

DSP

Digital
Signal
Processing

FACTS AND EQUIPMENT



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Digital Signal Processing

Facts and Equipment

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“No Compromise Communications”

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The Idea of Digital Sound Processing

Introduction. Digital Signal Processing (DSP) may soon revolutionize many aspects of the electronics industry. DSP will have much the same effect on electronics that personal computers have had on everyday life since the early 1980s. And part of that effect is due to the fact that DSP is computer-related.

You can expect DSP to affect applications as varied as medical electronics, diesel engine tune-ups, speech processing, long-distance telephone calls, music processing and recording, and television and video enhancement. This book mentions some of these applications, but it focuses mostly on the products and techniques used in high frequency two-way communications.

First, a few of the basics. We will discuss concepts of sound, sound retrieval, and sound transmission by radio. Then we will discuss how modern technology uses digital in accomplishing these same tasks.

Understanding Sound

We feel the need to save our sense experiences. For instance, we record photographs and video images, although we don't expect these mediums to reproduce exactly the original. The photograph and video screen containing an image of a cloud differ, of course, from a real cloud floating in the atmosphere.

But sound, heard through one of our basic senses, holds a special place in our lives because it allows us to communicate, protect ourselves from danger, and entertain ourselves.

And so, we save and retrieve our voices and our music on tape and disc, and we transmit them to other parts of the world via radio waves, wires, and cables. Anytime we trans-





mit, save, or retrieve a sound signal (which we call an audio signal), that signal must be changed into a storable form and then reconstituted into its former state so that we can understand it and enjoy it.

Hearing sound

The sound of the rain hitting the ground is a physical phenomenon. The rain drops hit the ground and cause air molecules to vibrate, to transmit through the air until their energy dissipates. If your ear is within range of the vibrations, the external parts of your ear will focus them so that they will travel down the ear canals to the ear drum and bones in the ears. Where the last bone connects to nerves, the physical vibrations become neural impulses, and your brain signals you that you hear the rain hitting the ground.

Those sound vibrations (called audio) travel in ripples, like ripples in a pond when you toss in a rock. Ripples of water will radiate out from the place that the rock splashed. The height (amplitude) of the ripples will decrease as they move farther away from the source of the splash. The amplitude of the ripples represents the loudness of the sound.

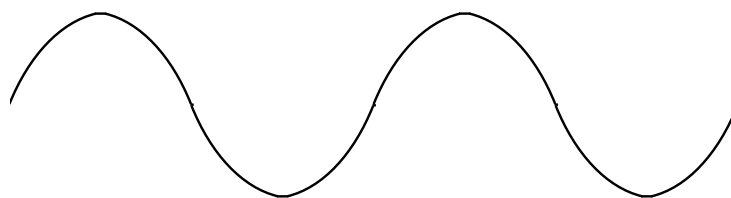


Figure 1 — Simple ripple form

Frequency. The measure of each ripple from peak to peak represents its frequency. The longer the measure, the lower the frequency (and the deeper the sound pitch). The shorter the measure, the higher the frequency (and the higher the sound pitch).

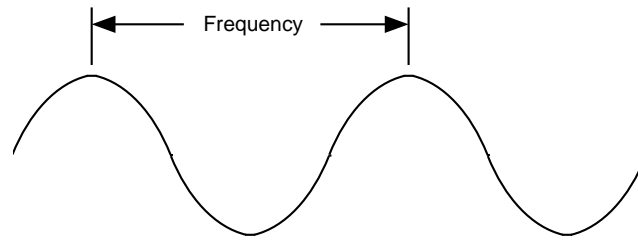


Figure 2 — Frequency of ripple from peak to peak

Amplitude. The measure of each ripple from peak to trough represents its loudness (amplitude). In between the peak and depth of the ripples, the level of the water is the same as it is throughout the rest of the pond.

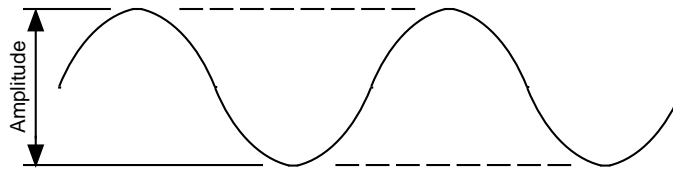


Figure 3 — Amplitude of ripple from peak to trough

Complex audio signals, however, look much different from those ripples on the pond. Whereas the pond ripples would resemble single-tone audio signals (like ones from a tone generator or tuning fork), complex sounds such as speech and the sound of musical instruments comprise many different waves that overlap and mix together, a much more jagged, complicated wave than any of those ripples on the pond.

Storing and Retrieving Sound

When a microphone picks up a sound, it changes the sound vibrations into electrical impulses. Inside the microphone, the sound waves strike a thin element (typically a *diaphragm* or *ribbon*). The movement of that element





through a magnetic field induces an electromagnetic signal that will travel to an amplifier to boost the amplitude of the tiny audio signals to a more usable level.

Storing sound. A phonograph record illustrates how the vibrational pattern from the microphone/amplifier translates those electromagnetic signals into physical vibrations. The vibrations, cut into the grooves of a vinyl disc, match the vibrations that the diaphragm made: waves that vary in amplitude and frequency.

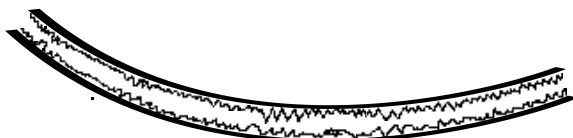


Figure 4 — Sound vibrations cut into the sides of a long-play recording groove

Retrieving sound. To reproduce the sounds cut into the vinyl record requires a phono cartridge very much like a microphone: it contains an element that moves within an electromagnetic field as the needle moves along in the grooves. The width (amplitude) of the groove controls the volume, and the rapidity (frequency) controls the pitch of the sound.

The electrical impulses from the phono cartridge travel to an amplifier, from which the strengthened signals travel to a speaker to be reproduced again as vibrations in the air. The electrical impulses cause the speaker voice coil to pump in and out, causing the speaker cone to vibrate just as the microphone element did, transmitting those vibrations through the air—to your waiting ear.

Transmitting and Receiving Sound by Radio

This book concerns DSP in radio technology, transmitting and receiving audio signals via the radio. This technology must address how to transmit a radio frequency signal that also

conveys an audio message. Consider that the typical voice signal ranges from about 100 to 5000 Hz (.1 to 5 kHz) while a typical radio signal might be transmitted on 7,200,000 Hz (7200 kHz—in the 40-meter amateur band). Somehow, the two signals have to be mixed together.

Modulation. One of the most common means to impress an audio signal on a radio signal is *amplitude modulation* (AM). The first component of the AM signal is the carrier. Just an “empty” radio signal that contains no audio, the *carrier* is called that because its only purpose is to carry an audio signal to receivers. A good way to hear a carrier is to tune in to the AM broadcast band and tune in to a radio station. When there is no audio and no static, you are hearing the carrier.

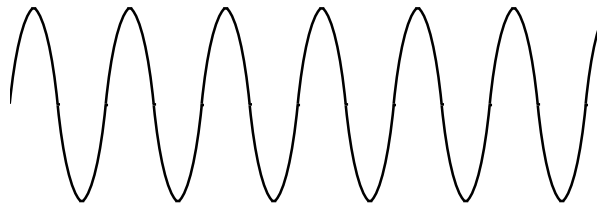


Figure 5 — A carrier signal without modulation

The amplitude-modulated signal has three basic components: the carrier, its upper sideband, and its lower sideband. When audio signals are added to an AM signal, the carrier frequency remains at the exact frequency of the radio signal.



Figure 6 — A carrier signal with modulation

The two audio signals, known as the *upper sideband* and the *lower side-band*, appear on either side of the carrier. The



upper sideband audio signal appears above the center of the carrier, and the lower sideband audio signal appears below the center of the carrier. As a result, if you tune your radio to the center of an AM radio station, the audio often won't be as strong as if you tune slightly to either side of the center.

Sidebands. If you look at one of the sidebands on an oscilloscope (a video presentation of signal shapes), it will look quite a bit like an actual voice signal. In single-sideband (SSB) radio transmission, the carrier and one of the sidebands are filtered out of the AM signal and eliminated. All that is transmitted is one of the audio sidebands.

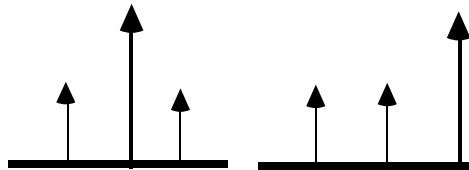


Figure 7 — All the energy is concentrated in the upper sideband (righthand diagram)

SSB transmission is important for two-way communications in the HF band. All of the power that once was used to amplify the carrier and two sidebands in an AM transmitter can now concentrate in the remaining single sideband. And now the SSB transmission requires only half the channel width. As a result, an SSB signal sounds almost 10 times louder than an equivalent AM signal. Because of its efficiency, ease of use, and good voice intelligibility, SSB is by far the most-used radio transmission on the HF bands.

The modulated signal moves from the transmitter out through the antenna and into the air. It travels through the atmosphere for dozens or even thousands of miles. When it is received by an antenna, the tiny radio signal passes into the receiver. In the receiver, the signal is amplified, filtered, and the audio deciphered. The deciphered audio signal goes through the same processes described in **Storing and Retrieving Sound**.

Processing Sound Digitally

The sound processing we have discussed so far is called *analog*, a system in which audio and radio waves mimic the sound waves they represent.

Digital signal processing changes analog audio signals into digital impulses, that is into millions of numbers which describe audio signals. The most common example of digital technology is the compact disc (CD). Every wave of sound is converted into binary code (1s and 0s). These numbers are transmitted in such a way that the audio wave is “built” from blocks of these numbers.

One way to think of these wave representations is to draw a mountain on a sheet of paper. That’s the analog signal. For the digital representation of this paper mountain, place the wooden squares from a Scrabble game in rows over top of the paper. With the wooden squares, you can represent the mountain that you drew on the paper, except that the edges of the block representation are blocky, not smooth. In actual digital audio, the numeric building blocks are so tiny that any blocky edges in the digital audio wave are undetectable.

Recording on Compact Discs

Although CD audio isn’t directly related to DSPs in high frequency radio use, CDs do offer a familiar example of digital

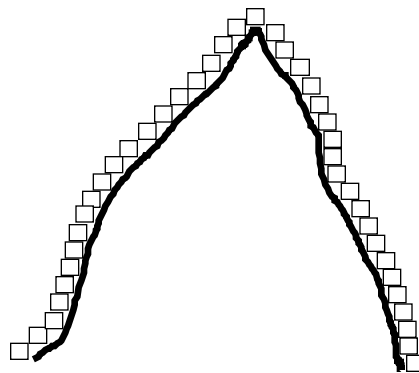


Figure 8 — Drawing of a mountain outlined in game tiles makes a blocky pattern





audio in the home. The music that is to be recorded onto a compact disc—simply a thin disc of aluminum that is encased in a plastic laminate to protect the recording—must be in a digital medium; that is, it must be converted into massive numbers of 1s and 0s. When the disc is recorded, much error-correcting data and system information (like track information and markers) also go onto the disc along with the music. All of this data must be retrievable, so the aluminum disc is etched with minuscule pits. The pitted and unpitted areas translate as the 1s and 0s that represent the data.

In place of needle and cartridge of the analog record player, a laser optical assembly retrieves the audio in a compact disc player. This low-powered laser fires at the tracks of the disc. The unpitted areas of the disc reflect its light back, but the pitted areas reflect almost nothing. This tremendously fast flickering of light is received by a photodetector that changes the light flickers into binary electrical impulses. These are then converted into analog impulses, which can be amplified and converted into sound by the speakers.

Sampling. Of course the analog-to-digital and digital-to-analog processes are extremely complicated—especially when you consider that such things as coding and sampling must also occur in the system. *Sampling* is the process by which the compact disc player retrieves an analog sound, then checks the digital source for its accuracy, then plays another sound. This cycling occurs 44,100 times per second (44.1 kHz), although many players now sample several times more than that per second to make sure that the information being received is accurate and not error-ridden. Such sampling at harmonic frequencies is known as *over-sampling*. Many of the high-cost compact disc players sample up to eight times the standard sample frequency.

Volume. *Relative sound volume* also needs to be considered. Every audio wave-form has a peak-to-peak length (the fre-

quency of the sound), which determines the pitch of the sound, and a height (the amplitude of the sound), which determines its volume. In order for the compact disc player to accurately reproduce music and not end up reproducing all of the frequencies at the same volume, the sound samples are quantified to a 16-bit number between 0 and 65,535. Every tiny piece of audio can be reproduced by the compact disc at any one of 65,536 different volume levels.

Compression. These codes that determine various aspects of the compact disc's sound and technical operations all require a vast amount of information. A full compact disc of approximately 74 minutes requires in the neighborhood of 34 million bits of information to produce. If this information was all held on a standard computer floppy disc, the selection would have to be placed on 48 5.25" discs or 25 3.5" discs. Using a compression code makes it possible for digital tapes and MiniDiscs to be digital and hold as much music as they do.

Conclusion

You have seen how a complex radio carrier wave and its audio signal can be filtered so only a sideband remains in use. And you have seen how audio signals can be converted to digital signals, in such forms as CDs.

In the next chapter, we look at the idea of filters that can make changes in waves—whether those waves are sound waves or radio frequency waves. And in Chapter 3, we look at how digital signals can be processed for radio transmitting and receiving.



The Idea of Analog Filtering

Analog filters are used for a wide variety of applications in electronics. One familiar application illustrates how filters work: speaker crossover networks.

Analog Filters in Audio

Speaker crossovers usually consist of three different types of filters that combine to channel audio to the proper speakers. The typical speaker arrangement comprises a woofer (low-frequency speaker), a midrange speaker, and a tweeter (high-frequency speaker) for each channel of a sound system. Filters make sure the appropriate audio frequencies at appropriate volume reach each speaker.

Crossover Network. The crossover consists of low-pass, high-pass, and bandpass filters at the speaker inputs. Each filter crops out certain frequencies and passes other frequencies.

Woofers. Most woofers are most effective in the several hundred Hz range, so the low-pass filter might be set at 500 Hz. All frequencies below 500 Hz (but little above that frequency) will pass to the woofer.

Tweeters. Similarly, most tweeters are effective above about 4 kHz, so the high-pass filter might be set at this frequency. All frequencies above 4 kHz (but little below that frequency) will pass to the tweeter.

Midrange. Midrange speakers use a more complicated filter—a bandpass filter, which combines high-pass and low-pass filters to set both a high-frequency and a low-frequency limit on the audio that passes through. This bandpass filter would pass all frequencies that were in an audio band above 400 Hz and below 4 kHz.

As a result of such filtering, these speakers produce good-

sounding audio and do not suffer damage from too much power being applied to the wrong speaker.

Cutoff. Some audio enthusiasts say that if the audio is cropped too sharply by the filters, it will sound sterile. So design of speaker crossover filters provides for a more gradual filtering. The low-pass filter, for example, does not cut off all audio at exactly 400 Hz. Rather it will gradually cutoff the audio over the course of several hundred Hz or more, passing everything below 400 Hz but gradually attenuating audio above 400 Hz.

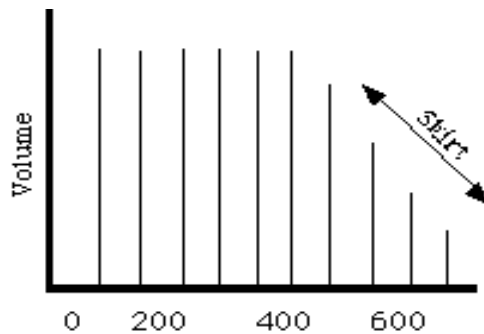


Figure 9 — This low-pass filter gradually attenuates frequencies above 400 Hz. Above 400 Hz is its “skirt.”

This slope of audio that is being attenuated by the filter is known as the *skirt*, which describes that slope in a graph of the filtered frequency.

Analog Filters in HF Radio

Standard HF radio filters are tunable bandpass filters. Bandpass filters trim off the upper and lower frequencies and pass signals within a certain range. The effect of a bandpass filter in radio is like the combination of a low-pass filter and a high-pass filter that passes audio to a midrange speaker. Unlike crossovers, the radio filters should have as close to straight skirts as possible. If they have wide skirts, audio from adjacent stations and noise from outside of the





radio signal will intrude on the tuned signal.

Therefore, radio bandpass filters are much more than a combination of low-pass and high-pass filters. With high-pass filters, one side of the skirt can easily be tuned; with low-pass filters, the other side of the output can be adjusted. Because the boundaries of these filters are not separately tunable, adjusting the values of the components in the bandpass filter will affect both sides of the filter's response.

Aside from the skirts of a filter's output wave form, the other components of this wave form are the area between the skirts—the pass band—and the area where no signal passes through the bandpass filter—the *stopband*.

Symmetry. Another principal characteristic of bandpass filters is that of *symmetry*. Drawing a hypothetical line down through the center of the bandpass waveform helps to see the symmetrical shape of the output (just like the skirts help to describe the filter characteristics).

To achieve a more symmetrical filter, most bandpass filters combine several bandpass filters. The wave forms of these filters mix together to form a composite passband wave form. As a result, these complex filters have virtually symmetrical outputs.

The ideal passband from a bandpass filter is a square wave in which nothing can be heard on either side of the passband, and the response across the top of the passband is straight and unattenuated.

Crystal filters. In order to improve the characteristics of passband filters, mechanical elements are often used instead of the traditional combination of capacitors and coils (inductors). Because of lower cost and better performance compared with capacitance-inductance bandpass filters, quartz crystal filters are often used in HF transceivers and communications receivers. The crystal filters are capable of steeper skirts than the standard inductor/capacitor filters, and they

also have more consistent quality.

Although analog filters are generally not variable at all, some of the older receivers had a “Crystal Phasing” control. This control was merely a tuning capacitor in the crystal filter which enabled the user to alter the shape of the band-pass wave form to reduce nearby interference.

Mechanical filters. A more dramatic improvement, which is covered further in the next section, is the mechanical filter. Mechanical filters, similar in design to crystal filters, use metal elements instead of quartz crystal elements. Mechanical filters are capable of much better characteristics than the crystal filters—steep skirts, nearly flat passband, and sharp stopband. But these filters are expensive to design and construct.

HF filters in practical applications

Communications receivers and modern-day transceivers must have several different filters. The filters allow the receiver to pass a certain band through the radio and to the speaker.

Wide bandpass. For a strong, high-fidelity AM signal, such as from some shortwave broadcast stations, a very wide (8 to 15 kHz) filter will allow you to enjoy the audio to its fullest. However, a wide filter such as this will permit adjacent-channel interference to pass through and will allow static to distort the signal.

Medium bandpass. So, for average AM broadcast listening, a medium-width filter (between 4 and 6 kHz) is best because it will keep out the static and interference, but will allow enough audio to pass through to be somewhat pleasant.

Narrow bandpass. For the narrow-width SSB voice signals, a filter only 2- or 3-kHz wide is usually used. The audio quality is fair for SSB, but is rather poor for listening to an





AM broadcast (the AM broadcast will sound “muddy” and will be difficult to decipher). For extremely narrow digital modes (such as Morse code), the filters used are typically between 0.1 and 1 kHz wide. At these widths, it is difficult to understand any voice communications; very little audio can pass through, except for the dots and dashes of Morse code.

Conclusion

You have seen how analog filters can make changes in waves—whether those waves are sound waves or radio frequency waves—to improve high fidelity audio performance and to improve radio reception by excluding unwanted frequencies and static. In the next chapter, we look at how digital signals can be processed for radio transmitting and receiving.

DSPs in HF Communications

Digital transmissions are nothing new. Morse code, which is a binary alphabet (dots and dashes instead of 1s and 0s), is approximately 100 years old. Another technological development that people assume is recent, facsimile transmission (FAX), had been successful in radio transmission nearly 70 years ago. But the high cost of technology made fax machines infeasible until the advent of the personal and business telephone-based fax machines in the 1980s.

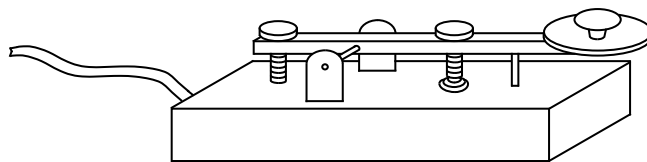


Figure 10 — Morse Code sending key

Like binary codes and facsimile, DSP has existed in theory since the early 20th century. DSP manipulates a digital signal. A box that digitally alters the acoustics of a symphony recorded on CD is a type of DSP. Equipment that digitally eliminates the time-delayed echo in telephone lines is another type of DSP.

Whatever their application, all DSPs use many of the same DSP microprocessor “chips.” The differences between the applications aren’t the DSPs alone; rather they are in what we program them to do. So the general category of DSP is extremely broad.

DSP Flow Chart

The flow chart of every basic application in which DSP is used is the same. An analog signal (either audio or video) enters the digital section of the equipment.





Sample and Hold. The first stage of the system is the *sample and hold*. The S/H circuit samples the signal and holds each sample briefly, for example the amplitude of the incoming signal at a specific time.

In the typical CD player, the sampling frequency is 44.1 kHz, which means that the amplitude of the incoming audio signal is sampled 44,100 times per second! The sampling rate for CD players is high because high-quality audio is more complex than telephone or HF communications, where the fidelity is often deliberately reduced to make the signals both easier to understand and more efficient. In these systems, the sampling rate will often be as low as 8 kHz.

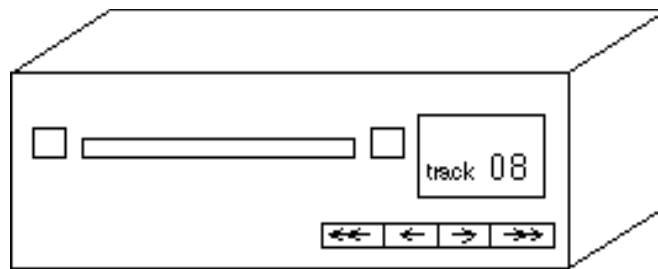


Figure 11 — A home CD player

The basic guideline for determining the sampling rate is that it must be at least twice the greatest frequency that you expect to reproduce. So, if the maximum frequency of the CD player audio is 20 kHz, two times this frequency (40 kHz) will still fall well within the “two times” guideline. For the 8-kHz sampling rate of the telephone system, you can expect that the highest frequency that can be reproduced is 4 kHz (near the top of the spectrum for the average voice frequency).

Analog to Digital. At the next stage, the analog-to-digital converter (ADC), the millions of tiny audio “slices” from the sample-and-hold circuit are converted into binary numbers.

ADCs operate in a variety of ways; some count with a “staircase” generator while others convert the analog voltage into a digital value with multiple comparators. The quality or usefulness of an ADC can be determined by its accuracy, complexity, and speed.



Several other methods for converting the data also exist; choice in methods depends on whether you want low cost, high-speed processing, or the ability to process massive amounts of data. The ADC selection is an important consideration at this point, but as technology advances and the prices decrease, it will become less a factor.

DSP. The actual DSP stage is next in the lineup. This chip—really a central processing unit commonly called a “computer chip”—might be programmed as a filter to reduce noise in a system, it might be programmed to produce or eliminate audio echo, it might be used to clarify a video signal, or it might be programmed to do any one of numerous other tasks.

Digital to Analog. The next stage of the DSP system is another that is used in standard digital audio applications, the digital-to-analog converter (DAC). The DAC does the same things as the ADC, only backwards. Its measures of quality (accuracy, complexity, and speed) are also the same as for the ADC. Like the ADC, it can also use a number of different methods to accomplish digital-to-analog conversion. In one type, the DAC counts digital pulses to determine the analog output. Others use such techniques as voltage or current conversion and oversampling to achieve the output. Like ADC converters, the problems in using DAC chips should decrease as the circuits become more complex and less expensive.

Low-pass filter. The output of the DAC is blocky waveform that would look like the Scrabble block mountain from earlier in this book, so that it is sometimes called a *staircase waveform*. Here the last section of the DSP (a low-pass filter) is used: it smooths out the rough stairs in the waveforms. This process sounds simple enough, but sometimes five or more different analog and digital stages are used in some smoothing filters.

DSP Evolution

Experimental use of DSPs in one form or another was occurring in the 1950s and 1960s. However, because of the enormous cost of early computers, this research was limited to large university and government research facilities. In the 1970s and 1980s, DSPs began to break away from the university and government centers, moving toward high-powered personal computers with central processing units, such as the Intel 8086 and 8088 semi-conductor chips.

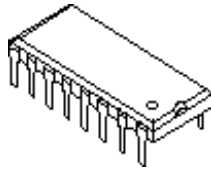


Figure 12 — A semi-conductor “chip”

Because the manufacturers of semiconductors realized the potential for DSP, they began to create specialized DSP chips that could perform signal processing faster and more efficiently than standard microprocessor chips. Today, companies such as Motorola, Texas Instruments, and Analog Devices have several hundred variations on their DSP chips, for differing applications and budgets.

As the technology of computer and DSP chips has increased in sophistication and the prices have dropped, several innovative companies have developed DSPs for use in different aspects of HF communications.

DSPs in Transmitting Applications

A number of advances in transmitter design and efficiency in HF communications have made use of DSP technology, but they do not have the same dramatic effect in cost or performance that DSPs make in receiver filter applications.

DSPs in Speech Processing. The speech processor in one transceiver is heavily intermeshed with its method of SSB modulation. This transceiver uses a system of low-pass and



high-pass filters to reduce the bandwidth of the voice signal and make the transmitter more efficient. The high-pass filter is adjustable so that the operator can choose from several different selections. This filtering will slightly alter the sound of the voice (make the voice sound stronger or tinier) and possibly help it cut through the static a bit better.

DSP in SSB Generation. One transceiver uses direct modulation to impose audio on the transmitted signal. Rather than using an analog filter to remove the unwanted sideband when creating a single-sideband signal, the transceiver uses a DSP. Because digital audio is mathematically based, its timing is almost perfect—perfect timing and audio control being essential for phase-based work.

DSP in Phase Delay

Aside from use in applications requiring delay, a phase-delay system can also entirely filter out a signal. This system eliminates a signal by adding another.

Out-of-Phase Signal. Because every audio signal has a positive cycle and a negative cycle, phase-shift sideband elimination works by inserting a duplicate of the original signal at exactly the opposite phase, a signal at exactly the same amplitude of the original. While the signal is in the positive cycle and its exact duplicate is in the negative cycle, the two waves cancel each other out and no signal remains. Because of its exactness, this system is much more precise than analog phase shifters, which sometimes allow trace amounts of the other sideband to remain in the signal.

Phase Shifting Networks. After the remaining modulated signal is limited in bandwidth by a low-pass filter, it runs through several phase-shifting networks to produce an SSB signal free from noise outside the band of voice frequencies. A digital filter suppresses the carrier so that only the SSB modulation wave passes out.



DSP in CW modulation

CW (Morse code) modulation is simple: turn the transmitter on; turn it off; turn it on; turn it off. Typically, the action of the transmitter being keyed on and off skews the waveform.

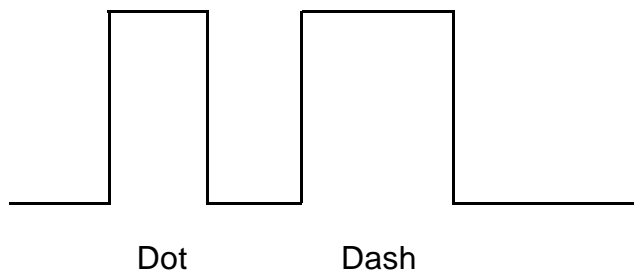


Figure 13 — “Perfect” CW modulation

A perfectly modulated Morse code signal would be a set of square waves. The beginning of the wave would rise instantly, stay steady for a length of time (determined by either a dot or a dash), then drop sharply down. Some Morse code transmitters “click” while in the CW mode, as a result of an improper waveform. This DSP system can eliminate any clicks and any other peculiar sounds that improperly modulated CW signals can make. The result is perfectly shaped Morse code. Of course modern transceivers in good working order rarely suffer noticeable modulation (or key on/off) problems.

DSPs in Receiving Applications

Because transmitting a powerful signal is only half of the game in HF communications, the real differences are these factors: patience, “good ears,” a great receiver, and an excellent antenna. DSP can’t help much with “good ears,” but it can dramatically improve the quality of a receiver for the operator who has been straining through the static and heterodynes for several hours.

Standard DSP filters. DSP filters in HF communications equipment are standard bandpass filters that pass a certain segment of the radio band through the radio into speakers or

headphones. These filters offer lower cost and improved flexibility over mechanical filters.

Analog. In analog filters, a number of standard parts, configurations, and equations determine the values of components. Analog filters take more of a hands-on approach to electronics; the electronics designer and user can actually see the effects of the work. The components have a direct impact on the electrical signals that pass through them.

Digital. Although digital filters are modeled after analog filters, and although their characteristics are based on analog filters, the design and applications of digital filters are entirely different. Digital filters employ a specialty DSP chip for each of the filtering functions. Instead of using separate components to control the filter functions, the bandpass filtering and other accessory functions are all controlled by programming instructions and equations in the chip. Rather than substituting parts for better performance (as in analog filter design), the digital filter designer programs better equations and instructions into the chip.

As a result, equations control and alter the binary numbers that pass through the DSP chip. The end result is that the numbers are converted back into tangible audio signals, which have been altered during the earlier binary numbers stage. In this respect, digital filter design is much more theoretical in approach than is analog design.

Programming. Because of the difference between analog and digital filter construction, the digital filters depend more on good programming than on good quality components. Of course, the circuits must be solid, but there are few differences between the important components in various digital filters—a filter could easily be changed from excellent to ineffective by merely changing its programming.

Because digital filters are both created and limited by their instructions, they can also be changed to anything, according





to their instructions. As a result, adjusting a variable resistor can continuously change the width of a bandpass filter. Also, changing some of the parameters within that filter changes some of its characteristics. This flexibility means that for less than the price of one good mechanical filter, a DSP company can develop the equivalent of dozens (or possibly hundreds) of different filters.

Continuously Variable DSP Filters. Until recently, filters have been single bandwidth (except for slight alterations in response from crystal phasing control). With the advent of real-time digital filters, the bandpass frequencies can now be changed in width, depending on the operator's particular receiving needs.

JPS Communications has developed a process for HF filters that is known as *dynamic peaking*. Like any DSP system, the received signal is constantly being sampled by the sample-and-hold portion of the analog-to-digital converter. But, in JPS's design, the DSP also works as a filter *while* it is monitoring the width of the signal that is being received. If the signal is narrow, the sample-and-hold checks it and automatically narrows the filter width. If the signal becomes wider, the sample and hold checks it and automatically widens the filter response so that the signal can easily be heard.

This sort of "smart filter" obviously depends on fast sampling times and accurate filter software. If the DSP was based on a slow-sampling DSP or on one of the older chips that didn't work in the real time, then the DSP would sample the signal and noticeably change the bandwidth at a point after the bandwidth of the signal had narrowed or widened.

As a result, if the DSP hardware reacted slowly, the received signal would be occasionally cut off at the beginning of words (because the bandwidth would still be narrow from the preceding pause) or it would be laced with bursts of

interference (because the bandwidth would still be wide from the preceding speech). Similar problems would occur if the software for the devices was even slightly inaccurate.

RF Attenuator. Some portable and modern solid-state receivers feature *RF attenuators*. (*RF* is radio frequency, the signals that your transceiver receives; *attenuation* is the weakening of signals.) Solid-state radios are prone to *overloading* from strong signals.

A strong signal will saturate the circuits which separate the audio signal from the carrier, causing that signal to be heard on several or possibly many frequencies. As a result, RF attenuators are used to decrease the strength of the signals into the radio. With the digitized audio of a DSP, this function can easily be programmed into the chip.

(RF attenuators can be handy if your receiver is a block away from another amateur operator who operates at the edge of the legal limit. Otherwise, if the transceiver *really needs* the RF attenuator for typical service, you might want to look into purchasing a transceiver with a better front end.)

DSP Filters: High-pass, Low-pass, Bandpass

High-pass, low-pass, and bandpass filters are often used in HF transceiver antenna input circuits for two purposes: to prevent strong out-of-band signals from saturating or overloading the receiver's front end and being heard throughout different bands; and to prevent adjacent-band signals from splattering over into other regions.

High-pass Filters. The most common high-powered local radio stations would be those in the AM broadcast band. The HF band is higher in frequency than the AM broadcast band, so these image signals could all be virtually eliminated with a high-pass filter. For example, if you live near a 10-kW AM broadcast station, you might have problems with hearing





that station as an image throughout the HF spectrum. If you are attempting to hear a weak signal, this overload can destroy your ability to hear the wanted station. A high-pass filter with a cut-off frequency around 1700 kHz can prevent these AM broadcast stations from interfering with the short-wave frequencies.

Low-pass Filters. U.S. television channel 2 at 55 MHz is just above the HF frequencies so these image signals could all be virtually eliminated with a low-pass filter. (In some parts of the world, television broadcasts at a frequency as low as 45 MHz.)

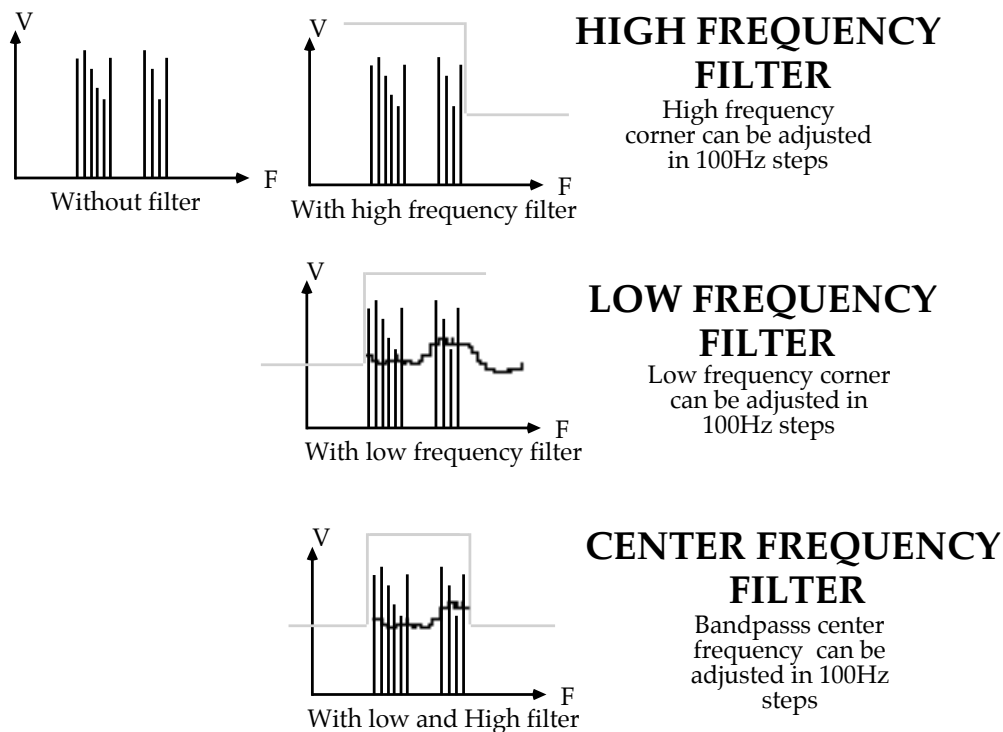


Figure 14 — Filters permit setting audio quality to personal preference

Bandpass Filters. Low-pass and high-pass filters are used less often than bandpass filters, however, to lock out unwanted signals. High power shortwave broadcast stations are on the air throughout the world. In the United States, 50-kW AM stations and 1 MW television stations broadcast at the edges of the amateur bands. In amateur radio transceivers, the typical solution would be to make the bandpass filter run from the bottom edge of the amateur band to its top edge. All tres-

passing signals would be virtually eliminated. Consequently, bandpass filters have become the mainstay of DSP use in HF equipment.

Of course, all of these filters to eliminate strong image signals in the receiver can be programmed into DSP chips. And because the cost for these extra filters is so low, they can be included in modern receivers—even though they were too expensive to be included in most earlier receivers.

Notch Filters

In HF communications, notch filters serve to eliminate nearby sources of interference. Notch filters are also known as *band-rejection filters* and *band-elimination filters*, names that provide an insight into their inner workings.

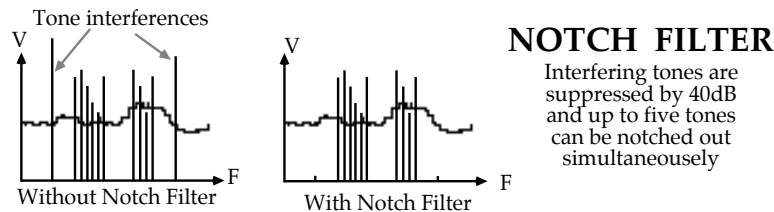


Figure 15 — The SGC Notch filter can suppress up to five tones at once

Instead of passing a tiny segment (or even a large segment) of the band through and rejecting all other signals, the notch filter rejects a tiny segment of the band and allows all other signals to pass through, unattenuated.

Band Interference. A notch filter can eliminate some interference within the band. A radio signal might be overwhelmed by Morse Code interference, but a notch filter on an analog receiver can tune out some of the interference. On many analog receivers, the notch filter settings provide little improvement. And even an excellent notch filter can reduce the interference of only one signal. The notch filters in the DSP designs often perform amazingly. Rather than just blocking out a nearby band segment, they act as true “killers” of whistle or heterodyne interference.





Heterodyne Interference. A heterodyne is a shrill tone that is caused when radio signals overlap. In the amateur bands, where nearly anyone can transmit nearly anywhere, heterodynes can cause a real problem, especially in the 80- and 40-meter amateur bands, where shortwave stations broadcasting in the AM mode can be readily heard. The mixture of AM and SSB signals results in an amateur band riddled with heterodynes. Heterodynes are not only unpleasant to listen to, but they can ruin an operator's ability to hear a signal; as a result, heterodynes have been one of the plagues of radio communications since its creation.

DSP notch filters are effective against heterodynes—most can be entirely eliminated. More importantly, they can eliminate several heterodynes at the same time. The DSP notch filter chip is programmed to eliminate all constant or slowly varying tones present in receiver or transceiver audio. In this sense, they behave differently than typical notch filters. If the digital notch filter can eliminate one of the worst enemy of the HF communications user, the heterodyne, we wonder what other miracle it can achieve—eliminate fading?

Digital AGC

Automatic gain control (AGC), also known as an automatic level control (ALC), is especially important when receiving wideband modes, such as AM, that are susceptible to fading. Because of fading, signals will quickly rise and fall in level.

AGCs level out only the amplitude of the signals that pass out of the receiver; therefore, they can easily be programmed into DSP chips. Because the technology for analog AGCs was already solid, the only real benefit of digital design is to save money in applications where a DSP chip is already being used: using a DSP simply for a digital AGC would be expensive.

Conclusion

You have seen how DSP has transformed the quality of HF communications in both transmit and receive. Next, we will look at available equipment which features DSP.



Available DSP HF equipment

As DSP technology is beginning to reach the marketplace, DSP products are finding their way into HF communications.



Figure 16 — SGC's DSP products: PowerTalk, SG-RM remote mobile head, and PowerClear.

DSP Transceivers

A number of transceivers currently on the market offer digital signal processing. This book, coming from SGC, designer and manufacturer of HF communications equipment, has been setting the stage for this biggest technological advance in two-way communications since the use of the SSB mode and the development of the single-unit transceiver.

SGC's SG-2000 PowerTalk. Presently, the equipment that



Figure 17 — The PowerTalk Transceiver

uses the state-of-the-art DSP filtering technology is the SG-2000 PowerTalk.

The SG-2000 PowerTalk offers the most DSP features for the lowest price. Some of the key DSP-related features of the SG-2000 PowerTalk are these:

- ADSP™ noise-reduction system
- SNS™ noise-reduction system
- First mobile/base HF transceiver with DSP
- First HF DSP system with visual display
- DSP filters can be programmed into separate memories
- Notch filter
- Eight preset DSP filter positions
- Variable high-pass, low-pass, and bandpass filters.
- Separate control head makes upgrade from SG-2000 to SG-2000 PowerTalk simple and inexpensive

ADSP™ noise reduction. ADSP (Adaptive Digital Signal Processing) is a particularly effective type of noise-reduction system to filter out unwanted noise in any signal being received. The DSP algorithm is “smart” and can “see” the dif-

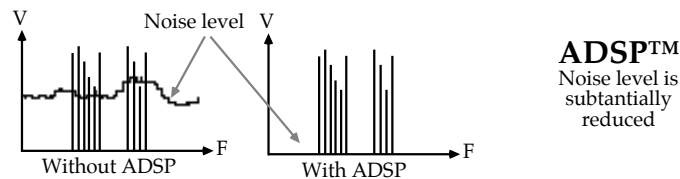


Figure 18 — SGC's ADSP substantially reduces noise level

ference between the signal being received and the accompanying white noise and static crashes. Then, it separates the two and passes only the received signal to the speaker.

SNS™ noise reduction. SNS (Spectral Noise Subtraction) is





revolutionary DSP noise reduction used only in the SG-2000 PowerTalk and in one of the DSP “black boxes.” Instead of the traditional method of filtering whereby signals are passed through a bandpass filter with a concrete shape, the SNS system acts more like a continuously variable bandpass filter.

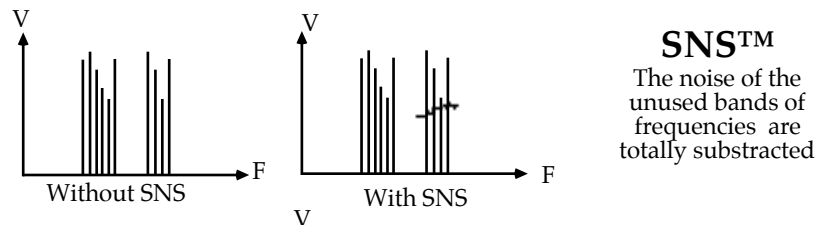


Figure 19 — SGC’s SNS subtracts spectral noise

With SNS noise reduction, the filter basically collapses against the radio signal (either voice or data). As a result, the receiver (and any interference during that audio) remains, but the noise between the bits of audio information is eliminated. (It’s a little like Dolby processing for high fidelity recording.)

First mobile DSP transceiver. Compared with other DSP transceivers, the SG-2000 PowerTalk is small (4.75” x 10” x 15”), light (12 lbs), and made specifically for 12-volt operation. On the road, on a boat, or on a DXpedition, where the conditions are much less than ideal, you will especially notice the benefits of the DSP functions.

Visual DSP filter display. None of the other HF DSP filters show you the exact settings of the filters. In a few cases, adjustable filters are controlled with rotary knobs with the increments marked around them.

In the SG-2000 PowerTalk, the filter positions (from 300 to 3000 Hz) are adjustable (in 100-Hz steps) and each step is displayed as an LED on the front panel. With this LED display system, you can immediately see the width and the exact frequency coverage of the filter that you are using at any given time. This system is particularly useful if you need to dial between many different frequencies and if the signals are of varying strengths and characteristics.



Programmable digital filters. Wishing to contact a station on a regular basis, you might find that a certain filter setting works well day after day for listening to that station. For your convenience, you can preset this filter setting into the radio memories (along with six other favorite filter settings). With a push of a button, you can immediately place the SG-2000 PowerTalk in your favorite filter position.

Pre-programmed filter settings. In addition to the enormous array of filter settings that you can create, eight standard settings are preprogrammed into the memories. Some of the most common of these positions are marked with LEDs for extra convenience.

Notch filter. The notch filter can locate and eliminate as many as five heterodynes at one time—many more than you will probably ever need to use!

Variable Bandpass, low-pass, and high-pass filters. The SG-2000 PowerTalk has variable bandpass, low-pass, and high-pass filters. These filters are one of the contributors to good radio reception. These accurately displayed, excellent variable filters could easily make the difference between a copyable signal and an unreadable signal amidst the noise.

Upgradable DSP head. Instead of buying a new transceiver for the DSP functions, you can simply purchase the SG-2000 PowerTalk head and place it on the SG-2000 transceiver case. Doing so could save you thousands of dollars over upgrading to a new PowerTalk transceiver.

Other Advantages. In addition to the DSP advantages of the SG-2000 PowerTalk, this model also has a number of other advantages:

Removable Head. Unlike other HF transceivers, the entire face plate (“head”) of the SG-2000 can be detached and used to operate the transceiver from remote locations—or in tandem with other heads. This feature is perfect for commercial and marine operation, or for club amateur stations where a



transceiver must be controlled from more than one location.

Simple design of front-panel controls. Instead of cramming dozens of tiny knobs and buttons on the front panel, SG-2000 PowerTalk displays only three knobs and a few rows of buttons. Not that the PowerTalk lacks features, but rather that it is so well designed that fewer buttons accomplish the same functions.

Even the DSP section of the PowerTalk—which features custom DSP memories, preprogrammed filter memories, a notch filter, a noise reducer, the SNS noise reducer, variable low-pass, high-pass, and bandpass filters, and a bypass function—requires only nine buttons. On the simplified panel, the buttons are large and spaced widely apart—there's little chance that you will misprogram the PowerTalk head. This simplified design is significant when you compare the SG-2000 Power-Talk with the many-knobbed alternatives.

High-power/small package. In spite of having the most flexible and highly developed DSP unit in any transceiver and being one of the highest-powered transceivers available (conservatively rated at 150 watts), the SG-2000 is small. As mentioned earlier, the SG-2000 PowerTalk is a mere 4.74" x 10" x 15" at 12 pounds. You get everything in a package that you can take anywhere.

Tested for high quality. No other transceivers advertise their testing procedures as SGC does. After it has been manufactured in the United States using high-quality components, every SG-2000 is factory-aligned. Then, each rig is keyed at full power into an open antenna for 10 seconds, then into a shorted antenna for another 10 seconds. Next, it is keyed for 24 straight hours in full-power CW. Each SG-2000 is then keyed on and off at 10-second intervals for 24 hours.

Finally, each SG-2000 is re-evaluated and all functions are verified to ensure that performance meets specifications.

After the SG-2000 passes these difficult tests, it may leave the factory. As a result of this quality, the SG-2000 is one of

the few amateur transceivers that is also type-accepted for commercial and marine service.

The bottom line is that the SG-2000 PowerTalk is one of the best-constructed, most flexible, most advanced, highest-powered, and easiest-to-use transceivers on the market. And the list price is just over half the price of the only other DSP transceivers.

Add-on DSP

Because DSP technology has become much more affordable, a number of different manufacturers have developed external DSP boxes to serve many of the same purposes as built-in DSP. Instead of connecting inside the radio, they connect between the headphone audio output jack and the headphones. All DSP conversions and alterations occur after the audio signal has passed out of the receiver. This makes DSP use and installation quite simple.

One of the major markets for the black boxes is radio amateurs. Combine these two features and you can assume that the target candidate will be a contest-entering amateur who is busily digging out weak, static-laden SSB and CW signals from the far corners of the world. Because the filters are intended for such difficult situations, they are typically narrow and effective for poor signal situations and not for solid, high-fidelity signals. Fortunately, the manufacturers of these boxes include easy pushbutton switches so that the filters can be quickly punched in and out.

Basic Features. Even when the DSP programming varies among the basic bandpass filters, the results are essentially the same. Because these filters create a square-wave filter response, most of the filter responses of the equipment on the market are good, and differences among them are slight. Although the boxes might vary in the number of features that it supplies, each box includes at least one of these three major features:





Variable bandpass filters. These filters, discussed in the book, are the key to DSP benefits in HF receiver design. Most of the digital filter “black boxes” are intended for use in high noise/weak signal conditions.

Notch filter. Notch filters are included in nearly every DSP black box; in fact, one of the DSP black boxes is solely a notch filter. Although some notch filters on the market are more effective than others, the most effective models are worth the price of an entire DSP filter unit for amateurs who regularly operate in the crowded 80- and 40-meter amateur bands.

Noise reduction. Unlike the different DSP bandpass filters on the market, the DSP noise-reduction techniques vary greatly. Unlike bandpass filters, which must come to a specific outcome, an engineer can take a wide variety of different routes to attack noise. Because of such differences, DSP black boxes vary in their effectiveness and even in the types of noise that they succeed in eliminating.

Advantages and disadvantages of DSP add-ons. If you plan to use DSP in conjunction with a transceiver, you could save some money by keeping your old transceiver and purchasing one of the DSP boxes. It’s less expensive than buying a DSP transceiver. You could greatly upgrade the capability of an old, out-moded transceiver by doing so.

However, none of the black boxes has a digital readout and adjustable high-pass, low-pass, and bandpass filters. (The DSP in the SG-2000 PowerTalk and in the PowerClear is arguably the best DSP unit that you can find anywhere.)

The DSP boxes are fine for fixed installations, where dozens of little accessories are stacked around the transceiver, but forget it for mobile operations. A DSP box sliding off of the dashboard or onto the deck of a vessel would be annoying. Also, the DSP boxes are active devices and they require power; either a 12-volt battery or an extra power line would have to run through the boat or vehicle for the separate DSP

box/transceiver mobile configuration. (With the SG-2000 PowerTalk, it's all built in.)

SGC's Add-on: PowerClear

SGC has just entered the market with its own “black box” (it's actually gray), the PowerClear. Offering all the audio DSP features of PowerTalk, it can be used with any radio (HF, VHF, UHF) or any voice and data link system, even noisy telephone lines.



Figure 20 — PowerClear— standing 3.65” high

It weighs 20 ounces and stands a mere 3.65” high, 6.65” wide, and 1.93” deep, unmounted. And yet it offers ADSP and SNS and memory features of PowerTalk, plus a built-in speaker, speaker jack, headphone jack, and volume control.

The built-in speaker permits the PowerClear to be used as an audio amplifier as well as a pre-amplifier. And the printed circuit board contains a larger number of components in a smaller space by means of four layers of circuits built into a single board. That, combined with “surface mounting” of components, permits a more dense packaging of components for more efficient use of space. That's how SGC has managed to make its PowerClear small but powerful.

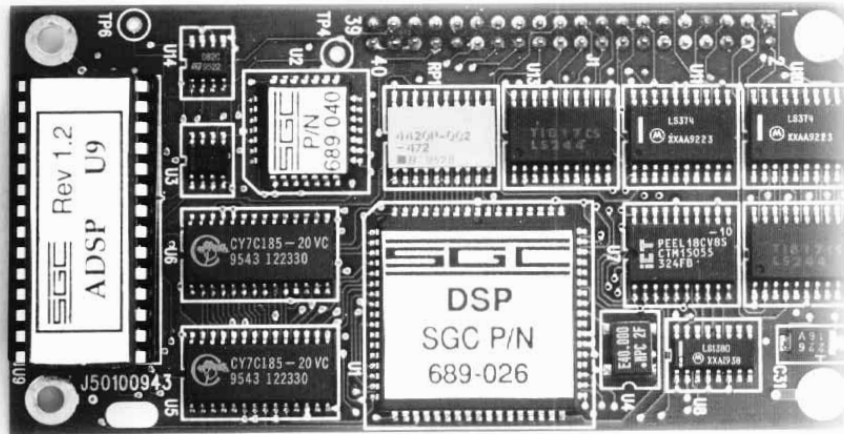




Using DSP HF Equipment

Digital filters and other types of digital processing are powerful but must be used, like any tools: where they are most effective. Misusing DSP technology could hinder rather than help reception.

Operating. If you would normally prefer a high-fidelity signal with a fair amount of noise to a muffled, low-fidelity signal with almost no noise, you would usually keep the filters



**Figure 21— Printed Circuit Board—
the Heart of PowerClear**

about as wide as you could stand. Because of their focus on noise reduction and tight filters, the DSP add-ons are often most effective in high-interference, weak-signal conditions.

Operating with DSP. Keeping DSPs switched out while listening to a station or net, you can punch in the filter and/or the noise reduction if the signal is a bit difficult to copy or if it is being degraded by a noise source. Sometimes DSP can reduce interference enough to significantly improve the understanding of a signal. Unfortunately, the filters are so narrow that they make general listening unpleasant. For receiving weak broadcast stations with some DSPs, the listening might even be a bit painful after a an hour or so—even if the DSP was effective.

The notch filter in some DSPs is unnoticeable until a hetero-



dyne is encountered; the filter wipes out the heterodyne—usually before you can even notice it. As a result, the notch filter is one accessory that can often be left in while scanning.

Unlike the standard passband filters and the notch filters, DSP noise reduction varies considerably from model to model. Some models do little more than reduce the gain of signal. One of the most effective noise reducers is often effective against constant sources of noise, but it also wipes out constant portions of the audio from the received signal.

However, one of the biggest problems with some noise reduction is that it makes the audio pulsate, as if it is coming in waves from the ocean. In many cases, this noise reduction benefits reception, but it does make it sound peculiar—and possibly annoying.

Operating with PowerTalk. Using the SG-2000 PowerTalk is different from using other DSP units. The filters are all digital, so it's not a matter of using or not using DSP. However, the bypass function does bypass the automatic ADSP processing and all of the other functions that you can choose.

To listen for SSB stations, start out by tuning through the bands with the bypass mode selected. If interference becomes a problem, switch out the bypass filter and choose a wide filter setting. For more "firepower," choose the noise-reduction systems only if necessary and try the preprogrammed memories. Try the user-controlled filters when the preprogrammed filters meet with no success.

Conclusion

Digital Signal Processing has arrived in the world of HF single sideband communications. Available in transceivers as well as in "add-on" units, it permits much more satisfying communication on today's crowded frequencies.

Finally, in Chapter 5, we will explore what the future holds for digital signal processing.



The Future of DSP

We have discussed DSP applications in sound and in radio communications. As DSP continues to be improved, it should find new, as yet unidentified applications.

HF Communications

DSP is the future of HF communications, not because the technology is new, complicated, computer-based—or even because this book is produced by a leader in DSP-based HF communication. The system will endure because it can produce better-than-ever results for lower-than-ever prices.

Hobbyists and experts seem to feel that as soon as DSP technology decreases in price, everyone will be using it. By the year 2000, most every receiver and transceiver on the market will use DSPs to improve performance and reduce cost.

New possibilities

But after digital filters, noise reduction systems, notch filters, and AGCs, noise-reduction systems still will require plenty of work, and they will surely improve in the future. And now that digital filters have been perfected, other interesting systems could be investigated. And so we speculate on the future.

Just as DSP converts all of the analog signals to digital data then back to analog signals, adding an interface to one of these pieces of DSP equipment should be a relatively simple task. With an interface, the data could be input to a computer, and once there, it could be used for a variety of applications.

Manipulation. The data could be manipulated by a specialty computer audio or editing program. The sound could either be altered for effect or “cleaned up” through a noise reduction program.





Storage. Data could be stored on a floppy disk or hard drive for perfect, low-cost copies. Also, it would be much easier to find audio clips on a computer disk than on a tape cassette: telling a computer to “go to” a segment instead of fast forwarding through as much as 60 minutes of recorded tape. Digital storage would be important for those who participate in emergency communications or for amateurs who want to save rare relayed messages.

Transmission. Depending on your needs, you might care to upload sound files from the radio to a computer network. For instance, a receiver could be placed in some remote land. To access it, you could link up with the receiver via a BBS or via the Internet.

Digital transmission. Voice and other sound material could be transmitted as digital information, giving up the analog modulation of carriers. (Even now, HF-SSB radio is being used to transmit data—not voice or CW but computerized, digitalized information—input not from a microphone but from a personal computer.) If audio signals can be digitized, they can be transmitted and received via the HF bands.

Because of the noise and fading, there would obviously be some receiving difficulties. However, during best-case conditions, the signals could theoretically sound as clear as an FM broadcast station or a compact disc. The possibilities for high-fidelity audio would be of more interest to broadcasters and program listeners than to two-way HF communications users.

Data to Computers. In the early 1980s, Radio Netherlands transmitted programs for the personal computer over short-wave. At that time, it sounded like a silly use of technology for technology’s sake. But now, computers and HF communications appear quite compatible.

The programs from Radio Netherlands could be recorded over the air to cassette and played back over one of several personal computers. This effort represented simple digital communications.

Other applications. DSPs could be used wherever reception or sound conditions were marginal: listening to long-distance telephone calls, listening to cellular telephones in marginal areas, serving as sophisticated equalizers in recording—or restoring— analog recordings.

Conclusion

The possibilities of DSP technology in HF communications are vast and the future is opening even greater potential. Stay tuned to SGC, the leader in two-way HF communications for the latest.



Glossary

40-meters	A band of frequencies (7MHz to 7.3MHz) with a wave of 40 meters (131 feet) long
AM broadcast band	A band ranging from 530 to 1605 KHz.
Amateur bands	HF frequencies of 1.8MHz to 29.7 MHz set aside for amateur radio operators.
Amplitude	The height of a radio or sound wave—loudness.
Amplitude Modulation	Adding information to an RF carrier by increasing and decreasing amplitude.
Analog	Representing data with physical quantities (a watch with hour and minute hands is an analog time display).
Binary	A system of numbers represented only by digits 0 and 1. (Contrast with decimal which uses digits 0 through 9.)
Capacitor	A device to store electrical energy.
Carrier	An unmodulated RF signal.
Chip	A wafer of semiconductor material used in an electronic circuit.
Copy	When radio operators hear and write down a message, they “copy.”
DXpedition	A contest in which amateur radio operators try to reach distant stations.
Frequency	The number of times per second a radio or soundwave oscillates. (See Hertz.)
Heterodyne	The frequency that results when two radio frequencies “beat” together (one frequency minus the second frequency = heterodyne).
Hertz	See Hz.
HF	A range of frequencies from 3 to 30 MHz.

Hz (Hertz)	A measure of frequency: one cycle per second
Inductor	A coil onto which voltage is imposed by another coil.
KHz	1000 Hertz
LED	Light-emitting diode: a semiconductor that lights up; used in digital displays.
MHz	1 million Hertz
Microprocessor	A computer processor contained on a chip.
Oscilloscope	A display of frequency on a cathode ray tube.
Phase-shift	Removing an unwanted frequency (or sideband) by imposing a mirror-image frequency so the two cancel each other.
RF	Radio frequency—such as a transmitter emits.
Transceiver	Radio transmitter and receiver combined in the same unit.



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