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Utilization of wireless technology for sound communication using TMS320 C6713

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Abstract. The development of communication technology, both on algorithms and devices has been so rapid. The application of communication devices that previously used radio signal (wireless) would be more efficient if cable (on wire) is used instead. With a large number of buildings with wide locations, there are still many spaces that are not reachable by communication devices. Therefore, it is necessary to add a new communication tool that can be used to convey information in the campus environment intended for announcements, from one place to another or from one room to another simultaneously, both with point to multipoint as well as point to point characteristic which is suitable for internal communication and emergency situations. This wireless network can be expanded without using a cable backbone. In addition, with the equal sharing of access rights, the network will be more effective because it can be done anywhere as long as there is still a network connection. Audio effects can be applied in real time using DSP (Digital Signal Processing) Starter Kit TMS320C6713 by implementing in the equipment. In the audio input signal of 0.050 Vpp there is no change in the frequency of the original signal that is 1 Khz there is a change in the intensity of the sound produced 0.364 Vpp, 2 kHz to 0.280 Vpp while 3 kHz to 0.236 Vpp. There is a change in amplitude while the output frequency approaches the original signal frequency.

1. Introduction

Communication is an important activity in activities in various places, especially on campus which consists of a group of people who jointly organize academic activities. In order to facilitate each element in the campus environment to communicate between elements required various communication tools. The function of this communication tool is vital so that often campus institutions have more than one type of communication tool.

Nowadays, the type of communication equipment and technological advancements are growing rapidly, making it possible for humans to still be able to communicate with each other even in different locations. Communication devices used are also as varied as intercoms, telephones or mobile phones or smartphones. short, for example between spaces in one building.

Of the various communication devices available, several communication tools such as telephone in the campus environment. But with a large number of buildings with large locations, there are still many rooms that are not affordable by means of communication. Therefore, it is necessary to add new communication tools that can be used to convey information or information in the campus environment intended for announcements, from one part to another or from one room to another simultaneously both point to multipoint and point to point suitable for internal communication and emergency situations.

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The tool that will be designed in this study has the advantages of being easy to install and operate, economical, does not require special operators to serve the connection, saves time and effort so as to increase employee productivity, wider communication channels, and can be utilized for information delivery.

In this research, will design and prototype a communication tool utilizing the wireless module facilities that are available. From the background that has been stated, can be formulated as follows: (1) How to determine the type of TMS that will be used? (2) How is the system integration of the supporting devices to be used?

Design and implement communication tools for announcements, from one part to another or from one room to another room either simultaneously or is point to multipoint or point to point suitable for internal communication and emergency situations.

2. Literature review

In a previous study about Wi-Fi enabled speech recognition controller nodes [1]. This study aims to record sound and interpret into text via WIFI. The process through a microphone receives sound, the received analog signal is converted into a digital signal and then transmitted via WIFI. Digital stream will be converted into text through a web service, stored and issued the specified command. This study uses the Arduino UNO platform integrated with the ESP8266 WIFI module. Microcontroller used ATmega328 which can sample analog to digital signals up to 20 MHz A sound sensor is also used to distinguish the original sound from room noise. Use of micro-SD cards to store and transfer audio and data via WIFI. Bluetooth is difficult to implement because of the low range. The use of WIFI was chosen because data can be transmitted via the internet directly with a further range and can reduce system power compared to Bluetooth. In this study, it is important to consider network security issues in the audio recording process.

Other research related to DSP boards using TMS320C6713 DSK examines the An Automatic Speech Recognition System on the DSP Board [2]. This study aims to design and implement automatic speech recognition and synthesis systems. This study uses the TMS320C6713 DSK produced by Texas Instrument (TI) as an independent platform. also integrated using MATLAB software.

In the development of the interface (interface) using Visual C #. Design for speech identification systems) is divided into two stages, namely: experiment and identification. Each given voice input is matched consecutively with the sound reference that has been stored in external memory and the recognition result is a match with a minimal distortion rate. MFCC and DTW algorithm is also used to find the same sound, by comparing the time sequence and the synthesis system on the software MATLAB is used to predict errors (error) in each frame.

In addition, Jean Jiang conducted research on Audio Processing with Channel Filtering using DSP Techniques [2]. This research discusses operating the sound system through the DSP board. The DSP board used is TMS320C6713. In addition to low (woofer) and high (tweeter) frequencies, the audio system is improved by adding speakers for mid-frequency and designed with a BPF filter to process in the middle region. Also given are 3 audio amplifiers for low, middle and high frequencies. Audio amplifier plus gain for power on the speaker, by modifying using the datasheet of the audio chip on the LM386 IC. Filters used are 3 types, namely low pass, band pass and high pass with Remez algorithm. 2 DSP boards are used to process sound signals in the audio frequency range. Audio data processing is enhanced with amplifiers on speakers in bass, mid-range and tweeters. To produce good sound, 3 different filters are used on bass, middle frequency and high frequency. The sound system is operated with a DSP board with 3 kinds of filters, low-pass, band-pass and high-pass. The system consists of high-quality amplifiers, 2 DSP boards and 3 set speakers. The system can produce quality sound at different frequencies. DSP boards are also used to filter audio that produces good sound quality.

Subsequent research discusses TMS320F28335 DSP programming using MATLAB simulink embedded coder: techniques and advancements [3]. This study provides a tutorial relating to digital signal programming (Digital Signal Processor) using Texas Instruments TMS320F28335 using code

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composer studio (CCS) version 6 and MATLAB Simulink embedded coder. The basic functions that can be used are related to modulation, analog digital conversion and proportional-integral controller.

3. Research methods

3.1. Research stages
Stages of research are intended to conduct detailed research in the manufacture of devices so that the results will be obtained in series. The research steps are as follows:
- Literature study on TSM DSK and Wi-Fi modules as the main components of this system. At this stage also determined the system specifications, programming used and network systems.
- Work system design, at this stage work steps will be made from making the device.
- The design of the design that is making the design of the work system design of the device to be made. At this stage the work steps of the TMS module and the Wi-Fi module are written in order to be able to carry out the communication system optimally.
- Component Selection, purchasing components for this research, this needs to be included because the device sought is still not widely sold in the market.
- Making the whole system, the making of the system as a whole is carried out the preparation of the system in accordance with the planning and design that has been made.
- System Testing, the system has been designed, tried to be implemented in the field.
- System Testing Results, at this stage, a check is made on the results of testing the system whether it is running as expected or not. If a program experiences a problem, a program error trace will be carried out so that the program runs as expected. And if the device is experiencing problems, it will be checked again against the device.
- System Analysis, at this stage an analysis of the work system between devices is performed, including programs that can run appropriately, devices can communicate (send data to each other with distances that can be reached for communication between devices) and find out end-to-end delay that occurs.
- Conclusion, aat this stage the conclusion is made from the results of research that has been done.

3.2. Research design
The design of the block diagram for the study is shown in Figure 1 below:

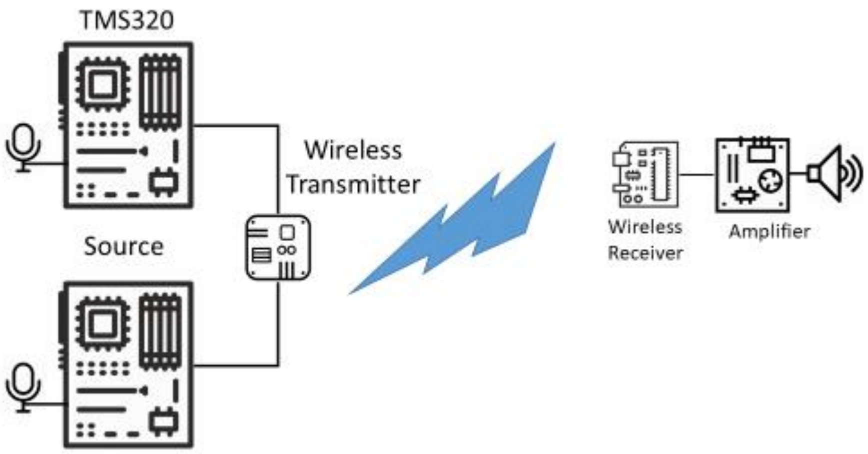

Figure 1. Device block diagram.

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Figure 1 explains the overall research block diagram, which consists of 1 set of Transceiver (Tx) that is connected using a Wireless connection with 1 set of Receiver (Rx). Each Tx and Rx consists of 1 microphone, 1 speaker, 1 TMS320C6713 Digital Signal Processing (DSP) module and 1 Wi-Fi module.

The selection of DSP TMS320C6713 is not without reason. This Texas Instrument DSP was developed as a low-cost platform but has high performance and is easy to implement because it is integrated with the components related to signal processing.

Sound that will be forwarded to its destination, will enter through the microphone on the Tx side, then processed from analog to digital form using the DSP TMC320C6713 module. Changing the shape of this signal is needed so that the voice signal can be forwarded to the Rx side using Wireless. On the Rx side, the voice signal received from Tx over Wireless, will be returned from digital to analog form so that it can be forwarded to the destination via the speaker.

4. Results and testing

The testing process is done after the program is ready to be implemented on DSK TMS430C6713. *The* necessary *hardware* is prepared as follows:

- 1 unit of Computer / Laptop
- DSK TMS320C6713*,* as the main device used to process sound into data and vice versa.
- Microphone, as a device for capturing sound waves.
- Active Speaker, as a device to strengthen the sound waves.
- Speaker, as a conversion device into sound waves
- USB and audio cable, as a liaison between existing devices.
- Wireless Module, as a link between the TMS320C6713 DSK, output with the audio receiver wirelessly.

4.1. Port - port on DSK and PCs used in the testing and data collection

DSK TMS320C6713

- Power Jack, serves to connect DSK with electric power.
- USB, functions to connect DSK with PC or laptop and as a media to embed programs from PC to DSK.
- Line in, serves to connect between the speaker port on a PC or laptop with DSK via cable media.
- Line out, functions to connect DSK with active speaker in the process of testing and retrieving data.

PC or laptop, USB, functions to connect PC or laptop with DSK and as a media to embed programs from PC to DSK. Active speaker, functions for the process of testing and retrieving data.

4.2. Data testing and retrieval process

In the testing process the input used is audio tone. With output in the form of sine signals which are tested using multi-Instrument and Active Speaker

The process of taking input data used is audio tone. With output in the form of sine signals which are tested using multi-Instrument and Active Speaker. Tests with a 1 KHz input signal frequency obtained by the TMS output amplitude of 0.05 Vpp. Ch.1 is the input signal from the function generator while Ch.2 is the output of the TMS in figure 2.

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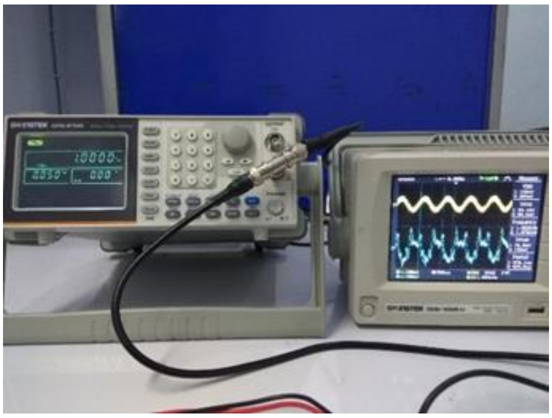

Figure 2. System testing for 0.05 vpp input signal input frequency 1 KHz.

As for the output from the wireless receiver as shown in Figure 3 and 4. Ch.1 is the input signal form from the function generator of the wireless transmitter while Ch. 2 is a wireless receiver output that is ready to be given an audio amplifier to get to the speakers.

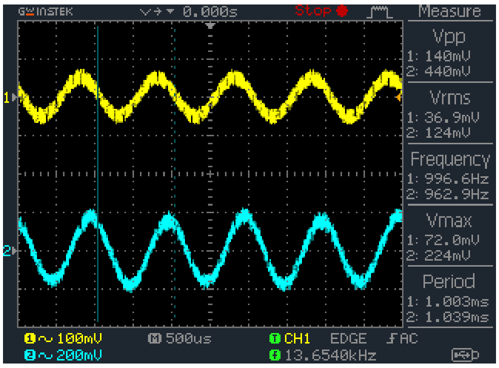

Figure 3. TMS320 input and output signals for 1 KHz frequency.

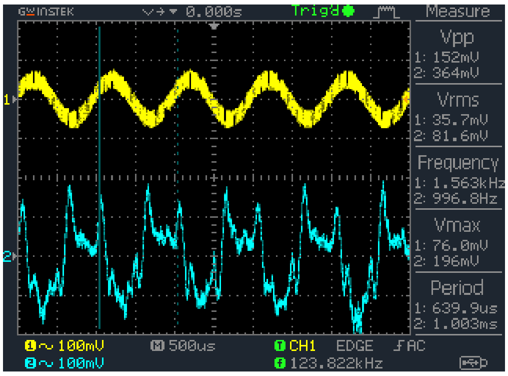

Figure 4. Input signal and output wireless receiver frequency of 1 KHz.

Next in Figure 5, a test with a 2 KHz input signal frequency obtained TMS output amplitude of 0.05 Vpp. Ch.1 is the input signal from the function generator while Ch.2 is the output of the TMS.

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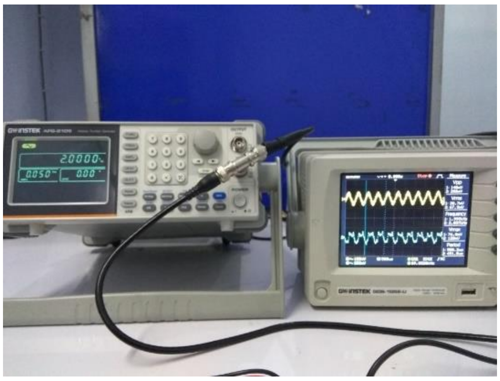

Figure 5. System testing for a 0.05 Vpp input signal input frequency of 2 KHz.

The output of the wireless receiving device is shown in Figure 6 and 7. Ch.1 is the input signal form from the function generator of the wireless transmitter while Ch. 2 is a wireless receiver output that is ready to be given an audio amplifier to get to the speakers.

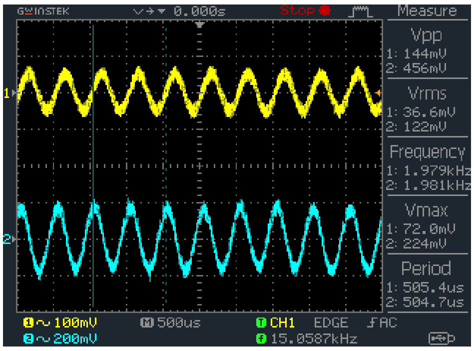

Figure 6. TMS320 input and output signals for 2 KHz frequency.

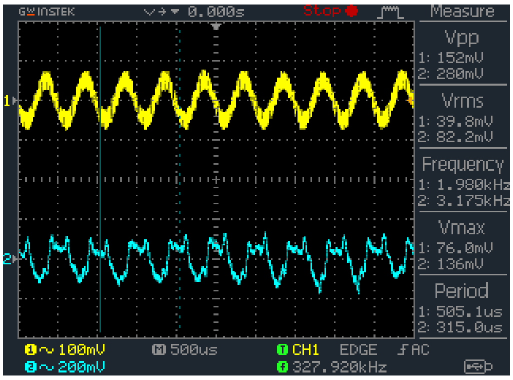

Figure 7. Input signal and output wireless receiver frequency of 2 KHz.

Then in Figure 8 testing with the frequency of the 3 KHz input signal obtained TMS output amplitude of 0.05 Vpp. Ch.1 is the input signal from the function generator while Ch.2 is the output of the TMS.

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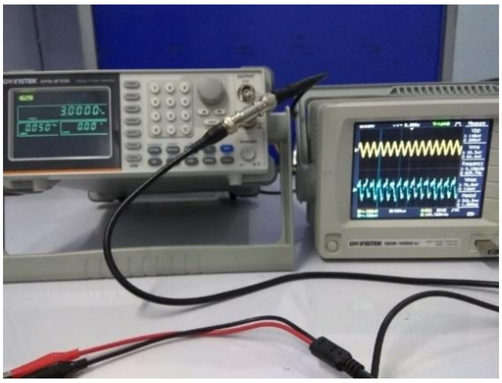

Figure 8. System testing for a 0.05 Vpp input signal frequency of 3 KHz.

The output of the wireless receiving device is shown in Figure 9 and 10. Ch.1 is the input signal form from the function generator of the wireless transmitter while Ch. 2 is a wireless receiver output that is ready to be given an audio amplifier to get to the speakers.

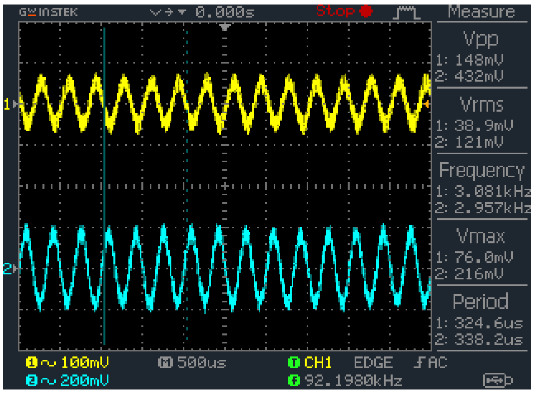

Figure 9. TMS320 input and output signals for 3 KHz frequency

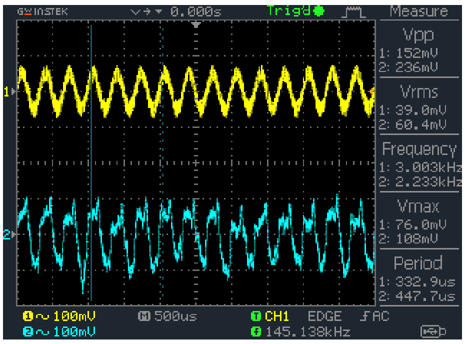

Figure 10. Wireless receiver signal input and output frequency of 3 KHz

The output of the wireless receiving device is shown in Figure 10. Ch.1 is the input signal form from the function generator of the wireless transmitter while Ch. 2 is a wireless receiver output that is ready to be given an audio amplifier to get to the speakers.

From all tests for the input amplitude of 0.05 Vpp with frequencies of 1 KHz, 2 KHz and 3 KHz, there was a change in signal shape while the real frequency remained.

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5. Conclusion

From the results of testing and analysis in this study several conclusions can be drawn as follows: 1) At the audio input signal 0.050 Vpp 1 Khz signal frequency there is a change in the sound amplitude produced 0.364 Vpp. 2) For the input signal with a frequency of 2 Khz there is a change in the amplitude of the sound produced 0.280 Vpp. 3) Input Signal is 3 Khz there are changes in the amplitude of the sound produced 0.236 Vpp. 4) There is a change in amplitude close to the frequency of the original signal.

After going through the planning, design and testing stages of this research, the system that has been made is not perfect and can still be developed even better. For development suggestions, we can provide the following: 1) The addition of support modules to be easily within TMS320C6713 module. 2) Making effects in real time using dynamic microphones. 3) Making MATLAB Simulink as a simulation tool.

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