

# Getting Started with Sound Card Digital Modes

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## Introduction

This presentation is about setting your Amateur radio station up to operate the wide range of modern radio software which uses the PC sound card to generate and receive signals. The most basic setting up information and hints are included. You will not need to take notes as a hand-out will be distributed with all the necessary information.

The range of operating modes now available is very wide, and I'll demonstrate some of them later. Here are some of the modes available:

|                            |                                  |
|----------------------------|----------------------------------|
| Morse                      | PSK31                            |
| Hellschreiber              | PSK-Hell and FM-Hell             |
| HF FAX                     | MFSK16, MFSK8                    |
| SSTV                       | MT63                             |
| RTTY                       | PSK63F, PSK125F                  |
| Weather Satellites         | QRSS                             |
| Oscilloscopes, Spectrum    | Time code receivers              |
| Analysers and spectrograms | Beacon monitors                  |
| Conventional FIR filters   | FFT filters and stereo spreaders |

## Requirements

- Pentium 100 PC or better with 16 MB RAM or more, 100 MB free hard disc, SVGA video
- Suitable 16 bit sound card, preferably a SB16 PNP or near clone. Later SB cards such as AWE32 and AWE64 are suitable, but watch drivers
- PC Operating System – Win95, Win98 (preferred), Win NT, Win 2000 (some driver difficulties), WIN ME (not recommended – many sound system differences. Newer operating systems will require special versions of the software, not practical until there are more users. A limited range of software is available for LINUX.
- HF SSB rig capable of running 100% duty cycle with high linearity at the intended power level. Must have 100 Hz step tuning or better, drift < 10 Hz/over and offset of < 10 Hz (preferably < 2 Hz). Most modern rigs are suitable for 25 Watts output, which is plenty for most modes.
- Set of connecting cables – one for transmit, another for receive, and an optional PTT interface if VOX operation is not used. All cables must be electrically isolated – the audio cables with transformers, the PTT cable by an opto-coupler or relay.

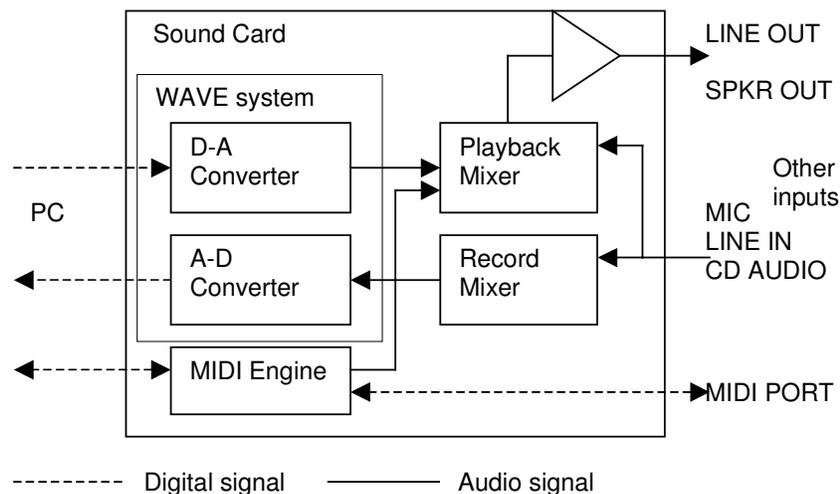
## Block Diagram of PC Sound System

The sound system of the PC is not well documented, and not well understood, but since it is right at the centre of activity for these modes, it is important to understand what goes on.

The PC of course can only handle numbers. It is a superb and very fast number cruncher. Thus, the audio input and output must be handled internally in a form the PC can understand. This is the main role of the sound card – to digitise incoming audio, and turn digital data streams back into audio. There are several different formats that can be handled by the PC, but for all the modes of interest to Amateurs, the wave data format is the one that is invariably used. No doubt you have heard of WAV files and some of the just about unlimited range of sounds that can be made. The WAV format is flexible and convenient, can be stored to disc and played back again, and can be generated mathematically.

The other important function of the sound card is to provide accurate timing. Modern PCs are designed for flexible operation and multi-tasking – running many programs at once, some the user knows about, and several that are hidden in the background. This makes the PC a poor bet for accurate timing in the millisecond range, since you cannot be sure when your program is running. Although it appears they all run at once, they are in fact sharing the processor, each getting small slices of its attention. The sound card is dedicated to the job of sampling the outside world and returning data to the outside world, and it can do this very accurately at a series of fixed rates, such as 8000 samples/sec or 11025 samples/sec.

The operating system and the sound card hardware also provide other features such as the ability to switch from different inputs and outputs and adjust levels. As time goes by these features get better and better, and other facilities are added, but of necessity the software needs to operate on a certain base level of functionality. What is described here is the generic SB16 card, which comes in many forms. It is important to note that the SB16 is *full duplex*, and many older cards were not. Some sound card modes depend on receiving and transmitting at the same time, internally at least, if not on the air.

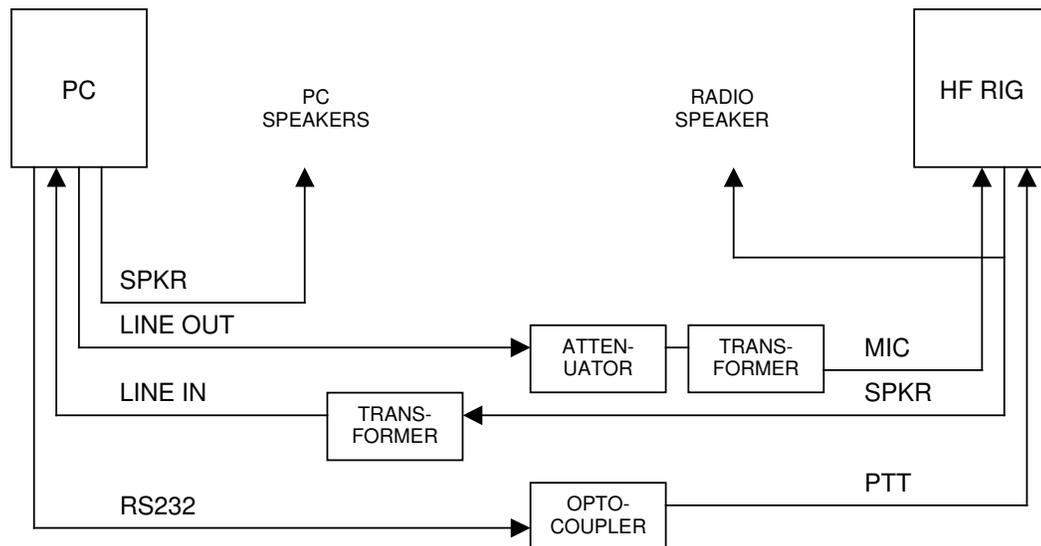


You won't need to know the details of what happens inside the sound card, with one important exception – it is important to understand how the mixers work. As can be seen from the above diagram, there are a number of inputs and outputs which can quite confusingly be used in both record and playback processes. In addition, the operating system provided in the PC is not exactly ideal, either in explaining how to make adjustments, or what software to use to make these adjustments.

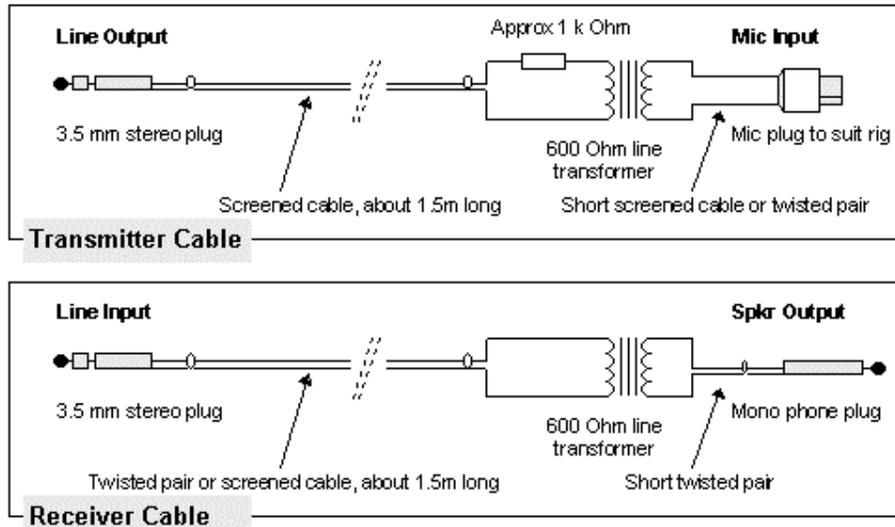
These will be explained later, but just for now you should realise that the RECORD mixer adjusts signals heading into your software, the receive software generally, and the PLAYBACK mixer adjusts signals heading for the rig and speakers, the transmit software and monitoring facilities. Sometimes the software connects the two systems together (DSP filters are a good example), but generally they are independent, so for example, while you have the system set to transmit and receive PSK31, you can't hear the receiver audio via the computer!

## Connecting Up, Cables

This is an area that can cause a lot of grief, but if attention is paid to the details, it is very straightforward. You *can* get started, at least on receive, with nothing more than an off-the-shelf or homebrew 3.5 mm stereo cable, plugged into the rig's speaker socket at one end, and the sound card LINE IN at the other, but I don't recommend it. There are serious risks involved in making DC connections between the rig and the PC. As well as RF going both ways (RF disrupting the computer, computer RFI disrupting the receiver) there is the very serious issue of ground loops. These can lead to high DC power currents from the rig circulating via the audio cable screen and the sound card, and AC hum from the PC getting into the rig, especially if a similar cable is used for transmit.



I have heard several horror stories of rigs and computers damaged, and even ZL1AN had the input circuitry of his new HF rig damaged in this way. The solution is simple – isolate the cables completely. This is simple to do and completely sets your mind at rest. The audio cables are best isolated with transformers – line transformers, old transistor radio interstage transformers, or new ones from an electronics retailer. DSE M0222 at \$4.50 is very small and quite suitable. Jaycar MM-2534 is similar and the price is the same. If you decide not to use VOX (most do use VOX, because it is simple and effective, and the software invariably supports it), use an opto-coupler or small relay to isolate the PTT signal from the PC. Unfortunately there is as yet no definitive standard connection for PTT signals at the PC end, although most software offers a choice.



As can be seen from the drawings, the cables are very simple, and you can easily make them or have a friend make them for you if your soldering skills are not up to it. I have a large bag of 8 pin Kenwood style microphone plugs if you need one. Many rigs have an accessory socket with both microphone input and speaker output on, which is really convenient.

The cables need not be stereo, but use stereo plugs at the PC end and wire the conductor to both channels. Most software uses the left channel, but it doesn't seem to matter.

A simple one-resistor attenuator is shown in the transmitter cable. This will probably not give sufficient attenuation, so connect a 200 Ohm pot across the primary of the transformer, and connect the 1k resistor to the tap. Once you are happy with the level, you could replace the pot with a fixed resistor (in the range 50 – 200 Ohm) and make all subsequent adjustments from the PC.

There are a few things to be aware of with accessory sockets – on some rigs, the TX audio input does not become live until the microphone is unplugged from the front. On other rigs, the microphone stays live even when the audio is sent in the accessory socket, so your muttering, coughing and key-pounding can be heard over the air! Other rigs also have no VOX operation from the accessory socket. All in all, the front panel mic socket, while perhaps not the most convenient input, is much the most reliable.

When setting up the receive cable, it is important to ensure that you can still hear the receiver audio via the rig, as you cannot rely on hearing it via the PC sound system. If you attempt to do this you will either have trouble with the VOX not tripping due to overload of the anti-vox by the RX audio, or the VOX will continually trip on incoming receiver audio. It is best to mute the incoming audio from the PC speaker system and rely on the rig's speaker. If you use an external speaker on the rig, simply connect the RX audio cable and transformer across the speaker.

An excellent alternative to home-made cables is one of the published controller designs, or a commercial rig controller. These units usually include full isolation, PTT, and automatic switching of the microphone, which is especially useful for SSTV, which is the one mode where voice is widely used. I also find the microphone very useful for voice assistance of beginner digi-operators. Perhaps the nicest commercial rig controller is the RIGBLASTER, which costs about US\$130, and comes with a good manual and a disc full of the latest sound card software. The RIGBLASTER is adaptable to just about every model of radio made, and includes clear instructions on how to set the jumpers for your radio.

## Setting up, Sound System Adjustments

Probably the most tricky software to set the audio for is PSK31, and once you have this correct, you won't have trouble with other modes. Unlike RTTY, SSTV or MFSK16, PSK31 (and MT63) require an extremely linear transmitter. This is because the transmission contains more than one tone at a time, and any distortion will cause intermodulation, excessive bandwidth, and poor reception at the other end, winning you no friends. However, setting up isn't difficult, provided you recognise the requirements.

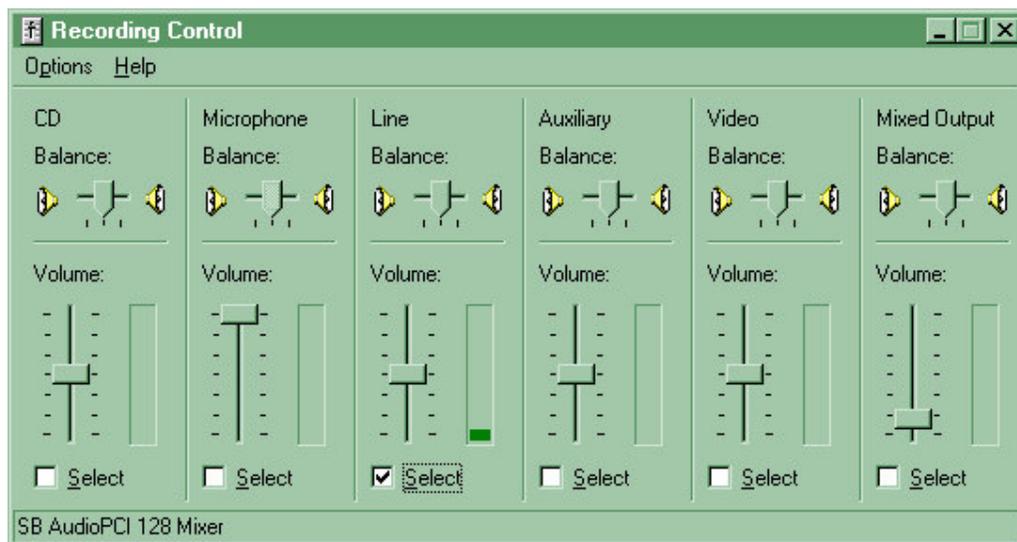
First, tune up your rig as normal, using an unmodulated carrier (CW mode), and note the maximum power output and ALC readings. Make sure the Speech Processor is OFF and it MUST STAY OFF. Set the microphone gain at the normal level used for SSB, for convenience. Transmit from the software, and slowly increase the drive (using the microphone gain and/or the mixer controls), watching the output power increase smoothly. If at any point the output stops increasing, back off. Do not exceed about 50% of the CW mode power output. If any ALC shows, reduce the drive again. For correct adjustment, the ALC MUST NOT OPERATE. This is because the ALC process is a modulation process designed to control voice and carrier signal levels, not digital modes. The PSK31 and MT63 signals contain significant AM components and these will be distorted by the ALC, causing the signal to be broad and difficult to copy.

High power is unimportant. For a start, most digital modes are very sensitive, and low power suffices, and in addition, most rigs will not tolerate continuous operation (which is typical of digital modes) at high power.

Once you have the settings correct, check that the signal is not distorting at the sound card (listen carefully to the PC speakers for high harmonics) and have a friend check your signal over the air. It may be best to drop the sound card output and reduce the attenuator at the mic input. This type of distortion is typically asymmetrical (clipping), and it can also occur in the microphone circuit of the rig. Clipping is a problem with all sound card modes, even RTTY and SSTV, and generates other components inside the SSB passband. For this reason, it is always a good idea to use high frequency tones, as the second harmonics will then be removed by the SSB filter in the transmitter.

Receiver adjustments are very simple – set a comfortable listening level at the receiver, and adjust the record mixer level for reliable operation. The adjustments are very uncritical.

It is important to understand and set the Record and Playback mixers correctly, or none of the software will work. This is the RECORD Mixer panel – used for adjusting the receiver settings:

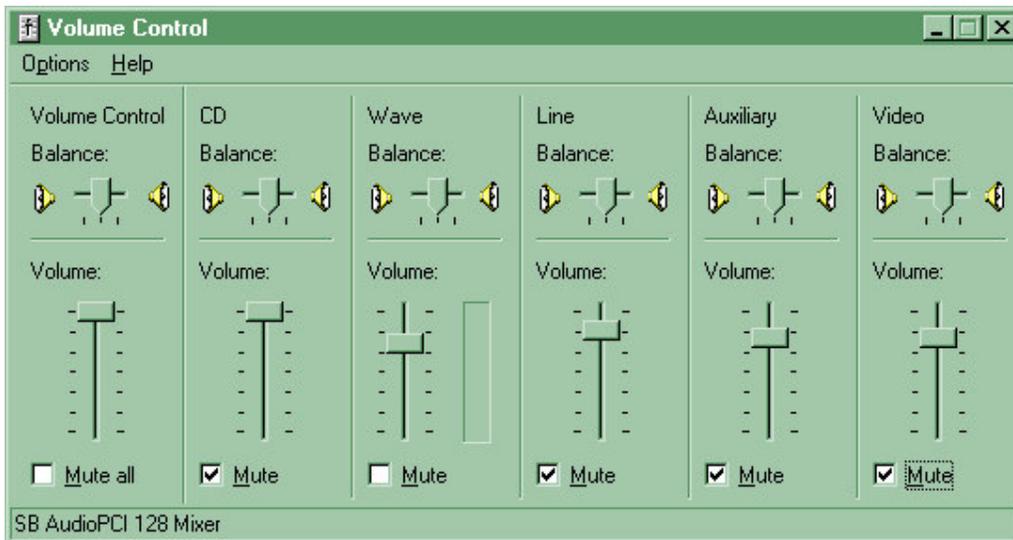


Not all mixers have the same list of inputs, but those shown are typical. Note that LINE is selected. Only one source can be selected at a time. You can use the Microphone input, but it has a preamp, and you will probably need an external attenuator in the cable to reduce the level sufficiently.

You MUST have the correct input (the one you connected the RX output to) selected, or the software will not work. Leave the level control half-way up and the balance centred. You can alter the level later if necessary.

If you use WIN95, you cannot access this mixer panel directly. Even in WIN98 it is hard to find unless the digital mode software does it for you. One guaranteed way is as follows:

1. Right click on the speaker in the system area (bottom right corner of screen) and then click on "Open Volume Controls". This will show the PLAYBACK volume controls.
2. Select Options/Properties and click on the "Recording" radio button. Check in the box below that the input you are using is selected, then press OK. The RECORDING mixer panel will then appear.
3. Make the adjustments as necessary. Leave the Recording mixer panel running so you can refer back to it to make adjustments.



Transmitter adjustments are made on the PLAYBACK mixer panel, ambiguously called the "Volume Controls". If you can, have both record and playback panels on the screen at the same time while making adjustments – you may find that some interact. For example, on some sound cards the WAVE record level and WAVE playback level are coupled together. If your sound card is not directly supported by the WIN95/98 Volume Control applets, your displays and adjustments may differ, but the effect should be the same. If you use different mixer software, and intend to call it from applications, you will need to tell the software which applet to use.

To adjust the transmitter drive (for the procedure mentioned previously) set the WAVE and VOLUME (the one on the left) about half way, and check "Mute" for ALL the other sources, especially the LINE INPUT, or the RX audio will trip the VOX. Make sure WAVE is not muted. Then set the levels at the transmitter, and if necessary change the attenuator on the cable. Finally, adjust the WAVE and VOLUME controls differentially to achieve the cleanest signal without distortion. It may be necessary to adjust the TX cable attenuator again.

A couple of other hints – ensure that system sounds are turned off, or your transmitter will inadvertently transmit noises when you least expect it to. In addition, some computers have speaker beeps redirected via the sound system, and these should also be disabled. Some Amateur sound card programs send status beeps via the speaker in addition to using the

sound system, and can be confused if the speaker beeps steal the sound system. This is because the WAVE section of the sound card can work only with one program at a time, for transmit and receive, or at the very most, one of each.

## Software and Sources

All the software I suggest is free. All of it is available via the internet. Most of it is on the demo computer, and I can easily demonstrate whatever you'd like to see. My recommendations are:

|               |   |
|---------------|---|
| Morse         | CWGET and CWTYPE by Sergei UA9OSV. The very best!               |
| PSK31         | Digipan by Nick UT2UZ, or STREAM by Nino IZ8BLY                 |
| Hellschreiber | IZ8BLY Hellschreiber by Nino IZ8BLY (also PSK-Hell and FM-Hell) |
| HF FAX        | WXSAT by Christian Bock (also for weather satellites)           |
| MFSK          | STREAM by Nino IZ8BLY (MFSK16, MFSK8, PSK63F etc)               |
| SSTV          | MMSSTV by Mako JE3HHT   |
| MT63          | IZ8BLY Terminal by Nino IZ8BLY                                  |
| RTTY          | MMTTY by Mako JE3HHT  |
| QRSS          | Argo by Alberto I2PHD and QRS by Rik ON7YD                      |

There is plenty of other software – see the references below.

| Source                            | URL  |
|-----------------------------------|--|
| Setting up IZ8BLY software        | <a href="http://www.qsl.net/zl1bpu/FUZZY/digital.html">www.qsl.net/zl1bpu/FUZZY/digital.html</a> |
| Digital Modes Glossary            | <a href="http://www.qsl.net/zl1bpu/FUZZY/digital.html">www.qsl.net/zl1bpu/FUZZY/digital.html</a> |
| MMTTY, MMSSTV                     | <a href="http://www.geocities.com/mmhamsoft">www.geocities.com/mmhamsoft</a>                     |
| Beaconsee, Radio Clock, GPS Time  | sapp.telepac.pt/coaa   |
| NL9222's Sound Card Software      | home.wanadoo.nl/~nl9222/software.htm   |
| Oliver Welp's Sound Card Software | <a href="http://www.muenster.de/~welp/sb.htm">www.muenster.de/~welp/sb.htm</a>                   |
| PSK31 Web Site                    | aintel.bi.ehu.es/psk31.htm   |
| PSK Warbler (80m)                 | <a href="http://www.njgrp.org/warbler">www.njgrp.org/warbler</a>                                 |
| IZ8BLY Software                   | iz8bly.sysonline.it/   |
| Fuzzy Modes, MT63, MFSK16         | www.qsl.net/zl1bpu   |