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Voice over Wireless Sensor Network (VoWSN) System: A Literature Survey

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Abstract: Recently, wireless sensor networks (WSNs) have been confirmed as an important low power embedded computing platform. As a consequence it has become a significant research area for many applications such as industrial control, environmental monitoring systems, intelligent agriculture, health monitoring. As one of the active research areas, voice application based sensor network is still new. The researchers efforts to exploit the WSN for transferring the voice led to emerge a new attitude refers to as Voice over WSN (VoWSN). This attitude opens up an interesting opportunity for numerous fields of applications based on human voice like voice communication system, monitoring and security system, and ambient intelligence. However, supporting multimedia transport like voice using these low cost, resource – limit sensor networks, is never a trivial task and the researches have been conducted widely. To cope the subject of VOWSN in its all aspects, this research reviews the state of the art and the major research challenges in algorithms, architectures, and requirements for voice over wireless sensor networks. Furthermore, This study presents a literature overview concerning the feasibility of designing and implementing the VoWSN system and assigns the problems of supporting the voice over WSNs despite sensor nodes only have limited processing and communication capabilities.

Keywords: WSN, Voice Communication, Ambient Intelligence, Embedded Systems, Resource Constrains

1. Introduction

Wireless Sensor Network is a technology consists of extensive number of distributed sensor nodes that is organized and operated independently. These sensor nodes are disseminated in the field to gather information relating to some phenomenon. Usually, the main characteristic of these sensor nodes is small in size with limited resources such as memory, processing capabilities and energy source. In its basic architecture, each sensor node consists of sensing unit, microcontroller (8-bit or 16-bit), ADC, memory, transceiver and power unit. Besides, these sensor nodes may likewise have extra components for example, a power generator, allocation finding system, and a mobilize which are application subordinate components [1]. A typical WSN system and architecture of sensor node is shown in Figure 1 [2]. As appeared in Figure 1, WSN system may comprise three classes of nodes: sensor nodes, routing/ forwarding nodes, anda gateway or a base station (BS). The wireless sensor nodegathersdata and transmits them through the routing nodes (in a single hop or multiple hop)to the base station(BS) for additionally processing.

The sensor nodes and the base station(BS) used the protocol stack that given in Figure 2 [1]. A protocol stack for WSNs must support their typical features and singularities. As shown in Figure 2, the protocol stack is much like the traditional protocol stack which combines the following layers (from the bottom of the stack): physical, data link, network, transport and application. The physical layer is responsible of simple and robust modulation, carrier frequency generation, data encryption and signal detection. The data link layer takes care of multiplexing of data streams, Media Access Control (MAC), frame detection and error control. The main features of the MAC protocol are being power aware and able to minimize collision with neighbors' broadcast since the environment where the sensor nodes are disseminated is noisy and sensor nodes can be mobile. The network layer is responsible of routing the data supplied by the transport layer between sensor nodes and sink using specific multi-hop wireless routing protocols. The transport layer helps to maintain the flow of data when the sensor networks application requires it. Finally, depending on the sensing tasks, different types of application software can be built and used on the application layer. Moreover, the protocol stack may also supports different planes: power management plane, mobility management plane and task management plane. These planes helps the sensor nodes coordinating the sensing tasks and lowering the overall energy consumption through monitoring the power, movement, and task distribution among the sensor nodes.

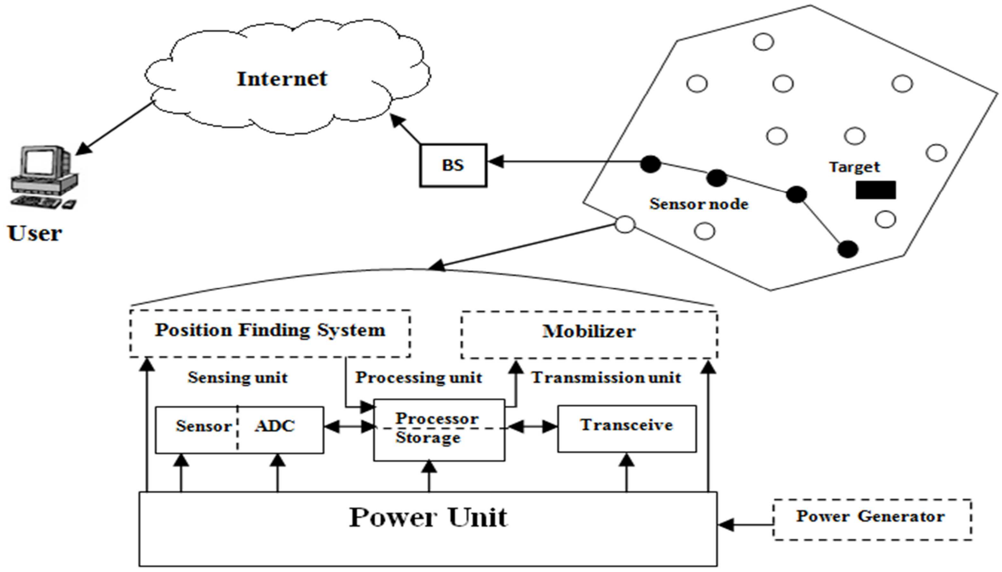

Figure 1. *A typical WSN architecture.*

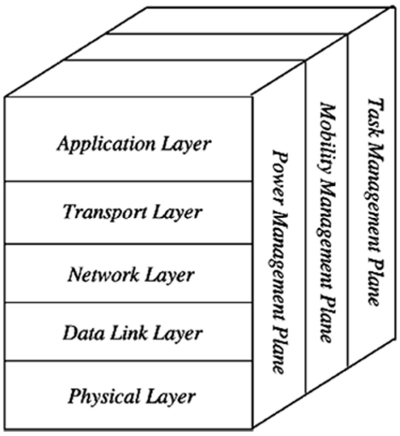

Figure 2. *Wireless Sensor Network protocol stack.*

In order to support a wide variety of applications, WSN platforms rely upon existing hardware and software components and may and may regularly utilizing some cross-layer optimizations. Typically, applications with particular necessities, can motivate custom design and optimization of the sensor nodes and the entire WSN platform. Such requirements are a substantial number of nodes, low cost and fast field deployment. As depicted earlier, the sensor nodes have many constrains such as, low computational power, limited energy and bandwidth. These constraints influence the deployment of a large number of sensor nodes that have raised many challenges to the management and design of wireless sensor networks. Generally, depending on the processing power and storage, the researchers have classified wireless node platform architectures into three categories [3]:

a. lightweight-class platforms: these platforms have small storage, low processing power capability and are generally furnished with fundamental communication components. Generally, these motes expend a very low amount of power. An example of a lightweight-class mote platform is the FireFly.
b. intermediate-class platforms: compared to the lightweight platforms, these platforms have better processing power and a more memory storage but at the same time are normally furnished with basic communications components. TelosB is an example of intermediate-class mote platform.
c. PDA-class platforms: these platforms usually used to process multimedia data which requires powerful processing capability and vast memory storage. Typically, these platforms support different operating systems and multiple radios. One drawback is that they consume more power. Stargate platform is An example of a PDA-class mote platform.

As shown in Figure 1, A typical node consists of four main components: The sensing unit, The processing unit, The communication unit and The power unit. In most configurations, the ADC, processor, memory and radio are integrated into a single system on a chip (SoC).

a. Sensing Unit: This unit consists of sensor nodes which responsible of sensing and collecting specific data of the area or environment. These sensors are typically low-powered passive devices. The analog sensed data collected is usually amplified before digitization by an analog-to-digital converter (ADC). Usually, the processor unit or the sensor package can perform these operations (amplification and digitization). However, choosing sensor package to perform these operations is preferred as the amplifier, filter and ADC can be better matched to the capabilities of each sensor and low-level analog signals are kept within the sensor unit which minimizes possible external interference and loss.

b. Processor Unit: At least, the WSN processor unit consists one very low-powered processing core and local memory. Basically, the functionality of processing unit, is to process data received from sensing unit, schedule tasks, execute the algorithms for data forwarding and control the functionality of other components in the sensor node. Generally, processor architectures are arranged into three classes: GPPs, DSPs and application-specific instruction set processors (ASIPs). Typically, a WSN mote contains at least one sufficient GPP as a basic processor for making for making straightforward computations, serialization and packetisation of data for transmission. It is also responsible for decoding and interpreting incoming messages. Processors those based on the ARM and MIPS architectures and the Atmel AVR are some examples of GPP processors. In contrast, DSPs are specialized processor architectures dedicated to implement complex processing algorithms for multimedia applications. In general, DSPs can use a floating-point or a fixed-point arithmetic format. For WSNs to be more energy efficient, DSPs would use a fixed-point format and are often optimized to perform operations such as fast Fourier transform (FFT) or digital filtering operations. Although GPPs and DSPs support a wide range of applications, ASIPs are optimized to support a single application and are often implemented by customizing the instruction set architecture (ISA) of a GPPs processors. Moreover, ASIPs are more energy efficient with a high computational performance than a GPP processor. Such examples of ASIPs processors customized for WSNs are for applications involving error correction, encryption and compression.

c. Communication Unit: This unit takes care of transmitting and receiving from/to the sensor node. For wireless communications, it is most regular to utilize radio frequency (RF) waves as communication method. Three common standards can be used for RF communication: IEEE 802.11 wireless LAN (normally termed Wi-Fi), Bluetooth and IEEE 802.15.4 (normally termed ZigBee^{TM}). ZigBee is a new wireless network technology with low-cost, low-rate, low-energy consumption and short distance transmission [4]. Table 1 lists a comparison of the RF communication standards.

d. Power Unit: This unit is responsible of managing power from the node power source. The node power source is typically a cell and can be increased with an energy-harvesting system from other potential sources such as light, vibration, heat and radio frequency signal. A cell is typically known by its chemistry type such as nickel metal hydride (NiMH), nickel cadmium (NiCd), zinc-carbon or alkaline(zinc-manganese dioxide) and lithium ion polymer (Li-ion polymer). Also, the power unit must be able to regulate the charging of the cell in systems consolidates both a cell and an energy-harvesting system.

Table 1. *A comparison of the RF communication standards.*

Market name	Wi-Fi ^{TM}	Bluetooth ^{TM}	ZigBee ^{TM}
Underlying standard	802.11b	802.15.1	802.15.4
Application focus	Web, email, video	Cable replacement	Monitoring & control
Battery life (days)	0.5-5	1-7	100-1,000+
Network size	32	16	100s to 1,000s
Bandwidth (k bits/s)	11,000+	720	20 - 250
Range m	1 - 30+	1 - 10+	1 - 1,000+
Network architecture	Star	Star	Mesh
Optimized for	Speed	Low-cost convenience	Reliability, low power, low cost, scalability

The rest of this paper is organized as follows: section 2 presents an introduction to the VoWSN system, section 3 describe the structure of VoWSN system, section 4 list the challenges related to the VoWSN system design, section 5 gives a comprehensive literature review related to the subject of VoWSN and section 6 provides conclusions and discussion.

2. Voice over Wireless Sensor Network (VoWSN)

Compared with traditional centralized sensing systems, the sensor nodes are very easy to deploy, and can be deployed very densely(hundreds or thousands). It is obvious that when multiple sensors work together, the information redundancy among the sensors and the redundant communication channels in the networks enable a better sensing coverage and provide more reliable information delivery [5] [6]. Moreover, the low-power consumption made WSNs very attractive for a broad variety of applications such as habitat monitoring, environmental monitoring, surveillance and emergency scenarios and much more.

The vast majority of the applications for WSN concentrated on simple sensing and reporting activities such as measuring temperature, humidity, or location of objects. In general, the main features of these applications are low bandwidth demands, and were usually delay tolerant. With the rapid improvement of WSN technology, there is a growing need to utilize this technique evolved towards more complex applications such as multimedia applications. As consequence, this led to emerge a new technology of WSN referred to as Wireless Multimedia Sensor Network(WMSN). WMSN is a special type of WSN which can sense and transfer multimedia data and also scalar data in real-time and non real-time [7]. In WMSNs nodes are interfaced with sensors for visual and sound data acquisition, such as Complementary Metal Oxide Semiconductor(CMOS) cameras and microphones. The emergence of this technology has drawn the consideration of the WSN research community towards exploiting it to support multimedia based applications such as video surveillance, environmental monitoring, target tracking and traffic management [6].

Compared to video camera, audio sensors(microphones) are cheap, easy to deploy, energy efficient. Moreover, the main property of audio sensors are that they don't rely upon a Line-Of-Sight(NLOS) which allow for omnidirectional sensing [8]. The property of NLOS in acoustic sensing give many advantages over the video camera like the independency from sensor position(not limited to a certain viewing angle), and easy to include many sensors into the environment to have a full coverage area of interest [8]. Furthermore, audio sensors are multipurpose tools. In addition to its ability to catch certain environmental information through sound, it also allows for the definition of specific events within the audio stream if the sensor node is equipped with a reasonable amount of processing power. The intelligent sensing capability of audio sensors together with communication capabilities makes them very interesting devices for a wide range of applications, where multimedia services are required.

Voice transmission is one of the multimedia applications that are recently being interested in the area of WSN [9]. The transmission of audio signals throughout a wireless network in which every its node is capable of sensing and interpreting events and processing jobs collaboratively with other nodes is called VoWSN [10]. With VoWSN, voice data captured with an event trigger can be transferred in an automated way. The advantages of VoWSN approach could be exceptionally profitable. Furthermore, when comparing a VoWSN, with the classic COTS analogical audio surveillance systems, VoWSNs could ensure more inescapability, reduced power utilization, and ability to perform local signal processing to enhance the communication quality as well as to reduce the bandwidth [11]. Recently, in order to accomplish all these potential advantages, the researchers efforts have been targeted toward the attempt to join the WSN world and the audio processing issues. With the ability of audio sensors of analyzing the audio signal and performing data signal processing locally, possible scenarios can be achieved like, secure ubiquitous home environment, emergency situations (people asking help), short- term intruder detection, voice activity detector, identify individual speakers, discriminate human speech and music, simple gender classifier and so on [12].

In this paper we attempt to take the topic of VoWSN in its all aspects. We discuss the state of the art and the major research challenges in architectures, and requirements for VoWSNs. Furthermore, This study presents a literature overview concerning the feasibility of designing and implementing of VoWSNs system and address the problems of supporting voice over WSNs despite nodes only have limited communication and processing capabilities.

3. Description of VoWSN Framework

3.1. Architecture of VoWSN System

For voice transmission system, the WSN shown in Figure 3 is depicted. This system include two types of sensor nodes; the first are the sensor nodes equipped with an auxiliary microcontroller and an audio sensor, and the others are sensor nodes that acting as simple routing sensors. As shown in Figure 3, the gap between the audio sensor node to the base station or sink node would be deployed with routing nodes to gather information through multi-hop routing [10]. The audio sensor nodes are configured to detected audio signals from their surrounding environment and then sampling these signals. After audio data acquired by the sensor nodes, the signals then transmitted back through the routing nodes to a high performance processing platform for further processing. This platform may be a single board computer.

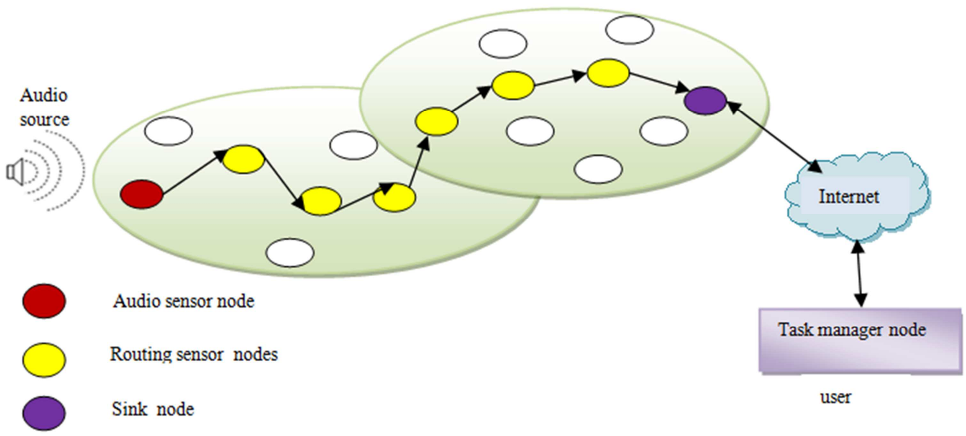

Figure 3. *Reference model of VoWSN for voice communication.*

As an example, the wireless communication system of the VoWSN based on ZigBee is shown in Figure 4 [13].

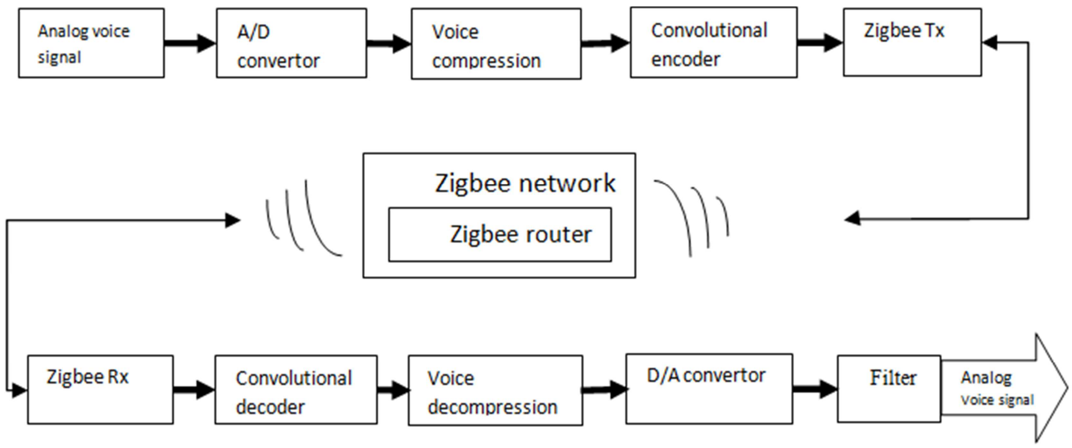

Figure 4. *VoWSN system functions block.*

Basically, ZigBee is designed for data communication and not for voice communication. However, its 250kbps bandwidth is sufficient to help the voice communication [4]. VoWSN solution based on ZigBee combines a low power microcontroller, wireless radio frequency(RF) transceiver and audio codec which integrate operational amplifier, power amplifier and filter. The data of voice signal is real-time, orderly and the data volume is big and is unable to realize transmission straightforwardly in the ZigBee network. Based on these characteristics, the transmitting end must realize the compression and sequencing at first, and operate the storage, packing (conforms to the ZigBee protocol), then transmit the data in the network through the RF modulation and the ZigBee route. In the receiving end, the opposite operation executed in order to get the original analog voice signal [13].

3.2. Structure of a VoWSN Node

Over the past few years numerous scientific and commercial node platforms have been proposed for use in VoWSNs. The node model shown in Figure 1 is also suitable to describe a node of a VoWSN. The limited resources of conventional WSN hardware platforms makes it difficult to use them in multimedia applications which require more resources in term of processing capabilities, memory, energy and bandwidth. Consequently, VoWSN node architecture may vary from that of conventional WSN node by adding some external devices such as external vocoder and audio sensor module. Also, the processing and communication units may differ from the conventional WSN in terms of processing capabilities, transmission bandwidth and so on.

However, the emergence of VoWSNs prompted change the function of the processor from its basic function to one that has to process more complex audio data such as to pre-process and compress the data from the audio sensors before transmission. Keeping in mind the end goal to accomplish these prerequisites, an alternate sort of low-power processing core with more processing power is required. Furthermore, when choosing the processing core other considerations have to be taken into account as it has enough I/O interface such as SPI (Serial Peripheral Interface Bus) and I2C (Inter-Integrated Circuit) or 1-wire. This interfaces are required for connecting various components of the system such as the external memory, ADCs and radio [3]. Generally, voice streaming applications require the utilization of compression algorithms to reduce the bandwidth. However, when using these data compression algorithms, the processing unit of sensor node requires more power to run the complex compression algorithms, which need lots of memory and time. So, an efficient data compression algorithm by maintaining the balance between the computational cost and the power saving from compression ratio is required. There are many audio compression techniques, such as ADPCM (Adaptive Differential Pulse Code Modulation), MP3 (Moving Picture Experts Group Audio Layer-3) and Ogg Vorbis [14]. As consequence, it is possible to use of an external vocoder to ease the burden on the precise microcontroller of carrying out audio processing and streaming operations [15].

As consequence to rapid advancements and scaling down in hardware, a single sensor device can be combined with audio collection modules such as equipping audio sensors with a host mote like Crossbow's MICA2 or MICAz [3]. The typical audio sensor module comprises of a microphone, amplifier, filters and an ADC. These audio sensors have to be low powered and small in size. There are two types of audio sensors which are suited for VoWSNs: Electret Condenser Microphone (ECM), or the Micro Electrical-Mechanical System (MEMS) microphone.

The ECM is a type of condenser microphone with a permanently charged electrets material fixed to the capsule back-plate. ECM microphones provide a good dynamic range and are found in many applications from high-quality recording to built-in microphones in telephones. ECM microphone is low cost, widely available and small in size making it suitable to be embedded into many applications. By contrast, the MEMS microphone is a solid state microphone that typically integrated with a matching ADC, power management, filter, hardware control and communication port. However, MEMS microphone has many advantages over the ECM microphone such as [3]:

a. Easy to be interfaced to the node processor utilizing a digital serial link.
b. MEMS microphones can accomplish better performance than equivalent ECMs in noisy environment.
c. More uniform part-to-part frequency response than ECMs(A random selection of MEMS microphones of the same type will have close indistinguishable response).

4. Challenges and Problems Related to Voice over WSN

As depicted in section 1, the main functions of a node in WSN are: sensing, computation, and communication. To realize voice based WSN, wireless sensor nodes should be able of catching voice, processing it and transmitting it over RF channel. Depending on the application, for the three main functions of a node in a VoWSNs system, nearly 80% of the power consumed is used in the transmission. In order to extend the lifetime of battery-powered sensors, communication needs to be minimized as much as possible [16]. Another challenge is that audio signal processing requires a large amount of memory for performing computations accurately and this is difficult for resource-constrained sensor nodes. Moreover, frequent high-speed real-time transmission of audio data produced significant challenges for WSN radio channel management, energy management and protocol design [17].

The main restrictions to use WSN in acoustical applications that must be taken into account when designing the system can be summarized in the following points [18]:

a. Bandwidth: The scarce communication bandwidth greatly limits data transmission between a sensor node and the processing center. WMSNs applications require a bandwidth that is higher than that supported by currently traditional sensors. The bandwidth bottleneck requires must carefully balancing the overhead of communication and computation.
b. Power: Multimedia applications characterized by high volumes of audio data, high transmission rates and extensive processing. This led to make WMSNs more power consuming than traditional WSNs. The sensor nodes are not connected to any wired energy sources. They are usually powered by battery but are expected to work for several months or even for years without recharging. So, minimizing the energy consumption is always a key concern during WMSNs operations.
c. In-node processing support: For VoWSN node, the role of an embedded processor is to compress data, signal analysis and features extraction and much more DSP functions. Due to the small size, current sensor nodes provide only very limited processing power. The limited processing capability renders existing voice processing algorithms hard to implement on individual sensors.
d. Cross-layer design: A successful enhancement of the considerable number of parameters include cross-layer protocol design going from Application to Physical Layer..
e. Hardware Constraints: Every sensor node, at least, needs to have a sensing unit, a processing unit, a transmission unit and a power supply unit. However, every additional operations related to multimedia application would increase cost, power consumption and physical size of the node. Thus, the tradeoff between extra functionality and cost and low-power requirements is always needed.
f. Limited storage space: Memories in a sensor node normally incorporate in-chip flash memory, RAM of a microcontroller and external flash memory. However, WSN has a limited storage space. With such a limited space, the audio processing algorithms implemented over sensor node should have not only low computational complexity but also low memory requirements.

To overcome the above restrictions, a great deal of effort has been spent towards enabling efficient VoWSN system implementation on existing WSN products platforms. Section 5 gives a comprehensive overview of the interest researches that try to optimize VoWSN over the last few years.

5. Literature Review

Researchers have started to exploit the versatility of WSNs with applications relating to acoustic signals. Rapidly, VoWSNs applications are gaining interest by researchers to develop new services. Later the researches evolved tried to resolve some problems addressed by implementing VoWSNs. In general, depending on the applications, research orientations of VoWSN can be categorized into two domains: voice communication system and Ambient Intelligence as shown in Figure 5.

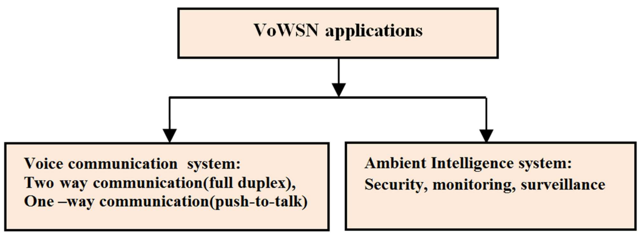

Figure 5. *VoWSN research orientation diagram.*

5.1. Voice Communication System

Voice streaming can be defined as the practice of delivering real-time voice through a network connection. This type of data transmission requires certain protocols for taking care of the order of data packets or other transmission types, to provide the end-user with on-demand content [19]. In general, voice streaming requires a buffering framework, a secure data stream platform and furthermore requires significant bandwidth to permit end users to listen to full audio files without interruption.

Different options to accomplish wireless voice communication system has developed for many years. A few of them are interphone, digital cordless, mobile telephone, Wi-Fi (VoIP), microwave, and satellite. These technologies have their disadvantages such as high cost, big power loss, and infrastructure based [20]. New approach for voice communication can be contributed by using WSN. Comparing with traditional infrastructure-based call system, Voice call WSN is infrastructure-less system [21]. A key advantage of Infrastructure-less nature of WSNs generally could offer many valuable advantages in reducing thecost and time associated of installing the infrastructure and its maintenance. Typically, sensor nodes are conveyed randomly, and are expected to self-organize to form a multi-hop network.

In some circumstance, the wireless voice communication is generally required, such as disaster zones and medical emergencies. Disasters pose challenges in ensuring reliable communications with a reasonable cost. Previously, before the advent of temporary GSM base stations, rescue teams have to deploy and handle VHF radios as well as high-cost satellite connections without packet-based local connectivity between the members [22]. In this environments VoWSNs system can make significant contributions in situational awareness by improving rescuers communication, and reacting to status changes, across the entire disaster. Usually when a disaster occurs, communications are significantly affected. Consequently, the rescuers might be unable to coordinate between each other. Thus, for this scenario, it is possible for rescuers to easily deploy low cost and low maintenance wireless motes throughout the disaster zone. However, several motes would suffer from loss or destruction and the others will be able to survive which can be used to enable communications between survivors and rescuers. The low-power features of the sensor nodes can help to guarantee the maximum life-time during the rescue mission. Also, their low cost feature makes their loss or destruction cost effective [23] [24].

Voice communication system based WSN can be categorized by two scenarios: two-way (full duplex) and one-way (push-to-talk (PTT) or half duplex). Initial researchers efforts were mainly focused on feasibility of implementing voice communication system based WSN (one of telecommunication value-added services) which falls under the streaming applications [25]), also, identify the problems to be solved related to combining voice communication with traditional WSN, and analyze several performance criteria.

The early attempt to implement a system for wireless voice communication over wireless sensor networks is developed utilizing the FireFly devices. Firefly is a low-cost wireless sensor network-based rescue device used in coal mine developed at Carnegie Mellon University [26]. Mangharam et al., built up a framework to support the transfer of one (two-way)voice stream based on the Fire-Fly nodes. Firefly nodes operate in a global TDMA schedule. Nonetheless, TDMA often causes a high overhead and performs ineffectively in response to network dynamics such as variation of traffic load and link/topology quality. They chose the Adaptive Differential Pulse Code Modulation (ADPCM) as a compression technique to reduce the data rate. The software compression is implemented on 8-bit fixed-point processor [27].

C. Wang et al., discussed the feasibility of supporting voice communications over ZigBee networks. Through simulation, they provided an evaluation of voice quality and performance of two types of voice communications: full-duplex voice over IP (VoIP) and half-duplex push-to-talk (PTT). They considered two metrics for measuring voice quality: the R-factor (a well known objective speech quality metric) for VoIP and packet-loss rate, delay, and jitter for PTT [28].

D. Brunelliet al., investigated the feasibility of ZigBee networks for low-rate voice streaming applications and proposed some improvements in the stack implementation then they analyzed streaming metrics such as throughput, packet loss and jitter in multi-hop topologies. Audio data is coded using 8bit A-law conversion, performed using an external ADPCM processor [29].

L. Teodoran et al., discussed the achievability of implementing a ZigBee Push-To-Talk application using commercial off-the-shelf (COTS) hardware. They analyzed important streaming metrics such as packet loss and evaluating Unicast and Broadcast addressing. They also analyzed the trade-off between audio quality and power consumption comparing different compression algorithms, which implemented using a programmable 8-bit processor. A programmable ADPCM processor provided using low-power audio CODECs where voice is sampled at 8KHz and encoded using 8 bit A-law [30].

M. Petracca, et al., analyzed the possibility of sending voice using WSN. They investigated the enhancement of speech quality by protecting the most perceptually important packets. They first evaluated the speech quality for a modified version of the ITU-T G.711 standard implemented by 16-bit microcontroller to fit the particular selected hardware. According to the MOS scale, their experimental results showed that the combination of the proposed perceptual marking algorithm and the selected hardware achieves good speech quality levels [23].

R. Alesii, et al., analyzed the key issues that must be taken into account to transfer voice over WSN and presented a project for an audio surveillance system. The system is assumed to be implemented by MicaZ wireless sensor nodes. They chose the Interactive Multimedia Association-Adaptive Differential Pulse Code Modulation (IMA-ADPCM) as a compression technique to reduce the data rate. The software compression is implemented on 8-bit fixed-point processor [11].

H. Rong-lin, et al., presented the architecture of the VoWSN and the hardware designing of Voice Node and Gateway Node. Different scenarios for Voice quality were tested such as indoor, corridor, corner and outdoor. They showed that good voice quality can be obtained in short distance or line of sight with no obstacle. ADPCM encoding was performed by digital signal processing circuit [21].

S. Kumar et al., proposed a MAC protocol for a streaming voice application with maximum data rate in a multi-hop path. They suggested adapting the frame time and avoiding the collisions to the streaming nodes by the neighboring nodes to achieve maximum data rate. The simulation results showed the possibility of transfer compressed voice across a large distance through multi-hop route. However, an explicit battery model has not included in the simulation model to account for energy constraints of the WSN nodes [31].

J. A. Kang et al., proposed an adaptive redundant speech transmission (ARST) approach for improving the perceived speech quality (PSQ) of speech streaming applications over wireless multimedia sensor networks (WMSNs). The suggested algorithm estimated the PSQ and the packet loss rate (PLR) from the received speech data. The adaptive multirate-narrowband (AMR-NB) speech codec and ITU-T Recommendation P.563 was used as a scalable speech codec [32].

O. Turkes, et al., investigated the quality of voice signals sent through the homogeneously multi-hop network. The basic characteristics of the voice samples used in the testbed are tuned with different in-network qualifications; so that they tried to propose a reasonable trade-off between voice quality and sensor network capabilities [10].

L. Herrera, et al., analyzed the performance of audio streaming over a multi-hop Low-Rate Wireless Personal Area Network. They investigated the feasibility of providing a bidirectional audio communication using a commercial IEEE 802.15.4 compliant platform. The research included a performance evaluation of different compression algorithms and software protocols to support voice transmission on a CC2530 System-on-Chip mote. The experimental results successfully achieved a maximum number of hops of a bidirectional single-route network under real-time voice quality constraints. The voice compression performed by the an external vocoder [33].

A. W. Rohankar, et al., described a real-time implementation of voice communication system using 802.15.4 radios. The main features of the proposed system were, Silence detection and soft ADPCM. It was shown that silence detection improves audio communication and performance bandwidth optimization over low bit-rate radios [34].

J. Kim, et al., designed a full-duplex TDMA/TDD voice mixer based on IEEE802.15.4 standard for multi-user VoSN system as group communication. They described the requirements of implementation the system such as MAC protocol, hardware configuration and synchronization among multiple users. The speech codec inside the VoWSN device was used for compressing voice data to 8 kbps with G.729a algorithm [35].

M. Petracca, et al., presented an algorithm for the perceptual selection of voice data to reduce the speech transmission bandwidth with a reasonable speech quality. they proposed a voice data protection technique based on speech perceptual importance. The ITU-T G.711 compression standard selected to be software performed by the MCU [36].

A. A. Prabahar, described how to make high performance VoWSN nodes using Digital Signal Processing (DSP) and how to implement speech processing algorithm on it. He introduced techniques to reduced high power consumption produces from combining DSP chip with WSN node. Customized algorithms for speech compression were performed by TMS320C5515 DSP chip [37].

D. Hollosi, et al., presented research implemented within the EU FP7 EAR-IT project. The project is working on the challenges of bringing acoustic sensing intelligence into two existing testbeds out of the EU FP7 FIRE projects Smart Santander and Hobnet. The experimental and simulation results for acoustic source localization utilizing audio sensing technology for WSN demonstrate that using audio method enables a wide range of applications and services with high social and technological value [8].

A. Geetha, designed a smart helmet as a mobile sensor node of ZigBee WSN, to collect environmental parameters such as temperature, humidity and illumination level of underground environment. Also a voice transmission system, based on the same low-rate ZigBee networks was designed. Using this system, the miners can communicate with control centers or with each other. G.726 voice codec algorithm which performed by the MCU, is used for compressing the speech signals which is a waveform speech coder uses ADPCM [38].

Y. Fu, et al., presented a wireless audio sensor network platform A-LNT. A systematic design and implementation of several elements were discussed such as, node hardware design, voice codec, network topology, hybrid MAC protocol design, address filtering, power management, radio channel allocation and clock synchronization [17].

H. Yoon, et al., presented a Voice over Sensor Networks (VoSN) system to be utilized for long distance communications. However, ZigBee has not been considered to interface with the long distance communications. The authors suggested a network configuration method to multi user and long distance telecommunication in WSN and measured the MOS value for voice quality [13].

Kh. Sahal, et al., proposed an integrated approach to be utilized for transmitting high quality voice data using low-cost 8-bit microcontroller and Off the Shelf components [39]. They aimed to develop a voice compression algorithm also comparable to ADPCM (Adaptive Differential Pulse Code Modulation), but less mathematically expensive to fit in low memory profile 8 bit microcontroller to achieve moderate voice quality which later achieved in [40].

V. Dubey, et al., presented the implementation of voice transmission system based on firefly nodes with high throughput and reduced overheads. Also, the author described the WSN QoS parameters within certain functional layers within a WSN application [9].

N. Preetha, et al., proposed system a zigbee network to be used for the transmission and reception of voice signals and forming a full duplex transmission. The zigbee used for transmission of voice signals was zigbee TR24A which supports voice communication and low cost. They used two zigbee modules one for the transmission and another for the reception. The voice signals from the microphone is processed by the microcontroller and transmitted through the zigbee. The receiver zigbee will receive the signal and processed by the receiver's controller and will be played in the speaker. By this proposed system they could enhance the security of the voice communication over wireless medium [41].

Table 2 summarized the surveyed Papers Research field.

Table 2. *The survey of Papers Research field.*

Ref. No.	Research field	Research methodology		Hardware
		HW	Sim.	
[27]	Implementing transfer of one (two-way) voice stream and investigating TDMA based network scheduling	√		Fire-Fly nodes
[28]	Investigation feasibility of voice communications over ZigBee full-duplex (VoIP) and half-duplex (PTT)		√	--------
[29]	Describing the audio streaming capability of a Zigbee-like network over the IEEE 802.15.4 framework	√		SoC CC2430+TLV320AIC1107 codec
[30]	Investigation the feasibility of transmitting voice over ZigBee compliant platforms using (COTS) hardware	√		SoC CC2430+ audio CODEC
[23]	Enhancing speech quality by protecting the most perceptually important packets	√		UTMOST platform MSP430F1611
[11]	Analyzing the fundamental issues to gain voice from a wireless sensor network (WSN)	√		MicaZ wireless sensor nodes
[21]	Presenting the architecture of the VoWSN and describing the hardware designing of Voice Node and Gateway Node	√		DSP processor
[31]	Presenting a MAC protocol for a streaming voice application		√	-----------
[32]	Proposing an adaptive redundant speech transmission (ARST) approach to improve perceived speech quality	√	√	-----------
[10]	Investigating the quality of voice signals sent through the homogeneously constructed multi-hop network	√		MSP430FG4618 microcontroller+ TMote Sky sensor
[33]	Evaluating performance of a bidirectional audio communication using a commercial IEEE 802.15.4 compliant platform	√		CC2530 SOC WSN mote
[34]	Implementing real-time audio communication using 802.15.4 radios	√		Jennic-5139 MCU kit
[35]	Designing a full-duplex TDMA/TDD voice mixer based on IEEE802.15.4 for multi-user VoSN system	√		ATmega128L MCU +CC2431 transceiver
[36]	Presenting an algorithm for perceptual selection of voice data to reduce the speech flow bandwidth	√		MCU
[37]	Designing a WSN node for the purpose of voice based applications.	√		TMS320C5515 DSP chip
[8]	Presenting research carried out within the EU FP7 EAR-IT project, to bring acoustic sensing intelligence to large-scale WSNs	√	√	Testbeds out of the EU FP7 FIRE projects and Hobnet
[38]	designing a smart helmet, as a mobile sensor node of ZigBee WSN for gathering parameters and voice transmission system	√		MCU PIC24FJ64GB106+codec Si3000
[17]	Presenting the key elements of implementing node hardware, voice codec, hybrid MAC protocol for an audio WSN platform A-LNT	√	√	MCU MSP430F2618+CC RF transceiver
[13]	Designing VoSN considered on long distance communications	√		DSP+ voice codec+CC2431
[39]	Proposing an integrated approach towards Transmission of High Quality of Voice Data using 8-bit Microcontroller	√		8-bit MCU
[9]	Implementing of voice transmission using firefly nodes	√		firefly nodes
[41]	Proposing a zigbee network for full duplex voice communication	√		PIC16f877 MCU

Table 2. *Continued.*

Ref. No.	Compression technique	Problems investigated	Results
[27]	software compression (ADPCM)	TDMA incurs a high overhead, performs poorly in response to network dynamics	Providing efficient support for applications with timing constraints
[28]	---------	Voice quality measuring by using the R-factor, packet-loss rate, delay, and jitter	WSN can support a limited range of voice services
[29]	External ADPCM processor	Measuring streaming metrics such as throughput, packet loss and jitter	Effective streaming over Zigbee are related to hardware improvements and to overcome drawbacks of the standard specifications
[30]	External ADPCM processor	Analyzing metrics such as packet loss, evaluating Unicast and Broadcast addressing	Trade-off between audio quality and power consumption comparing different compression algorithms
[23]	Software ITU-T G.711 standard	Evaluating perceptual importance of speech packets to perform single-hop & multi-hop data collection	Combination of the selected hardware and the proposed perceptual algorithm achieves good speech quality levels, according to the MOS

Ref. No.	Compression technique	Problems investigated	Results
[11]	IMA-ADPCM software compression	Multi-hop routing, using an 8,6 kHz sampling instead of 8 kHz	use of new versions of Tiny OS, saving energy, realizing an AGC software and improving the quality of the acquired signal.
[21]	Software ADPCM	Measuring voice quality of different scenarios indoor, corridor, corner and outdoor	Good voice quality can be obtained in short distance or line of sight with no obstacle
[31]	------------	Adapting the frame time and avoiding the collisions to the streaming nodes by the neighboring nodes	Possibility of compressed voice streaming across a large distance through multi-hop route
[32]	AMR-NB speech codec and ITU-T Recommendation P. 563	Estimating the PSQ and packet loss rate (PLR) from the received speech data	The ARST approach improves speech quality under packet loss conditions in WMSNs
[10]	-------------	Real time constrains while recording and transmitting voice data in segmented information packets	Aiming to assign different priorities to segments in order to satisfy expected quality of service. These inherent properties can be the key instruments in voice quality evaluation
[33]	External vocoder	Demonstrate different software protocols and compression algorithms for audio transmission	Establishing the maximum number of hops of a bidirectional single-route network under real- time voice quality constraints
[34]	Software ADPCM	Silence detection, soft ADPCM and data synchronization	Silence detection improves B.W. optimization performance over low bit-rate radios
[35]	The speech codec used with G.729a algorithm	Configuration of Hardware, MAC, and synchronization among multiple users	Expecting that a variety of applications can be developed using presented system
[36]	Software ITU-T G.711 compression standard	speech flow bandwidth and end-to-end speech quality	The perceptual selection algorithm could reach a 30.8% reduction in bandwidth occupancy
[37]	Software compression	Reduce the power consumption of DSP chip	Nodes based on DSP chip are suitable for acoustic applications based zigbee
[8]	------------	------------	A more intelligent usage of the audio modality enables a wide range of applications and services with high social and technological value
[38]	ADPCM compression by Si3000 voice codec	Integrating wireless speech communication system with environmental monitoring system	Reducing cost and reliable system with quick and easy installation
[17]	PCM algorithm by external CMX469 codec chip	Energy management methods such as address filtering and efficient power management	A-LNT is a light weight, low-power, low-speed, and high-performance WSN platform for multichannel real-time voice communications
[13]	External G.729 codec	A network configuration of multi user and long distance telecommunication & MOS value for quality	Good performance voice communication system and a long distance communication using WiFi
[39]	Software ADPCM	Using low cost 8 bit MCU and development bit swapping protocol to communicate 8 bit to 16 bit resolution over 8 bit ASCII	Development of a voice compression algorithm comparable to ADPCM, but less mathematically expensive to achieve moderate voice
[9]		WSN QoS parameters within certain functional layers	giving a model for voice transmission that has high throughput and reduced overheads
[41]		Implementing the transmission using SPI protocol to overcome delay in received signal	Enhancing the security of the voice communication over wireless medium

5.2. Ambient Intelligence

The term 'ambient intelligence' refers to an intelligent environment that is aware of the presence of a user and that is receptive to its needs. Persons are surrounded by a many distributed networks comprising sensors, computational devices and electronics [42]. Ambient Intelligence (AmI) applications need information about the surrounding environment which can be gathered using WSN [43]. Active communication between the user and the environment would based on automatic speech recognition (ASR) procedure. In ASR system the signal enhancement is very important because the background noise is of significant affect on the recognition performance [44]. Ambient intelligence (AmI) research orientations build upon advances in sensors and sensor networks, pervasive computing, and artificial intelligence [42].

Integrating speech recognition into a WSN system produced a new type of applications that have distributed configurations such as smart home automation, health care system, security and so on. Although, the speech recognition technology has been developed into advanced stages and assumed to has its own practical products, the speech recognition under the embedded system does not develop as fast as expected [45]. The speech recognition process of embedded system is realized by some research, but still has a lot of aspects which needs to be improved. However, a few research endeavors have integrated speech recognition technology into WSN systems and explored general framework practicality.

The earliest attempt of implementing speech recognition system on embedded resource constrains systems introduced by S. Phadke, et al.. They combines the aspects of both hardware and software design of implementing a speaker dependent, isolated word speech recognition system. They used modified Mel-scaled Frequency Cepstral Coefficients (MFCC) as feature extraction method and Dynamic Time Warping (DTW) as template matching process. The hardware was built around the industry standard TM320LF2407A DSP [46].

V. Berisha, et al., proposed a voice activity detector and a simple gender classifier algorithm. The algorithm has suggested to be used in a distributed acoustic sensing system. The algorithm made use of low-complexity audio features and a pre-trained regression tree to classify incoming speech by gender. The algorithm was implemented real-time on the MICAZ motes and the MTS310CA sensor board from Crossbow [47].

L. E. Palafox, presented a prototype voice capture application by using a wireless sensor network based on UC Berkley's MICAZ motes. The hardware and energy limitation represented an important obstacle in the development of the prototype implementation. A significant energy consumption savings achieved, thus, extending network lifetime [48]. Also, the authors in [49] extended their work towards a secure ubiquitous home environment. The approach depended on reducing data redundancy while keeping up node redundancy for reliability and providing at the same time basic security services without sacrificing a considerable amount of resources.

C. Shen, et al., presented the design and implementation of a distributed sensor network application for embedded, isolated-word, real-time speech recognition system. They adopted a parameterized-dataflow-based modeling approach to model the functionalities associated with sensing and processing of acoustic data. The associated embedded software implemented on an off-the-shelf sensor node platform (Texas Instrument/Chipcon CC2430 devices as the main processor and transceiver on all sensor nodes) that is equipped with an acoustic sensor and a TDMA access protocol was developed to manage the wireless channel [50].

F. Sutton, et al., demonstrated the implementation of a prototype architecture for automatic single word speech recognition on resource-constrained embedded devices. The experiments results showed that the prototype achieved a high average detection rate of 96%, while only dissipating 28.5 mW for continuous audio sampling and duty-cycled speech recognition. ARM 32-bit Cortex-M4 microcontroller was used to built the prototype. Audio signal acquisition was performed using a dedicated audio codec connected to a microphone [51].

Th. Soundhari, et al., surveyed hand-held devices equipped with Android app in WSN. The Speech recognition process was done using a commercial speech recognition engine (SVM Classifier). The speech recognition system would act as a hub within the Home to receive and route user commands to different control systems [52].

R. P. Raghava, et al., presented the design and implementation automatic single word speech recognition system on embedded devices. The words which are spell are stored in the operating system of Raspbian OS which is implemented on Raspberry Pi hardware kit of ARM 11 processor [53].

N. Fadzilah, et al., presented the development of a low cost remote home appliances control system based on speech recognition technique. The system focused on controlling fan and lamp wirelessly using Arduino Uno (Arduino UNO is a multi-purpose microcontroller board based on the ATmega328P) as the controller [54].

G. G. tolya, et al., analyzed ASR performance on utterances recorded by means of wireless sensors. The sound quality of utterances recorded by such sensors contrasts fundamentally from that of the larger audio data bases usually used for acoustic DNN (Deep Neural Network) training due to the small microphone installed on these devices. They could accomplish a5% improvement in terms of relative error reduction. They used Crossbow Iris MTS300 sensor board which equipped with a microphone [55].

Table 3 summarized the surveyed papers Research field.

Table 3. *The survey of papers Research field.*

Ref. No.#	Research field	Research methodology		Hardware	Results
		HW	Sim.		
[46]	Presenting a design of an Embedded Speech Recognition System	√	√	TMS320LF2407A DSP	Achieving high-speed recognition with maximum accuracy in minimum power and making the device portable
[47]	Proposing a voice activity detector and a simple gender classifier for use in a distributed acoustic sensing system	√		MICAZ motes and the MTS310CA sensor board from Crossbow	The algorithm makes used of low-complexity audio features extraction and a pre-trained regression tree. The algorithm is implemented in real time
[48] [49]	Presenting a prototype voice capture application and secure ubiquitous home environment by using a WSN	√		UC Berkley's MICAZ motes	The approach is energy-efficient compared to the capture-send approach used traditionally
[50]	Presenting the design and implementation of a distributed sensor network application for embedded, isolated-word, real-time speech recognition	√		Texas Instrument / Chipcon CC2430	Improving energy efficiency by distribute the overall computation workload across the network

Ref. No.#	Research field	Research methodology HW	Sim.	Hardware	Results
[51]	Demonstrating the design and implementation of a prototype hardware/software architecture for automatic single word speech recognition	√		ARM 32-bit Cortex-M4 microcontroller	The prototype achieve a high average detection rate of 96%, dissipating 28.5 mW for continuous audio sampling and duty-cycled speech recognition
[52]	Surveying hand-held devices equipped with Android app in WSN which allows controlling of devices using a Voice commands	√	√	Commercial speech recognition engine (SVM Classifier)	The speech recognition system act as a hub within the Home to receive and delegate user commands to Different switching and control systems
[53]	Presenting the design and implementation of prototype HW/SW architecture for ASR system on resource-constrained devices designed as a voice-activated extension of an existing wireless nurse call system	√		Raspberry Pi hardware kit of ARM 11 processor	The prototype achieves a high average detection rate of 96%, while only dissipating 28.5 mW for continuous audio sampling and duty-cycled speech recognition
[54]	Presenting the development of a low cost remote home control system using speech recognition	√		Arduino UNO ATmega328P+ Easy VR shield	Controlling fan and lamp wirelessly by applying speech recognition into the system
[55]	Examining ASR (based on DNN (Deep Neural Network) training) performance on utterances recorded via wireless sensors	√		Crossbow Iris MTS300 sensor board	Achieve a 5% improvement in terms of relative error reduction
[56]	Presenting a microcontroller based voice controlled home automation system using smart-phones	√		Smart phone+ Arduino Uno as the microcontroller	The system enabling users to have control over every appliance in the home with their voice

6. Conclusion

In this paper we address the challenges and problems related to the design and implementation of voice transmission over a resource constrains WSN through reviewing the most attached researches. The surveyed researches have been showed the applicability of voice transmission using WSN despite its limited resources. Also, it can be concluded that two categories of VoWSN applications can be distinguished: voice communication and ambient intelligence. Each of these applications has its own requirements and field of implementation with different scenarios.

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