How to get good sound quality before mastering

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Introduction:

In the tape prepare section, it explains different types of media, which use to record and submit for mastering. Now you want to produce a new recording from scratch, what can you do to maximize the audio quality and provide the best format for mastering session? This article provides some easy to understand procedures and theories, which may help you to create a good recording.

Label your tapes correctly:

When you finished a recorded session and send the tape for mixing, the tape is called “Session Tape”. Session tape is more often in multi-tracks formats in today recording standard. This format is then set to mixing session. This mulitchannel tape (or more often file base CD-Roms/DVD-roms) is usually in one of the following format:

(1) ADAT / Tascam DA88 / Analog 1” 16 channels/24 channels / Sony 3348/3324
(2) Protools / Logic / Nuendo / Pyramix interleaved wav or aiff files with EDL

The tape, which after mixing is called “Mixed Tape”. Mixed tape should be in stereo format or 6 channels for surround production. Many people, even audio engineers call the recorded and mixed tapes as “Mastered” Tapes. This creates confusion if the tapes need mastering procedure or have already been mastered. If you want to know more about differences between recording, mixing & mastering. The article “What is mastering ?” will be up in this site very soon.

Audiophile recording / mixing philosophy: Less is more.

When you are doing recording or mixing, try to minimize your signal path. If you are using a mixer, bypass all the un-used circuits through the signal path. Each of audio processing like EQ and compression add an active amplifier stage, which degrade the transparency of the recording. If allow, use outboard gears instead of using the mixer. Go direct from microphones to microphone preamps and direct record on tape. The word “DIRECT” is the key. For recording, select what you want and capture the cleanest signal is the first goal. This should be an open secret for all legendary engineers. If they have enough outboard gears, they do not use the mixer no matter it is a Amek or SSL. More often, they try to move the microphones in the studio rather using processing during recording. If you want to make recording, nothing more important than invest for good microphones and good microphone preamps.

Neumann has produced many great legendary microphones like M149 / M150 / U87
Sony G-800 has been used in many top pop music studios for vocal
The Brauner and Lawson microphones are new high end studio microphones.
We use Avalon Design M5 microphone preamplifiers for recording.
FM Acoustic, Millennium Media HV-3D, Forsell M2B are count as the most transparent microphone amplifiers.

For audiophile productions, always use the minimum microphones and least amount of signal processing. A pair of excellent stereo microphone is ideal for X/Y, MS or Blumlein recording. Brauner Tube VMS-1 is the most beautiful creation in the market today.

Pearl has MS-60 stereo microphone.
AKG legendary C-24 Tube microphones is often see in audiophile recordings.
DPA (B&K) is well known for stereo recording with its omni (4003) & cardioid (4011)
Royers stereo miking are used in our first HK Jazz production

How about the mixing stage?

For mixing, most studios use Protools for its multi-tracks flexibility and lots of plug-ins from many 3rd party manufacturers. We found that the best sounding mixes are came from analog consoles. (Of course there are exceptions!!) Our audiophile minds turn us to require a mixer when does not only mix many tracks into two tracks, it must be able to pass along the sonic and transparency of the original recording with very high headroom. The Vintage Neve console has a special sounding on its own which usually benefits the mix. Manley has launched its mixer, which mix 16 tracks into 2 tracks with audiophile signal path and tube output. In Hong Kong, legendary producer and engineer Michael Au uses Manley 16->2 mixer.

If we are going to build a room for mixing in the future, we will choose from Manley, Millennium Media or Fred Forsell design (Forsell Technology). Since 2002, there are more such mixing summing box appears to the market. The most well known items such as: SPL MixDream, Dangerous 2-Bus, Mixer etc. Dangerous Music is designed by Chris Muth who created all mastering console for Sterling Mastering facility. All of them share the features like huge headroom (+30dBu) and transparent audiophile signal path. They all are going to launch separate signal processing units to link with their mixer to become a full feature console. (You will need to get an automation system for these consoles).

Digital mixing is more ideal than analog since we just modify the data without any signal path. It is more likely adding numbers together, without the same type of signal degradation which provides by linking thousands of electronic components. But it is rare to find a digital console gives us the same sound quality from analog console. In our studio, we use very high precision (32bit or 64bit floating point) DSP for multi-tracks mixing to stereo. People in the industry are talking about the Pairs system from E-mu, Sony DMX-100 with 24bit/96kHz I/O and Sharc DSP for 32bit/40bit fix or floating point calculation. For higher end user, Sony Oxford produces very high quality sound from the recording mixed with it. What makes mixing with a real console better than inside a computer is the work flow and able to touch and adjust the mixing with human interface. It seems more often that new product - A console with works as only a DAW controller, all mixing and interfacing are happened inside the computer. The “ID” from Steinberg (Nuendo) and “ICON” from Digidesign are two most famous Intergrated Console at the moment. Of course there are some real consoles which also work well with DAW controller, SSL AWB900, System 5, Studer Vista. Modern audio production budget has dropped sufficiently, we will hardly see large format console in more places. In Hong Kong, two most interesting installations are Hong Kong Movie city, which designed by legendary acoustic designer Tom Hidley. And Cyberport with Studer Vista console.

Analog VS Digital?

Digital Audio has been improved so much compares to decades ago. The best digital recording sounds closer to the source than the best analog recordings from our opinions. In term of accuracy, best digital is the way to go. BUT, put in mind that working with digital audio requires a lot of focus in every step of the procedures, digital connections, DSP (digital signal processing) etc. There are tons of bad digital recordings, which lost all the tight sound images, depth, width and warmness. Those recordings sounds thin, harsh, without multi-dimensions. To avoid this happens, you must test all your digital gears, learn to use them properly, and not overdone the recording with them. We believe this is what all engineer doing everyday besides actually working on audio productions. We test all our new equipment and monitoring our signal path with scopes and testing equipment. We recently found one great digital mixing platform is Mobile I/O from Metric Halo. Combines Mobile I/O with Apple Powerbook can creates a powerful portable recording, mixing system. We have demo such system to more than 10,000 people in shows and playback hi-res 24/96 materials to external DAC.

Audiophile philosophy gives a good direction that if you use less processing, shorter signal path and the damage of the recording will decrease. Always try to use the microphone placement to fix the problems in the recording, you will learn to use less EQ or dynamics circuits. There are many audiophile productions out there which use NO signal processing at all....and they sounds wonderful~!

For CD mastering stage, 1/2” 30ips analog tape is still one of the best media because the mastering engineer can select to use analog processing (eg. tube equipment) to add the warm to the analog recording before the A to D conversion. Moreover, mastering faculty has mastering quality A/D converters which rare to find in the normal recording & mixing studios. Since A/D conversion is the most important factor for the quality of the digital audio, leave this process for mastering chain can improves the overall sound quality. To select analog or digital for recording depends on the studio setup, equipment and techniques. If you are familiar with analog equipment and have a good console to work with, stay in analog is the best method. If you have to work in digital, the following suggestions are worth to read.

How many BIT is enough? Why there are 24bit or even 32bit recording on CD? Is there any benefit? Compact Disc (Red Book CD) can only stores 16bit/44.1kHz data. The recording requires convert (re-dither) back to 16bit at the final stage even it was recorded on 20-24bit converters. By good re-dithering and noise-shaping techniques, we are able to create a recording, which has higher resolution than 16bit.

To determine how many bits the recording is, we use how many bit of A/D conversion that recording has recorded with. If a recording passed through a 20bit A/D converter, it is a 20bit recording. No matter how high resolution DSP (32bit / 64bit float, 48bit double precision) it has been processed, it is still a 20bit recording. From our point of view, the highest quality recording is now 24Bit, which has about 21Bit resolution. Why?

A 24bit converter has 144dB S/N which is almost impossible in the real world because of the thermal noise and temperature changes. The best 24bit A/D converters are around 130dB S/N, and mastering quality A/D converters have more than 120dB Dynamic Range. A proper A/D conversion requires not only S/N Dynamic Range measurement data. It requires the stability of the clocking, power supply design, and analog input stage design. If you are planning to purchase a high quality A/D converter, the followings are what we usually use in mastering studios.

Weiss ADC-1MKII, DAC-1, dCS 904, Elgar, dB Technology AD122-96 MKII, DA-924, HDCD Model ONE, TWO, Prism
We use both Weiss and dCS ADCs & DACs in PCM and DSD conversion. We found them are the best for us.

Digital Audio Processing can easily destroy the recording?

Analog processing does not work the same as digital audio processing, even though both analog/digital processing mean fade in/out, gain up/down, EQ, dynamics, reverb etc. Digital Signal Processing - D.S.P. DSP (Digital Signal Processing) involves in all digital audio processing. This means all DSP are algorithms, which calculate the digital data and modify it for the results. All DSP has a digital word length (or called “bit depth”), most of today DSP system like early Protools work in 24bit fixed point calculation, Sonic Solutions HD has double precision 48BIT data path. The PC NATIVE program, which uses CPU has 32bit floating point resolution.

There are debates about which calculation is better. 32bit-floating point is better then 24bit fixed point, but 48BIT double precision doesn’t mean better than 32bit float. Each of them has their own advantages/disadvantages, and depends on what kind of coding (function) does the DSP work. Daniel Weiss from Weiss Engineering told us the reason for them to choose Sharc from Analog Devices, is Sharc can code either in 40bit floating point or 32bit fixed point.

How long word length is enough?

It is hard to tell the real number. The longer word length has better quality from our experiments but using more DSP engine power. For serious high quality processing, DSP should contain 32bit and 48bit word length (double precision) calculation and output dithered to 24bit for the limit of digital I/O connection. On the Native world, algorithms should calculate in 64bit floating point and dithered to 32bit to store in host (Most PC can stores file in 32bit float) or dithered to 24bit fixed as well for digital output.

We talked with Glenn Zelniker from Z-system about the z-Q2 and the floating point calculation. He mentioned he has tried to increase arithmetic precision and found literally no measurable difference in SNR, THD+N, or any other performance metrics related to audible quality , but he also said that he will continue to research in this area and keep improving the wonderful Z-system designs.

For a 16bit CD recording, do we need more than 16bit word length for processing?

The answer is definitely YES. When you decide to you turn down even 0.01dB volumn of the digital source, you are performing a DSP calculation. To make a simplify example, the source has a volumn of 10.01 (2 decibels) and now you want to decrease 50% of the volumn. DSP will recalculate the 10.01 and divided it by 2 and gives 5.005 which is a 3 decibels output. If the calculation only works in 2 decibels, the 0.005 signals will be round off or cut away which means you will lose some signal in audio term.

This simple example shows the important role of the long word length. If the source is 16bit, the DSP should be at least calculate in 24bit, for better result, 48bit double precision is required. The above explanation gives us a good idea why the our CD recordings sound better in these years, besides the better designed A/D conversion, we are having higher precision DSP than before. For more information about fixed and floating point processing, please read another article “fixed VS. floating point”

Improper word length connections = Truncation

Truncation is a problem in digital audio processing. Especially many engineers are using DAW (Digital Audio Workstation) these days. They need to link many plug-ins as a DSP chain. When you are in this situation, you must test and check all your plug-ins and make sure they are work in correct bit depth.

Remember we received a mastering project from a local studio few years ago, the recording claims to use 24bit A/D conversion and all 24bit digital DSP for EQ, compression & limiting. When we listened to the sound of this mixed tapes, it was harsh and something wrong with it. I told the engineer and helped him to check his recording chain. Finally, we found out that the digital reverb, which in the 3rd section of the chain has 16bit truncation for the input & output. This means even his recording is recorded with 24bit A/D conversion, and performs with all 24bit DSP, once the final section of the chain has a “defected” 16bit output, it degrades the sound very much. One of the main reason that truncation becomes a big problem for digital audio recording, because you still hear the sound when a truncation is happening, you may not beware that there is a problem at all!

Few years ago, many engineers did not like the sound from the Protools, and finally found out the same reason that the mixing desk DSP are truncated to 16bit. This surely has been fixed by Protools, but still many engineers prefer to work with real mixing console, or the master class mixing engineers all went to AMD64 DAW with 32bit Nuendo system. To avoid truncation, you need to know better about your equipment or plug-ins. is it calculates in 24bit word length? If so, is the input & output of this DSP also support 24bit I/O? Some DSP calculate in 24bit but output 16bit only without a proper dithering. This decreases the precision of the data path. Be sure all your tools are working properly.

Digital Audio Level - How loud is enough?

How is the 0VU(+4) meter works with the dBFS digital meter ? For almost all commercial recordings, the answer will be “The louder, the better.” All producers and engineers want their recordings as loud as possible, to draw attention of the audience. But they do not understand that (or they do not care), their recordings can only draw the audience attention for the first few seconds, but it cannot last long. The real answer is actually on the opposite side, “The louder is the worse”. Audiophile community has a new theory in these few years. “The more you need to turn up the volumn of your pre-amplifier, that recording usually sounds better.”

For anyone who is going to make a recording, please do not try to make the recordings as loud as you can. This process should let the mastering engineer to decide and change for you transparently. The mastering engineer will match all the songs with the same loudness. (More detail on the article “What is mastering?”

For all audiophile & classical recording, use 0VU(+4dBm) = -20dBFS (Digital Meter) as standard. This means when the VU meter hits 0VU, it is actually -20dBFS from the digital peak meter. This gives 20dB dynamic range to work with. For most of the commercial recordings, they use -14dBFS as reference. For more details about this topic, we suggest you to read: Bob Katz’s AES Paper “An Integrated Approach to Metering, Monitoring, and Leveling Practices”

Mixed media - which one is the best?

For the most convenience and good quality media for CD mastering service, we suggest to send us 24bit (88.2kHz) *.wav file as standard. Disc based format has lots of advantage compares to tape based format. *.Wav or *.Aiff are universal sound files and can be open in almost all DAWs (Digital Audio Workstation). Our mastering studio has a high speed internet connection 24hours on for any incoming sound files. We found this is latest technology helps us transfer digital data by the internet and faster than shipping from the other side of the world. (eg. Asia)

When you are recording/mix your own recording in digital, some rules you may follow to create a better recording:

(1) Do not Fade in/out, change gain of the recording
(2) Use the least amount of digital DSP (EQ, Comp)
(3) Do not normalize/ limiting the recording and leave few dBs head room for mastering
(4) Only use basic edit to trim the sound file.
(5) List everything you want to change on a *.doc and save it with the 24bit *.wav files
(6) You have to purchase or rent a GOOD quality A/D converter and proper setup.
(7) Let 10 second recorded silence before & after the recording. Good Luck to your recording~!