



# Zeus-1

(ZS-1)

## Software Defined Transceiver

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# Foreword

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This operation manual will make you familiar with the operation of the digital transceiver ZS-1.

Described are the terminal layout of the sockets, the meaning of control LEDs, the interface, as well as operation of the software in the version ZeusRadio v2.8.1.

We recommend to read this manual carefully before installing the software and taking the device into operation.

# Introduction

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Thank you for choosing the transceiver ZS-1. We hope that you have fun while using your ZS-1.

The Transceiver ZS-1 works as a „direct sampling“ receiver, and also the transmitter signal is created digitally by digital synthesis. The characteristics of the transceiver will be determined in large part by the software or firmware, which is constantly being improved.

The Transceiver ZS-1 is designed for use in radio amateur stations, together with a PC using the operating systems Windows 8, Windows 7, Windows Vista and Windows XP.

Do not open the unit. It does not contain any parts needing maintenance. If you need help regarding technical matters, please contact our team:

[technik@ssb-electronic.de](mailto:technik@ssb-electronic.de)

**Packaging components (e.g. plastic film, polystyrene) can be dangerous for children. Danger of suffocation! Keep away from children.**



### **Disposal of your old appliance**

This product is covered by the European Community directive 2002/96/EC. 2.

All electrical and electronic products should be disposed of separately from the municipal waste stream via designated collection facilities appointed by the government or by the local authorities.

The correct disposal of your old appliance will help prevent potential negative consequences for the environment and the human health.

For more detailed information about the disposal of your old appliance, please consult your city office, waste disposal service or the shop where you purchased the product.

Within Germany, the above regulations are also valid for the disposal of batteries and accumulators accordingly.



### **Declaration of Conformity**

Herewith we declare that this product complies with all relevant regulations for the product:

**EN 301 489-1 V1.8.1 (2008-04)**

**EN 301 489-15 V1.2.1 (2002-08)**

**EN 301 783-2 V1.2.1 (2010-07).**

**EN 60950-1:2006/A1:2010**

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Tel.: +49(0)2941 – 93385 - 0

# Technical Data

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Sensitivity	-141 dBm (MDS)
IIP3	28 dBm
Output power (transmitter)	max. 15W
Bandwidth	up to 100 kHz
Spectrum bandwidth	up to 4 MHz
PC Interface	USB 2.0
Voltage	12-15 V
Current consumption (TX)	4 A

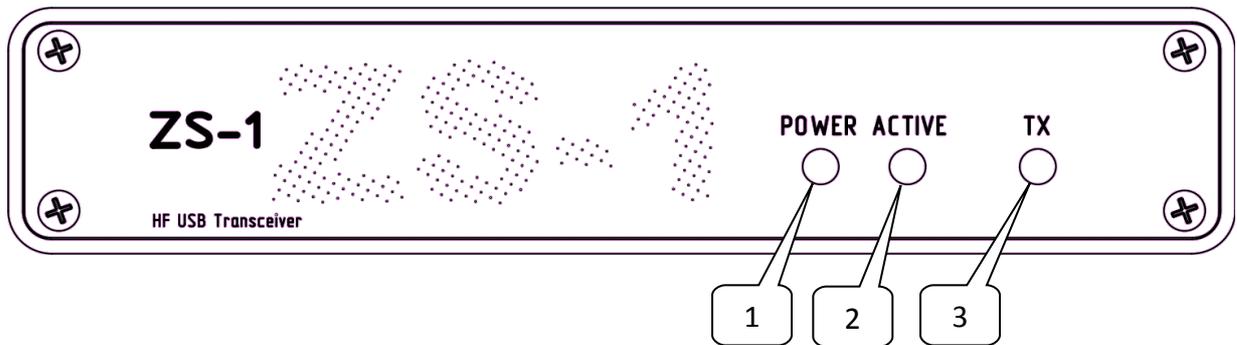
# Supplied accessories

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The Transceiver ZS-1 is supplied with the following accessories:

- |                            |       |
|----------------------------|-------|
| 1. USB 2.0 cable           | 1 pc. |
| 2. DC cable with connector | 1 pc. |
| 3. Ferrite core            | 1 pc. |
| 4. CD with software        | 1 pc. |

## Front panel



### 1 POWER

LED Indicator shows the existence of the supply voltage.

*Note: The indicator does not light up until the transceiver is connected to the PC.*

### 2 ACTIVE

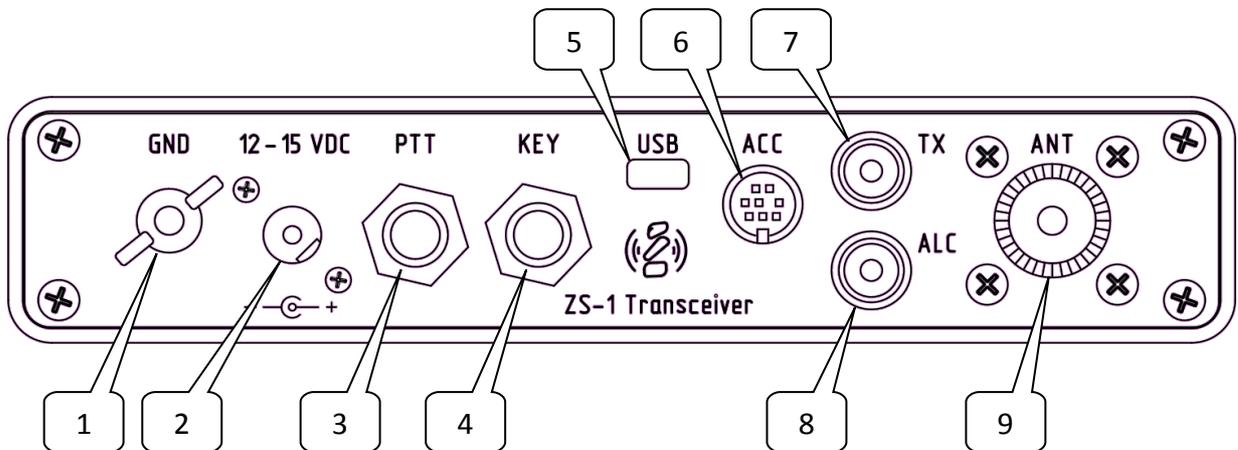
LED Indicator shows the operation mode of the transceiver.

When the ZeusRadio is not used or the USB cable is not connected, the LED indicator will go out and the transceiver is in the stand-by mode.

### 3 TX

LED Indicator for transmission mode.

## Rear panel

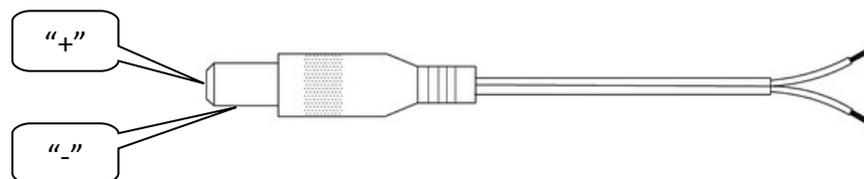


### 1 GND

Grounding terminal. Connect this terminal to a ground to prevent electrical shocks and ensure optimum performance.

### 2 12-15 VDC

Jack for the supply voltage. The inner contact of the connector has to be connected to the positive terminal of the power supply, the outer contact with negative one.



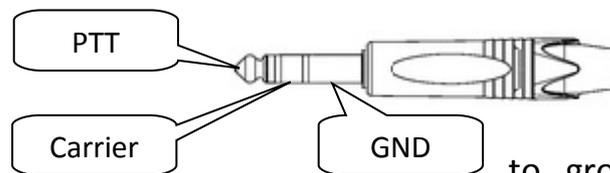
**WRONG POLARITY (REVERSED) WILL DAMAGE THE TRANSCEIVER.**

**BEFORE CONNECTING A POWER SUPPLY TO THE TRANSCEIVER,** please check that the both devices (transceiver and power supply) are grounded. A missing or improper ground connection can be the reason of electrical shock to the operator or can impair proper operation of the transceiver.

Please be sure that no electrical contact exists between the outer contact of the connector (negative terminal) and the transceiver chassis. This could lead to increased RFI.

### 3 PTT

This 6.35mm stereo input socket may be used to provide manual transmitter activation using a footswitch or other switching device.

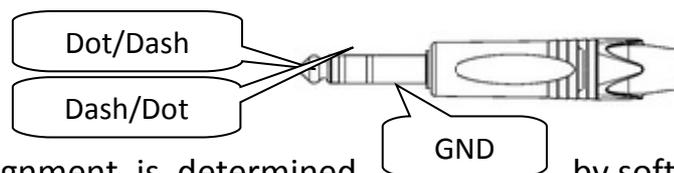


Close the center pin to ground to transmit a signal.

Close the middle contact to ground to transmit an unmodulated carrier for tuning purposes.

### 4 KEY

This 6.35mm stereo input socket accepts a CW key or keyer paddle. A 2-contact (mono) plug cannot be used in this socket.



The contact assignment is determined by software.

### 5 USB

USB 2.0 socket. Use this socket to connect the transceiver to your personal computer.

### 6 ACC

This mini-DIN socket is meant to control external devices, like power amplifiers, external preselector or an antenna switch. Switching algorithm of these 8 control lines (open collector outputs) is determined by software.

The maximum voltage is 24 V, the maximum current is 150 mA.

## **7 TX**

This RCA jack's center pin is closed to the ground (open collector output) while the transmitter is engaged. It may be used to control external linear amplifier.

The maximum voltage is 24 V, the maximum current is 150 mA.

## **8 ALC**

This RCA jack accepts negative-going external ALC (Automatic Level Control) voltage from a linear amplifier, to prevent over-excitation by the transceiver.

Acceptable input voltage range is 0 to -10 VDC.

## **9 ANT**

Connect your 50  $\Omega$  antenna here, using type-UHF (PL-259) plug.

# Precautions

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- The transceiver and other equipment has to be grounded; missing grounding can be the reason of electrical shock to the operator or can lead to malfunctions.
- Immediately turn the transceiver power OFF and remove the power cable if it emits an abnormal odor, sound or smoke.
- Do not open the transceiver. Unauthorized opening can cause the loss of warranty.
- Do not start to transmit without antenna. Broadcasting without antenna or end load can damage the transmission stage!
- Connect the transceiver only to a stabilized supply voltage with 12-15 V DC and at least 5 A. SSB-Electronic recommends the optional power supply **Art.-No. 9371**, as it was tested with the Transceiver in continuous operation
- Never apply more than 15 V DC to the 12-15 VDC jack on the transceivers rear panel. This could cause a fire or damage the transceiver
- Never expose the transceiver to rain, snow or any liquids.
- Do not use chemical agents such as benzine or alcohol when cleaning the transceiver, as they can damage the transceivers plastic surfaces and marking.
- Do not use or place the transceiver in areas with temperatures below 0 °C or above +50 °C.
- Avoid the formation of frost and condensation.
- Set the transceiver so that free air circulation is ensured. Do not place any other objects on the unit, to avoid overheating.
- Turn the transceiver power OFF and disconnect the DC power, antenna and USB cables when you will not use the transceiver for long period of time.

# Installation

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Select the location for the transceiver so the back panel with its sockets remains easily accessible. Free air circulation must be guaranteed; the top panel serves as a radiator and must remain free.

Once you selected the location for the transceiver, first of all connect the ground terminal. Verify that also all other devices connected with the transceiver (the power source, the PC, an amplifier etc.) and are grounded as well.

Missing or bad grounding increases the possibility of electrical shock to the operator, can cause increased noise level in receive mode and interference to home entertainment devices or laboratory test equipment during a transmission.

Connect the transceiver to a power source with an output voltage of 12-15 V.

The operating voltage of the transceiver may be in the range of 12 V to 15 V. However a higher voltage will result in higher power dissipation and transceivers temperature. During reception the transceiver needs about 0,5A, and supply voltage will not cause any heat generation changes. While transmitting, the current rises up to maximum 4 A; the transceivers temperature rises the faster, the higher supply voltage is.

When exceeding the limit of 60 °C the transmitter shuts down automatically.

Connect the antenna.

**BEFORE CONNECTING THE TRANSCEIVER TO AN USB 2.0 PORT OF YOUR PC, THE SOFTWARE HAS TO BE INSTALLED FIRST.**

The software can be found on the CD which is part of the delivery. The most recent version of the software will be made available for download on SSB-Electronic or PARS LLC web pages:

[www.ssb.de/product\\_info.php?info=p3407\\_Zeus-ZS-1-Transceiver](http://www.ssb.de/product_info.php?info=p3407_Zeus-ZS-1-Transceiver)

[www.zs-1.ru/index.php/downloads](http://www.zs-1.ru/index.php/downloads)

# Software installation

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For operating of the ZeusRadio software it is necessary that MicrosoftNetFramework service (v4.0 or higher) is installed on your PC.

Install this service by a double click on the file stored on CD:  
NetFramework\dotNetFx40\_Full\_x86\_x64.exe

To install the ZeusRadio software and the drivers for the ZS-1 transceiver start ZS-1\Setup.msi and follow the instructions on the screen.

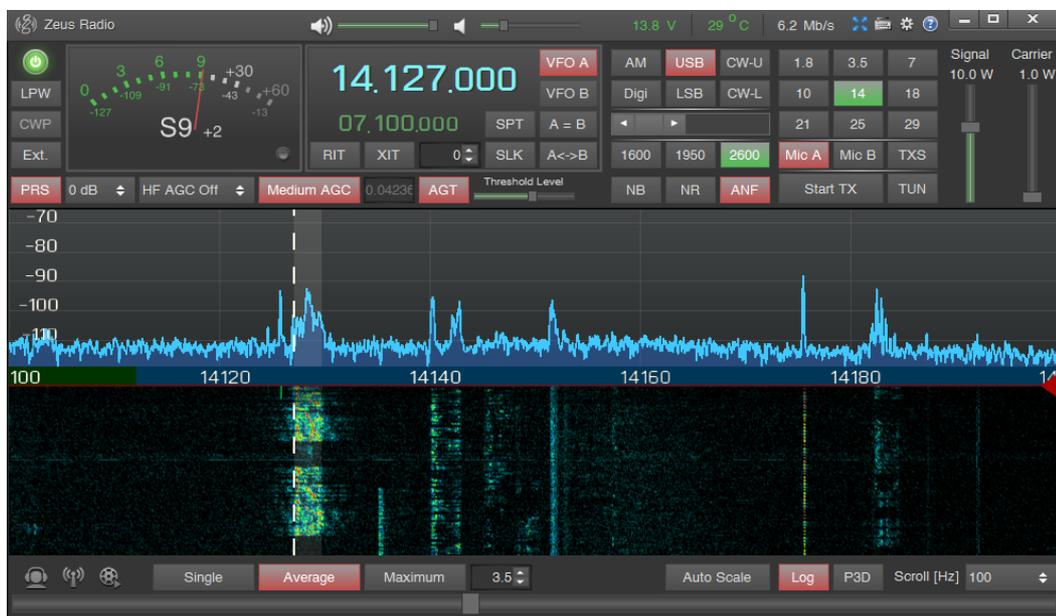
# Software description

In this document the functions and the Graphical User Interface of the software version 2.8.1 are described.

After installation of the software, three shortcuts will appear on the desktop of the PC: ZeusRadio (the ZS-1 operating software), IQ Player (player of recorded broadband IQ files) and HIQSDR (the operating software for HIQSDR transceivers).



After starting the operating software ZeusRadio the main window appears.

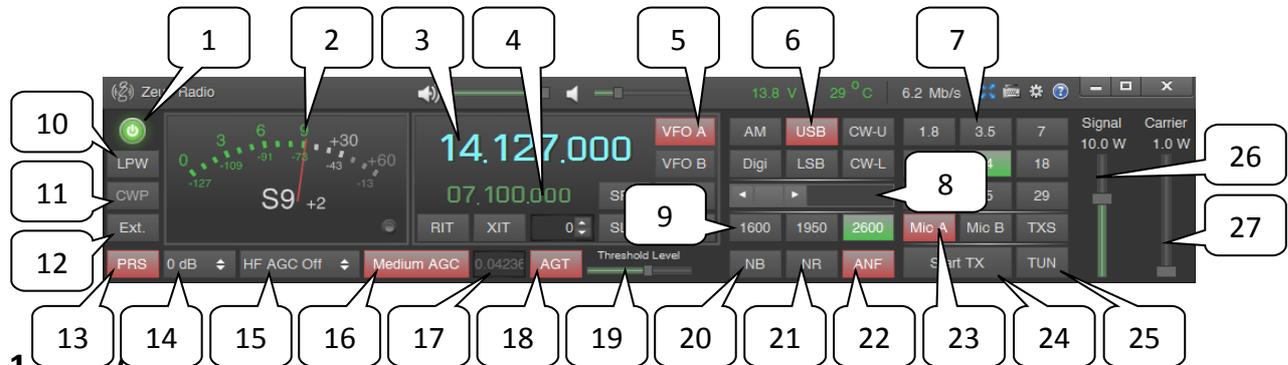


*Note: to start the software it is necessary that the transceiver is connected to the power supply and to the PC with the USB 2.0 cable that is part of the delivery.*

In the main window of the program the most important controls and functions of the operating modes are located.

Other controls are located in additional windows.

## Main window

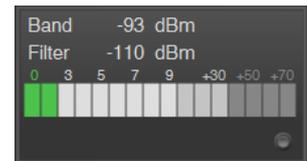


### 1 ON/OFF

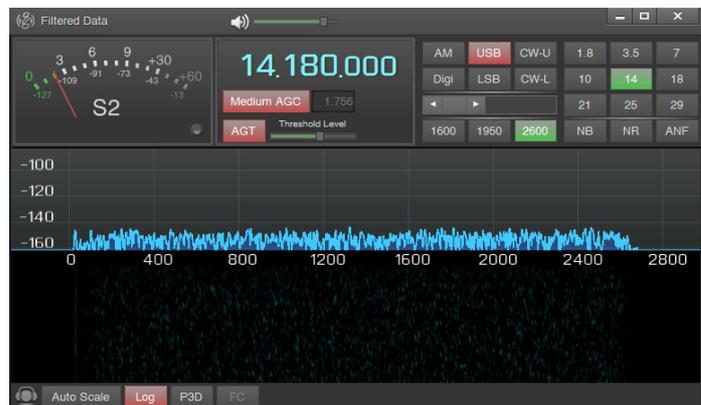
This button switches on and off the software.

### 2 Signal Level

The S-Meter. With a left-button mouse click on the S-Meter field the display changes from analog (arrow) to digital (LED-like) mode and the values of the signal level within the spectrum range as well as within the receiving filter bandwidth are shown in dBm. By repeatedly clicking the left mouse button on the S-Meter the display changes back to the analog mode.



With a right-button mouse click on the S-Meter field an additional window opens, which shows the spectrum of the received signal. The controls of this window are identical to the corresponding controls in the ZeusRadio main window.



In the lower right corner of the S-meter is an overload indicator that glows red when clipping of the analog-to-digital converter (ADC) occurs.

In transmit mode the S-meter displays the signal strength (PEP); the overload indicator shows the presence of such (clipping) in the transmission data path.

*Note: For the S-Meter as well as for level measurements in dBm a quasi-peak detector is used by default, what leads to a slightly higher readings, compared to measurements with the RMS detector.*

## **3 Primary VFO frequency**

## **4 Secondary VFO frequency**

## **5 VFO A, VFO B buttons**

Buttons for selection of a primary VFO.

## **6 Mode selection buttons**

In the actual version the following modes are implemented: AM, USB, LSB, CW-U and CW-L. In the actual version AM, USB, LSB, CW-U and CW-L are implemented. The button Digi is used for digital modes in conjunction with external programs. It works as memory button for audio input output settings, turns off the TX equalizer and compressor, as well as NB, NR, ANF and AGT functions.

*Note: Unlike the classical transceivers, where, for example, in CW-U mode it is possible to tune away from unwanted signal, which cannot be tuned away in CW-L mode, there is no difference in ZS-1 between CW-U and CW-L modes (main selection filters are realized digitally and do have symmetrical edges and an identical suppression to both sides of the pass band). These two modes only serve the convenience of the operator, in the case when the filter is moved out of the CW side tone range (the chosen CW tone is not in the center of the filter), one can easily switch between CW-L and CW-U, without changing the lower or upper filter limits to tune away from unwanted signal.*

## **7 Bands buttons**

## **8 Filter**

Slider to adjust the filter bandwidth.

*Note: The band filter can not be greater than the span of the transceiver and it can not exceed half of the selected sample rate audio.*

## **9 Memory buttons filter width**

Three memory buttons allow you to set the most commonly used band for the reception bandwidth for each mode (AM, SSB, CW) and quickly switch between them.

## **10 LPW button**

On/off switch for low power transmit mode. In this mode the final amplifier is not engaged during transmission. According to the output power slider setting, the antenna gets a signal of up to 40 mW.

*Note: In the LPW transmission is possible at any frequency within the HF band.*

## 11 CWP button

By clicking this button an additional window opens, which allows to tune different parameters of CW keying and to make simple QSO in a text mode.

*Note: Button is active only in CW-U and CW-L modes.*

## 12 Ext. button

By clicking this button an additional window with parameters for operating external devices opens.

## 13 PRS button

On/off switch for the preselector.

The transceiver ZS-1 has high selective band pass filters for the amateur radio bands. Out of the amateur radio bands no preselector is used; all settings are chosen automatically according to the selected frequency.

We recommend to keep the preselector turned on, to avoid overload caused by strong out-of-band signals. Switching off the preselector could be useful for measurement purposes or when using spectrum bandwidths exceeding 2 MHz.

## 14 Attenuator

Dropdown list for the attenuator. To achieve optimum performance without overload, choose the attenuator value according to the current reception conditions and signal levels. A manual selection of the attenuator value is not possible while using the High Frequency Automatic Gain Control (HF AGC).

## 15 HF AGC

Dropdown list for the HF AGC. This AGC analyzes the broadband HF signal and regulates the amplification coefficient according to the existing signal levels.

HF AGC Off – the HF AGC system is switched off. The operator chooses a suitable attenuation manually.

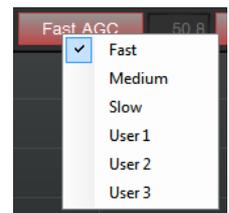
HF AGC Min – the HF AGC system uses an algorithm, which evaluates the noise level in selected band and switches the attenuator in a way that ensures the best possible signal/noise ratio. This setting usually will bring the best signal quality .

HF AGC Max – the HF AGC System uses an algorithm, which evaluates the input signal level and switches the attenuator in a way that avoids any overload of the receiver.

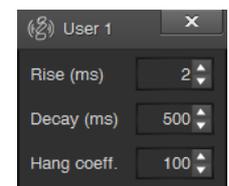
### 16 Audio AGC button

On/off switch for Audio AGC. This system compensates changing levels of the received signal. It should be switched off only in specific measurement tasks.

Audio AGC has three fixed modes (speed): Fast, Medium, Soft – and three tunable: User 1, User 2, User 3. Choosing one of these three modes is made by a right mouse click on the button. Mode is set separately for each demodulator (AM, SSB, CW).



By clicking with the right mouse button on one of the list items User 1-3 a window opens, giving access to settings of this AGC mode. Rise parameter determines the speed of response of the system at increased signal level, while Decay determines the speed of response of the system at reduced signal level. Hang coefficient is dimensionless and can reduce the AGC response to transients. The larger the Hang coefficient, the less AGC responds to the "clicks".



### 17 Audio AGC value

When the Audio AGC is disabled, the amplification factor can be adjusted manually (dB).

### 18, 19 AGT/MGT button, Threshold slider

On/off switch for adaptive or manual gain threshold (AGT or MGT) function and threshold level slider.

When AGT or MGT is switched off, the basic noise level (if there is no signal on selected frequency) will be raised to the value set by the volume control slider.

AGT regulates AGC threshold adaptively, based on the signal levels in the selected band and allows automatic reaction to changing signal levels within the band.

MGT allows limiting of the AGC factor manually to the specified value. This mode can be useful if one or several more powerful stations are in the band, as a selected AGC would substantially reduce the volume level.

# Software description

Main window

Switching between AGT and MGT is done by a right mouse click on the AGT/MGT button.

The AGT (or MGT switched on with a suitable threshold level) will provide the most comfortable reception experience.

## 20, 21, 22 NB, NR, ANF buttons

On/off switch for the noise blanker (NB), the noise reduction (NR) and the automatic notch filter (ANF) algorithms respectively.

With a right-button mouse click on the NB, NR or ANF button the horizontal slider appears, which allows to adjust parameters of the selected algorithm (NB, NR or ANF). Left position of the slider sets the maximum level of reduction/suppression.



*Note: The algorithm of suppression of impulse noise (Noise Blanker) not only removes the noise in the audio signal, but also on the spectrum and waterfall.*

## 23 Mic A, Mic B and TXS buttons

The buttons Mic A and Mic B allow quick changes between two different microphone characteristics for the transmit mode.

By clicking on the TXS (TX Settings) button an additional window with microphone settings opens.

## 24 Start TX button

Button to set the transceiver into the transmit mode. The signal of the chosen audio source of the PC (e.g. microphone input or virtual audio cable) will be transmitted.

The transceiver can be used for transmitting purposes only within the amateur radio bands, as long as the LPW mode is disabled.

Transmitting in the LPW mode is possible on any frequency.

*Note: By pressing the right mouse button on Start TX, the transceiver switches to transmit in carrier mode.*

*Note: For ZS-1 transceivers shipped to the United States of America the transmission is permitted only in the amateur radio frequency bands, both at full power and in the LPW mode.*

## 25 TUN button

On/off switch for a carrier transmission.

Transmitting in the LPW mode is possible on any frequency, full power is available only within the amateur radio bands.

This function can be used for tuning and adjustment purposes.

## 26, 27 Signal, Carrier sliders

Sliders for the output power adjustment. For signal and carrier respectively.

Slider positions are stored independently for each band and mode (AM, SSB, CW, DIGI), and also for the low power mode (LPW) and for an external amplifier.

*Note: Although the transceiver is designed to provide 15 W of output power, tested in continuous operation at this power and has an overheating protection system, it is advisable to set, for example, at half of the output power. This will decrease the signal by only 0.5 points on the S-meter, but will significantly reduce power consumption and heat dissipation, and the transmitted signal will be cleaner. If necessary, will also be an opportunity to increase the output power to the maximum.*



### 28 RIT, XIT buttons

RIT and XIT buttons turn on and off corresponding functions: "Receiver Incremental Tuning" and "Transmitter Incremental Tuning".

RIT – receiver frequency deviation related to the primary VFO frequency; the primary VFO frequency will be used for transmission.

XIT – transmitter frequency deviation related to the primary VFO frequency; the primary VFO frequency will be used for reception.

### 29 Deviation frequency

Input field for the frequency deviation (RIT/XIT) in Hz. The value can be adjusted with the mouse wheel also.

### 30 SPT button

On/off switch for the SPLIT function. When active, reception occurs on the frequency of the currently active (primary) VFO (with the chosen demodulator and filter bandwidth), while for transmissions the frequency of the other (secondary) VFO is used (with the demodulator and filter bandwidth chosen there).

### 31 SLK button

On/off switch for locking the frequency difference between the primary and the secondary VFO – Split Lock. When SLK is switched on, frequency changes of the active (primary) VFO are transferred synchronously to the secondary VFO. A frequency change of the secondary VFO will have no effect to the frequency of the active (primary) VFO.

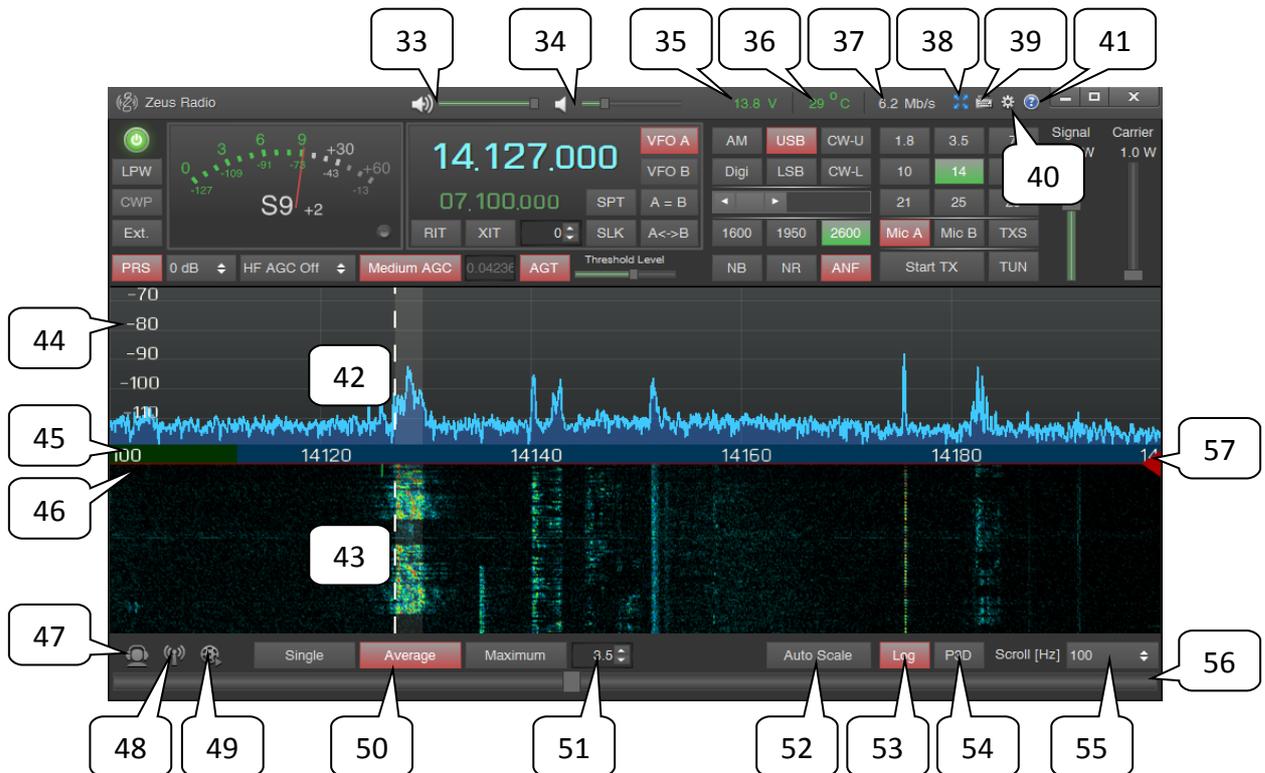
### 32 A=B, A<>B buttons

The button A=B causes an equalization of the secondary VFO frequency and the active (primary) VFO. The button A <> B causes a change of VFO frequencies: VFO A becomes VFO B, and vice versa.

These buttons do not change VFOs modes (AM, SSB, CW).

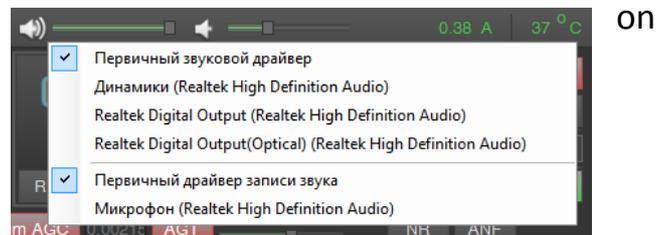
# Software description

Main window



## 33 Volume control slider

By pressing the right mouse button on the speaker icon, a list of select audio devices displays DirectSound or ASIO control panel used by the driver.



## 34 Monitor volume slider

Monitor signal in SSB and AM modes, output from that DSP unit, on which the loudspeaker icon is activated in the TX Signal Settings window.

## 35 Supply voltage and current consumption indicator

By clicking the display with the left mouse button the it changes between voltage and current consumption indication.

Exceeding the maximum values will lead to a warning message on the screen, the transceiver goes into the standby mode and the ZeusRadio software will be closed automatically.

If one of the parameters (voltage or current) falls below the die minimum permitted values, the color of the display changes from green to yellow. The transceiver will still work, but its parameters may be degraded, compared to the specifications.

In the case you encounter a current consumption out of the permitted range, please stop using the transceiver and contact your dealer.

### 36 Transceivers temperature indicator

The display informs about the mean temperature inside the transceiver and therefore it is a bit lower as the temperature of the transmitters final stage transistors. This way the temperatures of all sensitive components of the transceiver are controlled and the normal thermal state of the final stage ensured.

The background color of the temperature indicator changes in the same way as in the voltage and current indicator. If the maximum temperature values are exceeded, operation of the transceiver will be stopped.

### 37 Data transfer rate indicator

The indicator shows the data speed via USB interface.

The speed depends on the spectrum bandwidth. Because of the maximum resolution of the displays is used for bandwidths up to 100 kHz, the 100 kHz setting requires the maximum USB speed.

Bandwidth	Data speed	Bandwidth	Data speed
10 kHz	0,8 Mb/s	320 kHz	4,2 – 4,3 Mb/s
20 kHz	1,6 Mb/s	800 kHz	4,2 – 4,3 Mb/s
40 kHz	3,3 Mb/s	1,6 MHz	4,2 – 4,3 Mb/s
100 kHz	6,1 – 6,2 Mb/s	4 MHz	4,2 – 4,3 Mb/s
160 kHz	4,2 – 4,3 Mb/s		

For various reasons the data speed can be lower than needed. This means that the ZeusRadio software gets less data than needed for proper operation, which is manifested by audio dropouts and very high noise floor.

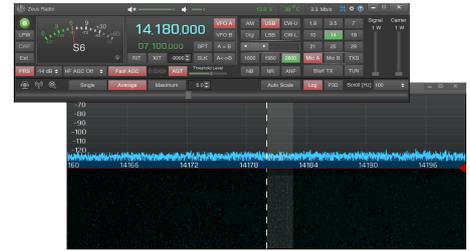
Also the use of a too long or inferior USB cable or a PC with an insufficient CPU may be the cause of too low transfer rates. Avoid using of any USB hubs (also active USB Hubs!).

# Software description

## Main window

### 38 Button for main window separation

This button allows to separate all of the controls from spectrum and waterfall. This way, for example, these two windows may be placed on different monitors. A further click will reunite both windows.



### 39 Hotkeys

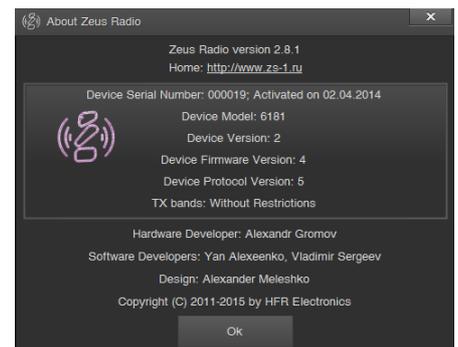
By clicking on the hotkeys button an additional window with keystrokes and MIDI controls settings opens.

### 40 Settings

By clicking on the settings button an additional window with important settings of the ZeusRadio software opens.

### 41 Info

By clicking on the info button an additional window with the basic information about transceiver and software opens.



### 42 Spectrum display

### 43 Waterfall display

### 44 The vertical axis of the spectrum

Depending on the chosen setting (linear or logarithmic) the values are shown in  $\mu\text{V}$  (microvolt) or dBm.

By moving the mouse up or down with the left button pressed the vertical scale of the spectrum will be adjusted.

By moving the mouse up or down with the right button pressed the reference value of the spectrum will be adjusted.

### 45 The horizontal axis of the spectrum (frequency axis)

Frequency values are displayed in kHz.

By moving the mouse to the left or right with the left button pressed the shown frequency range is shifted. The VFO frequency will not change as long as it remains within the visible frequency range.

By moving the mouse to the left or right with the left button pressed the displayed bandwidth can be changed continuously from 10 kHz up to 4 MHz.

## 46 Spectrum/waterfall ratio control field

By moving the line below the frequency axis upwards or downwards with the left mouse button pressed the vertical size of the spectrum and waterfall can be changed.

## 47 Audio signal recording button

By left-button mouse click an audio file recording is started. A recording includes received and transmitted signals. During a recording the button blinks. A further click will stop the recording.

By right-button mouse click a dialogue window will appear for defining the folder where an audio files will be stored. When selecting the folder it has to be ensured that the user has rights to write into this folder. Else no recordings can be stored.

It is possible to choose the file type of audio recordings: wav or ogg. OGG files are compressed and therefore smaller. WAV files are bigger, but are suited better for later analysis or processing.

## 48 IQ data recording button

By left-button mouse click an IQ data file recording is started. During a recording the button blinks. A further click will stop the recording. While recording IQ data, transmitting is not possible.

Recording of IQ data file is possible for spectrum bandwidths up to 100 kHz.

Recording can be replayed with the IQ Player software.

By right-button mouse click a dialogue window will appear for defining the folder where an IQ data files will be stored. When selecting the folder it has to be ensured that the user has rights to write into this folder. Else no recordings can be stored.

*Note: Recorded files can be huge. A one minute long recording of 100 kHz bandwidth will have a file size of 43 MB. A maximum file size of 200 MB is defined by software; as soon this value is reached, a new file is created automatically. Recording that consists of several files are treated as a single file by the IQ Player software and will be replayed without interruption.*

## 49 Button for audio file transmission

Clicking this button will replay an audio file; simultaneously this file is transmitted. This allows a comfortable modulation control for the partner station or repeat your call sign some times.

## 50 Single, Average, Maximum buttons

These buttons set the display mode of the spectrum:

Single — quick signal display, suited also for short time signals.

Average — display of averaged values; the illustration is smoother.

Maximum — display of maximum values.

## 51 Buffer field

This field sets the buffer size for spectrum processing.

If „Average“ or „Maximum“ is chosen for the spectrum display, averaging (or the definition of the maximum) process is done using the spectrum quantity set in this field.

## 52 Auto Scale button

On/off switch for the auto scaling function of the vertical spectrum axis.

For auto scaling, the average power of the displayed spectrum range is used, while the value is defined in a way that relatively weak stations can be seen while strong stations are present.

## 53 Log button

On/off switch for the logarithmic display of the spectrum.

## 54 P3D button

On/off switch for the pseudo 3D display of the spectrum.

## 55 Scroll field

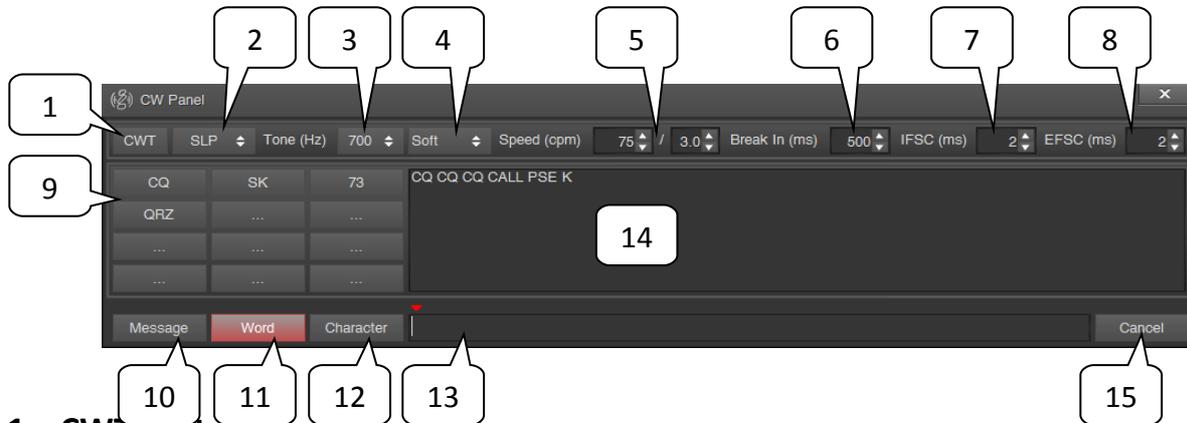
Input field for tuning steps of the VFO via the scroll wheel of the mouse. Predefined tuning steps are available, additionally tuning steps can be set via the keyboard.

## 56 Quick access bar for selecting a frequency

With the slider on this bar the transceiver can be tuned quickly with the mouse to any frequency, from low frequencies on the left to high frequencies on the right side of the bar.

## 57 Time Machine slider

## CW Panel window



### 1 CWT button

On/off switch for the telegraph key or paddle tuning mode (CW Test). When this setting is used, the transceiver, when set to CW-L or CW-U, will not switch to the transmitter mode, so tone, speed and other values can be set without transmitting any signal.

### 2 Key Type list

Dropdown list for mode selection of the automatic keyer.

KEY – straight key; closing the tip of the KEY connector to ground causes side tone generation.

SLP – single-lever paddle; closing the tip of the KEY connector to ground creates "dots", the middle one – "dashes".

SLP inv. – single-lever paddle with reversed contacts; closing the central pin of the KEY connector to ground creates "dashes", the middle one – "dots".

IP A – iambic paddle mode A; closing the central pin of the KEY connector to ground creates "dots", the middle one – "dashes".

IP A inv. – iambic paddle mode A with reversed contacts; closing the tip of the KEY connector to ground creates "dashes", the middle one – "dots".

IP B – iambic paddle mode B; closing the tip of the KEY connector to ground creates "dots", the middle one – "dashes".

IP B inv. – iambic paddle mode B with reversed contacts; closing the tip of the KEY connector to ground creates "dashes", the middle one – "dots".

### 3 Tone list

Dropdown list for selecting the frequency of the CW side tone in Hz (Pitch).

### 4 Hardness list

Dropdown list for selecting the "hardness" of CW tone.

Soft – "soft" CW signal (slowest slew rate).

Medium – medium hardness.

Hard – aggressive signal (fastest slew rate).

### 5 Speed

Fields for selecting manipulation speed and relation between dots and dashes.

### 6 Break In

Input field of the switchover time into the reception mode after finishing a transmission (Semi Break-in Delay).

*Note: It is necessary to determine this parameter according to the CW-speed. Too short break-in delay will cause a switchover even between dots and dashes and will lead to a faster wear of the changeover relay.*

### 7, 8 IFSC, EFSC

Input field for the first transmitted symbol length correction.

Due to RX-TX switchover delays in the transceiver or in an external linear amplifier, the first transmitted symbol may be distorted or „shortened“. To overcome such an effect, the duration of the first symbol can be corrected.

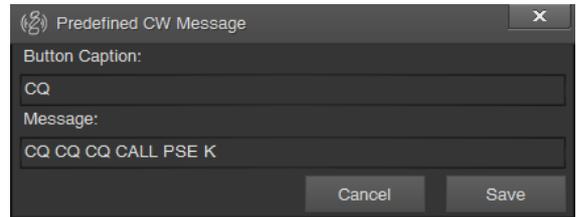
IFSC (Internal First Symbol Correction) – correction of the first symbol on transceivers switchover delay; a value of 2-3 ms is sufficient. It is possible, however, to set higher values for special applications.

EFSC (External First Symbol Correction) – correction of the first symbol on external devices switchover delay (linear amplifier, antenna switch etc.). This value should be set greater than the value for TX On Delay in the External control window.

### 9 Templates buttons

12 buttons with message templates. With a click on one of the buttons the corresponding text will be taken into the message input field. The content of these buttons may be defined by the user.

By pressing a right mouse button a new edit window opens. As well as the text of the template message, the button caption can be set by the user. After completion of a editing, the "Save" button has to be clicked.



### **10, 11, 12 Message, Word, Character buttons**

Buttons for selection of the CW message transmission mode:

Message – the message will be transmitted after a click on the „Send“ button or the „Enter“ key on the keyboard.

Word – the message will be transmitted after pressing the space bar, or the „Enter“ key, or after a click on any of the templates buttons.

Character – each input will be transmitted immediately.

*Note: By pressing the Esc key on your keyboard, the transmission of the message is interrupted.*

### **13 CW message input field**

### **14 History of transmitted messages**

### **15 Send/Cancel button**

After a click on this button „Send“ the text in the input field will be transmitted. A further click will cancel transmission.

## Settings window

The window is opened by clicking the Settings button. It contains three tabs: Main (common software parameters), Audio (audio parameters setup) and Server Mode (remote control setup).

### Main tab

In the Main tab, parameters are grouped into sections.

#### Background section

Here the background color of the spectrum can be defined (colors at the top of the spectrum area and at the bottom are set separately).

#### Frequency Plan section

On the horizontal (frequency-) axis of the spectrum a color scheme represents the modulation mode which is used preferably in the according frequency range. The color assignments are definable via dropdown menus.

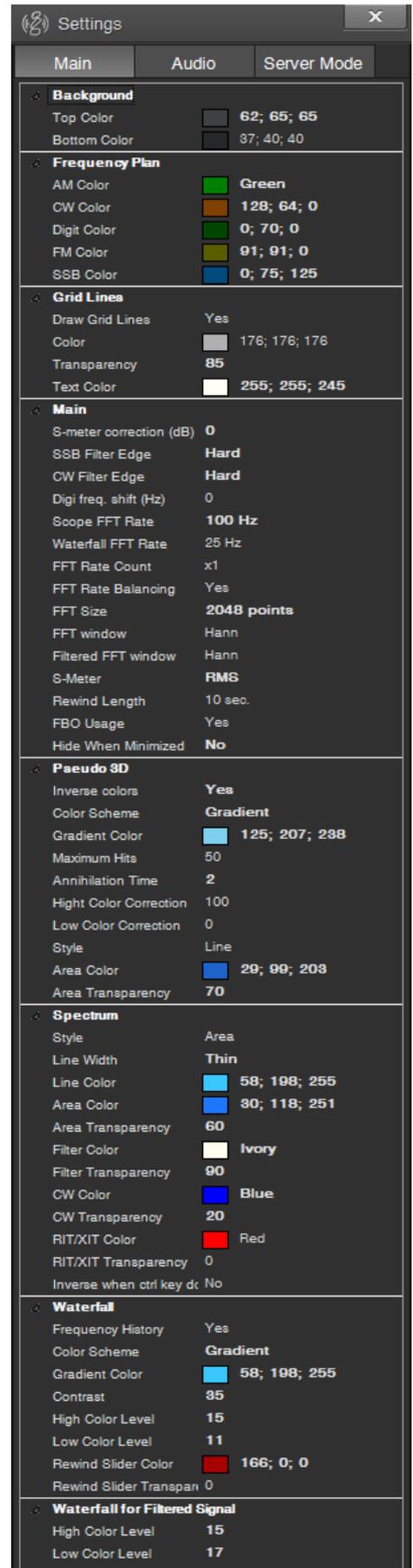
#### Grid Lines section

*Draw Grid Lines* - on/off switch for a raster display in the spectrum area.

*Color* — color of the raster.

*Transparency* — the transparency of the raster lines.

*Text Color* — text color of the frequency bar and the level (axis values).



# Software description

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Settings window (Main tab)

## Main section

Section with parameters of the digital signal processing and its display.

*S-meter correction (dB)* — correction value for S-meter readings.

*SSB Filter Edge* — sets the steepness of the filter edges in SSB and AM modes. Hard — high steepness, Medium — medium steepness, Soft — low steepness.

*Note: As higher the steepness is set, the bigger the latency between HF and audio signals will be, but the better the suppression of neighboring stations will become.*

*CW Filter Edge* — sets the steepness of the filter curves in CW mode (analog to SSB Filter Edge).

*TX Key* — sets the button of the keyboard for switching the transceiver in the transmit mode. One mouse click activates the transmitter, a further click will switch back into the receive mode.

*Note: Selected button will led the device in the transmit mode even if the program window is not the active window.*

*DIGI freq. shift* — adjustment of the frequency shift in mode DIGI (analogously to CW Tone).

*Scope FFT Rate* — refresh rate of the spectrum display.

*Waterfall FFT Rate* — refresh rate (speed) of the waterfall.

*FFT Rate Count* — preset of the number of calculated spectra per time unit. A high value requires more PC power, but the spectrum display will look more "active". This parameter has to be increased, to get most out of the function "Pseudo 3D".

*FFT Rate Balancing* — on/off switch for the increased number of calculated spectra for small spectrum bandwidths; the "Yes" setting requires more PC power, but the spectrum display will look more "active". This is most effective for spectrum bandwidths of 10 - 20 kHz.

*FFT Size* — spectrum resolution in points. A high value enlarges the resolution of the spectrum and waterfall but requires more PC power.

*Note: For spectrum bandwidths of more than 100 kHz the maximum resolution of the spectrum is 1024 points. If a higher value is selected, it will be used automatically for spectrum bandwidths of up to 100 kHz only.*

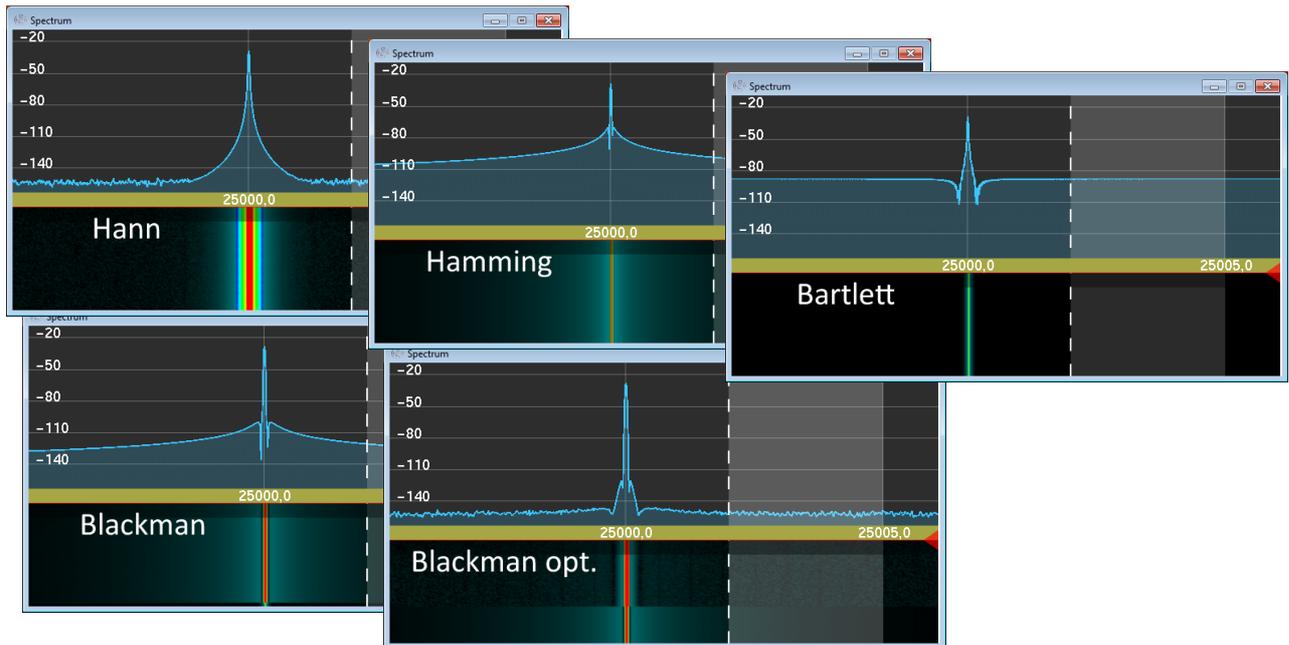
*FFT Window* — list for selecting the weighting window for the fast Fourier transform used for spectrum and waterfall display.

# Software description

## Settings window (Main tab)

In most cases, the Hann window will give the best results, by the aspect of main lobe width and sidelobes level.

*Note: The chosen weighting window for the fast Fourier transform has no impact on the characteristics of the digital filters, the S-Meter values or other signal processing algorithms. The selected weighting window only influences the spectrum and waterfall resolution.*



**S-meter** — selection of the detector used for S-meter. Peak – quasi-peak detector, RMS – root mean square detector.

**Rewind Length** — signal buffer size used in Time Machine function. ZeusRadio software must be closed to the changes to take effect.

**FBO Usage** — on/off switch for FBO usage. FBO is used by default and the best graphic performance is achieved. But in this case the waterfall can be missing on weak PCs and FBO usage should be turned off. ZeusRadio software must be closed for changes to take effect.

**Hide When Minimized** – show in the task bar or hide the program window to tray: If set to "No", the window will remain in the task bar after minimization.

### Pseudo 3D section

Here the parameters of the Pseudo 3D display are set.

**Inverse Colors** — on/off switch for the color scheme inversion.

**Color Scheme** — selection of color scheme.

**Gradient Color** — color selection for the „Gradient“ color scheme.

# Software description

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Settings window (Main tab)

*Maximum Hits* — preset how quick a maximum color is achieved.

*Annihilation Time* — the afterglow time setting.

*High Color Correction* — setting for the "upper" color level (the gradation downwards in the color scheme).

*Low Color Correction* — setting for the "lower" color level (the gradation upwards in the color scheme).

*Stile* — the style of the display: with (area) or without (line) colored filling below the spectrum.

*Area Color* — the color of the filling.

*Area Transparency* — the transparency of the filling.

## **Spectrum section**

Herewith the spectrum display can be adjusted.

*Style* — the style of the display: with (area) or without (line) colored filling below the spectrum.

*Line Width* — the width/thickness of the line.

*Line Color* — the color of the line.

*Area Color* — the color of the filling.

*Area Transparency* — the transparency of the filling.

*Filter Color* — the color of the frequency mark and the filter bandwidth on the spectrum display.

*Filter Transparency* — the transparency of the filter bandwidth.

*CW Color* — color of the CW side tone marker.

*CW Transparency* — the transparency of the CW side tone marker.

*RIT/XIT Color* — the color of the transmission frequency marker in RIT, XIT and SPLIT modes.

*RIT/XIT Transparency* — the transparency of the transmission frequency marker in RIT, XIT and SPLIT modes.

*Inverse when CTRL key down* — inversion of the CTRL key action for frequency tuning.

## **Waterfall section**

Herewith the waterfall display can be adjusted.

*Frequency History* — on/off switch for the horizontal shift of the waterfall display, synchronously to a change of the spectrum bandwidth.

*Color Scheme* — selection of color scheme.

# Software description

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Settings window (Main tab)

*Gradient Color* — the color selection for the „Gradient“ color scheme.

*Contrast* — adjustment of the waterfall contrast.

*High Color Level* — signal level, above which the signal is displayed with "maximum" color of the chosen color scheme.

*Low Color Level* — signal level, above which the signal is displayed with the "minimum" color of the chosen color scheme.

*Rewind Slider Color* — the color of the Time Machine slider.

*Rewind Slider Transparency* — the transparency of the Time Machine slider.

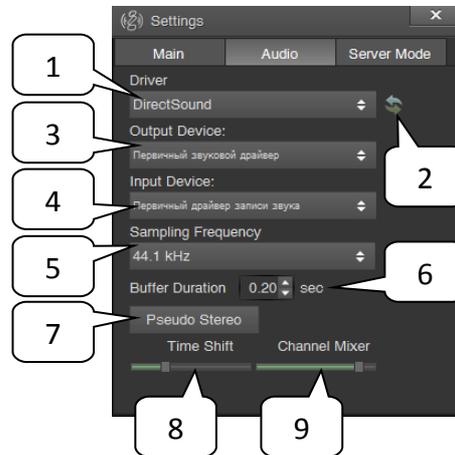
## **Waterfall for Filtered Signal section**

Herewith the waterfall display in the Filtered Signal window can be adjusted.

*High Color Level* — signal level, above which the signal is displayed with "maximum" color of the chosen color scheme.

*Low Color Level* — signal level, above which the signal is displayed with the "minimum" color of the chosen color scheme.

## Settings window, Audio tab



### 1 Driver list

Dropdown list for selecting the audio driver for audio devices.

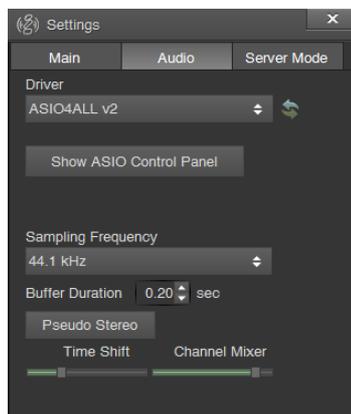
### 2 Button Refresh

Button to refresh the list of audio devices.

### 3, 4 Output Device, Input Device lists

Dropdown lists for selecting audio input and output. The content of these lists depends on the configuration and the installed software of the PC. The lists may contain analog and digital inputs/outputs of the sound card, as well as Virtual Audio Cables (special software).

*Note: These lists are present only if the DirectSound driver is selected. For an ASIO driver (if present on the system) instead of these two lists the button "Show ASIO Control Panel" is shown. After a mouse click the window for management of the chosen ASIO driver appears, which allows to select the available devices for the audio input and output.*



## 5 Sampling Frequency list

Dropdown list for selecting the audio sampling frequency.

## 6 Buffer Duration

Input field for the audio data buffer length. As higher the value is set, the bigger the time difference between HF and audio signals will be. The minimum value depends on the characteristics of the used audio device (sound card), but cannot be less than 0.08 sec.

## 7 Pseudo Stereo button

On/off switch for pseudo stereo mode.

*Note: This function is most useful when using a headphones. When using loudspeakers, the spatial effect may be less noticeable.*

## 8 Time Shift slider

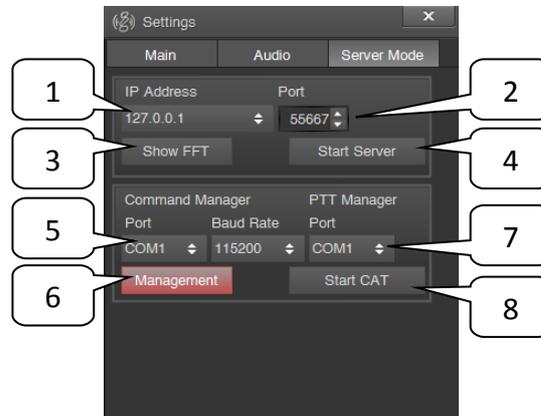
Adjustment of the audio signal delay. Delay is 0,2 ms when the slider is in the left position and 10 ms in the right position.

## 9 Channel Mixer slider

Adjustment of the signal distribution between the stereo channels. In the leftmost slider position the signal of the left channel is reproduced without any delay, while the signal of the right channel is heard with the delay that was set with the Time Shift slider. In the rightmost slider position the relation is reversed: the signal of the right channel is reproduced without any delay, while the signal of the left channel is heard with delay.

*Note: In fact the Time Shift slider sets the depth of the stereo effect, and the Channel Mixer slider regulates the location of the apparent source of the spatial sound.*

## Settings window, Server Mode tab



### 1 IP Address list

Dropdown List for selecting the servers IP address.

### 2 Port

Input field for the port number.

### 3 Show FFT button

On/off switch of local spectrum and waterfall display during remote operation.

*Note: When connected to a server (remote operation) the display of spectrum and waterfall in the ZeusRadio software is stopped. For enabling these functions, the Show FFT button has to be activated.*

### 4 Start Server button

This button starts/stops the server for remote operation.

### 5 Port, Baud Rate lists

Dropdown lists for the selecting of a COM port for the CAT Interface and choosing the connection speed.

### 6 Management button

On/off switch for transceiver control with the CAT Interface. When switched off, the chosen COM Port will transfer outgoing transceiver parameters (like changes of frequency or modulation mode etc.), but will not recognize control commands sent to the transceiver.

# Software description

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Settings window (Server Mode tab)

## **7 PTT Manager Port tab**

Dropdown list for selecting a COM Port for PTT control (change to the transmit mode) of the transceiver and for connection of a Morse key. For PTT the RTS signal, and for the Morse key the DTR signal of the chosen COM Port are used.

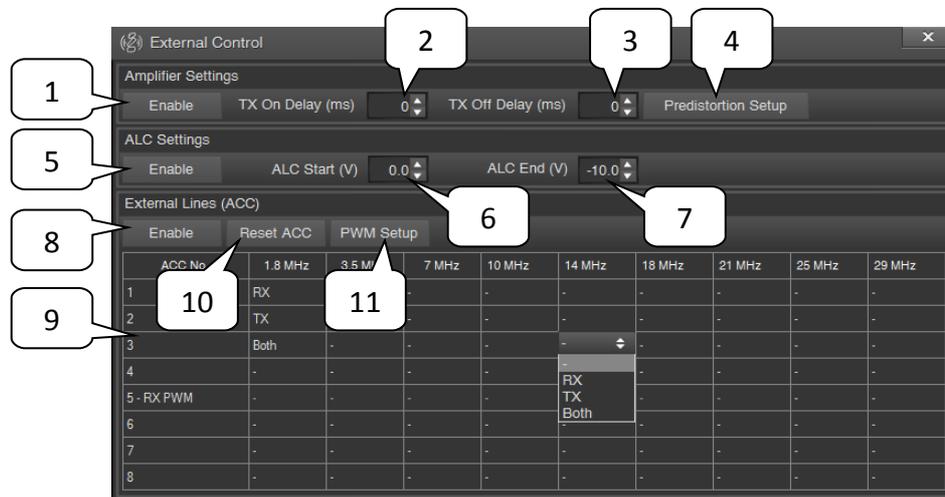
*Note: It may be used the same COM Port as for the CAT Interface.*

## **8 Start CAT button**

On/off switch for the CAT Interface (with the chosen parameters).

## External Control window

This window appears by clicking the Ext. button in the main window.



## Amplifier Settings

### 1 Enable button

On/off switch for the control of an external amplifier, which has to be connected to the TX socket at the rear panel of the transceiver. When the transceiver is switched into the transmit mode, the output is closed to ground.

### 2 TX On Delay field

Delay in ms for the activation of an external amplifier after switching the transceiver into the transmit mode.

*If this value is negative, the external power amplifier turns on first, and then the transceiver switches to transmit mode.*

### 3 TX Off Delay field

Delay in ms for the deactivation of an external amplifier after switching the transceiver into the receive mode.

*Note: If this value is negative, the external power amplifier turns off first, and then the transceiver goes into receive mode.*

### 4 Predistortion Setup Button

By a click, a window to set up predistortion algorithms appears.

## ALC Settings

### 5 Enable button

On/off switch for the Automatic Level Control (ALC).

### 6 ALC Start field

The maximum voltage at the ALC input is displayed, which is supposed to be admissible for the used external linear amplifier. If the voltage at ALC input is higher than this value, then the transceivers output power won't be reduced.

### 7 ALC End field

The minimum voltage at the ALC input, which will cause the transceivers zero output power.

*Note: The value in the ALC Start field should be higher than the value in the ALC End field. For example: ALC Start = -1 V, ALC End = -8 V.*

## ACC Settings

### 8 Enable button

On/off switch of the control for external devices connected to the ACC socket at the rear panel.

### 9 ACC table

In the table the status of all 8 lines for each frequency range can be set.

Dash – the line is switched off.

RX – the line is switched on during reception.

TX – the line is switched on during transmissions.

Both – the line is switched on permanently.

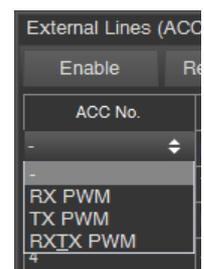
It is also possible to set pulse width modulation (PWM) on any line. To do this, click on the cell with the number of the line and select the PWM mode in the list that appears. 4 modes of PWM are implemented.

Dash - PWM is not used on this line, its state can be set up using the table.

RX PWM - PWM will be turned on in the receive mode.

TX PWM – PWM will be turned on in the transmit mode.

RX&TX PWM – PWM is turned on in both modes.



### 10 Reset ACC button

By pressing this button all the settings in the ACC table are reset to their initial state - all lines are switched off.

### 11 PWM Setup button

By a click, a window to set up PWM appears.

Temp button sets the mode, in which the parameters of the PWM depends on the temperature of the transceiver. It is possible to set the temperature (Min Temp), above which pulses appear with a predetermined duty cycle (From PWM) on the lines of the ACC connector. As the temperature increases the duty cycle will vary. And at the maximum temperature (Max Temp) will be equal to the value specified in the "To PWM" field.



Const button turns on constant duty cycle (the field PWM).

PWM parameters are specified separately for transmit and receive.

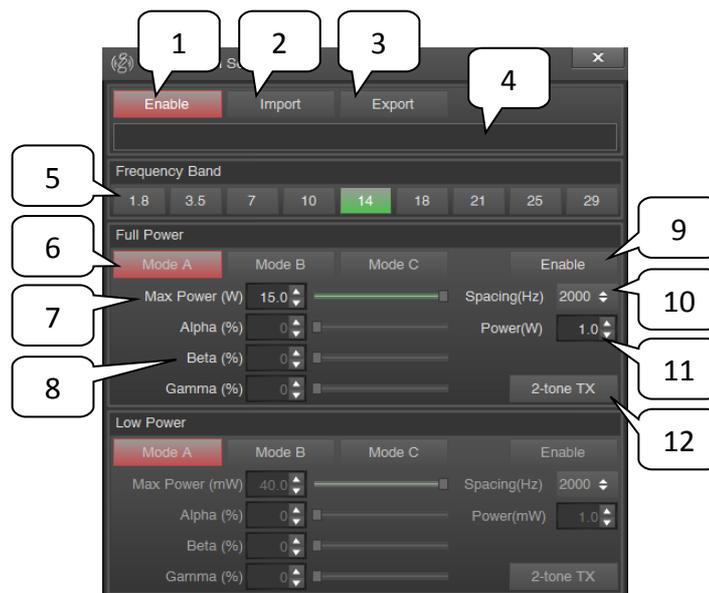
*Note: One example of the PWM control may be the speed of an external fan which can be used to prevent overheating of the transceiver or an external power amplifier, as well as to hold the transceiver's temperature at fixed value, which will guarantee a very high frequency stability of the internal reference oscillator.*

## Predistortion Setup

The Settings window for the predistortion of the transmission signal opens on pressing "Predistortion Setup" in the External Control window.

*Note: Since the main purpose of these algorithms is distortion compensation arising in external power amplifiers, while the compensation of the internal amplifier of the ZS-1 has already been made, the majority of the controls in the window become active only when the external amplifier control is enabled (Enable button in the Amplifier Settings section of the External Control window). The same applies to pre-emphasis of the transmitter signal.*

**WARNING** Misconfiguration of the predistortion transmission signal algorithms or use of them without an external amplifier can lead to a significant deterioration of the transmission signal!



### 1 Enable Button

Enable / disable the predistortion algorithm.

### 2 Import Button

Button to import configuration from a file.

### 3 Export Button

Button to export configuration to a file.

### **4 Comments Field**

Comment input field to a given set of settings.

### **5 Buttons for selection of band-dependant predistortion algorithms**

### **6 Mode A, B, C Buttons**

Button selects one of three predistortion algorithms.

### **7 Max Power Parameter**

Max Power parameter sets the maximum output power of the transceiver with an external amplifier in the selected frequency range, ie limits the maximum value for the ZS-1 Signal and Carrier power adjustment in the main program window.

### **8 Alfa, Beta, Gamma Parameters**

The coefficients of the predistortion algorithm for the selected frequency band.

For coarse adjustment parameter can be set by arrows up / down in the input field or rotate the mouse wheel. Keyboard input of a certain value is also possible.

### **9 Enable Button**

Enable / disable the predistortion algorithm in the selected frequency range.

### **10 Spacing List**

Drop-down list, selects the separation of the two tones.

### **11 Power Field**

Field for the power of two-tone signal (PEP).

### **12 2-tone TX Button**

On/off switch for two-tone signal transmission with given power and spacing between tones.

The controls located in the section intended for Low Power setting algorithms at a low output power (LPW), to be activated by pressing "LPW" in the main window.

## TX Signal Settings window

This window appears by clicking the TXS button in the main window.

In this window are controls for signal processing algorithms of the transmission signal, which are arranged downward in the order of signal flow.

Each block has its own output signal level display, overload (clipping) indicator and the speaker icon, which allows to monitor the output signal of this block.



### 1 Sample Recording Group

Clicking on the microphone icon starts recording audio from selected audio input. Button starts flashing. Pressing again stops the recording.

Clicking on the tape icon allows you to listen to a recording without any processing.

Here is also the recording duration indicator, (FD).

*Note: After you change the audio sample rate you need to record a new file.*

### 2 TX DSP Tuning Group

Clicking on the microphone icon starts signal processing algorithms for selected audio input and the result goes to the selected audio output. Thus, one can make adjustments of the algorithms in real-time.

# Software description

## TX Signal Settings window

Clicking on the tape icon starts the looped replay of the recorded audio sample, allowing to make adjustments of the signal processing algorithms using the audio sample recording.

Mic DSP A and Mic DSP B are the selection buttons for two separate sets of microphone settings.

### 3 Import and Export Settings Group

Pressing the Export button saves the current settings (Mic DSP A or B) to the file.

Press the Import button to read settings from the file.

The Reset button resets settings to its original state.

*Note: To save a set of settings, you can add a text label or specific name. By default sets are called Default MIC A and Default MIC B. This text can be replaced by any other.*

### 4 Microphone Gain Group

There is a button to turn on this processing unit (Enabled), and the input field for the microphone signal gain.

*Note: This block performs the function of amplifying the signal only in digital form, and does not control the specific functions of the sound card, if available.*

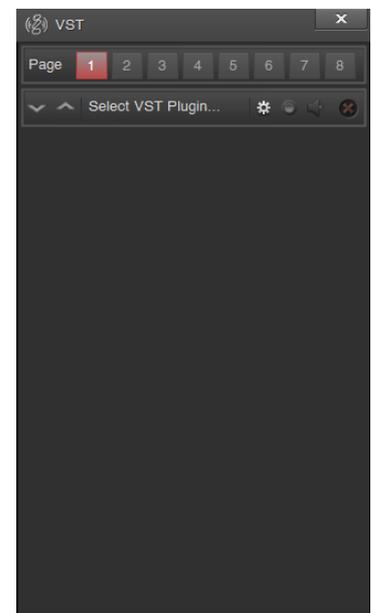
### 5 VST Plugins Group

Enabled button includes a set of the VST plug-ins in the processing path of the transmission signal.

A click on the View button opens the VST window.

In this window, you can build a processing path of the microphone signal of 96 series-connected VST plug-ins (8 pages with 12 plug-ins).

For each plug-in there are buttons to move it up or down the list, open the settings for the selected plug-in, its inclusion in the transmission path, and remove from the list.



## 6 Compressor Group

Dynamic Compressor button includes a dynamic compressor to the processing chain.

Noise Gate slider sets the level of the noise gate. If the signal level is lower than determined with this slider, the transmission coefficient of dynamic compressor becomes zero.

Knee Level slider - sets the level of the compressors knee. If the signal level is higher than determined with this slider, it begins to be suppressed and signals level remains at the level of the knee.

u-Law Compressor button includes a compressor, which applies the  $\mu$ -law to the processing chain. In the input field the level of compression is set in the range from 1 (no compression) to 255 (maximum compression).

*Note: The signal compression using the  $\mu$ -law allows to restore the signal at the receiver side, wherein a corresponding expander is to be used.*

## 7, 11 Equalizer Group

The Enabled button is the on/off switch of the equalizer.

By pressing the Reset button, the equalizer settings are reset to the initial state.

## 8 Filter

Slider to control transmission signal bandwidth in SSB and AM modes. By pressing the right mouse button on this slider a new window opens for fine tuning of the filter edges.

## 9 Level Correction Group

Audio AGC button - on/off switch for the AGC function.

When the AGC is off, the gain can be set manually (dB).

Decay slider to adjust the speed of the AGC reaction to a decrease a signal; as further to the left the slider is set, the quicker the AGC reacts.

AGT button - on/off switch for audio gain threshold (AGT) function. Threshold level is set by the Threshold Level slider.

## **10 AM/FM Settings Group**

In version ZeusRadio v.2.8.1 only the depth of amplitude modulation of the transmitted signal can be adjusted in the range from 10% to 150%.

*Note: Only used in AM mode; does not affect the signal transmission in the other modes.*

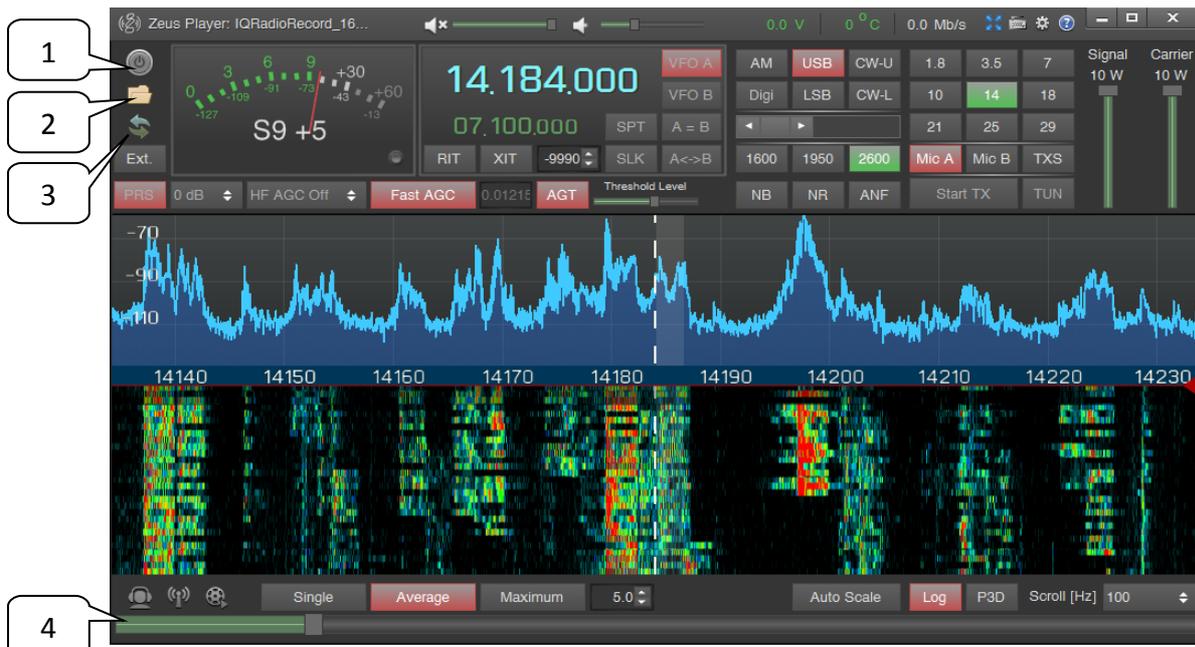
## **12 Spectrum display of the audio signal**

Visual control of the audio signal spectrum.

# Software description

## IQ Player

The separate software IQ Player replays recorded files with IQ Data. The window is similar to the main window of the ZeusRadio software and contains the same controls, with the some exceptions.



### 1 ON/OFF

Button for starting and pausing a replay.

### 2 Open

Button for selecting a file for replay. With a mouse click a file selection dialog window opens.

### 3 Cycle

On/off switch for looped replay of a file.

### 4 Playback scroll bar

The slider represents the status of replay within a file.

# Software description

## HIQSDR

ZeusRadio can be used to control the transceivers of the HIQSDR project.

*Note: Only RX mode is available with ZeusRadio v.2.8.1. TX mode and external devices control (LNA, PA, BPF, ATT) will be implemented in future versions.*

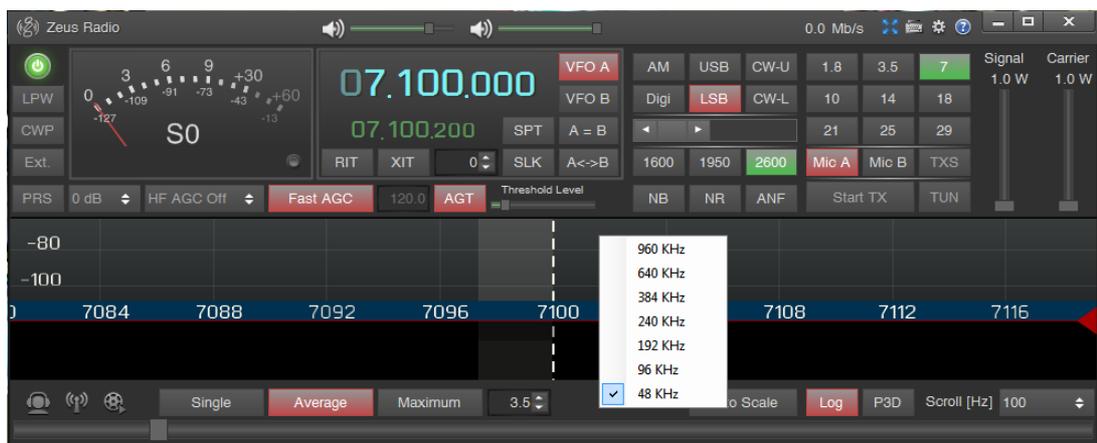
To control the HIQSDR the ZeusRadio.exe must be launched with the following parameters:

ZeusRadio.exe hiqsdrr ip=192.168.2.196 control=48248 rx=48247

Or just use the HIQSDR shortcut on the desktop, which appears after the ZeusRadio v.2.8.1 installation.

*Note: In the case of the remote operation or if the device works through another ports or has another IP address, it is necessary to set corresponding values of the launch parameters.*

The main window with HIQSDR is similar to the main window with ZS-1 and contains the same controls.



But there are some differences, caused by the differences in ZS-1 and HIQSDR devices.

LPW and PRS buttons are disabled due to the lack of these elements on the main board of the HIQSDR transceiver.

In current version also the following elements are disabled: CWP, Ext, attenuator control, HF AGC, Start TX and TUN. These controls will be functional after the TX mode implementation and external devices control update.

*Note: Soon the BPFs and LNA control will be realized through the External Control window.*

At the top of the window the bit rate is only displayed, while the temperature, supply voltage and current consumption are absent.

GETTING STARTED WITH HIQSDR UNDER ZEUSRADIO CONTROL THE S-METER MUST BE CALIBRATED. To do this, apply to the input of the transceiver the signal with known level and achieve appropriate S-meter readings by changing the S-meter correction parameter in the Settings window. More convenient to perform the calibration in the digital mode of the S-meter using the Filter value – the signal level in the filter expressed in dBm.

## Frequency setting

There are various options for tuning the frequency.

Turning the scroll wheel when the mouse pointer is set within the spectrum or waterfall will change the frequency of the active VFO corresponding to the scroll direction. Tuning step can be set in the Scroll field.

By the left mouse button click onto a signal within the spectrum or waterfall the according frequency is set. To tune on a station it is enough just to click on it and fine-adjust with the scroll wheel.

Each digit in the frequency display can be changed individually by positioning the mouse pointer and turning the scroll wheel, or by a click with the left mouse button (a click above the digit increases the value, a click below the digit reduces it).

If you hold down the Ctrl key and rotate the mouse wheel in the spectrum or a waterfall, or on one of the digits of the VFO frequency, then the change will not only have effect on the VFO frequency, but on the center of the displayed bandwidth. If in the Settings window the parameter "Inverse when CTRL key down" is set to Yes, then the rotation of the mouse wheel in the spectrum, waterfall, or on one of the digits of the VFO frequency is reversed, the VFO frequency and the center frequency of the span will change synchronously then without pressing CTRL

A click on the active VFO button centers the actual frequency to the middle of the spectrum.

Furthermore, the VFO frequency can be entered via the keyboard. Therefore the VFO frequency has to be marked with the right mouse button, the value has to be entered and the "Enter" key has to be pressed. The frequency of the inactive VFO in the field below may be altered in the same way.



For changing the frequency band, nine buttons with the amateur radio bands are available. In each band the last chosen settings (the frequency within the band, the modulation mode and the spectrum bandwidth) are stored.

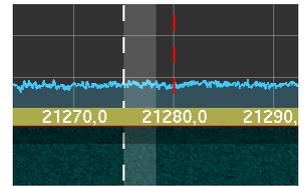
With the horizontal slider at the bottom of the window the transceiver can be set quickly to any frequency within the complete range.

## RIT, XIT and SPLIT modes

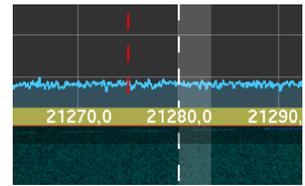
Usually the frequencies for receiving and transmitting will be identical. In this case, simply choose a VFO. If different frequencies should be used, three options are available: RIT, XIT and SPLIT.

When one of these modes is activated, in the spectrum display a red line with dots will appear, representing the transmission frequency.

RIT (Receive Incremental Tuning) changes the receiver frequency in the range of  $\pm 9990$  Hz, while the transmission frequency remains unchanged. In the screenshot an example is shown, where the VFO frequency is 21.280.000 Hz and the RIT frequency is -5000 Hz.



XIT (Transmit Incremental Tuning) changes the transmission frequency in the range of  $\pm 9990$  Hz, while the receiver frequency remains unchanged. In the screenshot an example is shown, where the VFO frequency is 21.280.000 Hz and the RIT frequency is -5000 Hz.



With activated SPLIT function (SPT) reception is on the frequency of the active VFO, the transmission will be on the frequency of the second VFO. If VFO B is active, reception is on the frequency of VFO B, while VFO A represents the transmission frequency.

VFO A and VFO B can be set to different bands and different modulation modes. For example, reception can be done on 21.280.000 Hz in SSB mode, while transmission is done on 7.050.000 Hz in CW mode. The according band button for transmission is highlighted in red.

With activated SLK (Split Lock), a frequency change of the active VFO (receiver frequency) will be taken over to the same extent to the frequency of the inactive VFO (transmitter frequency).

By clicking the A=B button the frequencies of VFO A and VFO B are synchronized. The chosen modulation modes of the VFOs are not influenced hereby.

By clicking the A <-> B button the assignments of the frequencies of VFO A and VFO B are reversed regarding reception and transmission. The chosen modulation modes of the VFOs are not influenced hereby.

# Basic operation

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## RIT, XIT and SPLIT modes

Changing the frequency of RIT / XIT offset possible in several ways.

You can enter the offset from the keyboard or adjust it by simply rotating the mouse wheel in the RIT/XIT offset field.

Also, holding down the Shift button on the keyboard, you can drag with the mouse the white dashed line (receiving frequency indicator) in RIT mode or the red dashed line (transmission frequency indicator) in XIT mode. Or simply holding Shift, click the mouse in the desired region of the spectrum - the frequency of reception or transmission will shift to this point of the spectrum.

The same way hold the Shift on the keyboard and use your mouse to move the transmission frequency in SPLIT mode.

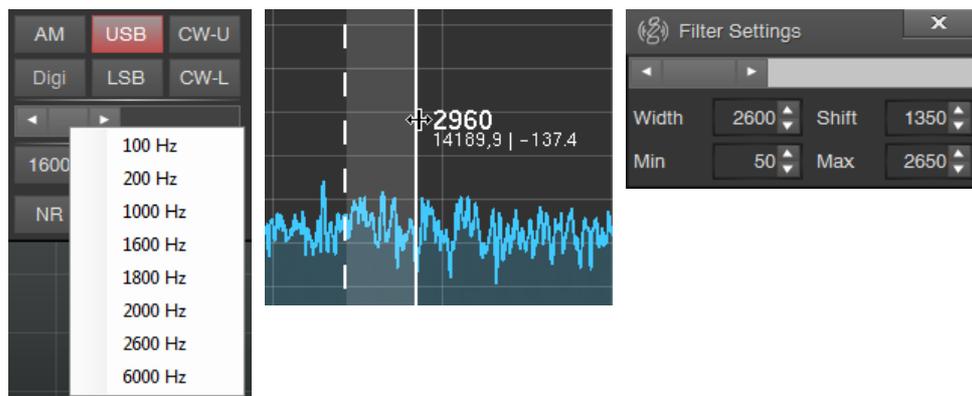
## The receiving filter bandwidth

The receiving filter bandwidth can easily be adjusted with the mouse.

The lower and upper filter edges can be adjusted separately, by moving the corresponding edges of the filter slider with the mouse. The whole filter can also be moved, by dragging the middle part of the filter slider. With a right button mouse click onto the middle part of the filter slider a list with predefined filter bandwidths will appear.

The filter edges and its bandwidth may be altered also by positioning the mouse cursor on the according filter slider part and turning the scroll wheel. Hold down the Ctrl key and rotate the mouse wheel with the cursor placed in the middle area of the slider to adjust the width of the filter without changing the center frequency.

The filter edges can be changed also by moving the filter edges directly in the display of the spectrum or the waterfall.



You can also use the three buttons with predefined values of the filter parameters. By pressing the right mouse button on any of them opens a window for adjusting the parameters of the button.

The minimum filter bandwidth is 50 Hz, the maximum bandwidth is limited to 10 kHz, but it cannot exceed half of the selected sample rate of the audio signal.

The filter bandwidth is stored separately for each mode (AM, SSB, CW).

In the Settings window the steepness of the filter edges can be predefined for the modulation modes CW and SSB (AM mode uses the same values as set for SSB). The higher the steepness is set, the better neighboring stations are suppressed, but the latency of the audio signal will become higher.

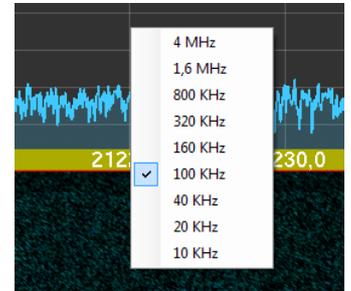
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## Display bandwidth

Like the receiving filter bandwidth, also the display bandwidth can be changed in different ways.

After pressing the right mouse button within the spectrum or the waterfall a list with bandwidths presets from 10 kHz to 4 MHz appears.

*Note: There is an important difference between the bandwidth of up to max. 100 kHz and above 160 kHz. When a display bandwidth of up to max. 100 kHz is selected, a continuous stream of IQ data is sent to the computer for further processing (filtering, demodulation, fourier-transform) and optional saving to hard disk. Data transfers with bandwidths of 160 kHz and more are regulated differently: a continuous narrow band stream of IQ data is sent to the computer for reception of the audio signal (filtering, demodulation), as well as data blocks for the fourier-transform (spectrum and waterfall display). For this reason, the maximum amount of spectrum size for bandwidths of 160 kHz and above is limited to 1024 points, and using the Time Machine feature is only possible without changing the receiver frequency (changes of the receiver frequency will force the signal buffer to be emptied, and the buffer filling starts again).*



The display bandwidth can be changed continuously by moving the frequency axis of the spectrum with the pressed right mouse button to the right or left.

Moving the frequency axis of the spectrum with the pressed left mouse button to the right or left will change the displayed range without affecting the bandwidth (the VFO frequency will not change as long as the frequency stays within the displayed range).

By reducing the displayed bandwidth the resolution of the spectrum and waterfall will be increased. At the same time more time is needed for signal accumulation and spectrum calculation. Hence the spectrum graph looks less fluent and changes jerkily. To compensate this effect, the function "FFT Rate Balancing" is implemented, which can be switched on in the Settings window. When switched on, the spectrum are calculated with a substantial overlap, what demands more CPU power.

## Time Machine

In ZeusRadio software a function is implemented, which cannot be found in classical transceivers. The Time Machine function allows to tune to a signal not only by frequency, but also by time. Any signal can be heard that is visible on the waterfall, so also short termed transmissions will not be missed.

In the initial position, the Time Machine slider is set to the upper position – the on-air signal will be reproduced. By moving the marker in the waterfall downwards with the mouse the point of time for reproduction of the signal changes (time shift). The maximum time shift is 2.5 minutes.



With a click of the right mouse button onto the "Time Machine" control it switches back to the initial position. This happens also by switching into the transmit mode.

Shortcuts are also provided for the Time Machine function.

Holding down the Ctrl key, the position of the marker can be set on the waterfall with the mouse click in the selected position of the waterfall.

Holding down the Ctrl and Alt keys, the position of the marker and the frequency of the active VFO can be set by the mouse click in the selected position of the waterfall.

If the marker was moved to a selected position, but the message is not copied, this signals can be replayed from the selected position again by pressing the Ctrl key and the right mouse button.

It is possible to replay any displayed on the waterfall signal only if a display bandwidth up to 100 kHz is selected. For bandwidths above 100 kHz only the time can be shifted (the signal is „rewound“ backwards), while the receiver frequency is not changed.

## CW mode

After activating the telegraph mode by clicking the buttons CW-L, CW-U, or in the CAT Interface, the CWP button became active and the CW key manipulation will lead to transmission of the according signals.

In the ZS-1 transceiver several variants of an automatic Morse keyer are implemented (Key Type dropdown list). Just connect the key to the KEY socket.

Independent from the chosen key type the transceiver will be set to the transmit mode by operating the key, and the signal transmission starts. The time interval for switching back into the receive mode can be defined in the Break In field. This way the Semi Break-in function (or VOX) is available.

The delay value in the Break In field has to be selected depending on the CW manipulation speed, so that the number of relay switches is minimized, preventing it from unnecessary stress.

With a PTT footswitch a manual switching between transmission and reception modes can be realized. Just as in the SSB mode, the PTT contact switches the transceiver in the transmit mode or the mode of carrier transmission. In this case, the value in the Break In field can be set down to 10 ms, while the relay switching between the symbols is prevented by the pressed PTT footswitch.

It is possible to set the CW mode parameters, to regulate the manipulator and to hear a transmission sample, without transmitting a signal. For this purpose, click the CWT button in the CW Panel window.

The quality of the CW side tone as well as its delay (not of the transmitted signal) depends upon the sound card characteristics and the used audio driver. The slightest delay and the highest quality signal can be achieved using ASIO drivers.

## SSB and AM signal transmission

For processing the signal in the SSB and AM modes algorithms are used, which allow to prepare the signal from the microphone to broadcast. The parameter settings for these algorithms are located in the TX Signal Settings window.

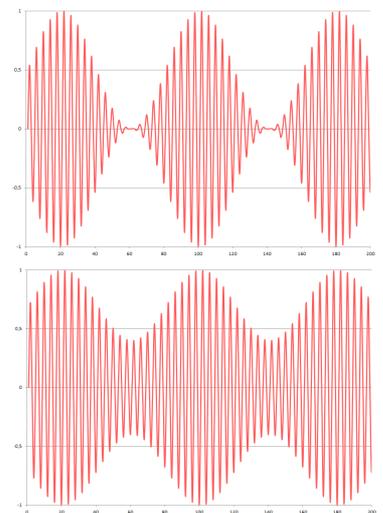
There are two sets of settings provided (Mic DSP A, Mic DSP B) between which you can switch quickly. For example, a set of settings for the cleanest signal and the second set for operation with a more aggressive modulation.

The transmission starts by clicking the Start TX button, or by pressing the corresponding keyboard key that is set in the Settings window, or by pressing a footswitch that was connected to the PTT connector, or by a PTT signal in the CAT interface.

During the transmission, the S-meter shows the peak output power of the signal (PEP). It is set with the output signal power slider.

In contrast to the SSB mode, where the carrier frequency and one sideband are suppressed, a feature exists in the AM mode, regarding the regulation of the signal output power.

When the output power of the transceiver is set to 10 W, it means also that the maximum value of the envelope power is 10 W. If a AM Depth value of 100% is selected at the same time, the carrier power is 2,5 W. And with 60% AM Depth the carrier power is 5 W.

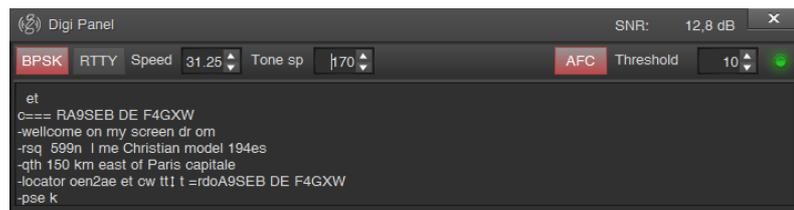


## Digital modes and VAC

ZeusRadio v2.8.1 has built in BPSK and RTTY demodulators. Future ZeusRadio versions will provide TX in this mode and all needed service functions.

Now quite a lot of manual operations is required for reception of signals.

By pressing the right mouse button on the DIGI button the additional Digi Panel window opens.

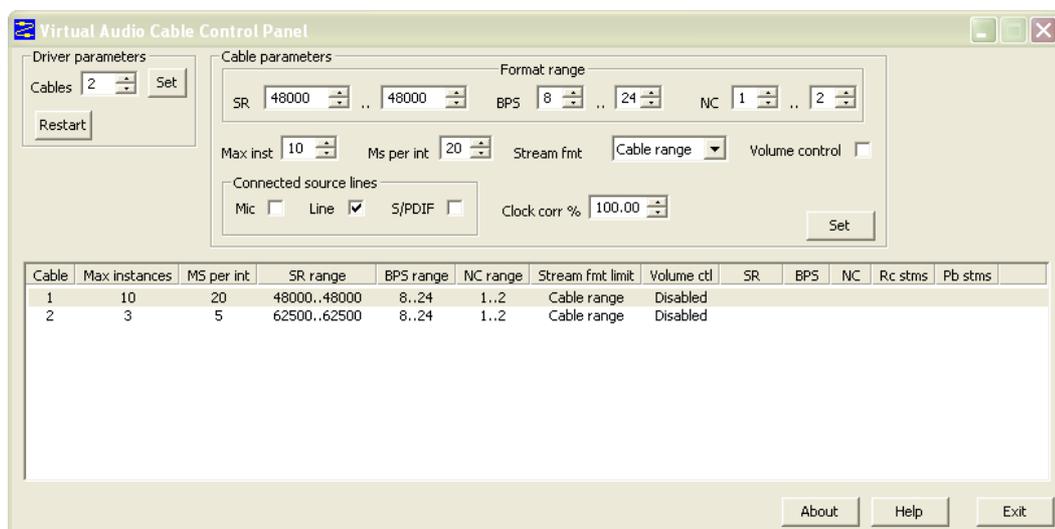


This window contains the mode selection buttons (BPSK, RTTY), the Speed input field, the Tone spacing input field, the automatic frequency control (AFC) button, the demodulator Threshold level input field and the green Active indicator. On the title bar the signal to noise ratio (SNR) is displayed.

A fully-functional and comfortable work in digital modes (not only BPSK and RTTY, but also any other) is possible using the Virtual Audio Cable (VAC) software. In this case the audio signal can be fed to an external decoding software, which also can supply the audio output for transmission.

For this purpose the VAC software has to be installed on the PC.

The most popular is VAC by Eugene Muzychenko:



Afterwards, one of the VAC channels has to be selected as audio input/output in the "Audio" tab, instead of the microphone or speaker used by default. Then the same VAC channel has to be selected as audio input/output in the respective decoding software.

The parameters of the transceiver (frequency, modes, etc.) has to be set by the CAT interface. At least one pair of comports have to be created (s. "Setting of the CAT interface").

To facilitate the work with digital modes, use the button Digi.

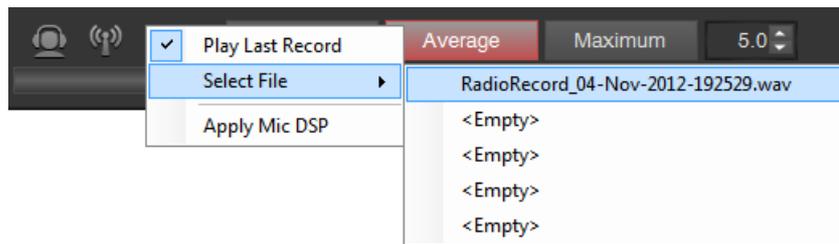
When you enable this button, the program ZeusRadio automatically switches to the last used audio input and output, audio sampling frequency, as well as AGC mode, RX filter width and the output level (volume). It also turns off the functions AGT, NB, NR, ANF and Pseudo Stereo.

Switching between Mic A and Mic B becomes unavailable, a control in the TX Signal Settings for processing the signal transmission does not interfere. After enabling Digi, the transmission path automatically disables EQ and compressors, turns on AGC with fast rise and slow decay, and the value of AGT is set to 10.

## Audio file transmission

When you click on the audio file playback button with the left mouse button, the transceiver is set to the transmit mode and playback of the audio file starts.

The selection of the file to be transferred is performed by clicking the button for audio file playback with the right mouse button



By default, the last modified file in the folder that is specified for audio files recording is selected for playback. With this selection, it is very easy to reproduce the previously received and recorded signal.

It is possible also to add any audio file to the five-line quick dial list. The entry of the file in any row is done by clicking on the respective row with the right mouse button. When clicking a file from the quick-dial list with the left mouse button, the file is selected for playback. If selecting the file from the list is done while holding down the Ctrl key, the file is played back immediately.

By default, replay of a file is done without any processing of the signal (filtering, compression, equalizer, etc.). However, processing of the signal may be switched on by activating Apply Mic DSP string. In this case, the current settings of the microphone signal processing algorithms (Mic DSP A, Mic DSP B) are applied to the signal during the file playback.

For replay, any 16-bit WAV file with no more than 5 minutes total length may be selected.

## Setting the transmit signal

In the TX Signal Settings window you will find the possibility to adjust the algorithms of digital signal processing of the microphone signal or any other audio source. In ZeusRadio four main functions are implemented: the signal level regulation, the signal peak factor reduction (compression), the equalizer, the transmission signal bandwidth adjustment. There is also a possibility of using VST plug-ins.

*Note: The settings of the broadcast signal does not affect the signal in the CW mode.*

*VST plug-ins are only supported with built-in GUI.*

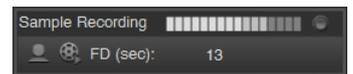
The processing units are arranged from top to bottom in the order corresponding to the signal passing through them.

Clicking on the speaker icon, which is included in each unit, the output signal of this unit can be monitored and thereby offers a consistent configuration of the transmission path.

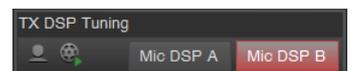


In addition, each unit has its own signal strength and overload (clipping) indicator. Resolution of the level indicator is 3 dB.

First of all a recording of an audio sample has to be started by clicking the microphone symbol in the Sample Recording field. During recording the text should be consistently spoken with the same volume as used in transmissions. A recording ends by a further click onto the microphone symbol, and it will end automatically done when the length of the recording reaches 20 seconds. The recording may be controlled by clicking the replay button.



It must also be stated which of the two sets of parameters should be regulated (Mic DSP A, Mic DSP B), and the button for looped playback of the recording has to be activated.



Further order of procedures may be different. Here is the one of them (without VST plugins):

- adjusting the Microphone Gain is necessary to achieve the level of the signal slightly in the red area which does not cause an overload;



# Software settings

## Setting the transmit signal

*Note: Microphone Gain carries only the digital signal amplification and does not control the hardware capabilities of the computer's sound card. If one has to set the values close to 20 dB to achieve the desired signal level readings, it is necessary to increase the gain of the microphone in the sound card settings, or use the microphone amplifier.*

- turn the dynamic compressor on, temporarily set the Noise Gate slider to the left (minimum gate threshold), and adjust the level of the compressors knee (Knee Level slider) to set the desired compression level, while controlling the signal by ear;

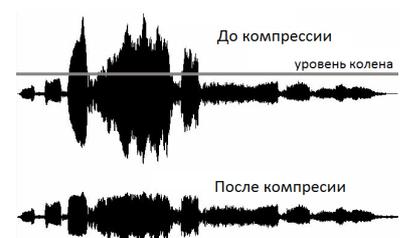
*Note: Dynamic compressor limits the volume of the signal at a predetermined level (knee). No significant difference in the sound may be noticeable, but the weak and loud sounds will differ less, the peak factor of the signal will be reduced and thus the average power of the signals is increased.*

- moving the Noise Gate slider from left to right, you can suppress the background noise during pauses in the signal; a further increase of the gate level may cause too much noise reduction, so weak sounds, especially the beginnings of the syllables, may be "swallowed";

- additionally, the  $\mu$ -law compressor can be switched on (or only this one); adjust the compression ratio in the range from 1 to 255 (1 - no compression, 255 - maximum compression);

*Note: When on the receiver side an expander is used that works with an  $\mu$ -law, it is possible to share these coefficients to restore the signal.*

- enabling the monitor signal from the output of the equalizer and, focusing on listening and the spectral shape, adjust the gain of the bands using the appropriate sliders;



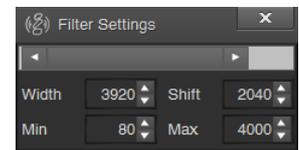
*Note: Most often one regulates treble, increase their levels, which leads to overloading of the output signal of the equalizer. To prevent overloading, reduce the Microphone Gain value or move all sliders of the equalizer to a lower level.*

# Software settings

## Setting the transmit signal

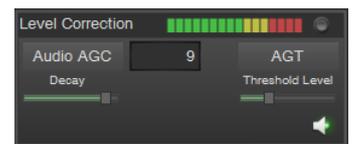
- set the required cut-off frequencies of the transmission filter, focusing on listening, and so, that the total width of the filter does not exceed the allowed value;

*Note: By pressing the right mouse button on the slider, the Filter Settings window appears, in which you can specify the filter parameters in digital form.*



- with AGC turned off, the gain must be picked up manually, so the output level is the highest but overload does not appear;

*Note: It is difficult to pick up the gain without overload and maximize the output power at the same time, as well as to control the level of the speech during the transmission. With the compressor turned on, this problem is greatly simplified. And this problem disappears when the AGC is turned on.*



- when working with the AGC, the Decay slider should be set closer to the right or to the far right, thus setting the slow response of the AGC to the signal level reduction and provide a more natural sound; you should also turn on the AGC value limiter (AGT button) and, by moving the Threshold Level slider, set its value to 6-10 dB higher than the minimum value of the coefficient of the AGC, which is displayed when you play a file or setting in real time.



*Note: The AGT actually sets the range of the AGC and provides no increase in noise level during the pauses.*

- the configuration of one of the two sets of processing parameters of the transmission signal (Mic DSP A, Mic DSP B) is done.

*Note: It may be that the AGC seems to be redundant, but it is recommended to use the AGC system. When working on the microphone signal, if the AGC restriction is set (AGT is on, the threshold level slider is set properly), it will let the AGC work correctly with higher signal levels and short-termed signal peaks, while with a switched off AGC blasting may occur.*

*In cooperation with external programs via the virtual audio cable the activated AGC allows that you do not have to worry about volume adjustments, so that the software receives an adequate signal.*

# Software settings

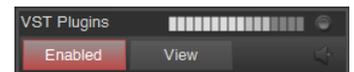
## Setting the transmit signal

A lot of alternatives for the signal processing algorithms (compressor, EQ, AGC) implemented in ZeusRadio can be found in a wide variety of VST plug-ins. Also, there are other features: De-Esser, Reverb, phase shifters, multiband compressors and so on. VST plug-ins can be a good alternative to racks of expensive equipment to build ESSB signal.

VST plug-ins are not a part of ZeusRadio and can be downloaded from the internet from various websites devoted to the subject, or from websites of software developers for sound processing.

In ZeusRadio v.2.8.1 plug-ins are only supported with built-in graphical user interface. Plugins without graphical interface can not be configured in ZeusRadio v.2.8.1.

The window for construction of the processing path based on VST plug-ins is opened by clicking the View button in the VST Plugins group in the TX Signal Settings Window.



After the ZeusRadio installation, the VST list does not contain any plugin. To add a plug-in into the processing circuitry it is necessary to press Select VST Plugin and select the plug-in file in, opened in Windows Explorer. After that, the plug-in will appear as a line in the list. Up to 96 plug-ins can be added, they will be available on 8 pages. The signal will pass through them downwards.

There is a set of control buttons for each plug-in:

- 1 – up/down arrows to move the plug-in in the processing chain (for a reordering of the plug-ins);
- 2 – plug-in settings window opening;
- 3 – on/off switch for the plug-in; if the button is not pressed (does not glow green), the signal passes this plug-in;
- 4 – sets the monitor signal output from this plug-in;
- 5 – button to delete the plug-in from the processing chain.



# Software settings

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## Setting the transmit signal

Examples of the usage and setting of the VST plug-ins can be found at [www.essb-labs.com](http://www.essb-labs.com) and others resources devoted to the subject of high-quality SSB signal and audio processing in general.

## Setting of audio devices

In the Audio tab of the Settings window the parameters for the audio input and output are set, as well as the audio devices for signal processing (decoding) programs. Incorrect settings can cause delays and faulty signals.

The most important parameter is the selection of the audio driver. It may be DirectSound or one of the ASIO drivers installed in the system.

The main difference between these two drivers is that when using the ASIO driver minimal audio signal delays are guaranteed, but only one program can use the selected device for sound input/output. That is, when using the ASIO driver in the ZeusRadio all other software cannot use this audio device. And vice versa, if another software already uses the audio device, ZeusRadio will not be able to connect to it.

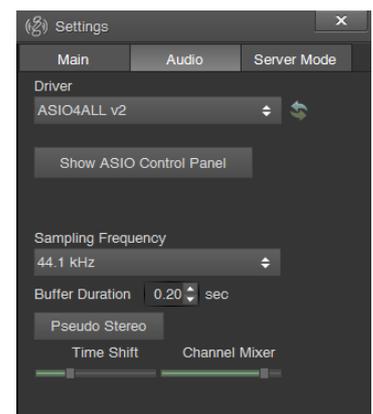
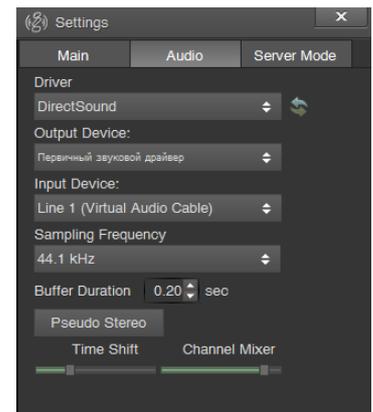
When using the DirectSound audio driver a delay may be slightly higher, but it still remains the opportunity to play audio from other software.

By default, the ZeusRadio software uses DirectSound and primary devices for audio input and output, i.e. the devices that are selected by default in the system as a sound recording/reproducers. Usually, it is the speaker and the microphone. But it is also possible to make the selection manually, for example, to select speakers for sound output (regardless of the system settings) and the virtual audio cable for sound input.

When using DirectSound, an important parameter is the Audio Buffer Duration. By default, a value of 0.2 seconds is set, which is suitable for the majority of systems. But in systems with high efficiency it is possible to reduce the sound delay by setting the Buffer Duration of, for example, 0.08 second. For systems with low efficiency a reduction of the buffer duration may lead to interruptions and pops of the audio signal.

To achieve minimum delay of the audio signal and for operation in telegraphy mode at high speeds, it is necessary to install and use the ASIO driver.

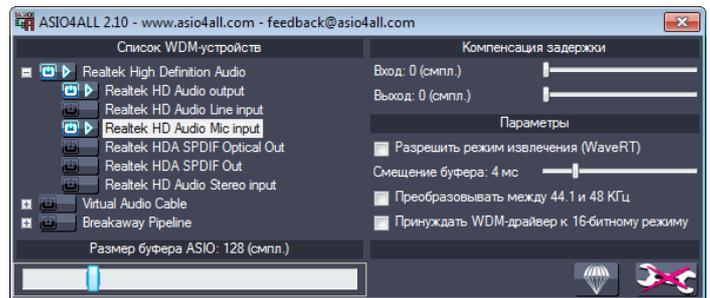
When selecting the ASIO driver, audio input and output selection is done through the driver settings window, which may look different and will be shown by clicking the Show ASIO Control Panel button.



# Software settings

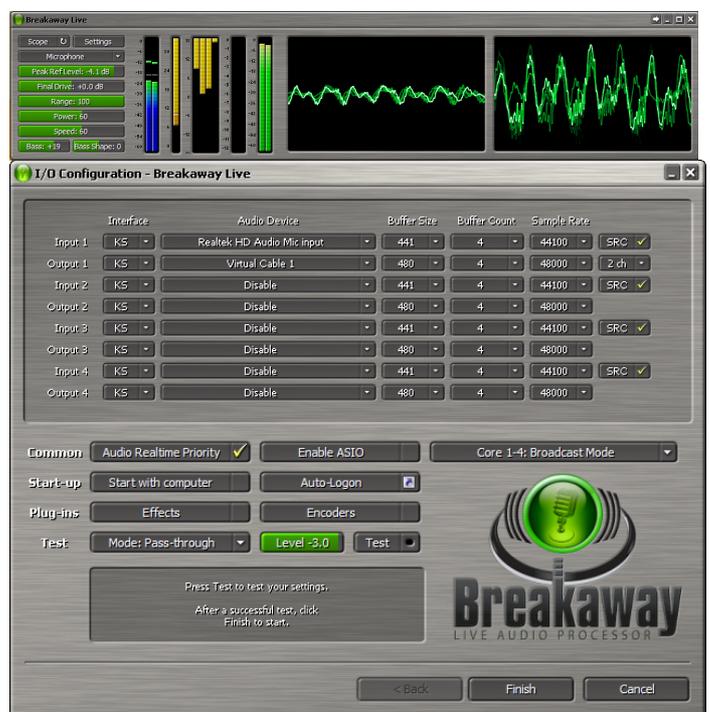
## Setting of audio devices

For example, such is the control panel of the ASIO4ALL driver (see picture). Here, the selection of audio input/output devices and the Buffer Duration setting is made (not in seconds but in samples of the signal).



As an example for the selection of audio input/output devices, we will consider the use of the Breakaway Live Audio Processor software. This program processes sound in real time and significantly improves the sound of the station.

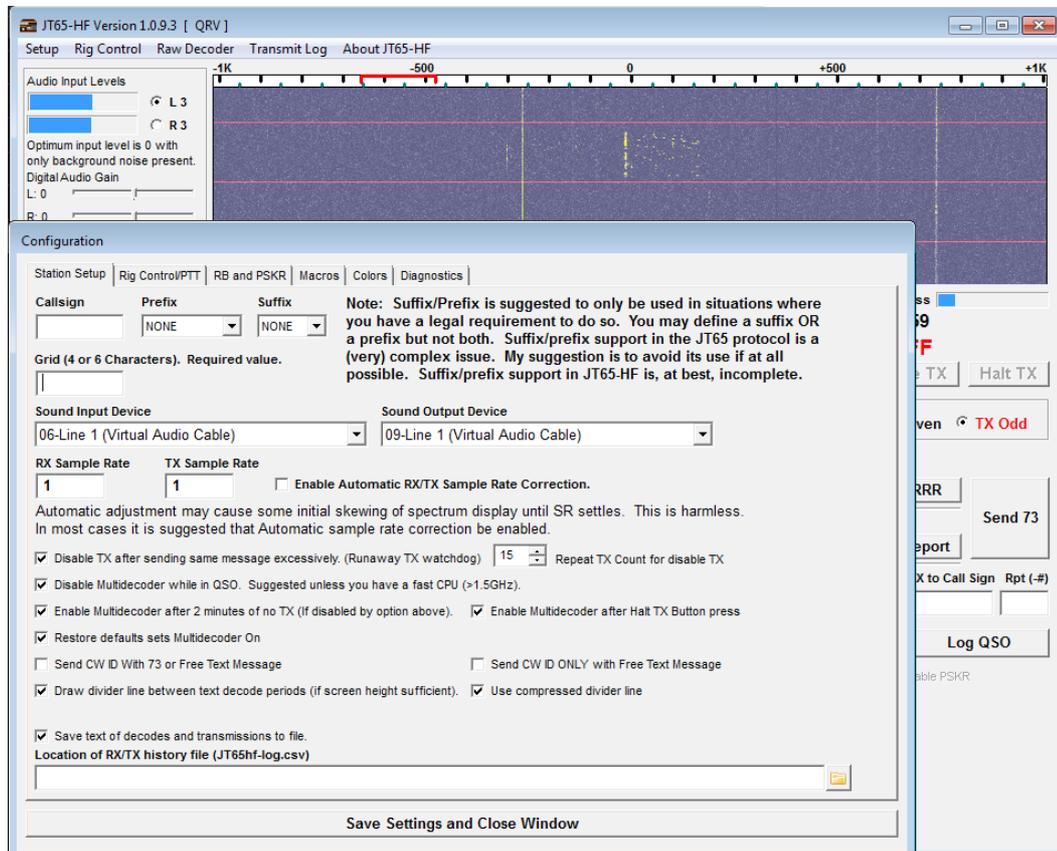
For use of this sound processor (like any other), you must select the virtual audio cable as input in ZeusRadio, and also the same virtual audio cable as output device in the sound processor settings, while the microphone is the input device.



# Software settings

## Setting of audio devices

A further example of audio input/output devices selection is working with digital modes, like JT65.



In this case, in ZeusRadio the virtual audio cable is used for audio output (for routing the signal to the JT65-HF software), as well as for input (for transmitting the JT65-HF signal).

The same settings have to be chosen in the JT65-HF software.

In this case also the CAT-Interface has to be configured to manage PTT line, the VFO frequency, etc.

## Transceiver frequency correction

The reference oscillator of the transceiver has a certain tolerance, which is also dependent on the transceiver's temperature. So transmit and receive frequencies can differ by 10 Hz to 20 Hz at a frequency of 10 MHz.

To correct this deviation, carry out a calibration with the frequency correction function. First open the Filtered Data window by clicking with the right mouse button on the S-meter and click the button FC.

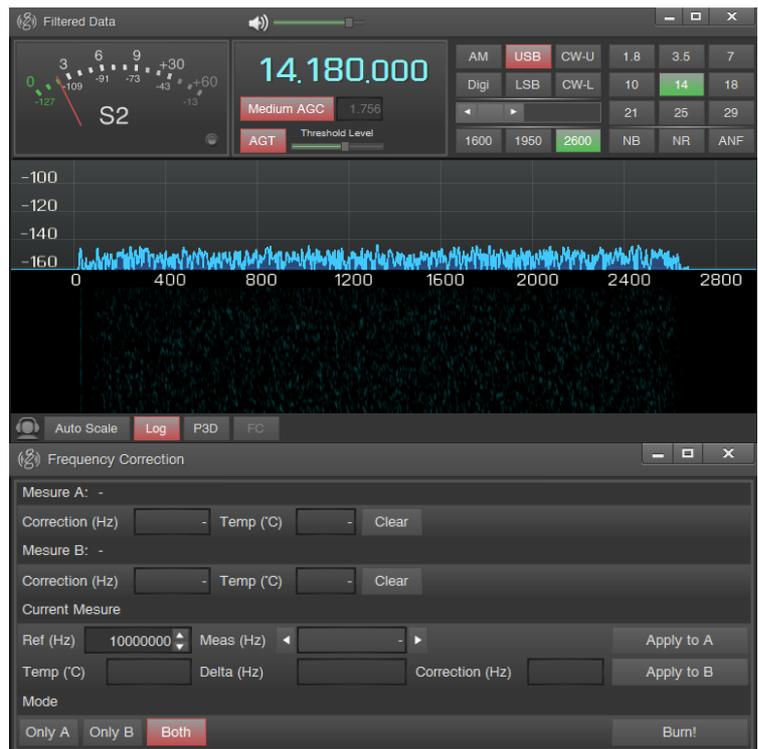
*Note: To enable the FC button, the Frequency Correction window must be opened, and a correction is possible only in the CW-L and U modes, up to a filter width of max. 100 Hz.*

The method involves measuring and correcting the reference oscillator's frequency with respect to the transceiver

temperature. The results are stored in the two correction memory cells Measure A and Measure B, and may be used either individually (buttons Only A, Only B) or together (Both button) for compensating the temperature instability of the reference oscillator.

*Note: The temperature compensation algorithm (button Both) is available only if the temperature difference in the measurements A and B is more than 3 degrees.*

You need an external reference oscillator with a nominal frequency (5 MHz, 10 MHz or any other) or use one of the reference frequencies on shortwave (4996 kHz, 9996 kHz, etc.)

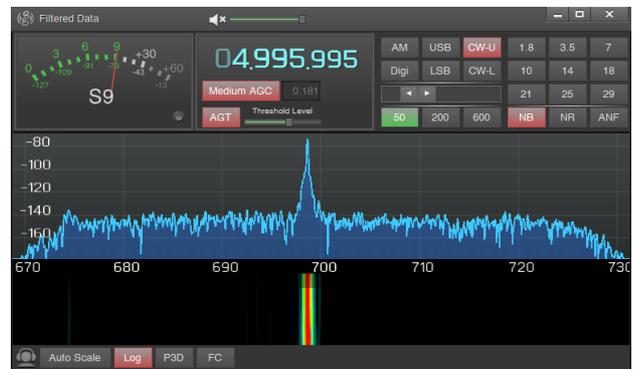
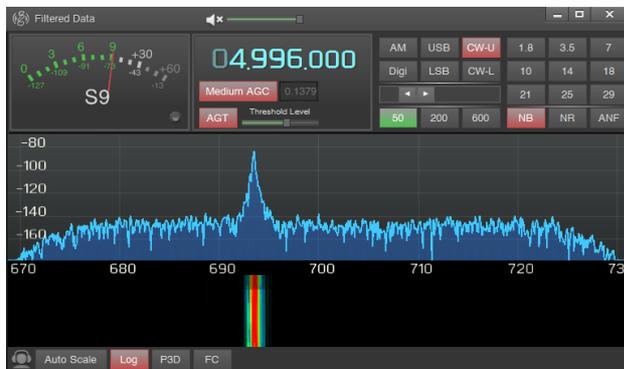


# Software settings

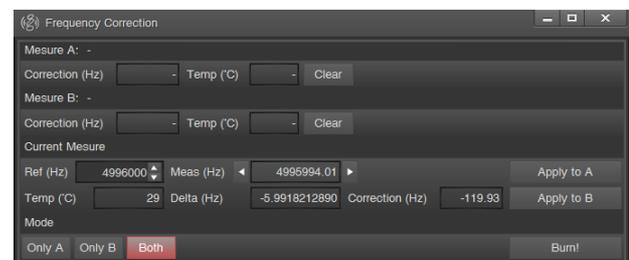
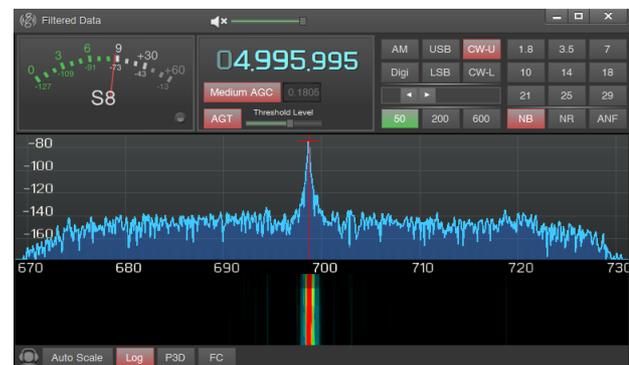
## Transceiver frequency correction

Frequency correction is recommended as follows:

- when using an external reference, switch it on, and connect it to the ZS-1 antenna input;
- start the software ZeusRadio, select the operation mode CW-U, set the filter bandwidth to 100 Hz or less, and adjust to the desired frequency - the frequency of the generator, or one of the HF reference frequencies (4996 kHz, 9996 kHz, etc.);
- open the window of the filtered data with a click of the right mouse button on the S-meter, wait for the structure of the spectrum graph and set the reference frequency closer to the center of the filter;



- click FC to open the Frequency Correction window, the accurate measurement of the frequency starts and the signal will be captured by a red marker in the Filtered Data window indicating the frequency, and in the Frequency Correction window the measured values are displayed: Ref - the value of the reference frequency (set by the user), Meas - the measured frequency, Temp - current temperature of the transceiver,



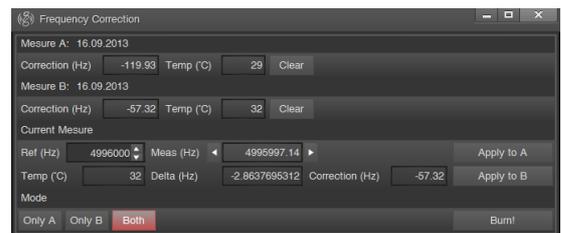
# Software settings

## Transceiver frequency correction

Delta - difference between the measured and the reference frequency, Correction - measured error of the transceiver's reference oscillator frequency;

*Note: If the spectrum contains several peaks, which is common for a HF signal, and the "wrong" peak is captured, then you can use the arrows on the edges of the Meas field to move the marker to the desired position; you can also change the transceiver's frequency so, that the "correct" peak will be closer to the filter center.*

- Click Apply to A to save the results of measurements in the cell Measure A;
- wait for an increase of the transceiver's temperature of at least 3° C, press Apply to B to save the second set of measurements in the cell Measure B;



- measured data will be stored in the software ZeusRadio, but can be saved to the internal transceiver memory by pressing the Burn button, (first the program must be stopped by pressing the ON/OFF button in the main window of the program);

- after the selection of correction mode (Only A - correction using only measurement A, Only B - correction using only measurement B, Both - temperature-dependent correction using both cells), the Frequency Correction and Filtered Data windows may be closed and you can work with the transceiver.



*Note: As long as the FC the window is open, frequency corrections are not applied, and no change in the main window is made.*

For best results when using the temperature-dependent correction, it is recommended to make the first measurement after the transceiver temperature is 5-7°C above ambient temperature, and the second measurement after a rise of the transceiver temperature of another 5-7°C.

# Software settings

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## Transceiver frequency correction

For temperature-dependent measurements, the correction factors are recalculated when the transceiver temperature changes by 1°C. That may lead to an abrupt change in frequency of the received signal in the range of 2 Hz at a receiving frequency of 10 MHz. For some applications, such jumps are not allowed, but is not noticeable by ear in SSB and CW and does not affect the results of decoding for the majority of digital amateur radio signals.

By a fan attached to the ACC connector the transceiver frequency can be stabilized and jumps in frequency can be avoided. After setting the threshold temperature and the fan speed in the External Control window, the transceiver temperature will be stabilized at a predetermined value, and changes in frequency of the reference oscillator over time will be negligible.

Due to the aging of the reference oscillator it is recommended to repeat the frequency correction once every 4 months during the first year of operation, and then approximately 2 times a year.

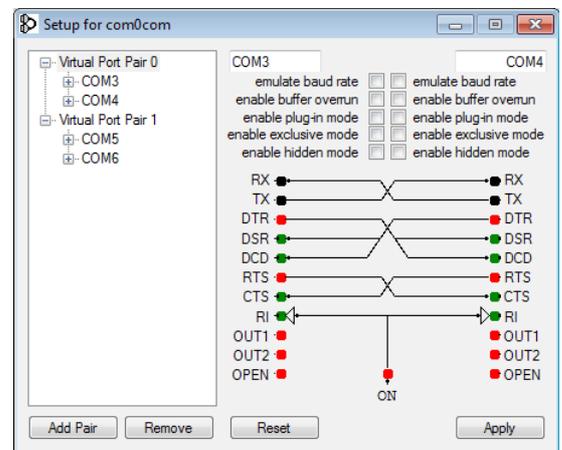
# Software settings

## Setting of the CAT interface

For controlling the transceiver with the COM port, the CAT interface is implemented in ZeusRadio, whose control is found in the Server tab of the Settings window.

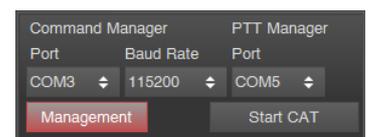
The transceiver may be controlled by a real COM port, which can present in the PC, or with a virtual one, created by a software. Virtual COM ports are used in a PC to exchange commands between different programs.

There are a lot of programs that implement virtual COM ports. One of them is com0com. Such programs create pairs of virtual COM ports that are connected with virtual cables. One of these ports should be selected in ZeusRadio, the second one – in the connected control software. In the picture, a variant of the software setting is shown, in which two pairs of virtual COM ports are installed.



*Note: For the operation of the ZeusRadio software it is necessary that the created ports have the default name of the form COM1, COM2, COMn, but not the names that are used in the system for real COM-Ports or ports created by other programs.*

The control of the transceiver can be done by two COM ports. One port can be used for a command transmission (Command Manager), the other one – for the management of PTT and CW signals (PTT Manager). One can select the same COM port for both functions, but it can cause software conflicts. It is recommended to use different COM ports, creating two virtual pairs.



For the start of the CAT interface you have to click the Start CAT button in ZeusRadio.

In ZeusRadio a command system is realized, which is compatible with Kenwood transceivers. That is why in the settings of the connected program, the following transceiver type has to be chosen: Kenwood TS-590.

# Software settings

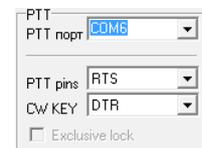
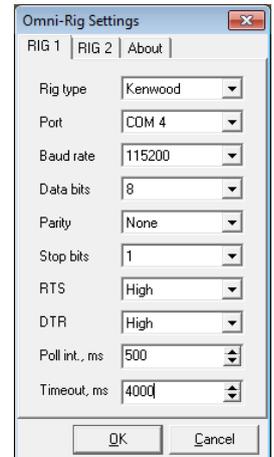
## Setting of the CAT interface

The COM port for command transfer must be chosen from the same pair that is selected in ZeusRadio (Command Manager). You also have to set the same transmission speed as well as further parameter values: data bits - 8, parity - none, stop bits - 1.

In the picture an example of the settings for the Omni-Rig software is shown.

*Note: The parameter values for Data Bits, Parity and Stop bits are hard-coded in the ZeusRadio software and cannot be changed.*

In the settings of the PTT ports please specify that the RTS signal is used to control the PTT line, and the DTR signal – to control the CW key line.

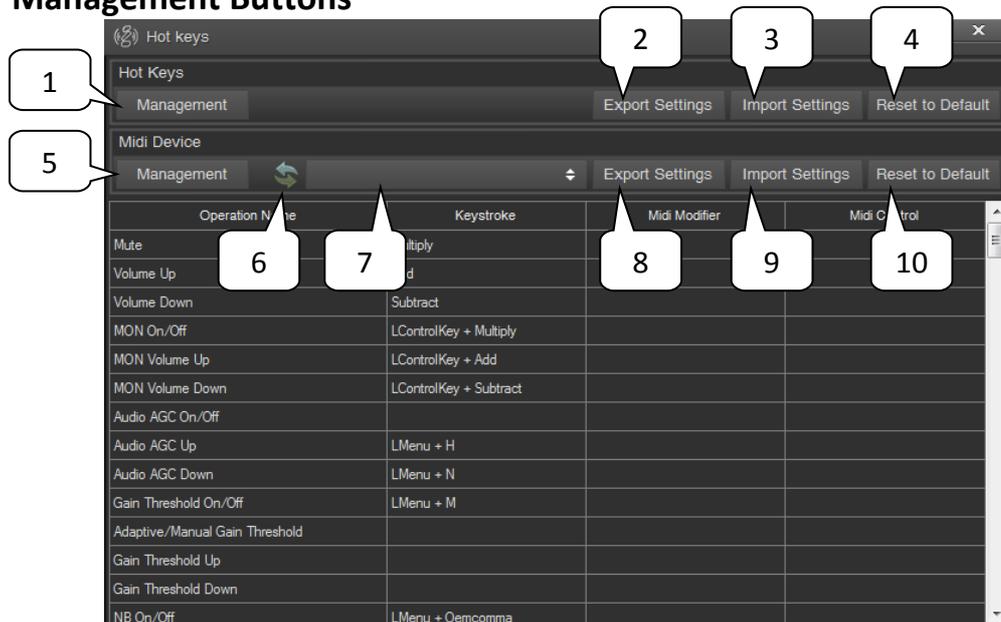


## Operation by hotkeys and MIDI devices

The ZS-1 transceiver can be controlled not only through the program interface ZeusRadio or CAT, but also by using hot keys, or any MIDI device or the Tmate2.

Settings window for hotkeys and assign functions to MIDI devices opens by pressing the keyboard icon in the upper right corner of the main program window.

### 1, 5 Management Buttons



Enable/disable buttons for program functions controlled by hotkeys and MIDI devices.

### 2, 8 Export Settings Buttons

Buttons to save the settings of hotkeys and MIDI devices to a file.

### 3, 9 Import Settings Buttons

Buttons to import settings for hotkeys and MIDI devices from a file.

### 4, 10 Export Settings Buttons

Reset Buttons to reset Hotkey and MIDI devices to its original state.

### 6 Refresh Button

Button to update the list of MIDI devices.

# Software settings

Operation by hotkeys and MIDI devices

## 7 Device List

Drop-down list to select a connected MIDI device to control the transceiver.

*Note: If a MIDI device connected to the PC, is not in this list, then click the button to refresh the list of MIDI devices.*

To assign a shortcut key or control MIDI devices for the desired software function, just click the left mouse button on the selected cell of the table (columns Keystroke and Midi Control) and then press the desired key combination or rotate / move, for example, the MIDI device controls.

Midi Modifier table column used to set modifiers. They can be used to allow to set the same, for example, slider or knob of the MIDI device, to control different functions of the software.

If there is only one tuning knob and two buttons, you can, for example, assign the frequency control for VFO A and VFO B to this single tuning knob, switching its function by pressing the corresponding modifier.

List of application functions available for control via keyboard or MIDI device contains almost everything you might need for external control:

Mute	– Muting the audio;
Volume Up/Down	– receiving volume control;
MON On/Off	– Mute CW monitor;
MON Volume Up/Down	– CW monitor volume control;
Audio AGC On/Off	– AGC switch;
Audio AGC Up/Down	– Audio AGC switch (Fast, Medium, Slow);
Gain Threshold On/Off	– AGT/MGT switch;
Adaptive/Manual Gain Threshold	– switching between AGT and MGT;
Gain Threshold Up/Down	– AGT/MGT level adjustment;
NB On/Off	– Noise blanker switch;
NB Up/Down	– Noise blanker level adjustment;
NR On/Off	– Noise reduction switch;
NR Up/Down	– Noise reduction level adjustment;
ANF On/Off	– Automatic notch filter switch;
ANF Up/Down	– Automatic notch filter level adjustment;

# Software settings

Operation by hotkeys and MIDI devices

Band Up/Down	– Band switching up / down;
1.8 MHz Band	– 1.8 MHz selector;
3.5 MHz Band	– 3.5 MHz selector;
7 MHz Band	– 7 MHz selector;
10 MHz Band	– 10 MHz selector;
14 MHz Band	– 14 MHz selector;
18 MHz Band	– 18 MHz selector;
21 MHz Band	– 21 MHz selector;
25 MHz Band	– 25 MHz selector;
29 MHz Band	– 29 MHz selector;
Toggle AM Mode	– insert mode AM;
Toggle USB Mode	– insert mode USB;
Toggle LSB Mode	– insert mode LSB;
Toggle CWU Mode	– insert mode CWU;
Toggle CWL Mode	– insert mode CWL;
DIGI	– insert mode DIGI;
Filter Preset A, B, C	– choice of reception filter width;
RX Filter Lower Edge Up/Down	– adjustment of the lower edge of the receiving filter;
RX Filter Upper Edge Up/Down	– adjustment of the upper edge of the receiving filter;
RX Filter Offset Up/Down	– shift adjustment of reception filter;
RX Filter Width Up/Down	– width adjustment of reception filter;
VFO A Frq. Up/Down	– adjust the VFO A frequency;
VFO B Frq. Up/Down	– adjust the VFO B frequency;
Bandwidth Up/Down	– viewing bandwidth adjustment;
Scroll Step Up/Down	– Frequency step adjustment;
VFO 1 MHz Up/Down	– change the tuning frequency by 1 MHz;
VFO 100 kHz Up/Down	– change the tuning frequency by 100 kHz;
VFO 10 kHz Up/Down	– change the tuning frequency by 10 kHz;
VFO 1 kHz Up/Down	– change the tuning frequency by 1 kHz;
VFO 100 Hz Up/Down	– change the tuning frequency by 100 Hz;
VFO 10 Hz Up/Down	– change the tuning frequency by 10 Hz;
Frequency Blocking On/Off	– lock receiving frequency;

# Software settings

Operation by hotkeys and MIDI devices

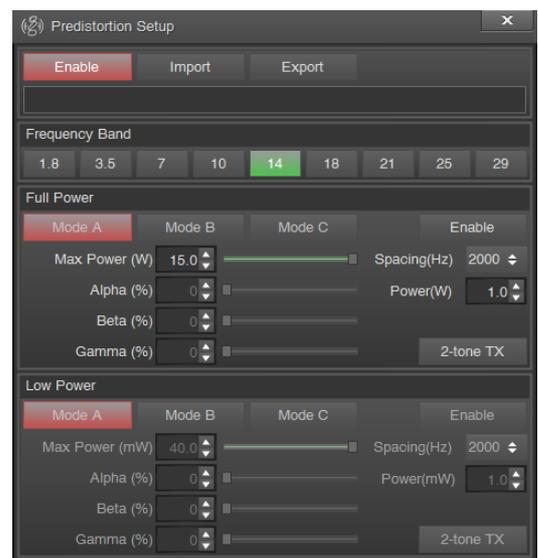
VFO A	– choose VFO A;
VFO B	– choose VFO A;
VFO A = B	– equating frequency VFO A and B;
VFO A <> B	– swaps the VFO A and B;
SPLIT, SPLIT Lock	– on / off functions for SPLIT, SPLIT Lock;
RIT On/Off, XIT On/Off	– on / off functions for RIT and XIT;
RIT/XIT Up/Down	– adjustment of RIT and XIT values;
PRS On/Off	– preselector setting;
Attenuator Up/Down	– attenuator setting;
HF AGC Up/Down	– RF AGC settings (Off, Max, Min);
Amplifier Control	– on / off control of external power amplifier;
TX	– switch to transmission mode;
TX PWR Up/Down	– output power adjustment;
Mic DSP A, Mic DSP B	– choice of microphone settings;
TUN	– switch to carrier mode;
TUN PWR Up/Down	– carrier power adjustment;
LPW On/Off	– switch for low power mode;
CWT On/Off	– switch for CW test;
CW Speed Up/Down	– CW speed control;
CW Tone Up/Down	– CW pitch adjustment;
CW Ratio Up/Down	– adjustment of dash / dot ratio;
CW Tone Hardness	– hardness of CW tone (Hard, Medium, Soft);
CW Break In Up/Down	– adjustment of Break-In Delay;
Record Audio	– start / stop recording an audio file;
Play Last Record	– transmit recorded file;
Record IQ	– start / stop recording an IQ file;
Time Machine Up/Down	– shift adjustment Time Machine;
Time Machine Reset	– reset Time Machine;
CW Template 1-12	– transmitting of CW template;
Setup CW Template 1-12	– editing of CW template.

## TX predistortion tuning

In the ZS-1 a predistortion transmission signal algorithm is implemented, which allows to reduce intermodulation components in the transmission signal. This algorithm is adjusted in the production process of each transceiver. In ZeusRadio v2.8.1 the ability to use this algorithm to compensate for the nonlinearity of one or more connected in series power amps or a transverter is implemented.

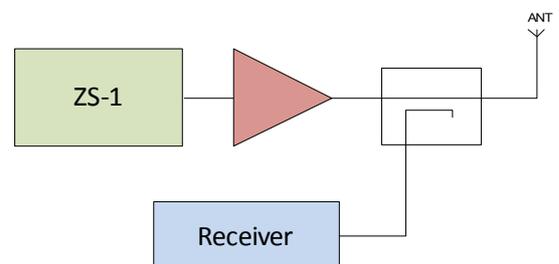
Open the setup window by clicking "Predistortion Setup" in the External Control window.

*Note: Since the main purpose of these algorithms is distortion compensation arising in external power amplifiers, while the compensation of the internal amplifier of the ZS-1 has already been made, the majority of the controls in the window become active only when the external amplifier control is enabled (Enable button in the Amplifier Settings section of the External Control window). The same applies to pre-emphasis of the transmitter signal.*



**WARNING** Misconfiguration of the predistortion transmission signal algorithms or use of them without an external amplifier can lead to a significant deterioration of the transmission signal!

To configure the predistortion algorithms a second receiver is needed. This receiver will be used to monitor the level of intermodulation products at the output of the amplifier. It can be connected to an additional antenna or a directional coupler at the output of the power amplifier. It is important to ensure that the signal level at the input of the reference receiver will not lead to its failure.



# Software settings

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## TX predistortion tuning

To avoid emission in the air and interference to other stations, it is recommended to initially use a 50 Ohm dummy load during a tuning process. But amplifier nonlinearity depends upon its load, so the results will be better after tuning with a real antenna.

The configuration process consists of setting the coefficients Alfa, Beta and Gamma for each operating frequency band by transmitting two tones (button 2-tone TX) and reduce the level of intermodulation products, which are observed by using the test receiver.

The parameters Alfa, Beta, Gamma from top to bottom are listed in order of importance. That is, after you've made all the basic settings (Power, Spacing, Mode, Max Power) and pressed two-tone TX, you should first achieve the minimum distortion by adjusting the Alfa, and then Beta. Then optimize by adjusting both values reciprocally. The Gamma factor is regulated at last; with most power amplifiers it gives a very weak effect.

Three versions of the predistortion algorithm (Mode A, B, C) are implemented to achieve the best results when working with different types of signals and amplifiers. The following features are available:

**Mode A** – algorithm has a lower accuracy of the coefficients, but is more effective in a wide band signal. That is, when you work in SSB or AM signal with broad bandwidth (more than 3 kHz), select this mode. This algorithm uses only Alfa and Beta, Max Power serves only for limiting the output power.

**Mode B** – algorithm has a greater accuracy of the coefficients, it can achieve a better result, but is less effective for broadband signals (over 3 kHz). Uses Max Power, Alfa, Beta.

Max Power is important here is not only to limit the power output, but also acts as a "scaling" factor for the predistortion algorithm, which is very important when the maximum output of the amplifier is achieved at low output levels of the ZS-1.

If your amplifier provides full power at 1W of input power, you need to set Max Power to 1W. Only then you can achieve a good result with predistortion. And also this setting will help you to avoid amplifier malfunction from overload – using this setting makes it impossible to set higher power levels with the power sliders in the main window.

# Software settings

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## TX predistortion tuning

**Mode C** – algorithm is a modification of the algorithm B. It adds the factor Gamma, which can improve the outcome for some types of amplifiers.

One of the results of the predistortion algorithm is a slight decrease in the output power of the amplifier. This is the price to pay for a cleaner signal. Though intermodulation products can be decreased by up to 30 dB.

If the Predistortion algorithm is configured correctly, it will not reduce the output power by more than 1 dB.

# ZS-1 Specifications

No	Parameter	Value
Receiver		
1	Frequency range	0,3 – 30 MHz
2	Sensitivity (MDS, preamp on)	-141 dBm
3	Sensitivity (MDS, preamp off)	-135 dBm
4	Blocking level (preamp on)	-22 dBm
5	Blocking level (preamp off)	-5 dBm
6	IIP2 (preamp off)	63 dBm
7	IIP3 (preamp off)	28 dBm
8	Attenuators	10, 20, 30 dB
9	Receiver bandwidth	up to 100 kHz
10	Viewing bandwidth	up to 4 MHz
Transmitter		
11	Frequency range	HF HAM bands: 10, 12, 15, 17, 20, 30, 40, 80, 160 m
12	Output power	15 W*
13	Harmonics level	-50 dB
14	Non harmonics level	-70 dB
15	TX bandwidth	up to 10 kHz
Common characteristics		
16	PC interface (data & control)	USB 2.0
17	Supply voltage	12 – 15 V
18	Supply current (receive)	0,5 A
19	Supply current (transmit)	4 A
20	Size	240 x 170 x 35 mm
21	Weight	1,2 kg

\* - the maximum output power of the transceiver ZS-1 in the range of 160 m, 80 m, 40 m, 20 m, 17 m, 15 m, 12 m and 10 m is 15 watts, but in the range of 30 m maximum output power is 8W.

Minimum system requirements: Windows XP/Vista/7/8 x32 / x64. USB2.0 Port.; Intel Core 2 Duo 1,5 GHz with 2GB RAM, OpenGL 1.5 or higher



SSB-Electronic wishes you much joy with your ZS-1!

You have questions, suggestions or comments?

Please let us know:

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or by telephone: 02941 - 93385-118

