

Guide to Military 2400 baud PSK waveforms on the PC Sound Device Modem

Version 1.02

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OVERVIEW

The goal of this document is to assist the reader in achieving the best results with the approach of using software defined modems for Military 2400 baud symbol rate PSK waveforms under MS-Windows on the PC Sound Device Modem (PCSDM).

Knowledge of the facts presented herein is important as everyone must test their systems and determine where they stand in meeting the requirements for use of the MS110A PSK software modem. Any system issues must be eliminated or mitigated as much as possible with goal to achieve the performance provided by software defined MS110A data modem PCSDM approach in lieu of using an embedded hardware modem approach.

The Military 2400 baud symbol rate PSK modems such as MIL-STD-188-110A (MS110A) and STANAG 4415 (S4415) now included in MS-DMT, along with MIL-STD-188-110B Appendix C, represent the basis of U.S. Military and U.S. Government HF data communications. MS110A and S4415 are only the first such Military data modems to be implemented as software defined modems in both the MS-DMT and MARS-ALE tools.

We need to achieve host computer operating systems that will support “Streaming Audio”, if not the software defined MS110A and S4415 modems will not be able to provide their stated performance in detecting and decoding transmitted signals. The goal of achieving Streaming Audio capable systems is a **MUST** before we can even think about pursuing more the complicated waveforms of MS110B Appendix C which provide higher speed throughput and some of which make use of QAM waveforms which put even more requirements on our software defined approach.

MS-DMT v2.00 meets the minimum performance requirements detailed in the military standard with consistent results. The MIL-STD performance requirements are met by mitigating Windows latency and using HF SSB radios with the proper characteristics and the use of a Professional Grade analog sound device such as an ASUS Essence STX II or an equivalent performing device, to include high end HF SDR radios using the best ADC/DAC devices. An Amateur Radio or

commercial grade SSB radio with an internal codec will come in a close second. The use of an external radio sound device interfaces with anal audio will provide decent results.

This document is distributed as “[READ_ME_FIRST_MIL-STD_DATA_MODEM.pdf](#)” as it’s the best place to start your reading during your learning curve. Many have requested a so called “Dummies Guide” to assist in attaining the full potential of the serial tone modem capability in the MS-DMT and MARS-ALE tools. However I do not consider the users of the tools to be dummies. I do however believe that there is a lot to take in when reading all the detailed documentation on the tools. Then there is the challenge involved with sound device selection and setup and the tailoring of the Windows OS to mitigate the latency issues. Where wireless networking and processes that make use of networking, along with Malware, Viruses and the like are the biggest latency offenders. If you are not running an Air Gap computer, become addicted to running the latency and anti-malware and anti-virus tools.

Also there is the configuration of the HF SSB transceiver to be addressed. Hopefully this document will serve to make the effort a bit easier, however it is **NOT** a substitute for reading all the other MS-DMT and MIL-STD data modem related documentation.

As such this document cuts to the chase in a clear and succinct approach with visual material in presenting what you need know and observe in your configuration to achieve the full performance provided by the MS110A modem. The simple truths regarding station configuration required to attain full 75bps performance with the MS110A software defined modem are presented herein. Once the host computer system has been squared away to provide full 75bps MS110A performance using the software modem, most HF SSB radios will provide full performance through 600bps. The higher data rates performance at that point is predicated on the characteristics of the HF SSB radio as detailed herein.

In addition the users of the tools must understand under what prevailing channel conditions which of the MS110A PSK waveforms and what interleave selection are best to utilize. In the absence of that knowledge the transmitting station would have no choice but to always select to use the 75bps data rate and LONG interleave as it will prevail under all channel conditions. However the knowledge of what conditions each data rate can deal with allows for speeding along the message traffic.

The focus in this document shall be the use of MS-Windows 7 SP1 and later operating systems, where use of Windows 10 is recommended. Both MS-DMT and MARS-ALE are specifically developed as native 32-bit Windows applications, where 64-bit MS-DMT is currently under development. Windows operating systems older than Windows 7 are laden with issues that mitigate overall software performance goals, Windows 10 is the least offensive desktop Windows in that regard to date.

All desktop Windows operating systems have latency issues which require mitigation, whereas Windows Embedded Standard is a much better choice as to DPC latency and in other areas of performance. However the bulk of the user base is fixated on the use of desktop Windows. It must be noted that the use of WINE under Linux or WINE or WINSKINS under MAC OS X have been used successfully with both MARS-ALE and MS-DMT in the past, however it is only native Windows that is the focus of development and testing. The author in testing under Linux in the past, has found Linux Zorin OS 8 to be the best flavor of Linux and WINE ready to go, where much better audio and latency characteristics were noted and thus better modem performance using the same PC hardware as used under Windows 7 Professional SP1 32 bit, especially in testing 4800bps un-coded performance.

The PC Sound Device Modem (PCSDM) herein is defined as an AC'97 compliant Sound Device where a device driver is required for its operation under the computer operating system. This document does not address the use of Amateur grade SDR transceivers that make use of on board ADC/DAC with virtual audio cable (VAC) interfacing. Such configurations when using high end radio models with high precision ADC/DAC devices, when all latency issues are resolved, will likely work very well. However the author has no hands on with such systems, as such he cannot write about them from the perspective of experience of having used them.

HF SSB RADIO

A lot of time and effort has gone into MARS-ALE and MS-DMT development in an effort to support all make/model HF SSB radios that are commonly used by MARS members. However any HF SSB radio that **DOES NOT** have a TXCO installed is **NO GOOD** for use with the MS110A software defined modem. As such support for radios without a TXCO standard or optional will in time will be removed from future versions of both tools as coded selections. The user will have to take it upon themselves via means such as MMI-RADIO to make use of radios that are deemed unsuitable.

Typically without the benefit of a TXCO the HF radio will drift too much during long data transmissions of anything but a very short message. The issues will be seen on the receiving end at the lower data rates sooner than use of the higher data rates predicated on what data rate the channel conditions will support. The need is to make use of all data rates from 75bps to 2400bps as channel conditions exist in moving message traffic. As 75bps is the “go to” selection to deal with weak signal and or poor channel conditions we cannot cater to those with poor system characteristics by just trying to make use of 300bps or 600bps all the time.

Also, HF SSB radios that **DO NOT** have wide enough SSB filters on transmit and receive with characteristics close enough to those specified in STANAG 4203⁽¹⁾ will result in degraded performance on both sending and receiving messages. Any use of SSB filters less than 2.4kHz in bandwidth, depending on the filter Group Delay characteristics, will result in too much attenuation and distortion of the PSK signal being transmitted or received.

1. For 2400 baud Military PSK waveforms STANAG 4203 specifies Radio bandpass of 2.75kHz (3kHz nominal military channel) is required where variations in attenuation at most are +/-2db and a Group Delay time over 80% of passband must not be more than .5ms.

Many HF SSB radios, especially Amateur grade, provide features on RX and TX that need to be disabled or for use with for MS110A and other Military waveforms. The common parameter settings required are: Squelch wide open, RF gain wide open, RF attenuator off and all DSP noise reduction off. The IF SHIFT needs to be set near 1800hz if not using a wide enough passband filter.

NOTE: Military complex PSK waveforms such as MS110A PSK are sequential noise-like signal structures. The MS110A 75bps (and STANAG 4415) is manmade pseudorandom-noise Walsh modulation or rather Direct Sequence Spread Spectrum (DSSS) within a base band 3kHz audio channel. Thus 75bps with these two modems are affected the most by DSP filter processing. If you have a local noise issue working find the source of that local noise and mitigate or eliminate it.

The HF SSB radio data port must be used for PC Sound Device interfacing for transmit and receive audio when not using an HF USB Codec based transceiver. The use of fixed data ports eliminates many issues, to include RX and TX audio levels being inadvertently changed, hot mic issues and the issue of radio Squelch which is undesirable as the modem wants to be constantly processing HF channel audio. If the current HF SSB radio does not have a dedicated data port you will need to get an up to date radio.

It is critical that HF radio fixed data port received audio output (typically 100 mV rms @ 600 ohms) be used in feeding audio into the PC sound device modem. The use of the PC Sound Device “Line In” is preferred over “Mic In” as Mic In is typically 30db hotter in most devices. No other ports on the radio should be used for RX or TX audio.

The RX audio level of the device used as the PCSDM must be adjusted to provide the MS110A modem a nominal -55db where a +/-20db dynamic range can be tolerated. The closer to -55db the better for dealing with the worst case channel conditions specified for performance of each data rate. The PCSDM audio levels once set **MUST BE** keep constant. A cheat sheet approach must be used to return to the proper settings using software level control or hardware sound device level control if either is changed.

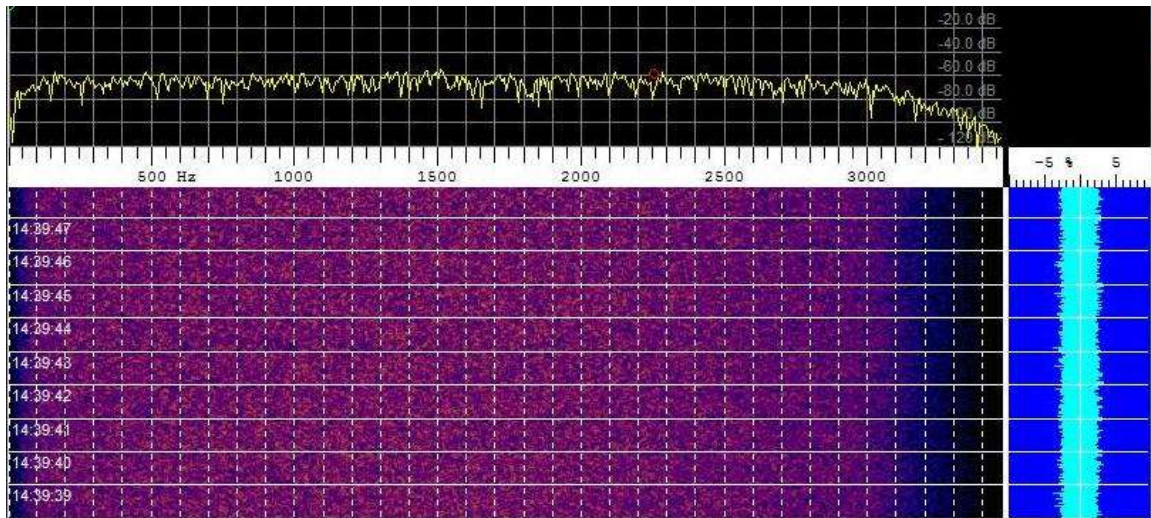
In Military circles the experts when referring to Military grade equipment write about modern single-carrier (serial tone) HF waveforms requiring tight digital and analog filters in order to meet current HF bandwidth allocations. They make it clear that this filtering process produces a variation in the peak-power of the waveform relative to the average-power and that the effect of this variation is that a back-off is required at the input to a power-amplifier (PA) when using a peak-power limited PA to avoid operating in the non-linear region of the PA. This is due to operating in the non-linear region will result in significant spectral regrowth and distortion of the waveform which makes decoding more difficult if not impossible. They recommend PA back-off typically between 3–6 dB depending on the amount of filtering and the modulation type used by the HF waveform. The net effect of this back-off is to reduce the on-air average transmit power by the back-off amount or to require a much larger PA in order to achieve a desired on-air average transmit power.

When making data transmissions with MS110A PSK using the average Amateur Radio grade transceiver it is critical that your radio does **NOT** display any amount of ALC on its ALC metering scale ever. Driving your radio into ALC with MS110A will result in distortion of the data transmission and it will be more difficult if not impossible to decode of your transmissions. The time constant of ALC at key down before ALC settles can even be a problem, especially with MS110B waveforms which is why many hardware modems provide for an initial key down ALC settling time delay.

If your Amateur Radio grade transceiver has no ALC metering or your radio do not have true ALC metering, the absolute maximum limit is 30-35% of your observed SSB Voice output power as viewed by the same RF power monitoring means when transmitting with MS110A. Maintaining drive level that results in less than 30% RF output will all but guarantee a clean TX signal using the average Amateur Radio grade transceiver. The use of Commercial, Maritime and Military grade HF SSB radios will support a greater percentage of RF output level than most Amateur Radio grade radios, how much more needs to be determined on a case by case basis.

Many of today's HF SSB radios in use by MARS members and those that monitor MARS activities have radio equipment that provide feedback at a glance as to whether your station is on frequency, frequency stable, overdriven or cleanly modulated and transmitting at the proper bandwidth. As anyone using an old clunker of a HF radio without a TXCO or having poor filtering for the task at hand will be identified as being part of the problem. Also, don't hesitate to ask your fellow MARS member to critique your stations' on-the-air signal quality.

The passband of your radio receiver can easily be viewed in the audio spectrum using any number of tools. The use of Spectrum Lab being this authors' preference. The passband depicted below is how your radio passband should appear when just monitoring noise with the radio on a dummy load and the Squelch wide open, RF gain wide open, RF attenuator off and all DSP noise reduction off . Please note the roll off that starts about 2.8kHz on the high end. **The image below is what you want your HF radio passband to look like.**



NOTE: The drive level from the radio's fixed level data port has been adjusted to about -55db feeding the software modems receiver. Predicated on the quality of the radio receiver audio and the PC sound device in use, the software modem can tolerate +/-20db range from that point under good channel conditions. However the closer to the mark the better for best results under all channel conditions.

The spectral display in the image above visually represents what needs to be achieved and can be achieved with many of today's Amateur grade radios out-of-the-box. The image above is actually that of a stock IC-7200, one of the least expensive HF SSB transceivers on the market with an internal USB codec that provides adequate results. The IC-7200 wide filter is selected for use with its internal USB codec being used for the modem in the above image. For older radios that come stock or have a TXCO option, the additional change of the radios SSB crystal filter may be all that is required to be compliant for full performance MS110A operation.

NOTE: The HF SSB radio AGC time constant must be configurable for less than 10ms. on desensitization and less than 25ms. on re-sensitization for full modem receiver performance to be achieved.

WINDOWS OS

The best results under desktop Windows will be had using any version of **Windows 7** and later. It is also recommended to use the latest driver available for the sound device, this usually reduces issues with sample rate error.

NOTE: If one is going to run the tools under Linux or MAC OS X, a late model computer loaded with RAM and latest versions of the given OS in question as well as WINE or WINSKINS or commercial tools as the case may be is recommended.

Even though both MS-DMT and MARS-ALE at present will both work under Windows XP versions, the use of XP should no longer be made due to poor latency, poor sound device drivers, security issues and lack of support from Microsoft that all contribute to degrade modem performance. At some point the support for XP shall be dropped in both MS-DMT and MARS-ALE. The use of **Windows Embedded Standard 7** (WES7) is highly recommended as a replacement for XP versions on older PC hardware that is not up to running desktop Windows 7. This has been proven using WES7 on a Dell GX-520 Intel Pentium 4 system. However WES7 is not an out-of-the-box install and go operating system, its boot image must be tailored for the target hardware, which requires a learning curve to achieve.

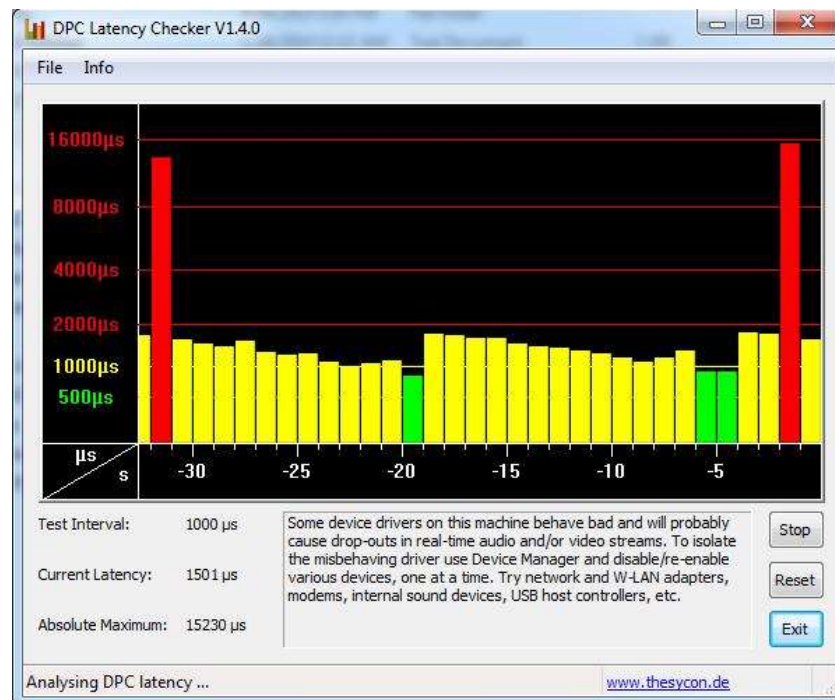
NOTE: At present the minimum the computer requirement for running MS-DMT (and soon for MARS-ALE) is one or more Intel Pentium 4 HT processors @ 3.06Ghz (some later Intel P4 models that had Hyper-Threading disabled and are not suitable) or compatible CPU supporting Hyper-Threading Streaming SIMD (Single instruction, multiple data) Extensions (SSE) 2 and a minimum of 1GB of physical RAM (with 2GB recommended) and 3GB virtual memory. The MS-DMT software will NOT work on a lesser PC hardware system.

All operating system energy saving modes **MUST** be disabled as we do not want the PCSDM to be put to sleep at any time. As many processes as possible that may be running on the computer should be terminated while using the MS110A modem as they are all potential sources of latency.

WINDOWS LATENCY

The Windows OS must be tested for Deferred Procedure Call (DPC) latency issues where excessive latency issues must be mitigated to within tolerances for real-time audio streaming applications for best results. Full details regarding and testing for DPC latency are found in the MS-DMT User Guide. Always perform a full reboot of Windows before testing latency.

The latency with desktop Windows 7 out-of-the-box is pretty good, even with wireless networking enabled, which is a real show stopper under XP versions. However there are things that are issues, the Windows anti-virus software as to it Malware protection was an issue as seen below on the version of Windows 7 Profession 32 bit unit under test (UUT) using the “DPC Latency Checker” tool: http://www.thesycon.de/deu/latency_check.shtml



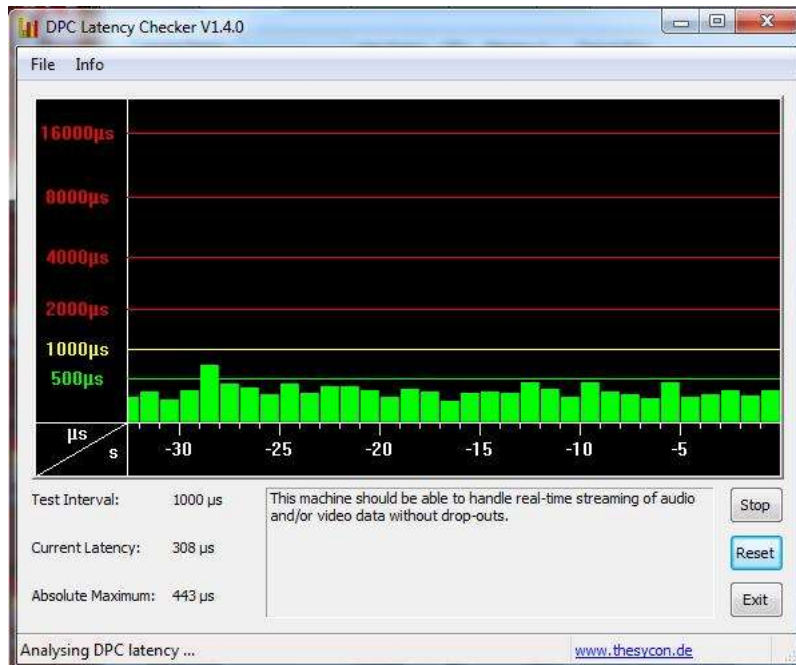
The use of the Resplendence Latency Monitor tool:

<http://www.resplendence.com/download/LatencyMon.exe>

The Latency Monitor tool; provides a processes display facility rapidly provides a view of the offending process as being the Microsoft Malware Protection (msmpeng.exe) process that at times was hitting well over 800 hard page faults as seen below.

Filename	Description	Hard page faults	PID	Company	Product
msmpeng.exe	Antimalware Service Executable	833	920	Microsoft Corporation	Microsoft Malware Protection
explorer.exe	Windows Explorer	291	1628	Microsoft Corporation	Microsoft Windows® Operating S...
latmon.exe	LatencyMon	17	7052	Resplendence Software Projects Sp...	LatencyMon

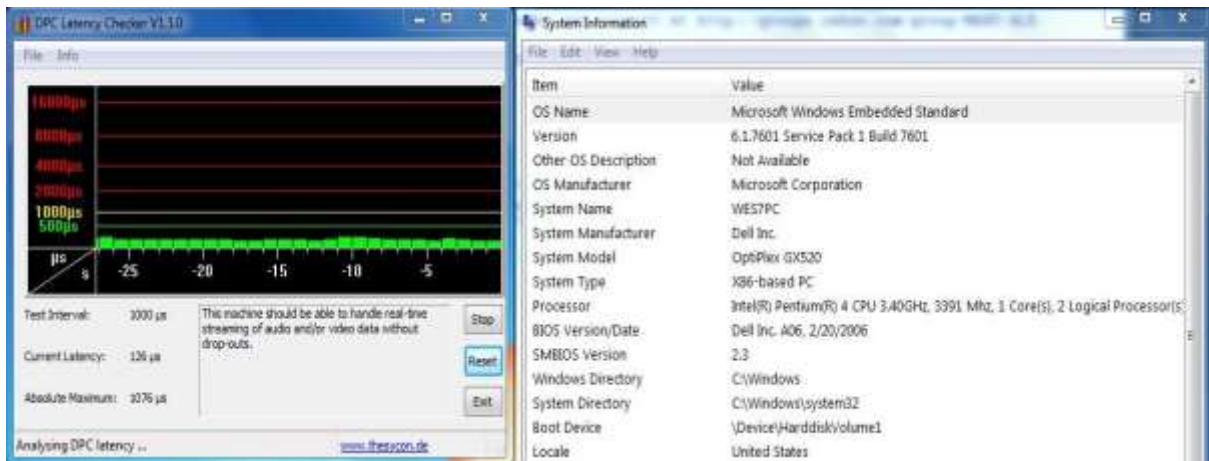
Changing the “Priority” of “msmpeng.exe” to LOW or disabling it using Windows Task Manager resulted in the following low latency on the laptop UUT:



The use of Windows 8.1 Tablet PC's have been found to have decent latency out-of-the-box as seen below on a low end Atom CPU based Winbook TW700 tablet as a worst case example.

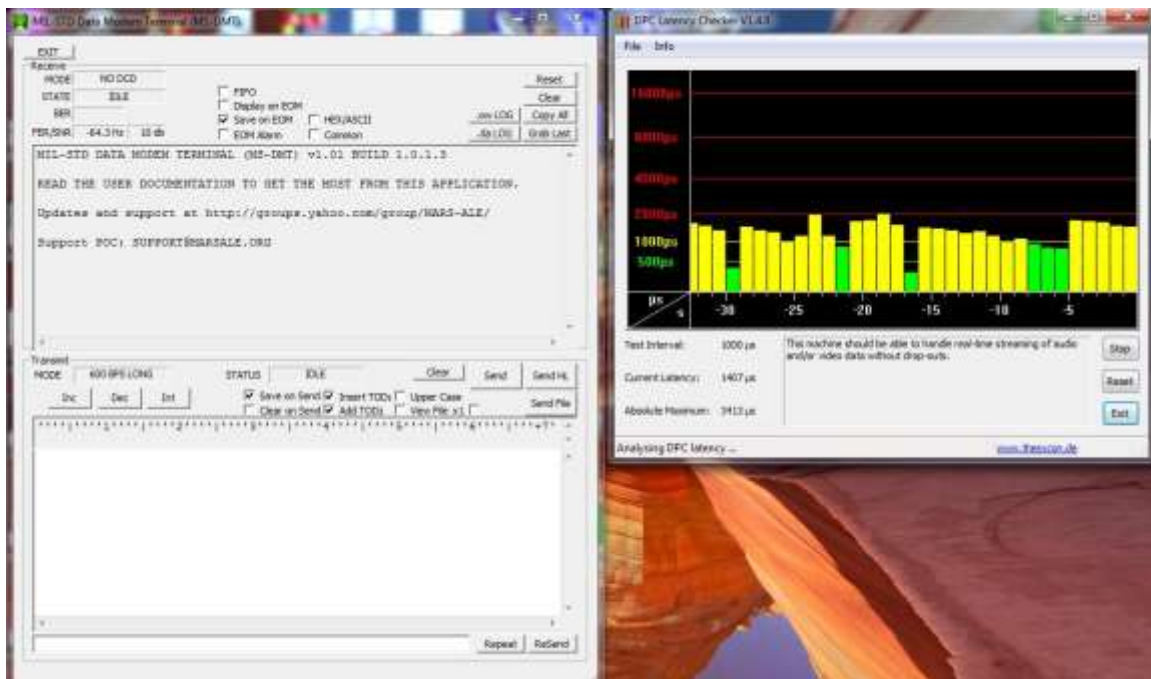


However the amazing low out-of-the-box latency as seen below when using WES7 is absolutely a perfect example of what is desired and required. In all testing of WES7 on two different PC's from an older Dell GX-520 P4 to a modern Atom based unit, nothing ever spikes the latency too high. Below is a screen cap from the GX-520 P4 system as a worst case example!



Any latency issues on any version of Windows that even briefly spikes into the RED using the visual display of the DPC tool when NOT running the software defined MS110A modem precludes its use on that computer system.

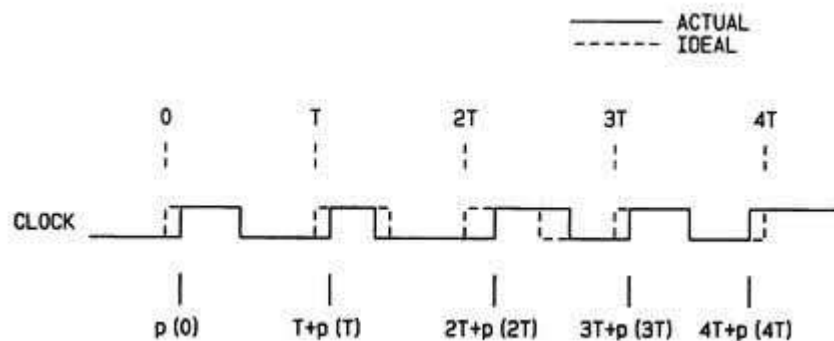
Here is the latest build of MS-DMT running where the worst case latency as seen on a Win7 32 bit laptop that can be tolerated is depicted.



PC SOUND DEVICE CHARACTERISTICS

The dynamic range and frequency response of the average sound device is more than adequate for low symbol rate software modem needs. However for our high 2400 baud symbol rate modem needs the sound device must adhere to more rigid characteristics, in particular:

- High Signal to Noise Ratio (DAC 100db SNR or greater and ADC 92db SNR or greater preferred) and Dynamic Range (DAC 95db SNR or greater and ADC 90db SNR or greater preferred) on input and output.
- Low Total Harmonic Distortion + Noise on input and output.
- An accurate, high stability sample clock (less than 10ppm error being best) on input and output.
- Low sample clock thermal drift stability.
- Sample clock with low deterministic jitter (where period jitter of less than 100 pico secs. (ps) RMS) is required for DAC/ADC data conversion timing.



Low jitter is a big issue in digital audio system related with timing variation in variable clock position. If you have high jitter in your system (ADC or DAC) then the actual result of the conversion process can be either too early or too late rather than ideal. Jitter is the timing variations of a set of signal edges from their ideal values. Jitters in clock signals are typically caused by noise or other disturbances in the system. Contributing factors include thermal noise, power supply variations, loading conditions, device noise, and interference coupled from nearby circuits.

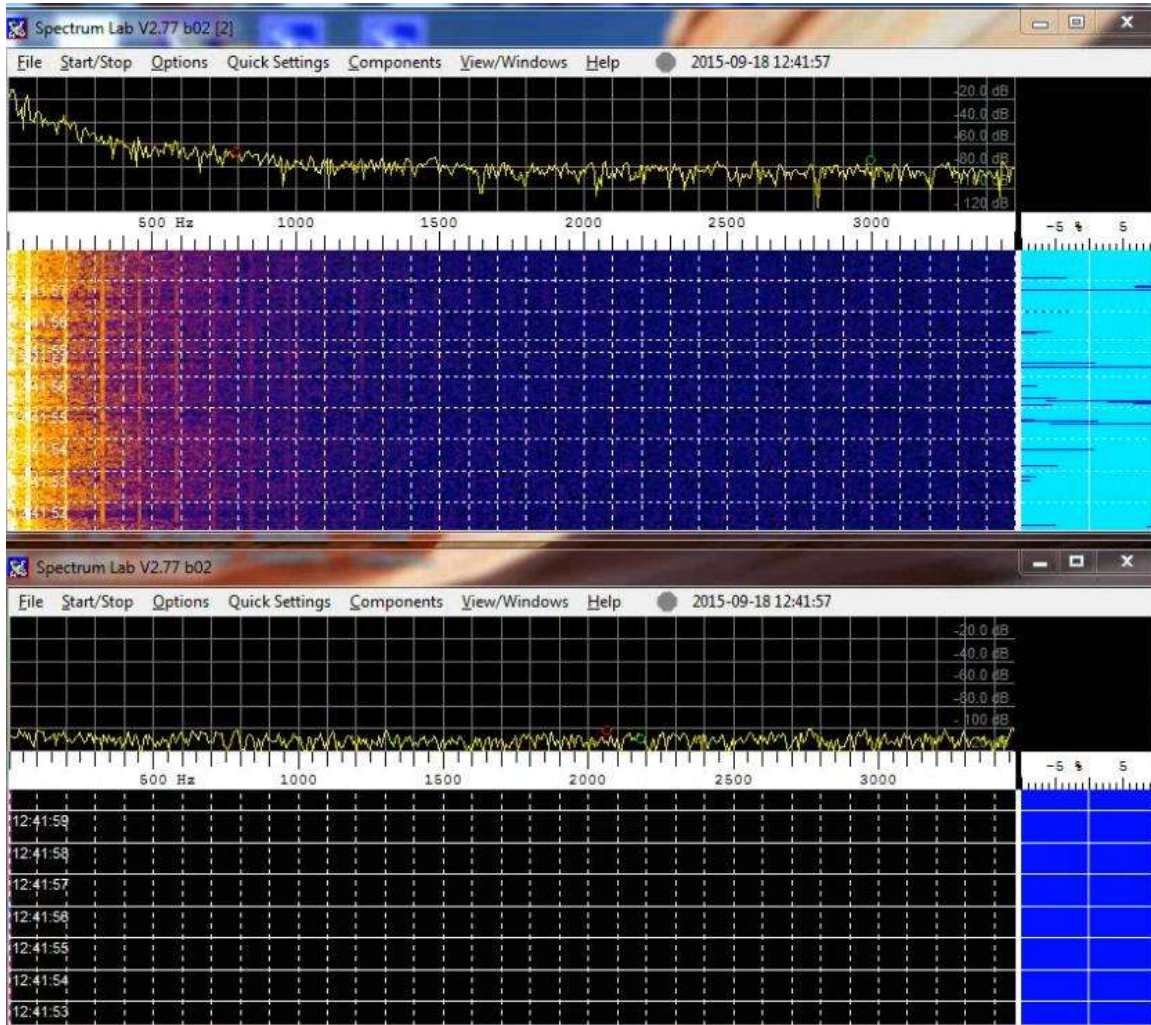
The sound device characteristics will determine its usefulness as a PC Sound Device Modem for use with the software defined modem due to the complex MS110A waveform and its constant 2400 baud symbol rate. An accurate and stable sample rate is critical for accurate Digital Signal Processing (DSP). Differences in sample clocks between stations, thermal clock drift and latency will lead to symbol decoding errors and loss of synchronization.

NOTE: Never have any external PC Sound Device Modem resting on top of equipment that heat and or cool and never have them in direct sun light, a stable thermal environment means less drift.

The screen cap (and video at the URL provided) below are of two external USB sound devices in closed loop baseband audio testing. As can be seen, the top device has artifacts being generated internally which raise the noise floor and mix in with the incoming MS-110A signal being

received causing the modem receiver to work harder. Whereas the bottom device exhibits the qualities desired of a USB sound device for our modem needs.

The bottom device in the image below has a higher Signal to Noise Ratio and higher Dynamic Range compared to the top device, thus the lower its noise floor and there are no artifacts being generated by the bottom device either. **The lower device is what you want your external PCSDM system to look like.**



www.n2ckh.com/MARS_ALE_FORUM/SideBySide.wmv

PC SOUND DEVICE

The interfacing between the HF SSB radio and the external sound device must be ground loop isolated using appropriated impedance matching isolation transformers between the ports. This is provided for internally in the design of many of today's HF SSB radios using the radio data port.

The use of the internal on the mother board chip-set based sound device as standard with today's computers is the worst choice for our modem needs and must **NOT** be used. This class of sound device has the worst jitter and noise and artifacts and usually very poor and unstable sample rates.

Any internal sound device from the PCI sound card or any externally inserted into the PC device such as PC-Card (PCMCIA) sound devices are likely to fall short of providing the modem with the required environment to provide full modem performance. The cabling between the PC and radio are often susceptible to RF interference and are often sources of EMI, which limit effective operation. There are some very expensive professional grade plug-in sound devices. If you have such a system in use and you are lucky enough to achieve full performance from its use on RX and TX then by all means stick with it.

As the goal is to achieve the full 75bps through 2400bps performance that the software defined MS110A modem is capable of providing, we must recommend using one of the best sound device class options. A PC Sound Device that is external to the PC which uses a high speed interface is the best choice in a sound device for our modem needs. The two main classes with up to date computers are USB 2.0 and 3.0 sound devices and Firewire 400 and Firewire 800 (IEEE-1395B) sound devices.

NOTE: All PC sound devices require drivers for their use under MS-Windows or any operating system. Not all sound devices are supported by their manufacturer with a driver. When it comes to USB devices, most just rely on use of the Generic USB driver provided by the OS. Many devices that the manufacturer does provide a driver for may not be supported under all common operating systems. In some cases there may be an operating systems supplied driver for the device when there is and when there is not a manufacturer driver is provided. In many cases the operating system driver may be a better choice as it may support less features, most if not all of which we want to disable for our modem use of the sound device. It is almost always the case that the latest driver for the device will provide for the lowest amount of sample rate error, this should always be tested when more than a few ppm or error exist on TX or RX with the device.

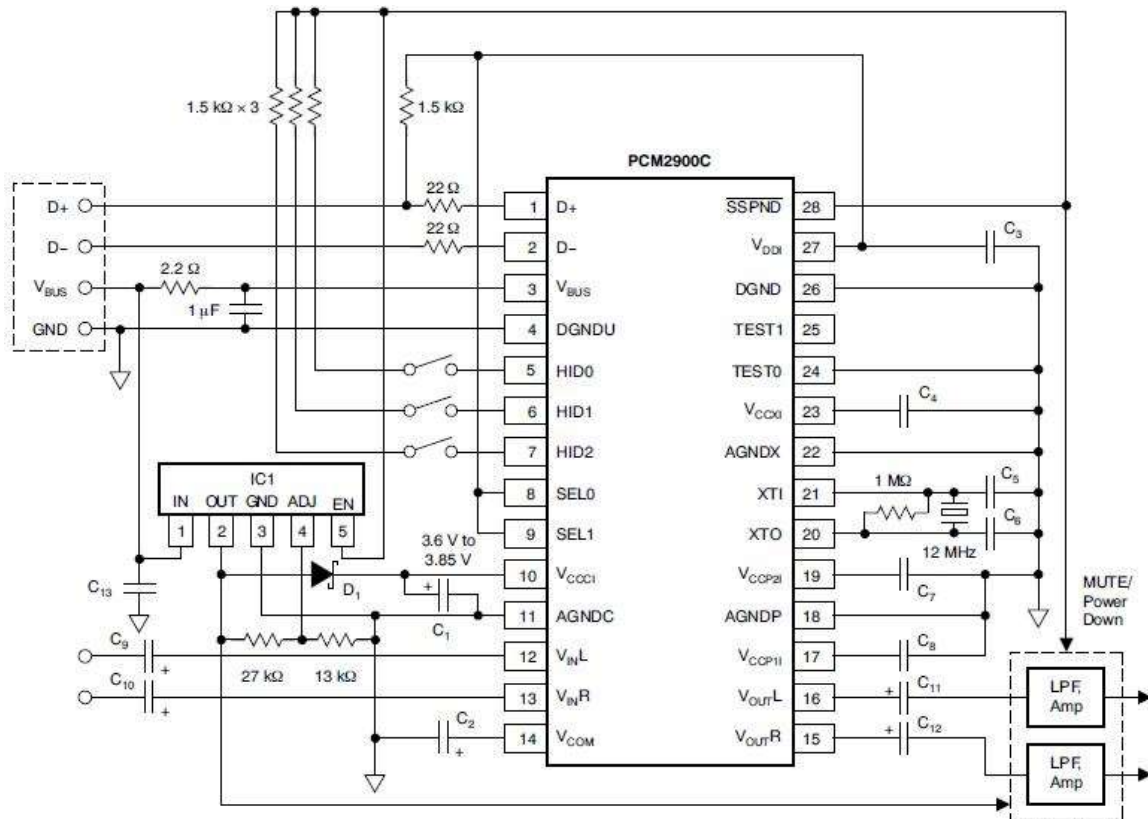
The cabling between these external devices and the radio are often susceptible to RF interference and are often sources of EMI, which limit effective operation.

The most common external USB sound devices in use are those inside COTS external radio interfaces costing around \$100-\$300USD or made from DIY kits or plans. The next most common USB device and gaining ground in models being offered from multiple manufacturers are the Amateur Radio grade and now some commercial grade HF transceivers with USB audio codec/CAT control port which can cost as little as \$1,000USD new for a single USB cabled solution.

The thing in common with both the external USB radio interface and the internal to the HF radio USB codec approach is that they both make use of a single inexpensive USB codec device and reference crystal. Most COTS external radio interface units and many HF SSB radios with USB codec ports are based on the Texas Instruments PCM2900/2902 family of very inexpensive devices. These devices on paper come close to meeting the desired Signal to Noise Ratio and Dynamic Range specifications desired, however some designs fall short in application.

The PCM2900B/2902B family was the first to be USB 2.0 compliant, however it is only the PCM2900C/2902C family and later that are MS-Windows compliant with the Windows Generic USB Device Driver when not supplying product specific driver. It is only the revision C and later PCM family devices that work properly under Windows without extra attention by the user in setup procedures.

The TI sourced image below is from the PCM2900C data sheet and illustrates a typical PCM2900C circuit for a high-performance application design that can be considered to meet the USB specification as a USB 2.0 compliant sound device product.



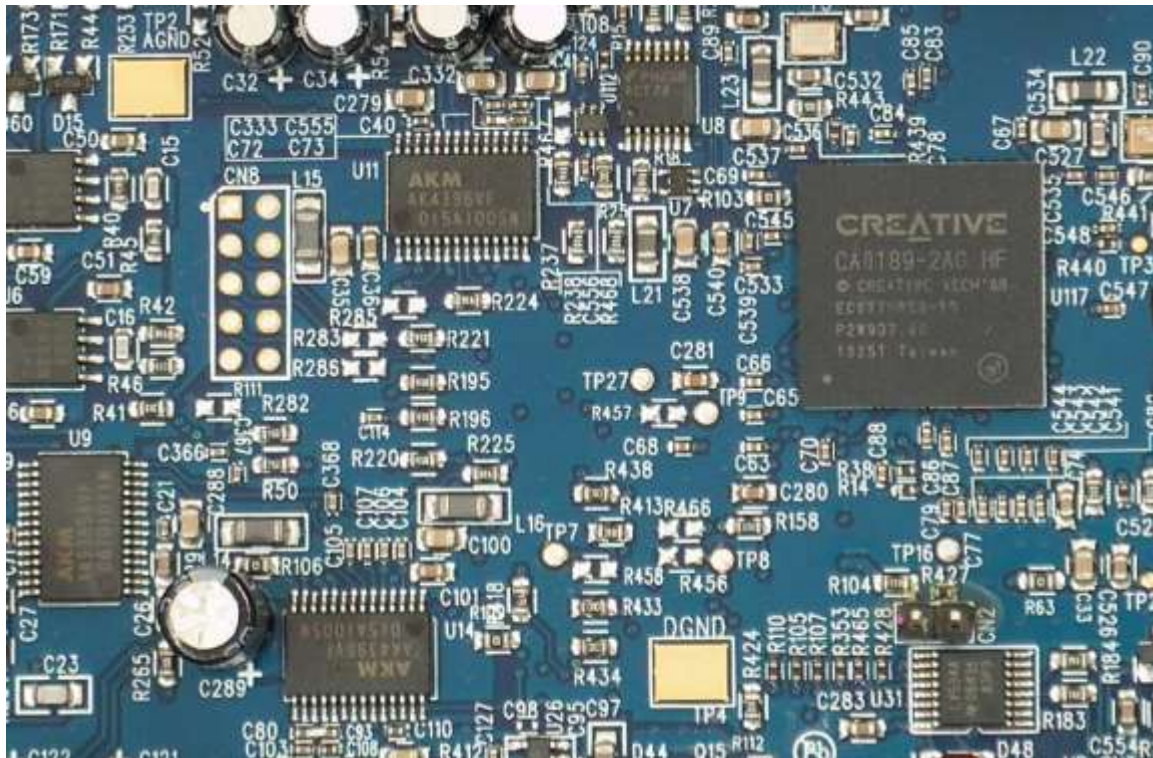
NOTE: C₁, C₂: 10 μF
 C₃, C₄, C₇, C₈, C₁₃: 1 μF (These capacitors must be less than 2 μF.)
 C₅, C₆: 10 pF to 33 pF (depending on crystal resonator)
 C₉, C₁₀, C₁₁, C₁₂: The capacitance may vary depending on design.
 IC1: REG103xA-A (TI) or equivalent. Analog performance may vary depending on IC1.
 D₁: Schottky barrier diode (V_F ≤ 350 mV at 10 mA, I_R ≤ 2 μA at 4 V)

What is missing from the circuit is the typical USB bridge and RS-232 port interface for radio CAT control and hardware PTT interfacing circuitry that would be part of a COTS radio interface product. Some COTS external interface units just provide additional PTT circuitry which can be as simple as an opto-isolator to an external source such as PC RS-232 DTR or RTS lines or an audio sampling VOX keying circuit.

When properly designed, regardless of make/model of codec device used, the inexpensive simple USB 2.0 codec device approach as depicted above can provide decent low noise audio quality and relatively accurate and stable sample rate for adequate support of our modem needs. The addition of radio interfacing options in an external interface adds single USB cable solution to the mix.

The next class of devices are the \$100 to \$300 range of external USB 2.0 and 3.0 consumer sound devices typical of the \$100 range “E-MU 0204 USB 2.0 Audio Interface”. This device class provides the next level of audio quality required for our modem needs. These devices make use of the custom system-on-chip approach coupled with ADC/DCA devices or Digital Signal Processor (DSP) and audio codec approach.

Here in close up inside an E-MU device using the Creative CA0189-2AG HF USB 2.0 system-on-chip device (<http://www.creative.com/oem/products/chips/ca0189.asp>) with AKM5385AVF AK ADC and AK4396VF DAC 24 bit devices with 114dB SNR for both DAC/ADC.



These devices meet or exceed the desired Signal to Noise Ratio and Dynamic Range specifications desired. Where for example the E-MU 0204 provides 113dB SNR for both DAC/ADC. These devices typically operate at sample rates beginning with 48kHz and provide up to 24 bit sampling or higher. These devices provide custom low latency device drivers which for the given make/model device may not be available across Windows, Linux and MAC OS X. Such consumer sound devices are usually wired for common 3.5mm stereo, XLR or other connectors typically found on consumer sound devices. Being consumer audio devices they provide no features in support of radio remote control or radio PTT, however most modern radios provide for CAT PTT and COTS interfaces for such support are readily available.

The next class of \$300-\$500USD+ Professional grade sound devices provide the best audio quality sound device for our modem requirements. These professional sound devices use custom Digital Signal Processors' (DSP) and Audio Codecs' and are optimized for low-latency multichannel sound recording and playback, including studio-grade fidelity and streaming audio applications. They either use high speed Firewire or USB 2.0 or 3.0 interfacing to the PC using custom low latency device drivers which for the given make/model device may not be available across Windows, Linux and MAC OS X. They typically operate at sample rates of 48kHz and

higher and typically offer 16 bit depth and higher data sampling. They are also usually wired for XLR and 1/4 inch connectors' verses common 3.5mm stereo or other connectors typically found on consumer sound devices. Being professional audio devices they provide no features in support of radio remote control or radio PTT, however most modern radios provide for CAT PTT and COTS interfaces for such support are readily available.

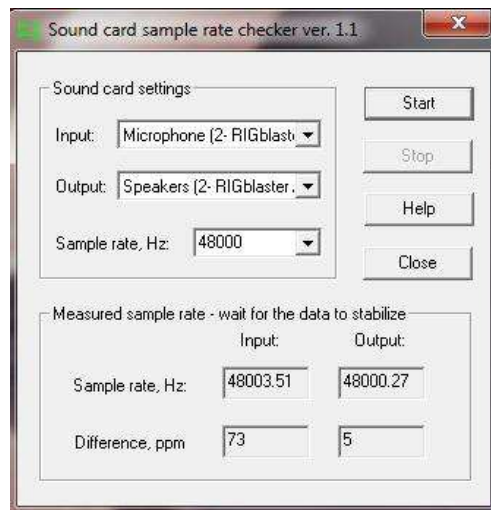
The ASUS Essence II which provides an onboard TXCO is the starting point COTS Professional grade device, from there other internal and external device are much more pricy

The MS-DMT developer has also taken the time to learn TI/BIOS DSP development in C language and Assembly coding and has worked with a TI development board to bring about a USB Composite Sound Device that provides for Professional Grade audio. This \$100USD device make use of standard USB generic drivers available under Windows, Linux and MacOS. Furthermore as the entire project code is controlled by the developer the operating sample clock can be calibrated to offset for USB port error. Based on the TI C5535 ezDSP developer board at present, the addition of a TBD daughter board will provide for all radio interfacing requirements. For more information see: www.n2ckh.com/MARS_ALE_FORUM/MS-PCSDM-OUT-OF-BOX-START.pdf and (<https://store.ti.com/TMDX5535EZDSP-C5535-eZdsp-USB-Stick-Development-Kit-P2615.aspx>)

SOUND DEVICE SAMPLE RATE

The symbol rate of MS110A at 2400 symbols-per-second (baud) needs to accurate to minimum of ± 10 parts per million (ppm) or ± 0.024 symbols-per-second between sending and receiving stations. The clock accuracy for generation of the 1800-Hz PSK carrier must be within ± 1 Hz.

The latest version of MixW (<http://mixw.net/index.php?j=downloads>) must be downloaded and installed to get the Check Sound Rate (CHECKSR.EXE) tool for use where the latest version is 1.1 as best I can tell.



The CHECKSR tool is rather simple in application, however there are things to take into consideration in its use. The selection of 3 parameters, the input and output sound device used as the PCSDM by naming convention and the sample rate to test at as seen above is all that is required. We are only interested in testing at 48000hz at this time.

The sound device itself needs to either have its input and output ports disconnected from the radio or the radio needs to be off. In the case of an HF radio USB codec, set the RF gain to minimum, the attenuator ON and pre-amp OFF with the radio connected to a dummy load.

You need to run it for 60 minutes after cold turn on of your PC system and/or sound device. Then you need to run it for 30 minutes on each additional USB port tested to determine which port is the best port as to sample rate error. During testing the sample rates are displayed in Hertz (Hz.) for input and output along with the difference in Parts Per Million (ppm) compared to desired rate, in our case 48000hz. As the test runs over time the sample rates displayed will be closer to 48000.00 as displayed to point where there is 1hz or preferable less movement. When you see 48000.nn being displayed and just the digits below the decimal point are in motion click "Stop". If after the time required or longer has pass and that has not occurred then you might as well click "Stop" anyhow as it's not going to get any better.

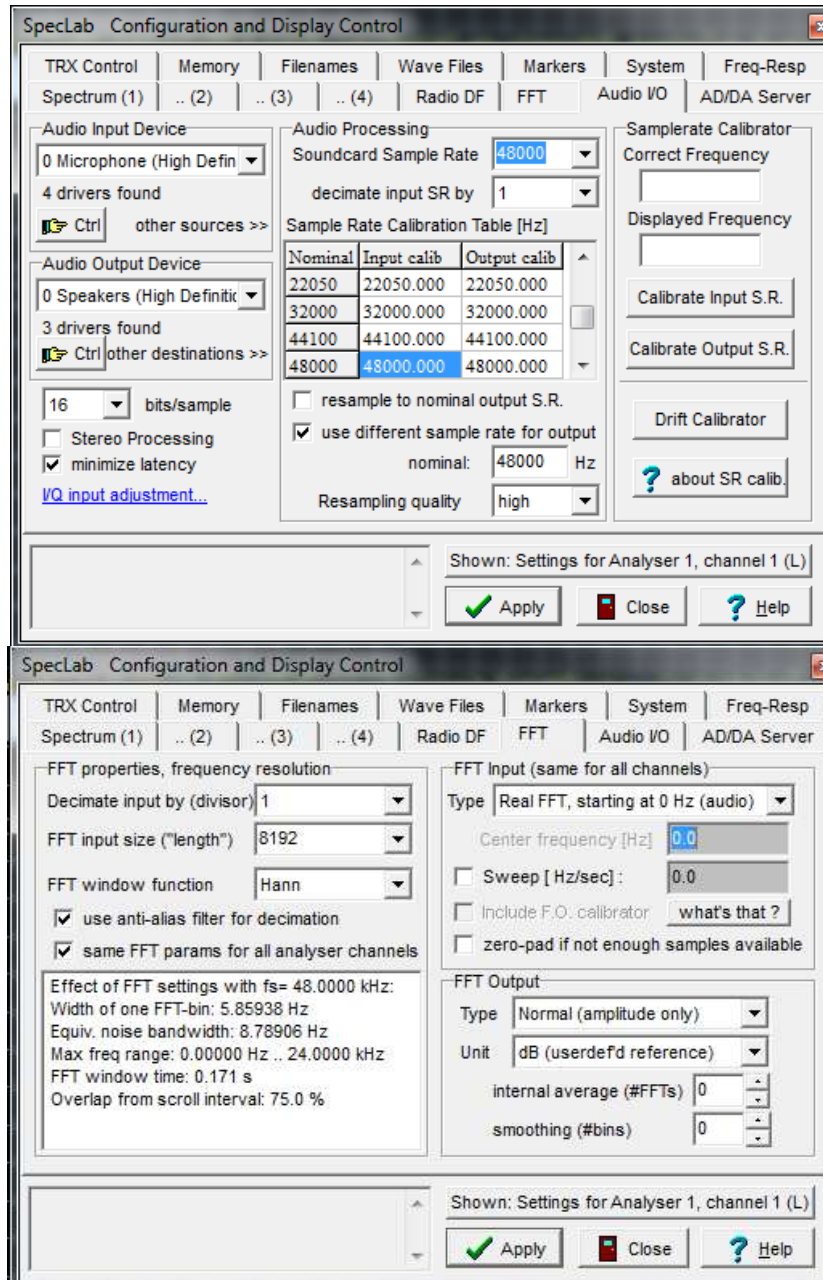
Also, many things can negatively affect its accuracy as to Windows events, even mouse movement let alone processes that may be running, its best used after system latency issues have been mitigated.

Sometimes right start of testing or many minutes later the readings will go sky high, this is often attributed to RFI of USB port 5v line voltage noise other factors unknown. You will need to click "Stop" and then "Start" again. If this continues to occur investigation will be in order. On really bad PC USB ports the 5v line may require elimination and power from regulated and filtered 5 or 5.1 volt D.C. source will be required. If this approach is required then you will likely find it will improve your sound device noise floor performance as well.

GOOD TO POOR SPECTRAL QUALITY EXAMPLES

This section provides a series of screen captures featuring various PC Sound Device units under test where the RX spectrum is displayed using Spectral Lab as a visual representation of Good to Poor quality examples for our modem needs.

The following Spectrum Lab parameters were used in testing:



Spectral Lab is a very power tool with many features, far more than can be covered in any detail herein. Installing the tool and just setting up the two pages of configuration depicted above will provide the required display characteristics for our basic needs.

The latest Spectrum Lab tool can be had at:

<http://dl4yhf.darc.de/spectral.html>

For additional information on the tools the following are recommended:

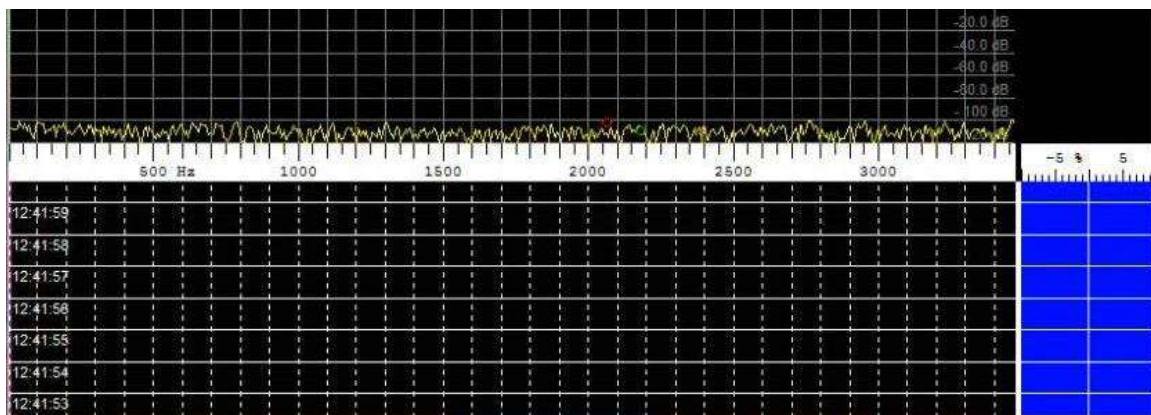
<http://www.qsl.net/dl4yhf/spectral.html>

<https://groups.yahoo.com/neo/groups/SpectrumLabUsers/info>

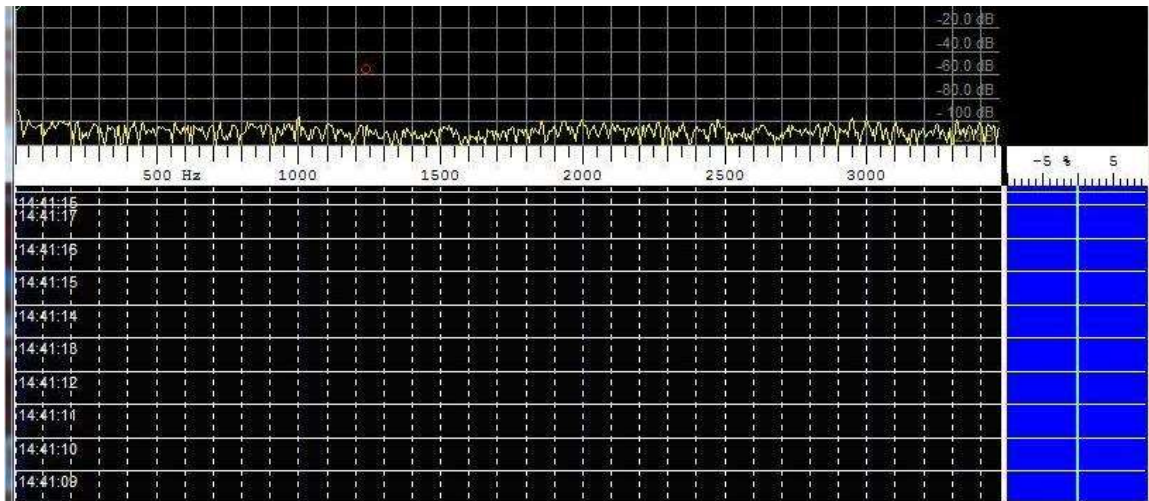
It is recommended that a comparison of your HF SSB radio passband be made to these images. If a USB codec radio is not being used, the fixed data port received audio output (typically 100 mV rms @ 600 ohms) must be used in feeding your PC sound device modem. The use of the PC Sound Device “Line In” is preferred over use of “Mic In” which is typically 30db hotter in most devices.

When it comes to the use of COTS external radio interfaces specifically made for Amateur Radio use, many that have been on the market for some time are known for certain issues. Some issues have been addressed by engineering changes over time and some have not by the various manufacturers. There are modifications that have been widely published on some units. However for many MARS members, making the modifications for a given device may be beyond their skills. If you find your unit has issues, a known hardware modification may cure the issue.

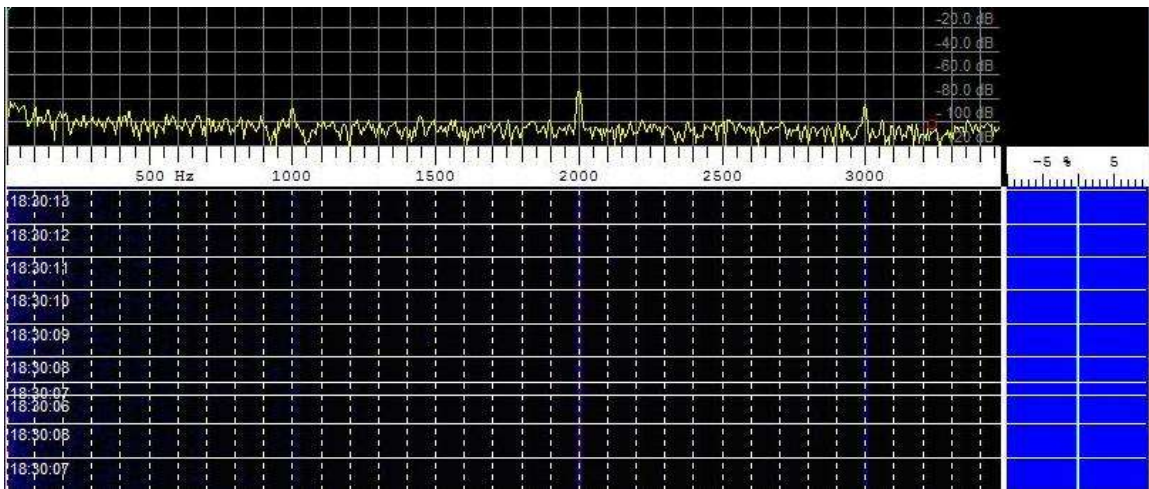
The use of each PC Sound Device's depicted herein are with the HF radio on a dummy load with the radio RF gain set to minimum, attenuator off and the preamp set off after having properly configured receive audio levels. In comparing your system the same procedures must apply.



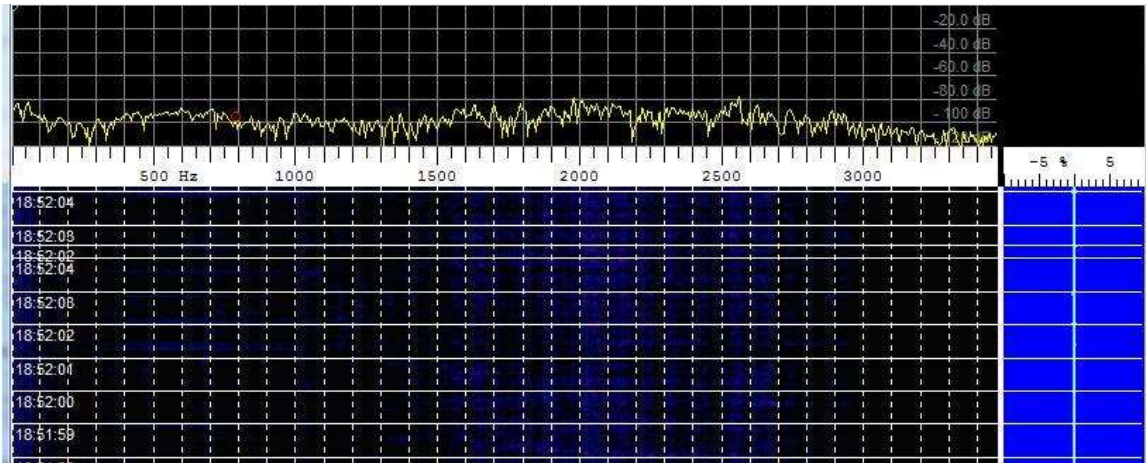
The spectrum above is that of an inexpensive external DSP/Codec based Composite USB sound device based on the TI C5535 ezDSP developer board. Note the flat passband response and low noise level of the passband and the lack of any artifacts in the passband above. This is typical of a Professional Sound Device. **This is what you want your external PCSDM system to look like.**



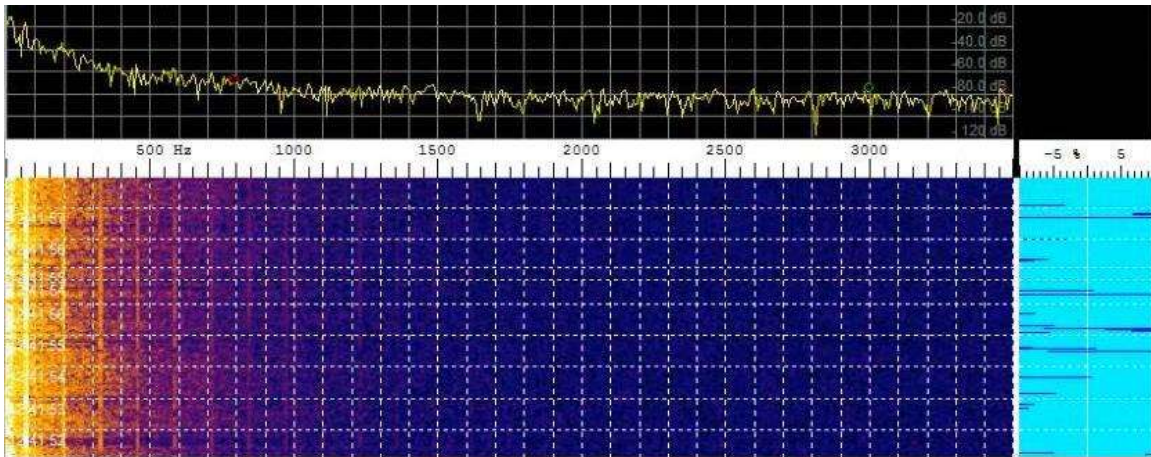
The spectrum above is that of an internal USB sound device of an IC-7200 HF SSB radio. Note the flat passband response and relatively low noise level of the passband and the lack of any artifacts. **This is what you want your HF radio USB codec based system to look like.**



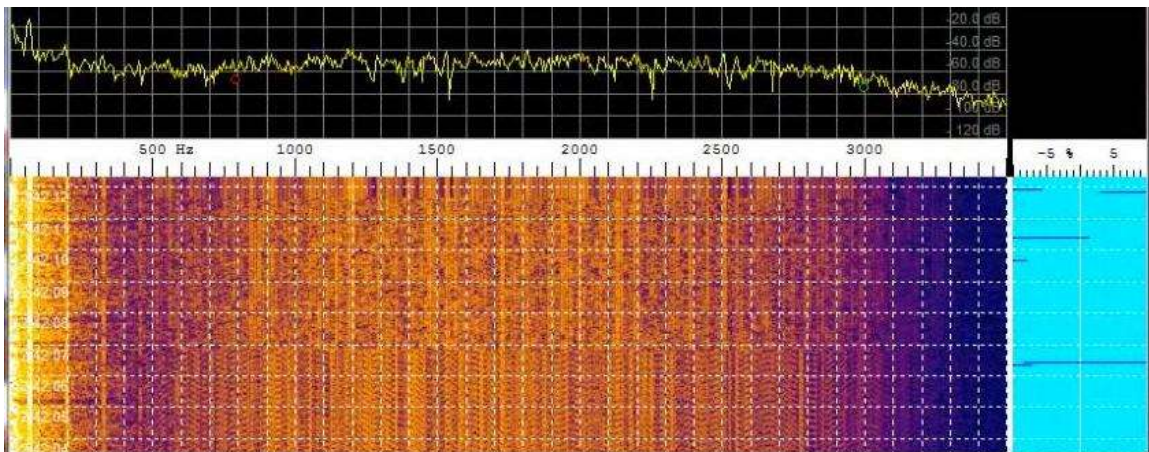
The spectrum above is that of an external USB sound device radio interface. Note the passband response and 3 pronounced artifacts at 1000, 2000 and 3000hz in passband. Also note the noise flow. **This is not an optimum pass band for our modem needs.**



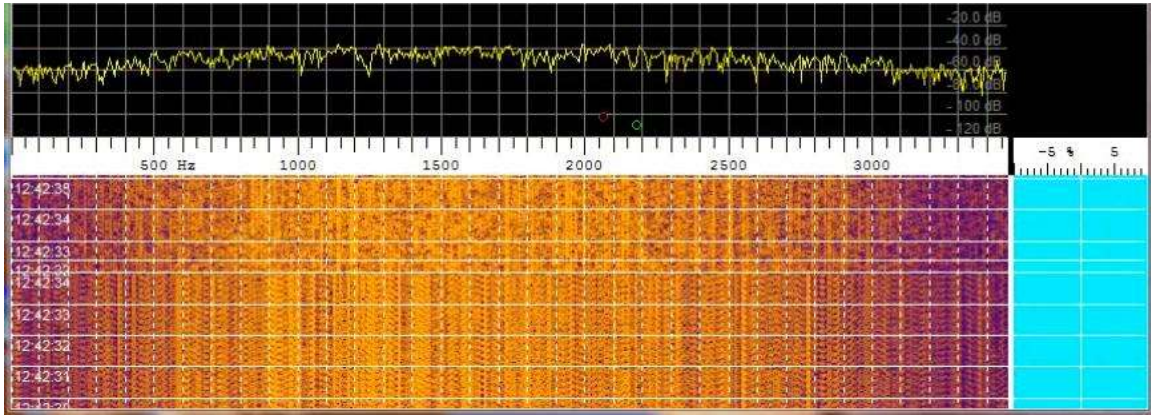
The spectrum above is that of a Laptop PC on board Codec device. Note the passband response with pronounced wide band artifacts and the noise floor. The use of on-board chip sets and many plug in PCI card sound devices will appear as above. **This is not an optimum pass band for our modem needs.**



The spectrum above is that of another external USB sound device radio interface. Note the passband response with artifacts being generated at the bottom of the spectrum and the resulting noise floor. **This is not an optimum pass band for our modem needs.**



As can be seen in the spectrum above, the attributes generated within the above USB sound device radio interface negatively affect the incoming MS110A waveform which causes the modem receiver to work harder which results in marginal performance under good, yet alone, severe channel conditions. Below is what the spectrum should look like during MS110A reception, the image below was captured while using the inexpensive external DSP/Codec sound device we started off using in this section.



For reference, below is the spectrum of the inexpensive external DSP/Codec sound device with the radio RF gain at minimum, attenuator and preamp off. **This is what you want your external PCSDM system to look like.**

