



**DRAFT**  
**MS110A**  
**Data Modem Terminal**  
**User Guide**  
**Version 1.00**

For use with  
Candidate #2 for Beta #2

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**WARNING:** Do **NOT** use greater than 2.8Khz filter on transmit or 3.0Khz on receive.

**NOTE:** Military data modem serial tone waveforms use 1200bps and 2400bps symbol rates which are **NOT** legal for Amateur Radio use under current FCC rules unless used for Digital Voice or Image transmissions.

**NOTE:** The release notes text file in the distribution .ZIP file will always contain the latest summary information on the software during the development cycle.

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## **PREFACE**

This software is being developed to provide the MARS member that cannot afford or otherwise acquire a hardware MIL-STD data modem an optional means of communications interoperability using the PC Sound Device as the modem hardware device.

The results from use of this software depend on the user adhering to all recommendations herein. It is **STRONGLY** recommended that all directions on configuring and testing your operating system latency, sound device sample clock error, setting up sound device levels and HF SSB radio parameters herein be followed to obtain the best possible results when using this software. In addition, although optional reading, it is highly recommended that the MARS member read the "MIL-STD/STANAG Data Modem Primer" document.

Those with PC Sound Device Modem (PCSDM) experience from using Amateur Radio software modem protocols commonly used in MARS must **NOT** make any assumptions that their existing Windows operating system configuration (or emulation using Wine or Winsteps or other), sound device and levels, HF SSB radio setup or PC-to-Radio interfacing as configured for Amateur digital communication applications previously used, will work properly in meeting the requirements of this software's supported military waveforms.

A MIL-STD/STANAG hardware modem was traditionally an Embedded System, designed specifically to perform its functions as a military modem. Embedded systems contain processing cores that are typically either Microcontrollers (MCU), Microprocessors (CPU), Digital Signal Processors (DSP) or Field Programmable Gate Arrays (FPGA) based. The key characteristic being that hardware based military modems have processing power dedicated to the processing requirements of the modem and are not shared among non related tasks. Since the embedded modems are dedicated to the specific modem tasks, design engineers have an optimized environment for consistent reliability and performance of their hardware modem firmware.

The new trend among some military hardware modem designers is based upon using a Unix or Linux embedded operating system due to low cost and ease of life cycle maintenance and ability to provide Software Development Kit (SDK) offerings for end user customization as an alternative to past embedded approaches. An example of a Linux based modem is the Rockwell-Collins Q96xx family of modems which even support direct video, keyboard and mouse for configuration setup and some user controlled operation without the need of an external computer which provide such user tools as Constellation and Channel Impulse Response displays and an SDK for additional user tool development that traditional military modems do not provide. However, even with these types of modems, the only tasking taking place by the embedded Linux OS is that dedicated to the modem use, thus we are still talking a dedicated, optimized environment for consistent reliability and performance of these military modems.

The software detailed herein represents a virtual MIL-STD data modem that does not benefit from having a dedicated environment in any aspect for its operation and is thus presented with a host of challenges to achieve the goal of meeting the minimum performance requirements as detailed in the military standards and with consistent results. However with a clean native Windows OS install, properly configured and maintained on a dedicated PC (or one setup for bootable USB 2.0 or greater Flash drive or external USB hard drive) for MARS communications use, this software already comes very close meeting minimum performance requirements across all data rates for the 110A modem, even though the software is still under development.

This software is much more demanding of the PC Sound Device, operating system and radio equipments vs. Amateur Radio digital communications applications which support low symbol rate data modes that make use of the PC Sound Device as the modem hardware. The complex Military serial tone PSK waveforms being supported by this software have much higher symbol rates, wider bandwidth and higher data rates than do the Amateur Radio, Commercial and Military FSK (RTTY) digital modes commonly used by MARS in the past and implemented on the PC Sound Device as the modem. Much more care must be taken in the configuration and maintenance of the PC operating system and the radio configuration for the best results when using this software.

The various common Military serial tone waveforms utilize either a constant 1200bps or the more common 2400bps symbol rate where the design of the IF passband filtering and radio receiver dynamic range determines how faithfully the radio reproduces complex signals. The MIL-STD/STANAG serial tone signals contain information in both phase state and amplitude value and occupy an instantaneous bandwidth as wide as 3Khz, requiring a minimum of 2.75Khz of IF filtering for optimal results. As the passband response of radio IF filtering alters both phase and amplitude of a complex signal, any marginal radio design considerations can cause degraded performance of these complex waveform signals. In transmitting these complex waveforms radio linearity is more critical as well where only full STANAG 4203<sup>[8]</sup> compliant filtering provides for the best results.

The use of 110A ST waveform for FEC only peer-to-peer, broadcast and multicast communications can most closely be compared to the use of MT-63 from a MARS user's perspective. They both make use of some common features technically and from outward appearance in application, such as operation at different data rates and interleaver settings. However they also differ greatly in numerous ways where MT-63 although influenced by Military standards, does not adhere to any standards and where its author developed the protocol to take into account all of the negative issues of implementing the protocol as a software modem running within the MS-Windows environment and on Ham radio equipments with narrow voice grade filters. The same is true of most other sound device protocols such as WINMOR, where even the use of VOX PTT latency was taken into account in the waveform protocol. This software however attempts to meet all the rigid requirements of the Military standards within a general computer operating system environment that is just not suited to doing so without additional effort made on the part of the end user.

A MIL-STD hardware based data modem has an accurate and stable sample clock and performs only one function. Whereas this software modem communications tool is being developed to work within the challenges of the MS-Windows pre-emptive operating system with no dedicated resources compared the traditional dedicated embedded hardware modem environment. There are many negative aspects of using a general purpose computer and MS-Windows or any similar multi-tasking operating system as a MIL-STD software modem platform. Aside from non modem related processes causing CPU loading and latency issues, the PC sound device sample clock accuracy, first program capture under some OS's and stability come into question as does its noise floor, deterministic phase jitter and other parameters. The MS-Windows environment needs to be tamed down to mitigate if not completely eliminate the demands on system resources and the latency issues that affect real-time audio streaming and RS-232 signaling timing for the best possible results when using this software, which some users may find a bit challenging to achieve.

## **OVERVIEW**

The M110A Data Modem Terminal (DMT) application is a software modem based communications terminal tool that is currently developed as a Microsoft Foundation Class (MFC) Multithreading 32 bit application designed to run under an MS-Windows 2000 SP4 or later (MS-Windows XP, Vista, Windows 7) operating system. The required PC hardware includes an appropriate PC Sound Device for the modem physical layer requirements and a means of RS-232 based or other, uninterrupted automatic radio push-to-talk (PTT) activated prior to and during data transmission. The software provides numerous features to aid in message handling, additional features such as Data Compression, Data Encryption and others will be added as required and in accordance with the specific Military standards dictated.

This software application functions as both a MIL-STD modem and data terminal and also supports external asynchronous terminal applications. The software provides MARS a MIL-STD-188-110A<sup>[1]</sup> (MS110A) compliant Serial Tone (ST) waveform modem based Message Terminal capability providing message composition and automated message storage to simplify MARS MS110A Forward Error Correction (FEC) message handling.

MIL-STD-188-110A details FSK Radio Teletype (RATT), PSK Single (Serial) tone and both 16 tone Differential Phase-Shift Keying (DPSK) and 39 tone Quadrature Differential Phase-shift Keying (QDPSK) parallel modems. At this time our main interest herein is detailed in MIL-STD-188-110A section 5.3.1.1, the 110A serial tone waveform. This serial tone waveform is the baseline for all current U.S. Military and Standard NATO waveforms and has been found to be superior to both the 16 tone DPSK and 39 tone QDPSK parallel waveforms to the point where the 16 tone waveform has become obsolete as of MIL-STD-188-110C<sup>[3]</sup>.

The software also serves Research and Development (R&D) requirements in pursuit of software modem development of additional MIL-STD and STANAG waveforms. MIL-STD-188-110A has been superseded by MIL-STD-188-110B<sup>[2]</sup> which has recently been superseded by MIL-STD-188-110C, however the 110A ST waveform specification has remained unchanged. The 110A ST waveform is one of a number of MIL-STD and NATO Standard (STANAG) ST waveforms, others of interest include MIL-STD-188-110B Appendix C, STANAG S4285<sup>[4]</sup>, S4415<sup>[5]</sup>, S4529<sup>[6]</sup> and S4539<sup>[7]</sup> which all has specific uses and are all planned for software modem development and integration into the DMT application.

In conjunction with a properly configured computer system and viable sound device and properly configured HF SSB transceiver as detailed herein, the M110A DMT software supports basic FEC based ASYNC (with EOM) or SYNC (with or without EOM) modes interoperable with hardware modems for FEC based ASCII message communications using the internal terminal or an external terminal application via the data port. Also supported is an asynchronous serial port compatible SYNC (with EOM) Over The Air (OTA) mode for transparent (TRANS) data port external communications applications in support of binary data streams. To communicate with hardware modems via any SYNC selection in the DMT requires the hardware modem used to be wired to an RS-232 Synchronous port and placed into SYNC mode or wired for RS-232 Asynchronous and placed into "High Speed ASYNC" mode if provided by the hardware modem.

This software has been tested against MIL-STD hardware modems under dumb terminal control for FEC operation using the hardware modems ASYNC and High Speed ASYNC (SYNC compatible over the air (OTA) mode) and found to be compatible. In addition the same external

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terminal software applications have been used with both the hardware modem and the MS110A DMT via the external data port in testing.

The 110A ST waveform and other MIL-STD/STANAG waveforms perform best with an HF SSB 100w or greater RF output radio having a Temperature Compensated Oscillator (TXCO) and IF passband of at least 2750Hz (nominal 3 kHz bandwidth) at the 2 dB points with low group delay as detailed in STANAG 4203 (and the radio performance specifications detailed in MIL-STD-188-141C<sup>[9]</sup>) which specifies the properties of military radios which receives the bassband audio signal (3 kHz bandwidth) from the modem and modulates it onto a carrier. At present, the bulk of MARS members will likely be limited to 600bps 110A ST operation with consistent results largely due to their radio filters not meeting S4203 parameters. However reliable 600bps FEC operation is a huge step forward for MARS FEC communications and the use of 110A ST FEC is just the first FEC ST protocol MARS has started using.

In keeping with MIL-STD-188-110B, Appendix E, a Rockwell MDM-3001 single channel modem with a two path HF channel simulator per CCIR-549-3<sup>[10]</sup> is used to verify compliance with the minimum serial tone modem performance requirements in a 3Khz AWGN channel. At present the beta candidate comes close to meeting the minimum performance requirements in testing with a single sound device looping back through the MDM-3001 to eliminate sound device sample clock error and phase distortion issues. This configuration presents the worst case as to noise floor and jitter and other PC based artifacts when testing is conducted using a laptop and AC'97 on-board sound device which is always used in addition to the best case use of an external USB sound device in all use of the HF channel simulator. Testing is conducted using Windows XP/Home, Windows XP Pro and Windows 7/64 where operating system latency issues are mitigated, such as disabling wireless networking and all other offending items. All comments regarding the software modem meeting minimum performance refer to testing with the HF Channel Simulator as the control environment and the single AC'97 on board sound device and minimum OS latency.

**CURRENT M110A DMT FEATURES**

- PC Sound Device for selection is provided by both Name and Number with no need to shutdown and restart the software.
- Supports multiple USB 2.0 or greater devices of the same or different make/model for clean selection. Where for example, if using two Signalink USB devices, the 2nd device shows up as "USB Codec 2".
- Provides automatic sample clock error correction that may mitigate 75ppm sample clock error when monitoring a hardware modem based stations transmissions. However, overall sample clock error of less than 20ppm is required for good results, less than 10ppm is required for excellent results with 1ppm or less sample clock error providing the best performance. Sample clock error is accumulative between stations, for example, if station A has +10ppm error and station B has -10ppm error, the overall error is 20ppm.
- MIL-STD-188-110A standard 1800hz PSK carrier is used by default. However optional PSK carrier selections used in many hardware modems of 1650hz and 1500hz are also provided where the use of the optional PSK carrier to center signals within narrower SSB filters must be offset from the dial frequency of stations using 1800hz by the difference.
- The MIL-STD-188-110A symbol rate is fixed at 2400bps requiring 2.7khz IF filters minimum to meet full performance specifications. Less than 2.7khz BW filters will impede the performance of data rates higher than 600bps.
- An external Data Port is provided in support of 3<sup>rd</sup> Party Applications for two-way communications via an RS-232 serial port for data, but not remote control at this time. The Data Port defaults at 19,200 baud with fixed parameters of 8N1 and CTS/RTS handshaking. Any baud rate from 1200 to 240,000 is supported for selection. Communication port support is limited to COM1..COM16 at present. The DTR line is asserted to indicate an active PTT state. The Data Port supports RS-232 Asynchronous interfacing providing both ASYNC and compatible SYNC Over-the-air (OTA) operation.
- The software is compatible with MIL-STD-188-110A hardware modem SYNC (EOM enabled or disabled) or ASYNC (EOM enabled) modes in support of FEC based ASCII dumb terminal messages using the internal terminal or Data Port based external terminals.
- Also provided is SYNC with EOM enabled transparent operation via the Data Port for external applications use for Binary, ASCII or mixed Binary/ASCII data streams.
- Modem receiver reset is provided via either or combination or EOM sequence, Flush Null Character and Reset on 100 errors per unknown data.
- Exceeds MIL-STD-188-110A modem receiver error handling of +/-75hz for all data rates. Meets Doppler spread of 3.5hz/sec slope at 75bps but not yet at the higher data rates.

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- Meets MIL-STD SNR AWGN 3Khz channel with multipath delay and fading parameters and doppler spread (but not slope rate aside from 75bps) at 150-600bps and comes close above 600bps.
- At present all messages are sent from the terminal via keyboard entry or by pasting only.
- At present both RS232 RTS or DTR and CAT PTT are sent at the same time.
- RTS or DTR is supported for hardware PTT where RTS is the default.
- RS-232 D.C. Power from DTR or RTS on which ever line is not selected for PTT.
- ICOM CAT PTT is at radio address 00h which supports all ICOM ham radio models.
- KENWOOD CAT PTT when selected uses TX; for PTT ON and RX; for PTT OFF, thus Kenwood radios with data port CAT PTT commands are not directly supported. MARS-ALE Radio Emulation (REM) is recommended for all other radio CAT PTT needs.
- ICOM or KENWOOD CAT PTT can be selected for use with MARS-ALE Radio Emulation (REM) CAT Radio Server to control any make/model radio selection and PTT type which that software supports.
- Radio CAT Comport defaults at 19,200 baud with fixed parameters of 8N1. Any baud rate from 1200 to 240,000 is supported for selection. Communication port support is limited to COM1..COM16 at present.
- At present the M110A ASYNC mode TX/RX data parameters are fixed at 8 data bits, No Parity, 1 Stop Bit. The M110A SYNC mode is always just 8 data bits.
- Supports mix and match of sound devices for RX/TX.
- All sound device output and input level settings for the modem are achieved using the Windows Volume Control Panels at present.
- Provides modem base band modulation adjustment from 30..90% with 75% the default.

## **FEATURES PLANNED FOR DEVELOPMENT**

- Additional supported modems, S4285 currently under development.
- Data Link Protocol layer support in addition to ASYNC and SYNC modes.
- Graphical Constellation and Channel Impulse Response (CIR) displays for both analysis of the radio equipment used to transmit and receive the waveforms and on going HF channel state analysis of multipath, fading and other conditions affecting the channel.
- Test Waveform (2400bps) to characterize signal path between modem/radio pairs to determine SNR, available bandwidth, radio ALC and AGC effects and distortion from non-linear effects with CIR display of I and Q data and other parametric information.



- Sync on Data (SoD) for late entry and recovery where known data is used to synchronize.
- Additional direct radio remote control support.

## **REFERENCES**

- [1] MIL-STD-188-110A, "INTEROPERABILITY AND PERFORMANCE STANDARDS FOR DATA MODEMS", 30 SEPTEMBER 1991
- [2] MIL-STD-188-110B, "INTEROPERABILITY AND PERFORMANCE STANDARDS FOR DATA MODEMS", 27 APRIL 2000
- [3] MIL-STD-188-110C, "INTEROPERABILITY AND PERFORMANCE STANDARDS FOR DATA MODEMS", 23 SEPTEMBER 2011
- [4] NATO STANAG 4285, "CHARACTERISTICS OF 1200/2400/3600 BITS PER SECOND SINGLE TONE MODULATORS/DEMODULATORS FOR HF RADIO LINKS", BRUSSELS, EDITION 1, 17 FEBRUARY 1989
- [5] NATO STANAG 4415, "CHARACTERISTICS OF A ROBUST, NON-HOPPING, SERIAL-TONE MODULATOR/DEMODULATOR FOR SEVERELY DEGRADED HF RADIO LINKS", EDITION 1, 21 JUNE 1999
- [6] NATO STANAG 4529, "CHARACTERISTICS FOR SINGLE TONE MODULATORS/DEMODULATORS FOR MARITIME HF RADIO LINKS WITH 1240 HZ BANDWIDTHS", EDITION 1, 20 JANUARY 1998
- [7] STANAG 4539 C3 (EDITION 1), TECHNICAL STANDARDS FOR NON-HOPPING HF COMMUNICATIONS WAVEFORMS, 8 JUNE 2006
- [8] STANAG 4203 C3 (EDITION 3), TECHNICAL STANDARDS FOR SINGLE CHANNEL HF RADIO EQUIPMENT, 27 APRIL 2007, ANNEX B and ANNEX C.
- [9] MIL-STD-188-141C, INTEROPERABILITY AND PERFORMANCE STANDARDS FOR MEDIUM AND HIGH FREQUENCY RADIO SYSTEMS, 25 July 2011
- [10] CCIR REGULATIONS, VOLUME III, ITU, "HF IONOSPHERIC CHANNEL SIMULATORS", CCIR REPORT 549-3, RECOMMENDATIONS AND REPORTS OF THE CCIR, ANNEX TO VOL. 3, PP. 47-58, GENEVA, 1990 0-7803-5538-5/99
- [11] W. N. FURMAN, J. W. NIETO, "UNDERSTANDING HF CHANNEL SIMULATOR REQUIREMENTS IN ORDER TO REDUCE HF MODEM PERFORMANCE MEASUREMENT VARIABILITY," PROCEEDINGS OF HF01, NORDIC HF CONFERENCE, FARO, SWEDEN, AUGUST 2001.

**NOTE:** Many of the above referenced documents can be found at many places on the Internet, to include:

<https://assist.daps.dla.mil/quicksearch/>

<http://groups.yahoo.com/group/MARS-ALE/files/>

## **INSTALLING M110A DMT FOR THE FIRST TIME**

The software may be installed to any Windows directory on your computer using Windows 2000 or Windows XP versions. At present the install program, INSTALL.EXE, by default installs to “C:\PROGRAM FILES\MARS-M110A-DMT\”. Do **NOT** install the software under “C:\PROGRAM FILES\” when using either MS-Vista, MS-Windows 7 or later due to file access restrictions imposed with these operating systems and be sure to install using Administrator rights. Also be sure to set the installed program .EXE file to run as an XP SP2 or later application.

The required installation steps are:

1. Either unzip the distribution archive file M110Ax\_FI.ZIP into a temporary directory or proceed to step 2 directly from the .ZIP file.
2. Run the INSTALL.EXE first and accept the license agreement, then follow all the on screen prompts through to completion.
3. Next the Microsoft VS 2008PRO redistribution install (vcredist\_x86.exe) will automatically start, accept the license agreement, then follow all the on screen prompts through to completion.

NOTE: If the INSTALL.EXE used is not the most recent, then before running the M110Ax.EXE file, acquire partial install archive (M110Ax\_PI.ZIP) and run the INSTALL.EXE or just unzip the contents of an update archive (M110Ax\_UD.ZIP) into the sub directory where M110Ax.EXE was originally installed on your computer.

The software is now installed and awaiting configuration, skip the next section and proceed to STARTING THE PROGRAM.

## **INSTALLING AN M110A DMT UPDATE**

The process for installation of a software update can be one of two methods detailed below.

**NOTE:** If you have NOT previously installed an earlier M110A DMT version, then you **MUST** run the current full install of M110A DMT before installing an update version.

1. With an existing M110A DMT installation that has been properly configured and tested, make sure you save the existing DMT.DAT file to a safe location as the update may have a changed DMT.DAT file structure, thus your existing DMT.DAT file will be considered corrupt and will be over written if present by the new version when started.
2. Either backup your existing M110A DMT installation before you modify it or create a new sub directory to copy over existing files with updated files.
3. Depending on the update distribution contents, either run the INSTALL.EXE contained in the partial install archive (M110Ax\_PI.ZIP) update or unzip the files contents of an update archive (M110Ax\_UD.ZIP) into the sub directory where M110Ax.EXE was originally installed. Users of Vista or Windows 7 should not installed into additional directories without first uninstalling as the log files in the originally directory will likely be used by both installations.

## TESTING FOR SYSTEM LATENCY

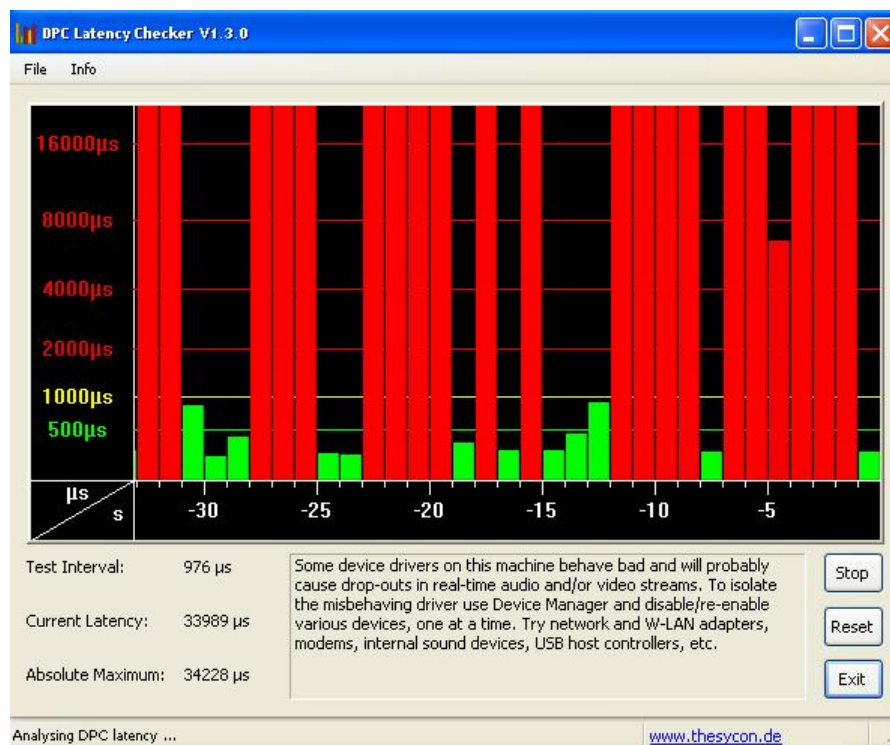
The levels of Windows latency must be tested and any excessive latency must be mitigated to within tolerances for real-time audio streaming applications. Windows is not a true real-time OS providing low-latency response to events, predictable time-based scheduling or explicit resource allocation. Windows was designed to maximize aggregate throughput and to achieve approximately fair sharing of resources.

When a Windows device driver cannot process data immediately in its Interrupt Service Routine (ISR) it schedules a Deferred Procedure Call (DPC), both of which operate at the highest priority code that runs in the system. Neither an ISR or DCP can be pre-empted by the operating system, they each run to process completion. ISRs and DPCs that run too often or too long degrade system performance by using too much CPU time and can cause audio timing issues, serial port timing issues, video pauses, erratic mouse behavior and numerous other system problems.

The recommended tool for audio latency testing of DPCs which can be used under Windows 2000 through Windows 7 can be downloaded from:

[http://www.thesycon.de/deu/latency\\_check.shtml](http://www.thesycon.de/deu/latency_check.shtml)

Here is the DPC tool is running on an XP Home system with excessive latency, which requires mitigation prior to use of the software.



The process of determining what process is the offending one requires motoring all processes with Task Manager to associate the timing of CPU loading or RAM increases with latency spikes and disabling the process to see the results on latency. When you look at Windows Task Manager and sort the running processes by CPU (Processor Utilization), the System Idle Process is almost always at the top of the list. What you may not know is that “process” is really a roll-up of several

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things. Among other things, included in that CPU number, is hardware interrupts and DPCs. You can see these two items by using the Microsoft “SysInternals” Process Explorer available here:

<http://technet.microsoft.com/en-us/sysinternals/bb896653.aspx>

After you have disabled a process watch DPC Latency Checker. When excessive latency values disappear you found the responsible device driver. If there are still exceptional large DPC latencies try the next process. You will likely see large spikes of short duration when opening and closing applications and or files and during mouse moment, continual such spikes would be a negative issue during use of the software. The M110A-DMT program itself while running will cause latency issues.

It has been found that network card drivers and wireless networking especially and various programs making use of network connections, such as many e-mail clients, are the largest issues regarding latency. Any plug-in devices are suspect, to include your sound device, including your on board sound device. Other negative processes affecting latency are firewalls, anti-virus software, anti-malware software, web browsers, email, screen savers, system hibernate, resources that power down, any process that automatically starts whenever it wants and other processes that load CPU and RAM. If a PC can not be dedicated to running the software it recommended that disabling wireless networking and anything found to be a latency issue and that a start up check list be created and checked prior to use of the software in disabling all such processes. A decent reference on many more issues and OS versions can be found at:

[http://www.tweakheadz.com/troubleshooting\\_audio\\_cards.htm](http://www.tweakheadz.com/troubleshooting_audio_cards.htm)

Below, the same XP box after disabling Wireless Networking, the amount of latency is now well within tolerance.



Whatever is found to be an issue must be disabled. If it is something that you feel you must have working during MIL-STD modem use of the PC you will need to find a substitute that is under 1000us in latency.

Another test program that could not run and document as it requires Vista or greater can be found at:

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

This tool is suppose to identify both DPC and ISR issues down to the offending process, thus it should be a better tool for those with Vista or greater to track down the processes that require disabling.

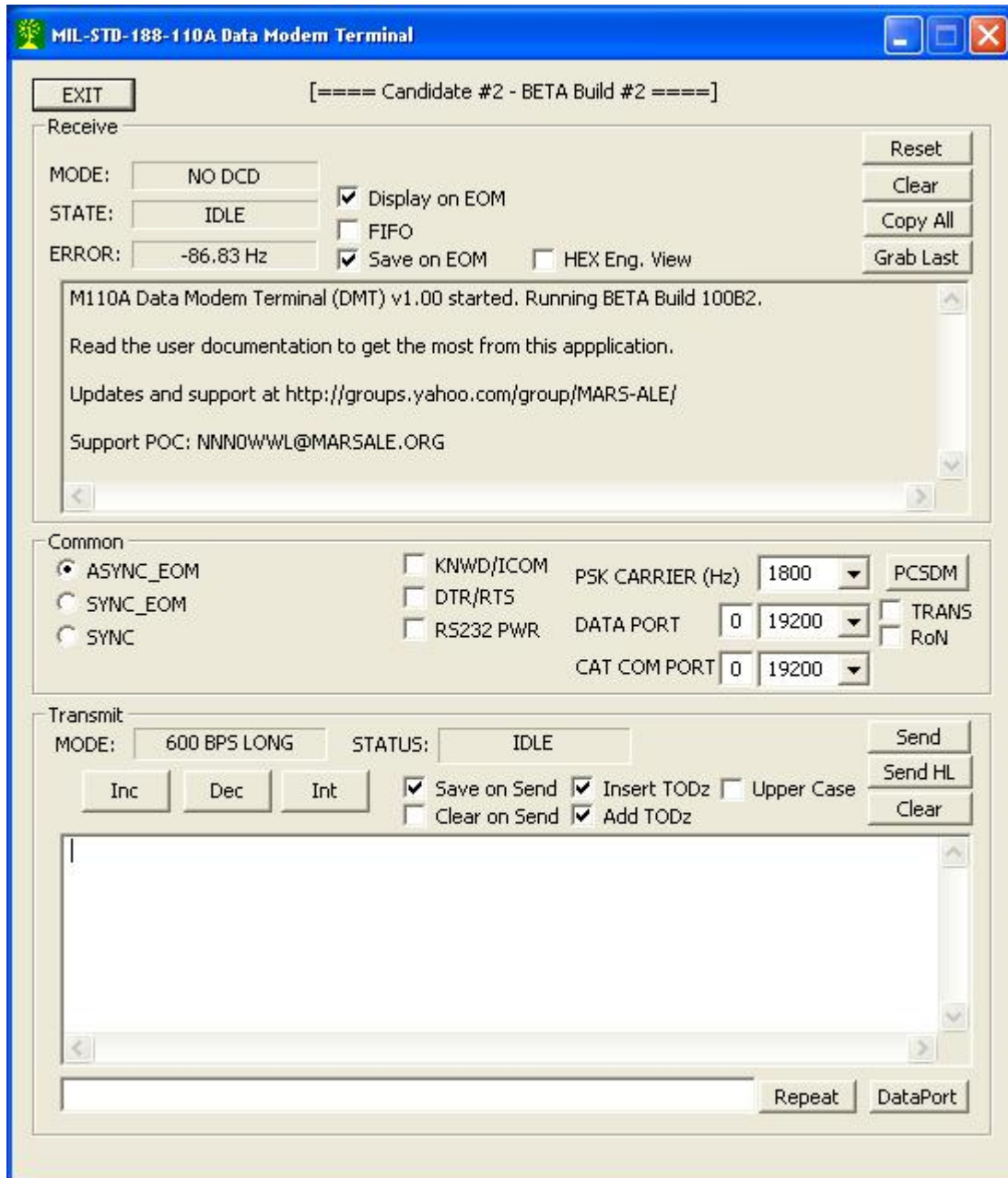
Under Windows XP to achieve better Latency navigate My Computer > Properties > Advanced > Advanced > Performance options. Optimize performance for Background Services, not Applications which can enhance sound device driver performance.



## STARTING THE PROGRAM

When the program is started for the first time after initial install or update it may generate a “Windows Security Alert”, if so just click just on “Unblock”. In addition, Anti-Virus or other such tools may also need adjusting to allow the M110ADMT.EXE executable to run.

When the software is started for the first time a DMT.DAT file will be created with default configuration settings, the main screen will appear as follows:

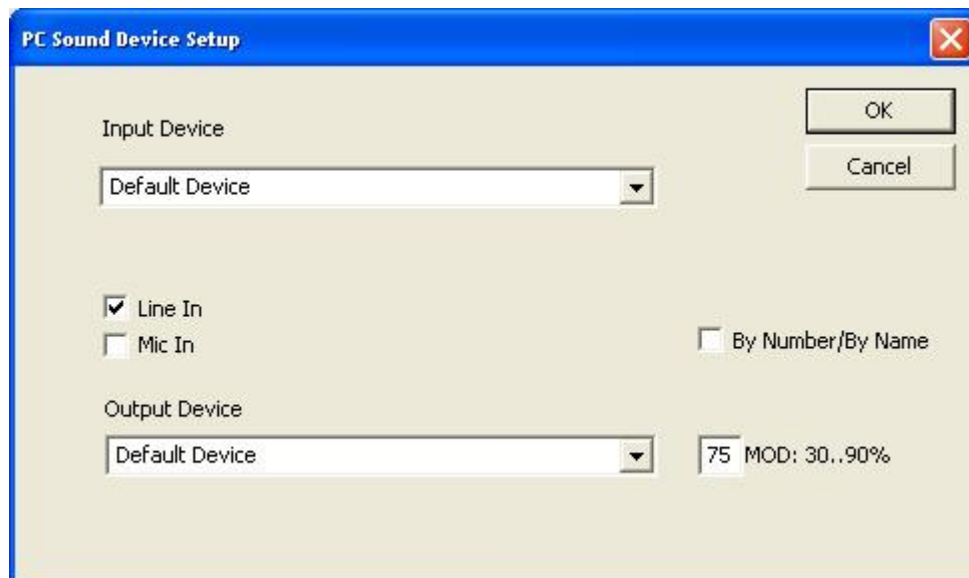


**WARNING:** Never use the DMT tool without using an external terminal application if the DATA PORT is not set to 0, otherwise internal display of incoming data will display very slowly.

**NOTE:** The program is not resizable, if running on a Netbook or other PC where resizing is required we recommend: <http://www.digitallis.co.uk/pc/downloads/ResizeEnable.zip> When ResizeEnable is started, the icon will show up in the taskbar near the clock. Then, window edges can be dragged larger or smaller. I use it often with some of the much-too-small Windows XP boxes, such as the one that shows up under Explorer's Tools > Options > File Types.

The main screen of the DMT software is divided into three panels from top to bottom; "Receive", "Common" and "Transmit" which segments the software into a clean user interface. The program is easy to use, just select the mode and parameters and paste and send your message. All user selected settings are saved each time program is properly terminated. The "EXIT" button is used to properly terminate the application.

The first configuration step should be the selection of your PC Sound Device, do so by either clicking on modem in the Common panel or by Right Clicking on the program bar at screen top and clicking on "PC Sound Device Setup", either methods will display the following screen:



This dialog supports the selection of the PC Sound Device to be used as the modem for the software. The device selected should be a sound device dedicated to radio communications and not the system default sound device.

At first use either Default Device or perhaps nothing will be displayed for the Input and or Output devices, either indicates the default device will be used. The devices can be displayed and selected by one of two means, by Number or by Name, where by Name is the default, to select by Number the "By Number/By Name" check box must be checked, which is usually only required for some sound device drivers under older versions of Windows. The same device should be used for both the Input and the Output device.

**NOTE:** Under Windows Vista or Windows 7 the sound card inputs are only detected when a cable or microphone is plugged in.



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By default Line In is selected, but can be changed. Line In is preferred over MIC In for the radio RX audio.

After selecting the device of interest click Ok, there is no need to shut down and restart the program

If your selected device is not displayed upon return to the dialog or on restart, then try the other “By Number/By Name” If neither methods work, there is a driver issue under your OS and either another driver will need to be tried or your device will need to be made the default sound device under Windows for now.

The setup of your RX and TX device levels at this time is via the Windows level panels for the sound device selected as the modem.

You may also want to maintain a database of sound device settings if you plan to make use of different sound device modem based communications applications and forego just setting the output sound panel level to max so that you return to the proper settings. The following tools are popular for this purpose:

- QuickMix (<http://www.ptpart.co.uk/assets/unsupported/quickmixin.zip>)
- RoMac Sound Card Manager (<http://www.romacsoftware.com/SoundManagement.htm>)

The next sections provide details to consider regarding sound device selection and configuration. Thereafter the program interface and features will be detailed.

## PC SOUND DEVICE SELECTION

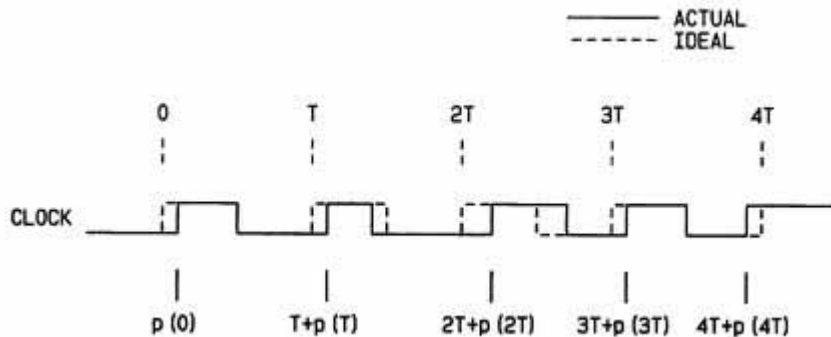
All PC Sound Devices (a.k.a. Sound Cards) are not created equal and all operating systems do not treat the same sound device equal. There are a number of important considerations when selecting the sound device for use as the hardware modem portion (PC Sound Device Modem) of this software solution.

This software requires an Audio Codec 97 (AC'97) compliant or better mono sound device with a native sample rate of 48k minimum at a 16 bit sample size minimum and at least 100-4,000hz flat frequency response. Details are provided on sound device driver configuration in the next section, which cover devices with greater raw maximum sample rate than 48k.

The dynamic range and frequency response of the average sound device is more than adequate for the software modem needs, however sound device must provide:

- An accurate sample clock on input and output
- Low sample clock thermal drift stability
- High Signal to Noise Ratio
- Total Harmonic Distortion + Noise
- Low deterministic phase jitter

Low deterministic phase jitter is a big issue in digital audio system related with timing variation in variable clock position. If you have big/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.



The sound device characteristics will determine its usefulness as a PC Sound Device Modem for use with the software due to the complex 110A waveform and its constant 2400bps symbol rate, decoding 110A and similar waveforms to be supported by the software. 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 decoding errors and loss of synchronization. Never have any external PC Sound Device Modem resting on top of equipments that heat and or cool and never have them in direct sun light, a stable thermal environment means less drift.

The following categories of sound device solutions will support different maximum reliable data rate operation with the range the specified channel conditions performance requirements of the standards as follows:

- On-the-Board sound device chipsets can often provide reliable support of 75bps to 600bps for 110A class waveform data rates at most due to noise floor and deterministic phase jitter. Interfaces and cabling between the PC and radio are often susceptible to RF interference and are also often sources of EMI.
- Plug-In (e.g. PCI card, PCMCIA CardBus Type II) sound devices can support 75bps to 600bps and often 1200bps for 110A class waveform data rates reliably. In some cases 2400bps may be found reliable. These devices have somewhat less noise and phase jitter than on-the-board devices. Interfaces and cabling between the PC and radio are often susceptible to RF interference and are often sources of EMI.
- Outboard sound devices (e.g. USB port (without built in hub) and IEEE 1394 (FireWire) sound devices) offer the lower noise and lower phase jitter than most internal devices. However USB devices do suffer from additional USB port latency issues. Be sure the USB sound device is on its own USB port, using high speed USB 2.0 or higher ports (USB 3.0 and eSATA/USB combo ports) and using short, double shielded USB cables also reduce any USB audio latency. According to Listen Inc., Firewire audio devices appear to have a problem with computers that use NEC or Agere Firewire chipsets, where Texas Instruments chipsets are preferred and Toshiba chipsets are found to work well. Radio communications interfaces with sound devices and the manufacturers cabling (e.g. US Navigator, Signalink-USB) designed specifically for use with HF radio communications will offer higher immunity to an RF environment than do general purpose sound devices in the same cost range. There are many expensive Outboard professional sound devices that are very well shielded and offer TXCO referenced sample clocks with many other outstanding features. Outboard devices should provide reliable operation through 2400bps for 110A class waveform data rates. Outboard sound devices should support the higher coded data rates of 110B Appendix C and S4539 waveforms when supported in the software.
- HF SSB radios providing USB port based internal Codec (e.g. IC-7200, TS-590) support via Windows device driver offer native 48K sample rate 16 bit support with even lower latency, noise and phase jitter than external USB sound devices. This approach offers an even higher immunity to RF and offers a single cable interface solution for audio and CAT control. They provide reliable operation through 2400bps for 110A class waveform data rates and should support the higher coded data rates of 110B Appendix C and S4539 waveforms when supported in the software.
- FlexRadio systems (post SDR-1000) with PowerSDR and Virtual Audio Cable (VAC) and TXCO provides audio in the digital domain utilizing high performance A/D and D/A converters instead of a conventional Windows driver based PC Sound Device which results in the most PC stable and cleanest PC digital mode operation possible and highest RF immunity and thus provides the most reliable support next to a hardware modem. These systems support 75bps thru 2400bps for 110A class waveform data rates and will likely support all the higher coded data rates of 110B Appendix C and S4539 waveforms when supported in the software. For FlexRadio setup see <http://w3jjj.com/m110a.html> and [http://groups.yahoo.com/group/MARS\\_Flexers/](http://groups.yahoo.com/group/MARS_Flexers/)

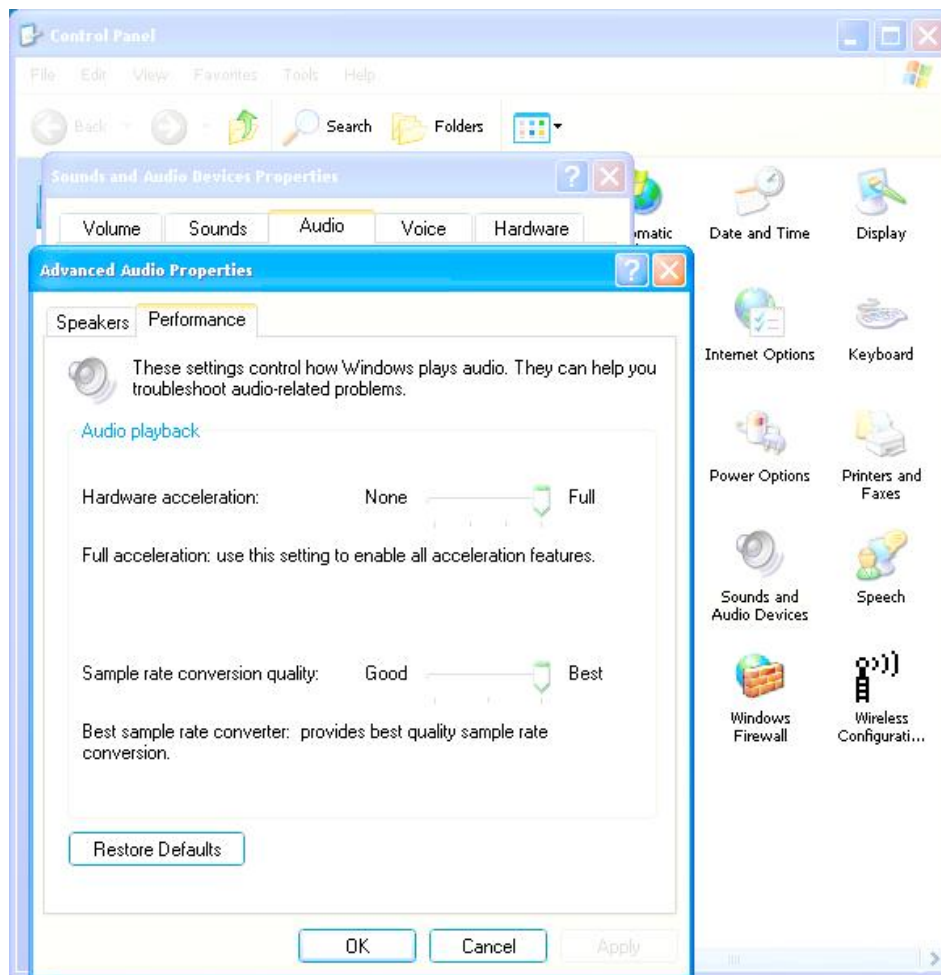
Based on these categories of sound devices, the common range of supported reliable data rates among all sound device types is 75bps-600bps. Moving away from On-the-Board and Plug-In sound devices raises the bar to the full range of 75bps to 2400bps for coded 110A class

waveforms, however the radio's SSB filter characteristics can also be a limiting factor to maximum data rate.

## PC SOUND DEVICE DRIVER

All sound devices under Windows (and most all operating systems) require a device driver. A driver is a low-level program that handles the data connections between the physical hardware and the operating system. Traditionally the manufacturers of sound devices provide drivers for their sound device hardware or family of sound device hardware. However, since Microsoft's Universal Audio Architecture (UAA) initiative, which supports HD Audio, FireWire and USB audio device class standards, a universal class driver by Microsoft can be used.

It is usually best to use the latest specific manufacturers driver for your sound device if one exists, where all features not required for modem use are disabled, if possible and other parameters are configured for the best modem use performance. The use of the latest driver may even lower your sound device sample clock error which is a critical parameter in the operation of this software.

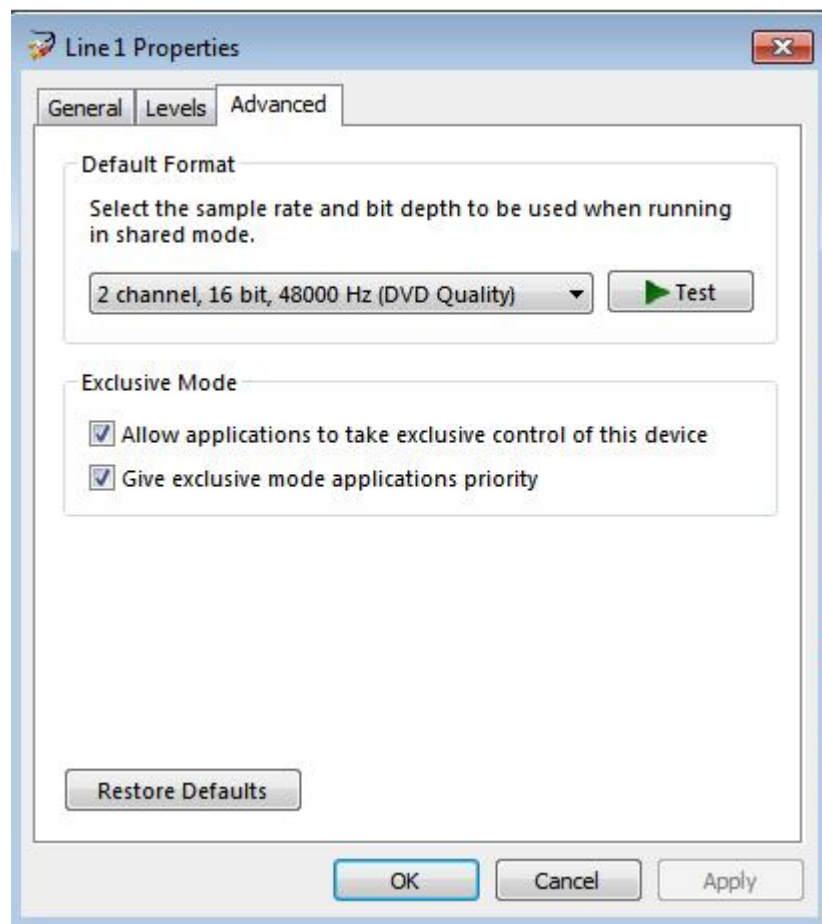


Although PC sound devices appear to support different sample rates, they actually have only one raw hardware sample rate which is the maximum rate that can be used for modem applications and achieve the effect of lower (and some times higher) sample rates by means of sample rate conversion. Such conversion is performed by the sound device driver and cannot be monitored or disabled by application software. The sample rate conversion may introduce certain low-level distortions to the signal which preclude usefulness for modem applications, this is especially true

of up conversion. In addition, most sound device drivers provide user selected levels of sample conversion to mitigate overhead in processing.

Should your sound device support a higher native maximum sample rate (e.g. 96k or better) you must select the best quality sample rate conversion in the sound device driver setup. However the use of sample conversion from a lower native raw sample rate, such as 44.1k or lower to achieve 48k must be avoided. In addition, should your sound device provide for Hardware Acceleration in its driver parameters, select full acceleration. To check these parameters select Control Panel > Sounds and Audio Devices > Audio > Advanced (Advanced Audio Properties) > Performance as seen below under Windows XP Home as seen above.

In addition, default sample rate selection under the driver properties is very important on Windows Vista and Windows 7 otherwise sample rate variance or "Error opening sound device" may occur, if your sound device driver (aside from USB port devices which are set to 48k default) provides for selection of 48k, select it as the default format. In Windows 7 the selection required as seen below is 2 channel, 16 bit, 48000Hz. Should there be any reason to run an additional program using the sound device during the use of this software, it too must use a 48k sample rate in Windows 7.



In Windows 7 share mode streams run in low-latency mode. The audio engine runs in pull mode with a significant reduction in latency. This is very useful for communication applications that require low audio stream latency for faster streaming.

At present this software uses one channel at a 16 bit sample size and 48k raw sample rate only and performs high quality sample conversion between 48k and the 9.6k sample rate required by the modem. Additional raw sample rates of 96k and 192k and additional sample sizes of 24 and 32 bits are planned for user selection so that the user can select the maximum native sample rate their sound device supports without software resampling coming into play and the best sample size. When the software provides greater sample size selections the user will need to note their sound device dynamic range to make the best choice.

Many 24-bit cards only have a dynamic range (Signal-to-Noise Ratio) of about 100 db, which actually corresponds to a resolution of only 17-bits. Whereas a 16-bit sample size theoretically provides 96 dB of dynamic range. In other words, the additional bits are used to digitize noise. Thus using 24-bit cards at the currently coded 16-bit resolution in this case is better than using 24-bit resolution as there is no additional noise. Thus when 24-bit and higher resolution is coded, it will be up to the user to determine what selection is best with their hardware.

**NOTE:** Under Windows 7 and likely later, the DMT application must be the first and preferably the only application running that uses the sound device. If the DMT application is not first and the first application must be using the required 48khz sample rate as used by the DMT, otherwise aliasing will occur which will degrade modem performance on both transmit and receive.

## **PC SOUND DEVICE MODEM TESTNG**

The requirement for synchronization clocks between 110A ST terminals is 10ppm of the sample clock. Clock accuracy for generation of the 1800-Hz carrier shall be within  $\pm 1$  Hz. The symbol rate of 2400 symbols-per-second requires accuracy of 0.024 (10 ppm) symbols-per-second. Thus the PC Sound Device sample clock accuracy is very important.

The software user needs to do everything possible to achieve the lowest possible PC Sound Device sample clock error (as well as low drift, low noise and low jitter, see Appendix E). There is no support for user entered sample clock error factors pertaining sound device sample clock error for calibration to the sound device deficiencies. The software is coded to provide nearly  $\pm 5$ hz (almost  $\pm 100$ ppm) of automatic raw sample rate error correction in the decimation/interpolation between the sound device raw 48000hz (48k) sample rate and the 9600hz (9.6k) sample rate of the modem. However, due to the constant 2400bps symbol rate, when high sample clock errors exist, the underlying phase error issues can not be overcome as is the case with low symbol rate modes typically used in Amateur Radio with sound device modems, thus the results expected from the 100ppm range of correction is not being had, about 75ppm error at most can be tolerated if working against a hardware modem based station.

If your sample clock error is not less than 75ppm, performance on RX below 300bps when a hardware modem is the source, will suffer the most and will likely be even worst with the average software modem as the transmitting source. Depending on the other sound device critical factors which can affect the higher data rates more than sample rate error, your data rate range may be limited to 300 and 600bps. Less than 20ppm overall sample clock error between stations is required for adequate results at 75bps with 0ppm of course being the best. Stations that transmit with greater than 20ppm add to the difficulties of those attempting reception. Lets take 10ppm error into account for two stations, both stations may experience clock error above or below 48000hz, say -10 ppm and +10ppm, when that error spread is both negative or positive for the two stations communicating, results are good, however when the two stations are on opposite sides of center, then results will be less due to the increased frequency spread and phase error. Should both stations be close to 0ppm as would be the case with a hardware modem, results will be optimal. Thus interoperability with a hardware modem user will usually always be better than working another software modem user unless they are both below 10ppm or have the same  $\pm$ ppm error from center.

The test program Check Sample Rate (CheckSR.EXE) from MixW is the simplest way to test sample rate error on both input and output. However making use of HF reception of WWV using various software programs is the most accurate way to test your RX sample rate error. Once your receive sample rate error is known, you can then perform a loop back test to determine your TX error.

When using a USB port based sound device, do not go through a hub unless the hub is one of the expensive TXCO or GPS referenced types (e.g. Maestro-GS22 and Maestro-iS22, see [http://www.fiberbyte.com/maestro\\_home.htm](http://www.fiberbyte.com/maestro_home.htm)) as the hub will likely introduce additional sample rate error. Check all USB ports directly on the PC to determine the port with the lowest sample rate error. If you still have a high sample rate error check the USB sound device on another PC if one is available to determine if perhaps it is the device itself that has the issue.

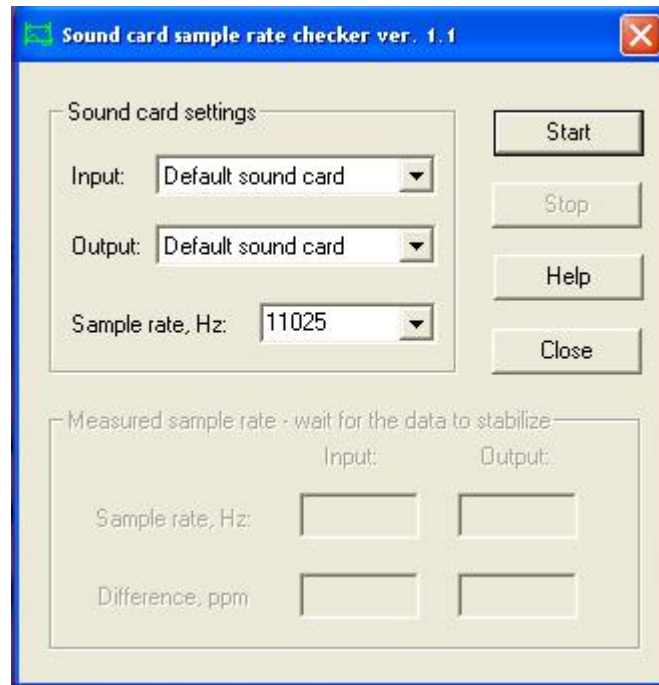


## MS110A DMT v1.00

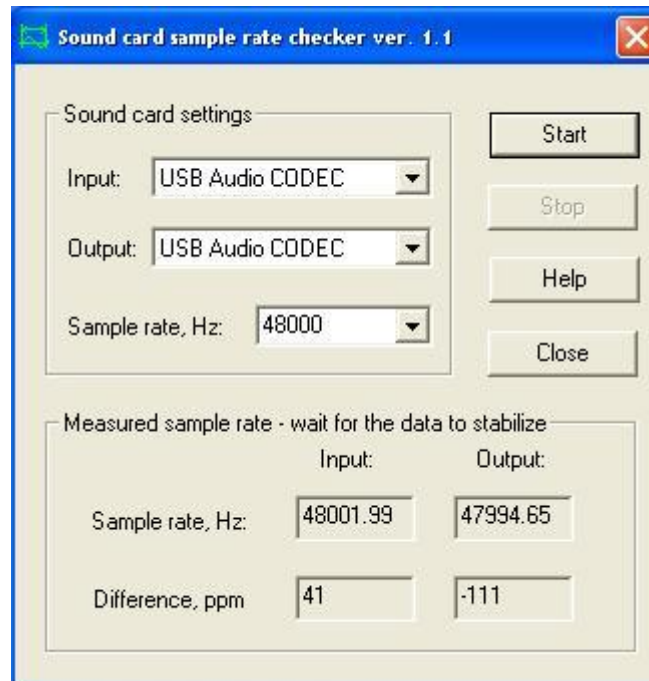
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Allow 20 minutes of warm up, then with and no sound device software running, not even the Windows sound panel and no audio input to the sound device, turn off screen saver, suspend and power saving modes, shutdown wireless networking and other latency causing processes and just leave the PC alone after starting the test. Run the CheckSR.EXE at least 30 minutes before clicking on stop.

**NOTE:** Windows Vista and later sound card inputs are only detected when a cable or microphone is plugged in.



The CheckSR program, upon startup selects the "Default sound card" and 11025hz as the sample rate. Select the Input and Output device of interest and 48000 for the test.



The resulting "Sample rate, Hz" must be less +/-5hz different from 48000 or the "Difference, ppm" must be less than +/-100ppm at 48000 for the PC Sound Device to be usable at all with this software and then only when the TX station is near 0.0ppm. In the example depicted above, 41ppm on RX is within usable tolerance and -111 on TX which is way out of tolerance for our needs.

The software supports the 48K sample rate only at present. However 96K and perhaps 192K may be supported in the future. Thus at some point, instead just using the pull down to select the 48K sample rate, you can delete the default 11025 that comes up and enter any desired sample rate manually.

**NOTE:** Some professional and even some lesser sound devices provide for an external clock source, however most devices do not. Many devices can be modified without much effort for referencing the sound device to an external frequency standard. The common 24.576Mhz (or other frequency) reference crystal used in sound devices can be swapped out for a more accurate crystal or better yet a TXCO reference can be soldered in place of the crystal. These practices have been used by many MARS members and others to achieve accurate and stable sample rates, however such practices are beyond the scope of this document.

## **LOOP BACK TEST**

There are a number of purposes of the loop back test:

- Verify that the operating system latency issues have been mitigated.
- Check the effects of sound device distortion, noise and jitter.
- Determine a relative starting point for RX and TX sound device level settings.
- Check the effects of cable wiring as to noise when the radio cables are jumpered for loop back testing.

Neither Sound Device Sample Rate Error or Stability enter into the loop back test as the same sound device is being used for TX and RX which equates to a 0ppm error. How your system performs during the loop back testing is an indication of how your system would perform using radios were you to have a 0ppm sample clock error.

The Loop Back test requires a jumper from the sound device Line Out to Line In or Mic In if required. The full benefit of the test is achieved when the actual cabling between the sound device and HF radio are jumpered for loop back verses just using a separate jumper. That being said, if there are issues when testing using the full cable set, then performing the tests with a separate jumper can be used instead of the cables to determine if system cables are an issue. It is usually with RF that issues with cables are the most prevalent.

Performing a loop back test requires two instances of the software running on the same PC. On both instances uncheck "Save on Send" and "Save on EOM". On the instance that will send the message, select the 4800bps uncoded data rate by clicking the "Inc" button in the lower Transmit panel displays "4800 BPS" in the MODE window.

As this is a loop back test, first send some short messages first and adjust TX level and RX level so that the ERROR window in the receiving instance is at its lowest reading which should be less than 0.25hz. Too much TX or RX audio or too little will cause the ERROR value to increase.

The loopbacktest.txt file in the distribution is used for the loop back test message using the 4800bps uncoded data rate as it will most easily reveal issues being as it has no FEC and no Interleaving.

Open the file with Notepad and copy the message and past it into the lower "Outgoing Message Window". Then click "Send" to start the test and do nothing else on the computer during the test. The message will take 1 minute and 10 seconds to complete if uninterrupted.

Review the loop back message on the RX instance of the software and verify that you displayed the message 100% correct. If 100%, your system (aside from possible sample clock error and drift) has passed the basic test.

You can repeat this test at all data rates using SHORT interleave, however 4800bps has more potential to detect issue's as it's the least robust. The message will take 2x plus for each lower data rate starting at 2400bps SHORT at 2 minutes and 22 seconds and 2 minutes 27 seconds for 2400bps LONG. The interleaving adds additional time beyond just the data rate change. If the 32kb test message does not pass at any data rate below 4800bps then your system has real issues requiring mitigation

## **SETTING UP TX AUDIO**

Select the same device for use as the Input and Output device using Line In and Line Out. Leave the Modulation level at the default 75% to start, only use it if the sound panel and other controls do not provide enough range in setting your radio ALC to zero. Do not use the radio MIC port for TX audio if your radio provides another port for data modem interfacing. Make sure that your MIC audio is not hot during data modem transmissions.

If possible, the TX SSB filter in the radio needs to come as close as possible to meeting STANAG 4203 requirements. If not the maximum data rate achievable will suffer. The design of the IF passband filtering determines how faithfully it reproduces complex signals. MIL-STD serial tone signals contain information in both phase state and amplitude value and occupy and instantaneous bandwidth as wide as 3Khz. The passband response alters both phase and amplitude of a complex signal. Any marginal design considerations can cause degraded performance of this complex waveform signals.

The ALC setting is significant in reducing distortion by maintaining linearity and thus improving performance. The TX Audio level for M110A DMT must not drive the radio into ALC. ALC can drive signals too high into the power amplifier, distorting the signal and possibly the modulating signal. This effectively limits the maximum SNR measured at the demodulator and may cause bit errors. For those with radios without ALC meters, no ALC typically equates to about 30-35% of your SSB voice RF output for data, however some Amateur grade and most Commercial and Military radios will often provide much more than 35%. Also be sure to power your radio from a clean, stable power source and if running off batteries, be sure that the voltage does not drop too low otherwise distorted audio and other issues may develop.

When not using the software for ALE follow on with MARS-ALE, it is recommended, if possible with your sound device interface controls, that the sound panel be used to set the TX level to max and then make use of an interfacing level control to adjust for proper TX level to allow for easy manual adjustment when changing frequency as typical Amateur Radio and Commercial Radio transceivers do not provide a constant RF output across 2-28Mhz. For ALE follow on use of the software with an ALE hardware radio, the proceeding applies. When used with MARS-ALE the process can be the same where you setup per the MARS-ALE docs with one level setting used for all channels and derive whatever output you get across 2-28Mhz. Alternately you can setup MARS-ALE per the docs to automatically adjust the sound control panel level for each frequency, in which case your interface control would be set close to maximum to achieve a constant RF output for each channel, this still requires no ALC for PSK and whatever ALC achieved for ALE FSK.

Any use of an Automatic ATU (AATU) must be made where the AATU has already been tuned to the frequency and is not being tuned by the data transmission at the start when the preamble is being sent, as such tuning will have a negative effect on the signal and thus the reception of the preamble portion of the transmission used for sync. Retuning by the AATU during transmission is also detrimental, as it will not only affect the data payload but also the known data used to maintain sync. In addition, take into account the ATU wattage rating and its duty cycles rating. For example the popular LDG tuners are rated in watts peak envelope power (PEP) and for ST modes you need to use less than half its rating as the ST modem is a high duty cycle mode, its nearly a continuous duty mode.

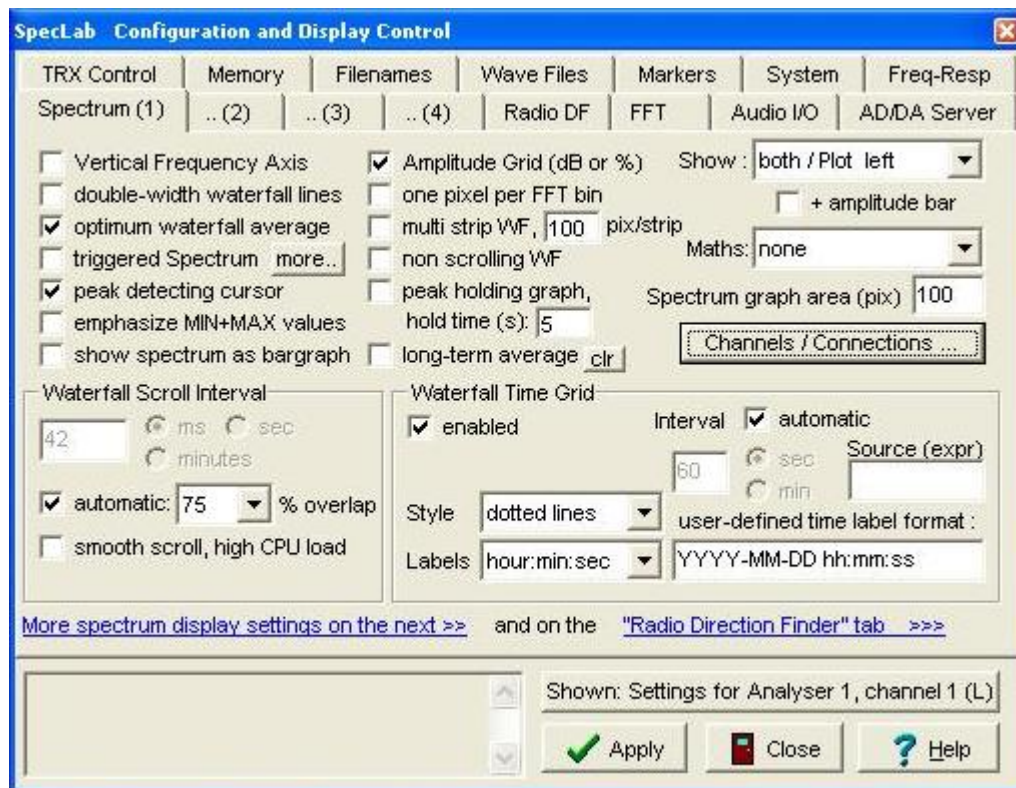
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Lastly, in a dire emergency situation use of this software via close acoustic coupling between a PC speaker output into the radio microphone will support sending a message, lower data rates of 300bps and below are recommended.

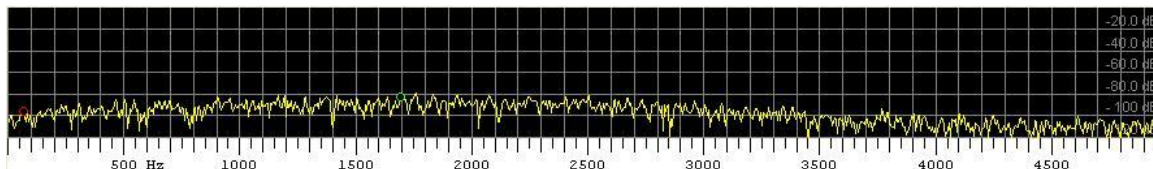
## SETTING UP RX AUDIO

Most HF radios provide a fixed radio line out at 100 mV rms @ 600 ohms either on a dedicated data modem port, on a separate port and often on the MIC port connector. Only make use of radio line out for the data modem RX audio. If you radio does not provide such a port, it is highly recommended that one be added by wiring to the high side of the radio volume control for a fixed output port regardless of the volume position.

Using Spectrum Lab (<http://www.qsl.net/dl4yhf/spectral1.html>) with a fresh install, select your sound device and then setup the Spectrum display by Right Clicking on the top part of the spectrum as follows:



Below is the fixed 100mv RX audio of an FT-890 with stock 2.4Khz filter with is IF Shift adjusted properly, the radio is on a dummy load, S0 on the meter. The fixed 100mv level requires adjusting your sound device line in level for -80db (with your radio on a dummy load).



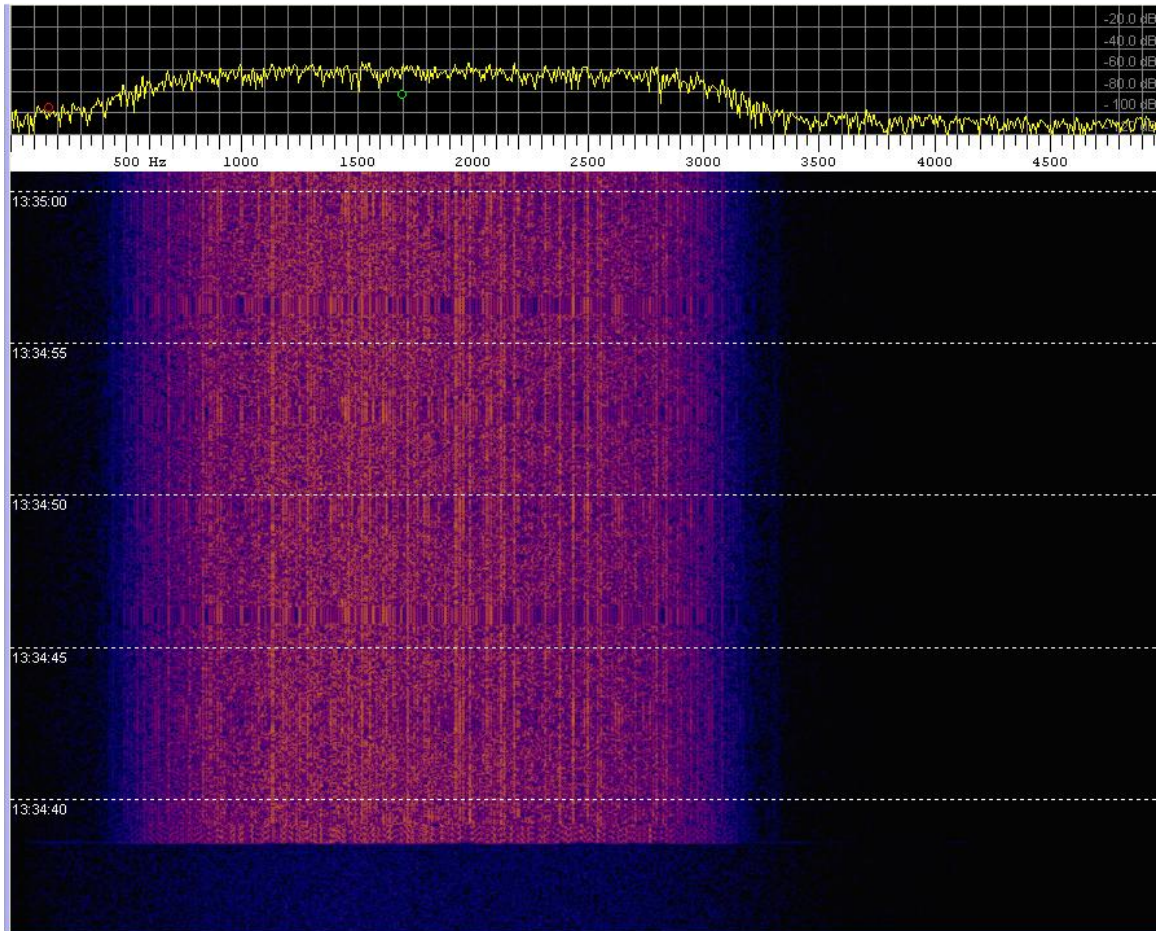
A typical incoming serial tone signal will cause an increase in radio line out by 25db regardless of actual S meter reading for an audible by ear signal. The capture below depicts a 110A 300bps signal from a Micom transceiver using its 3Khz data filter, the FT-890 receiver output displayed



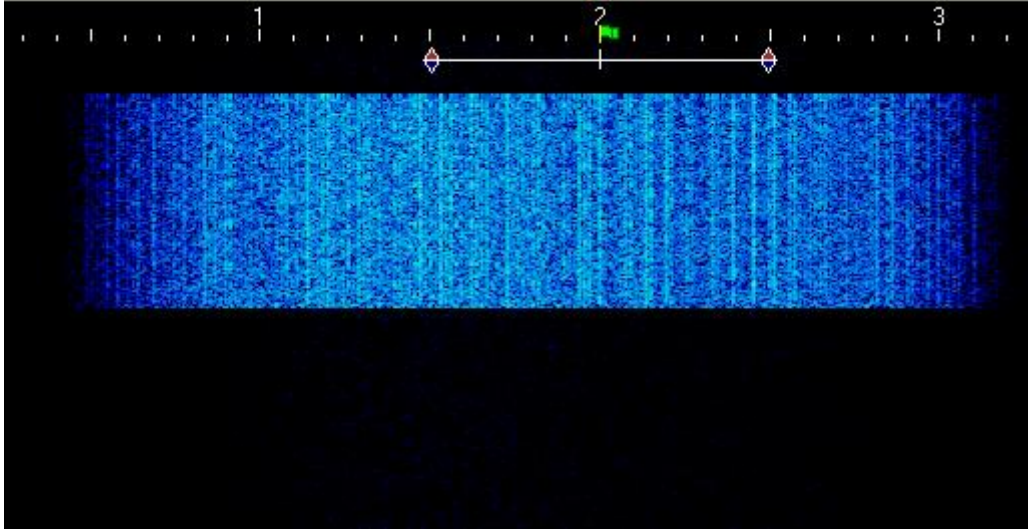
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is -55db. The actual S meter reading from the 110A signal transmitted at a power level to cause and audible signal registering around S0 and 110A signal transmitted at power levels to reach an S9 had no affect on the amplitude of the receivers fixed port output displayed.



Here is the same signal displayed using MixW with an 80db dynamic range setting and an average curve setting of normal.



You may also want to maintain a database of sound device settings if you plan to make use of different sound device modem based communications applications and forego just setting the input sound panel level to max so that you return to the proper settings. The following tools are popular for this purpose:

- QuickMix (<http://www.ptpart.co.uk/assets/unsupported/quickmixin.zip>)
- RoMac Sound Card Manager (<http://www.romacsoftware.com/SoundManagement.htm>)

Lastly, in a dire emergency situation use of this software via close acoustic coupling between a the PC Mic input from the radio speaker may support receiving a message, lower data rates of 300bps and below are recommended.



## **RADIO RECEIVER SETTINGS**

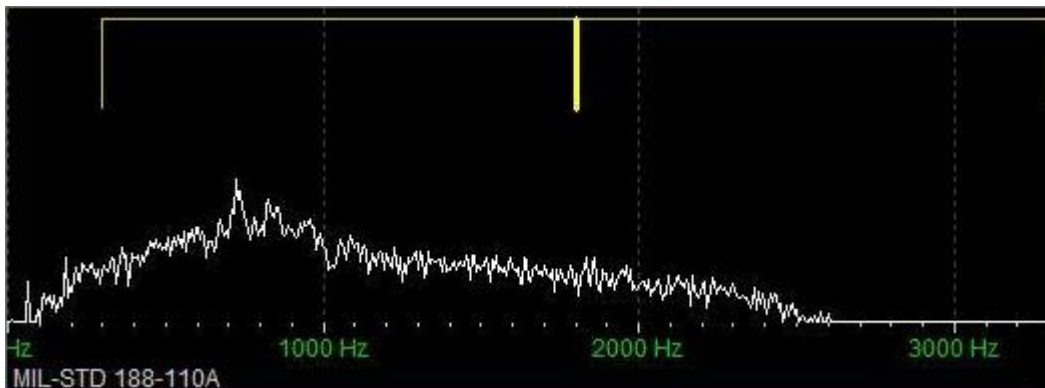
The 110A waveform occupies a bassband of 300-3300hz where STANAG 4203 (refer to Appendix F herein) specifies the IF filtering and properties of the radio which receives the audio signal (nominal 3 kHz bandwidth) from the modem and modulates it onto a carrier along with radio performance specifications that are also detailed in MIL-STD-188-141C.

Technical parameters specified in STANAG 4203 Edition 3 are:

- Radio frequency range
- Tuning
- Frequency accuracy and stability
- Frequency response and group delay time over a 3 kHz frequency band
- Phase noise
- Linearity
- Random emissions outside the frequency band and inter modulation products
- Operation mode (simplex/half-duplex)
- Switching time between transmitting and receiving
- Gain control
- Modulation of carrier wave

RX bandpass filtering is required such that, between 300 and 3,300 Hz, variations in attenuation are at most +2 dB and the group delay time over 80% of the passband must not vary by more than 0.5 ms. In addition the time constant of the automatic gain control (AGC) circuit must be more than 10ms on desensitization and less than 25 ms on resensitization. The frequency accuracy of the transmitter and receiver must be at least 10-6.

However most Amateur Radio and Commercial grade HF SSB transceivers only offer 2.4Khz filters, typically 300-2700hz. Such radios can be used with 110A at data rates of 600bps and less on TX and somewhat better on RX when using IF Shift or Passband tuning to achieve a wider RX bandwidth. Radios that have no IF SHIFT really need to have a wider filter installed. For reliable performance at data rates of 1200bps and higher the transmitting station must be using filters of 2.7-2.8Khz.



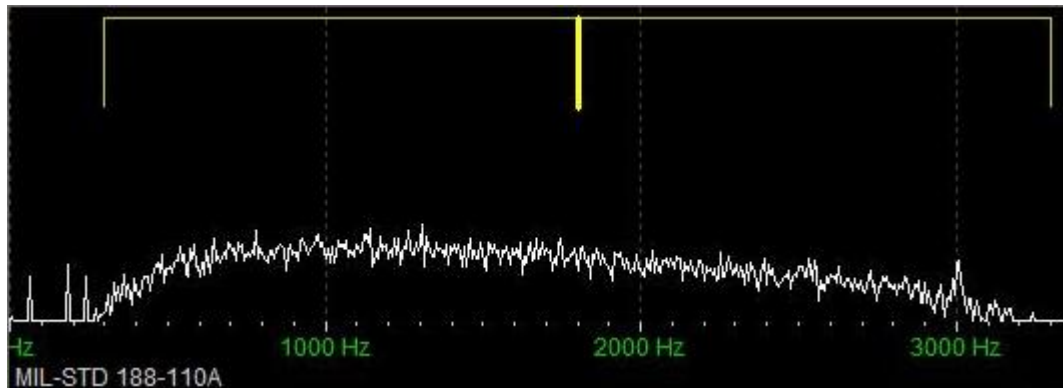
Above is the passband display of a TS-450 captured while monitoring a 4Mhz channel where its 2.4Khz first IF filter is selected and the 2nd filter is in THRU and the IF Shift set at center during

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the intercept of 110A from a station using 3Khz filtering on TX. The passband displayed below is the same TS-450 being and the same station transmitting, the difference being the IF Shift is now adjusted to above center to pass beyond 3Khz.

The vertical yellow marker in both images depicts the 110A standard 1800hz PSK carrier in the passband. Even the old TS-450 and its 2.4Khz filter when using IF shift can be adjusted to pass a full 3Khz 110A signal on RX for good decode. However this particular TS-450 will suffer as to the maximum data rates on TX as the passband will be restricted to the installed 2.4Khz filter.



Military radios run their receivers wide open, they do not provide the type of receiver features found in most Amateur and some Commercial radio equipments. In taking some of those commonly found features into consideration the following settings are recommended:

- IF BW as close to 2.8Khz on TX and RX as possible and no more than 3.0Khz on RX.
- IF Shift if equipped, set to 1800hz instead of 1500hz center if using filters under 2.8Khz.
- The longest receiver time constant of the automatic gain control (AGC) circuit must be used to prevent rapid gain changes arising from channel fades. More than 10ms on desensitization and less than 25ms on resensitization is required. Thus if AGC is selectable, SLOW or MEDIUM or perhaps even OFF should be used depending on the make/model radio.
- Receiver NOISE BLANKER for pulse type noise that is not DSP based can usually be ON.
- All receiver DSP features such as NOISE REDUCTION, DUAL WATCH should be OFF.
- Receiver NOTCH FILTERING of any type must be OFF.
- Receiver SQUELCH should be wide open. However it can be adjusted so as to quiet channels where the user suffers from a high noise floor to the point where the channel with the lowest noise level is squelched except when the system is being used for ALE for linking prior to the use of this software.
- Receiver RF GAIN should be full open, especially for unattended use, however the user can adjust it to quiet the receiver if needed. The same is true of using an ATTENUATOR.

- Some receivers that have additional IF stage filter selections where filters are at a low IF frequency and a high IF frequency, if selectable for bypass, the higher frequency IF filter should be bypassed. If not selectable for bypass, use the widest filter in the highest IF, such as the FM or AM filter and a filter closest to 2.8Khz at the lowest IF stage.
- Where one button on the radio drops or adds a stage of amplification, be it called PREAMP or other, the user can make use of this amplification stage as long as it does not cause non-linear characteristics or introduces an undue amount of noise that reduces the SNR of the channel.

## **M110A DMT CONFIGURATION**

The following screen captures and descriptions break down the software main screen into three panels and detail how to configure the M110A DMT tool and its parameters for proper operation.

**NOTE:** Any item that is grayed out all the time, is currently under development and/or not approved for use and thus not available at this time.

### **COMMON PANEL**

This panel contains parameters that aside from mode selection and perhaps PSK carrier are basically one time set and forget unless a hardware change is required.



### **Buttons**

The “Modem” button provides access to the Sound Device Configuration dialog. When clicked the “PC Sound Device Setup” dialog will be displayed.

### **Mode**

The mode selections of ASYNC\_EOM, SYNC\_EOM and SYNC (no EOM) are provided via radio buttons where ASYNC\_EOM is the default mode when the program is run for the first time or when ever the DMT.DAT file is missing or corrupted as ASYNC\_EOM is the most commonly used of the 110A modes. The mode selection radio buttons will change to a list box to support additional mode selections, which directly relate to additional MIL-STD or STANAG waveforms and possibly Data Link Protocols (DLP) are planned for addition. When the DATA PORT is active, which uses RS-232 Asynchronous interfacing via either a hardware port or VCP port, compatible SYNC mode operation is provided where start and stop bits are not sent over the air, there is no need to make use of a Synchronous Serial Port adapter as is required with conventional SYNC mode operation via hardware modems.

The mode selections when the DATA PORT is active (not 0) can be overridden by “TRANS” when checked, which places the tool into SYNC with EOM compatible mode of operation setup for a binary data streams rather than just ASCII. In addition, “TRANS” also bypasses the internal message and data windows during DATA PORT operation, thus all use of the modem is with the external software application only. In that regard, a user of the DMT application that is not using “TRANS” is compatible with when SYNC\_EOM for all ASCII data that is sent, and can view any non-ASCII data if “HEX Eng. View” is checked. However if the sending station sends any content that contains a Null (0x00) character while in SYNC\_EOM the modem receiver will reset at that point. Additional information of the “DATA PORT” follows herein.

If all stations are not set to the same mode, depending on the combination of mode selections between stations and what is being sent only gibberish may be displayed on the receiving station

terminal. Also, depending on the combination of modes in use, the receiver may run on with characters if the receiver is set for EOM use and the sending station is not, in which case the manual Reset will need to be used. This will be especially true of hardware modem users and “DATA PORT in TRANS” as the ASYNC\_EOM and SYNC\_EOM are coded to expect the EOM and deal with it properly but are also coded in a manner that if there is no EOM sent, the modem receiver will still reset properly.

The 110A SYNC mode is used with or without an EOM more commonly than ASYNC is used without the EOM and thus only SYNC mode is provided for use with and without EOM on transmit. A station that is using a hardware modem will know their configuration, which they can not change on the fly as can be done with software modems, thus a hardware modem based user may request data sent for them in SYNC vs. ASYNC and either with or without EOM. Many hardware modems do not support disabling of the EOM, however when they do and if they are used for ARQ and other applications, the modem will likely be setup for SYNC with EOM disabled.

The SYNC without EOM mode is the most used 110A hardware modem configuration when modems are setup for ARQ, FTP, Digital Voice (usually Secure Voice) and with Crypto devices. At present this software only directly supports basic FEC operation of standard ASCII character range (decimal 20..128) messages via the SYNC mode with or without EOM via its internal terminal. However the DATA PORT can also be setup for ASCII or Binary data streams. A SYNC\_EOM compatible mode is used via the Data Port when “TRANS” is checked, regardless of the mode selection made elsewhere. TRANS supports the use of external software applications and binary data streams over the full binary byte range of 0x00 through 0xFF where such applications can send ASCII, Binary or a mix of ASCII/Binary data as achieved with hardware modems.

### **PSK Carrier**

The PSK carrier selections provided will follow the mode selected as additional modes selections become available. For the 110A ST waveform the 1800hz selection is the standard PSK carrier and the default for 110A in the software. The 1650hz and 1500hz selections have been provided for 110A as optional non-standard selections as they are supported in many hardware modems for 110A.

All stations must either make use of the same PSK carrier or offset tune their frequency based on the difference between the PSK carrier that they are using and what has been directed for use on the given frequency. If either is not the case, the PSK carriers will not be aligned and communications will not occur.

The use of 1500hz supports maximum use of narrow SSB filters that don't meet S4203, its use is recommended for use during broadcast message reception and all digital net operation ONLY and not in mix Voice/DATA nets.

The use of data rates higher than 600bps is dependent on the TX station IF passband filtering. If not to STANAG 4203 requirements, the higher data rates will be negatively impacted on TX and thus also on RX, especially when using the 1800hz PSK carrier. Using a 1500hz PSK carrier on TX places more of the 300-3300hz ST waveform into the passband of IF filters that don't meet S4203 which improves the use of higher data rates when the TX frequency is offset. Likewise, if stations are using 1500hz on RX their filter will pass more of the waveform for reception and decoding.

### **CAT Com Port**

To make use of CAT PTT a com port between 1..16 must be entered, where 0 is entered for no CAT PTT. In addition a baud rate must be selected that matches the radio baud rate if direct control of the radio is configured rather than Radio Emulation via MARS-ALE. The port parameters at this time are set to 8 data bits, no parity and 1 stop bit and no handshaking.

### **Radio PTT**

By default serial support is OFF, this is achieved by setting the com port to 0. This is the mode for those using Signalink and other devices. You can also set the com port to 0 temporarily to run another program which is faster than shutting down to start another program also alleviates the possibility of shutting down and starting the next program too fast and having com port issue.

By default DTR/RTS is not checked, as such if the serial port is other than 0, RTS will be active, if checked, DTR will be active.

By default, CAT PTT, when the serial port is active is ICOM by KNWD/ICOM not being checked, if checked the CAT PTT is basic Kenwood. The ICOM radio address is 00h which works with all ICOM Ham models and MARS-ALE Radio Emulation (REM), see Appendix C for details.

Should an external interface be in use that requires D.C. power from the RS-232 port, check "RS232 Power", whichever line, RTS or DTR is not being used for PTT will be used for DTR power.

### **Data Port**

**WARNING:** Never use the DMT tool without using an external terminal application if the DATA PORT is not set to 0, otherwise internal display of incoming data will display very slowly.

The "DATA PORT" selection is in support of 3rd party applications use of the software's data modem capability on RX and TX where the M110A-DMT software can be setup and minimized if desired and all communications can be handled by the external application.

Use of the Data Port follows the selection of ASYNC\_EOM, SYNC\_EOM and SYNC modes available for selection to process the data port data stream as ASCII range (decimal 20 through 128) characters. In addition, when "TRANS" is checked, the mode will be locked into a compatible SYNC with EOM mode coded to pass a binary data stream, be it Binary, mixed Binary/ASCII or just ASCII.

If "RoN" is checked in addition to "TRANS", then in addition to resetting on the EOM, the modem will reset on "Null", this provides for modem reset in case the EOM is corrupted, missed or not sent. However, RoN can not be used if any binary data is being sent that contains even one ASCII value 0x00 Null character or else the message will prematurely cause a modem reset.

A Data Port COM port between 1..16 must be entered to activate the data port, where 0 is entered for no data port support. When there is no active external terminal application, the Data Port must

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be set to 0. If you have an active data port and no external terminal program is running, the display of data within the M110A-DMT will be hampered until you start an external terminal application at which point you may or may not display all received data. If allowed to continue receiving with an active Data Port and no external terminal application the program will eventually appear to fresh.

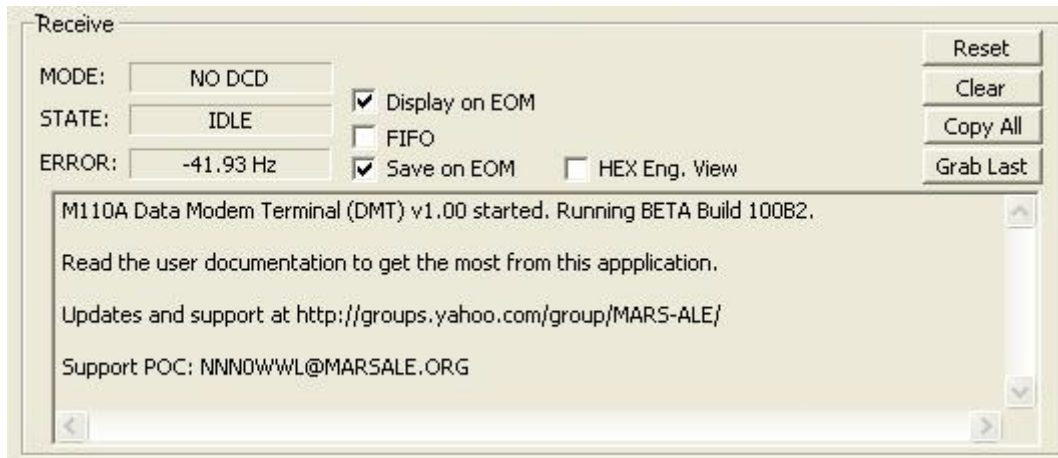
The Data Port parameters at this time are set to 8 data bits, no parity and 1 stop bit and CTS/RTS handshaking. Handshaking is supported via the VCP port, but not required of the 3<sup>rd</sup> party application. However it is recommended that all external applications be setup for RTS/CTS handshaking if supported for selection. The DTR line is asserted to indicate a high PTT state for any applications that require the monitoring of the PTT state.

The serial port being used for the Data Port is opened in binary mode with support through ASCII 0-255 and the modem itself will support passing binary range characters, however, at present the internal interface between the M110A-DMT only displays the normal printable ASCII character range. In that regard, a user of the DMT application that is not using "TRANS" is compatible with when SYNC\_EOM for all ASCII data that is sent, and can view any non-ASCII data if "HEX Eng. View" is checked. However if the sending station sends any content that contains a Null (0x00) character while in SYNC\_EOM the modem receiver will reset at that point.

The Data Port support is either accomplished by a paired (null modem) set of Virtual Serial Ports (VCP) where com0com or other 32 and 64 bit servers are recommend and where the 3rd party software is running on the same PC. It is recommended that overrun is selected as active with the VCP server. VCP servers tested that offer baud rate emulation do not seem to work, thus the selected baud rates between the DMT and external software do not seem to matter.

Alternately real serial ports can be used with a Null Modem via the same PC or separate PC's, the later being recommended when using real serial ports to minimize the effects of RS-232 interrupts on latency or when the M110A-DMT is running under another operating from the external terminal application. The selected port parameters much match between the DMT and external software and CTS/RTS handshaking is required.

When the data port is active, all data received is still displayed and treated the same within the M110A-DMT application to include RX logging except when "TRANS" is checked. All data sent and received via the data port passes through M110A-DMT application when the data port is active with no M110A-DTM parameters selected effecting the sending of data or the display of received data via the 3rd party application. The most recent message sent via the 3rd party application is retained by the M110A-DMT for later re-use by pressing the DataPort button, which pastes it into the M110A-DMT Outgoing Message window. 3rd party terminal applications may send unprintable characters (between ASCII decimal 0-31) any such characters when pasted for sending or when received will be automatically displayed as a box ( ) character within the M110A-DMT display.

**RECEIVE PANEL****Buttons**

“Reset” - Resets the Data Carrier Detect (DCD) state of the modem and STATE to IDLE which resets the Viterbi algorithm for decoding a bit stream and clears the ERROR field. The MODE will indicate “NO DCD” and the STATE will indicate “IDLE”. The modem receiver will return to hunting for a preamble stream to sync on. The use of this hard reset vs. a soft reset should be made when sync is lost during the Preamble or start of Data where nothing but gibberish is displayed, the displayed STATE will usually be “NO EOM” when this occurs. That way we are sure all aspects of the modem are 100% reset after the dramatic loss of sync.

“Clear” - Clears the Incoming Message Window and resets the scroll bars if they were automatically enabled on a received message due to size.

“Copy All” - A single click copy of the entire contents of the Incoming Message Window to the Windows Clipboard for pasting its contents into another application. This saves the right click menu selections or manual high lighting steps to copy from the message window.

“Grab Last” - Grabs the last message received and clears and replaces the outgoing message window contents. The grabbed message can then be edited for relay or highlighted for sending fills using "Send HL" as needed as rapidly as possible over other copying and pasting methods. Should there be no message to grab a message box is displayed to the user. The last received message is always available until another message is received no matter how many times "Grab Last" is used.

**Status**

“MODE” - Displays the Data Rate and Interleaver settings of the incoming data while data is being received and NO DCD when incoming data stops.

“STATE” - This is a work in progress, it displays the various states while awaiting detection of preamble sync, decoding data probes and payload data and completely decoding the payload through to the completion of the End Of Message (EOM) indicator. The current STATES that are displayed and there meaning are:



- **IDLE:** The modem receiver is hunting for a legitimate signal and PREAMBLE data to decode.
- **DATA:** The PREAMBLE has been found and is decoding as seen by the ERROR window values still changing, when frequency error values lock, the KNOWN/PROBE data and PAYLOAD data (consisting of the message) decoding starts taking place and the message (payload data if any) is then available.
- **NO EOM:** The message has been decoded through the EOM. This is displayed so fast and immediately switches back to IDLE that it will often not be caught by the naked eye. However, if it remains displayed then DCD has likely been lost and gibberish displayed to the Incoming Message Window if Display on EOM is not checked and the RESET button should be pressed for a hard reset to place the modem back into a known IDLE state.

“ERROR” - The Doppler Spread or Frequency Error has a display range of -999.99 to 999.99hz where the modem must properly decode an incoming signal that is off frequency by as much as +/- 75hz (including transmitter/receiver frequency variations and doppler spread). The modems RF frequency error correction in this software exceeds that requirement of the standard. The sample clock error does not enter into this error correction. The frequency error display during the hunting for a signal will periodically jump about with significant numbers, however when a legitimate signal is detected the display will be constantly changing and displaying lower and lower values as the preamble is being decoded and frequency error correction is being applied until the completion of that task and then it will lock onto displaying the last frequency error value until DCD is reset.

**NOTE:** There no need to touch up the RX frequency to deal with the error unless the error exceeds 75hz and DCD is being lost and then is must be done between rather than during transmissions.

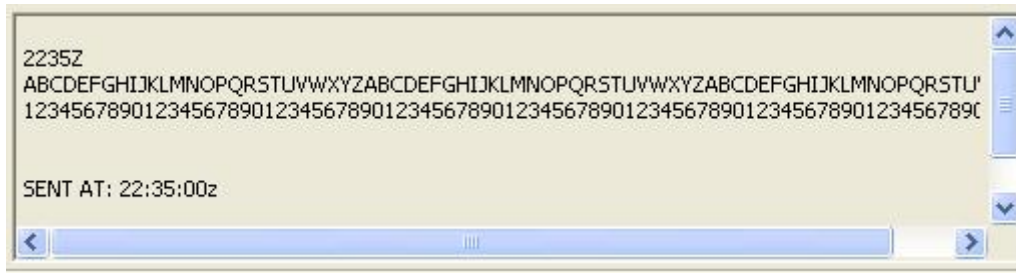
**NOTE:** Any tuning of frequency must be done prior to 110A transmissions. Do not try to tune in 110A as you would RTTY and other modes. You must be setup on the correct frequency at 110A transmit start. When a station is transmitting 110A and you touch up your RX frequency during the message being sent you will cause the software to work harder and you will likely loose DCD.

### **Incoming Message Window**

Any received message is displayed in the upper window of the Receive panel in the “Incoming Message Window”. The selection of “Display on EOM” determines if received messages are displayed continuously or one at a time and automatically cleared on the next message EOM received.

The RX message buffer has been set to the maximum of 2GB of system and virtual memory. This may present a problem for some Vista users if the buffer is allowed to get to full before hitting "Clear" based on some of the things I have read on CEdit controls and Vista. Should the message buffer ever hit the limit a message box will display an error message.

The font in use is the Microsoft Shell Dialog System Font which provides a fairly proportional display of incoming message text beyond 69 characters are displayed regardless of normal 96dpi size or greater being used.



### **Check Boxes**

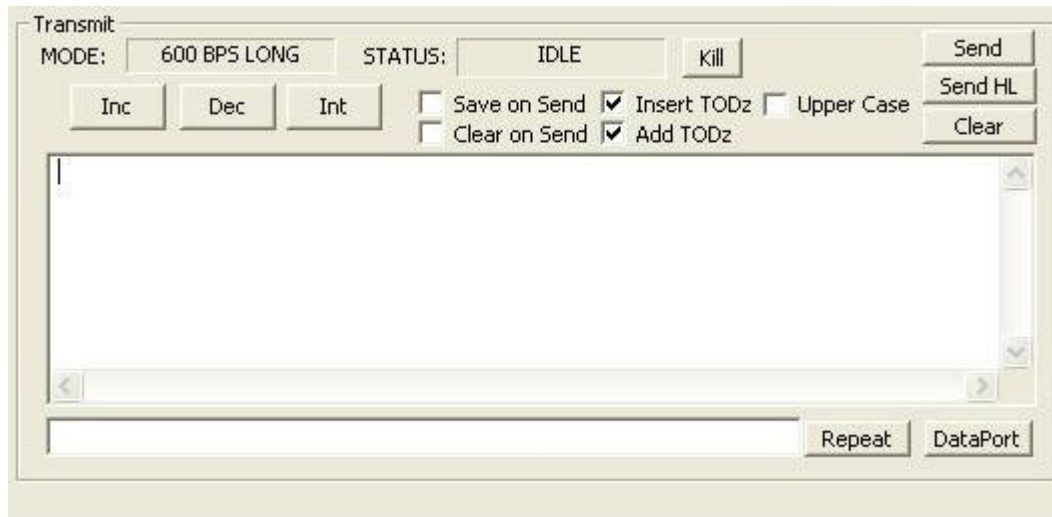
“Display on EOM” - When "Display on EOM" is checked, only the last message received that will be displayed, and done so all at one time on receipt of the End of Message (EOM) indicator. When "Display on EOM" is not checked, each character is displayed as received and scrolls smoothly line by line during reception of the entire message. Even though the scroll bars are currently active during this process, they really can not and should not be used until the entire message has been displayed.

“FIFO” - When "Display on EOM" is not checked and "FIFO" is checked, the latest message received shall be inserted at the top of the messages displayed buffer and done so character by character during decoding. Other, when “FIFO” is not checked, each message will be appended to the message buffer.

“Save on EOM” - When "Save on EOM" is checked, regardless of whether "Display on EOM" is checked or not, all messages received shall be saved to the “./dalog/” sub directory using a file naming convention of mmddyyyy.da in an ASCII text file that can only be read from start to end when M110A is not running. The file contents can be read at any time as the file is opened, written to and then closed. However if the file is opened for reading during incoming data, the file must be closed and reopened to see all the data written to file since it was opened for reading. See Appendix A for more details on the .da file.

“HEX Eng. View” – When “HEX Eng. View” is checked all data displayed in the Message Window and logged will be the HEX byte value for each data byte received. This feature for the user is of the most interest when monitoring binary data streams in SYNC or SYNC\_EOM mode, which can not be displayed as pure ASCII characters correctly or for third party developers. A good reference for the full ASCII character set is: <http://www.ascii-code.com/>

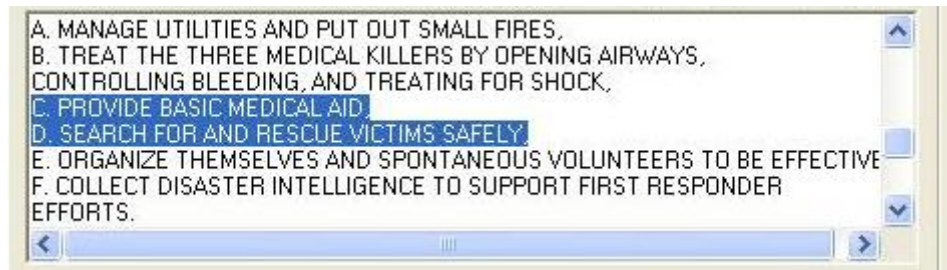
## **TRANSMIT PANEL**



### **Buttons**

“Send” - Sends the message located in the “Outgoing Message Window” if any message is present.

“Send HL” - Supports resending any part of a multi-line message in the Outgoing Message Window where for whatever reason, it is desired to only send part of the message. The user must highlight that portion of the message of interest as if to copy, once high lighted, clicking on "Send HL" instead of "Send" will transmit the highlighted part of the message as seen below.



“Clear” - Clears the Outgoing Message Window and resets the scroll bars.

“Repeat” - When clicked the last keyboard message sent during the current session will be pasted back to the keyboard message window for editing or immediate resending.

“DataPort” - Grabs the last message sent using by the 3 party software via the Data Port and paste is into the Outgoing Message window for editing and resending.

“Inc” - This button increments the data rate.

“Dec” - This button decrements the data rate.

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“Int’ - This button toggles the Interleaver between SHORT or LONG.

**NOTE:** The LONG interleave setting is more resistant to channel noise bursts and fades.

**NOTE:** There is no need for stations to be set to the same Data Rate or Interleave for RX as the 110A preamble tells all receiving terminals what mode is being sent for automatic configuration. Each side in two-way exchanges can make use of different data rate and interleave setting with no coordination.

### **Status**

“MODE” – This status window displays the current selected transmit mode is displayed. It follows the mode selected in the Common panel. For 110A the default TX mode when the program is run for the first time or when the DMT.DAT file is missing or corrupted is 600BPS LONG. The range of 110A TX data rates are: 75bps (compatible with S4415 Robust), 150bps, 300bps, 600bps, 1200bps, 2400bps coded for FEC and SHORT or LONG interleaver and 4800bps uncoded only.

“STATUS” – This status window displays the current transmit state, at present for 110A the states displayed are IDLE or TRANSMIT and BUFFERING. When the Data Port is active and receiving a message to transmit BUFFERING with flashing arrows “>” will be displayed as data is being buffered to send.

### **Outgoing Message Window**

A message is pasted or typed into the bottom “Outgoing Message Window” and is then sent by clicking on the "Send" button. The main outgoing message buffer size is limited to 32k (32,378) at this time. If a message is entered that exceeds the limit an error message will be displayed.

If a message is entered that is longer than will display in the window at one time, vertical scroll bars will automatically turn on for scrolling and will turn off when the "Clear" button is used. There is no LINE WRAP taking place, thus long lines will not wrap on the RX end.

When pasting messages be sure that the content has no hidden characters, its best to only copy and paste from an ASCII editor such as Notepad. Thus anything being copied from another source should first be pasted into Notepad and saved and then reopened to strip any non-ASCII content.

### **Check Boxes**

“Save on Send” - When "Save on Send" is checked, all messages sent shall be saved to the “./owlog/” sub directory in a mmddyyyy.ow ASCII text file that can only be read from start to end when M110A is not running, else only up the last time the program was run for the current Zulu Date shall be displayed. See Appendix B for more details on the .ow file.

“Clear on Send” - When "Clear on Send" is checked the outgoing message window will be cleared after the message has been sent.

“Insert TODz” - Per ACP-126 requirement "302. TIME OF TRANSMISSION INDICATOR - All transmissions are to include a time of transmission indicator. This is expressed as a time group in ZULU time and is to be the time the transmission commenced.”

Example A: Call  
(5 SPACES) (2CR) (LF)  
1234Z (2CR) (LF)"

Thus there is now a selection check box of "Insert TODz" which will upon sending the message insert the TOD in Zulu time. This selection and "Add TODz" can both be active at the same time and work with all available means of sending a message.

“Add TODz” - When "Add TOD" is checked, the Zulu time shall be appended to the out going message sent.

“Upper Case” - When checked any text typed or pasted, either manually or when using the “Grab Last” feature will be converted to upper case. Any existing text prior for all but “Grab Last” will remain in lower case.

### **Keyboarding Message Window**

For Peer-to-Peer rapid keyboarding vs. formal message traffic, this window supports the sending of up to a 1024 (1Kb) character message on the ENTER key stroke. At 600bps or greater it makes for fast turn around keyboarding.

Paste is also supported, however whatever is pasted is limited to 1024 characters and will terminate on any embedded carriage returns or line feeds. All entered or pasted text is always converted to Upper Case.

There is no LINE WRAP taking place, thus long lines will not wrap on the RX end. When the 1Kb limit has been reached with an entered message prior to sending, a message box warning will display, at which point hitting enter will allow sending what has been entered, or the message can be trimmed down to convey all of the information with edited wording.

"Save on Send" and "Add and Insert TOD" features do not apply to use of this message window. A carriage return and line feed is inserted and appended to what is sent.

The "Clear on Send" is not supported as the keyboard message window automatically clears on sending the message when Enter is pressed.

The “Repeat” button will refill the window with the last sent message for reediting or resending immediately.

## **APPENDIX A**

### **Data Rate and Interleaver Selection**

S4415 which only supports one data rate makes life simpler for the user and provides very robust operation under all channel conditions, but it is slow and thus not the best choice if the channel supports faster data rates. S4285 and S4529 which are not autobaud compliant require that both the TX and RX stations to be configured to the same data rate and interleaver settings in advance and thus do not support Adaptive ARQ not being autobaud. However due to their near constant re-sync known data transmitted during the payload, it makes them the better suited modes for FEC Broadcast applications.

However for FEC only use, the user must select the data rate and interleaver for FEC operation based on channel conditions manually. The use of serial tone modem waveforms in FEC modes by stations in attended operation, unlike in unattended guard channel operations permits a voice exchange to determine receive conditions on both ends, this holds true for Regional Broadcast use as well, as the NCS or directed sending station can poll the net for signal report. In two-way use of 110A serial tone FEC modes both stations can send at the same or different data rate and Interleaver settings, where it is best to just make use of the LONG interleaver setting to deal with any channel issues and minimum performance characteristics of the waveform data rates. For guard channel operations the broadcast station can only take into account TOD propagation characteristics for the wavelength being used, seasonal effects and minimum performance characteristics of the waveform data rates, thus 75-300bps should be used for CONUS wide or OCONUS broadcasts and 75-600bps for regional broadcasts.

The use of serial tone modem waveforms in FEC modes by stations in attended operation, unlike in unattended guard channel operations permits a voice exchange to determine receive conditions on both ends, this holds true for Regional Broadcast use as well, as the NCS or directed sending station can poll the net for signal report. In two-way use of 110A serial tone FEC modes both stations can send at the same or different data rate and Interleaver settings, where it is best to just make use of the LONG interleave setting to deal with any channel issues and minimum performance characteristics of the waveform data rates. For guard channel operations the broadcast station can only take into account TOD propagation characteristics for the wavelength being used, seasonal effects and minimum performance characteristics of the waveform data rates, thus 75-300bps should be used for CONUS wide or OCONUS broadcasts and 75-600bps for regional broadcasts.

Most MARS-to-MARS peer-to-peer and regional broadcast communications takes place within 2-12Mhz NVIS where the 3-7Mhz range sees the most use and where the 3 and 4Mhz range sees the bulk of the use and which has the highest noise levels and fading conditions next to 2Mhz. As such the recommended Interleaver setting is always LONG. Data rates beyond 600bps will not yield reliable good results even if one has an S4203 compliant radio system and hardware modem unless very good to excellent channel conditions exist, which can be determined if two-way contact with the audience stations is part of the scenario.

Below is the section from both MIL-STD-188-110B pertaining to 110A standards "Minimum Performance Requirements" taking into account the use of an S4203 compliant HF radio. Then further below is information regarding the calibration of S-meters which combined should give

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all users of 110A ASYNC an idea of how to best select the data rate and in consideration of prevailing and possibly changing channel conditions.

MIL-STD-188-110B, 5.3.2.5 Performance requirements.

The measured performance of the serial (single-tone) mode, using fixed-frequency operation and employing the maximum interleaving period, shall be equal to or better than the coded BER performance in table XX. Performance verification shall be tested using a baseband HF simulator patterned after the Watterson Model in accordance with International Telecommunications Union (ITU) Recommendation ITU-R F.520-2. The modeled multipath spread values and fading (two sigma) bandwidth (BW) values in table XX shall consist of two independent but equal average power Rayleigh paths. For frequency-hopping operation, an additional 2 dB in signal-to-noise ratio (SNR) shall be allowed.

**TABLE XX. Serial (single-tone) mode minimum performance.**

User bit rate	Channel Paths	Multipath (ms)	Fading (Note 1) BW (Hz)	SNR (Note 2) (dB)	Coded BER
4800	1 Fixed	-	-	17	1.0 E-3
4800	2 Fading	2	0.5	27	1.0 E-3
2400	1 Fixed	-	-	10	1.0 E-5
2400	2 Fading	2	1	18	1.0 E-5
2400	2 Fading	2	5	30	1.0 E-3
2400	2 Fading	5	1	30	1.0 E-5
1200	2 Fading	2	1	11	1.0 E-5
600	2 Fading	2	1	7	1.0 E-5
300	2 Fading	5	5	7	1.0 E-5
150	2 Fading	5	5	5	1.0 E-5
75	2 Fading	5	5	2	1.0 E-5

**NOTES:**

1. Per ITU-R F520-2.

2. Both signal and noise powers are measured in a 3-kHz bandwidth.

The signal to noise ratio (SNR) is defined as the ratio between the signal and noise levels, and is usually expressed in decibels (dB). 0 dB means the ratio is 1, the signal and noise power levels are the same. a 10 dB SNR means the signal power is 10 times the noise power, 20 dB means the signal is 100 times (it is a log based scale). These are for power values, for voltage ratios the SNR is twice the power value. A SNR of 0 dB would just be barely detectable, in practice you need a few dBs for even a weak signal, and a SNR of 30 or 40 dB is considered an excellent quality signal.

For a correlation to MIL-STD-188-110B 5.3.2.5 Performance requirements. TABLE XX. In terms of S meter reading only, the International Amateur Radio Union (IARU) Region 1 agreed on a technical recommendation for S Meter calibration for HF and VHF/UHF transceivers in 1981. IARU Region 1 Technical Recommendation R.1 defines S9 for the HF bands to be a receiver input power of -73 dBm. This is a level of 50 microvolts at the receiver's antenna input assuming the input impedance of the receiver is 50 ohms.

<b>S-reading</b>	<b>HF</b>		<b>Signal Generator emf</b>
	<b>μV (50Ω)</b>	<b>dBm</b>	<b>dB above 1μV</b>
<b>S9+10dB</b>	<b>160.0</b>	<b>-63</b>	<b>44</b>
<b>S9</b>	<b>50.2</b>	<b>-73</b>	<b>34</b>
<b>S8</b>	<b>25.1</b>	<b>-79</b>	<b>28</b>
<b>S7</b>	<b>12.6</b>	<b>-85</b>	<b>22</b>
<b>S6</b>	<b>6.3</b>	<b>-91</b>	<b>16</b>
<b>S5</b>	<b>3.2</b>	<b>-97</b>	<b>10</b>
<b>S4</b>	<b>1.6</b>	<b>-103</b>	<b>4</b>
<b>S3</b>	<b>0.8</b>	<b>-109</b>	<b>-2</b>
<b>S2</b>	<b>0.4</b>	<b>-115</b>	<b>-8</b>
<b>S1</b>	<b>0.2</b>	<b>-121</b>	<b>-14</b>



## **APPENDIX B**

### **Received Data Logging**

For “Save on EOM” the file naming convention is MMDDYYYY.da for example:

06082012.da

When "Save on EOM" is checked, all messages that come in with a legitimate EOM at the end will be saved to the current log file.

A log file is created or if existing for the current date opened when the program has received a message and the file is closed immediately after the data is saved.

The same log file will be used until the program allowed to run beyond the Zulu date, then a new log file is will be created at the next message received when logging is enable.

Each time the log file is opened during its life, the following statement is written to it:

FILE OPENED xxx-xx-xxxx xx:xx Greenwich Mean Time

For example, today’s files first entry is:

FILE OPENED Jun-08-2012 02:12 Greenwich Mean Time

Then later when the next message was received the entry is:

FILE OPENED Jun-08-2012 02:12 Greenwich Mean Time

For each “FILE OPENED” entry there should always be as corresponding “FILE CLOSED” entry, which should always be the last entry to a log file, such as:

FILE CLOSED Jun-08-2012 02:12 Greenwich Mean Time

In between “FILE OPENED” and “FILE CLOSED” entries with the same date/time stamps will be the message received, as the file is opened and written to and then closed, the date/time for these two events will most likely always be the same except for long messages, for example here is a complete example of a test message being received:

=====  
FILE OPENED Jun-08-2012 02:12 Greenwich Mean Time

=====

Received using Mode: SYNC\_EOM  
Received at Data Rate: 600 BPS LONG

**MS110A DMT v1.00**  
**DRAFT 09-10-12**

Received at Freq. Error: 0.04 Hz

\*\*\*Received Message Starts Here:

0202Z

THE QUICK BROWN FOX JUMPED OVER THE LAZY DOG'S BACK 1234567890 TIMES.  
RYRYRYRYRYRYRYRYRYRYRYRYRYRYRY

NNNN

SENT AT: 02:02:00z

\*\*\*Received Message Ends Here:

=====

FILE CLOSED Jun-08-2012 02:12 Greenwich Mean Time

=====

If the End of Message (EOM) indicator is displayed in the log as the three characters “iZ>” prior to the line “\*\*\*Received Message Ends Here:” the user is using an old Alpha build or Beta #1 and needs to upgrade.

In testing during numerous HF Channel Simulator sessions and real world testing to date, to include when stations were setup with the incorrect mode, such as ASYNC and SYNC, if the incoming signal is recognized as being a legitimate Preamble, when data starts, even if we immediately lose sync (NO DCD) and even if NO EOM is printed to the STATE display and the three EOM characters are not displayed, the log file has always properly closed for example here is a case where the sending station was using SYNC which resulted in NO DCD and NO EOM:

**MS110A DMT v1.00**  
**DRAFT 09-10-12**

=====

FILE OPENED Jun-08-2012 04:32 Greenwich Mean Time

=====

Received using Mode: SYNC\_EOM  
Received at Data Rate: 600 BPS LONG  
Received at Freq. Error: -0.06 Hz

\*\*\*Received Message Starts Here:

This is a test.

NNNN

\*\*\*Received Message Ends Here:

=====

FILE CLOSED Jun-08-2012 04:32 Greenwich Mean Time

=====

If the sending station has either or both "Insert TOD" or "Add TOD" enabled using M110A-DMT then each message received will have the Zulu TOD that the sending station sent the message inserted and or appended to the body of the message. The difference in time between the Zulu TOD in the body of the message if present and the FILE CLOSED Zulu TOD, assuming both stations have accurate TOD is a relative indicator of the duration of the message transmission time, but is not exact as time is only recorded down to minutes and not seconds.

As the log file is an ASCII file rather than a binary file, you can open them to access the information using Notepad or just about anything. Having the file open when the software writes to it does not cause any issues under MS Windows, running the software under a Windows emulation such as Wine may be another matter.

## **APPENDIX C**

### **Sent Data Logging**

When “Save on Send” is checked, each message sent using the “Send” and “Send HL” buttons the message text sent is saved to the current log file.

The log files are saved to the “..\owlog\” log directory using the file naming convention of MMDDYYYY.ow for example:

06082012.ow

The log will contain two types of sent messages, those sent normally and those sent as fill using the Send Highlighted feature, which will be labeled as “\*\*\*Sent Highlighted Message Starts Here:”, followed by the message sent and then “\*\*\*Sent Highlighted Message Ends Here:”

Here are some examples during testing.

=====

FILE OPENED Jun-07-2012 22:28 Greenwich Mean Time

Transmitted using ASYNC\_EOM

Transmitted using: 600 BPS LONG

\*\*\*Sent Message Starts Here:

2228Z

This is a test message

BT

NNNN

SENT AT: 22:28:00z

\*\*\*Sent Message Ends Here:

=====

FILE OPENED Jun-07-2012 04:44 Greenwich Mean Time

Transmitted using ASYNC\_EOM

Transmitted using: 4800 BPS UNCODED

\*\*\*Sent Highlighted Message Starts Here:

THE COMMUNITY EMERGENCY RESPONSE TEAM (CERT) PROGRAM IS AN  
ALL-RISK, ALL-HAZARD TRAINING. THIS VALUABLE COURSE IS DESIGNED  
TO HELP YOU PROTECT YOURSELF, YOUR FAMILY, YOUR NEIGHBORS AND  
YOUR NEIGHBORHOOD IN AN EMERGENCY SITUATION.

SENT AT: 04:44:00z

\*\*\*Sent Highlighted Message Ends Here:

## **APPENDIX D**

### **MARS-ALE Radio Emulation**

The use of the M110ATEST with MARS-ALE Radio Emulation (REM) provides CAT PTT access for any make/model radio supported by MARS-ALE that provides for CAT PTT (as detailed in MARS-ALE RHUG Appendix A) the steps for using MARS-ALE REM are:

1. Install MARS-ALE, the LICENSE.DAT file and setup a scan group of channels from either an edited .QRG file or manually, at least one channel setup for TX/RX is required for PTT to work. (NOTE: For radios like IC-7200 and TS-590 if their digital ports are being used, the mode must be USB-D instead of USB, see MARS-ALE RHUG Appendix A. for details).
2. Install a VCP server to setup a pair of paired VCP ports, e.g. COM12 and COM13. Such as Com0Com or the N8VM VCOM driver which is documented in RHUG Appendix C. You will need to use a VCP server that works with your Windows OS, the VCOM one seems to have issues with newer versions of MS-Windows I am told whereas Com0Com does not.
3. Configure MARS-ALE for either M110ATEST use after and ALE link where ALE is not used at all, per MARS-ALE RHUG Appendix C.

## APPENDIX E

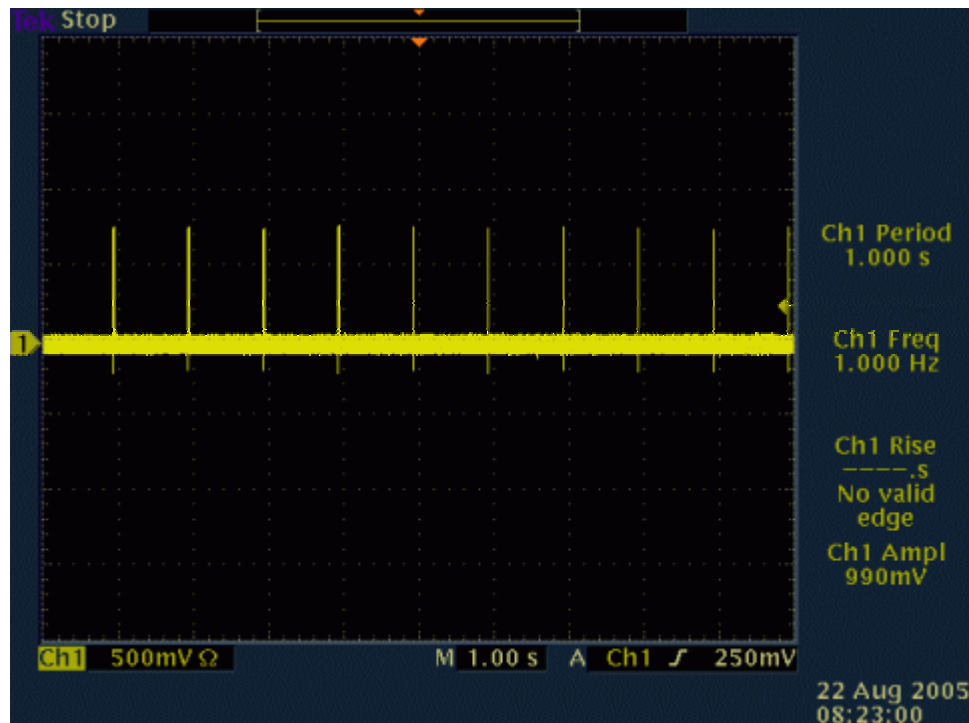
## Experiments with a PC sound card

The following information is sourced to LeapSecond.com and when last accessed for reference is found at <http://www.leapsecond.com/pages/sound-1pps/>

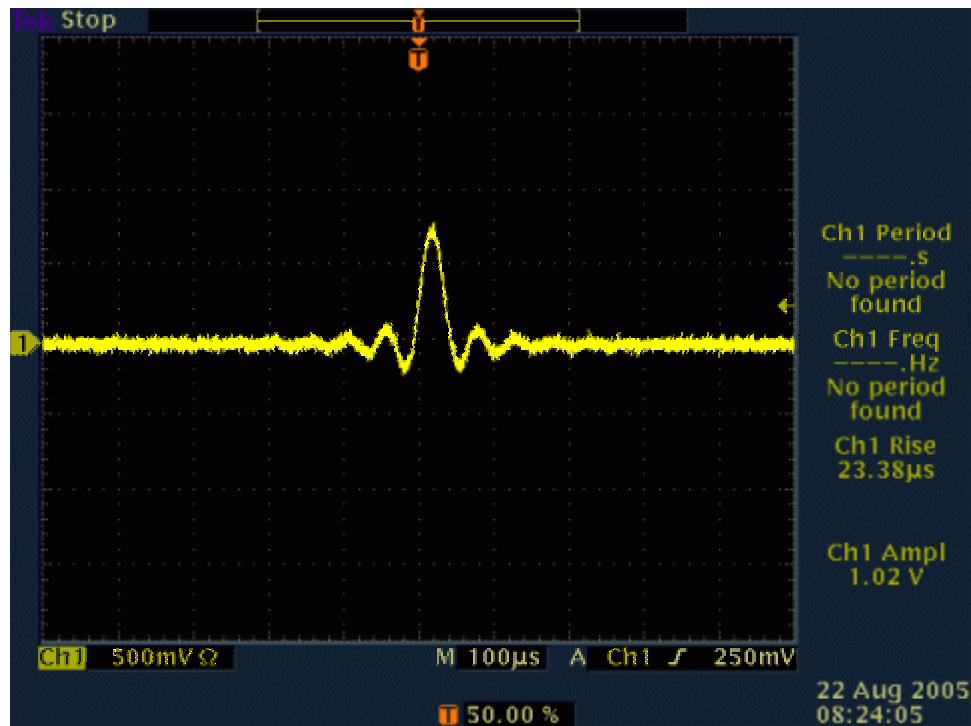
A PC contains many timing sources. The obvious one is time-of-day, which is implemented offline by a lithium battery powered 32 kHz CMOS clock chip and online by the continuous counting of crystal controlled periodic programmed interrupts. Neither is known for its great accuracy. Other timing sources include high-resolution timers, CPU clock cycle timers, serial port baud rate generators, video frame generation, and sound card time bases.

So how accurate is a typical PC sound card? How stable is the output? How would one measure this?

It would be easy to have a sound card generate, say, a 1 kHz tone and then measure the accuracy of that signal. Instead I wrote a simple tool that generates 1 PPS with a sound card. With a typical CD-audio sampling rate of 44.100 kHz (22.675  $\mu$ s per sample), the tool creates a repetitive waveform consisting of 44099 zero voltage samples and 1 non-zero voltage sample. The result is an audio output that is all quiet except for one sample each second. On an oscilloscope the output is a series of familiar-looking 1 PPS pulses below:



A closer look at one pulse, below, shows the rise-time is on the order of the width of one sample, around 20 microseconds.



The output impedance of a PC sound card headphone output is quite low so either 1M or 50R input termination can be used. For a TI counter use a 250 mV DC trigger level at 50R.

Measuring frequency accuracy and stability is then done as if this were any other 1 PPS timing source, such as an XO, TCXO, OCXO, WWVB, Loran-C, GPS, Rubidium, or Cesium clock frequency source.

Before we start let me guess the accuracy of a PC sound card is better than a PC time-of-day clock but not as good as any laboratory frequency source. Let me also guess, since it is quartz-based, that some frequency drift will be visible.

## RESULTS

Using a HP 53132A TI counter and a convenient GPS 1 PPS reference I recorded 1 PPS phase measurements for about 8 hours over night.

The data contained some jitter and phase jumps. This is probably due to my running it on my main PC rather than a quiet one. These have been removed in the plots below.

View of raw data from the TI counter:

```
C:\tvb\1hz>head -11 < log21345.txt
0.1389973689 s
0.1389532386 s
0.1389088184 s
0.1388645514 s
0.1388206957 s
0.1387757163 s
0.1387321822 s
```



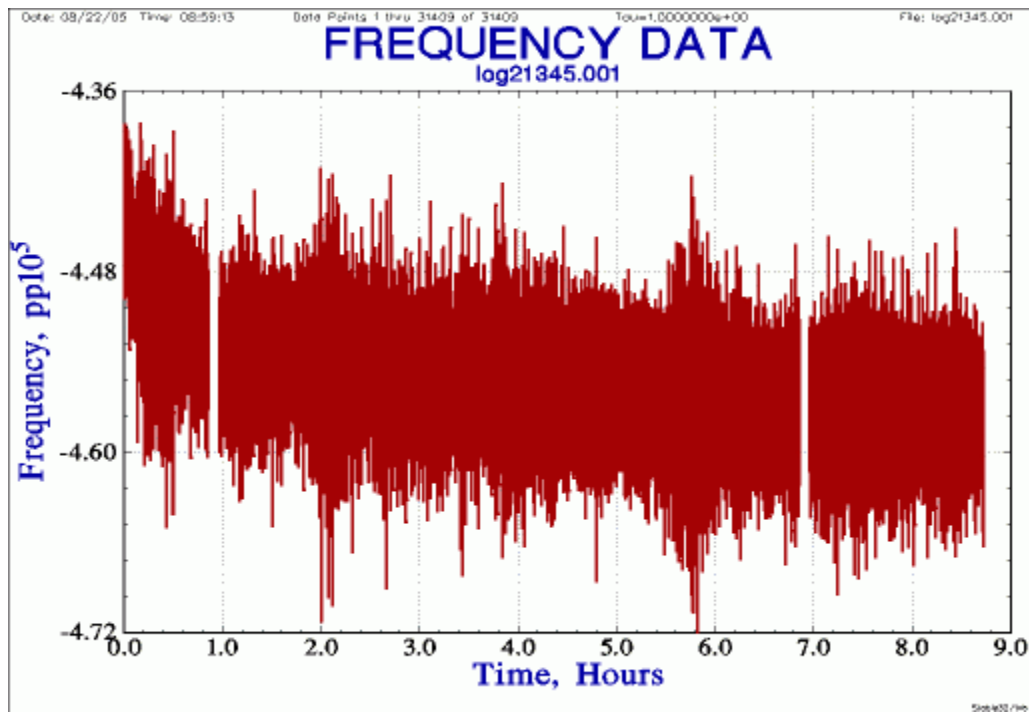
```
0.1386875740 s
0.1386436972 s
0.1385992097 s
0.1385551825 s
```

In just a few samples we can already see there is frequency error. If the sound card were perfectly accurate we would expect the same TI reading (phase error) between the reference 1 PPS and the sound card 1 PPS. Instead we can see that every second the sound card gets behind in the 5th decimal place. In fact with just 11 seconds of data we see the phase error is  $(0.1389973689 - 0.1385551825) = 0.0004421864$  seconds, which is 442 microseconds over 10 seconds, or 44 ppm.

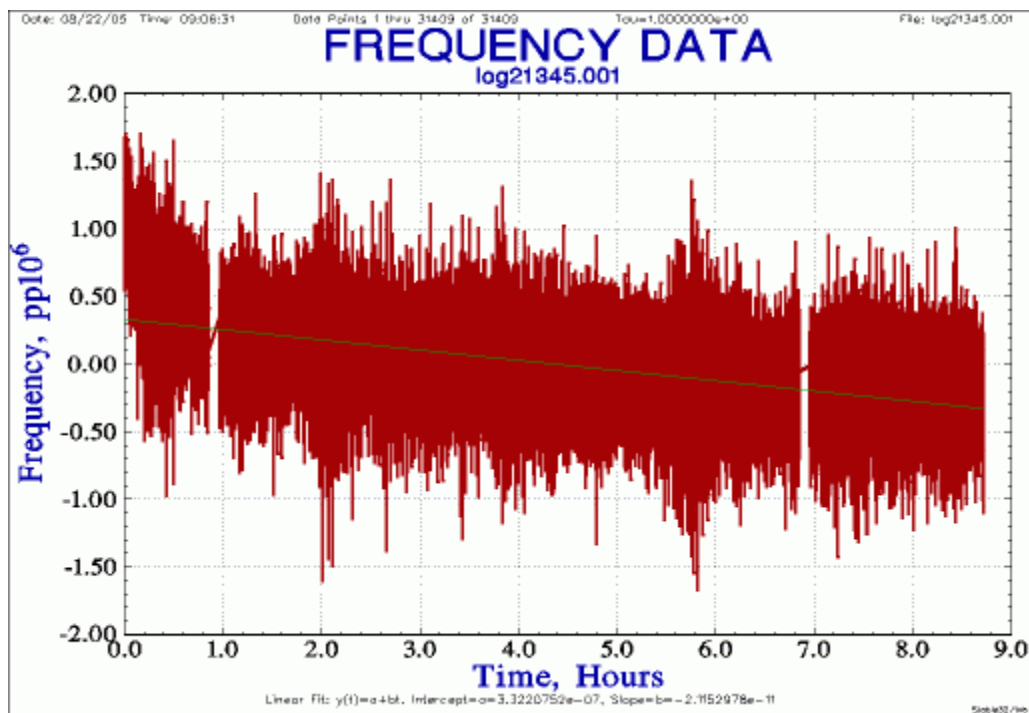
Below is a raw phase plot using Stable32. The main things to note is the gradual slope downward, the typical phase plot wrapping at one second, and a few glitches.



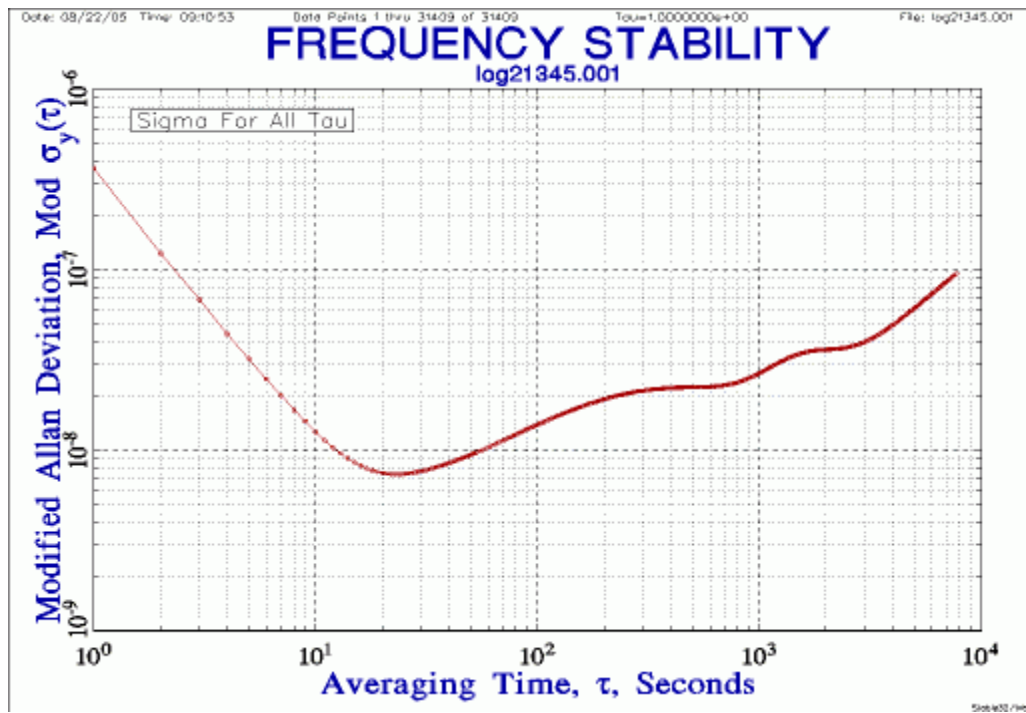
Cleaning up the data, below is a frequency plot. Note the frequency error is around  $-4.4\text{e-}5$ , which is 44 ppm (agreeing with the quick calculation from the raw phase data). Note also some irregular frequency noise, as well as a slight trend downward in frequency error.



Normalizing the above gives the plot below. The peak-to-peak frequency error is on the order of  $\pm 1$  ppm and the downward slope is  $-2.11\text{e-}11/\text{s}$  ( $\times 86400 = 1.82\text{e-}6/\text{day}$ ).



Finally, for this 8-hour run, below is the Allan deviation of the sound card:



## Conclusion

- The sound card on my desk PC is accurate to 44 ppm.
- It is stable to a few ppm over hours.
- And has a frequency drift rate of 1.8 ppm per day.

## **APPENDIX E**

### **SOFTWARE PORTABILITY**

This application at present is being developed as a Microsoft Foundation Class (MFC) C++ baseline built using the MS Visual Studio 2008 Professional compiler to build a 32 bit .EXE with WINVER targeted at the MS Windows Vista OS. The application runs on and has been tested on the following Windows OS versions common to MARS members use:

Windows 2000 Professional SP4  
Windows XP Home SP3  
Windows XP Pro SP3  
Windows XP Pro 64 bit SP2  
Windows Vista 32 bit SP2  
Windows Vista 64 bit SP2  
Windows 7 32 bit SP1  
Windows 7 64 bit SP1

It is planned to migrate the existing MFC baseline to the MS Visual Studio 2010 Professional compiler.

In the future MFC 64 bit .EXE build of this application may also be made.

At present the 32 bit MFC .EXE of this application has successfully been used under both Linux versions using Wine and Mac OS X Lion using both Wine and Wineskin.

For Linux, using Wine, the incoming data logging seems to require that the programs executable (.EXE) file be located in the dalog sub directory, which leaves out the use of outgoing logging, as that logging will not work if the .EXE is separated from the log location. As we progress an effort will be made to improve the operation under Wine.

For Mac OS X the "Wineskin" program at <http://wineskin.doh123.com/> is required. Create a Wrapper (name it M110A) that will run the application.

At some point the baseline will migrate away from MFC to using something like WxWidgets for GUI and PortAudio for the PCM audio and perhaps other libraries for Serial I/O etc. and become a cross platform portable baseline that will provide for native executables (.EXE) on other operating systems. Currently both Linux and Mac OS X support are planned.

All of the current development is done on Windows XP Home SP3 and testing is performed on it and Windows 2000 SP4 and Windows XP Professional SP3 and Windows 7 Premium 64 bit. We shall add Ubuntu 10.04 LTS 64 bit to the mix on the Win 7 PC for testing under Wine. Also desired is a MAC OS X Lion and Wineskin system added to the mix when possible.

## **APPENDIX F**

### **STANAG 4203**

The title of STANAG 4203 is “Technical standards for single channel HF radio equipment”.

The technical details can be read in STANAG 4203. In summary t=The standard specifies properties of the radio which receives the audio signal (3 kHz bandwidth) from the modem and modulates it onto a carrier. It is located at the physical layer of the OSI model, but is *not* a modem standard, in contrast to the other standards which are located at layer one of the HF House. It is a tri-service standard.

Because a range of new modem standards were developed in the 1990s, it became necessary to produce a new version of STANAG 4203 which was better adapted to the new modem standards. Edition 3 was issued for ratification in 2002 and has now been ratified. Without Edition 3 of this standard the new waveforms will not perform optimally. Most major HF suppliers who have implemented the new waveforms and who also manufacture a radio product have implemented STANAG 4203 Edition 3.

Technical parameters specified in STANAG 4203 Edition 3 are:

- Radio frequency range
- Tuning
- Frequency accuracy and stability
- Frequency response and group delay time over a 3 kHz frequency band
- Phase noise
- Linearity
- Random emissions outside the frequency band and intermodulation products
- Operation mode (simplex/half-duplex)
- Switching time between transmitting and receiving
- Gain control
- Modulation of carrier wave

The last item specifies how different types of signals shall be modulated onto a carrier wave, for example that SSBSC (Single Side Band Suppressed Carrier) should be used for analogue and digital speech, teleprinting (RATT) and data transmission.

The standard also describes parameters for Independent Side Band (ISB) 6 kHz operation. Two other standards which place stricter requirements than STANAG 4203 on the radio equipment are STANAG 4444 and STANAG 5511 (Tactical data Exchange – Link 11/Link 11B (Edition 4)).