

SDR-1000

OPERATING MANUAL

Version 1.6.2

SDR-1000 SOFTWARE DEFINED RADIO

Operating Manual

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Table of Contents

| | |
|---|----------|
| PREFACE | VIII |
| ACKNOWLEDGEMENTS..... | X |
| 1 GENERAL DESCRIPTION | 1 |
| 2 SPECIFICATIONS..... | 2 |
| 3 HARDWARE CONFIGURATION | 3 |
| STANDARD CONFIGURATION..... | 3 |
| High Performance Preamplifier..... | 3 |
| Fifth Order Low Pass Filters | 3 |
| Enhanced 1W PEP Driver | 4 |
| 28MHz Transverter IF Takeoff..... | 4 |
| Experimental Impulse Generator..... | 4 |
| Enclosure..... | 4 |
| UPGRADES..... | 5 |
| 2 Meter Transverter | 5 |
| 100W Power Amplifier..... | 5 |
| <i>Specifications</i> | 5 |
| Automatic Antenna Tuning Unit..... | 6 |
| Griffin PowerMate™ VFO Tuning Knob..... | 6 |
| Contour Designs Shuttle Pro V.2..... | 6 |
| External Clock Reference Option..... | 7 |
| USB To Parallel Adapter | 7 |
| 4 PLUG PINOUTS..... | 8 |
| EXTERNAL CONTROL CONNECTOR (X2)..... | 8 |
| Push To Talk (PTT) Input..... | 9 |
| Mute Receiver..... | 9 |
| External Linear Amplifier Keying..... | 9 |

TABLE OF CONTENTS

Universal Controller Board for Antenna and Transverter Control10

ENCLOSURE MICROPHONE CONNECTOR10

KEYER INPUT12

5 HARDWARE SETUP 13

6 SOFTWARE INSTALLATION & SETUP 19

INSTALL LATEST SOUND CARD DRIVERS 20

UPGRADING FROM AN EARLIER VERSION 20

POWERSDR EXECUTABLE INSTALLATION..... 20

POWERSDR SETUP WIZARD 25

SOUND CARD SETUP..... 34

POWERSDR CALIBRATION..... 35

 Frequency Calibration – DDS Clock Adjustment36

Auto Calibration.....36

Manual Calculation36

 Level Calibration – Spectrum & S-Meter36

 Image Null Calibration.....37

Automatic Calibration.....37

Manual Adjustment.....37

AUTOMATIC AMPLIFIER GAIN CALIBRATION..... 37

7 FRONT CONSOLE 41

(1) VFO A 42

(2) VFO B 43

(3) & (4) MULTIMETER 43

 RX Modes44

 TX Modes44

(5) BAND SELECTION & BAND STACKING MEMORIES 45

(6) MODE SELECTION..... 46

(7) FILTER CONTROLS..... 47

(8) CW SPEED (WPM) 47

(9) VARIABLE FILTER CONTROLS 47

(10) DSP CONTROLS..... 48

(11) VFO CONTROLS 48

(12) MEMORY AND SCANNER CONTROLS..... 49

(13) CPU %..... 52

(14) POWER (STANDBY/ON) 52

(15) MON (MONITOR) 53

(16) MOX (MANUALLY OPERATED TRANSMIT) 53

(17) MUT (MUTE)..... 53

(18) BIN (BINAURAL MODE) 53

(19) ATU MODES..... 54

(20) TUN (TUNE)..... 54

TABLE OF CONTENTS

| | |
|---|-----------|
| (21) PWR (TRANSMITTER POWER OUTPUT/TUNE POWER) | 54 |
| (22) AF (AUDIO FREQUENCY GAIN) | 55 |
| (23) SQL (SQUELCH) | 55 |
| (24) MIC (MICROPHONE GAIN) | 55 |
| (25) AGC (AUTOMATIC GAIN CONTROL) | 55 |
| (26) PREAMP | 56 |
| (27) DISPLAY MODE CONTROLS | 56 |
| Display Mode Descriptions | 56 |
| <i>Spectrum</i> | 57 |
| <i>Panadapter (Panoramic Adapter)</i> | 57 |
| <i>Histogram</i> | 58 |
| <i>Waterfall</i> | 59 |
| <i>Scope</i> | 59 |
| <i>Phase</i> | 60 |
| <i>Off</i> | 60 |
| <i>Cursor and Peak Position</i> | 60 |
| (28) DATE/TIME DISPLAY | 61 |
| (29) VFO LOCK | 61 |
| (30) SETUP FORM | 62 |
| (31) – (35) OPERATING FORMS | 62 |
| 8 SETUP FORM | 63 |
| GENERAL TAB | 64 |
| Hardware Config Sub-Tab..... | 64 |
| Radio Model..... | 64 |
| Hardware Setup..... | 65 |
| DDS..... | 66 |
| Options Sub-Tab..... | 67 |
| Options | 67 |
| Process Priority..... | 68 |
| Update Notification..... | 68 |
| Click Tune Offsets (Hz)..... | 69 |
| Auto Mute | 69 |
| Quick QSY | 69 |
| Main Console (Always On Top)..... | 70 |
| Filter Controls..... | 70 |
| Calibration Sub-Tab..... | 71 |
| Freq Cal..... | 71 |
| Level Cal | 71 |
| RX Image Reject Cal | 71 |
| AUDIO TAB | 72 |
| Sound Card Sub-Tab..... | 72 |
| Sound Card Selection | 72 |
| Buffer Size | 72 |
| Sample Rate..... | 73 |
| Line In Gain..... | 73 |
| Mic In Gain..... | 73 |
| Latency..... | 73 |
| Sound Card Output Voltage | 73 |

TABLE OF CONTENTS

| | |
|--------------------------------------|------------|
| Sound Card Setup Details | 74 |
| VAC Sub-Tab..... | 75 |
| Gain (dB)..... | 75 |
| Auto Enable..... | 76 |
| DISPLAY TAB | 76 |
| Spectrum Grid..... | 77 |
| Refresh Rates | 77 |
| Waterfall..... | 78 |
| Multimeter..... | 79 |
| Scope Mode | 79 |
| Polyphase FFT..... | 79 |
| DSP TAB..... | 80 |
| Options Sub-Tab..... | 80 |
| Noise Reduction Controls..... | 80 |
| Automatic Notch Filter Controls..... | 81 |
| Buffer Size | 82 |
| Noise Blanker | 82 |
| Noise Blanker 2..... | 82 |
| Keyer Sub-Tab | 86 |
| CW Pitch | 86 |
| Connections | 87 |
| Signal Shaping..... | 87 |
| Semi Break In | 88 |
| Options | 88 |
| Image Reject Sub-Tab | 89 |
| Receive Rejection | 89 |
| Transmit Rejection..... | 90 |
| AGC/ALC Sub-Tab | 91 |
| AGC | 91 |
| Leveler..... | 92 |
| ALC | 92 |
| TRANSMIT TAB..... | 93 |
| TX Profiles | 94 |
| Filter..... | 94 |
| Noise Gate | 95 |
| VOX..... | 95 |
| TX Monitor..... | 95 |
| Compression..... | 96 |
| AM Carrier Level..... | 96 |
| PA SETTINGS TAB..... | 97 |
| Gain By Band (dB) | 97 |
| EXT. CTRL TAB..... | 98 |
| APPEARANCE TAB..... | 99 |
| General..... | 100 |
| Display..... | 101 |
| Meter | 101 |
| KEYBOARD TAB..... | 102 |
| CAT CONTROL TAB | 103 |
| Cat Control..... | 104 |
| PIT Control | 105 |
| Test..... | 105 |

| | |
|---|------------|
| TESTS TAB | 107 |
| TX IMD Text | 107 |
| Audio Balance Test..... | 107 |
| X2..... | 108 |
| Impulse Test..... | 108 |
| Signal Generator..... | 108 |
| 9 OPERATING FORMS | 109 |
| (31) WAVE FORM | 110 |
| Playback | 110 |
| Playlist..... | 111 |
| Record..... | 111 |
| Options | 111 |
| Receive | 111 |
| Transmit..... | 111 |
| (32) EQUALIZER FORM | 112 |
| (33) XVTRS FORM | 113 |
| (34) CWX FORM | 114 |
| Standard CWX Controls..... | 114 |
| <i>CWX Memories</i> | 115 |
| <i>Special Characters</i> | 115 |
| Keyboard and Extended Controls | 116 |
| <i>Extended CWX Controls</i> | 116 |
| <i>Morse Definition Editor</i> | 117 |
| 10 OPERATION | 119 |
| POWER-UP PROCEDURE | 119 |
| POWER-DOWN PROCEDURE | 120 |
| TUNING METHODS | 120 |
| <i>Spectrum Drag and Click</i> | 120 |
| <i>Mouse Wheel</i> | 120 |
| <i>Mouse Wheel Hover</i> | 121 |
| <i>Spectrum Click Tuning</i> | 121 |
| <i>Keyboard Keys</i> | 121 |
| <i>USB Tuning Knob</i> | 121 |
| VOICE TRANSMISSION OPERATION | 122 |
| CW TRANSMISSION OPERATION | 126 |
| DIGITAL MODE OPERATION | 134 |
| CAT Control Setup..... | 137 |
| <i>Install N8VB's vCOM Driver</i> | 137 |
| <i>Configure the vCOM Port Pairs</i> | 143 |
| <i>Configure PowerSDR CAT Control</i> | 145 |
| <i>Configure PowerSDR Keyer Connections</i> | 146 |
| Virtual Sound Connection..... | 146 |
| <i>Create the Virtual Audio Cables</i> | 147 |
| <i>Setup VAC in PowerSDR</i> | 147 |
| Setting up Third Party Digital Programs..... | 148 |

TABLE OF CONTENTS

Using MixW with PowerSDR..... 149
Programs Needing to Connect to the Default Sound Device..... 154
ATU (ANTENNA TUNING UNIT) OPERATION156

11 TROUBLESHOOTING..... 158

 The software is oscillating between receive and transmit modes quickly. 158
 Relays do not click when starting the software or when cycling the software power button. 158
 Relays click, but I have no audio..... 163
 Relays click and I hear static, but no signals..... 166
 I hear signals, but they sound chopped up, like a motorboat. 166
 I hear signals, but cannot null the image. 167
 The signals are all off frequency..... 169
 The noise hiss when no signals are present is fatiguing. 169
 Software does not respond to PTT signals. 170
 The receiver works great, but I am getting no output power. 170
 I am getting some power out, but the power seems low (also, see above—no power issue). 171
 I am getting too much power out (>100W). 172
 I am getting signal reports that indicate echoing or distortion. 172
 I am getting signal reports that indicate I am transmitting DSB while in SSB modes..... 173
 I am getting an error message that says, “Error communicating with ATU.” 173
 It seems like the ATU is not tuning correctly. Is there a way to check? 173
 I get an error when I try to use my USB Adapter that says “USB Device not found. (-1)” 174

A REVISION HISTORY 175

Preface

Welcome to the exciting world of software defined radio. The SDR-1000 software defined transceiver is by nature ever (rapidly) evolving. It is therefore unlike most other transceivers, which once acquired, rarely if ever change.

Although the rapid development of the SDR-1000 can be exhilarating, it can also be somewhat daunting. When first confronted with an SDR-1000 and its PowerSDR operating software, the sheer number of connections, controls, and settings can seem mind boggling even to the most seasoned Ham radio operator. This operating manual attempts to both guide a user step by step through the setup process (both hardware and software) and to act as a reference once the radio has been set up. Additionally, the freely downloadable PowerSDR software will install with default settings that, in most cases, will require little adjustment. Any adjustments that you make are automatically saved and can be imported into an updated version of the software.

Due to the nature of the SDR-1000, the largest part of this operating manual, by far, will refer to software. The operating manual has numerous screenshots of windows and forms to detail the various steps. Although the manual describes the latest official release of the PowerSDR software, you may occasionally notice an earlier version identified in the title bar of a screenshot. This is because FlexRadio Systems has decided to only update a screenshot if it changes.

The first three chapters of this manual will give you an overview of the SDR-1000 radio. Chapters 4 and 5 will guide you through the hardware setup, which apart from the connections to the parallel (or USB) port and sound card of a computer, will not differ much from the transceivers to which you are most likely already acclimated.

Chapter 6 details the installation of the PowerSDR software and calibration routines. Once completed, you should be able to engage in your first QSO. For additional guidance on operating the SDR-1000 in various modes, you may want to refer to chapter 10: Operation. This is especially recommended for operating digital modes and CW, as this may be somewhat different from what you are accustomed to.

Chapters 7, 8, and 9 describe in detail the various controls on the Front Console, Setup Form and Operating Forms respectively. You may want to reference these chapters to familiarize yourself with the exact functioning of these controls and their often many options and settings.

P R E F A C E

Finally, chapter 11 is a troubleshooting guide which should help you through various issues you may face. If you cannot find an acceptable answer to an issue you may have, please do not hesitate to contact us either by email or by phone (both found on our website). You may also want to check out our website.

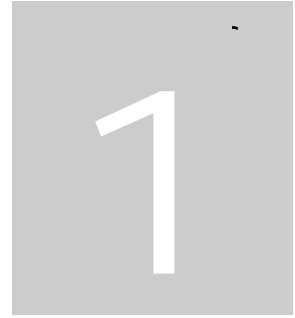
If you have any ideas on how to improve the SDR-1000, please feel free to contact us, or better still, to join our email reflector (see our website for details). Not only is the SDR-1000 a software defined radio; it is also a user defined radio.

FlexRadio Systems is committed to ensuring that your experience with the SDR-1000 will be one of the most enjoyable you have with Ham radio.

Acknowledgements

FlexRadio Systems could not be as successful, nor could the SDR-1000 radio be what it is today without the many selfless contributions of our users all over the world. These contributions have spanned and continue to span improvements to our hardware and software, ranging from bug reports and feature requests to actual design and implementation of certain functionality.

Identifying contributors by name would only risk leaving out others with equally valuable contributions. We therefore wish to suffice with a heartfelt thank you for your support and continued commitment.



General Description

The SDR-1000 Software Defined Radio is an open-software; 1 or 100W transceiver providing general coverage receive operation with amateur radio band transmission between 11kHz and 65MHz.

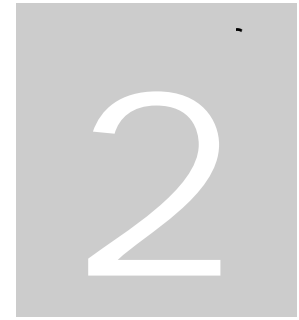
The FlexRadio SDR-1000 is a complete Software Defined Radio (SDR) transceiver designed for Amateur Radio use. It is the first fully open software transceiver that uses a personal computer (PC) for all Digital Signal Processing (DSP) and control functions of the radio. General coverage receiver operation is provided from 11kHz to 65MHz along with transmit capability on all licensed amateur bands within the coverage range. The theory behind the SDR-1000 is described in detail in a four part QEX magazine series entitled, “A Software Defined Radio for the Masses¹.” The articles are available for download from the FlexRadio website at www.flex-radio.com. There are also links on the site to other excellent reference materials related to SDR and DSP theory.

Since its introduction, additional hardware components have been added to further improve the functionality and performance of the SDR-1000. The PowerSDR[™] software, written in C#.NET, provides a full featured transceiver interface and digital signal processing for the SDR-1000. It is continuously improved over time through open source enhancements supplied by both customers and FlexRadio Systems.

The basic hardware consists of four boards in a stack within an enclosure. The hardware provides a LNA, filters, mixer, power, i/o interfacing and 1W PEP driver. Also provided is a standard interface to the optional Down East Microwave 2 meter low power transverter (intended for use in VHF, UHF, and Microwave systems), as well as an interface to the optional 100W power amplifier and the ATU (LDG Z-100). Additionally available is an optional interface for an external, customer supplied 10 MHz reference clock oscillator that may provide precision frequency control.

The Contour ShuttlePro v2 USB controller offers the ability to control the radio the way you want to without having to use a mouse or keyboard. With 15 programmable buttons and 2 concentric tuning dials (one spring loaded jog dial), this accessory offers an unparalleled level of programmable hardware control.

¹ G. Youngblood, “A Software Defined Radio for the Masses: Part 1, Part 2, Part 3, and Part 4,” QEX Jul/Aug 2002, Sep/Oct 2002, Nov/Dec 2002, and Mar/Apr 2003 respectively.



Specifications

(Specifications are subject to change without notice)

| | |
|--|---|
| Receiver Frequency Range | 11kHz-65MHz [†] |
| Transmitter Frequency Range | 160-6m Bands (optional 2m transverter) |
| IMD DR3 | 98dB |
| MDS | -132dBm |
| Minimum Tuning Step | 1Hz (DDS down to μ Hz) |
| DDS Clock | 200MHz, <1ps RMS jitter |
| Max. Receive Bandwidth | 48kHz (Limited by sound card and software) |
| Transmit Power | 100W HF 160-10m w/optional amp (See Products Page)** |
| Control Interface | PC Parallel Port (DB-25 connector) or optional USB |
| Rear Panel Control Outputs | TR relay and band control relays |
| Input Controls | PTT foot switch, Code Key, Microphone |
| Audio Interface | Line In, Line Out, Speaker Out, Microphone In |
| Enclosure Dimensions | 10" W x 10" D x 4.5" H (25.4cm x 25.4cm x 11.4cm) |
| PC Board Stack Dimensions (w/o RFE) | ~ 3"D x 4"W x 3"H (7.6cm x 10.2cm x 5.1cm) |
| Power Supply Requirement | 13.8VDC @ 1.25A (Max). 25A with optional 100W PA [†] |

* An optional filter bank is supplied that can be configured for bands below 160m

** 25W continuous carrier (100% duty cycle)

† We recommend the Astron SS-30 or SS-30M or equivalent power supply for the SDR-1000



Hardware Configuration

Standard Configuration

High Performance Preamplifier

The front end incorporates a Sirenza Microdevices SGA-6286 SiGe HBT MMIC amplifier with a noise figure of less than 2dB. With the INA baseband gain of the SDR-1000 set to 26dB, the overall receiver noise figure is approximately 4dB. The TRX board also allows 0dB INA gain setting for large signal handling capability. A 10dB fixed attenuator may be switched in front of the LNA to increase IP3 when needed. Overall there are four gain settings done in hardware to allow you to optimize the front end for your needs.

- ❑ The front end of the SDR-1000, provides the following key benefits:
- ❑ Lower receiver noise figure
- ❑ Reduces relative strength of DDS local oscillator spurs in relation to the signal of interest
- ❑ Lowers local oscillator radiation in relation to the signal of interest
- ❑ Fixed source/load impedance for the filters
- ❑ Fixed source impedance for the QSD (Quadrature Sampling Detector, IQ Mixer)

Fifth Order Low Pass Filters

Nine 5th order low pass filter banks are provided to cover all amateur bands from 160M-6M. Since the filters are low pass, general coverage reception is still provided. The filters improve 3rd harmonic rejection by more than 35dB over the original SDR-1000. There are BPF filters in front of the preamp board (in the signal chain) to provide overload protection for the LNA. The LNA is then followed by the low pass filters, which in turn drive the QSD (mixer) on the TRX board. A spare filter bank is provided for user configuration to allow improved coverage on other bands such as medium wave or VLF bands. While the SDR-1000 covers frequencies down to 11kHz, additional filtering is required to minimize response to odd order harmonics. The optional filter bank allows the user to customize the radio to add support for such frequencies.

Enhanced 1W PEP Driver

The low power model incorporates a 1W PEP driver amplifier. The driver is between the new front end low pass filters and the band pass filters on the BPF board, which yields increased harmonic suppression. The driver circuit also allows for amplifier stability when driving capacitive loads and increases the gain to reduce sound card drive requirements.

28MHz Transverter IF Takeoff

The RFE provides an internal connector for integration of the Down East Microwave 2M transverter IF (DEMI144-28FRS). Only a single coax cable is required for both RF and TR control. The SDR1K-ENC enclosure has mounting holes to allow either common or separate transmit/receive connections for transverters. You can even set up the SDR-1000 to provide a dual 144/28MHz IF to cover all of the VHF through microwave bands.

Key features of the DEMI144-28FRS are:

- ❑ 144-146MHz coverage
- ❑ Unity gain receive conversion
- ❑ 4dB NF maximum
- ❑ +17dBm double balanced mixer
- ❑ Three chamber helical filter
- ❑ 100mW nominal linear output

Experimental Impulse Generator

An experimental circuit has been incorporated that creates a 4.5ns pulse for equalization of the QSD and sound card quadrature path. DSP software will be written to calculate the impulse response of the circuit and automatically correct for phase and amplitude imbalances between the channels. In theory, this will provide dynamic opposite sideband rejection optimization.

Enclosure

The SDR1K-ENC enclosure is designed to accommodate expansion options including the Down East Microwave 2M IF transverter and the 100W PEP HF linear amplifier and LDG Z-100 automatic antenna tuning unit options. Rear panel connections are provided for the PC parallel control port, sound card interface, 7 external control lines, split or common transverter RF connections, and QRP and QRO RF connections. An AUX mounting hole is provided for the 10MHz reference clock option connector when precision frequency control is desired.

Upgrades

2 Meter Transverter

The DEM 144-28FRSK is a low power, high performance 144 MHz to 28 MHz transverter kit design to be used in conjunction with Flex Radio System's SDR-1000 Software Defined Radio transceiver. It is intended to be used as a 2nd conversion IF for microwave transverters. The 144-28FRSK has a nominal linear output power of 50 - 100 mW with the 28 MHz. IF drive provided by the SDR-1000. On the receive side, a high dynamic range amplifier, a high level double balanced mixer (+17.0 dBm) and a three chamber helical filter are employed to providing a over load proof, unity gain front end with superior selectivity. It is the same design as our high performance 2 meter transverter but without the GaAs FET front end that would provide excessive system gain not require in a microwave transverter system. The transverter is complete with all interfacing required to install in the SDR1K-ENC enclosure and operate with the SDR-1000.

The 144-28FRSK's 144 MHz input/output utilize BNC connectors (supplied in this kit) that are mounted in the existing holes in the SDR1K-ENC enclosure. The transverter also has a built in relay for external switching duties if required. It will "Shadow" the TR switch timing of the SDR-1000. The only external wiring required to operate a microwave transverter after the completion of this kit are a simple BNC cable or cables that will carry both TX and RX signals along with the keying voltage to activate the microwave transverters of choice transmit functions. It is a complete, simple to use kit and will provide the highest 2 meter to microwave transverter performance on the market today.

100W Power Amplifier

The SDR-100WPA is a 100 Watt HF linear power amplifier, designed exclusively for installation in SDR-1000 Software Defined Transceivers. Adding the 100WPA to existing radios required limited kit building skills including wire/coax preparation, connector assembly, soldering, and mechanical assembly.

Specifications

- ❑ Power output: 100W PEP, 50W on 60m
- ❑ Bands: 160, 80, 60, 40, 30, 20, 17, 15, 12, 10 meters
- ❑ Elliptical filters on all bands
- ❑ Drive requirements: <500mW for full output
- ❑ Output transistors: Rugged matched pair 2SC2879s
- ❑ Duty cycle: Designed for intermittent SSB service (25W continuous duty)
- ❑ Harmonic Content: < -43dBc
- ❑ Load tolerance: 2:1 SWR or better

- ❑ Software monitoring: Forward/Reverse Power and SWR
- ❑ SWR protection: automatic power reduction for SWR > 2:1
- ❑ DC power requirements: 13.8VDC @ 25A nominal for full output
- ❑ Internal fuse: 25A automotive mini fuse with reverse polarity protection
- ❑ Output: BNC 50 Ohm on rear panel
- ❑ Cooling: 80x80x15mm, 23dB-A fan provided
- ❑ Internal expansion connections: ATU RF In/Out and control

Automatic Antenna Tuning Unit

The SDR-ATU is a fully integrated automatic antenna-tuning unit across the entire HF range. It will tune dipoles, verticals, Yagis or virtually any coax-fed antenna. It will match a wide range of antennas and impedances. Features include:

- ❑ Microprocessor controlled, switched L tuning network
- ❑ Automatically matches antennas from 6-800 ohms impedance, or a 10:1 SWR, 3:1 on 6m
- ❑ Tunes from memory in less than one half second. A full tune will take from 1-6 seconds.
- ❑ 200 memories for almost instant return of previously tuned frequencies.
- ❑ Continuous coverage from 1.8 to 54 MHz
- ❑ Power range 0.1 to 125 Watts, 50 Watts on 6m

Griffin PowerMate™ VFO Tuning Knob

The attractive GCN-SLV Griffin Technology PowerMate control knob connects to a USB port to provide variable-rate VFO tuning for the SDR-1000. A simple click of the knob changes the tuning rate to any of four user configured tuning rates. It can even be programmed to set the volume by a press, hold, and turn of the knob. Made of high-quality machined aluminum, PowerMate feels like a solid volume knob pulled right off the front of a world-class stereo. Its heavy weight and tactile feel are a welcome departure from typical plastic USB peripherals.

Contour Designs Shuttle Pro V.2

The ShuttlePro V.2's sleek innovative design, pre-configured for many of today's popular applications, advanced programmability, and ease of use is in short, the perfect marriage of form and function.

The ShuttlePro V.2 is a powerful productivity enhancement tool for the PowerSDR software. The ShuttlePro V.2 even comes pre-configured for many of the leading applications. You can also easily customize the ShuttlePro V.2 for virtually any application.

The ShuttlePro V.2 is designed for ergonomic integrity and maximizing productivity, allowing one-hand access to the fully programmable buttons and jog/shuttle knob. Nine (9) of the buttons have removable keycaps for easy labeling and referencing. The inner ring or 'jog' rotates through 360 degrees and provides precision control. The outer black ring or 'shuttle' is rubberized and spring-loaded. It facilitates tuning and radio control. You can also use the jog and shuttle for many other purposes such as scrolling, volume control, and sequencing.

External Clock Reference Option

The SDR-CLK External Clock Reference Option kit provides the connection required to operate the DDS oscillator from an external 10MHz or 20MHz precision frequency reference. GPS locked oscillators may be used in this configuration to provide precise frequency control of the SDR-1000. The internal 200MHz oscillator must be removed to use the external reference clock.

Note: This option requires a customer provided precision clock source not provided by FlexRadio Systems.

USB To Parallel Adapter

The SDR-USB is a USB to parallel communications adapter, specially designed to control the SDR-1000. It allows the radio to be used with computers that do not have a parallel port available. The SDR-USB supports all of the control and status lines required by the SDR-1000 and has special firmware to offload hardware polling tasks from the PC. A separate download and installation of driver software is required.

Plug Pinouts

External Control Connector (X2)

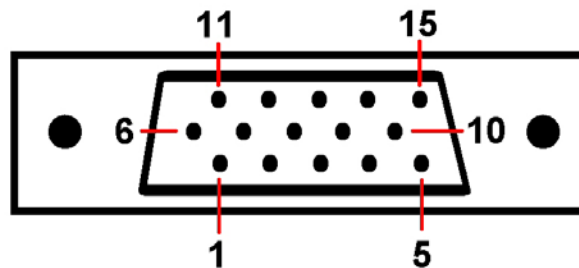


Figure 1: X2 Pin Diagram –Back Panel View

The 15 pin, high-density, female “D” connector on the rear panel (PIO board) allows open collector Darlington output control of up to seven external devices such as transverters, relays, antennas, power amplifiers, etc. It also includes two dedicated push-to-talk (PTT) inputs (Pins 10 & 11 – see below). The connector has 5VDC power at up to 50mA for powering external relays or logic. The open collector outputs may also be used with external power supplies up to a maximum of 50VDC. The maximum collector current for a single output is 500mA. A common output clamp diode return is provided (Pin 8) to handle relay transients. The table below details the pin connections. A simple way to make the cable/connector is to cut the end off a VGA PC monitor cable. This will eliminate the need to solder any connections on the 15-pin D connector.

Table 1: X2 Pin Connections

| Pin # | Function | Specification |
|-------|--------------------------|--|
| 1 | Control 1 | Open Collector – 50V, 500mA Max. |
| 2 | Control 2 | Open Collector – 50V, 500mA Max. |
| 3 | Control 3 | Open Collector – 50V, 500mA Max. |
| 4 | Control 4 | Open Collector – 50V, 500mA Max. |
| 5 | Control 5 | Open Collector – 50V, 500mA Max. |
| 6 | Control 6 | Open Collector – 50V, 500mA Max. |
| 7 | PTT Output (PA Control)* | Open Collector – 50V, 500mA Max. |
| 8 | Common Protection Diode | Connect to Relay + Supply |
| 9 | Not Connected | Not Connected |
| 10 | PTT Input (Secondary) | SSB/AM/FM only (no CW) – Ground this line to key transmitter |
| 11 | PTT Input (Main) [S3] | Works in all modes – Ground this line to key transmitter |
| 12 | Software Audio Mute** | Pull to ground to Mute the receiver. |
| 13 | Not Connected | Not Connected |
| 14 | Vcc | 5VDC Output @ 50mA |
| 15 | Ground | Signal Ground for all inputs/outputs |

* Requires enabling **X2 TR Sequencing** on the Setup Form - General Tab, Options Sub-Tab.

** Requires enabling **Auto Mute** on the Setup Form - General Tab, Options Sub-Tab.

Push To Talk (PTT) Input

The External Control (X2) connector has two PTT input connections as seen in Table 1 above. Pin 11 works in all modes, while Pin 10 works only in voice modes. Pin 11 is recommended for most applications. PTT is activated by grounding to Pin 15 either of the two respective pins.

Mute Receiver

The External Control (X2) connector has a pin dedicated to muting the receiver for multiple transceiver operation. Simply ground Pin 12 to Pin 15 to activate the software MUT (Mute) control. Note that this does not disconnect the coaxial connector from the receiver. External antenna switching is necessary to ensure that high power signals are not sent directly to the SDR-1000 receiver front end.

External Linear Amplifier Keying

External linear amplifiers may be keyed using Pin 7 on the External Control (X2) connector. Most amplifiers are switched by grounding its keying input. Connect Pin 7 (PTT Output) to the hot lead of keying input and Pin 15 to the amplifier ground. This output uses an open collector Darlington transistor switch that is rated at 500mA, 50VDC maximum. The Common Protection Diode (pin-8) should be connected to the positive lead

of the relay power supply. This will shunt the reverse voltage from the relay coil as it is turned off, thereby protecting the control transistors from transient voltages.

Universal Controller Board for Antenna and Transverter Control

The Universal Controller Board (UCB) is a customer developed and supplied accessory that connects to the External Control (X2) connector. It allows automatic software control of up to 16 antennas and/or transverters.

The Universal Controller Board is an extension of the SDR-1000 to enable additional control of external devices by the radio. The SDR-1000 X2 connector provides 6 open collector output pins for user defined functions. The purpose of the UCB is to accept data from the SDR-1000 via the X2 connector to be loaded into a 16X16 memory matrix on the UCB. Pins 1 through 4 of the X2 connector are used as an address field by the UCB to address one of 16 registers within the memory matrix. Each one of these UCB registers has 16 bits, of which any combination can be operational. Each bit in a register corresponds to a relay that will be picked.

Enclosure Microphone Connector

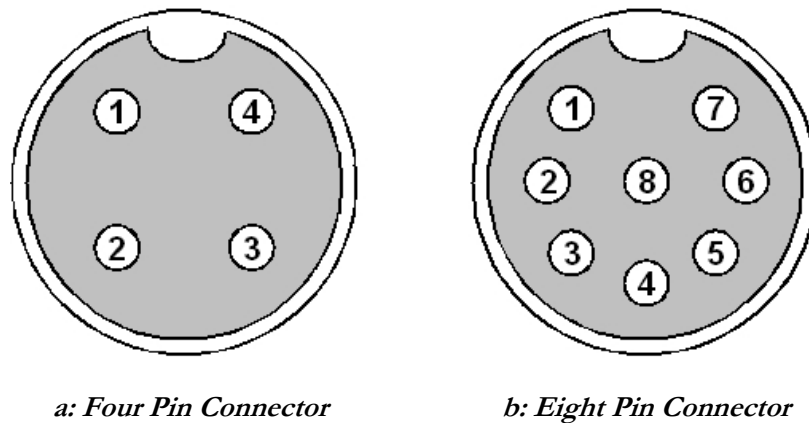


Figure 2: Microphone Connector Pin Diagram – Front Panel View

A front panel connector is provided for microphone and push-to-talk (PTT) input. Depending on when your radio shipped, the connector will be a 4 or 8 pin connector. See the tables below for the pinouts for each of these connectors. The PTT input must provide a contact closure to ground to activate the transmitter.

The 8-pin microphone configuration offers the ability to connect a balanced microphone through the front panel connector. Keep in mind that it is possible to plug a microphone directly into either of our recommended sound cards (Delta 44 and FireBox). They both offer 1/4" connectors and the FireBox has true balanced inputs along with XLR connectors and adjustable Mic Preamps.

Table 2 provides the 4-pin wiring configuration for radios shipped before February 24, 2005

Table 2: Oldest Microphone Connector Pin Description

| Pin # | Signal |
|-------|------------------|
| 1 | Ground |
| 2 | Microphone Audio |
| 3 | Push To Talk |
| 4 | No Connection |

Table 3 provides the 4-pin wiring configuration for radios shipped after February 24, 2005 but before January 11, 2006.

Table 3: Older Microphone Connector Pin Description

| Pin # | Signal |
|-------|---------|
| 1 | Mic (+) |
| 2 | Mic (-) |
| 3 | PTT (+) |
| 4 | PTT (-) |

Table 4 provides the 8-pin wiring configuration for radios shipped after January 11, 2006.

Table 4: New Microphone Connector Pin Description

| Pin # | Signal |
|-------|----------------------|
| 1 | Not Connected |
| 2 | Not Connected |
| 3 | Not Connected |
| 4 | Not Connected |
| 5 | Chassis GND (Shield) |
| 6 | PTT (+) |
| 7 | Mic (-) |
| 8 | Mic (+) |

We recommend use of the Heil microphones, especially the PR series; however, the HM-10 and Goldline microphones will also work well with the SDR-1000.

Use the procedure found in the Voice Transmission Operation section in order to match the microphone appropriately with the sound card and the DSP.

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Keyer Input

While the keyer input lines can be swapped on the CW Form, we are including this pinout for completeness.

Table 5: Keyer Plug Pinout

| Connector | Signal |
|-----------|--------|
| Tip | Dot |
| Ring | Dash |
| Sleeve | Common |

For a straight key input, use the following table.

Table 6: Straight Key Plug Pinout

| Connector | Signal |
|-----------|--------|
| Tip | Key |
| Ring | Key |
| Sleeve | Common |

Note: Using a mono 2-conductor type plug will cause problems as the Ring will be shorted to the Sleeve (Key to Common or Dash to Common depending on setup).

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Hardware Setup

The hardware setup process is as simple as making connections from the SDR-1000 enclosure to the hardware necessary to run the radio (power supply, antenna, parallel port, and sound card). Refer to the numerical callouts in Figure 3 below for the location of all radio connections.

Make sure the power switch on the front panel of the enclosure is in the Off (down – “O” on some models) position. Connect the power supply negative **(1)** and positive **(2)** leads to the black (–) and red (+) power supply connectors respectively. Ensure that correct polarity is maintained as reversing the cables can cause damage to the unit. The power supply must provide 13.8VDC at 25A (nominal) for the 100W transceiver and 1.5A for the 1W transceiver respectively.

Make a low impedance connection to earth (RF) ground **(8)** to properly protect the unit.

Connect the PC parallel port to the parallel connector **(10)** on the enclosure. Use a 25-pin, male-to-male parallel cable that has all 25 pins connected straight through. These are called RS-232 cables by some manufacturers. Both parallel and audio cables are available from FlexRadio Systems. Make sure the connection is firm as a partial parallel connection can cause sporadic frequency tuning and filter selection.

Note: FlexRadio Systems distinguishes between **recommended**, **legacy** and **unsupported** sound cards. Recommended cards are actively supported and used for system development. Legacy cards are those cards that were previously recommended and are now only passively supported. Unsupported cards include all other sound cards. From time to time FlexRadio Systems will adjust a sound card’s designation; the most up-to-date information is available on our website.

FlexRadio Systems currently recommends two sound cards for use with the SDR-1000: The M-Audio™ Delta 44 is the recommended PCI interface and the PreSonus FireBox is the recommended Firewire (1394) interface (ideal for laptop connection). Each of these professional grade sound cards offers multiple inputs and outputs which allows us to mix separate monitor and radio outputs as well as leave the radio and microphone connected to the inputs. The audio quality of these recommended cards is also superior to any of our legacy (previously recommended) cards as these legacy cards were all consumer grade. Please refer to the charts and tables on the specifications page of the website that show the superior dynamic range and noise performance of our

recommended cards. This is a clear-cut example of professional quality audio equipment (M-Audio) versus consumer quality equipment (Creative). The gap between the two is substantial.

Connect the sound card to the radio as follows. If using the M-Audio Delta 44, please consult the Delta 44 Quick Start Guide located at <http://www.flex-radio.com/delta44/delta44.htm>. A similar guide for the PreSonus FireBox may be found at <http://www.flex-radio.com/firebox/firebox.htm>. Otherwise, follow this guide: First, connect the Line In cable from the sound card Line Input (usually color coded blue) to the plug labeled “To Line In” (14) on the back of the radio. Next, connect the Speaker output from the sound card (usually color coded light green) to the plug labeled “To Line Out” (12). Make sure that if your sound card has multiple speaker outputs you use the one labeled Speaker 1 (or the “front” speakers).

Note: For certain sound cards, it is necessary to install the SDR-LPF/CBL low pass filter on the Line Out connection to prevent wideband noise transmission. Some sound cards, such as the Turtle Beach Santa Cruz and Audigy2 ZS, emit noise that peaks in the 100kHz range and must be suppressed to comply with FCC rules (-43dBc). Instructions for making the SDR-LPF/CBL can be found on the FlexRadio Private Download Page. The cable must be installed with the filter module nearest to the radio. Reversing the cable will not allow the filter to operate properly.

Then connect the Microphone Input from the sound card (usually color coded red) to the plug labeled “To Mic In” (15). Note that this is only necessary if using a microphone connected to the front panel of the enclosure. If using a PC mic, this can be plugged directly into the sound card. Finally, connect amplified speakers to the plug marked SPKR (13). **Note that using non-powered speakers or low impedance headphones will load down the transmitter drive to the radio, thereby preventing full power output. This is not an issue with 4-port cards such as the Delta 44 and the FireBox since the output does not run through the radio.**

Note: It is critical that the audio cables used to connect to the sound card are seated flush against the back panel of the enclosure to provide good contact for the stereo input and output. Failure to do so will prevent opposite sideband rejection on both reception and transmission. Cables with wide connector ends are the cause of this issue in most instances. Premium “Gold” cables are NOT recommended because they are typically overpriced and their connectors are usually too large.

If the 100W amplifier is included, connect the HF/6m antennas to the HF/PA (3) connector. **WHEN THE 100W AMPLIFIER IS INSTALLED, DO NOT CONNECT ANYTHING TO THE HF/50 MHZ QRP CONNECTOR.** This connector is in parallel with the input to the power amplifier. If the amplifier is not installed, connect the antenna to the HF/50 MHz QRP connector for the 1W output configuration.

If the DEMI144-28FRS 2m transverter is installed, its connection(s) go to **(6)** / **(7)**. Please see the transverter's installation instructions for connector options that allow for common or split TX/RX. Also, if using the external clock reference option, connect the reference signal to **(4)**.

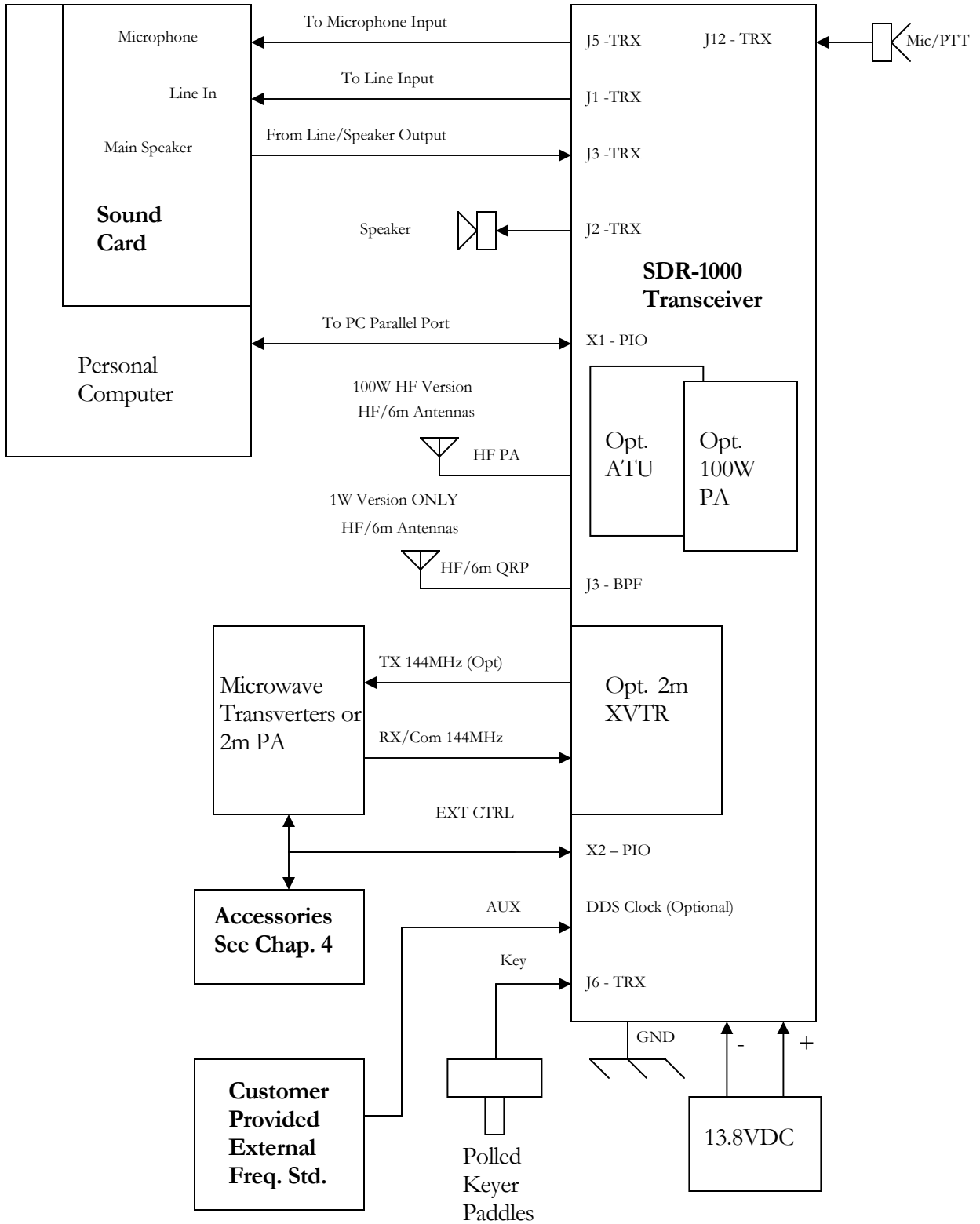
Note 1: If the 100W amplifier is installed, the AUX hole is covered by the amplifier hardware and the reference connector will be moved to another location (possibly to 6 or 7 if the transverter is not installed).

Note 2: If the amplifier is installed, it is necessary to uncheck the box that enables the PA on the Setup Forms General Tab before attempting to transmit QRP. If the amplifier is disabled in software, the 1W transmitter will be connected straight through to the HF/PA connector. The output on 6m is always QRP since the PA does not cover 6m.

Finally, connect any external hardware control lines (linear amplifiers, transverters, or relays) to the X2 connector **(9)** as defined in the section on Plug Pin Out above.

[Diagram needs to be redone in a style that shows the actual hardware and plugs including the UCB, USB, and LPF filter cable]

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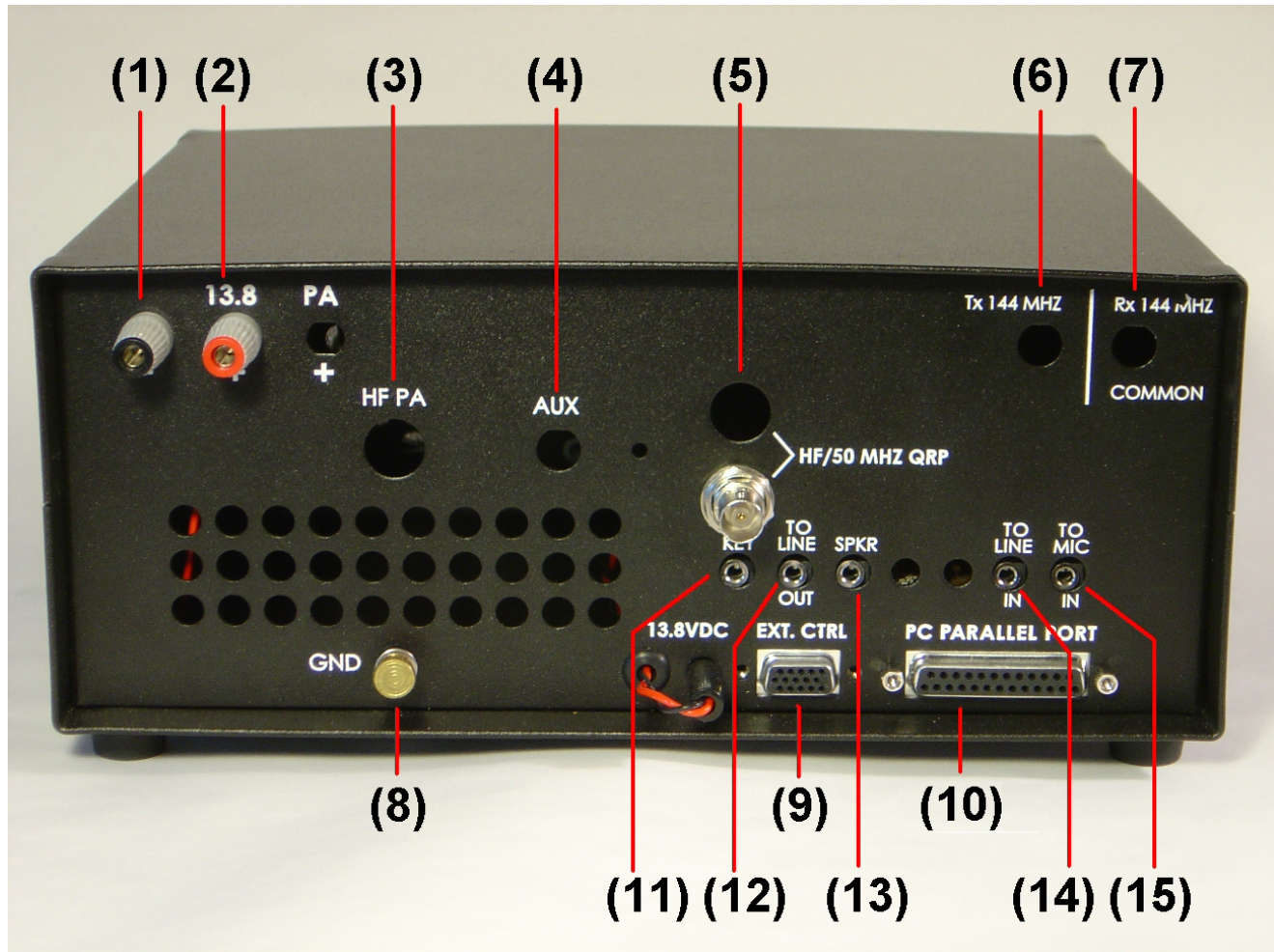


Figure 3: Back Panel of Enclosure

[Add new photo here with all connectors shown]

Connections are as follows:

- (1) Power Ground - #12AWG or greater for 100W version
- (2) 13.8VDC Power - #12AWG or greater for 100W version
- (3) 100W Power Amplifier Antenna Connector – **Use for all bands if 100W amplifier is installed**
- (4) Auxiliary Input Connector for optional External Reference Oscillator (this mounting hole cannot be used if 100W amplifier is installed)
- (5) 1W QRP Antenna Connector – **Use ONLY if 100W amplifier is NOT installed (when installed, this connector is in parallel with the amplifier input circuit.**

- (6) Two Meter Transverter Transmit Connector – Used for split transmit/receive operation (See DEMI144-28FRS installation instructions)
- (7) Two Meter Transverter Receive or Receive/Transmit Common Connector – (See DEMI144-28FRS installation instructions)
- (8) Earth (RF) Ground
- (9) X2 External Control Connector – See Table 1 for connections
- (10) Parallel Port Connector (to PC parallel port) – Optional SDR-USB USB to Parallel Adapter connects to this port
- (11) Code Key/Paddles Connector – See Tables 4 & 5 for connections
- (12) To Line Out (speaker output) on Sound Card – Green color coded. ***The SDR-LPF/CBL low pass filter cable is required for many sound cards on this connection to prevent wideband noise transmission.***
- (13) To Amplified Speaker or high impedance headphones (not to be used with the Delta 44 or FireBox)
- (14) To Line In on Sound Card – Blue color coded
- (15) To Microphone Input on Sound Card -- Pink or Magenta color coded.

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Software Installation & Setup

WARNING! Proper operation of the SDR-1000 depends on the use of a sound card that is officially recommended by FlexRadio Systems. Refer to the Specifications page on www.flex-radio.com to determine which sound cards are currently recommended. Use only the specific model numbers stated on the website because other models within the same family may not work properly with the radio. Officially recommended sound cards may be updated on the website without notice. If you have any question about the sound card you would like to use with the radio, please email support@flex-radio.com or call us at 512-250-8595.

NO WARRANTY IS IMPLIED WHEN THE SDR-1000 IS USED WITH ANY SOUND CARD OTHER THAN THOSE CURRENTLY RECOMMENDED AS STATED ON THE FLEXRADIO SYSTEMS WEBSITE. UNRECOMMENDED SOUND CARDS MAY OR MAY NOT WORK WITH THE SDR-1000. USE OF UNRECOMMENDED SOUND CARDS IS AT THE CUSTOMERS OWN RISK.

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Install Latest Sound Card Drivers

Download the latest sound card drivers from the manufacturer. Install the software using the “Driver Only” option if there is one.

- Note 1:** Using the “Driver Only” option (Audigy 2 ZS, and others) will prevent some software from loading programs that will engage audio effects such as Reverb or Echo by default. This causes problems with the receive and/or transmit performance of the radio. All such sound card audio effects should be turned off to prevent problems with the radio.
- Note 2:** If using a legacy sound card (or an unsupported sound card without native ASIO drivers) you will need to install ASIO4ALL, which can be found on the Downloads page of our website. If using one of our recommended sound cards (M-Audio Delta 44 or PreSonus FireBox), installing ASIO4ALL is unnecessary.

Upgrading From an Earlier Version

It is recommended that you leave older versions of the PowerSDR console installed when upgrading from a previous version. After reviewing the new version and verifying that your setup works, uninstalling previous versions is fine (but not necessary). Note that it is necessary to manually delete the database file (PowerSDR.mdb) from the application directory (usually c:\Program Files\FlexRadio Systems\PowerSDR x.y.z) in order to completely remove previous versions.

PowerSDR Executable Installation

Download the latest PowerSDR_x.y.z.zip to a directory on your hard drive (saving to the Desktop is recommended) and unzip the contents. Double click the Setup file (Setup.exe) to start the installation process. The PowerSDR installation will prompt you to install the .NET framework if it is not installed and it will point you to the appropriate web address for downloading as seen below.

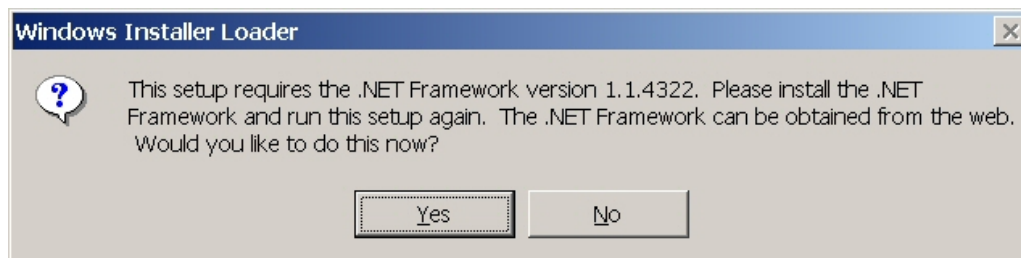


Figure 4: Prompt for .NET Framework

Follow the instructions to install the framework using the download from Microsoft's website and then restart the Setup.exe program. You should see the following screen.

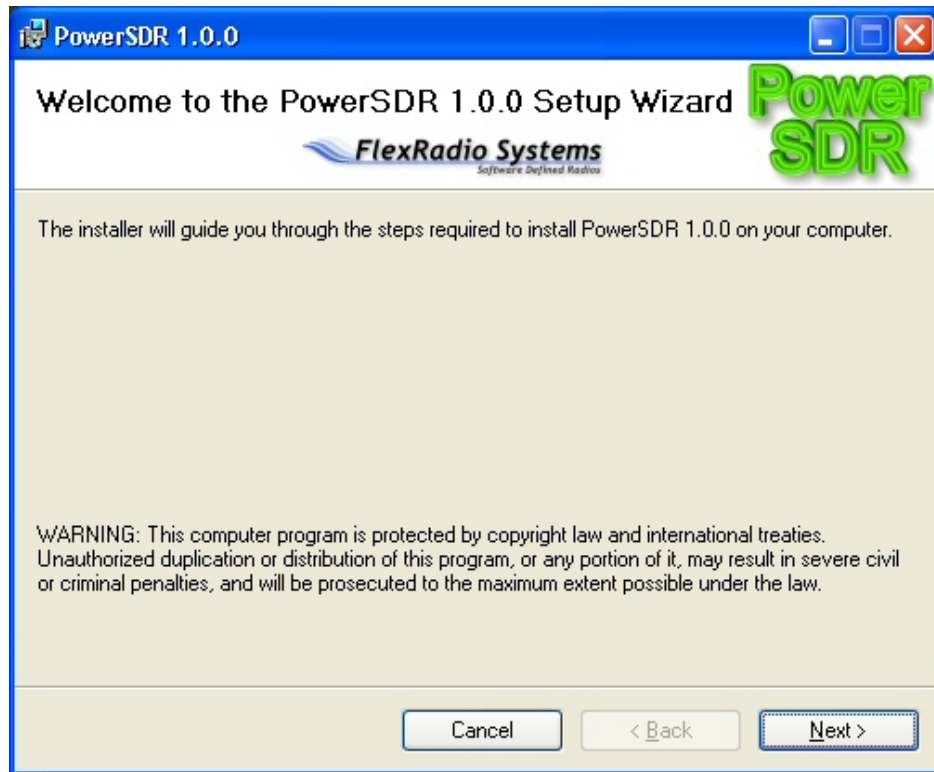


Figure 5: PowerSDR Installation Welcome Screen

Click the **Next** button to continue.

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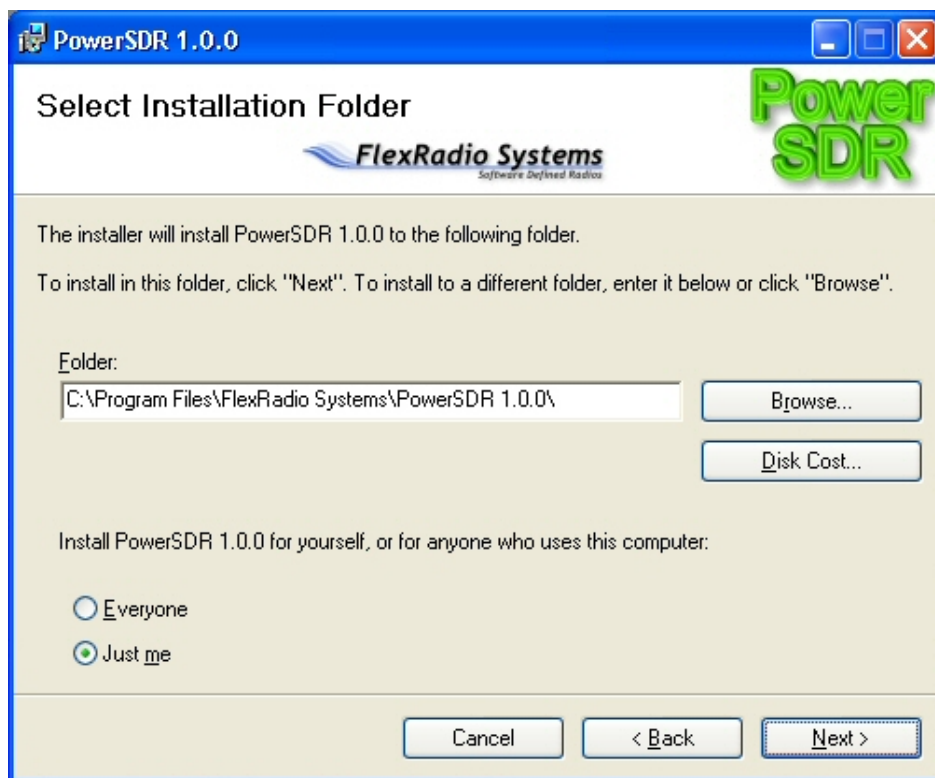


Figure 6: PowerSDR Installation Folder Selection

You can change the installation directory here, though we recommend you use the default for troubleshooting purposes. Click the **Next** button to continue.

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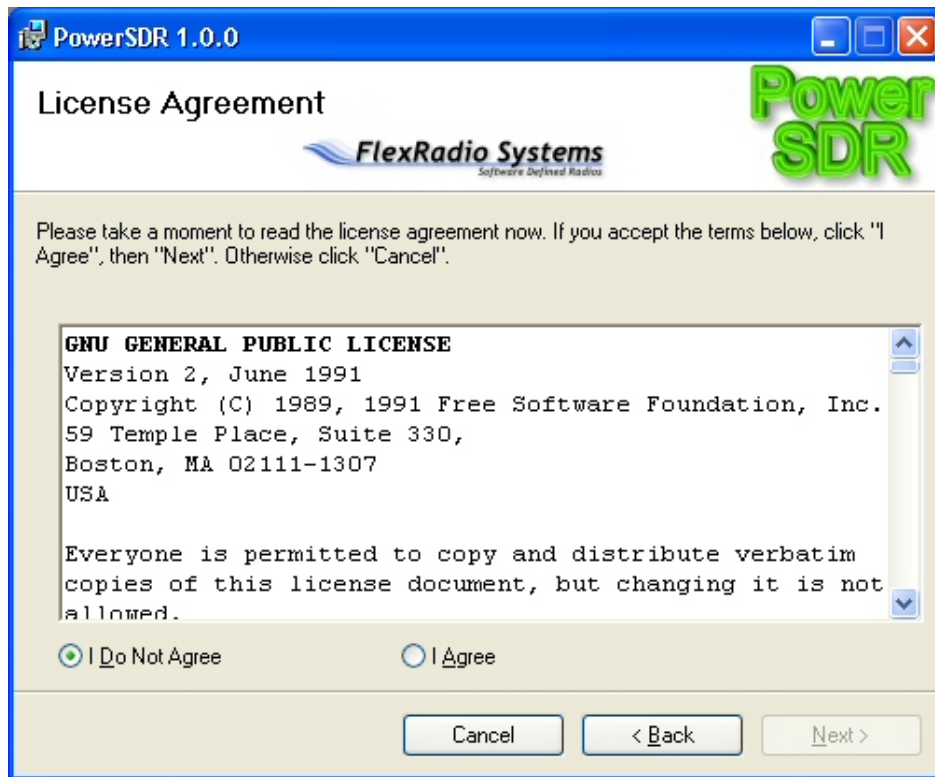


Figure 7: PowerSDR Installation License Agreement

Read the entire GNU Public License (wink), then click **I Agree** and click the **Next** button to continue.

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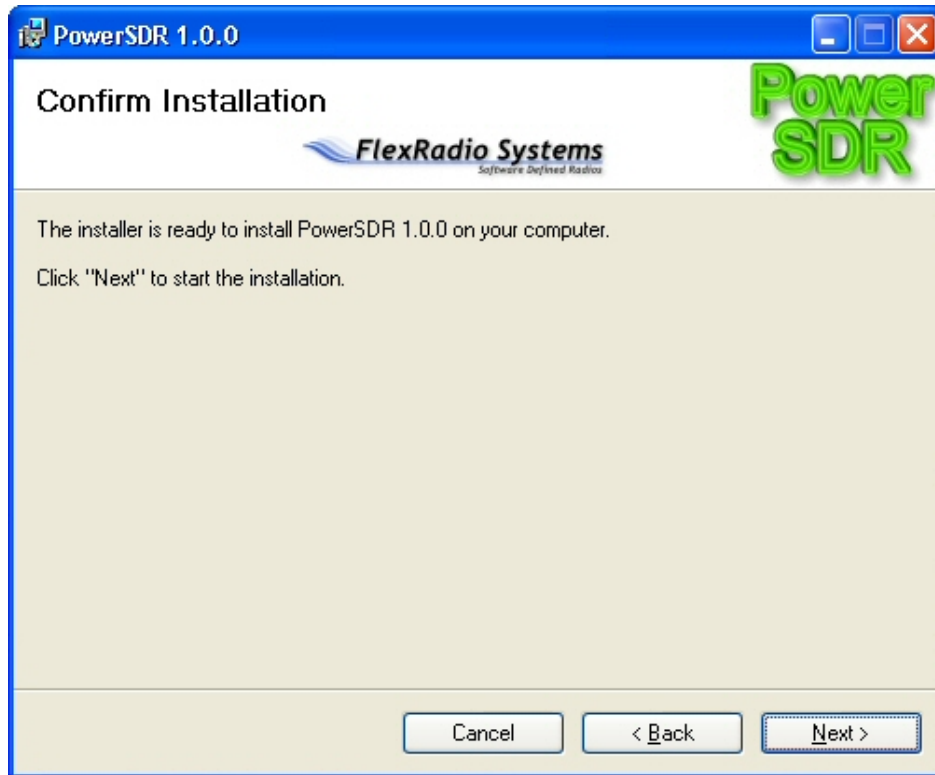


Figure 8: PowerSDR Installation Confirmation

Click the **Next** button to confirm these settings and to copy the necessary files to the selected install directory. Once the files have been copied, you will see the following screen.

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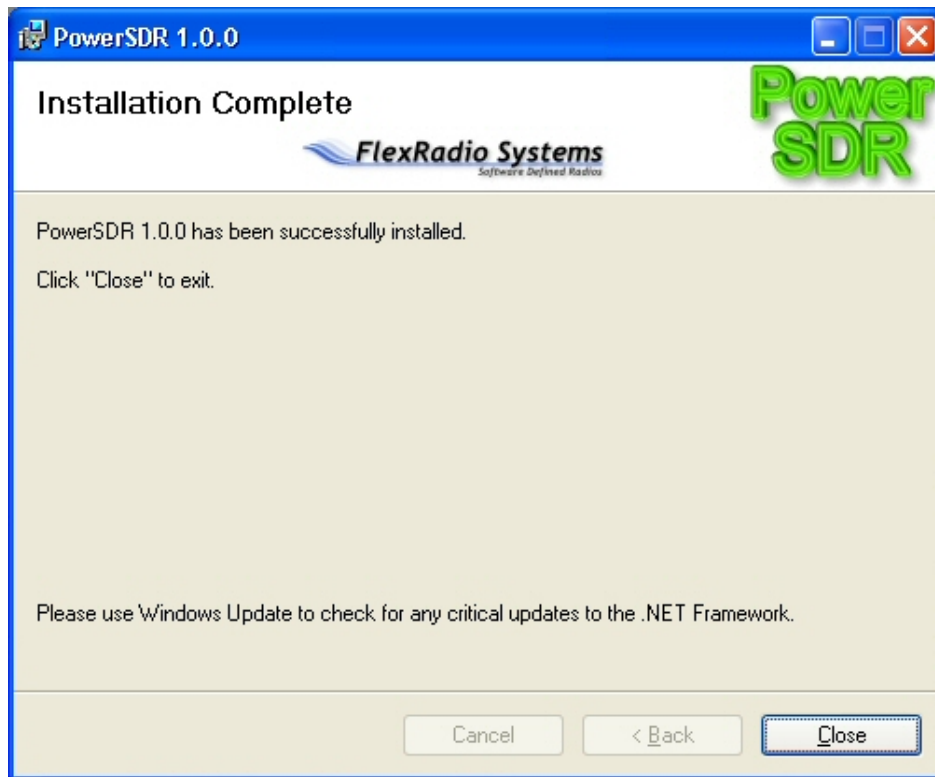


Figure 9: PowerSDR Installation Complete

Click the **Close** button to complete the installation and close the dialog.

PowerSDR Setup Wizard

Before powering up the radio hardware, load the PowerSDR console using the shortcut on the Desktop (or in the Start menu). When you run (a new release of) PowerSDR for the first time an optimization routine will run and the following screens will appear:

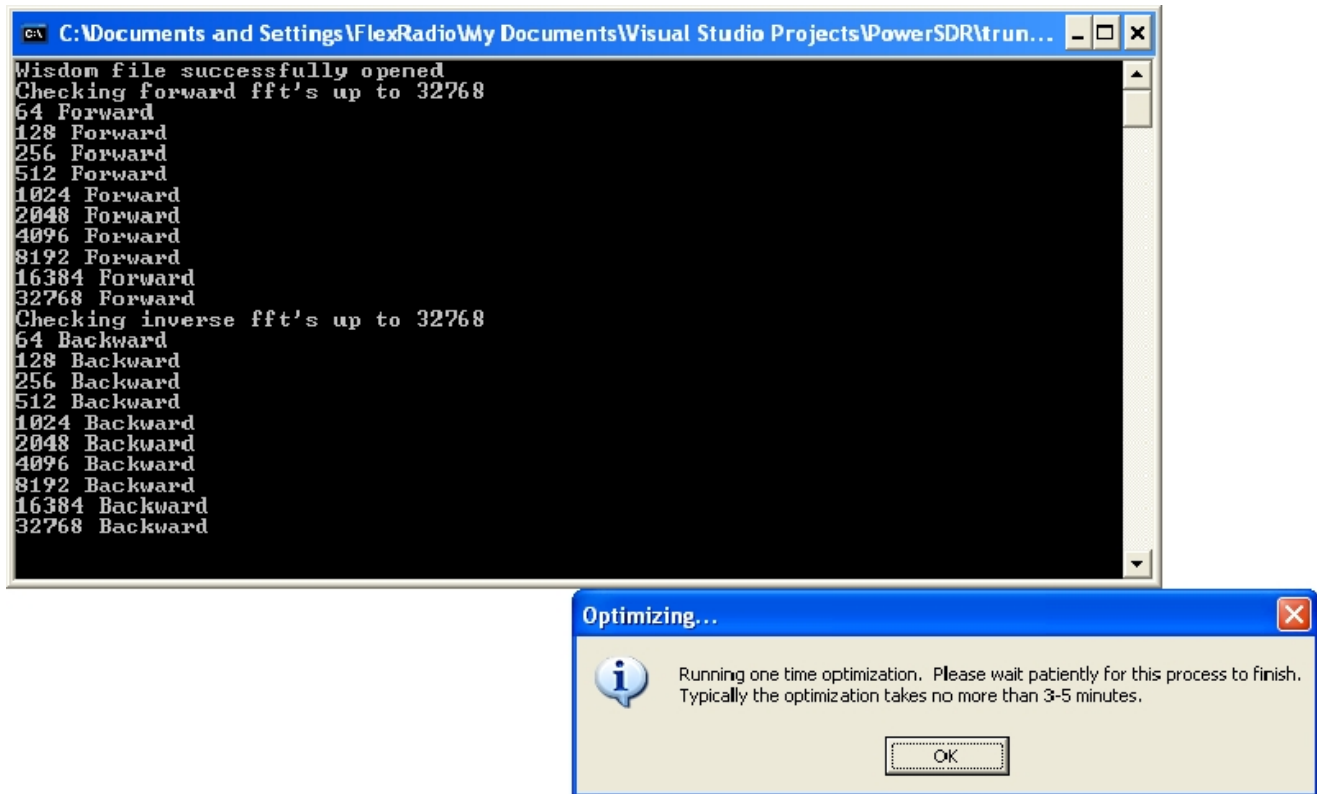


Figure 10: Optimization Routine

Click **OK** and let the routine run.

Note: This routine aims to optimize the FFT calculations for the environment (hardware and software) in which the calculations will be performed. For optimal performance, you should therefore close all applications you will normally not be running simultaneously with PowerSDR. The routine will save a file called FFTW to the directory in which PowerSDR resides. If you wish to run FFTW again, delete this file from the directory and start up PowerSDR.

When the routine has completed a brief startup sequence will follow, after which you should be greeted by the PowerSDR Setup Wizard as shown in the figure below.

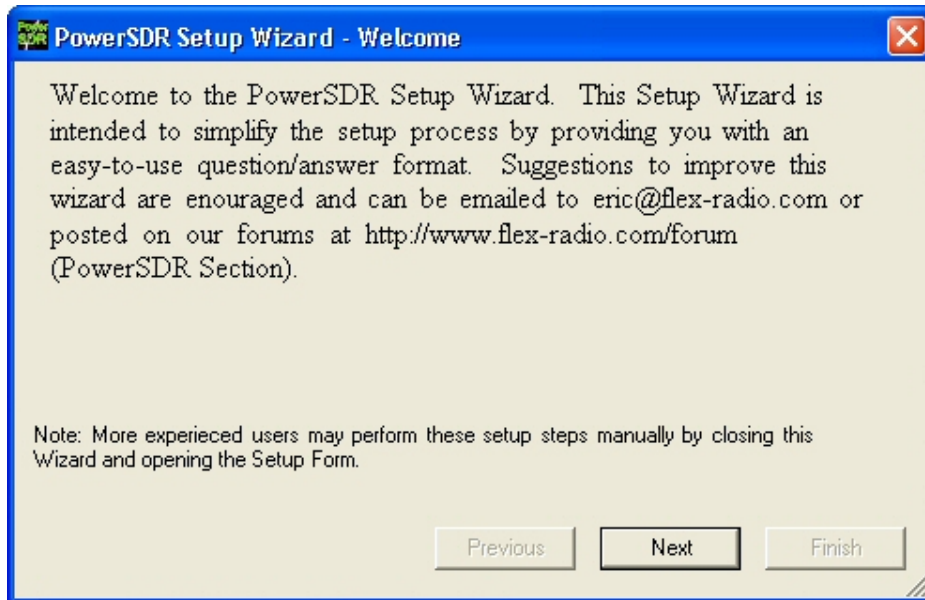


Figure 11: PowerSDR Setup Wizard Welcome

Click the **Next** button to continue.

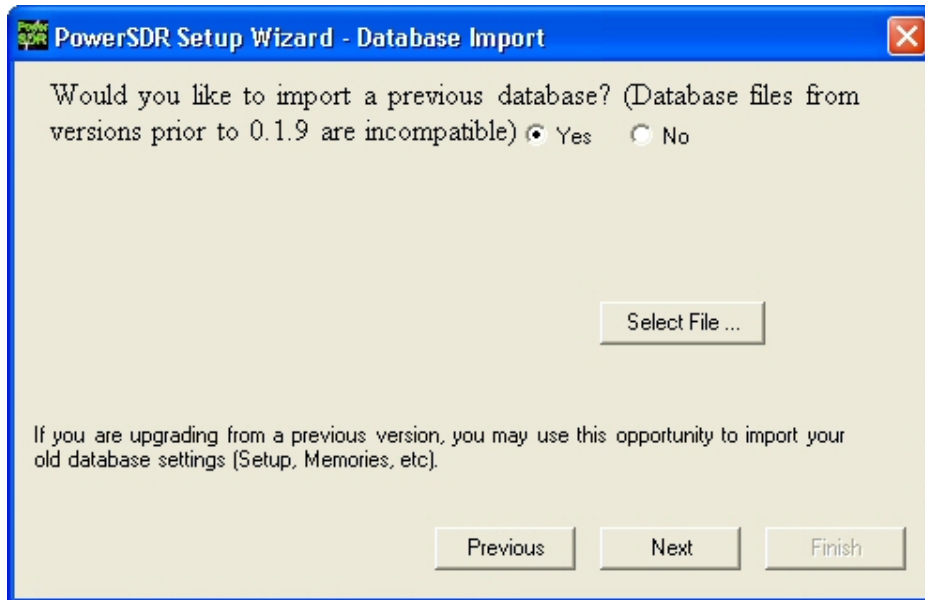


Figure 12: PowerSDR Setup Wizard - Database Import

Click the **Yes** or **No** button to indicate whether you would like to import a database from a previous version (assumes you are upgrading). If yes, click the **Select File ...** button and select the PowerSDR.mdb file from a

previous version (default directory is C:\Program Files\FlexRadio Systems\PowerSDR x.y.z\). A prompt will confirm that the database file was imported successfully as seen in the screenshot below. Click the **OK** button.

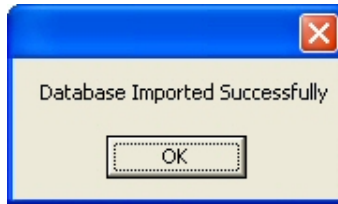


Figure 13: Successfully Imported Database Prompt

Click the **Next** button to continue. If you imported a database, the rest of the steps will be skipped and you will jump straight to the finished screen.

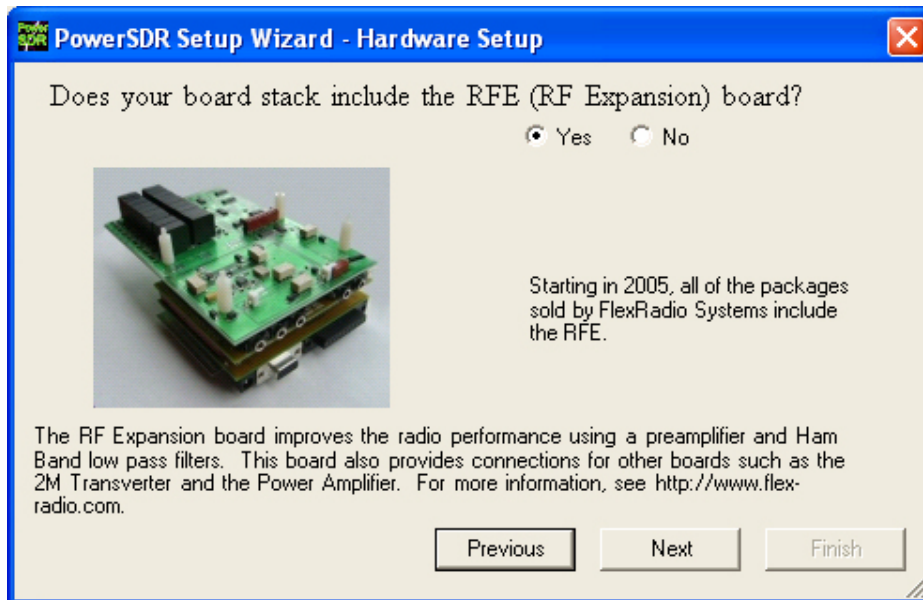


Figure 14: PowerSDR Setup Wizard – RFE

Select the **Yes** or **No** button to indicate whether your configuration includes the RF Expansion Board (RFE), and click the **Next** button to continue. Please note the configurations that include the factory installed RFE.

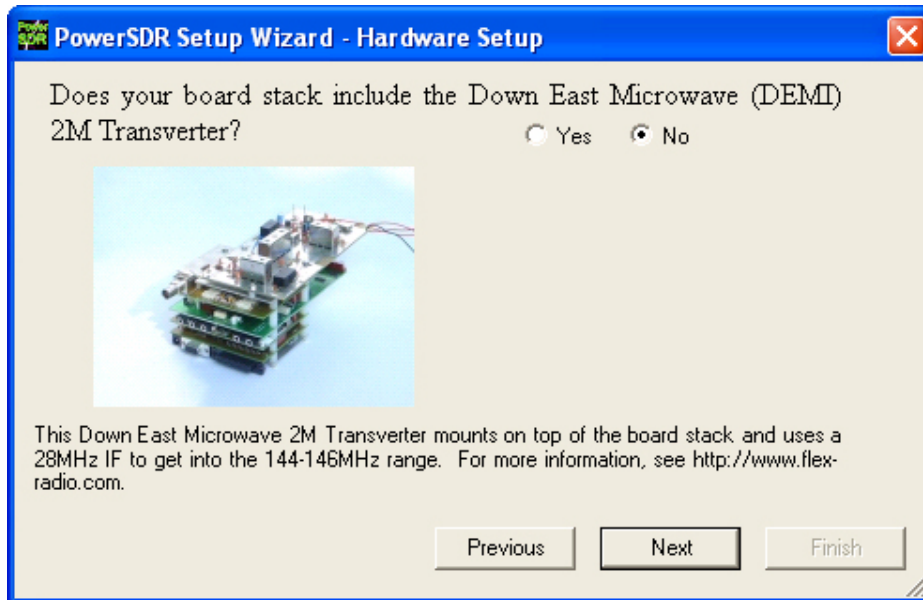


Figure 15: PowerSDR Setup Wizard - 2m XVTR

Select the **Yes** or **No** button to indicate whether your configuration includes the optional 2m transverter. If yes, select your model using the drop down box. Note that FlexRadio Systems only sells the DEMI144-28FRS model. It uses negative logic keying on the coax to the transverter. Click the **Next** button to continue.

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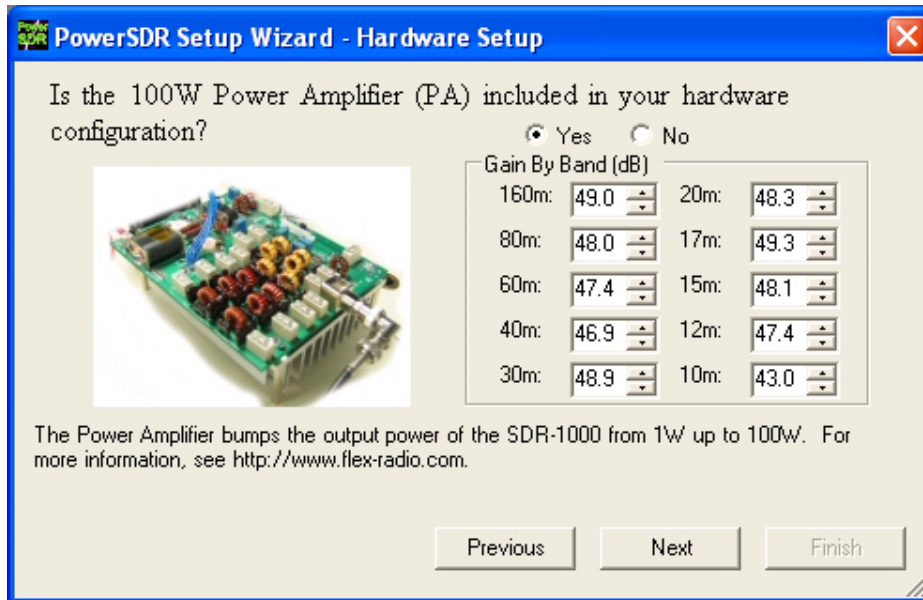


Figure 16: PowerSDR Setup Wizard – PA

Select the **Yes** or **No** button to indicate whether your configuration includes the 100W Power Amplifier. If yes, enter the gain values found on the sheet included with the amplifier shipment. Click the **Next** button to continue.

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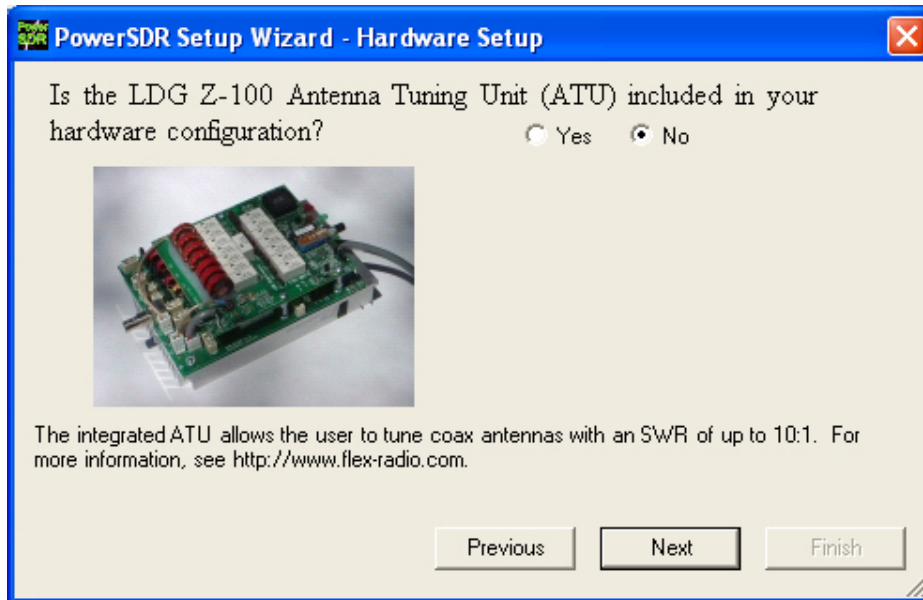


Figure 17: PowerSDR Setup Wizard – ATU

Click the **Yes** or **No** button to indicate whether you have the integrated ATU installed. Click **Next** to continue.



Figure 18: PowerSDR Setup Wizard - Ext. Clock

Click the **Yes** or **No** button to indicate if you will be using the optional external clock reference. If yes, select the frequency of your source using the drop down box (10 or 20MHz). Click **Next** to continue.



Figure 19: PowerSDR Setup Wizard - USB Adapter

Click the **Yes** or **No** button to indicate if you will be using the USB to Parallel adapter. Click **Next** to continue.

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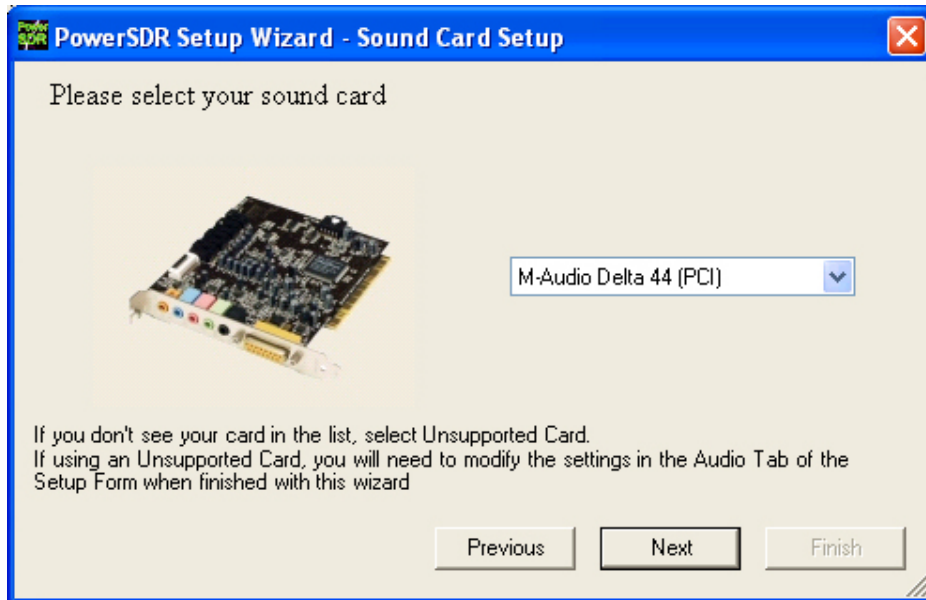


Figure 20: PowerSDR Setup Wizard - Sound Card

Select your sound card from the drop down box. If you are using a card other than the recommended or legacy cards listed on the FlexRadio Website Specifications page, then select the Unsupported Card option (this will require further setup after the wizard is complete). **Unsupported sound cards may be used at the customer's own risk. FlexRadio Systems cannot vouch for proper operation or support for such cards.** Click the **Next** button to continue.

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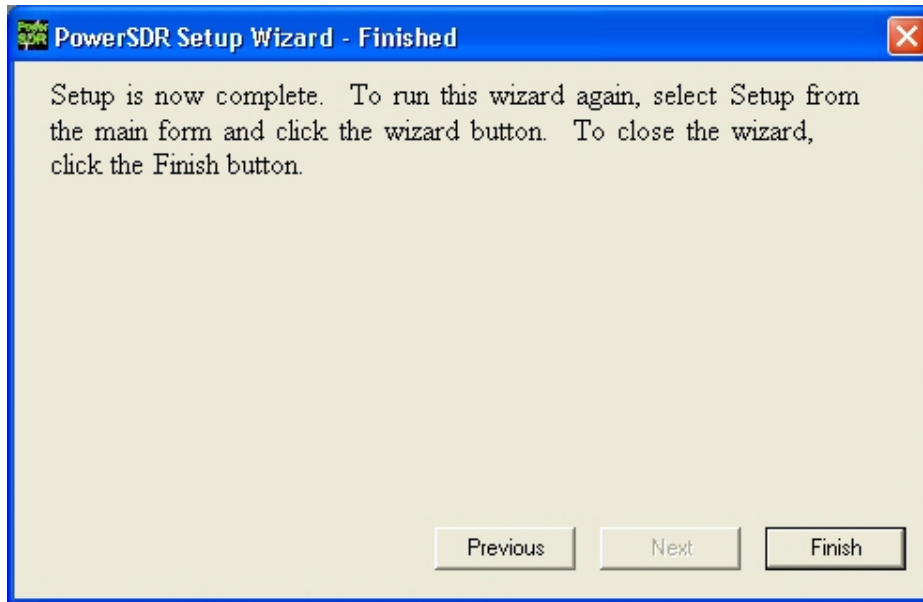


Figure 21: PowerSDR Setup Wizard – Finished

The Setup Wizard is now complete. Click the **Finish** button to complete the wizard.

Sound Card Setup

If using the M-Audio Delta 44, please refer to the Delta 44 Quick Start Guide located on our download page and at the following address: <http://www.flex-radio.com/delta44/delta44.htm>.

Otherwise, if using a consumer sound card, it may be necessary to do some more software setup. Open the software and look at the Audio Tab on the Setup Form. With the radio powered on and running, adjust the Line In Gain down from 50 until the noise floor just begins to rise. Adjust the value back up 3-4 and call that your optimal Line In gain. If the noise floor never rises, use a value of '1'. This will give you the least noise due to the sound card yielding more room for larger signals. This translates into higher dynamic range. Note that making any changes to the Line In gain will invalidate the calibration. So run make sure to calibrate after doing this procedure.

PowerSDR Calibration

Power on the radio by pressing the **Standby** button. Once you have verified that the radio is operational (try tuning a local AM station), it is time to calibrate the software. To bring up the calibration options, click the **Setup** menu button on the front panel. Once the Setup form is open, click the **General** tab. The calibration settings are on the Calibration subheading tab.

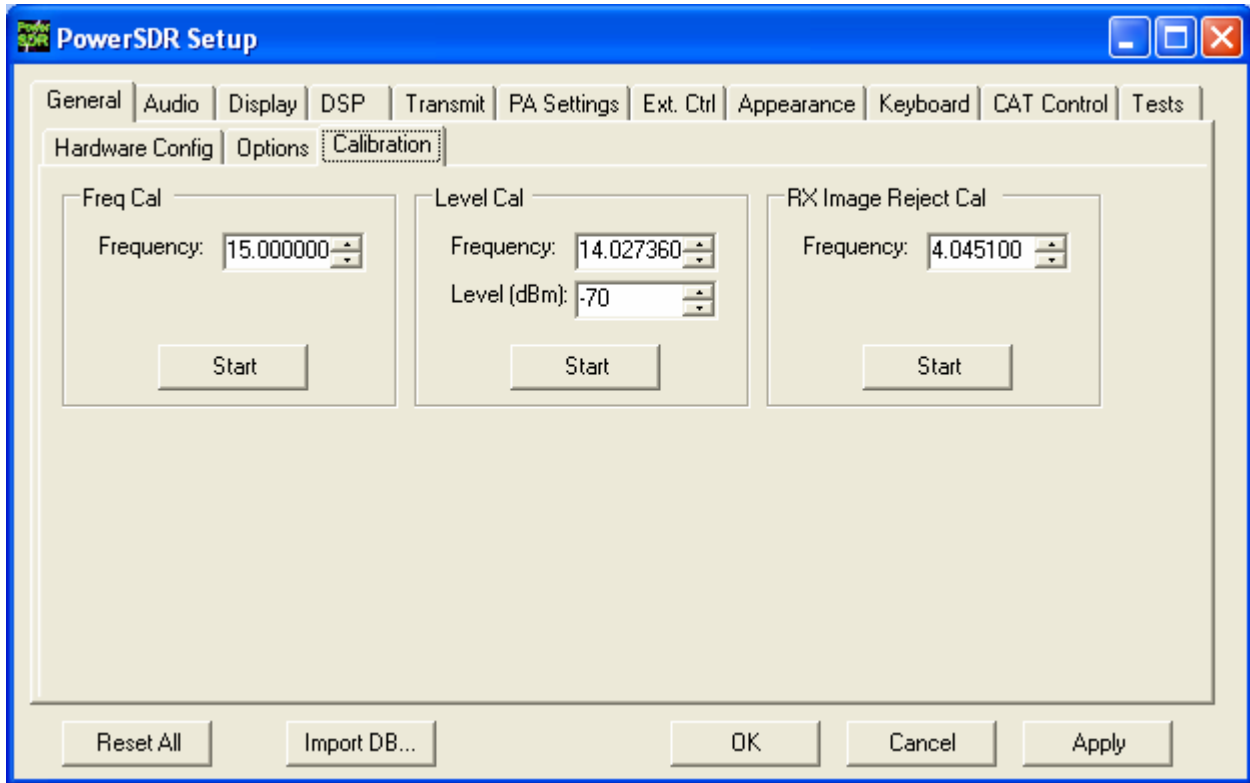


Figure 22: PowerSDR Calibration Settings

The three calibration routines available (Frequency, Level, and Image Null) are detailed in the sections below.

Note: Before calibrating the frequency, the radio needs at least a 15 minute warm-up period to ensure temperature drift is negligible.

Frequency Calibration – DDS Clock Adjustment

Auto Calibration

To calibrate the frequency, enter the frequency of the source in the entry box as seen in the figure above (e.g. 7.000000MHz). A calibrated signal generator is ideal, but a frequency standard station such as WWV may be used instead so long as its signal is at least 30dB above the noise floor. A separate transmitter may also be used as the signal source. We do not recommend calibrating on local AM stations as the correction may not be as accurate at such low frequencies. Once the frequency of the station or signal generator is entered, click the top **Start** button (across from the word Frequency) to run the calibration routine. Using the manual adjustment routine below may allow slightly better results due to fine adjustments.

Manual Calculation

The calculation for the DDS adjustment is not complicated. Divide the DDS frequency (200MHz) by the known signal frequency (say, 10MHz WWV) and then multiply by the difference between the known signal and the measured signal. For example, say you tune the VFO to 10MHz and the peak shows up at 9.999700MHz (difference is $10.0 - 9.999700 = +300\text{Hz}$). The Clock Offset would be $200/10 * 300 = 6000$. Plugging in 6000 into the Clock Offset control should zero beat the signal. The frequency offset is the actual clock oscillator offset from its nominal 200MHz output. The oscillator is rated at 100ppm, which yields a maximum offset of +/- 20kHz from 200MHz. Note that if the measured frequency were 10.000300MHz, the offset would be -6000. Fine adjustments may be made directly on the **Clock Offset** control on the General Tab of the Setup Form.

Level Calibration – Spectrum & S-Meter

For proper calibration of the spectrum and S-meter, some type of calibrated signal source is required. If no signal generator is available, Elecraft offers an inexpensive XG1 1uV and 50uV Receiver Test Set. To calibrate the visual spectrum and the S-Meter, set the signal generator to a known strong signal level such as -73 dBm (no less than 30dB above the noise floor) and tune the radio to place the carrier inside the filter passband. Enter the signal level of the generator in the Level (dB) text box as shown in the figure (e.g. -73dBm). If no calibrated signal source is available, you may use a second transceiver to either transmit a signal or to find a steady signal on the air (this will give a much less accurate level calibration). Without use of a calibrated signal source, the spectrum display and power meter may be off by as much as 10dB or more. Click the middle **Start** button across from the word Level to start the calibration. This calibration is accurate to within 0.1dB with respect to the source.

Image Null Calibration

Since the SDR-1000 has an analog front end, phase and amplitude imbalances can exist between the I and Q channels. Fortunately, this is a simple correction for the DSP software. The procedure for calibrating the receiver is described below. For a description of how to minimize the transmit image, look at the Setup Form chapter below in the section on Transmit Rejection.

Note: A signal generator or a separate transmitter with dummy load is required for this alignment.

Automatic Calibration

Increase the signal generator output level to -50dBm so that the image signal will be large enough to give optimal calibration. On the General tab of the Setup form, enter your signal generator frequency. Then click the lowest **Start** button next to the words “Image Null” to automatically find the optimal Phase and Gain settings. Using the settings found by the automatic adjustment as a starting point, manually adjusting the controls may offer slightly better results. Note that any other signals in the passband besides the image may hinder the routine from finding the best null.

Manual Adjustment

The controls for the Image Rejection are on the DSP tab of the Setup form. They allow fine adjustment of amplitude and phase to allow a null of the opposite sideband image in the time domain. The image can be found roughly 22kHz above a strong signal. The null actually moves around in a 3kHz window centered at Signal+22.050kHz if Spur Reduction is on. Alternately modify the Gain and Phase controls on the DSP Tab of the Setup Form to find the best possible null. Clicking on the track bar will give unit increment/decrements to the value and then tweaking can be done using the Up/Down control (for fractional component).

Automatic Amplifier Gain Calibration

CAUTION: The amplifier uses a fraction of the audio drive our recommended sound cards provide. For this reason, it is imperative that these steps be followed exactly to prevent overdriving the PA. Using a second sound card/PC with the PA can easily cause audio overdrive resulting in damage to the PA. Use extreme caution if using a non-standard setup. Contact FlexRadio Systems with any questions or concerns in this area.

The first step in setting up the PA is to determine the full-scale voltage output (Line Out) of the installed sound card. If using a recommended or legacy card, this value is automatically set to an average of the cards measured by FlexRadio Systems. If using an unsupported card, the following procedure is necessary:

Disconnect the To Line Out cable from the back of the SDR-1000 and leave the other end connected to the line output of the sound card. Turn the software power on and open the Setup Form. Click on the Audio Tab.

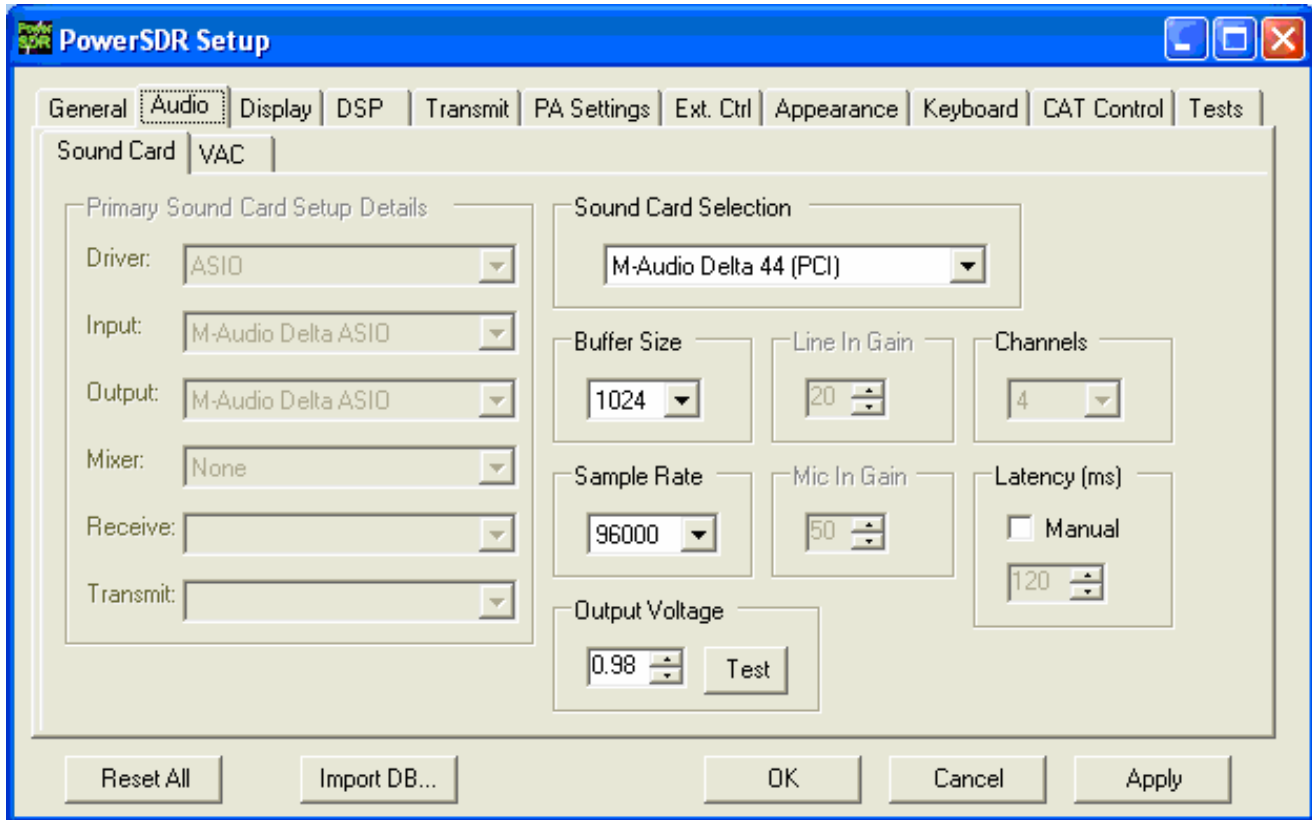


Figure 23: PowerSDR Setup – Output Voltage

Verify that the Line Out cable is disconnected from the radio and then click the **Test** button to continue. The following progress indicator will be shown.

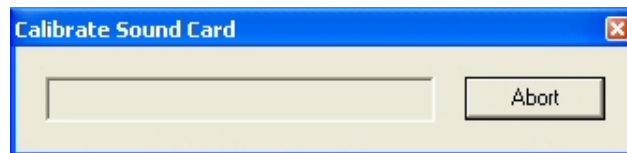


Figure 24: Calibrate Sound Card Progress Indicator

This will send a full-scale (maximum voltage) audio test signal to the sound card.

1. Connect an AC voltmeter between the tip and ground connections on the unconnected end of the cable.
2. Measure and note the AC voltage.
3. Click the **Abort** button to stop the full-scale output.
4. Enter the measured value in the control pictured above.
5. Press the **Apply** button to save the value to the database.
6. Reconnect the line output cable to the SDR-1000.

Note: If no dummy load is available the factory test values must be entered manually on the Setup>PA Settings using the factory test data provided with the amplifier in the **Gain By Band** section of the **PA Settings** tab.

Amplifier Gain Calibration can be performed by the user if a 100W resistive dummy load is available. This is recommended since slight differences in equipment (sound card, power supply, etc.) can give slightly different results. The following procedure automatically determines the RF signal chain gain (SDR-1000 and SDR-100WPA) by band in order to calibrate sound card drive requirements in the PowerSDR software. The resulting gain values, measured in dB, are stored in the PowerSDR database. All gain values are determined at 60W (default) output into a resistive dummy load.

WARNING! A 100W resistive dummy load must be connected to the amplifier output for the Automatic Amplifier Gain Calibration procedure. Failure to do so could cause damage to the amplifier.

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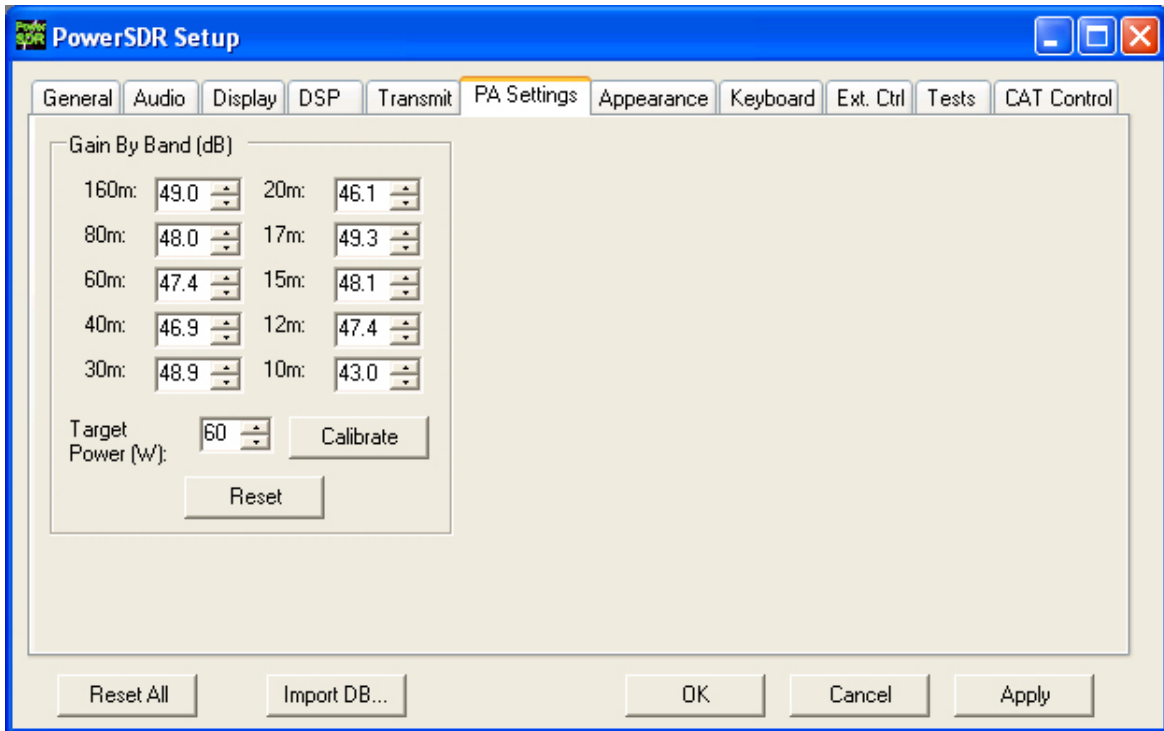


Figure 25: PA Settings Tab

Press the **Calibrate** button in the **Gain by Band** section of the **PA Settings** tab to start the calibration process. Through an iterative process, the software will transmit an estimated drive for 60W (default) output to the dummy load on each band. The power will be measured on the internal directional coupler to determine the gain for a given drive level. The software will apply any necessary correction and save the total system gain value for each band on the form. A progress bar will show the status of the calibration process. **An Abort button is available on the progress bar if needed to stop the calibration process.** If you get an error message during the calibration process, double check the sound card and antenna connections and try again. Press the Apply button when finished to save the values in the database.

Note: Be sure to use the Import Database function when installing future PowerSDR software releases in order to preserve the calibration values.

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7

Front Console

In this chapter and those following many types of software controls will be referred to. The myriad of various bells and whistles can sometimes be a bit overwhelming. The image shown below is a key that will help to introduce the basic controls for those less familiar with windows software.

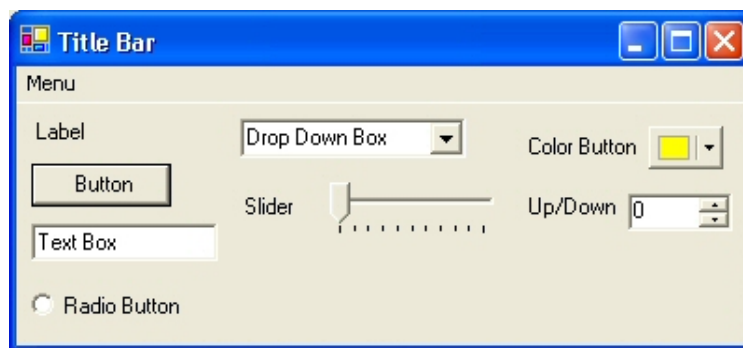


Figure 26: Control Key

- ❑ The **Form** refers to the entire window with the **Title Bar** showing the Form Name.
- ❑ The **Menu** is just under the Title Bar. Menu controls generally open other forms.
- ❑ **Labels** are callouts usually for other controls.
- ❑ **Buttons** can trigger events or act like an On/Off switch.
- ❑ **Text Boxes** allow text to be entered or displayed.
- ❑ **Radio Buttons** allow the user to choose between several options.
- ❑ **Drop Down Boxes** (also called Combo Boxes) enable the ability to offer many options without taking up as much window space as a Radio Button.
- ❑ **Sliders** allow easy modification of a numerical value.
- ❑ **Color Buttons** are used as color selectors. You can pick a generic color (yellow or green) or even make your own using the drop down menu.
- ❑ **Up/Down** controls are similar to a Text Box, but are limited to numeric input. They also have arrows for simple increment/decrement behavior.

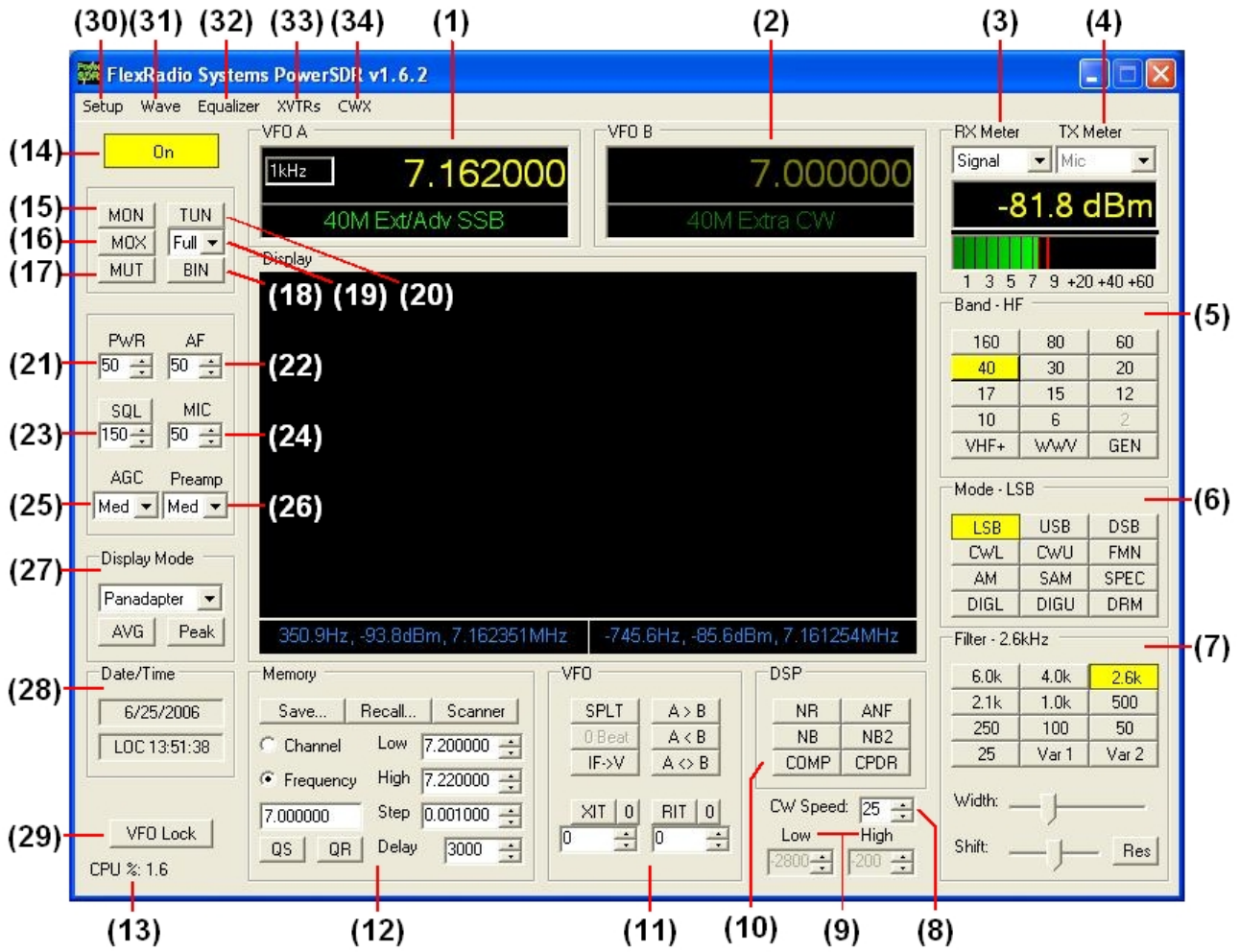


Figure 27: PowerSDR 1.6.2 Front Console

Note: The front console controls the basic facets of the radio: frequency, mode, filters, and display. In addition to these basic features, there are many other controls that are described in detail below.

(1) VFO A



Figure 28: VFOA

VFO A is the main tuning VFO for the radio. It consists of a Tuning Step rate, a frequency and a band description (related to the selected frequency). The Tuning Step displays the current tuning rate for using the mouse wheel (or Ctrl + Up/Down Arrow) to tune the radio. Rotating the mouse wheel away from you will increase the frequency by the step rate per click while rotating the wheel toward you will decrease the frequency. You can change the Tuning Step by clicking the mouse wheel button (or using Ctrl + Left/Right Arrow).

The frequency area is a simple Text Box and may be edited as such (click and drag highlighting, etc). Entering a numeric character (without any mouse interaction) is also a good way to change your frequency quickly. The underlined digit will show visually the digit that will be tuned when hovering over the frequency display. **See the Tuning Methods section in the Operation chapter for more details on how to tune.** Note that you can return to the previous frequency at any time by pressing the 'Escape' key.

The band text information underneath the frequency gives general information about the FCC Amateur bands. If not in an amateur band or a recognized frequency (such as WWV), the band information will display "Out Of Band" and will have a red background to call your attention to it. Note that this information is only a lookup in a database and has no bearing on the current operating mode. This information can be edited in the BandText table using Microsoft Access.

(2) VFO B



Figure 29: VFO B

The operation of VFO B is similar to that of VFO A. However, VFO B is used only in specific instances. When in split mode (SPLT button under VFO), VFO B determines the transmit frequency. Otherwise, it can be viewed as a storage container to copy VFO data to and from VFO A (see the VFO section below).

(3) & (4) Multimeter

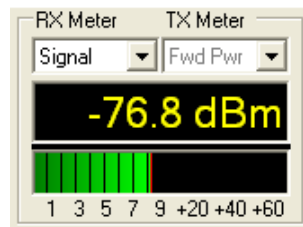


Figure 30: Multimeter

The multimeter functions as another way of feeding information back to the user. The two drop down boxes at the top allow selection of multimeter mode during receive and transmit modes independently.

The text display below the mode selection shows the digital data for whichever mode is active (Signal strength in the screenshot above). The display at the bottom of this section shows the data graphically using bar graph style graphics.

RX Modes

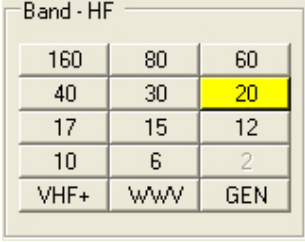
- ❑ **Signal (Signal Level):** Calculates the true RMS power of the current signal within the passband in dBm.
- ❑ **Sig Avg (Signal Average):** Calculates the true RMS power of a time averaged signal within the passband in dBm.
- ❑ **ADC L (Analog To Digital Left):** Calculates the level of Left input from the sound card in dBFS (decibel full scale).
- ❑ **ADC R (Analog To Digital Right):** Calculates the level of Right input from the sound card in dBFS (decibel full scale).
- ❑ **Off:** Used for debugging purposes or to save CPU cycles on slower machines.

TX Modes

- ❑ **Fwd Pow:** Reads out forward power minus reflected power in Watts according to the hardware ADC on the PA if it is installed. If using the 1W configuration, this shows the power output according to the DSP based on 1W full output. This is an average reading. Note that on SSB, the typical male voice peak to average ratio is 14dB. This means that an average meter will only read 4-10W (with a typical voice) when peaking at 100W when you do not have the compressor and/or compander turned enabled. With the ALC we are using, the average power. With these controls, very high average power can be tolerated.
- ❑ **Mic:** Reads modulation power from -20 dB to 3 dB. Ideal operation will peak around 0 dB and will rarely hit if ever hit 3 dB. If it is hitting 3 dB, the ALC is cutting back the power. Adjust the MIC control on the front panel to give more or less modulation.
- ❑ **EQ:** Reads the power following the equalizer. If the equalizer is not enabled, the equalizer power is equal to the Mic power.
- ❑ **Lev:** Reads the power following the leveler. The leveler attempts to level the voice coming from the microphone as the head and mouth change position relative to the microphone element. If the leveler is not enabled, this power is identical to EQ Out.
- ❑ **Lev Gain:** Reads the gain currently being applied by the level in dB.
- ❑ **Comp:** Reads the power after the compressor. If it is not enabled, it reads the same as Lev Out.

- ❑ **CPDR:** Reads the power after the compander. If it is not enabled, it reads the same as Comp Out.
- ❑ **ALC:** Reads the power after the ALC.
- ❑ **ALC Comp:** Reads the gain applied by the ALC algorithm. This number is always 1.0 or less so the gain is always ≤ 0 in dB. The minus sign is implicit.
- ❑ **Ref Pwr (PA Reflected Power):** Reads out reflected power in Watts according to the hardware ADC on the PA. (Only available with the PA)
- ❑ **SWR (PA Standing Wave Ratio):** Reads out the standing wave ratio as calculated by the hardware ADC on the PA. (Only available with the PA with TUN on)
- ❑ **Off:** Used for debugging purposes or to save CPU cycles on slower machines.

(5) Band Selection & Band Stacking Memories



| Band - HF | | |
|-----------|-----|-----|
| 160 | 80 | 60 |
| 40 | 30 | 20 |
| 17 | 15 | 12 |
| 10 | 6 | 2 |
| VHF+ | WWV | GEN |

Figure 31: Band Selection

The Band Selection controls actually perform multiple roles on this radio. The first role is easily seen by just tuning the VFO to a specific frequency. The band indicator will move to the appropriate band (GEN if not in one of the specific bands listed). This is used to quickly identify which band you are in, or when you are stepping over a band edge boundary.

The second role is a feature called Band Stacking Memories. A single memory is defined as a frequency, mode, and filter combination. Each band has several memories associated with it. Clicking on a band button repeatedly will cycle through the available memories repeating from the beginning after the last memory (hence the stacking memories). This is useful to quickly tune to various frequencies within a band. To replace one of the memories with the frequency, mode, and filter of your choice, first press the band button for the band memory you would like to modify. Then change the frequency, mode, and filter to the desired settings (the frequency must be in the band selected). Then simply press the band button again to save the values. Note that some band memory frequencies (such as 60m and WWV) are fixed in software and cannot be changed. The modified memories will be saved to the database upon graceful exit of the console. A crash will prevent changed memories from being saved in order to keep faulty data from making it into the database.

Note that the VHF+ button will swap between the typical HF bands and any configured transverters (via the XVTRs Form – see below). The VHF band buttons work in the same way, but for the frequencies within the transverter frequency range.

(6) Mode Selection



Figure 32: Mode Selection

The Mode Selection controls allow you to change the selected demodulation routine. Changing modes will select last frequency and filter used for that mode. Below is a list of the available modes:

- ❑ **LSB:** Lower Side Band
- ❑ **USB:** Upper Side Band
- ❑ **DSB:** Double Side Band
- ❑ **CWL:** CW Lower Side Band
- ❑ **CWU:** CW Upper Side Band
- ❑ **FMN:** Frequency Modulation (FM) Narrow
- ❑ **AM:** Amplitude Modulation Detection
- ❑ **SAM:** Synchronous (PLL) Amplitude Modulation Detection
- ❑ **SPEC:** Spectrum mode (DC IF, 48kHz bandwidth)
- ❑ **DIGL:** Digital Lower Side Band (Enables VAC if VAC Auto Enable is engaged)
- ❑ **DIGU:** Digital Upper Side Band (Enables VAC if VAC Auto Enable is engaged)
- ❑ **DRM:** Digital Radio Mondiale (requires licensed external demodulator software not available from FlexRadio Systems) (Enables VAC if VAC Auto Enable is engaged)

Hotkeys are available in the Setup Form, Keyboard Tab to cycle through the various modes using the keyboard.

(7) Filter Controls

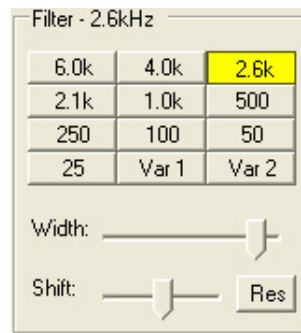


Figure 33: Filter Controls

Clicking on any of the labeled buttons in the top half of this section sets the filter bandwidth based on the selected mode. The filters below 1.0kHz are designed for use with CW modes, but are enabled for all modes. The Variable filters (**Var 1** and **Var 2**) allow custom filters to be designed using the **Low** and **High** controls described below. These filters are saved and are independent for each mode (i.e. you can save a different **Var 1** filter for **LSB** mode than for **AM** mode). The panadapter display mode is good for visualizing changes to variable filter controls. Hotkeys are available in the Keyboard Tab of the Setup Form to change the filters easily.

The Width and Shift controls give an easy way to manipulate the selected filter. See the description below of the IF->V button for info on how to adjust these after using them to tune a signal.

- ❑ **Filter Width:** Widens the filter as the slider is moved right, and narrows the filter as it is moved left.
- ❑ **Filter Shift:** Shifts the selected filter passband up or down from its normal center frequency. This can help to eliminate interference caused by signals in close proximity of the received signal. If the display is in Panadapter display mode, you can see the Filter Shift effects directly on the display.
- ❑ **Filter Shift Reset:** Returns the Filter Shift control to the default middle position and restores the filter to the original values (i.e. before the Filter Shift was used).

(8) CW Speed (WPM)

This controls the CW speed when using the internal keyer in Iambic mode.

(9) Variable Filter Controls

These adjust the filter low and high cuts.

- ❑ **Low:** Selects the low cutoff frequency for the filter. The value is the plus or minus offset from the center frequency as shown in the VFO display. In LSB and CWL, this frequency can be negative.

- ❑ **High:** Selects the high cutoff frequency for the filter. Note that in lower side band modes, this value can be negative.

(10) DSP Controls

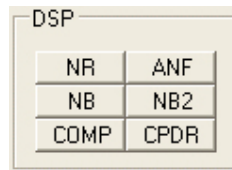


Figure 34: DSP Controls

These controls enable the DSP functions, which are described in detail in the Setup Forms DSP and Transmit Tabs.

- ❑ **NR (DSP Noise Reduction):** Activates the DSP Noise Reduction algorithm. *Note: Not available when running in Binaural (BIN) mode unless BLMS is checked on the setup form.*
- ❑ **ANF (Automatic Notch Filter):** Activates the Automatic Notch Filter algorithm. *Note: Not available when running in Binaural (BIN) mode unless BLMS is checked on the setup form.*
- ❑ **NB (Impulse Noise Blanker):** Activates the Noise Blanker algorithm.
- ❑ **NB2 (Mean Noise Blanker):** Activates the Mean Noise Blanker algorithm.
- ❑ **COMP (Transmit Compression) :** Activates the Transmit Feed Forward Compression algorithm.
- ❑ **CPDR (Compander) :** Activates the Compander algorithm.

(11) VFO Controls

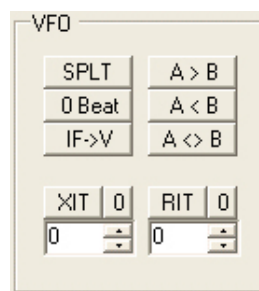


Figure 35: VFO Controls

- ❑ **SPLT (Split):** Enables Split operation using VFO B for the transmit frequency. The frequency text in VFO B will turn red to indicate that Split mode is active.

- ❑ **0 Beat:** Takes the peak signal and sets it to the center of the filter passband. An exception is made while in CW mode if the CW Pitch is within the passband. In this case, the signal is tuned to the CW Pitch. Note: Requires display to be in average (AVG) mode.
- ❑ **IF->V:** Takes any offset based on Filter Shift and the Filter Width and shifts it back to baseband. Useful when chasing a signal with the Filter Shift control.
- ❑ **A > B:** Transfers the contents of VFO **A** to VFO **B** (frequency, mode, and filter).
- ❑ **A < B:** Transfers the contents of VFO **B** to VFO **A** (frequency, mode, and filter).
- ❑ **A <> B:** Swaps the contents of VFO **A** and **B** (frequency, mode, and filter).
- ❑ **XIT (Transmit Incremental Tuning):** XIT may be used as a quick way to operate split at a specific offset. Clicking the push button enables this function. When enabled, the transmit frequency is increased from the VFO **A** frequency by the amount shown in Hz while leaving the receive frequency intact. Note that XIT cannot be used in combination with the SPLT function. Press the 0 button next to the XIT button to clear the XIT control to 0.
- ❑ **RIT (Receive Incremental Tuning):** Clicking the push button enables this function. When enabled, the receive frequency from the VFO **A** frequency by the amount shown in Hz while leaving the transmit frequency intact. Press the 0 button next to the RIT button to clear the RIT control to 0.

(12) Memory and Scanner Controls

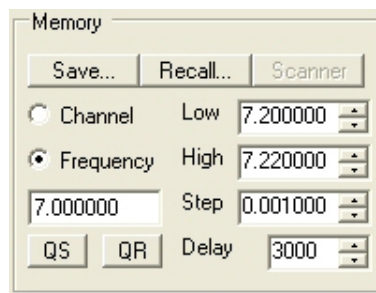


Figure 36: Memory and Scanner Controls

The Memory section allows storing and retrieving information such as frequency, mode, filter and various other settings. The scanner function allows scanning by Channel (stored memories) or between Frequencies as set in the Low, High, and Step settings. The delay specifies how long the receiver will stay at the current setting (channel or frequency modes).

- **Save...:** Opens the Save Memory Channel form as shown in the figure below.

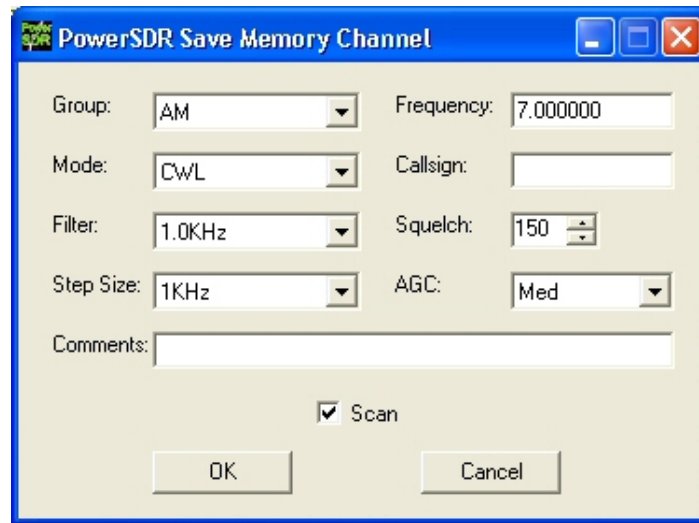


Figure 37: Save Memory Channel Form

The current **Mode**, **Filter**, **Step Size**, **Frequency**, **Squelch** and **AGC** settings are automatically transferred from the console. The **Group** Drop Down Box allows a further level of characterization of the type of entry. In the future this Group setting list will be customizable. The **Callsign** and **Comments** fields are free form and the user can enter details as desired. A Scan checkbox is provided and can be checked if the user wishes this entry to be scanned when performing a Channel scan (see below). Clicking the **OK** button will save the information shown above into the memory database before closing the form. Clicking **Cancel** simply closes the form (the data is not saved).

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- ❑ **Recall...**: Presents the user with a memory form with data from all previously stored memory locations (shown in the figure below).

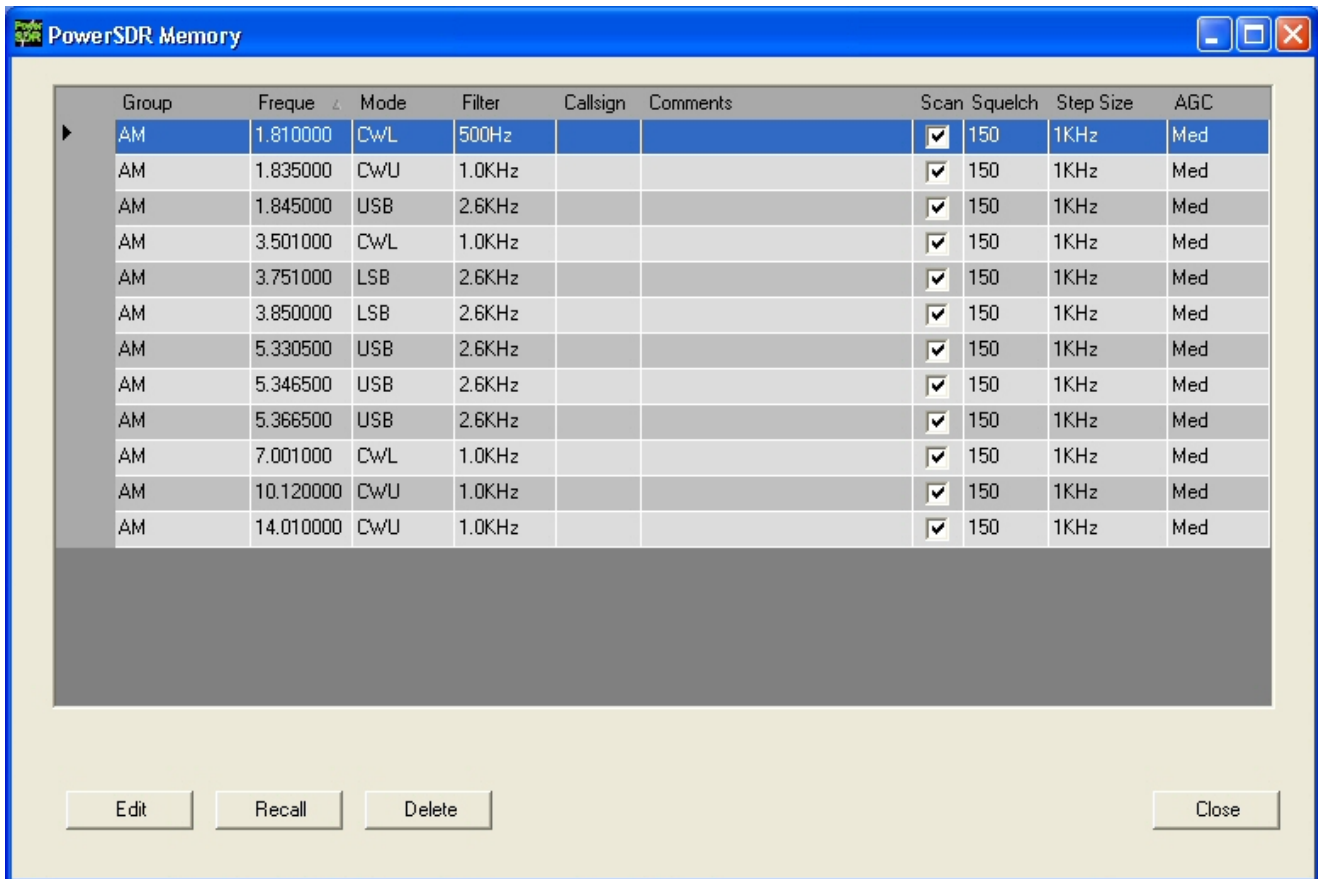


Figure 38: Memory Form

The data grid displays each memory that has been saved to the database. Clicking in the left hand margin will allow a particular memory to be selected. Clicking on the column titles will sort the data using the information in that row. Clicking the **Edit** button allows manual changes to be made to the saved memories. Make sure to click the button again when finished editing to prevent unintended changes from getting saved to the database. The **Recall** button will send the data in the memory to the front panel (i.e. restores a memory). You can also double-click on a row to accomplish this, though this method is less consistent. Clicking the **Delete** button will allow a memory to be removed from the database. A prompt will be shown to prevent unintended loss of memories.

- ❑ **Scanner**: Starts the scanning function in Channel or Frequency mode as determined by the controls described below. Currently the scanning function will only make one pass through the channels or frequencies.

- ❑ **Channel:** Selects the Channel scanning function. With this button active and the scanner is run, it will run through the list of memories from top to bottom. Note that you can sort the memory form prior to running the scanner to change the ordering.
- ❑ **Frequency:** Selects the Frequency scanning function. With this button active and the scanner is run, it will run through the frequencies from the selected Low frequency to the High frequency using the Step size as selected using the controls described below.
- ❑ **Low:** Selects the start frequency for the Frequency scanner option.
- ❑ **High:** Selects the stop frequency for the Frequency scanner option.
- ❑ **Step:** Selects the step size rate for the Frequency scanner option.
- ❑ **Delay:** Sets the delay in milliseconds between changing the memory or frequency (depending on mode) during the scanning operation.
- ❑ **QS:** Quick Save allows a frequency, mode, and filter to be quickly saved. The frequency will be shown in the box above the buttons.
- ❑ **QR:** Quick Restore restores the data saved by the most recent Quick Save (frequency, mode, and filter).

(13) CPU %

This displays the total CPU load as seen in the Task Manager under the Performance Tab. Note that running other applications will cause the CPU load to increase. If your CPU load is peaking at 100%, audio and possibly video artifacts will become noticeable. Closing additional applications and turning down some of the functions will allow systems that meet the requirements on the website to stay below the 100% mark.

(14) Power (Standby/On)

Switches the hardware and software between Standby and On modes. Note that the software Power button in the off (standby) state places the hardware in a lower power standby state (around 500-600mA), but does not turn off 12VDC power to the hardware. The front panel switch on the SDR-1000 enclosure removes power to the hardware. *It is recommended that the main power supply always be turned off when the radio is not in use.* The color of the POWER button indicates the current state:

- ❑ **Yellow** (default) – On/Receive
- ❑ **Red** – On/Transmit
- ❑ **Gray** – Off

If the SDR-1000 is connected properly and powered on, you will hear relays switch (click) as you cycle the software Power button.

Note: Please see the Operation chapter for info on best practices for Power On/Off sequences.

(15) MON (Monitor)

Allows the transmitted audio to be monitored through the receive speakers. The MON function is not available in AM, SAM, or FM modes as those modes are transmitted at an Intermediate Frequency (IF) of 11.025kHz. When operating voice modes, the MON feature will allow you to hear the effects of MIC gain, TX equalization, and compression being applied, and to adjust them in real time. When using the Delta 44 or the FireBox sound card, the AF control can be used to adjust the monitor volume.

Note: Due to audio buffering, you may experience a delay between spoken audio and the monitored audio. This latency can be reduced by using smaller sample buffers (see Setup Form, Audio and DSP Tabs).

(16) MOX (Manually Operated Transmit)

This button is active whenever the radio is in transmit mode and is used primarily for voice operation. MOX will not generate a CW carrier. To generate a carrier for tuning, refer to the tune (TUN) button description.

If the radio ever seems like it is stuck in transmit mode, try disabling the Push-To-Talk (PTT) function using the control on the Setup Form in the General Tab. The Disable PTT setting is necessary to use the PowerSDR software without the SDR-1000 attached to the parallel port (choose the Demo/No Hardware option on the Setup Form -> General Tab in this case)..

(17) MUT (Mute)

This button Mutes the speaker audio. The receiver may also be muted by pressing the * (asterisk) key on the keyboard. Also, refer to the Auto Mute option on the Setup Form -> General Tab.

(18) BIN (Binaural Mode)

The Binaural Receive function is a special feature of the SDR-1000 software implementation. It generates a pleasing effect as the two phased channels (I and Q) are mapped to the left and right audio channels. The phasing of the demodulated and filtered audio signal within the passband gives a stereo effect to the received signal. When tuning across CW signals, they will seem to move in “space” as you tune the radio. Many experienced operators feel that binaural receive gives them a competitive advantage under contest conditions where the effect allows them to more easily pick signals out of a pile up. Wearing headphones increases the effect of the stereo separation.

(19) ATU Modes

The LDG Z-100 ATU has three tuning modes: Bypass, Memory, and Full. Selecting the Bypass mode will instantly bypass the ATU. Selecting the Memory or Full modes requires the TUN button to be pressed in order to perform the tuning. Note that if VFO A is tuned outside of the band in which the radio was tuned, the ATU is automatically bypassed to protect the unit.

(20) TUN (Tune)

Without the ATU, this transmits a continuous (CW) carrier at the level set on the PWR (Power) control while the TUN mode is activated. It sets the power to the saved tune level (default 10W – 10% without PA) and outputs a tone at the CW Pitch. Any changes to the PWR control while the function is active are saved when the TUN button is turned off. This feature is used to simplify the antenna tuning process for proper load matching.

WARNING! DO NOT OPERATE THE 100W AMPLIFIER ABOVE 25W IN CONTINUOUS CARRIER MODES (E.G. CW, AM, FM, AND DIGITAL) FOR LONGER THAN 15 SECONDS. HIGHER LEVELS CAN CAUSE DAMAGE TO THE AMPLIFIER. THE AMPLIFIER IS RATED AT 100W PEP FOR SSB DUTY CYCLES.

With the ATU, this control combines with the Combo Box below it to perform the ATU tuning function. When the ATU is in Bypass mode, the TUN function operates just as if the ATU were not present. Note that the TUN button will stay selected (yellow default) after a tune cycle in Memory or Full mode. The button will become deselected (not colored) when it is bypassed whether it be due to tuning to another band or manually selecting Bypass from the ATU Mode control.

(21) PWR (Transmitter Power Output/Tune Power)

Adjusts the percentage of maximum power that will be available in transmit mode. A typical setting for the 1W SDR-1000 radio might be 50% (roughly 500mW), while a radio configured with the 100W PA may be operated at full power (100W) into a proper 50 ohm load. The power control may be adjusted while in either receive or transmit mode. *Note that the control doubles in function as the power level setting for the TUN (Tune) button described above. The Tune power may be adjusted while the TUN button is activated or by using the control on the Setup Form, Transmit Tab.* While great care is taken to ensure that this value is accurate and that selecting a PWR value will give approximately that amount when using the 100W PA, there are variances in the filter components and transmitter characteristics that make it difficult for this to be exactly right over the whole range (1-100). When using the 100W PA, these values are most likely to be accurate on the higher end. For example, it is not unusual if you select 10 on the PWR control, but only see something like 5W out. This was a design decision made to favor the higher power users. This will likely be improved in the future to a more accurate algorithm.

(22) AF (Audio Frequency Gain)

This control sets the monitor audio gain. The AF gain may also be adjusted by pressing the **+** (plus) and **-** (minus) keys on the numeric keypad. For best performance, the external speaker volume control should be set to the high end of the scale so that the AF control can be set to a lower value. If the AF control is set too high, it may cause feedback in the receiver.

(23) SQL (Squelch)

The push button enables the squelch function while the value below sets the threshold. The threshold is read in negative dB (i.e. a value of 110 would mean to set the threshold at -110dB). Squelch can be very useful to remove all noise from CW signals, especially in narrow filter settings.

(24) MIC (Microphone Gain)

Adjusts the software Microphone gain. This is a simple multiplier applied to the input samples when in transmit modes. This control is helpful for sound cards without mic preamps such as the Delta 44. Note that increasing the MIC control will also raise the amount of noise in the signal. A hardware preamp will give the best performance for amplifying microphone signals. Having said that, the software gain works very well in many setups. Voice modes are typically optimized when the peak reading on the TX Meter set to Mic mode reads just below 0dB.

(25) AGC (Automatic Gain Control)

Sets the receiver Automatic Gain Control setting to one of the following: Fixd (Fixed), Long, Slow, Med (Medium), Fast, or Custom. Custom mode uses the controls on the Setup Form -> DSP -> AGC/ALC Tab. The following table gives the control settings. For custom, set AGC/ALC tab in the DSP setup.

Table 7: AGC Setting Details

| Setting | Attack | Decay | Hang | Fast Hang |
|---------|--------|--------|-------|-----------|
| Fast | 2ms | 100ms | 100ms | 100ms |
| Med | 2ms | 250ms | 250ms | 100ms |
| Slow | 2ms | 500ms | 500ms | 100ms |
| Long | 2ms | 2000ms | 750ms | 100ms |

(26) Preamp

This control replaces the previous Gain and ATT controls and has four settings: Off, Low, Med (Medium), and High. See the table below for details on each setting. The typical setting for HF frequencies (and default for the software) is Med. Higher settings should be used for weak signal work while lower settings are ideal for large signal activity.

Note: When using the 2m transverter, **Low** and **High** are the only options available.

Table 8: Preamp Setting Details

| With RFE | | | | W/O RFE | |
|----------|------------|-------------|---------------|---------|-----------|
| Setting | ATT (10dB) | Gain (26dB) | Net Gain (dB) | Setting | Gain (dB) |
| Off | On | Off | -10 | Off | N/A |
| Low | Off | Off | 0 | Low | N/A |
| Med | On | On | 16 | Med | 26 |
| High | Off | On | 26 | High | 40 |

(27) Display Mode Controls

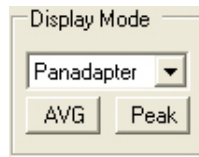


Figure 39: Display Mode Controls

Changes the way the data is represented on the main display. The display modes are detailed below with a snapshot of each mode. The Frames Per Second (FPS) rate of the main display can be modified using the Display Tab on the Setup Form.

- ❑ **AVG (Average):** Allows the visual data to be averaged over time (set in Setup -> Display). This will smooth out fast changing signals and is a good way to identify where real signals are located. This function must be active for the 0 Beat function to be enabled.
- ❑ **Peak:** Holds the peak value for each frequency in the display.

Display Mode Descriptions

Note 1: All displays show a 5MHz, -70dBm signal from an HP 8640B signal generator.

Note 2: The actual software display is crisper than the compressed images shown below.

There are four frequency domain display modes (Spectrum, Panadapter, Histogram, and Waterfall) and two time domain display modes (Scope and Phase). The colors of everything on the display (text, data line, background, etc) are completely customizable using the Appearance Tab on the Setup Form.

Spectrum

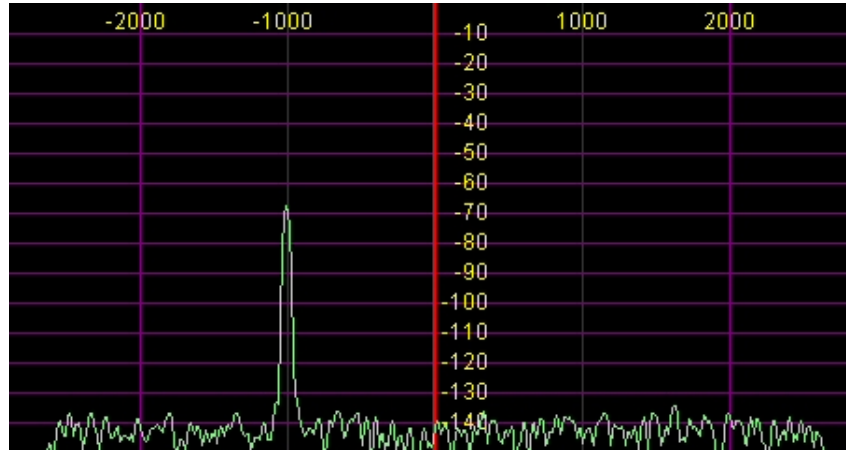


Figure 40: Spectrum Mode Display

This mode is the classical spectral view of the frequency with the ends of the display determined by the bandwidth of the filter. Rather than only using half the display window when in lower or upper sideband, we expand the display moving the 0Hz line to the left or right margin (DSB mode with a 2.6kHz filter shown).

Panadapter (Panoramic Adapter)

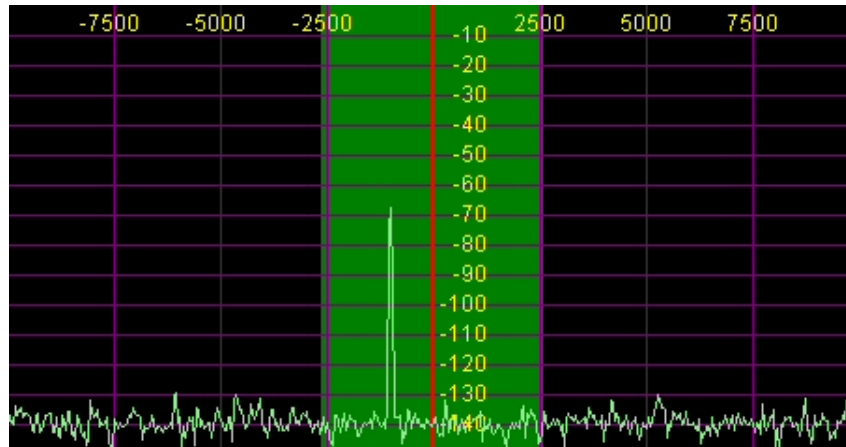


Figure 41 Panadapter Display Mode

The Panadapter mode is similar to the spectrum mode with a few differences. First, the display is a function of the sampling rate (20, 40, or 80kHz) no matter what filter bandwidth is selected. Second, the selected filter is shown as an overlay to help the user visualize the filter. The color of the overlay can be changed using the Appearance Tab of the Setup Form. This mode is useful because you hear only see the signals within the

passband but you can see in real time all signals within the displayed spectrum. This gives a much more complete picture of the surrounding areas of the band, especially when there is abundant signal activity (e.g. contest and DX situations). Also, while in this mode, both the signals and the filters may be dragged and dropped as desired.

Histogram

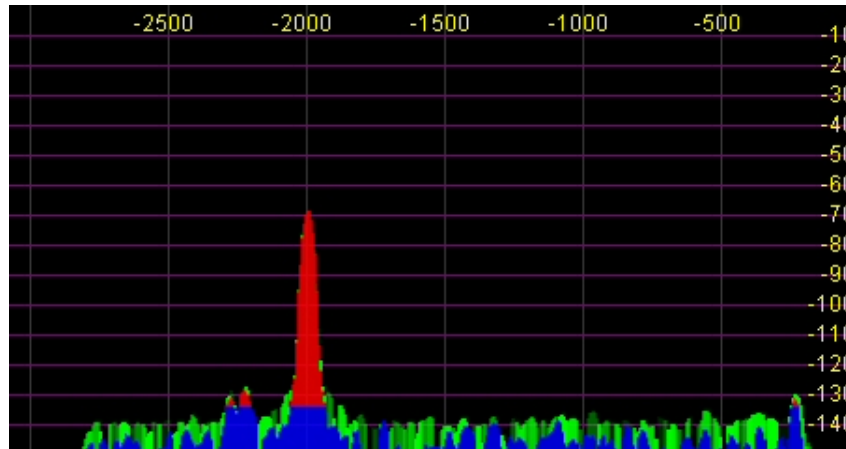


Figure 42: Histogram Display Mode

The histogram display mode is similar to the Spectrum mode, but instead of a single color data line, additional colored data is used. Blue signals are real-time (current) signals that are below a signal threshold (roughly below the average plus a small margin). The red signals are real-time (current) signals that are above that same threshold. The green signals are previous peaks on that same frequency that will fade as time goes by (a type of history, hence the name).

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Waterfall

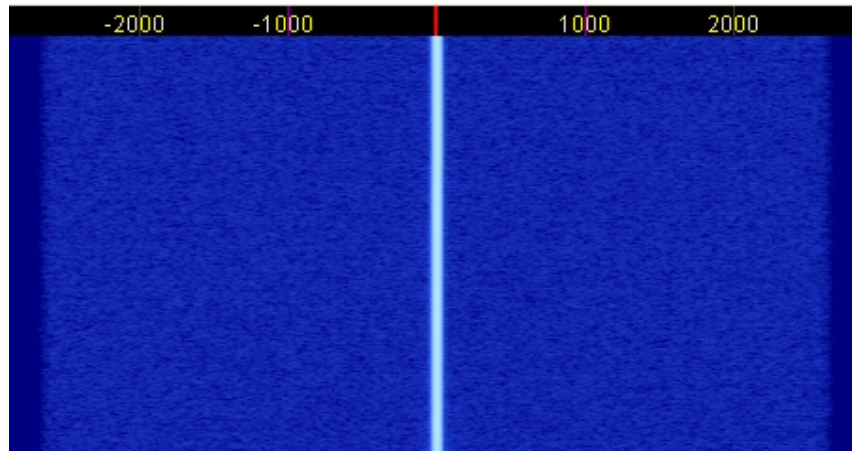


Figure 43: Waterfall Display Mode

The waterfall display mode gives a scrolling view of activity within the passband. This makes tracking narrow band signals much easier and can even allow visualization of CW signals at slower speeds (longer line is a dash, short line is a dot, no line is a pause). The entire waterfall display mode details are available for customization using the Display Tab on the Setup Form. This allows custom setting of the dynamic range and coloring for the display. See the Display Tab section in the Setup Form chapter below for more details.

Scope

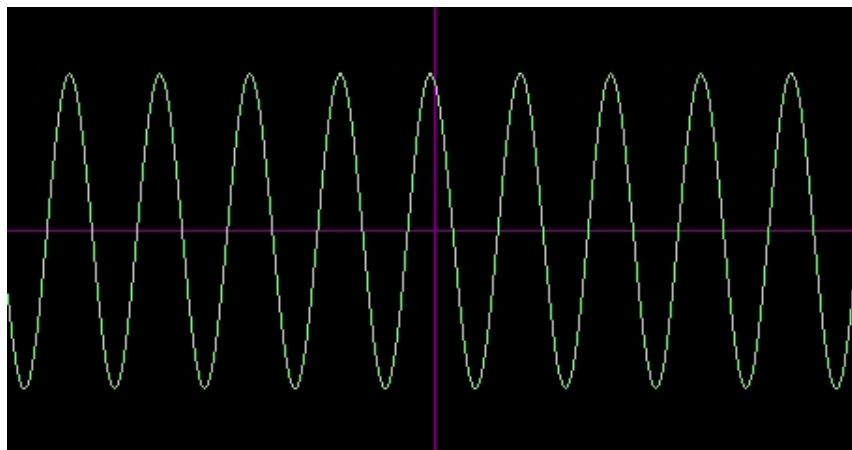


Figure 44: Scope Display Mode

The scope mode displays the time domain input data from a single channel. This enables a traditional waveform view of the input data. Shown is effectively a 1kHz tone (tuned to 7.001MHz with the signal at 7.0MHz). This mode is particularly useful in transmit mode to monitor your transmitted wave form. Use the Setup Form -> Display -> Scope Time control to adjust the frequency of the display.

Phase

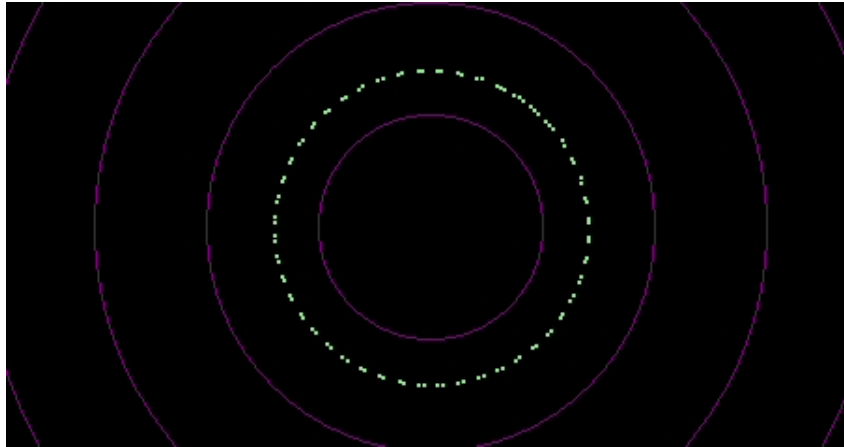


Figure 45: Phase Display Mode

The phase display maps the filtered I and Q (Left and Right) channels to the X and Y coordinate planes. This is useful for making sure the two channels are 90 degrees out of phase as they should be. There is also a **Phase2** display mode that uses the unfiltered data directly from the sound card Line Input with the same mapping. When a continuous carrier signal is received, the unfiltered data in Phase2 display mode should produce as near to a perfect circle as possible. If the circle distorts into an oval or a straight line, the input phase balance is off which would indicate a connection or hardware problem.

Off

The off mode is included mainly for debugging purposes, but can be used with slower systems to decrease the CPU load to more reasonable levels.

Cursor and Peak Position

There are two Text Boxes side by side under the display that are used to communicate cursor (left) and peak signal (right) information back to the user. The text is in the following format: [Offset from VFO], [Signal level], [frequency]. For example, in the screenshot shown, the signal is 1973.7Hz offset from the VFO frequency (7.998MHz). The signal level is -72.7dBm, and the signal is at 7.999974MHz (8.0MHz actually). Note that these values are fairly low resolution (hence the 26Hz error) due to the discrete nature of the pixel display.

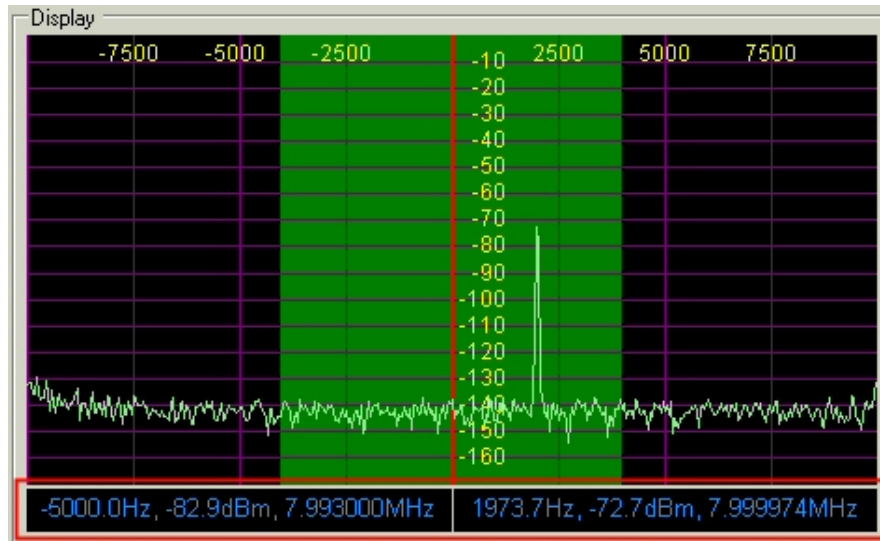


Figure 46: Display Position Features

In spectrum modes, you can click the right mouse button to bring up yellow crosshairs that span the width and height of the display. The crosshairs, along with the AVG control, allow easy measurement of signals on the display. Another feature of the yellow crosshairs is click tuning. Clicking the left mouse button with the yellow crosshairs tunes the VFO to the frequency indicated in the Peak Position textbox. This is an excellent way to tune CW signals as it will zero beat the tone to the CW pitch as selected in the Setup Form, DSP Tab.

(28) Date/Time Display

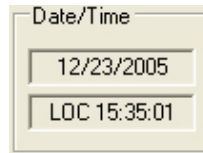


Figure 47: Date/Time Display

The date and time display can be especially helpful when taking screenshots, but can also be a quick reference to UTC time for those of us who are “time zone challenged.” Click inside the Date or Time area to switch between Local (LOC), UTC, and Off modes.

(29) VFO Lock

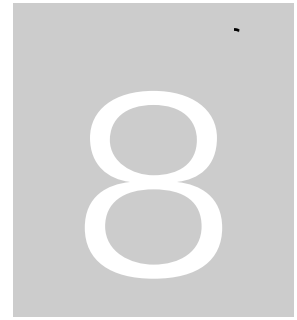
Keeps the frequency from being changed inadvertently. This is a handy feature to use while in a QSO to keep from accidentally losing the frequency due to clicking in the wrong area or hitting the wrong key on the keyboard.

(30) Setup Form

The Setup Form contains numerous controls for everything from the hardware configuration to transmit settings. Please refer to the next chapter for more detailed information.

(31) – (35) Operating Forms

Each of these items open forms, which are used while operating. Please refer to the Operating Forms chapter below for more detailed information on each one.



Setup Form

The Setup Form contains a vast assortment of controls from hardware Setup to detailed DSP options. A few of these controls are available on all Tabs of the Form. Descriptions of these controls will be followed by control descriptions for each individual Tab.

- ❑ **Reset All:** Resets all of the controls on all tabs to their default values. An exception is made for the hardware configuration controls. These will not be reset. A warning will pop up to ensure that accidentally pressing this button does not cause loss of settings.
- ❑ **Import DB (Database):** Allows a backwards-compatible database to be imported. The database saves all of the radio options and current state. To import another database, click this button and then browse to the directory of the database to import. If you have used the default directory during installation as recommended, the database file will be in C:\Program Files\FlexRadio Systems\PowerSDR x.y.z\. The database file is called PowerSDR.mdb. Double click this file and it will attempt to import all the settings. A confirmation message will let you know if the import was successful.
- ❑ **OK:** Saves the current values to the database and closes the form.
- ❑ **Cancel:** Reloads the values from the database into the setup form and closes the form. This button can be used to reverse unintended changes to the Setup controls.
- ❑ **Apply:** Saves the current values to the database immediately.

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General Tab

Hardware Config Sub-Tab

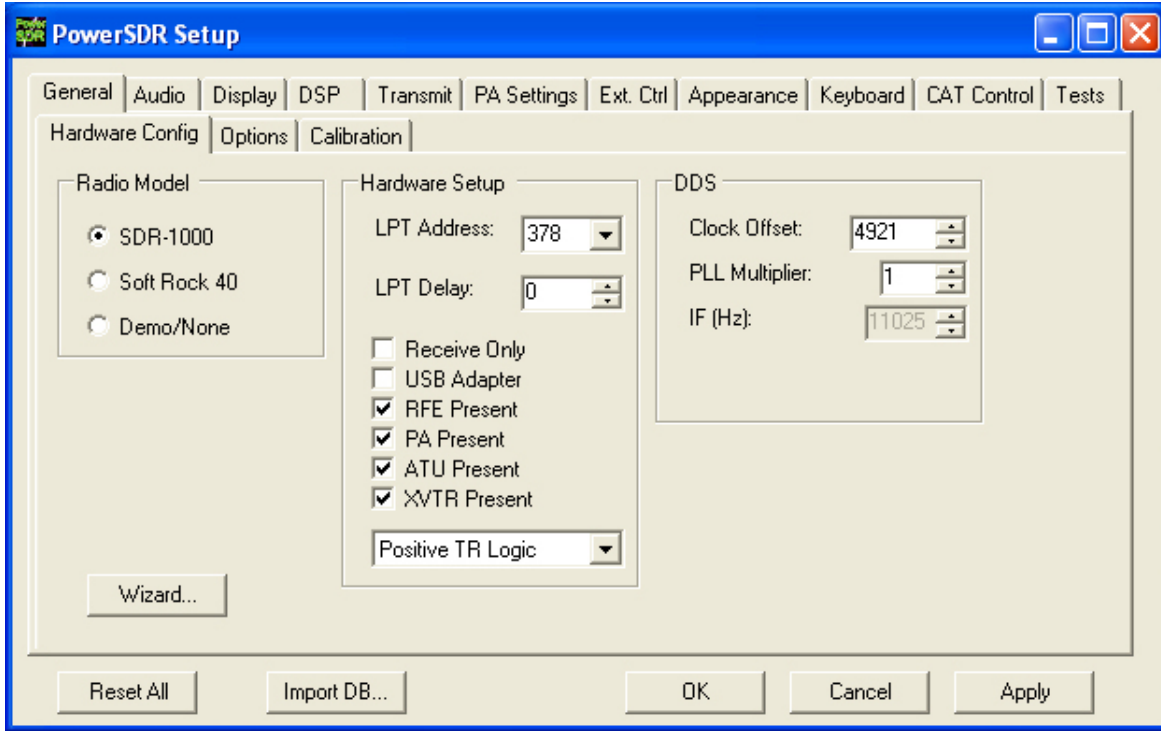


Figure 48: Setup Form - General Tab, Hardware Config Sub-Tab

Wizard...: Press this button to repeat the wizard that runs the first time you run the software. This is a handy way to update your configuration if your hardware has changed (i.e. added the RFE, PA, changed sound card, etc.).

Radio Model

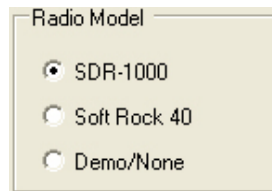


Figure 49: Radio Model Selection

Use this selection to choose the hardware (if any) that is connected to the computer running the PowerSDR software.

Hardware Setup

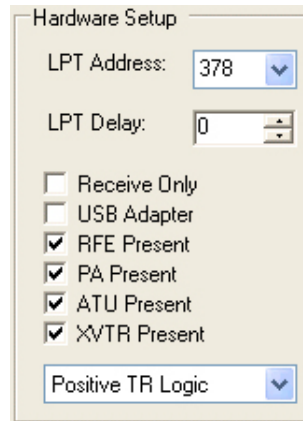


Figure 50: Hardware Setup

The Hardware Setup section allows the user to configure the software to match the SDR-1000 hardware configuration.

- ❑ **LPT Address:** The memory location of the parallel port used to connect the SDR-1000. This is nearly always 378 (LPT1) unless using an add-on parallel port card. If using an add-on card, note the memory address in the Device Manager and enter the value manually if it is not one of the options in the drop down box. *Warning: Changing this value without verifying your LPT address could keep the software from correctly communicating with the radio.*
- ❑ **LPT Delay:** Allows a delay to be added in milliseconds for longer/more capacitive cables. The default value of '0' should work fine for most six-foot cables. We only recommend changing this value if a long cable is necessary and control signals are sporadic. Increasing the value will add additional delay to each control signal sent. Note that this will slow each signal sent to the radio by the time selected. A delay value of 20ms could cause the radio to take nearly half a second (20ms x 24 signals = 480ms) to tune each time the frequency is changed. To use an old adage, a little bit goes a long way.
- ❑ **Receive Only:** Used to force the console to be used only in receive mode (disables transmit). This is useful for those with Receive Only hardware or simply those that want to ensure that they do not accidentally go into transmit mode.
- ❑ **USB Adapter:** Should be checked if using the USB to Parallel adapter available from FlexRadio Systems. An error message will be shown if the adapter is not plugged in and installed before attempting to use this option. *Warning: Once installed and running, do not unplug the USB adapter from the PC without first closing the software or unchecking this option. Doing so could cause the system to crash.*
- ❑ **RFE Present:** Should be checked if the RF Expansion board is installed in the stack. The RFE is installed in all current (2005) SDR-1000 packages. Note that if the RFE is not installed (and the box is not checked), none of the controls below it will be visible.

- ❑ **PA Present:** Should be checked if the FlexRadio Systems 100W PA is installed. Note that if the PA is not installed (i.e. the box is not checked), the ATU option will not be visible.
- ❑ **ATU Present:** Should be checked if the Automatic Antenna Tuning Unit for the SDR-1000 is installed.
- ❑ **XVTR Present:** Should be checked if the DEMI144-28FRS 2m transverter is installed. Make sure to select the appropriate TR logic using the combo box below. If you purchased the DEMI 2m XVTR (DEMI144-28FRS) before May 2005, it should be configured for negative TR logic. Otherwise (more recent orders) you should use positive TR logic.

DDS

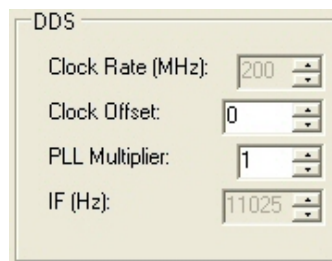


Figure 51: DDS

DDS Stands for Direct Digital Synthesis. The DDS chip in the SDR-1000 produces an analog sine wave at up to micro Hertz resolution. The DDS is used as a local oscillator to tune the radio.

- ❑ **Clock Rate (MHz):** Shows the clock rate of the DDS. This is currently locked as the clock rate (200MHz) is tied to the current hardware (Analog Devices AD9854).
- ❑ **Clock Offset:** Allows software corrections to be made due to the DDS clock oscillator not running at exactly 200MHz. The specification for the oscillator is 200MHz +/- 20kHz (100ppm). The correction routine is described above in Software Installation under PowerSDR Calibration (frequency calibration). Typically this value will be set automatically by the frequency calibration routine.
- ❑ **PLL Multiplier:** Unless the external clock option is installed, this value is always '1'. If using the external clock option, this value should represent the multiplier needed to get the external source up to the clock rate (200MHz). Therefore, if the source were 10MHz, the PLL Multiplier would be 20 (10MHz x 20 = 200MHz). ***Using a PLL Multiplier/external signal that equates to a clock value greater than 200MHz can cause heat damage to the DDS chip.***
- ❑ **IF (Hz):** Displays the Intermediate Frequency used in the software to avoid low frequency noise. This is currently locked to 11.025kHz.

Options Sub-Tab

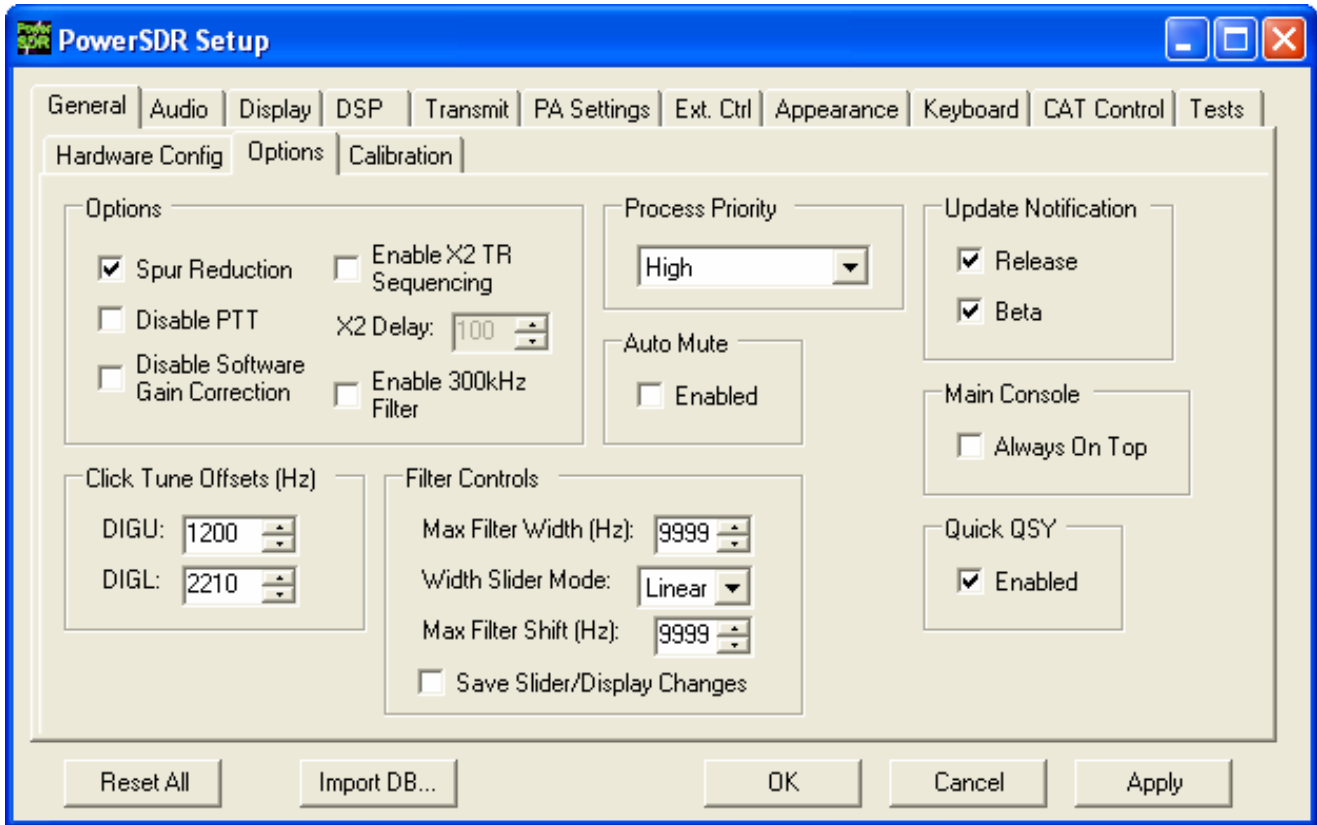


Figure 52: Setup Form - General Tab, Options Sub-Tab

Options

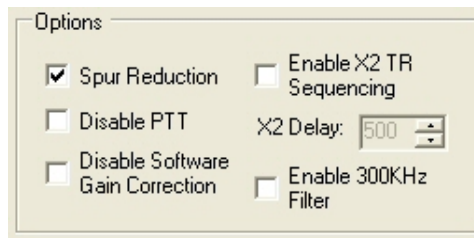


Figure 53: Options

- ❑ **Spur Reduction:** Adds the use of a software oscillator to avoid DDS frequencies known to have higher phase truncation related spurious response. In short, rather than tuning each frequency using the DDS (this is what happens when this option is turned off), the software tunes the DDS in 3kHz steps and does fine-tuning in software. Tuning in 3kHz steps also has the advantage of having to send

fewer command signals to the hardware. For a complete description of the spur reduction algorithm used, see [A Technical Tutorial On Digital Signal Synthesis](#) available from Analog Devices.

- ❑ **Disable PTT:** Disables the ability to use external Push-To-Talk lines to key the radio. It is necessary to use this function if evaluating the software without the parallel port terminated (i.e. without a radio) to prevent the software from engaging the PTT.
- ❑ **Disable Software Gain Correction:** Disables the compensation usually used to correct for the 0/26dB gain options as well as the 10dB attenuator on the RFE. The signal is then reflecting the signal to the sound card rather than what is at the antenna. This function is not recommended for normal operation because it invalidates the signal calibration of the spectrum display and multimeter.
- ❑ **Enable X2 TR Sequencing:** Enables X2 pin-7 transmitter sequencing for an external linear amplifier, relay, or transverter. The sequence for RX->TX is: MOX -> X2-7 goes high -> Delay -> SDR-1000 transmit relay gets set. The sequence for TX->RX is: MOX -> SDR-1000 transmit relay is reset -> Delay -> X2-7 goes low. Note that the delay is set in the following control. Check the box to enable this option and to set the delay.
- ❑ **X2 Delay:** Sets the delay in milliseconds when using the X2 sequencing. If no delay is desired, simply enter 0. Note that when using the X2 TR Sequencing option, transitions to and from transmit mode will both be delayed by the entered value.
- ❑ **Enable 300kHz Filter:** Engages the custom filter bank on the RFE on low frequencies. This is only useful if the custom filter bank is populated with a low frequency filter (not typical).

Process Priority

Sets the process priority for the PowerSDR process. Some users have reported that setting the priority higher than Normal can allow slower systems to perform more reliably and with smoother audio. While FlexRadio Systems recommends using the Normal setting, if you are experiencing audio glitches or are using a slower machine, moving this control up to Above Normal or High might improve the performance of the software. Note that FlexRadio Systems does not recommend using the Real Time setting as this could cause timing problems with the operating system.

Update Notification

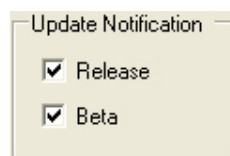


Figure 54: Update Notification Options

Use these checkboxes to enable a pop-up notification when new versions come out. Check the Beta box to receive update messages when new Beta versions are available. Similarly, check the Release box to receive update messages when official release versions become available.

Click Tune Offsets (Hz)

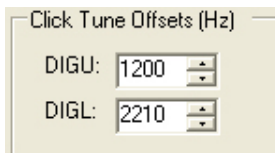


Figure 55: Tuning Options

- ❑ **DIGU:** Sets the offset in Hz to use when click tuning in DIGU mode. Defaulted to 1200 for SSTV.
- ❑ **DIGL:** Sets the offset in Hz to use when click tuning in DIGL mode. Defaulted to 2210 for RTTY.

Auto Mute

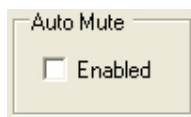


Figure 56: Auto Mute

Click the box to enable the Auto Mute option. When enabled, pulling pin X2-12 (see the chapter on Plug Pinouts) to ground will automatically activate the MUT (Mute) function to mute the receiver.

CAUTION: This is an audio only mute. It does not protect the RF circuitry and care should be taken not to transmit high power into the SDR-1000 antenna input as this could damage the unit. To use separate receivers and transmitters external switching for the antenna is still required.

Quick QSY

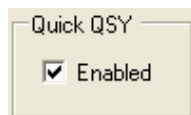


Figure 57: Quick QSY

This option enables the user to enable/disable the **Quick QSY** feature. When enabled, this feature allows the user to quickly enter a frequency in MHz and hit [Enter] to jump to that frequency. With this option disabled, using a mouse to tune or to click inside the VFO is the quickest way to change frequency. This feature is normally enabled by default, but can be disabled to prevent accidentally moving the VFO frequency due to accidental key presses.

Main Console (Always On Top)

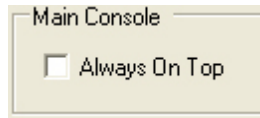


Figure 58: Always On Top

The **Always On Top** option is a common option for windows applications. With this option enabled, the main console will be painted on top of any other windows (even an active window).

Filter Controls

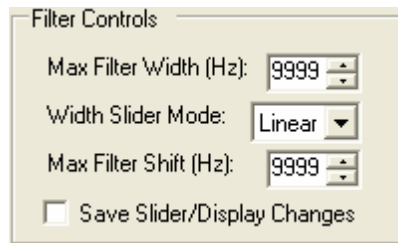


Figure 59: Always On Top

- ❑ **Max Filter Width:** Sets the maximum filter width to be set by the Filter Width slider on the front console.
- ❑ **Width Slider Mode:** Sets the mode to be used by the Width Slider. Linear, Log, and Log10 are the options. The log options offer more resolution on the smaller filter sizes.
- ❑ **Max Filter Shift:** Sets the maximum swing in Hz that the Filter Shift Slider on the front panel will allow in either direction.
- ❑ **Save Slider/Display Changes:** If checked, any changes to the filters made by the filter sliders or by using the click and drag on the filter edges on the display will be saved to the Variable filters and will be recalled as such. If not checked, the Var filters can only be changed by adjusting the Filter Low and High Cut controls on the front panel.

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Calibration Sub-Tab

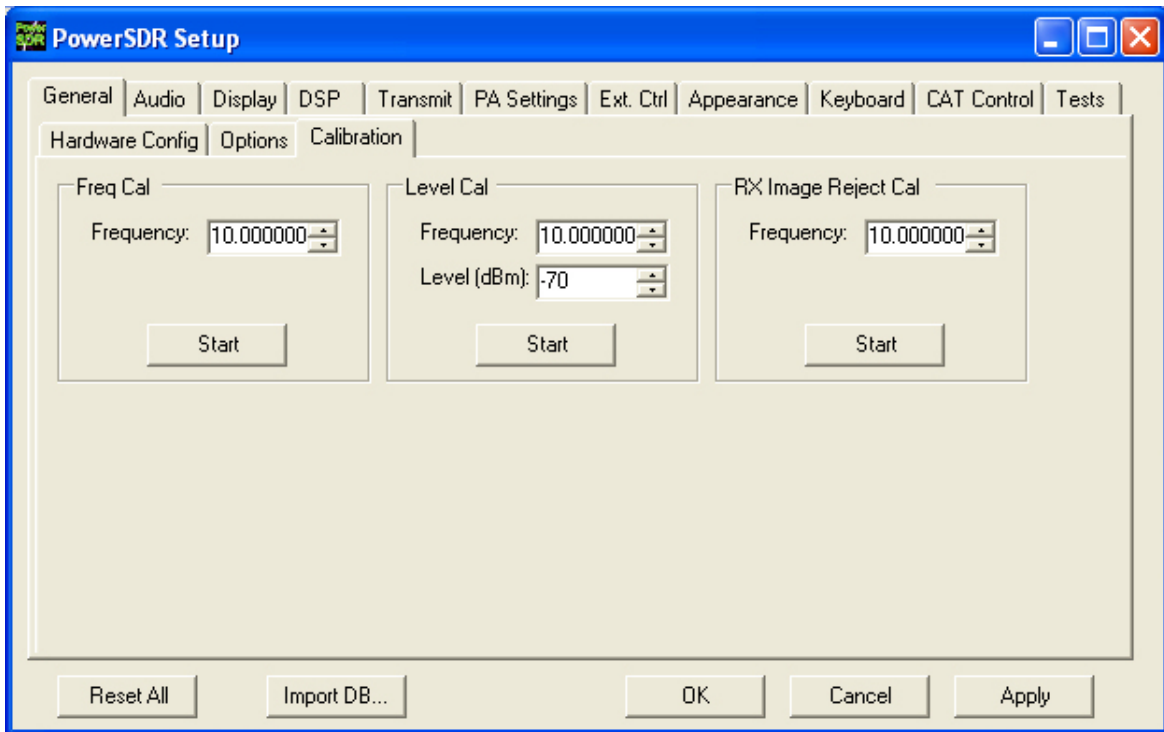


Figure 60: Setup Form - General Tab, Calibration Sub-Tab

The calibration section contains controls that are used to setup and start various calibration routines. For more information on calibrating, refer to the section in Software Installation called PowerSDR Calibration.

Freq Cal

Used to calibrate the frequency of the 200MHz oscillator in the SDR-1000. Enter the frequency of the signal source used for calibration in the control provided. The signal source can be a signal generator or a signal like WWV. Click the **Start** button to run the calibration.

Level Cal

Used to calibrate the power level in dBm for the display and the multimeter. Enter the frequency and level of the signal source used for calibration. The more accurate the signal source, the more accurate the display and multimeter will be. The multimeter is currently accurate with respect to the source to 0.1 dBm. Click the **Start** button to run the calibration.

RX Image Reject Cal

Used to calibrate the opposite sideband image rejection for the receiver. Enter the frequency of the signal source used to calibrate the rejection. Note that the calibration is done 22kHz above the signal frequency. Click the **Start** button to run the calibration.

Audio Tab

Sound Card Sub-Tab

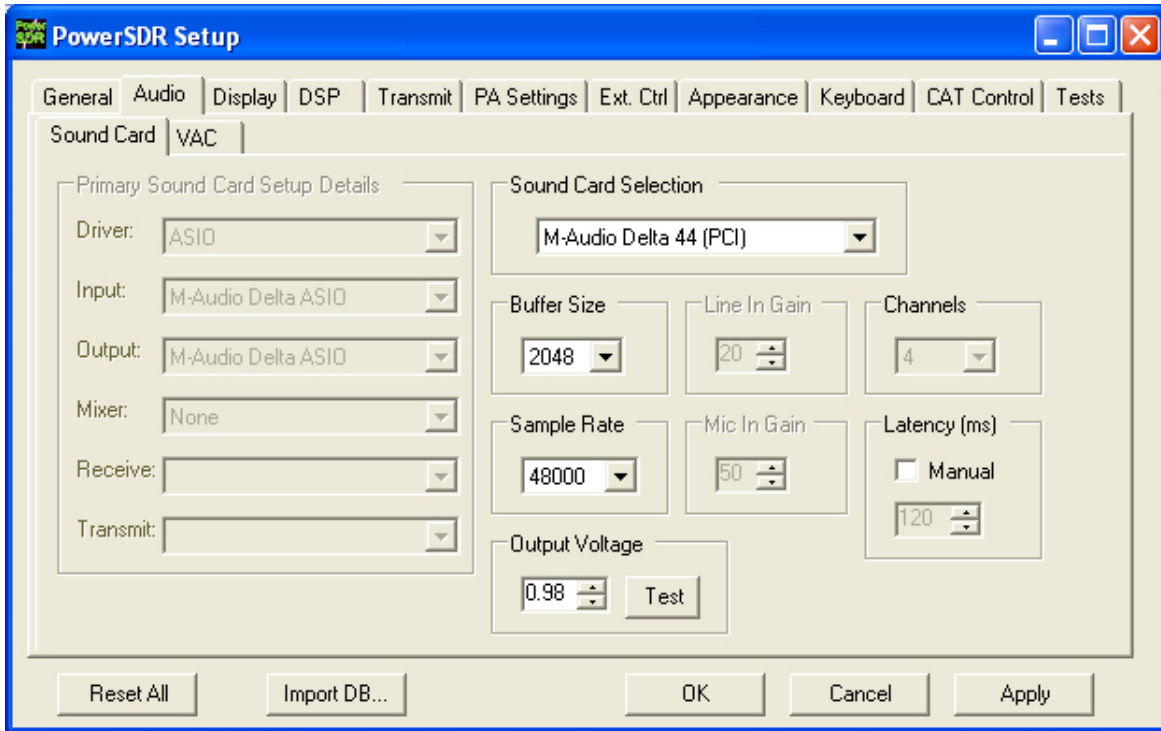


Figure 61: Setup Form - Audio Tab, Sound Card Sub-Tab

Due to space concerns, several of the Tabs (Audio and DSP) have been split into Sub-Tabs. Take care when changing the controls to pay attention which Tab (and Sub-Tab) you are on. This is especially true with the Audio Tab since many of the controls are similar.

Sound Card Selection

Allows simplified audio setup for recommended and legacy sound cards. Select the card you are using from the list. If your sound card is not in the list, select the Unsupported Card option. This will require additional setup using the controls in the Sound Card Setup Details section. Note that when using a recommended or legacy card the Primary Sound Card Setup Details are disabled (grayed out) as the values are defaulted for that specific card.

Buffer Size

Sets the size of the audio buffers in samples. Larger buffers mean more delay, but smoother audio. Smaller buffers yield less latency, but at the cost of CPU load. The 2048 sample buffer size means that a single buffer at

48kHz sampling rate is $2048/48000 = 42.7\text{ms}$ in length. Faster machines should be able to run 512 Sample buffers without issue. For best CW performance, set the audio buffer to 512 or less.

Sample Rate

The sample rate is locked to 48kHz unless your selected hardware supports higher modes (96 or 192kHz). Using these higher sampling rates will result in a wider panadapter display. For best CW performance set the sample rate to at least 96kHz

Line In Gain

Sets the Line In volume on the selected mixer for consumer grade cards. Note that supported cards have been tested and the optimal value is set by default. Selecting Unsupported Card on the Sound Card Selection will unlock this control. This control is not used for the Delta 44 or the FireBox.

Mic In Gain

Sets the Microphone volume on the selected mixer for consumer grade cards. This control should be set in order not to overload the sound card. The front panel MIC control can add software gain if using a professional grade card (Delta 44 or FireBox) or if your consumer card cannot provide adequate gain.

Latency

Using the manual option, the user may add additional latency (in milliseconds) to the audio buffering system for better audio performance. When the manual setting is off, the delay is set to 0ms. Note that some systems will have trouble with the manual setting on and values below around 15ms. We recommend using the default automatic setting for latency for best results.

Sound Card Output Voltage

This value is the measured single channel AC voltage on the sound card Line Output when running the test. To run the test, first unplug the To Line Out cable from the SDR-1000 enclosure. Click the **Test** button to begin sending the full range signal. Use an AC voltmeter to measure the voltage between the connector tip and the connector ground. Next, record the measured value in the box. Using a value that is higher or lower than your actual output will cause the power calibration to be inaccurate.

When using a supported card, this value is defaulted to an average of what we have measured across several cards. We have found that the variance from card to card (same model) is negligible (+/- 0.04V). For this reason, we recommend using the default settings if using a supported card. If your card measures significantly dissimilar to our values, please contact us for troubleshooting. The default values we use are shown in the table below.

Table 9: Sound Card Default Voltages

| Sound Card | Volts (AC) |
|-------------------------|------------|
| Turtle Beach Santa Cruz | 1.274 |
| Audigy/Audigy2 | 2.230 |
| Audigy 2 ZS | 2.230 |
| Creative Labs Extigy | 1.830 |
| Creative Labs MP3+ | 0.982 |
| M-Audio Delta 44 | 0.980 |
| PreSonus FireBox | 6.390 |

Sound Card Setup Details

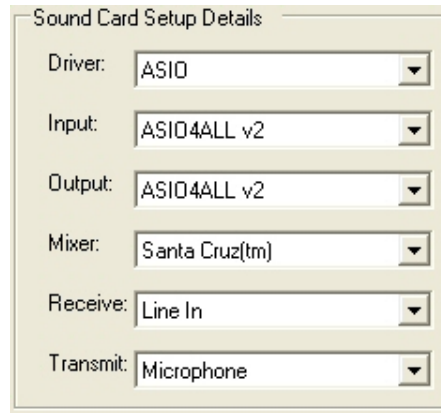


Figure 62: Sound Card Setup Details

If using a supported sound card, these controls will be set automatically. In order to change these settings, it is necessary to choose the Unsupported Card option from the Sound Card Selection drop down box.

- ❑ **Driver:** There are three audio drivers available through the PortAudio sound interface. These are in order from lowest to highest latency: ASIO, MME (multimedia extension), and DirectSound (part of the DirectX package). Note that lower latencies generally cause higher CPU loading. Therefore, it is likely that ASIO will take more CPU load than using MME or DirectSound. Selecting the Driver will populate the Input and Output Drop Down Boxes. ASIO is the recommended Driver type.
- ❑ **Input:** Selects the input audio device. While PortAudio allows different Input/Output devices, we recommend using the same device for both. The Mixer selection will attempt to follow the Input selection. However, if the naming is different, it will be necessary to manually select the mixer from the Mixer drop down box.
- ❑ **Output:** Selects the output audio device. Again, we recommend matching the Input/Output devices.
- ❑ **Mixer:** Selects the windows mixer device that will be used for volume control and transmit line switching. As noted above, this selection attempts to follow the Input selection. Manual selection is necessary if the driver name is different from the mixer name. Selecting the mixer device will populate the Receive and Transmit drop down boxes.

- ❑ **Receive:** Selects the audio line to be used in receive mode. This is typically Line In or Analog Mix.
- ❑ **Transmit:** Selects the audio line to be used in transmit mode. This is typically Microphone.

VAC Sub-Tab

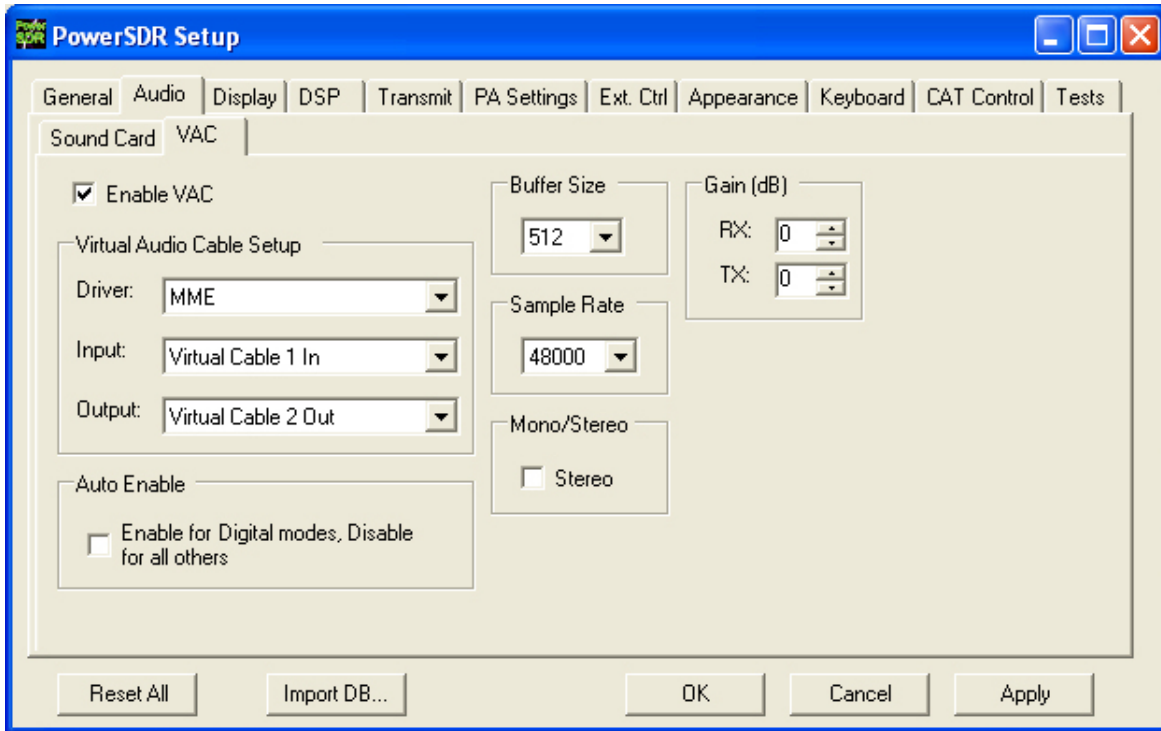


Figure 63: Setup Form - Audio Tab, VAC Sub-Tab

Use these controls to configure the VAC (Virtual Audio Cable) settings for use with the PowerSDR software. This is ideally for running digital modes, but can serve as another way to get audio in and out of the PowerSDR software. The Buffer Size, Sample Rate, and Setup details are all similar to those described for the sound card in the above section. Below we describe the unique controls on this form.

Gain (dB)

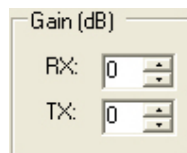


Figure 64: Gain (dB) Controls

These settings adjust the gain for signals coming in and out of the VAC interface. Use the RX control to adjust the audio level going to third party programs. Note that this control supercedes the front panel AF control for third party applications. Similarly, use the TX control to adjust the volume of audio coming from third party applications. Use this control instead of the MIC control on the front panel to calibrate transmit (adjust for 0 dB on the ALC meter).

Auto Enable

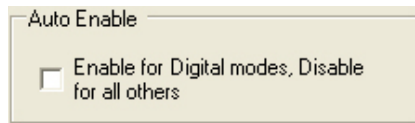


Figure 65: Auto Enable

Use this control to automatically enable VAC when in certain modes (DIGL, DIGU, DRM). This allows the user to easily switch between digital modes and SSB/AM/FM without having to change this control to enable/disable VAC.

Display Tab

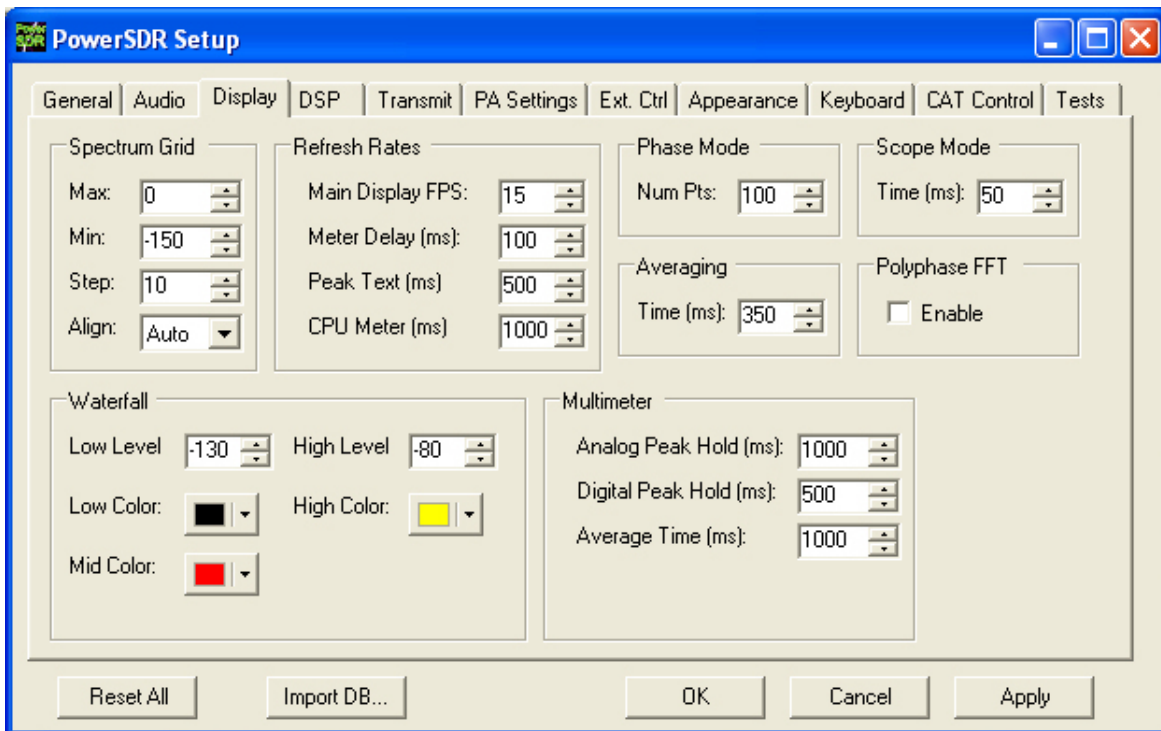


Figure 66: Setup Form - Display Tab

Spectrum Grid

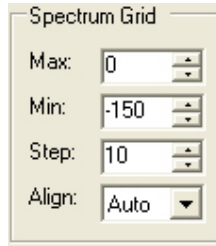


Figure 67: Spectrum Grid

The spectrum grid controls define the background grid shown in Spectrum, Histogram, and Panadapter modes.

- ❑ **Max:** The maximum signal shown on the spectrum display in dB (i.e. top of the display). Raising this value will expand the limits of the display while compressing the range shown.
- ❑ **Min:** The minimum signal shown on the spectrum display in dB (i.e. bottom of the display). Similar to the max setting, lowering this value will expand the limits of the display while compressing the range shown.
- ❑ **Step:** Spacing between vertical power spectra gridlines in dB. Some people may find the cross lines and callouts on the display distracting. This control can be used to spread the cross lines and callouts. Use the Align control to turn them off completely.
- ❑ **Align:** Sets the horizontal alignment of the spectrum grid level readouts. The Left, Center, Right, and Off settings are self explanatory. The Auto option works based on the filter selection and operates as it did in previous PowerSDR versions.

Refresh Rates

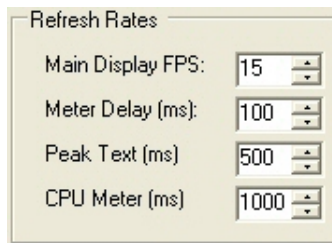


Figure 68: Refresh Rates

- ❑ **Main Display FPS:** Sets the update rate of the main display to the selected value in frames per second. Note that this is a good control to reduce if you are running on a slower machine to save CPU cycles. Raising this value will give faster updates at the cost of CPU load. Lowering the value will slow the display down.

- ❑ **Meter Delay (ms):** Sets the update rate of the multimeter on the front panel. This applies to the text and the graphical display. Raising the value increases the delay between updates and slows the meter down. Conversely, lowering the value will accelerate the meter.
- ❑ **Peak Text (ms):** Sets the update rate of the peak signal location text box located just beneath the display. Raising the value increases the delay between peak updates and slows the display down. Conversely, lowering the value will accelerate the updates.
- ❑ **CPU Meter (ms):** Sets the update rate of the CPU Meter in the lower left corner of the front panel. Raising the value will add more delay between updates while lowering the value will yield faster responses. Note that the CPU Meter value is for the entire system and not just the PowerSDR process.

Waterfall

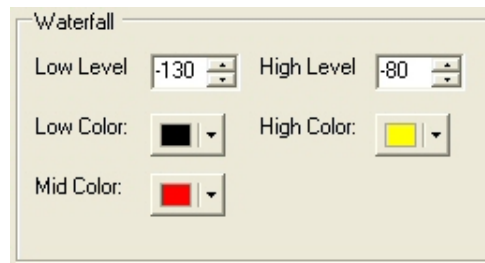


Figure 69: Waterfall

The waterfall display can be customized to look the way you like it in terms of dynamic range response and color coding. If a signal is at or below the Low Level, it will use the Low Color. If a signal is at or above the High Level, it will use the High Color. Signal levels in between the two levels will use a linear gradient between whichever level it is closer to (high or low) and the Mid Color.

- ❑ **Low Level:** The lower end of the dynamic range to view in dB. Signals at or below this level will use the Low Color.
- ❑ **High Level:** The high end of the dynamic range to view in dB. Signals at or above this level will use the High Color.
- ❑ **Low Color:** Color used if the signal level is at or below the Low Level.
- ❑ **High Color:** Color used if the signal level is at or above the High Level.
- ❑ **Mid Color:** Color used if the signal level is exactly between the Low and High Levels. A gradient is used between the Mid Color and either the High or Low Color (whichever is closer) if the value is not exactly in the middle of the selected levels.

Multimeter

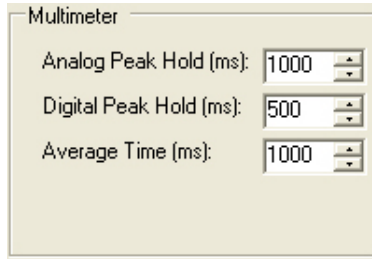


Figure 70: Multimeter Display options

- ❑ **Analog Peak Hold (ms):** The length of time in milliseconds to hold the peak on the analog bar meter. Raising this value will lengthen the hold time and peaks will be held for a longer period of time.
- ❑ **Digital Peak Hold (ms):** The length of time in milliseconds to hold the digital meter when using the “Peak Pow” option in transmit mode. Raising this value will lengthen the hold time and peaks will be held for a longer period of time.

Scope Mode

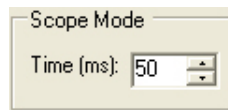


Figure 71: Scope Mode

This control adjusts the amount of time to be shown in one pass of the Scope Display Mode.

Polyphase FFT

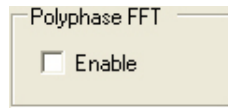


Figure 72: Polyphase FFT

Enable this feature to get sharper peaks in spectrum modes (Spectrum, Panadapter, Waterfall, Histogram). Expect in narrow display modes to see a reduction in the area of the “spike” of a tone by a factor of 1/4.

DSP Tab

Options Sub-Tab

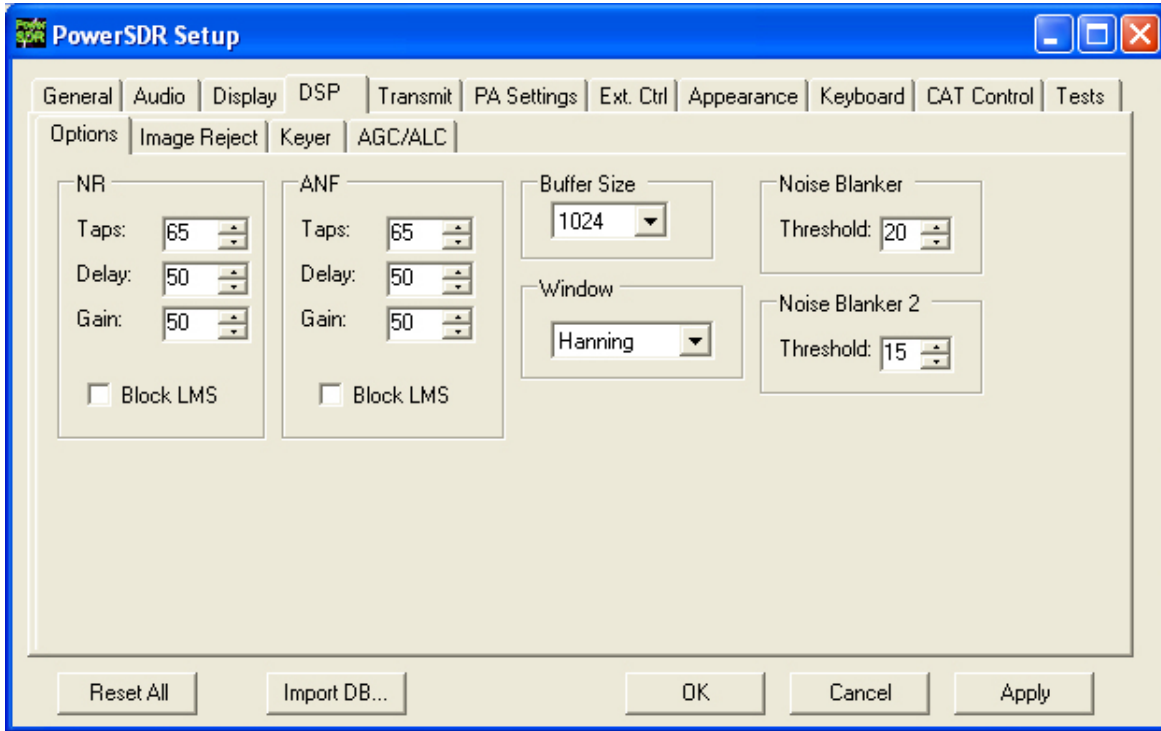


Figure 73: Setup Form - DSP Tab, Options Sub-Tab

Noise Reduction Controls

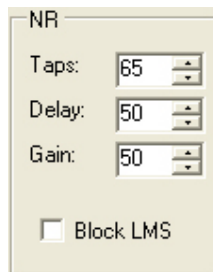


Figure 74: Noise Reduction Controls

Noise Reduction (**NR**) attempts automatic computation of a filter that maximizes the coherent or non-noise like signals and as a result, filters out the rest of the signal, which includes noise. It is best used for speech signals with a good signal to noise ratio or tones.

- ❑ **#Taps:** This determines the length of the computed filter. The longer the filter, the better the non-coherent signals (noise) will be cancelled. It also introduces latency equal to the number of filter taps divided by the sample rate in samples per second and is in addition to Delay (see below). The larger the number of taps, the longer it takes for the filter to converge but upon achieving convergence, the better the filter will be.
- ❑ **Delay:** Determines how far back you look in the signal before you begin to compute a coherent signal enhancement filter. With large delays, there is a higher the likelihood of detrimental affects to normal speech. Latency is also introduced that is equal to the Delay.
- ❑ **Gain:** Determines the adaptation rate of the filter. The larger the number, the faster the filter will converge *and the less stable it will be.*
- ❑ **Block LMS NR:** We have implemented a fast block LMS routine. This routine is faster than the old routine, gives more taps in the filter, and will work on I/Q signals which enables it to work in **BIN** mode.

Automatic Notch Filter Controls

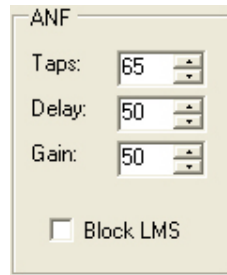


Figure 75: Automatic Notch Filter Controls

The Automatic Notch Filter (**ANF**) attempts automatic computation of a filter to remove one or more carrier tones that are interfering with the signal of interest.

- ❑ **#Taps:** This determines the length of the computed notch filter. The longer the filter, the larger the number of tones that can be cancelled and the more effective the cancellation will be. It also introduces latency (signal delay) equal to the number of filter taps divided by the sample rate in samples per second and is in addition to Delay (see below). The larger the number of taps, the longer it takes for the filter to converge but upon achieving convergence, the better the filter will be.
- ❑ **Delay:** Determines how far back you look in the signal before you begin to compute a cancellation filter. The larger the delay, the less the impact on normal speech, and the more likely the filter will be to concentrate only on longer term coherent signals such as carrier tones. There is latency introduced equal to the Delay.
- ❑ **Gain:** Determines the adaptation rate of the filter. The larger the number, the faster the filter will converge *and the less stable it will be.*

- ❑ **Block LMS NR:** We have implemented a fast block LMS routine. This routine is faster than the old routine, gives more taps in the filter, and will work on I/Q signals which enables it to work in **BIN** mode.

Buffer Size

This controls the size of the DSP buffers. Note that this is completely independent of the Audio buffer size. This controls the size of the FFT filter and therefore, the group delay (latency) through the digital filter. Higher values will give more latency and sharper filters. Lower values will allow nearly real time monitoring with filters that “roll off” (as opposed to the typical “brick wall” filters). Note that using lower values will negate the effects of some of the most narrow filters (25 or 50Hz).

Noise Blanker

This controls the detection threshold for impulse noise. If a signal exceeds this detection threshold, the sample will be set to zero and the filtering in the radio serves to interpolate through this zero sample. This noise blanker is identical in theory to those in traditional radios. The detection threshold in our noise blanker has the unique feature that they are signal strength dependent. This enables them to function properly at all signal levels.

This control is preferable when the spikes are very large in comparison to the average signal. However, when the spike is smaller, NB2 provides a much cleaner reconstruction of the signal since the signal is more likely to look like the mean. For this reason, the NB2 threshold should always be about four or five less than the NB threshold.

Noise Blanker 2

This controls the detection threshold for a pulse. If a signal, pulse or not, exceeds this detection threshold, the sample will be replaced by a computed estimation of what the signal sample should have been given an interpolation of the signal samples around it in time. By replacing the noise pulse with an interpolation of the signal, distortion is greatly reduced over that of traditional noise blankers.

When seeing lots of impulsive noise, the signal can really be damaged by being too aggressive with NB. However you still would like to remove the large pulses altogether before allowing the smoother acting blanker NB2 to operate. So if you are seeing lots of repetitive noise pulses, it is probably a good idea to use both NB and NB2. NB is adjusted to just begin to lower the noise from the pulses and then, NB2, with threshold four or five less, is turned on. The two together can then produce some spectacular noise reduction and increase intelligibility of the signal under severely adverse conditions.

Window

This selects the DSP windowing function that will be applied to the power spectrum in the main display when in Spectrum, Panadapter, Histogram, and Waterfall display modes. The default is Hanning, which is the best setting for many high-level signal measurement needs. The purpose of the windowing is to diminish bleed through to adjacent “bins” which results from a tone that is not right on the center frequency of one of the “bins” (or parallel filters) in the power spectrum calculation. The bleed through is caused by the use of the Fast Fourier Transform (FFT) to do the power spectrum. We need to use it for the sake of efficiency. That said, it

is important to understand that the FFT writes the data (mathematically speaking) on a circle and not on a line. So when the last sample meets the first sample as you write the data on a circle, it is very probable that it will not meet up or join in a continuous fashion. This discontinuity acts in exactly the same manner a key click causes a wide spectrum. The window is used to mitigate this key click like phenomenon. The worst for bleed through is Rectangular Window. The “best” for bleed through in our selections is Blackman-Harris. For further reading on windowing, please see the following links:

<http://astronomy.swin.edu.au/~pbourke/analysis/windows/>

<http://www.cg.tuwien.ac.at/studentwork/CESCG/CESCG99/TTheussl/node4.html>.

Let us consider each option and some of their characteristics.

- **Rectangular:** This means the data has no window applied. No shaping is applied to the incoming signal. As a result, you will have the greatest sensitivity in the power spectrum and the greatest bleed through or interference with adjacent bins.
- **Hanning, Hamming:** These windows are based on a raised cosine shape. In addition to providing continuity (Hanning) or near continuity (Hamming), these both provide a shape that makes the slopes of the signal agree at the beginning and the end of the signal buffer. This filter provides for both good sensitivity and less bleed through in the FFT.

$$\text{Hanning: } w[k] = 0.5 \left(1 - \cos \left(2\pi \frac{k}{n+1} \right) \right), \quad k = 1, \dots, n$$

$$\text{Hamming: } w[k+1] = 0.54 - 0.46 \cos \left(2\pi \frac{k}{n-1} \right), \quad k = 0, \dots, n-1$$

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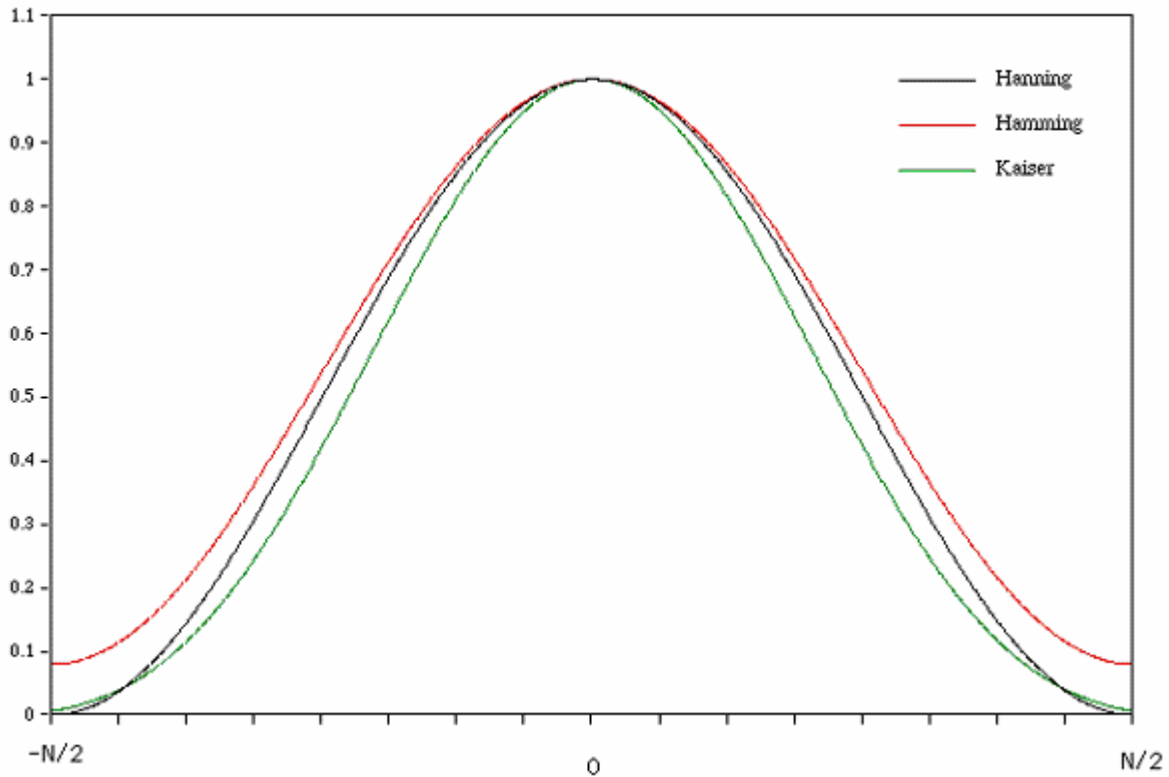


Figure 76 Hanning, Hamming and Kaiser Windows

- **Welch:** This applies a shaping function to the data before the FFT is computed. The figure gives its shape.

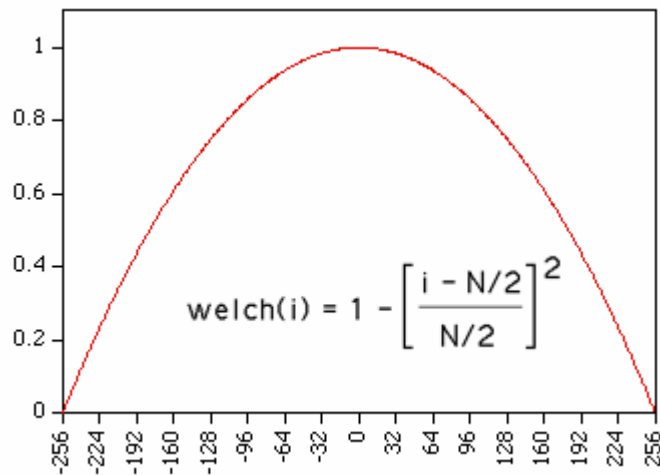


Figure 77: Welch Window

This picture is from a 512 long Welch Window. We actually use a 4096 long window but this does illustrate the qualitative features. Notice that at the left and the right end, we have forced the continuity by applying this tapering function. This continuous move to zero at the endpoints mitigates the “key-click” phenomenon that would otherwise result from the boxcar or rectangular window.

- **Blackman, Blackman-Harris:** We use Blackman-Harris to design all filters in the signal processing chain for all modes. This provides for minimal bleed through as we have described and the best shape factor for our needs in the overlap-save filtering routines. While it has some good features, these features are probably not ideal for the CW power spectrum *displays*. There is a penalty to pay for the smoothing near the ends afforded by the windows. If you have experience in the use of FFTs and the display of the power spectra that result, you know that a smaller FFT at a fixed sample rate causes a wider range of frequencies to be contained in one bin of the power spectral results. This means that a tone will look more like a large lobe or finger than a tone spike. The same phenomenon is present in the best windows. While the spreading of the “main lobe” of a tone is not as bad as taking an FFT of half the size, it is wider than one bin. The formulae for all of the Blackman filters are of this form:

$$f(x) = a_0 - a_1 \cos(\pi x) + a_2 \cos(2\pi x) - a_3 \cos(3\pi x)$$

The individual parameters by type:

- Blackman: $a_0 = 0.42, a_1 = -0.5, a_2 = 0.08, a_3 = 0$
 - Blackman Harris: $a_0 = 0.35875, a_1 = 0.48829, a_2 = 0.14128, a_3 = 0.01168$
 - Blackman Nuttall: $a_0 = 0.3635819, a_1 = 0.4891775, a_2 = 0.1365995, a_3 = 0.0106411$
- **Parzen, Bartlett, Exponential, and Riemann:** These windows are much less widely used and are included for completeness. They sometimes perform better for a particular application and some experimentation on the users part should be undertaken to find the window that gives the most pleasing display to you.

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Keyer Sub-Tab

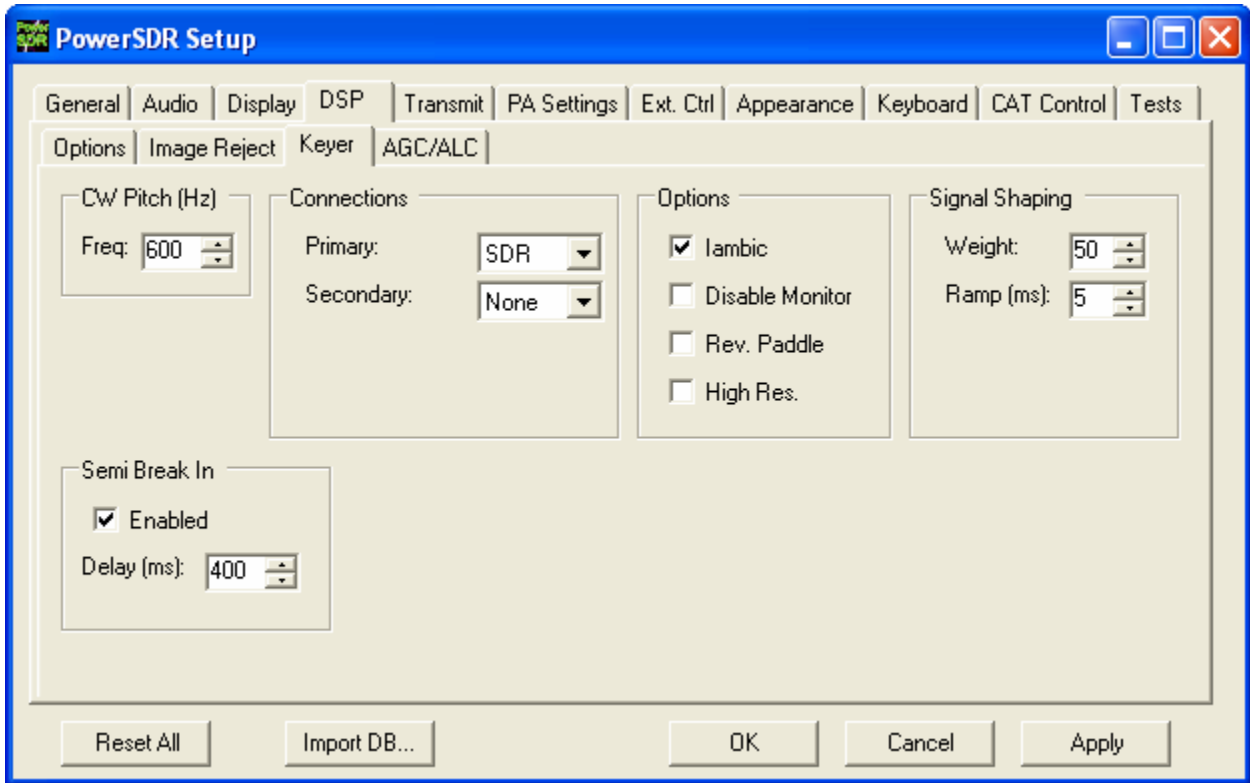


Figure 78: Setup Form - DSP Tab, Keyer Sub-Tab

Note: Modifying the Audio buffer size (as well as the soundcard latency or buffer settings) can help bring the latency of the internal keyer down to a minimum. Also, we suggest using the serial hookup for keying the radio via an external paddle or keyer. See the CW Operation section in the Operating chapter below for more information.

CW Pitch

This enables the user to set the desired audio frequency for CW listening at the center of the CW filters (1kHz and lower filters). This will determine the offset that is applied to the carrier in receive and transmit. The display will continue to read the actual carrier frequency (if the radio is calibrated) but the software will provide for an offset to get the desired CW tone. This pitch will determine the automated tuning frequency using the display and mouse “click tune” functions.

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Connections

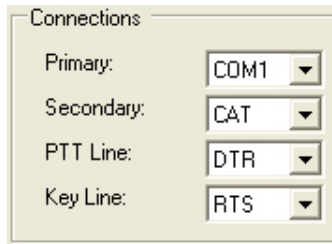


Figure 79: Connections

The internal keyer supports two inputs. We call these inputs the primary and secondary connections. The primary connection will override the secondary input. This was designed with the idea that the secondary connection might be used for automatic CW generation while the primary could be used with manual paddles to override the automatic output.

- ❑ **Primary:** Select the connection to be used for the primary connection. Selecting “SDR” allows use of the jack on the back of the SDR-1000.
- ❑ **Secondary:** Select the connection to be used for the secondary connection. Selecting CAT will allow use of the COM port that is currently being used by the CAT connection. This is useful as some programs allow both CAT commands and COM port line keying for CW. Note that the below two controls will not show up if None is selected.
- ❑ **PTT Line:** Select the COM port line used for PTT.
- ❑ **Key Line:** Select the COM port line to be used to activate the key.

Signal Shaping

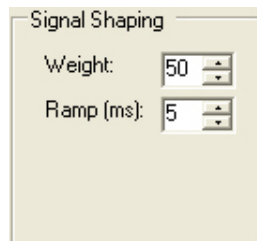


Figure 80: Signal Shaping Controls

- ❑ **Weight:** Sets the width ratio between the dot and dash.
- ❑ **Ramp:** Sets the length of the leading and trailing edge on the tones in milliseconds to avoid “key clicks.”

Semi Break In

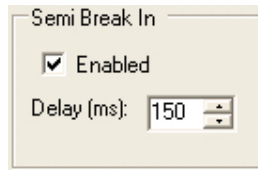


Figure 81: Internal Keyer Semi Break In Controls

- ❑ **Enabled:** Check this box to enable Semi Break In for the internal keyer.
- ❑ **Delay (ms):** Sets the amount of time between the last detected input and when the radio will drop back to receive mode. The smallest possible setting is 150 ms and it is recommended that this be set larger than 250ms. The SDR-1000 is not a QSK radio.

Options

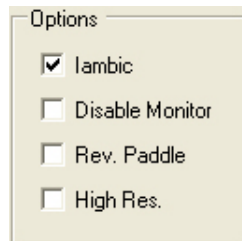


Figure 82: Internal Keyer Options

- ❑ **Iambic:** Check this box to enable Iambic mode “B” emulation. With the box unchecked, the key input will act like a straight key.
- ❑ **Disable Monitor:** The monitor is typically turned on when using Semi Break In with the internal keyer. In order to keep the monitor off, check this option.
- ❑ **Rev. Paddles:** Using this option will reverse the paddle inputs so that the dot becomes a dash and vice versa.

Image Reject Sub-Tab

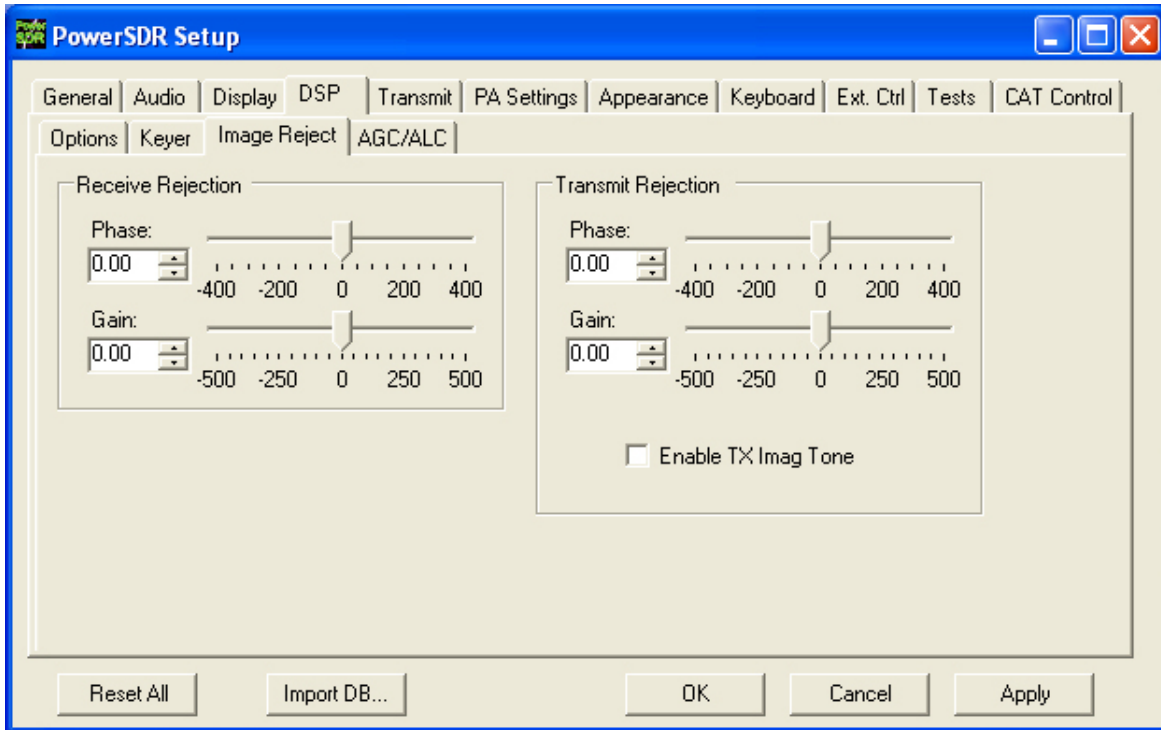


Figure 83: Setup Form - DSP Tab, Image Reject Sub-Tab

These controls enable the user to adjust the image to minimum in order to reject the opposite sideband. There is an automatic process for doing this on receive (signal source required). For transmit, the process is described below.

Receive Rejection

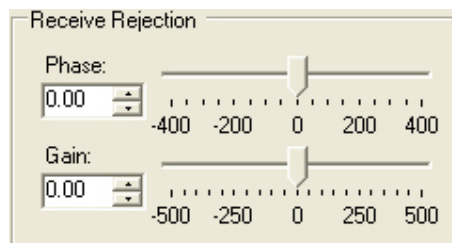


Figure 84: Receive Rejection

Image Rejection really means finding adjustments for phase angle and gain differences between the left and right channels in your stereo card (or the in-phase (I) and quadrature (Q) signals). This is done by a simple algorithm. The algorithm assumes that the phase offset and gain difference between the I and Q channels are constant across all bands and assumes that the channel the audio channel is linear across the sound card bandwidth. This works extremely well in practice.

- ❑ **Phase:** Sets the phase offset between the I and Q channels. Ideally, the phase angle of a tone in our passband will be 90 degrees different in the I and Q (right and left channels).
- ❑ **Gain:** Sets the amplitude offset between the I and Q channels. Ideally, the amplitude of a received tone will be equal in both I and Q (left and right) channels.

Transmit Rejection

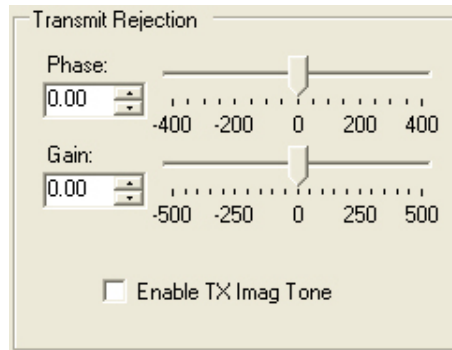


Figure 85: Transmit Rejection

Similarly to above, these controls enable the user to adjust the image rejection for the transmitter. This calibration requires external instruments. A spectrum analyzer is ideal but a second receiver should enable you to get satisfactory rejection levels. When you select Enable TX Image Tone, a tone is sent through the DSP. This only works in voice modes and should be done only in SSB because of the asymmetric nature of the SSB signal. Any small amount of image problem will likely be covered up in AM, SAM, and FMN by the symmetric nature of the signals. When the Enable button is selected, and the radio is in (say) USB, the MOX button will send a full strength tone through the DSP. The power this tone generates is controlled by the PWR setting on the front console. If the radio is in USB, look at the signal BELOW the carrier in either the spectrum analyzer or the second receiver. Using these controls, null the signals BELOW the carrier. If PowerSDR is set to LSB, minimize the images above the carrier. There is no way at this time to automate this procedure. It must be done manually.

AGC/ALC Sub-Tab

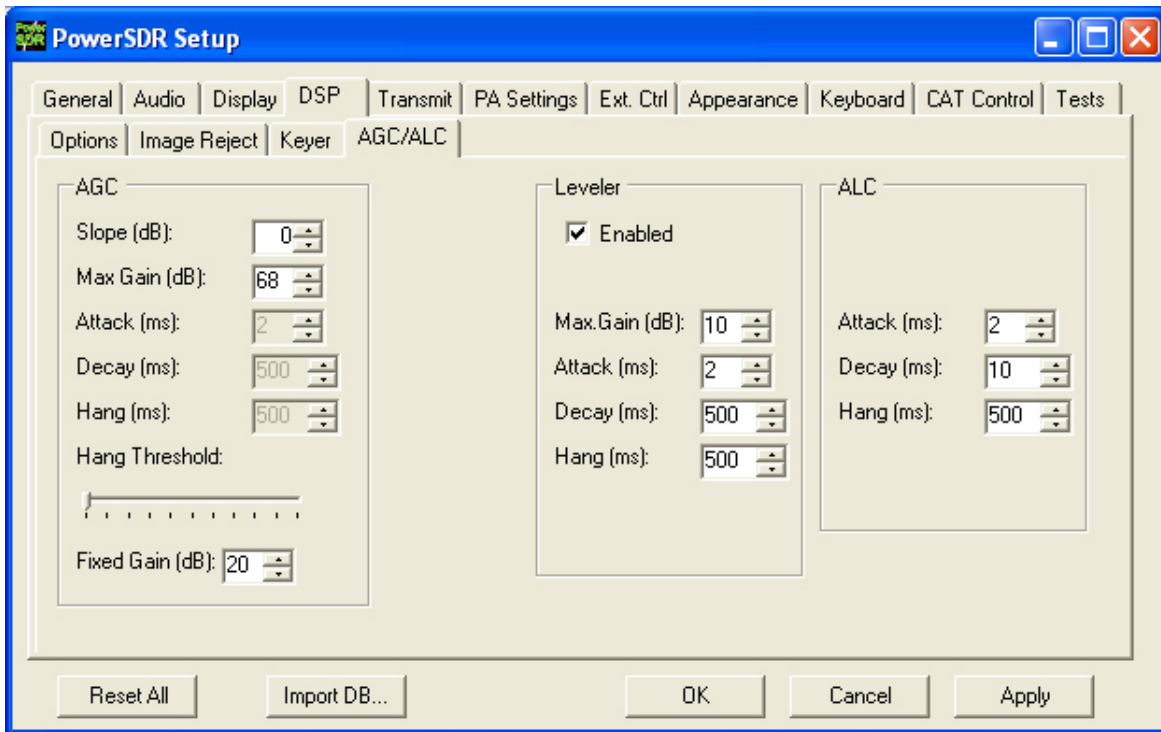


Figure 86: Setup Form - DSP Tab, AGC/ALC Sub-Tab

These controls allow the user to customize the AGC/Leveler/ALC to their own particular tastes.

AGC

The settings control the AGC **Custom** setting on the front panel.

- ❑ **Slope (dB):** Slope changes the rate at which gain increases with decreased signal. If you have the Slope set to 10 dB and the signal gets weaker by 1 dB, the gain will be increase by 10 dB. The increase is still limited by max gain. This provides different decay action in the AGC. Attack is unaffected. 0dB is default, 10 dB is maximum.
- ❑ **Max/Neg Gain (dB):** The maximum (or minimum) amount of gain the system is allowed to apply. On RX (AGC), this can be used to minimize ambient noise that may cause listening fatigue.
- ❑ **Attack (ms):** Note that in order for this and the two controls below it to be enabled, the AGC control on the front panel must be set to Custom.
- ❑ **Decay (ms):** This sets the time constant for the decay for the AGC. When a signal gets weaker, this determines how quickly the AGC will respond to the need for increased gain.

- ❑ **Hang (ms):** To keep the AGC system from doing too much up/down, an adjustable hang time is provided. After this time has expired, the Decay will then determine how quickly the AGC gain recovers.
- ❑ **Hang Threshold:** The hang will NOT occur if the signal is weaker than this threshold.
- ❑ **Fixed Gain:** When you choose Fixed AGC on the front panel, this number is used to amplify the signal.

Leveler

The Leveler is intended to even out the variations in input to your microphone caused by varying distance from or angle presented to it. It is an attempt to “level” the amplitude presented to the rest of the DSP audio processing. The leveler is *disabled* in DIGU and DIGL modes.

ALC

The ALC is what you would typically consider ALC to be in a transmitter. It is an attempt to prevent overdrive of the amplifier and the distortion that would accompany that. Because of the “dual channel” AGC algorithm we use, this ALC will allow very high average power while maintaining peaks at a controlled level. The Compressor and Compander (CMP and CPDR on front panel and on Setup-> Transmit tab) work very well with the ALC to increase average power without overdrive.

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Transmit Tab

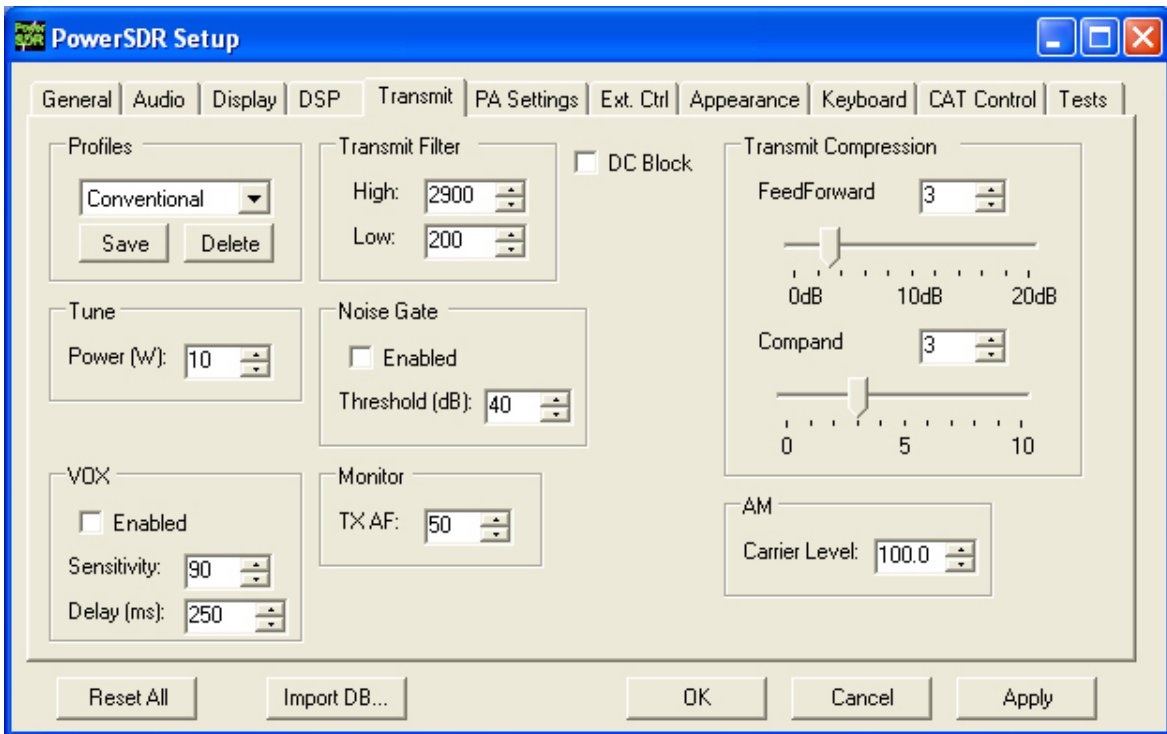


Figure 87: Setup Form Transmit Tab

The transmit Tab has controls that allow the user to tailor the transmit signal characteristics using features like compression and filtering.

- ❑ **DC Block:** Attempts to block any DC noise from entering the filter. This is used mainly with lower quality sound cards or on systems where there are grounding issues.
- ❑ **Tune Power:** Sets the power in Watts to be used whenever the TUN (Tune) function is used on the front panel (sets the PWR control). Changes made to the PWR level while TUN is active will be reflected in this control.

TX Profiles

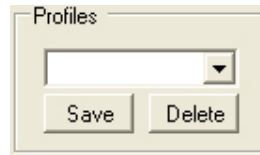


Figure 88: TX Profiles

The TX Profiles selection allows the user to save and restore the various TX settings with ease. The TX profile includes: EQ, Filter, Compression, MIC, PWR, COMP, CPDR, Leveler, and ALC settings. Simply press the **Save** button to save the current profile. You will be prompted for a name. To change profiles, simply select the named profile from the drop down list. To remove a profile, select it using the drop down menu, and then press the **Delete** button.

Filter

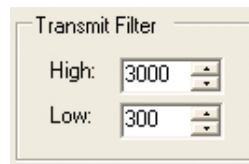


Figure 89: Transmit Filter Controls

- ❑ **High:** Controls the high cut of the Transmit Filter.
- ❑ **Low:** Controls the low cut of the Transmit Filter.

Note: You will receive a “good practice” warning if the transmit filter bandwidth exceeds 3kHz.

Noise Gate

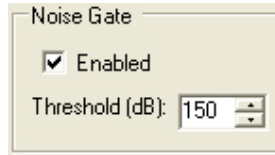


Figure 90: Noise Gate

- ❑ **Enabled:** Enables the transmit noise gate.
- ❑ **Threshold (dB):** The threshold below which the transmitter is silenced (gated). Works best when adjusted so that room noise will not trigger the transmitter, but it responds immediately upon speaking into the mic.

VOX

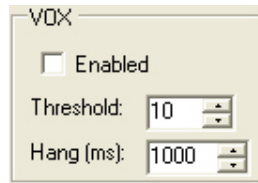


Figure 91: VOX

- ❑ **Enabled:** Enables/Disables the VOX operation.
- ❑ **Threshold:** The threshold above which the software automatically switches to transmit mode. Use this in combination with the Noise Gate for best results.
- ❑ **Hang (ms):** Time to stay in transmit after the last peak above the threshold.

Note: Due to current hardware limitations with the TR switch time, VOX will nearly always drop the first syllable during the transition.

TX Monitor

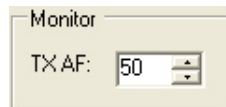


Figure 92: Transmit Monitor AF Control

Use the TX Monitor AF control to set the value that the AF control will use during transmit mode. Note that this is only functional with 4 port cards such as the Delta 44 and the FireBox.

Compression

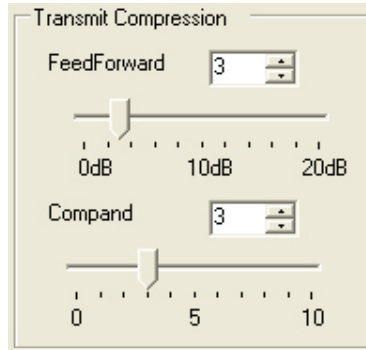


Figure 93: Transmit Compression

- ❑ **Feed Forward Compression:** Determines the “look forward” transmitter automatic gain setting. Unique to DSP implementations, it allows the required gain to anticipate the level before application rather than simply following the signal around. Raising the value will give higher average power without raising the peaks. It helps prevent pumping and popping with large signal onsets and allows for very high average power with low distortion. Note that depending on the microphone/sound card combination, higher values of compression may result in distortion on the output. This setting can be enabled by using the **COMP** button in the DSP group on the lower right hand section of the front console.

- ❑ **Compond:** The compond setting given here is really a compression setting. If you are familiar with “μLaw”, the compression that takes place is very similar. The setting here does not correspond to dB but to a slope of the compression algorithm near zero magnitude. In other words, how much are we pushing the signal up at small magnitudes. As the magnitude of the input signal goes up, less gain is applied and the fall off of the gain is exponential in its decay. With the goal of introducing fully componded SSB in the near future, this control was put in to the console to allow us to gain experience with this algorithm.

AM Carrier Level

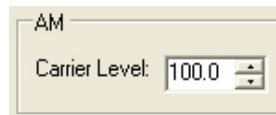


Figure 94: AM Carrier Level

The **Carrier Level** determines the percentage of carrier level to be applied to the transmit signal where 100% is one quarter of full power output (25W when PWR is set to 100). So a setting of 80 would yield roughly 16W when PWR is set to 100. This is useful as it allows the modulation to appear much stronger due to the weaker carrier.

PA Settings Tab

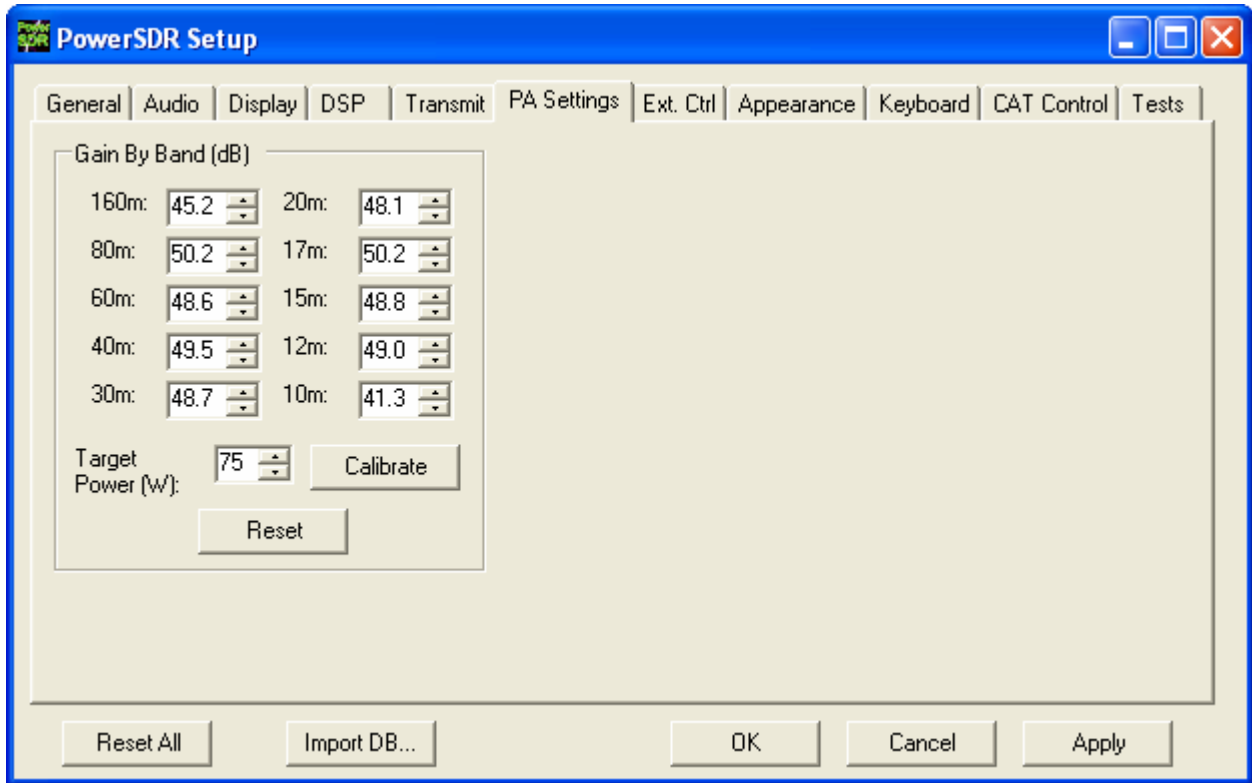


Figure 95: Setup Form - PA Settings Tab

Gain By Band (dB)

This shows the total hardware (radio + PA) signal chain gain. These controls are used to balance the output power across the ten supported amateur bands. A higher gain figure for the hardware (as shown) means a lower audio drive gain requirement.

- ❑ **Calibrate:** this button will allow those with 50-Ohm dummy loads to calibrate these values automatically. The calibration routine will utilize the **Calibration Point** as the target for the algorithm. Failure to use a dummy load with this test could cause damage to the amplifier. Those without dummy loads should simply plug in the values shipped with the unit.
- ❑ **Reset:** this button is included to reset all of the values to 48.0dB (low power output).

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Ext. Ctrl Tab

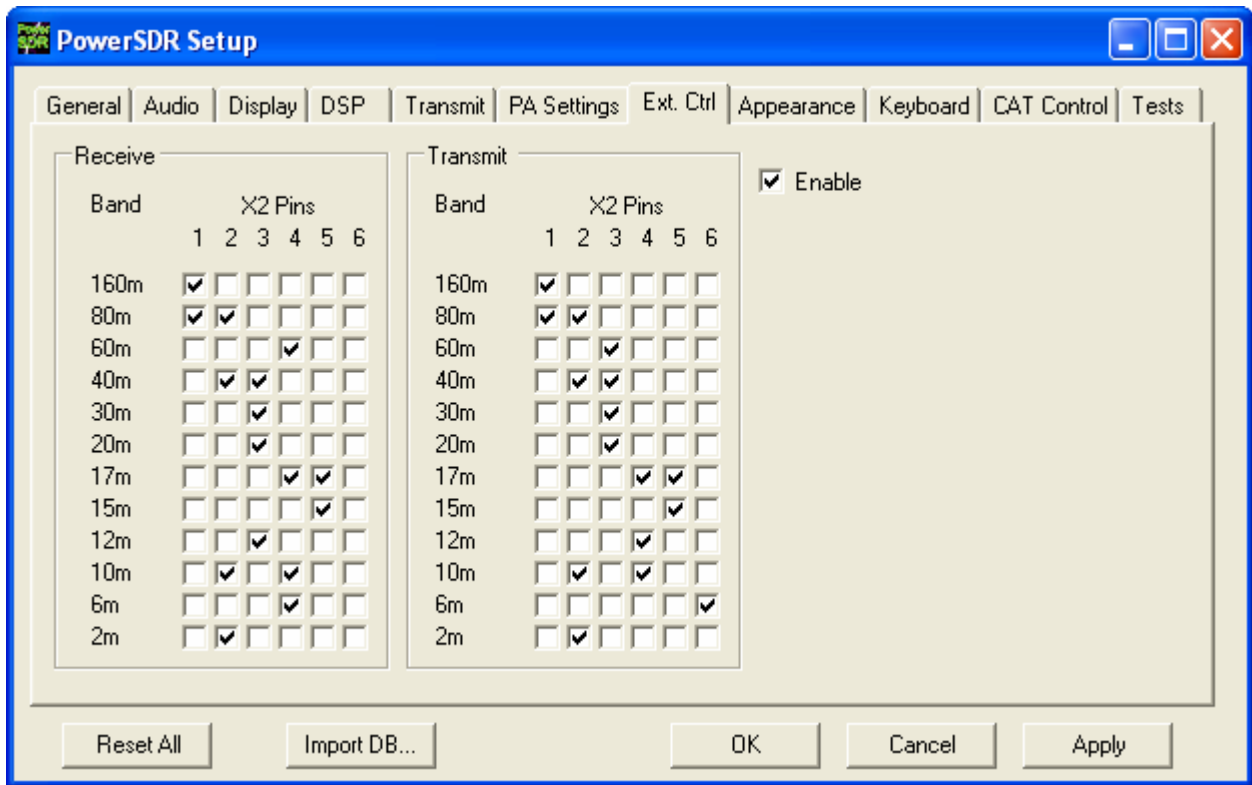


Figure 96: Setup Form - Ext. Control Tab

This form enables the user to map various bands to X2 pin outputs. This is a good way to control an external linear amplifier or transverter that needs to be switched in and out depending on what band you are on. Note that there are separate settings for transmit and receive. To use the controls, simply mark the boxes to indicate which pins should go logic high for that band. For example, if you check the 2 and 4 boxes on the 20m band on receive, those two pins will go high when in receive mode on that band.

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Appearance Tab

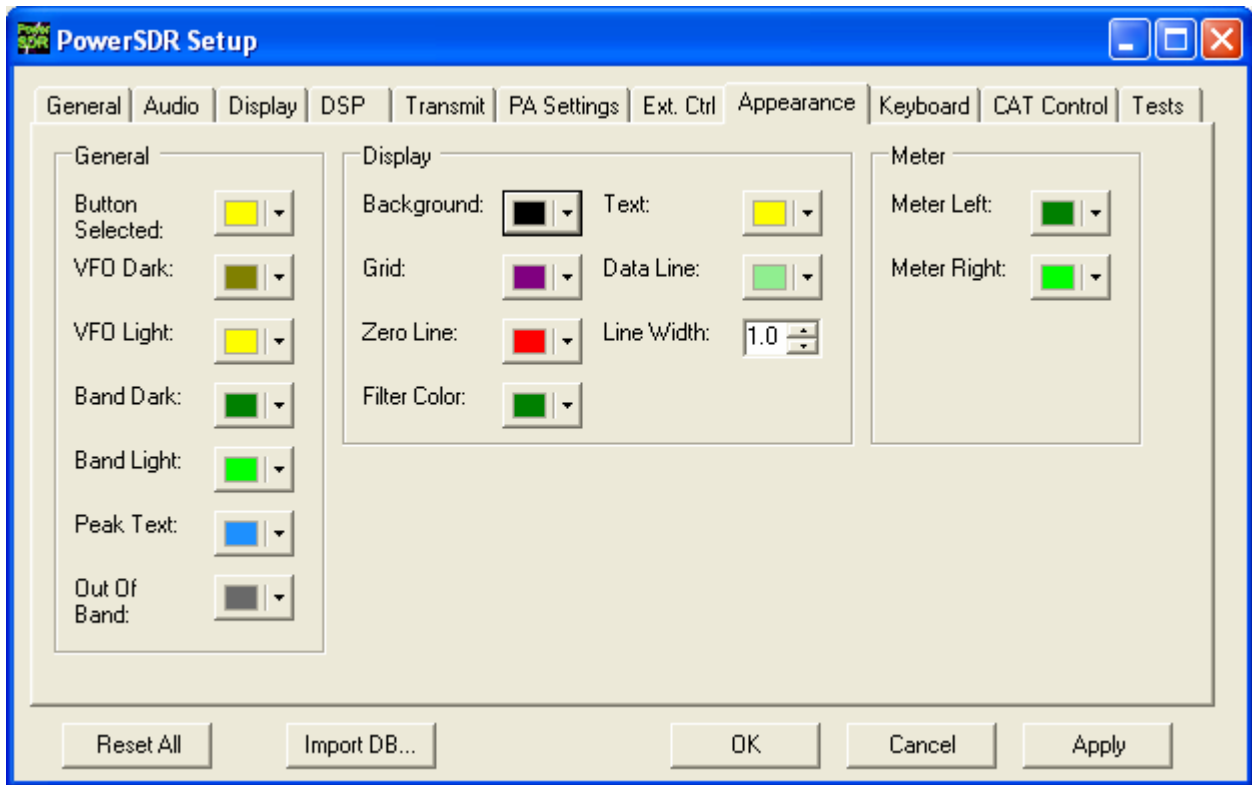


Figure 97: Setup Form Appearance Tab

The appearance controls allow the user to completely customize the appearance of the front console.

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General



Figure 98: Appearance - General

- ❑ **Button Selected:** The color of buttons when they are in the selected state.
- ❑ **VFO Dark:** The color of the text in the VFOs when they are inactive.
- ❑ **VFO Light:** The color of the text in the VFOs when they are active.
- ❑ **Band Dark:** The color of the band information text when that VFO is inactive.
- ❑ **Band Light:** The color of the band information text when that VFO is active.
- ❑ **Peak Text:** The color of the Peak signal location text located under the display on the right.
- ❑ **Out Of Band:** The background color for the VFO when outside of the amateur radio bands.

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Display



Figure 99: Appearance – Display

- ❑ **Background:** The color of the background for the display.
- ❑ **Grid:** The color for the grid in display modes where a grid is necessary.
- ❑ **Zero Line:** The color of the zero line in frequency display modes.
- ❑ **Filter Color:** The color of the filter overlay in the Panadapter display mode.
- ❑ **Text:** The color of the text on the display for frequency and level callouts.
- ❑ **Data Line:** The color of the actual data being displayed.
- ❑ **Line Width:** The width in pixels of the actual data being displayed.

Meter

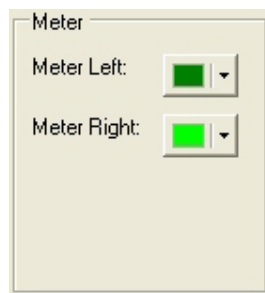


Figure 100: Appearance – Meter

- ❑ **Meter Left:** Color of the left side of the meter gradient.
- ❑ **Meter Right:** Color of the right side of the meter gradient.

Keyboard Tab

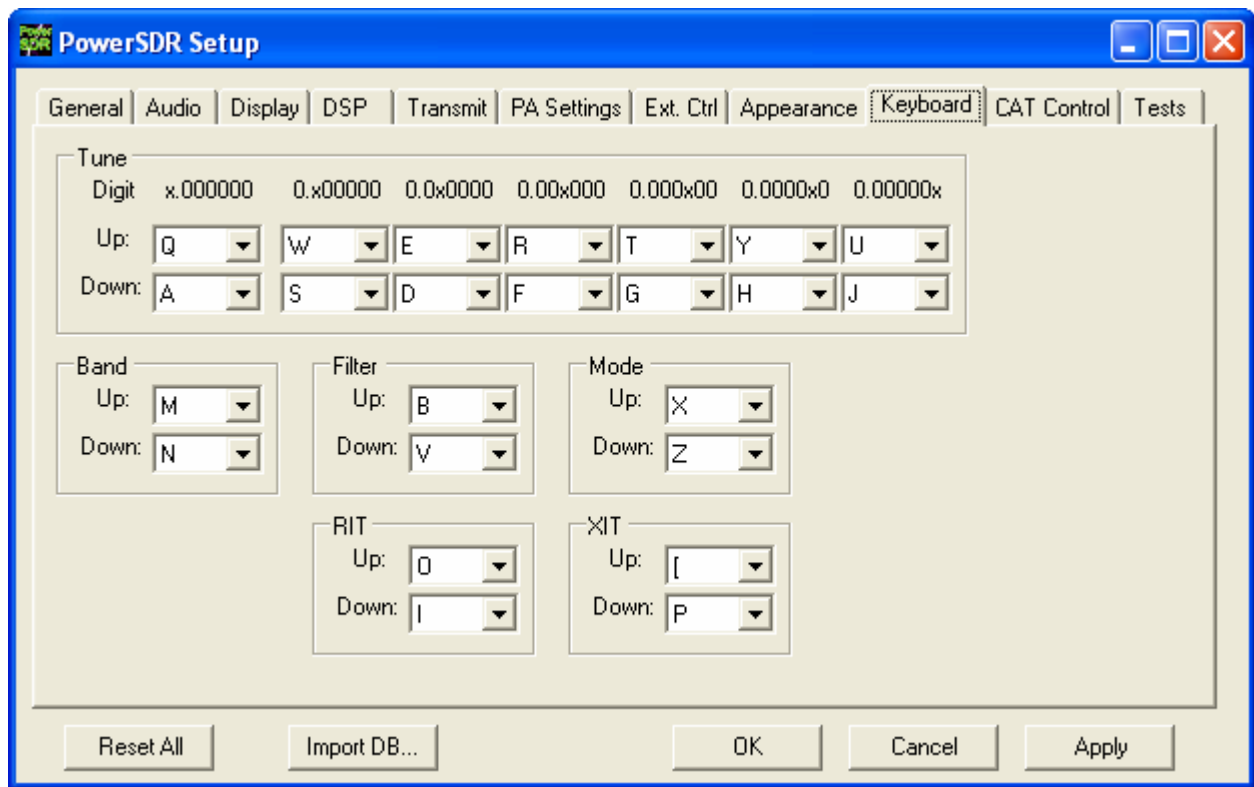


Figure 101: Setup Form - Keyboard Tab

The **Tune** mapping options allow you to tune each digit (with resolution to 1Hz) up or down using the key of your choice. The digit to be tuned is shown in the labels above the hot keys as an ‘x’. Similarly, you can map keys to change the **Band**, **Filter**, **Mode**, **RIT**, or **XIT** up or down using the drop down controls in the respective sections.

Note: Choosing any of the arrow keys will require using Alt + [arrow key].

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Cat Control Tab

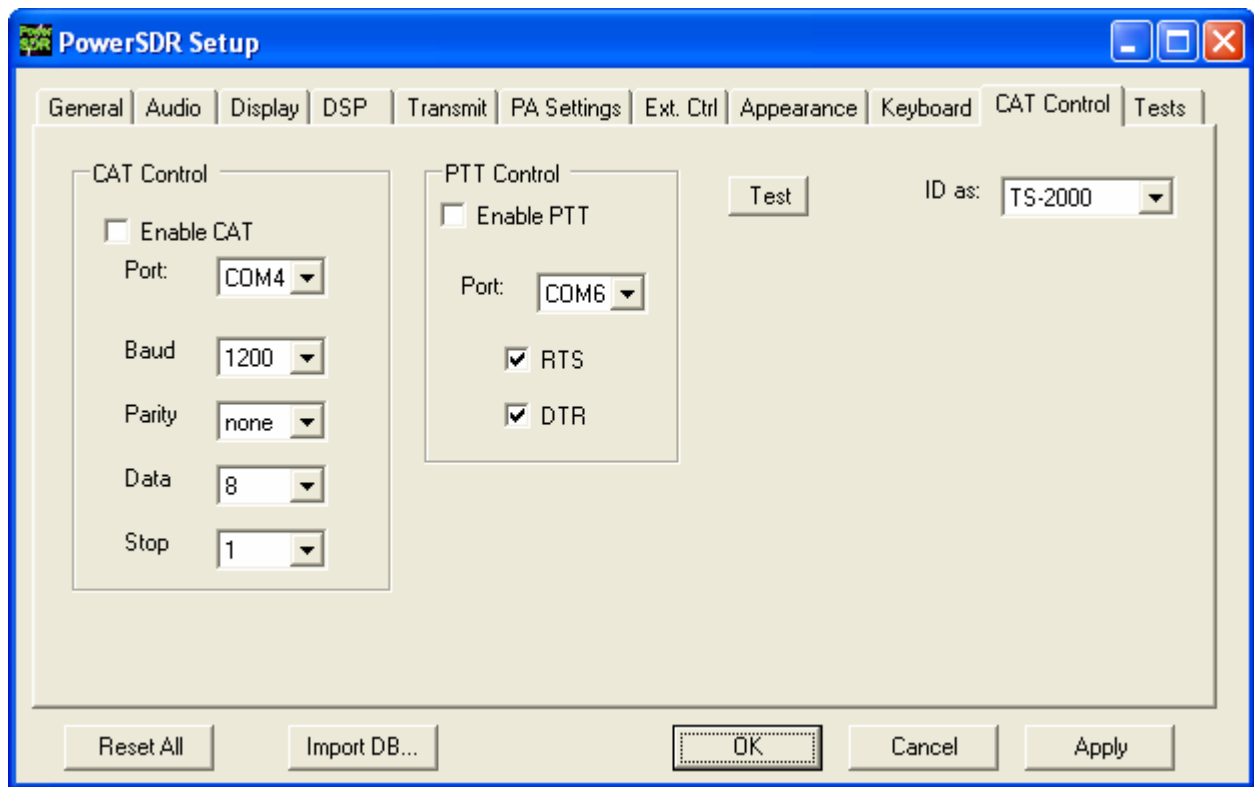
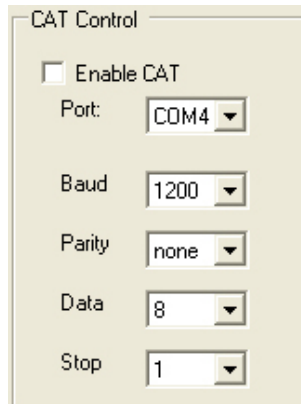


Figure 102: Setup Form - Cat Control Tab

The **CAT** (Computer Aided Transceiver) **Controls** enable the PowerSDR software to provide the user with a way to interface with third-party logging and remote control software. When used in conjunction with N8VB’s vCOM virtual serial port software, the possibilities for interaction with programs such as Ham Radio Deluxe (HRD), DXLab, N1MM Contest Logger, and numerous other software aids are realized.

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Cat Control



CAT Control

Enable CAT

Port: COM4

Baud: 1200

Parity: none

Data: 8

Stop: 1

Figure 103: CAT Control

- ❑ **Enable CAT:** Check this box to open the com port using the settings below. Note that you will need to uncheck this box in order to adjust the settings.
- ❑ **Port:** Com port number to use. If using N8VB's vCOM, note that you will use one end of the pair here and the other on the third party software (ex. HRD). The default ports are 4 and 5.
- ❑ **Baud:** The speed at which to transfer data.
- ❑ **Parity:** Sets whether to send a parity bit.
- ❑ **Data:** Sets how many data bits are sent with each byte.
- ❑ **Stop:** Sets whether to send a stop bit.

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PTT Control

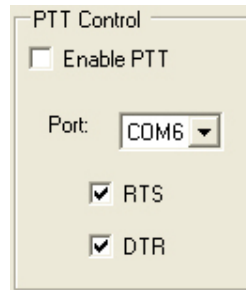


Figure 104: PTT Control

Some software utilizes a separate COM port than the CAT to activate PTT. These controls allow the user to configure this separate port to handle these signals.

- ❑ **Enable PTT:** Used to enable the hardware PTT Bit Bang.
- ❑ **Port:** The com port used for the PTT signal.
- ❑ **RTS:** Select this box to use the RTS line to engage PTT.
- ❑ **DTR:** Select this box to use the DTR line to engage PTT.

Test

Brings up the CAT Command Tester form as shown in the figure below. A valid CAT command may be entered in the CAT Command text box. Depending on the type of keyboard being used, the command will execute when a semicolon (the required CAT terminator) is entered, or when the “Enter” key is depressed, or when the “Execute” button is clicked. The CAT response will appear in the “CAT Response” text box.

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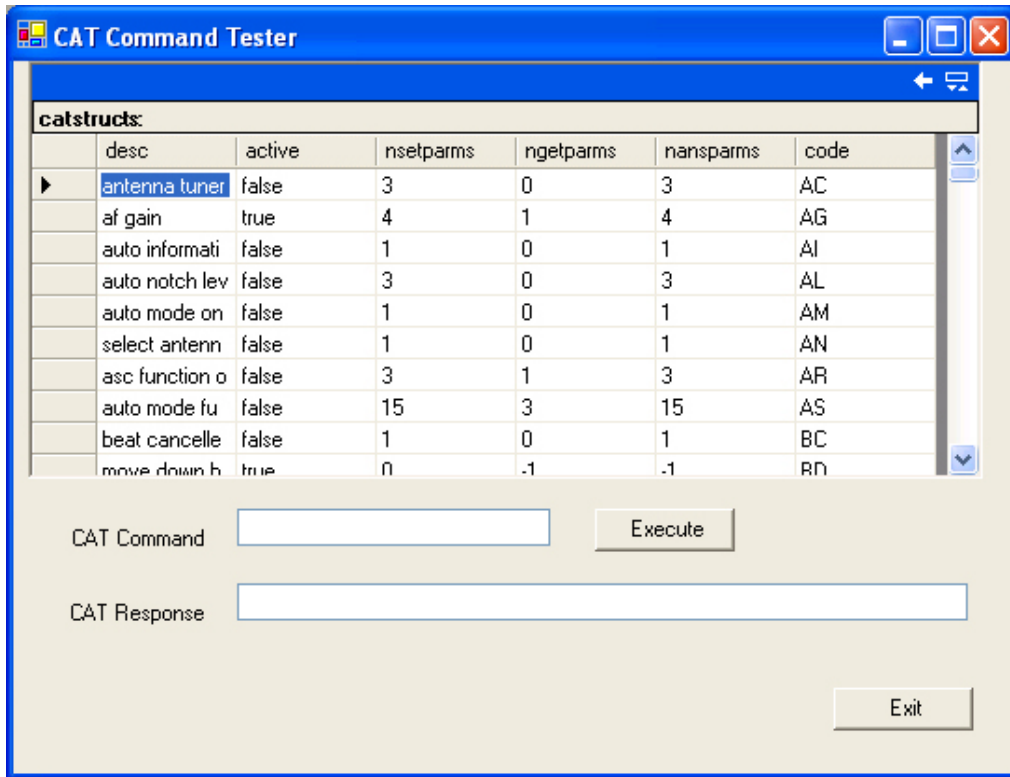


Figure 105: CAT Command Tester Form

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Tests Tab

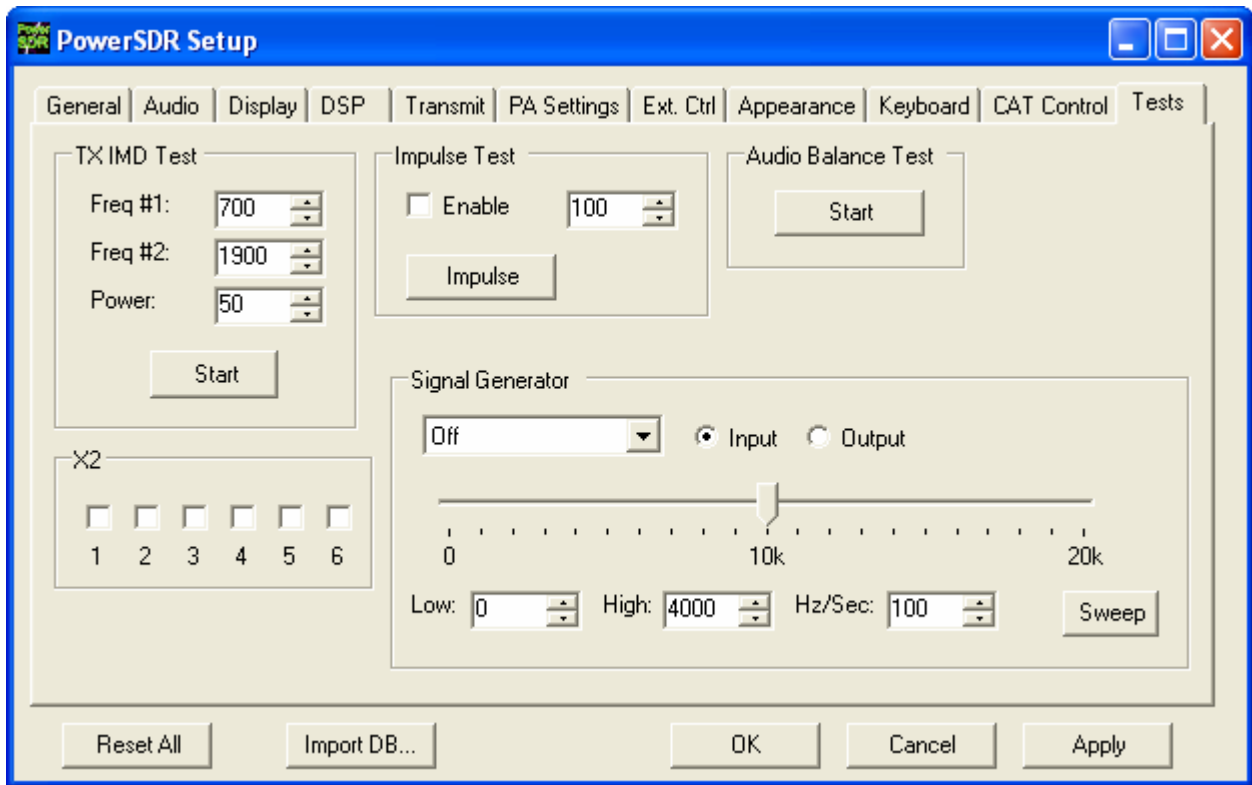


Figure 106: Setup Form - Tests Tab

TX IMD Test:

Used to test the two-tone IMD of the transmitter. To run the test, enter the two tone frequencies in the **Freq #1** and **Freq #2** controls. Enter the **Power** to be sent to the front panel PWR control. In practice, it generates equal amplitude tones at the entered frequencies. Click the **Start** button to begin sending a side-tone signal using the parameters entered. Manually adjust the Power control on this tab to set the tones to 6dB below PEP using a spectrum analyzer. Click the **Start** button again to stop the test and retrieve the PWR value. A single sweep function on a spectrum analyzer is an excellent tool to capture the output for analysis.

CAUTION: Do not leave the radio in two tone test mode for periods of longer than 15 seconds if using the power amplifier in order to avoid damage.

Audio Balance Test

Use this test to ensure that the Line Out cable is getting both output channels. The test will send a tone to each speaker and prompt to see if you hear audio in that one speaker. If you hear no audio or audio in both channels

during this test, then you may have a problem with the cable going to the plug labeled “To Line Out” on the enclosure.

X2

These controls were added in order to help test the Universal Controller Board (UCB). These will likely be removed once the UCB is completely integrated into the console.

Impulse Test

Used to test the Impulse response generator on the RFE board. Enter the number of iterations (impulses), click the Enable checkbox and then click the Impulse button to run the test.

Signal Generator

Used to generate signals for testing the PowerSDR software and connected hardware. Send whether the signal should be an input or an output and then select the type of signal. When finished, set the signal type to “Off” to allow the SDR-1000 hardware to once again be used.

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9

Operating Forms

This chapter describes each of the so-called operating forms. These forms are individually accessed by clicking on the relevant item to the right of Setup on the front console (see image below). For ease of reference, the numerical identifiers from the previous chapter on the Front Console are repeated in this chapter.



Figure 107: Operating Form Identifiers

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(31) Wave Form

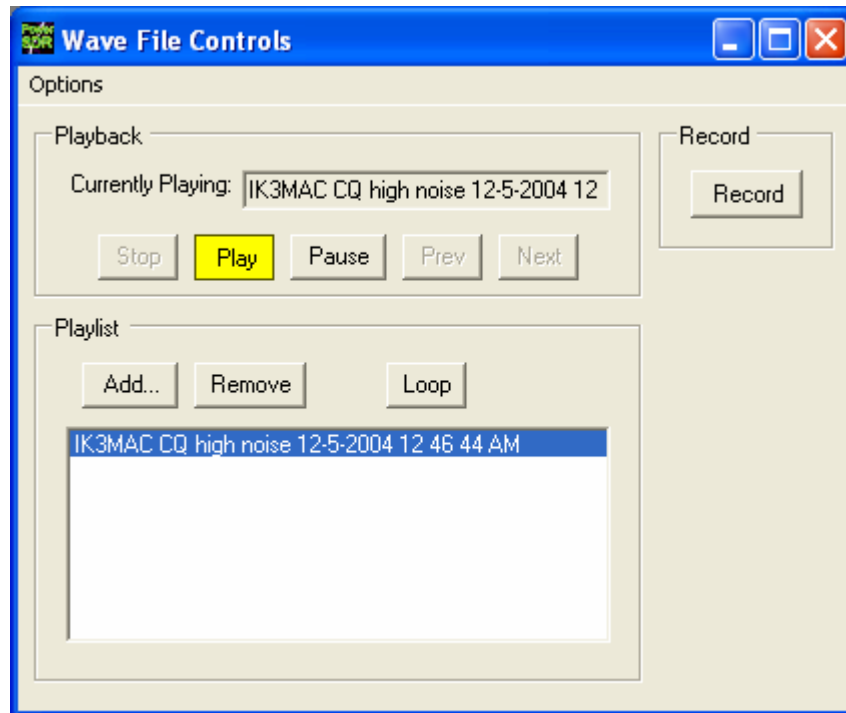


Figure 108: Wave Form

The wave form allows the user to record and playback up to 48kHz of bandwidth from the SDR-1000. To playback a file, simply click the Add... button in the playlist section and select the file. Then click the Play button while the power is on. To record the IF of the current station, click the record button to begin saving the data. When finished, press the record button again to complete the recording. The wave file will be date and time stamped automatically and saved in the default folder.

Playback

- ❑ **Currently Playing:** Displays the filename of the currently playing wave file.
- ❑ **Play:** Press this button to begin or stop playback of the current wave file. Note that clicking this button twice while a file is playing will restart the file.
- ❑ **Pause:** Pauses the wave file playback. Press once to pause and again to resume playback.
- ❑ **Prev (Previous):** When there is more than one file in the playlist, pressing this button will cause the previous file in the list to begin playing.
- ❑ **Next:** When there is more than one file in the playlist, pressing this button will cause the next file in the list to begin playing.

Playlist

- ❑ **Add:** Press this button to open a file menu to select wave file(s) to add to the playlist. Note that incompatible wave files will be removed from the list when they are played for the first time.
- ❑ **Remove:** Removes the currently selected file in the playlist. If the file is currently being played, then you will be prompted asking if you would like to stop playing the file and remove it from the list.
- ❑ **Loop:** When there is more than one file in the playlist, this option is enabled and allows playback to continue after finishing the last wave file in the list. At this point it will start playing the file at the top of the list.

Record

Press this button to begin recording a wave file. Press it again to complete the recording.

Options

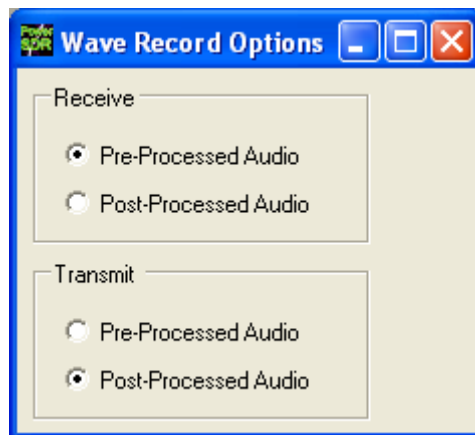


Figure 109: Wave Recording Options

The Wave Record Options can be used to modify what is recorded in receive and transmit modes.

Receive

- ❑ **Pre-Processed Audio** will record the whole 48kHz input. This is useful for playing back through the console at a later time.
- ❑ **Post-Processed Audio** will record just the filtered, AGC'd audio as you hear it coming out of the speaker. This is useful for playing back audio through a typical wav file player.

Transmit

- ❑ **Pre-Processed Audio** will capture the audio as it is seen at the microphone input without any filtering or compression.

- ❑ **Post-Processed Audio** allows the recording to capture the post-filtered data along with any compression or other features turned on for the transmit chain.

(32) Equalizer Form

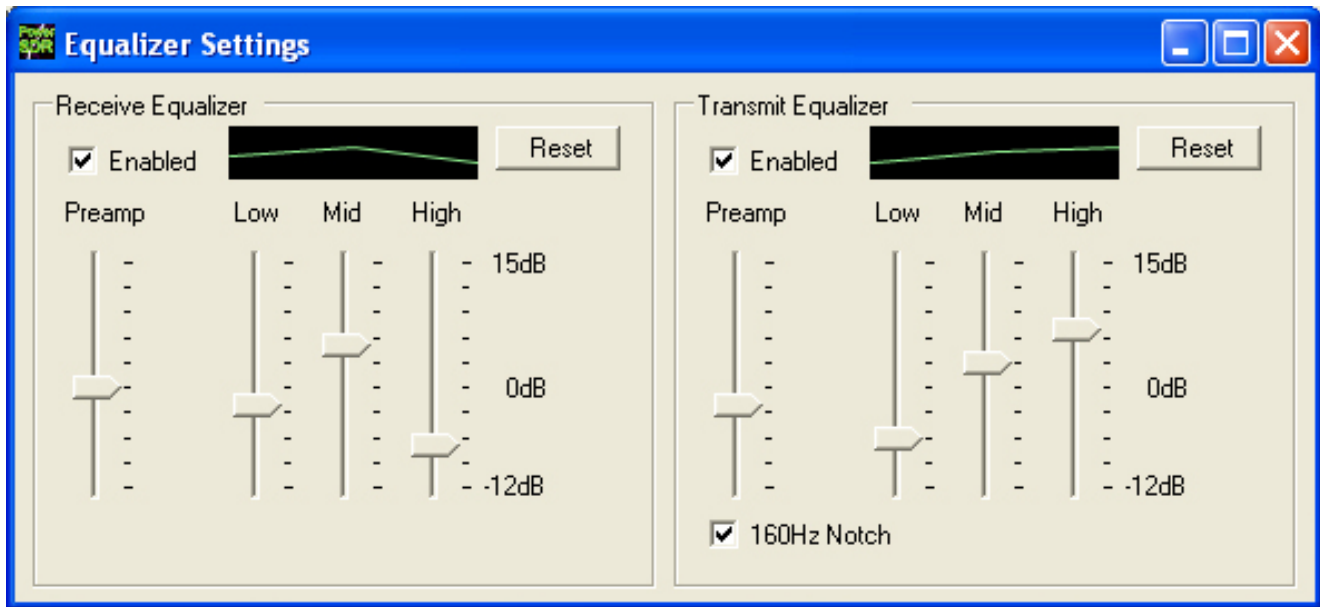


Figure 110: Equalizer Form

The Equalizer Form allows the receive and transmit audio to be modified by 3 band graphic settings. Use the sliders to adjust the gain for the frequency shown. The Preamp applies gain across the board. Easily compare the audio with and without the equalizer using the **Enabled** checkbox. The **Reset** button will reset all of the sliders to the 0dB position. For the TX Equalizer, there is an additional control that will turn on a 160 Hz notch filter. This gives about 12 dB of suppression at 160 Hz and is enabled only if TX equalizer is turned on. This is very fast in that it uses the same FFT Overlap Save approach used by the main filters.

(33) XVTRs Form

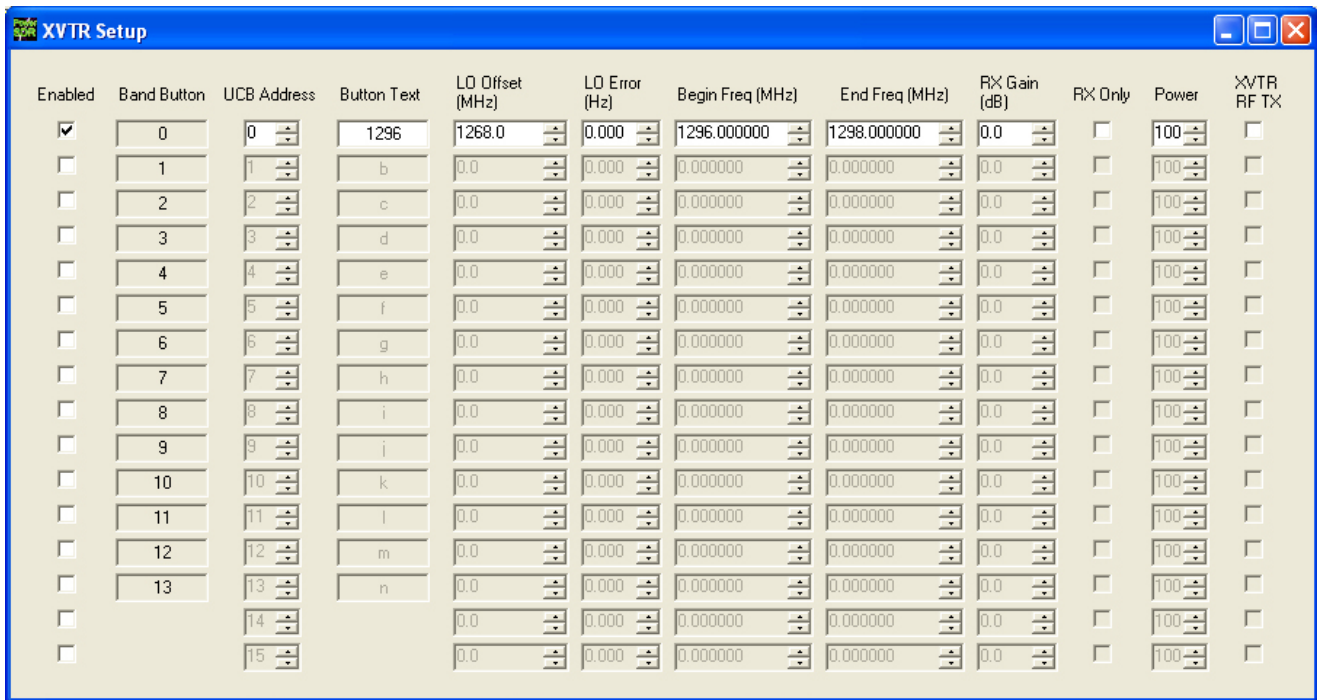


Figure 111: XVTR Setup Form

The XVTRs (transverters) Setup Form allows the user to configure up to 16 external transverters for use with the PowerSDR software.

- ❑ **Enabled:** Enables that particular Band Button with the options selected.
- ❑ **Band Button:** Band button to use for the particular transverter that is being configured.
- ❑ **UCB Address:** Used to set the X2 port for switching the external device (or a relay that controls the external device).
- ❑ **Button Text:** The text that will be shown on the Band Button.
- ❑ **LO Offset (MHz):** The difference between the transverter low frequency and the IF frequency. For example, on 2m, you might use 144-28MHz = 116.0.
- ❑ **LO Error (kHz):** This setting allows the user to correct for any error in the transverters oscillator.
- ❑ **Begin Frequency (MHz):** The lower frequency bound for the transverter.
- ❑ **End Frequency (MHz):** The upper frequency bound for the transverter.

- ❑ **RX Gain (dB):** Amount of gain to apply to the incoming signals to compensate for gain within the transverter.
- ❑ **RX Only:** If this box is checked, the radio will not transmit while in this configured band.
- ❑ **Power:** Sets the PWR control to this value whenever the VFO is within the configured band.
- ❑ **XVTR RF:** Determines whether to use the normal 1W RF output or the lower power XVTR output on the RFE board.

(34) CWX Form

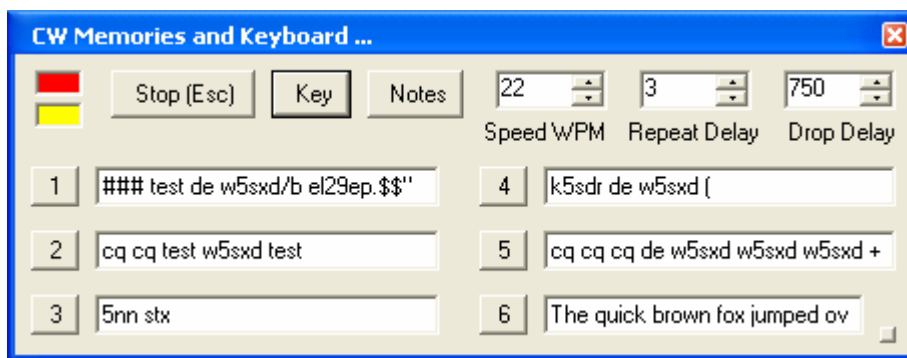


Figure 112: Standard CWX Form

The CWX form is designed to allow you to control automatic Morse code transmission and to send code from your computer keyboard. Pressing the **CWX** button on the main console menu opens this form. The radio must be in either **CWL** or **CWU** mode for transmission to take place.

Standard CWX Controls

- ❑ **Red Indicator:** The small red indicator shows when the radio is transmitting.
- ❑ **Yellow Indicator:** The yellow indicator shows when the key is being ‘pressed’ by the program.
- ❑ **Key:** Clicking this button will cause the radio to start transmitting a steady carrier for up to one minute. Clicking **Key** again will stop the steady sending.
- ❑ **Stop (ESC):** Clicking this button or the pressing the **Esc** key on your keyboard will terminate most functions and prepare the program to receive new orders or allow the iambic paddle to be used.
- ❑ **Notes:** Clicking this button will cause a small page of useful notes to be displayed. You can leave the notes up on the screen while you continue to work.

- ❑ **Speed WPM:** This control lets you set the speed of the Morse code being sent. The speed is computed by the standard PARIS method. The softness of the edges may be set by the **RAMP** control on the **Setup Form - DSP Tab, Keyer Sub-Tab**. The weighting is always 50% for the memory/keyboard keyer.

Note: This setting is separate from the control on the front console.

- ❑ **Repeat Delay:** This control specifies the amount of time that the keyer will wait when a special pause character is encountered.
- ❑ **Drop Delay:** This control specifies the amount of time that the semi-break in keying will wait before dropping the transmitter when there is no keying occurring.

CWX Memories

There are nine CWX memories, three of which are hidden on the extended form to the right (see below). Each memory can hold thousands of characters and in a standard single line text box. The numbered buttons to the left of each memory box may be clicked to start the message. A message may be stopped at any time by clicking the **Stop (Esc)** button or pressing the **Esc** key. Starting a message will seamlessly stop any current message or other automatically keyed transmissions from the radio, including the iambic paddle. The keyer will start the transmitter and send the Morse code for each character until the message ends, at which time the transmitter will shut down. Messages may be edited at any time but the changes will not take place until the memory is started again.

Special Characters

There are several predefined characters that provide non-Morse code functionality or to send familiar combinations like AR and SK (see table below). For beacon transmissions, the character # will send a long 23 element time dash comparable to a zero, with the key down the whole time. Multiple # characters can be strung together for longer continuous dashes. The \$ character works in a similar manner but generates a long 23 element time space. The “or ditto” character may be placed at the end of a message. When encountered, the keyer will shut down for the delay time set in the **Repeat Delay** control and then restart the message. This allows you to program a CQ and then listen for a reply with the radio back in receive. If the delay is set to zero then the message will simply repeat without the radio switching to receive. Six special combinations are preprogrammed, as shown in the table.

Table 10: Overview of Special Characters

| Special Character | Action |
|---------------------|--|
| # | Beacon - transmits 23 element "zero" time dash |
| \$ | Beacon - transmits 23 element time "space" |
| + | AR (. - . - .) |
| (| KN (- . - - .) |
| * | SK (... - . -) |
| ! | SN (... - .) |
| = | BT (- ... -) |
| \ | BK (- ... - . -) |
| & `) : ; < > [] ^ | User definable, up to any combination of 9 contiguous dots or dashes |

The remaining special characters & `) : ; < > [] and ^ are undefined and may be defined to produce any combination of nine contiguous dots and dashes. Characters that are undefined have no dots and dashes and are simply sent as a space.

Keyboard and Extended Controls

Click the little square button in the lower right corner of the form to expand it. When you do this, the remainder of the memories and controls will be visible including the keyboard window as shown below.

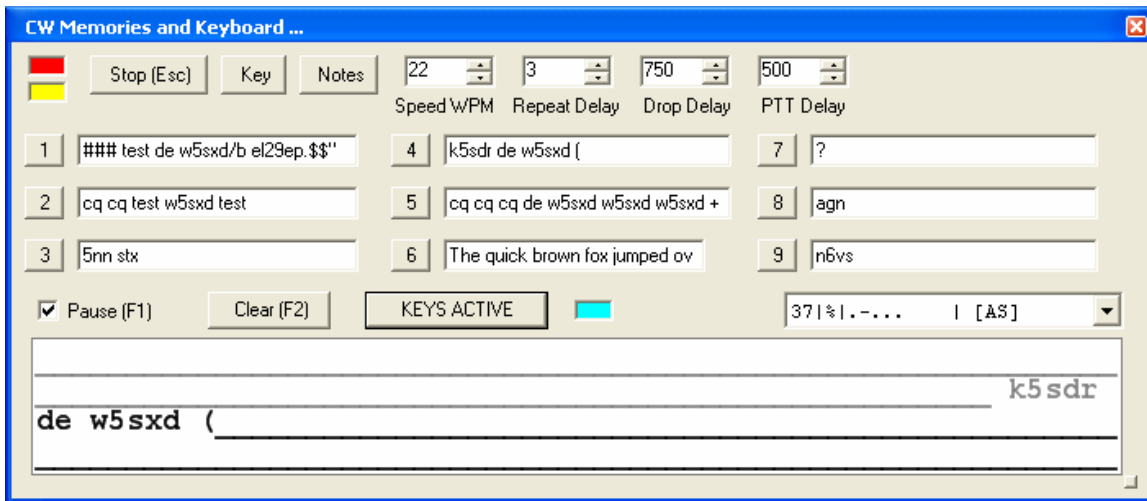


Figure 113: Extended CWX Form

Extended CWX Controls

- Keyboard area: the four-line text box at the bottom. Unsent characters are shown in black and sent characters in gray.

- ❑ **KEYS ACTIVE.** Clicking this button will enable you to use the keyboard to enter characters at the end of the unsent area. As soon as a character is typed it will be sent and then moved to the unsent area. If you type faster than the code is being sent, it will be buffered in the bottom area. **cyan indicator** shows that **Keys Active** is enabled.
 - The **Backspace** key will work in the unsent area.
 - Other editing keys like cut and paste are not implemented.
 - Pressing **Alt 1** to **Alt 9** or right clicking on the message number button will cause the numbered message memory to be copied into the unsent area just as if you had typed it.
 - The ditto character is ignored in the keyboard mode.
- ❑ **Pause (F1):** Clicking this button or the pressing the **F1** key will cause keyboard buffer sending to pause.
- ❑ **Clear (F2):** Clicking this button or pressing the **F2** key will clear the keyboard area and stop it sending. Of course, the **Stop (Esc)** button or the **Esc** key will do the same.
- ❑ **PTT Delay:** This control allows you to set the time delay between switching the radio to transmit and the first key closure.

Morse Definition Editor

The Morse definition editor allows you to define and even redefine almost all of the characters in the sixty-four-character set. The combo box control to the right of the **cyan indicator** lets you view and select any of the characters in the set. Each one is displayed as four fixed width fields separated by the | character. There are five special control characters that you may not change and they have an * in the element field.

Once you have selected the character that you wish to edit, left click to select and then right click to bring up the editor dialog.

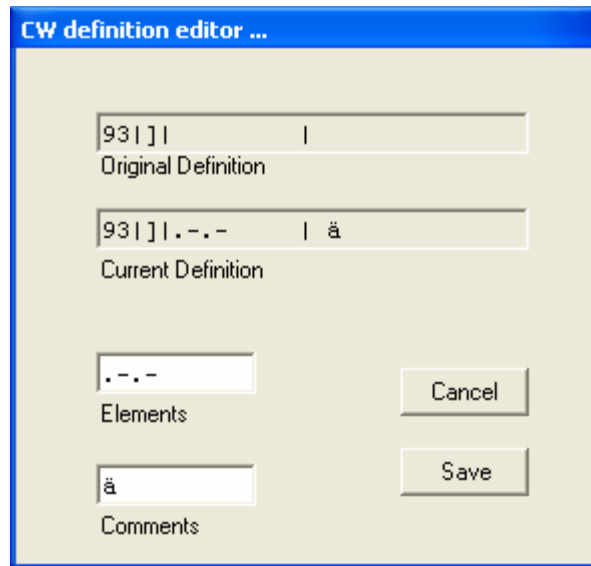


Figure 114: CW Definition Editor

The editor dialog example above shows the ‘]’ character being changed to send didahdidah which is the German code for **ä** (umlaut a). When your definition appears to be correct, click the **Save** button and the definition file will be resaved to the disk.

- ❑ The definition file is called **morsedef.txt** and can be found in the same folder as **PowerSDR.exe** and **PowerSDR.mdb**. It will not be automatically carried from one version to the next so if you make many changes you might want to save a copy elsewhere.
- ❑ The **morsedef.txt** file can be manually edited with a simple character editor like notepad (not Word), but the format must be followed *exactly*. Each line must be 26 characters long, not including the two end-of-line codes. It is not free-formatted. The line structure is: a two-digit number field (the decimal ASCII code) | a one-character code field | a nine character elements field | a ten-character comment field preceded by a space and followed by a carriage return and linefeed code. If you mess this file up too badly, simply close PowerSDR, delete **morsedef.txt**, and a clean, default copy will be created the next time you start. The editor makes simple changes relatively easy to do.

10

Operation

How to use the SDR-1000 in a typical operating environment.

This chapter is intended to provide the user with a clear understanding of how the SDR-1000 should be used when performing basic operations such as Powering Up or making a voice transmission. We chose to use this section in this way in lieu of listing all of the features of the radio since the feature list would essentially repeat the information given in the three preceding chapters. Please refer to those chapters for complete feature descriptions and how to use the controls.

Note 1: For consistency we have will use the same control identifiers as in the preceding Front Console chapter and for clarity we will leave out any identifiers of controls not referenced in the relevant section

Note: The procedures outlined in this chapter assume that the calibration routines outlined in the Calibration section of the Setup Form chapter above have already been carried out.

Power-Up Procedure

The SDR-1000 is unique in that the software does not have a way to cut the 13.8V power to the hardware unit. For this reason, we recommend the following procedure to ensure that the radio starts in a stable, safe state on Power Up:

1. Connect the SDR-1000 to the sound card, parallel port (or USB port if you are using the USB to parallel adapter), external key and power supply. Check the connections to ensure good contact.
2. Boot up the computer and make sure the PowerSDR software is ready to be launched (no hour glass cursor).
3. Turn on the power supply for the radio and flip the SDR-1000 Enclosure switch to the “On” (up) position. You should hear relays click as they receive power. The current reading before starting the software should be in the 1A range (possibly more depending on installed options).

4. Start up the PowerSDR software. You should hear the relays switch as they are reset to the “Stand By” setting. Make sure this step takes place fairly soon after step three as the relays should not be left in the higher current setting for long periods of time. The current reading after starting the software should be around 700mA.

Power-Down Procedure

The power-down sequence is almost the reverse of the power-up procedure.

1. Close the PowerSDR software by clicking on the “X” in the upper right hand corner of the Front Console. This will put the radio in the low power “Stand By” state.
2. Flip the SDR-1000 Enclosure power switch to the “Off” (down) position and turn off the power supply. Note that it is necessary to cut the power at the supply in order to completely power down the radio and ensure the amplifier circuitry is not in a bias state.

Always power down and disconnect the radio before running any other software that requires the use of the parallel port to prevent sending unintended settings to the radio. Such signals could cause damage to the unit.

Tuning Methods

Spectrum Drag and Click

The easiest way to tune signals on the display when in Panadapter display mode is to simply left click on the signal, and drag it into the filter area. You can also drag the displayed filter edges to adjust the filter width as well.

Mouse Wheel

A mouse wheel is the next easiest way to tune the radio. Click the mouse wheel or press Ctrl + Left or Right Arrow key to increase or decrease the tuning rate (shown in the upper left corner of **VFO A**). While the PowerSDR window is active, tune **VFO A** using the mouse wheel, with the mouse cursor anywhere on the screen. The frequency will change in steps equal to the selected tuning rate for each click of the mouse wheel.

Note: When the tuning rate equals 1kHz, tuning will snap up or down to the nearest 1kHz and will change in 1kHz steps with each click of the mouse wheel. E.g. if **VFO A** is tuned to say 14.000258MHz then with the tuning rate set to 1kHz, using the mouse wheel to tune up will first increase **VFO A** to 14.001000MHz and then to 14.002000MHz, and so on.

Hint: Holding down the Shift key while turning the mouse wheel will change **VFO A** at the next lower rate. E.g. if the tuning rate is set to 1kHz, then turning the mouse wheel while holding down the Shift key will change **VFO A** in 100Hz steps.

Mouse Wheel Hover

Hover with the mouse over a digit in either **VFO A** or **VFO B** and increase or decrease its value using the mouse wheel. The digit to be tuned will be underlined.

Spectrum Click Tuning

With the display set to Spectrum, Panadapter, Waterfall or Histogram, hover with the mouse over the display and right-click to cycle through yellow cross-hairs (tune **VFO A**), red cross-hairs (tune **VFO B**) or no cross-hairs (click tuning off). With the cross-hairs visible, hover over the desired signal in the display and click the left mouse button. The VFO will be immediately tuned to the frequency of the selected signal. When in the CW, AM, or FM modes, the VFO will tune the cursor frequency to the center of the filter pass band. In SSB modes, the VFO will tune to the carrier frequency for the sideband selected.

Hint: You can very quickly center a CW, AM or FM signal after click tuning it by clicking the **0 Beat (11)** button on the Front Console (make sure the display **AVG (27)** button is on to enable **0 Beat**).

Keyboard Keys

The following methods are available to use your keyboard to tune the VFO.

- ❑ **Numeric Keypad:** Key any frequency in MHz (e.g. 7.250) into the numeric keypad and hit enter to immediately jump to that frequency.
- ❑ **Mapped Keys:** You can map keys on your keyboard to tune each of the digits in **VFO A** using the **Keyboard Tab** on the **Setup Form**.
- ❑ **Ctrl +Arrow Keys:** Hold the Control key and press the
 - Up or Down Arrow key to tune **VFO A** up or down at the tuning rate (displayed in the upper left corner of **VFO A**).
 - Right or Left Arrow key to increase or decrease the tuning rate respectively.

USB Tuning Knob

Both the Griffin PowerMate and the Contour Designs Shuttle Pro v2 can be used to tune the radio. See the download page on the FlexRadio website for documents describing how to setup and use these controllers.

Voice Transmission Operation

The following procedure outlines how to quickly setup for voice transmission operation (SSB, AM, or FMN). If something in this procedure is unclear, please contact us, as we would like this to be as simple as possible.

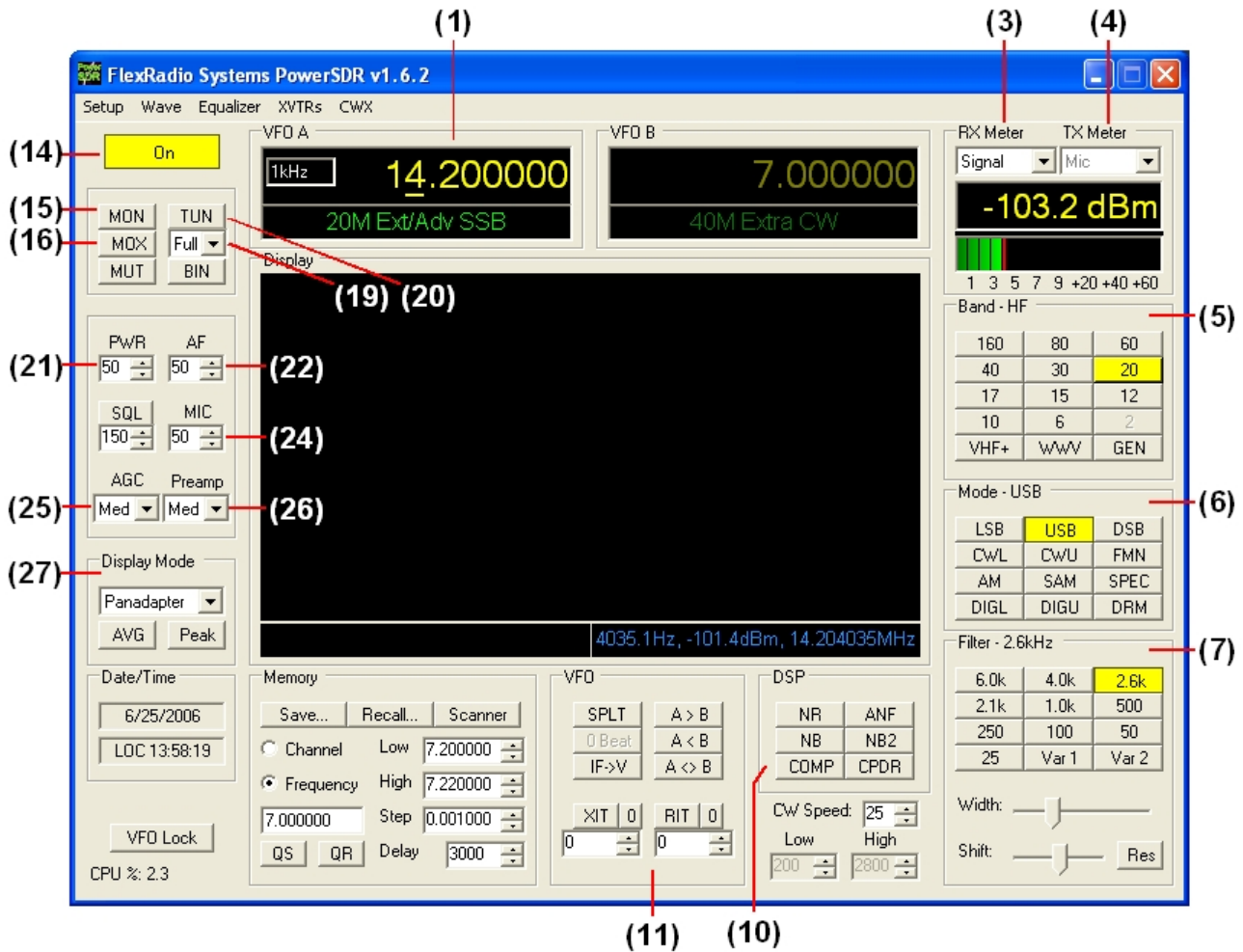


Figure 115 Front Panel Controls Used for Basic Voice Operation.

Please use the above figure to identify the controls referenced in the following step-by-step instruction.

1. Follow the Power Up Procedure, described above. Then click the **On** button (14). Set the following controls as specified. (These are the default settings, so if you have not changed them the settings will be correct).

Table 11: Initial Control Values for Voice Operation

| Ctrl | Value | Reference to Figure Above |
|---------------|------------|---------------------------|
| VFO A | 14.2MHz | (1) |
| RX Meter: | Signal | (3) |
| TX Meter: | MIC | (4) |
| BAND: | 20 | (5) |
| Mode: | USB | (6) |
| Filter: | 2.6kHz | (7) |
| COMP: | Off | (10) |
| CPDR | Off | (10) |
| XIT: | Off | (11) |
| RIT: | Off | (11) |
| PWR: | 50 | (21) |
| AF: | 50 | (22) |
| MIC: | 50 | (24) |
| AGC: | Med | (25) |
| Preamp: | Med | (26) |
| Display Mode: | Panadapter | (27) |

2. Connect a 50 ohm dummy load or tune **VFO A (1)** to a quiet frequency on the selected band. If using the 100W PA, use the **TUN** button **(20)** to verify power output on the TX Meter (Set to Fwd Pow). If using the integrated ATU, select **Full** on the ATU Mode selection **(19)** and then click the **TUN** button **(20)**. See the section below on ATU Operation for more on how to use the ATU.

CAUTION: If not using the ATU and not using a 50 ohm dummy load, ensure that the antenna presents a 50 ohm load with a low SWR or damage may occur to the SDR output transistors.

3. Press the Push-to-talk button on the microphone or click the **MOX** button **(19)** and speak into the microphone element to transmit your voice. Release the Push-to-talk button or click the **MOX** button **(16)** to switch the transceiver back to receive. If you do not see modulation on the spectrum, please reference the Troubleshooting chapter below.
4. Now that you can see the modulation on the spectrum, it is time to adjust the input. While transmitting, monitor the values with the **TX Meter (4)** set to **Mic**. Modify the **MIC (24)** setting until the **TX Meter** shows 0dB on peaks while talking in a normal voice at a normal distance from the microphone (above 0dB the signal will be compressed).
5. You are now ready to begin a QSO. If a 50 ohm dummy load was connected, connect an antenna in its place. Tune to the desired frequency using one of the methods outlined in the Tuning Methods section above.

6. Use the controls in the **Transmit Tab** of the **Setup Form** to adjust the transmitted audio further, including **Transmit Filter**, **Noise Gate**, **Transmit Compression** and **Comband**. Note that the compressor and compander may be enabled on the front panel (10). Generally, use the compressor to increase average power without adjusting the peaks. Detailed explanations of each of these transmit controls is included in the Setup Form chapter above.

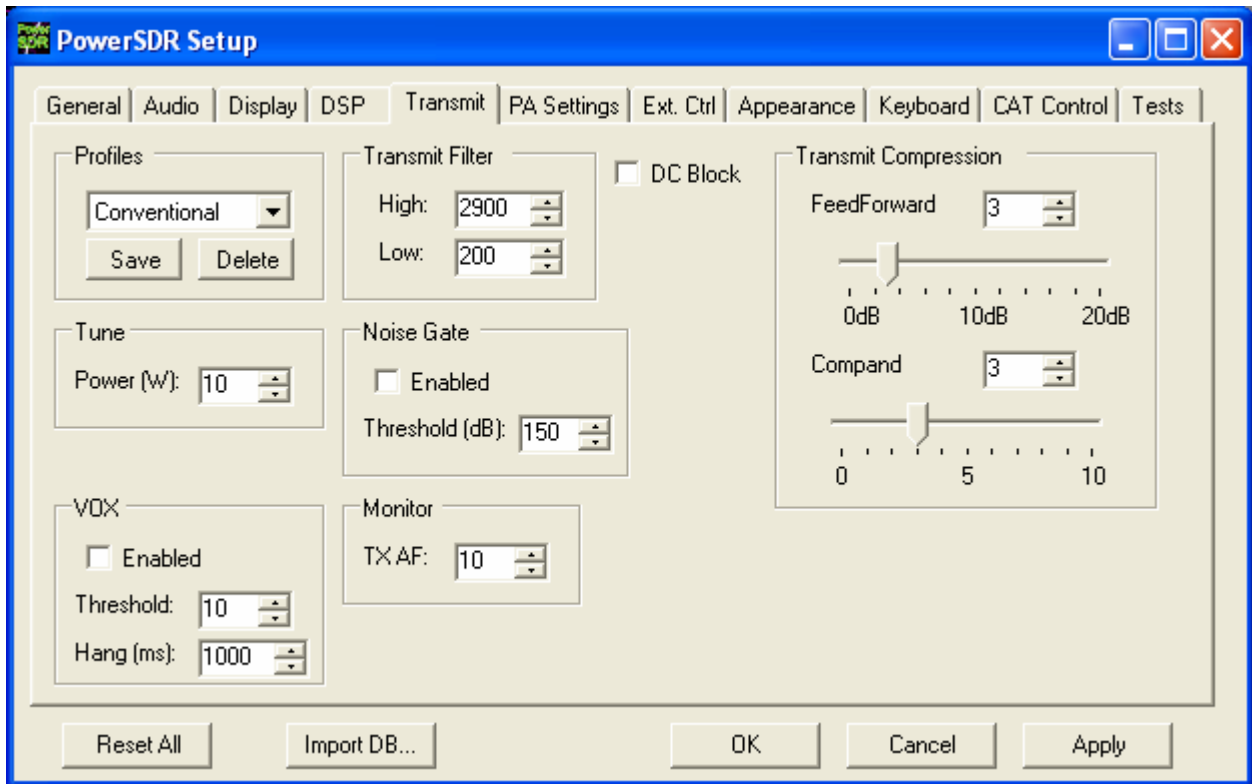


Figure 116: Setup Form Transmit Tab

Hint: For information on how to optimize your audio further, please refer to the Expert Setups page on our website.

7. In order to monitor voice transmissions, enable the **MON** button **(15)**. You may notice a delay due to buffering in the audio/DSP system. This processing delay is normal and is largest when using large buffer sizes. If you find this delay objectionable, try reducing the **Buffer Size** setting on the **Audio Tab** and/or **DSP Tab** of the **Setup Form** to decrease the delay. See the Setup Form chapter above for complete details of the ramifications of changing these controls.

Note: The delay is lengthened by ASIO4ALL and its external buffering outside of the PowerSDR software. Some settings on the ASIO4ALL control panel can be used to minimize the effective delay. Please refer to the ASIO4ALL manual for more info. The recommended M-Audio Delta 44 and PreSonus FireBox sound cards with their native ASIO drivers do not have this additional ASIO4ALL delay.

8. The **Fwd Pow** setting on the **TX Meter** will read out average power in Watts according to the DSP or the PA ADC (if the PA is installed). While the average has a short time constant, it is still an average and will not approach 1W (or 100W) in voice modes if calibrated properly even when the **PWR** control **(21)** is set to 100. This is also true when monitoring voice transmissions on an external wattmeter.

Note: The typical male voice has a peak to average power ratio of 14dB. Therefore a typical male voice transmission that is peaking at 100W will only average less than 10W. To raise the average power, use the **COMP** control **(10)** and the associated **FeedForward Compression** slider on the **Transmit Tab** of the **Setup Form** to increase the compression in 1dB steps. This must be done carefully and incrementally as adding too much compression can result in high levels of distortion. For further reading on the commonly misunderstood subject of peak versus average power, please see the following links:

<http://www.flex-radio.com/forums/viewtopic.php?p=6376#6376>

<http://www.arrl.org/tis/info/pdf/9505088.pdf>

CW Transmission Operation

The following procedure outlines how to quickly setup for CW transmissions using the Internal Keyer and paddles, an external keyer, the CWX-form or a third party program. For the latter, we will use MixW as an example.

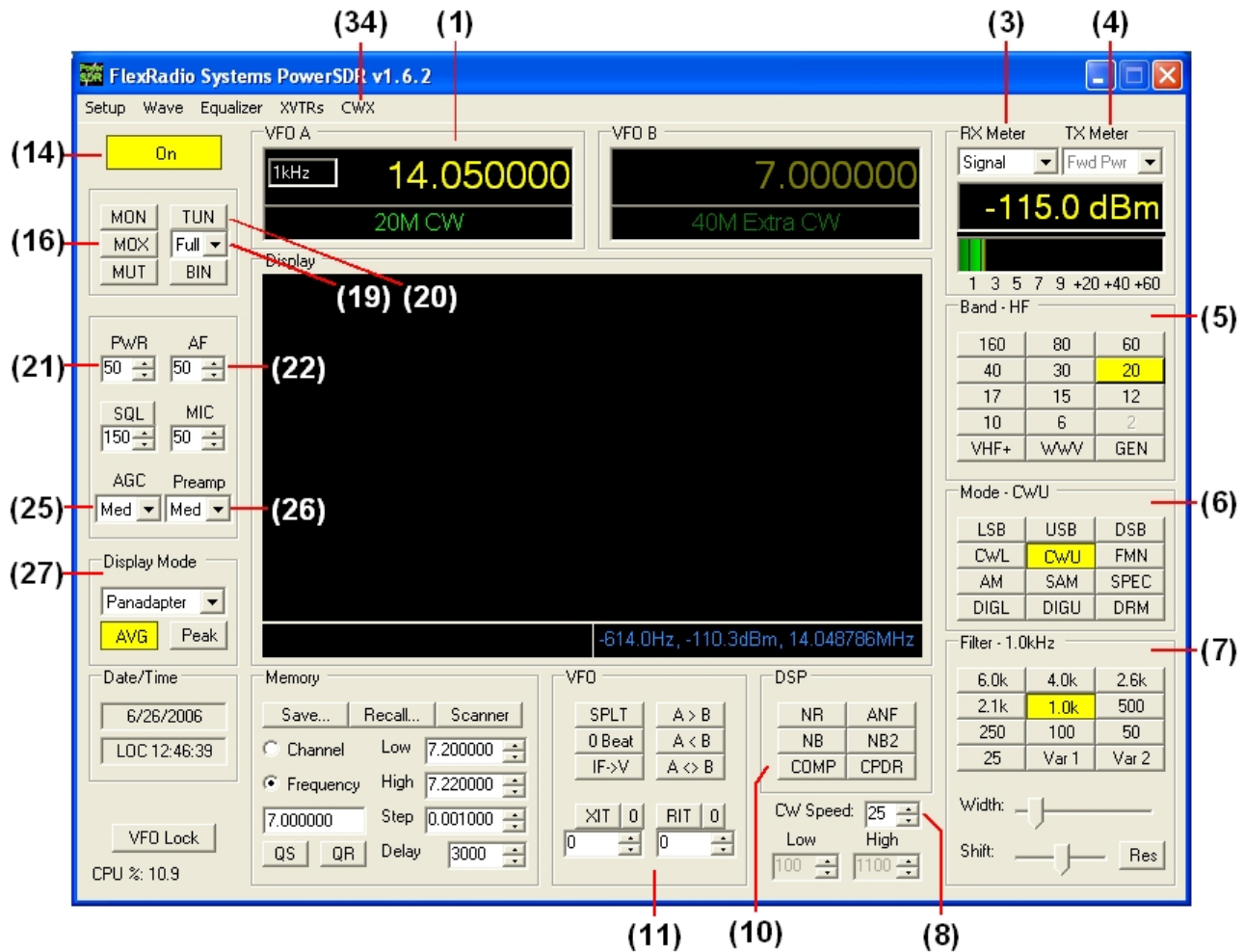


Figure 117: Front Panel Controls Used for CW Operation

Please use the above figure to identify the controls referenced in the following step-by-step instruction.

Initial Settings

1. Follow the Power Up Procedure, described above. Then click the **On** button (14). Set the following controls as specified.

Table 12: Initial Control Values for CW Transmission

| Ctrl | Value | Reference to Figure Above |
|---------------|------------|---------------------------|
| VFO A | 14.2MHz | (1) |
| RX Meter: | Signal | (3) |
| TX Meter: | Fwd Pwr | (4) |
| Band: | 20m | (5) |
| Mode: | CWU | (6) |
| Filter: | 1.0kHz | (7) |
| COMP: | Off | (10) |
| CPDR: | Off | (10) |
| XIT: | Off | (11) |
| RIT: | Off | (11) |
| PWR: | 50 | (21) |
| AF: | 50 | (22) |
| AGC: | Med | (25) |
| Preamp: | Med | (26) |
| Display Mode: | Panadapter | (27) |

2. Connect a 50 ohm dummy load or tune **VFO A (1)** to a quiet frequency on the selected band. If using the 100W PA, use the **TUN** button **(20)** to verify power output on the TX Meter (Set to Fwd Pow). If using the integrated ATU, select **Full** on the ATU Mode selection **(19)** and then click the **TUN** button **(20)**. See the section below on ATU Operation for more on how to use the ATU.

CAUTION: If not using the ATU and not using a 50 ohm dummy load, ensure the antenna presents a 50 ohm load with a low SWR or damage may occur to the SDR output transistor.

Hint : Adjust the Audio and Sound Card (Delta 44, Firebox) Latency/Buffer sizes as low as possible (e.g. 512 for the Delta 44 DMA buffer or 5ms for the FireBox latency) and the sampling rate to at least 96000 to minimize latency. Adjust the DSP Buffer size as high as possible (e.g. 2048) for maximum filter sharpness

Internal Keyer

If using PowerSDR's internal keyer, open the **Setup Form – DSP Tab, Keyer Sub-Tab** shown below.

[The rest of this page intentionally left blank]

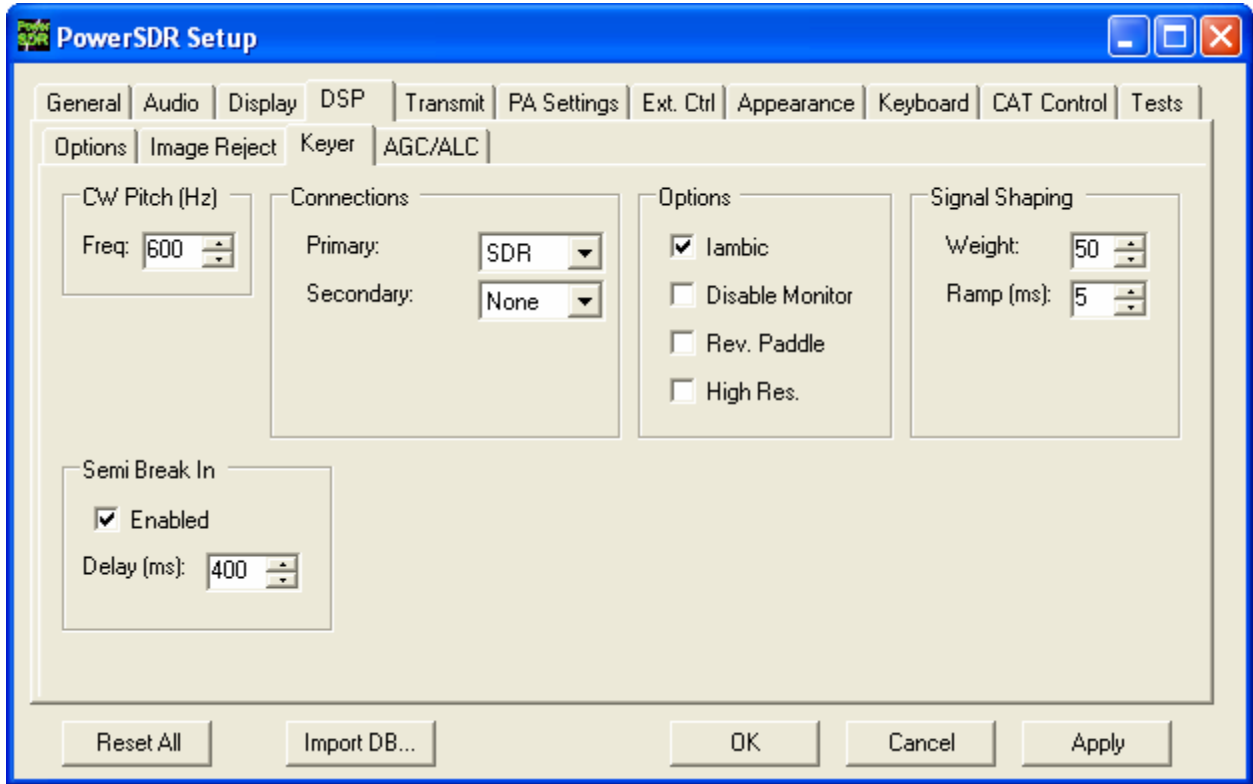


Figure 118: Setup Form – DSP Tab, Keyer Sub-Tab

1. If you have your paddles or external keyer connected to a COM port, set the **Primary Connection** to this **COM** port. Set the **Primary Connection** to **SDR** if you have your paddles or external keyer connected to the SDR keyer input plug (see preceding Plug Pin-Outs chapter).

Hint: For best CW performance, connect your paddles to a serial port using the following table to make the connections.

Table 13: Serial Port CW Pinout

| Pin | Connection |
|---------|------------|
| 4 (DTR) | Common |
| 6 (DSR) | Dot |
| 8 (CTS) | Dash |

2. In the **Options** section, check **Iambic** for Iambic mode or leave unchecked for a straight key. If the paddles seem reversed, check **Rev. Paddle**. The **Semi Break In** option allows the radio to be put into transmit mode simply with detection of keyer input. Set **Delay** to the amount of time between the last detected input and when the radio will switch back to receive.

3. Select the settings for **CW Pitch**, **Weight** and **Ramp** as desired (refer to the **Keyer Sub-Tab** section of the preceding Setup Form chapter for more detail).
4. If not using **Semi Break In**, click **MOX (16)** on the Front Console and begin transmitting using your paddles. If using **Semi Break In**, simply begin transmitting to key the radio. If using Iambic mode, adjust the speed with **CW Speed (8)** on the Front Console. If **Disable Monitor** is unchecked, you should hear the side tone. Verify with the **TX Meter (4)** set to **Fwd Pwr** that there is forward power.
5. You are now ready to begin a QSO. If a 50 ohm dummy load was connected, connect an antenna in its place. Tune to the desired frequency using one of the methods outlined in the Tuning Methods section above. Select either **CWL** or **CWU (6)** and proceed with the QSO.

External Keyer

1. If you have your external keyer connected to a COM port, set the **Primary Connection** to this **COM** port. Set the **Primary Connection** to **SDR** if you have your external keyer connected to the SDR keyer input plug (see preceding Plug Pin-Outs chapter).

Hint: For best CW performance, connect your external keyer to a serial port. See the table above on how to make the connections.

2. In the **Options** section, leave **Iambic** unchecked. If the your external keyer seems reversed, check **Rev. Paddle**. If using an external keyer with a sidetone, check the **Disable Monitor** checkbox to avoid hearing the side tone from the internal keyer. The **Semi Break In** option allows the radio to be put into transmit mode simply with detection of keyer input. Set **Delay** to the amount of time between the last detected input and when the radio will switch back to receive.
3. If using the sidetone from the internal keyer, select the settings for **CW Pitch**. Select the settings for **Weight** and **Ramp** as desired (refer to the **Keyer Sub-Tab** section of the preceding Setup Form chapter for more detail).
4. If not using **Semi Break In**, click **MOX (16)** on the Front Console and begin transmitting using your external keyer. If using **Semi Break In**, simply begin transmitting to key the radio. If **Disable Monitor** is unchecked, you should hear the side tone. Verify with the **TX Meter (4)** set to **Fwd Pwr** that there is forward power.
5. You are now ready to begin a QSO. If a 50 ohm dummy load was connected, connect an antenna in its place. Tune to the desired frequency using one of the methods outlined in the Tuning Methods section above. Select either **CWL** or **CWU (6)** and proceed with the QSO.

CWX Form

If you wish to send CW automatically, click on **CWX (34)** on the Front Console menu to open the CWX Form shown below. (Refer to the **CWX Form** section of the preceding Operating Forms chapter for more detail on how to use this form).

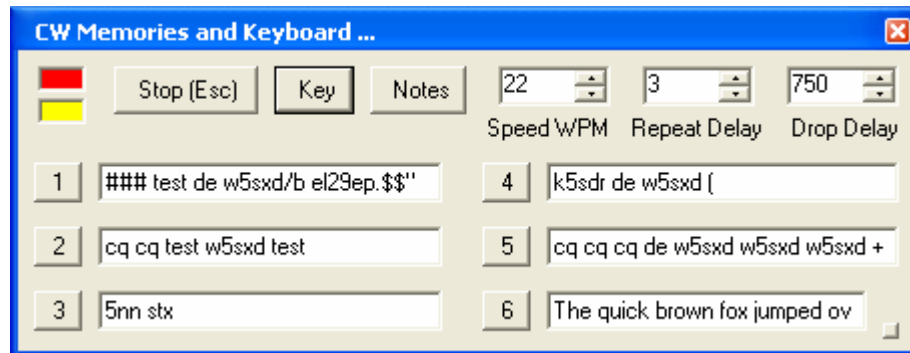


Figure 119: CWX Form

1. Click on one of the numbered buttons to start transmitting the corresponding CW sequence. Verify with the **TX Meter (4)** set to **Fwd Pwr** that there is forward power.
2. You are now ready to begin a QSO. If a 50 ohm dummy load was connected, connect an antenna in its place. Tune to the desired frequency using one of the methods outlined in the Tuning Methods section above. Select either **CWL** or **CWU (6)** and proceed with the QSO.

Third Party Program

To operate CW with a third party program, you will need to download and install N8VB's vCOM driver to create a virtual COM port pair through which PowerSDR can be connected to your third party program. The installation and setup of vCOM is described below in the Digital Mode Operation section of this chapter.

In the following we will use MixW as an illustrative example and we will assume the COM6-COM16 virtual COM port pair. We will also assume that PowerSDR is connected to COM6 and MixW to COM16 of this pair (see the Digital Mode Operation section below for details on how to do this)

1. In PowerSDR open the **Setup Form – DSP Tab, Keyer Sub-Tab** shown below to access the Internal Keyer controls

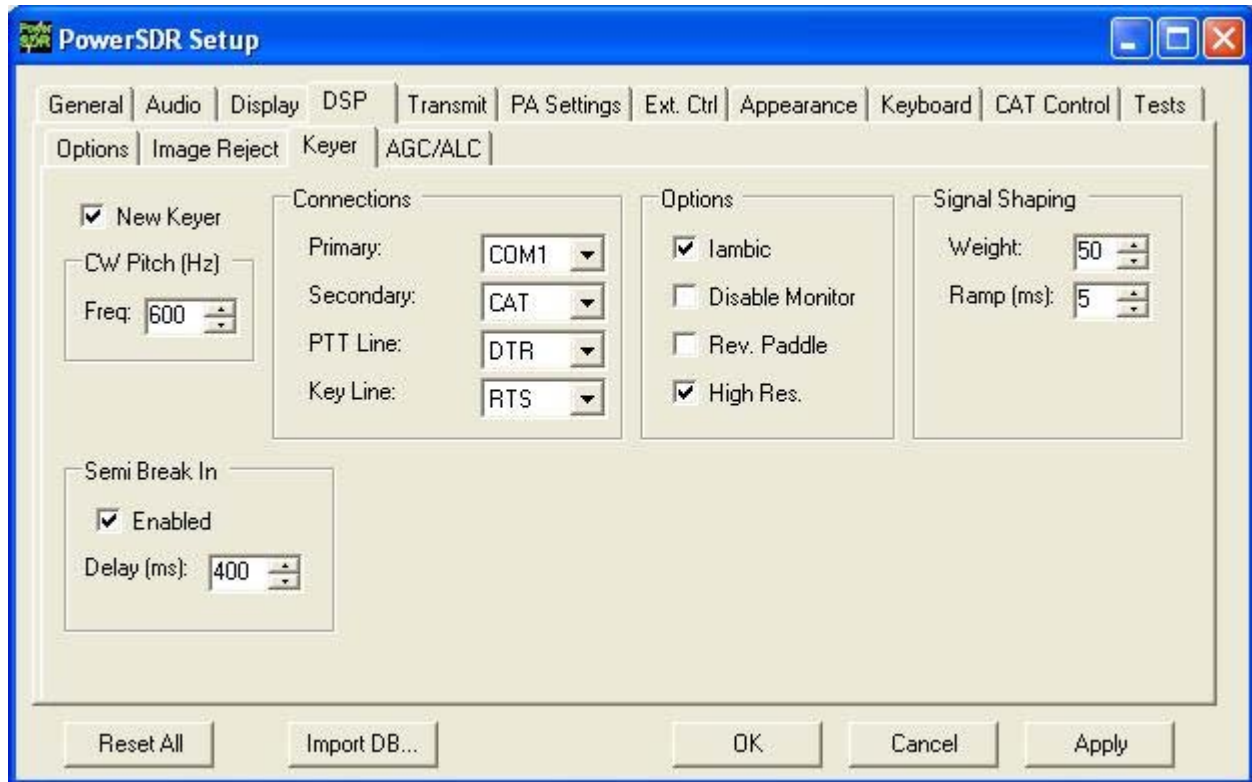


Figure 120: Setup Form – DSP Tab, Keyer Sub-Tab

2. Set **Secondary Connection** to **CAT** to use the same virtual COM port COM16, as selected on the **CAT Control TAB**. This will open up two additional selection boxes. Set **PTT Line** to **DTR** and **Key Line** to **RTS** as shown above.
3. In MixW, click **Configure** on the Menu bar and then select **TRCVR CAT/PTT** to open the following screen.

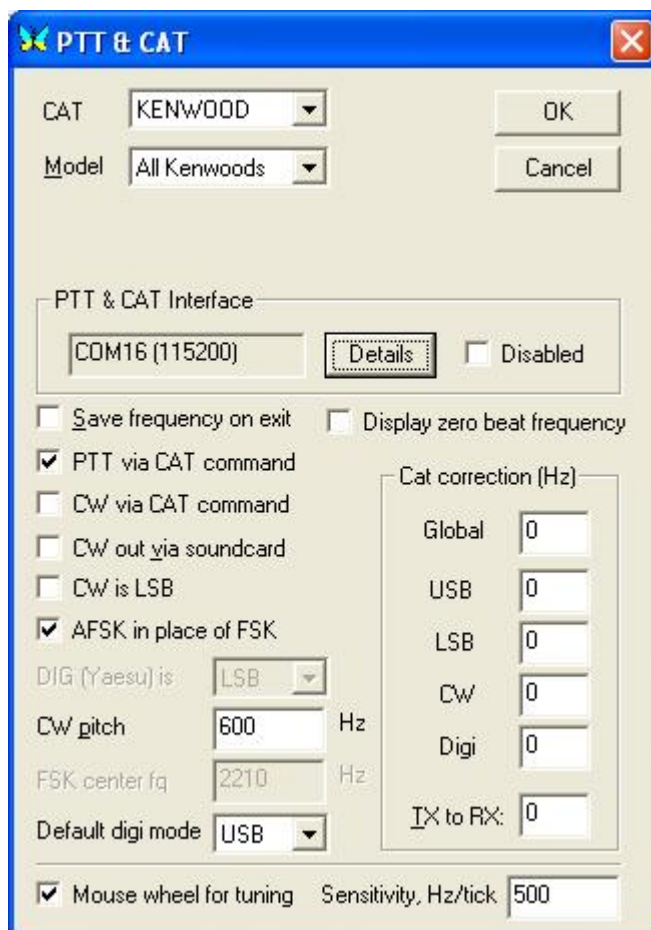


Figure 121: MixW PTT & CAT

4. On the **PTT & CAT Interface**, click on the **Details** button to open the following form:

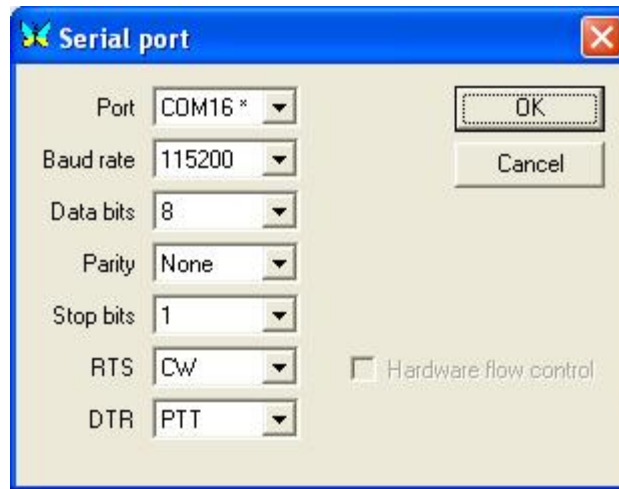


Figure 122: MixW Serial Port Details

4. Set the **Port** to **COM16**, **RTS** to **CW** and **DTR** to **PTT**. Click OK on this form and the previous form.
5. Change the mode in MixW to CW. Click on the **TX** button on MixW's main panel. It should key PowerSDR without generating a tone. Click **RX** in MixW and PowerSDR should return to receive. If you have entered your callsign in MixW, click on the **CQ** button. It should key the radio and produce Morse code calling CQ with your call sign. Verify with the **TX Meter (4)** set to **Fwd Pwr** that there is forward power.
6. You are now ready to begin a QSO. If a 50 ohm dummy load was connected, connect an antenna in its place. Tune to the desired frequency using one of the methods outlined in the Tuning Methods section above. Select either **CWL** or **CWU (6)** and proceed with the QSO.

Note:

CW VFO Frequency Offset

The VFO on the PowerSDR software is designed to show the zero beat of the CW tone using the selected CW Pitch. For this reason, when in the CW Modes, the center zero line is actually offset by the CW Pitch frequency. This allows click-tuning of CW signals as well as the traditional CW VFO readout. This also enables the VFO to remain constant when switching from CWL to CWU mode. Every effort possible is made to preserve a CW signal when switching between any SSB and CW modes.

Digital Mode Operation

To operate digital modes, PowerSDR needs to connect to third party digital mode programs with both CAT control and Audio connections. PowerSDR realizes the CAT control connection through N8VB's virtual COM port utility and the Audio connection through the Virtual Audio Cable (VAC) utility. We will explain later in this section how to install and setup each of these two utilities. First, however, we will outline how to operate digital modes with these utilities installed and setup.

Note: Throughout this section we will refer to *digital mode programs*, which also include logging programs. For the latter the CAT control section applies to enable reading and possibly also controlling PowerSDR's frequency, band, and operating mode. If the logging program includes a voice keyer, the VAC section may also be relevant.

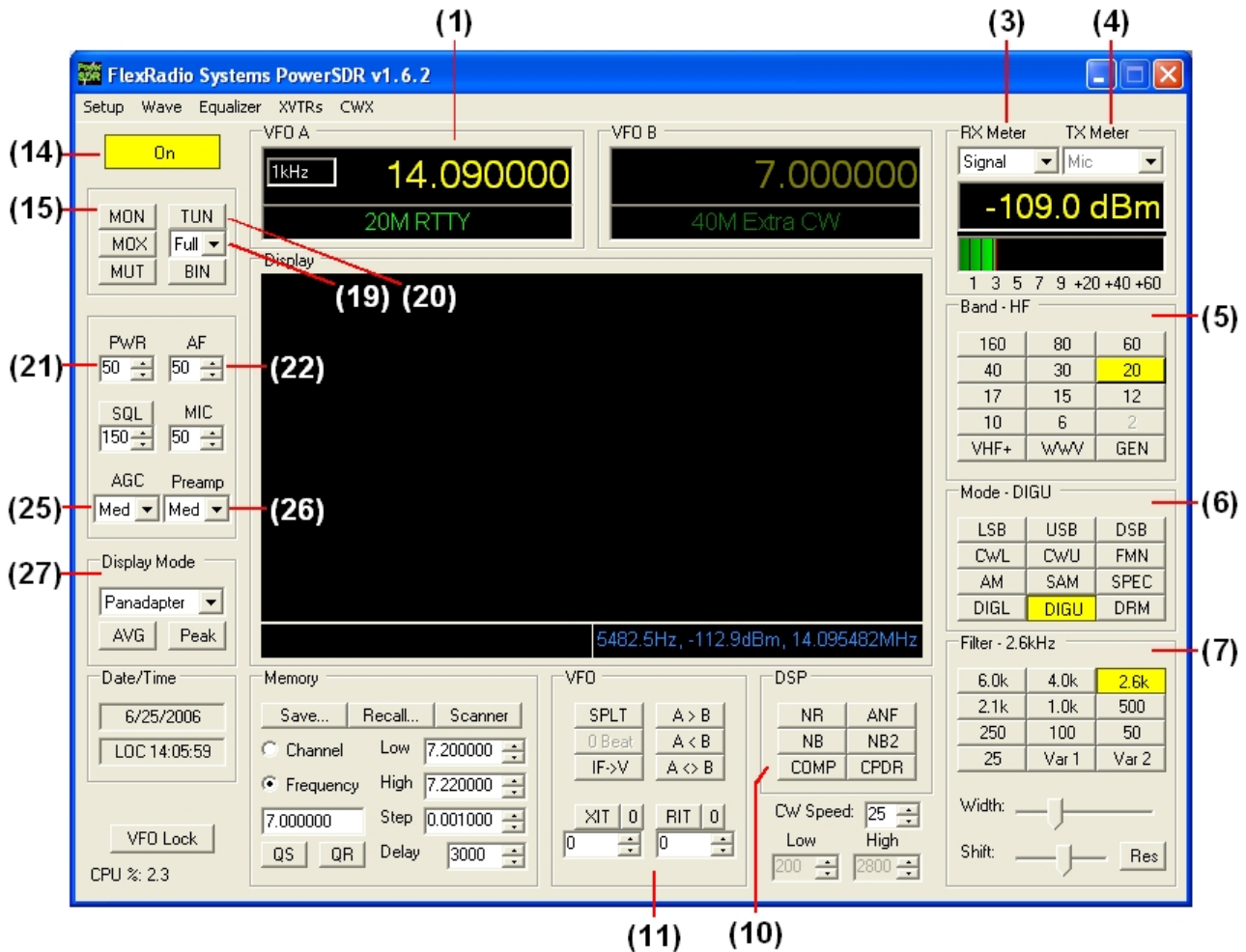


Figure 123: Front Panel Controls Used for Basic Digital Mode Operation

Please use the above figure to identify the controls referenced in the following step-by-step instruction. For consistency we have used the same control identifiers as in the preceding Front Console chapter and for clarity we have left out any identifiers of controls not referenced.

Note: This procedure assumes that the calibration routines outlined in the Calibration section of the Setup Form chapter above have already been carried out.

1. Follow the Power Up Procedure, described above. Then click the **On** button (14). Set the following controls as specified.

Table 14: Initial Control Values for Digital Mode Operation

| Ctrl | Value | Reference to Figure Above |
|---------------|------------|---------------------------|
| VFO A | 14.09MHz | (1) |
| RX Meter: | Signal | (3) |
| TX Meter: | MIC | (4) |
| BAND: | 20 | (5) |
| Mode: | DIGU | (6) |
| Filter: | 2.6kHz | (7) |
| COMP: | Off | (10) |
| CPDR | Off | (10) |
| XIT: | Off | (11) |
| RIT: | Off | (11) |
| PWR: | 50 | (21) |
| AF: | 50 | (22) |
| AGC: | Med | (25) |
| Preamp: | Med | (26) |
| Display Mode: | Panadapter | (27) |

2. Ensure that on the PowerSDR Setup Form **Cat Control** and **VAC** are enabled (if VAC is auto-enabled it will automatically be enabled when **DIGL**, **DIGU** or **DRM** (6) modes are selected).
3. Start up your digital mode program and ensure its CAT control and sound card selection are configured to connect to PowerSDR. If so, your digital mode program's frequency should correspond to PowerSDR's VFO frequency and it should be receiving Audio from PowerSDR. Use the **RX control** on the Setup Form – Audio Tab, VAC Sub-Tab to adjust the audio level going to your digital mode program (instead of the AF control on the Front Console).
4. Connect a 50 ohm dummy load or tune **VFO A** (1) to a quiet frequency on the selected band. If using the 100W PA, use the **TUN** button (20) to verify power output on the **TX Meter** (4) (Set to **Fwd Pow**). If using the integrated ATU, select **Full** on the **ATU Mode** selection (19) and then click the **TUN** button (20). See the section below on ATU Operation for more on how to use the ATU.

CAUTION: If not using the ATU and not using a 50 ohm dummy load, ensure that the antenna presents a 50 ohm load with a low SWR or damage may occur to the SDR output transistors.

5. Click on the Transmit button of your digital mode program. It should switch PowerSDR to transmit. Transmit a test signal (e.g. several CQ calls) in the mode you plan to operate and use the **TX control** on the Setup Form – Audio Tab, VAC Sub-Tab to adjust the volume of audio coming from your digital mode program (instead of the MIC control on the Front Console). Set the **TX Meter (6)** to **ALC** and adjust for 0dB to calibrate transmit. Click on Receive on your digital mode program and PowerSDR should switch back to receive.
6. You are now ready to begin a QSO. If a 50 ohm dummy load was connected, connect an antenna in its place. Tune to the desired frequency using one of the methods outlined in the Tuning Methods section above. Select either **DIGL** or **DIGU (6)** for lower or upper side band digital mode operation respectively. Select **DRM (6)** for DRM mode operation.

Note 1: The **DIGL**, **DIGU** and **DRM (6)** mode buttons bypass all signal processing in PowerSDR, except for AGC and Filtering. With **DIGL** and **DIGU** you have control over the filter width using the filter buttons **(7)**. **DRM** mode invokes a fixed 10kHz wide double side band filter.

Note 2: **DIGL** and **DIGU (6)** apply an offset when using Spectrum Click Tuning (see above). By default, the offsets are set to 1200 Hz (SSTV) in **DIGU** mode and 2210 Hz (RTTY) in **DIGL** mode respectively. These offsets can be modified on the Setup Form, General Tab – Options Sub-Tab.

Hint: When operating digital modes you have two options with regards to using filters. On the one hand you can use a wide band-pass filter in PowerSDR and use the filters within your digital mode program for selectivity. On the other hand you can home in on a specific signal with PowerSDR's filters and ignore the filters in your digital mode program. Although circumstances and operator preference will dictate which to use, many operators have found especially the second option to be of value to them.

If you already have vCOM and VAC up and running, you may ignore the following sub-sections.

We will now focus on installing and setting up N8VB's vCOM driver and VAC. We will then use MixW as an example on how to use these utilities to operate digital modes.

CAT Control Setup

The CAT control commands of PowerSDR are based on those of the Kenwood TS2000 and have been extended to cover PowerSDR's many unique features. Additionally, PowerSDR can provide a virtual COM port connection to third party software through N8VB's virtual COM port driver. Special thanks go to N8VB, K5KDN, and KD5TFD for their work on the CAT interface to make all of this work.

The following procedure outlines how to install and setup N8VB's virtual COM port driver.

Install N8VB's vCOM Driver

First download and install the vCOM Virtual Serial Port Driver from the Downloads page on the FlexRadio website. This program installs one or more pairs of virtual COM ports connected in null modem style. Locate the file you just downloaded (**N8VBvCOMSetup-226a.exe**) and double-click to start the Windows installer. The following screen will appear.



Figure 124: vCOM Installer Language

Select your language of choice and click **OK**. The welcome screen appears.

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Figure 125: vCOM Installer Welcome Screen.

Click the **Next** button to continue the driver installation.

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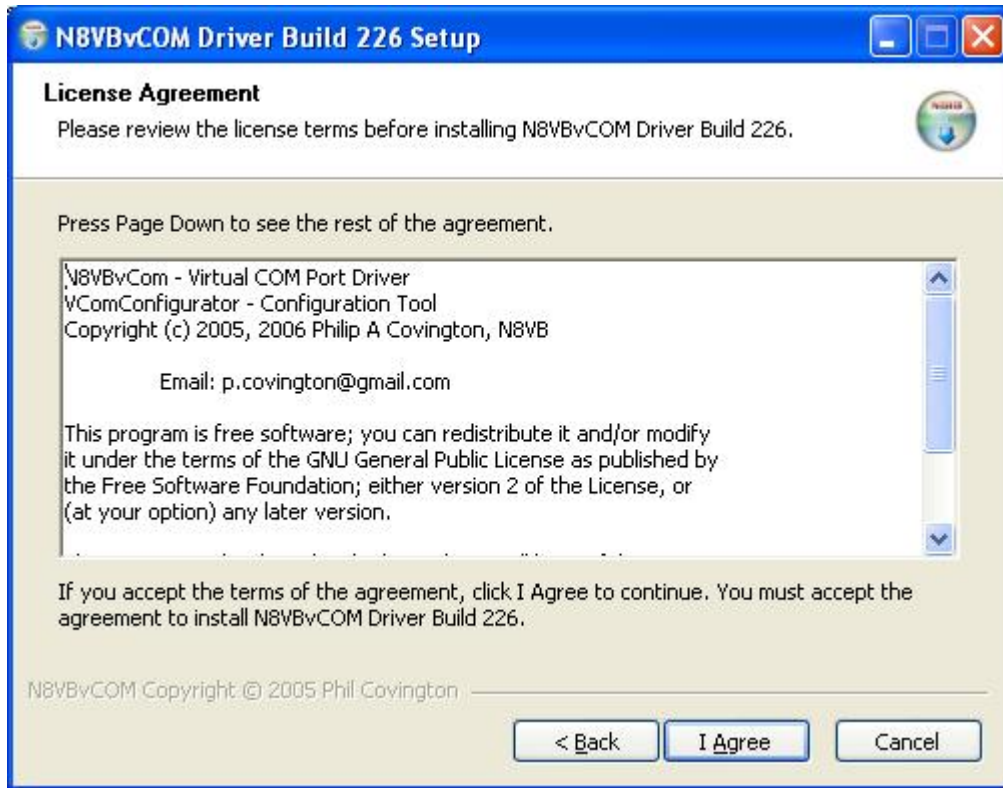


Figure 126: vCOM License Agreement.

Review the License Agreement and click the **I Agree** button.

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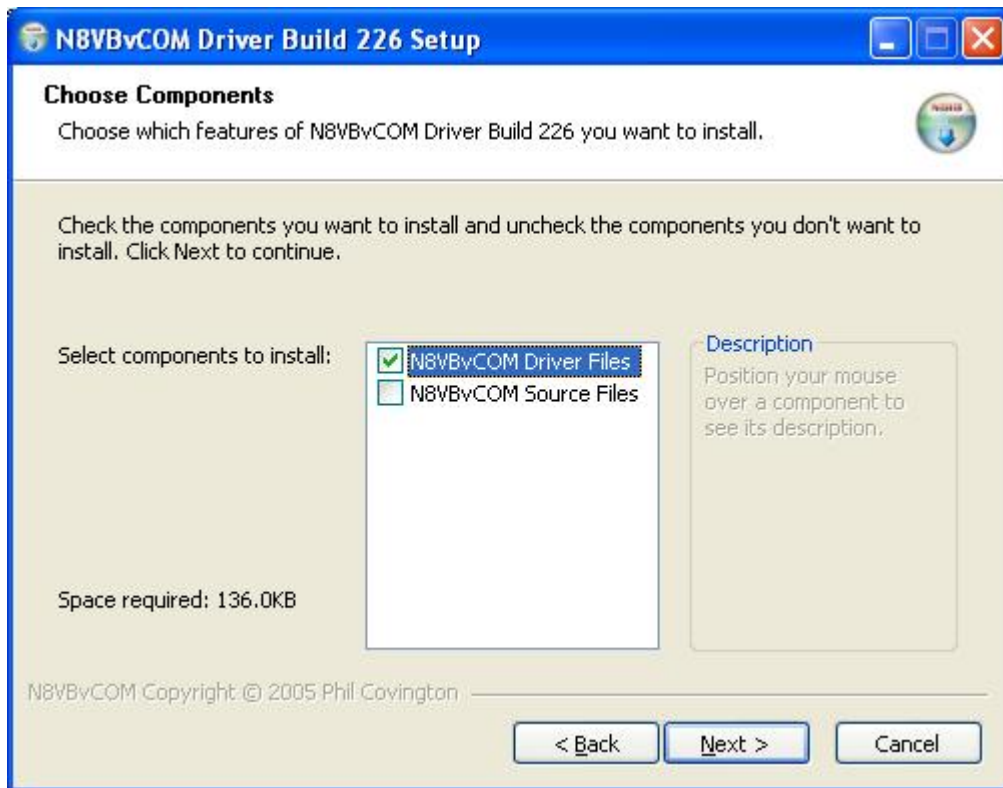


Figure 127: vCOM Installer Component Selection.

Select the components you wish to install and click the **Next** button.

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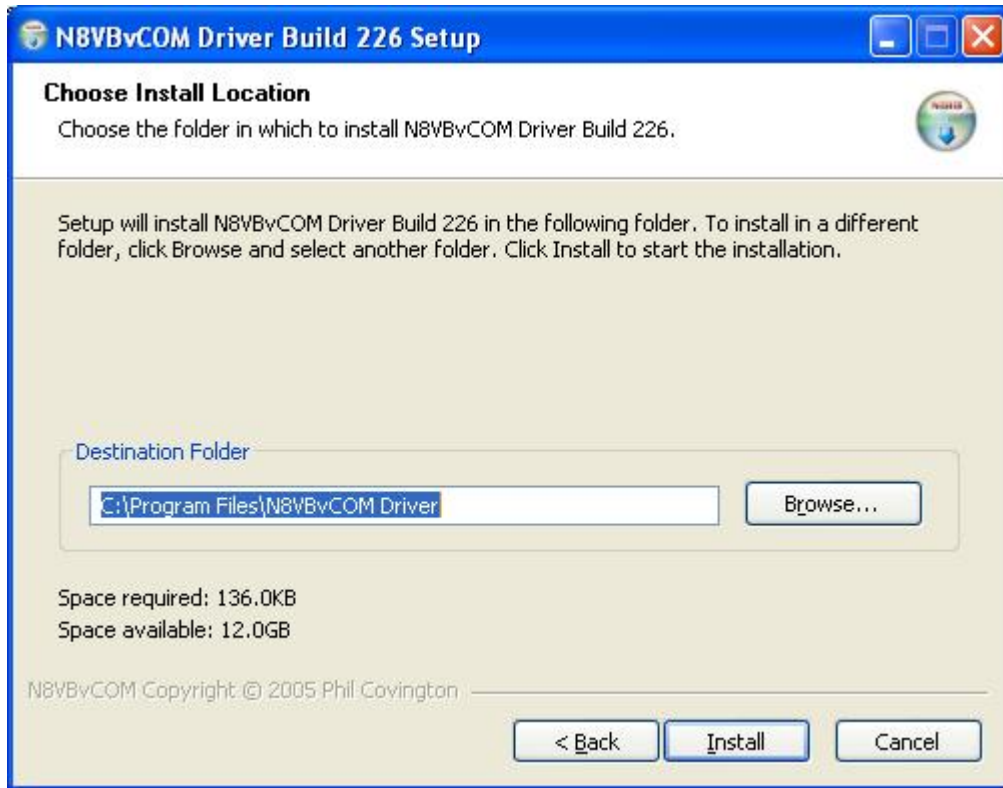


Figure 128: vCOM Install Location

Choose the folder in which to install the driver application in and click the **Install** button.

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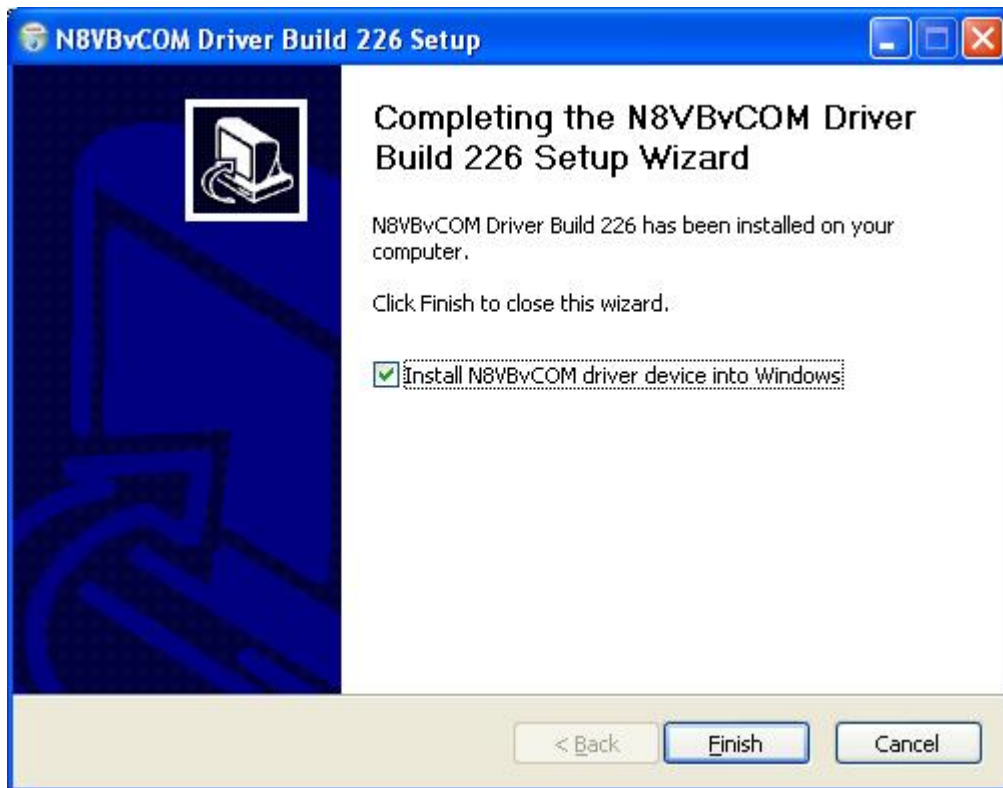


Figure 129: vCOM Installer Complete

Click the **Finish** button to complete the installation of vCOM. The virtual serial port driver will now be installed on your system. A Command Prompt window displaying status messages will open followed by the warning screen shown below.

[The rest of this page is intentionally left blank]



Figure 130: Hardware Installation Warning

Click on the **Continue Anyway** button to complete the installation.

Configure the vCOM Port Pairs

By default vCOM will install 4 COM port pairs. To view these pairs, to change the default or to remove the driver, locate and start the vCOMConfigurator application (Start>All Programs).

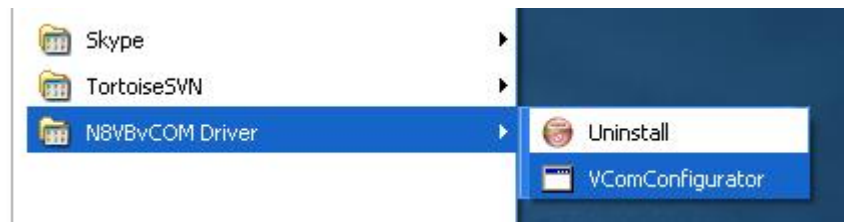


Figure 131: vCOMConfigurator Application

This will open the following screen.

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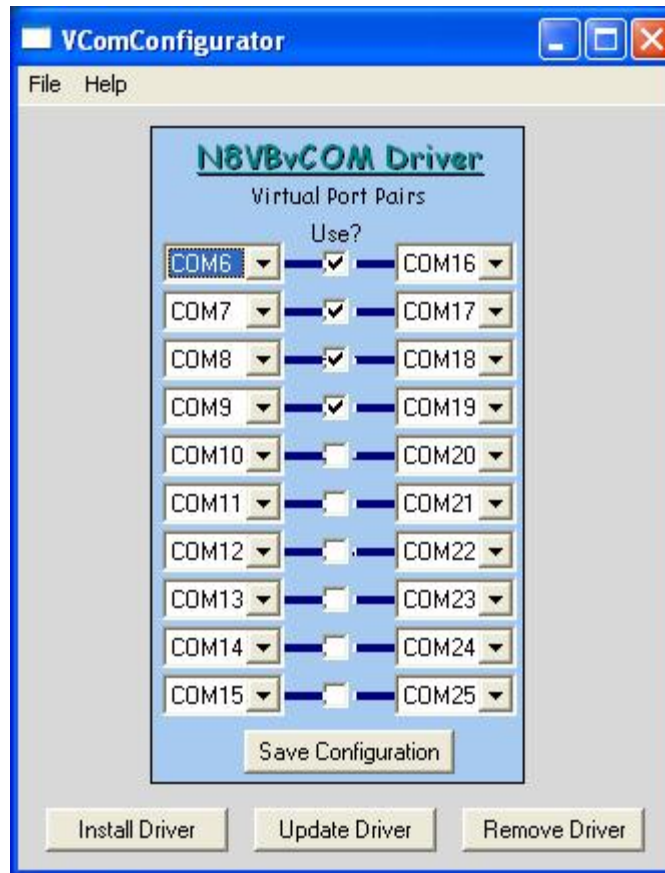


Figure 132: vCOMConfigurator

With vCOMConfigurator, you can select, unselect and modify the desired COM port pairs. Click the **Save Configuration** button to save your configuration and click **Update Driver** to update the driver with the new settings. From here you can also remove and install the driver. When you are finished, close vCOMConfigurator .

Now that the virtual COM port pairs are installed, we need to setup the applications to use them. We'll start with the PowerSDR software. Start up the program and pull up the CAT Control Tab on the Setup Form. For the purpose of this example, we will use the COM6-COM16 pair.

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Configure PowerSDR CAT Control

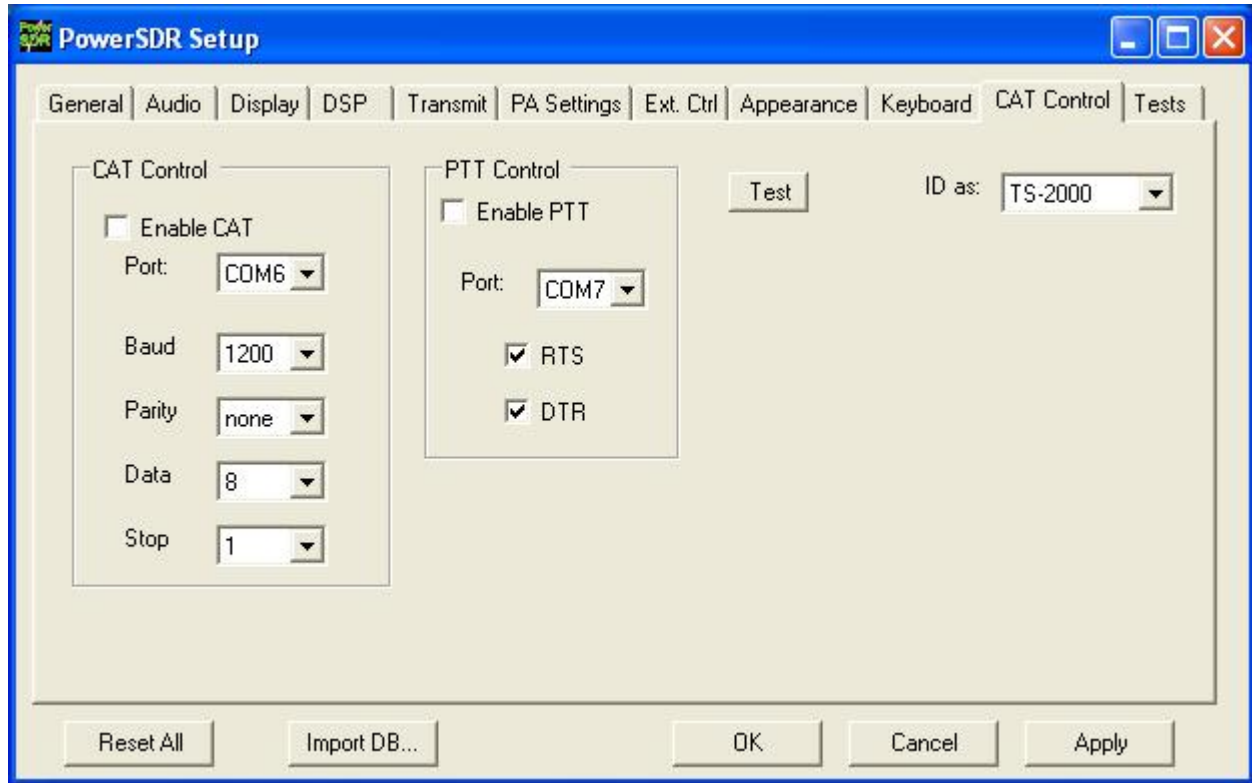


Figure 133: Setup Form Cat Control Tab

Set the controls in the **CAT Control** Group on the left side of this form to match those shown above to connect PowerSDR to COM 6 of the COM6-COM16 pair. (If you changed the COM port pair settings in vCOMConfigurator you will need to modify the COMx setting to match that change). Set the **ID** as **TS-2000**. Once all of the settings are correct, click the **Enable CAT** check box.

Note 1: As more external applications follow Ham Radio Deluxe to incorporate the Flex Radio SDR-1000 with its extended set of CAT controls, the ID can be changed to **SDR-1000**. However, in the meantime it is more prudent to select the **TS-2000**.

Note 2: Some third party applications do not allow you to connect to a high COM port number. In such a case change COM6 to COM16 above and connect the digital program to COM6. Alternatively, create a different virtual Com port pair, e.g. COM6-COM7.

Configure PowerSDR Keyer Connections

We can setup PowerSDR to connect to third party CW programs using the CAT port (i.e. the COM port used for CAT control - COM 6 in the previous figure). To do so, on the PowerSDR Setup Form, select the DSP Tab and then the Keyer Sub-Tab of the.

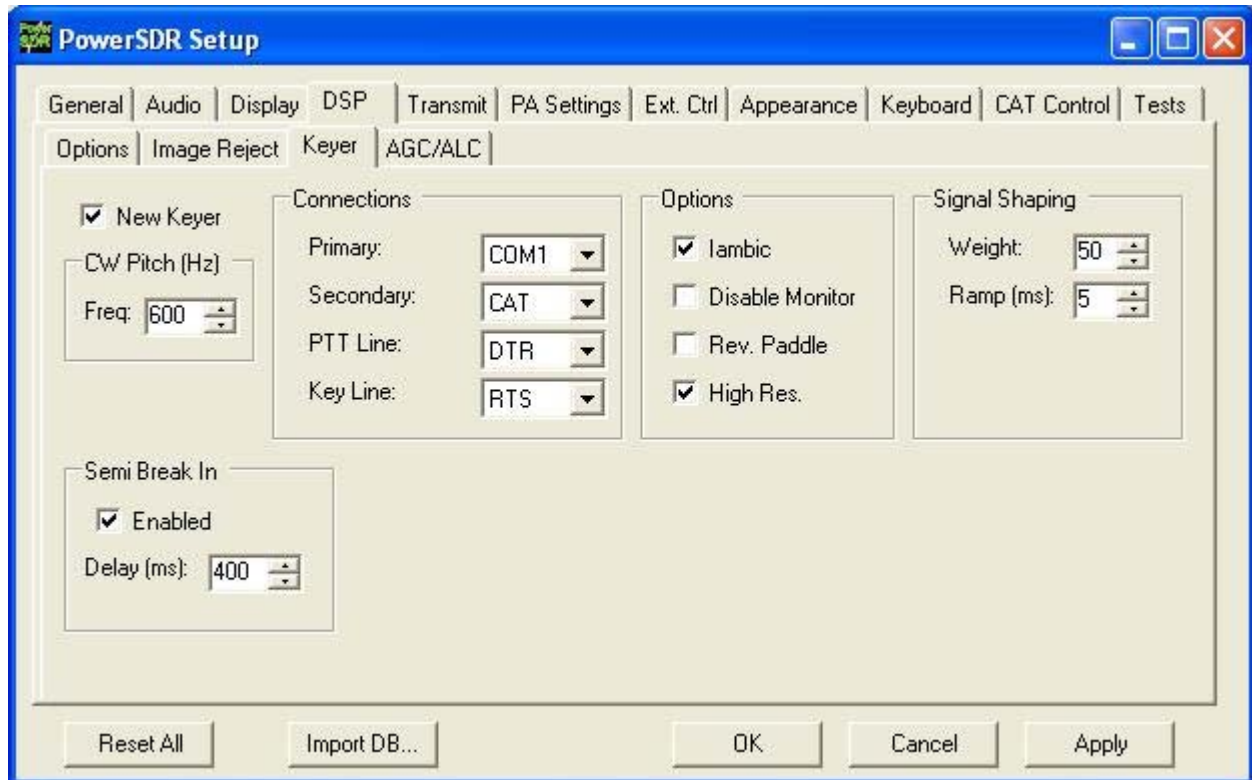


Figure 134: Setup Form - DSP Tab, Keyer Sub-Tab

Within the **Connections** Group, set **Primary** to the physical COM port you have your paddles or straight key connected to. Set **Secondary** (used for keyers and programs) to **CAT** to use the same (virtual) COM port used for CAT Control. Set **PTT Line** to **DTR** and **Key Line** to **RTS**. Note that CAT Control must be enabled (see the previous section) before this will work. Click **OK**.

Virtual Sound Connection

Although you can use a second sound card to connect to an external digital mode program, PowerSDR has the ability to use a virtual sound connection. To do so you will first need to download and install the third party Virtual Audio Cable (VAC) application from one of many sources. VAC is neither free nor open source.

Note: PowerSDR will work correctly with VAC versions 3.09 through 3.12. It will not work correctly with the more recent version 4.

In essence VAC enables the creation of so called digital virtual audio cables between two software applications. When setup correctly, these cables appear as input and output audio devices, as if they belonged to a sound card.

Create the Virtual Audio Cables

The first step is to create the two identical virtual cables. To do so open the Virtual Audio Control Panel.

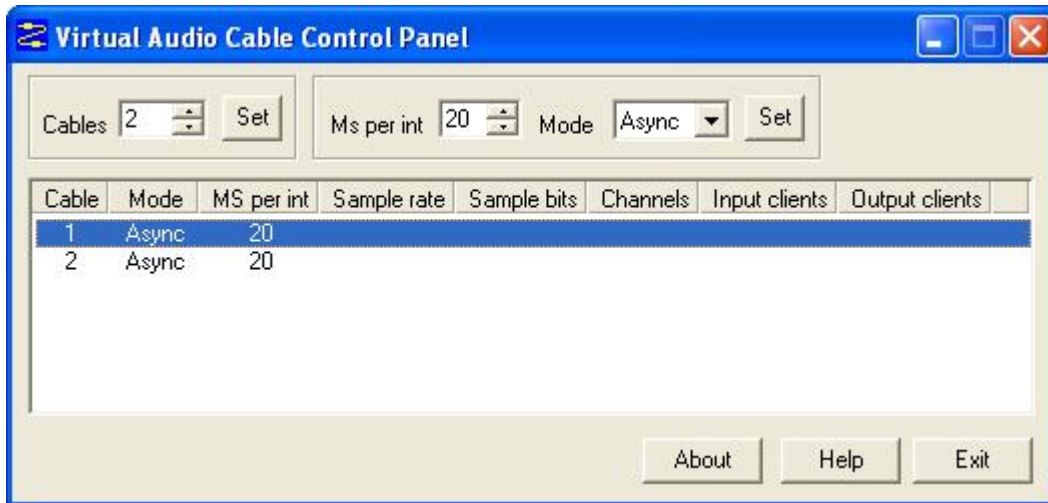


Figure 135: VAC Control Panel

Select 2 **Cables** and click **Set** – the 2 cables appear in the lower half of the screen (see the figure above). For each cable in turn select 20 **ms per interrupt** and Async **Mode** and click **Set**. Do not close the Control Panel just yet.

Setup VAC in PowerSDR

Next we need to connect PowerSDR to one side of each of these cables. Startup PowerSDR, but leave it in Standby. Open the Setup Form, click on the Audio Tab and then click on the VAC Sub-Tab.

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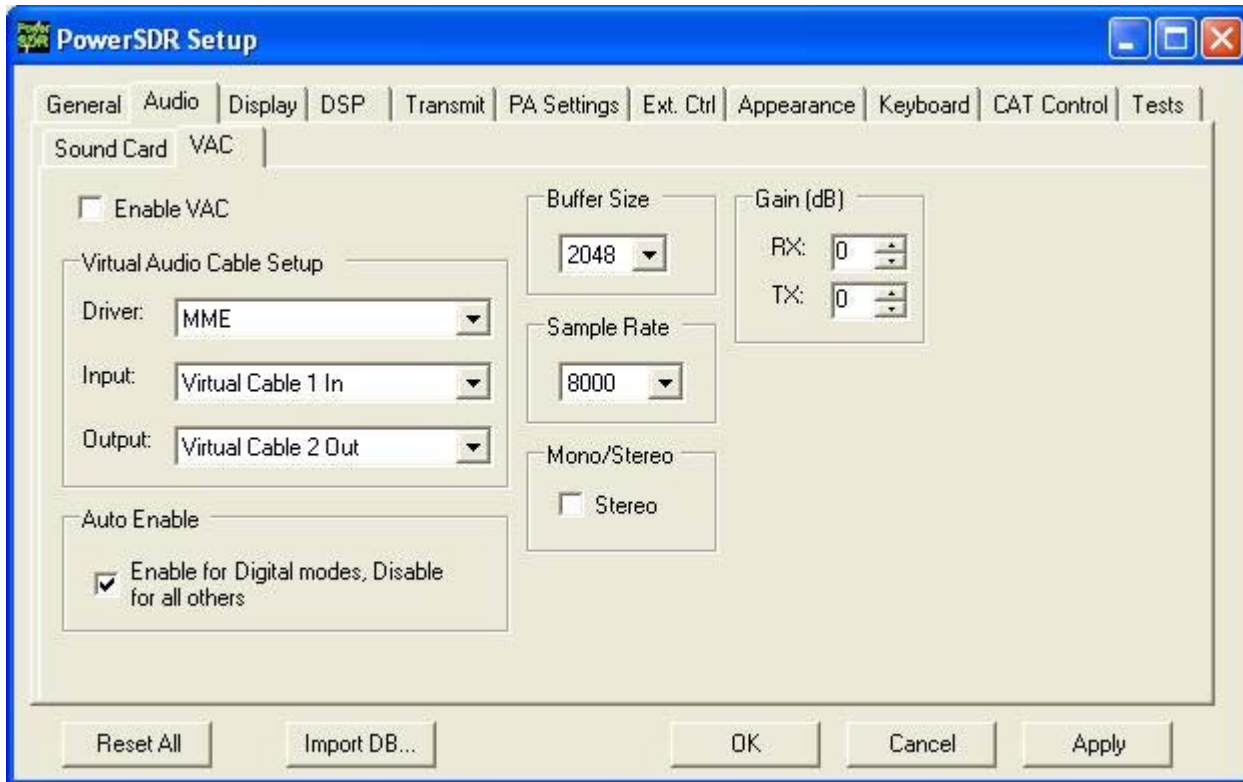


Figure 136: Setup Form - Audio Tab, VAC Sub-Tab

Configure the settings as shown above. Click Enable VAC to manually enable the virtual audio connection or click Auto Enable to automatically enable it when clicking the **DigL**, **DigU** or **DRM** mode button on the Front Console.

Note: The sample rate and **Mono/Stereo** mode settings shown in the figure above will work for most digital programs. However, especially in the case of **DRM** the sample rate will need to be increased to **48000** and the mode to **Stereo**.

Setting up Third Party Digital Programs

Each third party program has its own method of configuring CAT Control and selecting the COM port and the sound card. In the following, we will use MixW as an illustrative example. We will also discuss how to deal with digital programs that are only able to connect to the default sound device.

Note: Instructions for MMTTY, MMSSTV and WSJT 6 can be found in the **VAC Setup and Use Guide** by N4HY on the Downloads page of our website.

Using MixW with PowerSDR

Start MixW, which may be downloaded from <http://www.mixw.net/>.

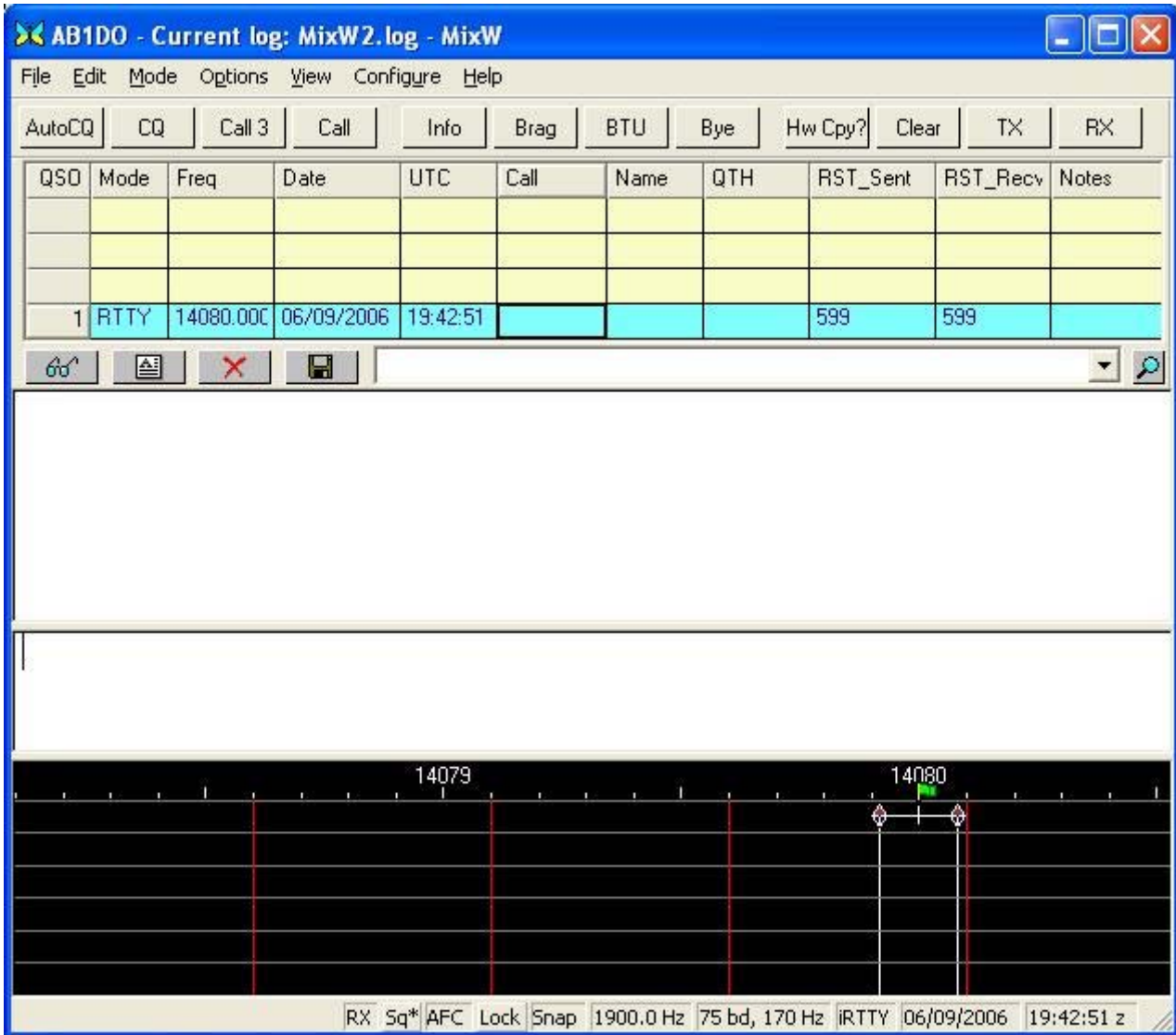


Figure 137: MixW Console

First we will configure MixW to connect through the virtual COM port pair to PowerSDR. Click **Configure** on the Menu bar and then select **TRCVR CAT/PTT** to open the following screen.

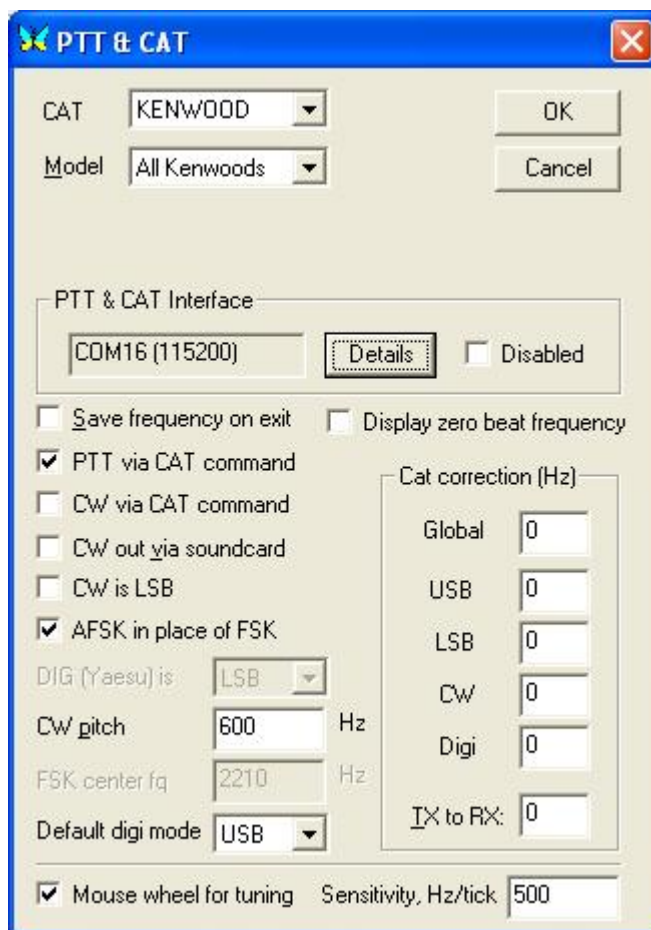


Figure 138: MixW PTT & CAT

Configure the controls as shown above. MixW does not recognize the extended SDR-1000 CAT controls yet. Therefore we set it up as if PowerSDR were a Kenwood radio (In MixW **All Kenwoods** is the only option available when **CAT** is set to **Kenwood**).

On the **PTT & CAT Interface**, click on the **Details** button to open the following form:

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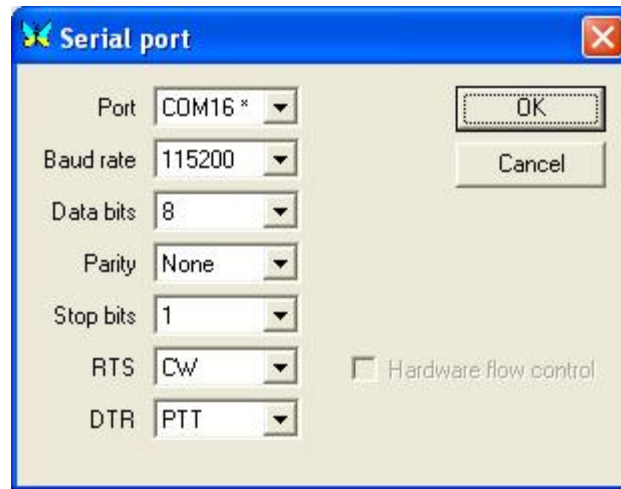


Figure 139: MixW Serial Port Settings

Recall that we are using the COM6-COM16 virtual COM port pair and that we setup PowerSDR to connect to the COM6 end of this pair (see the CAT Control Setup section above). We therefore now select **COM16** as the **Port** in the form above. Set the **Baud rate** to **115200** (it makes no difference what you set this to).

For **RTS** and **DTR** we need to match the settings on the PowerSDR Setup Form, Keyer Sub-Tab. Therefore set **RTS** to **CW** and **DTR** to **PTT** as shown above. Click **OK** on this form and then on the **PTT & CAT** form respectively to accept the settings and close the forms.

On the MixW front console, click again on **Configure** on the menu bar and this time select **Sound device settings...** to open the following form:

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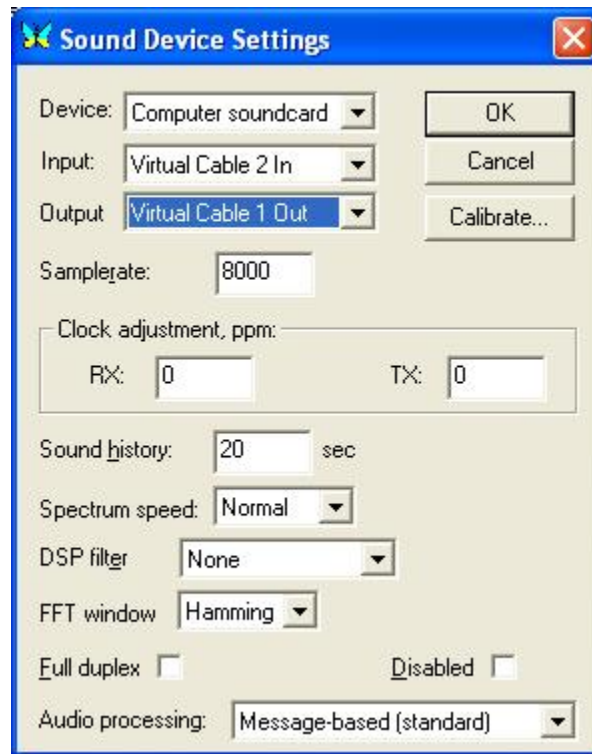


Figure 140: MixW Sound Device Settings

Recall that we previously created 2 virtual audio cables and then configured the PowerSDR VAC Input as Virtual Cable 1 In and Output as Virtual Cable 2 Out.

We now need to connect MixW to the other ends of those two virtual audio cables respectively. Therefore, set **Device** to **Computer sound card**, **Input** to **Virtual Cable 2 In** and **Output** to **Virtual Cable 1 Out** as shown above. If you change the **Samplerate** setting, MixW will tell you to exit and restart MixW.

Exit MixW for now so we can illustrate the functioning of the virtual audio cables.

Click the Standby/On button to turn PowerSDR on. When you do, the Virtual Audio Cable Control Panel should look like this:

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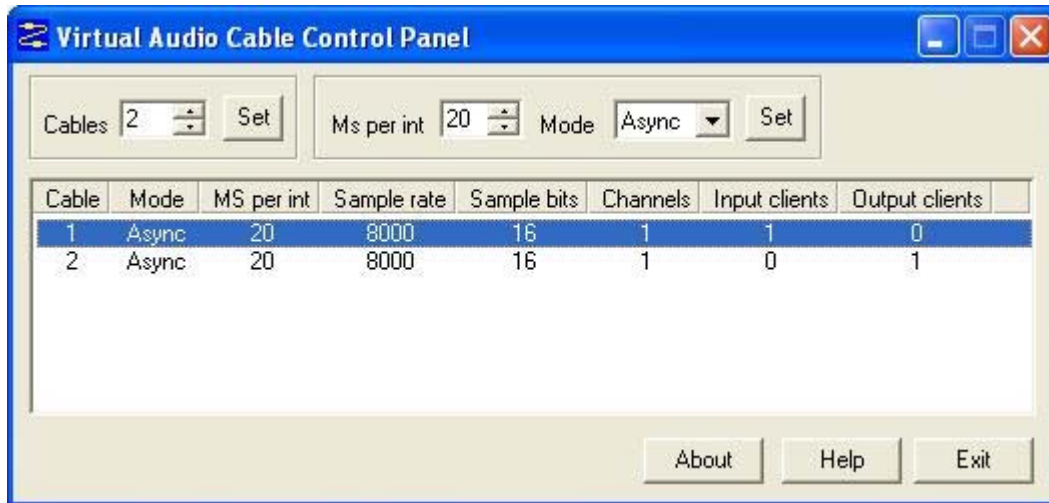


Figure 141: VAC Control Panel - Only PowerSDR Running

Notice that you have two cables. Each of them is a monaural channel (**Channels** = 1). You have one **Input** and one **Output client**. This is PowerSDR.

Now start MixW. You will see a subtle change in the VAC Control Panel:

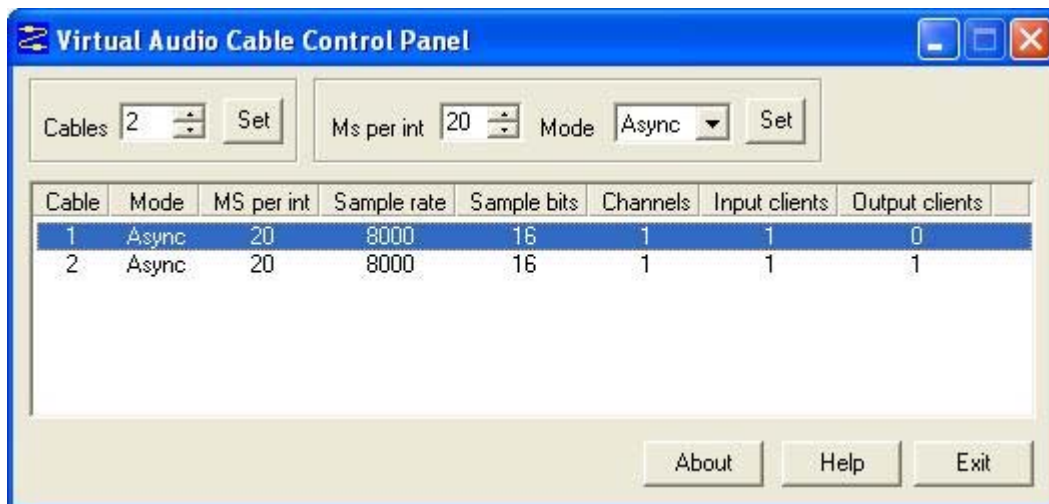


Figure 142: VAC Control Panel: PowerSDR Running and MixW Receiving

MixW is **receiving** audio from PowerSDR and is the **Input client** that is now present on Cable 2. If you transmit, this will drop and the first audio cable will get an output client. The MixW display should show signals and its frequency information should correspond to that of PowerSDR. Change frequency in either MixW or PowerSDR and the other will follow.

Change the mode in MixW to CW. Click on the **TX** button on MixW's main panel. It should key PowerSDR without generating a tone. Click **RX** in MixW and PowerSDR should return to receive. If you have entered your callsign in MixW, click on the **CQ** button. It should key the radio and produce Morse code calling CQ

with your call sign. Close the VAC Control Panel; you will no longer need this, unless you intend to create more.

Programs Needing to Connect to the Default Sound Device

Certain digital programs can only select the default sound card (or at best choose a single card by index). For example, MMTTY and MMSSTV choose by index. To enable PowerSDR to work with these programs you will need to change the system default sound card. VAC offers the ability to continue to use your other audio applications through its **Audio Repeater** utility, which we will also discuss.

To change the default sound card, in Windows click Start and then Control Panel. In Control Panel, double-click on **Sounds and Audio Devices** to open the Sounds and Audio Devices Properties Form:

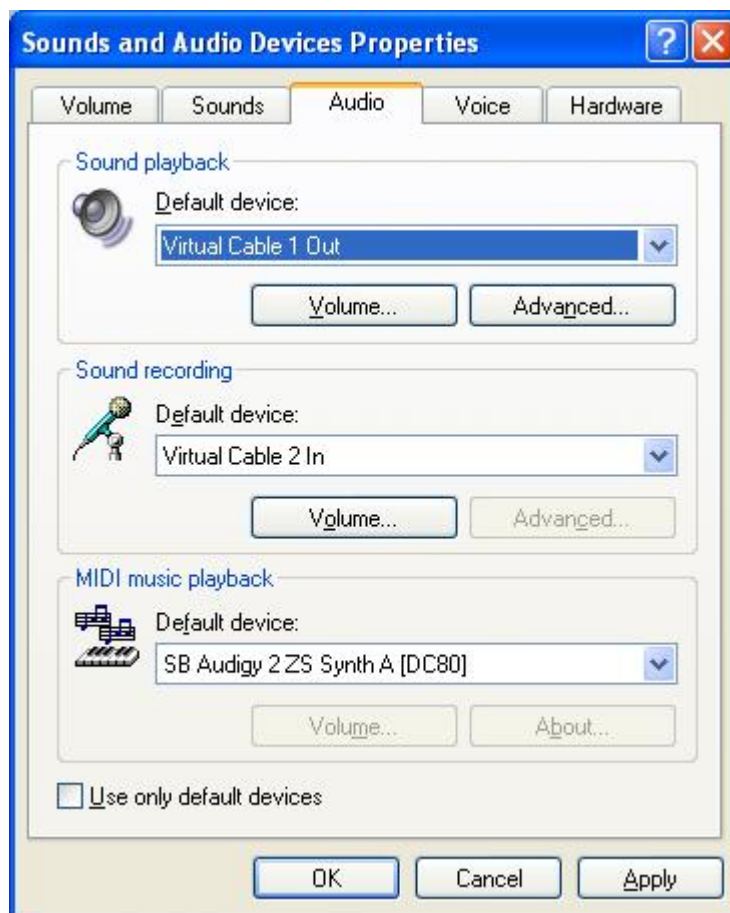


Figure 143: Sounds and Audio Devices Properties Form

Set up the **Default device** for **Sound playback** and **Sound recording** as shown above. (Midi playback is of no concern). Click **Apply** and then **OK**.

If you now setup your digital program to select the default sound card, it will work with PowerSDR.

To enable you to continue using other sound programs (MP3 players, etc) despite changing your default sound card settings, VAC comes with a utility called **Audio Repeater**. Locate Audio Repeater in the VAC program folder and double-click to start it. You should see the following screen:

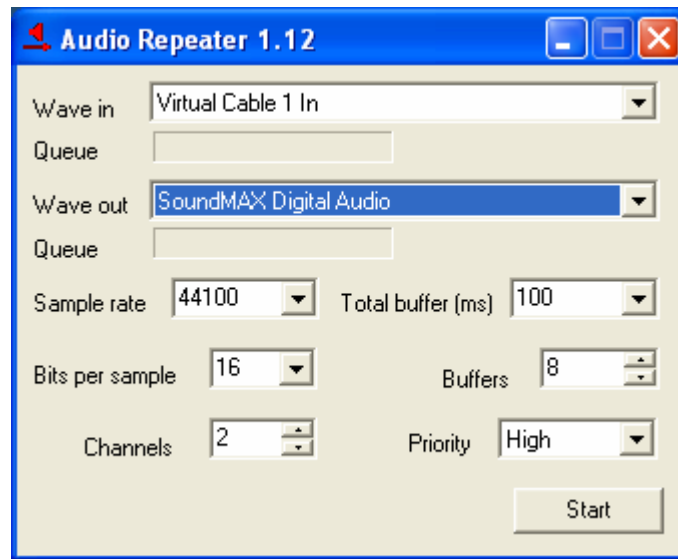


Figure 144: Audio Repeater Utility

To play back audio, set **Wave in** to **Virtual Cable 1 In** and **Wave Out** to the sound card you wish to use. You should keep the **Total buffer (ms)** as small as practicable to prevent long latency between the sound arriving at the virtual audio cable and it being played to the sound card.

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ATU (Antenna Tuning Unit) Operation

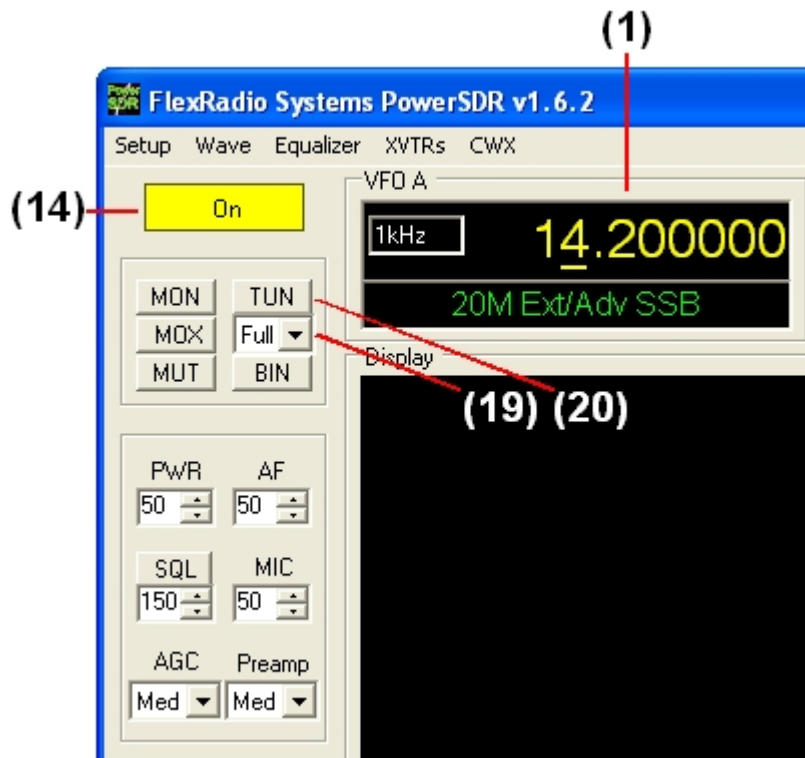


Figure 145: Front Panel Controls Used to Operate the ATU

Note: The ATU offered by FlexRadio is a customized version of the LDG Z-100. For more detailed information please refer to the Z-100 operating manual, which can be found at:

<http://www.ldgelectronics.com/Z-100%20operations%20manual.pdf>.

In the following we will describe how to operate the ATU using the PowerSDR software. Please use the above figure to identify the controls referenced.

1. Follow the Power Up Procedure, described above. Then click the **On** button (14).
2. Calibrate the 100W PA using the procedure discussed above in the Automatic Amplifier Gain Calibration section of the Software Setup chapter.
3. Select either the **Memory** or **Full** tuning mode on the **ATU Tuning Mode (19)** control.

Note 1: If the **ATU Tuning Mode (22)** control is not visible on the front panel, enable the **ATU Present** checkbox on the **Hardware Config Sub-Tab** of the **General Tab** of the **Setup Form**.

Note 2: **Bypass** will bypass the ATU. **Memory** will attempt to tune from memory and if no tuned setting is saved near the present frequency, the ATU will perform a full tuning cycle. **Full** will immediately perform a full tuning cycle, ignoring any previously saved settings.

4. Tune **VFO A (1)** to the frequency/band of choice.
5. Click the **TUN** button **(20)** to begin the tuning process.

CAUTION: If you change bands after tuning, the ATU is automatically put in **Bypass** mode for protection. Even if you just change and come back to the same band, you will need to tune again. If you proceed to transmit without retuning, you must present a 50 ohm load with low SWR or damage may occur to the SDR output transistors.

If you get an error message like the one shown below, the PowerSDR and the ATU are not communicating correctly. Please reference the Troubleshooting chapter below.



Figure 146: ATU Error Message

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Troubleshooting

This section is intended to be a collection of solutions to commonly seen problems with the SDR-1000. While this is not a comprehensive list of everything that could go wrong, there is a good chance that if you are having a problem, the solution may be here. If your problem is not dealt with here, please contact FlexRadio Systems using the info provided on the Contact page of the website.

The software is oscillating between receive and transmit modes quickly.

Possible Hardware Causes:

1. Parallel connection is not solid. Double check that the parallel port is making good contact on both ends. If running the PowerSDR software without hardware connected, it is necessary to check the Disable PTT checkbox on the General Tab of the Setup Form.

Relays do not click when starting the software or when cycling the software power button.

Possible Hardware Causes:

1. Power Supply/Enclosure switch is not connected/turned on. Verify that the power supply connection is secure and that it is powered up. Ensure that the front panel power switch on the enclosure is in the “On” position. Make sure the supply can handle the current load of the radio (1.25A w/o PA, 25A w/PA).
2. Parallel Cable problem. The parallel cable must be short (no more than 6 feet) and must have straight through connections on all 25 pins. Note that many cables with the same 25-pin D-connectors may not work properly because some of the pins are not connected or some are crossed (as used in null modem cables). Check the connections on both ends (PC and enclosure) to ensure that good contact is made. Low line drive could also be an issue. Verify that the signal levels on the parallel port are ~5V (when high) and that the rise time is fast and clean.

Possible Software Causes:

1. **Incorrect Configuration Settings.** Verify that your physical configuration (RFE, PA, XVTR, etc.) matches the Check Boxes in the Setup Form, General Tab. Failure to match these settings could cause the software to send the radio commands that it would not understand.
2. **LPT Address.** Verify that the parallel port address is set correctly in the software. This should almost always be 378 (Hex), but if using an add-on card for the parallel port, it could use a different address. You can verify the address using the following steps. Right Click on **My Computer** and select **Properties** to open the **System Properties** window.

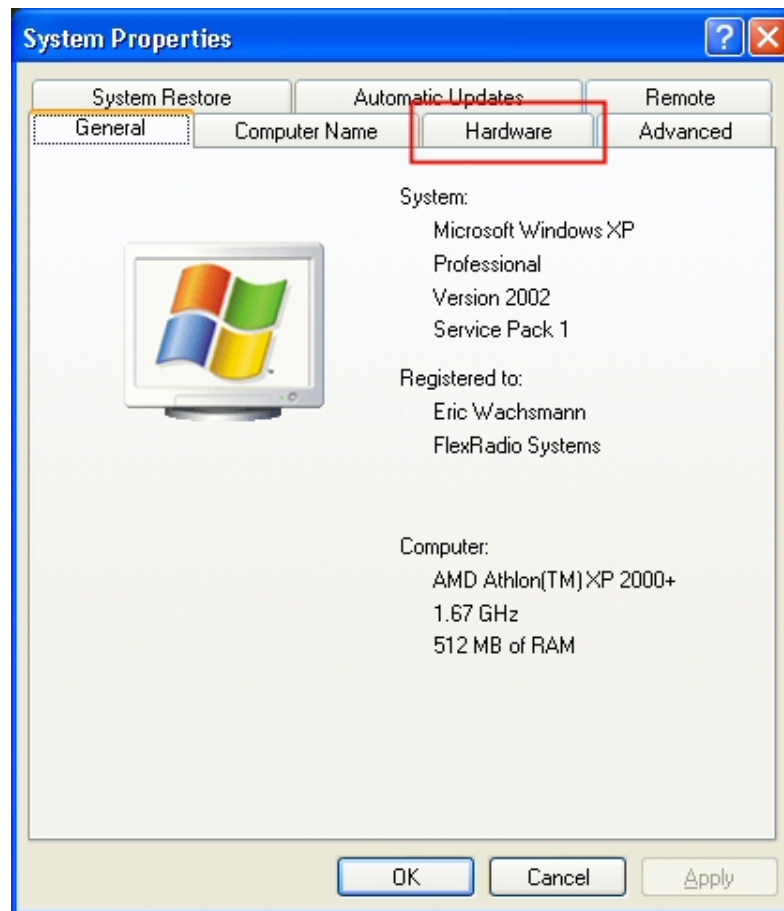


Figure 147: System Properties - General

Click the **Hardware** Tab.

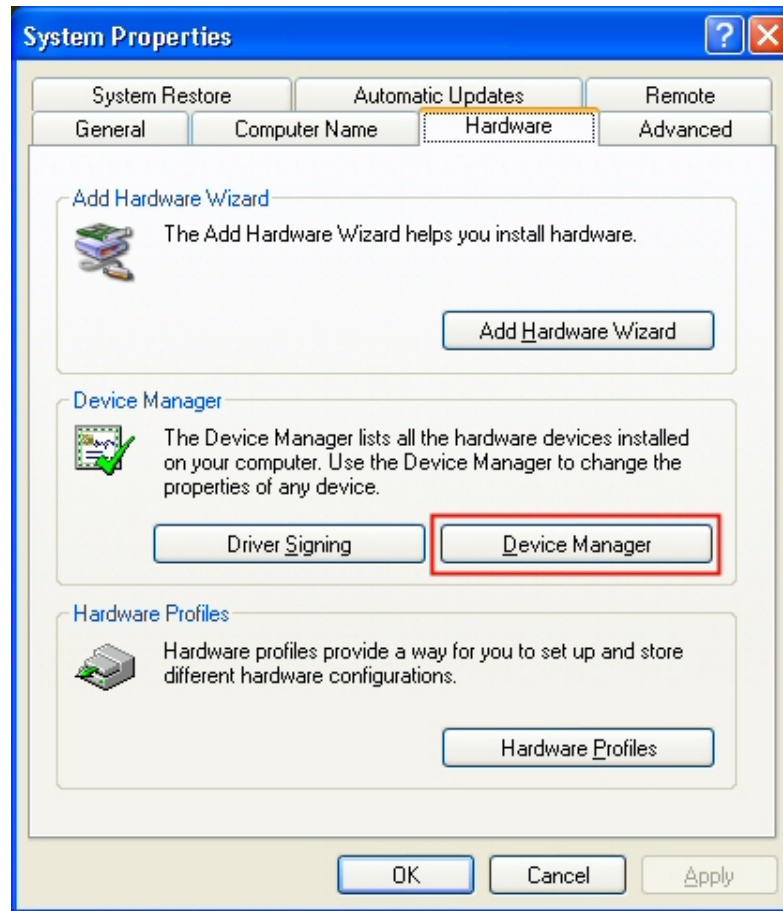


Figure 148: System Properties – Hardware

Click the **Device Manager** button.

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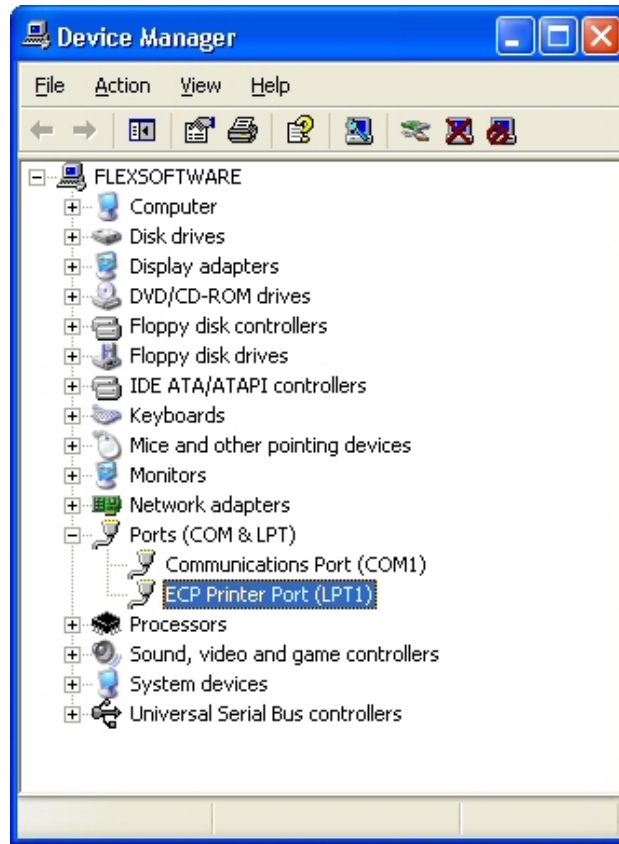


Figure 149: Device Manager

Expand the **Ports** menu. Right click on the device labeled **LPT1** (or LPT2, etc if using an add-on card) and select **Properties**.

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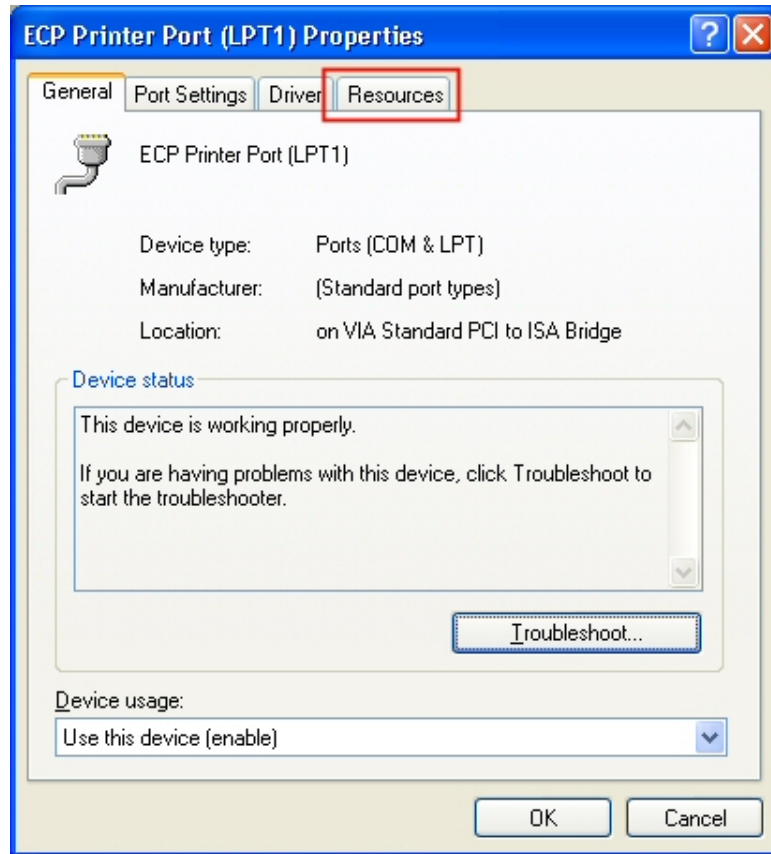


Figure 150: LPT1 Properties - General

Click on the **Resources** Tab.

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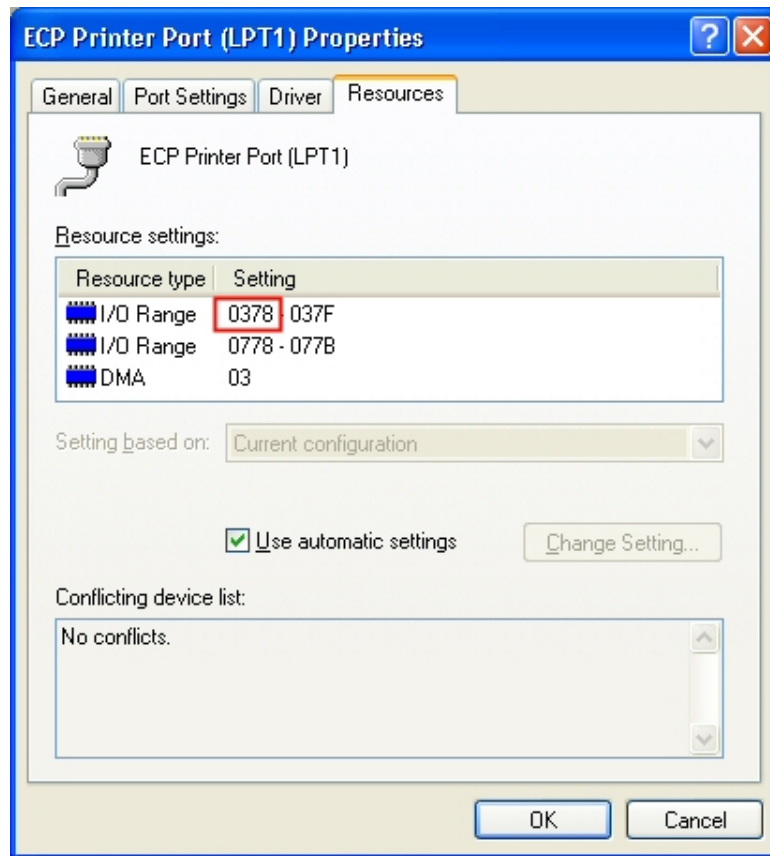


Figure 151: LPT1 Properties – Resources

The first number adjacent to the first I/O Range is the printer port address that should be used in the PowerSDR software.

Relays click, but I have no audio.

Possible Hardware Causes:

1. Powered speakers off or turned down. Ensure that powered (amplified) speakers are connected to the Speaker plug on the enclosure and that they are turned on and the volume level is not all the way down.
2. Audio cable connections. Check the To Line Out cable for contact and ensure that this is plugged into Speaker 1 or “Front” speaker connection on the sound card (usually light green, sometimes shown with a speaker with an arrow pointing out of it). You should be able to hear noise even without a radio connected.

Possible Software Causes:

1. Sound Card issues. Make sure to use a supported sound card as listed on the FlexRadio website (specifications page). These are the only cards we guarantee will work with the radio (others may work,

but we can not support them). Make sure that general Windows sounds can be heard through the speakers. Power down the radio first, then open up the **Control Panel** (Start -> Settings -> Control Panel). Select **Sounds and Audio Devices**.

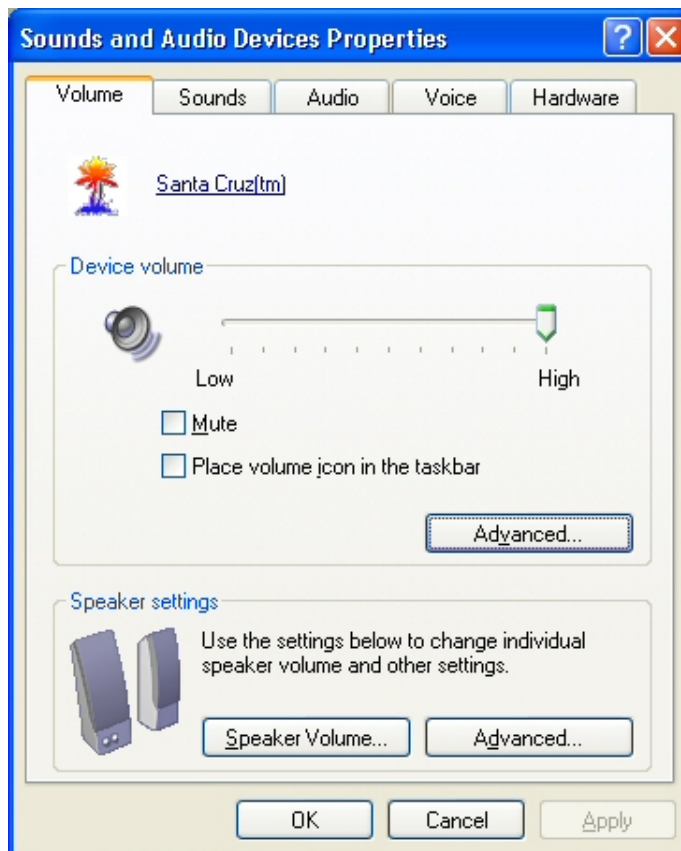


Figure 152: Sounds and Audio Devices – Volume

Select the **Audio** Tab and ensure that the card you are using is selected as the default playback and recording device. Then click the **Sounds** Tab.

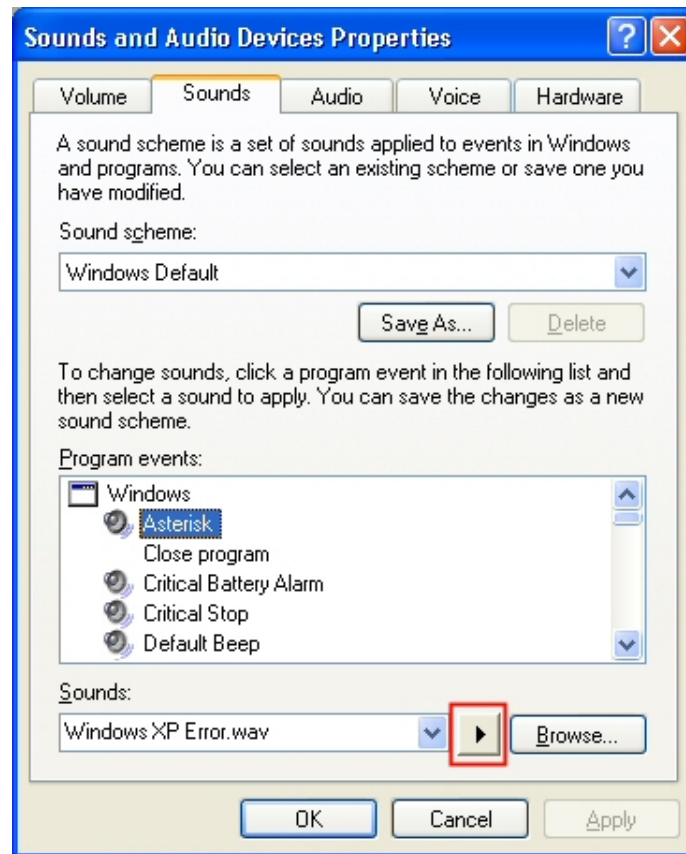


Figure 153: Sounds and Audio Devices – Sounds

Under Program events, select an event that has a speaker next to it (in this case, Asterisk). Click the **Play** button just to the left of **Browse**. If you do not hear any sounds, then something is wrong with the audio system. Try reloading the sound card driver and rebooting. The radio will not work without a functional sound card.

2. AF control is set too low. Setting the AF control too low could effectively mute the receiver. Try moving this control up to the default level of 50 if it has been turned down.
3. ASIO4ALL issues. If using a USB sound card (MP3+, Extigy, etc.), make sure that the selected device listed in the ASIO4ALL control panel is the USB sound card and not another audio device. This is a common problem as the menu selection seems to be lost when disconnecting the USB device or even when just rebooting.
4. PowerSDR Audio Setup. Make sure that the settings on the Audio Tab of the Setup Form reflect the sound card being used. If not using a sound card with native ASIO drivers (such as the M-Audio Delta 44 or the PreSonus FireBox), download and install ASIO4ALL before making the sound card selection in order to make sure the automatic settings will be correct. If using a supported card, select it from the list of devices using the Combo Box near the top left of the form. If using an unsupported card,

the detailed settings are left up to the user. Reference the Audio Tab description in the Setup Form chapter above for more information.

Relays click and I hear static, but no signals.

Possible Hardware Causes:

1. Audio cable connections. Make sure the To Line In cable is making firm contact on both ends. It should be connected to the Line In on the sound card (Blue, sometimes shown with a speaker with an arrow pointing into it). A good test to ensure the audio system is working is to unplug the cable from the To Line In plug on the enclosure and place your thumb over the end of the connector. The spectral display noise floor should “jump” each time the connector plug is touched.
2. Parallel Cable problem. As noted above, if the parallel cable is too long, some of the signals may make it through while others may not. This could cause sporadic tuning and/or relay switching that would prevent normal signals from being heard. Check the connections on both ends (PC and enclosure) to ensure that good contact is made. Low line drive could also be an issue. Verify that the signal levels on the parallel port are ~5V (when high) and that the rise time is fast and clean.
3. Mismatched board connectors. Check that the connections between the boards are all aligned. It is possible to be “off” one pin in either direction on several connectors and still be able to assemble the board stack. This could cause the wrong filter to be selected (among other issues) which would cause significant signal degradation.

Possible Software Causes:

1. PowerSDR Audio Setup. Make sure that the settings on the Audio Tab of the Setup Form reflect the sound card being used. If not using a sound card with native ASIO drivers (such as the M-Audio Delta 44 or the PreSonus FireBox), download and install ASIO4ALL before making the sound card selection in order to make sure the automatic settings will be correct. If using a supported card, select it from the list of devices using the Combo Box near the top left of the form. If using an unsupported card, the detailed settings are left up to the user. Reference the Audio Tab description in the Setup Form chapter above for more information.

I hear signals, but they sound chopped up, like a motorboat.

Possible Software Causes:

1. Delta 44 DMA Setting. If using the M-Audio Delta 44, make sure that the DMA Buffer Size on the Hardware Settings Tab of the Delta Control Panel is set correctly to 512 as shown in the Delta 44 Quick Start Guide. Some programs like MixW have been known to change this value to a much higher value like 2048. This causes problems with the PowerSDR Console. Note that if you are running both the Audio and DSP buffers lower than 512, it will be necessary to run the DMA size down to match this. Not all computers will be able to support settings this aggressive.

2. ASIO4ALL Buffer Settings. If using a card without native ASIO drivers and are having to use the ASIO4ALL application, double check the buffer settings. The KS buffers should be set to 2 or 3 and the Buffer size set to somewhere between 512 and 1024. Settings much larger than this will cause increased latency and can cause broken audio.
3. Other application. The PowerSDR Console is a real time audio application and as such it needs a good bit of CPU resources in order to keep up with the audio. Computer systems close to the minimum specification (800MHz) will have a hard time running other applications concurrently with the console. While faster machines may be able to keep up, we recommend that you close all other applications to get the best performance out of the PowerSDR Console.

I hear signals, but cannot null the image.

Possible Hardware Causes:

1. Partially plugged in audio cable. The software uses the left and right channels to feed the I (in phase) and Q (quadrature) signals into and out of the radio. Without both of these channels, opposite sideband rejection is not possible. Quite often this issue is caused by an audio cable that is not making good contact with the stereo plug on the radio (or the sound card). This results in only receiving a single channel of audio. Ensure that all stereo audio cables are completely connected on both ends. Gold plated cables with larger connectors are not recommended as they may prevent the cable from making good contact with the radio plugs due to the larger headers binding on the rear panel of the enclosure. You can monitor the input levels by comparing the RX Meter ADC L and ADC R levels. If they are more than 1.0dB apart, then there is likely a cable problem.
2. Stereo Line In required. As noted above, both the I and Q channels are required to run the radio. For this reason, a stereo input to the sound card is necessary. If using an unsupported sound card, double check that the input is indeed a stereo input.

Note: Microphone inputs are generally only mono connections even if they have a balance control associated with them. Laptops generally only have microphone inputs, so using an external USB sound card is recommended for most laptop configurations.

3. Reversed Line In cables. Some sound cards use separated cables for the left and right channels. If these cables are reversed, the image would actually be at 22kHz below the actual signal (rather than above). This is easily recognized by looking at what appears to be the actual signal (22kHz below the “Image”). When moving the actual signal up and down, observe whether the signal in the software follows the movements or moves in reverse. If the signal moves in reverse, then the channels are reversed and what you are looking at is the image (hence the reflected behavior).

4. **Bad Cable.** Occasionally, an audio cable will develop a short or open across one or both of the audio lines. This will cause issues similar to what happens if a cable is partially plugged in. This is easily tested using a Voltmeter to check that the tip, ring, and ground sections of each plug are electrically connected.

Possible Software Causes:

1. Mixer balance offset. Many sound cards have balance controls on the playback and record side of the mixer. If the recording balance is offset, one channel will be much stronger than the other and balancing the amplitude of the two channels will not be possible. Make sure the balance control is centered for all mixer controls. You can verify whether the audio coming into the sound card is balanced using the ADC L and ADC R settings on the multimeter display. These numbers should be within 1.0dB of each other. If not, the software compensation will not be adequate to resolve the difference.
2. Sound Card DSP active. Some sound cards come with audio effects enabled by default. These audio effects can cause problems with adjusting the image rejection accurately. Make sure to turn off all DSP functions. Use the “Driver Only” option when installing the sound card driver if that option exists.

The signals are all off frequency.**Possible Hardware Causes:**

1. If using the external clock option, make sure that the source is truly 10 or 20MHz. Any error in the source will be multiplied up to the DDS.

Possible Software Causes:

1. Run Frequency Calibration. The frequency calibration routine described in the Software Installation and Setup chapter fixes this very problem. The radio will likely be off by varying amounts (depending on the frequency) until this calibration is run. Note that some sound cards have error in the sampling rate clock (sometimes due to drift) that can partially be compensated using this calibration routine. For this reason, you may get better results by re-running the calibration after the sound card has been active for a few minutes.
2. External Clock Option Frequency Setting. Make sure that if you are using the external clock option that the source frequency is set correctly in the software (10 or 20MHz).

The noise hiss when no signals are present is fatiguing.**Possible Software Causes:**

1. If you are listening to anything other than very weak signals, the Max Gain setting (see Setup Form, DSP Tab) may be set too high. Decrease the value to prevent the AGC from amplifying the noise to annoying levels. Note that this may make very weak signals (close to the noise floor) harder to copy

Software does not respond to PTT signals.

Possible Hardware Causes:

1. Wiring problem. Double check the wiring for the Mic connector and/or the X2 PTT lines using the diagrams above in the chapter on Plug Pinouts.

Possible Software Causes:

1. Disable PTT is checked. Check this setting on the General Tab of the Setup Form.

The receiver works great, but I am getting no output power.

Possible Hardware Causes:

1. Sound card cable issues. Try clicking the **TUN** button on the front panel with the monitor on (**MON**) and see if you can hear the tone. If not, then double check the audio connections to the radio (specifically, the plug marked **To Line Out**). Refer to Figure 3 and the block diagram on the page before it for a reference.
2. Microphone Cable setup. If the **TUN** button test works, but voice transmission does not, double check that the microphone is setup correctly. If using a microphone connected to the front panel of the SDR-1000 enclosure, it is necessary to connect the plug on the back labeled **To Mic In** to the microphone jack on the sound card. If using a PC style microphone, simply plug the microphone into the sound card.
3. Speaker output loaded down. Using low impedance headphones or non-amplified speakers can cause the output line (which is connected to the transmit driver) to be loaded down. Use only amplified or high impedance speakers or headphones when connecting them to the **SPKR** plug on the back of the enclosure. Note that this is not an issue with 4-port cards like the M-Audio Delta 44 or the PreSonus FireBox since they have separate monitor and radio outputs.

Possible Software Causes:

1. PWR setting too low. Check the PWR setting on the left side of the front panel. If using the 100W PA, this should set the output to something near the value in watts (i.e. a setting of 65 should give a tone when tuning around 65W). If using the 1W configuration, this will set the output to a percentage of the audio drive where 100 represents full drive (~1W – depends on the sound card).
2. Mixer setting incorrect. If using a supported sound card, the mixer settings are setup automatically. However, if using a different sound card, it may be necessary to set these values up manually. Make sure that the Transmit selection shows Microphone (or whichever audio line the microphone uses).

I am getting some power out, but the power seems low (also, see above—no power issue).

Possible Hardware Causes:

1. I/Q imbalance on output. If the output cable connecting the sound card to the plug labeled **To Line Out** is damaged, one of the channels may not be as strong as the other. This will result in lower output power, little to no opposite sideband rejection, and distortion. A test is planned to be integrated into the PowerSDR console in the near future for this issue. Until then, with the radio power supply turned off, play an audio file (MP3, CD, etc), and use the sound card mixer controls to pan the audio from left to right. If one side sounds lower than the other, or is not there at all, the cable is likely damaged.
2. Low sound card power. If using the 1W configuration and an unsupported card, the sound card may not have sufficient power to drive the amplifier to 1W. At least 1V RMS is necessary to get full output of the 1W configuration. A procedure for measuring the output voltage of the sound card is included in the 100W PA calibration routine above.
3. PA Calibration. If using the 100W PA, it is necessary to enter the sound card voltage at full strength using the procedure described in the PA calibration routine above. This number is entered automatically for all supported sound cards, but checking this is a good idea as there can be small variations from card to card. If changes are made to the voltage setting, it is a good idea to re-run the PA calibration routine (using the 100W dummy load) to recalibrate the radio.
4. Drop in supply voltage. In order to reach full output, the radio needs to have near 13.8V from the supply. If the voltage drops more than 0.5V at the power terminals on keydown, then the amplifier may not be able to operate at full output. Make sure that at least #12 AWG wiring from the power supply to the terminals.

Possible Software Causes:

1. PWR setting too low. Check the PWR setting on the left side of the front panel. If using the 100W PA, this should set the output to something near the value in watts (i.e. a setting of 65 should give a tone when tuning around 65W). If using the 1W configuration, this will set the output to a percentage of the audio drive where 100 represents a full watt (depends on the sound card).
2. Calibration needed. It may be necessary to run the PA calibration routine to calibrate the amplifier using a 100W dummy load. This will set the audio drive through an iterative process using the ADC on board the PA for calculating correction factors by band.

I am getting too much power out (>100W).

Possible Software Causes:

1. Sound Card Voltage Setting. If using a recommended or legacy sound card, the recommended voltage setting will be set by default. If using an unsupported card, it is necessary to measure the voltage using the procedure described above in the PA Calibration section.
2. Delta 44 Output set to +4dBu. If using the M-Audio Delta 44, it is imperative to follow the Delta 44 Quick Start Guide (www.flex-radio.com/delta44/delta44.htm). One of the most critical settings is the output level. If this is set to +4dBu instead of the specified -10dBv, then the amplifier will be severely overdriven and damage could result.
3. PA Gain By Band is off. It could be that the Gain By Band settings for your Radio/PA/Sound Card are wrong. This could cause the output to be off (although it should not be much). Try re-running the PA Calibration and see if this corrects the problem.

I am getting signal reports that indicate echoing or distortion.

Possible Hardware Causes:

1. Audio feedback due to Monitor. Using the monitor during voice communications is not recommended as sensitive microphones can easily pick up amplified signals. Due to the processing delay, this can sound like a repeated, fading echo. If using the monitor during voice communications is absolutely necessary, headphones are recommended, though it is imperative to use high impedance headphones so as not to load down the drive signal.

Possible Software Causes:

4. Sound Card DSP/Effects. All sound card options and effects such as reverb should be turned off while using the SDR-1000. Any outside effect on the processed PowerSDR audio will cause distortion in the transmitted audio. To turn off DSP and/or effects, look carefully through the sound card mixer and any sound card software that was installed with the sound card driver. If possible, install only the Windows drivers when installing sound card software.
5. Microphone Boost Setting. Most consumer sound cards have a microphone boost option that will add 20dB to the microphone input. If this is turned on and you have a microphone that is putting out a strong signal, then you could be overloading the sound card. Try switching it off to see if the echoing/distortion stops. The Microphone Boost control can usually be found by bringing up the mixer and clicking the Advanced button to bring up another window. Note that it may be necessary to bring up the Recording mixer. To do this, select Options -> Properties on the menu. Then click the Recording radio button and click OK.

I am getting signal reports that indicate I am transmitting DSB while in SSB modes.

Possible Hardware Causes:

1. Line Out cable partially unplugged or damaged. Make sure that the cable going to the plug marked “To Line Out” is fully seated on both ends (enclosure and sound card). A single output channel (I or Q) will transmit in double sideband.

Possible Software Causes:

1. TX Image Reject is set incorrectly. Try resetting the controls on the DSP->Image Reject Tab of the Setup Form to zero for both Gain and Phase.

I am getting an error message that says, “Error communicating with ATU.”

Possible Hardware Causes:

1. Connector broken/not connected. This error is an indication that the software is not seeing the signals from the ATU unit. If you are comfortable opening the enclosure, take the cover off and verify that the larger cables connecting the ATU to the PA are making good contact.

Possible Software Causes:

1. ATU selected with hardware not present. Note that the software will not work if the ATU is selected without the ATU hardware connected.

It seems like the ATU is not tuning correctly. Is there a way to check?

Yes. Checking the tuning is as easy as taking off the top of the enclosure and monitoring the LEDs on the ATU. While tuning, you will see the red LED flash and you should hear the relays clicking until it finishes. If there is an error while tuning (like not enough RF), the red LED will flash. When the ATU is sent a Bypass command, the Green LED will flash three times. The following table details the feedback from the LEDs after a successful tune.

Table 15: ATU LED SWR Indicators

| SWR | Indicator |
|-----------|----------------|
| < 1.5 | Steady Green |
| 1.5 - 2.5 | Flashing Green |
| > 2.5 | No Green |

For more information about the ATU, please reference the manual at:

<http://www.lidgelectronics.com/Z-100%20operations%20manual.pdf>

I get an error when I try to use my USB Adapter that says “USB Device not found. (-1)”

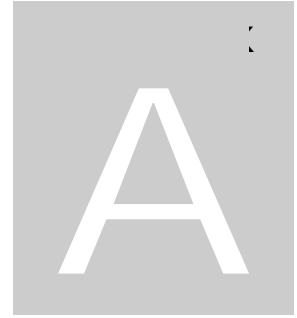
Possible Hardware Causes:

1. USB adapter not present. The USB adapter must be installed and plugged in to use this option. Make sure that the driver is loaded before checking the box.

Possible Software Causes:

1. Driver not loaded. The USB adapter requires that a third party driver be loaded before it will work with the PowerSDR software. This driver may be downloaded on our private download page. Install this driver before checking the USB Adapter box.

[The rest of this page has been left blank intentionally]



Revision History

- 2005-02-24: Updated PowerSDR front panel control descriptions to match current console.
- 2005-02-25: Added Microphone connector information.
- 2005-02-25: Added warning regarding unsupported sound cards.
- 2005-02-25: Added information regarding new Microphone wiring.
- 2005-03-24: Added description for USB To Parallel adapter.
- 2005-03-24: Expanded Plug Pinouts chapter.
- 2005-03-24: Added note in Hardware Setup chapter regarding low pass filter cable.
- 2005-03-24: Added color codes that match our cable set to the hardware setup and installation.
- 2005-03-25: Updated Control Descriptions for Release 1.2.0.
- 2005-03-25: Added more detailed info on how to use the controls and what to expect when you use them.
- 2005-03-30: Expanded Operation Chapter to include ATU, CW and changed/improved SSB to cover all voice modes.
- 2005-03-30: As requested, moved Griffin Powermate Support documentation to Appendix A.
- 2005-07-12: Updated manual for Release 1.4.0.
- 2005-07-12: Removed former Appendix A to separate document (Griffin Powermate Support).
- 2005-07-20: Updated manual for Release 1.4.3.
- 2006-01-10: Updated manual for Release 1.6.0.
- 2006-01-12: Modify manual for references to Delta 44. Added comments for FireBox and updated Microphone configurations.
- 2006-05-31: Restructured manual for consistency and easier accessibility: deleted and cleaned up sections, added 5 layers of headings, redefined headings and much more. Redesigned front cover.
- 2006-06-06: Restructured Chapter 7 Control Descriptions into 3 new chapters: 7-Front Console, 8-Setup Form and 9-Operating Forms. Visually highlighted distinction between Note, Hint, Caution and Warning blocks
- 2006-06-16: Updated Operation chapter (not yet CW) and expanded Digital Operations section to include VAC.
- 2006-06-26: Updated all front console images to 1.6.2, deleted Old Keyer and CW Form, renumbered remaining operating forms, updated CW section of Operation chapter, updated FlexRadio address and deleted ASIO4ALL chapter