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1 Attorney Docket No. 79941

2 SYSTEM AND METHOD FOR PROCESSING AN UNDERWATER ACOUSTIC SIGNAL
3 BY IDENTIFYING NONLINEARITY IN THE UNDERWATER ACOUSTIC SIGNAL
4

5 STATEMENT OF GOVERNMENT INTEREST

6 The invention described herein may be manufactured and used
7 by or for the Government of the United States of America for
8 governmental purposes without the payment of any royalties
9 thereon or therefore.
10

11 CROSS-REFERENCE TO RELATED PATENT APPLICATIONS

12 Not applicable.
13

14 BACKGROUND OF THE INVENTION

15 (1) Field Of The Invention

16 The present invention relates to systems and methods for
17 processing acoustic signals and particularly, to a system and
18 method for processing an underwater acoustic signal by
19 identifying nonlinearity (e.g., chaos) in the underwater signal.

20 (2) Description Of The Prior Art

21 Underwater acoustic signals, such as sonar signals, have
22 intrinsic nonlinear properties. Nonlinearity is a property of a
23 dynamical system whereby the evolution of its variables depends
24 on products of two or more of the present values of the

1 variables. Chaos is a special type of nonlinearity. A defining
2 characteristic for chaos to be present in a nonlinear system is
3 that the system contains at least one positive lyapunov
4 exponent. Nonlinearity in acoustics can arise from irregular
5 (i.e., nonlinear) boundary conditions of the propagation
6 channel, the target "echo" response, reverberant scattering
7 within the channel, or any combinations of these.

8 Underwater acoustic sensors and sonar systems typically
9 employ traditional linear processing methods. The intrinsic
10 nonlinear properties present in underwater sound propagation and
11 reflection cannot be detected using the traditional linear
12 processing techniques. A linear processor can detect first
13 order effects only. Linear processes are additive processes,
14 i.e., the additive property of superposition applies only to
15 linear processes. For nonlinear processes, the superposition
16 principle no longer holds up and one must revert to some
17 alternate means of signal analysis to best determine the
18 information contained in the signal. Although a conventional
19 method of signal analysis, such as a Fourier analysis, is very
20 robust for analysis of a signal for a linear process, such
21 methods are unable to make quantitative delineations in a signal
22 structure whenever there exists a dominant nonlinearity in the
23 signal of interest. Thus, the nonlinearity and chaos in the

24

1 acoustic signature cannot be exploited when using conventional
2 linear processing systems and methods.

3

4 SUMMARY OF THE INVENTION

5 Accordingly, one object of the present invention is to
6 process an underwater acoustic signal by identifying
7 nonlinearity (i.e., chaos) in the underwater acoustic signal.

8 In accordance with one aspect of the present invention, the
9 system and method of the present invention detects the
10 underwater acoustic signal and digitizes the underwater acoustic
11 signal to produce an acoustic time series representing the
12 underwater acoustic signal. The acoustic time series is
13 reconstructed using a phase space embedding algorithm to
14 generate a phase space embedded acoustic signal. A differential
15 radius signal is generated from the phase space embedded
16 acoustic signal using chaotic radius computations and
17 differential computations. Thresholds detected in the
18 differential radius signal represent nonlinear events hidden in
19 the underwater acoustic signal.

20 According to one embodiment of the system and method, the
21 phase space embedding algorithm is:

$$22 \quad Y(t) = \{S(n), S(n+T), S(n+2T), \dots, S(n+(m-1)T)\}$$

23 where $S(n)$ represents the acoustic time series for $n=1, 2, \dots, N$, T
24 is a time shift parameter, m is the number of dimensions in the

1 embedding space, and $Y(t)$ is a time dependent Euclidean vector
2 function in R^m . The dimension m can be determined using a False
3 Nearest neighbor (FNN) technique. The time shift parameter T
4 can be determined by computing average mutual information (AMI)
5 or by selective choice from a number of trial values for T .

6 In accordance with another aspect of the present invention,
7 a nonlinear signal processor processes the acoustic time series
8 representing the underwater acoustic signal. The nonlinear
9 signal processor comprises a phase space embedded signal
10 generator for reconstructing the acoustic time series using a
11 phase space embedding algorithm to generate a phase space
12 embedded acoustic signal. A chaotic radius processor computes a
13 chaotic radius for each point in phase space for the phase space
14 embedded acoustic signal producing a time series of chaotic
15 radius values. A differential radius processor computes a time
16 derivative of the time series of chaotic radius values to
17 produce the differential radius signal.

18

19 BRIEF DESCRIPTION OF THE DRAWINGS

20 These and other features and advantages of the present
21 invention will be better understood in view of the following
22 description of the invention taken together with the drawings
23 wherein:

1 FIG. 1 is a schematic block diagram of a system for
2 processing an underwater acoustic signal by identifying
3 nonlinearity, according to one embodiment of the present
4 invention;

5 FIG. 2 is a flow chart representing the method for
6 processing the underwater acoustic signal, according to one
7 embodiment of the present invention;

8 FIGS. 3A-3D are plots of phase space orbits of a linear
9 frequency modulated (LFM) signal having different embedding
10 delays T , according to one example of the present invention;

11 FIGS. 4A-4D are plots of phase space orbits of a LFM signal
12 having different signal-to-noise ratios, according to one
13 example of the present invention;

14 FIGS. 5A-5C are plots of an acoustic time series for three
15 different types of LFM signals, according to one example of the
16 present invention;

17 FIGS. 6A-6C are plots of a differential radius signal for
18 three different types of LFM signals where the embedding delay
19 $T=13$, according to one example of the present invention;

20 FIG. 7 is a phase plot of a LFM signal at time $t=.2s$ and
21 having a delay $T=13$, according to one example of the present
22 invention; and

FIGS. 8A-8C are plots of a differential radius signal for three different types of LFM signals where the embedding delay $T=28$, according to another example of the present invention.

DESCRIPTION OF THE PREFERRED EMBODIMENT

A signal processing system 10, FIG. 1, according to the present invention, is used to process an underwater acoustic signal 12 by identifying nonlinearity such as chaos in the signal 12. The signal processing system 10 uses a nonlinear dynamical systems technique referred to as the differential radius technique for identifying the nonlinearity in the acoustic signal 12, such as a sonar signal. In the exemplary embodiment, this approach to modeling sonar signal propagation is based on a deterministic characterization of sonar dynamics.

Although the exemplary embodiment is used for the detection and classification of a sonar signal, the present invention is applicable to any situation requiring detection and monitoring of underwater acoustic signals (e.g., bottom and target mapping) and applies to any underwater signal-of-interest having nonlinearity or chaos.

The signal processing system 10 includes a signal detector 14 that detects the acoustic signal 12 and converts the acoustic signal into an electronic signal. In one example, the signal detector 14 is a hydrophone or an array of hydrophones so that

1 the time series output of the measured acoustic fluctuations
2 represents element or beam level output. The signal processing
3 system 10 includes a digitizer 16 that digitizes the acoustic
4 signal (e.g., by time sampling the acoustic signal) to produce
5 an acoustic time series. The signal processing system 10 also
6 includes a nonlinear processor 20 that processes the digitized
7 signal or acoustic time series using a differential radius
8 technique for identifying nonlinearity (e.g., chaos) in the
9 acoustic signal 12.

10 The nonlinear signal processor 20 comprises a phase space
11 embedded signal generator 22, a chaotic radius processor 24, and
12 a differential radius processor 26. These components of the
13 nonlinear signal processor 20 can be implemented as discrete
14 structures, as software modules in a data processing system, or
15 as a combination of both. The nonlinear processor 20 processes
16 the acoustic time series and generates a differential radius
17 signal. A threshold detector 30 detects thresholds in the
18 differential radius signal, which represent nonlinear or chaotic
19 events hidden in the underwater acoustic signal 12.

20 The method for processing the underwater acoustic signal 12
21 is shown in FIG. 2. The underwater acoustic signal 12 is
22 detected (step 110), and digitized to produce the acoustic time
23 series (step 112). The acoustic time series is then
24 reconstructed to generate a phase space embedded signal (step

1 114). According to one exemplary embodiment, the state space
2 embedding algorithm can be described as follows:

$$3 \quad Y(t) = \{S(n), S(n+T), S(n+2T), \dots, S(n+(m-1)T)\} \quad (1)$$

4 where $S(n)$ represents the acoustic time series for $n=1,2,\dots,N$, T
5 is a time shift parameter, m is the number of dimensions in the
6 embedding space, and $Y(t)$ is a time dependent Euclidean vector
7 function in R^m . Each added delay (e.g., T , $2T$, $3T$, etc.) can be
8 thought of as a derivative of higher order. For a 2-D system,
9 the first derivative is captured, for a 3-D system, a second
10 derivative is captured, and so on until all of the derivatives
11 necessary for some prescribed n th order differential system are
12 obtained.

13 One method of determining the dimension m is by applying a
14 numerical technique called False Nearest Neighbors (FNN). One
15 preferred method of determining an optimal delay or time shift
16 parameter T is by computing the average mutual information
17 (AMI). According to another alternative, the parameter T is
18 computed using a value obtained by choosing from a number of
19 trial selections that maximally fill out orbits in the phase
20 space domain. The selection of an appropriate time delay value
21 T is described in greater detail below.

22 The chaotic radius for each point in the phase space
23 embedded signal is then computed, step 116. The chaotic radius
24 is defined as the Euclidean distance from the point of origin in

1 is defined as the Euclidean distance from the point of origin in
2 phase space to any given point along the phase space orbit. In
3 two dimensions, the chaotic radius is obtained by computing the
4 hypotenuse using the Pythagorean Theorem, in which values for
5 the legs of the triangle are given by the x and y vector
6 components of the point in phase space. According to one
7 example, a vector is drawn from the origin to a point in the
8 Euclidean phase space, and the vector magnitude is the chaotic
9 radius. Vectors are constructed from the origin to each
10 succeeding point in phase space, and the magnitude of each
11 successive vector yields a time sequence of chaotic radius
12 values that replaces the original acoustic time series.

13 The differential radius signal is then generated by
14 computing the time derivative of the chaotic radius values (step
15 118). The differential radius is computed from the time
16 sequence of chaotic radius values by computing the time
17 derivative, or rate at which points separate in phase space. In
18 one example, the differential radius can be approximated by
19 taking the difference of magnitude between adjacent points in
20 the time sequence of chaotic radius values.

21 Threshold values in the differential radius signal can be
22 detected (step 120) to allow nonlinear or chaotic events to be
23 identified. The differential radius signal indicates where
24 significant "jumps" occur in phase space and thus can be used to

1 monitor and track the entropy of an evolving dynamical system.
2 Because the differential radius, when plotted as a function of
3 time, is a measure of the separation rate of neighboring points
4 in phase space, it is also related to the Lyapunov exponent (the
5 characteristic parameter that is used to define chaos), and
6 thereby, to the entropy as well. The differential radius signal
7 provides an indication of events in which the state of the
8 system changes rapidly with respect to itself, such as when the
9 transmitted signal "echoes" off of a target (e.g., in time and
10 space).

11 Thus, the chaotic radius and differential radius
12 computations provide highly precise temporal and spatial
13 detection markers for evolving events as the time series
14 unfolds. Hidden or anomalous events can therefore be detected
15 in an acoustic time series. These events are associated with
16 rapidly changing evolutions in phase space, typical for a
17 variety of nonlinear and chaotic motions exhibited in a large
18 number of acoustic signatures. Target echo responses, man-made
19 and biologic anomalies in an otherwise ambient acoustic
20 background are examples of such evolutions.

21 The present invention is now described in greater detail in
22 connection with a simulated sonar signal $X(t)$, a linear
23 frequency modulated (LFM) wave form given by the following
24 expression:

$$X(t) = A \sin[2\pi(f_0 t + \alpha t^2)] + n(t) \quad (2)$$

where A is the amplitude level, f_0 is the start frequency, α is the change in frequency per second, t is the time duration, and $n(t)$ is additive noise. In this example, assume that the amplitude A is 1, the start frequency f_0 is 50 hertz, (Hz), the end frequency is 90 Hz, $\alpha = 20$ Hz per second, the time duration t is 1 second, and the sampling interval Δt is 1 millisecond.

FIGS. 3A-3D show how this LFM signal looks in phase space in two dimensions for four different values of the delay parameter T in the pure signal case having 0% additive noise or $n(t) = 0$. In the example shown, the abscissa is $X(t)$ and the ordinate is $X(t+T)$ for values of $T=1$ (FIG. 3A), $T=3$ (FIG. 3B), $T=10$ (FIG. 3C) and $T=13$ (FIG. 3D). Although the topology is similar at each value of the delay, T, the intricacy of the LFM signal is best revealed at a delay of $T=13$ sampling points (FIG. 3D). This LFM signal exhibits an almost-limit cycle behavior as in a simple sine or co-sine function, but unlike a sine wave, the LFM signal is made of several almost-connected sine waves. A limit cycle is an orbit that comes back on itself. If the LFM signal were to propagate in time over and over again continually, one would observe a repetition of combined frequency singular limit cycle orbit for each repeated cycle when plotted in phase space. This LFM signal having 0% additive

1 noise is dynamically linear in that there are no apparent
2 perturbations that force the LFM signal away from its natural
3 orbit.

4 FIGS. 4A-4D show the phase space orbits of the LFM signal
5 at different signal-to-noise ratios (SNR's): SNR = ∞ (FIG. 4A);
6 SNR = 20 dB (FIG. 4B); SNR = 17 dB (FIG. 4C); and SNR = 14 dB
7 (FIG. 4D). When noise is added to this signal, the phase space
8 images become more diffuse. At an SNR of 14 (FIG. 4D), for
9 example, the intricate helical pattern is totally obscured, yet
10 very rough near-limit cycle behavior can be observed. As more
11 noise is added to the LFM signal, the almost-limit cycle
12 appearance of the LFM signal eventually becomes unrecognizable.

13 FIGS. 5A-5C shows the time series for three simulated
14 conditions: (1) the pure LFM signal (FIG. 5A); (2) the LFM
15 signal plus 10% additive noise (FIG. 5B); and (3) a full wave
16 square law rectification of the pure LFM signal (FIG. 5C).
17 Condition (3) is a nonlinear analog of condition (1) due to the
18 squaring operation. By discriminating between a linear and non-
19 linear time evolution, the differential radius technique
20 discussed above is able to identify the nonlinearity in
21 condition (3).

22 The time series plots can be interpreted analytically
23 through trigonometric relationships as well as dynamically
24 through corresponding phase plots and differential radius plots.

1 Analytically, a trigonometric relationship exists between the
2 pure LFM signal (FIG. 5A) and the squaring of the same LFM
3 signal (FIG. 5C). Since the LFM signal is represented
4 analytically by a sine function, as indicated above in Equation
5 (2), the relationship between the two signals is made clear by
6 the following identity:

$$7 \quad \sin^2(\omega(t)) = \frac{1}{2} - \frac{1}{2} \cos(2\omega(t)) \quad (3)$$

8 This sine-squared signal is the superposition of two
9 signals, one a 0 Hz or "flat line" signal with constant
10 amplitude equaling 1/2, and the other, a cosine or phase-shifted
11 sine wave of twice the frequency ω and one-half the amplitude.
12 Multiplying two sine waves of different frequencies together is
13 equivalent to superposing two signals of differing frequencies
14 in which both superposed frequencies are non-zero.

15 The time series of the LFM signal plus additive noise (FIG.
16 5B) represents a small perturbation (i.e., 10% added noise) to
17 the pure LFM signal and looks almost the same. The dynamical
18 sensitivity to 10% additive noise can be seen by comparing the
19 phase space orbits for the pure LFM signal (FIG. 4A) and the LFM
20 signal with 10% added noise (FIG. 4B).

21 FIGS. 6A-6C show the differential radius signals for the
22 three times series across a one second window and with a time
23 delay $T=13$. Where the time delay $T=13$, the differential radius

1 of the pure LFM signal has a region where the differential
2 radius is very close to zero from time $t=0.18$ to $t=0.20$. During
3 this portion of the temporal evolution of the LFM signal, the
4 dynamical signal trajectory in phase space resides on a circle
5 of radius of approximately equal to one.

6 FIG. 7 shows a phase plot of the simulated LFM signal taken
7 around the interval $.18 < t < .2$ at a delay of $T=13$, which is hidden
8 in the more complicated phase plot shown in FIG. 3D. This phase
9 plot represents a typical circular (near-limit cycle) pattern in
10 which $T=13$ represents $1/4$ wavelength for a frequency of about 57
11 Hz, which is the same frequency being "swept out" by the LFM
12 signal in the time interval $.18 < t < .2$.

13 As the LFM signal evolves over time, there is a continual
14 shift in its frequency. The starting frequency of the simulated
15 signal is 50 Hz, and at .19 seconds from the starting time, the
16 frequency has advanced to 57 Hz. The chosen delay of $T=13$
17 corresponds to a time interval approximately equal to $\lambda/4$, where
18 λ is the associated period of the LFM 57 Hz frequency component.

19 Choosing a different delay T would require a corresponding
20 shift in frequency to obtain a circular geometry of the orbit.
21 For a delay of $T=13$ (FIG. 6A), the differential radius of the
22 pure LFM signal converges while in oscillation to zero around
23 $t=.2$ seconds and thereafter diverges while still in oscillation
24 to a maximum around $t=.7$ beyond which it again begins to

1 converge back to zero. In the differential radius of the LFM
2 signal with 10% additive noise (FIG. 6B), the trends in
3 differential radius fluctuation are very similar, but several
4 deflections are evident above and below the pattern for the pure
5 LFM signal. Thus there is an appearance of a higher degree of
6 randomness in the case of the noisy signal as compared to the
7 pure signal. This apparent increase in random behavior can be
8 associated with a corresponding increase in entropy between the
9 two signals.

10 The differential radius of the full wave square law
11 rectified signal (FIG. 6C) has approximately the same maximum in
12 comparison to the pure LFM signal (FIG. 6A) but exhibits
13 differences when comparing the minima. The differential radius
14 of the squared LFM signal has no zero convergence. The squaring
15 operation has pushed the limit cycle at 57 Hz away from the
16 origin to the extent that it no longer maintains, at least for
17 the short duration, a constant radius (i.e., a near perfect
18 circular geometry centered at the origin). Thus, the dynamical
19 distinction in differential radius between the pure LFM signal
20 and the squared LFM signal shows that the differential radius is
21 a useful analytical tool for identifying and discriminating
22 nonlinearities in a measured time sequence.

23 The implications of the choice of the delay T on the
24 dynamical evolution of the differential radius can be seen by

1 examining the corresponding limit cycle frequency component of
2 the LFM signal for a prescribed delay value T. For a given
3 value of T, the limit cycle period λ is related to T by the
4 following expression:

$$5 \quad \lambda/4 + n\lambda/2 = T \quad (4)$$

6 This expression shows that a circular limit cycle exists in a
7 phase space orbit if the embedding delay T is chosen to be 1/4
8 of the period of oscillatory wave plus any integer multiple 180
9 degree translation of the period. For example, a delay value of
10 $T=13$ corresponds to $\lambda=52$ msec or $f=19$ Hz for $n=0$. Since 19 Hz
11 lies outside of the band of the simulated LFM signal, an integer
12 multiple 180 degree translation must be identified that first
13 appears inside the band, according to equation (4). Where the
14 band of the signal described above goes from 50 to 90 Hz, the
15 integer value n that satisfies this requirement is $n=1$. Solving
16 for $f=1/\lambda$ where λ is obtained by solving the above equation (4),
17 one obtains $f=57$ Hz. Therefore, the 57 Hz component of the LFM
18 signal will have a differential radius value equaling zero
19 whenever the specified embedding delay T equals 13 sample units.

20 FIGS. 8A-8C show the differential radius signals for the
21 three times series with a time delay of $T=28$. According to
22 equation (4), when the delay T equals 28, $\lambda=112$ msec and $f = 9$ Hz
23 for $n=0$. Because 9 Hz lies outside the LFM band of 50 to 90 Hz,

1 one must choose an appropriate value for n . For example, a
2 value of $n=3$ yields $f=62$ Hz and a value of $n=4$ yields $f=81$ Hz,
3 which correspond identically to the null points in the
4 differential radius plot of the pure LFM signal (FIG. 8A) at .3
5 seconds and .75 seconds, respectively.

6 Although the examples described above use the values of 13
7 and 28 for the delay T , there are many selections of the delay T
8 for which the differential radius will equate to zero, and the
9 number of choices grows as the signal bandwidth grows. Although
10 the selection of the delay T is described above in reference to
11 the pure LFM signal, this applies also to the LFM signal with
12 noise and the squared LFM signal. An appropriate value for the
13 delay T should be selected to best quantify the dynamics
14 underlying the evolution of a particular signal of interest.
15 For the case of aperiodic or chaotic signals, the first minimum
16 of the average mutual information (AMI) is one of the preferred
17 methods for selecting the delay value T .

18 Accordingly, the nonlinear signal processing system and
19 method of the present invention is used in data processing of
20 active sonar signals for enhanced detection and classification
21 and identification of nonlinearity in the signal structure. The
22 enhancement techniques, namely the chaotic and differential
23 radius, when operated on an original acoustic time series
24 measurement, provide precise, temporal and spatial detection

1 markers for evolving events as the time series unfolds. This
2 allows the detection of hidden or anomalous events in an
3 acoustic time series, associated with rapidly changing
4 evolutions in phase space, typical for a variety of nonlinear
5 and chaotic motions exhibited in a large number of acoustic
6 signatures.

7 In light of the above, it is therefore understood that
8 the invention may be
9 practiced otherwise than as specifically described.

1 Attorney Docket No. 79941

2

3 SYSTEM AND METHOD FOR PROCESSING AN UNDERWATER ACOUSTIC SIGNAL
4 BY IDENTIFYING NONLINEARITY IN THE UNDERWATER ACOUSTIC SIGNAL

5 ABSTRACT OF THE DISCLOSURE

6 A nonlinear signal processing system and method is used to
7 identify nonlinearity (e.g., chaos) in underwater acoustic
8 signals, such as sonar signals. The system and method detects
9 the underwater acoustic signal and digitizes the underwater
10 acoustic signal to produce an acoustic time series. The
11 acoustic time series is reconstructed using a phase space
12 embedding algorithm to generate a phase space embedded acoustic
13 signal. A differential radius signal is generated from the
14 phase space embedded acoustic signal using chaotic radius
15 computations and differential radius computations. Thresholds
16 can be detected in the differential radius signal to reveal
17 nonlinear or chaotic events hidden in the underwater acoustic
18 signal.

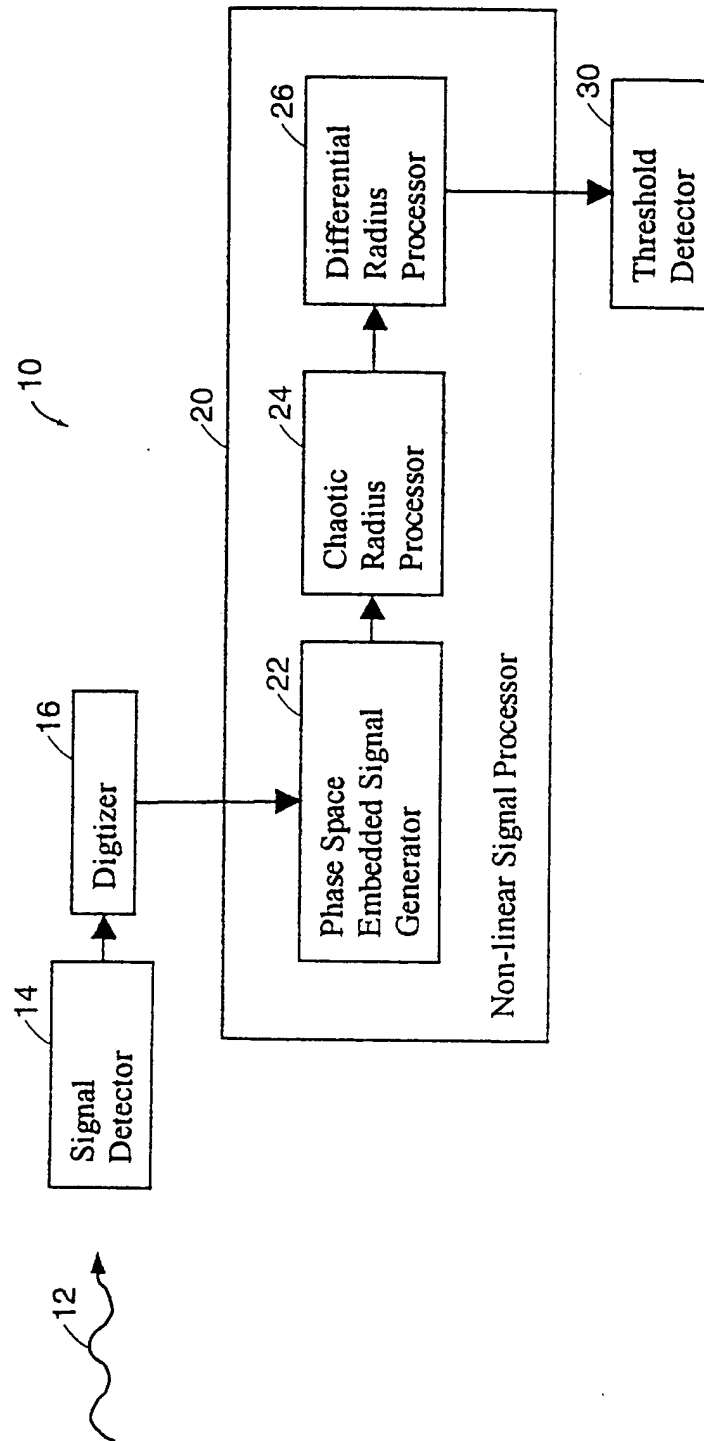


FIG. 1

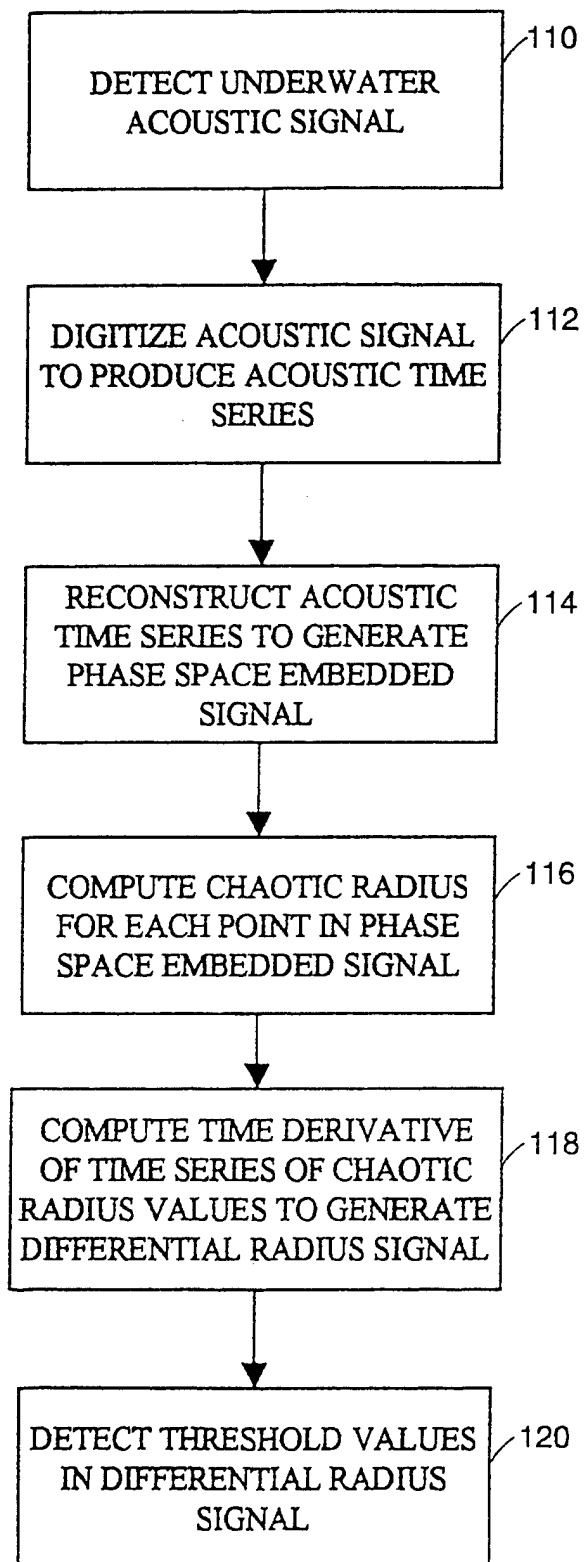


FIG. 2

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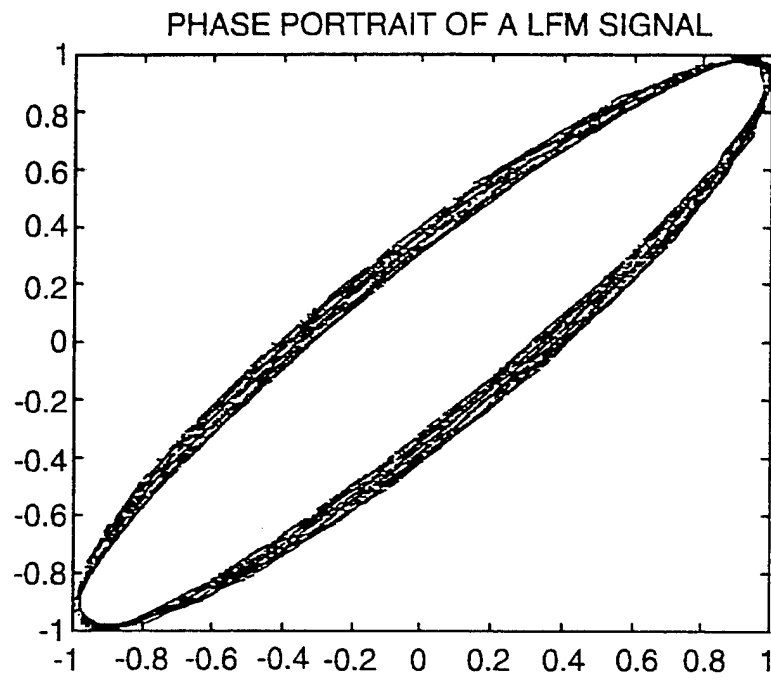


FIG. 3A

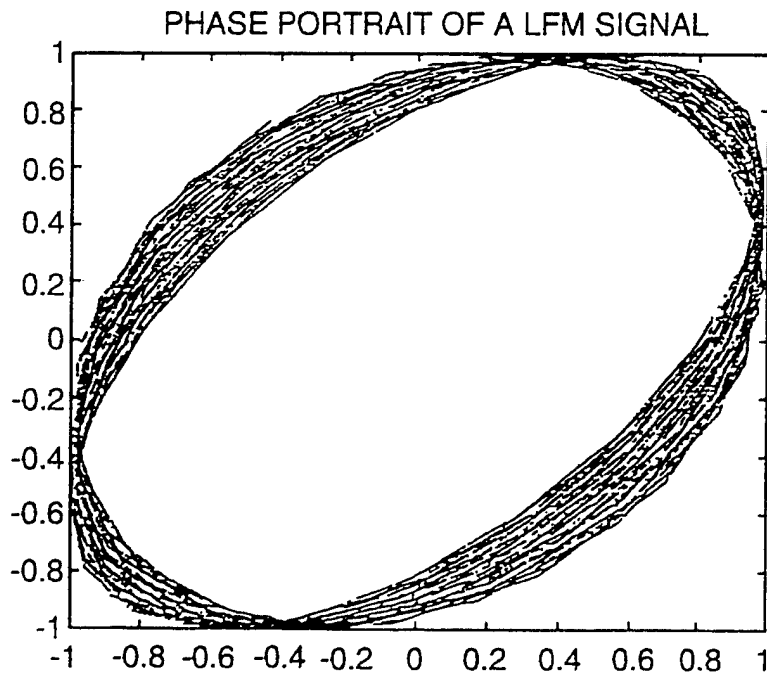


FIG. 3B

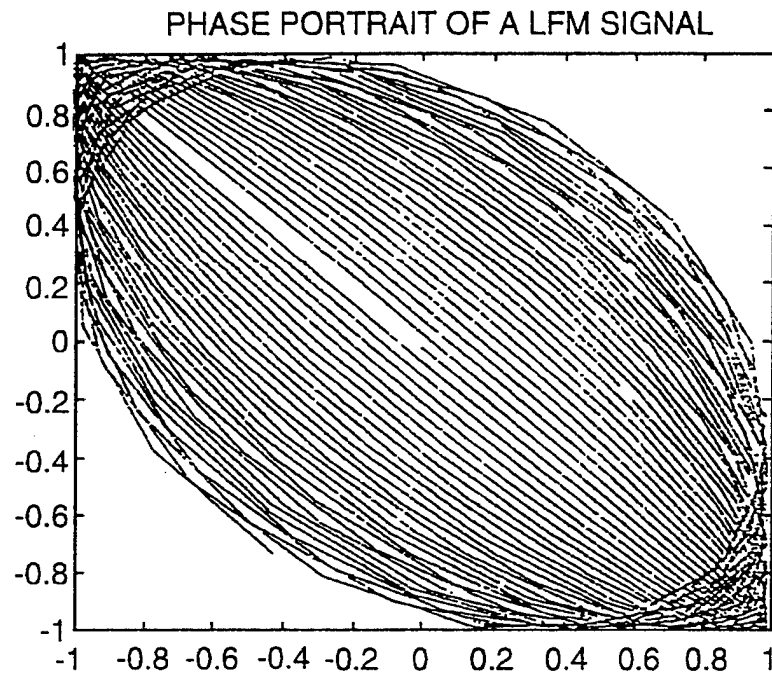


FIG. 3C

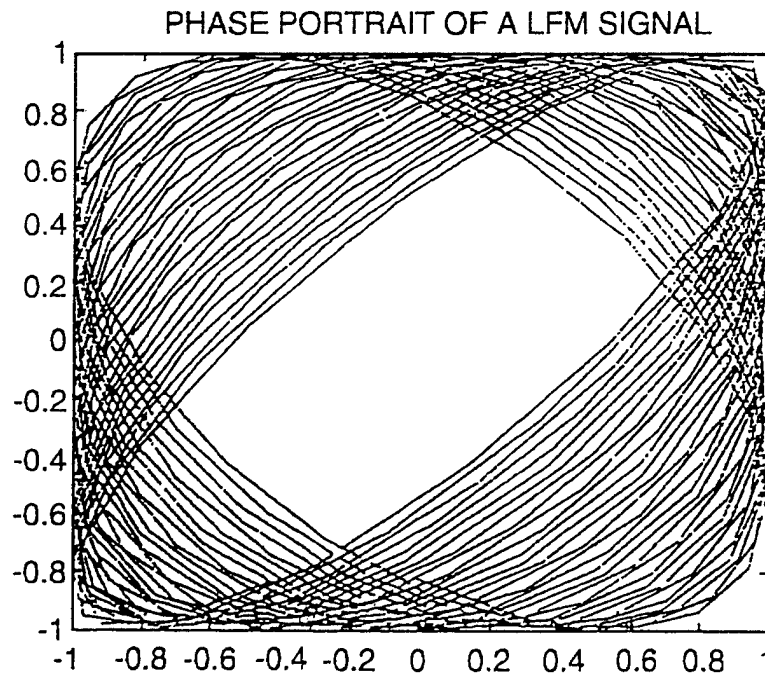


FIG. 3D

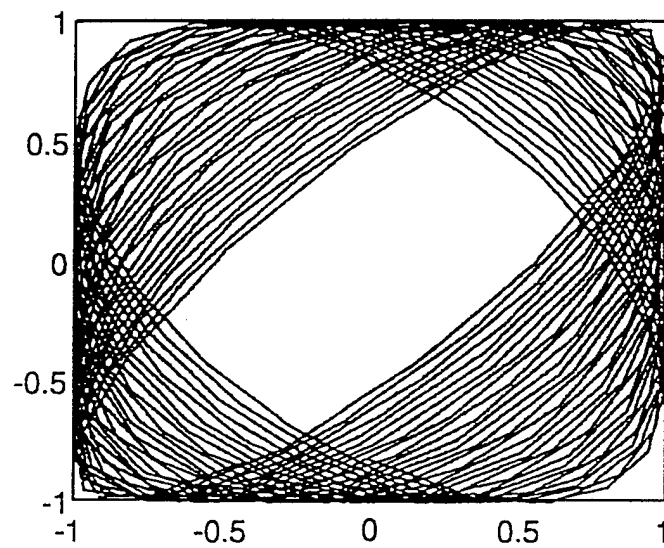


FIG. 4A

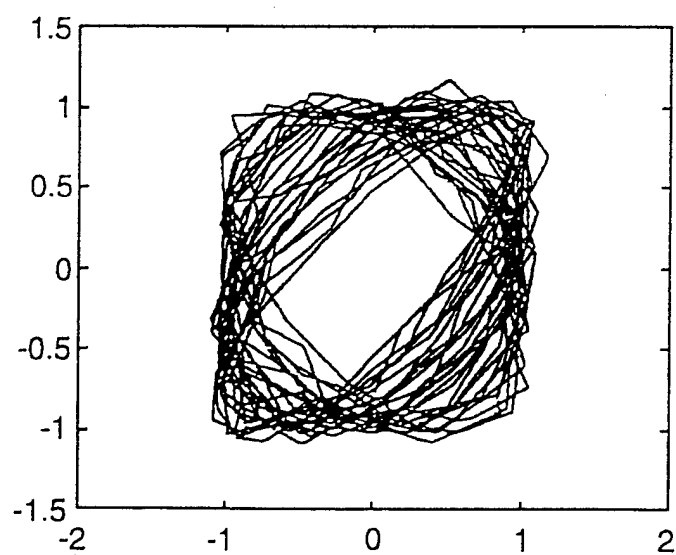


FIG. 4B

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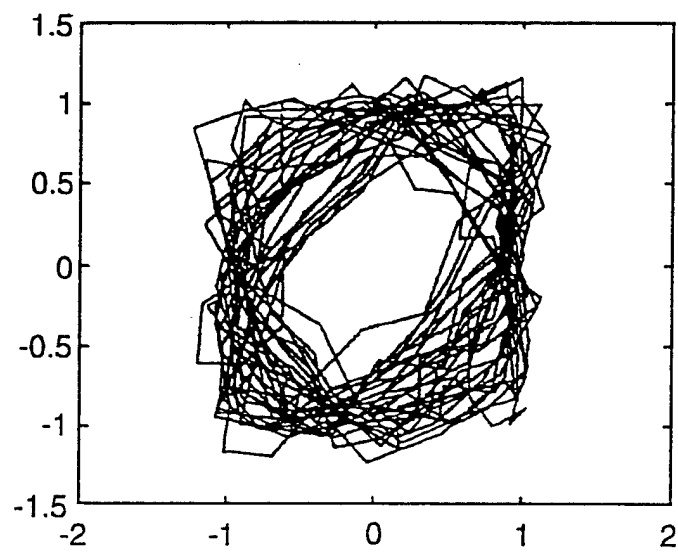


FIG. 4C

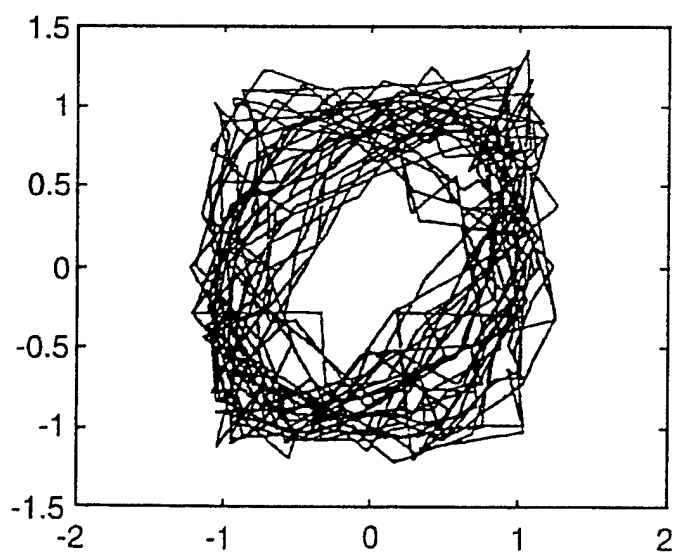


FIG. 4D

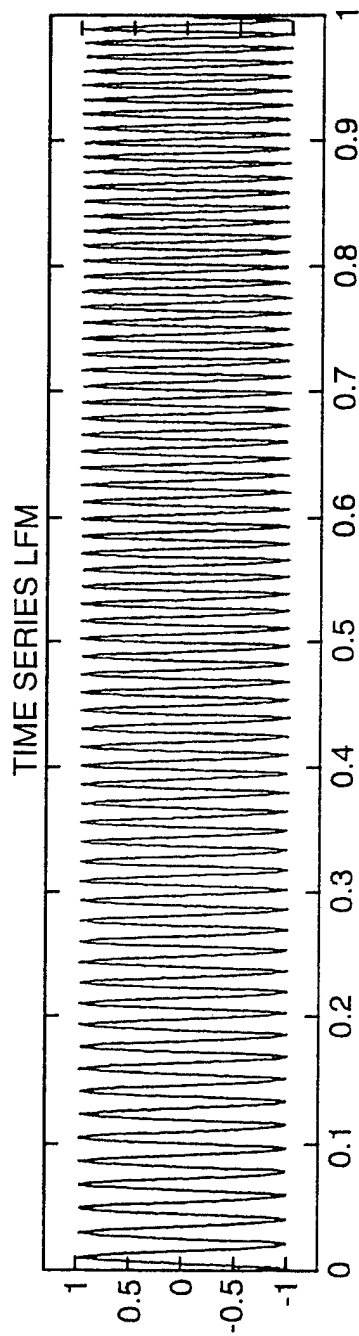


FIG. 5A

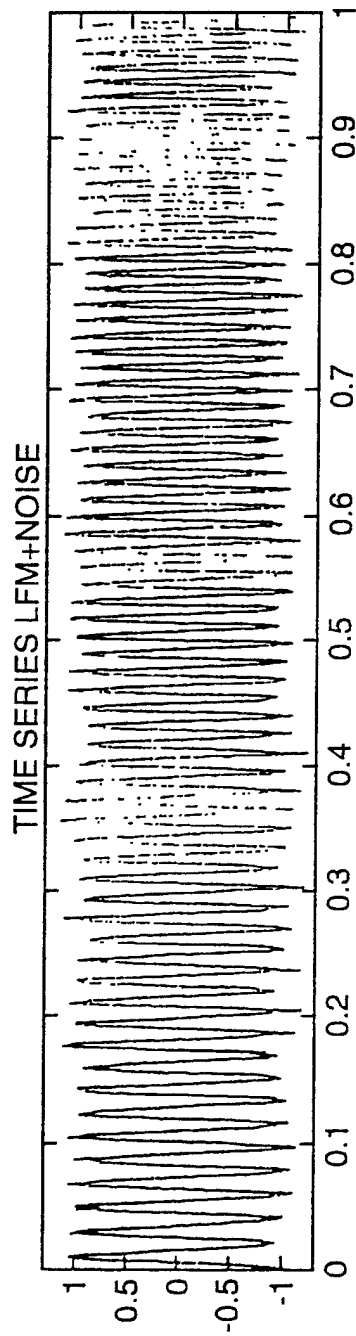


FIG. 5B

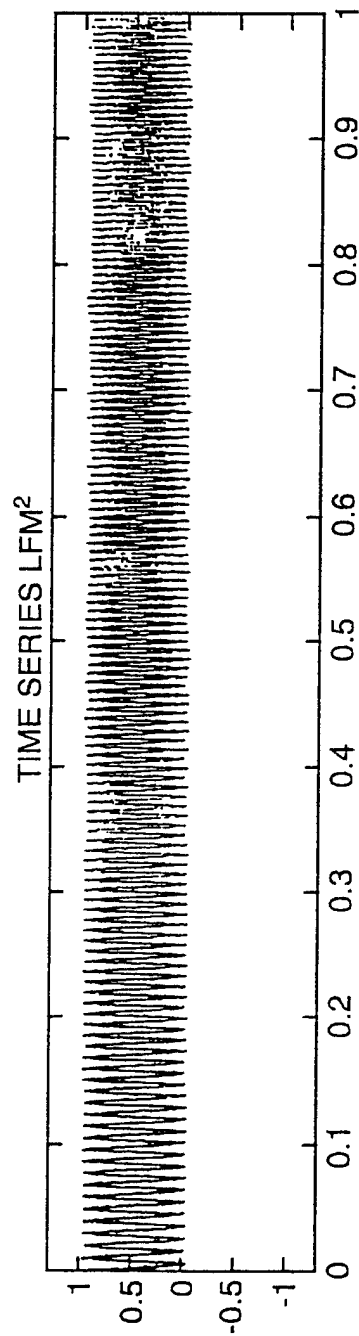
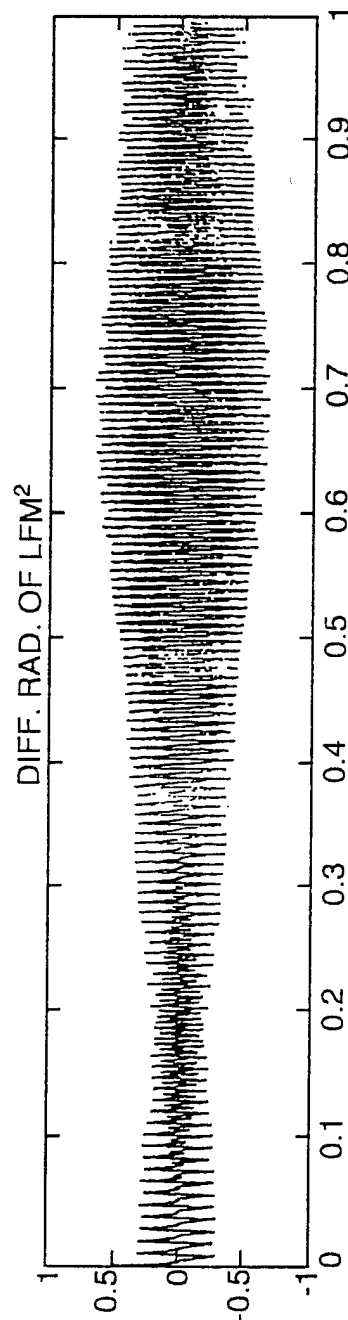
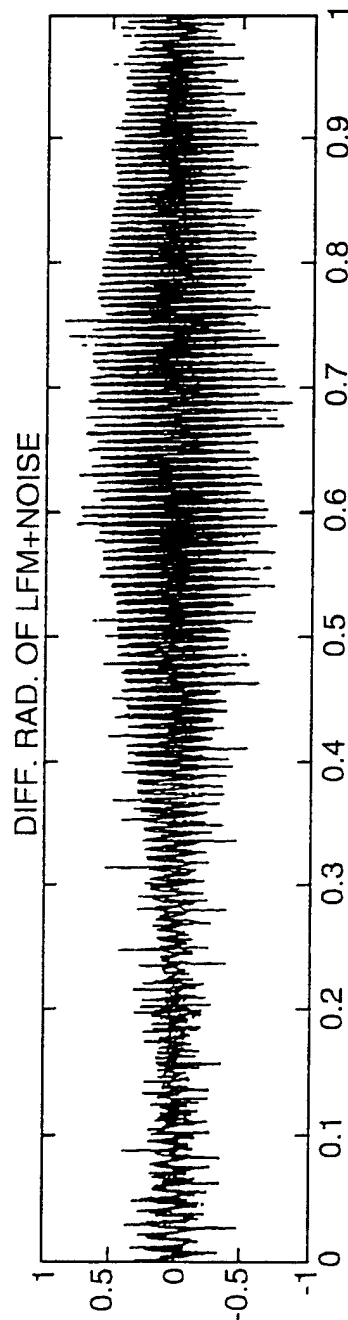
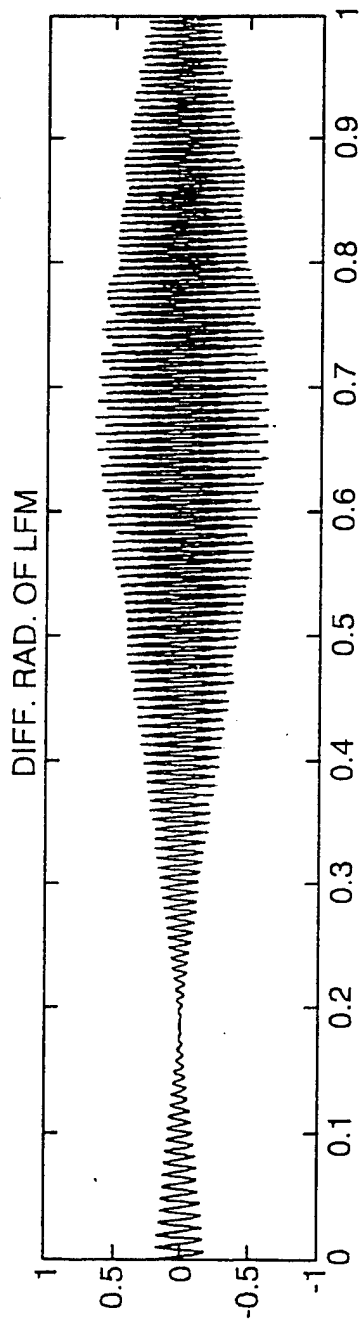


FIG. 5C



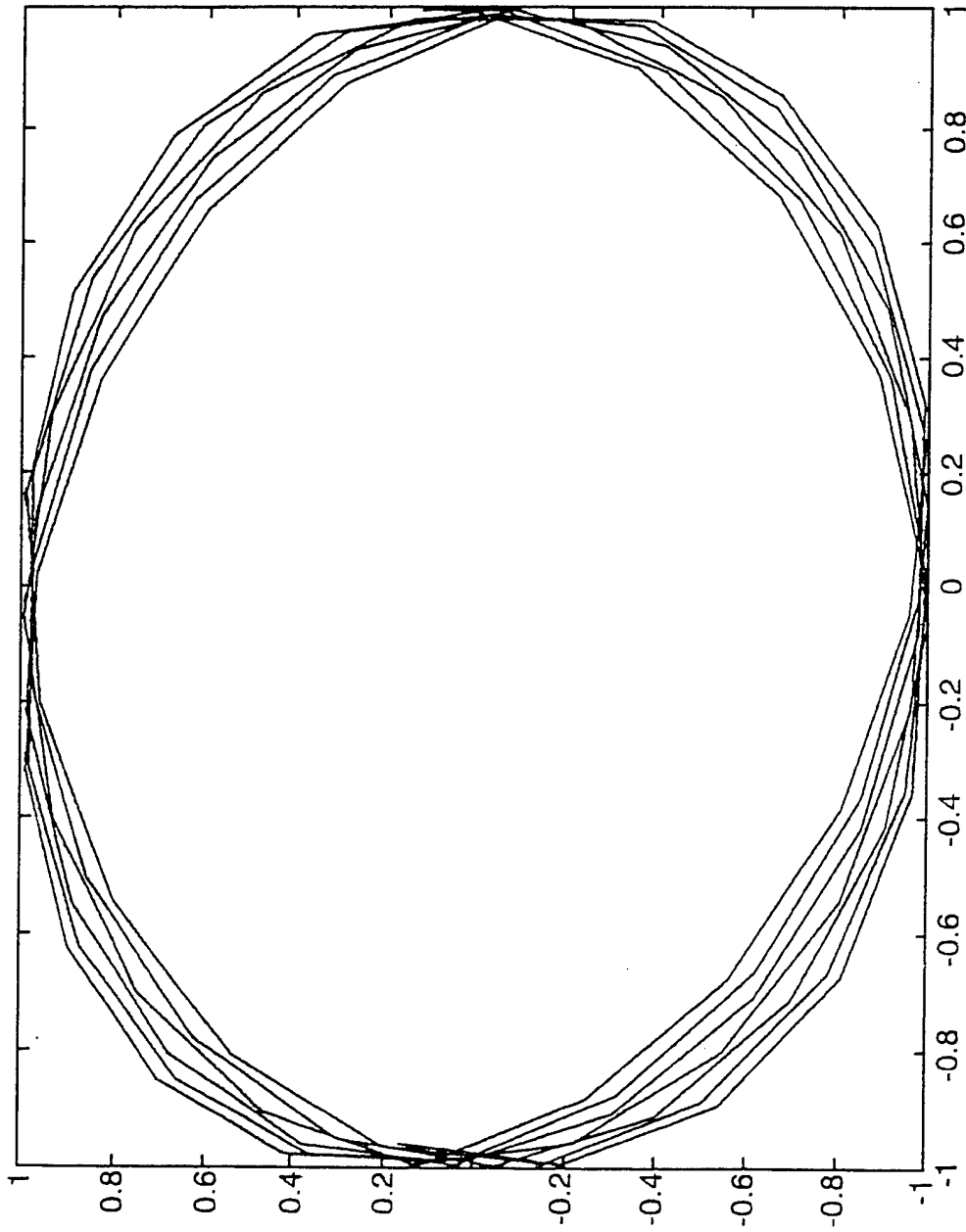


FIG. 7

