

DCF / HGA / RWM PC-Time Corrector.

The DCFHGATime.exe program was designed to receive time signals from the long-wave radio stations DCF39 (138830.4 Hz), HGA22 (135430.0 Hz) - RTTY or from the short-wave radio station RWM (4996 kHz, 9996 kHz, 14996 kHz) - CW (A1B). A description of the signal formats of these stations is given in the appendix. It is possible to correct the system time of the computer according to the exact time received from the air.

The second function of the program is measuring the frequency of the received signal, which is necessary to calibrate the receiver, sound card and spectrum analyzer program (SpecLab, Argo).

It is assumed that RX is in the mode of the upper sideband (USB). For an RWM radio station broadcasting in CW mode, reception is possible both in USB and LSB - all you need to do is to select such a receiver tuning frequency that the beat frequency of the audio signal was 1 kHz. That is, for example, to receive a frequency of 9996 kHz, the receiver must be tuned to 9995 kHz USB.

The audio signal from the receiver must be fed to the microphone or line-in sound card.

Interface description.



Dial - RX tuning frequency (zero beats). There are settings within the amateur LF-band 2200 m (135000 Hz, 135500 Hz and 136000 Hz). Wherein:

- 135000 - HGA22. The frequency of the “space” will be equal to 430 Hz, “mark” 770 Hz
- 135500 - DCF39. “Space” - 3330.4 Hz, “mark” - 3670.4 Hz. As a rule, only the “space” frequency will fall into the receiver's bandwidth. This is normal for the program, but the decoding quality will decrease.
- 136000 - DCF39. “Space” - 2830.4 Hz, “mark” - 3170.4 Hz. Here, also, the “mark” frequency may be somewhat beyond the bandwidth of the receiver or on its ramp.
- 9995 kHz - reception of a short-wave radio station RWM on one of three frequencies: 4996, 9996 or 14996 kHz. The program expects a CW signal at a frequency of 1000 Hz. Since the format of RWM broadcasts does not contain complete information about the date and time, it is possible to implement a time correction within one minute, by minute mark (key down 500

ms per 00 sec of each minute). Therefore, a preliminary correction of the time (up to a minute) must be performed manually.

Sound Input – to select the sound device to which the receiver is connected.

The sound input **level in dB** is displayed to the right of the Sound Input. For normal operation of the program, this value should be from -50 to -10 dB. In the event of input overload (more than -6 dB), the numbers will turn red.

Enable Time Correction - a check mark allows you to adjust the computer's system time. The time taken from the either is compared taking into account the time zone and the time of year (summer time) with the local time of the system, their difference **dT** is calculated. If dT does not exceed 1 second in modulus, the time is considered correct, the date and time indicator turns green, otherwise it turns red. In order to exclude incorrect correction of the system clock (for example, the received time stamp was decoded incorrectly), repeated reception (by default 5 times) of time stamps with the same dT is expected. The number of dT matches is displayed in the **cnt** field. So, if the system time requires correction and cnt = 5, then the program adjusts the computer's system clock.

Attention! In operating systems Win7, Win8, Win10, etc. DCFHGATime.exe should be **run as Administrator**, otherwise the time correction will not occur.

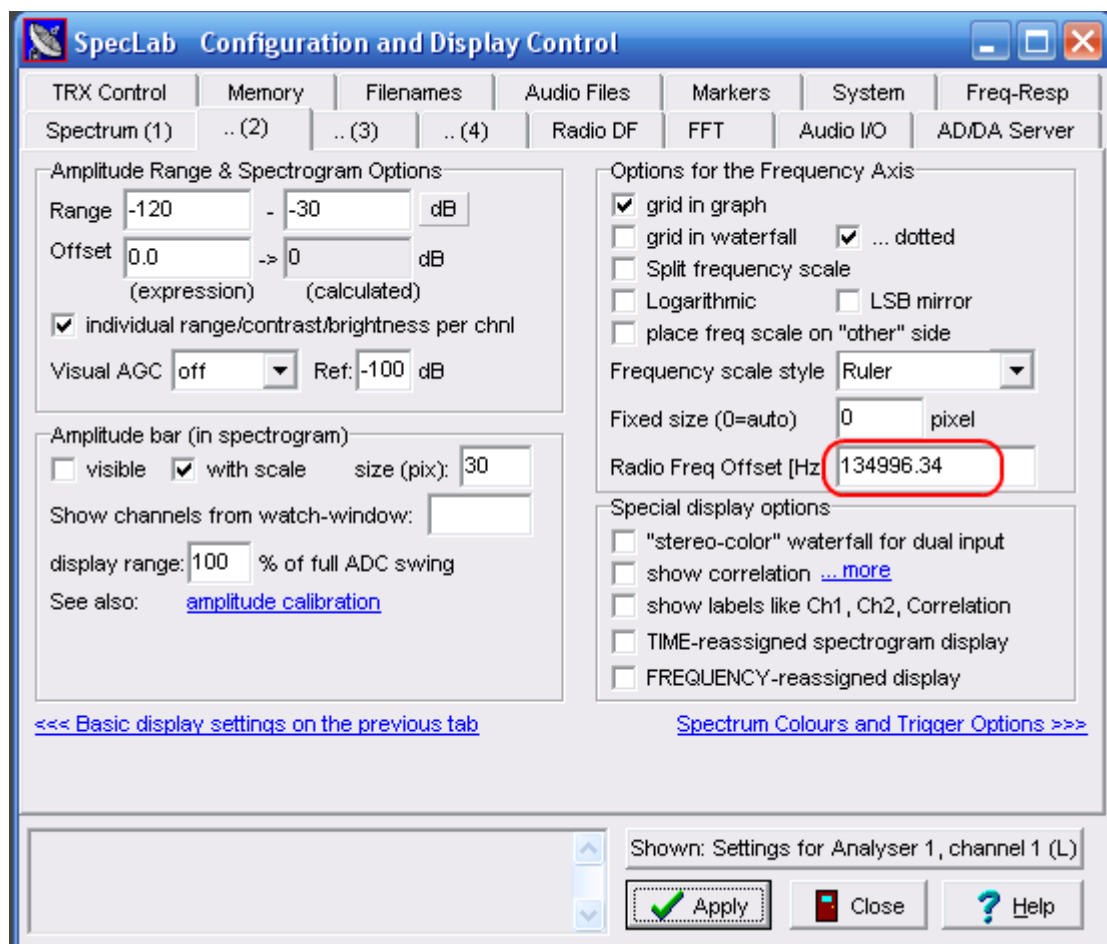
Check Parity of Data - check the checksum of the received timestamp. By default, a check mark is set. If the station signal is weak and decoding is bad, you can disable checksum checks, which will slightly increase the reception sensitivity, however, erroneous date / time decoding will occur. This is “not dangerous”, since each time, obviously, mismatched random dTs will be decoded, which means that the system clock will not be corrected.

dF is the measured frequency shift. The frequency shift is calculated according to the known frequency of the frequency of the DCF or HGA stop bursts (which are emitted most of the time), or by the RWM carrier:

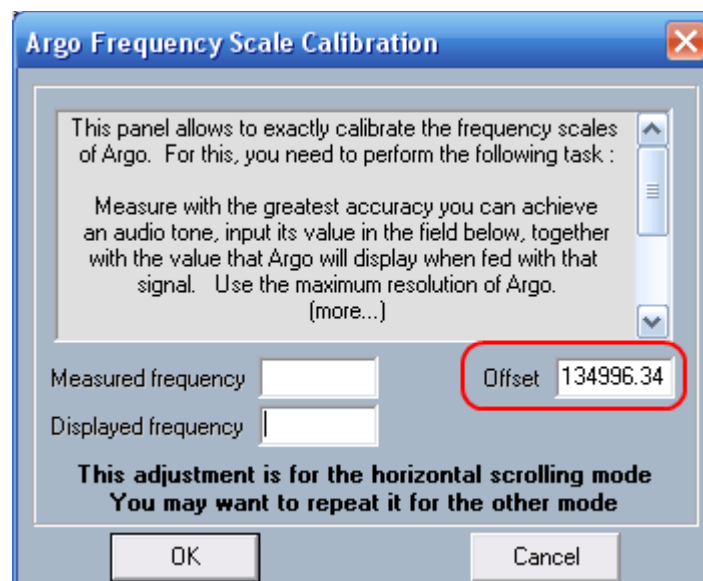
$$dF = F_{\text{measured}} - F_0,$$

where F_{measured} is the actually measured frequency, F_0 is its nominal.

Accordingly, the obtained dF value should be subtracted from the nominal of the Dial in SpectrumLab or Argo. For example, for the dF = 3.660 Hz shown in the screenshot, you should specify the value of Radio Freq Offset [Hz] = 135000 - 3.660 = 134996.34 Hz in the SPL settings.



Similarly in Argo: Menu-> Setup-> Calibration



The measurement results are updated every 0.12 seconds, the accumulation duration of one measurement row is 100 seconds (the accuracy of the frequency estimate is 0.01 Hz) Averaging over 1024

measurements is applied, the expectation dF and the assessment of the measurement accuracy (confidence interval) by criterion 3 sigma are displayed. Evaluation is considered good if the range of measurements does not exceed 0.01 Hz. Such accuracy on a stable, non-drifting frequency signal is achieved rather quickly: for example, it took 40 seconds for the HGA22 signal with SNR = 21 ... 25 dB, and about one or two minutes with SNR = 11..13 dB. On the contrary, if the local oscillator of the receiver is drifting slowly or wandering within fractions of Hz, even on a signal with SNR = 40 ... 50 dB, the frequency estimate will have an error of about 0.02..0.05 Hz (red color), the dF value will continuously slowly drift.

To be sure that the value is really accurate, you should wait 3-5 minutes.

The measured frequency shift, if the measurement error is less than 0.01 Hz, is placed once per minute in the **df.txt** file. This may be useful in the future for automatic calibration.

When switching the Dial, the dF measurements are reset and start again.

Date / time - the current system time of the computer, taking with the time zone. If time needs correction - the color is red, if the time is correct - green. If there is no reception of time stamps - blue.

In the upper right part, the signal-to-noise ratio **SNR** of the received signal is displayed. SNR is calculated as the ratio of the signal level from the output of the band-pass filter (50 Hz BW) of frequency counter Uf ("signal") to the level in the entire band U ("signal + noise") using the formula:

$$\text{SNR} = 20 \log (U_f / (U - U_f)) + 15$$

The normal operation of the DCF / HGA decoder starts at SNR > 15 ... 20 dB. For this parameter, the program is approximately equal to MixW.

Measuring the frequency shift and adjusting the seconds according to the RWM is also possible with significantly lower SNR levels > 3 dB.

On the right below the SNR, the levels from the outputs of the "space" and "mark" filters are displayed in the form of vertical bars. This allows you to visually judge the presence of a station signal on the air. Below in the presence of impulse noise, a lightning symbol appears briefly.

The **DCF-HGA-RWM** field with the name of the received station displays the demodulator / decoder operation:

- **gray**: no carrier
- **green**: receiving "space"
- **light green**: start of receiving DCF / HGA information or receiving a RWM minute label, or receiving a RWM carrier.

- **yellow**: reception of information ended with an error, or there is a manipulation (RWM) in the channel.

The status bar displays the decoded DCF-HGA telegrams or messages about the reception of RWM minute labels and the difference between the system and airtime dT.

Settings in the ini file of the program.

The program creates the INI file DCFHGATime.INI, which stores the following settings:

[FILE]	
PATH=TimeLog.txt	- the log file
DF_PATH=df.txt	- the file for measured frequency shift
[AUDIO]	
SND_INPUT=0	- selected sound device
[TIME_CORRECTION]	
ENABLE=1	- 1=yes, 0=no
EU_GMT_TIME_DIF=1	- difference between EU and GMT [hours]
COUNT_DT_EQUAL=5	- count of same decoded dT for time correction
[FREQ]	
DIAL=9995	- dial freq of RX (136000, 135500, 135000, 9995)
[DATA]	
PARITY_CHECK_ENABLE=1	- check parity of telegram DCF-HGA [1=yes, 0=no]

Some details of the implementation of the algorithm

The sampling frequency of the sound card is 12000 sps. The average signal level is calculated. If the sample exceeds the average level of the module more than 3 times, then it is replaced by zero (Noise Blanker). Next, we used the methods of our fathers and grandfathers - an RTTY demodulator on two filters with detectors, a frequency counter in the form of a filter, and a zero-crossing counter.

Filter options.

RTTY filters:

- 200 Hz BW,
- spacing between filters is 340 Hz.
- implementation - FIR filter, 65 taps.

CW Filter (for RWM):

- 50 Hz BW,
- FIR filter, 65 taps

Frequency counter Filter:

- 50 Hz BW
- IIR filter with 2 poles (oscillating circuit)

The outputs of the RTTY and CW filters are fed to an amplitude detector (fabs () function) and then to the simplest first-order low-pass filters (IIR filter of the form $y = 0.9y + 0.1x$) - an analogue of an RC-chain. These filters smooth the signal.

Then, the output of the first and second RTTY filters are subtracted, the difference sign gives an estimate of the transmitted parcel 1 or 0. ($s = \text{sign}(x_1 - x_2)$)

At the same time, the output of the 1st filter is compared to half of its smoothed amplitude. If the sample is more than half, this is 1. Otherwise it is 0. This is a telegraph (single-frequency, CW) demodulator ($s = \text{sign}(x_1 - y_1 / 2)$)

If, as mentioned above, the second frequency of the teletype signal is beyond the receiver bandwidth, then the output of the 1st filter will always prevail over the output of the second, then the result of their subtraction will give an information signal from many 11111111.

Now compare the outputs RTTY and CW demodulators. If RTTY goes 1, and CW goes 0, then the second frequency is really behind the receivers bandwidth and we will assume that 0 is transmitted in this case. If 1 and 1 are coming from both demodulators, then, of course, 1 is transmitted.

Next, the stream 1 and 0 enters to the median filter. This is a reversible counter, which increases when entering 1 and decreases when entering 0. Its account is limited to +10 and -10. If counter > 0 then output of median filter is 1, otherwise 0. Median filtering eliminates jitter and signal fragmentation.

The frequency meter counts the number of zero-crossing of the output signal of the filter (from + to - and from - to +). The second counter counts the total number of samples. A frequency estimate is calculated every 0.12 seconds.

$$F = 12000 * (N_{\text{zero}} / 2) / N,$$

where 12000 is the sampling frequency

Nzero - the number of zero-crossing - + and + -

N - number of samples

The counting continues for an interval of 100 seconds. Then, to protect the counter from

overflowing and tracking the frequency drift, the Nzero and N values are divided into two and the next 100 seconds are counted. Estimates of frequency, obtained every 0.12 seconds, are recorded in a ring buffer of 1024 cells. Thus, the buffer stores data for 2 minutes. Each time, F_{measured} is calculated as the average value of the estimates stored in the buffer and the standard deviation.

RWM, Moscow, Russia

Frequencies: 4996, 9996, 14996 kHz
 Carrier accuracy: $\Delta f / f < 1 \times 10^{-11}$
 Call sign: RWM
 Location: Moscow, 55° 48' n, 38° 18' e
 Operating hours: Continuous
 Power: 5 kW on 4996 and 9996 kHz, 8 kW on 14996 kHz
 Modulation: On-off keying (A1B)
 Identification signal: Call sign in Morse in minutes 09 and 39
 Programme: Schedule repeats every 30 minutes, see Table 13
 Time code: None

00m00s - 07m55s	30m00s - 37m55s	Unmodulated carrier
08m00s - 09m00s	38m00s - 39m00s	No transmission
09m00s - 09m55s	39m00s - 40m55s	"rwm rwm rwm ..." in Morse code (before 2004: "vvv cq cq cq de rwm rwm rwm")
10m00s - 19m55s	40m00s - 49m55s	1 Hz pulses with UT1-UTC code, see Figure 5. Pulse duration = 100 ms, minute pulse = 500 ms
20m00s - 29m55s	50m00s - 59m55s	10 Hz pulses. Duration = 20 ms, second pulse = 40 ms, minute pulse = 500 ms

Table 13. RWM hourly transmission schedule

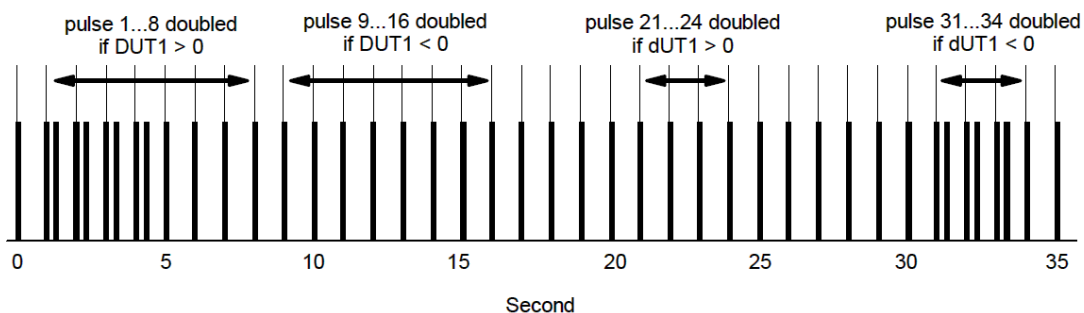


Figure 5. Coding of the deviation between astronomical time UT1 and atomic time UTC. The difference in seconds is given by $0.1 \times \text{DUT1} + 0.02 \times \text{dUT1}$. In this example, DUT1 is +4 and dUT1 is -3, hence $\text{UT1} - \text{UTC} = +0.34$ seconds.

