# Going Beyond 9600 Baud

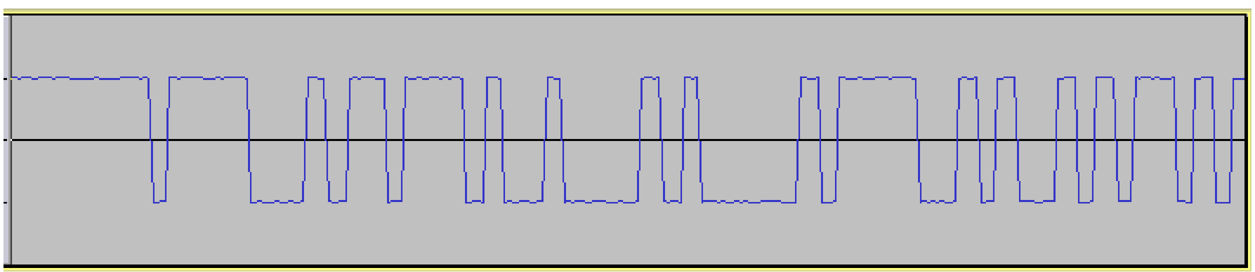
First rough draft - March 26, 2016

Improved PLL results – May 27, 2016

Recently there have been an increasing number of queries about making Dire Wolf handle data rates higher than 9600 baud.

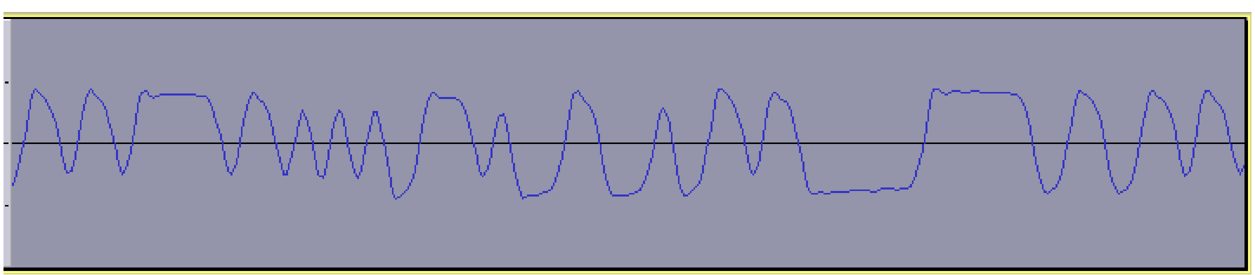
> *Note: The terms **baud** and **bits per second (bps)** are often used interchangeably. In this case they are the same number so it doesn't matter. When using more advanced modulation techniques, such as PSK, there is a difference and we need to be more careful about using the proper term. That is a subject for another place and another time.*

First, let's take a step back and review what a 9600 baud signal looks like. The ideal signal would look something like this with two voltage levels for the binary signal. Each pulse width is some integer multiple of 1/9600 second.



On the receiving end, we use a digital phase locked loop (DPLL), synchronized to the zero crossings, to determine where to sample for the data bits.

In the real world, signals end up looking like this after the bandpass limits of the transmitter and receiver.



The amplitude is not that important because we slice at the zero level to get a binary value. The jitter of the zero crossing points is very important because this is where we get our timing. If the sample rate is too low, we will get more jitter.

Dire Wolf version 1.3 and earlier placed artificial restrictions at 9600 bps for the data rate and 48000 Hz for the audio sample rate. Higher end “soundcards” can handle 96 kHz and sometimes 192 kHz. These limits, in software, have been lifted so we can start experimenting with higher rates. Of course, higher rates mean more CPU power is required.

My gut feeling is that that will need to increase the audio sample rate to get an adequate measure of the shorter pulse widths. Let’s do some experiments…

Eventually we will need to put radios in the middle, but we can gain some insights from simulation. “gen_packets” takes the place of the transmitter. “atest” is the receiver. There are no antennas, just an audio file between them. A typical test would look something like this:

```
$ gen_packets -r 44100 -B 9600 -n 100 -o test.wav
Audio sample rate set to 44100 samples / second.
Data rate set to 9600 bits / second.
Using scrambled baseband signal rather than AFSK.
Output file set to test.wav
built in message...

$ atest -B 9600 test.wav | grep "packets decoded in"
59 packets decoded in 0 seconds.
```

This generates 100 packets with increasing levels of random noise. In this case we see that 59 of them were decoded successfully. Let’s try this with other values and see what happens.

To play along at home, you will need the latest version from the “dev” branch. If you already have a git clone, do this:

```
git pull
git checkout dev
make clean
make
sudo make install
```

Initial observation, March 2016:

| Data Rate, bps (**-B** option) | Audio Samples per second (gen_packets **–r** option) | | | |
|---|---|---|---|---|
| | **44100** | **48000** | **96000** | **192000** |
| **9600** | 59 | 62 | 96 | |
| **14400** | 39 | 40 | 76 | 100 |
| **19200** | 12 | 26 | 64 | 96 |
| **28800** | 0 | 14 | 42 | 76 |
| **38400** | 0 | 0 | 25 | 64 |

We see a clear pattern here. Much better results with higher audio sample rates.

It looks like we want the audio sample rate to be at least 5 times the data rate to get good results. 10 times provides a large improvement.

This is not the final word on the subject, rather the starting point in our journey. There is some opportunity for fine tuning. On the transmit side we have a low pass filter to limit the bandwidth. On the receive end we have another low pass filter. We might be able to improve results by tweaking them. We can also adjust the DPLL inertia for less sensitivity to jitter.

Update May 2016:

Let's try the same test again after improving the data clock recovery PLL. gen_packets was temporarily adjusted to have a lower than normal noise level so don't compare the numbers directly with the above. The two tables below can be compared because the only difference is the improved clock recovery strategy which allows a lower sample rate to data rate ratio.

Old style PLL:

| Data Rate, bps (**-B** option) | Audio Samples per second (gen_packets **–r** option) | | | | |
|---|---|---|---|---|---|
| | **22050** | **44100** | **48000** | **96000** | **192000** |
| **9600** | 33 | 75 | 83 | 99 | |
| **14400** | 0 | 50 | 57 | 96 | 100 |
| **19200** | 0 | 33 | 33 | 82 | 99 |
| **28800** | 0 | 0 | 0 | 61 | 96 |
| **38400** | 0 | 0 | 0 | 31 | 82 |

New style PLL:

| Data Rate, bps (**-B** option) | Audio Samples per second (gen_packets **–r** option) | | | | |
|---|---|---|---|---|---|
| | **22050** | **44100** | **48000** | **96000** | **192000** |
| **9600** | 45 | 88 | 91 | 99 | |
| **14400** | 0 | 66 | 69 | 98 | 100 |
| **19200** | 0 | 45 | 66 | 88 | 99 |
| **28800** | 0 | 0 | 0 | 64 | 98 |
| **38400** | 0 | 0 | 0 | 67 | 88 |

For a **sample rate** to **data rate** ratio of

- 6.7 (turquoise) we are at the point of diminishing returns. Negligible improvement by increasing the sample rate.
- 5 (yellow) we went up from 82-83 to 88-91. Significant but not dramatic. This would be the lower end for best results.
- 2.5 (pink) we went up from 31-33 to 66-67. The number was doubled! Still you would want to avoid a ratio this low.
- 2.3 (green) we went from 33 to 45. Big improvement but still pretty bad.

Lessons learned:

- Don’t use 22k sample rate.
- Increasing sample rate from 44.1k to 48k can provide some improvement. (Note that the cheap USB audio adapters can usually handle these two sample rates.)
- Strive for a ratio of 5 or more between the audio sample rate and the data rate.

Of course, the big unknown is what the radios will do in the middle. If we optimize for best results in the simulated environment, we might make things worse in the real world.

## How do we know what sample rates are supported by the hardware?

Sometimes having device drivers hide the physical reality is not a good thing. Consider this example.

What audio devices do we have?

```
john@linux64:~$ arecord -l
**** List of CAPTURE Hardware Devices ****
card 0: Intel [HDA Intel], device 0: AD1984 Analog [AD1984 Analog]
  Subdevices: 1/1
  Subdevice #0: subdevice #0
card 0: Intel [HDA Intel], device 2: AD1984 Alt Analog [AD1984 Alt
Analog]
  Subdevices: 1/1
  Subdevice #0: subdevice #0
card 1: Device [C-Media USB Audio Device], device 0: USB Audio [USB
Audio]
  Subdevices: 1/1
  Subdevice #0: subdevice #0
```

"Card 0" is apparently this chip, [http://www.analog.com/media/en/technical-documentation/obsolete-data-sheets/AD1984.pdf](http://www.analog.com/media/en/technical-documentation/obsolete-data-sheets/AD1984.pdf) on the motherboard, which supports sample rates of 8, 11.025, 16, 22.05, 32, 44.1, 48, 88.2, 96, 176.4, and 192 kHz.

"Card 1" is a cheap C-Media USB-Audio adapter. When we read the spec sheet for the chip, we see that is it physically capable of only 44100 and 48000 samples per second. That's all.

Here we are asking card 1 for a sample rate of 192k, four times the physical limit, and the request is honored.

```
john@linux64:~$ direwolf -B 19200 -r 192000
Dire Wolf DEVELOPMENT version 1.4 A (Mar 26 2016)
Includes optional support for:  gpsd

Reading config file direwolf.conf
Audio device for both receive and transmit: plughw:1,0  (channel 0)
Channel 0: 19200 baud, K9NG/G3RUH, +, 192000 sample rate x 2.
The ratio of audio samples per sec (192000) to data rate in baud
(19200) is 10.0
This is a suitable ratio for good performance.
Note: PTT not configured for channel 0. (Ignore this if using VOX.)
Ready to accept AGW client application 0 on port 8000 ...
Use -p command line option to enable KISS pseudo terminal.
Ready to accept KISS client application on port 8001 ...
```

I suspect that the driver is performing a rate conversion and returning 4 samples to the application for every one sample from the hardware. This doesn't help us any. We need a way to find out about the physical capabilities of the hardware and make sure that the drivers are not hiding that reality.

Here is something interesting.  It's perfectly happy with any crazy sample rate.  Let's try 54321.   It works!

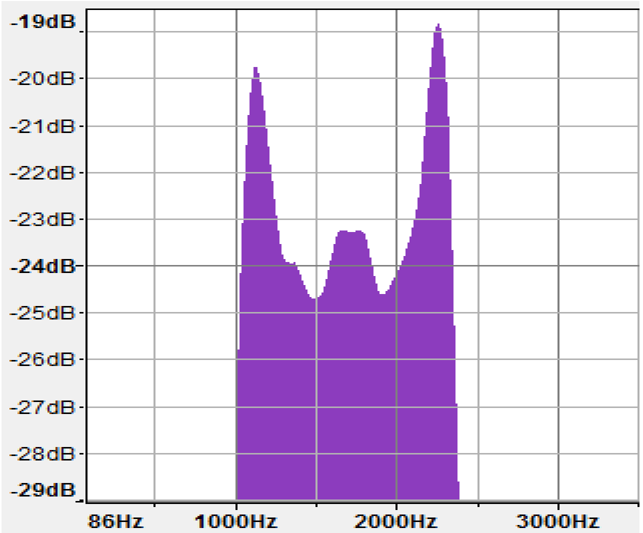
```
john@linux64:~$ direwolf  -r 54321
Dire Wolf DEVELOPMENT version 1.4 A (Mar 26 2016)
Includes optional support for:  gpsd

Reading config file direwolf.conf
Audio device for both receive and transmit: plughw:1,0  (channel 0)
Channel 0: 1200 baud, AFSK 1200 & 2200 Hz, E+, 54321 sample rate.
Note: PTT not configured for channel 0. (Ignore this if using VOX.)
Ready to accept AGW client application 0 on port 8000 ...
Ready to accept KISS client application on port 8001 ...
Use -p command line option to enable KISS pseudo terminal.

Digipeater W1MRA audio level = 1(0/0)   [NONE]   ||||||___
[0.2] N1OHZ>T2QT2T,W1MRA*,WIDE2-1:'cN]l <0x1c>-/
MIC-E, House, Unknown manufacturer, In Service
N 42 14.2400, W 071 50.6500, 0 MPH

Digipeater WIDE2 (probably UNCAN) audio level = 2(0/0)   [NONE]
_||||:___
[0.3] N1OHZ>T2QT2T,W1MRA,UNCAN,WIDE2*:'cN]l <0x1c>-/
MIC-E, House, Unknown manufacturer, In Service
N 42 14.2400, W 071 50.6500, 0 MPH
```

The application is getting 54321 samples per second.  The hardware is capable of only 44100 and 48000.

So, there is our next challenge.  How do we find out the actual hardware capabilities?  How do we make sure that the driver isn't deceiving us?   What bandpass restrictions are imposed by the analog amplification stages?

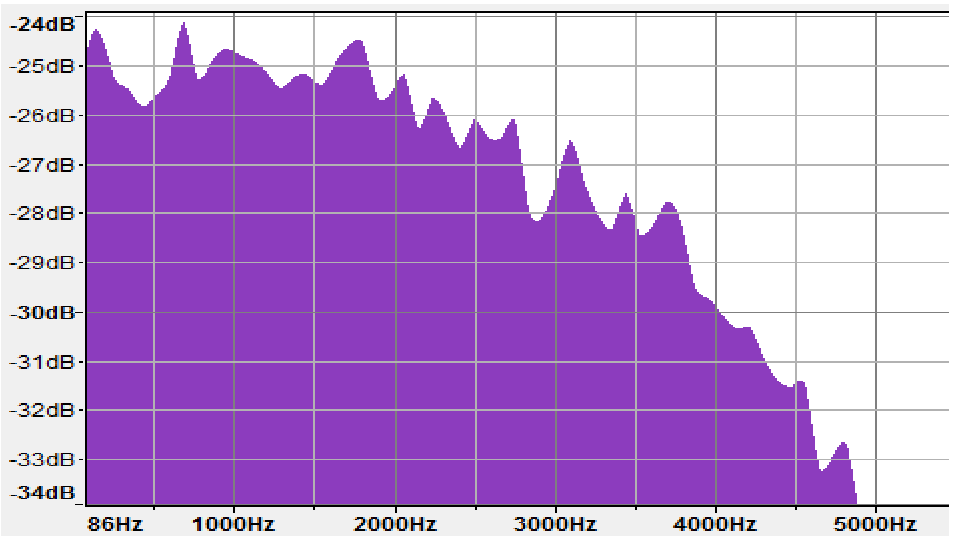
## Radio bandwidth

The spectrum for a 1200 baud AFSK signal looks like this:




This easily fits into the normal voice range and is not fussy about the radio audio characteristics.

The spectrum for a 9600 baud signal looks like this:




Note the peak at 4800 Hz. This corresponds to the maximum frequency of alternating 0 and 1 bits. The upper end of the audio passband needs to extend to at least 5000 Hz.

If you want to use 19.2k baud, double that. For 38.4k, double it again.

Circuits in the radio and your “soundcard” must have a bandwidth of at least half the baud rate.

It’s not as obvious how low we need to go.  There is a peak around 200 so I suspect that a high pass filter, intended  to keep CTCSS frequencies away from the speaker, would cause issues.

**IT WON’T WORK WITH THE MICROPHONE AND SPEAKER CONNECTIONS!**

The pre-emphasis and de-emphasis will distort the signal and make it completely unusable.

If your radio doesn’t have a suitable “data” connector, which bypasses the normal audio processing, you will need to do some surgery.

It probably won’t work if you have audio transformers in the middle.

## The next step on our journey

Questions:

1. How can we determine which sample rates are supported by the hardware?
2. How can we verify that we are receiving the native sample rate and the device driver is not performing a rate conversion?
3. How can we measure the bandpass characteristics of the soundcard?

Ideally: We would like a little application to assess the hardware capabilities. The user would connect a cable from the audio output to the audio input. The application would:

- Determine what native sample rates are available.
- Run a bandpass test for each of them.
- Plot the results.
- Provide some commentary on how it might perform for higher speeds.