34 All Kinds of Noise 7 January 2021

I'm Doug Fearn and this is My Take On Music Recording

Back in the days of magnetic tape recording, I was in a constant battle with noise. Tape machines have a signal to noise ratio of about 65dB. Compare that to modern digital recorders where the noise is so low, it isn't even a consideration.

We had to use all sorts of techniques to minimize the intrinsic noise from the tape. And the more tracks you used, the more the noise built up. This was not a huge problem with music that started loud and stayed loud to the end. But with music with a wide dynamic range, tape hiss would probably be audible in the quieter parts of the song.

Engineers back then had a variety of ways to minimize the noise. The easiest way was to keep the level up. Tape can take a lot of level before it sounds bad. Sure, the sound changes at higher levels, but many producers wanted that kind of sound. We learned to keep the level up as close to 0 VU as possible, and if peaks went above that, it wasn't a big deal. That's in contrast to digital where once you exceed digital zero, the sound turns very ugly very quickly.

That's why many of us who go back to the tape days have a difficult time breaking the old habit of maximizing the level when using a digital recording system. I think I am finally past that stage and I no longer feel uneasy when the level is only peaking at -12 or even less.

In addition to keeping the level up on tape, most of us also realized that we should do as much eq and processing as possible before the recorder. If you boosted the highs during the mix, for example, you would also bring up the tape hiss. Same for limiting. We got used to doing at least some processing right from the mic.

Another technique I used, although I don't know how common it was back then, was to boost the highs while recording beyond what the track needed. During the mix, the same high eq would be used to cut the highs to bring the response back to normal. Consequently, the noise level would be reduced.

However, after a lot of overdubbing, you sort of got used to that high-end boost and it was sometimes difficult to bring yourself to roll off the highs in the mix.

Dolby Labs developed their game-changing noise reduction system in the 1960s, and that made the noise much less of a problem. That was done by using multi-band compression on the audio going in, along with some boost in highs, and then expanding the audio with an expander and eq that exactly un-did the encoding used to record. The compression was only on low-level signals, so it did not have any effect on the louder portions of the recording.

It worked great, but it did require excellent maintenance and careful alignment on every reel of tape, or else you would hear the imperfect tracking on the playback. Studios with lousy maintenance hated Dolby noise reduction, but for those willing to attend to the details, all day, every day, the noise was reduced by a subjective 10-15dB, and it was totally transparent.

Near the end of the tape era, Dolby Labs developed an even better noise-reduction scheme that made tape at least as quiet as the 16-bit digital recorders of the day.

Unless you are using tape, that's just history. So, let's look at where noise comes from in modern recording situations, and how we can minimize it.

First, for this discussion, I define noise as any sound on the recording that shouldn't be there.

24-bit digital recording is extremely quiet and noise is not an issue. But when your recorder is virtually noise-free, other noises you might not have noticed before become apparent.

We can start with the bad news: There is no way to achieve absolutely no noise in our recordings. And one source we have no control over whatsoever is what is called thermal noise. This is a law of physics that says that the noise made by moving electrons, which we use in the amplification and manipulation in our **electronic** equipment, is unavoidable.

Where does this noise come from? Well it turns out that electrons flowing through any conductor or amplifying device will have a certain amount of noise, depending on the temperature. The only way to reduce this kind of noise is to cool down the equipment. A lot. For radio telescopes and deep-space data reception, the receiver "preamplifiers" are cooled in a liquid gas to as close absolute zero as possible. The noise improvement is enough to pull put incredibly weak signals that would otherwise be buried in the noise of the equipment.

That's not practical for our studio preamplifiers, but it sets a theoretical minimum amount of noise in our studios. This is highly simplified, but just to give you a sense of the lower limit of noise we can expect, across the audio spectrum, the lowest noise we can achieve is around -130dBm at normal room temperature. That's pretty quiet, and most modern mic preamps, tube or solid-state, can easily get within a few dB of that. The noise depends on the preamp gain, and by the time you are up to 70dB of gain, the inevitable noise may become a problem.

Noise is cumulative and comes from each device in our chain from mic to recorder. They add up arithmetically, so the lowest theoretical noise might be increased a few more dB by each eq, limiter, or other hardware device we add to the track.

I don't have a lot of experience with software plug-ins, but if a plug-in is based on sampling actual hardware, then at least some of the noise from the device is likely to be incorporated into the plug-in. This is usually not a problem, since presumably a lot of care was taken in the modeling process to make sure the noise is minimal.

Today, we can discount the noise component added by the recorder, but the analog-to-digital converters will also add a small amount of noise. Same with the digital-to-analog converters.

Still, today's equipment is quiet enough that the resulting noise should not become a problem.

However, it can be a problem if you do not pay attention to the levels at each stage of the process. This is commonly called "gain staging," but I prefer the term "gain structure."

This is really pretty simple, and there are only a couple of rules to follow.

First, apply as much gain as early in the chain as possible. Sure, there are going to be times when this is not the best strategy, but usually it works best to have most of your gain in the mic preamp. It also goes back to the microphone. For example, if the solid-state condenser mic has a built-in pad, do not use it if you don't need it. Some condenser mics will overload on extremely loud sounds, and that creates distortion that is not pleasant. So, use the pad if you have to – but make sure it is the mic that is overloading and not the mic preamp. How can we tell? Well, you have to know your gear and recognize the difference between mic electronics overload and mic preamp overload.

If your mic preamp has a meter or overload lights, that can help you determine this.

Note that vacuum tube condenser mics do not have pads because they would almost never be needed.

Next, if the mic preamp has a built-in pad you can select, don't use it if you don't need it. If you have to run the preamp gain very close to all the way turned down, that might be a situation to use the preamp input pad. But don't use it if you are getting the sound you like. Pads can change the sound of some mics. In my recording, I almost never use the preamp pad. Of course, I don't do much recording of screaming vocals or guitars, nor close-mic'd drums.

Next, adjust the gain of the mic preamp to a good level to your recorder. Some DAWS have adjustable gain or pads on their inputs, which should be set to minimum gain necessary to get a good level. Same with the pad – don't use it unless you have to. And you shouldn't have to if the preamp level is set properly.

This also gets into the reference levels set by your DAW or converter. These are set-and-forget adjustments that you may have made when you installed the converter. The default level setting may not be optimum for your workflow. You will have to experiment with this to get the best setting, but remember, no more gain than necessary.

If you are going through outboard gear before the converter, or as a hardware insert, use the same rules, if they have adjustable gain. For example, if your compressor/limiter has a make-up gain control, generally you want to run this at the level that gives the best level to your converter, not too low and not too high. Same with eq. Boosting frequencies with your outboard eq may make the level too high for the converter, but you would have to be doing something extreme for this to be a problem.

Paying attention to gain structure will not only reduce the noise as much as possible, but it should also optimize the dynamic range of your gear.

And speaking of gear, these concepts also apply when designing a piece of audio equipment. Any audio device that has gain stages also needs to have optimum gain structure. I want to make it nearly impossible for a user to make any of the D.W. Fearn products sound bad. And to achieve that, I have to be very careful how the various stages work together.

It reminds me of how airplanes were designed in the pre-computer days. Well, one aspect of airplane design, anyway.

Airplanes have to be a strong as possible, but also as light-weight as possible. The old way to do that was for an aeronautical engineer to use his education and experience to draw and fabricate his best-guess at the optimum design. Let's say it is an airplane wing.

They would then support the wing the same way it would attach to the rest of the airplane, and then start piling sandbags on the wing until it failed. Then they would analyze the pieces that broke and make those stronger. Any parts that didn't fail would be made lighter and less strong.

The process was repeated over and over until they got to a point where all the pieces of the wing failed at the same time, when the last sandbag was added. Now they had the strongest wing with the lowest weight possible.

Today, of course, this is all done on a computer. No need to break anything to get the solution.

I approach the gain structure in my audio designs the same way. I see what the first stage to overload is in the amplifier, and modify that stage so it doesn't overload. Then I see what the next stage to overload will be as the level is increased. And so on. The goal is to have all stages overload at the same very high input level.

I can also use computer modeling to help with this process, but I want to prove it in the real-world, too.

Of course, it is not quite that simple, because there are usually gain controls somewhere in the signal path, which will affect where things overload. But you get the idea. Think the same way in your studio setup and you can achieve the best performance from your gear.

Another noise source that falls within our broad definition is actual, unwanted acoustic noise. This can come from many sources, such as outside noise that leaks into the studio because of inadequate isolation. I covered that in a couple of previous episodes, but as a summary, it takes mass – heavy walls – that are perfectly airtight to reduce outside noise to a point where it won't be heard on your recording. The type of music you record also has a big effect on this, too. And remember that noise is cumulative, so each mic that is open, and each track that is overdubbed, can add another layer of noise. If you want more detail, listen to episode 4, "The Room Where Music is Recorded," and episode 13, "Recording In Improvised Spaces."

There can be acoustic noise sources within your studio, or near enough to cause audible noise on the recordings. The better your sound isolation, the more these other sources will become obvious.

Your HVAC system is probably the biggest source of noise in the studio. I don't want to go into HVAC design in this episode, but perhaps I will in the future. Just be aware that moving air always makes noise, and the way the minimize that noise is to use low velocity air. That means ductwork with a large cross section. And the grilles where the conditioned air comes out, and return ducts where the air leaves the room, should offer as little resistance to the air as possible. My solution was to not have any grilles at all. Painting the ducts flat black inside will make them less visible.

The same rules apply to the ductwork in the control room system. In my case, I have separate systems, but not much control over the air noise in the control room HVAC. So I turn it off during critical listening times.

The HVAC fan motors, the furnace, and air conditioning compressor may also be audible. Locate all that stuff outside the sound-isolating walls of your studio, if possible. In my current studio, that is not practical, so I simply shut off the HVAC system entirely while we are recording. Whenever there is a break of more than a few minutes, I turn it back on.

The nature of sound isolation and acoustic control in studios means that they are very well insulated thermally, which helps keep the space comfortable if you have to turn off the HVAC during recording.

Well-designed studios have separate HVAC systems for the studio and the control room. This is advantageous for many reasons, especially sound transmission between the rooms. This can get pretty complex in a large facility with multiple studios and control rooms.

If you are recording quiet music, you might be surprised what the mics pick up in the studio. In my case, the building's water supply comes from a well, and the electrical control for the well pump is pretty close to the mics in the studio. The pump itself is about 150 feet away and over 600 feet down in the earth, so the motor is never heard. I never hear any running water, but there is a relay that engages to send power to the pump. Every once in a while, there may be an audible click from the relay that gets into the recording. Not a big enough problem to ruin a take, but I can hear it.

I have designed home studios for some very serious musicians, and in one case, I had to design enough isolation for his basement studio so that the family with several kids could carry on their indoor activities without interfering noise in the studio. Fortunately, he had the budget to do it right. One of the more challenging aspects of this studio design and construction was a cast iron wastewater pipe that ran along the back wall of the studio. Despite heavy interior walls, there was still noise when a toilet was flushed. We solved that by wrapping the pipe in lead-loaded vinyl.

Some noise in the studio is unavoidable because you have human beings working in there. Chairs that squeak, even a tiny bit, can be a problem. And just people breathing adds to the noise floor in the room. Players with a lot of studio experience know how to keep this noise down, but what about the photographer or videographer documenting the session? Needless to say, cameras with mechanical shutters can't be used while recording. And if the visual staff is moving around during the recording, make sure they understand that they have to move silently. I've even had takes ruined by a professional video camera with a built-in cooling fan.

Microphones can also be noisy. Mics with active electronics, condensers, but also some ribbon mics, are going to contribute a certain amount of electronic noise.

Some mics are also susceptible to electrical noise in the studio environment. Condenser mics are pretty immune to this, but ribbon and dynamic mics can easily pick up electrical noise.

We can use the electric guitar as an example we are all familiar with. The nature of an electric guitar pickup makes it a pretty good detector of any electrical noise in the studio. You probably know this from watching the player turn this way and that to minimize that noise. Taking his fingers off the strings may also increase the noise, since that can also change the noise susceptibility.

Some players have developed a habit of turning down the volume knob on the guitar when they are not playing. That reduces the noise and is appreciated by everyone. It works well if the volume is normally going to be maximum, so it is simple to re-set it for the next take. But if the control is optimally set somewhere in the middle, the chances of the player setting it exactly the same each time can be pretty low. That not only changes the recording level, but it also changes the sound of the guitar, which may create problems if you have to make a composite of various takes.

Where does this noise come from? In the old days, the main source was florescent lights. In my studio, we have florescent lights, which are handy for setup and teardown, or for maintenance and cleaning. But for recording, we switch off the florescents and use incandescent lights instead. Incandescent lights do not generate any electrical noise, so despite their heat and inefficiency, they are the lighting method of choice for recording.

The incandescent lights are on dimmers. But not your typical household or even industrial dimmer. Those dimmers create intense electrical noise, although some are better than others. None of those type of dimmers is ever going to be totally quiet.

They work by chopping up the voltage into small slices. If the slices are wide, the lights are brighter. As the dimmer is turned down, the slices get narrower. Why is that a problem? Well, those slices are created by a high-speed oscillator, running at a frequency above most people's hearing range. But in the process, the dimmer creates terrible voltage spikes that radiate from the wiring for the lights, and that noise can be picked up by highly sensitive pickups in a guitar or bass, and by ribbon and dynamic mics.

The answer is to use rheostat dimmers. They are big and very expensive, but they make no noise at all. These dimmers are really just a big pot, like all the gain controls on analog gear. Scale that pot up to handle the many watts of lights, and you have a rheostat dimmer.

What about LED lights? I think using LED lightning in general is a good thing, since LED bulbs use far less power and generate only a fraction of the heat that an incandescent bulb does. They also last a lot longer.

“LED” stands for light-emitting-diode, which are solid-state devices that produce light when a DC voltage is applied. The voltage required by an LED is very low – a few volts at most. Each individual LED only puts out a small amount of light, so for illumination purposes, many separate LEDs are combined. And to make an LED bulb compatible with our 120 or 240 volt electrical system, the mains voltage must be converted to a much lower voltage.

The conventional way to take AC mains voltage and convert it to the low voltage DC for the LEDs is to use a transformer to lower the voltage, a rectifier to convert AC into DC, and a filtering system to make the rough DC into smooth DC. That requires large, relatively expensive parts. The converter to power an LED lightbulb would be larger and heavier than the bulb itself. Not practical.

But there is another smaller, cheaper way to do it, and that is to use a switched-mode power supply. These work differently, first converting the mains voltage to DC and using that DC to power an oscillator tuned above our hearing range. That oscillator output goes through a tiny transformer, which can be much smaller at the higher frequency. The output of the transformer is rectified again to change the voltage to DC. The efficiency is excellent and this type of supply generates very little heat.

There are several variants of the circuit described, but the end result is the same.

Switched-mode power supplies are sometimes simply called switching supplies. And they are ubiquitous in modern equipment. The little block charger for your phone is a switching supply. A conventional supply would be bigger than the phone.

Your computer, any wall charger, TVs and video monitors, and even your appliances all use switching supplies. Some pro audio equipment does, too, but I do not use them in our products. They are a great invention, which actually goes back to the 1960s but only since around 2000 have they become the power supply of choice when low cost and small size are the primary concern.

What's not to like? Well, basically, I hate all switching supplies. And the main reason is they are a huge source of electromagnetic interference, or EMI.

Our environment is highly polluted with the electrical noise these supplies generate. And this is a concern for audio devices, especially in the studio environment.

The electromagnetic interference they create comes about because, much like the cheap light dimmers I mentioned earlier, switching supplies use high frequencies above our hearing range that are chopped into tiny chunks to do their job. The waveforms generated are full of artifacts and harmonics, which can extend way into the GHz range. Switching supplies can cause interference to our sensitive audio circuits, but also to any wireless device. Your Wifi and cell phone could even be affected.

It is possible to build switching supplies that reduce the intensity of this noise, but it might cost a few cents more per supply, and for mass-produced consumer equipment, the manufacturer is not going to spend that extra money.

Sometimes you can reduce the amount of interference with a carefully designed filter that is placed in the electrical power lines. Needless to say, this is not practical for most of us to do on our own, and electricians are unlikely to have a clue about this. Unfiltered, the noise can travel throughout the entire building, via your AC wiring.

As an experiment, I tried using a conventional light dimmer on my studio lights. It was one that was advertised as having very low electromagnetic interference, so I thought it was worth a try. It wasn't cheap, either – around $60 as I recall.

Maybe it was quieter than most dimmers of that type, but it was far from acceptable in the studio environment.

Since the dimmers and the switching supplies use similar principles and generate similar noise, I think what I discovered will apply to both types of interference.

I used a portable AM radio as my noise detector. Although the AM broadcast band is way above audio frequencies, the noise is just as prevalent on the radio as it is in our audio devices.

I was not surprised that the noise was clearly audible on the radio when I was in the studio. But I wanted to see just how far it went. I started walking around the rest of the building, which has two floors above the studio level. The noise did not diminish at all, no matter where I went. It didn't disappear until I exited the building. This might not be typical in all situations, but our building has stucco outside walls, applied over metal lath. The lath acted as a Faraday shield, which blocks most radio frequencies.

The point is, the noise was everywhere, and didn't change in intensity anywhere inside the building.

This means that the source of interference, whether from light dimmers or switching power supplies, could come from anywhere in the building. Or even from an adjacent building.

Getting back to our LED lights, they work because a small switching supply is built into them. That supply converts the mains voltage, 120 or 240 volts, to the low voltage the LEDs need. In the process, they spew all sorts of electromagnetic noise into your studio. It will definitely affect guitar pickups, but also other pickup-based instruments like bass guitar or some types of electric pianos.

And it can get into your ribbon and dynamic mics. In a worst case, even condenser mics can pick up this noise.

What does this sound like in your monitor speakers? It's not unlike the noise you hear from a guitar pickup. It consists of many frequencies, which tend to have large spikes of voltage. And the character of the noise tends to change with the load on the power supply.

Well-designed and installed audio wiring in your studio should be largely immune to this noise, but it might not be if the wiring isn't done properly. More on that in an upcoming episode.

It is ironic that LEDs do not generate any noise at all. It's the power supply that is the problem. If we could run our lights on pure DC, they would be entirely acceptable. But there isn't much of a market for DC LED lighting, and most of it is designed for marine use, or in an RV. It's really expensive compared to conventional LED or incandescent bulbs, and requires special wiring. I may look into doing that in my studio at some point, but it's not exactly my highest priority in studio equipment.

Those switching supplies that power so much of our modern equipment are often designed to shut down when the device they are powering is disconnected. But some of those blocks and bricks continue to generate noise, even when nothing is connected to them, as long as they are plugged into a wall outlet.

The absolute worst noise I have encountered comes from rapid chargers for power tools.

What can we do about all this noise from these devices? Well, the simplistic answer is to not use them. That's impossible today. But you can minimize this source of noise by unplugging them when not in use, if practical. And keep them out of the studio and control room as much as practical.

That's really not possible, since every studio has computers and video monitors. Fortunately, most high-quality computer power supplies are pretty electrically quiet, and so are many screens. But keep in mind that they can cause noise problems. This can be an issue if you have mics in close proximity to your screens.

I ran into a problem one time while recording the voiceover track for a YouTube video. I was using a laptop computer as a teleprompter, and one of my ribbon mics. My semi-pro video camera was just above the laptop screen by about a foot.

I recorded a test segment and went back to the control room to listen. It was full of noise! I had the laptop on its power adapter, which of course was a switching supply. Running the laptop on its battery dropped the noise to less than half the level, but it was still obvious. It was the screen that was causing the noise.

I discovered that the video camera was also generating electromagnetic interference. I ran it on its batteries, too. Still too noisy. Ultimately, I changed the mic to a condenser and the noise problem was solved.

I can use the same ribbon mic just as close to the video screens in my control room without any noise.

Keep phone and other chargers out of the studio as much as possible. One thing I did in my studio was to put in a long strip of USB charging outlets, using a supply known to be quiet and with all the AC wiring inside of a steel conduit. This system produces zero detectible noise. I have to remind people in the studio to use those outlets and not their own charger. Everyone seems to come to a session with a dead phone battery. There is still potential interference from the charging cable going to the phone (or camera, etc.), but I supply good quality shielded charging cords for Apple and Android phones and interference has not been a problem.

Cell phones themselves can be a source of noise in the studio. You have certainly heard the raucous noise that a cell phone can induce into audio devices. This happens at intervals, when the phone needs to connect to the nearest cell tower to let it know where it is. I hear that noise all the time on radio and TV broadcasts, and podcasts, when the people involved fail to turn off their phones.

I suggest telling everyone to put their phones in airplane mode while in the studio. Just muting the ringer won’t stop the electrical noise.

The devices most susceptible to mobile phone noise are guitar pickups and similar, and lavalier mics, especially wireless ones. These use an unbalanced cable from mic to the transmitter, and unbalanced wiring is highly sensitive to interference. A guitar cable, for example, is unbalanced.

Another potential source of interference is solar panels. I’m all for solar energy, but the current systems radiate a huge amount of electromagnetic energy. I have not heard of these systems interfering with studio equipment, but it is certainly a possibility. Maybe it is happening but people do not suspect the solar panels.

Ironically, solar cells should be the most electrically quiet power source available. The panels produce pure DC. Really pure, like a battery. But once again it comes down to cost. To convey the low voltage DC from the panels to the storage batteries in an ideal system, very large copper wires must be used. That adds cost. So virtually all mass-produced solar systems use, you guessed it, switching power supplies at each panel. They boost the voltage instead of converting it to a lower voltage, as in the types of supplies I talked about earlier, but the principle is the same. That way, the system can use much smaller copper wires from the panels to the house wiring. That reduces the cost slightly.

But, of course, the switching supplies generate a lot of noise. In fact, at radio frequencies the interference can be heard for miles in some cases.

Unfortunately, there's not much you can do about it.

And speaking of radio frequencies, it is sometimes possible for our sensitive audio equipment to pick up radio transmissions of many types. Most obvious could be from a public service vehicle passing by your building while the driver speaks on the radio. Or you might get interference if your studio is located near the transmitter of an AM, FM, or TV station.

Poorly designed studio wiring and some deficient equipment can detect this transmission and convert it into audio that you can hear from your speakers. Sometimes it just sounds like white noise.

If you hear that kind of interference, there is something wrong in your setup.

But in extreme cases, even good cables and wiring may pick up radio frequency interference, or RFI. When I was developing our VT-1 mic preamp, a friend of mine wanted to try it out. I was happy to let him do that, especially because his studio was located in a section of Philadelphia where all the radio and TV transmitters were located. There were six huge towers over 1000 feet high, along with numerous other towers. About 30 FM and TV stations broadcast from there, radiated many megawatts of energy into the air. There was also a 50kW AM station there, along with just about every cellular service, pagers (this was a long time ago), public service, government, and private communications systems.

I knew this would be a real test of the VT-1. If it performed flawlessly there, I was sure it would do so anywhere.

My previous experience as an Amateur radio operator, plus a few years working in broadcasting, taught me a lot about protecting equipment from RFI. I incorporated that knowledge into the VT-1, and all the products after that. This also allowed the products breeze through the CE certification process, which subjects products to intense radio fields over a wide frequency range.

The studio owner told me that the VT-1 was the only piece of equipment he ever had that did not require a lot of work to eliminate the rf interference. By the way, he soon moved his studio to a more electrically quiet location.

One other potential source of noise might be a magnetic radiated field from some of your gear into more sensitive gear mounted in close proximity. Examples would be putting a mic preamp just above or below a big power amp in a rack. This could also happen if you had something with motors in it, like a tape machine, too close.

In this situation, the problem is that the power transformer in the power amp, or the motors in a tape machine, invariably have an alternating magnetic field around them. Good design minimizes this, but there is always some magnetic field radiated.

If this field impinges on sensitive circuitry in your nearby gear, you might hear this in the monitors as a hum in your monitor speakers. The hum will be at the mains frequency of 50 or 60 Hz. The solution is simple: put more space between those two pieces.

Poorly designed equipment can be more susceptible to this kind of noise.

By the way, in the episode on distortion, I mistakenly said that deteriorating filter capacitors can cause hum in the audio at the mains frequency. I realized I said that after the episode was published. The correct description of that hum is that it will be at twice the mains frequency, either 100 or 120 Hz.

This can help you determine whether the hum you hear is from a radiated magnetic field, at 50 or 60 Hz, or bad filter capacitors at 100 or 120 Hz.

Sometimes when recording an electric instrument with a DI or direct box, plus an amp in the studio, you will get a very obvious and annoying buzz at certain odd harmonics of the mains frequency. This can occur especially when the studio outlets and the control room equipment are on separate branch circuits from the electrical panel circuit-breaker box. I won't go into the technical explanation, but the usually effective solution is to use the "Ground Lift" switch on the DI or direct box.

For the same reason, in your control room, all equipment should be connected to the same branch circuit. That is, the same circuit breaker. That will ensure that you have the lowest risk of that type of noise in your audio equipment.

This noise is sometimes called "ground loop" noise, but that is not a technically accurate term.

It's probably not possible to run everything in the control room off of one 15 or 20 Amp circuit, so you will probably need at least two separate branch circuits. That can work OK if you make sure the two AC power wires run along exactly the same path for their entire distance, from breaker box to the equipment outlets, and the two wiring runs are kept close together.

In my control room, there are actually four separate branch runs. Three are for audio equipment and computers, and the fourth is just for lighting. I have no noise issues from the AC power at all. I do not use any kind of power conditioning or other manipulation. It's not needed.

One last source of noise is what I will call "bad contact noise." In episode 33, "All Kinds of Distortion," I talked about TRS patch panels and their potential to introduce distortion into your audio. In addition to distortion, patch panels can also produce noise when the connection is degraded by oxidation or worn-out jacks.

This can also occur if you have a bad cable or connector. It can also happen within your gear, especially if there are interior connectors that couple the audio from one part of the device to another, or to panel controls.

That's one reason why I have no connectors at all inside the products I design.

You've probably heard the sound a defective cable can create, especially when the connection is intermittent. When a shield connection is disrupted, it makes the cable much more susceptible to electromagnetic interference. And if it's a mic cable for a condenser microphone that uses 48 volt phantom power, the mic can intermittently lose power. And that creates a terrible noise that can actually damage your speakers if you have the monitors turned up loud.

But most of the time, the poor connection in any of the wiring will cause the level to fluctuate, along with a crackling or popping sound in the monitors. More on this in an upcoming episode.

With thousands of D.W. Fearn products in use around the world, we get a few customers each year who complain about noise that they attribute to a product we build. We try to help them do some troubleshooting over the phone or by email, to make certain that our product is the problem. If it seems probable that the noise is coming from one of our units, we get the customer to ship it back to us for repair.

I would estimate that over 80% of the time we cannot find any defect in the equipment. A return phone call typically results in the customer telling us, that, yes, it was a problem in the studio. Most often it is in the patchbay, but wiring and connectors are a close second.

I plan a future episode on troubleshooting problems in your studio, to help people track down these issues and correct them.

Noise can be a struggle to control, even though most of us have moved away from tape. Today, our noise sources are different, and perhaps more of a challenge than tape hiss. Understanding the sources of noise can help you minimize their effect on your recordings.

There is a transcript for this episode. If you want a written version, you can download a PDF version from dougfearn.com

And please keep the suggestions and comments coming. Your feedback helps me determine what I should talk about.

If there is sufficient interest, I am considering having an occasional question and answer episode. If you have something you would like me to answer, record it in your studio with your best equipment. In keeping with the high audio quality goal of my podcast, you can record your questions at 24-bit, 96kHz sample rate and send the file to dwfearn@dwfearn.com

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