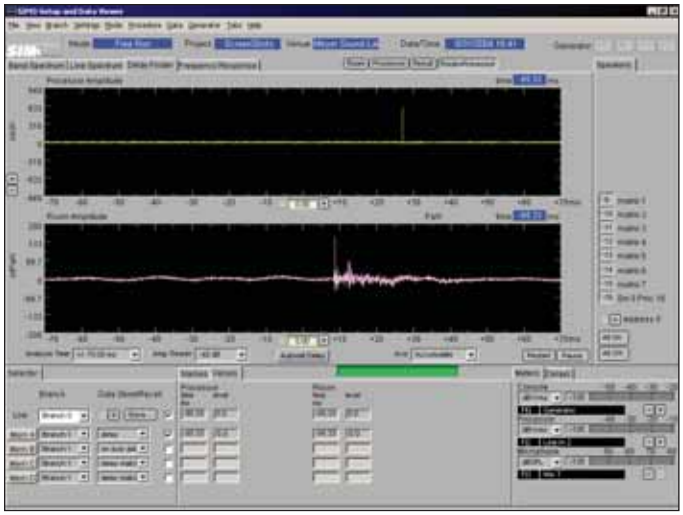
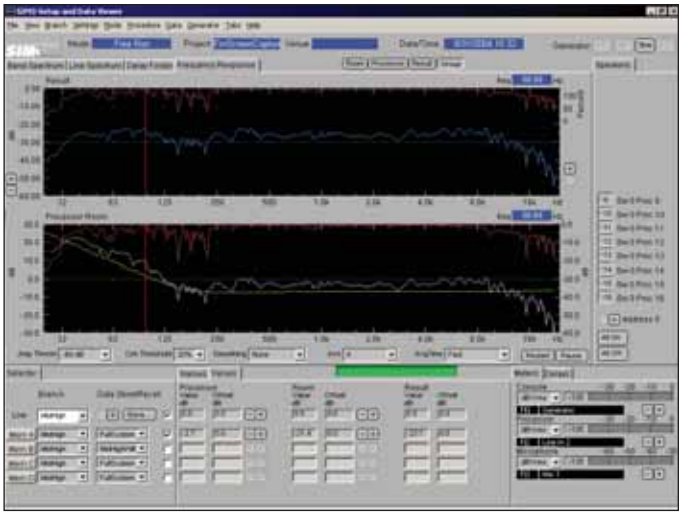
Walking the Hall

A guide to wirelessly tuning a loudspeaker system

By: Alan Hardiman





Two screengrabs from the Meyer SIM system, a popular device used in walking the hall.

Over the past quarter century, the technology underlying performance sound has grown increasingly complex, for both touring sound companies and owners of installed systems. In larger venues, it has become commonplace to distribute loudspeaker systems around the facility in a number of zones. In most cases, these zones are treated separately, in order to achieve specific results throughout the entire venue. As in most other areas of our lives, the application of computer-based solutions has brought new levels of sophistication to the science and art of integrating a performance sound system into a facility.

Nowhere is this more evident than in the increasingly visible practice of sound designers and technicians walking around a hall with a wireless tablet or laptop computer in hand, taking measurements and applying correction to the processors driving the amplifiers that, in turn, power loudspeakers. Referred to by some in the industry as "walking the hall," this has brought tangible improvements to performance sound for audiences everywhere, along with concomitant benefits for sound professionals and facility managers.

In this article, we'll take a look at the evolution of this practice and how the development of wireless technology has resulted in improvements in the measurement and control of performance sound systems. We'll include guidelines and recommendations, along with some tips for avoiding pitfalls and potentially career-limiting missteps.

The goal

The goal of sound system designers and technicians is to achieve high intelligibility with an even tonal balance throughout a facility, according to Martin Van Dijk, senior designer with the consulting firm Engineering Harmonics, Inc. of Toronto. It currently has some 85 projects at various stages of completion around the world, representing a total capital value of around $100 million in multimedia technology.

"In installations, you're trying to address a number of different functions. In many facilities, the objective is to integrate the loudspeaker systems into the listening experience, especially in concert halls and theatres where it's part of the acoustical environment," Van Dijk says. "This is in contrast to touring systems for shows such as rock concerts, where you put a big sound system into a hall. In order to do that, you need several locations for loudspeakers, and you need to be able to control them. With the new technology, we are fortunate in having new methods for addressing the zoning and tuning of loudspeaker systems. The objective is to present to the audience intelligible sound, even tonal balance, and the appropriate sound pressure levels, depending on the requirements.

"Most venues turn into multi-use facilities," Van Dijk continues. "An opera hall, for example, may be home to opera for only a portion of the year. The rest of the time, there are other events—sometimes two in a single day. There may be an opera rehearsal in the afternoon, and, in the evening, they'll have a different sort of performance or event, perhaps spoken word. Because the resident sound professional is not going to take the console out of the booth and put it out at the front of house, he's got to be able to walk the venue quickly and make adjustments to the sound system to ensure that it sounds right for the event. Eventually, he will establish presets for the system as he gets a

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sense of what the programming is. It may take several years to refine and store all the presets so that he can call them up on demand."

The goal for touring sound companies is similar, according to Jeff Berryman, senior scientist with Electro-Voice and director and sound designer with the international touring sound production and engineering company Jasonaudio. Contrary to popular opinion, he says, it isn't always about sound pressure levels.

"In some cases, even sound pressure level is important; in other cases, it isn't," notes Berryman. "In popular music concerts, for example, the value of even SPL for high-level rock concerts is overestimated. In fact, people buy seats based on how loud they think they're going to be. If you were to take a loud rock band and try to make it equally loud everywhere in the room, some people would be disappointed. It wouldn't be loud enough for people who bought the close-up seats, and, for people who bought faraway seats, it would be too loud. But they all want even tonal balance. On the other hand, if you go to a Barbra Streisand concert, almost everybody wants the same thing in all the seats. And in installed situations, typically you want the same SPL in every seat."

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Exploring both sides of the subject with Van Dijk and Berryman, I discover that, while they have never gone over this ground together before, they find themselves to be in complete agreement on almost every aspect of the theory and practice of walking the hall—a remarkable achievement in itself for two highly respected veterans of the industry.

How we got here

In the past, a sound tech—or, at best, two sound techs on walkie-talkies—would wander around a venue and listen to the sound in different zones, then relay instructions for adjusting the system based on what they heard. Early experiments with MIDI and the development of software scripts enabled control of devices such as BSS VariCurve equalizers via MIDI controllers, using standard MIDI cables. This provided a degree of freedom of movement and allowed for in-situ adjustments of the sound system, although it meant dragging a long MIDI cable around the hall. The dawn of wireless control finally broke when BSS developed a method of interfacing MIDI with a wireless microphone belt pack, which afforded almost complete mobility throughout a venue.

"We were using BSS's proprietary MIDI controller hooked up to their proprietary analog equalizers," recalls Berryman. "It was all done with proprietary system-exclusive (SYSEX) messages, and the ergonomics of the wireless handheld controller were very good. I don't think anything has ever surpassed that in terms of ergonomics. It was a dedicated piece of equipment that was very expensive for them to make. Eventually, they had to stop making it because the cost was just too high. They managed to extend it to control their digital processors as time went by, but, after a while, it just became too uneconomical to manufacture. By the end of its product lifetime, a replacement screen for it cost more than a complete laptop computer."

Toward the end of the 1990s, the increasing affordability of powerful PCs, along with the growing availability of programming tools for writing applications, led to the migration toward PC-based sound-system control. An important intermediate step was the introduction of a proprietary wireless technology that served as a bridge from wireless MIDI control to current WiFi technology. "PRAM Technologies developed a custom wireless interface that turned a PC serial port into a wireless serial port," says Berryman. "It allowed wireless control of a number of serially controlled devices which themselves were not wireless. You plugged in your RS-232, your laptop and PC, and you programmed it. The PRAM interface replaced the cable. It was a very niche sort of application that was relatively successful. Since then, PRAM has gone on to provide rack-mount WiFi hardware.

"The early device control systems allowed you to walk around the room with your controller in your hand, as opposed to having it fixed at the front of house," Berryman adds. "What they didn't give you, however, were measurement tools. You could adjust crossovers, equalizers, and delay, but you couldn't see in

quantitative terms what the sound in the hall was doing. That came later. Also, on the early systems the screen update rate was very slow—about once per second—so you couldn't watch VU meters bounce in real time. Those early connections could not support data rates that would enable real-time measurement systems to function very well," he said.

Enter Ethernet

Around 2000, the convergence of audio and computing brought new benefits to the industry, as several manufacturers released new products incorporating Ethernet technology. However, the specifications of Ethernet cable systems—chief among them length and handling limitations—have limited the speed of its ingress into professional and portable applications.

"Standard Cat5 Ethernet cable is not particularly friendly compared to normal audio cable cable. Its maximum pull rating of 25lbs. has been a limiting factor," says Van Dijk. "If you pull harder than 25lbs. on a typical Cat5 cable, you've exceeded its rating and it is no longer guaranteed to work. Also, the standard Cat5 connector is far too fragile for portable audio applications. Four years ago, Neutrik developed a Cat5 connector—the EtherCon—that handles like an XLR, but it's only within the last 12 months that Belden has developed a portable Ethernet cable that handles like a microphone cable and is strong enough for field use.

"A second factor that has limited the ingress of Ethernet technology into audio is that it's really hard for the average audio guy to diagnose what is wrong when it's not working," says Berryman. "When everything is plugged into hubs and switches and something over here can affect something over there, it is much harder to troubleshoot. Also, there are other things you need to understand, such as IP addresses. Many audio technicians still don't know about that sort of thing."

Ethernet-based audio control systems normally use a combination of wired and wireless (WiFi) Ethernet. As we have seen, wireless operation is liberating. However, WiFi introduces a new set of range and interference problems that can make system tuning anything but easy. With care and attention to detail, WiFi can be made to work well.

Recommendations for WiFi

Berryman notes that not all wireless systems are alike. "There are short-range wireless and long-range wireless. Short-range wireless will not serve you well in a large venue. There are consumer-grade routers and there are industrial-strength routers.

"Antennas are often neglected by inexperienced WiFi users," he continues. "Antennas for WiFi should be selected in the same way as for wireless microphone and monitoring systems. Audio pros should pay attention to gain, directivity, and location, and a low-loss type of antenna cable should be used. In general, diversity systems using two antennas will be required for reliable performance in large venues. And be sure to select and position

the antenna to make appropriate use of its pattern. For many venues, we've found it useful to use antennas designed for ceiling mounting, but orient them upside-down, so they cover the air space above and around the front-of-house position.

"Also, what is in the laptop or tablet matters a huge amount. Often, the built-in wireless equipment is not sufficiently robust and has a limited range. One easy way to get longer range is to use a plug-in PCMCIA card—some really good ones are available. I recommend choosing a PCMCIA card with one or two antenna jacks, so you can stick an external antenna onto your portable kit.

"Regarding speed, if you're really getting 10Mb per second, that's fine. Watch out, though—wireless systems auto-scale their bandwidths down when they start to get errors, often without telling you," Berryman warns. "You might be walking around the venue and notice your screen update rate slowing down. You don't know why until you go and look in your wireless control panel. Your connection might have scaled down to 2Mb per second or even slower."

Van Dijk notes that when communication between a tablet and host PC is lost and then re-established, certain applications do not reconnect. "If you are walking around a hall and need to go up to the balcony, you may lose the wireless connection when you walk out into the lobby and up the stairs. With some audio control systems, you can't reconnect without rebooting, and that is a real annoyance."

One way to minimize this problem is to configure the system so that all applications run on a dedicated host PC situated at the front-of-house position, and then set up the wireless tablet or laptop to control the host via remote terminal software—such as Microsoft Remote Desktop. Remote terminal software allows you to operate the host computer from the tablet or laptop, just as if you were sitting at front of house. With remote terminal software, if the WiFi connection is interrupted, the host PC continues unharmed, and maintains all of its relationships with the controlled devices. When the WiFi connection returns, the terminal software will almost always reconnect seamlessly to the host.

A second, major benefit of the remote terminal approach is that it facilitates the use of live measurement software, as we will see next.

Measurement

There are two fundamental methods of measuring the performance of a sound system. One is to input a standard test signal, such as pink noise, into the sound system, then measure the output with an RTA or similar tool. The second method is to input an arbitrary signal, such as music or other program material, into the sound system, and compare that with the system output. As far as walking the hall is concerned, there is a big difference between the two; in fact, the first method cannot be used at all during a show. Over the years, the comparison

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method has come to be accepted and trusted as yielding more useful results.

"If you are using a software comparison tool—and there are half a dozen of them on the market, including SMAART and SIM—there must be two signals coming into it, one of which is the reference, usually straight off the mixing console," says Berryman. "If your software is running on a handheld tablet 300' from the console, how does the reference signal reach into it? In fact, you don't need to get it there. If you use the remote terminal approach described above, the measurement software will be running on the host PC at the front-of-house position, so the reference signal will only have to travel a few feet."

"This is one of the advantages of using remote terminal software," he adds. "The second advantage is reliability, as we have already discussed. The third advantage is that if you use the front-of-house host computer as the source of test signals, such as pink noise, as well as your test listening tracks (your favorite tuning cuts), you can turn those on and off remotely. The fourth advantage is that, with both the measurement software and the system control software running on the same computer at the front of house, you can look at it in multiple windows on your terminal, flipping back and forth between measurement and control as you please. It's really quite a powerful technique.

"In cases where you're doing serious measurement, having a wireless laptop or tablet will only make sense if you can carry the measurement microphone around the room with you. You can do that by getting a wireless belt pack and measurement microphone that works with it and send the measured signal back to the measurement system at the front of house. Be careful about phantom power requirements and connector pinouts. For accurate low-frequency readings, you will need a special belt pack and receiver that allow you to switch companding off. You can gaffer-tape the microphone to the side of the tablet. In use, the microphone signal travels wirelessly to the front of house where it is input to the measurement system. The measurement system does its computation, and the picture comes back to you, enabling you to compare what's coming off the board down there with what your microphone is receiving up here in the balcony."

Van Dijk notes, "We use a Lectrosonics digital wireless system with a Beyer MM1 measurement microphone. Telex also makes a product that allows you to turn the companding on and off.

Ground rules for sound system control

"Don't fool around with preset factory crossover parameters," warns Van Dijk. "This includes delay, equalization, crossover filtering, *and* gain. You should assume that your loudspeaker system comes from the manufacturer with its internal delays already set correctly within each box. Some people make the mistake of trying to tweak the crossover parameters during the sound check. You can do that only under very well-defined conditions that you know, such as in a laboratory. Of course, you have to do it if you are hanging separate boxes for line arrays, mid-bass and low bass that were not designed together in a laboratory. But you should not attempt to adjust crossover parameters on the component level within each box."

Neither should adjustments be made to the level of different components within a box. "How many times have you heard people say 'Turn down the horns'?" Berryman asks. "If you want less treble, then turn down the treble with an equalizer. Don't turn down the horns, because when you do that, it changes the phase relationship at the crossover. Using crossover gain for tone control is like having an equalizer that ruins the directional pattern of your loudspeaker system. Who would want to use that if they knew that's what it did? And yet that's what 'Turn down the horns' does. The way tone color should be adjusted in the performance sound system is with very gentle curves. I find there's way too much narrow-band equalization being introduced into systems these days. And there's too much fooling around with crossover frequencies. If the physical array is arranged correctly—if the basic array is stepped right—then your ear will feel less of a need for narrow-band equalization," he said.

A good general rule to follow in tuning the system is to set architectural delays and SPL correctly before adjusting EQ. "Walking the hall enables you to go into all the different areas in the room and set SPL to even out the level throughout the venue," Van Dijk says. "The artistic part of using multiple loudspeaker zones lies in ensuring that you control the imaging to localize the apparent sound source to where it is supposed to be coming from. You have the Haas window to play around with in setting your delays, which is about 10 meters. If you get your loudspeakers sitting in that pocket, it will work well for you.

"Oddly enough, the quieter the production is, the more critical the alignment becomes, because localization becomes really apparent," Van Dijk adds. "It's almost counter-intuitive, because most people in a concert hall would say, 'Why should I spend so

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much on an expensive loudspeaker when the system is not going to be very loud?' Well, the quieter it is, the more you are going to notice it, so its response has to be more even, its distortion lower, and its articulate greater.

"For each zone, determining the point at which you measure the delay depends partly on the SPL—specifically, how many rows of seats that loudspeaker system will cover. You should usually measure from the center of each system's coverage area."

After adjusting architectural delays and SPL, attention can be turned to equalization. "It's not about the pursuit of 'flat,'" says Van Dijk. "It is about achieving an amplitude response in your system that is well-behaved throughout the entire frequency range and throughout the room. It means there will be gentle slopes in the response. If you can achieve that, you will have a good-sounding system. But if the system has been equalized wildly, then even though it might measure flat in the front of house, you'll find that, if you walk just a few feet off axis, it will sound completely different. The reality is, you are always working in an environment, so you have to work with the environment and you have to understand the environment."

"When you're equalizing the system, you have to make some assumptions: One is that the loudspeaker that you are using is good and has been set up correctly by the manufacturer," says Van Djk. Berryman adds, "A second assumption is that you have loaded into your crossover the correct presets and that they are all the same from one array to another—that is, that you have all loudspeakers running with the same set of settings. The third assumption is that you have checked that all your cables and connectors are wired in phase and all your amplifiers are working. If you've got that right, when you turn the system on you're going to find you don't need as much EQ you as you might think."

Van Dijk notes that the lion's share of problems can be traced to phase-reversed cables and connectors. "It's the simplest thing in the world, but getting polarity right is the simplest thing to screw up," he said.

Plan your work

As you look ahead to the process of tuning the room, it is helpful to plan your work and then work your plan. "Develop a discipline," Van Dijk advises. "Personally, I make up an agenda ahead of time of what I'm going to do, then walk through it. At Engineering Harmonics, we're looking at plans all the time, so that really makes sense for me. When I arrive at a hall, I've already figured out how I'm going to do it, and not just for myself. I'm also going to send the agenda to the contractor, who can then follow in my footsteps exactly, with the documentation I have given him and the data I have collected. I might only have an hour in a day that's available to do this in a hall. I've become more effective in my years at Engineering Harmonic,s because I've had to learn how to maximize my efficiency. When I do a show, I follow that same procedure and it works for me."

Hand-in-hand with planning goes the issue of file management. Once you're involved in walking the hall, you'll need to store measurements, curves, and other data with sufficient annotation that you can recall the location where each particular measurement was taken. If the measurement software does not offer an integrated filing system—and few do—you should prepare your own in advance.

"One of the coolest things about PC-driven measurement systems is that not only can you save the curves and notes from last night or even a year ago, but you can also save the parameter settings on the drive controllers," says Berryman. "You have to have a basic facility with naming conventions for files and folders and a good archiving method, because otherwise you're going to save a pile of useless information and you are never going to know what it is again."

"One rule for filing conventions: Don't be afraid to use lots of folders and put them inside other folders," he adds. "Give them all meaningful names and don't abbreviate much. Make sure you don't end up with one folder full of 500 individual files, all with arbitrary names. Be consistent in how you name files. Don't be afraid to put dates in file names with the full year, month, and day, all in numerals, as an eight-digit number in the format YYYYMMDD, so that when you sort, they sort in order."

When walking the hall, it is helpful to be able to communicate with someone at the front-of-house position, so carry a walkie-talkie. For example, you might want to selectively mute parts of the sound system but don't have sufficient wireless control to do that yourself. And as you go around to adjust delay times, following whatever agenda you have decided on, it's time to turn on the test signal.

A recommended test signal

"With a touring system, if you're using SMAART, SIM, or one of the comparative measurement tools before the show and the band is not onstage playing, you'll need to find some kind of test

signal. Pink noise is a good choice because it contains all frequencies," Berryman says. "However, other crew members working in the venue usually hate the sound of pink noise, and, to make it worse, many audio guys run the pink noise far louder than it needs to be. Some of the more considerate and sophisticated sound techs play music with pink noise mixed in at a lower level. This is a fine test signal to use, because music by itself does not contain all frequencies. You need something that's broadband, and that's what pink noise is. But pink noise by itself is unpleasant.

"To most people, this mix sounds like music playing from an FM station that's just a little bit too far away," adds Berryman. "And that is more considerate and generally creates fewer political problems, especially if you're playing it at a reasonable level. The only exception is that you have to play it fairly loud when adjusting low frequencies. But it's not a good idea to try to tweak the low frequencies before the show—anything above about 300Hz, absolutely, but you don't need much level for that."

Equalizing the system

Then you have to decide what you're going to equalize—and what you're not. "Woe be to the guy who sees a 15dB notch one-third of an octave wide and tries to get rid of it," warns Berryman. "You just cannot do that. You'll kill yourself. The thing about walking around with a wireless tablet is that you have a lot more opportunity to see bad stuff. If you have one microphone in only one position, then you only have to decide what to do about three or four bad things. When you go all around the venue, you find that every different place has a number of bad things, mostly due to cancellations. If you try to get rid of them all, you'll end up chasing your tail. Some things—cancellations, bounces, and room modes—can't be equalized, and you shouldn't even try. So deciding what not to do is really important."

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"There's really only one way to fix those things, and that's by moving the loudspeakers, but sometimes that just moves the bad spots, too," Van Dijk observes. "The devices that are contributing the most energy into the room need to be looked at first, and then you build out from that. In a left-center-right flown system, you might want to work with the center cluster first, if it's putting more into the room than either of the sides."

Because the passband of effective directivity is extended into lower frequencies with longer line arrays, Berryman strongly cautions against turning off any part of a line array when tuning the system: "You will just be reducing the directivity of the array at lower frequencies if you shorten the array by turning off some of the boxes. For instance, if you have a line array comprised of 15 boxes and you want to check out the top of the system from the balcony, beware of turning off all but the top five boxes. Lots of guys do that, but it only works above about 2,000Hz. Otherwise, you have just disabled your line array. This is a radically bad idea. When truncating an array you really have to know what frequencies to ignore. The more you eliminate, the higher the cut-off frequency. If you're not

experienced in acoustics, I recommend not turning off any part of a line array. You can turn off the subwoofer system of course, but not parts of the line array."

Training and experience

No software tool currently available can teach someone how to relate to a loudspeaker system in a venue. There is as yet no "SimuVenue" program to substitute for the physical, perceptual experience of tuning a performance sound system. Providing large venue training is expensive, simply because it requires large venues. Even though a person might have the tools, it still requires hours and hours of practice in a large venue to master what is fundamentally a learned trade.

"There are two mistakes that I see people make in tuning systems," says Berryman. "One is depending too much on the graph on the screen and not trusting your ears. The other is seeing something on the graph that shows up in measurement after measurement and refusing to believe it. So there's a balance that must be achieved between what you see and what you hear, and that comes only out of practice. And neither one of them is the final word."

"Someone with knowledge and experience and a good RTA can achieve great results because he can see behind it and understand what's making the data bounce up and down on the screen," says Van Dijk. "Using your ears and a really good RTA, you can get very close to what someone can achieve with extremely sophisticated comparison software tools like SMAART or SIM. You can get amazing results. I personally have seen this in A/B comparisons, with two guys equalizing the same system in the same venue using both approaches. And they come remarkably close to the same result.

"I find that customers really like the practice of walking the hall," he adds. "If something sounds odd to a facility manager or event director, the sound man can stand next to him with a tablet and adjust the system from right there. This provides two advantages. One, the sound man can hear what the boss is hearing. Two, the boss can see you doing your job, and you can interact with him as you're making adjustments. It's much more comfortable than trying to talk on an intercom. The personal contact facilitated by an onsite tablet is pretty cool."

Practical tips

A few practical recommendations are in order:

1. Use a dedicated front-of-house computer, run all the control programs in it, and make its RAM large in order to keep the system fast and usable. Generally, you're always running at least two applications: the measurement tool and the device you want to control. Also, make sure it has a good video system with a separate display adapter card that has its own video RAM. Otherwise, you will be hopping back and forth between screens really slowly. For Windows 2000 and Windows XP, 1GB would be a good size. For Windows Vista, 2GB.

2. You need great wireless communication. Wireless is the weakest link. Getting a faster tablet is not going to do much. Spend money instead on getting a great wireless system with a tablet that is not going to break when you drop it.

3. Resist the temptation to use your tablet or the front-of-house PC for other things. Leave it dedicated to this work. Don't play games on it and don't enable Windows Update on it.

4. Learn about Ethernet and IP addresses, and how to configure IP settings on all your computers. Understand the difference between a hub and a router, the cable length limitations for wired Ethernet, and the surprisingly restrictive rules for hooking up cascading hubs. Learn how to fix RJ-45 connectors. "You should be traveling with a little RJ-45 cable crimper. Get the ratcheting kind and learn how to use it. The good ones cost about $100. Also, buy a computer cable tester," Berryman suggests.

Recommended reading includes ESTA's 44-page document, "Recommended Practice for Ethernet Cabling Systems in Entertainment Lighting Applications" and its 22-page supplement covering 100Mb Ethernet titled "Supplement to the Recommended Practice for Ethernet Cabling Systems in Entertainment Lighting Applications" available online for $30 each at http://www.estafoundation.org/pubs/browse.php?. They are briefly summarized at http://www.esta.org/tsp/documents/published_docs.php (go to the bottom of the list).

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