Ham Audio Processing in a Nutshell – Just the Basics

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Foreword: I have been interested in better ham radio audio since the 1960's. As a broadcaster and voiceover artist, I have a professional interest in sounding the best I can on the air. As a former co-owner of a shortwave (HF) broadcast station, I also had to deal with the issue of high-quality audio firsthand as the one who engineered and maintained the station. I have applied professional audio processing techniques to my own ham station and get a lot of positive comments, along with questions about what I'm doing to achieve my sound. I thought it was time to write this paper to address the issue of better ham radio audio in an uncomplicated, simple to understand way, and am offering it in the hope that you too can improve your station sound, and for surprisingly little money.

Years ago, audio processing in ham radio was little known. In many cases, the famous Astatic D-104 crystal microphone ("The Lollipop") with its "communication sound" was the favorite method to achieve more effective audio. Then came circuits and devices such as volume compressors, limiters, and so-called "speech clippers", which were borrowed from commercial broadcasting and telecom services. These went further in giving the audio even more "punch", but also improved quality – if used correctly.

Today, audio processing is a ham specialty in itself. Those old techniques still exist, but a lot has happened to improve them, making it possible for the average ham to achieve broadcast-quality audio.

But there's a learning curve involved in getting all the audio processing together into what is termed the "audio processing chain" - i.e. the audio and processing devices being used before transmission. In this article, I'd like to acquaint you with the basics of audio processing theory - "just the basics" - and give you an idea of how a professional, broadcast-quality sound can be achieved – or at least approached - while at the same time taking into account the realities of typical narrow-band ham transmission and reception, especially on HF.

Before we get into the theories and gadgets, let's spend a moment on the radio, itself. Over the years, radio manufacturers have become aware of increased ham interest in transmitting better audio, and have been featuring transceivers with their own built-in audio processors, and some of them do an amazingly good job (some others fail miserably, though). You'll find these processors in both analogue and software-defined radios. A few of them are very expensive, while others are in the more affordable category. In other words, good audio does not have to cost the Earth.

Although it's beyond the scope of this article to recommend specific transceivers, certain makes and models have outstanding audio on their own – so good, in fact, that you may need nothing more than a good microphone to go with them. Listening to audiophiles, such as those who meet on 14.178 MHz is a good source of this information, as well as from individual specialist-hams. If you hear a station that sounds especially good, ask them what they're using and what they're doing to achieve their good audio.

If used carefully and ***in moderation***, built-in audio processors can produce excellent results by themselves. But remember, when adjusting them, "less is more". Read and follow the manual carefully, and avoid the temptation to turn all the controls wide-open. That can be disastrous, as we've all heard on the air.

You can have the world's best audio processing system, but the bandpass of your TX is the bottleneck that determines the fidelity of the transmitted audio. If your transmit bandpass is too narrow, your efforts will achieve little.

Before anything else, set your TX bandpass for a flat response between about 80 and 3,000 Hz, or as close to that as you can get. 3,000 – 3,200 Hz will provide a good quality audio at the high-end. (HF broadcasters are only transmitting a maximum of 4,500 Hz of audio and sometimes even less). Some will disagree, but in my experience, transmitting much more than 3,200 Hz on the crowded HF bands will usually make you more enemies than friends.

In my case, I installed a wider filter (the FL 70) in my ICOM 735 that now passes 3,000 Hz instead of the standard factory-supplied 2,300 Hz. I also introduce the audio signal directly into the balanced modulator using the ACC1 FSK input. Modulating the balanced modulator directly is the best method in older transceivers using mechanical filters, because it will usually be more linear than the mic input.

The most critical remaining parts of the audio chain are the microphone, and devices like equalizers, compressors, limiters and clippers. And then, there's the voice, itself.

Every part of an audio processing chain is important, but it starts with the voice. A lot of the impressive "sound" you hear from commercial broadcasters and hams with BC experience is the result of a naturally pleasant voice with the appropriate frequency register. These professionals also know how to project their voice and to enunciate carefully and clearly.

One of the biggest impediments to good voice quality is excessive "close-talking" of the mic. A standard professional rule is to keep back about 20 cm (8") and speak as though you were addressing someone across the room, but without straining the voice or shouting. Use a normal, but somewhat raised, voice. Since processing amplifies room noise, drapes and/or acoustic tiles may be required. It may also be necessary to relocate noisy equipment well away from the microphone. For example, I keep my linear amplifier in another part of the basement for this reason.

It's also important to speak slowly and carefully. What to you is your native language or dialect may be completely foreign and difficult to understand for many DX stations. Polish your accent. This is the fundamental starting point, and the voice I've just described will already sound better on any audio set-up. But there are electronic tricks that can help to make the audio even clearer and more articulate.

THE MICROPHONE: The importance of the mic is highly over-rated by many hams. There is no “silver bullet” that will suddenly make you sound like the VOA or the BBC. It is not the microphone alone that gives you a big sound. Any mic with a flat response (i.e. more or less flat between 80 and 12,000 Hz) should give you a good sound. Super-expensive microphones like Neumann and other exclusive condensers have little more than prestige value in most ham work. They are usually very sensitive and pick up more room and breath noise than a dynamic mic. But there are those who swear by them.

Electro Voice RE-20 microphone

Here I want to focus especially on the celebrated Electro Voice RE20 cardioid dynamic, which is often mysticized and revered. (I use one, but only because I got it for free). The reason the RE20 is the preferred broadcast mic today is because it has a very wide pickup axis (almost 180°), and because it is low in so-called “P-popping” and “proximity effect”. It also has low RF pickup.

In so many BC operations today, the presenter is a "self-op", meaning he has no control room technician and thus, has to do multitasking while on the air. This often results in a lot of body and head movement, so he's frequently “off-mic". This is where the RE20 excels – it's very forgiving in that respect. Its audio response is also very flat (important for FM stations and sophisticated audio processing equipment). Otherwise, there is no "magic" in the RE20. This mic by itself is not going to make you sound like God. There are many microphones that will do just as well on narrow-band HF. There's no need to pay $500 or more for a microphone. Here's why...

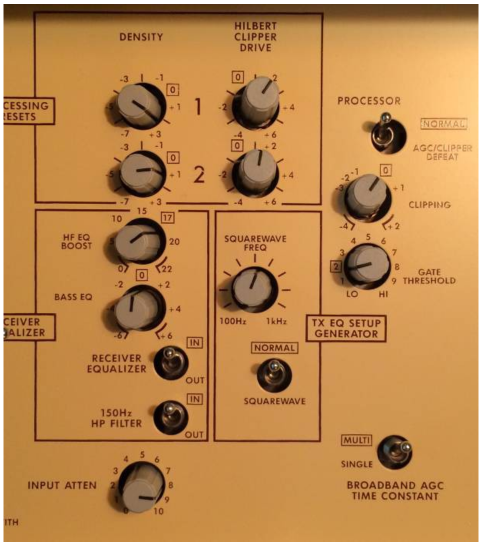
It's the processor that does the processing, not the mic !

The Orban Optimod-HF 9105A audio processor

This is what I use at HB9ASQ. The HF broadcast industry standard - the Orban Optimod-HF 9105A processor – was engineered specifically to sound best on a narrow-band shortwave radio, such as what we also use in HF ham radio. Typical SW/HF receivers have a bandpass of only 2.5 kHz.

Multi-band limiter meters of the 9105A

In a nutshell: the 9105A contains everything but the kitchen sink when it comes to audio processing for HF: Automatic Gain Control, noise gate, limiting and clipping of 6 discrete bands of frequencies (so-called “multi-band processing”, which processes each audio band separately and differently), a complex EQing matrix (which provides aggressive EQing in the mid and high-end audio range), and a clean-sounding aggregate clipper. There are very few audio processing techniques beside these, and it's all in one box.



9105A operator controls

Go to www.orban.com and download the multi-part Optimod 9105A manual. There's a wealth of general information and tips there, especially in section 3.

The only downside to the Optimod is the price - they cost a lot of money. (Mine was obtained for a fraction of the normal price by sheer luck and being in the right place at the right time). Incidentally, the 9105A has since been replaced by the Optimod 9300 and 9400 digital platforms, although many still prefer the 9105A analogue sound, including myself. The 9105A is still in regular service at the BBC, DW, RFI, VOA and other major HF broadcasters. Not surprising, since they are built for many years of continuous service.

But there are alternatives to the Orban Optimod for much less money, and they work very well. One of the most popular and inexpensive of these – and a favorite among budget-minded audiophiles - is the line of audio processors made by Behringer. I use a Behringer VX2496 all-in-one mic processor for my audio website and commercial voiceover production work, and the results are amazingly good – commercial quality. It also works well as a ham audio processor, although in most cases, it will require RFI protection. Rack mounting and good grounding, as well as adequate ferrite filtering of the audio and mains (AC) cables are usually needed in an RF environment, as is the case with some other semi-professional equipment. The VX2496 is no longer made, but can still be found, as well as older and newer Behringer processors of a similar type. An example is the VX2000. www.behringer.com/EN/Home.aspx

Behringer Ultravoice VX2496 mic processor

Now it's time to look at the key elements of the "audio chain" and how to adjust them correctly. The following functions may be in an all-in-one device, or in separate devices. However, the theory of adjustment is the same.

EQUALIZATION (EQ): Unless you're using an Optimod or similar professional device which equalizes the audio automatically, setting the EQ for your particular voice is probably the most difficult part of the set-up.

It's important to know that the most effective voice frequencies are between 300 and 3,000 Hz. Having said that, the so-called “presence range” (this gives the voice intelligibility and articulation) is found between 900 and 3,500 Hz. Adding about +3 to +8 dB of EQ boost in this area improves the clarity and presence of the voice and gives it brightness and sparkle. At the lower end of the audio scale are the so-called “warmth” (bass) frequencies. This range gives the voice a pleasant, mellow sound.

Beware: the low and mid-range are dangerous areas in HF communication, and are often greatly overdone by many hams!

Once again, the slogan to remember here is: “Less is more”. The low end – especially around 200 Hz and below 150 Hz – can make your audio sound muddy and unclear; these frequencies mask the higher presence frequencies. In fact, 180-220 Hz, and certainly below 150 Hz very often actually need an EQ ***reduction*** in the order of -3 to -8 dB, and sometimes even more, depending on the individual voice. On the other hand, a “thin” voice may actually need a low-end boost. Interestingly, the International Telecommunication Union (ITU) suggests a substantial roll-off below 150 Hz for HF broadcasting.

At this point, I want to make it clear that the purpose of this article is to help the average ham improve audio quality *within the confines of conventional amateur engineering practice.* There are some, however, who specialize in emulating the hi-fi sound of an FM broadcast station. This involves very complex processing techniques and wide-band transmission, and is commonly known as "Voodoo Sound". I do not do this myself and will not go into it here for that reason. For those who are interested, there are websites devoted entirely to that subject. Search the terms "voodoo audio" and "essb".

Individual voices vary greatly, and beyond the general advice above, there will be a need for on-air experimentation and fine-tuning of the EQ, preferably with the help of a cooperating station who understands audio processing. Listening to one's own signal is not recommended, because you will not hear what the distant receiver hears. It's also very wise to produce a sound that takes into account the narrow bandpass used by the majority of ham stations and not someone listening in wide-band. Consider the <u>typical</u> audience (as well as your own preferences, of course).

Ask for <u>honest</u> comment and criticism of your sound (remarks like "you sound great" or "you're BBC quality" are really not helpful, and often not even correct! You need more specifics). Also take into account DX conditions with weak signals, and not just "5-9+" conditions. There's a big difference!

The best way to evaluate your on-air sound is to get a "hard-wired" recording from a reliable source – i.e. from someone properly set up to make them.

COMPRESSION/LIMITING: The English Language, in particular, contains a great deal of dynamics (amplitude peaks and valleys). Typical British speech can have as much as 25 dB, while North American voices tend to have less - in the order of 15 to 18 dB. In transmission, it is important to have a high ***average*** modulation, as opposed to peaks, because the real communication power is in the average and not the instantaneous peak power (PEP). This is achieved by compressing the audio signal to keep the average modulation at a high level. This is often referred to as increasing the *density* of the audio.

A standard audio compressor value is a threshold of -18 to -20 dB and a ratio of 2:1 or 3:1, with fast attack and release times. This provides "punch" to the signal, and is essential for a smooth sound. Without compression or substantial limiting, your average modulation may be as little as 10 - 15 %. That's a lot of wasted communication power. This is also an effective way of making your signal sound louder without using a linear amplifier.

A word about Automatic Level Control (ALC): All transceivers have this limiting circuit that protects the final amplifier from being over-driven, and it can provide some audio compression effect as a by-product. But ALC's can overload and distort very easily. If this is the only form of compression available to you, then use the ALC in moderation and according to the manual. If you are using *external compression or limiting*, <u>avoid</u> TX ALC action by setting your level just below the transceiver's ALC threshold on voice peaks.

CLIPPING: Moderate clipping (producing a square wave audio signal by "clipping off" signal peaks) is another very effective way to improve average modulation and "punch". Anyone listening to shortwave (HF) broadcasting stations today is most likely hearing Orban Optimod-processed audio. Without the Optimod's unique clippers, it could never produce its distinctive sound. This technique was first used by the Voice of America (VOA) and Radio Free Europe and Radio Liberty (RFE / RL) during the Cold War to overcome jamming, and Orban adopted it in designing the Optimod-HF 9105A processor.

Basic clippers are simple to build, and circuits can be found in the usual reference sources. There are also a couple of old and new RF clippers around, but I don't consider them to be of sufficient "professional standard" in the context of this article, due to their narrow bandwidth.

A word of caution: Those favoring a more natural sound may not like the "controlled distortion" of clippers, and therefore would be advised not to use them. (The patented Optimod "Hilbert Clipper" is unique in that it is as effective as conventional RF clipping, but without the level of distortion associated with it).

SOFTWARE-BASED PROCESSING: In recent years, there have been great developments in computer software audio processing programs that can do everything the analogue equipment does. This is a newer specialty within a specialty, and although not the primary focus of this article, it would be a good idea to look into software audio processing as a viable alternative to conventional hardware. But whether digital or analogue, the same theories still apply.

THE "DX AUDIO" MYTH: There is a wide-spread belief that audio for DX or contest work has to be very narrow and shrill, or even distorted. I call it "meat grinder audio". We've all heard it. It's terrible to listen to, but even worse, it doesn't always do what is claimed or believed. On the contrary, the first audio frequencies to be lost under DX conditions are the lows and highs. Transmitting enough low end, in particular, can actually help in DX work. (That's what the "loudness contour" feature in your hi-fi system does when listening at low volume levels). This is why it's important to transmit a clean, balanced audio signal. I have often been in a pile-up, and the DX station told everybody else to stand by, because he wanted to work "that HB9 with the nice audio".

Warning: increasing the average modulation through audio processing can put a strain on transmitters and linear amplifiers, which may necessitate a reduction in RF power output to keep the power amplifier from being damaged. However, the loss of peak envelope power is offset by the increase in <u>average</u> power.

This is a complex, but fascinating, field and many of us have made it a hobby within a hobby. The audio processing specialists I know are happy to share knowledge and experience, and will be glad to help in on-the-air tests and adjustments. That includes me.

You can always contact me at contact@switzerlandinsound.com.

BEST OF LUCK AND 73!
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Bob Zanotti has been a licensed amateur radio operator since 1961, and has been interested in audio processing since that time. Bob was a full-time, on-air broadcast journalist with Swiss Radio International for 32 years, and was the co-host of the popular and award-winning SRI technical show "The Swiss Shortwave Merry-Go-Round" also known as "The Two Bobs", in which he and Bob Thomann HB9GX answered questions about shortwave and ham radio. HB9ASQ is now retired, but still active as a voiceover artist. Bob operates

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Thanks to Dan W1DAN for his editorial and layout assistance