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(54) **HYBRID HORN MICROPHONE**

(71) Applicant: **Cisco Technology, Inc.**, San Jose, CA (US)

(72) Inventors: **Rune Skramstad**, Drammen (NO); **Haohai Sun**, Nesbru (NO)

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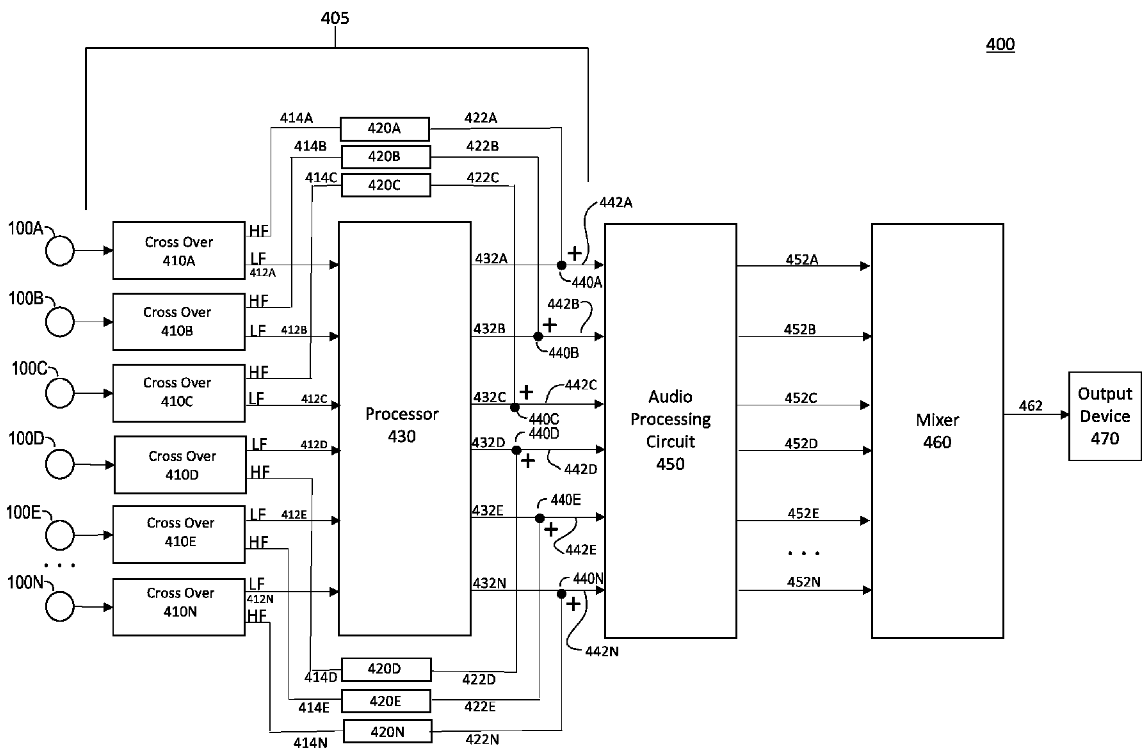
(57) **ABSTRACT**

The disclosed technology relates to a microphone array. The array comprises a plurality of microphones with each microphone having a horn portion. Each microphone of the array further comprises an instrument disposed at a distal end of the horn portion. Each instrument of the array is configured to convert sound waves into an electrical signal. The microphone array further comprises a beamforming signal processing circuit electrically coupled to each instrument and configured to create a plurality of beam signals based on respective electrical signals.



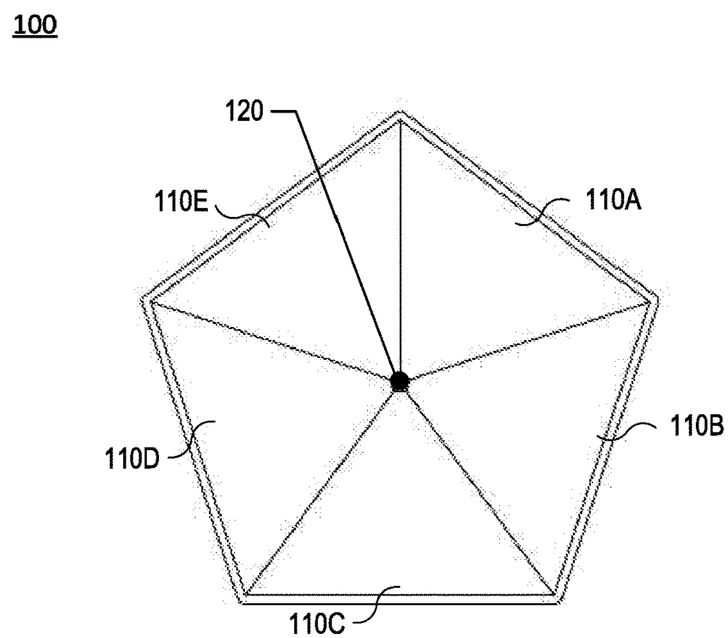


FIG. 1

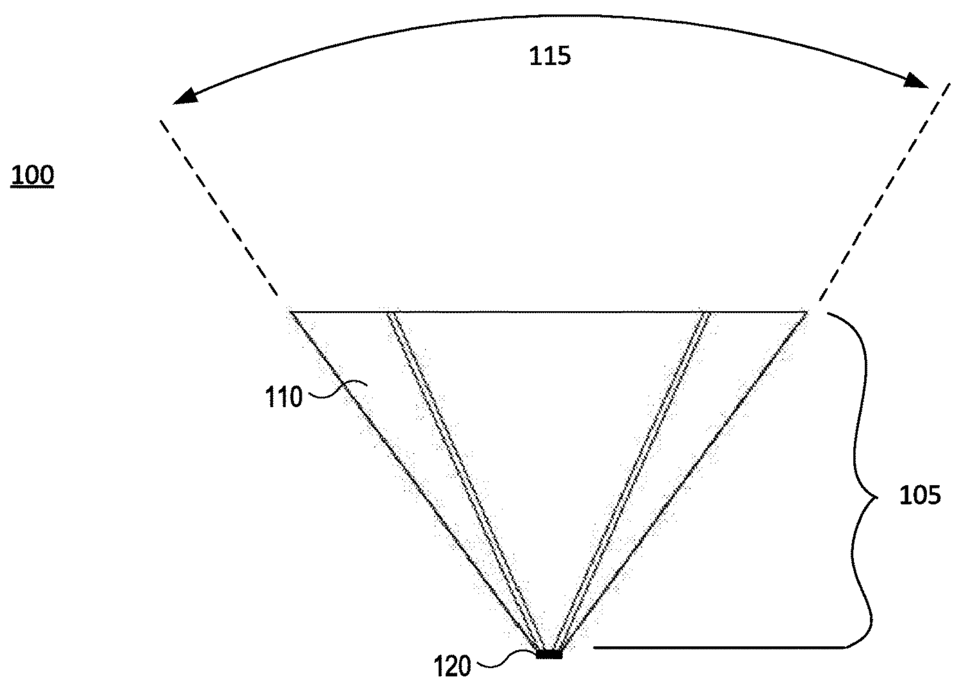


FIG. 2

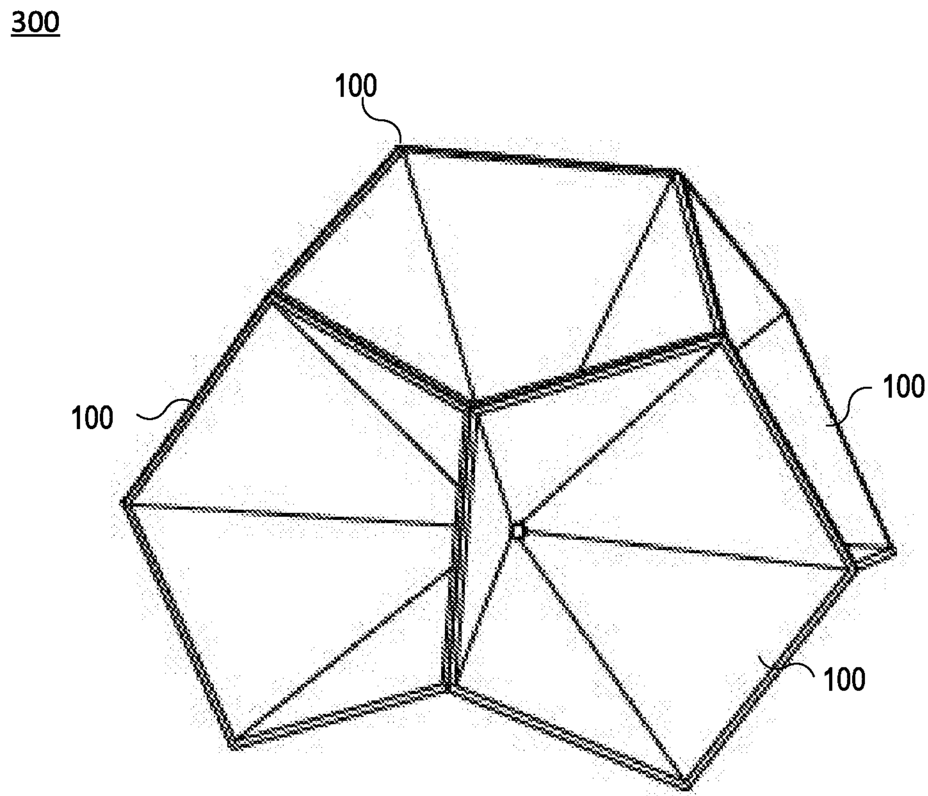


FIG. 3

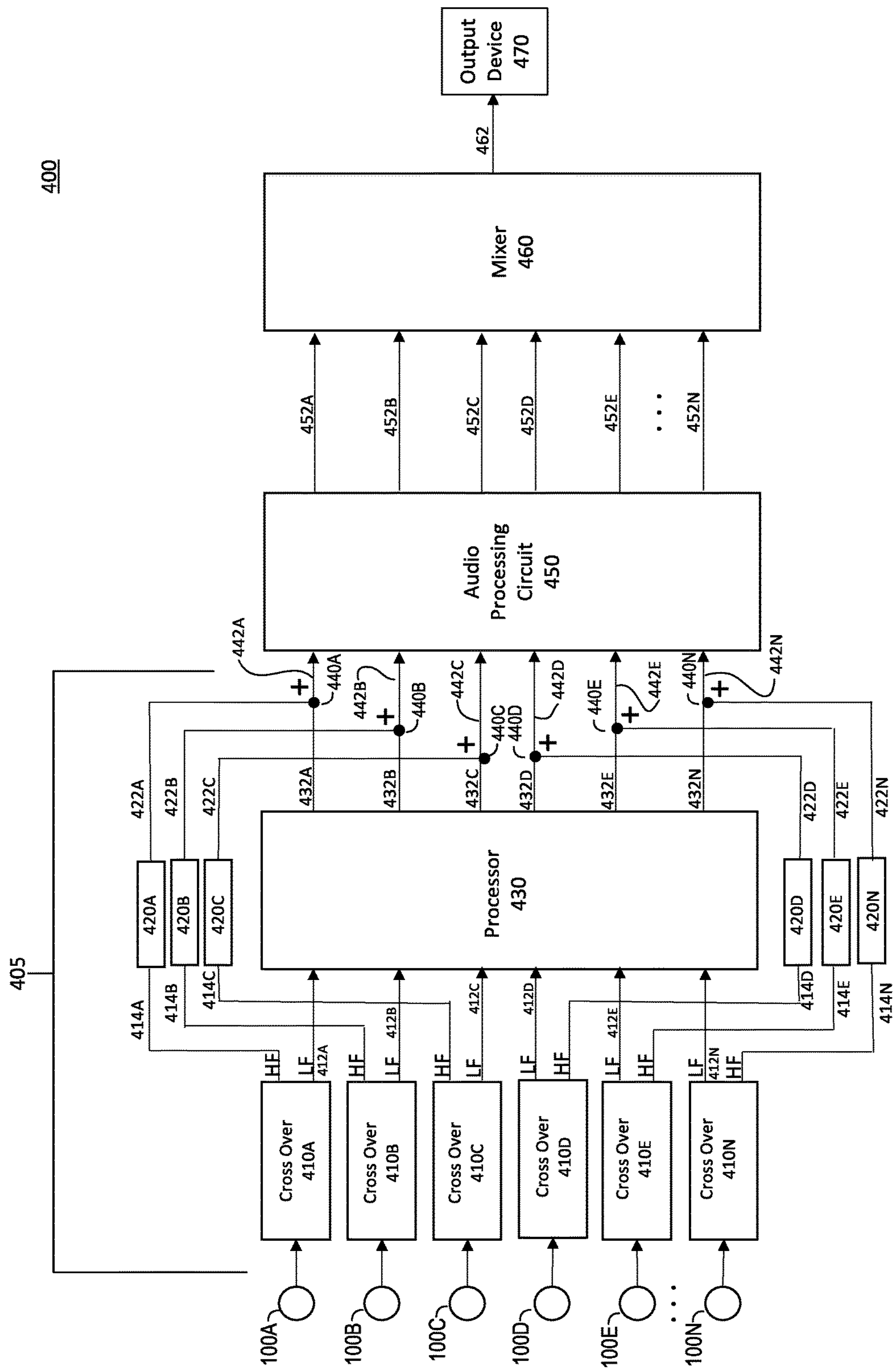

400
405
100A
100B
100C
100D
100E
100N
Cross Over 410A
Cross Over 410B
Cross Over 410C
Cross Over 410D
Cross Over 410E
Cross Over 410N
HF
LF
412A
412B
412C
412D
412E
412N
414A
414B
414C
414D
414E
414N
420A
420B
420C
420D
420E
420N
422A
422B
422C
422D
422E
422N
Processor 430
432A
432B
432C
432D
432E
432N
440A
440B
440C
440D
440E
440N
442A
442B
442C
442D
442E
442N
Audio Processing Circuit 450
452A
452B
452C
452D
452E
452N
Mixer 460
462
Output Device 470

FIG. 4

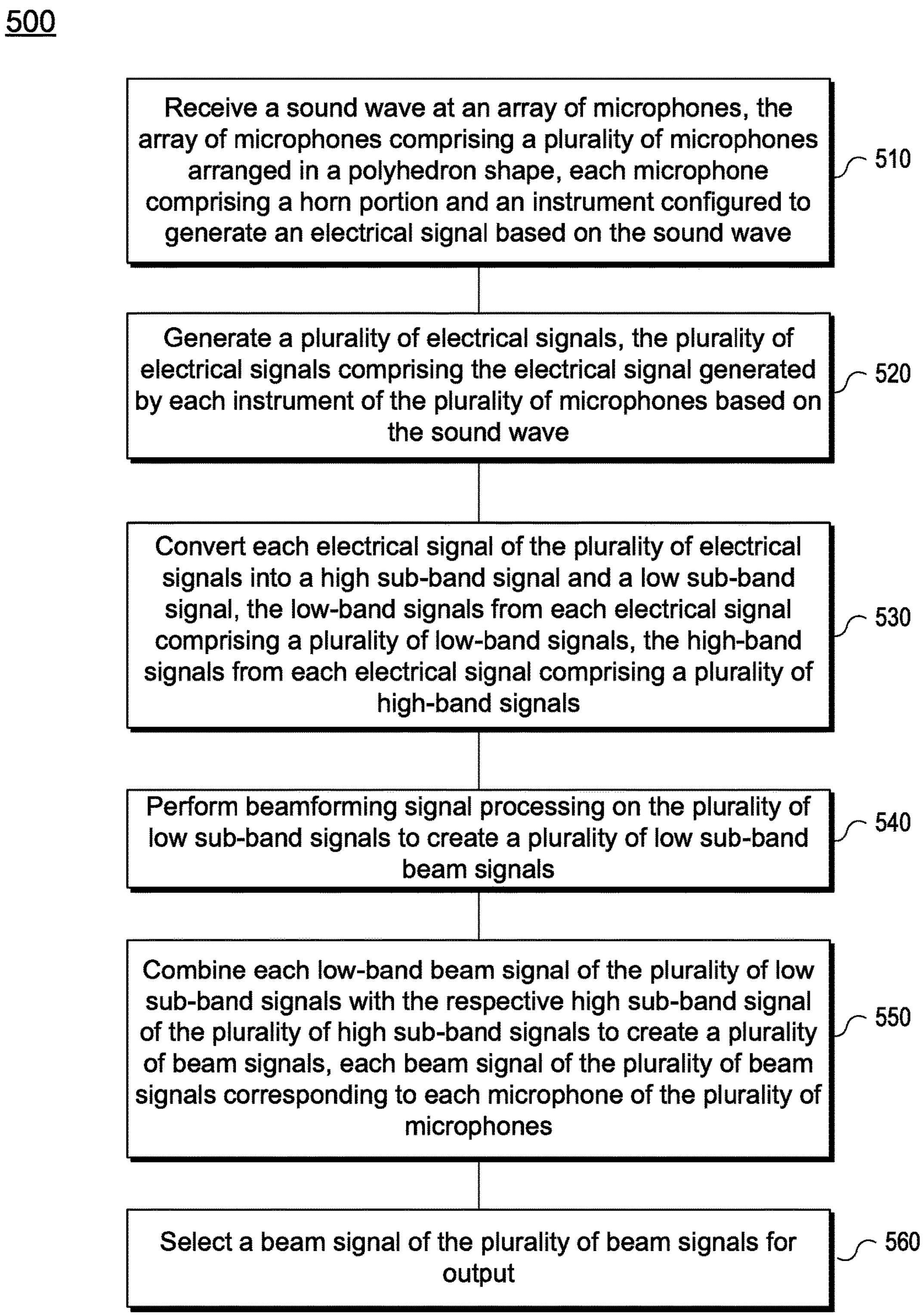

500
Receive a sound wave at an array of microphones, the array of microphones comprising a plurality of microphones arranged in a polyhedron shape, each microphone comprising a horn portion and an instrument configured to generate an electrical signal based on the sound wave
510
Generate a plurality of electrical signals, the plurality of electrical signals comprising the electrical signal generated by each instrument of the plurality of microphones based on the sound wave
520
Convert each electrical signal of the plurality of electrical signals into a high sub-band signal and a low sub-band signal, the low-band signals from each electrical signal comprising a plurality of low-band signals, the high-band signals from each electrical signal comprising a plurality of high-band signals
530
Perform beamforming signal processing on the plurality of low sub-band signals to create a plurality of low sub-band beam signals
540
Combine each low-band beam signal of the plurality of low sub-band signals with the respective high sub-band signal of the plurality of high sub-band signals to create a plurality of beam signals, each beam signal of the plurality of beam signals corresponding to each microphone of the plurality of microphones
550
Select a beam signal of the plurality of beam signals for output
560

FIG. 5

HYBRID HORN MICROPHONE

TECHNICAL FIELD

[0001] This present disclosure relates generally to microphones, and more particularly to a horn microphone utilizing beamforming signal processing.

BACKGROUND

[0002] A Microphone converts air pressure variations of a sound wave into an electrical signal. A variety of methods may be used to convert a sound wave into an electrical signal, such as use of a coil of wire with a diaphragm suspended in a magnetic field, use of a vibrating diaphragm as a capacitor plate, use of a crystal of piezoelectric material, or use of a permanently charged material. Conventional microphones may sense sound waves from all directions (e.g. omni microphone), in a 3D axis symmetric figure of eight pattern (e.g. dipole microphone), or primarily in one direction with a fairly large pickup pattern (e.g. cardioid, super cardioid and hyper cardioid microphones).

[0003] In audio and video conferencing applications involving multiple participants in a given location, unidirectional microphones are undesired. In addition, participants desire speech intelligibility and sound quality without requiring a multitude of microphones placed throughout a conference room. Placing a plurality of microphones in varying locations within a room requires among other things, lengthy cables, cable management, and additional hardware.

[0004] Further, conventional microphone arrays require sophisticated and costly hardware, significant computing performance, complex processing, and may nonetheless lack adequate sound quality when compared to use of multiple microphones placed throughout a room. Moreover, conventional microphone arrays may experience processing artifacts caused by high-frequency spatial aliasing issues.

BRIEF DESCRIPTION OF THE DRAWINGS

[0005] The embodiments herein may be better understood by referring to the following description in conjunction with the accompanying drawings in which like reference numerals indicate identical or functionally similar elements. Understanding that these drawings depict only exemplary embodiments of the disclosure and are not therefore to be considered to be limiting of its scope, the principles herein are described and explained with additional specificity and detail through the use of the accompanying drawings in which:

[0006] FIG. **1** is a top view of a hybrid horn microphone, in accordance with various aspects of the subject technology.

[0007] FIG. **2** is a front view of a hybrid horn microphone, in accordance with various aspects of the subject technology.

[0008] FIG. **3** is a perspective view of a hybrid horn microphone array, in accordance with various aspects of the subject technology.

[0009] FIG. **4** depicts a hybrid horn microphone array processing block diagram, in accordance with various aspects of the subject technology.

[0010] FIG. **5** depicts an example method for processing signals representing sound waves, in accordance with various aspects of the subject technology.

DESCRIPTION OF EXAMPLE EMBODIMENTS

[0011] The detailed description set forth below is intended as a description of various configurations of embodiments and is not intended to represent the only configurations in which the subject matter of this disclosure can be practiced. The appended drawings are incorporated herein and constitute a part of the detailed description. The detailed description includes specific details for the purpose of providing a more thorough understanding of the subject matter of this disclosure. However, it will be clear and apparent that the subject matter of this disclosure is not limited to the specific details set forth herein and may be practiced without these details. In some instances, structures and components are shown in block diagram form in order to avoid obscuring the concepts of the subject matter of this disclosure.

Overview

[0012] Conventional microphones may sense sound waves from all directions (e.g. omni microphone), in a 3D axis symmetric figure of eight pattern (e.g. dipole microphone), or primarily in one direction with a fairly large pickup pattern (e.g. cardioid, super cardioid and hyper cardioid microphones). In applications where sensing of sound from various locations may be required, an array of microphones may be positioned in a central location, such as on the middle of a table in a room. Conventional microphone arrays require sophisticated and costly hardware, significant computing performance, complex processing, and may lack adequate sound quality when compared to use of multiple microphones placed throughout a room or assigned to individual participants or users. In addition, conventional microphone arrays may have a shorter critical distance, that is, the distance in which the microphone array may adequately sense sound due to the sound pressure level of the direct sound and the reverberant sound being equal when dealing with a directional source, when compared to the hybrid horn microphone of the subject technology. Moreover, a conventional microphone array may experience processing artifacts caused by high-frequency spatial aliasing issues.

[0013] The disclosed technology addresses the need in the art for providing a high-sensitive and anti-aliasing microphone by combining horn technology and beamforming signal processing. In an array configuration, the hybrid horn microphone of the subject technology requires less processing power compared to conventional microphone arrays. In addition, the hybrid microphone of the subject technology has a higher signal to noise ratio and less high frequency spatial-aliasing issues than other implementations. The hybrid horn microphone array of the subject technology also has a longer critical distance and increased sound quality compared to conventional microphone arrays.

[0014] In addition, the hybrid horn microphone array of the subject technology does not require multiple arrays, may utilize a single output cable, and may be installed in a single location in a room, such as on or near the ceiling. There is no need for multiple microphones to be located, installed and wired throughout a room. Further, users do not need to reposition table microphones to improve sound quality as the subject technology is capable of processing audio signals to create high quality sound.

DETAILED DESCRIPTION

[0015] Various aspects of the disclosure are discussed in detail below. While specific implementations are discussed, it should be understood that this is done for illustration purposes only. A person skilled in the relevant art will recognize that other components and configurations may be used without parting from the spirit and scope of the disclosure.

[0016] FIG. **1** is a top view of a hybrid horn microphone **100**, in accordance with various aspects of the subject technology. Microphone **100** comprises a horn portion that is formed by a plurality of planar surfaces **110**A-E. The planar surfaces **110**A-E are arranged in a converging orientation to form a shape having a first opening on a proximal end and a second opening on a distal end, the second opening at the distal end being smaller in area than the first opening at the proximal end.

[0017] The plurality of planar surfaces **110** may be substantially planar and devoid of curvature such that a cross-sectional area of the horn portion from the proximal end to the distal end decreases at a constant rate. In some aspects, the planar surfaces may include curvature such that the cross-sectional area of the horn portion from the proximal end to the distal end decreases with varying rates.

[0018] The plurality of planar surfaces **110** may be made of polymer, composite, metal, alloys, or a combination thereof. It is understood that other materials may be used to form the horn portion without deviating from the scope of the subject technology.

[0019] Each planar surface **110** of the plurality of planar surfaces **110**A-E may have substantially the same thickness. The thickness of each planar surface **110** may be 0.13", 0.25", 0.38", or 0.5". It is understood that the planar surfaces **110** may have other values for thickness without departing from the scope of the subject technology.

[0020] In some aspects, the length of the planar surface **110** may range from 4-6 inches, 6-8 inches, 8-10 inches, 10-12 inches or 12-14 inches. It is understood that the planar surface **110** may have a longer length without departing from the scope of the subject technology. In one aspect, a width of the planar surface is similar to the length of the planar surface.

[0021] In one aspect, the horn portion may be formed by a single component, folded, cast, or molded into the desired shape. For example, the horn portion may comprise sheet metal folded into a pentagonal pyramid having five planar surfaces **110**A-E. In another aspect, the horn portion may be assembled from multiple components with each component comprising the planar surface **110**.

[0022] FIG. **2** is a front view of the hybrid horn microphone **100**, in accordance with various aspects of the subject technology. The microphone **100** includes an instrument **120** disposed at the distal end of the horn portion **105**. The distal end is located where the planar surfaces **110**A-E converge to form a narrow opening. The instrument **120** is configured to detect sound waves and convert air pressure variations of a sound wave into an electrical signal. The instrument **120** may comprise an electret microphone. An electret microphone is a type of electrostatic capacitor-based microphone.

[0023] Sound waves emitted by a source, such as a user speaking at a telephonic or video conference, are directed or reflected towards the horn portion **105** and are directed to the instrument **120** by the shape of the planar surfaces **110**A-E. In one aspect, the size and shape of the horn portion **105** correlates to a frequency range or bandwidth of the sound waves desired for detection.

[0024] In another aspect, by utilizing the horn portion **105**, the microphone **100** detects and senses sound waves directionally. That is, the microphone **100** is capable of detecting sound waves from a source located within a detection range **115**, while minimizing detection of sound waves from other sources that may be located at different locations from the source, outside of the detection range **115**. By utilizing the horn portion **105**, the microphone **100** is also able to prevent detection of ambient noise (typically greater than 10 dB) coming from sources located outside of the detection range. In one aspect, the horn portion **105** of the microphone **100** significantly reduces detection of sound waves coming from angles outside of the direction of the microphone **100** because the sound waves from outside the direction of the microphone **100** are reflected away from the instrument **120** by the horn portion **105**. In another aspect, for sound waves coming from a source located within the detection range **115** of the microphone **100**, a Signal to Noise Ratio (SNR) of the sound wave is significantly higher (generally 9 dB or more) than conventional microphones resulting in increased sound quality. In one aspect, for sound waves coming from a source within the detection range **115**, the microphone **100** has a very high directivity at frequencies above 2 kHz.

[0025] In some aspects, the horn portion **105** may have various shapes formed by the planar surfaces **110**. For example, the shape of the horn portion **105** formed by the plurality of planar surfaces **110** may comprise a triangular pyramid having three interior faces. In another example, the shape of the horn portion **105** formed by the plurality of planar surfaces **110** may comprise a square pyramid having four interior faces. In yet another example, the shape of the horn portion **105** formed by the plurality of planar surfaces **110** may comprise a pentagonal pyramid having five interior faces. In another example, the shape of the horn portion **105** formed by the plurality of planar surfaces **110** may comprise a hexagonal pyramid having six interior faces. In yet another example, the shape of the horn portion **105** formed by the plurality of planar surfaces **110** may comprise a heptagonal pyramid having seven interior faces. In another example, the shape of the horn portion **105** formed by the plurality of planar surfaces **110** may comprise an octagonal pyramid having eight interior faces. It is further understood that other shapes may be formed by the plurality of planar surfaces **110** as desired by a person of ordinary skill in the art.

[0026] FIG. **3** is a perspective view of a hybrid horn microphone array **300**, in accordance with various aspects of the subject technology. In some aspects, the horn microphone **100** may be arranged in an array **300** to receive sound waves from one or more sources located within an area, such as a conference room. For example, the array **300** of microphones **100** may be arranged to form a polyhedron shape, such as a full dodecahedron that may be formed by arranging twelve microphones **100** into a full sphere dodecahedron arrangement. In another example, the polyhedron shape may comprise a half dodecahedron that may be formed by arranging six microphones **100** into a half dodecahedron arrangement (as shown in FIG. **3**). In yet another example, the polyhedron shape may comprise a quarter dodecahedron formed by arranging three microphones **100** into a quarter dodecahedron arrangement. It is understood that the array **300** may comprise other shapes and may be formed of a multitude of microphones **100**, including up to 120 microphones **100**. In one aspect, the higher the number of microphones **100** comprising the array, the narrower the detection of sound waves from the source.

[0027] Each microphone **100** of the array **300** is pointed at a different direction, as shown in FIG. **3**. In some aspects, by forming the array **300** with the plurality of microphones **100** arranged so that each microphone **100** is pointed at a different direction, each microphone **100** is configured to detect sound waves from the direction the microphone is pointed.

[0028] FIG. **4** depicts a hybrid horn microphone array processing block diagram **400**, in accordance with various aspects of the subject technology. The microphone array **300** (shown in FIG. **3**) may further comprise the hybrid horn microphone array processing block diagram **400** to process the electrical signals generated by the instrument **120** (shown in FIGS. **1** and **2**) of each microphone **100**. In one aspect, the functions and operations depicted in the hybrid horn microphone array processing block diagram **400** may be performed by components mounted to the array **300**, components located at a remote location, or at an output device as discussed further below.

[0029] The hybrid horn microphone array processing block diagram **400** comprises a beamforming signal processing circuit **405** for creating a high-sensitivity and anti-aliasing microphone array **300**. The beamforming signal processing circuit **405** is electrically coupled to each microphone **100** and is configured to receive the electrical signals from each instrument **120**. The beamforming signal processing circuit **405** is further configured to create beam signals corresponding to each microphone **100** based on the respective electrical signals. In some aspects, the beam signals are indicative of a location of a source of the sound waves detected by each microphone **100**.

[0030] The beamforming signal processing circuit **405** comprises a crossover filter **410**, a delaying circuit **420**, a processor **430**, and a mixer **440**. Each electrical signal from the microphones **100**A-N passes through respective cross over filters **410**A-N. Each crossover filter **410**A-N is configured to convert the respective electrical signals from the microphone **100**A-N to a first signal **412** and a second signal **414**, with the first and second signals, **412** and **414** respectively, having different frequencies or sub-bands. For example, the frequency of each respective first signal **412** may be below 2 kHz and the frequency of each respective second signal **414** may be above 2 kHz. In one aspect, the crossover frequency can be adapted to the size of the horn portion **105** (as shown in FIG. **2**) of the microphone **100** in the array **300**.

[0031] For example, with reference to a first microphone **100**A, the electrical signal from the microphone **100**A is received by the cross over filter **410**A. The cross over filter **410**A converts the electrical signal from the microphone **100**A into a first signal **412**A (Low Frequency or LF) and a second signal **414**A (High Frequency or HF). With reference to a second microphone **100**B, the electrical signal from the microphone **100**B is received by the cross over filter **410**B. The cross over filter **410**B converts the electrical signal from the microphone **100**B into a first signal **412**B (Low Frequency or LF) and a second signal **414**B (High Frequency or HF). With reference to a third microphone **100**C, the electrical signal from the microphone **100**C is received by the cross over filter **410**C. The cross over filter **410**C converts the electrical signal from the microphone **100**C into a first signal **412**C (Low Frequency or LF) and a second signal **414**C (High Frequency or HF). With reference to a fourth microphone **100**D, the electrical signal from the microphone **100**D is received by the cross over filter **410**D. The cross over filter **410**D converts the electrical signal from the microphone **100**D into a first signal **412**D (Low Frequency or LF) and a second signal **414**D (High Frequency or HF). With reference to a fifth microphone **100**E, the electrical signal from the microphone **100**E is received by the cross over filter **410**E. The cross over filter **410**E converts the electrical signal from the microphone **100**E into a first signal **412**E (Low Frequency or LF) and a second signal **414**E (High Frequency or HF). In some aspects, any number of microphones **100**N may be connected to the beamforming signal processing circuit **405**, including the cross over filter **410**N to convert the electrical signal from the microphone **100**N into a first signal **412**N and a second signal **414**N, without departing from the scope of the subject technology.

[0032] The delaying circuit **420** is configured to delay the second signal **414** from the crossover filter **410** to create a delayed second signal **422**. In some aspects, the delaying circuit is configured to sufficiently delay the second signal **414** so that upon mixing by the mixer **440**, as discussed further below, the mixed signal is sufficiently aligned. Each second signal **414**A-N from the respective cross over filters **410**A-N is received by corresponding delaying circuits **420**A-N to create respective delayed second signals **422**A-N.

[0033] For example, with reference to the first microphone **100**A, the second signal **414**A from the cross over filter **410**A is received by the delaying circuit **420**A. The delaying circuit **420**A delays the second signal **414**A to create a delayed second signal **422**A. With reference to the second microphone **100**B, the second signal **414**B from the cross over filter **410**B is received by the delaying circuit **420**B. The delaying circuit **420**B delays the second signal **414**B to create a delayed second signal **422**B. With reference to the third microphone **100**C, the second signal **414**C from the cross over filter **410**C is received by the delaying circuit **420**C. The delaying circuit **420**C delays the second signal **414**C to create a delayed second signal **422**C. With reference to the fourth microphone **100**D, the second signal **414**D from the cross over filter **410**D is received by the delaying circuit **420**D. The delaying circuit **420**D delays the second signal **414**D to create a delayed second signal **422**D. With reference to the fifth microphone **100**E, the second signal **414**E from the cross over filter **410**E is received by the delaying circuit **420**E. The delaying circuit **420**E delays the second signal **414**E to create a delayed second signal **422**E. In some aspects, any number of microphones **100**N may be connected to the beamforming signal processing circuit **405**, including the delaying circuit **420**N to delay the second signal **414**N and create a delayed second signal **422**N, without departing from the scope of the subject technology.

[0034] The processor **430** may be configured to downsample the first signal **412** from the crossover filter **410** to create a downsampled first signal, process the downsampled first signal to create a processed first signal that is indicative of the location of the source of the sound waves detected by the microphone **100**, and upsample the processed first signal to create an upsampled first signal **432**. Each first signal **412**A-N from the respective cross over filters **410**A-N is received by the processor **430** to create the processed first signal **432**A-N.

[0035] In some aspects, the processor **430** utilizes beamforming signal processing techniques to process the first signals **412**A-N. Beam forming signal processing may be used to extract sound sources in an area or room. This may be achieved by combining elements in a phased array in such a way that signals at particular angles experience constructive interference while others experience destructive interference.

[0036] In one aspect, because the horn portion **105** (as shown in FIG. **2**) of the microphone **100** significantly reduces detection of sound waves coming from angles outside of the direction of the microphone **100**, provides a high SNR for sound waves coming from a source located within the detection range **115** (as shown in FIG. **2**), and provides a very high directivity at frequencies above 2 kHz; no processing is required by the processor **430** for the second signals **414**A-N. In one aspect, because no processing is required for the second signals **414**A-N, spatial aliasing issues are avoided.

[0037] The processor **430** may downsample each of the first signals **412**A-N to a lower sampling rate such as from 48 kHz to 4 kHz, which may significantly reduce computational complexity by 90%. The processor **430** may then filter and sum (or weight and sum in the frequency domain) each of the first signals **412**A-N to create respective processed first signals representing acoustic beams pointing in the direction of each respective microphone. In another example, the processer **430** may use spherical harmonics theory or sound field models to create respective processed first signals representing acoustic beams pointing in the direction of each respective microphone. In one aspect, the processor **430** may measure the array response vectors for various sound arrival angles in an anechoic chamber. In another aspect, the processor **430** may implement various types of beam pattern synthesis/optimization or machine learning. The processor **430** may then upsample the processed first signals to obtain respective upsampled first signals **432** with a desired sampling rate.

[0038] For example, with reference to the first microphone **100**A, the first signal **412**A from the cross over filter **410**A is received by the processor **430**. The processor **430** may downsample the first signal **412**A to create a first downsampled first signal. The processor **430** may then filter and sum (or weight and sum in the frequency domain) the first downsampled first signal to create a first processed first signal representing an acoustic beam pointing in the direction of microphone **100**A. The first processed first signal indicative of the location of the source of the sound waves detected by the microphone **100**A. The processor **430** may then upsample the first processed first signal to obtain an upsampled first signal **432**A. With respect to the second microphone **100**B, the first signal **412**B from the cross over filter **410**B is received by the processor **430**. The processor **430** may downsample the first signal **412**B to create a second downsampled first signal. The processor **430** may then filter and sum (or weight and sum in the frequency domain) the second downsampled first signal to create a second processed first signal representing an acoustic beam pointing in the direction of microphone **100**B. The second processed first signal indicative of the location of the source of the sound waves detected by the microphone **100**B. The processor **430** may then upsample the second processed first signal to obtain an upsampled first signal **432**B. With respect to the third microphone **100**C, the first signal **412**C from the cross over filter **410**C is received by the processor **430**. The processor **430** may downsample the first signal **412**C to create a third downsampled first signal. The processor **430** may then filter and sum (or weight and sum in the frequency domain) the third downsampled first signal to create a third processed first signal representing an acoustic beam pointing in the direction of microphone **100**C. The third processed first signal indicative of the location of the source of the sound waves detected by the microphone **100**C. The processor **430** may then upsample the third processed first signal to obtain an upsampled first signal **432**C. With respect to the fourth microphone **100**D, the first signal **412**D from the cross over filter **410**D is received by the processor **430**. The processor **430** may downsample the first signal **412**D to create a fourth downsampled first signal. The processor **430** may then filter and sum (or weight and sum in the frequency domain) the fourth downsampled first signal to create a fourth processed first signal representing an acoustic beam pointing in the direction of microphone **100**D. The fourth processed first signal indicative of the location of the source of the sound waves detected by the microphone **100**D. The processor **430** may then upsample the fourth processed first signal to obtain an upsampled first signal **432**D. With respect to the fifth microphone **100**E, the first signal **412**E from the cross over filter **410**E is received by the processor **430**. The processor **430** may downsample the first signal **412**E to create a fifth downsampled first signal. The processor **430** may then filter and sum (or weight and sum in the frequency domain) the fifth downsampled first signal to create a fifth processed first signal representing an acoustic beam pointing in the direction of microphone **100**E. The fifth processed first signal indicative of the location of the source of the sound waves detected by the microphone **100**E. The processor **430** may then upsample the fifth processed first signal to obtain an upsampled first signal **432**E. In some aspects, any number of microphones **100**N may be connected to the beamforming signal processing circuit **405**, including the processor **430** to downsample, process and upsample the first signal **412**N and create a upsampled first signal **432**N, without departing from the scope of the subject technology.

[0039] The mixer **440** is configured to combine the upsampled first signal **432** from the processor **430** and the delayed second signal **422** from the delaying circuit **420** to create a full-band beam signal **442**. Each upsampled first signal **432**A-N and delayed second signal **422**A-N from the respective delaying circuits **420**A-N is received by corresponding mixers **440**A-N to create respective full-band beam signals **442**A-N.

[0040] For example, with reference to the first microphone **100**A, the upsampled first signal **432**A from the processor **430** and the delayed second signal **422**A from the delaying circuit **420**A is received by the mixer **440**A. The mixer **440**A combines the upsampled first signal **432**A and the delayed second signal **422**A to create a beam signal **442**A. With reference to the second microphone **100**B, the upsampled first signal **432**B from the processor **430** and the delayed second signal **422**B from the delaying circuit **420**B is received by the mixer **440**B. The mixer **440**B combines the upsampled first signal **432**B and the delayed second signal **422**B to create a beam signal **442**B. With reference to the third microphone **100**C, the upsampled first signal **432**C from the processor **430** and the delayed second signal **422**C from the delaying circuit **420**C is received by the mixer **440**C. The mixer **440**C combines the upsampled first signal

432C and the delayed second signal **422**C to create a beam signal **442**C. With reference to the fourth microphone **100**D, the upsampled first signal **432**D from the processor **430** and the delayed second signal **422**D from the delaying circuit **420**D is received by the mixer **440**D. The mixer **440**D combines the upsampled first signal **432**D and the delayed second signal **422**D to create a beam signal **442**D. With reference to the second microphone **100**E, the upsampled first signal **432**E from the processor **430** and the delayed second signal **422**E from the delaying circuit **420**E is received by the mixer **440**E. The mixer **440**E combines the upsampled first signal **432**E and the delayed second signal **422**E to create a beam signal **442**E. In some aspects, any number of microphones **100**N may be connected to the beamforming signal processing circuit **405**, including the mixer **440**N to combine the upsampled first signal **432**N and delayed second signal **422**N to create the beam signal **442**N, without departing from the scope of the subject technology.

[0041] The hybrid horn microphone array processing block diagram **400** may further comprise an audio processing circuit **450**. The audio processing circuit **450** may be configured to receive each of the beam signals **442**A-N and perform at least one of an echo control filter, a reverberation filter, or a noise reduction filter, to improve the quality of the beam signals **442**A-N and create pre-mixed beam signals **452**A-N.

[0042] For example, with reference to the first microphone **100**A, the beam signal **442**A from the mixer **440**A is received by the audio processing circuit **450**. The audio processing circuit **450** performs operations such as echo modification, reverberation adjustment, or noise reduction, to improve the quality of the beam signal **442**A, and thereby create a pre-mixed beam signal **452**A. With reference to the second microphone **100**B, the beam signal **442**B from the mixer **440**B is received by the audio processing circuit **450**. The audio processing circuit **450** performs operations such as echo modification, reverberation adjustment, or noise reduction, to improve the quality of the beam signal **442**B, and thereby create a pre-mixed beam signal **452**B. With reference to the third microphone **100**C, the beam signal **442**C from the mixer **440**C is received by the audio processing circuit **450**. The audio processing circuit **450** performs operations such as echo modification, reverberation adjustment, or noise reduction, to improve the quality of the beam signal **442**C, and thereby create a pre-mixed beam signal **452**C. With reference to the fourth microphone **100**D, the beam signal **442**D from the mixer **440**D is received by the audio processing circuit **450**. The audio processing circuit **450** performs operations such as echo modification, reverberation adjustment, or noise reduction, to improve the quality of the beam signal **442**D, and thereby create a pre-mixed beam signal **452**D. With reference to the fifth microphone **100**E, the beam signal **442**E from the mixer **440**E is received by the audio processing circuit **450**. The audio processing circuit **450** performs operations such as echo modification, reverberation adjustment, or noise reduction, to improve the quality of the beam signal **442**E, and thereby create a pre-mixed beam signal **452**E. In some aspects, any number of microphones **100**N may be connected to the audio processing circuit **450** to improve the quality of the beam signal **442**N and create pre-mixed beam signal **452**N, without departing from the scope of the subject technology.

[0043] The hybrid horn microphone array processing block diagram **400** may further comprise an automatic mixer **460**. The automatic mixer **460** may be configured to receive the plurality of pre-mixed beam signals **452**A-N and identify one or more beam signals from the plurality of beam signals **452**A-N to output to an output device **470** based on a characteristic of the beam signal **452**A-N. The characteristic of the beam signal **452**A-N may include, for example, quality, level, clarity, strength, SNR, signal to reverberation ratio, amplitude, wavelength, frequency, or phase. In some aspects, the mixer **460** may be configured to review each incoming pre-mix beam signal **452**A-N, identify one or more beam signals **452**A-N based on one or more characteristic of the beam signals **452**A-N, select the one or more beam signals **452**A-N, isolate signals representing speech, filter low signals that may not represent speech, and transmit an output signal **462** to the output device **470**. In one aspect, the mixer **460** may utilize audio selection techniques to generate the desired audio output signal **462** (e.g., mono, stereo, surround).

[0044] The output device **470** is configured to receive the output signal **462** from the mixer and may comprise a set top box, console, visual output device (e.g., monitor, television, display), or audio output device (e.g., speaker).

[0045] FIG. **5** depicts an example method **500** for processing signals representing sound waves, in accordance with various aspects of the subject technology. It should be understood that, for any process discussed herein, there can be additional, fewer, or alternative steps performed in similar or alternative orders, or in parallel, within the scope of the various embodiments unless otherwise stated.

[0046] At operation **510**, a sound wave is received at an array of microphones. The array of microphones comprise a plurality of microphones arranged in a polyhedron shape, as shown for example, in FIG. **3**. Each microphone may comprise a horn portion and an instrument, the instrument configured to generate an electrical signal based on the sound wave. The horn portion may comprise a plurality of planar surfaces that are arranged to form the polyhedron shape.

[0047] At operation **520**, a plurality of electrical signals are generated based on the received sound wave. The plurality of electrical signals comprise the electrical signal generated by each instrument of the plurality of microphones.

[0048] At operation **530**, each electrical signal of the plurality of electrical signals is converted into a high subband signal and a low sub-band signal. The electrical signal generated by each instrument and microphone, is thus converted to two signals, the high sub-band signal and the low sub-band signal. Each of the low-band signals, together, comprise a plurality of low-band signals. Similarly, each of the high-band signals, together, comprise a plurality of high-band signals.

[0049] At operation **540**, beamforming signal processing is performed on the plurality of low sub-band signals to create a plurality of low sub-band beam signals. Stated differently, each of the low-band signals undergoes beamforming signal processing to thereby create a low sub-band beam signal. As described above, beamforming signal processing may comprise use of spherical harmonics theory or sound field models, use of array response vectors for various sound arrival angles in an anechoic chamber, and/or use of various types of beam pattern synthesis/optimization or machine learning.

[0050] At operation **550**, each low-band beam signal of the plurality of low sub-band signals is combined with the respective high sub-band signal of the plurality of high sub-band signals to create a plurality of beam signals. Each beam signal of the plurality of beam signals corresponds to each microphone of the plurality of microphones of the array.

[0051] At operation **560**, one or more beam signals of the plurality of beam signals is elected for output to an output device.

[0052] The functions described above can be implemented using computer-executable instructions that are stored or otherwise available from computer readable media. Such instructions can comprise, for example, instructions and data which cause or otherwise configure a general purpose computer, special purpose computer, or special purpose processing device to perform a certain function or group of functions. The computer executable instructions may be, for example, binaries, intermediate format instructions such as assembly language, firmware, or source code. Examples of computer-readable media that may be used to store instructions, information used, and/or information created during methods according to described examples include magnetic or optical disks, flash memory, USB devices provided with non-volatile memory, networked storage devices, and so on.

[0053] Devices implementing the functions and operations according to these disclosures may comprise hardware, firmware and/or software, and can take any of a variety of form factors. Typical examples of such form factors include laptops, smart phones, small form factor personal computers, personal digital assistants, rackmount devices, standalone devices, and so on. Functionality described herein also can be embodied in peripherals or add-in cards. Such functionality can also be implemented on a circuit board among different chips or different processes executing in a single device, by way of further example.

[0054] The instructions, media for conveying such instructions, computing resources for executing them, and other structures for supporting such computing resources are means for providing the functions described in these disclosures.

[0055] Although a variety of examples and other information was used to explain aspects within the scope of the appended claims, no limitation of the claims should be implied based on particular features or arrangements in such examples, as one of ordinary skill would be able to use these examples to derive a wide variety of implementations. Further and although some subject matter may have been described in language specific to examples of structural features and/or method steps, it is to be understood that the subject matter defined in the appended claims is not necessarily limited to these described features or acts. For example, such functionality can be distributed differently or performed in components other than those identified herein. Rather, the described features and steps are disclosed as examples of components of systems and methods within the scope of the appended claims.

1. A system for converting sound waves, the system comprising:

an array of microphones, the array comprising a plurality of microphones, each microphone of the plurality of microphones comprising:

a horn portion comprising a plurality of surfaces, the surfaces arranged in a converging orientation to form a shape having a first opening at a proximal end and a second opening at a distal end, the second opening at the distal end being smaller in area than the first opening at the proximal end; and

an instrument disposed at the distal end of the horn portion, the instrument configured to convert sound waves into an electrical signal;

the microphones of the array are radially disposed around a central point and oriented to direct received sound waves to that central point; and

a beamforming signal processing circuit electrically coupled to each instrument of the plurality of microphones and configured to create a plurality of beam signals based on the respective electrical signals of each instrument.

2. The system of claim **1**, wherein the beamforming signal processing circuit comprises a crossover filter, a processor, a delaying circuit, and a mixer.

3. The system of claim **2**, wherein the crossover filter is configured to convert the electrical signal from each instrument of the plurality of microphones to respective first signals and second signals.

4. The system of claim **3**, wherein the processor is configured to:

downsample each of the first signals from the crossover filter to create respective downsampled first signals;

process each of the downsampled first signals to create respective processed first signals, the processed first signals indicative of a location of the source of the sound waves detected by the respective instrument; and

upsampled each of the processed first signals to create respective upsampled first signals.

5. The system of claim **4**, wherein the delaying circuit is configured to delay each of the second signals from the crossover filter to create respective delayed second signals.

6. The system of claim **5**, wherein the mixer is configured to combine each of the upsampled first signals from the processor with corresponding delayed second signals from the delaying circuit to create the plurality of beam signals.

7. The system of claim **1**, further comprising an audio processing circuit, the audio processing circuit configured to perform at least one of an echo control filer, a reverberation filter, or a noise reduction filter, to the plurality of beam signals from the beamforming signal processing circuit.

8. The system of claim **1**, wherein the shape of the horn portion formed by the plurality of surfaces comprises a square pyramid having four interior faces.

9. The system of claim **1**, wherein the shape of the horn portion formed by the plurality of surfaces comprises a pentagonal pyramid having five interior faces.

10. The system of claim **1**, wherein the shape of the horn portion formed by the plurality of surfaces comprises a hexagonal pyramid having six interior faces.

11. The system of claim **1**, wherein each beam signal of the plurality of beam signals is indicative of a location of a source of the sound waves detected by each respective instrument.

12. A microphone array comprising:

a plurality of microphones arranged to form an array, the microphones of the array being radially disposed around a central point and oriented to direct received sound waves to that central point, each microphone of the plurality of microphones comprising;

a horn portion comprising a plurality of planar surfaces, the planar surfaces arranged in a converging orientation to form a shape having a first opening on a proximal end and a second opening on a distal end, the second opening on the distal end being smaller in area than the first opening on the proximal end; and

an instrument disposed on the distal end of the horn portion, the instrument configured to detect sound waves and convert sound waves into an electrical signal;

a beamforming signal processing circuit electrically coupled to each instrument of plurality of microphones, the beamforming signal processing circuit configured to:

receive a plurality of electrical signals, the plurality of electrical signals comprising the electrical signal from each microphone of the plurality of microphones; and

create a plurality of beam signals based on the plurality of electric signals each beam signal of the plurality of beam signals corresponding to the electrical signal from each microphone of the plurality of microphones.

13. The microphone array of claim 12, wherein the beamforming signal processing circuit comprises a crossover filter, a processor, a delaying circuit, and a mixer.

14. The microphone array of claim 12, further comprising an audio processing circuit, the audio processing circuit configured to perform at least one of an echo control filter, a reverberation filter, or a noise reduction filter, to the plurality of beam signals from the beamforming signal processing circuit.

15. The microphone array of claim 12, further comprising an automatic mixer, the automatic mixer configured to receive the plurality of beam signals and identify a beam signal from the plurality of beam signals based on a characteristic of the beam signal.

16. The microphone array of claim 12, wherein the shape of the horn portion of each microphone of the plurality of microphones comprises a pentagonal pyramid having five interior faces.

17. The microphone array of claim 12, wherein the array comprises a polyhedron shape.

18. The microphone array of claim 17, wherein the polyhedron shape comprises a half dodecahedron.

19. The microphone array of claim 12, wherein each beam signal is indicative of a location of a source of the sound waves detected by each microphone of the plurality of microphones.

20. A method for creating a plurality of beam signals, the method comprising:

receiving a sound wave at an array of microphones, the array of microphones comprising a plurality of microphones radially disposed around a central point and oriented to direct received sound waves to that central point; each microphone comprising a horn portion and an instrument, the instrument configured to generate an electrical signal based on the sound wave;

generating a plurality of electrical signals based on the received sound wave, the plurality of electrical signals comprising the electrical signal generated by each instrument of the plurality of microphones;

converting each electrical signal of the plurality of electrical signals into a high sub-band signal and a low sub-band signal, the low sub-band signals from each electrical signal comprising a plurality of low sub-band signals, the high sub-band signals from each electrical signal comprising a plurality of high sub-band signals;

performing beamforming signal processing on the plurality of low sub-band signals to create a plurality of low sub-band beam signals;

combining each low-band beam signal of the plurality of low sub-band signals with the respective high sub-band signal of the plurality of high sub-band signals to create a plurality of beam signals, each beam signal of the plurality of beam signals corresponding to each microphone of the plurality of microphones of the array; and

selecting an output beam signal from the plurality of beam signals for output to an output device.

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