Recording Guide

by Roger Nichols

It takes more than just plugging in a microphone and pushing a record button to make a good recording.

Every link in the chain is as important as the ones around it.

The following is a rough outline of the things you need to think about when recording and mixing a project. The explanations are short as they are only meant to jog your memory.

Setup:

How many people (musicians) will be in the recording room and how will they be arranged

What Instruments will they be playing and what special requirements need to be met

How big is the room (or rooms). If sharing an isolation room, consider grouping of instruments for least adverse leakage.

Isolation between instruments should be considered. Is some of what is being recorded going to be replaced (stand in vocalist, not the real solo, etc.) Determine how best to isolate the instruments (baffles, Tube Traps, blankets, foam, plywood).

Cables:

You can never have too many cables or adapters. All cables must have previously been ascertained to be in proper working order. Cables that have been previously suspect and checked to find nothing wrong should be labeled as such until they have successfully worked in a session. (Tthis is in case of a cable problem, the first cable to check would be a previously faulty cable.) Anticipate problems as much as possible.

Microphones:

Choice of microphones. What mics are available for the session? What mics are specifically requested by the client? Are there notes from previous sessions with the same musicians that pointed out a unique requirement or a mic that worked rather well in a particular situation.

Impedance must be matched. (Lo, Hi, Inline Xformer) Thin sounding microphones (reduced low frequency response) usually means that the impedance is not matched properly. Connections must be matched. (XLR, 1/4", DIN, Teuschel). Polarity must be matched (Pin 2 hot - Pin 3 hot?)

Phantom requirements must be ascertained. If mics are split to multiple consoles, such as live performances, only one console should provide phantom power. Make sure that mic splitter will pass phantom (some will not). If console does not provide the proper phantom voltage, use external pass through phantom power modules. If the microphone has it's own power supply, make sure that phantom is turned off to that mic, otherwise distortion and noise may result. (Guaranteed in some instances)

Can't have phantom on with unbalanced microphone.

Try direct box for synths and electric instruments. Try different direct boxes (they are like microphones and have a coloration of their own). Active (phantom powered) direct boxes may not have ground isolation capabilities and may cause ground loops (which result in a buzz or hum). Some consoles will let you pluginstruments in directly. (check for impedance matching.)

Pickups on acoustic instrument can be added in with the microphone sound.

Mic patterns must be chosen properly for the job. Understand proximity effect in microphones.

Microphone placement may not be the same each time you record in a similar situation. It may depend on the individual player or instrument.

Listen for reflections off of music stands (use foam or towel) when recording vocals. Listen for extraneous noises from squeeky chairs or rattling instruments.

Speakers and monitoring:

What do they sound like. Have you heard these speakers before in a different environment? Does the control room color the sound of the speakers so that you must compensate for that difference?

Placement of speakers in control room may effect the way they sound (experiment with different placements)

Try not to use speakers (instead of earphones) for monitoring in the studio during recording. The leakage will hamper an otherwise good recording. If this must be done, there are methods whereby two speakers are fed a MONO signal and placed out of phase. The microphone is then placed in the phase null between the speakers. Extreme caution should be taken when employing this method.

Use distribution system for multiple headphones, don't just parallel a bunch of headphones from the console or cassette machine headphone outputs.

"More Me" headphone distribution systems for individual mixes to each musician can help the recording process immensely.

Console:

Trim vs. volume control. Don't clip the input. In some situations it may be desirable to set all faders to "zero" and establish the initial recording level with the input trims. This provides an instant graphical representation if microphone levels change drastically during the recording (the fader is pulled down or up from it's reference)

Check the sound through the console, select the correctrouting path, with inserts disabled.

With no EQ, listen. Does it sound good, is it muffled, scratchy, far away,or boomy?

Is the mic facing the wrong way, (this happens often) or are you listening to wrong input.

Impedance mismatch between the console input and the source. (line, mic, instrument)

Bad cord, connection, patch bay, patched in wrong hole, patchbay normal not broken- mic going too many places at once

Balance vs. unbalanced - pin2 vs. pin 3 (unbal pin 3 at one end + unbal pin 2 = SHORT)

Bad instrument - change try another

Bad playing technique or position - try something else, face Mecca.

Move the mic a little - start with close micing, then move the mic away

Acoustic guitars, pianos, Bass, Standup bass, Drums

Go in the room and listen to the instrument with a finger in one ear.

Ask the player - chances are he has recorded this instrument before and has some idea as a starting place.

If he says "This is what I do all the time and it always sounded good before" then there is probably something else wrong.

Tape Machine:

Machine on input
Monitor through the machine (good idea in case you are overloading the machine input) make sure that whole signal path is working right. (what you see on the meter may not be what you think is going there.

Listen to output of machine with no music playing. Listen for hums, crackles or buzzes. If the meter is reading something, then there is probably a hum or other noise that you didn't notice.

What kind of metering
Digital metering is the most accurate
Peak meters second best
Analog VU meters, depend on what music is playing (click, hi hat, organ, etc.) Percussive instruments should indicate lower on the VU meter for proper recording level.

Don't forget to make sure that you are using the correct tape for the machine. Bias, tape stiffness & head wrap.

Noise reduction dbX Dolby A, B, C, S, SR

Don't use noise reduction on the SMPTE track. Make the SMPTE track one of the edge tracks. (cross-talk). Don't use noise reduction on digital recordings.

Autolocate is a nice feature. Also autopunch, cycling etc.

Recording:

Start the machine in plenty of time before the song begins. Allows machine to get up to speed. Allows plenty of SMPTE for future lockups.

Let the machine keep running a little while after the take. In case you want to add something at the end, or cross-fade into the next tune, or?

Make sure you have enough tape for the take you are about to record. If what you are recording is longer than a reel of tape, plan a break in the music for changing tape or get a second machine for A/B rolling. (the second machine is placed into record before the first machine runs out of tape.

Overdubs:

Must be able to monitor output of machine

Good earphone mix.
Try not to use speakers to monitor during overdubs

Test punch in capabilities of machine Punch during sustained playing & punch right on beat. Play back. See if there is glitch and see if there is any delay. If delay, modify punch technique accordingly.

If big glitch, don't punch during sustains or punch on back beat or someplace that will mask punch.

Effects - EQ:

Equalizers - change the tonal characteristics of the audio. They have at least bass and treble controls. Most desirable is four band sweepable parametric EQ.

Graphic equalizer. Usually 5 to 31 frequency bands, each fixed in frequency. Usually with slide pots to show a graphic representation of the frequency curve.

Peak vs. Shelving EQ.

Tuning EQ by EAR

Use EQ to:
Compensate for low listening levels
Make the blend between different instruments more pleasing
Compensate for bad frequency response in some device
Reduce noise
Special effects like telephone voice
Reduce apparent leakage between instruments

Effects - Compressors & Limiters:

Compressors keep levels more constant by automatically detecting level changes above a set level and riding the gain.

Use compressors on individual instruments, not mix. It will be less audible.

Attack time settings determine the "punchiness" of the instrument. Peaks get through before the compressor actually clamps down. Faster attack will make for a smoother sound.

Limiters are faster than compressors and are there to LIMIT the amount of signal passing. These are usually there to protect equipment such as radio transmitters or speakers from overloading.

Effects - Noise Gate:

Noise gates work like a soft on-off switch. As the level of the sound gets below a set point, the signal is turned off, blocking any residual noise that may creep through. If not set properly they can be worse than the noise.

Effects - Delays & Echoes:

A delay by itself has no effect but to delay the signal. When the delay is heard mixed with the original signal, we have a sometimes more interesting sound.

Echo & reverb units control the amount of feedback sent to the input of the delay as well as the number of taps off of the delay line. These signals mix together to form artificial reverberation as found in different size enclosed spaces.

Doubling (recording the same instrument playing the same part twice) can be simulated by using a delay of from 9 to 30 milliseconds. This fattens up vocals and instruments and can make it sound like there was more than one instrument playing the same part.

Short delays can also add fake ambiance to a recording that was too dead sounding.

Chorusing is caused by modulating the delay time. This modulation causes a change in pitch as well as a change in the delay time. This produces a wavy effect in the sound.

Flanging effects are created by using a delay of 10 to 20 milliseconds and changing the delay amount slowly between those two parameters. The delayed signal mixes with the original signal and some of the frequencies are out of phase with each other and cancel or augment each other. A change in delay time changes the frequency that is affected.

Harmonizers - Octave dividers - Aural Exciters:

Harmonizers are used for pitch shifting effects. They can be used to fix bad notes in some cases, or to add harmonies in other cases.

Octave dividers add an additional tone one to two octaves below the original signal. This can fatten up an otherwise wimpy bass or kick drum.

Aural Exciters work by adding slight distortion and phase shift to the signal. This can brighten up an otherwise dull sounding instrument. They usually work the best if there is a rich overtone sequence present in the original sound. The work great on snare drums, background vocals, and string pads.

Combining tracks:

On analog machines do as little as possible. If you have to combine vocals - record a bunch, combine to one track - record next bunch - combine to next track. At all costs avoid bouncing to adjacent tracks (feedback). Watch out for track next to SMPTE.

Combination digital and analog machine. Record vocals on ADAT & combine to one track on analog deck. Same quality as one original recording on analog machine. Adjacent tracks not a worry on digital machines.

Comping tracks:

Recording multiple tracks and combining to make one master track. If you can do it on the digital deck, it is better. If you must do it on analog deck, try not to do it multiple times.

If the way you work is to try 4 takes, then comp, try 4 more then comp again. Don't use the comp track as a component and bounce to new track, try to take any pieces and comp them into the existing comp track. That way comp track will never be more than one generation down. (Make a safety track if you have trouble punching in tight spots.

Mixing:

Clean up tracks. Erase unwanted material (with the supervision of the producer). Mixing will be easier

Make a cue sheet reminding you when to make what moves

Levels. Different DAT machines use different reference level.
What does reference level mean? analog vs. digital.

In analog recording, “Zero” is a level reference at which there is 3% harmonic distortion. Above this level there will be more distortion but a better signal to noise ratio. Audio contains peaks which may be above this zero reference by as much as 20dB. Analog tape compresses this information and records it with more harmonic distortion, but for the small instance that the peak lasts, this may not be a problem. If recordings are made at a lower level, the distortion figures are lower, but the signal is dropping into the noise floor of the tape.

In digital recording, “Zero” is the level above which no additional information can be recorded. This results in hard clipping of the sound. Anything above “Zero” is not recorded. A reference level of 18dB below “Zero” allows room for peaks in the audio to be recorded without clipping. Because the noise floor is so low in digital (98dB below “Zero”) having a reference at -18dB does not really effect the quality of the recording.

Echo. Don't use too much of a good thing. Use just enough to provide the ambience or effect necessary.

Effects. If you have empty tracks available on the multi track tape, record the effects to free up equipment for something else, or to save time in re-mixing.

Limiters (use on record and playback - different ratios) Effects on vocals should be kept to a minimum.

Panning and stereo placement should be determined by the final destination of the mix (TV, video game, Surround Sound, CD, CD Rom). Keep in mind the center buildup phenomenon. Avoid placing something all the way to one side. (keep in mind stereo listening and being able to hear from opposite side of the room)

Mix machines:

Analog 2 track
- Revox, Tascam. Otari, Ampex, Sony, Studer
- 30 ips vs. 15 ips vs. 7 1/2 ips
- 1/2 inch vs. 1/4 inch
- Dolby vs. Non Dolby
- dbX and other noise reduction.
- Center track time code
- Cleaning of machines

DAT
- 44.1kHz vs. 48kHz
- Emphasis on or off
- External or built in converters
- Type of DAT tape Computer backup DAT tape, Apogee, HHB.
- Don't use 3hour tapes unless machine is designed for it
- Cleaning of machines. Use DAT cleaning tapes properly.
- Input pause wears the heads.

CD-R
- Marantz, Carver, Yamaha, Studer, Phillips, Micromega.

Cassette

- No comment

Mixing back to two tracks of multi-track
- Multitrack 48kHz or 44.1kHz? Stuck with whatever multi is.
- Digital or analog.
- Updating mix without remixing

Sample Rate Conversion:

To get from one sample rate to the other or VSO final mix?
Alesis AI-1
Roland SRC-2
N-Vision
Z-Sys
Analog out - in

Editing:

To change the arrangement of the song
To assemble all of the tunes in order for distribution or going to mastering

Razor Blade editing

Hard disk editing Akai, Sound Tools, Sonic Solutions, Turtle Beach
Roland, SADiE, RADAR, etc.

Optical disk editors AKAI, Sony PCM 9000

DAT editors Sony, Otari, Fostex
Music editing vs. assembly editing

Pause editing DAT is a NO-NO unless plenty of time between cuts.

Editing for vinyl records (South America etc.)

Editing for cassette master

Pre-Mastering:

Assemble in the correct order with proper spacing
Don't do pause edits on DAT machines unless 5 seconds around edit
(If that is the only way, let mastering do it)
Consistent levels (If you don't do it, Mastering will have to)
EQ All of the selections should have similar tonal quality

When you are done, Make a Digital Copy. Don't send your only tape

Some plants can accept CD-R as master. It must not be Multi-Session
All Plants accept Sony 1630
Some plant will accept DAT(not if they have to edit)

Include accurate timing sheet (where you want each cut to start

Make them send you a ref (plant or mastering facility)

If everything done (eq, levels, editing) copy DAT to 1630
PQ codes on tape? or PQ time sheet. Music @ 3:00 into tape

If you can afford it, good idea to let mastering facility EQ and level correct your tape. You want your product to be competitive with everyone else so it has to sound as good. Third party reference is good.

Think about breaks for cassette. Second side should be shortest

Labels:

Multi track labeling of boxes and track sheets.

DAT labels & J cards
Cassette
CD labels.

Keeping notes:

Keeping good notes. Which mic on which instrument.
Which sequence was used to print to tape
What was the tempo
Which SMPTE interface was used to drive sequencer
What kind of direct box was used through what preamp?
Was instrument delayed? if so, which delay and by how much?
What kind of tape was used and what was the machine set up for
What was the reference level for recording.
What reference tape was used to set up machine
What reference tape and levels were used for Mix?
What effect units were used and what were the settings?
Limiter settings for vocals or whatever?
If you printed alternate mixes, what were the differences?
Were they printed at different levels or different VSO settings?

This should get you started in the right direction. If there are any further questions, mail them to me at EQ magazine or Internet rnichols@ping.com or Compuserve (70241,1142)
Thanks, Roger Nichols