



FreeDV Quick Start Guide V 1.1.0

Here is a one page doc that should get you started on HF DV.

For detailed instructions, see the detailed document.

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1. **Hardware:** For TX and RX, two sound devices are required. One internal sound card, a USB sound interface -or- a transceiver with a built-in sound device will be used for the data. The second device may be “Gamer” or equivalent USB headset for the voice I/O. Any modern PC with USB ports and a sound card will work. If you just want to “listen to digital voice,” then only one soundcard/device is needed.
2. **Install** FreeDV with the Windows installer. Select the two sound devices and test them with Audio Config tools. Next, select a conventional com port for PTT or a serial device using USB with Hamlib. In Tools>Options, add your call sign/name/location. Keep this text entry as short as possible.
3. **Radio DSP:** Turn off TX and RX DSP processing. Think flat BP filters, no speech compression. Turn off the radio’s TX monitor.
3. **Tuning:** Set the radio dial on frequency (try 14.236), then click on “Start.” FreeDV needs about 3dB SNR to decode without voice dropouts. The small Red Marker at bottom of the Waterfall must be on the BPSK Pilot tone carrier (normally at 1500Hz) to obtain sync. Use the mouse to move the Red Marker. Do not re-tune once sync has been established. For more frequencies, visit <http://qso.freedv.org> QSO Finder.
4. **Power:** Run only 20% of the radio’s rated output. Peak power is much higher. Avoid ALC action.
5. **Mode:** Start with the *default* “1600.” Use 700 mode for low SNR signals and “DX.”
6. **Equalizer:** Bring up the mid-range (treble) gain and lower the bass for the mic and speaker.
7. **Voice decoding** begins immediately after the received signal is “tuned in” (in sync). Expect sync around 3dB SNR and good voice decoding around 4dB SNR for 1600 (even lower for 700 **mode** but with lower audio quality). The higher **the SNR is**, the better for all modes...
8. **Squelch:** Start with the default of 1.5 (dB). Open (**0 dB**) under extreme signal conditions.
9. **Split** operation allows changing the receive frequency without affecting the transmit frequency.
10. **Analog** mode bypasses DV and routes audio for SSB and analog frequency monitoring.
11. **Videos:** “Microsoft Windows Users Quick Start” and “Video Guide for Microsoft Windows Users” are available at FreeDV.org Please take a few minutes and watch these instructional videos.

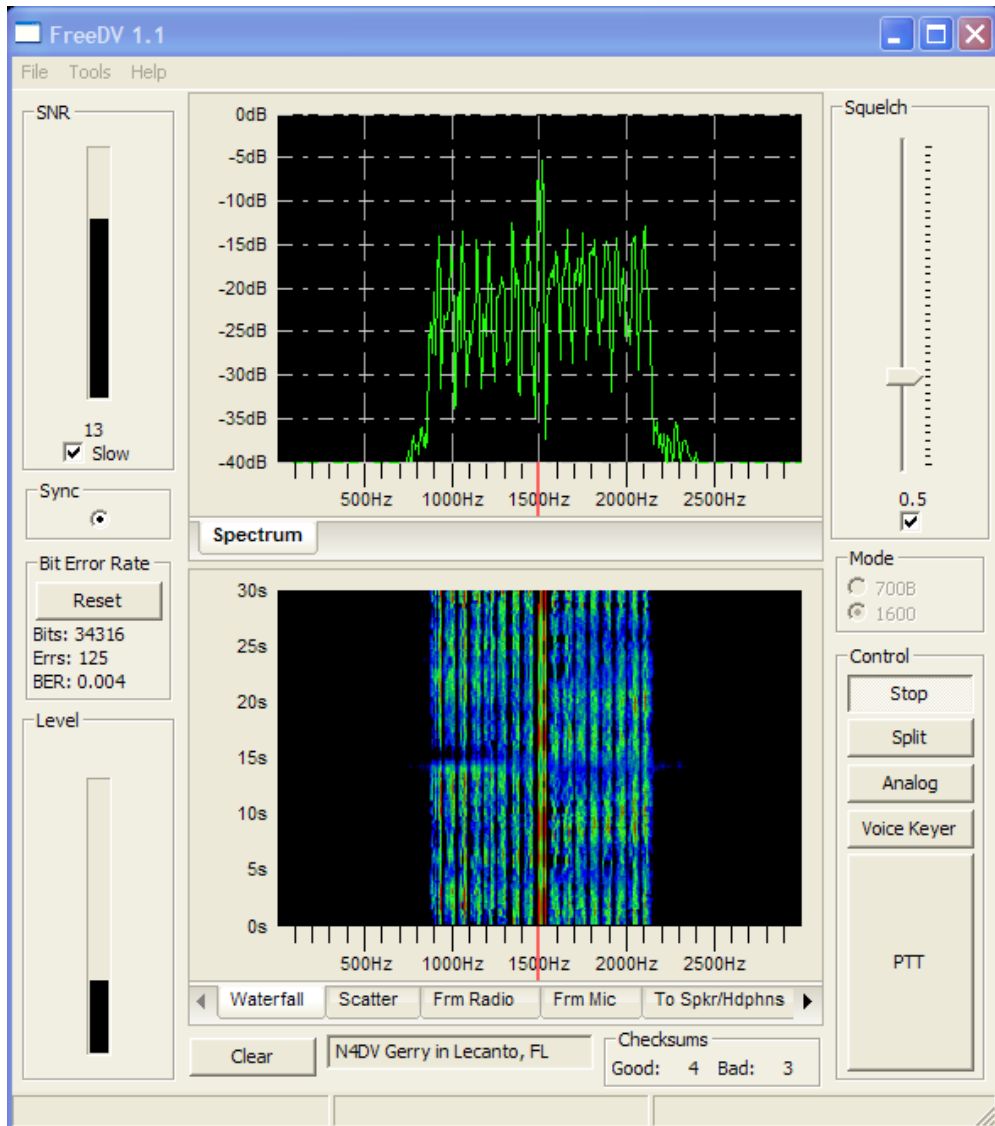


FreeDV...Digital Voice for HF

Detailed Document v 1.1.0 15-Oct-2015

The screen shots in this document are taken from the Windows version of FreeDV(1.1.0). When running on other platforms such as MAC, Linux or Flex, or earlier or later versions of FreeDV the screens may look different.

If you are using or thinking of buying the SM1000 hardware device, please refer to the documentation supplied with that device and on the rowtel.com website for further guidance.



FreeDV Graphic User Interface showing waterfall and Spectrum displays

Table of Contents

Hardware:.....	4
Software:.....	6
Configuration steps:.....	7
PTT Configuration.....	8
Voice Keyer.....	9
Identification Text Configuration	9
Audio Configuration.....	10
Tools - Filter and Equalizers.....	11
Tools - Recording Features.....	12
Receiving Digital Voice:	13
Received signal displays.....	15
Transmitting Digital Voice:	18
Operating Tips:.....	20

FreeDV is a HF Digital Voice program for Windows, Linux and Apple utilizing an FDM modem and the open source low bit rate CODEC2. Compatible with SSB radios, FreeDV provides communications voice audio occupying less than half the bandwidth of conventional SSB. FreeDV may also be used with other radio **transmission modes** (FM and AM) for conversion to a Digital Radio.

Some Technical Specs:

- Codec2 voice codec (vocoder) and DQPSK and COPSK modems
- 1.25 kHz spectrum bandwidth (half SSB) with 75 Hz carrier spacing for the default mode.
- FreeDV 1600 (default) mode: 1275 bit/s voice coding, 25 bit/s text for call sign ID, 300 bit/s FEC, 16x50 baud QPSK carriers, Differential QPSK demodulation
- FreeDV 700 mode*: 650 bit/s voice coding, 50 bit/s text for call sign ID, no FEC, 14x75 baud QPSK carriers, frequency diversity to combat fading, coherent QPSK demodulation, 1.5KHz BW
- No interleaving in time, resulting in low latency, fast synchronization and quick fade recovery.
- 44.1 or 48kHz sample rate sound card compatible

*References in this document to 700 mode refer to the version B of that mode at present. As 700 mode is being developed later versions 700C etc. are likely to appear in later versions of FreeDV.

Please refer to ["FreeDV Specification"](http://freedv.org) at freedv.org for additional details.

["Guide to microphones for FreeDV"](#) and ["Video Guide for Microsoft Windows Users"](#) videos are linked to from the freedv.org website. These are excellent source of help for installing and operating FreeDV, take a few minutes to view these before starting the setup.

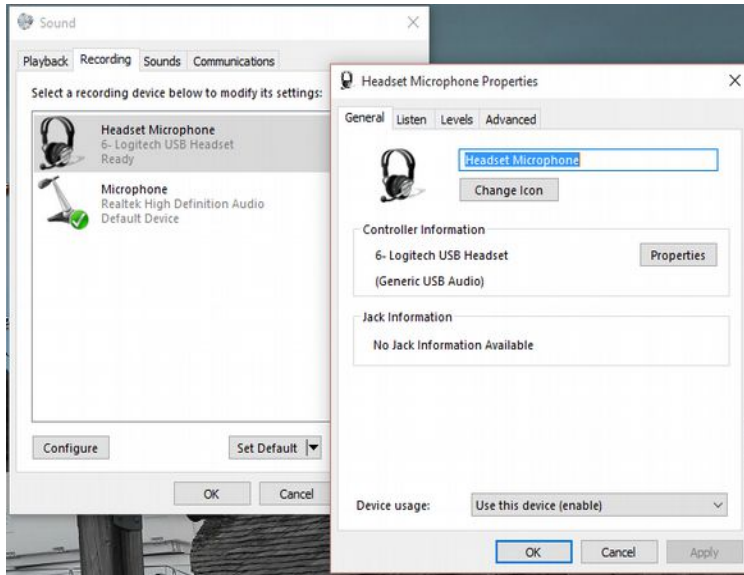
Hardware:



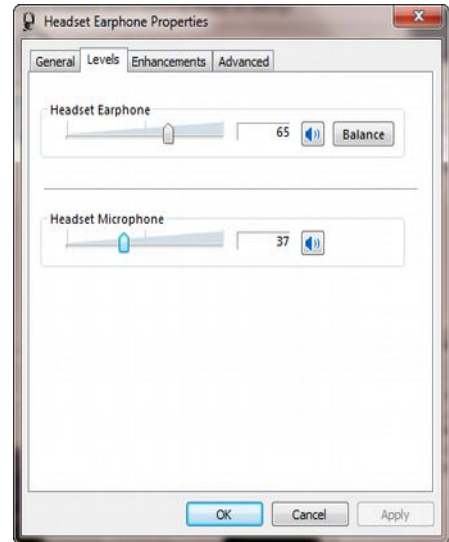
Sound card and PTT.

The control setup is the same as with other digital modes such as PSK31 except DV requires a "PC Gamer" or equivalent USB headset for voice input and received audio output. As an alternative to the headset, a second sound card (physical card in a desktop PC or an external USB "dongle") with a microphone and speaker may be used. Like other digital modes, a radio to PC interface, either homemade or purchased, is necessary to provide isolation and relay control for PTT. To "listen only," just one sound card is needed.

After connecting/installing the sound devices, identify and locate their associated Windows Mixers and slider controls. In **Windows 7** go to the sound mixer program via the Control panel, in **Windows 8/10** right click the speaker icon in the tray and select "recording devices" or "playback devices" as required.



Windows 8/10



Windows 7.

Make sure you understand which recording audio device is the input from the rig (received audio) and which from your microphone and which playback device output is going back to the rig (transmit audio) and which will be the decoded digital audio that you will hear in your headphones or through a speaker. By selecting properties and then level, you can adjust the level.

If you are currently using a digital mode such as PSK31, the transmitter and receiver levels should be relatively close, but you will still need to set the levels on the “second audio device” (USB Headset or Microphone plus speaker).

Advanced Windows user notes:

Audio - If you have a current model laptop, it will not have the 3.5mm audio input and output jacks needed by some of the rig interface systems. In this case you need to add a USB audio dongle for the Mic and headphone plugs from the rig interface to plug into. If you have a modern rig with a USB port on the rig, you will only need a USB cable to connect directly from the rig to a USB port on the laptop or desktop PC.

In all cases it is possible that when you use more than one USB Audio Interface, the names can be very similar (or even the same). If you wish to change the name Windows uses for the USB Audio ports, it is documented how to here: <http://www.eightforums.com/customization/15321-tutorial-how-change-device-names-device-manager.html> - note this action requires changes to the Windows registry – perform with due care!




PTT - If your digital rig interface, PTT switching interface uses an early Prolific USB-Serial interface chip, you will need to manually install the Windows Vista driver for it if you are using Windows 7 or later editions of Windows as the latest driver disables early Prolific chips (trying to stop clones from working, but disabling original parts as well). If you have bought your rig interface recently, this problem should not occur.

Software:

FreeDV for Windows will execute and run on any current home PC or laptop with a dual core or higher Intel/AMD CPU. Virtual Audio Cables (VAC) commonly used with SDR radios are compatible with the FreeDV software. Dependent upon your hardware interface and radio, PTT rig control may be configured for VOX, Hamlib or a serial port. Flex, HPSDR and other SDRs may use a virtual com port pair. Note Flex Users: Flex GUI does not include ability to change Squelch (fixed at around 3-4dB).

Under the **download** section on the freedv.org web page, you will find the latest released code for 32 and 64 bit Windows and other operating systems. If you are running a 64 bit version of Windows on your PC, select the 64 bit, otherwise select the 32 bit version. If you are not sure whether the Windows version on your PC is 32 bit or 64 bit, select the 32 bit version.

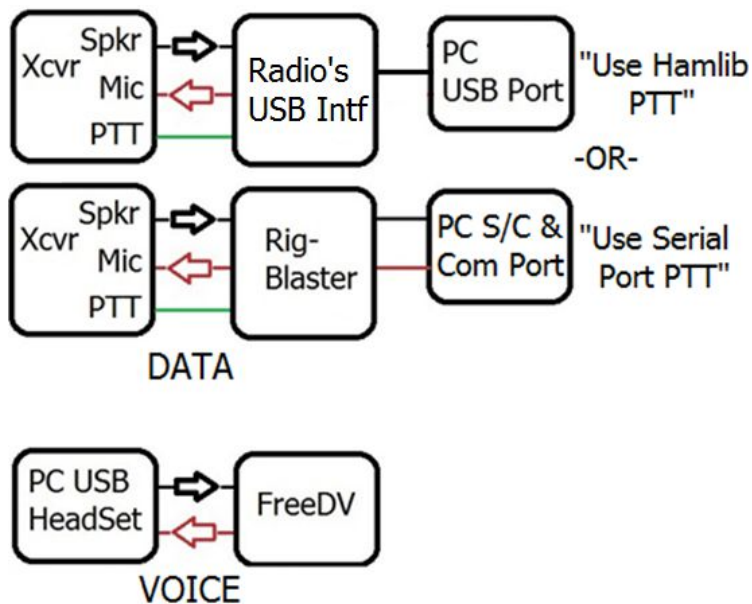


 bin	9/30/2015 1:48 PM	File folder
 log.txt	9/30/2015 1:54 PM	Text Document
 Uninstall.exe	9/30/2015 1:48 PM	Application

This will install the Windows software including an uninstall.exe file. The bin folder will contain dll's, freedv.exe, and the PTT Hamlib dll's for CAT control. No other files will be created after configuration. After installation, execute FreeDV from the start menu.

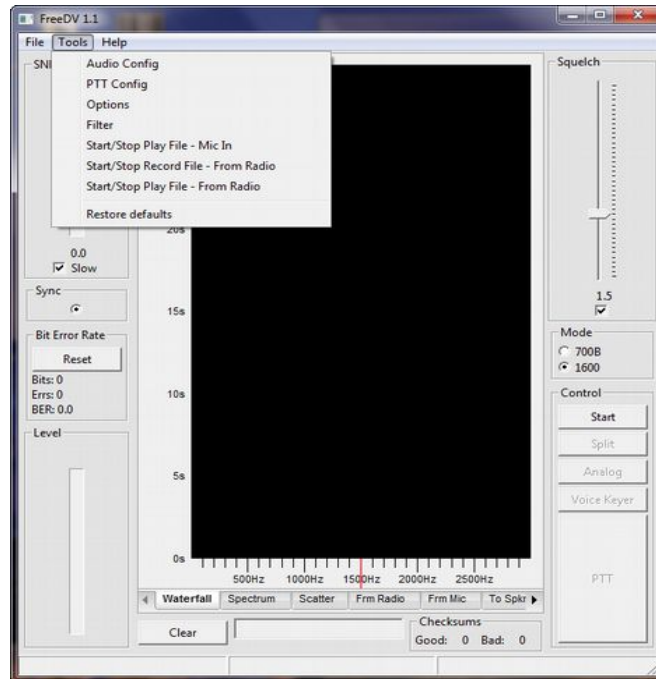
Configuration steps:

Connectivity - Rig to PC interface. For conventional SSB radios, connect the interface you have according to the supplier's documentation. When configuring the PTT option in FreeDV, and you have a Signalink™ device configure it's USB port to use Hamlib PTT control otherwise select the serial port of the control cable. Connect your "second audio device" that you will use to talk into and hear decoded DV. This can be a USB headset or a separate sound card or USB audio "dongle" with a microphone and speaker.

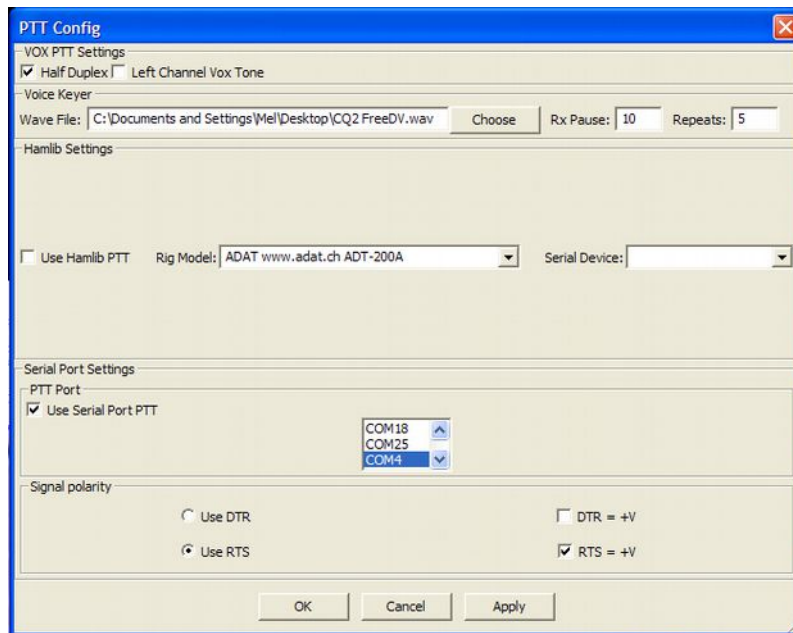


PC to Radio Interface Connections

PTT Configuration



Tools Pull down Menu



PTT Configuration page

Under Tools, go to “PTT Config” and select the Com Port for the radio’s PTT control (recommended over use of VOX control). Either Hamlib PTT or conventional Serial Port Settings may be used. Select the Serial Device or COM port used for PTT on the interface. “RTS= +V” is normally used for rig control. If the radio is keyed into transmit after clicking “Apply,” change the polarity. Then click OK. If this still does not control the PTT, try selecting DTR or combination of both.

Voice Keyer

The PTT config panel is also used to set up the Voice Keyer if you wish to use it. A voice keyer can be useful while running tests or in a contest. Think of it like a continuous loop tape. You will need to record the audio you wish to send as a WAV file using whatever program you have available on your PC to do this. *Note:* the audio **MUST** be recorded with an 8K sample rate or it will not work correctly with FreeDV. See “Voice Keyer” recording information under [Receiving Digital Voice](#) within this document.

Identification Text Configuration

The screenshot shows the 'Options' dialog box with the following settings:

- Test Frames:** Enable, Channel Noise
- FreeDV 700 Clipping:** Enable
- Txt Msg:** DD5LP in JN58la
- Text Encoding:** Use Checksum on Rx
- Event Processing:**
 - Enable System Calls Syscall Spam Timer: 10
 - Regular Expressions to Process Events:**

s=(.*)	curl http://qso.freedom.org/cgi-bin/onspot.cgi? s=\1
--------	---
 - Log of Events and Responses:** (Empty text area)
- UDP Control Port:** Enable UDP Control Port UDP Port Number: 3000

Buttons at the bottom: OK, Cancel, Apply

Options page

Under Tools, go to “Options.” In the Txt Msg field enter your call sign and any optional information (name, location, etc.) This will be sent to the receiving station along with the voice. To test the flatness of the TX radio’s filters, checking “Test Frames” sends data to the receiving station for evaluation using the “Test Frame Error” window. Do not check this for normal DV contacts. FreeDV 700 clipping is enabled by default. The Event Processing section is for use by advanced users and not covered in this documentation. If you wish to work with this option, please refer to the people on the Googlegroups Digital Voice forum for help and advice.

Audio Configuration

Audio Config

From Radio

Device	ID	API	Default Sample Rate
Microsoft Sound Mapper - Input	0	MME	44100
Realtek HD Audio Input	1	MME	44100
Virtual Cable 1	2	MME	44100
Virtual Cable 2	3	MME	44100
Virtual Cable 3	4	MME	44100
Virtual Cable 4	5	MME	44100
Virtual Cable 5	6	MME	44100
Virtual Audio Streaming	7	MME	44100
USB Audio	8	MME	44100
none			

From Xcvt's Speaker

All "devices" must be set to same rate!

Device: Virtual Cable 2 (3) Sample Rate: 48000

To Speaker/Headphones

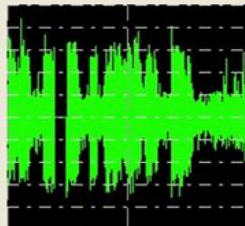

Device	ID	API	Default Sample Rate
Microsoft Sound Mapper - Output	9	MME	44100
Realtek HD Audio output	10	MME	44100
Virtual Cable 1	11	MME	44100
Virtual Cable 2	12	MME	44100
Virtual Cable 3	13	MME	44100
Virtual Cable 4	14	MME	44100
Virtual Cable 5	15	MME	44100
Virtual Audio Streaming	16	MME	44100
USB Audio	17	MME	44100
none			

To Headset Speaker

Device: Microsoft Sound Mapper - Output (9) Sample Rate: 48000

Receive Transmit API Info

Refresh OK Cancel Apply

Receive

Audio Config

From Microphone

Device	ID	API	Default Sample Rate
Microsoft Sound Mapper - Input	0	MME	44100
Realtek HD Audio Input	1	MME	44100
Virtual Cable 1	2	MME	44100
Virtual Cable 2	3	MME	44100
Virtual Cable 3	4	MME	44100
Virtual Cable 4	5	MME	44100
Virtual Cable 5	6	MME	44100
Virtual Audio Streaming	7	MME	44100
USB Audio	8	MME	44100
none			

From Headset MIC

All "devices" must be set to same rate!

Device: USB Audio (8) Sample Rate: 48000

To Radio

Device	ID	API	Default Sample Rate
Microsoft Sound Mapper - Output	9	MME	44100
Realtek HD Audio output	10	MME	44100
Virtual Cable 1	11	MME	44100
Virtual Cable 2	12	MME	44100
Virtual Cable 3	13	MME	44100
Virtual Cable 4	14	MME	44100
Virtual Cable 5	15	MME	44100
Virtual Audio Streaming	16	MME	44100
USB Audio	17	MME	44100
none			

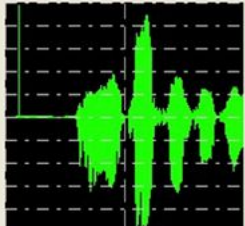
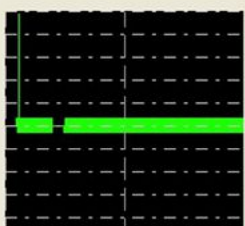
To XCVR MIC

(Set all to 48000 sample rate)

Device: Virtual Cable 1 (11) Sample Rate: 48000

Receive Transmit API Info

Refresh OK Cancel Apply

Transmit

Go to the Tools pull down menu and select “Audio Config.”. Sound cards/USB adapters, VAC and/or USB headsets installed on the PC will be listed. Select the PC’s internal sound card or USB-audio dongle interface for the “From/To Radio” receive and transmit data. Select the USB headset or USB-audio dongle (if using a microphone / speaker rather than a headset) for the “To Speaker/Headphones – From Microphone” voice out and input.

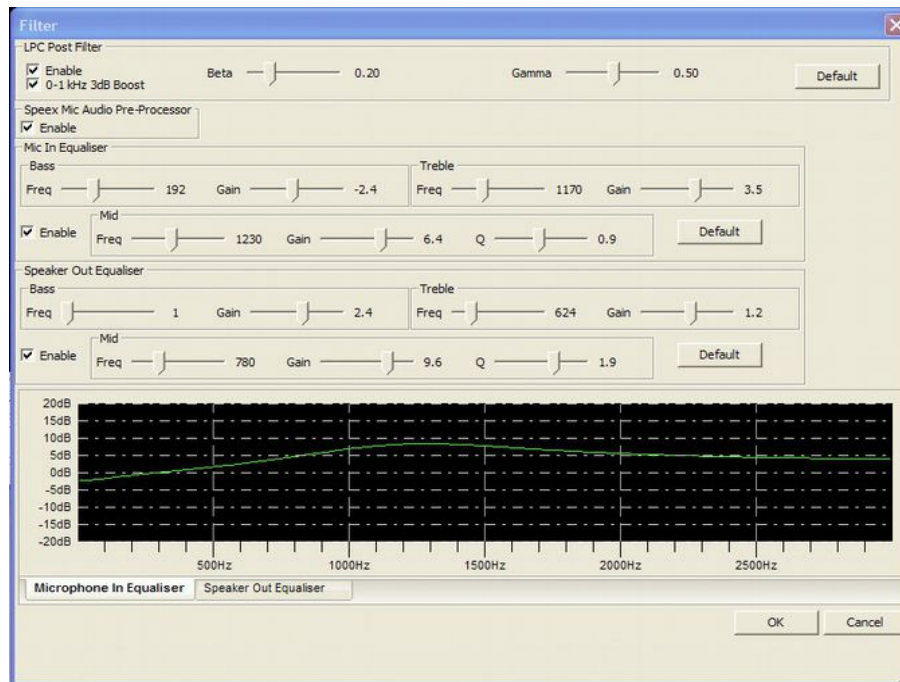
To test the selection, click on the “Rec 2s” tab, look for a 2 second deflection in the scope (from the radio’s speaker output audio or from speaking into the mic). Next, click on the corresponding output device and listen for an 800Hz tone followed by a deflection in the scope. If audio levels have not been previously set for use with another digital mode like PSK31, then they will need adjusting to avoid over or under driving your MIC input of your radio or over-driving the input of the sound card.

See [“Receiving Digital Voice”](#) and [“Transmitting Digital Voice”](#) topics later in this document for these adjustments. The API tab shows Port Audio info.

Note:

Alternative rig connection: If you have a "9600" digital input and output on your radio (usually on a round DIN ACC connector), this is the best connection for every digital mode. Your radio (usually in a menu) should be configured for 9600 or "no pre-emphasis/de-emphasis" if it has this available. If the radio's configuration menu has a 1200/9600 setting, leave it permanently on 9600. If not available, FreeDV will still work fine with Mic/speaker connections. FreeDV must be “stopped” to change Audio and PTT configurations.

Tools - Filter and Equalizers



Under Tools, go to “Filter.” Adjustment of the Equalizer will depend upon the mic your voice characteristics. Testing has found voice quality may be improved with most mics by raising the mid-range frequency and Gain. Similar changes for the Speaker equalization may show improvements also. Adjustment of the LPC Post Filter may be adjusted to improve the subjective speech quality. On-the-air reports and experimentation will be needed to make final adjustments.

Tools - Recording Features

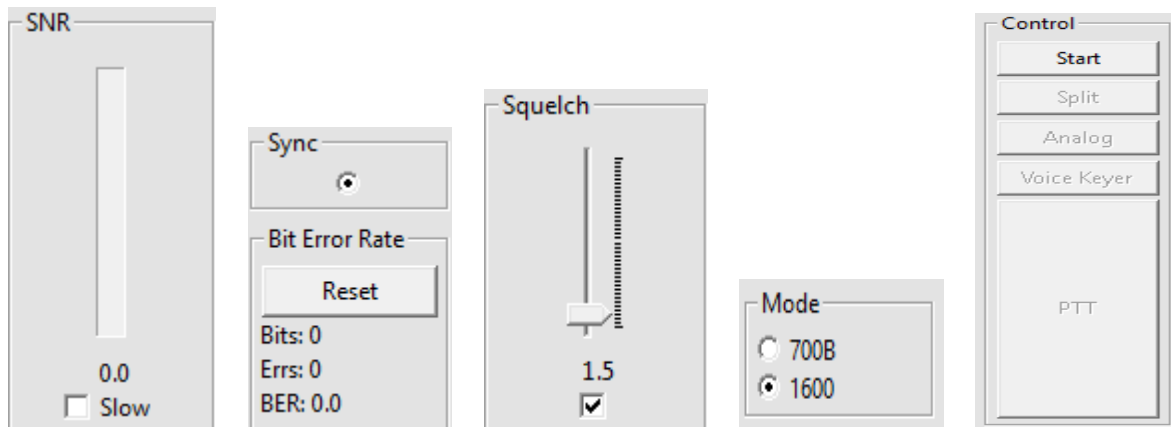
Playing File: k3dcc to Mic Input

Recording File: N4DV From Radio

Playing File: wa6nut into From Radio

Under Tools, options for Recording and Playback are available. Currently, the record/playback files (Mic in) must be sampled at 8 kHz using 16 bit samples. Off the air DV may be recorded and played back through the PC speakers. Recordings that are to be played back into the radio mic input need to be 16 bit 8KHz sampled WAV files, others rates will not work (this applies to the voice keyer input as well). These dialog boxes will appear in the lower left hand corner below the options call sign/name text box while running. The recordings are 30 seconds in length which is the same time as a DV signal requires to move from the bottom to the top of the waterfall display.

Receiving Digital Voice:



SNR

BER

Receive Squelch

Mode

Control

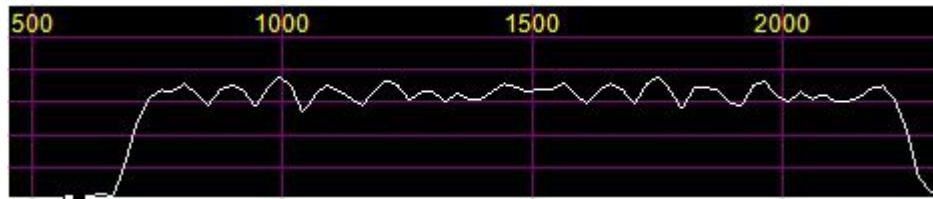
SNR: (Signal to Noise Ratio) is shown both in a visual indicator and numerically. Since SNR estimation will continually fluctuate (sometimes rapidly) over an HF channel, it can be averaged using the “Slow” option. An SNR of approximately 3dB is required to begin decoding voice. This would be a rather low level noisy signal so some dropouts can be expected. Dependent on your receiving location’s noise floor, expect good decoding at S4-5. To judge the quality of the data signal being received, expect an SNR of >15 to be indicated while receiving an S9 signal (again, this depends on your noise floor). Multi-path distortion, QRM and fading have an adverse effect on the SNR. Providing a “SNR Report” is a good way to report the received signal level. The higher the dB value shown, the better.

Sync: The dot appears when the received signal is in sync and can be decoded. Sync is derived from the BPSK (binary phase shift keying) pilot tone signal in the center of the FDM carriers. The small Red Marker at the bottom of the display (defaults at 1500 Hz) should be centered in between the BPSK signal “tracks.” (It can be within +/- 200Hz and still provide sync).

BER: (Bit Error Rate) is an end-to-end performance assessment of the transmitter, receiver and the propagation between them. (This is only available on 1600 mode). BER is the percentage of errors defined by the formula:

$$\text{Bit Error Rate, BER} = \frac{\text{Number of errors}}{\text{Total number of bits sent}}$$

Squelch: When box is checked squelch is activated. The recommended default value is 1.5dB. Experience has found decoding of low SNR signals is possible in multipath/fades and QRM by taking advantage of the fast recovery/sync of FreeDV and your “brain’s” FEC. Experiment with the squelch level in different signal and noise conditions. This setting correlates with the received signal SNR.

**700 Mode****1600 Mode**

Spectrum Occupancy by Mode

Mode: 700x mode works better in marginal conditions at the expense of audio quality. CODEC2 is running at a very low bit rate (650 bit/s) for voice encoding giving it 3dB coding gain. 700B uses no FEC and 7 x 2 identical carriers for frequency diversity across 1.5Khz to combat deep multipath fading

1600 mode (the default) has 300 bits of FEC in CODEC2 allowing up to a 10% BER before failing. This is currently the “de-facto” mode in use. It is not as robust, but has a narrower BW and better voice quality.

Control:

Start/Stop FreeDV must be in “Start” mode to begin receiving. With no FreeDV signal present, the waterfall will display band noise indicating audio is being routed to FreeDV’s input. If no noise is observed, slowly raise the audio level using Windows sound level on the audio channel from the rig until the level is blue in color. A signal will have more amplitude and depending upon its strength, the carriers will change color and become lighter blue to red. Strong interference (SSB) and strong noise (QRN) will be red in color.

Split Allows operation on two different frequencies within the audio pass band and is useful if you find the station you are working does not have their carriers centered on 1500 Hz – think of it like an RIT control on an SSB transceiver. Note: Once the signal is in sync no additional tuning should be made.

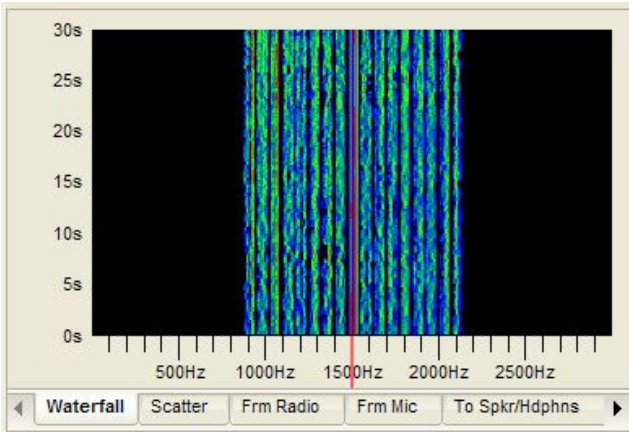
Analog This function switches off digital encoding and decoding so that you can communicate with a station on frequency running standard SSB (i.e. not DV).

Voice Keyer This turns on a preconfigured recording and sends it as many times as you have programmed, with a gap between where it switches to receive and switches back to receive once all requested repeats have been completed. Once a valid signal is received (response to your CQ), the Voice keyer will terminate. One easy way to make a wave file recording uses the *Windows* “Sound Recorder.” Set the recorder to PCM Format and 8.000Khz, 8 bit Mono. With a PC microphone, make the “CQ” recording. *Note:* Loudness/Level is important and may take a couple tries to get the level correct to avoid over driving FreeDV’s “From Mic” level. Save the file on your hard drive and then enter location in Tools>PTT Config Voice Keyer “Wave File.”

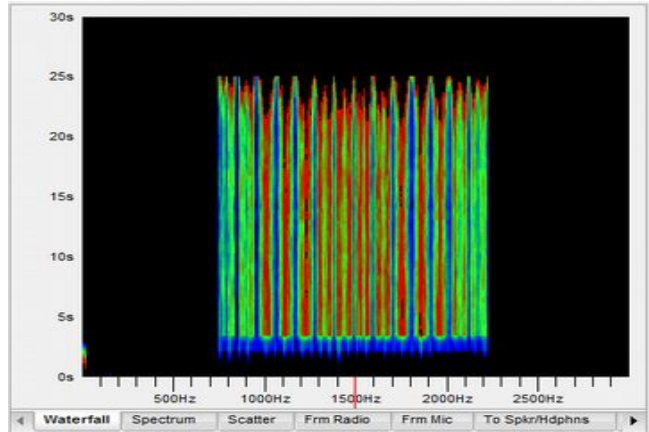
PTT may be activated with either clicking this button or pressing the PC space bar.

Note: The FreeDV GUI must be in focus to control PTT with the space bar.

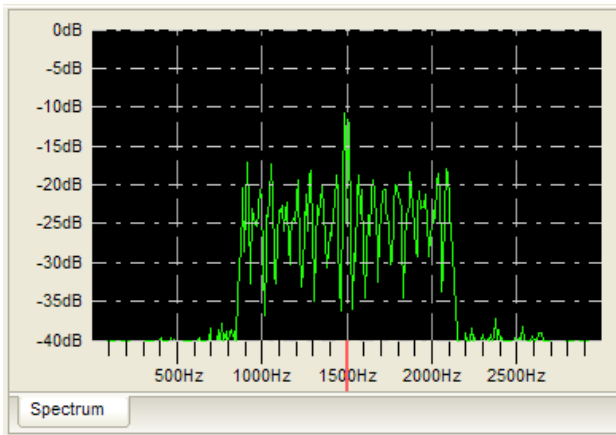
Received signal displays.



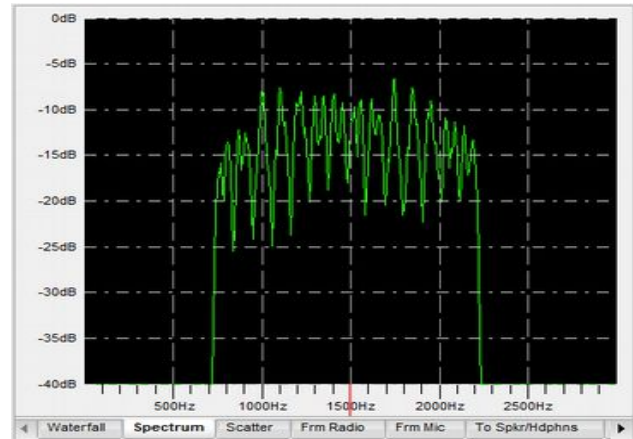
Default 1600 Mode Waterfall
 16 carrier FDM signal with 75Hz carrier spacing
 Red marker showing sync on DQPSK Pilot tone



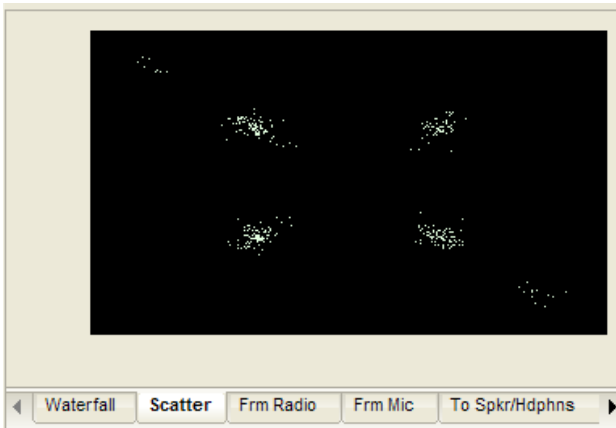
700B Mode Waterfall
 14 carrier FDM signal with 75 Hz carrier spacing
 Change in color (amplitude) indicates fading



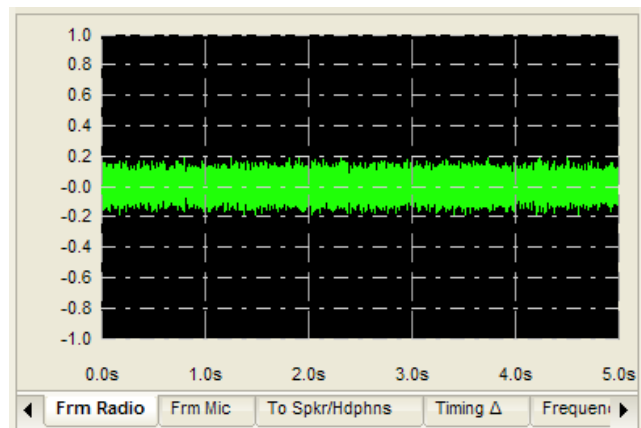
1600 Spectrum
 16 carrier FDM signal with 75Hz carrier spacing
 with sync on center of Pilot tone at 1500Hz



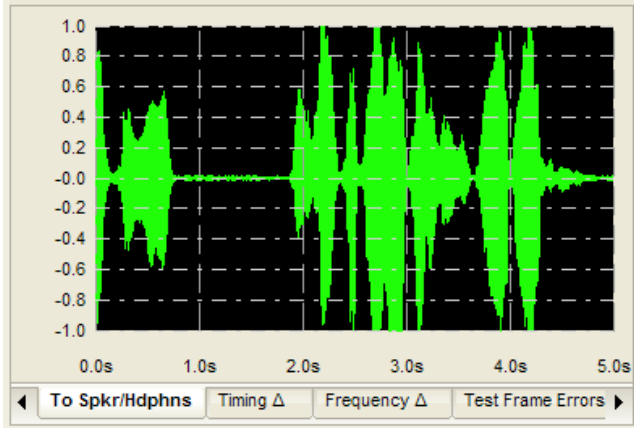
700B Spectrum
 14 carrier FDM signal with 75 Hz carrier spacing
 and frequency diversity to combat fading.



Scatter Plot
 H/V points of the 4 QPSK phases with a "tight cluster" = good signal while fading = X shape

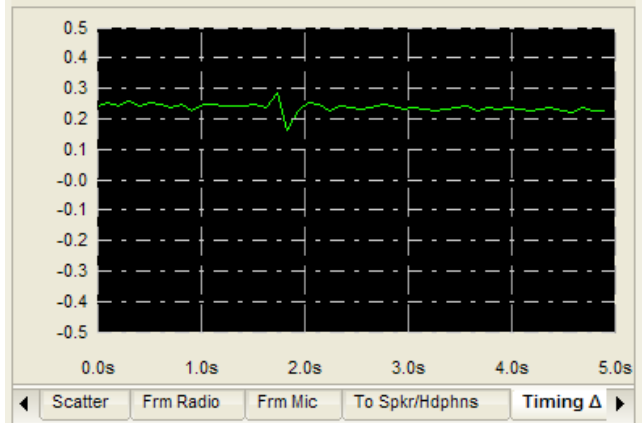


From Radio
 Carrier audio level received at the input of the soundcard.



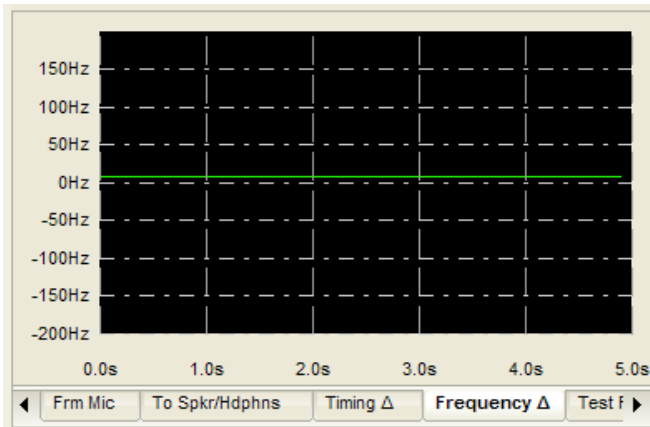
Audio to Speaker/Headphones

Level of audio feeding the USB headset or Speaker. Adjust for max peaks ± 1



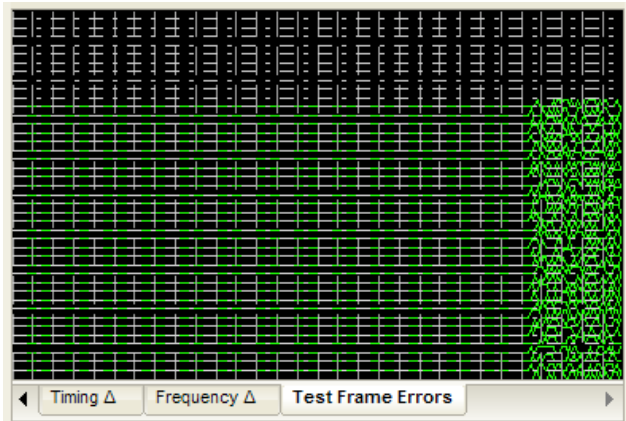
Timing Delta

Plot showing estimate of the best sampling time. Changes indicate difference in TX/RX clock.



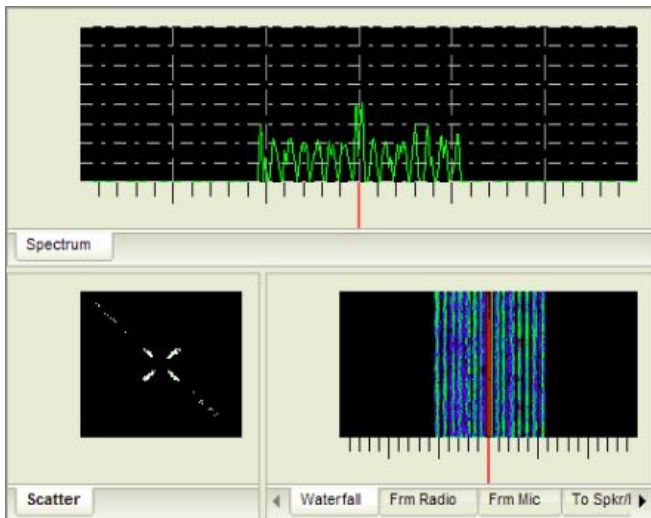
Frequency Delta

Frequency difference between receiving station and transmitting station frequency.

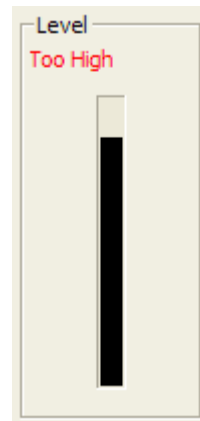


Test Frame Errors

Display of a test frame received. Straight lines are carriers with no errors. No signal on right.



Multiple Signal Displays



Audio Level Indicator

Multiple Signal Displays: Various displays are available to view the received signal. The default display is the waterfall. Click on the tabs below the current display to enable other displays. On transmit; the “Frm Mic” scope display is used for showing input audio from the Mic.

To add a display, left click on the display tab and drag it vertically to the top. Next, release the left click button when a light blue background is showing. Displays can be arranged in various horizontal and vertical positions.

Note: The added displays are not remembered, neither is any re-sizing of the overall program window, when the program is exited, the settings have to be repeated when the program is restarted.

The waterfall color may be changed to monochrome or all blue hues with a *right click* inside the display window. Dark areas (called notches) indicate presence of multipath fading. Under these conditions (which are very common on HF), the spectrum display will show the 14 carriers constantly changing in amplitude.

To obtain sync with the transmitted signal, precise (<200Hz) tuning is required but easily accomplished using the mouse for “click tuning” while observing the Waterfall or Spectrum displays. First, tune the receiver dial to the operating frequency (try 14.236 USB). If a signal is present with an SNR of at least 2-3dB, decoding should start when the Red Marker at the bottom of the display is within the BPSK “tracks” (same waterfall appearance as PSK31) sync signal. If the Red Marker is not centered, then move the mouse cursor to the center and left click. Instantly, the signal should sync and voice will be decoded and sent to the headset or speakers. Once in sync, the radio’s frequency dial should not be moved. FreeDV’s AFC will track small drifts of the signal. Normally, the sync should be centered in the display at 1500 Hz to ensure the FDM signal for all modes is within the Tx/Rx band pass of the transceiver. *Note:* For the 700 mode, there is no center pilot/sync tone. Here, the sync is spread out across all the carriers (7+7 duplicated). The tuning is the same using the RedMarker centered inside signal. The tuning window, however it is only 60Hz wide.

The input level from the radio to FreeDV should be adjusted using the Windows level control so the waterfall FDM carriers are mostly blue to green in color. If the level is set too high, the “Level” bubble will flash a warning. A slightly higher level may be needed for the Spectrum display. Stronger signals in the waterfall will display the carriers in different shades of blue, green, and red. Multipath selective fading results in a various darker “patterns” displayed in the waterfall as the amplitude of the carriers are being reduced.

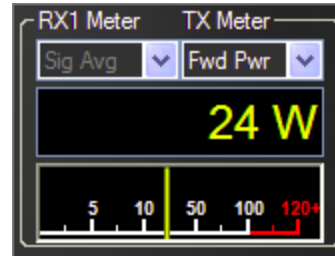
Note: When using a PC interfaced to a normal transceiver, Window’s level “sliders” are used for all level adjustments. For SDR radios with Virtual Audio Cables such as PowerSDR, use the application’s software adjustments (VAC Rx / Tx Gain).

Transmitting Digital Voice:

Power Settings (graphics from Flex SDR):



Keep ALC < 0



For typical 100w radio
run < 25w average power

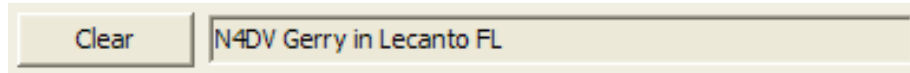


FreeDV Transmit Display (Microphone input level)

Verify the radio is operating at full output on SSB. The final output drive will be set using the sound device selected in Tools>Auto Config>Transmit>To radio. If not started, click on “Start” and then “PTT.” Quickly, adjust the RF output level and then watch the ‘Frm Mic’ display while speaking in a normal voice. Adjust for 75% *maximum* deflection. Any voice input too high will cause clipping. Adjust the RF power for approximately 25w average with a 100w transceiver. This is an important setting to avoid distortion and lowering SNR at the receiving station. If necessary, reduce the drive level so that NO ALC action takes place. With no voice input, the “thin” center baseline on the display *should not show any deflection*. If it does, this indicates unwanted noise is being picked up (i.e. loud fans) within the ham shack or RF getting into the system and under these conditions your audio will sound bad!

Note: The DV signal has a high peak-to-average-power (PAPR) ratio meaning the RF amplifier needs additional “head-room” for the multi-carrier’s higher peak power. Otherwise these peaks will be distorted and cause errors in the data sent and received then passed to the CODEC2 at the receive end for decoding. The peaks are quite fast and a conventional “peak reading” watt meter cannot accurately read the peak power. The average power output should not exceed 20-25% of your SSB transmitter/amplifier. Dependent on the power meter in use, “average” power will show some fluctuation in its readings.

Tip: Call sign/name text data is being sent (repeated continuously during transmit) at a low bit rate (25bit/sec to 50 bit/sec). When initializing a contact, transmit for 10-15 seconds minimum to ensure the receiving station receives your call and location in the text box.



Text Box for Call ID/QTH info

The call sign/name/location text is sent in a continuous loop and displayed in the text box. Use Tools>Options to enter your info for transmitting. The “Clear” button may be used to clear the box. *Note:* Under Tools>Options, Text Encoding box (*Use Checksum on Rx*) may be checked to receive “error free” Text information. However, depending on the SNR, this may have to be received several times before decoded without errors. The status is shown on the GUI (Checksums “Good” - “Bad”) In the future, error free data received here will be used to trigger “Events” and in turn “Process Events” locally and across the ‘net for interaction with web applications such as the FreeDV QSO Finder

Operating Tips:

Where to Begin...

Start by looking for activity on 20m, 14.236 USB. FreeDV's FDM carriers make a distinctive noise but are half the width of SSB. Avoid confusing the digital EasyPal picture transfer OFDM signals on 14.233 with DV. "FreeDV QSO Finder (<http://qso.freedv.org/>) written by John, K7VE is available on-line for the DV community. Current stations on the air, frequencies in use and a chat line are available. Just log on with your call sign (you need to be defined at hamqth.com before you will be allowed access, so set up your details there first). The list of suggested FreeDV frequencies are shown at the bottom of the page. You can also enter your own frequency in the box. All users who have entered a frequency (Frequency Selector) will be displayed on the left hand side of the page. Look here to see where to listen first. A call sign look up using hamqth.com is available for stations logged in. Just click on their call sign. Thank you, John! Currently, approximately 90% of the QSOs are using the default 1600 mode so look for it first. The differences are noticeable in the appearance of the waterfall. This 700"x" mode will continue to see changes to improve the robustness and voice quality so check *FreeDV.org* for updates.

Calling CQ...

Call sign/name text data is being sent (repeated continuously) at a low bit rate (25bit/sec) during transmit. When calling CQ or initializing a contact (calling a station), transmit for 10-15 seconds minimum to ensure the receiving station receives your call and location in the text box. Keep this in mind when making a recording for the Voice Recorder.

RFI and ambient shack noise...

RFI will cause noise and annoying sounds to be transmitted and decoded along with the voice. If experienced, a few ferrites on the interface cables and at the mic input may help remove this annoyance for the listener. This is very important otherwise the voice quality can be seriously degraded.

Voice quality is dependent on the type of microphone but low cost PC "electret" mics can provide good voice audio. Logitech and Altec Lansing HS are known to work fine. The FreeDV equalizer (under tools/filter) can improve the audio quality.

Minimize ambient noise pick up by lowering the mic level as low as possible while speaking in a normal level *close to the mic element*. Loud fans or other non-voice audio in the background may be picked up and transmitted as noise. Microphones in PC Laptops are generally not recommended. To check for noise pickup, watch the center green base line on the Tx display when *not* speaking into the mic. It will not show any deflection if no input "noise" is detected. Under Tools>Filter the Speex Mic Audio Pre-Processor is effective in removing noisy mics and noisy sound cards/USB. Under normal conditions, this option should remain enabled (default).

More information...

Want more information on FreeDV? Check freedv.org often and the links including David Rowe's posts at http://www.rowetel.com/blog/?page_id=452 Like FreeDV? Please consider a donation at freedv.org for the developers.

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