TOA
EXES-6000
SYSTEM

PRINCIPLE

OF

OPERATION

AND

TROUBLESHOOTING

TOA Corporation
KOBE, JAPAN

PREFACE

This booklet readily explains a working principle of the EXES-6000 system from a standpoint of the entire system and so, does not refer to the details of the circuits, CPU's or IC's. It is compiled with an emphasis placed on the descriptions of audio and dial signal flows that are a base of the system, and the timing of their relative switches, which are explained by using block diagrams of actual circuits. The booklet is so compiled that the basic items considered necessary, such as link, PAM, time division switching, etc. are first explained priot to entering into the explanations of the actual systems with the aim of making readers knowledgeable on the system by degrees.

TABLE OF CONTENTS

I. BASIC ITEMS

1. Hands-free System

1-1. Basic Principle of the Hands-free System

Fig. 1 shows a basic principle of the hands-free system that permits duplex conversation to be made between the calling and called parties without using their hands to lift their station handsets.

Feedback produced between the microphone and the speaker by the following reasons, however, makes it impossible for the system to be in actual use:

a) Both the speaker and microphone are housed in a small case, causing them to be located in close proximity to each other.
b) Speaker audio output level is high.
c) Microphone is very sensitive.

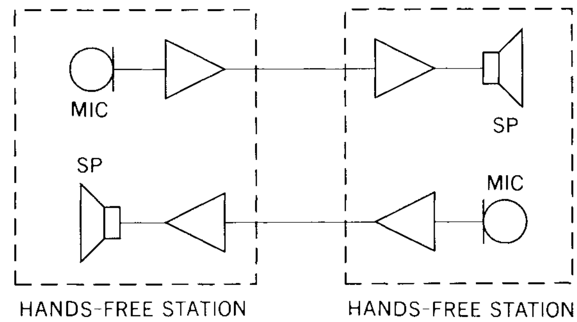


Fig.1 Hands-free System

1-2. Voice Switch

To make the hands-free half-duplex conversation possible by preventing occurrence of such feedback problem, Toa EXES-6000 system uses voice switch in its exchange that is able to detect signal strength information from both lines and to automatically select the speech line of higher signal level, while disconnecting another speech line of the lower signal level. See Fig. 2.

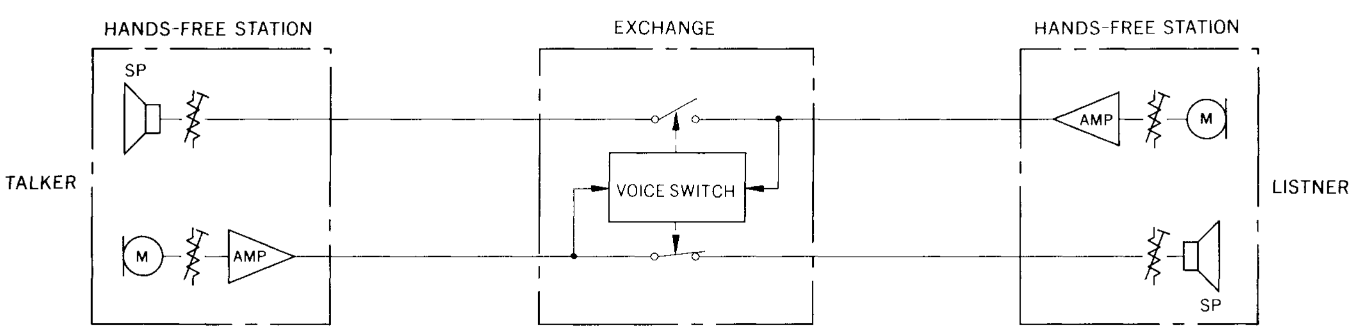


Fig.2 Principle of the Voice Switch

2. Concept of Speech Link

2-1. Manual Exchange

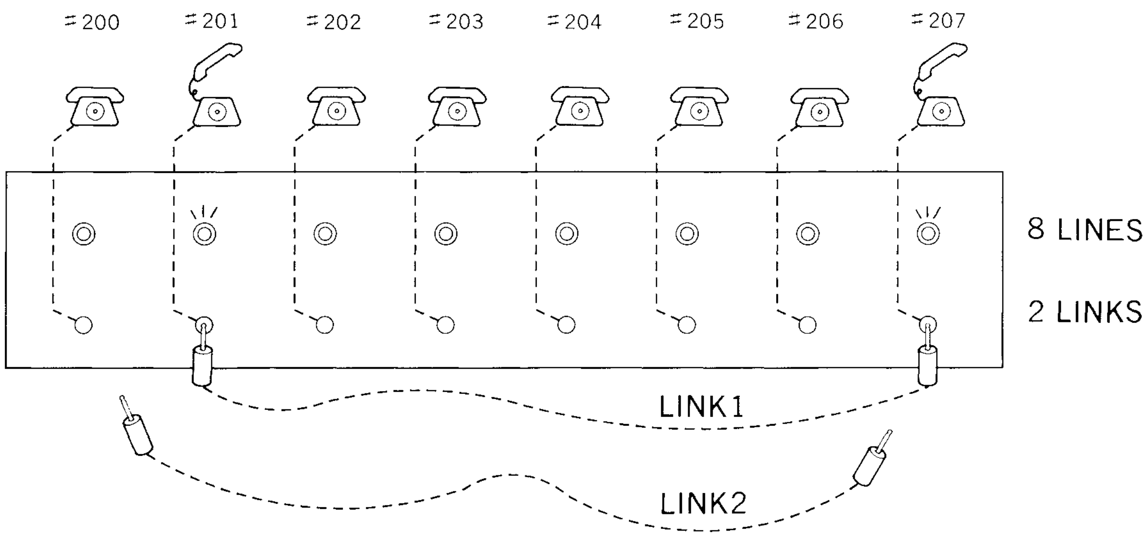


Fig.3 Manually Operated Exchange (8 lines/2 links)

Fig. 3 shows the simplest telephone system. When an attendant connects between 2 stations with a strap, speech path can be set up between them. The strap represents a link, and the number of links can be increased as the strap increases in number. You may presume that the attendant functions as switch in this system.

2-2. Crossbar System

Fig. 4 shows the crossbar method which corresponds to space division method in the classification of speech path. The space division method is mentioned in the next section 2-3. This method, however, has both the advantage and the disadvantage; the advantage is that all stations in a system can participate in the conversation at the same time, while the disadvantage is that an increase in the number of stations will require greater number of switches as is evidenced by an example of the EXES-6000 system with 128 stations where the required number of switches totals

$128^2 - 128 = 16{,}256$ pieces.

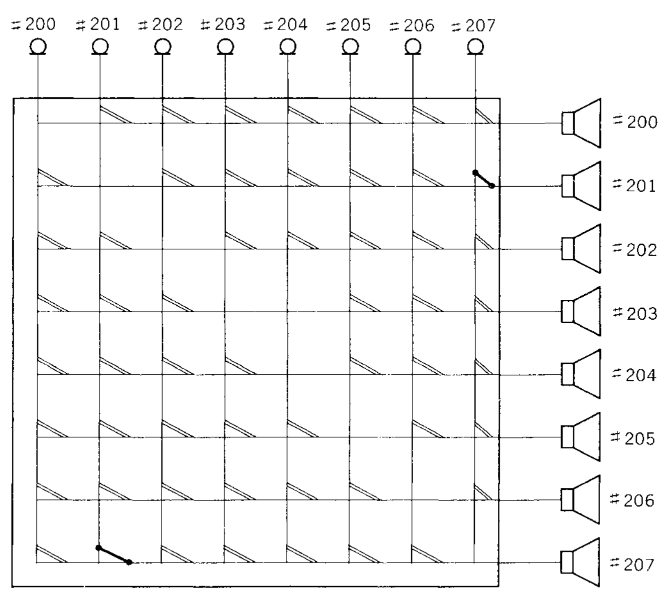


Fig.4 Crossbar System Exchange (8 lines/full links)

2-3. Space Division and Time Division Methods

a. Space Division Method

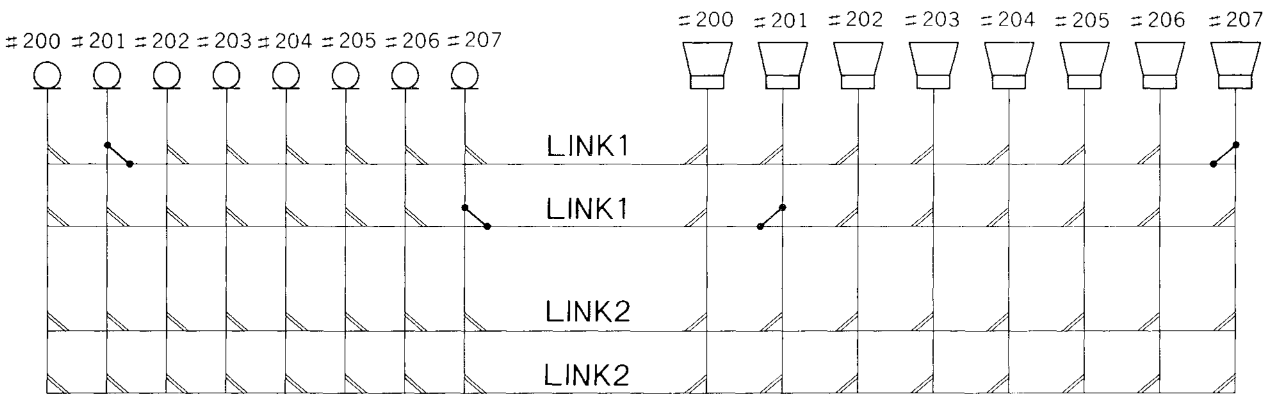


Fig.5 Space Division Method (8 lines/2 links)

The method employed in Toa EX-16 system. With this method, the system can do with the relatively small number of switches if it is the small system. But if the system consists of many stations, more switches are required than with the time division method referred to in the next section. Supposing the EXES-6000 system (128 stations/16 links) was designed with this method, it would require (2 x 128) x (16 x 2) = 8192 switches in total.

b. Time Division Method

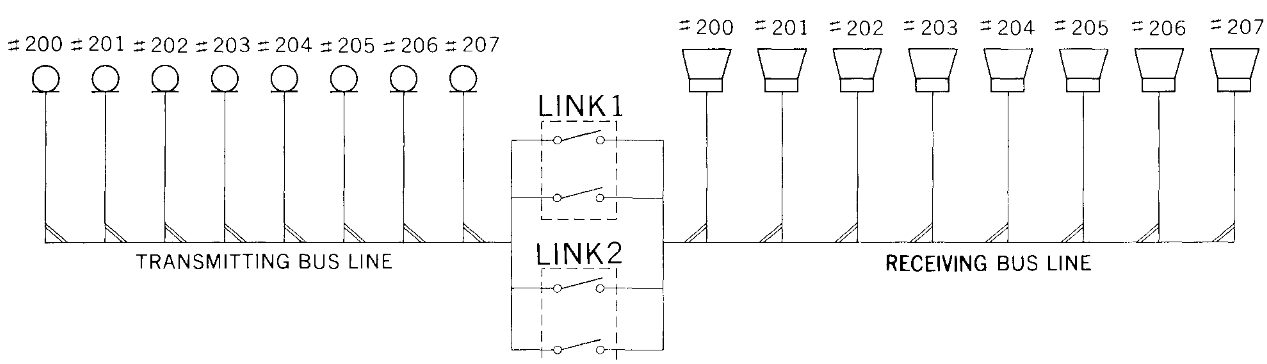


Fig.6 Time Division Method (8 lines/2 links)

The method adopted in the EXES-6000 system. All stations are connected by one of each of transmitting and receiving bus lines, we call "highway", and the speech signals are multiplexed on the bus lines by means of pulses of different timing. If we compare this method with the space division method by taking the same example of the EXES-6000 system (128 stations/ 16 links), the required number of switches is 128 x 2 + 16 x 4 = 320 pieces (this is the actually required number and not in agreement with a calculation made from the above Figure. For details refer to later explanations.), which is far smaller than that required with the space division method.

3. Principle of Time Division

3-1. PAM (Pulse Amplitude Modulation) Signal

In order to convert an analog signal such as speech into a pulse stream, a circuit must sample it at periodic intervals. The amplitude of the pulses sampled is proportional to the amplitude of the original signal at the sampling instant. This process is called Pulse Amplitude Modulation or simply PAM.

3-2. PAM Signal Generating Process

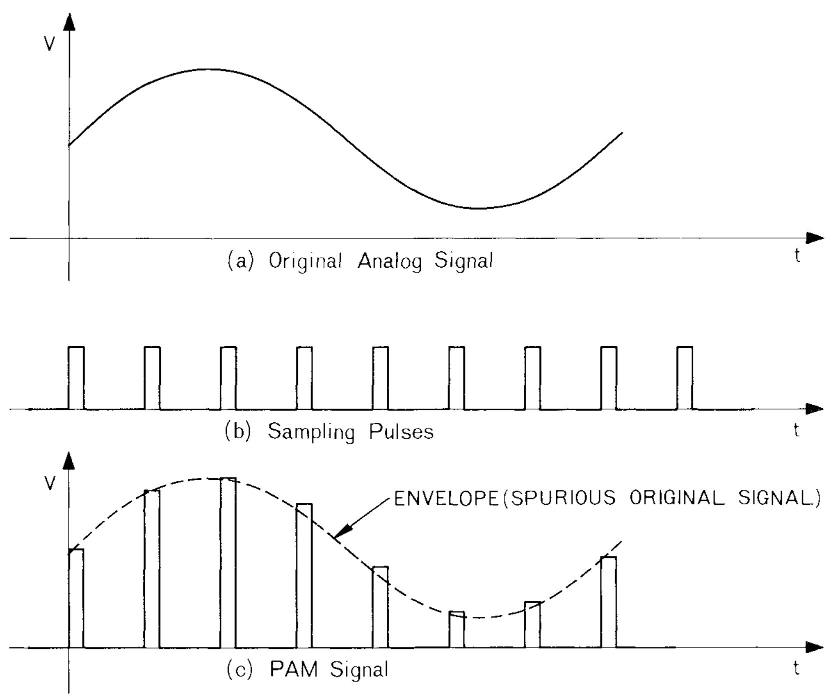


Fig.7 Waveforms for PAM Signal

After being modulated by the sampling pulses having periodic intervals (Fig. 7-(b)), the analog signal (Fig. 7-(a)) is converted into such PAM signal as is shown in Fig. 7-(c). However, the real pulse intervals are so short that the "spurious" original analog signal in Fig. 7-(c) and the original signal in Fig. 7-(a) are almost identical.

Fig. 8 shows the PAM signal generating process. In its block diagram, the analog signal (Fig. 7-(a)) applied to the input is chopped by the ON and OFF operation of the switch. The switching is controlled by the sampling pulses with their high level causing the switch on and their low level the switch off. Through this process, the PAM signal (Fig. 7-(c)) is delivered to the output.

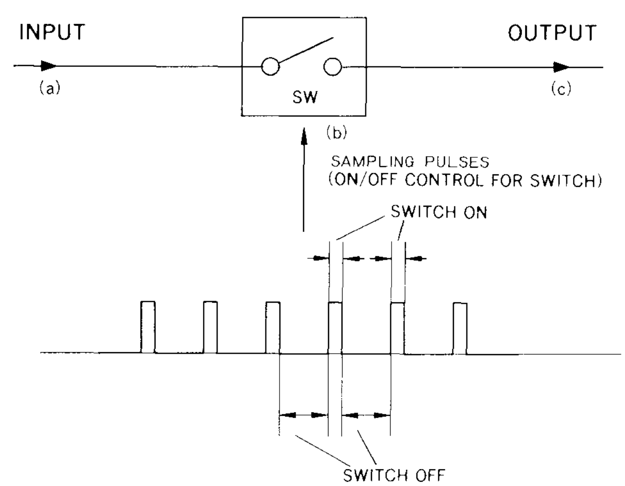
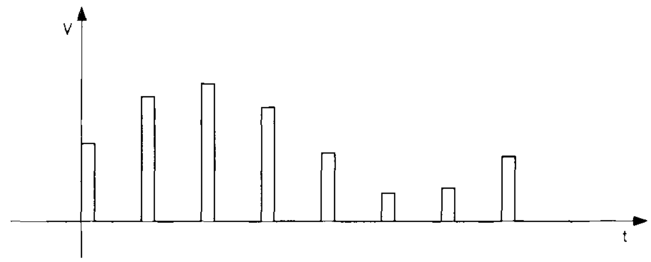


Fig.8 PAM Signal Generating Process

3-3. Demodulation of PAM Signal

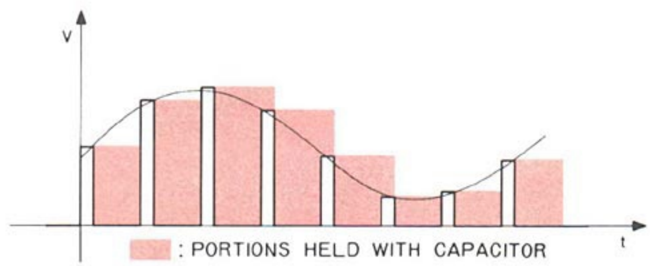
Both holding capacitor and lowpass filter are necessary to demodulate the PAM signal.

(1) PAM signal potential of Fig. 9-(1) at each sampling instant is held with the holding capacitor.



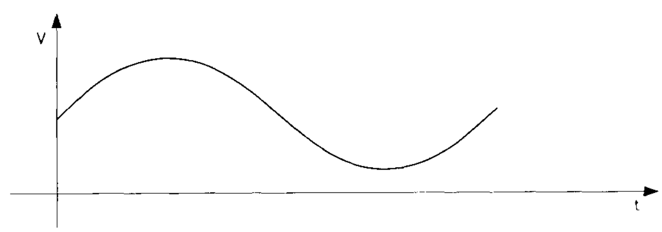
(1) PAM Signal

(2) Sampling pulse elements contained in the waveforms of Fig. 9-(2) are eliminated through the lowpass filter.



(2) Waveform When PAM Signal Potential Is Held by Holding Capacitor

(3) Through processes (1) and (2), the PAM signal is demodulated into such a waveform as is shown in Fig. 9-(3) which is identical with the original analog signal.



(3) Demodulated Analog Signal

Fig.9 Demodulation of PAM Signal

3-4. Sampling Pulse

If a band-limited signal is sampled by pulses having regular intervals of time and a frequency equal to or higher than twice the highest significant frequency, then the sample contains all the information of the original signal.

4. Principle of Time Division Multiplex

4-1. Time Division Multiplex

The time division multiplex is a method to divide various signals by time and to place such time divided signals on a single common line called highway.

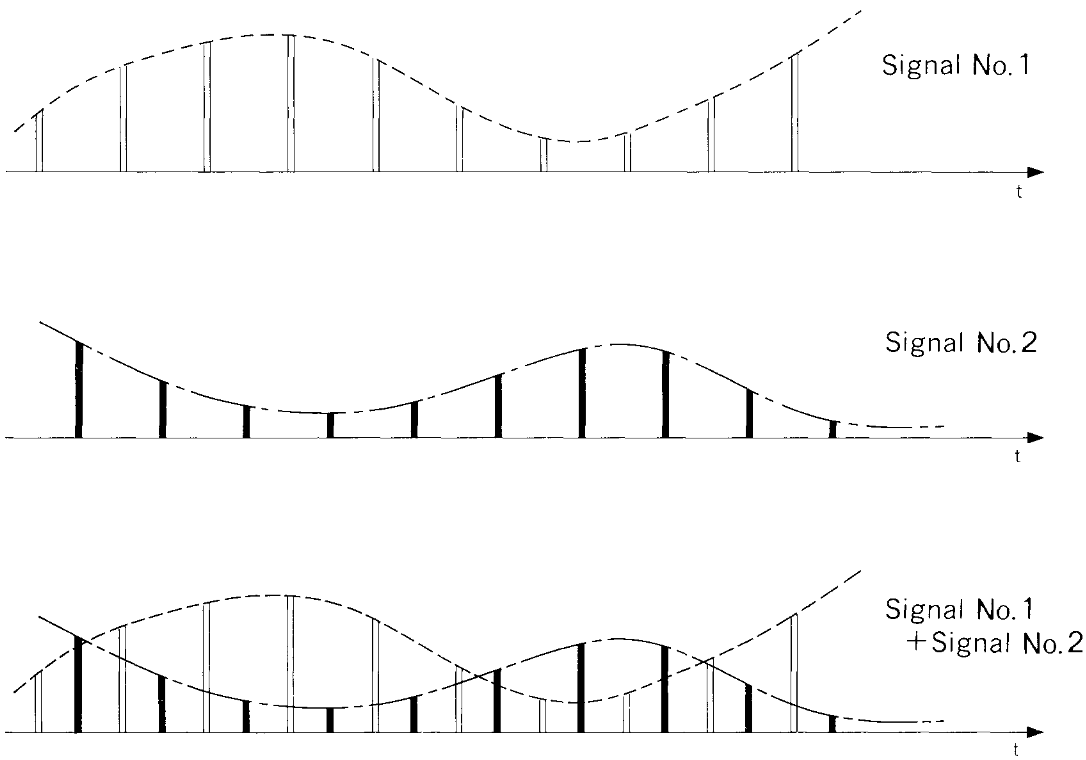


Fig. 10 Time Division Multiplex

4-2. Access to Time Division Multiplex

In Fig. 11-(a) at the right hand side, consider just how many lines are needed to transmit 4 different signals A, B, C and D to the outputs 1, 2, 3 and 4, respectively.

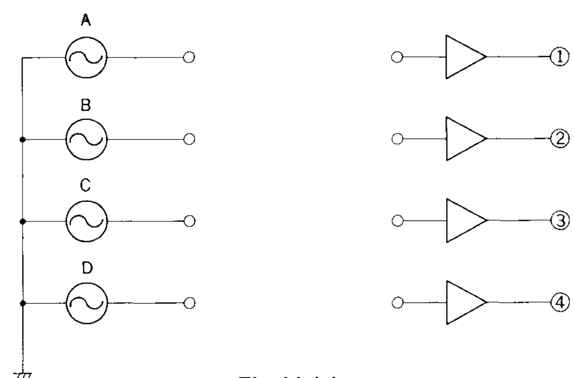


Fig.11-(a)

(1) The simplest way is to connect between them as indicated by dotted lines in Fig. 11-(b) at the right hand side. This method, however, involves such problem as that the more different signals are used, the more grows the number of lines.

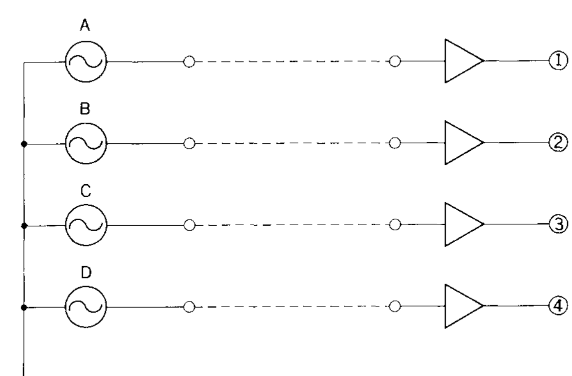


Fig.11-(b)

(2) Another connection example like one in Fig. 11-(c) can be considered when you wish to develop the previous connection in order to reduce the number of lines. The idea of this connection is that switches 1 and 2 are synchronized with each other, through which signals A to D are transmitted in this order to each corresponding output. In this event, the signals delivered to the outputs 1 through 4 are the RAM signal described in section 3-2.

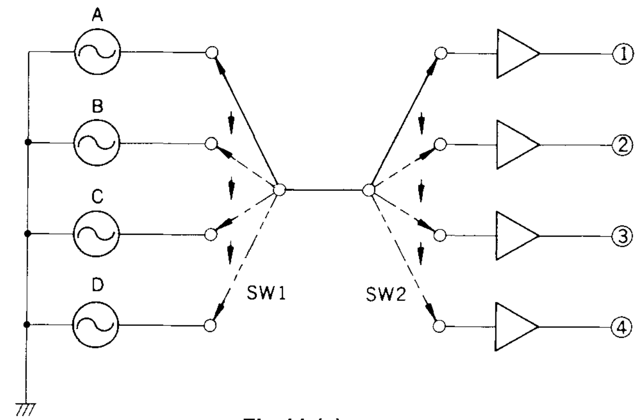


Fig.11-(c)

(3) In the example of Fig. 11-(c), one switch has many contact points, but an actual system uses an analog switch as shown in Fig. 11-(d), instead of each contact point so that an individual pair of analog switches (x and y) is sequenced in the following order:
When the analog switches x_1 and y_1 synchronized with each other are on, the signal A is allowed to go through to the output 1, and likewise when x_2 and y_2 are on, the signal B is delivered to its corresponding output 2. This continues till the signal D is sent out to its output 4, and is again repeated from the beginning.

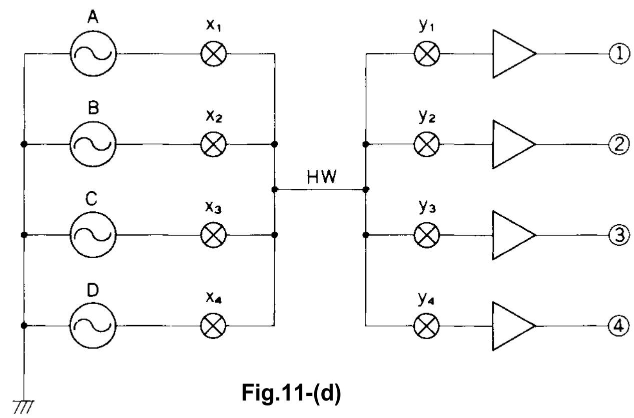


Fig.11-(d)

Here we show the timing of each switch used in the system of Fig. 11-(d).

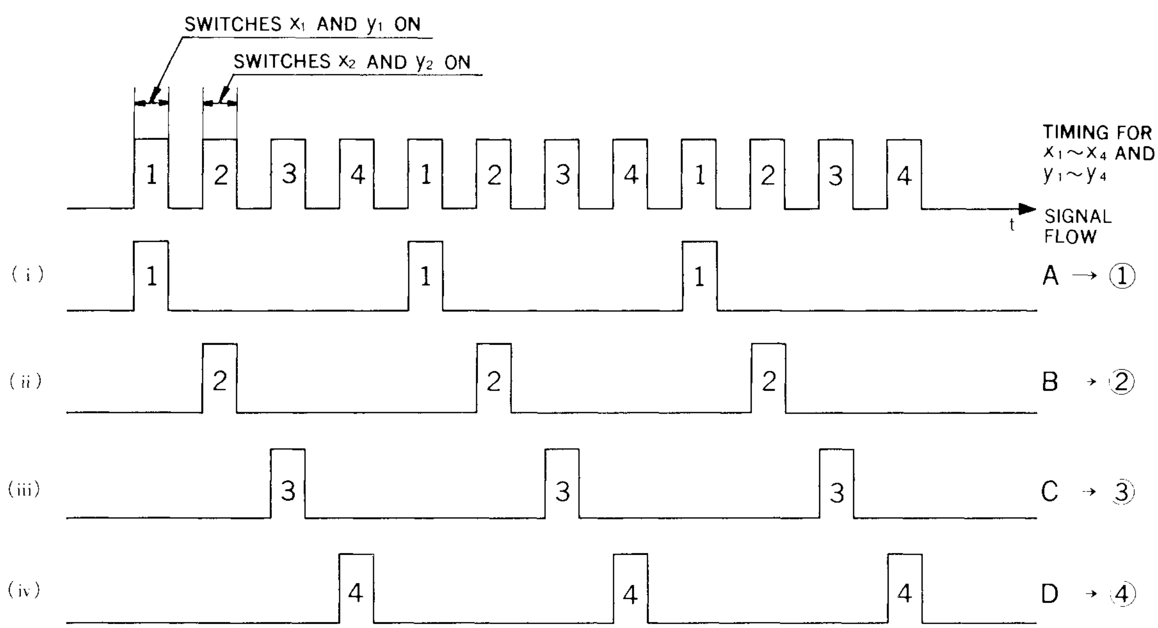


Fig.12 Timing Chart

In the above timing chart, 4 different PAM signals of Fig. 11-(d) are on a highway (H W line), the state of which is called "being time division multiplexed". In Fig. 11-(d), the signals are quadplexed.

Also, the frequencies (time intervals) of sampled pulses (i) through (iv) are all the same, which are the sampling frequency described in section 3-4.

4-3. Time Division Exchange

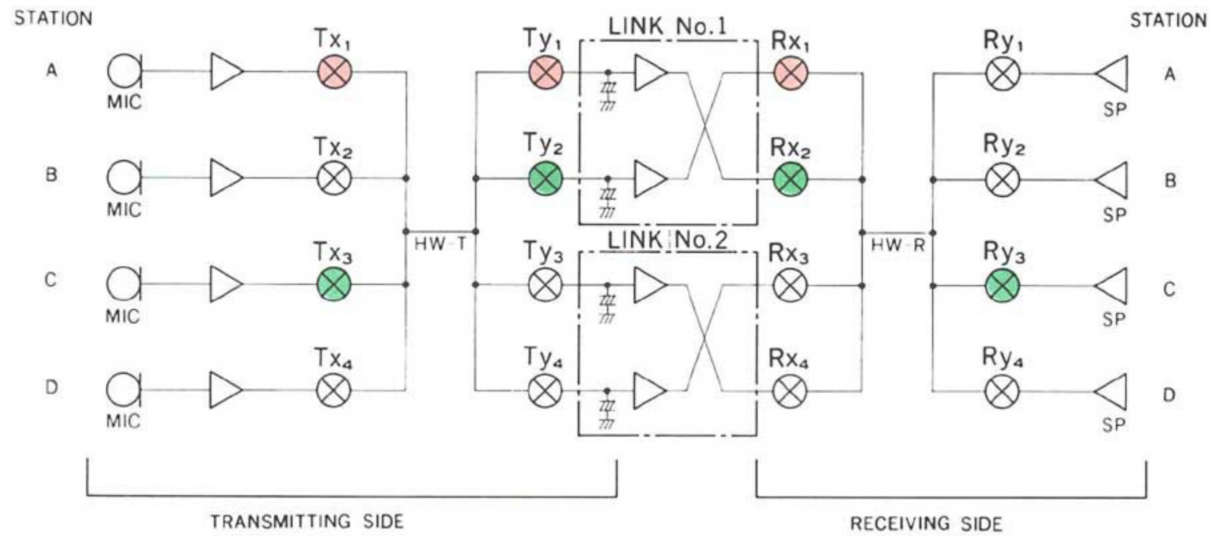


Fig.13 Time Division Exchange Block Diagram (4 lines/2 links)

Shown below is the timing of each analog switch when a conversation is being made between stations A and C using LINK No.1 in the above Fig. 13. (Voice signal flow: Mic A → Speaker C, Mic C → Speaker A)

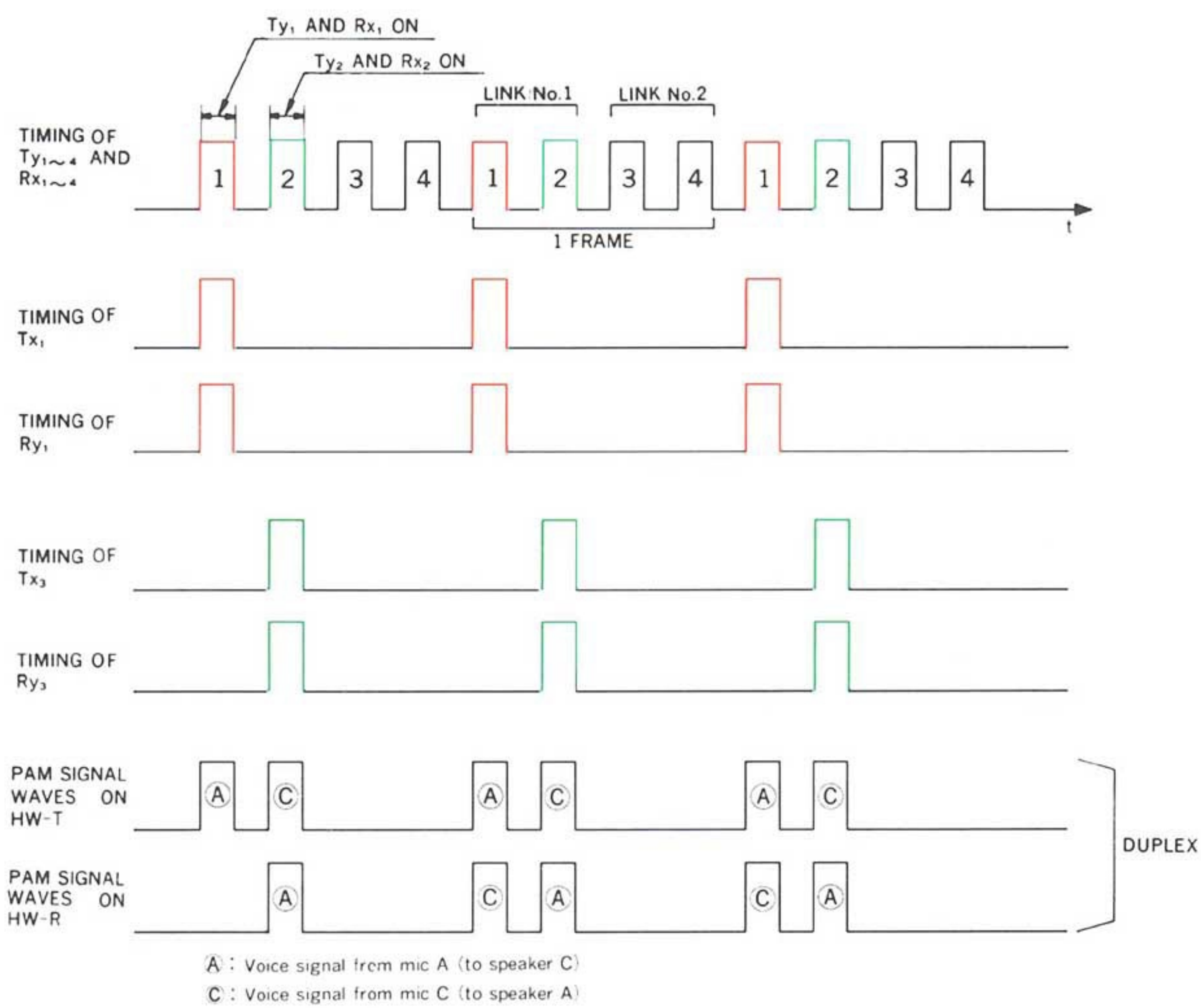


Fig.14 Timing Diagram

Here we show the voice signal flow of Fig. 13 where stations A and B are conversing with each other. For timing of each switch, refer to Fig. 14.

— Timing "1" —

(1) Switches Tx_1 and Ry_1 close at the timing "1" among the timings of Ty_{1-4} and Rx_{1-4}. (Marked in red in the Figs. 13 and 14)

(2) The voice signal from the microphone of station A is transmitted to LINK No. 1 via Tx_1, HW-T line and Ty_1. The signal between Tx_1 and Ty_1 is the PAM signal. Now that Rx_2 is left open at this timing, the signal delivered to LINK No. 1 will not go beyond this LINK where the signal is held with a holding capacitor in order to be demodulated into the original voice signal. See section 3-3 "Demodulation of PAM Signal".

— Timing "2" —

(3) Switches Tx_3 and Ry_3 close at this timing. (Marked in green in the Figs. 13 and 14) The signal from station A held at LINK No. 1 is transmitted to the speaker of station C via Rx_2, HW-R and Ry_3. The signal between Rx_2 and Ry_3 is the PAM signal. The signal from the microphone of station C is, in the same manner as described in (1) and (2) above, delivered to LINK No. 1 and held there.

— Second Timing "1" —

(4) The signal from station C held at LINK No. 1 is sent out to the speaker of station A in the same manner as in (3) above.

- Since the timings "3" and "4" have no relations with LINK No. 1, they are omitted from this example.

II. VOICE SIGNAL AND DIAL SIGNAL FLOWS

1. Voice Signal Flow

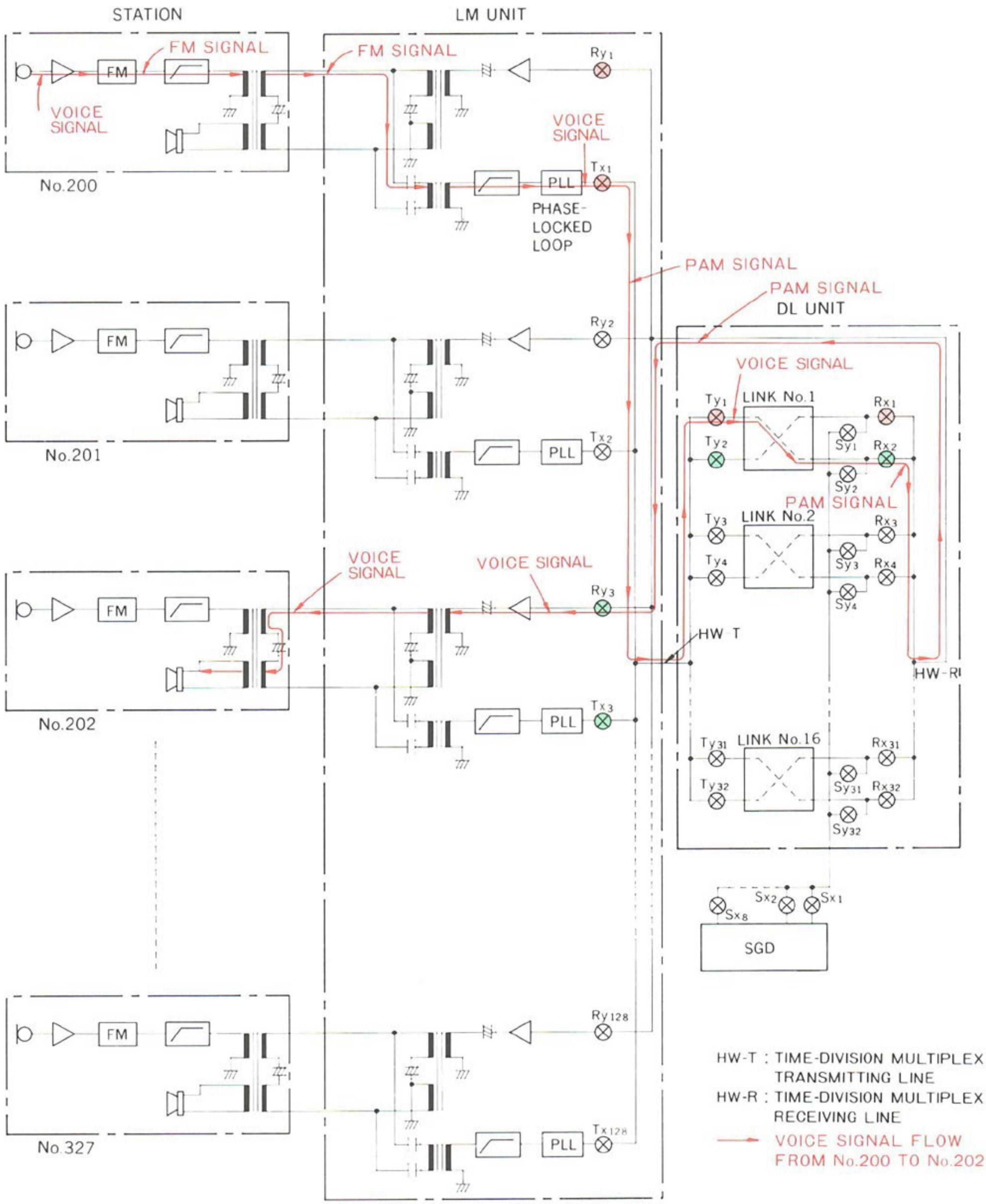


Fig. 15 Voice Signal Flow

Assume that stations No. 200 and No. 202 are in the conversation mode through link No. 1.

a) In this event, voice signal flows from station No. 200 to station No. 202 as follows:

The voice signal is transmitted to the LM unit after being frequency-modulated in station No. 200 ..... demodulated into the original analog signal in the PLL circuit (analog switches Tx_1, Ty_1, Rx_1 and Ry_1 close at the same timing) ..... passes through Tx_1 ..... enters the DL unit through highway HW-T (in link No. 1, cross-connection is made between Ty_1 and Rx_2 and between Ty_2 and Rx_1) ..... because Rx_2 stays open, the voice signal does not pass through highway HW-R and is held in the link by a holding capacitor ..... At the next timing Tx_1, Ty_1, Rx_1, and Ry_1 open, and Tx_3, Ty_2, Rx_2, Ry_3 close ..... the voice signal held in the link passes through Rx_2 ..... enters the LM unit through HW-R ..... passes through Ry_3, the only switch that remains closed among ones connected to HW-R ..... sent out to station No. 202 connected in base frequency band after being amplified in the LM unit ..... can be heard from the speaker of station No. 202

b) The timing chart below shows actual switch timings and PAM signal wave forms on HW-T and HW-R in EXES-6000 systems when the simultaneous conversation is being made between stations No. 200 and No. 202.

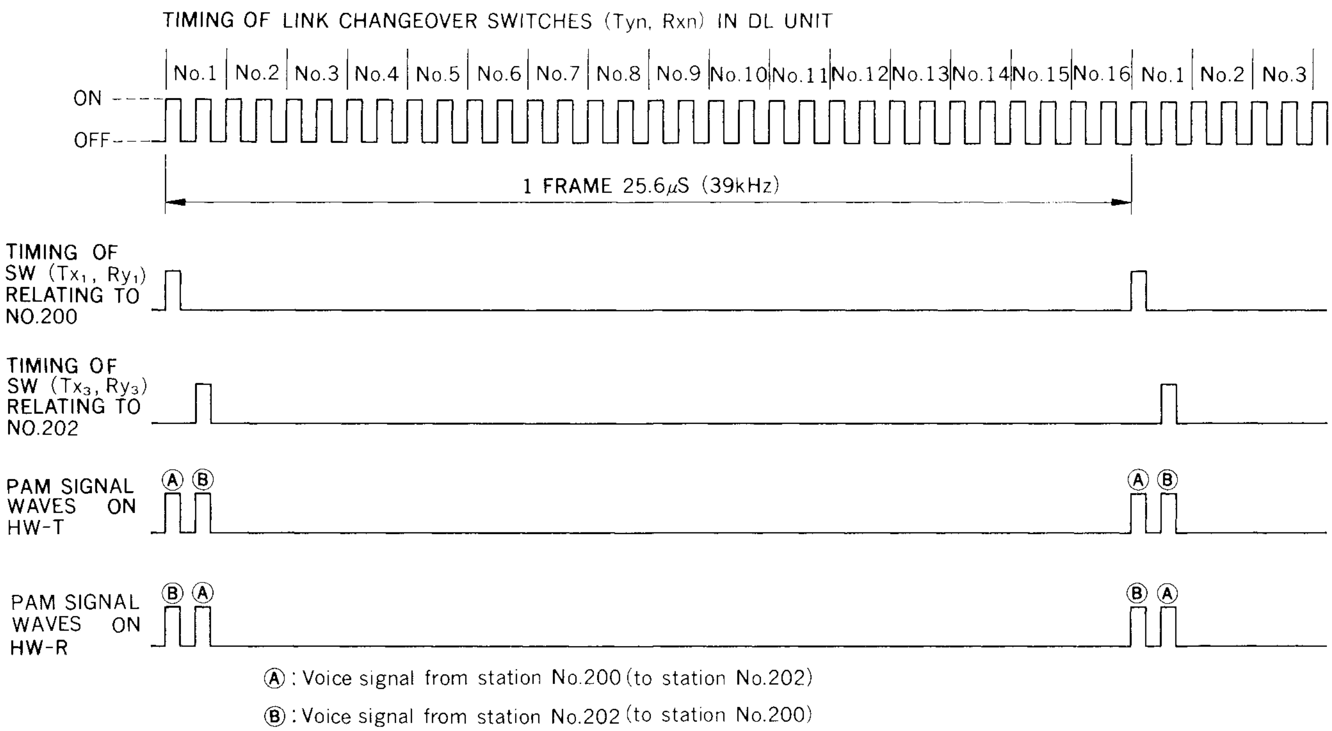


Fig. 16 Timing Chart in Conversation Between Stations No.200 and No.202

2. Dial Signal Flow

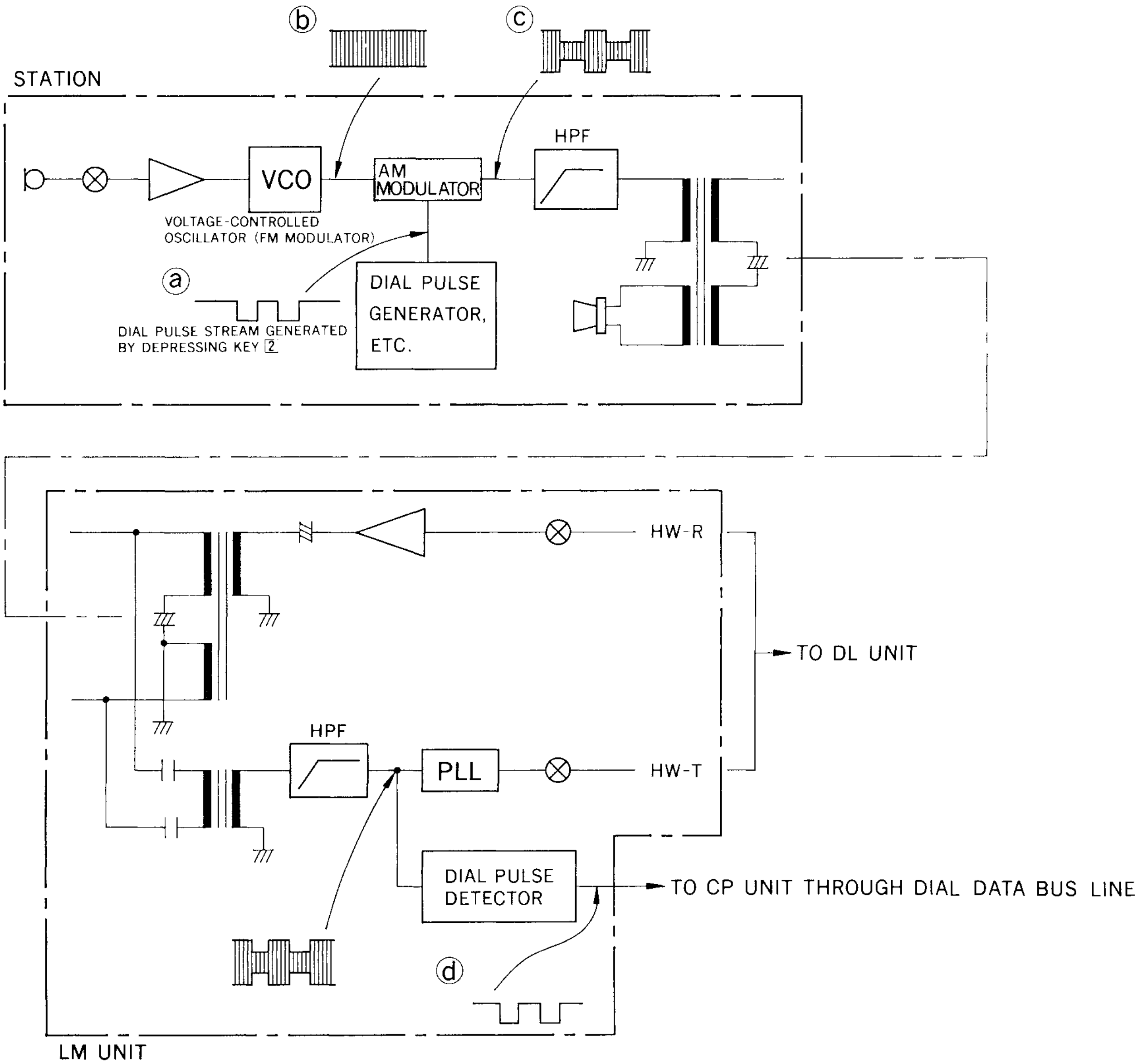


Fig. 17 Dial Signal Flow

Station dialing operations cause the pulse stream ⓐ corresponding to each key depressed to be generated in the dial pulse generator, which amplitude-modulates the FM carrier ⓑ. The amplitude-modulated signal ⓒ is sent out to the LM unit of the exchange.

The dial streams are demodulated into the original pulses with a dial pulse detector in the LM unit, and then transmitted to the CP unit through the dial data bus line.

3. Dial Signal Receiving System

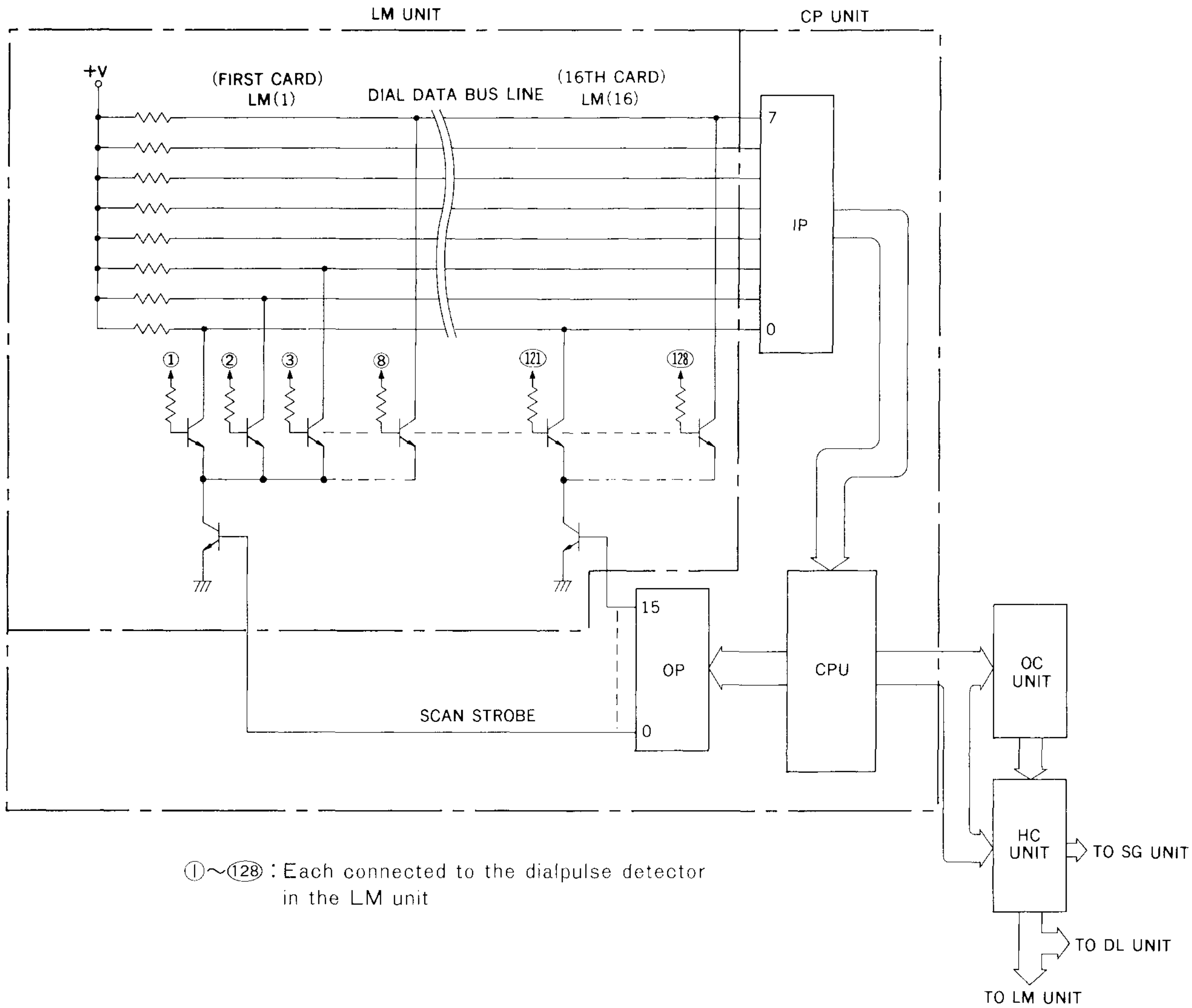
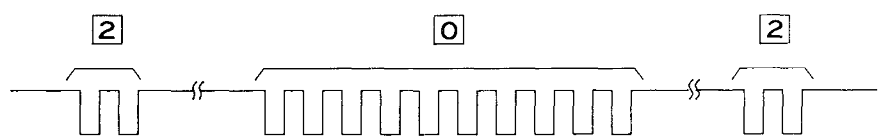


Fig. 18 Dial Signal Receiving System

Let's assume that 202 (station number) is dialed at station No. 200. The dial pulse detector located in the LM unit and intended for station No. 200, as explained just before, delivers the following dial pulses to the output.



These pulse streams comprising "2", "10" and "2" pulses enter ① in the above Figure, and are inverted by a transistor. They then pass through the dial data bus line, and go into Input Port (IP) No. 0 of the CP unit. The CP unit is cyclically sending from its Output Port (OP) the scan strobe signals to each of the 16 LM units in a serial manner (LM1 through LM16, again LM1 through LM16 ...... this is repeated), and reads dial data of 8 stations contained in each LM unit in a fraction of second simultaneously with the transmission of the scan strobe signals.

The serial pulse streams that went into IP No. 0 are transmitted to both the OC unit and HC unit after being converted into parallel codes. In the HC unit, the data from the CP unit is written into both the line memory and signal memory according to the address specified by the OC unit. Station numbers of the stations in converstations and the status of each link are written in the line memory, of which contents become the control signals for the analog switches of the LM unit.

Signal codes, PIT or non-PTT mode and the status of each link are written in the signal memory, of which contents become the control signals for the analog switches of the SG unit.

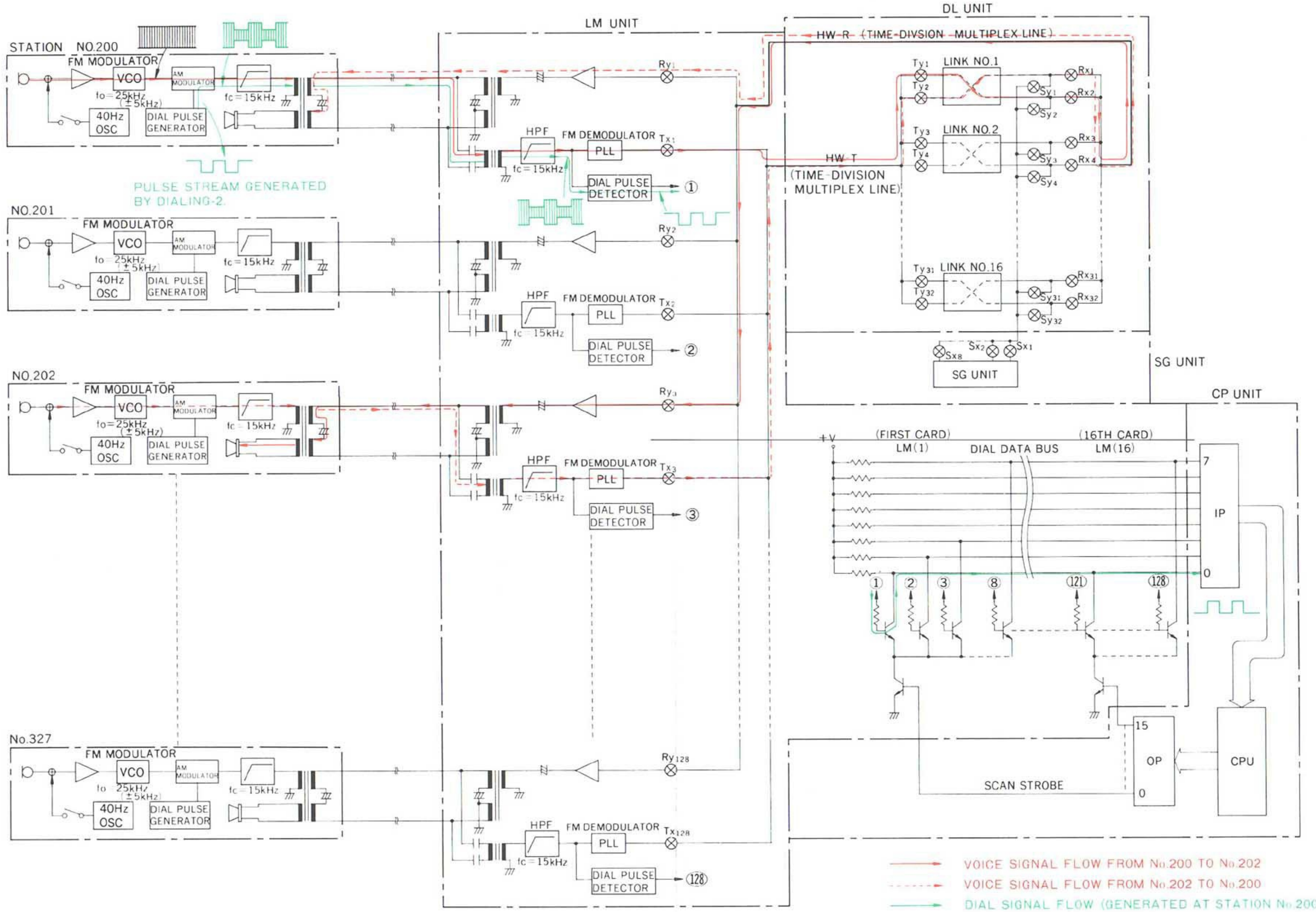


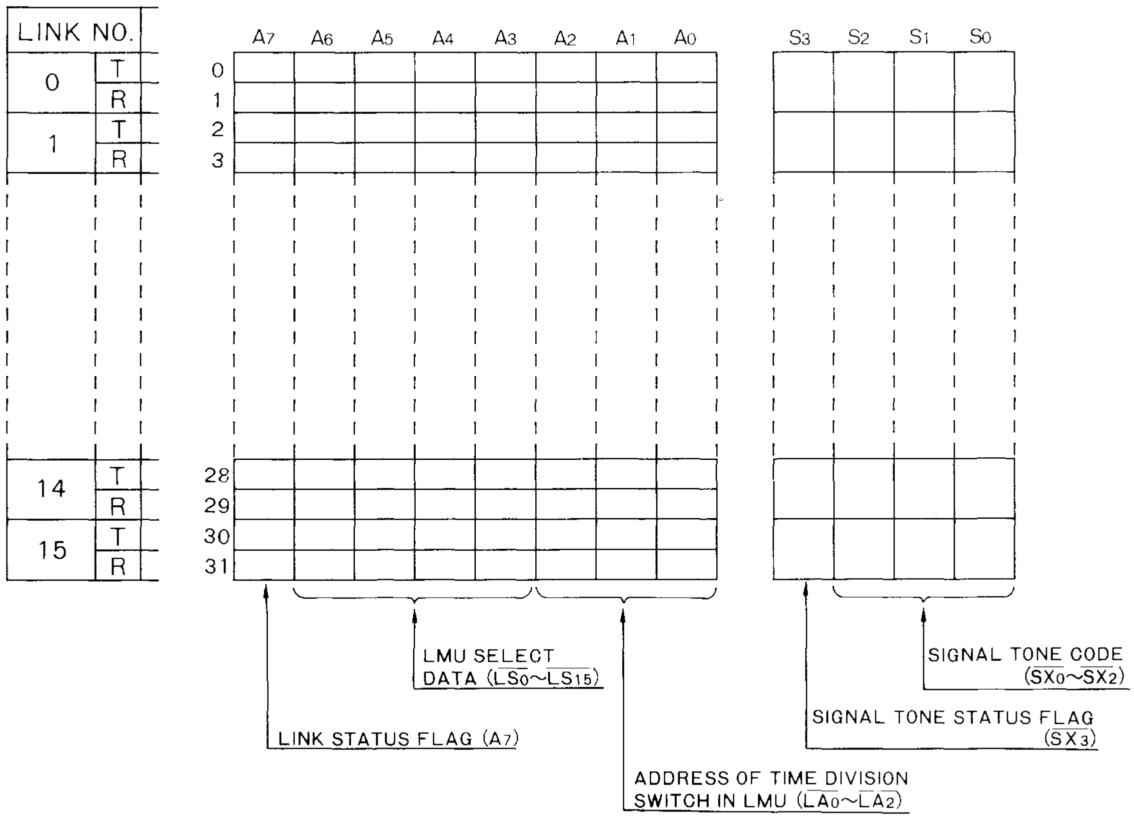
Fig. 19 Voice and Dial Signal Flows

4. Line Memory and Signal Memory

Line memory is a 32 words by 8 bits memory formed from 4 RAM's of 16 words by 4 bits. In this line memory, station numbers of the stations engaged in coversations are written into the addresses that correspond to the link to be used in the order of a calling side (T) and called side (R). The data is accessed and read cyclically word byword in synchronization with the link change-over switches (Ty_1-Ty_{32}, Rx_1-Rx_{32}) in the DL unit and controls the corresponding time division switch in the LM unit according to the contents of the data.

Signal memory is a 16 words by 4 bits RAM, in which signal codes or PTT/non-PTT mode each link uses are written. Signal switches (Sy_1-Sy_{32}) of the DL unit's links and time division switches (Sx_1-Sx_8) in the SG unit are controlled with the data read out in the same manner as that of the line memory.

Memory maps of the line memory and signal memory are as follows:



Contents of the line memory when stations No. 200 and No. 202 are engaged in a conversation using link No. 1.

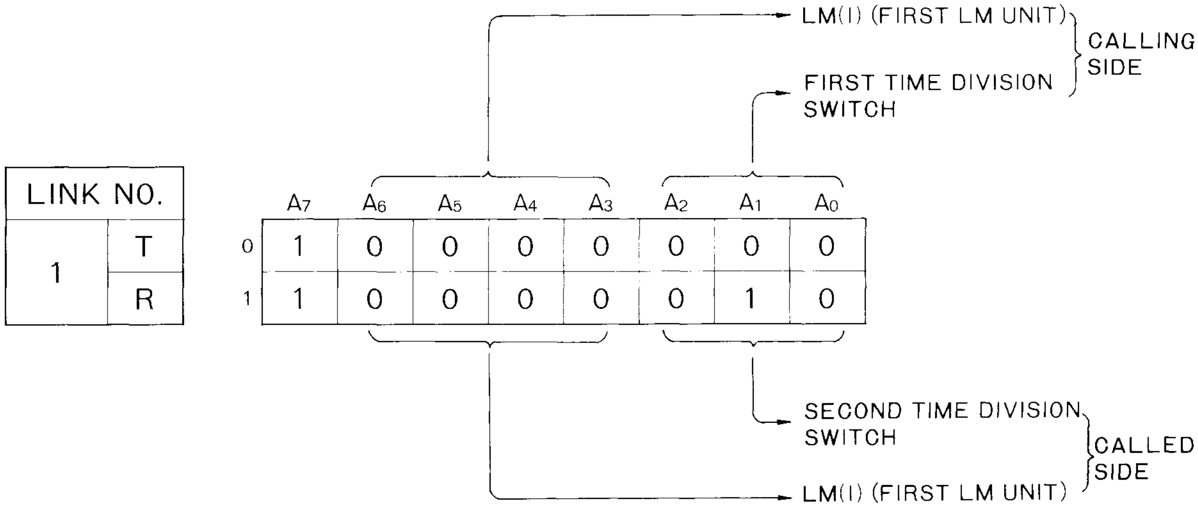


Table of Service Signal Tones

		S_3	S_2	S_1	S_0
Signal Tone Transmission Mode	Calling tone	1	0	0	0
	Privacy tone	1	0	0	1
	Busy tone	1	0	1	0
	Dial tone	1	0	1	1
	Zone paging preannouncement tone	1	1	0	0
	All-call paging preannouncement tone	1	1	0	1
	Priority	1	1	1	0
	Holding tone/ Confirmation tone	1	1	1	1
PTT Mode	T → R	0	0	0	1
	R → T	0	0	1	0
	Duplex	0	0	0	0

III. VOICE SWITCH CIRCUIT

It is already mentioned voice switch is necessary for the hands-free conversation. Here, we explain the principle of the voice switch circuit used in the EXES-6000 system. The voice switch circuit mentioned here is an improved version of one that is used in the EXES-5000 system.

The voice switch is employed in the DL unit and its operations differ depending on the conversation mode (hands-free or handset) between the stations.

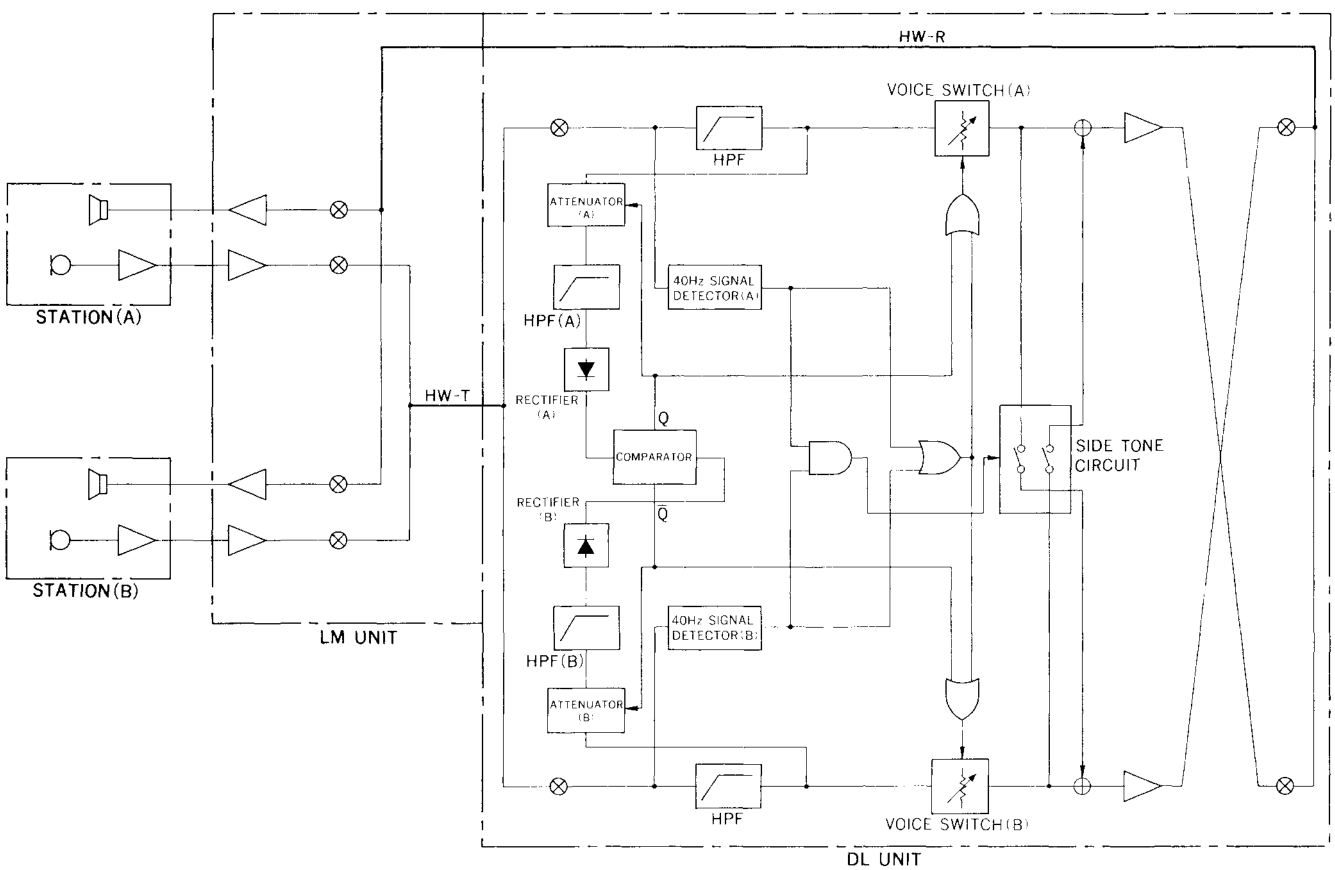


Fig. 20 Block Diagram of Voice Switch and Peripheral Circuit

1. Operation Principle of the Voice Switch Circuit

1-1. Hands-free to Hands-free Mode

Given below are operation procedures of the voice switch when stations A and B are talking to each other in the hands-free mode and sound pressure level at a mic of station A is higher than that at station B. See Fig. 22

(1) A signal flow from mic A:
ATT(A)→HPF(A)→RECTIFIER(A)→COMPARATOR
Similarly, a signal from mic B flows as follows:
ATT(B)→HPF(B)→RECTIFIER(B)→COMPARATOR

(2) A comparator compares signal strength from stations A and B.

(3) Since a signal level from station A is higher than that from station B, switch SW(A) closes and SW(B) opens, thus permitting transmission of the signal from mic A (signal A) to speaker B.
At the same time, ATT(B) is activated and the signal from mic B attenuated by ATT(B) is transmitted to the comparator. In this event, ATT(A) is not activated. The signal A, therefore, is sent to the comparator without being attenuated.

(4) Therefore, the signal A that is fed from speaker B back to its mic is sent to the comparator after being attenuated in ATT(B), allowing not only signal A to be still kept higher at the comparator but also the status of (3) above to be maintained.

* The EXES-6000 system uses FET's for the switches SW(A) and (B). This makes transient response at switching longer than that of the EXES-5000.

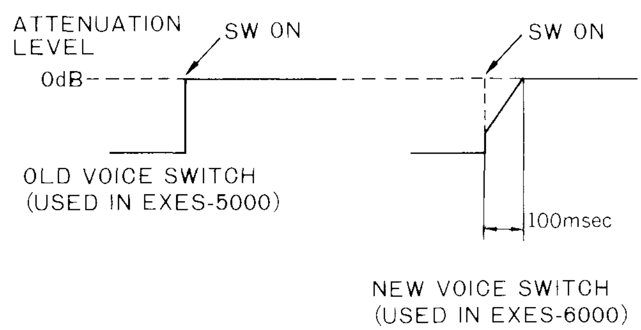


Fig. 21 Voice Switch Transient Response

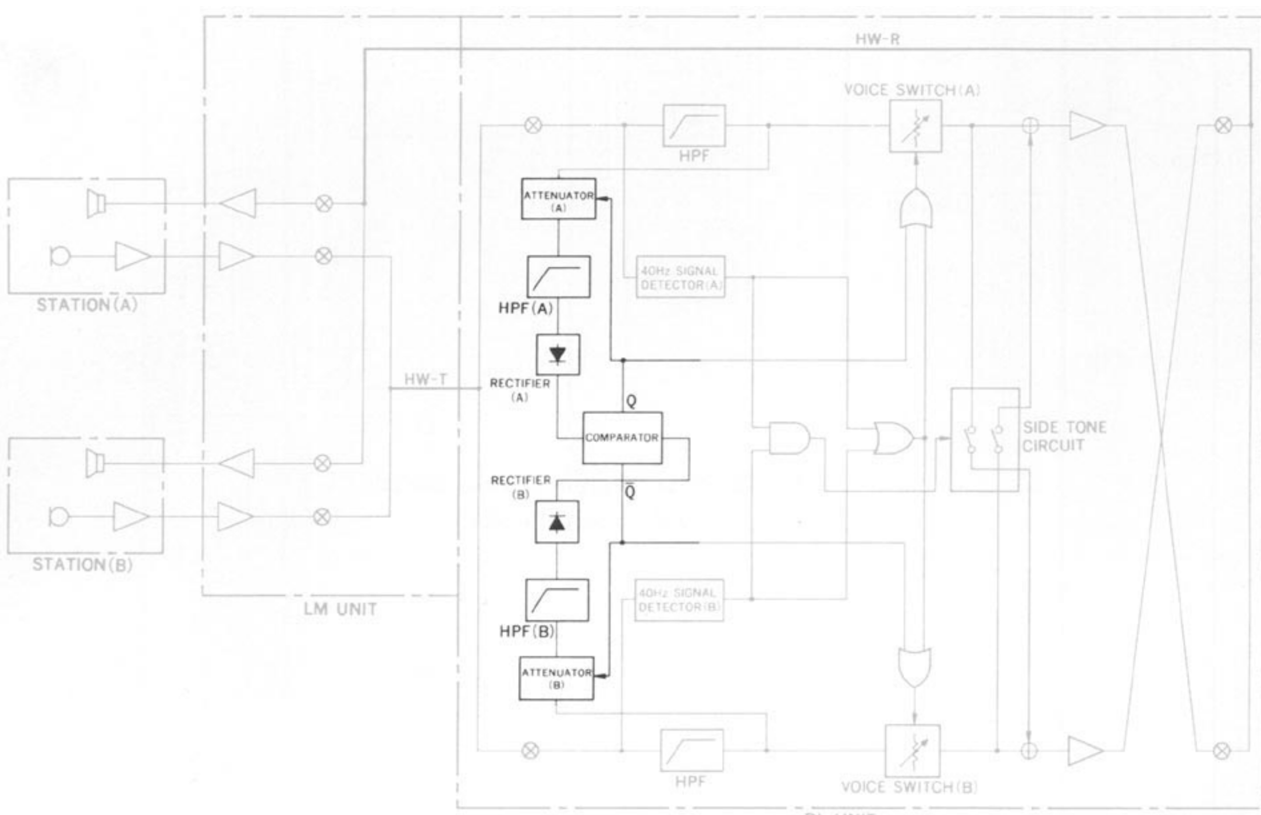


Fig. 22 Main Circuit Activated in Hands-Free to Hands-Free Mode

1-2. Handset to Hands-free Mode

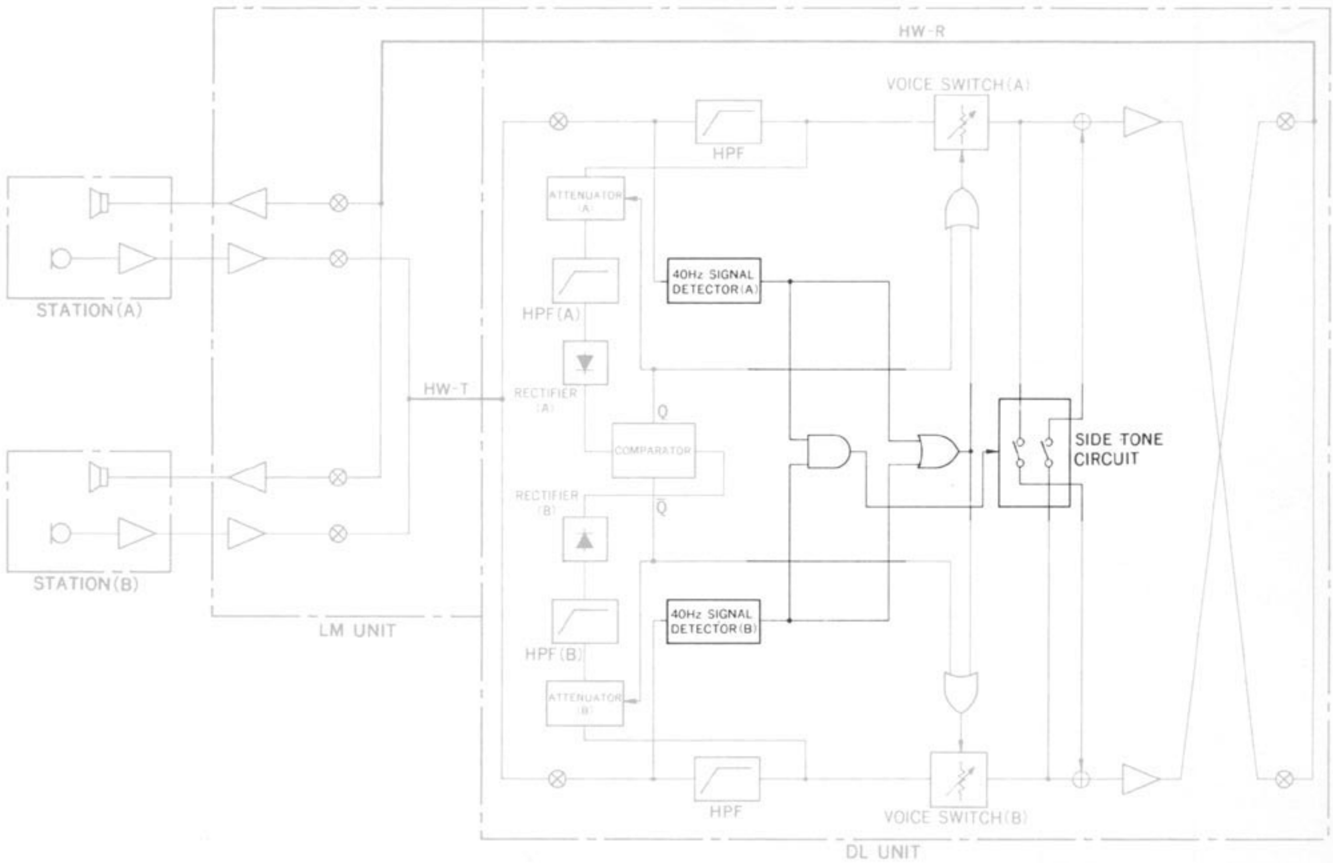


Fig. 23 Main Circuit Activated in Handset to Hands-Free or Handset to Handset Mode

Assume that in Fig. 23, station A in the handset mode is connected to station B in the hands-free mode.

(1) A 40Hz handset mode signal and voice signal from station A are transmitted to the DL unit via the LM unit and HW-T.

(2) Only the voice signal is sent out from station B, which is similarly transmitted to the DL unit.

(3) The 40Hz handset mode signal from station A preceeds to 40Hz SIGNAL DETECTOR (A), and its output level becomes high, while the voice signal is sent to VOICE SWITCH (A) through a high-pass filter (HPF).

(4) The voice signal from station B passes through HPF and proceeds to VOICE SWITCH (B). Since no 40Hz signal is sent out from station B, the output level of 40Hz SIGNAL DETECTOR (B) is kept low. The status of logic elements after this stage is shown in Fig. 24.

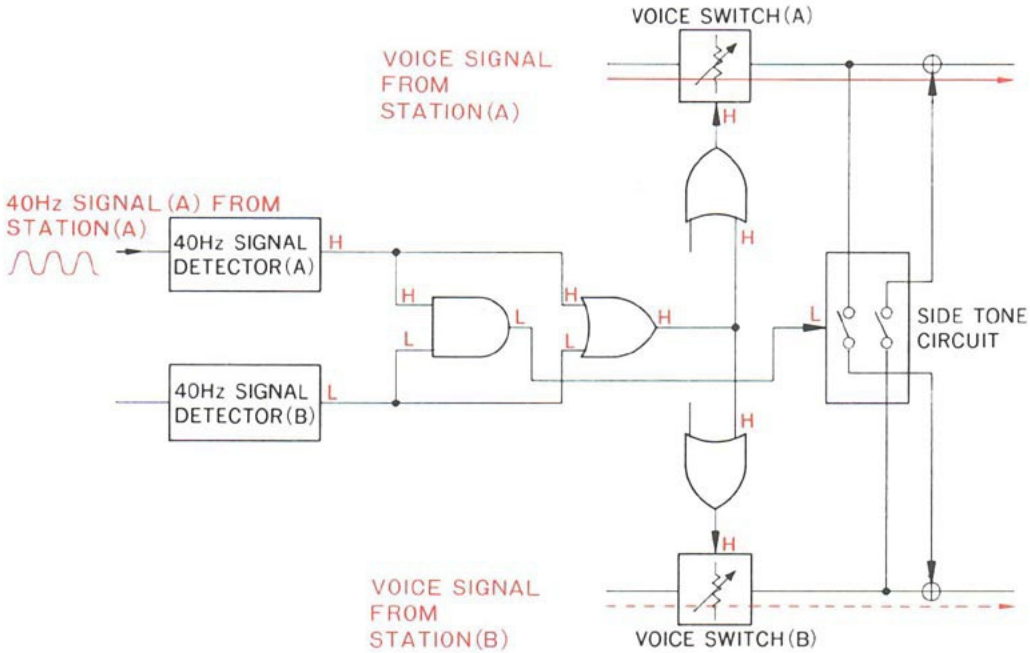


Fig. 24 Status of Logic and Voice Signal Flow in Handset to Hands-Free Mode

At last, both VOICE SWITCHES (A) and (B) close, providing full duplex conversation. In this event, SIDE TONE CIRCUIT is not activated.

The side tone circuit is a circuit that enables a talker using a station handset to hear his own voice through a handset ear speaker. In the case of the conversation mode used in this section, a voice spoken into the mic of station A comes out of the speaker of station B. But now that station B is in the hands-free mode, that voice from the speaker is fed back to its mic, which in turn can be heard through the handset ear speaker of station A, thus eliminating the necessity of activating the side tone circuit. See Fig. 25.

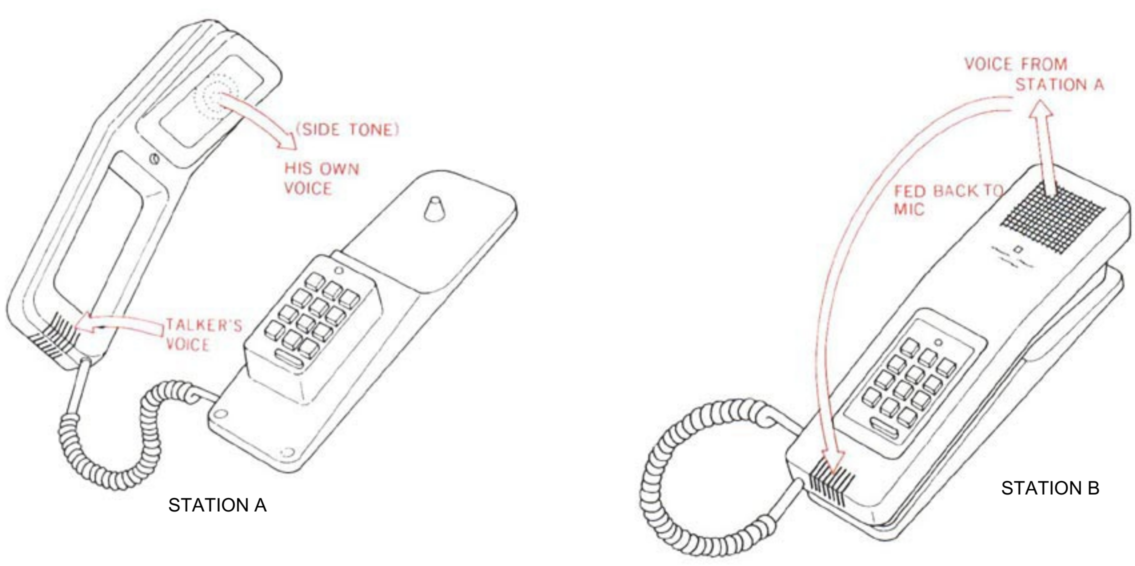


Fig. 25 Side Tone on Handset to Hands-Free Conversation

1-3. Handset to Handset Mode

Assume that in Fig. 23, both stations A and B are in the handset mode and connected.

(1) A 40Hz handset mode signal and voice signal are transmitted from the both stations to the DL unit via the LM unit and HW-T.

(2) The 40Hz handset mode signal (A) proceeds to 40Hz SIGNAL DETECTOR (A) and another 40Hz signal (B) to 40Hz SIGNAL DETECTOR (B), resulting in a high level output at the both detectors. Similarly, each of the both voice signals is transmitted to VOICE SWITCH (A) and (B), respectively via HPF. The status of logic elements after this stage are shown in Fig. 26.

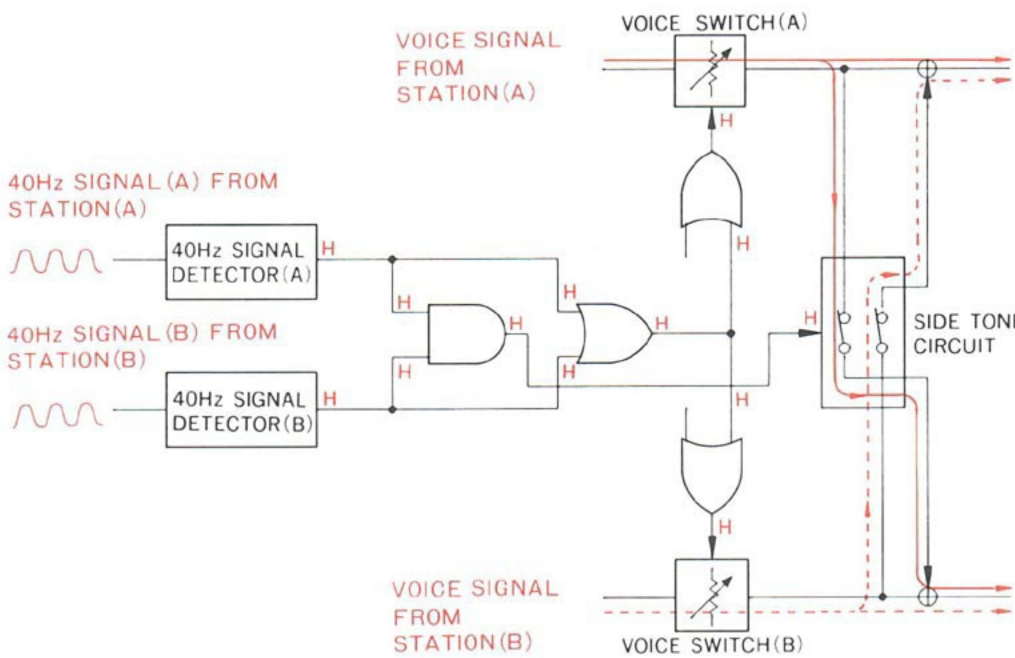


Fig. 26 Status of Logic and Voice Signal Flow in Handset to Handset Mode

Both VOICE SWITCHES (A) and (B) close, providing full duplex conversation. In this event, the side tone circuit is activated, and the both stations (A) and (B) enable its talker to hear his own voice from its handset ear speaker.

2. Performance Comparison between New (Used in the EXES-6000) and Old (used in the EXES-5000) Voice Switches

2-1. Voice Switch Features of Duplex Link Unit (DLU-52, DLU-11A, CPU-16 etc.) for the EXES-5000, the EXES-1000 and the EX-16

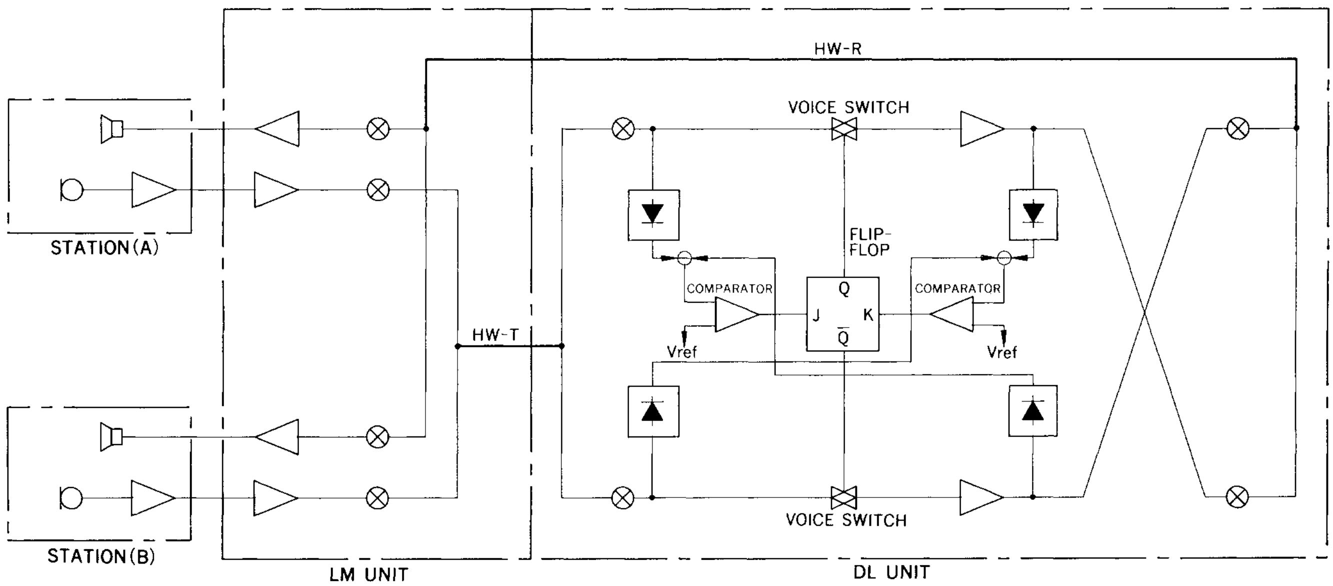


Fig. 27 Old Voice Switch Circuit

- Quick voice switch response permits easy interruption in conversation, resulting in high interruption performance in low noise area, while in high noise condition, the switching action is unstable.
- Usable under up to 60-phon noise level
- Subject to pulsive noise (typewriter, etc.), and unstable operation in conversation.
- Acoustic feedback by echo or reverbration may cause the circuit to malfunction.

2-2. Voice Switch Features of Duplex Link Unit (DL-62/62A) for the EXES-6000

- Improved voice signal detection circuit assures high performance against pulsive noise and malfunction due to acoustic feedback.
- Sensitivity of the voice signal detection circuit is controlled automatically according to the difference of the noise levels between stations. So, conversation is possible under the noise levels as high as 70 to 75 phon.
- Transient response of the voice switch improved allows comfortable interruption in conversation.

IV. PAGING SIGNAL FLOW

1. Paging System and Signal Flow

In the case of normal conversation, the speech path system is formed by stations, LM unit and DL unit as shown in Fig. 28.

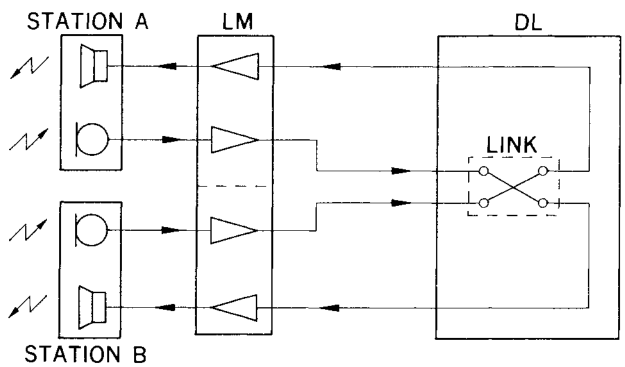


Fig. 28 Voice Signal Flow between Stations

To have the paging function in the EXES-6000 system, use a PA amplifier and external speaker instead of station B in Fig. 28. The PI unit is required as interface for this purpose. Refer to Fig.29.

When paging, just make a call from the station normally.

1-1. Paging by an External PA Amplifer (Zone Paging)

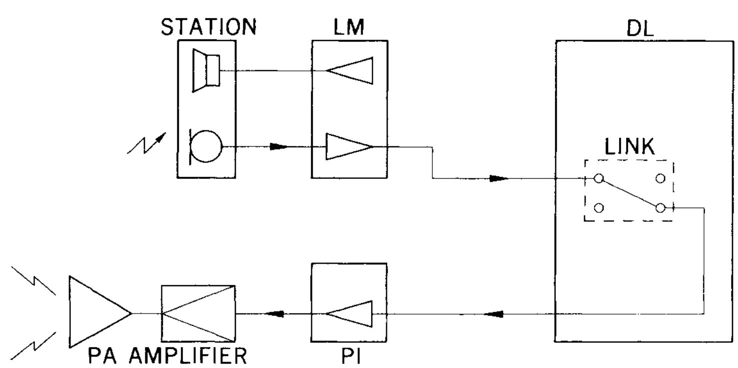


Fig. 29 Voice Signal Flow on External Speaker Paging

1-2. Station (Group) Paging

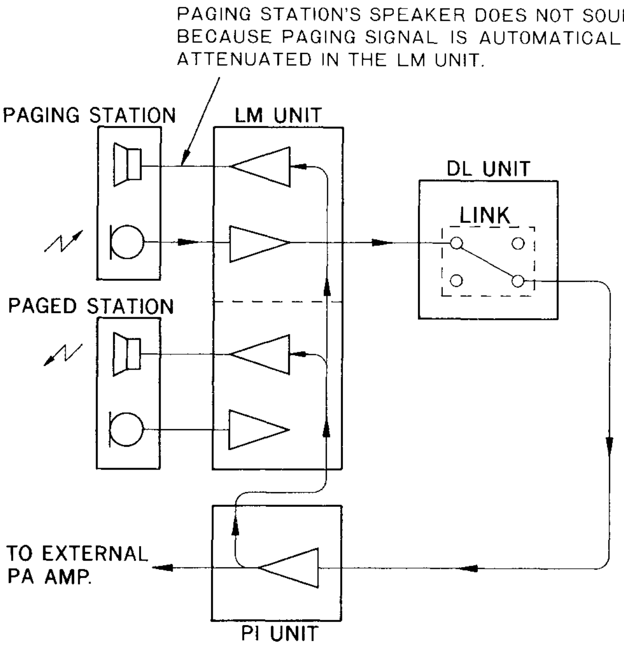


Fig. 30 Voice Signal Flow of Station Paging and Zone Paging

Besides the paging by an external PA amplifier, it is possible to page using the station speakers. In this event, only one-way speech path is formed from the paging station mic to the paged station speakers. See Fig. 30.

1-3. Block Diagram of Paging Interface (PI) Unit

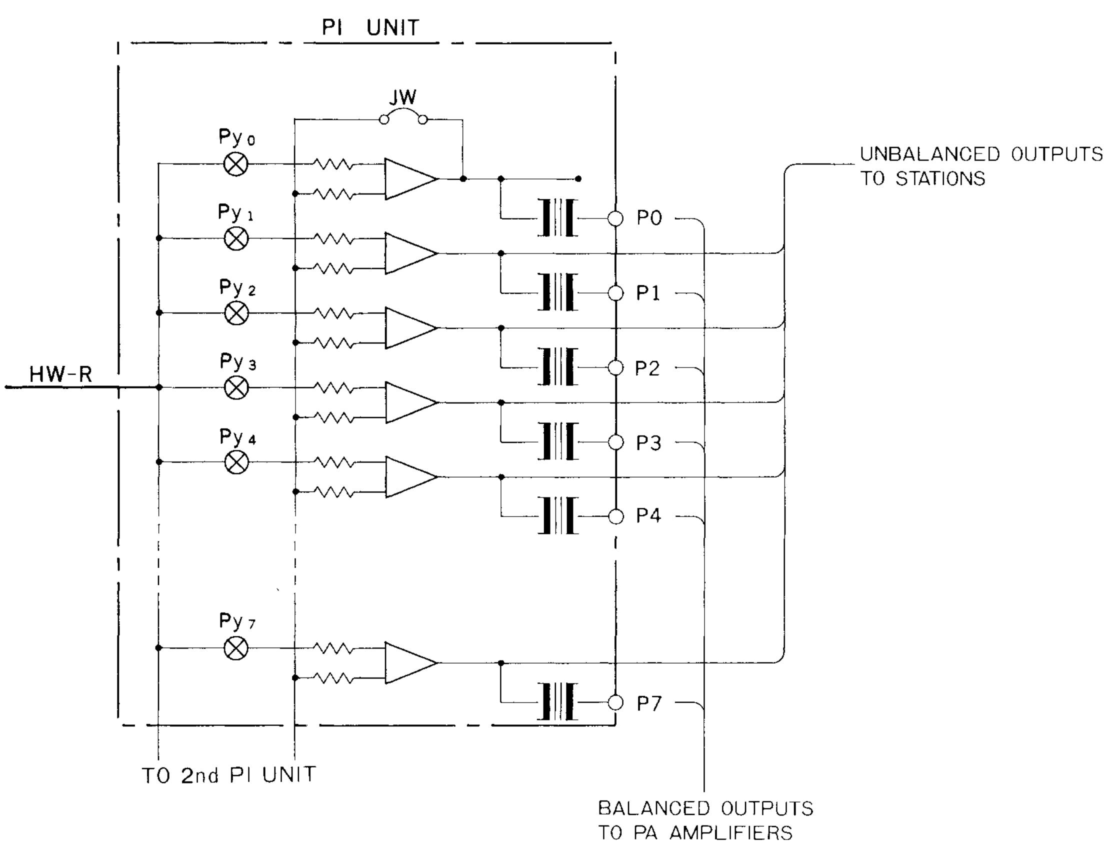


Fig. 31 Block Diagram of PI Unit (PI-62)

Fig. 31 shows block diagram of the PI unit. One PI unit is provided with 3 paging-zone outputs in PI-60 and 8 paging-zone outputs (all call + 7 zones) in PI-62. Designed so that a maximum of 1 (for EX-600) or 2 (for EX-610/620) or 4 (for EX-630) PI units can be mounted, the EXES-6000 can have all call plus up to 3 paging zones (for EX-600) or all call plus up to 15 paging zones (for EX-610/EX-620) or all call plus up to 31 paging zones (for EX-630).
In Fig. 31, the paging signal (PAM signal) from HW-R passes through analog switches and appears at each zone output (P0, P1-P7). The paging signal that has reached to P0 also proceeds to each zone output (P1-P7) through jumper wire (JW) and becomes an all call.

An incoming paging signal takes 2 separate courses at each zone output of the PI unit, and one goes to each output section in the LM unit via an assignment part of PI-60 (for EX-600) or an assignment plug (for EX-610/EX-620) or station assignment unit (SA) (for EX-630) utilized for users to assign zones for station paging, while another to a PA amplifier for external paging via an output transformer. In addition to this, relay make contacts are provided as output for the PA amplifier to allow remote ON-OFF operation of the PA amplifier.

The output P0 is seldom used practically because it is used for all call purpose. When using 2 or more PI units, by cutting a jumper wire (JW) on the second unit or its subsequent units, the output P0 of the unit can be used as one of zone outputs.

1-4. Paging Signal Flow

Fig. 32 shows a single LM unit in which zones are divided into 2 by cutting JP6:

Zone 1 (P1): Stations No. 200 — No. 203
Zone 2 (P2): Stations No. 204 — No. 207

(A) Here, supposing paging is made from station No. 202 over zone 2 by using Link No. 1 Fig. 32

(1) the voice from mic of station No. 202 passes through Tx_3 ... HW-T ... Ty_1 and is transmitted to Link No. 1, where it is held.

(2) Rx_2 and Py_2 close at the next timing, and the voice that has been held branches into 2 outputs after passing through Rx_2 ... HW-R ... Py_2: one goes to an external PA amplifier for zone 2 via transformer, and the other is sent back to the LM unit and transmitted to the speaker of each station (No. 204 through No.207) after passing through JP7 through JP9, R522, R622, R722 and R822.

(B) When all-call paging is made from station No. 204 by using Link No. 1 (Fig. 33),

(1) The voice from mic of station No. 204 passes through Tx_5 ...HW-T ... Ty_1, and is transmitted to link No. 1, where it is held.

(2) Rx_2 and Py_0 close at the next timing, and the voice that has been held passes through Rx_2 ... HW-R ... Py_0.

(3) The signal sent to P_0 is routed to all zones (No. 1 through No. 15) via jumper wire (JW). Then the signal to each zone is divided into 2 signals, and one goes to an external PA amplifier for each zone through transformer, and the other is sent back to the LM unit, and then to the speaker of stations No. 200 to No. 203 through JP3 to JP5 and to the speakers of stations No. 204 to No. 207 through JP7 to JP9.

Here, station No. 204 is a paging station. Lest the voice from this station should come into its speaker, it is so designed that the voice is attenuated in the LM unit. This also applies to the stations in conversation. When any station in the same paging group does not need to be paged, this can be accomplished by cutting the resistor (R122/R222.../R822) in the LM unit for that station.

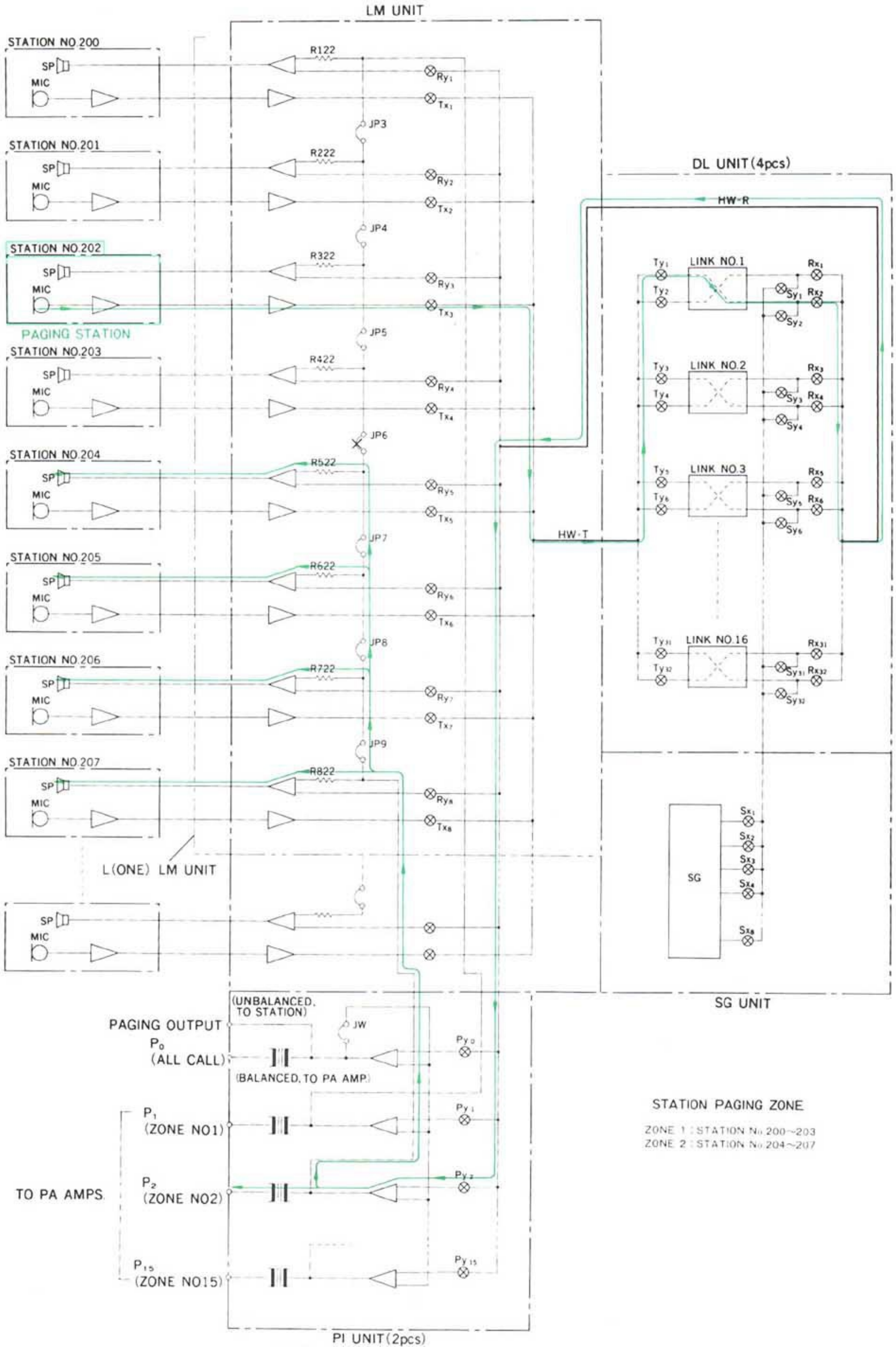


Fig. 32 Zone Paging Signal Flow

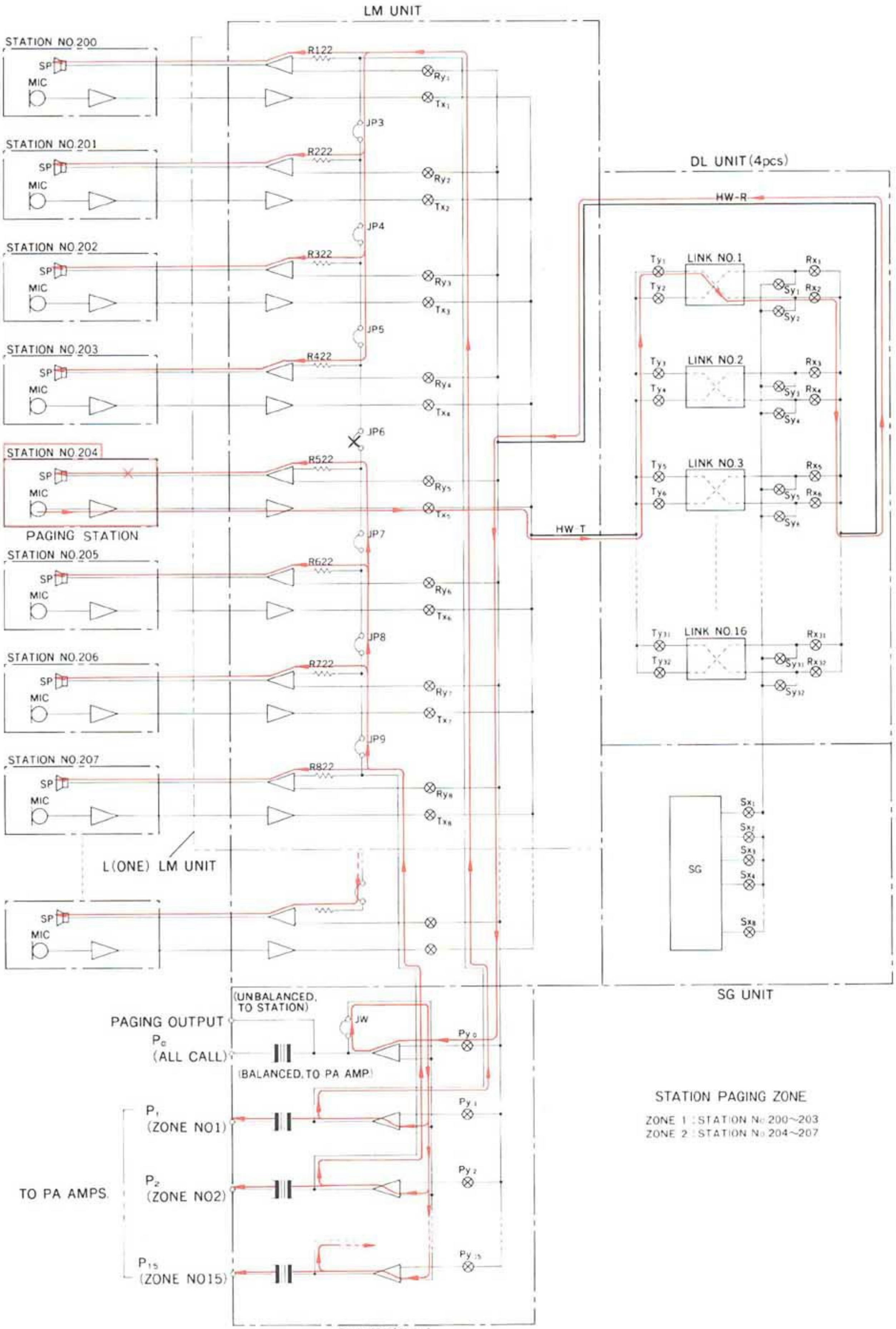


Fig. 33 All-Call Paging Signal Flow

2. Connection for Station Paging (Group Paging)

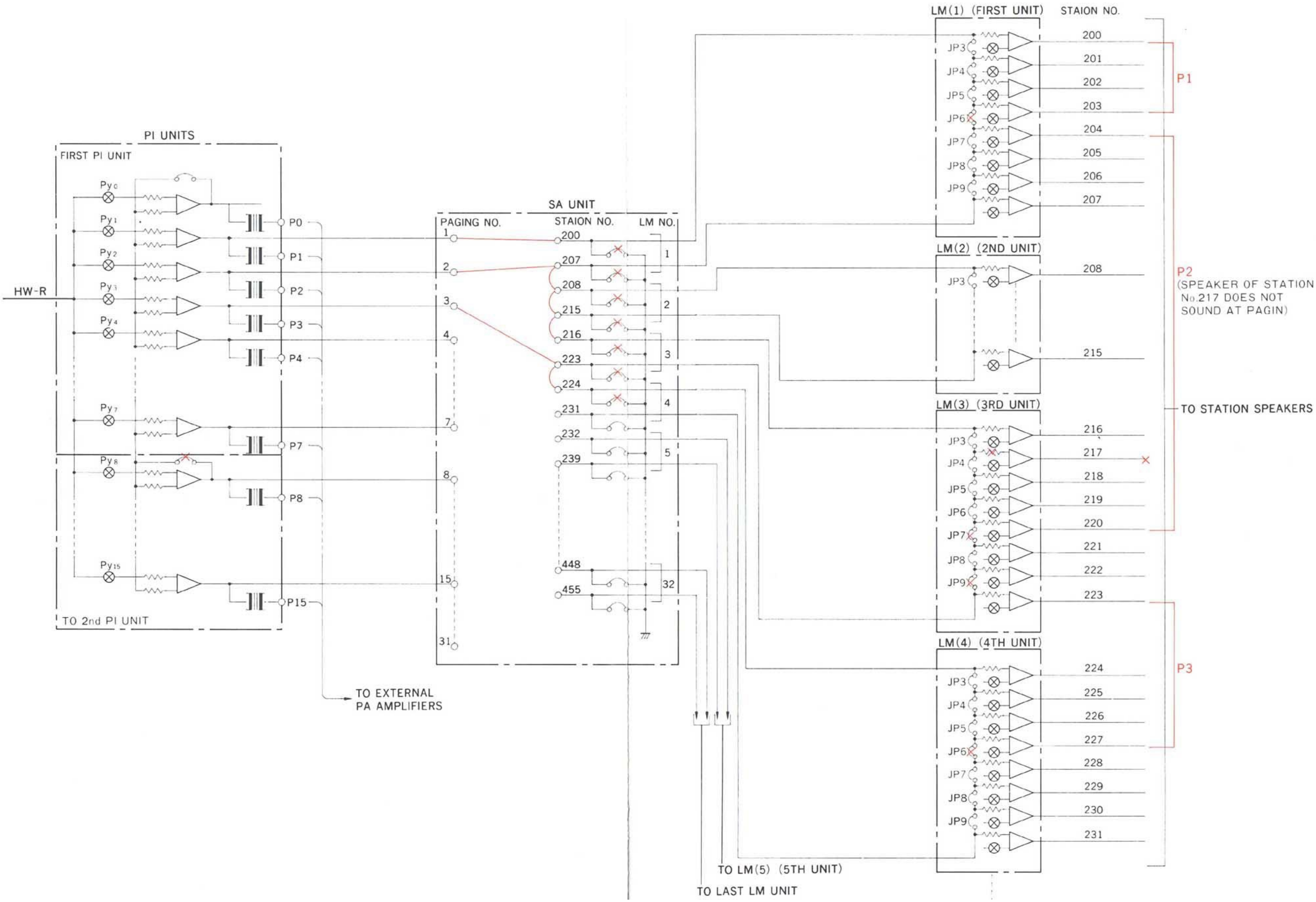


Fig. 34 Station Paging Zone Assignment (EX-630)

Assigning each station to the designated paging zone will require the assignment plug for the EX-610 and EX-620, and the SA unit (SA-64) for the EX-630. Supose that 4 LM units, LM(1) to LM(4) (32 stations, No.200 through No.231) are used in a system, with stations No. 200 to No. 203 assigned to zone P1, stations No. 204 to No. 220 to zone P2 and stations No. 223 to No. 227 to zone P3. See Fig. 34. The station number for each paging group should be consecutive numbers.

(1) To separate each zone, cut jumper wire JP6 of LM(1), JP7 and JP9 of LM(3) and JP6 of LM(4). When the paging group has a station which does not require to be paged, remove the resistor in the LM unit for that station. In this example, the resistor in LM(3) for station No. 217 is cut.

(2) Make connections for paging zone assignment in the SA unit.

(2)-1 The exchange is so designed that the paging outputs of the PI unit are connected to the inputs PAGING No. 1 to No. 31 of the SA unit, and the paging inputs of the LM unit to the outputs LM No. 1 to No. 32 of the SA unit. Therefore, each input of the SA unit needs to be connected to the first or last station number of the LM unit. In this example, the input PAGING No. 1 is connected to station No. 200, PAGING No. 2 to station No. 207 and PAGING No. 3 to station No. 223.

(2)-2 When a certain paging zone extends to more than one (1) LM unit, connect between the last and first inputs of the LM units involved. In this example, STATION No. 207, No. 208, No. 215 and No. 216 are connected for zone P2 because it extends to 3 LM units. Another connection is made between STATION No. 223 and No. 224 for zone P3 because it extends to 2 LM units.

(2)-3 Because each paging input of the LM units is grounded through jumper wire in the SA unit to prevent cross-talk, cut the jumper wires (in the SA unit) corresponding to the LM units used for paging. In this example, the following jumper wires in the SA unit are cut:
No. 200 and No. 207 for LM No. 1
No. 208 and No. 215 for LM No. 2
No. 216 and No. 223 for LM No. 3
No. 224 for LM No. 4

As far as LM unit (4) is concerned, when the paging group is assigned as shown in Fig.34, its corresponding jumper wire No. 231 in the SA unit should not be cut.

(3) When using 2 or more PI units, cut the all-call paging jumper wire in the second and its subsequent PI units.

V. PRINCIPLE OF CONFERENCE FUNCTION

Fig. 35 shows an example where 4 stations (A) to (D) are engaged in conference using the conference function (CL unit is necessary).

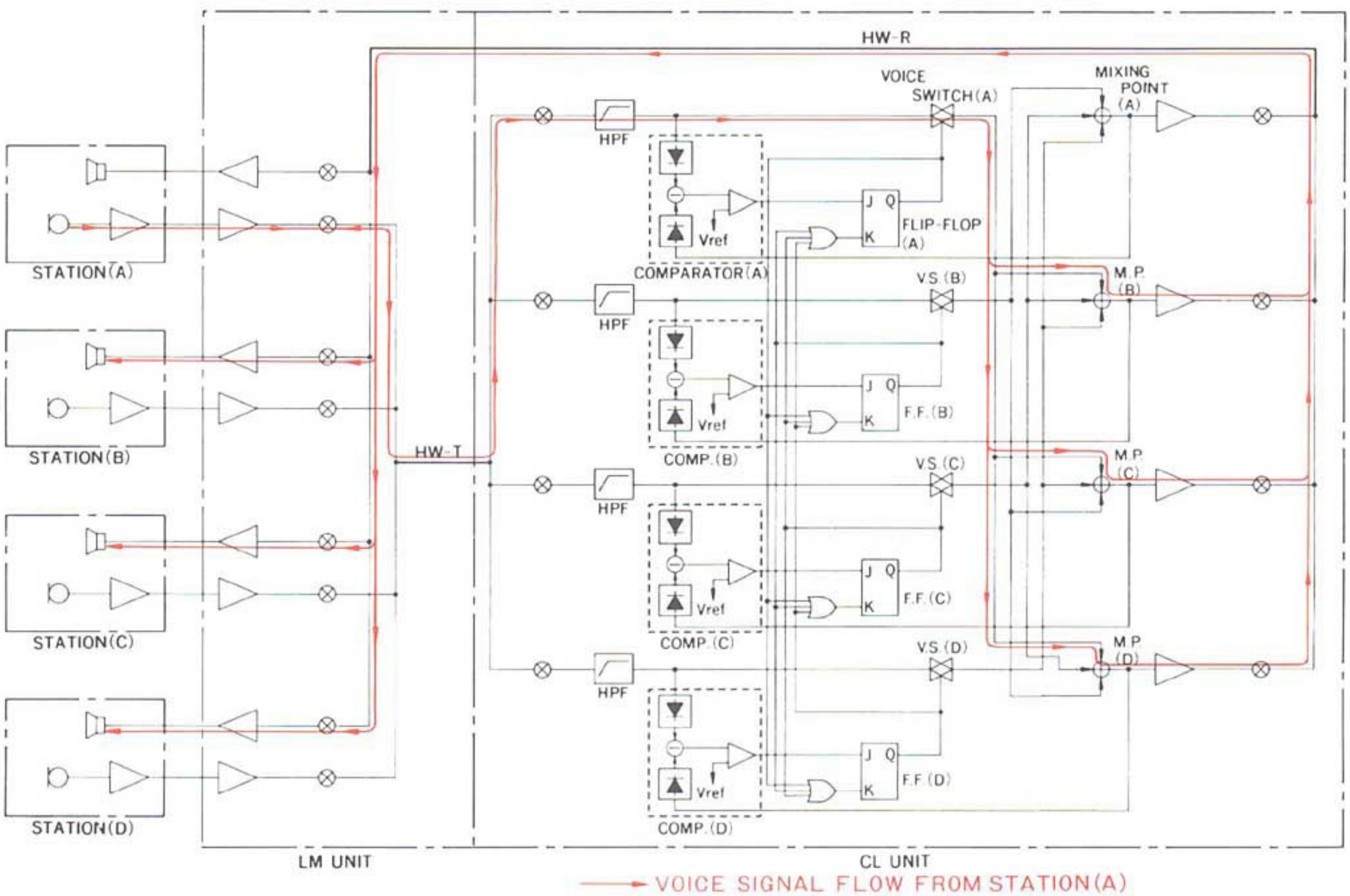


Fig. 35 Block Diagram of CL Unit

1. Example when One Station is in Conference

Supposing station (A) alone is speaking, the voice signal flows as follows:

Timing 1. Station (A)'s mic ... HW-T ... HPF ... Voice switch (A) ... (Voice is held)
Timing 2. Mixing point (B) ... HW-R ... Station (B)'s speaker
Timing 3. Mixing point (C) ... HW-R ... Station (C)'s speaker
Timing 4. Mixing point (D) ... HW-R ... Station (D)'s speaker

In this instance, the voice switch (A) is ON. We will explain why it is so.

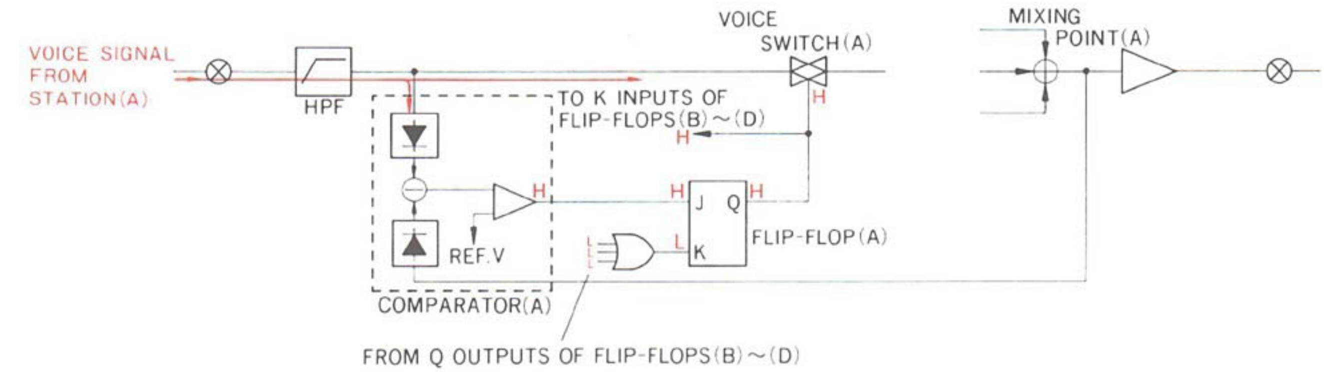


Fig. 36 Control Section of Voice Switch (A)

In Fig. 36, the voice signal that passed through HPF takes two courses: one goes to the voice switch and the other to the comparator (A). The comparator (A) compares the signal strength between the signal from station (A) and one from the mixing point (A), and provides "H" to "J" input of Flip-flop (A) if the signal level from station (A) is higher. In this event, no signal appears at the mixing point (A) because no other stations than station (A) are speaking. That is, the "J" input of Flip-flop (A) is "H" and so is the "Q" output, thus causing the voice switch (A) to be ON. In this event, since the "Q" output is connected to the "K" inputs of other Flip-flops (B) to (D), these "K" inputs all become "H" and the "Q" outputs of Flip-flops (B) to (D) are made "L". As a result, voice switch (A) alone turns on and other voice switches (B) to (D) are off.

Consider another example where both stations (A) and (D) are speaking. (See Fig. 37).

2. Example when Two Station are in Conference

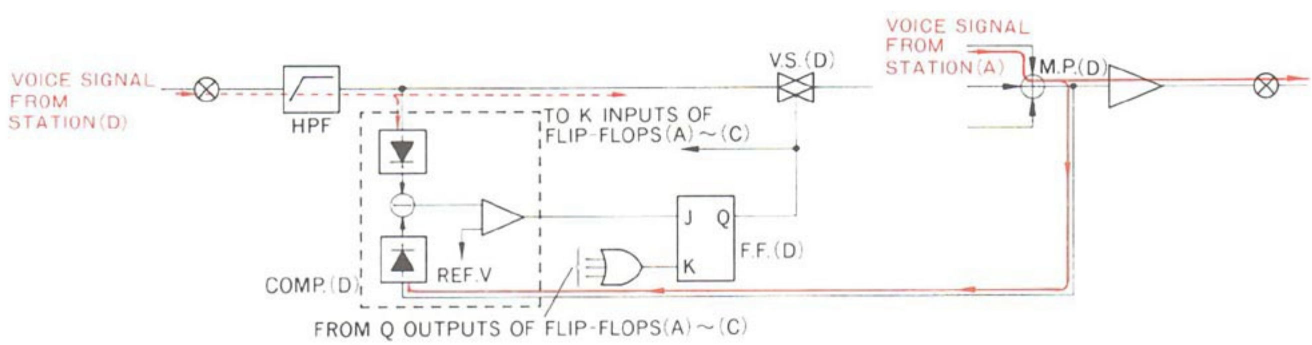


Fig. 37 Control Section of Voice Switch (D)

The voice signal from station (D) which passed through HPF is transmitted to both voice switch (D) and comparator (D). At the same time, the voice signal from station (A) that appears at the mixing point (D) is sent to the comparator (D). The comparator (D) compares the signal strength between these signals from both stations. If the signal level from station (D) is higher, the voice switch (D) turns on and the signal flows from station (D) to stations (A), (B) and (C).

To sum up, when the conference function (CL unit) is used, only one of the four voice switches is activated just as in the case of hands-free to hands-free conversation mode. The exceptions are that the CL unit has no side tone circuit and that the number of links corresponding to the DL unit is two.

VI. TIE-LINE SYSTEM

A maximum number of stations that can be used in the EXES-6000 System in 32 for the EX-600, 64 for the EX-610, 128 for the EX-620 and 256 for the EX-630. But it is possible to increase the total number of stations by connecting two or three exchanges (tie-line connection), except EX-600 which has no tie line function. The explanation is given to the operation of the Tie-line interface (TI unit) and flows of both voice and dial signals in the tie-lined system.

In Fig. 38, suppose conversation begins after station No. 200 connected to exchange A places a call to station No. 470 connected to exchange B.

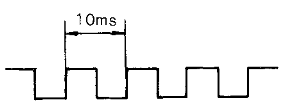
1. Dial Signal Flow

Trace green line in Fig. 38.
Dial pulse stream generated in station No. 200 is sent to the LM. It is detected by the dial pulse detector in the LM, and transmitted to the CP.

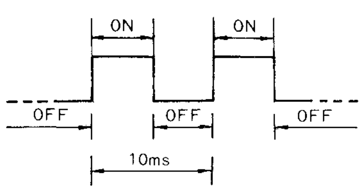
Station key number	1	2	3	4	7	8	9	0	·	C	PTT
Pulse number	1	2	3	4	7	8	9	10	11	12	16or more

The CP generates a pulse stream corresponding to the dial pulse stream and outputs it through OP (Output Port) of the OC (Output Control Unit) to the TI.

When the key 4 is dialed, the pulse stream is generated from the CP as shown to the right.

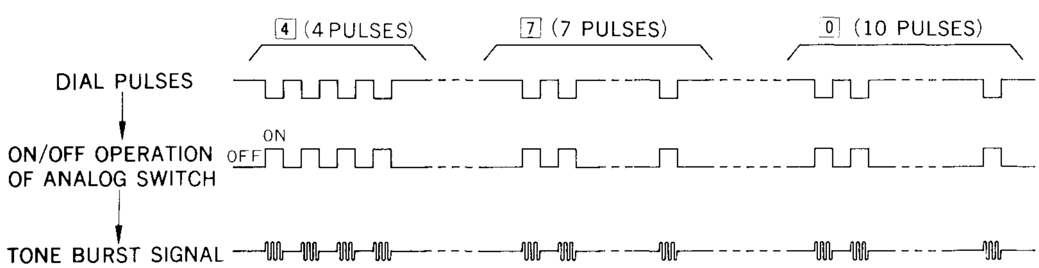


The pulse stream from the CP in turn causes analog switches in the TI to turn on and off at 10ms intervals (100Hz) for the number of the pulses.



The signal of 2.6kHz is oscillated in the data transmitting section of the TI, which turns into tone burst signal through on/off operations of the analog switches.

The Figure shown below is the process to obtain the tone burst signal when a station dials the keys 4 7 0



The tone burst signal sent to the dial receiving section of the TI in exchange B via a tie-line link is then transmitted to the CP through the BPF (Band-pass Filter), amplifier and rectifier (Amp. & Rect. in the Figure), after being converted into the original dial pulse stream by the dial pulse converter. The CP processes the dial pulse stream, and orders the HC to turn on or off the time division switch for the speech path.

In Fig. 38, a red line shown the voice signal flow from station No. 200 to station No. 470, and a green dotted line represents a dial pulse stream flow from station No. 470 to station No. 200.

2. Voice Signal Flow

See red lines in Fig. 38.
Assume that conversation flows from station No. 200 to station No. 470.

In exchange A,
(1) the voice signal flow when both Tx_1 and Ty_1 close:
Station No. 200 ... Tx_1 ... HW-T ... Ty_1 ... link No. 1 (voice is held)
(2) the voice signal flow when both Rx_2 and Ly_1 close:
Link No. 1 ... Rx_2 ... HW-R ... Ly_1 …TI

The PAM signal is demodulated into the analog voice signal in the TI unit, which passes through tie-line link No. 1 and flows to the data receiving (audio signal transmitting) section of the TI in exchange A. It is then transmitted to the data transmitting (audio signal receiving) section of the TI in exchange B.

In exchange B,
(1) the voice signal flow when both Lx_1 and Ty_1 close:
Lx_1 ... HW-T ... Ty_1 ... link No. 1 (voice is held)
(2) the voice signal flow when both Rx_2 and Ry_1 close:
Link No. 1 ... Rx_2… HW-R ... By_1 ... Station No.470

A red dotted line in Fig. 38 represents the voice signal flow from station No. 470 to station No. 200.

* The signal that flows in the tie-line link is not the PAM signal but the analog voice signal (base band). The on/off timing of time division switch for forming the speech path in exchange A is completely independent of that in exchange B.

A single TI unit has 8 each of data transmitting and receiving sections, meaning 8 tie-line links. When one (1) TI unit is mounted on each exchange, there are 8 speech links between exchanges A and B.

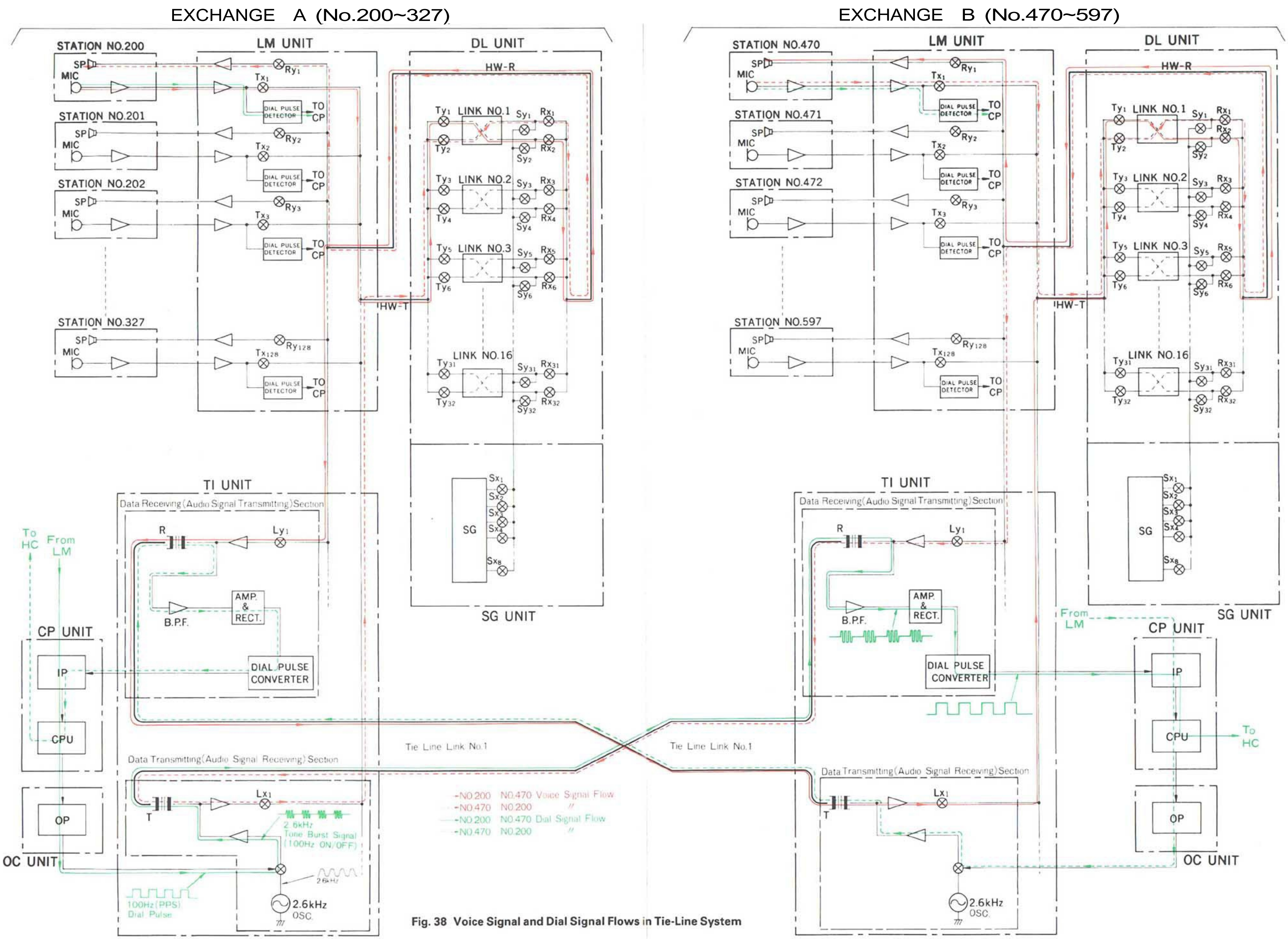
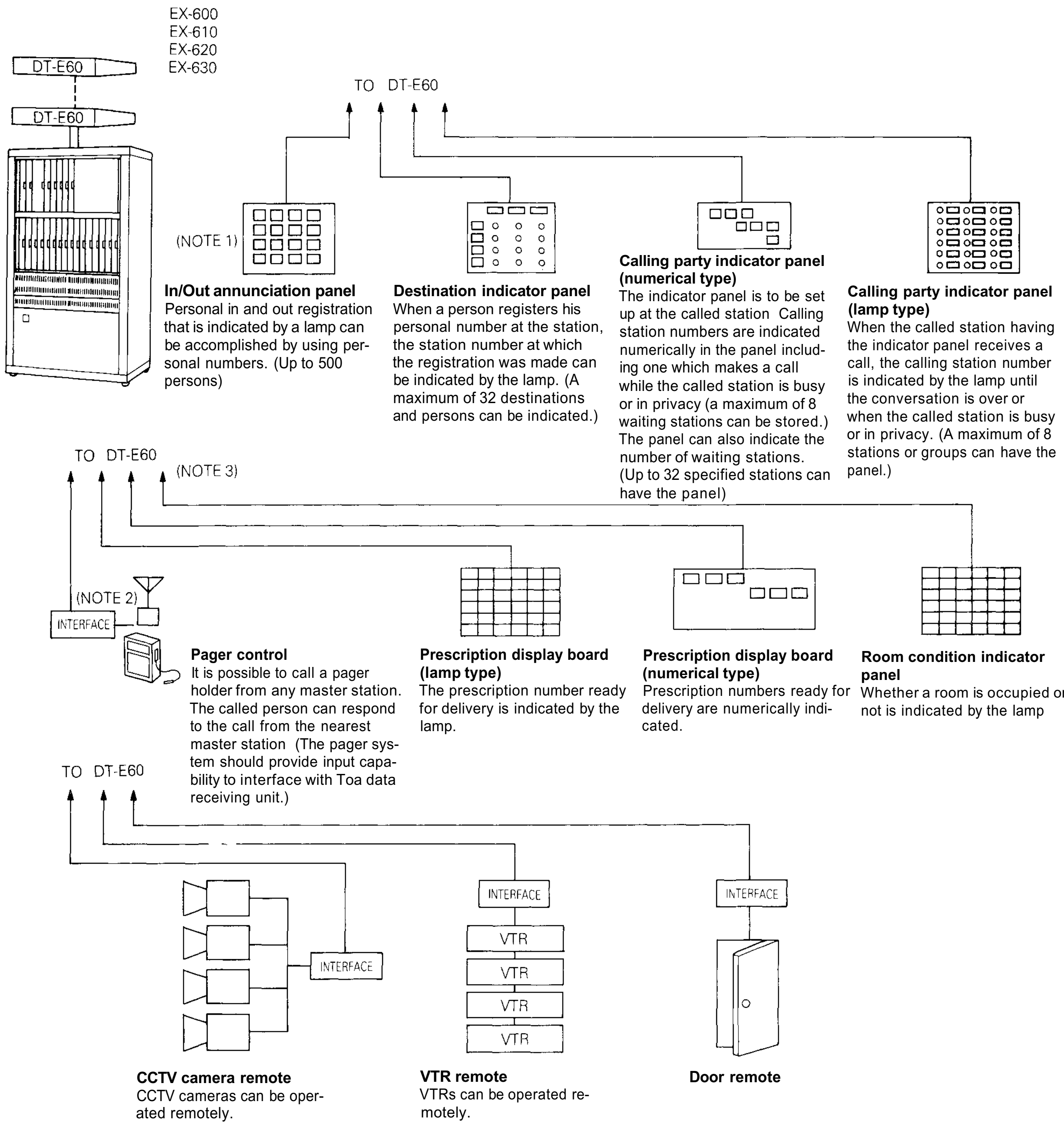


Fig. 38 Voice Signal and Dial Signal Flows in Tie-Line System

VII. DATA TRANSMITTING AND RECEIVING SYSTEM

By using the Data Transmitting Unit (DT-E60, DT-E11G or DT-E11) and Data Receiving Unit (DR-B61) in the EXES-6000 System, the ON/OFF operation of various indicators and the controls of other equipment are possible from any master station. This is an optional function. Fig. 39 shows the applications.

Note. Refer to Installation Manual of each CP unit for functions of these units.



Notes

(1) Each indicator panel needs to be made by using the Data Receiving Unit DR-B61.
(2) An interface with the DR-B61 built inside is necessary for each remote operation.
(3) Connect between the Data Transmitting Unit DT-E60 and the Data Receiving Unit DR-B61 with 2 non-polar wires.

Fig. 39 Applications of Data Transmitting and Receiving System

The data transmitting unit (DT) converts the data received from the CP unit in the exchange, and transmits the converted data to the data receiving unit (DR). After receiving the data, the DR will control 32 relays contained in it. The DR-B61 is a printed circuit board assembly, and users need to provide desired indicators or controllers by using it.

1. Function Setup (DIP switch setting)

1-1. DIP Switch Setting for the Data Transmitting Unit DT-E60, ET-E11G or DT-E11

A maximum of 16 data transmitting units can be connected in parallel per exchange. It is possible to determine the channel number of the unit by setting the DIP switch on the PC board inside the unit. See Fig. 40.

The function that corresponds with the channel number varies according to the CP unit to be used. So, refer to Tables 1 through 4 when selecting the desired function with the DIP switch.

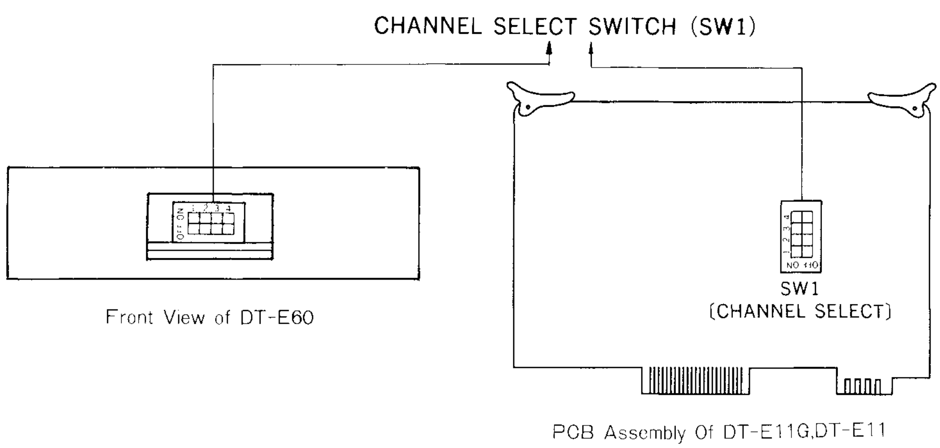


Fig. 40 Data Transmitting Unit

Note. The number of channels available from each type of CP unit:
CP-60 (EX-600) .... CH. 0 ~ CH. 1
CP-62 (EX-610/620) .... CH. 0 ~ CH. 11
CP-63 (EX-610/620 tie-line) .... CH. 0 ~ CH. 3
CP-64 (EX-630) .... CH. 1 ~ CH. 15

In any case, a maximum number of the DT units that can be used is 16, and the channel number of each DT unit is freely selectable. (It is possible to use all the 16 DT units for Channel No. 1.)

Table 1. CHANNELS VS FUNCTIONS FOR CP-60

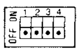
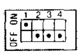
CHANNEL SELECTION	WORD NO.	FUNCTIONS	DESCRIPTION	APPLICATION
CH.0 	0~7	One-shot Make Output (100 contacts)	One-shot make output is delivered for 1 to 2 seconds when a function code is dialed.	• ITV camera select • VTR control
	8~13	8 Selectable Make Output (9 units)	One contact out of 8 selectable make outputs is obtained for about 1 or 2 seconds.	• VTR control
	14~20	In/Out Annunciation (100 persons)	Personal in and out registration can be accomplished at any Master station by using personal numbers max. 100 IN/OUT annunciations may be done.	• Use of In/Out annunciation panel with personel with personal number function.
	21~27	Pager Control Output (100 pagers)	Make output (100 contacts) is available for pager control.	• Pager
CH.1 	0~15	Calling Indication (32 stations)	Up to 32 stations calling one (1) same station (or 1 group having stations of consecutive numbers) can be indicated with lamp.	• A maximum of 8 groups of called stations can have a panel which is common within each group.
	16~31	Calling and called party indication (32 stations)	A maximum of 32 stations in the conversation mode including called stations can be indicated with lamp.	• Can be divided into up to 8 groups.

Table 2. CHANNELS VS FUNCTIONS FOR CP-62

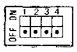
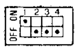
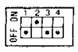
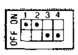
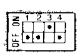
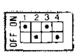
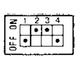
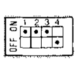
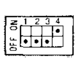
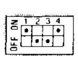
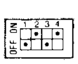
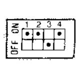
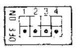
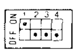
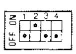
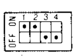
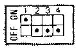
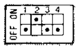
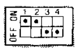
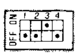
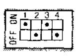
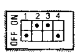
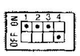
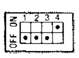
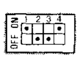
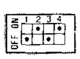
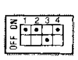
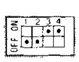
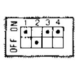
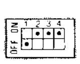
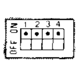
CHANNEL SELECTION	FUNCTIONS	DESCRIPTION	APPLICATION
CH.0 	IN/OUT Annunciation (500 persons)	Personal in and out registration can be accomplished at any Master station by using personal numbers Max. 500 IN/OUT annunciations may be done.	• IN/OUT Annunciation
CH.1 	Make/Break Output (512/100 contacts)	Make/Break contacts can be available at any Master station.	• Door Remote • IN/OUT Annunciation
CH. 2 	One-shot Make Output (500/50 contacts)	One-shot make contacts can be available at any Master station.	• ITV camera select • VTR control
CH. 3 	(1) 4 Decimal digits output (9 units)	Indicate by 7 segments LEDs.	• Prescription annunciation
	(2) Decimal Output (9 units)	10 Selectable Decimal Outputs are available with 7 segments LEDs.	• Room condition indication
	(3) 8 Selectable Make Output. (9 units)	One contact out of 8 selectable make outputs is obtained. "Clear" operation makes all 8 relays break.	• Destination indication
	(4) Pager Control Output (100 pagers)	Make output (100 contacts) is available for pager control.	• Pager
	(5) 8 Selectable One-shot Make Output (9 unit)	One contact out of 8 selectable make outputs is obtained for about 1 or 2 seconds.	• VTR control
CH. 4 	Decimal Output (99 units)	10 Selectable Decimal Outputs are available with 7 segments LEDs.	• Room condition indication • Destination indication
CH. 5 	8 selectable make Output (64 units)	One contact out of 8 selectable make outputs is obtained. "Clear" operation makes all 8 relays break.	• Room condition indication • Destination indication
CH. 6 	Calling Party Indication Numerical-type (1)	When a station with a Display Board is called, calling party number is indicated until the conversation is over and also when the called station is busy or in privary.	• The number of called stations are No.201 ~ No.216.
CH. 7 	Calling Party Indication Numerical-type (2)		• The number of called stations are No.217~No.232.
CH. 8 	Calling Party Indication (One Station; One Lamp)	Max. 128 Calling station numbers can be indicated when designated called station with Display Board is called. The numbers of called stations having an indication panel can be programmed at No.200 station.	• The group number of called station(s). No. 1~4
CH. 9 	Calling Party Indication (One Station; One Lamp)		• The group number of called station(s).No.5~8
CH. 10 	Destination Indication (1)	When a person makes his own Personal Number Programming at the station, the station number at which the registration was made can be indicated by the lamp.	• Personal number No. 1000~ 1015
CH. 11 	Destination Indication (2)		• Personal number No. 1016~1032

Table 3. CHANNELS VS FUNCTIONS FOR CP-63

CHANNEL SELECTION	FUNCTIONS	DESCRIPTION	APPLICATION
CH. 0 	IN/OUT Annunciation	Personel in and out registration can be accomplished at any Master station by using personal numbers. Max. 500 IN/OUT annunciations may be done. (All the 3 exchanges provided the same indication)	• IN/OUT Annunciation
CH. 1 	(1) One-shot Make Output (50 contacts)	One-shot make contacts can be available at any Master station. *1	• ITV camera selection • VTR control
	(2) Make/Break Output (100 contacts)	Make/Break contacts can be available at any Master station. *1	• Door Remote • IN/OUT Annunciation
	(3) 8 Selectable Make Output (9 unit blocks)	One contact out of 8 selectable make outputs is obtained. "Clear" operation makes all 8 relays break. *1	• Destination indication • VTR control
	(4) Decimal Output (9 unit blocks)	10 Selectable Decimal Outputs are available with 7 segments LEDs. *1	• Room condition indication.
	(5) 4 Decimal digits output (9 unit blocks)	Indicate by 7 segments LEDs. *1	• Prescription annunciation
	(6) Pager Control Output (64 contacts)	Make output (64 contacts) are available for pager control. *2	• Pager
CH.2 	Calling Party Indication (1) (One Station; One Lamp)	Max. 120-Calling station numbers can be indicated when designated called station with Display Board is called. The numbers of called stations having an indication panel can be programmed at No. 200 station. (Only the calling stations within the same exchange can be indicated by a lamp)	• The group number of called station(s). No. 1 ~ 4
CH.3 	Calling Party Indication (2) (One Station; One Lamp)		• The group number of called station(s). No. 5 ~ 8

Note.
*1. Each exchange has an independent control system, and it is impossible to control the Data Transmitting Unit of the other exchange from the station connected to the different exchange.
*2. Can only be connected to the exchange A (Station No. 200 ~ 327). It is impossible to call the pagers from any station not connected to the exchange A. However, the response to a pager call is possible from any station regardless of the exchange it is connected to.

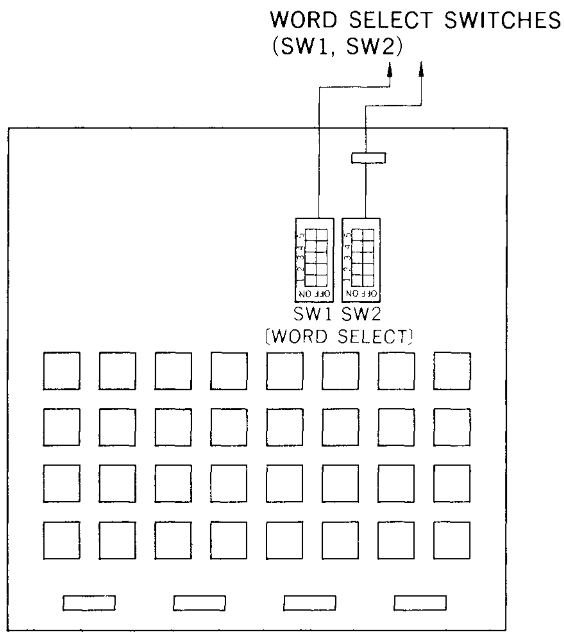
Table 4. CHANNELS VS FUNCTIONS FOR CP-64

CHANNEL SELECTION	FUNCTIONS	DESCRIPTION	APPLICATION
CH. 1 	Make/Break Output (512/100 contacts)	Make/Break contacts can be available at any Master station.	• Door Remote • IN/OUT Annunciation
CH. 2 	One-shot Make Output (500/50 contacts)	One-shot make contacts can be available at any Master station.	• ITV camera select • VTR control
CH. 3 	(1) 4 Decimal digits output (9 units)	Indicate by 7 segments LEDs.	• Prescription annunciation
	(2) Decimal Output (9 units)	10 Selectable Decimal Outputs are available with 7 segments LEDs.	• Room condition indication
	(3) 8 Selectable Make Output. (9 units)	One contact out of 8 selectable make outputs is obtained. "Clear" operation makes all 8 relays break.	• Destination indication
	(4) Pager Control Output (100 pagers)	Make output (100 contacts) is available for pager control.	• Pager
	(5) 8 Selectable One-shot Make Output (9 unit)	One contact out of 8 selectable make outputs is obtained for about 1 or 2 seconds.	• VTR control
CH. 4 	Decimal Output (99 units)	10 Selectable Decimal Outputs are available with 7 segments LEDs.	• Room condition indication • Destination indication
CH. 5 	8 selectable make Output (64 units)	One contact out of 8 selectable make outputs is obtained. "Clear" operation makes all 8 relays break.	• Room condition indication • Destination indication
CH.6 	Calling Party Indication Numerical-type (1)	When a station with a Display Board is called, calling party number is indicated until the conversation is over and also when the called station is busy or in privary.	• The number of called stations are No.201~No.216.
CH. 7 	Calling Party Indication Numerical-type (2)		• The number of called stations are No.217~No.232.
CH. 8 	Calling Party Indication (One Station; One Lamp) (1)	Max. 256 Calling station numbers can be indicated when designated called station with Display Board is called. The numbers of called stations having an indication panel can be programmed at No.200 station.	• The group number of called station(s). No. 1~2
CH. 9 	Calling Party Indication (One Station; One Lamp) (2)		• The group number of called station(s). No. 3~4
CH. 10 	Calling Party Indication (One Station; One Lamp) (3)		• The group number of called station(s). No. 5~6
CH. 11 	Calling Party Indication (One Station; One Lamp) (4)		• The group number of called station(s). No. 5~6
CH. 12 	Destination Indication (1)	When a person makes his own Personal Number Programming at the station, the station number at which the registration was made can be indicated by the lamp.	• Personal number No. 1000~1015
CH. 13 	Destination Indication (2)		• Personal number No. 1016~1031
CH. 14 	In/Out Annunciation (1)	Personal in and out registration can be accomplished at any Master station by using personal numbers Max. 1000 IN/OUT annunciations may be done.	• Personal number No.1000~1503 (504 persons)
CH. 15 	In/Out Annunciation (2)		• Personal number No.1504~1999 (496 persons)

1-2. DIP Switch Setting for the Data Receiving Unit DR-B61

The DR-B61 will control 32 relays after receiving the data of 512 bits (16 bits × 32 words) from the data transmitting unit. For example, word 1 covers No. 0 through No. 15 and word 2 No. 16 through No. 31 when the IN/OUT annunciation panel is used.

One (1) word consists of 16 bits (16 relays), and each DR unit has a 2-word capacity.



DR-B61 (PCB ASSEMBLY)

There are 32 words (WD.0 — WD.31), and any word setting is possible including a case where all the 16 DR units have word 0. Since the output data (function) corresponding to each channel varies according to the CP unit to be used, refer to the EXPLANATION OF DATA RECEIVING UNIT OUTPUT DATA of Installation Handbook for CP unit when setting the word number of each DR unit with the DIP switches of the DR unit, it is possible to connect up to 50 DR units to one (1) DT unit. This makes a 100-word capacity, but a maximum number of types of word selectable is still 32. When a system uses 50 DR units, for example, such word arrangement is possible:

WD.0 — WD.29 × 1 each .... 30 words
WD.30 × 35 .... 35 words
WD.31 × 35 .... 35 words
totaling 100 words

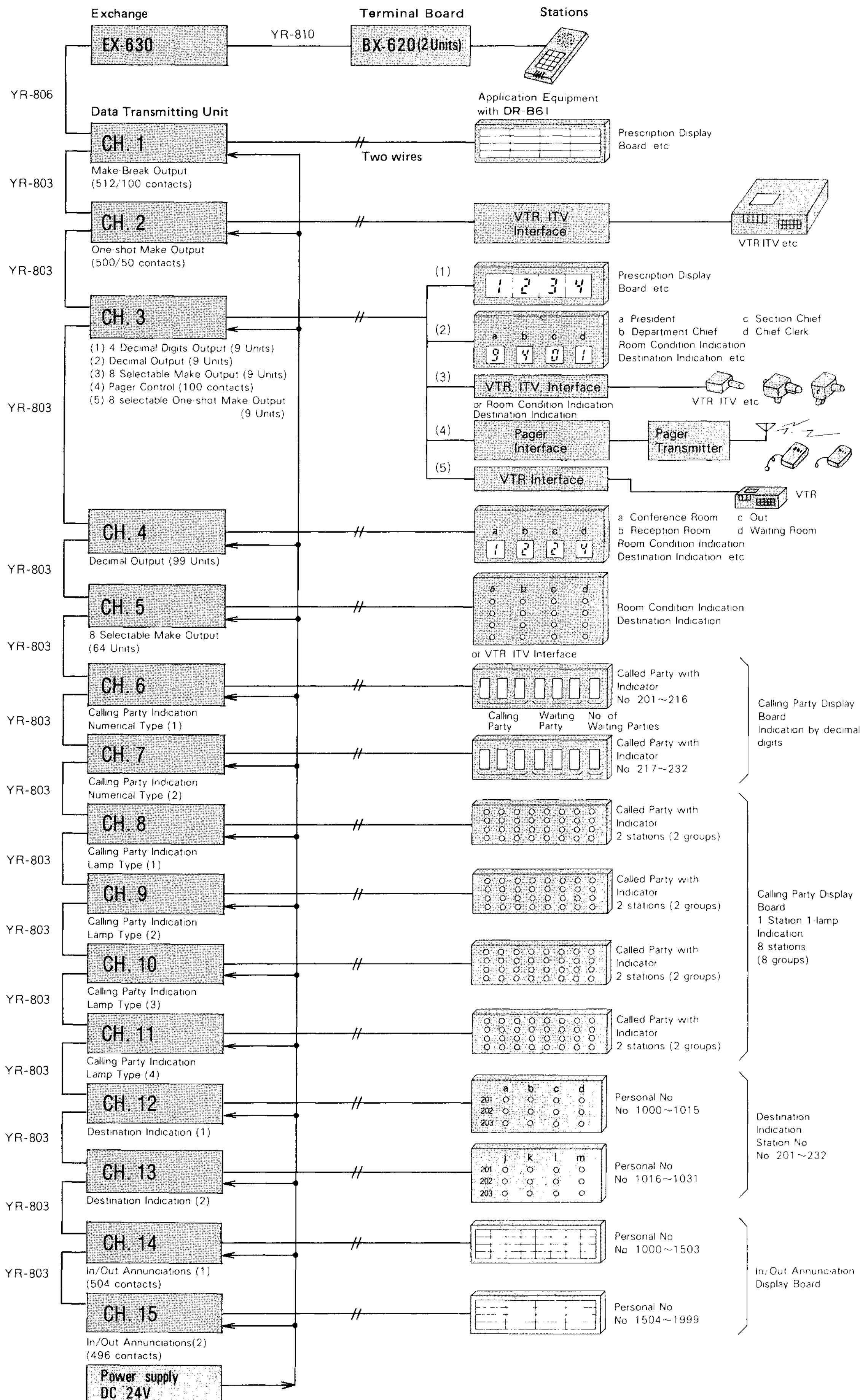


Fig. 41 System Block Diagram of Data Transmitting and Receiving Units in CP-64 System

2. Principle of Data Transmitting and Receiving System

The CPU inside the CP unit has such memory that stores the data as is shown in Fig. 42. When the specific function code is dialed at station, the 1-word data is stored in a designated area. The 1-word data consists of 16 bits, and the 1-bit data is expressed as 0 or 1 in binary code, with 0 set for relay OFF and 1 for relay ON. The simplest way of controlling the indicator or controller by the content of the data is to directly connect between each 1-bit data and the device via a cable. The total number of transmission wires in this case is equal to the number of bit data plus 1 (common line). A system using the CP-64 requires
$(16 \times 32) \times 15 + 1 = 7{,}681$ wires.

The data transmitting and receiving system employed in the EXES-6000 mentioned hereafter requires only 2 transmission wires instead of above tremendous number of wires.

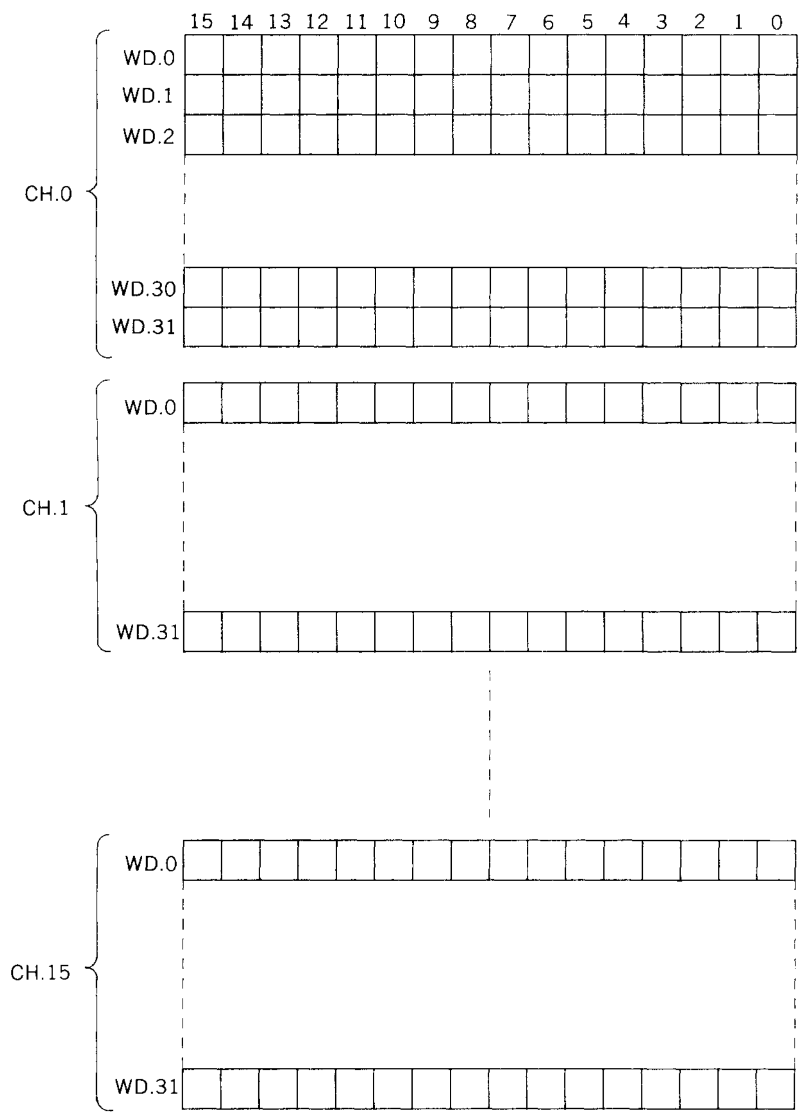


Fig. 42 Memory Map in CPU

Suppose that the data in Figs. 43 and 44 (1) is stored at an address of WD.1 (Word 1) of Channel 1 of the CPU memory inside the CP unit.

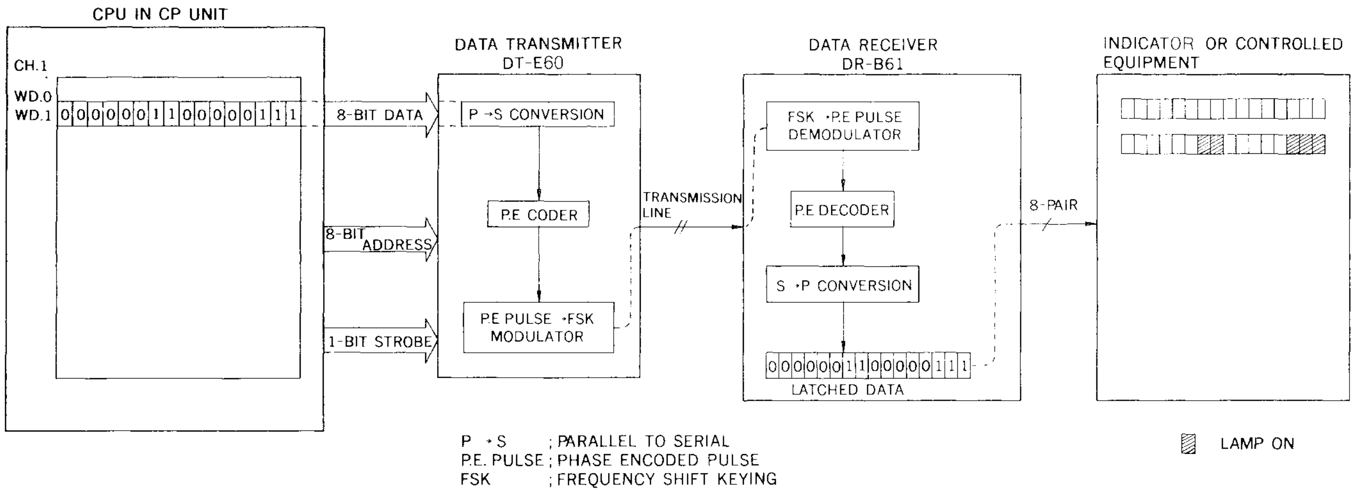


Fig. 43 System Block Diagram of Data Transmission and Reception

The 1-word data consisting of 16 bits is transmitted to the DT unit via the 8-bit data bus. Firstly, the lower 8-bit data (0th bit ~ 7th bit) is transmitted, and secondly, upper 8-bit data (8th bit ~ 15th bit). Thirdly, the 5-bit data representing data address (word number) inside the memory is transmitted.

In the DT unit, each time these data are transmitted, they are latched at the address instructed by the CPU. The 8-bit address data from the CPU contains channel number and memory address (latch IC) inside the DT unit. After the DT unit receives a complete set of data (② of Fig. 44), the data is parallel-to-serial converted and phase-encoded (P.E.). (See ③ of Fig. 44.) The start and the end data are added to the beginning and the end of the P.E. data, respectively so that the DR unit may know the start and the end of data reception. (See ③ of Fig. 44.) The P.E. data (pulse) is transmitted to the DR unit after its high and low levels are converted into frequencies of 28kHz and 18kHz, respectively (that is, after being FSK modulated). See ④ of Fig. 44.

The DR unit goes through the process reverse to that of the DT unit, and as a result, it obtains the same data as that of the CPU memory of the CP unit. The finally obtained data activates the relays' contacts of the DR unit, thus permitting controls of the indicator or controller connected to the DR unit.

Above explanation is just about the data processing for (1) word. All the 512-word data are also always sent out to the DT unit on a word-by-word basis regardless of the existence of the data in the words, and are processed in the aforementioned manner. The data of each of the words 0 through 31 is transmitted from the DT unit to the DR unit at time intervals of 0.32 seconds.

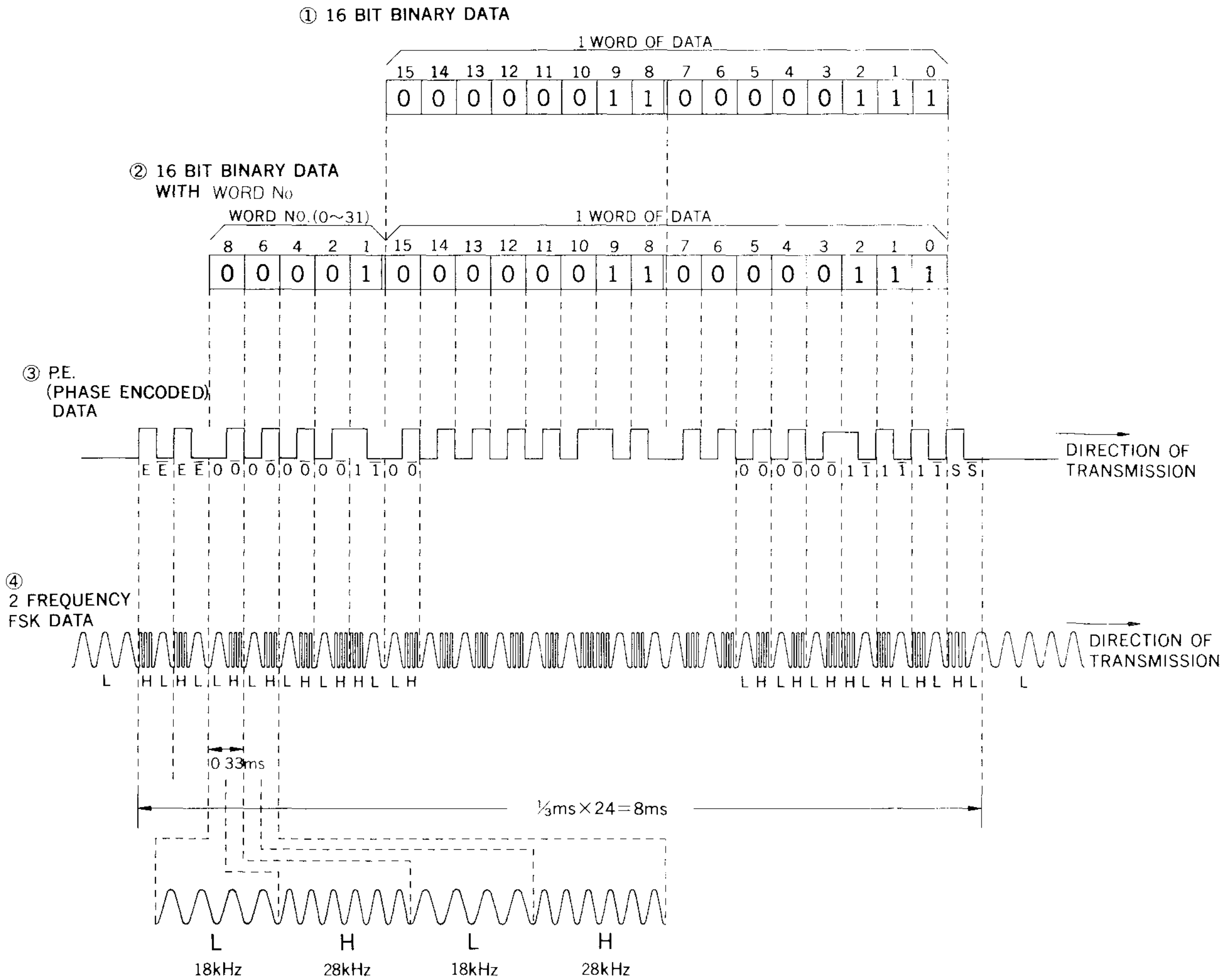


Fig. 44 Contents of Data from CPU and Conversion Process into Transmitting Signal (FSK)

VIII. TROUBLESHOOTING

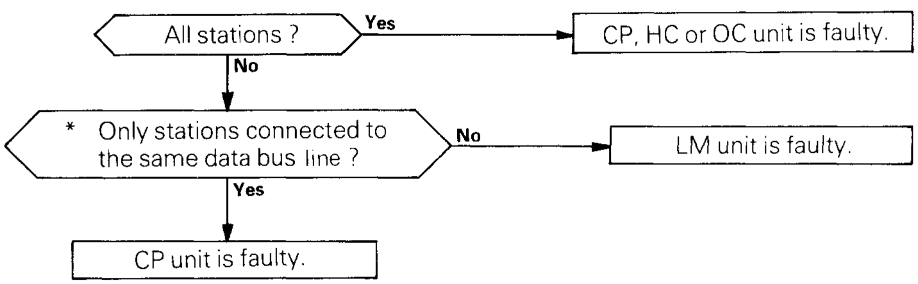
This chapter deals with how to find in the field the defective card in the system already installed, according to symptoms of defect. Now that the field repair is limited to the replacement of the card, it is not referred to how to repair the defective card at the component level.

Each symptom is given on conditions that power supply and stations work normally and that the following are properly arranged:

- Initial programming setup
- CP unit's DIP switch setting
- Mounting of each card on its designated location in a cabinet rack
- Wirings

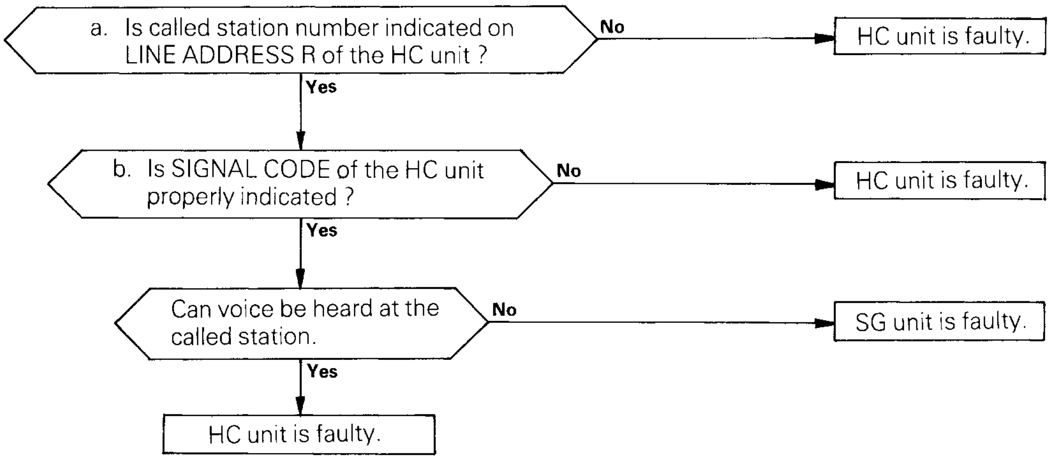
In order to find whether or not the station is faulty, it is recommended to replace the original station with normal one to see if the same symptom will result. "XX unit is faulty" that appears in the flow chart means the type of card that is the most suspicious for the cause. If replacing the card still does not remedy the symptom, try to find the other suspicious card by following the signal flow from the input through reference to the foregoing sections of this booklet. This also applies to the other symptoms not mentioned in this troubleshooting.

Symptom 1. Cannot dial.



* The dial data transmitted from the station finally comes into the CP unit's input port through the dial data bus line of the LM unit. The data bus consists of 8 lines, which are connected to the input ports No. 0 through No. 7, respectively. Each bus line corresponds to the station numbers that skip every 8 numbers; for example, the input port No. 0 corresponds to the stations No. 200, No. 208, No. 216 .... and the input port No. 1 to the stations No. 201, No. 209, No. 217 and so on. Therefore, when dialing is impossible from a certain station, whether or not the CP unit is faulty can be determined by dialing from stations of which numbers increase or decrease by every 8 from the faulty station number.

Symptom 2. Call tone does not sound.



How to check for a. and b. above.
It is possible to confirm the called station number and the type of service signal tone by the lamp status on HC unit.

(1) How to find the called station number (① in Fig. 45)

Example for HC-62

64	32	16	8	4	2	1	
○	●	●	○	○	○	●	R

●: Lighting

Called station number: 249 = 200 + 32 + 16 + 1

Note. The lamp status is indicated by the binary code.
The lamp for station No. 200 does not light.

(2) Service signal tones (② in Fig. 45)

●: Lighting

SIGNAL CODE F	4	2	1	TONE AND SPEECH MODE
○	○	○	○	During Conversation
○	○	○	●	Press-to-talk (R → T)
○	○	●	○	Press-to-talk (T → R)
○	○	●	●	OFF duration on Continuous Calling
●	○	○	○	Calling
●	○	○	●	Privacy/Disconnected
●	○	●	○	Busy
●	○	●	●	Dialing
●	●	○	○	Zone paging
●	●	○	●	All call paging
●	●	●	○	Priority/Executive priority
●	●	●	●	Registration/Call holding/Mic-off

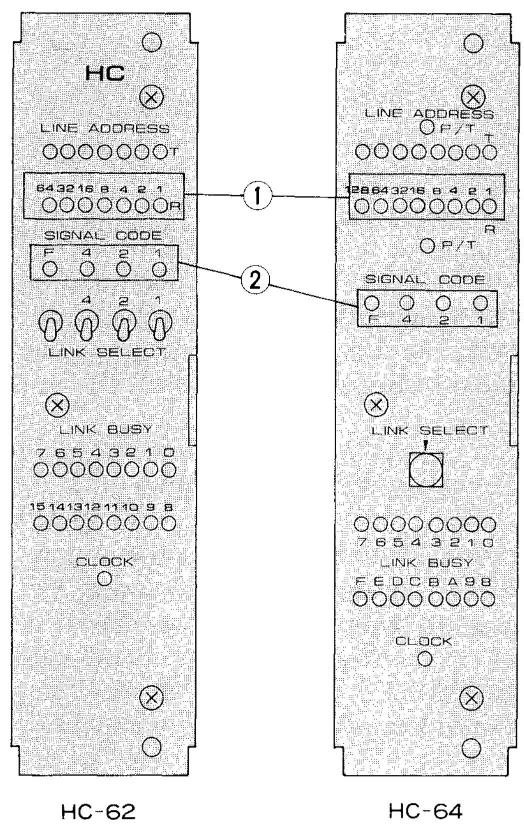
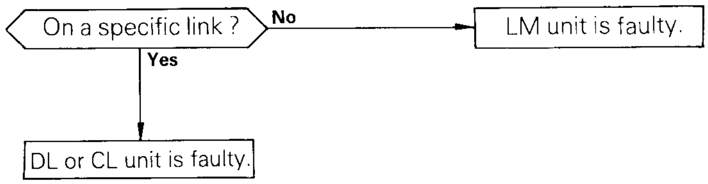


Fig. 46 Called Station Number and Signal Code Indications on HC Units

Symptom 3. Call tone is normal, but voice is distorted.



Test procedures

a. Prepare a couple of normal stations.
b. Make a call from one station to another.
c. Confirm the occupied link number with the LINK BUSY lamps of the HC unit. See Fig. 46.
d. Check the voice distortion by actually making conversation.
e. Terminate the conversation by depressing C button on the calling station.
f. Follow the procedures (b) to (e) for 16 times (the total number of links) by using the same stations. In this event, the LINK BUSY lamps must light in numerical order.

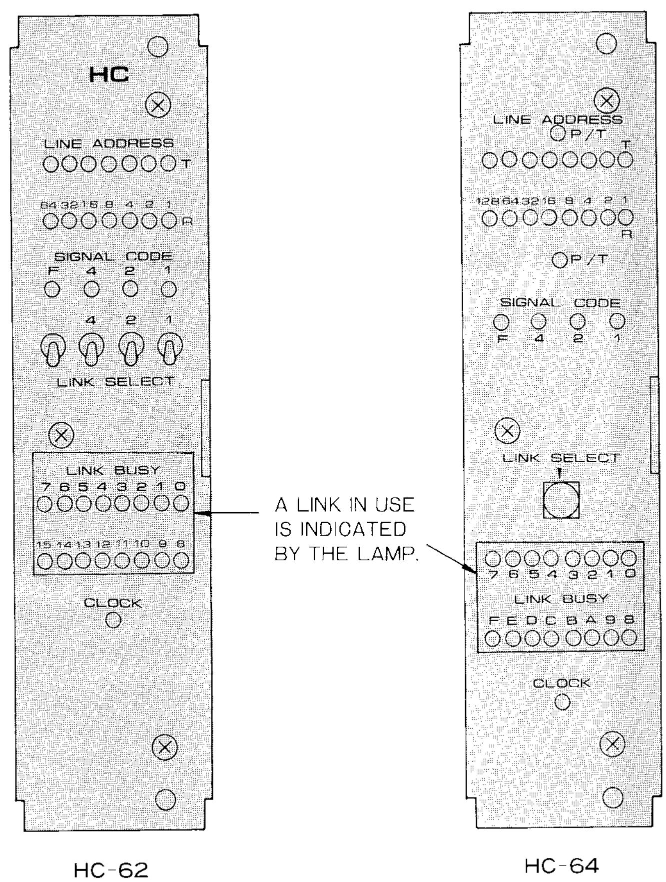
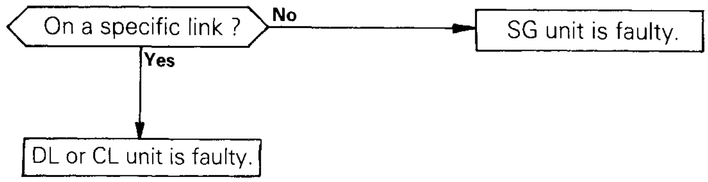


Fig. 46 Link Busy Indication on HC Units

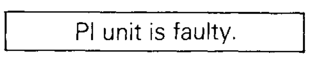
Symptom 4. Call tone is distorted.



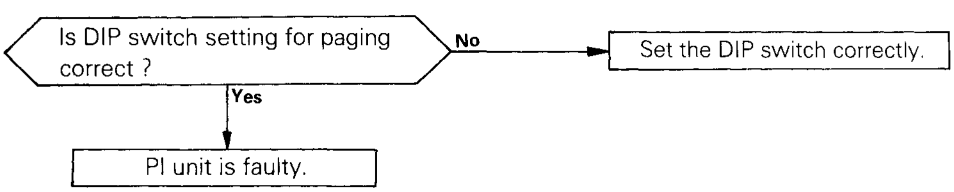
Test procedures

a. Prepare a pair of normal stations.
b. Make a call from one station to another by means of continuous calling tone,
c. Confirm the occupied link number with the LINK BUSY lamp of the HC unit. See Fig. 46.
d. Check to see if this calling tone is distorted.
e. Terminate the call by depressing C button on the calling station,
f. Follow the procedures (b) to (e) for 16 times (the total number of links) by using the same stations. In this event, the LINK BUSY lamps must light in numerical order.

Symptom 5. Conversation is normal, but paging voice is distorted.



Symptom 6. Cannot make PA paging.



Confirm the proper DIP switch setting by referring to Installation Handbook CP-60, CP-62, CP-63 or CP-64.

GLOSSARY

BASEBAND SIGNALING
Transmission of a signal at its original frequencies, i.e., a signal not changed by modulation.

BUS
A transmission channel for data from one of several sources to one of several destinations. Only one transmission via the bus is possible at the time.

CARRIER
A continuous frequency capable of being modulated, or impressed with a second (information carrying) signal.

CL (Conference Link Unit)
Operates under the control of the HC unit to connect or disconnect the links for conversation among up to 4 parties and conversation between 2 parties. One CL unit is provided with 1 conference link and 2 speech links.

CP (Central Processing Unit)
Reads out the exchange procedures written into the memory (ROM) and collates the data from stations for processing in succession.

CROSS-BAR SWITCH
A switch having a plurality of vertical paths, a plurality of horizontal paths, and electromagnetically operated mechanical means for interconnecting any one of the vertical paths with any of the horizontal paths.

CROSS-BAR SYSTEM
A type of line-switching system which uses cross-bar switches.

DL (Duplex Link Unit)
Operates under the control of the HC unit to connect or disconnect the individual links for conversation between two stations, changing over from the hands-free conversation to the full duplex conversation mode, or vice versa.

DUPLEX CONVERSATION SYSTEM
At any two stations in a Private Branch Exchange (PBX) system and an Intercom System mutual outgoing/incoming and private conversation are possible.

DUPLEX TRANSMISSION
Simultaneous two-way independent transmission in both directions. Also called full-duplex transmission.

FLAG
Any of various types of indications used for identification, such as a work mark, or a character that signals the occurrence of some condition, such as the end of a word.

FREQUENCY SHIFT KEYING (FSK)
A form of frequency modulation used especially in telegraph and facsimile transmission, in which the modulating wave shifts the output frequency between predetermined values corresponding to the frequencies of correlated sources. Abbreviated FSK.

HALF-DUPLEX TRANSMISSION
A transmission system in either direction but not both directions simultaneously.
Also called semi-duplex transmission.

HC (Highway Control Unit)
Turns on and off the time division switches at the LM, DL and SG units according to instructions received from the CP unit. The link busy indicator lamps on the front panel display the number of the link in use.

HIGHWAY
A common path for conveying multiplexed PAM signals in time-division switching system.

LINK
A connection between two stations.

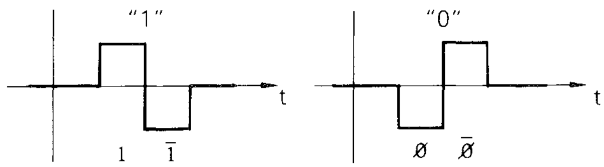
LM (Line Modem Unit)
Comprises a modulator to transmit signals from the stations to the speech link, a demodulator to send out signals from the speech link to the station, a dial receiver and a scanning circuit that scans the station "Privacy ON/OFF" conditions and "Handset ON/OFF" conditions. Up to 8 stations can be connected to one LM unit.

OC (Output Control Unit)
Stores temporarily the output data from the CP unit and distributes them to each unit.

PARALLEL DATA
The data transmitted or processed parallelly and simultaneously, or constituted of bits arranged in parallel at a time. To transfer 8-bit data parallelly, for instance, 8 lines are necessary.

PARALLEL TO SERIAL CONVERSION
Opposite to serial to parallel conversion.
See "SERIAL TO PARALLEL CONVERSION".

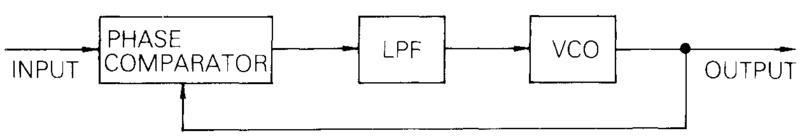
PHASE ENCODING
In general, a method to record binary data on magnetic recording media. The bit "1" or "0" is recorded by the inversion of magnetic flux having different phases, as shown below.



For instance, the change of magnetic flux from negative to positive is recorded as "1", and the change from positive to negative as "0". This method applies to data transmitting and receiving system in EXES-6000 system.

PI (Paging Interface Unit)
Provides paging functions of PA paging and station paging. It sends out both paging voice signals and relay Make signals for power remote control of the PA amplifier.

PLL (Phase-Locked Loop) CIRCUIT
Comprises a phase comparator, an LPF (Low-Pass Filter) and a VCO (Voltage Controlled Oscillator). The phase comparator compares the phase of input signal with that of the output signal from VCO, then automatically controls the VCO so that both phases, or both frequencies, coincide.



SA (Station Paging Assignment Plug/Unit)
Distributes each of paging outputs to each LM unit.
SA plug is used in EX-610/620, and SA unit in EX-630.

SCAN STROBE SIGNAL

A control signal by which the CPU reads the dial data on the dial data bus at an optimum timing when the CPU scans the data.

SERIAL DATA

The data in which bits are transmitted one after another when the data is processed or transferred. The bits constituting the data are arranged in serial at instants. So they can be transmitted on a signal line.

SERIAL TO PARALLEL CONVERSION

In general, a CPU processes parallel data, while a transmitting device or a simple terminal equipment processes serial data. When the data from the transmitting device or terminal equipment is transfered to the CPU, the serial data is necessary to be converted into the parallel data. This is called Serial-To-Parallel Conversion.

SG (Signal Generating and Distributing Unit)

Composed of 8 kinds of signal generators (calling, privacy/disconnected, busy, dialing, zone paging, all call, priority/executive priority, registration confirmation/call holding/mic-off) and distributors. This unit distributes under HC control the required service signal tones to the individual links.

SIDE TONE

The sound of the talker's own voice heard from his station handset.

SPACE-DIVISION SWITCHING SYSTEM

A type of switching system in which a speech signal is transmitted by spatial branched out connection line.

SPEECH PATH

In the multiplexed carrier system or radio system, a line on which voice signal transmitting is possible.

TI (Tie-line Interface Unit)

When tie-line system is built for connecting 2 or 3 exchanges, this unit transmits and receives the audio signals and dial data among the exchanges.

TIME-DIVISION SWITCHING SYSTEM

A type of electronic switching system in which input signals on lines and links are sampled cyclically, and each active input is associated with the desired output for a specific phase of the period.

VCO (Voltage Controlled Oscillator)

A oscillator whose oscillating frequency is varied with applied voltage.

WORD

A symbol sequence which stores or processes it as a unit, and has not less than one meaning.
For DR unit (DR-B61); 1 word = 16 bits.

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