

# **RECORDING THE FLAMENCO GUITAR**

**By**

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## Introduction

Welcome to recording 101. Our objective is not to overwhelm you with all the technicalities of audio recording, rather give a sound overview of the various types of equipment and software available for home recording.

First, let's set one thing straight: we don't need to spend thousands of dollars to achieve remarkable quality recordings! Like most every item in the high tech world, prices are coming down, and quality is going up at a rapid pace. With our desktop PC, and some simple add-on equipment and software, we can achieve results that would be unheard of in major recording studios not that many years ago. We will be focusing primarily on hard-disk recording to our PC, but most of what we will cover will be of value for other recording methods.

Before you start loading up your computer with expensive goodies, it's wise to first determine your recording objectives. Do you plan to make simple recordings for practice purposes? Do you plan to post files on Foro Flamenco for critiquing? Or do you plan to go all out and create a demo CD? Each will demand a different level of system, and in turn expense. If you are just starting to purchase equipment, hopefully we can guide you in the right direction to achieve your objectives and not break the bank. Remember we're here to play Flamenco, not to be recording engineers.

In the pages to come, we will be giving you a brief overview of what to look for in equipment and software, followed by set-up examples, ranging from basic, mid, and more advanced systems. Even at that, the higher end systems should set you back no more than a \$1,000, plus the cost of the PC. Trust us on this, you can spend thousands and thousands of dollars on equipment, and not be able to hear a great difference between the \$1,000 setup. Buying recording equipment is somewhat like buying stereo playback equipment. After you reach a certain price level, you have to spend thousands more for minimal improvement.

Most recording engineers will agree that recording the acoustic guitar is very difficult, but flamenco guitar brings added challenges. As we all are aware, the flamenco guitar can be a wild beast with a very high dynamic range that sometimes needs controlling. Let's now take a look at the various hardware and software.

We have limited our examples to hardware, software and equipment we own and use. There are many equivalents to our examples.

## Equipment Overview

### Microphones

Microphones are likely the most debated subject in recording. Just look on-line or walk into an audio store and you will soon have information overload.

The two types of microphones we will use most in recording the flamenco guitar are “Dynamic” and “Condenser”. Most stage performers use Dynamic microphones for their incredible durability. Stage mics get dropped, thrown, kicked and everything else you can think of. With the durability of dynamic’s, comes a loss of sensitivity and frequency response. The Shure Company is world famous for their moderately priced dynamic stage mics. Most every garage band, along with professional musicians and recording engineers, own several Shure SM58 (\$100) or SM57 (\$80) microphones. Although not the best choice for our needs, dynamic mics will do a pretty good job at a very low cash outlay.

To really achieve studio quality, we need to step up to a Condenser mic. Condensers can be quite fragile, and should be used in the studio only. Condensers offer much more sensitivity and frequency response than Dynamic’s. But, one of the most important aspects of condenser’s is very low noise level. Remember, every piece of equipment, software and cable in our system adds noise. Rule number “one”. You must control noise, to produce good recordings. Microphones produce what is called “self noise”. For obvious reasons, the lower the amount of noise a microphone produces, the better.

Condenser microphones are made in three different types, or what is called “capsule size”. Inside the microphone is the capsule, or disk, that range in size from small, medium, and large diameter. Each has its positives and negatives. In general, the larger the capsule, the flatter the frequency response, but comes a negative of over boomy basses if not positioned properly. All in all, a medium or large condenser is a great workhorse in the studio.

In just the last few years, manufactures have brought many fine Condensers to the market at unheard of prices. New methods of construction have reduced prices dramatically. Pretty good quality Condenser’s are now being made available for as little as \$70 to \$100. The sky is the limit. You can spend as much as four or five thousand for top of the line studio Condenser’s. You will likely need to be at about \$200 on up for top-level performance. A couple of very highly rated mics in this price range is the Rode NT1 (large capsule), and the Audio Technica AT3035 (large capsule). Most Condenser microphones need a source of external power called “phantom power”, that is usually supplied by a mixer console, which we will cover in a bit.

One thing to look for in the purchase of a Condenser microphone is what is called a “Cardioid Pattern”. The term Cardioid refers to the pickup pattern. Think of the shape of a heart (thus the name cardioid). All this means to us, is a cardioid microphone will pick up much more from the front, rather than the back. This is very important for most of us home recorders. Most of our rooms are far from ideal acoustically. A cardioid microphone will allow us to somewhat direct what does and does not get picked up, helping to reduce unwanted room reflections and noise.

Just as important as the choice of microphones, is the placement of the microphone in our recording studio. In most cases we will be using what is called “close miking”. The term “close miking” refers to the distance we place the microphone from the guitar. Most “close miking” is done by placing the mic twelve to eighteen inches from the guitar. Close miking helps reduce unwanted room reflections and noise. The downside with “close miking” is a rather acoustically dry sound. This is OK, as we can enhance the recording to add life to it by utilizing a bit of compression, equalization, and reverb (more on this later).

The three most common microphone placements for the flamenco guitar are at the sound hole, behind the bridge, and at or near the twelfth fret. You would think, because most of the sound comes from the sound hole, this would be the best position. With some microphones it may very well be, but “large capsule” condenser mics can have what is called “proximity effect”. By directing the mic at the sound hole (proximity), you may experience an undesirable, boomy sound in the basses. You should experiment with various placements to see which you like best. As a rule, placement at the neck will reduce bass and enhance the trebles. Good results may be achieved by placing the mic at the neck, but directing it back towards the sound hole. You just need to experiment.

### **Preamps**

If you are using an inexpensive dynamic microphone, you may be able to plug it directly into your sound cards “Mic In” jack, and get an OK recording without a preamp. If you are working with a condenser mic, you will need to run it into a preamp. Condensers have very low output that need to be boosted before it gets to the sound card. With a condenser, you will either run it into a stand alone (dedicated) preamp, or the “MIC IN” jack of a mixer board. Most mixers have built in preamps, although of questionable quality, that will get you by to keep cost down.

In our opinion, a good preamp is one of the best places to put your money. A good preamp won't make a bad mic good, but can dramatically enhance a quality mic over a mixers built-in preamp. Many preamps that are coming on the market also have built in “Limiter” capabilities. Digital recording is unforgiving in regards to distortion. A “limiter” will automatically reduce the preamp gain, eliminating too hot of a signal to reach the recorder. Digital recording requires a very hot signal going to the recorder. You should set your preamps gain controls to near distortion level.

In recent years, we have seen a major movement back to the old analog, vacuum tube systems, which were the norm many years ago. Many people believe the tone quality of digital recordings to be almost “too” clean, or “too” sterile sounding. It's ironic when you realize that vacuum tubes actually add distortion to our recordings. Now we've been saying up to this point that we want to reduce distortion at all cost, and now we are going to add distortion! For whatever reason, the human brain tends to like the type of distortion that vacuum tubes add to a recording. If used sparingly, tubes tend to warm the tone of our recordings. In our opinion, flamenco guitar recordings can be greatly enhanced with the use of a tube preamp.

Most of the lower cost tube preamps are actually hybrid units, using solid-state circuitry, with the addition of tubes to add the desired distortion. ART and Berhinger produce some very fine low cost tube preamps, and in some cases with built-in limiters. Following are two audio clips recorded with a built-in mixer pre amp and with a tube pre amp.

[Mixer Preamp](#)

[Tube Preamp](#)

## **Compressor/Limiter**

If your preamp does not have a built-in limiter, a combination hardware (Compressor/Limiter) can be placed after the preamp to serve the same purpose. Many compressors also have a “limiter” circuit added to help stop signal peaks from reaching the sound card. We will be covering the use of compressors later.

## **Mixer**

The mixer, or mixer board, is the next step in what is called “the chain” or signal path. The first thing a mixer will do is supply phantom power to our microphone if needed. The mixer will do just what it says it does. If you are recording from multiple sources (two mics.etc), the mixer will combine these signals to be sent via the “Master Out” to the sound card. Although not usually advised, you can also alter the tone (equalization or EQ) of the signal on the mixer. In most cases it is better to not alter the equalization on the mixer, as once you record an altered signal, that’s it. Once the recording has been made, if you want to utilize equalization you can do so inside your recording software, preserving your original copy.

If you are using a hardware reverb unit (rather than software), you can use the “Auxiliary Send or Out” outputs from the mixer to the reverb unit, to be returned from the reverb to the “Auxiliary In” on the mixer.

As we have mentioned, the built-in microphone preamps inside most mixers are of questionable quality, but will give good results before upgrading to a dedicated preamp. One of the main purposes of the mixer is to adjust the volume levels going into the sound card. As we have mentioned, you want to maintain as hot of a signal at all stages as possible without clipping (distorting) the signal. You should adjust the volume levels on the mixer to produce a signal on your “Recording VU” meter that stays at the top of its range, without signal clipping.

## **Soundcards**

Like so many things, Soundcards come in a wide price range. Also, price does not always dictate quality. Use what you have or can afford. Let your budget be your guide just like any other purchase.

Some motherboards have built in Soundcards and these will be fine for some folks. Many motherboards will allow more than one Soundcard being plugged in at the same time while others will not.

The SoundBlaster 16 is an old timer and yet can do a good job of recording audio. Making a choice of which card to use depends upon your requirements and how much you have to spend. A good recording can be made with an inexpensive Soundcard. Good is a subjective term and may or may not please others. Remember you are trying to please yourself.

For the purpose of this discussion, we will break Soundcards into two classifications. Examples will be given of those cards we have used. Later recorded examples will be available for you to access so you can compare the results of some of the equipment and recording techniques we are discussing.

1. Soundcards
2. Professional Audio cards.

Examples of Soundcards:

1. ESS Sound (motherboard)
2. SoundBlaster 16
3. SoundBlaster Audigy
4. SoundBlaster Extigy USB

Examples of Professional Audio cards.

1. M-Audio 2496
2. M-Audio Delta 44
3. M-Audio 1010

There are many Soundcards on the market so you should do your homework. Be sure of the return policy of the store you purchase equipment from.

Check the specifications and requirements of the Soundcard to be sure they are within the limits of your computer.

## Software for Audio Recording

### Sound Recorder

Windows PC's come with a little program under Accessories – Entertainment – Sound Recorder. This program is very limited, but will allow you to record up to one minute of audio.

Comments: It meets the needs as a simple recording utility without the ability to edit.

### Magix 5.0

This program is relatively easy to use and can be purchased for about \$10 from places like Guitar Center. It also has a few built in editing tools.

Comments: Magix 5.0 is limited to Windows 98 machines and is not bad for making recordings of more than one minute, which it does well.

### Magix 7.0 Studio Deluxe

There is also a Magix 7.0 without the “deluxe” features. Also, you can add additional effects software packages from different sources. You can do multi track recording by recording one track, saving it and then recording additional tracks, one at a time. It is a 16 bit full duplex program.

Comments: The deluxe version is not as easy to use as version 5.0 but it does have many nice features for editing such as Compression and a long list of others.

<http://www.magix.com/select.html>

The product numbers have changed to 2004.

### N Track

A free demo version is available from the Internet. This is a shareware program and if you want to register it and use all its features it will cost \$45 for the 16 bit version and an additional \$24 for the 24 bit version. Add ons such as Compressor and Equalizer (EQ) are also available for \$20 each. Simultaneous multi track recordings may be made. This will allow multiple microphones and additional sources to be recorded live.

Comments: This is an easy to use program and gives very good results.

<http://www.fasoft.com/>

## **Cubase SX**

This is a professional software-recording package with many features. It is meant for the serious amateur or professional. There is little that this program cannot do in the world of software recording and editing.

Comments: It takes time to learn how to use this program. If you do not use this program on a regular basis you will have a problem remembering how to use all its features. If you make your living with audio recording this is a great program. Also if you are a serious amateur this program is worth looking into.

<http://www.steinberg.net/en/index.php>

## **Format Conversion**

Software packages have add ons or built in conversion features and functions. To convert a Wave file (common PC audio format) into a file that can be reduced in size to send over the Internet, MP3 is very popular. A good external program to convert Wave to MP3 and MP3 to Wave is db Power Amp. It is free and works very well!

<http://www.dbpoweramp.com/>

## **Monitors, Speakers and Computers for Audio Recording**

### **Monitors**

Sometimes called reference quality speakers, monitors are used in recording studios to reproduce sound accurately. One well-known manufacturer of monitors is Tannoy. For the serious recording studio, monitors are a must. We would not suggest monitors for the average user as they are very expensive. If your professional career depends upon audio recording you will use monitors! Otherwise a good set of speakers will be all you require to listen and enjoy sound produced by your computer.

### **Speakers**

Computer Speakers come in many flavors. Let your pocket book be your guide. You can have acceptable speakers that came with your computer or upgrade to something more expensive. The choice is yours.

### **Earphones**

Earphones can be very useful for recording and editing. Many types are available for this purpose. Earphones are not meant to replace monitors or speakers.

## **Cables**

Use the shortest length possible. If your budget can afford it buy the best cables you can. This will help reduce the pickup of unwanted radiation and improve the signal strength while reducing noise, from one piece of equipment to another.

## **Computers**

This discussion is limited to PC's and does not include the Macintosh. Just about any PC can record sound. The average computer may come with a sound card build into the motherboard. Some motherboards allow additional audio cards to be used and others do not. As an example there is one series of motherboards that has a simple jumper that is removed to allow one or more additional audio cards to be used.

A good soundcard can make a great difference in a computer audio recording.

## **Digital Audio Workstations**

Some PC's are designed to record audio and are optimized for that purpose. Such a computer is called a Digital Audio Workstation (DAW).

Technology changes so rapidly it is hard to keep up with what is current. Here are some things to look for in a DAW or to improve your present computer.

1. Motherboard Intel D875PBZLK W/800 MHZ FSB (Front Side Bus)  
(512KB L2 Cache) with 2.8 Ghz Intel CPU
2. Hard disk - Western Digital 80GB ATA/100 7,200rpm with 8MB Buffer for Audio Data (AMS Removable HD Case)
3. Memory - 512MB (2 x 256MB) PC3200 DDR
4. CD - Plextor 48X IDE CD-RW
5. Chassis:
  - Antec SONATA (#AN08) With 380W Power Supply
  - Zalman CNPS7000 ALCU Ultra Quiet CPU Heatsink
  - Zalman Fanmate Speed Controller
  - Round ATA 133 24" 2 devices IDE Cable

A DAW attempts to optimize audio recording by having a system that minimizes system noises, buffer data to improve CD recording, and eliminate problems such as IRQ conflicts.

## Equalization (EQ)

Equalization or EQ is the process of altering the tonal qualities of your recording. EQ'ing is typically done at the mixer board or within your recording software. We believe it is advisable to not alter the EQ settings on your mixer board. Remember, the mixer is placed before the sound card. By altering the EQ at the mixer you have recorded these alterations, which may be difficult to undue. Better, in our view, is to make your EQ alterations within your recording software. N Track software offers a very user friendly equalizer plug-in for about \$20.

When altering the tonal qualities of a recording, two methods are normally employed. The first is what is called “additive”. Additive refers to raising the volume of a specific frequency, or band of frequencies. The second method is called “subtractive”. Subtractive is the opposite of additive, in that you will reduce the volume of various frequencies.

With an “additive” process, if we want to boost a particular tone, we will raise, or increase the sliders on our EQ board. The negative with utilizing an “additive” process, is the potential for increasing background noise. As a general rule, it is advisable to utilize a “subtractive” method if possible. In a “subtractive” method, we will reduce the undesirable frequencies, which in effect, increases the preferred frequencies. In other words, if we want to boost the bass and trebles, we would reduce the mid range frequencies, which effectively boost the bass and trebles.

The rule of thumb with EQ, is “a little goes a long way”. Here are three sound clips to give you an idea what various levels of EQ sound like.

- 1) No EQ
- 2) Modest levels of treble and bass boost
- 3) High levels of treble and bass boost

## Compression

Compression is a very useful tool in recording the flamenco guitar. As we are all aware, the flamenco guitar has an incredible dynamic range, versus its more sedate sister, the classical guitar. In a live performance, we all are aware that the dynamics are what we desire. But, in a recorded environment, the flamencos dynamics may need to be tamed a bit. In an uncompressed recording, we will likely hear major shifts in volume from soft passages to all out rasgueados. Years ago, recording engineers would attempt to level out these volume variations by doing what was called “riding the faders”. The engineer would literally have his hands on the volume controls (fader), and would attempt to reduce the level of high volume passages. As you can tell, this method left a lot to be desired.

Modern compressors act in the same way as the recording engineers did, but instantaneously. Literally, the instant a passages volume reaches a preset level called the “threshold”, within milliseconds the volume is reduced by what is called the “ratio”. “Ratio” refers to the amount of compression that takes place once the level exceeds the threshold level. The higher the “Ratio”, the more compression takes place. A ratio of 1:1 would be zero compression. A level of 20:1 would be at the extreme. Ratio simply means that for every unit of volume (dB) exceeding the threshold, the volume will be compressed by the ratio setting (1:2, 1:5, etc.). We will normally utilize ratios from 1:3 to as high as 1:8, to help tame the high dynamics.

One of the problems with over compression is raising the noise level (noise floor) in the low volume passages. In many old recordings that have been digitally remastered, you can plainly hear the increase in volume (and noise) when the performer reaches a quiet section.

Compressors come in hardware and software versions. Most of us will be utilizing software compressors within our recording software. Following are sound clips of compression.

- 1) No compression
- 2) Modest compression
- 3) High compression
- 4) High Compression with high noise floor

## Reverb

Reverb, or reverberation units are what gives our recordings the final polish. Most of the recording we will be doing will be in a very dry, poor acoustic environment. Reverb will add the sense of depth and space to our recordings. Reverb can be achieved with hardware as well as software. Most of us, due to cost and convenience, will utilize software reverb inside our recording program.

It's likely that more professional recording studios use reverb equipment from the Lexicon Company than all others combined. In the last few years, Lexicon has brought to market very reasonably priced hardware units that have the same reverb qualities of more expensive units, without many of the bells and whistles.

The objective of reverb is not to overpower the recording, but rather to enhance. Many professionally recorded CD's tend to utilize excessive levels of reverb. The engineers' adage seems to be "if a little is good, a bunch is better" when it comes to reverb. We believe reverb, if used sparingly, can add dramatically to an otherwise dry recording.

The problem we have encountered with many software reverbs is a preponderance for very high default settings. You will likely achieve more pleasing results, by reducing the default level settings. We have also experienced what is called "a long reverb tail". "Reverb tail" refers to how fast the reverberations decay or die down. Again, we find most software reverbs have too long of a reverb tail, which tends to muddy our recordings. Some software reverbs allow you to adjust the "room size", which will reduce the reverb tail. One of the best methods to determine the characteristics of each reverb is to play a cord and instantly mute it. This method will help you adjust the reverb settings. Following are three sound files recorded from N Track reverb. Note how each has its own characteristics.

1) [N Track Bright Room reverb](#)

2) [N Track Large Hall reverb](#)

3) [N Track Large Hall 2](#)

Another issue with reverb, is the coloring of the sound. All reverbs will alter the tone in one way or another (good or bad). Following are four sound files recorded from N Track reverb.

1) [N Track Bright Room reverb](#)

2) [N Track Large Hall reverb](#)

3) [N Track Large Hall 2 reverb set at 50%](#)

4) [N Track Metal Tank reverb](#)

## Hooking it all up

Now that we have all the parts for our system, let's hook them all together. One of the basic rules in cabling is, buy the best you can, and keep them short. OK, here's the steps:

- 1) Microphone to preamp. Microphone will go direct to soundcard "Mic In" if you are not using a preamp and mixer. Microphone will go to "Mixer, Mic In" if are not using a preamp.
- 2) Preamp "Line Out" to the compressor/limiter if you are using one, otherwise, direct to mixer "Line In"
- 3) Mixer "Line Out or Master Out" to soundcard "Line In"
- 4) If you are using a hardware reverb, you would normally use the "Auxiliary Send" on the mixer to the "In Line" on the reverb unit. From the reverb unit "Line Out" to the "Auxiliary In" on the mixer.

The important points are to keep your cables as short as possible to reduce the introduction of noise. Also, if you are miking direct to the sound card you will plug into the "Mic In" on the card. If you are running a preamp (dedicated or mixer), you will plug into the "Line In" on the sound card.

Another application of the Sound Cards Line Input is to make a recording on your computer from any device that is line level compliant, such as a CD player, tape recorder, etc. You can convert formats and edit with software. We suggest you do this with material that is not protected by copyright.

## Examples

Following are examples of four setups ranging from very basic to more elaborate.

### Basic setup

#### Basic Level One

The "Basic" setup utilizes a low cost dynamic microphone plugged directly into a modest quality sound card that is built into the motherboard utilizing Windows Sound Recorder. 8 bit/22.05 kHz

[Windows recorder sound clip](#)

#### Basic Level Two

The "Basic level two" utilizes the same setup as above with the addition of upgraded recording software. These clips have been recorded with N Track software.

- 1) [No editing](#)
- 2) [With compression only](#)
- 3) [Final mix with EQ, Compression, and Reverb](#)

## **Mid Range**

The “Mid Range” setup utilizes a modest priced microphone, mid range sound card with the addition of a preamp and N Track software. 16 bit/41.4 kHz

- 1) No editing
- 2) Limiting
- 3) With compression only
- 4) Final mix with EQ, Limiting, Compression, and Reverb

## **Higher End**

The “Higher End” setup utilizes a quality condenser microphone, fed to a tube preamp with limiter, into a high quality sound card recorded with N Track software. 24 bit/96 kHz

- 1) No editing
- 2) With EQ only
- 3) With compression only
- 4) With reverb only
- 5) Final mix with EQ, Limiting, Compression, and Reverb

## **The Art of Editing**

Now that we have the basics down, lets briefly cover the final editing process. The reason we say “briefly”, is the fact that editing could, and does, fill many volumes. Many fine schools across the country offer degree programs in audio engineering, so the scope of this article will be the tip of the iceberg.

Granted, we must be familiar with the mechanics of editing, but the reality is, the editing process in many ways is “Art”. In a professional recording studio, the recording of the music is likely to be the fastest and easiest part of the whole process. Its been said that Sabicas made the majority of his recordings with one take. Once the recording is done, the real work begins. In many cases the recording for a CD is done within days, only to be released to the public after months and months in the editing stage.

We will now take you through the process that was used to record the above clips in the “Higher End” setup.

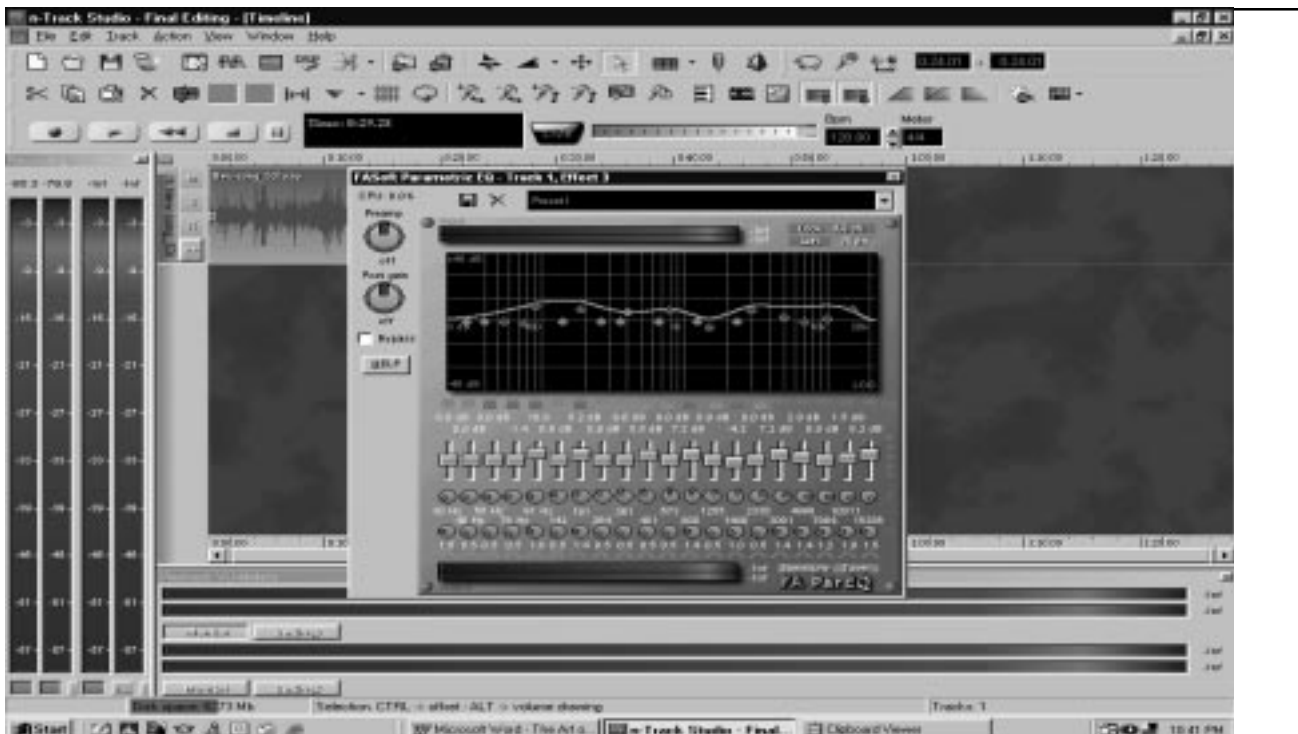
A single (mono) Audio Technica (AT3035) large condenser microphone was fed to a Behringer T1953 Tube Utagain preamp. The preamp was fed into a Behringer AutoCom Pro XL MDX 1600 Compressor and then into a Behringer Eurorack MX 602A mixer. The mixer was fed into an M-Audio MD 2496 sound card. Recording software was N Track, with N Track Reverb, Equalizer, and Compressor Plug-ins.

This recording was close miked with the microphone placed at the twelfth fret, angled back toward the sound hole at about eighteen inches from the guitar. The tube preamp was set to near clipping level. The tube phase of the preamp was set to “Maximum” warmth. The compressor’s “Limiter” circuit was used to keep the signal from clipping going into the sound card. No compression was used at the recording stage. The equalizers on the mixer board were set to neutral (no gain or boost). The “Main” fader on the mixer was used to control the signal level to the sound card, which was adjusted to near clipping level on the N Track recording VU meter.

If you listen to the first clip (No editing), you can tell that the recordings dynamics are too high. That is, the low volume sections are too low, and the high volume areas, too high. It also is very dull as to trebles and bass and needs enhancements. You can also tell that the recording is very dry, with very little natural room reverb, which also needs enhancement. As this recording was done with all N Track software, all references will be to N Track products. This recording was done at the 24-bit level. You must have a sound card that can record at this level, and purchase the 24-bit upgrade from N Track.

This file was saved to the hard drive before any editing was done. This is very important! You always want to keep an unedited copy saved...just in case.

The first stage in editing of this file was done with the N Track “Parametric EQ” graphic equalizer plug-in that is available for an additional \$20. Following is a “screen shot” of the equalizer settings for this recording:



Following are the approximate equalizer settings:

- 1) A 10 dB boost at 90 Hz
- 2) A 8 dB boost at 190 Hz
- 3) A 7 dB boost at 800 Hz
- 4) A -4 dB reduction at 1,500 Hz
- 5) A 7 dB boost at 3,000 Hz
- 6) A 3 dB boost at 5,000 Hz
- 7) A 1.5 dB boost at 10,000 Hz
- 8) A 8 dB boost at 15,000 Hz

Now, if you set your equalizer to these settings it won't look the same as the above shot. The reason is, you need to adjust the "band width" of each of the above frequency settings. If you adjust your equalizer as above you will see a bunch of narrow frequency spikes, that need to be widened. That's what the little knobs at the bottom of the equalizer are for. These knobs are called "Q" controls. The "Q" controls adjust the band of frequencies that will be boosted or reduced. Following are the settings for the lower (Q) knobs:

- 1) 90 Hz set at 1.8
- 2) 190 Hz set at 1.4
- 3) 800 Hz set at 1.4
- 4) 1,500 Hz set at 1.0
- 5) 3,000 Hz set at 1.4
- 6) 5,000 Hz set at 1.4
- 7) 10,000 Hz set at 1.0
- 8) 15,000 Hz set at 1.5

Leave all of the little green icons at the bottom as "peaks". These are used to set the type of equalization. The best thing to do is keep it simple with these settings. After you have all the above settings you should save them as a "pre-set", which you can do at the top of the equalizer, with the "save" icon.

You may remember in our discussion under the "Equalizer" section, we mentioned that it is usually better to "cut" frequencies rather than "boost" them. The primary reason for cutting is to keep the noise level down. When you boost frequencies you are introducing additional noise. This file was made with "boost" in most frequencies. Due to the incredibly low noise levels of the hardware components being used, the minor increase in noise at the EQ stage is acceptable, as boosting will give you additional tonal control.

The next step in editing was to add some reverb with the N Track reverb plug-in. This plug-in comes with N Track at no additional charge and is as good or better than most that you have to buy. The reverb that was used was "Large Hall 2". This reverb, with minor adjustments is very realistic. The only adjustment made to the "default" settings, were to reduce the "Room size" a bit, and set the "Wet" control to -13.0 dB. All other settings were left at the default. As above, these settings have been saved as a "Pre-set".

The next step was to add some compression with the N Track plug-in that is available for an additional \$20. This is a very nice sounding compressor, and is a must have, along with the N Track equalizer. Proper use of compression can really pull a recording together, reducing the dynamics to an acceptable level. The function of compression is to reduce level of the high volume passages, and increase the level of the low volume passages. The problem is, if you use too much compression, you literally squeeze the life out of your recording.

The setting that was used for this recording was the “Soft Knee” default, with no other adjustments. Be sure and click the “Compensate Gain” box at the bottom of the compressor. This setting will automatically increase the level of the low volume passages, to make-up for the reduction in gain of the high volume sections.

The next step was to use the “Fade” icons at the top of the N Track screen. Just highlight the beginning section you want to fade-in, then click the “fade-in” icon. The same process is used for the fade out. The last step was to convert the Wave file to the MP3 format for Internet distribution.

That’s all that was done with this recording, and as you can hear in the final version, was quite dramatic.

#### 1) No Editing

#### 2) Final mix

You can literally spend hours and hours making adjustments, with questionable results. One area that may be of interest is the editing of the recording on a note by note, or section by section. Play a bad note? Don’t record the whole piece again, just replace the bad note. The same can be done with a complete section. It’s interesting to note that many professional recordings are actually recorded in segments, to be later edited and assembled in the studio. A good engineer can replace one bad note and we will never be aware. One performer plays or sings off key. Again, no problem, simply raise or lower the offending pitch. You can do some pretty amazing editing with basic recording programs such as N Track. For more advanced features, programs such as “Cool Edit” allow much more control in editing. Cool Edit will allow import of a Wave file that was recorded from another recording program.

Following are equalizer settings for recording the flamenco guitar. These settings will give you a good starting point. It would be wise to save them as a “pre set”, then make additional adjustments to your liking, making additional “pre sets” as you go.

- 1) 100 Hz boost with a Q of 1.0
- 2) 200 Hz boost with a Q of 1.4
- 3) 800 Hz boost with a Q of 1.0
- 4) 1,500 Hz reduce with a Q of 1.0
- 5) 3,000 Hz boost with a Q of 1.4
- 6) 5,000 Hz boost with a Q of 1.4
- 7) 10,000 Hz boost with a Q of 1.0
- 8) 15,000 Hz boost with Q of 1.0

## Some Final Thoughts

- 1) Always save your original recording before you edit.
- 2) Your computer will lock-up 55 ½ minutes after you start editing, give or take 55 ½ minutes!
- 3) Save your work every few minutes.
- 4) Repeat steps (1) and (3)
- 5) Don't take this recording stuff too seriously.
- 6) Have fun and post your recordings to Foro Flamenco

## Converting to MP3

Now that you have your recording and would like to share it on Foro Flamenco, your next step is to convert it to an MP3 file. Most recording software save your recordings to what is called a Wave file, which really is the extension (.WAV) to the file name. Recordings in Wave format will give you very high quality. The problem is, Wave files can be very, very large, which will likely take significant time to transfer via the internet.

The most widely recognized file format for Internet file sharing of music is called MP3 or MP3 coded files. Some very smart programmers figured out a way to compress a Wave file and still maintain reasonable quality. A two meg Wave file can be compressed to a 250K file and still sound acceptable. Some recording software have MP3 conversion programs built into them. If not, you will need to obtain a conversion program. These programs are readily available free or at a small cost.

We are concerned with recording in Wave format and conversion to MP 3 to send audio files over the Internet. This is a compromise but it is not a serious issue except for those who have “golden ears”.

Controversy exists about which bit and sampling rates should be used. Some people like to record at 24 bit and 96 kHz and then mix down or convert to 16 bit 44.1 kHz or 48 kHz. CD quality is 16 bit at 44.1 kHz. The argument is that MP 3 is a lossy conversion methodology and cannot compare to CD quality regardless of the bit and sampling rates employed. In the end you should be trying to please yourself.

The higher the bit and sampling rate the “greater the fidelity” is a general rule. Along with increased bit and sampling rate you must include greater file size. The question you should ask yourself is what is more important and acceptable to you? Greater fidelity and smaller file size are in conflict with each other.

There is also an argument by some about the mathematics involved in mixing down from different standards to another value. Does it really work as advertised? Or is there a mismatch when going from 24 bit 96 kHz to 16 bit 44.1 kHz? Mathematics may tell you one thing and your ears something else.

MP 3 attains almost a 10 to 1 compression of the original wave file and is dependent upon the bit and sampling rate. That means for a given wave file size you can achieve a file almost 10 times smaller when converted to MP 3. You will lose some audio quality but that may not be an important issue. The ratio saved is dependent upon the bit and sampling rates used. The higher the bit and sampling rates used for recording the less the savings of file size.

It is said that humans can hear as high as 20 kHz. The sampling rate according to the Sampling Theorem should be twice as high as the highest frequency of interest. Because filters are used to limit all frequencies over 20 kHz in this example, a minimum frequency of 44.1 kHz was chosen for CD recording.

What do you buy with a sampling rate of more than 44.1 kHz? You could argue the point but high fidelity (accurate reproduction of the original signal) is the goal.

What about bit rates? The higher the bit rate the higher the audio quality. A higher bit rate allows recording more information within a given spectrum.

Now all of this may be more than you really wanted to know. It also just begins to scratch the surface of the subject. In the end you should try different combinations of your hardware and software and analyze the end results.

## **Stereo, Taking Your Recordings to the Next Level**

Now that you have some pretty good equipment and happy with your mono recordings, you might consider moving up to stereo. Stereo opens up a whole dimension in recording as well as additional issues. For that “3 D”, “live feeling”, nothing can compare to stereo. Let’s face it, we hear in stereo, so why not record in stereo.

Basically, everything we hear is in stereo. When a car horn honks, the sound travels to each of our ears at a slightly different time. Our brain has the amazing ability to determine where a sound comes from, from this slight time delay. Stereo recording works in a similar fashion. By placing two microphones apart from each other, they will in turn receive a sound at a slightly different time. When these two signals or channels are played back through speakers or headphones, our ears detect this miniscule time difference, which gives us the stereo effect.

All of our discussions to this point have dealt with mono or monaural recording. In mono recording we only need one microphone, one mic cable, one mic preamp, and one limiter. We need to double all of these components for stereo recording. If your budget can afford it, you might consider initially buying a two-channel preamp (and compressor/ limiter , if not built in) if you plan to record in stereo in the future. You will also need a sound card that can record two separate channels at the same time, so think ahead before buying.

There are several stereo miking methods being employed, but we can boil it down to two for recording the flamenco guitar. The first is called X-Y miking. With X-Y miking we place two matching microphones very close to each other, pointed at around ninety degree angle to each other. This method works well to reduce what is called “phase” problems. When a sound reaches two microphones at different times, the signal can be out of phase, which can cancel desired signals. Phase problems can reduce bass, or eliminate the sound in the center of the stereo field. Many microphone manufactures produce stereo microphones, which contain two capsules in one housing

The second, and most widely used method, is called “spaced pair” miking. Two matching microphones are placed in front of the guitar at a distance apart. Normally one is pointed at or behind the bridge, and the other at the twelfth fret (possibly angled toward the sound hole). To reduce phasing problems you should adhere to the 3-to-1 rule. The 3-to-1 rule states that you should maintain the spread of the pair at least three times the distance the microphones are from the guitar. In other words, if the microphones are 12 inches from the guitar, they should be 36 inches apart or more.

Another technique to cancel phase problems is to use a pre amp such as the ART TPS, which allows the phase of either microphone to be reversed.

A couple of things you want to do is keep the settings the same on both channels of your preamp, limiter and mixer. The second is to turn the left channel “Pan” hard to the left and the right channel hard to the right on your mixer. This will assure you of keeping a wide stereo field.

Following are two files recorded in mono and stereo.

- 1) [Mono](#)
- 2) [Stereo](#)

## **Burning to a CD**

You may also want to save your music to a CD for sharing with friends. The process of saving to a CD is called “burning”. As with MP3, many recording programs will have built in capabilities to burn direct to a CD. If your recording software does not support direct burning, you will need to obtain CD burner software. Again, as with MP3 software, CD burning software is readily available free or at very little cost.

Of course you will also need a CD drive in your computer that allows you to burn CD’s. These types of CD drives are called “Burners” and have become very inexpensive in the last few years.

So there you have it. We hope we have been able to give you some guidance without confusing you too much. As you have seen, very nice recordings can be achieved with a minimal outlay.

OK, it’s up to you. Let’s all share our love of flamenco by sharing our recordings.

It always helps to go to [www.google.com](http://www.google.com) and search for answers to questions you may have.

Pat and Tom

## Glossary

- **Amplitude** - The distance above and below the centerline of a waveform
- **Dynamic Range** - The lowest and highest signals volumes of a signal
- **Frequency** - The rate at which a soundwave repeats any number of cycles within one second.
- **Headroom** - Maximum amount of level a system can take before you cause distortion. A system can be an amplifier, a pre-amp, digital or audio tape, a console etc.
- **Hertz** - A frequency measurement unit (in cycles per second)
- **I/O** - This stands for input/output. It is the most basic part of a console and refers to the input output module. When you see that a console is 32x8x2, the number 32 is the number of I/O modules.
- **Masking** - When a louder sound covers up a softer sound of similar frequency.
- **Noise Floor** - The inherent noise of a device
- **On-Axis** - If you envision a polar axis around a microphone, on-axis would be at zero degrees. Or at the front of the mic. This is the area where the microphone's frequency response is best. Off-axis is at the back of the microphone.
- **Semi-Tone**: Also called a half-step. It is an increment of pitch equivalent to one fret up or down on a guitar. For instance from F to F Sharp is a semi-tone.
- **Signal to Noise Ratio**: This is the relationship between the signal you're recording or bringing into a console or device and the inherent noise of the tape medium and/or device itself.
- **SPL** - Sound Pressure Level. The measurement taken of audio level at one point. Measured in decibels.
- **Transient** - A short duration, high level peak. Such as a hand clap or snare drum hit.
- **Unity Gain** - In the simplest terms it means, what you put into this gain stage (fader etc..) you're getting out the same level. You're not adding or subtracting from the signal.
- **Velocity** - The speed of sound in air, 1130 ft/sec at 70 degrees F.
- **Wavelength** - The actual size of a soundwave. Low frequency waves are larger than high frequency waves and are omni-directional. High frequency waves are uni-directional.
- **x-y technique** - The stereo miking technique where two microphones' capsules are placed in close proximity and are splayed at 90 degree to 110 degree angles as in the x-y axes on a graph.
- **3-to-1 Rule** - Stereo mic placement technique that states: For every unit of distance away from the sound source, the mics must be at least 3 units of distance apart. This applies only to equidistant close miking of a sound source.

Notes: