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# Computer operations

A fully equipped hamradio operator can set up SSTV reception in an hour. Start with creating a connection between a transceiver and a sound card using a compatible cable. Then, download relevant SSTV related software and tune into 14.230 MHz using upper-sideband (USB). There is round the clock SSTV activity around this frequency.

The available software for SSTV operations are very similar and they all provide the same basic functions, of course with different comfort levels. Some of them are intuitive and well-arranged, while others require studying a manual. There is the possibility to choose from more variants and see what works best for you. Every operator must know how to carry out these basic tasks:

- ▷ configuring an accurate sampling frequency for reception and transmission,
- ▷ setting proper sound card volume levels,
- ▷ using the tuning indicator,
- ▷ manually change the SSTV mode,
- ▷ loading and saving images in common graphic formats,
- ▷ creating an image gallery for transmission and
- ▷ adding text into transmitted images.

## 7.1 Hardware configuration

Take your time in selecting a suitable configuration for the computer in your hamshack. The unpleasant fact is that, operating systems and distributed software constantly require better CPUs and more memory. For SSTV operations it is still possible to use obsolete PCs with old 386 processors. The *Hamcomm* modem and

some other popular programs from the nineties run on DOS. In this case, 4 MB of RAM, a hundred megabytes hard disk and SVGA graphics card with 256 colors are enough. But graphic cards with 32 or 64 thousand colors are more suitable.

To use a sound card, a good PC is necessary. The PC should be configured with at least a 150MHz Pentium, 64 MB RAM and few gigabytes hard disk and a graphics card that supports at least a 1024×768 resolution in 64 thousands or 16.7 million color mode. The minimal operating system is Windows 95 OSR2, but some new programs may not run in Win95. I recommend having better hardware but you don't need the latest mega-hyper model.

There are also software products for Mac OS X and for GNU/Linux, but the largest selection is for Microsoft Windows.

## 7.2 Using a sound card as a modem

A sound card is standard PC equipment these days. It can be used as a music player for multimedia, games and recording. The main application of a sound card in hamshacks is as a MODEM. The modem – MODulator/DEModulator allows computer information to be transmitted and received over physical media like radio waves or telephone lines. The modem translates analog signals to digital data and vice versa.

### 7.2.1 Sound processing in PCs

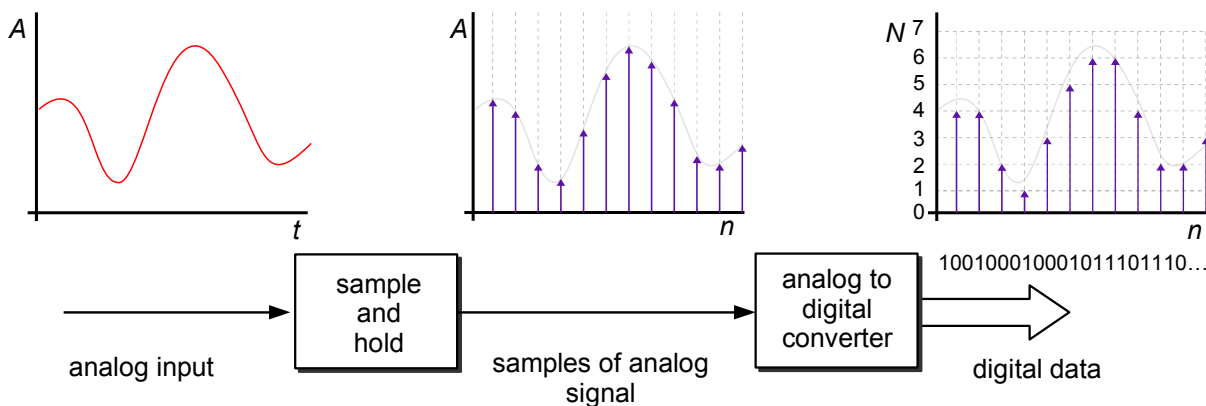
To allow the computer to work with sound signals, the signal must be converted into a format suitable for data processing (digital or discrete signal).

#### 7.2.1.1 Sampling

The digital conversion process begins with *sampling*. Sampling is an activity which periodically scans the current value of the analog signal. For example, this happens 11,025 times per second or depends on a user defined sample rate supported by the sound card. The sampling frequency of sound cards ranges from 8 kHz suitable for internet telephony up to 96 kHz designated for more exacting requirements of recording studios.

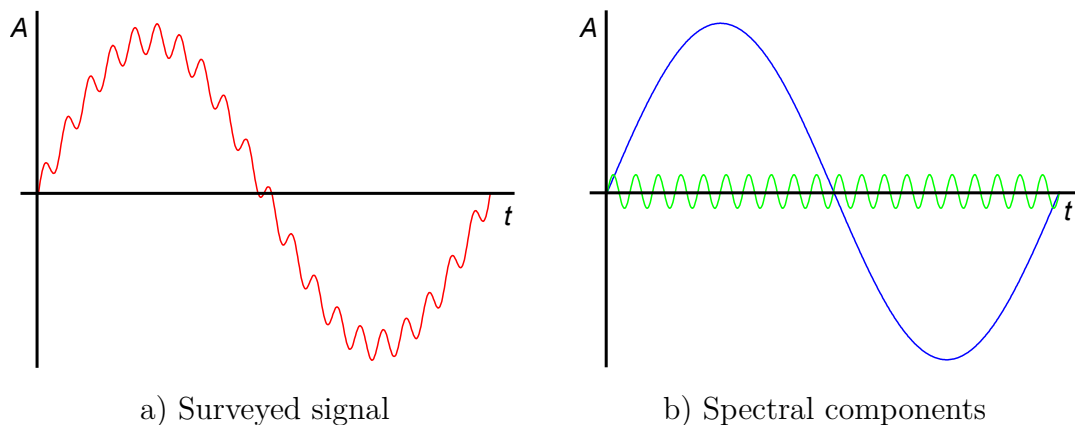
The answer to the question of what sampling frequency should be used is given by Shannon's theorem (also known as Nyquist-Kotělnik-Shannon theorem). It defines that a signal continuous in time, containing spectral components with the highest frequency  $f_{max}$ , can be clearly reconstructed from a sequence of evenly spaced samples with a sampling frequency  $f_s$  greater than double  $f_{max}$ :

$$f_s > 2f_{max}$$



**Figure 7.1:** The conversion of an analog signal into numeric data.

You can see the importance of Shannon's theorem in the example below. The signal in **figure 7.2a** expresses the dependency between time  $t$  and amplitude  $A$ . Using Fourier's analysis, we can find (see chapter ??), that the signal contains two harmonic components, showed in **7.2b**.



**Figure 7.2:** Example of signal.

By using Fourier's transformation, the signal can also be expressed as a dependency between amplitude  $A$  and frequency  $f$  – *signal spectrum*. Both frequency components are evident  $f_1$  and  $f_2 = f_{max}$  in the signal in **fig. 7.3**.

For explicit signal reconstruction the condition  $f_s > 2f_{max}$  must be satisfied, see **fig. 7.4**. If a sampling frequency is lower than  $2f_{max}$  then the higher frequency components causes signal distortion. This error is called *aliasing*.

For the sampling of common narrow band signals like SSTV, RTTY, PSK31 or WEFAX, which are transferred via SSB channel with a bandwidth of about 2500—3000 Hz; a sample rate of 11,025 Hz is enough.

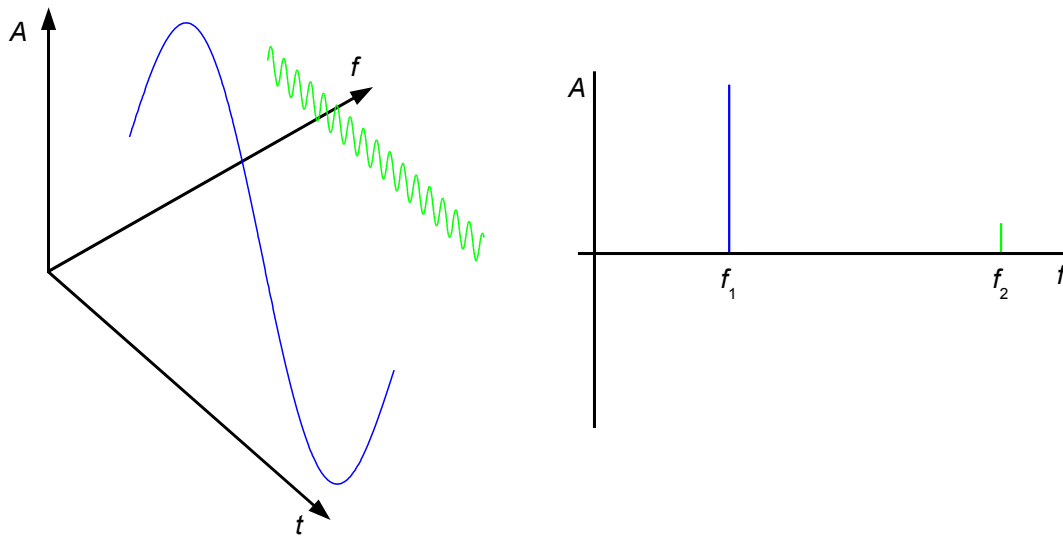


Figure 7.3: The frequency spectrum of signal in **fig. 7.2a**

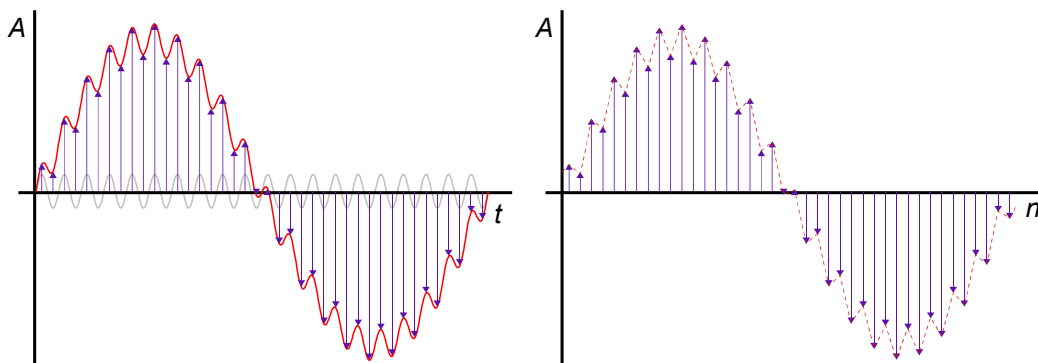
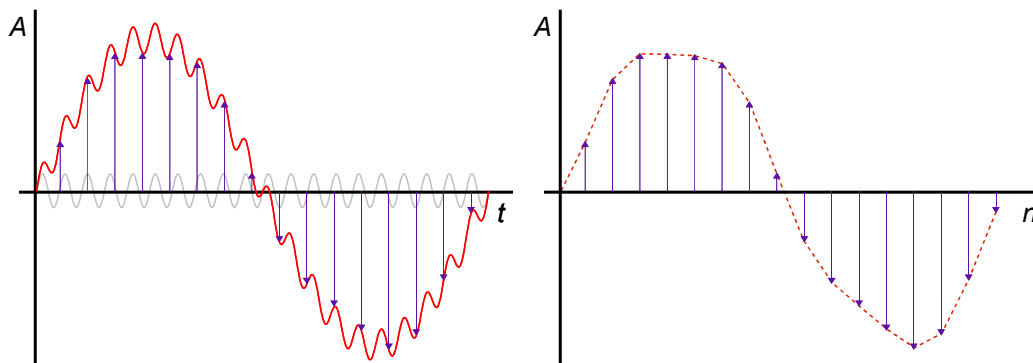


Figure 7.4: The signal sampled with a frequency higher than  $2f_{max}$

## 7.2.2 Analog-to-digital conversion

The analog signal sample continue into the analog-to-digital (A/D) converter. The current value of the signal is converted into digital data in this device. Some A/D converters work with a resolution 8 or 16 bits according to the type or settings of a sound card. The resolution of an A/D converter indicates the accuracy of signal amplitude scan in a defined range, for 8 bits it is  $2^8 = 256$  values and for 16 it is 65,536.

A constrained resolution of the A/D converter causes a *quantization* error. E.g. for an 8 bit converter processing voltage range 0–5.0 V it's the error  $5.0/(2^8 - 1) \doteq 0.02$  V. The 8bit converter cannot distinguish between voltage levels lower than 0.02 V. So for input voltage 3.111 V it could find a corresponding numeric



**Figure 7.5:** The sample rate does not meet the Shannon's theorem.

value  $10011110_2 \approx 3.098 \text{ V}$  or  $10011111_2 \approx 3.118 \text{ V}$  because less significant bits are influenced by the quantization error. The size of the error can be decreased by greater resolution of A/D converter. For our purposes the 16bit resolution is acceptable.

A modern sound card could be equipped with a digital signal processor (DSP), which extends card functions e.g. for filtering or data compression during recording so it can lighten the load of the main computer CPU. E.g. *Sound Blaster Live!* contains a programmable DSP labeled *EMU10K1*.

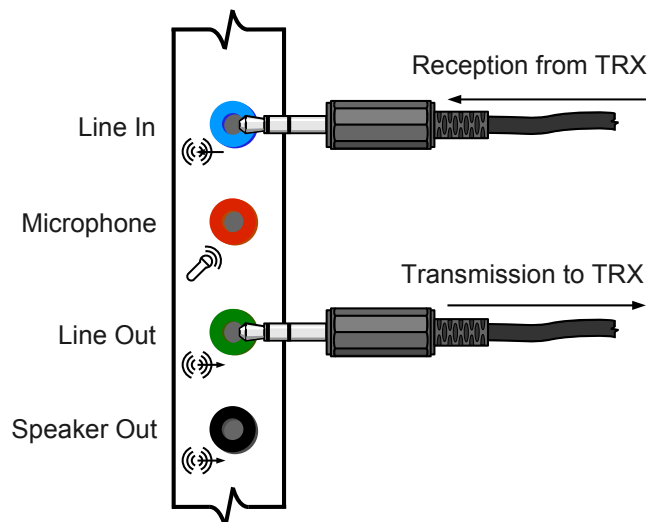
The choice on which sound card to use, depends solely on the user's preferences and intentions. With the rise of cheap Software Defined Radio (SDR) solutions, always choose a sound card with a higher sampling rate and analog anti-aliasing circuits at the input. Many PCs come with an integrated sound card on the motherboard.

### 7.2.3 Interface between TRX and PC

The basic interface is made of shielded cables and 3.5mm jack plugs. A reception cable connects sound card input *Line In* and TRX headphones output or output for external speakers. To use the sound card microphone input, use the TRX output often labeled as *AF OUT* with an impedance of about  $10 \text{ k}\Omega$  which gives a maximum voltage output of  $100 \text{ mV}$ . This output could also be used for interfacing tape decks or audio amplifiers. A sound card's microphone input has an automatic gain controller (AGC) for better recording and it is possible to connect dynamic microphones with impedance ranges from  $600$  to  $10.000 \Omega$ .

For the transmission it is possible to use *Line Out* with impedance of about  $600 \Omega$ . The *Line Out* can be connected to the microphone input of the TRX or a rear panel connector like *PATCH IN*.

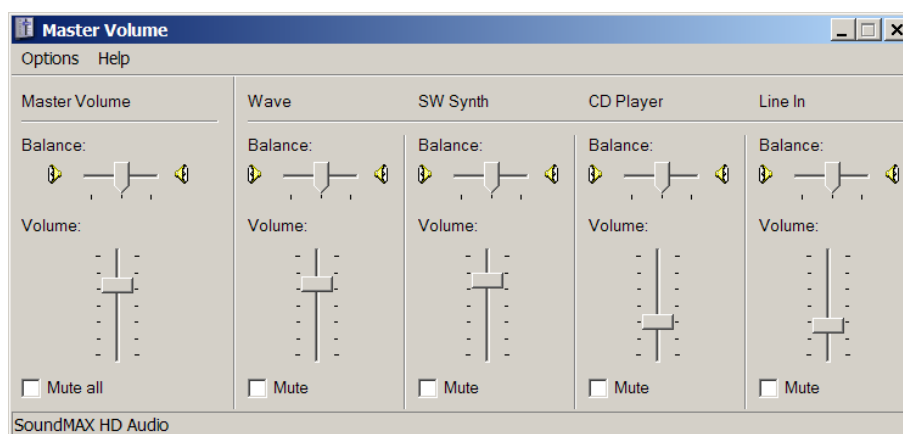
Some transceivers have a feature that the microphone input and rear panel input are interconnected, so it is necessary to disconnect the microphone during AFSK



**Figure 7.6:** Basic interface between transceiver and sound card.

transmission, because noise in the hamshack could interfere with the sound card's signal! Check your TRX instruction manual for relevant information.

The last thing you need to set up, is the audio level for received and transmitted signals. This can be done using operating system tools. The level of transmitted signal should be about 2/3 of maximum volume levels. The signal should not be too attenuated or over-excited and distorted. You can detect it by monitoring the outgoing signals. For the incoming signal you can set the proper level directly on the TRX and check the input level in your SSTV software.



**Figure 7.7:** Software volume control.

After the audio mixer configuration, it is useful to save your sound card settings (so you can easily restore it before SSTV sessions). A program such as *QuickMix* can easily store your settings. This is important because other PC program can change sound card levels.

<http://quickmix.softpedia.com/>

### 7.2.4 PTT control

The PTT (Push-To-Talk) button switches between reception and transmission. There are several methods used for PTT control:

1. Manual PTT switching; this method is not very elegant, but can be used for initial experiments.
2. Automatic switching can provide the TRX with a VOX feature, when the TRX is automatically keyed by signal in the input. A disadvantage of this method may be that its reaction is not immediate, so in case of digital modes the beginning states of transmission or SSTV vertical synchronization can be lost. Keep in mind that operating systems often produce malicious sounds that could accidentally key the transmitter.
3. Computer controlled automatic PTT switching; all SSTV programs support PTT control over a simple serial port (COM, RS-232) circuit. The circuit contains one switching transistor or opto-isolator and few passive parts. See **schematic 7.8** for details. The control signal is connected to RTS pin (7 at Cannon DB9 connector, 4 at DB25) or DTR (4 at DB9, 20 at DB25), the selected pin can be changed by software configuration. The ground is on both serial ports wired on pin 5 at DB9 and 7 at DB25.

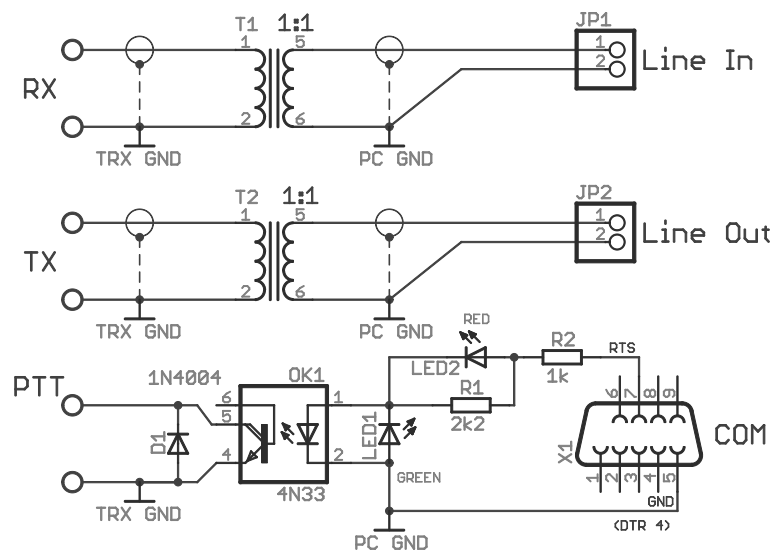
A large number of handheld TRXs use the same connector for both microphone input and PTT. In this case an audio signal should be galvanically separated by the capacitor at about 100 nF and the PTT signal can be connected by a resistor; whose resistance can be found in the TRX instructions or you can connect a trimming resistor at about 15 k $\Omega$  and test the max. value when TRX switching.

4. Some transceivers support control over the serial port. The CAT (Computer Aided Transceiver) interface can provide PTT switching. Commands can also be sent over the CAT interface e.g. for tuning, mode control, etc. This method must be supported by the installed software, for example MixW can control some TRXes so it is not needed to practically touch the TRX buttons.

What can you do if your computer is not equipped with a serial port? Some motherboard manufacturers build only one serial port and notebook manufacturers don't provide any. If this happens, you can use VOX or buy a USB/RS232 interface. Some programs also support similar switching circuits as described before but on parallel port (LPT).

### 7.2.5 Eliminate supply noise

A computer and a TRX can have slightly different electrical potential and in this case the direct connection causes annoying noise within the communication channel. It is possible to remove the noise by the galvanic separation of both devices. The path of the audio signal should go through the galvanic transformer and PTT control switch with an opto-isolator, e.g. 4N25, 4N33, etc. You might need to change R2 to a lower value if the opto-isolator is not switched on properly when serial port PTT's signal is active.



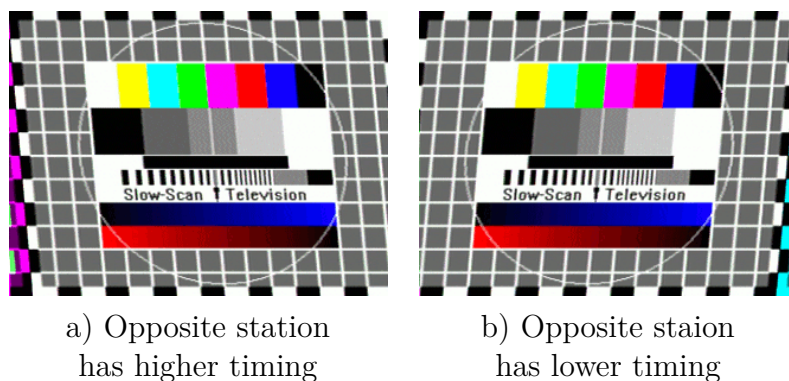
**Figure 7.8:** The galvanic separation of transceiver and sound card.

## 7.3 Timing oscillator configuration

There is a description of synchronous (free-run) SSTV system in section ???. Horizontal synchronization pulses (syncs) are detected only at the beginning of reception and after synchronization, a reception device stops detect syncs and receive with free-run scan. Due to this there are excessive requirements for accurate timing of corresponding stations.



If the timing slightly differs then images are distorted – inaccurate timing causes image *slant*. You can see image slant for 0.01 % timing difference in **fig. 7.9**. If a transmitting station has higher timing (and reception lower) the image slants to the right (**7.9a**) in opposite situation to the left (**7.9b**).



**Figure 7.9:** Image slant distortion when inaccurate timing is used for free-run modes.

However timing derived from sample rate is not used to be exactly 11,025.00 Hz, but often can differ up to few tenths of percent for each piece of hardware. For speech and music processing it doesn't matter, but in free-run transmission of SSTV it causes problems.

The configuration of accurate timing/sample rate for reception and transmission apart must be done to meet the strict requirements for synchronous SSTV broadcast. Your signal must be acceptable for any SSTV device.

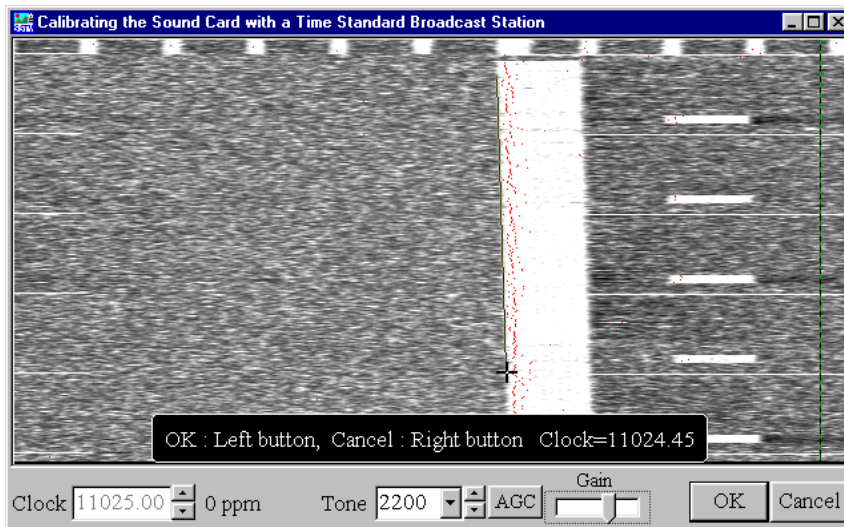
All SSTV program are equipped with a tool for the timing configuration.

It is possible to receive SSTV signal from a band and by an edge of received image set the timing – program will automatically compute timing deviation. This way has a disadvantage, because not all SSTV stations has proper transmit timing. This is caused by offset between receiving and transmitting timing.

Much more precise way is use of shortwave broadcast of timing normal. Programs are equipped with special reception option, which displays spectrum in a second cycles. For timing setting just tune to the frequency of broadcast and leave to plot received pulses for several minutes.

The usable transmitter is a Moscow station RWM operating on frequencies 4,996.0, 9,996.0, 14,996.0 kHz with 8 kilowatts power. So it can be nicely received in Europe/Asia region. Select CW mode and tune your receiver directly to one of station frequencies. The unmodulated carrier is transmitted between 0. and 8. minute of an hour, telegraphy identification goes from 9. minute and then the timing signal will continue. Pulses in intervals 1/60 and 1 Hz goes between 10. and 20. minute

and 10Hz pulses goes between 20. and 30. minute. This is repeated every 30 minutes.



**Figure 7.10:** The configuration of accurate timing with RMW reception in MMSSTV.

The reception of WWV station is the next possibility. This station broadcast timing pulses and announcement on frequencies 2,500.0, 5,000.0, 10,000.0, 15,000.0, 20,000.0 kHz and uses double sideband (DSB) modulation. You can receive it with AM mode selected. The WWV operates from the North America, Fort Collins in Colorado. The used power ranges from 2.5 to 10 kW.

There is yet another way with WEFAX station reception, because these stations must have accurate timing too due to synchronous transfer.

The deviance error you should measure use to be expressed like absolute value of actual frequency, e.g.  $f = 11024,45$  Hz or like deviance from  $f_s$  the  $\Delta_f = -0,55$  Hz. Some program this measure in *parts per million (ppm)* unit. The ppm deviance can be computed:

$$\Delta = \frac{\Delta_f}{f_s} \cdot 10^6.$$

For the  $f = 11024,45$  Hz the deviance in ppm is:

$$\Delta = \frac{\Delta_f}{f_s} \cdot 10^6 = \frac{-0,55}{11025,00} \cdot 10^6 \text{ ppm} \doteq -50 \text{ ppm}.$$

### 7.3.1 Transmit timing offset

There is necessity to configure transmit timing *TX offset* after the precise configuration of reception sample rate, when received SSTV images are not slanted. It

is important for your own transmission, because inaccurate transmit timing causes image slant on reception side.

Some programs makes possible to monitor outgoing SSTV signals, so with this feedback it is practicable to check the TX offset – deviance between reception and transmission sample rate. The feedback can be internal or external. *External feedback* needs to connect Line Out and Line In with cable and it requires a sound card with full-duplex mode enabled. By this way you can set *TX offset* precisely on your own.

The *internal feedback* doing almost the same, but no external cables is needed. But some sound cards support only software feedback, so you will find zero deviance, but it is not real fact! Then the TX offset setting must be done with external feedback or with opposite station help. You need to disable any automatic corrections of received signals in this way.

Anyway you need to make “dry run” QSO before your first CQ. This helps you to uncover possible problems with TX offset, supply noise, audio level, etc.

The TX offset issue is often pretty messy. You can notice that some software running concurrent with your SSTV program can influence sound cart output and then the change of sample rate occurs. Even the simple Volume Control tool can do this. So it is useful to stop unnecessary program running in the background. Especially programs that can influence sound card output or decreases stability of Microsoft Windows.

You may notice a strange behaviour if you are user of modern sound card with full duplex mode enabled with several output channels with a support of different sample rates for each channel. This is for example *SB Live! Value*. I have noticed that my TX offset randomly changes! I have this experiences with SB Live! Value and I found that another radio amateurs has same. You can try to set other sample rate than 11,025.0 Hz in this case, if this doesn't load your computer too much. For example try 48,000.0 Hz, this value is fixed sample rate (see your card user's guide) and best results you can achieve with using of this value or its half or quarter – 24,000.0 kHz or 12,000.0 kHz. When you change this value you need to recalibrate your accurate timing again.

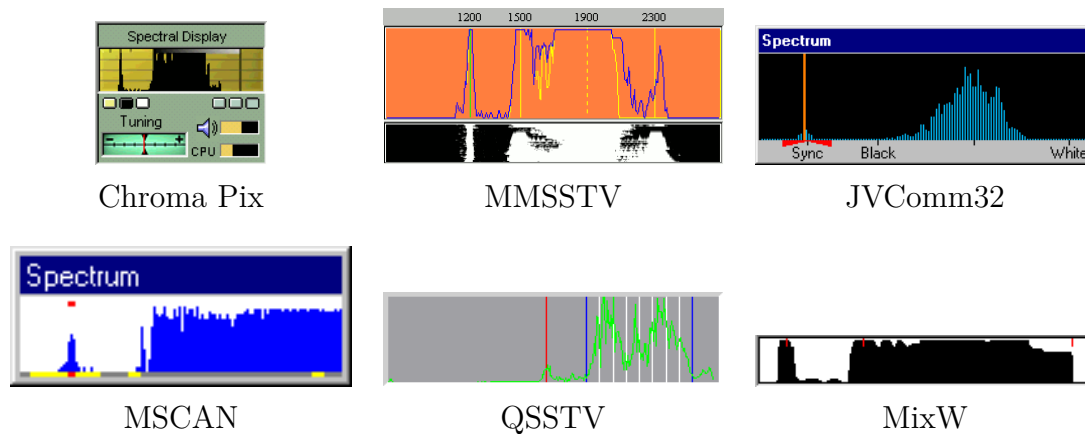
To avoid these problems you can constantly monitor the outgoing signal through the external feedback with the TRX monitor enabled and before your today first transmission you will check that everything is fine. It's unpleasant that problems often occurs during QSO and then you will stunned by counterpart replay images.

## 7.4 SSTV tuning

First of all, we need to find SSTV stations by listening near calling frequencies. Thanks to typical SSTV sound and clattering of syncs it is not a problem to distinguish between SSTV and other communication modes.

Every SSTV program is equipped with precise tuning indicators – spectroscopes, see fig. 7.11. The spectroscope shows frequency band from 1000 Hz to 2500 Hz with marks for critical frequencies – 1200 Hz for syncs, 1500 Hz and 2300 Hz for the video signal.

Is is possible to simple detect band of video signal and syncs during clear reception. Rotate the tuning knob to achieve that all important frequencies are aligned in spectroscope display.



**Figure 7.11:** Spectroscopes in common SSTV programs.

## 7.5 Video digitalization

The video digitizer should be additional equipment of SSTV station. The device can convert output signal from camera into computer form. There is a great choice of many different devices with varying capabilities, parameters and price. You can choose some webcams, frame grabbers, TV cards or digital cameras. Then your broadcast will not be limited only to pre-prepared images and you will have a lot more fun with live transmission.

The cheapest option are web cameras, they are equipped with a low-resolution CCD and low-cost optics, but provided quality is suitable for SSTV.

An another option is a TV card with video input. This possibility is more expensive because you must connect an external camera where the choices ranges from cheap CCTV black and white or color CCD cameras up to professional studio equipment.

## 7.6 Software for Windows

### 7.6.1 List of programs

#### 7.6.1.1 SSTV software

- ▷ Chroma Pix – <http://www.barberdsp.com/>
- ▷ JVComm32 – <http://www.jvcomm.de/>
- ▷ MMSSTV – <http://mmhamsoft.ham-radio.ch/>
- ▷ MSCAN SSTV, Meteo – <http://www.mscan.com/>
- ▷ Winskan, SSTV32 – [http://webpages.charter.net/jamie\\_5/](http://webpages.charter.net/jamie_5/)
- ▷ W95SSTV – <http://www.barberdsp.com/w95sstv/w95sstv.htm>

#### 7.6.1.2 Digital mode software with SSTV support

- ▷ MULTIPSK – [http://f6cte.free.fr/index\\_anglais.htm](http://f6cte.free.fr/index_anglais.htm)
- ▷ MixW – <http://www.mixw.net/>

#### 7.6.1.3 Software for dedicated interfaces

This programs don't use sound card as modes but special interfaces.

- ▷ Bonito Radiocom – <http://www.computer-int.com/rc.htm>
- ▷ Roy 1 – [http://www.roy1.com/dvb\\_ham/dvb\\_1.htm](http://www.roy1.com/dvb_ham/dvb_1.htm)
- ▷ Wraase SC-4 – <http://www.wesacom.de/sstv/>