Spread Spectrum Scene

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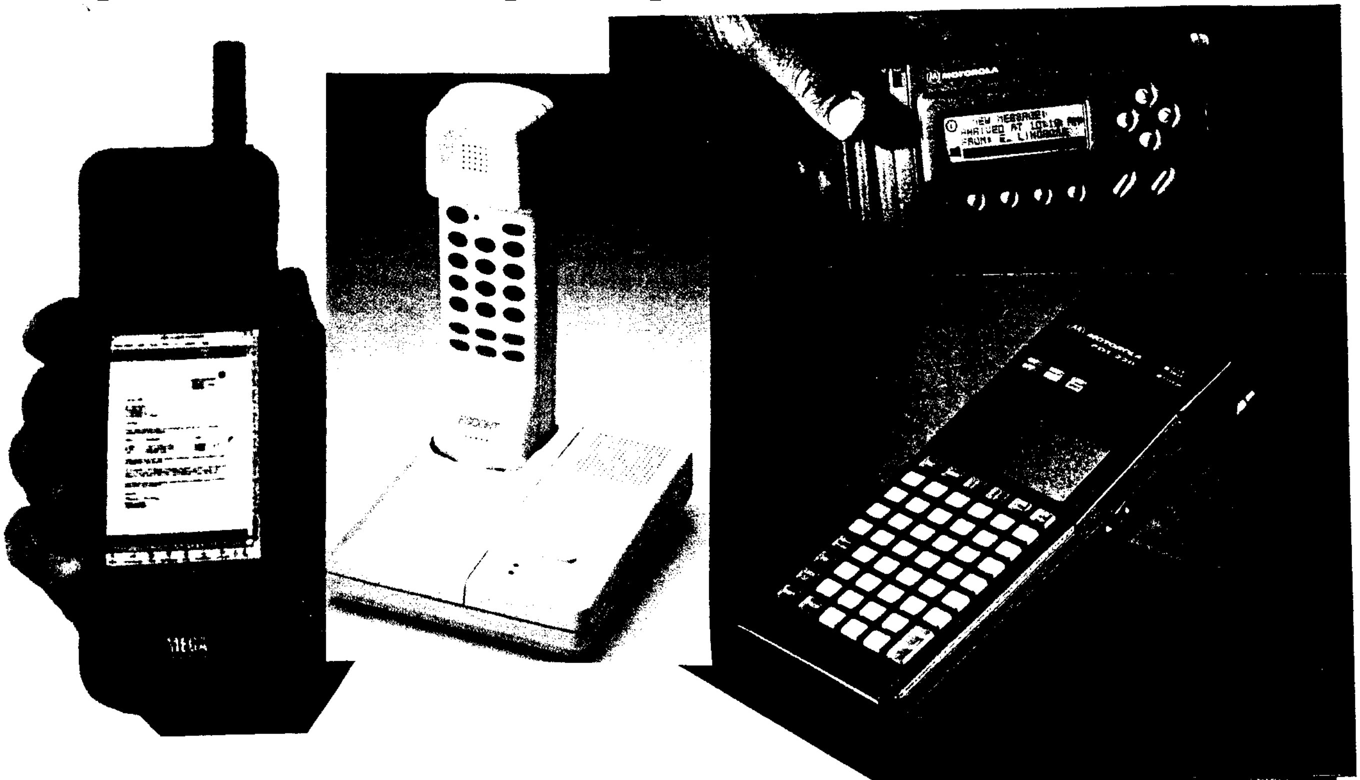
Special Issue on New Spread Spectrum, LAN and PCN Products



What's Inside

The Future of Wireless LANs and PCNs

Some of the latest wireless products for LANs and PCNs are pictured here, Do these products indicate the future direction of Spread Spectrum applications?

Are we about to enter the era of the PDA (Personal Digital Assistant, or Appliance)? Or are these sleek new products just more misguided marketing ideas that represent sidetracks to the direction of progress in this business? This month's editorial discusses these and other important issues for Spread Spectrum's future.

see EDITORIAL page 2

Are We Ready For This? More New Products Inside Spread Spectrum Scene

SPREAD SPECTRUM SCENE is dedicated to the Spread Spectrum professional and is committed to being the primary source for the latest news and information about the growth, regulation, and opportunities in this emerging science.

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Rumors & Ramblings

● Telxon Corporation announced that its joint project with Wal-Mart Stores, Inc. to implement the wireless spread spectrum radio communication network has been successfully installed in record time in all of Wal-Mart's 1,804 stores nationwide.

● Telxon Corporation also announced a unique plan with ARDIS, a joint venture of IBM and Motorola, to allow field service organizations to test the power of handheld microprocessors and wide area wireless data networks.

● Bell Atlantic and Cellular Data Inc. are installing a new wireless data system based initially in Baltimore. Bell Atlantic is also working with IBM and Westinghouse Electric to provide other facets of the proposed new service.

● Heard a good rumor -- want to "leak" some info to your competition -- call our 800 number and we may print it!

Decipherings

MAKE VOYAGES.
ATTEMPT THEM.
THERE-S NOTHING ELSE.

- Tennessee Williams-

EDITORIAL

Whither Wireless LANs and PCNs?

The front cover of our rather late, special October issue shows some of the latest hardware for LANs and PCNs. Are these products representative of the future direction of applied spectrum technology? We attempt to answer this interesting question in this issue and in this editorial.

First let me say that this special October issue is devoted to what industry is currently doing with spread spectrum technology. You will see quite a few new products and some interesting advertisements in this issue. Some of these products will undoubtedly find a market niche. Other products, that are less well marketed, conceived or funded will probably go the way of the dinosaur. Honestly, even my crystal ball is a little foggy when it comes to picking the survivors from the dogs in this business!

Given the tough highly competitive marketplace out there and the fact that no major segment of the consumer market has demanded large volumes of these products yet, this marketplace is like a horse race run on an old fairgrounds' dirt track during the off season -- a lot of nags competing with a few dark horses. Are we ready yet to see a shake out of manufacturers and products? Well, not really -- some 31 vendors or interested parties sent the FCC comments on the proposed revisions to Part 15 spread spectrum rules, some 2 1/2 years ago. Now only about 15 of those companies are still in business. Thus, shake outs seem to be a definite on-going process, not always related to the marketplace or product mix.

As we all have noticed, our economy is a mess! It may be tough for any new, currently non-essential technical gadget to gain market acceptance. Everybody says things will be better after the election. But, I wonder! Just maybe, the economic malaise we are suffering is due more to our gadget orientation, our fascination with high technology and B-school MBA company Presidents/CEO's, who have forgotten (or never knew) the art of critical thinking.

This country needs to get back to basics: we need to create jobs! This means manufacturing jobs -- not service sector jobs. We need to stop exporting raw materials and resume exporting and dominating the industries of the world through high technology. **Forget the frivolous -- come up with products that improve productivity, reduce costs and till a real world need in an environmentally sound way!**

Maybe when engineers and scientists run our companies, like they do in Europe (not B-school MBAs) -- just maybe, we'll have a chance again. Until then, guessing the direction of SS or any other technology is just a waste of time!

There were two very meaningful conferences in the last **6** weeks or so: (1) The Wireless Data Conference and Exposition, Oct. 13-15, in Santa Clara and (2) The IEEE sponsored First Annual Wireless LAN Implementation Conference, Sept. 17-18, in Dayton. If any of you are interested, next month we'll highlight the most interesting papers from one or both shows. Call, write or FAX your wishes.

The Aerial

by Peter Onnigian, P.E., W6QEU

Gain Over What?

Recently I gave a talk on Yagi antennas at a national amateur radio convention. With nearly 200 in attendance, it became apparent in the Q and A session, many were antenna engineers and technicians, some from large defense firms. It was sad to hear a few did not know the difference between an isotropic (dBi) and a half wave dipole (dBd) reference. They also had difficulty understanding why the gain of the same antenna may be expressed as a higher number in dBi than in dBd!

Antenna Gain References

To understand an isotropic antenna, imagine the radiator totally enclosed in a hollow sphere. The radiation from its center is distributed uniformly over the interior surface of the sphere. This uniform radiation is said to be isotropic by definition.

Assume further, it takes one watt of power to cover the entire surface of the sphere with 100 milliwatts of intensity. If we were to illuminate only a small portion of the sphere with the same surface intensity, it follows that the radiation source power required would be much less than the one watt required to illuminate the entire sphere. In fact, a dipole would illuminate a wide band only on the sphere with the same 100 milliwatts intensity.

In fact, it would require only 0.61 watts of power for the same intensity as required to coat the entire sphere. This reduction to 61 percent by a dipole is equal to -2.146 dB, rounded off to -2.15 dB. Thus the isotropic sphere has 2.15 dB more gain since it requires more power to illuminate the entire sphere! Gain is the ratio of the maximum radiation in a given direction to the maximum radiation produced in the same direction from a reference antenna both with the same input power.

Another definition: The directivity is the antenna's ability to concentrate radiation in a particular direction. Useful antennas exhibit some directivity unlike an isotropic, which radiates equally in all direction. As stated in last month's column, an isotropic antenna exists only as a mathematical model, and is not realizable in practice.

The gain of an antenna is a basic property and is frequently used as the figure of merit. Gain is the directivity of an antenna, less the various losses inherent in it. These include IR, dielectric, VSWR mis-match, undesired side lobes, front to back ratios and other losses.

Gain Numbers

Our interest is the relative gain of commercially available antennas. The common practice expresses gain in decibels relative to that of a half wave dipole. This gain is expressed as dBd, that is decibels over a dipole. However this does not hold true for all those available in the market place.

For example antenna brand X is rated as 8.5 dBd. This is equal to brand H antenna which is rated as 10.65 dBi [8.5 + 2.15 = 10.65] Misleading'!

Aerial from page 3 - To some yes, Very misleading!

Suppose you have to choose an antenna for the higher gain. They are both priced equally. Would you pick the one rated at 10.0 dBi or the one rated 9.15 dBd? To find the answer, level the reference. Either subtract 2.15 dB from the isotropic to equal dBd, or add 2.15 dB to the one rated in dBd. Don't compare dBi with dBd. That's comparing apples with oranges. Most engineers compare with the dipole reference, so subtract 2.15 from the dBi value, to equal 7.85 dBd. The antenna rated 9.15 dBd has 1.3 dB higher gain than the one rated as 10.0 dBi.

If the manufacturer insists on playing the numbers game and rates his antennas in dBi, then you should normalize it to dBd to get the true value.

Truth in Labeling

There is nothing that beats measuring antennas on a certified and accredited antenna range. While expensive, the numbers are accurate. This procedure of range measurement eliminates the sales hype and gives the buyer/user confirmation of antenna gain numbers.

Unfortunately in the spread spectrum and burglar alarm fields there are very unscrupulous manufacturers whose antenna gains are highly over rated. Then there are those that don't indicate if the gain stated is simply guess work, measured or developed by the sales department! The worst offense is stating gain in dB without reference to isotropic or dipole. Maybe it's gain over a wet noodle!

Technical Trends in Education

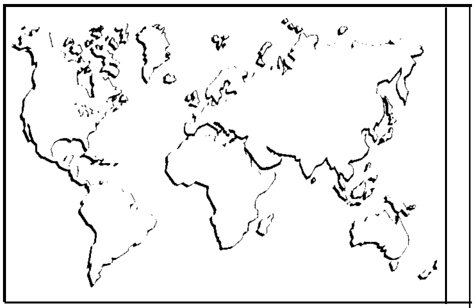
by Tom Diskin

Technical Trends In Education

Some years ago when I was a budding electronics hobbyist, I remember reading an article in a leading electronics magazine entitled "Me Technician, You Engineer." This article carefully delineated the separate tasks of both technicians and engineers, as well as how they must work together to achieve their common goals. I believe the author's goal in writing this piece was to emphasize that these two job titles are both mutually exclusive of each other, while at the same time highly dependent in terms of the end product they would jointly produce.

Around the same time, a paper was delivered to electronics instructors in California at a meeting of the Electronics Instructors Council on November 2 1, 1959 entitled "Building Status for Technicians." In his presentation, Mr. Charles R. Mulkey of Monterey Peninsula College criticized those who describe technicians as "assistants to engineers" or one who "...works for an engineer or under the close supervision of an engineer." He maintained that technicians, like engineers, create, design and build, and in some cases do the job well enough that they are often more highly praised by their companies than the best engineers. In some cases, senior technicians often act as consultants to engineers, for they are the members of the team who understand the materials, processes and limitations of techniques needed to successfully complete a project. Mr. Mulkey further stated that the technician "...may in reality be the functioning part of the engineer. He may be the engineer's only real connection with a real world. He is very often the 'practical' brain of the engineer. He is, above all, a man who knows the new materials and the new techniques, and he knows and understands their limitations." In other words, he is the one who really applies the technology of our modern Western civilization, with understanding of its scientific basis and a practical knowledge of its limitations.

While we may take issue with some of Mr. Mulkey's statements, we must temper this with the understanding that these statements were made nearly 33 years ago! How much has the status of the technician changed in the ensuing years'? While we have added many related job titles, such as "Engineering Technician", "Field Service Technician", and the like, has the relationship See Education page 29



International Scene

Germany recently deregulated and re-structured the Postal Administration. The Deutsche Bundespost (DBP) now has the Ministry of Posts and Telecommunications (MPT) with three separate departments in charge of all telecom.

The British "Duopoly" of telecom includes BT plc and Mercury Communications Ltd. The UK Office of Telecommunications (Oftel) started a "duopoly review" process that initiates a new competitive era in the British telecom industry.

The French have also re-vamped their century old telecom laws. It's still a government monopoly, but value-added services, cellular telephones and terminal equipment can be sold competitively.

Washington Scene

Well, since the election is so close, Congress went home for the year. They left a lot of work undone! Both S. 73, the Amateur Radio Spectrum Conservation Bill and S. 218, the Emerging Technologies Initiative died from inaction. Maybe next year they will really get to work -- especially if we see some new faces in Congress because of the election!

The FCC has been pretty busy lately, however. They gave Pioneer's Preferences to: American Personal Communications, Cox Enterprises of Atlanta and Omnipoint Communications of Colorado Springs (R. C. "Bob" Dixon's company). All of these companies were rewarded with an entitlement to a license in the market of their choice for PCN services.

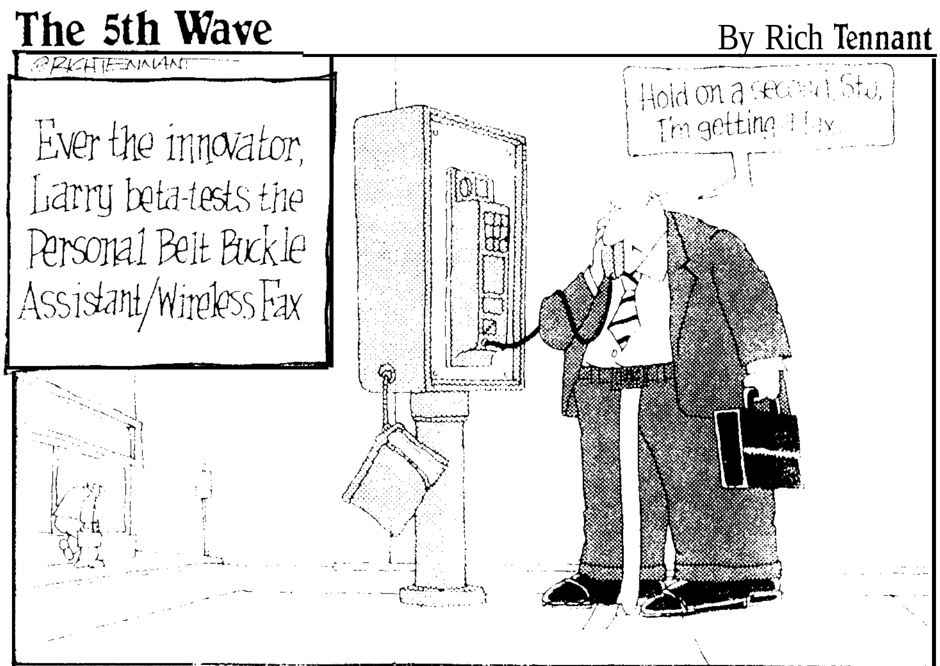
The FCC also issued a 97 page NPRM on the PCN/PCS services. So many companies are more than eager to enter this $35 to $40 Billion a year market (according to a recent Arthur D. Little report) that the FCC just can't wait for congressional direction and enabling legislation. Look for some of the current FCC commissioners to take jobs in the PCN industry, just after the election!

DSP
for Spread Spectrum
by Matthew Johnson

In my last column, I promised to write on HDTV. Unfortunately, I made it sound like I would cover much more in a single column than makes sense, so this will be the first of several columns on HDTV

So what is HDTV and what is it doing in a DSP column? HDTV is a generic name for any one of a number of schemes for transmitting a television signal with about twice the horizontal resolution and twice the vertical resolution of the NTSC system we use for color television in the US today. Since NTSC requires 6 MHz/channel (and that is with a very narrow guardband) doubling both the horizontal & vertical resolution demands 2x2x6 = 24

see DSP page 27



SECRET SPREAD SPECTRUM SIGNALS

In our last column we discussed some of the available literature on this fascinating subject. We found an absolute dearth of useful information about "Secret Signals." No reader has bothered to share any special information with us either.

Is there really anything that can be done to explore these signals we have seen and know they are on the air? The answer is Yes and No! Yes there is some general tutorial information that can be shared with our readers. But the answer is also no, because to give you detailed, how-to information on cracking these signals would be a Federal crime.

So, we come to the first real meat in this column: *How do you (theoretically) try to obtain useful information from "secret" spread spectrum signals, when you do not know the exact code being used?* The following information should be useful in answering this question -- any coded system (or cipher for that matter) can be attacked by a straightforward crypto-analytic attack, whether you know the code, key, or are authorized or not! Whether the attack is successful depends on many variables -- but, let us say that given patience, adequate computing power (and some customized software) and just a bit of luck you may find that today's commercial secret signals and even NBS standard encryption are merely child's play!

The basic principle to apply is called a "known clear text" crypto-analytic attack. The idea is very simple: somehow, you obtain a sample of "clear text" -- which could be a picture, a valid user ID code, an authentication key, a valid billing number / account number, etc. You may actually have to subscribe to the service for a short period to obtain the valid "clear text." However, once you have obtained a sufficiently long sample of "clear text" that you have reason to believe will be sent again sometime in the future - you are ready to attack the system.

Most commercial systems, re-key (or change keys/codes) infrequently, usually sometime in the late evening / early morning or even only once a week. So the same keys (read: spreading codes) are used for many hours or days. This presents the opportunity to record large amounts of encoded stuff for later analysis.

The analysis is very, very simple: you are looking for a repeat of your known "clear text." Of course it would help if you knew when the known "clear text" was going to be transmitted again -- but this is not absolutely necessary, is it?

You simply correlate linear samples of incoming demodulated data with your "clear text," if something that looks like a repeating, fixed code pops out, ***voila - you found it!*** Next month we will give a few more theoretical details.

Beginners Box

FEDERAL COMMUNICATIONS COMMISSION RULES - PART 15
Paragraph 15.249
OPERATION WITHIN THE BANDS 902-928 MHz, 2400-2483.5 MHz, 5725-5875 MHz, AND 24.0-24.25 GHz.

(a) The field strength of emissions from intentional radiators operated within these frequency bands shall comply with the following:

Fundamental frequency	Field strength of fundamental (millivolts/meter)
902-928 MHz	50
2400-24X3.5 MHz	50
5725-5875 MHz	SO
24.0-24.25 GHz	250

Fundamental frequency	Field strength of harmonics (millivolts/meter)
902-928 MHz	500
2400-24X3.5 MHz	500
5725-5875 MHz	500
24.0-24.25 GHz	2500

(b) Field strength limits are specified at a distance of 3 meters.

(c) Emissions radlated outside of the specified frequency bands, except for harmonics, shall be attenuated by at least SO dB below the level of the fundamental or to the general radiated emission limits In 15.209 whichever Is the lesser attenuation.

(d) As shown In 15.35(b), for frequencies above 1000 MHz, the above field strength limits are based on average limits. However, the peak field strength of any emission shall not exceed the maximum permitted average limits specified above by more than 20 dB under any condition of modulation.

(e) Parties considering the manufacture, Importation, marketing or operation of equipment under this section should also note the requirement In 15.37(d).

Thanks to:

Rules Service Company
Rockville, MD - (301) 424-9402

A 16Kbs Full Duplex Spread Spectrum RF Data Link

by Dan Doberstein, President DKD Instruments

This is the first in a series of columns describing a full duplex Direct Sequence RF data link using the cordless telephone chip set. Although this frequency band is not allocated for spread spectrum use for very low power uses it is legal and with a couple of mixers and associated LO's the frequencies used can be any that are currently allocated for Spread Spectrum.

The system block diagram is shown in figure 1. The MC145168 and the MC3362 form the heart of the system. The MC145 168 is a Dual PLL that locks both the transmitter VCO and the first LO of the receiver portion. The frequencies used for transmit and receive are burnt into the internal ROM. This means you cannot change the frequencies used for xmitt and receive to any arbitrary ones desired but will have to live with the ones provided for. A table of 16 transmit and receive pairs are available in the 45 to 49 Mhz band. The channel select inputs are used to select which channel pair you are on.

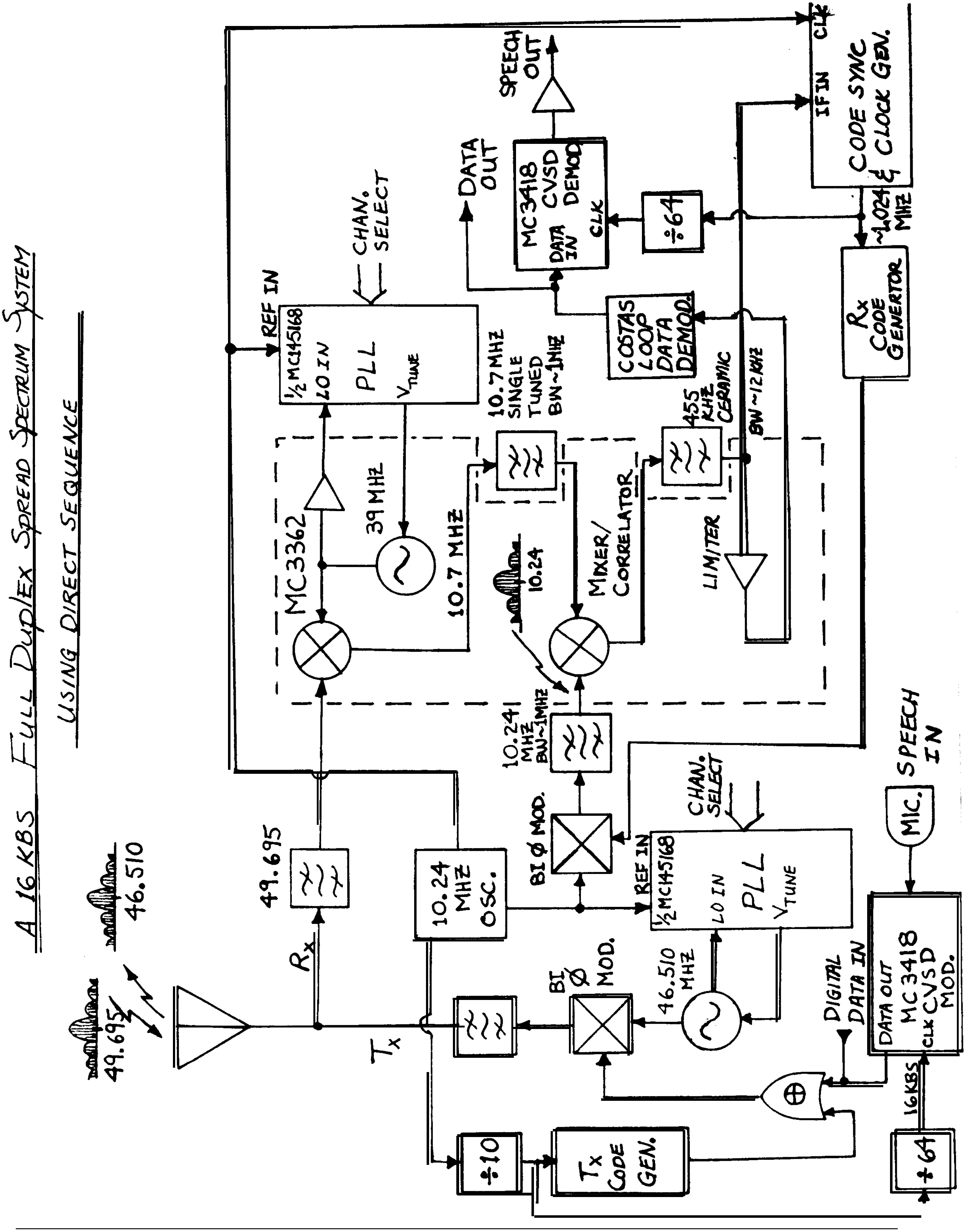
The MC3362 is a single chip dual conversion receiver that was designed for the cellular and cordless telephone applications. In this design it used to convert the received signal to the first IF of 10.7 Mhz and then to the 2nd IF of 455 KHz. A single tuned bandpass is used for the 1 st IF of 10.7 Mhz as a wide bandwidth is needed here due to the fact that signal is not despread yet. The chip rate of about 1 Mhz dictates the approximate 1 Mhz BW of this filter. The 10.7 is then mixed with the 10.24 Mhz LO which is Bi-phased modulated with a replica of the transmitted phase code. Assuming correlation and code lock the signal is now narrow band and is passed through the 455 Khz ceramic filter. This filter has about a 12 Khz BW. This is just wide enough to pass the 16Kbs digital data stream. the data stream can be user data or using the CVSD chip speech data. The data is bi-phase modulated onto the carrier after being exclusive OR'ed with the spreading code. After correlation the data can be recovered at the receiver end using the costas loop data demodulator.

The MC341 8 chip is a Continuously Variable Slope Demodulator/Modulator(CVSD). It takes in speech or other analog signals and converts them to a serial bit stream. You can choose the bit rate by changing the supplied clock. A bit rate of 16KBS is a little marginal but intelligible speech can be sent. The use of this chip reduces the bit rate for speech considerably over straight ahead application of sampling at say 8 bits every 125 usec. One of the features of this chip is its ability to modulate AND demodulate the serial bit stream.

One of the design goals for this system was to minimize the number of crystal sources used. This has cost as well as practical benefits. The PLL's, 2nd LO, Code Clocks and Data Clocks all are locked to or derived directly from one crystal oscillator at 10.24 Mhz. This multiple use of one frequency reference simplifies the hardware considerably, rather than having several xtal oscillators to worry about one good one is used. Another spin off of the entire direct sequence/coherent architecture is that the data clock is AUTOMATICALLY recovered once code lock is achieved. This simplifies the data recovery problem considerably.

There are two code generators for each station, one for the transmitted data and one for correlating with the incoming received signal. The 10.24 Mhz is divided by 10 for the transmitter code generator. The receiver code generator is clocked by the Code Sync and Clock Generator circuity. The code used for transmit WILL NOT be the same as for the received code,i.e. one code pair will be used for each station pair. The user will be able to specify which code pair is to be used by changing jumpers on the code generators. The use of different codes for xmit and receive helps with the rejection problems of a shared antenna.

The Code Sync and Clock Generator circuitry provides a modulated version of the IO.24 Mhz reference divided by 10 to give approximately a 1.024 Mhz code clock for the receiver code generator. The 1.024 Mhz clock is modulated in such a way as to keep the receivers code aligned with the incoming code from transmitter at other station. At the heart of the code sync method is the tried and true tau dither technique. We will discuss the details of the implementation of this technique next month.



A 16 KBS Full Duplex Spread Spectrum System
Using Direct Sequence
49.695
46.510
Rx
Tx
49.695
10.24 MHz OSC.
BI Ø MOD.
46.510 MHz
BI Ø MOD.
10.241 MHz BW~1MHz
CHAN. SELECT
REF IN
1/2 MC145168
LO IN
PLL
VTUNE
MIC.
SPEECH IN
DIGITAL DATA IN
DATA OUT
MC3418 CVSD MOD.
CLK
16KBS
÷64
÷10
Tx CODE GEN.
MC3362
39 MHz
10.7 MHz
10.24
MIXER/ CORRELATOR
10.7 MHz SINGLE TUNED BW~1MHz
455 KHz CERAMIC
BW~12 KHz
LIMITER
COSTAS LOOP DATA DEMOD.
DATA OUT
MC3418 CVSD DEMOD
DATA IN
CLK
÷64
SPEECH OUT
Rx CODE GENERATOR
~1.024 MHz
IF IN
CLK
CODE SYNC & CLOCK GEN.

Low power for mobile communications equipment

*The **UMA1014T** single-chip frequency synthesizer is a **low-**power device which provides a cost-effective solution to frequency synthesis in battery-powered mobile communications equipment – such as cellphone handsets, PMR (Private Mobile Radio) and pagers. It is an integral part of the Philips **chipset** for mobile communications equipment and complements the **company's** existing range of mobile radio **ICs** from Europe and the USA.*

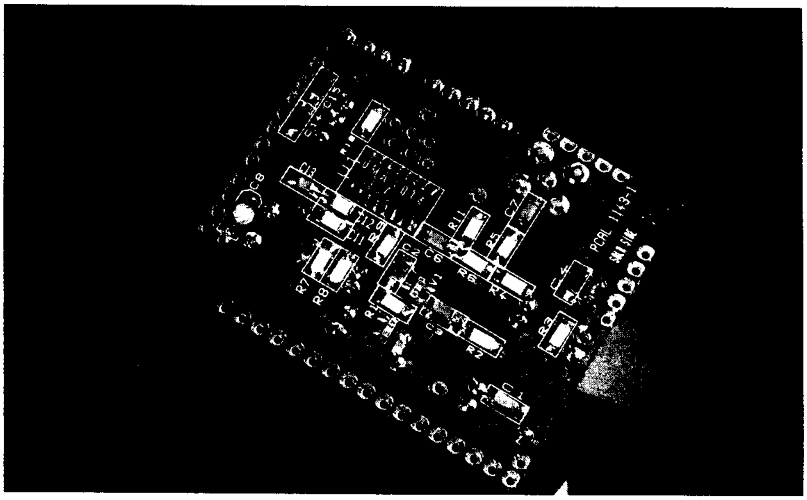
The UMA1 014T provides in a single chip all the functions required in an RF frequency synthesizer. Functions integrated into the device include an oscillator circuit, prescalers and programmable dividers. These are necessary to 'frequency divide' the reference and RF frequency inputs – thereby completely eliminating the interfacing problems associated with external prescaling. The divided down reference and RF frequencies are then compared in an on-chip phase/ frequency comparator. In addition, the UMA1014T has an I²C-bus interface so that it can be controlled by a system microcontroller.

The charge pump output of the comparator, which charges/discharges an external capacitor to provide the control voltage for a voltage-controlled RF oscillator (VCO). settles rapidly after frequency changes. Only minimal passive filtering is required between the comparator output and the VCO.

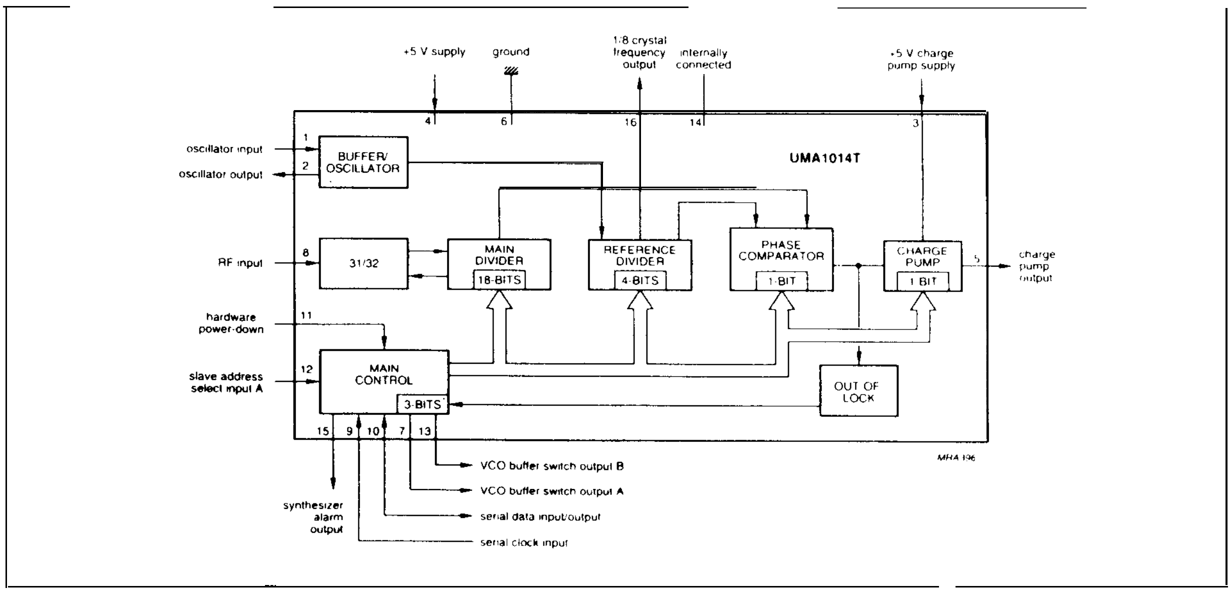
Finally, the excellent stability of the charge pump current-both to changes in temperature and output voltage – allows the UMA1014T to be used in accurate and stable phase-locked-loop circuits containing very few external components.

The UMA1014T is available 'in a 16-pin small outline plastic package and has a current consumption of 13 mA when active and only 2.5 mA when in power-down mode, which contributes significantly to extended battery life in portable equipment.

An extensive applrcations report describes the operation of the UMA1014T. It also gives detailed theory for calculating the values of the loop elements, describes applrcations and gives sample printed circuit board layouts.



***Providing a cost-effective solution to frequency synthesis** in **battery-powered** mobile communications **equipment – the** UMA **10**14T chip.*

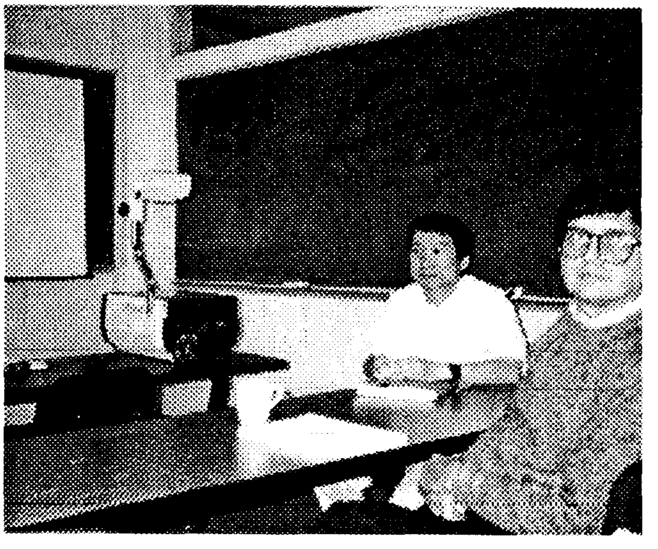


***Block diagram of the** UMA1014T single-chip **frequency** synthesizer*

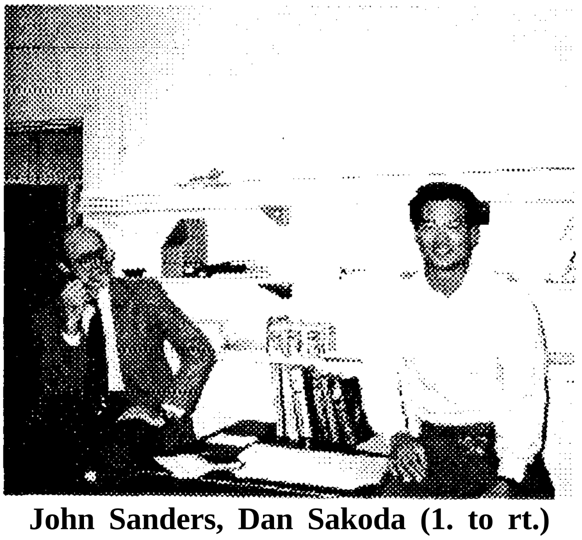
Acknowledgements for the NPS PANSAT Article Series

SSS wishes to gratefully acknowledge the help, encouragement and co-operation of Mr. John Sanders, Deputy Director of Public Affairs at NPS. Professors Rudy Panholzer, Ed Euler and Tri Ha were also extremely helpful in the preparation and review of these articles. Finally, we would like to thank Mr. Dan Sakoda, PANSAT Systems Engineer for showing us around the ultra-modern shop, lab and classroom facilities at NPS.

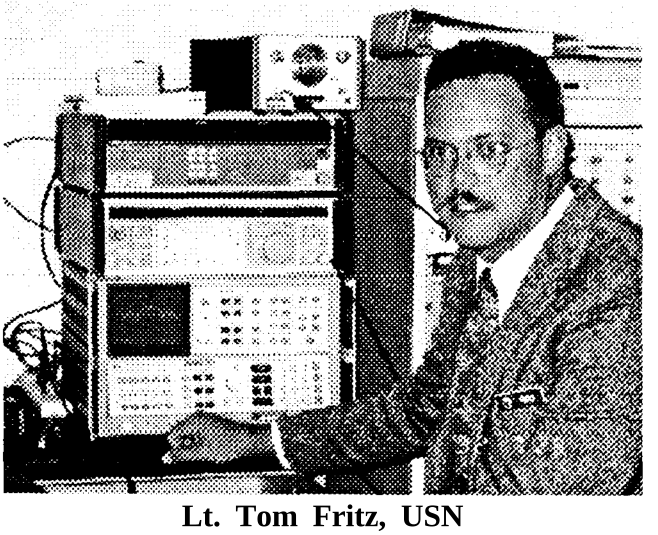
Prof. Rudy Panholzer (center).



Prof. Tri Ha, NPS grad student (1. to rt.)



John Sanders, Dan Sakoda (1. to rt.)



Lt. Tom Fritz, USN

Introduction to The Navy's PANSAT Project - Part 2

by Randy Roberts, KC6YJY & RF/SS Director

The author visited the Naval Postgraduate School (NPS) in Monterey, California on July 28, 1992. Randy was graciously hosted by Professor Rudy Panholzer, PANSAT Principal Investigator and his staff Most of our questions about PANSAT and how it will operate were answered at a weekly development/project status review meeting chaired by PANSAT Project Lead, Professor Ed Euler The following concluding article highlights the mission, current status and plans for the 1995 launch of PANSAT.

PANSAT Project Recap

PANSAT is the acronym for "Petite Amateur Navy Satellite". The Navy's experiment is listed as Experiment Number NPS-901. Last month we presented the basic details of the PANSAT project. This month we conclude the description of the Project. The basic concept of PANSAT is to store and forward a packetized uplink message until the destination station is in view of the satellite and then re-transmit the message on the downlink. Thus PANSAT will be functionally another Pacsat -- the big difference is the signals needed to access the satellite will be Spread Spectrum (SS), with direct sequence modulation. The primary military objective of the experiment is to significantly enhance the education of military officers in the NPS's Space Systems Curricula by the design, fabrication, testing and operation of a low-cost, low earth orbit (LEO), digital communications satellite.

The actual PANSAT signal design is still being studied and the ground station modem design is the subject of the following NPS graduate students' Masters Theses. The preliminary thesis titles are:

"BPSK SS Modem Design," Lt. T. Fritz, USN

"QPSK SS Modem Design," Lt. J. Mckinstry, USN

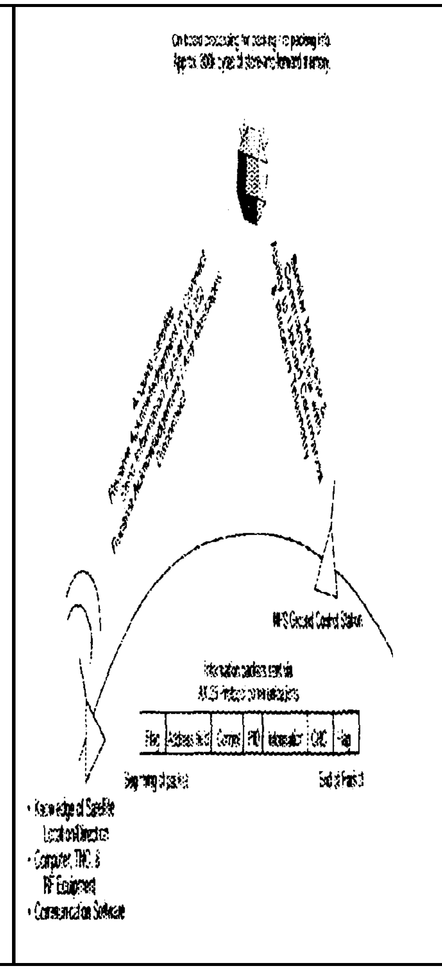
"4-FSK SS Modem Design," Lt. T. Murray, USN

"DPSK SS Modem Design," Lt. L. Rasnick, USN

The project is currently in the phases of design, analysis, parameter optimization and circuit breadboarding. A design review is being held this October to finalize many of the design details of all the satellite's subsystems.

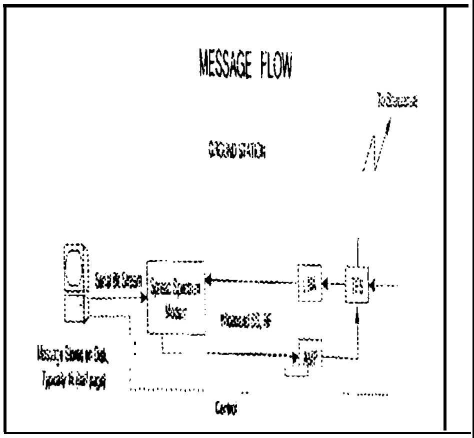
PANSAT DESIGN DETAILS

The major spacecraft subsystems include: (1) Structure and Configuration; (2) Power Control Unit; (3) Spacecraft Control Unit; and (4) Communication Control Unit.

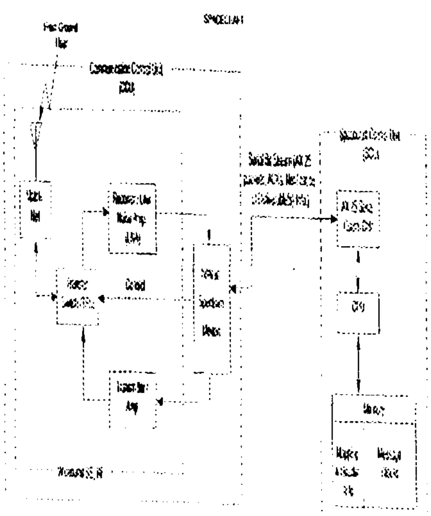


The Communication Control Unit (CCU) will utilize an 8088 micro-processor with watchdog timer and up-loaded commands. The communications payload will comply with current FCC part 97 rules for amateur radio SS operations and will provide 10^{-5} bit error rates and will utilize redundant (mutually exclusive) SS modems. The telemetry and command (TLM/TCM) functions will use a simple non-spread BPSK modem (NPS use only).

The PANSAT solar cell power generation subsystem will output 24.5 Watts, minimum average electrical power at End of Life, even with a tumbling spacecraft. The onboard power bus will be an unregulated 12 Volts.



The physical satellite will be a 26-sided polyhedron with 17 solar sub-panels. The PANSAT is sized for a Space Shuttle GAS launch, but is amenable to launches on vehicles of opportunity including: Pegasus, Delta II, Titan II and Scout.

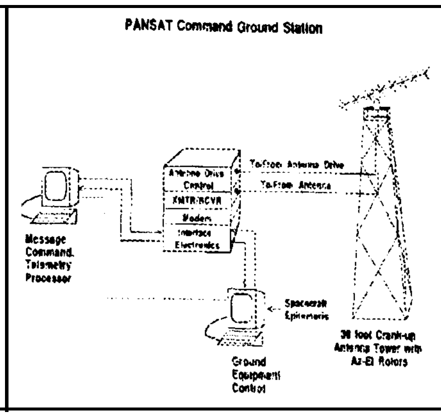


A 5 Watt (+7 dBW) satellite transmitter with a 0 dBi satellite antenna is envisioned. The ground based user station can use a 1 Watt (0 dBW) transmitter with a 1/4 wave ground plane (+2.15 dBi) or a 1/2 wave crossed dipole turnstile array would be even better.

The satellite antenna will be a four element Tangential Turnstile with circular polarization. The pattern simulations so far suggest nulls less than 3 dB deep and polarization losses less than 3 dB are possible. The 26 sided PANSAT will continually tumble, and any more complicated antenna than this 4 element circular one is just not feasible in a low cost satellite.

PANSAT Ground Segment

The figure above shows the preliminary concept for NPS's ground control facility for PANSAT. It is not much different than a well equipped amateur radio OSCAR 13 station. NPS will however, be the only ground control station for PANSAT.



During this author's visit to NPS, Lt. Russell Gottfiied USN, presented the preliminary results of his network communication model simulation and described the results of an informal amateur radio poll taken by the local radio club. Several key parameters of PANSAT's design (including onboard memory size, average message length and processor loading) are a function of the- user traffic loading. Further, the very nature of an SS demod-remod satellite communication payload makes certain limits as to how many users can access the bird during any given pass. In fact, the bird may appear to be blocked for traffic if too many ground users try to access PANSAT at one time. Since only a single user "connects" to PANSAT at a time, queuing delays of up to several seconds are possible with even moderate traffic loadings. Lt. Gottfi-ied did an excellent job of modelling and simulating all of these complex interactions. His main findings are that messages should be a maximum of 2 to 4 Kilobytes long and that simultaneous use by a world wide ham community of 200 to 400 users should provide optimum PANSAT performance.

Where will 200 to 400 space equipped, spread spectrum modem equipped hams come from? Inherent in the research into modem techniques being done at NPS is the desire to design a simple, low cost ground user modem/RF package. This author will follow these developments and report on them in the future. NPS wants to release to the public domain, all information necessary to build ground user station modem/RF equipment for PANSAT. Perhaps SSS or another magazine will publish these details and even offer a kit for potential PANSAT enthusiasts. In the mean time, start thinking about spread spectrum, learn something about it and get ready to enter Ham Radio's 21st century.

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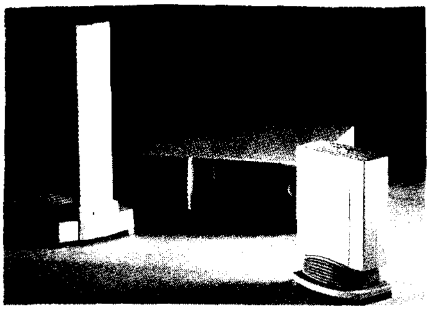
More New Products



FreePort operates seamlessly with existing wired LA Ns.

Contact: Windata, Inc.. IO Bearfoot Rd., Northborough. MA 01532, (508)393-3330: fax (508) 393-3694.

FirLAN, a IO-Mbps wireless Ethernet LAN, consists of two tranceivers and a hub. Able to run transparentl: across all Ethernet platforms and media, FirLAN allows seamless integration into existing wired LANs.

The ET300 transceiver has a 90-foot range and supports up to 29 Ethernet connections. An extended-range ET330 transceiver has a range of 300 feet. The EH360 hub covers a 90-degree area and supports as many as 250 transceivers.

Communications, Inc., 102-21 Antares Dr., Ottawa, Ontario. Canada K2E 7T8, (613) 723-l 103; fax (613) 723-6895.

The Granite Links system is based on a hand-held personal data communicator.

µP vs. DSP - What's the Difference?

The main distinction between a general purpose microprocessor (Intel 808x, 80x86, Motorola 680xx, etc.) and a DSP engine (TI TMS320xx, Motorola DSP560xx, AT&T DSP-xx, etc.) is replacing the microprocessor's built-in software routines (microcode) for numerical calculations with dedicated hardware circuits. This makes math computations much faster.

For example, a microcoded multiply requires 12-41 clock cycles in an Intel 80386. A TMS32025 performs the same multiply in l-cycle, thanks to a built-in modified Booth algorithm multiplier circuit.

DSP chips add extra buses, registers, and other goodies to further speed up the math. The performance improvements are so dramatic that microprocessor manufacturers are making their new µP chips more and more "DSP-like".

DSP chip makers, pressured by more sophisticated applications and the demand for high-level language support, are making their devices more and more "VP-like" by including flexible memory addressing and management, once the domain of the µPs.

The development of faster pipelined RISC microprocessors on the one hand, and sophisticated floating-point DSP chips on the other hand, means that the distinction between µP and DSP will grow smaller as time goes on.

*(If all of this confuses you, read the tutorial on DSP Engine Architectures, **scheduled for** our next issue.)*

The TMS320E15 at a Glance.

The **Texas Instruments TMS320E15 is an EPROM (electrically-programmable read-only memory) version of the f&-generation TMS32010 DSP engine. The part was introduced in 1987.**

Arithmetic: Fixed-point.

Precision:	**13/16-bit multiply (MPYK/MPY), for .012%/.0015% accuracy. 32-bit accumulator.**
Speed:	**160-nS instruction cycle (6.25 MIPS) with 25-MHz clock (TMS320E15-25).**
Memory:	**4096 16-bits on-chip program &ROM, 256 x 16-bits on-chip data RAM, and 4096 x 16-bits external program or data memory.**
Price:	**$120/ $86.10 (US) (qty 1/ 1000) for 25-MHz version.**

(Thanks to HamPute, May/June 1992)

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DSP Tutorial

Modulation, Fourier Series, and Sampling Theory

Some Ground Rules.

One of the horrors of using a new technology is that you have to learn some new theory to know what you're doing. DSP theory, in its "pure" form can be very complicated. The derivation of DSP involves the kind of heavy-duty mathematics that the "Einsteins" of this world really enjoy.

Good for them, bad for us.

As it turns out, much of the theory behind what we do in amateur radio involves some heavy math. Fortunately, most of it can be boiled down to some fairly simple "rules of thumb" that make the technology useful to us. We didn't have to study electron ballistics in order to use vacuum tubes. Understandiig the physics of hole and electron migration, and the Ebers-Moll equations, wasn't really necessary to bias a transistor. This theory has been "canned" into forms that we can use. So it is with DSP.

Of course, by streamlining the theory you lose the ability to rigorously prove things as you present them. Many items have to be accepted on "faith". While this drives engineers and mathematicians crazy, most hams have learned to live with it. *The ARRL Handbook* has done a wonderful job of streamlining theory for amateur use. We hope to follow in the same spirit.

DSP replaces traditional components by mathematical equations. As a result, the math underpinning of DSP can't be completely eliminated

---

We will assume that you can remember a little of your high-school algebra and trigonometry. We will jog your memory as we go. Beyond that, we will spare you the rigorous mathematical proofs. Hams learn by doing. We want you to be "doing" DSP as quickly as possible.

Let's begin . . .

*The **Road** to Sampling Theory.*

Sampling theory is what links the analog and digital worlds. We *could* just present it in "here it is" fashion, but we won't. The concepts behind it are too important to gloss over. Instead we will build up to it, step-by-step, so that you will have a "feel" for it. So that you can *use* DSP, not just parrot the buzzwords.

If You Can't Handle It, Map It!

Mathematics is a tool that we use to represent the world around us. It is a language. A way to look at things. If a given mathematical "environment", or *domain* becomes awkward when dealing with a physical reality, we are perfectly free to re-defme the domain and look at the reality through different eyes. We ***transform*** the problem from one domain to the other, or *map* it. Mathematicians have been doing this for centuries.

As amateurs, we use an oscilloscope to look at signals in the *time domain. The* 'scope plots signal amplitude versus time. Unfortunately, things like harmonic distortion and sideband splatter are hard to see on an oscilloscope. Instead, we use a spectrum analyzer to look at the signals in the *frequency domain.* Were looking at the same signal, only through different eyes (*See Figure 1*).

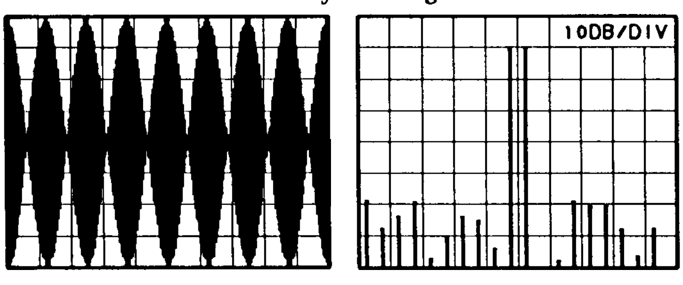

*Figure 1. **Two-tone** SSB with 5% **Crossover***

We all learned mapping in high school. Remember the quadratic equation? It is shown as equation pair (1) and (2). Everything was rosy until b^2-$4ac$ became less than zero! How do you take the square root of a negative number? You map the problem to a new domain, of course!

$$a \cdot x^2 + b \cdot x + c = 0 \qquad (1)$$

$$x = \frac{-b \pm \sqrt{b^2 - 4 \cdot a \cdot c}}{2 \cdot a} \qquad (2)$$

Imagine if there were a square root of -1. We know it doesn't really exist, but just imagine. Let's defme this *imaginary number **i** as the square* root of -1, whatever that is. We could then map all possible solutions to the quadratic equation in this new domain of real *and imaginary* numbers. Put the real numbers along a horizontal axis, and imaginary along a vertical axis,

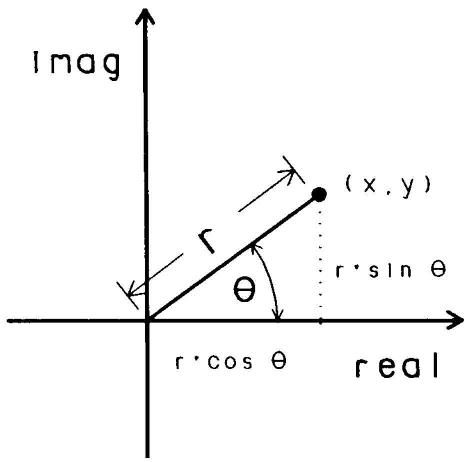

Figure 2 Coordinate Mapping

and you have *rectangular or Cartesian awrdinatee,* as shown in *Figure 2.* Every number can be represented as a real part and an imaginary part:

c = (real part) + *i*·(imaginary part)

These numbers are also called *complex **numbers**.* "*i*" is called the *imaginary operator.* Since we use "*i*" to represent current in electronics, we can use "*j*" as the imaginary operator to avoid confusion.

We found out in high school that dealing with rectangular coordinates got haii when doing things like multiplying and dividing. Solution: map it again!

We know that multiplying and dividing numbers with exponents is easy (you just add or subtract the exponents). 18th century Swiss mathematician Leonhard Euler proved that:

$$e^{j\theta} = \cos(\theta) + j\cdot\sin(\theta) \qquad (3)$$

This equation is known as *Euler's identity*. If we draw a line from the origin (0,0) to our number, we can represent it as a ray "*r*", or distance from the origin, and an angle with the horizontal x-axis "θ". We have *polar coordinates*. The trigonometry involved is shown in Figure 2. $e^{j\theta}$ is the polar operator.

Polar representation is widely used in signal processing. "*r*" is the amplitude of the signal, while "θ" is the phase at any instant in time. In case you forgot? "*e*" is the base of the natural logarithmic system, and equals 2.718281828... Remember that the different domains represent the same thing (the signal). This is all just mathematical smoke and mirrors!

Negative Frequencies?

In Euler's identity, θ represents a fixed phase angle. Each complete rotation, or cycle, represents 360-degrees, *or* 2·π radians. To represent a continuous signal, we express it in the form:

$$A\cdot e^{j\omega t}\text{, where} \qquad (4)$$

A = the amplitude of the signal

$\omega = 2\cdot\pi\cdot f$
= the frequency in radians/second

t = time

Okay, so far so good. But look what happens when we solve Euler's equation for cos(ωt), which is a simple sine wave:

$$\cos(\omega t) = \frac{e^{j\omega t} + e^{-j\omega t}}{2} \qquad (5)$$

, where

$$e^{-j\omega t} = \cos(\omega t) - j\cdot\sin(\omega t) \qquad (6)$$

Equation (6) is the *complex conjugate* of $e^{j\omega t}$. It represents a *negative* frequency! What on earth is a negative frequency? Nobody really knows. It's just like an imaginary number. Mathematical smoke and mirrors. But it makes the math work.

Let's illustrate by looking at something more familiar:

Mixing and Simple Modulation.

How do you build a mixer in DSP? *Just multiply!* *No* toroids, no hot carrier diodes, no FETs, simply multiply the signal by a carrier, and you're done.

Let's follow this one case through to illustrate how negative frequencies work in the math. Remember *from* algebra that you derived things by substituting, expanding all of the terms, collecting terms, and substituting back:

$$\cos\left[\omega_c\cdot t\right]\cdot\cos\left[\omega_s\cdot t\right] =$$

$$\frac{e^{j\cdot\omega_c\cdot t} + e^{-j\cdot\omega_c\cdot t}}{2}\cdot\frac{e^{j\cdot\omega_s\cdot t} + e^{-j\cdot\omega_s\cdot t}}{2} =$$

$$\frac{e^{j\left[\omega_c+\omega_s\right]t} + e^{-j\left[\omega_c+\omega_s\right]t}}{4} + \frac{e^{j\left[\omega_c-\omega_s\right]t} + e^{-j\left[\omega_c-\omega_s\right]t}}{4} =$$

$$\frac{\cos\left[\left[\omega_c+\omega_s\right]t\right]}{2}\ \text{(USB)} + \frac{\cos\left[\left[\omega_c-\omega_s\right]t\right]}{2}\ \text{(LSB)} \qquad (7)$$

Equation (7) is the *modulation equation*. It states, mathematically, that a perfect mixer produces the sum and difference frequencies of the incoming signals. The negative frequency components generate the difference frequencies.

Our DSP mixer is a double-sideband (DSB) modulator. The signal frequency ω_s is the baseband audio component.

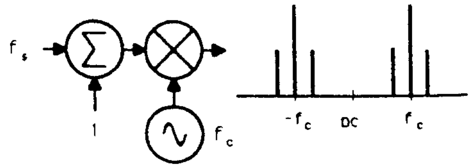

Figure 3. AM Generation

A DSP multiply is equivalent to a perfect "double-balanced" mixer. No input components appear at the output!

How do you make AM? Add DC. As shown in *Figure 3* and equation (8), the result is ideal 100% modulation. This is what Kenwood does in the TS-950SD.

$$\left[1 + \cos\left[\omega_s\cdot t\right]\right]\cdot\cos\left[\omega_c\cdot t\right] =$$

$$\cos\left[\omega_c t\right] + \ \text{(CARRIER)}$$

$$\frac{\cos\left[\left[\omega_c+\omega_s\right]t\right]}{2} + \ \text{(USB)}$$

$$\frac{\cos\left[\left[\omega_c-\omega_s\right]t\right]}{2}\ \text{(LSB)} \qquad (8)$$

We'll *cover* other modulation techniques in a future tutorial. For now, let's get back on the road to sampling theory.

Linear Systems: Superposition.

So far, we've demonstrated how to process a carrier and a sinusoidal signal component. But actual signals contain *many* components. Doesn't this complicate things? Not if the *system* is *linear.*

A *system* is any combination of signal processing functions. It can include mixers, filters, detectors, what have you. *Linear systems*, by definition, adhere to the principle of *superposition.*

Superposition means that if signals are added together at the input, the output will be the sum of the system's response to each of the input signals. We can superimpose the individual responses. *There is no interaction among the signals.*

A ham receiver, for example, is supposed to be a linear system. Its input consists of signals throughout the radio spectrum. It's output response to a particular signal should not be distorted by other signals in the spectrum. There is no interaction *(intermod) among* the incoming signals. When a receiver stage overloads, intermod occurs and the system is, by definition, non-linear.

Mixers, filters, phase-shift networks, and amplifiers are all linear, so long as there is no clipping or compression. Detectors may or may not be linear. A diode, for example, is not linear. You may be able to restore linearity, however, if you can filter out the intermod components.

Luckily, most DSP involves linear circuits. Superposition applies. We can treat each component of a signal independently of all the other components, and then add the individual results to get the total output response. Divide and conquer. Now if we only knew how to process waveforms other than sinusoids ...

The Fourier Series.

Jean Baptiste Joseph Fourier to the rescue! This 18th century French mathematician devoted much of his career to mathematically linking the time and frequency domains. He proved that any periodic, or repeating, waveform is equivalent to a sum of sines and cosines:

$$f(t) = \frac{a_0}{2} + \sum_n a_n \cdot \cos\left[\frac{2\cdot\pi\cdot n\cdot t}{P}\right] + \sum_n b_n \cdot \sin\left[\frac{2\cdot\pi\cdot n\cdot t}{P}\right] \qquad (9)$$

,where $n = 1, 2, \ldots, \infty$ and:

$$a_n = \frac{2}{P}\int_0^P f(t)\cdot\cos\left[\frac{2\cdot\pi\cdot n\cdot t}{P}\right] dt$$

$$b_n = \frac{2}{P}\int_0^P f(t)\cdot\sin\left[\frac{2\cdot\pi\cdot n\cdot t}{P}\right] dt$$

The $\sum$ symbol in equation (9) is just a shorthand notation for adding a large number of terms together. We'll elaborate on this notation in future tutorials. The $\int$ symbol represents integration (from calculus). You don't have to learn calculus to use the Fourier series, since the series for a wide variety of waveforms have been tabulated in a number of references. In the equations, P represents the period of the waveform, and t is time.

Figure 4 illustrates how a periodic

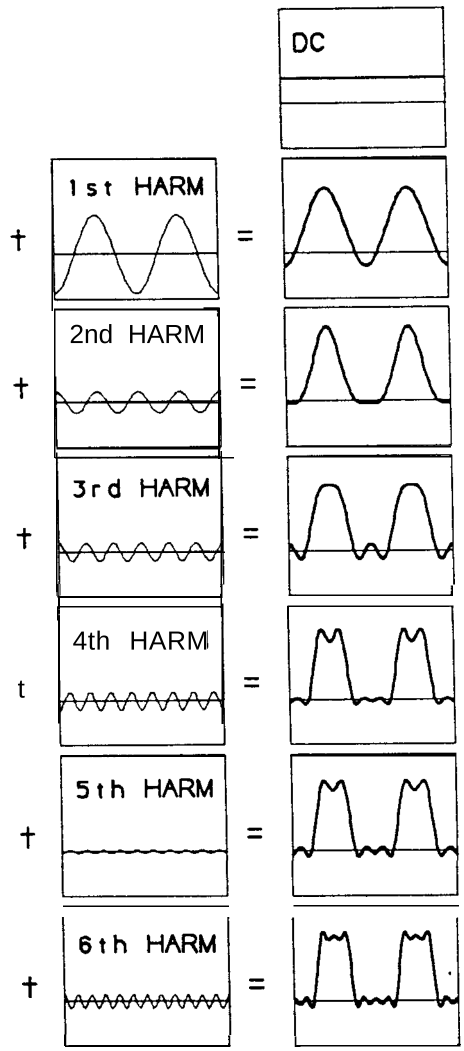

*Figure 4. **Constructing** a Periodic Pulse Train from **Sinusoids***

waveform can be constructed by adding sinusoidal components. This should give you an intuitive "feel" for what's going on.

The fundamental frequency ("first" harmonic) of the sequence is always the inverse of the waveform period (Z/P). Which harmonics you get depends on the *symmetry* of the

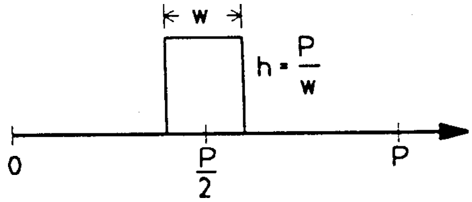

Figure 5. *Unit **Pulses***
(Power = 1.0)

waveform. You may also get a DC term, depending on symmetry.

Let's look more closely at one special case: a "unit" pulse train *(Figure 5)*, with period P, width w, and area/period (power) = 1. The Fourier series for this waveform is:

$$1 + 2\cdot\sum_n (-1)^n \cdot \frac{\sin\left[\frac{\pi n w}{P}\right]}{\frac{\pi n w}{P}} \cdot \cos\left[\frac{2\pi n t}{P}\right]$$

$$\text{,if amplitude} = \frac{P}{w} \qquad (10)$$

Remember that $(-1)^n$ equals +1 if n is even, and -1 if n is odd. The spectrum "envelope" follows the form sin(X)/X, where X is $[\pi\cdot n\cdot w/P]$ in this case. This envelope is important whenever "wide" pulses are used in DSP. The frequency rolloff must be corrected in this case. $\sin(\pi X)/(\pi X)$ is plotted in *Figure 6.*

As the pulse width gets narrower, the sin(X)/X envelope broadens. The extreme case is when the pulse width is "zero". Periodic zero-width pulses are called an ***impulse** train.* With the help of some limit theory (*don't worry* about it) equation (10) becomes:

$$1 + 2\cdot\sum_n (-1)^n \cdot\cos\left[\frac{2\pi n t}{P}\right] \qquad (11)$$

but equation (5) tells us that the $\cos(2\cdot\pi\cdot n\cdot t/P)$ equals a positive and negative frequency component, each of amplitude 1/2. Result:

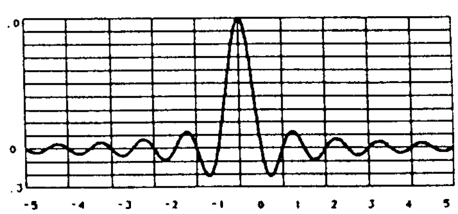

*Figure 6. **Sin(πX)/(πX)***

A periodic impulse tmin contains all harmonic components, including a LX term, and has a "flat" spectrum envelope: all components have the same amplitude.

Figure 7 shows this graphically.

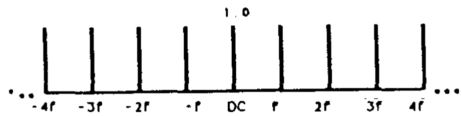

Figure 7. Unit ***Impulse Train*** *Spectrum*

Recap.

Let's recall where we have traveled thus far ..

- We reviewed the concept of mapping and domains, and transforming problems from one domain to another using operators.

*We looked at rectangular and polar coordinates as ways to map signals.

- Euler's identity introduces the concept of negative frequencies.

*We derived the modulation equation as a practical application of Euler's approach. This is the first trigonometric identity in our "bag of tricks".

- We learned that we can work with complex signals one frequency component at a time, thanks to superposition.

- The Fourier equations allow us to represent any periodic signal as a sum of sinusoids.

- One very special case is the periodic impulse train.

- All of the above applies equally well to analog or digital signals.

We told you all of that so that we can tell you this . . .

Sampling Theory

An analog signal is *continuous:* there are an infinite number of points on the waveform, each having an infinite number of possible values. Digital computers can only handle *discrete signal8* (finite number of points, or ***samples***), With ***quantized levels*** (finite number of possible values). How do we bridge this gap?

Sampling theory was developed in the 1930s and 1940s to determine what happens when you represent a continuous signal by a finite number of samples. We'll deal with the effects of quantizing the signal levels in a future tutorial.

H. Nyquist proved that *if a bandlimited analog signal is sampled at a periodic rate that is at least twice the highest frequency contained in the signal, the analog signal can be perfectly reconstructed by an ideal lowpass filter.* This is known a8 the *sampling theorem,* and the minimum sample rate is called the *Nyquist rate.*

The sampling theorem says that a sequence of numbers can contain all of the information we need to process and reconstruct the signal. The infinite number of points between the sampling points contain *redundant* information!

Sampling means that we extract the signal's value at discrete points in time *(Figure 8).* ***This is exactly*** *the same a8 multiplying the* ***signal*** *by* a *periodic impulse* ***train.*** We know the spectrum of a periodic impulse train, we know the modulation equation, and we know about superposition. Therefore:

> ***When a signal is sampled at a periodic rate, the resulting spectrum is obtained by adding and subtracting all frequency*** *components within the eignal* to *all harmonic8 of the sampling rate (including a* ***DC*** *term).*

Figure 9 illustrates the sampling theorem. The image components, created by mixing the baseband signal and the impulse train, are called ***aliasing*** by DSP engineers. Signal components are *aliased to*

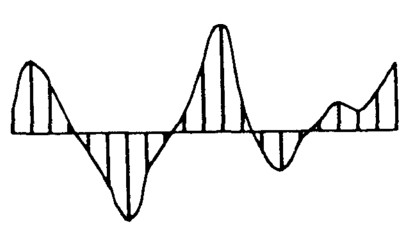

Figure 8. Sampling

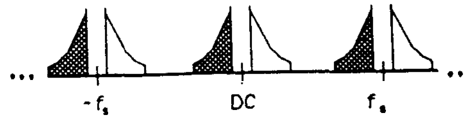

Figure 9. ***Nyquist*** *Sampling*

other frequencies (just like a criminal aliases hi name when he skips town). Aliasing is the same as images in your receiver, or the slow-motion effect obtained by viewing high-speed action with a strobe light.

The Nyquist theorem requires the sampling rate to be high enough to avoid overlap of the aliased spectra. Then you can reconstruct the signal with a lowpass filter.

For ham applications, the Nyquist rate is sometimes overkill. *Figure 10* shows what happens if you sample *narrowband* signals at a low sampling rate. The spectra are interleaved, but they might not

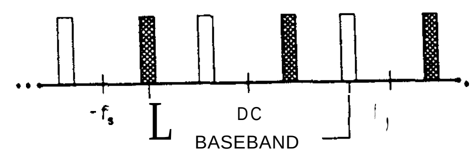

Figure 10. Narrvwband ***Sampling***

overlap. You could then reconstruct the baseband signal with suitable *bandpass* filters. This is what Kenwood does in the TS-950SD.

One word of caution: signal "bandwidth" here is considered to be the width between the *stopband* (typically -40 dB to -80 dB) points, not the passband (-6 dB) width! As you work with DSP, you are free to change the sampling rate as *long a8 you* do not allow the stopband points to overlap. There are advantages in doing this . . .

- NX6R

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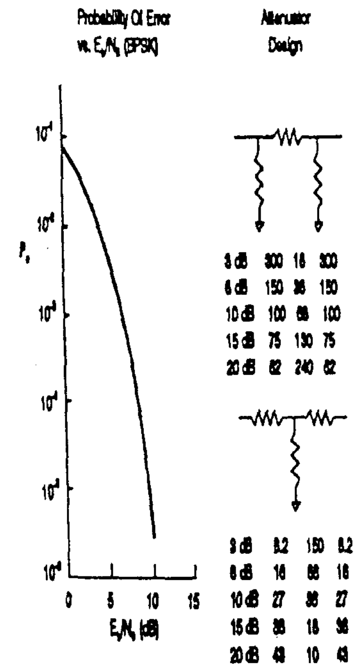
DSP from page 5 - MHz if we were to stick with the NTSC scheme. This is clearly unacceptable, so much effort has been expended in trying to reduce the bandwidth requirement, with some success.

These efforts have yielded two basic classes of schemes: analog, such as Japan's MUSE, and digtal, such as the four US proposals among the five schemes currently under FCC consideration for the US HDTV standard. The fifth is a narrow bandwidth version of MUSE (6MHz). It would be embarassing for the US electronics industry if MUSE wins the competition! The digital schemes are Zenith & AT&T's DSC-HDTV, General Instruments DigiCipher, Sarnoff s ADTV, ATVA and an MIT proposal.

This column will discuss only the MUSE system, since it allows me to develop the terminology needed to describe the all-digital schemes, and there is already a family of DSP processor chips tailored specifically for MUSE on the market. I should then be in a good position to explain why the FCC has taken a lot of flak from some very professional engineers for hastening the entire approval process with significant bias for all digital schemes.

Next month I will present a block diagram of the processes discussed here. I will also give more details about today's digital HDTV schemes and how they fit into continuing political/FCC approval battleground.

Beginner's Box



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Education from page 4 - between technicians and engineers really changed all that much in the past three decades?

I recently had the opportunity to review a new book on this subject which is due to be released in November. Entitled Becoming An Electronics Technician: Securing Your High Tech Future by Ronald Reis, this new book describes the electronics industry as it exists today. It explores the background and history of the electronics field, current avenues or research and development and the trends that will shape the future of the industry. Of particular interest is a chapter which describes the technician in the workplace of the nineties with respect to the electronics engineer, technologist and assembler. It shows students how the career they've chosen fits into the overall workplace picture and what type of relationships they can expect to find once they're on the job. This book gives a clear, concise look at the electronics industry, how to prepare for that first job as a technician, and most importantly, how to continue technical growth to stay abreast of new trends and developments. The book is being published by Merrill Publishing, of Macmillan Publishing, New York.

Next month I will explore current trends in education across the country which address the incorporation of technical preparation courses for **all students.** Stay tuned!

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November SSS Preview

Conclusion: Conclusion of Dan Doberstein's Idea Article on a General Purpose SS Radio (GPSSR) for commercial, industrial or ham usc, designed around the 49 MHz cordless telephone chips.

Interesting: More on DSP, "Secret" Signals, The Aerial and Technical Education in our columns.

Tutorial: More hands-on tutorial materaii in Technical Tricks and the feature article on low power 915 MHz links.

News: Latest news on Spread Spectrum regulatory, new product and marketing developments.

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