Transmitter Bandwidth and Splatter

Technical Comments about Transmitter Intermodulation, Distortion, and ESSB (or Hi Fi Wide Fi SSB) audio or over equalization ***(Tnx W8JI.com)***

Bandwidth
The most common cause of excessive width in a SSB or ESSB signal is not just the bandwidth or audio fidelity applied to the microphone input or filter bandwidth, but includes several factors. Let's look at some of the common causes of excess bandwidth.

When checking for splatter or reporting splatter, we should be sure:

*We know **our receiver is** good*
We are not using a noise blanker
*We actually understand what **normal bandwidth is***
We do not have excessive receiver IM Distortion

Distortion products are a major cause of excessive bandwidth. **Many forms of distortion products causing splatter cannot be heard when listening closely on frequency, and cannot be detected with a oscilloscope displaying the envelope!** The splatter (excessive bandwidth) we observe most commonly comes from ***odd-order products***.

Note: ***Some ESSB or High Fidelity good audio proponents think the bandwidth of filters sets the bandwidth of their signal. They also think they can hear the distortion that causes splatter, and if they "sound clean" on frequency to friends they have no splatter bothering off-frequency neighbors. We will see why this isn't true as we go through this article.***

Mixing between various tones in the RF signal caused by even the slightest amount of non-linearity (just as in audio systems) will generate new frequencies. With RF, something is different. Odd-order mixing in the SSB RF system creates new "off-channel" distortion products. By saying "off-channel", I mean outside the passband of the filters of a receiver tuned to your operating frequency.

The above distinction is very important. I don't want to be offensive to anyone, but the fact is many "audio experts" claim they or their friends would know they have splatter. They base this claim on the fact they listen closely to audio quality through good receivers, and would notice splatter-causing distortion on the desired signal.

The cold hard fact is we actually can't hear troublesome odd-order IM products listening ***ON FREQUENCY*** even when they are at offensive levels no matter how "golden" our ears, and we probably cannot detect non-linearity that causes low-level IM distortion with a scope. A fraction of a percent non-linearity can create noticeable harmful adjacent channel QRM. No one could even hear distortion 30 dB down from desired audio unless they were listening above or below the other station's main signal. IM 30dB down can be devastating to QSO's on the adjacent channel up or down.

Again, there's nothing wrong with wanting to listen to high fidelity audio as long as we don't take up three or more normal communications channels on a crowded band to do it.

Remember these rules of thumb for IM3 bandwidth:

The maximum frequency spacing of new intermodulation products is the difference between the lowest and highest pitched tones modulating the transmitter.

The total bandwidth occupied by a SSB signal, when we include IM3 products, is approximately ***three times*** the audio bandwidth of the system.

What is Odd-Order Intermodulation?

In SSB systems, the ***RF harmonic*** of an RF carrier created by one audio pitch or tone can mix with the ***third harmonic*** of another RF carrier caused by *another* modulating pitch or tone. The action or mechanism of this mixing is much like the mixing in an intentional frequency conversion scheme. As a matter of fact, all of our regular receivers have a stage intentionally driven into non-linearity (saturation) by a local oscillator. A controlled local oscillator frequency is combined in a mixer to produce a sum and difference frequency with any signal frequencies. We don't hear the distortion caused by this non-linear mixer because it primarily develops out-of-band or off-frequency mixing products that are filtered out or rejected by tuned circuits. A well-designed high dynamic range mixer will actually terminate these unwanted frequencies in a dummy load, so they do not reflect back into the mixer!

Our SSB rigs include a multitude of amplifier stages conducting much less than 360 degrees (as well as some mediocre mixers or frequency converters). Most stages are class AB. The full sine wave isn't amplified, only a portion of it.

Linearity

The most important factor is linearity, which can be expressed as a transfer function of input vs. output level. If transfer function was perfect, an X percent input change would produce an identical output power percentage change. In that case IM would not be much of an issue.

Unfortunately manufacturers and designers compromise linearity in transmitters to reduce expense, size, heat, and power requirements. Transmitter IMD performance has taken a backseat, and we have come to accept poor IMD performance as a way of life. If you have any doubt about this, look at equipment test reviews published in QST or manufacturer's specs. Most newer radios have odd-order transmitter IMD products in the 30 dB below PEP range. (Numbers like that would be considered terrible in receivers.)

My old Collins KWM-2 measured -47dB below PEP for IM3, a new IC-756 I tested was -30dB. The TS-870, often used by ESSB proponents, has IM3 as low as -20dB PEP on some bands. That's about 400 times more power in the off-channel distortion power levels in the 870S compared to an old 1950's Collins rig! Most interesting is a TS870S sounds great on frequency. Off-frequency is where we notice the distortion. I can detect spits and splats as far as 20kHz away from normal modest strength SSB stations when they use some of the poorer modern transceivers and NORMAL audio bandpass.

How odd-IM Frequencies are Created

Amplitude linearity describes how closely the input to output transfer response (gain) of an amplifier (or mixer) resembles a straight line. When an amplifier's input level increases by a certain percentage, its output level must increase by the same percentage, otherwise distortion is produced. The deviation from a straight-line can be represented by a power series. When a single carrier input signal is substituted into the above expression, the output waveform will contain the original carrier and harmonic distortion products. For most communications applications, (with bandwidths of less than an octave), harmonics can be eliminated by filtering.

SSB is different. We can consider each speech "tone" a carrier that changes amplitude and frequency with our voice. In other words the SSB generator in our rigs simply upconverts baseband audio applied to the microphone input to radio frequencies. When more than one audio input tone is present, more than one RF output "carrier" varying in frequency and level is present. Beat products are produced in the vicinity of these RF output "carriers". The new signals are known as intermodulation distortion (IMD) products. ***They are located at frequency intervals equal to the separations of the desired carriers.*** Filtering cannot eliminate IMD products, because the IMD is located on the same frequency, or nearby the desired output signals.

As two or more pure RF "tones" or carriers from our voices pass through a less-than-perfectly linear stage or component, harmonics are generated. It doesn't matter if devices are push-pull or single-ended, it doesn't matter if we can't hear the distortion, it is always there to some extent. Various harmonics created, even though greatly attenuated, mix with the fundamental desired tones and other harmonics of those tones. Most undesired products fall far outside the band we are using and are easily cleaned up. Unfortunately some products fall in band, just outside the desired occupied bandwidth. These are the odd-order products. The odd-order products are the problems that are difficult or impossible to filter.

We identify troublesome products by the harmonic relationship of the tones being mixed. We have a lowest order harmful product creating splatter***.*** The ***third-order product*** is the lowest order product that is a problem. This is where the 2nd harmonic of one tone mixes with the fundamental of the other desired tone. This is called the ***third-order product*** because mixing of **2 times F1** and one times F2 makes the new undesired signal. The harmful products are 2F1-F2 where F1 can be either tone, as can F2.

There are also other products. Consider the fifth order intermodulation product, or "IM5". This mix is caused by 2*F1 minus 3*F2. It's called the fifth-order because it is a combination of a second harmonic and third harmonic, and 2nd + 3rd is "5".

Looking at a Practical Transmitter

An 1850 kHz LSB transmitter modulated with audio tones of 500 Hz and 3000 Hz would have main signals at 1849.5 and 1847 kHz. This is the carrier or dial frequency of 1850, minus 500 Hz for one tone and 3 kHz for the other tone. The third-order IM products of this particular configuration will fall at:

(1849.5*2)-1847 = 1852 kHz

(1847*2)-1849.5 = 1844.5 kHz

You can see we have two new frequencies at 1852 and 1844.7 kHz, both OUTSIDE the bandwidth occupied by desired pure channel tones of 1847 kHz to 1849.5 kHz. This is why Golden Ears, no matter how good, can't hear splatter-causing distortion listening to the desired signal! It only bothers the other people up or down the band. The fifth-order products would fall at:

(1849.5*3)-(1847*2)=1854.5kHz

(1847*3)-(1849.5*2)=1842kHz

You can see every increase in order spreads the signal another (F1-F2) up and down the band. In the above case F1-F2 is 2.5 kHz. The 7th order product would be 2.5 kHz above and below the 5th order products. Any odd product adds bandwidth to the signal that is outside the passband of the original audio!

What if most of our speech energy is at 1849 and 1848, in the 1000-2000 Hz frequency range? In this case the stronger IM3 products would be (1849*2)-1848 = 1850 and (1848*2)-1849 = 1847 kHz. The IM3 extends the bandwidth only 1 kHz lower than the highest pitched LSB tone! The more we restrict bass, the less overall distortion bandwidth we have!

You won't find this on many (if any) ESSB web sites, but the worse case for bandwidth and splatter is when bass and treble are simultaneously increased! This gives the widest spread between <u>strong</u> frequencies in the RF signal, moving IM products the greatest possible distance from our "carrier" frequency. This is true even when we use good filters, and low distortion audio chains driving the transmitter.

If somebody above or below you is running enhanced bass you will almost certainly notice greatly increased adjacent channel interference problems, and they aren't likely to be your receiver's fault. Enhanced bass increases distortion product bandwidth, and do let anyone kid you...modern SSB transceivers all add very noticeable distortion products in the RF sections. Most tetrode grid-driven amplifiers are also much worse than most cathode driven amplifiers, just look at reviews.

Even-order Distortion

If the product is direct mixing or even-order mixing, mixing would be (1*F1)-(1*F2), (3*F1)-(1*F2), (2*F1)+(2*F2), etc. The "harmonic order" in the mixing would total an even number. Let's try that:

(1849.5*1)-(1847*1)=2.5 kHz. 2.5 kHz is well outside the passband of the transmitter and antenna! It isn't even RF anymore.

(1849.5*1)+(1847*1)= 3696.5 kHz, again well outside TX passband.

(1849.5*3)-(1847*1)=3701.5 kHz again outside the passband of the antenna and transmitter's RF section. This was the fourth order.

Even-order mixing clearly isn't a problem in RF transmitters. This is the reason why push-pull RF amplifiers don't help audible distortion and don't help bandwidth. Push-pull designs do reduce harmonic content (distortion). This in turn relaxes output filter requirements, but output filtering is generally is a non-issue anyway. Even a simple pi can provide adequate harmonic suppression.

Keep these rules in mind:

- ***Any increase in frequency difference between the highest and lowest modulation frequency increases BW greatly.***
- ***An increase in level increases the strength of the IM product in even greater proportion than we might expect.***

Odd-order products create most of our SSB headaches. This occurs because odd-order products can fall outside the normal passband of a typical SSB transmitter, spreading unwanted and undesirable distortion energy throughout adjacent voice channels. This is why Enhanced SSB or increased bass and treble is a very poor idea on crowded bands.

Three Greatest Sins

The three greatest sins creating unnecessary bandwidth are:

1. ***Turning up a radio's internal power or "drive limit" pot.***
2. ***Enhancing Bass and Treble***
3. ***Under-loading an amplifier***

Radios

Some modern radios are especially poor, even when operated at rated power levels. For example, the TS-2000 and IC-756 series radios are particularly bad on 160 meters. I can hear some of these radios, when signals are strong, producing weak spurious emissions (splatter) 10-20 kHz away from the operating frequency. If you look at ARRL test reports of transmitters, you will see many radios are just a bit above class-C performance levels. (Remember the ARRL uses dB below PEP, which improves results by 6 dB compared to commercial test methods.)

IM3 levels of -30 dB are really very poor. An old KWM2 I tested was -47 dB using ARRL standards. Compared to something like an IC-756, the Collins had about ***50 times LESS power*** in total adjacent channel distortion products!

Compression and Processing

The multiple tones in our voices contain low and high pitched tones that mix. The strength of the distortion products outside the desired communications channel depends heavily on the average power level of low and high pitches and the frequency spread between the lowest and highest pitches modulating the transmitter. Since the average level of lowest and highest modulating frequencies increase with speech processing, any form of speech processing or compression (even ALC) increases off-channel average IM power levels.

Processing is a bit of a double-edged sword. It isn't all bad. Processing that controls peak levels reduces chances of overdriving stages following the processing. Decreasing the ratio of peak to average power produces a more steady load on power supplies. It also can prevent later stages from limiting or clipping. Although processing brings the average power level of lows and highs up, it can also decreases overdrive problems.

Light or modest processing is actually beneficial in reducing splatter!

ALC

ALC is normally plagued with inherent problems. Filters in radios add group delay (the signal takes noticeable time to move through filters), and the ALC loop adds a time-delay of its own. The result can be a leading edge signal power overshoot, which often shows up as an adjacent channel "spit" or "pop" on leading or rising edges of voice or CW.

Some rigs like the early 775DSP and IC706 are plagued with very high levels of overshoot. A new 775DSP I had actually overshot to around 300 watts on leading edges for a few milliseconds. Kenwood and other radios also have this problem. This problem is not only harmful for bandwidth, it also can damage amplifiers. The problem often gets *WORSE* as power level is turned down!

Gain should be set so ALC just starts to take effect, if a drive control is available. Rigs like the FT1000D include a "Drive" adjustment.

WideFi or Enhanced Audio

Enhanced SSB audio is a generally bad idea, since it adds and boosts unnecessary lows and highs. Audio response flattening brings levels of unnecessary low and high frequencies up, and this rapidly increases power level in unwanted off-frequency products compared to normal communications audio. The frequency difference between lows and highs is wider, so the "junk" extends further than normal. The level of lows and highs are significantly stronger than levels in normal communications audio, and this makes IM products much stronger. As a matter of fact, energy in IM products does not follow a linear increase as base and treble are increased! ***Unwanted power on adjacent channels increases at several times the rate of the power increase in base and treble!***

Make no mistake about it, enhanced SSB or Hi-fi SSB audio, even with perfect "brick wall" filtering, is always going to have ***significantly*** more unwanted energy on adjacent channels when compared to regular communications audio through the same system.

Many of the radios popular with the ESSB crowd are among poorer radios for IM performance! My own opinion is Hi-Fi audio is OK on emptier bands, but we should do all we can to discourage enhanced audio on crowded bands or near weak signal areas.

Transmitter Tests

The standard transmitter test is IM3 or higher order products. The general test uses two tones of equal level. If it is a radio, the two tones are fed into the audio port. If it is an amplifier, the tones must generally be from two separate transmitters generating steady carriers. The carriers are mixed through a combiner and used to drive the amplifier. The reason we use two separate transmitters in the test is most amplifiers, especially cathode driven triodes, are far cleaner than most modern radios. The test radio's two-tone IM would establish the IM distortion limit (not the amplifier, in most cases.

Confusing Conflicting Standards

In most commercial tests, we find the maximum power of one tone and compare that level to the level of third-order spurious signals created in the transmitter. The ARRL for some reason adopted a different reference. The ARRL compares PEAK power of both tones to the spurious, rather than the level of one test tone to the spurious. This inflates transmitter IM results, making everything appear 6dB better than the standard dB-below-one-tone test used commercially.

If you look at Eimac data sheets, the IM specs are dB-below-one-tone of the two tones. If you look at data sheets from other sources, they might use dB below PEP. One manufacturer's US importer of Russian tetrodes used dB below PEP to compare the quality of their product to Eimac.

Unfortunately Eimac used dB below one tone, while the other tetrode test used dB below PEP. This results in a 6dB change in results. The tetrode manufacturer wrongly proclaimed their tetrodes "cleaner" when in fact they were not. If you look at QST tests of ETO, QRO, and Ameritron amps you will see the 3CX800's in the AL800H greatly surpass the 4CX800's in the other amps for IM3 and IM5. This confusion is a clear example of how mixed standards gets us in trouble.

It's a Poor Test Anyway

Two-tone IM3 (or higher orders, like IM5) transmitter tests generally show us the very ***BEST*** a transmitter will do. In general, two-tone tests are pretty poor tests for system designed to process speech. The two-tone test uses two steady signals (normally with wide spacing), but actual modulation has many frequencies that vary at syllabic rates.

In two-tone tests, the test spacing is generally a few kHz. The varying load on power and bias supplies is at the separation of the two frequencies. Small capacitors filter the time-varying load, while the long term (or low frequency) dynamic load remains constant. Two-tone tests do ***NOT*** show power supply deficiencies.

Slow variations in speech level load and unload power and bias supplies. This causes supplies to "wobble around". The conclusion of some is that screen or bias regulation is "unimportant", but that conclusion is mostly rooted in the fact the tester actually used a flawed test method that does not show low frequency dynamic regulation problems. We can't test for distortion created by poor low-frequency dynamic regulation when the test method provides a constant load on the supplies!

A Better Test

There are two tests that are better. One test is an adjacent channel power tests, with normal voice modulation of the transmitter, another test I developed uses a three-tone signal.

The three tone test injects a third low-frequency tone into the system. The third tone is anything from a warble to a low-pitched hum causing a slow variation in power levels of the two major tones. The analyzer reads the peak amplitudes of mixing in higher frequency tones while the level is varied at a syllabic to low pitched audio rate. The low pitch amplitude modulation is varied in frequency until the worse case IM is produced.

The variation causes the power to change at a speech rate, testing supply regulation effects on wide spaced distortion at all important frequencies for speech.

The adjacent channel power test simply uses normal voice operation and compares the long term peak power in an adjacent channel to the peak power in the desired channel.

Either test gives a much more reliable indication of transmitter bandwidth than a two-tone test.

Remember, a two-tone test is generally a "best case" scenario.

Summary

A normal two-tone test is not really very effective in measuring a SSB signal, because there is no slow dynamic change typical of a voice. Voltage regulation problems are masked and do not show up when a two-tone test is used, because load currents all average the same amount. Filter capacitors and other energy storage components mask any voltage regulation problems. We can, as a general rule, be confident the actual SSB voice performance is LESS than a two-tone test indicates.

A large improvement occurs with a three-tone test, when levels are varied at syllabic rates as well as higher frequencies in the voice range. This tests voltage regulation problems otherwise hid in a two-tone test, because all speech frequencies ranges are included. The most accurate test, of course, is with actual speech. The FCC now requires some commercial radios used on congested bands to be tested with actual speech.

We can help ease spectrum pollution by:

- **Strongly discouraging use of enhanced bass and treble on crowded bands. It has no place on crowded band or near weak signals. Wide-Fi is selfish and inconsiderate even when it uses a 3kHz filter because of the increase in level and frequency spread of IM products.**

- **Discouraging and chastising people would turn the power limit or drive limit control inside radios up. This is CB behavior! There isn't a radio made that will tolerate a user-increase in power limit without a serious degradation in IM performance. If you have a friend who peaks up the power control inside a radio, tell him why it is bad. Radios are bad enough without removing even more headroom. Even the best transistors cannot be driven more than about half of saturated power before IM becomes unacceptable.**

- **Making sure we use processing and ALC, but only at modest levels. We should just see the needles start to show compression.**
- **Making sure we tune amplifiers correctly, and avoiding 12 volt transistor amplifiers or grid-driven tetrode amplifiers whenever possible.**