

PACKET STATUS REGISTER



President's Corner

As I write this, Dayton is fast approaching (which our beloved editor Stan keeps reminding me of)! I'll try to bring you up to date with recent happenings in TAPRville.

We have just sent out the first shipment of tested Penelope boards, the latest project in the ongoing HPSDR saga. Penny, as she is called, is a 1/2 Watt HF exciter. Details are at <http://hpsdr.org/penelope.html>. This board plugs into the Atlas backplane and communicates with Janus and Ozy, the sound-card replacements that we introduced at Dayton last year. The members of the HPSDR community are busy working on Mercury, "a direct sampling front end" that will enable direct sampling of the spectrum from 0-65 MHz. There will be updates regarding these and other aspects of HPSDR at the TAPR and HPSDR forums at the

Hamvention.

Our TAPR forum will be on the first day, Friday, from 0915 to 1045 in Room 1.

We have a packed and very tight schedule as we continue to fight for enough time to present all that is new and "hot" in the digital world. Nevertheless, if we don't cover it deeply enough for you in the forum, then we'll be available at the TAPR booth to answer questions and hear your ideas for interesting things that we should get involved in.

Speaking of the booth, we will be moving – and we'll have our backs to the AMSAT booth. I'm sure that Rick and the AMSAT staff will rest easier knowing that we have their backs!

And speaking of our associates at AMSAT, we will be having our second joint banquet with AMSAT this year. It will be on Friday night, and the details will be found elsewhere in the PSR, but the important thing to note is that you MUST book in

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advance by calling Martha at the AMSAT office at 301-589-6062. Cutoff date is May 12, as she will be traveling.

Just to keep things hopping in the digital world, the D-STAR folks will be hosting a get-together on Friday night too, so I'm sure I'll be running over there after our banquet. It will be at the Drury Inn, 6616 Miller Lane, from 7:00 PM to 10:00 PM – but I hope that it will still be in full swing by the time that I get there!

As always, I welcome your e-mail with comments and suggestions, and I hope to see many of you at our booths (455 to 458).

73,

Dave, VE3GYQ/W8

Spencerville, OH

Mail to: ve3gyq@tapr.org

###

TAPR Plans for the Dayton Hamvention

By John Koster, W9DDD, w9ddd@tapr.org

The 2008 Dayton Hamvention is May 16-18, and TAPR will be present on a number of fronts.

TAPR's BOOTH SPACE

See us at Booths 455-458, behind AMSAT. The TAPR booth is located in the Ball Arena of the Hara Arena

TAPR DIGITAL FORUM

The TAPR/Digital Forum will be from 9:15 AM to 10:45 PM on Friday, May 16 in Room 1. The moderator will be David Toth, VE3GYQ.

9:30 to 9:45 AM – Update on Time Projects and OHL by John Ackermann, N8UR

9:45 to 10:05 AM – Design Considerations for an HPSDR Time Reference by John Ackermann, N8UR, and Phil Harman, VK6APH

10:05 to 10:15 AM – How to Submit a Project for Consideration by TAPR by Steve Bible, N7HPR

10:15 to 10:25 AM – Manufacturing for the HPSDR Community: An Update on Penelope

and Mercury by Steve Bible, N7HPR, and Scott Cowling, WA2DFI

10:25 to 10:35 AM – Update on Digital Voice by Mel Whitten, K0PFX

10:35 to 10:45 AM – USRP 2008 by Matt Ettus, N2MJI

2008 TAPR/AMSAT BANQUET

The second annual joint TAPR/AMSAT Banquet will be held Friday evening, May 16, 2008 at the Kohler Presidential Banquet Center, Kettering, OH.

Reservations are required and must be made by Monday, May 12, 2008.

The price for the Banquet is \$25 per person. Tickets can be picked up at the AMSAT booth at Hamvention on Friday, or at the door. Please contact Martha (martha@amsat.org) at the AMSAT office for information or call the office to make reservations. You can reach Martha from 10:00 AM to 6:00 PM EDT at (301) 589-6062, or in the US toll free at (888) 322-6728. Presidents Club Gold members should let Martha know if they are planning to attend the banquet.

6:30 PM – Doors open and Cash bar is available with Beer, Wine, Liquor and soft drinks.

7:15 PM – Buffet Dinner service begins. The Center has a reputation for good food and service. The Banquet will be in the Lincoln or Kennedy room.

Menu

Fresh mixed green salad with assorted dressings

Roast Prime Rib of Beef au jus - carved on site

Marinated Roasted Garlic Rosemary Chicken Breast in lemon butter sauce

Salmon with Newburg sauce

Whipped potatoes

Normandy blended green beans

Fresh fruit bowl

Roll and butter

Coffee/ iced tea/ water

Assorted pies

###

2008 ARRL/TAPR DCC: Call for Papers

Technical papers are solicited for presentation at the 27th annual ARRL/TAPR (<http://www.tapr.org/>) Digital Communications Conference (DCC), Friday-Sunday, September 26-28, in Chicago, Illinois (<http://www.tapr.org/dcc.html>). Papers will also be published in the Conference Proceedings. Authors do not need to attend the conference to have their papers included in the Proceedings. The submission deadline is July 31.

The ARRL/TAPR Digital Communications Conference is an international forum for technically minded radio amateurs to meet and present new ideas and techniques. Paper/presentation topic areas include – but are not limited to – software defined radio (SDR), digital voice, digital satellite communication, digital signal processing (DSP), HF digital modes, adapting IEEE 802.11 systems for Amateur Radio, Global Positioning System (GPS), Automatic Position Reporting System (APRS), Linux in Amateur Radio, AX.25 updates and Internet operability with Amateur Radio networks.

Submit papers to Maty Weinberg, KB1EIB, ARRL, 225 Main St, Newington, CT 06111 or via e-mail (maty@arrl.org). Papers will be published exactly as submitted, and authors will retain all rights. ARRL will provide additional information on the 2008 DCC as it becomes available.

###

Chicago – “Your Kind of Town”
for the 2008 ARRL/TAPR Digital
Communications Conference

September 26-28

Chicago plays host to the largest gathering of Amateur Radio digital enthusiasts in the country. Make your reservations now for three days of education and camaraderie, including a Sunday seminar on Software Defined Radio by Phil Harman, VK6APH.

See the Digital Communications Conference site on the Web at www.tapr.org/dcc/ or call TAPR at **972-671-8277** to make your reservations today.

DStar DV Sensitivity vs. Analog Sensitivity

By Mark Miller, N5RFX

I have heard many conversations about increased range with the DStar system and decided to test this out for myself. The claims of increased range should correlate with the sensitivity of the receiver. I found however that the noise free reception of DStar DV occurs over a larger range than an equivalent strength analog signal.

FM RECEIVER SENSITIVITY

The sensitivity of an FM receiver is given as the power at the antenna needed to provide a 12 dB SINAD. SINAD pronounced by some as "sine add" and some as "sin add" is the ratio of the signal plus noise, and distortion over the noise, and distortion. When expressed in dB the ratio at sensitivity is 12 dB SINAD. A signal generator is frequency modulated with a 1000 Hz tone (sometimes 1004 Hz) and the modulated R.F. is brought to the antenna port of the FM receiver. The deviation is set for 3.3 kHz. The audio output from the FM receiver is brought to a device to measure SINAD. Sometimes this is a device called a SINADDER, distortion analyzer,

or transmission impairment test set. The R.F. level is adjusted until the instrument measures 12 dB SINAD.

With digital signals SINAD is not a good measurement of sensitivity, because typically the SINAD of a digital receiver is quite high until a certain point where the audio signal disappears. There is no gradual falling off of the signal like there is with analog signals. Figure 1 shows the results of sensitivity tests performed on an ID-800. The ID-800 in wide FM mode is intended for FM transmissions with a 3 to 5 kHz deviation. At the -102 dBm point the analog FM receiver is at its highest SINAD. The 12 dB SINAD of this receiver occurs at -122 dBm. In Digital Voice mode (DV) the signal drops off at -120dBm. This means that the analog FM receiver is about 3 dB better in sensitivity than the DV receiver. 12dB SINAD signals are typically considered difficult copy. What we consider to be a full quieting FM signal is one where the SINAD is nearly 30 dB. The 27 dB SINAD point in Figure 1 occurs when the DV signal stops or

is unintelligible.

DV ADVANTAGE

The DV signal has a steady noise level to -119 dBm and drops off at -120 dBm. The analog FM signal SINAD begins to drop at -102 dBm. Between -102 and -119 dBm DV has a SINAD advantage over analog FM. The advantage occurs over a 17 to 18 dB range. When noise free signals are desirable, DStar digital voice can meet this requirement with a 17dB to 18dB increase in the range that noise free operation can occur. For weak signal work, the analog FM signal will prevail.

CONCLUSION

Trading 2 dB of sensitivity for a 17dB increase in nearly noise free reception is an advantage of DStar over analog FM. When weak signal reception is necessary, the analog signal will provide better performance.

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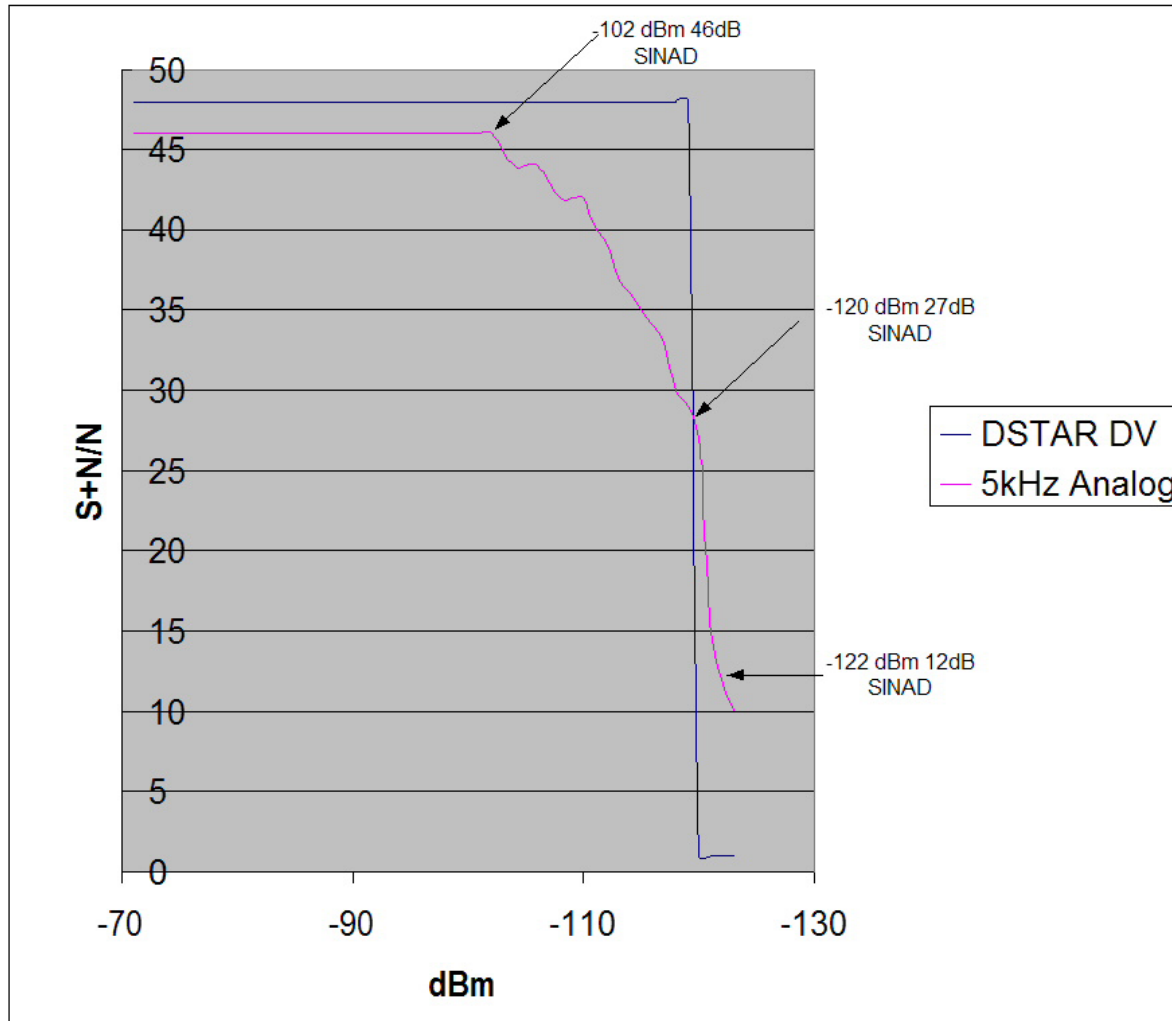


FIGURE 1 DSTAR VS. ANALOG FM SENSITIVITY

EcomScs

By John Blowsky, KB2SCS

EcomScs is a new packet radio e-mail client.

Everything that you can do with your Internet e-mail client you can do with *EcomScs*.

EcomScs uses the Packet Radio BBS system to transport its e-mails.

It will connect to your home BBS and check to see if you have mail waiting for you. If it finds any, it will automatically download your mail to the Inbox directory on your PC. After it downloads your mail from your home BBS, *Ecomscs* will then check to see if you have any mail in your Outbox directory. If you do, it will then automatically upload your out going e-mails to your Home BBS.

EcomScs is fully Mime compliant and uses Base64 to transport your attachments.

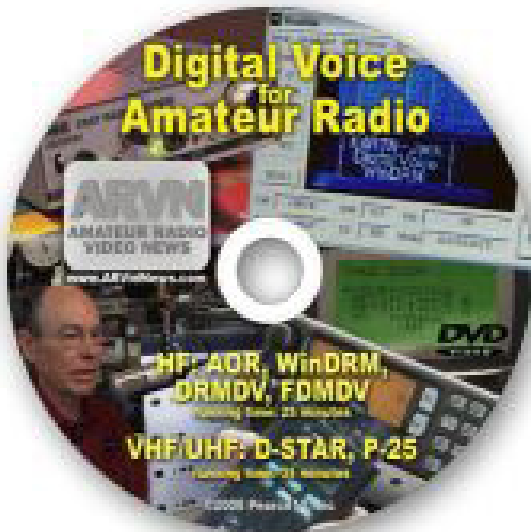
Please go to <http://www.qsl.net/kb2scs> to learn more about *EcomScs*

Please notice on the web page the URL for the Google Group for *EcomScs*.

###

Digital Voice for Amateur Radio DVD

By Mel Whitten, KØPFX



Gary Pearce, KN4AQ of Amateur Radio News has just completed a program on the various Digital Voice programs in use today for both HF and VHF/UHF. These programs include the HF DRM derivatives WinDRM/DRMDV, a sound byte of the new FDM DV and AOR's 9000 series DV modems. For VHF and above, DV is described using P-25 and D-Star. The hour plus long DVD provides an introduction to all these modes plus on-the-air demonstrations. For an 8 minute preview see <http://www.arvideonews.com>.

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WRITE EARLY AND WRITE OFTEN



Packet Status Register (PSR) is looking for a few good writers, particularly ham radio operators working on the digital side of our hobby, who would like to publicize their activities here.

You don't have to be Shakespeare to contribute to *PSR* and you don't have to use Microsoft Word to compose your thoughts. The *PSR* editorial staff can handle just about any text and graphic format, so don't be afraid to submit whatever you have to walou@tapr.org.

The deadline for the next issue of *PSR*, the Digital Communications Conference issue, is July 31, so write early and write often.

###

Digimode Identifiers

By Patrick Lindecker, F6CTE

INTRODUCTION

The identification of a digital amateur mode (commonly called digimode) can't be done automatically by software. This is due to the diversity of the modulations used (BPSK, QPSK, FSK, MFSK, IFK, OOK...), speeds used (from 1 to 9600 bauds), codings used (convolutional, Reed Solomon or other). Also, it may have several transmissions of different type on the same bandwidth.

However, it can be identified by mode, but for a given modulation and within a sharp bandwidth (as for BPSK in Multipsk). It should be remembered that the RadioRaft software of François Guillet, F6FLT which permits automatic identification of many FSK modes (for one transmission on the bandwidth).

It is also possible to quantify speed modulation or shift.

To conclude, the problem of automatic identification of digimodes, even if exciting, cannot be solved at the present time (perhaps, it will be solved in the future with processings based on artificial intelligence algorithms).

Some modes are simple to identify, either because

their frequency is well known (BPSK31 on 14070 KHz, for example) or because the visual signature on the waterfall or the acoustic signature is characteristic (RTTY 45 bauds for example).

But for the other digimodes (more or less exotic), it is very difficult to identify the mode or the sub-mode used, with the simple visual and acoustic traces, therefore, the need of digimodes identifiers.

At the present time, the only official identifier is the one which defines the analogical SSTV sub mode used (Robot 36, Martin 1...). It works perfectly but it is limited to SSTV.

In general, it is possible to send (at 20 wpm on Multipsk) a small CW text before each digimode transmission. But this is rarely used. Moreover, CW is not always understood and, even if understood, the latency time before the cerebral decoding occurs, causes one to lose the beginning of the message.

Here are two new identifiers of modes (RS ID and Video ID) which could simplify the identification of digimode transmissions, if they would be more widely used..

Reed-Solomon identifier ("RS ID") of mode and frequency

MAIN USE

The "RS" ("RS" for "Reed-Solomon") identifier allows the automatic identification of any digital transmission done in one of the RX/TX modes handled by Multipsk (103 modes and sub-modes in version 4.8, from BPSK31 to JT65-C) plus the FDMDV mode from Cesco (HB9TLK). Two events occur: the mode used is detected and the central frequency of the RS ID, which is also the central frequency of the identified mode, is determined with a precision of 2.7 Hz.

As soon as this identifier is received, Multipsk switches on the received mode and frequency and decodes immediately the QSO in progress or the call (CQ). This identifier is transmitted in 1.4 sec and has a bandwidth of 172 Hz. Its detection is done down to a Signal to Noise ratio of about -13 dB, so with a sensitivity equal or better than the majority of the digital modes (RTTY, PSK31...), except several modes as PSK10, PSKAM10, THROB, THROBX or JT65.

Note: consequently, it could appear that the RS ID be detected but the call or the QSO could not be decoded due to a too weak signal.

This identifier can be transmitted, first, before each general call or prior to each answer in a QSO.

The search is, in general, done in the bandwidth 200-2500 Hz. For this bandwidth, the equivalent CPU load is about 200 MHz.

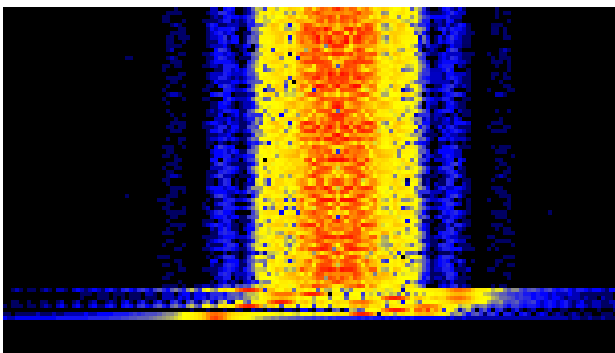
At the present time (27th of March 2008) two programs offer the RS ID:

- * PocketDigi by Vojtech (OK1IAK)
- * Multipsk by the author

RS ID programs sources are available either from Vojtech (OK1IAK) or from the author.

The list of the RS ID identifiers (in form of numbers) associated to digital modes is managed by the author.

For example, here is the PSK63 RS ID (root) received just before the PSK63 transmission itself (trunk).



PRINCIPLE

Each mode corresponds to a number which is transformed in a particular Reed-Solomon sequence. This RS coding (RS(k=4,t=6)) is defined by the parameters k=4 (number of bits per symbol), n=15 (=2^k-1, number of symbols by RS sequence) and t=6 (maximum number of errors which could be theoretically fixed). It means that each RS sequence is composed of 15 symbols of 4 bits, among which 3 (=n-2t) carry data. In other words, 12 bits (3x4) are available to define the mode number. Even if the maximum number of errors which could be fixed is equal to 6, it is limited to one correction so as to have a negligible probability of false detection

(by increasing the Hamming distance between any random sequence and the selected sequences). Consequently, the number of possibilities would be equal to 4096 (2¹²).

However, it has been conservatively supposed that two RS IDs could be sent successively and that two RS IDs could be also sent in juxtaposed frequencies. As false RS ID detection with part of one and part of the other (either in the time domain or the frequency domain) must be avoided, it has been determined a sub-set of RS ID which are really independent (“orthogonal”) from each other, i.e., two RS ID sequences can’t be mixed up so as to produce a valid but wrong RS ID code. This sub-set is only composed of 272 possibilities. It is the first choice.

Each symbol is transmitted in a MFSK modulation. There are 16 possibilities of frequencies separated by 11025/1024=10.766 Hz, each symbol transmission being done on only one frequency for a duration equal to 1024/11025x1000=92.88 ms. Therefore, the 15 symbols are transmitted in 15x1024/11025=1,393 s.

The decoding is done in a “soft decision” way, so without any hard-decision Reed Solomon decoding algorithms, the decoding gain being equal of about

2 dB. It means that all the possibilities are explored (“brute force” algorithm type).

However, Vojtech (OK1IAK) uses also a “hashing” technique to accelerate the calculation.

For each semi-step of time (46.44 ms) and for each semi-step of frequency (5.38 Hz), it is determined if a RS ID has been transmitted during the last 1.393 second. So each second, about 8500 (depending on the selected bandwidth) possible RS IDs are tested (depending on the bandwidth). As the probability of false detection is almost nil; there is no problem to test so many possibilities.

The analysis is based on FFTs (Fast Fourier transform) on 2048 points at 11025 samples/sec, regularly done at each semi-step of time (46.44 ms).

Note: it could be imaginable to sample at 44100 “Hz” to detect any RS ID within a bandwidth of 20 KHz.

As it is a free error transmission, in presence of an identifier, there are two solutions:

* Either the RS ID identifier is not received because the signal is too weak,

* Or it is received and it is correct, the probability of detection of a wrong identifier being almost nil.

TRANSMISSION

In all modes (including MT63, SSTV, Fax, FELD HELL...), an identifier can be transmitted. It is sent on the average transmission frequency, except in SSTV where it is sent on the synchronization peak as displayed on the waterfall (between 1150 and 1400 Hz).

The video identifier of mode (“Video ID”)

MAIN USE

The mode label and/or other information is sent in CMT Hell (Hellsreiber) before the main transmission. This identifier will be visible in the “waterfall” of the receiving station.

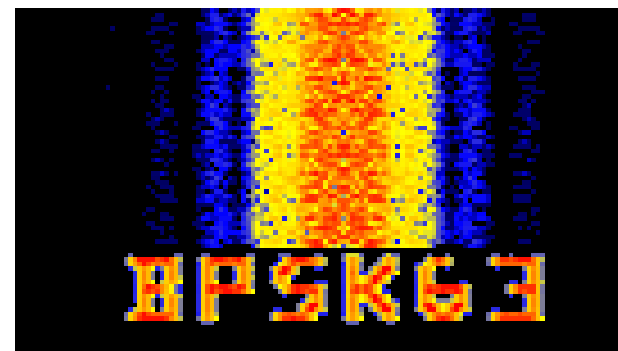
The identifier can be transmitted either:

- in a horizontal shape : the transmission is quick but not powerful,
- or in a vertical shape: the transmission is slow but is more powerful because more concentrated (much less carriers).

For example, just before the general call (“CQ”) in BPSK63, “BPSK63” will be sent automatically on CMT Hell. The receiving station will see “BPSK63” displaying on his “waterfall” and the operator will

switch immediately to the BPSK63 mode (the switching is not automatic as with the RS ID).

Here is what is seen in the waterfall (Video ID followed by the BPSK63 transmission itself).



At the present time (27th of March 2008) three programs propose the video ID:

- * FIDigi by Dave, W1HKJ
- * Digipan by Skip, KH6TY
- * Multipsk by the author

PRINCIPLE

The initial idea came from Henri (F6BAZ), who advocated (in December 2005) that the waterfall be

used to display the name of the mode in CMT Hell (see <http://f6baz.free.fr/windrm4.jpg>).

Each character is transformed in capital letter, if necessary, then in pixels matrix according to the selected font (see hereafter).

Apart from the MT63 (which is a particular case), the transmission is done at 11025 samples/second. The FFT is done on 4096 samples but with 2048 samples of the previous FFT buffer so with new 2048 points only. This comes to double the display speed, with a light dependency between successive pixels. Consequently, between 2 pixels, it will be found intervals of frequency and time of respectively:

$$* 11025/4096 = 2.691 \text{ Hz,}$$

$$* 2048/11025 = 0.1858 \text{ second.}$$

The pixels are transmitted in the Concurrent MT Hell mode (for mode details, see the site of Murray, ZL1BPU: http://www.qsl.net/zl1bpu/FUZZY/MT_intro.htm).

To sum up, the transmission is done on many juxtaposed carriers sent in parallel. The presence or the absence of a carrier determines either a trace (dot) on the waterfall or an absence of trace. So each line will be composed of a set of dots and a set of lines

will constitute a set of characters.

Note: to keep linear (without any overload), each carrier will be limited in amplitude.

The mode name is centered in the middle of the mode bandwidth (see the example about BPSK63 above).

If the transmission is reversed (for FSK, MFSK and QPSK modulations), the name is also reversed (it will be written from right to left). Remember normally, in HF, digimodes are transmitted in USB not in LSB, and this independently from the frequency.

Mode names will be transmitted in their standard name: «BPSK31», «BPSK63», «MFSK16», «PAX2», «FAX»..., if they are not ambiguous.

However, there exists several special cases. For example, MT63 name will be «MT63 bandwidth interleaving» type with for the bandwidth: «500», «1K» (for «1000») or «2K» (for «2000»), for the interleaving «VST» («Very short»), «ST» («Short») or «LG» («Long»)....so a maximum of 12 characters («MT63 500 VST», for example). See the Multipsk help for more details about the way to express mode names.

The characters can be sent in different Hellsreiber fonts («Hell 80 double», «Feld Hell double, normal», «Feld Hell double, bold»).

Relatively to the transmitted mean power:

The transmitted mean power will be weak due to the fact that it is necessary not to saturate the signal to keep the text readable. Thus, the more there will be «horizontal» pixels to transmit, the less the mean power will be important. However, the human capacity to read a very noisy text compensates for this weak power.

(Thanks to Bill Duffy, KA0VXK, for proofreading this text.)

###

WT4M's Octopus Console

By John A. Ficke, WT4M

Big thanks out to Scott (N1VG) and Jason (KE4NYV) for letting my XYL know where I am at all times. (Yikes!) Scott for designing the OpenTracker and Jason for hooking me at the CARS SpringFest!

Since I already had a radio box built into my truck, I wanted to have console type controls for the OT1x. This outlines what I did to accomplish that end (more or less successfully).

First I assembled the OT1x, all in one sitting. I convinced myself that it would be best to test it the following day, after getting some much needed sleep. But in the end my curiosity won and I foolishly pressed testing in the same evening. Wouldn't you know it; I hooked up power with the polarity reversed! The 5 volt rectifier was not very happy with this situation and surrendered after a few milliseconds I suppose.

Fortunately the rest of OT1x is a tough cookie and persevered! After a quick trip to Rat Shack and a little more solder it was back up and running like a champ.

After some steep learning curves with APRS SSIDs, I am now transmitting as WT4M-14. Not that I am a purist, but I believe Bob Bruninga (WB4APR) should have the final word. I started with just WT4M and then WT4M-9

(as Bob listed 14 as "truckers" somewhere or another). I had assumed this meant of the more than two axle kind. The whole SSID thing is kind of murky, so probably anything goes, but I think I will stick with 14 for my Silverado...

By the by, I found that findu.com left a lot to be desired as far as tracking goes. After some extensive searching I found this awesome site in Helsinki! If you haven't been there yet, once you use it, I'd doubt if you would ever use any of the others...

In order to accomplish my "console" I needed to modify three OT1x lines on X1. I wanted to keep the box as close to original design as possible, but definitely wanted to have a console LED!

I am not certain if I had to modify Pin 8 to JP3 or not, it may have suited my purpose as it was, but I didn't want to chance it.

OT1x X1 Pin 2 (Change to LED OUTPUT):

I didn't need the "COR/SQUELCH INPUT" for the HTX-212, so I cut the trace that leads to the junction of the RX variable resistor and C3. I then jumpered from X1 pin 2 to the junction of R4 and U1 pin 3. According to the specs for U1, pin 3 will provide up to 15mA. Placing the 600Ω resistor will limit current to 8mA for the external LED (a

total of 13mA between the external and OT1x LEDs). I couldn't find a 600Ω resistor, so I used two 1kΩ in parallel and a 100Ω in series.

OT1x X1 Pin 8 (Change to PWR CONTROL):

I also had no need for the PTT INPUT. I cut the trace from X1 pin 8 to the junction of D2 and D3. Then jumpered pin 8 to JP3 "PTT INPUT OR RELAY OUT".

OT1x X1 Pin 9 (Change to XMT REQ):

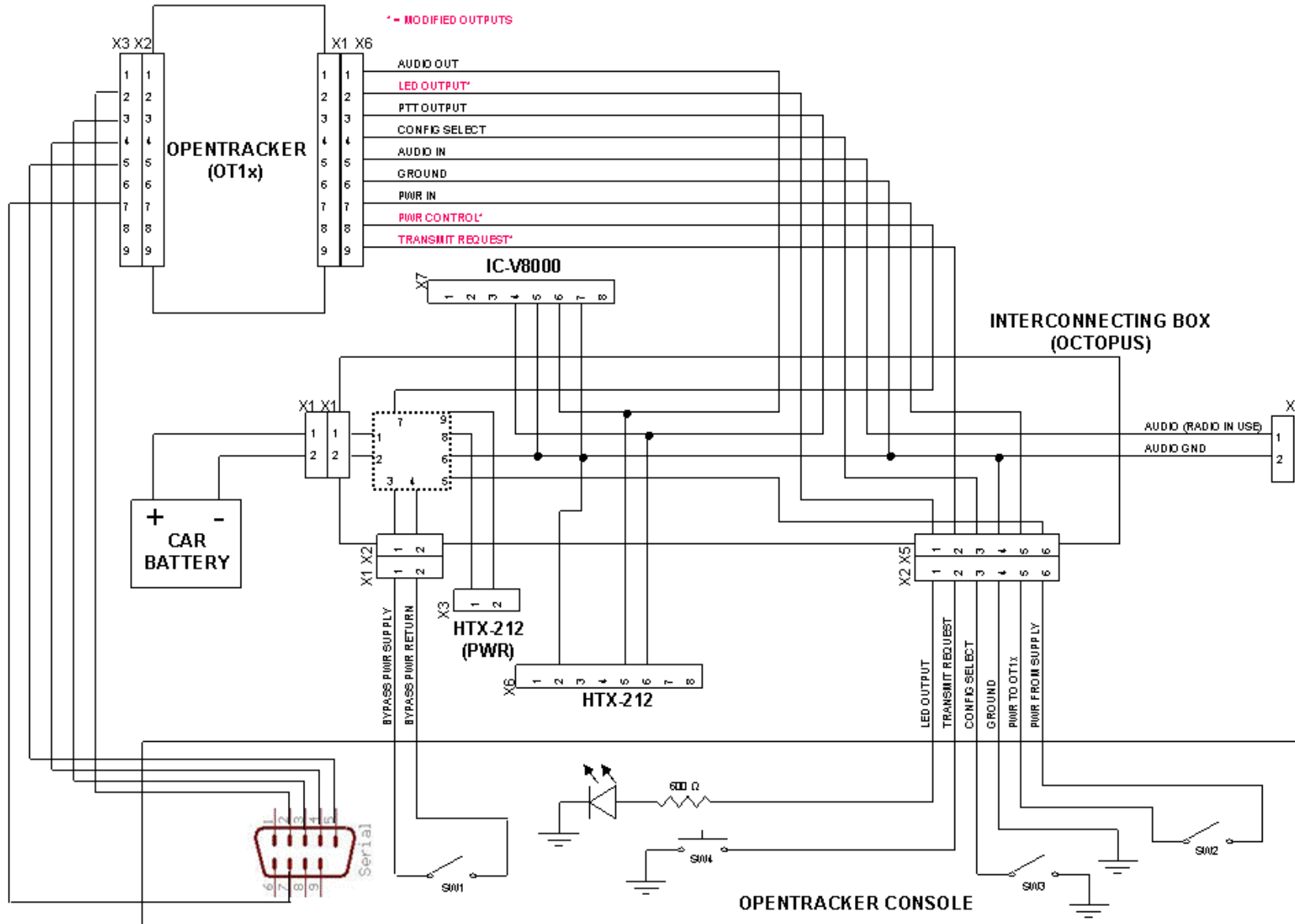
Pin 9 was a no connection pin, so no need to cut any traces, just jumper from JP4 "IRQ".

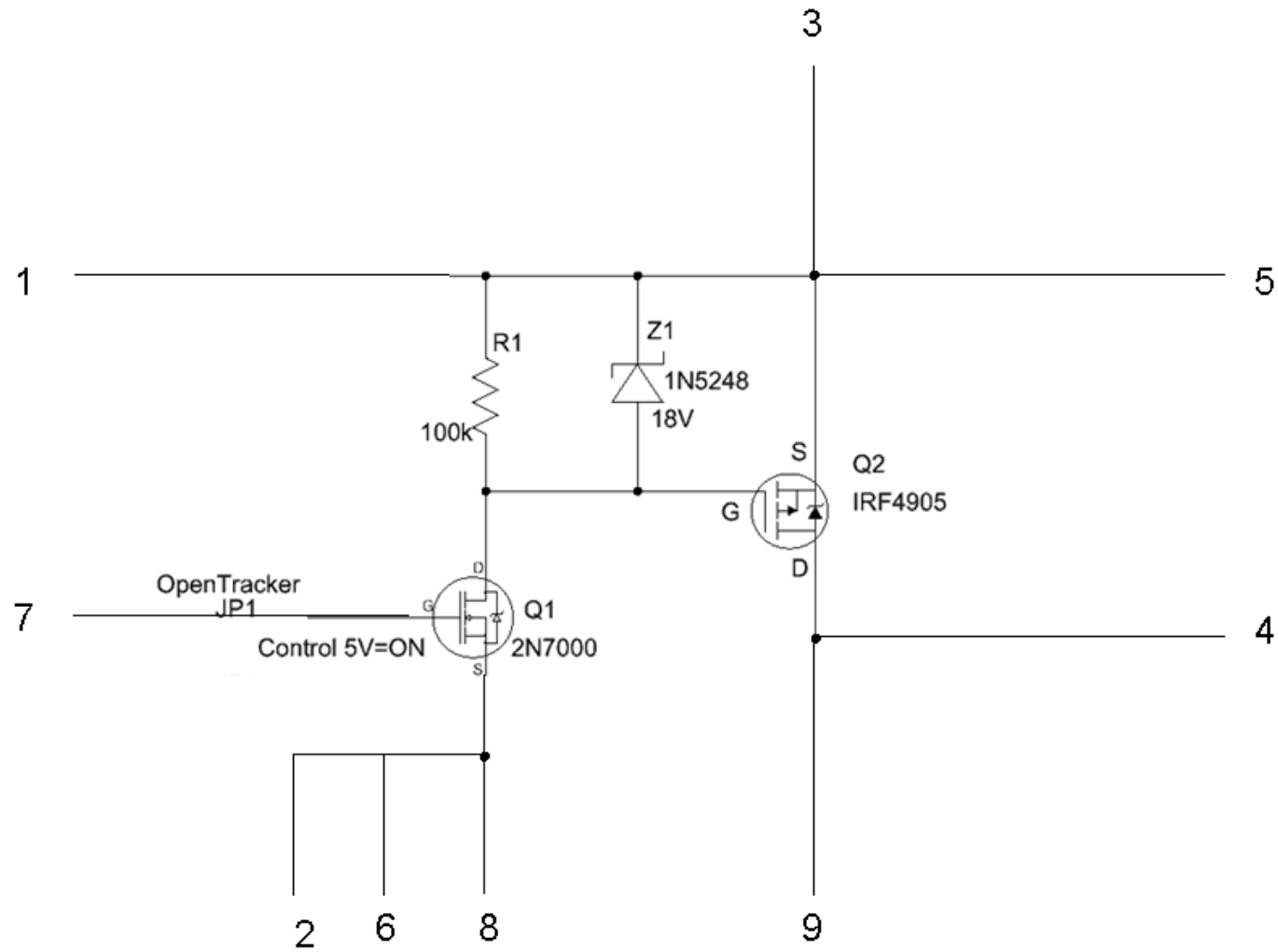
Notes:

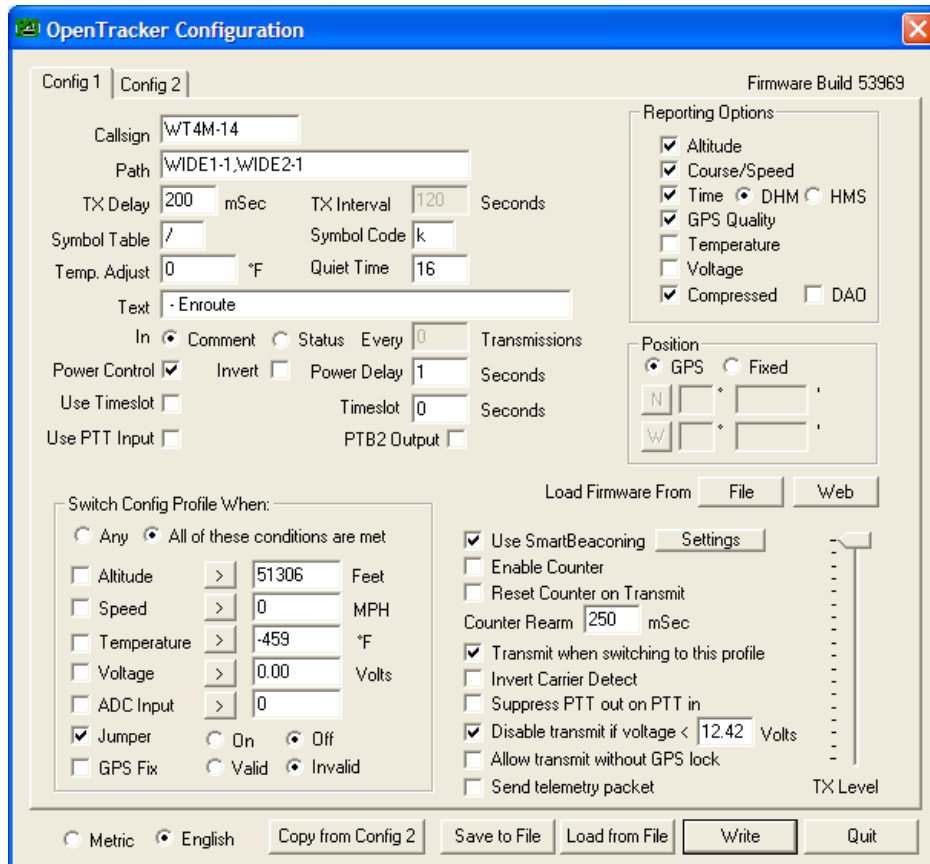
1. The rating of the switches is inconsequential, except for SW1, which must be rated for as many amps as your rig is going to draw. In my case I used a 10 amp rated switch (for the HTX at high power, though I rarely use it).

2. I wrangled over LED specifics, and ended up using a blue one from Rat Shack, rated at 5 volts and 30 mA. I thought that it might be very dim, but is just bright enough not to blind me at night.

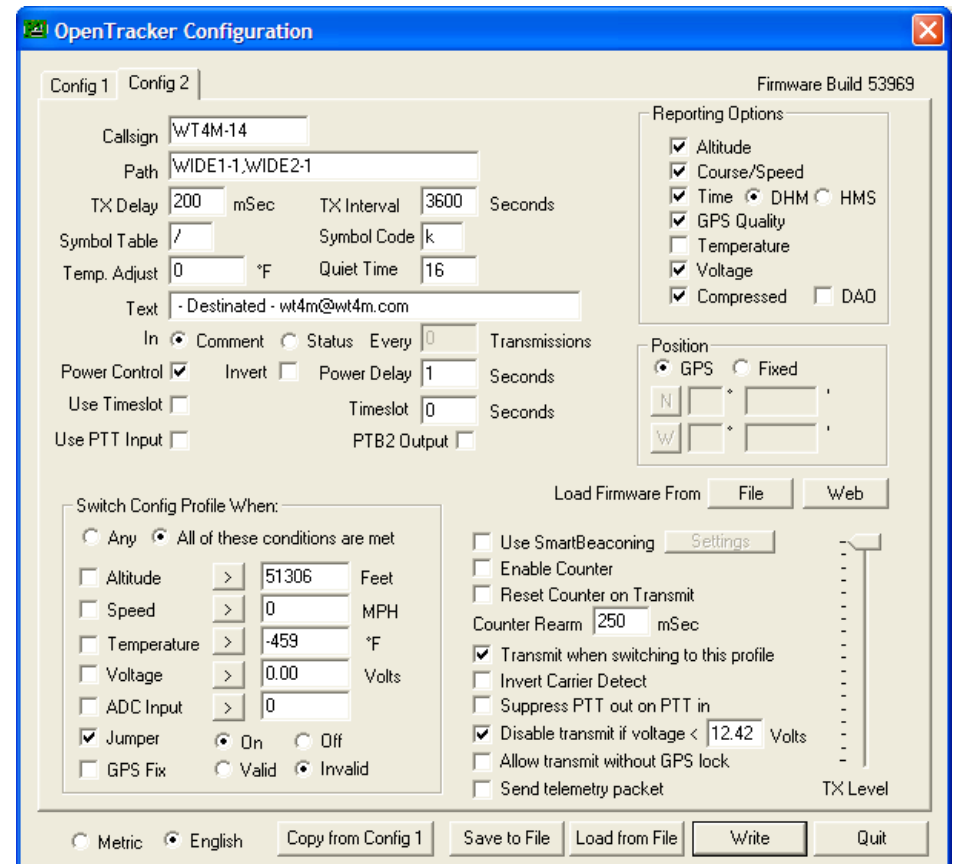
My Diagram (thanks to Juergen and ZL1VFO for the power control):







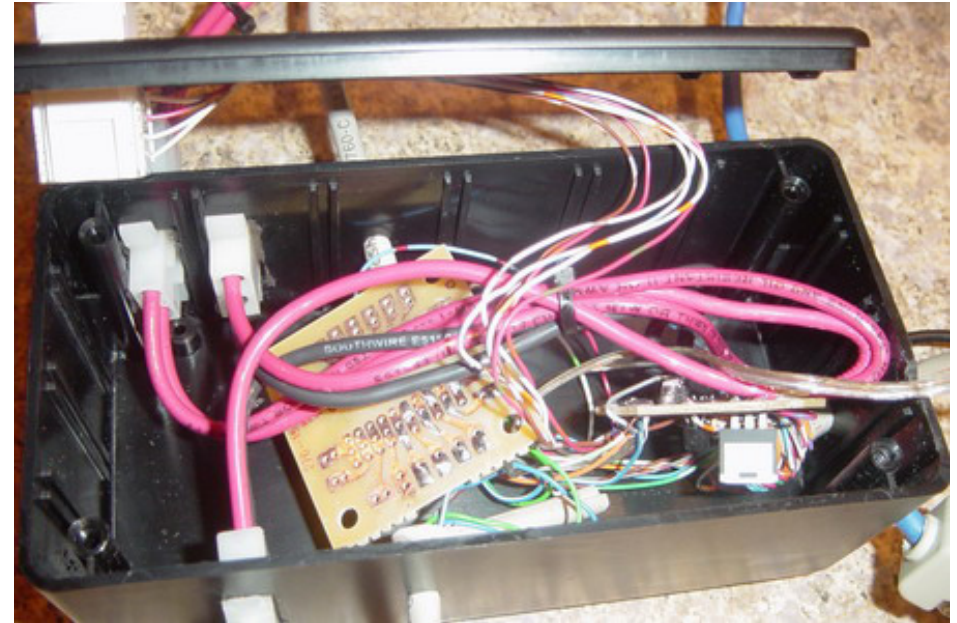
CONFIG 1 (WHEN I AM ON MY WAY SOMEWHERE)



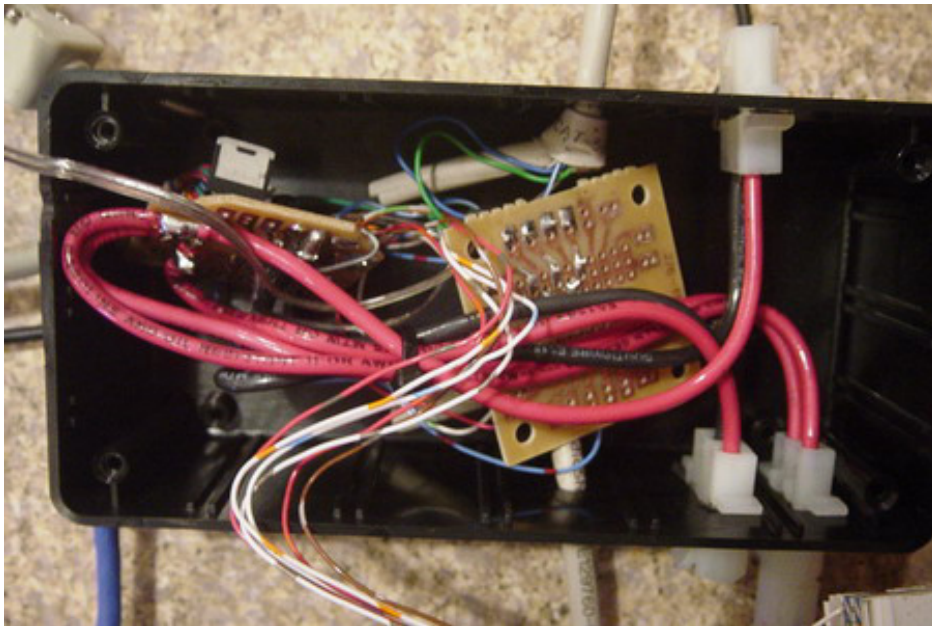
CONFIG 2 (DESTINATED - TRANSMITS ONCE AN HOUR)



**OK, so I AM NOT THE MOST ORGANIZED FELLOW IN THE WORLD...
WHAT COUNTS IS THE END RESULT... RIGHT? (NOT ACCORDING TO THE
XYL!)**



I HAVE NAMED THIS CONTRAPTION THE OCTOPUS!



ANOTHER SHOT OF THE ELUSIVE SEA CREATURE...



THE CONSOLE... NOT BEAUTIFUL, BUT IT WORKS...

THIS SHOT IS UPSIDE DOWN. I HAD ORIGINALLY PLANNED TO INSTALL IT THIS WAY, BUT MADE ASSUMPTIONS ABOUT THE SWITCH POSITIONS THAT TURNED OUT TO BE THE OPPOSITE OF WHAT I EXPECTED. HENCE THE 180 DEGREE INSTALLATION... (HOMER WOULD SAY, "DOH!")



EVERYTHING READY TO GO!



THE FINAL INSTALLATION...





ANOTHER SHOT... OT1X CURRENTLY CONTROLS THE HTX-212, BUT IF I UNPLUG IT I CAN PLUG INTO THE V8000.

WITH THE GPS CONNECTED...

DESTINATED!

IF YOU HAVE ANY QUESTIONS PLEASE DROP ME A LINE! AT wt4m@wt4m.com

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phone 972-671-TAPR (8277)

fax: 972-671-8716

e-mail tapr@tapr.org

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PSR Packet Status Register Editor:

Stan Horzepa, WA1LOU

One Glen Avenue, Wolcott, CT 06716-1442 USA

phone 203-879-1348

e-mail wallou@tapr.org

TAPR Officers:

President: David Toth, VE3GYQ, ve3gyq@tapr.org

Vice President: Steve Bible, N7HPR, n7hpr@tapr.org

Secretary: Stan Horzepa, WA1LOU, wallou@tapr.org

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David Toth, VE3GYQ, 2010, ve3gyq@tapr.org

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