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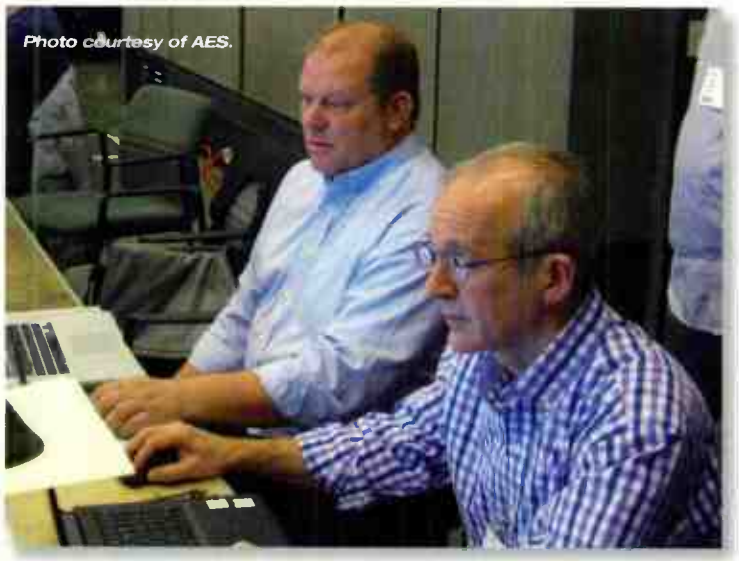
Photo courtesy of AES.

Wheat Goes To Washington For AES67 Plugfest

Wheatstone's engineers arrived at NPR's headquarters in Washington, DC in November with a WheatNet-IP audio network to participate in the second AES67 plugfest. This plugfest was a follow up to the AES67 system compatibility testing conducted in Munich last year, and provided for further testing on multicast as well as unicast streaming.

AES67 requires support for both multicast and unicast streaming, the former of which needs the Session Initiation Protocol (SIP) for connection management. A number of products participating in the plugfest support unicast and SIP, including our WheatNet-IP audio network.

Thirteen products were tested, with AES67 implementations varying from software on a PC to hardware-based FPGA solutions.

According to a preliminary AES report summing up the plugfest, "Although these tests involved a growing number of devices compared to the previous plugfest, a majority of unicast streams interoperated successfully." However, because SIP interoperability was not achieved in some cases, the report suggests that an SIP technical overview and recommendation be published prior to subsequent AES67 plugfests in order to ensure the best possible conditions for SIP interoperability.

Multicast interoperability was also thoroughly tested during the plugfest, and according to the preliminary report, "most combinations (94%) were successful. Many of the receivers were able to interoperate despite some conformance issues."

The plugfest took place in November to confirm the interoperability of various products according to the AES67 standard that was first published in 2013 and revised in 2015. AES67 requires interoperability with linear PCM audio coding, a sampling frequency of 48 kHz, 16 or 24 bits-per-sample, 1 to 8 audio channels (2-channel stereo presumed to dominate), and a packet time of 1 ms.

The next plugfest is expected in 2016 in the U.K.

For more IP Audio News: INN31.wheatstone.com

IP Audio, Par For Australian Open Course

By George Biagioni

George Biagioni is IT Director for Crocmedia, an independent syndicator of sports content located in Victoria, Australia.

We recently returned from the 2015 Australian Open Golf Tournament, where my crew and I spent the better part of a week making the rounds and reporting live to spectators there as well as to listeners tuning in to sports radio station, SEN, in Melbourne and SportFM 9.13 in Perth. This marks the second year for Australian Open Radio, a temporary low-power station that Golf Australia contracted my company to set up in order to bring fans closer to the action. This special-event broadcast presented some unique challenges, and therefore required a most interesting mixture of technology to reach the ears at the tournament as well as those listening elsewhere.

To learn how we made it all work using 4G iPhone 6s with Report-IT, Tieline Genie distribution, a 5W transmitter, and Wheatstone IP audio networking, audio processing and IP console...

...go to INN31.wheatstone.com

Your Question Answered

Q. What are the benefits of multiband voice processing?

A. Multiband processing on voice can help in many different ways. It can help voice cut through in audio challenged media like AM and low bitrate streams. It can also smooth out differences in voice textures between multiple hosts using the same studio/microphone. With news talk formats moving towards higher quality mediums like FM and FM-HD channels, the tailoring of your talent's audio using multiband mic processing can help increase TSL. Finally, multiband voice processing helps your jock cut through when talking over loud CHR and rock recordings.

For more IP Audio News: INN31.wheatstone.com

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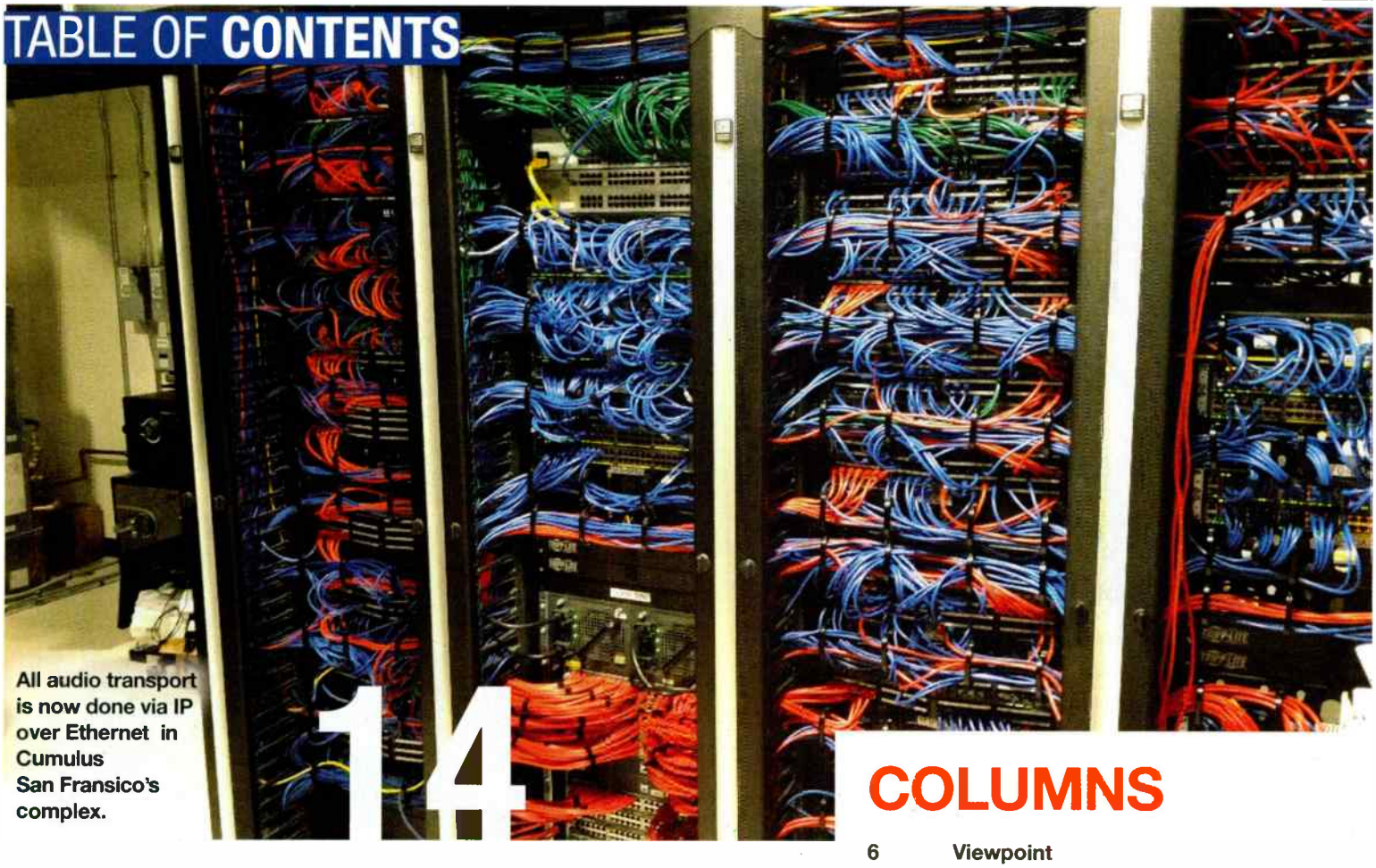

All audio transport is now done via IP over Ethernet in Cumulus San Fransico's complex.

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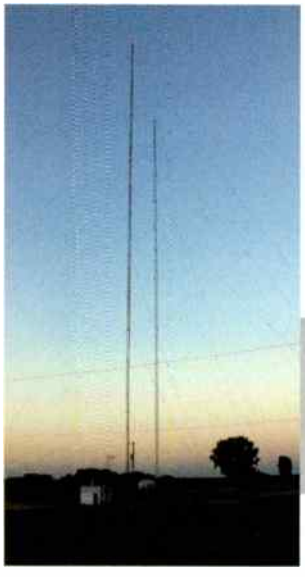

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COLUMNS

DEPARTMENTS

On the cover: Cumulus San Fransico's Studio K, featuring an Axis Fusion console, Axia power stations (in the rack below) and a Telos Vset12.

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- Initiate audio playback from hot buttons
- Run macro command from hot buttons
- Secure access to your system

VIEWPOINT

Did You Survive Winter's Stress Test?

As I write this, New York, Washington, D.C., and much of the eastern seaboard is still digging out from January's Winter Storm Jonas. I'm sure many of you spent much of your time over that weekend tending to your station(s).

Now that the blizzard has passed, think of the silver lining: It was a great opportunity to stress test your main and backup systems!

The big natural disasters give us a chance to see how well our systems were able to withstand the wrath of nature. As a fellow natural disaster veteran, I recommend you do a couple of things, before too much time goes by:

- Hold a departmental debriefing meeting to gather facts about what worked — and what didn't
- Make up an action list of steps needed to take care of all the systems that didn't live up to expectations
- Draft another list of ways that the next (inevitable) disaster can be handled more effectively

As I said, don't wait too long before the debriefs and brainstorming sessions regarding what could be done better next time; it's best to do so while the details are fresh in your memory. After all, the next storm could be right around the corner.

In this issue, we have three "hands-on" articles designed to help you deal with typical needs of a radio station.

Chris Cottingham is back with his series on remote access; he demonstrates how to configure a firewall to allow for remote connections to an AoIP codec. There's nothing difficult about that — though you may not realize it until you get it done the first time.

Aaron Read of Rhode Island Public Radio describes how to use dual IP links in order to drastically reduce cost, while increasing the reliability of AoIP connections.

And in Tech Tips we continue our series on rehabilitating old vacuum tube transmitters; we're looking at the final amplifier input circuit in detail this time.

We all like to look at "shiny, new stuff" too, right? Mark Greenhouse gives us a detailed look at the new and excellent Cumulus Media facility in San Francisco. Its location holds a special place in my heart.

If you find yourself with capital to spend on a new on-air phone system, read our Trends in Technology article on that subject. We've included recording and playback systems as well. Recorders aren't just digital; they're networked now, and that can change your workflow considerably.

Lee Petro's FCC Update column tackles what can happen to stations that don't take all necessary steps to comply with both the letter and the spirit of the FCC rules. The lesson applies to all broadcast companies — big and small.

Installed on the last page of this month's issue you will once again find the Wandering Engineer's Sign Off. He (or she) poses the question: What attracts us to, and keeps us interested in, a vocation that, perhaps, values our highly-developed skills less than others would? Is it because the vocation of broadcast engineering is so closely related to the avocation of ham radio? You be the judge. I've said too much already.

Doug Irwin, CPBE AMD DRB | Technical Editor

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THE RADIO TECHNOLOGY LEADER

February 2016 | Vol. 22 No. 2

EDITORIAL
Technical Editor: Doug Irwin, CPBE DRB AMD
Managing Editor: Emily Reigart

CONTRIBUTORS
Fred Baumgartner, Russ Berger, Chris Cottingham, Lee Petro, Jeremy Ruck, Fardau van Neerden, The Wandering Engineer, Chris Wygal

CORPORATE
President and CEO: Steve Palm
Chief Financial Officer: Paul Mastronardi
Controller: Rick Ng
Group Circulation Director: Denise Robbins
Vice President of Web Development: Robert Ames

VIDEO/BROADCAST GROUP
Executive Vice President: Carmel King
Vice President of Sales/Group Publisher: Eric Trabb

ADMINISTRATION AND PRODUCTION
Editorial Director: Paul J. McLane
Production Manager: Lisa McIntosh
Advertising Coordinator: Caroline Freeland

CIRCULATION
Group Director, Audience Development: Meg Estevez
Circulation Manager: Kwentin Keenan
Circulation Coordinator: Michele Fonville

ADVERTISING SALES REPRESENTATIVES
Publisher, U.S. Sales: Steven Bell
sbell@radiomagonline.com | 212-378-0400 x519

International Sales Manager: Rafaella Calabrese
rcalabrese@nbmedia.com | +39 320 8911938

Japan: Eiji Yoshikawa
callems@world.odn.ne.jp | +81 3 3327 5759

Asia-Pacific: Wengong Wang
wwg@imaschina.com | +86 755 83862930/40/50

Member: American Business Media

A NewBay Media Publication

NewBay Media

NewBay Media, LLC
28 East 28th Street, 12th floor
New York, NY 10016

SUBSCRIPTIONS: Free and controlled circulation to qualified subscribers. Customer Service can be reached at: newbay@computerfulfillment.com or by calling 888-266-5828 (USA only) or 978-667-0352 (Outside US) or write us at Radio Magazine, P.O. Box 1884, Lowell, MA 01853, USA. Back issues are available by calling Customer Service.

LIST RENTAL: 914-925-2449
danny.grubert@lakegroupmedia.com

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Radio, Volume 22, Number 22, (ISSN 1542-0620) is published monthly by NewBay Media LLC, 28 East 28th Street, 12th floor, New York, NY 10016. Periodical postage paid at New York, NY and additional mailing offices. Postmaster: Send address changes to Radio, PO Box 1884, Lowell, MA 01853.

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Resurrect the Input Circuit Of a Tube Transmitter

by Doug Irwin, CPBE AMD DRB

In the first three installments of this series, we discussed whether or not an old transmitter was even worth fixing, followed by some techniques I've used to repair high-voltage power supplies, followed by analysis and repair of control circuitry. We've worked our way up to the RF amplifier deck. Now, start by working on the input circuit.

At this point, you should be reasonably certain the power supplies are functioning correctly, and that the control circuitry is functioning correctly. However, you won't be 100 percent sure until you're able to check all of the transmitter's functionality, at the nominal power expected.

DESIGN CONSIDERATIONS

Likely you know that the vacuum tube amplifier stage has fairly high input impedance; but because the transmitter input standard is 50 ohms, an impedance matching network is necessitated in the design. The circuit is frequency-dependent as well. Its job is to take RF power coming in from a 50-ohm source and generate a much higher peak-to-peak voltage on its output, connected to the control grid, so that plate current is pulsed at carrier frequency. So in trying to resurrect the input circuit of an old tube transmitter, we need to be certain the following conditions are indicated or met:

- Control grid bias power supply has the correct voltage.
- RF is being coupled all the way through the input matching network. This is indicated by control grid current in the transmitter metering.
- The reflected power, back to the driving source (your exciter), is not too high.

WE NEED YOUR TIPS

Tech tips may be suitable to earn SBE recertification credits. Send your tips to radio@RadioMagOnline.com.

The final amplifier input section (below the tube socket) should be clean and inspected for loose or broken parts as part of the rebuild process.

BIAS POWER SUPPLY

Review the control grid bias power supply function again. This voltage biases the tube at cutoff or beyond, so that the static plate current drawn, with no RF drive, is low. There might be a variable resistor in the circuit to set that voltage: See factory test data, or circuit description, to determine the correct voltage.

Setting it too high (too negative) will require more drive from the exciter; setting it too low (not negative enough) will allow more static plate current, thus lowering the overall efficiency. There will likely be part of the control circuitry that tests for the presence of this power supply.

INPUT MATCHING NETWORK

An unfortunate reality in the use of vacuum tubes is that, right from the factory, they are not all exactly the same. The inter-grid capacitances will be slightly different from tube-to-tube.

Additionally, when you consider the manufacturing process, each transmitter needs to be built to work anywhere from 88 to 108 MHz. As I mentioned, the input matching network is frequency-dependent and needs to be tuned to the transmitter frequency; thus, the transmitter design calls for elements that provide for tuning: Variable capacitors, variable inductors, or both.

It's also very important to realize that these parts might be slightly different from one another, depending on what part of the FM band the transmitter was ordered at originally. If you need to change frequencies on this old transmitter, you'll need to be certain that the input network components will work for the part of the FM band to which you're moving. If you are moving a transmitter to 88.5 MHz, as an example, from its original frequency of 107.1 MHz, it might not work without key components in the input circuit being replaced. There again, you'll need the manual so that you can check this.

During its former life, this transmitter had thousands and thousands of hours with air blowing against these input circuit components. Naturally, they accumulated dirt. The first thing you should do as part of the RF deck project is to very carefully clean and inspect the components, looking for obvious problems like failed (and broken) components. After that, make sure the front-panel controls actually work: when you crank input tuning and/or loading controls, see that the associated components in the input circuit are actually moving, and in both directions.

Assuming all of the componentry looks normal, when you energize the plate circuit

of this transmitter, you should be able to "peak" the grid current, using the input tuning and/or loading controls. At the same time the load that this input circuit presents to the driving source should be close enough to 50 ohms that the source can actually operate.

All RF gasketing should be clean and in place. More next month.

REFLECTED POWER FROM INPUT CIRCUIT

Aside from providing the correct peak-to-peak voltage necessary to drive the input (control grid) circuit, the input matching network must "match" that impedance back to something that looks reasonably like 50 ohms to your exciter.

If not, your exciter's reflected power metering and control will likely either shut it off, or at minimum, ramp its power back. When this happens you lose drive to the transmitter itself. When tuning the input circuit, do your best to find the point at which drive is maximized and reflected power back at the exciter is minimized. It would be ideal if the two occurred at the same exact spot in the tuning settings — but my experience is that this never happens. A compromise between the two is always in order, but I always give precedence to grid drive.

Another common problem is associated with the warm-up period of the tube. It presents a different impedance to the output of the matching network as it warms up, finally settling in after a certain amount of time. While this is happening, the matching network presents a different load impedance to the driving source.

You will likely be tuning up this input matching circuit after the tube has warmed up. Just know that after it cools all the way down that it might not be presenting such a good load to the exciter. If it is off too far, the transmitter might not come up at all, at worst, or it may come up slowly, as the tube warms up.

This is obviously not a good situation, considering you want to use this transmitter as a backup, right? To get around this, you can do one of two things: Either find a compromise setting between "cold start" and "warmed up" or preferably, buy an RF isolator, with enough power handling capability, and insert it between the exciter output and the transmitter input.

Installation of the isolator provides the added advantage of being able to completely emphasize the peaking of the grid current, since you'll not have to worry nearly as much about reflected power back at the exciter.

Next month: the amplifier output side.

Irwin is RF engineer/project manager for Clear Channel Los Angeles. Contact him at doug@dougirwin.net.

APT SureStreamer Won't Hiccup

by Aaron Read

Transmitting audio over the Internet is a reality for almost every radio station these days, and Rhode Island Public Radio is no exception.

In fact, we use it even for mission-critical purposes like getting our main programming audio from our studios to our transmitters in Seekonk — WELH(FM) — and South County — WRNI(FM). But while audio-over-the Internet is so ubiquitous that everyone takes it for granted, it's important to remember that, strictly speaking, it's a terrible idea and the Internet was never designed for it.

That's where nifty devices like the new APT SureStreamer come in.

By "never designed," I mean that the core design characteristic of the Internet is resiliency. If a nuclear war happens and half the network that made up ARPANET (the military precursor to the Internet) was destroyed, the packet-based nature of the data flowing across it would keep things working. And use of TCP meant the far end would recognize any missing packets and ask for them to be resent. This is great if you're downloading a file or sending an email, but it can be problematic for streaming media, like audio or video.

STREAMING SOLUTIONS

There are several solutions to this thorny problem. First is that instead of TCP/IP, we use User Datagram Protocol. UDP doesn't have a bi-directional nature. Lost a packet? Too bad! That packet is dead to us! Leave him behind! Without that bi-directional nature, you have far, far less overhead and processing time needed. UDP works fine even under far poorer network conditions.

You can also use some clever tricks like Forward Error Correction and Reliable UDP. Both add a little more delay (but not so much that it's a problem), and they make the overall system much better at successfully getting packets from Point A to Point B.

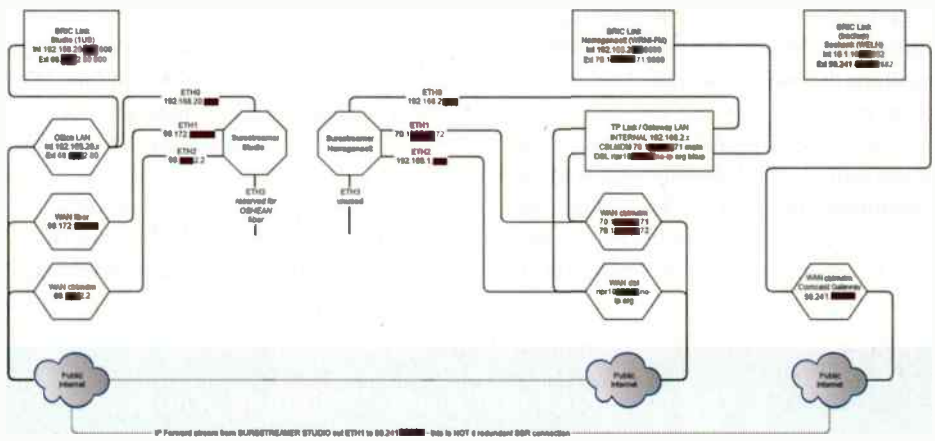


Fig. 1: Network diagram of the RIPR SureStreamer system.

You could also throw money at the problem by investing in more reliable infrastructure, like fiber-optic Internet delivery; and once you've got that in place, you can make use of Quality of Service technologies, like MPLS, which direct the network to give your packets a certain priority from end to end, thus better ensuring they arrive on-time and in-order. These technologies are quite expensive, though, often costing $1,000 to $4,000/month. Although, for that much money, you tend to get much better customer service during a problem!

Even with a pricey fiber line, things can still go wrong — very wrong sometimes. A tree can fall on a telephone pole and snap all the wires. A construction worker can dig right through the wrong conduit and destroy your fiber line. Even the best service contract in the world isn't going to change the fact that it'll be hours, if not days, before that cable can be repaired.

Enter the APT SureStreamer. Instead of relying on any one Internet connection, and trying to make that connection as reliable as it can be, the SureStreamer aggregates up to four separate Internet connections, all simultaneously. Put a SureStreamer at either end, plug your codec (in our case, Comrex BRIC Links) into the local LAN, and the local LAN into the SureStreamer. Then connect as many Internet Service Provider connections as you want into the remaining three Ethernet jacks. You'll want at least two, although unless the third follows a physically redundant path (e.g. the same backhoe can't "fade" all your ISP links at once) then two is all you really need.

Lost a bunch of packets on the fiber line? No big deal — exact copies of those packets made it just fine over the cable-modem. The audio didn't even hiccup!

In late July 2015, RIPR acquired the first

Worldcast APT SureStreamer

two APT SureStreamers in the entire USA for demo units/beta testing. It took a lot of fiddling but we eventually got them set up properly, thanks to APT's help. Once the SureStreamers are connected to each other, you just configure your audio codec to send a Realtime Transport Protocol stream to the local SureStream and *voila*! It comes out the other end for your codec to accept.

On the studio end, we have a Cox fiber-optic Internet connection, and a Cox Business cable modem. On the transmitter end, there's another Cox Business cable modem and a Verizon DSL line. This is important because cable modems don't work when power is out in the area; even if you've got a generator on-site (as we do) the cable modems need power on the poles for the equipment to function. DSL does not; as long as there's power at the local central office the DSL will function. Similarly, even though fiber optics are usually quite reliable, they do need to be taken down sometimes for maintenance. (See Fig. 1, our network diagram.)

The important thing to remember is this: If you lose connectivity on either IP link, the SureStreamers don't care. They just chug right along, and the audio comes across the BRIC Links with nary a hiccup. APT's own tests show zero or near-zero actual packet lost over months and even years of testing, because even though each connection will lose some packets here and there, they almost never lose the same packets at the same time.

I've verified this myself by simply unplugging the network from one of the ISP's while watching the network statistics page on BRIC Link. You can see a screen capture of it, above in Fig. 2.

The row of blue columns in the middle shows the last 60 seconds of network performance in terms of target delay, jitter, and actual delay. Below that is a row that's currently black. Black is good — if there were frame loss (dropped packets) you'd see red columns, and in fact if you look closely right above the "T-30 Seconds" in the middle, you'll see a tiny smidge of red, indicating a slight frame loss, not enough to cause an audio outage or distortion. It's hard to see, but flush with the bottom is a tan/brown line indicating the frame loss at the far end of the connection is also at 0 percent.

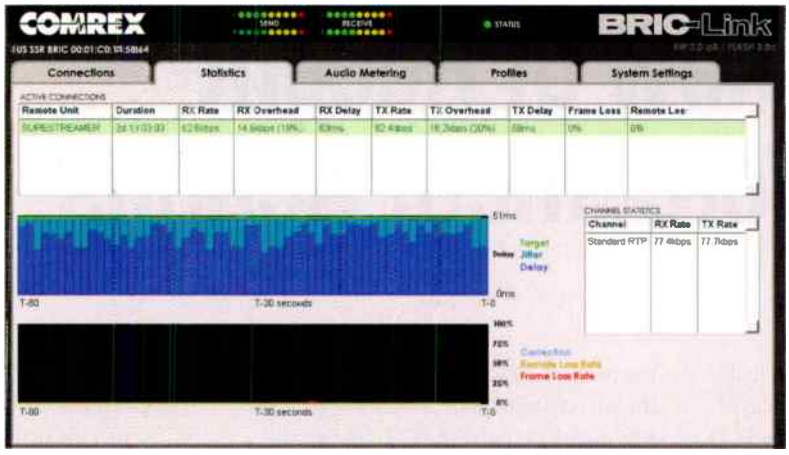


Fig. 2: Network Connection Statistic page of the BRIC Link at 1 Union Station.

So to summarize, RIPR now has a system that can deliver better reliability than an expensive fiber optic line at the Narragansett tower, and is using a mere cable modem and DSL line. Instead of spending $1,500/month we're spending more like $200/month for both connections.

Technically, we are still vulnerable, since the DSL and cable modem both come via telephone poles along Point Judith Road and Westmoreland Road to the tower. A falling tree could wreck our whole day, although that's always been a concern with WRNI(FM) ... even when we had a Verizon PAS circuit for our STL (Verizon discontinued the service in March 2015).

At the moment the software does not support any 3G/4G cellphone Internet modems for use via the USB ports, but it is planned for a future firmware release. I'm keeping my fingers crossed for some time in 2016.

Read is the director of IT and engineering for Rhode Island Public Radio.

Review of Recent Enforcement Actions

by Lee Petro

While the rest of Washington had cleared out during the holidays, the enforcement elves at the FCC were busy at work. Two recent actions are noteworthy because they touch on the need for full disclosure when filing an application, and the need for broadcast licensees to fully disclose the sponsor of advertising.

First, the Enforcement Bureau entered into a Consent Decree with Cumulus Radio for $540,000 in connection with advertisements found to be lacking full disclosure of the sponsor. Specifically, the licensee had run advertisements regarding the Northern Pass electric project in New Hampshire on a station during a five-month period. The bureau took issue with the advertisements because, while they mentioned some variation of the Northern Pass Project, they did not specifically state that the electric company — Northern Pass Transmission, LLC — had placed the ads.

The bureau noted that the FCC's sponsorship identification rules require disclosure whenever "money, service or other valuable consideration" is paid or promised to the broadcast station. It determined that the reference to just the name of the project did not adequately identify the party placing the advertisements. Because the advertisement ran 178 times during the five-month period, the FCC imposed the penalty and also required Cumulus to develop a company-wide compliance program.

Next, the Media Bureau issued a Notice of Apparent Liability in the amount of $9,000 to a licensee who made incorrect certifications on applications submitted to the FCC. Another broadcaster filed a petition to deny the assignment of the FM translator, alleging the seller had falsely certified construction on a tower that had been destroyed eight years earlier. The petitioner also alleged that station was constructed at variance from its authorized parameters.

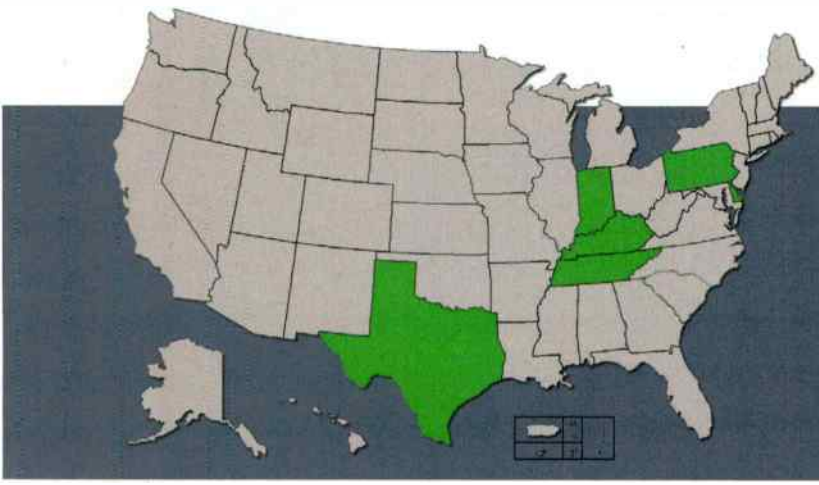
In response, the licensee confirmed that the tower site referenced in its construction permit and license authorization was incorrect. According to the licensee, the tower site was originally specified in its 2003 FM translator filing window short-form application. When it filed its application in 2013 to implement the proposal, it merely carried over the information, and provided a photograph of a nearby tower it assumed to be the one specified in the application. The licensee also confirmed that the station operated at variance, but just briefly before the station was taken silent to make further modifications.

In reviewing the matter, the Media Bureau noted that each of the applications about which the petitioner had complained were final authorizations, so the FCC would not re-open consideration of the applications. However, the bureau noted that it still could issue a forfeiture against the licensee for the false certifications contained in its license authorization during the term of its license.

Addressing the merits of the argument, the bureau found that the licensee provided incorrect information (i.e., a false certification), which warranted a $5,000 forfeiture. Because the station operated at variance from its authorized facility, the bureau added $4,000. Finally, the bureau admonished the licensee for supplying information without having a reasonable basis for believing it was accurate. The licensee will need to provide a copy of the FCC's determination with every construction permit application and license application for five years.

The takeaway is that the FCC expects licensees to take all necessary steps to comply with both the letter and the spirit of its rules. The FCC consistently expresses its lack of patience when licensees are found in noncompliance of their rules, especially when this arises from a licensee failing to make accurate disclosures.

Petro is of counsel at Drinker Biddle & Reath LLP. Email: lee.petro@dbr.com.



DATELINE

March 21, 2016 – Comments in AM Revitalization FNPRM are due.

April 1, 2016 – Broadcast Mid-Term Reports due for Radio stations with 11 or more employees in Indiana, Kentucky and Tennessee.

April 1, 2016 – Annual EEO Public File Reports placed in stations' public inspection files in Delaware, Indiana, Kentucky, Pennsylvania, Tennessee and Texas.

April 1, 2016 – Biennial Ownership Reports for non-commercial stations in Delaware, Indiana, Kentucky, Pennsylvania and Tennessee.

TIC Automates Modem Reboots With DM Engineering's Silence Sense Jr.

The Talking Information Center, located in Marshfield, Mass., was established in 1978 and has been servicing the visually-impaired and print-impaired communities for 36 years. TIC operates with the help of more than 200 volunteers in their Marshfield studio alone.

Access to the TIC service is provided by a network of radio and cable TV outlets in collaboration with five affiliates across the state of Massachusetts. TIC also streams its programming 24 hours per day, seven days per week, 365 days per year.

Frank Stas of the Talking Info Center was having a problem with a TIC Internet modem locking up and not coming back online, thereby losing audio. He would then have to drive over 40 miles to reset the power to the equipment. He reached out to Dave Mandelbaum, of DM Engineering, seeking help for this problem.

To solve this problem, Mandelbaum suggested DM Engineering's Silence Sense Jr.

The Silence Sense Jr. is a microprocessor-based system that accepts single-ended audio with a user- adjustable input level control. It features adjustable silence detect time duration (from 7 seconds up to 4 minutes maximum, jumper selectable), with switch-selectable momentary or continuous relay contact closure output. The unit automatically resets with reapplication of audio, and is provided with battery backup and an AC power adapter.

Mandelbaum suggested that TIC should use the normally-closed option of the Silence Sense Jr., with the "momentary" output option selected, connected to a Studio Solid State Relay Pack, which will switch 115VAC up to a 5 amp load. This system senses the lack of audio after a user pre-set period, and then momentarily switches off the power feed, causing the affected equipment to re-boot properly.

For Stas' application DM Engineering increased the momentary time constant from one second to four to assure a proper power reset. Now, after a modem lockup and loss of audio, this process will be happen automatically, saving Frank the time and trouble of a long trip to the transmitter site.

MORE INFO.

You can learn more about Silence Sense Jr., and other products from DM Engineering, by visiting their website at *http://www.dmengineering.com/pages/products.htm*

Cumulus San Francisco Complex Features Historic Stations, Bayview

by Mark Greenhouse

Cumulus Media Inc. is an American broadcasting company founded in 1996. Headquartered in Atlanta, it also owns Westwood One and reaches 245 million people weekly through its 567 stations in 120 cities.

The broadcasting company's San Francisco cluster is the consolidation of some very well recognized brands: FM stations KFOG (104.5) and KSAN (107.7 The Bone), NASH FM 92.3; and on the AM band 680 KNBR (on-air since 1922), 1050 KTCT, KGO 810 News, and KSFO 560.

Local management, engineering, IT and program directors all contributed to the design, which was then drafted by Ohio architectural firm SHP Leading Design. Turner Construction of Atlanta began work on the project in October 2014; high-voltage/low-voltage engineering was installed by Cupertino Electric.

IP-based from end to end, the entire infrastructure is Axia Fusion consoles with Axia Powerstations.The first broadcast aired from the complex on Feb. 24, 2015.

THE BUILDOUT

When Cumulus acquired Citadel Broadcasting Corp. in 2011, it shoehorned more stations into the existing Citadel building on San Francisco's Hawthorne Street. Due to that lease's impeding expiration, and the requirement for more space, another location was definitely needed: 750 Battery Street was selected.

Two blocks from the bay (you can view it from the top floors), the structure commands the corner of Battery and Broadway in San Francisco's North Beach area. Univision's Bay Area radio cluster was in full operation there up until mid-2014; they had taken over the space from Clear Channel in 2001, which had occupied the building since 1990.

Though many decisions needed to be made during the construction phase — like that of not gutting the entire space and outfitting the newsroom in bulletproof glass — the total time from the first walkthrough to the first delivery of equipment was only four months.

"Having the opportunity to do a buildout of a complete cluster, this facility breaks many molds for the traditional radio studio buildout," said Corporate Director of Engineering/ Telecommunications Michael Gay, who served

The KGO 810 talk studio. Custom furniture by Omnirax.

KGO 810 Newsroom featuring Axia RAQ consoles.


The KGO 810 Newsroom.

as senior systems and IT infrastructure design engineer and co-project manager for the new facilities. "The more we got into it, the more we realized we were actually building a data center that plays music on the air.

"All audio transport within the plant is facilitated via IP. All of this IP gear is connected to a central carrier class Cisco switch which handles thousands of multicast channels of uncompressed real time audio. All music is stored on hard drive and played out via IP —no sound cards in any machine. All mixing consoles are really just control surfaces for software mix engines performing the mixing of IP audio sources on an embedded hardware platform.

"Most audio cabling — analog or digital — never went more that10 feet within the plant before being converted to IP and the only audio ever leaving a studio was in the IP multicast realm. Welcome to the new world!"

PHYSICAL PLANT

The complex contains a media lab; one large performance space with control room (which is equipped with a Midas 32 console); a newsroom and six dedicated production rooms.

The broadcast studios include:

- **KFOG** — one control room
- **KSAN** — one control room, one update studio/production room
- **NASH FM** (a San Jose property run out of the cluster) — one control room
- **KNBR 680** — one control room, one talk studio, one update studio
- **1050 KTCT** — one control room, one multi-use talk/production room
- **KGO 810 News** — one control room, one

KFOG's air studio with Axia Fusion console and integrated VX controller.

talk studio, one update studio

- **KSFO 560 Talk** — one control room, and one multi-use talk/production room

CONNECTIVITY

Gay chose a Cisco Catalyst 4510 chassis-based router with line cards and supervisor cards. Though a single device, it is modular in nature, loaded with dual power supplies and gigabit PoE line cards. Given all sources and destinations the matrix is about 3000 x 3000. GPIO logic is wired in this manner as well, with pin-to-pin routing over the Axia Livewire network.

The news desks utilize Axia RAQ consoles with QOR.32 engines, each providing 32 I/O connections. RAQ has six rotary faders with OLED channel options displays, two stereo mixing buses and preview (cue) bus, a high-resolution OLED meter display with switchable VU/PPM ballistics, and monitor/headphone controls for auditioning program buses or the two assignable external monitor source selections.

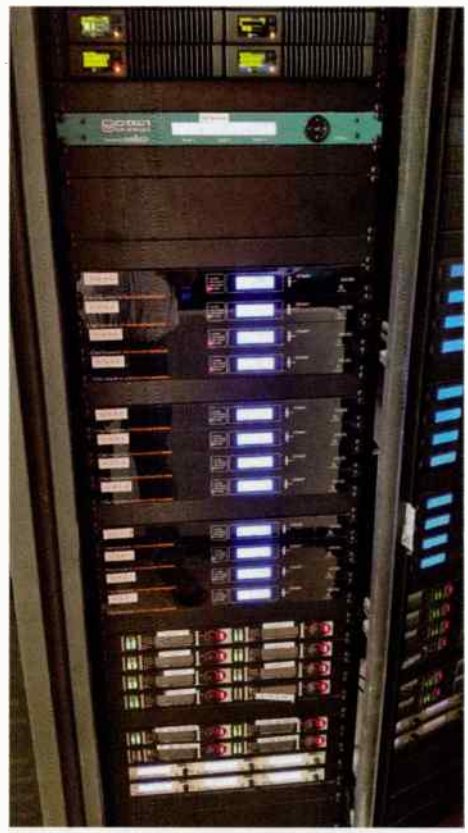
Wheatstone M2 mic processors manage the Electro Voice RE27 dynamic mics and Genelec 8020B speakers keep the audio fidelity high. Avid HD Pro Tools is the primary DAW. CD, DAT, MiniDisc, analog open reel and cassette tape decks and turntables are all available.

Much to my delight, I found a couple Eventide Ultra-Harmonizers in a control rack.

AUTOMATION

When I asked Gay what automation he had chosen for the complex, he replied "BSI's OP-X, of course!" A little digging revealed why — BSI is a wholly owned subsidiary of Cumulus Media.

OP-X adheres to the Microsoft Windows Multimedia wave specification, so the look and feel of the software is very close to Microsoft



A stack of DaySequerra monitoring receivers in the technical operations center.

Windows programs that are already familiar to operators and talent. Additionally, it works with standard PC parts and equipment.

Among the numerous modules is the File Server which keeps track of audio files — their locations, their attributes, the stations they belong to and the data transfers to and from the audio servers in the system. An audio server is the actual player for the OP-X system with one located in every station. Each audio server connects to the singular file server which serves all the studios in the cluster. This server also controls satellite receivers and other outboard audio switching devices. The file server obtains the settings for each station, device configurations, satellite clocks, audio files and program logs.

The Studio Client permits users to operate OP-X from the studio as well as control room. As the most-used module in the system, all controls are operable from both touch-screen and mouse. From here, one can create hot keys to play sound effects, jingles, weather beds, etc., customize skins (the look of the interface to personal taste).

Cumulus SFO technical operations center racks.

The Voice Track Editor allows not only the ability to pre-record voiceovers, but also access to the voice-track position in the fade-out of one tune and pre-programming the start of the next one. Even ducking volume is adjustable from this module. Remote Voice Tracking permits creation of voice tracks for a station in a remote location from your location (both equipped with OP-X) via an FTP client.

The Edit Mode module permits playlist editing while on the air.

Five additional modules — Clock Builder, Import-Merge, File Manager, Serial Device Server and Info Editor — complete the package.

COMMUNICATIONS AND TELEPHONY

There is an Axia Intercom in each studio and at every one of the 22 newsroom positions. A Mitel 3300 phone system, XDS-PRO satellite receivers, Telos VX studio phone systems, Zephyr Xstreams (ISDN codecs), Zephyr Z/IP One IP codecs and Comrex Hotline (POTS codec) broadcast codecs round out the communication support.

Finally, the primary studio-to-transmitter

NEW...

ARRAKIS SYSTEMS INC.

ARC-8 Blue

Bluetooth, wireless, Cell phone interface & Stereo streaming

- Eight channels
- Stereo Program output
- 2 mic, 4 stereo line, PC, & Phone in
- USB interface for play and record from a PC
- Mix minus in-out for an external Telephone hybrid
- BOTH balanced and unbalanced inputs and output for flexibility

The BEST of both analog and digital. With the NEW Bluetooth built in, the ARC-8 features a wireless Cell phone interface for telephone talk shows and wireless audio streaming from any Bluetooth device such as an MP3 player. The ARC-8 Blue also features a built in USB sound card to play and record in digital directly from both Windows & Apple *PCs. With BOTH balanced and unbalanced inputs and outputs, the ARC-8 works out of the box in any Radio application.

NO Hybrid required

long life switches,
conductive plastic faders,
Socketed ICs for fast on site repair.

Fast, easy installation:
- RCAs for unbalanced IO,
- Cat5s for balanced inputs,
- XLRs for mics & Program output

Bluetooth enabled !!! The #1 selling ARC-8 advanced Radio console has gone 'Blue'... Bluetooth enabled that is! This means that Channel seven on the console can be paired to any Bluetooth enabled audio device such as your Cell phone, MP3 player, MP3 recorder-editor, and more. Just pair your cell phone to the ARC-8 and answer your incoming calls with the 'Call' button and drop them with the 'Drop' button, just like a standard phone hybrid. The Caller receives the console bus mix ('minus' the caller audio) so there is no feedback. OR... you can pair any Bluetooth enabled audio device such as an Ipad, Tablet device, or MP3 player and stream full bandwidth, high quality stereo (A2DP) audio to channel seven on the console. If a paired Cell phone call comes in, then the stereo stream is dropped and the call can be answered. While an external hybrid is not needed, Arrakis does all of this while still allowing an external phone hybrid to be connected if desired. Your imagination is the only limit with this amazing console !!!

list price $1,099

*USB sound card supports USB HID compatible PCs

www.arrakis-systems.com 970.461.0730

systems are Harris Intraplex T1 shelves, with a secondary links composed of APT Horizon gear using an MPLS network.

TECHNICAL FURNITURE

One of the most important considerations of any installation, and one that gets little promotion, is the furniture that houses the equipment.

Omnirax Furniture Co. of Sausalito, Calif., continued their relationship with Cumulus. According to Cumulus Senior Vice President Engineering and IT Gary Kline, Omnirax is Cumulus' "go to" studio furniture designer and fabricator. The two organizations have completed over 20 projects together in the past six years alone.

David Holland, partner and designer at Omnirax, described the project as "a great big dance" with Kline pulling the strings from above, Gay engineering the drawings, and Vice President of Engineering Martin Stabbert being on-point, orchestrating the physical installation.

The performance space during construction.

Regional Broadcast Engineering Director Yancy McNair contributed to the furniture design and integration as well.

Holland started with a 2D plan; once approved, went to 3D, and from there everything was computer modeled one-to-one. Every piece of furniture in every room is custom, and each a one of a kind — for all studios, the newsroom and the performance studio/control room. This required that each to be fully assembled, tested, set up, broken down, documented and then staged for shipping. Once delivered, McNair saw to the reassembly and installation.

107.7 The Bone on-air studio, featuring Omnirax custom furniture.

750 Battery Street has a long and storied involvement with radio broadcasting. The first stations there were strictly analog; cart machines and analog consoles were the order of the day. Twenty-five years hence, the evolution in radio broadcast technology is clearly exemplified by Cumulus' new facility in San Francisco.

Greenhouse is an audio recording studio owner and a radio broadcast technician. He recorded the current music themes for National Public Radio's "All Things Considered."

These Phone Interfaces Play to Radio's Strengths

by Doug Irwin, CPBE AMD DRB

Telephone calls have long played an important role in radio programming. They give the medium a sense of immediacy and excitement, mainly because they can be generated and added in to a show very quickly.

I can't say they're unique to radio. Still, I don't think there's any other media that gets more out of live phone calls.

It's always best to play to a strength; and for this reason, the ability to air live phone calls, or alternatively to record and play them back later — is a very important feature in a radio studio. This isn't something you'll see with pure-play streaming services or social media — so radio needs to keep doing it well. Let's take a look at some telephone interface products.

The STAC VIP plus STAC VIP controller

AEQ is a European broadcast equipment manufacturer that has been making on-air telephone systems for over 20 years. Their third generation system, Systel, is a multi-line, multiple studio system that can make direct use of SIP trunks in addition to lines brought in from the PSTN.

A Systel phone system can be based on either the IP4 or the IP12. IP4 is a single RU device that accommodates up to four SIP trunks and includes three audio inputs, three audio outputs and a provision for the telephone set. IP12 is a two RU device, with 12 audio inputs and outputs, that shares up to 12 SIP trunks with up to four studios through simple analog or digital cabling. Both IP4 and IP12 are compatible with Asterisk PBX. Encryption algorithms include G.711, G.722, G.726 and G.729. As such, the IP4 and IP12 are N/ACIP-compatible.

The telephone is used to respond to calls in conjunction with Web browser that can be operated from any PC. AEQ has also created a customizable user interface for iPads and tablets. Multiple control locations can be configured to allow talent, engineers, and producers access to the system functions. Systel can also be configured so that certain incoming trunks show up on specific outputs (and thus console faders). Caller audio levels can be configured to be adjusted via the user-interface, or with reliance on console faders.

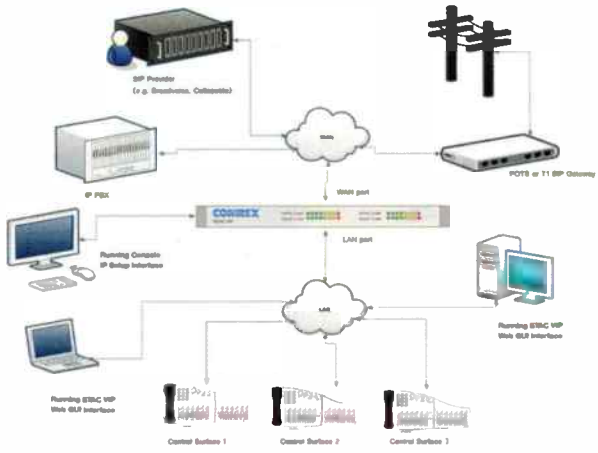
Like the other VoIP phone systems we'll cover, Systel can access PSTN lines via an FXO gateway. This is a device that terminates POTS, ISDN or ISDN PRI circuits, and converts them to SIP trunks for use by the VoIP system. Some examples of manufacturers are Cisco, Linksys and Grandstream. Naturally you'll want to use the FXO gateway recommended by whatever VoIP phone system manufacturer you chose.

Comrex is perhaps best known for its AoIP codec line, but they've been in the business of telephone interfacing for broadcast for many years. Comrex has a line of telephone hybrids in addition to STAC and STAC VIP — their on-air phone systems. STAC VIP is their latest on-air telephone interface system that can take calls from traditional POTS lines, HD Voice-capable telephones and smartphone apps, as well as those from a local PBX.

VoIP calls utilize Session Initiation Protocol to establish a connection between two callers, and SIP also negotiates which audio codec will be used for the call. When a call is placed to STAC VIP from a VoIP telephone, soft codec or even a smartphone with a SIP app, STAC VIP will negotiate to find the highest quality codec supported by the caller's phone. The VoIP calls don't require that much bandwidth; STAC VIP can process up to 12 incoming VoIP calls on a

single DSL line utilizing telephone grade G.729 audio compression.

STAC VIP accommodates POTS lines by use of VoIP Gateway devices (also known as an FXO gateway) which are commercially available from several manufacturers. In this fashion, existing



Basic STAC VIP one-line diagram

POTS/PSTN, ISDN and T1/E1 telephone lines to be converted to a SIP compatible VoIP trunk for connection to the STAC VIP Mainframe.

Numerous broadcast facilities depend on IP-based PBX systems to deliver telephone calls throughout their offices. STAC VIP accommodates extension sharing and integration with SIP-based PBX systems.

STAC IP is an integrated call screening and control interface that runs natively on STAC VIP. Using a Web browser, the password-protected graphical user interface provides basic call screening functionality, such as caller name, location and notes, and caller ID information when available.

STAC VIP utilizes the same STAC control surface which provides indication of phone line and caller status by way colored LEDs and button layout. Two features worth noting are "Busy All Lines" and control of the "Auto Attendant" feature, which answers calls and plays out a custom message before parking calls on hold. STAC CS connects to the mainframe via IP, so it can be located anywhere IP connectivity is possible.The STAC VIP Mainframe is the central point where all outside calls meet with the STAC VIP's control, management and audio interface functions. Audio connections include AES audio in and out along with Line 1 and Line 2 in and out on XLR. Two independent Hold Audio inputs are provided, allowing STAC VIP to be shared between studios. Two Ethernet ports provide for WAN connections to all telephone line sources and LAN connections for local STAC VIP Control Surfaces and local interface control. Up to four Control Surfaces can be supported per mainframe.

In addition to STAC and STAC VIP, Comrex also offers a line of stand-alone, single RU hybrids: The DH20 (single hybrid); the DH22 (dual-hybrid); and the DH30, a single-hybrid with additional audio processing features. Each of the three use digital hybrids that continuously adapt to telephone line conditions; each has AGC for caller audio, as well as ducking. Additional audio processing for the DH30 includes acoustic echo cancelling for applications where open speakers and microphones are used, and adjustable compression, downward expansion, three-band equalization and bass boost.

JK Audio offers an extensive line of telephone interface equipment for live radio and podcasting applications. Let's take a look at the Innkeeper line of hybrids: Innkeeper 1X; Innkeeper 2; and Innkeeper 4. They're all digital hybrids used to interface POTS lines.

The Innkeeper 1X has three balanced audio connections: mic/line switchable XLR input, XLR caller output and an additional, user-defined XLR output. Additionally, it has a front-panel headphone jack, and a rear-apron output meant to drive a monitor speaker directly. It has front panel controls, but it can be remotely controlled in one of two ways: The Guest Module 1

allows remote Call, Drop and DTMF dialing (by way of rear-apron terminal block remote control connections). Alternatively, the optional RIU-IP can be used for control. (This is an accessory for the Innkeeper 1X as well as Innkeeper 2 and Innkeeper 4.) RIU-IP contains a Web server which allows the user to send and receive control data through their Web browser. RIU-IP has an Ethernet connection for access, and uses RS-232 to communicate with the hybrids.

Also included as standard features are AGC; call ducking (by 9 dB when the local host speaks); and an auto-answer function that allows the Innkeeper to work as an IFB coupler in live, remote news gathering applications.

Innkeepers 2 and 4 are single rack unit devices including two and four independent hybrids respectively. Each hybrid has an independent send input, and a "master" send input,

JK Audio Innkeeper 1X digital hybrid.

that will drive each hybrid input, is included as well. (Line-level only.) Each has front panel controls but are also accessible via RS-232 and/or the RIU-IP for IP access.

The **Telos** VX is the company's flagship telephone interface and talk show system. VX connects to traditional POTS and ISDN telephone lines via standard FXO gateways; but, it also connects to VoIP-based PBX systems and SIP Trunking circuits to take advantage of Internet-delivered phone services. VX uses Ethernet as its connection backbone, likely reducing the cost of phone system installation, maintenance and cabling. By way of Ethernet, VX shares phone lines among studios and connects system components. This also makes VX naturally scalable.

VX connects directly to Axia IP-Audio networks, connecting multiple channels of audio and control via a single CAT-5 cable. Optional VX interfaces break out audio into analog and digital formats, along with GPIO logic commands. Some of its other features include:

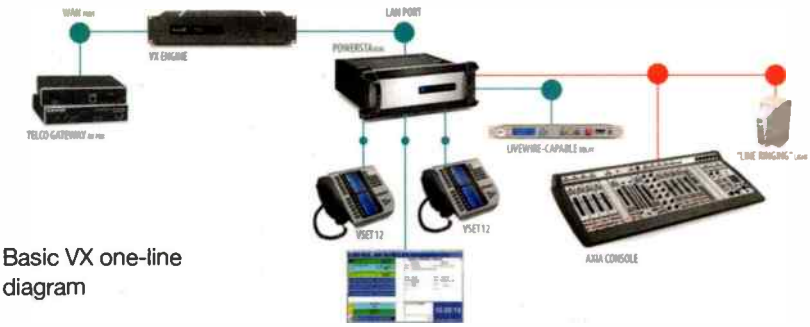
- Support for G.722 codec enables high-fidelity phone calls from SIP clients
- Modular, scalable system can be expanded to manage a network of up to 20 studios, each with a dedicated Program-On-Hold input.
- System capacity of up to 48 standard phone lines; supports up to 250 SIP numbers.
- Up to 16 hybrids, with as many as 48 active calls (up to four per hybrid), may be placed on-air concurrently.
- Using Axia AoIP, multiple phone lines — each with a dedicated hybrid — can automatically map to individual console faders.
- VX includes analog, AES/EBU and GPIO

Telos VX components: VSet6, VSet12, VX Engine

interfaces. Audio interfaces feature 48 kHz sampling rate and 24-bit A/D converters with 256x oversampling.

- Dynamic line management enables rapid re-allocation of call-in lines to studios requiring increased capacity.
- Dynamic EQ, AGC, adjustable caller ducking and send-and-receive-audio dynamics processing by Omnia.
- Acoustic echo cancellation from Fraunhofer IIS to mitigate open-speaker feedback.

The VSet12 phone controller is an IP-based phone set with two large, high-contrast color LCD panels that provide line status and caller information. (VSet6 and VSet1 are smaller versions.) Calls can be selected, held, and dropped in the way to which operators have grown accustomed. Also included is Broadcast Bionic's XScreen Lite call-screening software, which will run on any networked PC. An integrated recorder/editor allows for the recording of calls.



Basic VX one-line diagram

In addition to the VX system, Telos also offers the Hx6, a six-line Talk show system. Hx6 works with POTS or ISDN phone lines and comes equipped with two Telos digital hybrids, each with independent AGC, noise gate, and call-ducking dynamics. The hybrids include Telos' Digital Dynamic EQ, a multi-band equalizer that analyzes and adjusts caller audio. There's also echo cancellation and adaptation routines and a pitch shifter that reduce the possibility of feedback when calls are taken in open-speaker situations.

The Telos Vset12

By way of the rear apron Livewire connection, Hx6 puts audio, hybrid control and mix-minus for all six phone lines onto a CAT-5 cable. The Ethernet connection also provides access for remote call-screening applications that run on a producer's PC. Control is provided by the Telos VSet6 six-line phone controller (sold separately).

If you are simply looking for a POTS hybrid, you could consider the Telos Hx1 or Hx2 (dual-hybrid version). Both are single rack-unit digital hybrids, including the following features:

- Wide-range AGC and noise gate by Omnia, with adjustable gain settings
- Pitch shifter help prevent feedback in situations where open speakers are required
- Adjustable caller override improves performance and allows you to individualize the degree to which the announcer ducks the caller audio
- Dynamic EQ and adjustable leveler EQ High and EQ Low display meters
- Separate Send and Receive level meters
- Auto-Answer with selectable ring count
- Input/output via analog XLR (input switchable between Mic or Line levels)

RECORDING AND EDITING

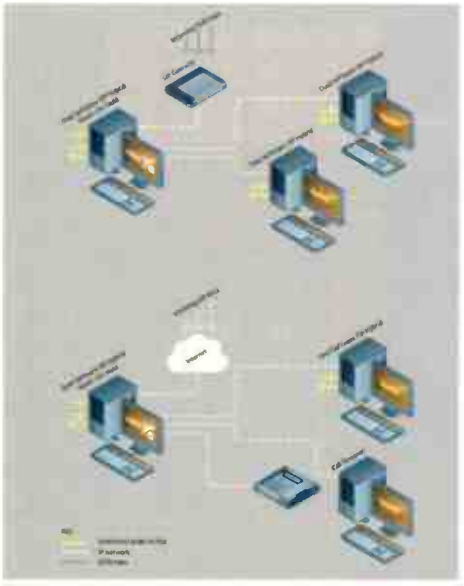
Picking up phone calls and placing them on-air live is one thing — but recording them for editing and later playback is another important aspect in the workflow of a typical radio station. Let's take a look at a couple of systems that are designed just for that.

Broadcast Bionics XScreen runs on a PC and provides a user-interface that gives jocks and call-screeners a convenient way to manage calls (and callers). For starters, there's a drop-and-drag database of callers, a phonebook and visual warnings of persistent or nuisance callers. An upgrade to the full XScreen client software (from the free version) adds even more features: An extended call history, an enhanced phonebook, prize management, and GPIO functionality.

XScreen is designed to integrate with the two

Telos multi-line systems (VX and the Hx6) that were mentioned already; and XScreen, deployed as part of a Livewire network, enables call recording, editing and console integration directly over the network.

PhoneBOX Solo is software from Broadcast Bionics that allows you to use a PC as a call routing system. The PC's soundcard is used as a dual hybrid, with EQ, dynamics, recording, playback and editing functionality. Two versions are available — up to six or up to 12 lines for a single studio, with the option of adding on up to two further positions (knows as Buddies) for a second studio and producer or call screener. This is a VoIP system, so you'll need SIP trunks coming in from your provider, or alternatively, you can



Simple application of PhoneBOX Solo

obtain an FXO gateway to convert circuits from the PSTN to SIP lines. Here are some of the more important features of PhoneBOX solo:

- **Call Log.** See who's called and when — drag and drop numbers back on to a line and call them right back.
- **Number Directory.** Store all valuable numbers; drag and drop them on to a line and call out immediately.
- **Caller rating.** Classify callers by screening them and scoring them for suitability to a particular show.
- **Call recording.** Every call taken on PhoneBOX solo is automatically, and the last five calls are saved. Talent and caller are recorded to different tracks. Use the integrated editor to prep calls for air.

VoxPro is a probably the most well-known digital audio recorder and editor for control and news room use. It has many recording and editing capabilities and provides a platform for creating on-air content in a tight timeframe. You likely know that the developer of VoxPro — Audion — was recently purchased by **Wheatstone**; but even before this development, VoxPro could be integrated in to the WheatNet-IP audio network environment.

VoxPro5 is the latest version, and it will run on Windows 7, 8 or 8.1 (32- or 64-bit). Recording and editing is done on two tracks. All popular file formats, including M4A (iTunes), MP3, MP2, WAV, AIFF and WMA files can be imported and exported individually or in groups.

An interesting evolution in the design and operation of VoxPro is represented in its networking capabilities. (Long gone are the days of sneakernet.) The networking of VoxPro5 workstations is automatic; two or more computers running VoxPro on the station's LAN will automatically find each other, swap information and connect, allowing users to access their password-protected accounts from those workstations.

However, you can configure VoxPro to operate in standalone mode, as though there were no other VoxPro workstations on your LAN; or, you can be selective in determining which remote VoxPro workstations will be allowed to share accounts with other computers running VoxPro. In this fashion, clusters of VoxPro workstations belonging to one station can be logically separated from those belonging to a different station while keeping everyone on the same LAN.

An unlimited number of users can access password-protected folders and files within a given

Voxpro control surface, and user-interface

user group in addition to the files they personally create. VoxPro users can choose any folder anywhere on the station's LAN and have VoxPro check that folder every 30 seconds for any incoming media files. If a file is detected it automatically gets "sucked-up" and into the users current folder. This is a very handy feature for doing remote broadcasts that are not quite in real time.

Phone interactivity remains a very important feature of many shows and podcasts. Getting it right — meaning that callers sound intelligible to listeners — and jocks, board-ops and show hosts use the equipment correctly and effectively — is a big part of broadcast engineer's job.

accept UDP data on port 9000.

Now at the studio we simply configure our Comrex Bric link to connect to 206.169.185.1 port 9000 or use the Dynamic DNS name (see more on that below) and port 9000. Once we configure this remote connection, the audio should flow between the units. (See Fig. 3).

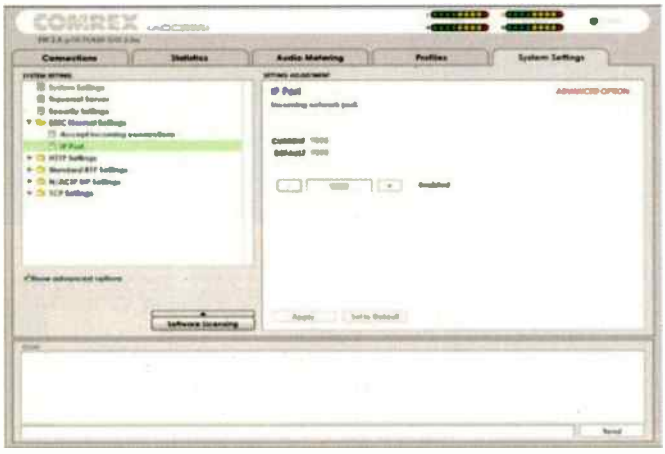


Fig. 2: Comrex BRIC system settings

DYNAMIC ADDRESSING

There are a couple of things that can happen with plain old run-of-the-mill internet access, similar to what you would get at home, that you need to be aware of.

As I already mentioned, you need to know two particular IP addresses for this configuration: The public-facing, "external" IP address, and the LAN address of the device that is the object of the port-forward. If you are not sure of what your external IP is, browse a website, such as *IPchicken.com* or *speedtest.net*, to see what external IP address has been assigned to you.

Unless you have paid extra for a business class Internet connection or static IP, you more than likely have a dynamic external IP address. This means that this external address can change at any time. Why is that a problem? Imagine if you wanted to call your friend across town and his phone number changed at random times. How would you ever be able to get ahold of him? Well the answer is simple: You would hire a service that would tell you what the number is each time you wish to call him.

This service is called dynamic DNS. If you do not have a static external IP address, then you will need to utilize a dynamic DNS service to ensure that you have remote access. Dynamic DNS does not change the configuration of the remote end; dynamic DNS has an agent that is loaded on a computer in the local network that sends the dynamic DNS service the external IP of the network every 15 minutes. Instead of accessing your remote network via the external IP address directly, you use the dynamic DNS name that you created when the service was setup. This takes care of the issues caused by the use of a non-static IP address on your external network.

Either way, you will need to know the external IP address or the dynamic DNS name to access your network remotely.

The second or internal IP address needs to have an IP address that never changes as well. Most of the networks that are setup with Linksys or Netgear routers utilize Dynamic Host Configuration Protocol. When a computer or other "host" device is first turned on it will broadcast a request for an IP address on the LAN. The DHCP server will in turn respond with an IP address for its use.

This makes deploying a network easy but causes issues with remote access, because every time the device is restarted there is a chance that this IP can change. Port forwarding requires that a static internal address be configured. Again, this is done by way of the router's management page; in the DHCP settings you will find a place to configure the DHCP range. Set the range for, say, .100 to .254. You can then set all hosts that need static addresses to have their last octet below .100.

What type of transport is going to be utilized? I have covered UDP and TCP data types in a previous article (Radio October 2015). What a person needs to know about this is simple: Most of the time it is TCP, but if you are working with audio or video then it will probably be UDP. The transport type will need to be setup when a port forward is configured.

Both Linksys and Netgear routers have a simple drop down menu selection for this. The drop down has options for TCP, UDP or both. There is no real harm in leaving it as both. When in doubt, use TCP. Most manufactures of equipment that have instructions for remotely accessing their devices will list what type of transport is needed.

TESTING YOUR REMOTE ACCESS

After a port forward has been successfully configured, you must test it. The testing is either pass or fail. It will either work or not at all. If you are having trouble with the remote access, start with troubleshooting the port forwarding information. Do I have the right port? Do I have the right transport? Almost all remote access issues arise from incorrect firewall configuration information.

Every remote access solution is different. The software used to access your remote access solution will depend on what you are trying to access. The Comrex Bric link requires that you access it with a Web browser. Solutions by Burk remote controls require the use of their Autopilot software. The point is that you will need to use the right software solution to test your remote access setup.

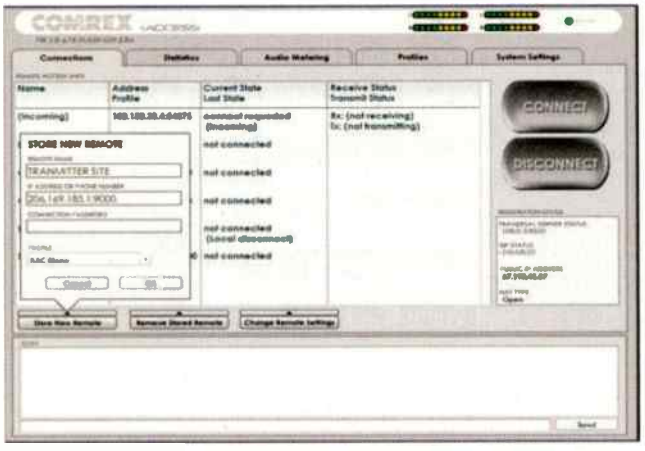


Fig. 3: Configuring a new remote connection

Make sure you read the documentation and use the right software!

Cottingham is the chief engineer of KFMK(FM) in Austin, Texas.

RF MONITOR | LBA

The FieldSense 2.0 is now available for shipping. The personal RF monitor is designed to protect personnel working near broadcast and telecommunication transmitting antennas from EMF exposure, with this new version including updated features like increased frequency range, a fall detection alarm and visible LED lights. This version features an operating range of 50 MHz-6 GHz. FieldSense 2.0 also measures both E & H fields, with six orthogonal E & H field probes; has a shaped response to measure in accordance with FCC, SC6 and INCIRP exposure limits; and a E & H field data logger, which can accessed by a USB connection.

www.lbagroup.com

AUDIO INTERFACES | STUDIO TECHNOLOGIES

Studio Technologies has upgraded its Model 46A and 47A audio interfaces, which are now available for shipping. The new versions offer improved circuity and software in a more compact, lightweight package from previous iterations.

Designed for sports and special-event broadcasting, the Model 46A provides two 2-channel party-line interfaces that provide direct connection of user belt packs and related devices. The 46A features 4-wire inputs and outputs that are directly compatible with matrix intercom systems that are standard in live-event broadcasting and production facilities. Other features include a lower audio noise floor and improved noise immunity.

The Model 47A provides dual 2- to 4-wire analog audio interfaces, but now adds power and audio on both channels of both party-line interfaces, equaling a total of four powered PL channels. It also offers an improved audio noise floor and enhanced immunity from ESD than its predecessor.

Both also now feature improved PL output currents for an additional margin for connected devices. Revised internal power supplies and a new generation of microcontroller-integrated circuitry are also part of the upgrade.

www.studio-tech.com

TOOL BAG | PLATINUM TOOLS

Platinum Tools' Big Mouth Tool Bag is now available for shipping.

Featuring six interior and exterior pockets, the tool bag is water resistant and comes with nylon stitching. It is able to be collapsed flat when not in use.

www.platinumtools.com

MONITORING SYSTEM | INOVONICS BROADCAST

It's take your pick with Inovonics new INOmini RackPack combos. Users can choose from a selection of INOmini monitoring options and combine them for a customized monitoring system. Among the variety of models that users can choose from are AES, RDS, FM, AM, HD and Internet Radio. The INOmini RackPack comes with three INOminis or two 610s that are fully mounted in the 1U Rack Shelf with Jumper Power Cables and one Power Supply.

www.inovonicsbroadcast.com

OSCILLOSCOPE | ROHDE & SCHWARZ

Rohde & Schwarz has unveiled a new handheld oscilloscope that combines five instruments in a compact, mobile form for installation and maintenance work. The system includes isolated inputs and communications interfaces; in addition it meets CAT IV standards and can carry out measurements on low voltage installation sources up to 600 V. This portable oscilloscope comes with a large format capacitive touchscreen, large buttons and a multifunction wheel for parameter adjustment. It also has R&S Scope Rider's IP51 certified housing for environmental protection. The system also features a WLAN interface. R&S will offer the Scope Rider as a four channel or two channel instrument, with the two channel version featuring a digital multimeter. The system will have bandwidths of 60 MHz, 100 MHz, 200 MHz, 350 MHz and 500 MHz.

www.rohde-schwarz.com

NEWSROOM SYSTEM | WIREREADY

WireReady has announced a number of updates to its slate of products, with the new gear offering the ability capture RSS feeds, play both mp2 and mp3 files, send RDS info and remote control automation.

Customers using the WireReady newsroom system with a support plan can now request a free capture app upgrade, which will allow them to connect with news providers who distribute via RSS feeds. The capture engine pulls in the story text and any linked audio files into the system automatically.

The company has also announced an update to WireReady V12 and up, now allowing them to play mp2 and mp3 files, in addition to WAV files. Customers with current support contracts may download software upgrades on the company's website.

The RDSReady system is now available as well. The system takes metadata feeds from automation systems, reformats them, merges them, switch between satellite and live automated feeds, and distribute multiple outgoing formats for all devices and service providers. The system can take and send data via file/LAN, TCP/UDP, RS-232/422, and also pass data via URL.

In addition, ControlReady V12 is now available and allows a user to create up to 1,000 alpha numeric-based actions that can be triggered by a phone or other devices.

www.wireready.com

HEADPHONES | FOSTEX

A trio of new headphones from Fostex has been announced. The TR Series premium professional dynamic headphones — made up of the TR-70, TR-80 and TR-90 — offer a response range of 5Hz to 35kHz, accurate sound reproduction and plenty of audio level. The new headphones come with a new 40mm Fostex dynamic headphone driver, specially tuned housings, and slide bars.

The TR-70 has a fully open design; TR-80 has a fully-closed back; and the TR-90 has a semi-open design. Each model is available in either an 80 ohm version or a 250 ohm version.

All models come with two detachable cables, a 1/4-inch to 3.5mm adapter and both regular and thick ear pads.

www.fostex.com

This advertiser index is a service to readers. Every effort is made to ensure accuracy, but *Radio* magazine cannot assume responsibility for errors or omissions.

Of Hyperbands and Hams

by the Wandering Engineer

We don't talk often of the Hyperband or the role amateur radio plays in our profession. While many have tried, including me (it seems so easy) no one has ever adequately put words around why we often have such affection for a profession that — when laid out in black and white — really doesn't stack up to what the skill set might bring in rewards elsewhere.

Broadcast engineers often stay in the profession for a lifetime. Maybe you are the kind whose heart flutters at night when cold bridging a tower and the field intensity meter brings in your co-channel acquaintances like a ton of bricks, messing up your null. Maybe you reminisce about the days when a car radio, console radio — or your "special" DC-30 MHz radio — brought in stations from places you may never visit.

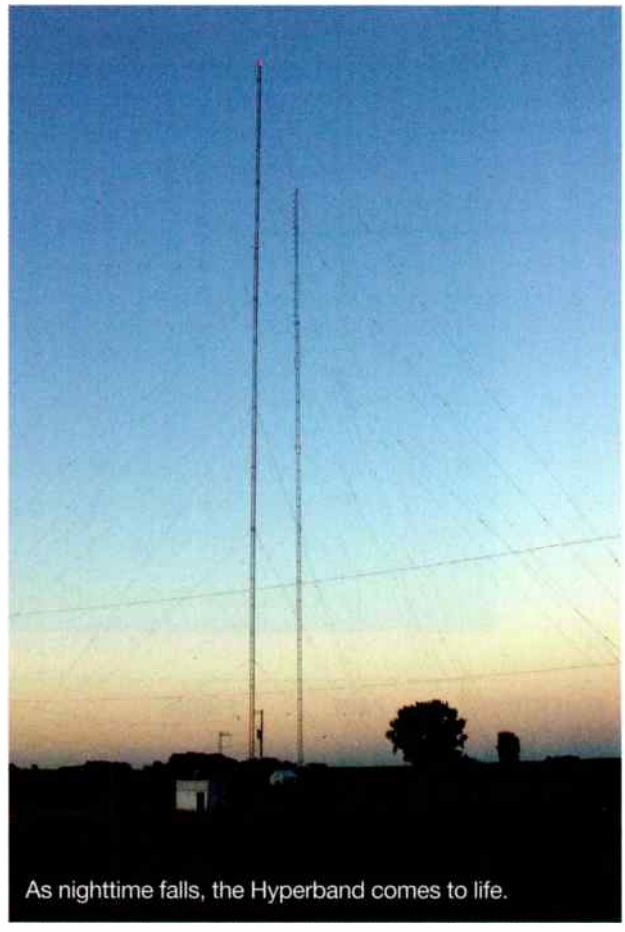
As a Ham, you may have discovered the Hyperband while playing with 160-meter receive antennas. The 1610-1700 kHz Hyperband is next door to the 1800-2000 MHz Ham band. In Ham vernacular, 160-meters is the "gentleman's band" or "top band" for all those romantic radio reasons. It's the place where a guy or gal with desire to restore a legacy AM rig can legally use it. Practically, it is Ham radio's only medium wave allocation (5W EIRP at 472-479 kHz hardly counts).

Before the AM birth control era, IBOC, narrow IFs and switching power supplies, listening to AM radio was a different experience. When 250 W was big, before there were clear channel blasters, it was really different. The Hyperband is the edge of the observable radio universe, where you can look back to that smidgeon of time after the big bang of radio where there are only a half-dozen tiny transmitters on any given frequency at night.

There are very few directional antennas or IBOC signals in the Hyperband. With a quiet antenna and a good radio in a good location you can hear a noise floor that's close to nature's — something that is impossible within 550-1600 kHz. Want a radio challenge? See if you can log each of the seven or so stations on each 10 kHz channel.

Hams are just broadcast engineers who still wonder wide-eyed at what less power than a hairdryer can do. It's easy to slip across the 100 kHz gap that separates the broadcast engineer who isn't a Ham from the Ham who isn't a broadcast engineer. If you think it's the size of the toys, keep in mind that either side of the gap it is 1 kW and the Ham actually has more antenna options and Single Side Band.

All our lives in this profession, it's been about staying ahead of the technology curve and never looking back. The first with a computer, editor, solid state rig… the first with every new toy. Every now and again, it's good to turn away from the ever steepening learning curve and look back at what made this career fun. Most of us have a radio that has become an ornament, a trophy. Some of us built "all-American-five" tube derivative kits that Heath and Allied Knight-Kit sold. Others built or bought solid state versions of the same when we wanted to know all about electrons and hear what all was out there. Every now and again, blow out the dust, replace a cap or tube and light her up.



As nighttime falls, the Hyperband comes to life.

Scott Fybush

It used to be all about "arts and education." Our current concentration on STEM — science, technology, engineering and math — isn't a bad thing, but at the expense of art, it's a more barren life and career. As I stand out here on a crisp cold night, the Milky Way and the meter light on my FIM all that can be seen, I've decided I don't feel at all guilty or embarrassed about slipping away from taking that monitoring point reading into the Hyperband where a thousand watts a thousand miles away bounces the meter wildly.

There is still radio magic out there.

The Wandering Engineer is an industry stalwart who has been in broadcasting since the days of Marconi and Tesla. He gives his thoughts on the current state of broadcast engineering and the broadcast engineer.

NEW for NAB 2016

Simple-IP

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- 8 in, 8 out mono (4 in, 4 out stereo)
- Analog or AES3 digital models
- Not an IP Console
- Works with your current studios
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- Uses off the shelf Switches
- 1RU rack mount cabinet
- Super high performance !

Simple-IP is a one rack unit AoIP box with 8 mono inputs and 8 outputs (or 4 by 4 stereo pairs). Available in 2 models: analog & AES digital. Uses standard off the shelf ethernet 'switches' for connectivity. And uses the world standard DANTE ethernet AoIP protocol.

Simple-IP is plug and play with CAT5e wiring to all balanced Arrakis ARC or MARC series analog consoles, and ideal for Arrakis X-mixer digital consoles.

Simple-IP solves wiring problems in old and new facilities alike.

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Using the world standard DANTE ethernet audio protocol, Arrakis has built the first cost effective ethernet audio networking solution for the average radio broadcaster (and professional AV installer). It replaces the multipair cables and punch blocks that link radio studios with a single CAT5e ethernet cable. The base product, with 4 stereo inputs and 4 stereo outputs (analog & digital models), is the perfect size for the average studio and costs only $999 list price. Put one in each studio and one or more in your equipment room, then run CAT5e ethernet cable and your station's audio routing is DONE! Need more ins/outs now or later, then just add another box in the studio without adding any new wiring. Need to change what audio is connected to where? No more punch blocks to play with. Just a few mouse clicks on the easy to use 'DANTE Controller' software and you are done.

www.arrakis-systems.com

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