Why ruin what was good ? Because, the most important is **coherence**, even if you need to sacrifice some audio purity... When watching a movie, your ears will accept shuffle, bad quality if it's constant. After some minutes, you will not notice them anymore. But if the quality is always changing from scenes to scenes and sentences to sentences, your listening experience will be greatly disturbing during all the watching.

Conclusion is, if you have to sound bad, always sound bad ! Even if it's obviously better to sound good.

Musical illustration

Music Non Stop

This is a bit off-topic but I wanted to write a short chapter about music in videos.

I've noticed that many brick films were featuring non-stop dialogs and that there was music only during the credits. I think any director should manage some pauses in its movies and include a break between some scenes. In regular movies, this is the moment to place a piece of music. In a soundtrack, I distinguish two types of music : the themes and the backgrounds. The themes are used to announce a place, a landscape or a character, they are not used along with dialogs except monologues sometimes. Themes are generally the music we recalled when thinking back on a movie.

Backgrounds are intended to be played low. There are just there to fill the silence during the dialogs. They are neither strong nor violent (kind of mall music).

For an average audience (!), it's hard to concentrate on several things at a time. So either you use a superb piece of music with few foleys, no background ambience and no dialogs, or you have dialogs, foleys, ambience and very low music or no music at all. Last thing, camera movements love music.

A very clever use of music can be seen in the movies directed by Quentin Tarantino, Guy Ritchie, David Lynch or Steven Soderbergh to name a few. If you're more into classical, have a look on Peter Jackson, George Lucas or Ridley Scott movies.

Decomposing Composers

Unless you work with your personal composer like many Hollywood directors do, you will have to find some suitable music for your movie.

Obviously, if you're doing a remake, you will pick up the original score. Don't forget that using that music is subject to copyright laws. If you plan to distribute it widely, you may upset someone – especially with Disney.

Some companies are dedicated to music editing : they work with various (unknown) composers, and can send you free CDs by a simple request. You only pay for what you use i.e. nothing for a private use. This kind of music can be heard in many TV programs, adverts or in malls. In France, we call it "musique au mètre" (music by the meter), as a reference to the textile world. I guess there must be hundreds of these companies near Hollywood.

Another way to find free quality music is to browse the web to find generous artists (www.mp3.com, or me, and so on). Probably they will let you use their music in exchange of a line in the end credits.

Try to avoid using midifiles unless you intentionally want to sound cheap.

Conclusion

Whatever your material, whatever your experience, whatever your knowledge, HAVE FUN !

Thanks to Doctor John, The Police, Jacques Dutronc, Perry Blake, Pink Floyd, Simon & Garfunkel, Hooverphonic, Black Sabbath, Kenji Kawai, Bruce Springsteen, Alan Parsons, Metallica, David Bowie, The Beatles, Antonio Carlos Jobim, King Crimson, Kraftwerk and The Monty Pythons for all those titles.

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