Radio Guide

Radio's Technology Forum

September 1995

The System Works!

Radio Guide Toll-Free BBS

Our toll-free BBS is officially on-line now. In last month's issue, you may remember the problems I had getting the toll-free number installed. A few days ago I received a bill for the first month's service. No phone call, no notification that it was connected — just a bill! So, I called the number on the invoice and, sure enough, the 800 number was functional. Not willing to leave it alone, I called the business center, and their reply was, "We told you it would be installed within 60 days."

Yesterday, I discovered the BBS telco loop to the C.O. had been "cut" — no dial tone, no voltage. Is it just me, or does it seem like when they install one service, they take one away; something like a zero-sum process. I'm going to publish the new number anyway, since I have full confidence that the line will be repaired by the time you read this.

**The new toll-free BBS number is:
800-811-4045**

The BBS is set up to receive **Radio Shopper** classified ads, **Radio Guide** tech tips and articles, and subscription requests for both publications.

You'll also find a Library of technical and useful information. If you need a hard-to-find part or schematic, or just have a technical problem that needs solving, we will post it in the Library.

A Broadcast Internet Service

**Our new Internet e-mail address is:
Radio@broadcast.net**

We are involved in a new Internet project, and you can read more about it on page 16. As this service grows, we will continue to keep you informed on how to access, and make use of, this new broadcast Internet service. There's nothing like it, and it promises to be something special. We'll keep you posted.

Here's How to Reach Us . . .

**INTERNET: Radio@broadcast.net
COMPUSERVE: 71203,2341
TOLL-FREE BBS: 800-811-4045
TELEPHONE: 507-280-9668
24-HOUR FAX: 507-280-9143**

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Audio Polarity Testing

Radio Guide

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Radio Guide Publication

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The Radio Forum

By Ray Topp — Editor/Publisher

New EBS/EAS Test Script

Here is the text of the new, shortened EBS/EAS Script. It is provided courtesy of ***SBE Chapter 17 of Minnesota****, and was supplied to them by Don Hepplemann, Minnesota EBS/EAS coordinator . . . editor*

Revised Weekly Test Script For EBS/EAS System Tests

1. Discontinue normal broadcasting.

2. Broadcast this announcement:

"The following is a test of the Emergency Broadcast System."

3. Transmit the attention signal.

Broadcast the attention signal for 8 seconds as specified in Section 11.32 of the new EAS rules (47 CFR 11.32).

4. Broadcast this announcement:

" This station is testing its emergency Broadcast System equipment. The EBS will soon be replaced with the Emergency Alert System. The EAS will provide timely emergency warnings. This station serves the [insert EBS/EAS Local Area Name] area. This concludes this Emergency Broadcast System Test.

5. Resume regular programming.

If a longer test is desired, lengthen either the transmission time of the attention signal and/or use the script below in place of the script in step 4.

"This station is testing its Emergency Broadcast System equipment. The EBS will soon be replaced with the Emergency Alert System. The EAS will provide timely emergency warnings. If this had been an actual emergency, such as [insert the types of emergencies likely to occur in the station's coverage area], the attention signal you just heard would have been followed by an official warning or alert information. This station serves the [insert EBS/EAS Local Area name] area. This concludes this Emergency Broadcast System test."

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CCA/CSI Xmtr Tips

Mark Pallock — MBE Enterprises
[818-882-9475]

For everyone who has an older model CCA FM transmitter (D Series), or 12kW and higher power CSI transmitter, that uses 3-inch transmission line, check the bullet support bracket at the very top of the PA box (it supports the bullet that leaves the top of the transmitter). There should be at least 1 inch of Teflon showing between the bullet and the end of the bracket. If not, you should either remove the bracket or get a replacement unit.

This is important because the impedance at this point changes and is not 50 ohms, and it *must* be 50 ohms for the transmitter to make power, especially in the "D" series of transmitters. In the CSI, the wrong bracket could cause arcing due to circulating currents in the PA box.

In the "D" series of transmitters, removing this bracket is no small feat, because in many instances the top of the transmitter has to be removed and the bolts that hold the bracket in place also have to be removed.

Harris FM25K Tips

Bruce Musso — WROQ, Greenvile, SC
[803-242-0101]

Clean Meters

When trying to clean the meters on a Harris 25K transmitter, most of the meters are easy to reach — except the multimeter in the center of the transmitter. To clean this one, you will need to open the exciter hatch on the front so you can reach this meter. After opening the hatch, find a small piece of banding strap (like they use on lumber). Wrap a piece of paper towel around it and tape itl.

Lightly spray it with some glass cleaner and gently insert it between the meter glass and the glass mounted to the front of the transmitter. I've done this within the past weeks and it works very well, without tearing the whole transmitter apart to do it.

IPA Power

If you are having a problem with the IPA forward/reflected power, try changing the 2.5K, 50 watt screen resistor, located in the back of the main cabinet. Call the Harris parts department and get two (2) 5K, 50 watt resistors, and parallel them together.

This will give you 2.5K at 100 watts, instead of the single 2.5K at 50 watts. This should help, if you are experiencing more reflected power that you would like.

STL Path Analysis

Jim Somich — Radio Engineering Services, Broadview Heights, Ohio [216-546-0967]

This is the first of a series of columns written to address the challenge of being a radio broadcast engineer in the turbulent present and the uncertain future. With the demise of station engineering departments, we independent engineers have taken up the slack and met the challenge of keeping our stations on the air and sounding good. It is a tribute to the inventiveness and creativity of most contract engineers that the broadcast industry has made a relatively smooth transition from in-house to contract engineering.

The purpose of this column, and those to follow, is to provide an avenue of communication between us engineers: the unsung heroes of the broadcasting industry. This column will be as much yours as it is mine. I hope it can become a practical, interactive forum that is long overdue.

My forte is practical, nuts-and-bolts engineering. Let's leave the information superhighway and cyberspace to others. I want to make this column a source of ideas and inspiration for all of you. Whether we are discussing a new product that can make our lives easier, or a quick fix for a chronic problem, my mission is to make your job a little easier and fun. Feel free to call me anytime at 216-546-0967. I will be glad to discuss anything that is on your mind.

STL Construction

This month we will discuss the design of microwave links. A high reliability link for STL or intercity relay purposes requires careful planning. If you never had the opportunity to build such a link, you should find what follows to be of interest. Moseley and Marti will perform path surveys for equipment purchasers at no cost, but it is still important to know how to do it for yourself.

Topo Maps

The first step in a path survey is to acquire 7.5 minute series topographic maps covering both ends of your hop and all of the terrain in between. These maps are available from many local map dealers or from the U.S. Geological Survey, Washington D.C. 20242. An index is available that will help you determine the maps necessary for your project. Each quadrangle (map) is identified my name, i.e. Ripley Quadrangle.

You will need some earth curvature paper for plotting your path from the maps. 1.33X earth radius paper is available from K&E at most drafting shops, or you can use the past page of the 1995 Marti Catalog (Marti can be reached at 817-645-9163.) and run off as many copies as you need on your copier.

Lay Out Your Path

Lay out the maps on a large smooth surface. If more than one map is required, trim the edges so that you can butt them tightly against each other. There is no overlap between adjacent quadrangles. Tape the maps together at the junction, so they cannot slip.

Mark both ends of your path on the maps using landmarks and roads. If you know the coordinates of both sites, this will make it easier to find your spots. Coordinates are printed along the sides of the topos.

Draw a straight line between your marked locations. From the length of this line, you can determine your path length from the topo scale.

Plot Your Path

Now you will need the earth curvature paper to plot your path. The center of the earth curvature paper (0 miles) is halfway point of your path. For example, if your path is exactly 10 miles long, to 0 mile marker is 5 miles from either end of your path. Mark your transmitting and receiving points on the paper by measuring out from the center.

The vertical designations on the curvature paper are in feet above mean sea level (AMSL). This corresponds with the elevation marking on the topo maps. Brown lines on the topo maps connect areas of the same AMSL. At each end of your path, add your antenna height to the elevation AMSL and plot it on the curvature paper. Draw a straight line between the two antennas to represent the path.

The bottom line on the curvature paper represents the earth's surface. On long paths with low antennas, the earth itself can block your path. In these cases, it is necessary to increase antenna height. You can also calculate the distance to the radio horizon by the formula:

$$D = 1.23 \left(\sqrt{Ht} + \sqrt{Hr} \right)$$

Where: **D is the distance to the horizon, in miles.**
Ht is the height of the transmitting antenna.
Hr is the height of the receiving antenna.

If the length of your path is greater than D, you must raise the height of one or both antennas to clear the horizon.

Now, follow the line of your path on your topo maps. Place a hash mark at each point where you cross a major elevation line (one that is marked in AMSL).Then, using the map scale, make the mileage to each hash mark. Transfer this information to the earth curvature paper and connect the dots.

(continued on page 5)

STL Path Analysis

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You now have an accurate representation of the terrain that exists between your two points. It is normal to allow another 50 feet or so for trees. Also, if there is a tall building in your path, it may not show on the topo. If you suspect a man-made obstruction, get out your binoculars on a clear day, and try to sight the path.

Fresnel Clearance

In addition to obstacle clearance for your microwave path, you must also allow for Fresnel clearance. Microwave energy beamed between two points actually takes an infinite number of paths, depending on atmospheric and weather conditions, as well as terrain. Refraction may result in either upward or downward directions. Energy arriving over the various paths is categorized into Fresnel Zones. These zones are numbered to correspond with circles of different radii centered on the direct line between antennas. The zone with the smallest radius is the first Fresnel zone. This is the only zone you must take into consideration when designing a microwave path. A value of 0.6 times the radius of the first Fresnel zone normally is taken as the absolute minimum clearance.

The radius of the first Fresnel zone (at the point of a major obstruction in the path) may be calculated from the equation:

$$R = 72\sqrt{AB/Pf}$$

Where: **R** is the radius in feet.

- **A** is distance from one end of the path to the point of obstruction, in miles.
- **B** is the distance from the other end of the path to the point of obstruction, in miles.
- **P** is the total path length, in miles.
- **f** is the frequency in GHz.

Multiply R by 0.6, and plot this on your earth curvature paper. In most cases, you will only be concerned about Fresnel clearance below your beam, but remember that Fresnel zones exist on the top and sides of your path also. For example, if you are trying to shoot between two buildings, the first Fresnel zone to the side could be a problem.

At this point, you will know if you have a good path or not. If you are in trouble, the only way out may be to increase the height of your antennas.

The Link Budget

After you have plotted and approved your path, the last step is to prepare your link budget. This will tell you if you have enough gain in your system to provide full receiver quieting. Let's look at a typical link budget:

Transmitter Power Output = 38.5 dBm
Transmitter Antenna Gain = 19.65 dBi
Receiver Antenna Gain = 19.65 dBi
Total Gains = 77.8 dBm
(Gain figures obtained from manufacture's literature.)

Path Distance = 8.77 miles
Frequency = 950.0 MHz
Path Loss = 115.01 dB
Free Space path loss, caused by the spreading of the microwave beam, can be calculated by the formula:
Free Space Loss (dB) = 36.6 + 20log F + 20log D
(Where D is in miles and F is in MHz)

Transmitter Line Loss = 2.1 dB
Receiver Line Loss = 2.43 dB
Connector Loss = 1.0 dB
(Allow 0.5 dB per connector.)
Other loss = 0.0 dB
(Diplexers, filters, multiplexers, etc.)

Total Loss = 120.54 dBm
Received Signal = -42.74 dBm
(Subtract total loss from total gain.)
Signal for 60 dB quieting of receiver = **-66.0 dBm**
(Get from STL manufacturer's Literature.)
Fade Margin = 23.26 dB

Reliability

An STL must be 99.99% reliable. This means that out of a 24-hour program day (86,400 seconds) the signal could deteriorate to something less than normal program quality for a period of 9 seconds. This 9-second period would not necessarily occur all at one time, but could be accumulative over a 24-hour period. It's interesting to note that a reduction to a 99.9% reliability factor would allow some 87 seconds in the same program day to deteriorate to less than standard, which is unsatisfactory for STL use.

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STL Path Analysis

(continued from page 5)

You can calculate the reliability by the formula:

$$a \times b \times 2.5 \times 10^{-6} \times f \times D^3 \times 10^{-0.1F}$$

Where: a = 4, for very smooth terrain, including over water.
a = 1, for average terrain with some roughness.
a = 0.25, for mountainous, very rough or dry regions.

b = 0.5, for Gulf Coast or similar low, humid regions.
b = 0.25, normal, interior, temperate/northern regions.
b = 0.125, for mountainous or very dry regions.

f = Frequency in GHz

D = Path Length in miles

F = Fade margin to minimum acceptable point, in dB.

For example, the 23.26 dB fade margin in our example, at 950 MHz (0.95 GHz), over average terrain of 8.77 miles, in an area of normal humidity = 0.0000188 = 0.0018% = 99.998% reliability.

This meets our requirement for STL path reliability.

It is important to realize that this formula does not take rainfall into consideration. In area where heavy rainfall is normal, this figure will be degraded somewhat.

Obtain the Required Forms

Now you are ready to apply for your new link. You can obtain forms from the FCC by calling 800-418-3676. You will get a recorded message asking you to leave specific information, including the forms you want to order and your name and mailing address. You will normally receive your forms within ten days. Certain forms are also available on the FCC Form Fax-Back service at 202-418-0177. An index of available publications can be faxed to you by request from the 800 number. You must know the code for forms you want faxed to you; these are listed in the index.

If this is your first STL, you will need to request three publications: Form 313 itself, the instruction pamphlet for form 313, and the private radio bureau filing guide. I suggest you request a dozen or so 313's and keep them on file. Form 313 is used to apply for a license for a studio transmitter link (STL) or a remote pickup link (RPU). 313 will ask for your transmitter output power, antenna input power and effective radiated power. This can be calculated from your path study and from manufacturer's literature.

FCC forms can be confusing, but my rule-of-thumb is to answer every question with as much information as possible and never used the words "on file." Err on the side of giving too much information. And if you really get stumped, you can always call your FCC regional office.

A Great Little Video/FM Composite DA Amplifier

Frank Berry — St. Petersburg, Florida [813-577-0041]

I originally designed this DA to provide additional video outputs from WTVT's "Live Trucks." The design performed so well that I'm now building a bunch of them for use in the field wherever we need more video outputs. Additionally, these are excellent distribution amplifiers for FM composite signals, when it's necessary to feed the output of your stereo generator or composite STL to more than one transmitter. These distribution amplifiers can be operated from a single nine volt battery or from a "wall wart," making them extremely versatile.

The DA can be broken down into two sub-systems. The equalized amplifier/driver and the output amplifiers. I'll also explain the single ended-to-bipolar supply.

The entire amplifier is designed around the Analog Devices AD847, high speed opamp. The AD847 has a closed loop frequency response extending to 50 MHz and a very low output impedance, making it ideal for use as a high quality video or composite amplifier. It's also stable when driving very long cables with high capacitance. In this design, the frequency response extends from DC to 18 MHz.

The Input Amplifier

The input portion of the DA consists of a 10K gain potentiometer which feeds the non-inverting input of the AD847. A frequency-compensating feedback loop consists of a 330 ohm resistor from the output to the inverting input of the opamp and a 100 resistor with a parallel 47 pF fixed capacitor and a 12-100 pF variable capacitor from the inverting input to common ground. The compensating network extends the high frequency response. A fixed 100 pF capacitor can be substituted for the two capacitors shown in the schematic, since 100 pF is only slightly greater than the optimum capacitance for this stage.

The Line Drivers

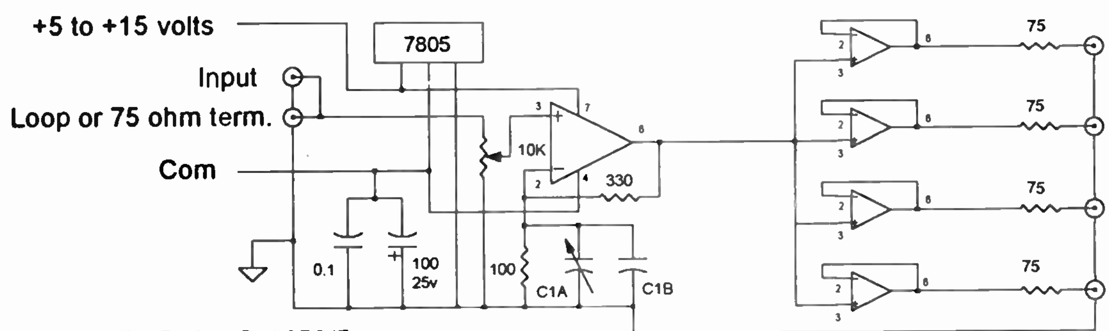
The output amplifiers are simply non-inverting, unity gain output drivers. I have put a 75 ohm resistor in series with each output. These resistors help to extend the high frequency response when feeding highly capacitive loads and will damp any signal reflections caused by non-terminated coaxial line. This damping action prevents the reflections from appearing on other outputs. It is not necessary to terminate the unused outputs.

Powering the Amplifier

Ideally, this amplifier should be powered by +/- 5 to 15 volt bipolar supply. In the absence of a true bipolar supply, you can simulate + and - voltages by using a simple three-terminal voltage regulator to provide the common ground reference, as long as both sides of the power supply are floating. If you use a three-terminal voltage regulator to provide a simulated center tap, be sure to use the output of this IC as your amplifier's chassis ground.

Construction Hints

Since this distribution amplifier is designed for a flat frequency response from DC to daylight, it's important to keep your internal wires a short as possible. It's also important to bypass the +/- power supply lines to common ground with 0.1 mFd capacitors, very close to the power pins of the input amplifier chip. Keep the feedback network as far away from the input/output lines as possible. When using this design as a composite DA, you may wish to increase the maximum output voltage swing. You can do this by changing the three-terminal voltage regulator to a 7812, and using a 24 to 30 volt outlet-mounted power supply. But in no case should you exceed +/- 18 volts on the AD847 chips.



All IC's Analog Devices Part AD847.
Adjust 10K pot for unity gain when input is terminated.
C1A is a fixed 47pF ceramic capacitor. C1B is a 12-100 pF variable capacitor, adjusted for flat HF response.
If you have no way of testing HF response, replaced C1A/B with fixed 100 pF capacitor.

Video/FM Composite DA Amplifier

Filament Fingers

John Bredesen — Chief Engineer, KLCC, Eugene OR [503-726-2224 or e-mail jab@efn.org]

In March of this year I wrote about the process of "filament voltage management." Using that technique and an obviously good tube from Econco, KLCC recently passed the 30,000 hour mark on a 4CX15,000 operating at 23.4 kW in a Continental 816R-4. The tube is rated at 6.3 filament volts and I had been able to reduce it to 5.3 and comfortably maintain the required reserve emission required to hit 105% TPO. One reason I was able to go that low, without much risk of dropping below the "knee," was because I had installed a 2 kVA Sola constant voltage transformer for the filament circuit and was thereby able to operate very close to the knee.

The "knee" to which I'm referring is that point on the filament voltage/cathode current curve where the temperature of the filament structure is just able to supply the required cathode current. If the temperature drops below that, electron deprivation occurs and you can't make TPO. If the temperature is much above the knee, cathode life is shortened.

The remainder of this tale falls into the "A Funny Thing Happened On The Way To The Forum" category.

With that as preamble, I was not too surprised when the filament voltage had to be raised about a month ago to maintain output power. With over 30,000 hours, the tube didn't owe us anything. Raising the voltage to 5.8 made everything sweet again.

Several weeks later the phone rang at 11 p.m. with that always welcomed message: "The transmitter is off the air!" We ran through some steps over the phone to no avail. 20 minutes later I entered the transmitter room to find all was well. Frank, as we've named our Continental, was happy at 98% power and no overload lights were showing.

The observed conditions, while we were off the air, had been that plate voltage was about 500 volts higher than normal and there was no plate current. The most likely suspect was exciter or driver failure, but these appeared fine when we went off the air the next day! While standing there scratching the head, up comes the transmitter just as pretty as you please. Oh yes, I should add that the filament voltage meter registered 6.0 volts, instead of the 5.8 V where I had left it, but dropped to 5.8 V when it decided to return to the air.

By this time, I suspect you've guessed that the filament was going out on the final; everything pointed to that as the immediate source of the problem. Later that night, after we signed off, I pulled the tube and saw the ugliest sight I've seen in a long time. The inner set of ring fingers (one side of the filament circuit) in the coaxial socket had practically self destructed from heating. Near the end of its useful life, a 4CX15,000 will draw upwards of 180 amps in the filament.

Apparently the resistance was just a little too high between the fingers and the tube, and the resultant heating destroyed the finger stock. With no other choice available, I cleaned the tube contact surface with ScotchBrite and bent the remaining fingers out a little bit to give greater contact pressure. I put the tube back in, and crossed my fingers. It worked! Continental was able to get the replacement contact assembly to me later that day, and I replaced it the next day while running on our back-up transmitter.

Continental said that the most common failure in that socket is the inner ring because it has the smallest surface area of the two filament contacts, resulting in the highest current density. I was cautioned NOT TO PUT THE OLD TUBE BACK IN SERVICE, after the installation of the new contact assembly, because it probably would result in the destruction of the new finger set!

When a socket fails in this manner, very small surface irregularities are left on the tube contact area which can result in a poor mechanical connection, and therefore high resistance. The destruction process can begin all over. Fortunately we had received our replacement tube to put into service.

One interesting observation was that after I cleaned the base of the old tube and had increased spring pressure of the remaining fingers, I was able to reduce filament voltage down to 5.5 while maintaining full power. It would seem that the tube "failure" which I had observed over the previous month was caused by the increased voltage drop across the socket and not because of a change of filament characteristics. Econco has told me to send the tube back to them for evaluation and if it still has adequate life, they will refinish the tube base and return it for a spare.

On page 11 of the Aug-95 issue of **Radio Guide**, I presented some incorrect information, which I need to correct. I was writing about air conditioning a transmitter suite and stated that the total heating load presented to the room would be about 40,000 watts if the exhaust were to be included. That is an error because the total input power to the transmitter is less than that! Also, the BTU conversion formula values were "transposed." The correct formula is: 1 watt-hour = 3.415 BTU, and <u>not</u> 3.415 watt-hours = 1 BTU, as shown.

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Filament Fingers

(continued from page 8)

The correct waste heat figure for a Continental running at 25 kW RF output is 13.7 kW, assuming total air recirculation. (These are figures from the Continental 816R-4 instruction book.) Using the correct formula of 1 watt-hour = 3.415 BTU, this gives us 46,786 BTUs (13,700 watts X 3.145 BTU = 46,786 BTU). Assuming 12,000 BTU per ton of cooling, we would now need approximately 4 tons of cooling.

My thanks to Paul E. Burt for calling my attention to the error. Paul is the GM of KLSU and Director of Engineering of KQXL, WYCT and WXOK, all in Louisiana. He also made me aware of another way of doing an effective job of air conditioning a transmitter room. Let me quote Paul directly.

". . . at WYCT, Clinton, LA, we continuously run 1200-1500 c.f.m. (exhaust numbers from the BE FM-30A book) of outside air into a special . . . TRANE A/C, and then positively pressurize the building with the A/C's output (the only way out is through the BE transmitter). It drops dirty, wet, 95-100 degree Louisiana air down to a dry 75-80 degrees, and all we have to do is change filters religiously, and clean the condenser coil once a year."

"Our A/C guy first told me that 'with our humidity, it'll always be frozen up.' The trick was to point out to him that we wanted whatever the local Oil and Chemical plants use to keep their process buildings cool — with their multitudes of exhaust vents — but yet still positively pressurized. We told him to find out who makes the units and ask to see if they have something [a little smaller]!"

"Well, he made some calls and found this unit, which was designed for something like 'zero air flow,' or high-back pressure . . . or something. In any case, the unit just *pours* the water out, runs all the time, and keeps my once-cranky BE rock-solid..."

I had a reminder recently about what good climbers mice can be, especially when they want to stay warm. We have an equipment rack with STL equipment housed in a small cement block building on top of a 4,000 ft mountain. One warm day about 6 weeks ago, one of the microwave transmitters failed. When I got there I found that mice had used the front part of the top cover of the failed unit as their personal latrine.

On the rear of the top cover is a series of elongated perforations positioned over the PA heat sink located inside. When I opened it up, I found the heat sink fins were packed with what looked like grass clippings, which effectively eliminated the heat sinking function. The resultant heat build-up caused the RF output module to fail.

This piece of equipment is located about 5 feet above floor level. The front of the rack was not enclosed with blank panels, although there is a close fitting door on the back. Apparently the mice simply climbed the cables up to the level they desired and set up housekeeping. Needless to say, I've put a panel in place before cold weather sets in.

ISDN For Broadcasters — Part 1

Reprinted with permission from Comrex Corporation, 65 Nonset Path, Acton, MA 01720

This information is designed to help the novice (and perhaps the not-so-novice) take the mystery out of audio on digital phone lines. At first glance, digital telephony seems pretty complex but once you get past all those acronyms, it is really very manageable.

ISDN (Integrated Services Digital Network) is the latest evolution of telephone technology. It allows total digital transmission from end to end on a dial-up service. ISDN is also a hugely complex and evolving set of standards that allows for a dazzling array of functions and features, including packet switched data, circuit switched data and voice, and a host of supplementary services. However, before your yawn turns into a nap, allow us to focus on what ISDN can do for you . . . the broadcaster.

In a nutshell, ISDN makes it possible to transmit high quality audio at amazingly low cost. And since ISDN is a dial-up service, you pay only for the time you use (at rates often no more that a regular telephone call.) In recent months, the worldwide availability of ISDN has been increasing exponentially, allowing high quality digital connections to be generated at will across town..or around the world. Although this book is designed to give a general background of ISDN with a focus on broadcast, we realize that much of the specific ordering and terminal adapter information has a definite North American slant.

Here are a few audio broadcast applications that are already becoming "mainstream."

What Can ISDN do for ME?

Digital Audio Codec Applications

What you can do with ISDN and digital audio codecs is remarkable and people are dreaming up new applications every day. This chapter will outline some basic applications but the limit to what you can do is only in your mind. For more information on an application, feel free to contact us directly at our toll free number.

Live Remotes

Live remotes are a tradition at many radio stations and can be extremely profitable, but the best way to get the audio back to the station from the car dealership or shopping mall can be variable. For places where ISDN is inexpensive and easy to get, it is an excellent option. The advantage of using ISDN is not only the outstanding quality program sent to the station but the equally high quality return feed for PA systems at the remote site.

Voice Overs

ISDN is an ideal tool for the voice over business. Voice over talent needs to be able to deliver their voice to a number of different sites quickly and inexpensively. In the past, the voice over had to be recorded in a studio and then shipped via overnight delivery to the station. With an ISDN line and codec, this is accomplished with ease. Talent can be in one part of the world and the station or recording facility in another. The sound quality is just like you were standing there.

Sports Feeds

The National Basketball Association leaguewide conversion to ISDN in 1992 was one of the first major applications of this technology. Since then, many professional and college leagues have made cooperative arrangements to install ISDN lines into each sports venue and at each radio rights holder's location. They find that ISDN not only gives them better quality, it also saves them money over other backhauling alternatives.

Audio Supplement to Educational Videoteleconferencing

Not all affordable videoteleconferencing systems provide good quality audio and some do not provide full two way audio. Imagine trying to learn to speak German, long distance without the best possible audio quality. A codec in the classroom can make all the difference, allowing all participants of a class to hear each other clearly at all times.

Weather/News Feeds

Digital audio codecs allow easy retrieval of news or weather from a news bureau or weather service. Great quality can easily and inexpensively be achieved. By using ISDN to transport the voice from the service bureau to the local station, the perception is created that the station has hired its own weather staff. The full duplex nature of the service helps here — the weathermen and the local DJ can conduct a natural conversation via the ISDN link.

Stereo Music

Do you want to broadcast a concert live from a jazz club? Transporting the audio back to the station is no problem with a ISO/MPEG Layer II digital audio codec. You can deliver 20 kHz stereo real time over ISDN.

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Remotes Using an RPU to an ISDN Site to the Studio

Wireless RPS's are a common way for broadcasters to do remotes. The difficulty is that these systems are dependent on line-of-sight transmission and often their range does not match the station coverage area. In many cases, there are not enough unused frequencies for RPU repeater sites. These problems can be solved by installing an ISDN repeater system at strategic locations. Each ISDN repeater site would have an RPU receiver hooked to an audio codec and ISDN line.

These sites can be unattended, with the station simply dialing the ISDN number closest to the remote in progress.

Remotes Using Wireless Modems

Using a portable spread-spectrum wireless modem allows greater mobility for a remote. The host could be in a moving vehicle or some distance away from the ISDN drop and broadcast the remote. How about a remote by the swan boat pond in Boston Garden? No problem!

Satellites with Codecs

Sometimes the broadcast must occur where there is no access to digital lines (or any phone lines for that matter). In this instance a good alternative is a digital audio codec with a portable satellite uplink. Then the program can be sent from the satellite downlink to the station via digital lines. A great alternative when broadcasting from Zambia (or Luxor)!

The Evolution of ISDN

Few subjects have managed to alienate the masses and to confuse the steadfast more thoroughly than digital telephony. The subject matter is not technically or theoretically difficult, but rather the technology has developed to a large extent apart from the traditional audio and broadcast disciplines. It wasn't always so, of course. The patchbays and dBms and 600 ohm loads known well by the broadcaster were developed for telephony. But like an ancient cousin, telephony has learned to speak a different language and therefore requires translation. Hence this book.

The most important lesson will be the immense impact this technology can have (and is having) on the audio industry. Audio on ISDN removes many boundaries previously constraining broadcasters and program producers. A revolution is underway:

Geography is no longer a limitation in audio programming. Audio broadcasts may be generated anywhere in the world.

The low up-front and operating costs of ISDN makes it accessible to virtually everyone.

Digital vs. Analog

In general, an analog signal is classified as having an unlimited number of quantifications i.e. as it's amplitude moves from one level to the next it takes on an infinite number of tiny discrete values. A digital signal generally has two discrete values. An analog signal is easily recognized with respect to the information it represents: The voltage across a pair of speaker wires rises with audio level. A digital signal simply toggles the two discrete values representing coded information. A light switch would be a typical digital device and a light dimmer would be a closer example of an analog device.

Analog and digital signals have their pros and cons when it comes to telephone transmission. Analog signals are simpler to distribute and, until recent advances in integrated circuit technology, less expensive. They also require less bandwidth for transmission. The main problem with analog signals is the susceptibility to noise. Every time an analog signal is amplified, transmitted or stored within the network, noise is added. This can add up in the case of long distance transmissions to the point where the signal is severely degraded. Once noise has affected an analog signal, it is difficult or impossible to remove.

Digital signals, on the other hand, take at least twice as much transmission bandwidth as their analog counterparts. They require conversion to and from the analog domain, a complex and expensive procedure. But their resistance to degradation by noise makes them the cornerstone of the modern telephone network. This is because the receiver need only differentiate between the two states which the transmitter is sending. It can reproduce this state "noise free" to the next receiver in order to continue transmission.

Digital telephone channels are not completely immune to the effects of noise. If the digital channel's noise level rises to the point where the receiver has difficulty interpreting the current state of the digital signal, errors can occur. We usually specify the quality of a digital channel in terms of its Bit Error Rate or BER. Any decent digital telephone link should produce less than one error in every million bits, or have a BER of 1x10E-6.

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ISDN For Broadcasters

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The Telephone Network

The telephone network is divided into two basic parts: the local loop (the pair of wires running from the user's premises to the nearest telephone company central office) and the rest of the network, which switches and transports voice audio through the telephone network hierarchy to the central office closest to the other end of the call. The division here is logical: technology has affected each of these two sections in a dramatically different way.

First, the local loop. This is the part which, until recently, was unaffected by technology. When you lift the handset of most telephones, signalling takes place and dial tone is delivered much the way it was thirty years ago. When you speak into the handset, the microphone converts your voice to an analog electrical signal, sent down the pair of wires to the central office. Ring signals, dial pulses, and even touch tones work in the analog domain.

The rest of the telephone network has undergone a revolution. Once just a series of mechanical cross-connect switches, it evolved into crossbar matrices and, most recently, digital carrier systems and switches. A carrier system is the way in which the telephone company transports multiple voice channels between network offices. These carrier systems are called "trunks," which are the actual "pipes" used to transport these channels.

The modern telephone network carries too much traffic for the phone company to dedicate separate wires for each call. Carrier systems allow "multiplexing" (combining multiple audio channels into a single medium -- a pair of wires or a fiber optic cable). Initially, multiplexing took place in the analog domain. Voice channels could be "stacked" on a trunk, much like radio stations across the dial. But digital multiplexing systems were soon introduced, and as their price came down they became more widely implemented than the analog types.

The most common type of digital carrier system is DS1, and the physical trunk on which it is implemented is called T1. By taking one bit at a time from each of 24 different digital voice channels, T1 can carry those voice channels along a single pair of wires (in one direction only). Two pair are required for full duplex transmission.

So the telephone company had digital trunks, but analog switches. Inevitably, switches were introduced which could pass the digital voice channels economically. Today, most telephone calls are analog only along the local loop. A digital/analog and analog/digital conversion takes place at the end of the pair of wires, before your call even touches the telephone network.

Turning Analog to Digital

Early telephone engineers determined a limited frequency band was necessary in order for voice to be understandable on the telephone network. Of course, the narrower the voiceband, the more channels that could be crammed into the carrier systems. This is why telephone calls do not sound like our CD players. Typically, the telephone channel is filtered so frequencies above about 3300 Hz are attenuated. Also, the low end of telephone audio is usually cut below 300 Hz to prevent irritating power line "hum" being induced to the channel.

Now some technical details on the conversion of a telephone channel from analog to digital. The first step of the conversion process involves taking samples of the input audio waveform at discrete times. For reasons we won't go into, it is necessary to sample the input analog audio at least 8000 times every second to preserve the 300 to 3300 Hz frequency band. This produces 8000 numbers each second representing the amplitude of the analog audio at each of those discrete times. These numbers may be of varying lengths and resolutions, so they must be rounded to the nearest fixed value allowed by the converter, which usually allows the sample to take one of 256 discrete values. This procedure is called quantization. In order to convert this number to a digital sequence, it is changed into a binary number. In binary, eight bits are required to represent 256 values, so each sample is converted into an eight bit binary number.

This process is called Pulse Code Modulation, and takes place in much the same way as in any type of digital recording device, like a DAT machine. Since the telephone channel can only transmit one bit at a time, these eight bit words must be fed serially down the telephone channel. Doing the math, sending eight bit words at a rate of 8000 per second creates a serial data stream running at 64000 bits/second, or 64 kb/s. This magic number will appear often in our discussions.

But, you may say, my CD player works at 44.1 kHz, a speed slower than that discussed here. Why is the audio quality so superior? Because 44.1 kHz is the SAMPLING rate of the CD player, not the serial data rate. In our terms, a CD quantizes two samples per period (for stereo) to 16 bits each. This creates a serial data stream of 44100 x 2 x 16 or over 1.5 million bits per second, a rate which would require 24 digital telephone channels to transmit.

The final concept is robbed bit signalling. With these data streams travelling around its network, the telephone company must have a way to monitor the status of each one. On many older carrier systems, the phone company occasionally steals the last bit of an eight bit word. By doing this, it can send slower speed status information

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along the same channel without seriously degrading the voice quality of the call. For this reason, only seven of the eight bits in each word is guaranteed to arrive on the other end of the link uncorrupted. Since 1/8 of the data becomes unreliable, the actual throughput of the telephone channel becomes 56 Kb/s. This is another magic number you will see often in our discussions. Most of the newer carrier systems, and all those outside North America, are unaffected by this limitation.

Digital Direct to the User

The vast majority of the length the data travels is purely digital. Only the "last mile" on each end requires the sophisticated conversions. While the telephone network carries 64 kb/s most of the distance, these conversions cause the data throughput to be limited to about 9.6 kb/s. Even using the best modem available today, the highest throughput would only be 28.8 kb/s-- and only on the best quality connections.

It seemed there should be a way to take advantage of the high speed data stream running almost the entire length of the connection. If the telephone company could only supply the consumer with the same bitstream it uses, instead of converting it to analog before sending it, data capacity could be increased by a factor of 6!

For several years the telephone company has provided direct access to a digital phone service. In North America it is called DDS (Dedicated Digital Service). It brings a 64 kb/s data link (56 kb/s in most locations due to robbed bit signalling) between two fixed locations. This type of service can be less expensive than dedicated analog lines or satellite channels which are utilized full time. A DDS circuit required a Data Service Unit (DSU) to allow the audio codec to interface with the digital line.

DDS is a dedicated service, i.e. both ends of the link are connected at all times. A service which may be dialed and disconnected is called a switched service. So, the logical next step was a switched DDS service. Switched 56 (SW56) became very common in the late 80s and early 90s. Technically very similar to DDS, it adds a signalling method for dialing, answering and disconnecting calls. Switched 56 is a North American "precursor" to ISDN.

Next Month:

We'll continue with, Part 2 of ISDN for Broadcasters. You can order the complete book from Comrex, free of charge, by calling them at 800-237-1776.

Digital FM Exciters: Spectral Considerations

Andy Butler — B&G Consultants, Annandale, Virginia [703-739-5474]

I've delayed writing this article several times because I don't want it to use **Radio Guide** to disseminate information that could possibly influence you in making a purchase decision that could favor a company that is my client. So, please note that I am a product consultant to Broadcast Electronics. This situation definitely influences the information that comes my way and that information filter definitely affects my perceptions and opinions. Having said that, I will also say that I try to maintain a balanced, professional attitude and I hope at least one of you will whack me if I slip.

The issue at hand is digital exciters. A number of companies have introduced such products recently and there will be a few more added before the end of the year. These boxes are generally good performers that can offer a performance boast by allowing you to go digital at your studio output and avoid the extra noise and distortion of an analogue STL. This convenience combined with good audio performance specifications, and tight modulation control due to digital level processing in the exciter, make an attractive package.

For most users the results are good, but there are a limited number of cases where a common characteristic of the first generation digital exciters is causing problems. The situation is very simple. Due to the high cost of the sampling components, none of the current batch of digital exciters generates its signal directly on frequency. They all generate the basic digital signal at some lower frequency, then either user a mixer or some other frequency multiplication to upconvert to the final carrier frequency.

Due to the fast rise times of the sampling waveforms, the initial process that creates the digital signal also generates an abundance of sub frequencies and harmonics. These are heavily filtered before the upconversion, but they are still present and the non-linearities in the upconverting process tend to preserve if not increase their number and level. Most of the digital exciters use several stages of filtering through the RF stages to bring these products within the FCC RF mask.

The spectral contents of the output waveform from these exciters is well within FCC limits. Unfortunately there are substantially higher levels of spurious signals close to the carrier frequency than most analogue exciters in common use. This won't be a problem for 95% of users. It will be of interest, but not a critical problem, for most of the rest, however it has caused several good friends some head scratching, if not direct grief. Simply being aware of the characteristic could have helped them.

I won't belabor the point by quoting examples in detail, but I will offer some general guidelines for the following cautionary situations:

Combined or Master Antenna Systems — The guys who designed these systems had to assume a spectral model for the average transmitter output. Most of them used average analogue-based boxes as the model. As a result, they paid a lot of attention to spurious products that are at harmonic frequencies, well removed from carrier, but didn't consider the rejection and reflection of hash near the desired carrier(s). Adding a device with a radically different spectral signature has caused some unexpected results. If you live in a shared system, watch carefully for combiner performance changes when you install a new exciter.

Multi-Service Shared Sites — A lot of stations are owners or tenants at a site that includes two-way, paging, cellular or other general communications services. Keeping the resulting mix of frequencies clean is often tough. The addition of close-to-carrier extras can change the overall intermodulation profile dramatically.

Nearby Sensitive Sites — Changing the spectral content of your signal may suddenly make you a problem for aircraft or other services that are located nearby in both frequency and distance.

Narrow Band IPA or Final Amplifier Stages — Most of us really don't know the bandpass of each stage in our transmitter. Adding some new elements to the spectral mix may change the performance of a stage substantially. Given these concerns, what should you do? Simply knowing about the situation is your best weapon. Try to secure the use of a good spectrum analyzer before you put your new exciter on line. Watch for funny changes in transmitter tuning as you bring it up with the new exciter. Involve the manager of a community antenna system or fellow tenants at a shared site in your checkout. They'll be a lot more cooperative if they don't have to play detective to find out what has changed.

If you do note problems, you may be able to solve them by inserting an RF pad between the exciter and the IPA. (Some of these pads are found in older transmitters, whose companies have sprouted new exciters). If that doesn't help, discuss the situation with the manufacturer. Their field service people have often sorted out similar situations.

Audio-Metrics CD-10/E

Ron Oler — Harris Allied [317-962-8596]

Removal of Transport Unit

The following is provided to allow testing of the CD player's transport, outside of the chassis, without the need for extraneous cables.

Here is the procedure:

1. Disconnect AC mains cord.
2. Remove the Top cover, via three (3) screws on each side, near the top.
3. Position the Transport units at the front of the player, just behind the Display.
4. Position the Servo Circuit Board atop the Transport unit.
5. Disconnect the large ribbon cable from the Servo Board.
6. Disconnect the two (2) red/white cables from the Display Board.
7. Disconnect the small gray cable from the Servo Board. This will be tied with the two red/white cables.
8. Locate the 3/16" round hole in the aluminum deck-plate near the front of the Transport unit and loosen the Phillips head screw below that hole.
9. Grasp the Servo Board and slide the Transport unit back about 1/4" and remove the entire Transport unit by lifting it first from the rear.
10. Place the Transport unit to the right side of the player and connect the large ribbon cable to the Servo Board, and the two red/white cables to the display — the small gray cable may remain disconnected.
11. The player may now be operated with the transport unit outside of the player.

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