How we capture the intent of your affirmation: We use a short time Fourier transform, which means that we essentially break up the sample into a number of smaller samples which can be analyzed and broken into a sum of sinusoidal's. However it is not enough to simply break down the sample into a non-overlapping set of smaller samples, there is some framing that must be done, in our processing the sample windowing is overlapped by 75%. This provides for a smoother scaled output signal without the large number of signal artifacts, which would otherwise be present at the boundaries of our processing sample size. The processing sample size is set at 20ms which provides for a small enough sample so that we can use a Short Time Fourier Transform to generate our scaling data since over 20ms of time the signal will not significantly change in the time domain.

Using the STFT (Short Time Fourier Transform) we generate a Frequency Domain analysis of the signal by generating an array of bin frequency analyses. Our bin frequencies are separated by 48hz to provide maximum resolution in the Frequency Domain. We probe our 20ms sample for each of our bin frequencies resulting in a Magnitude, Frequency and Phase result.

We then do some additional processing to manage phase shifts, which occur due to the fact that our input sample frequencies are not spaced exactly 48Hz apart. When a sample frequency participates in more than one bin frequency probe the phase of the resulting output will shift. We take this into account in our processing by the use of an algorithm designed to take the phase difference in our bin processing output and apply it to the Magnitude of the frequency and shifting the phase of the output to be coherent with the expected phase.

Then it is a simple matter to take the median frequency domain analysis of the input sample and our target frequency and arrive at a scaling factor. This scaling factor is applied to the frequency result of our processing. We then process the results of our processing using an Inverse Fourier Transform which basically takes our processed set of sinusoidal frequencies and regenerates a complex wave form that has been frequency shifted. We use this Alpha - Theta - Delta information to imprint the Structured Water and create the cellular frequency programs.

Overriding A/D Sampling Limitations: Our application takes audio samples at a rate of 44100 samples per second with an amplitude resolution of 16 bits giving 65536 discreet amplitude steps per sample. This full CD quality sampling rate ensures that all available frequency and amplitude information in the voice is collected and analyzed. Sampling at this rate results in a data set that is able to represent frequency information where the Nyquist frequency is 22050 kHz, well above the range of human speech.

Analysis: Our application applies a standard Fast Fourier Transform to the mathematical representation of the voice sample data to convert the information in the time domain as it is represented by the sample data collected from the user to a data structure representing the same information in the frequency domain. This is an industry standard analysis function used by all the spectrum analysis tools available today.

We supplement the utility and resolution of the FFT (Fast Fourier Transform) by the use of a specialized and custom arithmetical mathematics library that allows for a far greater degree of resolution than currently available in commercial math libraries. Our application also applies a variant of the FFT algorithm to the input data called the Goertzel Transform. The Goertzel Transform is mathematically related to the FFT but acts on only a single frequency, allowing us to apply a different algorithm to the same data and increasing again the accuracy of our analysis. The combination of these two algorithms is unique to our approach and to this writer's knowledge is not used commercially in any other product.

Both the FFT algorithm and Goertzel algorithm we have developed are modified to work against an intermediate data representation that expands and extrapolates the data contained within the voice sample. This is required due to the way that these algorithms work. Both algorithms result in a series of bins each bin contains two complex numbers that can be further manipulated mathematically to produce a frequency/intensity value. It is this value that is used subsequently in our analysis algorithm.

Due to mathematical constraints the size and thus resolution of this set of bins is one half of the sample size. An analysis set size of 1024 samples will result in the entire frequency domain map spanning only 512 bins; each of these bins therefore will contain information regarding 43.06 Hz of the frequency spectrum - obviously very low resolution. This is the type of frequency domain analysis used by media player visualizations and by some other spectrum analyzers on the market.

Our application uses a technique whereby the output range is vastly increased resulting in an output structure that contains over 1,099,511,627,776 bins. These bins are mathematically represented with a proprietary format and method that requires virtually no storage on the sample processing computer. This representation allows us to analyze voice data at a resolution which would otherwise require more storage per sample window than is present on any modern day computer. Our sample resolution results in each bin containing frequency information about .00000002005 (2.005E-8) Hz of the frequency spectrum - as you can see this allows us to more accurately gain information about the frequency spectrum of a sample since each bin represents such a small section of the entire spectrum.

Comparison with Hardware Spectrum Analyzers: It is difficult to compare our mathematical approach to a hardware based approach simply because of the limitations of the hardware based method. Hardware methods have a resolution that depends on the cost and complexity of the circuitry used to generate the frequency domain data. Hardware based approaches use a resonant filter circuit for each bin that filters out intensity information not configured for that filter. For each individual frequency the hardware system analyses there must be a single corresponding circuit. Due to the physical nature of these circuits there is a small upper limit on the number of bins that a hardware based system is able to provide whereas our software based system is virtual in nature and relies on mathematical concepts for it's representation and analysis allowing us practically unlimited resolution.

Synthesis and Remapping: Our synthesis engine is also mathematically based on trigonomic functions that output waveform data directly and allow us to modify and control the phasing of individual components of the synthesized audio. Other applications rely on wave table synthesis whereby the output waveform is stored in small chunks (the wave table) and simply copied out to the output data. Wave table synthesis is faster but results in aliasing of output data as a result of scaling which must take place to generate waveforms of a different frequency than what is stored in the wave table. Our method generates a smoother, more natural sounding output. Being able to modify the phasing of component waveforms also allows us to generate with great precision beating of the signals.

It is this beat frequency generation that results in the great impact our system has on the user. By the application of a proprietary algorithm we are able to tune the standing wave generated inside the user's brain. A standing wave is an interference pattern generated when two or more waveforms interact. The important thing about standing waves is that they apply energy to a single spot continuously whereas a regular waveform applies energy only for a brief period during each cycle. Manipulation of the phasing of the component signals allows us to generate standing waves inside the neural circuitry of the user's brain to initiate and sustain immensely powerful change.

However, our system does not simply beat two frequencies; the output waveforms are complex and contain more than simply two waveforms. We generate a complicated interference pattern comprised of more than 6 waveforms and the interference pattern thus generated exhibits dynamic shifting in four dimensional space (the three spatial dimensions and time). By the use of a phasing equation we are able to manipulate the Scalarwave energy construct so that it maximizes the impact on the receiving system - the user.

Conclusion: "Our system is by far the most accurate and reliable system available. It melds the science of mathematics and sound to produce a system that mediates change with a precision unprecedented by any other system. Other systems rely on simple monotone frequency generation, low resolution analysis, basic tonal analysis and generally do not offer the complexity required to mediate change within the user. When coupled with proprietary Scalar Vortex Technology, this system is unbeatable." **Leslie J. Marshall (M. Sc.) July 24, 2006**