Pyramidal Frequency Search

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One of the early challenges in the AUV competition is detecting and locking on to the pinger signal. This is complicated by the fact that there are two pingers, one in each half of the competition arena. In the 2006 competition, each pinger signal lasted for 1.3 milli-seconds at 2-second intervals (approximately). The brute force approach is to collect hydrophone signals for 2.2 seconds to be sure of capturing the signals and then analyze the results. With a sampling frequency of 500 KHz, this requires a mere megabyte of memory and then some. A typical DSP might have about 32 kilobytes. One answer is to simply add more memory external to the DSP. The downside to this approach is that most of the hydrophone samples are just noise and the DSP then has to spend copious amounts of time searching the data to find the proverbial needle in a haystack. Clearly, a more intelligent method of searching is indicated.

Determining the frequencies present within a given signal is achieved by means of the Fourier Transform. For a sampled signal, we must use the Discrete Fourier Transform as expressed in the formulae

$$ReX[k] = \sum_{i=0}^{N-1} x[i]\cos(2\pi k\, i / N) \quad \text{and} \quad ImX[k] = \sum_{i=0}^{N-1} x[i]\sin(2\pi k\, i / N)$$

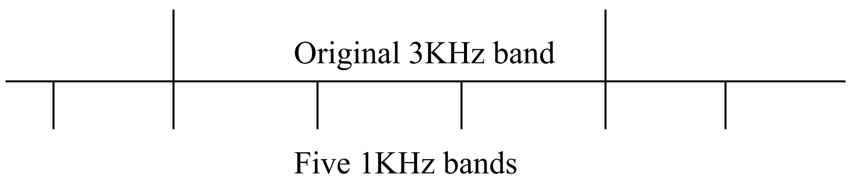
where N is the number of samples to be processed, i is the index of each sample in turn, and k is the number of complete cycles that fit exactly into the time period occupied by the N samples. The frequency corresponding to k is derived from the sampling frequency and the number of samples. *ReX*[] and *ImX*[] are the real and imaginary components of the derived frequency spectrum and are orthogonal to each other. The results may also be expressed in polar form as magnitude and phase components.

For the convenience of this discussion, we will set the sampling frequency at 768 KHz and choose the initial size of the sample set to be 256 samples. We are looking for a signal in the 20 to 40 KHz range and lasting about 1.3 milliseconds. With these settings, the pinger signal will appear in about 1000 samples and each "point" (band, bin, or bucket) in the frequency spectrum covers 768 KHz / 256 = 3 KHz. Most of the frequency spectrum is of no interest, so there is no point in calculating it. We are interested only in the points 6 through 13 inclusive covering frequencies from 18 KHz up to 42 KHz. Note that 8 points x 256 samples is the break-even point between the partial Discrete Fourier Transform (DFT) and the Fast Fourier Transform (FFT). This process has four possible results: (a) no point has a value significantly higher than any of the others,

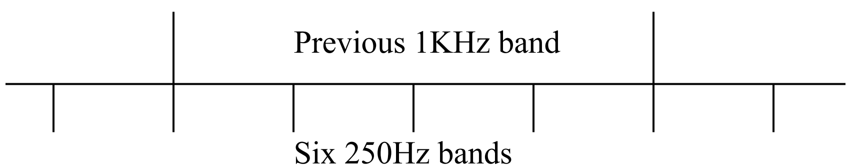
(b) one point alone has a value (peak) significantly higher than the others
(c) two adjacent points have values significantly higher than the others, and
(d) two separated points have values significantly higher than the others

In the first case, there is no pinger signal in this block of samples, so it is discarded and the same procedure is repeated with the next block of 256 samples. This constitutes the search part of the process – repeatedly testing for the onset of the pinger signal. In the other three cases, we do the same test on the next three blocks of 256 points; we expect these blocks to contain the bulk of the pinger signal. If the original peak disappears, it was a false start, so scanning starts again with the next block of samples. If a second peak appears, we have detected the onset of the signal from the second pinger, so we then accept up to three additional blocks of samples so that we have the bulk of the second signal in addition to the first. When all the testing is complete, we are left with one of the three remaining cases, b, c, or d stated above.

Considering the single-peak case (b) first, we calculate the partial DFT for the 768 points (in blocks 2, 3 and 4), but only for the points corresponding to the original peak plus one point on either side. Note that each "point" in this frequency spectrum covers 768 KHz / 768 = 1 KHz, so there are 5 point total to compute.



As before, we may find a single peak in one band, two peaks in adjacent bands, or two peaks in separated bands. For now, we will stay with the single-peak case. The next step is to add nine blocks of zeros after the 768 legitimate samples and compute the partial DFT for 3072 samples, but only for the points corresponding to the previous peak plus one point on either side. Each "point" in this frequency spectrum covers 768 KHz / 3072 = 250 Hz, so there are 6 point total to compute.



This process can be repeated until the desired level of precision is reached. In some applications, a high degree of precision is required to avoid undesirable side effects in signals generated by the system. Note that no additional legitimate samples are brought into the mix because we know in advance that we have already captured most of the pinger signal – there is no more data to collect. When the signal frequency has been determined with sufficient accuracy, we turn our attention back to the time domain to find the exact length of the pinger burst.

At any stage in the hierarchy, if we find two peaks in adjacent bands, the procedure is the same, but more points have to be calculated so that both of the bands are covered plus one new point on either side. At the first stepdown, this would mean 8 points instead of the 5 stated above. The same rationale carries all the way down the hierarchy. The one remaining case is two peaks separated by an intervening band. For this, we do two separate subsearches, each identical to the single-peak subsearch outlined above.

When all the data collected has been analyzed sufficiently, it may be that only one pinger frequency has been determined. The search then resumes from the beginning again and continues until a second frequency has been detected or until enough time has elapsed that we know that there is no second pinger. These methods can also be used when there are more than two pingers and some general information about them is known.

> *This is the end of this White Paper. The use of these techniques without proper acknowledgement of the author in all written works will cause your sub to be cursed and sink to the bottom of the competition arena. Be sure to read the companion White Papers on the hydrophones mathematical model, Hydrophone Sampling Techniques, and Synchronous Fourier Transforms. Coming soon to a website near you!*