Update information
and complete list of files for this site

My name is Leif Åsbrink. Since 1961, I have been a radio amateur, and all the time my interest has been DX on 144 MHz, and in particular equipment designed for this purpose. On this site I try to share some of my experiences in designing and building equipment for VHF long distance communication.

The primary location is at http://www.sm5bsz.com/index.htm
There are two complete mirror sites. One in Europe http://www.g7rau.co.uk/sm5bsz/index.htm supplied by G7RAU Dave and one in the USA http://nitehawk.com/sm5bsz/index.htm supplied by W6/PA0ZN Rein.

The digital revolution. New possibilities for the experimenter.

In the old days (before about 1970) amateurs built their own equipment. Experimenting was a natural part of the hobby - inspired by what others did one had to try to use the parts from one's own "junk box" to do something similar or hopefully better.

With the commercial availability of the modern amateur SSB transceiver, the need for experimenting disappeared. It is not easy to design and build equipment that can compete with the commercial units. The best way to get a well performing station has long been to buy a SSB transceiver. Only very few real enthusiasts have been building their own receivers and transmitters.

In the new century, after about year 2005, the situation is becoming different. By use of simple equipment, well suited for home building, the radio signal can be moved into the digital world. Once the signal is available in digital form, a whole new world of experimentation is opened. The computer can do everything we did before using analog circuits. Experimenting with different filter characteristics, AGC (automatic gain control), AFC (automatic frequency control) can be done in software at no cost (except the experimenters time.)

The new possibilities in interference reduction, treating different kinds of interference as signals, each one received with a digital receiver that optimises the S/N for the particular interference opens a whole new world of experiments with radio recei vers that will allow reception of signals that can not be received at all with a conventional SSB transceiver.

Linrad is a free computer program that works on standard PC computers. (IBM-compatible, x86) Linrad was developed under the Linux operating system but it is available also for Microsoft Windows as well as for FreeBSD. The same package of source code files will compile under all the different operating systems to produce an executable file. (The .exe file for Windows can be downloaded directly.)

Use this link if you are a newcomer to Linrad: Linrad for newcomers. The other links about Linrad contain very much information and may be hard to understand without having spent quite some time with Linrad.

Linrad receives one or two IF passbands in digital form. Performance is determined by the hardware; the program analyzes whatever data supplied in digital form. Linrad can be used to process the analog audio output from a conventional SSB radio or any other linear receiver with a bandwidth that can be handled by the soundcard of the computer. With two conventional receivers having common oscillators Linrad can process two analog channels with a stereo sound card and use them for adaptive polarization or adaptive beam forming. Linrad can also use two audio channels (stereo) to process the I/Q signal pair produced by a quadrature mixer in a direct conversion radio. In this case four audio channels are needed for 2 IF channels.

Linrad was from start, before year 2000, designed for use with radio A/D converters although such hardware was not yet available at an affordable cost. The first commercial such hardware designed for amateurs was the SDR-14 which appeared in 2004. Today there is a large number of direct conversion receivers. This link SDR Hardware Tested with Linrad gives data for SDR-14, Perseus, G31DDC, SDR-IP, ELAD, WSE, Softrock and others.

The WSE converters were developed before any direct sampling hardware was available. Linrad can be used as a high performance spectrum analyzer with the WSE units as well as with any other high performance hardware. This link some high resolution spectra demonstrates the capabilities of WSE/Linrad as a narrowband spectrum analyzer in the year 2004. Transatlantic carriers on 1030 kHz shows high resolution waterfall graphs captured with a whip antenna in Eskilstune (near Stockholm) Sweden in the night Sept 03 to Sept 04 2010. One can see the carriers of about 20 different stations, but not even the strongest one is strong enough to allow AM detection.

It is convenient to have a set of reference files containing the radio signals in digital form on the hard disk. Such files can be used for optimisation of algorithms as well as for comparing different SDR packages. The SDR data file library contains data files with various mixtures of signals and interferences as well as typical loudspeaker output files from Linrad as well as from other SDR packages to the extent others have made them available to me. This link Missing SDR recordings has screen dumps and other info about interference problems that could be solved by use of appropriate SDR algorithms but where suitable recordings have not been made available.

Fundamentally there is no difference between analog and digital receivers. Both kinds have the same fundamental problems with dynamic range and spur suppression. All the new methods for combatting interference have their equivalent analog counterparts.

Here is a general discussion on radio receivers which is intended to resolve common misunderstandings and to explain how one can make sure that a receive system is properly optimized. The well known problems of low noise and dynamic range are present in digital as well as analog circuits.

A software defined radio like Linrad, Winrad and many others have a common problem in that the input data stream is not derived from the same crystal oscillator as the loudspeaker output. It is necessary to use a variable resampling rate to accommodate for the frequency deviation from the nominal rate. This link frequency stability of the loudspeaker output in Linrad, Winrad and Perseus shows the frequency stability of the loudspeaker output when a very stable signal is sent into a Perseus HF receiver.

Starting with version 02.36 Linrad has a transmitter as well as a receiver. The Linrad transmitter and receiver are intended to be operated simultaneously and the operator should be able to listen while transmitting in SSB mode because the the transmitter is muted during those short intervals in the speech when the voice level is low. The Linrad transmitter is currently in an early experimental stage.

Comparing different SDR hardware.

The conventional way of comparing different radio receivers is to make measurements of various properties like MDS (minimum discernible signal, directly related to noise floor or NF = noise figure.) Dynamic range (many definitions exist.) Fequency stability, image rejection and many other properties that might be relevant to users.

With SDR technology, some properties depend on the hardware while other depend on the software. The fundamental properties of various SDR hardware can be demonstrated by running several hardware simultaneously with the same input signal in simultaneously executing independent instances of Linrad that run on the same computer. Differences between hardware become directly visible on the screen. In Oct 2010 Linrad can use SDR-14, SDR-IQ, Perseus, SDR-IP, Excalibur and systems that use a soundcard. Starting with version 03-24 (Aug 2011) Linrad can also use the Winrad ExtIO_xxx.dll files that exist for many hardware for use under Linux. Libraries written with the same specification and named libExtIO_xxx.so (possibly with a version number) are supported under Linux.

This link Direct comparison of different SDR hardware running simultaneously on the same input signal. shows the pretty large differences that exist between different hardware. The screen dumps show very clearly to what extent differences in hardware performance affect the visibility of signals under the various difficult circumstances under which the tests are made. I hope that browsing the different screens will give a better basis for taking desicions on cost/performance issues for amateurs who are interested in buying SDR hardware. It should be noted, of course, that the software supplied by the different manufacturers can be a more important reason to prefer one hardware rather than another. The tests shown here are all with Linrad set to process the digital data in the same way and thus to enlighten the fundamental hardware properties.

Comparing different SDR software.

The SDR software use filters. Those filters may or may not have a stop band attenuation that matches the hardware performance. That can be tested with a .wav recording containing a simulated strong signal with a noise floor well below the noise floor of todays hardware. This link: Filter performance in perseus.exe, Winrad, Linrad, Excalibur, QS1R, SpectraVue, SDR-RADIO, HDSDR and Studio1 Shows tests with a 500 kHz recording. It shows that the dynamic range limitations may be in software rather than in hardware. Such limitations can easily be resolved in future software releases, the page will updated when I get information that significant improvements have been made to the software. PowerSDR needs a test file at 192 kHz. The filters in PowerSDR. shows the result of an equivalent test of the PowerSDR software. Rocky needs a test file at 96 kHz. The filters in Rocky. shows the result of an equivalent test of the Rocky software.

AGC (automatic gain control) is a very important function in receivers. Modern receivers, SDRs as well as conventional receivers with a DSP for the final filtering and AGC often use unsuitable algorithms that make the AGC system far too sensitive to short pulses. This link: AGC in receivers. Theory and examples. describes why old-fashioned analog receivers work so well and why modern receivers sometimes work less well. It also gives links to test files and loudspeaker output files from Linrad in which a properly working AGC was implemented in Linrad-03.20 and later. The test files can be played on the major SDR software and I hope that AGC problems will soon be history. With digital processing we can do much better than what was possible in analog reeivers. This link A practical AGC test contains test files that can be run with many SDR software. The files contain a weak and a strong SSB signal and impulse noise. Performance differs significantly between SDR software..... Some analog receivers are also tested with an RF signal generated from the test recording.

Time delay in receivers.

The total time delay from antenna to loudspeaker has to be below 50 ms or so to allow a CW operator to work at normal speed while listening between dots and dashes (full QSK operation). The delay has a theoretical minimum value that depends on the bandwidth and shape factor of the filter in use. In analog receivers the minimum bandwidth usable for full QSK is in the order of 250 Hz. In software designed receivers there is an additional time delay caused by input and output buffers as well as by the actual computations. There could be many other reasons for additional delays in an SDR. This link: Time delay in FT-1000 and Linrad shows measurements at a bandwidth of 250 Hz.

The sound system of a modern computer is a complicated thing. There may be buffers, equalizers, resamplers, mixers and perhaps other things that operate silently and invisible to the innocent user. There are also different kinds of drive routines for each operating system. By use of Portaudio Linrad-03.09 and later versions can use many kinds of drive routines with Microsoft Windows: MME, Direct Sound, ASIO, WDM-KS and WASAPI. Under Linux the two kinds of drivers, ALSA and OSS can be accessed either directly or through Portaudio.

This link: time delays in PC sound systems with Linrad-03.12 shows that there are combinations of hardware and software that give negligible delay due to buffering. The total delay is the sum of three parts:

1.) Input hardware and associated drive routines.
2.) Processing delay in the SDR software.
3.) Output hardware and associated drive routines.

Click on the links to see studies of the three different parts that sum up to the total delay from antenna to loudspeaker.

Detectors

Different modulation methods require different detectors. There are two distinct exceptions: Morse coded CW and voice SSB. Those two modes do not require a detector in the receiver, detection is normally done in the brains of the operator and the radio receiver is just shifting the frequency from RF to audio while filtering out everything outside the desired passband. Surely we can put a detector in a SDR. CW Skimmer is one example. One could also include some kind of voice recognition software to detect SSB. The so called product detector is just a frequency mixer. Not a detector.

AM modulation requires a detector. In the analog world we would use a diode, the digital coulterpart would be to take the absolute value. We often have our signals represented as an analytic baseband signal (an I,Q pair) and then we would compute the power as the sum of the squares of I and Q. Then the amplitude would be computed as the square root of the power.

AM can be detected with synchronous detection. One would use two frequency mixers with a 90 degree phase shift and feed them with a local oscillator that matches the carrier frequency. If the signal is a pure AM signal, all the energy from the sidebands would appear in the I channel while the Q channel would be zero. In real life with multi-path propagation it could be different. Listening in stereo headphones may improve readability since the two sidebands will never cancel in both ears. Synchronous detection may also aid in the detection of two AM stations that share the same RF channel. These links show how synchronous AM detection works in Linrad:
Close spaced AM stations with overlapping sidebands
West Coast USA at night

FM also needs a detector. In the analog world it is far more difficult than the AM detector and so is the case in a software defined radio. The FM detector should be insensitive to amplitude variations. It should look only at the frequency (or phase vs time) of the signal. Look here for the FM detectors in Linrad Setup for FM mode

Wideband FM in the 88 to 108 MHz band is a special case. The detected signal contains information above the audible frequency range. Theere is a pilot tone at 19 kHz, a DSB signal containing stereo information and a channel with digital data known as RDS. There may me more information channels. The high modulation frequencies in combination with the high frequency swing produces higher order sidebands on the FM signal. For a DX-er who wants to hear a DX in the neighbouring channel of a strong local FM station this is annoying. The higher order sidebands can however be computed from the first order modulation information so it should be possible to produce reception of FM in the neighbouring channel at relative signal levels that would have made it impossible even in the best analog receiver. This link Detecting FM with neighbour channel suppression will show step by step how Linrad is used as a tool to build new things into Linrad itself. In this case an "antisplatter" function for wideband FM. (Whether the result will turn out to be useful is another story. The answer is totally unknown when this text is written in March 2010.)

MAP65 and Linrad

The program MAP65 by K1JT can be used to receive all the transmissions in the digital mode JT65 within about 90 kHz bandwidth. MAP65 can receive data from Linrad on the network in the TIMF2 format. The TIMF2 data is the wideband signal corrected by whatever calibration data the user has stored and with the noise blanker as adjusted within Linrad.

Linrad can send TIMF2 data as 16 bit integers or as 32 bit floating point data. In case 16 bit data is used one has to be careful to set the digital levels correctly to avoid loss of sensitivity due to quantization noise. Digital modes are sensitive to errors in the drive routines that could cause occasional loss of data because digital modes require precise timing. This is something one should make certain does not occur. This link MAP65 with Linrad is intended to assist MAP65 users to get the best possible performance out of their systems.

Antennas

The antenna - the link between the electronics and free space - is of course the most important piece of equipment for the serious VHF amateur.

On 144 MHz, maximum gain is often the right criterion for selecting the best antenna. On higher bands clean pattern and/or low losses are more important since, for receive, gain divided by temperature, G/T is the true figure of merit. The optimum antenna is a compromise. High gain is always good, but depending on the local surroundings and the propagation mode, other factors may be more important and lead to trade off some gain in exchange for other good things.

Antennas can be optimized for maximum gain by a least squares fitting of the radiation pattern to a desired radiation pattern (zero in all directions except forward !!) It is trivial to add extra equations in the least squares fit to improve G/T, F/B, efficiency, impedance or anything else that the computer can extract from the antenna model. This optimization method is convergent - there is only one optimum design for each set of design criteria. Theory, software and hints on how to design, build and verify high performance antennas should contain everything you need to make your own optimum design.

The 2SA13 is an antenna designed for general purpose usage as a four-stack of cross polarised antennas on 144 MHz. Look here for all details on the development of the 2SA13 antenna

Low noise amplifiers.

Most radio applications do not need low noise amplifiers. The noise from the antenna is usually room temperature or above so a noise temperature a bit below room temperature is usually fully adequate. In space communication (satellites, EME or radio astronomy) it is different. The antenna temperature, the summed noise from losses in the antenna and the noise picked up by the antenna can be very low, only a few degrees Kelvin.

Today, with GAAS FETs, one can get very low noise temperatures in amplifiers. The usual figure of merit for a low noise amplifier is the noise figure, **NF** which is a convenient way of expressing noise temperatures. The NF is the degradation in dB in a room temperature system when using the particular amplifier compared to an ideal, totally noise-free amplifier. Optimizing low noise amplifiers and measuring their true performance is non-trivial. There are standard instruments designed for the telecom industry but they are not designed for the measurement of very low noise figures. They are intended for easy and reproducible measurements of amplifiers used in terrestrial communication.

This link: Tweaking and measuring low noise amplifiers demonstrates how one can tweak low noise amplifiers for optimum performance as well as how one can do absolute NF measurements with good accuracy using simple equipment within reach of the average radio amateur.

Traditional NF meters give large uncertainties for low noise figures. By using them in unconventional ways we can improve the accuracy. This link: Comparing NF measurements uses results from EME 2012 and compares them with data obtained with Linrad using ice and boiling water.

It seems that simple amateur measurements can be made significantly more accurate than measurements with standard instruments. This link Precision Measurements of noise figures is a report on attempts to reduce errors and evaluate uncertainties in NF measurements.

On microwave bands the losses in connectors, adapters, relays and cables may cause a significant loss of S/N. There are two kinds of losses however, mismatch losses and dissipative losses and they behave very differently. A relay with high losses might be better than a relay with small losses. The data we normally read from data sheets is the sum of both kinds of losses while only the dissipative losses degrade the S/N significantly. Have a look here: Losses in adapters, cables, relays and connectors

Relative NF measurements can be made with great accuracy in several ways as is shown in the above links. It is however non-trivial to find the zero point on microwave bands. This link A comparison of 1296 MHz amplifiers at the EME 2013 meeting in Orebro. is an attempt to establish a relative NF scale by which low noise amplifiers can be compared to those amplifiers that were measured at the meeting.

The experiments linked to in this section have shown an unexpectedly close agreement between noise source calibration and noise temperatures determined by independent methods. This link: comparing three different noise heads. is an attempt to get some more insight in NF measurement accuracies.

Cleaning connectors can cause problems if the cleaning agent is not allowed to dry out completely. a study on connector cleaning

Polarisation

Polarisation is by convention horizontal for DX work at VHF. In some propagation modes, e.g. aurora and maybe sporadic E, the polarisation plane may twist due to Faraday rotation, and in EME the polarisation plane may twist due to purely geometrical reasons as well. Fast switching of polarisation, as well as different polarisation for transmit and receive is very useful in these propagation modes.

Crossed yagi antennas may be used for fast and independent switching of TX and RX polarisation. Look at Electronic Polarisation Control for hints on several ways to make good use of crossed yagi systems. Of course the same methods apply equally well to feed horns or any other antenna with two orthogonal polarisations.

With cross yagi antennas, it is important to make sure that the two orthogonal parts really are orthogonal. Look here for more info and some NEC simulations. How to calibrate an adjustable polarisation antenna If the two polarization planes of a cross yagi are not quite orthogonal, the sky noise in the two channels will be correlated to some extent. How to check a cross yagi system for orthogonality using Linrad.

Filters

Particularly for EME, very narrow CW filters may be useful. If you have access to old-fashioned calibrator crystals - use two for this high performance Narrow Filter with 2 * 100 kHz Calibrator X-tals.

If you prefer to use more modern technology look at Sliding FFT and DSP Filtering. This section describes part of the system I am currently using for EME and tries to explain why this method is equivalent to the use of several conventional DSP filters at different frequencies - where the computer continuously selects the best one.

In my opinion the best weak signal communication mode is morse telegraphy, so this mode, CW is the only mode I use on VHF. There are different opinions on what is the best way to receive weak CW signals, some use an ordinary SSB type filter, and others use narrow filters of different kinds. Here is my personal experience in listening to weak CW signals and some examples: demonstration with audio and spectrograms how a weak EME signal sounds with different kinds of filtering.

Look here for a description of my old TMS320C25 EME system and some audio file examples of typical EME signals as they reach my head phones.

As an example of my 1998 system using a 200MHz Pentium (with MMX) look at and listen to the signals from EL2RL, a really weak EME signal. Another example of a rally weak signal is 8J1RL

Computers allow all sorts of interesting experiments. It is well known that an EME signal is about 300Hz wide at 10GHz because of the different doppler shifts from different reflection points on the moon. On 144MHz the EME signal is much narrower than one would expect from the frequency ratio. Look here bandwidth measurements of a continuous wave reflected off the moon for an experiment that clearly demonstrates that two different reflection types are present.

Another example of a computer experiment is shown here: 25W emitted from single 10 element yagi detected via EME using only 4x14 elements

PC software project

The very fast development of digital technology has not only made my dedicated hardware (TMS320C25 with 100ns RAM) obsolete. My first generation PC receiver for MS-DOS is also becoming obsolete. It was written using Watcom C and it works only with "old" computers. There is no support for modern screens and the mouse has to be a serial mouse. The MS-DOS package has served well a few years but for the future an environment where the hardware drivers are outside the radio software will prevent the DSP radio from becoming obsolete so quickly.

I am currently working on a new DSP radio package. This time the system is designed for flexibility so it can be used for many different combinations of computers, A/D boards and analog radio circuitry. The platform is Linux and the package will typically operate with a 486 computer together with a conventional SSB receiver as the minimum configuration. The current high end operation is with a 4-channel 96kHz A/D board and a Pentium III providing nearly 2 x 90kHz of useful signal bandwidth in a direct conversion configuration (stereo for two antennas).

The LINUX PC-radio for Intel platforms will be continuously upgraded to show various aspects of digital radio processing and how they are implemented in the dsp package. The Linux PC-radio is not designed for VHF weak signal only. It is very flexible and designed to accommodate routines for all radio communication modes on all frequency bands.

Linrad is the new name for the LINUX PC-radio since July 2001.

Noise Blankers

A good noise blanker may improve station performance a lot. It is a good idea to have several noise blankers inserted at different points along the Rx signal path. At times when there is no strong signal present on the band, a wideband noise blanker can do absolutely fantastic things - it may even remove computer spurs!!! (This is experimental, not only a theoretical idea). Look here for hints on Noise blankers

Power Amplifiers

To be successful in two way DX communication, you need a good power amplifier. Vacuum tubes are no longer the natural choice for high power. Semiconductors may be more attractive today. Used high power tubes may however be obtained at low cost so tube amplifiers can still be an attractive way to get high power. If you like to design an amplifier of your own, or if you have something commercial to rebuild look here for Building High Power Amplifiers

As amateurs, we often use surplus tubes. They may be brand new, but if they have been stored for many years, they should be reconditioned before use. The vacuum is gradually deteriorating over time if a tube is left on the shelf. Here is a procedure to recondition power tubes that may decrease the risk of arcing due to poor vacuum.

Power amplifiers are often referenced as "linears", meaning linear amplifiers. To my knowledge, few amateurs run their power amplifiers in class C because a class C amplifier is far from linear and can not be used for amplification of SSB signals. In CW-mode the class C amplifier will give the same output as Class AB with much less heating of the anode because of the higher plate efficiency. This helps thermal stability, and reduces irritation among neighbors because of twinkling lights due to mains voltage variations.

It is simple and straight-forward to run an amplifier with variable class biasing by use of a grid resistor. Look here for more information on this and other aspects of power supplies for high power amplifiers.

Electronic antenna relay

PIN diodes can be used as antenna relays at high power levels on VHF. An about $5 device, UM9415, can handle power levels of several kW continuous (key down) carrier. To listen between dots and dashes is useful in contests and during major openings. Look here for details on my High power antenna relay using PIN diodes

By recording the echoes when transmitting dots at meteor scatter speed, aurora, meteors and airplanes can be seen. Look at some typical graphs Aurora and other Echoes With Narrow Band Radar

Dynamic range

For amateurs in densely populated areas dynamic range is probably the most important aspect of the rig. Today, I do not worry since I now live in a rural area, but 15 years ago, the poor design of commercially available transceivers was the dominating problem in my efforts to work distant stations on 144MHz.

In order to make my own situation better, I persuaded my neighbor amateurs to allow me to modify their rigs, and the result was a significant improvement in DX possibilities - and a series of articles. I still get questions about these articles now and then, so now they are available here:

Introduction: English / / Deutsch
Modifications for TS-700: English / / Deutsch / / Swedish
Modifications for IC-211/IC-245: English / / Deutsch
Modifications for FT-221: English / / Deutsch / / Swedish
Modifications for FT-225: Swedish / / English
Keying clicks with FT221 as an example. Perfect high speed ms keying can be done without any keying clicks. Swedish / / English / / Deutsch
Modifications for FT-736: English

AM keying clicks are directly related to the keying envelope waveform. For a good theoretical treatment, look at the article by Kevin Schmidt ***Spectral Analysis of a CW keying pulse*** which you can find at the W9CF links page. In case this link has become outdated you can download the article in pdf format from here with the permission from the author: click.pdf (162770 bytes) The article ***On the Occupied Bandwidth of CW Emissions*** by Doug Smith which you can find at the KF6DX site is more practical and shows the same thing. In case the link has become outdated, there is a copy here with the author's permission: On the Occupied Bandwidth by KF6DX This article also points out that the ARRL handbook has a treatment of optimum keying waveforms that is limited to envelopes formed by a single RC link. It is unfortunate that the treatment in the ARRL handbook is outdated and misleading. Keying with an exponential waveform belongs in the vacuum tube era 50 years ago when cathode keying was normal - but even then better solutions with LC filters were used.

Keying clicks are often produced by amplitude modulation, morse code is in itself amplitude modulation and it is obvious from the links above how the shape of the RF envelope is related to the frequency spectrum in the case of pure amplitude modulation. In the real world, the transmitter frequency may be disturbed at the moment of key closure or release. If the frequency/phase modulation contains high frequencies, there will be FM keying clicks that are invisible in the RF envelope waveform. Worst case is when a VCO looses locking for a while at keydown, but phase modulation can be caused by many mechanisms. Have a look here Keying clicks in the time domain AM and FM

Amateur transceivers often use ALC to improve the average to peak power ratio by having a short time constant for the ALC. This is discussed to some extent in the KF6DX article. It is not good practise, ALC should be a safety precaution only, a circuit (TGC) that puts the peak power just below the maximum safe level and not an AM modulator that modulates the SSB signal to make the power more constant over time in a millisecond time scale while adding wideband modulation sidebands. Click here for spectra, and a discussion of this problem, the main reason for splatter on the amateur bands. The abominable ALC.

Like around 1970, when new digital technology started to become popular (the frequency synthesizer) the coming years will probably bring many digital solutions that have inadequate dynamic range performance for many situations.

Dynamic range properties are characterized differently and often incompletely, which makes it difficult to compare measurements made at different places. This link Measuring receiver dynamic range suggests how data could be presented in an unambiguous way. The measurements required to make a fair comparison between digital and analog receivers are also discussed. The third order intermodulation performance of a receiver is often fully characterized by IP3 only. The link gives a simple theory with some oscilloscope images showing intermodulation waveforms. There is also a discussion about precision measurements and the small deviations from the simple theory that can be observed in the FT1000D.

Here is performance data of modern transceivers. The link contains dynamic range data for a number of modern amateur transceivers both in receive and transmit mode.

Amateur transceivers often use ALC to improve the average to peak power ratio by having a short time constant for the ALC. This is not good practise, ALC should be a safety precaution only, a circuit that puts the peak power just below the maximum safe level and not an AM modulator that modulates the SSB signal to make the power more constant over time in a millisecond time scale while adding wideband modulation sidebands. Click here for spectra, and a discussion of this problem, the main reason for splatter on the amateur bands. The abominable ALC. There is of course much more severe splatter caused by operator errors, but for properly operated transmitters ALC related splatter is more problematic than splatter generated by amplifier non-linearities, this is due to inadequate design and I hope it will not be like this in new amateur transceivers. The ALC problems are closely related to power regulation. Excessive power output from the IC706MKIIG in low power mode is another problem owners of this rig should be aware of.

It should be clear from the above links that my personal opinion is that transceiver dynamic range characteristics has to be better specified and measured differently from how it has been done traditionally. The below articles that have been published in **DUBUS** in English and German as well as in CQ VHF or in QEX give detailed descriptions of the problems and how to deal with them.

Receiver Dynamic Range DUBUS 4/2003, pp 9 - 39. Also in DUBUS *TECHNIK VI* and CQ VHF in two parts. Part 1 Fall 2004 and part 2 Winter 2005.

Transmitter Testing DUBUS 2/2004, pp 9 - 45. Also in CQ VHF in two parts. Part 1 Spring 2005 and part 2 Summer 2005.

Real life dynamic range of modern amateur transceivers DUBUS 2/2005, pp 22 - 37. Also in CQ VHF Fall 2005.

Blocking Dynamic Range in Receivers QEX Mar/Apr 2006, pp 35 - 39.

IMD in Digital Receivers QEX Nov/Dec 2006, pp 18 - 22.

Band Pollution from Amateur Transmitters DUBUS 3 2013, pp 76 - 84.

Speech processing

A human voice has a very high crest factor, a high peak power compared to the average power. Our communication channels are limited by a peak amplitude that must not be exceeded and for that reason it is favourable to distort a voice signal in a way that reduces the crest factor without degrading intelligibility too much. Here is an article on the subject: DUBUS 4 2005 The article is based on findings with voicelab

Sideband noise in oscillators.

Sideband noise measurements on oscillators are not necessarily accurate just because professional instruments are used. Overtones may cause large errors. This link Sideband noise in oscillators shows measurements on five different oscillators. Some good ones and some with common design errors. It is an investigation motivated by lack of agreement between measurements done with amateur equipment and those done with the FSUP from Rohde and Schwarz. The link shows how to measure sideband noise within a few tenths of a dB with amateur equipment. Links from the page discusses the design of really good oscillators as well as common design errors. Linrad is a two channel receiver. With appropriate hardware it can be used to produce correlation spectra which can be used to extend the dynamic range for sideband noise measurements by 20 dB or more. Details here: Using average correlation spectra for sideband noise measurements with Linrad

EME signals received with a huge antenna

During the 2001 ARRL EME contest I brought equipment to Tobbe, SM5FRH to make recordings of the EME signals from his array of 32 X-yagis. The recordings cover about 50% of 5 hours time (9 gigabytes). This link ARRL2001 contains information extracted from these recordings as well as links to the raw data (compressed) This material gives a very good picture of 144 MHz EME signals, how their amplitude and polarisation varies with time.

The debate about Digital modes vs CW in weak signal communication.

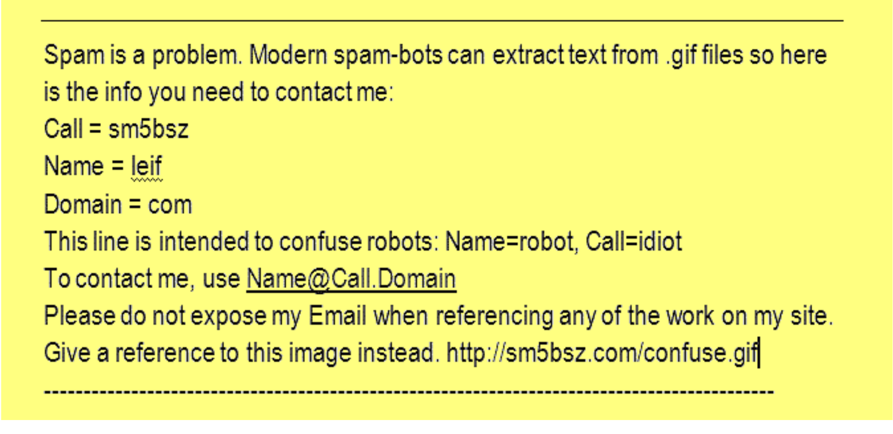
New digital modes can be made much more sensitive than Morse coded CW. This has made it very much easier to complete a DXCC diploma on EME. Some operators who already made DXCC on 144 MHz or above the hard way with Morse coded CW are not very enthusiastic about the situation. A heated debate has been the result.

The arguments in the debate are of two kinds. One is about fairness of competition. Bicycle vs Ferrari so to speak. The other kind of arguments in the debate is about integrity and validity of contacts and it is technical in nature.

The first issue, fairness of competition is politics. How to compromise between legitimate but opposite interests between different groups. Politics is best handled with civilized discussions. Here is my contribution: Mixed contests are needed in EME

On technical issues there is right or wrong. Establishing what is true and false is possible and once that has been done no more discussion is needed.

Since I have some limitations in how much time I can spend with amateur radio - and experimenting is my main interest - I do not promise to answer every E-mail that arrives in my mailbox



Certainly I will try to answer any questions about the material presented at the sm5bsz site. If you have found errors, please send an E-mail so I can make corrections.