Effective Directivity for Shortwave Reception by DSP

Learn how the author uses two small loop antennas and synchronized receivers with DSP software to reduce or eliminate interfering signals.

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Radio amateurs living in a suburban or residential location are faced with some limitations. The most important limitations are the space for placing shortwave antennas, a high ambient man-made-noise level and the significant possibility of local interference. On the top bands, large directional antennas are hardly an option. Given these preconditions, we can optimize shortwave reception by balancing antennas as well as possible, and by placing them as far as possible from local interfering sources. In some cases, a local interfering source can be attenuated by using the null in the radiation pattern of a small "magnetic" loop antenna. With the help of noise cancelling and a second antenna, a single local interfering source can always be suppressed. Not much more is practically feasible with analogue means in a residential location, however.[1]

How can digital signal processing (DSP) help to improve shortwave reception, whereby we do not wish to limit ourselves to one or more local interfering sources? It turns out that DSP, using time difference of arrival (TDOA), in combination with two active small loop antennas and two synchronized receivers, is capable of creating an effective directivity, which is comparable to large directional antennas.

Frequency Domain Processing

As an example, we will take a close look at the content of an SSB signal, as shown in Figure 1. The bandwidth needed for a voice channel is around 3 kHz. During every fraction of a second however, only a part of that channel is used. There are always only a limited number of frequency components present.

Depending on the speech content in the signal, the channel will contain different frequency components every following moment. Every interfering source, whether local or not, will also contain its own frequency components at any moment. The chance that frequency components of different sources will overlap one another is very small. See Figure 2.

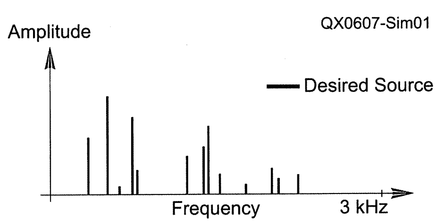

Figure 1 — Limited number of frequency components per unit of time of a single source.

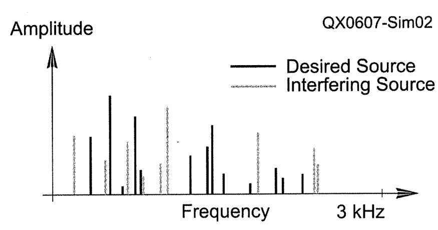

Figure 2 — Non-overlapping frequency components from two sources.

If we are able to select only the desired frequency components, this is comparable with a reduction in bandwidth. Those parts of the spectrum in which we are not interested are cut off — just as with a CW filter, but within the channel. This assumes that each frequency component always comes from just one source. This is of course, not entirely true, as there is noise from the antenna preamplifiers and atmospheric noise over the whole band in variable amplitude. There is always a small chance that more than one source will contribute to a frequency component. We will deal with this later in more detail.

Distinguishing Frequency Components Using TDOA

How can we distinguish the desired frequency components from the interfering frequency components for the purpose of selection in the frequency domain? How do the various signal sources differ?

We can look for a difference in the typical characteristics per type of signal source. CW, SSB and ambient noise have different characteristics. The assumptions one must make for this do not always hold true. An interfering source can, for example, have the characteristics of speech or can even be speech.

A practical and more useable difference is seen in the direction (azimuth and elevation) from which the signals arrive. Direction translates, when using two or more antennas, into time difference of arrival (TDOA) or phase differences and in ratios of amplitudes.

We see the same effective difference whether or not signals are circularly polarized, and the sense of this circular

[1]Notes appear on page 45.

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polarization. We will also see this as a phase difference later on.

We can now choose what we are going to do with each frequency component based on phase difference and/or amplitude ratio. We limit ourselves to discrimination based on phase difference alone. There are two reasons for this. First, because the phase difference mainly provides the information on the direction, and second, because the processing is carried out at the audio, and not at the intermediate frequency level. As a consequence of the AGC, it is not possible to determine the real amplitude ratio. It is mainly due to the AGC that this sort of processing must actually be carried out at the intermediate frequency level.

From Direction and Polarization to Phase Difference

We need a minimum of two antennas to achieve a directional phase difference. Two antennas that are small with respect to the wavelength are a good choice. See Figure 3. For example, active small "magnetic" loop antennas work well (see Note 1). They are wideband, optimally balanced and there is a negligible mutual coupling between both antennas.

Loop Antennas in the Same Direction

It is obvious we are about to use both loop antennas as an array, as we are mainly interested in the direction from which the signals enter. In this array, all antennas must have identical radiation patterns and the polarization must also be the same in all

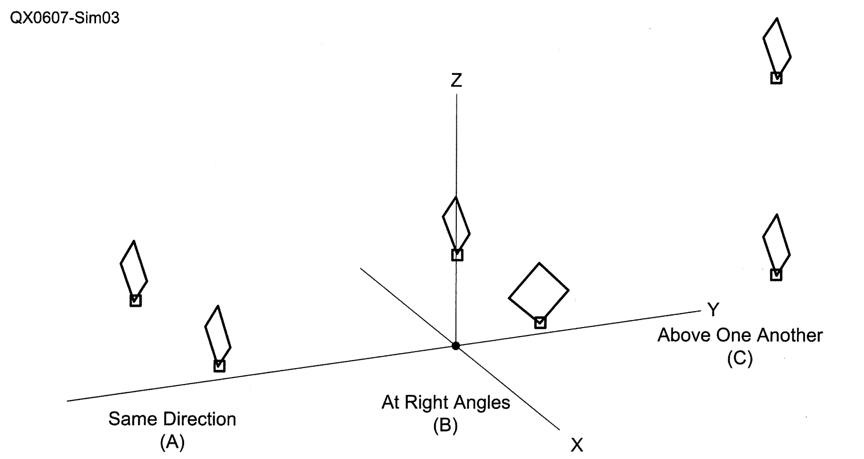

Figure 3 — Three configurations with the antennas a) in the same direction b) at right angles to one another and c) above one another.

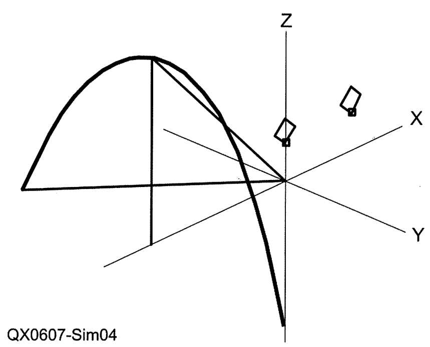

Figure 4 — Two antennas show an arc of directions with the same difference in phase.

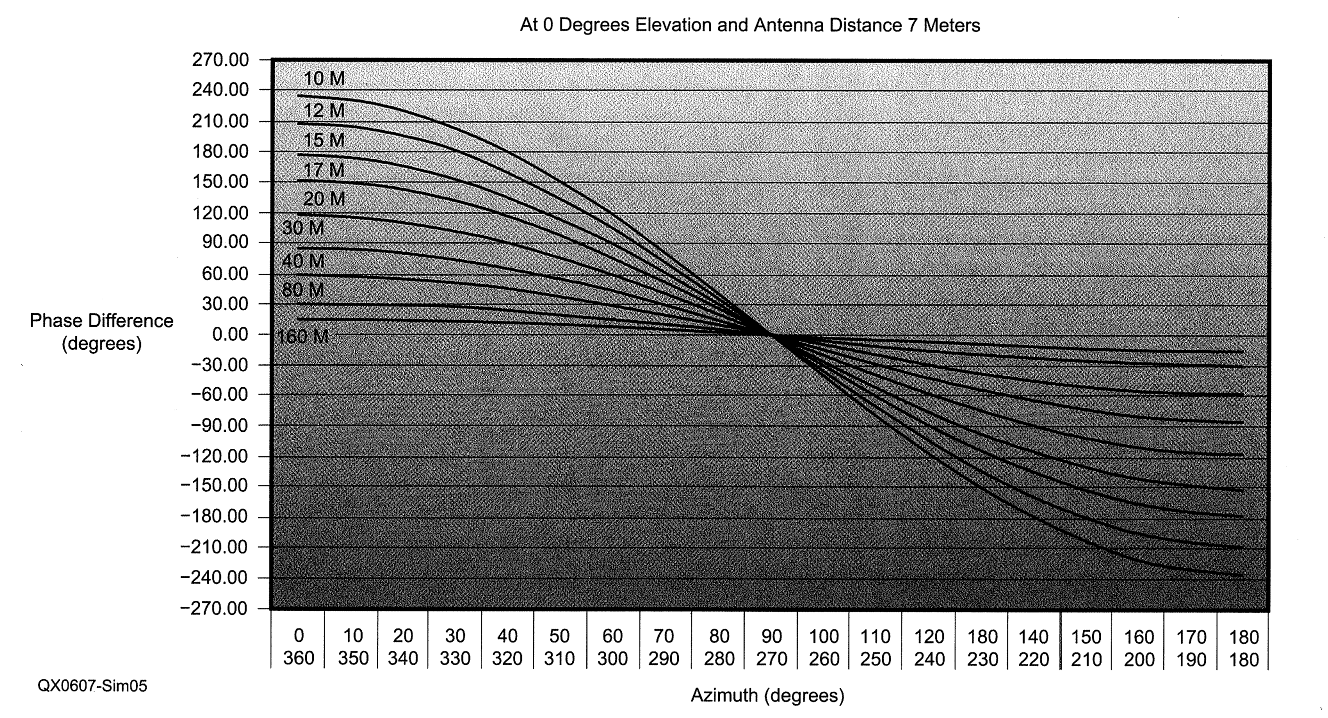

Figure 5 — The phase difference as a function of the azimuth direction for a 0° elevation angle.

directions. The resulting radiation pattern of an array is determined by adding the fields of the antennas with suitable amplitude ratios and phase differences.

In our case, however, we are not going to add the fields, but decompose one field into two components (signals) with their own amplitude and phase. In this specific case, we look at the difference in phase between the signals received by both antennas. This actually concerns time difference of arrival (delay) differences. One antenna receives the signal earlier in time than the other antenna. We may convert this into a phase, as the bandwidth of the channel is small with respect to the carrier frequency.

This phase difference is simple to calculate. We already know the wavelength of the carrier frequency. Now we only have to work out the difference in distance between both antennas for the respective direction. A wavelength corresponds to 360° (a complete sine period). The phase difference is therefore this 360° times the ratio of the difference in distance and the wavelength. Depending on which antenna first receives the signal, this phase difference will be positive or negative.

Remember, the direction also includes elevation as well as azimuth. We get the same phase difference for an arc of direction, as shown in Figure 4.

Each arc is formed by all directions having the same angle with the axis on which both antennas are situated (X-axis). We can therefore make no distinction between signals on the opposite side of this axis. We need more antennas for this. Only in the extension of this axis and at 0° elevation do we find a single point where the phase difference is at a maximum. Figure 5 shows the phase difference as a function of the azimuth direction for an elevation angle of 0° and an antenna distance of 7 m for each band. With increasing elevation angle, these phase differences become smaller. Apart from the phase difference, we therefore also need the elevation angle in order to determine the azimuth direction. We cannot determine this elevation angle with two antennas, and can only estimate this based on the distance to the transmitter.

Loop Antennas at Right Angles to One Another

Here also, we are still looking only at the far field. When both antennas are at a relatively short distance from one another, a wave will arrive almost simultaneously. Ground waves and local interfering sources are linearly polarized. With small antennas, this can only result in a phase difference of 0° or 180°. I will elaborate on this in the "Near Field" section.

Radio amateurs normally do not dwell on it, but sky waves can be circularly (mostly elliptical) polarized. At 160 and 80 meters, this circular polarization can be highly constant and predictable. This leads to phase differences of about –90° or +90°. On the higher-frequency bands, polarization is normally linear and variable in direction. The phase difference wanders unpredictably over the entire 360°.

We can make use of this circular polarization. At 160 and 80 meters, there are now unexpected possibilities, as it is difficult on a residential location for those bands to have sufficient distance between both loop antennas. With the loop antennas in the same direction, the maximum phase difference on those bands is then quite small. In the case of local stations, this is further reduced by the high elevation angle (NVIS). The smaller the phase difference, the more difficult it is to separate the signals. Thanks to circular polarization, and with the loop antennas at right angles to one another, we can still separate local interfering sources and ground-wave signals from desired sky-wave signals.

Loop Antennas Above One Another

It would be a fine thing if we could select the signals based on the elevation angle. Local interference arrives at a very low elevation angle. The same, however, also applies to the true DX signals. Many other signals enter under a steeper elevation angle. These can then be separated from local interference.

This is simple in free space. You place the two antennas at a distance above one another. Depending on the elevation angle, we measure a phase difference.

In practice, the (loop) antennas are generally, with respect to the wavelength l, placed on a low height (< ¼ l) above the ground. The elevation angle cannot be easily determined with two antennas in this condition.

Figure 6 shows the contribution made by ground reflection. We find this picture in Chapter 3 of *The ARRL Antenna Book* and other Amateur Radio handbooks.[2] The wave reflected by the ground appears to have been received or transmitted by a mirror antenna in the ground. Let us assume the ground reflects 100%. Then the mirror antenna receives the same signal strength, but at a distance BC, later. The real antenna receives the total of both signals, the direct and the reflected wave. The net phase is then precisely between the phases of the real and the mirror antenna. This is exactly the phase that we receive at a height of 0 meters (half the distance A – B between the real and the mirror antenna).

The phase we measure is therefore independent of the height at which we place the antenna. In reality, the ground is not ideal and we will measure phase differences. But these will be of no use until we have placed one antenna at a height of ¼ λ or higher. On a residential location, neighbors may find as little as 10 meters to be too high. This option will therefore be mainly applicable for the higher-frequency bands, 30 meters and higher.

Near Field and the Phase Difference

Up until now, we have mainly dealt with the far field. In a residential location, however, we can, for local interfering sources, easily be dealing with the near field. For the boundary between far and near field, a distance of 1/6 λ may be maintained, but this

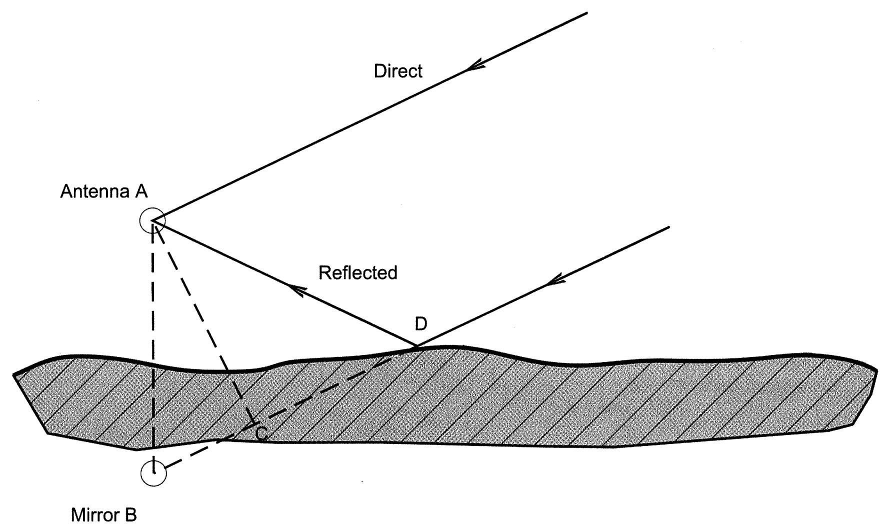

Figure 6 — Phase with low level antennas independent of height as a result of far field ground reflection.

is not a strict limit and must sometimes be set at 1 λ or more. The dimensions from the source with respect to the wavelength are very relevant in this matter. The near field radiation is completely different from that of the far field and the relationship between phase difference and direction is no longer obvious. Unexpected phase differences can occur, which, however, remain constant in time!

With the loop antennas at right angles to one another, this translates into a phase difference that is no longer in the 0° or 180° range. The phase difference can have any value for a large-sized local source. Overlap in phase difference of a local interfering source and a desired source can then occur by chance.

If we set the loop antennas parallel in the same direction, we can see something similar. The phase difference can be larger than we expect based on the distance between the antennas. This is favourable, as the local interfering source and a desired source cannot overlap in this case.

In this context, it should also be mentioned that coupling with objects (such as antennas) in the near vicinity can influence the phase difference. This coupling can normally be minimized by turning the loop antenna(s).

Overlap in Phase Difference

There is a chance that an interfering source gives the same phase difference as a desired source. Both sources can then no longer be separated based on phase difference. This phase difference applies with loop antennas in the same direction in the far field for an arc of directions and the chance of this is thus higher than with a very large directional antenna (beam). See Figure 4. By giving another azimuth direction to the axis on which both antennas are situated, we also give this arc another direction. This can solve overlap in phase difference. A practical solution to this is a third antenna on another axis direction. With interfering sources of relatively large dimensions with respect to the wavelength in the near field, all unexpected phase differences can occur. Turning both loop antennas can then sometimes solve overlap.

Elliptical Polarization as a Result of Propagation

Elliptical or circular polarization can be found in ionograms. Figure 7 shows an ionogram of the Belgian Dourbes.[3] The signals reflected by the ionosphere are shown in two shadings.

We can imagine a linear polarized field, such as that of a dipole for example, as consisting of the sum of a right-hand and a left-hand circularly polarized component. Together, they form the linearly polarized field. In the ionograms, we see both circularly polarized components as the ordinary and extraordinary wave. If now, due to propagation by the ionosphere, for example, the left-hand circular component is attenuated, a right-hand circular field remains.

On the top bands, the extraordinary wave (light) normally seems to have a higher attenuation. In these cases, the ordinary wave (dark) is dominant. We will then be left with a circular (normally elliptical) polarized field. This effect in propagation is of course not constant and depends on propagation, on the time of day, the time of year and solar activity.

There is, however, not much information about this in normal handbooks or on the Internet. On the 80-meter band, the ordinary wave appears to dominate during the day (phase difference around –90°) and the signals are normally circularly polarized. As it gets dark, this moves to linear polarization with a once dominating extraordinary wave (phase difference around +90°). On the 160-meter band, we at least see circular po-

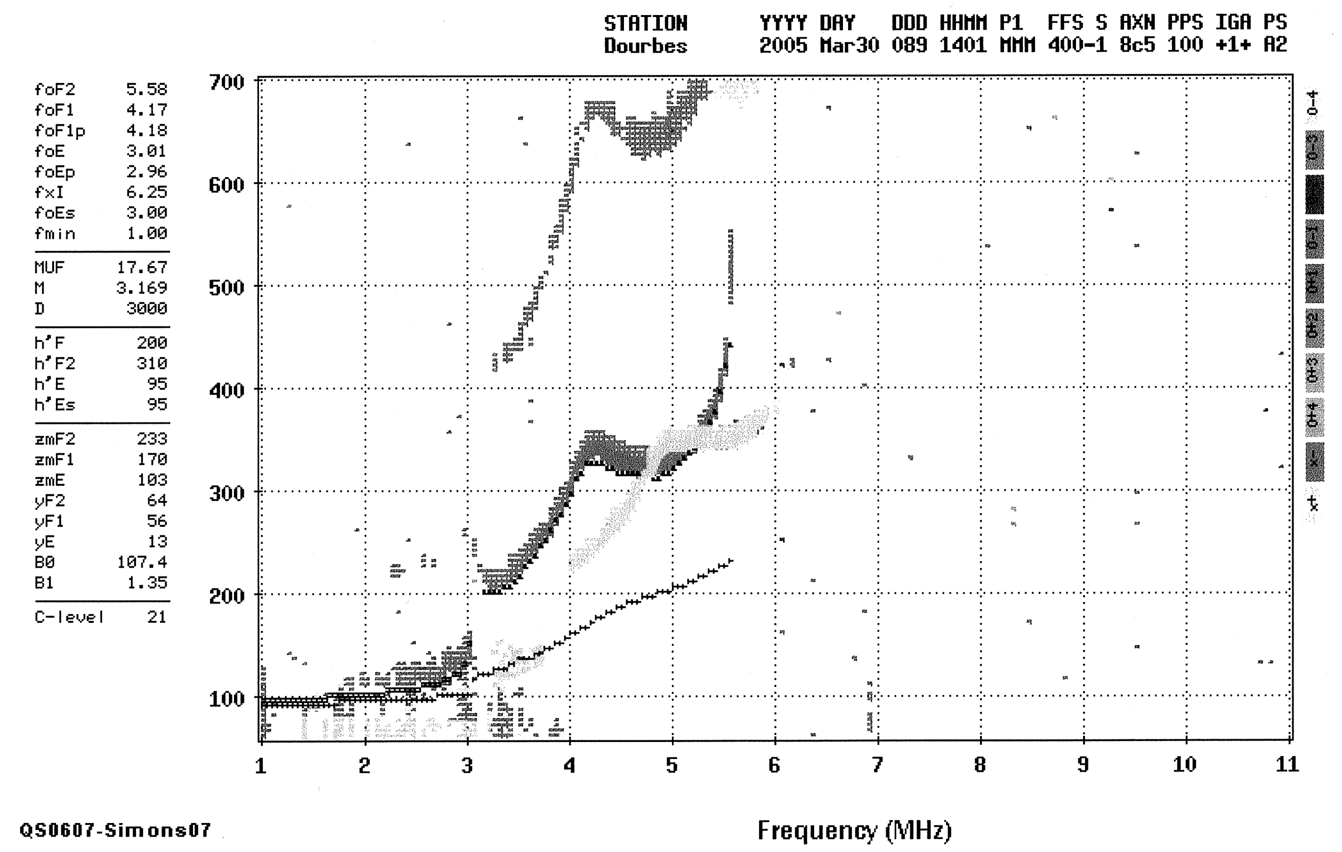

Figure 7 — Dourbes' ionogram with ordinary (light) and extraordinary (dark) reflections.

larization in the evening. Ground wave propagation is mainly vertically polarized, not circular. I have found no information as to how this is for DX signals. This will undoubtedly depend on whether or not propagation takes place by daylight. The Prolab program uses this division into ordinary and extraordinary wave and should be able to state this for DX.[4]

Phase Selectivity in DSP Equals Directivity

What can DSP do with a frequency component based on the phase difference? The most obvious thing is to pass 100% of those frequency components with a phase difference that corresponds with the desired direction. All other components are not passed, but suppressed 100%. You thus create a very effective directivity. In practice, you must state two boundaries between which the phase difference of the frequency component must be in order to be passed. The width of this phase window depends among other things on the constancy of the direction from which the signal arrives. The smaller this window, the more undesired signals will be suppressed and the stronger the directivity is. Phase selectivity equals directivity.

If we have one or several interfering sources, we can also opt to suppress associated phase differences. In fact, we can therefore suppress more than one interfering source 100%. For each interfering source, we must select a matching window. The smaller the window, the smaller the chance that desired signals will also be suppressed.

Everything coming outside or within the scope of a window can be suppressed 100%. We have an infinite front-to-back ratio, which can sometimes be too good. We may not hear the signals at all that we want to or should hear. We can make the level of suppression adjustable for this purpose, depending on what is desired.

We can freely select the form of the window, as shown in Figure 8. It is not necessary to pass or suppress 100% of the frequency components that have a phase which comes within a window. We can make this dependent on the phase difference with respect to the middle of the window. The window does not have to be continuous. We may also place a notch in the window against interference that shows about the same phase difference as the desired signal.

The windows are controlled manually, just like a rotator. The center of the window is set to the corresponding desired direction. Depending on whether the source in that direction is the desired signal or an interfering signal, the signal is either suppressed or passed. The width of the window is adjusted for best readability.

Overlap in Frequency

In the beginning, it was assumed that the frequency components of various sources do not overlap one another. In fact, there is, of course, always a chance that one or more components do overlap. This possibility is not constant but depends on the coincidental frequency content of the sources and the frequency resolution in the digital processing. The consequences depend on the amplitude ratios and the width of the phase window. The source delivering the strongest contribution at that moment has the strongest influence on the net phase difference of the total of both overlapping components.

If the contribution from the desired source is strongest, the component will be considered desirable. The contribution from the interfering source will then be unjustly passed, based on the phase difference. This will manifest itself in a lower suppression of the interfering source.

If the contribution of the interfering source is strongest, the component will be considered interference. The contribution of the desired source will then be unjustly suppressed based on the phase difference. If we make the window very small, we can hear this with speech in combination with noise as a loss in the high tones. The high tones are normally lower in amplitude. Weaker components of a desired source are generally less relevant. With the combination of several overlapping frequency components and whereby the contribution of the interfering source to these components is stronger than those of the desired source, we will miss much of the desired source. Depending on the coincidental overlap, fragments occur whereby the desired signal fails partially or completely.

Noise Sources

The assumption is that every frequency component comes from one single source. This assumption is always disrupted by the presence of noise, as noise exists over the entire bandwidth.

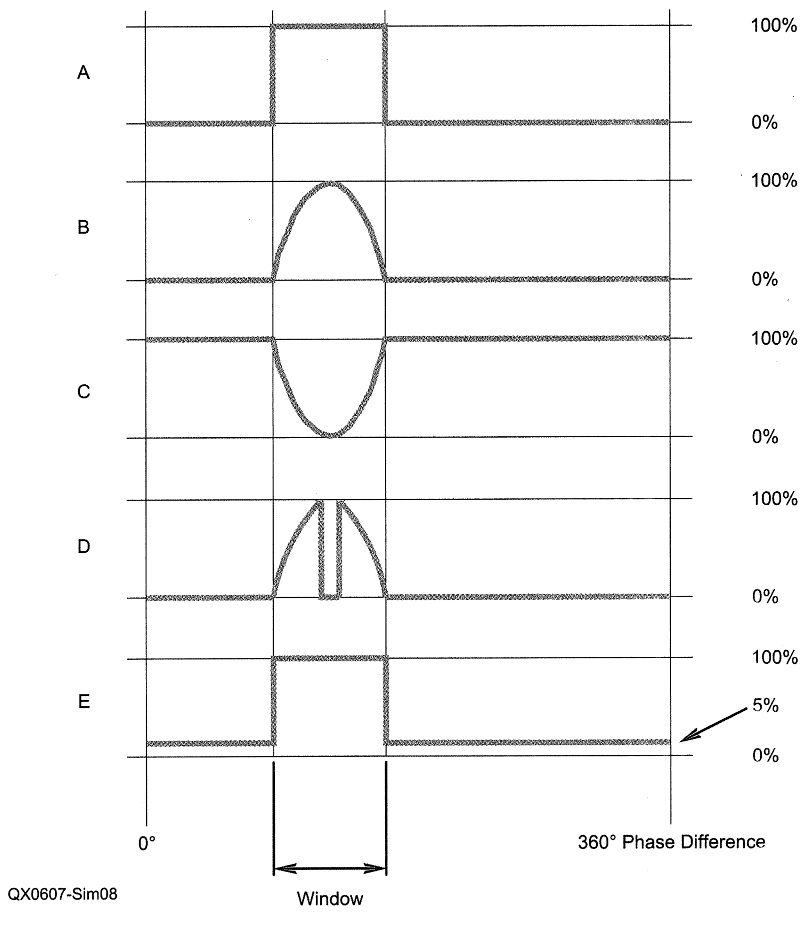

Figure 8 — In the window A) pass 100% B) pass depending on phase difference C) suppress depending on phase difference D) pass with notch e) pass 100% with 95% suppression outside the window.

We can distinguish four noise sources, man-made noise, atmospheric noise, antenna preamplifier noise and receiver noise. Man-made noise can be viewed by the large number of sources as random noise unless it concerns a distinct dominant local source.

The noise from both antenna preamplifiers and the receiver noise from both receivers are uncorrelated and can therefore have any random phase difference between 0° and 360°. Atmospheric noise and man-made noise are difficult to distinguish. Both antennas receive this noise with a difference in the time of arrival, and corresponding phase difference. The maximum possible phase difference depends on the distance between the antennas and the wavelength, as is the case with a normal desired signal. Each phase difference is also associated with an arc of directions. We therefore do not see this noise over the entire 360° but with a certain distribution over a part of this, as shown in Figure 5.

With the antennas at right angles to one another, man-made noise (ground wave) will be around 0° and 180°. Atmospheric noise can be circularly polarized and have a dominant phase difference around –90°. Linearly polarized atmospheric noise can also be distributed over the entire 360° (variable linear polarization), however.

Consequences of Noise

Were there not any noise, we would be able to determine the phase difference very accurately for each frequency component. In practice, however, each frequency component consists of a contribution of the desired or undesired signal and a contribution of the noise. This noise causes phase noise. The maximum deviation in the phase difference resulting from the noise increases as the signal gets smaller with respect to the noise. Therefore, suppression of a weaker interfering signal demands a wider phase window. If an interfering signal is roughly as strong as the noise, the need to suppress it will also be lower. Parallel to this is the fact that a stronger interfering signal needs a smaller window to be suppressed. So this works out the right way.

As the signal-to-noise ratio decreases, the achievable directivity decreases as a result of the necessary wider phase window. This is a disadvantage for weaker desired signals, as you would require of all things, stronger directivity for this.

For local (normally weaker) interference, it is possible that one of the antennas picks up the interference at a lower level than the noise level. One receiver then receives frequency components from this interference and the other only noise components. The phase difference is then dominated by these noise components. The result is that the interference components are spread out over 360°. The result is that interference with regard to phase difference is similar to noise.

Noise Reduction

Antenna amplifier noise and receiver noise can be found over the entire 360° phase difference. Atmospheric noise and man-made noise appear over a smaller range, defined by the distance between the antennas and the operating frequency. The amount of noise we pass is determined by the part we pass via the phase window. The width and form of this window determines how much noise is effectively passed. But also the amplitude ratios of the various noise sources is of influence, as it determines the distribution of the phase difference over 360°.

For the antenna amplifier noise and the receiver noise, reduction can increase considerably. A phase window of 45° will already show a reduction of 9 dB. A local interfering source, which is received by only one of the loop antennas, will also undergo similar reduction. Atmospheric and man-made noise will be reduced less. This noise is not distributed over the entire 360° but is concentrated within a smaller area.

Hardware

Figure 9 shows the total system. Both antennas are connected to the FT1000D. The FT1000D has two synchronized receivers, a main receiver and a sub receiver. Unfortu-

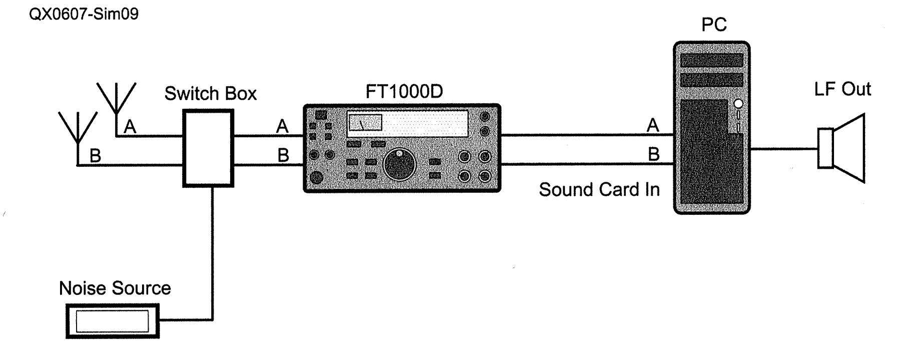

Figure 9 — The total system.

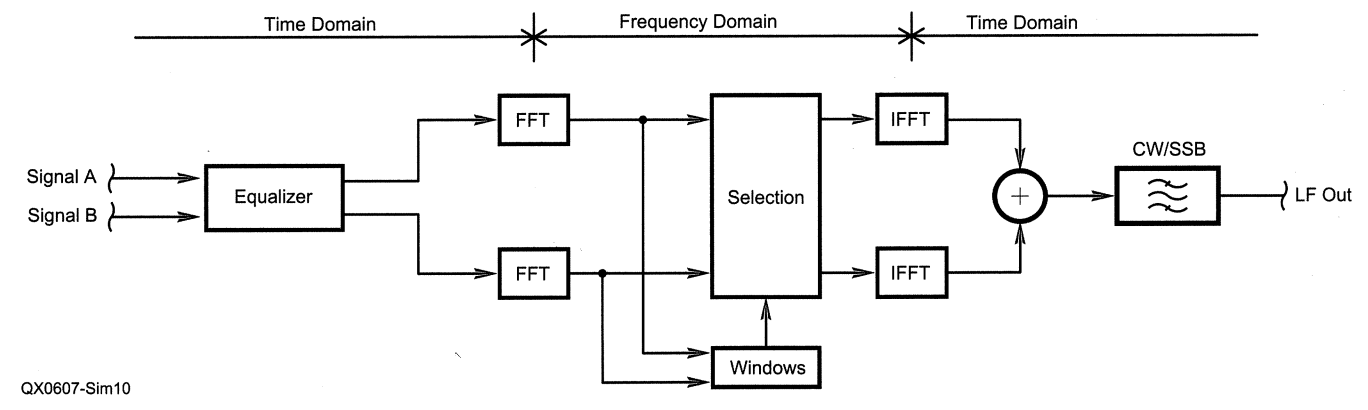

Figure 10 — Block diagram of the digital signal processing.

nately, both receivers are not identical. In the DSP we have to take into account the fact that the frequency and phase characteristics are different and that the phase difference between both receivers is constant in time, but unknown. A modification has been fitted in order to tune in both receivers via the main tuning button.[5]

The audio signal from both receivers is acquired via the sound card. No extreme specifications are demanded of the sound card as the receivers already carry out the AGC. A disadvantage is that software buffering is required on input and output for the uninterrupted processing of sound. This leads to an extra non-functional delay of 0.2 seconds. Apart from the fact that this is not ideal for tuning, it handicaps QSK telegraphy. For real-time processing, a 3-GHz Pentium PC is required.

The noise source is used to equalize both receivers in the DSP regarding frequency characteristics. For this, we connect the noise signal to both receivers at the same time.

An adaptive filter then corrects the audio signal from the sub receiver.

DSP Functionality

The key to the digital signal processing lies in the FFT (fast Fourier transform).[6] See Figure 10. The FFT can be seen as a large number of bandpass filters. With this, we can divide the audio band from 0 Hz to 4 kHz in 256 bands of 15.625 Hz. Because the bandpass filters overlap each other somewhat (spectral leakage), the net frequency resolution becomes about 30 Hz. If we increase the FFT frequency resolution, this will be associated with a longer time window. The frequency content of the sources varies in time and the chance of a contribution from an interfering source into one of these bands will remain roughly the same. But the average amplitude of this contribution decreases accordingly. So, a higher frequency resolution gives progressively better results. A four times higher resolution (7.5 Hz) is feasible at the cost of an increased time delay in the processing.

Apart from amplitude, the FFT also calculates the phase of each frequency component and it is therefore simple to calculate the phase difference. In the "windows" block, the software determines if the phase difference is within a window, and is calculated, depending on the form and function of a window, how strong the respective component must be passed or suppressed. The "select" block will calculate the final strength of the frequency components. The "IFFT" (inverse FFT) block converts it into a signal in the time domain.

With the "equalize" block, at the beginning of the block diagram, we can correct for differences in both receivers using an adaptive filter. With the noise source connected to both receivers at the same time, an adaptive filter adjusts the frequency characteristic in such a way that the sub receiver signal equals the signal from the main receiver.

The CW-filter has a somewhat hidden function. We decide per unit of time whether a frequency component may be passed. A passed frequency component during this unit of time has a constant amplitude output and is hard switched on and off. This produces the same effect (key-clicks) as that of a hard switched CW signal. With this CW filter, we reduce these "clicks." With the bandwidth used for SSB, this phenomenon is not audible.

Programming in Matlab/Simulink

Matlab is a most suitable environment in which to develop a DSP application.[7] It demands the necessary knowledge in the areas of DSP and mathematics, so it is not really meant for beginners. The block diagram in Figure 10 was programmed with Simulink. Simulink uses ready-made blocks in which basic functions are pre-programmed. This is handy as it is not necessary to reinvent the wheel. Another significant advantage is that Matlab is capable of producing an executable program, which increases processing speed considerably. This translates into the possibility of carrying out more complex processing in real time. It would be a little too much to explain the processing in greater detail in this article, but I will quote some figures. There are 4 FFTs and 4 IFFTs, each 512 (up to 1024 or 2048) samples long executed with a sample frequency of 8 kHz. The adaptive filter is of the least-mean-square (LMS) type.

Figure 11 shows a screenshot. The control panel is shown in the bottom half and is programmed using Labview.[8] For two windows, the phase difference (–180° / –90°) and the window width (0° / 30°) can be adjusted. There are switches to equalize both receivers, for selecting between CW or SSB, for selecting the main-receiver or DSP output, for passing or suppressing signals in the windows and for placing a notch in the window or not. There are settings for the depth of the suppression of the undesired frequency components and for an amplitude level below which frequency components (for example noise) will not be passed. Also, noise cancelling and an AGC function are implemented. An S-meter indicates the audio-signal strength and the AGC gain.

Two graphs show the phase difference of the frequency components (0 to 4 kHz) and a cumulative histogram of phase differences (upper right). There is also a setting for an amplitude level below which frequency components are not used for both graphs. The phase difference fitting the peak in the histogram (–87°) is shown separately in numerical terms. This makes it easier to adjust the windows.

Limitations

Practice shows there is only a small chance that frequency components of various sources overlap each other. The digital signal processing is based on this assumption.

This assumption also constitutes the major limitation, however. With the combination of many overlapping frequency components, whereby the contribution of the interfering source in these components is stronger than that of the desired source, the results described below are not feasible. In these cases, we can always fall back to the nulling properties of the loop antennas and to analog or digital (DSP) noise cancelling. (See Note 1.) In the DSP, we can also increase the frequency resolution from 30 Hz to, for example, 7.5 Hz. This gives progressively better results, especially for CW. The extra delay in the processing, however, is very noticeable. A 30 Hz net frequency resolution seems a good compromise.

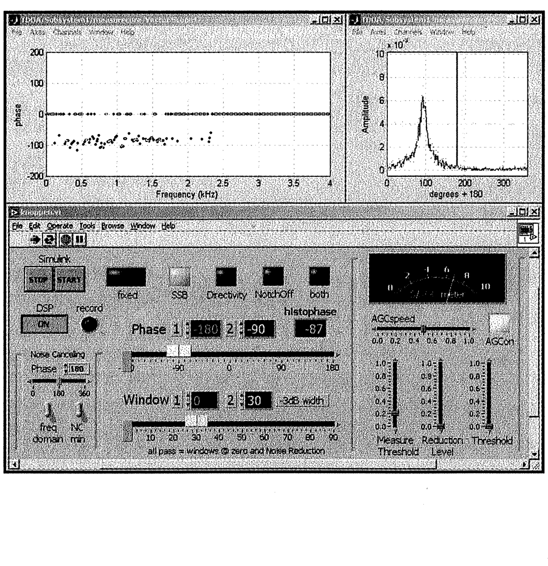

Figure 11 — Screenshot of the control panel and graphs. The control panel is shown in the bottom half and is programmed using Labview.

Despite the fact that the digital signal processing has not been carried out on the intermediate frequency level, it turns out to be very effective. The need for processing on the intermediate frequency level will only become noticeable if the interfering signal is much stronger than the desired signal. The desired signal will then be modulated too intensely by the AGC. It is better to control the AGC manually. This last limitation is not fundamental, however.

Results

The directivity achieved with DSP and the two loop antennas in the same direction is very convincing. Weak signals will be easier to copy with the noise reduction, although the useable directivity decreases with decreasing signal-to-noise ratio. Local and nonlocal interfering signals are easy to suppress. Even at 80 meters, and with an antenna distance of just 7 meters, stations coming from different directions can be distinguished. At 40 meters for example, it is often quite possible to separate G (west) from DL (east) stations here in PA. This gets even better on the higher bands because the distance between the antennas increases with respect to the wavelength.

For the 80 meter and 160 meter bands, the use of circularly (elliptically) polarized signals is very effective. The loop antennas are placed at right angles to one another for this purpose. If there is stable circular polarization, we can select the desired signals via the associated phase difference (–90°) resulting in a very quiet band. This is normally the case by day at 80 meters. Around and after sunset, the polarization sense on the 80 meter band may sometimes turn around and we see a phase difference of around +90°. In some cases, local PA stations distinguish themselves from DX stations at 80 meters in the sense of this circular polarization. The associated phase differences are then opposite and we can separate these stations very well. If polarization becomes linear at 80 meters after sunset, the phase difference will switch between 0° and 180°. Only local interfering sources coincidentally showing another phase difference can then be suppressed. For linear polarization, it is better to place the antennas in the same direction and to use directivity.

To what extent placing the loop antennas above one another can be used for selection on the elevation angle has not yet been tested. There should be added value from the fact that DSP adjustments are not needed as often.

Audio samples tell more than words and figures. That's why there are some audio samples on my Web site (see Note 1) that give an idea of the effectiveness of this digital signal processing.

Comparing Noise Cancelling Techniques

In principle with DSP, the same noise cancelling can be performed as, for example, the MFJ-1025 or ANC-4 perform on high frequency signals. DSP can cancel interference much better and faster using adaptive filters. When cancelling a local interfering source, two different antennas can be used.

Although a lot is possible with adaptive filters, there are limitations in the level of noise reduction achievable with two antennas used as a receiving array. In such an array, the signals from the antennas are summed after applying a complex weight to the signals. When using two antennas there is a maximum of just one interfering source or one arc of direction to be suppressed. Figure 12 shows an example of a possible 3-dimensional radiation pattern of an array of two loops, as shown in Figure 3A. The arc of nulls can be very sharp, but the beam is always broad and the achievable directivity is limited. When the distance between the loops gets smaller compared to the wavelength ($<1/8\ \lambda$), the gain decreases significantly.[9, 10, 11, 12] The sharp null makes it difficult to suppress interfering sky wave signals, because on shortwave the direction of arrival is constantly changing over time. The advantage is that such an array can operate if two or more signals share the same frequency.

The frequency domain processing using time difference of arrival or the phase difference for selecting frequency components as described in this article, behaves much differently. Beams and nulls are in arcs of direction as shown in Figure 4, but very sharp beams and very broad nulls are possible. Also, multiple beams and multiple nulls are possible. The achievable directivity is only limited by the signal-to-noise ratio. The gain is not affected when the distance between the loops gets smaller compared to the wavelength. A further advantage is that it is possible to profit by elliptical polarization.

All in all, each technique has its own advantages and disadvantages. There is not one single best technique. The best you can do is have all techniques at your disposal. Digital Signal Processing makes this feasible.

Epilogue

A lot more is certainly possible in the field of DSP. A challenge is carrying out this

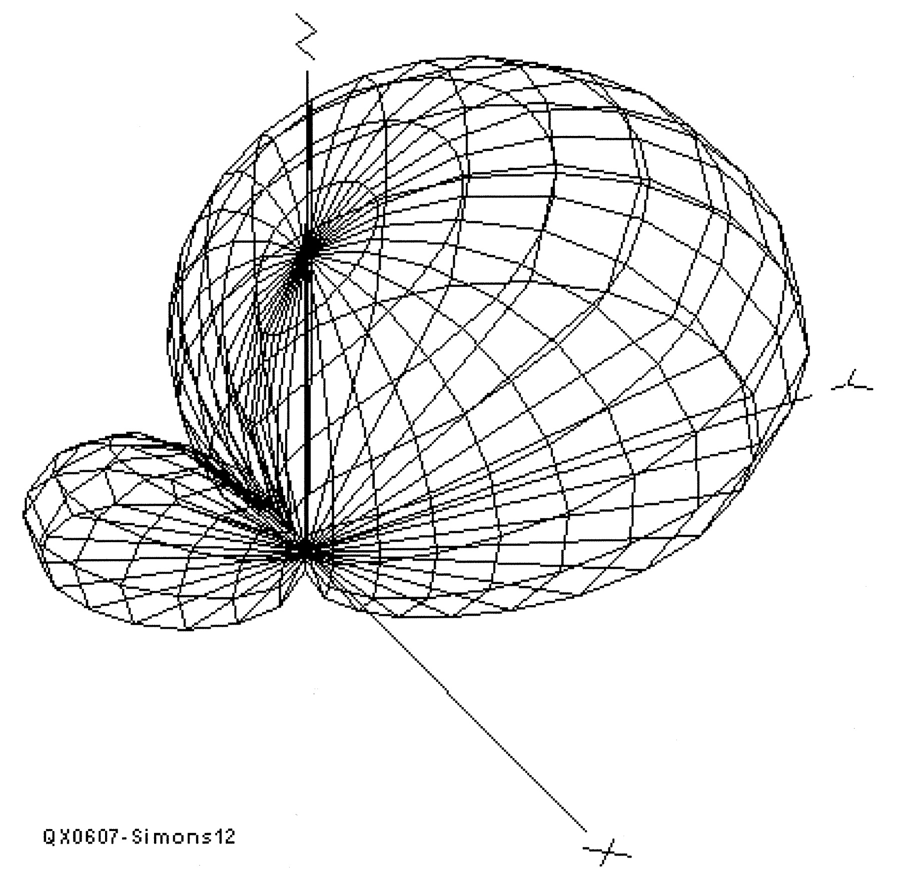

Figure 12 — A three-dimensional radiation pattern of an array of two vertical small loops, using summing of signals.

processing on the intermediate frequency level of two synchronized receivers. By "synchronized," I do not just mean exactly the same frequency, but also a fixed phase difference (phase coherent). With the FT1000D, the phase difference is changed after each frequency tuning and also after each receive/transmit/receive cycle. The Flexradio concept is maybe the most promising concept.[13]

Another challenge is extending the number of antennas and receivers. This would mean we can make a better selection of the desired direction.

Instead of loop antennas, other antennas, such as ground-planes or dipoles can be used. The associated radiation patterns each have their advantages and disadvantages. When making this choice, remember also the necessity for good balance and the level of mutual coupling between the antennas.

For reducing BPL (Broadband over Power Line) interference, this set-up is certainly no solution. BPL fills the entire band and, in case of mass use, comes from every direction.

Notes

[1] PAØSIM (audio samples) on the Internet: **http://home.plex.nl/~jmsi/**.

[2] R. Dean Straw, N6BV, *The ARRL Antenna Book*, 20th edition, The American Radio Relay League, 2003, p 3-11. *The ARRL Antenna Book* is available from your ARRL dealer or the ARRL Bookstore, ARRL order no. 9043. Telephone 860-594-0355 or toll free in the US 888-277-5289; **www.arrl.org/shop/**; **pubsales@arrl.org**.

[3] Dourbes: **http://digisonde.oma.be/**.

[4] Prolab: **www.spacew.com/www/proplab.html**.

[5] Modification FT1000D: **www.angelfire.com/md/k3ky/page61.html**.

[6] See the discussion about The Fourier Transform at: **http://users.rowan.edu/~polikar/WAVELETS/WTpart2.html**.

[7] Matlab/Simulink: **www.mathworks.com/**.

[8] Labview: **www.ni.com/labview/**.

[9] D. Smith, KF6DX, "Introduction to Adaptive Beamforming," *QEX*, Nov/Dec 2000, pp 50 – 55.

[10] B.Widrow and S. Stearns, *Adaptive Signal Processing*, Prentice-Hall, Englewood Cliffs, NJ, 1985.

[11] J. Devoldere, ON4UN, *Low-Band DXing*, Fourth Edition, The American Radio Relay League, 2005, Chapter 7. *Low-Band DXing* is available from your ARRL dealer or the ARRL Bookstore, ARRL order no. 9140. Telephone 860-594-0355 or toll free in the US 888-277-5289; **www.arrl.org/shop/**; **pubsales@arrl.org**.

[12] D. Smith, KF6DX, *Digital Signal Processing Technology*, The American Radio Relay League, 2001, Chapter 12. *Digital Signal Processing Technology* is available from your ARRL dealer or the ARRL Bookstore, ARRL order no. 8195. Telephone 860-594-0355 or toll free in the US 888-277-5289; **www.arrl.org/shop/**; **pubsales@arrl.org**.

[13] Flexradio: **www.flex-radio.com/**.

Jan Simons, PAØSIM has been a licensed radio amateur since 1975. He works as an analog and digital electronic engineer for a printer and copier manufacturer in the Netherlands. Designing mixed signal ASICs is a part of his job. As a radio amateur he especially enjoys working DX and CW on shortwave. Dealing with QRM is a real challenge when living in a residential location. Improving reception on shortwave has been a major activity for the last few years.

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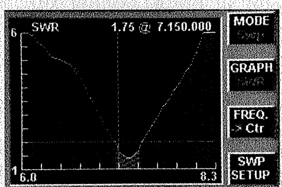

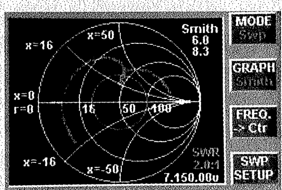

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