sound advice

Advice for Audio Professionals

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USB Or Firewire?!?

by Alec Watson

I have received some e-mails lately asking whether to go with Firewire or USB interfaces, here are some thoughts: I find that Firewire has more stability on older systems. This may, in part, be due to the fact that you have your printer, cordless mouse, keyboard, hub, external hard drive, camera, USB coffee maker … all plugged into the USB ports while the Firewire port sits empty. If I were to choose one system over another, that would be why. A Firewire port will likely be dedicated to your music interface allowing for uninterrupted bandwidth to and from your processor.

USB microphones will be a big hit this year – I don't think I would run out and buy 12 of these and then plug them into a couple of USB hubs and then record my band – though conceivably you *should* be able to do this. But if you just want a simple rig for getting sound into the computer, these little techie gifts from the Computers Gods are awesome. I first reviewed the Samson USB mic last year, and although it isn't likely to replace any of my studio mics anytime soon, value for money, it was excellent. Apparently I wasn't the only one who thought this, as there are all sorts of manufacturers who will be peddling USB mics this year. If you just need to get an acoustic instrument or vocal into the computer, you will be hard pressed to beat the ease of use and price of these.

The greatest innovation, possibly of all time, I have saved for last (insert echo-y deep announcer voice) Roland's V-series MIDI accordion! You think I am joking don't you? I am, a bit. Seriously however, this thing rules! (www.roland.com/products/en/FR-7/). For all the keyboard players and programmers out there who have to twiddle, for hours, with all sorts of knobs to get their keyboards to play horns or strings, this unassuming little machine could change programming as we know it. The biggest reason a keyboard can't recreate a realistic horn or sax sound is the fact that the keyboard is a percussion instrument. Much like a drum, the envelope of the sound attacks hard and then decays; an accordion however IS a wind instrument, one with keys. You can play horn arrangements that sound fantastic on this little machine – accordion geeks of the world rejoice! For all those years that you have been picked on and laughed at, the lead guitarist and keyboard player are now going to be secretly envious! You're still not likely to get any action after the show, but oh yes, the lead guitarist *will* secretly be plotting against you, while he is getting some action!

Alec Watson is a producer/engineer that works from his destination studio on Vancouver Island. His parents couldn't afford to get him a piano when he was a child; they got him an accordion; the rest, including his career, is history.

Don't Shout Before You Speak

by Jim Yakabuski

The lights go down. The dry ice creeps over the front of the stage. The crowd is frantic as a low rumble builds and builds until the ceiling tiles are falling out of the roof and people are ready to run from the building. Just as you think you can't take it any more, the rumble builds to a deafening, throbbing crescendo and then is abruptly cut off by blinding light and a band on stage that sounds as if it is playing through a transistor radio.

Sound familiar? Hey, it has happened to me. The darned intro tape can kill you every time. And why is it that bands always want to use something that has 4 Hz in it to open the show? Go figure.

The problem that causes this discrepancy in level is usually SPL reference. During the afternoon when you soundchecked the band in an empty room the volume of the intro tape seemed quite substantial. But after an opening act and the roar of the audience as the house lights go off, you find yourself pushing the level of that intro tape higher and higher, leaving the band to come out sounding less than impressive.

You need to establish the maximum level that the intro "rumble" DAT can go before it upstages your band's first song power level, and not be freaked out if it doesn't sound loud enough as it's rolling. It's better to start out with the intro sounding a bit low and the band sounding a little loud than the other way around. I refuse to let all the frequency bands through when this type of tape is handed to me. If the bottom end of the band doesn't usually live in the 30-40 Hz region for most of the show, then I'm going to high-pass my DAT intro tape to at least 40 or 50 Hz. You want the audience to remember the first note the band plays with an overwhelmed feeling, so let it be good and powerful. Don't let a silly tape that was produced and mixed at Skywalker Ranch give your sub-bass speakers too much of a workout before the real deal comes on stage. Save the best for last and lighten up on intro overload.

This article is taken from Jim Yakabuski's book entitled Professional Sound Reinforcement Techniques. *The book is published by MixBooks, an imprint of artistpro.com.*

Setting Levels For Digital Recording Can Take More Thinking … Than You Might Think

by Alec Watson

Every once in a while I get a little carried away with trying to think up some technical gem; this time, for a change, I thought it might be nice to take things back to basics – real basics – but not necessarily real simple. Sure, it's good to know the "hows" (how to do this and that), but more importantly it is better to know the "whys".

We all know (in digital) that the green lights on the record level meters are good and the red lights are BAD. Did you know however that there are different *shades* of red? Okay, well there aren't really, but some nasty red lights on tracks *are* more acceptable than others…

Of course, we could simply avoid any red lights at all (and the point of this article) by setting levels really low; so why not be extra conservative when setting record levels? The reason we are trying to maximize levels is two-fold; both reasons having to do with noise. There is an inherent noise floor in a preamp that the microphone is plugged into; there is also a noise floor present in the analog to digital converter that the preamp is plugged into. The A to D conversion process also suffers from a different type of noise, "quantization noise", that can become an issue when recording digitally at low levels. The noise floor of a preamp or A to D converter can be heard as a hiss; whereas quantization noise is more of a digital artifact that is present when an instrument being recorded is so quiet that it is barely moving the meters. So, in an ideal world we are simply trying to get our recorded signal as loud as possible above the noise floor of the electronic circuits and digital conversion process.

Life isn't always so simple though is it? It turns out that many mic preamps, especially the tube variety, have a "sweet spot". There are often three amplification or electronic stages on the way from a microphone to a digital recording medium. The first stage can be found in the microphone itself. A condenser microphone will often have a "pad" where you can attenuate the volume of the incoming signal. If you find your source sounds distorted no matter how low the preamp is set, your source could be distorting your microphone.

The next amplification stage is the mic preamp. This is usually the piece of gear where you get to trust your eyes AND ears. Many mic preamps have little red lights to tell you whether they are distorting or not; sometimes a little red light on the mic-pre can make the track sound more aggressive or fatter; be very careful with this though as there is no "undo" button for a distorted signal. You might find you can add a bit of "grit" later in the digital plug-in world.

Don't be afraid to trust your ears; no you don't want lots of little red lights in your digital recordings, but don't go stopping a take if "the magic is happening" and your ears say it's okay. It is probably much better to have a flawed recording of a great performance than a technically perfect recording of something completely unremarkable (Hmmm, I am suddenly thinking of some people's records…).

Alec Watson is a producer engineer that works from his destination studio sitting atop the Georgia Strait – not to be confused with George Straight. Visit him online at www.alecwatson.com.

Darth Vader You Don't Need

by Jim Yakabuski

Effects can take an average show and add all the glitz and sparkle that make a great show. If you have a nice, acoustically dead environment to mix in your choice of effects, it can make or break your mix. One of the bad habits I've found myself getting into over the years is checking effects returns during the day and then assuming the parameters are not going to change at all before the show. You can be pretty sure that all is okay with most effects units because you physically have to call up edit parameter pages to get in there and muck around with things, but some units have parameter adjustment wheels on the front, and they have been known to get bumped here and there which can cause some embarrassing moments.

One unit in particular that I am speaking of is a certain kind of harmonizer. It has a spinning wheel on the front panel that is very easy to turn, and on one occasion I just happened to catch myself before I made a horrendous mistake. The last thing that I was editing that afternoon was the pitch of the harmonizer. Without changing that edit page, the wheel got bumped later on in the day and just before show time I happened to listen to my effects returns and catch the mistake before the show started. If I had not, the two lovely ladies who were singing backup vocals for the show would have resembled Darth Vader much more than their normal sweet-sounding southern selves. The edit wheel had spun down and the pitch dropped considerably.

Another way that you can get caught is if someone, like an opening act engineer, makes adjustments to your effects during his show and forgets to tell you, or you forget to recall your program or parameters. Usually these days most opening acts get their own effects gear, but in many club situations everybody is sharing effects. Be sure to store your settings, and in those types of situations, double-check that nothing is out of whack. It's a good habit to get into. Unfortunately, aside from the benefit that all these programmable units have given us, they can catch us once in a while because we rely too much on their stability. Add this to your pre-show checklist. And use the force!

This article is excerpted from Jim Yakabuski's book entitled Professional Sound Reinforcement Techniques. *The book is published by MixBooks, an imprint of www.artistpro.com. You can also find the book online at www.mixbooks.com and www.musicbooksplus.com.*

Audio Phasing: Part I

by Al Whale

The speed of sound is approximately 1,130' per second in air, depending on the actual air temperature. Therefore, if you have a 1,130 Hz tone it will complete one full cycle in one 1'.

Now suppose that you have a tone generator feeding a speaker, with two microphones as shown below (image A). Distance affects the phase. If the mics were both the same distance from the speaker, they would be in phase and would add together. The resulting tone would be twice the level (6dB) of either tone (image B). Similarly, if the second mic was 1' further away from the speaker, the two sources would still be in phase and would again add together. If the second mic was only 6" further away from the speaker than the first mic, the two sources would now be out of phase. This would cause the tones to cancel (image C). For the next section return to the previous setting, mic #2 is 1' further away from the source than mic #1.

Frequency also affects the phase. At 565 Hz (1,130 hz/2) the tone will now complete a full cycle in 2'. As seen in the following example, the two tones now arrive out of phase and thus cancel (image D). At 1,695 Hz (1,130 Hz x 1.5) the two tones also arrive out of phase and cancel. However at 2,260 Hz (1,130 Hz x 2) the two tones arrive in phase and thus add. This effect, known as comb filtering, can be shown to repeat all the way up the frequency band.

The following graph shows the resultant gain verses frequency (image E). Note that when the two signals are equal, if they are exactly in phase they add 6dB, but if they are exactly out of phase, they totally cancel. In an actual situation, the effects would probably not be as pronounced, since the levels from the two mics would seldom be exactly equal. One good example of this situation is when two mics are (mistakenly) placed on each side of a lectern, with the idea that they will pick up the audio regardless of which way the speaker turns. This will result in poor sound quality. As the speaker turns his head, one mic can be closer than the other, thus introducing the comb filtering. Comb filtering will produce a hollow, diffuse, and thin sound.

Pick up the February issue of *PS* for Audio Phasing: Part II.

Al Whale is a Broadcast Technologist and Assistant Chief Engineer at CHBC-TV. He has also set up and operated sound systems and taught sound in many church settings. Reach him at awhale@chbc.com.

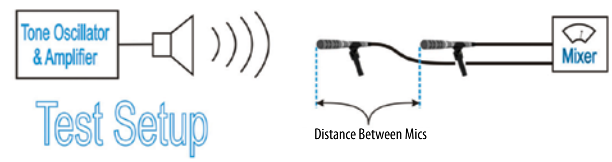


Image A

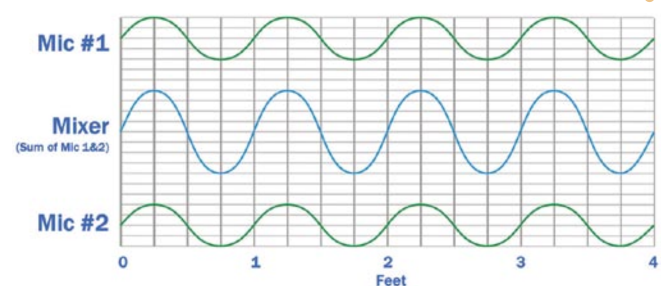


Image B

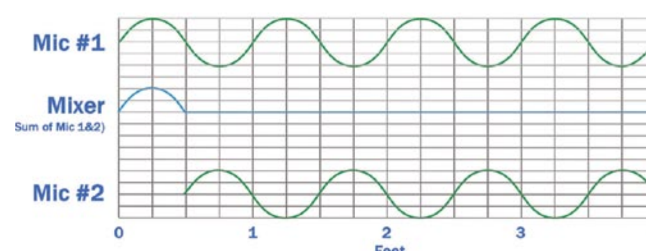


Image C

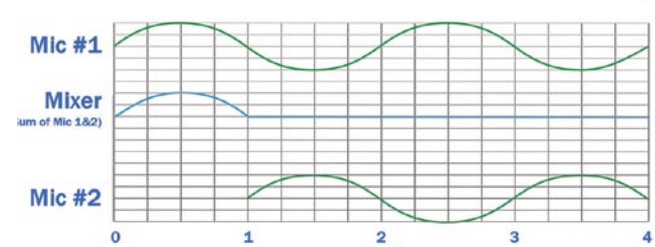


Image D

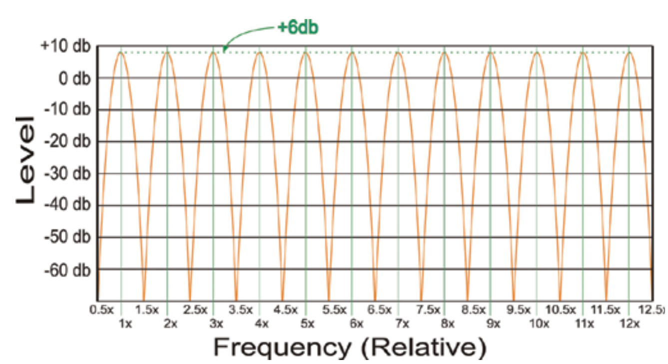


Image E

Miking The Snare Drum

by Tim Crich

For the best snare drum sound, using a properly tuned and professional drumkit is paramount. Whether the band is Death Metal From Saskatoon or The Polka Pals 'n' Gals, the drums will be the backbone of the recording.

Start with a dynamic mic, as it can handle the high transient levels of the snare drum and a solid, stable mic stand. Position the mic off-axis with the rest of the drums to minimize leakage. Aim the mic directly at the point of impact – where the tip of the stick makes contact with the drum. Look down the barrel and line up the placement.

Of course, place the mic where the player can't accidentally whack it. Expecting a drummer not to hit a poorly placed mic is like asking a record producer not to order sushi; sooner or later, it's going to happen. It's your fault if the drummer hits the mic with the drumstick, not his.

For more crack, maybe place a second mic with a different quality, such as a crisper high end, alongside the first. Keep these two mic capsules as close together as possible because two mics on any one source can create phasing issues. Perhaps add a third (switched out-of-phase) mic underneath the drum aimed up at the snares. Get the best sound using mic choice, placement, and level before reaching for the equalizer.

If possible, record the individual snare drum tracks on your digital recorder, and analyze the sound waves. Work on moving the mics around so, when recorded, all the drums are in total phase. Good luck!

Tim Crich is a recording engineer/writer living in Vancouver. His credits include The Rolling Stones, John Lennon, Billy Joel, Bon Jovi, KISS, and lots more. Watch for Tim Crich's Assistant Engineers Handbook 2nd Edition *coming soon. Reach him at tcrich@intergate.ca, www.aehandbook.com.*

Challenges In Recording 5.1

by Michael Nunan

As the great bulk of television production begins to make the transition to HDTV, and casts a wary eye on the notion of 5.1 surround sound – we're faced with a quality versus quantity dilemma.

On one hand, virtually all of the cameras that we're likely to employ in the acquisition of HD pictures feature excellent digital PCM audio recording. That's a "check" in the quality column. On the other hand, even the most expensive HD field cameras have a disturbing number of audio connectors on the back … two! Even on cameras that utilize videotape formats that natively feature four (and more) audio channels, the default configuration of the camera usually features only two inputs! That leaves us with a significant quantity problem.

All other things being equal, I can readily suggest several microphone techniques, or microphone systems which will do a great job of capturing multi-channel sound. Even after accounting for the requirement that these systems be rugged, simple to use, and field-portable – there's still a number of options to choose from. Alas, in the world of factual TV production – all is not equal. While the last few years have seen an increasing number of multi-track location recorders appear on the market, their use assumes as a prerequisite that double-system recording is easily accommodated by the production. Unfortunately, most television production (that isn't dramatic in nature) still relies on a single-system workflow (where the camera is the only recording device on location, responsible for recording both picture and sound) and therein lays the challenge. Until producers are adequately convinced of the merit of authentic surround production, most will be unwilling to undertake the cost and complexity of changing the way they normally work. The alternative is for manufacturers to start making camera systems which have native multi-channel audio capabilities, which will allow us to honour the tradition of single-system production, while still satisfying our requirement for higher track counts from the field.

Either way, the thing we need most is for the audio community to become much more vocal in lobbying for change in the world of TV. Those of us who love sound and understand the crucial role it can play are already convinced … we need to stop preaching to the choir, and start educating everyone around us.

Michael F. Nunan is the Post Sound Supervisor at CTV Television Inc. Reach him at mnunan@ctv.ca.

Rich's Rights To Recording Electric Guitar

by Richard Chycki

I've been fortunate enough to record a number of legendary-status guitar players like Aerosmith's Joe Perry and Rush's Alex Lifeson. Watching them work is truly an inspiring and educational opportunity; artists like these have accrued a wealth of real-world experience in manifesting instantly recognizable guitar tones. Being the captor of these tones, I'll share some tips about recording electric guitars.

Right tools for the job: This is a no-brainer but is a common miss. Select gear and tone that works for the song and put your individuality into it. Want to get the right tone? Listen to it. Really. That means pointing the speaker right at your head, not blowing across your knees while you stand in front of a half-stack. Off-axis settings are brittle and don't sit well in a mix.

Right mics: While there are a myriad of possibilities for miking an amp, I've had great success with a few favourite mics. First is the venerable Shure SM57. I've tried the Shure Beta 57 and, while it sounds similar, the polar pattern is so tight that finding the sweet spot in front of the speaker can be quite a mission. Other mics I commonly use include the Sennheiser 421, the Sennheiser 409, and the Earthworks SR30. Special mention goes to the Royer 121 ribbon mic. This workhorse mic sounds amazing for almost any electric guitar purpose from country to metal and the specially designed ribbon element won't fry from the high SPL of close-miking an amp on 11.

Right place at the right time: Personally, I prefer to record guitars in more of a dead environment, although I've been known to track in extremely live environments (Joe Perry's tiled bathroom for one) for effect. In all situations I have the amp lifted well off the floor to avoid troublesome reflections, and I don't use anything hollow that could resonate (like a roadcase).

Right phase: For multi-miking, it's important that the phase relationship between the mics remain consistent. Liberal testing of phase using the console's phase flip button is a necessity when blending mics. For mics placed at various distances from an amp, comb filtering can result from the phase shift due to the longer time the sound takes to reach the more distant mic. Fortunately, a small company in the Los Angeles, CA area called Little Labs has a device called an IBP (In-Between Phase). It can shift the phase to any degree from 0 to 180 so it's a simple task of dialing the mics into exact phase.
Happy recording!

Richard Chycki is currently recording a new CD for Rush and has worked with Aerosmith, Mick Jagger, Seal, Pink, and many others in the past. Reach him at info@mixland.ca.

Audio Phasing: Part II

by Al Whale

Comb filtering, which produces a hollow, diffuse, and thin sound, will occur with one microphone receiving the same sound from two sources. A common example of this is shown below. If the microphone had been closer, the difference in the direct path and the reflected path would have been greater, thus the reflected path's reduced level would have had less effect. Also the reflected source volume would have been less if the floor had been carpeted.

Methods of correction:

1. Keep the vocal audio mix low into the monitor.
2. Handhold or place the microphone closer to the singer.

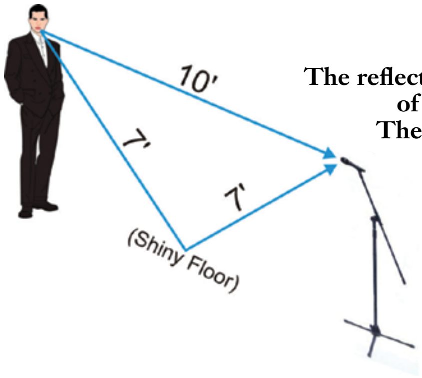
While the monitor helps the singer, as the monitor's gain is increased, the resulting vocal will be more muffled. Many professionals use in-ear monitors to eliminate this effect. Although not popular with the performers, using music only on the monitors (no vocal) will also minimize comb filtering. Often, the house audio suffers when trying to improve the monitoring for the performers.

This article was prompted after I attended several performances in which the music was excellent, however the dialogue was difficult to understand. Most of the production crews knew the script so well that they were unaware of the problems. If you asked the audience, they would probably say that they thoroughly enjoyed the music. If you were more specific and asked them about the script, they probably would be unable to answer. The comb effect of excessive use of stage monitoring would mush the dialogue so that the audience (which doesn't know the words) would be unable to understand them. If the performers are trying to tell a story, they basically miss the goal and only provide enjoyable music.

Ideas to reduce comb filtering:

- Reduce the number of paths from the same audio source.
- Fewer microphones.
- Reduce the possibility of reflections.
- Reduce the relative amplitude of the additional paths.
- Increase the difference in path lengths, thus the secondary path will have more attenuation.
- Use absorbent material.
- Use directional qualities of the microphones.

The following sites assisted in this article: Calculations of attenuation over distance www.mcsquared.com/dbframe.html; calculations of distances www.pagetutor.com/trigcalc/trig.html.

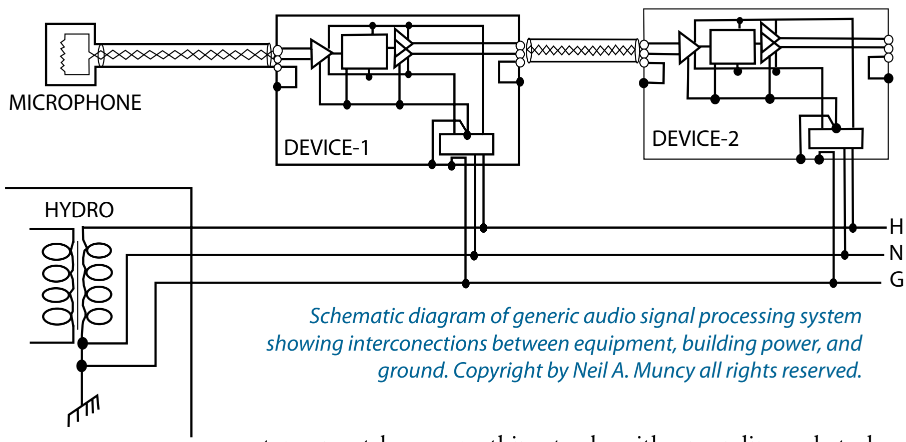


The reflected source is -3dB of the direct source. The comb filter effect will be present.

Al Whale is a Broadcast Technologist and Assistant Chief Engineer at CHBC-TV. He has also set up and operated sound systems and taught sound in many church settings. Reach him at awhale@chbc.com.

GROUNDING, SHIELDING, HUMS, BUZZES, & THINGS THAT GO ZAP! IN YOUR SOUND SYSTEM

by Neil A. Muncy



Schematic diagram of generic audio signal processing system showing interconections between equipment, building power, and ground. Copyright by Neil A. Muncy all rights reserved.

Noise susceptibility (or the lack thereof) in audio systems is a function of two principal factors: shielding, and the "pin-1 problem." The endless conversations concerning this matter inevitably involve earth "grounding," a subject which has been around for so long (200+ years) that it has devolved into a sea of confusion, misinformation, and mythology, even though it is completely dictated by easily understandable, basic physics.

Conventional grounding mythology would have one believe that electronic systems of all kinds must be robustly connected to earth ground in order to properly function – audio signal processing systems in particular. The grounding reality is that airplanes, motor vehicles, laptop computers, blasters, etc. seem to work just fine without connections to earth ground. Nevertheless, A/V systems of all kinds are considered exempt.

According to the conventional mythologists, "noise in audio systems must have something to do with grounding, what else could it be?" The bad news is that the short answer to this question would fill up this entire issue many times over. The good news is that on the *Professional Sound* website, www.professional-sound.com, a long list of reference material will be found. In addition, the June 1995 issue of the Journal of the Audio Engineering Society, entitled "Shields and Grounds," includes seven papers which directly address this matter. Go to www.aes.org, and look up "Special Publications." It's available as freeware to anyone for $15 US, less if you're an AES Member … it may also be downloadable. It won't take you long to realize that the conventional mythologists just might be wrong!

Neil Muncy has been around since the days when recorded sound was analog mono and vacuum tubes ruled the audio landscape. He has been a consultant in the audio field for many years, and can be contacted by email at: nmuncy@allstream.net.theaters.

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5 Tips For Stalking, Managing, & Capturing Rogue Sounds With Traps & Baffles

by Russ Berger

Employing Sound Traps and Baffles is much like hunting.

1. Know your hunting grounds: Before the hunt, know and understand your acoustical environment. Once you bound a space with walls, a floor, and a ceiling, you've committed acoustics. The boundaries of your space define the low frequency modal response and set limitations for the ambient decay time. Wonderful programs and countless texts have been written that clearly describe the process for analyzing, predicting, and managing acoustical boundary conditions.

Once you understand your environment you will better know how rogue sounds behave in the space; you can better identify where problems might lie and devise a trap to capture the problem.

2. Put the traps where the beavers are: Place traps to capture rogue sound much like you'd place traps for beavers. Placing beaver traps on the ceiling will do you little good, just like placing acoustical traps where the sound you want to capture doesn't exist. Beavers pretty much live their lives along the floor plane. But rogue sounds live in the three dimensional world, so successful hunting can be achieved if the traps are placed in proximity to boundaries and intersections.

3. Be sure your passive trap is big enough to capture your game. Lower frequencies require larger and deeper traps to control and manage long wavelength rogue sounds.

4. Know how many you want to trap: Trapping one beaver vs. an entire colony will require different methods. The effective trap absorption efficiency is proportional to the area of coverage.

5. Conceal the trap: A good looking studio always seems to sound a little better. Integrate your traps into the architecture and along with those rogue sounds you'll catch new clients.

Bonus Tip #6: go to www.RBDG.com – Russ Berger is Owner of Russ Berger Design Group (RBDG), which is a design and consulting firm that combines expertise in acoustics, architecture, and interiors to create technical environments and buildings for recording studios, broadcast facilities, creative production spaces, and home.

Everything Is An Amplifier Part I

by Bryan Martin

The basic building block in audio is the amplifier. When the word is mentioned most of us have the image of a power amp pop into our heads. They get all the press because they are the largest and most glamorous of the species, but what about the myriad of smaller and forgotten gain stages that occurred before the signal arrives at this last power stage? They are largely anonymous and taken for granted, but determine the quality of recorded sound.

Almost every knob on a piece of audio equipment is controlling a specific amplifier stage. And the farther we get away from the basic understanding of this simple entity, the farther we get away from knowing how to maximize its sonic potential.

The steady advancement of technology has served to obscure their very existence. Amplifiers have become so small and commonplace that they have virtually disappeared from human consciousness. Just look at the iPod nano – that thing is loaded with amplifiers, all crammed onto a little chip and powered by another sliver of technology.

The implementation of each gain stage, individually, and then as a complete amplifier, determines the sound quality of a piece of audio equipment. This fact seems to have been largely lost in the mysteries of time. Most people don't even realize the devices they use even contain an amplifier. It just works.

In recent years, there has been an onslaught of multi-function units, recording channels, and the like. With the recording business moving from a professional to a consumer market, manufacturers are trying to offer the most features for the price. This looks great on the outside, but there is a large cost on the inside. The quantity of functions within a unit is usually inversely related to its sound quality. Fundamentally, it is difficult to design a good-sounding, multi-function unit, because every gain stage comes with the constraints of its implementation.

The requirements of a gain stage are:

1. Its gain coefficient. (With a coefficient of 10,1 V input will give 10 V output.)
2. Bandwidth. For audio we generally want to double the range of human hearing (20 Hz to 20 kHz) so that would be 40 kHz to insure good transient response.
3. Input impedance.
4. Output impedance. (Generally we want to have the output impedance of the previous stage low in relation to the stage that it is driving to minimize the losses in the coupling between the two stages.)
5. Maximum output signal before clipping.
6. Maximum input signal before clipping.

Pick up the August issue of *PS* for Part II.

Bryan Martin owns Sonosphere Mastering. Over his 20+ year career he has worked with David Byrne, Rufus Wainwright, Max Roach, Run DMC, and White Zombie. He can be contacted via e-mail at bryan@sonosphere.ca or on the web at www.sonosphere.ca.

TIPS *On Getting Killer Drum Sounds*

by Nick Blagona

Engineers spend more time getting drum sounds than any other instrument. I've seen situations where days have been spent getting a drum sound. Kits are changed, heads are changed, cymbals are changed, heads are taped up or un-taped, mics are selected and changed, the kit is placed in various parts of the studio, head damping devices are used, mini pads are cut up and placed on heads, and on it goes. The poor drummer keeps hitting his kick, snare, and toms … by the end of this, he or she is back in rehab.

Here's my approach for a great drum sound. My recommendations for drum mics: Sennheiser MD 421s, Shure 57, and some Neumann 87s. I like using the Neve 1081 console in Studio 1 at Metalworks, so all frequencies mentioned here are from the 1081s. I find that padding down the preamp as low as you can go with the fader up gives me the best result. Having the mic pres all the way down gives me very little leakage from the cymbals to the toms and hi-hat to the snare.

THE KICK

Mic the kick drum with a Sennheiser 421, throwing a sandbag in the drum helps to dampen out any overtones. The mic should be placed right at the beater. I also use a Yamaha NS10 woofer as my second mic, placed where the front skin used to be. I record this flat since it has the perfect frequency response.

For the 421, give it +3 at 82 Hz for bottom and +4 at 6.8 K for added attack.

THE SNARE

For the snare drum, use the Shure SM57 at a 45- to 60-degree angle about an inch or two above the head pointing it at the centre of the snare.

+2 at 82 Hz, -2 to -4 at 820 Hz, and +4 at 6.8 K for crispness. If you like the idea of miking under the snare for some rattle and hum, use an AKG 414 in a tight pattern under the stands.

TOMS

Mic all three toms with the 421s set at about a 45-degree angle to the centre of the tom. I usually add some 8.2 K.

OVERHEADS

For the overheads use U 87s. Place the mics about 16" over the cymbals' centres and towed out at about 45 degrees. I usually record them flat.

HI-HAT

An AKG 451, pointing at the centre.

Nick Blagona has recorded The Bee Gees, Chicago, The Police, The Tea Party, Alexisonfire, Deep Purple, and many others. Please go to www.nickblagona.com for more details.

Star Grounds, Loop Areas, & Electrical Safety In Project Studios, Edit Suites, & Other Compact Audio Installations Part II

by Neil A. Muncy

A Star grounding scheme, in which all equipment in an installation is bonded to a central ground hub, can be useful for minimizing low frequency common mode voltages between various pieces of equipment if it's properly implemented. If not properly implemented, star grounding can result in performance, which in some cases is actually worse than that resulting from a completely haphazard approach.

Any secondary grounding system installed in parallel with already existing equipment U-Ground conductors in an installation has the instant effect of causing far more potential ground loops between equipment than would otherwise exist. Sometimes it makes a difference, sometimes it doesn't. The $64 question is whether it reliably, and without exception, makes noise go away permanently and completely. Not likely.

A popular Star Grounding practice involves using separate ground wires to bond all equipment in the ensemble to a central hub, and then connecting this hub to a dedicated earth-grounding terminal, which is not bonded to the main building ground system. This practice is very dangerous and is completely illegal in the context of North American Electrical Codes.

One connection between an ensemble of equipment and building ground is all that is needed to make the system safe in terms of both the letter and intent of applicable electrical codes. Most installations usually involve more than one AC power circuit, whether actually required due to the size of the total load or not.

What is not considered in such a scenario is how long and by what path(s) the power circuits and their respective equipment ground conductors take before they get back together at the breaker panel. Just because two outlets are within a few feet of each other does not necessarily mean that they are on the same circuit.

In smallish installations in which all equipment is in one area/room and the longest audio cables are perhaps less than 100' in length, and assuming that the breaker panel is somewhere else in the building, a very effective approach is to arrange to have all of the power circuits end up at a point in one box in the middle of the equipment ensemble. Very often, this middle point would be in the floor trench under the tabletop of the producer's table equipment cabinet behind where the engineer/producer sits.

Install as many circuits as you think you need. What this scheme buys you is that by bringing all circuits into one multi-gang outlet box, all of the associated equipment ground conductors (one per circuit) also end up in the same box, all bonded together as prescribed by code. This star point becomes your one connection back to building ground, with the added advantage that now you have a demonstrably lower impedance path back to building ground by virtue of having X paralleled equipment ground conductors.

From this central box, 3-wire branch circuits are then run out to each grouping of equipment. If at all possible, all of these circuits should be in one continuous raceway/conduit, so that the associated equipment ground conductors are daisy chained throughout the facility. This ensures that the total length of the equipment ground conductors between different equipment locations within the room is as short as possible. For only a few circuits, series-connected power bars are acceptable for this application, but use good ones and try to stay away from conventional "Surge Protected" ones, which employ Metal Oxide Varistors (MOVs) – they have been known to start fires when they ultimately outlive their service life. This ensures that the total length of the equipment ground conductors between different equipment locations within the room is as short as possible. As simple as this seems, this

approach may eliminate enough residual noise so as to end the effort to go any further.

"OK wise guy, so what happens when I then run shielded audio cables all over the place?" you ask. "Don't I end up with a big bunch of ground loops anyway?" Yes you do. Minimize the areas of the resulting ground loops by selectively cutting cable shields at one end or the other, the One-End-Only (OEO) approach. This is a simple way of smothering the symptoms of Pin-1 problems, and while in larger systems it may be required for other reasons, it's usually not necessary in a small installation – besides which, it's a pain in the ass and you can't do it anyway in unbalanced single-conductor shielded cable installations for reasons which should be obvious. What you can do to minimize these loop areas is to simply run all of the low-level audio cables parallel and adjacent to your new branch power cables.

Be sure to pick up the October issue of *PS* where Muncy delves into the Pin-1 problem and other RFI solutions.

Neil Muncy has been around since the days when recorded sound was analog mono and vacuum tubes ruled the audio landscape. He has been a consultant in the audio field for many years, and can be contacted by e-mail at: nmuncy@allstream.net.

EVERYTHING IS AN AMPLIFIER PART II

by Bryan Martin

Once all of the requirements of a gain stage are met [as outlined in the June issue], the designer will then select the ideal requirements for a specific stage, but achieving these requirements rarely happens in the real world. Attaining one design goal is often at the expense of another. There will be limitations imposed by the characteristics of the gain device chosen, economics, physics, and a host of other factors. And as the number of stages increase, so does the difficulty in bringing them all into an optimum specification.

Great sounding amplifiers require high-quality components. Transformers used in power supplies and for audio I/O are both large and expensive. Quality coupling capacitors, gain devices, and hardware all drive up equipment costs. The classic and highly sought after Neve modules have large, expensive power supplies, plenty of transformers, and build quality of impeccable craftsmanship. You will also notice that these Neve consoles have a far simpler layout and less options than the later generation Neve V Series, SSLs, and the like. Generally, a very well-implemented, simple gain path will always out perform a complex one – and negates the need for further processing.

Pick up the October issue of *PS* for Part III.

Bryan Martin owns Sonosphere Mastering. Over his 20+ year career he has worked with David Byrne, Rufus Wainwright, Max Roach, Run DMC, and White Zombie. He can be contacted via e-mail at bryan@sonosphere.ca or on the web at www.sonosphere.ca.

Star Grounds, Loop Areas, & Electrical Safety In Project Studios, Edit Suites, & Other Compact Audio Installations • Part III

by Neil A. Muncy

Still have noise left? If you've reworked your power as described in previous issues, you've done everything you need to do to make your power and grounding system safe and legal.

The Pin-1 problem is a term coined to describe the almost universal practice employed by most audio equipment manufacturers, in which the old-fashioned (pre-1970) method of connecting cable shield terminals (Pin-1s) on I/O connectors directly to the chassis at the point of entry has given way to connecting Pin-1s to some convenient nearby ground circuit trace on the motherboard. The consequence of this practice is that the moment you connect a cable, you have just attached an antenna to the most sensitive inner workings of your equipment! See the AES publication [1] for how to do a Pin-1 test, and suggestions on how to deal with the consequences.

Once you uncover Pin-1 problems, send the manufacturer a letter/e-mail outlining your observations. Surveys conducted by the author suggest that only about 10 per cent of all the equipment presently in use in the audio industry is demonstrably free of Pin-1 problems. If the manufacturer in question doesn't respond, or implies that you've gone bonkers, tell them that you are going to sell off the offending equipment and buy an equivalent unit from another manufacturer who has seen the light. That should get their attention. If not, you now know whom you're dealing with.

If you still have RF Interference (RFI) problems, start looking for equipment with less than major Pin-1 problems. Just because a piece of equipment doesn't exhibit a significant Pin-1 problem at powerline frequencies doesn't guarantee that it will not be susceptible to RFI. A piece of ground wire a couple of inches long inside a piece of equipment, which is employed to internally chassis ground Pin-1(s) can be a very effective re-radiator from well below 100 MHz to the upper limit of the RF spectrum. An RF signal generator can be utilized for this type of Pin-1 test.

This scenario will make your system virtually immune to farfield magnetically coupled interference. Wall warts, line lumps, and power transformers in your gear are all sources of strong extreme nearfield magnetic field energy, which will also cause hum problems if you aren't careful. Locate wall warts, line lumps, and anything else that has big power supply as far away from your low level equipment as practical. Make use of the Inverse Square Law, which dictates that as you increase the distance between a source of interference and the "victim" equipment and cables, the strength of the interference decreases as the square of the distance. In other words, in this case an inch is (almost) as good as a mile.

Pick up the December issue of PS *for Muncy's conclusion and his invaluable tips on MOV surge suppressors.*

[1] The June 1995 issue of the AES Journal, *Shields & Grounds* reprinted as a Special Publication by the Audio Engineering Society. On the web at: www.aes.org.

Neil Muncy has been around since the days when recorded sound was analog mono and vacuum tubes ruled the audio landscape. He has been a consultant in the audio field for many years. E-mail: nmuncy@allstream.net.

Everything Is An Amplifier • Part III

by Bryan Martin

Every amplifier has a sound. Mankind is still searching for the audio grail of a "straight wire with gain." What a great amplifier does is transfer the maximum amount of the information from its input to its output with as little damage as possible. This translates into full bandwidth, wide dynamics, and undamaged transients: the essentials of great sounding reproduction.

In the brave new world of the 21st century, technology has brought powerful tools to the everyday. Recording studios live in a laptop, and declining are the great temples of sound recording and the monks who populate them. We take music and technology for granted. We want it all in a bundle. And very few people have the privilege of experiencing music in an ideal listening environment. Ear-buds, iTunes, and laptop speakers are a pale copy of a breathtaking audio system. As the audio chain gets dumbed down, there is all the more reason to give recorded sound the best possible vehicle on its way to immortality. Because after it is committed to a stream of digits, the road it takes back to sound will be challenging.

Equalization, compression, and the like are often reached for in an attempt to correct a sound source that is lacking. I have always been baffled by manufacturers who package a mic preamp, EQ, and compressor all in on package. If the mic preamp was good in the first place, then why the need for the compressor and EQ to fix the sound coming out of it? Note: Manufacturers spout specs and tech-speak, which may sound impressive, but to the educated reader is often contradictory or plain rubbish.

Audio specs are like accounting: you can make them look like whatever you want. But specs don't translate into good sound. There are plenty of horrible-sounding units out there with amazing specs. To cheaply achieve good bandwidth, hideous mechanisms are employed in the signal path. Using a large amount of negative feedback will drive the bandwidth into the nether regions of the sub and supersonics, and also completely kill the sound quality. People listen with their eyes these days, not their ears. How often do we find ourselves staring at the waveform while it plays back out of a workstation? It's become a reflex almost totally associated with the listening experience. The box looks great; it has to sound great. But that is not always the case.

Bryan Martin owns Sonosphere Mastering. Over his 20+-year career he has worked with David Byrne, Rufus Wainwright, Max Roach, Run DMC, and White Zombie. E-mail: bryan@sonosphere.ca, www.sonosphere.ca.

Star Grounds, Loop Areas, & Electrical Safety In Project Studios, Edit Suites, & Other Compact Audio Installations Part IV

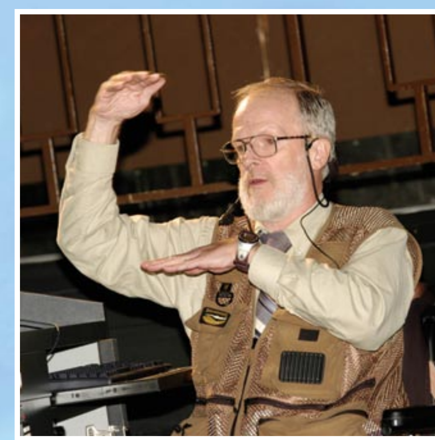
by Neil A. Muncy

Surge Suppressors are widely advertised as the answer to noise and interference problems in all kinds of systems. Consider a few points. First, as mentioned in previous issues, conventional Metal Oxide Varistors (MOV) surge suppressors incorporated into power bars are in widespread use. Unfortunately, unless they are built to a robust standard, which many of the older ones weren't, they may constitute a serious fire hazard, because when MOVs fail, they often get hot enough to melt the plastic housing of a typical power bar long before the fuse or circuit breaker operates. (Murphy at work!) Newer ones must meet a considerable more demanding UL/CSA specification, and are supposedly safer.

Let's suppose the computer(s) in your installation are fed by a power bar with a built-in MOV. When a surge comes along, the "bad stuff" is diverted into the equipment ground conductor and supposedly finds its way back to the service entrance. If the equipment ground path is more than a few feet in length, the natural inductance of the equipment ground conductor will be enough to significantly limit the flow of high-frequency noise current, which is what transients and surges are made of in the first place. Instead of getting rid of surge energy, what happens is that for the duration of the event the entire computer systems' ground reference voltage goes up towards the level of the surge itself, which can be hundreds of volts – if not more.

If the computer is sitting there all by itself and is not connected to any other equipment, this problem may be more academic than real. But if the computer is connected to something else, and the rest of the studio equipment is either not on a MOV surge suppressor fed by the same power circuit, which feeds the computer, or worse yet, is fed by a different power circuit altogether, during a surge there can be sufficiently high voltages between the computer's "protected" ground reference and an "unprotected" studio equipment ground reference to cause major noise and even permanent damage.

If you are absolutely convinced that you need MOV surge suppression, the best way to minimize this problem is to first configure your studio power as described above, and then use the same kind of MOV suppressor on each power circuit feeding the room. Connect all of them to the central hub of your power distribution system, and then run all branch circuits from there. A much better solution is a new Series Mode surge suppressor technology, which does



not contaminate equipment grounds. A bit more expensive than good MOVs, but much safer and much more effective in the long run. You can find out about it at www.surgex.com.

Getting rid of noise in audio systems is nothing more than applied Good Engineering Practice (GEP), the formula for which is: BP + GOCHS = GEP (Basic Physics + Good Old Common Horse Sense). The proponents of alternative esoteric grounding schemes would do well to keep in mind that Mother Nature wrote the original script for the show – and she don't do re-writes!

Neil Muncy has been around since the days when recorded sound was analog mono and vacuum tubes ruled the audio landscape. He has been a consultant in the audio field for many years, and can be contacted by e-mail at: nmuncy@allstream.net.

Chris' Plug-in Script

In today's digital audio recording environment there are a wide variety of plug-ins to choose from, ranging from homemade EQs to $1,000-plus bundles. All can be useful if used in the proper applications.

The most consistently useful plug-ins for me seems to be in the WAVES bundle, including Renaissance Strip. The EQs and compressors are predictable and always do what you require without adding too much colouration to the sound.

Another great plug-in I use a lot is Isotope Trash. It's by far the most in-depth distortion plug-in I have ever heard. It takes a little more fiddling with the controls to get the sound you want, but will yield great results in the end. It works fine on bass as well.

For reverb, I still prefer the classical outboard digital reverbs like the Lexicon 480 and 960, but Rverb and TLspace are also superior sounding plug-ins. I also have found Dverb is useful from time to time … but don't tell anyone I said that.

So, basically, if you're able to afford it, go with WAVES. If not, the Digidesign plug-ins can do you just fine. For cool effects and nice distortions, check out the Isotope series.

Chris Crerar is an Engineer at Metalworks Studios. Visit www.metalworksstudios.com.

The Art Of Mastering: Part 2

by Marisa T. Déry

A few years ago, I wrote an article for *PS* about the art of mastering and how it was evolving. No longer are we there just to make sure that the technical restrictions of the record era are in check; we are now an important part of the creative chain.

Lately, however, I have been seeing an alarming trend: people who, thinking that they can bypass any formal training in engineering, are buying mastering software and instantly calling themselves a "mastering engineer."

This is a dangerous trend.

Firstly, the mastering engineer brings a fresh pair of ears to a project that probably feels like it took an eternity to make. Having that unbiased perspective is priceless to any project.

Secondly, the mastering engineer is not only the last of what seems to be an interminable parade of engineers, but he or she is also a skilled technician/editor/musician who has spent many hours listening to music, and understands what people want to hear in their music, and how they want to hear it. They understand why a Latin mix should be bright and why a hip hop track needs to be bassy.

Mastering is understanding every item in your toolbox and knowing when to use it, how to use it, and even whether to use it. If one doesn't understand the principles of compression, how can one possibly use a compressor properly? If one doesn't truly understand "Q", also known as bandwidth, how can one properly equalize a mix without phase cancellation? Improperly mastered music sounds over-compressed, out of phase, and has too many highs and too many lows. And it's distorted.

This distortion is my biggest concern.

Because the music is so terribly over-compressed – thanks to plug-ins like the Ultramaximizer and others similar to it – one gets tired after just a few songs because of ear fatigue; without peaks or valleys in a song, the ear becomes physically tired and listening to the music become tedious. In addition, when one crosses the line with that software trying to make it louder and LOUDER, there simply isn't any more room for the sound file to fill, and it begins to distort. It is at that point that the output just isn't musical anymore. It's noise.

I am not against all the software that is now generally and affordably available to all; it is a wonderful tool for writers, musicians, and engineers.

When one spends the time learning about how to use these tools properly, as does a mastering engineer, it is amazing how wonderfully clear and professional-sounding music can be. After such a long process, wouldn't you want your project to sound its best?

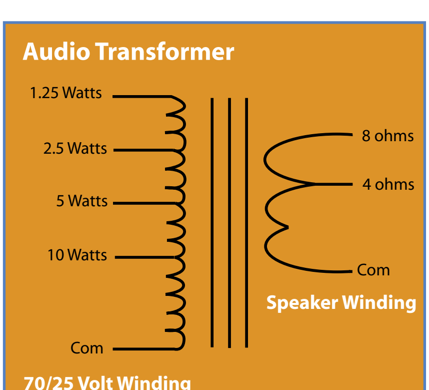
Marisa T. Déry, a native of Ottawa, ON, is the owner and engineer for Tamar Mastering in Boston, MA. A graduate of Berklee College of Music, her clients have included The Mighty Mighty Bosstones, Tugboat Annie, and RUSHYA. She has also mastered soundtracks and TV scores that have appeared on ESPN, TLC, Animal Planet, and in the Boston Film Festival. Also, she currently works in the Audio Preservation Studio at Harvard University. For more information, check out www.tamarmastering.com.

70-V Audio Distribution: Part 1

by Al Whale

When required to connect speakers over large areas with non-interacting area controls like halls or classrooms, the 70-V system is ideal, although it should be noted that some places consider 70-V systems to be unsafe, so the 25-V system is used instead.

Transformers are used at each speaker location to convert from the 70(25)-V system to the speaker impedance (eg. 8 ohms).



The 70(25)-V line from the amplifier is applied to the input of the transformer. The input selected is based on the maximum power needed from the speaker. Each speaker location comes from this same 70(25) volt line source (in parallel). The sum of the power setting of all transformers used should be less than the maximum power of the amplifier.

Be warned: if the total is over the maximum setting, the amplifier will be overloaded and there will no longer be a constant output. Switching a group of speakers in this situation will then affect the other speakers.

$W = E \times I \qquad I = \frac{V}{R}$

Therefore $W = \frac{E^2}{R}$

W = Watts E = Voltage
I = Current R = Resistance

For 70 volt systems

$E^2 = 70 \times 70 = 4900$
(or approximately 5000)

Therefore use

$W = \frac{5000}{R}$ or $R = \frac{5000}{W}$

From the above calculations, the 10-watt tap will be 500 ohms, and the 5-watt tap will be 1,000 ohms. When wiring, a smaller gauge wire can be used to go long distances without affecting the audio due to line loss.

Example: If the total load on the 70-V line is 100 watts, from the above formula, the impedance would be 50 ohms. Using the practice of 5% max, the wire would have to be under 2.5 ohms. Checking wire tables, for 50-ft. run, the wire would only need to be #22 gauge (1.614 ohms). For 500 ft., the wire would be #12 gauge (1.588 ohms). This is far easier than using 8-ohm lines – #16 & #6 gauge, respectively.

Al Whale is Broadcast Technologist and Assistant Chief Engineer at CHBC-TV. He also performs maintenance, design, and installation set-up. He has operated and taught sound in many church settings. Visit Al's website at: www.whalco.ca.

sound advice

Best Practices In Disk Keeping For Maximum Performance: Part 1

by Scott Leif

In a world filled with high-throughput applications such as those found in video editing, colour correction, audio mixing, and uncompressed playback, degraded disk performance can cost users thousands of dollars a day in lost revenue, a well-known but not frequently addressed area of content management.

Applications and workflow require that users add and delete content on a regular basis, sometimes as often as daily. This practice scatters bits and bytes of data all over the disks or RAID sets, which leads to negative impact on both read and write performance, and can result in erratic playback, as well as erratic application behavior.

To better understand the grey area of disk performance, we must first understand how a disk drive works when it comes to reading and writing. Let's take a look...

A disk drive writes data to blocks on the platters of the drives. The blocks begin at the innermost (centre) part of the platter and disperse out from there in a sequential order as data is written. In a RAID environment where multiple disks are used, data is typically written one block at a time, one disk at a time. So, if you can imagine the first block being "0", data would be written to block "0" of the first drive, then block "0" of the second drive, and so on. The more data that gets written, the further out from the centre of the platter it goes.

Since the blocks at the inner part of the rotating disks reach the heads of the disk drives faster than the blocks at the outer edge of the platter, there is a measurable performance difference in both reads and writes as the disks become more full. That is because data written to blocks further away from the centre require a longer length of time to access.

Fragmentation occurs when data is deleted and new data is written in place of the deleted blocks. Again the disk drive writes from the inside out and the first bit of data will be written to the first available block. When this occurs, the data effectively gets written out of sequence, which causes additional latency in both reads and writes. Compound that with a well-filled drive, and you have a recipe for a major slowdown.

In the film and post-production markets, where large sequences of frames are played back sequentially, fragmentation can be a huge headache for any user. It's very commonplace for facilities to add and delete hundreds of gigabytes and even terabytes a day. This practice, without the proper maintenance, has a tendency to degrade disk performance by more than 50 per cent. Pro audio applications such as Pro Tools can have the same effect on disk performance and can be equally as sensitive to fragmentation.

Next issue, Scott will offer practices you can adopt to keep your storage devices working at optimum level.

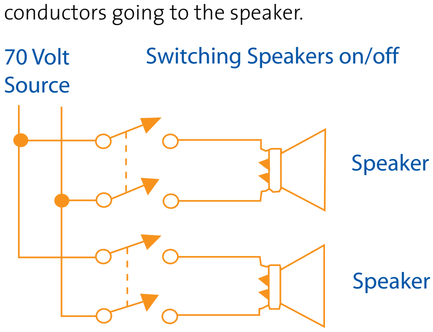
Scott Leif is President and CTO of Globalstor Data Corp., a leading storage technology provider for the professional audio and video, post-production, government, medical, education, and military industries. He is responsible for designing high-performance storage servers and storage area networks that are widely recognized among film and video, post-production, and audio professionals.

70-V Audio Distribution: Part 2

by Al Whale

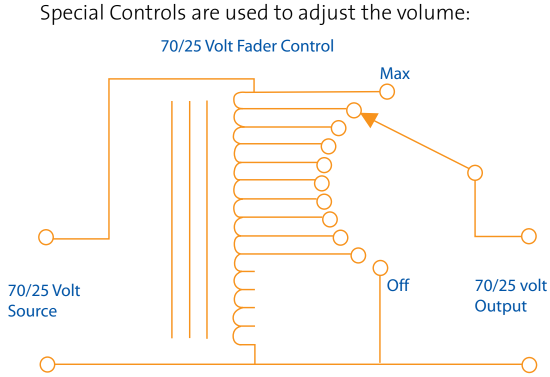
Switching And Controlling The Speaker Zones

Speakers can be individually switched (on/off) or switched in zones (such as several speakers in a hall). It is a good practice to switch both conductors going to the speaker.

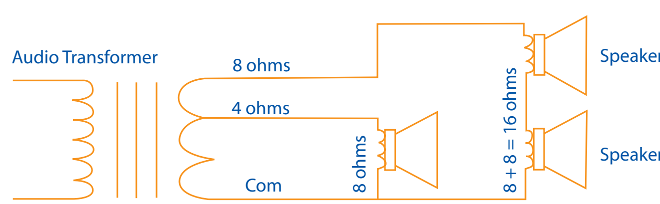


If designed properly, switching speakers on and off should not affect the levels in other speaker locations.

Special Controls are used to adjust the volume:



These controls will reduce the volume to the systems that they feed, still at 70 (25)-V. Although not usually done, use of series/parallel circuits can still be used on the 8-ohm output side of the transformer to limit the number of transformers used in low power situations.



An interesting method to wire three speakers:
the 8 ohm tap is half loaded & the 4 ohm tap is half loaded.
The tranformer sees 2x half loads i.e. a full load.

Summary

70 (25)-V systems are designed for multiple speakers in remote locations. For higher-power systems, use of the standard impedances (16/8/4 ohms) is still the best course. The 70 (25)-V system's frequency response is affected by the quality of the transformers used. A transformer rated at a maximum of 10 watts will probably saturate, giving distortion, at a far lower value if the frequency is 60 Hz. This will only be evident for loud low-frequency situations. If the system is designed with plenty of headroom, this will probably not be an issue.

Al Whale is Broadcast Technologist and Assistant Chief Engineer at CHBC-TV. He also performs maintenance, design, and installation set-up. He has operated and taught sound in many church settings. Visit Al's website at: www.whalco.ca.

sound advice

Best Practices In Disk Keeping For Maximum Performance: Part 2

by Scott Leif

Last issue, Scott offered some introductory information on disk drives and the headaches of fragmentation. Here's some advice on how to keep your disks running efficiently and effectively.

There are many tools available for dealing with fragmentation; some are even included with many typical operating systems used today such as Windows, which has an application built right in called Disk Defragmenter. Such tools can analyze the disks or RAID volumes to determine just how scattered the data is, where the blank blocks or sectors are, and then will proceed to reorganize the data so that it's in sequential order beginning with the innermost blocks, moving the blank sectors to the outer ends of the disks where they belong.

Many of these defragmentation tools will allow users to schedule defrags so they are done when the system is not in use. In many cases, defrags should be performed on a daily basis, depending on how much data or content is removed and written in relationship to how full the disks are. The fuller the disks, the more frequently a system may need to be defragged. By defragmenting, your system could find renewed performance.

Another tool that can have a big impact on performance, especially write performance on a system, is virus software. Most of us have had, at one time or another, the misfortune of dealing with a virus and have been forced to be prepared for future scares. What we do not realize is that the very software we rely heavily on to combat these threats can severely impact our business productivity, especially in write performance-sensitive applications such as uncompressed video capture as well as professional audio software.

This happens because the virus software wants to scan each file as it enters the system as well as when the file is opened, and that process is time consuming. Even a millisecond can cause a drop frame or a write delay. Either can require a user to start over only to have it happen again. Not only can this be frustrating and costly, but also confusing. Not being aware of the issue could cause a software problem to look like a hardware problem. Of course, the solution can be as simple as disabling the software during the use of performance-sensitive applications.

As the industry continues to change, so too will the issues and demands facing disk management. In response, more and more solutions will become available for overcoming such issues. Having the knowledge and appropriate resources in place before encountering any obstacles can protect your day-to-day operations and essentially ensure streamlined content management and productivity.

Scott Leif is President and CTO of Globalstor Data Corp., a leading storage technology provider for the professional audio and video, post-production, government, medical, education, and military industries. He is responsible for designing high-performance storage servers and storage area networks widely recognized among film and video, post-production, and audio professionals.

Live Webcasting From The Streets To The Corporate World: Part 1

by Brad Marshall

So you have been asked to do a live webcast for your company, friend, community, or local band. You ask yourself: "What's a live webcast?"

Don't worry! I am here to help you out. I've done live webcasts on Queen St. in Toronto, as well as the ivory towers of the corporate world. I started my career doing community events, which meant that I had no access to T1 or fibre connections with incredible amounts of bandwidth. Instead, I learned from the beginning how to do quality webcasts using standard household DSL and cable broadband. I started this work in 2002 when webcasting, or "live internet streaming" was in its infancy. Okay, let's get you started.

If you're in live production, you're already 75 per cent of the way there and that's good news, but before you can commit to doing a webcast for your over-excited parties, you must check the available bandwidth at your webcast location. Bring your laptop or make sure there is a computer onsite that is accessible to you.

Connect to the network and go to www.speedtest.net. This online speed test tool is fun and accurate. It's like looking at the speedometer on your car – and it can be a real adrenaline rush. Click on the geographical area that is near to you. Don't worry about the download – pay attention to the upload! Upload is important because you're taking your broadcast out of your location to a multimedia server. Upload speeds can be from 100 kbps to 1000 kbps or higher depending on where you are. Do this test several times, using different locations, then average the numbers. If your upload speed is 200 kbps (average) then you should be webcasting at 100 kbps. Why? Bandwidth fluctuates, and if you are broadcasting at 200kbps and there is a fluctuation, your webcast will be kicked offline, so you need to leave some room.

You are not quite ready yet. If you are in someone's home it's fine to disconnect any computers on the network before your broadcast. If you are in a corporate environment, you may be behind five firewalls and a suspicious IT department. Go to the IT department and explain what you are going to be doing. Make sure you understand the culture of the network and how it's used. Also, if you have to obtain a static IP (Internet Protocol) address, which you may need to do depending on the type of webcast you're doing, only the IT department can do this and it may take time to sort out.

Warning! If you test your connection two weeks before your broadcast and everything is fine don't assume when you return that it will still be set up for you. IT departments have huge responsibilities and things can change while you are gone. One more thing: Do not do a webcast using a wireless network ... just don't go there.

Brad Marshall is the Owner of Popular Minority Productions, which brings live events to the Internet (www.popularminority.com). He is currently writing a 10-week course on Live Webcasting to the Internet for Conestoga College in Kitchener, ON. He can be reached anytime at: brad@popularminority.com.

sound advice

Location CD Recording: Miking Techniques – Part 1

by Earl McCluskie

Location recording of non-live events has its pros and cons. On the pro side are natural acoustics, a unique sonic character that can give the recording a distinctive sound, prestige from the name of the facility, and sometimes lower rental costs. On the con side are external noises, little or no control over the early reflections and reverberation, difficulty isolating musical elements, and less than ideal control room monitoring conditions.

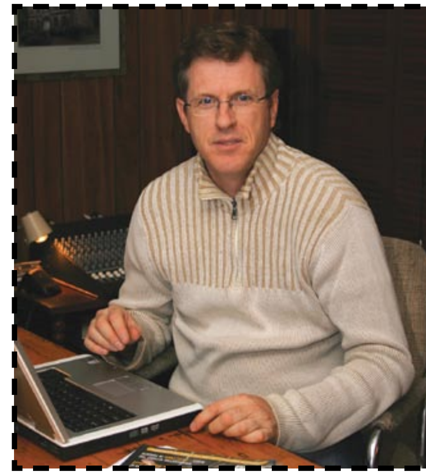
If the cons can be overcome, or ways to successfully deal with them found, good recordings can be made. These recordings do not have to be limited to just classical recordings, which typically are recorded in natural acoustics, or "live performance" environments. As an example, a 40-voice choir backed by piano, bass, and drums singing contemporary jazz-influenced music can be successfully recorded in a natural ambience.

The choir sound that one would naturally pick up in a church or concert hall using mic techniques associated with classical choral recording would have a significant amount of ambience and depth, suitable for that style of music, but not with the sort of warmth and presence that is associated with a contemporary "pop" sound. A good hall acoustic has a life and character that only the best studios can emulate, and so it is often worth finding a way to capture this sound.

Close-miking the choir would defeat the advantage of the hall by suppressing its natural attractive acoustic. Even the best cardioid pattern mics have significant colourations resulting from their uneven off-axis response, and these often do not compliment the room acoustics. A carefully-placed array of three or four omni mics over the choir can produce a natural-sounding pickup.

Make sure that choir members are as equally distant from the mics as possible, with the lower voices singing directly on-axis to the mic, and the higher voices projecting slightly below the 0-degree axis of the mics. The distance between the mic array and the choir will also depend on the ratio of direct to early reflection balance that sounds best.

Use two additional omni mics placed behind the choir to pick up the warmth of the choir, and give additional boost to lower male voices, which tend to be more omni-directional.



Earl McCluskie is a producer/engineer and Owner of Chestnut Hall Music, a music production company based in the Waterloo region of Ontario. The company specializes in location CD recording, both live and session. Recent projects have included Vancouver-based composer Timothy Corlis with the DaCapo Chamber Singers and the Guelph Symphony Orchestra.

In-Ear Monitors: Tips & Tricks – Part 2

by Keith Gordon

Picking up where we left off last issue, there is another important psychoacoustic effect to remember when working with IEMs. When IEMs are used in both ears, there is a "stereophonic" effect (stereo mix not required, just both ears used) known as binaural summation which yields a perceived 6dB increase in volume without any change in the level of either ear's input volume. This means the left and right IEM are each outputting 90dB SPL, but when both IEMs are inserted, our brain sums them together and we hear an equivalent 96dB SPL, yet without the hearing damage associated with those extra 6dB SPL.

You can try this experiment yourself by turning on your MP3 player, setting a level, and putting in one earbud. When you add the second, you will notice a substantial jump in level. The practical upside of this is to always use both IEMs and not just one like so many performers I see on TV. It makes me cringe to think of how much louder they are blasting their IEMs to get the same volume. Even worse, if they are using floor wedges in an attempt to get the "best of both worlds," they will be blasting their open ear too.

I learned another trick for wireless IEM users from Mike Prowda, monitor engineer for Nine Inch Nails and David Bowie. Prowda likes to use a compressor and limiter before the wireless transmission stage. Wireless systems have fairly narrow bandwidths in which each channel operates, so to best exploit what is available, it is important to aggressively compress and limit the signal before the wireless stage to keep it from overloading while at the same time not leaving any dynamic range unused – and therefore wasted.

This is similar to the approach radio stations take with their transmissions, using multi-band compressors and limiters to deal with different frequencies separately so that the overall energy level is controlled while not making the music sound overly squashed. At the time, Prowda was using Aphex Dominators, though there are similar units that can also handle this multi-band compression approach. For those of you without access to advanced tools such as this, try experimenting with whatever compression you do have before the wireless stage to see if you can find improvement. If you have any questions, please drop me a line.

Keith Gordon is a veteran audio engineer who helped develop a DSP-based hardware/software IEM system (inearsounddesign.com) in conjunction with Westone Laboratories. He can be reached at keithgordonca@gmail.com.

sound advice

Location CD Recording: Miking & Mixing Techniques – Part 2

by Earl McCluskie

When recording, for example, a contemporary choir backed by piano, bass, and drums in a natural hall environment, the backup will be picked up by the choir mics, and will sound boomy and unfocused if not carefully controlled. Fortunately, most halls have a built-in solution: reception spaces and other rooms with doors opening into the hall. The drums and bass can often be located here. If the bass is acoustic, some sound will enter the hall, but considerably less than if the instrument was in the hall. Communication for the instrumentalists, and conductor if necessary, can be achieved with headphone fold-back and video monitors.

Typically in such a space, one would mic the piano by putting mics inside the piano, closing the lid, and perhaps even encasing the piano in packing blankets. This produces a distinctive sound, but does not take advantage of the natural piano sound in the room. Instead, position the piano with as much distance from the choir as is possible, and balance its pickup with the leakage into choir mics. During sound check, experiment with your post-production plans (EQ, compression, etc), as getting the right balance has to happen now.

Miking technique alone will not give you the kind of control you need to produce a full "studio" sound. Once you have captured a full choir sound, you will find that the room characteristics will define the choir sound as being in a natural acoustic space, and this will not balance well with the drums and bass.

An old trick used for "fattening" up the sound of a guitar involves double tracking the sound source, applying compression and expansion to one track, and then mixing the two together. The choir pickup will have unneeded bottom end from the omni mics, so in the processed track, much of this can be rolled off. You can also narrow the stereo width of the processed version of the choir, using the unprocessed original to create a sense of depth and width. You can delay this track as well, although care must be taken with possible phase cancellations, leading to an unnatural choir sound. Also, any processing done to the choir sound will impact the piano sound, and vice versa.

The sound of the hall has now become an integral part of the choir sound, and can be blended with the backup ensemble tracks, sweetened with appropriate reverb.

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Four Key Tips For Mixing An Outdoor Festival

by Pete Bartlett

Bartlett with the Digidesign Venue

1. Well, outdoors or indoors, you apply your modus to your mix. We all have our tricks we've picked up along the way, and so just because you're outside doesn't mean this should change.

2. It's always nice not to be fighting with some dreadful hall or bad mix position, which is helpful, but it means that outside there are less excuses.

3. If you have a good system tech (the guys at Toronto's Virgin Festival were superb), you have to trust him.

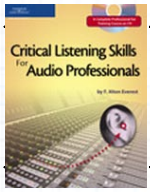
4. Most of all, remember you're not mixing for audio engineers – you're mixing for kids who have the record and want that same experience, only bigger, better, and louder. Be bold.

I don't dig the "chin scratching" static-sensible mix, where people are going, "Hmm ... nice mix." Give me excitement any day. I try to remember the way I felt when I was 17 (that was some time ago), when your life depended on this gig. If the kids walk out saying, "That was fucking amazing," then the job is done...

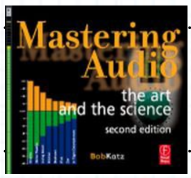
Pete Bartlett is FOH engineer for UK-based indie rockers Bloc Party, recently in Toronto for the Virgin Festival. He can be reached at fohpete@aol.com.

RECOMMENDED READING

ASSISTANT ENGINEERS HANDBOOK – SECOND EDITION BY TIM CRICH
Packed with Proven Recording Studio Secrets. Key Priorities for Before, During and After the Session. Required reading in dozens of audio engineering courses at college, institutes and universities across North America.
http://musicbooksplus.com/bassistant-engineers-handbook-second-edition-p-10893.html

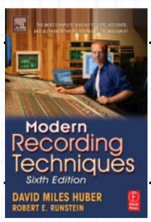


CRITICAL LISTENING SKILLS FOR AUDIO PROFESSIONALS BY F. ALTON EVEREST
With this course you can acquire the audio discernment skills of a seasoned recording engineer by studying this course at your own pace, in your own home.
http://musicbooksplus.com/bcritical-listening-skills-audio-professionalsb-p-7162.html

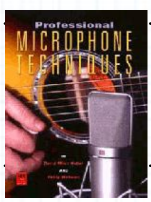


MASTERING AUDIO – THE ART AND SCIENCE, SECOND EDITION BY BOB KATZ
Mastering Audio gives you a thorough introduction to the unique procedures and technical issues involved in mastering.
http://musicbooksplus.com/bmastering-audio-science-second-editionb-p-9278.html

THE MIXING ENGINEER'S HANDBOOK, SECOND EDITION BY BOBBY OWSINSKI
You will learn about the history and evolution of mixing, various mixing styles, the six elements of a mix, the rules for arrangement and how they impact your mix, where to build your mix from, and mixing tips and tricks for every genre of music.
http://musicbooksplus.com/bthe-mixing-engineers-handbook-second-editionb-p-7161.html

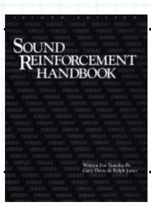


MODERN RECORDING TECHNIQUES, SIXTH EDITION BY DAVID MILES HUBER & ROBERT A RUNSTEIN
Modern Recording Techniques provides everything you need to master the tools and day-to-day practice of music recording and production.
http://musicbooksplus.com/bmodern-recording-techniques-sixth-editionb-p-5758.html



PROFESSIONAL MICROPHONE TECHNIQUES BY DAVID MILES HUBER AND PHILIP WILLIAMS
Professional Microphone Techniques intensely focuses on microphone usage for dozens of different instruments as well as vocals, amplifiers, Leslie cabinets and more.
http://musicbooksplus.com/bprofessional-microphone-techniquesb-p-1396.html

PROFESSIONAL SOUND REINFORCEMENT TECHNIQUES BY JIM YAKABUSKI
Professional Sound Reinforcement Techniques gives unique insight into a wide variety of general and specific live sound topics, from PA system setup and band politics to zone equalization and signal processing.
http://musicbooksplus.com/professional-sound-reinforcement-techniques-p-2882.html



THE SOUND REINFORCEMENT HANDBOOK BY GARY DAVIS & RALPH JONES
The Sound Reinforcement Handbook features information on both the audio theory involved and the practical applications of that theory, explaining everything from microphones to loudspeakers.
http://musicbooksplus.com/bthe-sound-reinforcement-handbookb-p-458.html