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FIBRE OPTIC TELEPHONE - CENTRAL SWITCHING

AN OPTICAL FIBRE TELEPHONE SYSTEM
(CENTRAL SWITCHING AND LOGIC)

by

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LIST OF ILLUSTRATIONS

GLOSSARY OF SYMBOLS

Symbol	Description of Signal
${}_I A_X$	audio output from receiver X, and input to switching circuitry
BT_X	busy tone to X's audio line
$BUSY_X$	control signal to put busy tone onto X's audio line
${}_Y BUSY_X$	control signal signifying that X should get a busy tone because he has dialed Y
C_{OX}	output of X's dial pulse comparator
${}_X C_Y$	control signal gnnerated by station X dialing station Y
D_X	control signal signifying that someone has dialed X's number (D_X identifies X as called party)
${}_X D_Y$	goes HI when X dials Y (inverted output of X's 7442 decoder)
DP_X	dial pulses on audio output of X's receiver
f_{RX}	subcarrier receiving frequency of X's phone set
f_{TX}	subcarrier transmitting frequency of X's phone set
${}_A I_X$	inhibit signal from A to all stations other than A, preventing these stations from calling A
M_X	audio input to X's transmitter (modulator)
$M2_X$	decimal "2" output signal of X's 7442 decoder
MZ_X	decimal zero output signal of X's 7442 decoder
R_{SX}	X's sidetone resistor
$RING_X$	control signal to put ringtone on X's audio line
${}_X S_Y$	DELAYed version of ${}_X C_Y$

Symbol	Description of Signal
S_{XY}	control signal to close audio switches connecting X and Y
T_X	control signal to turn on X's C.P. transmitter

CHAPTER 1

INTRODUCTION

One of the main advantages of fibre optics is the large potential increase in information carrying capacity. Because of its higher frequency, light offers an increase of four orders of magnitude over microwave transmission. As well as this, fibre optics is advantageous for other reasons. These include crosstalk immunity, ground loop immunity, E.M.I. immunity, small size and weight, and longer repeater spacing due to the fibres low loss and wide bandwidth.

As an example of the state of the art of optical fibre communications, a system was installed in Chicago in 1977 over a ten kilometer length. It used injection lasers driven at a rate of 44.7 megabits per second. The fibre cable contained 24 fibres, each fibre having the capacity to carry 672 one way voice signals.

Many such systems have been recently built, and evaluation reports are just beginning to appear. Many areas and problems remain to be investigated. For instance, a recent article in the IEEE Journal of Cable Television points out the need for more research into optical splitters and taps. This need was one of the primary motivations for the present project.

For our M.Eng. on-campus project, my associates and I have chosen the design, assembly and preliminary evaluation of a telephone system designed for short or medium distance transmission over optical fibre.

For a successful completion of the project, this system was required to meet the following specifications:

1. Each end of a two party conversation must be able to transmit and receive simultaneously, as is the case with most public telephone systems. Such an arrangement is referred to as a duplex system.
2. This system should have most of the other characteristics of a commercial telephone system, such as:

a) there be a system of audio cues (dial tone, busy tone, ringing tone, etc.)

b) that no adjustments or technical knowledge be required of the user

c) that all of the interconnections be made automatically and be based on a simple number system

d) that all conversations be inaccessible to other parties

3. The system should be compact, and rely on portable power supplies at each station.
4. The system should make use of optical (as opposed to electrical) types or splitters, employ fibres bidirectionally (rather than using separate fibres for transmitting and receiving), and have all signals carried over a single fibre (i.e. a multiplexed system).
5. Due to the varying numbers of taps, different positions of stations, etc., the system should be able to handle a 60 dB optical loss between any transmitting and receiving points. As well, the receivers must be able to handle multiple signals differing by up to 40 db in optical strength.
6. Although only 3 stations are required to demonstrate a basic tele-

phone system, the parameters and design specifications are to be based on a system with up to 100 stations.

7. It should be possible to change cable lengths and interchange stations without extensive electronic adjustments.

The format of the optical signals was chosen to be analog. This was done for simplicity and because of numerous technical problems encountered when trying to use a clock sync signal over the same link as the data, and over varying lengths of travel. The analog electrical data signals would intensity modulate the light from a suitable optical source. This is different from amplitude modulation in that the modulated intensity can never be less than zero. If all of the optical sources are to operate on the same optical wavelength, then the multiplexing must be done on the electrical data signal. This is done by using radio frequency subcarriers of different frequency as the different data channels. For the sake of simplicity, the voice signals would be amplitude modulated onto the appropriate RF subcarriers. Because of the availability of off the shelf components for the citizen's band frequency range (26.5 MHz - 27.3 MHz), the subcarrier frequencies were chosen from this range.

Once the subcarrier multiplexing had been chosen, a number of solutions become possible to the problem of establishing a conversation link between stations. One solution is to have each station contain the logic necessary to establish a link, provide the audio cues, etc. and the link is made by the station initiating the call. For an N station system, this can be implemented by each station containing its own unique transmitting subcarrier channel but N possible receiving channels. The caller then receives on the called party's transmitting channel and tells the called party which receiving channel he should use. An N transmitter/1 receiver station cannot be used since a busy signal could not be produced.

Another approach is to use a separate "central" switching station where the links are made. The caller then dials the central station, identifying the called party and the central station activates the appropriate switched connections to link the caller and the called party. It is likely that there would be enough optical reflection in the system such that a significant fraction of the signal transmitted from a station will return to the receiver. If the receiving channel and transmitting channel for each station are the same, this reflected signal could swamp out the conversation. This could be avoided by using a talk/listen switch, but this would violate requirement #1. Instead, separate transmit and receive channels could be used. This is the case with the finished system. Now, 2N subcarrier cahnnels are needed for an N station system.

This second approach was chosen for a number of reasons. With the central processor (C.P.) system, extra stations can be added without changing the existing stations. The individual stations are also much simpler, and this system as a whole is more easily adaptable for linkup with a commercial telephone system. Also, it is the cheapest to produce.

This C.P. version is similar to a commercial telephone system. On the commercial system, the various "stations" produce different voice signals on different wires leading to the central switching unit. These signals are all in the same frequency range (audio baseband), and the central switching unit puts the voice signals onto the appropriate outgoing wire. In this optical system, the different voice signals occupy different frequencies, but all on the same "wire". The C.P. then switches the voice signals onto the appropriate outgoing frequency.

The present system used near infrared LED's as the optical source. Either PIN photodiodes or avalanche photodiodes could be chosen as the detector. To avoid the high voltage power supplies required by an avalanche photodiode, PIN photodiodes were chosen.

Initially, multifibre optical cables were to have been used because the availability of couplers and the ease of making splitters. However, the recent development of fused splitters and couplers for single fibres offer lower loss and more flexability in splitting ratios. For these reasons, single fibre cable was used. Because of the short distances over which this cable would be used, dispersion is not a problem. Therefore, the choice between graded index or step index fibre was made only on availability.

There are three possible layout configruations for the optical cable: the star, the line (main trunk) and the tree (a combination of the star and the line).

In the star system, the losses increase linearly with N and all of the signals are roughly the same amplitude at a given receiver. However, the layout is not very flexible, and requires a larger quantity of fibre than the other two layouts.

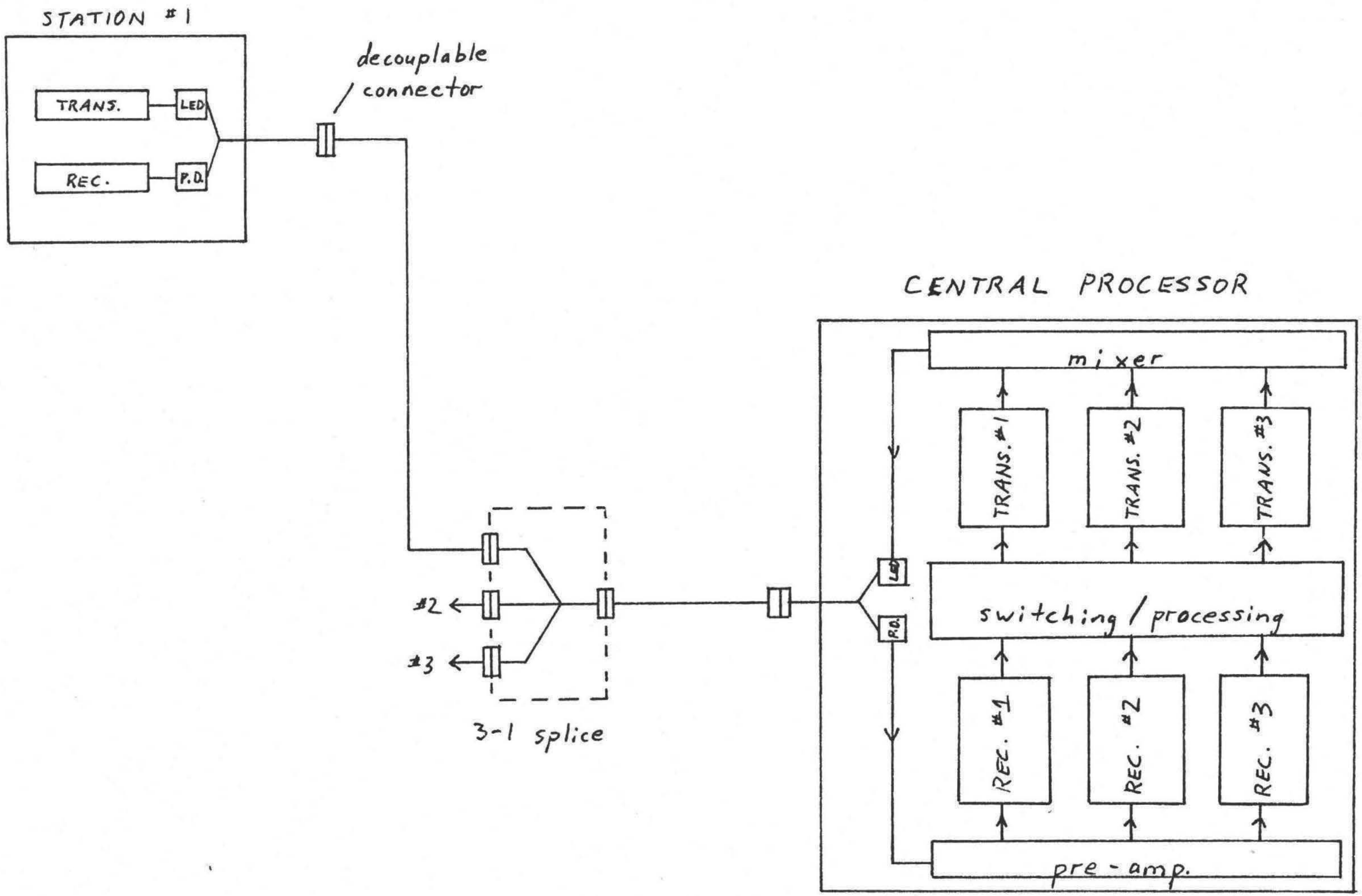


Fig. 1 Layout of Optical Fibre Telephone System

The linear, or main trunk, configuration requires the least amount of cable. However the dynamic range of the signals and the loss at the weakest station increase exponentially with N, due to consecutive tapping.

The tree configuration, which is normally used for CATV, has most of the advantages of the star and linear configurations. This was the configuration chosen.

As shown in Fig. 1, the geometry of the fibre in this tree layout takes the shape of a multi-branched "Y". The central processor which does the switching between the various subcarriers is situated at the base of the main arm of the "Y". A single fibre enters each of the telephone stations and the central processor. Inside these units, a fused fibre splice divides the light into two separate fibres. At the end of one of these fibre segments is an LED. Suitably modulated light is produced here and propagates out through the 2-1 splice into the tree. The other fibre segment runs into a photodiode. These detect the signals that have been modulated onto the light coming off the tree.

Each telephone station has an optical receiver consisting of a photodiode followed by an A.M. radio receiver and an optical transmitter consisting of an infrared LED at the output of an A.M. radio transmitter. A telephone station identifies the station to which he wants to speak by means of an ordinary telephone dial switch. This switch is used to interrupt the subcarrier coming from the calling station. The C.P. senses these interruptions and makes the necessary switched connections.

The central processor also uses a photodiode in its optical receiver and an LED in its optical transmitter. However, here there

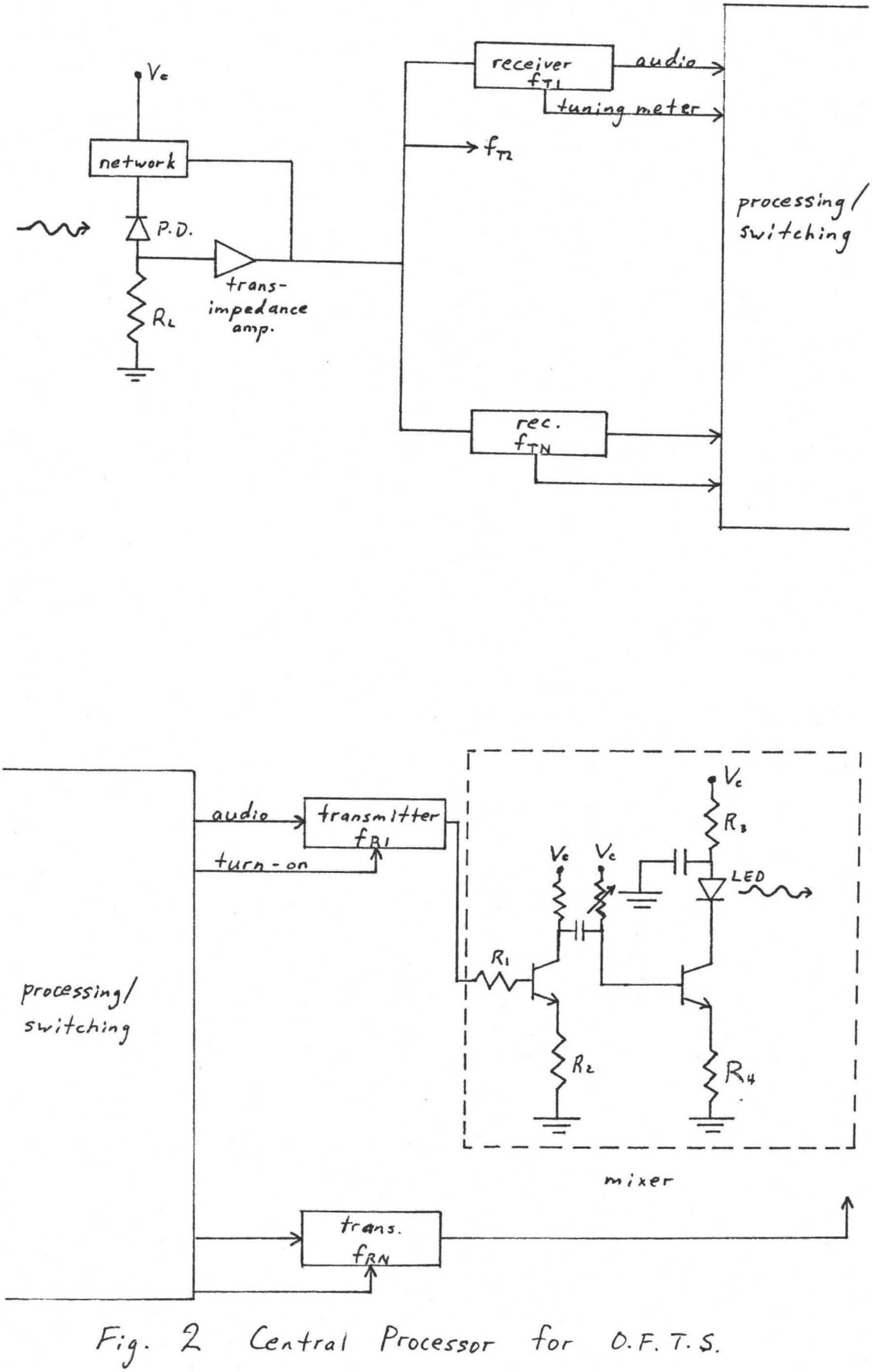


Fig. 2 Central Processor for O.F.T.S.

are N radio receivers and N radio transmitters interfacing with these opto-electronic components. In the C.P., there is a receiver/transmitter pair corresponding to each of the N stations. The C.P. receiver for a station is tuned to the subcarrier that the station transmits on and its C.P. transmitter is tuned to the subcarrier that the station receives on. The central processor is shown in Fig. 2.

Between the radio receiver bank in the C.P. and the radio transmitter bank, there is a switching network. When one station dials the phone number of another station, the C.P. counts the subcarrier interruptions. With this information, the proper audio switch is closed to connect the audio output of the C.P. receiver corresponding to the calling station with the audio input of the C.P. transmitter corresponding to the called station. Another switch is simultaneously closed to connect the reverse path. The same control signal that operates the audio switches also turns on the C.P.'s transmitter to the called party. When the called party's phone senses this subcarrier turn on, the phone starts ringing. When the called party answers his phone, the ringing stops and two way conversation takes place through the C.P. As well as the basic network of audio switches, the C.P. also contains the TTL logic controlling the switching and the various audio cues (dial tone, busy tone, ring tone). This logic is such that these tones are heard under the same circumstances as in a commercial telephone system.

The work on this project was divided into three sections:

(1) switching logic and peripherals - J. Goodwin

(2) optical fibre, splitters and couplers - G. Duck

(3) analog electronics - A. Jurenas

Parts 1, 2 and 3 are described in detail in the reports of Goodwin, Duck and Jurenas. The system with three telephones, C.P., power supplies and interconnections is operational, and is now undergoing detailed evaluation. The latter work is being done by V. Tzannidakis, who will report on it at a later date.

CHAPTER 2

The Central Processor

The circuitry contained in the central processing unit is designed to simulate the behaviour of an ordinary telephone system. It switches calls, ensures uninterrupted conversations and has status tones, i.e. dial tone, busy tone, etc. The inputs to this "audio system" are the voice outputs of the individual receivers, some means of sensing whether or not a particular user's "phone" is in the cradle and signals coding for the number that was dialed. The outputs of this system are the audio inputs to the transmitters and some sort of signal to cause the called party's "phone" to ring. The following is a description of the circuitry that was used to perform these functions.

i Switching Matrix

The first consideration in a communications system as described above is the ability to make the connection between the caller and the called party. Each "phone" station has only one transmitting frequency (f_{TX}) and one receiving frequency (f_{RX}). Therefore there must be some way to identify the party to be called so that the proper connection can be made. This is the same situation as in a regular telephone system where all conversations are baseband, but originating from different wires. In both cases, a dial switch can be used to identify the called party by using a pre-arranged "phone number". A commercial telephone dial switch is normally in the closed position. Upon its release after dialing a digit, the switches opens and closes rapidly. The number of togglings of the switch corresponds to the digit that was dialed. As mentioned earlier, the dial is used to interrupt the radio subcarrier. These RF interruptions are counted to produce a signal which then activates the proper switched connections between the caller and the called party.

Each station in the telephone system must have the ability to be connected with each of the other stations. The least sophisticated control circuitry consists of N(N-1) hard-wired switched connections, when N is the number of stations. Each switch in this matrix is normally open, and closes when a control signal is applied to it. As in a telephone system, the caller controls the connections for the conversation. Therefore the switch (es) which connect stations A and B must be able to be activated by either A or B, depending on which is the caller at that time.

The layout of a segment of a matrix employing single switched connections is shown in Fig. 3. If switch A-B has been closed by either A or B, the output from the central processor's (C.P.) receiver that picks up A's transmitting frequency (f_{TA}) would be connected through the switch to the input of the transmitter to B (f_{RB}). The other half of the conversation goes through the switch in the other direction. The signals going through these switches could be either audio or the intermediate frequency (I.F.) output of the receivers. No audio output from the C.P. to the external world is needed. It would therefore

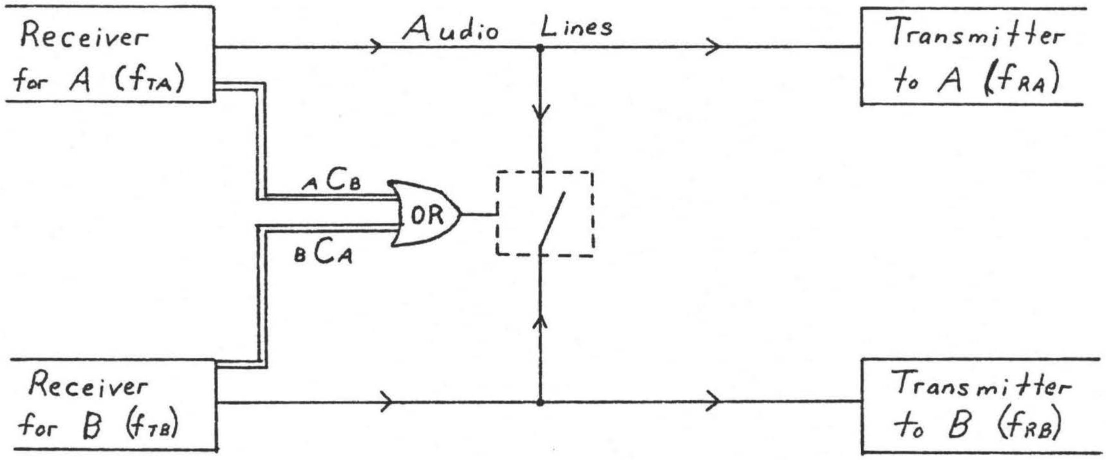


Fig. 3 Single Switch Connection

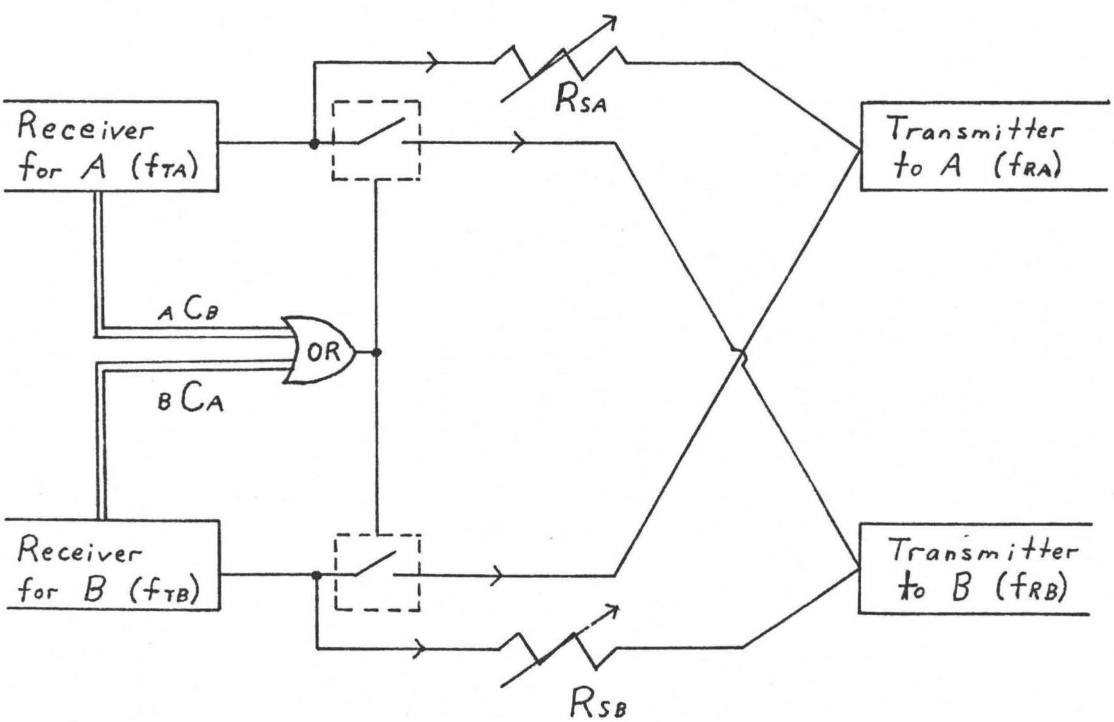


Fig. 4 Double Switch Connection

be possible to operate the switching at I.F. and avoid any extra circuitry to produce audio in the C.P. One complication with this method is that the various status tones i.e. dial tone, which are produced in the C.P., would have to be produced at I.F. However, for large N, there could be a considerable saving in circuitry by switching at I.F.

The N = 3 system was constructed with the switching matrix operating at audio frequency. The deciding factor in this was standardization. If the audio output is used in the C.P., its receiver and transmitter modules can be identical to those in the individual stations.

Instead of the 1-switch connection matrix, a 2-switch version was used in the working system (Fig. 4). The reason for this is that a sidetone is much easier to implement in the 2-switch version. The sidetone in an ordinary telephone system is produced and regulated in the individual telephones by means of a bulky transformer. In Fig. 4, it can be seen that the sidetones are produced by a single resistor. This approach reduces the complexity of the individual stations. As well, the removal of the sidetone transformer from the commercial telephone frees some valuable space. In practice, the sidetone is produced by simply adjusting R_{SX} until the proper level is reached. The audio from A back to A will then have the proper attenuation relative to the audio from B to A. This occurs when R_{SX} is several times the audio input impedance of the transmitter.

The switching elements used in the working system are CD4016 CMOS quadbilateral switches. These are produced in 14-pin integrated circuit chips with four switches per chip. The control inputs can operate from TTL logic and supply levels to yield a closed resistance of about 3 KΩ for the switches.

ii Inhibit Signal

At first, it was thought that the hardware implementation of the logic controlling the switching matrix would be quite simple. Eventually this had to be modified somewhat as described at the end of this section. It was also decided that all receivers in the system would always be on but that the transmitters would be turned on only as needed. With this in mind, a typical conversation (A calling B) would proceed as follows:

1) A picks up his telephone from the cradle - this turns on the transmitter in A's station (operating at f_{TA}).
2) A's reciver in the C.P. (at f_{TA}) senses the subcarrier at f_{TA} turn on - this turns on the C.P.'s transmitter to A (operating at f_{RA}).
3) A dials B's number - the C.P. receiver at f_{TA} counts the interruptions of this subcarrier - after the dial has been completed, this counter has identified B as the party to be called, and it actuates two signals - one to close the audio switches connecting A and B, and another signal to turn on the C.P.'s transmitter to B (operating at f_{RB}).
4) B's station receiver senses the f_{RB} subcarrier come on and this causes B's telephone set to ring.
5) B answers the phone by lifting the handset from the cradle - this stops the ringing and turns on B's transmitter to the C.P. (operating

at f_{TB}) - The conversation is now connected. -

6) If B hangs up first, his transmitter to the C.P. is shut off, and A no longer hears anything (but the switched connections are still being held closed by A).
7) If A hands up first, his transmitter to the C.P. is shut off - this causes A's counter in the C.P. to reset and so the audio switches to B relax back to their open state - this means B no longer hears anything but sidetone even though his C.P. transmitter is working.

Besides this simple procedure, the system must react to a 3rd party in the same manner as a regular telephone system. For instance, if there is a conversation between A and B, and C dials B's numbers, C must not be able to interrupt the conversation or listen to it. As well, C should get a busy tone to inform him that B is busy. To implement this, when a station such as B is engaged in a conversation (as sensed by its C.P. receiver picking up the f_{TB} subcarrier from the station) a signal could be sent to the C.P. receiver of all other stations to prevent any interruption of the conversation. This signal would go to the counters following the receivers and would block the control signals which are dependent on the counter output. For instance, if C had dialed B, the "inhibit" signal from B would block C's control signal and prevent it from operating any audio switches. The presence of the inhibit signal from B,and C's counter output identifying B could be used to produce a busy tone for C.

An additional refinement needs to be made to the logic described above. If it is implemented exactly as described above, a problem will result because the called party will inhibit the callers control signal as well as those of the 3rd parties it is intended to inhibit. A solution to this problem is to have the counter output blocked only if the inhibit signal is applied before that count appears. If the count appears, and an inhibit signal is subsequently applied by the called party, the inhibit signal will have no effect. The logical implementation of this process - a "first come-first served" latch - is shown in Fig. 5.

The functioning of this latch is easily explained. If ${}_{B}I_{A}$ is LO (not inhibited), point G is LO and so gate Q_1 is enabled. Therefore ${}_{A}C_{B} = {}_{A}D_{B}$ so that if A dials B and ${}_{A}D_{B}$ goes HI, the connecting audio switches are activated by the control signal ${}_{A}C_{B}$, and B's transmitter is also turned on. When this dial has occurred (i.e. ${}_{A}D_{B}$ goes HI), we see that ${}_{A}C_{B}$ disables gate Q_2 so that the inhibit ${}_{B}I_{A}$ will have no effect when it goes HI. If initially ${}_{B}I_{A}$ is HI and ${}_{A}D_{B}$ is LO, G is HI and so gate Q_1 is disabled. This will be the case when B is in a conversation with a station other than A. If A then dials B and ${}_{A}D_{B}$ goes HI, C remains LO because Q_1 is disabled. Since G is still HI, $BUSY_A$ goes HI after this dial and this causes A to hear a busy tone.

iii Counting Circuitry

As was mentioned earlier, the telephone dial is used to interrupt the R.F. subcarrier from the station to the C.P. As the R.F. is interrupted, the decay of its amplitude over a finite period of time

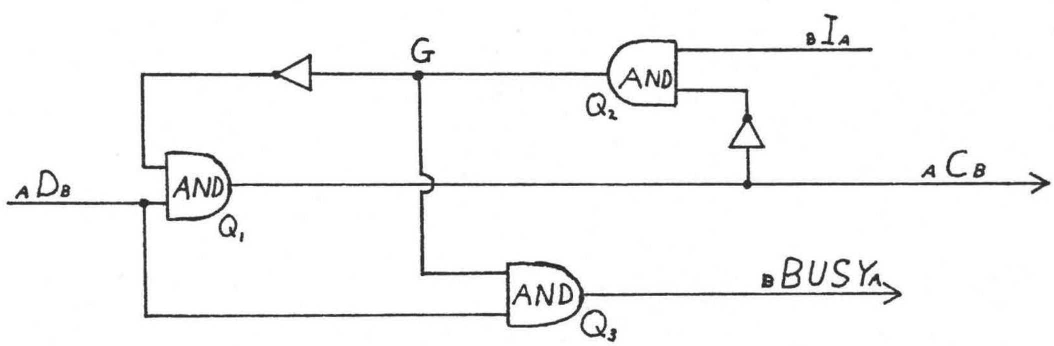


Fig. 5 Inhibit/Control Circuit

gives a finite bandwidth to the subcarrier. At the C.P. receiver, this appears as a large spike at the audio output. It is these transient spikes that are actually counted in the C.P. to determine the number that was dialed.

Basically, the transient spike counting circuitry consists of a shaping network at the audio output to detect and shape the audio-spike into a smooth pulse for counting. Following the shaping network is a comparator to take the smoothed audio spike and produce a large, square pulse. The 339 quad comparator that is used has a specified response time of about 1 microsecond. These pulses are to be counted by a 7493 binary counter, which takes TTL signals as input. The 339 output would respond too slowly to the 7493 and could lead to an erroneous count. For this reason, the comparator output is put into a 7413 Schmitt trigger whose TTL compatible output then goes into the counter. The binary output of the 7493 counter is converted to decimal form by

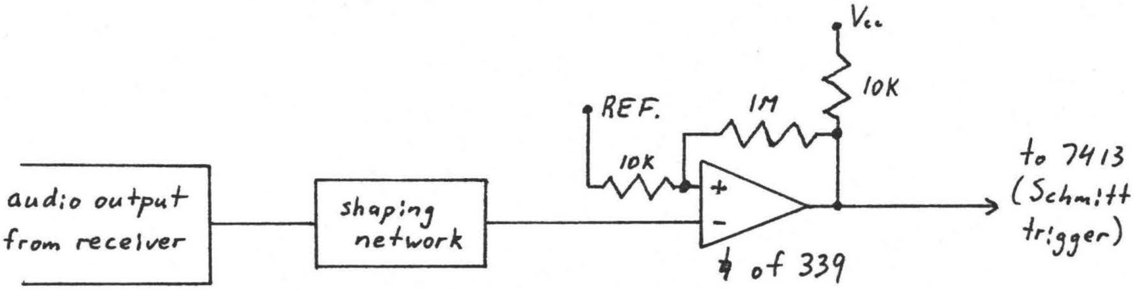


Fig. 6a Dial Pulse Detecting Circuit

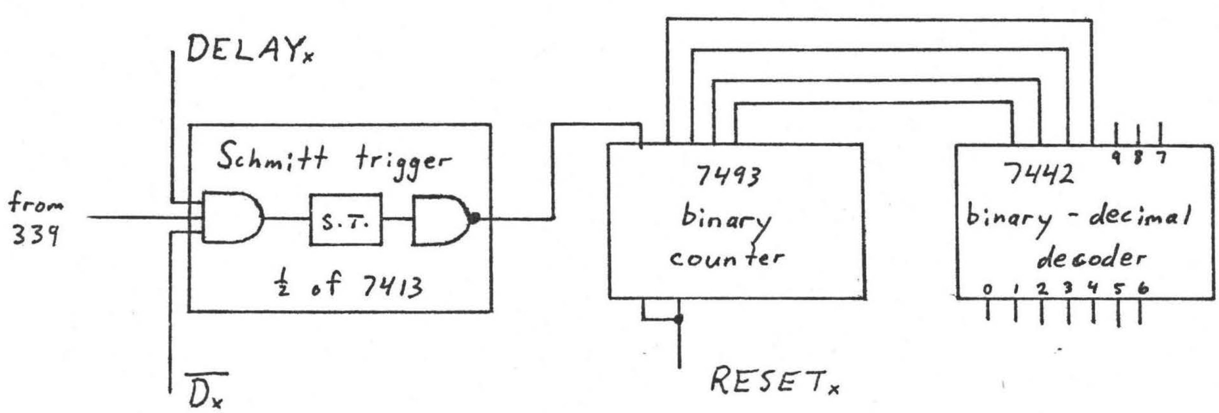


Fig. 6b Counting Circuitry

by a binary-decimal decoder.

The entire counting circuit is shown in Fig. 6a,b. The 7413 Schmitt trigger has an inverting stage in it. An order that the input pulses to the 7493 counter have the same sense as the audio transients, an inverting comparator circuit is used. This circuit was built with a feedback path to provide about 50 mV of hysteresis for increased noise immunity. The reference level for this comparator is set at about 500 mV. The decimal outputs of the 7442 decodes are used to provide the xDy signals for the inhibit circuit. The sense of the 7442 outputs is inverted i.e. if a binary "5" is input, the decimal "5" output would go LO while all of the other decimal outputs would stay HI. Signal xDy is actually just the inverted level of the decimal "y" output of "x"'s decoder.

The extra signals shown in Fig. 6b (i.e. DELAYx, RESETx and $\overline{Dx}$) will be explained later. For the purposes of counting, the DELAYx and $\overline{Dx}$ can be taken as HI and RESETx as LO (such that the counter is enabled).

iv DELAY Signal

The $_XD_Y$ signal cannot be used directly as the control signal for the connection between X and Y. If this was the case, then all of the other stations which are not inhibiting at the time and whose station number lies between zero and the number that is dialed will be successively called as the audio spikes appear one after the other. Audio switches to these stations will then be briefly activated as the

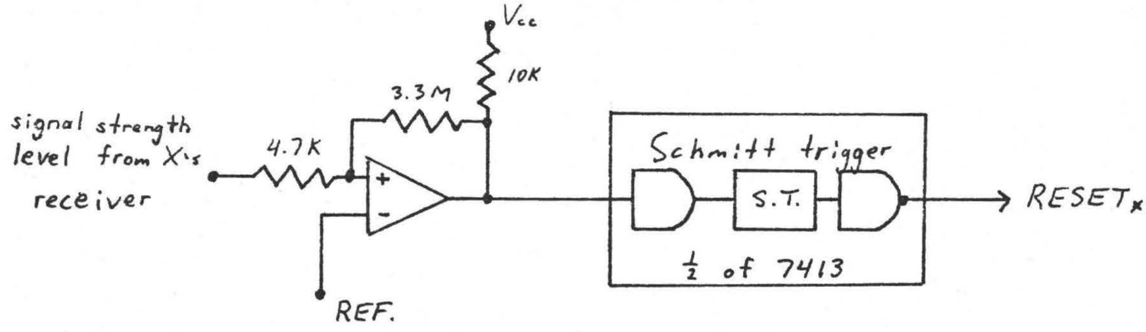


Fig. 7 Generation of RESET Signal

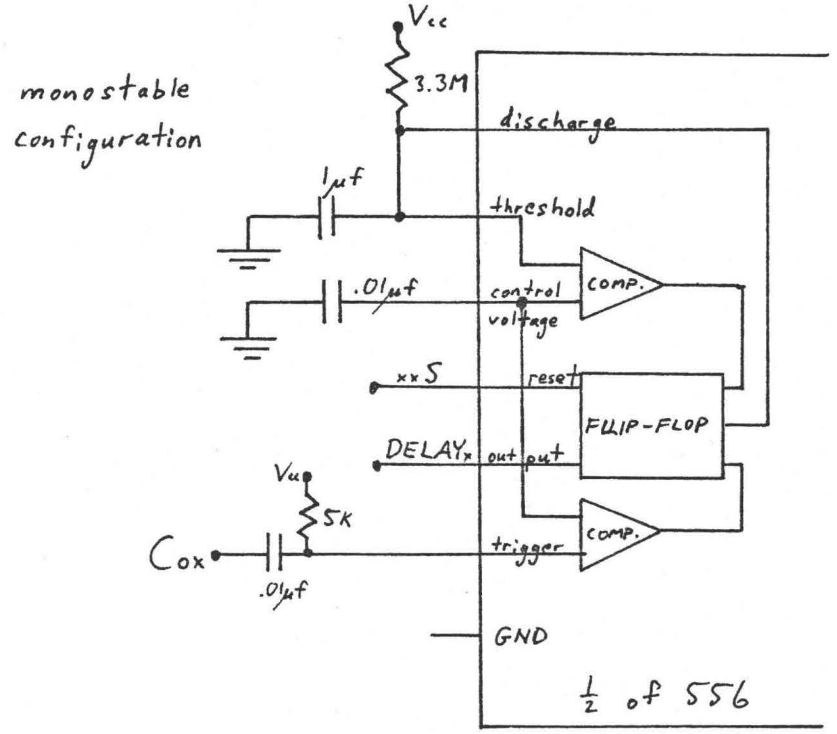


Fig. 8 Generation of DELAY Signal

count charges. As well, transmitters will be turned on, causing phones to ring. One way to avoid this situation is to delay the effects of these control signals until such time that it is certain that all of the dial pulses have arrived. A delay of 2 to 3 seconds should be sufficient for this. The DELAY version of the control signal ${}_{A}C_{B}$ is designated as ${}_{A}S_{B}$ (A's signal to control the audio switches to B).

As well as delaying the control signal, this delay should be used to supply a gating signal to the counting circuitry. This circuitry would be enabled for only the period of the 2 to 3 second delay. The purpose of this is to prevent any spurious counting activated by the audio of the conversation after the dial has been established. In order to count all of the dial pulses, this DELAY signal must be activated by the first dial pulse. This signal is applied to one of the AND gate inputs of the Schmitt trigger, as shown in Fig. 6b.

The 7493 counter counts on the falling side of a pulse applied to its input. In view of this, and the fact that the DELAY signal is applied as an AND gate input, this signal should behave as follows:

1) $DELAY_X$ should normally be LO
2) $DELAY_X$ should be activated to go HI on the rising side of the first dial pulse received - it must react quickly enough to be in the HI state by the falling side of the first dial pulse (so that the first pulse may be counted)
3) After a 2 to 3 second period, $DELAY_X$ should go LO again

A linear 555 timer chip could be used to generate such a signal. Fig. 8 shows how this I.C. is used to generate the required signal. The 555 is triggered by negative-going pulses. It is necessary to trigger the 555 at the start of the first count pulse so that DELAY is HI before the falling side of the first count pulse arrives. To do this, the 555 is triggered from the counting circuits comparator output. It is an inverting comparator, and so the initial rising side of audio spike will cause a negative-going transition at the comparator output. The RC combination in front of the trigger turns this negative going transition into a short negative going pulse which is applied to the trigger input. The value of RC is chosen to be short compared to the duration of the audio spikes.

The DELAY signal described above, when applied to the Schmitt trigger chip, would be sufficient protection against any spurious counting except for one thing. Any audio transient which is large enough to exceed the computer's reference will trigger a new DELAY gate signal in the same way as a dial pulse. This problem could be solved if the activation of $DELAY_X$ could be blocked after the train of dial pulses has been allowed in i.e. after the original $DELAY_X$ had gone LO again. If A is engaged in a conversation, one of the signals ${}_AS_X$ will be HI, and it would not have gone HI until after the original DELAY gate had passed. Therefore, the signal ${}_{AA}S = {}_AS_B + {}_AS_C$ applied to the RESET input of the 555 will prevent the further triggering of a DELAY signal. (The 555 reset must be LO to allow triggering). This triggering would only be allowed again after the caller has hung up. This resets the counter, and so any ${}_AS_X$ would then go LO.

The called party does not have this protection from spurious

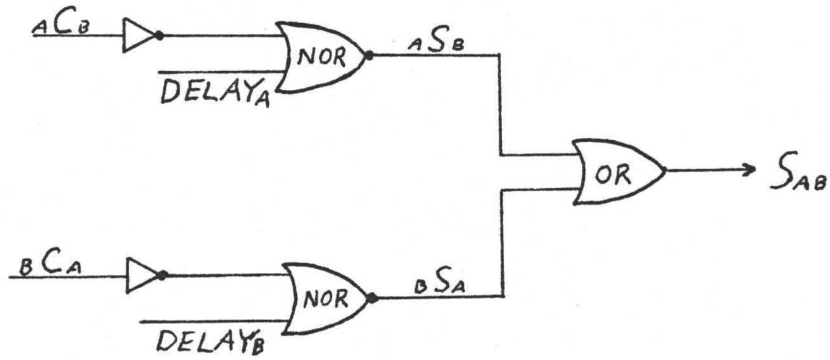


Fig. 9: Generation of Signal Sab

counting. Since he does not use the dial, he never activates a DELAY signal. However, it is required that this counter stay at zero. Although he is inhibited by the caller, any other station could be reached by a spurious count. To prevent this, his counting circuit should be disabled if someone is calling him. To implement this for station C, a signal $\overline{D_C} = \overline{{}_AC_C + {}_BC_C}$ is applied to one of his Schmitt triggers AND gate inputs, as shown in Fig. 6b.

As was mentioned in an earlier section, it is necessary to allow either of two stations to control both of the audio switches between them. A logical network to do this and to allow each of the input control signals to be independently delayed is shown in Fig. 9. S_{AB} is now the control signal that is applied to the switches of the 2-switch connection shown in Fig. 4.

V RESET Signal

To enable the counting of the dial pulses, the RESET inputs of the 7493 counter must be LO. This input must stay LO during the conversation to hold the count which controls the audio switches. After the caller hands up, the RESET inputs should go HI to reset the counter and prepare it for the next call.

The RESET signal in the existing system is derived from the 3088AE radio chips output for a tuning meter. Since the receiving frequency is fixed by the CB crystals, this output can be used as a "signal strength indicator". These levels were measured to be 70 mV for no input signal and 120 mV for an unattenuated signal as input. To produce RESET signal levels that are TTL compatible, the signal strength indicator level is applied to a 339 comparator. The other (negative) input is the reference, set at 100 mV. As in the case of the counting circuitry, the comparator output here is put through a 7413 Schmitt trigger to speed the transition up to TTL compatible rates. The output of the Schmitt trigger is then the actual RESET signal. In the individual stations, the signal strength indicator level is used to activate the ringing of the phones. This is explained in APPENDIX A.

As well as resetting the dial counters, the RESET signal is used in the production of the inhibit signal. As mentioned earlier, we want an inhibit signal to be sent out whenever a station is involved in a conversation. Even greater protection can be obtained if the inhibit signal is activated as soon as the caller picks up the phone. One way of sensing this is to have the inhibit signal activated by the signal strength indicator of the callers receiver. When the caller picks up his phone, his C.P. receiver senses the subcarrier come on and the signal strength indicator level goes HI. This will then cause the signal to go HI. Because of the way in which the RESET signal was produced from the signal strength indicator level, the inhibit signal could simply be the inverse of the RESET signal.

The above logic for generating the inhibit signal would be adequate for the caller. However, if A is calling B, this means that the inhibit signal from B to the other stations (${}_{B}I_{X}$) would not go HI until B answered his phone. This means that a 3rd party could successfully dial B while B's phone is already ringing due to A. To avoid this, B's inhibit signal should be HI when his CP signal strength indicator goes its high state or when someone else is calling him. This last condition is logically implemented by ${}_{A}Q_{B} + {}_{C}Q_{B}$ + etc. On the actual hardware, the inhibit signal is produced by:

$${}_{B}I_{X} = \overline{\mathrm{RESET}_{B} \cdot \overline{{}_{A}Q_{B}} \cdot \overline{{}_{C}Q_{B}}}$$

This can be shown to be equivalent to the logic described above.

vi Ring Latch

When one of the stations dials another number, we would like the called party's phone to ring and the caller should hear a ring tone. The ringing of the called party's phone should occur when the RF subcarrier for that station is sensed to be coming from the C.P. The ringing should be off while the called party's phone is off the hook. The commercial phone set is equiped with cradle switches, one of which could mechanically break the ringing circuit when the phone

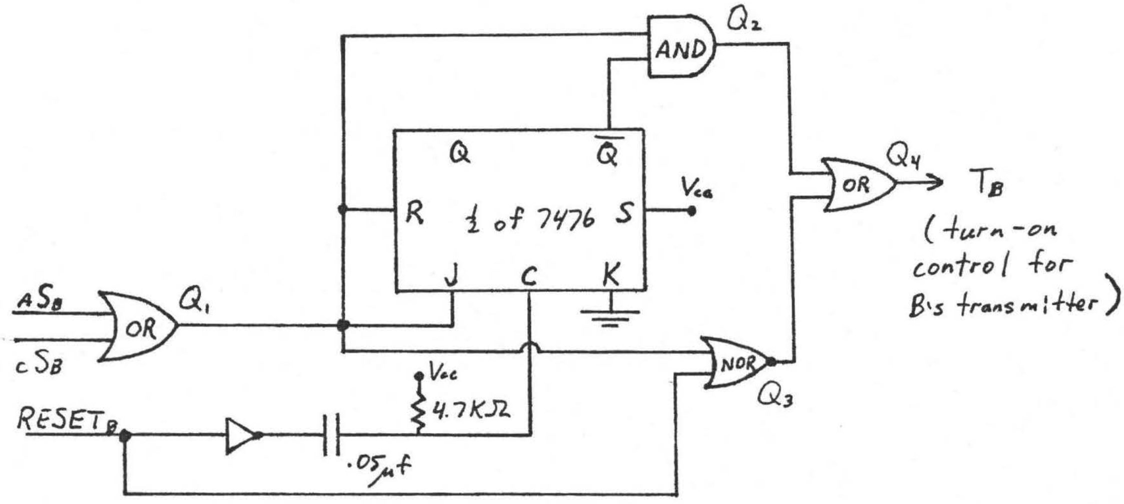


Fig. 10 Turn-on Latch Circuit

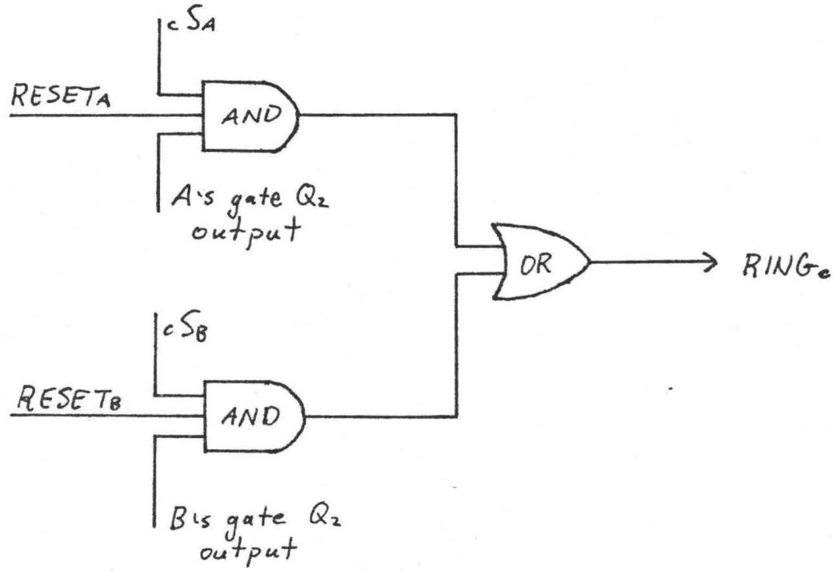


Fig. 11 Ring Tone Control

is picked up.

The actual ringing tone that the caller hears need not be synchronized with the ringing of the called party's phone, but it must be heard at the same time. The ring tone should be heard after the connection has been made to the called party, but only until the called party picks up the phone. In terms of existing signals, a station should hear a ring tone when it has an uninhibited control signal going out AND when there is no RF subcarrier coming from the called party's station (phone has not yet been picked up to turn on transmitter). A circuit implementing this logic is shown in Fig. 11. The RESET signal of the called station signifies whether or not the subcarrier from the called station is on. The third input, the called party's gate Q_2 output, will be explained later, and may be taken as being HI for now.

If only two inputs to the AND gates of Fig. 11 were used i.e. ${}_{A}S_{B}$ and RESET_{B} there would be a problem if the called party hung up first. The cradle switch would then be closed again and the subcarrier to the C.P. would shut off. Now the same logic conditions are met as at the start of the call. The caller would hear a ring tone and the called party's phone would ring. To correct this, some method is needed to decide whether or not the called party had picked up the phone earlier, and to enable the ring tone to the caller and the ringing of the called party's phone only if he had not. A latch of some sort could be used to enable the phone ringing and the ring tone when it is in the RESET state. This latch should be SET by the called party hanging up his phone and RESET by the caller hanging up his phone. If this is done, the phone ringing and the ring tone to the caller are blocked by the SET latch after the called party hangs up.

In order to reduce the amount of logic in the individual stations, the latches and associated circuitry were constructed in the C.P. The latch blocks the ringing of the called party's phone by blocking the subcarrier to it (which initiates the ringing). The latch then becomes part of the transmitter turn-on logic. Fig. 10 shows the latch circuit that was used.

This circuit produces the turn-on signal for a C.P. transmitter. In the case of the caller, we want his C.P. transmitter to turn on when the subcarrier from his station reaches the C.P. For instance, if A is the caller, this will be signalled by $RESET_A$ going LO. From Fig. 10, we see that when $RESET_A$ goes LO, Q_3's output goes HI and the turn-on signal for A's transmitter then goes HI (T_A). Note that the output of gate Q_1 is used to insure that A is the caller. That is, Q_1's output will be LO if and only if A is the caller i.e. no other station is calling him (both ${}_CS_A$ and ${}_BS_A$ are LO). If A was not the caller at that time, Q_3's output would be kept LO by Q_1.

If A is the called party, the latch itself must be used to make the turn-on signal behave properly. The 7476 is a J-K flip-flop. That is, on the falling side of a clock pulse at C_1 the data at the J, K inputs are transferred to the Q_1 $\overline{Q}$ outputs. (In the case where both J and K are LO, the flip-flop is locked and the outputs won't change as a clock pulse is applied). When the set input (S) goes LO, the 7476 goes into the set state (Q is HI, $\overline{Q}$ is LO) and when the reset input (R) goes LO, the 7476 goes into the reset state (Q is LO, $\overline{Q}$ is HI). When another station dials A, the output of gate Q_1 will go from LO to HI. Since there originally was a LO at the R input, the 7476 was in the reset state ($\overline{Q}$ was HI). Now R and J have gone HI, but the outputs stay the same because no clock pulse has been applied. Therefore the output of gate Q_2 will go HI, and the turn-on signal for A's transmitter goes HI. It can be seen from the diagram that a negative going clock pulse will appear only when $\overline{RESET_A}$ goes LO i.e. when A hangs up his phone. With A's turn-on signal now HI, his station senses the subcarrier from the C.P. and A's phone rings. When A picks up the phone, the subcarrier to the C.P. goes on and $\overline{RESET_A}$ goes HI. This does not affect the outputs of the 7476, and so the transmitter stays on. If the caller hangs up first, his counter resets and the output of gate Q_1 goes LO, turning off the transmitter to the called party. If the called party hangs up first, $\overline{RESET_A}$ goes LO and the signal to the clock input causes the J, K data to be transferred to Q_1 $\overline{Q}$. $\overline{Q}$ is now LO, and the transmitter to the called party turns off. Independent of what the called party does, it will stay like this until the caller hangs up, causing Q_1's output to go LO, and resetting the 7476. The circuit is now ready for another calling cycle.

Using the 7476, it is now possible to prevent the called party's phone from ringing again if he hangs up first. To prevent the caller from hearing a ring tone again under these conditions, the ring tone control logic should also take advantage of this circuitry. A logic network which does this is shown in Fig. 11. From this, it can be seen that C will hear a ring tone if he has dialed A (${}_CS_A$ is HI), AND A's phone is on the hook ($RESET_A$ is HI), AND A has not already picked up his phone and then replaced it (gate Q_2 output of Fig. 10 is HI). If the called party has just hung up, the first two conditions are met but the 7476 is now set so that the gate Q_2 output is LO. C will not hear a ring tone in this case.

vii Dial Tone

As in the conventional phone system, the caller should hear a dial tone after he has picked up his phone, but before he has started to dial. The simplest logic would then be to have the dial tone for a caller switched in when his dial pulse counter has a count of zero. This means that there is a dial tone to the caller before he picks up the phone, but since his C.P. transmitter is off, it wouldn't be heard.

This logic accounts only for the caller. The called party would always be hearing a dial tone, since his dial pulse counter is not used. To account for both cases, the logic should be that a dial tone is heard when the dial pulse counter reads zero AND that station is not being called by another station. This last condition means that, as required, the called party never hears a dial tone. Therefore, the logical implementation of the dial tone control signal is:

$$(\text{dial tone control})_X = \overline{(\text{zero count})_X} \cdot \overline{({}_AQ_X + {}_BQ_X + \ldots)}$$

The signal $(\text{zero count})_X$ is the signal available at the decimal zero output of the 7442 decoder. This is LO when the count is zero.

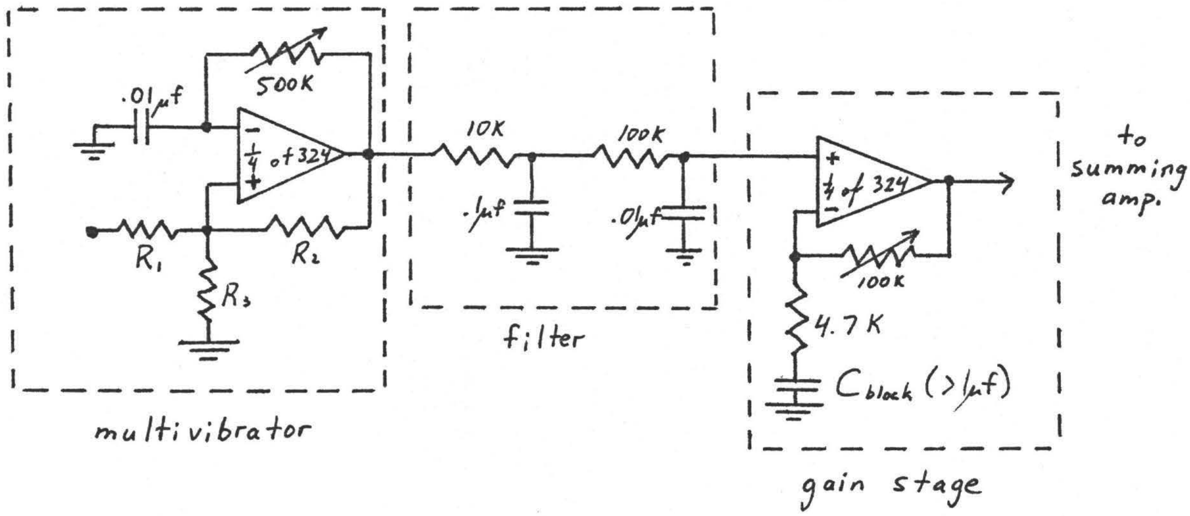


Fig. 12a Sine Wave Generator

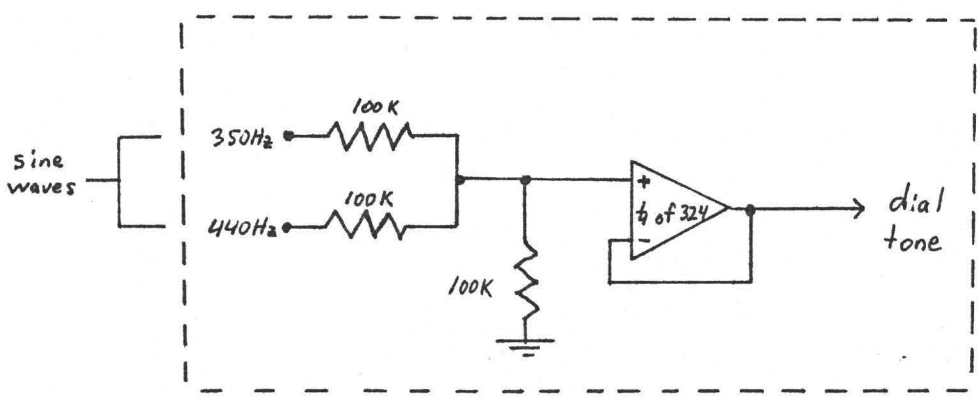


Fig. 12b Example of Summing Amplifier

viii Tone Generation

This optical fibre telephone system has busy, dial and ringing tones similar to those in a conventional telephone system. According to Talley[3], these tones consist of two separate frequencies:

Tone	Frequencies
dial tone	350 Hz plus 440 Hz
ringing tone	440 Hz plus 480 Hz
busy tone	480 Hz plus 620 Hz

As in a conventional telephone system, the busy tone and the ringing tone are interrupted. The individual frequencies are produced by square wave oscillators followed by low pass filters to suppress all but the fundamental frequency.

Four square wave oscillators for the four frequencies were constructed using a single OP-AMP each (see Fig. 12a). The OP-AMP used was a quad 324 chip. In Fig. 12a, the frequency of oscillation is determined by RC and the duty cycle is determined by R_1, R_2 and R_3. If the high state of the output went right to + 5 V, then a 50% duty cycle would be obtained with $R_1 = R_2 = R_3$. The oscillators should be run at 50% duty cycle because then the even order harmonics disappear and only the third, fifth, etc. are left. This makes it easier to isolate just the fundamental by using low-pass filters. When one of these oscillators was first built, a 61% duty cycle was observed. This caused a bothersome amount of even order harmonics. This duty cycle is due to the fact that the 324 rises to only 3.9 V in its high state. Fortunately, the duty cycle could be brought back to 50% by increasing R_1.

As was mentioned above, the frequncy of oscillation is determined by RC. Fixed 0.01 μf capacitors were installed for all four of the oscillators and a 500 K pot was used to provide a variable R. The different pots were adjusted to generate the four required frequencies (350 Hz, 440 Hz, 480 Hz and 620 Hz).

The output of the square wave oscillators feeds into a simple two stage RC low pass filter. Each stage has 6 db/octave attenuation for maximum isolation of the fundamental. The component values in the second stage are such that the first stage is only lightly loaded. For a 50% duty cycle wave, this two stage filter will produce an acceptable sine wave output at the oscillation frequency. This filter is shown in Fig. 12a.

An OP-AMP gain stage is needed after the filter for two reasons. First, a voltage gain of ten to twenty is needed to compensate for the attenuation of the fundamental by the low pass filter. Also, the OP-AMP will prevent the relatively low impedance of the audio inputs to the transmitter from loading the filter excessively. The DC component of the square wave oscillator output will come through the low pass filter unattenuated. For this reason, the gain required after the filter to recover the fundamental's amplitude will be more than enough to saturate the OP-AMP with DC. Capacitor C_{block} in the gain stage is used to provide a gain of one for the DC, while the gain for the fundamental is determined by the R_2/R_1 ratio. The gains of these final stages are adjusted so that all four frequencies are at 1.4 volt (peak-to-peak) with 2.4 volts of DC.

These frequencies must now be summed to produce the proper tones. This can be done simply with a single OP-AMP as shown in Fig. 12b. The output, being one half of the sum of the inputs, has approximately the same amplitude characteristics as each of the inputs.

Finally, the busy tone and the ringing tone must be interrupted. This is done with a multivibrator (constructed from a 556 timer chip) controlling a series switch at the output of the summing stages for these tones. The frequency and duty cycle of these interruptions are adjusted until the familiar tone is heard.

CHAPTER 3

Conclusions

For other larger systems, some changes would have to be made to the central processors switching logic. The present system uses commercially available two and three input gates. Larger systems would have to cascade these gates to perform some of the logic functions when $N > 3$. A more efficient, but initially more expensive approach would be to use customized multi-input gates. Microprocessors might also be a useful way to implement the switching logic for large systems.

When laying out the circuitry for large systems, customized circuit boards should be used to allow as much as possible of the circuitry to be modular (by station). This would save alot of time in debugging and servicing. If the market is large enough to warrant the cost, some of the logic functions i.e. inhibit circuit, ring latch would be integrated onto one chip. In fact, except for a few functions which combine signals from different stations, it may be possible to put all of the switching logic for one station on a single chip.

For systems with $N > 10$, the dialing and switching must be done in decades. The process of counting audio transients is well suited to the decade scheme. With one counter per decade, the counting and subsequent switching for high order digits could be done with the audio transient pulses passing through audio switches that had already been activated by the lower order digits. For very large systems, decade switching (especially on a demand basis) would result in a large reduction in switching circuitry per station.

APPENDIX A

Ringing of Stations Phone

The phones at the individual stations should ring when the sub-carrier to that station is sensed to be coming from the C.P. i.e. when the signal strength indicator of that station's receiver goes into its high state. As was mentioned earlier in the section on the RESET signal, the signal strength indicator level ranges between 70 mV and 120 mV. As for the RESET signal, a comparator is used here to detect the transition of the signal strength indicator from its low state to its high state. This is shown in Fig. Aa. One consideration here that didn't appear for the RESET signals in the C.P. is the reference signal for the comparator. In the C.P., this reference is taken from a resistive potential divider between the 5 V line and GND. The 5 V voltage regulator will hold the supply line very close to 5.0 V, i.e. within a few percent. However, the power supply for the individual stations is a 12 V D.C. battery. This battery voltage will drop from about 13 V to about 11 V as it is discharged. If a simple resistive potential divider was used here, the percent variation in the reference level would be a significant fraction of the range of the signal strength indicator level. Again, a 5 V regulator could be used, but this adds unnecessarily to the complexity and expense of the stations. Instead any Germanium diode biased to beyond its turn-on point (i.e. where the diode voltage has started to saturate at about 0.3 V) could be used to provide a more stable reference. This is shown in Fig. Aa.

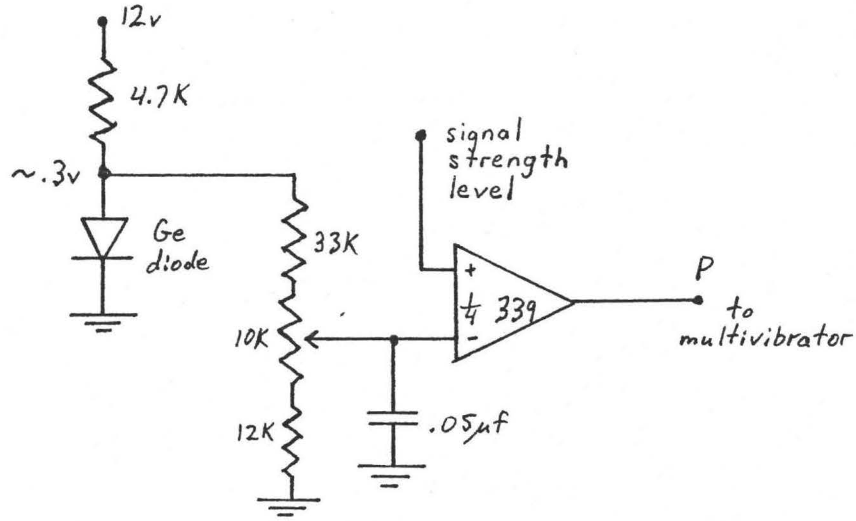


Fig. Aa Signal Strength Comparator

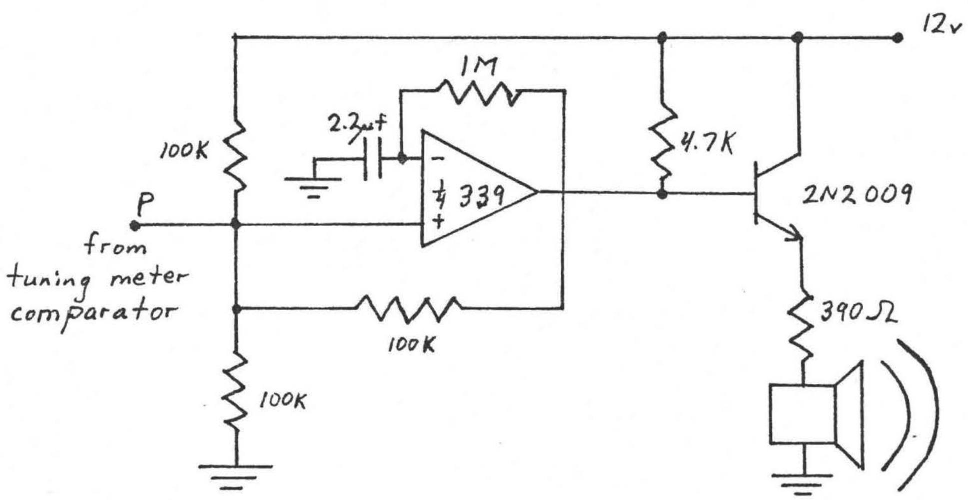


Fig. Ab Buzzer Drive Circuit

The circuitry as shown reduces the percentage variation of the reference level to about one quarter of that of the battery voltage. A 0.05 μf capacitor is needed at the negative input of the comparator to decouple any oscillation from the reference.

The output of the comparator is applied to the positive input of a multivibrator similar to the one described in the tone generation section. This multivibrator has a period of roughly four sec., to provide the "ringing" of the phone. When the comparator output goes low, the multivibrator is held off. This is the case when no subcarrier is coming from the C.P. and the signal strength indicator level is low. When the comparator output goes high, it looks like an open circuit to the multivibrator, and the ringing occurs normally.

A major problem with the ringing of the phone was finding a bell or buzzer that was loud enough to be installed inside the phone. The device must also be physically small because most of the space inside the phone set was already taken up by the receiver and transmitter. A small DC buzzer was found that seemed to meet these requirements. The Star SMB-06 buzzer is approximately 1 inch by 1/2 inch by 1/2 inch. With about 6 V applied, it draws 20 mA and has a sound output of 80 db at 20 cm. The buzzer drew too much current to operate directly from the output of the multivibrator. As shown in Fig. Ab, a transistor stage was then added to drive the buzzer. Using a quad 339 comparator for the comparator and multivibrator stages, the circuitry of Fig. A(a,b) was compactly constructed. The circuitry was installed on its own perforated board, approximately one inch across and one half inch high. The buzzer itself was firmly bolted to the baseplate of the phone.

APPENDIX B

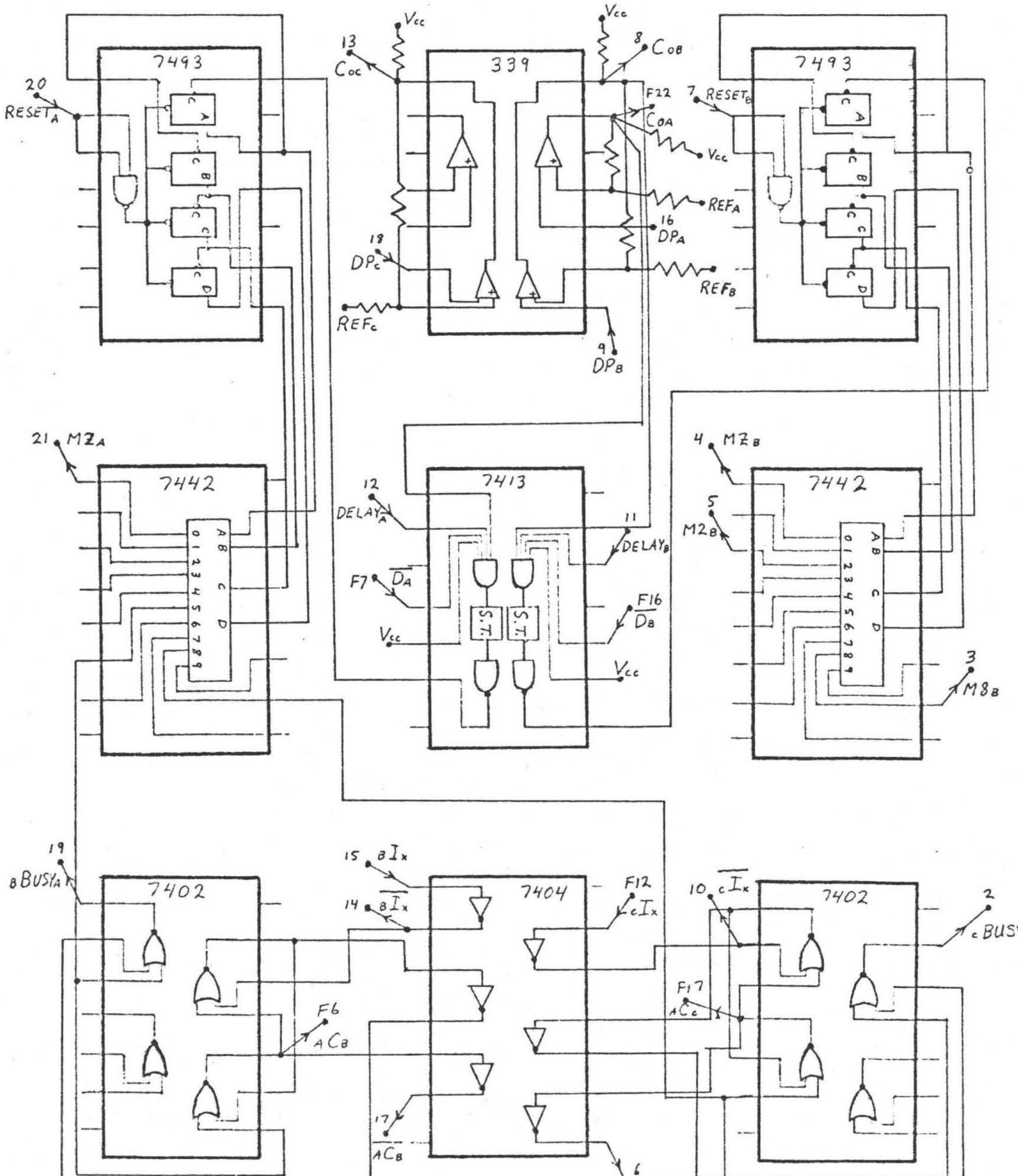
Circuit Layout and Trouble-shooting Guide

The circuitry for this portion of the system physically occupies five Realistic edge connecting circuit boards arranged in a stack. A separate board is used to generate the various tones. Layout diagrams are given here for the five stacked boards. Interconnections between boards are done by means of 44 pin edge connectors.

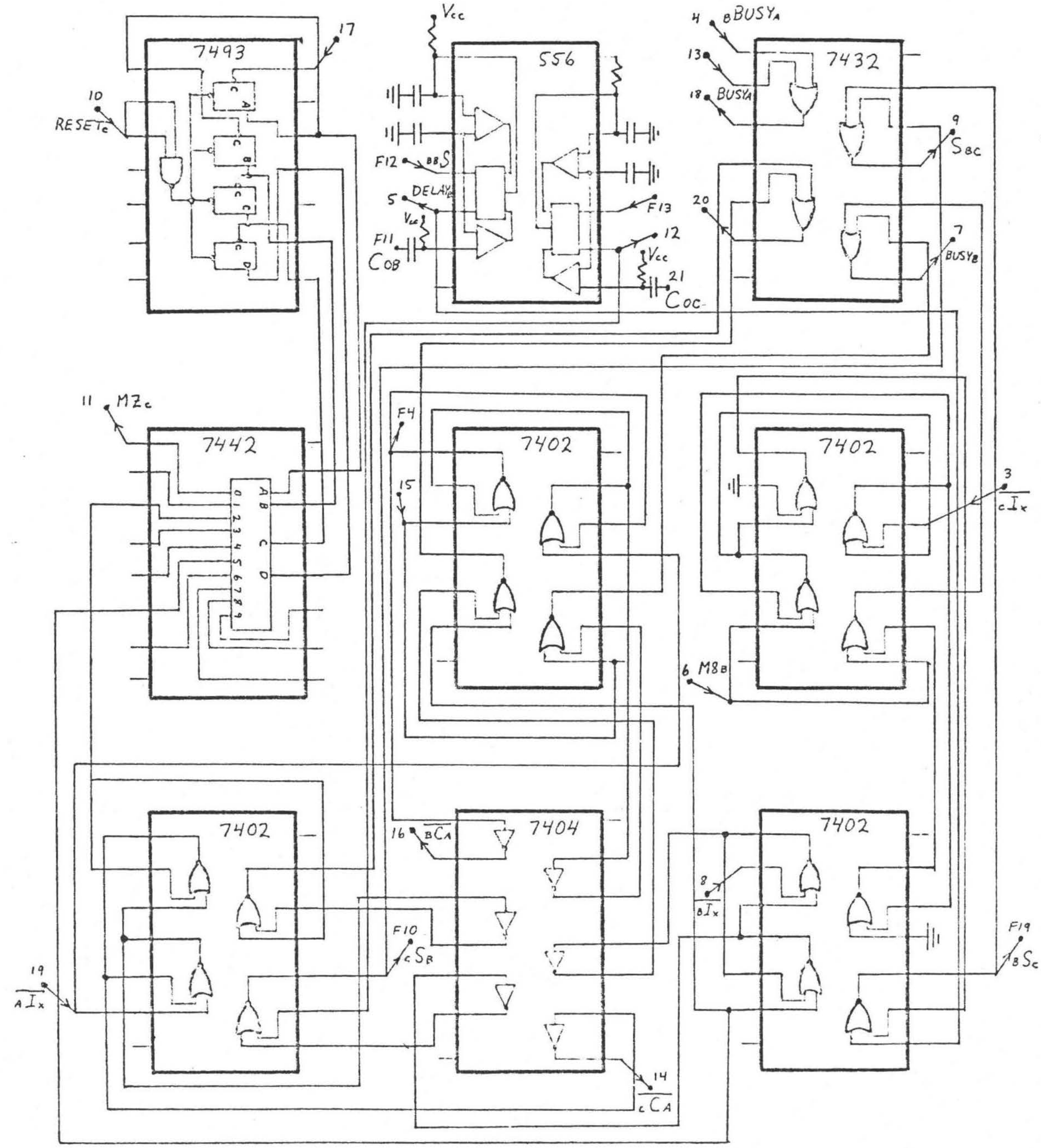
The layout diagrams shown in this section are viewed from the uncoppered site, with the edge connections towards the top of the page. The five boards in the stack are numbered starting at the top. Signals leaving one board through its edge connector are numbered and labelled, if possible, on the layout diagrams. The numbering of the pins on the edge connector is done as 1-22 from left to right for the terminals that are seen from the coppered (back) side of the board, when the edge connector is at the top. The 22 connections on the front of the board are numbered F1-F22 in a similar fashion. For all five boards, the 5 V supply (V_{cc} is on pin #1 and ground (GND) is at pin #22.

As mentioned earlier, labels are usually listed with any signal that makes an external connection with the board. Having external signals labelled by their logical significance (i.e. $RESET_A$) is advantageous during trouble-shooting, and especially debugging. However, if there is insufficient space for a label to be written or if the signals logical significance is not immediately apparent, the labels are omitted from the layout diagrams. These labels are then listed with

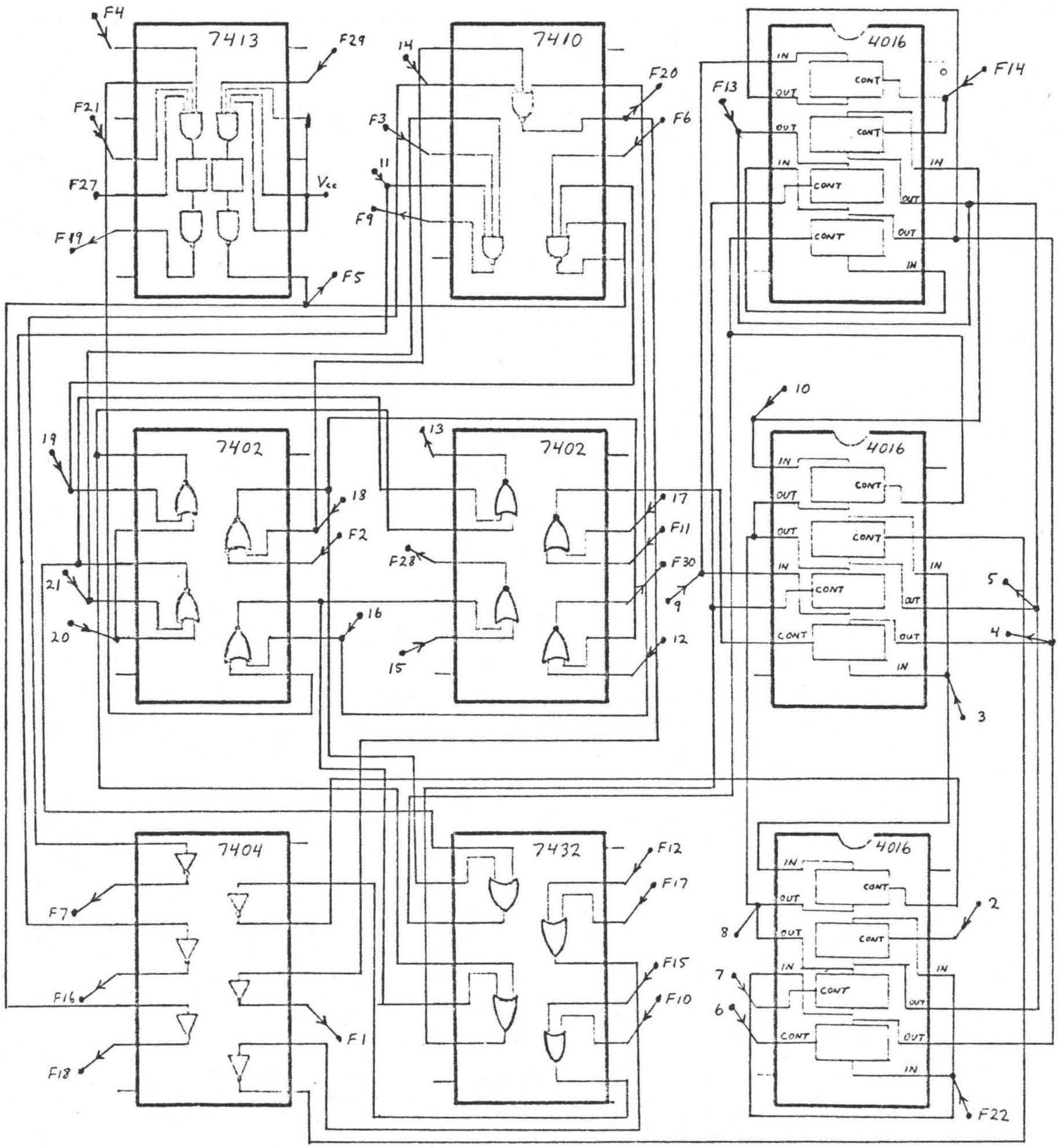
Board #1


7493
339
7493
20 RESET_A
13 C_OC
Vcc
8 C_OB
F22 C_OA
7 RESET_B
REF_A
16 DP_A
REF_B
18 DP_C
REF_C
9 DP_B
21 MZ_A
7442
7413
7442
4 MZ_B
5 M2_B
12 DELAY_A
11 DELAY_B
F7 D_A
F16 D_B
S.T.
3 M8_B
19 B BUSY_A
7402
15 B I_x
14 B I_x
7404
F12 C I_x
10 C I_x
7402
2 C BUSY_A
F17 A C_C
F6 A C_B
17 A C_B
6 A C_C

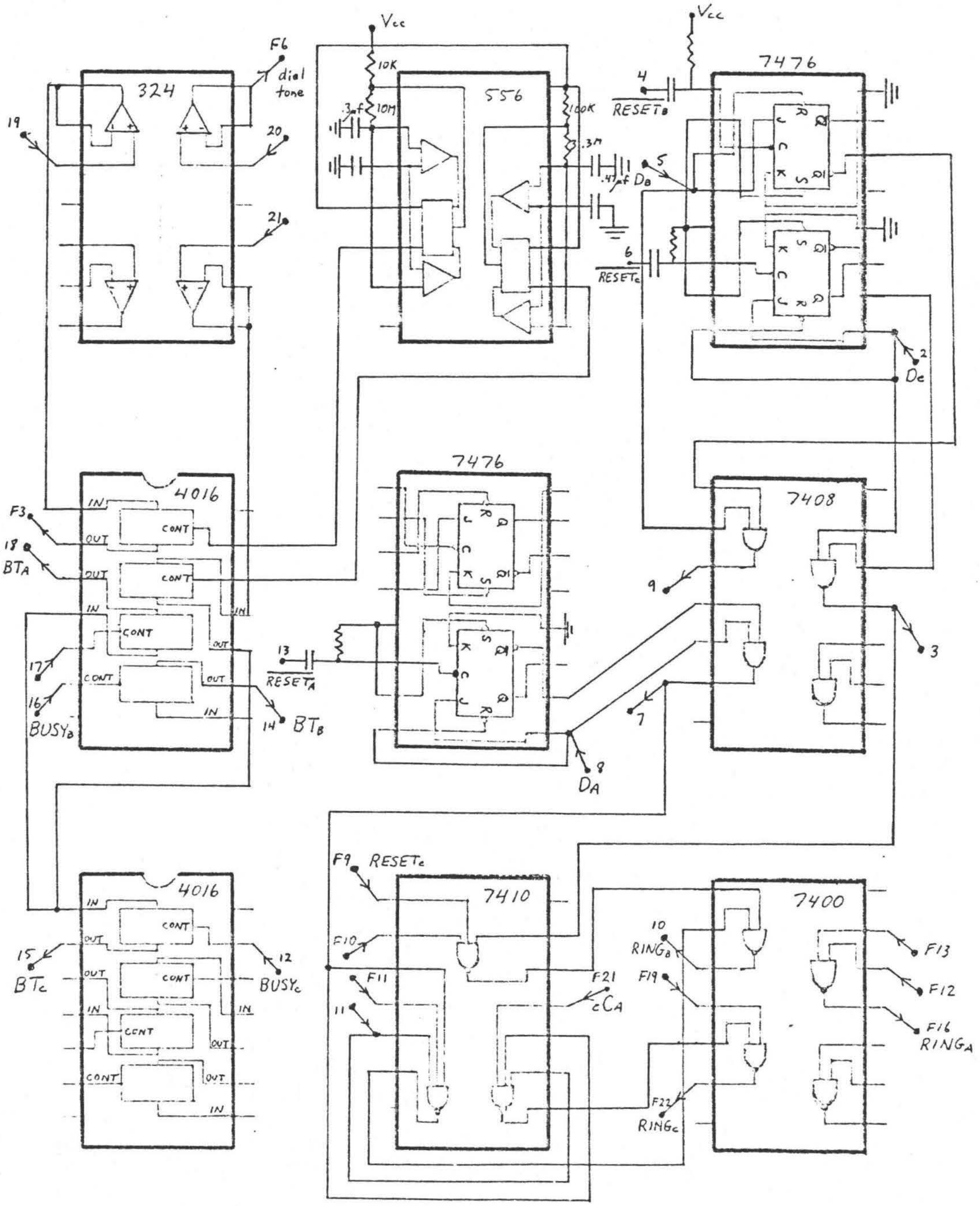
Board #2


7493
556
7432
7442
7402
7402
7402
7404
7402
Vcc
RESETc
MZc
BUSYA
BUSYB
SBC
DELAY
F11
COB
F12
F13
COC
F4
F10
F19
M8B
cSB
BSc
CA

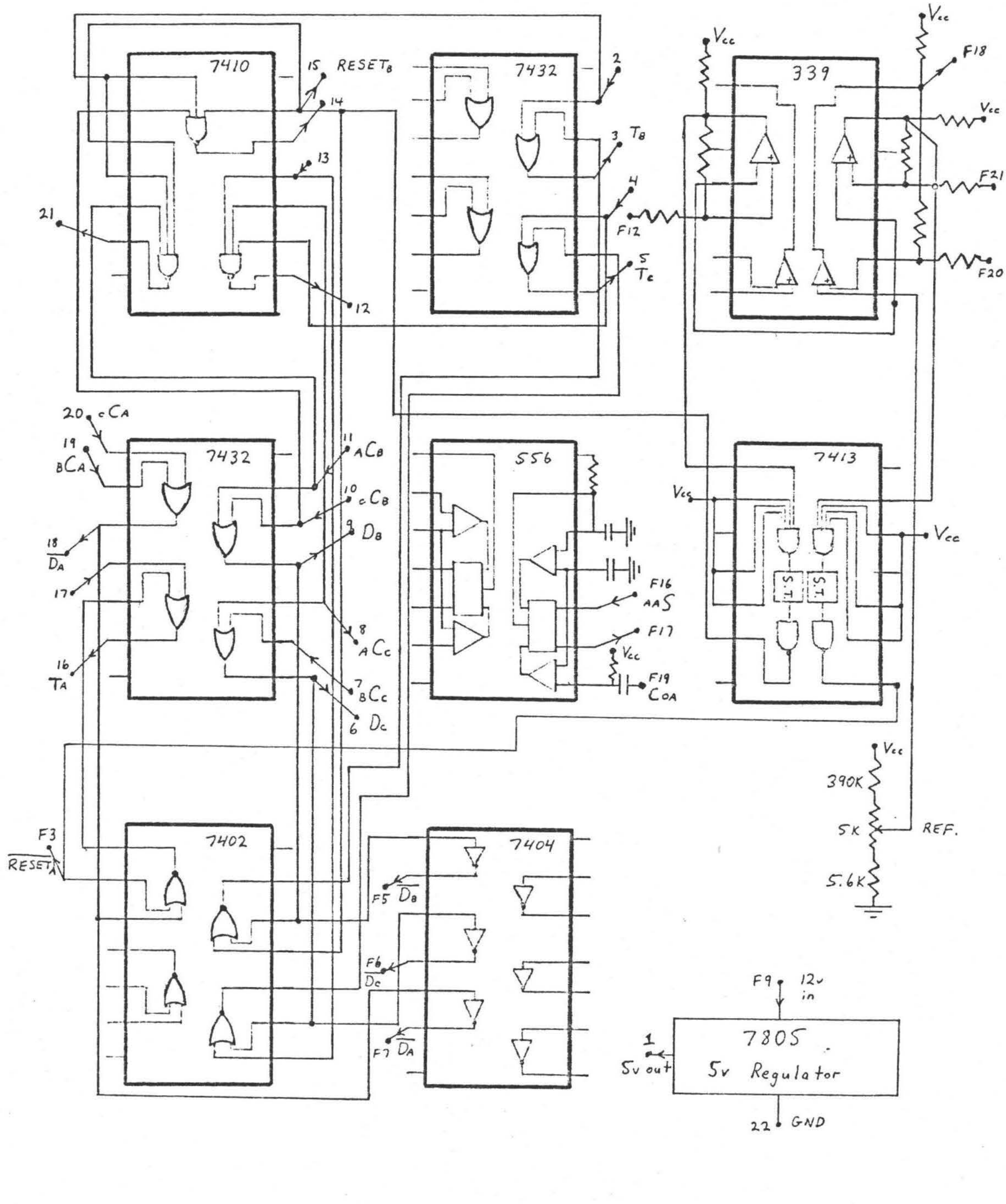
Board #3


7413
7410
4016
7402
7402
4016
7404
7432
4016
F4
F29
F21
F27
Vcc
F19
F5
14
F3
11
F9
F20
F6
F13
F14
10
19
18
F2
13
17
F11
F28
F30
21
20
16
15
9
12
5
4
3
F7
F12
F17
F16
F15
F10
F1
F18
8
2
7
6
F22

Board #4


324
F6
dial tone
19
20
21
556
Vcc
10K
10M
.3µf
100K
3.3M
.47µf
7476
RESET_B
RESET_C
D_B
D_C
4016
F3
18
BT_A
17
16
BUSY_B
14
BT_B
7476
13
RESET_A
D_A
8
7408
9
3
7
4016
15
BT_C
12
BUSY_C
7410
F9
RESET_C
F10
F11
11
F21
CC_A
7400
10
RING_B
F19
F22
RING_C
F13
F12
F16
RING_A

Board #5


7410
RESET
7432
339
Vcc
F18
F21
F20
F12
TB
TC
7432
ACB
CCB
DB
ACC
BCC
DC
CCA
BCA
DA
TA
556
F16
AAS
F17
F19
COA
7413
S.T.
390K
5K
5.6K
REF.
7402
RESET
7404
F5
F6
F7
7805
5v Regulator
5v out
12v in
GND

the appropriate edge connection numbers in the section immediately following the layout diagrams. The wired interconnections were made such that stations A, B and C have phone numbers 2, 5 and 8 respectively.

To relieve some of the congestion in these layout diagrams, the V_{cc} and GND connections for the individual chips are not shown. For the same reason, component values are not shown for those components that appear in "standard" circuits ($DELAY_X$ generation, dial pulse counting comparator, signal strength indicator and the turn-on latch). Component values for these circuits can be found by referring to the appropriate schematic diagram in the main body of this report.

Because of the simplicity of the sixth board on which the various tones are generated, no diagram is shown for this. The layout of the four circuits for generating the four frequencies closely follows that of Fig. 12a, b. This "tone board" holds two 324 chips, containing eight OP-AMPs. For each frequency the first two OP-AMPs in the sequence shown in Fig. 12a are situated on the tone board. The remainder of the circuitry is on the 4th board. These OP-AMPs are paired on the opposite side of the long axis of the 324 chips.

Omitted Edge Card Connections

Board #2

12	$DELAY_C$	13	${}_{C}BUSY_A$
15	$M2_B$	17	C's counter input
20	$BUSY_C$	F4	${}_{B}C_A$
F13	${}_{CC}S$		

Board #3

1	GND	2	$RING_C$
3	dial tone	4	M_B
5	M_C	6	$RING_B$
7	$RING_A$	8	M_A
9	$I^A{}_C$	10	$I^A{}_B$
11	$RESET_B$	12	$B^S{}_C$
13	AA^S	14	$RESET_A$
15	$C^S{}_B$	16	$\overline{C^C{}_A}$
17	D_B	18	$\overline{B^C{}_A}$
19	$\overline{A^C{}_C}$	20	$DELAY_A$
21	$\overline{A^C{}_B}$	22	V_{CC}
F1	$\overline{A^I{}_X}$	F2	$DELAY_B$
F3	$\overline{C^C{}_B}$	F4	C_{OC}
F5	$RESET_C$	F6	$\overline{B^C{}_C}$
F7	$\overline{RESET_A}$	F8	$C^I{}_X$
F9	$B^I{}_X$	F10	MZ_A
F11	MZ_B	F12	MZ_C
F13	$I^A{}_A$	F14	S_{BC}
F15	D_A	F16	$\overline{RESET_B}$
F17	D_C	F18	$\overline{RESET_C}$
F19	C's counter input	F20	$A^I{}_X$
F21	$DELAY_C$	F22	ringtone
F23	$B^S{}_A$	F24	$C^S{}_A$
F25	$A^S{}_B$	F26	$A^S{}_C$

F27	$\overline{D_C}$	F28	$_{CC}S$
F29	C's signal strength indicator output	F30	$_{BB}S$

Board #4

3	to 5th board (pin #4)	7	to 5th board (pin #17)
9	to 5th board (pin #2)	11	$RESET_A$
19	from tone board (ring)	20	from tone board (dial)
21	from tone board (busy)	F 3	ring tone
F10	$_BC_C$	F11	$_BC_A$
F12	to 5th board (pin #12)	F13	to 5th board (pin #21)
F19	to 5th board (pin #14)		

Board #5

2	to 4th board (pin #9)	4	to 4th board (pin #3)
12	to 4th board (pin #F12)	13	$RESET_C$
14	to 4th board (pin #F19)	17	to 4th board (pin #7)
21	to 4th board (pin #F13)	F12	signal strength level of B's receiver
F17	$DELAY_A$	F18	C's signal strength indicator output
F20	C's receiver F21 signal strength level	F21	A's receiver signal strength level

REFERENCES

1) Fairchild TTL Data Book.

2) National Semiconductor Linear Data Book.

3) E. Talley, "Basic Electronic Switching for Telephone Systems", Hayden, 1975.