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(54) **Title**: MULTIPLE OUTPUT MULTIPLE USER METHODS AND/OR SYSTEMS OF UNDERWATER ACCOUSTIC COMMUNICATION

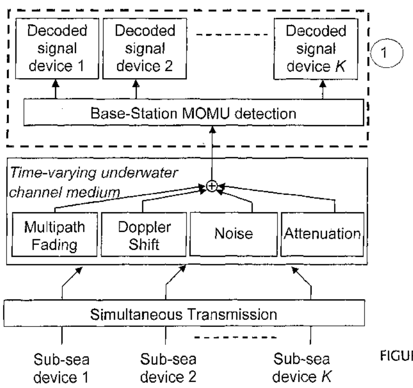

FIGURE 5

(57) **Abstract**: A method of signal processing for underwater acoustic communication between N acoustic transmitters and at least one M remotely located acoustic signal receiver, comprises the following steps: configuring said transmitters and said receiver to share a common frequency band; ranking signals representing the power of at least two signal transmissions received by said receiver from said transmitters; employing N adaptive feedforward equalisers with a soft base band; whereby the method equalises Inter-Symbol Interference from multipath propagation and phase fluctuations; configuring a (N-I) adaptive feedforward equaliser to output a soft-decision to a N adaptive feedforward equaliser; and cancelling Multiple Access Interference generated from simultaneous asynchronous or synchronous reception of a plurality of signals.

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MULTIPLE OUTPUT MULTIPLE USER METHODS AND/OR SYSTEMS OF UNDERWATER ACCOUSTIC COMMUNICATION

Field of the Invention

The invention relates to methods and/or systems of underwater acoustic communication as well as receivers, transmitters, and base units configured to carry out the methods.

Background and review of Art known to the Applicant

Underwater communications, employing sound propagation, is an area that is an immense task beset by many underwater environmental problems.

The receiver usually observes a superposition of signals sent by the active transmitters, as shown in Figure 1. The invention seeks to improve the performance in handling the processing of signals and overcoming or at least minimising the problems which arise from multiple user underwater acoustic communication.

Figure 2 shows an underwater virtual network with a base station 1, sub-sea units such as sub-sea unit 2, crawlers such as crawler 3, and a ROV/AUV 4. Downlink signals and uplink signals are shown as arrows such as arrows 5 and 6 respectively. The acoustic signal is symbolised by symbols such as symbol 7. Apart from resolving the problem of ISI (inter-symbol interference) arising from multipath propagation, Doppler shifts, environmental noise etc, the receiver in such scenarios has the additional task of mitigating the effects of co-channel interference from other users in the network system.

Whilst a wide variety of prior art embodiments exist, the closest prior art embodiments are described in detail in the following publication: H. K. Yeo, B. S. Sharif, A. E. Adams and O. R. Hinton, "Implementation of Multi-user Detection Strategies for Coherent Underwater Acoustic Communication", IEEE Journal of Ocean Engineering, vol. 27, Issue 1, pg. 17-27, January 2002. The contents of this publication are incorporated by reference.

The proposed multi-user detection systems and signal processing systems in accordance with these prior art embodiments are complex and require extensive processing. One of the objects of the invention is to improve the performance of the detection process and the signal processing.

The following further prior art documents are acknowledged: 1) Impact of ocean variability on coherent underwater acoustic communication during the Kaual experiment (KauaiEx) J. Acoust. Soc. Am. 123(2), February 2008 (Song et al); 2) US6813219; 3) US6819630; 4) Underwater communication link with iterative equalization IEEE 2006 (Oeberg et al.); 5) US2007/0071077; 6) US5301167; 7) WO94/10629; 8) US5905721; 9) US6985749; 10) BRADY, D and Preisig J C, "Underwater Acoustic Communication" Download from http://allegro.mit.edu/~gww/pdf/1998-poor-wornell-ph-ch8.pdf, on 10 Aug 2009. Created 19 Dec 2002. See Section 8.4.3 (pp 369-372); 11) Calvo, E and Stojanovic, M, "Efficient Channel-Estimation-Based Multiuser Detection for Underwater CDMA Systems", IEEE Journal of Oceanic Engineering, Vol. 33, No. 4, Oct 2008, pp502-512; 12) Gupta, C; Mumtaz, T; Zaman, M; Papandreou-Suppappola, A, "Wideband chirp modulation for FH-CDMA wireless systems: coherent and non-coherent receiver structures", *Communications 2003. ICC '03. IEEE International Conference on*, vol.4, no., pp. 2455-2459 vol.4, 11-15 May 2003; 13) Stojanovic, M and Freitag, L, "Multiuser undersea acoustic communications in the presence of multipath propagation", Proc of Oceans 2001, 5-8 Nov 2001, pp2165-2169. See Abstract and Fig. 2; 14) Hursky, P et al, "Comparison of two underwater acoustic communications techniques for multi-user access", Acoustical Society of America Journal, Vol. 115, Issue 5, p2469, May 2001; 15) Song A, Badiey M, Song HC, Hodgkiss WS, Porter MB, KauaiEx Group. Impact of ocean variability on coherent underwater acoustic communications during the Kaual experiment (KauaiEx). The Journal of the Acoustical Society of America. 2008 Feb; 123(2): 856-65.

Summary of the Invention

In a first broad independent aspect, the invention provides a method of signal processing for underwater acoustic communication between N acoustic transmitters and at least one M remotely located acoustic signal receiver, comprising the following steps: configuring said transmitters and said receiver to share a common frequency band; ranking signals representing the power of at least two signal transmissions received by said receiver from said transmitters; employing N adaptive feedforward equalisers with a soft base band; whereby the method equalises Inter-Symbol Interference from multipath propagation and phase fluctuations; configuring a (N-1) adaptive feedforward equaliser to output a soft-decision to a N adaptive feedforward equaliser; and cancelling Multiple Access Interference generated from simultaneous asynchronous or synchronous reception of a plurality of signals.

By contrast, the closest prior art uses throughout adaptive decision feedback equalisation instead of adaptive feed forward equalisers as in the first broad independent aspect.

Similarly, the document "Impact of Ocean Variability on Coherent Underwater Acoustic Communications During the Kaual Experiment" also fails to propose adaptive feed forward equalisers and in particular it fails to suggest a plurality of adaptive feed forward equalisers with a soft base band. It also fails to suggest configuring an (N-1) adaptive feed forward equaliser to output a soft-decision to an N-adaptive feed forward equaliser. It actually only suggests a one transmitter by one receiver system. There is no disclosure of multiple remote transmitters communicating with a receiver. It suggests throughout the use of a decision feedback equaliser (DFE) see for example column 1 on page 856, Figure 3, column 1 of page 859, column 1 of page 860.

US6813219 and US6819630 also use decision feedback equalisers throughout. There is no disclosure of employing N-adaptive feed forward equalisers with a soft base band; whereby the method equalises Inter-Symbol Interference from multipath propagation and phase fluctuations; configuring an (N-1) adaptive feed forward equaliser to output a soft decision to an N-adaptive feed forward equaliser.

By employing adaptive feed forward equalisers of the form suggested in the main claim, the processing complexity is considerably reduced which leads to a performance which is three to four times more efficient than the performance of the prior art embodiments detailed in the closest prior art.

In a subsidiary aspect, N is greater or equal to 2.

In a further subsidiary aspect, the method comprises the step of observing a signal contained within said signal transmission to determine if it has expanded or contracted within a predetermined time period.

In a subsidiary aspect, the method incorporates the steps of broadcasting signals to a selection of underwater devices to increase power transmission for reception.

In a subsidiary aspect, the method comprises the step of switching between multi-element and single element receiver output modes dependent upon the evaluation of power, multi-path, Doppler Shift and noise.

In a subsidiary aspect, the method comprises no step of broadcasting to transmitters to increase power transmission; and the steps of sending receiver signals to a MOMU (Multi-Output Multi-User) detector and returning a signal to individual transmitters.

In a subsidiary aspect, the method comprises the step of transmitting an identifying code utilising a hybrid of Pseudo-Random Binary Sequences and Linear Frequency Modulation (LFM).

In a second broad independent aspect, the invention provides a method of underwater acoustic signal transmission comprising the steps of arranging a number of distinct underwater acoustic devices with transmitters for acoustically transmitting to a base station with multiple-output hydrophones; and transmitting signals from said devices to said base unit over a common frequency bandwidth.

In a third broad independent aspect, the invention provides a method of underwater acoustic communication with a base station and a plurality of underwater devices forming a network of underwater receivers and transmitters, comprising the steps of broadcasting signals to a plurality of underwater devices from a base station; detecting power received for each underwater response; ranking said devices in terms of power; and broadcasting signals to a selection of underwater devices to increase power transmission for reception.

In a fourth broad independent aspect, the invention provides a method of underwater acoustic communication with a base station and a plurality of underwater devices forming a network of underwater receivers and transmitters, comprising the steps of broadcasting signals to a plurality of underwater devices from a base station; and switching between multi-element and single element receiver output modes dependent upon the evaluation of power, multi-path and noise.

In a fifth broad independent aspect, the invention provides a method of underwater acoustic communication with a base station and a plurality of underwater devices forming a network of underwater receivers and transmitters, comprising no step of broadcasting to transmitters to increase power transmission; and comprising the steps of sending receiver signals to a MOMU detector and returning a signal to individual transmitters.

In a sixth broad independent aspect, the invention provides a method of underwater acoustic communication with a base station and a plurality of underwater devices forming a network of underwater receivers and transmitters, comprising the step of transmitting an identifying code utilising a hybrid of Pseudo-Random Binary Sequences and Linear Frequency Modulation (LFM).

In a subsidiary aspect, the method comprises the step of back-to-back mis-matched filtering.

In a further subsidiary aspect, the method comprises the step of power ranking by measuring the correlation peak output for each transmitter.

In a seventh broad independent aspect, the invention provides a method of underwater signal processing comprising the steps of: receiving a signal; employing a window for observing a change of frequency component for a base-band symbol period which is carrier modulated to a passband signal; and employing a fast butterfly FFT (Fast Fourier Transform) for a time-domain window containing samples to determine a passband frequency; whereby Doppler shift is determined.

In a subsidiary aspect, said window takes the form substantially as defined in equation (12).

In a further subsidiary aspect, said fast butterfly FFT takes the form substantially as defined in equation (13).

In a further subsidiary aspect, said method further comprises the steps of 1) compensating by adding or substracting a frequency component; and then 2) down-mixing to baseband signals.

In an eighth broad independent aspect, said method of underwater signal processing comprises the step of providing one or more adaptive feedforward equalisers.

In a subsidiary aspect, said method comprises the steps of providing said adaptive feedforward equalisers with equaliser taps; sending predetermined training sequences for adapting said equaliser taps' weights; and switching to a decision directed mode.

In a further subsidiary aspect, said adaptive feedforward equaliser is configured to have a complex output substantially as defined in equation (14).

In a further subsidiary aspect, said adaptive feedforward equaliser defines a symbol error estimation substantially as in equation (15).

In a further subsidiary aspect, said adaptive feedforward equaliser defines a mean square error substantially as in equation (16).

In a further subsidiary aspect, said adaptive feedforward equaliser is configured to have a soft decision complex output based on maximum likelihood estimation where a probability density function is derived from a series of computed complex outputs as in equation (17) and the maximum likelihood estimation of said soft decision complex output is derived from equation (18).

In a further subsidiary aspect, said method incorporates a hard decision complex output operating in a first mode suitable for single user operation; and a second mode suitable for multi-user operation.

In a further subsidiary aspect, said first mode suitable for single user operation incorporates a data mode based on the adaptive decision of a user; the power estimate is set to zero; the maximum likelihood decision is set to zero; and the soft base-band decision is set to zero.

In a further subsidiary aspect, said second mode suitable for multi-user operation incorporates a multiple access interference cancellation step substantially based on equation (18) for soft base-band decision and substantially based on equation (19) for hard base-band decision.

In a further subsidiary aspect, said method incorporates interference cancellation steps substantially based on any one of equations (20) to (26).

In a ninth broad independent aspect, the invention provides a method of underwater signal processing, comprising the steps of: determining an input vector for single or multiple channel inputs to each user depending upon channel conditions; determining initial bits/symbols for N users from their corresponding adaptive feedforward equaliser unit whilst selecting a power estimation; and setting soft-decision base-band estimate to zero; feeding a power estimate and a complex soft decision base-band output from a first user to a second user's decoding block; regenerating bits/symbols with a weighting factor and phase correction; substracting the regenerated bits/symbols from a received signal to obtain a modified received signal; passing said modified received signal to a further stage for the removal of a further user signal; repeating the process of decision estimation, regeneration, weighting and interference cancellation for N-1 stages.

In a subsidiary aspect, the method further comprises the step of employing a buffer window to store the time reference or time delay estimation for each user.

In a further subsidiary aspect, a signal vector is fed in a soft-decision base-band parallel interference cancellation process which substantially takes the form of equation (20).

In a further subsidiary aspect, the adaptive feedforward equaliser derives retrievable information which for a user is decoded by summation of maximum likelihood at each stage according to equation (21).

In a further subsidiary aspect, the method comprises the step of obtaining energy statistics from received signals to rank users in descending power.

In a further subsidiary aspect, the method further comprises the step of neglecting users which are weaker than an intended or a stronger user.

In a further subsidiary aspect, the method further comprises the step of linearly relating the time complexity per bit/symbol to the number of users in the system.

In a tenth independent aspect, the invention provides a method of underwater signal processing, comprising the steps of:

obtaining energy statistics from received signals to rank users in descending power; performing adaptive symbol estimation of the strongest user; regenerating and cancelling estimated result from received signal; passing substracted received signal to the next weaker user for decoding; repeating the preceding steps to decode signals of other users; providing a feedback loop to cancel out summed effects from other users; summing, regenerating, weighting with phase correction and cancelling output decisions of a stronger user from other users to obtain a substracted signal; and decoding the subtracted signal whilst assuming that only background noise is present.

In an eleventh broad independent aspect, the method of underwater signal processing, comprises the steps of: obtaining energy statistics from received signals; randomly selecting a first user if equal energy statistic are obtained for each user; performing adaptive symbol estimation of the first user; regenerating and cancelling estimated result from received signal to obtain substracted received signal; passing substracted received signal to the next user for decoding; repeating the preceding steps to decode signals of other users; providing a feedback loop to cancel out summed effects from other users; summing, regenerating, weighting with phase correction and cancelling output decisions of said first user from other users to obtain a substracted signal; and decoding the subtracted signal whilst assuming that only background noise is present.

In a twelfth broad independent aspect, the invention provides an underwater communication system, for communication between N acoustic signal transmitter or transmitters and M remotely located acoustic signal receiver or receivers to form an underwater network comprising: means for said transmitter and said receiver to share a common frequency band for at least one signal transmission; means for ranking signals representing the power of at least two signal transmissions received by said receiver from said transmitter; means for said receiver to identify said transmitter; means for observing the information contained within said signal transmission to determine if said signal expanded or contracted within a predetermined time period; means for equalising Inter-Symbol Interference from multipath propagation and phase fluctuations; and means for Multiple-User Detection within said network.

In a subsidiary aspect, said transmitter and said receiver are configured to assign a predetermined code for establishing a transmission channel between said transmitter and said receiver over said frequency band.

In a further subsidiary aspect, said transmitter is configured to increase the power for said information transmission when said received ranked signals are below a predetermined power threshold.

In a further subsidiary aspect, said receiver is configured not to broadcast to transmitters to increase power transmission; the receiver incorporates a MOMU detector; said receiver is configured to send received signals to said detector and return a signal to individual transmitters.

In a further subsidiary aspect, said transmitter transmits an identifying code to said receiver.

In a further subsidiary aspect, said identifying code utilises a hybrid of Pseudo-Random Binary Sequences and Linear Frequency Modulation.

In a further subsidiary aspect, said means for equalising Inter-Symbol Interference further comprises a training means and a decision means.

In a further subsidiary aspect, said training means comprises a predetermined training code which simulates at least one received signal to initially configure said means for equalising Inter-Symbol Interference.

In a further subsidiary aspect, said decision means receives a plurality of transmitted symbols embedded within said signal; said symbols represent said information contained within said received signals for subsequent adaptation and equalisation.

In a further subsidiary aspect, said means for Multiple-User Detection further comprises a means for cancelling Multiple Access Interference generated within said network.

In a further subsidiary aspect, said means for cancelling Multiple Access Interference is configured to utilise said received transmitted information; said received information is weighted and reconstructed to form said received signal.

In a further subsidiary aspect, said received transmitted information is weighted and reconstructed to form said received signal is repeated over a plurality of stages.

In a further subsidiary aspect, the means for cancelling Multiple Access Interference within said network is MOMU soft-decision Parallel Interference Cancellation.

In a further subsidiary aspect, said means for cancelling Multiple Access Interference is configured to obtain the highest ranked signal representing a received signal; said highest ranked signal is multiplied with a weighting factor with phase correction and is then subtracted from the received signal.

In a further subsidiary aspect, said means for cancelling Multiple Access Interference is configured to be repeated over a plurality of stages; each stage processes the next ranked signal which is weaker than the previous signal.

In a further subsidiary aspect, said means for cancelling Multiple Access Interference within said network is MOMU soft-decision Successive Interference Cancellation.

In a further subsidiary aspect, one or more decision means are summed together, multiplied with a weighting factor, subtracted from the received signal and fed back to the highest ranked signal for Multiple Access Interference cancellation.

In a further subsidiary aspect, said means for cancelling Multiple Access Interference is configured to be repeated for each ranked signal.

In a further subsidiary aspect, said means for cancelling Multiple Access Interference is MOMU soft-decision Recursive Interference Cancellation.

In a thirteenth broad independent aspect, the invention provides an underwater device configured to operate the method and/or system of any of the preceding aspects.

Brief description of the figures

Detailed description of the figures

When referring to figure 1, an underwater acoustic network consists of a sub-surface device (base-station 1), communicating acoustically to and from a plurality of mobile or fixed sub-sea devices (2, 3, 4) of an underwater network. The downlink transmission 6 (base-station to sub-sea devices) consists of a command and control where the base-station will send a broadcast signal to a plurality of subsea devices to transmit wireless acoustic signal to the base-station. In the up-link transmission 5, sub-sea devices users transmit acoustic information over a time-varying channel. The base-station 1 is then required to demodulate the received data for each user as if there were only one user present, while treating the others as additive noise. The communication operation between base-station and sub-sea devices is interchangeable. Each node is bi-directional in an embodiment of the invention. However, this idea is beset by underwater environmental issues like multipath (giving rise to Inter-Symbol Interference ISI), signal attenuation, Doppler Shift, noise etc and system issues like co-channel interferences or multiple access interferences (MAI).

A Multiple Output Multiple User (MOMU) method and device is provided for resolving underwater environmental artefacts and asynchronous multiple access communication operating in the same frequency band in the physical layer.

System Model

In an uplink asynchronous communication model, shown in Figure 4, each user is observed to be arriving asynchronously at the base-station; this process is applicable or identical to down-link communication. The data field may consist either of 2 blocks:

1) The data field of the transmitted data consists of a header, S_k, user identification ID_k, training sequence, T_k , and D_k, variable data length. The header is used for initial "coarse" time synchronisation or clock recovery, and the training sequence is used to provide initial training to adapt the equaliser tap weights.

2) Without the training sequence, this is to allow events where signal drops off and the receiver needs to self-adapt to the data stream

Reference can be made to figure 3.

The received signal at the base-station can be expressed in complex form, $r(t)$ where

$$r(t) = \sum_{k=1}^{K} a_k h_k b_k\,(\mathrm{t} - \tau_k) + \mathrm{n(t)} \tag{1}$$

where $a_k(t)$, $h_k(t)$, $b_k(t)$ and τ_k denote, for each user k, the received amplitude, channel transfer function, transmitted bit/symbol sequence and time delay respectively, and $n(t)$ is the Additive White Gaussian Noise (AWGN). In order to model an asynchronous reception, consideration is given to the transmitted bit/symbol stream, b_k, of the kth user which takes the form

$$b_k \in \left\{\pm {}^{1}\!/_{\sqrt{2}};\ \pm {}^{1}\!/_{\sqrt{2}};\right\} \tag{2}$$

Thus generalising (1) becomes

$$r(t) = \sum_{k=1}^{K} \sum_{\mathrm{j=-M}}^{\mathrm{M}} a_k h_k b_k^j\,(\mathrm{t} - \mathrm{jT} - \tau_k) + \mathrm{n(t)} \tag{3}$$

where T is the bit/symbol duration. Symbol-epoch offsets are defined with respect to an arbitrary origin, $\tau_{\mathrm{ref}} = 0$, which is the time origin of the first detected user at the base-station, as shown in Figure 4.

Physical System

This section presents the design and development of several novel Multi-Output Multi-User (MOMU) detection strategy embodiments of the inventions for underwater acoustic communications, Figure 5 shows the system level architecture of the MOMU receiver and Figure 6 shows the top-level physical architecture of the MOMU receiver.

Figure 6 shows the MOMU system block diagram

Figure 7 shows the analog front end architecture of the system

Figure 8 shows the system block diagram of the down-mixing for the single output receiver

Figure 9 shows the block diagram of the single output receiver

Multiple distinct sub-sea devices transmitting over a same frequency bandwidth or distinct bandwidth, thereby increasing data throughput (Embodiment of the Invention)

In an embodiment of the invention, multiple distinct subsea acoustic devices transmits individual data stream acoustically via a projector to the base-station Multiple-Output hydrophones. Each distinct subsea device shares a common same bandwidth for transmission, it is possible that the same bandwidth may be multiplexed and split into multiple carriers to reduce the propagation delay spread within the symbol period for each user bandwidth. The acoustic streams transmitted by the subsea devices go through the underwater channel medium which can be defined by as a matrix channel which consists of multiple paths between the subsea devices transmit projector and the multiple-output receive hydrophones at the base-station. The simplified equation for the received vector, r, at the base-station from equation (1) is

$$r = \sum_{k=1}^{K} H_k b_k + n_k \qquad \textbf{(4)}$$

where H_k, b_k and n_k are the underwater channel matrix, transmitted acoustic data and noise vector for K users respectively. The number of multipath solutions or resolution of multipath at the base-station will depends on the physical limitation of the hydrophones available, L, Figure 10. The number of L hydrophones, L – 1 equates the number of solutions for multipath resolution.

ID detection

Channel Capacity

For the number of user capacity contain within the underwater network, an embodiment of the invention can operate with a closed loop or an open loop defined as follows:

1) Closed loop – Base-station will broadcast to plurality of subsea devices and detects the power received for each sub-sea response which will vary in distances from the base-station. The base-station will then rank the power distribution of each user via conventional Singular Value Decomposition (SVD) technique and power allocation via Waterfilling technique. The base-station then re-broadcast to subsea devices that

are further away to increase power transmission for reception. This ensures that the received power levels for each user at the base-station are in par, if not almost in par. However this leads to a shorter operation life if the subsea devices are operating on battery pack (application specific). However, if the subsea devices are attached to external entity with power source, this technique will be applicable. The signals are then fed into the base-station Multiple-Output Multiple-User (MOMU) detector to decode the signal and return the original signal to the individual users

$$Capacity_{Closed\_loop} = E\left[\max_{Q} log_2 det(I + USU^H)\right] \quad (5)$$

2) Open loop - Base-station will broadcast to plurality of subsea devices and detects the power received for each sub-sea response which will vary in distances from the base-station. The base-station will then rank the power distribution of each user via conventional Singular Value Decomposition (SVD) technique and power allocation via Waterfilling technique. However, the base-station will not re-broadcast to subsea devices that are further away to increase power transmission for reception. This process will be applicable for subsea devices operating with battery packs. The power levels for each user at the base-station are not in par, or unequal. The received signal will be send into the base-station MOMU detector within the base-station to decode the signal and return the original signal to the individual users.

$$Capacity_{Open\_loop} = \max_{Q} E[log_2 det(I + HH^H)] \quad (6)$$

Hybrid Linear Frequency Modulation (LFM) and Pseudo-Random Binary Sequences (Embodiment of the Invention)

An embodiment of the invention utilises a hybrid Linear Frequency Modulation (LFM), chirp signal – with wide ambiguity function against Doppler Shift and Pseudo-Random Binary Sequences (PRBS) like Gold, Kasami code, or other PRBS sequence with good auto-correlation and low cross-correlation properties to transmit signals.

The individual properties of LFM chirp signal (1954, US Patent 375/285 ; 333/14; 333/20; 333/28R; 342/201; 375/306; 380/32; 455/111) and PRBScodes are well known and it is not the intention here to reiterate their properties. These disclosures are incorporated by reference.

However, applying the individual LFM and PRBS code in underwater communication are beset by several limitations:

1) LFM chirp signal – although LFM has wide ambiguity function against Doppler shift effect, which is more Doppler Shift tolerant. In order to achieve a low cross-correlation factor in multi-user/network environment, a high bandwidth is required to provide enough orthogonal bands for chirp signals so that interference can be minimised and cross-correlation factor will be low. The feature of high bandwidth is not possible in a bandwidth limited underwater communication environment that operates in only a Kilo Hertz (KHz) bandwidth.

2) PRBS like Gold or Kasami or other code – exhibits high auto-correlation yet low cross-correlation properties. However, Doppler tolerance, $f_{DOPPLER}$, of PRBS code is limited by its bandwith, B_w

$$f_{DOPPLER} = \frac{1}{B_w}$$

Reference to figure 21 can be made.

An embodiment of the invention is to employ – a hybrid LFM/PRBS Code, with high auto-correlation and low-crosscorrelation factor. A baseband PRBS code 3 chip sequence n = ‘0 1 0’ (top of figure 21), which can be $n=2^m-1$ chips, is synthesized into a modulated signal (second view of figure 21) and LFM signal (third view of figure 21). The individual synthesized signals of PRBS code, with high auto-correlation and low-crosscorrelation factor, and LFM are combined prior to transmission (bottom of figure 21).

Users ID strength detection and ranking via mis-matched filtering (Embodiment of the Invention)

The objective of the ID detection by the base-station is to identify the sub-sea devices that have transmitted acoustically to the base-station. Each subsea–device transmits a carrier modulated PRBS code, Pseudo-Random Binary Sequences, which consists of a hybrid combination of Linear Frequency Modulation (LFM) and PRBS code, which offers good auto-correlation, low cross-correlation factor and wide ambiguity function against Doppler shift.

In the event that the base-station receiver knows a priori the sub-sea device that is transmitting back a data stream, the base-station can employ detection of the ID based on auto-correlation:

$$M_i = \sum_{i=1}^{j} P_k P_{k+i} \tag{7}$$

An ideal code sequence to employ for ID sequence detection is a 13 chip Barker Code [+1 +1 +1 +1 +1 -1 -1 +1 +1 -1 +1 -1 +1] which gives the following auto-correlation output.
Baker code is just an example of ID sequence where one can change the PRBS, with good auto-correlation and low cross-correlation properties accordingly.

Figure 11 shows the element M_i is the main lobe, peak at index 13, while the remaining side lobes are zeros. This maximum peak, with zeros side lobes ensures that probability of false detection is kept to minimum, ie near 0.

This is an ideal case of ID sequence detection for a single user ID detection. However, due to the fact that the Barker code only has 1 PRBS with such properties, it is not possible to allocate different ID's for each users. The alternative is to employ mis-match filtering.

In a cross-correlation, *R*, of 2 or more sequences, *C* and *D* of period *i* is

$$R_i = \sum_{i=1}^{j} C_k D_{k+i} \tag{8}$$

Taking *C* as a sequence to be cross-correlated with *D* which is composed of complex numbers, the sequences *C* and *D* are related by the weighting factor, *T*.

$$D_i = T_i C_i \tag{9}$$

In this instance, D is considered a "mis-matched filter" to C and the normalised filter D can be expressed as,

$$\sum_{i=1}^{j} D_i^2 = j \tag{10}$$

Figure 12 shows the cross-correlation of a hybrid 2 PRBS codes/LFM (2 users) 15 chips sequences, it can be seen that the side lobes for each sides of index 13 are high, which increases the probability of false detection once more asynchronous users reception are in the system as the cross-correlation factor will grow with each user, although the theoretical users capacity (not the scope of this invention) with a 15 chip sequence is $2^N - 1 = 2^4 - 1 = 15$. This is an example of a 15 chips hybrid sequence PRBS code/LFM, the system is configured to be adaptable by for example adopting 2^n-1 chips depending on network capacity.

However, employing a back-to-back mis-matched filtering, the sidelobes within the two peaks are ensured to be zero, see Figure 13. This ensures and allows:

1) False detections for multiple user IDs are kept to minima by averaging the side lobes values between the 2 peaks
2) Propagation delay spread (multipath) in the time-domain via cross-correlation mainly contain the channel transfer function, $h(t)$.

Power Ranking (Embodiment of the Invention)

Figure 14 shows the header for each user data stream consisting of a N series of back-to-back PRBS hybrid Code/LFM codes to differentiate its ID, Pseudo-Random Binary Sequences. Power ranking is measured by the correlation peak output for each user.

Doppler Shift

Doppler Shift arises from the physical movement of either the sub-sea device or base-station due to the underwater time-varying current. The time-domain passband signal will either expand or contract in time, this is equivalent to the change in carried frequency. A one-way Doppler Shift from either side of the acoustic transmission entity can be expressed as

$$\Delta f = \frac{v}{c} f \tag{11}$$

Doppler Shift Compensation (Embodiment of the Invention)

The acoustic transmission time from the base-station to the sub-sea devices last usually for few hundred milliseconds and Doppler Shift is considered minimum. However the transmission time from the sub-sea device to the base-station can last for a few seconds to tens of seconds per burst, therefore Doppler Shift compensation is considered for the uplink path, although it is applicable for both directions.

A Doppler Shift compensation is proposed here. For each base-band symbol period, T_s, which is carrier modulated to a passband signal, when undergoing Doppler Shift, it will either expand or contract with time. A window is used to observe this change of frequency component (Figure 15),

$$T_G = T_s + 2\Delta T_s \tag{12}$$

The term $2\Delta T_s$ denotes a partial past symbol period and future symbol period.

For each time-domain window containing $r(n)$ samples, the frequency content can be determined by performing a fast Butterfly FFT such as fixed point embedded algorithm within a general DSP,

$$R^*(k) = \sum_{n=0}^{N-1} r^*(n) W_N^{*kn} \tag{13}$$

where $W_N^{*kn} = e^{-j^{2\pi}/_N}$. The passband frequency determined by the FFT will be the Δf in (11), or Doppler Shift. This frequency drift will be added or subtracted from f to perform the down-mixing of the complex signals.

Multiple-Output Multiple-User (MOMU) architecture

Multiple-Output Adaptive Feedforward Equaliser (Embodiment of invention)

1) Multiple-Output (MO) adaptive Feedforward Equaliser
2) Soft–Decision Output for Multi-User detection strategy – i.) Single-User mode – set to zero. ii.) Multi-user mode - Summation of Multi-User (MU) Soft-Decision interference cancellation from interfering users back to intended user for signal cancellation
3) Hard-Decision Output – i.) Single-User mode – Detected data output sequence ii.) Multi-User mode - Summation of Hard-Decision fed back to intended user for hard-decision detection.

As each subsea units/remote devices are transmitting narrow bandwidth data each simultaneously. The requirement for an adaptive DFE (Decision Feedback Equaliser) is not required for 2 reasons:

1) The uplink bandwidth for each remote devices is narrow, therefore the channel propagation delay spread (or multipath) will tend not to exceed each symbol period, T_s
2) Adopting a DFE in a multi-user environment will introduce feedback erroneous cancellation that will propagate for several symbols long when the data packet of an unintended user arrives. This is due to the DFE having a sudden surge in the MSE (Mean Square Error) and will need to adapt the filters coefficients accordingly.

The primary task of the adaptive Feedforward Equaliser (FE), shown in Figure 16, is to equalise the ISI (Inter-symbol interference) which result from multipath propagation and phase fluctuations. The single-output feedforward equalisers for each user are used to remove ISI and provide phase compensation.

The operation of the adaptive FE is divided into two phases – the training mode and decision directed mode. In the training mode, a short priori known training sequence which is embedded in the receiver system is used as the desired signals to provide initial training for adapting the equaliser tap weights. At the end of the training mode, the equaliser would have attained convergence close to the optimal values, convergence is considered to be obtained when the MSE is below 3dB point. In the event where a training sequence is not available, the system will adapt blindly to minimise the Mean Square Error. The receiver then switches to the decision directed mode where the detected symbols are treated as the desired signal for further adaptation and equalisation so that variations in the channel can be tracked.

At time nT_s, where $T_s < {}^{1}\!/_{Bw}$, the complex output of the adaptive FE for user K is

$$a_k^*(n) = f_k^* X_k^*(n) \tag{14}$$

where f_k^* and X_k^* are the complex filter coefficients, nT_s spaced samples buffered in the feedforward filters for user K at time nT_s.

The symbol error estimation can be defined as

$$e_K^*(n) = \hat{d}_K^*(n) - a_K^*(n) \tag{15}$$

where $\hat{d}_K^*(n)$ and $a_K^*(n)$ are the hard-decision decision is the pre-decision variable. The corresponding mean square error (MSE) is depicted as

$$MSE_K = E\{|e_K^2(n)|\} \tag{16}$$

The soft decision complex output, $\tilde{s}_k^*$, is based on Maximum Likelihood Estimation (MLE), where a probability density function, $Z_{\tilde{s}_k^*}$, is derived from $a_{k(n)}^*$, a series of computed complex output from the adaptive feedforward equaliser from user k, $a_{k1}^*, a_{k2}^*, \ldots\ldots a_{k(m-1)}^*$,

$$Z_{\tilde{s}_k^*}(a_{k1}^*, a_{k2}^*, \ldots\ldots a_{k(m-1)}^* | \tilde{s}_k^*) \tag{17}$$

and the MLE of complex $\tilde{s}_k^*$, is determined from

$$L(\tilde{s}_k^*) = \sum_{n=1}^{m} logZ_{\tilde{s}_k^*}\left(a_{k(n)}^* \middle| \tilde{s}_k^*\right) \tag{18}$$

The hard decision complex output, $\hat{d}_k^*$ operates in 2 modes:

1) Single user mode –

   i.) Phase 1 - Training mode, $\hat{d}_k^*$ complex output is based on the priori known training sequence of user k.

ii.) Phase 2 - Data mode, $\hat{d}_k^*$ complex output is based on the adaptive decision of user *k*.

iii.) In single user mode, the Power Estimate is set to '0', thus the input to the feedforward filter only contains $r_{L_{Base}}^*(n)$ complex signal.

iv.) MLE $L(\hat{d}_k^*)$ set to zero

v.) Soft base-band decision $\tilde{s}_k^*(n)$ set to zero

2) Multi user mode –

i.) Phase 1 – Training mode, $\hat{d}_k^*$ complex output is based on the priori known training sequence of user k.

ii.) Phase 2 – MAI cancellation, $\hat{d}_k^*$ complex output based on equation (18) for *M*th stage interference cancellation.

$$L(\hat{d}_k^*) = sgn\left(\sum_{n=1}^{m} logY_{\hat{d}_k^*}\left(a_{k(n)}^* | \hat{d}_k^*\right)\right) \tag{19}$$

where $Y_{\hat{d}_k^*}$ is the probability density function derived from a series of $Y_{\hat{d}_k^*}(a_{k1}^*, a_{k2}^*, \ldots\ldots a_{k(m-1)}^* | \hat{d}_k^*)$.

The Multiple Output architecture comprises of a series of receiver of Figure 16, shown in Figure 17.

Multiple Output Multiple-User (MOMU) Detection Strategy

Apart from the problems accrued from time-varying multipath propagation and Doppler fluctuations, the capacity and performance of a network system is also limited by multiple access interference (MAI). As the Multiple-Output adaptive Feedforward equaliser does not take into account the existence of MAI from other users, by which each user in the system is detected separately without regard for other users. The effect of MAI will become substantial as the number of interferences or power differences increases in the network system. A better detection strategy is one of Multiple Output Multi-User Detection (MOMU). Here, the information of multiple users is used jointly in order to better detect each individual user in the system. By utilising MOMU algorithms, there is significant added benefit in providing reliable communication in a network system.

In this section, 3 methods of MOMU schemes for multiple access interference cancellation are proposed and developed:

i.) MOMU Soft decision base-band Parallel Interference Cancellation (SDBB-PIC)

ii.) MOMU Soft decision base-band Successive Interference Cancellation (SDBB-SIC)

iii.) MOMU Soft decision base-band Recursive Interference Cancellation (SDBB-RIC).

MOMU Soft Decision Base-Band Parallel Interference Cancellation (SDBB-PIC) MU Strategy (Embodiment of the Invention)

Multi-user receiver structure based on parallel interference cancellation (PIC) estimates and subtracts all the MAI for k users concurrently. The performance of PIC can be improved by performing an initial partial cancellation. The partial cancellation involves multiplying the estimated symbol of each user with a factor less than unity prior to any interference cancellation. This takes into account the fact that the tentative decisions made in the earlier stages are less reliable than those of the later stages. However, the act of employing these PIC schemes in underwater acoustic communications is unrealistic since these MUD structures, developed in mobile communications, do not take into account the predominant effects of ISI, phase fluctuations and Doppler effects encountered in the underwater acoustic environment.

The prior art pass-band PIC MUD structure based on weighted parallel interference cancellation (PIC) adopts a DFE (decision feedback equaliser) for each user. The complex hard decision output for each user is weighted and used to re-constructs the pass-band signal for parallel interference cancellation. However, there are 2 limitations with such prior art MUD detection with increased users.

i.) With the increase in users, the probability of the feedback path of the DFE propagating errors will increase during the first stage of MAI cancellation, therefore increasing the MSE for each user and it will take a longer time for the adaptive filters to converge.

ii.) With the increased feedback propagated error via the DFE for each user, the re-generated pass-band carrier frequency component may be distorted which then results in signal phase cancellation.

By contrast, in an embodiment of the invention, a *Mth*-stage MOMU receiver structure based on soft decision base-band weighted parallel interference cancellation (PIC) with adaptive FE is proposed.

Input to Stage 1

The input vector, $r^*_{L\_Base}(n)$ and $r^*_{Base}(n)$, are the single channel L input or multiple channel input to each of the user, depending on the channel conditions.

The conditions of selecting single channel or multiple channels are based on the following:

a.) Power ranking (invention 4) – determine how many users are detected.

b.) Time domain channel impulse response – based on 10.log(Time Domain Impulse Response), shown right of Figure 22.

c.) Signal to Noise Ratio (SNR) measurement – based on a measurement of background noise and signal level.

Output from Stage 1

In stage 1 of the soft-decision base-band PIC structure, the initial bits/symbols for all users $k = 1,2,3, \ldots\ldots K$, are estimated from the corresponding adaptive FE units. The power estimation, P_w, for all users and the soft-decision base-band estimate, $\tilde{s}^*_K$, is set to zero.

Input to Stage 2 till Stages (M-1)

In stage 2 to (M-1), a circular rotation for each user is performed within the DSP by which each user is rotated by one step for MAI cancellation. For clarity sake, the power estimate, $P_{w,1}$, and complex soft decision base-band output, $\tilde{s}^*_1(n)$, from user 1 is fed into user 2 decoding block.

The estimated bit/symbol decisions are then regenerated with a weighting factor and phase correction. The regenerated signal is then subtracted from the received signal at the receiver array elements. The modified received signal, having one fewer interfering signals, is then passed to the next stage for processing and the removal of a further user signal. This process of parallel decision estimation, regeneration, weighting and interference cancellation is repeated for M stages, where the last stage are where all MAI have been removed between users. The T_b^{M-1} term in Figure 18 denotes the time delay of the received signal to be summed with the regenerated MAI signals of other users at the *(M-1)th* stage.

A buffer window is used to store the time reference or time delay estimation for each user for asynchronous reception. At stage M-1, the signal vector that is fed either to single element input vector, $\mathrm{r}^*_{\mathrm{L\_Base}}(\mathrm{n})$ or array elements $\mathrm{r}^*_{\mathrm{Base}}(\mathrm{n})$ of user K, at stage K-1, in the soft-decision base-band parallel interference cancellation (SDBB-PIC) in stage K-1, is expressed as:

$$X_{k,M-1} = \sum_{k=1}^{K} a_k^{*l} h_k^{*l} b_k^{*l}(t - nT_b^{M-1}) - \sum_{k=1}^{K-1} P_{w.k}^{*}(n) \, . \, L\left(\sum_{m=1}^{M-1} \sum_{k=1}^{K-1} \tilde{s}_{k,m}^{*}(n)\right) + \rho(n) \quad \textbf{(20)}$$

where the second term and third term are the power-estimation, soft-decision for all users, $k = 1,2,3,\ldots(K-1)$, and the residual noise, respectively.

At stage K, the retrieved information from the adaptive FE for user K from the output of stage M is decoded by the summation of maximum likelihood at each stage,

$$\hat{d}_K^*(n) = L\left(\sum_{m=1}^{M-1} \hat{d}_{K,m}^*(n)\right) = sgn\left(\sum_{n=1}^{m} logY_{\hat{d}_k^*}\left(a_{k(n)}^* \middle| \hat{d}_k^*\right)\right) \quad (21)$$

In order for a new set of signals to be regenerated for better data estimation in the next stage, the PIC structure assumes that the decision of the previous stage has been estimated correctly. Therefore any estimation error, contributed by any user, will degenerate the removal of MAI for other users. This problem arises in a "near-far" scenario, where the received signal for the weak user, coupled with the strong MAI from the other users that are fed into the adaptive feedforward equaliser (FE) structure, will encounter difficulties in estimating the data correctly. Therefore, it can be seen that the PIC receiver structure is superior in a well-power-controlled channel, where all signals from separate users are at an almost equal power level.

***MOMU* Soft Decision Base-Band Successive Interference Cancellation (Embodiment of the Invention)**

By contrast to the adaptive PIC receiver structure, the successive interference cancellation (SIC) strategy uses a successive approach towards MAI cancellation. If a decision has been made about the interfering strong user's bit/symbol, then the interfering signal can be regenerated at the receiver and subtracted from the received signal. The resulting subtracted signal should then be free from the strong interfering signal. However, this assumption relies greatly on the accuracy of the decision made for the interfering signal; if this decision is incorrect it will double the contribution of the interfering weaker signal. Once the interfering signal is stripped away from the received signal, the signal processing side of the receiver takes the view that the resulting signal contains one fewer users. The process is then repeated with the other weaker users, until the last user (weakest user) has been demodulated. A prior art SIC structure uses decisions produced by single-user matched filters, which neglects the presence of the interfering signals. This approach has been reported to perform well in a near-far situation under AWGN.

The SIC receiver structure, proposed here employs a soft decision base-band weighted parallel interference cancellation (SDBB_SIC) as shown in Figure 19. It is assumed, without loss of generality, that the powers of $k = 1,2, \ldots,K$ users are in descending order and that perfect delay estimation is achieved. In this case, the strongest user, K $= 1$, correct bit/symbol decision is regenerated by multiplying with a weight factor, $P_{w,k}$, with phase correction and is then subtracted from the received signal. Therefore, this approach aims to remove the strongest MAI from the received signal. The detector then makes a decision for the next strongest user ($k = 2$) from the subtracted signal. This process of decision-making, regeneration, weighting, phase correction, and cancellation from the received signal continues until the weakest or last user, K, has been decoded. The retrieved information symbol from the output of user K adaptive FE in stage M can also be determined from equations (20) - (21), however, it should be noted that the MAI reductions are performed successively.

The technique of removing the MAI of the strongest user from the received signal aids in the estimation of signals for weaker users. Therefore SIC can be seen to be superior in a non-well-power-controlled channel. However, one prime disadvantage of such system is that the strongest user does not benefit from the reduction of MAI, which means that the summed MAI effects from all other weaker users will, to a certain degree, affect the correct data estimation for the strongest user. The embodiment presented here for the adaptive MOMU SIC strategy can be described in the following algorithmic form:

1) Obtain energy statistics from the received signal to rank users in descending power.
2) Perform adaptive symbol estimation of strongest user (amplitude and phase).
3) Estimated result is regenerated and cancelled from received signal.
4) Subtracted received signal is passed to the next weaker user for decoding.
5) Repeat 1) to 4) until the last or weakest user K is decoded.

The practical implementation features of the MOMU SIC can be summarised as follows:

1) Prior to adaptive signal processing, knowledge of the received power for all users in the network cell is required so that interference cancellation can be performed successively. Any errors in the estimation translate directly into additive interference for further decision making for weaker users.
2) Users weaker than the intended user are neglected for SIC.
3) The delay time for demodulation for SIC grows linearly with the number of users.
4) Time complexity per bit/symbol is linearly related to the number of users in the system.

Adopting the SIC strategy has a number of general advantages. Firstly, the receiver has the best chance of estimating the correct decision for the strongest user in the system. Secondly, removing the strongest user in the system gives the most benefit to the remaining weaker users. The SIC structure can be considered to be effective if the received power for users are widely variable. A major shortcoming of the adaptive SIC processor is that its performance is asymmetric, where users of equal received power are demodulated with disparate reliability. This is the opposite of the adaptive PIC, which means that the summed MAI effects from all other weaker users will, to a certain degree, affect correct data estimation.

***MOMU* Soft Decision Base-Band Recursive Interference Cancellation** (An embodiment of the Invention)

The embodiment presented here is based on MOMU soft decision base-band recursive interference cancellation (SDBB-RIC), shown in Figure 20. In the case of unequal power reception, the receiver, having a priori knowledge of K users, first detects and obtains statistics from the received signal, r(t), to rank the users in order of descending power. The selector then switches to the corresponding adaptive FE of the detected strongest user. Subsequently, the output decision of the strongest user is regenerated, multiplied by the weighting factor, $P_{w,k}$, with phase correction and is then cancelled from the received signal. The subtracted received signal is then passed to the next strongest user for decoding as if the received signal consists of K – 1 user. This process is repeated until the last (weakest) user has been decoded. The distinctive feature of the SDBB-RIC structure, as compared to the SIC structure, is the feedback loop in Figure 20. This allows the strongest user to cancel out the summed effects from other users. The output decisions from all other users are summed, regenerated, weighted with phase correction and cancelled from the received signal. With the subtracted signal, the strongest user is decoded again, with the assumption that only background noise is present. Decoding for the rest of the users is then performed for a predefined number of loops.

In the case of equal power reception, all users are placed with the same priority and the selector switches to the first available user for adaptive FE estimation. The procedure of decoding of K equal power users is the same as that of unequal power reception, except there is now only an arbitrary priority between users.

Assuming again that the power of $k = 1, 2, ..., K$ users are in descending order and there is perfect delay estimation. The information symbols for user K are retrieved successively as described in previous section, this aspect is identical to the SDBB-SIC structure. During the loop back, the output symbol decisions for users $k = 2, 3, ..., K$, are summed, regenerated and cancelled from the received signal. The subtracted signal that is fed-back to user 1 can then be decoded free from MAI of other users. The output from the adaptive FE of user 1 at $(M+1)$th stage can be expressed as

$$\hat{d}^*_{1,(M+1)} = L\left(\textstyle\sum_{m=2}^{M} \hat{d}^*_{1,m}(n)\right) \tag{22}$$

where the signal, that is fed to single element single element input vector, $r^*_{L_{Base}}(n)$ or array elements $r^*_{Base}(n)$ of user 1, is

$$X_{1,M+1} = \sum_{k=1}^{K} a_k^{*l} h_k^{*l} b_k^{*l}(t - nT_b^{M+1}) - \sum_{k=2}^{K} P^*_{w.k}(n)\,.\,L\left(\sum_{m=2}^{M}\sum_{k=2}^{K} \tilde{s}^*_{k,m}(n)\right) + \rho(n) \tag{23}$$

Expanding (22) leads to

$$\hat{d}^*_{1,(M+1)} = L\left(\sum_{m=2}^{M} \hat{d}^*_{1,m}(n)\right) = sgn\left(\sum\nolimits_{n=1}^{m} logY_{\hat{d}^*_1}\left(a^*_{k(n)}\middle|\hat{d}^*_1\right) - c_{1,M}(n)\right)$$

(24)

where $c_{1,M}(n)$ is for the removal of postcursor ISI for user 1 at stage (M + 1). At stage 2M, for a single loop-back, the retrieved information for user K is depicted as:

$$\hat{d}^*_{K,2M}(n) = sgn\left(\sum\nolimits_{n=1}^{m} logY_{\hat{d}^*_1}\left(a^*_{k(n)}\middle|\hat{d}^*_1\right) - c_{K,(2M-1)}(n)\right)$$

(25)

where the single element input vector, $r^*_{L\_Base}(n)$ or array elements $r^*_{Base}(n)$ of user K, is

$$X_{K,2M-1} = \sum_{k=1}^{K} a_k^{*l} h_k^{*l} b_k^{*l}(t - nT_b^{M+1}) - \sum_{k=1}^{K-1} P^*_{w.k}(n) \,.\, L\left(\sum_{m=2}^{2M} \sum_{k=1}^{K-1} \tilde{s}^*_{k,m}(n)\right) + \rho(n)$$

(26)

Operation of this detector can be described in the following algorithmic form:

i) Obtain statistics from the received signal to rank users in descending power. If the received powers are equal, switch to the adaptive FE for the first available user. Process steps ii) to vii) of the algorithmic flow.

ii) Perform adaptive FE symbol estimation of the strongest/first user (amplitude and phase).

iii) Estimated result is regenerated and cancelled from received signal.

iv) Subtracted received signal is passed to the next strongest user adaptive FE for decoding.

v) Repeat i) to iv) until the last or weakest user K is decoded.

vi) Decisions of all subsequent users are then summed, regenerated, cancelled from the received signal and fed back to the strongest user for MAI cancellation.

vii) Repeat ii) to v) where regeneration and MAI cancellation is performed for the next weaker user, k = 2, 3, 4, ... K for a pre-defined number of iterations.

The advantages of implementing the adaptive SDBB-RIC structure are threefold. Firstly, the SDBB-RIC structure offers the flexibility to self-adapt to handle equal or unequal power reception. In the case of equal power reception, the SDBB-RIC structure operates identically as the PIC structure. Whereas in an unequal power reception, the recursive loop back feature allows the strongest user to benefit from the reduction of MAI from other users. Secondly, with the MAI reduced, the adaptive FE block that is incorporated for each user can then effectively cope with the multipath fading propagation and inter-symbol interference (ISI). And finally, implementing the SDBB-RIC structure can effectively tackle the problem of power control inefficiency in horizontal-link communication. Therefore, the adaptive SDBB-RIC MUD structure manifests itself to be a superior candidate for implementation in both well-power-controlled and non-well-power-controlled channels. From the analysis, the adaptive SDBB-PIC receiver structure has a processing load of $(K \times M)$, whereas the adaptive SDBB-SIC receiver structure has only a processing load of M. Although the adaptive SDBB-RIC receiver structure has an increased load, $(2M)$, compared to the SDBB-SIC structure, it requires a much lower computational load than the SDBB-PIC structure. One major gain of the SDBB-RIC over the SDBB-SIC MUD structure is that users weaker than the intended user are accounted for by the loop-back feature.

Claims

1. A method of signal processing for underwater acoustic communication between N acoustic transmitters and at least one M remotely located acoustic signal receiver, comprising the following steps:

configuring said transmitters and said receiver to share a common frequency band;

ranking signals representing the power of at least two signal transmissions received by said receiver from said transmitters;

employing N adaptive feedforward equalisers with a soft base band; whereby the method equalises Inter-Symbol Interference from multipath propagation and phase fluctuations;

configuring a (N-1) adaptive feedforward equaliser to output a soft-decision to a N adaptive feedforward equaliser; and

cancelling Multiple Access Interference generated from simultaneous asynchronous or synchronous reception of a plurality of signals.

2. A method according to claim 1, comprising the steps of broadcasting signals to a selection of underwater devices to increase power transmission for reception.

3. A method according to either of the preceding claims, further comprising the step of switching between multi-element and single element receiver output modes dependent upon the evaluation of power, multi-path, Doppler Shift and noise.

4. A method according to either claim 1 or claim 3, comprising no step of broadcasting to transmitters to increase power transmission; and the steps of sending receiver signals to a MOMU (Multi-Output Multi-User) detector and returning a signal to individual transmitters.

5. A method according to any of the preceding claims, comprising the step of transmitting an identifying code utilising a hybrid of Pseudo-Random Binary Sequences and Linear Frequency Modulation (LFM).

6. A method according to claim 5, comprising the step of back-to-back mis-matched filtering.

7. A method according to any of the preceding claims comprising the steps of receiving a signal; employing a window for observing a change of frequency component for a base-band symbol period which is carrier modulated to a passband signal; and employing a fast butterfly FFT (Fast Fourier Transform) for a time-domain window containing samples to determine a passband frequency; whereby Doppler shift is determined.

8. A method according to claim 7, wherein said window takes the form substantially as defined in equation (12).

9. A method according to either of claims 7 and 8, wherein said fast butterfly FFT takes the form substantially as defined in equation (13).

10. A method according to any of claims 7 to 9, wherein said method further comprises the steps of 1) compensating by adding or subtractinga frequency component; and then 2) down-mixing to baseband signals.

11. A method according to any of the preceding claims, comprising the steps of providing said adaptive feedforward equalisers with equaliser taps; sending predetermined training sequences for adapting said equaliser taps' weights; and switching to a decision directed mode.

12. A method according to any of the preceding claims, wherein said adaptive feedforward equalisers are configured to have a complex output substantially as defined in equation (14).

13. A method according to any of the preceding claims, wherein said adaptive feedforward equalisers define a symbol error estimation substantially as in equation (15).

14. A method according to any of the preceding claims, wherein said adaptive feedforward equaliser defines a mean square error substantially as in equation (16).

15. A method according to any of the preceding claims, wherein said adaptive feedforward equalisers are configured to have a soft decision complex output based on maximum likelihood estimation where a probability density function is derived from a series of computed complex outputs as in equation (17) and the maximum likelihood estimation of said soft decision complex output is derived from equation (18).

16. A method according to any of the preceding claims, wherein said adaptive feedforward equaliser incorporates a hard decision complex output operating in a first mode suitable for single user operation; and a second mode for multi-user operation.

17. A method according to claim 16, wherein said first mode suitable for single user operation incorporates a data mode based on the adaptive decision of a user; the power estimate is set to zero; the maximum likelihood decision is set to zero; and the soft base-band decision is set to zero.

18. A method according to claim 16, wherein said second mode suitable for multi-user operation incorporates a multiple access interference cancellation step substantially based on equation (18) for soft base-band decision and substantially based on equation (19) for hard base-band decision.

19. A method according to claim 16, wherein said method incorporates interference cancellation steps substantially based on any one of equations (20) to (26).

20. A method according to any of the preceding claims, comprising the steps of determining an input vector for single or multiple channel inputs to each user depending upon channel conditions; determining initial bits/symbols for N users from their corresponding adaptive feedforward equaliser whilst selecting a power estimation; and setting soft-decision base-band estimate to zero; feeding a power estimate and a complex soft decision base-band output from a first user to a second user's decoding block; regenerating bits/symbols with a weighting factor and phase correction; subtracting the regenerated bits/symbols from a received signal to obtain a modified received signal; passing said modified received signal to a further stage for the removal of a further user signal; repeating the process of decision estimation, regeneration, weighting and interference cancellation for N-1 stages.

21. A method according to claim 20, further comprising the step of employing a buffer window to store the time reference or time delay estimation for each user.

22. A method according to either of the preceding claims 20 and 21, wherein a signal vector is fed in a soft-decision base-band parallel interference cancellation process which substantially takes the form of equation (20).

23. A method according to claim 22, wherein the adaptive feedforward equaliser derives retrievable information which for a user is decoded by summation of maximum likelihood at each stage according to equation (21).

24. A method according to any of claims 20 to 23, further comprising the step of obtaining energy statistics from received signals to rank users in descending power and feeding in a soft-decision base-band successive interference cancellation process or recursive interference cancellation process.

25. A method according to claim 24, further comprising the step of neglecting users which are weaker than the strongest user.

26. A method according to either of the preceding claims 24 and 25, further comprising the step of linearly relating the time complexity per bit/symbol to the number of users in the system.

27. A method according to claim 1, wherein the method comprises the steps of obtaining energy statics from received signals to rank users in descending power; performing adaptive symbol estimation of the strongest user; regenerating and cancelling estimated results from received signal; passing subtracted received signal to the next weaker user for decoding; repeating the preceding steps to decode signals of other users; providing a feedback loop to cancel out summed effects from other users; summing, regenerating, weighting with phase correction and cancelling output decisions of a stronger user from other users to obtain a subtracted signal; and decoding the subtracted signal whilst assuming that only background noise is present.

28. An underwater device configured to operate the method of any of the preceding claims.

29. Signal processing apparatus configured to operate in accordance with the method of any of claims 1 to 27.

30. Computer software which configures a signal processing apparatus to operate according to the method of any of claims 1 to 27.

31. An underwater communication system substantially as hereinbefore described with reference to and/or as illustrated in any appropriate of the accompanying text and/or figures.

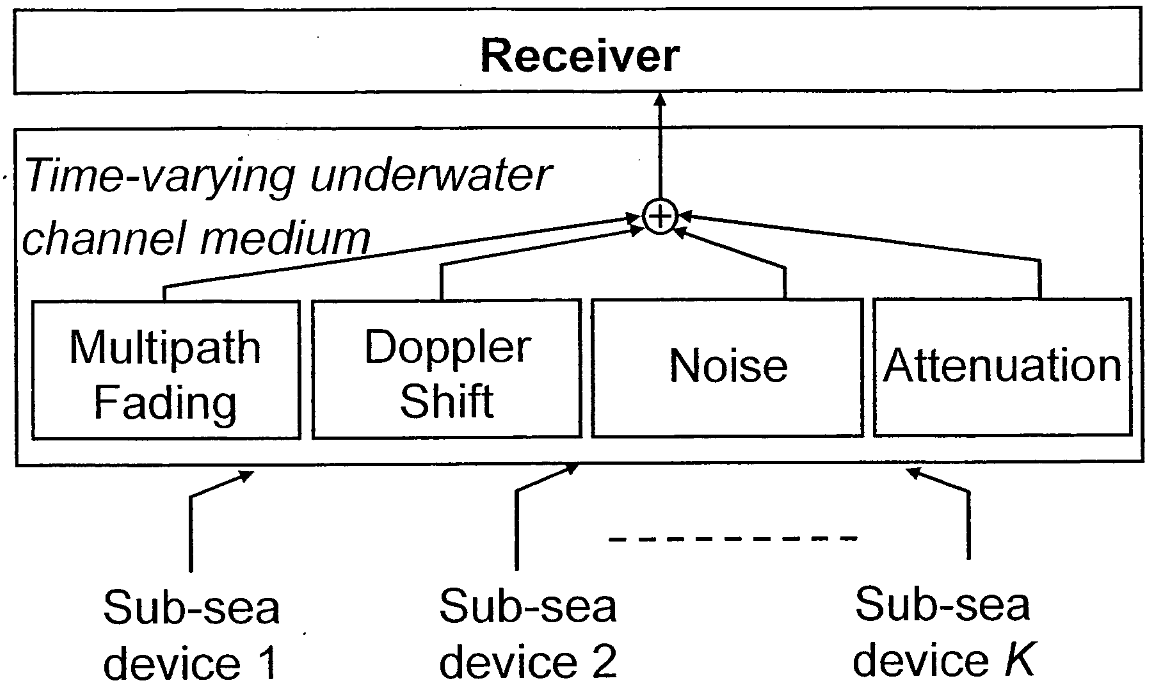


FIGURE 1

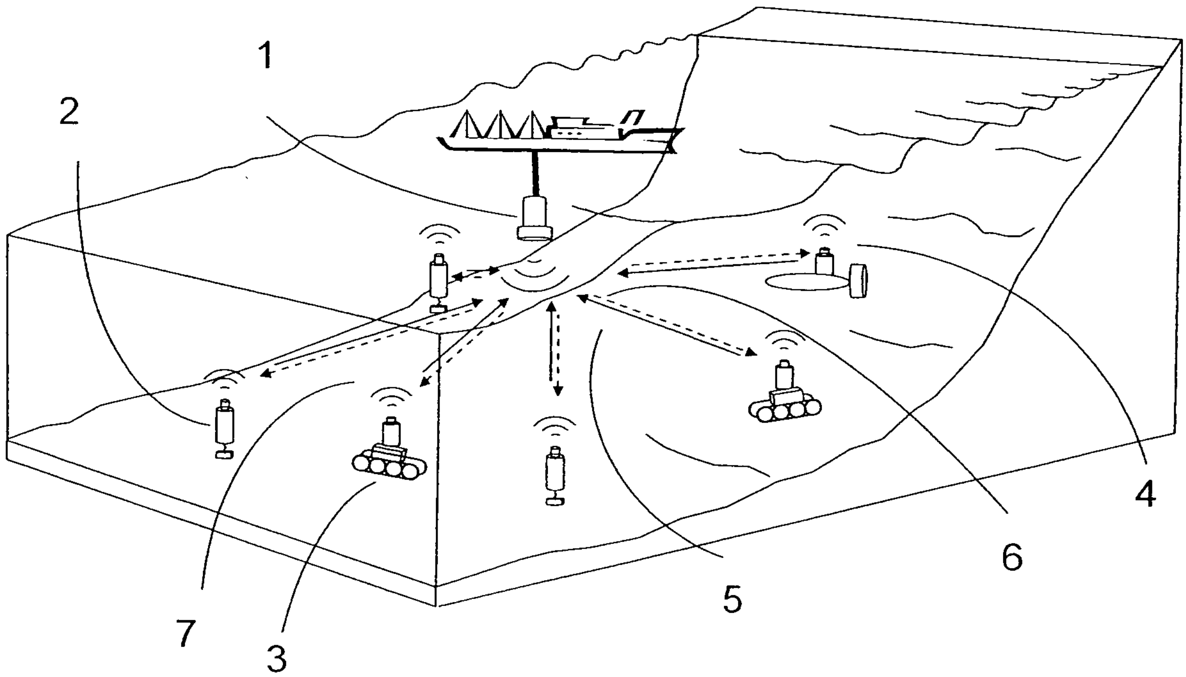


FIGURE 2

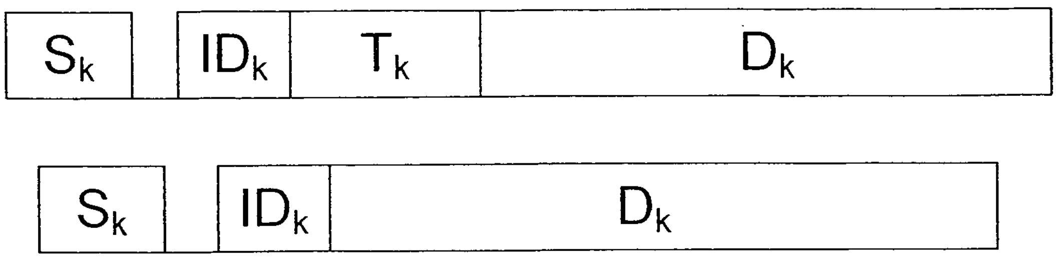


FIGURE 3

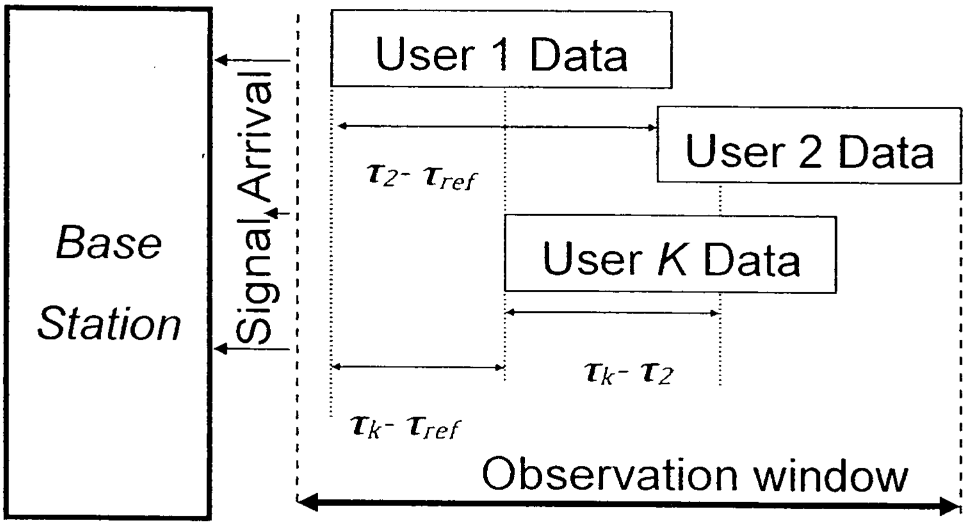


FIGURE 4

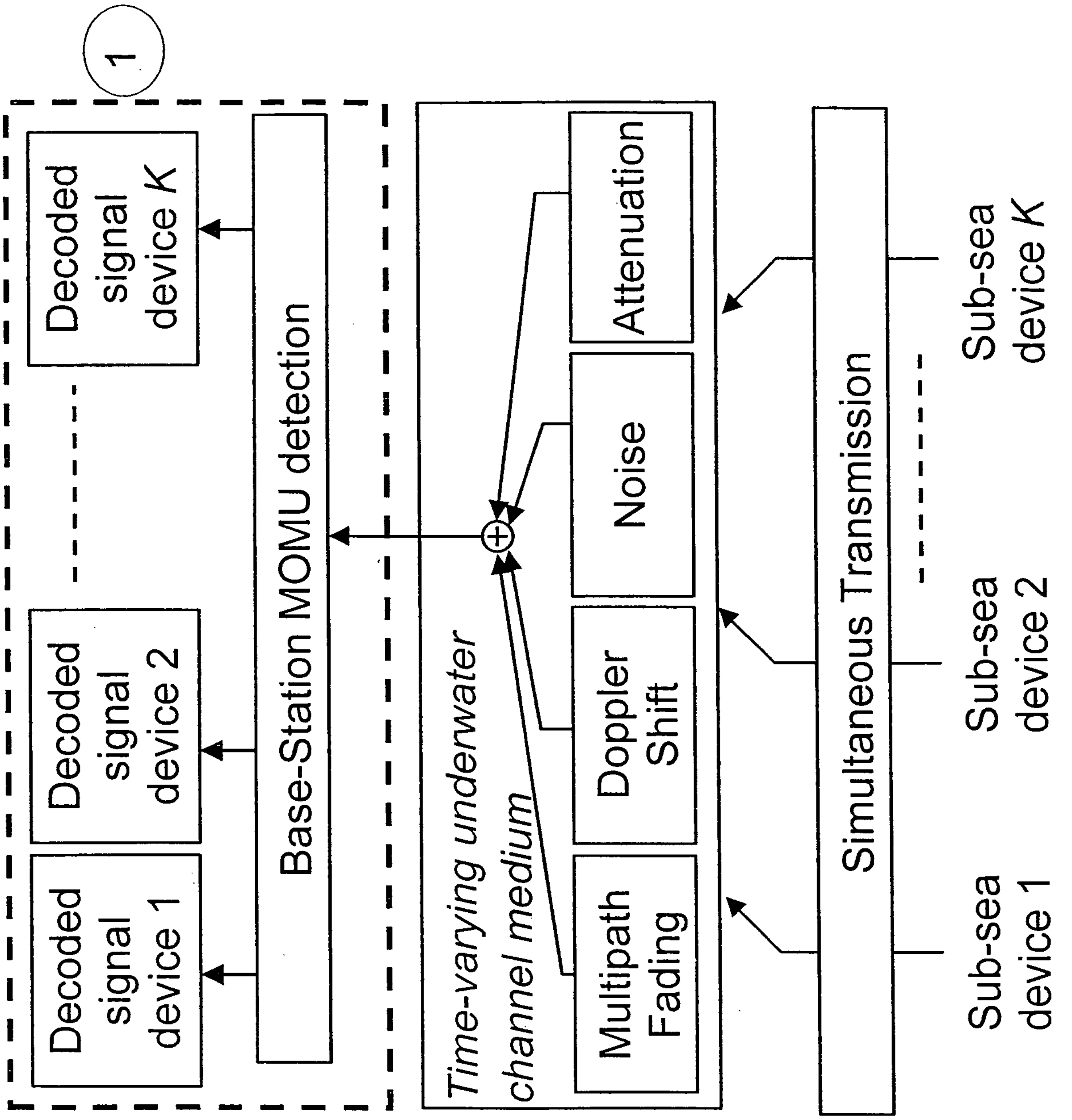

1
Decoded signal device 1
Decoded signal device 2
Decoded signal device K
Base-Station MOMU detection
Time-varying underwater channel medium
Multipath Fading
Doppler Shift
Noise
Attenuation
Simultaneous Transmission
Sub-sea device 1
Sub-sea device 2
Sub-sea device K

FIGURE 5

4 / 19

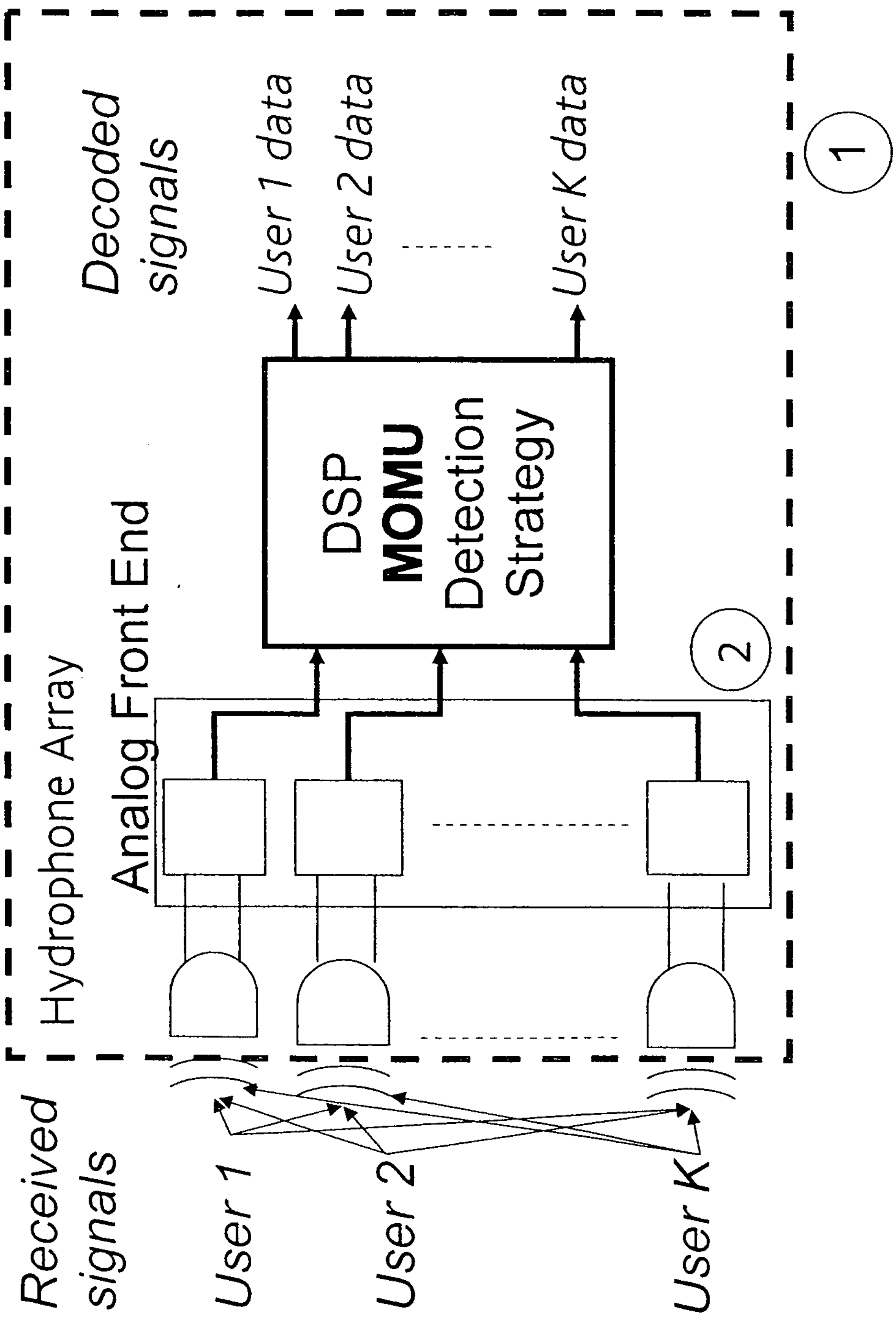

Decoded signals
User 1 data
User 2 data
User K data
1
DSP
MOMU
Detection
Strategy
Hydrophone Array
Analog Front End
2
Received signals
User 1
User 2
User K

FIGURE 6

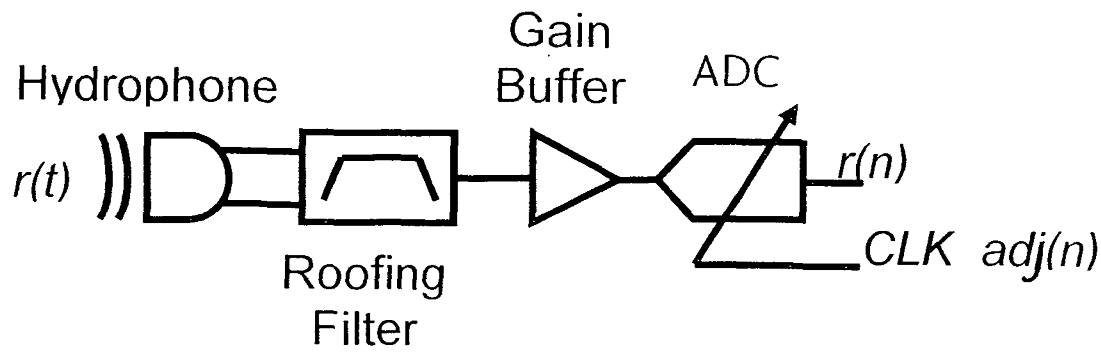


FIGURE 7

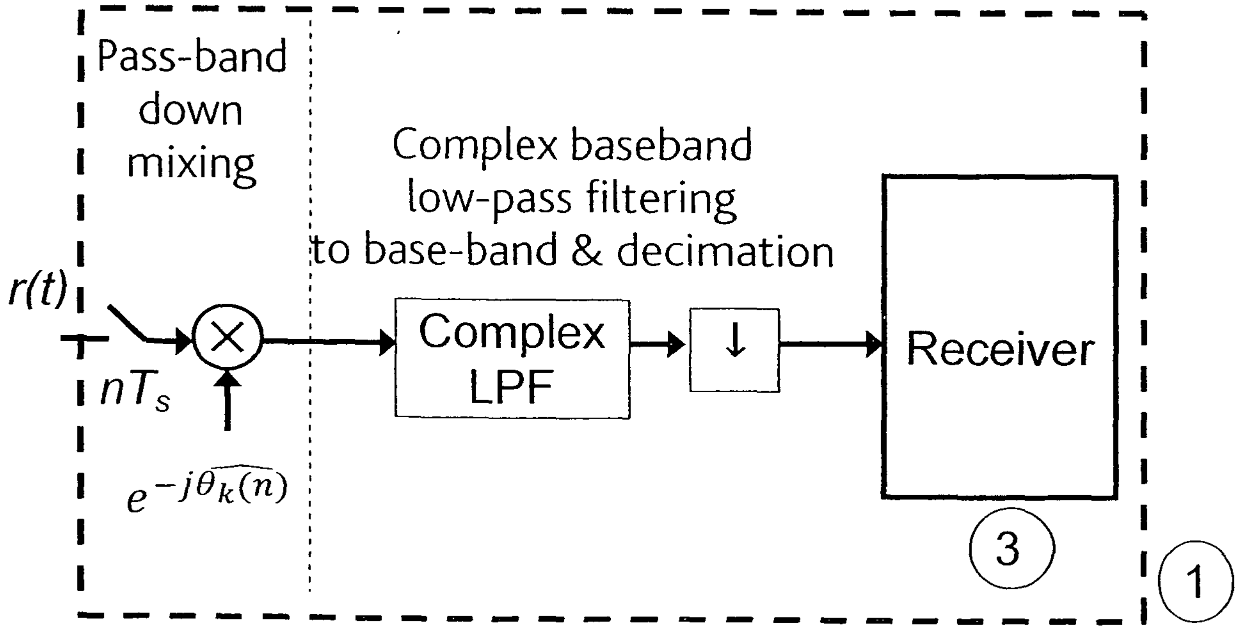


FIGURE 8

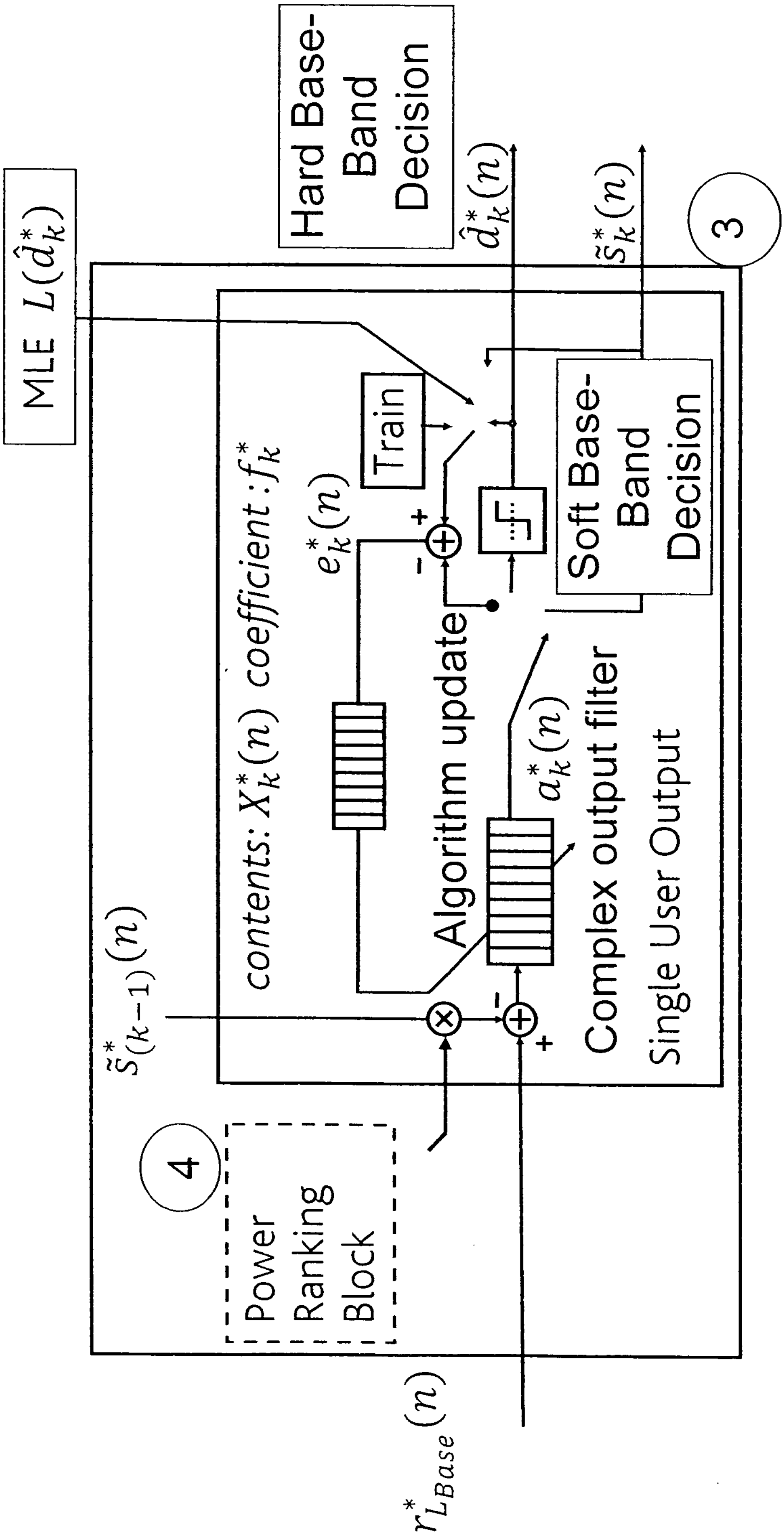

MLE $L(\hat{d}_k^*)$
Hard Base-Band Decision
$\hat{d}_k^*(n)$
$\tilde{s}_k^*(n)$
3
Train
contents: $X_k^*(n)$ coefficient : f_k^*
$e_k^*(n)$
+
-
Soft Base-Band Decision
Algorithm update
$a_k^*(n)$
Complex output filter
Single User Output
$\tilde{s}_{(k-1)}^*(n)$
-
+
4
Power Ranking Block
$r_{L_{Base}}^*(n)$

FIGURE 9

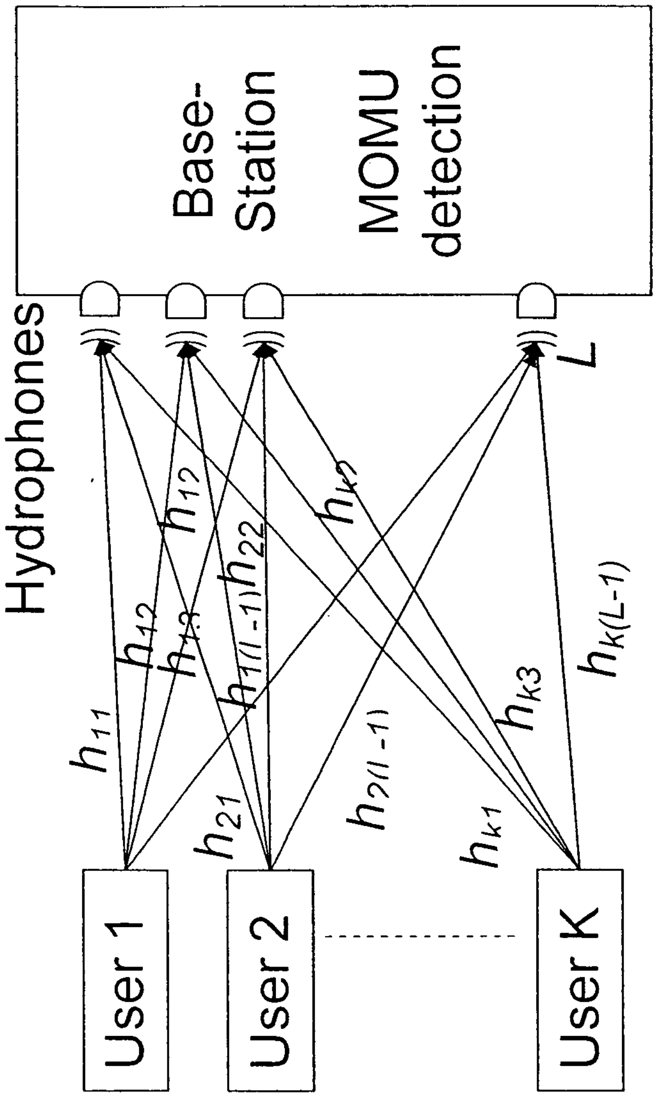

Base-
Station
MOMU
detection
Hydrophones
L
User 1
User 2
User K
h_{11}
h_{21}
h_{22}
h_{k1}
h_{k3}
$h_{K(L-1)}$

FIGURE 10

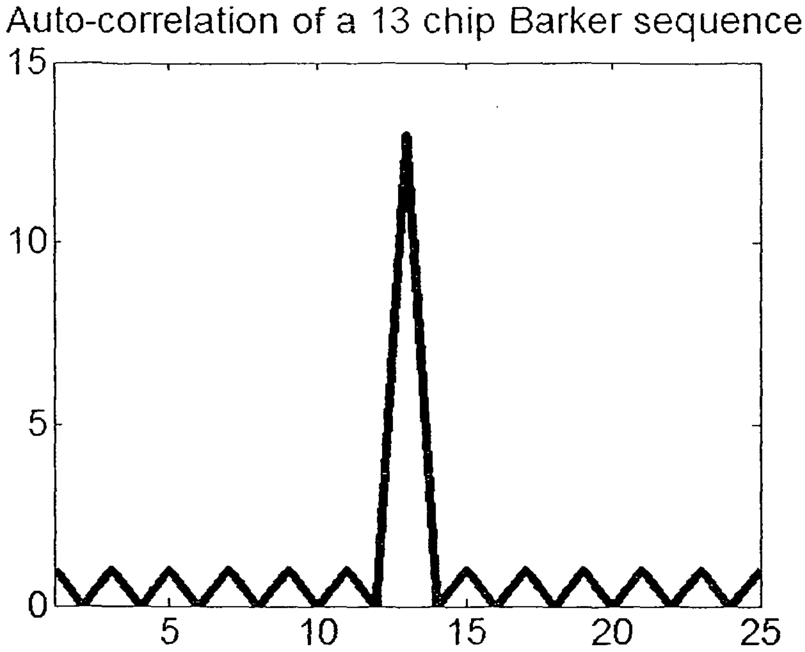


FIGURE 11

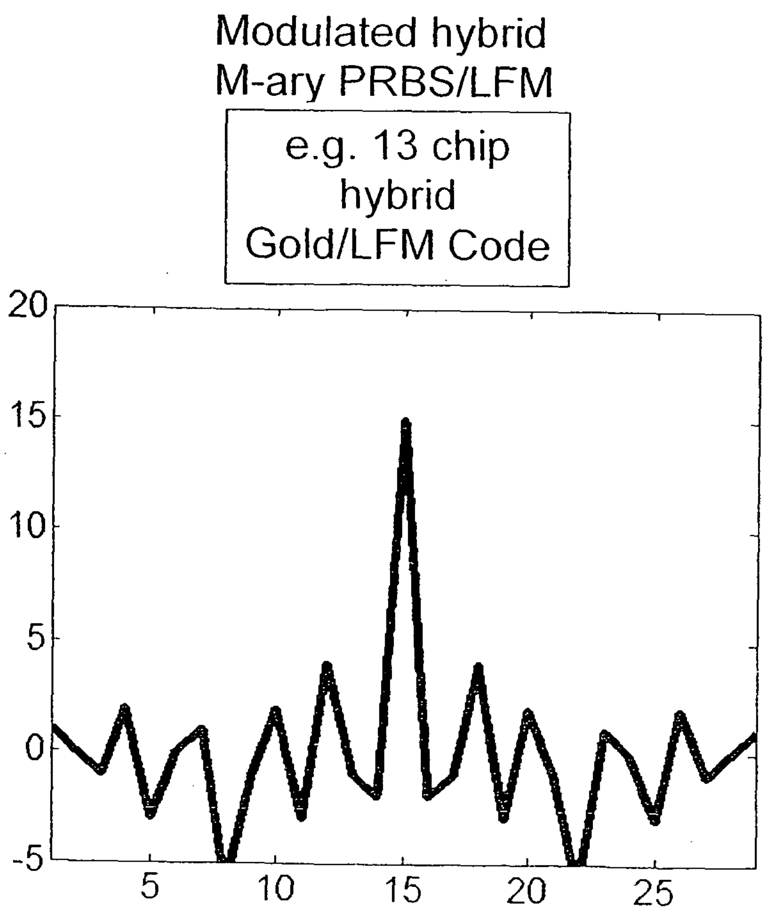


FIGURE 12

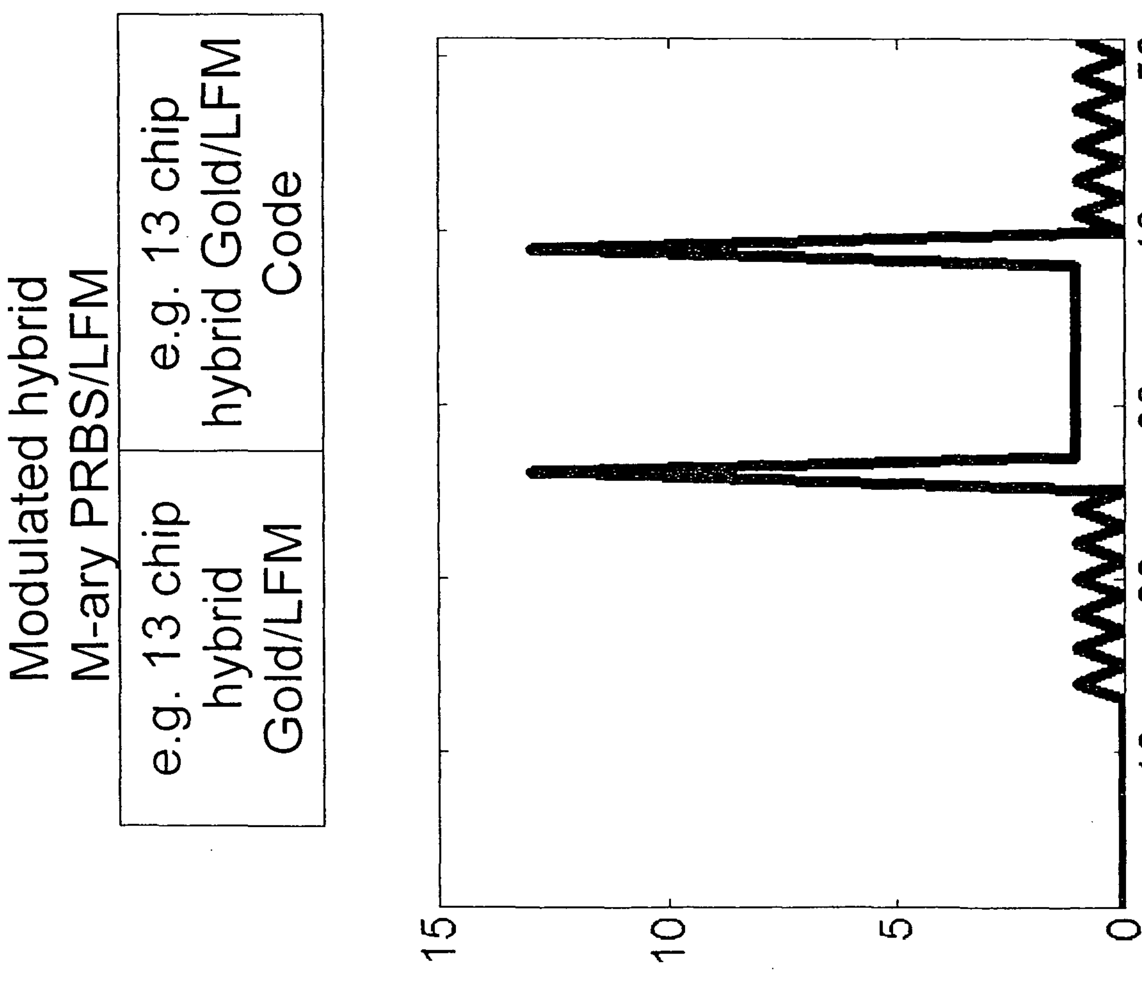


FIGURE 13

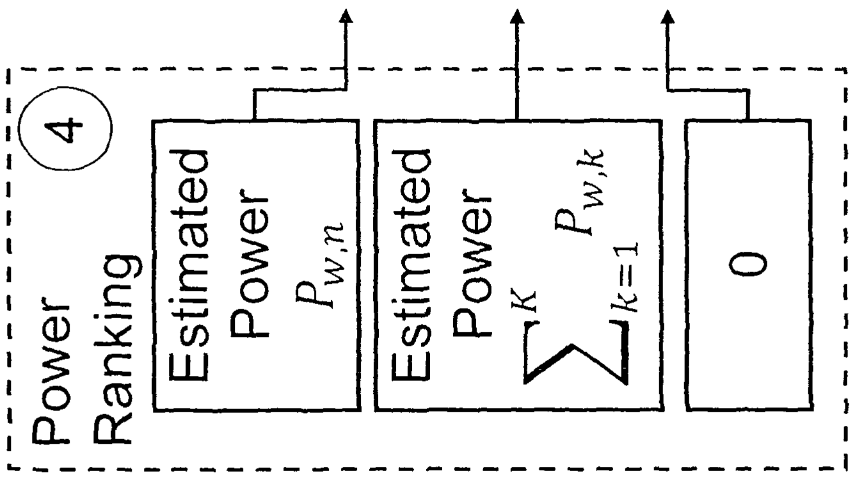


FIGURE 14

Past Symbol	Current Symbol	Future Symbol
Partial Period $-\Delta T_S$	Symbol Period ΔT_S	Partial Period $+\Delta T_S$

FIGURE 15

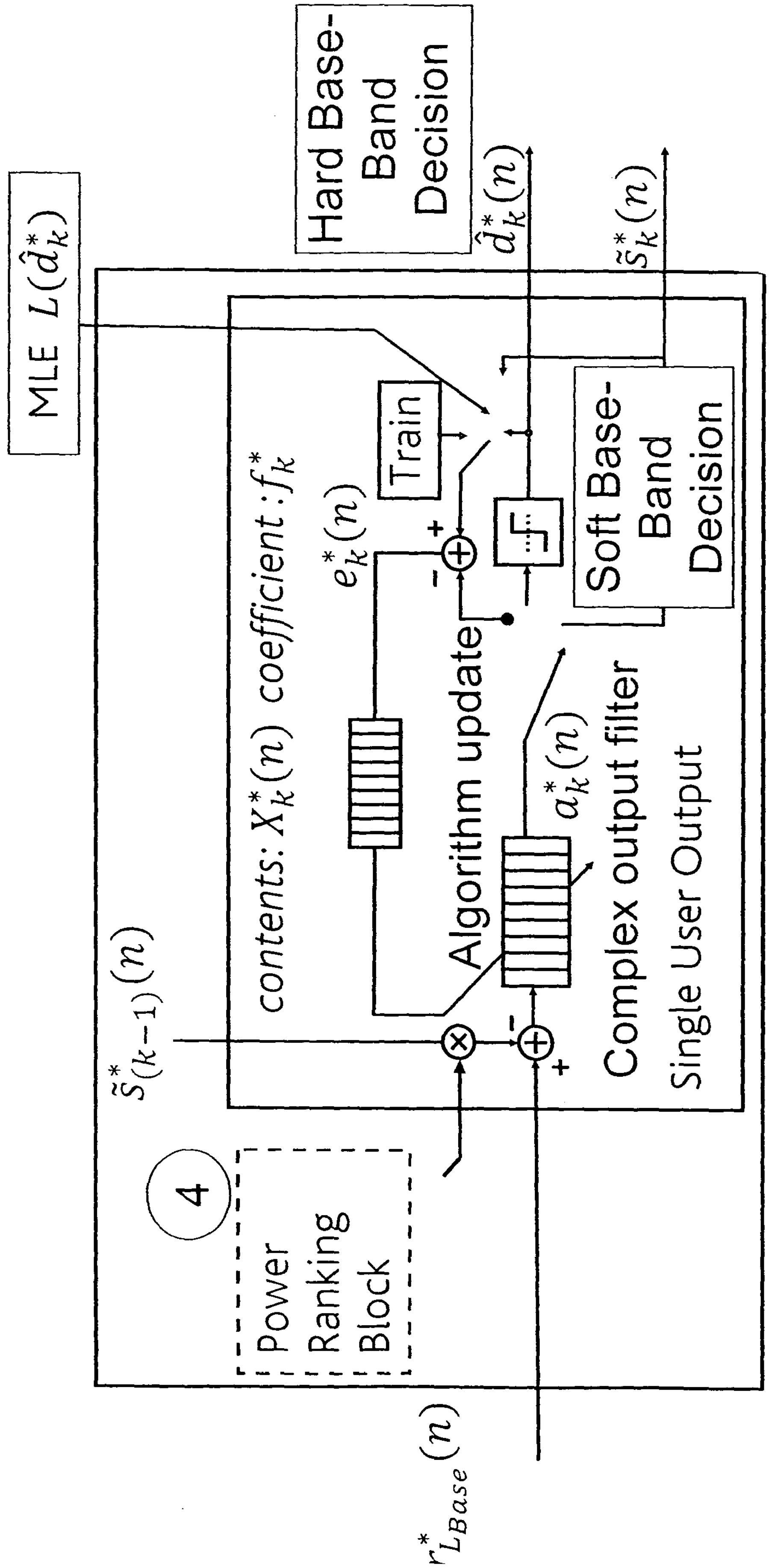

Hard Base-Band Decision
MLE $L(\hat{d}_k^*)$
$\hat{d}_k^*(n)$
$\tilde{s}_k^*(n)$
Train
Soft Base-Band Decision
contents: $X_k^*(n)$ coefficient : f_k^*
$e_k^*(n)$
Algorithm update
$a_k^*(n)$
Complex output filter
Single User Output
$\tilde{s}_{(k-1)}^*(n)$
4
Power Ranking Block
$r_{L_{Base}}^*(n)$

FIGURE 16

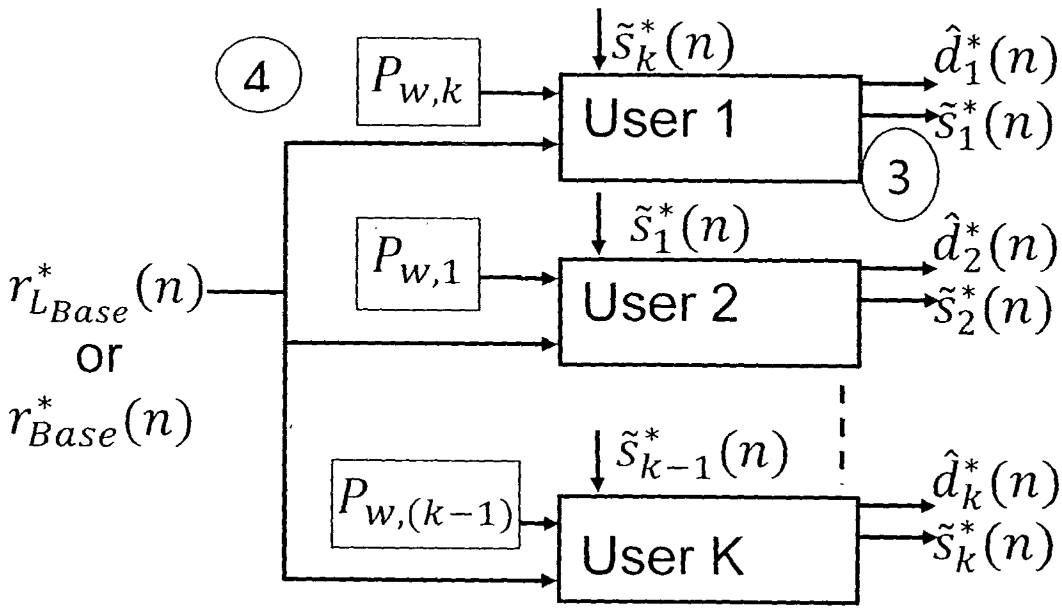

4
$P_{w,k}$
$\tilde{s}_k^*(n)$
User 1
$\hat{d}_1^*(n)$
$\tilde{s}_1^*(n)$
3
$P_{w,1}$
$\tilde{s}_1^*(n)$
User 2
$\hat{d}_2^*(n)$
$\tilde{s}_2^*(n)$
$r_{L_{Base}}^*(n)$
or
$r_{Base}^*(n)$
$P_{w,(k-1)}$
$\tilde{s}_{k-1}^*(n)$
User K
$\hat{d}_k^*(n)$
$\tilde{s}_k^*(n)$

FIGURE 17

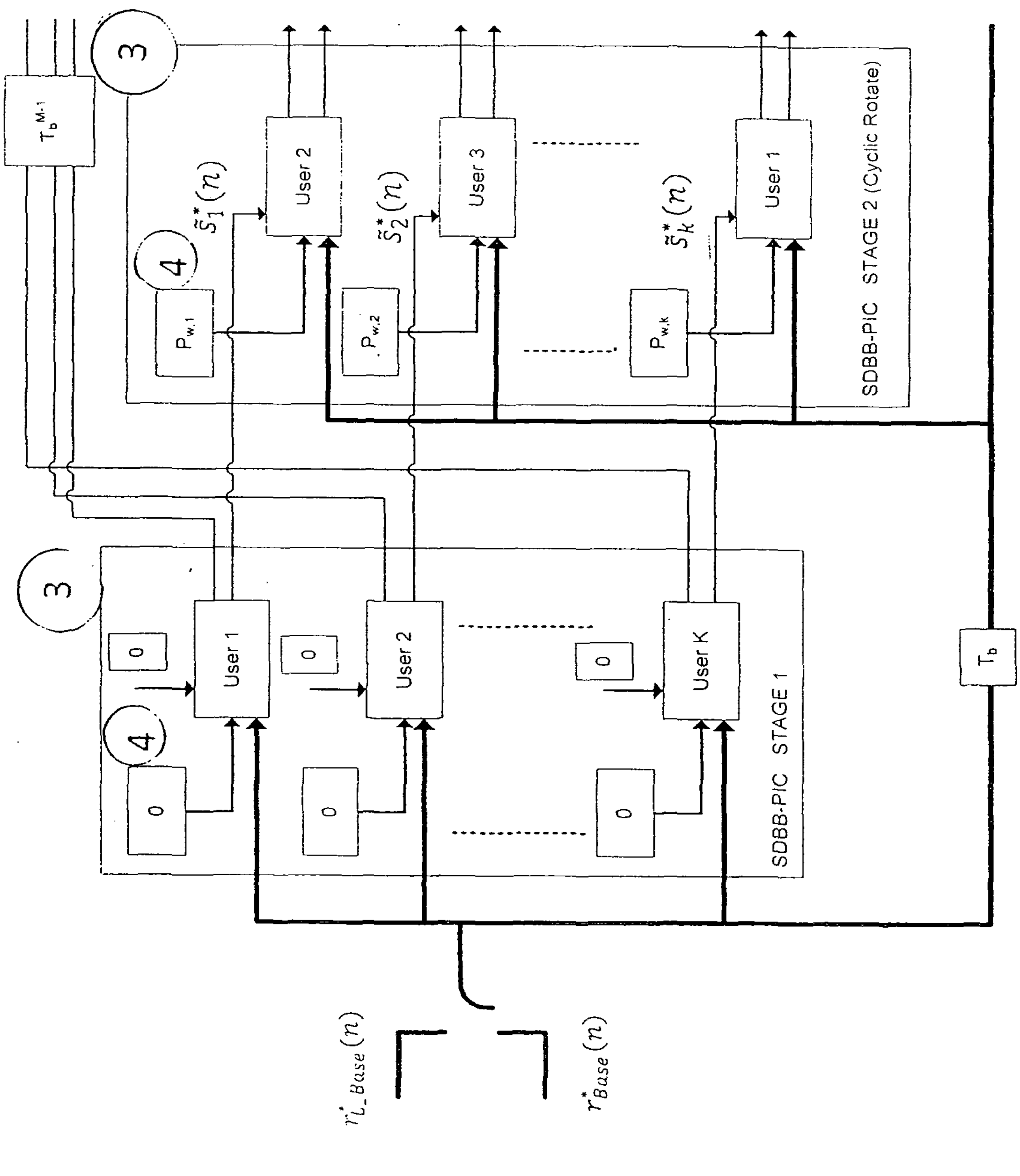


FIGURE 18 (PART I)

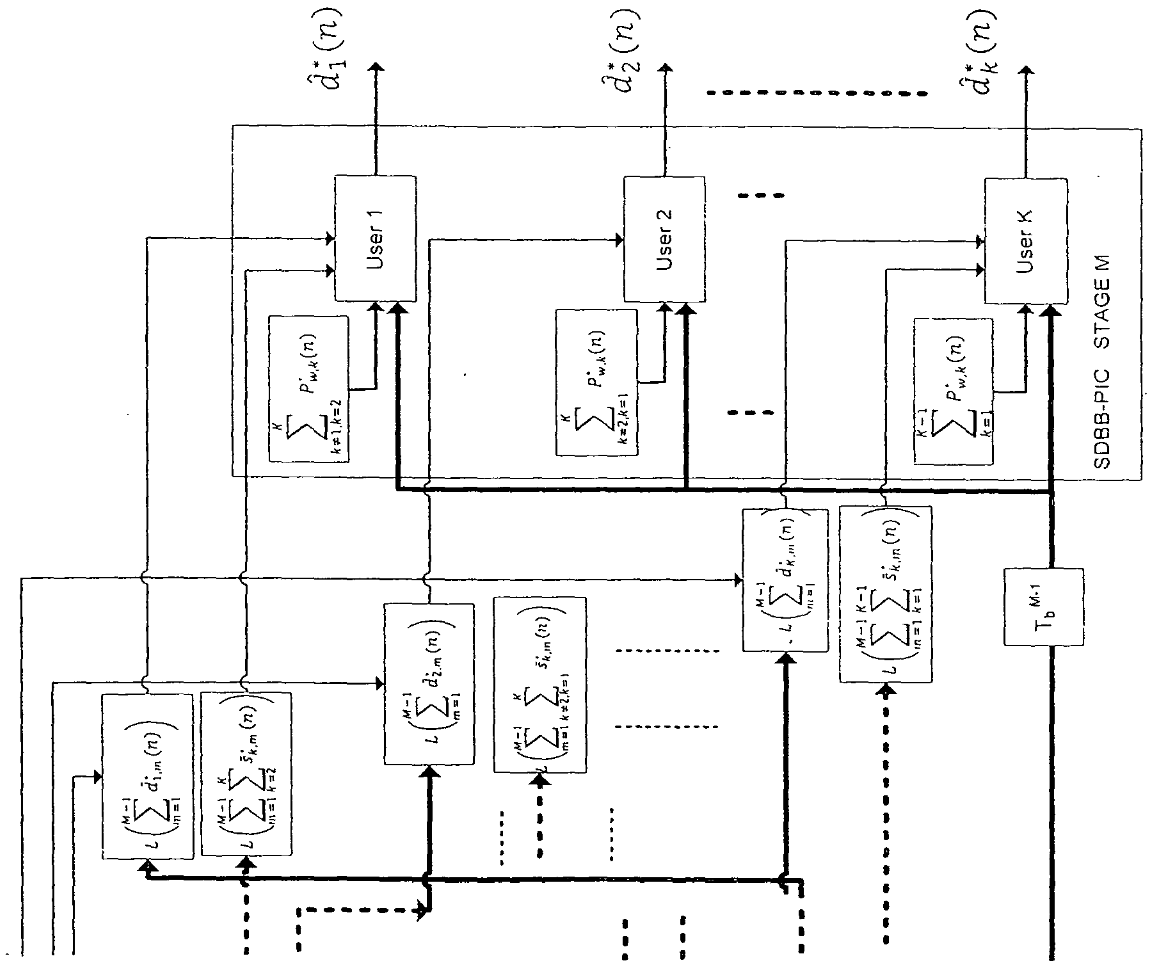


FIGURE 18 (PART II)

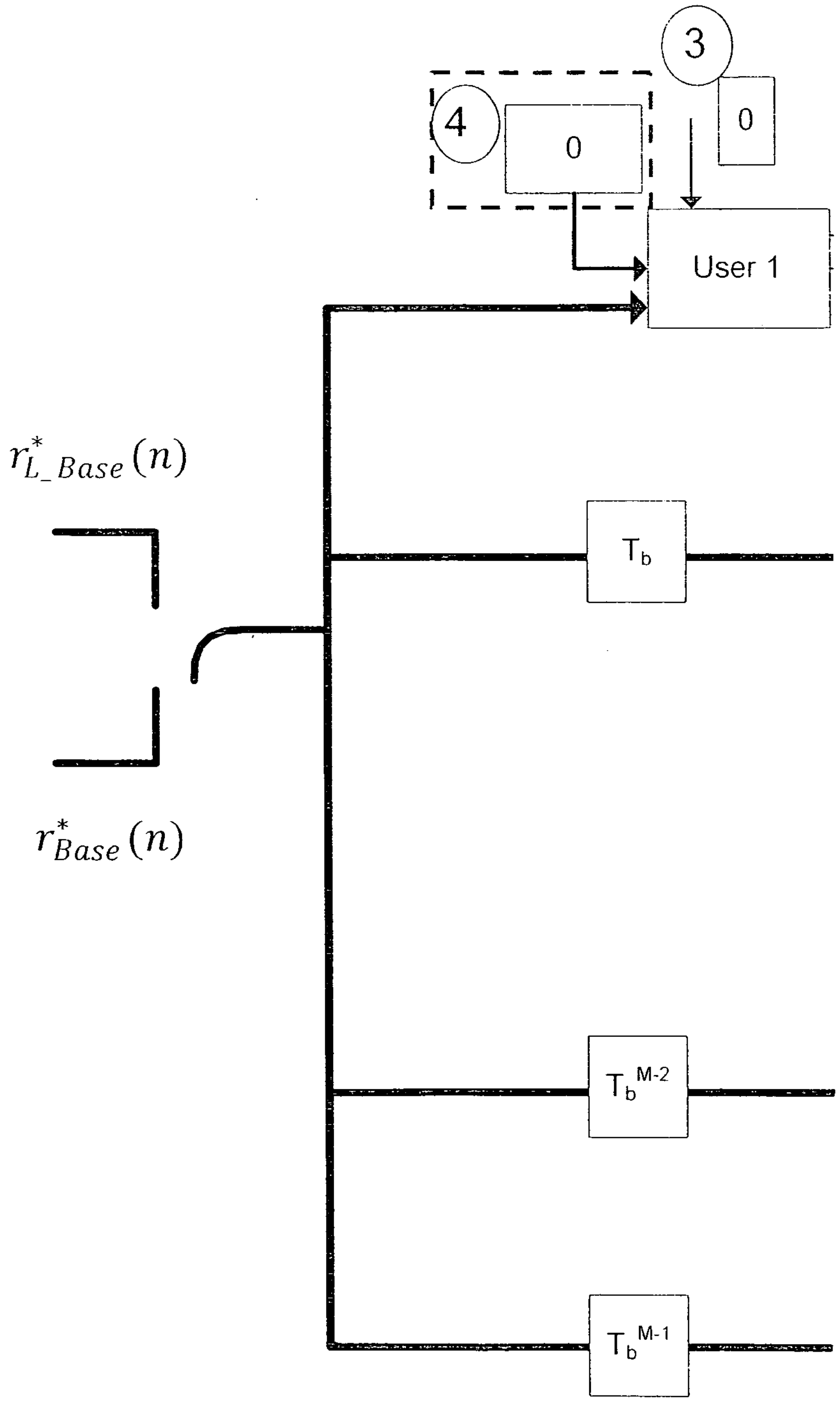


FIGURE 19 (PART I)

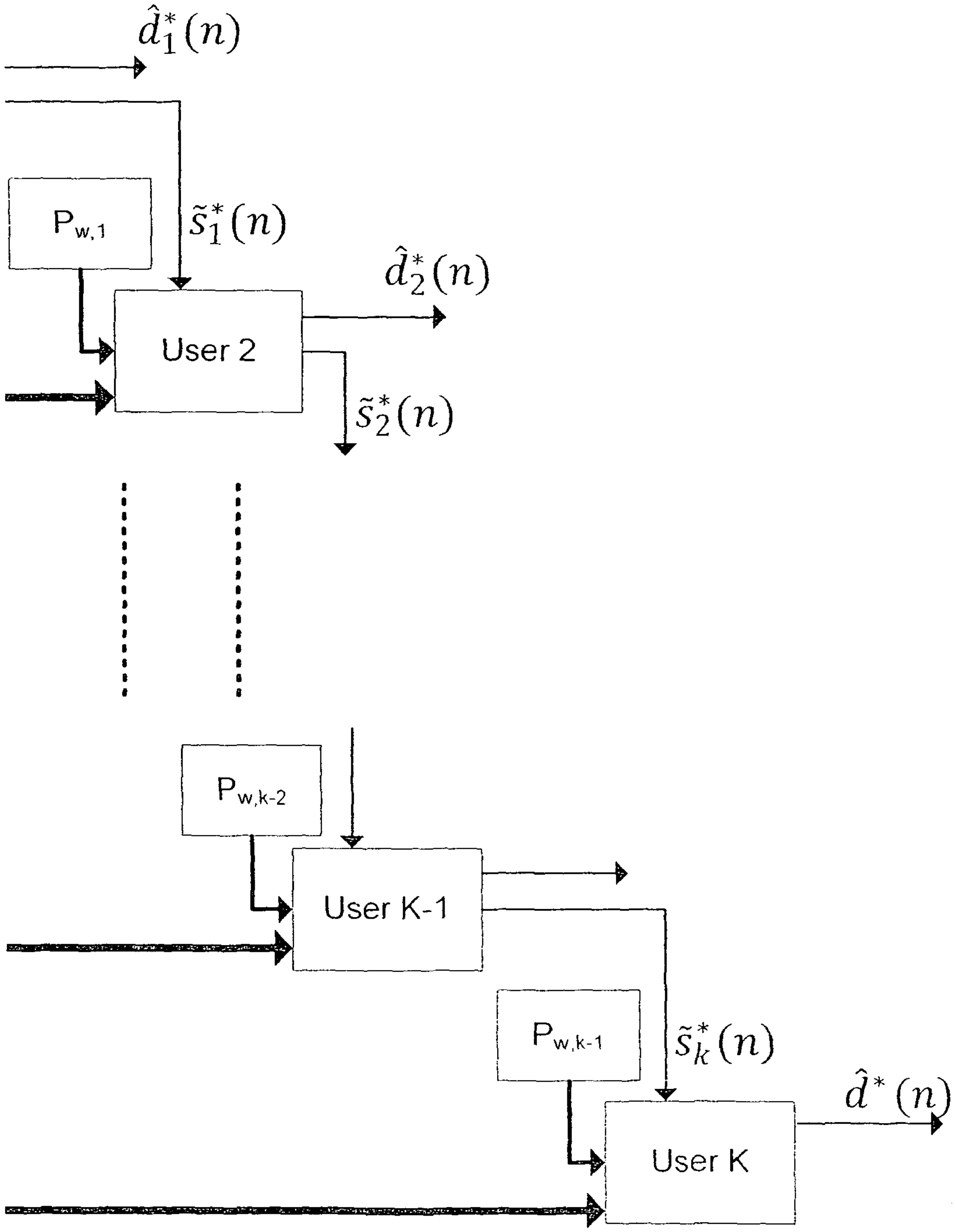
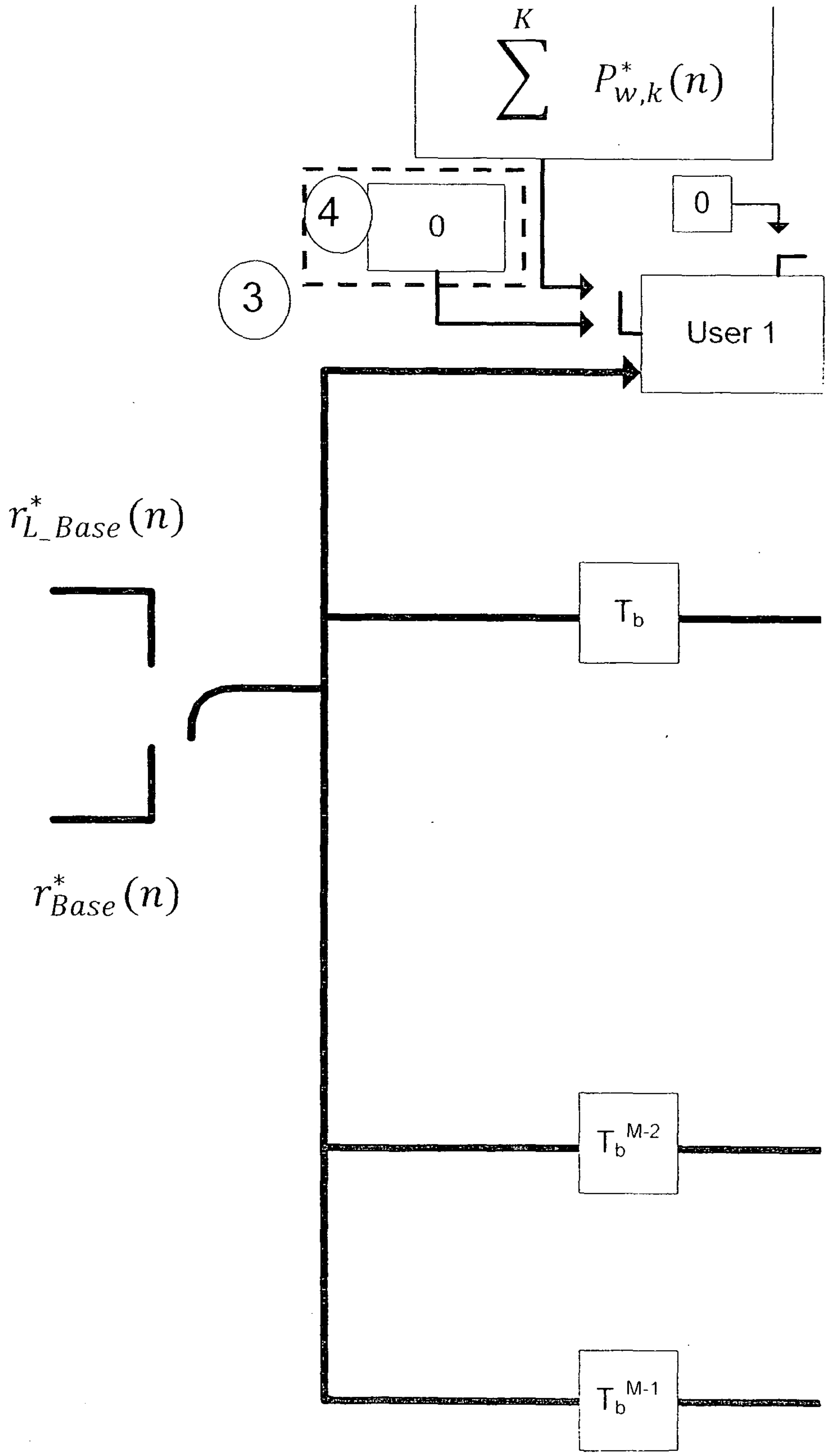


FIGURE 19 (PART II)


$\sum^{K} P_{w,k}^{*}(n)$
4
0
3
0
User 1
$r_{L\_Base}^{*}(n)$
T_b
$r_{Base}^{*}(n)$
T_b^{M-2}
T_b^{M-1}

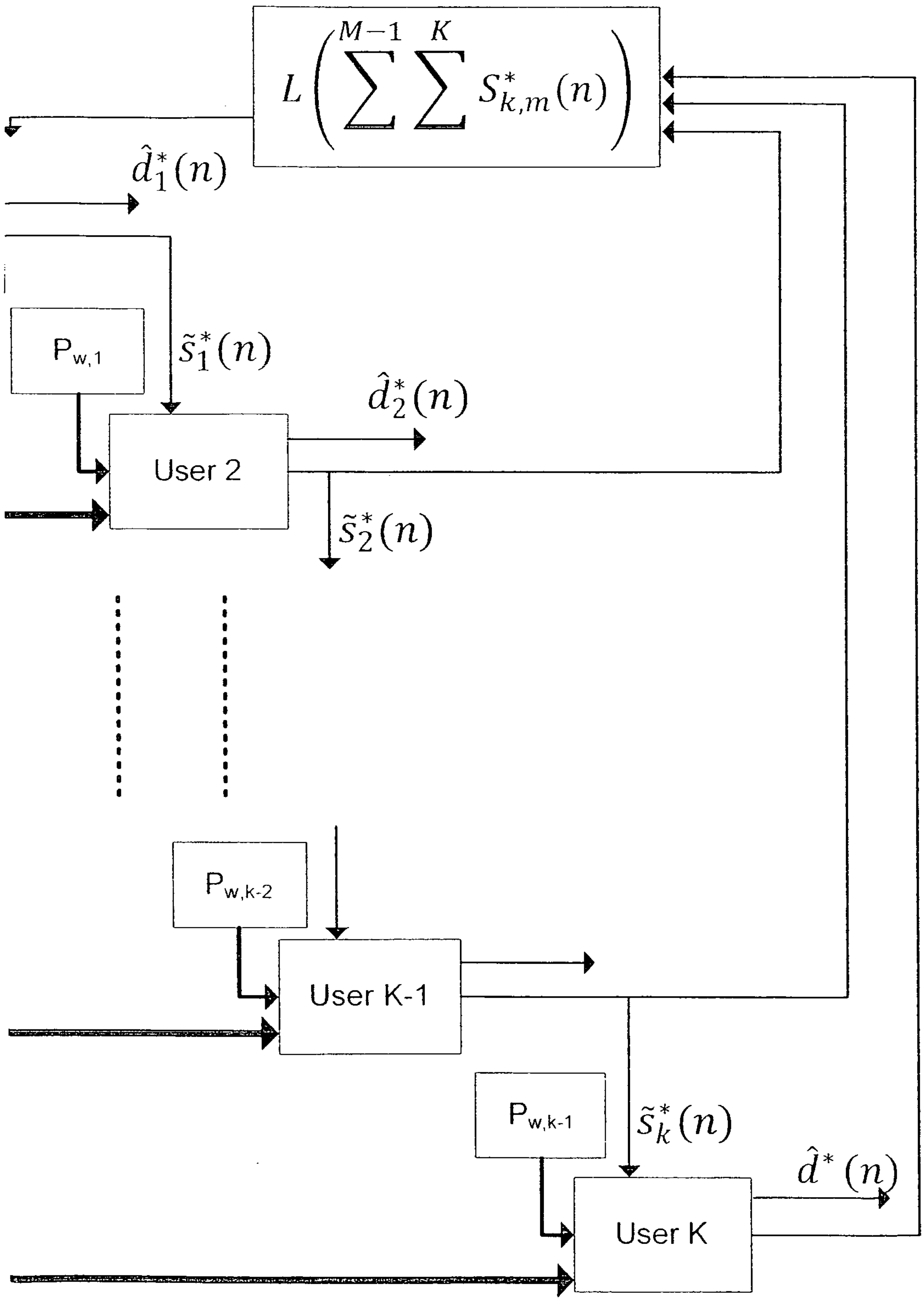


FIGURE 20 (PART II)

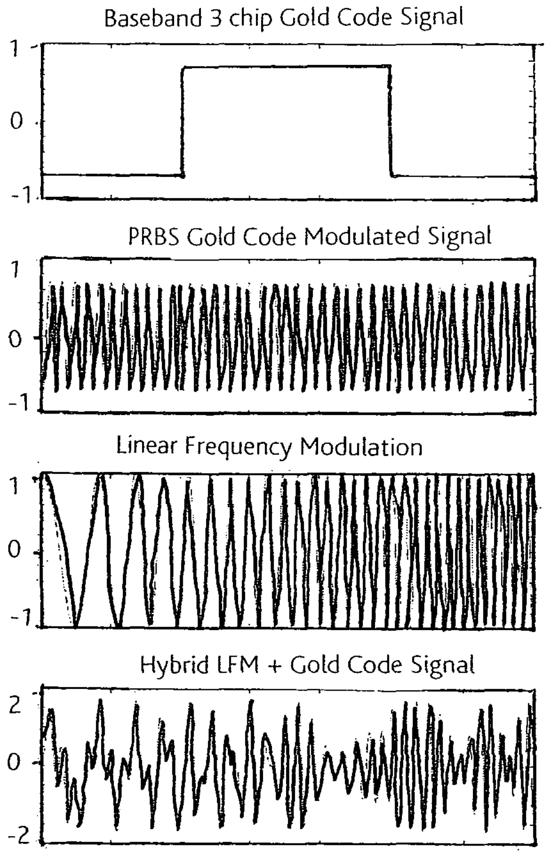


FIGURE 21

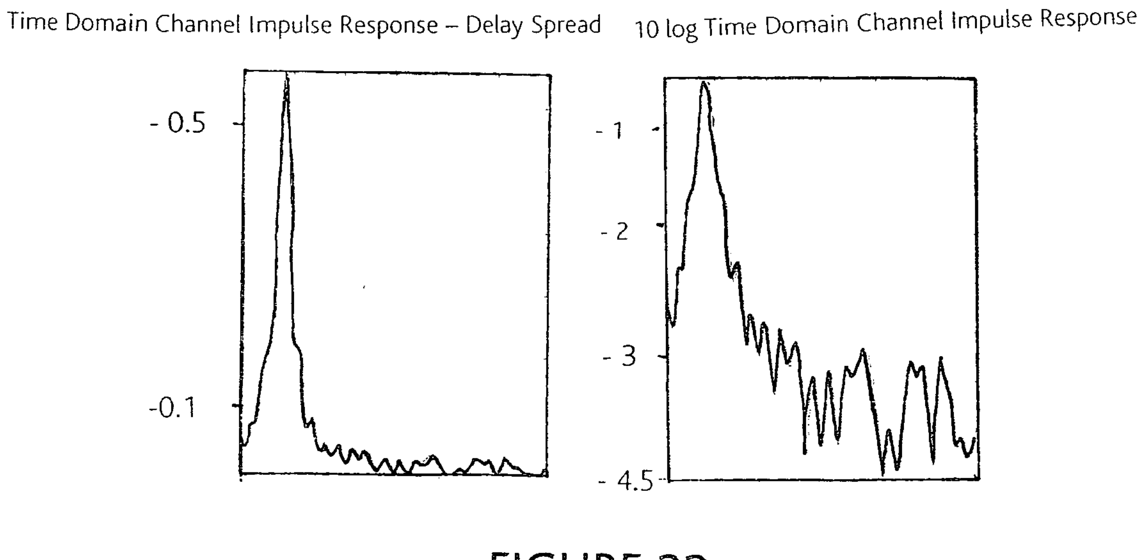


FIGURE 22