

# 7-station intercom uses 1 master

**BUILD** *this all-transistor setup.  
Binary-type switching makes  
7-station, all-master system practical*

By **ROBERT D. HOCHBERG**

The two most disturbing problems in home intercommunication systems are that substations can't call other substations, and that most people just don't have the patience to wait for the amplifier tubes to warm up. The common and quite expensive solution to these problems has been to use all-master systems requiring multiconductor cables and ac sources at each station location, and to allow the five or more amplifiers to operate continuously. The intercom system described here solves these problems by combining the economy of the master-sub systems with the versatility of the all-master systems.

•It provides for calls between any

two of its seven stations. •It uses a single, transistor amplifier. •It partially solves the multi-cable problem with a binary type coded signaling system. •It provides for a station located outside the front door. •It needs no outside power, though the batteries may be replaced by a well-filtered power supply if desired. Even then, the unit will require ac power at only one location.

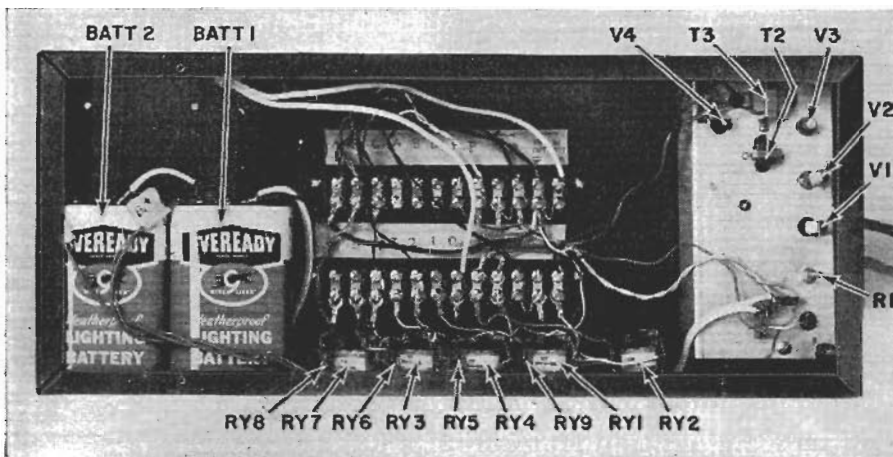
In purpose, this intercom is similar to "Intercom Super Duper Model I" (RADIO-ELECTRONICS, January 1961). In approach, however, the two are quite different. The Model I uses six vacuum tubes, three of which are in the relay exchange circuit. The proper connections are made through rather critical biasing of the relay control-tube grids.

My unit uses just four low-drain transistors—which draw current only when the unit is in operation—and an exchange circuit which has no tubes or transistors.

The heart of the system is a control cabinet containing the amplifier, exchange relays and batteries. It can be located in an attic, under a stairway or in just about any other accessible but out-of-sight location. One of the two batteries controls the exchange relays and the first two amplifier stages. The other battery controls only the output stage of the amplifier. With this arrangement, both batteries should last about the same length of time. Cables run from the control cabinet to each of the masters. Each master unit uses one speaker which doubles as a microphone, one two-pole seven-position selector switch, and two four-pole double-throw lever switches as function selectors.

## The binary exchange circuit

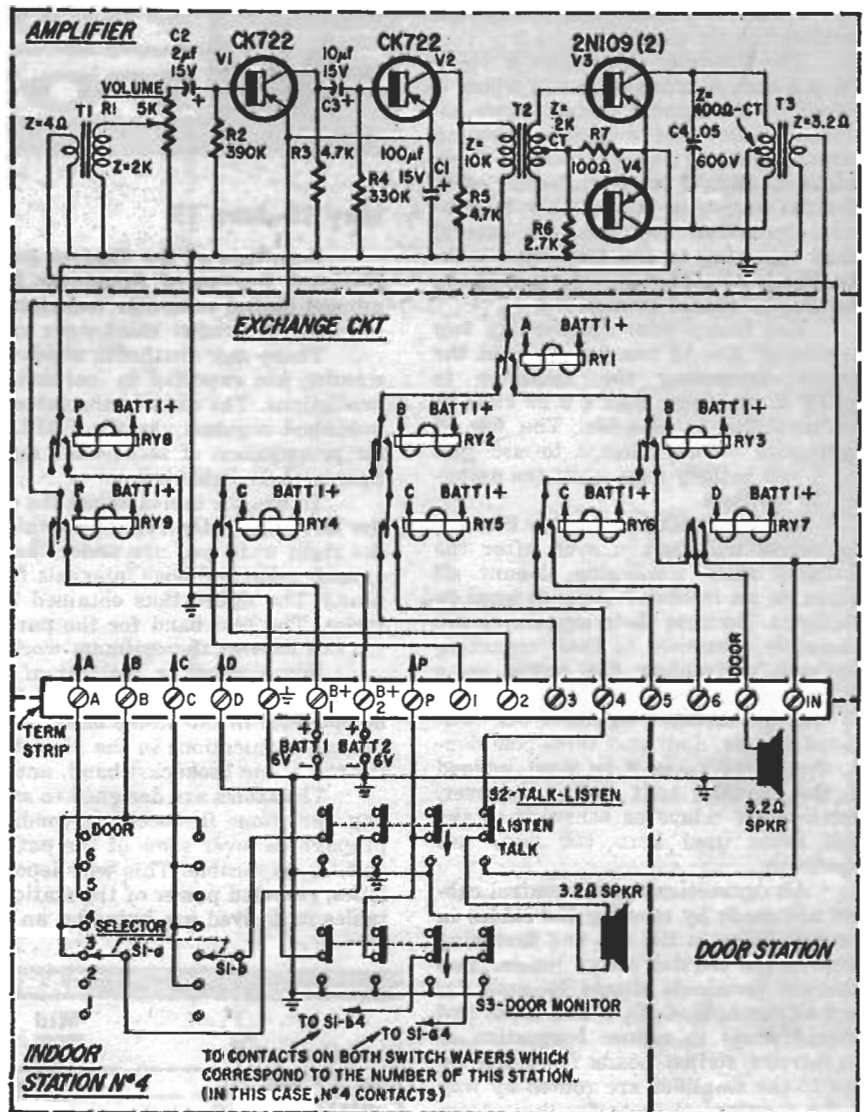
As shown in the schematic, the hot amplifier output lead is connected to the armature of RY1, an spdt relay. Depending on the position of the armature, the signal may be fed either to RY2 or RY3. (Since RY1 and RY2 are activated by the same signal, a single two-pole double-throw relay may be used instead.) From RY2 the signal may be sent to RY4 or RY5, and from RY3 it may be sent to RY6 or RY7. (Here a three-pole double throw relay may be substituted for RY4, RY5 and RY6). RY4, RY5, RY6 and RY7 channel the signal to masters 1,2,3,4,5,6 and DOOR. Thus, by energizing the proper



Switching relays and transistor amplifier are inside this case.

R1—pot. 5,000 ohms  
 R2—390,000 ohms  
 R3, R5—4,700 ohms  
 R4—330,000 ohms  
 R6—2,700 ohms  
 R7—100 ohms  
 All resistors 1/2-watt 10%  
 C1—100  $\mu$ f, 15 volts, electrolytic  
 C2—2  $\mu$ f, 15 volts, electrolytic  
 C3—10  $\mu$ f, 15 volts, electrolytic  
 C4—05  $\mu$ f, 600 volts, ceramic disc  
 BATT 1 BATT 2—6 volts (NEDA 915, specify type with binding posts)  
 RY1, RY7—spdt sensitive relay (Sigma IIF-1000-G/SIL or equivalent)  
 RY2, RY3—spdt sensitive relay (Sigma IIF-1000-G/SIL or equivalent) (These two units can be replaced with a single relay bearing dpdt contacts.)  
 RY4, RY5, RY6—spdt sensitive relay (Sigma IIF-1000-G/SIL or equivalent) (These three units can be replaced with a single relay bearing 3-pole double-throw contacts.)  
 RY7, RY8—spdt relay sensitive (Sigma IIF-1000-G/SIL or equivalent) (These two units can be replaced with a single relay bearing dpdt contacts.)  
 \*S1—rotary switch, 2 poles, 7 positions (Mallory 3229J or equivalent)  
 \*S2—lever switch, 4 poles, double throw (Centralab 1457 or equivalent)  
 \*S3—lever switch, 4 poles, double throw (Centralab 1457 or equivalent)  
 T1—output transformer; primary, 2,000 ohms; secondary, 4 ohms (Universal output transformer used in reverse)  
 T2—driver transformer; primary, 10,000 ohms; secondary, 2,000 ohms, ct (Lafayette TR-98 or equivalent)  
 T3—output transformer; primary, 400 ohms, ct; secondary, 3.2 ohms (Lafayette TR-109 or equivalent)  
 V1, V2—CK722  
 V3—2N109  
 \*Speaker, 3.2 ohms, 5-inch diameter  
 †Speaker, 3.2 ohms, 5-inch diameter (for door station)  
 \*Case, for remote station, 7 x 5/2 x 3 inches (Lowell P35X or equivalent)  
 †Panel, for remote station, 8 3/4 x 7 1/4 inches (Lowell C5XD or equivalent)  
 †Case, for door unit, 7 x 5/2 x 3 inches (Lowell P35X or equivalent)  
 †Panel, for door unit, 8 3/4 x 7 1/4 inches (Lowell C5RX or equivalent)  
 Chassis, for amplifier, 1 1/4 x 6 1/8 x 3 inches  
 Case, for amplifier and relays and batteries, 16 x 6 1/2 x 3 inches or larger  
 \*For each indoor remote you will need one of each of these components.  
 †These units are made by Lowell (Lowell Mfg. Co., 3030 Laclede Station Rd., St. Louis 17, Mo.) as well as other manufacturers. The units specified have a brushed-brass finish. A variety of other arrangements are available, in assorted colors. It might be wise to get the manufacturers catalog and select units to suit your decor.

Circuit of the intercom amplifier, switching circuits, a remote station and the door station.



exchange relays, the output signal may be fed to any desired station.

Now let's assume someone at Station 4 wants to call someone at Station 6. He sets his selector switch to 6 and depresses his talk button. Three things happen: First, his speaker is connected across the input of the amplifier and disconnected from its normal position on its output relay. Second, P is grounded, energizing the two relays which supply current to the amplifier. Finally, leads A and C are grounded through the selector switch, activating RY1, RY4, RY5 and RY6 in the control cabinet. These channel the output of the amplifier to the speaker of Station 6.

If the doorbell rings, the person answering switches the nearest station to 7 and depresses the talk button to be heard outside the front door. To hear a reply, he has merely to push his DOOR MONITOR button. This sends power to the amplifier, connects his speaker through the exchange relays to the amplifier output, and connects the door station through RY7 to the amplifier input. The DOOR MONITOR button eliminates the need for any controls on the front-door unit.

By now you should be familiar enough with the switching circuit to

see that only one two-way conversation can be handled at a time. However, this disadvantage is balanced by the facts that only eight-conductor cables are needed from the masters to the control cabinet, and that there is only one amplifier to power.

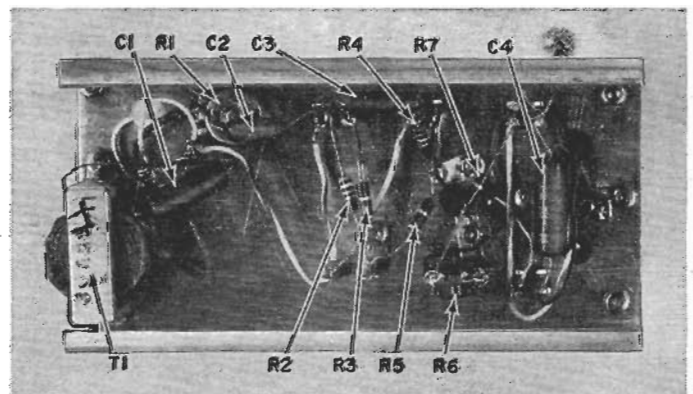
The greatest advantages of the binary exchange circuit are in a system with many stations. This 7-station system requires 3 wires to operate the exchange relays. A 64-station, all-master system would require only 6 wires to

operate the relays. There is, though, a practical limitation—a 64-station system would require 63 spdt relays.

### Transistor amplifier

The four-transistor amplifier uses CK722's for the first two stages and 2N109's in the push-pull output circuit. Its reproduction fidelity is equal to that of most three-tube intercom amplifiers. (If desired, the unit can be made into an interesting and good-sounding phono amplifier simply by removing the input

Closeup view of the underside of the amplifier chassis.



transformer and connecting a magnetic cartridge in its place.)

The amplifier is built on a 1 1/4 x 6 1/2 x 3-inch aluminum chassis which is placed in the control cabinet after assembly. Follow the schematic, keeping output leads as far as possible from the input. Mount the transformer cores at right angles to one another to minimize inductive feedback. Be careful when soldering to the transistor sockets. Beads of solder may form between the closely spaced prongs.

Use heavy wire (preferably two lengths of No. 18 run parallel) in the circuit connecting the amplifier to BATT 2, and keep BATT 2 as close to the amplifier as possible. The use of light wire or an attempt to use just one 6-volt battery may cause the amplifier to oscillate.

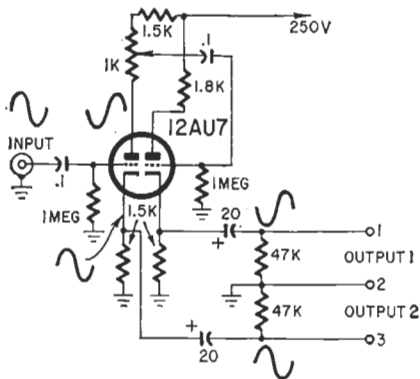
Bend the relay springs a little so the relays will pull in even after the batteries start weakening. Mount all relays on an insulator such as wood or Plexiglas. Because their armatures are internally connected to their mounting brackets, assembling the relays on a metal chassis would, in effect, connect all the armatures together. As was stated earlier, dpdt and three-pole double-throw relays may be used instead of the parallel spdt units. However, multicontact relays as sensitive as the spdt types used here (50 mw) are expensive.

All connections to the control cabinet are made by running the cables in through holes in the top and fastening them to the barrier strips inside. Two adjacent terminals should be used for each of the A, B, C, D, P and input and ground wires to reduce congestion at the barrier strips. Leads from the relays to the amplifier are routed by way of the terminal strip to facilitate later removal of relay bank or amplifier. END

### DUAL CATHODE FOLLOWER

Sometimes we need a way of reversing the polarity of a low-impedance signal or supplying two equal signals 180° out of phase from low-impedance sources. Here is a circuit from *Revista de Informacion Electronica* (Madrid, Spain) that does the trick.

Phase relationships are shown on



the diagram. The input triode is a phase splitter and cathode follower. The 1,000-ohm potentiometer is used to balance the outputs. Balance can be checked with an audio vtvm or scope.

# SW PROPAGATION FORECAST

May 15-June 15

By STANLEY LEINWOLL\*

Scientists of the Central Radio Propagation Laboratory (CRPL) of the National Bureau of Standards have after 5 years of research, successfully applied digital computer techniques in preparing the world contour maps that are used to predict short-wave radio propagation conditions.

These new methods, which will be available for general use in several months, are expected to increase the accuracy of high-frequency propagation predictions. The data in the tables below, derived from the basic contour maps published regularly by the CRPL, show the optimum frequency in megacycles for propagation of short-wave signals between the locations shown during the time periods indicated.

To use the tables, select the one most suitable for your location, read down the left side to the region in which you are interested, then follow the line to the right until you are under the appropriate time. (Time is given at the top of each table in 2-hour intervals from midnight to 10 pm, in your local standard time.) The figure thus obtained is the optimum working frequency, in megacycles. The best band for the particular service in which you are interested is the one nearest the optimum working frequency.

For example, a resident of San Francisco would use the Western USA table. At noon, Pacific Standard Time, signals to and from the Far East would be optimum in the 13-mc band. A radio amateur would be most likely to establish communications in the 20-meter (14-mc) band, while a listener would first try the 11-mc broadcast band, and follow this with the 15-mc band.

The tables are designed to serve primarily as a general guide, since day to day variations in receiving conditions can be considerable. At certain hours, propagation over some of the paths given in the tables may be extremely difficult, or impossible. This will depend on the type of service, antenna characteristics, radiated power of the station, etc. The curves from which the data in the tables is derived are based on an effective radiated power of 10 kw.

### EASTERN US to:

	Mid	2	4	6	8	10	Noon	2	4	6	8	10
West Europe	8	8	8	10	11	12	12	13	13	13	12	10
East Europe	8	8	8	9	10	11	12	12	12	11	9	8
Central America	12	10	11	13	15	15	16	17	16	15	14	13
South America	12	9	12	15	17	17	17	17	17	16	14	13
Near East	8	7	8	10	11	12	12	13	13	12	11	10
North Africa	8	7	8	11	12	13	13	14	14	14	12	10
South & Central Africa	7	9	10	13	15	16	16	16	15	14	8	8
Far East	9	8	8	8	10	11	11	11	12	12	11	10
Australia & New Zealand	12	11	11	10	10	10	9	15	18	18	18	15

### WESTERN US to:

West Europe	8	7	9	11	12	12	12	13	13	11	10	8
East Europe	8	8	8	11	11	12	12	12	11	10	10	10
Central America	13	11	10	13	15	15	16	17	17	16	15	13
South America	9	10	15	16	17	17	17	17	15	14	13	13
Near East	8	9	10	11	12	13	13	13	11	11	9	9
North Africa	7	9	11	12	12	13	13	13	12	11	10	8
South & Central Africa	8	10	12	13	13	14	14	15	15	13	10	8
Far East	9	7	9	12	11	12	13	13	13	14	13	11
Australia & New Zealand	12	12	10	10	10	16	18	19	18	17	13	13

### CENTRAL US to:

West Europe	8	8	8	10	11	11	12	12	11	10	9	9
East Europe	7	7	7	8	10	10	10	10	10	9	9	9
Central America	13	11	12	15	17	18	19	19	19	18	15	13
South America	10	8	10	15	16	17	17	17	16	16	14	12
North Africa	7	7	8	10	11	12	12	12	12	11	10	9
South & Central Africa	9	8	10	13	13	14	14	13	11	8	8	9
Far East	13	13	9	9	12	12	13	14	14	14	14	14
South Asia	12	9	8	9	12	12	12	12	12	13	13	13
Australia & New Zealand	13	13	11	10	10	14	19	20	20	20	20	16

\*Radio-frequency and propagation manager, RADIO FREE EUROPE.



# LOUDSPEAKER INTERCOM SYSTEMS

By RAY A. SHIVER

Principles of operation and features provided by this widely used method of voice communication for the home and industry.

**L**LOUDSPEAKER intercom systems are used extensively in home and industry for rapid, effective voice communication. They vary in size and complexity from the simple two-station system with one master and one remote unit up to the large multiple master and remote systems such as would be found in a complex industrial installation.

The principle of operation is the same for all types of systems, whether they be simple or complex. Basically, the voice is used to actuate a microphone or a PM speaker, the output of which is amplified by an audio amplifier which, in turn, operates a loudspeaker at some remote point. In order to be an intercommunicating system, some means must be provided for a return signal to permit a two-way exchange of information. This is generally accomplished by switching the amplifier input and output to provide both an incoming and an outgoing signal. In this manner one amplifier can be used and each speaker can be employed alternately as a microphone for the outgoing signal and as a loudspeaker for the incoming signal. Most systems use this switching principle (a few special cases will be covered in a later section) in some form and it can be very simple or complex, depending on the special features desired.

## Simple Two-Station Unit

Fig. 1 illustrates the operation of a simple two-station sys-

tem consisting of one master and one remote (or "slave") station. The audio amplifier can actually be any type of amplifier capable of driving the loudspeakers to the desired listening level. Amplifiers employing vacuum tubes and transistors are commonly used.

The circuitry is generally designed to limit the operation range to the voice frequencies (about 300 to 5000 cps) which permits the use of compact components. Often a.c.-d.c. circuitry is employed in the lower powered units which permits a further reduction in size by eliminating a power transformer. For ordinary home or office use a fraction of a watt of output power will usually suffice. An industrial plant with a high noise level may require a unit with several watts of output power and special loudspeakers to insure effective intercommunication.

Referring to the circuit of Fig. 1A,  $S_1$  is the talk-listen switch and is shown in the "listen" position which permits the local loudspeaker  $LS_1$  to receive the incoming signal from the remote speaker  $LS_2$ . The local speaker is connected to the amplifier output by means of switch contacts 1A and 2A. The remote speaker, which in this case is being used as a microphone, is connected to the amplifier input by means of switch contacts 1B and 2B.

In Fig. 1B, the situation is just the reverse. Switch contacts 1A and 3A connect the amplifier output to the remote



speaker for the outgoing signal and contacts 1B and 3B connect the local speaker to the amplifier input thereby permitting it to be used as the microphone. The talk-listen switch is normally spring-loaded in the "talk" position so that it automatically returns to the "listen" position when finger pressure is removed from the knob.

The impedance of loudspeakers used for intercom systems is generally 45 ohms. This permits longer wire runs between stations with a minimum signal loss due to wire resistance, as would be the case with a 4- or 8-ohm speaker. It also permits several remote stations to be paralleled without dropping the total impedance to a value which would create a serious mismatch to the amplifier output.

However, when the speaker is being used as a microphone, some means must be devised to step up the impedance to a value suitable for the high input impedance of vacuum-tube amplifiers. This is accomplished by adding a microphone-to-grid transformer at the input of the amplifier. It should be well shielded with a high permeability material to prevent magnetic coupling with other components on the chassis, especially power transformers where they are used.

### Systems with Privacy Features

The system described thus far is "non-private," that is, the person operating the master unit can listen in to the remote station at will. Obviously, this is not always desirable, especially in schools and commercial applications. In order to prevent this, some modification must be made to our simple system.

Fig. 2 shows how this can be accomplished. Note that the remote station now requires three wires instead of two. The dotted lines between switch contacts 2B and the remote speaker line show the modification necessary for private operation. With this connection removed there is no longer an electrical path between the remote station and the amplifier input when the talk-listen switch is in the "listen" position. Note, however, that when the switch is in the "talk" position, the unit will function normally. All that remains to complete the private system is to provide a means for the remote station to answer a call. This is done with a privacy switch at the remote unit and a third wire which bypasses the talk-listen switch at the master unit and connects directly to the input

of the amplifier that is utilized in the intercom system. As can be seen from the diagram, the privacy switch disconnects the remote speaker from the normal station line and connects it to the privacy line when returning a call. This switch is generally spring-loaded like the talk-listen switch in the master unit. In this manner privacy has been gained for the remote station, but not without cost.

The disadvantage of a private system is that the operation at the remote unit is no longer "hands free" but requires manual operation of a switch in order to return a call from the master unit. This could be a serious disadvantage if the remote speaker were located, for example, at a loading dock which would require a worker to leave his job and perhaps

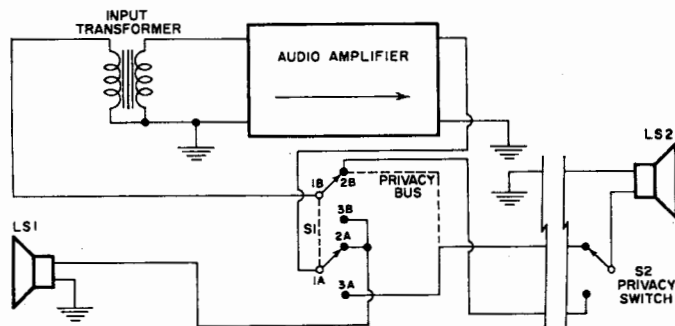


Fig. 2. For privacy feature a three-wire interconnection is used.

walk a considerable distance in order to answer a call. For this reason, this type of operation is usually confined to desk-top installations.

### Balanced & Unbalanced Line Systems

It may have been noticed in the systems we have described thus far that one side of the input and output transformers is returned to ground, providing a common path between the two. This is an unbalanced system and is quite extensively used in master units for systems where only one master station and one or more remotes are used. This type of circuitry simplifies construction since only half the switching needed for balanced-line operation is required. However, this system does not exhibit the noise-concealing properties of a balanced-line system and for this reason shielded lines are generally required if extraneous noise pickup and hum are to be kept to a minimum. This is especially important with units utilizing the privacy feature since the privacy line is unterminated and makes an effective antenna if not well shielded. Multiple master operation is not recommended for this type of system because of its inherent high level of crosstalk.

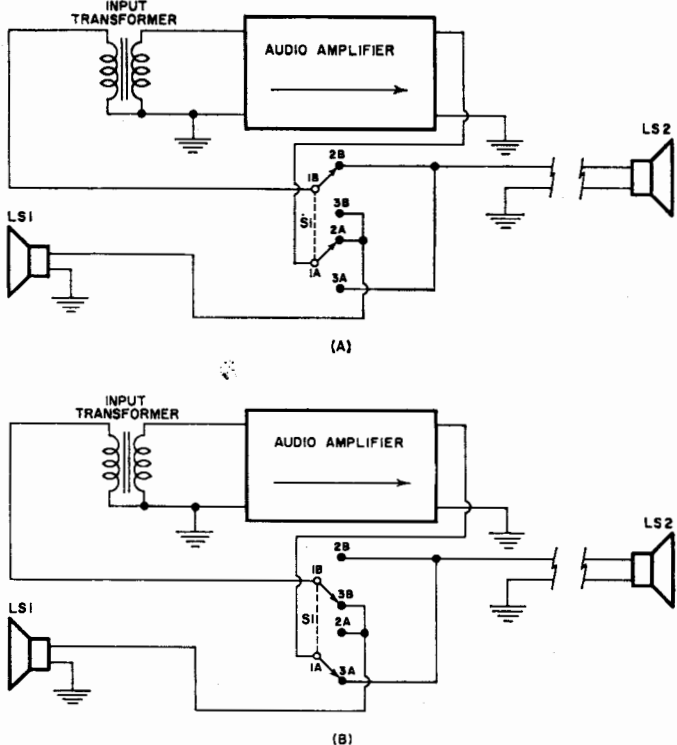
Balanced-line construction is generally employed in systems where two or more master units are used and they are required to communicate with each other. Wiring for such systems usually consists of multiple conductor twisted-pair cable that requires no shielding. Since there is no common connection between input and output circuits of the amplifiers in such a system, crosstalk is greatly reduced or eliminated entirely.

### Call-In Systems

The addition of the privacy feature to the system of Fig. 2 really accomplishes a two-fold purpose. In addition to maintaining privacy, switch S<sub>2</sub> in the remote unit may also be used to initiate a call to the master station. This is not possible in the system of Fig. 1 unless the remote station were monitored continuously which, ordinarily, is not desirable.

This type of call-in system is known as "voice call-in" and is quite commonly used in unbalanced-line systems. It should be pointed out that if the non-private connection is left in the circuit between switch contacts 2B and the remote speaker line, the call-in feature will still be retained if the unit is equipped with a standby switch.

Fig. 1. Simple 2-station system during (A) "listen" and (B) "talk."



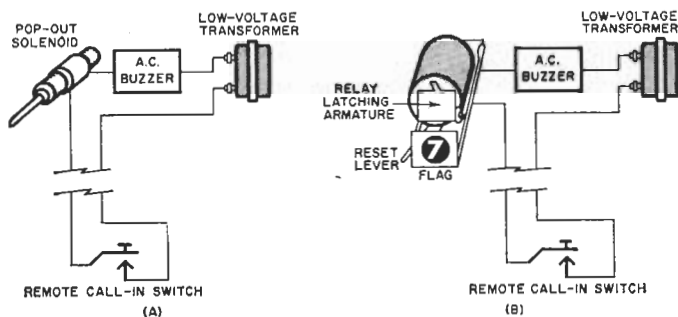


Fig. 3. (A) Pop-out solenoid and (B) relay drop-flag annunciators.

One disadvantage of voice call-in is that the calling party must identify his station in systems consisting of several remotes in order for the called party to select the proper station for returning the call. If the called party does not happen to be present when a call is initiated, he has no way of knowing, upon his return, that he has received a call nor the identity of the calling party. This problem can be eliminated by the use of an annunciator call-in system. Two types are generally employed: the pop-out or the drop-flag. The pop-out system uses a small solenoid for each station. This pops out when energized thus identifying the calling party. In order to re-set the unit, a plunger must be pushed in manually. In the drop-flag type a relay trips a small identifying flag. To re-set this unit, the flag is simply pushed up until it latches in the original position. Lamps are sometimes used as call-in indicators but they are less reliable due to bulb failure. Fig. 3 is a diagram of each type of annunciator call-in. An audible buzzer is generally used with each type of system to attract the attention of the called party.

### Selective Systems

In Fig. 2 we have advanced our simple system to include privacy and remote-call origination. We may now wish to add several more remote stations to expand our communications network. This can be accomplished by the addition of a station-selector switch, S, in Fig. 4. The addition of this switch will allow us to select any one of several stations as desired.

For a balanced-line system, the switch would have to have two poles and twice the number of contacts since both sides of the line would be switched. As shown in Fig. 4, in the unbalanced system the privacy and call-in line can be extended to each additional remote station by simply paralleling the lines at the master station. Ordinarily the remote stations all have to be set up as either private or non-private since the manner in which the switch is connected at the master station will affect all remote stations equally.

Station-selector switches can be of the rotary wafer type, push-button, or slide-switch types. The first two are preferable since they have self-wiping contacts and give long, trouble-free service. Push-button switches have the advantage of providing more convenient station selection since in the rotary type it is often necessary to step through several switch positions in order to arrive at the desired station.

### Intermixed System: Masters & Remotes

In the system of Fig. 4 it is evident that remote stations A, B, and C can establish communication with the master station but not with each other. There are many cases where such a system would be entirely adequate. In a typical example, let us suppose it has been established that station A has need for contacting stations B and C. We must now replace station A with a master unit if this is to be accomplished.

Now both master stations can converse with remote stations B and C but additional switching must be provided if the master units want to communicate with each other. This involves the addition of another position on the talk-listen switch and, basically, converts the master to a remote station when not in use. This is shown in the schematic diagram of

Fig. 5 for a balanced-line system. Note the increased complexity of switching involved to provide this facility in a balanced-line system. The additional switch position, usually designated "off" or "standby," disconnects the speaker from the amplifier in the master station and connects it to the pair of terminals marked "X" in the diagram. This pair of terminals is called the "home line" for the unit and offers a direct connection to the speaker in the master unit when the talk-listen switch is in the "off" position. Thus the master stations can communicate with each other as well as with the remote stations. However, it can be seen that in a system of this type the privacy and voice call-in feature cannot be extended to the remote stations since the speakers in the master stations are disconnected from their respective amplifiers when the units are in the "standby" position. In fact, the amplifiers are usually turned off by an extra set of contacts on the talk-listen switch which breaks the "B+" lead when the switch is in the standby position. This will provide minimum power consumption and heating when the unit is not in actual use.

A system of this type is generally used where the need is for a limited number of master stations and the bulk of the

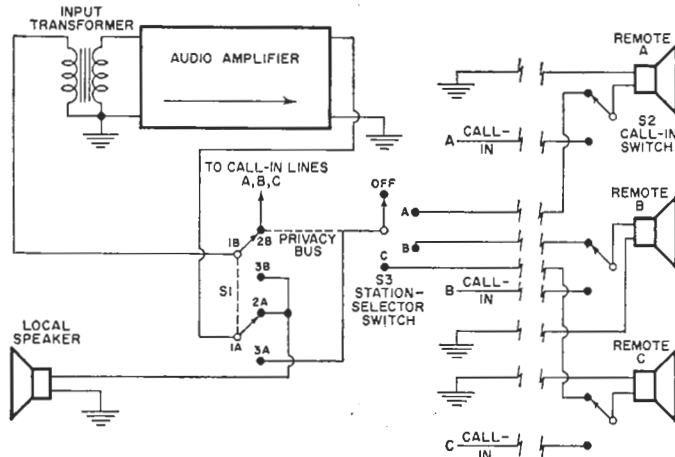


Fig. 4. The addition of a station-selector switching arrangement.

system consists of remote units. If a call-in system is employed, one of the annunciator types is generally used.

### Master-to-Master System

The all-master type of system is designed for use where each station in the system must be able to contact every other station. The basic switching system is shown in Fig. 6. Note that the home-line switching is similar to Fig. 5 except that only a two-position switch is used. The normal "listen" position, utilizing the amplifier, is not needed in an all-master system. Instead, an "off" position is used as "listen."

Referring to Fig. 6, when the unit is in the "talk" position, operation is quite straightforward. When the master switch is in the "listen" position, the amplifier is disconnected and the home-line terminals are connected to the local speaker. Note that the level control is only in the circuit when the master is in the "listen" position. This permits one setting of the control for desired loudness at a master station to serve for all incoming calls if the gain of all the master amplifiers in the system is approximately equal, which is generally the case.

One important feature of the all-master system is the 100 per-cent trunkage feature. This is an old telephone term meaning there can be half the number of simultaneous conversations going as there are units in the system. Two units, of course, are required to make one conversation path. This feature can be very worthwhile in a system subject to heavy usage.

In a system of this type it is not possible for a station to be monitored by another station as long as the talk-listen switch

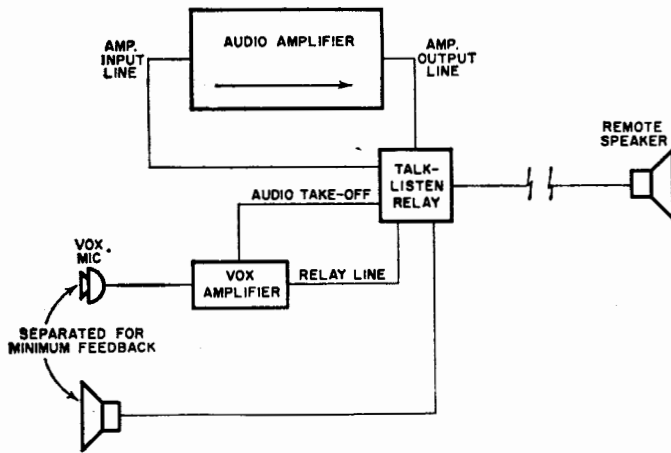


Fig. 7. The use of a "vox" (voice-operated switching) circuit.

loudspeaker. A loud ringing bell or horn is used to signal the called party.

Operation in an explosive environment poses another requirement for an intercom system. Components for such a system must be enclosed in special cases or containers that are air-tight. Systems of this type may be found in chemical plants, in the presence of explosives or explosive gases, and in hospital operating rooms.

Another special feature often desired in industrial systems is high-level voice paging. One line can be selected at one or all master stations to drive a booster amplifier which provides the power for the paging speakers. This is diagrammed in Fig. 9. The pad is usually necessary to drop the level from the intercom amplifier to a value that will not overdrive the booster amplifier. Such a circuit, of course, is one-way only since it is not possible to receive through the booster amplifier used.

Other special features often in demand are foot-operated talk-listen switches for "hands-free" operation, handsets instead of loudspeakers for additional privacy, combination intercom and background music systems which provide for dual usage of the intercom speakers, and all-call facilities which permit all stations to be monitored or called simultaneously. There is literally no end to the combination of special features and circuitry that can be provided to meet the requirements of just about any loudspeaker intercommunications system.

### Systems for Home Use, Wireless Systems

Intercom systems designed for home use have several unique features not generally found in commercial units. Several master stations may be required in a home system but, due to the limited service involved, usually one amplifier is provided for the complete system. This permits only one master unit to be in service at a time or, in other words, it is a "one trunk system." Applications requiring more than one trunk would dictate the use of a commercial-type system.

Since the off-duty time of a home system is much greater than the in-service time, the amplifier can be disabled when not in use. For this reason filament-type vacuum tubes or transistors are generally used in the circuitry. This permits instant service without waiting for a warm-up period and conserves power and component life when the system is not in use.

Master stations in most systems of this type can be set for either private or non-private operation. Remote stations, as a rule, are non-private although in some systems they may be wired for voice call-in. Where voice call-in is not used, a push-button is usually available for the remote unit to permit signaling the master by a buzzer or bell.

For ordinary home use a fraction of a watt of audio power is usually adequate for a good listening level hence amplifiers can be of very compact construction and, in most cases, will

be flush mounted in an ordinary stud wall constructed of 2 by 4's.

In some cases where existing construction makes a wired system impractical, a wireless intercom system may be considered. Only the existing electrical wiring is required for a system of this type. Low-frequency r.f. energy is used as the communications medium and this is coupled to the electrical wiring.

Basically a unit of this type is a low-powered r.f. transmitter and receiver operating in the range from about 75 to 350 kc. Multi-purpose tubes are generally employed much as they are in transceiver circuitry. Utilizing this type of construction, along with the small amount of r.f. energy required (usually only a few milliwatts), the unit can be quite compact—often no larger than a comparable wired unit.

Multi-channel units are available with up to twelve individual communications channels. To minimize crosstalk between adjacent channels they are usually spaced at least 25-kc. apart.

Operating characteristics for this system can be compared to the all-master wired system. Voice call-in, 100 per-cent

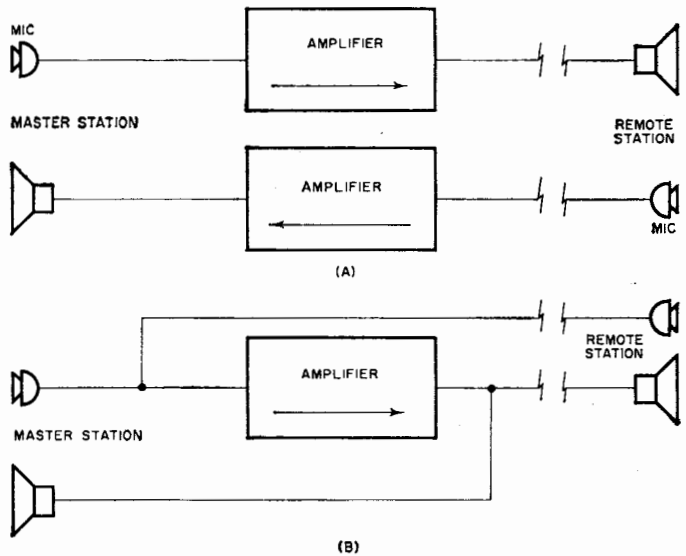


Fig. 8. Simultaneous system overcomes drawbacks of "vox" system.

trunkage, and privacy are all included. Difficulties are sometimes encountered with r.f. carrier intercom systems when they are operated on power lines that are common with large motors or transformers. The large shunt capacity often associated with these devices tends to bypass the r.f. signal to ground and, in some cases, will not permit reliable operation. In addition, they do not work too well on polyphase power distribution systems since the signal is shunted in the same manner when attempting to cross from one power leg to another. For these reasons, wireless systems cannot be used interchangeably with wired systems but are capable of reliable service when used under the proper conditions.

Although all possible combinations and types of systems cannot be covered in a single article, the author has tried to suggest the basic types and the various features available. Intercoms of the types described include those most commonly used and most reliable in their operation.

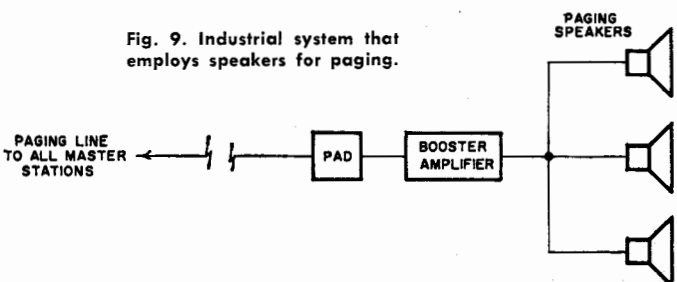
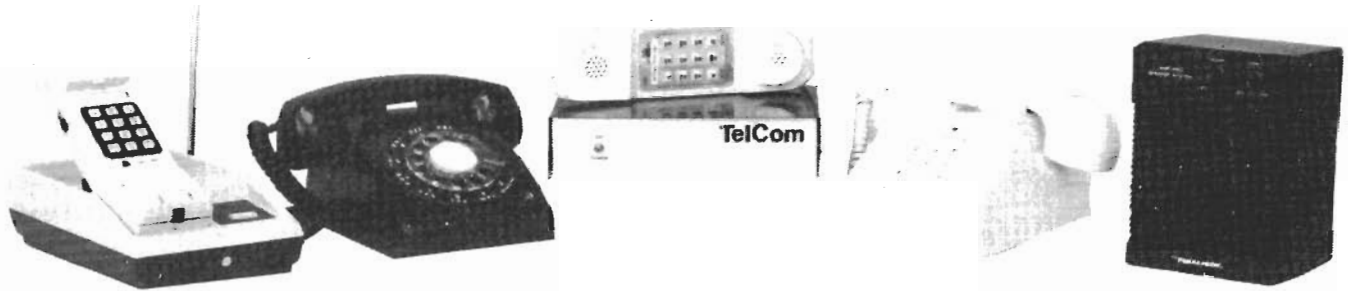


Fig. 9. Industrial system that employs speakers for paging.

# BUILD THIS

*A home or small-business intercom needn't be expensive. Our economical system uses standard pulse-dial telephones, and has options for a PA system—with music!*



## NINE-STATION INTERCOM

DWIGHT MORRISON

INSTALLING A BUSINESS-TELEPHONE SYSTEM can be an expensive proposition. But if you have several unused pulse-dial telephones lying around, you can use them and cut costs dramatically by building our simple control center (for about \$50). The control center provides a switchboard-like function allowing any one of nine phones to call any other. In addition, one station can be set up as the office PA system; to make an announcement from any station, just dial the station assigned to the PA. The PA system can provide background music (from any source) while no announcement is being made.

### System features

Nearly any telephone with a pulse-dial output may be connected to the control center. For example, a standard rotary-dial telephone, a pushbutton phone, a speakerphone, or even a cordless phone could be used. (See this story's lead photo.) Each telephone may be located as far as 2000 feet from the control center; interconnections are made with standard four-conductor 22-gauge telephone wire.

Standard telephone-ring generators require a special 90-volt, 20-Hz power supply. Building such a circuit is expensive as well as difficult because components are hard to obtain. Therefore, the control center generates its own special ring and dial tones.

The control center provides a LIGHT output to illuminate a "busy" LED installed in each station. All LED's light up whenever any station is off hook.

### How to use it

Operating the intercom is simple. Pick up the phone at any station; you'll hear a

dial tone in the earpiece, and all LED's will light. Now you can dial any station; while dialing occurs, all of the LED's will flash. After dialing, you'll be able to hear the ringback tone in the earpiece. In addition, the station you dialed will buzz for about one second. If no one answers at that station, you can call again without hanging up.

If someone at a different station picks up his phone, he can join in the conversation. That "party-line" effect can be used for simulating a conference call. For example, if two people are talking and decide that they'd like to include a third, they should hang up, and one should call the third. After giving him time to answer, the one who hung up can pick up his phone and join the conversation.

### Circuit operation

The schematic diagram of the control center is shown in Fig. 1. The heart of the circuit is an M-959 IC, manufactured by Teltone (P. O. Box 657, 10801-120th Avenue N. E., Kirkland, WA 98033-0657). The M-959 is a CMOS device that counts dial pulses and provides an encoded binary representation of those pulses. For example, if the digit nine were dialed, the D<sub>0</sub>-D<sub>3</sub> outputs would contain logic levels 10 01. The M-959 also provides a logic-level indication of hook status (OH) at pin 13.

When any station goes off hook, current flows through the coils of relays RY10 and RY11, so their contacts close. Relay RY10 supplies power to the LED's in each station, and relay RY11 grounds the LC (Loop Current) input of IC1. That forces the OH output to go high.

The SR flip-flop composed of IC4-c and IC4-d does not change state, but the

astable oscillator composed of IC5-a and IC5-b turns on because both OH and pin 10 of IC4-c are high. That astable is what provides the dial tone, which is fed to the talk circuit via R12 and C12.

When a digit is dialed, relays RY10 and RY11 "follow" the dial pulses—i. e., their contacts make and break a number of times according to the digit that was dialed. After dialing stops, IC1 places the encoded binary digit on pins 8-11. In addition, IC1's STB output (pin 12) goes high for 200 ms.

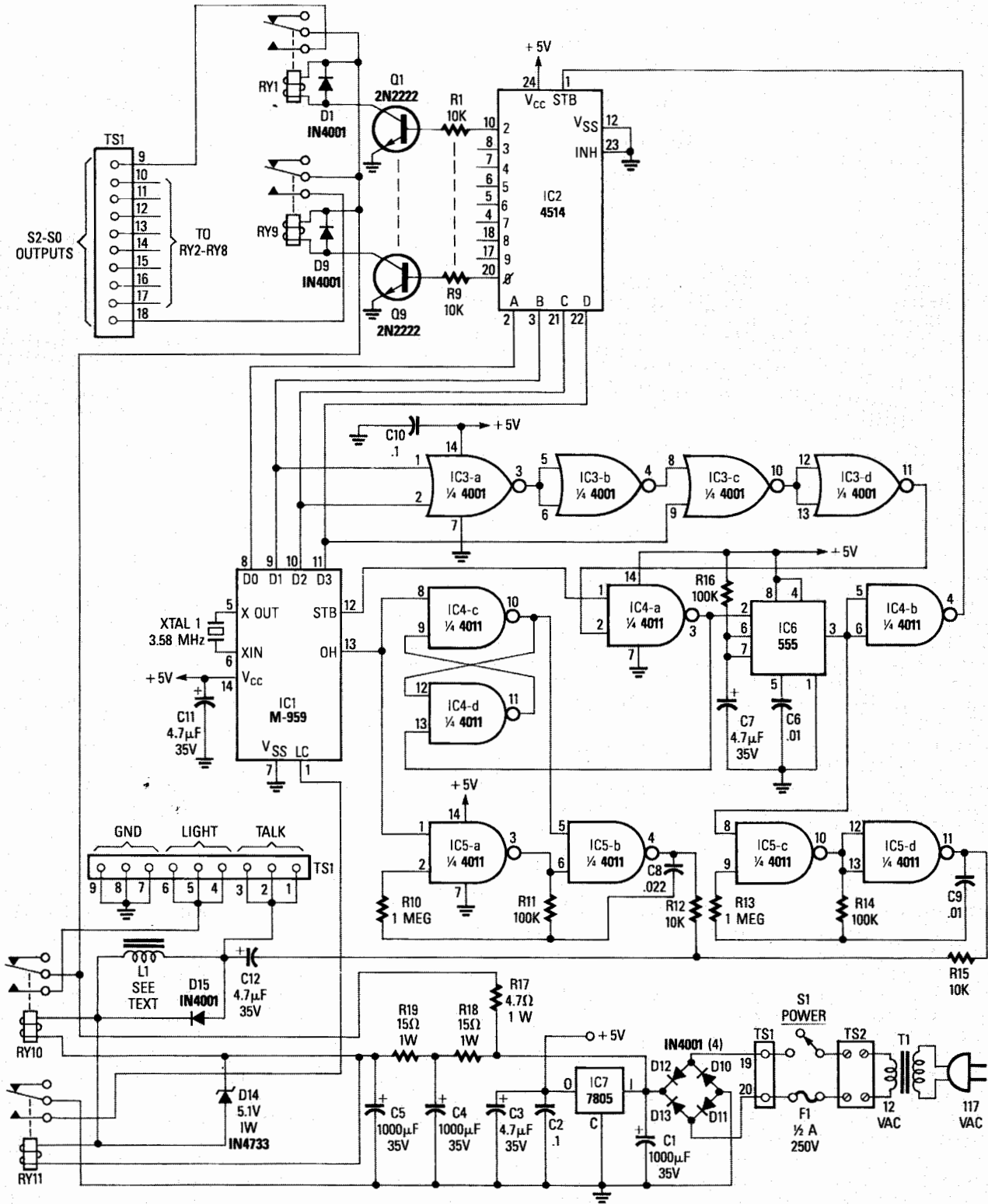
There is no station one in this system. The reason is that the circuit cannot distinguish well between going off hook and dialing the digit one. The four gates of IC3 are used to ensure that only stations greater than one are called. Those gates are set up as a three-input OR gate. The inputs of the gate are connected to the D<sub>1</sub>-D<sub>3</sub> outputs of IC1, so any station greater than one will cause the output of IC3-d to go high. That signal is NAND-ed with the STB output of IC1 by IC4-a, and the combined signal is used to trigger IC6, a 555 timer operated in the one-shot mode.

With the component values shown in Fig. 1, the output of the 555 will remain high for about one second. The output of IC4-a also resets the IC4-c/IC4-d flip-flop, thereby disabling the ring generator (IC5-a and IC5-b).

The 555's output (pin 3) enables the astable composed of IC5-c and IC5-d, which oscillates at a frequency of about 1000 Hz. That astable supplies the dial tone, which is fed to the talk circuit via R15 and C12.

In addition, the 555's output is inverted by IC4-b and that signal is used to strobe the binary outputs of IC1 into IC2, a 4-





**FIG. 1—A-HALF-DOZEN IC'S** and a dozen relays comprise most of the circuitry of the intercom. Note that only two of the relay-output circuits are shown (R1-Q1-D1-RY1 and R9-Q9-D9-RY9); the other seven outputs from IC2 are wired in a similar manner.

to-16 line decoder that works as follows. Assume that station two was dialed. Then pin 10 of IC2 will go high and turn on transistor Q1, which will in turn enable relay RY1. At that point, 16 volts will be present at pin 9, the S2 output, of TS1, the 20-terminal strip. That voltage would then

drive the buzzer in station two.

When the conversation ends, both parties hang up. Then relays RY10 and RY11 will de-energize. That will cause all the LED's in the circuit to extinguish, and it will allow IC1's LC input to float high.

Diodes D10-D13 rectify the 12-volt AC

input; IC7, a 7805 regulator provides +5-volts for the logic IC's. Resistors R17, R18, and R19 provide +16-volts to operate the relays and the buzzers. Coil L1, which is actually the primary of a 500-ohm audio transformer, filters power-supply hum from the talk circuit.

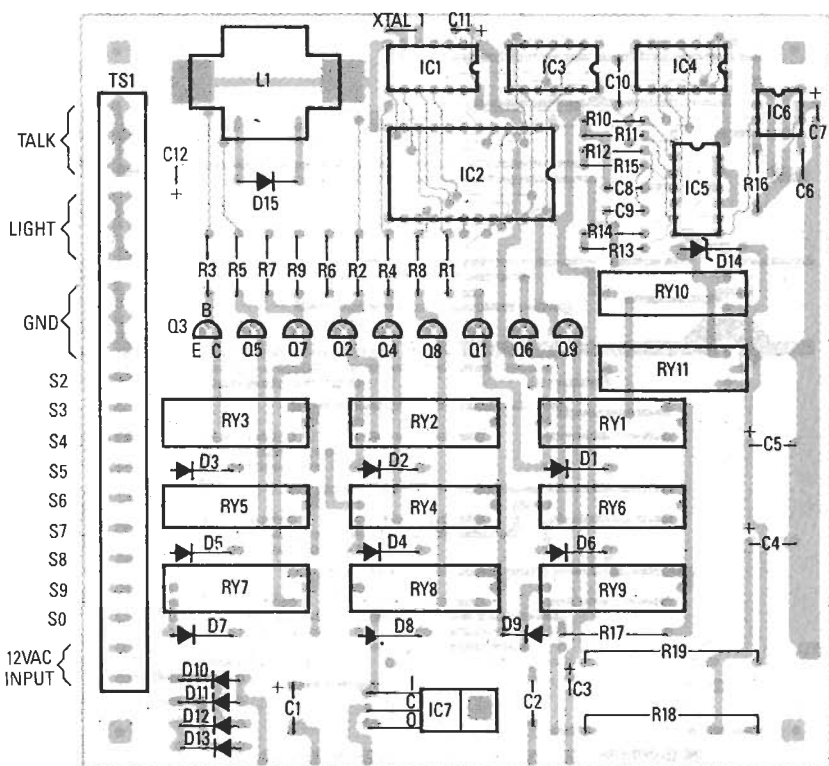


FIG. 2—MOUNT ALL COMPONENTS as shown here. The terminal strip (TS1) is composed of ten dual terminal strips.

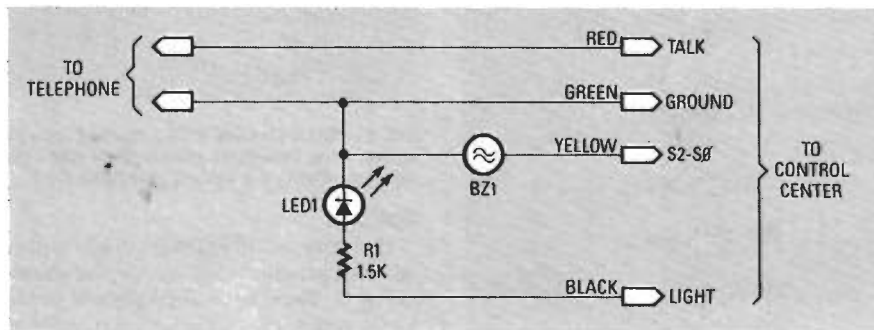


FIG. 3—AN LED, A BUZZER, and a resistor should be mounted in each telephone.

### Assembly

Terminal strip TS1 is composed of ten terminal-strip pairs; each PC-board pin and screw terminal is located 0.2" from its neighbor. Radio Shack and Digi-key (P. O. Box 677, Thief River Falls, MN 56701) sell connectors from different manufacturers that will fit the board.

Foil patterns for etching a suitable PC board are shown in PC Service. Beware that the board is double-sided, so, if you etch your own board, you'll have to make provision for soldering all component leads (about 200) on both sides of the board. Alternatively, you can buy a board from the source mentioned in the Parts List. The commercially available board has plated-through holes.

Whether you buy or build your PC board, check it carefully for shorted and open traces before mounting any components. Then, after solving any problems, solder the components to the board ac-

ording to Fig. 2. Mount the low-profile components and IC sockets first.

Don't insert the IC's in their sockets until you perform the initial check-out. The exception is IC7 (the 7805); bend its legs 90° so that it rests flat against the PC board. Secure its tab to the board.

Lay the PC board on a non-conductive surface and connect a 12-volt AC source to the board through a ½-amp fuse. Measure the voltage at the input terminal of IC7; it should be about 16-volts DC. The output of IC7 should be between 4.8- and 5.2-volts DC. If those voltages are correct, remove power and insert the IC's in the appropriate sockets.

### Testing

You'll need to add a buzzer, an LED, and a resistor to (at least) two telephone sets as shown in Fig. 3. Four wires connect each phone to the control center: three for the talk, light, and ground circuits, and one from each phone to the

### PARTS LIST

All resistors are ¼-watt 5% unless otherwise noted.

R1-R9, R12, R15—10,000 ohms  
 R10, R13—1 megohm  
 R11, R14, R16—100,000 ohms  
 R17—4.7 ohms, 1 watt  
 R18, R19—15 ohms, 1 watt

**Capacitors**  
 C1, C4, C5—1000 µF, 35 volts, electrolytic  
 C2, C10—0.1 µF, ceramic disk  
 C3, C7, C11, C12—4.7 µF, 35 volts, electrolytic  
 C6, C9—0.01 µF, ceramic disk  
 C8—0.022 µF, ceramic disk

**Semiconductors**  
 IC1—M-959 dial-pulse counter (Teltono)  
 IC2—4514 4-to-16 line decoder  
 IC3—4001 quad NOR gate  
 IC4, IC5—4011 quad NAND gate  
 IC6—555 timer  
 IC7—7805 5-volt regulator  
 D1-D13, D15—1N4001 rectifier  
 D14—1N4733 5.1-volt, 1-watt Zener diode  
 Q1-Q9—2N2222

**Other components**  
 F1—0.5 amp, 250 volts  
 RY1-RY9—reed relay, 12 volt  
 RY10, RY11—reed relay, 5 volt  
 L1—500-ohm audio transformer (see text)  
 S1—SPST toggle  
 T1—12-volt, 0.5-amp wall transformer  
 TS1—20-position terminal strip (see text)  
 TS2—2-position barrier block  
 XTAL—3.58 MHz, color-burst

**Miscellaneous: Enclosure, fuse holder, telephone wire, 12-volt piezo-electric buzzers, LED's, and resistors for telephone stations, etc.**

Ordering info  
**Note: The following are available from COM-TECH, 1856 S. Highland, Jackson, TN 38301: PC board, IC1 (M-959), IC2, and L1, \$34.95; 12-volt, 0.5-amp wall transformer, \$11.95; 12-volt DC latching relay with 3-amp contacts, \$19.95. All orders add \$3 (\$5 Canada) for shipping. Foreign orders must include U. S. funds and \$8.00 for shipping. Tennessee residents add 7% sales tax.**

proper s2-s0 output. At the phone end, the red and green wires going to the telephone should connect in parallel with the corresponding wires already in the telephone. The other two wires are connected to the LED and the buzzer as shown.

With power disconnected, connect two telephones to the control center. Apply power, and lift the handset from the base of one telephone. All the busy LED's should light up, and a dial tone should be heard through the handset. Dial the number corresponding to the other telephone. The busy LED's should flash during dialing. After dialing, a ringback tone should be heard over the handset, and the other phone should buzz, both for about one second. Lift the handset from the other phone, if all is well, you should be able to communicate over the two handsets.

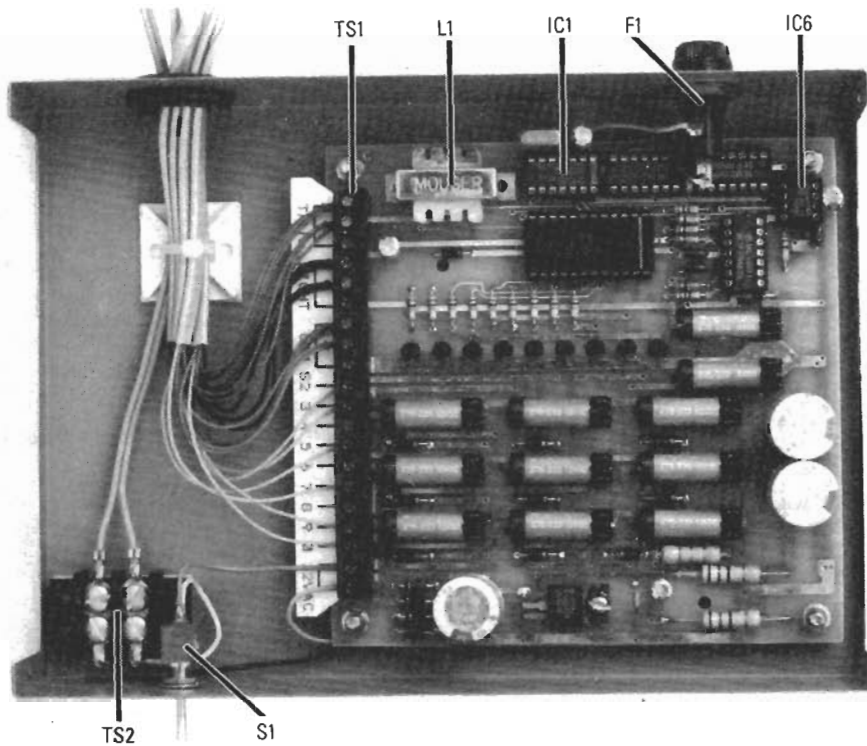


FIG. 4—MOUNT THE PC BOARD, fuse, and power switch as shown here. Run the interconnecting wires through a hole in the rear of the box; insulate the hole with a large grommet.

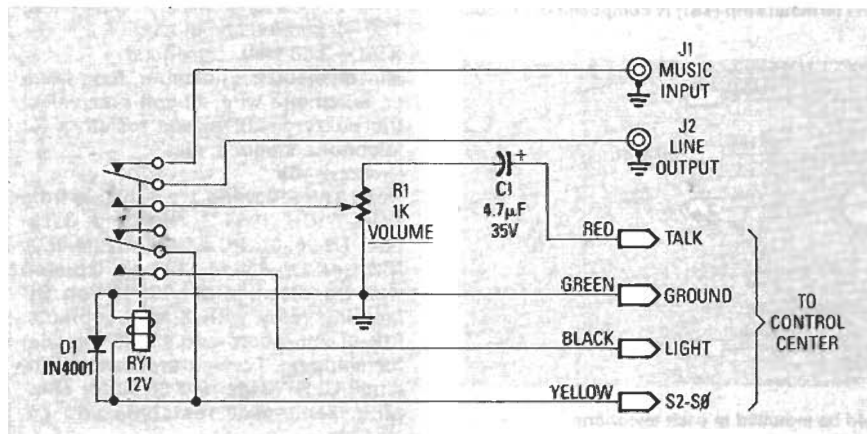


FIG. 5—ADD A PAGING/MUSIC circuit as shown here. Connect the music input to the output of a transistor radio or another suitable source.

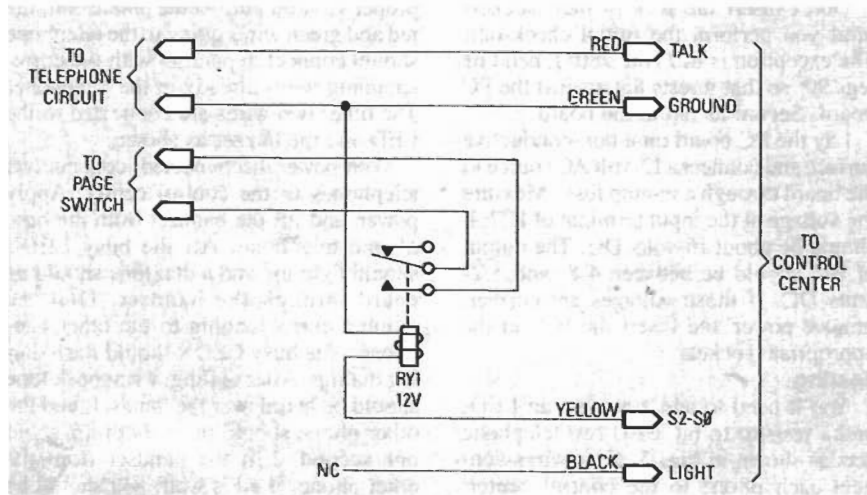


FIG. 6—CONNECT A WIRELESS PHONE as shown here.

To verify that all other channels are working correctly, repeat the above test while connecting one test phone to each of the s2-s0 pins of TS1.

### Installation

If a problem is detected during testing, track down the source and correct it. When all circuits work, mount the PC board in a suitable enclosure, as shown in Fig. 4.

Then you'll want to wire your premises. Each telephone can be located as far as 2000 feet from the control center. When installing cable, don't route it near high-voltage AC wiring because hum could be induced in the system.

All red wires from each telephone should be connected together and to the talk terminals of TS1. Likewise, all green wires should go to the ground, and the black wires to the light terminals of TS1. Each of the yellow wires should be separately connected to one of the s2-s0 terminals of TS1.

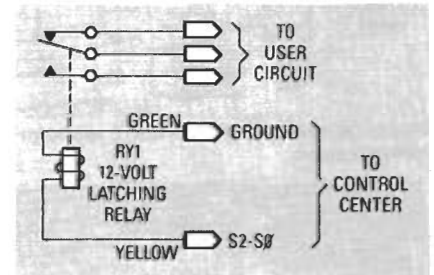


FIG. 7—YOU CAN CONNECT a latching relay as shown here. Dialing its station once turns the relay on; dialing it a second time turns it off.

### Options

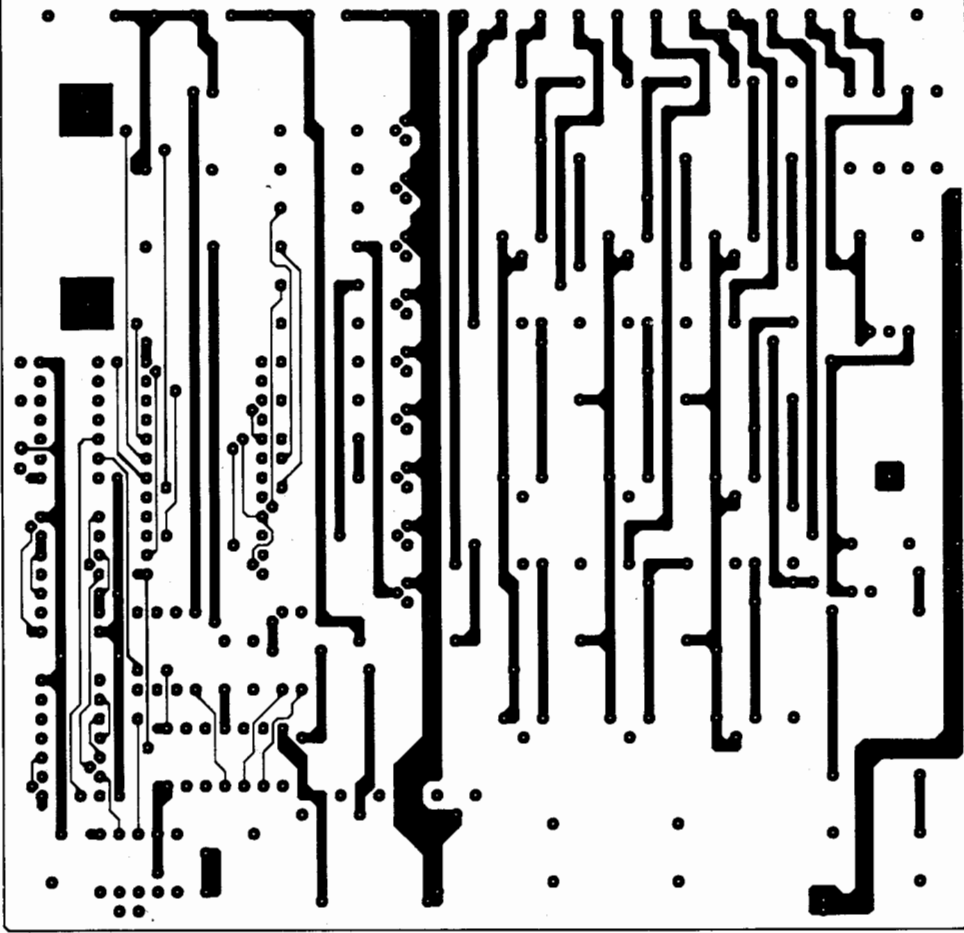
One very useful option is to add a paging/background music circuit, as shown in Fig. 5. The relay is normally off, so the music source (J1) is connected directly to the line output (J2) through the upper set of contacts. But when that station is called, the relay activates, and the talk circuit is connected to the line output.

The s2-s0 outputs of the control center can be connected to devices other than telephones. For example, you could use one output to activate the "page" feature of a cordless phone. You could page someone carrying a cordless phone by connecting the paging switch in the base station to the control center as shown in Fig. 6.

Another option is to connect a latching relay to one control center output as shown in Fig. 7. Each time the station that circuit is connected to is called, the output of the relay would change state.

There are many other uses for the circuit. For example, you could control the latching relay from a cordless telephone to provide remote control of an electric garage-door opener. Many other uses will doubtless occur to you as you ponder the possibilities!

R-E



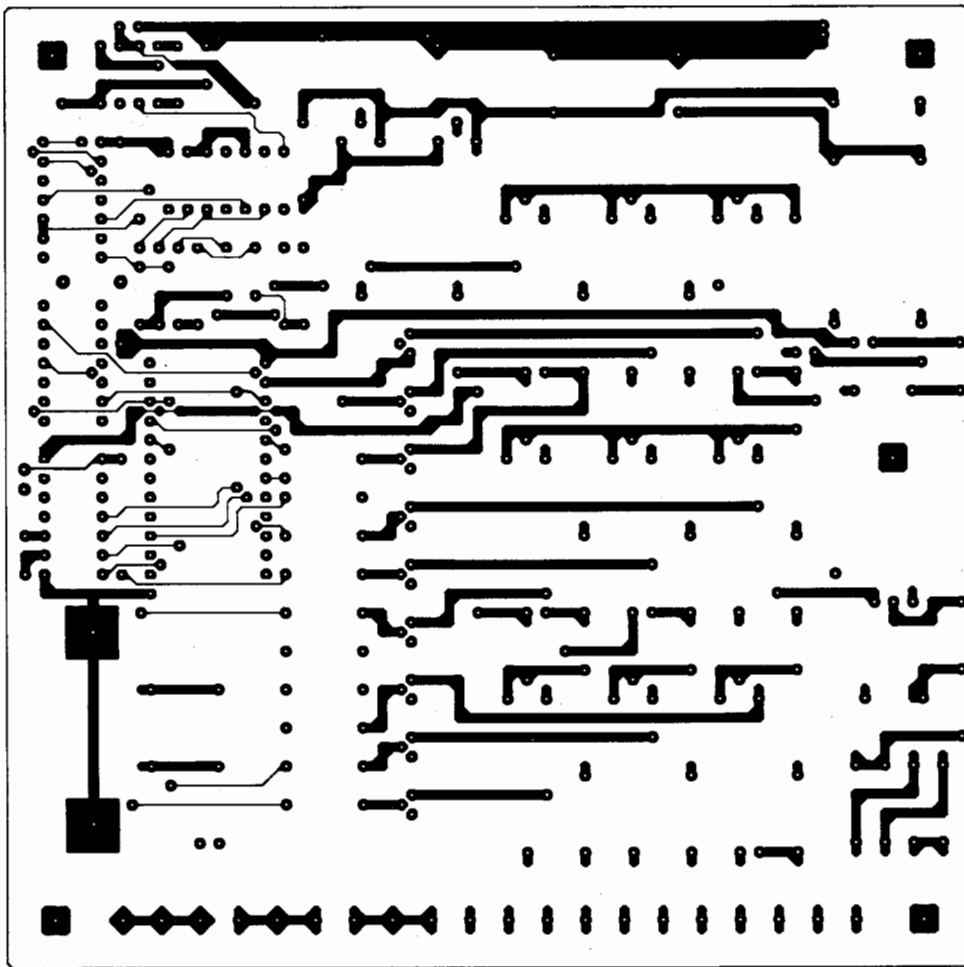
5 INCHES

One of the most difficult tasks in building any construction project featured in **Radio-Electronics** is making the PC board using just the foil pattern provided with the article. Well, we're doing something about it.

We've moved all the foil patterns to this new section where they're printed by themselves, full sized, with nothing on the back side of the page. What that means for you is that the printed page can be used directly to produce PC boards!

**Note:** The patterns provided can be used directly only for *direct positive photoresist methods*.

**COMPONENT SIDE** of the telephone intercom.



5 INCHES

**SOLDER SIDE** of the telephone intercom.

# HOBBY CORNER



EARL "DOC" SAVAGE, K4SDS,  
HOBBY EDITOR

## Intercom modifications

OVER THE PAST FEW MONTHS WE RECEIVED several inquiries concerning intercoms: It appears that a number of you wish to modify them to suit your specific needs. So, this month, we'll take a look at a couple of the changes that you want to make and see what must be done to make those modifications work.

Our first request comes from Dick Ryszykow (NY), who wants to know how a speaker can be made to function as a microphone as well as a speaker. Well, all speakers are actually crude microphones. You can prove that to yourself by taking your stereo headphones and using them as microphones for your tape deck. In order to use a speaker more efficiently as a microphone, however, you have to be concerned about impedance matching. The impedance-matching de-

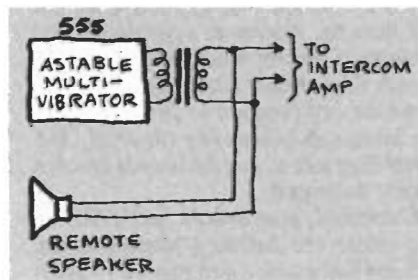


FIG. 2

vice usually used is an audio transformer. The high-impedance (say, 1000 ohms) side of the transformer is connected to the amplifier and the low side (say, 8 ohms) to the speaker. Thus, both devices "see" the proper impedance and they work well together. But how can we use a speaker as an input device or microphone?

First, consider that a microphone must produce electrical signals corresponding to the audio

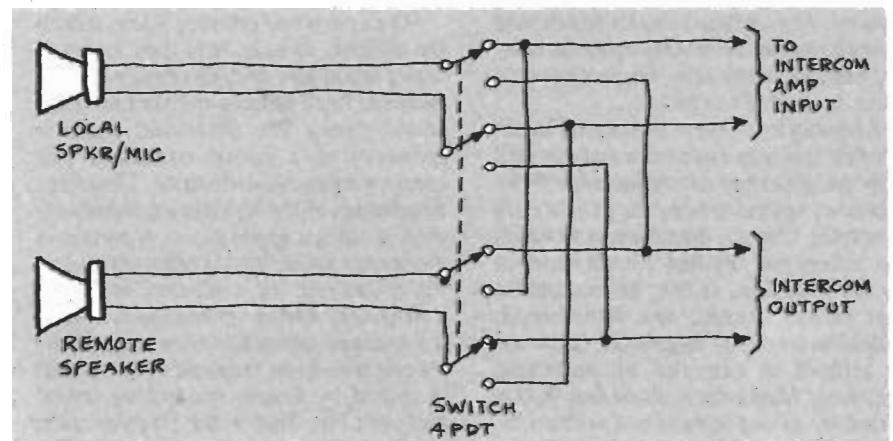


FIG. 3

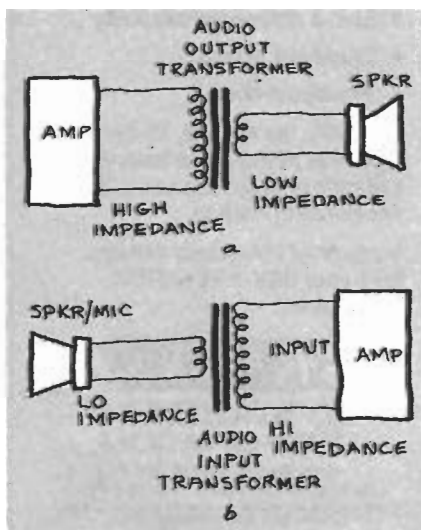


FIG. 1

vice usually used is an audio transformer.

Speakers are low-impedance devices, normally on the order of 3.2 to 16 ohms. Most amplifiers, whether for intercoms or whatever, have high-impedance input and output circuits. In order to avoid distortion and signal losses, the impedances of the speaker and amplifier-output must be matched.

You are familiar with the use of an output transformer between an amplifier and speaker, as shown in Fig. 1-a. The high-impedance (say,

1000 ohms) side of the transformer is connected to the amplifier and the low side (say, 8 ohms) to the speaker. Thus, both devices "see" the proper impedance and they work well together. But how can we use a speaker as an input device or microphone?

First, consider that a microphone must produce electrical signals corresponding to the audio



plifier. Remember, the impedances must be matched. Figure 1-b shows how an audio-input transformer is used to match two different impedances.

You may have noticed that the transformers in diagrams 1-a and 1-b of the figure are alike, except that their primaries and secondaries are reversed. Actually, the only difference between input and output transformers is the name—you can use an output transformer as an input device and vice-versa. A transformer doesn't care whether a signal is going from its high side to its low side or the other way around—just hook it up in the direction you need!

That information can be very handy for things other than intercoms. For instance, a few years ago, I needed to record the sounds from a large area that was some distance from the tape machine. Of course, I could have used a standard microphone and a long run of microphone cable, but that stuff is expensive and the cable has fairly high signal losses. So instead, I simply connected an output transformer "backwards" (with high impedance side going to the recorder) and ran a long length of old lamp cord between it and a speaker.

My "microphone" (in this case the speaker) was quite sensitive and the recording was perfect. That setup can be used for recording anything, for instance bird calls, from a remote location!

### Multiple inputs

David O'Brien (MN) would like to add several call stations to his intercom and have each of them give a distinctive sound, so that he can tell which is being used. There are a number of ways to accomplish that, David, but I'll show you one that does not require fooling around with the intercom's amplifier wiring.

Take a look at Fig. 2; it shows an astable multivibrator (oscillator) connected across the remote speaker/microphone wires of an intercom. (The oscillator circuit is not given because of the large number of 555 oscillators that have appeared in past "Hobby Corner" columns and other places.)

When the oscillator is turned

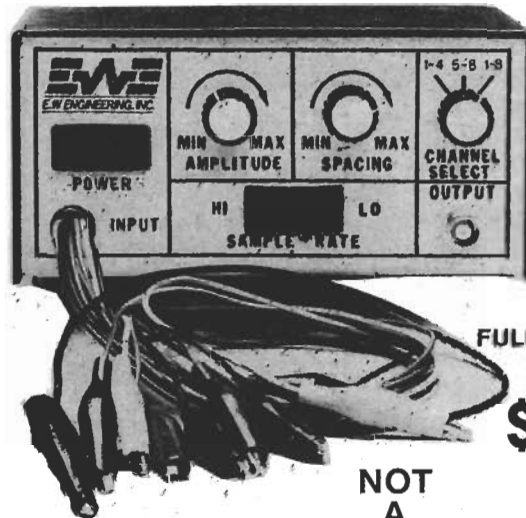
on, an audio signal is input to the intercom and you'll hear it just as you would a voice in the remote speaker. By making the frequency of the oscillator adjustable, and using several at different stations, you can tell from which location the tone is coming. For example, you could put an oscillator with a high pitch at the front door and one with a low pitch at the back. Any practical number of oscillators may be used at various locations to tell you which one is activated. Just

remember: You must be able to distinguish between the tones.

Although Fig. 2 shows that we used a transformer as coupling device between the oscillator's output and the intercom wires, you'll have to do a bit of experimenting to determine just what to put in the circuit there. It depends on the input circuit of the amplifier and the number of oscillators you add. Direct connection *might* work but you'll most likely have to add a

*continued on page 94*

## VIEW 8 TRACES ON YOUR SINGLE OR DUAL TRACE SCOPE WITH THIS LOW COST DEVICE!!



Now you no longer have to spend thousands on an expensive multi-trace oscilloscope - our single trace Hitachi scope combined with this module will allow you to view up to 8 simultaneously occurring analog or digital (or both) signals in their real time and amplitude relationship. The MPX 101 may be used on any oscilloscope, whether single, dual or multiple traces. Its low cost makes it a particular favorite for designers, testers, hobbyists and repairmen who want to compare and analyze displayed signals in a timing diagram format. The controls on the front panel of the metal case allow you to vary amplitude and spacing of the displayed signals.

**MODEL MPX101  
FULLY ASSEMBLED & TESTED!**

**\$99<sup>88</sup>**  
FULL 1 YEAR REPLACEMENT WARRANTY

- Made In The United States -

### SPECIFICATIONS

Inputs: 8 signals plus ground via 9 input leads terminated with alligator clips  
Bandwidth:  $\pm 1$  dB to 5 MHz  
Impedance: 10.9 K  
Input Voltage:  $\pm 5$  V peak (diode clamped to  $\pm 5$  Volt supplies)  
Output: Staircase waveform summed with input signals, 0-800 mV full scale  
Step Amplitude: Variable 0 to 150 mV/step  
Signal Voltage: Variable 0 to

150 mV/step @ 5V Input  
Multiplex Rate: Switch selectable, 40 KHz or 4 KHz  
Impedance: 50 Ohms  
Power: 105-135 VAC @ 1 V a  
Dimensions: 6.25" x 3.25" x 4.75" (WxHxD)  
Operating Temperature: 0-40°C  
Weight: 1 lb. 10.5 oz.  
Warranty: one year full replacement warranty from date of purchase  
Lighted on/off power switch  
Wood grain finished metal case

**DISTRIBUTOR AND REPRESENTATIVE INQUIRIES INVITED**

**EWE**  
E.W. ENGINEERING, INC.

VISA, MASTERCARD, AMEX TELEPHONE ORDERS ACCEPTED!  
6 Herman Drive, E. Granby, CT 06026 □ 203/651-0285

CIRCLE 82 ON FREE INFORMATION CARD

# Bilateral speaker networks form switchless intercom

by Frank Kasparec  
St. Poelten, Austria

Only one transducer—a dynamic loudspeaker—is required at each station of this intercom to permit transfer of audio information in both directions simultaneously. Having no need for push-to-talk switches, the circuit is less costly and less bulky than conventional transceiver units. Undesired acoustic distortion normally encountered in this type of transmission system is eliminated by using a simple phase-compensation network.

As described for the receiver portion of the unit, audio signals at the input are converted into their acoustical equivalent,  $S_1$ , by the speaker after passing through the RC network made up of  $C_1$  and resistors  $R_3$  to  $R_7$ , which is required to offset the frequency-dependent phase shift created by the speaker's inductance. When properly set, potentiometer  $R_6$  also cancels the feedback (talkback) voltage appearing at the noninverting input of the 741 differential amplifier, which is normally used to amplify the electrical equivalent of the acoustic vibrations,  $S_2$ , hitting the cone of the speaker in the transmit mode.

As expected, the compensation network responds similarly to voltages emanating from the speaker's coil, acting to minimize phase distortion at  $V_{out}$ . In this case, however, input signals to the differential amplifier are in the millivolt range. The 741 op amp provides a gain of  $R_9/R_8$  and thus the needed amplification at  $V_{out}$ . Note that in most cases a power stage will be required following the 741 op amp to energize a loudspeaker.

The design of the phase-shift network, although easy, must be done with care if acoustic feedback is to be reduced to a minimum. The phase differences as seen by the differential amplifier must be canceled, and thus  $(R_1 + R_2)/Z_1 = Z_{C1}/(R_3 + R_4/2)$ , where  $Z_1 = jX_{L1}$ ,  $Z_{C1}$  is the reactance of capacitor  $C_1$ , and  $L_1$  is the speaker's inductance. It is also assumed that  $R_2$  is approximately equal to  $R_1$ , and  $R_3$  and  $R_4$  are much smaller than  $R_5$ – $R_7$ . For the general purpose speaker,  $R_1$  will be about 4 or 8 ohms. The above equation reduces to  $(R_1 + R_2)(R_3 + R_4/2) = L_1/C_1$ . At the same time, for direct current balance at the output of the op amp,  $R_2/R_1 = (R_7 + R_6/2)/(R_5 + R_6/2)$ .

To determine the element values in the phase-compensation network, and to estimate the frequency response of the unit, it is necessary to measure both the loudspeaker's dc resistance and its inductance. The best way to find the dc resistance is to use a digital ohmmeter. To determine coil inductance, it is necessary to apply an audio signal (preferably at  $f = 1$  kHz) to the loudspeaker, as shown in the inset, and to measure  $A$ , the ratio of the root-mean-square output voltage to the rms input voltage. The coil inductance is then given by:

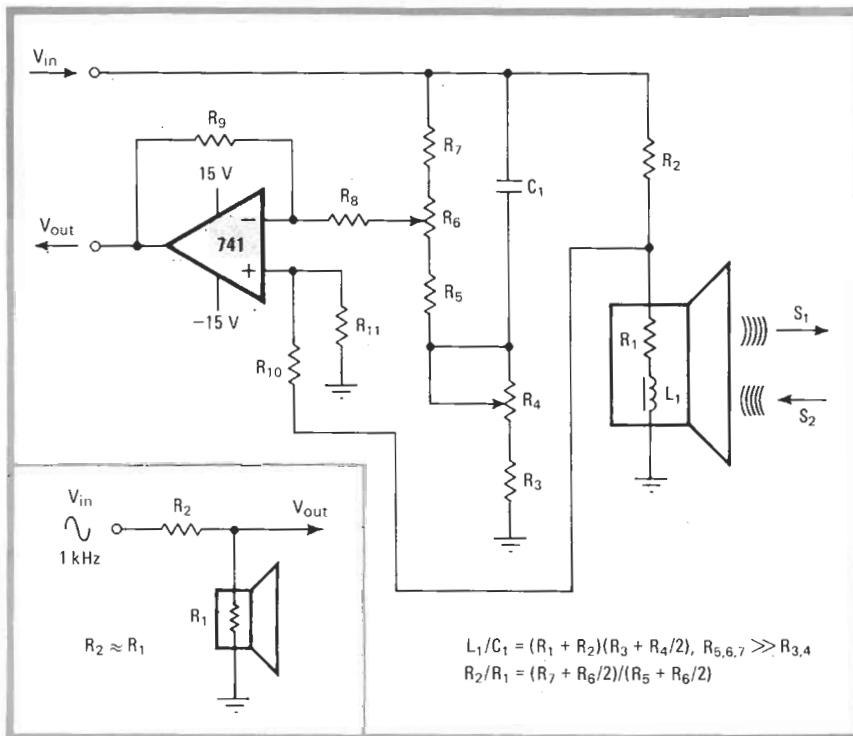
$$L_1 = [R_1^2 - A^2(R_1 + R_2)^2 / 4\pi^2 f^2 (A^2 - 1)]^{1/2}$$

Rearranging the equation:

$$A = [R_1^2 + 4\pi^2 f^2 L_1^2 / (R_1 + R_2)^2 + 4\pi^2 f^2 L_1^2]^{1/2}$$

and it can be seen that the speaker response tends toward a high-pass characteristic. Because  $L_1$  is generally negligible, the frequency response is largely flat over the audio range.

Adjustment of the circuit is simple.  $R_4$  is initially set to null the output of the differential amp under no-signal conditions. A square wave is then applied at  $V_{in}$  and  $R_6$  set to minimize  $V_{out}$ . □



**Double duty.** One-transducer intercom station using dynamic loudspeaker minimizes cost and bulk of conventional units. Compensation network ( $R_3$ – $R_7$ );  $C_1$  reduces phase distortion created by speaker. Speaker inductance may be easily measured (see inset) in order to set component values in compensation network.

## Two wire intercom

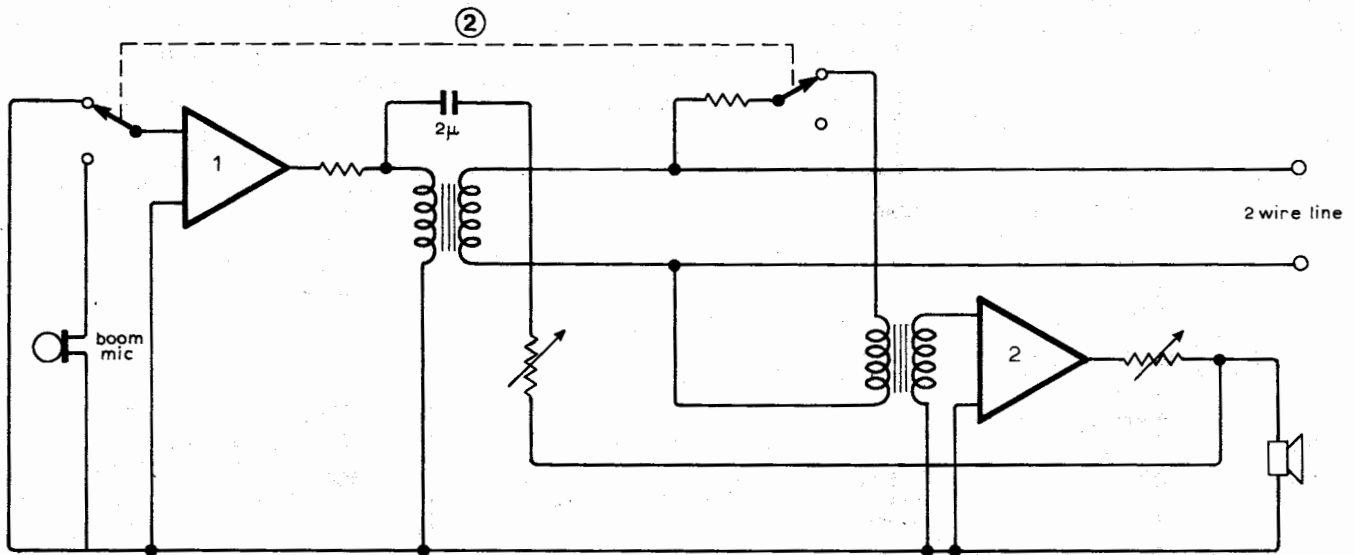
In telephone circuitry the multi-tapped inductor at the telephone-receiver end converts the two wire line into an effective four wire system to give side tone control. A similar principle using only one pair of wires per station in a multichannel intercom network is possible, but obtaining side tone control without using inductors and v.d.rs is difficult due to unavoidable coupling between the receiver and transmitter amplifiers which causes feedback. This

circuit solves the problem of obtaining side tone control and does not suffer from instability. Receiver amplifier 2 is disconnected from the balanced lines when the switch is operated, and a receiver path is connected via the  $2\mu\text{F}$  capacitor. This allows side tone control and retains other messages on the lines at a lower level. Amplifiers such as the TDA1054 can be used because they contain a compression circuit. The presence of multiple signals on the

balanced lines does not seriously alter the listening level at the earpiece when such compression amplifiers are used.

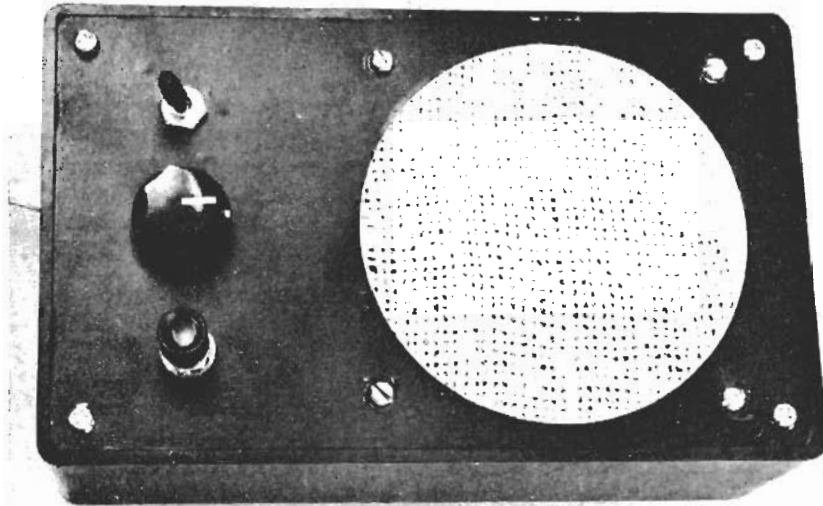
The above circuit has been tried on a 20 student intercommunication network in a language laboratory with satisfactory results. No further amplification of the signal via the  $2\mu\text{F}$  capacitor was found necessary. The isolating transformers have 1:1,  $10\text{k}\Omega$  windings.

K. Soma,  
Singapore



# SIMPLE INTERCOMS

Three intercom systems using the LM380 audio IC.



AN INTERCOM SYSTEM is not only one of the most useful projects that one can build, it is also one of the easiest.

Commercial intercoms are of course readily available — often at prices even lower than one could build the units for oneself. Nevertheless a project of this nature is still more than justifiable for not only does it provide valuable

experience but the system can also be built to suit one's exact needs.

In this unit, as with most simple intercoms, a speaker doubles as both microphone and loudspeaker, its role being changed from one to the other by a pushbutton 'talk/listen' switch.

As a loudspeaker is not particularly efficient when used as a microphone, we have used a step-up transformer

and LM 380 integrated circuit amplifier.

Whilst the circuit of a suitable power supply is shown (Fig. 4) current drain is low and battery operation may be used if the unit is not used very much.

## Construction

Construction is very simple indeed and, as there are very few components, we suggest that the amplifier be built onto matrix board or similar. A heatsink is not required for the LM380, when working into a 15 ohm, be sure to obtain speakers having this impedance. Higher impedances will result in much lower power output, and lower impedance speakers (eg 8 ohms) will require the use of a heatsink.

The internal layout of the prototype unit is shown in Fig. 5. Note that we used the system connections shown in Fig. 2. This system requires (easily obtainable) single-pole push buttons but requires three interconnecting wires between station 1 and station 2. It also has the advantage that an

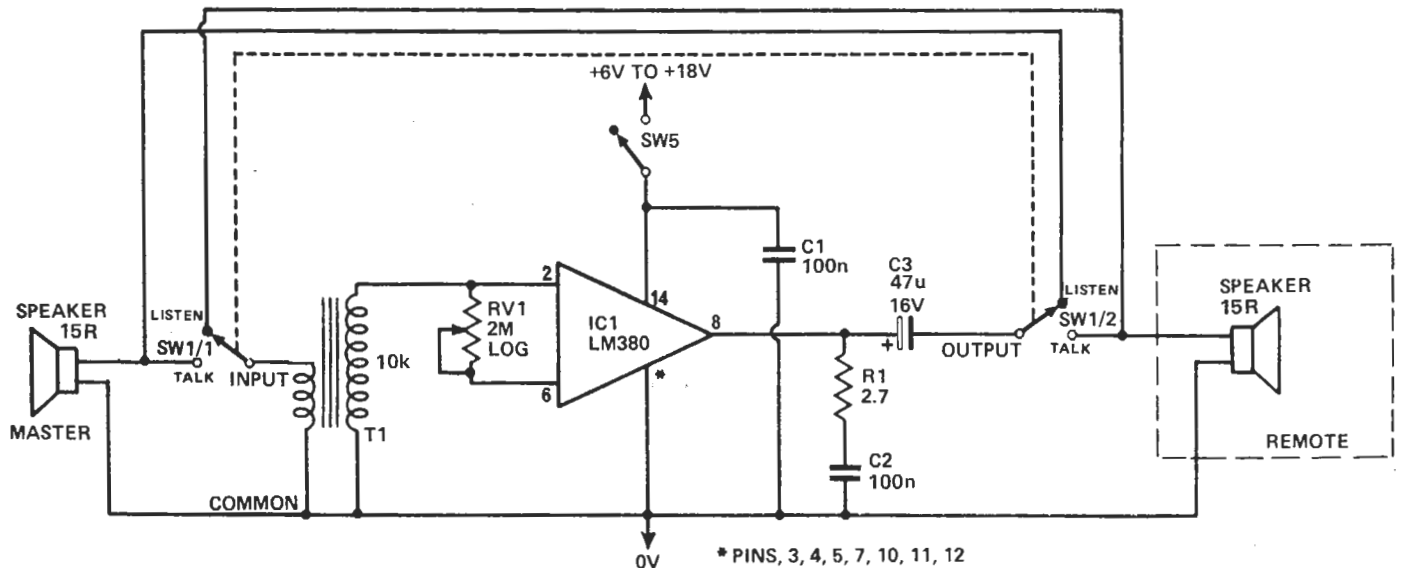


Fig. 1. About the simplest possible intercom, in this arrangement the master station listens to remote station at all times until TALK button is pressed.

output is heard only when the TALK button at either station is pressed.

The system shown in Fig. 1 only requires two wires to the remote station but has the disadvantage that station 1 is always listening to station 2 except when the talk button is pressed.

If two remote stations are required the system illustrated in Fig. 3 should be used. Again this requires three wires to each of the remote stations. Switch SW1 is the push-to-talk button and SW2 selects the required remote station.

We used a small 9 volt battery to power our unit but, if continuous use is expected, it would be wiser to use a larger battery. For example, the standby current is about 3 mA which would result in only about 150 hours operation from a small battery. The best battery system would probably be two 6 volt lantern batteries connected in series. Alternatively a simple power supply such as shown in Fig. 4 could be used. ●

## HOW IT WORKS

The (approximately) 1 millivolt output from the speaker acts as a microphone, applying its output to T1. The transformer is a standard audio type used in reverse.

The output of T1 is connected directly to the non-inverting (+) input of the LM380 (pin 2) and also, via potentiometer RV1, to the (-) input. Since the input resistance of the IC is about 150 k, the signal level at the negative input is dependant upon the setting of RV1.

The IC, as with all differential amplifiers, amplifies the *difference* in signal level between its two inputs, pins 2 and 6. Thus RV1 effectively acts as a volume control.

With the connections shown on Fig. 1, the remote station speaker acts as a microphone, applying its output to T1. The output of T1 is amplified by the IC and applied to the MASTER speaker. Thus the master station is listening to the remote station at all times other than when SW1 is pressed.

When SW1 is pressed the master speaker becomes the microphone and the remote speaker receives the amplified signal from the IC.

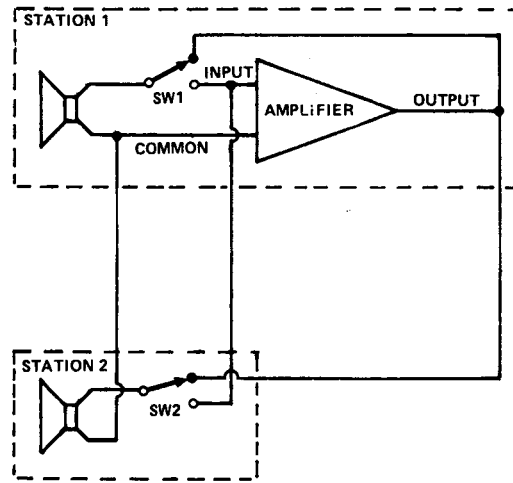


Fig. 2. This system gives privacy to station 2 but requires three wires between stations. The amplifier circuit is identical to that in Fig. 1.

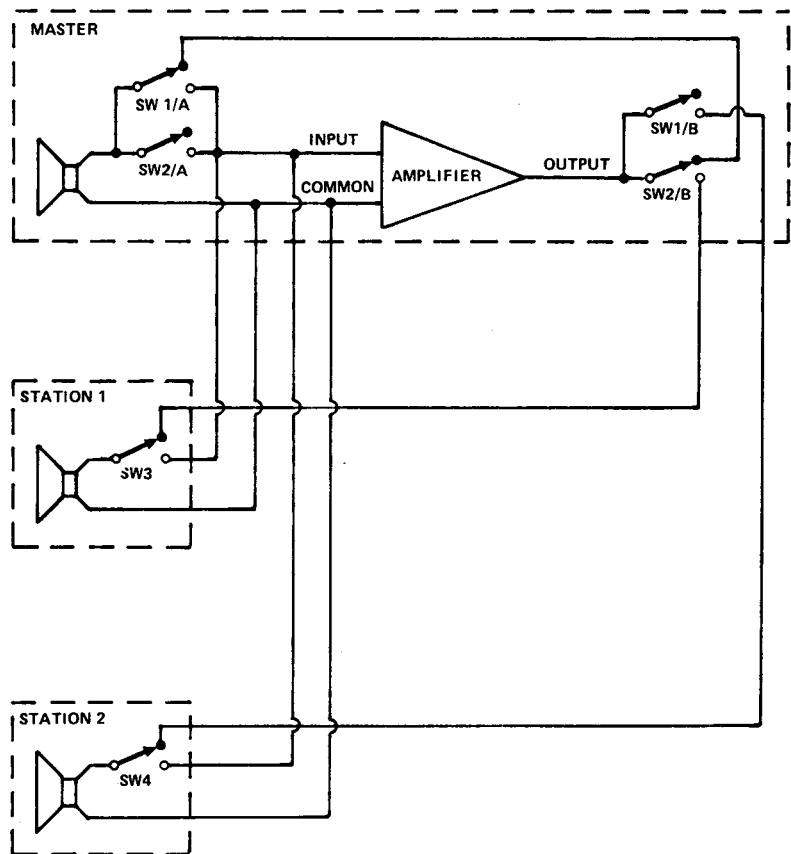


Fig. 3. An extension of the previous system to a master and two slaves requires three wires from the master to each slave, and a more complex switching arrangement.

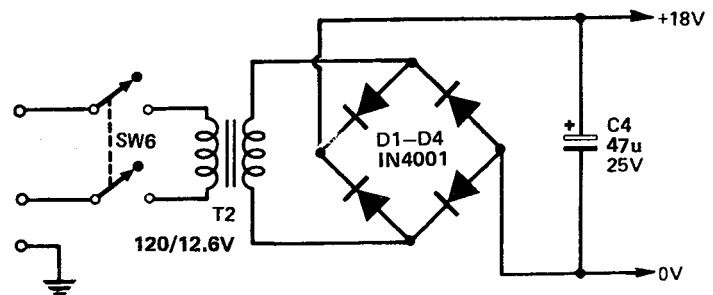


Fig. 4. A simple power supply for use with the intercom.



## PARTS LIST

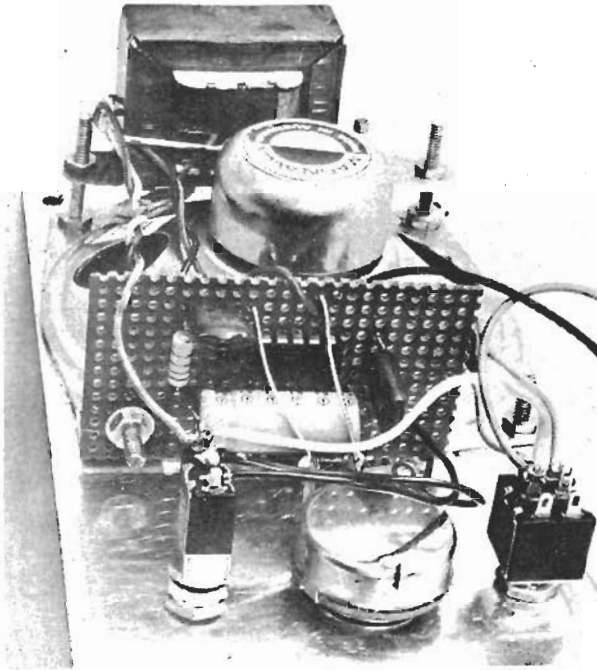


Fig. 5. The method of assembly may readily be seen from this internal view.

R1	2.7 ohm 1/2 watt 5%
C1, C2,	100n polyester
C3	47U 16 volt-electrolytic
IC1	LM380
T1	10k to 8R audio transformer
RV1	2M log rotary

	System 1	System 2	System 3
SW1	DPDT pushbutton	SPDT pushbutton	DPDT pushbutton
SW2	—	SPDT pushbutton	DPDT pushbutton
SW3	—	—	SPDT pushbutton
SW4	—	—	SPDT pushbutton
SW5*	SPDT toggle	SPST toggle	SPST toggle

Loudspeakers 15 ohm 3" diameter  
Six to 12 volt battery. \*  
Plastic or metal box, piece of matrix board, bolts and nuts etc.  
\* required for battery versions only.

### Power Supply

D1-D4	IN4001 or similar
C4	470u 25 volt electrolytic
C5	25u 25 volt electrolytic
T2	120/12.6 volt
SW6	DPST toggle

## if you met these people, would you know they had Cystic Fibrosis?



Children and adults with CF often look as healthy as you. To stay that way they go through a rigorous program of medical treatment and therapy every day.

CF is inherited and incurable.

You know it's good to be alive... so do CF people.

With your help research will find a cure or control.

**take action**  
**MAKE CF PEOPLE**  
**WINNERS**



Canadian Cystic Fibrosis Foundation

# CIRCUIT IDEAS

## Multiple station two-way intercom

This circuit shows a four-station, two-way intercom, where any station can communicate in privacy with any one of the others. Each two-station link-up is assigned a code, three bits being sufficient as there are six possible link-ups. The appropriate code is selected by Sw 1-4, and is generated at each station. All the station codes are "OR-ed" by IC<sub>3</sub> and decoded by IC<sub>4</sub> to drive a matrix of analogue switches which couple the appropriate audio inputs and outputs. Code 000 is allocated to a system-free status, indicated by l.e.ds 1 and 4 being on. A system-busy status is indicated by the l.e.ds flashing. When a code is

### Components

IC <sub>1</sub>	CD4071	IC <sub>3</sub>	CD4011
IC <sub>2</sub>	CD4081	IC <sub>9,10</sub>	CD4025
IC <sub>3</sub>	CD4075	A <sub>1</sub>	LM380
IC <sub>4</sub>	CD4028	A <sub>2</sub>	741
IC <sub>5,6,7</sub>	CD4016		

Station links	Code
1 to 2	001
1 to 3	010
1 to 4	011
2 to 3	100
2 to 4	101
3 to 4	110

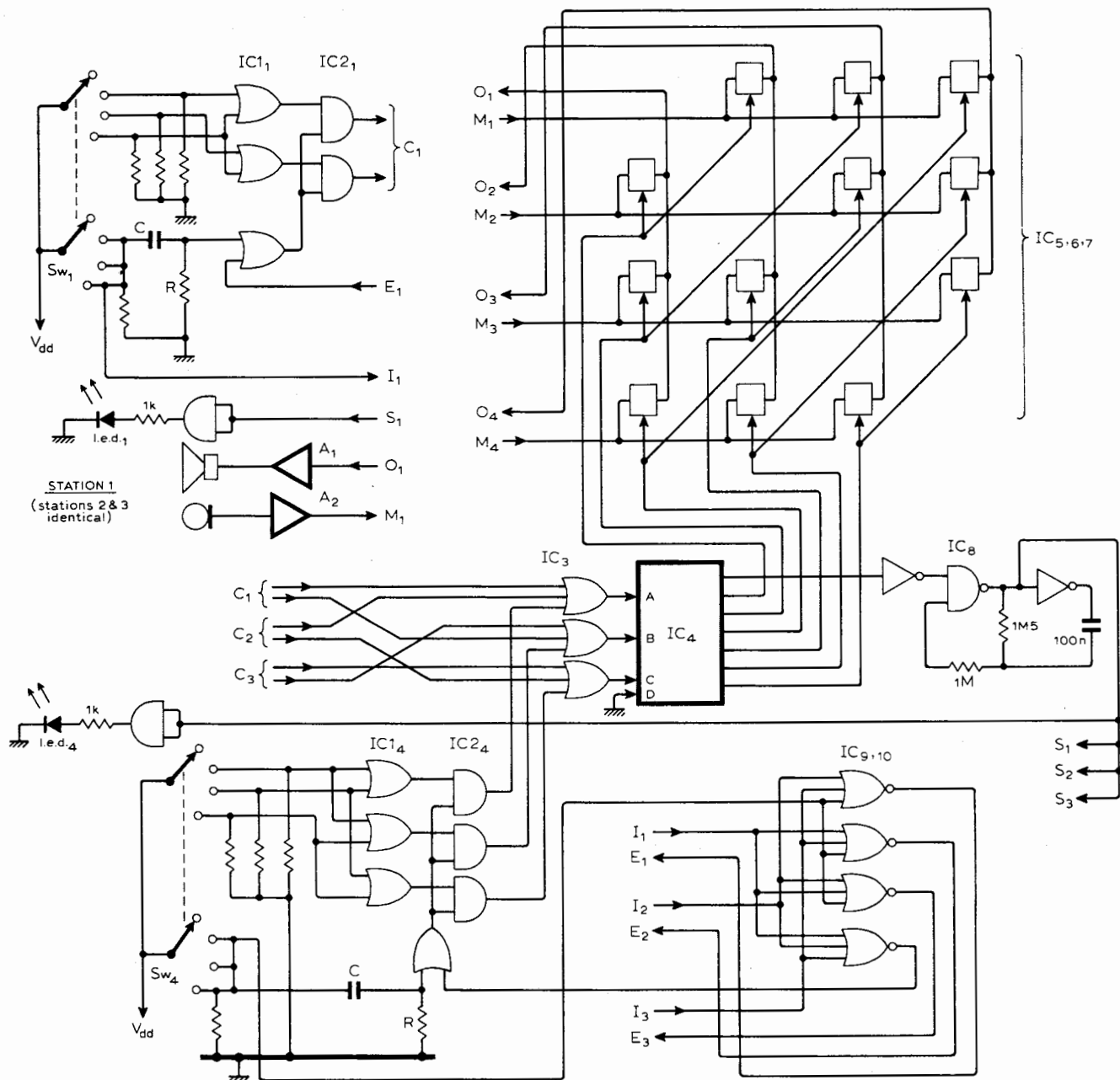
Electronics housed in station 4 as all three bits are used.

selected, the station inhibit output is taken high and this forces the enable inputs on all other stations low, thus

preventing any further codes being generated at the station outputs. However, if a station wishes to use the system and selects any of the other stations while the system is busy, it will flash a code for a time determined by CR thus interrupting the established link.

If the electronics are housed in one station, only two code wires are required to the other three. The system can be easily expanded up to six stations, where there are fifteen possible link-ups, by using a 4-bit code and a CD4514, 4-to-16 line decoder with an enlarged matrix of analogue switches.

B. Voynovich,  
Norwood,  
Middx.



# Five-state LED display monitors paging system

by D. F. Fleshen  
Springfield, Va.

This paging station circuit uses a single red- or green-light-emitting diode to alert the user to any of five distinct paging conditions. It is a simple, extremely easy-to-build monitor designed to be part of a large paging system. Two relays and a 555 timer send a signal to the LED (a Monsanto MV5491 or Xciton XC5491) to produce the following signals:

- Off (no power) indicates that paging has been cut off to that station or the region in which it is located.
- Steady green signifies that the station is operational, but is not being paged.
- Steady red is the individual station's paging signal.

- Flashing green tells the user that all stations in the system are being alerted.
- Flashing red signals an emergency situation to all monitoring stations.

An external signal triggering relay A controls the color of the LED, changing it from its green (idle) condition to red. Relay B is excited when all stations are to be called. This second relay puts the 555 timer in the astable mode, changing the LED's usual dc state to an on-off oscillation of about 7.5 hertz at a duty cycle of 50%. The frequency of oscillation can be adjusted by suitably selecting  $R_1$  and  $C_1$ . In the emergency red flashing mode, both relays must be tripped by external signals.

This circuit can readily be adapted to signal a single panel-mounted lamp. To derive the five operating modes, the relay contacts must be replaced by the contacts on a suitably wired rotary switch.

Supply voltage for the 555 may vary from 9 to 15 volts; with a 12-v supply, current drain is about 40 milliamperes. □

