Design of Wireless Audio Exchange Based on nRF module

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ABSTRACT - ***A Adaptive pulse code modulation (ADPCM) approach is demonstrated for voice communication in the environment of wireless sensor networks. The architecture of the Wireless exchange Based on RF module is presented and the hardware designing of Voice Node and Gateway Node are described in detail. Voice quality of different scenarios such as indoor, corridor, corner and outdoor are tested. It is resulted that good voice quality can be obtained in short distance or line of sight with no obstacle. The Routing Node and Gateway Node is introduced to relay in the case of long distance or with obstacles in the channel path .***

Key Words: *Wireless Sensor Networks, Time Division Duplex, Voice Communication*

1. INTRODUTION

Wireless Sensor Networks (WSN) is widely used in Monitoring, industrial and agricultural production, military probe and so on. In recently years, with the development of multimedia communications technology, audio sensor networks is applied to transmit short distance audio signals for target locating and tracking.[1-6]R. Alesii F researched the problems related to the transport of an audio signal through a wireless channel and sensor nodes and presented a project for an audio surveillance system in reference [6]. In many researches for acoustic locating and tracking, the audio signal transmission way is simplex[7-10]

With the widely application of modern mobile communication, low cost short distance voice communication is rarely focused on. A new approach for short distance voice call can be contributed by using wireless sensor networks. The voice call system using WSN is called Wireless exchange Based on RF module (WAESN)[6]

In some circumstance, the voice WSN is widely needed, such as medical emergency call. Comparing to traditional medical emergency call system, Wireless exchange Based on RF module have many outstanding advantages in functional completeness, mobility, flexibility, cost and construction period. Usually, there are two duplex way for short distance wireless communications:

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2. NETWORK ARCHITECTURE

The Wireless exchange Based on RF module we designed is consisted of voice node (or handheld node, refined node), routing node(or reduced node), gateway node and call center. The network topology is constructed by NRF24L01 protocol. The voice node includes voice module, NRF24L01 wireless communication module, microphone and headphones(or speaker). The routing function is executed only by the routing node when the distance of the both speakers is too long to guarantee the QoS, since only the wireless communication module is contained. When one subnet of Wireless exchange Based on RF module is far away from an other subnet, the gateway node can be used to connect both sides of speaking. A Basestation module and a NRF24L01 module is integrated in the gateway node. The call center is designed to deal with communications between voice nodes, to display and store the calling voice nodes information and call information. Figure 1 shows the architecture of wireless sensor network for voice call.

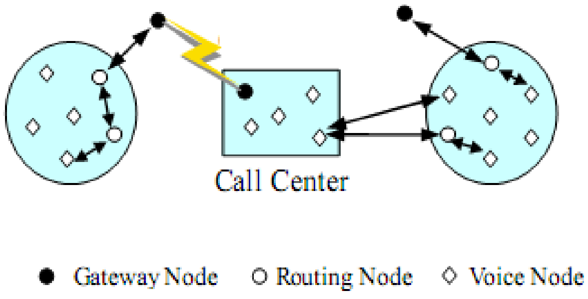

Figure 1.

3. HARDWARE DESIGN

3.1. Voice Node

The voice node is consisted of voice circuit, digital signal processing circuit, NRF24L01 module, power, buttons and status indicators. The structure and composition of voice node is shown in Figure 2.

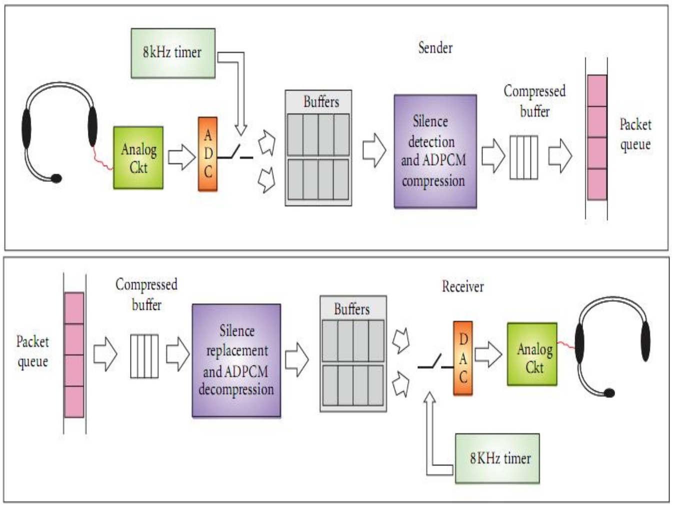

Figure 2.

The Voice analog signal amplification, filtering, acquisition, quantization, encoding and decoding, A/D and D/A conversion and power amplification are completed in the voice signal process circuit. And the main function of digital signal processor circuit is to complete real-time voice digital signal processing including ADPCM encoding to reduce data rate and interleaving to resist to wireless channel fading. The NRF24L01 chip NRF24L01 as wireless communication MCU, antenna and other circuit components are contained in NRF24L01 module. The function of node circuit control, wireless data transmission and receive is accomplished by the NRF24L01 module. Figure 3.

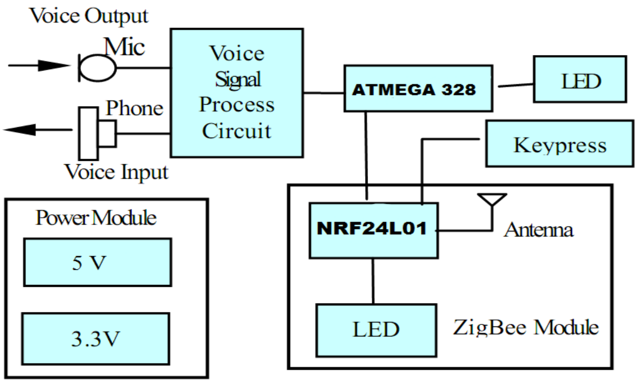

Figure 3.

3.2. ROUTING NODE

The routing node is function reduced as showing in Figure 4, only consisting of NRF24L01 wireless communication module and power supply.

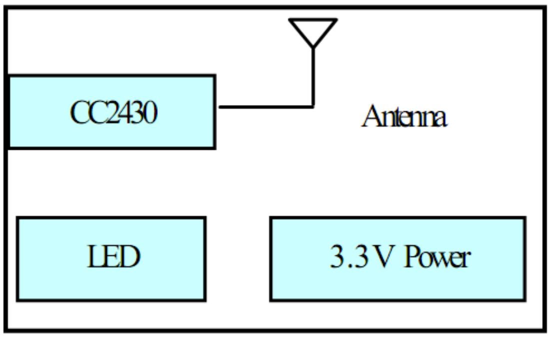

Figure. 4

3.3 GATEWAY NODE

Gateway node is consisted of NRF24L01 module, ATMEGA 328 processing circuit and power circuit. The structure of gateway node is shown in the figure 5. The NRF24L01 networks and NRF networks is connected by the gateway node. The NRF24L01 module in gateway node, voice node and routing node is identical. Ardunio embedded processor chip of ATMEGA 328 is used as the MCU of Gateway node, which with rich interfaces and simple peripheral circuits. Two voltage of 4.2V and 3.3V power output are provided to the, NRF24L01 module and ATMEGA 328 processor.

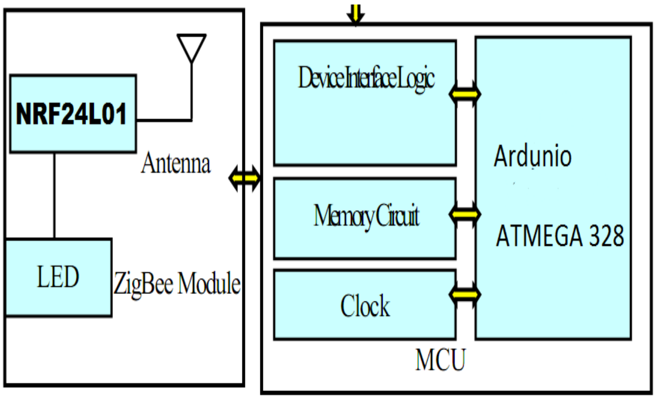

Figure 5 Gateway Node Structure

4. VOICE SIGNAL PROCESSING AND ENCODING

Voice signal processing and source coding are needed to complete the wireless voice communication in the voice node. Waveform coding, parametric coding and hybrid coding are the mainly used audio source coding method. The voice quality of waveform coding is better and the algorithm complexity of it is low than the other two.

The coding rate of waveform coding method Adaptive Differential Pulse Code (ADPCM) is 32kb/s. With the same voice quality, the rate of Pulse Code Modulation (PCM) is 64kb/s. Though the compression rate of parameter coding is relatively high, but with poor voice quality and higher complexity which requires higher processor performance.

The dedicated v.coder is usually used for designing parameter coding circuit. Which would greatly increase the cost and difficulty for future updating? The analog voice is digitalized at the first step. Then, before the data transmission, the 16bit digital voice signal is processed in turn through the order of ADPCM coding, voice buffering, interleaving, data buffering. In the end of receiver, the inverse steps are executed as shown in the figure 6

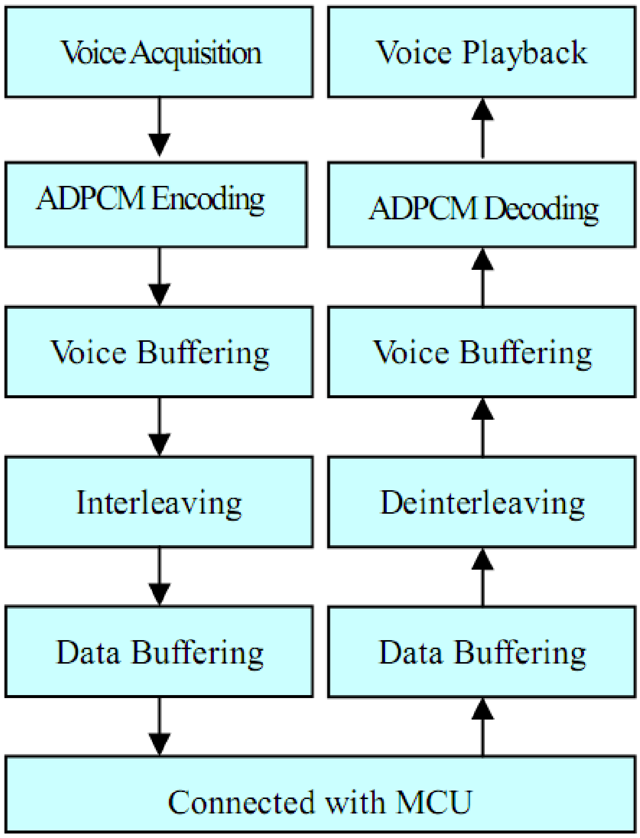

Figure 6 Voice Signal Processing and Coding

4.1 VOICE OVER REAL TIME-LINK PROTOCOL

In order to deliver voice as a primary means of communication under emergency conditions, we needed to support 2-way interactive voice, 1-way push-to-talk messaging and a voice mail application where a user could broadcast a voice snippet at moderate speed to nodes on the periphery of the network. We focus on 2-way voice communication as it is the most stringent in terms of end-to-end delay and throughput requirements.

We choose the Adaptive Differential Pulse Code Modulation (ADPCM) waveform codec as it provided us with a set of low transmission data rates, was easy to implement on an 8-bit fixed-point architecture and has low processing overhead. ADPCM is simpler than advanced low bit-rate vocoders and can complete encoding and decoding in a relatively short time.

The principle of ADPCM is to predict the current signal value from the previous values and to transmit only the difference between the real and the predicted value. As the dynamic range of this difference is small, we are able to get a compression ratio of 8:1. The microphone was sampled at 4KHz and output an 8-bit value which was compressed to a 4-bit, 3-bit or 2-bit ADPCM code. This enabled us to reduce a full-rate uncompressed voice stream of 64Kbps into a 16, 12 or 8Kbps compressed stream. We chose to sample the microphone at a rate lower than the normal 8KHz because introduced the data rate by 50% and did not degrade the voice quality significantly.

4.2 ARDUNIO VOICE CODECS

Ardunio currently supports raw audio sampling, 16Kbps,12Kbps and 8Kbps ADPCM encoding. Each ADPCM unit is encoded at the time the microphone is sampled. Ardunio maintains a double buffer to store the encoded data between transmission intervals. Table 2 lists the number of concur- rent unidirectional raw audio and ADPCM streams supported between a node and the gateway across multiple hops.

We compare its performance with GSM as it is an efficient codec that can also be implemented in 8-bit fixed point with moderate processing overhead. The number of unique slots which repeat are given by 2r, where r is the RT-Link rate. For example, rate 3 features an 8 slot repetition interval. A node may be only scheduled to transmit on slots that are separated by at least 3 slots so as to facilitate pipelining in the presence of the hidden terminals [25]. For rates 1 and 2, where a node transmits on every or every other slot respectively, only single-hop communication is possible. For rate 1, every 6ms slot is used to forward a voice packet to a receiver. In 6ms, 24 bytes of raw audio or 12 bytes of ADPCM-1 or 6 bytes of ADPCM-3 are captured.

ADPCM- 1 is able to pack 9 concurrent flows in the 112-byte payload every 6ms. As voice can be pipelined along a chain of nodes when at least 3 unique slots are available [25], ADPCM-1, ADPCM-2, ADPCM-2 and GSM-1 are able to support bi directional voice across multiple hops. For rate 3, a node along a chain transmits only once every 4 slots and hence captures 24ms of voice data. At rate 4, a node along a chain may transmit once every 4 slots for bi-directional streams or once every 8 slots for unidirectional streams.

With the network schedules used in Section 7, a node transmits a packet every 4 slots for bi-directional traffic. As each node is assigned a slot unique in its 2-hop neighborhood, both its neighbors are in receive mode during its transmit slot. Thus a node is able to concatenate both neighbors data in one packet and send it as a single transmission.

4.3 NETWORK SCHEDULING

Given a sensor network topology, our goal is to schedule at least one interactive bi-directional audio stream from any connected point in the network to the gateway while unobtrusively allowing sensing tasks to continue normal network operations. Given a connected graph G = {V,E}, we find a schedule such that from any V to the root node, there is a path composed of nodes each of which transmits every n slots. N defines the data rate of the network and hence governs which audio encoding schemes are best suited for the system.

To accommodate interactive voice, we must ensure that packet latency is symmetric between upstream and downstream communications and that packets arrive within acceptable timeliness bounds. The end-to-end latency of an audio stream is a function of the TDMA slot rate as well as the number of hops. Interactive voice requires an end-to-end latency of 250ms or less, beyond which users notice a drop in interaction quality. The goal for voice scheduling is therefore to minimize the number of unique slots to maximize n and to ensure the ordering of the slots results in balanced end-to-end delay in both directions of the flow. Scheduling of the network is performed in two phases to cater to voice streaming and simultaneously to other network applications.

First, time slots for audio streams are reserved. Next, the remaining slots are used to schedule lower data rate tasks such as sensor data reporting. It is important to schedule the audio streams first since these require strict latency bounds. Since our system initially only requires a single audio stream at a time, the two-hop coloring constraint associated with arbitrary communication scheduling is relaxed. In Fig- ure 7(a), we see that for sensor data aggregation and forwarding to avoid the hidden terminal problem and be collision free, each node requires a slot assignment which is unique in its 2-hop range.

However, in Figure 7(b), we see that if only one voice connection is required at a time, the system requires a single upstream flow and nodes at each level in the tree can utilize the same schedule. if a path is scheduled for a flow then multiple concurrent streams can be scheduled for slot rates 1 through 3 and redundant streams can be scheduled for slot rates 4 and 5. In our deployment within the coal mine, as mining groups are few and far between, we scheduled the network to support a single end-to-end voice stream from

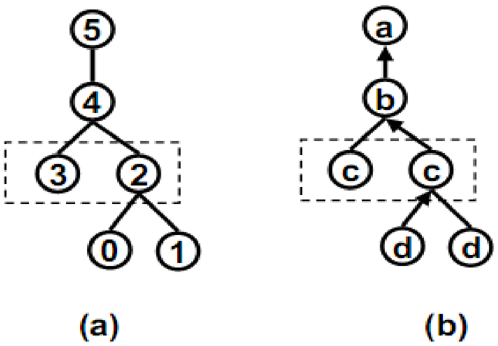

Figure 7. *Slot assignment for (a) sensor sampling (with minimum latency to the gateway) and (b) simple streaming (with minimum latency for a single flow to the gateway)*

5. CONCLUSION

In this paper, the NRF NRF24L01 module is applied to achieve duplex voice communications in wireless sensor networks environment. The designing of voice node and gateway node are subscribed in detail. The source coding method of ADPCM is used by voice code. To against the burst fading of wireless channel , the interleaving is introduced. The voice quality of different scenarios are tested. Good voice quality is obtained in short distance or line of sight with no obstacle. Routing node and gateway node can be used to relay in the case of long distance or with obstacles in the channel path.

6. REFERENCE

[1] Honggang Wang, Hempel, M., Dongming Peng,t6 et al. Index-Based Selective Audio Encryption for Wireless Multimedia Sensor Networks[J]. IEEE Transactions on Multimedia, 2010, 12(3), pp: 215 –

[2] Akyildiz, I.F., Melodia, T., Chowdhury, K.R.. Wireless Multimedia Sensor Networks: Applications and Testbed[J]. Proceedings of the IEEE, 2008,96(10), pp: 1588 – 1605.

[3] Atif Sharif, Vidyasagar Potdar, Elizabeth Chang. Wireless Multimedia Sensor Network Technology: A Survey[C]. 7th IEEE International Conference on Digital Object Identifier, 2009, pp: 606 – 613.

[4] Hui Dong, Jiangang Lu, Youxian Sun. Distributed Audio Coding in Wireless Sensor Networks[C]. International Conference on Computational Intelligence and Security, 2006, pp: 1695 – 1699.

[5] Kushwaha, M., Songhwai Oh Amundson, I. Koutsoukos, X. et al. Target tracking in urban environments using audio-video signal processing in heterogeneous wireless sensor networks[C].Signals, Systems and Computers, 2008, pp: 1606 – 1610.

[6] R. Alesii, F. Graziosi, L. Pomante, C. Rinaldi. Exploiting WSN for Audio Surveillance Applications: the VoWSN Approach[C]. 11th Euromicro Conference On Digital System Design: Architectures, Methods and Tools. 2008, pp: 520 – 524.

[7] Visar Berisha, Homin Kwon, Andreas Spanias. Real-Time Acoustic Monitoring Using Wireless Sensor Motes[C]. Proceedings of 2006 IEEE International Symposium on Circuits and Systems, pp: 847-850

[8] Craig S. MacInnes. The Localization of Distributed Acoustic Sensors[J].IEEE Journal Of Oceanic Engineering, 30(4), OCTOBER 2005, pp: 920-923.

[9] Zhao, Jingrong, Wang, Ke. Wireless locating method for point of impact based on acoustic technique[C].Proceedings of the 2009 International Conference on Communication Software and Networks, ICCSN 2009, pp: 353-357.

[10] Zhang, Jinsong Walpola, Malaka Roelant, David et al. Self-organization of unattended wireless acoustic sensor networks for ground target tracking[J].Pervasive and Mobile Computing, 2009, 5(2), pp: 148-164.