**Tips for improved image reception**

If copying a black and white image;
Place a fixed notch (if available) midway between the tones, this can increase signal to noise.

Run some slight DSP noise reduction if available, this can help clean up the product.

When copying a BW image, make sure to not use greyscale mode.

If copying a black and white or greyscale image;
Make sure to form a filter just a bit beyond the tones, if you can form the filters.

If using audio to feed the decoder, make sure you don't overdrive the decoder, nor starve it for signal. Keep in mind that running audio inputs/mixer gain at max isn't always the best idea.

Try various AGC settings to see what helps or hurts production. In noisy conditions like static crashes, you want the AGC action to quickly recover the signal so fast AGC might be best. When the signal is above the noise med or slow AGC will be better. If fast fading is taking place, try them all to see which is best suited, but my guess is fast will be best.

Some decoders, both hardware and software, have options to choose either an am or FM demodulator. FM demod might be more effective because FM rejects amplitude noise, but FM might bring a higher dropout level, if you have the option, try them both and see what works best.

Software decoders often have demodulation options as above, and some offer DSP or MS (Mark and Space) as well. WCODE offers these options, for example. For sigs where you have only two tones such as BW fax or BDSK, MS usually will result in better copy, but DSP is still an option to try.

Some decoders, such as the previously mentioned WCODE, offer user adjustable bandpass filtration. This is very nice as it helps the decoder focus only on the desired signal and not band noise. Don't be tempted to only pass the peaks of each tone, as that forms a filter whose swr goes sky high at the filter edges and more or less increases noise at those peaks - this is called "ringing".
[https://en.wikipedia.org/wiki/Ringing\_(signal](https://en.wikipedia.org/wiki/Ringing_%28signal))
<https://en.wikipedia.org/wiki/Ringing_artifacts>
<https://dsp.stackexchange.com/questions/2170/why-do-i-see-ringing-in-the-output-of-a-digital-filter-with-a-narrow-transition>

In analog filters, ringing is due the SWR effect, SWR is how filters work - low SWR in the passband, increasing SWR at the high and low cutoffs, without SWR there'd be no filtering going on. So that explains why there's ringing in analog filters, but doesn't explain DSP filters very well as they have no analog components hence no SWR. The issue in DSP filters is, in my theory, due to the DSP equivalent to SWR, poles and zeros. And poles have an interesting thing called infinite gain.
We all know there can't really be infinite gain, right?
Anyway here's more on the subject, and I know you'll be up late at night studying this mystery;
<http://web.mit.edu/2.14/www/Handouts/PoleZero.pdf>
<https://www.dsprelated.com/freebooks/filters/Pole_Zero_Analysis_I.html>
<https://en.wikipedia.org/wiki/Pole%E2%80%93zero_plot>

Here's an example of the above in practice;
The Icom IC-756Pro II sitting before me has an internal Baudot decoder as well as a screen on which to display the output. The Icom IC-756Pro II, as well as most other Icom DSP rigs, has both analog and DSP filtration. I was monitoring an RTTY contest on the HAM bands a few months back with bespoke Pro II and had the filters set to just pass the 170Hz BFSK tones, copy was ok but there were random characters being generated on band noise, which due the filter skirts being at the mark and space tones, noise aka energy was increased at those frequencies. As the decoder doesn't know noise from signal, it responds on any noise that breaks the threshold level for any given signal.

The FSK Peak filter, an otherwise excellent feature, increased the noise even more, making for even more extraneous characters being printed.

The solution was to open the filter to 500Hz, where the SWR created noise peaks were far removed from the decoder tones, and perfect copy resulted, I could even reduce the threshold for decode for weak signal work. Another issue is that analog filters can have varying group delay, meaning the high pass end of the filter may slow the signal down while the lowpass end may slow the signal down even more. The human ear can tolerate huge amounts of group delay and still extract intelligence from a given signal, but digital decoders expect group delay to be equal across the passband. Good thing fax decoders simply display what they're fed, huh.

You can see this with your own eyes if you've a decoder with a "tuning eye" or "constellation eye pattern display". You'll see distinct noise peaks that mimic an actual signal displayed in the tuning eye, opening the filter so these noise peaks are moved outside the decoder MS tones will result in much better copy. I know while this example is for BFSK Baudot, it applies to fax and more or less any other mode regardless.
Filters have a side effect of make everything "look" like a desired signal, noise or not. Also, they don't care.

Well, not only do you have group delay and SWR issues in analog filters you also have something completely out of anyone's control and unrelated to filtering - multipath.
<https://en.wikipedia.org/wiki/Multipath_propagation>
Our friend multipath is always with us, from minimal to extremes of 10ms or more, and this has an effect on the goal of clean fax production. Humans deal with multipath easily enough say in the case of an am signal, you get fade or peaks and continue listening with minor annoyance, but digital decoders do not like multipath at all - they demand zero group delay (and no bit smearing aka multipath) in their input signals.

Now back to fax production.

Image slant can be reduced by adjusting sound card timing or by the decode app itself. It's best to adjust sound card timing as that then corrects the timing for all other modes in that decoder too.
Adjusting sound card timing is time consuming and can be a pain but the payoff is worth it. Many decoder apps use fax mode to help adjust the timing, just keep tweaking the timing until the image is perfectly aligned with the border and then save that timing. Most apps have detailed instructions on how best to do this, some apps even include test tones to assist.

Using the app to adjust slant is ok too, some call it phasing or some such but it all does the same job. The prob is you'll have to do it for each fax signal more or less when you could have spent a little time adjusting the clock for the app itself, which will benefit most every other decoder mode, especially the PSK modes.

If your system sound card, onboard or not, is of the hi def variety, there should be no error, and if you've adjusted the clock timing prior to installing hi def audio, save the setting offline and then reset the clock to 1.000.
 If there is error even with hi def, try to reduce it with clock correction. Sadly, each app will usually require its own correction factor, but the first clock correction attempt should get you an idea as to which way to go timewise in the subsequent apps.

The following url is a link to some helpful hints on better fax production.
<https://www.sdrplay.com/community/viewtopic.php?f=5&t=3861>
The post is in reference to the SDRPlay RSP SDR but the fix found on page two of the thread should be of help to everyone who suffers from degraded images on otherwise strong, clear, and stable sigs. After making the vac line the playback device and setting it in properties to 16bit 48KHz sample rate (sound card max sample rate in my system) the borders are sharp and clean, no more frittering. What I've discovered if you use vac to pipe the output of your receiver to a decoder, make sure everything is agreed on such as sample rates and bit depths. If you have one input using 44KHz and playback using 48KHz there's a translation going on that shouldn't be there that will cause timing issues.

On the use of vac and other sound handling routines;
I found vac to be very hit and miss as to when and what apps actually saw the vac virtual soundcard, as well as getting settings in its control panel to take and stick.
Not only that, for what ever reason, my decoders didn't show much if any benefit from going from a literal soundcard to a conjectural one. All digital modes should benefit from having the purest input possible and a conjectural soundcard is supposed to do just this.
I also had trouble setting playback and input devices as well as other irritating issues with vac, and read there was another free to use virtual soundcard; VB cable, so was more than willing to try it
Got it and got it going, very simple, only needed me to chase down the playback bitrate and depth once, and the WCODE decoder is now working better than ever before. So much so that the decoders that bested WCODE prior to VB are now left in the dust with WCODE hearing and decoding sigs the other decoders didn't. The tables have turned.

Regarding synch loss, I found that if I changed focus during reception of some faxes, say closing a browser or opening a new tab and loading a page in the same browser, synch was lost on the fax regardless of signal strength. So if your system is like mine, don't be messing around with opening and closing a bunch of windows when faxing. Try it for yourself to see if your system causes synch loss when opening and closing apps or browsing.