Advances in Science and Technology, Vol. 2, No.2, September/October 2008, pp114-121.

Design and Simulation of a Microprocessor-Based Audio Teleconferencing System

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(Submitted: March 20, 2007; Accepted: September 19, 2008)

Abstract

The design and simulation of a microprocessor controlled audio–teleconferencing system is presented. Top–down design technique is adopted and the system is divided into modules. The modules with the exception of the power supply and the mixer interact by means of the 8085A microprocessor. A mixer circuit is incorporated to combine the inputs from all the microphones to the system and to give the same output to the individual speakers so as to enable participants have discussion in a more natural atmosphere. The design is intended to accommodate sixteen people in a group discussion with three groups operating simultaneously. However, the system can be readily expanded far beyond this capacity with minimum components increase because it is software based. "**Add - on**" bridging technique which allows a conference participant, to call up others to add each to the conference is employed. The design was implemented and tests carried out in the laboratory to demonstrate the workability of the design. Software development for the system was based on the INTEL 8085A microprocessor instructions and the program written in Assembly language. System operation was simulated using the Microprocessor Applications Trainer (MAT 385) and the results show the program to work.

Keywords: Teleconferencing, Audio, Microprocessor, Bridges, Add-on, Simulation

1.0 Introduction

Teleconferencing is defined as group communication through electronic means (Johnsen 1984). Wikipedia Encyclopedia 2008 has defined teleconferencing as the live exchange of mass articulation of information among persons ad machines remote from one another linked by a telecommunications system usually over the phone line. The telecommunication system may support the teleconferences by providing one or more of the following, audio, video, and/or data services (see Military Standards 2008). It is a generic term for the whole range of electronic meeting aids. Teleconferencing need not be a "meeting" in the same sense as face–to–face meeting but it attempts to provide as much as possible the natural atmosphere that exists when people come together for the purposes of having discussion. With it, widely separated people can conduct a meeting by typing messages at their terminals (Tanenbaum 2002). Attendees may leave at will and find out what they missed when they come back. Tanenbaum 2002 has used the term teleconferencing for a variety of telecommunications media ranging from:

- Voice only conferencing – where only audio signals are being transmitted.
- Voice with still pictures or voice with motion picture – where audio with fixed or moving images are transmitted simultaneously.
- Electronic mail which is oriented towards person–to–person communication and
- Computer conferencing which supports conferencing at which participants are not present simultaneously.

Teleconferencing has a spectrum of options with no clear indication that any of the options is superior, though there are definite price trade–offs between the transmission bandwidths required as well as various equipment configurations (Johnsen and Vallee 1979). Therefore, a tendency exists to view audio teleconferencing as bottom of the line. The obvious cost difference makes this a logical observation. In any case, audio teleconferencing is as good as and sometimes better than motion video teleconferencing for specific applications. Some of such applications where audio teleconferencing is attractive are:

- Voice only conferencing – where only audio When there is no restriction on when and

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how long a user is entitled to use it.

- Voice only conferencing – where only audio When the fund available to the user cannot support video teleconferencing.
- Voice only conferencing – where only audio Where the people involved in the conference are already familiar.

In this paper, a microprocessor is used to control the connection of individuals in order to have an audio conference. These individuals are physically separated but in an area of interest say a university environment. The design supports more than one such conference going on at a time. The use of a microprocessor for the control reduces the number of components required for the system design. Also, since the system can be programmed to meet individual user need, the use of a microprocessor provides the best means of achieving the connection.

2.0 Audio Teleconferencing

Although videoconferencing is making headway, audio teleconferencing continues to deliver on the promise of long distance communications for education and business. The greatest limitation to electronic mail is the computer terminal it requires. Audio teleconferencing eliminates this need since it can be used through a touch tone telephone. A number of audio teleconferencing systems are already in use. Two most widely used are the ones developed by Robert Springer in the United States of America (see Springer, Schmal and Gavenman) and Yamaguchi and his colleagues in Japan (see Yamaguchi and Wada 1986). The design by Robert Springer features the use of telephone lines on commercial audio bridges to support multi–point communications. Horn and Sharma 1994 have reported typical operations of audio bridges.

2.1 Components of Teleconferencing System

The components which make up a teleconferencing system include audio–visual terminal equipment, bridging equipment and telephone lines or other transmission media to interconnect these facilities. A variety of shared and dedicated communication services that can be used to link together the remotely located participants of a teleconferencing system abound. The public switched telephone network (PSTN) can be used to interconnect audio systems. Satellite transmission facilities can be used to interconnect point–to–point or point–to-multipoint compressed full motion video locations (Maher 1988).

When audiographic terminal equipment is interconnected via telephone line, a teleconferencing bridge is the means by which multiple locations are tied together during a meeting. Two basic types of teleconferencing bridge for terminal equipment that inter work with the telephone network exist namely, a premises bridge and a network bridge. A premises bridge is located on the premise of a teleconferencing service vendor while a network bridge is embedded in the public telephone network and usually provides for echo control and automatic volume level control.

A bridge may have one or more of the following features (Fischell and Maher 1987):

- The "***Meet–Me***" feature which allows individuals and groups to dial into the bridge to participate in a scheduled conference.
- The "***Add–On***" feature which allows a conference participant (or designated attendant) to call up others to add each to the conference.
- The "***Dial–Out***" feature which requires a conference operator to dial out to bring participants into a conference.
- The "***Blast–Up***" feature which uses a computer to simultaneously dial–out the numbers of the conference participants and add those that answer to the conference.

The design in this paper utilizes the advantage of the ***add–on*** system. One advantage of the ***add–on*** bridging system over others is that an unwanted person cannot link his or her line to the bridge and listen to the conference.

2.2 Design Consideration

In the design reported in this paper, a lot of factors were considered before components were chosen with a view to making the design cheap, reliable, efficient and work effectively. Compromise was therefore drawn between conflicting factors. The major considerations in the choice of components were cost and availability. Where options were available reliability and familiarity were considered. The choice of circuit was based mainly on simplicity to enhance quick and easy implementation. In some cases, system performance and quality were given priority.

The selection of the microprocessor was based on availability, ease of use and the availability of complementary hardware. An 8–bit Intel™ 8085A was considered suitable for the range of this work (see Intel Corporation 1983).

2.3 System Layout

The block diagram of the audio teleconferencing system is shown in Figure 1. Top–down design approach was used and highly interrelated parts of the system are organized in modules. The modules are:

(a) Processing unit
(b) Keyboard unit
(c) Decoding unit
(d) Switching unit
(e) Mixer unit

2.4 Processing Unit

The processing unit consists of the microprocessor, power–on reset circuitry and the crystal oscillator, the memory and memory decoder chips and the programmable input and output units. Address lines $AD_0 - AD_{15}$ form the 16–bit address bus, while $AD_0 - AD_7$ form the multiplexed address/data bus. $AD_0 - AD_{10}$ are connected to the memory unit where they are used to achieve the required memory map. Address lines $A_{11} - A_{15}$ are connected to the data and enable lines of the 3–to–8 line decoder chip which is an Intel™ 8205 compatible with the 8085 microprocessor. The address latch enable ALE pin of the microprocessor is connected to the corresponding ALE pins of the memory chips. This is to distinguish when data is placed on the multiplexed data/address line and when the bus is carrying address signal. The IO/$\overline{M}$, CLK, $\overline{WR}$, $\overline{RD}$ pins of the microprocessor are connected to the corresponding pins of the memory chips. The reset out pin of the microprocessor is extended to the reset–in of the memory chips while the outputs of the 8205 chip are connected to the chip enable of the memory chips.

The memory mapping of the system is as follows:

0000H – 07FFH as EPROM 1 (2KByte)
0800H – 0FFFH as EPROM 2 (2KByte)
1000H – 10FFH as RAM 256bytes

When a memory location within the address range 0000H – 07FFH is addressed, pin 15 of the micro-

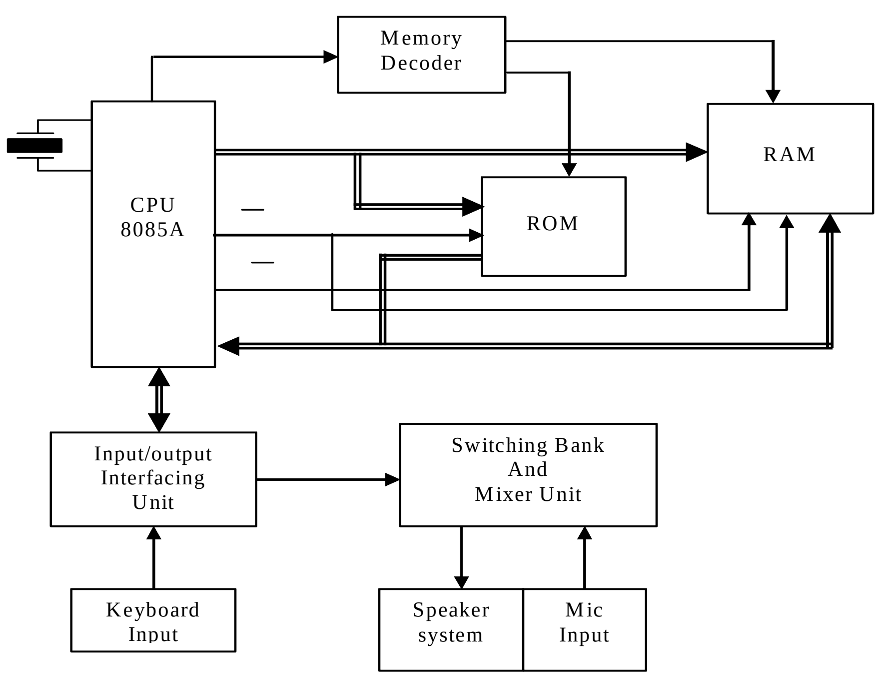

Figure 1: System block diagram

processor will go low thus enabling data to be read from EPROM 1. Similarly, when a memory location within the range of 0800H – 0FFFH is addressed, pin 14 of the 8205 chip, the IO/$\overline{M}$ and the $\overline{RD}$ outputs of the microprocessor will go low, thereby enabling data to be read from EPROM 2. Memory locations 0000H – 00CEH were used for the system software. The central processing unit is shown in Figure 2.

The memory chips 8155 and 8755 contain the input/output (I/O) ports. The logic value of the IO/$\overline{M}$ determines whether the address refers to memory or I/O port. The address and value of IO/ are latched on the falling edge of the ALE. The ports can be configured to serve as an input or output port as shown in Figure 3 (Feedback Instruments Limited 1983).

The Intel 8155 and 8755 require single 8-bit command information to initialize the ports. Each of the 8-bit byte is assigned some command significance as shown in figure 4. The command byte is transferred to the 8155 or 8755 using a specific address and

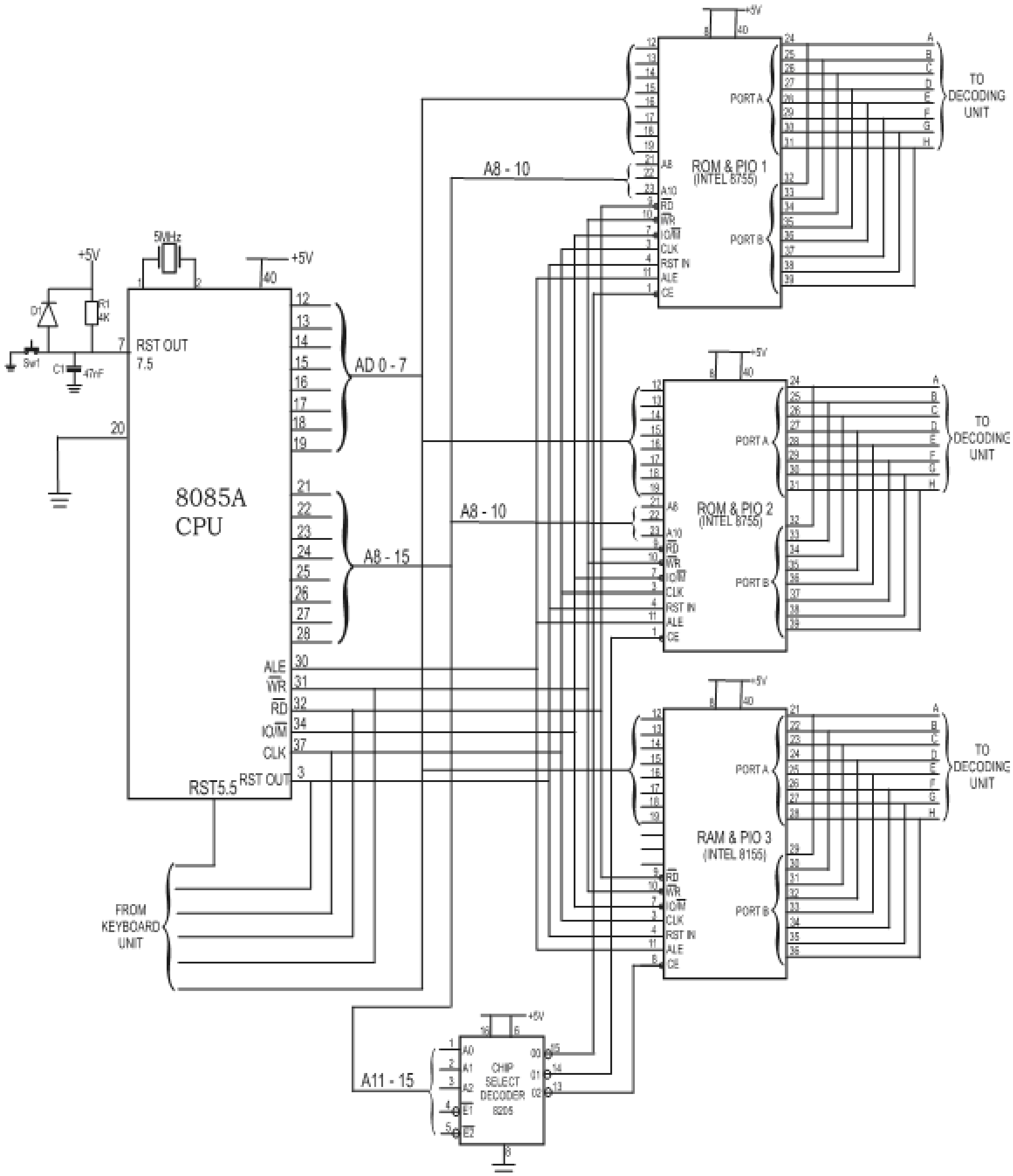

Figure 2: Processing Unit

the programmed output instruction *OUT*. Thus the instruction *OUT address* transfers the contents of the processor A–register to the address input/output device. Table 1 shows information on how the ports are programmed.

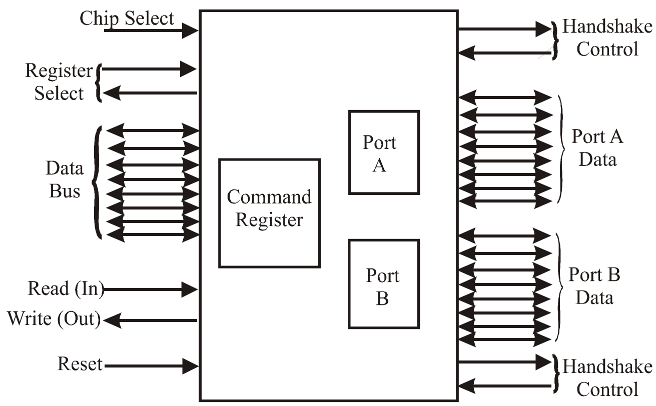

Figure 3: Port initialization information

Table 1: Port Programming Information

Chip	Port Address	Function
8755	00	Command status register
ROM	00	Port A (8 bit I/O)
PIO 1	01	Port B (8 bit I/O)
8755	08	Command status register
ROM	09	Port A (8 bit I/O)
PIO 2	0A	Port B (8 bit I/O)
8155	10	Command status register
RAM	11	Port A (8 bit I/O)
PIO 3	12	Port B (8 bit I/O)

In this design, port A is configured as an output and port B as input port. Port A lines are connected to Port B lines in such a way that the output is equal to the input.

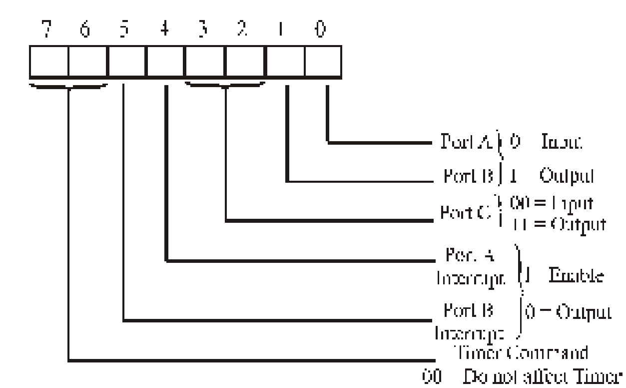

Figure 4: 8-bit command information

2.5 Keyboard Encoding Unit

This unit consists of the matrix keyboard located at the users end and the programmable keyboard interface (Intel 8279 chip), the keyboard decoder (74LS156) which is a dual 2–to-4 line decoder configured to work as a 3–to–8 line decoder. The line enhancing chips are located in the users end and the central location. When the system is switched on, the 8279 chip starts to scan the keyboard. If any key is pressed, then the interfacing chip recognizes the key and generates the appropriate binary digits. After two keys have been pressed, the binary equivalent of the two keys in 8 bit first-in first-out (FIFO) set the interrupt output line to the microprocessor for processing.

2.6 Decoding Unit

The decoding unit makes use of six 74LS154 chips which is a 4 line–to–16 line decoder to select a total of 80 lines. Figure 5 shows the decoding unit with the decoder chips connected in stack form to achieve the required number of lines. Each output line of the decoder is connected to a latching device (Intel 8212) so as to hold on the output immediately it is asserted. The outputs of the latching chips are each connected to the control inputs of the switching unit.

2.7 Switching Unit

The switching unit serves to make proper connections of the lines by linking up the appropriate speakers and microphones to the mixer. This is achieved by the use of the MC14053B chips. The MC14053B analogue multiplexer chip is a digitally controlled analogue switch that implements a-triple single-pole double-throw operations. Figure 6 shows the internal switching arrangement of the MC14053B Chip and the external connections.

2.8 Mixer Unit

The mixer is simply a summing amplifier, the output voltage being a vector sum of the input voltages. The most common way of mixing audio signals is by the parallel combination and is employed in the work being reported here. The mixer consists of three sub units namely audio pre-amplifier, summing amplifier and audio amplifier. The audio pre-amplifier makes

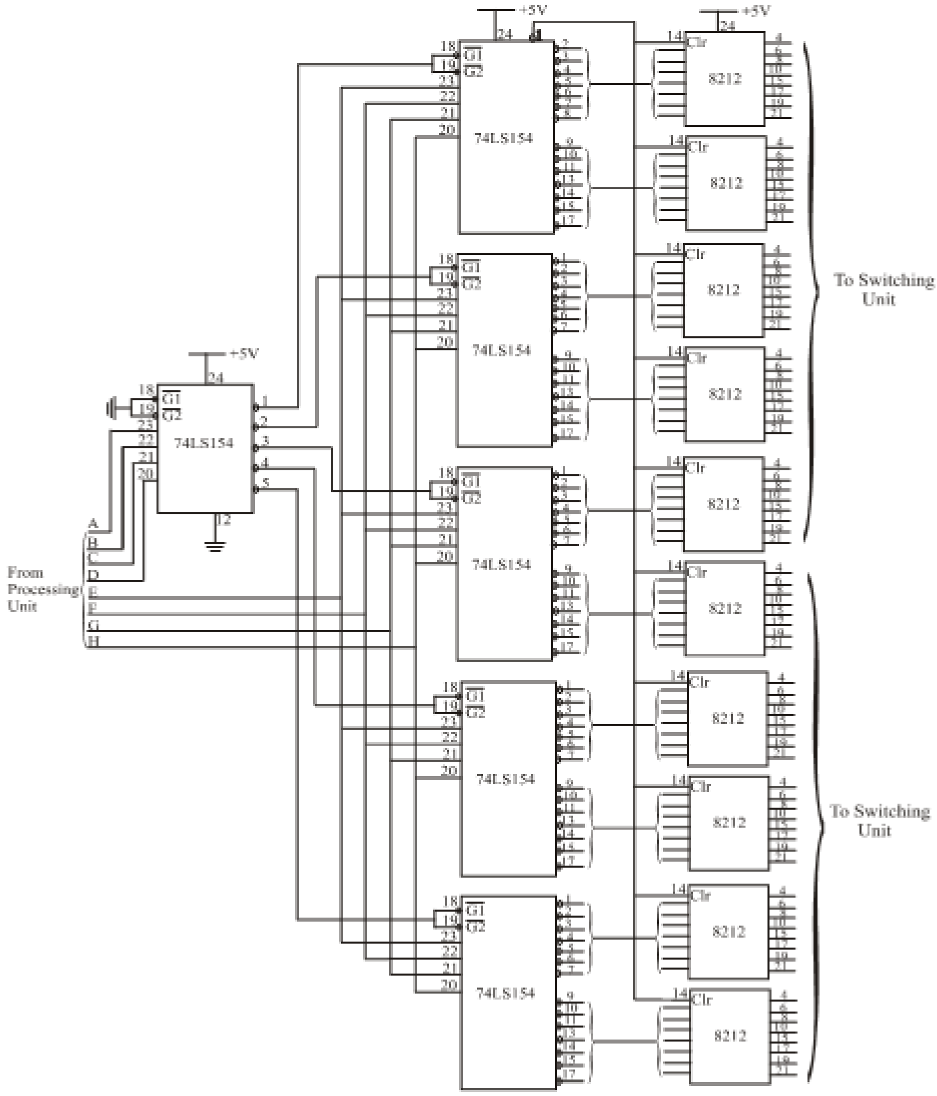

Figure 5: Line selector (decoding) unit

use of a two-stage transistor amplifier using BC148 connected in the common-emitter configuration (Bishop 1979; Meadows 1978). The choice of this configuration is based on the fact that the input impedance of the microphone is of medium range value.

2.9 Program Development

The program is written in assembly language for Intel 8085A CPU and is made up of two main sections — the main routine and the keyboard read routine.

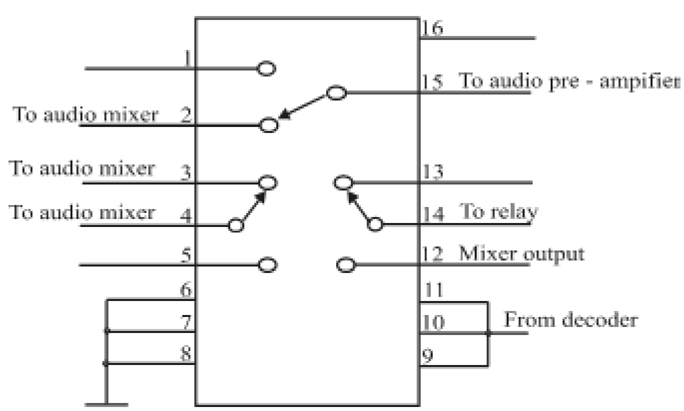

Figure 6: Internal switching arrangement of MC14053B

At power on or reset, the program starts from the memory address 0000. It starts first by initializing programmable input-output (PIO) port 1. Port A is initialized as an output while port B is initialized as an input port. Also, the command/status register is initialized as well as the stack pointer. The main program starts by setting the maximum number of people that are required to participate in a group discussion.

The keyboards are polled by unmasking the interrupts and enabling them. If any data is found, it is stored in the location 10F5H which has been set aside as the input buffer address. Also, as the data is being sent to the output, a look-up table is created where these data are being stored for further reference. Group connection is terminated automatically when the numbers of connections made are up to sixteen and manually when FFH is pressed in the keyboard. The program also checks the calls already made before connecting the second and third groups. At the end of any group discussion, the code combinations 00, 01 and 02 are used to disconnect all the engaged lines for groups 1, 2 and 3 respectively. The program for the keyboard read routine is designed in such a way that when a key is pressed, the value is stored in location 10FEH. The value in this location is then moved to the accumulator when needed.

2.10 System Operation

The system starts operating once its power supply switch is set. The program starts at location 0000, thus the keyboard is continuously scanned for inputs. By pressing the appropriate keys on the key-board, a string of binary digits appear at the output port. These binary digits are fed to the decoding unit which selects one out of eighty lines. The selected output is latched to the switching unit which in turn connects the appropriate line corresponding to the keys pressed on the keyboard.

The system continues to connect up lines until a total of sixteen lines are connected or the end of connection signal (FFH) is received. When any of these occurs the second group connections is automatically set and is ready to make connections. The procedure is repeated.

At the end of the group discussion, 00 is pressed in the keyboard by the participant that convened the meeting. This action clears or disconnects all the connections made on the group one. Similarly 01 and 10 disconnects all the connections made to groups two and three respectively. This is achieved by connecting the outputs of the decoders corresponding to these numbers to the *clear* inputs of the latches i.e. the 8212 chips.

A voice activated switch is incorporated in the design. This serves to isolate the participant's speaker when he is speaking in order to avoid the microphone picking up the output of the speaker as this may cause echo. Many other ways of suppressing echo and leakage currents in teleconferencing systems abound (see http://svconline.com/mag/avinstall_audio_teleconferencing_2/). The voice activated switch in this design makes use of the BT 151 thyristor.

2.11 Program Testing

The assembly language program was written using the Intel™ 8085A instructions set and was tested on the Microprocessor Applications Trainer (MAT 385). The program was loaded in the ROM of the MAT 385 starting from location 8000. The MAT 385 has one input and one output port that can be used without external interfacing. As a result of this, the program was modified so as to show all the outputs in port 22 and to use port 21 as the input port.

The switches of port 21 were arbitrarily set and the program was executed. Then two keys were pressed and the binary equivalent of the keys in first–in first–out order was displayed in port 22. This was repeated many times to ascertain the credibility of the program.

3.0 Conclusions

Teleconferencing system is already a well developed technology that enhances productivity both in industry and academic environments. Audio teleconferencing provides a reliable and cheap teleconferencing system suitable for use by many different groups of people in an area of common interest. The choice of the Intel 8085A CPU as opposed to other types of CPU's was based on availability rather than performance or reliability.

The attractive feature of the system designed in this work is that it is open ended. The modular design of this system gives room for future modifications and expansion because of the software base of the design. Only inputs and output ports are the hardware necessary for future expansion of the system. Further work will focus on interfacing this system to the Nigeria telephone network for wider area coverage.

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