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(71) Demandeur/Applicant:
RIBIC, ZLATAN, AT

(72) Inventeur/Inventor:
RIBIC, ZLATAN, AT

(74) Agent: MARKS & CLERK

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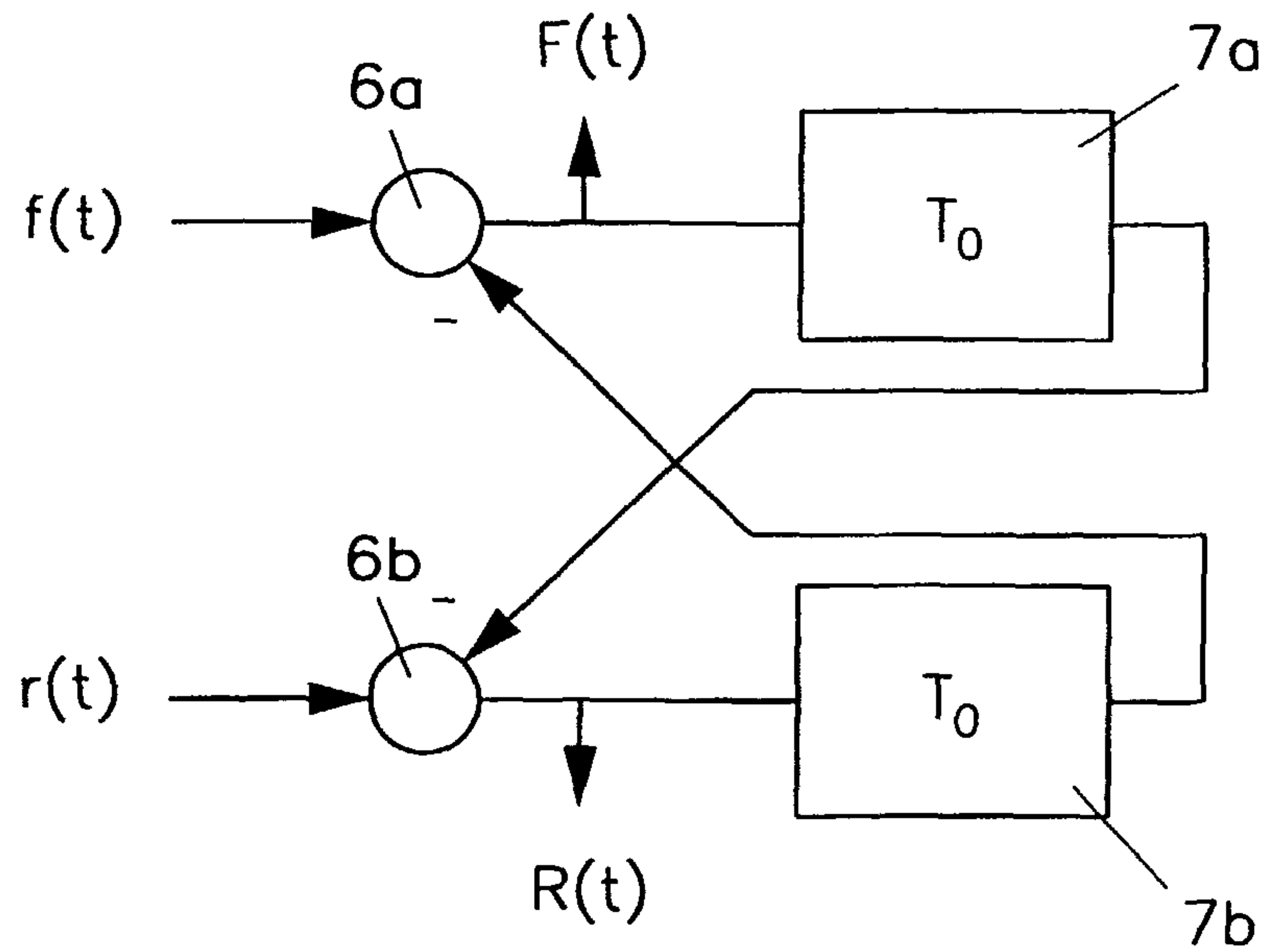

(57) Abrégé/Abstract:
The invention relates to a method for picking up sound consisting of the following steps: providing at least two essentially omnidirectional microphones (1a, 1b, 1c) or membranes (9a, 9b) which have a mutual distance (d) shorter than a typical wave length of the sound wave; combining these microphones (1a, 1b, 1c) or membranes (9a, 9b) to obtain directional signals (F(t), R(t)) depending on the direction (3) of sound; processing the directional signals (F(t), R(t)) to modify the directional pattern of the signals.

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(71) Applicant and
(72) Inventor: **RIBIC, Zlatan** [AT/AT]; Anton Baumgartner Strasse 44/A8/052, A-1232 Vienna (AT).

(74) Agent: **BABELUK, Michael**; Mariahilfer Gürtel 39/17, A-1150 Wien (AT).

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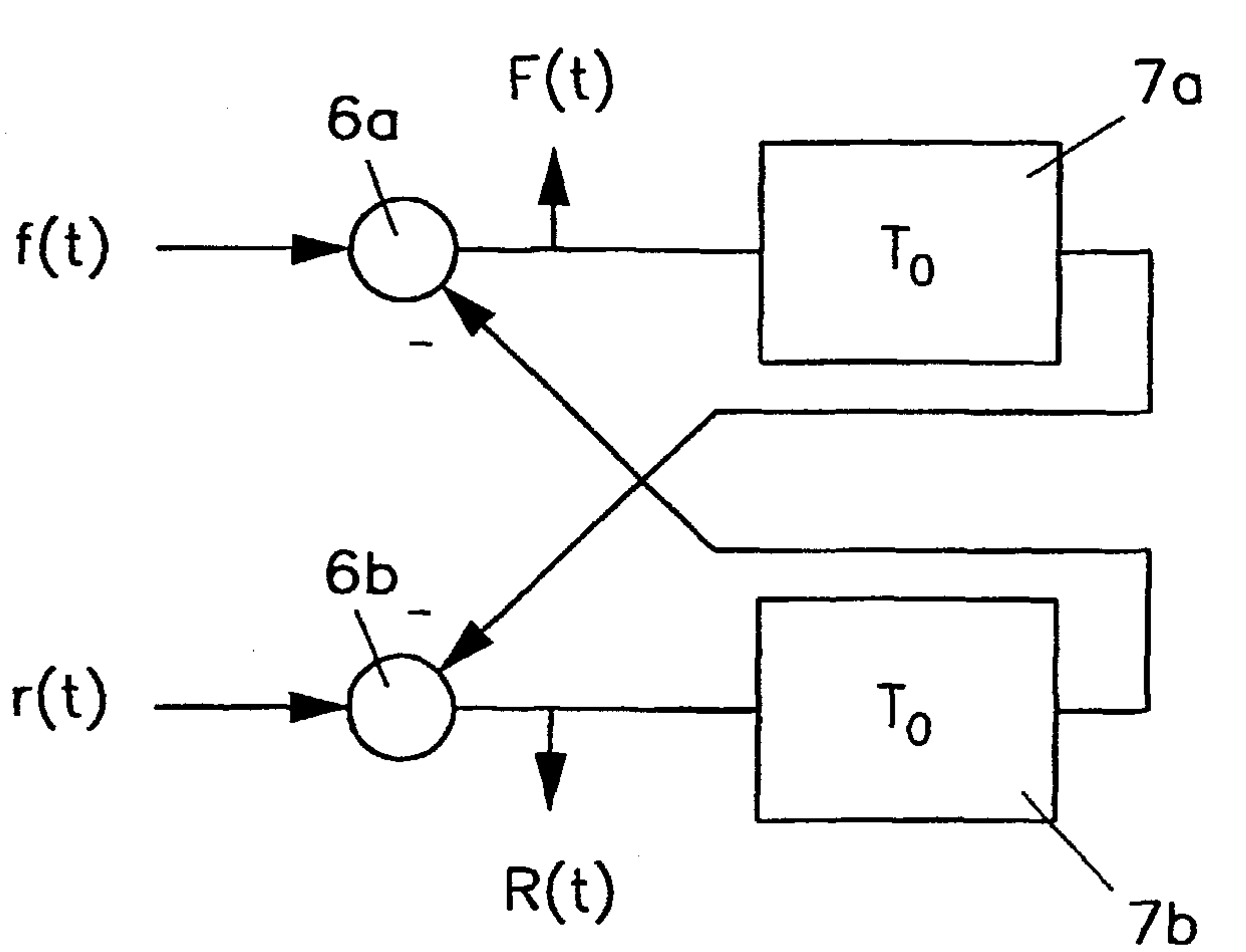
(54) Title: METHOD AND APPARATUS FOR PICKING UP SOUND



(57) Abstract: The invention relates to a method for picking up sound consisting of the following steps: providing at least two essentially omnidirectional microphones (1a, 1b, 1c) or membranes (9a, 9b) which have a mutual distance (d) shorter than a typical wave length of the sound wave; combining these microphones (1a, 1b, 1c) or membranes (9a, 9b) to obtain directional signals (F(t), R(t)) depending on the direction (3) of sound; processing the directional signals (F(t), R(t)) to modify the directional pattern of the signals.

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Method and apparatus for picking up sound

The invention relates to a method and an apparatus for picking up sound.

In a hearing aid, sound is picked up, amplified and at the end transformed to sound again. In most cases omnidirectional microphones are used for picking up sound. However, in case of omnidirectional microphones the problem occurs that ambient noise is picked up in the same way. It is known to enhance the quality of signal transmission by processing a signal picked up by the hearing aid. For example, it is known to split the signal into a certain number of frequency bands and to amplify preferably those frequency ranges in which the useful information (for example speech) is contained and to suppress those frequency ranges in which usually ambient noise is contained. Such signal processing is very effective if the frequency of ambient noise is different from the typical frequencies of speech. There is little help in the so-called "party situation", in which the useful signal is speech of one person and noise consists of speech of a lot of other persons. To overcome this problem it has been proposed to use directional microphones with a cardioid or hyper-cardioid characteristic. In such cases sound of sources in front of the person wearing the hearing aid is amplified and sound from other directions is suppressed. Directional microphones are often used in these situations, but they have several serious disadvantages. For instance, the directional microphones are bulky, usually have higher equivalent input noise, and are extremely sensitive to wind. The situation becomes even more problematic when stereo or surround record is required. Then, it is necessary to use more microphones. US-A 5,214,709 teaches that usually pressure gradient microphones are used to pick up the sound at two points with a certain distance to obtain a directional recording pattern. The largest disadvantage of the simple small directional microphones is that they measure air velocity, not sound pressure, therefore their frequency response for the sound pressure has a +6dB/octave slope. This means that their pressure sensitivity in the range of low frequencies is much lower than at high frequencies. If inverse filtering is applied the own microphone noise is also amplified on the low frequencies and the signal to noise ratio remains as bad as it was before the filtering. The second problem is that if the directional microphone is realized with two omnidirectional pressure microphones, their matching is critical and their frequency characteristic depends very much on the incoming sound direction. Therefore, the inverse filtering is not recommended and can have a negative effect. Because of the mentioned reasons omnidirectional pressure microphones with linear frequency response and a good signal to microphone noise ratio on whole frequency range are mostly used for peaceful and silent environments. When the noise level is high, the directionality is introduced, and since the signal level is high, the signal to microphone noise ratio is not important.

Furthermore, US-A 5,214,907 describes a hearing aid which can be continuously regulated between an omnidirectional characteristic and a unidirectional characteristic. The special advantage of this solution is that at least in the omnidirectional mode a linear frequency response can be obtained.

It is further known from M. Hackl, H. A. Müller: Taschenbuch der technischen Akustik, Springer 1959 to use double membrane systems for obtaining a directional recording pattern. Such systems are used in studios and professional applications. However, due to losses caused by membrane mass and friction the real capabilities are partially limited. It is not known to use such systems for hearing aids.

Some documents, e.g. EP 690 659 A, EP 869 697 A or US 3,109,066 A disclose microphone systems in which the signals of microphones are delayed and these delayed signals are mixed to original signals of the microphones. In that way a cardioide pattern can be obtained for example. However, such feed forward solutions show a frequency response for the sound pressure having a +6dB/octave slope. Generally this disadvantage con be partially overcome by selective amplification of the signals, but then noise is amplified too and the signal/noise ratio is deteriorated.

It is an object of the present invention to avoid the above disadvantages and to develop a method and a system which allows picking up sound with a directional sensitivity which is essentially independent of the frequency. Furthermore, it should be possible to control directionality continuously between a unidirectional and an omnidirectional characteristic and/or to change the direction or the type of the response.

The method of the invention is characterized by the steps of claim 1. Experiments have shown that with such a method a directional signal can be obtained which has a high quality and which in its behaviour is essentially independent of the frequency of the input signals. Depending on different parameters to be chosen a cardioid, hyper-cardioid or other directional characteristic can be obtained.

It has to be noted that a typical distance between the first and second microphone is in the range of 1 cm or less. This is small compared to the typical wavelength of sound which is in the range of several centimeters up to 15 meters.

The principal difference of the invention to prior art is that it is a sort of feedback solution, that is not the signal of a microphone is delayed but a composite signal which contains already delayed information. It has been found that in that way no dependency on frequency is observed.

In a preferred embodiment of the invention two subtractors are provided, each of which is connected with a microphone to feed a positive input to the subtractor, and wherein the output of each subtractor is delayed for a predetermined time and sent as negative input to the other subtractor. The output of the first subtractor represents a first directional signal and the output of the second subtractor represents a second directional signal. The maximum gain of the first signal is obtained when the source of sound is situated on the prolongation of the connecting line between the two microphones. The maximum gain of the other signal is obtained when the source of sound is on the same line in the other direction.

The above method relates primarily to the discrimination of the direction of sound. Based upon this method it is possible to analyse the signals obtained to further enhance the quality for a person wearing a hearing aid for example. One possible signal processing is to mix the first signal and the second signal. If for example both signals have the form of a cardioid with the maximum in opposite direction, a signal with a hyper-cardioid pattern can be obtained by mixing these two signals in a predetermined relation. It can be shown that a hyper-cardioid pattern has advantages compared to a cardioid pattern in the field of hearing aids especially in noisy situations. Furthermore, it is possible to split the first signal and the second signal into sets of signals in different frequency ranges. Depending on an analysis of the sound in each frequency range different strategies can be chosen to select a proper directional pattern and a suitable amplification or suppression. For example, it is possible to have a strong directional pattern in the frequency bandes in which the useful information of speech is contained whereas in other frequency bandes a more or less omnidirectional pattern prevails. This is an advantage since warning signals or the like should be noticed from all directions.

The present invention relates further to an apparatus for picking up sound with at least two essentially omnidirectional microphones, each of which is connected with an input port of a subtractor, a delaying unit with an input port connected with an output port of a first subtractor for delaying the output signal for a predetermined time. According to the invention an output port of the delaying unit is connected with a negative input port of a second subtractor.

According to a preferred embodiment of the invention three microphones are provided wherein the signals of the second and the third microphone are mixed in an adder, with an output port of which being connected to the second subtractor. This allows shifting the direction of maximum gain within a given angle.

In an alternative embodiment of the invention three microphones and three discrimination units are provided wherein the first microphone is connected to an input port of the second and the third discrimination unit, the second microphone is connected to an input port of the first and the third discrimination unit, and the third microphone is connected to an input port of the first and the second discrimination unit. In this way three sets of output signals are obtained so that there are six signals whose direction of maximum gain is different from each other. By mixing these output signals these directions may be shifted to any predetermined direction.

Preferably, more than three microphones are provided which are arranged at the corners of a polygone or polyeder and wherein a set of several discrimination units is provided, each of which is connected to a pair of microphone. In case of an arrangement in the form of a polygone all directions within the plane in which the polygone is situated can be discriminated. If the microphones are arranged at the corners of a polyeder the directions in threedimensional space may be discriminated. At least four microphones have to be arranged on the corners of a tetraeder.

A very strong directional pattern like shotgun microphones with a length of 50 cm or more with a characteristic like a long telephoto lens in photography may be obtained if at least three microphones are provided which are arranged on a straight line and wherein a first and a second microphone is connected with the input ports of a first discrimination unit, and the second and the third microphone is connected to the input ports of a second discrimination unit and wherein a third discrimination unit is provided, the input ports of which are connected to an output port of the first and the second discrimination unit and wherein a fourth discrimination unit is provided, the input ports of which are connected to the other output ports of the first and the second discrimination unit.

The invention is now described further by some examples shown in the drawings. The drawings show:

Fig. 1 a block diagram of an embodiment of the invention,

Fig. 2 a circuit diagram of the essential part of the invention,

Fig. 3 a schematical view of a double membran microphone,

Figs. 4a and 4b circuit diagrams of two variants of a further embodiment of the invention,

Fig. 5 a circuit diagram of yet another embodiment of the invention,

Fig. 6 a detailed circuit diagram of another embodiment,

Fig. 7 a block diagram of a further embodiment of the invention,

Figs. 8, 9 and 10 typical directional patterns obtained by methods according to the invention.

Fig. 1 shows that sound is picked up by two omnidirectional microphones 1a, 1b. The first microphone 1a produces an electrical signal f(t) and the second microphone 1b produces an electrical signal r(t). When the microphones 1a, 1b are identical, signals f(t) und r(t) are identical with the exception of a phase difference resulting from the different time of the sound approaching the microphones 1a, 1b. The signals of the microphones 1a, 1b fulfill the following equation:

$$r(t) = f(t - \frac{d}{c}\cos\varphi) \qquad (1)$$

wherein d represents the distance between the microphones 1a and 1b, c sound velocity and φ the angle between the direction 3 of sound approaching and the connection line 2 between the microphones 1a and 1b.

Block 4 represents a discrimination unit to which signals f(t) and r(t) are sent. The outputs of the discrimination circuit 4 are designated F(t) and R(t). The amplitude of F(t) and R(t) depends on angle φ wherein a cardioid pattern is obtained for example. That means that the amplitude A of signals F and R corresponds to equation 2:

$$A = \frac{A_0}{2}(1 + \cos\varphi) \qquad (2)$$

A_0 represents the maximum amplitude obtained if the source of sound is on the connection line 2 between microphones 1a and 1b, which means that the maximum amplitude of F(t) is at $\varphi = 0$ and of R(t) at $\varphi = \pi$.

Signals F(t) and R(t) are processed further in the processing unit 5, the output of which is designated with FF(t) and RR(t).

In Fig. 2 the discrimination unit 4 is explained further. The first signal f(t) is sent into a first subtractor 6a, the output of which is delayed in a delaying unit 7a for a predetermined time T_0. Signal r(t) is sent to a second subtractor 6b, the output of which is sent to a second delaying unit 7b, which in the same way delays the signal for a time T_0. Furthermore, the output of the first delaying unit 7a is sent as a negative input to the second subtractor 6b, and the output of the second delaying unit 7b is sent as a negative input to the first subtractor 6a. The output signals F(t) and R(t) of the circuit of Fig. 2 are obtained as outputs of the first and the second subtractors 6a, 6b respectively. The following equations 3, 4 represent the circuit of Fig. 2 mathematically:

$$F(t) = f(t) - R(t - T_0) \tag{3}$$

$$R(t) = r(t) - F(t - T_0) \tag{4}$$

A system according Fig. 2 simulates an ideal double membrane microphone as shown in Fig. 3. A cylindrical housing 8 is closed by a first membrane 9a and a second membrane 9b. The distance d between membranes 9a and 9b is chosen according equation (5):

$$d = cT_0 \tag{5}$$

In this case signal F(t) can be obtained from first membrane 9a and signal R(t) can be obtained from membrane 9b. It has to be noted that the similarity between the double membrane microphone and the circuit of Fig. 2 applies only to the ideal case. In reality results differ considerably due to friction, membrane mass and other effects.

The above system operates at the limit of stability. To obtain a stable system a small damping effect is necessary for the feedback signals. Therefore the above equations (3) and (4) are modified to:

$$F(t) = f(t) - (1 - \varepsilon)R(t - T_0) \tag{3a}$$

$$R(t) = r(t) - (1 - \varepsilon)F(t - T_0) \tag{4a}$$

with $\varepsilon << 1$, being a constant ensuring stability.

It is obvious that the circuit of Fig. 2 only corresponds to a double membrane microphone when the delay T_O is equal for the delaying units 7a and 7b. It is an advantage of the circuit of Fig. 2 that it is possible to have different delays T_{0a} and T_{0b} in the delaying units 7a and 7b respectively to obtain different output functions F(t) and R(t).

In the above embodiments the direction in which the maximum gain is obtained is defined by the connecting line between microphones 1a and 1b. The embodiments of Fig. 4a and 4b make it possible to shift the direction in which the maximum gain is obtained without moving microphones. In Fig. 4a as well as in Fig. 4b three microphones 1a, 1b, 1c are arranged at the corners of a triangle. In the embodiment of Fig. 4a signals of microphones 1b and 1c are mixed in an adder 10. The output of the adder 10 is obtained according to the following equation (6):

$$r(t) = (1-\alpha)r_1(t) + \alpha r_2(t) \qquad (6)$$

With $0 \leq \alpha \leq 1$.

The processing of signals F(t) und R(t) occurs according to Fig. 2. For $\alpha = 0$ the maximum gain for F(t) is obtained for sound approaching in direction 3b according the connecting line between microphones 1a and 1b. On the other hand, if $\alpha = 1$ maximum gain for F(t) is obtained for signals approaching in direction 3c according the connection line between microphones 1a and 1c. For other values of α the maximum is obtained for sound approaching along a direction between arrows 3b and 3c.

In the embodiment of Fig. 4b there are three discrimination units 4a, 4b and 4c, each of which is connected to a single pair out of three microphones 1a, 1b, 1c. Since microphones 1a, 1b, 1c are arranged at the corners of an equilateral triangle, the maximum of the output functions of discrimination unit 4c is obtained in directions 1 and 7 indicated by clock 11. Maximum gain of discrimination unit 4a is obtained for directions 9 and 3 and the maximum gain of discrimination unit 4a is obtained for directions 11 and 5. The arrangement of Fig. 4b produces a set of six output signals which are excellent for recording sound with high discrimination of the direction of sound. For example, in a concert hall it is possible to pick up sound with only one small arrangement of three microphones contained in the housing of one conventional microphone with the possibility of recording on six channels giving an excellent surround impression. The directions mentioned above can be changed in a continuous way similar to embodiment shown in Fig. 4a for example by mixing output function F from discrimination unit 4c with output function F from discrimination unit 4a. In this way the maximum gain can be directed to any direction between 1 and 3 on clock 11.

If four microphones (not shown) are arranged at the corners of a tetraeder the directions of the maximum gain can not only be changed within a plane but also in three dimensional space.

The above embodiments have a directional pattern of first order. With an embodiment of Fig. 5 it is possible to obtain a directional pattern of higher order. In this case three microphones 1a, 1b, 1c are arranged on a straight line. A first discrimination unit 4a processes signals of the first and the second microphone 1a, 1b respectively. A second discrimination unit 4b processes signals of the second and the third microphones 1b and 1c respectively. Front signal F_1 of the first discrimination unit 4a and front signal F_2 of the second discrimination unit 4b is sent into a third discrimination unit 4c. Rear signal R_1 of the first discrimination unit 4a and rear signal R_2 of the second discrimination unit 4b are sent to a fourth discrimination unit 4d. All discrimination units 4a, 4b, 4c and 4d of Fig. 5 are essentially identical. From third discrimination unit 4c a signal FF is obtained which represents a front signal of second order. In the same way a signal RR is obtained from the fourth discrimination unit 4d which represents a rear signal of second order. These signals show a more distinctive directional pattern than signals F and R of the circuit of Fig. 2.

With the circuit of Fig. 5 it is possible to obtain a very high directionality of signals which is necessary in cases in which sound of a certain source is to be picked up without disturbance by ambient noise.

In Fig. 6 a detailed circuit of the invention is shown in which the method of the invention is realized as an essentially analogue circuit. Microphones 1a, 1b are small electret pressure microphones as used in hearing aids. After amplification signals are led to the subtractors 6 consisting of inverters and adders. Delaying units 7a, 7b are realised by followers and switches driven by signals Q and Q' obtained from a clock generator 12. Low pass filters and mixing units for the signals F and R are contained in block 13.

Alternatively it is of course possible to process the signals of the microphones by digital processing.

Fig. 7 shows a block diagram in which a set of a certain number of microphones 1a, 1b, 1c, ... 1z are arranged at the corners of a polygone or a threedimensional polyeder for example. After digitization in an A/D-converter 19 a n-dimensional discrimination unit 14 produces a set of signals. If the discrimination unit 14 consists of one discrimination unit of the type of Fig. 2 for each pair of signals, a set of n (n - 1) directional signals for n microphones 1a, 1b, 1c, ... 1z are obtained. In an analysing unit 15 signals are analysed and eventually feedback information 16 is given back to discrimination unit 14 for controlling signal processing. Further signals of discrimination unit 14 are sent to a mixing unit 18 which is also controlled by analysing unit 15. The number of output signals 17 can be chosen according to the necessary channels for recording the signal.

In Fig. 8 the result of numerical simulation is shown for different values of T_0. T_0 is chosen according the equation (7):

$$T_0 = k\frac{d}{c} \qquad (7)$$

with k being a proportionality constant, d the distance between the two microphones, and c sound velocity. In case of k = 1 the double membrane microphone of Fig. 3 is simulated so that a cardioid pattern (line 20) is obtained. For smaller values of k a hypercardioid pattern is obtained as shown with lines 21, 22, 23 and 24 for values of k = 0.8; k = 0.6; k = 0.4; and k = 0.2.

Fig. 9 shows the directional pattern for a signal processing according the following equation (8):

$$FF(t) = (1-\alpha)F(t) + \alpha R(t) \qquad (8)$$

$$RR(t) = (1-\alpha)R(t) + \alpha F(t) \tag{9}$$

For $\alpha = 0$ a cardioid pattern is obtained shown with line 31. For bigger values of α line 32, 33, 34, 35, 36 and 37 respectively are obtained. Line 37 represents an ideal omnidirectional pattern for $\alpha = \frac{1}{2}$. In Fig. 9 k was set to 1.

Fig. 10 shows the result with the same signal processing as in Fig. 9 according equations (8), (9) but with a value of k = 0.5. Beginning with a hypercardioid 41 lines 42, 43, 44, 45 and 46 are obtained for increasing values of α, wherein for $\alpha = \frac{1}{2}$ an omnidirectional pattern according to line 46 is obtained.

The present invention allows picking up sound with a directional sensitivity without frequency response or directional pattern being dependent on frequency of sound. Furthermore, it is easy to vary the directional pattern from cardioid to hyper-cardioid, bi-directional and even to omnidirectional pattern without moving parts mechanically.

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CLAIMS

1. A method for picking up sound consisting of the following steps:
   - providing at least two essentially omnidirectional microphones (1a, 1b, 1c) which have a distance (d) shorter than a typical wave length of the sound wave;
   - obtaining a first electrical signal (f(t)) from a first microphone (1a) representing the output of this microphone (1a);
   - obtaining a second electrical signal (r(t)) from at least one other microphone (1b, 1c) representing the output of this microphone (1b, 1c);
   - supplying the first electrical signal (f(t)) to a first subtractor (6a) as a first input;
   - supplying the second electrical signal (r(t)) to a second subtractor (6b) as a first input;
   - obtaining an output of the first subtractor (6a) and delaying this output for a first predetermined time (T_0);
   - obtaining an output of the second subtractor (6b) and delaying this output for a second predetermined time (T_0);
   - supplying the delayed signals of each subtractor (6a, 6b) to another subtractor (6b, 6a);
   - obtaining the output of one subtractor (6a, 6b) as a directional signal (F(t), R(t)).

2. A method of claim 1, wherein two subtractors (6a, 6b) are provided each of which is connected with a microphone (1a, 1b) to feed a positive input to the subtractor (6a, 6b), and wherein the output of each subtractor (6a, 6b) is delayed for a predetermined time (T_0) and sent as negative input to the other subtractor (6b, 6a).

3. A method of one of claims 1 or 2, wherein the output signals (F(t), R(t)) of the subtractors (6a, 6b) are analysed and mixed depending on the result of the analysis.

4. A method of one of claims 1 to 3, wherein signals of two microphones (1b, 1c) are mixed and the result of the mixing is sent into the second subtractor (6b).

5. A method of one of claims 1 to 3, wherein three microphones (1a, 1b, 1c) are provided and the signals of each pair of two microphones (1a, 1b; 1b, 1c; 1c, 1a) out of three are processed according to one of claims 2 to 4.

6. An apparatus for picking up sound with at least two essentially omnidirectional microphones (1a, 1b, 1c), a first and a second subtractor (6a, 6b) each of which having an input port connected with a first and at least another microphone (1a, 1b) respectively, a first and a second delaying unit (7a, 7b) having input ports connected with output ports of the first and the second subtractor (6a, 6b) respectively, for delaying the output signals (F(t), R(t)) a predetermined time, wherein an output port of the first delaying unit (7a, 7b) is connected to a negative input port of the second subtractor (6b) and wherein an output port of the second delaying unit (7a, 7b) is connected to a negative input port of the first subtractor (6a).

7. An apparatus of claim 6 wherein, wherein a sound processing unit (5) is provided to modify the directional pattern of the signals (F(t), R(t)).

8. An apparatus of one of claim 6 or 7, wherein two microphones (1a, 1b) are connected with a first and a second subtractor (6a, 6b), respectively.

9. An apparatus of one of claim 6 or 7, wherein three microphones (1a, 1b, 1c) are provided and wherein the signals of the second and the third microphone (1b, 1c) are mixed in an adder (10), an output port of which is connected to the second subtractor (6b).

10. An apparatus of one of claim 6 or 7, wherein three microphones (1a, 1b, 1c) and three discrimination units (4a, 4b, 4c) are provided, wherein the first microphone (1a) is connected to an input port of the second and the third discrimination unit (4b, 4c), the second microphone (1b) is connected to an input port of the first and the third discrimination unit (4a, 4c), and the third microphone (1c) is connected to an input port of the first and the second discrimination unit (4a, 4b).

11. An apparatus of one of claims 6 or 7, wherein more than three microphones (1a, 1b, 1c, ... 1z) are provided which are arranged at the corners of a polygone or polyeder and wherein a set of several discrimination units is provided, each of which is connected to a pair of microphones.

12. An apparatus of one of claims 6 or 7, wherein at least three microphones (1a, 1b, 1c) are provided which are arranged on a straight line and wherein a first and a second microphone (1a, 1b) is connected with the input ports of a first discrimination unit (4a), and the second and the third microphone (1b, 1c) is connected to the input ports of a second discrimination unit (4b) and wherein a third discrimination unit (4c) is provided, the input ports of which are connected to an output port of the first and the second

discrimination units (4a, 4b) and wherein preferably a fourth discrimination unit (4d) is provided, the input ports of which are connected to the other output ports of the first and the second discrimination units (4a, 4b).

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Fig. 1

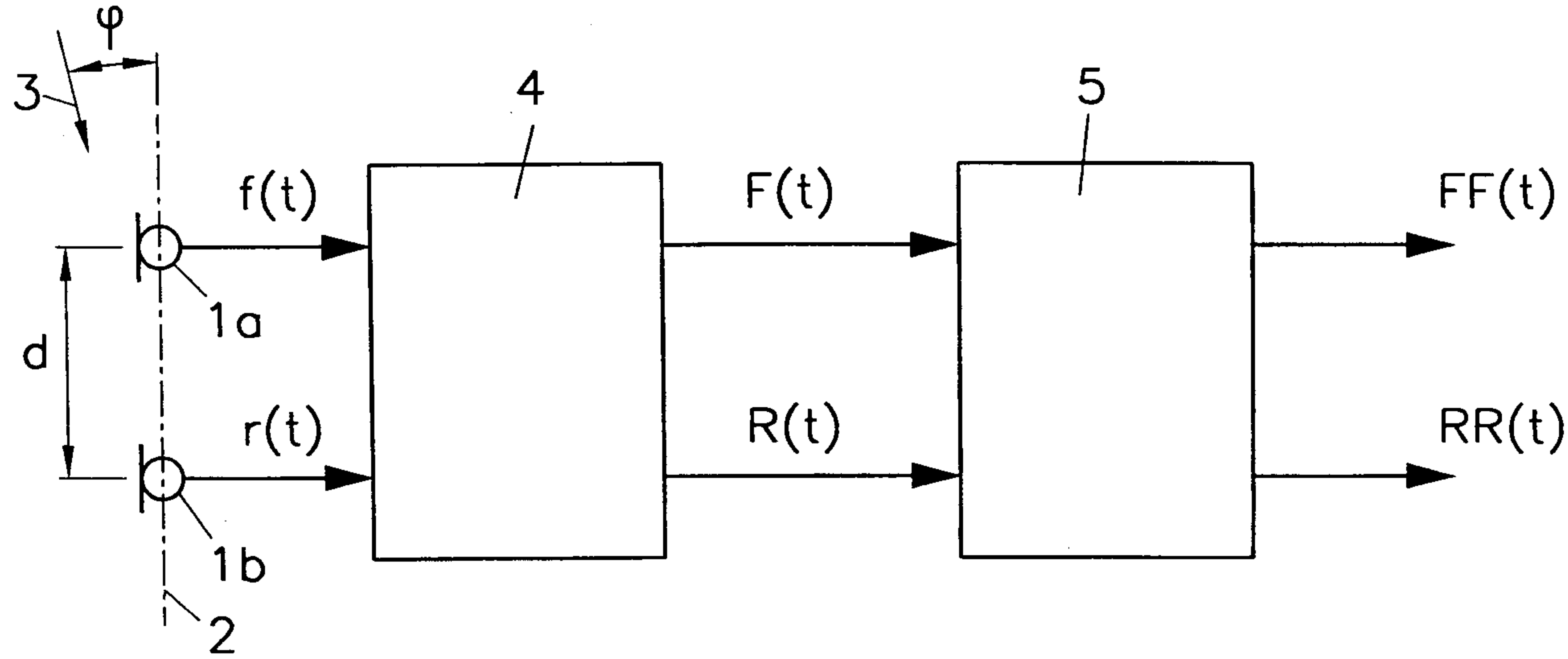


Fig. 2

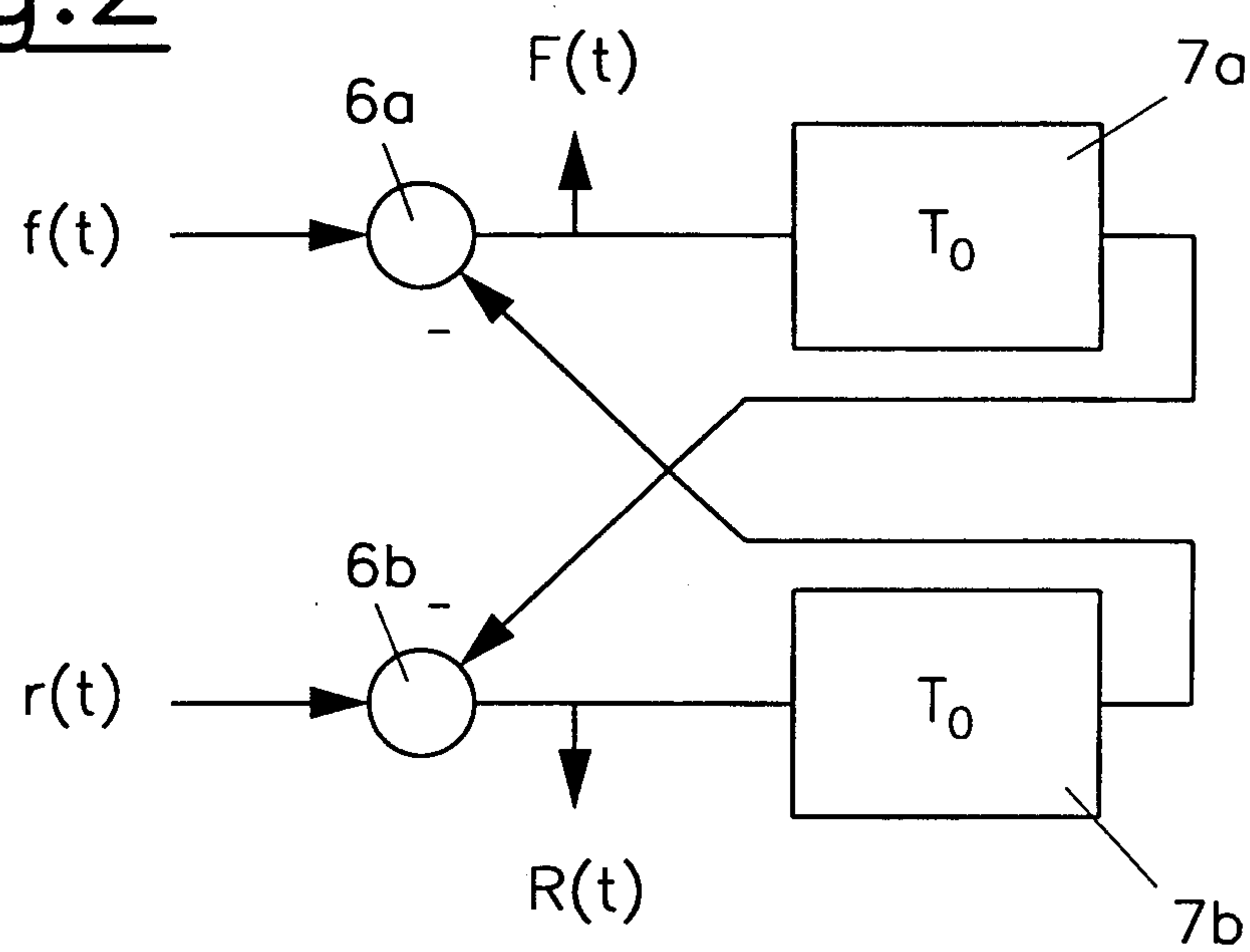


Fig. 3

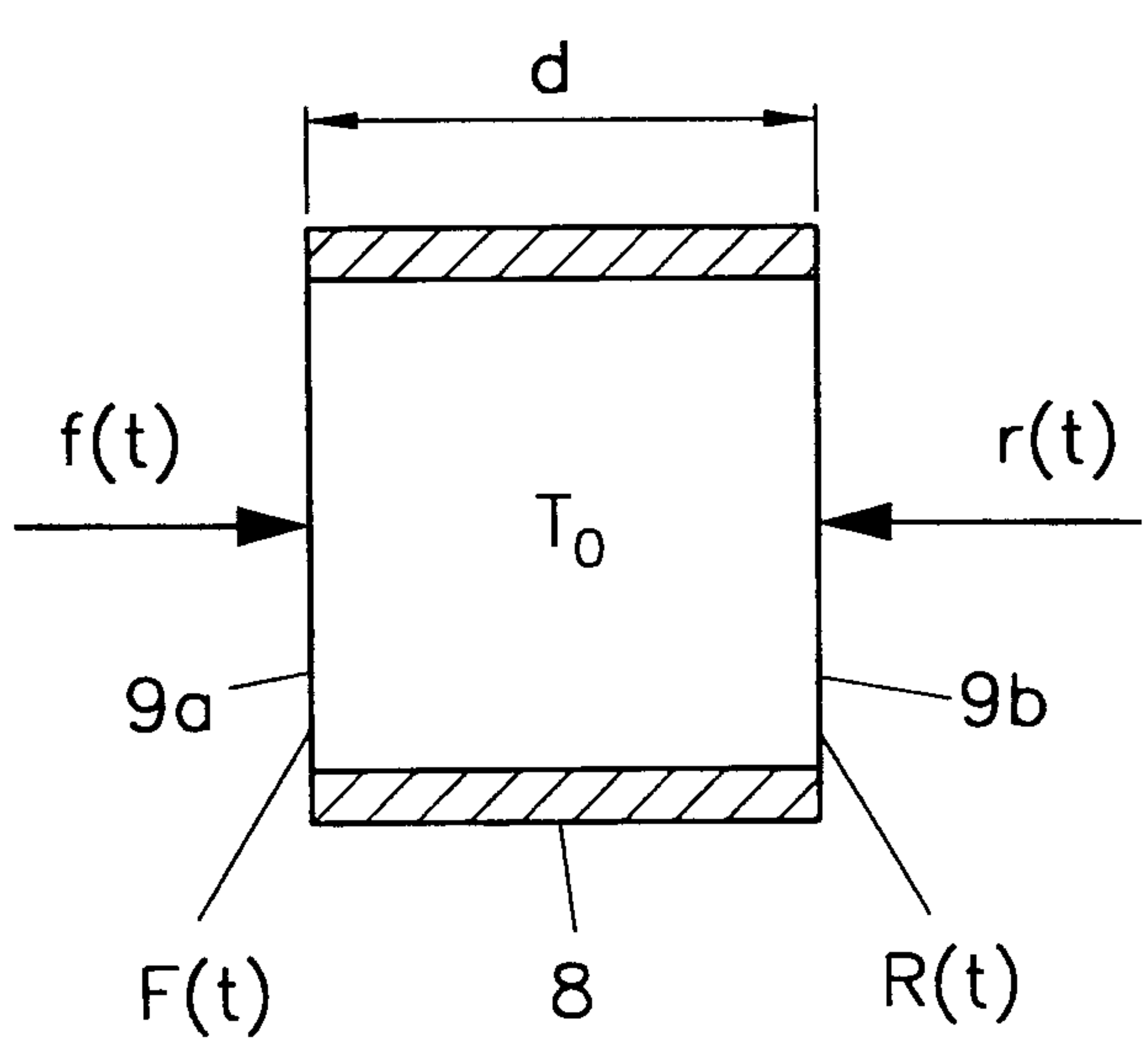
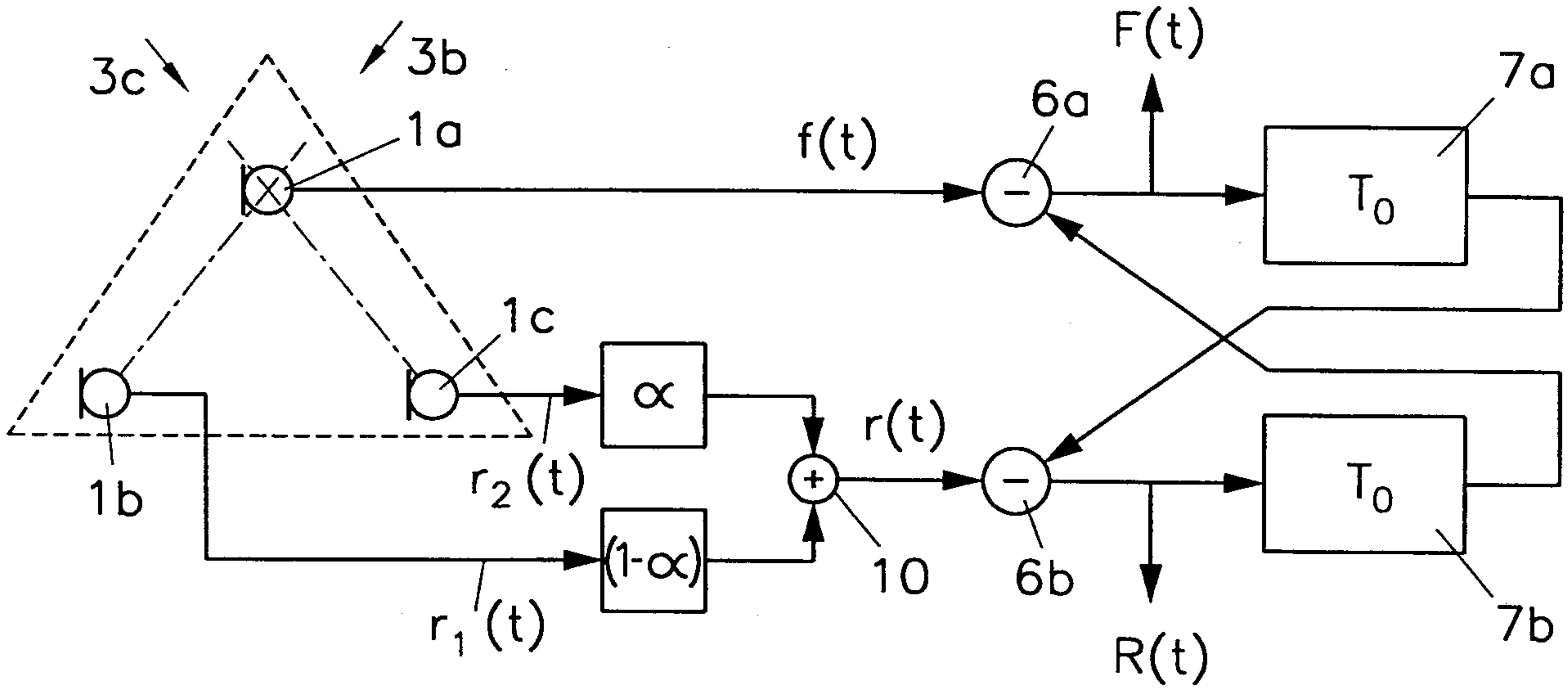


Fig.4a



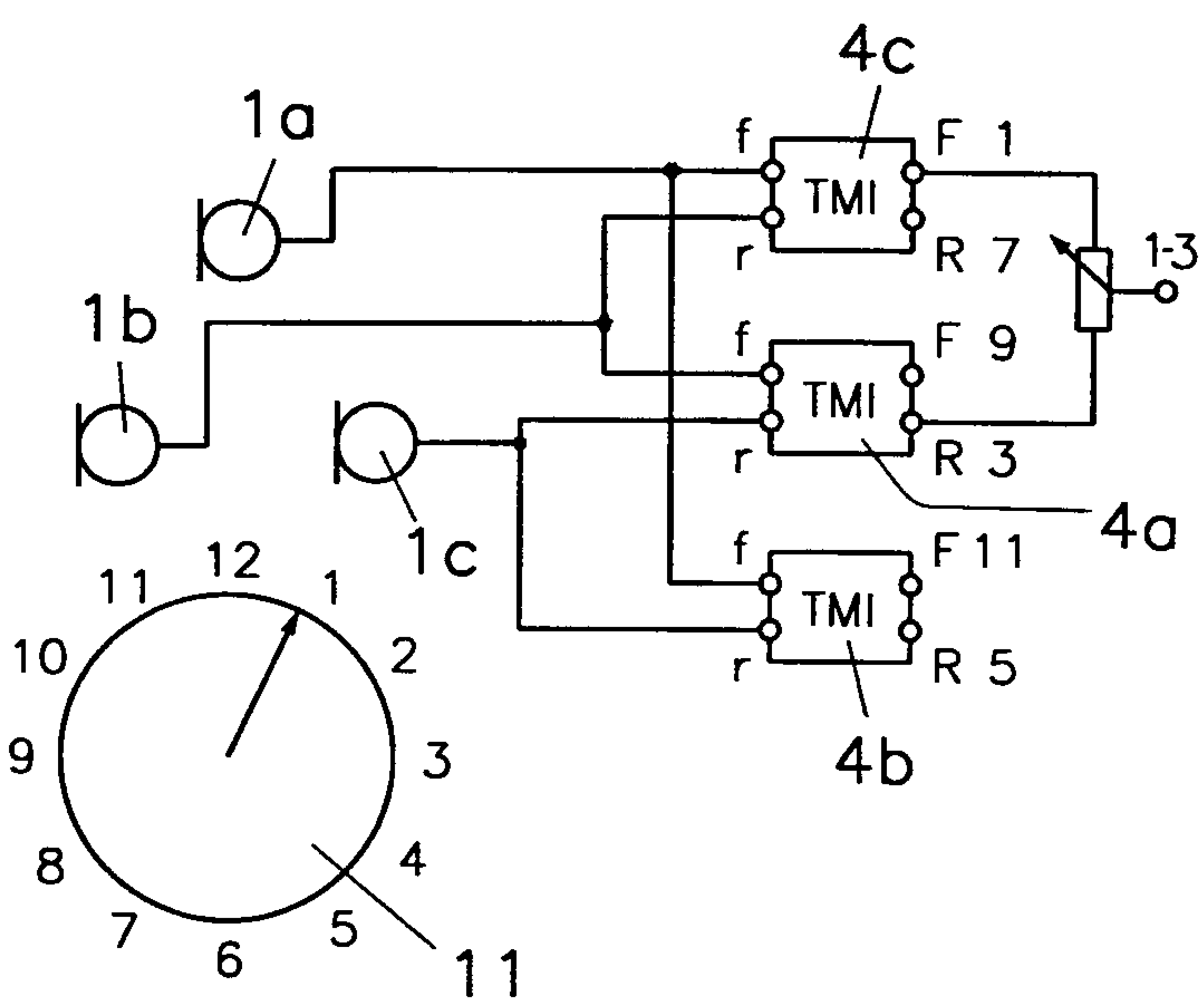


Fig.4b

Fig.5

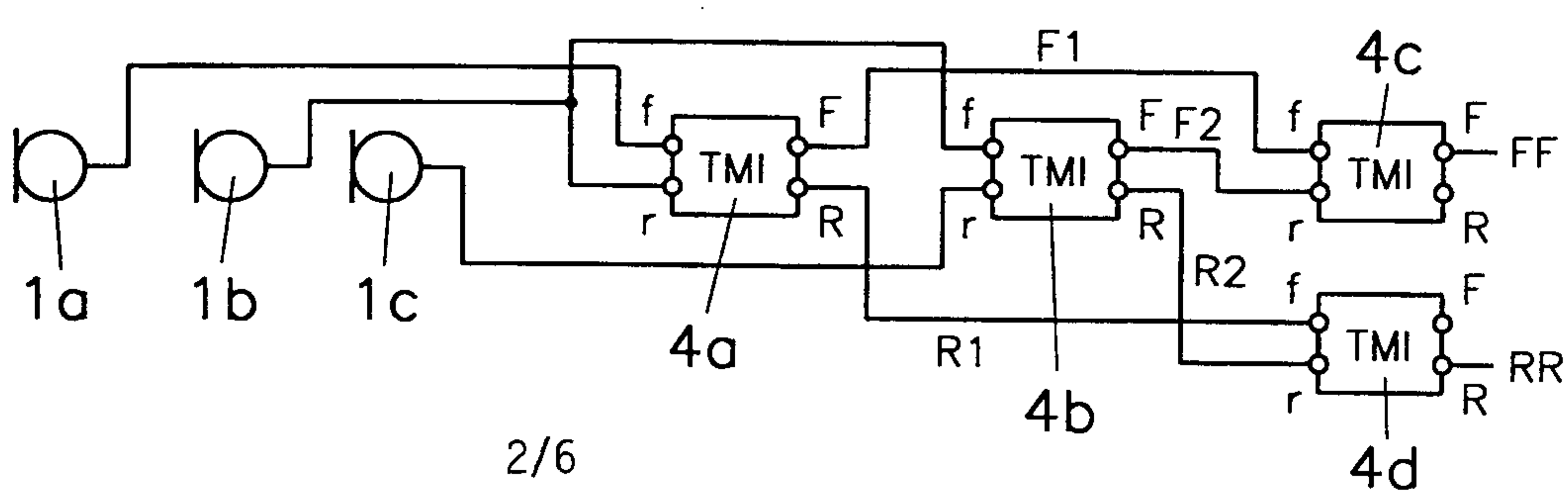


Fig.6

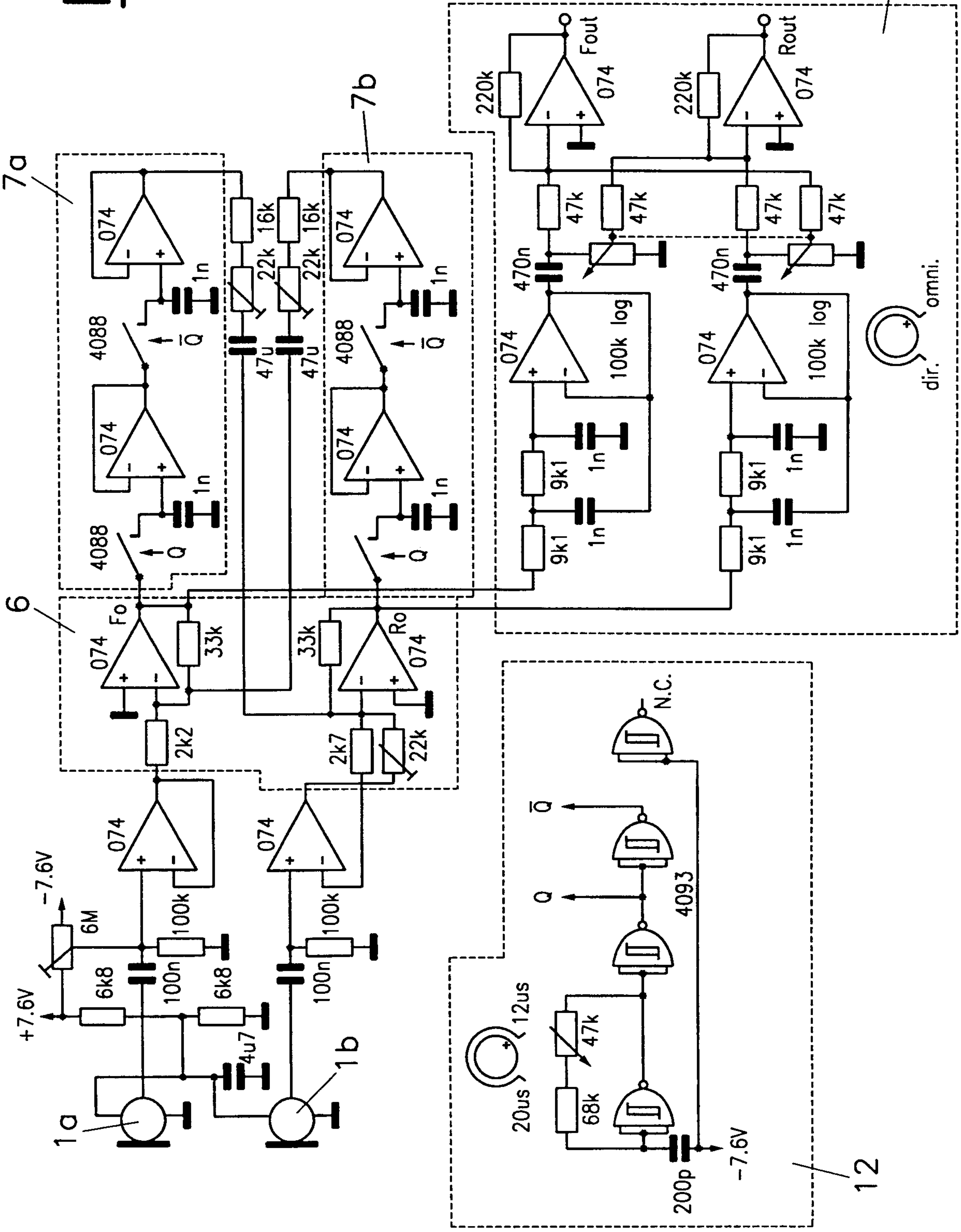

1a
1b
6
7a
7b
12
13
074
4088
4093
Fo
Ro
Fout
Rout
Q
N.C.
+7.6V
−7.6V
6M
6k8
100k
100n
4u7
2k2
2k7
22k
33k
16k
47u
1n
9k1
100k log
470n
47k
220k
68k
200p
20us
12us
dir.
omni.

Fig.7

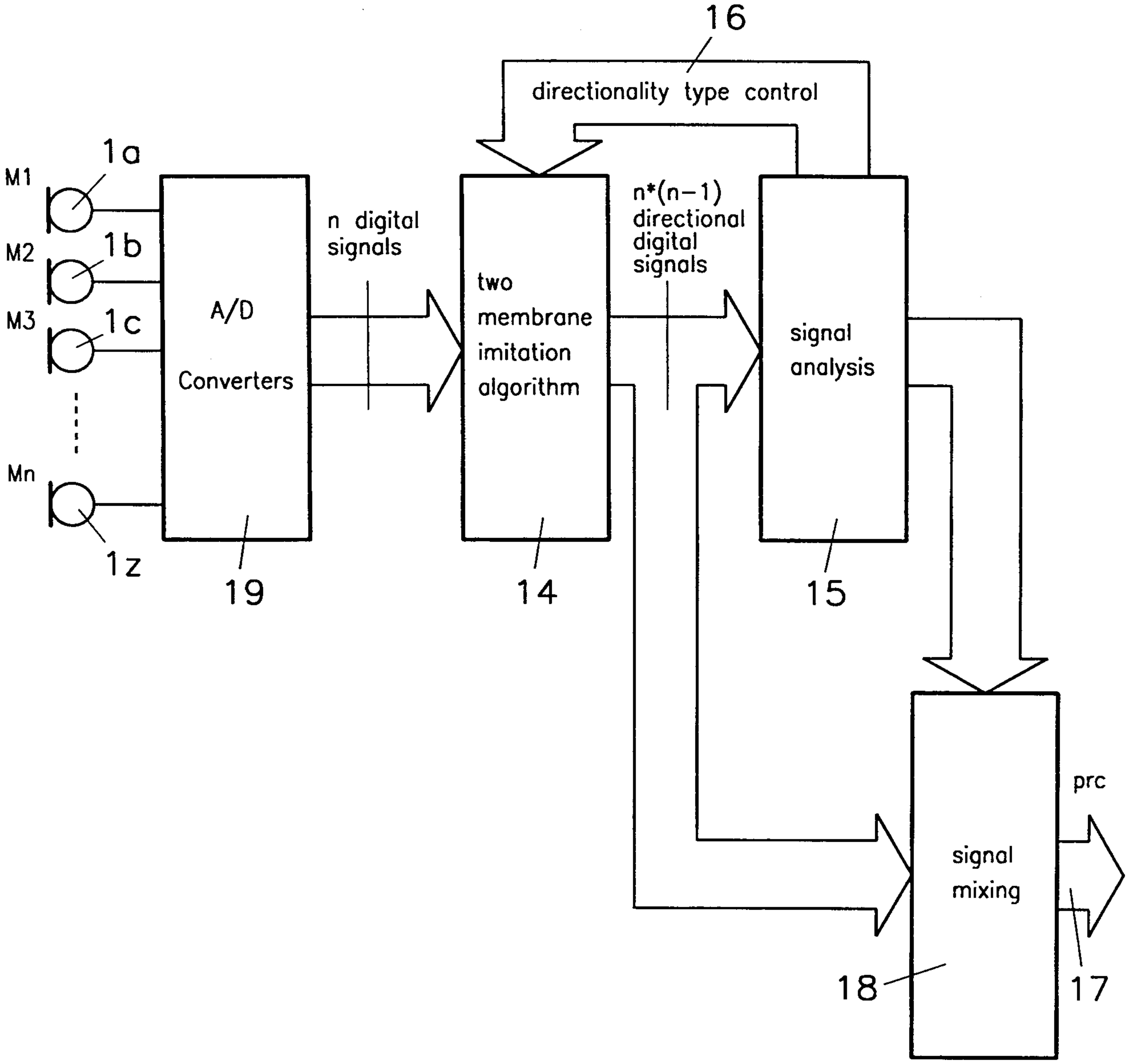
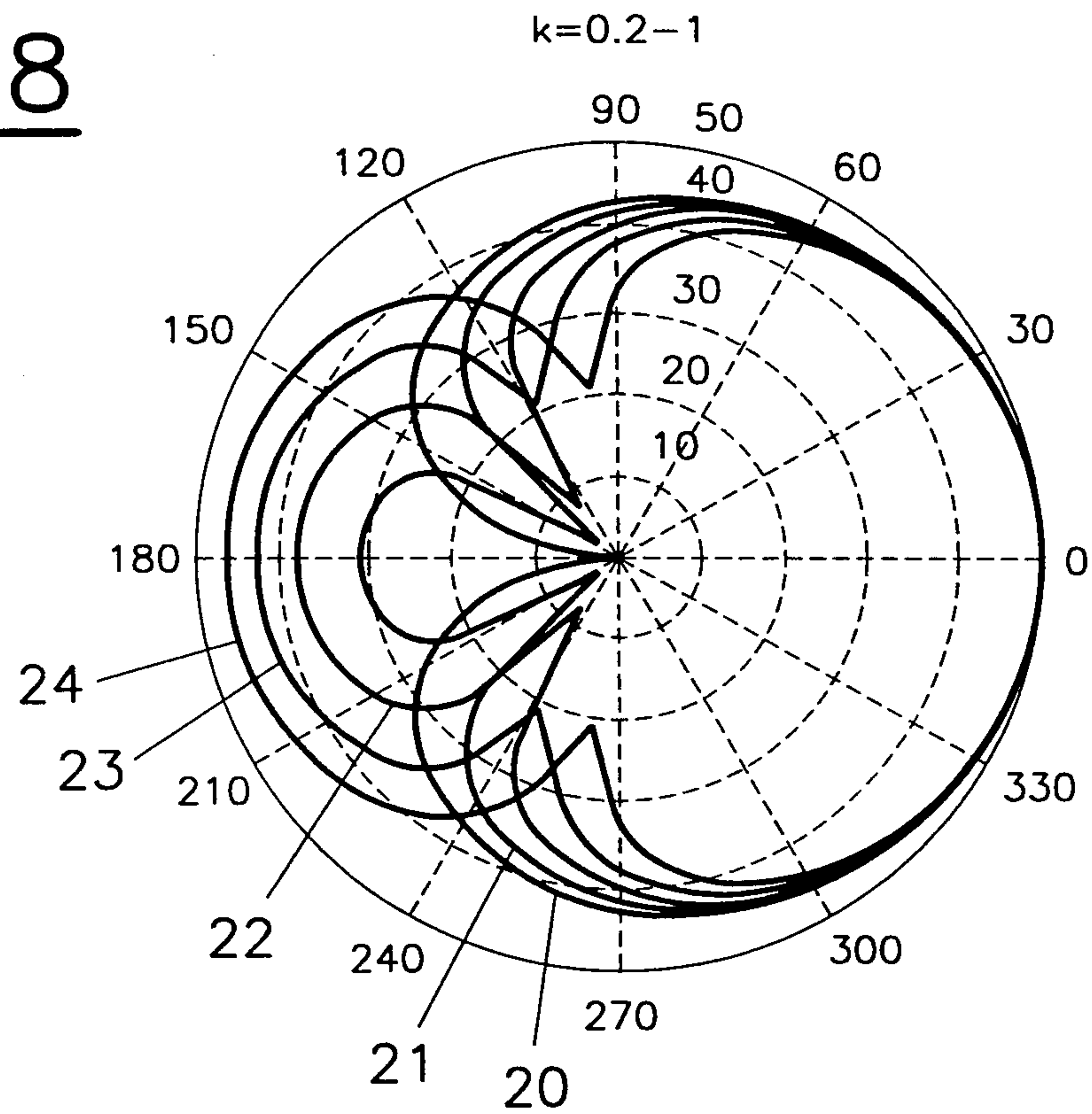

16
directionality type control
1a
M1
M2
1b
M3
1c
Mn
1z
A/D
Converters
19
n digital
signals
two
membrane
imitation
algorithm
14
n*(n-1)
directional
digital
signals
signal
analysis
15
signal
mixing
18
prc
17

Fig.8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90
50
120
40
60
30
150
30
20
10
180
0
24
23
210
330
22
240
300
270
21
20

Fig.9

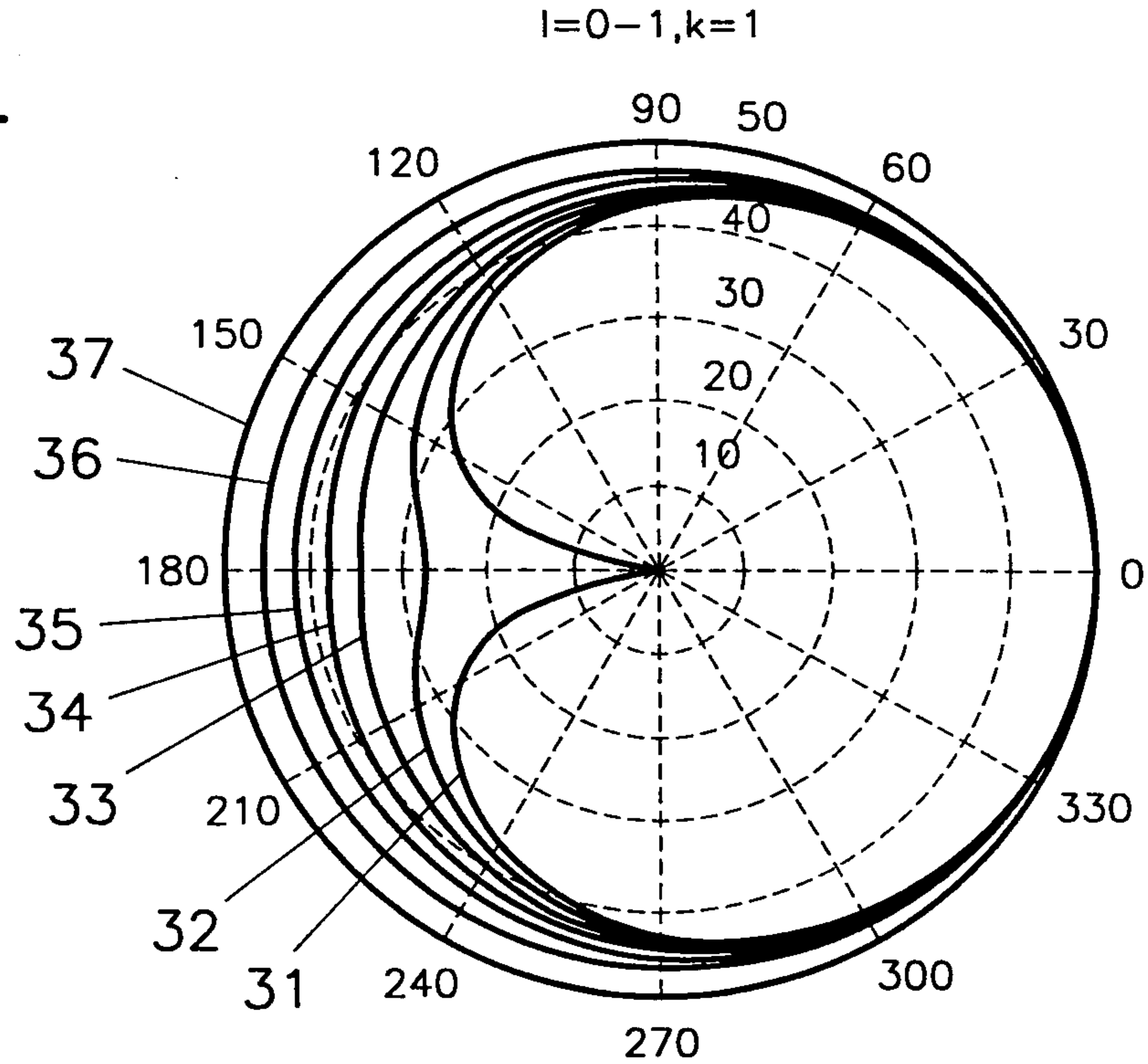
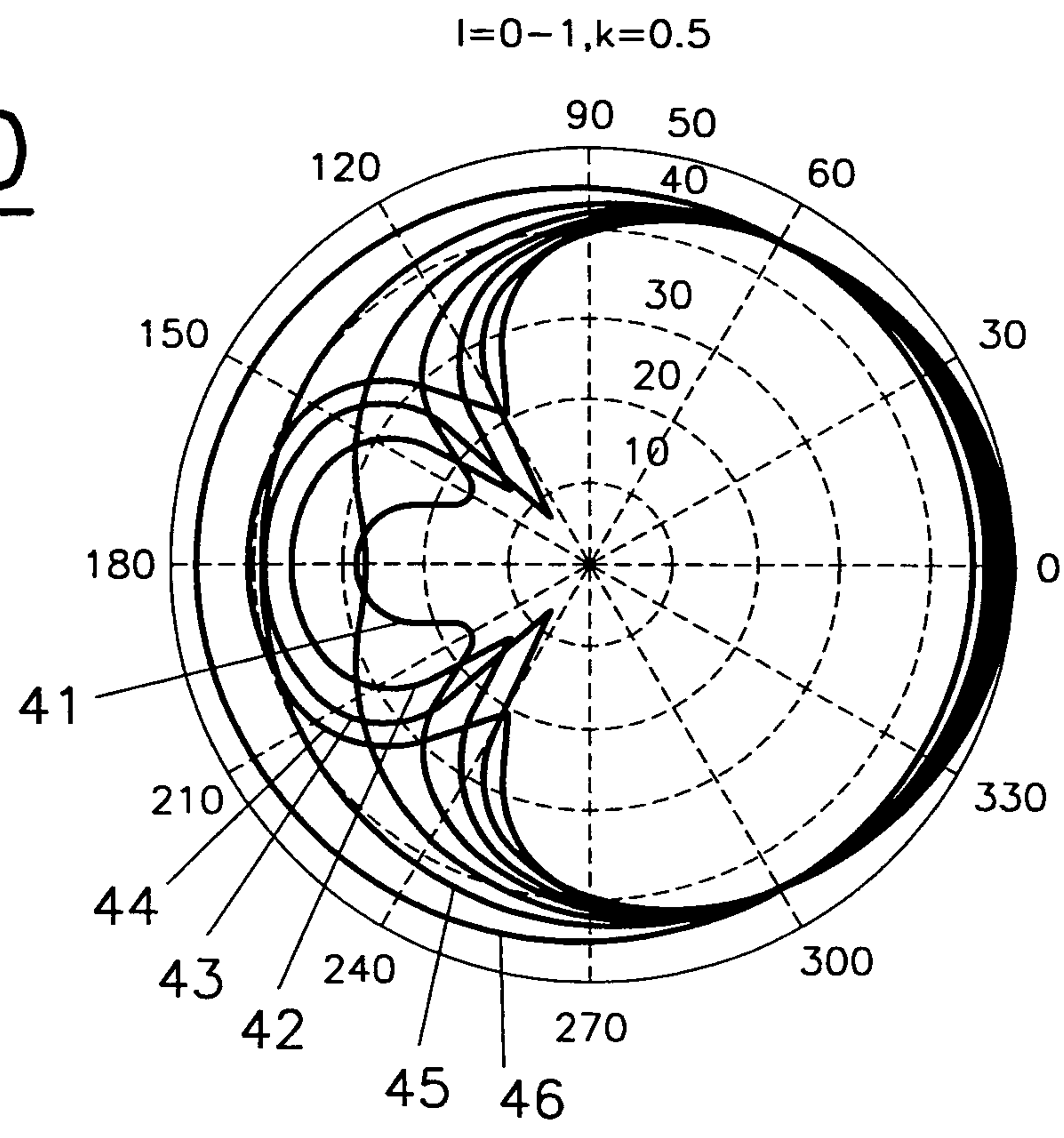

l=0-1,k=1
90
50
120
60
40
30
150
30
20
10
37
36
180
0
35
34
33
210
330
32
31
240
300
270

Fig.10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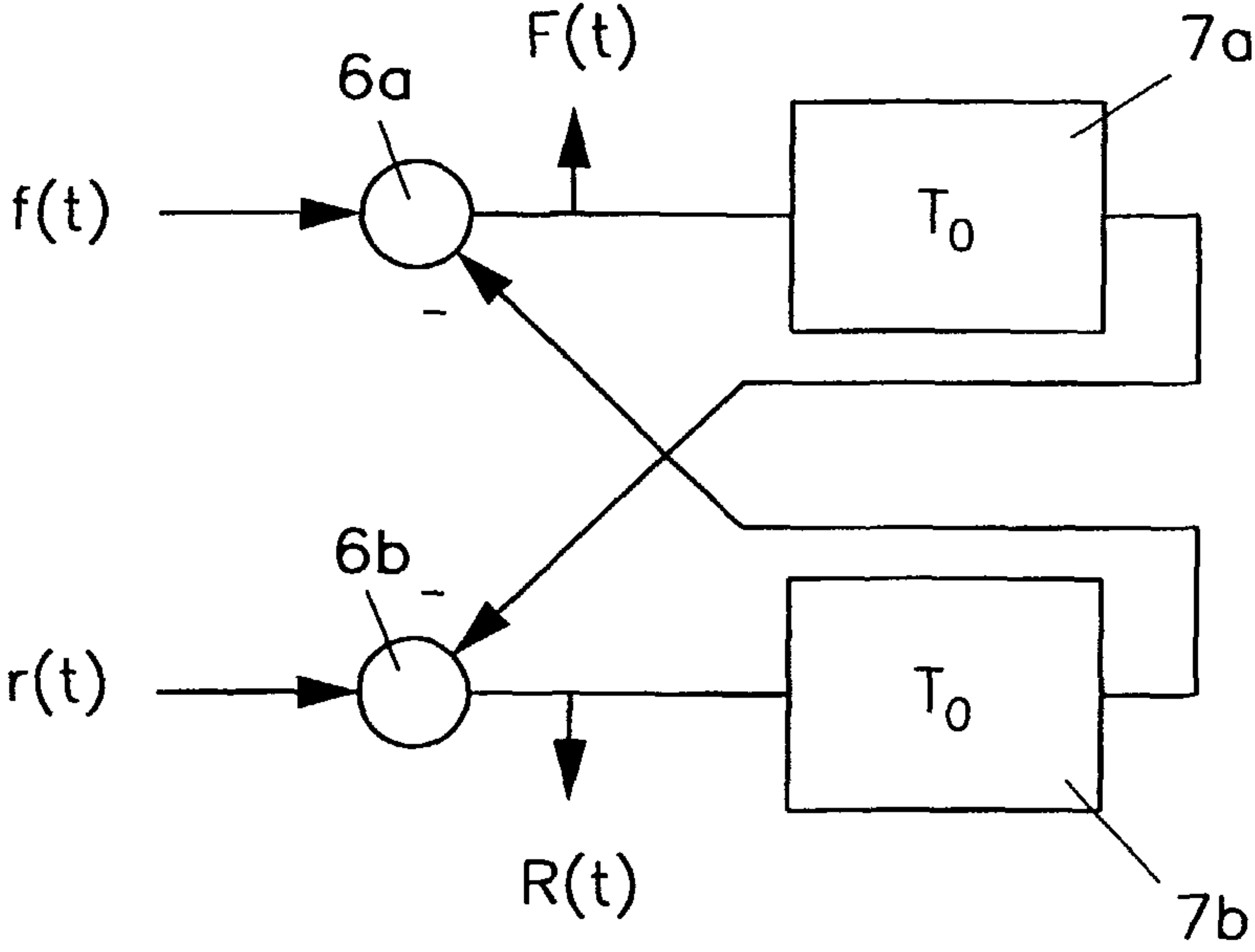
120
40
60
30
150
30
20
10
180
0
41
210
330
44
43
240
300
42
270
45
46


F(t)
7a
6a
f(t)
T_0
6b
r(t)
T_0
R(t)
7b