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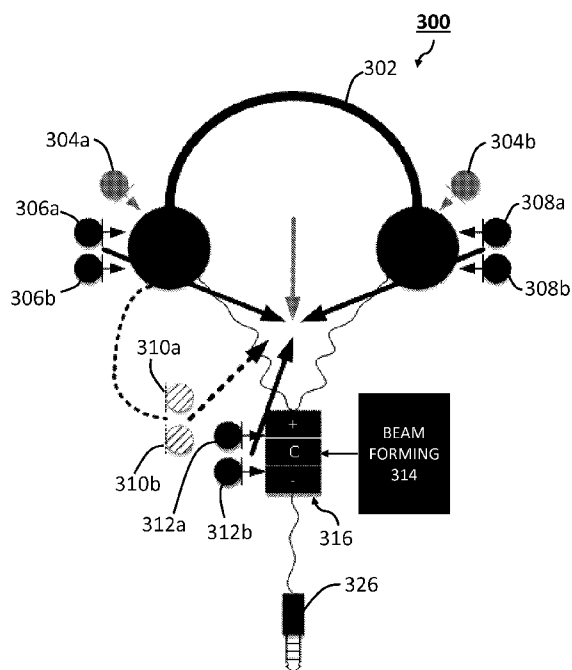


FIGURE 3

(57) Abstract: Instead of using directional microphones and boom arms, beamforming technology can be used to improve audio quality. Beam forming, traditionally, requires a separate power source to power multiple microphones and a digital signal processor. When the microphone jack of telephones and mobile electronic devices offers a maximum of 1 milliwatt of power, it becomes a challenge to provide beam forming on a headset. To work within an ultra-low power budget, a system having a special integrated circuit can implement beam forming with two or more microphones, e.g., for a headset. The integrated circuit can include a localized bias can be provided to reduce the power delivered to the microphones. With some left-over in the power budget, the integrated circuit can include beam forming circuitry that processes signals from the microphones to enable a beam formed signal to be generated and transmitted back via the microphone jack.

LOW POWER UPLINK NOISE CANCELLATION

CROSS-REFERENCE TO RELATED APPLICATIONS

[0001] This application claims priority to U.S. Provisional Application Serial No. 62/023,544, entitled "LOW POWER UPLINK NOISE CANCELLATION", filed on July 11, 2014, which is hereby incorporated by reference.

TECHNICAL FIELD

[0002] The present invention relates to the field of integrated circuits, in particular to headsets with low power uplink noise cancellation.

BACKGROUND

[0003] Signals received at a telephone (e.g., mobile phones) microphone, such as a microphone on a headset, often include significant environment noise, compromising the audibility of the speaker. One conventional solution to decreasing the noise in the signal is using directional microphones, which are sensitive to sounds from a specified direction. Another conventional solution is the use of boom arms to position the microphone close to the speaker's mouth. However, directional microphones are expensive, and boom arms are unwieldy and often undesirable to users.

OVERVIEW

[0004] Instead of using directional microphones and boom arms, beamforming technology can be used to improve audio quality. Beam forming, traditionally, requires a separate power source to power multiple microphones and a digital signal processor. When the microphone jack of telephones and mobile electronic devices offers a maximum of 1 milliwatt of power, it becomes a challenge to provide beam forming on a headset. To work within an ultra-low power budget, a system having a special integrated circuit can implement beam forming with two or more microphones, e.g., for a headset. The integrated circuit can include a localized bias can be provided to reduce the power delivered to the microphones. With some left-over in the power budget, the integrated circuit can include beam forming

circuitry that processes signals from the microphones to enable a beam formed signal to be generated and transmitted back via the microphone jack.

BRIEF DESCRIPTION OF THE DRAWINGS

[0005] To provide a more complete understanding of the present disclosure and features and advantages thereof, reference is made to the following description, taken in conjunction with the accompanying FIGUREs, wherein like reference numerals represent like parts, in which:

[0006] FIGURE 1 is a diagram illustrating a headset and microphone;

[0007] FIGURE 2 is a diagram illustrating a headset, microphones, a beam forming circuit, and a power source;

[0008] FIGURE 3 is a diagram illustrating a headset and microphones, according to some embodiments of the disclosure, according to some embodiments of the disclosure;

[0009] FIGURE 4 is a diagram illustrating a low power noise cancellation system including microphones and beam forming technology, according to some embodiments of the disclosure; and

[0010] FIGURE 5 is a diagram illustrating a low power noise cancellation method, according to some embodiments of the disclosure;

[0011] FIGURE 6 is a diagram illustrating a low power noise cancellation system of FIGURE 4 with wind noise detection and/or cancellation enhancements, according to some embodiments of the disclosure;

[0012] FIGURE 7 is a diagram illustrating a low power noise cancellation method having wind noise detection and/or cancellation, according to some embodiments of the disclosure; and

[0013] FIGURE 8 is a diagram illustrating a method for programming the low power noise cancellation system, according to some embodiments of the disclosure.

DESCRIPTION OF EXAMPLE EMBODIMENTS OF THE DISCLOSURE**[0014]** Challenges to providing beam forming

[0015] One obstacle to using beam forming with multiple microphones to enhance an audio signal is that a standard telephone headset supports only one microphone. According to one approach, an external device, such as a headset, including two or more microphones is used. Beam forming is performed at the external device on the analog signals, and the multiple microphone signals are converted into one virtual microphone signal that represents the focused audio beam from the beam forming system. The single, beam formed virtual microphone signal is input to the telephone.

[0016] While conventional systems exist for converting multiple microphone signals into one virtual microphone signal through beam forming, these systems all consume high amounts of power. Additionally, conventional beam forming system perform beam forming after digitalization of the signals. Most existing external beam forming solutions consist of a digital signal processing system combined with an analog front-end and an analog back-end to enable the integration into the headset system. Thus, in these existing systems, the incoming analog signal is converted to a digital signal for digital signal processing and beam forming, and after processing, the signal is converted to an analog signal.

[0017] Conventional systems require a separate power source, since a typical phone system, such as a cellphone, can deliver a maximum of 1 milliwatt (mW) into the microphone bias line. Laptops and tablets similarly deliver a maximum of 1 mW or power into the microphone bias line. Phone systems are designed to make available this amount of power because it is sufficient for powering the JFET amplifier embedded into a typical microphone.

[0018] FIGURE 1 is a diagram 100 illustrating a conventional headset 102 and microphone 104. The microphone is connected through a unit 106 to headphones 108 and a microphone plug 110. When the microphone plug 110 is connected to a microphone jack of a device, a signal received at the microphone 104 is sent to the device, and sound from the device is output to the headphones 108. Additionally, the microphone jack can output up to one milliwatt of power to the headset 102, which is sufficient to power the JFET amplifier of the microphone 104.

[0019] FIGURE 2 is a diagram 200 illustrating a headset 208, microphones 204a, 204b, a beam forming circuit 212, and a power source 214. The power source can be a battery, and/or AC power. A signal received at the microphones 204a and 204b can be amplified at the beam forming circuit 212. The (conventional) beam forming circuit 212 uses digital processing and consumes power from the power source 214.

[0020] Providing beamforming on an ultra-low power budget

[0021] Systems and methods for selectively amplifying a signal received at a microphone are provided. In particular, systems and methods are provided for applying beam forming technology to signals from multiple microphones. The application of beam forming to signals received from two or more microphones allows for selective amplification of an audio signal from a selected direction. Beam forming is performed on the incoming analog signals. According to one feature, this allows multiple microphones to act like a directional microphone.

[0022] According to one implementation, systems and methods are provided for using beam forming with multiple microphones to enhance an audio signal using a maximum of 1 mW of power. The systems and methods can be used to enhance the audio signal of a standard telephone.

[0023] FIGURE 3 is a diagram 300 illustrating a headset 302, with two or more microphones (including two or more of microphones 304a, 304b, 306a, 306b, 308a, 308b, 310a, 310b, and 312a, 312b), and a beam forming circuit 314, according to some implementations of the disclosure. The microphones 304a, 304b, 306a, 306b, 308a, 308b, 310a, 310b, and 312a, 312b can be positioned at various locations around the headset 302, and are each provided for sensing audio and outputting a signal. Using a few different microphone placement options drives a large number of potential beam directions. To support many different microphone positions, a large degree of freedom is used in configuring the beam forming circuit 314.

[0024] The beam forming circuit 314 can be provided as a part of an integrated circuit on the headset 302. In one example, the beam forming circuit 314 can be provided in unit 316, which is provided along the cabling which connects the headset 302 with the microphone plug 326. Unit 316 can enclose the integrated circuit having the beam forming circuit 312 and

electrical connections/wiring for the headphones and microphones. In another example, the beam forming circuit 314 can be provided in an ear piece of the headset 302, or some other part of the device seen in diagram 300.

[0025] FIGURE 4 is a diagram illustrating a low power noise cancellation system 400 including microphones 402a, 402b and beam forming technology, according to some implementations of the disclosure. As shown in FIGURE 4, the system 400 includes local microphones 402a, 402b, a microphone bias 404, microphone pre-amplifiers 406a, 406b, a microphone matching filter 408, an Xtal amplifier 410, beam steering filters 412a, 412b, a summation circuit 414, a virtual microphone amplifier 416, a low dropout regulator (LDO) 418, a filter coefficient fetcher 420 (e.g., "METAL PROGRAMMABLE DEFAULT CONFIGURATION + I2C SELFBOOT"), and a memory block 422 (e.g., "I2C EEPROM"). The local microphones 402a, 402b are each connected to a pre-amplifier 406a, 406b, and both microphones 402a, 402b are connected to the bias 404. The pre-amplifiers 406a, 406b are connected to the microphone matching filter 408, which is connected to the beam steering filters 412a, 412b. The beam-steering filters 412a, 412b are connected to the summation circuit 414, such that the output of each steering filter 412a, 412b is input to the summation circuit 414. The summation circuit 414 connects to the virtual microphone amplifier 416, which connects to the low dropout regulator (LDO) 418. The xtal amplifier 410 also connects to the matching filter 408, and to the filter coefficient fetcher 420.

[0026] In some embodiments, a low power system (e.g., the low power noise cancellation system 400) for improving audio quality comprises a first and second microphones (e.g., microphones 402a, 402b, or more) for sensing audio and outputting a first and second received signal, and an integrated circuit comprising: first and second pre-amplifiers (e.g., microphone pre-amplifiers 406a, 406b) for amplifying the first and second received signals; and a beam forming circuit for combining the first and second amplified signals and outputting a beam formed signal. The low power system operates on less than one milliwatt of power. For instance, the power can be supplied by a telephone system, or a mobile electronic device, a computing device, etc.

[0027] Rather than relying primarily on digital signal processing to provide beam forming, the system 400 includes Sampled Analog Technology, which uses complex, configurable, and low power adaptive analog filters for low-powered beam forming of the

analog signal. Sampled Analog Technology is Analog Devices technology, and is described in greater detail in U.S. Patent No. 8,188,753, U.S. Patent No. 8,547,272, U.S. Patent No. 8,717,094, and U.S. Patent No. US 8,736,361, which are assigned to Analog Devices and incorporated by reference in their entirety. Sampled Analog Technology allows analog signals to be processed by analog circuitry (e.g., switched capacitor circuits) through charge sharing and redistribution. Advantageously, filters and mathematical operations can be performed on analog signals without the use of digital circuitry, or the need for an analog front end for digitizing the analog signals into digital signals. Furthermore, the filters and mathematical operations can be performed with much lower power with analog circuitry than their digital counterparts. The analog circuitry can also be much smaller in size/area compared to their digital counterparts.

[0028] The system 400 is a flexible uplink noise cancellation system that is low power enough that it can operate on the <1 mW microphone bias supply supported by all telephone systems. In some embodiments, the system 400 is configurable or programmable to suit the particular configuration of the headset, specifications of the microphone, etc.

[0029] These components (aside from the microphones themselves) of the low power system (e.g., system 400 or other systems described herein) can be provided on an integrated circuit that is embeddable on a headset. The system 400 preferably consumes less than one milliwatt of power. Part of the power budget is used for powering the microphones (e.g., 402a and 402b), and part of the power budget is used for powering the integrated circuit. As a result, the headset can provide beam forming without a large digital signal processor, analog front end, and a separate power source. The resulting integrated circuit having these components of the system preferably has a small form factor, making it suitable to be embedded even in the most lightweight headsets.

[0030] Microphone management

[0031] One step in developing a system that consumes less than one milliwatt of power is to lower the power consumed by the microphones 402a, 402b, thus, starving the microphones. If the microphone(s) can be starved, two or more microphones and an integrated circuit for improving the audio quality can be powered without the need to provide a separate power source.

[0032] According to one implementation, local management of the microphones 402a, 402b can be utilized to decrease microphone power consumption. Instead of running the microphones 402a, 402b off the BIAS provided from the microphone jack (not shown), the provided BIAS is input to a localized BIAS 404 that supplies the attached microphones 402a, 402b with less current (and less power) than the provided BIAS. The localized BIAS 404 can provide first and second microphones (e.g., microphones 402a, 402b or other microphone(s)) with a constant, but lower, amount of power.

[0033] When supplied with a smaller amount of power, the resulting output swing from the microphones' JFET amplifiers is lower. The system 400 compensates for the lower output swing of the JFET amplifiers with local amplification using the microphone preamplifiers 406a, 406b, which amplifies received signals from the microphones. For instance, the integrated circuit of the low power system can include first and second pre-amplifiers (e.g., microphone preamplifiers 406a, 406b) for amplifying first and second received signals received from first and second microphones for sensing audio (e.g., microphones 402a and 402b, or other microphones).

[0034] Since the uplink system 400 is positioned locally by the microphones 402a, 402b (e.g., on ear pieces, in a piece of the headset, along cabling of the headset, in unit on the cabling of the headset), the microphones 402a, 402b drive a lower load than usual in a telephone headset. In one example, the distance between the other components of the system 400 and the microphones 402a, 402b is less than 2 mm, while a typical headset includes about 3 feet of cable between the microphones and any noise cancellation or signal enhancing technology. The short distance between the microphones 402a, 402b and the other components of the system 400 allows the system 400 to lower the power consumed by the microphones 402a, 402b, pre-amplifiers 406a, 406b, and bias 404 from 1mW per microphone 402a, 402b to less than 400uW (microwatts) per microphone 402a, 402b.

[0035] In some implementations, one of the microphones 402a, 402b may be at one ear piece on a headset along with the beam forming components of the system 400, and the other one of the microphones 402a, 402b may be at another location on the headset. In one example, one of the microphones 402a is at one ear piece, and the other microphone 402b is at the other ear piece. In another example, one of the microphones 402a is at one ear piece on a headset, and the other one of the microphones 402b is positioned locally, closer to the

speaker's mouth. In some cases, one or more of the microphones may be positioned along the cabling of the headset (e.g., on cables, in unit 316 of FIGURE 3). Thus, in various implementations, the distance between the beam forming components of the system 400 (e.g., beam forming circuitry 314 of FIGURE 3) and one of the microphones may be closer to one foot. Total power consumption of, e.g., the microphones 402a, 402b and system 400, is maintained at less than 1 mW.

[0036] The bias provided from the microphone jack is input over the microphone bias line. The bias is also known as the microphone output/microphone signal as seen from the microphone. The microphone input (as seen from the receiver) has a DC bias applied to it. In various examples, the microphone bias line may also be referred to as the microphone output, the microphone signal, and the microphone input. The microphone bias line carries the signal recorded by the microphone back towards to the microphone receiver/amplifier. The microphone bias line also provides a DC voltage which traditionally is used to keep a charger of a condenser microphone.

[0037] There are many different types of microphones all of which use different DC voltage levels to build enough charge across the capacitor/condenser be able to generate a voltage based on a measure or sense audio, i.e., acoustical waveform. For example, electret microphones contain a built-in FET amplifier that performs pre-amplification of the signal generated across the minute cap/condenser found in miniature microphones used for headsets/cellphones.

[0038] For a typical microphone used in headset/tables/cellphones/car linings the DC voltages can be between 1.8V and 3.3V, with 2.2V being the typical value. The DC voltage is generated by a microphone bias generator. The microphone bias generator is part of the microphone input as part of the microphone receiver circuit.

[0039] In typical applications the microphone receiver circuit has a 2.2k ohm resistor in series with the microphone bias generator. The microphone input in the microphone receiver circuit also includes an AC coupling capacitor in series with the microphone receiver circuit input amplifier. Because of the series resistance between the microphone bias generator and the microphone bias line, there is a natural limit to the amount of power that can be drawn from the microphone bias generator. This limit can be calculated as the typical microphone DC bias voltage level of 2.2V divided by the typical microphone bias generator

series resistor of 2.2kOhm, to be 1mW. Thus, there is a power limitation for the amount of power that can be available and used by circuitry for improving audio quality (e.g., the uplink noise cancellation circuit).

[0040] Beam forming: microphone matching

[0041] The next step is to check that the attached microphones 402a, 402b are matched, by providing a microphone matching filter 408 in the beam forming circuit. Any mismatch in the microphones 402a, 402b amplitude or phase will result in the beam being off. The delay sum beam former of the system 400 depends on the microphones being matched. According to one implementation, the microphone matching filter 408 matches the amplitude and phase of the received signals from a first and second microphones (e.g., microphones 402a, 402b or some other microphone(s) if applicable), respectively. According to another implementation, the microphone matching filter 408 matches a gain of the first amplified signal (from the first preamplifier) with a gain of the second amplified signal (from the second preamplifier) and outputting first and second matched signals. The microphone matching filter 408 is preferably implemented with Sampled Analog Technology. Accordingly, the microphone matching filter 408 includes analog circuitry for processing the first and second amplified signals to provide microphone matching.

[0042] In some embodiments, the microphone matching filter 408 is a static matching filter, where parameters for matching the first and second microphones (e.g., gain, phase, etc.) are set during before use (e.g., during manufacture or test). In some embodiments, the microphone matching filter 408 is a dynamic matching filter, which changes/adapts parameters for matching the first and second microphones during use. In some embodiments, the microphone matching filter 408 can both a static and dynamic matching filter, where parameters are programmed before use and the same parameters can be changed/adapted during use.

[0043] According to some implementations, microphone matching, specifically dynamic microphone matching, can be turned off due to wind noise. Wind noise can cause the microphone matching filter 408 to track the wind, interfering with tracking of a speaker or other target signal. According to one implementation, wind noise can be detected by the system 400. In one example, the system 400 uses a low pass filter to detect wind noise. In

one implementation, when wind noise exceeds a selected threshold, the system 400 turns off the microphone matching circuit. In another example, a front-end component including a high pass filter can be added to the system 400 to remove the wind noise, which tends to occur at lower frequencies. The high pass filter can have a dynamic cut-off frequency, with the cut-off frequency selected based on the severity of the wind noise. According to one feature, in order to maintain target signal quality, the frequency value of the cut-off frequency is set as low as possible. In some implementations, wind noise removal is an alternative to beam forming. In other implementations, wind noise removal is an alternative to microphone matching. An embodiment of wind noise detection and/or cancellation is described in greater detail in later passages.

[0044] Beam forming: delay sum beam former

[0045] A beam forming algorithm is implemented in analog circuitry, including switch capacitor circuit that includes steering filters 412a, 412b implemented as sub-sample phase delay finite impulse response (FIR) filters, and a summation circuit 414. Accordingly, the beam forming circuit further includes first and second steering filters (e.g., steering filters 412a, 412b) for (1) receiving first and second matched signals and (2) outputting first and second steered signals. The first and second steering filters can be sub-sample phase delay finite impulse response filters, preferably implemented with Sampled Analog Technology. The beam forming circuit further includes a summation circuit (e.g., summation circuit 414) for combining the first and second steered signals. The summation circuit outputs a beam formed signal. With Sampled Analog Technology, the first and second matched signals are analog signals, and the beam forming circuitry having the first and second steering filter and summation circuit 414 comprises analog circuitry for processing and combining the analog signal.

[0046] The output of the beam former from the summation circuit 414 is passed through a virtual microphone amplifier 416 for amplifying the beam formed signal. The virtual microphone amplifier 414 is adjusted to down stage the output signal from the summation circuit 414 to a level consistent with the expected amplitude found in a microphone JFET amplifier. The resulting signal (the signal output from the virtual microphone amplifier 416)

is an AC signal, and is coupled to the phone's microphone bias line to send the beam formed signal back to the phone itself.

[0047] Support blocks

[0048] A number of support blocks are used in the uplink noise cancellation system 400. One support block is an LDO 418, used between the phone microphone BIAS signal and the internal BIAS 404 supply of the system 400. The incoming BIAS from the microphone jack can vary depending on the vendor and the load, and the LDO 418 ensures that the system 400 has a stable supply. As the BIAS voltage can vary from 1.8V to 3.3V in conventional systems, depending on loading, sizing of series resistor, selection of audio codec, and BIAS settings, the LDO ensures that the system 400 has a constant supply voltage independent of the actual current drawn from the BIAS line. Since the result of the beam former is fed back into the phone BIAS line, which in turn feeds the system 400, an isolation circuit is used to provide some level of Power Supply Rejection Ratio (PSRR) to avoid electro acoustic feedback. According to one feature, the LDO 418 provides a PSRR that is high enough that the audio signal output to the BIAS is dampened and will not adversely affect performance of the system 400.

[0049] Another support block is the filter coefficient fetcher 420. In one example, the filter coefficient fetcher 420 is an I2C/SPI coefficient reader. According to one implementation, the uplink noise cancellation system uses configurable analog filters, and Original Design Manufacturers (ODMs) may change the coefficients of the microphone matching circuit 408, steering filters 412a, 412b, microphone pre-amp 406a, 406b gains and final amp 416 stage gain. This configurability provides the system 400 with the flexibility of a digital signal processing system, but with the power and performance of an analog signal processing system.

[0050] A further support block is the xtal amplifier 410. The xtal amplifier 410 is used by the Sampled Analog filters and allows the ODM to control how much power is used by the uplink noise cancellation system 400. Additionally, the xtal amplifier 410 allows the ODM to tune where the potential folded down products of the discrete analog output signal will be positioned. One property of the Sample Analog Technology is that the resulting filtered analog

signal is a non-continuous and discrete time analog signal. The discrete time analog signal contains “steps” with a frequency that matches the frequency of the xtal frequency. Depending on the ADC that samples the discrete analog signal, the broadband overtones caused by the discrete steps can fold down into the audio domain. To compensate for this, the ODM can choose to pick a xtal frequency that is high enough that any overtones that fold down do so outside the audible frequency band.

[0051] Low power noise cancellation method

[0052] FIGURE 5 is a diagram illustrating a low power noise cancellation method 500, according to some embodiments of the disclosure. At step 502, a signal is received at first and second microphones. At step 504, the received signals are amplified at first and second pre-amplifiers. At step 506, the amplified signals are combined at a beam forming circuit. In some embodiments. The method can include receiving, by first and second pre-amplifiers, a first and second received signals from first and second microphones for sensing audio, and amplifying, by a beam forming circuitry, first and second received signals, combining first and second amplified signals, and outputting a beam formed signal. The method operates on less than one milliwatt of power, and may include providing, by a localized bias, power to the first and second microphones.

[0053] The received and amplified signals are analog signals, and the beam-forming unit processes and combines the analog signals using analog circuitry based on Sampled Analog Technology. According to one implementation, the method 500 includes matching a gain of the first amplified signal with a gain of the second amplified signal at a microphone matching filter, and outputting by the microphone matching filter first and second matched signals.

[0054] In some embodiments, the method can further include receiving, by first and second steering filters, first and second matched signals, and outputting, by the first and second steering filters, first and second steered signals.

[0055] In some embodiments, the method can further include combining, by a summation circuit, the first and second steered signals, and outputting, by the summation circuit, a beam formed signal.

[0056] According to another implementation, the combined first and second matched signals are an output signal, and further comprising adjusting the output signal at an amplifier. For instance, the beam formed signal from the summation circuit can be amplified by an amplifier.

[0057] According to a further implementation, the method 500 includes removing wind noise from the received signals by a wind noise detection circuit and a wind noise removal circuit.

[0058] Ambient noise cancellation

[0059] When three or more microphones can be powered within the low power budget, beam forming can be applied to two or more microphones, while one microphone can be provide another function, e.g., measure environmental or ambient noise. For instance, a low power system can include a first and second microphones are used for beam forming to improve audio quality, and a third microphone is positioned away from the desired source (e.g., ear bud, head band, etc.). The third microphone can be used to provide additional improvement on audio quality by detecting or sensing ambient noise, i.e., undesired audio. The audio signal from the third microphone (or a filtered version thereof) can be attenuated, e.g., with an amplifier or suitable analog circuitry, since the audio signal from the third microphone can include the desired signal. Attenuation, or some form of filtering, can remove or reduce part of the desired signal in the audio signal from the third microphone. The attenuated signal can be subtracted from the signal generated from the first and second microphones (e.g., the beam formed signal), effectively providing an ambient noise removal circuit, for additional audio quality improvement.

[0060] Advanced wind noise detector and wind noise cancellation

[0061] In a low power system, beam forming can be applied to two microphones, and a third microphone can assist in detecting wind noise. The third microphone can be away from the desired source (e.g., away from the mouth). Wind noise usually an uncorrelated noise source, meaning it affects one microphone very differently from another microphone in a different position when wind is present. Based on this assumption, it is possible to find correlation between (1) either the first microphone and/or the second microphone, and (2)

the third microphone. Due to the positioning of the third microphone being located away from the first and/or second microphone, there is a delay path between (1) and (2). To determine correlation, either the signal from (1) or the signal from (2) is delayed so there is a relative delay between the two. Then, it is possible to assess whether there is correlation between one signal and the delayed signal. If there is correlation, it is likely that speech is present. If there is no correlation, it is likely wind noise is present. If there is no correlation, one may attenuate the signal being transmitted back to the telephone to dampen the signal and noise therein.

[0062] In some embodiments, the wind noise detector circuit performs cross correlation between microphone(s) near mouth (sensing desired source) and microphone near ear (sensing undesired source). The microphone near the ear can have very low quality, since it is used for cross correlation and wind noise sensing. The cross correlation may involve applying a delay for one of the signals to time wise align the two signals, since there is acoustic path delay between the two. Then, the cross correlation can correlate to see if speech is present in both sources. For instance, it is possible to check how well they correlate in some narrow band frequency (e.g., speech spectrum). A filter can be applied to focus on the narrow band frequency, and the cross correlation can assess how well the energy content in the narrow band frequency correlate between the two. If there is no correlation, the beam formed signal can be attenuated, e.g., with amplifier, before it is transmitted to the telephone. If there is correlation, no attenuation on the beam formed signal is needed.

[0063] FIGURE 6 is a diagram illustrating a low power noise cancellation system of FIGURE 4 with wind noise detection and/or cancellation enhancements, according to some embodiments of the disclosure. The low power noise cancellation system includes many parts seen in FIGURE 4, but further includes a wind noise detection circuit 602. The wind noise detection circuit can use two of the following: the microphone signals (e.g. first and second matched signals (outputs of the microphone matching filter 408), the beam formed signal, a signal generated by a third microphone 612. Accordingly, the low power system 400 may include a wind noise detection circuit (e.g., wind noise detection circuit 602) for detecting wind noise based on two of the following as first and second input signals: the first matched signal, the second match signal, the beam formed signal, and a third received signal from a third microphone. Other suitable pairs of signals may be used for wind noise detection.

[0064] The wind noise detection circuit 602 may include a first low pass filter (e.g., LPF 606) for filtering the first input signal and outputting a first filtered signal, a second low pass filter (e.g., LPF 608) for filtering the second input signal and outputting a second filtered signal. The first and second low pass filters helps narrow the band of interest to a limited frequency band, since wind noise is a low frequency noise, and speech in a low frequency band. The wind noise detection circuit 602 can further include cross correlation detector (e.g., X-COR 610) for correlating energy of the first filtered signal and energy of a delayed version of the second filtered signal. If correlation is detected, wind noise is assumed to be absent. If no correlation is detected, wind noise is assumed to be present, and one or more actions can be taken in response to detecting wind noise.

[0065] In some embodiments, the signal being transmitted back to the telephone (or some other device on the receiving end) is attenuated, e.g., with an amplifier, in response to detecting wind noise.

[0066] In some embodiments, if wind noise is detected in a particular signal, the signal can be used for wind noise removal by subtracting the particular signal (or an attenuated version of the particular signal, or an upsampled version of the particular signal) from the audio signal being transmitted back to the telephone.

[0067] When wind noise is present, the microphone matching filter 408 having dynamic gain adjustment can be negatively affected by the wind noise (or some other uncorrelated noise source for that matter), leading to improper microphone matching. In some embodiments, the cross correlation detector (e.g., X-COR 610) signals to the microphone matching filter to halt matching in response to detecting uncorrelated energy. Advantageously, the wind noise detector can ensure the microphone matching filter would not improperly match the microphones in the presence of uncorrelated noise. The microphone matching filter can remain “off” until wind noise is no longer detected.

[0068] In some embodiments, the low power system further comprises a wind noise removal circuit for removing wind noise in response to detecting wind noise. The wind noise removal circuit can include a high pass filter having a dynamic cut-off frequency (e.g., filters 604 and 614). The cut-off frequency can be controlled dynamically with a control loop. The high pass filter can filter out low frequencies of the first and/or second amplified signals. Depending on the implementation, the high pass filter can be provided to filter any one or

more of the following: the first and/or second matched signals, the first and/or second steered signals, the beam formed signal, and the amplified beam formed signal. Since wind noise is a low frequency noise, the high pass filter cut-off frequency can be moved up to reduce wind noise. By providing a control loop, the wind noise removal circuit can slowly move the cut-off frequency back down when wind noise is longer present. A leaky decay, e.g., implemented with a relaxation time constant, can be provided in the control loop. Accordingly, the wind noise removal circuit can dynamically adjusting the cut-off frequency of the high pass filter based on an output of the wind noise detection circuit (and a suitable time constant).

[0069] FIGURE 7 is a diagram illustrating a low power noise cancellation method 700 having wind noise detection and/or cancellation, according to some embodiments of the disclosure. In task 702, a wind noise detection circuit, cross correlate two signals (e.g., time-wise aligned based on the delay of an acoustic path between two microphone signals) to detect whether wind noise is present. In check 704, the wind noise detection circuit determines if wind noise is present based on the correlation. In task 706, if wind noise is detected, the microphone matching filter is turned off. In task 708, if wind noise is detected, a wind noise removal circuit can be signaled, activated, or triggered to remove wind noise. If wind noise is not detected, the method returns to task 702.

[0070] For purposes of finding correlation between two audio signals, the quality of the microphone or the audio signal being used can be very low, since the wind noise detection circuit is concerned primarily on the energy information in the signal. Accordingly, audio fidelity is not an important requirement. This means the microphone can be starved to receive as little power as possible, and/or the microphone can be of very low quality. In some embodiments, the audio signal can be of a very low resolution (e.g., if a low power analog-to-digital converter, such as an 8-bit successive approximation register analog-to-digital converter is included in the integrated circuit). Even a very low resolution, slowly sampled audio signal is sufficient for correlation. If one of the audio signal is to be used for subtraction to remove the wind noise later on, the audio signal may need to be of a better quality so as to not introduce additional noise in the signal path.

[0071] Programming the integrated circuit

[0072] As mentioned previously in FIGURE 4, the integrated circuit can be programmed using a filter coefficient fetcher 420 (e.g., “METAL PROGRAMMABLE DEFAULT CONFIGURATION + I2C SELFBOOT”), and a memory block 422 (e.g., “I2C EEPROM”). Many of the filters can be programmed, even the analog circuitry. For instance, the steering filters (sub-sample phase delay finite impulse response filters) of the beam forming circuitry can be has a set of programmable coefficients, so that when the delay is to be changed, the coefficients can be updated to implement a different delay. Other parts or settings of the system can also be programmed/configured, e.g., (pre-amp) gain values, bias, etc. The physical configuration and positioning of the microphones, types of microphones, the position of the desired source with respect to the microphones, the number of microphones, and so on, are examples of factors which may prompt different sets of coefficients and settings to be used. When headsets are mass manufactured for a variety of configurations, it is beneficial to have a single integrated circuit which can be programmed and/or tuned to work with a wide variety of configurations. Advantageously, the same chip can be used for all kinds of headsets.

[0073] To provide configurability, the integrated circuit can include a mechanism, e.g., a filter coefficient fetcher and a memory, which can allow coefficients to be set for a given configuration. The mechanism can utilize the microphone wire as a communication channel (e.g., as a single wire communication channel with respect to ground). Accordingly, the mechanism can receive commands for setting the coefficients, commands for selecting a set of coefficients, and/or values for setting the coefficients. At product definition time, the proper coefficients can be set so that the integrated circuit is programmed for the product. The coefficients can be burned into memory, e.g., memory block 422 of FIGURE 4 and 6, and one time programmable memory and/or non-volatile memory, and usable by the integrated circuit for low power noise cancellation (e.g., beam forming, wind noise cancellation, etc.). The memory can include memory internally provided in the integrated circuit; in some cases, the memory can include memory external to the integrated circuit. If desired, the coefficients can be programmed into shadow registers in volatile memory during the use of the integrated circuit to fine tune the coefficients being used by the integrated circuit.

[0074] FIGURE 8 is a diagram illustrating a method 800 for programming the low power noise cancellation system, according to some embodiments of the disclosure. In task 802, a filter coefficient fetcher can detect if the integrated circuit is in an initialization state (e.g., a state where coefficients have not been programmed yet or coefficients need to be programmed). The detection task may be skipped, or may be implicit, if circuitry in the integrated circuit (e.g., switch circuitry which changes behavior based on the initialization state) is set in such a way to allow external access or signals to be transmitted to the integrated circuit via one or more pins/inputs (as opposed to the circuitry being set in such a way that the one or more pins/inputs are in a high impedance state).

[0075] In task 804, when the integrated circuit is in an initialization state, the filter coefficient fetcher can receive coefficients over a communication channel, e.g., provided by one or more wires connecting pins/inputs of the integrated circuit to a connector/jack. The one or more wires can include the microphone wire, left audio wire, right audio wire, ground/bias wire, or any suitable wire/conductor or combination of wires/conductors allowed by the jack (such as a 3.5 mm jack). Alternatively, the filter coefficient fetcher can receive information for selecting a set of pre-programmed coefficients over the communication channel.

[0076] In task 806, the integrated circuit can write the coefficients into memory. Alternatively, the filter coefficient fetcher can (e.g., burn fuses) to select a set of pre-programmed coefficients.

[0077] In task 808, the filter coefficient fetcher can update the initialization state to a state which indicates coefficients have been programmed or set. At the next power up of the system, the filter coefficient fetcher would find the initialization state to be set to indicate that the coefficients have been programmed already and does not initiate a method to program the integrated circuit nor allow the integrated circuit to be programmed.

[0078] As illustrated by the method of FIGURE 8, the programming can be carried out only once. In some other cases, the programming can be carried two or three or more times if desired (e.g., if testing can be performed to test different sets of coefficients and using the best set of coefficients). In some cases, the filter coefficient fetcher can listen periodically, or at certain predetermined times, for information over the communication channel, to allow further programming of the coefficients can be carried out. Generally speaking, the filter

coefficient fetcher can be a coefficient fetcher for fetching coefficients usable for any one or more components of the low power uplink noise cancellation system, even for coefficients which are not associated with a filter.

[0079] In one implementation, the integrated circuit implementing, e.g., the functionalities of beam forming and so forth, can have a filter coefficient fetcher that would allow the integrated circuit to start initially in a programmable state using one or more analog signals, e.g., communications signals. After programming, the integrated circuit would then start up as a (purely) analog device.

[0080] A programming scheme as such allows manufacturers to program the device through, e.g., a 3.5mm jack after full manufacture (even after the integrated circuit is embedded/encased in the headset). In some embodiments, the integrated circuit can be configured, during the initialization state, to listen over a communication channel provided over, e.g., a 3.5 mm jack, for programming commands and/or coefficient information. Typically, the jack is a TRRS (short for “Tip, Ring 1, Ring 2, and Sleeve) connector, a four-conductor jack. A TRRS connector has four conductors: Left audio, Right audio, BIAS/Microphone, and Ground (GND), and the four conductors conduct signals over “wires” which are connected to four respective pins/inputs of the integrated circuit. Several configurations are possible for implementing such communication channel, since different combinations of conductors are available for carrying data from a programming device to the integrated circuit.

[0081] In one implementation, the Left audio and Right audio wires can be used for programming, or tuning of coefficients for the integrated circuit. Such implementation allows for an analog feedback path via the Ground/BIAS lines. After tuning/final programming, the integrated circuit, having switch circuitry, can switch the programming pins (pins connected to Left audio and Right audio wires) into high impedance input pins. The advantage of using the Left audio and Right audio wires is the ability to use the standard I²C (short for Inter IC) protocol, which is readily supported by many electronic devices. Accordingly, it would be easy for programming device to communicate via the jack using I²C. Furthermore, I²C only requires that the integrated circuit to provide the pull down, while the pull-ups can be provided in the programming device. The supply reference can be the DC BIAS level used to power integrated circuit.

[0082] In one example, the initial state of the device comprises the following:

- Left audio == SCL (short for Serial Clock Line)
- Right audio == SDA (short for Serial Data Line)
- GND == GND
- BIAS == BIAS/Mic

[0083] In this initial state, the integrated circuit provides full access to either internal, and/or external non-volatile memories (NVM). For an external NVM, the I²C signals may pass through the integrated circuit to program the external NVM, which can be achieved via an on-chip switch circuit in the integrated circuit.

[0084] For implementing a communication channel that can be used for programming/tuning of the integrated circuit during the initialization state, the chip can include the following switch circuitry for routing signals over the (programming) pins to circuitry which processes I²C signals. The switch circuitry can include a switch between the Left audio pin/input to the pin/input of I²C circuitry (within the integrated circuit), and a switch between the Right audio pin/input to the pin/input of I²C circuitry (within the integrated circuit):

- Left audio → switch → I²C slave clock pin
- Right audio → switch → I²C slave data pin

[0085] The switch can be implemented as either an analog switch circuit or as a high impedance driver. Preferably, the switch comprises an analog switch circuit, since an analog switch circuit would provide the best isolation. Typically, the programming pins, no matter which pins are chosen, needs to be capable of high voltage isolation, because the speaker drivers in telephones are at least 5 V, or sometimes boosted to 8-9 V. The switch is preferably capable of withstanding these voltages without engaging ESD (short for Electrostatic Discharge) protection circuits. With an analog switch circuit, this problem is considerably easier to solve when compared to a high impedance driver design.

[0086] After programming has been finalized, the filter coefficients fetcher can set the initialization (or boot) state of the switch to indicate the integrated circuit is no longer in the initialization state, e.g., set the switches of the switch circuitry to open. Accordingly, the circuitry prevents further programming of the integrated circuit when the switches is open

(indicating the integrated circuit is no longer in the initialization state by means of the pins/inputs being in a high impedance state).

[0087] For external NVM support, a further switch can be added to short Left audio pin/input and Right audio pin/input to an external SCL and SDA port.

[0088] For debugging, the circuitry can further include one or more switches which controllable by an (optional) external override signal.

[0089] Other implementations and applications

[0090] Many other variations and implementations of the system 400 are possible. In one example, the system 400 can include more microphones. Adding more microphones narrows the beam and can improve focus of the beam. In another example, the system 400 can be used as a null focus system to remove a source of noise. A null focus system includes another level of adaption, as the beam has to search for the highest noise source and remove it. In a further example, the system 400 can be adapted as multi-beam forming system. In particular, adding multiple steering filters allows more than one beam to be active. Having more than one active beam allows for an adaptive search for the loudest speaker and/or noise, and the focus beam can be steered accordingly. For example, the focus beam can be steered to enhance a speaker signal or cancel noise.

[0091] In one implementation, one or more additional beams can be added to the system 400, and the system 400 can be used as a handsfree beam former or radar. By adding a minimum of a second beam, the system 400 can have one beam locked on an active speaker while another beam searches for another speaker. An identification algorithm can use an added LMS/RMS circuit to determine whether the current focus beam is louder than the scanned source.

[0092] In one example, the system 400 can be used for a hands-free voice search system. The hands-free voice searcher can be further extended by added support for formant finding. Formant finding can be done by filtering the beam into a number of formant bands and determining vowel activity to determine whether a given sound source is speech or noise.

[0093] In another example, the system 400 can be used for automotive null focus. In a car cabin, the location of speakers is often known and beam forming is currently only used to focus on the one or more speakers. However, a major challenge in automotive applications

is the multiple sources of noise, such as engine noise, wind-noise, and road noise. By utilizing dynamic null steering, these noise sources can be dynamically removed, depending on speed, weather and road conditions.

[0094] In one implementation, the system 400 can be used for ultrasound. The same rules that cover audible sound also apply to ultrasound. Ultrasound uses echolocation for detection and imaging. Ultrasound machines utilize arrays of microphones and speakers to perform excitation and recording of echoes from the item being investigated. Ultrasound analysis is performed in 2MHz-18MHz range making it impractical to perform AD conversion in the transducer as the data rates can become unwieldy. Thus, focusing in ultrasound has been limited to mechanical focusing by moving speakers and/or microphones. An analog beam former as implemented in the system 400 allows receiver based focusing while retaining the properties of an analog system.

[0095] In one example, an ultrasound machine uses an array of 128 microphones. Low-powered Sampled Analog beam forming can be added in front of each microphone to focus pairs or groups of microphones on an area of interest. According to one example, the area of focus is cone-shaped. The information from the array of microphones can be compressed using beam forming technology to make it appear like a virtual microphone in an analog domain. In various applications, null-steering or beam-steering can be used to track or focus on something in an ultrasound image. Further details on using the system 400 for ultrasound are included in Tab A below.

[0096] In another example, the system 400 can be used as a sound scanner. As the low power analog solution of system 400 is easily integratable into a home security system, it can be used to scan for sources of sound. The advantage of a scanner vs. an omnidirectional microphone is that a sound scanner using the system 400 can actively ignore things like fridges that will occasionally fire off sound in a given home environment. In further examples, the system 400 can be used as a sound scanner in other environments, such as to factories, in which it can actively ignore machine noise.

[0097] Additionally, according to various applications, although a delay sum beam forming algorithm is shown in the system 400, any beam forming algorithm may be used in accordance with a low power analog system for noise cancellation in received signals.

[0098] The technology described herein can be applicable where beam forming or any audio quality improvement is to be provided with a (limited, ultra) low power budget. Generally speaking, applications include mobile devices, electronic devices, and/or computing devices where the power being supplied via the microphone/audio/headset jack is minimal. Other applications include where headsets are used and where improved audio quality is desired. These applications include home audio systems, audio/video systems, home theater systems, professional audio equipment, gaming systems, call center audio equipment, telephone systems, communication systems having headsets, wireless headsets (Bluetooth headsets), wired headsets, etc.

[0099] Variations and implementations

[0100] In the discussions of the embodiments above, amplifiers, switches, digital core, transistors, capacitors, clocks, DFFs, dividers, inductors, resistors and/or other components can readily be replaced, substituted, or otherwise modified in order to accommodate particular circuitry needs. Moreover, it should be noted that the use of complementary electronic devices, hardware, software, etc. offer an equally viable option for implementing the teachings of the present disclosure.

[0101] In another example embodiment, the electrical circuits of the FIGURES may be implemented as stand-alone modules (e.g., a device with associated components and circuitry configured to perform a specific application or function) or implemented as plug-in modules into application specific hardware of electronic devices. Note that particular embodiments of the present disclosure may be readily included in a system on chip (SOC) package, either in part, or in whole. An SOC represents an IC that integrates components of a computer or other electronic system into a single chip. It may contain digital, analog, mixed-signal, and often radio frequency functions: all of which may be provided on a single chip substrate. Other embodiments may include a multi-chip-module (MCM), with a plurality of separate ICs located within a single electronic package and configured to interact closely with each other through the electronic package. In various other embodiments, the amplification functionalities may be implemented in one or more silicon cores in Application Specific Integrated Circuits (ASICs), Field Programmable Gate Arrays (FPGAs), and other semiconductor chips.

[0102] It is also imperative to note that all of the specifications, dimensions, and relationships outlined herein (e.g., the number of processors, logic operations, etc.) have only been offered for purposes of example and teaching only. Such information may be varied considerably without departing from the spirit of the present disclosure, or the scope of the appended claims. The specifications apply only to one non-limiting example and, accordingly, they should be construed as such. In the foregoing description, example embodiments have been described with reference to particular processor and/or component arrangements. Various modifications and changes may be made to such embodiments without departing from the scope of the appended claims. The description and drawings are, accordingly, to be regarded in an illustrative rather than in a restrictive sense.

[0103] Note that the activities discussed above with reference to the FIGURES are applicable to any integrated circuits that involve signal processing, particularly those that can execute specialized software programs, or carry out algorithms, some of which may be associated with processing digitized real-time data. Preferably, the activities discussed above with reference to the FIGURES are applicable to analog circuits that processes analog signals with Sampled Analog Technology. Sampled Analog Technology may be used as an alternative to, or in addition to, digital signal processing.

[0104] Note that with the numerous examples provided herein, interaction may be described in terms of two, three, four, or more electrical components. However, this has been done for purposes of clarity and example only. It should be appreciated that the system can be consolidated in any suitable manner. Along similar design alternatives, any of the illustrated components, modules, and elements of the FIGURES may be combined in various possible configurations, all of which are clearly within the broad scope of this Specification. In certain cases, it may be easier to describe one or more of the functionalities of a given set of flows by only referencing a limited number of electrical elements. It should be appreciated that the electrical circuits of the FIGURES and its teachings are readily scalable and can accommodate a large number of components, as well as more complicated/sophisticated arrangements and configurations. Accordingly, the examples provided should not limit the scope or inhibit the broad teachings of the electrical circuits as potentially applied to a myriad of other architectures.

[0105] Note that in this Specification, references to various features (e.g., elements, structures, modules, components, steps, operations, characteristics, etc.) included in “one embodiment”, “example embodiment”, “an embodiment”, “another embodiment”, “some embodiments”, “various embodiments”, “other embodiments”, “alternative embodiment”, and the like are intended to mean that any such features are included in one or more embodiments of the present disclosure, but may or may not necessarily be combined in the same embodiments.

[0106] It is also important to note that the functions related to microphones, beam forming circuits and transducers, such as ultrasound transducers, illustrate only some of the possible microphone and beam forming circuit functions that may be executed by, or within, systems illustrated in the FIGURES. Some of these operations may be deleted or removed where appropriate, or these operations may be modified or changed considerably without departing from the scope of the present disclosure. In addition, the timing of these operations may be altered considerably. The preceding operational flows have been offered for purposes of example and discussion. Substantial flexibility is provided by embodiments described herein in that any suitable arrangements, chronologies, configurations, and timing mechanisms may be provided without departing from the teachings of the present disclosure.

[0107] Numerous other changes, substitutions, variations, alterations, and modifications may be ascertained to one skilled in the art and it is intended that the present disclosure encompass all such changes, substitutions, variations, alterations, and modifications as falling within the scope of the appended claims. Note that all optional features of the apparatus described above may also be implemented with respect to the method or process described herein and specifics in the examples may be used anywhere in one or more embodiments.

[0108] Example 1. A system for amplifying a signal, comprising: first and second microphones for receiving the signal and outputting a first received signal and a second received signal, first and second pre-amplifiers for amplifying the first and second received signals, and a beam forming circuitry for combining the first and second amplified signals, wherein the system operates on less than one milliwatt of power.

[0109] Example 2. The system of example 1, wherein the power is supplied by a telephone system.

[0110] Example 3. The system of example 1, wherein each of the first and second microphones consumes less than 400 microwatts of power.

[0111] Example 4. The system of example 1, further comprising a localized bias for providing the power to the first and second microphones.

[0112] Example 5. The system of example 1, wherein the beam forming circuitry includes: first and second steering filters for receiving the first and second matched signals and outputting first and second steered signals, and a summation circuit for combining the first and second steered signals.

[0113] Example 6. The system of example 5, wherein the steering filters are sub-sample phase delay finite impulse response filters.

[0114] Example 7. The system of example 5, wherein the summation circuit outputs a beam formed signal, and further comprising an amplifier for amplifying the beam formed signal.

[0115] Example 8. The system of example 1, wherein the first and second amplified signals are analog signals, and the beam forming circuitry uses sampled analog technology to combine the analog signals.

[0116] Example 9. The system of example 1, further comprising a microphone matching filter for matching a gain of the first received signal with a gain of the second received signal and outputting first and second matched signals.

[0117] Example 10. The system of example 1, further comprising a wind noise detection circuit for detecting wind noise and a wind noise removal circuit for removing wind noise.

[0118] Example 11. The system of example 10, wherein the wind noise removal circuit is a high pass filter having a dynamic cut-off frequency.

[0119] Example 12. A method for amplifying a signal, comprising: receiving the signal at first and second microphones, amplifying first and second received signals at first and second pre-amplifiers, and combining first and second amplified signals at a beam forming circuitry, wherein the method operates on less than one milliwatt of power.

[0120] Example 13. The method of example 12, further comprising matching a gain of the first amplified signal with a gain of the second amplified signal at a microphone matching filter.

[0121] Example 14. The method of example 12, wherein the first and second amplified signals are analog signals, and combining the first and second amplified signals includes using sampled analog technology to combine the analog signals.

[0122] Example 15. The method of example 12, wherein the combined first and second matched signals are an output signal, and further comprising adjusting the output signal at an amplifier.

CLAIMS:

1. A low power system for improving audio quality, comprising:
first and second microphones for sensing audio and outputting a first and second received signal;
an integrated circuit comprising:
first and second pre-amplifiers for amplifying the first and second received signals and outputting first and second amplified signals; and
a beam forming circuit for combining the first and second amplified signals and outputting a beam formed signal;
wherein the low power system operates on less than one milliwatt of power.
2. The low power system of Claim 1, wherein the power is supplied by a telephone system.
3. The low power system of Claim 1 or 2, wherein each of the first and second microphones consumes less than 400 microwatts of power.
4. The low power system of any one of the above Claims, further comprising a localized bias for providing power to the first and second microphones.
5. The low power system of any one of the above Claims, wherein the beam forming circuit includes:
microphone matching filter for matching a gain of the first amplified signal with a gain of the second amplified signal and outputting first and second matched signals.
6. The low power system of Claim 5, wherein the beam forming circuit further includes:
first and second steering filters for (1) receiving first and second matched signals and (2) outputting first and second steered signals.

7. The low power system of Claim 6, wherein the first and second steering filters are sub-sample phase delay finite impulse response filters.

8. The low power system of Claim 6 or 7, wherein the beam forming circuit further includes:

a summation circuit for combining the first and second steered signals.

9. The low power system of Claim 8, wherein:
the summation circuit outputs the beam formed signal.

10. The low power system of Claim 9, further comprising
an amplifier for amplifying the beam formed signal.

11. The low power system of any one of the above Claims, wherein:
the first and second amplified signals are analog signals; and
the beam forming circuitry comprises analog circuitry for processing and combining the analog signals.

12. The low power system of any one of Claims 5-10, further comprising:
a wind noise detection circuit for detecting wind noise based on two of the following as first and second input signals: the first matched signal, the second match signals, the beam formed signal, and a third received signal from a third microphone.

13. The low power system of Claim 12, wherein the wind noise detection circuit comprises:

a first low pass filter for filtering the first input signal and outputting a first filtered signal;

a second low pass filter for filtering the second input signal and outputting a second filtered signal; and

a cross correlation detector for correlating energy of the first filtered signal and energy of a delayed version of the second filtered signal.

14. The low power system of Claim 12 or 13, wherein the cross correlation detector signals to the microphone matching filter to halt matching in response to detecting uncorrelated energy.

15. The low power system of any one of Claims 12-14, further comprising:
a wind noise removal circuit for removing wind noise in response to detecting wind noise.

16. The low power system of Claim 15, wherein:
the wind noise removal circuit comprises a high pass filter having a dynamic cut-off frequency.

17. A method for improving audio quality using a low power system, comprising:
receiving, by first and second pre-amplifiers, a first and second received signals from first and second microphones for sensing audio; and
amplifying, by a beam forming circuitry, first and second received signals, combining first and second amplified signals, and outputting a beam formed signal;
wherein the method operates on less than one milliwatt of power.

18. The method of claim 17, further comprising:
providing, by a localized bias, power to the first and second microphones.

19. The method of Claim 17, further comprising:
matching, by a microphone matching filter, a gain of the first amplified signal with a gain of the second amplified signal; and
outputting, by the microphone matching filter, first and second matched signals.

20. The method of Claim 19, further comprising:
receiving, by first and second steering filters, first and second matched signals; and
outputting, by the first and second steering filters, first and second steered signals.

21. The method of Claim 20, further comprising:
combining, by a summation circuit, the first and second steered signals; and
outputting, by the summation circuit, the beam formed signal.
22. The method of Claim 21, further comprising:
amplifying, by an amplifier, the beam formed signal.
23. The method of any one of Claims 17-22, wherein:
the first and second amplified signals are analog signals; and
combining the first and second amplified signals includes processing and combining
the analog signals using analog circuitry.
24. The method of any one of Claims 17-23, wherein the combined first and
second matched signals are an output signal, and the method further comprises adjusting the
output signal at an amplifier.
25. The method of any one of Claims 17-24, further comprising:
detecting wind noise, by a wind noise detection circuit, based on two of the following
as first and second input signals: the first matched signal, the second match signal, the beam
formed signal, and a third received signal from a third microphone.
26. The method of Claim 25, further comprising:
filtering, by a first low pass filter, the first input signal;
outputting, by the first low pass filter, a first filtered signal;
filtering, by a second low pass filter, filtering the second input signal;
outputting, by the second low pass filter, a second filtered signal;
correlating, by a cross correlation detector, energy of the first filtered signal and
energy of a delayed version of the second filtered signal.

27. The method of Claim 25 or 26, further comprising:
signaling, by the cross correlation detector, to the microphone matching filter to halt matching in response to detecting uncorrelated energy.
28. The method of any one of Claims 25-27, further comprising:
removing, by a wind noise removal circuit, wind noise in response to detecting wind noise.
29. The method of Claim 28, further comprising:
dynamically adjusting, by the wind noise removal circuit, a cut-off frequency of a high pass filter based on an output of the wind noise detection circuit.
30. An integrated circuit for improving audio quality, comprising:
first and second pre-amplifiers for amplifying the first and second received signals from first and second microphones configured to sense audio;
a beam forming circuit for combining the first and second amplified signals and outputting a beam formed signal; and
a localized bias for powering the first and second microphones;
wherein the integrated circuit and the first and second microphones operates on less than one milliwatt of power.
31. The integrated circuit of Claim 30, further comprising:
a filter coefficient fetcher for:
in response to the integrated circuit being in an initialization state, receive coefficients and/or information for selecting a set of pre-programmed coefficients over a communication channel provided over one or more of: microphone wire and speaker wire; and
writing the coefficients into a memory and/or burning fuses to select the set of pre-programmed coefficients.

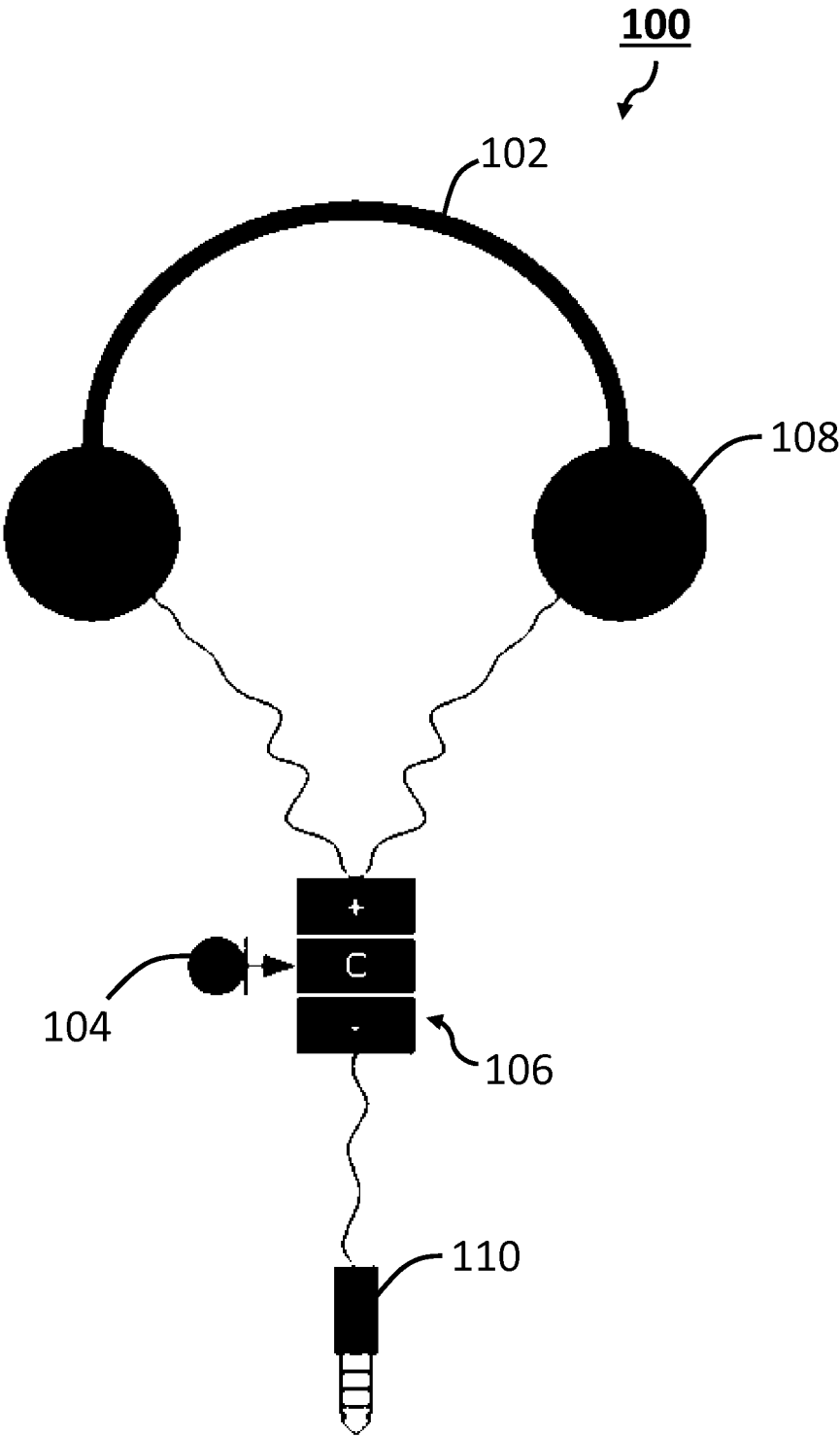


FIGURE 1

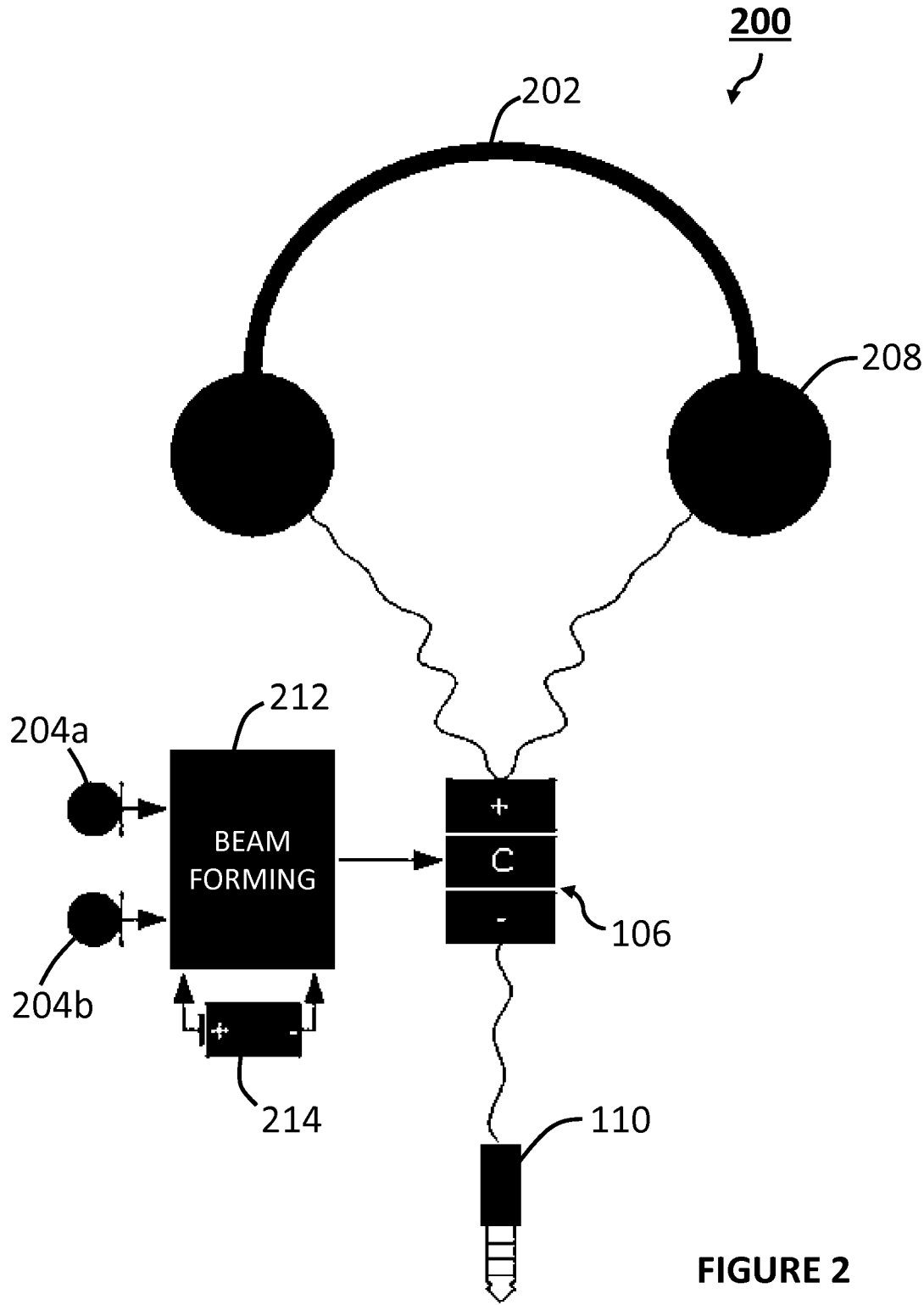


FIGURE 2

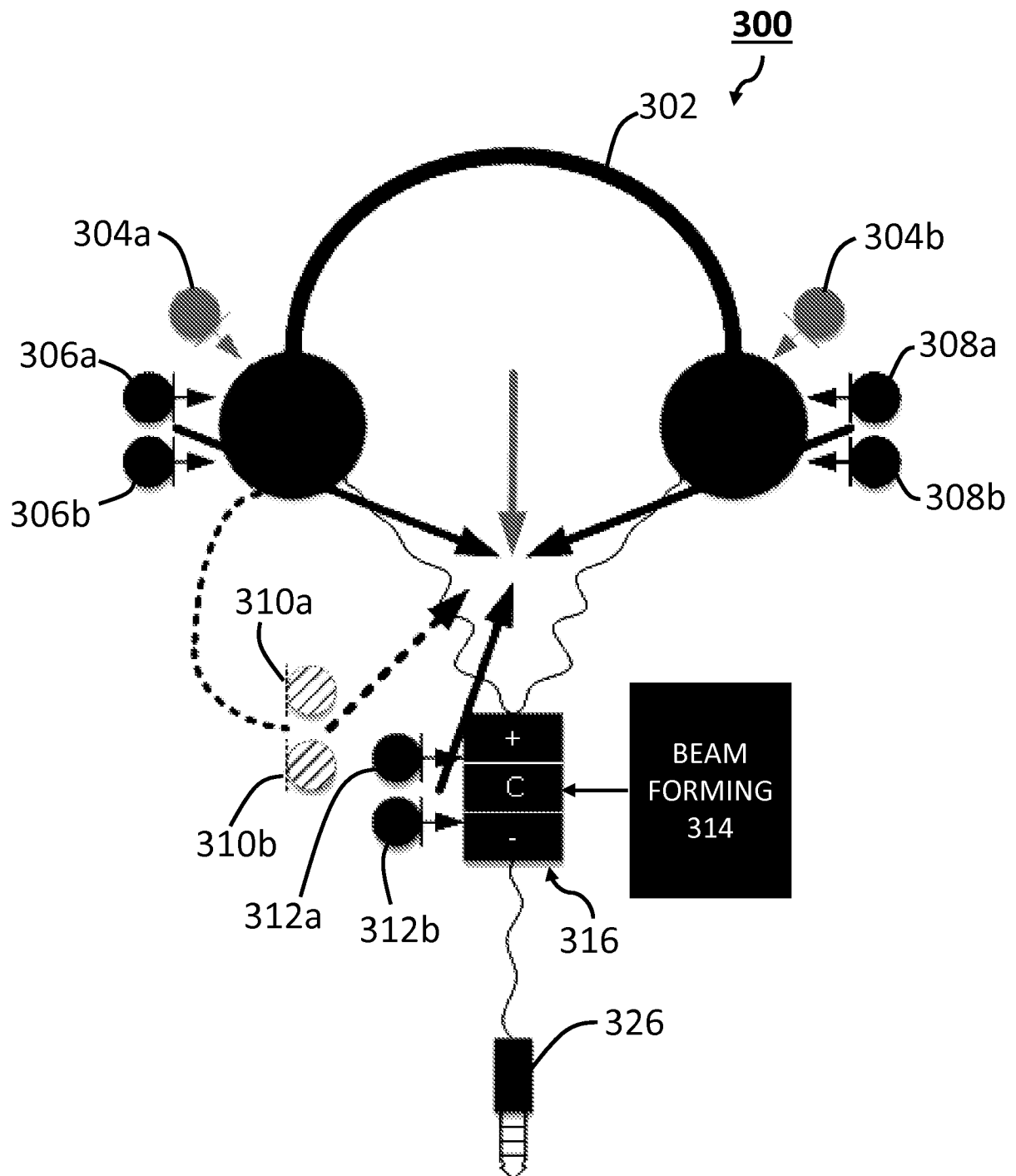
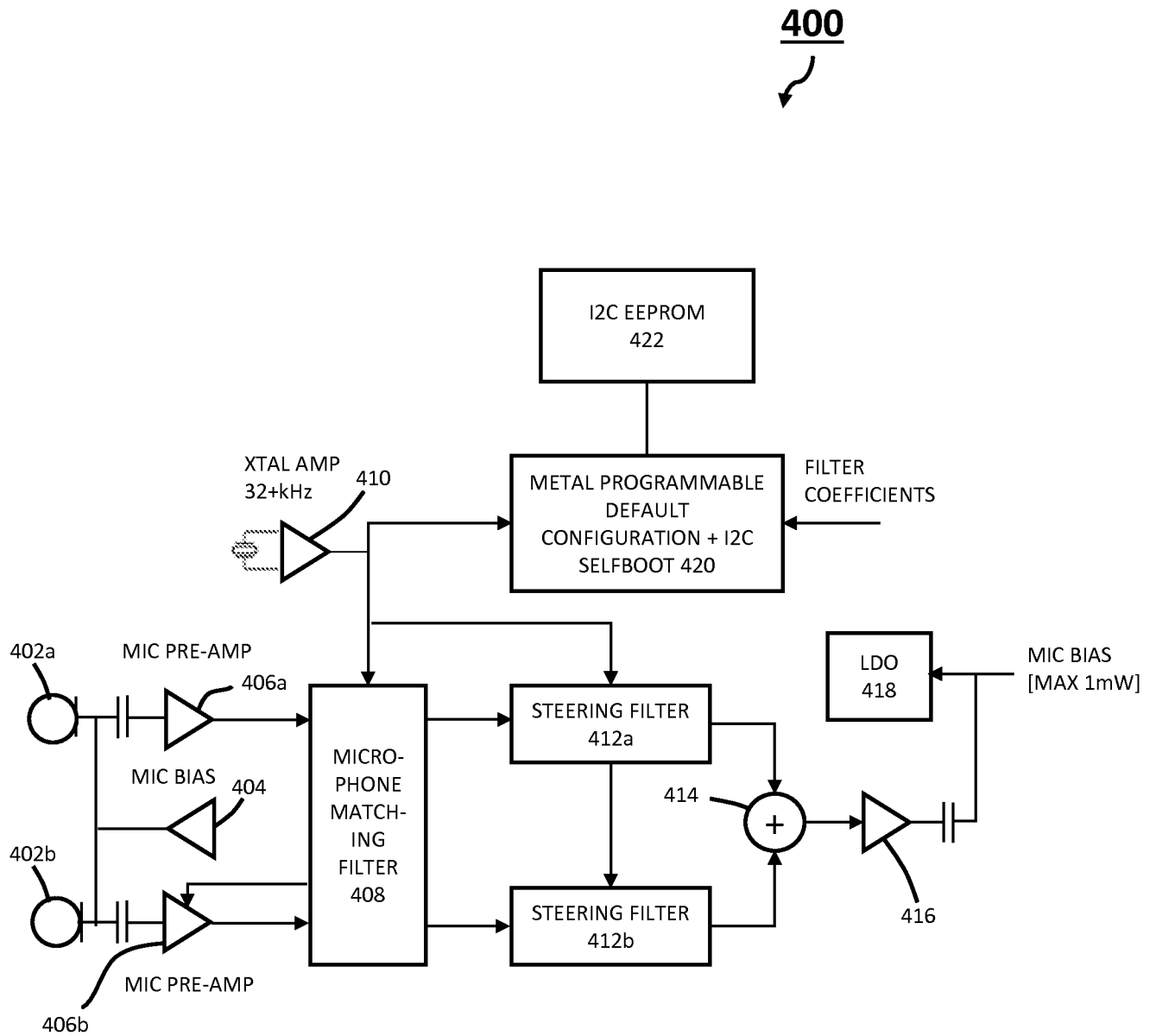
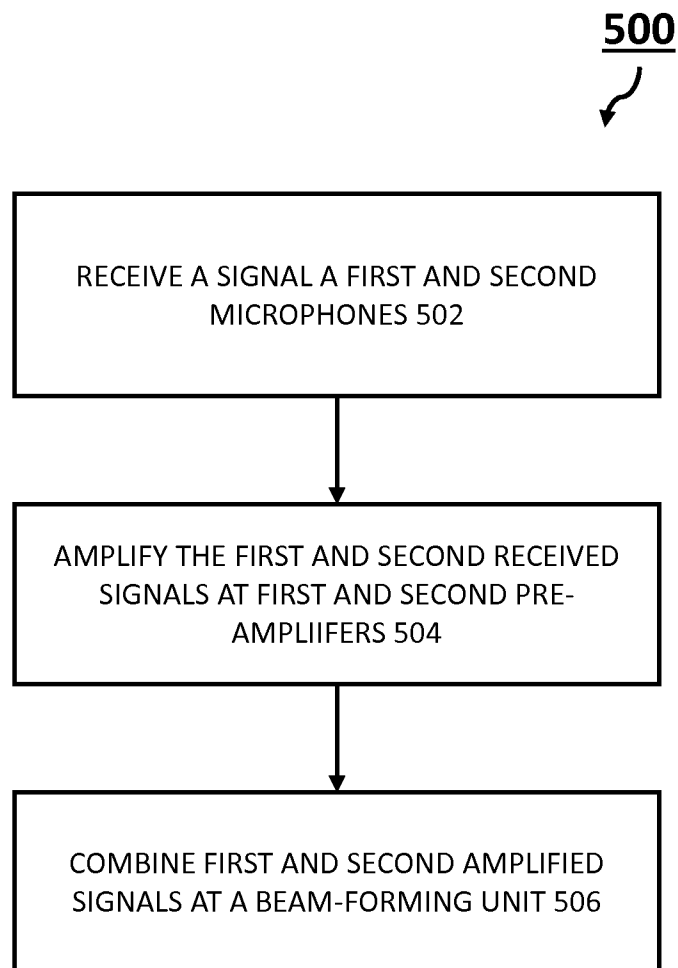
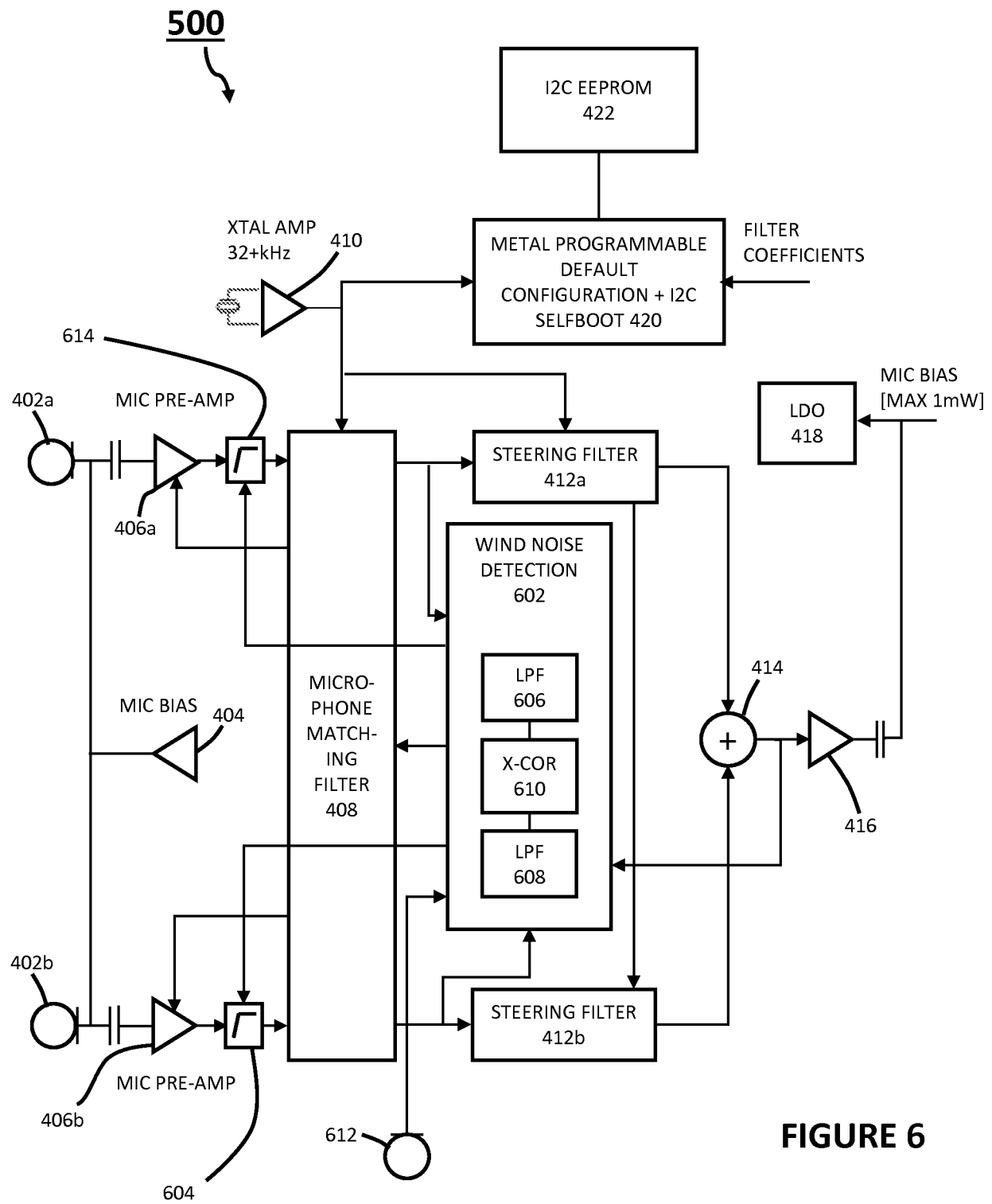


FIGURE 3

**FIGURE 4**

**FIGURE 5**

**FIGURE 6**

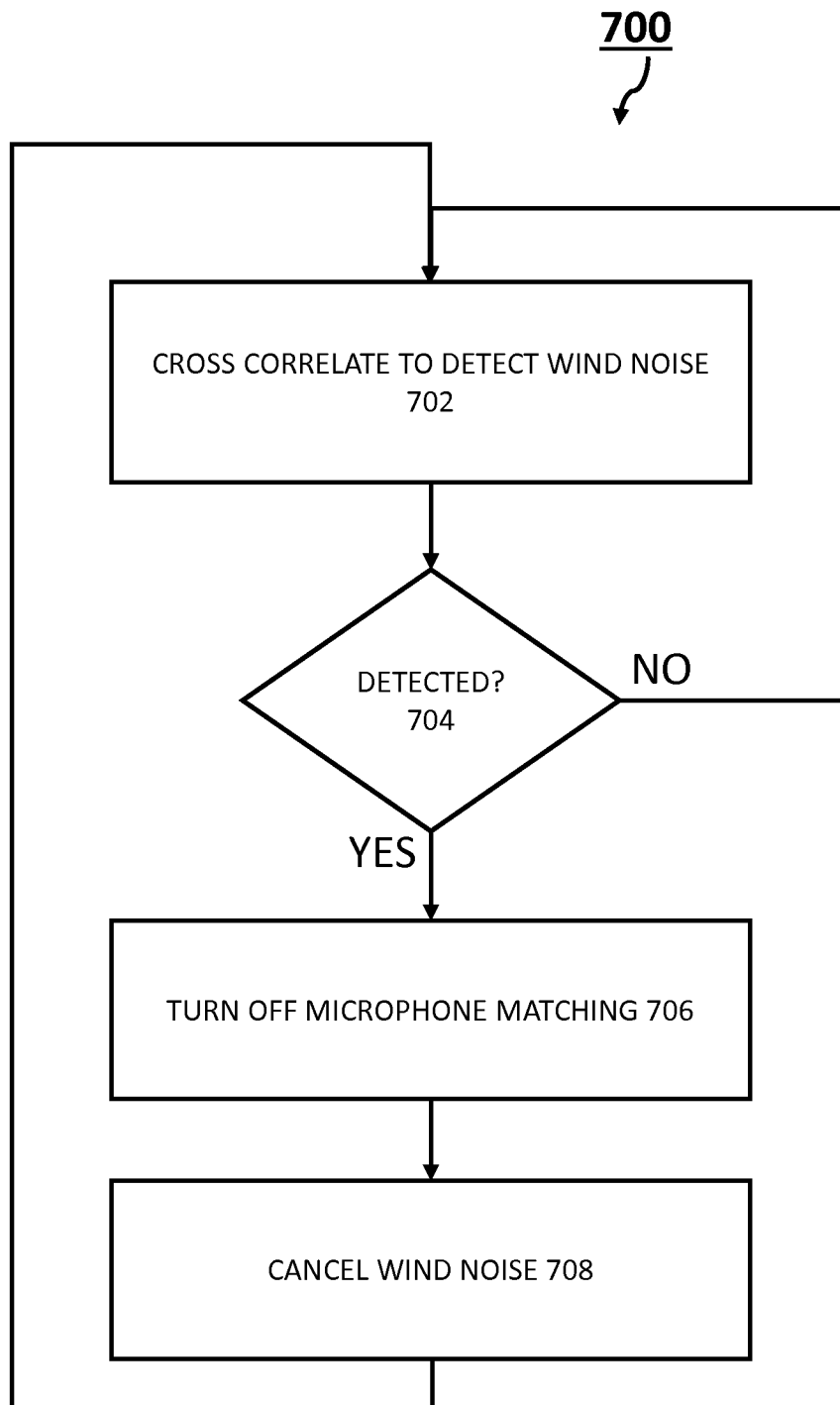
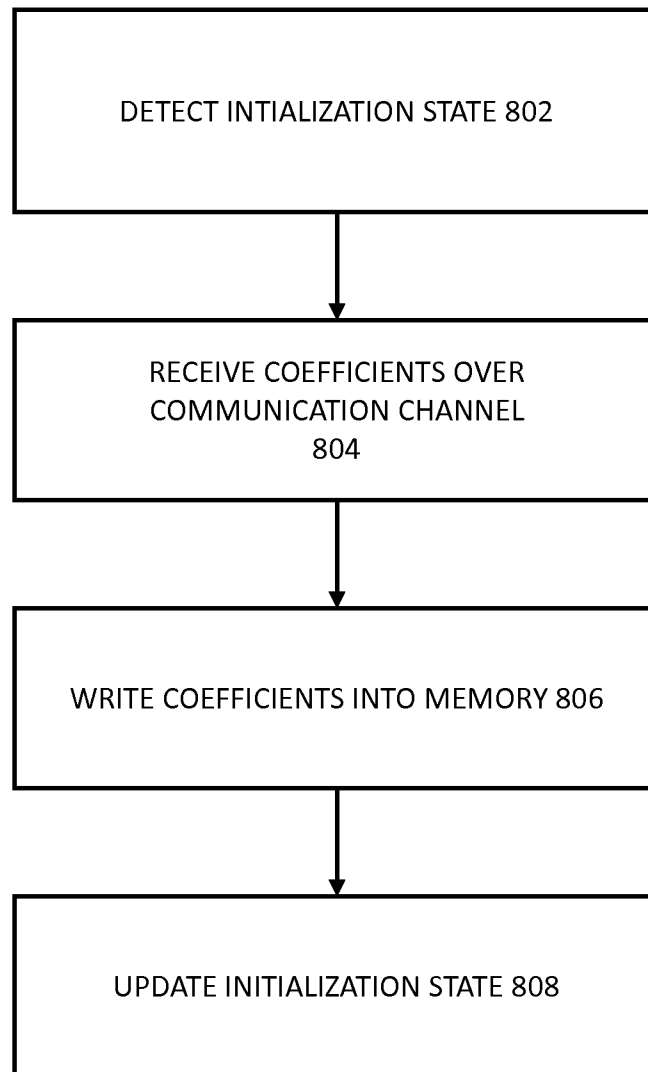


FIGURE 7

800**FIGURE 8**

INTERNATIONAL SEARCH REPORT

International application No.
PCT/US2015/039334**A. CLASSIFICATION OF SUBJECT MATTER****H04R 3/04(2006.01)i, H04R 1/10(2006.01)i**

According to International Patent Classification (IPC) or to both national classification and IPC

B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)

H04R 3/04; H04R 25/00; H04N 5/232; H04M 9/08; G06F 17/00; H04N 5/92; H04R 1/10

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched

Korean utility models and applications for utility models

Japanese utility models and applications for utility models

Electronic data base consulted during the international search (name of data base and, where practicable, search terms used)

eKOMPASS(KIPO internal) & Keywords: power, beam, forming, amplifier, microphone, filter, steer, summation, combination, analog, noise

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category*	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
Y	US 8184801 B1 (MATTI HAMALAINEN) 22 May 2012 See column 9, line 7 - column 12, line 25, column 13, line 12 - column 14, line 15 and figures 2-4.	1-31
Y	US 2007-0165875 A1 (BEHROOZ REZVANI et al.) 19 July 2007 See paragraphs [0015]-[0021] and figures 1-4.	1-31
A	US 2011-0144779 A1 (CORNELLI PIETRE JANSE et al.) 16 June 2011 See abstract, paragraphs [0099]-[0129] and figures 3-6.	1-31
A	US 2011-0044481 A1 (FRANCOIS MARQUIS et al.) 24 February 2011 See abstract, paragraphs [0043]-[0057] and figures 1-5.	1-31
A	KR 10-2014-0011614 A (MIGHTYWORKS) 29 January 2014 See abstract, paragraphs [0045]-[0080] and figures 1-4.	1-31



Further documents are listed in the continuation of Box C.



See patent family annex.

* Special categories of cited documents:

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"P" document published prior to the international filing date but later than the priority date claimed

"T" later document published after the international filing date or priority date and not in conflict with the application but cited to understand the principle or theory underlying the invention

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"&" document member of the same patent family

Date of the actual completion of the international search

16 October 2015 (16.10.2015)

Date of mailing of the international search report

16 October 2015 (16.10.2015)

Name and mailing address of the ISA/KR

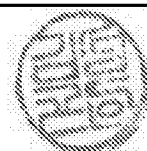
International Application Division
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Republic of Korea

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INTERNATIONAL SEARCH REPORT

Information on patent family members

International application No.

PCT/US2015/039334

Patent document cited in search report	Publication date	Patent family member(s)	Publication date
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