INDOORS DATA COMMUNICATIONS USING AIRBORNE ULTRASOUND

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ABSTRACT

We have tested an indoor data communications system based on ultrasound which is the core of an indoor positioning system. Unlike most other such systems, it relies on ultrasound alone. No radio or infrared channel nor any tether is used for the communications. The main hardware components are a tag and a detector. The detector has an Ethernet interface and uses digital signal processing to cope with the acoustic environment and its noise, reverberations and Doppler shift.

The attainable range is 10-20 meters and by making a comparison with the range for speech, we find that the range predictions are consistent with our experience. The channel efficiency of the system is found to be somewhat less than for human speech which also has to deal with a similar environment. This comparison is done by using the Shannon channel capacity theorem.

1. INTRODUCTION

The wireless communications field started with the discovery of the long range propagation characteristics of radio waves. Based on sonar development, underwater acoustic communications was then developed [1-3]. Airborne sound or ultrasound has not been used very much for communications. This is natural, as communications by means of electromagnetic waves performs better than acoustic communications in all but a very few applications.

These can be found in areas such as room-based asset, file or personnel tracking. These applications take advantage of the fact that ultrasound will not leave a closed room, thus making room-based positioning simpler than with a radio system. The health-care industry may be one of the first to utilize these possibilities. There are also applications involving positioning in context-aware computing and in ubiquitous computing [4]. Low bit rate device-to-device or device-to-human communications is another application [5, 6].

No analysis of the maximum possible range of such systems exists, and neither does an analysis of the bit rate when background noise level, Doppler shifts, and reverberations are considered. This paper presents such results for the first time by comparison to the properties of speech communications.

This paper starts by describing the features of the system and its subcomponents. The major challenges in designing a reliable ultrasonic communications system are then outlined. Finally a discussion of attainable bit rates by comparison with the Shannon channel capacity theorem and a discussion of maximum range are given.

2. SYSTEM DESCRIPTION

The discussion in this paper is based on the development of a set of ultrasonic tags and detectors that have been successfully used for equipment and asset tracking at several locations. The system will transmit reliably over a distance of 10-20 meters indoors. This means that most rooms in typical institutions can be covered with a single detection node. The experience from a pilot installation at the Norwegian National Hospital in Oslo, Norway, with 31 detectors and about 200 tags is described in [7].

5.1. Tags

The ultrasonic tag's circuit board is shown in Fig. 1. Its footprint is the same size as its power source, a 3.6 Volts AA lithium battery. The tag has a microcontroller, an ultrasound transducer which is used both for reception and transmission, a movement sensor and an optional optical tampering sensor. The tag will transmit its own unique ID, its battery status and error detection bits.

5.2 Detector

The detector processing unit (DPU) contains a digital signal processor, and an Ethernet controller and interface, as shown in Fig. 2. It is powered from a mains adapter. It can process inputs from up to 8 detectors.

These detectors consist of an ultrasound microphone and some analog electronics. They can be mounted wherever it is convenient such as in the ceiling or on the wall, or integrated in wall panels. The architecture is shown in Fig. 3. All movement data is collected in a central server and is accessible from client terminals where the

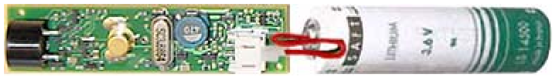

Figure 1 Tag electronics with battery.

Figure 2 Detector Processor Unit (DPU)

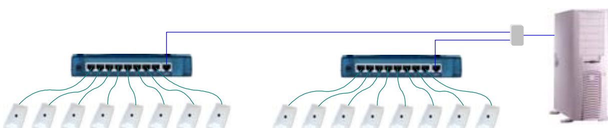

Figure 3 Detector architecture showing 16 detectors connected to 2 DPU's linked via Ethernet to a PC

database can be accessed in different ways according to the need of the specific installation.

One of the unique features of the system is that all data transmission takes place using ultrasound alone. Unlike other comparable systems such as ActiveBat [8], Cricket [9], and Dolphin [10] that rely on radio for multiple access control, triggering or data communications, there is no need for a side channel in our system. Multiple access control is accomplished by having an ultrasound receiver in each tag that ensures that the likelihood of two tags transmitting simultaneously is small. Each tag will listen for a clear channel before it attempts to transmit using a carrier sense, multiple access (CSMA) protocol.

4. CHALLENGES IN ULTRASONIC COMMUNICATIONS

The main challenges in ultrasonic airborne communications are due to the large Doppler shifts, the reverberation, and the background noise. The signal processing algorithms of the DPU have been designed to deal with each of these issues as discussed here.

4.1 Doppler Shift

The propagation speed in air is only 340 m/s which is almost 1:1,000,000 compared to radio and a little more than 1:4 compared to underwater sound. The largest platform velocities one can think of for radio communications, such as interplanetary spacecraft at 21,000 km/hr generate a relative Doppler shift that is equivalent to only 24 m/hr platform velocity for airborne ultrasound. Compared to a typical velocity for walking of 6 km/hr (our design target), it is therefore clear that Doppler shifts are greater in airborne ultrasound than in any other medium. But fortunately, the nature of the shift is not as in underwater communications where there is also Doppler spread due to surface or platform motion.

4.2 Reverberation

Reverberation is the lingering of sound in a room once the source of the sound has ceased producing. In building acoustics, the reverberation time is defined at the -60 dB point and typical values are:

- 0.3 sec - "dead" sound.
- 1 sec - clearer articulation of speech.
- 3.5 sec - richer musical sound, difficult for speech.

In a communications system, -60 dB is very conservative, so smaller values can be used in design. Typical values are 50 – 300 ms. The larger values should be used for communications in e.g. a long corridor with concrete walls, floor, and ceiling. This ensures that the range will be limited by the background noise and not by the reverberations. This was our design criterion and it is assumed for the rest of this paper.

4.3 Noise

The noise level at audible frequencies is:

10dB - Soft rustling of leaves
20dB - Whisper at 1 meter
30dB - Very soft music
40dB - Quiet office or residential area
50dB - Large office background noise
60dB - Normal conversation, background music
70dB - Freeway traffic, TV audio

The values are given in SPL (Sound Pressure Level) which is dB relative to 20 μPa.

At ultrasound frequencies, the only background noise measurement that we are aware of is that by Bass and Bolen from 1985 [11]. They measured a level of 70–80 dB SPL in the range 20–60 kHz in an industrial environment (3 kHz bandwidth), with grinding producing a level of 80 dB. Air tools are even worse and may produce levels up to 100 dB SPL 1.2 m from the source. We will use the 75 dB level later for comparison and need the equivalent spectral density which is 75 dB-10 log(3000) = 40.2 dB/Hz.

With the proliferation of switch-mode power supplies, a new noise source has become common. They often operate in the range 20–60 kHz, and may generate an almost sinusoidal ultrasonic noise. Fluorescent lamps and computer monitors are especially bad with this respect. Any reliable ultrasonic communications system needs to be designed with robustness against such noise.

5. COMPARISON WITH SPEECH COMMUNICATIONS

Speech is a communication form which operates in the same medium as our system, therefore it is of interest to compare the two both with respect to achievable data rate and range.

5.1 Data Rate

In speech, a typical data rate for the raw speech is 50 bits/sec, and with prosodic information (i.e. such as details of intonation, mood of the speaker, or even the sex of the speaker) it increases to about 200 bits/sec [12]. Communications quality speech can be transferred in a bandwidth of about 300 – 2300 Hz. Therefore the efficiency in terms of bit rate per bandwidth is in the order of 50/2000 = 0.025 bits/s/Hz.

The efficiency, C/W, is a measure of how well the channel is utilized and comes from Shannon's information capacity theorem:

$$C = W \log_2(1 + SNR) \tag{1}$$

Here C is the rate in bits/s, W is the bandwidth in Hz, and SNR is the signal to noise ratio. For the sake of comparison, a mobile communications system like GSM has a C/W in the order of 0.5-1 bits/s/Hz, and an analog modem in a standard fixed line telephone connection achieves C/W in the order of 10 bits/s/Hz indicating better and better utilization of the bandwidth. Speech communications is therefore a means of communications that utilizes the channel much less efficiently than typical electromagnetic communications systems. This is due to the low quality of the acoustic channel and the redundancy built into speech that makes it robust to movement, reverberations and noise. It should be remarked that the acoustic channel in this respect is a much more difficult medium than the wireless radio channel or the mobile telephone channel.

Now let's compute the data rate that one would expect to get from an airborne ultrasonic communications system. We will assume that a transducer with f_0=40 kHz transducer is used, as they are readily available. Typical bandwidth (-6 dB) is 10% or W=4 kHz. A simple, but robust, modulation such as FSK is used, and the system should have a large tolerance to Doppler shift. If the maximum velocity is +/-v, the maximum Doppler shift is +/-f_0v/c and the minimum distance between each transmitted frequency is $2f_0v/c$. Thus the number of frequency pairs that can be used for FSK is:

$$M = \frac{W}{4 f_0 v / c} \tag{2}$$

Using c=340 m/s and v=2.1 m/s (7.6 km/h) one can use only M=4 pairs. The efficiency is then:

$$C / W = M \cdot C_0 / W \tag{3}$$

Here C_0 is the transmission rate per pair of frequencies.

In order to have an efficiency similar to that of spoken speech, C_0 must be:

$$C_0 = (C / W)_{speech} W / M \tag{4}$$

This yields a value of C_0=25 bits/s per pair and a total bit rate of $C=MC_0$ = 100 bits/s. This estimate assumes that each pair of frequencies can be reused every $1/C_0$ seconds or after 40 ms, which is usually too little. A fourfold increase in reverberation design value (160ms) and a decrease of maximum velocity to only 0.5 m/s also gives a system which has similar efficiency as speech. However, our applications of ultrasonic communications systems required tolerance to larger Doppler shifts than that, and thus a corresponding lower efficiency.

5.1 Range

The maximum sound pressure level (SPL) for male speech is about SL=88 dB SPL.

Attenuation of ultrasound in air increases with frequency, and usually in a quadratic manner. The absolute attenuation also increases with humidity, and temperature and pressure also play a role. For the audible frequencies, attenuation is not so large and spherical spreading is often the dominant effect for limiting the range, but at higher frequencies the attenuation also has to be taken into account. Spherical spreading implies that intensity decreases by 6 dB per doubling of distance. The performance of communications systems can be analyzed using a calculation of the received signal to noise ratio using the passive sonar equation:

$$SL - PL - NL > DT \tag{5}$$

The meaning of the symbols is:

SL – Source level in dB at range R_0 (usually R_0=1 m)
PL – Propagation loss, consisting of spreading loss, $6\log_2(R/R_0)$ and attenuation αR.
NL – Noise level

DT – Detection threshold in dB

The threshold for detection is in the range 15-20 dB [13], and here the more conservative value will be used. An analysis of speech for SL = 88 dB and a noise level, NS = 30... 40 dB in eq. (1) gives:

$$88 - 6\log_2 R - NS > 20 \qquad (6)$$

The maximum range is 25 m for 40 dB noise level and 80 m for 30 dB noise level, which seems to agree with common sense. Thus we are confident that the equation can be used for ball park predictions of range.

A similar range calculation for a 40 kHz communications system is now done based on:

- Source level SL=115 dB SPL at 1 meter
- Propagation loss consists of spherical spreading and attenuation which increases with relative humidity from α=0.27 dB/m (0% RH) to a maximum of 1.25 dB/m for 40% RH, and then falls to 0.89 dB/m at 100% RH.
- Processing bandwidth BW=25 Hz
- Noise level is 75 dB SPL measured in 3000 Hz or 40.2 dB/Hz.

In the worst case (40% RH), the range can be found from

$$115 - 6\log_2 R - 1.25R - (40.2 + 10\log(25)) > 20 \qquad (7)$$

This equation has a solution for a range of 14.3 meters which is close to the minimum range guaranteed in practice. In a 0% relative humidity environment, the range would increase to 36 meters, and if in addition the noise reduces by 10 dB, the estimated range will be almost 60 meters. This shows the large dependency on environmental factors that are outside of the control of those who deploy these systems.

6. CONCLUSION

This paper has outlined the main elements and experience with an ultrasonic indoors communications system. The performance of the system has been estimated by comparing with speech both for bit rate calculation and range estimation. Although there are many uncertainties in these first attempts at estimating performance parameters, the system has been found to be consistent with these predictions.

Our present system has a fixed detection threshold, and we have shown that it is desirable to design a system which adapts to the environment. Such a system could achieve a much higher range under favorable conditions, while maintaining robustness under less favorable conditions. It should also be possible to increase datarate since the channel efficiency is only 25% of that of speech.

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