Using frequency-selective multiband compression for improved HF voice communications

Technical Report

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I) Introduction

information messaging is performed on a global level, using various methods to convey point-to-point messages. HF voice communications (utilizing radio waves in the 2-30mHz frequency spectrum) are used globally for transmission and reception of information when the need arises, particularly in situations where other means of communication may not be available. Such communications rely on many uncontrollable factors, such as: 1) atmospheric conditions, 2) environment, 3) equipment availability, 4) reliability of communications medium, 5) operating conditions, and several others.

HF messaging using SSB (single sideband) modulation is a common mode for voice communications, is more efficient than conventional AM or FM, and is well accepted on a global level. A wide range of equipment has been and continues to be designed for such communications, and is a well proven method with successful results. Government, business, hobbyist, and experimental communications utilize such equipment with reliable performance, and have so for decades. Continued advances in technology allow for better performance, efficiency, and cost of HF communications equipment, which results in improved reliability, reduced resource needs, and dependability.

II) Sound primer

Sound waves are pressure waves, in which electrical energy is converted to mechanical energy with the use of transducers, such as microphones and speakers. The pressure waves enter the ear, which converts the pressure waves into vibrations, which then get converted to electrical signals to the brain.

The human ear is capable of hearing sound waves in the acoustic range of 20-20000 Hertz. Several factors, such as age, hearing loss, gender, and physiological factors can cause this range to be limited, however the human voice is limited to a much narrower range, typically 200-4000 Hertz.

In wired telephone communications, equipment uses the voice frequency band from approximately 300 to 3500 Hz.[1] This same voice band is used on HF communications.

Adult males typically have a fundamental frequency from 85 to 155 Hz, while adult females range from 165 to 255 Hz.[1] Although the fundamental frequency of most speech falls below the bottom of the voice frequency band, voice harmonics make up for the missing fundamental to create the intelligible tones for the human voice.

Since pressure is a physical force measured in Pascals (similar to Newtons), and sound pressure level (SPL) is measured in decibels(dB) of SPL, where approximately 2 Pascal = 100dB SPL (20log function). To give relative examples of SPL levels, 0dB SPL is identified as the 'threshold of hearing', 50dB SPL= normal conversation between two persons, and 130dB SPL is identified as the 'threshold of pain' (equivalent to a musical trumpet measured at 1M from the bell of the instrument). Sound pressure measurements can also be 'weighted', in which filters are added to the measuring instrument, for the purpose of limiting or shaping the measurements of specified frequency ranges. For example, "A-weighted" measurements involve sharp filtering of lower frequencies of the hearing range (as well as some high frequency filtering), and "C-weighted" measurements are primarily a 'flat' response (almost no filtering). For the purposes of this paper, only unweighted measurements shall be discussed.

Further complications to the human ear involve the uneven frequency response at various sound pressure levels. Scientists Fletcher and Munson determined that the human ear responds to frequencies differently at various SPL. They discovered that at low SPL levels, midrange frequencies(1-3kHz) are more prominent, while the low and high frequency ranges seem to appear much lower to the listener. Conversely, at high SPL levels, the lower and higher frequencies are stronger, while the midrange appear lower. They published the now well-known '*Fletcher-Munson curves of equal loudness'*, along with their studies (see figure 1). This phenomenon became a marketing tool for many home and commercial audiophile equipment, and included circuitry and controls to enhance the effects of this listening issue to make the audio content appear more pleasing to the listener. Remember the 'loudness' button on many home stereos? The purpose was to modify the frequency response of the audio for various volume levels so that the tonal quality of the musical content was enhanced.

Although the unequal loudness issue is most prevalent in listening to musical content, it is not as much an issue with voice content. However, the same techniques used to enhance musical content are similar to the techniques used for voice content, in a limited fashion. These enhancements are also used in two-way HF radio communications, but not necessarily to the best of the technological art.

Hearing loss can further deteriorate speech intelligibility. The listener with such a condition finds it very difficult and frustrating to understand a voice communication when suffering such a disease. Enhancements to the audio spectrum, such as hearing improvement devices, custom auditory products, and other listening skills can enhance the performance of the hearing-impaired, but the initial problem remains.

III) Intelligibility

In radio communications, the information passed from the sender to the recipient may be critical, and sometimes life-saving, while other information may only involve casual communications. In either case, conveyance of the information depends on intelligibility of the signal, particularly when using voice as the method of conveyance. Most importantly, in two-way communications, the voice must be intelligible in both directions, so that all parties to the communication can understand the message passed from point to point, with minimal issues.

Under optimum conditions, outside forms of interference are not present, which cause the intelligibility of the communications to be disrupted. However, optimum conditions may not always be present, and reliability of communications becomes diminished. Such disruptions can include:

- signal fading,
- noise and static,
- interference from other adjacent-channel signals,
- poor quality transmission,
- insufficient signal path propagation.

Any or all of these factors can make communications unintelligible, and results in failed messages. With modern technology, the use of Digital Signal Processing (DSP) can improve some of the anomalies that occur during voice communications. However, unless specifically addressed for each operating environment, preselected DSP technology still has limitations that still require human intervention.

Due to the fact that voice SSB HF communications uses an almost identical audio response range as the wired telephone industry (bandwidth of 300-3500 Hz), vocal intelligibility is somewhat restricted, and the SSB voice communications industry develop their products to be limited to this frequency response range. This results in manufacturers of support equipment (microphones, audio processing, equalization, compression, and others) to operate most efficiently within the limitation of this bandwidth. Furthermore, government regulating agencies that place conditional rules on transmission parameters (such as ‘occupied bandwidth’), frequency response, and spectral purity) also must be maintained for legal compliance. Considering all of the conditions for successful transmissions, every optimization step needs to be considered for best performance.

IV) Microphone performance and response vs intelligibility

Microphones are used every day and are accepted as an industry-standard device for daily needs, with little concern to their presence and design by the casual user. Microphones are relied upon for personal communications, social events, musical performances, communications, and many other needed services. Microphones have become a dependent and lasting device to transfer vocal information into the medium in which they are used. Microphones play an important role that many users pass off as just another useful simple device, while not appreciating the necessity for proper performance of such a device when safety of life is involved.

The technology that microphones utilize converts sound pressure waves into electrical energy. Early microphones used contained carbon granules in an electrically charged field, and the vibration of the granules from the voice would cause the granules to vibrate, which would generate an electrical voltage proportional to the frequency and amplitude of the voice. Later, speakers were turned into microphones, where the diaphragm was attached to a coil, wrapped around a magnetic field, and a voltage was created from the vibrating coil moving across the magnetic field.

Dynamic microphones were then created from this same concept, with improved materials for improved frequency response. This type of microphone is widely used in both the professional and consumer marketplace. Crystal microphones utilize a stretched metallic diaphragm, attached to a point-source piezoelectric element. These types of microphones were developed for their crisp, higher frequency performance sought after in the radio communications industry.

The most current and very common microphone used in smartphones, computers, and many industries is the condenser microphone, in which a polarized capacitive field creates a voltage proportional to the vocal frequency and amplitude. This technology results in superior performance at an extremely low cost investment.

Other microphone technologies exist, but for the purpose of this paper, dynamic and condenser mics will be primarily discussed.

Microphone manufactures also make considerable effort in the development of the acoustic properties of the microphone, which affects performance and desired frequency 'shaping'. Some manufacturers tailor the microphone for specific applications, i.e. 'noise-cancelling' mics, lapel mics, handheld, etc., which provides very exclusive use in certain applications.

Microphones exhibit a phenomenon known as *proximity effect*. This causes the frequency response to behave in such a way that speaking closer to the microphone causes the lower frequencies to be enhanced, and the voice will sound 'deeper' and 'bassy', particularly below 200Hz (3). Although this effect may not have a large effect on HF Communications, it can enhance the lower frequencies of voice transmissions in some cases.

HF Communication equipment typically use 'factory' microphones, that may have certain limitations that third-party communications microphone manufacturers may not have, and some users tend to rely on better quality microphones than what the equipment manufacturer supplies as a 'standard' microphone. The third-party microphones may offer improved frequency response, frequency tailoring to optimize the transmit medium, amplification to increase the volume, and adjustable frequency equalization for selectable frequency enhancement.

Another issue is changes in the performance of the microphone when connected to the transmitter. So that the maximum performance can be achieved from the microphone, the device that the microphone connects to should not deteriorate the microphones frequency response and designed output level.

Impedance is an electrical term that involves resistance, and reactance in an electrical circuit. The resistance is based on Ohm's law, where maximum current flows when the resistance is at a minimum at a given voltage, in a DC circuit. Reactance is the AC equivalent of resistance, which involves capacitance and inductive properties. When the capacitive and inductive reactance's cancel, the total reactance is also cancelled, and the only remaining element of the impedance is the pure resistance. Any reactance in the circuit causes the AC component to change due to this effect.

When dealing with microphones, adding resistance or reactance to the microphone causes a 'loading' effect on the microphone, which can change the frequency response, output amplitude, or both. The design of a transmitter should be such that the input impedance (load) of the transmitter is either much greater than the microphone impedance (source), or the impedance should be matched between the microphone and transmitter. Although the matching concept was widely utilized on microphones for many decades, modern designs allow for load impedances are very high (known as 'bridging' input impedances), so that connecting a microphone will not deteriorate the performance of the microphone. Even though manufacturers of HF Communications equipment use this method, some manufacturers tailor the impedance and frequency response of the microphone input circuitry so that their 'stock' microphone provides the best combination. Unfortunately, this may limit the user's ability to select a different microphone if desired, as it may not perform as expected due to the tailoring. The change in performance can also lead to reduced intelligibility of the transmitted audio communications.

V) Receiver audio performance and noise reduction techniques

HF Communication receivers operate over a very wide dynamic range. In weak signal SSB reception, s/n ratios as low as 6dB can still result in 90% intelligibility (4), however this does not include the effects of selective fading that may occur during HF communications, due to atmospheric conditions, which worsens intelligibility by a considerable margin.

An SSB signal (single-sideband suppressed carrier) generates a signal dependent on the amplitude of the transmitted signal, and only on one sideband (through filtering). When there is no audio being generated, there is no signal transmitted. To recover the audio at the receiving end, an artificial carrier is generated in the receiver, offset by the corresponding sideband being received. This carrier 'demodulates' the incoming SSB signal into the audio baseband though a demodulator stage.

The 'shape' of the SSB filter in the receiver determines the received audio bandwidth. Vintage receivers used technology that limited the quality of the received signal, and reception intelligibility may have been limited. Modern SSB technology has made significant improvements in filtering technology, optimizing the transmission to the best it can receive the signal.

Not considering operating conditions, both transmitter and receiver specifications should be optimized for best performance for optimum intelligibility. Once this is achieved, operating conditions would have less effect on intelligibility.

Modern-day technology utilizes digital signal processing (DSP) for enhanced reception performance. Communications equipment only a few decades ago dwarfs the performance of a DSP-enhanced receiver, and the result is a significant improvement in SSB signal reception, particularly in noise reduction. Intelligibility is much improved, resulting in more dependable communications. Notwithstanding this, there remains a considerable quantity of equipment still in use that does not have the advantage of such technology, yet still operates as reliably as the equipment can provide. Some industries are slow to change or upgrade equipment, either by budgetary restrictions, or regulations that limit selection availability. By continuance of use, this equipment may have limitations that cannot be overcome, due to the inherent design of the products. Therefore, external technology may be necessary to overcome limitations, both on transmitter and receiver products.

VI) Compression / speech processing

One technique that is widely used in voice HF Communications is the use of audio compression, or speech processors.
First, some discussion on *noise floor*. The rough definition is the noise (usually in the form of a 'hiss'- sometimes mis- referred to as 'white noise') that can be heard in a medium between the pauses in the audio content. For example, a 78 LP record has a typical noise floor between 35-45dB below the signal peaks, a 33-1/3 LP could achieve -55dB, high-quality reel-to-reel tape equipment can achieve up to -65dB, and Compact Disks are typically -96dB below signal peak. The human ear can typically perceive a noise floor of -60dB, and barely perceive a noise floor at -70dB. Noise floors below -80dB may be only discernable by audio 'purists', but very rarely. Of course, this under ideal conditions, in a 'quiet' room, with no external noises or sounds.

In the HF communications industry, recorded musical content noise floor levels would be perfect, but can never be achieved in the analog medium in which HF communications exists. Remembering the 6dB s/n ratio from above HF minimum, a ‘comfortable’ listening scenario for HF communications would be useable with at least 30db s/n, and optimum with 40dB s/n(4).

From previous discussion, SSB transmission is dependent on the highest audio level for peak reception. Because the receiver operates over a wide dynamic range, the gain is maximized when no input signal to the receiver occurs, and when a strong signal occurs, the receiver has to dynamically reduce the gain (using automatic gain control [AGC] circuitry) so that audio overload does not occur in the electronics. The use of AGC allows both strong and weak signals to be within a tolerable audio level, so that the operator is not constantly re-adjusting the controls for such a wide range of received signal strengths. This AGC action also reduces the dynamic range of the noise floor, as the noise is maximum when the signal is weakest, since the receiver is operating at maximum gain. When the received signal is strong, the gain is reduced, and the noise floor is greatly reduced as well, producing a much greater s/n ratio on stronger signals.

The AGC circuit utilizes signal level ‘compression’, reducing the gain with higher input signals, and increasing the gain with lower input signals automatically. The ‘attack’ and ‘release’ time of the gain change is optimized so that the gain does not change instantaneously, but rather slowly (in seconds or many milliseconds), so that the effect doesn’t sound like the gain is increased faster than what the ear can tolerate during pauses in SSB transmissions.

Audio compression in HF transmitters is very similar to the AGC action in a receiver. The purpose is to act like a quick responding gain control device, such that lower-level audio can be increased to match the to the audio peak level, effectively increasing the overall volume of the audio. The result is a louder audio communication, which can assist during weak signal conditions. This also reduces the need for the operator to speak loudly versus softly, as the compression process will balance the loud and soft volume of the voice.

Audio compression also results in audio *limiting*, such that the audio ‘peaks’ can be held to a specified level so that distortion from overmodulation will be reduced. This limiting action reduces unwanted RF emissions from the byproducts of overmodulation; unwanted RF emissions can cause harmful interference to other radio communications.

Both the threshold level and peak levels of an audio compressor circuit can be adjusted so that the compressor does not increase the lower-level volumes excessively, as well as set the peak limit point accordingly. The compression ratio affects how much of the audio is limited with a specified input level range, and the threshold is the level where the audio begins to hold the gain to the limit point.

<u>Audio expansion/audio gating</u>

One of the drawbacks of audio compression is a significant increase in background noise. During audio compression, the background noise can match the level of the soft voice passages of the operator, which can be annoying and make the message difficult to understand.

Audio *gating* is a term that refers to a removal of audio compression below a threshold point. This effect essentially ‘shuts off’ the audio during pauses in the spoken message, which drops the background noise below the level of the compressor. This reduces the effects of background noise during compressed audio, and is used in environments where background noises may create intelligibility issues in a voice transmission. Noise Gating operates like an on-off switch, removing background noise entering the microphone when no voice is speaking.

A more subtle method of removing background noise is using *audio expansion.* With this process, the audio 'gate' process is replaced with a gradual (usually in fractions of a few seconds) lowering in compression level until the audio gain is reduced to no audio output. This is a more pleasing sound to the listener, and is commonly used in professional audio applications.

Audio compression and expansion have limitations. Too much compression can cause a fatiguing sound, extremely fast compression 'release time' can amplify vocal breath sounds, mis-adjusted setting of an expansion threshold can cut off portions of the lower-level voice communications, etc. Proper adjustment will reveal the best balance when using compression and expansion simultaneously.

VII) Audio Equalization

Audio equalization is the process in which electronic circuitry adjusts the levels of various frequency 'bands', such that certain frequency ranges can be enhanced or reduced. Audio equalization is used both in adjustable and fixed instances. Some examples include:

- The treble and bass control adjustments on many consumer audio devices,
- Professional audio, musical instruments, broadcast, sound reinforcement,
- Car stereos, 'boom boxes', smart phones, personal computers, televisions, radios, etc,
- Communications equipment (low-pass audio filtering, hum elimination, static reduction, etc.)

Many of the uses are subjective in nature (for personal listening pleasure), while others fit a particular need based on the application. For example, the terms *pre-emphasis* and *de-emphasis* are used in FM radio communications, for the purpose of improving signal-to-noise ratios of the audio signal. These are fixed value electronic components that boost higher-frequency audio components during transmission, then cut the higher-frequency audio components inversely to the pre-emphasized audio curve. The purpose is to reduce the FM noise when the signal is not at full quieting without sacrificing audio quality. The pre-emphasis has to be controlled with limiting or a form of frequency-selective audio compression at the transmit end, such that emphasized higher-frequency audio components do not cause overmodulation of the transmitted signal.

Being that the human ear does not hear all audio frequencies at all levels, the use of equalization is typically used to enhance the audio tonal quality of the audio content. Equalizer devices are usually divided into frequency 'bands', where the center frequency of each band has the greatest impact, and adjacent frequencies have a 'bell-shaped' response centered around the center frequency. The equalizer has sufficient 'overlap' between the center frequencies, so that adjacent frequencies will not be lost between bands. Since the audio spectrum is divided into a range between 20-20000 Hz, each equalizer center frequency band is set on 'octave' spacings (30Hz, 60Hz, 120Hz ,250Hz, 500Hz, etc), so that the bands are evenly divided across the audio spectrum. Some professional equalizers (particularly in sound reinforcement and feedback elimination) are set on half or third-octave bands, so that more distinct frequency adjustments can be achieved.

In some applications, the use of adjustable-frequency equalization is used. This is particularly useful when dealing with specific sound issues, like audio 'notching' of very distinct frequencies(to control audio feedback during live performances), or enhancement of a particular frequency for a designated purpose. In all cases, equalization controls can be set to either 'boost' or 'cut' a frequency band, which in effect, increases or decreases a particular range of frequencies in the audio spectrum. An infinite combination of adjustments can be made for any particular desired sonic effect. The use of adjustable-frequency equalization is a form of *parametric equalization*, since both the frequency, bandwidth, and gain of the frequency band can be varied. Alternately, *graphic equalizers* are fixed-frequency equalizers with the ability to only control the gain of each frequency band.

Equalizers can cut and boost frequencies within a specified gain range. Typically, +/- 12dB is a common range for most equalizer bands, with some slightly greater or less. Keep in mind that 6dB is a doubling of the audio level, so a typical equalizer can boost/cut up to four times the center band during adjustment. If you start boosting more of the center frequencies, the total gain increases even more. This is important to remember, because if the audio content already has boosted frequencies on the band that is already boosted, the output level will approach close to or above the maximum output level of the equalizer, and peak clipping can occur, which will cause distortion to the audio signal. This is especially important to remember during live sound, as 6dB of audio change requires four times the available power from the speaker amplification system, which could easily reach the point of over-driving and serious distortion. Therefore, it is important set the controls on an equalizer with this in mind. In many cases, it may be better to cut the undesired frequencies instead of boosting the desired ones. This method is commonly used for adjusting the problem of feedback in a live sound system, as the offending frequencies are cut so that the gain is greatly reduced when the sound system approaches feedback thresholds. Automatic feedback control devices operate on this principle; sensing the feedback frequency, and electronically moving the center frequency until it matches the offending frequency, then reducing the gain and/or bandwidth of the filter until the feedback is eliminated.

With regard to HF SSB communications, the equalization parameters will fall within the voice frequency band(300-3500Hz), so equalization is limited to less than five octaves.

Some of the more contemporary HF communications transceivers, particularly the amateur radio industry, have incorporated adjustable equalization into the microphone circuitry of their equipment, which allows the user additional audio tailoring of the microphone audio.

Audio equalization is also used to enhance portions of the audio spectrum that are difficult for some human ears to hear properly. Hearing loss typically affects the higher frequencies of the audio spectrum from being detected by the ear. In more severe cases, the higher portion of the voice band can also be difficult to hear. This causes challenges to the listener, particularly when intelligibility is lost. Electronic devices can boost the missing frequencies, and hearing-impaired persons can attain some hearing relief once these missing frequencies are enhanced. These devices amplify portions of the audio frequency bands that are typically diminished, and some devices have adjustments for variations in the frequency settings for optimum personal enhancement.

VIII) Multiband compression vs single-band compression

Audio compression results in reduction of audio dynamics, as well as limiting maximum audio levels. Since the audio spectrum covers 10 octaves, the compressor has to operate over this same range. Single-band compression is very common, but has its own anomalies. One particular issue is the attack and release time constants used in the gain control portion of the compressor. The lower portion of the audio spectrum responds quite differently from the higher end of the spectrum, and designers of audio compressors have to consider these effects when designing the time constants. Referring to figure (D) audio compressor block, the audio input passes through a variable gain stage, controlled by a detector connected in a feedback fashion, so that the increase in signal from the variable amplifier stage is sampled by the detector, which creates a control voltage that decreases the gain of the variable amplifier stage, until the gain does not exceed the threshold of the detector. Increased input signal will cause additional compression, and reduced gain will cause the control voltage to be reduced, which increases the gain of the amplifier. This, the varying input level will maintain a relatively constant output, within limitations of the electronics.

Since the detector stage produces a variable voltage, the voltage must be free of AC components from the audio sine waves (otherwise severe distortion of the control voltage would severely modulate and garble the audio output). Filtering of the AC component is done though rectification and filtering, which creates a time constant. This time constant is the ‘attack’ (how quickly the control voltage reacts and ‘release’ (the time to discharge the filtered signal so that the gain increases to normal). Since the time constant has to contain sufficient filtering to act on the entire audio spectrum, a minimum attack and release time must be calculated. Since all frequencies will respond differently to the time constant, there will be an averaging of the time constant based on the audio frequency energy being sent to the time constant. If the release time is set too quickly, the lower frequencies will sound un-natural (like a ‘pumping’ sound), and if set too slowly, the higher frequencies will be less responsive, due to the slow reaction time of the gain control. Therefore, a fixed time constant will never be a perfect solution for the entire audio spectrum and across all audio applications.

Many professional audio compressors contain adjustable attack and release time settings, so that the optimum sound effects can be achieved by the user. Audio compression is commonly used in various degrees; from recording studio mic limiting, to special effects, to peak audio control and overload protection in public address systems. On the consumer end, audio compression may be used in portable recording devices for unmanaged audio control.

IX Frequency-selective compression

In some audio processing situations, the lower frequency energy may be stronger than high-frequency energy, which will trigger a broadband compressor to act upon the detected signal to begin compressing the audio. When it does, high frequencies are compressed along with the lower frequencies that trigged the compressor. This will cause the energy in the higher frequencies to become less pronounced, even though the lower frequencies contain most of the energy being compressed.

Even though you can boost the high frequencies with EQ to make up for what was lost, a more natural solution is to compress only the lower frequencies (and reduce their gain somewhat), while leaving the higher frequencies uncompressed so that the they are more pronounced from the compressed audio. This technique is known as ‘frequency-selective’ compression.

In some cases, the frequency ranges can be adjusted, such that specific ranges (or ‘bands’) of frequencies are individually compressed for maximum effect, and variations in the frequency windows can be optimized for the best perceived sonic result. The term ‘dynamic equalization’(5) can be applied to some devices that adjust the amount of gain based on the audio content, and was used by RCA in the 1960’s under the term ‘Dynagroove’. The result was enhanced, lively-sounding recordings, resulting in greater listener pleasure.

With Frequency-selective compression, increased energy in a specific band will not be subject to the additional energy being boosted, because the compression will hold the gain of the boosted energy of a given frequency range to a safe level.

X Multiband audio compression

Multiband audio compression is a technique in which the audio spectrum is split up into multiple, fixed-frequency bands, so that individual compression of each band can be obtained, then the output of each compression band is electronically mixed together into the final product. The result of multiband compression separates the audio bands with more unified control, and the sonic quality of the original content is more distinct, with less artifacts from the broadband compression processes.

The use of multiband compression is widely used in the radio broadcast industry; where the radio stations attempt to achieve the greatest amount of perceived loudness for their listeners. This technique was first developed as a two-band approach, and has increased up to ten bands of individual frequency-selective compression bands. The belief that “louder means stronger’ has many mis-conceptions, but is accepted as an industry practice in the radio broadcast industry.

XI Combining multi-band compression with frequency-selective compression

Multiband compression (MBC) and frequency-selective compression (FSC) techniques have both positive and negative effects when used individually.

With MBC, the frequencies of each band are pre-set, and the user is unable to adjust the passbands of each compression stage. Whatever energy is present in the center of the passband is the limitation to the processed audio. MBC, however is used widely in the broadcast industry, where the preset bands will result in already proven results.

With FCS, there is some limitation to how many bands use this technique, and the user must be careful in adjusting the bands so that the effect does not ‘over-process’ a specific range of frequencies. This can also lead to very destructive audio quality if set improperly.

XII Advanced audio processing technology

With the onset of digital signal processing (DSP) technology, modern-day audio processing hardware can be reduced to microprocessor-controlled DSP devices. DSP utilizes high-speed mathematical functions to perform audio filtering, gain control, and audio tailoring based on user settings. With the use of such technology, a wide range of adjustments can be obtained without the use of multiple analog controls that would be needed in older technology, and user presets can provide an infinite range of adjustments and settings.

XIII Summary:

Audio processing for HF Communications is becoming a standard for high-end HF equipment, with the additional control of audio tailoring found in such devices. This does leave a gap in the lower end technology products, for which there is a need for such product. Some users want to get the best quality and performance from their signal, while others are happy with simply making a contact to a foreign land. In either case, audio quality is paramount in intelligibility and reliability, and the use of various methods of audio enhancement can be achieved. However, careful selection and considerations must be made for the best results.

XIV Figures:

FIGURE 1: Audio spectrum:

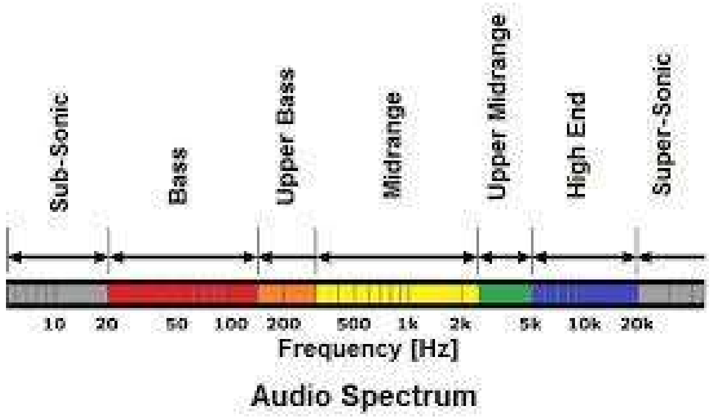
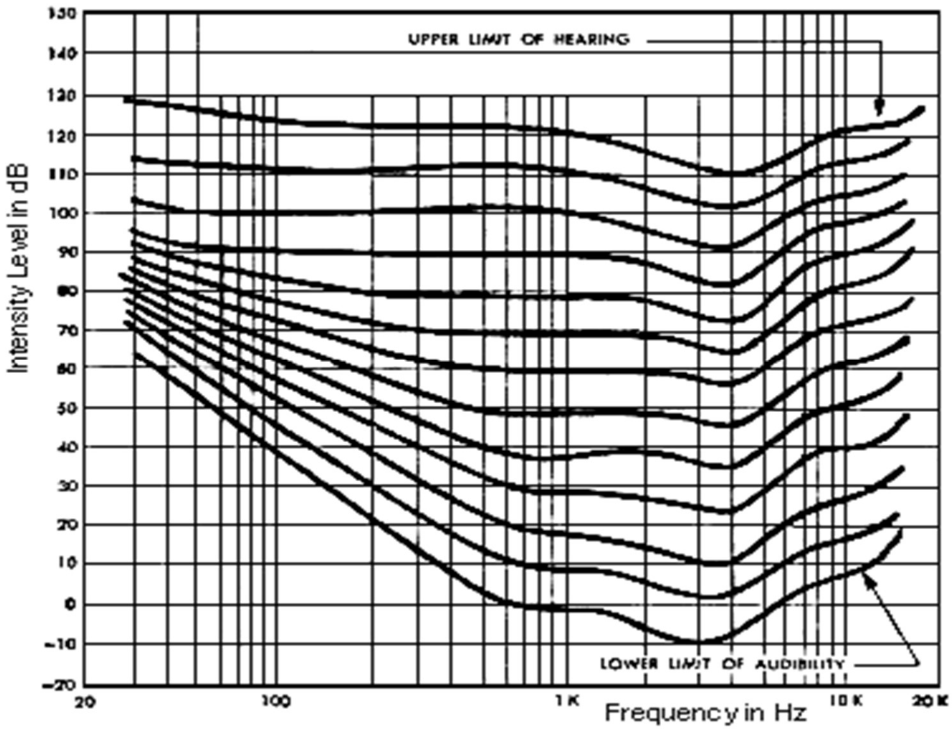


Figure 2: Fletcher-Munson Equal loudness curves:



XV References:

(1) Wikipedia (2022). *Voice frequencies*, https://en.wikipedia.org/wiki/Voice_frequency
(2) Leonard, Bill (2020). *Using Audio Equalizers to Improve SSB Communications*, www.naØtc.org/
(3) Neumann (website). *Proximity Effect* www.neumann.com/homestudio/en/what-is-the-proximity-effect
(4) ESSA ERL 131-ITS 92. *Required Signal-to-Noise Ratios for HF Communication Systems,* https://apps.dtic.mil/sti/pdfs/AD0697579.pdf
(5) Trapani, Jim; Richter, Don *Signal Processing using Dynamic Equalization,* AES Convention: 79 (October 1985) Paper Number: 2309