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RADIOWORLD

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NewBay Media

SureStream: An Improved IP Audio Method

Multiple Paths Enhance Performance Over Unmanaged IP Networks

BY HARTMUT FOERSTER AND TONY PETERLE

Hartmut Foerster is product/sales manager for WorldNet Oslo. Tony Peterle is manager of Worldcast Systems Inc. in Miami.

WHITEPAPER

The use of Internet Protocol and various Ethernet standards to transmit audio from point to point is an important capability in broadcast, and the use of IP networks for audio transmission is increasing. Within any given broadcast facility, and also to transmit audio between distant facilities, IP is fast replacing traditional RF and synchronous type links (E1, T1, ISDN) for both studio-to-transmitter links (STLs) and studio-to-studio communications.

If the IP network being used is "in house" — either within a single facility or on some kind of private network between facilities (VPN, MPLS, etc.) — the network can be managed. Established routes can be defined between the various nodes on the network, and Quality of Service (QoS) controls can be enforced that will regulate the traffic on any network segment, with (presumably) the highest priority given to the audio packets, so that they may arrive at the destination on time and in sequence.

However, these types of "managed" networks can be expensive to deploy and maintain, particularly when sending audio over some distance, such as between cities or to different parts of a region or country. For these types of links, the public Internet offers an

(continued on page 14)

Thermal Imaging Saves Time and Money on Transmission Line Failures

Non-Intrusive Camera Allows Inspection While Station Operates

FROM THE FIELD

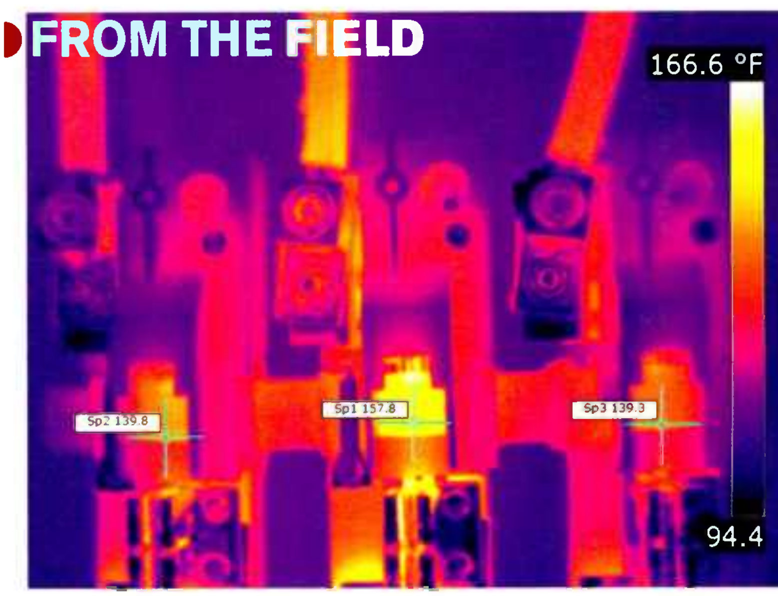

Fig. 1: Overheating terminal in the electrical service panel. Note the temperature 20 degrees higher compared to other terminals.

BY CHRIS MURRAY

Throughout day-to-day operations in a broadcast facility lies a part of the broadcast chain often overlooked. Antenna and feed-line systems are the last link in the broadcast distribution network, but they are often the most difficult to troubleshoot and maintain.

Broadcast engineers do the best they can with the limited tools available. There are ways of tracking and maintaining various computer networks and

(continued on page 20)

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The Great Translator War

The FCC Plays Solomon to Resolve Fight with LPFM

BY MICHAEL LECLAIR

It appears that the FCC is in the final stages of resolution of the long-running battle of LPFM vs. translators. According to the Fletcher, Heald and Hildreth August 2012 blog, the Office of Management and Budget has now approved the proposals as outlined in the Fourth Report and Order published in March 2012. The guidelines of that Order allow the FCC to award as many of the backlogged translators licenses as possible, within limits that leave spectrum available for Low-Power FM licenses in the top 150 markets.

The commission inadvertently began the conflict in 2003 when they opened up a window for translator licenses without setting any public policy limits on applications. No technical requirements were in place to even determine if an application could legitimately be built. It was the heyday of anti-regulatory fervor applied to an industry that had been intensely regulated since its inception, and the crowd went wild. With plenty of advance warning, applicants prepared a flood of applications for every conceivable geographic corner and population center. Why not, if the government was giving away licenses for the FM band, then at the peak of equity-fueled valuation?

Any entity, whether or not a broadcaster, was allowed to file for translators. This invited outside speculators to put in barely literate applications with the hope of selling any properties thus obtained back to broadcasters at a profit. It also fostered a vision of national radio networks via a phalanx of translators, satellite fed.

When more than 13,000 translator applications (some entities filed several thousand each) landed in the FCC's inbox, it was apparent that the whole process had been a mistake. Broadcasters who had applied for translators for traditional signal enhancement purposes found themselves swarmed with mutually exclusive applications, including multiple applications from one entity that were themselves mutually exclusive. But most importantly, everyone knew that some kind of line had been crossed in terms of spectrum. If all the applications had been allowed to proceed, this window marked the end of any available spectrum for future Low-Power FM or translator services where any population existed to be served. That was it — the whole enchilada would have been given away.

It was the heyday of anti-regulatory fervor applied to an industry that had been intensely regulated since its inception, and the crowd went wild.

THE BACKLASH

Acting on complaints, the FCC froze 6,500 or so of the translator applications and settled down to the task of trying to come up with a reasonable fix. In the end, it took some help from Congress and nine years to get where we are today. Both sides will get some of what they want.

Entities that filed in the 2003 window will be allowed to keep up to 50 of their applications, but the rest must be withdrawn. The freeze will be removed allowing applications to proceed immediately. However, there's still a catch.

The FCC spent years looking at the various broadcast markets and evaluating potential channels for LPFM stations. Each market was broken into fine grids for analysis. Locating new LPFM opportunities was assisted by the congressionally mandated relaxation of rules on spacing of third-adjacent stations. LPFM will be given at least this level of preference or protection: If a market only has room for LPFM or a translator, all translator applications in that market will be dismissed.

If the measure of the wisdom of a compromise is the degree to which it is disliked by both parties, then the FCC has been wise in its deliberations. Certainly hundreds, if not thousands, of new translators will be authorized in the near future as the freeze is lifted. Many AM stations may be in the market for one of these new licenses to supplement their coverage at night. We can expect that soon after this backlog has been processed that a new window for LPFMs will open, which has been a Congressional goal for many years. But even with the relaxed interference rules, only a relatively small number of channels remain for LPFM.

Those with visions of national networks are also unhappy; but at least to this observer, the idea of using a class of stations designed to augment local services and fill in coverage gaps was a poorly conceived goal at best. The Telecommunications Act of 1996 essentially deregulated ownership limits on radio licenses. Those looking to build national networks can do so without having to resort to secondary channels a few watts at a time. Certainly, giving away such a valuable quantity of spectrum to one or two players was never the goal of the FCC when it opened the flood gates in 2003.

IS THIS THE END OF AN ERA?

Significantly, in the opening section of the Fourth Report and Order, the FCC notes that the clearing of the 2003 translator window followed by a subsequent LPFM window are likely to be virtually the last of their kind. "We have determined, based on these studies, that the next LPFM window presents a critical, and indeed possibly a last, opportunity to nurture and promote a community radio service that can respond to unmet listener needs and underserved communities in many urban areas" [from paragraph 2 of the introduction]. As is generally true with most occasions when the FCC gets stuck on an issue, the outcome represents a watershed moment. We appear to be reaching the limit for FM with regards to creating new stations in areas with actual population.

All of which begs the question: Will the FCC ever agree to add more spectrum to the FM band in the face of what continues to be an overwhelming demand? The TV 6 band would make a nice addition to a service that continues to grow and is ready to take on even more.

Michael LeClair is chief engineer for radio stations WBUR(AM/FM) in Boston; he has been technical editor of Radio World Engineering Extra since its inception in 2005.

THIS ISSUE

AUGUST 22, 2012

Use Antenna Modeling to Evaluate Potential Reradiators

You Only Get One Bite at the Apple. Careful Analysis Will Better Protect Your Station

BY CRIS ALEXANDER

Any broadcast engineer who operates an AM station in a built-up area periodically gets the notifications: A wireless or other tower close to your AM antenna or array is being constructed or modified.

Several of these cross my desk every month. I have long been a proponent of not ignoring these. You really have only one chance to head off any potential pattern distortion issues that might result from the construction or modification of a nearby structure.

While it is a rare thing for any reradiator to fill the nulls in a directional pattern so that actual interference is caused, such pattern distortion does occur, and unless he catches it during the notification/construction phase, the AM licensee could well wind up on the hook for the costs of fixing it.

I have been personally involved in a situation on Mt. Scott in Portland, Ore., where several reradiators were allowed to distort the pattern over years. Now one of the tower owners is refusing to play ball, leaving me with the options of paying for the detuning of the reradiator myself or living with a "permanent STA."

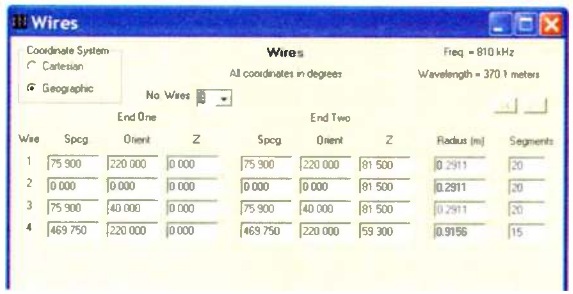

Fig. 1: Array geometry moment method model for three tower in-line array.

The point is that it's important to evaluate thoroughly the potential effect of reradiators or modifications thereto before they become a de facto part of your directional array. You only get one bite at the apple.

Through the years, we have used a number of tools to evaluate reradiators. Most were simply formulae that we used to calculate the worst-case reradiation from a structure based on that structure's aperture and the incident field. And while these methods did provide some idea of what kind of reradiation we could expect, we didn't really have a way of evaluating the impact of a particular reradiated field on the various radials (or the pattern as a whole) of an AM antenna or array.

Moment method modeling offers a tool that is coming into use for this purpose. It's what I use if there is any doubt whatsoever.

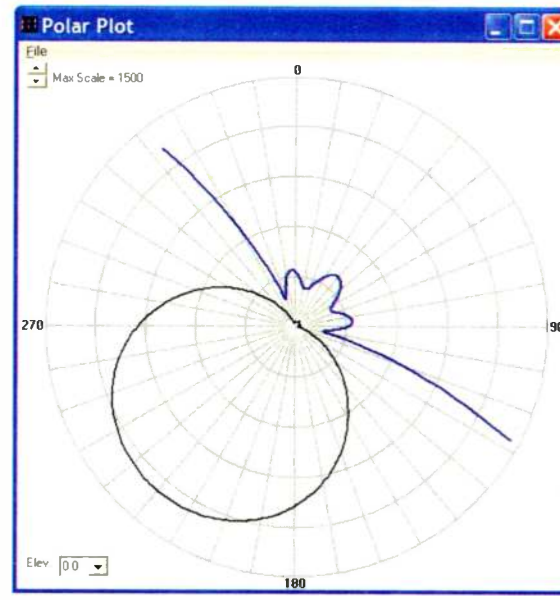

Fig. 2: Polar plot of modeled pattern.

A THEORETICAL EXAMPLE

To see how this works, let's take a typical AM directional array and pattern, in this case a three-tower in-line array on 810 kHz.

Fig. 1 shows the geometry of this array in the

(continued on page 6)

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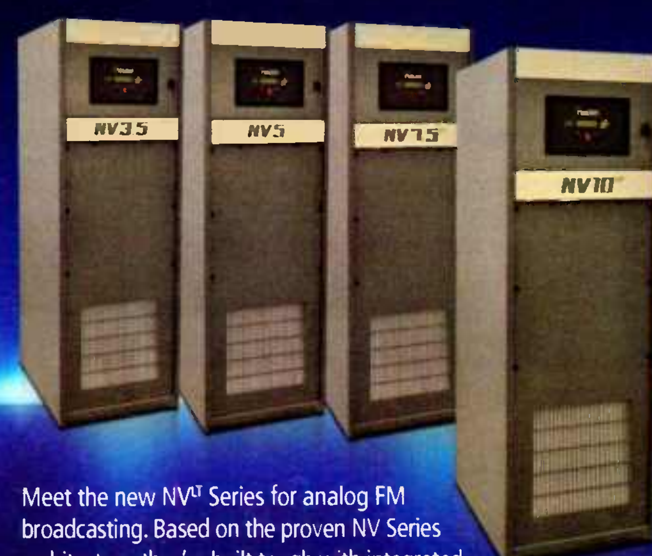

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MODELING

(continued from page 4)

moment method model, and Fig. 2 shows the model output polar plot of the pattern. Note that there are four deep nulls, at 15, 65, 99.5 and 340.5 degrees. The standard pattern 15- and 65-degree pair null depth is 23.6 mV/m, and the 99.5- and 340.5-degree pair null depth is 20.9 mV/m.

We can model the E-field of the array along a radial path and plot those values to see what the inverse distance field (IDF) of the pattern looks like on that radial. Fig. 3 shows such a graph for one of the null radials of this pattern. Note that the predicted E-field points all plot following an inverse-distance function well below the standard pattern IDF line. Remember that the standard pattern IDF is the maximum radiation permitted in a particular azimuth by the FCC for that array and pattern.

This gives us a base line for what the directional pattern and the 15-degree radial should look like for this array without external influencing factors (like reradiators).

Now we can add another tower to the model to see what effect that structure might have on the directional pattern. To make things interesting, let's make this tower a 200-foot monopole with an average radius of 36 inches, and let's place it at the worst possible location, 3/10 of a mile from the center of the array and right in the main lobe (220 degrees). Fig. 4 shows the model geometry for this additional tower.

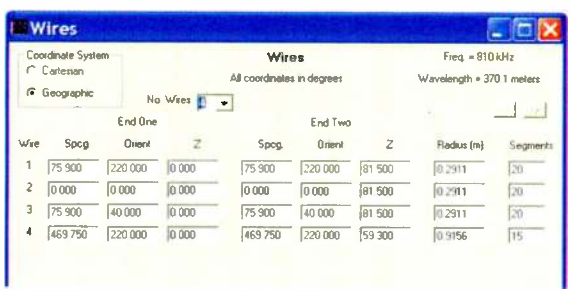

Fig. 4: Array geometry moment method with 200-foot reradiator in main lobe at 220 degrees.

To find the distance to the potential reradiator in electrical degrees, I took the distance (0.3 miles), converted it to meters, divided by the 370.1 meter wavelength and then multiplied by 360.

To get the height in electrical degrees, I followed the same procedure, converting the 200-foot height to meters, dividing by the 370.1 meter wavelength and multiplying by 360.

Note that I used 15 segments for the model on this additional tower, and I chose that value to keep the segment length for that tower roughly the same as the segment length on the other towers in the array. This is a basic principle of moment method modeling that should be followed as much as possible.

Running the model to determine the E-field along that same 15-degree null radial yields the graph shown in Fig. 5. Note that the model-predicted E-field radiation along that radial is above the standard pattern inverse-distance line all the way. The nearby monopole becomes, in effect, another element in the station's directional array, and the reradiation from that monopole produces sufficient field to skew the pattern out of its standard pattern limits. In the real world, this monopole would require detuning.

The above example really is worst case, with the reradiator located very close to the directional array

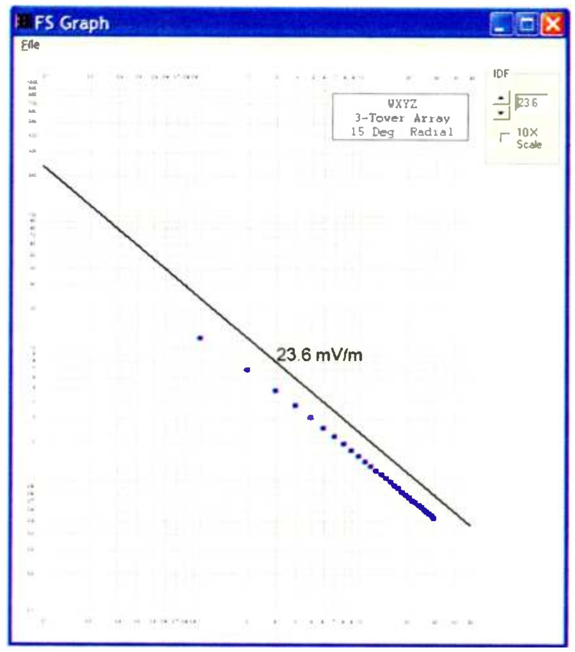

Fig. 3: Field-strength graph of modeled E-field on 15-degree null radial.

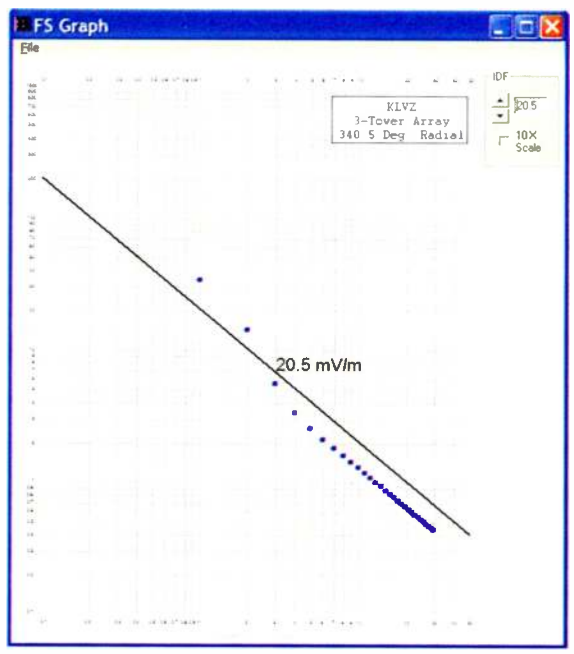

Fig. 5: Field-strength graph of modeled E-field on null radial. Note that the reradiation has pushed the inverse distance field above standard pattern.

and right in the main lobe. Once in a while, that kind of thing happens; but more commonly you get a tower located nearby and off to the side of the array.

One of the things I learned early on in my career is that if the energy is not there, a structure cannot reradiate. While that is true, even a relatively low incident field often will produce some reradiation, and even though that reradiation will most often not adversely affect the far-field (and thus will not produce interference to the co- and adjacent-channel stations that are being protected), it will many times have close-in effects.

MISLEADING MONITOR POINTS

Consider another case where this same monopole is located not in the main lobe but off the side of the array. In this example, I moved the monopole around to 290 degrees at 0.5 km and re-ran the model, this time plotting the E-field along the 340.5-degree null radial. Fig. 6 shows the resulting plot.

Note that the predicted far-field IDF is well below the standard pattern IDF line, so no interference is likely to result. Note also that close in, inside 2 km, the predicted field is well in excess of the standard pattern IDF. This is significant because this is the area where many conventionally proofed stations will have their monitor points located. A set of pre-/post-construction monitor point measurements in this case would likely show a problem when there isn't one.

While I haven't shown it here, the opposite also can happen, usually when the reradiator is in the main lobe or some other high-energy part of the pattern. The plot will show the close-in points way below the line, sometimes as low as 50 percent of theoretical, while the far-field points are well above the standard pattern IDF line. A set of pre-/post-construction monitor point measurements in such a case may well show that everything is great when in fact it isn't.

The point of all this is that it is difficult to determine with any certainty that a potential reradiator is or is not significantly distorting a station's directional pattern by pre-/post-construction monitor point measurements alone. And yet this is the methodology that is most often used!

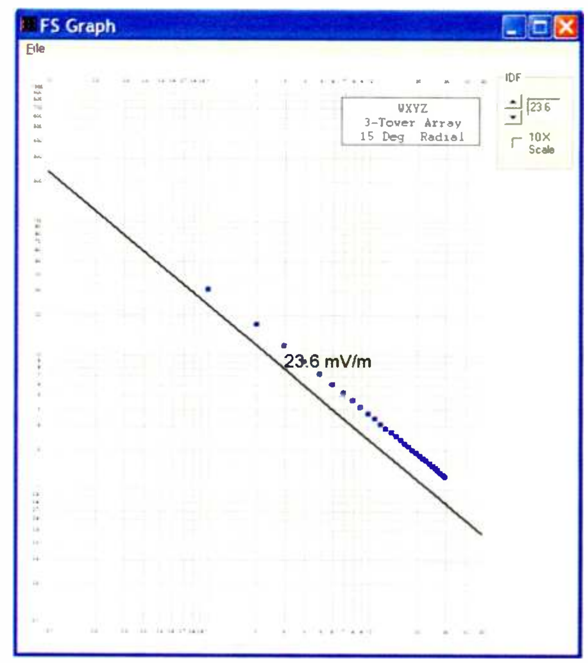

Fig. 6: Field-strength graph of modeled E-field on another null radial. Note the close-in points above the standard pattern IDF line and far-field points below.

A moment method model is a good way to determine what we should expect out in the real world, and if things are close, the question can always be settled with a set of radial measurements (not just one point on the radial).

Wireless operators naturally want to take the least costly path that still complies with the letter of the FCC rules. The burden really is on the AM station licensee to press the issue and insist on a higher level of analysis and/or proof. A properly constructed moment method model is a handy thing to have in your toolbox for this purpose.

Cris Alexander is the director of engineering at Crawford Broadcasting and a past recipient of SBE's Broadcast Engineer of the Year award.

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Sale

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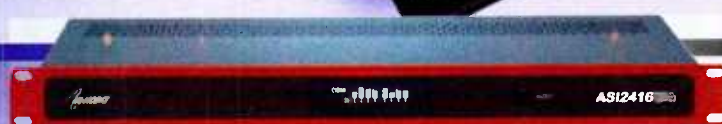

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ClearOS Is a Firewall Winner

Free Software Provides Solutions to ISP Overload

BY TODD DIXON

Contributor Stephen Poole writes:

In this issue, it's time to address something that all of us face. No matter how much Internet bandwidth you buy from your Internet service provider, your employees will find a way to max it out. Another very real problem is so-called "malware," from viruses to Trojans to scam sites that want to trick you into entering personal information.

My assistant Todd Dixon is an absolute whiz at finding free, downloadable solutions to problems like these. I'm going to turn it over to him and let him tell you about the ClearOS firewall.

MAXIMIZE THE INTERNET PIPE

At our Crawford Broadcasting cluster in Birmingham, Ala., we had been looking for a way to maximize the Internet bandwidth coming in and out of our building. We wanted a way to increase our usable bandwidth while not sacrificing service. We had researched getting different Internet providers with more bandwidth only to find that their services weren't available in our area.

We have three Internet audio streams, equipment that increasingly relies on the Internet to function and employees who need Internet content with Flash media, Java and other plug-ins. We had to find a solution.

We knew that part of the answer would be an Internet firewall between our DSL modems and the rest of our network. The term Internet "firewall" may be unfamiliar one. In fact, a better one might be Internet "filter." Don't confuse "Internet firewall" with the little blue box that allows everyone in the office to get on the Internet. A good firewall will be a computer with software designed to allow the parts of the Internet that are good and essential for your business and block the parts that aren't.

At its core, an Internet firewall not only will strain out material unsuited for your work environment, but will also keep data on your network safe from malware. These Internet downloadable programs kill data and employee productivity when the computer needs to be taken offline so the malware can be removed. A firewall also should provide a way for secure, remote access to computers on your network from the Internet. If you've been dealing with these types of problems related to your network, your best friend is about to become a firewall.

We had tried several open-source firewalls that used the Linux operating system, but they were difficult to install, seemed to actually slow our already-limited Internet bandwidth and were difficult to maintain after installation. We felt like we were searching for the impossible: a firewall that was dead simple to install, easy to maintain and would grow with us as we continued to expand our Internet presence.

Then we found a Linux distribution called ClearOS (*www.clearfoundation.com*), a free 700 MB download. Based on Red Hat Enterprise Linux, it was developed to turn any computer into a full-featured, easy-to-use firewall.

Once you've burned the downloaded ISO onto a blank CD-ROM, you simply find a middle-of-the-road machine with two network interface cards (NICs). A machine with 2 GB of RAM and a 3 GHz processor can protect between 50 and 200 employees. If you don't have near that many connections onsite, you can get away with a computer with even less horsepower. A 20 GB hard drive is more than plenty to handle the install and the logs necessary for the firewall to run properly. Ensure that the machine is able to boot from the CD, insert the CD and fire it up; the installer will start running.

In the event that you have never done anything like this before, the install will destroy anything that was previously on the hard drive. It will all be overwritten by the new install of the ClearOS system.

Before you start setting up the firewall, you will need the provisioning information from your ISP: IP address, network mask, password (if needed) and so on. This will go on the first network card; ClearOS calls the Internet side the "DSL" connection (some firewall solutions call it the 'red' side). See Fig. 1.

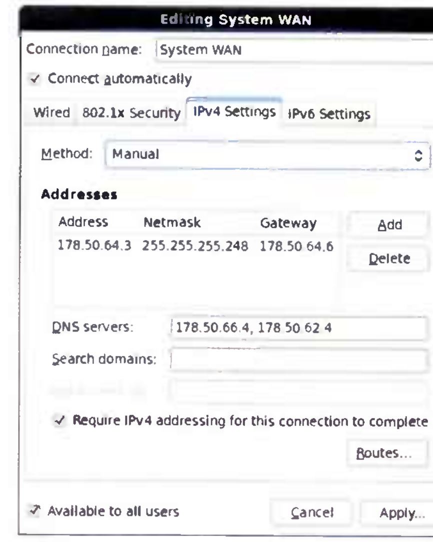


Fig. 1: Setting up the WAN (Internet) side.

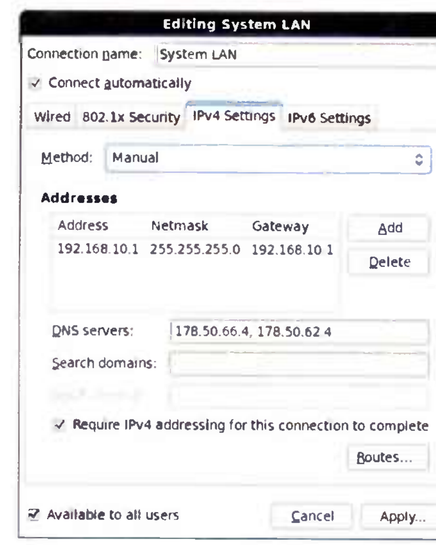


Fig. 2: Setting up the LAN (internal network) side.

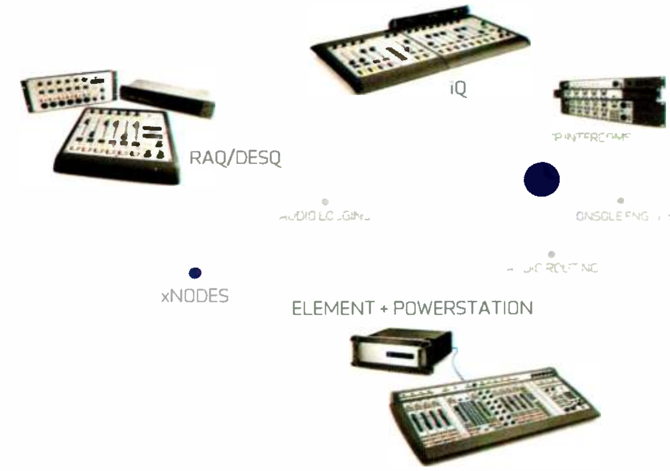
The second network card is called the "LAN" (often called the 'green' side) connection and is for your internal network. On this side, you will set up your networking parameters. See Fig. 2.

We were primarily interested in content filtering, using the built-in Web proxy server and the reporting features built into ClearOS. In a nutshell, the content filter checks every page that is requested against a predefined list of "not safe for work" (NSFW) sites. If a page on the list is requested, it blocks the user from being able to see it. Of course, the content filter can be set to be as granular as you would like it to be. You can "whitelist" (always let through no matter what the content) or "blacklist" (never let through). Surprisingly, we have not had to adjust the default filter much with whitelist or blacklist at all. The filter is courtesy of noted anti-spam service SpamAssassin.

The Web proxy server saves valuable bandwidth, by caching (placing

(continued on page 10)

NETWORKING xNODES



MEET THE NEW AXIA xNODES!

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Everybody knows that Axia introduced broadcasters to IP-Audio in 2003. 3,000 studios and 30,000 connected devices later, the competitors who said "it'll never work" are now eating their words. How do you follow up that sort of success? If you're us, you open up a case of Monster and go back to work. So we did.

The result: Axia xNodes, smart new AoIP interfaces that transform your audio devices into an intelligent network. Use them to turn analog, digital or mic-level signals into routable IP-Audio, with associated GPIO logic. They're so advanced, they won two major awards at their NAB debut.

WHAT CAN YOU DO WITH THEM?
HERE ARE A FEW IDEAS.

» **BUILD A ROUTING SWITCHER.** One stand alone xNode is an 8x8 (4x4 stereo) routing switcher. Connect 8 xNodes to a switch and make a 64x64 routing switcher. Need more I/O? Connect more xNodes. Like all Ethernet-based networks, Axia systems are naturally scalable, up to 10,000 stereo signals (plus logic).

» **STL OVER IP.** Today's cluttered RF spectrum makes IP a great alternative. Put an xNode at either end of a fiber run, OC-3 circuit or a pair of inexpensive Ethernet radios to send eight channels of uncompressed audio to your TX – and get eight channels of audio backhaul too.

» **SAY SO LONG TO SOUND CARDS.** PCI, PCIe, USB3, FireWire... who needs 'em? Load the Axia IP-Audio Driver onto your PC workstation and connect it to an xNode to get eight professional balanced outputs and eight inputs. Use an industry-standard DB-25 breakout cable for pro XLR connections. You'll get studio-quality audio and save some green, too.

» **ADD MICS TO THE MIX.** xNodes make awesome multiple Mic preamps. They have ultra-low-noise, ultra high-headroom, studio grade preamps with selectable Phantom power. Put your Mics in, bring your analog line level out. And that IP-Audio network jack? Ready to be used whenever you upgrade to a full IP-Audio network.

» **MAKE AN A/D/A.** Take one analog and one AES/EBU xNode and rack-mount them side by side. Voila! Eight precision A/D converters and eight precision D/A converters, in just 1RU. Studio-grade, 48 kHz, 24-bit Delta-Sigma A/D and D/A convertors, with 256x oversampling, make difference you can hear.

» **SLIM DOWN YOUR SNAKE.** Connect two analog or AES xNodes with a single Ethernet cable for an instant 8x8, bi-directional snake and bid the multi pair bundle goodbye. Add a few more xNodes on each end for a 16x16, 32x32 or 64x64 snake. Use off-the-shelf media converters for long-haul fiber connections.

RJ45 OR DB-25? xNodes give you I/O both ways so you can choose whichever industry-standard breakout cable you prefer.

XNODES ARE SMALL. Mount them on your wall, under the counter — mount 'em on the ceiling if you like. Optional rack- and wall-mount kits provide plenty of options.

CONFIDENCE METERS on every xNode mean you'll never have to wonder where the audio's at. Audio presence and levels are both displayed at a glance.

INFORMATION OVERLOAD? Not here. Sharp, high-rez OLED displays put all the information you need right on the front panel, without the need for a distracting multi colored lightshow.

xNODES WORK WITH BOTH LIVEWIRE AND RAVENNA AoIP networks — making them compatible with IP-Audio gear from over 40 major broadcast companies.

Livewire RAVENNA

NO NOISY FANS HERE. Front-mounted heat sink keeps xNodes calm, cool and collected using air-conditioned studio air (instead of that hot air in the back of the rack).

FAST, ONE-BUTTON SETUP. Hit the switch and plug 'em in — your xNodes will be streaming audio in under 30 seconds.

MONO OR STEREO ROUTING. Choose from 8-in, 8-out mono operation or 4-in, 4-out stereo. Both signals intermix seamlessly on your Axia network.

NOT AT THE OFFICE? No problem, built-in webserver lets you manage an xNode from anywhere. Or, use Axia iProbe software to manage your entire facility – back-up and restore settings, automatically update software and more.

DUAL ETHERNET PORTS for redundant network links. The overnight jock kicks out a connection? No problem, the other one takes over so your programming never skips a beat.

xNODES HAVE AUTORANGING INTERNAL POWER SUPPLIES, but can use PoE (Power over Ethernet) too. Perfect for those out-of-the way places where a power cable is inconvenient. Hook 'em both up for redundant, auto-switching backup power.

VERY VERSATILE. 5 different xNodes handle nearly any signal type: AES/EBU, Analog, Microphone and GPIO. xNodes are perfect when you've got a lot of one audio type to work with. But what if you need a little of everything? This is the Mixed Signal xNode. Think of it as your utility MVP, with a switchable Mic/line input, 2 dedicated analog ins, 3 analog outs, a digital AES/EBU input and output, and 2 GPIO logic ports.

TWO xNODES MOUNT SIDE-BY-SIDE, so you can create your own custom mix of I/O types within a single rack space. Pair up an AES/EBU xNode with a microphone xNode, or match a GPIO xNode with an analog unit. Or combine a couple of Mixed Signal xNodes for the ultimate mix of mic, analog, AES3, Analog and logic I/O.

AxiaAudio.com/xNodes

AXIA

FIREWALL

(continued from page 8)

into memory) a large number of commonly visited sites (*google.com*, for instance) so that the same page doesn't have to be downloaded again and again. Lastly, the reporting features contained in ClearOS allow us to get a clear picture of our network usage. From overall usage to individual users, we can clearly see the worst offenders and pay them a visit. These three features alone have increased the efficiency with which we use our limited bandwidth. See Fig. 3.

OPTIONAL FEATURES

The ClearOS firewall also contains a full-fledged mail server on it, if you have been considering hosting your own mail server but didn't think it was possible. This is a POP3/SMTP server with spam, malware and virus protection included. Webmail is also a part of the mail package so that your co-workers may check email from anywhere that an Internet connection is available.

One of the final features that really endeared us to ClearOS is the MultiWAN functionality. By adding a third NIC and some basic configuration, you can add a second DSL line to increase your bandwidth while the firewall continues to perform its duties on both connections to the Internet. ClearOS load balances both connections in whatever ratio you want them to be used with the added benefit that if one of the DSL lines goes down, ClearOS automatically switches all traffic to the remaining one. This way, you can at least stay online until the problem is resolved.

You are probably well aware that content filtering and a mail server, with anti-spam and virus protection, can come with a hefty price tag. ClearOS is actually free and provides the basic updates at no charge. If you're willing to pay a bit, you can receive more frequent updates, spam and virus definitions. We pay about $120 per year for frequent anti-virus and anti-malware updates. We are pretty well versed in Linux and don't require support backup, but should you need support, ClearOS has packages between $80 and $500.

The great thing about ClearOS is that the software modules (both free and paid) allow you to really tailor a solution that completely fits your needs. If you run the mail server portion of the software, you can opt for the increase in the service level and pay the fee for one of their support packages. Their support packages are modest compared to the price of hosting email through an Internet service on a per-month, per-user basis. Since we chose to only use the firewall for content filtering and Web proxy duties, daily updates are not as crucial to our operations so we opted for the free monthly updates.

If you have been struggling to get the most out of your limited Internet service, an Internet firewall is definitely the best way to do it. With four installs already in place in Birmingham and another market, we have found that ClearOS has done it for us — with both simplicity and strength.

Stephen M. Poole, CBRE-AMD, CBNT, is market chief engineer, and Todd Dixon is assistant engineer at Crawford Broadcasting in Birmingham, Ala.

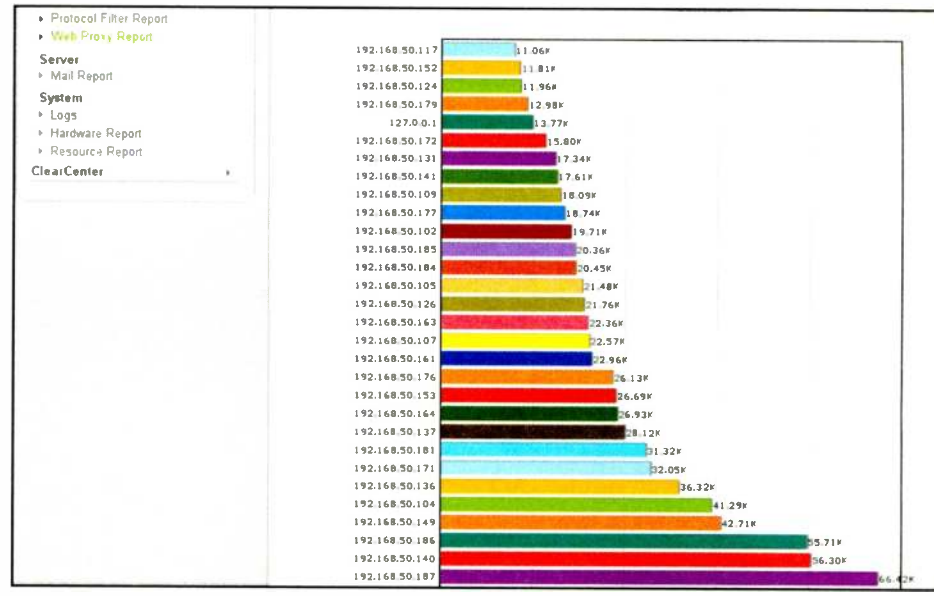

Fig. 3: An example of ClearOS reporting, showing bandwidth consumed by each user.

Drops jaws. Not audio.

Broadband Internet is everywhere. Which makes it ideal for live remotes.

Unfortunately, the internet is also notoriously erratic. Even if you're lucky enough to get a good connection, it might deteriorate during your show. So you dial back the bitrate, sacrificing sound quality to play it safe.

There is a better way.

Meet the amazing new Z/IP ONE codec.

Telos and Fraunhofer collaborated to develop a unique coding algorithm that adapts to changing network conditions on-the-fly.

If your connection quality drops, the Z/IP ONE uses error concealment, elastic buffers, and bitrate adjustments to keep audio flowing at maximum quality.

The Z/IP ONE now makes it possible to use the internet for great sounding remotes.

The convenience of the Internet. The sound of Telos.

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Telos is the trusted World Leader in single-line hybrids and multi-line phone systems.

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With more than 20,000 in daily use around the world, Zephyr is The Best Way to Hear from Everywhere.

STREAMING AUDIO

Hardware and software products for processing, encoding and streaming your audio content.

AUDIO LOGGING

PC Software for archiving and logging all of your stations audio.

this simple setup runs rings around any other AoIP network - at an

Meet the LX-24...Wheatstone's flagship, multi-award-winning advanced modular networkable console control surface

The design initiative behind the LX-24 was to create the world's finest control surface. The result is a console that redefines the entire genre. The LX-24 is an intelligent surface that can store and recall all your settings. Its totally modular design lets you configure it exactly as you like – you can even hot-swap modules at any time without having to reconfigure.

Assign any source of any type anywhere on your network to any fader. Each input channel can be assigned to four stereo busses, plus four pre/post-selectable aux sends, a stereo CUE bus, four mix-minuses and the panel's own bus-minus. Full Vorsis EQ and Dynamics let you sculpt and control your sound with the quality of the finest dedicated outboard processors. The visually-stunning meter bridge features up to four sets of bright, high resolution LED meters, as well as circular LED displays for auxiliary send levels and pan control. A digital count-up/count-down timer is also included.

The LX-24 is advanced in ways that can make a HUGE difference in your capabilities. But it's also immediately familiar to anyone who has ever sat behind a board at a radio station. Use it to make your programming the best it can be. Just plug it into your WheatNet-IP Intelligent Network – with it, and the BLADES across the page, you can, dare we say it, rule the world.

THE LX-24 CONSOLE CONTROL SURFACE FEATURES

Low-profile table-top design - no cutout required

Meter bridge with up to four bright, high-res LED meter sets

Control room and headphone outputs with level control and source selection

Two independent studio outputs

Stereo cue speakers and amplifier, built-into meter bridge

Onboard VGA and USB-Mouse connectors

Event storage (snapshots) and recall

Each input channel features:
- Four stereo bus assigns
- Four pre/post-fader aux sends
- Four mix-minuses
- Bus-Minus
- Source name display
- A/B source selector
- 2 programmable buttons
- Vorsis EQ and Dynamics including 4-band parametric EQ, High- and Low-Pass filters, Compressor and Expander/Noise gate

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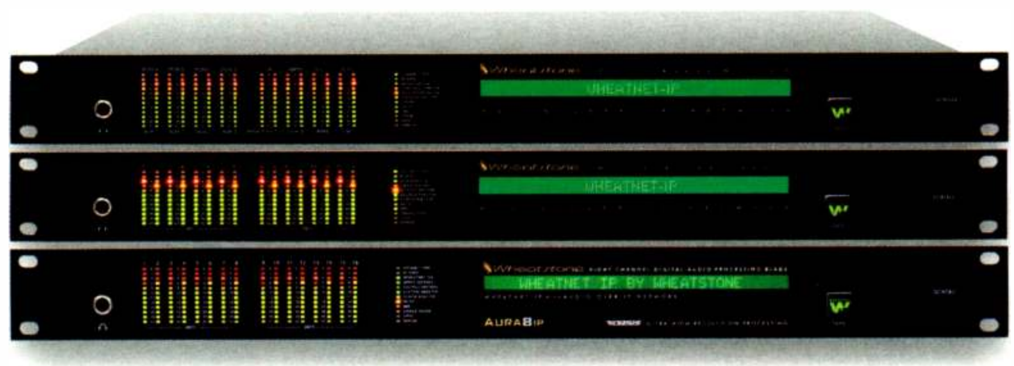
T NETWORK

-ice. it's called The WheatNet-IP Intelligent Network, and it rules.



Our BLADES carry out your orders network-wide at Gigabit Ethernet speeds - no bottlenecks

As an integral part of the WheatNet-IP Intelligent Network, BLADES interface, move, bend, shape, route and control everything you want to do with your audio. If it's audio, a BLADE will handle it - at lightning speed.

Use them organically with our control surfaces, run them from our Glass-E software wherever you have internet access, or control them from the front panels. BLADES make your life incredibly easy and secure.

As you need more functionality, just plug in more BLADES - they come in configurations to handle whatever you need (analog, digital, a/d, mic, MADI). Each BLADE is self-configuring and has the DNA of the entire self-healing network.

With BLADES, you can do everything from a simple (or complex, if you like) snake to STL-over-IP to full-on multi-studio/facility networking - even processing. And because of Wheatstone's partnership with the top suppliers of automation and remote gear, you'll have control over your entire system right from WheatNet-IP. Ruling the world has never been easier.

And this is ALL the extra stuff you need to wire-up the Intelligent Network:

Four CAT-6 cables and a low-cost switch that handles the gigabit speed WheatNet-IP runs at. Let's do the math - plug in eight connectors, power up a console and three BLADES, add your audio and you are ready to rock, roll and rule the radio world. Brilliant, you ask? Nah - just really, really intelligent.

Want to know more?

WheatNet-IP outperforms the other AoIP systems exponentially and is, by far, the most reliable network you can get. Log onto wheatip.com. There is a world of *real* information there. Or, give us a call. There's nothing we like better than talking about this stuff.

See us at IBC 2012 - Stand 14.119

EVERY BLADE FEATURES

Two 8x2 stereo virtual Utility Mixers that can be used for a wide range of applications; for example, using Wheatstone's ACI Automation Control Interface, your automation system can control the mix for satellite or local insertion switching

Front panel bar graph meters switchable to display source input level or destination output level after gain trim

Front panel routing control — any system source to any destination on that BLADE

Front panel headphone jack with source select and level control — monitor any system source

Flexible GPI logic — 12 universal logic ports, programmable as inputs or outputs, routable throughout the entire system

Built-in web server so you can configure and control locally or remotely without having to run dedicated software

SNMP messaging for alerts

Silence detection on each output that can trigger alarms or make a routing change

Silent — no fans — can safely be located in a studio with live mics

Wheatstone

SURESTREAM

(continued from page 1)

inexpensive means to deliver audio, although it is not without its challenges, primarily the fact that the public Internet is an "unmanaged" network.

The lack of IP network management such as QoS mechanisms, traffic balancing or any intercommunication between packet source and packet destination usually results in unreliable and interrupted data streams with large variations of data transmission quality.

Typical phenomena on packetized audio streams in unmanaged networks can be as follows:

Typical phenomena of audio transmission over unmanaged IP networks

1. A high and variable rate of delay jitter caused by bottlenecks on backbone routers, particularly if they operate very close to their overload limit
2. Loss of single audio packets due to network inconsistencies
3. Loss of a cluster of audio packets, sometimes called a "burst" loss
4. Loss of connection, temporary network failure
5. Packets arrive out of sequence due to variations in network latency

Dropped packets are lost and cannot be replicated without mechanisms like Forward Error Correction (FEC). However, even implementing an effective FEC mechanism cannot completely prevent the loss of packets beyond the FEC design limitations. FEC also is costly in terms of delay time between audio input and playout, and it significantly increases the bidirectional network bandwidth required to deliver audio successfully.

THE LIMITS OF UNMANAGED CONNECTIONS

The definition of an unmanaged network is that it operates without management. On unmanaged networks, no corrective action will take place should the traffic on any given segment exceed the nominal load capacity. The unmanaged network will react with an overload condition and start to drop packets due to buffer overloads on router queues, and the packet latency will increase significantly. Conditions like this can be seen as the worst-case scenario for real-time media content streamed over the network.

The audio IP industry has proposed many possible mechanisms to cope with these conditions when a network fails after reaching a certain level of error rate in the transmission chain. The intelligence and control provided by protocols (RCTP/RTSP/DCCP/FEC/QoS, etc.) can help to assure a secure transmission if the network supports these mechanisms, but this is not the case on the public Internet.

Sending broadcast-quality audio over public Internet connections requires a simple and pragmatic approach to counteract the drawbacks of what is otherwise a basically attractive and cost-effective network for professional audio transmission. Effectively, the Internet could replace the use of E1, T1 or ISDN links for professional audio broadcast if only a reliable technical approach could be deployed to overcome the common drawbacks of using this unmanaged public network.

THE INTERNET VS. SYNCHRONOUS CONNECTIONS

The benefits of the Internet as an audio transport backbone are many, some of which are obvious. It is globally and locally available nearly everywhere on Earth. It is cost-effective to access, even as Internet payload capacity and speeds have increased.

The Internet allows communication across virtually any distance and can be leased for short periods of time (such as weeks or months if necessary). Due to an architecture that spans a "web" of interconnection points, the Internet is also self-healing to a degree, which offers some protection against the failure of a single segment in a communications link.

The aim is to make those benefits available for audio transmission from studio to transmitter, studio to studio, or from outside locations or venues where professional broadcasters might currently transmit program audio over synchronized networks like ISDN.

In comparison, synchronized networks offer highly reliable and uninterrupted streaming capacity with a low and constant latency. The bit rate and audio quality on a synchronous network does not dynamically change during transmission and is suitable for almost all low or reduced bitrate audio formats.

The design goal is to make the best characteristics for synchronous networks a reality for transmission of audio over unmanaged IP networks.

PACKETS AND MULTIPLE ROUTES

Due to the nature of the packetized network, a sender is not physically connected to a data receiver as typical in synchronized networks. All IP packets include both the source and the destination addresses, but the exact route that a packet will travel through the IP network is determined dynamically by the network, and therefore is not predictable. The route depends on the actual network conditions and decisions made dynamically by network router protocols (RIP) in order to balance the overall payload as effectively as possible throughout a network segment.

The lack of a direct connection results in situations where two program streams could be directed on different routes from the same source location to the same destination. One stream could travel via route A, and the other through a very different route B. Even individual packets from the same program stream could be sent via different routes. Furthermore, it is not unusual to find that, for example, program A might demonstrate satisfactory performance while program B frequently drops out and has lots of buffer jitter and unacceptable audio.

Recognizing the fact that two data streams from the same source to the same destination can show such different performance characteristics, one is guided to a simple theorem: Using potentially *both* routes A and B for each of the program streams could increase the reliability of both program feeds. Fig. 1 (on page 16) shows the principal of a real world configuration. We call this SureStream — the capability of sending multiple component streams to provide seamless redundancy and reliability for audio transmission over unmanaged IP networks.

THE SURESTREAM ENCODER

A single program audio stream can, when replicated, appear on the Internet as separate streams generated from different sources on different network segments which will then be routed separately. If the packet sequence number generated by the RTP protocol is synchronized properly on all of these replicated streams, they can then be recombined at the receiving end into the original data, with any packets missing from one stream replaced by packets from one of the other streams. But note the packet sequence numbers are the only data point that should be congruent — all other source information must be unique to each stream in order to increase the randomness of the routing in the network. When SureStream is employed, that randomness equals resilience.

APT provides in its products the ability to generate as many redundant streams as required, up to the payload capacity of the IP hardware. The overall reliability increases with each stream added to the mix. Ideally, these streams should be

(continued on page 16)

THE NEW MATRIX

A complete range of handheld audio measurement tools from NTI. You may never need any others.

XL2 Analyzer

A high performance audio analyzer and acoustics analyzer together in one efficient battery-powered package. (acoustics measurements require optional microphone) Designed to close the gap between handheld and benchtop instruments in performance and features, it functions as an audio/distortion analyzer with frequency counter; octave & 1/3 octave real time analyzer (optionally up to 1/12 octave); calibrated sound level meter; FFT spectrum analyzer; polarity tester; scope mode signal viewer; delay time and RT60 reverberation analyzer; and optionally as an STI-PA speech intelligibility analyzer.

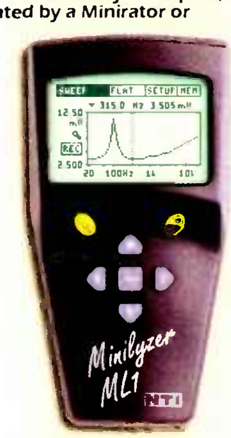
- Broadcast & Studio engineering
- Autoranging audio analyzer for service
- Studio and external acoustics measurements
- Installations, Remotes and Live Venues

Incorporating:

- High (-100 dB) audio performance
- Ergonometric package with brilliant hi-res backlit display
- RTA, Distortion Analyzer and FFT Spectrum analyzer in one
- Balanced and unbalanced audio inputs
- Monitor speaker, headphone output, tripod mount and shock jacket
- .WAV file recording for later reference
- Data & setup memory storage and recall on micro SD card

ML1 Minilyzer
Analog Audio Analyzer

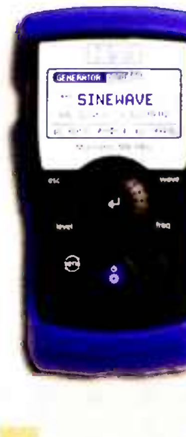
The ML1 Minilyzer is a full function high performance audio analyzer and signal monitor that fits in the palm of your hand. The comprehensive feature set includes standard measurements of level, frequency and THD+N, plus VU+ PPM meter mode, scope mode, a 1/3 octave analyzer and the ability to acquire, measure and display external response sweeps generated by a Minirator or other external generator.



Add the optional MiniLINK USB computer interface and Windows-based software and you may store all tests on the instrument for download to your PC, as well as send commands and display real time results to and from the analyzer.

- Measure Level, Frequency, Polarity
- Automatic THD+N and individual harmonic measurements k2 → k5
- VU + PPM meter/monitor
- 1/3 octave analyzer
- Requires optional MiniSPL microphone for SPL & acoustic RTA measurements
- Frequency/time sweeps
- Scope mode
- Measure signal balance error
- Selectable units for level measurements

MR-PRO Minirator
High performance
Analog Audio Generator + Impedance/Phantom/Cable measurements



The MR-PRO Minirator is the senior partner to the MR2 below, with added features and higher performance. Both generators feature an ergonomic instrument package & operation, balanced and unbalanced outputs, and a full range of signals.

- High (+18 dBu) output level & <-96 dB residual THD
- Low distortion sine waves
- Programmable stepped or glide sweep
- Pink & white noise
- Polarity & delay test signals
- User-generated custom test signals & generator setups
- Impedance measurement & speaker power calculation
- Impedance measurement of the connected device
- Phantom power voltage measurement
- Cable tester and signal balance measurement
- Protective shock jacket

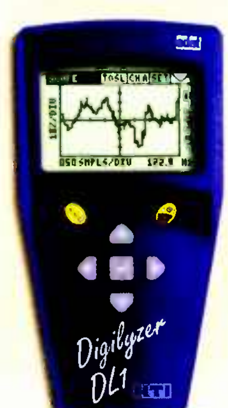
MR2 Minirator
Analog Audio Generator

The MR2 pocket-sized analog audio generator is the successor to the legendary MR1 Minirator. It is the behind-the-scenes star of thousands of live performances, recordings and remote feeds. With its new convenient thumbwheel control system and ergonometric package design, the MR2 is the compact and economical choice for a high performance analog source.

- Intuitive operation via thumbwheel and "short-cut" buttons
- New higher output level (+8 dBu) & low distortion
- Programmable Swept (chirp) and Stepped sweeps
- Sine waves
- Pink & White noise
- Polarity & Delay test signals
- Illuminated Mute button
- Set outputs in Volts, dBu or dBV
- Hi-res backlit display
- Balanced and unbalanced outputs

DL1 Digilyzer
Digital Audio Analyzer

A handheld digital audio analyzer with the measurement power & functions of more expensive instruments, the DL1 Digilyzer analyzes and measures both the digital carrier signal (AES/EBU, SPDIF or ADAT) as well as embedded digital audio. In addition, the DL1 functions as a smart monitor and digital level meter for tracking down signals around the studio. Plugged into either an analog or digital signal line, it automatically detects and measures digital signals or informs if you connect to an analog line. In addition to customary audio, carrier and status bit measurements, the DL1 also includes a comprehensive event logging capability.



- AES/EBU, SPDIF, ADAT signals
- 32k to 96k digital sample rates
- Measure digital carrier level, frequency
- Status/User bits
- Event logging
- Bit statistics
- VU + PPM level meter for the embedded audio
- Monitor DA converter and headphone/speaker amp
- Audio scope mode

DR2 Digirator
Digital Audio Generator

The DR2 Digirator not only generates digital audio in stereo & surround, it is a channel transparency and delay tester as well, all condensed into a handheld package. Delivering performance & functionality challenging any digital audio generator made today, it produces all common audio test signals with sampling frequencies up to 192 kHz and resolution up to 24 bit. The Digirator features a multi-format sync-input allowing the instrument to be synchronized to video and audio signals. In addition to standard two-channel digital audio, the DR2 can source a comprehensive set of surround signals.

- AES3, SPDIF, TOSLink, ADAT outputs
- 24 bit 2 channel digital audio up to 192 kHz SR
- Sine wave with stepped & continuous sweeps; White & Pink Noise; Polarity & Delay test signals
- Dolby D, D+, E, Pro-Logic II, DTS and DTS-HR surround signals
- Channel Transparency measurement
- I/O Delay Measurement
- Sync to AES3, DARS, word clock & video black burst
- User-generated test signal files

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SURESTREAM

(continued from page 14)

delivered to two different networks via two physical IP ports on the encoder. This further increases the robustness and reliability of the overall stream. In practice, a single or multiple component streams could be generated on each port. The concept also works on codecs that have only one physical port but with slightly less effectiveness; there is no way to protect against a "Loss of Connection" (LoC) event.

More processes are performed to optimize each packet. The Statistical Diversity Generator that is part of SureStream ensures that a replicated packet, even with a duplicated RTP sequence number, appears as a unique packet, rather than being an identical copy. On a highly dynamic network like the Internet this additional optimization process can have a significant impact on performance.

THE SURESTREAM DECODER

Once an encoder has generated duplicated streams intended to reach the same destination with the same payload, the decoder must cope with a massive number of redundant packets arriving via different network routes. APT's audio codecs with SureStream activated are able to compose a reconstructed stream with the correct packet sequence from multiple incoming streams.

On a first-in-first-out basis the re-sequencer/combiner generates a single packet sequence by reading packets from all streams with identical content. Whatever packet arrives first will be inserted into the reconstructed sequence; all other redundant packets will be dropped.

Together with an adjustable delay jitter compensation buffer, this is a highly efficient approach to cope with the behavior of unmanaged networks like the public Internet or G3/G4 services.

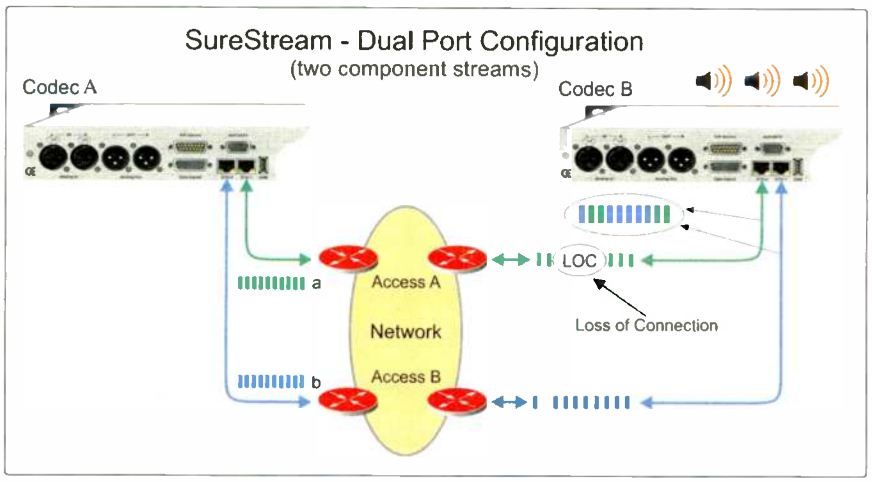

Fig. 1: The principle of redundant IP packet streaming on two IP ports.

AUDIO PERFORMANCE TESTS

APT engineers have performed audio transmission tests using consumer-type ADSL access points connected to SureStream-enabled codecs at each end. These DSL access points were selected from different DSL providers to increase the chance of being routed through different network segments. The results show the recombined packet sequence was never interrupted nor any audio glitch recorded during the period of the test, while individual component streams showed LoC errors and a substantial number of dropped packets.

REAL-WORLD TEST #1 – APT BELFAST OFFICE TO WORLDCAST OFFICE IN MIAMI, USING OSLO CODEC

For this test, the link was established using two WorldNet Oslo codec frames, connected to two different DSL Internet providers in each country. Very generic and inexpensive DSL services were chosen. The streams were configured using the Enhanced apt-X algorithm at 384 kbps with a 512 byte packet size, resulting in an audio data rate with packetization overhead of approximately 410 kbps (including an allowance for control data). This configuration will provide a frequency response of 22.5 kHz, a suitable quality for digital broadcast including DAB and iBiquity HD Radio. The test lasted over 96 hours.

The fact that the test was set-up in this way does not imply any restriction for other configuration options. It is a common setting and proves the principal impact of SureStream. A subsequent test setup investigated both the usability for 5.1 surround sound transmissions over the public Internet and the benchmark figures of the delay-jitter buffer settings.

Fig. 2 shows the performance monitor screen of the WorldNet Oslo on the

(continued on page 18)

SURESTREAM

(continued from page 16)

receiving end. This test was carried out with the first component stream on route 0 (IP port 5004) and a second component stream on route 1 (IP port 5006). The streams (routes) were received by two ADSL access points from different providers on both physical Ethernet ports of the IP transport card. Note that streams can also be duplicated on the same ADSL link if required (single IP port), however this doubles the bandwidth requirement on the DSL, and offers no protection against LoC events.

The third stream, on route #4, represents the reconstructed stream derived from both component streams on route 0 and route 1. This stream is configured only to display the effect of the SureStream technology. This third stream has no impact on the performance or on any other operational behavior and would typically not be configured for normal operation.

STREAM 0

Stream 0 is the first component stream and shows more than 23 million packets received. It also shows that this route has dropped 1,599 packets; this number of dropped packets is not unusual for audio sent via the public Internet over four days of use. Furthermore, this stream

Stream Port		Packets	Bytes	Dropped Pkts	Reseq Activity	LOC	Dup Pkts
0	5004	23753567	658885710	1599	0 (0.00%)	3	0

shows three LoC errors. An LoC is a total stream interruption. These errors might reflect the fact that DSL circuits are often dropped and re-synchronized by the service provider once every 24 hours (usually around midnight).

In summary, each dropped packet would at minimum result in audio glitches or artifacts, and could cause a complete muting of the audio — this streaming performance would not be acceptable to a professional broadcaster.

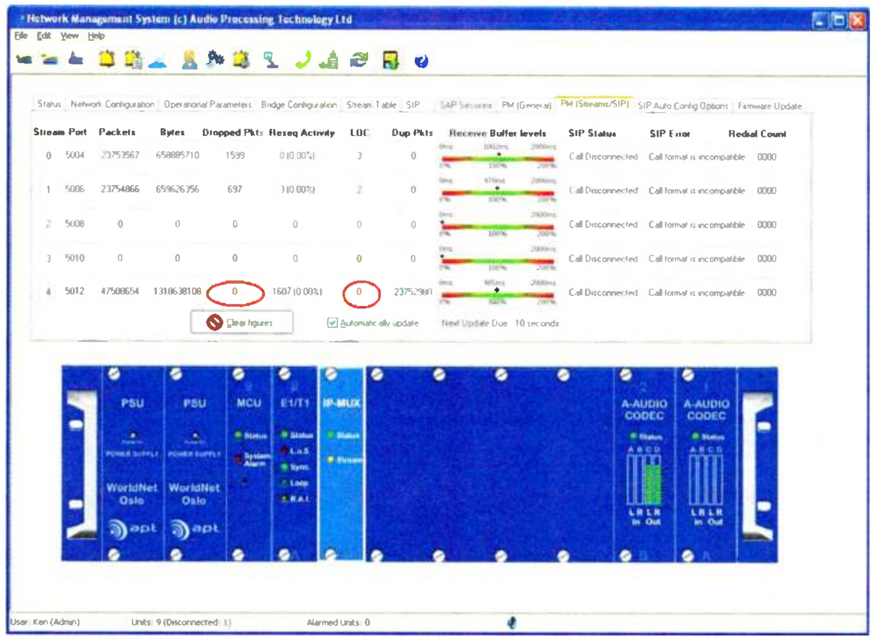

Fig. 2: Performance figures of the component and the reconstructed streams (WorldNet Oslo).

STREAM 1

Stream 1 is the second content stream (with identical content) and shows a

1	5006	23754866	659626356	697	3 (0.00%)	2	0

similar number of received packets. It also shows that this route has dropped 697 packets and has had two LOC errors.

In terms of results, this stream performs better than stream 0 but still has stream interruptions and a large number of dropped packets.

RECONSTITUTED STREAM 4

Again, this third stream has no impact on the transmission performance and is just configured to help visualize the effect of the SureStream solution. A nor-

4	5012	47508654	1318638108	0	1607 (0.00%)	0	23752980

mal operational setup wouldn't need the third stream, but it helps to monitor the performance of each component stream (here, stream 0 and stream 1). The reconstituted stream is the output of the SureStream engine, which reads the content of each packet on all component streams, seamlessly reconstructing the audio program that is fed to the audio card.

The reconstructed stream does not show any dropped packets or any LoC errors. Instead, it shows a huge number of duplicated packets that were discarded by the SureStream engine. Furthermore, it shows that 1,607 of the packets received from both component streams needed to be reordered by the resequencer before playout.

REAL-WORLD TEST #2 – LONG-TERM TEST BETWEEN BELFAST AND MIAMI WITH HORIZON NEXTGEN UNITS

Following a series of shorter tests, a long-term test was initiated between the Worldcast APT offices in Belfast and Miami. For these tests, the stereo Horizon NextGen codecs were used. Again, two basic DSL connections from different providers were linked to the two Ethernet ports on the codecs at each end of the connection. See Fig. 3.

For this test, three component streams were configured, this time at a lower but still common bit rate. We used the Enhanced apt-X algorithm with 16-bit coding passing 15 kHz stereo audio. The base rate for this algorithm is approximately 256 kbps, which results in around 280 kbps with packetization overhead.

Again, the results are obvious and impressive: over 25 million packets sent (17 million duplicate, or about 8 million payload). The receive codec resequenced 8,688 packets that arrived out of order. The component streams suffered a total of 20 LoC errors and more than 15,000 lost packets in all — and SureStream was able to recover all but one packet, with zero LoC errors. In terms of reliability, the loss of one packet out of 8 million equates to a 99.99999 percent success rate, or "seven nines" — better reliability than many providers will guarantee on even a T1 or ISDN line, accomplished using affordable equipment on low-cost public Internet connections.

PERFECT IP AUDIO OVER IMPERFECT NETWORKS

When determining design goals for the SureStream technology, the APT engineers focused on achieving the performance benchmarks set by synchronous connections — to deliver broadcast-quality audio over basic open public Internet connections, without any variations in audio bandwidth or delay/latency, no restrictions in what audio algorithms could be used, and achieve a transmission reliability of 99.99 percent or better.

With SureStream, those goals have been achieved. While no transport mechanism can be 100 percent perfect all the time we can safely say that SureStream does in fact deliver "Perfect audio over imperfect networks."

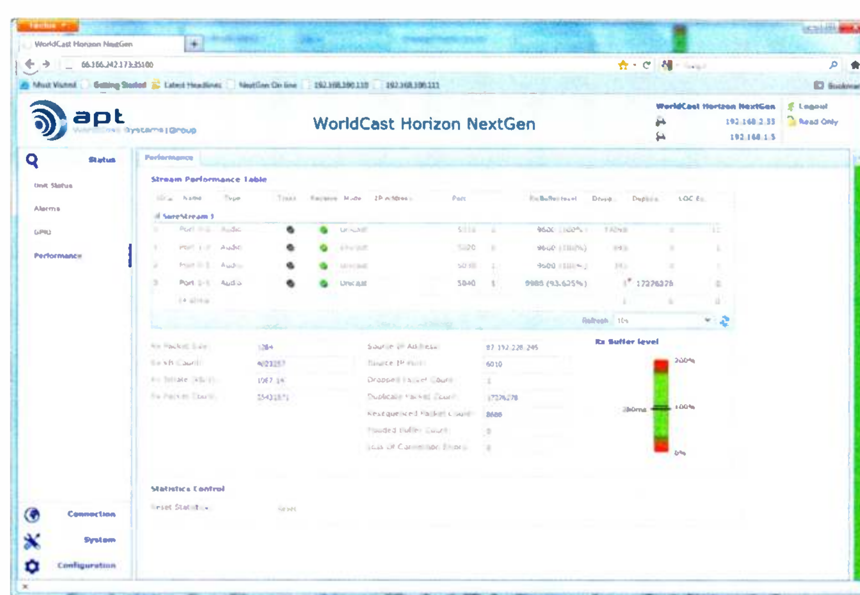

Fig. 3: Long term test results.

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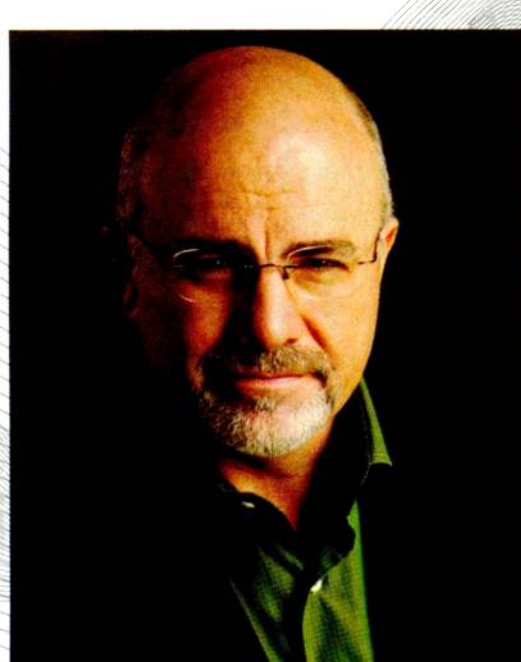

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THERMAL IMAGING

(continued from page 1)

transmission systems in a broadcast facility, and some maintenance can even be performed in a systematic manner thanks to modern technology.

However, antenna and feed-line systems still remain difficult to troubleshoot effectively. To make matters worse, when antenna problems do arise, there is often little that can be done to rectify the problem quickly without engaging an expensive tower crew to investigate. So the question becomes, "How do we troubleshoot problems that are out of reach and difficult to see?"

In February, this question was posed rather suddenly to McKenzie River Broadcasting in Eugene, Ore. McKenzie River Broadcasting combines two of their signals, KKNU(FM) and KMGE(FM), with a high-level combiner. KKNU and KMGE contribute a total of 50 kilowatts from a branch combiner into a 4-inch rigid feed line and antenna system. That month, I began to notice intermittent VSWR overloads from high-speed "Watt Watcher" detectors. Both transmitters were displaying VSWR overload and PA screen overload a few times per week — indications of a problem in the transmission system. Clearly something was arcing and causing a temporary short circuit. Where could it be?

Early stages of arcing often are difficult to detect and even more difficult to pinpoint. There are only a few places to look, but a direct mechanical inspection is time-consuming, possibly requires a tower rigger and requires the stations to be off the air. Arcing became more regular in a short amount of time until the transmitters were tripping off the air several times a day.

The combiner and associated components were dismissed first by a physical inspection and looking for areas of excess heat. Options for detection and repair became limited once the problem was tracked to the tower transmission line and antenna system. A time-domain reflectometer at ground level is only helpful if there is a total burnout resulting in an open circuit and that wasn't the case yet. Of course, a total burnout is also the worst-case scenario for the antenna system, and the one thing McKenzie River Broadcasting wanted to avoid. Both FM signals for McKenzie River Broadcasting function into a single broadband antenna at the top of a 600-foot tower, so a failure would take both stations off the air at the same time.

Conventionally, after a problem of this sort is tracked to the antenna system, the only solution is to remove and inspect the entire feed line, which for McKenzie River Broadcasting is 600 feet of 4-inch rigid line. If the line sections look adequate, then the antenna would be taken down for inspection. This would be an expensive and a time-consuming project. It would have required the other FM stations on the tower to operate at safe power levels for days.

Woods Communications offered an alternative solution. Tom Woods has been a broadcast engineer for some 20 years. McKenzie River Broadcasting hired him to help identify the antenna system failure.

In 2011, Woods had acquired a professional thermal imaging camera. Woods knowledge of antenna systems, combined with his ability to climb towers, led him to believe that thermal imaging could be used to help identify the problem without having to resort to taking down the transmission line or antenna.

Woods believed the thermal imaging camera provided a way to locate an imminent burnout. The idea was to use a thermal camera and map the antenna and rigid line system temperatures in hopes of finding discrepancies with the system that would indicate a pending failure. This would require the other high-power FM and TV stations on the tower to only operate at the low and safe power level for only an hour.

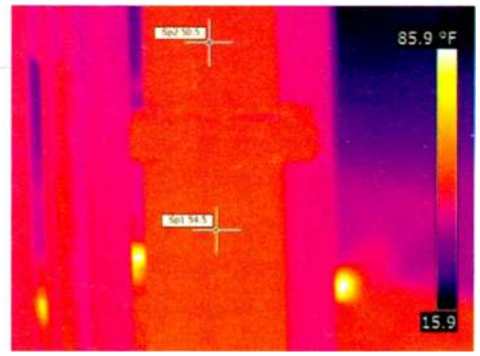

Fig. 2: Typical rigid-line flange connection. Temperature was consistently about 4 degrees higher on the low side of the flanges but this indicated normal operation.

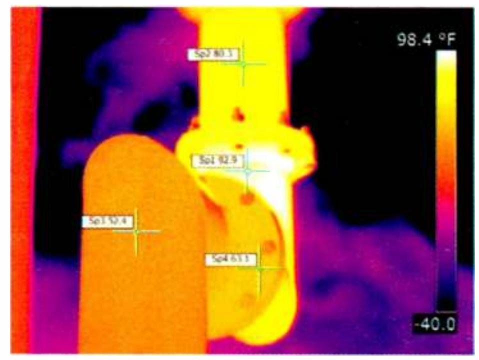

Fig. 3: Thermal image of an elbow at 540 feet.

THERMAL MEASUREMENT BASICS

Thermal technology works on the theory that all objects emit heat. An infrared camera is a non-contact device that detects infrared energy (heat) and converts it into an electronic signal, which is then processed to produce a thermal image on a video monitor and perform temperature calculations. Heat sensed by an infrared camera can be precisely quantified, or measured, allowing you to not only monitor thermal performance but also identify and evaluate the relative severity of heat-related problems.

For this application, Woods Communications used a FLIR T-400. We started temperature measurements in the transmitter suite and immediately found a lurking problem, although it wasn't the problem we were looking for. Fig. 1 (on page 1) shows an overheated contactor in KKNU's three-phase disconnect panel. This excess heat eventually will cause the metal to fatigue, get even hotter and eventually fail. I can now schedule time to replace this panel before a failure occurs.

From there, measurements of the branch combiner and all rigid feed line paths to the tower were taken. Although no further weaknesses were discovered inside the building, the information gathered provides a frame of reference for future problems.

Next, Tom and I took our efforts outside. Fig. 2 depicts one of the sections near the bottom of the tower. Thanks to the T-400, it was easy to see temperature both above and below each flange. The ambient temperature on the ground that day was 46 degrees. The temperature on the exterior of the line measured at about 52 degrees at the bottom and incrementally decreased further up the tower as ambient temperature dropped. However, the temperature difference between the top and

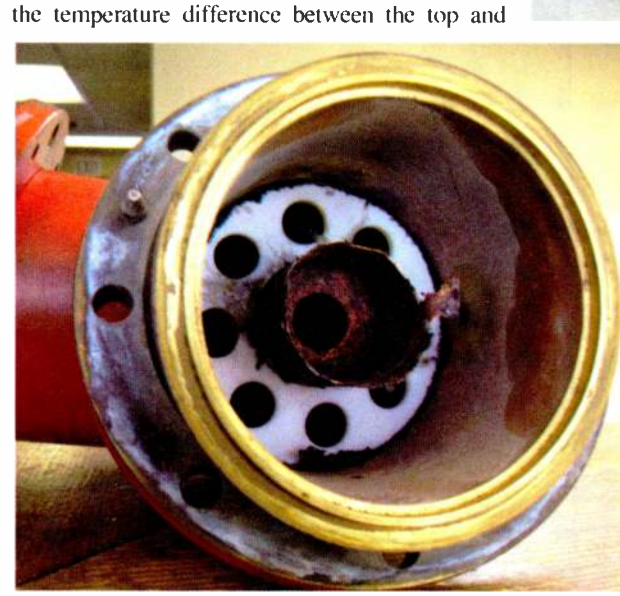

Fig. 4: This internal connection at the elbow is nearly gone — the entire end is burned off. Note the metal fragments piled up on the insulator — why the arcing gets worse after the first event.

bottom of the bullet remained consistent at 2 to 4 degrees. This temperature difference is attributed to infrared reflection from the bottom of the flange down on to the line. I noticed that the bottom of each bullet was slightly warmer than the top. I suspect that this has to do with the ability of the inner conductor above the bullet to sink heat away from the bullet.

After scanning 48 sections of rigid line, the problem finally was discovered at the 540-foot level in an elbow. Fig. 3 depicts the largest temperature difference. Heat at the elbow measured around 90 degrees. The weakness in McKenzie River Broadcasting's antenna system had been identified.

The elbow was later removed and replaced, and McKenzie River Broadcasting quickly returned to full power. Fig. 4 shows the damage inside the elbow. A close look reveals the arcing path from the inner to the outer conductor. Although there wasn't a complete burnout, severe heating damage to the bullet is visible in Fig. 5.

The survey took about 90 minutes and repairs were completed in less than six hours. Repairs would have been more extensive and costly in terms of lost air time had the problem worsened to the point of a complete burnout.

Fig. 5: The line bullet is heavily discolored, indicating overheating.

Using infrared photography to survey your entire antenna system while it is in operation is a tremendous troubleshooting tool that can prevent future catastrophic failures. The use of thermal technology has been invaluable to Woods Communications in the work of identifying weaknesses that can't otherwise be observed.

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METER

(continued from page 22)

$V_s = V_m$; and
$I_s \times R_s = I_m \times R_m$

We can solve for R_s (shunt resistance) by rebalancing the equation:

$$R_s = (I_m \times R_m) / I_s$$

Substituting $I_{in} - I_m$ for I_s yields:

$$R_s = (I_m \times R_m) / (I_{in} - I_m)$$

Let's give it a try using that typical 200 ohm value for the ammeter resistance value.

$$R_s = (0.001 \text{ amp} \times 200 \text{ ohms}) / (1 \text{ amp} - 0.001 \text{ amp})$$
$$R_s = 0.2 \text{ volt} / 0.999 \text{ amps}$$
$$R_s = 0.2002002 \text{ ohms}$$

At this low a resistance value, the pragmatics of creating a pure and exact resistance at that small value are important, as even stray resistance of the connection wires can affect measuring accuracy. Special wire and bar stock are made just for the purpose of achieving small resistance values with high current handling. Most often a manganin and/or constantan alloy that exhibits a low coefficient of resistance change with temperature is used for highest accuracy across the current range of interest. Most full-line meter manufacturers usually have shunts available off the shelf to multiply their ammeters scales in standard values.

In the example above, if ultra accuracy is needed, a 2 ohm shunt would be used and a small value "trim" pot can be added in series with the meter (see Fig. 3). Small in our case would be a 25 to 50 ohm (usually multi-turn) pot. Referenced against laboratory grade instruments, the trim pot would be adjusted to match and double checked at full scale as well as several intermediate values of current.

The next SBE certification exams will be given in the local SBE chapters Nov. 2–16. Closing date for signing up is Sept. 14. If you are interested and ready to take the exams, we strongly suggest that you sign up ASAP, as the next exams are scheduled for Feb. 8–18, 2013.

Charles "Buc" Fitch, P.E., CPBE, AMD, is a frequent contributor to Radio World. Graphics were drawn by Victor Osorio, the full-time CE at HRRZ AM and HRKD FM in Juticalpa, Olancho, Honduras. Reach him at victor@radiojuticalpa.com.

Missed some Certification Corners or want to review them for your next exam? Find past Certification Corner articles under the Columns tab at radioworld.com.

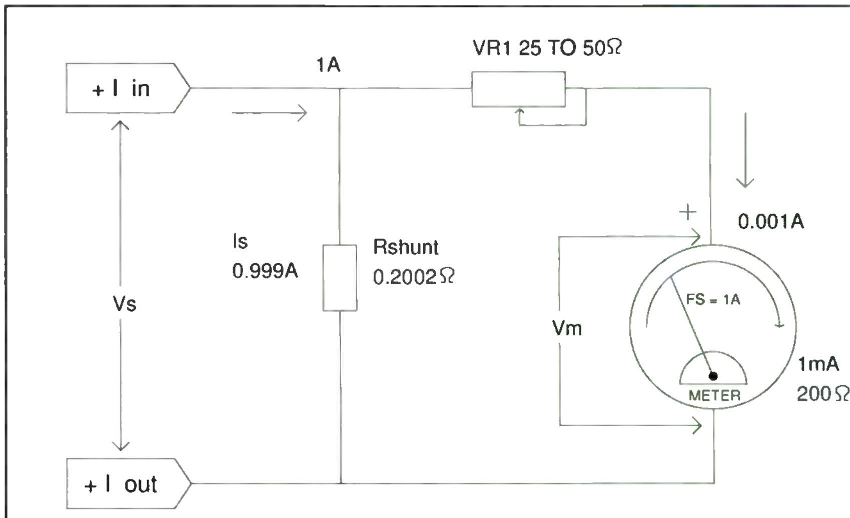

Fig. 3: Variable Shunt Resistor

The Mismatch Patch

Question for next time

(Exam level: CBNT)

A cabling impedance mismatch can be caused by which of the following?

a. Using RJ45 connectors on Cat-6 cable.
b. Running cable in an overhead cable tray near electrical conductors.
c. Nicking cable conductors when stripping.
d. Mixing shielded and unshielded twisted pair cable in the same segment.
e. Not following ITE/EIA/IEEE standard UTP-101 regards wire twist format (i.e. left overhand).

The Flexible Meter

Circuit Design Allows Simple Meter To Measure a Wide Range of Values

Meter, Meter on the Wall

Question from the June 13 issue
(Exam level: CBRE)

The analog DC ammeter with an actual full-scale value of 10 milliamperes in your transmitter that measures 1 ampere of plate current has failed and the only linear meter that fits in the same space available is a 1 mA unit. Could you use this available meter as a replacement?

a. Yes, with a series resistor of 10 k
b. Yes, with a series and parallel resistor both 10 k
c. No
d. No because the scale is set by the coil wind count
e. Yes, with a select parallel resistor across the meter contacts

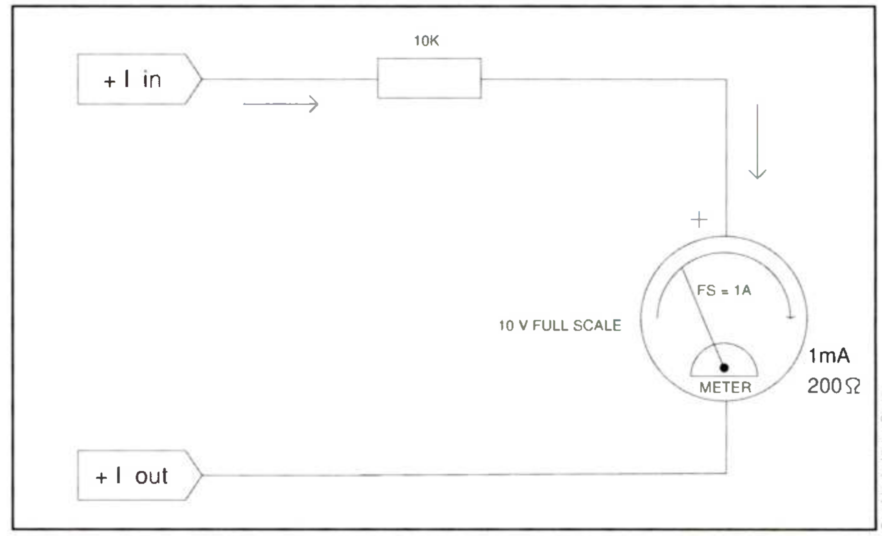

Fig. 1: Basic Meter Circuit With Series Resistor for Voltage Measurements

Graphics by Victor Osorio

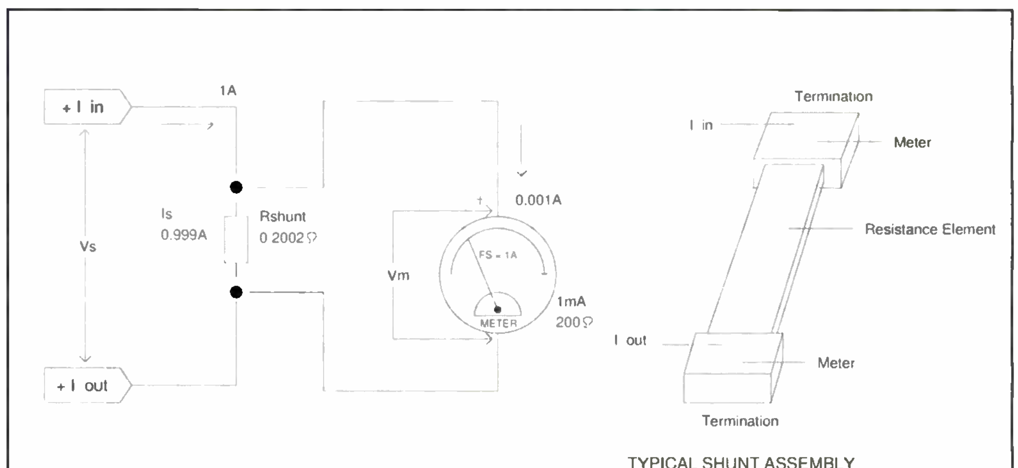

Fig. 2: Shunt Resistor for Current Measurement

BY C.S. FITCH, P.E.

SBE certification is the emblem of professionalism in broadcast engineering. To help you get in the exam frame of mind, this column poses typical questions. Although similar in style and content to the exam questions, these are not from past exams nor will they be on future exams in this exact form.

Tentatively, let's assume that the meter has no resistance, which will simplify our discussion for the moment. We'll come back to this, I promise.

In the ultimate analysis, all analog DC meters are actually displaying current flowing through the meter coil that deflects the pointer. With that in mind, answer (a) would really be creating a meter that responds to voltage.

In Fig. 1, we have voltage impressed between the top of the series resistor and the negative terminal of our meter. The values shown of 10 volts and a series resistor of 10 k would cause a current flow of 1 mA or full scale on our replacement meter. When you buy a voltage meter, this series resistor is inside already.

To keep the parts acquisition count down, most transmitter manufacturers use the same model milliammeter in every position and a variety of this series resistor scheme to create the appropriate scale values for voltages. For example, a full scale of 3000 volts using a 1 mA meter would require a 3 megohm series resistor. Preferably, a precision resistor would be used in this instance to enhance accuracy.

The answer (b), with at least one of the 10 k resistors being in series, would make it impossible to use this meter as an ammeter to represent 1 ampere as that 10 k resistor would drop any impressed voltage notably. If you throw in the voltage divider action of that shunt 10 k resistor, the simple reality is that this arrangement is just a more complicated "voltage variable" version of Fig. 1.

The answer (c) is reserved for someone who either has no knowledge of the topic or didn't get a good night's sleep the night before the test! One should always come to SBE exams well rested and with a clear mind. This is the last choice to go with until all other options have been eliminated.

The wind count mentioned in answer (d) is a factor in creating the small DC resistance of the meter (that we decided above to discount momentarily), but is just one element involved in determining the full scale value.

HOW IT WORKS

To answer the question, (e) is correct, as we can increase the value displayed on this 1 mA meter to be calibrated to 1 amp FS with a parallel resistor, known as a shunt, across the meter contacts.

How do we determine this shunt resistor, which will route all current above 1 ma around the meter leaving just an analogous, representative current to be displayed on the meter?

Meters do have resistance, and since we now need to consider parallel paths for current flow, we really need to know the actual value of the meter coil resistance. A typical value for a 1 mA meter is about 200 ohms. If you don't know the meter resistance, you'll have to ascertain what it is.

In most cases, it is easier to calculate (or measure) the voltage across the meter movement for full-scale display. If you don't know the resistance, it can be measured with a digital multimeter. The current from most digital multimeters is low enough not to cause damage to the meter, but the pointer may swing rather violently. Connect with reverse polarity to minimize the risk of bending the pointer.

Fig. 2 shows our 1 mA meter with a shunt resistor across the meter contacts. We've assumed our meter's internal resistance is 200 ohms. The voltage across the meter and the shunt is in parallel so V_s (= voltage across the shunt) is equal to V_m (= voltage across the meter winding). But we also know that V_s must be equal to the current through the shunt times the shunt resistance and that V_m must be equal to the current through the meter times the meter resistance.

In shorthand here are the relationships:

(continued on page 21)