

# THE BELL SYSTEM TECHNICAL JOURNAL

DEVOTED TO THE SCIENTIFIC AND ENGINEERING  
ASPECTS OF ELECTRICAL COMMUNICATION

---

Volume 50

February 1971

Number 2

---

Copyright © 1971, American Telephone and Telegraph Company

## The *Picturephone*<sup>®</sup> System

### Foreword

The first public demonstration of two-way video telephony took place on April 9, 1930, when representatives of the press were shown a system operating between the Bell Telephone Laboratories building at 463 West Street and A.T.&T. Co. headquarters at 195 Broadway in New York City.\* Dr. Herbert E. Ives, then Director of Electro-Optic Research for Bell Laboratories, said of this event, "The latest development to be demonstrated is that of two-way television as an adjunct to the telephone."<sup>1</sup> The following day a New York City newspaper reported:

"Yesterday we saw a much more highly developed form of television demonstrated by the Bell Telephone Laboratories. It was two-way television. We sat in a booth at No. 195 Broadway and conversed with . . . [a person in another] booth at the Bell Laboratories. . . . Each was visible to the other, there being no telephone mouthpiece to mar the image. The speech was very clear. An inoffensive blue light was shot across the face of the speaker from the camera's eye and picked up

---

\* This demonstration of a two-way system had been preceded by a demonstration of a one-way system between New York and Washington, D. C., on April 7, 1927.

by other batteries around the booth. And yet this marvelous invention is still in the laboratory stage, according to the Bell engineers."<sup>2</sup>

The interest in video telephony continued after Dr. Ives' demonstration, but this was a case where feasibility was demonstrated long before the technology became available to transform the idea into a practical reality. That transformation depended, as the articles in this issue show, on technology developed primarily during the 1960s, although it was apparent late in the 1950s that the necessary technology was not far away.

Dr. Ives understood clearly that his was just a demonstration of feasibility, but once the possibility of bringing people face-to-face in long distance communications had been demonstrated, it hardly seemed reasonable that it would not in due course be brought out of the laboratory into commercial service. Nor does it seem unreasonable to expect that *Picturephone*<sup>®</sup> service will eventually become an accepted, normal mode of person-to-person communication and that, with its adjuncts, it will give us new means for communication between people and machines. And just as telephone service—and the network that grew up to provide it—changed the pattern of our lives and the conduct of our business, so we can expect *Picturephone* service and its network to have impacts that we can only begin to appreciate.

This special issue of *The Bell System Technical Journal* describes *Picturephone* service as it exists today. As it becomes available to more and more people, new uses will be provided. We are already looking forward to methods of providing high-resolution images of graphic material, and to *Picturephone* service in color as a logical evolutionary step. Beyond these will be other steps to make the service more economical and to make the service and its network more useful.

The articles in this issue have been written by some of the people responsible for making *Picturephone* service a reality. The issue is dedicated to all of the many people without whose contributions it would still be in the laboratory stage.

JACK A. BAIRD

#### REFERENCES

1. Ives, Herbert E., "Two-Way Television," Bell Laboratories RECORD, 8, No. 9 (May 1930), pp. 399-404.
2. The New York Daily Mirror, April 10, 1930.

# The Picturephone<sup>®</sup> System:

## The Network

By IRWIN DORROS

(Manuscript received June 8, 1970)

A Picturephone<sup>®</sup> network will take maximum advantage of existing telephone equipment. To provide service, a new videotelephone set will be added to a Touch-Tone<sup>®</sup> telephone installation. Video access lines (loops) to the serving central office will be provided using additional regular wire pairs, appropriately equalized. Arrangements are being provided for extending PBX and key telephone customer switching arrangements to have a Picturephone capability. Central office switching will be accomplished by auxiliary video switches under the control of ordinary switching machines. Short-haul trunks will be provided on ordinary trunk cables equipped with equalizers as for loops. Long-haul trunks will make use of existing and planned systems, with additional equipment needed only at the terminals.

The network was designed primarily for face-to-face communications, but two additional services are being offered at the outset: data communications at 460.8 kb/s and interfacing arrangements for access to a customer's computer. Other new services are in the offing.

### I. INTRODUCTION

*Picturephone* service adds a new dimension to telephone communication—that of sight. This face-to-face communication capability will be realized by taking maximum advantage of the existing nationwide telephone network.<sup>1</sup>

Although the advantage of more complete communication between people with *Picturephone* service is readily apparent, the switched *Picturephone* network will be useful in additional ways. (i) Simple graphic material, such as line drawings will be displayed and, if higher resolution is desired, slowly scanned images will be transmitted. (ii) The network will be used to communicate with a computer. High-speed data will be transmitted between business machines in the megabit-per-second range. (iii) Moreover, *Picturephone* service will be used to communicate among groups of people in conference rooms or by a

number of individual persons at many different locations. (iv) It will even be used for surveillance of events at distant locations. Some of these additional uses will be possible at the outset of service; the rest will come later. There will surely be more applications, some yet to be thought of.

Important as these additional uses might become, the *Picturephone* network is expected to be used predominantly for face-to-face communication in much the same way as the telephone network is used predominantly for voice communication. Further, each of the additional uses, while important to a particular segment of the market, requires its own set of system parameters to match the system to the application best. There is no single system that will satisfy all applications of video communication, if cost-utility factors are considered. For these reasons, initial emphasis has been placed on designing the network for face-to-face performance. The other applications were kept in mind and those which did not seriously compromise the performance of the system for face-to-face communications have been accommodated.

The *Picturephone* network will complement the telephone network. That is, existing services will remain in place; telephones will continue to be used for setting up calls; card-dialers, speakerphones, and other extra telephone features will be retained; billing-recording procedures will not change; and customers will choose between a regular voice telephone call and a *Picturephone* call when they originate a call. This so-called vertical extension of existing services will take maximum advantage of existing equipment.

The first customers will be from the business community. This observation has affected the physical design of the *Picturephone* set, placed special emphasis on the use of customer switching equipment, and guided the design of the relatively short access lines (loops) which connect most business customers with the serving central office.

The articles in this issue cover the initial design of the systems of equipment that will provide *Picturephone* service. Innovations that will add to the initial service capability and reduce cost are already in the design stage and more are being studied. Future articles in this *Journal* and elsewhere will report on new improvements as they emerge.

This article presents an overview of the systems; the next article reports on system parameters and how they have come about. The remainder of the issue is organized by major communication system functions: stations, transmission, central office switching, customer switching, and finally, data and computer display equipment.

## II. THE SERVICE, THE CUSTOMER, AND THE STATION

The most important advantage of *Picturephone* service is that the user will be able to *see* as well as *hear* the person with whom he is talking. Moreover, to simulate as closely as possible the naturalness of a face-to-face conversation, the customer's hands remain free as in a normal conversation. This feature is provided by the addition of a microphone and loudspeaker. However, a telephone handset can be used when audio privacy is desired.

The equipment required at the customer's location is packaged in four parts, three of which are shown in Fig. 1.<sup>2</sup> The picture *display unit* contains the camera, picture tube, loudspeaker, and complex electronics to drive them. A small *control pad* contains the user controls and the microphone. These and any standard *Touch-Tone* telephone are easily accessible in front of the user. A service unit, containing the power supply, logic and control circuits, and transmission equalizing circuits, is mounted out of sight in any location up to 85 feet away. Normal lighting is adequate.

The *Touch-Tone* telephone is used for *Picturephone* service because there are 12 symbols available rather than the 10 symbols provided by rotary dials. The 12th button, labeled #, in the lower right-hand corner of the *Touch-Tone* dial, is used to designate a *Picturephone* call. The customer simply depresses this button and then, in most cases, dials the number of the station he is calling. No new numbering plan is necessary, since the switching will recognize the symbol # and control the equipment to make the video connection.

In cases where telephone company operator assistance is required on a *Picturephone* call, the symbol # is depressed before the digits normally used to reach the operator. The operator responds on an audio-only basis. An appropriate fixed image, however, will appear on the customer's screen. Initially, the 3CL switchboard will be equipped for *Picturephone* operator service; the Traffic Service Position System (TSPS)<sup>3</sup> will be added later. Recorded announcements, such as "no such number" or "your call didn't go through," will also be provided on an audio-only basis.

Incoming *Picturephone* calls are identified by a distinctive ring created by a new tone ringer in the display unit. If the customer has a key telephone, the key corresponding to the called line lights red to identify a *Picturephone* call and white for a voice-only call. The call is normally answered in the hands-free mode by operating a switch on the control pad.

Normally, the *Picturephone* customer sits about three feet away

from the display unit. He sees the other person on a  $5 \times 5\frac{1}{2}$  inch ( $12.5 \times 14.0$  cm) screen. The images are displayed 30 times per second. Each display, or frame, consisting of 251 active scanning lines, is divided into two fields. The odd lines are first displayed in one field and then the even lines are interlaced in the succeeding field. This technique makes best use of the one-MHz transmission bandwidth, as is described in detail in the article on system standards.<sup>4</sup>

An additional provision permits the display of simple graphic material. A mirror, built into the display unit, is flipped out in front of the camera lens. Graphic material is placed beneath the mirror on the tabletop in front of the display unit and the camera focuses on the plane of the table.

Lines equipped for *Picturephone* service may also have voice-only extensions, such as a secretary's pickup. Although video calls can be



Fig. 1—The station installation, showing three of the four components: the existing telephone, a display unit, and a control pad. (The service unit is not shown.)

answered on such an extension, the caller would not see anyone until the call is picked up on a *Picturephone* set.

Full video service can be supplied to a PBX attendant if the customer wishes. As a further option, a fixed image, such as a company trademark, can be displayed to the caller while the attendant handles the call. In each of these situations, however, the communication system sets up a full two-way video circuit, and hence the customer will be charged at *Picturephone* rates.

Business communication services today range from simple direct lines to more complex PBX and Centrex arrangements. PBX service features include dial intercommunication and attendant service. In the case of Centrex, the service features include direct inward dialing to telephones and identified outward dialing. To assure continuation of the availability of these existing business services, new *Picturephone* key telephone units offering pickup, hold, and intercom service will be provided. Business customers who currently receive their telephone service from 701 and 757 PBXs, No. 101 ESS, or No. 5 crossbar systems, will be able to add *Picturephone* service and retain all of their major PBX and Centrex features. *Picturephone* service with other new PBX systems and with the No. 1 ESS system will be introduced at a later stage of the service.

### III. LOCAL AND LONG-HAUL TRANSMISSION

The video signal leaving the *Picturephone* service unit of a station is a line-by-line electrical analog of the luminance of the scanned image interspersed with appropriate horizontal and vertical synchronizing pulses. The higher-frequency components of the luminance signal are pre-emphasized to approximately match the expected composite noise encountered in transmission. This technique improves the signal-to-noise ratio after de-emphasis at the distant station. The pre-emphasized time waveform is illustrated in Fig. 2.

*Picturephone* service requires no modification to the existing two-wire telephone loop; conventional calls and the voice portion of *Picturephone* calls use these wires. Two additional pairs of wires in standard telephone cables are assigned for the video signals—one pair for transmission in each direction. To compensate for the attenuation at video frequencies, equalizers are inserted at about one-mile intervals along the additional pairs. The ON-OFF switch-hook signals, *Touch-Tone* dialing signals and ringing signals, as well as the voice portion of *Picturephone* calls, are all transmitted over the voice pair.

The basic local arrangement for direct lines is shown in Fig. 3.

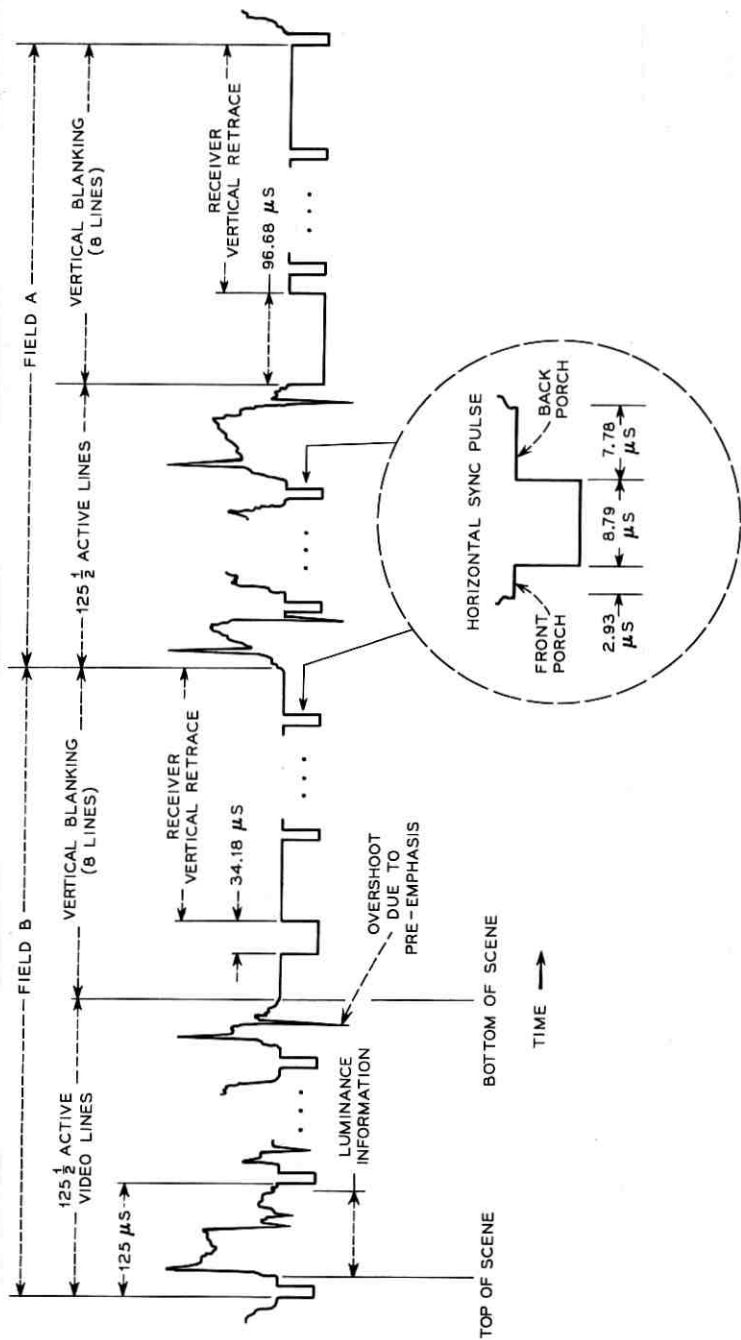


Fig. 2—Composite video signal. (Only video, not sync pulses, is pre-emphasized; line rate, 8 kHz; field rate, 59.93 Hz; video positive on top, as shown.)



More elaborate multistation arrangements on the customer's premises are served by similar loop arrangements. For PBXs, such six-wire loop arrangements serve as PBX trunks with the voice pair used exclusively for *Picturephone* calls.

Similarly, trunks to distant switching offices use six wires in much the same fashion. For the initial years of service, the equalizers in loops and trunks will be of the same design. Later, new trunk equalizers and loop equalizers, each with increased capabilities, will be available. The primary improvement in the later equalizers will be the capability to automatically compensate for cable transmission variations due to temperature changes. This will greatly increase the distance range of analog transmission in a local area. The mileage ranges are given in the article on the transmission plan.<sup>5</sup>

For transmission beyond the local area, which may initially be a business district but will later be an entire metropolitan area, the analog *Picturephone* signal will be encoded in digital form. This transformation involves sampling at the Nyquist rate of about 2 MHz and then encoding the amplitudes of the *differences* between successive samples into 3-bit binary codes. The bit rate is thus about six megabits per second. The equipment at one terminal that encodes in one direction and decodes in the other is called a *codec*.

Digital transmission is desirable because essentially all of the impairment in the process occurs due to quantization errors in the coding and the corresponding decoding (coding noise). Impairments can be controlled by an appropriate choice of the coding algorithm. Once encoded, the digital stream is transmitted, with regeneration at appropriate points in the path, in such a way that the received stream is

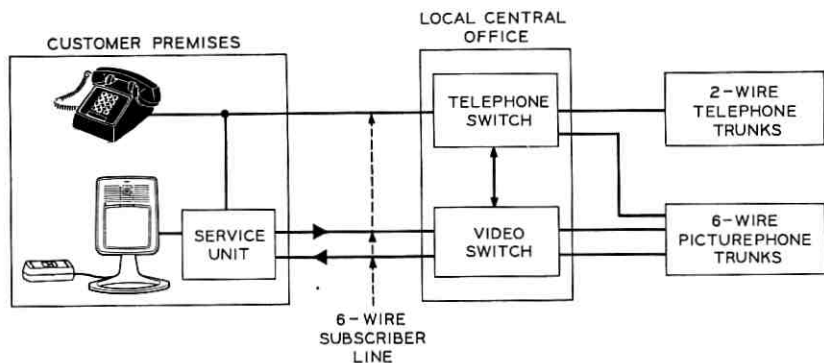


Fig. 3—Basic local arrangement.

nearly identical to the transmitted stream regardless of the distance transmitted.<sup>6,7</sup> The digit errors and the timing jitter introduced in transmission are controlled so as to have insignificant effect.

A switching and transmission plan has been devised which limits a signal in a complete connection to a single encoding and decoding to avoid an accumulation of coding noise. The signal stays in analog form in the local area. Once encoded for longer-haul transmission, it stays in digital form until it reaches the local area of the called station. Thus, the signals are switched in analog form in the local areas and in digital form in the long-haul plant.

In the initial years of *Picturephone* service, voice and signaling information will be transmitted on conventional carrier-derived voice-frequency channels. Later, as newly developed equipment enters the field, voice and signaling information will be converted to digital form and multiplexed together with the video signal into a composite bit stream.

The basic line rate for encoded *Picturephone* signals, with or without multiplexed voice and signaling, is 6.312 Mb/s.<sup>8</sup> The T2 digital transmission line, soon to be introduced, transmits at a 6.312-Mb/s rate and is, in fact, the reason for selecting this particular signal encoding rate. The T2 line will operate up to several hundred miles over wire pairs equipped with regenerative repeaters.

Digital transmission systems now being developed for a broad range of communications will ultimately take over the long-haul transmission of *Picturephone* signals. The first system to transmit digital bit streams for long-haul transmission will be the TD-2 microwave radio relay system. A pair of radio channels, one in each direction and each having a 20-MHz bandwidth, will be equipped to transmit 20.2-Mb/s streams carrying three coded *Picturephone* signals. Any or all of the ten pairs of radio channels on a TD-2 system may be so equipped.

The L-4 coaxial cable carrier system will also be equipped to transmit bit streams. In this case, a 13.29-Mb/s stream, carrying two 6.312-Mb/s signals, will be transmitted in place of one of the six mastergroups of voice channels (600 such channels per mastergroup) on a pair of coaxial units in a cable. Any or all of the six mastergroups on a pair of coaxials may be so equipped. Thus the TD-2 and L-4 systems, already in existence, will meet the long-haul needs of *Picturephone* channels for the first few years.

The TD-3 system, a solid-state successor to TD-2, will also be capable of carrying the 20.2-Mb/s pulse stream per radio channel. The L-5 system, now in development, will be equipped to handle 13.29-Mb/s pulse streams in place of a mastergroup, as in the L-4 system.

Higher bit rates, carrying more *Picturephone* signals in place of several mastergroups, are also contemplated on L-5 for greater efficiency. Still later in time, other coaxial systems and a new waveguide system will carry *Picturephone* signals with still greater efficiency.

Current research has shown good prospects for reducing the required bit rate from 6.312 Mb/s to significantly lower rates by taking advantage of the inherent frame-to-frame redundancy.<sup>9</sup> This is a future prospect, however, and will not be available in the initial system.

The picture that is finally viewed at a *Picturephone* station contains impairments introduced by each part of the built-up connection. The end-to-end impairments are controlled by holding each part of the connection within specified limits. Each of these has been assigned a numerical end-to-end maximum value with a specified portion allocated to stations, loops, trunks, etc.<sup>5</sup>

#### IV. CENTRAL OFFICE SWITCHING

At the local central office, the voice pair is connected to the existing telephone switch in the conventional way. The video pairs, however, are connected to a separate four-wire video switch which is under the control of the existing telephone switching machine.

As *Picturephone* service is first offered, No. 5 crossbar switching machines will be modified to switch video calls with up to a maximum of 3200 line appearances and 400 trunk appearances. The capability of providing *Picturephone* service will be added to No. 1 ESS later. *Picturephone* service for customers served from step-by-step, panel, or No. 1 crossbar offices will be routed to a nearby No. 5 crossbar or, later, to a No. 1 ESS office.

Whenever a customer dials a call, the common control equipment in the switching machine recognizes the digits and establishes a talking path between the lines and trunks. For voice-only calls, the existing two-wire telephone switch makes the connection. When the special prefix, #, is dialed, indicating a *Picturephone* call, the talking path is established through the two-wire telephone switch, and a path is established simultaneously through the four-wire video switch to the trunk side of the switching matrix. There, the audio and video paths form a composite six-wire appearance. For intra-office calls, the six-wire *Picturephone* signal returns through the switches to another line. For calls to a distant central office, however, the path is established over a six-wire *Picturephone* trunk. The audio portion of the six-wire trunk is dedicated to *Picturephone* traffic and is never used for voice-only telephone traffic.

What has been described above is the means for switching *Picturephone* signals in *analog* form. *Picturephone* signals will be switched in this way in the local serving central office. Initially, some of these same offices will also provide the toll switching function, also in analog form. The digital encoding and decoding are carried out within the interoffice trunks. Later, the bit streams will be switched at toll centers. Initially, No. 5 crossbar systems will function as the toll centers, and later, electronic toll switching machines will assume this function.

There are some cases where it will be economically attractive to add a switch remote from the serving central office. The large business customer having a number of *Picturephone* stations located a few miles from the serving central office is such an example. In this case, a wideband remote switch (WBRS) can be provided that acts as a concentrator to reduce the required number of video links between the customer and the serving central office. Only a portion of the control circuitry is at the WBRS. Control circuits in the central office direct the switch over regular telephone pairs, which serve as the control link.

Figure 4 illustrates the *Picturephone* switching hierarchy. Note that the toll center may switch either analog or digital signals. Actually, although not shown, a toll center or a primary center may be equipped to switch both analog and digital signals in separate switch matrices, with codecs in the interswitch paths. The choice is based on local trunking needs and economics. Figure 4 also illustrates a use of a WBRS and a wired-through connection at a local telephone end-office not equipped for *Picturephone* switching. Although not shown, trunks may exist between any pair of switching offices in the diagram (except that the WBRS connects only to its homing office).

An illustrative local area configuration is shown in Fig. 5. Analog loops and trunks are shown with local and toll analog switching. Also shown is how they interface with digital trunks and bit stream switching.

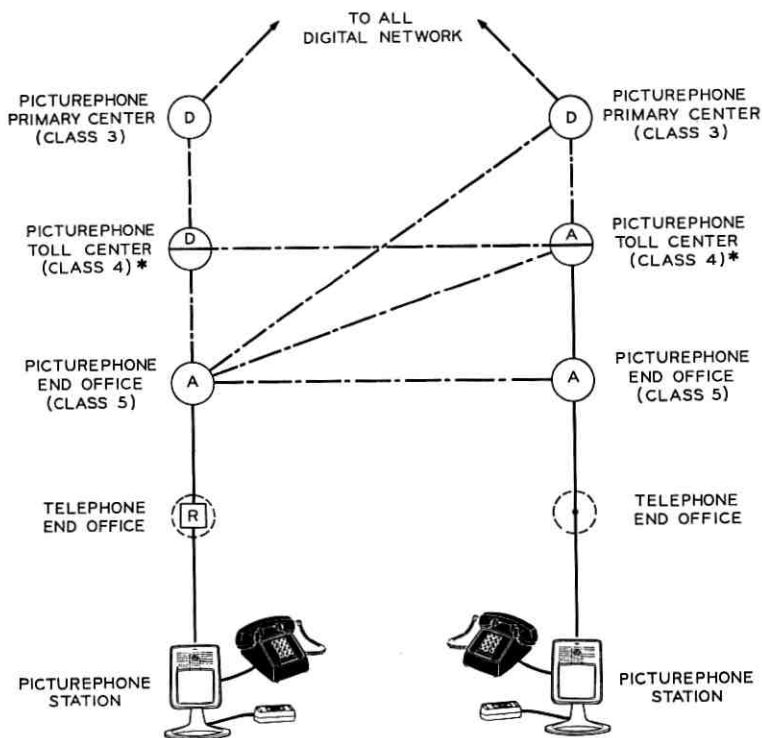
## V. OTHER SERVICES

*Picturephone* service will provide a one-MHz switched network, which will be capable of providing more than "see-as-you-talk" communications. Two obvious and important additional capabilities are being implemented from the start: digital data transmission and interaction with a computer.

For initial service, the *Picturephone* network will transmit and re-

ceive synchronous digital data through a data set at the station, much like *Data-Phone*® data communications service on the voice network. Initially, transmission of digital data is limited to a rate of 460.8 kb/s by the coding processes in the trunks, but improvements are planned to extend the maximum bit rate to 1.344 Mb/s.

The capability of interacting with a computer from a *Picturephone* set is provided by means of a special display data set, which appears to the computer as a voice-frequency time-sharing access line and to the switched network as a regular *Picturephone* set. The special display data set communicates with the computer in the format of a voiceband data set, such as those currently used in time-shared sys-



\* The toll center may be either A or D. The two cases shown are illustrative.

Fig. 4—*Picturephone* switching hierarchy. [ R wideband remote switch (WBRS); -·-·- analog or digital transmission; — analog only; - - - - digital only; D = digital switch; A = analog switch.]

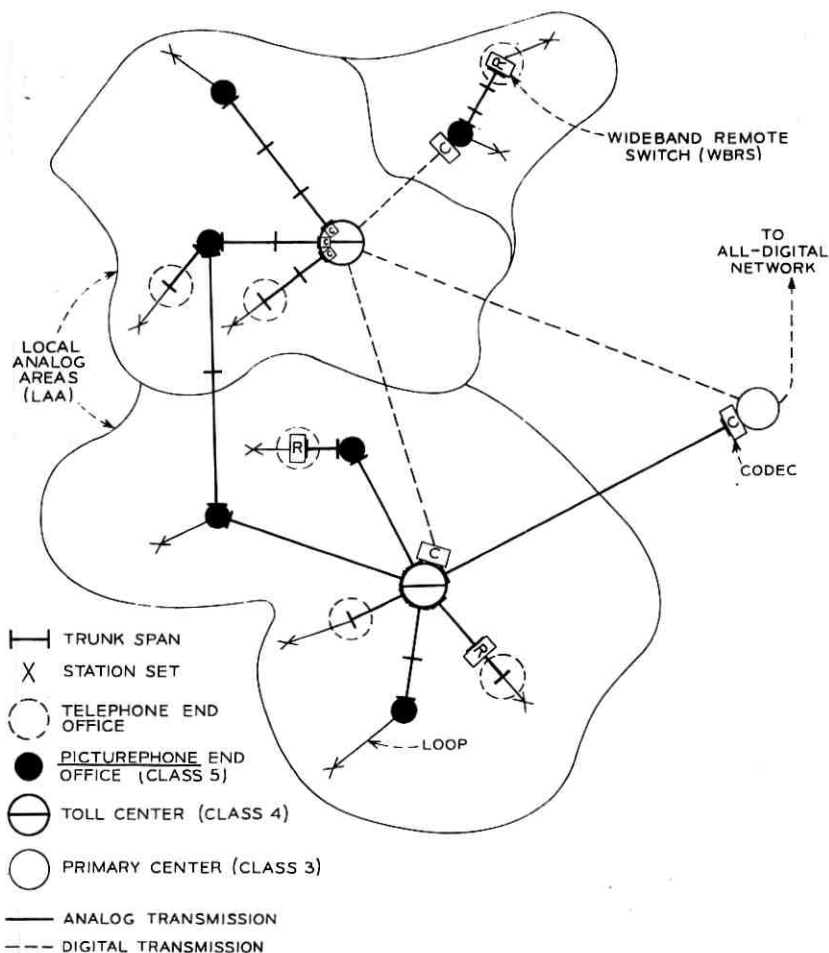


Fig. 5—Illustrative local area configurations.

tems. The data set generates an analog video signal from ASCII-coded\* computer information for display on the *Picturephone* set. The data set also converts the tone signals from the *Touch-Tone* telephone to ASCII code to interrogate the computer. Thus, a user can dial a regular *Picturephone* number, reach a customer's computer, and interact using the user's display tube and the *Touch-Tone* dial with no additional

\* American National Standards Code for Information Interchange (ANS X 3.4—1968).

equipment at the station location. An optional extension of this capability, using an ASCII keyboard, is under way.

Other uses will come in time, many by customers interconnecting their own station arrangements. It should be made clear that although the *Picturephone* signal occupies a nominal one-MHz bandwidth, the switched network cannot be used to transmit arbitrary one-MHz signals because of power constraints, spectral distribution constraints, and signal processing constraints in the digital coding. The connecting arrangements for customer owned equipment are described in an available technical reference.<sup>10</sup>

One other service that should be mentioned here is multipoint conferencing, involving many conferees, each at a different location. A Voice Operated Video Conference System (VOVCOS) is planned for introduction a few years after the initial service offering. In this system, a voice-actuated switch causes the picture of the person talking to be seen by all other conferees whose sets are connected to the special VOVCOS conference bridge. Such a system is necessary to extend the voice conferencing services now offered, to the *Picturephone* network, where the new question of who sees whom arises.

Two more obvious extensions of the service are still in the active research stage: full color service and a high-resolution graphics capability. In the latter, high resolution will be obtained within the same bandwidth by sacrificing response to animation. It is expected that in time, these service capabilities will be added.

## REFERENCES

1. Bell Laboratories RECORD, Special Issue on *Picturephone*® Service, 47, No. 5 (May/June 1969).
2. Cagle, W. B., Stokes, R. R., and Wright, B. A., "The *Picturephone*® System: 2C Video Telephone Station Set," B.S.T.J., this issue, pp. 271-312.
3. "Traffic Service Position System," special issue of the B.S.T.J., 49, No. 10 (December 1970), pp. 2417-2734.
4. Crater, T. V., "The *Picturephone*® System: Service Standards," B.S.T.J., this issue, pp. 235-269.
5. Brown, H. E., "The *Picturephone*® System: Transmission Plan," B.S.T.J., this issue, pp. 351-394.
6. Oliver, B. M., Pierce, J. R., and Shannon, C. E., "The Philosophy of PCM," Proc. IRE, 36 (November 1948), pp. 1324-1331.
7. Kelley, R. A., and Andrews, F. T., "Challenge in Transmission," Science & Technology, No. 76 (April 1968), pp. 55-62.
8. Davis, J. H., "A 6.3 Megabit Digital Repeated Line," 1969 International Communications Conference, Boulder, Colorado, Convention Record, p. 34.9.
9. Mounts, F. W., "Conditional Replenishment: A Promising Technique for Video Transmission," Bell Laboratories Record, 48, No. 4 (April 1970), pp. 110-115.
10. Bell System Technical Reference, "*Picturephone*® Connecting Arrangement-PVF, Interface Specifications," September 1970.





# The Picturephone® System:

## Service Standards

By T. V. CRATER

(Manuscript received September 25, 1970)

*We describe in this paper the fundamental requirements of a visual telephone service. These lead to a basic physical configuration of the station equipment. The picture standards are chosen to provide the visual adjunct at no greater cost than is necessary to secure most of the possible enhancement of direct conversation. Transmission standards are established with the objective of limiting to an acceptable range the difference in quality between the image as viewed at the originating station and as received over the longest connection possible in the network.*

### I. THE FUNDAMENTAL REQUIREMENTS FOR Picturephone® SERVICE

The previous paper<sup>1</sup> described Picturephone service as primarily designed for face-to-face conversation. Put very simply, we are interested in improving telephone communication by making it possible for the two parties to see as well as hear each other. The notion of seeing what is going on at some distant point by means of electrical signals transmitted over wires is as old as the telephone itself.<sup>2</sup> The early work on television was in fact directed toward a wired point-to-point visual service.

Although television developed as a broadcast service, it brought into being most of the technology necessary to make visual telephone service technically feasible. Economically, however, the techniques developed for television are in many respects inappropriate for a face-to-face service. The subscriber's station must be equipped with a camera, a receiver and voice equipment, packaged for great durability and safety, and manufactured initially in relatively modest quantities; this suggests an equipment cost several times that of a home television receiver. The subscriber needs a private two-way channel to the nearest switching machine, but short-distance cable systems used for television today are costly, especially for the initial era of very light development

of the service. Finally, long-distance television transmission today, requiring usually an entire microwave radio channel and attentive maintenance, is prohibitively costly for a face-to-face service to achieve wide public acceptance.

The key aspects of the design and implementation of a *Picturephone* system therefore pivot around questions of cost. Succeeding papers of this issue will describe the means chosen, in each part of the system, to minimize initial and maintenance costs while meeting standards of service. We have set those standards so that the quality will be no better than is really needed, particularly in the more cost-sensitive aspects. In doing this we have tried to ensure that the standards chosen are fully adequate for a face-to-face service in the foreseeable future, since it may be extremely costly to upgrade them later. In the beginning, the usefulness of *Picturephone* service will be limited by its cost; eventually, it will be limited by its inadequacy relative to something else. The choice of the standards will determine the length of its era.

### 1.1 *The Experimental Basis for the Standards*

The subjective factors involved in establishing video standards are sufficiently complex that a comprehensive theory is not available. Thus the weighting of economic and technical factors tends at every step to require subjective testing as well. Moreover, a set of standards, once tentatively established, must be tried under realistic conditions, to test the overall effectiveness and utility of the service they define. It is consequently the author's privilege to report on the results of a series of studies, test programs and trials, extending over a period of a decade and a half, and supported by the efforts of dozens of people. In what follows, we attempt to abstract an outline in a logical sequence of the bases for the separate but interrelated choices that make up the standards.

It is not possible in many cases to refer to the literature for a description of the test or trial alluded to, since most of the work at Bell Laboratories on visual telephone standards has not yet been published. As a partial substitute, it may be helpful to review here briefly the major study and test programs which have contributed to these standards.

The work may be considered to have started in 1954 with an investigation by W. E. Kock, F. K. Becker, R. L. Miller and others of the possibilities of a visual adjunct consisting of a series of snapshots of the distant party, produced at the rate, in one realization, of one every two seconds.<sup>3-5</sup> Although this approach was not pursued beyond the

demonstration of its technical feasibility, the work excited new interest in the notion of a visual adjunct to telephony. In 1956 A. D. Hall, with J. D. Gabbe, H. Cravis, and later M. W. Baldwin, Jr., J. E. Abate, the author and others, began an engineering study of the economic and subjective factors involved, with the objective of establishing requirements for station equipment. At about the same time W. T. Wintringham, with R. L. Eilenberger, R. L. Miller and later R. C. Brainard, F. W. Mounts, E. F. Brown and others, began experiments with methods of efficient digital encoding of the video signal. This group also investigated standards in order to characterize the signal which might be encoded. Their work has continued to influence standards at each step.

By 1960 an initial service definition was available, calling for a 0.5-MHz video signal which could be transmitted and switched at acceptable costs. A station instrument development was initiated by a group under the direction of L. A. Meacham. The resulting instrument,<sup>6</sup> referred to as the Mod I *Picturephone* station set, was installed in an 8-station switched network as an exhibit at the New York World's Fair in 1964.<sup>7,8</sup> Visitors chosen by a random-sampling procedure were given an opportunity to make calls between booths of the exhibit, and their reactions to the experience, in respect to the utility of the service and the appropriateness of various features of the set design, were elicited by means of a questionnaire. Observations were also made of the learning process as visitors used the equipment, to determine whether any human-engineering improvements were indicated.

In 1965 Mod I sets were installed in 28 Bell Laboratories offices at Murray Hill and Holmdel, New Jersey, with switching and transmission equipment of preliminary design, and with telemetry equipment to record the duration of each phase of every call and all operations of electrical controls. Questionnaires were also used to determine subjective reactions. A separate trial was conducted at offices of the Union Carbide Corporation in New York and Chicago during the same year, in which 35 stations were installed and statistics of both local and intercity traffic were recorded.<sup>8</sup>

On the basis of results of these trials and of continuing laboratory experiments and economic studies, standards for an improved station set, referred to as Mod II, were established in 1966. This station set, described in another paper in this issue,<sup>9</sup> was tested in a trial in the Westinghouse Corporation's offices in New York City and Pittsburgh in 1969. It is also in regular experimental operation in the Murray Hill-Holmdel network, now expanded to include offices of the American Telephone and Telegraph and Western Electric Companies in New

York City. With modifications resulting in part from these tests, the Mod II set has been employed in commercial service since July 1, 1970, when *Picturephone* service was instituted in the Golden Triangle area of Pittsburgh, Pennsylvania.

### 1.2 *Basic Picture Requirements*

In view of the high cost of television transmission, it would be prodigal to provide the picture at broadcast television standards. The prospect of using fewer scanning lines and less bandwidth is very attractive. With good maintenance of station equipment and of wire transmission systems, picture quality can be kept close to the design objective; much of the possible quality of broadcast television is lost in the radio broadcast path and in poor maintenance of the home television receiver. The reduction of bandwidth which is possible is limited by a built-in human resistance to very fuzzy images and to images which flicker. As will be seen, a minimum bandwidth of several hundred kilohertz is found to be necessary.

A large reduction of bandwidth is possible if the display of the image in continuous motion is abandoned. In the visual adjunct experiments carried out in the years 1954 through 1956 by W. E. Kock, F. K. Becker, R. L. Miller and others, the picture elements were transmitted over a separate telephone circuit and stored at the receiving end until the snapshot was complete, and then displayed while information was accumulated for the next image. This adjunct permitted inspection of the other party, but the disjointed series of facial images, each one displayed while its successor was made, did not provide the enhancement of telephone conversation which occurs with full motion portrayal.

We have chosen to give the system sufficient bandwidth to provide full motion capability, adequate, for example, for lip reading, and resolution sufficient for a life-like image of the face.

Most of the visual enhancement of telephone communication requires a monochrome image only; the elements of it are smiles, frowns, averted glances, broad grins, expressions of shock, dismay, amusement, or sympathy. There are however additional values in the naturalness of an image transmitted in color. These have not been deemed sufficient to justify delaying the service until problems of cost of the color station instrument can be resolved. They do suggest the consideration of transmission capacity in the light of the needs of a future compatible color system.

### 1.3 Basic Operational Objectives

Since *Picturephone* service is to be an extension of telephone service, it must operate in typical telephone environments. To avoid duplication of equipment, visual telephone and ordinary telephone calls should be made from the same station instrument. To simplify operation as much as possible, the station address for *Picturephone* service should be the same as for telephone service, except for a prefix to indicate that a call is to include the visual adjunct.

On the other hand, the user will not want to give up other special telephone services which he may already have. This requires that the applique equipment for the visual adjunct be compatible with any of the many types of *Touch-Tone*<sup>®</sup> telephone station instruments now in use in the Bell System.

Telephone service should not in any way be diminished when the visual channel is added, and there is one respect in which it should be expanded. It is appropriate to supply hands-free speakerphone audio with *Picturephone* service because the handset becomes a barrier in the optical path and detracts from the feeling of presence. The handset, of course, remains available for use when privacy is important or room noise is disturbing. The speakerphone, on the other hand, is available for telephone calls as well.

In use, the subscriber should be able to make either a telephone or a video telephone call to another *Picturephone* station, or a telephone call to any telephone station, using in any case the same telephone instrument. If he elects to make a visual call, the video channel is provided at the beginning and is available throughout; this makes it possible for the system to select an available video channel and check to ensure that it and the called subscriber line are functioning before making the connection. It will not be possible to summon the video adjunct midway through a telephone call or drop it midway through a video call. Since any telephone line may become a *Picturephone* line, it may have telephone-only extensions; visual calls may be originated or received on these, although the picture will not be seen.

We have discussed only the use of the service for face-to-face conversation. It is also suitable for the transmission of graphic information, such as pencil sketches, diagrams, pictures and some printed material. The video display and broadband channel are ideal for interaction with a computer. The network of broadband channels will be useful for the transmission of data at very high speeds. These applications are described in other papers in this issue. The standards, how-

ever, are based on the face-to-face application, and only such minor modifications have been permitted on behalf of other applications as will not significantly affect the cost of the face-to-face use.

These elementary considerations lead to a concept of a station instrument which can be added to the existing telephone set, to operate as simply and with as little disruption of the normal telephone environment as possible. No more transmission capacity need be provided than is necessary to achieve the enhancement which vision adds to face-to-face conversation, without visual strain, nervous strain or discomfort.

## II. CONFIGURATION AND PHYSICAL DIMENSIONS OF THE STATION EQUIPMENT

In setting standards to obtain fully adequate visual images at minimum cost, video parameters such as number of scanning lines, resolution, and the allowable degree of picture impairment of each kind are of primary importance because they determine the costs of transmission systems. These requirements must be evaluated in terms of a rather specific station configuration, with particular respect to viewing distance, picture size, and picture aspect ratio.

To continue with the discussion of standards, we therefore look next at the station configuration. Given the initial requirement that the equipment be designed for use at the close quarters provided by a desk or table, there are some constraints that must be accommodated.

### 2.1 *Imaging the Viewer*

For the parties to the conversation to enjoy a normal visual exchange, they should each remain reliably in view of the other, and they should be able to "look each other in the eye." The first of these needs might be satisfied by a camera arranged to follow the user so as to keep him centered in the field of view. This is presently considered unattractive on technical, economic, esthetic and psychological grounds. Some means is therefore desirable to help the user stay in view. The second ideally requires that the camera be located in effect where the screen is, at about the bridge of the nose of the image.

Both of these needs are quite well satisfied by putting the camera behind a half-silvered mirror which reflects the image of the display tube so it can be seen by the user.<sup>10</sup> The result is to box in the optical path to the screen so that it appears to be at the far end of a short tunnel. In order to see the entire image, the user stays within the camera

field. Eye contact is also very good. A schematic diagram of such a station set is shown in Fig. 1; an instrument using this principle was constructed and tested at Bell Laboratories in 1963 by R. L. Eilenberger.

This arrangement makes it quite difficult for others in the room to see the image. While the resulting privacy is an advantage in some cases, it is a source of frustration when the user wants to introduce a second person to the distant party, or when he wants to demonstrate the service to others. The mirror also requires an increase in bulk. For these reasons the simpler configuration shown in Fig. 2 is preferred for general use. The viewer looks directly at the screen and the camera is located at as small an angle from the screen as possible. The eye contact requirement remains a factor in dimensioning the instrument; this question is discussed further in the paper on the station set in this issue. The eventual development of a station set using the split optical path is by no means ruled out by the standards we have chosen, and such a set would have marked advantages for the user who wants to exclude the distractions of a busy environment.

## 2.2 *The Camera Field of View*

Somewhat related to the choice of the open screen is the question of the field of view. Since the major visual clues in conversation are facial expression and movements of the head, eyes, and lips, the minimum bandwidth would be required for an image of the head only. With the open screen, the conscious effort required of the user to stay in so constrained a position would make him quite uncomfortable. Experiments conducted by J. D. Gabbe and others in 1956 and thereafter and by R. L. Eilenberger and others about the same time,

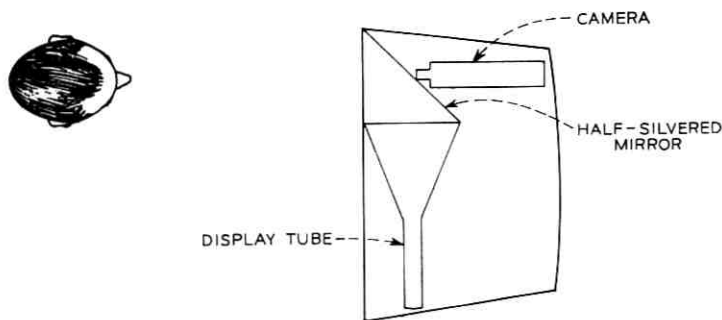


Fig. 1—Station set with coaxial display and camera optical paths—top view.



Fig. 2—Station set with open display.

quickly established the desirability of the head-and-shoulders view with the open-screen configuration. This view not only permits a necessary degree of freedom of motion, but also allows the distant party to see some of the surrounding environment, shows additional visual signals such as hand and arm motions and shrugs, and provides an esthetically pleasing picture. Experience with the Mod I set at the New York World's Fair in 1964 confirmed the advantages of the head-and-shoulders view.

The substantial savings in transmission cost might justify the head-only picture in the early years of the service when costs are paramount. In the long term, however, the provision of the head-and-shoulders view is considered necessary to assure the full adequacy of the service for normal face-to-face conversation.

### 2.3 *Picture Size and Viewing Distance*

The ratio of viewing distance to screen size is closely related to the number of scanning lines and to the horizontal resolution, and in fact to many other picture quality standards. This is because the user wants to be close enough to see all of the useful picture detail but far enough away to be undisturbed by the line structure and other visual effects due to the scanning process. In the visual telephone application viewing distance tends to be the independent variable, leaving picture size and scanning standards to be chosen in the light of the cost of bandwidth in transmission.

Since *Picturephone* service must operate in the relatively small spaces in which the telephone is used, the available viewing distance is limited. For a desk of ordinary size, a distance greater than about 0.9 meter is awkward. Even this is too much for many telephone situations. On the other hand, shorter viewing distances tend to degrade the image obtained by the camera, which must be located as near the display as practicable. Short camera ranges tend to increase the eye contact angle and to introduce distortion effects due to per-



spective. A short camera range also deprives the user of leeway for his normal forward and backward movements. A user who leans forward 0.3 meter from a normal range of 0.9 meter may still transmit an acceptable picture, but the effect of moving from 0.6 meter to 0.3 meter is found to be almost grotesque.

A longer camera range also reduces the height at which the camera must be located, and therefore leads to a more compact set. Since the head is to appear near the center of the picture, the camera must either be located near the level of the head or be tilted upward. Tilting the camera tends to bring overhead lights into the picture, and if carried to extremes, makes walls and bookcases appear to be leaning backward and produces a distorted view of the face. The further away the camera is, the lower it can be placed and still remain within a given tilt-angle limitation.

Because of these factors, confirmed by experience with the Mod I set, a viewing distance of 0.92 meter (36 inches) has been chosen, to place the instrument as far away as possible while keeping it on the desk or table at which the user is sitting. This standard has been retained through the several trials of the two station set models.

The choice of picture size is somewhat more complicated. A primary factor is the desirability of a compact subscriber's set. However, if the user's esthetic or psychological reaction is that the picture as seen at the design viewing distance is "too small," he may tend to compensate by moving forward. In experiments at Bell Laboratories in 1958, R. L. Eilenberger found that at a viewing distance of only 0.66 meter, a picture of about 0.013 square meter (20 square inches) was rejected by test subjects in favor of larger pictures.

An upper limit to the size is imposed by the fact that the desk is also used for other activities. It appears however that esthetic preferences restrict picture size more than desk space might. Mr. Eilenberger found preferred picture heights ranging from 0.14 meter (5.5 inches) to 0.16 meter (6.2 inches) at viewing distances from 0.66 meter (26 inches) to 1.07 meter (42 inches). His tests used a 525-line picture and the height preferences obtained are therefore based on esthetic preferences rather than raster visibility. In tests of pictures with limited bandwidth made at Bell Laboratories, J. D. Gabbe and A. C. Bandini found that with a 245-line picture of 0.20-meter (8-inch) height, subjects considered a 1-meter viewing distance too close, while with a 0.15-meter ( $5\frac{3}{4}$  inch) picture of the same number of lines, the 1-meter distance was entirely satisfactory.

Optimizing the picture at minimum bandwidth leads to a similar

picture height. The picture height should be chosen to make the transmitted information fully available to a person with normal vision. The viewer tends to approach the picture until the scanning structure becomes sufficiently obtrusive that there is no advantage in moving closer. If the distance at which this happens is 0.92 meter (36 inches), the optimum viewing range will coincide with the chosen camera range. It will be shown in Section III that this leads to a picture height of about 0.13 meter (5 inches).

The choice of picture height cannot be entirely divorced from the choice of aspect ratio. Studies of preferred aspect ratio for a head-and-shoulders view, based upon viewing the picture, were not in complete agreement, a 3:4 aspect ratio (width to height) being found in one study and about 1:1 in another. The Mod I station set used the 3:4 ratio, with a 0.11-meter (4-3/8 inch) by 0.15-meter (5-3/4 inch) picture. Trials of this equipment at the New York World's Fair in 1965, at the Union Carbide Corporation in New York and Chicago in 1965, and at Bell Laboratories branches in Murray Hill and Holmdel, New Jersey, in 1965 and later, called attention to the user's problem of staying within the camera field of view. A choice of the 4:3 aspect ratio of entertainment television would solve this problem quite completely, but the bandwidth would then be increased by a factor of 16/9 for a given resolution of the actual user image.

These factors were compromised in the Mod II set by the choice of the 1.1:1 aspect ratio; this frame is more consistent with test results on optimum aspect ratio for viewing, provides adequate freedom for the user and is economical of bandwidth. At the same time, the picture height was reduced from 0.15 meter to the 0.13 meter mentioned above. This reduced the visibility of the line structure. It also reduced the increase in the contribution of the display tube to bulk, from 47 percent to 10 percent, and permitted reduction of the eye contact angle. The Westinghouse and Bell Laboratories trial results with the Mod II design indicate that these dimensions are entirely satisfactory.

#### *2.4 Audio and Controls*

In a video telephone call, the telephone handset is not only a visual obstruction, concealing important parts of the face, but also burdens the hands, interfering with normal gesturing and the manipulation of objects to be displayed. For this reason a microphone and speaker are considered essential for every station. The handset must also be available for those occasions when privacy or background noise makes the speakerphone arrangement undesirable.

Provision for viewing oneself has been found very desirable. Use of this feature is typically high in the first few months as a new user, not yet confident in the use of other visual cues, checks frequently to see whether he is properly framed in the picture. To secure proper framing, it must be possible for the user to adjust both the azimuth and elevation of the camera. Provision for shutting off the outgoing picture is necessary for obvious reasons; it is desirable to provide an electronic means for this rather than require the user to obstruct the camera view. A means of alerting the user, while the station is being rung, that the call is a *Picturephone* call, is also desirable.

These features imply a number of controls which is close to the minimum. Their realization is described in the paper on the station set in this issue.<sup>9</sup>

### III. OPTIMIZATION OF PICTURE PARAMETERS

The choices to be made in selecting picture standards are so numerous, and interact in so many ways with each other and with the cost of providing the service, that it is difficult even to propose a logical study sequence which would lead to a unique set of standards. We have already seen that the picture size cannot be established without some reference to the scanning standards. In what follows, the bases of the major choices will be outlined.

#### 3.1 *Brightness and Field Rate*

In a *sequential* scanning pattern, the picture is scanned line-by-line from top to bottom. The resulting field of horizontal lines contains all of the information to be transmitted about one frame. In a *two-to-one interlaced* pattern, the picture is scanned twice, the second time between the first set of scanning lines, so that two *fields* are required to complete one *frame*. With either pattern, the gross visual effect of a field is that of a moving pulse of light, decaying in accordance with the characteristics of the phosphor while the scanning beam traverses the picture. If the field rate is too low the picture appears to flicker. The lowest frequency at which flicker disappears depends on several factors but varies approximately as the logarithm of the highlight luminance. Under typical television conditions the threshold of perception of flicker at 50 fields per second occurs at a highlight brightness of about 30 foot-lamberts, while at 60 fields per second it occurs at 180 foot-lamberts.<sup>11</sup> In tests at Bell Laboratories in 1966, E. F. Brown found a highlight brightness of about 80 foot-lamberts to be preferred for *Picturephone* service. This would permit a field rate of

56 Hz under typical television conditions, but to allow for variable circumstances it is desirable to establish a somewhat higher field rate.

Interference from power lines at 60 Hz and its harmonics has usually been a factor in the choice of field rate for any video service. Interference at a frequency differing from the field frequency by 10 Hz requires about 12 dB more suppression than one differing by only 0.5 Hz.<sup>12</sup> Recent tests at Bell Laboratories by D. B. Robinson, Jr., show that if the two frequencies are very nearly equal, the suppression required may be as much as 20 dB less than at 0.5-Hz difference. With the advent of solid-state electronics and the development of improved clamping circuits, however, these results are of less significance than formerly.

Both fluorescent and incandescent lights have a fluctuating component at twice the power frequency. This interacts with the camera to produce a flicker if the power and field frequencies are different. Means of mitigating this effect have not been investigated, since flicker at the field rate provides sufficient motivation to retain a field rate near 60 Hz.

To obtain 250 active lines per frame a line repetition rate somewhat larger than  $30 \times 250$  is required because of the need for vertical retrace time. The resulting line repetition frequency is in the neighborhood of 8 kHz. There is no reason why it should not be made exactly 8 kHz, and there may be some future advantage in doing so, in the digital transmission plant, for example, where the sampling frequency for voice signals is 8 kHz. Dividing 8000 by 30 gives a number close to 267 for the total lines per frame, including those lost in blanking. To make it exactly 267 (the odd number is necessary for interlacing), the frame rate has been made 29.9625 Hz.

### 3.2 *Interlaced Versus Sequential Scanning*

Although 60 fields per second are necessary to eliminate flicker, a much lower rate is sufficient for motion. Indeed, under ordinary conditions of room lighting and screen brightness, 30 pictures per second are indistinguishable from 60 if flicker is suppressed, for example, by displaying each picture twice.<sup>13</sup> Two-to-one interlace is therefore suggested. By this means the bandwidth required for a given horizontal resolution, vertical resolution, and field rate is reduced by half. However, the bandwidth saving in this case comes at the cost of a loss in overall subjective quality, because interlace introduces some undesirable visual effects. These are more noticeable in *Picturephone* service than in broadcast television because the angular subtense of

the spacing between lines is larger in the former at normal viewing range.

If all but two of the raster lines are masked, only one is scanned in each field, so that a single line may be seen jumping back and forth at a 30-Hz rate. This effect, called interline flicker, occurs at horizontal brightness boundaries in the picture material and in small areas of high brightness.

If an object in the picture moves up or down at a rate such as to pass one scanning line every sixtieth of a second, the raster appears to break down into the lines of a single field, moving up or down at the same rate. The effect, called subjective line pairing, may be quite striking. In *Picturephone* service, object motion at just the right rate is unusual, although the effect can be invoked voluntarily at close viewing ranges by scanning the screen slowly upward or downward. Momentary eye motions up or down, however, can cause subjective line pairing to occur long enough that the alternate-line pattern emerges, although the apparent motion may not be seen. If the receiver is experimentally turned on its side so that the scanning is vertical, the effect is enhanced and the picture may appear to break up at almost every glance, apparently because involuntary horizontal eye movements are more frequent than vertical ones.

The net result of these effects is to give the appearance of a somewhat "busy" or noisy picture, compared to a sequentially scanned picture. It is appropriate to ask whether the interlaced picture is in fact subjectively better than a sequential picture transmitted over the same bandwidth. Test results indicate that it is, but for face-to-face *Picturephone* applications, not as much as might be expected. In a series of carefully designed experiments, E. F. Brown found that the actual bandwidth advantage was surprisingly low.<sup>14</sup> He used as a reference an interlaced 225-line, 0.13-meter (5-inch) square picture at various brightness levels and determined the bandwidth required for a subjectively equivalent sequentially scanned picture at the same brightness levels. The bandwidth reduction with interlace ranged from 37 percent at 40 foot-lamberts of highlight luminance to only 6 percent at 100 foot-lamberts. At the preferred highlight luminance of 80 foot-lamberts, the reduction under the conditions of the experiment was 16 percent.

Although the advantage is unexpectedly small, it is nonetheless desirable to accept it. As described in the previous paper in this issue, the video transmission from subscriber to central office utilizes telephone wire pairs with equalized amplifiers at regular intervals. The

cost of this link depends largely on the number of amplifiers required, which in turn is closely related to the bandwidth. The possible advantage in sequential scanning is in long distance transmission, where future digital encoding systems may take advantage of the similarity between successive frames to economize on bit rate. These may require less storage with sequential scanning. However, the cost savings in trunks fall short of the additional cost in subscriber loops. We have therefore retained the interlace.

### 3.3 *The Optimum Raster with Interlace*

With interlaced scanning, a minimum of about 250 lines has been found to be necessary for adequate portrayal of the head-and-shoulders image; about six or seven lines then portray the eyes and eight to ten the lips. We want to choose the line spacing so that the user will be able to see all of the detail easily at 0.92 meter but will find the scanning structure annoying at closer range. The appropriate angular subtense, with interlaced scanning, for the distance between centers of adjacent scanning lines, is found to be about 2 minutes of arc. This leads to about 20 lines per cm at 0.9 meter, so that the 250-line picture requires a 0.13-meter height.

The Mod I station set, developed for experimental trials, was designed to hold bandwidth to a minimum. Since the picture was relatively narrow in width, the height was set at 0.15 meter (5¾ inches) to keep the overall size adequate at 0.92 meter (36 inches), but to conserve bandwidth the number of visible scanning lines was kept at about 250. The line spacing then substantially exceeded 2 minutes of arc. Subjective line pairing and interline flicker effects were objectionable. With the change to the 1.1:1 aspect ratio for Mod II, and the reduction of the height to 0.13 meter, the angular subtense was reduced to slightly less than 2 minutes of arc. The resulting picture is more nearly optimized at the design viewing distance.

### 3.4 *Bandwidth and Horizontal Resolution*

With the number of visible lines fixed at about 250, the choice of bandwidth affects only the horizontal resolution. Since the vertical resolution has been optimized for the chosen viewing distance, horizontal resolution better than the vertical would be to some extent wasted. The viewer would have to "look through" the distractions of the scanning pattern discussed in the previous section in order to observe the fine-grained horizontal detail. This effectively sets an upper limit on the bandwidth with 250 lines, since additional band-

width would appropriately be spent on increasing the number of lines. For a horizontal resolution equal to the vertical, and a 1.1:1 aspect ratio, we need, to a first approximation, 275 picture elements in a scanning line, or 137.5 cycles of the highest frequency. Allowing for total horizontal and vertical retrace time of about 23 percent of scanning time, the upper band limit frequency is given by

$$f \cong 250(137.5)(1.23)(30) \cong 1.27 \text{ MHz.}$$

The actual frequency for 1:1 horizontal-to-vertical resolution ratio is less, about 1 MHz. This is because the full vertical resolution corresponding to 20 lines per cm is not realized, by the ratio of the Kell Factor.<sup>15</sup>

At the other extreme, it is undesirable to make the horizontal resolution less than half the vertical resolution, because the overall subjective sharpness of a given horizontal-vertical resolution product decreases when the ratio is less than 0.5 or more than 2.<sup>16</sup> For this reason bandwidths less than about 0.5 MHz would also tend to be inefficient.

Within this range, the value of picture resolution must be weighed against the cost of bandwidth in transmission. The considerations involved cannot be examined here. It is perhaps worth pointing out, however, that in long distance digital trunks, the cost dependence on bandwidth is less severe than might be expected. This is because it is desirable for efficient digital encoding to take advantage of the correlation between successive samples, by differential feedback or other means.<sup>17</sup> When the bandwidth is reduced, for example from 1.0 MHz to 0.5 MHz, and the sampling rate with it, the more widely spaced samples are more nearly independent, and the sample differences are a larger fraction of the sample amplitudes. This requires an increase in the number of bits per sample to obtain the same signal-to-noise ratio (S/N). In addition, in the 1.0 MHz case the noise in the upper half of the bandwidth offers little impairment, as shown by the subjective weighting curve described in the next section, whereas in the 0.5 MHz picture most of the noise contributes to impairments. Therefore the S/N for the 0.5 MHz systems must be higher than for the 1.0 MHz system, for equal subjective noise impairment. The net effect is that the digital transmission rate with the 1.0 MHz bandwidth is reduced only about 25 percent with the 0.5-MHz bandwidth.

We have chosen the 1.0-MHz bandwidth and the nominal 1:1 resolution ratio. In return for the higher cost, the wider bandwidth provides assurance of resolution adequate for face-to-face service for

the foreseeable future. With the addition of color, for example, the bandwidth should still provide adequate horizontal resolution.

### 3.5 Receiver Roll-Off Characteristic

The choice of the 1-MHz bandwidth implies that the amplitude and phase response of analog transmission channels will be controlled only within that band. Since the cathode-ray tube is inherently capable of displaying signals at much higher frequencies, it is essential to prevent either components of the camera signal or interference at frequencies above 1.0 MHz from reaching the display. Although the high-frequency signal components could be suppressed equally well either at the camera or the receiver, for maximum noise and interference suppression it is desirable to put all of this "roll-off" attenuation at the receiver. Figure 3 shows the circuit configuration.\*

To keep signal energy beyond 1 MHz sufficiently unnoticeable in the picture, we have found it sufficient if the overall frequency response, from visual scene to receiver screen, is down 20 dB at 1 MHz, and more at higher frequencies. One might expect that the maximum resolution within the 1-MHz band would be obtained by using a phase-equalized sharp-cutoff filter to get 20-dB suppression at the band edge. Unfortunately the ringing thus produced is subjectively unacceptable in the picture. To obtain a rapid roll-off in frequency response without ringing, a filter whose impulse response is approximately a gaussian density function may be used.

With sufficiently large delay  $\tau$ , a filter can be designed so that for values of  $t$  in the neighborhood of  $\tau$  the impulse response  $g(t)$  is approximated quite well by

$$g(t) = \frac{\omega_0}{\sqrt{2\pi}} \exp \{ -\omega_0^2(t - \tau)^2/2 \}. \quad (1)$$

The corresponding frequency response,  $G(\omega)$ , is given by

$$G(\omega) = \exp (-i\omega\tau - \omega^2/2\omega_0^2). \quad (2)$$

For the present purpose the delay implied by the linear-phase term may be ignored.

\* As is discussed in the paper on the transmission plan in this issue,<sup>18</sup> pre- and de-emphasis networks are included in the circuit to suppress interference within the band. Although these circuits are physically located in the station set, they may be regarded in this discussion as part of the transmission channel and therefore are not shown in Fig. 3.



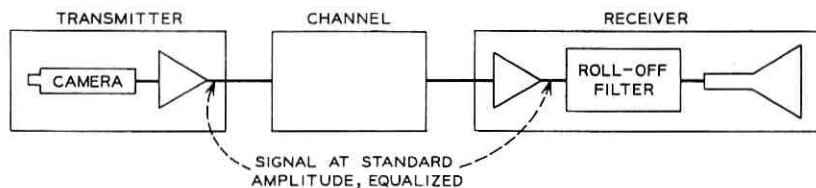


Fig. 3—Circuit configuration with roll-off filter.

To use this filter as the roll-off filter in Fig. 3 the value of  $\omega_0$  is to be chosen so that

$$G(2\pi \times 10^6) = 0.1G(0). \quad (3)$$

Assume now that any roll-off in the camera due to aperture or other effects is compensated in the camera circuitry, and that the display is similarly compensated in the receiver electronics if necessary. Assume further that the transmission channel has unity gain and linear phase over the band; departures from this ideal are considered in the next section. Consider the response  $h(t)$  of the filter to the camera output signal when a vertical black-white boundary in the picture is scanned. This is

$$h(t) = \frac{\omega_0}{\sqrt{2\pi}} \int_{-\infty}^t \exp(-\omega_0^2 x^2 / 2) dx. \quad (4)$$

The received picture will therefore shade monotonically from black to white. However, it is not the most pleasing picture that can be transmitted within a given bandwidth. A subjective improvement is obtained by "crispening," that is, introducing an overshoot preceding and following the transition.<sup>19</sup> This not only shortens the actual rise time at the transition, but also provides an effect of greater resolution by increasing the contrast across the boundary. It is done by subtracting the second derivative of the signal from itself. Figure 4 shows the roll-off filter with this addition. It is convenient to represent the crispened signal with the expression

$$s(t) = h(t) - \frac{k}{\omega_0^2} h''(t). \quad (5)$$

With this representation the overshoot peaks are found to be at

$$t = \pm \sqrt{\frac{1+k}{k}} \left( \frac{1}{\omega_0} \right). \quad (6)$$

The amount of the peak overshoot, as a fraction  $p$  of the step amplitude, is given by

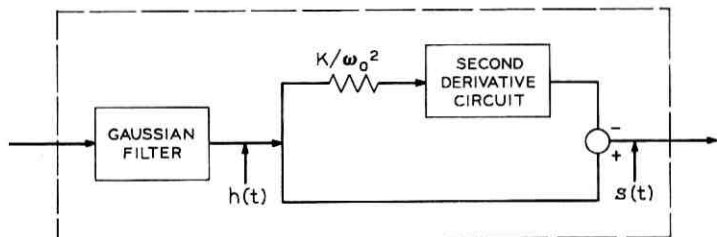


Fig. 4—Gaussian roll-off filter with crispening.

$$p = \frac{1}{\sqrt{2\pi}} \left\{ \sqrt{k(1+k)} \exp\left(-\frac{1+k}{2k}\right) - \int_{-\infty}^{-\sqrt{(1+k)/k}} \exp(-u^2/2) du \right\}. \quad (7)$$

The amount of the overshoot in this representation therefore depends only on  $k$ . We find  $k$  to give the desired overshoot and then choose  $\omega_0$  so that the frequency response  $S(\omega)$  corresponding to  $s(t)$  and given by

$$S(\omega) = 1 + k \left( \frac{\omega^2}{\omega_0^2} \right) \exp(-\omega^2/2\omega_0^2), \quad (8)$$

will be equal to 0.1 at 1 MHz.

Since the crispening technique provides more gain at higher frequencies within the band, it enhances noise. Nonetheless subjective tests carried out by E. F. Brown indicate a preference for about 12 percent overshoot in the presence of noise.<sup>20</sup> The effects of equalization error in transmission, however, make a smaller value desirable. This is because gain changes due to temperature variation on the telephone pairs used for connections to the central office tend to be greatest at the higher frequencies. When the cable is at a higher temperature than the one for which it was equalized, an additional loss increasing with frequency is imposed; when it is colder, there is an incremental gain increasing with frequency. The amount of overshoot for which the picture is about equally impaired with the maximum permissible high-frequency loss and gain deviations was found in studies by M. W. Baldwin, Jr., and H. G. Suyderhoud at Bell Laboratories in 1967 to be about 4 percent. They also found this to be the amount of overshoot which makes the effects of high frequency response deviations most tolerable.

Four percent overshoot corresponds to a value of  $k$  equal to 0.5292. For 20-dB attenuation at 1 MHz, the appropriate value of  $\omega_0$  is

$2\pi(355, 920)$  radians/second. The resulting time-domain response at an abrupt brightness boundary is shown in Fig. 5; the frequency response is shown in Fig. 6. This is therefore the scene-to-screen response of the system, exclusive of transmission channel effects.

To allow tolerances for the design of the filter, the maximum and minimum overshoot values are set at 4.5 and 3.5 percent respectively. The corresponding values of  $k$ , when substituted into equation (7), give upper and lower values at each frequency within which the frequency response is required to lie. The station set receiver is also required to meet the equalization requirement (echo rating) given in the paper on the transmission plan in this issue, as explained in the next section.

#### IV. PICTURE TRANSMISSION STANDARDS

The parameters given in the previous section define the quality of the picture seen when a station transmitter is connected to a station receiver through a connecting link that is free of noise, interference, and distortion. A complete specification of the picture quality requires a statement of the impairment which is permitted in transmission.

In establishing the basic picture standards, we were free to consider

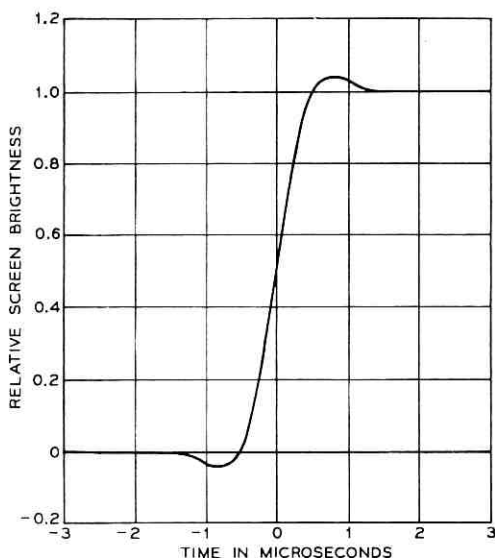


Fig. 5—Response of display at an abrupt brightness boundary.

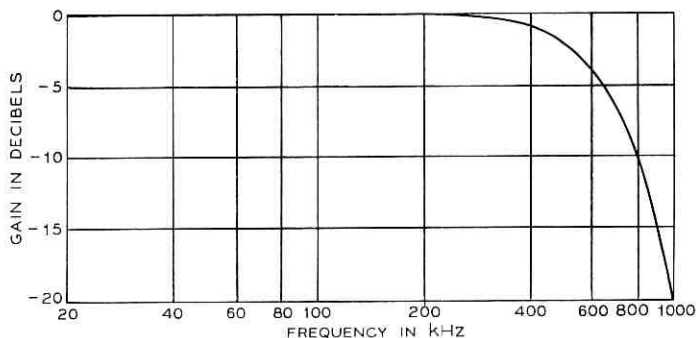


Fig. 6—Overall frequency response, visual scene to receiver screen.

a wide range of values of each, in order finally to establish a service quality sufficient to secure most of the possible enhancement of communication at no greater cost than necessary. With these standards established, the latitude remaining for transmission impairment is limited. This is because the customer will compare the quality of the picture at any given time with the quality of the best *Picturephone* visual telephone image he has had occasion to see. The objective in setting transmission standards is therefore to ensure that the quality of the picture transmitted over the longest connection is unobjectionable compared to the unimpaired picture.

#### 4.1 Analog Transmission

Analog transmission may produce identifiable effects in the picture due to several different types of linear distortion and of interference. Since baseband amplifiers must ordinarily be ac-coupled, there is in any connection an accumulated low-frequency roll-off. Clamping in the receiver mitigates this,<sup>21</sup> but there is a residual effect known as "tilt." The accumulated effects of imperfect equalization in many links produce distortions of various types depending upon the value of the net equalization error at each frequency. Single-frequency interference, such as a radio broadcast carrier, tends to produce moving wave patterns. Connections through central offices may pick up brief high-amplitude impulses due to switching transients in telephone lines. Crosstalk from one video channel to another may cause a second image to appear, usually moving relative to the first, and often slowly enough to be identifiable if the degree of coupling were not controlled. Interference from power lines, at 60 Hz and its harmonics, produces moving bars in the picture. Thermal noise in amplifiers, and the accumulation

of all other interferences which are individually too low in amplitude to be identifiable, produce random noise on the screen, its appearance depending on the average power spectrum of the noise.

The approach taken to these analog impairments is to characterize them, define the method of measurement, and determine by subjective testing the amount of each impairment which is acceptable in a connection of the maximum number of links. These amounts may then be allocated among the various switches and analog transmission systems as described in the paper on the transmission plan in this issue. The station transmitter and receiver also get an allocation of some types of impairment, as do the analog portions of the digital encoder and decoder.

The basis of this subjective testing is a comment scale. With the 7-point comment scale, for example, the subject is asked which of the following comments applies to the amount of a specific impairment in the picture he is seeing:

- (i) Not perceptible.
- (ii) Just perceptible.
- (iii) Definitely perceptible but only slight impairment to picture.
- (iv) Impairment to picture but not objectionable.
- (v) Somewhat objectionable.
- (vi) Definitely objectionable.
- (vii) Extremely objectionable.

The subjects chosen are technical people who are not involved in video communications work. The resulting data are processed to estimate the amount of impairment at which 95 percent of the user population would rate the picture comment three or better. The transmission objective will be to make the overwhelming majority of circuits meet such a 95 percent point or better with respect to each impairment.

#### 4.1.1 *Equalization Error and Echo Rating*

As explained in the preceding paper in this issue, the subscriber's connection to the central office utilizes telephone pairs with equalizers at intervals of the order of a mile, as do short interoffice trunks. Since equalization at each amplifier is not perfect, and since temperature changes in the cables introduce additional equalization error, departures from zero gain and linear phase occur and accumulate link by link. Those due to cable loss variations resulting from temperature changes may add systematically. As a result equalization error is commonly the limiting factor in baseband analog transmission.

Since equalization error has different subjective effects depending on its distribution in frequency within the band, it is desirable to define a figure of merit for equalization which increases monotonically with increasing subjective acceptability. This requires relating subjective impairment to particular features of the response as characterized by measurements. The figure of merit will be of greatest usefulness if it is so defined that the figure of merit for two links in tandem can be determined, or at least estimated, by combining in some way the figures for the two links considered separately.

A means of doing this was originally proposed at Bell Laboratories in 1949 by S. Doba, Jr., in a study which was not published. A formulation following his approach, as elaborated recently by H. G. Suyderhoud and R. Piater, may be outlined as follows.

Suppose  $f(t)$  to be the response to a unit impulse of an ideal circuit having zero gain and linear phase throughout the band  $(0, f_0)$ , and arbitrarily large attenuation elsewhere. The actual response  $h(t)$  of a given circuit to a unit impulse may be written as:

$$h(t) = \sum_{n=-\infty}^{\infty} C_n f\left(t - \frac{n}{2f_0}\right), \quad (9)$$

where  $C_n$  is proportional to  $h(n/2f_0)$ . The series may be truncated at suitable values of  $n$  denoted by  $\pm M$ . Since a small amount of overall time delay is not an impairment and an amplitude error may be corrected at the station by automatic gain control, we may choose the time origin and the amplitude of  $h(t)$  so that  $C_0 = 1$ ,  $C_n < 1$ ,  $n \neq 0$ . Then the error response is given approximately by:

$$h_e(t) = \sum_{n=-M}^M C_n f\left(t - \frac{n}{2f_0}\right) - f(t). \quad (10)$$

Equation (10) expresses the difference between the desired response to a unit impulse and the actual response as a sum of preceding and following "echoes" of the desired signal of amplitude  $C_n$ . If we think of the unit impulse as representing, by its amplitude, an element of the picture,  $h_e(t)$  represents the extent to which this element is spread to the left and right in transmission. Some of the echo amplitudes may be negative, so that the corresponding displaced picture information undergoes a change in sign.

The impairment introduced into the picture by a single echo has been studied in connection with television by S. Doba, A. D. Fowler and H. N. Christopher,<sup>22</sup> and P. Mertz.<sup>23</sup> The impairment in the case of visual telephone images was studied by R. Piater at Bell Laboratories in

1969. These studies show generally that the impairment increases with displacement from the picture, that is, with increasing  $|n|$ , as well as with amplitude, that is, with the magnitudes of the coefficients  $C_n$ . This suggests weighting the echoes with coefficients  $W_n$  and summing to determine the total power of the weighted error transient.

To secure additional freedom in matching the weighted error to subjective test results, each echo may also be frequency weighted. Conceptually this is done by passing impulses of amplitudes  $W_n C_n$  separately through filters with impulse responses denoted by  $g_n(t)$ , and combining the responses. The resulting weighted transient is given by

$$h_w(t) = \sum_{n=-M}^M W_n C_n g_n\left(t - \frac{n}{2f_0}\right), \quad \text{with } C_0 = 0. \quad (11)$$

The total error power, relative to the desired signal power, is obtained by squaring and integrating this signal. We get:

$$P_e = \sum_{n=-M}^M \sum_{m=-M}^M C_n C_m W_n W_m k_{nm}, \quad (12)$$

where

$$k_{nm} = \int_{-\infty}^{\infty} g_n\left(t - \frac{n}{2f_0}\right) g_m\left(t - \frac{m}{2f_0}\right) dt. \quad (13)$$

The problem of defining  $P_e$  so that it increases monotonically as impairment increases is therefore reduced to choosing the matrix elements  $A_{nm} = W_n W_m k_{nm}$ . This must be done by analysis of the results of subjective tests of pictures impaired by circuits for which the values of  $C_n$  are accurately controlled. Although a preliminary result has been obtained, this problem is still under investigation. The description above is somewhat simplified, particularly with respect to the method of normalizing the amplitude and of choosing the time origin.

Since the desired received signal power, in response to a unit impulse input, has been set at one unit by adjusting the received level so that  $C_0 = 1$ , the power  $P_e$  represents the relative error power. The desired figure of merit is the level in dB of the error power relative to the signal power. This has come to be called the echo rating, in reference to the procedure of weighting the echoes, although this term is somewhat misleading since individually identifiable echoes are unusual in *Picturephone* transmission. We have:

$$ER = 10 \log P_e . \quad (14)$$

The echo rating objective presently established is  $-26$  dB.

If the equalization errors of several links are random and independent, the values of  $P_e$  for the links can be added to determine the overall error power and hence the overall echo rating. Where the equalization error is systematic, the square roots of the separate error powers must be added to determine the square root of the overall error power. On these bases it is possible to allocate echo rating to every analog link in the network, including the station set and analog portions of the codec.<sup>18</sup>

The echo-rating figure of merit is designed to rate circuits in which the error transient associated with the transmission of a unit impulse (which is to say a picture element) is less than one scanning line in duration. Equalization error which is confined to a few kilohertz of bandwidth and is sufficiently severe, or which varies cyclically across the bandwidth, with a period of a few kilohertz, will not be correctly evaluated. In the analog transmission facilities planned for *Picture-phone* service there are no networks or network components which should generate these effects. They should therefore occur quite infrequently. If necessary however the echo rating concept can in the future be extended to include them.

#### 4.1.2 Low-Frequency Cutoff and "Tilt" Impairment

Baseband amplifiers and the baseband portions of encoders and modulators are usually ac-coupled through capacitive coupling circuits. The impairment due to the resulting low-frequency cutoff, known as "tilt," accumulates approximately linearly in a connected circuit, so that if one link has one percent tilt and a second has two percent, the two together will have three percent. The tilt allowable in a maximum-length connection is therefore allocated among subscriber loops, switching machines and trunks on the basis of linear addition.

The gross effects of low-frequency cutoff are removed by clamping the signal at the horizontal synchronizing pulses in the station receiver; tilt is the residual impairment. Suppose for example that the picture consists of a white rectangle against a grey background. The average signal voltage of the all-grey scanning lines is lower than those which scan the rectangle. A plot of average voltage across a line therefore shows rectangular pulses during the field scanning time. The coupling circuit transmits these pulses with the characteristic decay toward the average value. The result is that the baseline of the synchronizing pulses wanders as shown in Fig. 7. This would produce



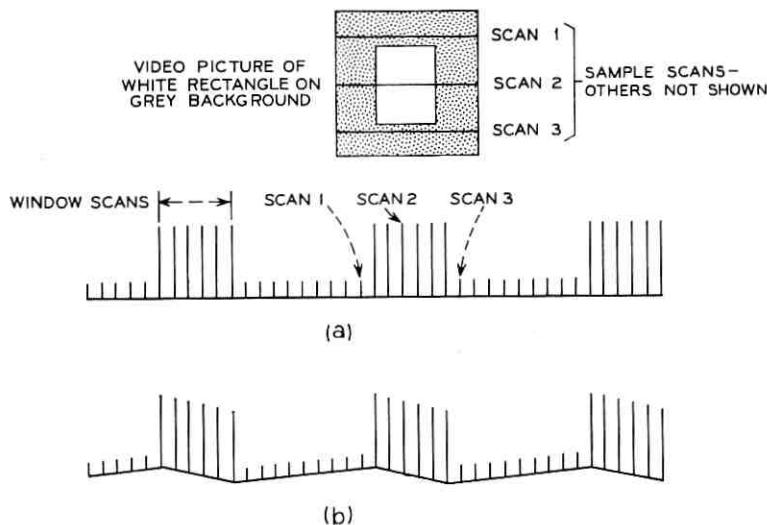


Fig. 7—Effect of low-frequency cutoff on composite video signal, before clamping. Simplified composite video signal (a) before low-frequency cutoff and (b) after low-frequency cutoff.

shading from top to bottom in the picture, except for the clamp, which restores the baseline. The slope or "tilt" remains in the video during each line, however. Figure 8 shows the signal plotted for scanning above, through, and below the white rectangle, assuming the signal clamped at the beginning of each line. The grey value drifts downward when scanning through the rectangle; this means that a shadow will appear on either side of the white rectangle, but more visibly to the right of it.<sup>21</sup> The effect is most noticeable when a white object is moved about in the picture.

Tilt is defined as the decay in the response to a voltage step, measured over the first 100 microseconds, expressed in percent of the step height. Tests made at Bell Laboratories by W. Ohnsorg in 1969 indicate that pictures transmitted over circuits with 10 percent tilt will be rated comment 3 or better by 95 percent of the user population.

#### 4.1.3 Random Noise

Random noise interference consists of the sum of thermal noise and all those other interferences appearing in the signal which are at too low a level to be separately identified on the screen. The amplitude distribution is nominally gaussian within the dynamic range of the channel. The spectrum may vary widely depending on the characteristics of the transmission systems through which the signal has

passed and other circumstances. The appearance depends on the spectrum. White noise peaks tend to be more visible than black; as a result the appearance is sometimes likened to falling snow. If the noise is preponderantly at low frequencies the "flakes" appear as horizontal streaks; at high frequencies as instantaneous white dots.

The S/N is referred to a point at which the signal is correctly equalized but has not passed through the roll-off filter in the receiver (see Fig. 3). It is defined as

$$S/N = 20 \log (p/n) \quad (15)$$

in which  $p$  is the peak-to-peak composite signal voltage and  $n$  is the rms value of the noise voltage.

The S/N at which 95 percent of the user population are expected to rate the picture comment three or better depends on the spectrum. The roll-off filter in the receiver suppresses noise at the higher frequencies, and at higher frequencies noise is less visible anyway. An "eye-weighting" curve, found by M. W. Baldwin, Jr., at Bell Laboratories in 1967, is shown in Fig. 9; this represents the relative impairment

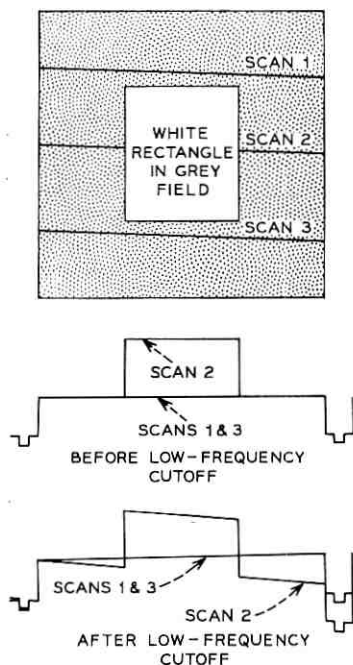


Fig. 8—Effect of low-frequency cutoff on video signal representing one scanning line, after clamping.

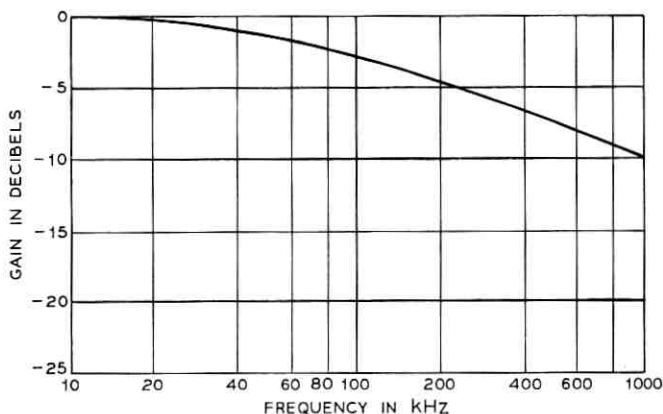


Fig. 9—Subjective noise weighting curve for video signal.

due to noise as a function of its position in the spectrum. Figure 10 shows the total weighting when the receiver roll-off is also taken into account.

The weighting curve of Fig. 10 is used in the same way as in television practice.<sup>24</sup> To determine the weighted S/N for random noise having a given spectrum, the noise may be passed through a filter whose amplitude response approximates the weighting curve of Fig. 10, and the weighted value of  $n$  measured with a true rms voltmeter. The equivalent operation may be performed numerically. The 95 percent point for the signal-to-weighted-noise ratio is 52 dB. For example, a noise flat over the 1-MHz band with S/N = 47 dB can be shown to have a signal-to-weighted-noise ratio of 52 dB.

#### 4.1.4 Switching Noise

The use of telephone pairs for subscriber loops and short trunks exposes the video signal to the interference sources common in telephone switching of offices. Chief among these is switching noise. The opening of contacts attached to a telephone pair carrying direct current may produce a train of transients of large amplitude containing energy distributed over several MHz. The inevitable crosstalk coupling allows some of this energy to be transferred from telephone pairs to video telephone pairs in the same cable. The resulting interfering transient may reach amplitudes of the order of a volt and this together with its brief duration has led to the use of the term "impulse noise". The transient is typically fairly complex, consisting of a series of separate rapidly decaying oscillations. The duration of each may be 5 to 50  $\mu$ s,

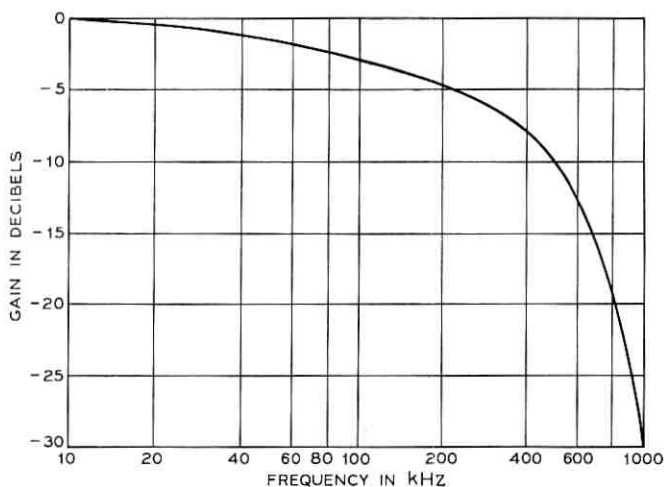


Fig. 10—Noise weighting curve, including effect of receiver roll-off.

they may occur at intervals of 20 to 200 $\mu$ s, and the entire train may last on the order of a millisecond. On the screen the appearance is typically that of a scattering of white dots and small blotches.

Statistically, these transients are very infrequent. With their short duration they contribute very little to total random noise power in spite of their high amplitude. They must therefore be subject to a separate set of requirements. The method of objective measurement must be devised to be representative of their ability to impair the signal.

The means used is to sample the noise on the idle channel at a 10-MHz rate and detect and count samples of amplitude greater than a given threshold. Since the dc content is zero the maximum count is  $5 \times 10^6$  per second. The actual count, as a fraction of the maximum, over a sufficiently long period, is the estimated probability  $P$  of the noise exceeding the threshold.

Switching noise coupled onto a cable pair is subject to the frequency shaping of the equalizing amplifier at the central office. The gain characteristic of this amplifier is adjusted to equalize the cable section behind it, which may range in length from a few hundred feet to a mile or more. To take into account the resulting differences in switching noises as received at the line terminal, the noise is weighted by passing it through a weighting filter before making the measurement. The weighting which is applicable was found by R. M. Lund at Bell

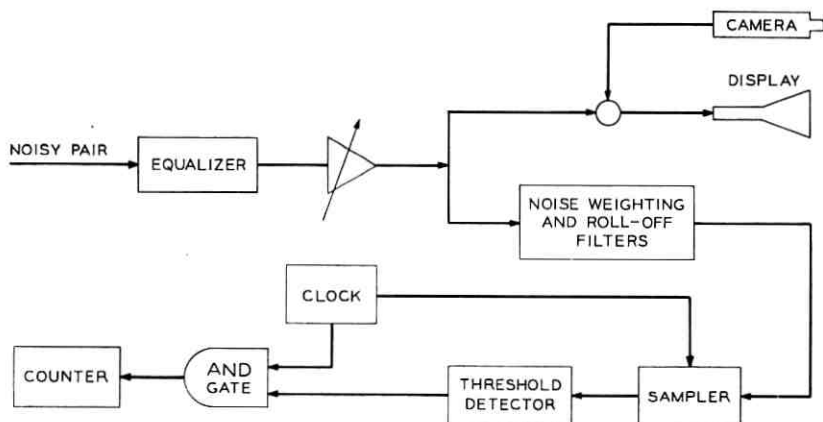


Fig. 11—Measurement arrangement for switching noise.

Laboratories in 1967 to be very closely the curve of Fig. 10, for random noise. The measurement arrangement is shown in Fig. 11. Lund found that noise counts at a threshold level 33 dB below the peak-to-peak composite signal amplitude were best correlated with subjective evaluations of this impairment.

The user may not evaluate the noise he sees over a long period. His attention is more likely to be drawn to switching noise when it is particularly bad, during an interval of the order of a minute. The number of samples which exceed the threshold during such a short interval is a random variable.

Lund has found that the logarithm  $X$  of the number of samples which exceed the threshold in one minute is normally distributed, with mean and standard deviation related to the value of  $P$ . This may be combined with the probability that a user will rate the impairment denoted by  $X$  comment three or better, to determine the probability that a user chosen at random, viewing the picture during an interval chosen at random on a pair whose weighted threshold probability is  $P$  will rate it comment three or better. With  $P = 6 \times 10^{-5}$  the comment three score is 95 percent.

Because of the difficulty of obtaining and using long-term averages in a changing environment, and some uncertainty resulting from basing noise characterizations on a very limited sample of cables and offices, the requirement has arbitrarily been reduced to  $1.5 \times 10^{-5}$ , representing a comment three score of 99 percent on the basis of available data.

This is then the requirement on the long-term average fraction of weighted switching noise samples which may exceed the  $-33$  dB threshold.

#### 4.1.5 *Single-frequency Noise and Power Hum*

The single-frequency S/N is defined as in equation (15) for random noise,

$$S/N = 20 \log (p/n) \quad (15)$$

in which  $p$  is the peak-to-peak composite signal voltage at a point at which the signal is correctly equalized and before it has passed through the receiver roll-off characteristic, and  $n$  is the rms value of the sinusoidal interference.

Single-frequency interference at low frequencies which are exact multiples of the field rate of the particular camera and receiver under test produces a fixed bar pattern. At low multiples the bars are horizontal. As the frequency of the interference is made to depart from a multiple of the field rate the bars begin to move. The interference is most impairing when its frequency is about 10 Hz different from the field frequency.

Figure 12 shows the estimated S/N at which the impairment will be rated comment three by 95 percent of the user population, for the range of power hum frequencies, as determined recently by D. B. Robinson, Jr., at Bell Laboratories. Robinson's studies show that frequencies 10 Hz different from the field or frame harmonics continue to provide the locally most severe impairment throughout the band. However, these impairment maxima vary cyclically. They are about 10 dB more severe at multiples of the line scanning rate than at odd multiples of half the line rate.

The curve of Fig. 12 was obtained using a receiver without clamping. The improvement due to clamping cannot be determined by taking into account the measured clamping effectiveness, because effects due to low-frequency interference remain in the picture even with perfect clamping, much as in the case of low-frequency roll-off.

Figure 13 shows the envelope of maxima of the single-frequency S/N corresponding to comment three for the Mod II station set. This curve was obtained by fitting a curve to points at frequencies about 10 Hz different from harmonics of the field or frame frequency and near harmonics of the line rate. At lower frequencies the requirement is diminished by clamping, at higher frequencies by the roll-off filter of the station set and some additional eye weighting.

#### 4.1.6 Crosstalk

The telephone cables used for transmission of the *Picturephone* video signal typically exhibit crosstalk coupling from one pair to another. The more important coupling effects are described elsewhere,<sup>25</sup> but may be briefly summarized here. In "far-end" crosstalk the desired and undesired signals are subject to the same amplifier gain and transmission loss, except for the crosstalk coupling loss. The coupling is randomly distributed along the cable. The net effect is that of a single capacitor coupling the transmitter in one circuit to the receiver in the other. "Near-end" crosstalk occurs only when both directions of transmission are in the same cable sheath. The undesired signal is coupled through many paths, each involving a different transmission loss down the cable and back. The resulting frequency characteristic, aside from the receiving amplifier gain, is a random variation about a 4.5-dB per octave trend line of loss decreasing with frequency, compared to a consistent 6-dB per octave for far-end crosstalk with no additional amplification involved.

Tests conducted at Bell Laboratories in 1966 by J. H. Gentry and others indicated that subjective effects of the two types of crosstalk

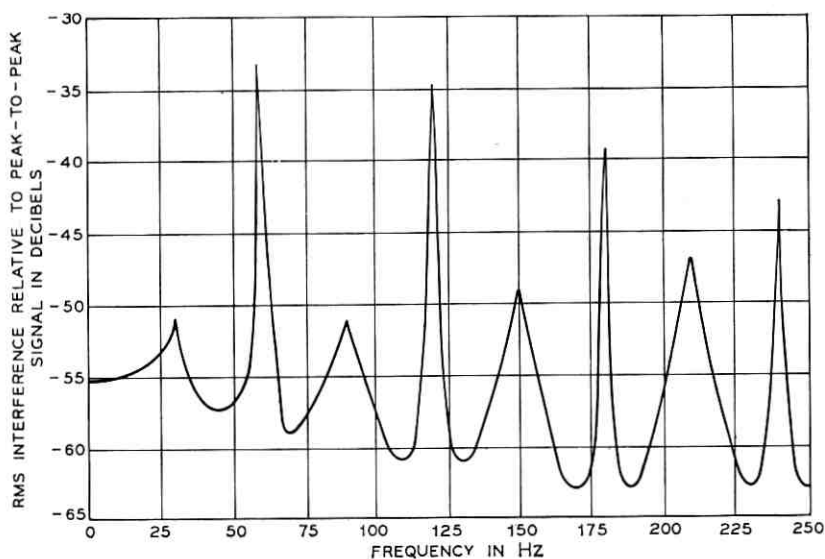


Fig. 12—Single-frequency interference for comment three impairment at ac-power harmonic frequencies.

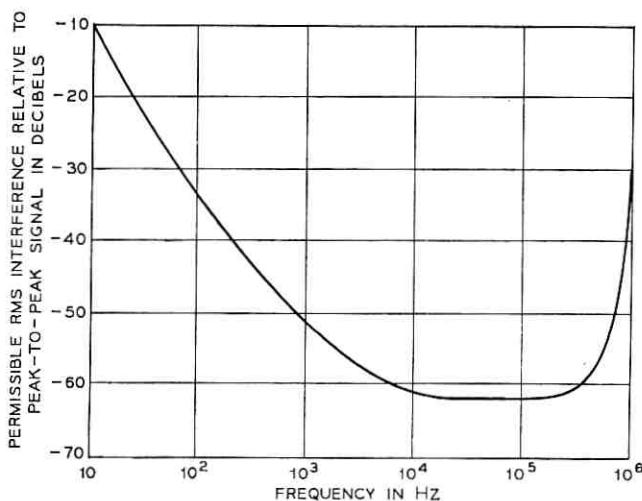


Fig. 13—Envelope of minima of permissible single-frequency interference.

coupling may be equated if the loss at the 150-kHz point of the near-end trend line, less the amplifier gain, is equal to the 150-kHz far-end coupling loss. Therefore the 150-kHz point is used to evaluate any coupling characteristic.

Since the unwanted image has in effect been differentiated by the crosstalk coupling, vertical boundaries between black and white areas tend to be accentuated and appear as lines in the picture, moving horizontally. The number of times per second that the coupled image passes across the screen is equal to the difference between the horizontal scanning frequencies of the connected and the interfering transmitters. The most objectionable rate, determined by J. Orr in tests made at Bell Laboratories in 1969, is about 0.2 passes per second. At this rate, the estimated coupling loss at 150 kHz required to get a rating of comment three or better from 95 percent of the user population is 45 dB, and this is therefore the requirement for any single interferer. Crosstalk also contributes to random noise, and the sum of all crosstalk interference, together with all other random noise sources, must meet the weighted random noise requirement given in Section 4.1.3.

#### 4.2 Digital Transmission

The impairment introduced by the use of digital transmission facilities consists of quantization noise, pulse jitter, and the effects of regeneration errors in transmission. (The analog portions of the coding



terminals may also contribute to the impairments described in Section 4.1.)

Quantization noise occurs in the differential feedback encoder because the video-sample differences, whose amplitudes occupy a continuum of values over the dynamic range of the signal, must each be assigned one of a small number of values. The resulting quantizing noise has an appearance similar to random noise except at vertical or diagonal brightness boundaries, where close examination usually reveals a slightly pulsating appearance.<sup>17</sup> This "edge busyness" must be traded off against the random noise effect in the encoder design. It is difficult to quantify objectively, and the optimum encoder design is best obtained by visual comparison. An *a priori* requirement has therefore not been placed on quantization noise, although an allotment has been made for the random noise component.<sup>18</sup>

Pulse jitter is an effect in which the pulse rate is alternately speeded up and slowed down, accordion-like. It occurs because the regenerators are timed from the incoming pulse train and are therefore to some extent affected by the information content. Jitter may be removed to any desired degree by buffering and retiming the signal, and the requirements therefore do not affect the basic system design. At present, jitter requirements have not been formulated.

A regeneration error in transmission occurs when noise, interference, and overlapping adjacent pulses combine to operate the regenerator when no pulse was transmitted, or cause it to fail to regenerate a transmitted pulse. The differential feedback decoder stores the resulting noise pulse in its feedback loop, so that its effects may be extended over a substantial part of a scanning line. Since the white errors are more visible than the black, the subjective effect is that of an occasional horizontal white streak along a scanning line. The majority of these are missed in observation.

The subjective effect of pulse regeneration errors depends to some extent upon the encoding algorithm used. With the differential feedback encoder<sup>17</sup> preliminary observations indicate that an error rate of  $10^{-6}$  would introduce negligible impairment. A requirement of  $3.3 \times 10^{-7}$  can be met by the proposed network.<sup>18</sup> This allows some margin for changes in the coding algorithm and for the possibly more stringent requirements of network applications other than visual telephone.

## V. SUMMARY

The basic standards of *Picturephone* service have been established with the objective of providing the visual adjunct at no greater cost

than is necessary to secure most of the available enhancement of direct conversation, in an instrument which can be associated with an ordinary telephone and used with a minimum of rearrangement of the home or office environment. The standards do not preclude the future application of color or the use of the network for a wide variety of services other than face-to-face conversation. Transmission standards are based on the principle that the quality of the picture after transmission through the longest possible connection should not be objectionable by comparison with the unimpaired picture.

#### VI. ACKNOWLEDGMENTS

Contributors to this work are very numerous. It is a pleasure to call specific attention, however, to the leadership of Mr. A. D. Hall, the author's original mentor in this field, and to the contributions of Mr. W. T. Wintringham to some of the earlier work described here.

#### REFERENCES

1. Dorros, I., "The *Picturephone*® System: The Network," B.S.T.J., this issue, pp. 221-233.
2. Garrett, M. A., and Mumford, A. H., "The History of Television," Proc. IEE, London, 99, No. 17, Part IIIA (1952), pp. 25-42.
3. "Telephones That Can See," The Reporter, September, 1956, pp. 10-11.
4. Kock, W. E., and Miller, R. L., "Two-Way Television Over Telephone Lines," Patent No. 2,895,005, July 14, 1959.
5. Clark, M. A., Miller, R. L., and Sears, R. W., "Two-Way Television Over Telephone Lines," Patent No. 2,922,843, January 26, 1960.
6. Meacham, L. A., "The *Picturephone*," IEEE Trans. Broadcasting, BC-12, No. 1 (June 12, 1966), pp. 37-52.
7. Hall, A. D., "Experiments with *Picturephone* Service," Bell Laboratories Record 42, No. 4 (April 1964), pp. 114-120.
8. Carson, D. N., "The Evolution of *Picturephone* Service," Bell Laboratories Record 46, No. 9 (October 1968), pp. 284-291.
9. Cagle, W. B., Stokes, R. R., and Wright, B. A., "The *Picturephone*® System: 2C Video Telephone Station Set," B.S.T.J., this issue, pp. 271-312.
10. Rosenthal, A. H., Patent No. 2,420,198, issued May 6, 1947.
11. Fink, D. G., *Television Engineering Handbook*, New York: McGraw-Hill Book Company, 1957, p. 1-27.
12. Fowler, A. D., "Low-Frequency Interference in Television Pictures," Proc. IRE, 39, (October 1951), pp. 1332-1336.
13. Brainard, R. C., Mounts, F. W., and Presada, B., "Low Resolution TV: Subjective Effects of Frame Repetition and Picture Replenishment," B.S.T.J., 46, No. 1 (January 1967), pp. 261-271.
14. Brown, E. F., "Low Resolution TV: Subjective Comparison of Interlaced and Non-Interlaced Pictures," B.S.T.J., 46, No. 1 (January 1967), pp. 199-232.
15. Fink, D. G., Editor, *Television Engineering Handbook*, New York: McGraw-Hill Book Company, 1957, pp. 8-6 to 8-23.
16. Baldwin, M. W., Jr., "The Subjective Sharpness of Television Images," B.S.T.J., 19, No. 4 (October 1940), p. 563.
17. Millard, J. B., and Maunsell, H. I., "The *Picturephone*® System: Digital Encoding of the Video Signal," B.S.T.J., this issue, pp. 459-479.

18. Brown, H. E., "The *Picturephone*® System: Transmission Plan," B.S.T.J., this issue, pp. 351-394.
19. Brown, E. F., "A New Crispening Circuit for Television Images," J. SMPTE, 72, No. 11 (November 1963), pp. 849-853.
20. Brown, E. F., "Television: The Subjective Effects of Filter Ringing Transients," J. SMPTE, 78, No. 4, (April 1969), pp. 249-255.
21. Doba, S., Jr., and Rieke, J. W., "Clampers in Video Transmission," Trans. Amer. Inst. Elec. Eng., 69, Pt. 1 (1950), pp. 477-487.
22. Fowler, A. D., and Christopher, H. N., "Effective Sum of Multiple Echoes in Television," J. SMPTE, 58, No. 6 (June 1952), pp. 491-500.
23. Mertz, P., "Influence of Echoes on Television Transmission," J. SMPTE, 60, No. 5 (May 1953), pp. 572-596.
24. Barstow, J. M., and Christopher, H. N., "Measurement of Random Monochrome Video Interference," Trans. A.I.E.E., Communication and Electronics, 72, Part I (1953), pp. 735-741.
25. Bunin, B. J., Hirsch, R. B., and Olsen, R. E., "The *Picturephone*® System: Crosstalk Considerations in the Transmission of Analog Signals on Paired Cable," B.S.T.J., this issue, pp. 427-457.



## **The Station**



## The *Picturephone*<sup>®</sup> System:

# 2C Video Telephone Station Set

By W. B. CAGLE, R. R. STOKES, and B. A. WRIGHT

(Manuscript received October 14, 1970)

*This article describes the station set designed for use in commercial Picturephone<sup>®</sup> service. Coded the 2C Video Telephone Station, it consists of three new pieces of apparatus—the 1A Display Unit, the 1A Service Unit and the 72A Control Unit. We discuss details of the electrical and physical design.*

### I. INTRODUCTION AND OBJECTIVES

The station set is that part of the *Picturephone* system with which the customer interacts directly and is the principal basis for his reactions to the service. The set should provide the video camera and display functions, hands-free audio facilities, and the necessary customer controls in an attractive, easy-to-use, and economical form. The fundamental design requirements for the 2C Video Telephone Station, commonly known as the Mod II *Picturephone* Set, evolved from laboratory and field studies as discussed in other papers in this issue.<sup>1,2</sup> From these and other studies, the following general objectives were formulated:

- (i) The *Picturephone* station equipment should initially be designed for desk-top location and should work in conjunction with any standard 12-button *Touch-Tone*<sup>®</sup> telephone.
- (ii) The set should incorporate a built-in speakerphone with ON, OFF, and VOLUME controls.
- (iii) The camera and display should utilize a 1-MHz video signal containing 251 visible scanning lines in a two-to-one interlaced, 30 frames per second format.
- (iv) The camera should be physically centered above the display tube which should provide a picture with an 11 by 10 aspect ratio.

- (v) The user will sit about 0.9 meters in front of the set and should be able to operate the major controls from this position.
- (vi) There should be controls for changing the display brightness, viewing the outgoing picture and shutting off the camera for video privacy.
- (vii) The set should contain simple means for customer control of the camera field of view for size, elevation and azimuth.
- (viii) The camera should automatically compensate for a wide range of ambient illumination covering the levels to be encountered in offices.
- (ix) The set should utilize local 110 V ac power and telephone service should not be impaired during power failure.
- (x) A convenient means for transmitting graphical material should be incorporated.

The resulting station equipment designed to meet these objectives consists of a display unit and control unit which, together with a standard *Touch-Tone* telephone, are located on the user's desk. These units interface with the network through a service unit which can be remotely located with a cable run of 26 meters maximum length.

A 55D Control Unit is also required for speakerphone operation. This unit is connected into the *Picturephone* station equipment at the service unit. The 55D Control Unit has more received signal handling capability than the 55B used with the 3A Speakerphone.<sup>3</sup>

The 1A Display Unit (Fig. 1) contains the camera and display tubes and associated transmitter and receiver electronics, the loudspeaker, and the tone sounder for distinctive *Picturephone* call alerting. While designed primarily for face-to-face communication, it includes a built-in capability for showing graphics placed on the desk in front of the set. The display unit design is discussed in detail in Section II of the paper.

The 72A Control Unit (Fig. 2) contains the user controls and the microphone for speakerphone operation. The ON, OFF and VOLUME controls are for hands-free audio operation on telephone and *Picturephone* calls. The microphone can be muted for audio privacy by holding the QUIET button depressed. The VU-SELF and PRIVACY switches provide for checking the transmitted picture and for shutting off the camera for video privacy. A distinctive horizontal bar pattern is transmitted when the PRIVACY switch is operated. The ZOOM and HEIGHT controls allow the user to adjust the size and elevation of his



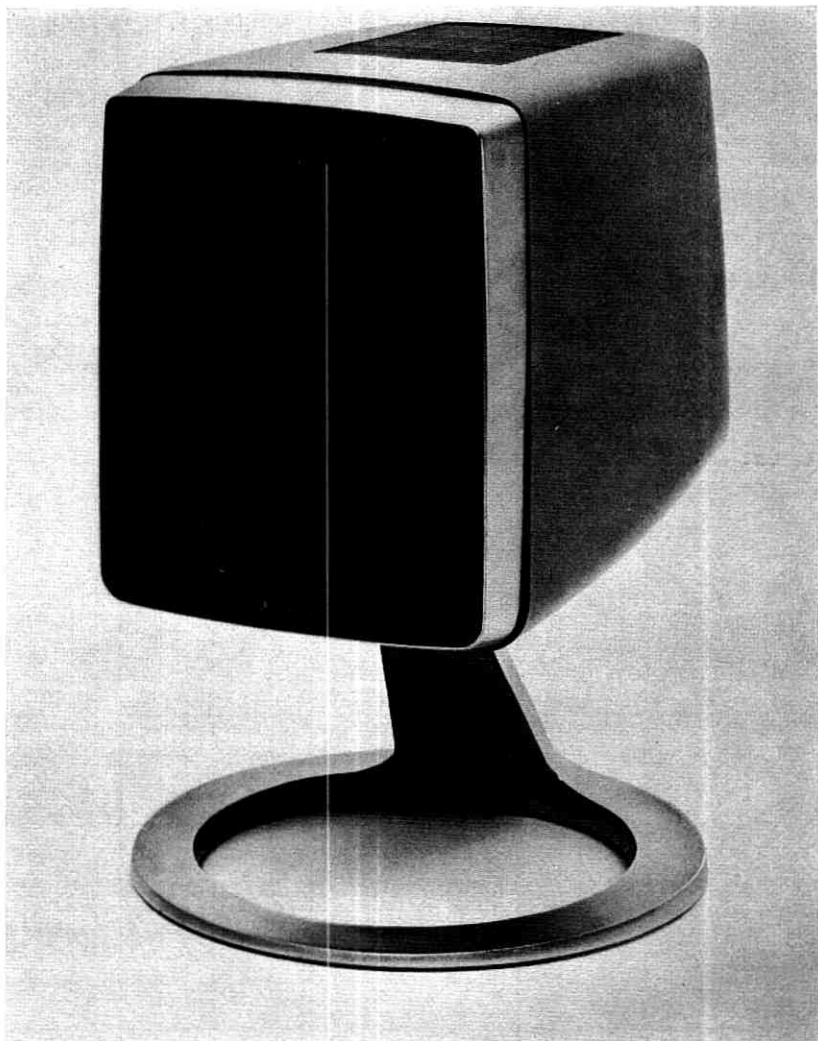


Fig. 1—1A Display Unit.

camera's field of view. The BRIGHT control permits adjustment of the display brightness for best subjective performance. The design of the control unit is described in Section III.

The 1A Service Unit (Fig. 3) contains the station power supply, supervisory and control logic, and the video line/loop termination, protection and maintenance electronics. All installer adjustments and

options are located in the service unit to simplify installation and maintenance. The service unit design is described in Section IV.

## II. DISPLAY UNIT

### 2.1 *General Factors*

#### 2.1.1 *Lens Considerations*

The display unit is a desk-top instrument designed to accommodate the heights of 99 percent of the American male and female adult population when seated. For convenience of desk use and other reasons, a 92-cm viewing distance has been established.<sup>2</sup> A survey of subjects using the Mod I *Picturephone* set showed that users seldom remain at a fixed viewing distance, but move from a near position of 50 cm to a distant position of 150 cm, with an average distance of 92 cm. This established an object focal distance of 92 cm for the camera lens and permitted the display unit to be positioned beyond the central working area of a desk.

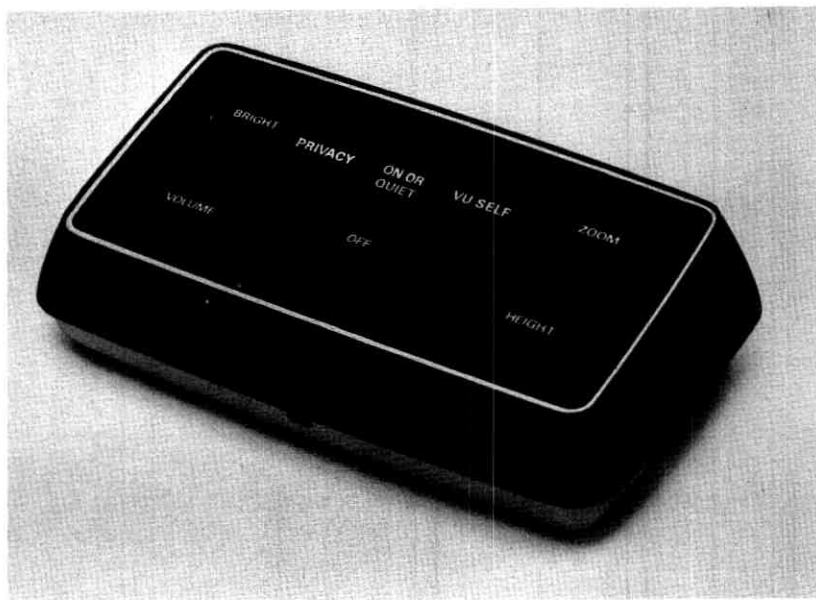


Fig. 2—72A Control Unit.

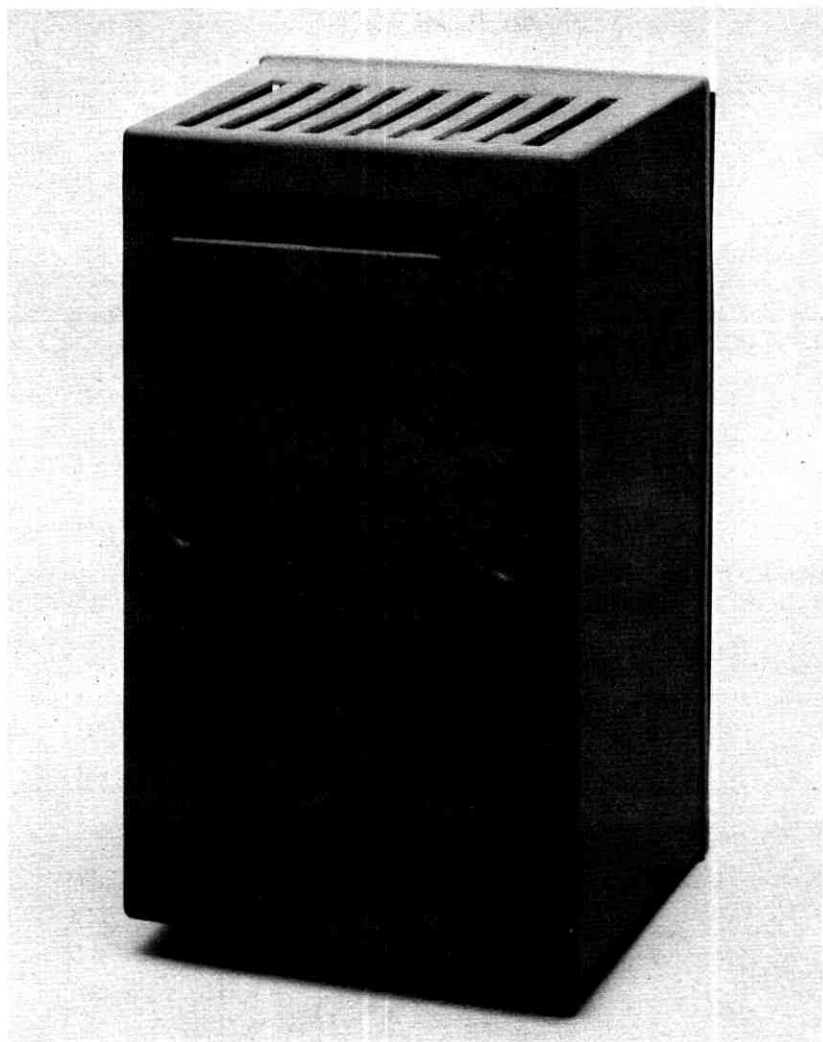


Fig. 3—1A Service Unit (cover on).

An aperture stop of  $f/2.8$  was selected as a compromise between depth of field consideration and the advantages of a small lens. The resulting front glass diameter of 1.65 cm is consistent with styling, graphics, eye contact angle, cost and size objectives. Above an illumination of four foot-lamberts, an automatic iris progressively closes

and thereby increases the depth of field until the 50- to 150-cm range of user motion cited above is accommodated with an illumination of 32 foot-lamberts. For higher light levels, the iris continues to stop down until a maximum design illumination of 520 foot-lamberts is reached. The range of illumination over which the set works exceeds the range of light levels recommended by the Illuminating Engineering Society<sup>4</sup> for office applications.

The lens continuously views the maximum object field. Zooming is accomplished electrically by reducing the raster size on the camera target to scan a portion of the wide angle field. Similarly, adjustment of the effective optical axis for users of different heights is obtained without moving the lens by displacing the reduced raster vertically over the full image field height.

Because of the method used for height adjustment, the maximum object field height is determined by the range of customer heights. Referring to Fig. 4, the 99 percentile large man seated upright measures 68.8 cm from the desk to the top of his head, while the 1 percentile small woman seated relaxed measured 20.4 cm from desk to chin.<sup>5</sup> This range of 48.4 cm is supplemented with vertical margins to frame the subjects and allow for variations in desk-to-chair height differences, yielding a desired vertical field height of 60.7 cm. To accommodate this object field height at a 92-cm viewing distance with an 11 to 10 aspect ratio, a wide angle lens was required with a 53° diagonal taking angle (37° vertical taking angle).

An inverted telephoto lens was designed to reduce the radial varia-

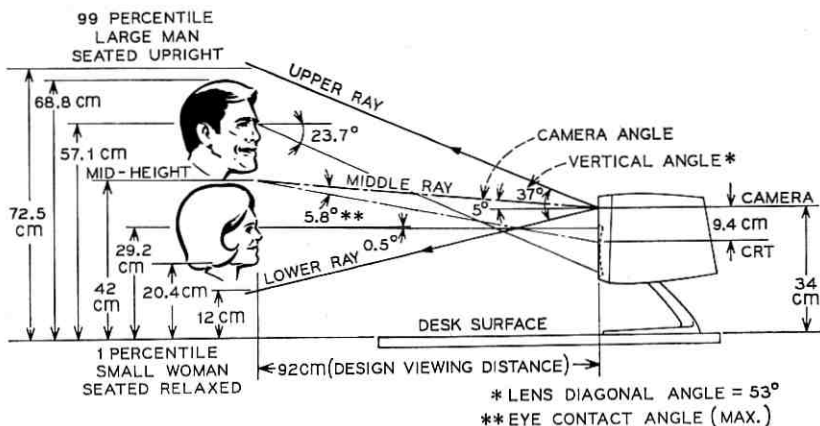


Fig. 4—Human factor considerations for tube placement.

tion in relative illumination typically experienced in a lens due to  $\cos^4$  losses.<sup>6</sup> The six-element lens has a paraxial transmission factor of approximately 80 percent with relative illumination at the wide angle radius falling to approximately 52 percent. In addition, the modulation transfer function (MTF) was shaped differently from that of a conventional photographic lens to maximize the signal for frequencies below 1 MHz. At no point does the MTF fall below 78 percent. With a focal length of 16.35 mm and a magnification ratio of 1/57, the lens produces an image of 1.57-cm diameter, which is equal to the diagonal of the 1.16-cm by 1.06-cm image format on the camera target.

### 2.1.1.2 Tube Placement

The camera is located 33.4 cm above the desk and tilted 5° upwards for the central ray to intercept the center of the desired object field (42 cm above desk height) at the design viewing distance of 92 cm. The placement is a compromise between undue set height and excessive camera tilt which would introduce perspective distortion and the problem of intercepting ceiling lights.

The display tube is placed directly beneath the camera to minimize the eye contact angle. Because the user views the display rather than the camera, the transmitted picture shows the fixation point of the eyes displaced downward. Measurements<sup>7</sup> show that the loss of eye contact is more difficult to perceive on the vertical axis than on the horizontal. The loss of eye contact on the vertical axis is perceptible at about 3° and objectionable at 12°. Fortunately, the center of interest in a display tube—the place where the majority of people prefer to position their eyes in the picture—occurs 10 percent above the tube center. The separation between the camera and the display center of interest is 9.8 cm, as determined by the tube sizes. At a viewing distance of 92 cm, this results in an eye contact angle of 5.8°, which is perceptible but acceptable.

The *Picturephone* screen displays about 250 lines. To obtain satisfactory subjective picture quality at the preferred viewing distance, the display tube height was set at 12.7 cm. The 11 to 10 width-to-height aspect ratio resulted in a 14-cm-wide display. At a viewing distance of 92 cm, the entire display remains within the central 10° arc of best perception<sup>5</sup> (16 cm wide) and permits total viewing without eye motion.

This configuration provides comfortable head inclination for viewing. For a seated person, the typical line of sight is 15° below the horizontal

and the comfortable head tilt is  $\pm 15^\circ$  about that line.<sup>5</sup> Therefore, the display tube should be placed such that the top edge is below the horizontal eye level for the small woman and the bottom edge is less than  $30^\circ$  below the horizontal for the tall man. Within this 24.9-cm range, the display screen placement resulting from the aforementioned considerations achieved acceptable downward angles of  $0.5^\circ$  for the small woman and  $23.7^\circ$  for the large man.

A simple, visor-mounted mirror on the front of the camera provides a capability for transmitting graphics. When the visor is manually lifted, as shown in Fig. 5, the lens automatically refocuses to the plane

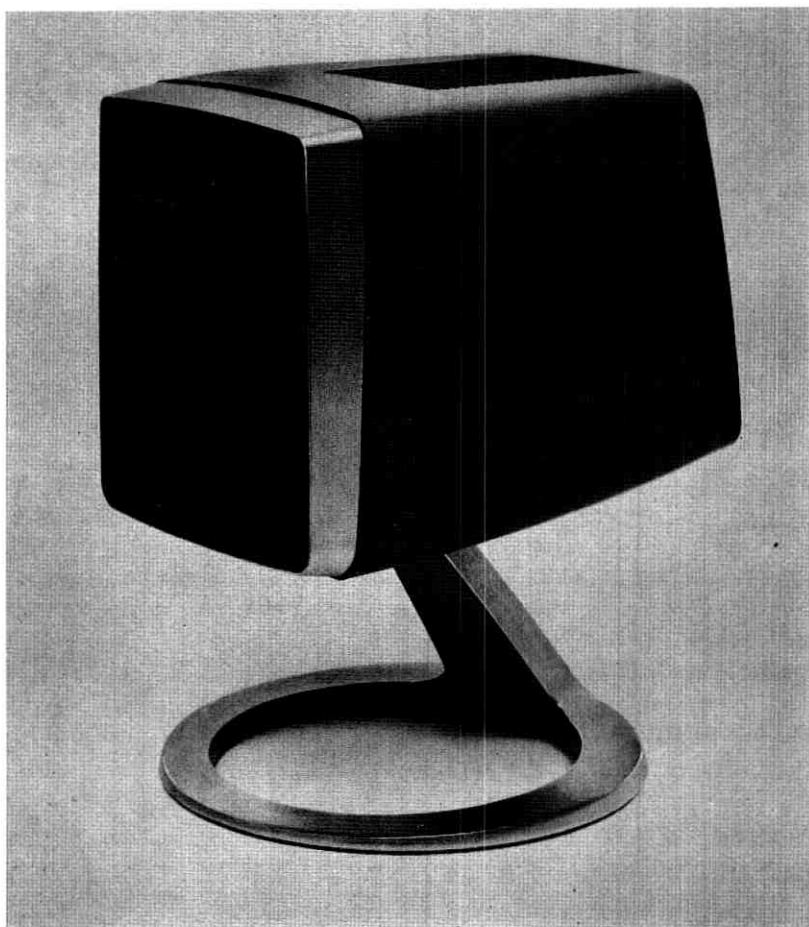


Fig. 5—Graphics visor in open position.

of the desk and a relay reverses the camera horizontal scanning to compensate for the optical inversion caused by the mirror. In addition, the camera raster is automatically centered and changed to the narrow angle mode to increase resolution and hold keystone distortion to 7.5 percent. The graphics object field is slightly trapezoidal due to keystone distortion with a height of 15.5 cm and an average width of 17.3 cm.

### 2.1.3 *Appearance Design*

The appearance design was done in close collaboration between the industrial design firm of Henry Dreyfuss Associates and Bell Telephone Laboratories. The resulting display unit, shown in Fig. 1, consists of a head assembly measuring approximately 19 cm wide, 25 cm high and 30 cm deep, sitting on the sloping pedestal of a ringstand. The aesthetic objective was to avoid a tall and bulky set, prescribed by the tube locations. This was approached by having the head apparently float on the pedestal and by exposing the desk through the ringstand.

The bezel accommodates the display tube, camera lens, graphics mode visor, and grillwork. A loudspeaker is mounted behind the grill for hands-free speakerphone service. The bezel face is painted with a rubbery black finish which reflects less than 2 percent of the incident light and minimizes undesired glare. The face is framed by an accentuated edge which extends beyond any projecting part of the face. This permits the unit to be stably set on the face for removing the housing. The bezel frame is chrome plated with a satin finish to complement the ringstand and to provide a neutral color isolation band between the black face and the variety of business office colors proposed for the rear housing: gray, white, black, green, beige and ivory. In addition, the frame is contoured to provide visual clues to help the user stay centered on camera.

The cover is a one-piece plastic molding attached to the frame with concealed fasteners. The top and bottom surfaces include grills for vertical air flow for cooling.

The ringstand is a satin finish, chrome plated die casting with a hollow-core, skewed pedestal. The cords enter the base at the desk level so the head can freely pan 330° without cord drag. This arrangement also allows for the base to be oriented to achieve the best drape of the cords over the edge of the desk. The ringstand diameter was set such that the outer edge serves as a document positioning aid for the graphics mode, and the pedestal top was located at the center of the ringstand to maintain this feature in any panning position.

## 2.2 Transmitter System

The video telephone transmitter converts an optical image to a composite video signal for transmission over the *Picturephone* network. The transmitter system must automatically adapt to the wide range of illumination normally encountered in an office or home environment, provide the user with the desired controls over his transmitted picture, and process the signal in a manner that will allow a subjectively pleasing display on a *Picturephone* receiver.

### 2.2.1 Camera Tube

The principal component of the transmitter is an electrostatic-focus, magnetic deflection Silicon Diode Array Camera Tube<sup>8,9</sup> designed specifically for the *Picturephone* application. The photosensitive target in this tube is not damaged by high-intensity light sources, has improved sensitivity over conventional targets, and can be selectively scanned to achieve electronic zooming.

Since the spectral response\* of this tube peaks in the 800- to 900-nanometer region, a Schott 2-mm KG3 filter is used to reduce excessive infrared response. With this filter the typical tube sensitivity is approximately 1750 nA per foot-candle in response to tungsten illumination at 2854°K and about 750 nA per foot-candle in response to fluorescent illumination. To compensate for the display tube gamma† of 1.9 and achieve a subjectively pleasing picture, a correction circuit is incorporated in the transmitter video processing circuitry to modify the unity gamma of the camera tube to about 0.7. The combination lens-tube optical sensitivity is about 50 nA per foot-lambert (incandescent) at the maximum lens aperture of  $f/2.8$ . With the 200 nA nominal peak-to-peak signal current at which the tube is operated, scenes with highlight luminances from 520 foot-lamberts to 4 foot-lamberts can be accommodated over the  $f/32$  to  $f/2.8$  iris range. Illumination levels down to 2 foot-lamberts are accommodated by adding increased sensitivity in the form of electronic gain at the expense of signal-to-noise ratio (S/N).

The zoom and height adjustment features are obtained by using the deflection circuits to vary the scanned area and position on the camera tube target. In the wide angle mode the transmitted video

\* Spectral response refers to the output current per unit input light energy as a function of wavelength.

† Gamma as used here is the slope of the transfer function curve relating the logarithmic optical signal and the logarithmic electrical signal (electrical to optical for the display tube and optical to electrical for the camera tube).<sup>10</sup>



signal corresponds to a 1.16-cm by 1.06-cm scanned raster on the tube target. To zoom, the horizontal and vertical deflection currents are simultaneously reduced up to 1/1.6, yielding a raster size of 0.73 cm by 0.66 cm at the narrowest angle and achieving a continuously adjustable zoom over an area ratio of 2.5 to 1. A constant aspect ratio is maintained during zooming by controlling the amplitudes of the horizontal and vertical sawtooth generators in the deflection circuits from a single zoom potentiometer. The height control potentiometer positions the raster vertically between the extremities of the wide angle scan by varying the dc current in the vertical deflection yoke. The user height and zoom potentiometers are coupled to restrict the height adjustment range as a function of zoomed position to remain within the active target area.

The variable scan method of zooming is inexpensive to implement, but system performance is poorer when operating the camera tube in this manner because of sensitivity and resolution considerations. Tube sensitivity, and thus the illumination range which the set can accommodate, is reduced in proportion to the scanned area since less charge per frame is available from the target. The effects of zoom and the spectral content of the illumination upon set sensitivity are summarized in Table I.

Camera tube resolution is limited by the finite size of the scanning beam. This results in an electrical response or modulation transfer function which is near gaussian in shape. When the tube is operated in the wide angle mode, the response to an optical sinusoidal pattern which gives a 1-MHz temporal frequency is between 30 percent and 50 percent (relative to the flat or low-frequency portion of the response curve) with a mean of 38 percent. When the set is used in the narrow angle mode, the optical pattern must be a higher spatial frequency to produce the same temporal frequency. Thus the fixed beam size is larger relative to the pattern it is reading. This produces a temporal MTF shifted in frequency and results in a 1-MHz tube response between 12 percent and 25 percent with a mean of 16 percent. To partially correct for these effects, a linear phase horizontal aperture correction circuit with an 8-dB gain at 1 MHz is incorporated into the camera preamplifier.

The pre-aligned camera tube package is shown in the transmitter block diagram (Fig. 6). The circuits required for tube bias and deflection adjustments have been included in the camera tube package to allow interchanging of tube packages without set alignment. To facilitate this interchangeability, the set provides precision deflection

currents to the camera by means of ramp generators and feedback drive circuits which sense the current in the deflection yokes. The currents are set initially to within 0.2 percent and will vary no more than 0.8 percent with temperature and aging. The linearity of the two sweep currents due to initial tolerance, temperature, and aging will be within 0.5 percent displacement error.<sup>11</sup> Adjustments for variations in horizontal and vertical deflection sensitivity in the tube are accomplished by varying the feedback error signal to the deflection circuits. Raster centering adjustments are implemented with a dc voltage control of the deflection circuits. To eliminate a white border around the displayed picture, referred to as "edge effect," and caused by charge migration from the unscanned to scanned portion of the target, about 2.6 percent of the video information on each side and 0.8 percent on the top and bottom of the scanned raster is blanked out of the transmitted signal. To eliminate vertical ringing bars in the video display, caused by horizontal yoke retrace signals coupling into the vertical yoke, the vertical yoke is center tapped and damped with a RC network.

The series connected camera tube and CRT filaments are each operated at 8 V (1 watt) with 2 percent regulation to enhance tube life. When the set is idle, these heaters are each maintained at 0.8 watts to achieve a picture in less than one second.

Although the silicon camera tube has made it possible to provide the user with adequate *Picturephone* performance, temperature and time dependent spatial dark current variations in the electrostatic focus version of the tube limit the set life and require operation in ambients below 30°C.<sup>9</sup> Studies<sup>9,12</sup> indicate that lower operating voltages will increase tube life appreciably and, consequently, a magnetic focus

TABLE I—CAMERA SYSTEM OPTICAL SENSITIVITY

Illumination	Zoom Setting	Operational Range in Terms of Highlight Luminance (foot-lamberts)		
		Minimum	$f/2.8^*$	Maximum
Incandescent	Wide Angle	2	4	520
	Narrow Angle	5	10	1300
Fluorescent	Wide Angle	4.7	9.4	1220
	Narrow Angle	11.8	23.5	3050

\* Cutover point between AGC and AIC operation.

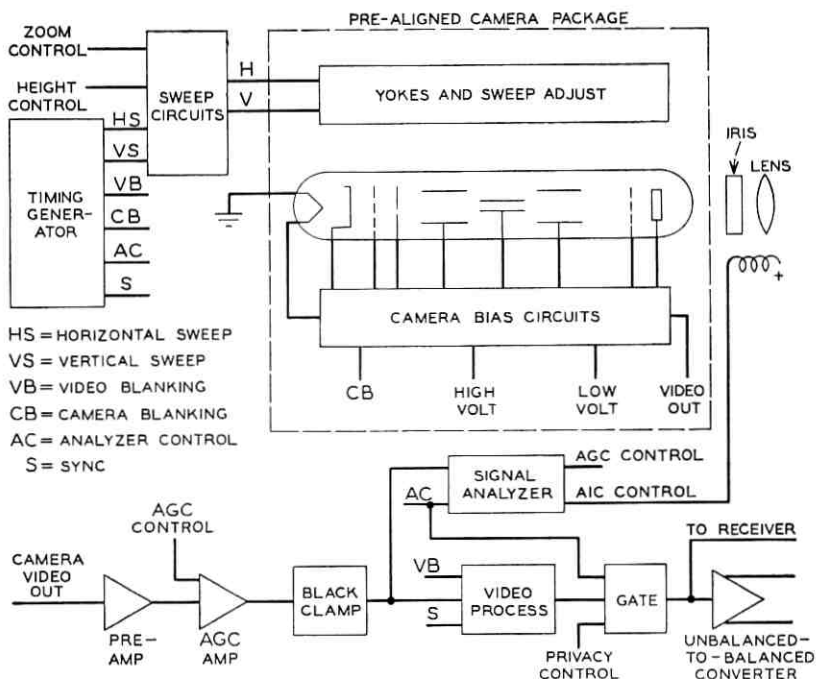


Fig. 6—Transmitter block diagram.

tube with lower operating voltages is being designed and will be used in later sets.

### 2.2.2 Video Circuits

The video circuits in the *Picturephone* transmitter (Fig. 6) amplify, process and control the video signal from the camera tube, add synchronization pulses to form a composite video signal and convert this signal to a balanced format for transmission over the analog loop.

The *Picturephone* timing generator derives the six timing waveforms (Fig. 7) required for transmitter operation by means of a 512 kHz  $\pm 100$  ppm crystal controlled oscillator and appropriate countdown and logic circuitry.<sup>8</sup> The 13.67- $\mu$ s horizontal sweep and 750- $\mu$ s vertical drive waveforms are used to control the respective transmitter deflection circuits. Blanking of the camera tube beam during retrace is provided with the camera blanking signal which is a logical combination of the sweep waveforms. Horizontal timing information at the 8000-Hz  $\pm 100$  ppm line rate is obtained by combining the 8.79- $\mu$ s synchronization pulse, 19.5- $\mu$ s video blanking pulse and the video signal. The

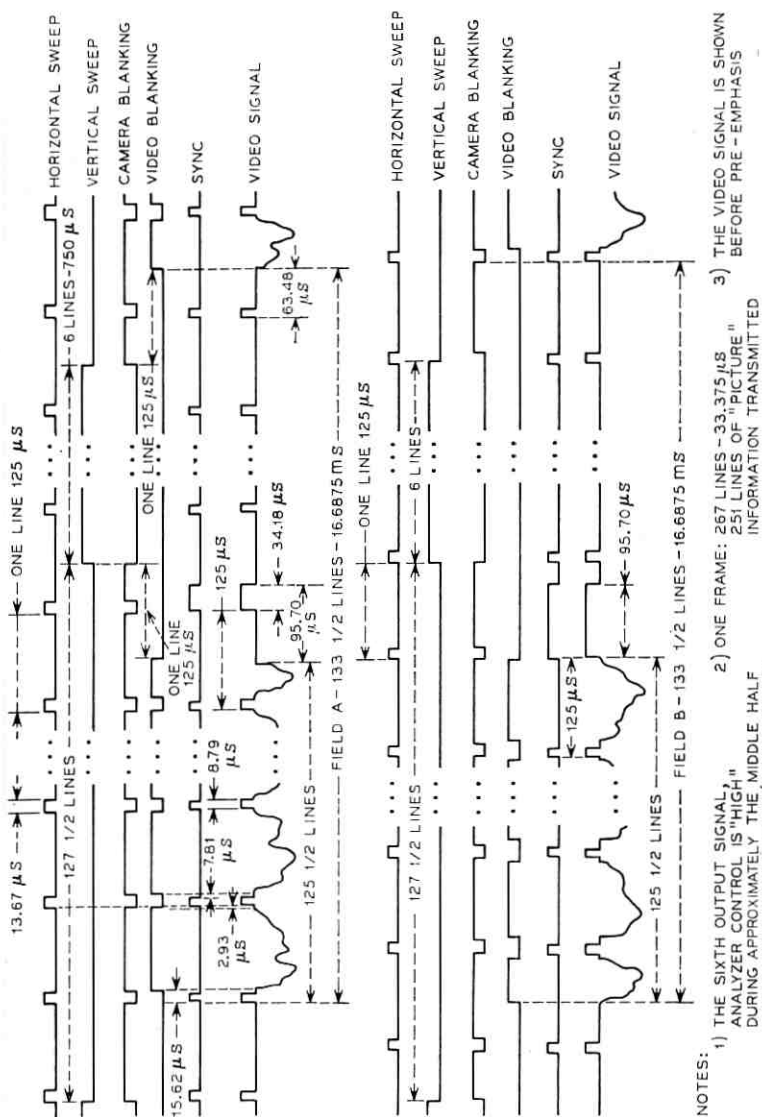


Fig. 7—Picturephone timing format.

analyzer control (AC) signal (not shown in Fig. 7) is positive during the middle half of each field and is used to generate a distinctive video signal for the privacy mode and to gate the signal analyzer circuit as discussed later.

The camera preamplifier and AGC circuit (Fig. 6) amplify the 200-nA peak-to-peak video signal current ( $i_{\text{signal}}$ ) from the camera tube to a voltage level of approximately 5 V peak-to-peak. At this level it is possible to clamp the video signal with negligible impairment.

A low input impedance, cascode type, preamplifier input stage is employed to negate the effect of stray input capacitance and to achieve a flat 1-MHz frequency response. This amplifier employs overall feedback to achieve a nominal 62-dB peak-to-peak composite video signal to weighted-wideband-rms-noise ratio.\* Weighted S/N can be characterized<sup>13</sup> by:

$$S/N|_{\text{weighted}} = 20 \log \frac{1.33 i_{\text{signal}}}{\left\{ \int_0^{\infty} [(I_{eq})_B + I_{eq}(f)]^2 r^2(f) W^2(f) df \right\}^{\frac{1}{2}}} = 62 \text{ dB}$$

where  $(I_{eq})_B$  is the shot noise of the camera beam current,  $I_{eq}(f)$  is the equivalent preamplifier noise current referred to the amplifier input,  $r^2(f)$  is the power transfer function of the station set exclusive of the preamplifier and  $W^2(f)$  is a weighting function<sup>2</sup> to account for the subjective effect of the human eye. With a perfect preamplifier,  $I_{eq}(f)$  equals zero and the weighted S/N would be limited to about 68 dB by the shot noise of the camera tube beam current. A voltage-controlled, variable-gain AGC amplifier employing a field effect transistor (FET) variollosser adds up to 6-dB additional gain under low illumination conditions at a sacrifice of 6-dB S/N performance.

For proper insertion of the synchronization information and analysis of the video signal for automatic level control, it is necessary to re-establish the dc component of the video signal by clamping. The black clamp circuit is gated to sample the video signal during the 105.5- $\mu$ s interval shown in Fig. 8 to establish the darkest portion of the scene at a fixed dc reference. This darkest portion of the signal is processed as black in the gamma correction circuitry and is inserted into the composite signal as black, or at the voltage level of the video blanking pulse. By setting the darkest portion of the scene to black

\* A 62-dB weighted S/N corresponds to a weighted noise power of -54 dBm at the output of a *Picturephone* transmitter when the transmitter pre-emphasis is removed.

it is possible to obtain a subjectively pleasing display and to utilize the full dynamic range of the transmission system independent of scene content.

After clamping, the video-processing circuit combines the 19.5- $\mu$ s video blanking pulse and the 8.79- $\mu$ s synchronization pulse from the timing generator<sup>8</sup> to form the composite video signal shown in Fig. 8. The amplitude of the synchronization pulse relative to porch or pedestal level is used for AGC control in the receiver and is thus maintained at a precise value of  $200 \pm 10$  mV. In order to reduce the susceptibility of the video signal to radio frequency interference and crosstalk during transmission, the video portion of the composite signal is pre-emphasized by 17 dB at 1 MHz relative to 1 kHz.<sup>14</sup> This amount of high-frequency peaking can cause clipping of large, rapid transitions in the video signal in the analog loop equalizers under allowable positive gain-frequency deviations. The subjective impairment of such clipping is acceptable, but if this same clipping were applied to pre-emphasized synchronization pulses, it would impair the recovery of accurate timing

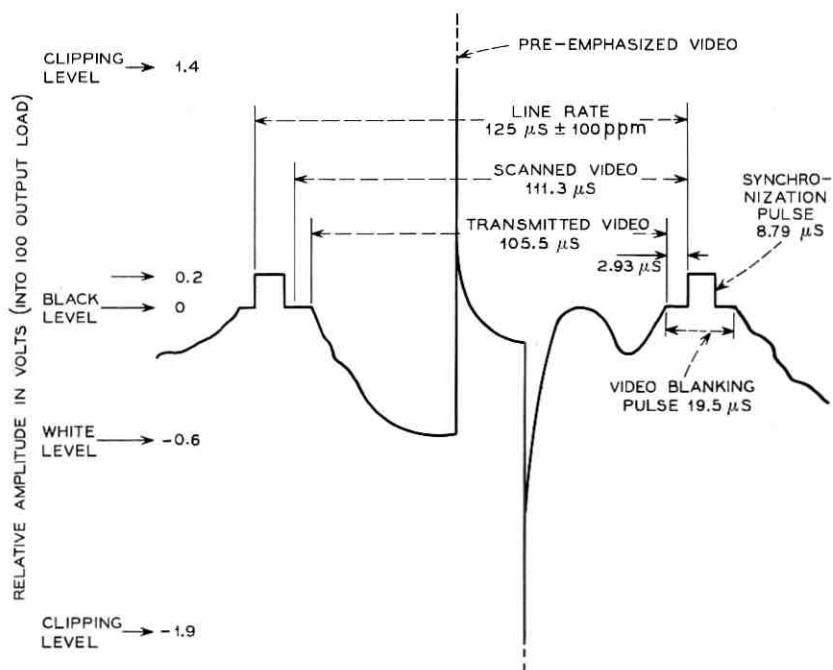


Fig. 8—Picturephone line format.

information at the receiver. A rather simple means is utilized to selectively pre-emphasize the video signal and not the sync signal by integrating the pre-emphasis shaping into the sync insertion portion of the circuit. By introducing controlled clipping of the positive and negative signal excursions in the video processing circuit of the set (Fig. 8), the dynamic range requirement of the unbalanced-to-balance (U-B) converter circuit is reduced. The controlled clipping employed is less severe than the clipping that occurs during transmission.

To implement the subscriber's video privacy feature, a gate circuit, under control of the user's PRIVACY switch, selects either the normal video signal or the ac signal for transmission over the video loop and to the receiver for self-viewing.

The unity gain U-B converter circuit converts the unbalanced video signal from the gate circuit to a balanced signal format for transmission over the video pair. This circuit, which is similar to those used in the transmission equalizers,<sup>15</sup> provides a precise termination of the video pair to minimize longitudinal-to-transverse signal conversion. Conventional carbon blocks connected to the balanced pair and voltage limiting diodes in the output circuit of the converter protect this circuit against transient voltage and lightning surges. These loop protection and termination functions are physically located in the service unit while the active circuitry is in the display unit.

The overall optical-input-to-electrical-output sinusoidal frequency response of the transmitter, excluding pre-emphasis, is nominally flat to 1 MHz. Low-frequency response is characterized by less than 0.3 percent tilt in 100  $\mu$ s to an input step voltage. By carefully controlling capacitor charging in the video amplification circuits during turn-on, a usable video signal is obtained in less than one second while still maintaining excellent low-frequency response.

### 2.2.3 Automatic Gain-Iris Control

In order to achieve a nearly constant video signal level over a 260-to-1 range of illumination, the set utilizes a zero-order, saturating-type, feedback control system. This system employs a combination iris and AGC amplifier (AGIC system) to achieve an advantageous trade off between depth-of-field and signal-to-noise performance. The iris and variable gain amplifier are controlled by analyzing the peak video signal amplitude in the middle 50 percent of the scene and the average amplitude over the full scene.

The basic operation of the AGIC system can be explained with the aid of Fig. 9. At high illumination levels the iris is operated near the

upper end of its  $f/2.8$  to  $f/32$  range, and the gain control amplifier is providing a constant gain and S/N. As the light level of the scene decreases, the iris opens to provide more light to the camera to maintain a constant video signal level. The AGC-AIC crossover point occurs at about 4 foot-lamberts scene highlight luminance (for the wide angle case shown here) where the iris reaches its maximum opening of  $f/2.8$ . The AGC makes up for further reduction in illumination to 2 foot-lamberts by adding up to 6 dB of video amplification. An important feature of this combination gain-iris mode of operation is the continual improvement of set performance as illumination levels are increased. In the AGC range, increased illumination improves the signal-to-noise performance. At the 4-foot-lambert AGC-AIC crossover point, the desired 62-dB signal-to-noise performance is obtained. Increased illumination decreases the iris opening to achieve increased depth of field rather than excess signal-to-noise performance. The effective depth of field is approximately 50 cm at  $f/2.8$  and increases to about 2.5 m at  $f/8$ .

The signal analyzer circuit (Fig. 6) detects the video signal from the black clamp on a combination peak-average basis to obtain an error voltage, compares this error voltage to a precision reference

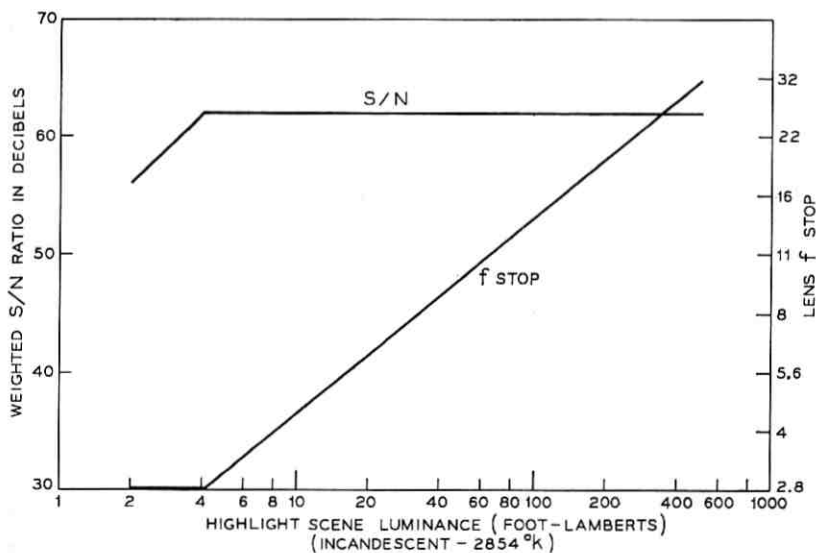


Fig. 9—Transmitter AGIC performance.



voltage and filters the resultant comparator output to derive a dc voltage for control of the iris and AGC amplifier.

The peak-average detector in the signal analyzer is arranged by means of the AC signal to ignore signal peaks in the top and bottom quarters of the scene. This reduces the loss of facial contrast that would occur if the video signal peaks from ceiling lights and white shirts were allowed to set the signal level. With a combination of peak detection over a portion of the scene and average detection over the whole scene the contrast ratio of the transmitted signal will be a function of the scene content. For normal scenes the detector sets the transmitter video output at a nominal 0.6 V peak-to-peak. With black-on-white graphics material, such as a typewritten page, the video output signal is about 0.4 V peak-to-peak. This reduced contrast lowers the display highlight brightness to enhance readability and reduces the impairment caused by transmission clipping of pre-emphasized high-frequency graphics signals.

An RC integrator having a three-second time constant is included at the output of the comparator. This filter inhibits level corrections for short-term scene changes that might occur when a person walks through the camera field of view. Since the iris is included in the AGIC feedback loop, a low friction, nonsticking iris design is essential to guarantee loop stability. With the specially designed *Picturephone* iris, loop stability is dominated by the three-second integrator time constant.

### 2.3 Receiver System

The receiver of the *Picturephone* set processes the incoming composite video signal to produce a high-quality, subjectively pleasing optical display. This processing includes several functions: frequency shaping of the video information to optimize subjective display quality, automatic gain control (AGC) to maintain a display with nearly constant contrast independent of flat gain deviations in transmission, and automatic phase control of the extracted timing information to achieve a stable display. In addition, the receiver system provides the user with the necessary controls over his display and derives the high voltages for camera and display tube operation. A functional block diagram of the *Picturephone* receiver is shown in Fig. 10.

#### 2.3.1 Video Circuits

The nominal 0.56 V peak-to-peak composite video signal received from the output of the receiving transmission equalizer [in the service unit or key telephone system (KTS)] is amplified and converted to an

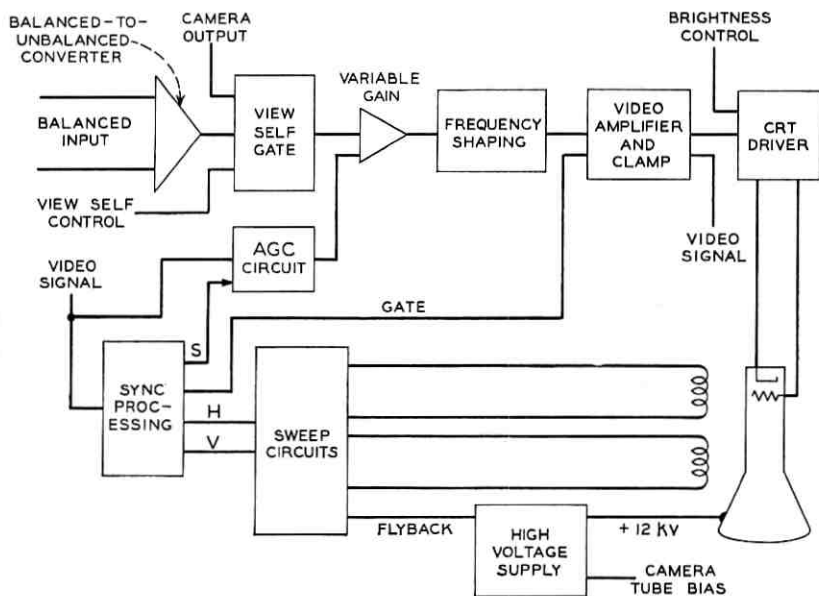


Fig. 10—Receiver block diagram.

unbalanced format by the balanced-to-unbalanced (B-U) converter. This differential amplifier provides a common mode rejection better than 30 dB over the 1-MHz frequency band to minimize the effects of longitudinal or common mode interference signals. The view-self gate circuit selects, under the control of the customer, either the user's camera output signal or his received signal for processing and display. This gate circuit provides a minimum 50 dB of crosstalk rejection over the 1-MHz frequency band which reduces crosstalk below perceptibility.

The receiver employs a sampling type AGC system, comprised of a variable gain amplifier and an AGC circuit, to maintain a constant display over a  $\pm 5$ -dB range of input signal levels which can result from flat gain variations in transmission. This system is arranged to sample the synchronization pulse amplitude, relative to the "porch" level, in order to adjust the gain independent of the video information.

After clamping and de-emphasis of the composite video signal, the AGC circuit samples this signal for  $4.7 \mu\text{s}$  during the "back porch" interval of each horizontal blanking pulse to obtain a measure of the voltage amplitude of the synchronization pulse. This voltage is compared with a precision reference and the resulting comparator output

is filtered to derive a dc voltage for control of the variable gain AGC amplifier. An active multiplier type variable attenuator is employed in this amplifier to maintain the output within  $\pm 0.5$  dB of the nominal signal over the full range of input signal variations. The AGC system provides the proper gain setting in less than one frame time in response to the full 10-dB input signal change, and can therefore accommodate switched conferencing systems without objectionable picture flashes.

One of the problems that had to be overcome in the design of the AGC is the susceptibility to lockup in an incorrect gain setting when large transient input signal changes occur. For rapid increases in signal, the linear amplifiers saturate while for rapid signal reductions improper separation of the synchronization pulses from the composite signal may occur. Both conditions can produce false timing information and result in improper sampling of the video signal. With certain video signals, the derived sample voltage corrects the gain in the wrong direction and thereby produces lockup. To overcome this lockup condition, a special circuit senses saturation or improper timing and overrides the AGC action to properly correct the gain. The lockup protection circuit also senses the absence of an input signal and is used to control the application of a gray raster to the display tube under this condition.

The frequency shaping network de-emphasizes the composite video signal and returns the pre-emphasized transmitter signal to its nominal spectrum. This reduces the effects of transmission noise and interference. Additional response shaping to a prescribed<sup>2</sup> high-frequency rolloff is employed in the receiver to produce a subjectively pleasing picture and to further mask the effect of impulse noise, radio frequency interference and high frequency equalization deviations accumulated during transmission. This response shaping is added by a crispened rolloff filter (Fig. 11, curve c). When this crispened response is added to the 4-dB equivalent electrical rolloff, due to CRT beam size, the nominal overall optics response of the station set is as shown in curve d of Fig. 11. This overall response shape can be electrically characterized in the time domain. For a step input signal, the output transition is symmetric with a 4 percent single pre-shoot and post-shoot and exhibits a 10 to 90 percent rise time of  $0.7 \mu\text{s}$ . The overshoots emphasize the picture contrast at transitions to "crispen" the picture and produce a more pleasing display.<sup>2</sup> The electrical low-frequency response of the receiver is characterized by less than 0.1 percent tilt in  $100 \mu\text{s}$  in response to a unit step input voltage.

The balanced-to-unbalanced converter, view-self gate, AGC amplifier and frequency shaping network are realized in a single hybrid

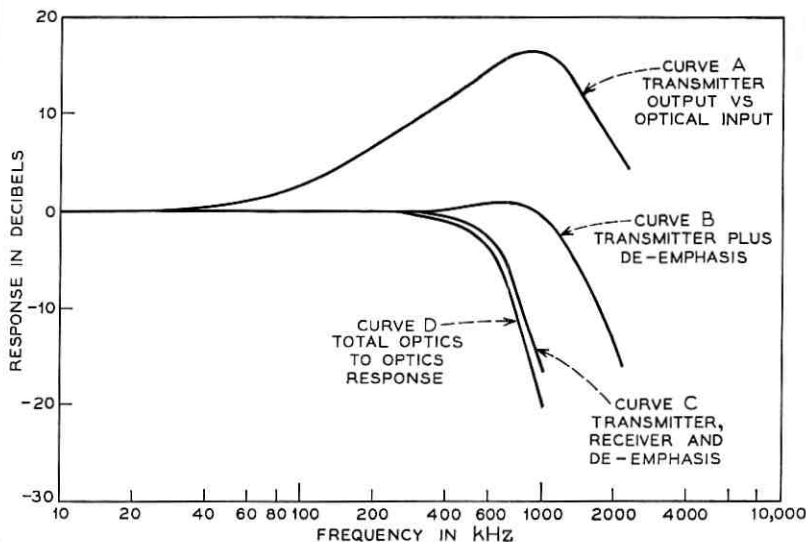


Fig. 11—Picturephone video frequency response.

integrated circuit mounted on a ceramic board containing tantalum film capacitors and resistors.<sup>8</sup>

After frequency shaping, the signal undergoes further processing in a feedback type synchronous amplifier clamp. This circuit clamps on the tips of the synchronization pulses to reduce the effects of system low-frequency rolloff and low-frequency crosstalk (power hum at 60 Hz is reduced by 35 dB).

The CRT driver circuit provides the video signal for the CRT grid (G1), brightness control and vertical retrace blanking. The G1 drive circuit has an adjustable gain which is matched in set manufacture to the tube transconductance to reduce contrast variability. User control of the display highlight brightness over a 40- to 110-foot-lambert range is accomplished by varying the tube cathode voltage. In the absence of an input signal the circuitry provides a gray raster display to indicate that the set is operational and to eliminate transient picture flashes during call setup. To protect the CRT from phosphor damage, the circuitry is arranged to rapidly cut off the beam when the set is turned off. In addition, limiting diodes are used to prevent circuit damage from internal tube arcing.

### 2.3.2 Synchronization, Sweep and High Voltage Circuits

The sync processing circuitry removes the timing information from the composite video signal, separates the horizontal and vertical

timing and provides stable noise insensitive synchronization pulses to the sweep circuits. These synchronization circuits are designed to provide a stable display without user adjustment over the life and operating extremes of the set.

Sync separation from the video signal is obtained by amplitude discrimination at the midlevel of the synchronization pulses. To improve the stability of timing information, separate sync amplification and clamping of the AGC corrected signal are employed. A Schmitt trigger gating circuit, which requires a 2- $\mu$ s sync separator output signal to indicate valid horizontal timing, is used to guard against impulse noise. A second Schmitt trigger circuit, which is arranged to require a 25- $\mu$ s signal, is used to separate the 34.18- $\mu$ s and 93.70- $\mu$ s vertical pulses (Fig. 7) from the horizontal pulses. A conventional APC voltage-controlled oscillator is used to lock on the recovered horizontal timing information and reduce timing jitter due to transmission and coding noise. The voltage-controlled oscillator is realized with silicon integrated circuit chips mounted on a tantalum thin film substrate.<sup>8</sup> Both the horizontal and vertical oscillators are arranged to free run to provide raster scanning in the absence of an input signal.

The horizontal and vertical sweep circuits provide corrected deflection currents to the CRT yokes to compensate for the S-distortion<sup>10</sup> which results from the nonspherical face of the CRT. In addition, the five-ampere horizontal current is provided with correction to cancel the distortion caused by ohmic losses in the yoke. Linearity adjustments are not necessary due to the utilization of precision components. Protection of the horizontal sweep transistor, which switches the five-ampere deflection current, is accomplished by arranging this transistor to turn off if there is a loss of timing. The display overscan margin is kept below five percent to increase the information content of the display, particularly for computer-access readouts. Aspect ratio error is held to less than one percent and geometric distortion does not exceed two percent.

A flyback type high-voltage power supply is integrated into the horizontal sweep circuit to provide the required CRT and camera tube bias voltages. This circuit is provided with a safety interlock to disable these voltages when the display unit housing is removed. The 12 kV ultor potential is derived by rectifying the third-harmonic-tuned flyback pulse, at the secondary of the flyback transformer, with a 1X2B vacuum tube diode. To maintain precise raster size and aspect ratio the 12 kV is regulated to within two percent by means of a corona discharge regulator. To reduce shock hazard, a 3 G $\Omega$  resistor discharges the 500 pF CRT aquadag capacitance when the set is turned

off. Radiation measurements show that the set does not produce any detectable radiation above the natural background (0.02 mR/hr).

The camera tube and CRT bias voltages are obtained by rectification of primary voltages from the flyback transformer. The CRT control grid voltage is adjusted during final set alignment to match the CRT voltage-cutoff characteristic.

Extreme care was taken in the design and mechanical layout of the high-voltage and horizontal-sweep circuits to reduce the coupling of these high energy circuits into the transmitter. Direct electrical and electromagnetic coupling was reduced by providing these circuits with a separate low-voltage regulator, segregating power feeds, eliminating circulating currents and damping flyback pulse ringing.

#### 2.4 *Mechanical Design*

The display unit without its cover is shown in Fig. 12. It is assembled from a number of interchangeable modular subassemblies as shown in Fig. 13 and discussed individually below.

##### 2.4.1 *Structural Framework*

Two aluminum castings of 360 alloy are used as the primary structural members to achieve low cost, uniform assembly and electrical shielding. The front casting accommodates the CRT-yoke assembly and aids in the shielding of the camera from the yoke fields. The rear casting shields the camera from the high energy sweep circuits and high-voltage power supply. A third nonstructural casting is used to encapsulate the flyback transformer. Collectively, the castings provide an 18-dB advantage in electrical isolation over that achieved with a conventional framework. This is important since the camera signal current is unbalanced and approximately 100 dB less than the CRT deflection current and the two tubes are physically adjacent in the display unit.

##### 2.4.2 *High-Voltage Assembly*

The high-voltage assembly consists of the flyback transformer, rectifier tube, ultor voltage regulator and associated circuitry. As mentioned, the transformer uses a separate 360 aluminum alloy casting for ease of potting and curing. For safety, the casting is grounded and the 12-kV anode lead is provided with a back-up screw lock to prevent inadvertent disconnection. Solventless silicone resin is used as the potting material with silicone room temperature vulcanizing

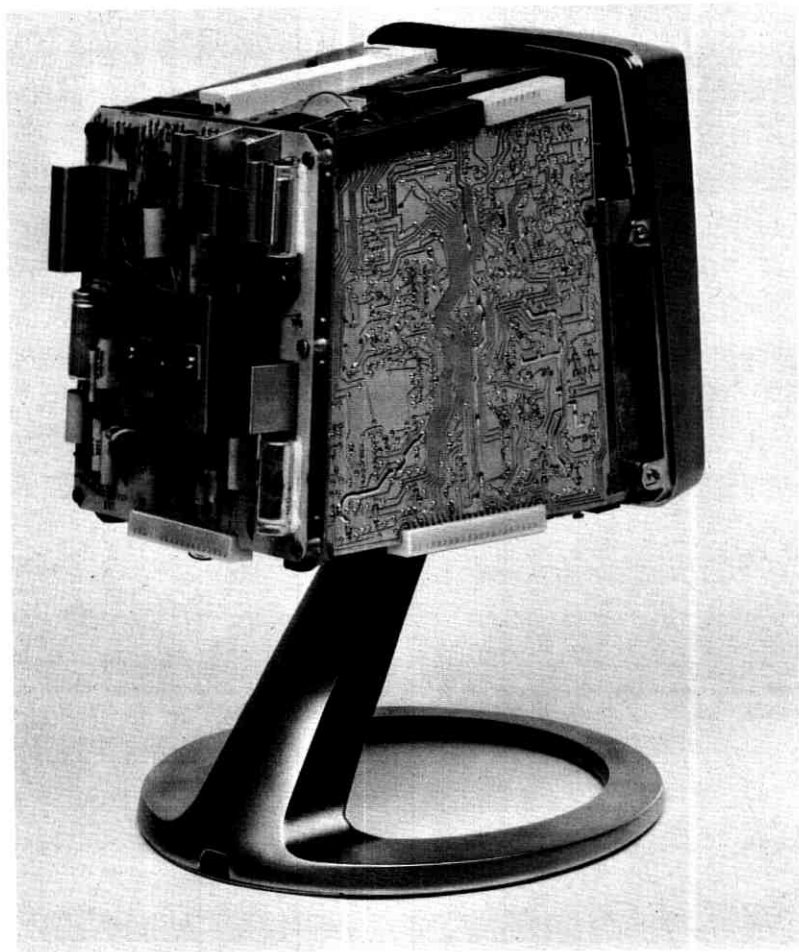


Fig. 12—1A Display Unit (less cover).

(RTV) rubber coatings at the critical interfaces to achieve arc suppression, corona prevention and 8-kHz noise control.

The horizontal sweep rate of 8 kHz is in the audible range and components in the horizontal deflection circuit produce a subjectively objectionable hum. This is true of the flyback transformer and several inductors, which have ferrite cores and expand magnetostrictively at the 8-kHz rate, as well as a laminated core choke and several high-voltage capacitors which vibrate electrostatically. The problem was

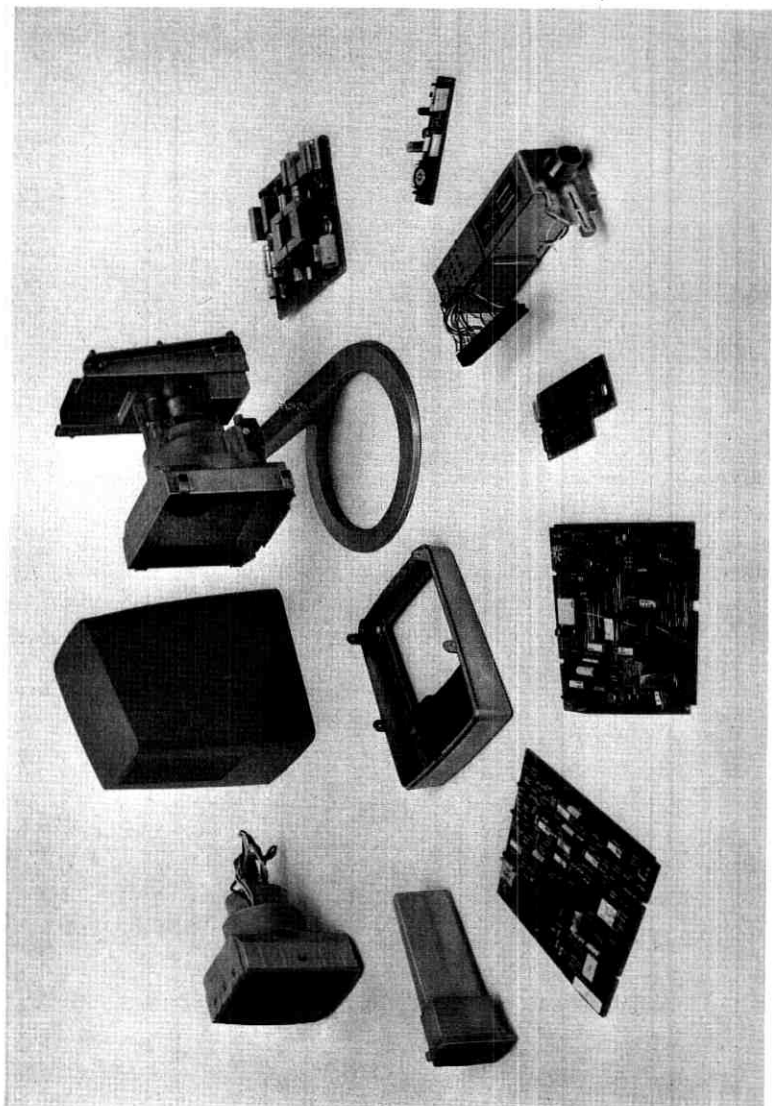


Fig. 13-1A Display Unit (subassemblies).



resolved by enclosing each offending component in an oversized stiff case to achieve an air shunt between the noise source and case.

#### 2.4.3 465A Display Tube<sup>8</sup>

The packaged display tube is designed to fit into a CRT casting which has limited rear access to the alignment magnets and anode, yoke and stem leads. The tube funnel region and most of the stem is encapsulated with rigid polyurethane foam. This permits pre-centering of the display tube in the foam, gives better dimensional control than that obtained with the glass tube envelope, allows the use of molded deformation wedges to compensate for tolerance build up when mounting, and holds the prealigned yoke in position to avoid slippage or subsequent misadjustment. The foam also provides two-fold protection: to the user by reducing implosion danger and to the tube itself by reducing handling damage.<sup>10</sup> Customer safety from accidental tube implosion is further improved with a safety panel bonded on the face with clear polyester. The panel has an etched surface to disperse incident light and a transmission factor of 30 percent to improve contrast by having reflected light attenuated twice. The refractive index of the polyester was selected close to that of the glasses used to minimize ghost images at the interfaces. Nonbrowning glass was used for the tube face to resist electron bombardment discoloration. The anode lead is attached to the tube to minimize moisture, dust and safety problems. Likewise, the stem and yoke have flying leads which are stress relieved to minimize lead breakage.

#### 2.4.4 466A Camera Tube

The camera tube is packaged in a rectangular mu-metal shield for interference control. An aluminum front plate assembly mounts the lens and iris. Effort was made to keep the camera temperature low since the target dark current doubles for each 9°C rise and thereby reduces the useful dynamic range of the tube. This was partially achieved by connecting the target end of the tube to the cast zinc bezel to take advantage of its high-thermal capacity. The gain of this thermal shunt is degraded by 3°C due to the need for electrical and vibrational isolation between the tube and bezel. The camera temperature rises exponentially during continuous operation, reaching a steady state value of 11°C above ambient after 5 hours.

The camera biasing circuits have been included on two small printed wiring boards in the package.<sup>8</sup> A separate metal shield to accommodate the video preamplifier is attached to the camera package.

### 2.4.5 Iris

The iris, functioning as an element in the video feedback loop, must be stable with time. To accommodate this, an iris having no mechanical friction was designed<sup>17</sup> using the principles described below.

Referring to Fig. 14, two phosphor-bronze cantilever arms, A and B, are mounted to base C and connected by a thin platinum-nickel wire, D. The ends of the cantilever arms form the mating halves of the aperture. By foreshortening the thin wire, the arms are pulled together to open the aperture. This is achieved without mechanical friction by using a taut-band suspended meter movement, E, with a double post actuator, F, attached to it. Since the thin wire is in equilibrium, the tension up and down is equal. As the actuator rotates, the wire progressively wraps around the posts without sliding, foreshortening its length without relative motion, hence without mechanical friction between parts. Worst-case accelerated life testing of 6 units resulted in an average of 7 million operations to failure with a worst-case MTBF of 6000 hours.

The aperture halves are shaped to provide the desired optical-electrical-mechanical transfer function. When off, the iris is closed to its smallest stop, approximately  $f/32$ .

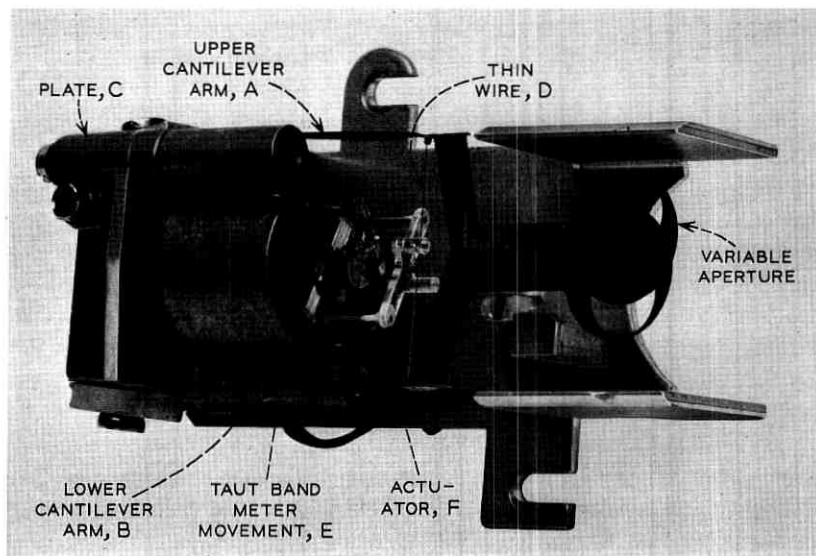


Fig. 14—Iris assembly.

The iris assembly is mounted to the camera plate with the aperture halves entering a cutout in the side of the lens. Thus the lens moves relative to the iris blades and is at the optimum nodal location in only one position. This degradation in iris location was evaluated using a lens design computer program and found to be an acceptable compromise for the advantage of permanently mounting the iris.

#### *2.3.6 Printed Wiring Boards*

The three major printed wiring boards average 210 square cm each and are made of 0.16-cm-thick G-10 glass epoxy material for strength. The transmitter and receiver boards are retained at four points with the circuit paths facing out to protect against stripping components when the cover is installed.

The printed wiring boards are single sided for reliability and lower cost. Low-resistance straps are used where necessary to avoid double-sided boards with plated-through holes. Wherever possible, components are mounted vertically to achieve improved heat transfer and large components are secured with an RTV adhesive to prevent possible damage during shipment. The wiring side is coated with an acrylic spray finish for moisture and fungus control.

The wiring boards are interconnected by right angle, edge mounted connectors. A common mounting block permits inserting a family of gold-plated, beryllium-copper contact members having different gauge wires.

#### *2.4.7 Shielding and Grounding*

Considerable attention was given to the mechanical design to reduce electrical interference to an acceptable level. Some of the steps taken to reduce interference have been discussed. In addition, the topographical layout utilized the available space to separate the receiver and transmitter circuits as much as possible. The printed wiring boards were laid out with special attention given to signal level progression, types of adjacent circuitry, parts orientation, low impedance ground paths, and connector land assignments. Equal care was exercised in the harness design to separate and shield wires which were subject to crosstalk.

#### *2.4.8 Bezel and Graphics*

The bezel is a zinc die casting to permit satin finish, chrome plating of the frame. The front area is painted black for glare control and good contrast. The bezel mounts to the CRT casting, retains the CRT and provides a mounting surface and graphics facility for the camera.

A capability for displaying graphics is obtained by lifting a spring loaded visor. This inserts a mirror in the optical path of the camera and directs the field of view to the desk top. Precise mirror orientation can be made in manufacture with a front adjustment.

The use of a mirror in the graphics mode introduces an optical inversion which is corrected by reversing the horizontal yoke leads. This is initiated by a visor arm actuating a switch. Simultaneously the raster is centered and shifted to the narrow angle. To reduce electrical interference, the switch and lens actuators for refocusing in the graphics mode are insulated from the bezel visor.

#### 2.4.9 *Audio*

A 6.35-cm-diameter speaker is located behind the protective grill on the right-hand side. This relatively small speaker has a 300-cubic-cm back chamber to extend its low-frequency response. It is driven by an auxiliary audio amplifier which adds 6 dB of gain with bass boost frequency shaping.

The output of the speaker introduces considerable radiated and mechanically coupled vibration into the structural framework, requiring isolation of the camera to reduce microphonics below a perceptible level in the video signal.

The tone ringer is mounted on the rear casting and the alerting sound emanates from the vents on the bottom of the housing.

#### 2.4.10 *Panning Mechanism and Manual Tilt*

The structural framework of the display unit attaches to a tilting mechanism for manually adjusting the elevation during initial installation over a 10° vertical range and a panning mechanism for rotating the head through 330°. The thrust bearing design utilizes a differential load near the center of gravity to reduce the torque required to rotate the head. The design accommodates large dimensional variation due to manufacturing tolerance and wear.

#### 2.4.11 *Ringstand*

The ringstand is a satin finished, chrome plated zinc die casting which weighs 2.15 kg and helps lower the center of gravity of the set. To protect against a set being tipped over or pulled off the desk by the cord, the ringstand includes a dual base.<sup>18</sup> The inner surface in contact with the desk is a material with a high coefficient of friction which resists slipping. However, for impacts above the center of gravity, this material holds and becomes an objectionable fulcrum about which the entire set tips. Since desk top slipping is less damaging than tip-

ping, an overhanging edge with a low coefficient of friction is provided. After the set rotates through a small angle after impact, the second surface touches and provides slipping rather than continued tipping.

#### 2.4.12 *Harness and Cord*

The display unit harness passes through the panning mechanism and the hollow core of the ringstand pedestal and terminates in a connector to mate with the cord. The harness can be readily replaced since the end connector can be pulled through the pedestal and panning mechanism. Since the head rotates and the ringstand is stationary, fatigue tests were run on the harness to determine a lacing and mounting arrangement to achieve lead life beyond 100,000 panning operations.

The interconnecting cord is color coordinated to the sets and is of a "V" construction with a 1.2-meter section of 26-lead cordage going to the control unit and a 3-meter section of 50-lead cordage going to the service unit. The 50-lead sections terminate in connectors and permit the use of B25A extension cable. The junction of the "V" construction enters the bottom of the ringstand to connect with the harness. To connect the set, the harness connector is pulled slightly from the base for cord insertion. Thereafter, the mated connector pair is pushed back into the base where it cannot open due to the pedestal width. Subsequently, a cover plate—designed to avoid damage to the desk—is attached to retain the connectors.

#### 2.4.13 *Complete Assembly*

The complete display unit assembly weighs 11 kilograms. In final testing the unit successfully passed a 35 g shock test with six repetitions of a 6.5-millisecond sawtooth pulse in each direction of three mutually perpendicular planes, for a total of 36 drops. Two different tests were used to confirm its vibration endurance. In the first test, the unit was sinusoidally vibrated at 0.05 inches double amplitude from 10 to 31 Hz and at 2.5 g from 31 to 200 Hz at a sweep rate of approximately 4.5 octaves per minute for 60 minutes in each of three mutually perpendicular planes. In the second test, a random input of  $0.04 \text{ g}^2/\text{Hz}$  (equivalent to 2.75 rms g level), clipped at  $3\sigma$  (8.25 g rms), was applied over a bandwidth of 10 to 200 Hz for 20 minutes in each of three mutually perpendicular planes. Experience gained in actual shipment confirms the validity of these laboratory results. In operation, the unit functions satisfactorily in a relative humidity of 95 percent and reaches an average steady-state temperature of approximately  $20^\circ\text{F}$  above ambient.

### III. CONTROL UNIT

#### 3.1 *General Factors*

To bring the controls and audio input closer to the customer than the display unit, a separate, small control unit is provided as shown in Fig. 2. This supplements the 12-button *Touch-Tone* telephone set which is necessary for establishing calls for regular telephone service and for privacy when the speakerphone feature is not desired.

The control unit has a family resemblance to the display unit, using a slightly concave black face with an accentuated edge around the perimeter. The molded plastic cover is available in the six matching business office colors. A satin finish aluminum base completes the aesthetic parallel. The composite design achieves a low profile by the use of rocker switches and thumbwheel controls.

#### 3.2 *Audio*

The AF-1 electromagnetic transmitter developed for this application is located behind the small port on the front of the control unit. The microphone measures 2.2 cm in diameter and 0.89 cm in depth. It is suspended in a rubber isolation boot which positively seals the microphone input to the base while providing isolation from microphonic noise above 40 Hz. The housing-to-base gap is sealed on each side of the microphone to avoid deleterious chamber effects.

The microphone is flexibly connected to the audio preamplifier which provides frequency shaping and 75 dB of audio gain. The preamp shares a multipurpose wiring board, shown in Fig. 15, with the video controls and function switches.

#### 3.3 *Mechanical Design*

The rocker type function switches move shorting wipers across commutator lands on the printed wiring board. This eliminates all wires, soldered connections and conventional contact adjustments, resulting in low cost and high reliability. The rocker switches also eliminate the need for lamps since the state of the switch is indicated by its position. The single exception is the ON/OFF switch where compatibility with the 3A Speakerphone<sup>s</sup> circuitry required momentary makes and breaks with the subsequent need for an alerting lamp.

To avoid inadvertently leaving the PRIVACY or VU-SELF switches in the operated position, mechanical linking is provided between the switches such that they are returned to their normal states when the OFF button is actuated.

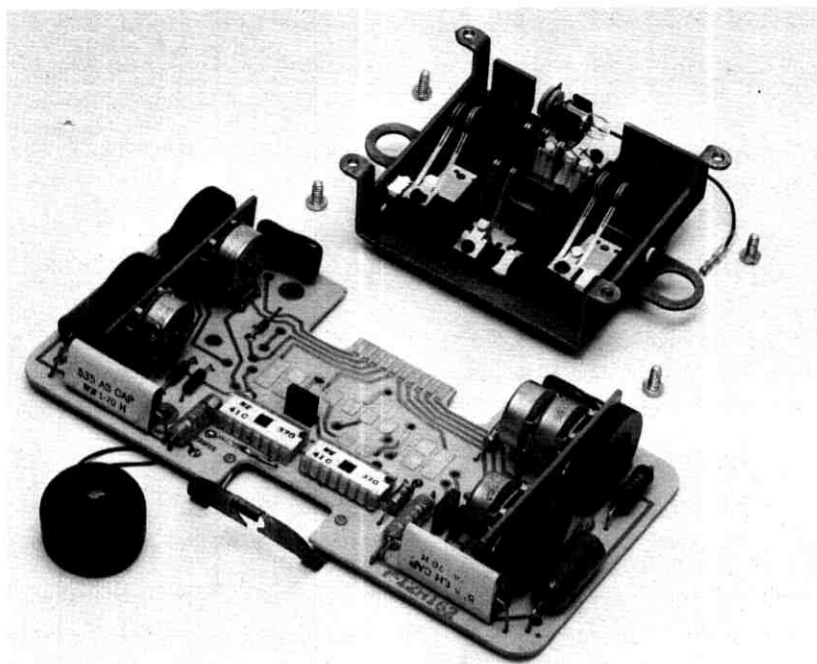


Fig. 15-72A Control Unit (subassemblies).

All of the switches and potentiometers mount directly to the printed wiring board for alignment accuracy and the mounting cord connects directly to gold plated, printed lands. For space reasons, a double-sided, G-10 glass epoxy board was used. The cover and base clamp on opposite sides of the wiring board and the complete assembly is held together with two screws. The complete assembly weighs 142 grams.

#### IV. SERVICE UNIT

##### 4.1 General Factors

The 1A Service Unit shown in Fig. 3 is the third major assembly required for *Picturephone* service. Since the customer does not interact with it directly, it is designed to be mounted remotely with a cable run up to 26 meters from the display unit. It has a stipple finished gray housing measuring 30 cm high, 18.1 cm wide and 15.9 cm deep, and is

planned for vertical installation to minimize the projected floor area required. Since this is the only unit connected to a 110 V power source, the 7.7-kg assembly mounts to a metal wall plate and has a fire resistant cover for safety. All adjustments and class of service options are located here to avoid the necessity for opening the display and control units during installation.

## 4.2 Circuitry

### 4.2.1 Functional Description

The service unit contains the power circuits which convert the subscriber's local 110 V to several well regulated dc voltages of lower potential for use in the set; control circuits which detect and process the audio and video supervision signals to control the display unit; and a transmission equalizer<sup>15</sup> which adds frequency shaped gain to correct for the insertion loss of the receiving video pair. A functional block diagram of a service unit as arranged for a non-key telephone installation is shown in Fig. 16.

The basic system plan for station supervision of the video telephone involves the use of normal off-hook and ringing derived from the audio

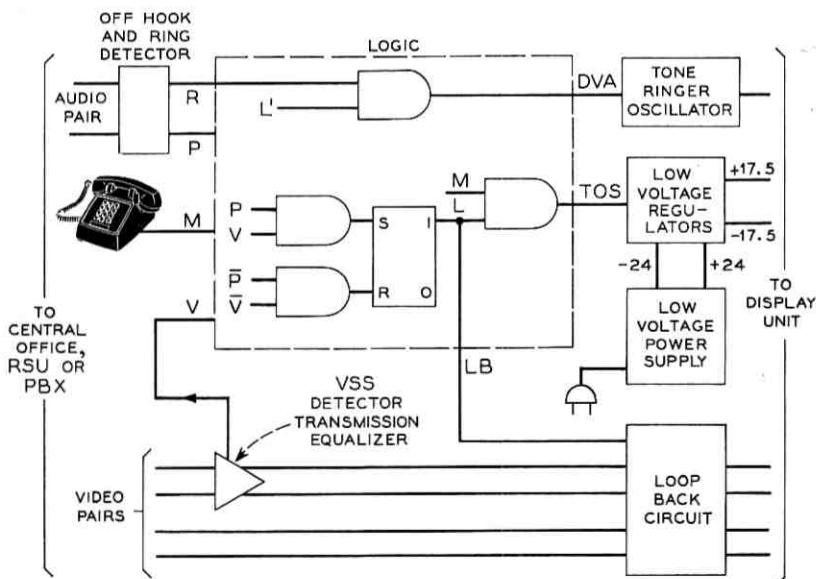


Fig. 16—Service unit block diagram (for single line arrangement).



pair and a video supervision signal (VSS) derived from the receiving video pair. An off-hook and ringing detector and a VSS detector sense the audio and video supervision signals to provide ring, off-hook and video indications to the set logic. The logic circuitry combines these input signals and the main station switchhook indication to derive three control signals required for set operation. The distinctive video alerting signal (DVA) controls a tone ringer oscillator which drives the sounder in the display unit for audible alerting on video calls. A turn-on-set (TOS) signal activates the low-voltage regulators in the service unit which power the circuits in the display and control units. The loop-back (LB) signal controls the maintenance loop-back circuit which provides an equalized loop-back of the video pairs to the central office for maintenance purposes and also connects together the balanced input and output of the display unit when the set is idle.

#### 4.2.2 Control Circuits

The circuits required for proper logical control of the display unit are: the off-hook and ringing detector, VSS detector, loop-back circuit and the service unit logic circuitry.

The off-hook and ringing detector<sup>s</sup> consists of a high input impedance differential amplifier which bridges the audio pair and two separate detector circuits. This amplifier responds to voltage changes at its input and provides the detectors with complementary outputs which follow the input voltage changes. The ringing detector reshapes the ringing signal to provide a square wave logic signal (R) that follows the 20-Hz ringing. The off-hook detector is arranged to provide a dc output indication (P) under low voltage input conditions to the amplifier and thus indicates the switchhook status of both the main station and audio extensions. Low pass filtering is used to eliminate response due to the ringing signal. A 60-ms minimum time delay in the off-hook to on-hook indication is included to guard against response to transient excitation of the audio pair.

A VSS detector, which is included as part of the transmission equalizer, samples the VSS or the video signal on the receive pair at an unbalanced equalized point in the equalizer. The 8-kHz content of this signal is filtered and rectified to derive a logic signal (V). The detector has a Q of 40 and a response threshold to a sinusoidal input signal of 10 mV rms. Although this detector will respond to all video signals as well as VSS, the system arrangements never rely on using the video signal in place of VSS for control of the set.

Station logic is summarized in the state diagram (Fig. 17) which

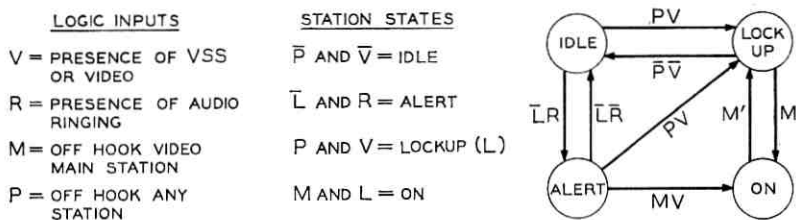


Fig. 17—Picturephone logic state diagram.

gives the relationship between the P, M, R, and V logic inputs and the ON, ALERT, LOCKUP, and IDLE station states. The detailed interaction of the station and the switching system during calling procedures is outlined in Ref. 19. Since the VSS signal is never maintained after the called party answers, the set logic must supply memory of a video call. This memory or video lockup state (L) is implemented with a set-reset type flip-flop which is set by the simultaneous occurrence of any audio or video party off-hook (P) and video supervision (V). The station set remains in the lockup state until all audio and video stations on the video line are on-hook ( $\bar{P}$ ) and there is no V indication ( $\bar{V}$ ), at which time the flip-flop is reset and the set is returned to the idle state. At any time during lockup the video circuitry may be turned on when the main video station goes off-hook (M).

During the idle state the tone ringer may be activated with the ringing logic signal (R) which contains the 20-Hz interruption to properly modulate the tone ringer circuits. The 60-ms delay in the off-hook to on-hook transition delays the enabling of the tone ringer upon return to idle ( $\bar{L}$ ). This avoids ring tap which would otherwise occur because of extraneous response of the ringing detector to the on-hook indication. The volume of the tone ringer may be adjusted by the installer to one of three levels to accommodate customer preference and room acoustics.

The maintenance loop-back is on during the idle state and removed during lockup. The loop-back is not replaced prior to entering the idle state because of the possibility of a calling party receiving his own picture through the called party's loop-back. Loop-back is implemented with a relay circuit. The switching systems are arranged to perform a continuity test through the station loop-back prior to setting up a video call. During a local power failure the continuity test will fail because of the unpowered equalizer, and therefore call completion and video charging are inhibited. A *Picturephone* sub-

scriber's normal telephone service is not interrupted by such a power failure although the speakerphone is inoperative.

The service unit circuitry arrangements just described are compatible with single-line video service and will allow the connection of one video unit and the normal complement of audio extensions. Service requirements for multivideo pickup of a single video line and/or multi-line access by a single station are accommodated with a video KTS.<sup>20</sup>

In a station set which operates behind a KTS, the service unit is equipped with different plug-in circuit boards to provide a simplified arrangement (Fig. 18). The audio and video supervision detection, logic and loop-back functions are removed from the service unit and the transmission equalizer is replaced by a passive build-out network. Set turn-on is controlled directly from the KTS by the TOS lead which, as before, activates the low voltage regulators. The DVA signal is standard KTS ringing applied to the tone ringer via a balanced ringing pair when distinctive alerting is required. An indication of a loss of station set power is provided to the KTS by means of the PF control lead. In such an event the KTS cancels call completions by arranging for continuity loop-back test failure as was done in the single video-line arrangements. The build-out network is selected at the time of set installation from one of four possible networks so as to achieve a nearly constant electrical transmission loss between the KTS and the station. This loss is pre-equalized in the KTS equalizer.<sup>15</sup>

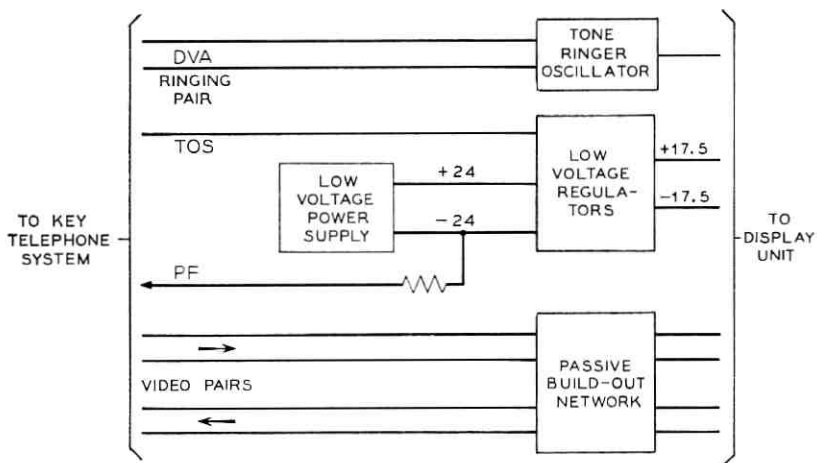


Fig. 18—Service unit block diagram (for key telephone arrangement).

#### 4.2.3 Power Circuits

The primary power supplies for the station set are located in the service unit and convert the subscriber's 110 V local ac power to low voltages for display unit operation. This location insures customer safety, since 110 V wiring to the subscriber's display unit is not required, and convenient installation, since it is generally possible to find an existing power outlet within a 26-meter cable run to the display unit. To achieve the well regulated voltages required for the video circuits over the 24 V range of line voltage variations, a ferro-resonant power supply is used. The service unit has been submitted to Underwriters' Laboratories for listing.

A pair of series-type voltage regulators derive the more precise voltages required for video circuit operation from the  $\pm 24$  V ferro-resonant power supply. Solid-state low-power on-off switching of the video units by the set logic is accomplished in the regulators by controlling the base current in the series regulating transistors. By performing more precise regulation in the service unit it is possible to reduce overall display unit power dissipation which is 4 watts in standby and 40 watts in service. The service unit dissipates 21 watts in standby and 35 watts in service.

Pair assignment in the 50-pair interconnecting cable between the service unit and display unit was made to reduce transmit-to-receive video crosstalk and audio-to-video crosstalk. This assignment includes separate power feeds and grounds for the receiver and transmitter circuits. Pair quantity in the power feeds is allocated to maintain one-volt regulation at the display unit over all interconnecting cable length variations. Inadvertent disconnection of the interconnecting cable or a loss of either service unit fuse is sensed by a relay circuit in the service unit which automatically switches off all dc power. This inhibits the successful completion of the continuity loop-back test and thereby signals the switching system to cancel the completion of calls under these conditions.

#### 4.3 Mechanical Design

The primary framework for the service unit is a one-piece *U*-shaped structural member made of 0.236-cm, cold-rolled steel with a zinc-chromate finish. Mounting tabs, ventilation scoops, stiffening ribs, printed wiring board mounting posts and miscellaneous sub-bracketry are formed from the principal part to provide a simple, low-cost assembly. Structural integrity is maintained by bridging the center with a spacer and the open ends at top and bottom with braces. The 3.6-kg ferro-resonant transformer and other large components of the

power supply are mounted on the stiff center section of the *U*-channel to resist deformation under shock loading. In manufacture, the power supply is wired as shown in Fig. 19 using 300 V test, flame-retardant wire.

The  $\pm 24$  V secondary voltages are brought to a central printed wiring board, CP-1, for distribution. CP-1 physically mounts on the open end of the *U*-channel and interconnects the right, central and left zones. Consequently, all terminations are performed on CP-1 for signal as well as power distribution. The board includes a 50-pin female connector for the display unit cord, a 21 screw connection field to accommodate the loop and speakerphone leads, two fuses, test points and screw lifters. On the left side of CP-1, the gold-plated connector land accommodate either the FW-1 circuit pack for key telephone installations or the FW-2 circuit pack for single-line installations. The right-side lands accommodate the 939-type cable equalizer or appropriate 877-type pad-out networks, depending on the installation. This approach to circuit interconnection utilizes the physical arrangement to lower cost and increase reliability by reducing connectors and harnesses. The assembly without cover is shown in Fig. 20. With any complement of boards, all terminals, fuses, screw lifters, adjustments and test points are available from the front.

The cover is made of 25 percent glass-filled polyester which meets the fire-resistant codes. All labels are molded in. The cover has venting to accommodate the heat dissipating circuitry. Under maximum power dissipation, the temperature rise is 25°C above room ambient.

## V. CONCLUSION

The 2C Video Telephone Station is now in production by Western Electric Company and is being used in initial *Picturephone* service. As manufacturing and field experience accumulates, normal engineering improvements will be made to reduce cost and improve reliability. Different versions of station equipment will be required in the future as *Picturephone* service grows, such as designs for residential applications, conference rooms, and public booths. In addition, future designs may incorporate slow-scan techniques for better graphics transmission, and eventually color *Picturephone* sets will become a practical possibility.

## VI. ACKNOWLEDGMENTS

The design of the *Picturephone* set has been a collaborative effort of many colleagues whose contributions are gratefully acknowledged.

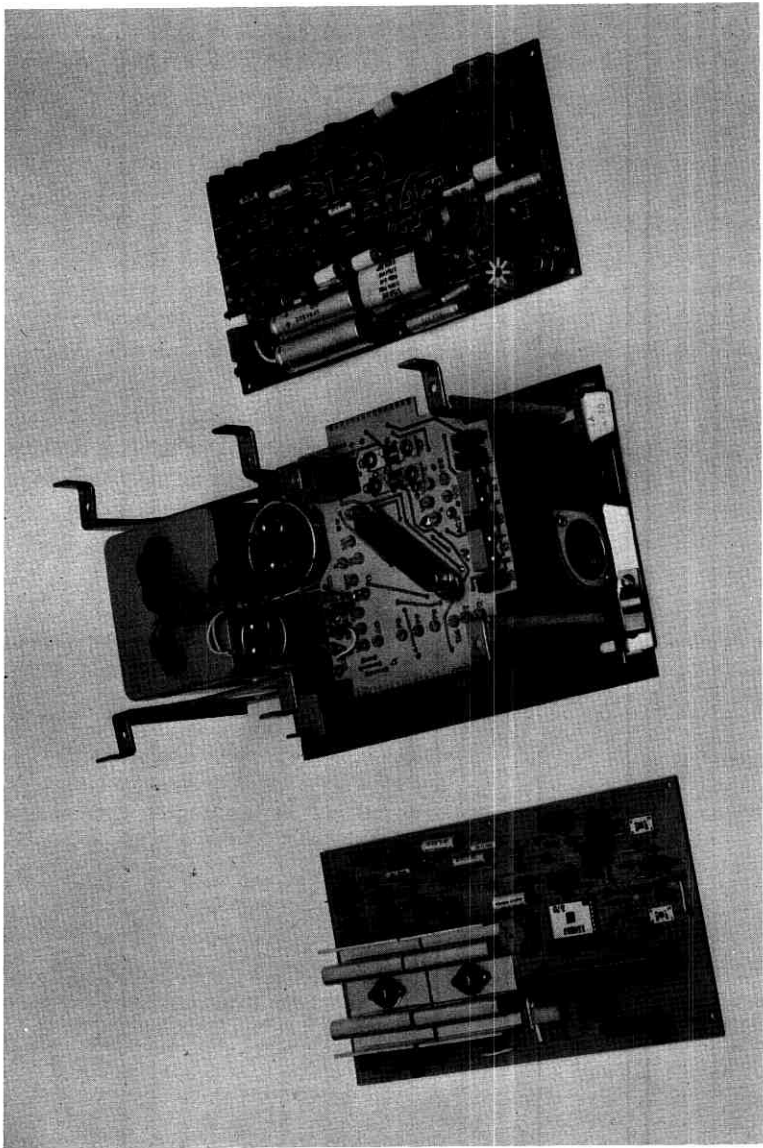


Fig. 19-1A Service Unit (subassemblies).

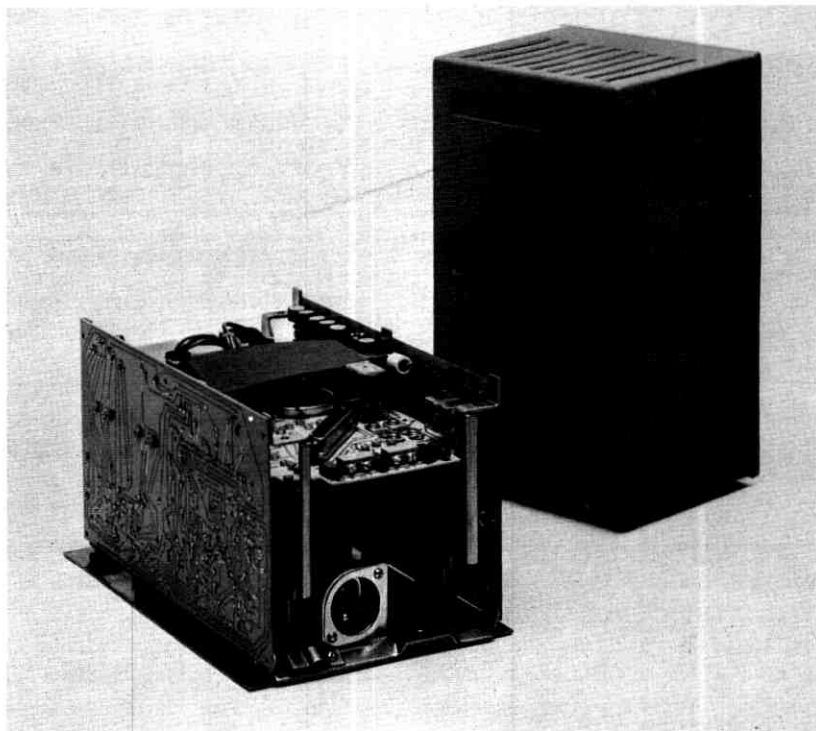


Fig. 20—1A Service Unit (less cover).

The completion of the design on schedule required and received outstanding cooperation from many technical areas and the shops and drafting organizations at Bell Laboratories. The enlightened leadership of C. G. Davis and F. F. Romanow and the technical and supervisory contributions of A. M. Gordon, S. C. Kitsopoulos, V. E. Munson, H. T. Shippen and W. Thommen are especially acknowledged.

#### REFERENCES

1. Dorros, I., "The Picturephone® System: The Network," B.S.T.J., this issue, pp. 221-233.
2. Crater, T. V., "The Picturephone® System: Service Standards," B.S.T.J., this issue, pp. 235-269.
3. Clemency, W. F., and Goodale, W. D. Jr., "Functional Design of a Voice-Switched Speakerphone," B.S.T.J., 40, No. 3 (May 1961), pp. 649-668.
4. Kaufman, J. E. (Ed.), *Illuminating Engineering Society Lighting Handbook*, 4th Edition, New York: Illuminating Engineering Society, 1966, pp. 9-55.
5. Dreyfuss, H., *The Measure of Man: Human Factors in Design*, New York: Whitney Library of Design, 1959, pp. H1-H2.

6. Kingslake, R., *Applied Optics & Optical Engineering, Vol. 2*, New York: Academic Press, 1965, p. 219.
7. Nagata, K., and Miura, A., "A Study of Picture Quality and Video Telephone," 1970 IEEE International Convention Digest, March 1970, pp. 250-251.
8. Gordon, A. M., and Singleton, J. B., "The Picturephone® System: Station Set Components," *B.S.T.J.*, this issue, pp. 313-349.
9. *Proceedings of the Camera Tube Symposium*, Bell Telephone Laboratories, Murray Hill, N. J., December 4, 1969.
10. Fink, D. G., *Television Engineering Handbook*, First Edition, New York: McGraw-Hill, 1957, pp. 1-50, pp. 6-58.
11. Millman, J., and Taub, H., *Pulse Digital and Switching Waveforms*, New York: McGraw-Hill, 1965, p. 515.
12. Von Ohlsen, L. H., "Soft X Ray Effects upon Silicon-Diode Arrays Aged in Camera Tubes," *IEEE Journal of Solid-State Circuits*, Vol. SC-5, No. 5, October 1970, pp. 261-265.
13. Thommen, W., "Noise Characterization of a Television Camera Amplifier," *Proceedings of the IEEE*, 54, No. 12 (December 1966), pp. 1953-1954.
14. Brown, H. E., "The Picturephone® System: Transmission Plan," *B.S.T.J.*, this issue, pp. 351-394.
15. Brown, J. M., "The Picturephone® System: Baseband Video Transmission on Loops and Short-Haul Trunks," *B.S.T.J.*, this issue, pp. 395-425.
16. Hamilton, J. O., and Wachs, M. H., "A Plastic Encapsulation for the Picturephone Display Tube," 1969 IEEE International Conference, Electrical Insulation Group (G-32) Special Session, New York, N. Y., March 28, 1969.
17. Kitsopoulos, S. C., Stokes, R. R., and Thommen, W., "An Automatic Electroresponsive Light Regulator Utilizing Taut-Band Meter Movement," U. S. Patent No. 3,561,847, filed June 5, 1968, issued February 9, 1971.
18. Stokes, R. R., "Apparatus Base Configuration for Reduction of Sliding and Tipping," U. S. Patent 3,462,111, filed December 13, 1967, issued August 19, 1969.
19. Urich, J. F., "The Picturephone® System: Switching Plan," *B.S.T.J.*, this issue, pp. 521-531.
20. Daskalakis, A., "The Picturephone® System: Key Telephone Systems," *B.S.T.J.*, this issue, pp. 567-584.



## The *Picturephone*<sup>®</sup> System:

# Station Set Components

By A. M. GORDON and J. B. SINGLETON

(Manuscript received October 27, 1970)

*The development of the Picturephone<sup>®</sup> station set required the special design of many new electronic components. These include new camera and display tubes for opto-electro conversion, 18 integrated circuits using both silicon and tantalum technologies, and several additional discrete devices. This article reviews the development of these custom designed electronic components.*

### I. INTRODUCTION

The *Picturephone* station set<sup>1</sup> is one of the most sophisticated items of station equipment ever provided a Bell System customer for switched two-way communications. In concept, the *Picturephone* station set resembles a commercial closed circuit transmitter and receiver. However, the operating conditions are markedly different and impose special requirements on many of the components. To provide the optical to electrical conversion in the transmitter and the inverse in the receiver, as well as the audio and control functions, requires over 900 electrical components in the station set. These range from simple passive discrete resistors to the silicon diode array camera tube which contains over 500,000 diodes in its approximately 0.2 square inch active target area.

Combinations of hybrid integrated circuits utilizing both silicon and tantalum technologies, silicon monolithic circuits, discrete solid-state devices, and discrete passive components are incorporated on the various circuit boards. Additional components consist of the camera and display tubes and two active non-solid-state devices which are used in the high-voltage power supply. The active components used in the display, control, and service units are shown in Fig. 1.

To develop the complex station set circuitry in an orderly fashion, it was necessary to decide early in the design which portions of the circuitry were to be realized in integrated form. Some circuits re-

quired the precision tuning available with tantalum thin-film technology and were designed using hybrid integrated circuits. Other circuits were uneconomical in size and/or cost to be built in discrete form and their designs were realized in monolithic silicon form. In still other circuits it was found that though they could be realized in discrete form, it would be advantageous to utilize the matched characteristics available on silicon chips. Thus integrated circuit designs were used where it made best technical or economic sense to use them. Out of this design effort, undertaken jointly by the Systems Development and Device groups, 33 silicon integrated circuits (SIC) of 18 different designs evolved for *Picturephone* station set use. The most widely used SIC is the building block circuit which is used in seven different applications.

A simplified block diagram of the *Picturephone* station set is shown in Figs. 2, 3, and 4. Those circuits which utilize hybrid and monolithic integrated circuits are marked accordingly.

Several discrete components were custom designed to operate under the special conditions found in the *Picturephone* station set. Included

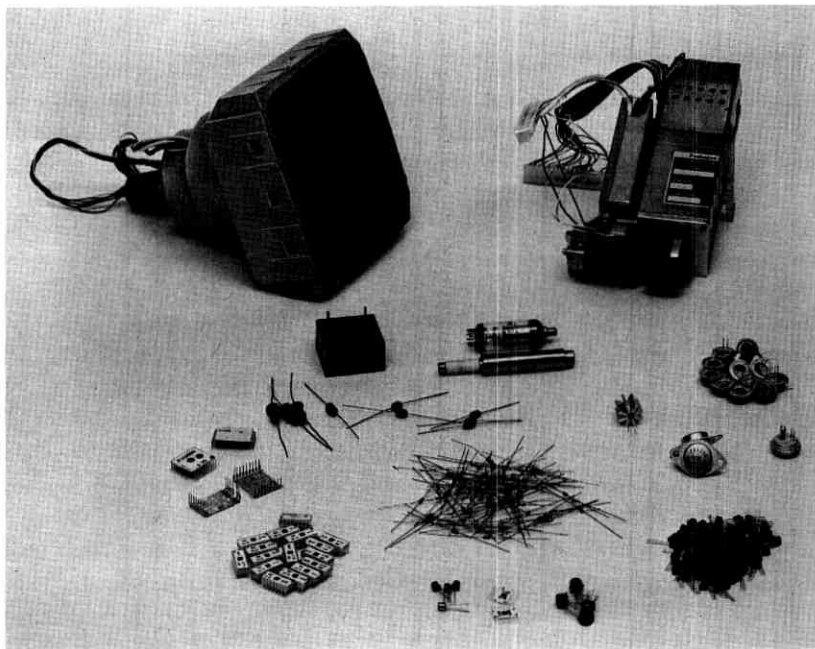


Fig. 1—Station set active components.

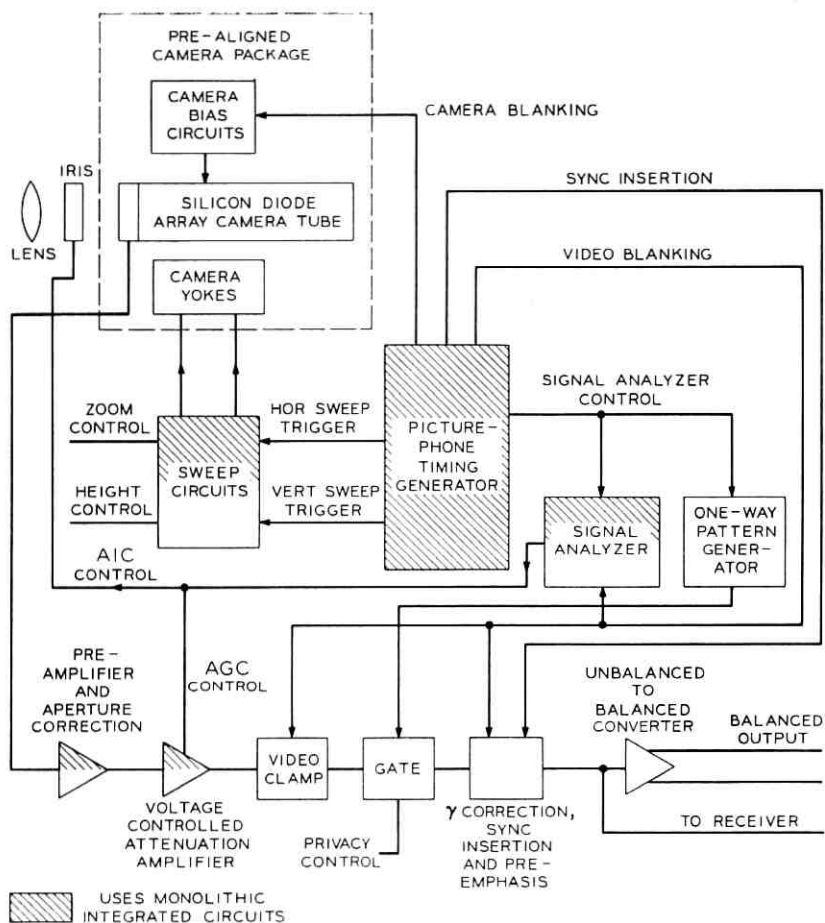


Fig. 2—Transmitter block diagram.

in this category are bipolar transistors, junction field effect transistors, a packaged display tube and a completely interchangeable packaged camera tube. This article will highlight those devices which were custom designed and existing devices which required special characterization for use in the station set.

## 11. CAMERA TUBE

Several unique properties are required of a camera tube for use in *Picturephone* service.<sup>2-4</sup> Life and reliability must be consistent with

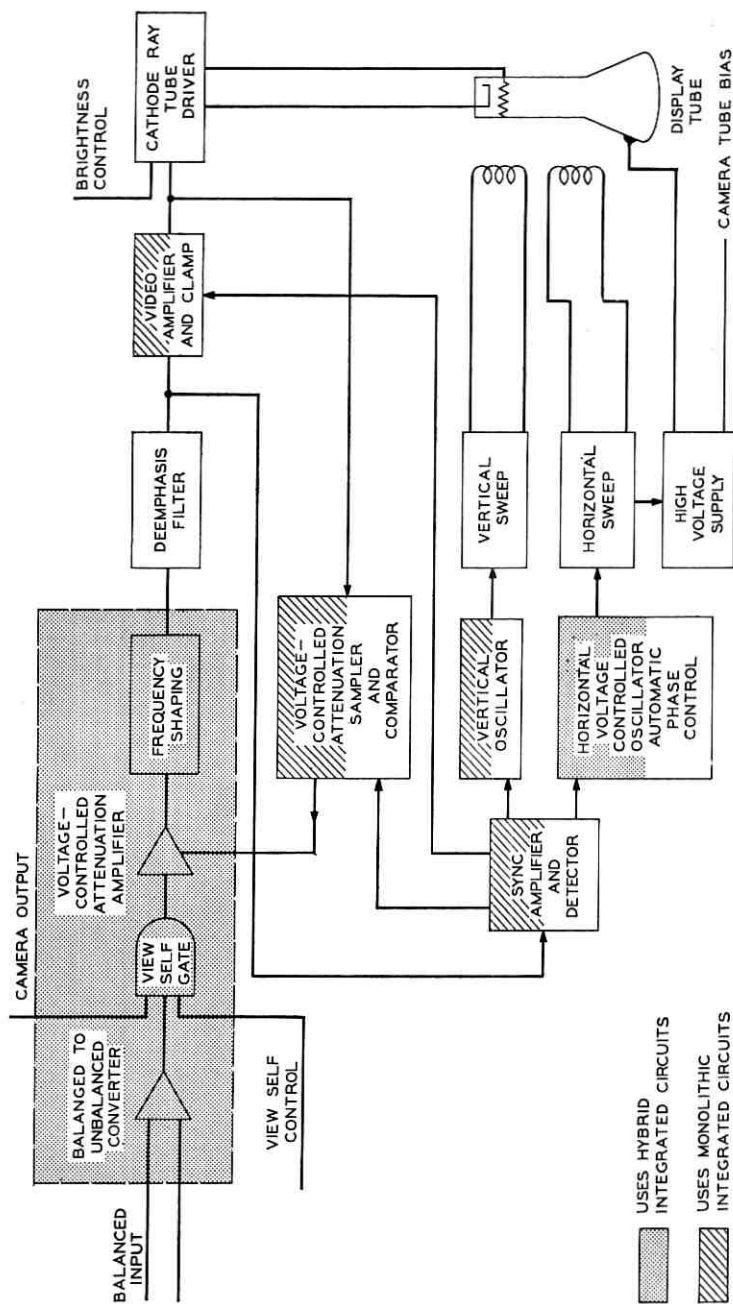


Fig. 3—Receiver block diagram.

Bell System standards. The camera tube must operate in environments where there are bright lights and high temperatures. It must withstand mechanical shock and have the ability to react rapidly when called upon. In meeting these objectives, the required power must be kept to a minimum and the manufacturing cost must be sufficiently low to justify use in *Picturephone* service.

These requirements are fulfilled in the design of the *Picturephone* camera tube. Unlike previous camera tubes which use an amorphous substance as the photoconducting element, this design uses an array of diodes ( $\sim 525,000$  in the scanned area) on a single crystal slice of silicon. This approach results in a target shown in Fig. 5 having the following advantages compared to an amorphous target:

- (i) It is not damaged by any reasonable light source.
- (ii) Sensitivity is improved by a factor of at least 3 to 1.
- (iii) It allows the scan size to be electronically changed without adverse effects.
- (iv) It possesses a long life.

Figure 6 shows the camera-tube structure which has been designed to hold the silicon target so that it will withstand mechanical shock in excess of 50 G and will be maintained at the temperature of the station set front panel. The electron gun and beam focusing electrodes

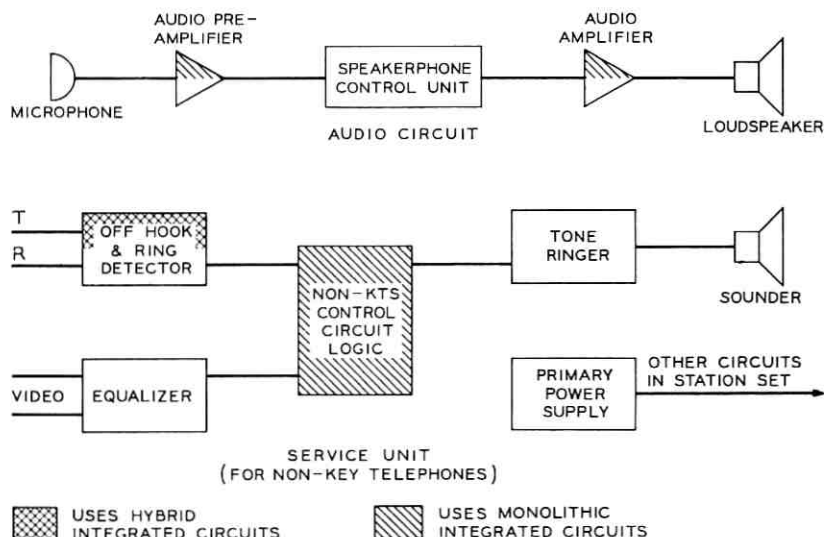


Fig. 4—Block diagram of audio circuit and service unit.

are assembled by brazing the metallic parts to a high alumina ceramic. Halide-free hard glass is used for the envelope and seals are made directly between glass and metal. This technique contrasts with that used in most similar devices, in which a soft metal seal (such as indium) forms an intermediate layer between glass and metal. An advantage of the direct seal between glass and metal is that high processing temperatures can be used which enhance long cathode life.

The cathode has been designed to match the long life expected of the silicon target. A thin layer of nickel is used to coat the electron emissive cathode particles so that the resulting cathode can be operated at a lower temperature for a given current. Use of this coating technique along with the proper nickel base alloy provides a cathode with greater than five-year life expectancy.

Manufacturing techniques and facilities have been designed to enhance reliability. Targets are processed and tubes assembled within laminar flow work benches which are installed within a room fed with filtered air. Assembly operators are garbed in a manner to minimize contamination of the work area. Final tube exhaust is carried out on an oil-free high vacuum station, with ultimate pressure at the time of tube pinch off of about  $2 \times 10^{-8}$  torr.

The physical design philosophy used in the station set has been to provide all units in modular form. Manufacture of the camera tube follows this concept. The required circuits are mounted within the camera package at the tube manufacturing plant. They include deflection coils, magnetic shielding, preadjustments for deflection currents,

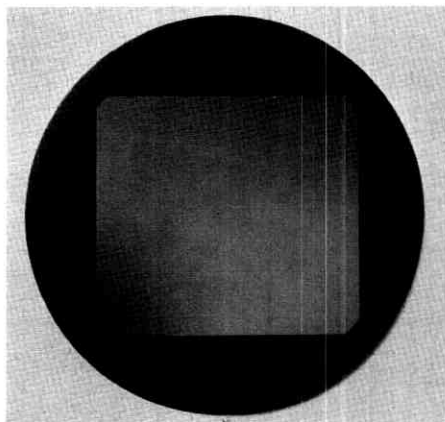


Fig. 5—Camera tube target.

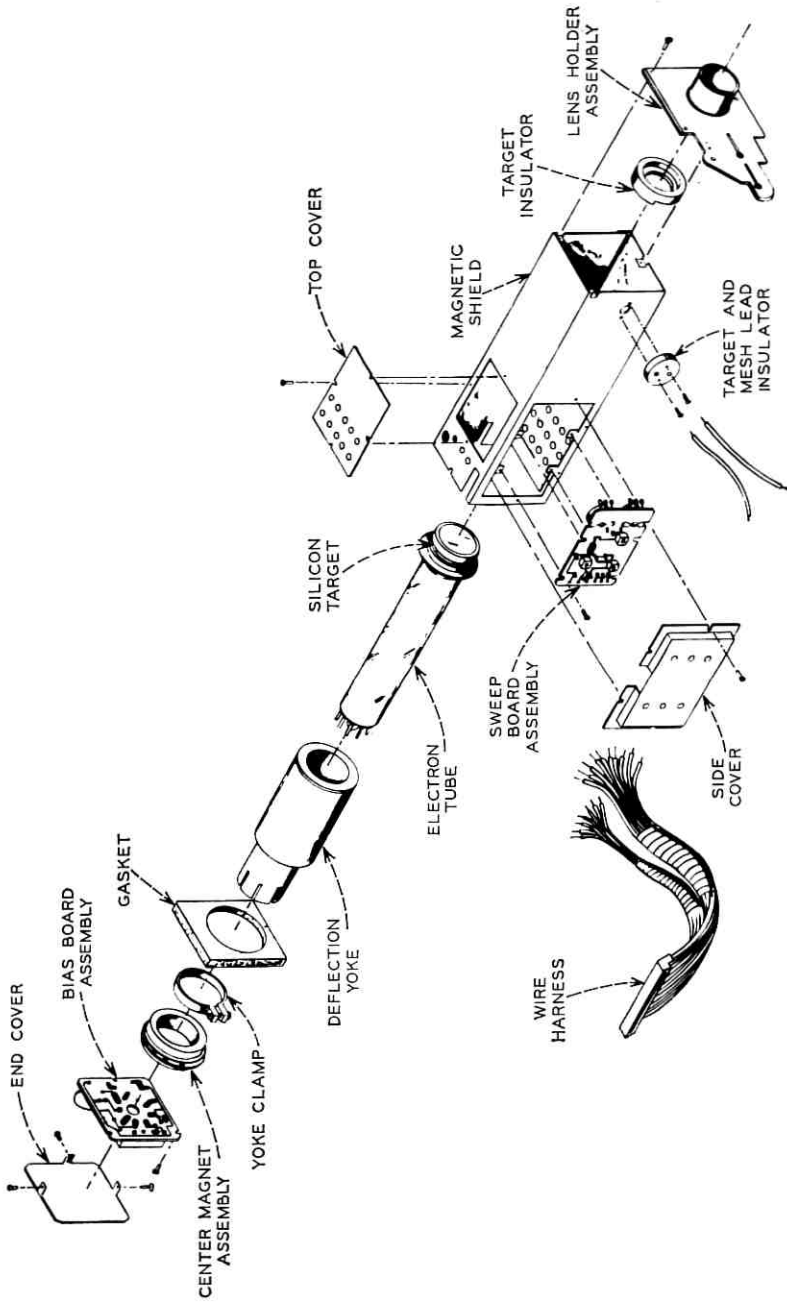


Fig. 6—Close-up view of the camera tube.

raster centering and all tube electrode voltages as shown in Fig. 7. Adjustment of the camera tube will not be necessary when it is mounted in the station set.

The camera tube operates at a peak signal current of 225 nA and has a maximum signal output in excess of 300 nA. At room temperature the dark signal current is about 10 nA. For incandescent light the average sensitivity is 175 nA per 0.1 footcandle. Operation of the set begins almost instantly when a call is initiated or answered; this performance is attained by maintaining the tube at 90 percent of its operating heater voltage during standby periods.

### III. DISPLAY TUBE

Requirements on the display tube are also somewhat more stringent than would be found in normal commercial applications. In addition to providing long, unattended life, and shock resistance, the

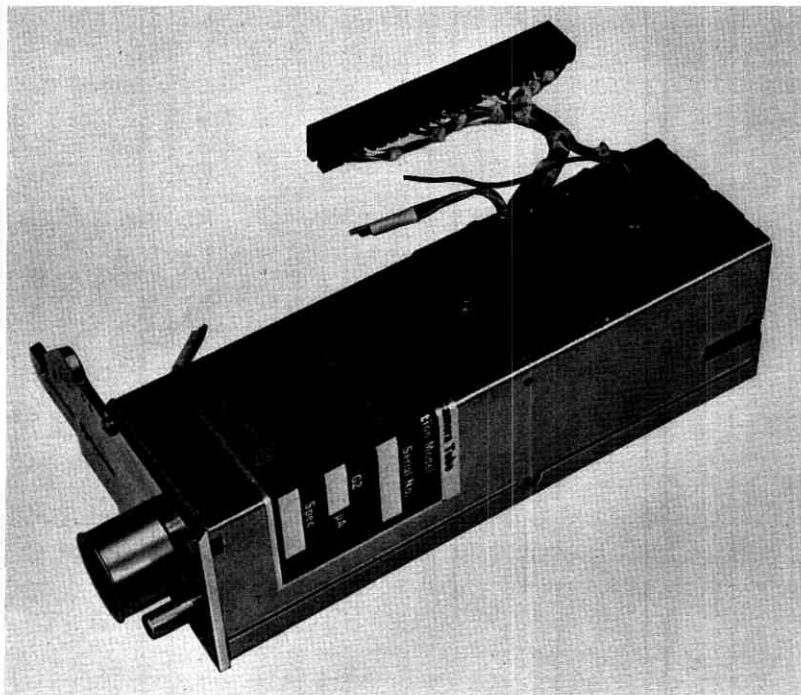


Fig. 7—Assembled camera tube.



potential hazard of the envelope imploding must be considered for personnel protection. Implosion protection is provided by applying a laminated safety panel over the exposed face of the tube. Additional implosion protection is provided for the funnel portion of the tube by completely encapsulating it in a polyurethane jacket. Since the cured polyurethane material adheres to the glass envelope, it prevents collapse of the funnel upon fracture of the display tube.

The display tube is manufactured with a non-browning glass envelope. This insures that long-term bombardment of the glass by the high-voltage electron beam will not result in a glass color change which might alter the spectral quality of the emitted light. Conventional settling techniques are used to deposit a white phosphor. To increase radiated light and to protect the phosphor layer from ion damage, the bulb is then aluminized.

Assembly of the electron gun consisting of stainless steel parts is performed on laminar flow benches. Cathode techniques, as described for the camera tube, have been used to maximize life capability. To further extend operating life the average cathode current is held to a minimum.

In keeping with the modular concept, the *Picturephone* display tube package shown in Fig. 8 contains the deflection yoke and centering

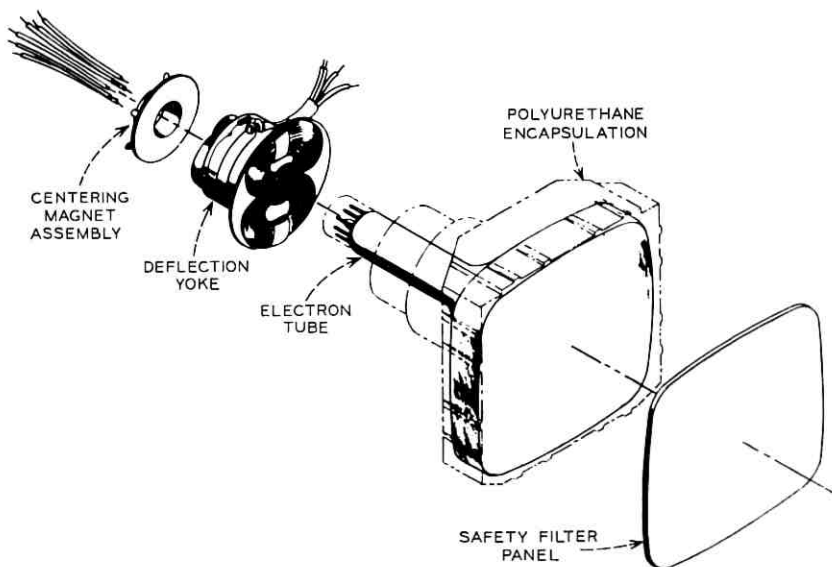


Fig. 8—Close-up view of display tube.

components. The entire package is assembled at the tube manufacturing plant. To improve picture contrast in very bright ambient light, the laminated face plate which is applied over the front of the tube for implosion protection incorporates a 30 percent light transmission filter. With this face plate, less than 9 percent of the incident room light is combined with the video display, yet the highlight brightness can exceed 100 foot lamberts. Finally, the entire assembly shown in Fig. 9 is encapsulated in the previously mentioned polyurethane jacket. This technique, in addition to providing protection from flying glass in the event of an implosion of the bulb, also establishes a firm and reproducible mechanical outline for mounting within the station set. The finished unit provides a package of considerable physical strength and replacement of a display tube requires only adjustment of raster size and centering.

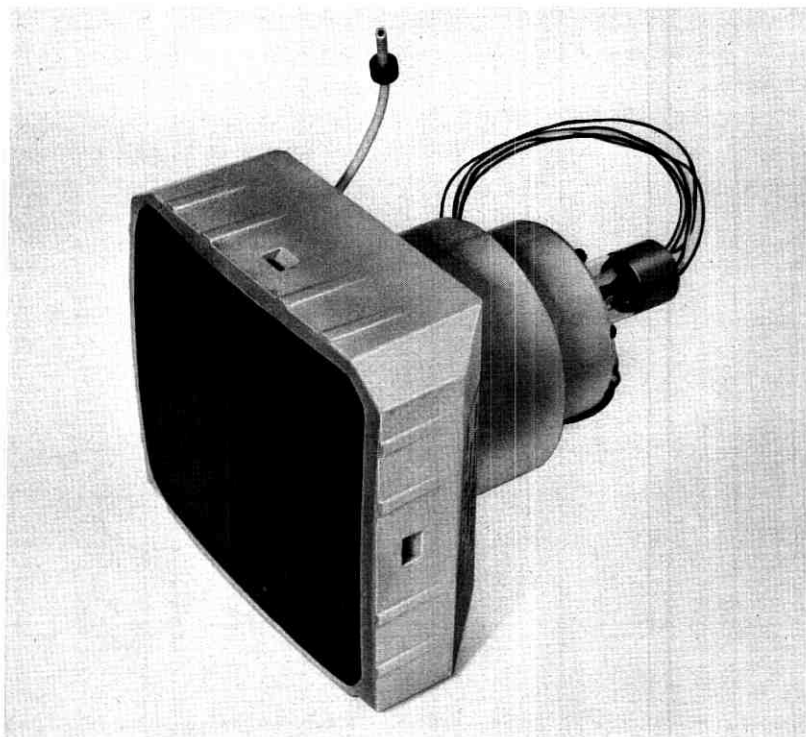


Fig. 9—Assembled display tube.

IV. *Picturephone* TIMING GENERATOR

Locked 2 : 1 interlace and all the timing functions required for transmitter operation are provided by the *Picturephone* timing generator. This circuit consists of a crystal controlled oscillator and five resistor-transistor logic (RTL) circuits mounted on a one-square-inch ceramic interconnection board as shown in Fig. 10. In addition to the 376 transistors, 2 diodes, and 535 resistors which are contained in the six beam-leaded chips, the circuit uses an externally mounted crystal to maintain a stability of better than  $\pm 100$  ppm for all timing intervals. This precision allows the pull-in range of the receiver automatic phase control (APC) circuit to be reduced and permits a substantial reduction in the capacity and cost of the "elastic store" used in the digital coding system. A 12-cell store is used in the present coding system<sup>5</sup> as compared to the 30-cell store used in the previous design.

The timing functions provided include the following:

(i) Horizontal and vertical sweep triggers. These are used for initiating retrace of the beam in the camera.

(ii) Camera blanking. This provides cut off of the camera beam during retrace.

(iii) Video blanking. This permits disabling of the linear signal-processing chain during the nonvideo intervals.

(iv) Sync insertion. This is the composite sync signal that will be mixed with the video information for transmission.

(v) Signal analyzer control. This output is used to examine the center half of the picture for light intensity of the scene for setting the automatic gain and iris control (AGIC) of the camera. A distinct pattern which is transmitted when a subscriber goes into the "Privacy" mode is also derived from this signal.

Figure 11 shows the relationship between these various timing waveforms (except the signal analyzer control) and the transmitted composite video. The timing generator derives these timing functions from the crystal controlled oscillator operating at 512 kHz and digital countdown circuitry. 8-kHz horizontal and 59.925-Hz vertical signals are generated by judicious interconnection of the output signals from the various stages of the countdown circuitry.

A block diagram of the timing generator is shown in Fig. 12. The two-stage counter and the first four-stage counter divide the 512-kHz oscillator frequency down to the 8-kHz horizontal line rate. A 16-kHz signal from the first four-stage counter is used to drive the second

four-stage counter. By means of recycling techniques, the second and third four-stage counters divide down to the 59.925-Hz field rate. Various timing intervals available from the intermediate stages of the countdown circuitry are used in the mixing circuits to derive the timing wave forms shown in Fig. 11. The precision of all these signals is derived from the inherent stability of the 512-kHz oscillator.

Low power (1–2 mW/gate), high packing density (85 mils<sup>2</sup>/gate) and high immunity to power supply variations (2–3 volts) and temperature variations (0–65°C) were the design considerations for the logic circuits. The oscillator circuit was designed around minimum crystal requirements and provides a square wave output with low sensitivity to power supply variations (4–10 volts) and temperature variations (0–65°C). No external capacitors, inductors or manufacturing adjustments are required for oscillator operation.

The total circuit has been realized on six beam-leaded silicon nitride passivated junction isolated chips which are:

- (i) Crystal controlled oscillator and output buffer circuit
- (ii) Two-stage counter and mixing circuit.

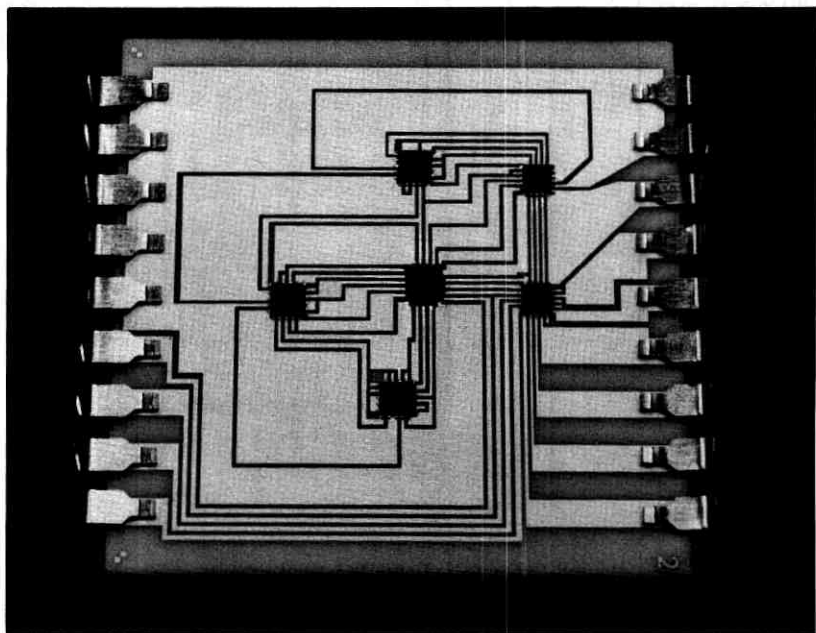


Fig. 10—Picturephone timing generator board.

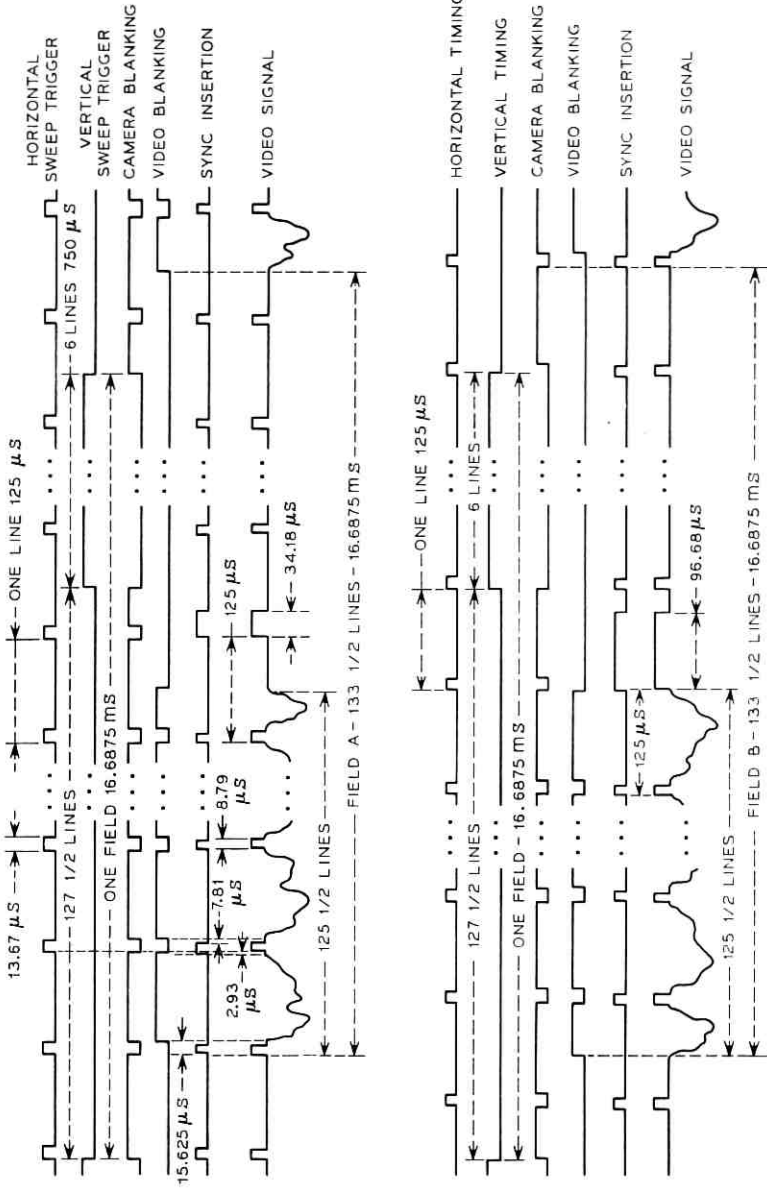


Fig. 11—Timing signals. (The sixth output, signal analyzer control, is "high" during approximately the middle half of the active scan time. One frame: 267 lines—33.375 ms, 251 lines of "picture" information transmitted.)

(iii, iv, v) Four-stage counter circuits.

(vi) Recycling and output circuit.

#### 4.1 Crystal Controlled Oscillator and Output Buffer Circuit

The crystal oscillator circuit provides the fundamental 512-kHz frequency for the timing generator. Its output consists of a symmetrical square wave suitable for driving the integrated logic circuits which count down to the horizontal line and vertical field rates. The oscillator frequency typically deviates from the nominal less than 40 ppm for all causes, including manufacturing tolerances of the amplifier circuit and crystal, a 40°C temperature range, and power supply voltage variations of  $\pm 40$  percent.

In addition to the oscillator circuit, buffer transistors for five of the logic outputs are incorporated on the same integrated circuit chip. A total of 13 transistors, 2 diodes, and 24 resistors are contained on the  $54 \times 54$  mil chip which constitutes this silicon integrated circuit. The total resistance is 43 k $\Omega$  and the power dissipation is about 70 mW.

The crystal unit consists of a rectangular DT cut quartz plate with dimensions of 0.350 by 0.140 inches and a thickness of 9 mils. It resonates in a width shear mode which provides a better temperature coefficient than that obtained with square, face shear resonators. Two header wires are attached to the nodal points at the center of the major surfaces of the plate and brought down to a TO-5 header base. Encapsulation of the crystal is achieved by cold welding a 0.6 inch high metal can to the header.

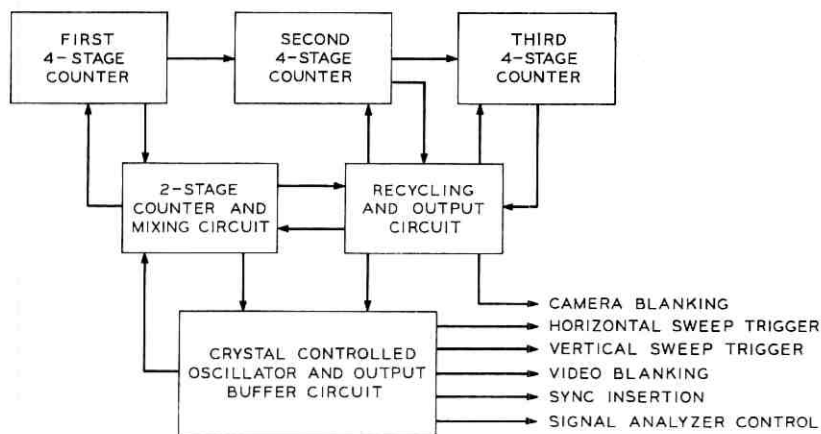


Fig. 12—Block diagram of the *Picturephone* timing generator.

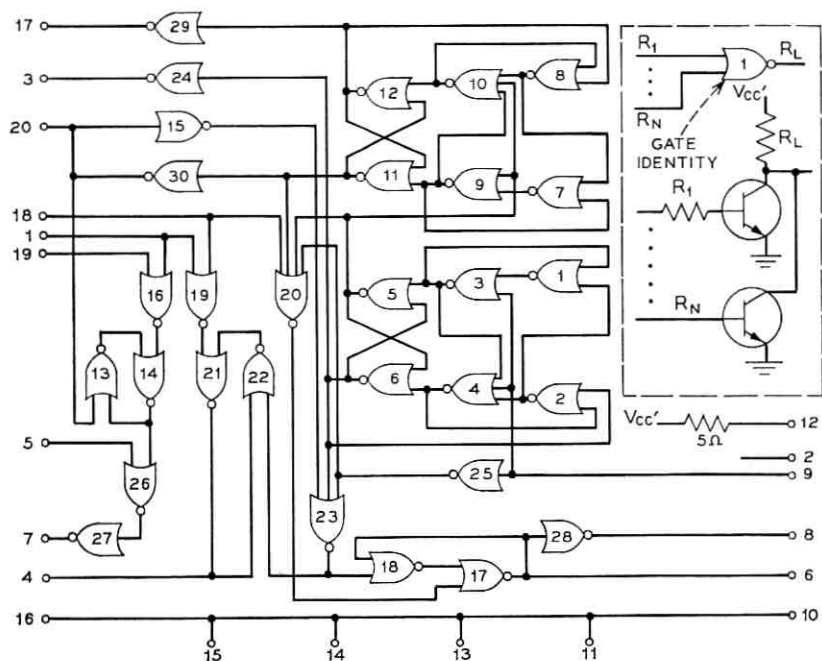


Fig. 13—Logic diagram of two-stage counter and mixing circuit.

#### 4.2 Two-Stage Counter and Mixing Circuit

RTL is used in this circuit to derive several logic functions including dividing the input 512-kHz square wave signal by 2 and 4. The horizontal sweep trigger and video blanking signals are also supplied by this circuit.

As shown in Fig. 13, this circuit contains 30 logic gates composed of 58 transistors and 80 resistors on a  $54 \times 54$  mil chip. Nominal power dissipation at 2.5 V is 50 mW. Each gate was individually designed for fanout to utilize minimum resistor values (therefore low area) for a given power consumption.

#### 4.3 Four-Stage Counter

Three four-stage counter circuits are used in the timing generator. Each chip has 38 logic gates containing 79 transistors and 111 resistors. RTL is used to divide the input square wave signal by 2, 4, 8 and 16. Provision is made for stuffing pulses to control recycling and for gating various state outputs. Nominal power dissipation for the  $62 \times$

62 mil chip is 60 mW at 2.5 V. A photograph and logic diagram of the circuit are shown in Figs. 14 and 15.

#### 4.4 Recycling and Output Circuit

Several functions including camera blanking, signal analyzer control, vertical sweep trigger and the sync insertion signal are derived in this logic circuit. The camera blanking output drives external circuitry directly whereas the other signals drive the output buffer circuits. As shown in Fig. 16, this RTL circuit contains 36 gates composed of 68 transistors and 98 resistors on a  $62 \times 62$  mil chip. Nominal power dissipation at 2.5 V is 60 mW.

### V. RECEIVER VIDEO PROCESSOR

In the block diagram of the *Picturephone* station set receiver (Fig. 3), the receiver video processor circuit is enclosed by a dashed line.

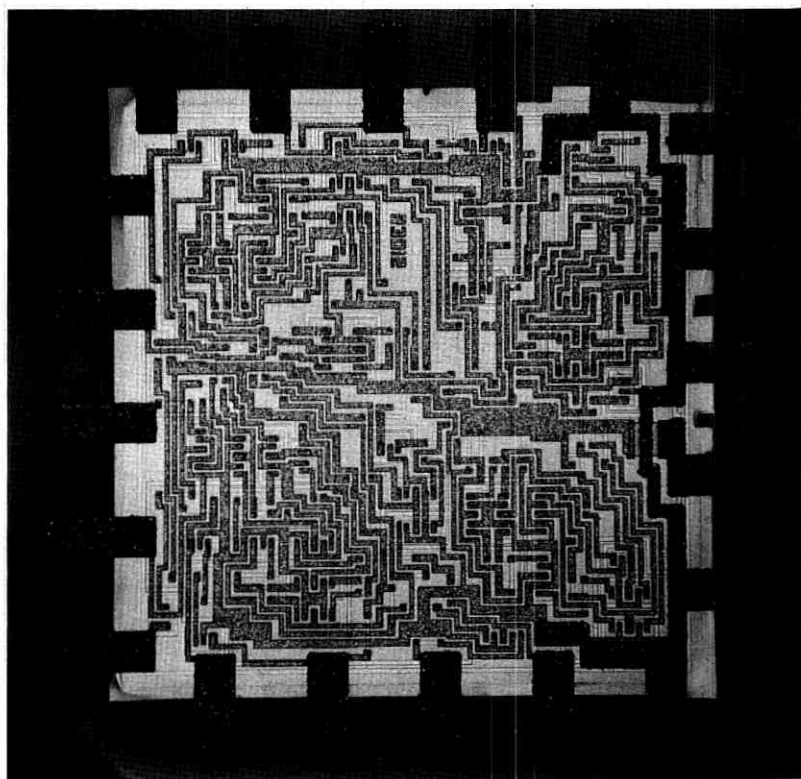


Fig. 14—Four-stage counter.



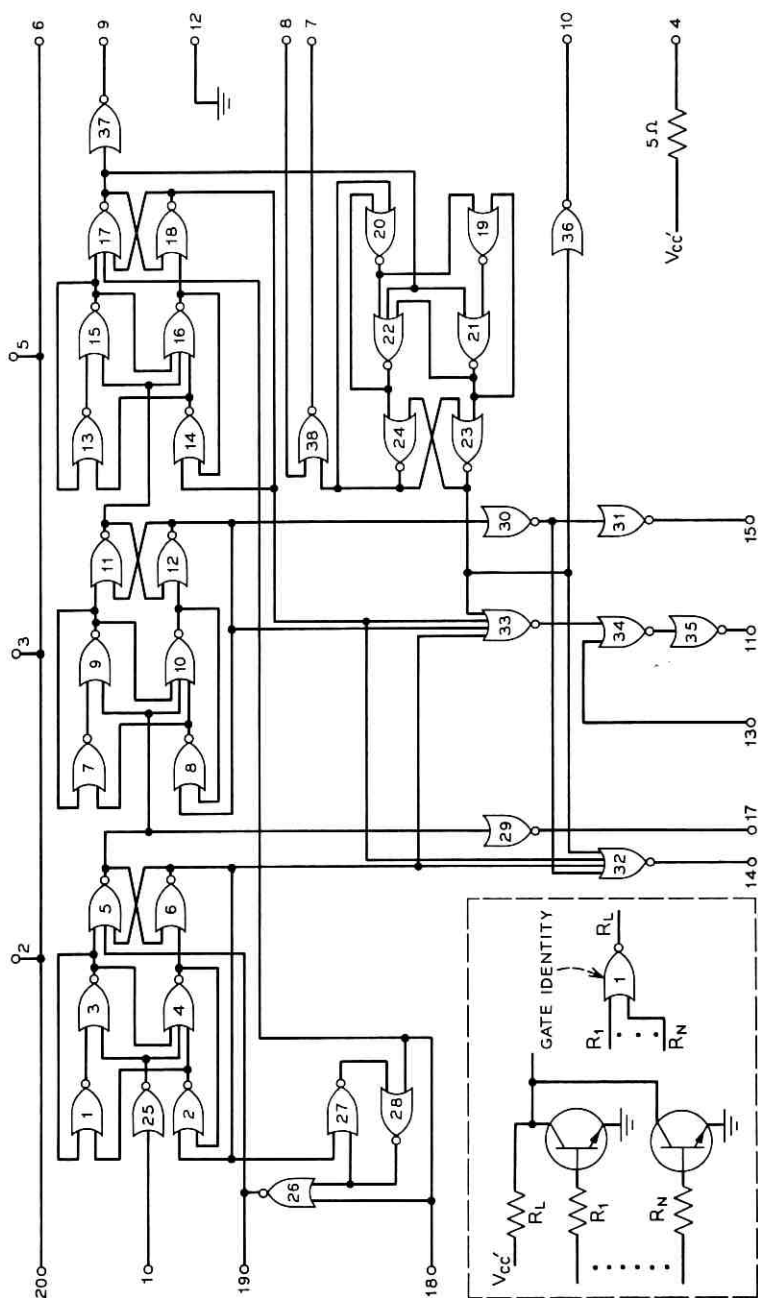


Fig. 15—Logic diagram of four-stage counter.

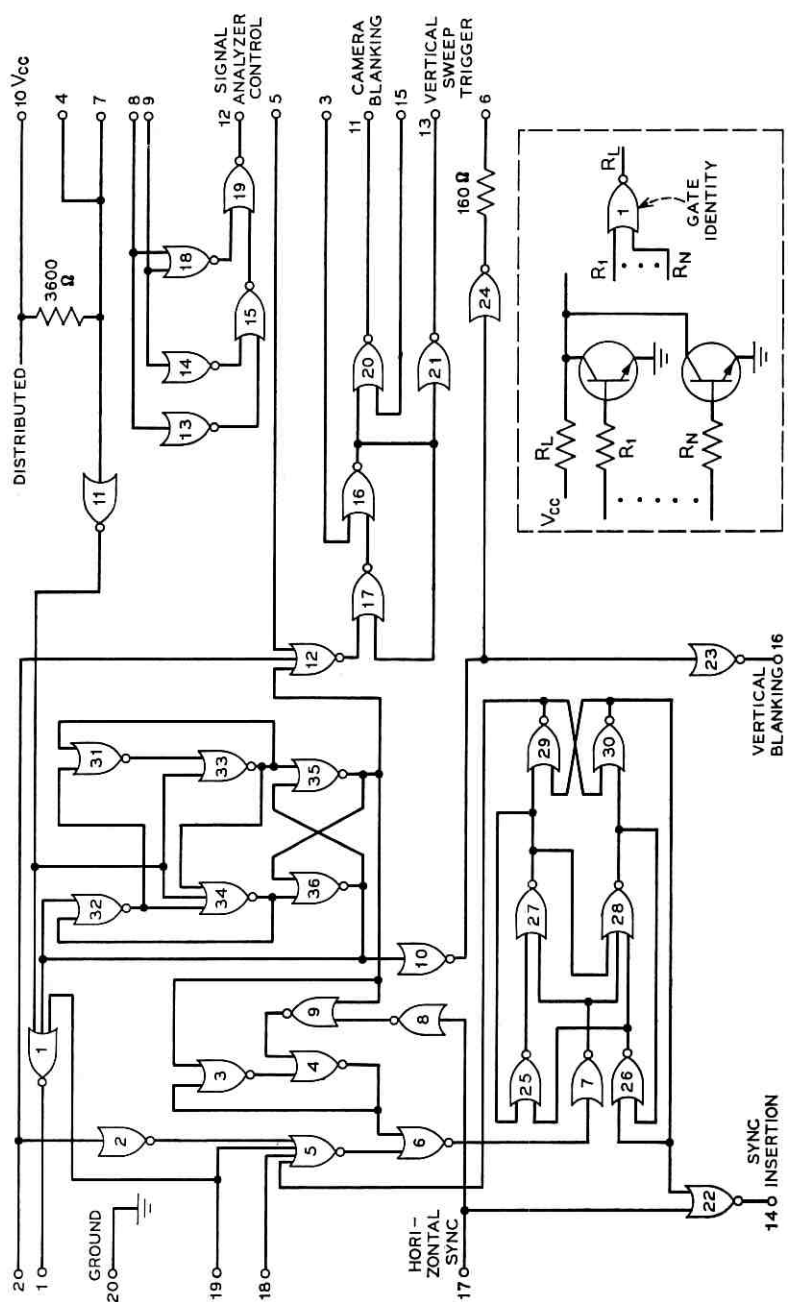


Fig. 16—Logic diagram of recycling and output circuit.

This hybrid integrated circuit (HIC) performs these functions in the receiver of the *Picturephone* station set: (i) balanced-to-unbalanced conversion (balun), (ii) video switching, (iii) AGC (voltage-controlled attenuation), and (iv) frequency shaping (bandwidth control and signal crispening).

A photograph of this HIC is shown in Fig. 17. Two beam-leaded silicon integrated circuits, mounted on a ceramic board which includes tantalum thin-film capacitors and resistors on one substrate constitute the total circuit. The balun, video switch, and variable gain stage are integrated on one monolithic silicon chip. Eleven transistors and 11 biasing resistors are incorporated in the second chip as part of the frequency shaping filter. The resistors and capacitors which control the frequency response are on the thin-film substrate.

At the subscriber's premises the analog video signal is received on a balanced twisted pair transmission line.<sup>6</sup> *Picturephone* signals are transmitted in a balanced manner to increase immunity from interference signals. Conversion to an unbalanced signal is performed in the first stage of the video processor by a differential amplifier.

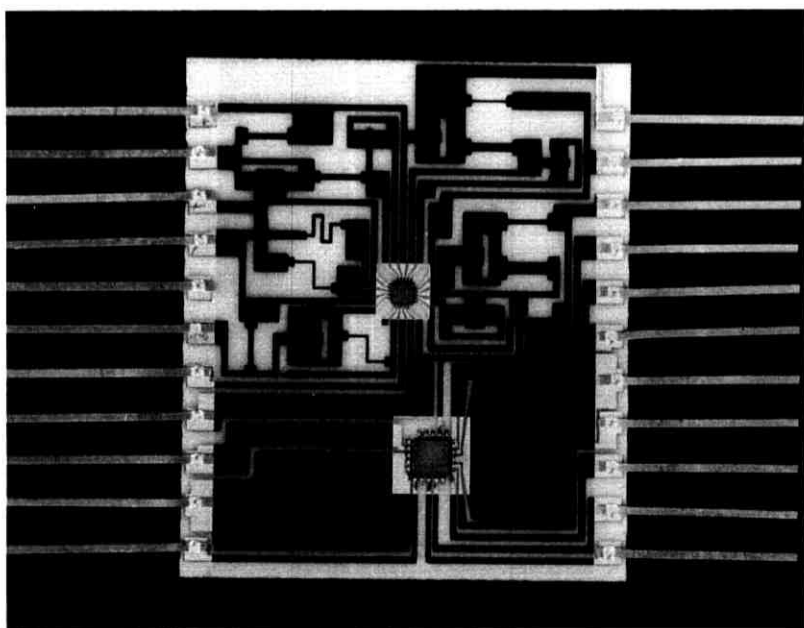


Fig. 17—Video processor board.

The video switch allows the user to view either the outgoing or incoming picture. He selects the mode of operation with his control unit.

Variable attenuation is provided in the last stage of the first SIC. An attenuation range of more than 12 dB is achieved by utilizing the logarithmic ratio of current to voltage in transistors. Total harmonic distortion is less than five percent over the attenuation range for a two-volt peak-to-peak input signal. The three functions have been designed into a  $64 \times 64$  mil beam-leaded junction isolated silicon chip. It contains 24 transistors, 9 diodes and 54 resistors totaling 147 k $\Omega$  and dissipates 160 mW.

An active roll-off filter is used to limit bandwidth and reduce transmission noise. This consists of a six-pole Thomson filter which approximates the ideal linear phase characteristic of a gaussian filter.<sup>7</sup> The crispening circuit accentuates contrast transitions in the displayed picture, which give the effect of an extended bandwidth. Emitter-followers and biasing resistors which isolate the filter stages are on the second SIC. The resistors and capacitors which control the response are of tantalum composition on a ceramic substrate. Final values for the tantalum-film resistors are determined by running a successive-approximation computer program for each production circuit. In effect, each production filter is resynthesized before the chips are bonded down to insure that the particular silicon-tantalum combination is capable of meeting performance requirements. The resistors are then trimmed to the values determined by the computer and finally the silicon chips are bonded to the substrate and the completed assembly is measured. The frequency response characteristic of the filter is shown in Fig. 18. The maximum allowable error is  $\pm 0.5$  dB over the frequency range of 0.1 to 1.1 MHz.

In addition to the transistors and biasing resistors for the filter, the second SIC contains metallization paths which interconnect some of the beams to provide crossovers for the external circuit.

## VI. VOLTAGE CONTROLLED OSCILLATOR

A voltage controlled oscillator (VCO) which is synchronized with the incoming horizontal signal provides the drive for the receiver horizontal sweep and high voltage supply. An automatic phase control (APC) circuit is employed to reduce jitter and provide immunity from noise pulses. The circuit is in hybrid integrated form on a 0.6 square inch ceramic substrate containing tantalum resistors, 2 silicon chips

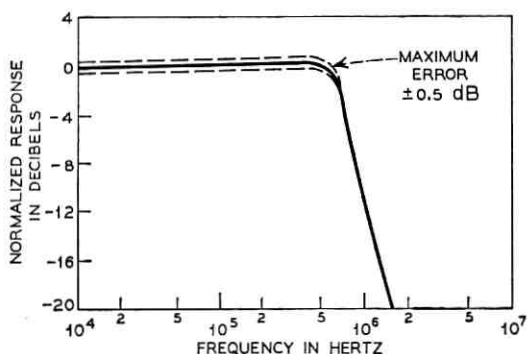


Fig. 18—Filter response.

and a tantalum RC network fabricated on glass for adjusting the oscillator timing intervals. A photograph of this circuit is shown in Fig. 19.

Figure 20 shows the VCO circuit diagram including the integrated and discrete components. Basically, the VCO consists of an oscillator under the control of an APC circuit. This oscillator is composed of one SIC and a tantalum thin-film resistor-capacitor network and functions as a temperature compensated astable multivibrator. A  $42 \times$

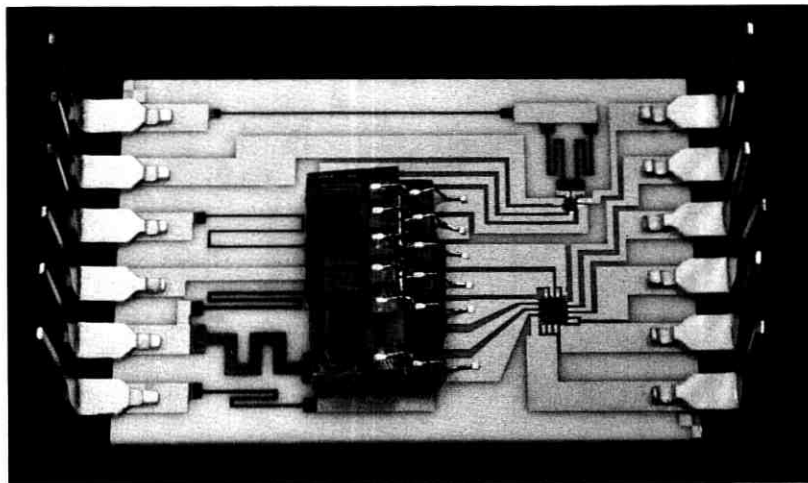


Fig. 19—Voltage controlled oscillator.

42 mil beam-leaded junction isolated chip containing 8 transistors, 5 diodes, and 10 resistors totaling 39 k $\Omega$  constitute this SIC. Nominal power dissipation is 70 mW.

A sawtooth type phase comparator is used in the APC circuit to derive a control voltage from the phase difference between the incoming horizontal sync signal and the horizontal signal derived in the receiver oscillator. The control voltage developed in the phase lag filter is near ground potential and can be used directly to control the oscillator but with the penalty of affecting the time constants of the oscillator. By introducing the level shifter shown in Fig. 20, in which the resistive elements are in tantalum thin-film form, the effects of interconnecting the discrete and integrated circuits are buffered out. Essentially unity gain for the control voltage and a dc shift of approximately +5 volts is achieved with the level shifter. A matched pair of transistors is used for the two active elements of the level shifter to take advantage of their matched  $V_{BE}$ 's and temperature characteristics.

In manufacture, the oscillator RC network and the resistors for the level shifter are trim anodized to value, except for  $R_{1G}$  and  $R_{4G}$ . These two elements are adjusted after the two silicon chips and the glass sandwich containing the tantalum components for the oscillator are mounted on the ceramic interconnection board which also contains

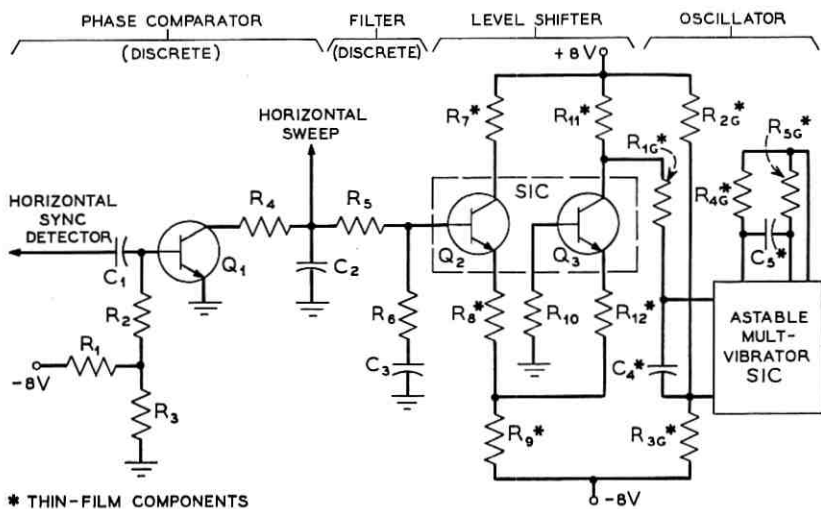


Fig. 20—Voltage controlled oscillator.

the resistors for the level shifter.  $R_{4G}$  is trim anodized to provide a 19.5- $\mu$ s ( $\pm 1$  percent) timing interval and  $R_{1G}$  is trim anodized to provide an 8050-Hz ( $\pm 0.5$  percent) repetition rate. Both inputs of the level shifter are terminated in separate 23.7-k $\Omega$  resistors to ground while these adjustments are made. In normal operation, the control voltage will experience very small deviations. The sensitivity of the oscillator is such that a change in control voltage of  $\pm 0.1$  volt results in a change in frequency of approximately  $\pm 80$  Hz.

#### VII. OFF-HOOK AND RING DETECTOR

This circuit, shown in Fig. 21, is bridged across the balanced telephone line to detect an off-hook condition and to detect ringing. Whenever a video call is received, the output from the ring detector is combined logically with other signals to activate a ringing signal generator. The circuit is included in the service unit for station sets used with non-key telephones; for key telephones this function is performed in the common key equipment. Thin-film technology is used because the comparator circuit requires careful balancing to detect the transverse signals and provide good common mode rejection. A well balanced amplifier is achieved by trim anodizing tantalum nitride thin-film resistors to  $\pm 1$  percent in combination with a matched pair of transistors.

The resistor values required range from 200  $\Omega$  to 121 k $\Omega$  a ratio of 1 : 605. Design considerations for the low value resistors differ from those of the high value resistors. However, by appropriate selection of line widths and sheet resistivity, it is possible to fabricate all resistors from a single uniform tantalum nitride film. Because two 121-k $\Omega$  resistors are required, a final sheet resistance of 100  $\Omega$  per square and 5 mil wide paths were chosen to keep the required surface area as small as possible. Thirty mil wide paths were chosen for the 200- $\Omega$  resistors to provide an effective area for anodization. Once the final sheet resistance had been chosen as 100  $\Omega$  per square, the initial sheet resistance required for the sputtered tantalum nitride material was calculated to be approximately 50  $\Omega$  per square. This allowed for a large enough anodizing voltage to provide the benefit of oxide protection and an adequate trimming range. Individual resistors for all 15 circuit modules on a substrate are trim anodized to  $\pm 1$  percent in the absence of the SICs. The 15 circuits are then separated, the integrated circuits are appliquéd, and the completed modules tested. Figure 22 shows a 0.8-square-inch completed circuit with lead frames attached.

## VIII. OPERATIONAL AMPLIFIERS

## 8.1 High Output Voltage Swing Operational Amplifiers

Three operational amplifiers were designed to meet the frequency response and output voltage swing requirements for eight applications in the *Picturephone* station set. All three codes have a beam-leaded silicon chip of the same basic design bonded to a 16 lead dual-in-line ceramic substrate. Three different test specifications are utilized to reflect the various requirements for the eight applications.

The basic circuit is designed with optional Darlington inputs and operates from  $\pm 8.0$ -volt power supplies. In one operational amplifier, the Darlington inputs are externally connected to provide high input impedance ( $> 1 \text{ M}\Omega$ ), low bias current, and low offset current ( $< 20 \text{ nAde}$ ). These characteristics are needed for applications in both the camera horizontal and vertical sweep circuits. In addition, this operational amplifier provides a minimum output drive current of 10 mA into a 600- $\Omega$  load for the vertical sweep circuit.

Another version of the operational amplifier is used in the receiver video and sync amplifier circuits and in the camera video amplifier. In

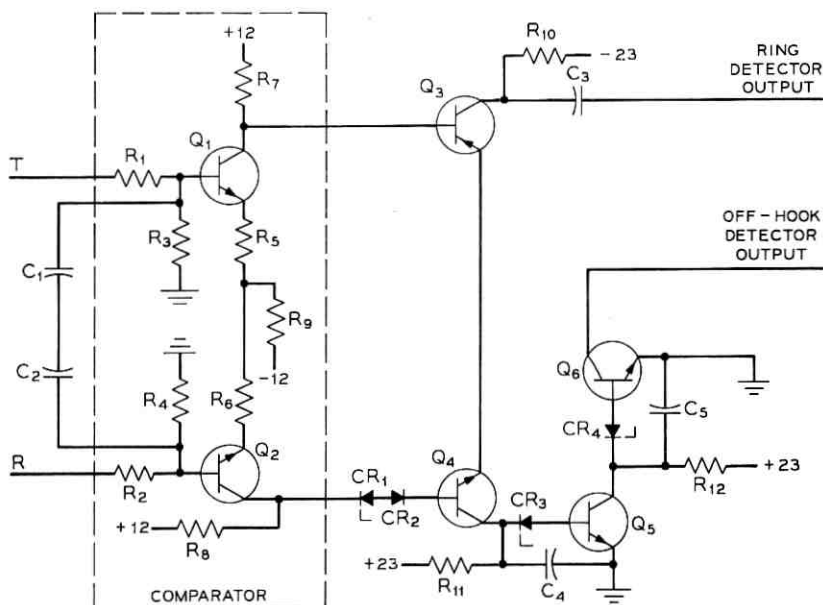


Fig. 21—Off-hook and ring detector schematic.



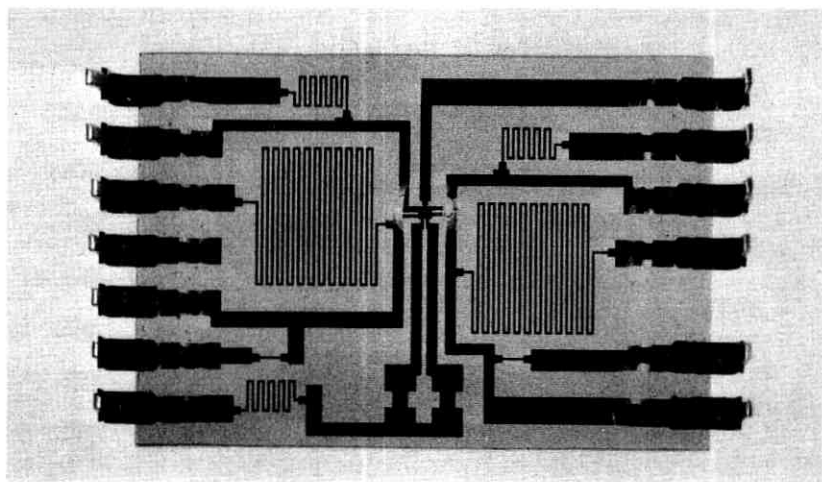


Fig. 22—Off-hook and ring detector comparator.

order to achieve a stable, flat (to  $\pm 0.1$  dB) frequency response beyond 1 MHz with a 12-volt peak-to-peak output swing, the non-Darlington input connections are used. Additionally, both this and the previous version of this operational amplifier provide a low offset voltage ( $< 3$  mVdc).

The third version of this operational amplifier is used in the audio portion of the *Picturephone* station set. Two of these operational amplifiers are used in the audio preamplifier and one in the auxiliary amplifier for the speech circuit. The non-Darlington inputs are used to insure high open loop gain (typically 82 dB). Noise voltage with C-message weighting is less than  $-114$  dBV.

All of these operational amplifiers are fabricated using junction isolation and beam-lead, sealed junction technology on a  $46 \times 46$  mil chip with 16 beam leads. Each chip has 14 npn transistors and 14 resistors totaling 64.2 k $\Omega$ . A picture of the chip is shown in Fig. 23.

### 8.2 General Purpose Operational Amplifier

This operational amplifier was designed for general Bell System use. As used in the *Picturephone* station set, it serves as a voltage comparator in the transmitter automatic gain and iris control (AGIC) circuit. The amplifier is mounted in a dual-in-line package (DIP) and as a voltage comparator has a closed loop gain of 32 dB. Pertinent properties of this device which are required for application in the AGIC circuit are:

Input bias current	$< 2.2 \mu\text{A}$ ,
Power supply voltages	$\geq \pm 12 \text{ V}$ ,
Common mode input voltage	$> 4.5 \text{ V}$ ,
Output swing	$\pm 10 \text{ V}$ .

To increase the positive output swing capability and insure full closure of the iris (when required), the output stage is bypassed. This is accomplished by tying several leads together via the metallization on the ceramic portion of the DIP. Bypassing the output stage also reduces the power dissipation on the chip from 290 mW to 240 mW.

#### IX. BUILDING BLOCK SIC

To meet the need for multiple matched transistors in the station set circuitry, a building block SIC was developed. This SIC con-

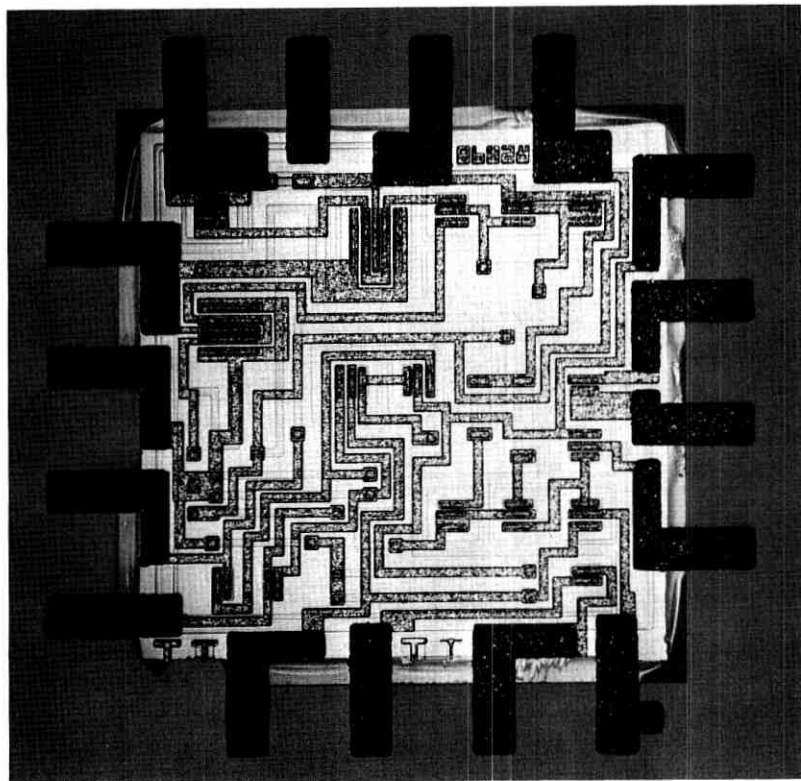


Fig. 23—High output voltage swing operational amplifier.

sists of six transistors connected in the following fashion: Two of the transistors are connected as a Darlington pair, two transistors are connected as a cascode pair and the elements of two transistors are brought out separately. This SIC is used in seven station set circuits:

- (i) camera pre-amp,
- (ii) transmitter sweep circuits,
- (iii) receiver video clamp,
- (iv) receiver sync clamp-AGC lock up protection,
- (v) receiver AGC comparator,
- (vi) receiver AGC comparator—video clamp, and
- (vii) receiver vertical sync.

Beam-lead sealed junction technology is used to fabricate this circuit which is supplied in a DIP.

#### X. AUDIO POWER AMPLIFIER

An audio power amplifier SIC was designed for use as the station set speaker driver. This circuit consists of a differential input stage, level shifter, and power output stage capable of outputs in the  $\frac{3}{4}$  watt range. Internal feedback resistors are used to set the gain of the amplifier but this can be adjusted with an external resistor. The high swing output stage operates in class AB and utilizes matched high power interdigitated transistors capable of peak currents of 500 mA. Approximately 9 mA of quiescent current flows in the entire amplifier.

Sealed junction technology is used to fabricate this SIC on a  $64 \times 68$  mil chip. Five large area high power interdigitated transistors, three medium power transistors, seven low power transistors, and 13 resistors totaling 44.8 k $\Omega$  constitute the circuitry. A photograph of this chip is shown in Fig. 24. The chip is mounted in a 10-lead TO-55 type header with a molybdenum heat spreader. When mounted in the station set a heat sink is attached to aid in heat dissipation.

#### XI. VOLTAGE REGULATOR

A precision hybrid integrated voltage regulator was specifically designed for the *Picturephone* station set. Three of these integrated circuits are used in each station set; two on the receiver circuit board and one on the transmitter circuit board. Essentially all the linear and synchronization circuits on the receiver circuit board utilize the  $\pm 8$  volts provided by its two integrated regulators. This includes the

following circuits: balun-video gate-AGC, roll-off filter and crispening circuit, video display amplifier and clamp, timing amplifier and clamp, vertical and horizontal synchronization circuits, AGC comparator, AGC sampling circuit, AGC lock-up protection circuit, sync separator and the sync gate. The regulator on the transmitter circuit board provides +8 volts to the gamma correction, pre-emphasis, sync insertion, vertical and horizontal sweep drive, and crystal oscillator circuits, and to the camera tube and display tube filaments.

A schematic of the regulator SIC is shown in Fig. 25. Basically it consists of a reference diode (D4), an error amplifier (Q4-Q5), and an output stage (Q1-Q2). Currents of approximately 0.5 mA each are provided to terminals 2 and 10 from external circuitry. Since the output current of this SIC is rated at 30 mA, an external medium power transistor is utilized with each one to achieve the desired higher output current needed in the *Picturephone* station set. By inserting the output device within the feedback loop of the error amplifier, the regulating properties of the SIC are maintained.

To assure almost instant operation of the *Picturephone* station set

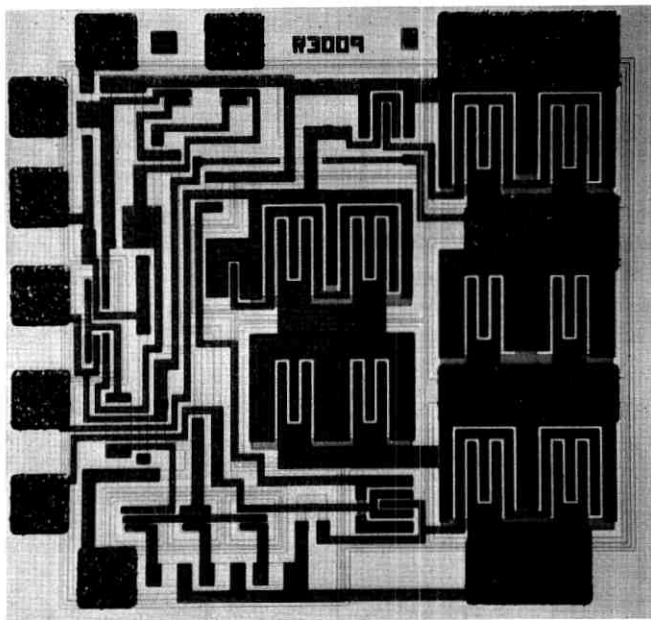


Fig. 24—Audio power amplifier.

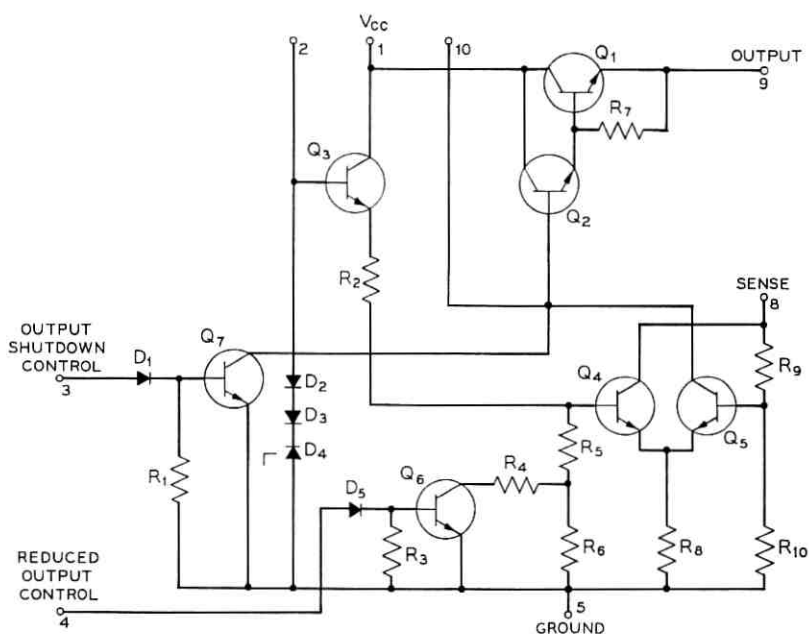


Fig. 25—Voltage regulator.

when a call is initiated or answered, the filament voltages of the camera and display tubes are maintained at 90 percent of their operating voltages during standby periods. This is achieved by supplying a positive control signal (during standby) to terminal 4 of the SIC used on the transmitter board.

During manufacture, the output voltage is set to 8 volts  $\pm 1$  percent for simultaneous worst-case conditions of line, load, and temperature variations. The output impedance is less than 1.5  $\Omega$ , the ripple attenuation is greater than 23 dB, and the output noise (10 Hz to 10 MHz) is less than 0.15 mV rms over the operating temperature range of 10° - 65°C.

The precise output voltage is achieved by matching a monolithic integrated circuit chip to a discrete reference diode. Both devices are eutectic bonded to a molybdenum platform in a 10 lead TO-55 package. The molybdenum platform insures temperature matching of the reference diode and the monolithic SIC. Both the reference diode and the SIC utilize standard nitride sealed junction and beam-lead technologies.

## XII. QUAD "NAND" GATE

The off-hook and ring detector used with non-key telephones was described earlier. Its outputs drive logic circuits which activate the proper equipment. To perform these logic functions, two flip-flops, one three-input "NAND" gate, one two-input "NAND" gate and two inverters are required. Using "NAND" type logic, these functions are obtained very efficiently with two identical SICs without any external components.

The SICs used are one of a family of diode transistor logic (DTL) units developed for the 810A PBX to provide positive logic "NAND" functions. It is especially suitable for use in the *Picturephone* station set because of the following properties.

- (i) It has a high dc noise margin because of its relatively high switching threshold of 1.8 volts.
- (ii) It operates at relatively slow speed with a propagation delay of  $0.5 \mu\text{s}$ . This makes the unit less sensitive to high frequency noise pulses.

This SIC is a  $54 \times 54$  mil beam-leaded junction isolated chip mounted on a ceramic substrate (0.625 inches  $\times$  0.35 inches substrate with 16 external leads). When used to its full capability, it provides three two-input and one expandable three-input gate each having a maximum fan out of five. Figure 26 shows the logic diagram and basic gate schematics for this SIC. As used in the *Picturephone* station set with a 4.7-volt power supply, the unit dissipates 110 mW.

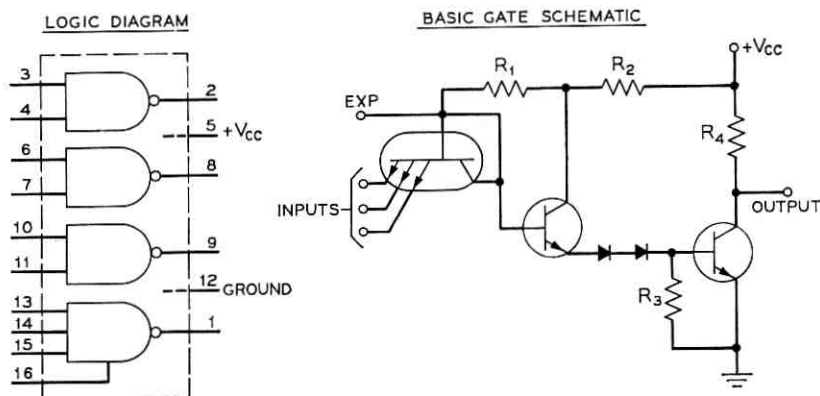


Fig. 26—Quad "NAND" gate logic and schematic of gate.

## XIII. DISCRETE DEVICES

Most of the more than 200 discrete solid-state devices in the station set are general purpose Bell System units originally designed for other projects. In addition, several new discrete devices were required to meet special operating conditions in the *Picturephone* station set. These will be described in this section.

13.1 *Power Transistor*

In order to hold size variations to a minimum, the horizontal sweep circuit of the receiver requires about 1.0 ampere from a well regulated 18-volt supply. An npn transistor with the following properties is required for the output stage of the regulator:

$$\begin{aligned} \text{With } I_e &= 1\text{A,} & h_{FE} &\geq 50; \\ \text{and } T_j &= 25^\circ\text{C,} & V_{CE}(\text{SAT}) &< 0.80 \text{ V,} \\ & & V_{CE}(\text{SUS}) &\geq 20 \text{ V;} \\ & & \text{Power Dissipation} &\geq 5 \text{ watts.} \end{aligned}$$

A power transistor was designed to meet these requirements. It consists of a planar silicon chip with 700 mils of emitter perimeter and 3,300 square mils of base-emitter junction area. The device is housed in a TO-32 package which has a junction to case thermal resistance of  $2^\circ\text{C}$  per watt. This provides a power capability considerably in excess of that required for this application.

An epitaxial silicon layer was not used in the design of this device to avoid potential secondary breakdown problems. A value of resistivity was chosen to allow the current gain and sustain voltage requirements to both be satisfied. To improve reliability, an in-process power screen for two hours is performed to remove early failures caused by hot spots. Reliability tests on screened product indicate that failure rates of less than 10 FITS may be expected for this device as used in the *Picturephone* station set.

13.2 *Medium Power Silicon pnp Transistor*

Both the horizontal sweep circuit in the transmitter and the vertical sweep circuit in the receiver employ push-pull output circuits for driving their respective yokes. Each output stage requires a complementary pair of medium power devices. Existing Bell System transistors were found to be satisfactory for the npn applications. A medium power device was designed for the pnp applications and utilizes a planar epitaxial structure with silicon nitride protection and beam-lead

metallurgy contacts. The transistor chip is mounted in a TO-38 package which has a junction to case thermal resistance of  $6.5^{\circ}\text{C}$  per watt. Basically this transistor is an enlarged version of an existing low-power npn transistor designed to meet higher current and power dissipation requirements.

### 13.3 High-Voltage Silicon Transistor

The station set display tube is driven with the video signal applied to its grid. Two pnp transistors with their collectors tied together constitute the driver stage. One transistor provides the video signal under normal conditions and the second device, in the absence of an incoming signal, provides a gray raster signal to the receiver. For this application, the transistors must meet the following end-of-life requirements:

$$\begin{aligned} BV_{\text{CEO}} &\geq 90 \text{ V at } T = 0 \text{ to } 65^{\circ}\text{C}, \\ h_{\text{FE}} &\geq 20 \text{ at } 0.1 \leq I_o \leq 5 \text{ mA and } V_{\text{CE}} \geq 10 \text{ V}, \\ h_{i_o} &\geq 20 \text{ at } I_o \text{ of } 0.1 \text{ mA and } 6 \text{ mA at } 1 \text{ MHz}, \\ I_{\text{CBO}} &< 100 \text{ nA at } V_{\text{CE}} = 5 \text{ V}, \\ V_{\text{CE}}(\text{SAT}) &< 0.5 \text{ V}, \\ C_{\text{ob}} &< 3 \text{ pF at } V_{\text{CB}} = 5 \text{ V}, \\ \text{Power dissipation} &\text{—less than } 180 \text{ mW peak power} \\ &\text{and } 130 \text{ mW normal power.} \end{aligned}$$

The device developed for these applications utilizes the basic structure of an existing low-power npn transistor. By making appropriate changes in the impurity profile, the high-voltage characteristics and low-current gain capability were achieved. A 6-mil-thick wafer with  $10 \Omega\text{-cm}$  resistivity was used for the starting material. This planar epitaxial silicon transistor uses beam-lead metallurgy contacts and silicon nitride protection. Initially, the devices were provided in TO-18 packages, but later units have been encapsulated in plastic similar to the TO-92 package.

### 13.4 Field Effect Transistor

An n-channel silicon planar epitaxial junction field effect transistor (JFET) is used as a shunt variollosser in the transmitter AGIC circuitry. The signal from the integrated voltage comparator is applied to both the iris and the gate of the JFET. Under high-light conditions the signal from the comparator circuit decreases the bias on the iris control which in turn closes the iris. Simultaneously this signal maintains the JFET in its saturation region and provides minimum gain in the video amplifier. As the scene luminance is decreased,



the iris opens to its widest position. For further reductions in scene luminance, the JFET comes out of its saturation region and serves as a higher shunt resistor for the amplifier input so as to provide up to 6 dB of additional gain.

This JFET is mechanically the same and electrically similar to a JFET developed for use in the N2 and N3 Carrier Systems. For this *Picturephone* application, the device meets tighter requirements on pinch off voltage,  $R_{DS}$ , and saturation drain current.

An additional JFET is used in the camera tube preamplifier circuit and satisfies the requirements of low noise, high-input impedance and low capacitance needed for this application. Except for specifications on  $V_{GS}$ ,  $I_{DSS}$ , and  $g_{fs}$ , these devices are also similar to units developed for the N2 and N3 Carrier Systems.

#### XIV. PURCHASED DEVICES

Several devices used in the receiver horizontal sweep circuit and high-voltage supply are not manufactured by Western Electric Company. These devices are purchased from various suppliers to meet Western Electric specifications. Although these devices are similar to units manufactured by these suppliers for the industrial market, the specifications reflect electrical parameters and reliability consistent with Bell System standards. These devices are discussed below.

##### 14.1 Sweep Transistor

This device is a high-voltage germanium alloy pnp power transistor which is housed in a TO-3 package and has a junction to case thermal resistance of  $1.5^{\circ}\text{C}$  per watt. It is used for two applications in the receiver horizontal sweep circuit as shown in Fig. 27. In one application (Q1), the device serves as a switch to reverse the sweep current in the horizontal yoke. When the transistor is turned off, the flyback pulse of 250 volts appears across the device and the primary winding of the high-voltage transformer which develops the ultor voltage. To insure against excess power dissipation in the transistor at the start of the flyback pulse, the transistor must turn off quickly. The current gain and saturation voltage limits were imposed by worst-case operating conditions.

Pertinent end-of-life requirements are given below:

$$I_{CES} < -24 \text{ mA at } V_{CE} = -310\text{V,}$$

$$V_{CE}(\text{SAT}) < -0.5\text{V at } I_C = -8\text{A and } I_B = -0.4\text{A,}$$

$$h_{FE} > 20 \text{ at } I_C = -8\text{A and } V_{CE} = 0.75\text{V,}$$

$$t_{OFF} < 0.75 \mu\text{s for } I_{C1} = -8\text{A, } I_{B1} = -0.4\text{A, and } I_{B2} = 0.5\text{A.}$$

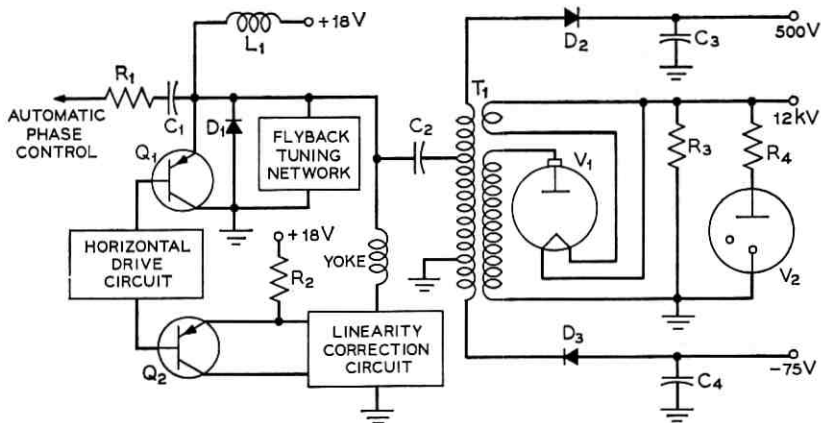


Fig. 27—Receiver horizontal sweep and high-voltage supply.

In the second application ( $Q_2$ ), the device serves as a switch in the linearity correction circuit. A one-ampere current pulse is conducted by the device during the horizontal retrace interval. Requirements imposed by this application are readily met by the specifications listed above.

#### 14.2 Electron Tube Diode

The flyback pulse that appears on the primary of the high-voltage transformer ( $T_1$ ) shown in Fig. 27 is stepped up to 16 kV (open-circuit) on the secondary winding. This pulse is then rectified to develop the dc ultor voltage for the display tube. Rectification is performed by a miniature filamentary high-voltage electron tube diode ( $V_1$ ). Filament power for the tube is provided by a single winding on the secondary of the transformer. Since the tube has a directly heated cathode, the filament is maintained at the dc cathode voltage. An electron tube rather than a solid-state device was chosen for this application because of the faster switching time, lower cost, and adequate reliability provided by the tube.

#### 14.3 Corona Regulator

The display tube picture has a requirement of  $\pm 1$  percent on horizontal size and aspect ratio. Since the raster size is inversely proportional to the square root of the ultor voltage, this potential must possess a stability of  $\pm 2$  percent. This stability was achieved by em-

ploying a gas filled high-voltage regulator (V2) as shown in the circuitry of Fig. 27.

As shown in Fig. 28, the tube consists of a cylindrical metal shell (cathode) with a projecting ceramic insulator which is capped with a metal terminal (anode) designed for plug-in connection with a mating receptacle. The breakdown voltage of the device is controlled by the internal gas pressure. For voltages below breakdown, no current flows. If the voltage is slightly above the breakdown voltage, considerable undesirable noise is generated as current starts to flow. Also, voltage spikes referred to as "pips" can occur. As used in the *Picturephone* station set, the tube must draw greater than  $25 \mu\text{A}$  to get beyond this noise region. A series resistor is employed to prevent the tube from reaching its glow discharge region ( $> 1000 \mu\text{A}$ ) due either to supply voltage transients or pips.

In the station set, the tube is operated in the corona mode of discharge with a current range of  $25 \mu\text{A}$  to  $200 \mu\text{A}$ . The higher current occurs for a dark displayed picture and the lower current occurs for a bright picture. For this range of current, the voltage variation is at most 250 volts. The breakdown voltage of the tube at  $75 \mu\text{A}$  is specified as  $12.0 \pm 0.3 \text{ kV}$ . During manufacture of a station set, the regulator tube is installed and the horizontal size and aspect ratios are set within specifications by adjusting their respective controls. As the displayed picture brightness is varied the regulator tube maintains the size and aspect ratio within tolerance.

#### XV. PASSIVE COMPONENTS

Although solid-state technology has been utilized throughout the

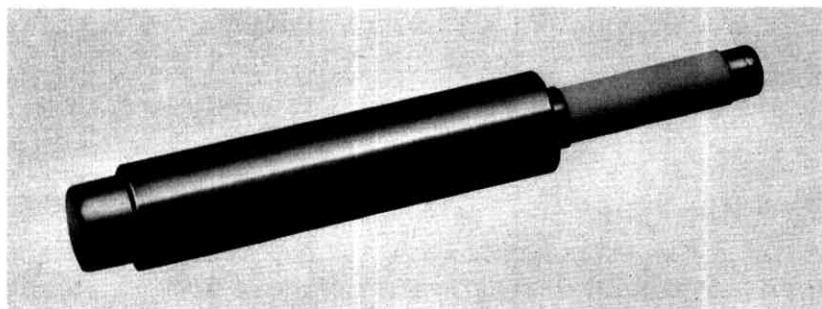


Fig. 28—Corona regulator.

station set design, a substantial number (over 650) of discrete passive elements are employed.

Because of size and cost considerations, the use of inductors has been kept to a minimum. Resistors and capacitors are either coded Western Electric components or are obtained from other suppliers under Western Electric specification. Two types of resistors are of special interest. The first is a 3000 M $\Omega$  high voltage  $\pm 5$  percent metal oxide film resistor. By shunting it across the 12-kV ultor supply, it serves as a bleeder resistor to remove the charge from the ultor supply after the set is turned off.

The second type is the tantalum resistors constituting over half of the discrete resistors used in the station set. Each consists of a tantalum nitride film of appropriate thickness and geometric pattern sputtered on the embossed face of an alkaline earth porcelain substrate. The substrate dimensions are 0.35 inches by 0.10 inches plus two external leads. These resistors are rated at  $\frac{1}{8}$  watt at 70°C and  $\frac{1}{4}$  watt at 40°C and are available with  $\pm 3$  percent and  $\pm 1$  percent tolerances. Because of their small size, low cost, and good stability ( $< 1$  percent change in resistance after 20 years), they provide an excellent means for obtaining accurate signal levels.

#### XVI. CONCLUSIONS

Development of the *Picturephone* station set has required the design of a considerable number of state of the art solid-state and electron-tube devices to meet critical performance and cost objectives. Where available, satisfactory existing devices were used. Integrated circuits were used judiciously to take full advantage of their attributes.

#### XVII. ACKNOWLEDGMENTS

Developing components on schedule for the station set has been a demanding task which has required the closest type of cooperation between Systems Development and Device Development areas. The number of people involved in both areas has been large and we acknowledge their dedication and cooperation in making the *Picturephone* station set possible.

#### REFERENCES

1. Cagle, W. B., Stokes, R. R., and Wright, B. A., "The *Picturephone*® System: 2C Video Telephone Station Set," B.S.T.J., this issue, pp. 271-312.
2. Crowell, M. H., Buck, T. M., Labuda, E. F., Dalton, J. V., and Walsch, E. J., "A Camera Tube with a Silicon Diode Array Target," B.S.T.J., 46, No. 2 (February, 1967), pp. 491-495.

3. Crowell, M. H., and Labuda, E. F., "Silicon Diode Array Camera Tube," B.S.T.J., 48, No. 5 (May-June 1969), pp. 1481-1528.
4. "Proceedings of the Camera Tube Symposium," Bell Telephone Laboratories, Murray Hill, N. J., December 4, 1969.
5. Maunsell, H. I., and Millard, J. B., "The *Picturephone*® System: Digital Encoding of the Video Signal," B.S.T.J., this issue, pp. 459-479.
6. Brown, J. M., "The *Picturephone*® System: Baseband Video Transmission on Loops and Short-Haul Trunks," B.S.T.J., this issue, pp. 395-425.
7. Crater, T. V., "The *Picturephone*® System: Service Standards," B.S.T.J., this issue, pp. 235-269.



**Transmission**





# The Picturephone® System:

## Transmission Plan

By H. E. BROWN

(Manuscript received October 1, 1970)

*An important step in preparing for a nationwide, switched, videotelephone service is the formulation of a transmission plan for the video and audio portion of the network. The transmission plan described in this paper features digital transmission for the long-haul portion of the network and analog transmission for the local portion, close to the customer. The performance of each of the components of the network, that is, station equipment, local loops, trunks and switches, is discussed for the network as it is expected to be configured initially and as it might evolve through the 1970s.*

### I. INTRODUCTION

The signal standards and performance objectives for the overall system discussed in previous papers<sup>1,2</sup> were arrived at through a series of subjective tests and system studies which indicated that the resultant picture quality would be acceptable to the user. Given the end-to-end transmission objectives, the intent in formulating a transmission plan for the network is to strike a balance between the cost and performance of each of the components: the station equipment, loops, local switching and trunking, and long-haul switching and trunking. Impairments are allocated to each of these components such that the overall cost of the service is minimized, taking into account that the loop connecting the station equipment to the local central office is normally dedicated to a single customer, whereas the switching equipment and trunks are shared by hundreds or perhaps thousands of customers.

### II. NETWORK CONFIGURATION

An orderly description of the transmission plan starts with the switching hierarchy,<sup>1,3</sup> which will closely resemble that of the telephone network. The ultimate number of switching levels in the hier-

archy will depend on the relative cost of switching and transmission equipment and the number of subscribers connected to the network. As the network grows and the traffic between various switching nodes increases, high usage or direct trunks will be used. In what follows, the hierarchy is described as it is expected to develop in the mid-to-late seventies. Differences between it and the network in the initial years of service are noted where appropriate.

A call may be viewed as encountering three major pieces of the network: the digital portion and two local analog areas, one at each end of a connection (Fig. 1). For the initial years, the digital portion will consist strictly of transmission facilities, i.e., switching of the video or audio signal in digital form (bitstream switching) is not envisioned. Decoding and recoding to switch the video signal in analog

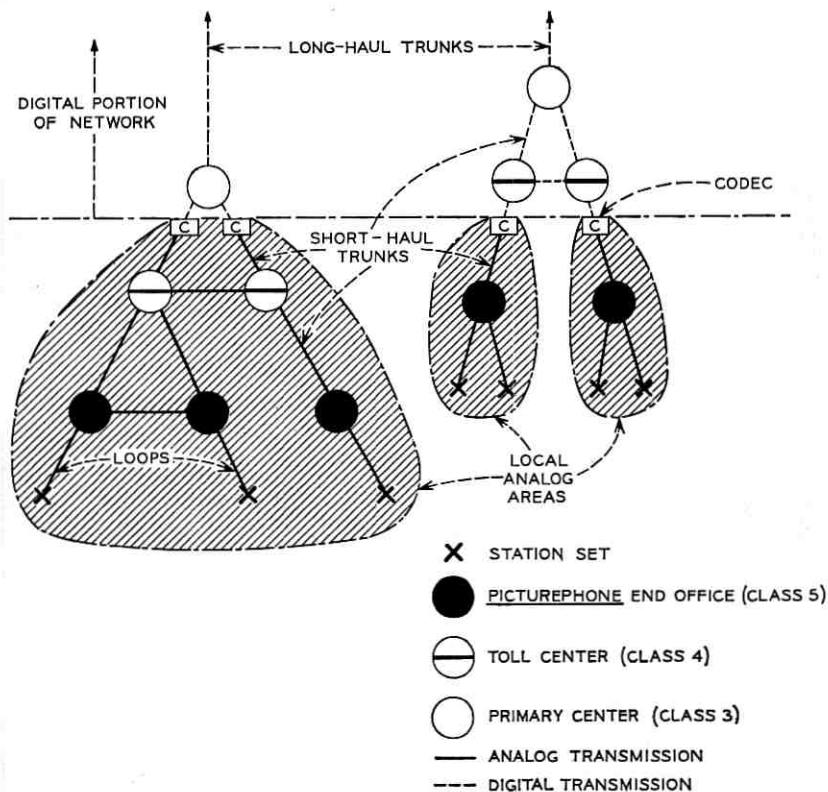


Fig. 1—Picturephone® network.

form is not permitted since the noise that would result from multiple encodings and decodings of the signal would exceed the noise allocation discussed in Section 3.3. Because the noise introduced by multiple encodings and decodings exceeds the amount allocated, a network plan has been formulated which uses bitstream switches rather than analog switches for the long-haul portion of the network. The plan requires that the video signal, once it has been encoded, be transmitted and switched in digital form until it reaches a point where it can be decoded and delivered to its final destination point over analog facilities. Initially, the audio channel, which contains the voice, signaling and supervision information, will be transmitted as a separate signal over any one of a number of facilities already capable of transmitting these signals. In the mid-1970s, with the advent of bitstream switches, the audio channel is expected to be encoded and multiplexed with the encoded video signal to produce a composite digital signal.

Digital transmission over analog and digital facilities is attractive for long-haul transmission of the video and audio signals for several reasons:<sup>4</sup> (i) Virtually all of the impairment in the transmission link occurs in the terminal equipment as the signal is converted from analog to digital and from digital to analog.<sup>5</sup> Very little impairment is introduced in the transmission path; hence, the performance of the digital portion of the network is essentially independent of distance. (ii) Digital facilities now being developed are expected to compete economically with analog facilities for voice service and be significantly cheaper for *Picturephone* service since the video signal requires the equivalent capacity of fewer voice channels on the digital facilities.<sup>4</sup> (iii) Although a bit rate of 6.312 Mb/s will be used initially for transmitting the video information in digital form, the potential exists to reduce the bit rate by a factor of from two to four. A reduction of this magnitude appears possible through the use of redundancy-removal techniques during encoding which take advantage of the similarity between successive lines or frames of video information.<sup>6,7</sup>

The interface between the analog and digital portions of the network is through an A/D converter called a codec (for coder-decoder).<sup>5</sup> The initial service codec samples the incoming analog video signal at the Nyquist rate of about two mega-samples/s and encodes the amplitudes of the differences between successive samples into three-bit binary codes. The resultant bit rate, including miscellaneous bits for framing and maintenance, is 6.312 Mb/s. This rate is the same as that of the T2 digital repeated line.<sup>4,8</sup>

The T2 line, which utilizes wire pairs equipped with regenerative

repeaters, is a digital facility designed for distances up to several hundred miles. Its major application in the *Picturephone* network will be as feeders between the bitstream switches planned to be introduced in future years and the long-haul facilities, and as short-haul trunks where the distance exceeds that permitted for analog transmission.

In the initial years, two systems will be available for transmitting the digitalized *Picturephone* signal over long-haul facilities. The first utilizes the TD-2 microwave radio relay system.<sup>9</sup> Terminal equipment, designated M2R, multiplexes three 6.312-Mb/s coded *Picturephone* signals into a 20.2-Mb/s bitstream. Each 20.2-Mb/s bitstream is transmitted over a single radio channel. There is one channel for each direction of transmission. The second system will make use of the L-4 coaxial cable carrier system.<sup>10</sup> In this system, L Mastergroup Digital (LMD) terminal equipment multiplexes two 6.312-Mb/s coded *Picturephone* signals into a 13.29-Mb/s bitstream suitable for transmission over any one of the six mastergroups on a L-4 coaxial unit in a cable. Thus, one pair of coaxials will be capable of transmitting 12 two-way *Picturephone* signals. In both systems, digital regeneration will be required—about every 400 miles along the route for the TD-2 system and about every 300 miles for the L-4 system.

The local analog area (LAA) is defined as that portion of the network which is wholly interconnectable by analog facilities (Fig. 1). It comprises the customer's station set, loop, end office, toll center,<sup>\*11</sup> short-haul trunks and codec. Figure 1 illustrates two ways in which the codec may be used. In the LAA on the left of Fig. 1, the codec is between the toll center and the next higher level in the hierarchy. In the two LAAs on the right of Fig. 1, the codec is between the end office and the toll center, in which case the toll center would utilize a bitstream switch. Other alternatives are possible; for example, the codec could be part of a digital high-usage trunk group to an office in a distant LAA. Figure 2 shows illustrative local area configurations.

In the early years of service, to keep the cost of the local area equipment low, maximum use of existing plant will be made. This avoids the placement of expensive new cable plant for the exclusive use of *Picturephone* service. Further, the use of existing plant avoids the long lead time required to order and install new cable plant and gives the telephone companies the ability to respond rapidly to new service requests. For the loop and most short-haul trunks, this implies baseband video transmission on paired cable, the same type cable that

\*In future years, the toll center could be a bitstream switch and would be considered part of the digital portion of the network.

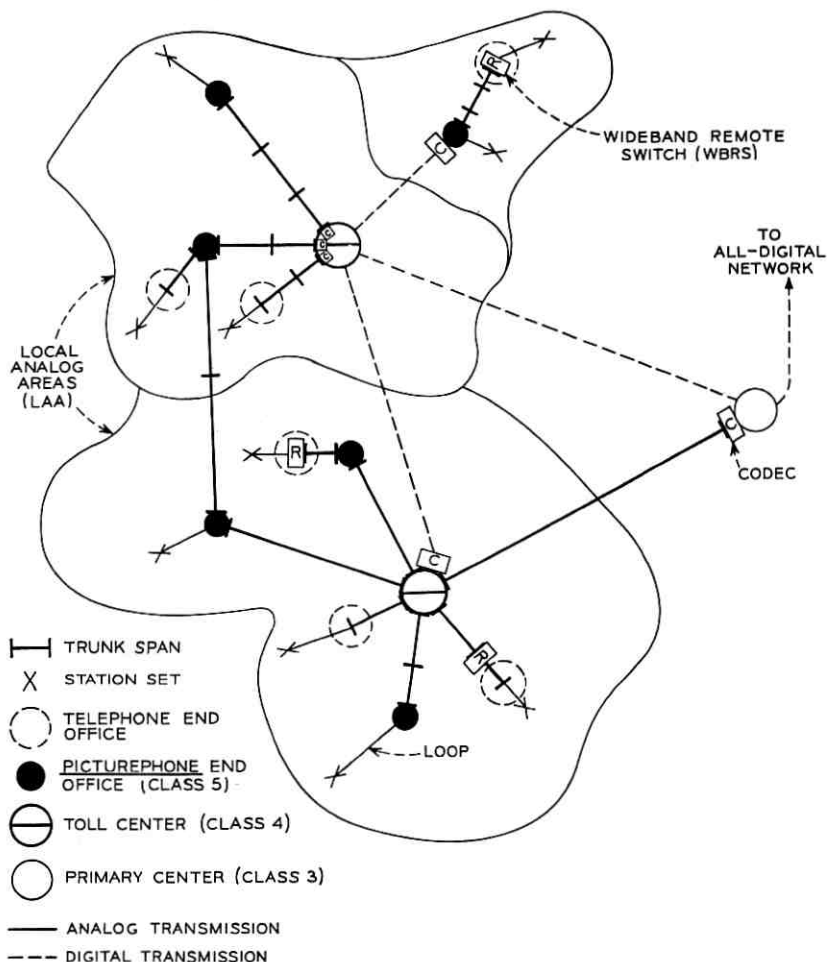


Fig. 2—Illustrative local area configurations.

is used for providing telephone service, with equalizers placed at regular intervals along the cable.<sup>12</sup>

Initially, the equalizers designed for the loops will also be used for local trunks. These equalizers can be used on underground cable of any gauge from 16 to 26 gauge. Aerial cable is not permitted with this design for two reasons. First, diurnal changes in the temperature of aerial cable result in excessive gain and phase deviations from flat gain and linear phase versus frequency. Second, the signal level on the cable is not high enough to override interference from broadcast radio stations.

The use of the loop equalizer for trunks in the initial years has several advantages. Among them is the ability of the equalizer to handle many gauges which gives the engineer planning the local analog area the ability to pick from the available cables the one that best fits his needs. This is of prime importance in the initial years when, on underground cable, the lack of temperature equalization will limit trunk lengths to only a few miles on 22-gauge cable,<sup>12</sup> the prevalent gauge in the trunk network. The ability to use 16-gauge cable on the other hand, permits trunk lengths several times those possible with 22 gauge. Another advantage is that engineering, installation, and maintenance personnel encounter only one basic design of cable equalizer. This equipment permits development of a modestly sized serving area of about six miles in diameter.\*

Extensive expansion of the service is planned and is expected to be made possible by the availability of a second generation of loop and trunk equalizers for local area transmission. These equalizers will take advantage of automatic temperature compensation and increased dynamic range, enabling analog transmission on aerial and underground cable over an area up to about 46 miles in diameter.†

This second generation of equalizers for loops will employ different designs than those for trunks for several reasons. One is that the allocation of impairments to the loop is such that short loops will not require automatic temperature compensation. This avoids the costs involved in providing and maintaining the more complex equipment associated with automatic temperature compensation. Another is that the cost of the trunks is shared among many users; hence, more sophisticated circuitry, capable of a higher level of performance, is justified. Second generation trunks will employ equalizers spaced at intervals up to 6 kft, with automatic temperature compensation in every equalizer, regardless of the length of the trunk.

### III. ALLOCATION OF VIDEO IMPAIRMENTS

The previous sections briefly outlined the switching hierarchy for the *Picturephone* network and the role of the transmission facilities that will be used to interconnect the switching machines. Planning a network and allocating impairments to its individual components require that the network be reviewed as it might ultimately be configured. Thus, a description of a mature network has been included

---

\* One mile for each loop and four miles of analog trunking.

† Three miles for each loop and up to about 40 miles of analog trunking.

in Section II. In the early years, the network will have fewer switching levels, not all the serving areas will be interconnected, loops and trunks will be short and the customers, for the most part, will be concentrated with fairly low density in the downtown areas of major cities. Initially, only analog switching machines will be available. The allocation of impairments discussed in the sections to follow is consistent with the network as it is expected to develop initially and as it is expected to be ultimately configured. It recognizes that future generations of equipment will have expanded capability for transmitting *Picturephone* signals at baseband on paired cable. Also, it recognizes that coding of *Picturephone* signals at bit rates less than 6.312 Mb/s is probable in the future and should have considerable impact on the cost of long-haul transmission.

The allocation of impairments, as given here, represents a snapshot in time of the best balance between the cost and performance of each of the components of the network. These allocations are continually being reviewed as experience is gained and advances are made in technology.

The end-to-end transmission objectives for the video portion of the *Picturephone* network are discussed in a companion article.<sup>2</sup> These objectives were allocated to the components of the network in such a way as to insure that they will seldom be exceeded (no more than one percent of the connections) even on the longest connection. This is accomplished by first establishing the configuration of the longest connection and then apportioning the end-to-end objective to each component.

The impairments of interest may be divided into two categories: those associated with the digital portion of the network and those associated with the analog portion of the network (Fig. 1).

The digital portion of the network will be considered first. The maximum length connection in this portion is assumed to consist of 4500 miles of digital transmission facilities and ten passes through bitstream switches. This permits eight bitstream switches in tandem, with two passes through two of them for operator-handled calls. Thus, the network is configured in a hierarchical manner with a readiness to offer intercity service when and where required to meet the demand.

Digital transmission can affect the quality of the video signal by introducing errors and timing jitter. Errors are introduced at regeneration points along the transmission path where noise, interference from pulses in adjacent time slots, and deviations in the pulse positions combine to cause the regenerators to falsely reproduce the informa-

tion in particular pulse slots. Timing jitter occurs when the time between adjacent pulse positions is altered due to imperfect timing recovery in the regenerators.<sup>13</sup>

The plan for the digital portion of the network assumes the availability of digital channels between all of the major cities in the mid and late 1970s. These digital channels are being designed to transmit a number of different signals in addition to *Picturephone* signals: for example, voice, data and television. Since *Picturephone* signals are not controlling in the establishment of objectives for these digital channels, room is left to allocate some of the overall impairment allowed for *Picturephone* service to the bitstream switches. The error rate allocation for the digital portion of the *Picturephone* network is shown in Table I. The error rate is allocated to two classes of transmission facilities, short- and long-haul, and to the bitstream switching machines. In a companion paper,<sup>2</sup> it is given that an error rate of  $10^{-6}$

TABLE I—ERROR RATE ALLOCATION FOR THE DIGITAL PORTION OF THE NETWORK

Transmission Facilities			
Contributor	Distance (Miles)	Error Rate	Percent*
Long-haul facilities	4000	$2.6 \times 10^{-7}$	95
Short-haul facilities	500	$0.4 \times 10^{-7}$	95
Switching Machines <sup>†,‡</sup>			
		Error Rate	% of Connections
		$3 \times 10^{-9}$	95
		$3 \times 10^{-7}$	99.95

\* The distribution of error rate for a set of transmission facilities is to be considered in two dimensions: percent of time during which a given facility exceeds the stated error rate, and percent of all facilities which exceed, on a long-term average basis, the stated error rate. The former is a function of the likelihood of coincidence of additive, generally time-variant, impairments. The latter is a function of design variations, not generally time variant. In general, short-haul facilities are more subject to design variations; hence, the allocation can be interpreted as percent-of-facilities. Long-haul facilities are more likely to experience time related effects; therefore, the allocation can be interpreted as percent-of-time.

<sup>†</sup> Each machine.

<sup>‡</sup> Error rate on a long-term average for a given connection. The major variation in the error-rate performance of the switch is expected to be caused by differences in path length through the switch and the susceptibility of a particular path to crosstalk interference.



appears to introduce negligible impairment on a *Picturephone* connection. The allocations of Table I are considered adequate to assure that virtually 100 percent of calls of reasonable duration will never experience an error rate greater than  $10^{-6}$ . Further review of the relation between *Picturephone* error-rate requirements and digital transmission and switching error-rate objectives is expected as experience with digital *Picturephone* connections is gained. The actual error-rate performance of a connection will depend on the number of bitstream switches and the mix and length of long-haul and short-haul facilities.

A requirement for jitter has not been established at this time. Circuits for removing a nominal amount of jitter are being built into the long-haul systems and the resulting jitter control is expected to be more than adequate. Should field experience indicate the need for additional de-jitterizing, it will be possible to add the additional circuitry to these systems without affecting the basic design.

The connection on which the allocation for the analog portion of the network is based (maximum connection) is shown in Fig. 3. The maximum analog connection at each end of a *Picturephone* connection consists of the codec, two analog switching machines, three trunk spans, a loop and a station set. In the maximum connection, the loop consists of a wideband remote switch (WBRS), customer switching equipment and interconnecting transmission facilities. The WBRS concentrates the video facilities, reducing the number of video transmission links between it and the *Picturephone* end office. Where a number of customers are served from a telephone end office that is not a *Picturephone* end office, it is generally economical to place a WBRS in the telephone end office. Where a sufficient number of stations are physically close to each other, it is economical to place the WBRS closer to the stations.

The allocation is on the basis of an equal amount for each half of the connection. Six passes through analog switching offices are assumed in the maximum length connection; this includes the four indicated in Fig. 3 and two more to account for operator calls, which require two passes through a switching office. The second generation trunk facilities are being designed to allow up to three 36-kft spans at each end of a connection. The rules for engineering the trunking network permit the three spans to be used in a variety of ways as shown in Fig. 2.

The allocation of the analog impairments is summarized in Table II and Fig. 4 and discussed below. The discussion focuses on the components of the network which have a major impact on the allocation.

A further sub-allocation of transmission impairments to the components of the loop is given in Table III. Figure 3 shows five different loop-serving configurations. As in the case of the analog network, the connection with the maximum number of components is used to sub-allocate the impairments. The controlling configuration is the station behind a key telephone system (KTS),<sup>14</sup> behind a private branch exchange (PBX)<sup>15,16</sup> or a WBRs.

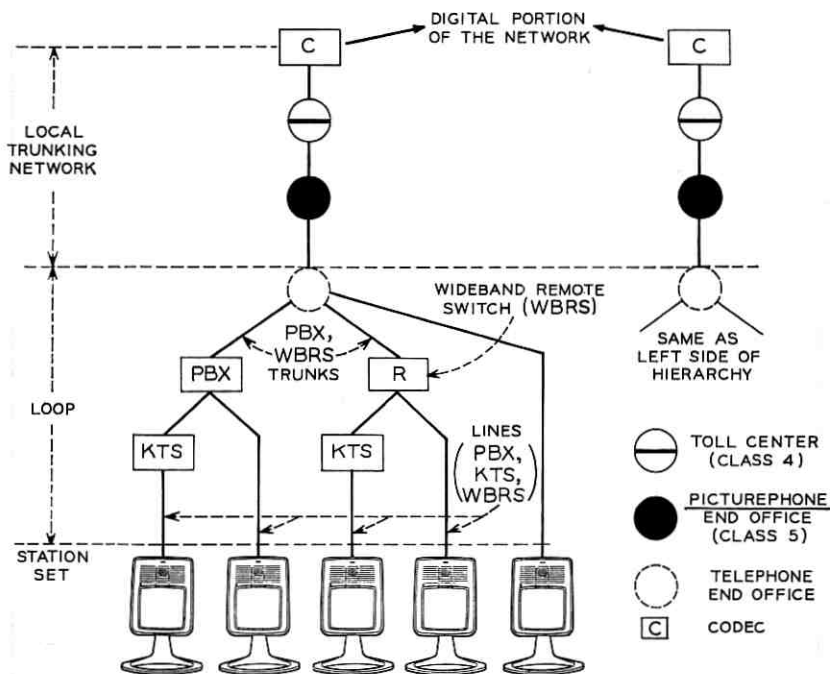


Fig. 3—Local analog area.

Echo rating,<sup>2,17,18</sup> random noise, crosstalk and flat gain have been allocated on a statistical basis. A mean and standard deviation for the performance of each of the components of the analog network is specified, and the performance of the maximum connection is estimated by summing\* the distributions for each of the components. The allocation is on the basis that 99 percent of the maximum connections should meet the objectives specified in Ref. 2. The 95 percent point on the

\*Power summation of log normal distributions for echo rating, random noise and crosstalk except that, in the initial years, echo rating is allocated on a voltage basis (see Section 3.2).

TABLE II—ALLOCATION OF IMPAIRMENTS—ANALOG PORTION OF THE Picturephone NETWORK

	Maximum in Full Network†	Echo Rating (dB)		Random Noise* (dBm <sup>0</sup> )		Self Crosstalk at 150 KHz <sup>0</sup> (dB)		Worst Disturbance at 150 KHz <sup>0</sup> (dB)	Impulse Noise* Probability of exceeding plus or minus 18 millivolts <sup>0</sup>	Flat Gain Variation (dB)		Low Fre- quency Hum†† (dB)	Power Hum (dBm <sup>0</sup> ) 60 Hz 120 Hz 180 Hz
		95% Mean	σ	95% Mean	σ	95% Mean	σ			95% Mean	σ		
STATION SET	1	-35	-37.5 1.5	-54	-56.5 1.5	55	57.5 1.5	45	N/A	±0.6	0 0.3	0.5%	-39 -45 -48
LOOP‡	5 (3)	-31 (-37)	-37.5 4.0	-55	-61.5 4.0	52	66.5 8.8	45	3.0 × 10 <sup>-4</sup>	{ +2.2 } { -3.2 }	{ +1.0 } { -2.0 }	1.5%	-28 -34 -37
SWITCHING OFFICE	6	-48	-54.5 4.0	-62	-65.3 2.0	65	69.9 3.0	45	1.0 × 10 <sup>-4</sup>	±0.2	0 0.1	0.1%	-38 -44 -47
ANALOG TRUNK SPAN	6 (3)	-37 (-45)	-40.3 2.0	-58	-64.5 4.0	57	68.0 6.7	45	0.5 × 10 <sup>-4</sup>	±0.6	0 0.3	0.9%	-33 -39 -42
CODEC	1	-40	-42.5 1.5	-46		60	62.5 1.5	45	N/A	±0.2	0 0.1	0.3%	N/A N/A N/A
OVERALL SYSTEM OBJECTIVES		-26‡		-44‡		45‡		45	1.5 × 10 <sup>-4</sup>	{ +5.0 } { -7.0 }		10%	-22 -28 -31

\* Weighted Noise—Use weighting curve, Fig. 5.

\*\* Referenced to 0 PTLF (see Section 3.1).

† Table III allocates to each component of a LOOP.

‡ Allocation is to each appearance of the component, e.g., one trunk span.

§ 99% of the maximum connections are expected to meet this objective.

() Allocation for the initial service period only, before automatic equalizers are available. The -37 dB for the loop is for each loop. The -43 dB for the analog trunk span assumes only one span at each end of the connection.

Δ No variation is assumed in this allocation.

N/A = not applicable.

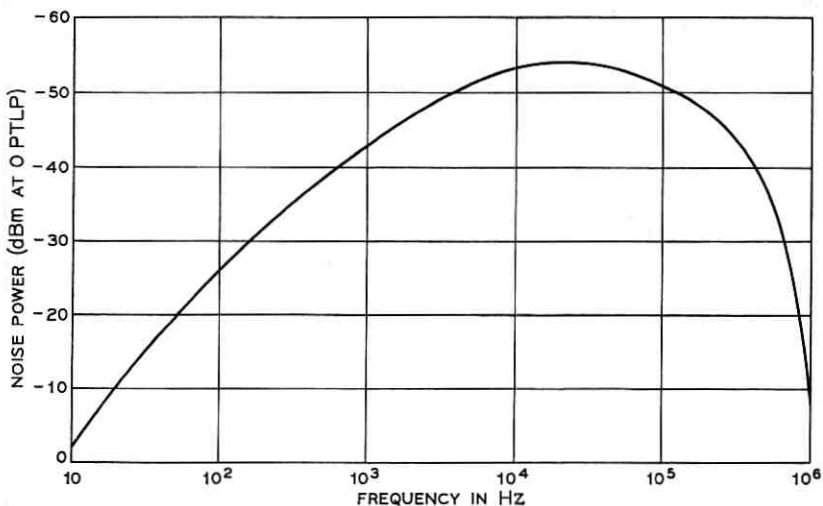


Fig. 4—Single frequency interference objective.

performance distribution for each of the components has significance in that it is the point used during the design stage to specify the minimum level of performance which should be achieved in essentially all situations.

### 3.1 Video Transmission Level Plan

The objectives for random noise, self crosstalk, worst-disturber crosstalk, impulse noise, power hum and single frequency interference

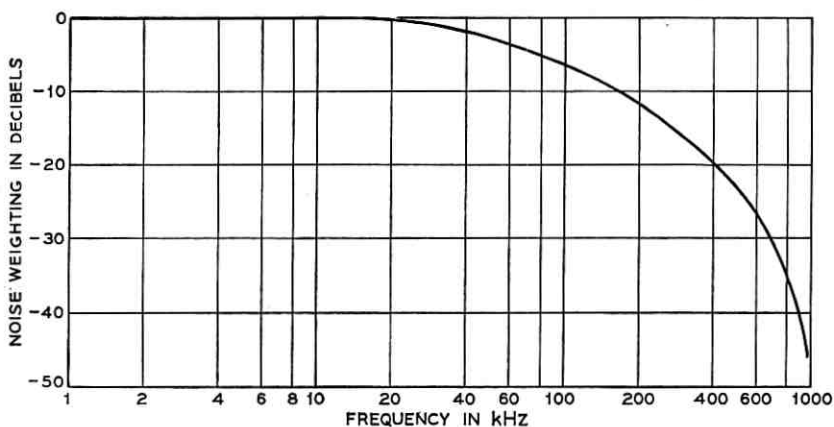


Fig. 5—Noise weighting curve (including de-emphasis, roll-off and eye weighting).

TABLE III—SUB-ALLOCATION OF IMPAIRMENTS—LOOP PORTION OF THE NETWORK

	Echo Rating (dB)		Random Noise* (dBm <sup>**</sup> )		Self Crosstalk (loss in dB at 150 kHz <sup>**</sup> )		Worst Disturber Crosstalk (loss in dB at 150 kHz <sup>**</sup> )	Impulse Noise (probability of noise exceeding voltage plus or minus 18 millivolts <sup>**</sup> )	Flat Gain Variation (dB)		Low Frequency Roll-off (dB)	Power Hum (dBm <sup>**</sup> ) 60 Hz 120 Hz 180 Hz
	95%	Mean $\sigma$	95%	Mean $\sigma$	95%	Mean $\sigma$			95%	Mean $\sigma$		
KTS	-50.0	-52.5 1.5	-60.0	-62.5 1.5	58.0	59.6 1.0	45	$0.5 \times 10^{-4}$	$\left. \begin{array}{l} 0 \\ -0.4 \\ -0.1 \\ -0.5 \end{array} \right\}$	N/A	-40 -47 -50	
PBX OR WBRS	-50.0	-51.6 1.0	-61.0	-62.6 1.0	62.0	63.6 1.0	45	$1.0 \times 10^{-4}$	$\left. \begin{array}{l} -0.1 \\ -0.5 \end{array} \right\}$	N/A	-38 -44 -47	
TRANS- MISSION†	-31.1	-34.7 2.2	-56.7	-60.2 2.1	52.7	64.7 7.3	45	$1.5 \times 10^{-4}$	$\pm 2.5$	$\pm 1.5$ 0.5	1.5%	-29 -35 -38
OVERALL LOOP	-31‡		-55‡		52‡		45	$3 \times 10^{-4}$	$\left. \begin{array}{l} +2.2 \\ -3.2 \end{array} \right\}$	1.5%	-28 -34 -37	

\* Weighted Noise—Use weighting curve, Fig. 5.

\*\* Referenced to 0 PTLP (see Section 3.1).

† Total for KTS line, PBX or WBRS line, and PBX or WBRS trunk.

‡ 95% of the loops are expected to meet this objective.

N/A = not applicable.

are referenced to a point in the system designated as the *Zero Picturephone* Transmission Level Point (0 PTLP). This reference point, 0 PTLP, is defined as the output of the central office loop equalizer in the direction of transmission from the station set to the central office. In the initial years of service (when first generation equalizers will be used), 0 PTLP will also appear at other points in the system, as indicated in Fig. 6. The level plan for the latter years of service (second generation equalizers) has not been established. Consideration is being given to increasing the transmission level of certain of the equalizers in the loop and trunk to gain immunity from interference from carrier systems and broadcast radio stations. The level plan for the latter years of service is expected to retain 0 PTLP at the output of the equalizers facing the PBX and central office switches.

The nominal signal characteristics at the 0 PTLP are covered in more detail in Ref. 19.

To decrease the susceptibility of the video signal to interference during transmission over the *Picturephone* network, the video signal is pre-emphasized before it enters the network and de-emphasized after passing through the network. Since the end-to-end objectives in Ref. 2 are referred to the input to a reference station set receiver which does not contain a de-emphasis filter, the objectives<sup>2</sup> must be adjusted before they can be allocated. Appendix A contains a discussion of the pre-emphasis plan, how it is implemented, the advantages of this

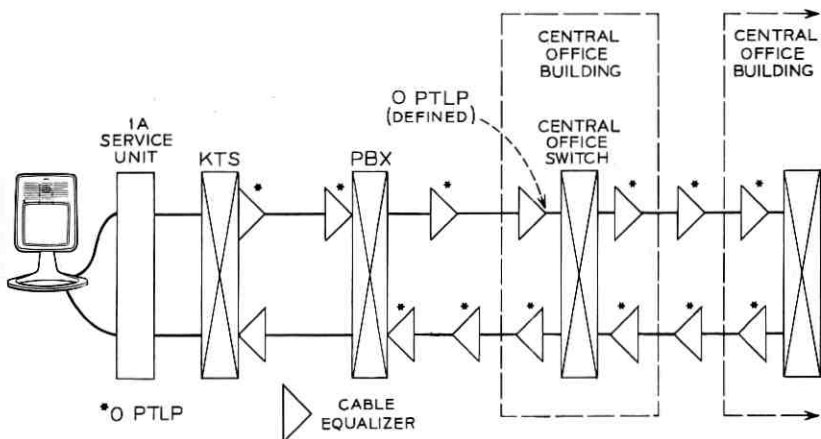


Fig. 6—Transmission level plan (initial service).

form of signal shaping and the technique for adjusting the objectives in Ref. 2 to account for de-emphasis.

### 3.2 Echo Rating

Echo rating (ER) is a technique for evaluating the effects of gain and phase deviations, from nominal, of a video communications channel. It is applicable to the portion of the frequency spectrum from about 20 kHz to 1000 kHz. Gain and phase deviations below the line frequency are governed by the objective for tilt which is covered later in this paper. There is no requirement at present for line time distortion (around the 8-kHz region).

Echo rating plays a very important role in the establishment of the transmission plan for the local analog area where the video signal, in analog form, is susceptible to gain and phase deviations introduced during transmission. The following are illustrations of the extent to which ER controls the gain and phase characteristic of a video channel.

The first example, Fig. 7, illustrates, as a function of cable length, the echo rating of a number of types of cable normally found in the local analog area. In this figure, it is assumed that neither the transmission loss nor the phase of the cable is equalized. To illustrate the

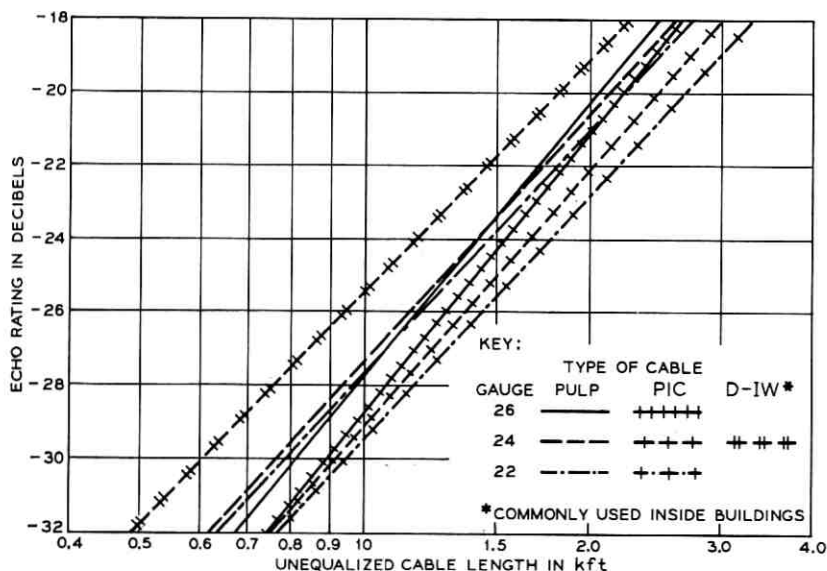


Fig. 7—Echo rating versus unequalized cable length.

use of the figure, assume the unequalized cable is allocated the entire end-to-end echo-rating objective of  $-26$  dB; then, a cross-country connection could have no more than about 1200 feet of unequalized cable in the analog portion of the connection. This includes, for example, cable on the customer's premises and path-length variation through analog switching offices.

In the next example, Fig. 8, it is assumed that a length of cable is perfectly equalized at some nominal value of cable temperature but that the cable temperature is allowed to vary from nominal without an appropriate change in equalizer characteristic from that at nominal temperature. For example, assume 18 kft of 26-gauge cable pairs subjected to a temperature variation of  $\pm 25^\circ\text{F}$  from nominal. This results in an equalized length times unequalized temperature change of  $18 \times 25$  or 450 kft  $^\circ\text{F}$ . Entering the abscissa of Fig. 8 at 450 gives an echo rating of  $-26$  dB for 26-gauge cable pairs.

The next three examples illustrate the effect of one or more bridge taps\* on echo rating. The assumption is made that the cable is perfectly equalized without bridge taps. The bridge tap is then added, causing distortion in the gain and phase characteristics of the cable. The cable is assumed to be 26 gauge. Figure 9a shows the echo rating of a single bridge tap as a function of its length. Note that the echo

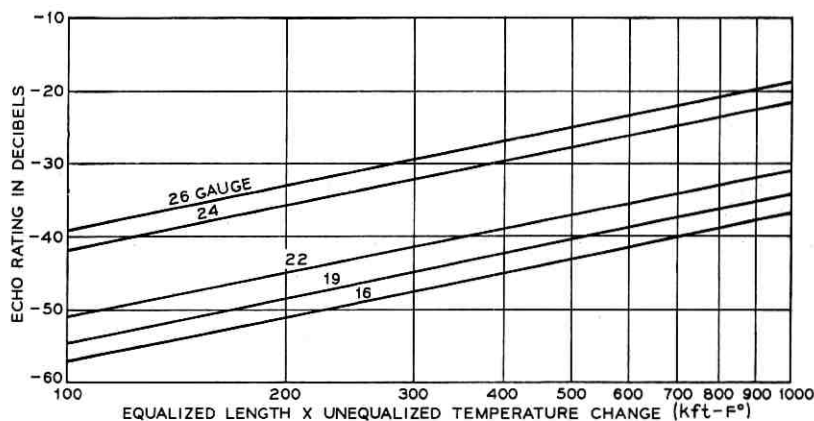


Fig. 8—Effect of change in cable temperature on echo rating.

\* A bridge tap, as used here, is a wire pair (bridged pair) one end of which is connected across the transmission pair. The distant end of the bridged pair is considered to be unterminated.



rating of a single bridge tap decreases 8 dB with each doubling of its length until the bridge tap is about 100 feet long. The reason for this is that bridge taps under 100 feet in length produce a similar loss characteristic that smoothly increases with frequency and results in a single close-in echo in the picture, appearing to the viewer as a loss of resolution. Bridge taps longer than 100 feet produce a ripple in the video band which may be represented as a multiplicity of echoes in the video picture, each echo having a different time delay from the main signal and, hence, producing a different subjective effect.<sup>2</sup>

Figure 9b shows the echo rating of a set of bridge taps of equal length connected in parallel at the same point in the equalized cable. Although the number of bridge taps in the set is an integer, the points representing the echo rating have been connected to simplify the figure. For multiple bridge taps, each less than 100 feet, the echo rating of the set of multiple taps decreases about 8 dB for each doubling of the number of bridge taps in the set.

Figure 9c illustrates yet another point regarding bridge taps. This figure is derived from Figs. 9a and 9b and illustrates the echo rating of a set of parallel bridge taps as a function of their cumulative length. Note that, as long as each bridge tap in the set is less than about 100 feet in length, the echo rating of the set is essentially dependent only on the total cumulative length of the bridge taps in the set.

These examples illustrate how lengths of cable, temperature variations and bridge taps, which are of little importance in the telephone network, are significant in the design of the *Picturephone* network.

Two echo-rating allocations are given for the loop and trunk spans in Table II. The numbers in parentheses apply to the initial service period when static equalizers without automatic temperature compensation are available. Note that they are significantly more stringent than those which apply to the period beyond initial service. The reason is that the echo-rating performance of the initial service loops and trunk spans will be controlled primarily by phase distortion and by the effects of temperature variation, both of which accumulate linearly with distance. Since the seasons of the year and, hence, the extremes of the temperature range occur roughly simultaneously in all parts of the country, the impairments introduced into the various network components by changes in temperature will tend to add in phase. Thus, for the initial few years, the impairments in the loops and trunks have been assumed to add together on a voltage basis, resulting in an echo-rating allocation of -27 dB. This is then added to the rest

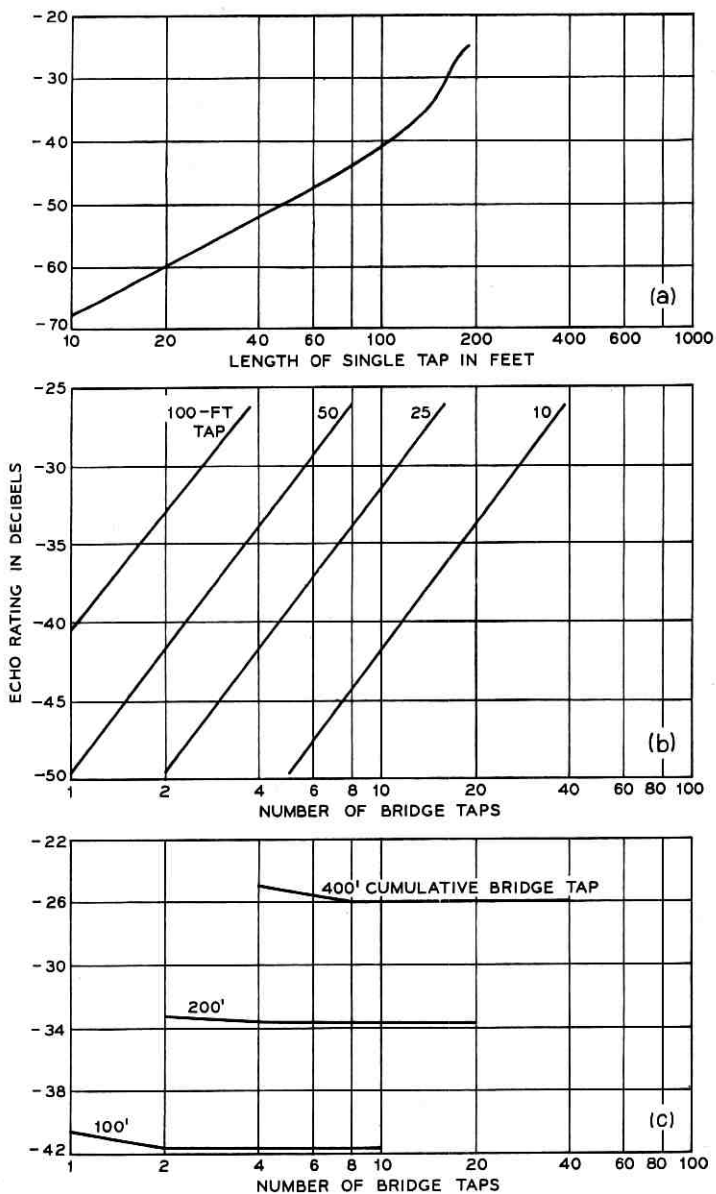


Fig. 9—(a) Echo rating of a 26-gauge equalized cable pair with a single bridge tap (unequalized). (b) Echo rating of a 26-gauge equalized cable pair with multiple bridge taps (unequalized). (c) Echo rating of a 26-gauge equalized cable pair with cumulative bridge taps (unequalized).

of the network ( $-33$  dB) on a power basis to meet an end-to-end performance objective of  $-26$  dB. For service after the initial period, all components are assumed to add on a power basis.

The loop is given the major portion of the allocation since its cost is quite sensitive to the echo-rating allocation and not shared by many customers.

The allocation to the switching office results in a controlled floor plan to minimize the distance between equipment bays and, hence, minimize transmission loss variability.<sup>20</sup>

The allocation to the trunk span is considerably less than that given to the loop since the cost of the trunk span is shared among many customers.

The remaining allocations are essentially determined by practical limitations in equipment design and are not affected significantly by system-design tradeoffs. The allocation to the station set reflects the challenges associated with installing and maintaining equipment on the customer's premises. The allocation to the codec is intended as a control on the accuracy of its input and output filters. The WBRS has a more stringent allocation than the central office switch because it has fewer switching stages and is smaller in size, subjecting the video signal to less path loss variation.

An allocation plan, such as is outlined above, which fixes the performance of each of the components of the network, has many advantages in that it results in fixed maintenance limits, uniform engineering procedures and a minimum of record keeping for the operating companies. For the initial years, however, it is recognized that a more flexible allocation plan, which permits tradeoffs of the allocation between loops and trunks where appropriate, will permit a larger local serving area. For example, in serving areas where no analog trunking exists, it is permissible for the loops to assume the allocation for half of the end-to-end connection. In the early years, this permits a 50-percent increase in loop lengths. As the network grows and the challenges of administering and maintaining the network increase, the need for such flexibility must be carefully studied and balanced against the administrative and maintenance complications.

Since echo rating is a major factor in local area planning and, hence, a strong factor in the determination of the cost of local service, the end-to-end objective and the allocations are under constant review.

### 3.3 *Random Noise*

The random-noise objective is split into two parts: one for the codec and the other for the remaining components of the network. The

codec allocation is based on the experimental performance of the codec described in a companion paper.<sup>5</sup> The allocation to the remaining portion of the network is such that it adds only a nominal amount to the codec noise.

The transmitting portion of the station set automatically corrects for a wide range of lighting levels.<sup>19</sup> For brightly lighted subjects, the aperture of the iris is reduced to prevent excessive light from hitting the target. For dimly lighted subjects, the iris is opened to its extreme position, and the gain at the camera output is increased to maintain the proper signal level at the station set output. Noise generated at the camera target and in the preamplifier at the output of the target is enhanced by the additional gain that is inserted at low light levels. The allocation given to the station set is one which is met for normal room lighting and is exceeded for very low light levels. However, the effect of low light levels on overall noise is minimal, since the noise performance of the network is dominated by the codec.

Noise in baseband loops and trunks is primarily introduced through crosstalk from other transmission systems in the same cable.<sup>21</sup> These systems may be carrying voice, data or *Picturephone* signals. A major source of noise in the trunks is expected to be interference from T1 lines, which are used extensively in metropolitan areas where *Picturephone* baseband equalizers will find their major application. Since both T1 and the baseband equalizer system for *Picturephone* trunks will vie for newly placed cable plant, the noise allocation to the trunk span must be great enough to permit compatibility of the two systems in the same cable.

### 3.4 *Self-Crosstalk*

Self-crosstalk is the interference due to the coupling between the two directions of transmission of a connection. The major portion of the allocation is given to the station set and to the baseband loops and trunk spans. The low signal levels present in the station set at the camera output make it extremely difficult and expensive to insure a high self-crosstalk loss. Even with the allocation shown, extensive shielding and very careful design are required. The apportionment between loops and trunk spans reflects the fact that it is more difficult to achieve separation between the two directions of transmission in the loop, where the number of pairs in loop cables is often less than that in trunk cables and the pairs are frequently reassigned to provide service to new customers. In the loop, particularly in the portion on the customer's premises, care must be taken in the selection of cables,

since the two directions of transmission will generally be in the same small cable where physical separation is impractical.

The suballocation within the loop (Table III) illustrates this by giving a significant portion of the allocation to the transmission facilities. The remaining allocations reflect the expected performance of the switching equipment. The self-crosstalk objective is being reviewed; hence, the listed allocations are subject to change.

### 3.5 Worst-Disturber Crosstalk

Intersystem crosstalk is, in general, controlled by the noise requirement discussed in Section 3.3. On the other hand, interference from one *Picturephone* signal to another is controlled by worst-disturber crosstalk. Worst-disturber crosstalk occurs when one particular interferer dominates over all others and appears as a distinctive pattern on a display tube. Therefore, a more stringent objective than that set in Section 3.3 is required. This objective is not allocated since the probability is very low that the same interferer will dominate in more than one component of the network.

### 3.6 Impulse Noise

The impulse-noise allocation given here is based on experience gained through a number of trials of *Picturephone* service. It is expected that the various components of the network will not experience difficulty meeting these objectives.

### 3.7 Flat Gain Variation

The station-set receiver is designed to accept a signal from the network provided that the signal level is not greater than 5 dB above or 7dB below the nominal design signal level. An automatic gain control circuit in the station set receiver<sup>19</sup> corrects the signal level to meet the end-to-end objectives of  $\pm 0.5$  dB from camera to display. The major source of gain variation is in the loop, where temperature variations are expected to be controlling. On underground cable, maximum deviations of gain due to changes in temperature tend to occur simultaneously with the highest temperature occurring in the summer and the lowest in the winter. To control the gain performance at the extremes of the seasonal swings, the mean of the loop allocation has been specified at two points, mid-summer and mid-winter. The specification of the objective in this manner may turn out to be unduly severe for aerial cable, since the diurnal variations are more random and the regulating system in later designs will tend to further randomize the effects of temperature.

### 3.8 *Low-Frequency Roll-Off (Tilt)*

The tilt objective makes uneconomical the use of transformers in the network. Without transformers, powering of the intermediate equalizers in the loop and trunk systems becomes a major challenge to permit powering over the cable pair and at the same time provide an acceptable termination to the cable pair to maintain good tilt and echo rating performance. Consequently, these systems are given a large portion of the allocation. The loop receives the largest portion of the allocation because of its greater impact on system cost.

### 3.9 *Single Frequency Interference*

The end-to-end objective is given in Fig. 4. This objective is not allocated to the components of the network, since it is unlikely that exactly the same tone will appear in more than one component of the network (except for power hum; see Section 3.10). Multiple single frequency tones tend to appear as noise and are controlled by the random noise objective given in Section 3.3. The single frequency interference objective has its greatest impact in the loop, where aerial cables are particularly susceptible to interference from broadcast radio stations.

Trunk spans can be exposed to single frequency interference from voice band signaling tones and carrier pilots, but the crosstalk loss between cable pairs at these frequencies is expected to give sufficient protection.

### 3.10 *Power Hum*

The overall requirement at each frequency is obtained from Fig. 4 and is then allocated to each portion of the network. Power hum is allocated, since it is quite likely that this interference will be experienced in various places throughout the network. The major source of power hum is anticipated in the loops, where aerial cables are quite susceptible to power line pickup. Analog trunks are also susceptible, but to a lesser degree, since the cables will generally be in underground conduit. Experience in trials of *Picturephone* service indicates that power hum will not be a major problem.

## IV. AUDIO TRANSMISSION PLAN

### 4.1 *General Features*

#### 4.1.1 *Introduction*

In the initial years of *Picturephone* service, the audio channel, which is used for transmitting voice, signaling and supervision infor-

mation, will utilize the same facilities as those presently used in the DDD network. Beginning with the introduction of bitstream switching in the mid-1970s, the information in the audio channel is expected to be encoded at the point in the hierarchy where the video signal is encoded. The encoded audio signal would then be multiplexed with the digitalized video signal to form a composite 6.312-Mb/s bitstream which would be transmitted and switched in digital form in the digital portion of the network.

The telephone loop facility presently used to connect to the nationwide DDD network will also be used as the audio channel for *Picturephone* calls. The loop transmission plan is identical to that used for providing telephone service.

The audio channel associated with *Picturephone* trunks between local offices (direct trunks) is used only for *Picturephone* calls but is being designed according to the same transmission plan as that used for direct DDD trunks.

Trunks between *Picturephone* end offices and toll offices (toll-connecting trunks) and between toll offices (intertoll trunks) will be dedicated to *Picturephone* use and will follow a transmission plan specifically designed for *Picturephone* service. The switching and transmission of audio signals in digital format on a built-up connection necessitates that all transmission loss be inserted in the analog portion of the network. This plan represents a significant departure from the via net loss (VNL) plan used in the DDD network, wherein every trunk is assigned a specific loss according to its physical length. Thus, in the toll portion of the *Picturephone* network, the loss is essentially independent of distance and number of trunks in the connection. This approach is compatible with bitstream switching, is simple and hence easier to administer and maintain, is lower in cost, and has better performance than a plan based on VNL design which would require decoding and encoding of the audio signal at every toll-switching point.

This section is devoted to a discussion of the transmission plan for this, the toll portion of the audio part of the *Picturephone* network.

#### 4.1.2 Objectives and Assumptions

The primary objective for the quality of audio transmission on *Picturephone* calls is to provide a level of performance equal to or better than that of the DDD telephone network. To see how this is accomplished, it is first necessary to explore some of the basic features of the transmission plan and how they affect audio quality. The

transmission plan assumes digital transmission and bitstream switching of the audio signal at Class 3 and higher switching centers, and at Class 4 offices when economically attractive. The plan also assumes that *Picturephone* service is primarily a handsfree service, i.e., speakerphones will be used on most *Picturephone* calls; but at the same time, the plan recognizes that acceptable performance must be provided when a handset is used.

The transmission of the audio information in digital form over the same facility as is used for the video information avoids the cost associated with a separate audio facility and, because digital transmission is used, results in essentially noise-free transmission, independent of distance. Further, the administration of the composite video-audio trunk is simplified.

Switching the voice information in digital form avoids degrading effects from reflections which result from impedance irregularities. The common control equipment in local telephone switching machines, such as No. 5 crossbar, which employ two-wire switching of the voice information on DDD calls, is used to control the operation of the four-wire video switch for both the local and toll portion of the *Picturephone* hierarchy. Bitstream switching avoids the need for decoding the voice signal, passing it through a hybrid, switching it on a two-wire basis at each office in the hierarchy and encoding it for transmission between switching points. Thus, impedance mismatches at the point of conversion to two-wire are avoided, eliminating the associated costs of controlling the reflections that otherwise would result. The avoidance of echoes from reflections in the audio path at the bitstream switches also permits the digital transmission facilities to be operated at zero loss. A nominal loss which is required in the toll network to insure gain stability, to control echoes on calls involving handsets, and to control noise originating in the trunk facilities is inserted in the toll-connecting trunks. This approach results in essentially a fixed transmission loss between local switching offices, independent of the number of trunks in the connection or the length of the trunks. Where both the toll-connecting and intertoll trunks are switched through bitstream switches, the loss objective from local office to local office is 6 dB. Where the connection involves analog toll-connecting trunks or stations served directly from toll offices, a different loss is incorporated to reflect the resultant differences in susceptibility to echoes and to provide adequate singing margins. These differences are covered in more detail later in the paper. In addition to avoiding reflections in the audio path, bitstream switching avoids



the accumulation of coding noise that would otherwise result from frequent encoding and decoding of the voice signal.

The use of speakerphone on most *Picturephone* calls has a significant effect on the transmission path since either the speakerphone transmitter or receiver at each end has an extra 15 dB of voice-switched loss inserted:<sup>22</sup> this essentially eliminates audio echo as an impairment when both ends of a connection are using a speakerphone. The voice-switched loss is also present in the receive path when a talker is using a speakerphone and the listener is using a handset.

#### 4.1.3 Development of the Picturephone Audio Transmission Plan

The following sections describe the audio transmission loss plan designed for *Picturephone* service. First, the fixed end-office-to-end-office (EO-to-EO) loss requirements for a connection with all toll switching in the bitstream format are established. The choice of loss is shown to be based on several performance criteria. Next, this analysis is extended to the combined analog/digital toll switching environment to determine the loss design for trunks which are switched in analog format at the Class 4 office. The design is structured on a set of constraints for audio switching and a set of performance and maintenance guidelines consistent with satisfying the overall audio transmission requirements. Finally, the EO-to-EO losses are tabulated for various types of connections.

### 4.2 Audio Transmission Loss Plan (Bitstream Toll Switching)

#### 4.2.1 Transmission Considerations

The audio performance of the network was evaluated from four points of view: gain stability, echo, received volume and received noise. To accomplish this, certain assumptions are made regarding the ex-

TABLE IV—ASSUMED VALUES OF AUDIO TRANSMISSION PARAMETERS

	Mean	$\sigma$
Talker volume-speakerphone (originating end office)	-17 VU	5.0 dB
Talker volume-handset (originating end office)	-16.3 VU	6.4 dB
Loop loss (1300 Hz)	4.8 dB	1.8 dB
Noise (all digital trunks)	23 dB <sub>rnc0</sub>	3.0 dB
Noise (VF trunks—0 to 8 miles)	10 dB <sub>rnc0</sub>	7.0 dB
Loop return loss (speakerphone)	13 dB	2.0 dB
Loop return loss (handset)	11 dB	2.2 dB
Acoustic coupling (at speakerphone terminals)	10 dB	2.0 dB

pected performance of the network. These are shown in Table IV.

Under these assumptions, a minimum loss of 6 dB is required between local *Picturephone* switching end offices to insure adequate gain stability on four-wire facilities and to avoid a near singing condition which would produce a hollowness in the speech quality. Another factor which motivates in the direction of high transmission loss is the desire to minimize the need for echo suppressors. The use of echo suppressors on a facility increases the cost of the facility and introduces an impairment by permitting voice signals to pass in only one direction at a time. The allowable round-trip delay of a connection for which suppressors are not required increases as loss is added to the echo path.<sup>23</sup>

The upper bound on the amount of loss in the network is set by two factors: (i) contrast in the received volume between local area *Picturephone* calls and local DDD calls, and (ii) the absolute level of received volume. Since all the loss in the network must be inserted in the signal path where the signal is in analog format, the loss is present in all connections, whether they are part of a local call or part of a long distance call. This may be compared to the DDD network, where the loss is distributed throughout the network. This difference in administering the loss gives rise to the possibility of a higher loss on local *Picturephone* calls than on local DDD calls. Hence, the contrast between them must be considered. Where a speakerphone is involved at the receiving end, it is possible to compensate for a portion of the connection loss by adjusting the gain in the receiving path of the speakerphone, thereby reducing contrast and maintaining adequate volume. Compensating for loss in this manner, however, results in an increase in speakerphone received noise and the amount of loss in the speakerphone that must be switched in and out as users alternately talk and listen.

Since the received volume on calls involving a handset at the receiving end cannot be adjusted, as in the case of a speakerphone, calls involving a handset control the upper bound on the amount of loss that can be inserted.

In balancing the above factors to arrive at a value for the transmission loss of the network, four talker/listener situations were considered: (1) speakerphone talking to speakerphone, (2) speakerphone talking to handset, (3) handset talking to speakerphone, and (4) handset talking to handset.

Figure 10 illustrates the effect of transmission loss on the number of calls that will be judged good or better with respect to received volume at the indicated transmission loss when the receiving end is

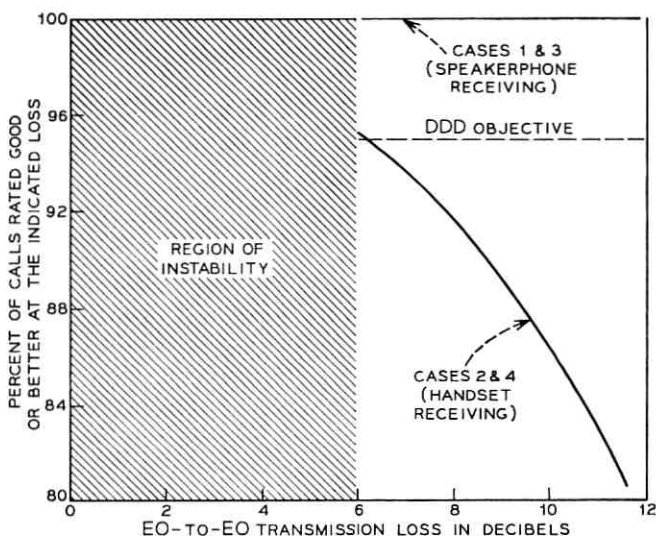


Fig. 10—Received volume performance as a function of the transmission loss.

a handset (Cases 2 and 4 above). When the receiving end is a speakerphone (Cases 1 and 3), as will be the case for a majority (estimated at 95 percent) of the calls, the volume control permits the customer to adjust the received volume to his preferred level. Hence all Case 1 and 3 calls will be judged good or better. The received volume grade of service<sup>23,24</sup> averaged over all calls can be calculated by assuming a fixed transmission loss (as will be the case when all toll offices are bit-stream switches) and weighting the performance of the two classes of calls by their probability of occurrence. For example, with a 6-dB EO-to-EO transmission loss, the grade of service is  $(0.95)(100) + (0.05)(95)$ ; hence 99.75 percent of the calls will be rated good or better. This may be compared with the objective for the DDD network of 95 percent good or better<sup>25</sup> (dotted line on Fig. 10).

#### 4.2.2 Echo Performance

Assuming that a 6-dB EO-to-EO loss is acceptable with respect to received volume, it remains to examine the resultant echo performance. Figure 11 illustrates the echo performance when the EO-to-EO transmission loss is 6 dB. For round-trip delays of 30 to 40 ms, the maximum amount of delay expected on intraregional\* calls, the re-

\* Regional areas for *Picturephone* calling are defined as analogous to those for telephone calling, in that a region "centers" on a high-level switching point called a regional center. However, *Picturephone* calling regions may not necessarily coincide with telephone calling regions.

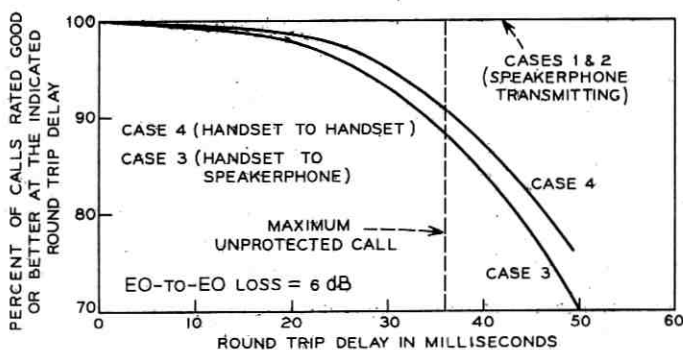


Fig. 11—Echo performance as a function of the round-trip delay.

sultant quality is rated good or better by about 90 percent of the observers for Case 3 and 4 calls. The echo performance for Case 1 and 2 calls will always be acceptable because of the switched loss in the speakerphone. This quality of service, taken over all cases, was judged acceptable and a value of 36 ms was chosen as the maximum value of round-trip delay that could be permitted without echo suppressors. This plan may be compared with the DDD requirement that echo suppression be used on all connections with a round-trip delay greater than 45 ms.<sup>23</sup> The difference results from the lower (fixed) EO-to-EO loss in the *Picturephone* network.

The following rules for the *Picturephone* network resulted:

- (i) Echo suppressors are used on all interregional final trunks interconnecting the highest class offices in different regions.
- (ii) Echo suppressors are also required on all interregional high-usage trunks having circuit lengths greater than:
  - (a) 1800 miles (25 ms\*) between two Class 4 offices and,
  - (b) 1000 miles (15 ms\*) for all other interregional high-usage trunks.

#### 4.2.3 Noise Performance

An important feature of the audio plan for the network with bitstream switching at all toll offices results from the fact that all the loss is inserted at the receiving end of the connection. This results in maximum suppression of noise (in the absence of speech) originating

\* Round-trip delay.

in the trunk facilities. The manner in which this loss is inserted is discussed in Section 4.3.

#### 4.2.4 Summary of Performance

Table V summarizes the expected performance of the network for the four cases under consideration. The column headed noise/volume gives the performance when both noise and volume are considered together. Note that the results are the same as when only volume is considered, indicating that the expected noise level is low enough not to affect the quality. The majority of the calls (those involving speakerphones at both ends of the connection) should have adequate received volume and be reasonably free from echo and noise.

TABLE V—AUDIO PLAN—TRANSMISSION PERFORMANCE

Talker/listener	Assumed % of Con- nections	% of Calls Rated Good or Better		
		Volume	Noise/Volume	Echo
Speakerphone/speakerphone	95%	100%	100%	100%
Speakerphone/handset	5%	95%	95%	100%
Handset/speakerphone		100%	100%	88%*. <sup>†</sup>
Handset/handset		95%	95%	91%*

\* Percent of customers who will be satisfied at the maximum unprotected round-trip delay of 36 ms. Where the round-trip delay exceeds 36 ms, echo suppression will be used. Since these cases represent only a small percentage of the connections and most connections will have a round-trip delay less than 36 ms, the actual percentage of total customers who will be satisfied is greater than that shown.

<sup>†</sup> In this case, the echo occurs through two paths, impedance mismatches and acoustical feedback from speaker to microphone.

#### 4.3 Transmission Loss Plan (Combined Analog and Bitstream Toll Switching)

The previous loss plan and discussion of performance assumed that the toll-connecting trunks (TCT) and intertoll trunks (ITT) are switched through bitstream switches at the toll offices, Class 4 and higher, with analog switching only at the Class 5 office. This will not always be the case, since in the initial years, analog switching of the video as well as the voice at Class 4 offices and stations served directly by analog Class 4 offices will be commonplace. This section describes the ways in which portions of the *Picturephone* network might evolve over the years. Such a description is a necessary prelude to a discussion of the way in which the audio loss plan is implemented.

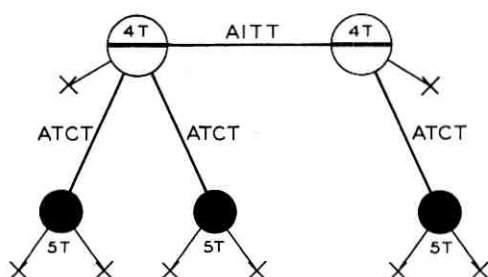


Fig. 12—Audio transmission in the analog-only video network. (Notes: X, *Picturephone* station; —, analog transmission facility; ATCT, analog toll-connecting trunk; AITT, analog intertoll trunk; 4T, 5T, telephone switch at *Picturephone* Class 4, 5 offices.)

#### 4.3.1 Evolution of the Analog/Digital *Picturephone* Network

4.3.1.1 *Audio Transmission in the Analog-Only Video Network.* Figure 12 illustrates possible configurations of the network prior to the introduction of bitstream switching. The 4T and 5T designations represent the telephone switches at the *Picturephone* Class 4 and 5 office respectively which switch the *Picturephone* audio channel and control the switching of the video signals through associated video switches. The trunk designation A in ATCT and AITT signifies that both ends of the trunk terminate in analog switches.

4.3.1.2. *Audio Transmission in the Combined Analog and Digital Picturephone Hierarchy.* When bitstream switching is introduced, additional alternatives for structuring the network become available as shown in Fig. 13. The Class 4 video switch may be a bitstream switch, an analog switch or a combination analog/bitstream switch, as indicated. Intertoll trunks may be digital (DITT) or, when one or both ends terminate in analog Class 4s, analog/digital (A/D ITT) or analog (AITT) respectively. Toll-connecting trunks may be analog (ATCT) or analog/digital (A/D TCT) as indicated.

#### 4.3.2 Audio Loss Plan for Class 4 Analog Switching

4.3.2.1 *Objectives.* The loss plan to be discussed is predicated on providing 6 dB of EO-to-EO loss when all toll switching is via bitstream switches (see Section 4.2). When analog switched trunks are used, their loss design is such that the performance of any combination of analog and digital trunks is consistent with that of digitally switched trunks in terms of echo protection and volume. Furthermore, from an administrative standpoint, it is desirable to have uniform loss in the two direc-

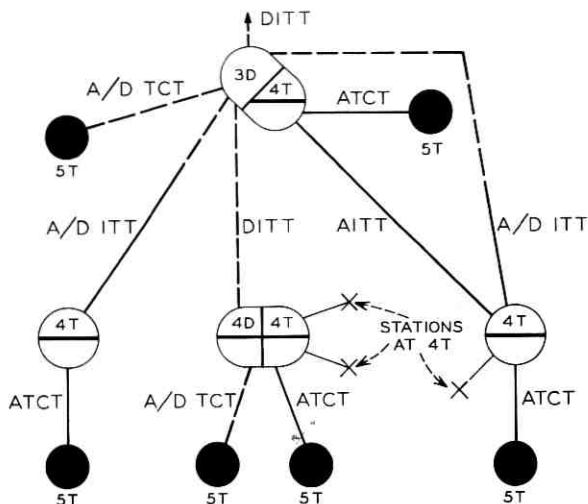


Fig. 13—Audio transmission in the combined analog and digital *Picturephone* hierarchy. (Notes: —, analog transmission facility; - - -, digital transmission facility; ATCT, analog toll-connecting trunk; A/D TCT, toll-connecting trunk, terminating analog at 5T, digital other end; AITT, analog intertoll trunk; A/D ITT, intertoll trunk, terminating analog, one end, digital, other end; DITT, digital intertoll trunk; 3D,4D, Class 3,4 bitstream switching offices; 4T,5T, telephone switch at *Picturephone* Class 4,5 offices.)

tions of transmission and uniform loss pads for all trunk terminations. For maintenance reasons, it is desirable to provide uniform test levels at maintenance points, independent of trunk type (A, A/D or D), and to provide maintenance procedures consistent with DDD practices.

4.3.2.2 *Constraints.* In developing the loss plan, the following constraints for switching and transmission of the audio channel are invoked:

- (i) When the video signal is transmitted digitally, the voice is also transmitted digitally. (Prior to bitstream switching, the audio will use separate facilities.)
- (ii) When the switching office is a pure bitstream switching office, the audio channel is switched through the bitstream switch.
- (iii) When the switching office is a combination office (switches both analog and digital signals), the audio channel is switched in:
  - (a) Digital form when both trunks to be switched together terminate in digital form in the office, e.g., A/D TCT — DITT, A/D TCT — A/D TCT, DITT — DITT.
  - (b) Analog form when either or both of the trunks to be switched together terminate in analog form at the office, e.g.,

ATCT - ATCT, ATCT - AITT, ATCT - A/D TCT,  
ATCT - DITT.

- (c) Analog form when the call terminates to a station directly served by the office.

4.3.2.3 *Loss Plan.* Application of these constraints leads to an overall EO-to-EO loss plan illustrated in Fig. 14 (the loss values are discussed below). Note that this figure is essentially a more detailed version of Fig. 13 with identical connections, and that the connections of Fig. 12 are a subset of those of Fig. 14. Hence, Fig. 14 illustrates the various connections that may occur—and the performance of which must be considered—in the analog/digital *Picturephone* hierarchy. To aid tracing these connections, offices and stations have been identified by a circled reference letter.

#### 4.3.3 *Loss Value*

The actual values of the losses shown are based on the following performance and maintenance objectives:

- (i) For standardization of levels at maintenance test points:
- Assuming that the level at the transmitting side of the Class 5T switch is 0 TLP, the level at the input to the carrier (CXR) or line terminal (TERM) in the 5T office should be -16 TLP.
  - Assuming that the level at the transmitting side of the Class 4 and higher switches is -2 TLP, the level at the input to the carrier (CXR) or line terminal (TERM) in the same office should be -16 TLP.

Note that (a) and (b) above determine the amount of loss in the transmission path from the switch to the CXR or TERM at all offices.

- As noted in Section 4.2, the loss for any A/D TCT - A/D TCT connection is 6 dB.
- Intertoll connections involving ATCTs should have 1 dB of additional loss over that for a pure digital connection for each Class 4T office involved in the connection. Thus, an ATCT - ITT - A/D TCT should have 7 dB of loss and an ATCT - ITT - ATCT should have 8 dB of loss. This additional loss protects against the reflections due to impedance mismatches at the 4T office.
- A station at a 4T is, from the point of view of echo performance, equivalent to a station at a 5T which connects to a 4D via an A/D TCT. Hence, a connection involving a station at a 4T



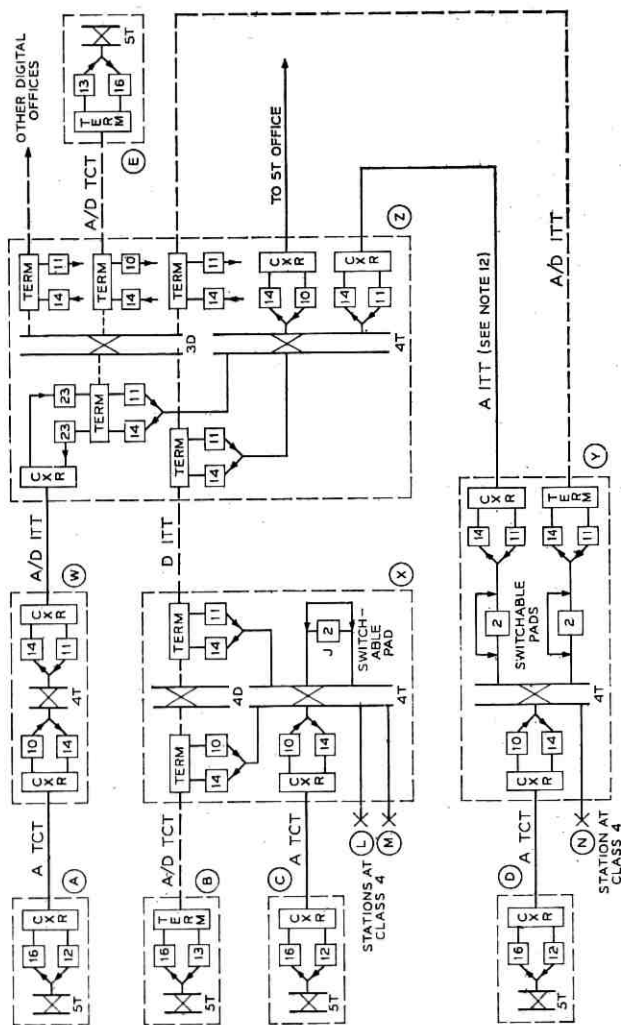


Fig. 14—Audio loss plan. (Notes: 1. \_\_\_\_\_, analog transmission. 2. \_\_\_\_\_, digital transmission. 3. 3D, 4D, denote bitstream switches which switch the audio in digital form multiplexed with video. 4. 4T, 5T, denote audio (POTS) switching networks to which the associated wideband analog and/or bitstream switches are slaved. 5. Stations and 2-dB pads at office Z are not shown. 6. Facility gains: a. 23 dB on each analog trunk, b. 23 dB once on a through-switched digital connection. 7. [16] loss value at trunk termination (includes 4-dB hybrid loss). 8. "CXR" denotes carrier terminals such as those now in use in the DDD network. 9. "TERM" denotes special terminals being developed for the *Picturephone* network to interface switching and transmission facilities. 10. Trunk and line appearances not all shown. 11. J, junctor trunk (two line and two trunk appearances). 12. High-usage trunk—may connect only to TCTs or stations at office Z.

should have the same loss as that indicated in the corresponding row or column under A/D TCT.

- (v) Testing and administrative procedures at the Class 4 office should be uniform for all TCTs.

With these objectives taken together, it is possible to derive the loss to be provided at the transmitting and receiving end of each trunk shown in Fig. 14. This is accomplished in Appendix B. The loss values shown consist of two parts: hybrid loss (4 dB) and pad loss. For example, the 16-dB loss value at the Class 5 office transmit end employs a 12-dB pad in conjunction with the hybrid.

One minor disadvantage resulting from the set of loss values thus derived is that the loss from the Class 4 office to the Class 5 office differs between ATCTs and A/D TCTs. This nonuniformity in administration and testing is a necessary compromise to achieve the specified EO-to-EO losses.

#### 4.3.4 Connection Losses

4.3.4.1 *Connections Using Intertoll Trunks.* The losses for various connections throughout the network are given in Table VI. Note that these losses are consistent with the performance objectives of Section 4.3.3. Included in the table is a listing of typical paths for each type of connection with reference letters keyed to offices and stations of Fig. 14.

4.3.4.2 *Connections Not Using Intertoll Trunks.* With the loss plan for connections using intertoll trunks established and appropriate loss values assigned to trunks, it remains to examine the losses found on connections not involving intertoll trunks. This is done in Table VII, again including representative paths for each type of connection as represented by the offices and stations of Fig. 14. Note that, with the exception of A/D TCT - A/D TCT connections, all involve analog switching at the Class 4 office. Since no ITTs are involved, the round-trip delay is short enough to eliminate echo as a performance criterion; hence, no loss must be added to these connections to guard against reflections. An exception to this is the case of a long A/D TCT used as an inter-regional high-usage or direct trunk; the echo on such a trunk will be controlled by echo suppressors.

Note that the table does not have symmetry across the diagonal as does the table for connections involving intertoll trunks. For A/D TCT-ATCT connections, this asymmetry results from the difference in loss from the Class 4 to the Class 5 office for the two types of TCTs, ATCT and A/D TCT, as noted in Section 4.3.3. Similarly,

TABLE VI—EO-TO-EO AUDIO TRANSMISSION LOSS  
(For calls via intertoll trunks)

Transmission Loss (dB)	
From/To	Station at 4T
ATCT	7
A/D TCT	6
Station At 4T	6

Connection Path (refer to Fig. 14)		
ATCT	A/D TCT	Station at 4T
C-XX*-Z-Y-D	D-Y-Z-X-B	D-Y-Z-X <sup>(2dB)</sup> †X*-L
	B-X-Z-E	B-X-Z-Z <sup>(2dB)</sup> †Y-N
Station at 4T		L-X <sup>(2dB)</sup> †X*-Z-Z <sup>(2dB)</sup> †Y-N

\* The double letter indicates that the audio is switched twice through the 4T network. This is necessary to maintain the correspondence between video and audio switching when slaved video switches are used. One audio switch path corresponds to the bitstream video switch path, the other to the analog video switch path.

† Denotes that a 2-dB pad is switched into the connection when terminating at a station at the 4T.

TABLE VII—EO-TO-EO AUDIO TRANSMISSION LOSS  
(For calls not involving intertoll trunks)

Transmission Loss dB			
From/To	ATCT	A/D TCT	Station at 4T
ATCT	6	7	3
A/D TCT	6	6	5
Station at 4T	3	6	0

Connection Path (refer to Fig. 14)			
	ATCT	A/D TCT	Station at 4T
ATCT	C-X-C	C-XX*-B	C-X-L
A/D TCT		B-X-B	B-X <sub>(pad)</sub> <sup>(2dB)</sup> *X-L
Station at 4T			L-X-M

\* Refer to footnotes of Table VI.

asymmetry exists on calls involving stations at 4Ts connected to A/D TCTs due to the difference in loss in the two directions of the A/D TCT when switched analog at the Class 4 office.

A connection between stations served by the same Class 4T office is classified as an intraoffice call and, therefore, has zero trunk loss assigned.

#### 4.3.5 Junctor Trunks

Connections at a combined analog/digital Class 4 office—e.g., office X of Fig. 14—between (i) an ATCT or a station at the 4T, and (ii) an A/D TCT or a DITT result in two passes of the audio signal through the 4T network. These passes correspond to the video passes through the wideband analog and bitstream switches. Associated with the video codec which interfaces these two video-switching paths is a codec junctor trunk at the 4T network (denoted by "J" in Fig. 14). The purpose of this trunk is to set up the appropriate audio paths through the 4T network and to provide the control for the switching of the video. A switchable pad is provided at the junctor trunk for connections terminating to a station at the 4T (see Appendix B).

## V. REMARKS

The allocation of impairments and the resultant network transmission plan described here for the *Picturephone* network are the result of many system studies to evaluate and compare alternative plans. Studies of system tradeoffs between performance, cost and features were carried on throughout the planning stage with compromises made, where necessary, to produce a viable plan. The plan has now passed through the study stage and the design stage and is being implemented in the field.\* The network transmission plan has a certain uniqueness associated with it. Never before has a network of this magnitude (a nationwide, switched, broadband network with a projected growth up to one million stations in ten years) been planned so thoroughly. It covers the evolution of the network from one station set to over a million sets, from local service in one city to network service nationwide, from today's technology to tomorrow's technology.

It is inevitable that there will be revisions as new ideas are generated, motivated by the continuing drive to expand the utility of the service and the network, to improve the transmission quality of the network and to reduce its cost.

Color *Picturephone* service, higher resolution graphics and full voice-video conferencing are challenges now, but they also will become a reality. Means for more efficiently transmitting such broadband signals in the local area and across the country are being studied for possible inclusion in the transmission plan. Although the task of planning a network is never finished, the plan described here is believed to be a big first step.

## VI. ACKNOWLEDGMENT

Many people have contributed to the formulation of the transmission plan for *Picturephone* service, some of whom are authors of companion papers. The author particularly acknowledges the work of G. W. Aughenbaugh who investigated the effects of bridge taps; Ezra Cohen who carried out the studies that resulted in the allocation of impairments; A. J. Ciesielka and H. R. Mehan, who conducted the studies that resulted in the pre-emphasis plan described herein; J. A. Mines who originated the audio level plan; and J. A. Schick and E. E. Lewis who were instrumental in the development of the plan for the digital portion of the network.

---

\*Commercial *Picturephone* service was initiated in the Golden Triangle section of Pittsburgh, Pennsylvania, on July 1, 1970.

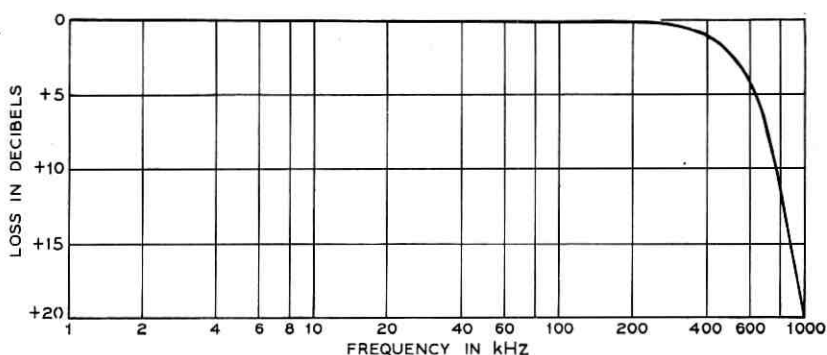


Fig. 15—Station set receiver rolloff—without de-emphasis.

#### APPENDIX A

##### Signal Shaping

Certain of the end-to-end performance objectives (random noise, self crosstalk, worst-disturber crosstalk, impulse noise, power hum and single frequency) as given in a companion paper<sup>2</sup> are referenced to the peak-to-peak signal voltage at the input to a reference station set with a roll-off filter characteristic as shown in Fig. 15. The objectives as allocated in *this* paper are referenced to 0 PTLP, which differs from the reference point used in Ref. 2, both in frequency shaping and in level.

The translation to 0 PTLP is made by first adding the shaping introduced by the de-emphasis network in the 2C station set (see Fig.

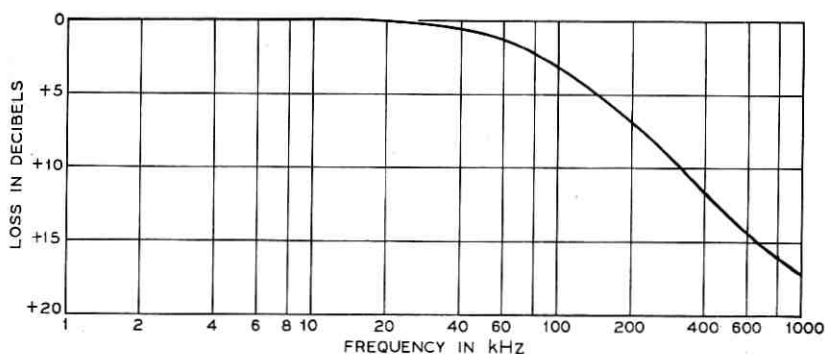


Fig. 16—De-emphasis characteristic.

16) to the weighting curve given in Ref. 2. This results in the weighting curve in Fig. 5 which is used for the allocations in this paper. Next, the interference power, ( $P_i$ ), in dBm into 100Ω at 0 PTLP is calculated from the objectives in Ref. 2, which are given in terms of  $S/N = 20 \log$  (Peak-Peak Signal Voltage/RMS Interference Voltage). It may be shown that the interference power, ( $P_i$ ), in dBm at 0 PTLP is

$$P_i \text{ (dBmO)} = 10 - S/N + 20 \log (\text{Peak-Peak Signal Voltage}).$$

For this calculation, the nominal peak-peak signal voltage at 0 PTLP is assumed to be 0.8 volts.<sup>19</sup> The interference power, in dBm, at 0 PTLP is then equal to

$$P_i \text{ (dBmO)} = +8 - S/N.$$

Pre-emphasis is used in the station-set transmitter to enhance the signal-to-interference ratio in the network. De-emphasis is used in the station-set receiver to restore the signal to its original form. The actual pre-emphasis frequency characteristic chosen tends to equalize the susceptibility of the video signal to various types of noise as the signal is transmitted over analog facilities.

A number of factors enter into the consideration of an optimum pre-emphasis frequency characteristic. They are: the level of each of the interfering signals, the actual *Picturephone* signal level at the point where it combines with the interference, the band roll-off filtering in the station-set receiver and finally, the transmission performance objectives which take into account, subjectively, the eye's relative sensitivity to interference in various portions of the frequency band.

The significant sources of interference are power hum (60 Hz, 120 Hz, 180 Hz), broadcast radio, carrier systems (T1 digital system, subscriber carrier, N carrier), other *Picturephone* channels and impulse noise from central office switching machines and other nearby equipment.

The video signal level is lowest and most susceptible to interference at the input to a baseband equalizer equalizing a maximum length cable section.<sup>12</sup> At this point, the video signal has passed through a loss characteristic similar to that shown in Fig. 17. To equalize this loss characteristic and achieve an insertion loss that is flat with frequency, the equalizer adds 42 dB of high frequency gain relative to that at low frequencies. Since the interference passes through the same equalizer, it also receives enhanced gain at the high frequencies. This makes the video signal susceptible to interference at the high end of the frequency band. Offsetting this is the frequency roll-off in the sta-

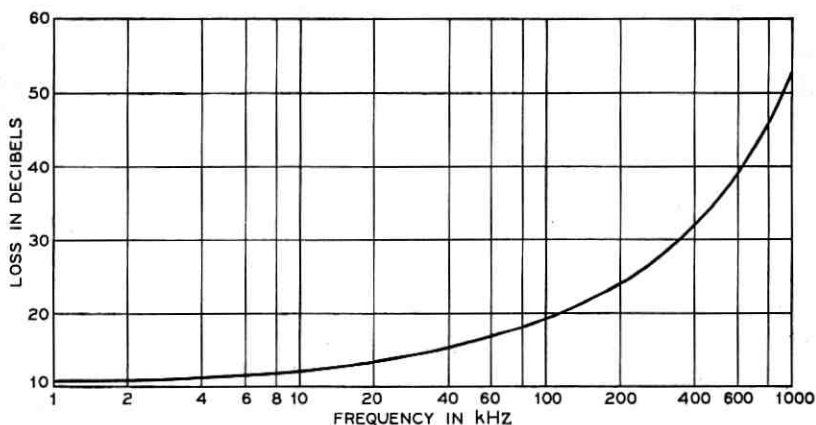


Fig. 17—Cable loss versus frequency—6000 feet of 26-gauge pulp.

tion-set receiver which adds 20 dB of attenuation at the high end of the frequency band relative to low frequencies.

The transmission objectives reflect the viewer's relative sensitivity to the various types of interference. Generally, the viewer is more sensitive to low-frequency interference than high-frequency interference, more to interference which is moving across the screen than to interference which is stationary and more to interference that produces a definite pattern on the screen than to interference that is random, such as noise.

The net effect of weighing all these factors was to establish the need for additional suppression at frequencies in the band above 100 kHz to overcome interference from broadcast radio, T1 carrier and impulse noise; hence, the need for pre-emphasis in this portion of the frequency band. Subjective tests showed that the peaks of a pre-emphasized video signal (high-frequency pre-emphasis only) could be clipped appreciably without noticeably impairing the video signal. The worst clipping

TABLE VIII—INTERFERENCE SUPPRESSION BY PRE/DE-EMPHASIS

Interference	Suppression (dB)
Impulse Noise	6 dB
T1 Carrier	13 dB
Radio Frequencies	12 to 15 dB
Sync Pulse Crosstalk	11 dB
Power Hum	-2 dB



occurs on large, rapid black-to-white or white-to-black transitions when the transmission path is in a state of positive misalignment (a condition where the gain-frequency characteristic deviates from nominal in the gain direction). The impairment appears as a softening of the clipped edge transitions. This softening offsets the harshness which occurs when the transmission path has a positive misalignment that increases smoothly with frequency, such as occurs when the cables used for analog transmission are at minimum temperature.

In summary, pre-emphasis has several advantages. It decreases susceptibility to high-frequency interference without any appreciable loss of low-frequency noise margin\* and without any increase in dynamic range. An indication of the magnitude of the improvement is given in Table VIII. The amount of suppression for impulse noise was determined from subjective tests with actual impulse noise. For the other sources, the amount of suppression is calculated from Fig. 16.

The manner in which pre-emphasis is introduced into the signal path is worth noting. The pre-emphasis shaping is applied only to the video signal; the sync pulse is not pre-emphasized.<sup>19</sup> This has two significant advantages over pre-emphasizing the composite video signal. First, the sync pulse is not exposed to the distortion associated with clipping, leaving the sync pulse undistorted for use in timing recovery and automatic gain control in the codec and station-set receiver. Second, inter-system crosstalk from the sync pulse of the interfering system into the interfered system is suppressed by the de-emphasis network.

The manner in which the clipping is introduced is also worth noting. The signal peaks generated by the pre-emphasis network are partially clipped in the station-set transmitter<sup>19</sup> at a level such that negligible impairment is introduced into the video signal. Clipping in the station set prevents excessive signal levels from entering the cable plant and coupling to other pairs. The equalizers in the network clip at 1.5-volts peak above average referred to 0 PTLP. If the gain characteristic of the transmission channel is at its nominal value, the additional clipping introduced by the equalizers is not detectable on most scenes. If the gain of the channel is greater than nominal, additional clipping occurs in the analog equalizers, with the amount of clipping dependent on the amount of positive gain and the shape of the gain as a function of frequency. Noticeable clipping occurs only in graphic scenes, where

---

\* The video signal level was decreased 2 dB over that which would be possible if de-emphasis were not used.

rapid black/white transitions are common, but only when the channel is in a condition of substantial positive misalignment.

## APPENDIX B

### *Derivation of Audio Loss Values*

In this appendix the audio plan loss values shown in Fig. 14 are derived. The derivation is based on the performance and maintenance objectives of Section 4.3.3.

#### B.1 *Transmit Losses*

To meet objective (ia) (see Section 4.3.3), the loss value between the 5T switch and the carrier input (for ATCT) or line terminal (for A/D TCT) is 16 dB, consisting of the 4-dB hybrid loss and a 12-dB pad. Similarly, to meet objective (ib), the loss value between the 4T switch and carrier or line terminal inputs is 14 dB (hybrid plus 10 dB pad).

#### B.2 *Receive Loss—A/D TCT at Class 5*

The line terminal to line terminal gain is 23 dB, similar to standard carrier systems. In order to provide the overall loss of 6 dB [objective (ii)], with the transmit loss fixed at 16 dB, the receive loss must be 13 dB ( $16 + 13 - 23 = 6$ ), consisting of the hybrid loss plus a 9-dB pad.

#### B.3 *Receive Loss—All Intertoll Trunks*

Consider objective (iv). A connection terminating at a station at the 4T includes a 2-dB switched pad at the 4T\*, thereby providing the proper loss from the 4T via the distant A/D TCT to the distant 5T ( $2 + 14 - 23 + 13 = 6$ ). This leads to a loss of 11 dB on the receive side of the intertoll trunk so that the overall loss remains at 6 dB ( $16 - 23 + 11 + 2 = 6$ ) in the connection from the distant 5T to the station at the 4T. Note that this also provides the required loss for a connection between stations at separate 4Ts ( $2 + 14 - 23 + 11 + 2 = 6$ ).

#### B.4 *Receive Loss—ATCTs*

Consider objective (iii). The 7-dB connection loss going from the A/D TCT via ITT to the ATCT is achieved by providing a 12-dB

\* The 2-dB value is also justified in that it is a standard pad configuration and easily administered.

receive loss at the Class 5 end of the ATCT, taking into consideration the ITT receive loss computed in Section B.3 and the 23-dB gain of the carrier system ( $16 - 23 + 11 + 14 - 23 + 12 = 7$ ). Similarly, it is found that the ATCT receive loss at the Class 4 must be 10 dB ( $16 - 23 + 10 + 14 - 23 + 13 = 7$ ) to meet the objective in the ATCT-ITT-A/D TCT direction. Note that the 8-dB loss for an ATCT-ITT-ATCT connection is also achieved ( $16 - 23 + 10 + 14 - 23 + 11 + 14 - 23 + 12 = 8$ ). Again, the 12-dB and 10-dB losses consist of the 4-dB hybrid loss and 8-dB and 6-dB pads respectively.

#### B.5 Receive Loss—A/D TCT at Class 5

To achieve objective (v), this loss is chosen to be the same as the receive loss of the ATCT, viz., 10 dB. This results in unequal losses in the two directions of transmission when a A/D TCT terminates analog at a Class 4 and leads to the asymmetry for some of the connections in Table VII.

#### B.6 Pad Control

To provide the required loss for connections between a DITT or A/DTCT and a station at the 4T, codec junctor trunks are equipped with switchable 2-dB pads. The pad is "in" when the termination is to a station, otherwise the pad is "out."

To provide a loss equal to that for the DITT connection described above, an AITT or A/D ITT which can connect to a station at the 4T is equipped with a switchable 2-dB pad which is "in" when the connection is to a station at the 4T.

When an ATCT connects directly to a station at a 4T (i.e., no ITT is involved), switchable pads are not used, resulting in lower loss than the preceding connections. Echo is not a problem on this type of a connection since the ATCT is distance limited by video transmission constraints. In contrast to this, the digital facilities may provide service to an end office at a great distance. The greater delay, together with the degrading effect on return loss of two passes through the 4T switch to connect to a station, requires the added echo protection afforded by the 2-dB pad.

#### REFERENCES

1. Dorros, I., "The Picturephone® System: The Network," B.S.T.J., this issue, pp. 221-233.
2. Crater, T. V., "The Picturephone® System: Service Standards," B.S.T.J., this issue, pp. 235-269.
3. Brown, H. E., "Network Planning for Picturephone® Service," Proc. Nat. Elec. Conf., 25 (December 1969), pp. 501-504.

4. Boyd, R. C., and Hathaway, S. D., "Long Distance Video Transmission for *Picturephone*® Service," Proc. Nat. Elec. Conf., 25 (December 1969), pp. 511-515.
5. Millard, J. B., and Maunsell, H. I. G., "The *Picturephone*® System: Digital Encoding of the Video System," B.S.T.J., this issue, pp. 459-479.
6. Mounts, F. W., "A Video Encoding System Using Conditional Picture-Element Replenishment," B.S.T.J., 48, No. 7 (September 1969), pp. 2445-2554.
7. Mounts, F. W., "Conditional Replenishment: A Promising Technique for Video Transmission," Bell Telephone Laboratories RECORD, 48, No. 4 (April 1970), pp. 110-115.
8. Davis, J. H., "A 6.3 Megabit Digital Repeated Line," 1969 International Communications Conference, Boulder, Colorado, Convention Record, pp. 34-9.
9. Broderick, C. W., "The *Picturephone*® System: A Digital Transmission System for TD-2 Radio," B.S.T.J., this issue, pp. 481-499.
10. Gunn, J. F., Ronne, J. S., and Weller, D. C., "The *Picturephone*® System: Mastergroup Digital Transmission on Modern Coaxial Systems," B.S.T.J., this issue, pp. 501-520.
11. Urich, J. F., "The *Picturephone*® System: Switching Plan," B.S.T.J., this issue, pp. 521-531.
12. Brown, J. M., "The *Picturephone*® System: Baseband Video Transmission on Loops and Short-Haul Trunks," B.S.T.J., this issue, pp. 395-425.
13. Byrne, C. J., Karafin, B. J., and Robinson, D. B., Jr., "Systematic Jitter in a Chain of Digital Regenerators," B.S.T.J., 42, No. 6 (November 1963), pp. 2679-2714.
14. Daskalakis, A., "The *Picturephone*® System: Key Telephone Systems," B.S.T.J., this issue, pp. 567-584.
15. Anderson, H. P., "The *Picturephone*® System: The 850A PBX," B.S.T.J., this issue, pp. 585-604.
16. Brown, D. W., Horvath, J. R., Paxton, T. S., "The *Picturephone*® System: No. 101 ESS," B.S.T.J., this issue, pp. 605-620.
17. Wheeler, H. A., "The Interpretation of Amplitude and Phase Distortion in Terms of Paired Echoes," Proc. IRE, June 1939.
18. Fowler, A. D., and Christopher, H. N., "Effective Sum of Multiple Echoes in Television," J. Soc. Motion Picture and Television Engineers, 58 (June 1952).
19. Cagle, W. B., Stokes, R. R., and Wright, B. A., "The *Picturephone*® System: 2C Video Telephone Station Set," B.S.T.J., this issue, pp. 271-312.
20. Burgess, P. N., and Stichel, J. E., "The *Picturephone*® System: Central Office Switching," B.S.T.J., this issue, pp. 533-552.
21. Bunin, B. J., Hirsch, R. B., Olsen, R. E., "The *Picturephone*® System: Cross-talk Considerations in the Transmission Signals on Paired Cable," B.S.T.J., this issue, pp. 427-457.
22. Clemency, W. F., and Goodale, W. D., Jr., "Functional Design of a Voice-Switched Speakerphone," B.S.T.J., 40, No. 3 (May 1961), pp. 649-668.
23. Members of the Technical Staff, Bell Telephone Laboratories, *Transmission System Design*, Fourth Edition, Winston-Salem, N. C.: Western Electric Company, 1970, pp. 45, 61, 62.
24. Lewinski, D. A., "A New Objective for Message Circuit Noise," B.S.T.J., 43, No. 2 (March 1964), pp. 719-740.
25. Coolidge, O. H., and Reier, G. C., "An Appraisal of Received Telephone Speech Volume," B.S.T.J., 38, No. 3 (May 1959), pp. 877-897.

## The Picturephone<sup>®</sup> System:

# Baseband Video Transmission on Loops and Short-Haul Trunks

By J. M. BROWN

(Manuscript received September 25, 1970)

*Local Picturephone<sup>®</sup> service will be provided using present telephone cable facilities. Three cable pairs are required for each Picturephone call, one for audio and two for video transmission. Underground cables will be used extensively and aerial facilities to a limited extent. A cable equalizer will provide shaped gain to compensate for cable loss from 5 Hz to 1 MHz. This article discusses the plan for providing baseband video transmission on paired cable facilities, the cable equalizer design features, the important rules for engineering the video transmission system, and the maintenance features.*

### I. INTRODUCTION

For *Picturephone* service, baseband video signals are being transmitted on customer loops and short-haul interoffice trunks along routes similar to the ones used for present telephone service. It is economically attractive to use ordinary pairs of wires in the existing cables for video transmission. One pair is used for each direction of video transmission in addition to the pair used for the audio signal.

The basic element needed to provide video transmission is a cable equalizer, which provides shaped gain to compensate for cable loss. Present designs which are the subject of this paper are intended primarily for underground 22-, 24-, and 26-gauge paper pulp-insulated conductors. Underground cables are preferred since they are subject to less temperature variation and interference than aerial cables. Future systems now being designed will have temperature regulation and higher transmission levels, permitting the use of longer underground loops and trunks and the use of aerial facilities.

## II. LOCAL VIDEO TRANSMISSION SYSTEM

The typical baseband video transmission system of Fig. 1 shows two *Picturephone* station sets interconnected by means of *Picturephone* loop and trunk facilities and two video switching offices. The audio portion of the *Picturephone* system is not shown nor discussed in this paper. Fixed shaped gain cable equalizers, designated with an "F" in Fig. 1, are used in the outgoing direction of transmission from switching offices to equalize for the carefully controlled office cabling and switching loss. Adjustable shaped gain cable equalizers, designated with an "arrow" in Fig. 1, compensate for a wide range of cable losses. The cable equalizer associated with the station set is mounted within the station set service unit as discussed in Ref. 1.

The video switching office transmission plan is shown in greater detail in Fig. 2. The video loop and trunk pairs are brought into the office through the main distributing frame (MDF) to the incoming and outgoing cable equalizers and then to the wideband distributing frame (WBDF) which is used for switch load balancing purposes. While loops connect to both the horizontal (H) and vertical (V) connections on the MDF in order to provide flexibility, trunks connect to

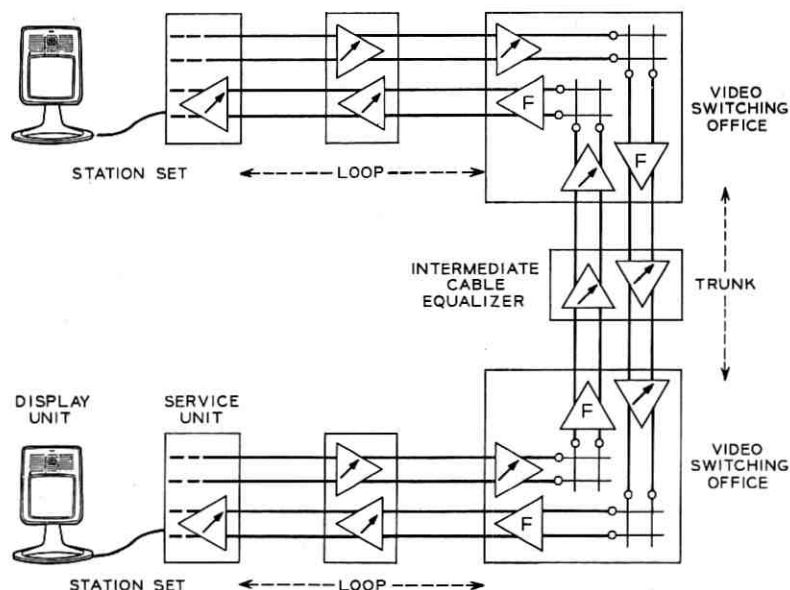


Fig. 1—Local video transmission system.

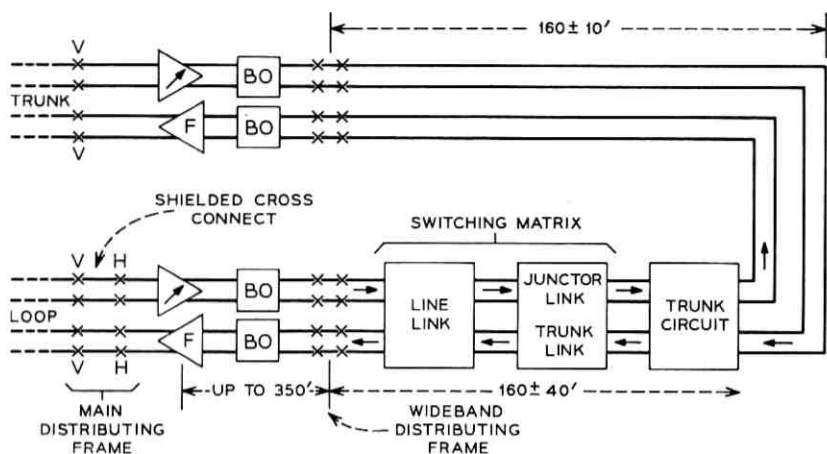


Fig. 2—Video switching office transmission plan.

only the verticals on the MDF since trunk rearrangements are not normally required. The cable length from the central office cable equalizers (COCE) to the WBDF is electrically built out to  $350 \pm 50$  feet of 24-gauge cabling by means of the block labeled (BO). The loop transmission path from the WBDF through the three-stage video switching matrix to the trunk circuit is carefully controlled to  $160 \pm 40$  feet. Likewise the trunk cabling from the WBDF to the trunk circuit is also carefully controlled to  $160 \pm 10$  feet. Consequently, the transmission loss from the incoming loop cable equalizers through the central office to the outgoing loop or trunk cable equalizers is equivalent to the sum of  $(350 + 160 + 160 + 350)$  or 1020 feet of cabling. This central office transmission loss is compensated by the fixed shaped gain outgoing central office cable equalizer designated with an "F". The video switching office is discussed in more detail in Ref. 2.

Other situations, shown in Fig. 3, which require the use of cable equalizers include video transmission through customer located switching equipment, such as Wideband Remote Switches (WBRS),<sup>2</sup> Key Telephone Systems (KTS),<sup>3</sup> or Private Branch Exchanges (PBX)<sup>4</sup>. For video lines from zero to approximately 400 feet of 24-gauge cable between the station set and the KTS, passive station set build out networks (BO) are used instead of cable equalizers in order to build out these lines to a fixed amount of shaped loss. For the direction of transmission from the station set toward the KTS, this loss is post equalized by the KTS cable equalizer on the central office side of the

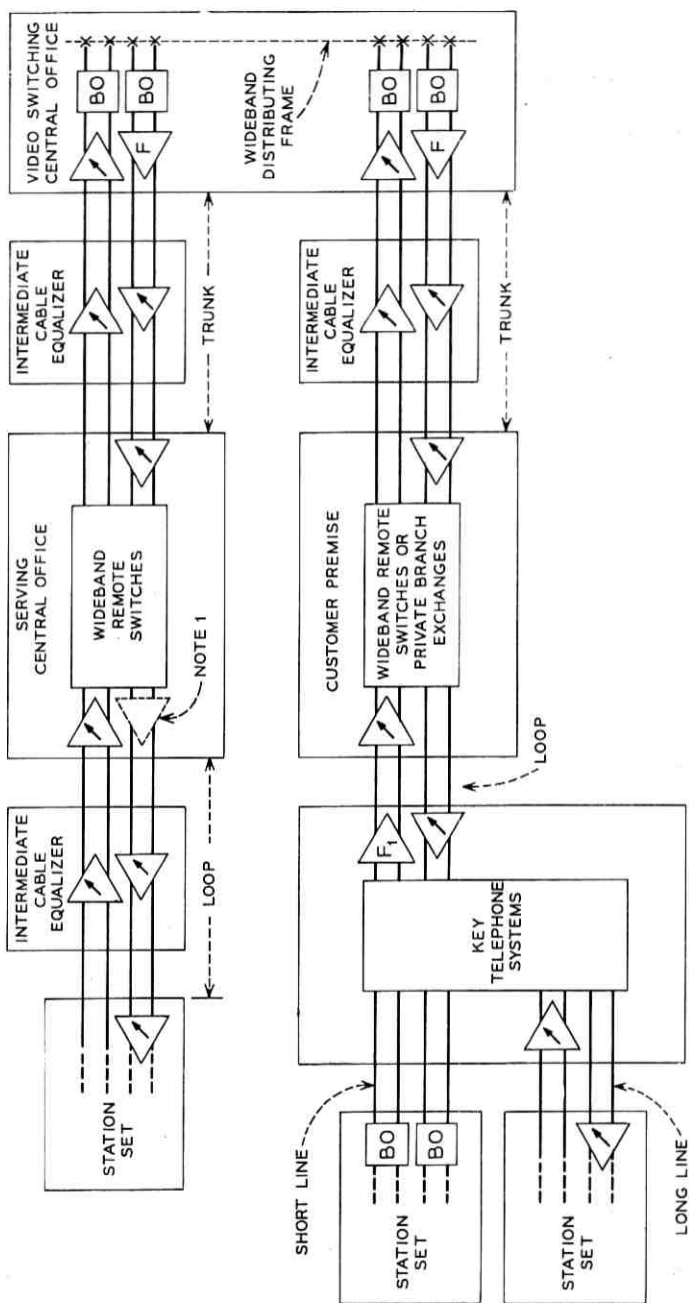


Fig. 3—Video transmission plan. (Note 1—used for supplying simplex power to ICE and for fault location.)



KTS designated  $F_1$ . The KTS cable equalizer on the central office side of the KTS designated with an arrow provides fixed pre-equalization for this built out KTS line loss for transmission from the KTS to the station set as well as adjustable equalization for the preceding loop cable section. For station sets beyond the range of the station set buildout networks, a station set cable equalizer and an additional KTS cable equalizer are required, as shown in Fig. 3.

A cable equalizer is required on all video loops and trunks coming into a PBX. Since PBX switching and transmission loss is negligible, no cable equalizers are required on outgoing loops and trunks.

A WBRs may also be used between the video switching office and the *Picturephone* station set. A WBRs is an extension of the video switching office and under its direct control but remotely located on the customer's premises or in another switching office not equipped for video switching. It is used to reduce video transmission costs by concentrating a number of loops into a lesser number of trunks connected to the controlling video switching office. The video transmission plan for a WBRs is essentially the same as for a PBX.

A customer's serving central office for telephone service may not be the same office used for switching his video calls, particularly in the early years of *Picturephone* service. These serving central offices may contain intermediate cable equalizers (ICE) or WBRs which are used for the video portion of a *Picturephone* call. The facility which interconnects the serving central office with the video switching office is engineered to video trunk standards<sup>5</sup> in a manner similar to that used for baseband video trunks interconnecting two video switching offices. Such planning allows a graceful growth pattern; serving central office video equipment may progress from ICE to WBRs, and then ultimately the office may become a video switching office itself.

Loop transmission facilities are used to interconnect a PBX or WBRs on a customer's premises with a video switching office and are called trunks, although they are engineered<sup>5</sup> as a loop. All other transmission facilities not already specifically discussed (see Fig. 3) are engineered as loop facilities. Details of the engineering of the loop and trunk video transmission system are discussed in Section VI.

### III. CABLE EQUALIZER DESIGN OBJECTIVES

The cable equalizer was designed to compensate primarily for the loss of 22-, 24- and 26-gauge pulp-insulated cable pairs, or mixtures thereof. The maximum cable equalizer gain was limited to 54 dB at

1 MHz. The criterion which was used to determine this maximum allowable equalizer gain was that the near-end crosstalk loss of the cable should be greater than cable equalizer gain by 14 dB for 99.9 percent of the installations. Consequently, the equalized cable section length is limited to approximately 6 kilofeet (kft) of 26-gauge cable and 8 kft of 22-gauge cable. This limit on cable section lengths is discussed more completely in Section 6.3.

The number of cable sections that may be cascaded to form one loop is, in practice, limited to from one to four sections due to the effects of changing cable insertion loss caused by changing ambient temperatures. In order to determine individual cable section design objectives such as tilt and random noise performance<sup>5</sup> from the loop objectives, it was assumed that a maximum of four cable sections comprise a loop. Powering capabilities and maintenance features were also planned on this basis.

The equalization objective was to compensate for the insertion loss of a cable section to within  $\pm 0.1$  dB from essentially dc to 500 kHz, with the limits increasing to  $\pm 0.3$  dB at 1 MHz. The amplitude response at the low frequency end of the band is determined by the loop tilt objective.<sup>5</sup> The tilt objective controls the amount by which a dc signal may change over a given interval of time when passed through an ac coupled facility. This requirement controls the allowed change in brightness of one horizontal scan line caused by the *Picturephone* transmission system itself. For a loop tilt objective of 1.5 percent per 100  $\mu$ s and a four-section loop with five cable equalizers, each cable equalizer must satisfy a tilt requirement of 0.3 percent. We assume that only one RC low frequency coupling network is dominant. This assumption requires an RC time constant of at least 33 ms or a cable equalizer frequency response which is 3 dB down at 4.8 Hz.

The sinusoidal signal-handling objective for the cable equalizers was set at a nominal 0.75 V peak over the 1-MHz frequency band with a 6-dB margin against overload. The 6-dB margin allows for system misalignment and for changing cable insertion loss due to changing ambient temperature. The cable equalizer gain stability objective for a  $-40^{\circ}\text{F}$  to  $+140^{\circ}\text{F}$  temperature environment was set at  $\pm 0.2$  dB at 1 kHz and  $\pm 0.3$  dB at 1 MHz. The cable equalizers were also required to withstand lightning surges which are limited to 600 V by carbon block protectors.

The presence of large longitudinal signals on video pairs, principally 60-Hz signals, may overload the cable equalizer input network or reverse bias the Zener diode used to generate the cable equalizer

powering voltage. The cable equalizer was required to tolerate a 10-mA peak longitudinal current and present a longitudinal impedance of 100 ohms at the input (equivalent to 1-V peak at the input) and 25 ohms at the output of a locally powered cable equalizer. The corresponding maximum open circuit longitudinal pair voltage that can be tolerated is approximately 2-V peak per mile for 26-gauge cable and 1-V peak per mile for 22-gauge cable. For the underground facilities that are contemplated for the early years of *Picturephone* service, it is unlikely that longitudinal signals will exceed this level.

The weighted random noise objective<sup>5</sup> is -65 dBV\* across a 100-ohm termination at the system zero *Picturephone* transmission level point (0 PTLP) for a loop. This noise objective includes cable equalizer noise and random interference from other systems coupled by means of cable crosstalk paths. The noise objective per loop due to only cable equalizers was set at -80 dBV, allowing a 15-dB margin for noise from other sources. Assuming four equalized cable sections per loop, the weighted noise objective per cable equalizer becomes -86 dBV. Combining the weighted noise characteristic<sup>5</sup> with the cable equalizer gain characteristic yields a frequency response with 13 dB of gain which is 3 dB down at 4.8 Hz and 500 kHz. The theoretical noise performance of an ideal noiseless cable equalizer with this indicated weighted frequency characteristic is -114 dBV. Hence, the cable equalizer noise figure objective is 28 dB.

The longitudinal balance (common mode rejection) of the cable equalizer input network<sup>†</sup> should be somewhat better than the longitudinal balance obtained from paired cable facilities so that cable facility performance is limiting. A longitudinal disturbance on one pair in a cable may be coupled to a *Picturephone* pair through the longitudinal-to-longitudinal near-end crosstalk path (with loss  $L_{LL}$ ) and then converted to an unwanted transverse signal because of the finite balance of the input network. Even if the input network balance were infinite, an unwanted transverse signal is produced by coupling through the longitudinal-to-transverse near-end crosstalk path (with loss  $L_{LT}$ ) of the cable itself. Consequently the input network balance ( $BAL_{in}$ ) is given by equation (1); a 10-dB margin has been included to insure that cable balance is controlling at least 95 percent of the time.

$$BAL_{in} = L_{LT} - L_{LL} + 10 \text{ dB.} \quad (1)$$

\* In this paper, dBV is defined as dB with respect to 1 V rms.

† Longitudinal Balance (dB) = Transverse Gain (dB) - Longitudinal Gain (dB).

Equation (1) was used to determine the input network balance objective from 30 kHz to 1 MHz and is shown as part of Fig. 4.

For the frequency range from 600 Hz to 30 kHz, the single frequency interference (SFI) requirement<sup>5</sup> of -65 dBV at system 0 PTLP was used to determine the input network balance objective. If the maximum longitudinal signal (LS) present at the input of the cable equalizer is -10 dBV\* and the maximum cable equalizer gain  $G$  is 10 dB for the 600 kHz to 30 kHz frequency range, then the input network balanced should be 65 dB as given by equation (2).

$$\text{BAL}_{in} = \text{LS} - \text{SFI} + G \quad (2)$$

For the low frequency range from 60 Hz to 600 Hz, the input network balance should be such that a 60-Hz longitudinal signal at -3 dBV, the maximum tolerable longitudinal signal, should result in a transverse signal no greater than the -38 dBV power hum requirement<sup>5</sup> for a loop facility. Assuming a cable equalizer low frequency gain of 10 dB, the input network balance at 60 Hz should be 45 dB. The composite balance objective for the input network is shown in Fig. 4.

The balance objective for the balanced output network of the cable equalizer is important to prevent coupling of a video signal from one video pair to another or to another type of transmission system. Any unbalance of the output network allows transverse signals to be converted to longitudinal signals and then coupled to other pairs via the longitudinal-to-transverse near-end crosstalk path with loss  $L_{LT}$ . Even if the output network balance were infinite, a transverse signal on one pair would cause a transverse signal on another pair via the transverse-to-transverse near-end crosstalk path with loss  $L_{TT}$ . Consequently the balance of the output network should be greater than  $L_{TT} - L_{LT}$  plus a 10-dB margin, resulting in a 20-dB balance objective for the output network for most of the frequency band of interest.

Output network balance objectives more stringent than the 20-dB balance discussed above are required for remotely powered intermediate cable equalizers which are floating with respect to earth ground. The feedback path around a floating cable equalizer, shown in Fig. 5, is one of two possible feedback paths which involves the

---

\* Most of the longitudinal signal tolerance of the cable equalizer should be reserved for 60-Hz type pickup, and harmonics thereof. Hence it is reasonable to require that all other types of longitudinal interference on cable pairs are at least 7 dB down from the 1 V peak longitudinal signal tolerance level, or -10 dBV or less.

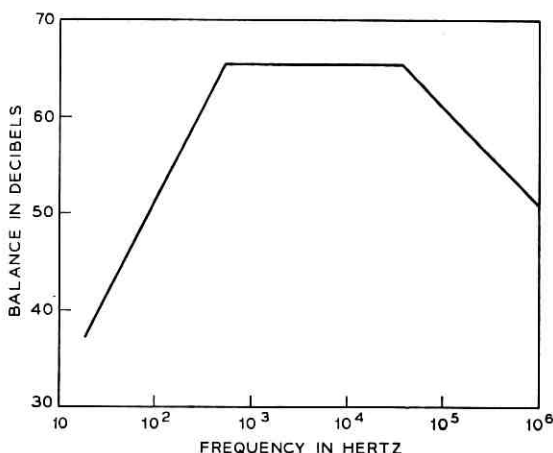


Fig. 4—Input network balance objective.

balance of input and output networks and the longitudinal cable impedance which is referenced to earth ground. Analysis of these two feedback paths yields essentially the same balance requirements for the output networks; hence, only the situation shown in Fig. 5 will be discussed here.

Because of the finite balance of the output network, a transverse signal at the cable equalizer output is converted to a longitudinal signal with loss  $BAL_{out}$  and is coupled to the input of the cable equalizer through the longitudinal cable impedance at the input and output of the cable equalizers. This longitudinal signal is then converted to a transverse signal by the unbalance of the input network with an additional loss  $BAL_{in}$  and then amplified by up to 54 dB, the maximum cable equalizer gain at 1 MHz. Consequently,  $BAL_{out} + BAL_{in}$  should be significantly greater than 54 dB at 1 MHz. If the input network balance is 50 dB at 1 MHz, then the output network balance should be much greater than  $54 - 50$  or 4 dB. The output network balance requirement was chosen as 40 dB over the 1-MHz band of interest primarily because this amount of balance is readily achievable, as indicated in the next section, and provides ample attenuation to suppress the unwanted feedback.

#### IV. CABLE EQUALIZER DESIGN

The cable equalizer consists basically of an input network, a static equalizer, and an output network as shown in Fig. 6.

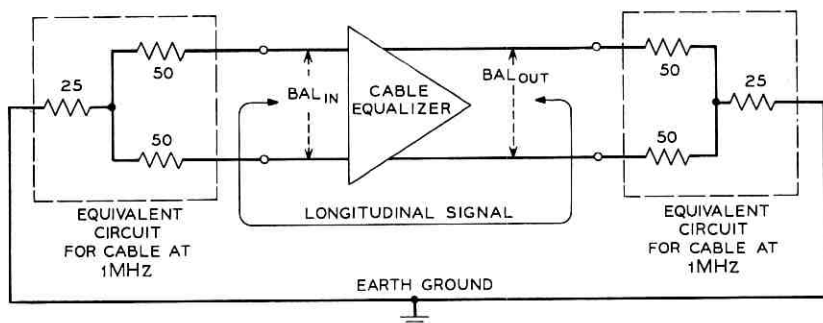


Fig. 5—Feedback path around a simplex powered cable equalizer with finite input and output balance.

The input network is a high impedance differential amplifier which converts the incoming balanced signal to the unbalanced mode for equalization. This network is capacitively coupled to the cable and provides a fixed center tapped 100-ohm resistive termination to the cable. This 100-ohm termination was chosen because the characteristic impedance of pulp cable pairs is approximately 100 ohms above 100 kHz. It is this coupling network which primarily determines the 4.8-Hz 3-dB corner frequency of the cable equalizer. A flat gain

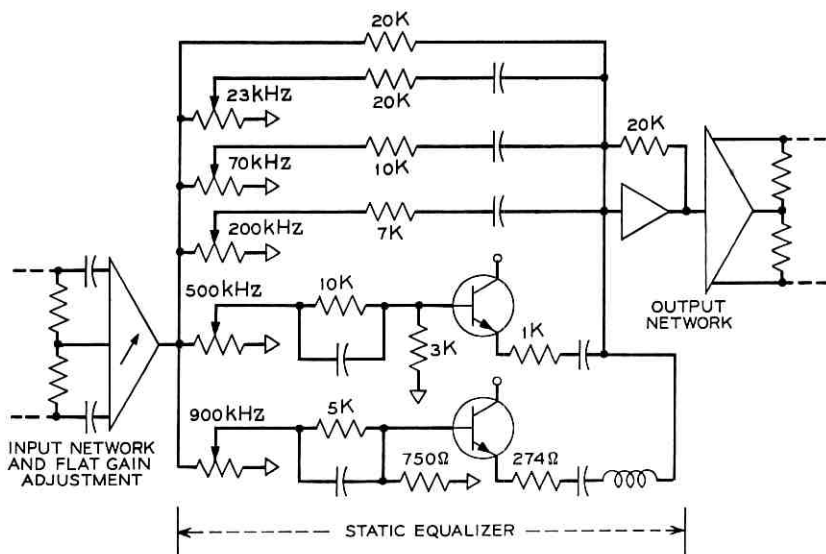


Fig. 6—Cable equalizer.

adjustment is provided with a  $-6$  dB to  $+10$  dB gain range. The worst-case longitudinal balance (common mode rejection) of the input network essentially meets the balance objectives as outlined in the previous section and shown in Fig. 7.

The output network is a dc coupled, unity gain, differential power amplifier which converts the unbalanced static equalizer output signal to the balanced mode for transmission on the cable. It provides a fixed center tapped 100-ohm resistive termination to the cable. It requires 60 mA of dc bias current to produce a signal handling capability of 1.5 volts peak above average into a 100-ohm load. The typical longitudinal balance of the output network exceeds the 40-dB balance requirement from dc to 1 MHz as shown in Fig. 8.

The static equalizer is basically a shaped gain amplifier that compensates for the high frequency loss in the transmission line. The desired gain shape is achieved by combining the currents from six parallel paths by means of an operational amplifier. The magnitude of the current through five of the paths is controlled by five potentiometers, giving a high degree of flexibility in compensating for a wide variety of cable loss shapes. Each potentiometer is identified with its respective adjustment frequency, and each potentiometer drives a network whose critical frequencies occur in the vicinity of its adjustment frequency. These critical frequencies were chosen as 23 kHz, 70 kHz, 200 kHz, 500 kHz, and 900 kHz. The critical frequencies increase and the impedance levels decrease for successive paths,

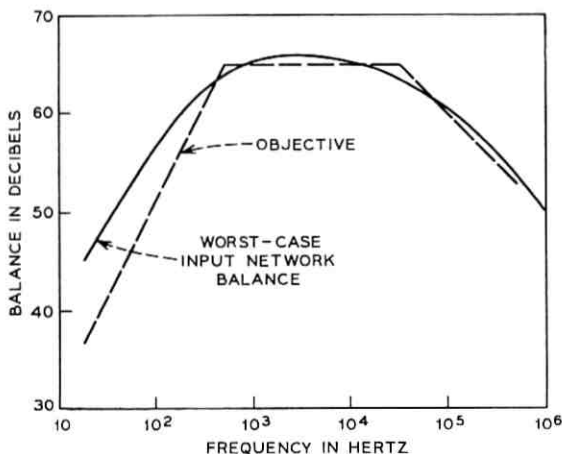


Fig. 7—Worst-case input network balance characteristic.

thereby providing greater gain at higher frequencies. Greater gain slopes are achieved at the higher frequencies by providing a cascade of RC networks.

In designing the six parallel paths that form the heart of the static equalizer, it is important to guarantee that this overall adjustable network is a minimum-phase network for any expected potentiometer setting in order to provide the proper phase equalization. A sufficient condition is that no two parallel paths shall have a phase difference of  $180^\circ$  or greater in their output currents at any frequency. This sufficient condition is easily satisfied by the adjustable network.

The adjustable static equalizer has a gain range of from 0 to 23 dB at 1 MHz. Not shown in Fig. 6 are two additional cascaded stages of fixed shaped gain with critical frequencies in the 200-kHz to 1-MHz frequency range which may be added by screw switches, one for medium length loops and both for long loops. They provide additional gain of 7 dB and 14 dB at 1 MHz respectively, and lesser amounts at lower frequencies. The use of these screw options reduces the range of shaped gain required for the adjustable static equalizer and helps in obtaining the steep (on the order of 18 dB/octave) gain versus frequency slope required for long cable sections at high frequencies. It would be difficult to achieve an 18 dB/octave slope entirely from

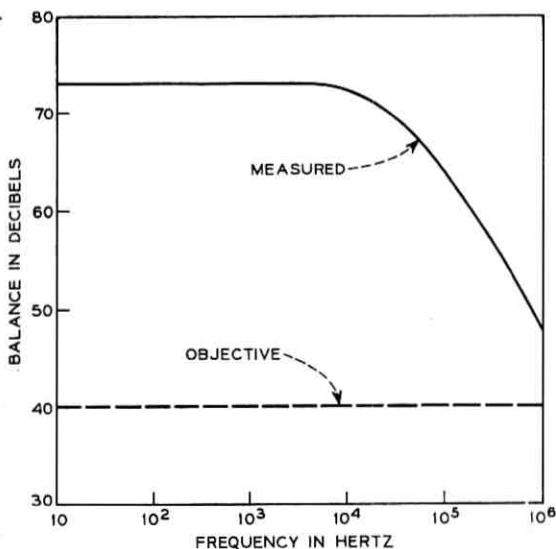


Fig. 8—Typical measured output network balance characteristic.



the adjustable network and still meet the sufficient condition for minimum phase.

The pole and zero locations for the individual parallel paths within the adjustable static equalizer and the two fixed shaped gain networks were first determined roughly by laboratory experimentation so that the equalizer would perform properly for any cable length up to the maximum length for both 22- and 26-gauge pulp cable. The performance criteria were gain range at the five test frequencies and closeness of fit between test frequencies. After the static equalizer configuration was determined experimentally, the optimum adjustment frequencies and network values were determined by computer simulation. By this process it was found that five potentiometers were required within the adjustable static equalizer in order to achieve the equalization design objective for a section of cable of  $\pm 0.1$  dB up to 500 kHz and increasing to  $\pm 0.3$  dB at 1 MHz.

Thus, by means of five potentiometers and two screw options within the static equalizer and the 10 dB of adjustable flat gain associated with the input network, gains from 0 to 10 dB at low frequencies and 0 dB to 54 dB at 1 MHz are achieved, sufficient to equalize up to about 6 kft of 26-gauge pulp cable and 8 kft of 22-gauge cable.

Also included in the static equalizer is a 3-pole low-pass Butterworth filter which attenuates out-of-band frequencies but has a negligible effect on the amplitude response up to 1 MHz. This filter helps prevent system oscillations in the vicinity of 1.3 to 2.0 MHz due to cable near-end crosstalk, as discussed in Section 6.3.

Alignment is accomplished by transmitting six frequencies, one at a time, over the section of cable to be equalized. A low frequency tone at 1 kHz is used to adjust the flat gain of the input network and five tones (from 23 kHz to 900 kHz) are used to adjust the five potentiometers within the static equalizer to obtain an equalized cable section frequency response flat to 1 MHz and approximately 3 dB down at 4.8 Hz and 1.3 MHz. The five static equalizer adjustments are not completely independent, however, and it is necessary to cycle through the alignment frequencies about six times to obtain an alignment accuracy of better than 0.05 dB at the alignment frequencies. This accuracy of 0.05 dB is readily achievable and was chosen so that the misalignment caused by potentiometer maladjustment is small compared to the total facility misalignment error caused by such things as cable equalizer gain drift due to ambient temperature changes, the absolute accuracy of the test instruments used during alignment, and the effects of temperature changes on cable insertion loss.

The amplitude and phase response of an equalized 6-kft section of 26-gauge pulp cable is shown in Fig. 9. The asterisks in Fig. 9 indicate the alignment frequencies. The error in the amplitude response is less than 0.1 dB up to 600 kHz and less than 0.2 dB up to 1 MHz. The phase distortion of Fig. 9 is due principally to the sharp cutoff above 1 MHz of an equalized cable section. The time delay distortion is about 75 ns at 600 kHz and 150 ns at 800 kHz. As an example of the significance of this amount of time delay distortion, a cascade of two to three such sections of cable, i.e., 15 kft of 26-gauge, uses the entire loop echo rating allocation as discussed in Section 6.2 and Ref. 5. A later version of the cable equalizer will have less time-delay distortion.

Other characteristics of the cable equalizer include a measured weighted noise of  $-90$  dBV corresponding to a noise figure of 24 dB which exceeds the objective discussed in Section III by 4 dB. The overload performance of a typical cable equalizer is plotted in Fig. 10 for peak sinusoidal signal amplitude versus the signal-to-harmonic distortion ratio (S/H.D.). For a 0.75-V peak sinusoid which is the nominal transmission level for the cable equalizer and is 6 dB below the clipping level, the S/H.D. is 55 dB.

The cable equalizer was designed for a  $-40^{\circ}\text{F}$  to  $+140^{\circ}\text{F}$  temperature environment. Since the cable equalizer does not contain temperature regulation, the temperature environment for cable itself is restricted to the considerably narrower limits for underground or

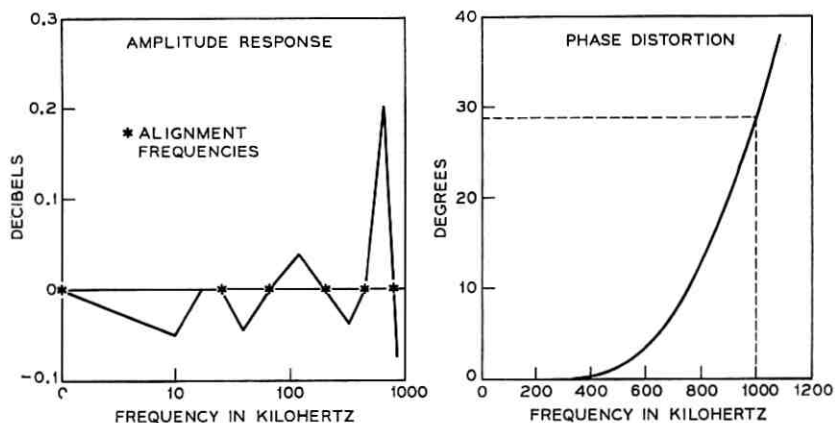


Fig. 9—Typical measured frequency response-equalized 6-kft 26-gauge pulp cable.

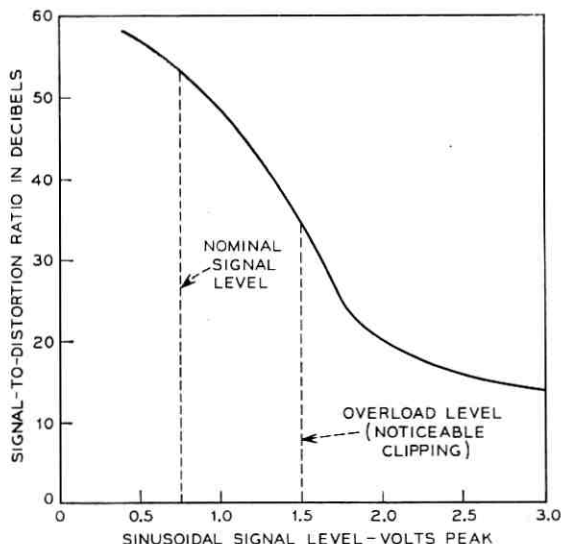


Fig. 10—Typical measured cable equalizer overload performance.

buried cables of about 30°F to 80°F. The effect of changing temperature on the gain of a typical cable equalizer at 1 kHz and 900 kHz is shown in Fig. 11. The effect of temperature change at other frequencies between 1 kHz and 900 kHz tends to lie within these two curves. Gain changes are less than the objective of 0.2 dB at 1 kHz and 0.3 at 900 kHz over this temperature range. For a restricted temperature range of 30°F to 120°F, gain changes are within 0.1 dB at all frequencies.

Preliminary studies of reliability indicate a mean-time-to-failure of 40 years for a one-way cable equalizer. For a loop shown in Fig. 1 which includes a two-way central office cable equalizer, a two-way intermediate cable equalizer, a one-way station set cable equalizer, and associated power supplies, the mean-time-to-loop-failure is approximately seven years.

A one-way cable equalizer which mounts in a PBX, a WBRS, a central office, and in a manhole consists of two printed wiring cards 6" wide by 7" long that are sandwiched together so that all components are toward the interior of the sandwich and the printed wiring is exterior as shown in Fig. 12. The resulting packages mount on 1½" centers. Each one-way cable equalizer consists of 40 transistors, 13 diodes, 40 varistors used primarily for lightning protection, 120 re-

sistors, 40 capacitors, two inductors, and six potentiometers. Cable equalizers for Key Systems are similar electrically and physically (7.5" wide, 5" long, and 1.5" thick). For station sets, the cable equalizer is also similar electrically, but all components are mounted on one printed wiring card 5.5" wide and 10.5" long. A typical central office bay with cable equalizers is shown in Fig. 13.

#### V. POWERING AND LIGHTNING PROTECTION

Each one-way cable equalizer requires approximately  $\frac{1}{8}$  ampere at  $-24$  volts or approximately 3 watts. It may be locally powered or, for the case of ICEs, it may also be simplex powered from the central office or a customer located WBRs. ICEs in manhole apparatus cases may be simplex powered only; ICEs between a PBX and a KTS or station set may be locally powered only. A simplex powered loop is shown in Fig. 14.

A  $+130$ -volt battery potential is applied through a current regulator (CR) to the center tap of the incoming central office loop or trunk cable equalizer input network and then to the cable pair (this cable equalizer is powered from the local regulated  $-24$  V supply). Powering current travels over the transmission pair in the direction opposite that of video transmission. At the next equalizer location, the powering current is routed through the center tap of the output network, used to power that cable equalizer, and then applied to the next cable section through the center tap of the input network.

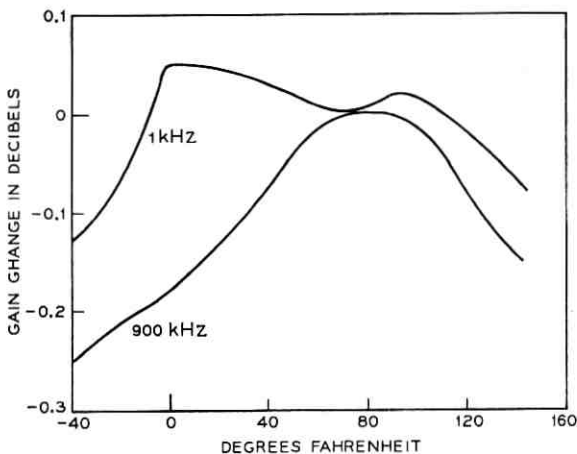


Fig. 11—Typical measured cable equalizer gain variation with temperature.

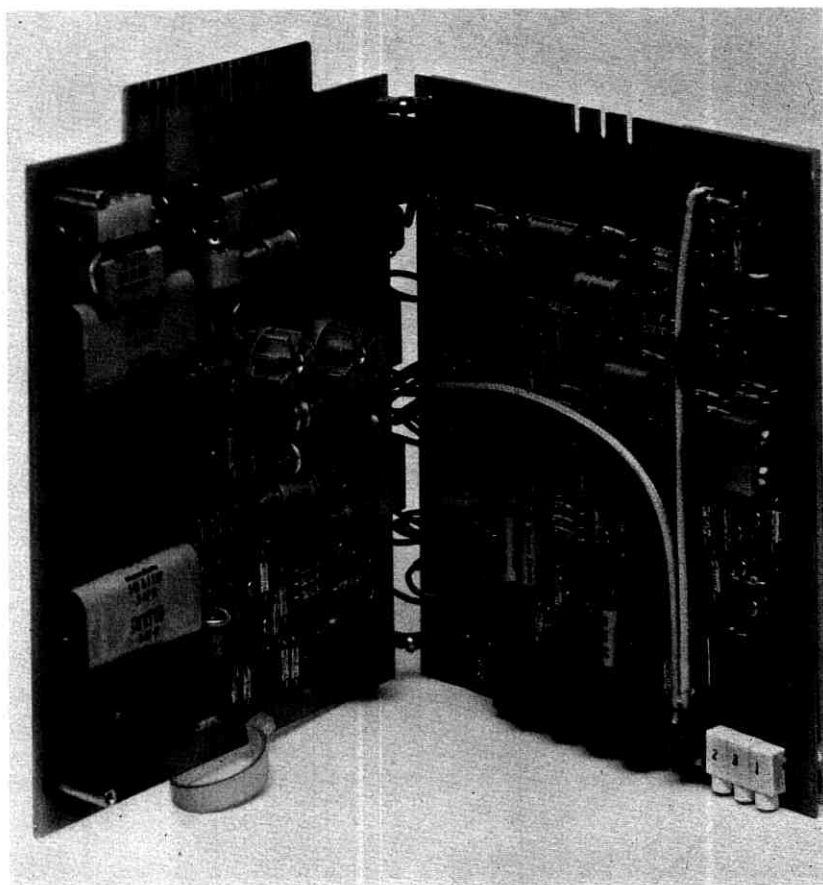


Fig. 12—Cable equalizer—open view.

At the last intermediate cable equalizer, the powering current is looped back from one video pair to the other, used to power the remaining loop cable equalizers, returned to the central office where it powers the outgoing central office equalizer, and then returned to ground,  $-48$  V, or  $-130$  V, depending on the number of ICEs and the length and gauge of the loop facility. Up to three intermediate cable equalizers may be powered in series.

Table I tabulates the simplex powering limitations in terms of cable gauge and length from the central office to the last intermediate cable equalizer. The total loop length consists of the simplex powered section plus the last section which is not simplex powered. The final

loop configuration must simultaneously satisfy three separate conditions: simplex powering limitations shown in Table I, cable equalizer spacing as dictated by engineering considerations discussed in Section 6.3, and maximum loop or trunk length limitations as dictated by the allowed transmission impairments as discussed in Section 6.2.

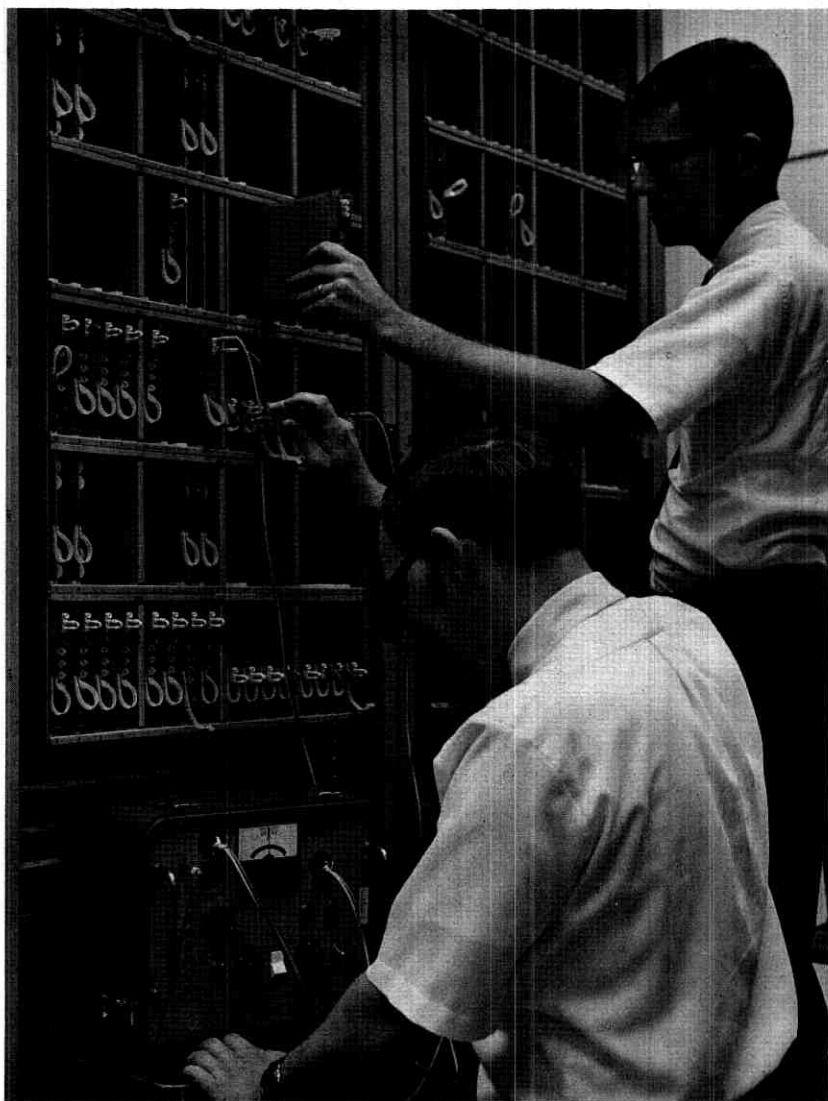


Fig. 13—Central office cable equalizer bay.

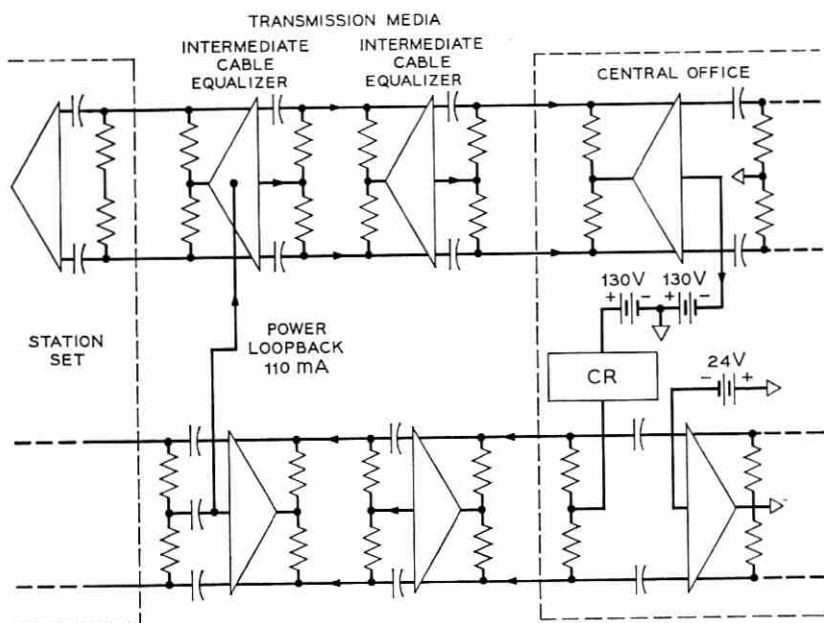


Fig. 14—Simplex powering.

Cable equalizer powering and lightning protection features are shown in Fig. 15 for a simplex powered cable equalizer. A similar arrangement is used for local powering from  $-24$  V. A Zener diode connected between the center tapped 100-ohm input and output terminating resistors is used to generate the 14.5 V required to power the cable equalizer circuitry. A total of 20 varistor diodes at the input and 20 at the output are used to provide low voltage lightning protection at the cable equalizer input and output. Twenty diodes are required in order to prevent unwanted conduction by dc bias voltages and ac video signals. Standard carbon blocks or gas tubes are used for high voltage lightning protection.

## VI. ENGINEERING THE TRANSMISSION SYSTEM

*Picturephone* service will be provided using present telephone cable facilities for transmitting baseband video signals on customer loops and short-haul interoffice trunks. These facilities have been installed over the years primarily for an economical and flexible voice-frequency communications network. Planning, engineering, installation, and ad-

ministrative considerations different from those for voice grade service are required for *Picturephone* service because of the more stringent transmission requirements.

### 6.1 General Cable Facility Considerations

The cables recommended for baseband video transmission are buried or underground 22-, 24-, and 26-gauge pulp-insulated 0.083  $\mu\text{F}/\text{mile}$  cable, which are subject to less temperature variation and radio frequency interference than aerial cables. While not a preferred type, 19-gauge pulp-insulated 0.083  $\mu\text{F}/\text{mile}$  cable may also be used. Cables of the unit type construction are preferred because of superior cross-talk properties. Cable sections placed prior to 1940 are to be avoided, if possible, since higher attenuation and impedance irregularities may result from paraffin filled splices, plugging compound, and lower insulation resistance at splice joints. If necessary 19-, 22-, 24-, and 26-gauge 0.083  $\mu\text{F}/\text{mile}$  plastic insulated cables (PIC) may also be used.

In those few cases requiring cable facilities so long that they cannot be served on the above indicated cable pairs, 16-gauge cable used for commercial TV transmission may also be used to provide *Picturephone* service.

Mixtures of the above cable gauges with the same capacitance characteristic are permitted in a video telephone facility. Cable equalizers are not required for isolation purposes at the gauge change interface.

All load coils and build-out capacitors must be removed from cable pairs used for video telephone service. Any bridge tap in excess of 100 feet must be removed; in addition, the sum of all bridged taps on a loop or analog trunk should not exceed 200 feet.

Tests at Bell Laboratories have indicated that video cable pairs without at least 15 mA of dc current flowing must have soldered or

TABLE I—SIMPLEX-POWERING LIMITATION

Gauge	Maximum Distance in kft to Last Simplex Powered ICE		
	One ICE (+130V GND)	Two ICES ( $\pm 130\text{V}$ )	Three ICES ( $\pm 130\text{V}$ )
26	6	11	9
24	10	18	14
22	16	29	23
19	32	58	46
16	64	117	92



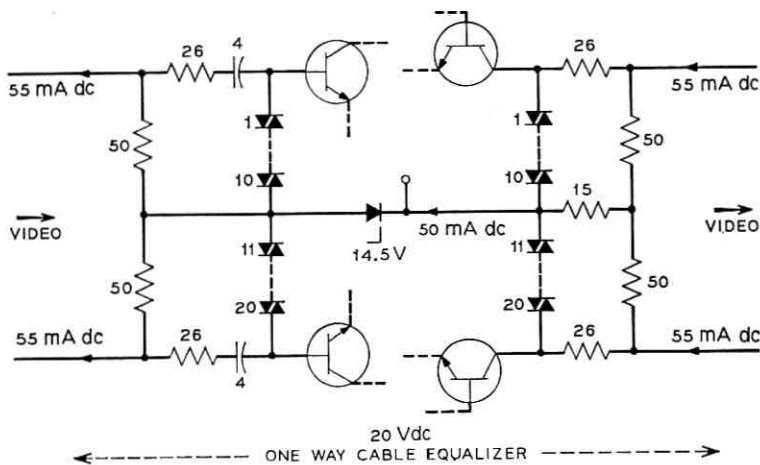


Fig. 15—Powering and lightning protection.

crimped splices in order to guarantee low-resistance joints. Hand twisted splices tend to form high-resistance joints in time without dc current or without high ac potentials such as 20-Hz ringing signals. For video telephone pairs, soldered joints or crimped splices are required in most instances, i.e., for those facilities without simplex-powered intermediate cable equalizers (ICEs) and the last section of those simplex-powered ICEs.

Wideband data systems and carrier systems such as T1, T2, and N cannot be transmitted in the same unit, complement, or splicing group with video telephone pairs. When unit integrity is maintained, *Picturephone* pairs may be in units adjacent to such facilities only if the direction of transmission is the same. Regardless of the direction of transmission, *Picturephone* pairs may be in units nonadjacent to such facilities. Where unit integrity is not maintained, separate sheaths are required to separate *Picturephone* circuits from the above systems.

Thermal noise and impulse noise are expected to be within the acceptable levels as outlined in Ref. 5. Radio frequency interference (RFI) from commercial radio broadcast stations is also expected to be within acceptable levels for single frequency interference<sup>5</sup> on underground or buried facilities providing that all branch pairs entering a sheath containing *Picturephone* circuits are likewise underground or buried. Aerial pairs for video circuits are permitted for distances up to 500 feet only if there is a continuous, unbroken sheath on the aerial portion and if the aerial portion is approximately one mile or more

from a radio station broadcasting antenna. In addition, branch pairs entering a sheath containing video telephone circuits must satisfy the aerial cable rule to a distance of 6 kft from the branch point. These aerial cable restrictions will be relaxed for future systems having higher transmission levels.

Unbalanced private line circuits widely used for dc telegraph, burglar alarms, and clock synchronization may act like antennas and bring RFI into cables with video telephone circuits. Consequently these unbalanced circuits should not be in the same unit with *Picturephone* circuits, assuming unit integrity. If unit integrity is not maintained, the unbalanced circuits should not be in the same sheath with *Picturephone* circuits.

Also because of RFI, no more than 200 feet of unshielded inside wiring, not contained in metal pipe or building raceway, may be used in a reinforced concrete or structural steel building which itself provides shielding. In buildings not providing shielding, such as wooden structures, all inside wiring cable should be shielded.

### 6.2 Loop and Trunk Lengths

Baseband video loop and trunk lengths are presently limited by two dominant factors, phase distortion discussed previously and the effects of temperature on cable insertion loss as shown in Fig. 16 for typical 22- and 26-gauge pulp cable pairs. To evaluate the effect of these transmission impairments, a technique known as "echo rating"<sup>6</sup> is employed.

The echo rating for the effects of temperature change on cable insertion loss can be expressed by equation (3), where echo rating (ER) is in dB, the magnitude of temperature change (T) in degrees Fahrenheit, cable length (L) in kilofeet\*, and the subscript on (L) representing cable gauge.

$$ER_T = -79 + 20 \log_{10} \left[ T \cdot L_{26} + T \cdot \frac{L_{24}}{1.4} + T \cdot \frac{L_{22}}{4} + T \cdot \frac{L_{19}}{6} + T \cdot \frac{L_{16}}{8} \right]. \quad (3)$$

Similarly, equation (4) shows the echo rating result due to phase distortion as a function of cable gauge and length.

$$ER_\phi = -45 + 20 \log_{10} (1/6) \left[ L_{26} + \frac{L_{24}}{1.2} + \frac{L_{22}}{1.5} + \frac{L_{19}}{2.25} + \frac{L_{16}}{5} \right]. \quad (4)$$

\* Equations (3) and (4) are for pulp insulated pairs for 19- through 26-gauge cables and plastic insulation for 16-gauge cables.

The combined effect of these two impairments is such that the individually computed echo ratings add together on a power basis as shown by equation (5).

$$ER_{TOTAL} = ER_{\phi} + 10 \log_{10} \left[ 1 + \log_{10}^{-1} \left( \frac{ER_T - ER_{\phi}}{10} \right) \right]. \quad (5)$$

As an example, assume a total facility length of 15 kft of 26-gauge pulp cable installed at mean temperature and hence subjected to a 25°F temperature change. Computations show that

$$ER_T = -27.5 \text{ dB},$$

$$ER_{\phi} = -37 \text{ dB},$$

$$ER_{TOTAL} = -27 \text{ dB}.$$

The end-to-end *Picturephone* system echo rating objective<sup>6</sup> is -26 dB with -27 dB allocated to baseband transmission on loops and trunks and -33 dB allocated to all other system impairments such as switching, station sets, and codecs. Consequently, the 15 kft of 26-gauge cable installed at mean temperature uses the entire transmission echo rating allocation of -27 dB.

Since the effects of phase distortion and changing cable insertion loss due to changing temperature on echo rating are the same for loops and trunks, loop and trunk echo rating impairments add together

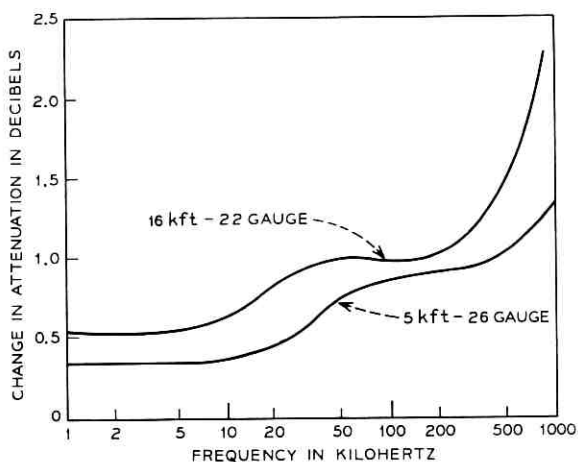


Fig. 16—Typical measured effects of temperature on cable attenuation for a 25°F temperature change.

on a voltage (correlated) basis. The transmission impairment allocation for echo rating is divided so that approximately one-third is allocated to each of the two loops and to the sum of the baseband trunks in an entire end-to-end *Picturephone* connection. Based on an echo rating<sup>5</sup> of -27 dB, each loop and the sum of all trunks are designed to meet a -37 dB echo rating. The resulting loop and trunk half-connection\* lengths are tabulated in Tables II and III as a function

TABLE II—DESIGN LENGTH OF LONGEST LOOP\*

Gauge	Length in kft <sup>†</sup> For Given Temperature Variation From Time of Alignment <sup>‡</sup>			
	50°F (No temp. realignment 1 visit)	25°F (Recommended; mean temp. 2 visits)	12.5°F (Seasonal 4 visits/yr)	2.5°F (No temp. variation 1 visit)
26	2.7 (11) <sup>§</sup>	5.1 (6)	8.8 (3)	15
24	3.7 (8)	6.9 (4)	12 (2)	18
22	9.7 (3)	16 (2)	21 (1)	24
19	14 (2)	24 (1)	31	35
16	20 (1)	37	58	78

\* For present cable equalizer designs only.

<sup>†</sup> Lengths must be reduced by 10 percent per splice in cable sections placed prior to 1940.

<sup>‡</sup> Alignment procedure based on 50°F maximum temperature variation assumed for underground cable.

<sup>§</sup> Number in parentheses is the percentage reduction in length required per 100 feet of aerial cable, up to a maximum of 500 feet.

of gauge and temperature variation. For example, to meet a loop echo rating of -37 dB, loops are limited to 5.1 kft of 26-gauge pulp cable or 16 kft of 22-gauge cable, assuming mean temperature alignment, i.e., a 25°F temperature change on underground facilities. A trunk half-connection is limited to 10 kft of 22-gauge cable, assuming quarterly alignment to minimize temperature effects, i.e., 12.5°F temperature change on underground facilities. These allowed lengths will be increased significantly by the use of a later version of the cable equalizer with less phase distortion and with temperature regulation, i.e., from 5.1 to 15 kft for 26-gauge loops and from 10 kft to 100 kft for 22-gauge trunk half connections.

\* Each end-to-end *Picturephone* connection is allowed two loops and two trunk half-connections. Each trunk half-connection has an echo rating of -43 dB.

TABLE III—DESIGN LENGTH OF LONGEST TRUNK HALF-CONNECTION\*

Gauge	Length in kft <sup>†</sup> For Given Temperature Variation From Time of Alignment <sup>‡</sup>			
	50°F (No temp. realignment 1 visit)	25°F (Mean temp. 2 visits)	12.5°F (Recommended; seasonal 4 visits/yr)	2.5°F (No temp. variation 1 visit)
26	1.3 (22) <sup>§</sup>	2.5 (12)	4.4 (6)	7.5
24	1.8 (17)	3.5 (8)	5.8 (4)	9.1
22	4.8 (6)	7.9 (3)	10 (2)	12
19	7.2 (4)	12 (2)	15 (1)	18
16	10 (3)	19 (1)	29	39

\* For present cable equalizer designs only.

<sup>†</sup> Lengths must be reduced by 20 percent per splice in cable sections placed prior to 1940. Lengths are reduced 30 percent if baseband on TD-2 is used as one trunk facility.

<sup>‡</sup> Alignment procedure based on 50°F maximum temperature variation assumed for underground cable.

<sup>§</sup> Number in parentheses is the percentage reduction in length required per 100 feet of aerial cable, up to a maximum of 500 feet.

### 6.3 Cable Equalizer Spacing

Cable equalizer spacing, as opposed to total loop length discussed above, is limited by the crosstalk properties of the cable. When cable equalizers for both directions of transmission are installed at the same physical location, there is some possibility that the output of one can feed into the input of the other via cable near-end crosstalk (NEXT) paths and cause "singing." This is, of course, most likely to occur at the higher frequencies where equalizer gain must be high and crosstalk loss is low. This crosstalk situation is shown in Fig. 17 for two loops connected to the same video switching office by cable pairs within one cable sheath. The arrows indicate near-end crosstalk paths coupling the output of one cable equalizer to the input of another. The magnitude of the crosstalk problem is a function of crosstalk loss and cable equalizer gain and, hence, cable equalizer spacings.

As shown in Fig. 17, two loops, each with one ICE, have four possible NEXT paths, any one of which may cause "singing." For the maximum length loops with three ICEs, eight such paths exist. If the system objective is a 14-dB margin against oscillation 99 percent of the time at 1 MHz, then each of the eight paths should have this 14-dB margin approximately  $(1 - 0.01/8)$  or 99.875 percent of the time. An approximation is made here that one and only one of the

eight possible crosstalk paths dominates for the worst case of only a 14-dB margin against oscillation. The exact analysis of this situation, which is discussed in Ref. 7 and considers the sum of all eight paths taken together, yields essentially the same result.

Assuming a log normal NEXT distribution, this 99.875 percent represents the  $3\sigma$  point on the distribution. For single unit operation, the NEXT crosstalk distribution has a mean of 74 dB and a standard deviation of 8.8 dB. Consequently, the 99.875 percent point on the NEXT distribution is  $74 - 27$  dB or 47 dB of loss. Subtracting 14 dB for margin against oscillation gives a maximum equalizer gain of 33 dB at 1 MHz. Similar considerations for adjacent unit operation give a maximum equalizer gain of approximately 54 dB at 1 MHz. Consequently, the maximum cable equalizer spacings may be from 3.6 to 5.8 kft for 26-gauge cable pairs or from 4.9 to 8.0 kft for 22-gauge cable, depending on which cable pairs are used, i.e., whether the pairs for the opposite direction of video transmission are in the same unit or adjacent units. Special circumstances, like high impulse noise levels coupled through NEXT paths from audio pairs to video pairs, may dictate shorter spacings.

A tabulation of the maximum equalizer spacing for opposite directions of video transmission in the same unit and separate units is shown in Table IV for various cable gauges. Since cable NEXT loss is limiting for same unit and adjacent unit operation, one would

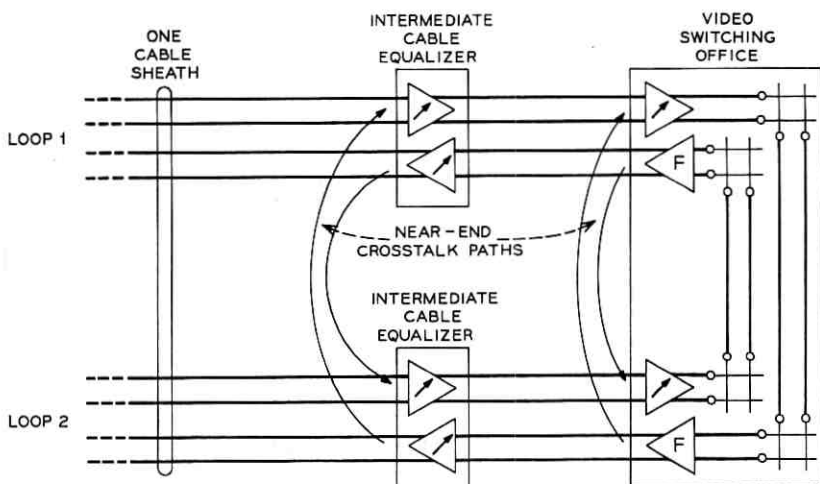


Fig. 17—Effect of cable near-end crosstalk on cable equalizer spacing.

TABLE IV—MAXIMUM LOOP EQUALIZER SPACINGS\*

Opposite Directions of Video Transmission in:		
Gauge	Same unit (kft)	Separate unit (kft)
26	3.6	5.8
24	4.1	6.7
22	4.9	8.0
19	7.3	12
16	—	27

\* If steam pipes should parallel a route, reduce spacings by 7 percent; if layer-type cable, reduce by 15 percent.

normally expect that if nonadjacent units or separate sheaths were used, a greater equalizer spacing would be possible. However, available cable equalizer gain as discussed in Section IV limits equalizer spacings to the same numbers derived on the basis of adjacent unit NEXT for 19-, 22-, 24- and 26-gauge pulp cables. For 16-gauge cable, which consists of individually shielded pairs, cable equalizer spacing is limited by the maximum equalizer gain capability and not by cross-talk considerations.

#### 6.4 Pair Selection and Installation of Cable Equalizers

For the first step in providing video pairs for *Picturephone* service, it is recommended that entire cable units or complements be chosen, using assignment and cable records, in accordance with the video transmission engineering rules discussed in Section 6.1. Because of the complexity involved in selecting and conditioning video pairs, it is not economically feasible to select just one pair at a time as the demand arises, but rather entire complements in anticipation of *Picturephone* demand. Reserving pair complements for *Picturephone* service does not mean that these pairs can be used only for video, but that the temporary use of these reserved pairs for other systems must be compatible with the engineering rules. The choice of a cable route to provide *Picturephone* service may not be the shortest route; the economics may be such that the total cost of the shortest route, including cable plus conditioning may exceed the total cost of a longer route which requires less conditioning.

The selected complements must then be "conditioned" for video transmission. Conditioning includes, for example, cable rearrangements necessary to remove incompatible systems, such as T or N carrier,

from the chosen complements and the removal of all load coils, build out capacitors, and excessively long bridged taps from pairs selected for video transmission.

It is necessary to verify the pair acceptability by performing measurements on at least a representative sample of pairs within a complement. The standard tests used for telephone service are performed first and include dc tests for insulation resistance and foreign EMF. Additionally, low frequency longitudinal and transverse noise tests are made. These noise tests are important to insure that the levels of 60-Hz pickup and harmonics thereof do not exceed the allowable levels as discussed in Section III and Ref. 5.

The last step in determining the pair acceptability is to determine whether unrecorded bridged-taps, load coils, and other sources of impairments are present. The presence of such impairments is most easily determined by equalizing the selected cable pairs by means of a cable equalizer packaged within a portable test set. Gross transmission impairments result in the inability to align a cable section at the six alignment frequencies. The presence of a load coil which essentially inhibits transmission above 4 kHz is one example of a gross impairment. Less severe impairments may permit the equalization of a cable section at the six frequencies but still result in unacceptable performance because of ripples in the amplitude response between alignment frequencies. Bridged-taps, build-out capacitors, and parafin splices may cause these ripples in the amplitude response. The presence of such ripples is determined by making a swept frequency measurement between the highest alignment frequencies of 500 kHz and 900 kHz.

If cable pairs chosen for video transmission do not pass the pair suitability tests outlined above, either alternate pairs are chosen or the source of impairment must be located and corrected. To locate a source of ripple impairment, a commercially available pulse echo test set is used to measure the time delay between the main transmitted pulse and the returned echo and hence the distance to the impairment. Because of the loss of cable pairs at frequencies up to 1 MHz, pulse echo tests may have to be made from both ends of the cable pairs.

After choosing the pairs for video transmission, conditioning them, and verifying their acceptability, the cable equalizers are then installed and aligned. Prior to placing the equalizers in their respective housings along the video pairs, the various screw switch settings and plug-in networks are chosen and set. Then the potentiometers are



adjusted with the cable equalizers in place and working. The test set shown in Fig. 13 is used to align the cable equalizers. The recommended method of alignment requires a craftsman with a test set simultaneously at each cable equalizer location along a video pair. An order wire allows voice communication among the various locations.

#### VII. MAINTENANCE FEATURES

Prior to the completion of a *Picturephone* call, a continuity test is performed by the video switching office on video loops.<sup>2,8</sup> A 12-kHz signal is transmitted from the switching office and is returned to the office via a loopback within the station set, PBX, or KTS. This loopback is removed by an off-hook condition at the station set and by the detection of a sync signal transmitted from the switching office to the station set, PBX, or KTS.

The quality of loop equalization may be checked using the loopback feature. Broadband test signals transmitted from the test center<sup>8</sup> at a switching office are returned to the office via the loopback, thereby allowing the quality of video transmission to be determined for both directions of transmission in tandem. Such a test cannot isolate troubles to one of the two video pairs or to a particular section of cable.

A fault locating network (FL1 and FL2 of Fig. 18) contained in the central office and intermediate cable equalizers allows the selective interrogation from the central office of a particular cable equalizer in order to determine if catastrophic cable equalizer failure has occurred.

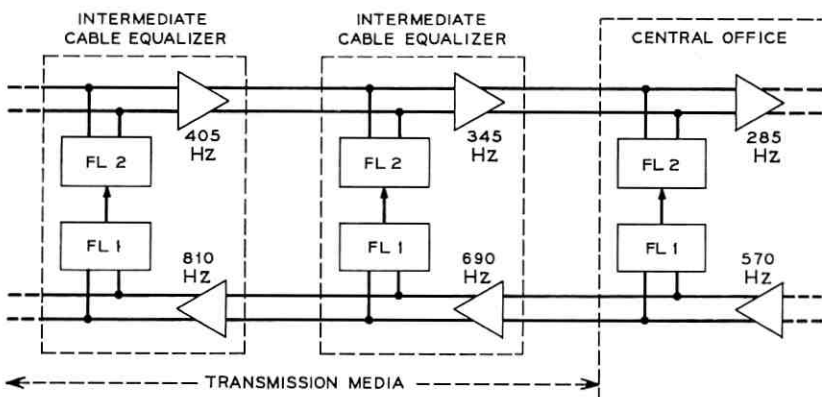


Fig. 18—Fault location.

The fault locating network in the outgoing cable equalizers contains a filter tuned to one of four audio frequencies in the range from 500 Hz to 1000 Hz, which allows the selective interrogation of a particular cable equalizer. A Schmitt trigger is actuated if the filter output is of sufficient amplitude. The complementary incoming cable equalizer contains a bistable multivibrator which is triggered by every other pulse obtained from the differentiated output of the Schmitt trigger. The multivibrator output, a square wave of frequency one half the interrogating frequency, is then transmitted back to the central office on the incoming video pair. Detection of this half frequency signal at the central office indicates that a particular section of the loop has no gross impairment at low frequency. This test cannot be used to detect a degradation of the high frequency amplitude response.

If the dc current flowing in the video pairs for powering intermediate cable equalizers is interrupted, the fault-locating network will be inoperative. But it is possible to localize the failure to a particular section by removing the central office cable equalizers and applying a 48-volt supply as shown in Fig. 19. This causes dc current to flow in the direction opposite to that of the loop powering current. The reverse voltage across the two video pairs at an equalizer causes a diode to conduct current from one video pair to the other through a resistor. The amount of loop current flowing gives an indication of the

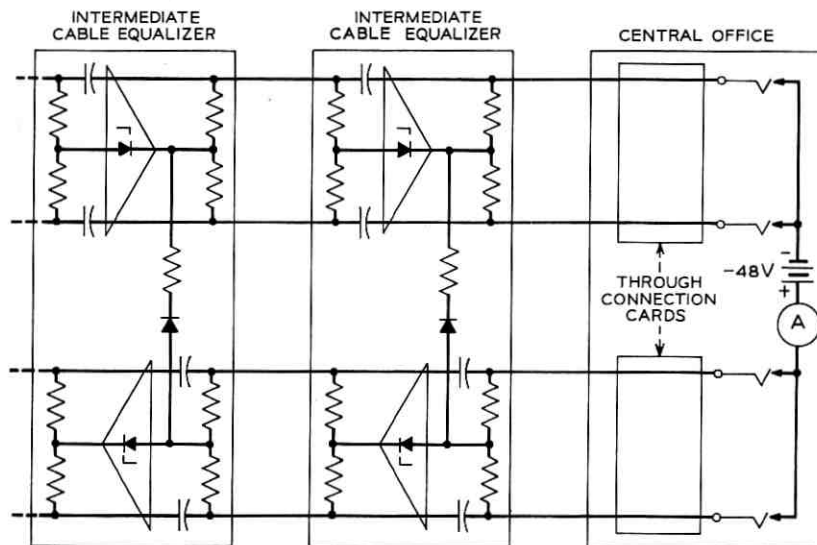


Fig. 19—Arrangement for locating open circuits in the video path.

number of bridging resistors and hence the number of "good" sections by counting toward the station set from the central office. This test will provide no useful information if only one conductor of the video pair is open since this dc test current can still flow, causing additional resistors to be bridged across the video pairs which indicate "good" sections. In addition, this test for locating open circuits may not be used if ICEs are locally powered.

#### VIII. SUMMARY

A means for providing baseband video transmission on customer loops and on short haul trunks using present telephone cable facilities has been described. A cable equalizer which provides adjustable shaped gain up to a maximum of 54 dB at 1 MHz is required approximately every mile. Special procedures will be required to condition the telephone pairs for video transmission.

#### IX. ACKNOWLEDGMENTS

Numerous people have contributed to the design of the baseband video transmission system described in this paper. In particular, the author wishes to acknowledge the electrical design contributions of S. J. Brolin, L. C. Sanders, H. C. Bond, and R. J. Welborn, assisted by J. H. Beck, R. R. DeSimone, and T. A. Sickman; the physical design contributions of F. H. King, W. C. Jurgens, and R. R. Fornilli; the system computer simulation contributions of F. L. Schwartz, F. M. Stumpf, and Mrs. A. F. Rogers; the system engineering contributions of G. W. Aughenbaugh; and the work of R. S. Farbanish and G. E. Harrington on pair acceptability testing.

#### REFERENCES

1. Cagle, W. B., Stokes, R. R., and Wright, B. A., "The Picturephone® System: 2C Video Telephone Station Set," B.S.T.J. this issue, pp. 271-312.
2. Burgess, P. N., and Stickel, J. E., "The Picturephone® System: Central Office Switching," B.S.T.J., this issue, pp. 533-552.
3. Daskalakis, A., "The Picturephone® System: Key Telephone Systems," B.S.T.J., this issue, pp. 567-584.
4. Anderson, H. P., "The Picturephone® System: 850A PBX," B.S.T.J., this issue, pp. 585-604.
5. Brown, H. E., "The Picturephone® System: Transmission Plan," B.S.T.J., this issue, pp. 351-394.
6. Crater, T. V., "The Picturephone® System: Service Standards," B.S.T.J., this issue, pp. 235-269.
7. Bunin, B. J., Hirsch, R. B., and Olsen, R. E., "The Picturephone® System: Crosstalk Considerations in the Transmission of Analog Signals on Paired Cable," B.S.T.J., this issue, pp. 427-457.
8. Doughtery, H. J., Peterson, E. B., and Schachtman, M. G., "The Picturephone® System: Maintenance Plan," B.S.T.J., this issue, pp. 621-644.



## The *Picturephone*<sup>®</sup> System:

# Crosstalk Considerations in the Transmission of Analog Signals on Paired Cable

By B. J. BUNIN, R. B. HIRSCH, and R. E. OLSEN

(Manuscript received October 2, 1970)

*Crosstalk is an important consideration in the transmission of analog baseband Picturephone<sup>®</sup> signals on paired cable. Crosstalk due to the worst disturbing Picturephone signal can cause a distinctive visible interference pattern. Crosstalk interference from other Picturephone and wide-band systems, such as T1 and T2 digital lines, contributes to random noise. In addition, feedback via crosstalk coupling may cause spurious oscillation of cable equalizers. Methods are determined to control these impairments by placing restrictions on equalizer spacing and cable pair assignment.*

### I. INTRODUCTION

Exchange cable pairs will provide the medium for transmission of the analog baseband *Picturephone* signal over subscriber loops and most local trunks.<sup>1</sup> The unwanted coupling of energy, or crosstalk, between pairs within a cable places important limitations on the engineering of these equalized transmission lines.

#### 1.1 *Types of Impairments*

Three types of impairments in the *Picturephone* signal which can occur in the presence of crosstalk are considered in this article. They are listed in Table I along with the source of the impairment. Near-end and far-end crosstalk coupling paths are shown in Fig. 1.<sup>2</sup>

##### 1.1.1 *Worst-Disturber Interference*

The first impairment is caused by interference from the single most prominent *Picturephone* interfering signal among all those present, and is called worst-disturber interference. The signal from the worst

disturber, when passed through a crosstalk coupling path, will appear differentiated and superimposed on the desired signal. Since the two signals are not synchronized, the interference will drift across the screen when the desired signal is displayed. For example, if the sync pulses are the most detrimental part of an interfering *Picturephone* signal, the interference will appear as two vertical lines, one black and one white, drifting across the screen.

### 1.1.2 Random Noise Interference

Even if no signal is visible as a worst disturber, each of the interfering *Picturephone* signals will contribute unwanted energy to the desired signal. This energy must be considered as a contribution to the overall system random noise, which is the second impairment whose effects are studied. The interfering signals do not necessarily have to be of the *Picturephone* type, but could be of any wideband system. In this article we consider crosstalk coupling of noise from *Picturephone*, T1 line, and T2 line signals. Interference from the digital systems is studied on trunk facilities only, since carrier systems are not usually present on subscriber loop cable.

Additional sources of interference, such as  $N$  carrier systems exist, but are not considered in this article. Furthermore, the effect of interference on other systems, caused by the coupling of energy from *Picturephone* signals, and subsequent limitations on this energy are not discussed here.

### 1.1.3 Equalizer Singing

Finally, we investigate a third impairment due to crosstalk, referred to as singing. Singing, or spurious oscillation, can occur around the

TABLE I—IMPAIRMENTS DUE TO CROSSTALK

Impairment	Source of Impairment
(i) Worst disturber	Interference from oppositely-directed <i>Picturephone</i> signals via NEXT*
(ii) Random noise	Interference from <i>Picturephone</i> signals via NEXT and FEXT* Interference from T1 and T2 signals via NEXT and FEXT
(iii) Equalizer singing	Feedback around two oppositely directed equalizers via the two NEXT paths coupling them

\* NEXT and FEXT stand for near-end and far-end crosstalk, respectively.

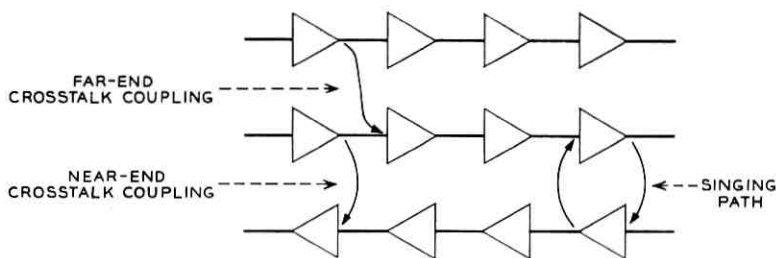


Fig. 1—Crosstalk and singing paths.

path formed by two oppositely-directed *Picturephone* equalizers and the two near-end crosstalk paths coupling the output of one equalizer to the input of the other, as shown in Fig. 1.

### 1.2 Measures to Control the Effects of Crosstalk

The effects of near-end crosstalk can be mitigated by reducing equalizer spacing and thus the required equalizer gain. The signal-to-interference ratio is then increased by the amount the equalizer gain is reduced. If the required spacing becomes too small to be economical, additional near-end crosstalk loss can be realized by segregating the opposite directions of transmission into separate cable units. Further reductions in interference may be achieved by limiting the number of interfering systems.

Equal-level far-end crosstalk is independent of equalizer spacing. This interference can be reduced, however, by limiting the overall line length, and by placing restrictions on the number of interfering systems, and on their placement within the cable sheath.

## II. RESULTS AND CONCLUSIONS

We present methods of analyzing the effects of crosstalk which can be used in writing engineering rules. The worst-disturber impairment is analyzed in Section IV by finding the probability that the minimum pair-to-pair crosstalk coupling loss is less than the requirement. Random noise interference is computed in Section V by passing the interfering spectra through several shaping filters and then integrating the output to find the total effective interfering power. The probability of singing, or of being close enough to singing to affect echo rating, is computed by a Monte Carlo technique in Section VI.

As an example of the use of these methods, they are applied to find the allowable equalizer spacings using the requirements and assump-

TABLE II—LIMITS ON EQUALIZER SPACING

Type and Source of Impairment	Coupling Path	Allowable Loop Equalizer Spacing (kft) (26-gauge pulp cable)		Allowable Trunk Equalizer Spacing (kft)* (22-gauge pulp cable)	
		Within-Unit	Adjacent-Unit	Within-Unit	Adjacent-Unit
(i) Worst disturber	NEXT	4.6	>6.0	>6.0	>6.0
(ii) Random noise	(a) Picture-phone signals	NEXT	>6.0	3.6	>6.0
		FEXT†	>6.0	6.0	>6.0
	(b) T1 signals	NEXT		0	4.7
	FEXT†		0.3	>6.0	
(c) T2 signals	NEXT			2.7	>6.0
	FEXT†			4.3	>6.0
(iii) Singing	NEXT	3.6	5.8	4.4	>6.0

\* This applies to the trunk equalizer now being developed.

† Interference coupled through the FEXT path limits overall length but is independent of equalizer spacing. The spacings are listed for consistency and are based on allowable overall length divided into 4 and 6 equal sections for loops and trunk spans respectively.

tions stated in Section III. The results are given in Table II for the indicated source and crosstalk coupling path. The figures for trunks are for the trunk facilities being developed, not for the equalizers to be used initially.<sup>1</sup> The limitations differ somewhat between the two. Other design considerations restrict equalizer spacings to 6 kft on the assumed transmission media. Therefore, spacings beyond this value are not given in the table.

The controlling impairment for loops is singing or near-singing and that for trunks is random noise interference from T1 lines. Thus for both loops and trunks the desire to keep the number of equalizers to a minimum by using long equalizer spacings is offset by the necessity to limit equalizer gain to reduce the effects of near-end crosstalk. The details of the allowable spacings are given below.

Loop equalizers may be spaced 5.8 kft apart on 26-gauge cable if opposite directions of transmission are in separate cable units. When opposite directions of transmission must be placed in the same cable unit, 3.6 kft separation is possible with acceptable probability of degradation in the gain and phase characteristics due to near-singing.



To obtain 6-kft trunk equalizer spacing on 22-gauge cable, opposite directions of *Picturephone* transmission must be in separate cable units and there may be no T1 or T2 systems in any cable unit containing *Picturephone* systems. In addition, opposite directions of T1 and *Picturephone* transmission must be in non-adjacent units. Equalizer spacing of 4.7 kft may be obtained if opposite directions of T1 and *Picturephone* transmission are in adjacent units. If opposite directions of *Picturephone* transmission must be placed in the same cable unit, interference due to random noise may be controlled by maintaining an equalizer spacing no greater than 3.6 kft. In order to achieve this spacing of within-unit and oppositely directed trunk equalizers, it is important to control T1 and T2 interference by keeping *Picturephone* lines and the digital (T1 and T2) lines in separate cable units.

### III. ASSUMPTIONS OF THE ANALYSIS

#### 3.1 Cable Characteristics

Pulp-insulated cable of unit-type construction is assumed to be the medium over which analog *Picturephone* signals are transmitted. The 900-pair cable of this type is made up of eighteen 50-pair cable units, as shown in Fig. 2. Adjacent and alternate units are indicated on the figure. The loss characteristics of 26- and 22-gauge pulp-insulated cable pairs, which are assumed in this study to be the facilities for loop and trunk transmission respectively, are given in Fig. 3. Experimentally obtained crosstalk data for 22-gauge pulp-insulated cable are given in Table III.<sup>8</sup> The same values of coupling loss are used for 26-gauge cable. The loss, in dB, of both near-end and far-end crosstalk coupling is assumed to be normally distributed to  $3.5\sigma$  with means that decrease with frequency at 4.5 and 6.0 dB/octave, respec-

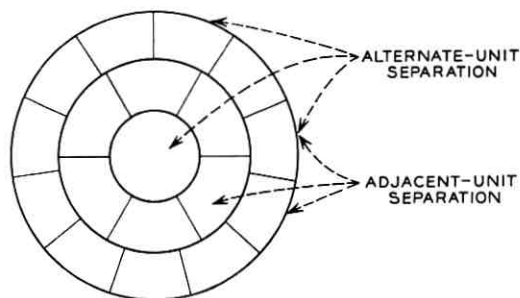


Fig. 2—Construction of unit-type cable.

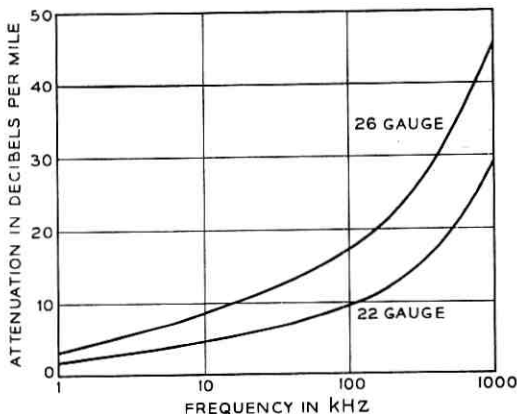


Fig. 3—Loss versus frequency of pulp-insulated cable pairs.

tively.<sup>4</sup> The standard deviation of the distribution is assumed to be independent of frequency.

### 3.2 Requirements

The requirements given by H. E. Brown<sup>5</sup> result from subjectively determined limits allocated among the various segments of the circuit that make up the complete connection. The equal-level crosstalk loss which allows the interference from the worst-disturbing signal to be noticeable but not objectionable is 45 dB at 150 kHz.<sup>6</sup> This applies to both loops and trunks as well as to all other parts of the end-to-end transmission path. The requirement for random noise on loops is given as a distribution with mean of  $-61.6$  dBm and a standard deviation of 4.0 dB. Although Brown further allocates the loop noise requirement among the systems that may compose a loop, the loop facility here is assumed to be a direct connection between central office and subscriber and is therefore allocated the entire requirement. On trunks the mean and standard deviation per span\* are specified as  $-64.6$  dBm and 4.0 dB respectively. The minimum singing margin (see Appendix B) required to insure that an equalized line has an acceptably small probability of singing is given as 10 dB with probability of 0.99.

### 3.3 Equivalent Number of Disturbers

The detrimental effects of near-end crosstalk occur primarily at equalizer locations. Therefore the equivalent number of disturbers

\* Defined in Fig. 2 of Ref. 5.

( $nq$ ) that must be considered is equal to the number of interfering signals at each equalizer location,  $n$ , multiplied by the number of equalizer sections,  $q$ .

Equal-level far-end crosstalk coupling is computed for the overall line length since its effects are independent of equalizer spacing. The number of equivalent disturbers is therefore equal to  $n$ , the number of actual disturbers.

In the computations that follow, it is assumed that a disturbed pair in a given 50-pair cable unit may be subjected to at most  $n = 25$  oppositely-directed, within-unit disturbers at any one equalizer location. It is further assumed that loops and trunk spans will be composed of at most  $q = 4$  and 6 equalizer sections, respectively. If all pairs within a cable unit are transmitting in the same direction, interference on any disturbed pair, due to the effects of far-end crosstalk, is calculated assuming  $n = 49$  within-unit disturbers. When disturbing and disturbed pairs are in adjacent cable units, the interference is that computed from  $n = 50$  disturbers. All computations assume that the equalizers of the disturbing and disturbed circuits are located at the same point.

### 3.4 Traffic Activity

A traffic activity factor, defined as the percentage of pairs active of those equipped for *Picturephone* transmission, is assumed for loops and trunks. The factor for loops is chosen to be 0.1 whereas that used for trunks is conservatively taken as 1.0.

## IV. WORST-DISTURBER INTERFERENCE

The analysis of worst-disturber interference is given in Appendix A. It is shown that the equalizer spacing is dependent upon the sub-

TABLE III—CROSSTALK COUPLING LOSS AT 1 MHz FOR 22-GAUGE PULP-INSULATED CABLE

Pair Separation	Near-End (>1000 ft)		Far-End (1000 ft)	
	Mean (dB)	Std. Dev. (dB)	Mean (dB)	Std. Dev. (dB)
Within-unit	74	8.8	78	12
Adjacent-unit	93	6.7	94	11
Alternate-unit	105	6.7	103	11

jective effect of this interference, the number of equivalent disturbers, and the traffic activity.

The equal-level loss required to make this type of interference definitely noticeable but not objectionable, denoted by  $L_{REQ}$ , has been subjectively determined to be 45 dB at 150 kHz (Section 3.2). As is shown in Fig. 4, a disturbing signal coupled through the near-end crosstalk path is subject to the gain of the equalizer of the disturbed circuit. Denoting this gain in dB by  $G$ , the required cable crosstalk coupling loss,  $C_\ell$ , is equal to  $(L_{REQ} + G)$  dB at 150 kHz.

Knowing the cumulative probability distribution of the pair-to-pair crosstalk loss,  $F_X(x)$ , the following result (derived in Appendix A) can be obtained.

Assume that in a cable made up of  $q$  equalizer sections, there are  $n$  possible interferers and  $n$  possible disturbed circuits, each of which is active with probability  $p$ . Then the probability that the minimum pair-to-pair loss between the  $n$  interferers and a disturbed circuit chosen at random is less than the required  $C_\ell$  is given by

$$F_m(C_\ell) = 1 - [1 - (1 - a)p]^n \quad (1)$$

where

$$a = [1 - F_X(C_\ell)]^q \quad (2)$$

and  $F_m$  is the distribution of the minimum pair-to-pair loss.

This relation is used to obtain the curves of Fig. 5, where the probability of exceeding the requirement on loops is given as a function of equalizer spacing. The equivalent number of disturbers is treated as a parameter. If a criterion is chosen that limits the probability of exceeding the worst-disturber requirement to 5 percent, and the number of within-unit equivalent disturbers is assumed to be  $25 \times 4 = 100$ , the maximum equalizer spacing is found to be 4.6 kft.

Worst-disturber interference is not a problem on trunks even for the maximum trunk equalizer spacing of 6 kft. This follows since the gain of the trunk equalizer for 22-gauge cable is less than that supplied on the same length 26-gauge loop, and therefore a smaller coupling loss,  $C_\ell$ , is required.

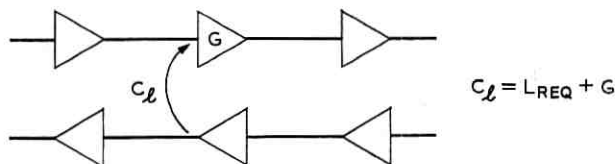


Fig. 4—Crosstalk loss,  $C_\ell$ , required to meet the worst-disturber requirement,  $L_{REQ}$ .

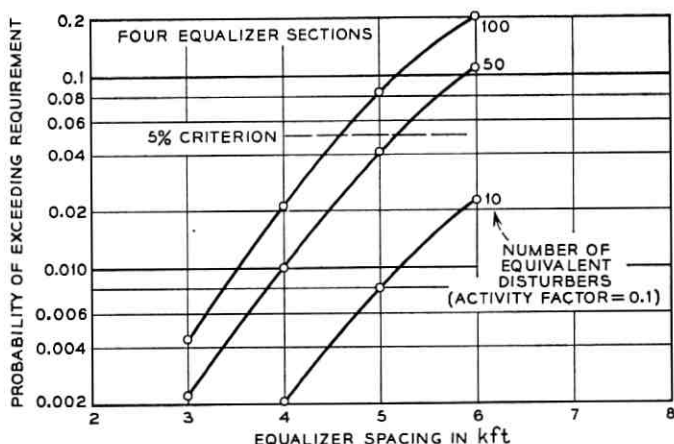


Fig. 5—Probability of exceeding the worst-disturber requirement on loops due to within-unit, near-end crosstalk.

#### V. RANDOM NOISE INTERFERENCE

Even when the effects of crosstalk are not evident as a distinct pattern produced by a worst disturber, as discussed in Section IV, wide-band systems operating within the cable contribute unwanted energy through crosstalk coupling paths to a *Picturephone* circuit. This energy must be taken into account, and is treated here as random noise.

The effects of random noise on equalizer spacing and pair placement are described in the following sections. The models used to determine the received noise power due to near-end and far-end crosstalk coupling of *Picturephone*, T1 line, and T2 line signals are shown in Figs. 6a and 6b respectively. The equalizer gain is not included in Fig. 6b because in the far-end crosstalk case, the disturbing signal is exposed to cable loss equal to the gain supplied by the equalizer of the disturbed line. As shown, the interfering source is coupled through a crosstalk path, assumed to be that described in Section III. The *Picturephone* equalizer gain (included in the near-end case only) is assumed to match the loss, at nominal temperature, of a given length of 26- or 22-gauge pulp cable. Deemphasis is used to provide increased immunity to high frequency interference<sup>5</sup> and noise weighting is included to account for subjective weighting of noise by the human eye. The characteristic of station set roll-off, used to enhance picture quality, attenuates the received signal at the high end of the band

and, therefore, must be included. The characteristics of noise weighting plus roll-off, and deemphasis are shown in Fig. 7.

From Fig. 6a the mean noise power at any frequency due to interference coupled through the near-end path is given by

$$N(f) = S(f)X(f)G(f) D(f)W(f)R(f) \quad (3)$$

where  $S(f)$  is the interfering power spectrum,  $X(f)$  the mean crosstalk loss function,  $G(f)$  the equalizer gain, and  $D(f)$ ,  $W(f)$  and  $R(f)$  the deemphasis, noise weighting, and station set roll-off characteristics respectively. The expression for noise coupled through the far-end crosstalk path is the same, except for the gain factor,  $G(f)$ , which is not included. Numerical integration of equation (3) yields the mean noise power on the disturbed pair due to one disturber. The amplitude distribution of  $N(f)$  is based on the lognormal distribution of crosstalk loss, and is therefore also normal in dB. The probability distribution of the noise power accumulated from  $nq$  equivalent disturbers is found by convolving the lognormal distribution  $nq$  times.<sup>7</sup> The noise power exceeded with some given probability can then be determined.

The noise produced through far-end crosstalk coupling is calculated by first finding the far-end crosstalk loss distribution for the total loop or trunk span length of interest and then applying the model of Fig. 6b. The noise due to  $n$  actual disturbers is determined by convolving the amplitude distribution of the noise generated by a single disturber  $n$  times.

### 5.1 Interference From Picturephone Signals

Results are naturally dependent upon the source of the interference, the *Picturephone* signal. In this study the preemphasized *Picturephone*

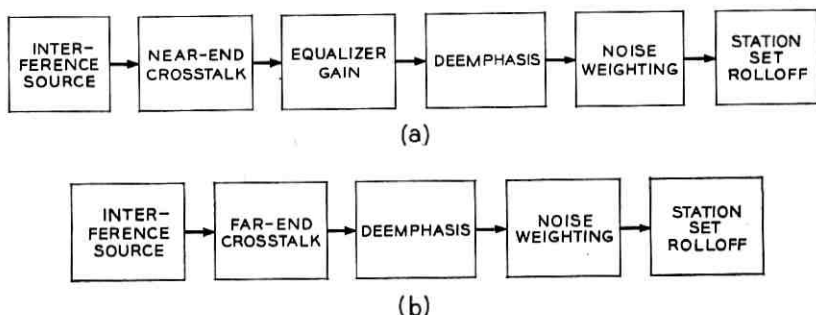


Fig. 6—Model used to compute random noise interference coupled through (a) the near-end crosstalk path and (b) the far-end crosstalk path.

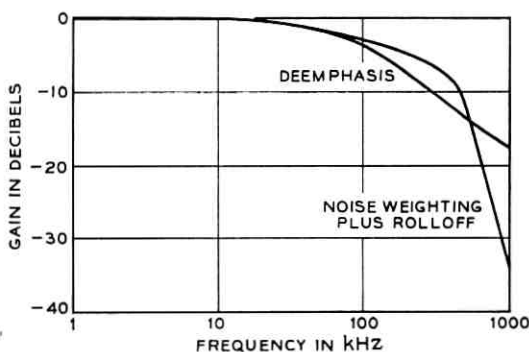


Fig. 7—Gain-frequency characteristics of noise weighting plus station set roll-off, and deemphasis network.

spectral envelope is assumed flat to a frequency of 15 kHz, and decreasing at a 6 dB/octave rate thereafter. The spectrum is scaled to produce a total signal power of 0 dBm.

Only the video information portion of the *Picturephone* signal is preemphasized while the synchronizing pulses, the dominant part of the signal, remain unaltered. The spectral envelope of the preemphasized and unpreemphasized signals are therefore very much the same.

#### 5.1.1 *Picturephone Interference on Loops*

The mean noise power due to *Picturephone* interference on loops is given as a function of equalizer spacing in Fig. 8. The average number of equivalent within-unit disturbers is shown as a parameter. The exact number of equivalent disturbers cannot be specified since each disturbed pair is assumed active with probability  $p$ . The average number is equal to  $nqp$  where  $n$  and  $q$  are as previously defined and  $p$  is the traffic activity factor, equal to 0.1 in the loop case.

Using the requirement (Section 3.2) that the mean noise power on loops should not exceed  $-61.6$  dBm and considering the effect of 100 equivalent disturbers, it is seen that the noise produced is within that allowed even for equalizer spacing of greater than 6 kft. If it is possible to restrict each cable unit to carry only one direction of transmission, the noise produced on a given disturbed pair is primarily due to the effects of within-unit far-end crosstalk coupling. This noise is well within the requirement and results in no limitation on total loop length.

### 5.1.2 Picturephone Interference on Trunks

With the availability of the trunk facilities under development, trunks will be limited to 22-gauge pulp-insulated cable and trunk spans will be composed of a maximum of six equalizer sections of at most six kft each. The 22-gauge cable has less loss than 26-gauge and therefore with identical equalizer spacing requires less gain. Hence 22-gauge trunks have some advantage relative to 26-gauge loops in near-end crosstalk coupling. The advantage, however, is reduced by the increased activity on trunks (a traffic activity factor of 1.0 is assumed).

The allowable trunk equalizer spacings are derived using the assumption that one of the several sources of noise (*Picturephone*, T1, T2) dominates and may, therefore, be allocated the entire trunk span noise objective. If, in engineering a trunk span, it is found that the total noise from two or more sources exceeds the requirement and that the contribution from each of the sources is comparable, the noise power allocated to the span will have to be further allocated among the sources of noise, and the equalizer spacings or the number of disturbers reduced so that the total noise produced is within that permitted.

Figure 9 shows the mean within-unit near-end crosstalk noise power as a function of trunk equalizer spacing with the equivalent number of disturbers,  $nq$ , a parameter. As shown, with 150 equivalent dis-

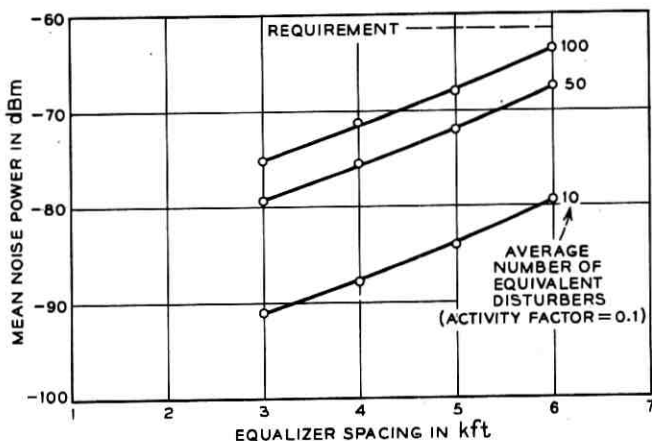


Fig. 8—Mean noise power on loops due to *Picturephone* interference coupled via within-unit, near-end crosstalk.



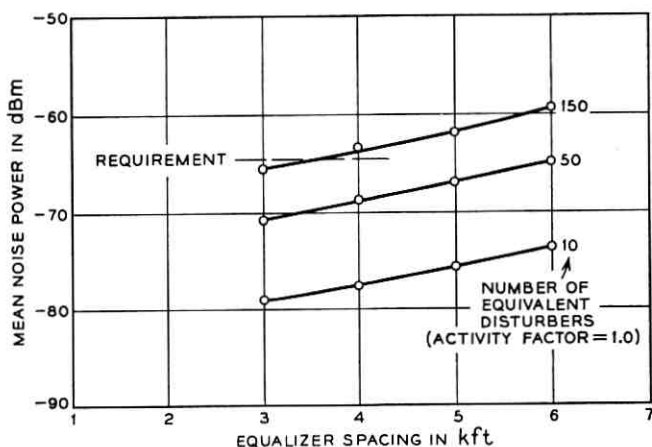


Fig. 9—Mean noise power on trunk spans due to *Picturephone* interference coupled via within-unit, near-end crosstalk.

turburs, the noise produced is within the requirement of  $-64.6$  dBm only if the equalizer spacing is limited to 3.6 kft. The crosstalk advantage derived with adjacent-unit separation of opposite directions of transmission is sufficient to control the interference created by near-end crosstalk coupling of *Picturephone* signals and therefore allow maximum equalizer spacings. The noise on a trunk span of a given overall length, due to  $n$  actual disturbers coupled through the within-unit far-end crosstalk path is shown in Fig. 10. With the maximum of 49 actual disturbers the noise produced on a 36-kft trunk span just meets the requirement.

### 5.2 Interference from Digital Carrier Systems

Concern has been raised over the possibility of degradation of the *Picturephone* signal by noise crosstalked from the digital T1 and T2 lines. This is possible since T1 is found largely in urban areas and often on 22-gauge cable, the same environment and medium in which *Picturephone* analog trunks will initially be installed.<sup>8</sup> Although T2 signals will initially be transmitted on a specially constructed low capacitance cable, it is planned that future T2 designs will allow transmission over 22-gauge pulp cable.<sup>9</sup> The study of interference on *Picturephone* lines caused by T2, located within the same 22-gauge pulp cable sheath, is therefore included.

Compatibility with T1 carrier is of particular concern because of its widespread use and, as shown in Fig. 11, the fact that the maxi-

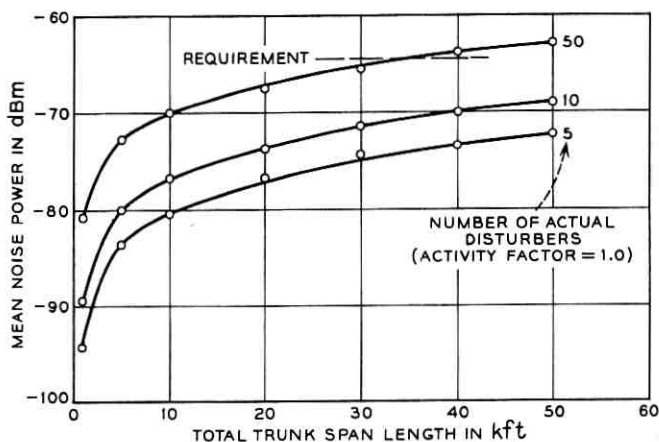


Fig. 10—Mean noise power on trunk spans due to *Picturephone* interference coupled via within-unit, far-end crosstalk.

imum energy of the typical T1 spectrum falls at 772 kHz, well within the *Picturephone* band. The 0 to 1 MHz portion of the typical T2 spectrum is also shown in Fig. 11. The noise produced by crosstalk of these digital signals on the *Picturephone* line is computed by again using the models of Fig. 6.

The mean noise power due to within-unit near-end and far-end crosstalk of T1 into *Picturephone* lines is given in Figs. 12 and 13

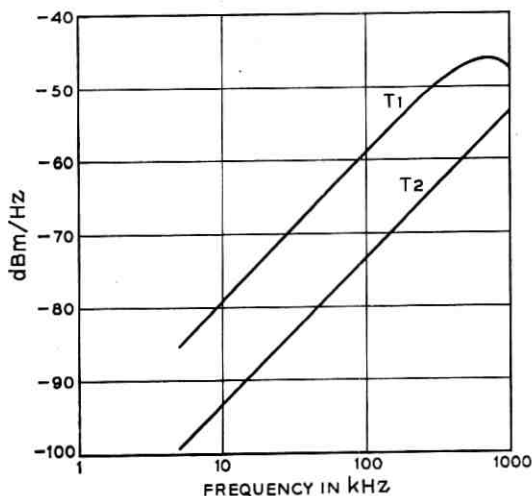


Fig. 11—Spectra of typical T1 and T2 signals.

respectively. As shown in Fig. 12, the noise generated by 150 equivalent disturbers coupled through the near-end path would not meet the requirement even if no equalizer were used and no gain employed. Similarly, Fig. 13 shows that the noise produced by 49 actual disturbers coupled through the within-unit, far-end path is within the requirement only if the total trunk span length is limited to 1.5 kft. Although adjacent-unit separation of opposite directions of T1 and *Picturephone* transmission provides some advantage, it is not sufficient to allow 6-kft equalizer spacing. As shown in Fig. 14, 300 equivalent adjacent-unit T1 disturbers (see Section 3.3) coupled through the near-end crosstalk path yield a noise power within the requirement only if the equalizer spacing is limited to 4.7 kft. If full 6-kft spacing is desired, non-adjacent-unit separation (see Fig. 2) of opposite directions of T1 and *Picturephone* transmission is required. The curves of Fig. 15, however, indicate that adjacent-unit separation of like directions of T1 and *Picturephone* transmission is sufficient to control interference from 50 disturbers coupled through the far-end path and hence allow maximum total trunk span length.

Figures 16 and 17 present the mean noise power generated by crosstalk of T2 signals. Again assuming 150 equivalent disturbers coupled through the within-unit near-end path, it is seen from Fig. 16 that the resulting noise power is within that allowed when equalizer spacing is limited to 2.7 kft. Total trunk span length must be limited to 26 kft, as indicated in Fig. 17, if the noise produced by 49 actual dis-

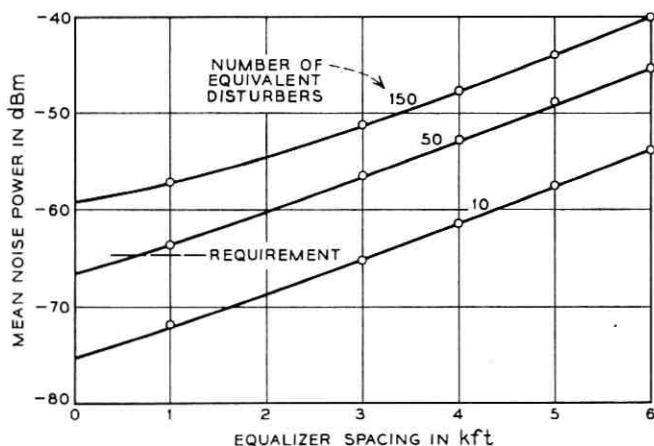


Fig. 12—Mean noise power due to T1 interference coupled via within-unit, near-end crosstalk.

turburs coupled through the within-unit far-end path is to be within that required. Interference from T2 signals can be controlled and 6-kft equalizer spacing and 36-kft trunk span length obtained if T2 and *Picturephone* signals are restricted to separate cable units.

## VI. SINGING OF BASEBAND EQUALIZERS

### 6.1 The Feedback Mechanism

The problem of providing an adequate singing margin for *Picturephone* equalizers is aggravated by the fact that cable loss must be equalized beyond the 1-MHz signal bandwidth. This must be done so the inband phase nonlinearity associated with the gain roll-off is small enough to meet the echo rating requirements.<sup>1,5</sup> The higher the frequency to which the cable is equalized, the more linear the inband phase can be made. However, this also makes the equalizer more likely to oscillate because the equalizer gain increases, while the cross-talk loss decreases with frequency. Thus equalized *Picturephone* lines may provide acceptable crosstalk loss in the 1-MHz signal bandwidth, while allowing the equalizer gain and the loss at the tail of the crosstalk loss distribution to be comparable beyond 1 MHz. This allows the possibility of singing.

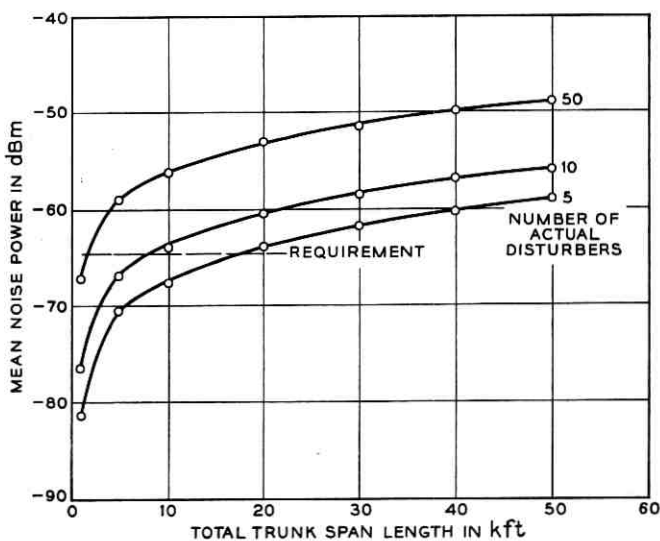


Fig. 13—Mean noise power due to T1 interference coupled via within-unit, far-end crosstalk.

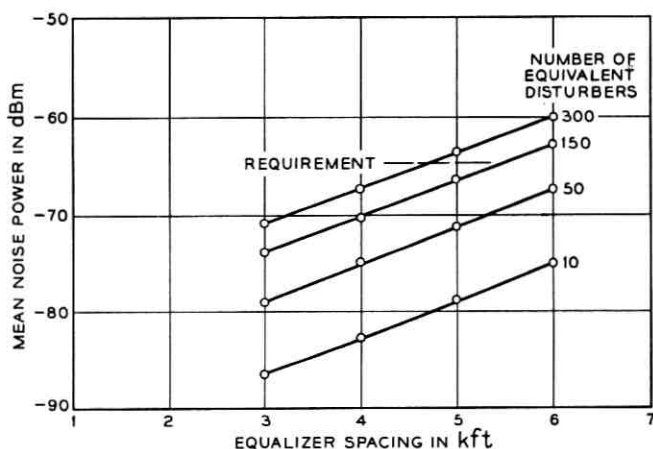


Fig. 14—Mean noise power due to T1 interference coupled via adjacent-unit, near-end crosstalk.

Figure 18 shows the mean NEXT loss and the gains of the loop and the trunk equalizers versus frequency. The significant parameter as far as singing is concerned is the difference between the crosstalk loss and the equalizer gain, denoted  $CR$ , the average of which is also plotted. This difference is half the average loss of the feedback loop around a pair of oppositely-directed equalizers. Since the crosstalk loss is random, the gain of the feedback loop is random. Thus, a pair of equalizers

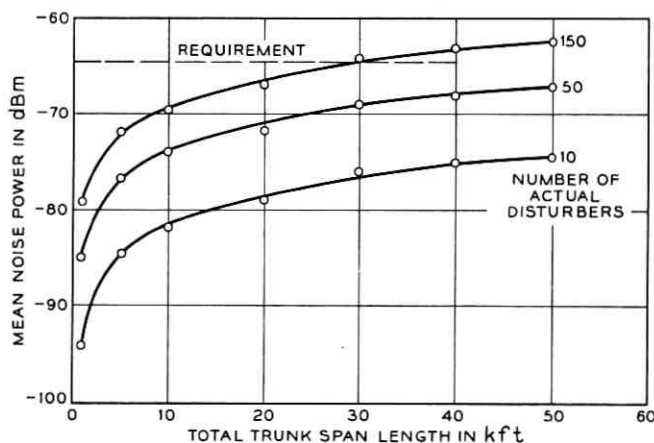


Fig. 15—Mean noise power due to T1 interference coupled via adjacent-unit, far-end crosstalk.

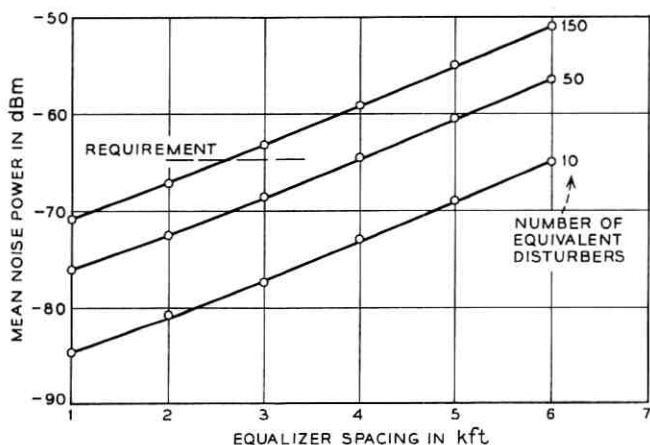


Fig. 16—Mean noise power due to T2 interference coupled via within-unit, near-end crosstalk.

could sing if the loss around the feedback path happened to be less than one when the phase was a multiple of  $2\pi$ . Consider, for example, the probability that a single pair of oppositely-directed, 6-kft, 26-gauge loop equalizers, when installed in the same unit of a pulp cable picked at random, will have loop gain greater than unity at 1.3 MHz. It is equal to the probability that a normal random variable of mean  $2 \times CR_{1.3 \text{ MHz}} = 2 \times 16 = 32 \text{ dB}$  and standard deviation  $\sqrt{2} \times 8.8$

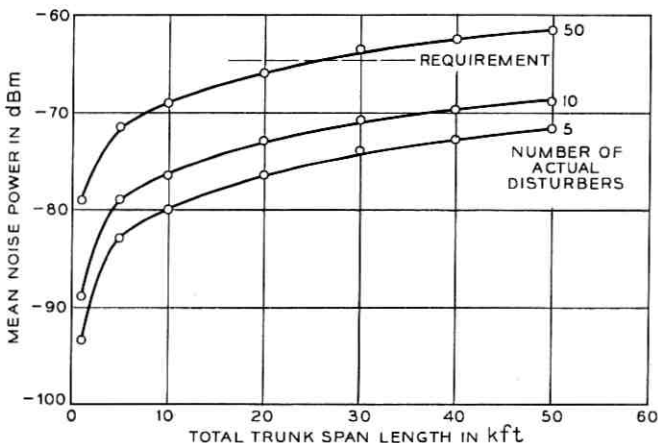


Fig. 17—Mean noise power due to T2 interference coupled via within-unit, far-end crosstalk.

(Table III) = 12.4 dB is less than 0 dB, which is 0.5 percent. The possibility of singing can be decreased by reducing the equalizer spacing. Increasing the crosstalk loss has the same effect on the probability of singing and, in fact, the possibility of singing via NEXT can be eliminated by placing the opposite directions of transmission in different cables. Two-cable operation may not always be practical, however, especially on loops. In addition, singing would still be possible with two-cable operation via near-end-near-end-interaction crosstalk (NE-NE-IXT), as shown in Fig. 19. However, NEXT is the limiting singing path in single-cable operation so NE-NE-IXT will not be considered further. In two-cable operation, the equalizers are limited by their available gain rather than singing considerations.

Thus far we have considered feedback around a single 2-way equalizer. However, in a cable with many parallel equalized lines, each with several intermediate equalizers in tandem, there are a

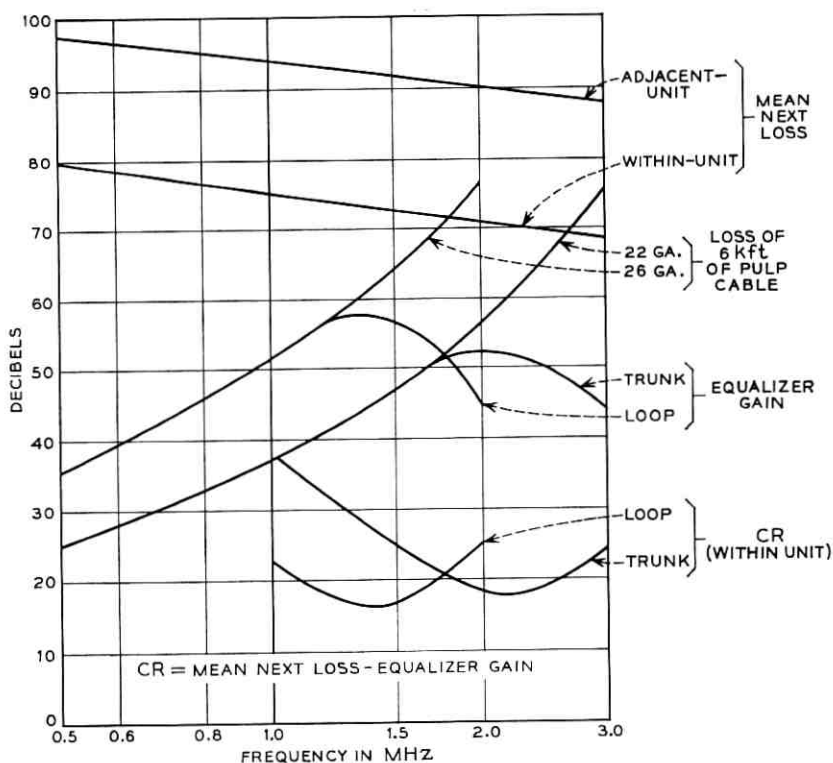


Fig. 18—Equalizer gains and crosstalk loss.

multitude of feedback paths. The feedback path for each equalizer includes all the oppositely-directed equalizers at a site, thus increasing the loop gain. Furthermore, a single feedback loop can include two separate equalizer sites. Two of the many possible paths are shown in Fig. 19.

The two directions of a *Picturephone* equalized line will be looped back on themselves through low-loss connections in several situations. For example, when a person calls his neighbor, one direction of one line may be looped through the zero-loss switching office to the opposite direction of the other line in the same cable as shown in Fig. 20. Such loop-backs act, in effect, like additional low-loss crosstalk paths and thus increase the probability of singing.

### 6.2 Method of Calculation of Singing Margin

A random variable called the singing margin, defined in Appendix B, allows one to estimate how close an equalized line is to singing. In order to calculate the distribution of the singing margin when there are several equalized lines in a single cable, a matrix formulation of the effects of near-end crosstalk is also presented in Appendix B. This model, which allows one to calculate the singing margin taking into account all near-end singing paths, has been implemented in a computer program. The program is used as part of a Monte Carlo procedure to calculate the distribution of the singing margin. The crosstalk is simulated by pseudo-random variables with lognormal ampli-

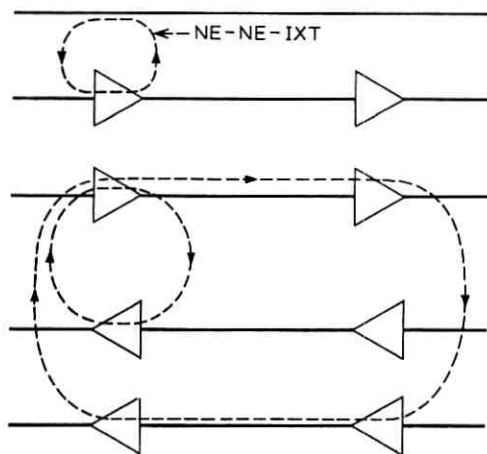


Fig. 19—Some possible singing paths.



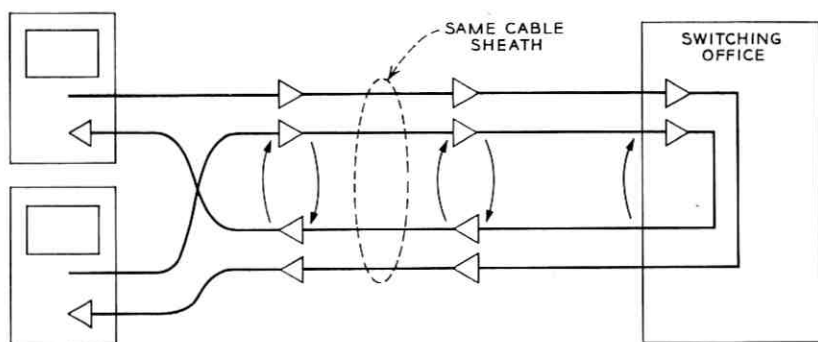


Fig. 20—Loopback of a line through a switching machine.

tude density and uniform phase density. The effect of this crosstalk on the through transmission of an equalized line with specified parameters is calculated on a computer many times, each with new crosstalk samples. This Monte Carlo procedure allows one to estimate the probability that a line is close to singing and to determine the effects of various parameters of the line on this probability.

### 6.3 Limitations Necessary to Avoid Singing

#### 6.3.1 Required Singing Margin

The criterion used to insure that an equalized line has an acceptably small probability of singing is to require its singing margin to be at least 10 dB with probability 0.99. The small probability of exceeding the limit is used because singing is a severe degradation. The 10-dB value for the limit is adopted for computational convenience, as described in Appendix B. The method of analysis and the criteria of acceptability described here are somewhat different from those mentioned in Ref. 1. The results of these two approaches are quite similar however. Through the use of the computer program this 10-dB requirement can be translated into a requirement on the minimum  $CR$ , i.e. the minimum difference between the mean crosstalk loss and the repeater gain. As shown in Fig. 21, the required  $CR$  in dB is proportional to the logarithms of the number of intermediate equalizers in tandem and the number of systems in the cable. Once the required  $CR$  has been determined, it can be translated into a requirement on equalizer spacing for the type crosstalk being encountered. Since singing paths can traverse a switching office, one must consider the possibility of singing for an entire connection.

The difference between crosstalk loss and equalizer gain must also be limited inband, in order to avoid degradation of a line's echo rating due to gain and phase deviations caused by feedback via crosstalk. As a guideline, the inband difference should, at all frequencies, exceed the minimum difference required for singing protection by 10 dB for loops and 15 dB for trunk spans. These limits are roughly those at which a line's echo rating would begin to be degraded. Either the inband difference or the singing requirement can be the limiting one, depending mostly on the gain roll-off characteristic of the equalizer in question.

### 6.3.2 Trunk Equalizer Spacing

For 25 within-unit trunk spans, each with five intermediate equalizers in tandem, the required  $CR$ , found from Fig. 21, is about 31.5 dB. From Fig. 18, the trunk equalizer  $CR$  at the worst frequency is about 19 dB for 6-kft spacing. By scaling the equalizer gain, we find that the spacing would have to be reduced to 4.4 kft to raise the  $CR$  to the required level. The same method applied to 50 adjacent-unit trunk spans indicates that there is adequate singing margin with 6-kft equalizer spacing because of the increased crosstalk loss between adjacent units.

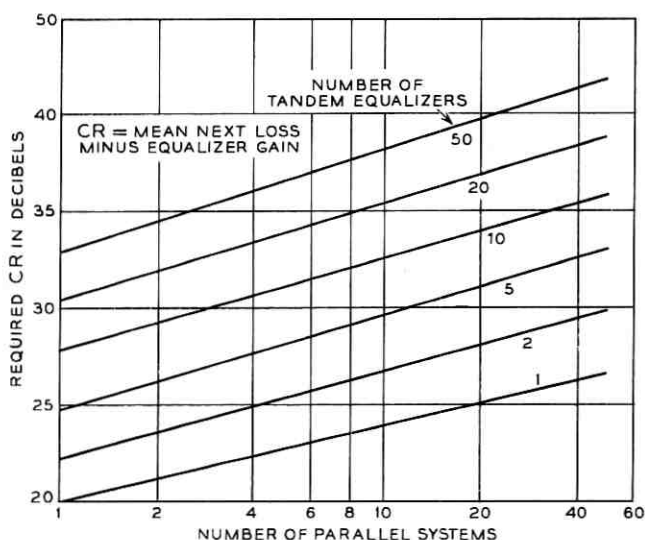


Fig. 21—System parameters required to give adequate singing protection.

Another possible singing situation is with the maximum of six analog trunk spans connected in tandem. This could result in 35 intermediate equalizers. Assuming that only one system would be in parallel over the entire length of such a connection, it can be seen from Fig. 21 that the required  $CR$  is again about 31.5 dB. Hence, the equalizer spacings derived above, 4.4 kft for within-unit operation and 6 kft for adjacent-unit operation, will provide an adequate singing margin for this connection also.

Because the gain roll-off occurs so far out of band, the singing requirements are dominant over the inband requirements.

### 6.3.3 Loop Equalizer Spacing

The controlling limitation on loop equalizer spacing is due to the possibility of connecting two lines in the same cable through an office switch. Such loopbacks change the functional dependence of  $CR$  on the numbers of systems and equalizers from that shown in Fig. 21. Thus, new computations including the effects of a loopback were made to determine the required  $CR$  on loops.

For adjacent unit operation on the maximum length line composed of three intermediate equalizers with 6-kft spacing, the singing margin is adequate even with two lines looped back. To provide singing protection for within-unit operation, the equalizer spacing must be reduced to 4.2 kft. Since the singing margin with a loopback is very insensitive to the number of systems, this suffices for 25 systems also, the maximum number possible. However, this reduction to 4.2-kft spacing only provides adequate singing protection; it does not give adequate protection against echo rating degradation. To do this, the equalizer spacing must further be reduced to about 3.6 kft. This increases the one percent point of the distribution of the singing margin at 1 MHz to 20 dB for two systems looped back through a switching office. Out-of-band loss in the office equalizers could increase the minimum spacing required by singing limitations from 4.2 kft to perhaps 4.4 kft, the limit with no loopback. Such loss cannot be used inband, however, so it cannot affect the 3.6-kft limit due to echo rating protection. This limit can be increased only by improving the crosstalk separation.

## VII. ACKNOWLEDGMENTS

We wish to acknowledge valuable discussions with many of our colleagues at Bell Laboratories. In particular, those with T. V. Crater

on worst-disturber interference, M. W. Baldwin, Jr., on random noise weighting, and L. Hochgraf on carrier system interference have been especially helpful.

## APPENDIX A

### *Worst-Disturber Interference*

#### *A.1 Characteristics of Worst-Disturber Interference*

When a *Picturephone* signal is coupled via a crosstalk path into another cable pair, it is approximately differentiated. Sharp transitions in the interfering signal appear as spikes in the disturbed pair. In particular, the sync pulses at the end of each scan line of the video signal represent step functions displaced by the horizontal blanking time. Differentiating both step functions results in two delta functions, one positive and one negative, in the disturbed video signal. Hence each scan line in the disturbed signal will have a positive and negative delta function superimposed upon it. When displayed, the interference will manifest itself as a white vertical line and a black vertical line. Since the oscillators generating the interfering and disturbed signals will not be at precisely the same frequency, the interference pattern will drift across the raster of the displayed disturbed signal. The loss required to render this interference just perceptible is denoted by  $L_{REQ}$  and is equal to 45 dB at 150 kHz. (See Section 3.2).

#### *A.2 Minimum Pair-to-Pair Loss Distribution for $n$ Interfering Signals*

Let  $F_X(x)$  be the pair-to-pair coupling loss distribution between one interferer and one disturbed signal. We are interested in obtaining the distribution of the minimum loss when there are  $n$  interferers,  $F_m(x)$ . Let the probability density corresponding to  $F_X(x)$  be  $f_X(x)$ . Then the probability that a pair-to-pair loss is in the interval  $(x, x + dx)$  is  $f_X(x) dx$ . For a given disturbed pair, the probability that a given coupling loss is in this interval and all other losses are greater is given by

$$p(x) = [f_X(x) dx][1 - F_X(x)]^{n-1}. \quad (4)$$

Since there are  $n$  possible disturbing pairs, the probability that the coupling loss between the given disturbed pair and any disturbing pair is in the interval  $(x, x + dx)$  and all other coupling losses are greater, is the sum of  $n$  terms of the form (4), or

$$np(x) = n[f(x) dx][1 - F_x(x)]^{n-1}. \quad (5)$$

The associated distribution function,  $F_m(x)$ , is the integral of equation (5) or

$$F_m(x) = 1 - [1 - F_x(x)]^n. \quad (6)$$

Since the given disturbed pair is equally likely to be any of the  $n$  possible disturbed pairs, this is the required result.

### A.3 Effect of Repeater Spacing

We have denoted the required loss as  $L_{REQ}$  at a frequency of  $f_L$  Hertz. Suppose at this frequency the repeater gain is  $G$  dB. The gain  $G$  depends on the length of cable that the repeater must equalize. To meet the requirement, the cable crosstalk loss must be, as shown in Fig. 4, equal to  $L_{REQ} + G$ . Equation (6) can be used to calculate the probability that the coupling loss between a disturbed pair and the worst single interferer is less than  $L_{REQ} + G$ .

### A.4 Cable Activity, and the Effect of Several Disturbers

With  $n$  two-way systems operating in a cable, there are  $n$  interferers and  $n$  disturbed circuits. Assume that during the busy hour each system initiates  $p$  erlangs of traffic. If there is a negligible probability of blocking, this can be interpreted by stating that the probability that a given system is in operation is  $p$ . The probability that  $j$  systems are in operation is then

$$P(j) = C_i^n p^j (1 - p)^{n-j} \quad (7)$$

where

$$C_i^n = n! / (n - j)! j! \quad (8)$$

is the binomial coefficient.

If the cable is made up of  $q$  equalizer sections, then when one interferer is active, the disturbed circuit is subjected to  $q$  interferences. Hence with  $j$  interferers active over the cable of  $q$  sections, the minimum pair-to-pair loss distribution becomes, from (6),

$$F_m(x) = 1 - [1 - F_x(x)]^{jq}. \quad (9)$$

Combining equations (4) and (6) according to the law of conditional probabilities gives

$$\text{Prob}(L < C_i) = \sum_{j=1}^n P(j) \{1 - [1 - F_x(C_i)]^{jq}\} \quad (10)$$

where  $\text{Prob}(L < C_t)$  is the probability that the crosstalk loss  $L$  is less than the required  $C_t$ .

This can be written in closed form by using the binomial expansion to give

$$\text{Prob}(L < C_t) = F_m(C_t) = 1 - [1 - (1 - a)p]^n \quad (11)$$

where

$$a = [1 - F_x(C_t)]^q. \quad (12)$$

## APPENDIX B

### *Calculation of Singing Margin*

#### *B.1 Criterion for Near-Singing Condition*

There are many near-end crosstalk paths between equalized lines in a cable. Since the closed paths thus formed around a single equalizer may be considered as many positive feedback loops, a single open-loop gain, including crosstalk, is not defined. Hence such quantities as gain or phase margin are not appropriate. The criterion adopted as an indication of closeness to singing is the deviation from unity of the normalized closed loop gain in the presence of crosstalk, denoted  $G_n \exp(j\theta)$ . The normalizing gain is that in the absence of crosstalk. The larger the magnitude of the gain change due to crosstalk, the closer the line is to singing. We shall define the log of this quantity to be the singing margin, i.e.,

$$SM = -20 \log | G_n \exp(j\theta) - 1 |. \quad (13)$$

A plot of the singing margin versus the loop gain is shown for three types of feedback configurations in Fig. 22. When there is no feedback,  $SM = +\infty$ ; singing corresponds to  $SM = -\infty$ . Near the point where singing occurs, the singing margin changes very rapidly. Note that  $SM = 0$ , contrary to intuition, does not indicate singing in this context.

Note, also, that the singing margin of an equalized line is a random variable, since it depends on crosstalk losses which are themselves random variables. Thus, we cannot guarantee that a line will not sing; we can only insure that its chances of singing are below some value. Another important fact about the singing margin is that it is a function of frequency. In order that a line not sing, its margin must be greater than  $-\infty$  for all frequencies.

Computing such an overall probability by combining the probabili-

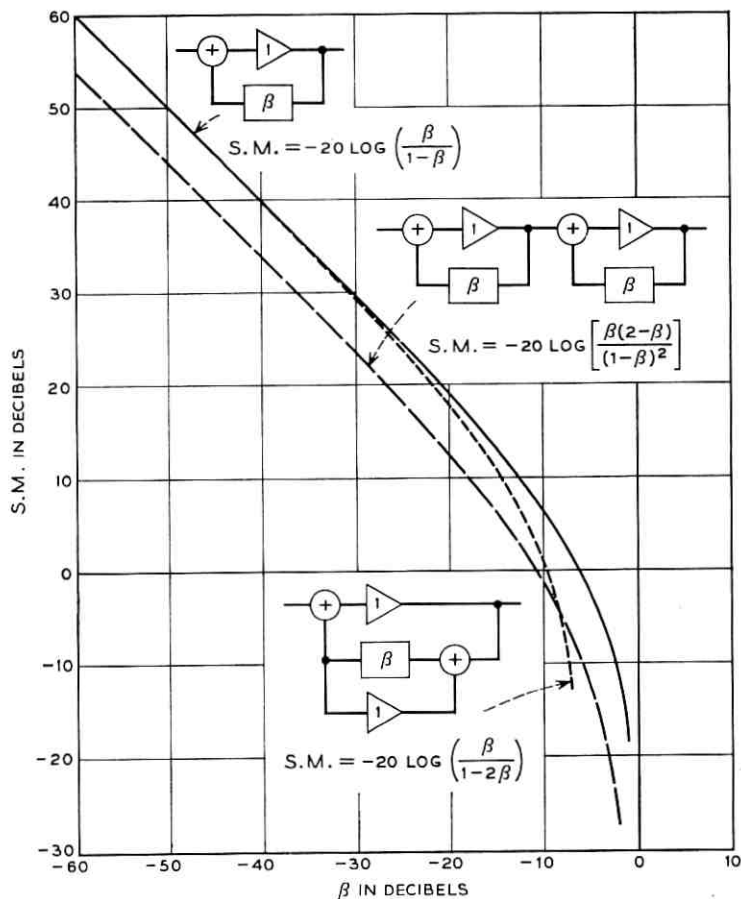


Fig. 22—Singing margin versus loop gain for several feedback configurations.

ties computed at many frequencies using a Monte Carlo technique is time-consuming and inaccurate. Instead, it can be shown that it is reasonable to assume that the overall probability of singing is less than one percent if the one percent point of the distribution of singing margin is at least 10 dB at the worst frequency.

## B.2 Calculation of the Singing Margin

### B.2.1 Only One Intermediate Equalizer

We are interested in the singing margin of an equalized line in a cable containing  $n$  identical, 2-way equalized lines. Let  $c_{jk}(b, c)$ , a complex number, denote the near-end crosstalk voltage transfer func-

tion from point  $b$  of system  $k$  to point  $c$  of system  $j$ . Let  $v_i(b)$ , also complex, represent the Fourier transform of the voltage at point  $b$  of system  $i$ . Then the voltage due to crosstalk from point  $b$  to point  $c$  in a particular system, say  $r$ , is the sum of all the voltages at point  $b$  times the crosstalk gain between each of them and point  $b$  of system  $r$ , i.e.,

$$v_r(c) = \sum_{k=1}^n c_{r,k}(b, c)v_k(b). \quad (14)$$

We can simultaneously write the equations for all the voltages at point  $b$  due to crosstalk from point  $a$  by utilizing matrix notation. Let  $\mathbf{V}(b)$  be a column matrix of  $n$  elements, the  $k$ th of which is  $v_k(b)$ . Let  $\mathbf{C}(b, c)$  be an  $n \times n$  matrix, the element of its  $j$ th row and  $k$ th column being  $c_{j,k}(b, c)$ . Then

$$\mathbf{V}(c) = \mathbf{C}(b, c)\mathbf{V}(b). \quad (15)$$

Referring to Fig. 23, what we are interested in is a relationship of the form

$$\mathbf{V}(b) = \mathbf{T}(a, b)\mathbf{V}(a), \quad (16)$$

where  $\mathbf{T}(a, b)$  is an  $n \times n$  matrix which relates the input and output voltages, including all the effects of feedback via near-end crosstalk

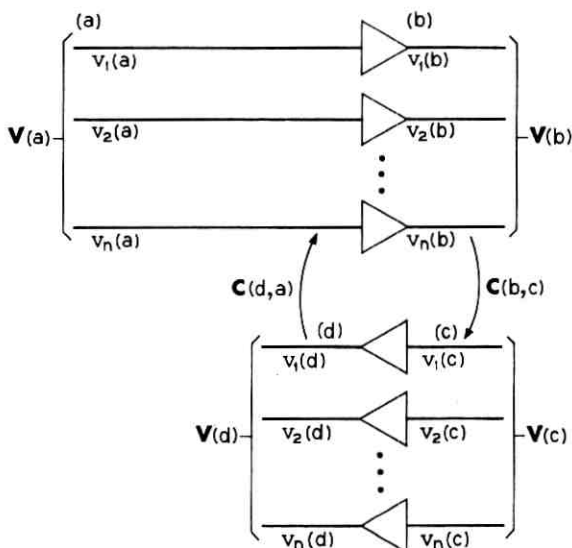


Fig. 23—Vector notation for crosstalk.



paths and the oppositely directed equalizers. Without crosstalk,

$$\mathbf{T}(a, b) = ga\mathbf{I} = q\mathbf{I}, \quad (17)$$

where  $g$  is the equalizer gain,  $a$  is the cable gain,  $q$  is the overall gain of an equalizer section, and  $\mathbf{I}$  is the identity matrix. (Throughout this appendix it should be understood that the quantities involved are functions of frequency, although this functional dependence is not made explicit.)

To find  $\mathbf{T}$  in the presence of crosstalk, referring to Fig. 23, note that

$$\mathbf{V}(b) = q\mathbf{I}\mathbf{V}(a) + g\mathbf{I}\mathbf{C}(d, a)g\mathbf{I}\mathbf{C}(b, c)\mathbf{V}(b). \quad (18)$$

Rearranging and factoring out  $\mathbf{V}(b)$  gives

$$\mathbf{V}(b) = q[\mathbf{I} - g^2\mathbf{C}(d, a)\mathbf{C}(b, c)]^{-1}\mathbf{V}(a), \quad (19)$$

where the exponent indicates the inverse of the matrix. Thus, in the presence of feedback,

$$\mathbf{T}(a, b) = q[\mathbf{I} - g^2\mathbf{C}(d, a)\mathbf{C}(b, c)]^{-1}. \quad (20)$$

If we are interested in the output of the  $k$ th system,  $v_k(b)$ , due only to an input to that system,  $v_k(a)$ , it is given by

$$v_k(b) = t_{kk}(a, b) v_k(a) \quad (21)$$

where  $t_{kk}(a, b)$  is the  $k$ th diagonal element of  $\mathbf{T}(a, b)$ . In other words, the diagonal elements of the  $n \times n$  matrix  $\mathbf{T}$  are the transfer functions of the individual equalized lines, including the effects of feedback via near-end crosstalk paths and oppositely directed equalizers.

### B.2.2 Several Intermediate Equalizers in Tandem

When one considers a cable containing  $n$  two-way equalized lines, each having  $m$  intermediate equalizers in tandem, the problem of calculating the singing margin becomes even more complicated because feedback paths are possible between the equalizers in tandem. There are  $2mn^2$  possible near-end crosstalk paths in such a configuration. These form a total of  $\frac{1}{2}n^2m(m+1)$  partially-overlapping, closed loops.

Let us find an expression for  $\mathbf{T}(0, m) \equiv \mathbf{T}(m)$ , the  $n \times n$  matrix relating the inputs and outputs for one direction of the equalized lines in a cable containing  $n$  two-way systems, each with  $m$  intermediate equalizers in tandem. The crosstalk matrices  $\mathbf{C}_u(k)$  and  $\mathbf{C}_d(k)$  apply to the locations and in the directions shown in Fig. 24. We must solve the following set of equations which apply to that figure:

$$\mathbf{V}(k) = q\mathbf{V}(k-1) + g^2\mathbf{C}_u(k) \sum_{i=k}^m q^{i-k}\mathbf{C}_d(i)\mathbf{V}(i), \quad k = 1, 2, \dots, m. \quad (22)$$

Define

$$\mathbf{P}(k) = [\mathbf{I} - g\mathbf{Q}(k)\mathbf{C}_d(k)]^{-1}; \quad (23)$$

$$\mathbf{Q}(k) = q^2\mathbf{P}(k-1)\mathbf{Q}(k-1) + g\mathbf{C}_u(k), \quad k = 2, 3, \dots, m; \quad (24)$$

$$\mathbf{Q}(1) = g\mathbf{C}_u(1). \quad (25)$$

Then, beginning with the equation for  $\mathbf{V}(1)$ , we find

$$[\mathbf{I} - g\mathbf{Q}(1)\mathbf{C}_d(1)]\mathbf{V}(1) = q\mathbf{V}(0) + g\mathbf{Q}(1) \sum_{i=2}^m q^{i-1}\mathbf{C}_d(i)\mathbf{V}(i) \quad (26)$$

whence

$$\mathbf{V}(1) = q\mathbf{P}(1)\mathbf{V}(0) + gq\mathbf{P}(1)\mathbf{Q}(1) \sum_{i=2}^m q^{i-2}\mathbf{C}_d(i)\mathbf{V}(i). \quad (27)$$

Now, substituting this expression for  $\mathbf{V}(1)$  into the equation for  $\mathbf{V}(2)$ , we get

$$\begin{aligned} \mathbf{V}(2) = q^2\mathbf{P}(1)\mathbf{V}(0) + q^2g\mathbf{P}(1)\mathbf{Q}(1) \sum_{i=2}^m q^{i-2}\mathbf{C}_d(i)\mathbf{V}(i) \\ + g^2\mathbf{C}_u(2) \sum_{i=2}^m q^{i-2}\mathbf{C}_d(i)\mathbf{V}(i); \quad (28) \end{aligned}$$

$$\begin{aligned} [\mathbf{I} - g\mathbf{Q}(2)\mathbf{C}_d(2)]\mathbf{V}(2) = q^2\mathbf{P}(1)\mathbf{V}(0) + g\mathbf{Q}(2) \sum_{i=3}^m q^{i-2}\mathbf{C}_d(i)\mathbf{V}(i); \\ \mathbf{V}(2) = q^2\mathbf{P}(2)\mathbf{P}(1)\mathbf{V}(0) + gq\mathbf{P}(2)\mathbf{Q}(2) \sum_{i=3}^m q^{i-3}\mathbf{C}_d(i)\mathbf{V}(i). \quad (29) \end{aligned}$$

This expression for  $\mathbf{V}(2)$  is now substituted into the equation for  $\mathbf{V}(3)$ . The process can be continued until we get

$$\begin{aligned} \mathbf{V}(m) = q^m\mathbf{P}(m-1) \dots \mathbf{P}(1)\mathbf{V}(0) \\ + g[q^2\mathbf{P}(m-1)\mathbf{Q}(m-1) + g\mathbf{C}_u(m)]\mathbf{C}_d(m)\mathbf{V}(m); \quad (30) \end{aligned}$$

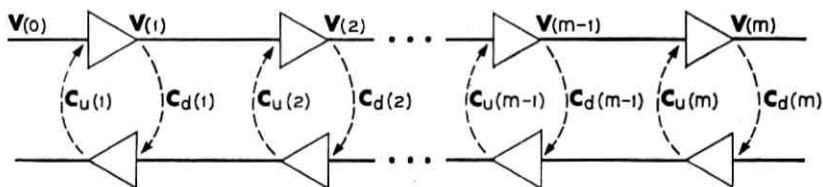


Fig. 24—Vector line of  $n$  parallel systems each with  $m$  intermediate equalizers in tandem.

$$V(m) = q^m P(m) P(m-1) \cdots P(1) V(0). \quad (31)$$

Hence

$$T(m) = q^m P(m) P(m-1) \cdots P(1), \quad (32)$$

where  $q^m$  is the gain of the overall equalized line. This gives a formula for calculating the transfer functions, including the effects of feedback via near-end crosstalk through equalizers transmitting in the opposite direction, of each of  $n$  systems with  $m$  intermediate equalizers in tandem. This allows calculation of the normalized gain, from which the singing margin is calculated by equation (13).

### B.2.3 Loopbacks

Occasionally a two-way equalized line is looped around at a terminal, i.e., the output of an incoming line is connected to the input of the outgoing line in the same cable, as shown in Fig. 20. This can be handled by simply setting the crosstalk matrices at these locations equal to identity matrices times the gain of the loopback.

### REFERENCES

1. Brown, J. M., "The Picturephone® System: Baseband Video Transmission on Loops and Short-Haul Trunks," B.S.T.J., this issue, pp. 395-425.
2. Bell Labs Staff, *Transmission Systems for Communications*, New York: Bell Telephone Laboratories, Inc., 1970, 4th ed., Chapter 11.
3. Stumpf, F. M., "High Frequency Crosstalk Characteristics of Exchange Grade Pulp Cable and its Effects on T2 Transmission Design," Proc. Tel. Cable Crosstalk Symp., Bell Laboratories, Baltimore, Md., 1966.
4. Henneberger, T. C., and Fagen, M. D., "Comparative Transmission Characteristics of Polyethylene Insulated and Paper Insulated Communication Cables," A.I.E.E. Trans., Part I—Communications and Electronics, No. 60, March 1962, pp. 27-33.
5. Brown, H. E., "The Picturephone® System: Transmission Plan," B.S.T.J., this issue, pp. 351-394.
6. Crater, T. V., "The Picturephone® System: Service Standards," B.S.T.J., this issue, pp. 235-269.
7. Näsell, I., "Some Properties of Power Sums of Truncated Normal Random Variables," B.S.T.J., 46, No. 9 (November 1967), pp. 2091-2110.
8. Cravis, H., and Crater, T. V., "Engineering of T1 Carrier System Repeated Lines," B.S.T.J., 42, No. 2 (March 1963), pp. 431-486.
9. Davis, J. H., "A 6.3 Megabit Digital Repeated Line," 1969 International Communications Conference, Boulder, Colorado, Convention Record, p. 34.9.



## The Picturephone<sup>®</sup> System:

# Digital Encoding of the Video Signal

By J. B. MILLARD and H. I. MAUNSELL

(Manuscript received September 4, 1970)

*We describe in this paper the theory and implementation of a differential pulse code modulation system that is employed in the Picturephone<sup>®</sup> network. Established and original coding techniques have been combined to allow transmission of a coded one-megahertz analog signal over a 6.312-megabit-per-second digital channel.*

### I. INTRODUCTION

Pulse code modulation (PCM) is used for processing of analog signals for digital transmission. Quantizing noise and increased bandwidth are the penalties for the noise immunity inherent in digital transmission systems.

Differential systems<sup>1</sup> such as delta modulation<sup>2-4</sup> and the more general differential pulse code modulation (DPCM) have been investigated because of the ability to shape the spectrum of the noise and to take advantage of the sample-to-sample correlation in the signal. In a series of papers in 1952, B. M. Oliver,<sup>5</sup> E. R. Kretzmer<sup>6</sup> and C. W. Harrison<sup>7</sup> considered the use of linear prediction for "decorrelation" of video signals. In 1956 W. F. Schreiber<sup>8</sup> performed entropy measurements on television signals to determine the predominant redundancies present. In 1958 R. E. Graham<sup>9</sup> demonstrated by computer simulation the feasibility of using 3-bit DPCM for the television transmission of still black and white pictures. More recently J. B. O'Neal, Jr.; has analyzed delta modulation<sup>10</sup> and DPCM<sup>11</sup> for the transmission of video signals. The noise structure of DPCM systems in slope overload has been investigated by E. N. Protonotarios.<sup>12</sup> Subjective evaluations have been recorded by R. C. Brainard.<sup>13</sup> These studies have led the way for the application of DPCM to *Picturephone* transmission.

### II. DIGITAL PROCESSING OF THE VIDEO SIGNAL

#### 2.1 Pulse Code Modulation

The conventional method of digital processing of analog signals is PCM. The signal,  $s(t)$ , is sampled,  $s(nT)$ , quantized to discrete levels,

$s_q(nT)$ , and encoded,  $Z(nT/N)$ , in a binary form for transmission at the digital line rate (see Fig. 1).

The main source of impairment in the decoded and filtered signal is noise introduced at the quantizer.

$$n_q(nT) = s(nT) - s_q(nT). \quad (1)$$

This quantizing noise takes the form of background noise of a generally random nature. In addition, as the number of bits per sample is reduced, the background noise level rises, and contouring patterns appear in the video picture.

## 2.2 Predictive Feedback Systems

Independent processing of each sample is assumed for PCM systems. For signal sources having statistical properties like those of white noise, zero sample-to-sample correlation, this is appropriate. However, most video signals are highly correlated not only from sample-to-sample but from line-to-line and from frame-to-frame as well.

In predictive quantizing systems the value of the signal is predicted (predictor  $P$ ) at each sampling instant from its recorded past history (Store) and only the difference,  $e(nT)$ , between the signal sample and its predicted value,  $s_p(nT)$ , is quantized,  $e_q(nT)$ , and transmitted (see Fig. 2).

The *Picturephone* signal has a 1-MHz bandwidth with an 8-kHz line rate. There are 271 lines per field with a 2:1 interlace. Table I shows the minimum amount of storage required (transmitter and receiver) for various feedback systems assuming that information for use in the predictor,  $P$ , is stored with an accuracy of eight bits per sample.

The costs and complexities of systems containing large stores weighed heavily against the use of frame-to-frame correlation<sup>14</sup> at the time the design of a codec for the *Picturephone* network was begun. The work of O'Neal on television signals, whose line-to-line statistics are much the same as those of *Picturephone* signals, demon-

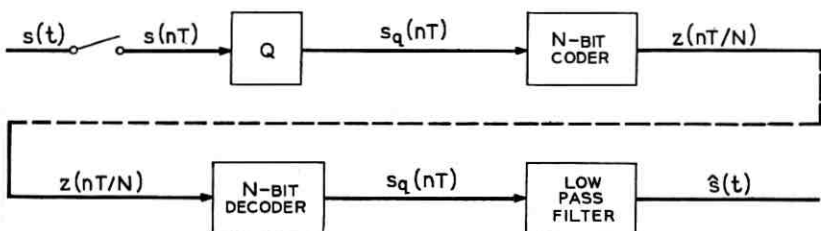


Fig. 1—Pulse code modulation (PCM) system.

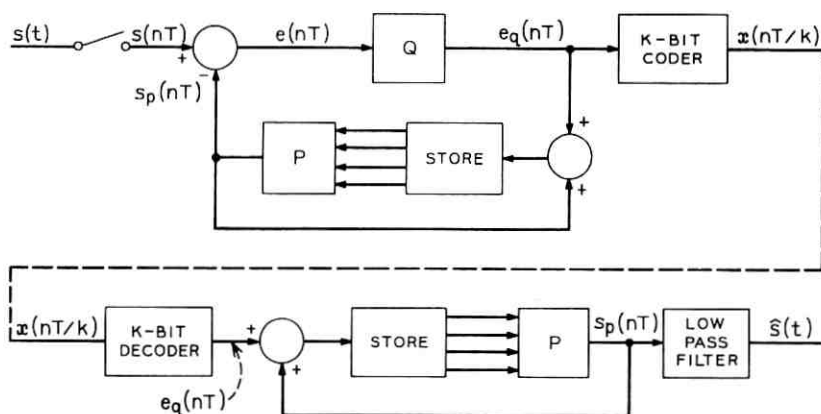


Fig. 2—Predictive quantizing system.

stated that only about a 2-dB gain in signal-to-noise ratio (S/N) could be achieved using linear prediction if horizontal and vertical correlations were approximately equal. Somewhat less could be achieved with a 2:1 interlace format. The design effort was therefore focused on sample-to-sample decorrelation procedures.

For sample-to-sample predictive systems, the store takes the form of a weighted tapped delay line ( $T$  is the sampling interval) with the weights,  $a_i$ , to be determined by the statistics of the signal (see Fig. 3).\*

$$s_p(nT) = \sum_{i=1}^{\ell} a_i \{e_a[(n-i)T] + s_p[(n-i)T]\} \quad (2)$$

where the  $a_i$ ;  $i = 1, 2, \dots, \ell$  satisfy the equations

$$R_{0i} = a_1 R_{1i} + a_2 R_{2i} + \dots + a_{\ell} R_{\ell i} \quad (3)$$

and  $R_{ij}$  is the correlation function,

$$R_{ij} = E\{s[(n-i)T]s[(n-j)T]\}. \quad (4)$$

Measurements<sup>9</sup> have shown that, for samples taken on the same line, television signals have an autocorrelation function which is very nearly Laplacean.<sup>†</sup>

\* The assumption here is that the quantizer makes very little difference in the statistical properties of the error signal, an assumption which becomes more nearly correct as the number of quantizing levels is increased. Note that an exact analysis is impossible using linear prediction techniques with the non-linear quantizer in the network.

† As with all statistical properties, care should be exercised in interpreting this result. Although a typical subject on the average will exhibit this type of correlation, there are likely to be regions of the picture where much less sample-to-sample correlation is present.

TABLE I—FRAME STORAGE REQUIRED FOR  
VARIOUS PREDICTIVE SYSTEMS

Frame Correlation Predictor	1,084,000 bits
Line Correlation Predictor	4,000 bits
Sample Correlation Predictor	16 bits

$$R_s(\tau) \approx \exp(-\alpha |\tau|). \quad (5)$$

For this type of source, the optimum solution to minimize the mean square error between input and output (predicted signal) is one where all the  $a_i = 0$  except  $a_1$ .<sup>11</sup> This means that only the previous sample is effective in decorrelating the signal. The predictive feedback coder then takes the form shown in Fig. 4. A measure of the validity of the assumptions and the effectiveness of the decorrelation process is the amount of correlation removed from the video signal and the amount remaining in the error signal. The adjacent sample correlation coefficient

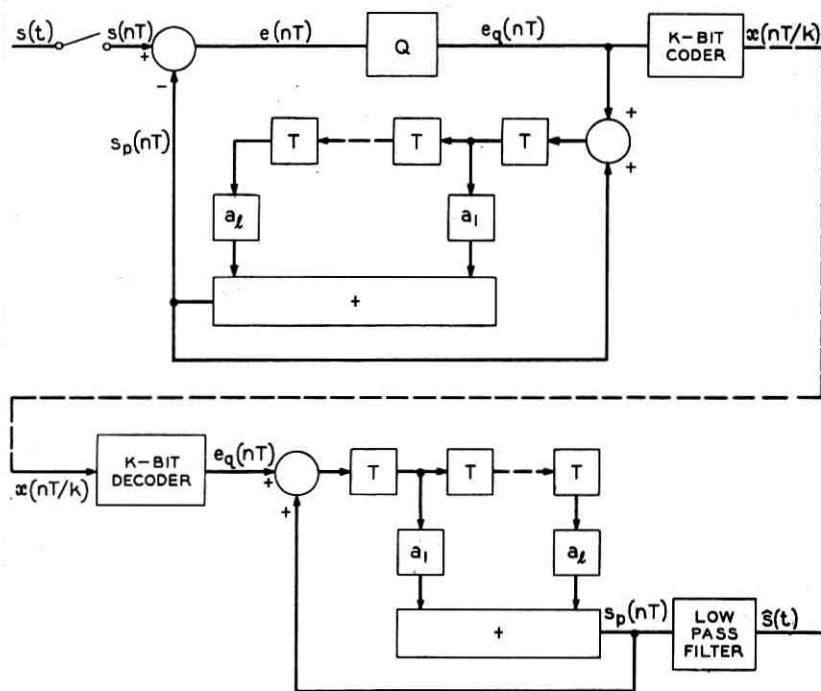


Fig. 3—Sample-to-sample predictive quantizing system using a tapped delay line.



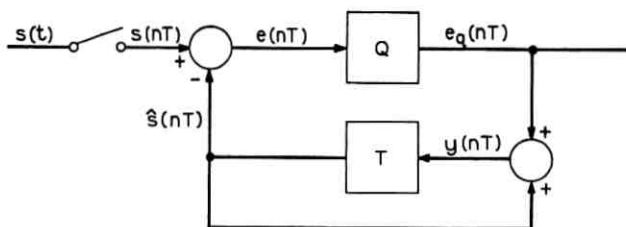


Fig. 4—One-tap predictive quantizing coder.

$$\rho_s = \frac{E\{s(t)s(t+T)\}}{E\{s^2(t)\}} \quad (6)$$

was calculated from sample statistics and found to be 0.93 for the still picture "Rosalie" as referred to in Section 2.6. The adjacent sample correlation coefficient

$$\rho_e = \frac{E\{e_q[nT]e_q[(n-1)T]\}}{E\{e_q^2(nT)\}} \quad (7)$$

for the same picture is 0.4407 as calculated from laboratory data for the three-bit DPCM coder using the quantizer of Fig. 5. This shows that considerable redundancy has been removed from the signal but that some sample-to-sample correlation remains.

### 2.3 Quantizing Noise

To demonstrate that the noise introduced in predictive feedback systems can be determined by analyzing the error at the quantizer itself, consider the one-tap system of Fig. 4. The quantizing error  $QE$  is given by

$$QE = e_q(nT) - e(nT) \quad (8)$$

where

$$e(nT) = s(nT) - \hat{s}(nT), \quad (9)$$

$$e_q(nT) = y(nT) - \hat{s}(nT), \quad (10)$$

$$y(nT) = \hat{s}[(n+1)T]. \quad (11)$$

Substituting equations (9), (10) and (11) into (8) gives

$$QE = s(nT) - \hat{s}[(n+1)T] \quad (12)$$

which is the system error in transmitting the signal in the absence of any channel errors. The system noise can therefore be determined

directly from the characteristic of the error signal and a knowledge of the quantizer. The manner in which the quantizing noise manifests itself as picture impairments will be discussed in later sections.

#### 2.4 Differential Pulse Code Modulation

Consider the single tap predictive feedback system shown in Fig. 4. The feedback signal,  $\hat{s}(nT)$ , is given by

$$\hat{s}(nT) = \sum_{i=1}^{\infty} e_o[(n-i)T]. \quad (13)$$

If a loss factor  $\beta < 1$  is introduced into the delay section then equation (13) is modified to give

$$\hat{s}(nT) = \sum_{i=1}^{\infty} \beta^i e_o[(n-i)T]. \quad (14)$$

Past history is weighted giving greater importance to the more recent samples. In practice, the feedback network in the *Picturephone* codec is replaced by an analog network which consists of an integrator,  $F$ , with leak (see Fig. 6);

$$F(s) = \frac{K}{s + \alpha}, \quad (15)$$

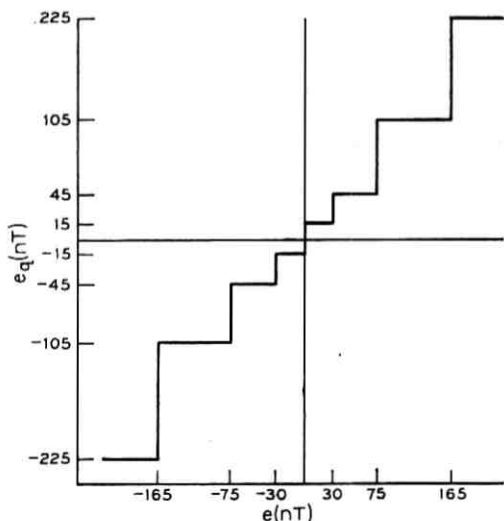


Fig. 5—Three-bit DPCM quantizer characteristic.

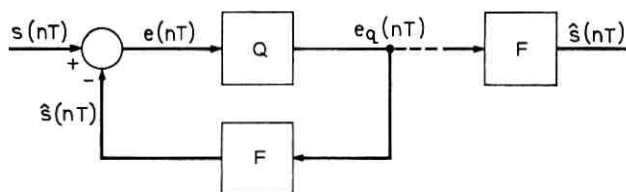


Fig. 6—DPCM system with feedback integrator.

where the  $\beta$  [equation (14)] and  $\alpha$  [equation (15)] are related in the following manner:

$$\beta \approx \exp(-\alpha T). \quad (16)$$

A system organized in this fashion is a DPCM system. A differential system very nearly transmits the differences between successive samples rather than the samples themselves. The receiver accumulates the received difference and reconstructs an approximation to the original waveform. The presence of a quantizer provides the potential for an accumulation of systematic quantizing error. The feedback loop around the quantizer prevents this accumulation.

The feedback network used in the *Picturephone* coder has a  $27\mu\text{s}$  time constant. For subjective effect the time constant was made as large as possible without unduly magnifying the effects of transmission errors on the viewed picture. This is discussed further in Section 2.8.

### 2.5 Subjective Considerations and Noise Shaping

In DPCM, the subjective effect of a video observer can be taken into account and the spectrum of the noise shaped accordingly. As Graham points out,<sup>9</sup> the "concept of predictive quantizing as a truly perceptual coding technique" is one "in which the primary object is not necessarily accurate prediction all or almost all of the time, which frequently is impossible, but only accurate prediction in those signal regions where the observer perception is high." It is known<sup>9</sup> that the eye is more tolerant of noise located at black-white interfaces than in flat regions of the picture. Interfaces are characterized by large values of the difference signal and flat regions by small values. It is therefore possible to use companding on the difference signal to locate the noise in region of the picture where that noise is least objectionable.

Companding is accomplished through the use of a nonlinear quantizing characteristic providing a fine structure and a high S/N for low level difference signals (flat regions) and a coarse structure with its lower S/N for large difference signals (interfaces). The nonlinear

characteristic used in the *Picturephone* codec is shown in Fig. 5. The choice of this particular set of step sizes will be discussed in a later section.

Because of the subjective aspects of rating picture quality, S/N is not a sufficient measure of system performance. However, it is useful to compare the theoretical S/N performances of PCM and DPCM. Based on some simplifying assumptions\* O'Neal<sup>11</sup> has shown that for the optimum one tap predictive filter the S/N improvement for DPCM over PCM is related to the sample-to-sample correlation of the signal,  $R_s(T)$ , in the following manner.

$$\text{S/N IMPROVEMENT} = 10 \log \left[ \frac{1}{1 - R_s^2(T)} \right]. \quad (17)$$

For video signals with only a moderate amount of detail this results in an improvement of approximately 12 dB which is equivalent to two bits of quantizing information per sample.

### 2.6 Signal Impairments

DPCM systems are subject to a unique set of subjective impairments. Among these are overload noise, granular noise, contouring and edge busyness, all resulting from the quantizing process (see Section 2.3) but different in appearance to the observer.

Unlike PCM systems which are amplitude limited, DPCM systems overload on slope. If the sample-to-sample difference is greater than the largest quantizer step, then the system is said to be in slope overload. A system exhibiting slope overload effectively reduces the horizontal resolution of the scene being processed.

In low detail regions of the picture, the noise power is determined by the rather fine quantizing structure designed for small differences. This is termed the granular noise region. Figure 7 shows a typical video waveform having an overload region and a granular noise region.

In the *Picturephone* codec the small step is 15 mV and the large step is 225 mV on a full scale video signal of  $\pm 750$  mV (cf. Fig. 5). The resolution error for any brightness change is plus or minus one-half of a small step. The slope handling capability of the codec is 450 mV per  $\mu\text{s}$ .

\* The assumptions on which the theory is based are that (i) the distribution of the error signal can be approximated by the Laplace density function:

$$P(e) = \frac{1}{\sqrt{2}\sigma_e} \exp\left(-\frac{\sqrt{2}}{\sigma_e} |e|\right),$$

(ii) the quantizer is optimized to the statistics of the error signal and (iii) the probability of overload is small.

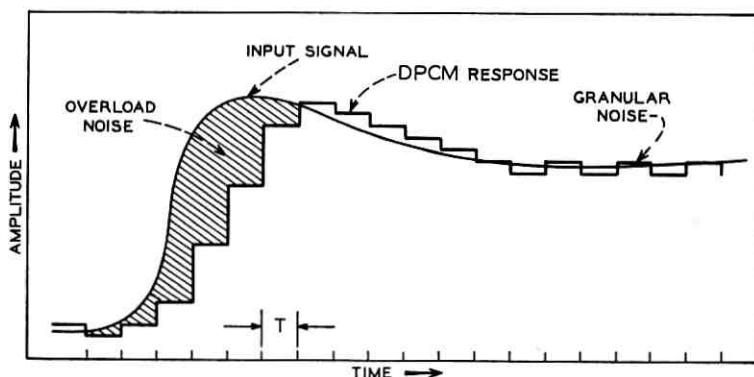


Fig. 7—Illustrated video waveform and DPCM response showing an overload region and a granular noise region.

Figure 8 shows a typical picture ("Rosalie") uncoded, (see Fig. 8a), and exhibiting slope overload after three bit encoding (see Fig. 8b). This is most noticeable as smearing on the ear ring (white-to-black interface).

If the small steps are too large, the strong correlation between the quantizing noise and the signal can lead to visible patterns that are subjectively objectionable. In flat regions of the picture (zero sample-to-sample difference), the DPCM feedback loop oscillates using alternately positive and negative small steps. The integrator leak breaks up this pattern and if the leak time constant is too small, leak contouring can occur. This is illustrated for two different values of video level in Fig. 9. The dashed component of the signal is the leak contour.

Even in the absence of leak, contouring patterns can occur in sloping regions of the video signal. A slope which slightly exceeds (falls short of) the ability of a particular step size to follow it will require periodic correction, through the use of the next larger (smaller) step. If this waveform is repeated through several lines (strong vertical correlation in the picture) and if the differences between the adjacent steps in question are too large, a visible contour pattern will result. Figure 10 demonstrates this effect.

Figure 11 is a photograph of an oscilloscope trace showing a leak contouring pattern. A well-designed three bit codec does not produce visible contours. This picture resulted from changing the leak time constant from  $27 \mu\text{s}$  to  $5 \mu\text{s}$ .

The requirements for minimizing the preceding impairments are in conflict, given a fixed number of steps in the quantizer. Granular noise requirements specify a fine quantizing structure. Slope overload and



Fig. 8—(a) Analog *Picturephone* signal "Rosalie." (b) Photograph of 3-bit DPCM encoded *Picturephone* signal "Rosalie." Exhibit of slope overload.

hence horizontal resolution requirements specify a large outer step. Contouring and, as will be shown below, edge busyness, can both be reduced through the use of a uniform spacing of the steps. Quantizer design is therefore a matter of finding the proper compromise among the subjective effects of the various impairments.

### 2.7 Synchronization and Edge Busyness

Since the system was designed to concentrate the noise in the high frequencies corresponding to the interfaces or edges in the picture, the resulting degradation becomes a limiting factor. Quantizing noise at slightly nonvertical edges takes the form of "stairsteps." This is due to the discrete nature of the sampling process. If in going from frame-to-frame in a still picture there are changes in the response of the

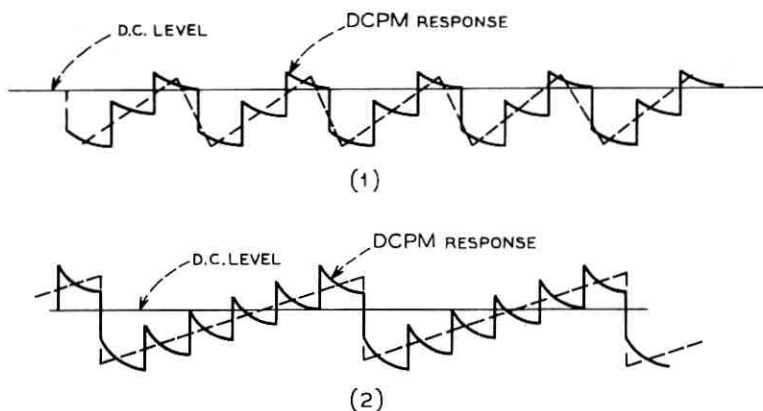


Fig. 9—Illustration of leak contouring patterns.

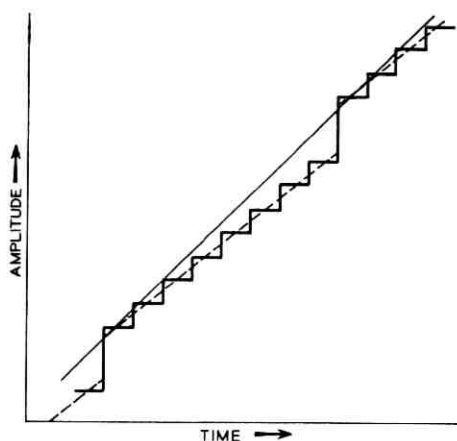


Fig. 10—Illustration of contouring pattern.

DPCM loop, it follows that the edge noise will shift. This gives rise to the degradation known as edge busyness.

Considerable improvement is realized by phase locking the sampling clock in the coder to the line rate of the video source. This ensures that each sample is taken in the same horizontal position from line-to-line and from frame-to-frame. Edge busyness is also a function of noise, quantizer characteristic, threshold tolerances and past history due to the memory in the feedback loop. It follows that synchronization will substantially improve picture quality but will not eliminate edge busyness completely. Figure 12 illustrates a busy edge pattern

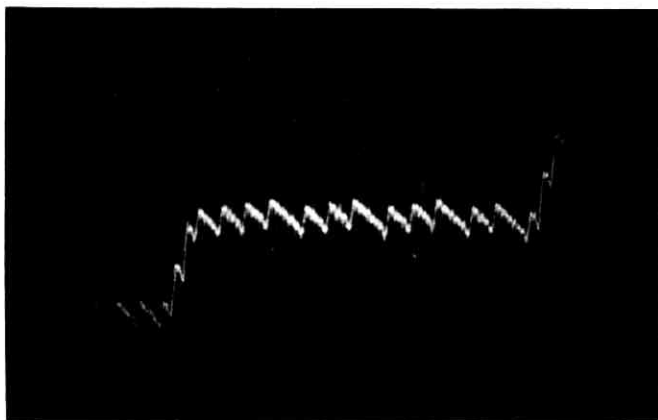


Fig. 11—Photograph of oscilloscope trace of leak contouring in the 3-bit DPCM encoded *Picturephone* signal "Gray Scale" ( $5\mu\text{s}$  leak time constant).

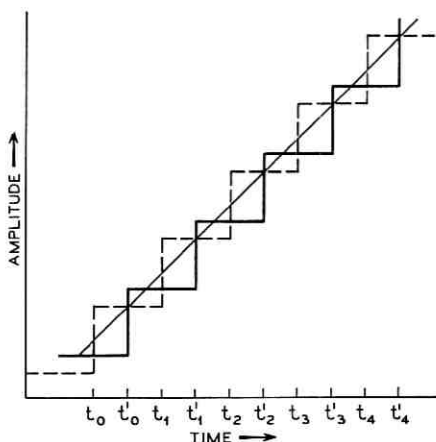


Fig. 12—Illustration of edge busyness.

caused by differences in past history and in clock phase. Figure 13 is a photograph of an oscilloscope trace showing a "busy edge." The different paths taken in making the black-to-white transition are clearly visible in this multiple trace picture.

Of the above factors, the quantizing structure and the leak time constant are under the control of the designer. Since edge busyness results from the variability in the selection of the set of steps to approximate the input ramp voltage it follows that minimizing the horizontal difference between the two responses will minimize the edge busyness in a mean square sense. The horizontal differences are minimized by making the differences between adjacent steps as small as possible. This would be accomplished through the use of a uniform quantizer. However, the existence of the other impairments with their conflicting requirements makes the use of a uniform quantizer impractical as stated above.

### 2.8 Effects of Channel Noise

Since differential encoding removes redundancy in the signal source it provides for more efficient transmission of information. However, the removal of redundancy from a signal will in general, make the signal more susceptible to channel noise.\*<sup>15,16</sup>

For DPCM systems the degradation due to digit errors takes the form of horizontal streaks in the picture. Each streak originates at a decoded sample difference which is incorrect due to a transmission

\* Since the transmission is in digital form, the susceptibility is to bit errors in the channel.



error. Decoder memory causes the effect of a channel error to propagate through successive reconstructed samples for a period of time determined by the leak time constant. The visible effect is therefore lessened by using an integrator with a small leak time constant. The creation of leak contour patterns is a lower bound for reduction in the size of the leak time constant.

### III. CIRCUIT DESIGN

Figure 14 is a block diagram of the overall circuit. The transmitting and receiving circuits are each made up of three major parts, the analog interface, the digital interface and the DPCM coder or decoder circuits.

The input signal may be a pre-emphasized 1-MHz video signal such as that sent from one *Picturephone* station set to another or it may be a 460.8 kb/s binary data signal sharing the same facilities. Level variations as large as  $\pm 6$  dB about the nominal level are acceptable and will be corrected to within  $\pm 0.1$  dB before coding.

The digital signals are B6ZS signals of the proper format for transmission on a T2 facility.<sup>17</sup> The circuits are matched to a 100-ohm balanced line and sufficient equalization is provided in the decoder to compensate for a 22-gauge cable pair of up to 2000 feet.

#### 3.1 *The Transmitting Portion Analog Interface*

All input signals, video or data, pass through common input circuits (Fig. 15), which provide 40 dB of longitudinal suppression and which

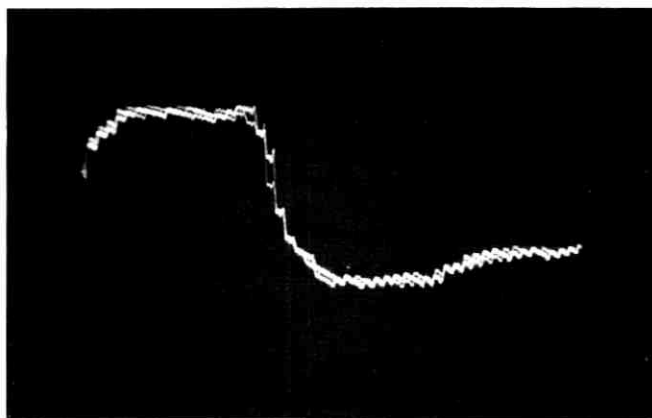


Fig. 13—Picture of oscilloscope trace of edge busyness in 3-bit DPCM encoded *Picturephone* signal "Rosalie."

drive a phase-linear 6-pole low-pass filter with a gaussian roll-off 6 dB down at frequency  $f = 1$  MHz,

$$\text{Filter loss} = 20 \log_{10} \exp(0.69f) \text{ dB.} \quad (18)$$

After filtering, the signal is processed in one of three ways. Picture signals, which are recognized by the 8-kHz sync format, must be de-emphasized<sup>18</sup> in order to obtain a normal video waveform, similar to that at the output of a *Picturephone* camera tube and suitable for coding with a DPCM coder; the de-emphasis circuits have the inverse characteristics of the pre-emphasis circuits used in the station set.

$$\text{De-emphasis loss} = 20 \log_{10} \left| \frac{10(jf + 0.1)}{jf + 1} \right| \text{ dB,} \quad (19)$$

where  $f$  is the frequency in MHz.

The signal is then clamped and passed through automatic gain control (AGC) circuits which maintain a constant 0.375-volt sync-tip to back-porch voltage; thus at the input to the DPCM coder the video signal has a nominal level of 1.5 volts peak-to-peak. Binary data are controlled by a separate suitable AGC circuit with 7.2 dB less gain which adjusts the nominal level to the coder to be 0.9 volts peak-to-peak. Low-level signals, specifically the 12-kHz tone used for testing the channel continuity, pass through fixed gain circuitry. Information as to the presence and nature of the received signal is multiplexed onto the output bit stream to control functions at the decoder that are complementary to those in the coder.

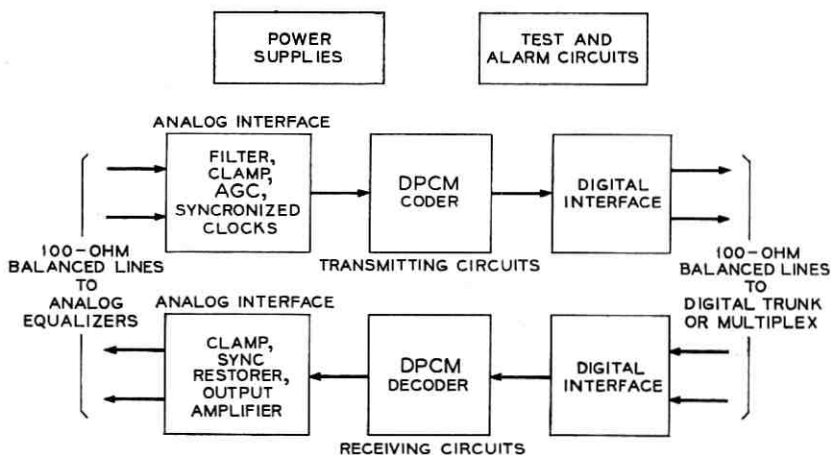


Fig. 14—Codec terminals.

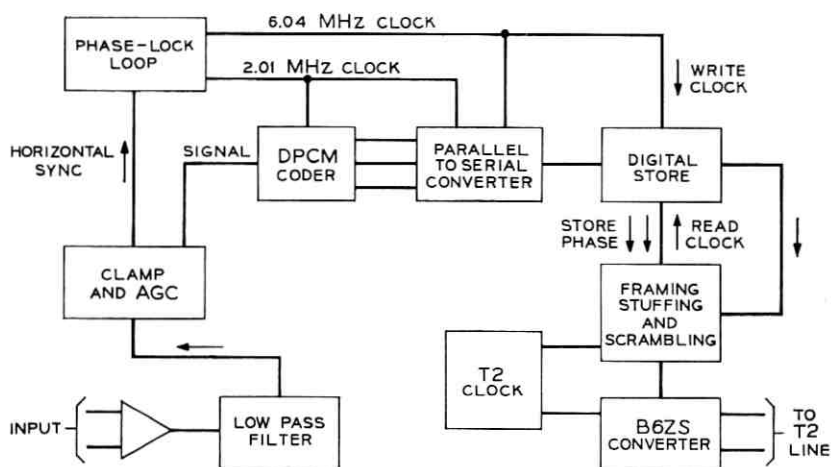


Fig. 15—Block diagram of the transmitting circuit.

The basic clock frequency for the coder circuits, 6.048 MHz, is an exact multiple of three times the *Picturephone* line frequency, 8 kHz, (it is the 252nd harmonic). This allows the sampling clock to be synchronized to the *Picturephone* line rate so that the rectangular video picture is sampled with a uniform grid. This 6.048-MHz frequency is generated by the voltage controlled oscillator (VCO) in a phase-locked loop and is divided by three to obtain the 3-phase 2.016-MHz sampling clock that drives the actual coder. The 6.048-MHz frequency is counted down by a factor of 12 in the digital store addressing circuits and the resulting frequency is counted down by an additional factor of 63 to obtain an 8-kHz square wave. This signal is one input to a phase comparator whose other input comes from the sync separator within the video clamp circuits; the output of the phase comparator is used to control the frequency of the VCO. In the absence of external noise, any individual sample of the picture is made at the same temporal point along the same line with an accuracy of less than ten nanoseconds, a figure that corresponds to less than 0.001 inches displacement across a *Picturephone* display tube. This synchronization process minimizes the busy edge effects. The clock is not synchronized to the data signals.

### 3.2 The DPCM Coder

The differential pulse code modulator circuits that are used in the codec are shown in Fig. 16. At each sampling instant of time the video input is compared directly against the decoded video output signal

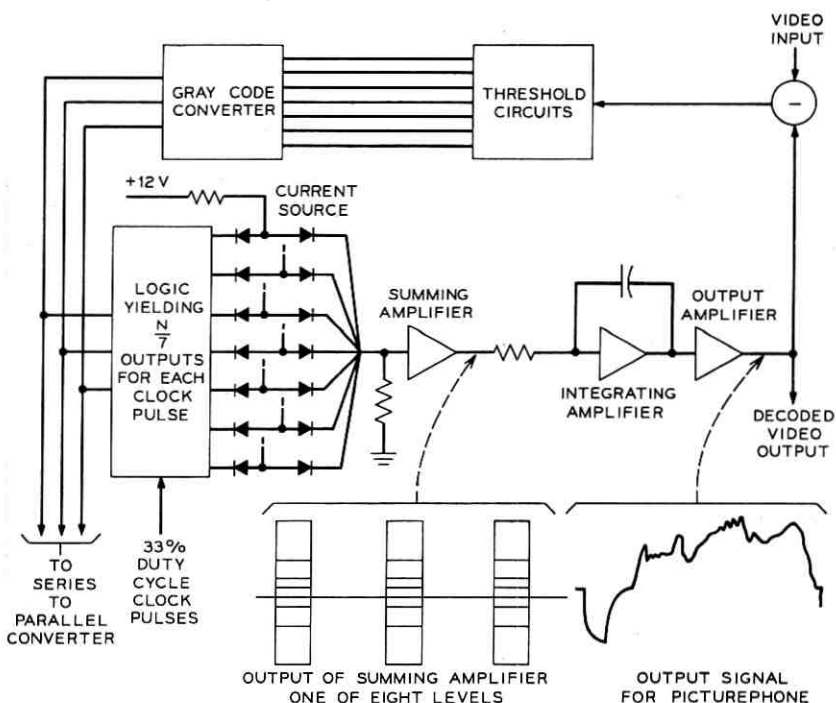


Fig. 16—DPCM coder.

last transmitted. The output of the difference amplifier is connected in parallel to the inputs of seven threshold detectors set to  $0$ ,  $\pm 2$  percent,  $\pm 5$  percent and  $\pm 11$  percent of the nominal 1.5 volt peak-to-peak video input signal (cf. Fig. 5). The cyclic coded outputs of these detectors are the output of the coder and are taken to the digital-to-analog circuit shown in the lower half of the circuit of Fig. 16 whose logic, gating, summation, integration and amplification circuits produce a decoded video signal that is a close replica of the original input signal. The various circuit functions of the digital-to-analog circuitry are kept separate to avoid undesirable voltage interactions and the duty cycle is set at 33 percent to avoid similar trouble in the time domain. The required (nonlinear) companding characteristic is achieved by making the current sources appropriate for output voltage changes,  $e_c(nT)$  of magnitude  $\pm 1$  percent,  $\pm 3$  percent,  $\pm 7$  percent, or  $\pm 15$  percent of the peak-to-peak quantizer input video signal (cf. Fig. 5).

Low-speed data signals<sup>19</sup> are coded in exactly the same manner as video signals. However the magnitude of the data signals is such that

the largest step size is exactly 25 percent of the total peak-to-peak excursions. This ensures that each binary transition can be coded within four sampling intervals, a time interval that is slightly less than the period of 460.8 kb/s data. Between binary transitions extensive use is made of the smallest steps whose magnitude is only  $\pm 1.7$  percent of the binary peak-to-peak signal.

### 3.3 *The Transmitting Portion Digital Interface*

After conversion to a serial bit stream, the coder output bit stream passes through a digital store (see Fig. 15), with a capacity of four 3-bit words each of which occupies a fixed position within the memory. The information is not taken out of the store at the steady 6.048 MHz rate but at a 6.312 MHz rate that is sometimes interrupted so that the bit stream can be framed. The framing process must allow the inclusion of three signaling elements and must make adjustments for small variations in the frequency of the sampling clock which is synchronized to the video horizontal scanning frequency. The three signaling elements are transmitted as zeros except when

$BZ = 1$ , if a signal is detected at the coder input, or

$PD = 1$ , if a video signal is being coded, or

$RF = 1$ , where  $RF$  is a store framing signal.

There are two types of 108-bit words  $R$  and  $S$  where

$$\{(\text{Bit } 1 = 1), (\text{Bit } 33 = 1), (\text{Bit } 65 = 1), (\text{Bit } 92 = BZ)\} \in R,$$

and

$$\{(\text{Bit } 1 = 0), (\text{Bit } 33 = 0), (\text{Bit } 65 = 0), \\ (\text{Bit } 76 = PD), (\text{Bit } 92 = RF)\} \in S,$$

and in each case all other bits or elements of the word carry DPCM information. The number of information bits is different for the two words and which word is sent is determined by monitoring the relative phases of the elastic store write and read addressing.

The output bit stream passes through a one-cell scrambler to remove any preponderance of 1's or 0's and is converted to B6ZS format for transmission on the T2 line. The scrambling is performed by delaying the output through a one-cell shift register, the input of which is the modulo-2 sum (exclusive-OR function) of the input and the output of the shift register.

### 3.4 *The Receiving Portion Digital Interface*

After clock recovery in a T2 receiver the unipolar signal is recovered from the B6ZS input signal and is descrambled to yield the

108-bit words clocked at a 6.312-MHz rate (see Fig. 17). The descrambler contains a one-cell shift register the input and output of which are applied as inputs to an exclusive-OR logic circuit to derive the output data stream. The bit stream is passed through a 12-cell store using a write clock that is obtained by analysis of the 108-bit words and which clock only DPCM information bits into the store. The signaling bits are recovered and the digital store framing information is used to ensure that the four 3-bit words are placed in the decoder digital store in the same position that they were taken out of the coder digital store.

A jitter-free regenerated clock is needed for reading data from the store and for use in the series-to-parallel converter and the decoder; the 6.048-MHz VCO clock is locked to the incoming signal by means of a phase-lock loop which includes the divide-by-12 circuits used for addressing the digital store. After passing through the digital store, the bit stream is processed in a serial-to-parallel converter which provides the required inputs for the DPCM decoder.

### 3.5 The DPCM Decoder and Output Interface

The decoder consists of a digital-to-analog circuit exactly the same as that used in the coder and shown in the lower portion of the circuit in Fig. 16. Since the circuits at the two ends of the digital transmission system are of identical design, an accurate representation of the input signal is assured. Although not shown in the figure, a resistive leak is provided across the integrator capacitor in each circuit; the leak is necessary to ensure decay of the effects of digital errors and mistracking and a time constant value of 27 microseconds is used.

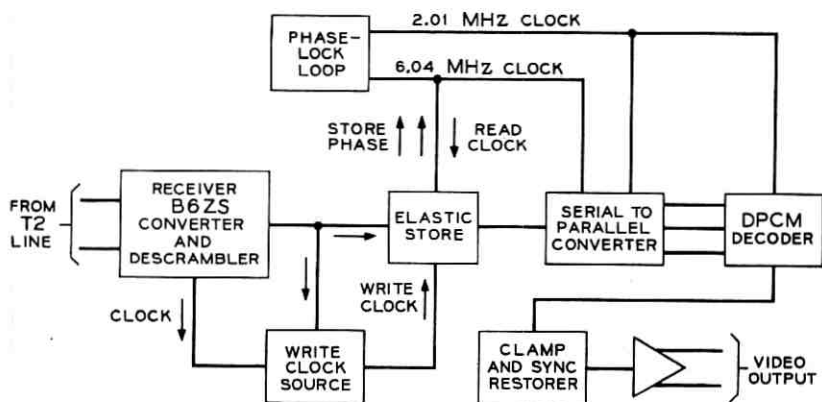


Fig. 17—Block diagram of the receiving circuit.

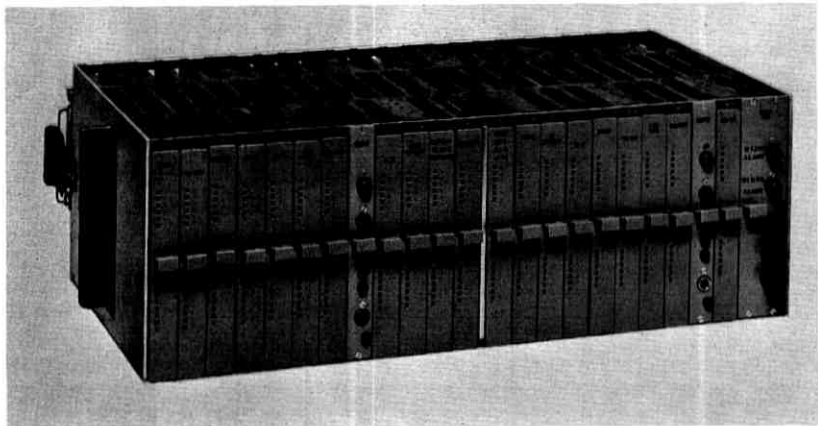


Fig. 18—The codec shelf.

At the output of the decoder, data mode signals are adjusted to the correct level and then passed to the balanced amplifier that drives the 100-ohm line. For video signals some additional processing of the video signal is necessary after pre-emphasis because the station set dc restorer would be adversely affected by the quantizing noise at the tips of the sync pulses which might cause low-frequency beats in the *Picturephone* clamp circuits.<sup>20</sup> Accordingly a circuit is included that attenuates the high frequency quantizing noise during the occurrence of sync pulses.

### 3.6 Equipment Design

The circuits for the two-way codec terminal are placed on 21 printed circuit packs located in a 7-inch-high shelf as shown in Fig. 18. Four of the cards carry the circuits that do the actual coding and decoding of the video signal, two of eight test cards provide power and alarm circuitry and the remaining cards provide the interface and synchronization circuits.

## IV. EVALUATION

Many codecs have been built and tested; six were in service on the TD-2 radio relay route between New York and Pittsburgh for a 6-month period in 1969. Even with the additional noise introduced by the route, a subjective evaluation of the quantizing noise showed less than 1 dB of degradation in the field.

After the trial some changes were made in the design used in the

first service model codecs to provide more flexible usage and to conform to new *Picturephone* standards. Circuits were added to the input and output interfaces to allow 460.8 kb/s data to be processed, to activate the appropriate signal paths and to distinguish between video and data signals. The new *Picturephone* set employs crystal controlled frequency standards allowing the use of the small buffer stores and the simple phase-lock loops that have been described above. The sync format has also been changed. However, no fundamental change was made to the coding algorithm or to the digital interface. Work is continuing to seek improved coding techniques for use in the *Picturephone* network.

#### V. ACKNOWLEDGMENT

The authors acknowledge that the work involved in the development of the *Picturephone* codec was done by all the members of the Digital Transmission Design Department.

#### REFERENCES

1. Cutler, C. C., "Differential Quantization of Communication Signals," Patent No. 2,605,361, July 29, 1952 (applied for June 29, 1950).
2. de Jager, F., "Delta Modulation, A Method of PCM Transmission Using a 1-Unit Code," Phillips Res., No. 7 (December 1952), pp. 442-446.
3. Aaron, M. R., Fleischman, J. S., McDonald, R. A., and Protonotarios, E. N. "Response of Delta Modulation to Gaussian Signals," B.S.T.J., 48, No. 5 (May-June 1969), pp. 1167-1196.
4. Abate, J. E., "Linear and Adaptive Delta Modulation," Proc. IEEE, 55, No. 3 (March 1967), pp. 298-308.
5. Oliver, B. M., "Efficient Coding," B.S.T.J., 31, No. 4 (July 1952), pp. 724-750.
6. Kretzmer, E. R., "Statistics of Television Signals," B.S.T.J., 31, No. 4 (July 1952), pp. 751-763.
7. Harrison, C. W., "Experiments with Linear Prediction in Television," B.S.T.J., 31, No. 4 (July 1952), pp. 764-783.
8. Schreiber, W. F., "The Measurement of Third Order Probability Distributions of Television Signals," I.R.E. Trans. on Inform. Theory, IT-2, No. 3 (September 1956), pp. 94-105.
9. Graham, R. E., "Predictive Quantizing of Television Signals," IRE Westcon Conv. Record, Part 4, (August 1958), pp. 147-156.
10. O'Neal, J. B., Jr., "Delta Modulation Quantizing Noise Analytical and Computer Simulation Results for Gaussian and Television Inputs," B.S.T.J., 45, No. 1 (January 1966), pp. 117-141.
11. O'Neal, J. F., Jr., "Predictive Quantizing Systems (Differential Pulse Code Modulation) for the Transmission of Television Signals," B.S.T.J., 45, No. 5 (May-June 1966), pp. 689-721.
12. Protonotarios, E. N., "Slope Overload Noise in Differential Pulse Code Modulation Systems," B.S.T.J., 46, No. 9 (November 1967), pp. 2119-2161.
13. Brainard, R. C., "Subjective Evaluation of PCM Noise-Feedback Coder for Television," Proc. IEEE, 55, No. 3 (March 1967), pp. 346-353.
14. Mounts, F. W., "Conditional Replenishment: A Promising Technique for Video Transmission," Bell Laboratories Record, 48, No. 4 (April 1970), pp. 110-121.



15. Fano, R. M., *Transmission of Information, A Statistical Theory of Communications*, Cambridge, Massachusetts: The M.I.T. Press, 1961.
16. Wozencraft, J. M., and Jacobs, I. M., *Principles of Communication Engineering*, New York: John Wiley & Sons, Inc., 1967.
17. Johannes, V. I., Kaim, A. G., and Walzman, T., "Bipolar Pulse Transmission with Zero Extraction," *IEEE Trans. Commun. Technology, COM-17*, No. 2 (April 1969), pp. 303-310.
18. Brown, H. E., "The Picturephone® System: Transmission Plan," *B.S.T.J.*, this issue, pp. 351-394.
19. Bender, E. C., and Howson, R. D., "The Picturephone® System: Wideband Data Service," *B.S.T.J.*, this issue, pp. 667-681.
20. Doba, S., Jr. and Richey, J. W., "Clampers in Video Transmission," *Trans. AIEE*, 69, P1-1 (December 1960), pp. 477-287.



## The *Picturephone*<sup>®</sup> System:

# A Digital Transmission System for TD-2 Radio

By C. W. BRODERICK

(Manuscript received September 2, 1970)

*A terminal for transmitting digital signals over TD-2 microwave radio relay routes has been designed and evaluated in a recent field trial. This paper details some of the radio system characteristics which influenced the design, describes the major features of the terminal, and reports the trial results. The terminal will be used in the toll network to provide inter-city Picturephone<sup>®</sup> trunks and data circuits.*

### I. INTRODUCTION

The TD-2 Radio Relay System is the major trunk carrier system in the Bell transcontinental network, interconnecting every major metropolitan center in the United States.<sup>1</sup> Therefore, as digital transmission grows, it is natural to consider how TD-2 might be used to distribute these signals.

A terminal that applies the digital bit stream to TD-2 has been designed to transmit at a rate of 20.2 megabits per second (Mb/s) using a four-level, 10.1-megabaud pulse format to modulate standard TD-2 terminal equipment. The input is arranged to accept up to three 6.312-Mb/s bipolar streams from such sources as *Picturephone* coders and data sets. Design features include the use of pulse stuffing for multiplex synchronization, dedicated pulses for timing recovery, a transversal filter for optimizing the received eye pattern and the use of self-synchronizing scramblers for testing.<sup>2,3</sup> The signal is regenerated at baseband about every 400 miles to limit the accumulation of unequalizable pulse distortion.

A number of digital terminals have been constructed and tested in conjunction with the New York-Pittsburgh *Picturephone* product trial. Test results on this 450-mile trial route, using one regenerator station, have indicated that performance of better than one error in  $10^{10}$  bits,

in the absence of significant fading, has been realized. The error rate increases to about one error in  $10^6$  bits when the received signal on any radio hop fades to the point where transmission is switched automatically to a protection radio frequency channel.

## II. TD-2 MICROWAVE SYSTEM

The TD-2 microwave radio system has been described previously.<sup>1</sup> It operates in the 3,700- to 4,200-MHz band, with the 20-MHz-wide channels spaced 20 MHz apart. The system employs frequency modulation of an IF carrier centered at 70 MHz, prior to conversion to the microwave spectrum. The baseband frequency response is constant to within 0.5 dB to about 5 MHz for a 300-mile system.

The reliability of the radio channel is maintained to a high degree through the availability of automatic switching to standby protection channels in the event of a signal-to-noise ratio (S/N) below minimum requirements.<sup>4</sup>

## III. DESIGN CONSIDERATIONS

It was desired to use the existing TD-2 microwave equipment including the 3A FM terminals and to make the digitally modulated microwave channel compatible with regular analog message and television signals on adjacent microwave channels.<sup>5,6</sup> This makes it possible to use the existing TD-2 nationwide network in providing *Picturephone* toll trunking and digital data circuits and it imposes the following signal constraints:

(i) The baseband signal must not interfere with the radio protection switching system which uses a 7-MHz pilot tone and a 9-MHz noise slot. This is accomplished by using a baseband transmitter filter whose cutoff frequency is below 9 MHz.

(ii) The IF signal spectrum must not interfere with the noise slot frequency or the top message frequency of the adjacent channel. This limits the peak frequency deviation.

(iii) The limiting S/N is determined by the protection switching system initiator setting used for analog message service.

(iv) The digital signal must be regenerated to limit the unequalizable baseband distortion encountered in RF transmission.

The coded *Picturephone* signal has a 6.312-Mb/s rate.<sup>7</sup> This rate was chosen to match a standard line rate which is evolving as part of the Bell System's digital transmission network and which will be used by the T2 digital line. The digital transmission system capacity therefore was chosen in increments of this 6.312-Mb/s rate.

The initial use of the terminals, in support of the *Picturephone* product trial between New York and Pittsburgh required an error rate of  $< 10^{-6}$ . A plot of the required S/N for fixed error rates and of the available S/N due to the above restrictions, both as a function of the system bit rate, is shown in Fig. 1. The available S/N is calculated at the fading threshold of the radio protection switching system. To meet the signal constraints above, the peak frequency deviation and therefore the available S/N must decrease as the system bit rate increases.

The digital signal eye pattern distortion increases with increasing system bit rate due to the sharper rolloff in the baseband filtering. Therefore, the required S/N for constant error rate increases as the system bit rate increases. From the intersection of the required S/N curve and the available S/N curve, it can be seen that the system capacity is about 21 megabits for an error probability of  $10^{-6}$  at the fading threshold of the radio protection switching system. The digital terminal input therefore was set at three 6.312-Mb/s lines. The system parameters are specified in Table I.

The controlling signal constraint at the intersection in Fig. 1 is IF signal spectral overlap onto the noise slot frequency of the adjacent

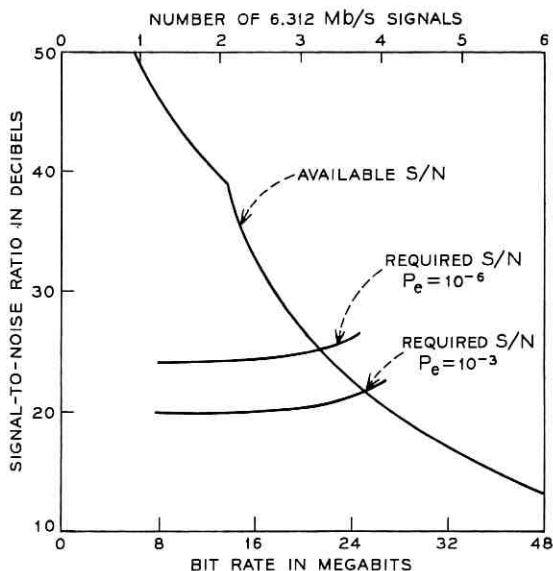


Fig. 1—Available S/N and required S/N for constant probability of error,  $P_e$ , versus system bit rate.

channel. This is a form of near-end adjacent channel interference involving a receiver at a center frequency 20 MHz away from a transmitter. The interference path includes the side-to-side coupling of adjacent horn reflector antennas and the loss in waveguide runs from transmitter to antenna and antenna to receiver. The coupling loss is typically about 100 dB which results in a reduction of fade margin equal to 0.8 dB. In some circumstances siting difficulties lead to coupling losses that are several dB smaller, and therefore a further reduction in fade margin.<sup>8</sup>

#### IV. TERMINALS

##### 4.1 *Transmitting Terminal*

A simplified block diagram of the transmitting terminal is shown in Fig. 2 with typical waveforms. The digital transmitter accepts three 6.312-Mb/s pulse streams in parallel and multiplexes them into a single serial pulse stream.

TABLE I—DIGITAL TRANSMISSION SYSTEM PARAMETERS

1. Transmitting Digital Terminal	
Input	Three 6.312-Mb/s lines in bipolar format
Output	10.1-megabaud, 4-level signal band limited to 7.5 MHz at -15.7 dBm on 124 $\Omega$ balanced pair
2. Receiving Digital Terminal	
Input	10.1-megabaud, 4-level signal at +0.4 dBm on 124 $\Omega$ balanced pair
Peak baseband signal-to-rms noise ratio (worst case) at detector input	27 dB (includes effect of receiver filter)
Output	Three 6.312 Mb/s lines in bipolar format Error rate: $10^{-10}$ nominal; $10^{-8}$ at protection switching point
3. TD-2 Microwave Radio	
RMS carrier-to-noise ratio at protection switching point (37.5-dB fade)	96 dB/Hz
99% power bandwidth for digital modulation	12 MHz

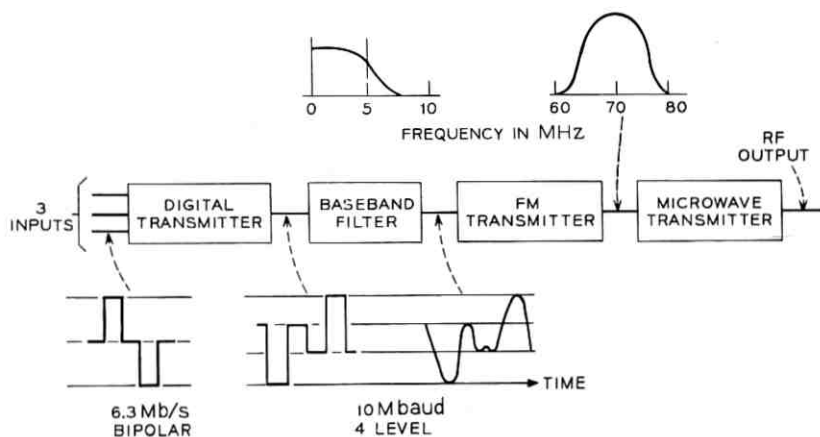


Fig. 2—Digital transmission over microwave radio transmitting terminal.

The output of the digital transmitter is a 10.1-megabaud, four-level signal which is made up of information pulses, framing pulses, and synchronization pulses. This composite signal is then shaped by the baseband filter into the signal which will deviate the FM transmitter. The output spectrum of the FM transmitter, centered around 70 MHz, is shown in a simplified form.

#### 4.2 Receiving Terminals

At the receiving terminal, the digital signal is recovered from the FM terminal receiver at baseband, filtered, equalized and demultiplexed into three 6.312-Mb/s signals. A simplified block diagram of the receiving terminal, together with its appropriate waveforms, is shown in Fig. 3. The functional operation of the receiver is essentially the inverse of the transmitter.

#### 4.3 Filters

The overall shaping and filtering of the four-level, 10.1-megabaud baseband signal is obtained by splitting the baseband filtering and locating a portion of it in the transmitting terminal to band limit the signal and the remainder in the receiving terminal to limit noise. The combined transmission characteristic of the baseband filters produces a pulse with a raised cosine spectrum which has a high tolerance to interference.<sup>9</sup>

The transmitter baseband filter characteristic is given by

$$H_t(f) = \begin{cases} \frac{\pi f/f_B}{\sin(\pi f/f_B)}, & |f| \leq f_{B/2} - f_x; \\ \frac{\pi f/f_B}{\sin(\pi f/f_B)} \cos\left(\frac{\pi}{4} \frac{f - (f_{B/2} - f_x)}{f_x}\right), & f_{B/2} - f_x < |f| < f_{B/2} + f_x. \end{cases}$$

The receiver baseband filter characteristic is given by

$$H_r(f) = \begin{cases} 1, & |f| \leq f_{B/2} - f_x; \\ \cos\left[\frac{\pi}{4} \frac{f - (f_{B/2} - f_x)}{f_x}\right], & f_{B/2} - f_x < |f| < f_{B/2} + f_x; \end{cases}$$

where  $f_B = 10.1$  MHz.

For the 50 percent rolloff filter used in these terminals,  $f_x$  equals  $f_{B/4}$ . The  $x/\sin x$  term in the transmitter filter characteristic is a correction for the spectrum of the square-wave input.

#### V. BASIC FUNCTIONS—TRANSMITTER

The digital transmitter block of Fig. 2 is shown in expanded block diagram form in Fig. 4. The functions of bit stream synchronization, multiplexing, and modulation shown in Fig. 4 are described below.

To time multiplex the three independent 6.312-Mb/s inputs, each input must be synchronized to the same clock frequency. This is accomplished by the method of pulse stuffing. Each input is written into a separate eight-cell elastic store under the control of an input

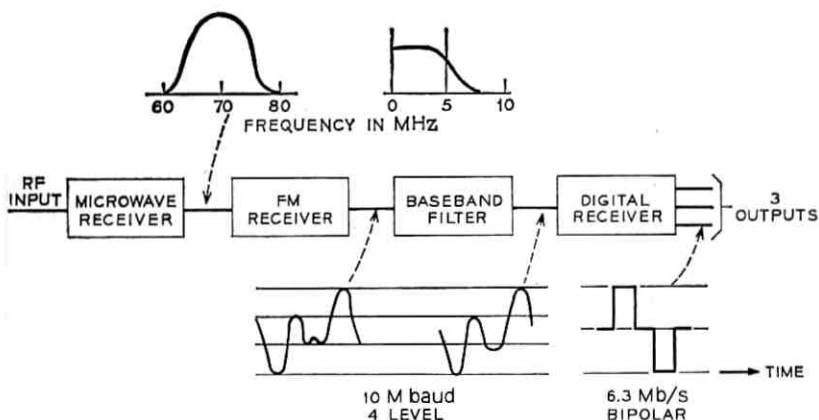


Fig. 3—Digital transmission over microwave radio receiving terminal.



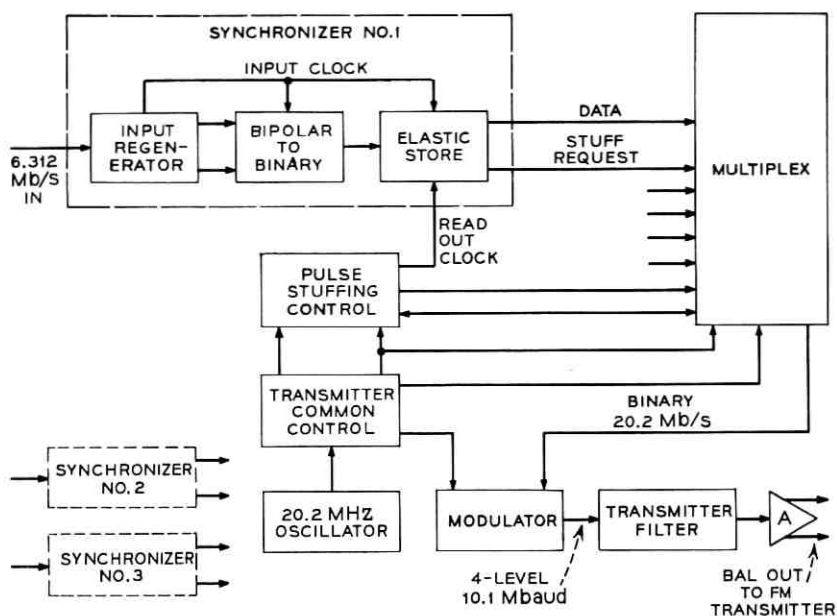


Fig. 4—Block diagram of digital transmitter.

clock derived from the average frequency of the respective input signal. The input frequency is  $6.312 \text{ MHz} \pm 30 \text{ ppm}$  which is a tolerance of about  $\pm 190 \text{ Hz}$ . The reading function for each synchronizer is performed by one of three read clocks generated in the pulse stuffing control. Each of these read clocks over a long period of time has exactly the same number of clock pulses as there are information pulses written into the storage cells of the associated elastic store. Furthermore, each read clock has been blanked appropriately to allow the insertion of the framing and synchronizing pulses. Sixteen bits out of every 256-bit frame are allocated for these purposes. The nominal pulse stuffing rate is 500 stuff pulses per second. This rate is variable from a minimum of zero to a maximum rate of one stuff every 12 frames. The actual synchronization is performed by the pulse stuffing control through the monitoring of stuff requests generated by nearly empty elastic stores. These stuff requests reach the pulse stuffing control via the multiplex. All of the periodic waveforms required to perform the functions of synchronizing, multiplexing and modulating are generated in the transmitter common control. The basic system clock frequency is  $20.2 \text{ MHz} \pm 30 \text{ ppm}$  and is generated by the 20.2-MHz crystal oscillator.

The multiplex accepts the three properly synchronized information

pulse streams with their associated stuff request pulses. It also receives the required periodic waveforms from the pulse stuffing control and the transmitter common control and time multiplexes them together with the information pulses and stuff pulses into a 20.2-Mb/s binary stream. Included in this stream are the necessary control signals to allow the receiver to remove the added stuff pulses.

The final digital operation in the chain is to modulate the 20.2-Mb/s binary signal to a four-level, 10.1-megabaud signal. This takes place in two steps. The 20.2-Mb/s binary signal is first converted into two parallel 10.1-Mb/s binary streams which are then converted into a single, four-level 10.1-megabaud signal. A simplified schematic diagram of the modulator is shown in Fig. 5. The serial bit pairs,  $B_1B_2$ , are mapped to pairs of parallel bits,  $D_1/D_2$ , through the functions;

$$D_1 = \overline{B_1}$$

and

$$D_2 = \overline{B_1 \oplus B_2}.$$

The parallel bit pairs then control the emitter-coupled transistors which route the proper amount of current from the current generators into the load resistor to produce the four-level output signal.

The digital signal format at the binary output of the multiplex and the corresponding four-level signal out of the modulator are illustrated in Fig. 6. The contents of one frame are shown, including dedicated

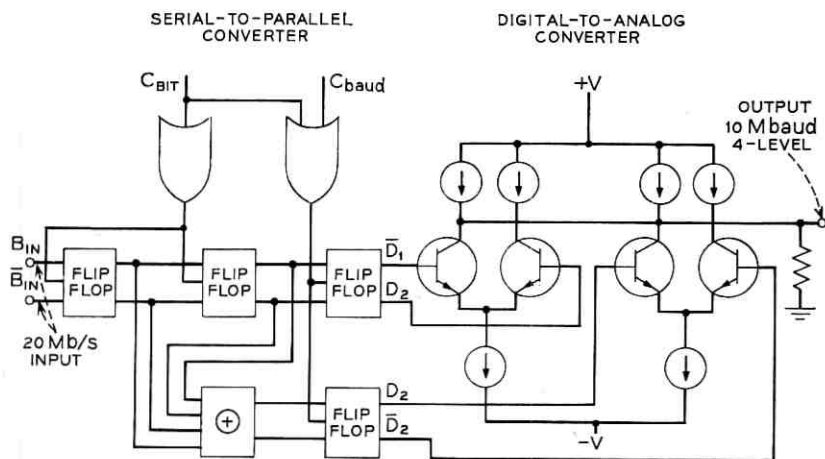


Fig. 5—Modulator.  $C_{\text{BAUD}} = 10.1\text{-MHz}$  baud clock;  $C_{\text{BIT}} = 20.2\text{-MHz}$  bit clock;  $B_{\text{IN}}, \overline{B_{\text{IN}}} = 20.2\text{-Mb/s}$  binary bit stream and its complement;  $D_1 = \overline{B_1}$ ;  $D_2 = \overline{B_1 \oplus B_2}$ .

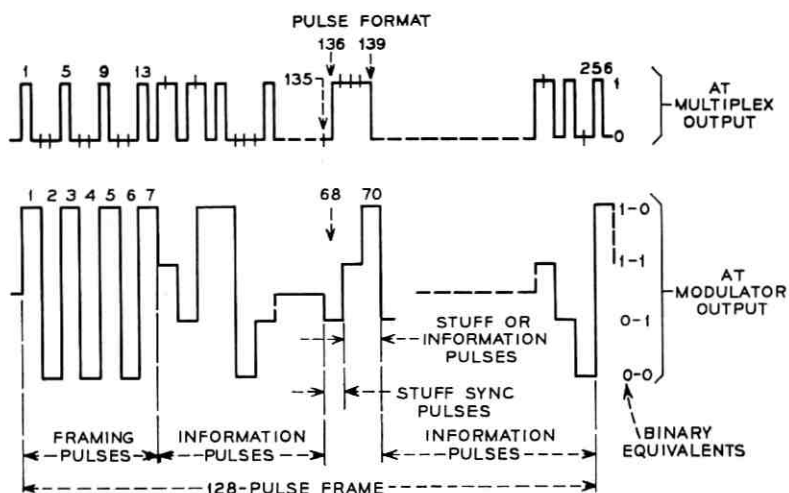


Fig. 6—Pulse format of digital signal.

framing pulses, stuff sync pulses, stuff pulses, and information pulses. The stuff sync pulses of 12 successive frames make up a triply redundant word which indicates to the receiver when and where stuff pulses are inserted for each of the three information channels. From the modulator, the signal passes through the transmitting filter, which restricts the baseband frequencies and provides the desired pulse shaping, and then to an FM transmitter through an unbalanced-to-balanced amplifier.

#### VI. BASIC FUNCTIONS—RECEIVER

The digital receiver block of Fig. 3 is shown in expanded block diagram form in Fig. 7. At the digital receiver, the four-level, 10.1-megabaud signal, from the FM receiver, is converted to an unbalanced signal and then passes first through the receiver baseband filter and into the transversal filter. The baseband filter, together with the filter in the transmitting terminal, provides optimum shaping of the baseband response characteristic. The transversal filter simplified schematic is shown in Fig. 8. The input signal passes through a 12-section tapped delay line. The gain of each tap, except for the center tap, is manually adjustable so that the output signal  $S_0(t)$  is given by

$$S_0(t) = \sum_{n=-6}^{+6} K_n S_i(t - n\tau),$$

where

$$K_0 = 1, \quad -1 < K_n < +1$$

and

$$S_i(t - n\tau)$$

is the input signal at the  $n$ th tap.

Adjustment of the tap gains,  $K_n$ , compensates for the nonfading distortion encountered by the signal in transmission over the radio channels. The equalized output signal drives the digital section of the receiver.

The operation of the receiver is, in general, the inverse of the transmitter. The receiver converts the 10.1-megabaud, four-level input signal into three 6.312-Mb/s pulse streams by first demodulating the signal into a 20.2-Mb/s binary data stream, and then separating and adjusting the timing of, or desynchronizing, the three individual output signals.

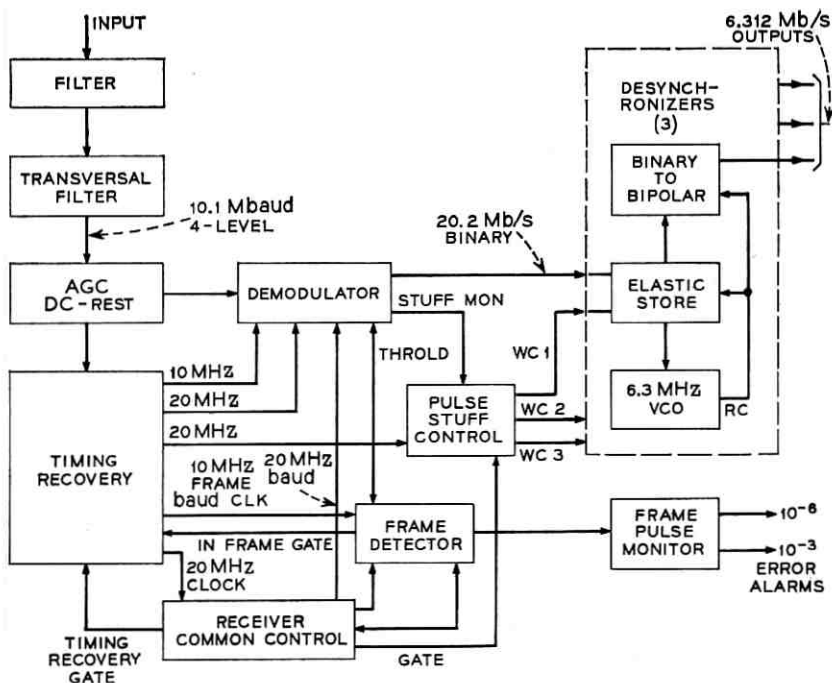


Fig. 7—Block diagram of receiving terminal.

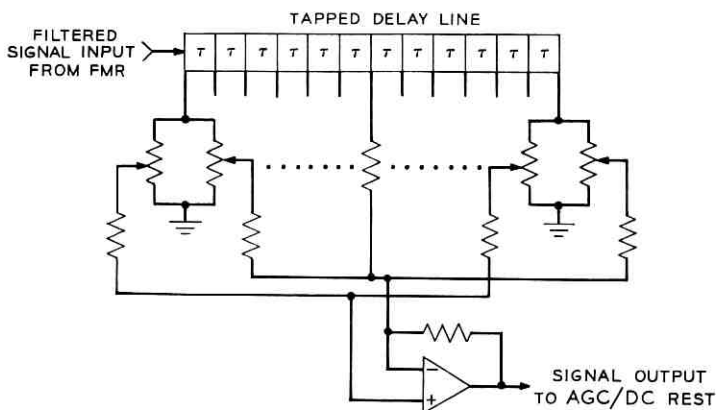


Fig. 8—Transversal filter.

Demodulation is accomplished through the AGC/DC restoration, timing recovery, demodulator, frame detector, and receiver common control circuitry. The AGC/DC restoration circuit reduces the effect of input signal level variations and assures that the signal is at the proper dc level.

The timing recovery circuit produces 10.1-MHz sinewave outputs which are synchronous with the 10.1-megabaud input signal. Timing is initially recovered using the complete signal. However, when the receiver is in frame, only selected zero crossings of the dedicated framing pulses (Fig. 6) are used. This results in a more stable recovered clock because the dedicated pulses always have the same pattern. This clock is converted to 20.2 MHz by the frequency doubler circuitry.

The demodulator converts the 10.1-megabaud, four-level signal into a 20.2-Mb/s binary data stream. A simplified schematic is shown in Fig. 9. The 10.1-MHz sample clock is converted into a narrow strobe pulse of about 15-ns width that operates the sample-and-hold flip-flops at the output of the threshold detectors. There are three threshold detectors, each of which consists of two stages of emitter-coupled transistor pairs. High-speed silicon transistors ( $f_t = 4$  GHz) are used in the last stage to minimize errors caused by timing inaccuracies. The outputs of the sample-and-hold flip-flops are converted logically in the parallel-to-serial converter to a 20.2-Mb/s binary data stream.

The relative phase of the 10.1-MHz sample clock and the 10.1-megabaud signal is adjustable to permit sampling to occur at the minimum distortion point of the eye pattern.

The frame detector is used to establish proper framing of the receiver common control. When the framing pulses are in the demodulator sample-and-hold circuit, a fixed pattern appears at the frame detector. To establish initial framing when the system is put into service, or to reinitiate framing when an outage occurs, the dedicated pulse pat-

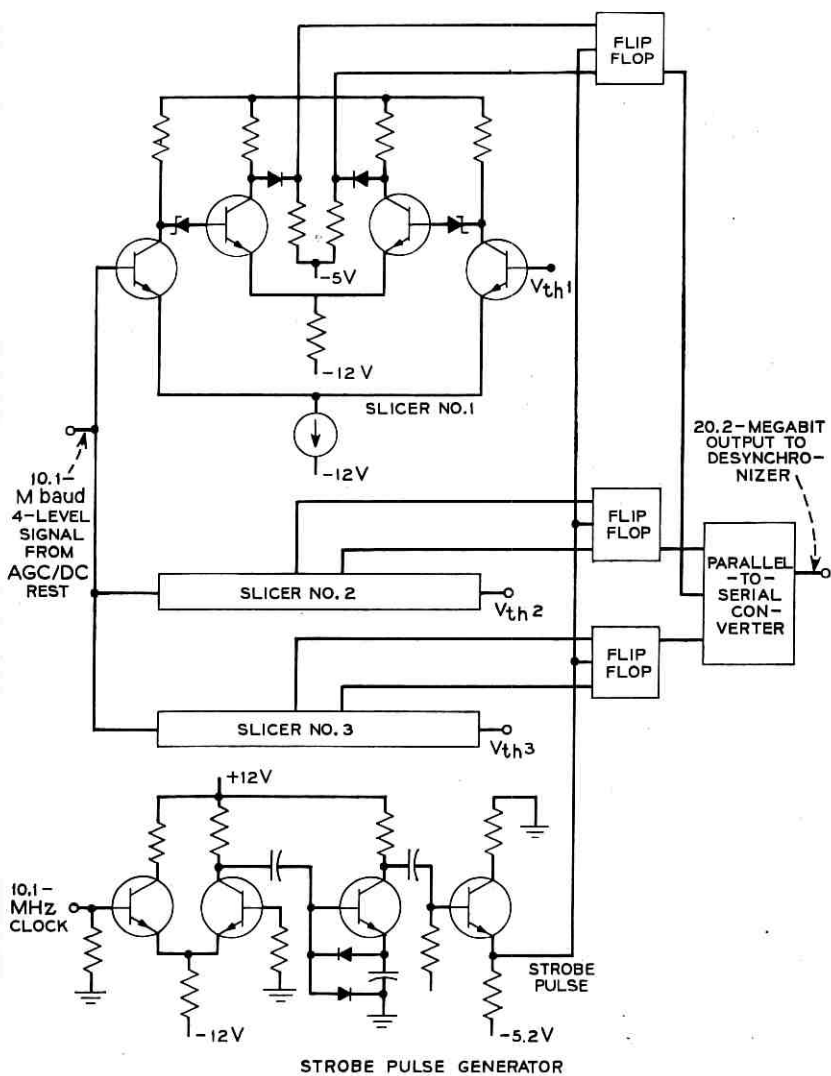


Fig. 9—Demodulator.

tern must be detected correctly in two successive signal frames. The master counter in the receiver common control will then be automatically set to its proper phase with respect to the input signal stream. In the absence of errors in the incoming four-level pulse stream, the in-frame condition is achieved in no more than 25 microseconds. Once the in-frame condition is established, the time of occurrence of the dedicated pulse pattern as seen by the frame detector is compared with the time of occurrence of a frame sync pulse from the receiver common control. If they are in phase, a signal appears on the in-frame gate lead to the timing recovery. If they are not in phase during four successive signal frames, a frame reset pulse resets the common control and continues to reset until the correct framing pattern is noted again in two successive frames. In addition, when an error occurs in the framing pulses, a signal is sent to the framing pulse error monitor.

The framing pulse error monitor keeps track of the rate of errors of the framing pulses. When the framing pulse error rate is equivalent to a data error rate less than one in  $10^6$  bits, performance is considered acceptable. When the equivalent error rate becomes greater than one in  $10^6$  bits, an alarm is activated as described in Section 10.2.

Circuits used for separating and adjusting the timing of the three channels are the pulse stuffing control, receiver common control, and three sets of desynchronizers each set consisting of an elastic store, voltage-controlled oscillator, and binary-to-bipolar converter.

The purpose of the pulse stuffing control in the receiver is to provide the three write clock (WC) signals, each clock having pulses corresponding to a different information channel. In addition, the pulse stuffing control determines when a channel has an added stuff pulse to be ignored. Information as to whether a stuff has occurred is contained in two dedicated "Sync" bits of the 20.2-Mb/s binary data stream. When the presence of a stuff pulse is indicated by the sync bits, the appropriate pulse of the output clock is blanked in the time slot where the stuff pulse has been inserted by the digital transmitter. Only one channel will have been stuffed in any single frame.

Data is written into the elastic store of desynchronizer channel No. 1 by WC1. Thus only those bits belonging to channel No. 1 get into this elastic store. The data is then read from the elastic store by the smoothed read clock generated by the voltage-controlled oscillator. This is a crystal controlled oscillator, the frequency of which is locked to the average frequency of WC1. The smoothed binary data stream out of the elastic store is finally converted back to bipolar format.

## VII. ERROR-RATE MEASUREMENT

To measure the system error performance, a pseudo-random word generator called a scrambler is used with a complementary descrambler. As shown in Fig. 10, the scrambler consists of a 15-stage shift register with feedback connections that cause it to generate a word of 32,767 bits in length when the input is "0." The descrambler has the same basic configuration and performs the inverse function, i.e., when the descrambler input is the 32,767-bit word, the output of the descrambler is "0." Errors in the 32,767-bit word will cause a pulse to appear at the descrambler output.

The frequency of operation of the scrambler-descrambler combination is determined by the clock frequency that is supplied to it. To test the overall system error rate, a 20.2-MHz clock with appropriate blanking is used and the scrambler output drives the transmitter multiplex. The receiver demodulator drives the descrambler and the output of the descrambler is monitored to measure the error rate. To test a single 6.312-Mb/s channel, an appropriately blanked 6.312-

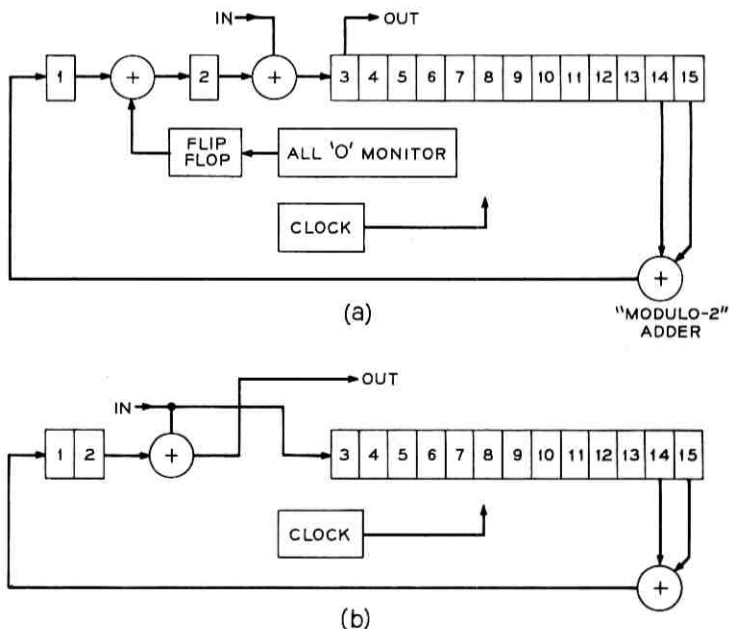


Fig. 10—Word generator: (a) scrambler, and (b) descrambler.



MHz clock is used and the scrambler replaces the elastic store in the channel being tested. The receiver elastic store in the same channel is replaced by the descrambler and the output of the descrambler is monitored to measure the single-channel error rate. Since this last test does not check the pulse-stuffing operation in the channel being tested, a third test mode, which uses a scrambler driven by an asynchronous 6.312-MHz clock, is used. The output of this test source drives a single 6.312-Mb/s channel, replacing the normal input. The descrambler in the receiver is used in the binary-to-bipolar converter position. In this way the pulse stuffing operation of a single 6.312-Mb/s channel can be tested.

#### VIII. DISCUSSION OF FIELD TEST RESULTS

A *Picturephone* product trial was conducted between New York and Pittsburgh from February 1969 to August 1969. As part of the product trial, digital terminals were installed at New York and at Pittsburgh with a digital regenerator at Alma, New York. The digital terminals were in operation from July 1968 to August 1969 and during the first 6 months of this period, before the product trial started, error-rate measurements were made. Data accumulated during the test period have indicated that the performance has exceeded the original design objectives. Error performance over several periods of two to three days has indicated an error rate of  $10^{-9}$  to  $10^{-11}$ . Over 99.5 percent of the 10-second samples recorded were error free during a total of over 200 hours of measuring time. Among the small percentage of samples containing errors, about 90 percent contained five or less binary digit errors, which corresponds to an error rate of less than  $10^{-7}$  errors per bit.

A plot of actual error-rate results on the radio circuit between Pittsburgh and New York, obtained by injecting equivalent noise at baseband in the digital receiver to simulate a fading condition, is shown in Fig. 11. The center curve is the overall circuit with the regenerator in at Alma and equalized for overall optimum performance. The other curve shows the performance when the regenerator at Alma was bypassed. No change in equalization was made between the two conditions.

A photograph of an actual eye pattern is shown in Fig. 12. The eye closure due to intersymbol interference is clearly evident.

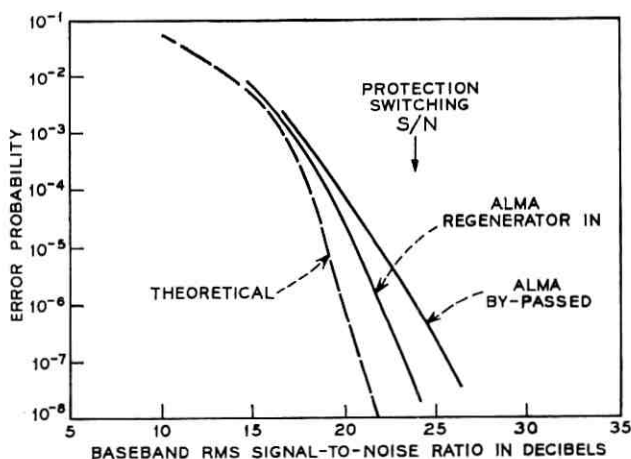


Fig. 11—Error rate versus baseband S/N.

## IX. PHYSICAL DESIGN ASPECTS

### 9.1 General Description

The digital terminal shelf includes a digital transmitter and receiver and is shown in Fig. 13. It is constructed of three 17.8-cm-high (7-inch) aluminum castings and is designed for relay rack-type mounting on standard 58-cm-wide (23-inch) duct type bays. Plug-in printed

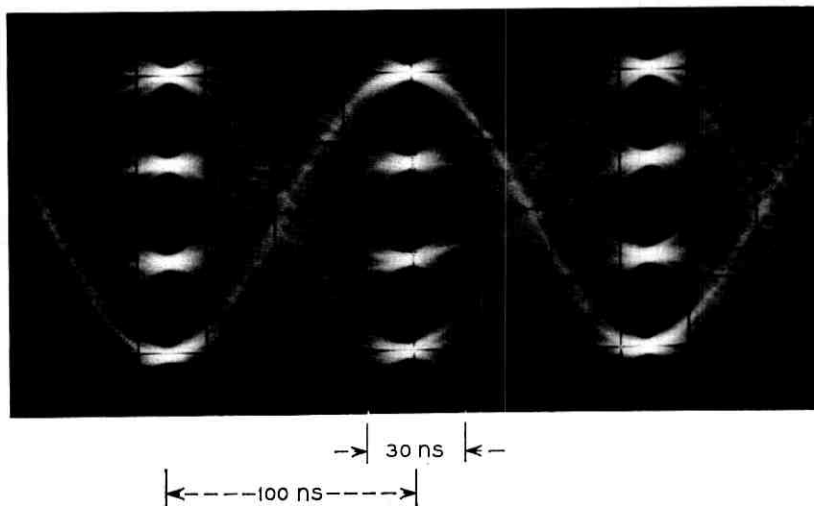


Fig. 12—Eye pattern.

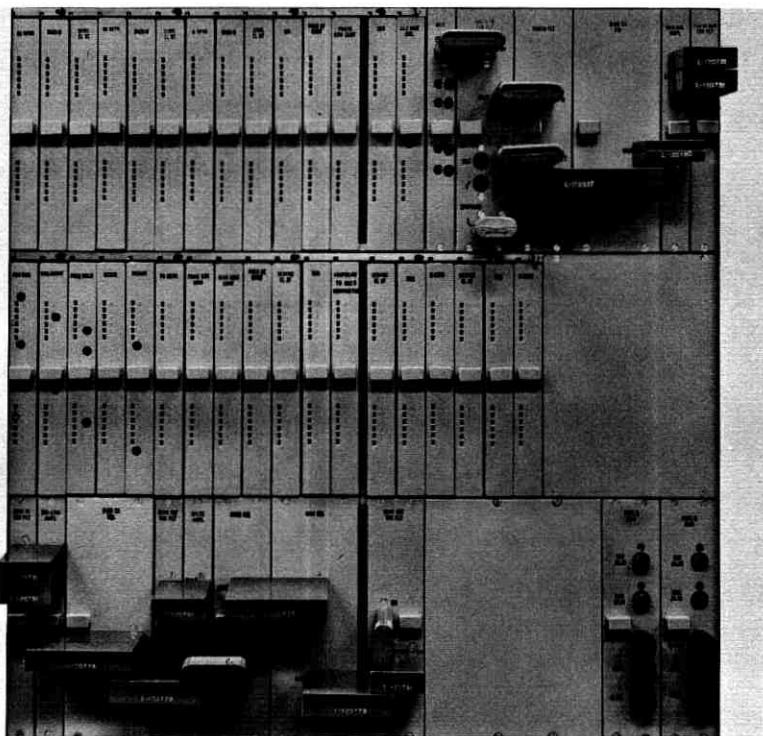


Fig. 13—Digital terminal shelf.

circuit boards with a nominal size of 16.5 cm by 27.9 cm (6.5 inches by 11 inches) mounted with horizontal spacings in multiples of 2.16 cm (0.85 inches) are used for individual circuits.

### 9.2 Maintenance Considerations

Monitor jacks for the diagnosis of trouble and routine servicing are provided on the test and alarm panel. Manual switching of a failed digital terminal shelf is provided for at terminal locations. A failed digital regenerator may be bypassed at IF using a 223B switch.

Use of plug-in circuit packs permits quick repair of a shelf.

## X. POWER AND ALARM ARRANGEMENTS

### 10.1 Power Supplies

Three dc voltages are required for the operation of digital transmitters, receivers, and regenerators. They are +12V, -12V, and -5.2V.

An ac distribution panel located at the top of the bay feeds each power supply. To prevent the interruption of service resulting from the failure of a power supply, two supplies are provided for each voltage. They are wired in parallel via a diode bridge as shown in Fig. 14. A failed power supply unit may be replaced without disrupting service.

### 10.2 Alarm Arrangements

All alarm conditions register as a major alarm on the office alarm system. The bay alarm on the test and alarm panel is transferred to either a "DIG CKT" or an appropriate "PWR" alarm indication when the "ACO RST" button on the panel is operated.

A "PWR" alarm indication results from an overvoltage turn off of a dc supply or from the tripping of a power supply circuit breaker on the ac distribution panel.

A "DIG CKT" alarm indication is further subdivided into a major alarm or a minor alarm by an indication on the appropriate fuse and alarm circuit pack. A major alarm results when the signal error rate is greater than  $10^{-3}$  error/bit or from a blown fuse in the fuse and alarm circuit pack. A minor alarm results when the signal error rate is greater than  $10^{-6}$  error/bit.

## XI. CONCLUSION

A four-level, 10.1-megabaud digital transmission system consisting of a digital terminal transmitter, a digital regenerator and a digital

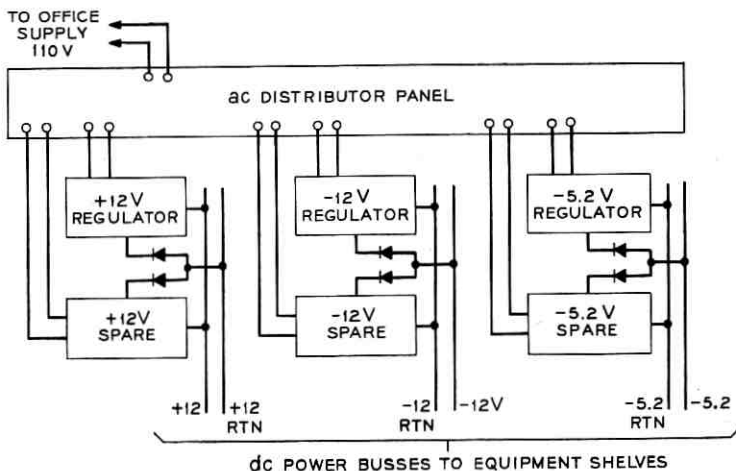


Fig. 14—Power supply arrangement.

terminal receiver capable of transmitting digital signals at a rate of 20.2 million bits per second over the TD-2 microwave system has been built and has undergone a successful field trial. An interim physical design permits early availability to the operating companies.

Operating over a radio route maintained under standard plant maintenance procedures, an error probability of the order of  $10^{-10}$  error/bit has been realized. Experience on this route indicates the compatibility of a broadband digital system with other analog systems on the same radio route.

## XII. ACKNOWLEDGMENT

The digital terminals were developed by the members of the Digital Transmission Design Department.

## REFERENCES

1. Roetken, A. A., Smith, K. D., and Friis, R. W., "The TD-2 Microwave Radio System," B.S.T.J., 30, No. 4 (October 1951), pp. 1041-1077.
2. Witt, F. J., "An Experimental 224 Mb/s Digital Multiplexer-Demultiplexer Using Pulse Stuffing Synchronization," B.S.T.J., 44, No. 9 (November 1965), pp. 1843-1885.
3. Savage, J. E., "Some Simple Self-Synchronizing Digital Data Scramblers," B.S.T.J., 46, No. 2 (February 1967), pp. 449-487.
4. Griffiths, H. D., and Nedelka, J., "100A Protection Switching System," B.S.T.J., 44, No. 10 (December 1965), pp. 2295-2236.
5. Barry, J. F., Gammie, J., Lentz, N. E., and Salvage, R. C., "3A FM Terminal Transmitter and Receiver," B.S.T.J., 47, No. 7 (September 1968), pp. 1423-1458.
6. Curtis, H. E., Collins, T. R. D., and Jamison, B. C., "Interstitial Channels for Doubling TD-2 Radio System Capacity," B.S.T.J., 39, No. 6 (November 1960), pp. 1505-1528.
7. Millard, J. B., and Maunsell, H. I., "The Picturephone® System: Digital Encoding of the Video Signal," B.S.T.J., this issue, pp. 459-479.
8. Hathaway, S. D., Hensel, W. G., Jordan, D. R., and Prime, R. C., "TD-3 Microwave Radio Relay System," B.S.T.J., 47, No. 7 (September 1968), pp. 1143-1188.
9. Bennett, W. R., and Davey, J. R., *Data Transmission*, New York: McGraw-Hill, 1965, chapters 5 and 7.



## The *Picturephone*<sup>®</sup> System:

# Mastergroup Digital Transmission on Modern Coaxial Systems

By J. F. GUNN, J. S. RONNE, and D. C. WELLER

(Manuscript received September 4, 1970)

*We describe in this paper a system currently under development that will adapt the L-4, and later the L-5, coaxial system for digital transmission in mastergroup bands. The terminal equipment will combine two 6.312 Mb/s digital signals and, hence, two Picturephone<sup>®</sup> signals in digital form for transmission on the coaxial systems along with analog message channels in other mastergroups. The overall system includes regenerative repeaters at intervals of not more than 300 miles to realize the error rate objective of  $2.6 \times 10^{-7}$ , or better, for distances up to 4000 miles.*

### I. INTRODUCTION

The introduction of intercity *Picturephone* service will stimulate growth in the amount of digital information requiring transmission over long distances. It is important to provide for this growth and for the increasing capacity required for high-speed data in an evolutionary manner. This has motivated the development of terminal equipment to adapt existing microwave radio<sup>1</sup> and analog coaxial facilities for the transmission of digital signals.

The L-4 coaxial line is an excellent medium for an efficient hybrid transmission system.<sup>2</sup> A high signal-to-noise ratio (S/N), excellent linearity, and well-defined and stable transmission characteristics make it very attractive for multilevel pulse transmission. A mastergroup band provides a convenient subdivision for frequency multiplexing high-speed digital and analog signals. The mastergroup band has a bandwidth of 2.52 MHz and conventionally provides for 600 frequency division multiplexed voice channels. A mastergroup band is also capable of carrying two time-division multiplexed 6.312-Mb/s signals and, thus, can handle two digitalized *Picturephone* signals. It is planned that the digital mastergroup signals will be transmitted over the L-5 coaxial system that is currently under development.

The maximum digital capacity of an L-4 route employing master-group digital equipment is equivalent to 108 digital *Picturephone* trunks. (Each of the six mastergroups per coaxial tube could carry two 6.312-Mb/s signals, and there are nine working two-way tube pairs per line, the remaining pair being reserved for protection.) The mastergroup digital capacity of the new L-5 system, employing a new 22-tube coaxial line and transmitting 15 mastergroups per tube, is 300 digital *Picturephone* trunks.

The L-mastergroup digital system, together with the microwave radio system, will provide the Bell System the long-haul, digital capacity it will need in the early 1970s. Digital signals can be routed to any of the metropolitan areas served by L-4 and L-5 coaxial cable routes.

The terminal equipment being developed to adapt L-mastergroup bands for digital transmission is the subject of this paper. The equipment employs the techniques of time division multiplexing, digital processing, vestigial sideband modulation, adaptive equalization, analog-to-digital conversion and digital regeneration.

## II. DESIGN OBJECTIVES

Design objectives have been established to ensure performance compatibility with the evolving digital hierarchy and with the L-4 and L-5 Coaxial Systems. The bit rate should be sufficient to accommodate an integral number of 6.312-Mb/s signals. The error rate for a 4000-mile mastergroup digital connection should not exceed  $2.6 \times 10^{-7}$  errors-per-bit more than 5 percent of the time.<sup>3</sup> The timing jitter of 6.312-Mb/s digital signals delivered from receiving mastergroup digital terminals should be no greater than that presented to the corresponding transmitting terminal. In addition, a mastergroup digital signal should not degrade the coaxial system noise performance established by the analog mastergroups. The digital equipment should be compatible with the standard blocking, branching, and frogging rules of the coaxial system and should also be compatible with standard facility switching, restoration, alarm, and maintenance procedures. Thus, the means should be provided for in-service monitoring of the digital facility that allows for quick identification and correction of trouble conditions.

## III. SIGNAL FORMAT

The class IV partial-response signaling format<sup>4,5</sup> exhibits a number of characteristics that make it well suited for use in the L-Master-



group Digital (LMD) System. Foremost is the power distribution as a function of frequency of the baseband signal which has nulls at dc and at a frequency equal to one-half the symbol rate. These nulls permit the transmission of both a carrier pilot and a timing pilot along with the signal, which greatly facilitates coherent demodulation and accurate timing of the received signal. Since the power distribution rolls off gradually near the band edges, the signal is less sensitive to the severe edge-of-band phase distortion introduced by the channel filters located in the LMD terminal.

The partial-response signaling rate can be nearly twice the bandwidth of the channel since satisfactory performance is not dependent upon the provision of a large (50 or 100 percent) excess bandwidth for roll-off (Nyquist) shaping as is often the case for a full-response format. Indeed, satisfactory performance has been obtained in partial-response digital transmission systems with no roll-off band.

The partial-response coding format has the further advantage that violations in the code can be monitored after the received signal has been sampled. The number of violations in the random data signal that occur per unit time provides a measure of the error performance for the system.<sup>6</sup> Coding of the signal in the LMD System is administered in a way that allows the error performance of each regenerative repeater section to be monitored independently. In this way, a trouble condition in any one of the repeater sections can be easily isolated to the appropriate section. The violation detector is also used in a control loop to correct for errors in the sampling time.

#### IV. INFORMATION RATE

The probability of error per bit for an  $n$ -level, class IV partial response signal, transmitted over a channel with optimum raised-cosine shaping and no intersymbol interference is

$$P_{IV} = \left( \frac{2}{\log_2 N} \right) \left( 1 - \frac{1}{N^2} \right) Q \left( \sqrt{\frac{3P_s}{2(N^2 - 1)P_n}} \right)$$

where

$$Q(x) = \frac{1}{\sqrt{2\pi}} \int_x^\infty \exp\left(-\frac{t^2}{2}\right) dt,$$

$P_s$  = channel signal power,

$P_n$  = channel noise power in the Nyquist band,

$\log_2 N$  = number of bits conveyed per pulse, and

$n = 2N - 1$  = number of coded levels transmitted.

The usual assumption has been made that only errors caused by gaussian noise which produce transitions to levels adjacent to the transmitted level need be considered. Thus, with Gray coding, the bit error rate will be  $1/\log_2 N$  times the symbol error rate as indicated in the expression for  $P_{IV}$ .

The LMD signal must be digitally regenerated in order to prevent noise and other forms of distortion from accumulating over a 4000-mile L-4 or L-5 system and becoming too great to permit efficient use of the mastergroup bandwidth. Because of their size and complexity, the regenerative repeaters must be located only at L-carrier mainstations. In a 4000-mile system, there will be approximately 16 LMD repeaters having a maximum separation of 300 miles (two mainstation sections).

The noise power falling within the Nyquist band at the input to a regenerative repeater,  $W_N(L)$  is given by

$$W_N(L) = W - 88 + \log(B_N/B_M) - 10 \log(L_c/L)$$

where

- $W$  = expected noise power in 4000 miles of L-4 (40 dBm0),
- $B_N$  = Nyquist bandwidth (2.215 MHz),
- $B_M$  = message channel bandwidth (3 kHz),
- $L_c$  = maximum LMD System circuit length (4000 miles), and
- $L$  = repeater spacing ( $\leq 300$  miles).

By substitution, the worst-case noise power at the repeater is  $-30$  dBm0.

The distribution of voltage for 600-message channels, after modulation into a mastergroup band, is approximately gaussian.<sup>7</sup> The voltage distribution for a mastergroup digital signal is bounded, and randomizing of the digital signal ensures that the peak power level in a mastergroup band is well controlled. With these considerations in mind, the power level of a mastergroup digital signal has been set equal to the average of a message mastergroup, or  $P_s = +11$  dBm0. This is a conservative choice intended to ensure that a mastergroup digital signal contribute no greater intermodulation noise than a message mastergroup. This establishes the worst-case S/N of the digital signal at the input to a digital repeater at 41 dB.

The  $2.6 \times 10^{-7}$  error rate objective for a 4000-mile connection corresponds to an error rate of approximately  $2 \times 10^{-8}$  per repeater section. The S/N that is required to just achieve  $2 \times 10^{-8}$  error performance can be computed for a 15-level partial response signal from the

expression for  $P_{IV}$  and is approximately 31 dB. The 10-dB difference between the computed signal-to-noise requirement and the 41-dB ratio expected is the margin that may be allocated to impairments arising from channel imperfections such as amplitude and phase distortion, timing jitter, signal-level variations, quadrature distortion, and imperfect multilevel to binary conversion. Thus, the 15-level class IV partial response signal combines adequate margin for transmission impairments with good bandwidth efficiency. The pulse rate has been set at 4.43 megabauds which results in a bit rate of 13.29 Mb/s. The LMD System can conveniently handle two 6.312-Mb/s signals plus additional capacity for multiplex framing, bit synchronization and miscellaneous control signals.

#### V. L-MASTERGROUP DIGITAL CHANNEL

The LMD equipment required to adapt the L-4 or L-5 Carrier System for transmission of mastergroup bandwidth digital signals is located only at mainstations. This arrangement not only facilitates maintenance but readily permits the multiplexing of both digital mastergroups and message mastergroups in the same cable. A block diagram of a typical digital channel is given in Fig. 1. The terminal digitally multiplexes two 6.312-Mb/s signals and modulates the combined signal into one of the L-4 mastergroup bands. At intervals up to 300 miles, the repeaters demodulate the line frequency signal to baseband where it is digitally regenerated and modulated back into an appropriate mastergroup band.

Figure 2 is a block diagram of the terminal and associated multiplex equipment. In the figure, the Combiners, the Terminal and the M2L are separated because in the actual equipment they are separate for maintenance reasons. The M2L time multiplexes (and demultiplexes) two 6.312-Mb/s signals. In the LMD Terminal, the digital processor maps the binary signal into 4-bit partial-response encoded words and presents them to the binary-to-multilevel converter at a rate of 4.43 megabauds. The binary-to-multilevel and multilevel-to-binary converters translate the 4-bit words to and from the 15-level, PAM format. The adaptive equalizer in the receiver provides mop-up equalization on a continuous basis. It corrects for residual amplitude and phase distortion in the terminal networks and also compensates for time varying distortion in the repeated line, coherent demodulators, and timing recovery circuits. A set of modulators and demodulators (modems) is used to frequency multiplex up to six LMD signals for application to the coaxial transmission system. Thus, message con-

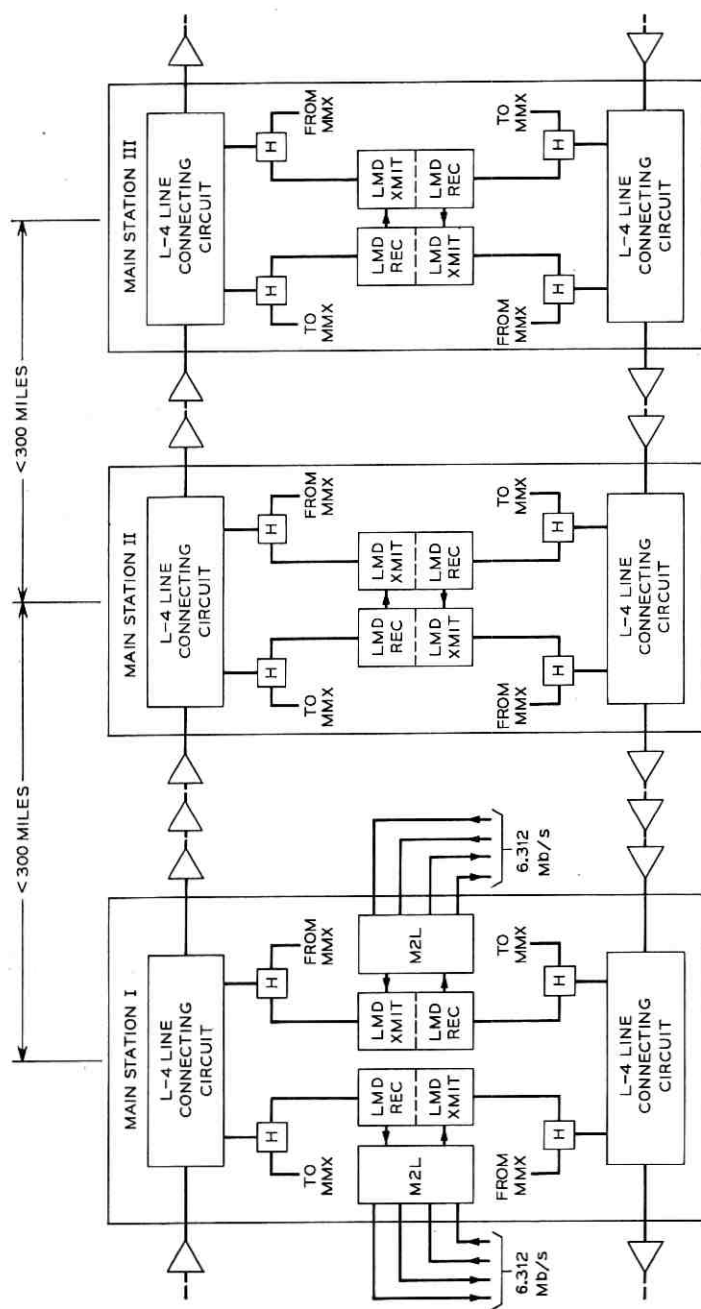


Fig. 1—Message mastergroups from the Mastergroup Multiplex (MMX) or digital signals (6.312 Mb/s) may be added or dropped at main stations. In this example, main stations II and III are regeneration points for an LMD channel, while main station I provides terminals for LMD channels in both directions.

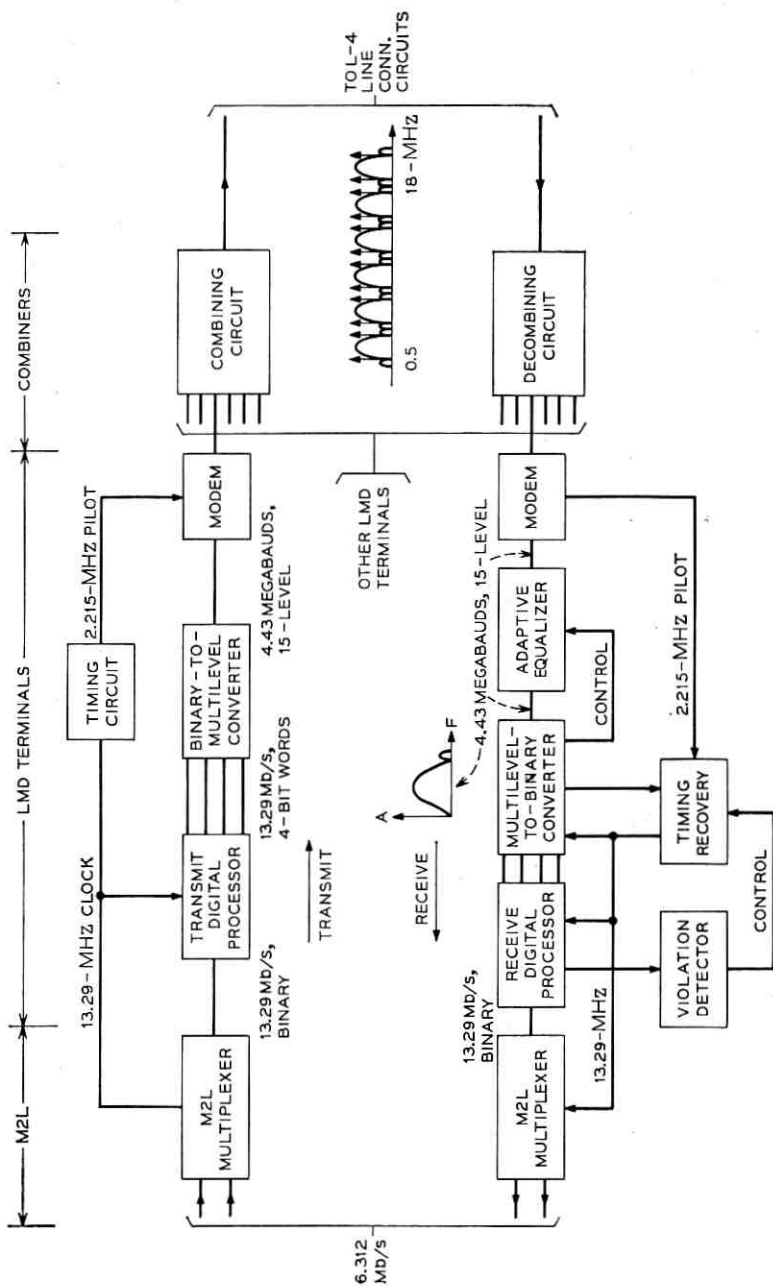


Fig. 2—L-4 line Mastergroup digital signals for a particular coaxial pair are combined, and the composite signal is added to the L-4 line.

veying mastergroups and digital mastergroups can be combined in any convenient assignment.

An LMD regenerative repeater consists of terminals connected at the 13.29-Mb/s interface as indicated in Fig. 3. (The M2L is not required in the repeater.) The repeater demodulates the signal to base-band, converts the 15-level signal to a regenerated 13.29 Mb/s, two-level signal. This basic pulse stream is then converted into a new 15-level signal and modulated into an appropriate mastergroup band. The repeater need not modulate the signal into the same mastergroup band that supplied the input. Thus, a repeater can interconnect any two mastergroups so that frequency frogging can be employed to accommodate the standard L-4 or L-5 blocking, branching, and frogging rules.

The functions and major design features of each of the basic blocks of equipment will be examined in more detail in the following paragraphs.

#### VI. M2L TIME DIVISION MULTIPLEXER

The time division multiplexer combines two 6.312-Mb/s signals into a single binary stream at 13.29 Mb/s. The multiplexing operation is complicated by the fact that the two 6.312-Mb/s streams are not always exactly at their nominal speed. The technique known as pulse stuffing<sup>8</sup> is used to buffer against rate variations in an adaptive manner. The 13.29-MHz terminal clock supplies more time slots per unit time than required by the input signals (Fig. 4). Pulses that convey no information are inserted into a number of these extra time slots on an optional basis, depending on deviations of the incoming signals from the nominal 6.312 Mb/s. The remaining extra time slots are used to convey control signals to the receiver so that it can delete the extra pulses and separate the two signals in synchronization with the transmitter.

#### VII. DIGITAL PROCESSOR

The digital processor is designed to perform two basic functions: (i) randomize the digital information to prevent discrete frequencies from appearing in the transmitted signal above specified levels, and (ii) encode the signal to achieve the desired spectral nulls in such a way as to prevent error propagation at the receiver.

A block diagram of the digital processor is given in Fig. 5. The serial-to-parallel converter groups the randomized 13.29-Mb/s binary

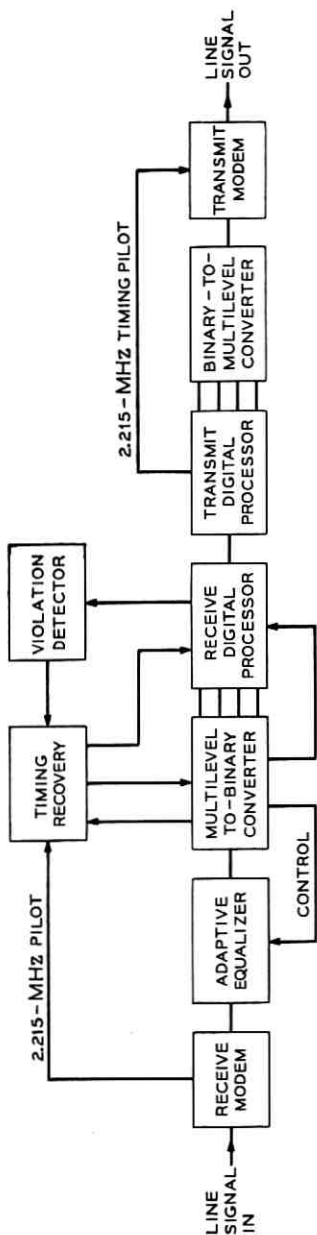


Fig. 3—An LMD repeater comprises two LMD terminals, interconnected at the 1329-Mb/s interface. Only one direction of transmission is shown.

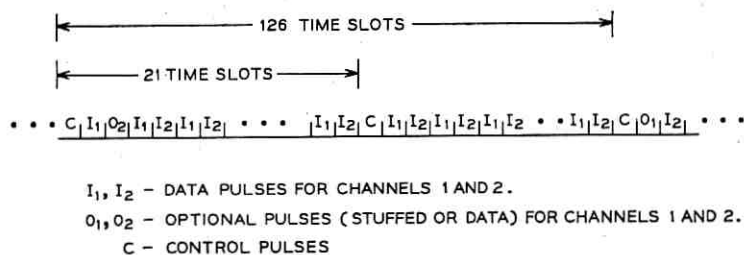


Fig. 4—In the time division line format, pulses are time multiplexed with data pulses into every 21st time slot. After six control pulses have been transmitted, an optional pulse is provided for stuffing in one of the data channels if required. Stuffing may take place in the other data channel following the next 126 time slot sequence.

stream into 3-bit words. Each of the  $2^3 = 8$  possible words is uniquely associated with an integer in the range 0 to 7, according to a Gray code. The Gray code ensures that a single level error made at the receiver in estimating the transmitted level will result in only a single bit error. The integer generated in the  $n$ th time slot is represented by the symbol  $D_n$ .

The encoded sequence,  $\{B_n\}$ , is obtained from the original sequence,  $\{D_n\}$ , by applying a pair of operations given by the following equations:

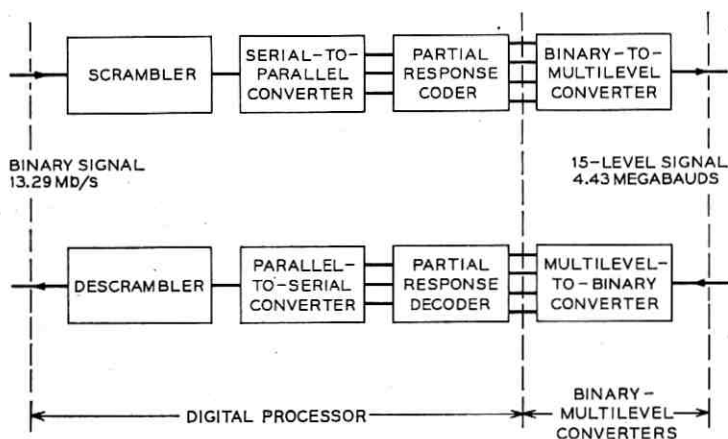


Fig. 5—The digital processing and binary-multilevel conversions indicated perform the digital-analog translations.



$$A_n = (D_n + A_{n-2}) \text{ modulo } 8, \quad (1)$$

$$B_n = A_n - A_{n-2}. \quad (2)$$

By substitution, the overall relation becomes

$$B_n = (D_n + A_{n-2}) \text{ mod } 8 - A_{n-2}. \quad (3)$$

If  $(D_n + A_{n-2}) \leq 7$ , then

$$B_n = D_n.$$

If  $(D_n + A_{n-2}) > 7$ , then

$$B_n = D_n - 8.$$

It follows that the symbols,  $B_n$ , can assume any one of the 15 integer values in the range  $-7$  to  $+7$ . The operation in equation (2) has the transfer function,  $1 - \exp(-j4\pi fT)$ , which has the desired zeroes at dc and the Nyquist frequency,  $(1/2T) = 2.215$  MHz. The operation in equation (1) takes account of the intersymbol dependence established by equation (2) so that the decoding operation of the receiver is independent of previously generated symbols,

$$D_n = (B_n) \text{ modulo } 8. \quad (4)$$

The operation in equation (3) is performed by the digital processor using binary elements. The output of the processor is a sequence of 4-bit binary words which represent the symbols  $B_n$ .

The digital processor includes a modulo-scrambler to randomize the digital signal sufficiently to suppress single tones. Discrete frequencies in the LMD signal can be generated by repetitive sequences of input data. *Picturephone* signals, for example, will have concentrations of power at multiples of frame rate, line rate, and sampling rate, plus lines in the spectrum due to periodic components in the original video. Single frequency tones arising from an LMD terminal or repeater and falling out-of-band in adjacent message channels should not exceed  $-72$  dBm0 in order to be inaudible. If complete protection against all pathological digital sequences were attempted, the out-of-band suppression requirements of the transmitting filters would have to be excessive. It is important to minimize these suppression requirements since they strongly influence the in-band phase distortion. Single tones higher than  $-14$  dBm0 can cause intelligible crosstalk if their frequencies fall at or near multiples of 4 kHz. Intermodulation of a message signal with such a tone may appear as undistorted speech in another channel.

## VIII. BINARY-TO-MULTILEVEL CONVERTER

The binary-to-multilevel converter generates a voltage pulse that can achieve any one of 15 equally spaced levels for each 4-bit word delivered by the digital processor. As shown in Fig. 6, a weighted current gating circuit is used to perform this function. The multilevel signal is the input to the modem.

## IX. MODEM

Since the partial response signal format is generated in the digital processor, the channel networks must be designed to minimize intersymbol interference by satisfying the Nyquist criterion as closely as possible.<sup>9</sup> The amount of bandwidth that is available for the digital signal is somewhat greater than the 2.52 MHz allocated for each analog mastergroup because of the guard space that has been provided between mastergroups. This guard space was originally provided in the message mastergroup multiplex for the skirts of the separation filters.<sup>2</sup> The corresponding skirts of the LMD separation filters can be used to shape the vestigial and roll-off regions of the digital signal. Although, in principle, vestigial and roll-off bands are not required for the partial response signal, they serve several purposes: (i) substantially improve

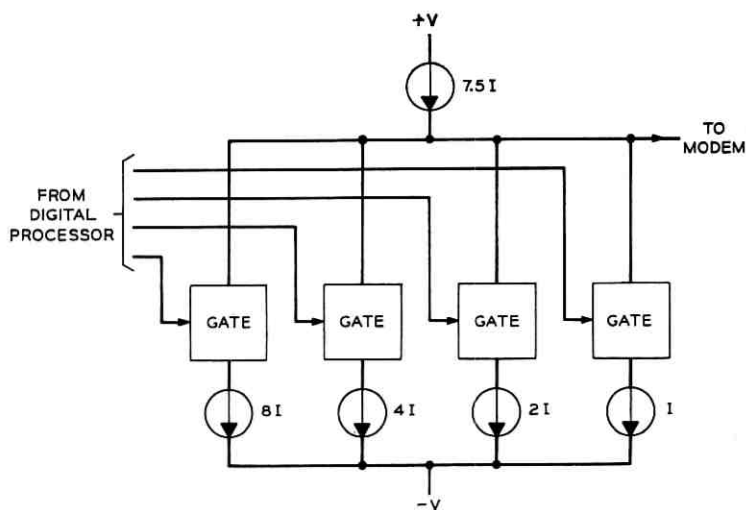


Fig. 6—The binary-to-multilevel converter receives the partial-response encoded signal from the digital processor on four leads and converts it to a 15-level signal for modulation into the appropriate mastergroup band.

network realizability over single sideband, no roll-off networks; (ii) permit gentle filter shaping which will minimize in-band amplitude and phase distortion; (iii) provide somewhat wider "eyes" which are less subject to timing jitter distortion; and (iv) reduce signal overshoot and corresponding LMD circuit linearity requirements, especially important in the M/B converter.

The analog portion of the LMD terminal contains two steps of modulation for all but mastergroup 2 (MG-2) as indicated in Figs. 2 and 7. The 4.43-megabaud, 15-level, partial response signal is first modulated into the MG-2 band. The second step of modulation translates the MG-2 signal into one of the other L-4 mastergroup bands. This modulation system has the advantage that all in-band signal shaping and most of the out-of-band discrimination is provided by the networks associated with the first step of modulation, and hence, this difficult job of network design has to be done only once. New synthesis techniques were developed to optimize the filter designs on the basis of a digital mean-square error criterion.<sup>10</sup> Mastergroup 2 was selected for the first step of modulation because it is surrounded by the narrowest guardbands and, therefore, requires the smallest vestigial and roll-off bandwidths. One further advantage of the two-step modulation technique is that MG-1 can be included in the basic system plan. Two steps of modulation are required for MG 1 to prevent signal leak from impairing performance since the baseband spectrum overlaps the MG-1 frequency band.

The first step of modulation utilizes vestigial sideband modulation with coherent detection. The second step of modulation is a potential source of additional frequency offset that could increase the phase error in the coherent detector. A new carrier recovery circuit has been developed for the LMD terminal, however, which significantly reduces the frequency offset without requiring more stable oscillators. A block diagram of the new circuit is given in Fig. 8, together with the equations that illustrate its operation.

#### X. ADAPTIVE EQUALIZER

An adaptive equalizer receives the bandlimited signal from the demodulator and filters it to minimize intersymbol interference. This type of equalizer incorporates a tapped delay line and electronically controlled tap attenuators, as indicated in Fig. 9.

The transfer function for the equalizer is given by

$$H_E(f) = \left[ \sum_{k=-K}^K C_k \exp(-j2\pi f k T) \right] \exp(-j2\pi f K T).$$

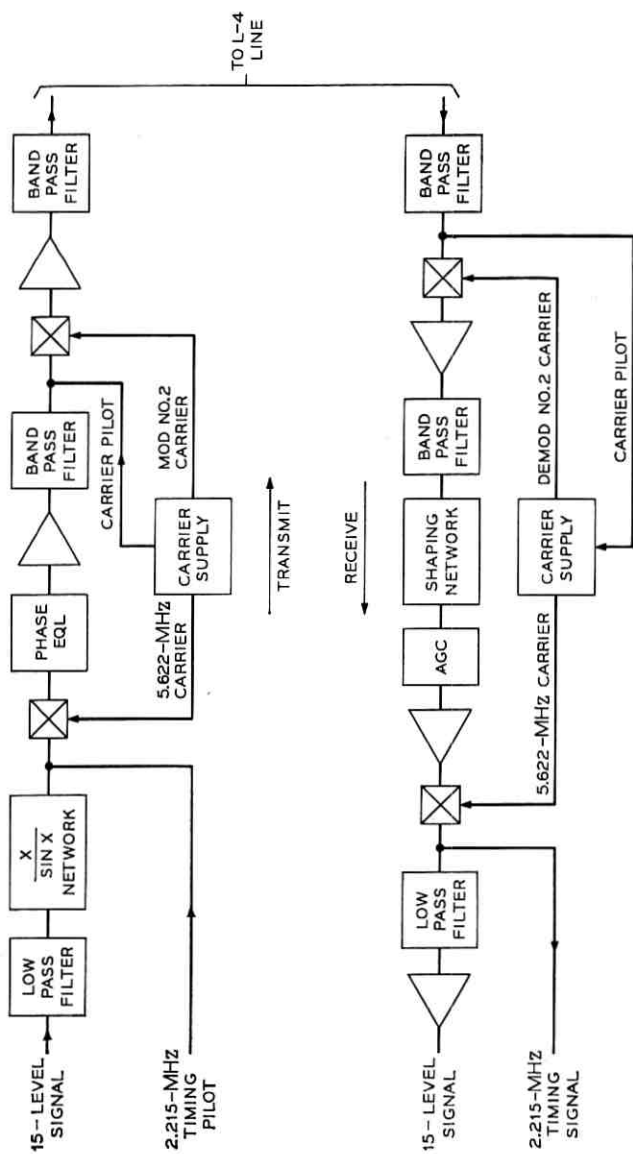


Fig. 7—The LMD modem utilizes two steps of modulation for all but MG-2.

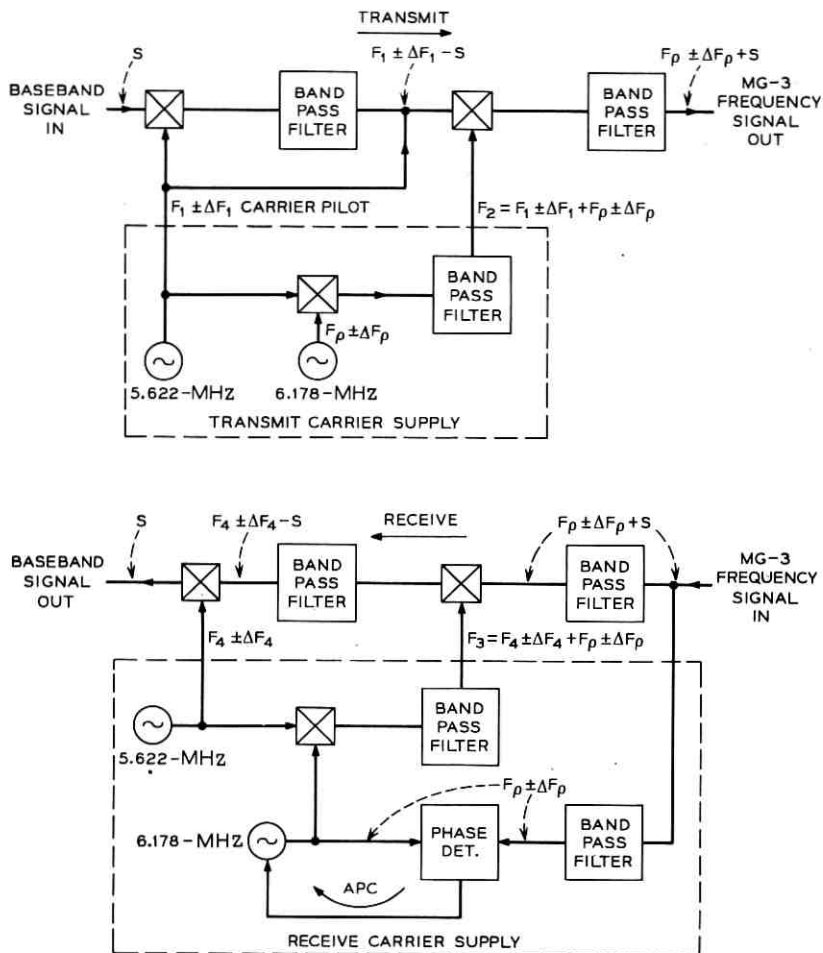


Fig. 8—The carrier recovery circuit as it applies to MG-3 is shown.

The factor in brackets shows the equalizer to be a finite dimension Fourier series synthesizer in the frequency domain. The equalizer is set to approximate the inverse of that portion of the overall channel characteristic that contributes intersymbol interference. The state of the Fourier coefficients,  $C_k$ , is continuously updated to compensate for changes in channel characteristics. This adaptive control is based on an algorithm which uses statistical estimates of the sampled pulse response of the equalizer as control information.<sup>11</sup>

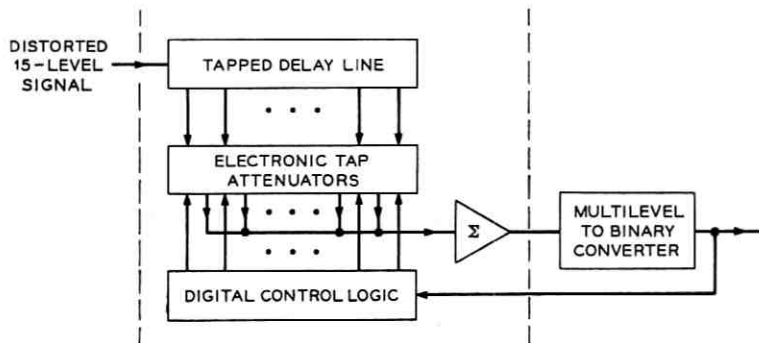


Fig. 9—The adaptive equalizer receives the band-limited signal from the demodulator and filters it to minimize intersymbol interference.

#### XI. MULTILEVEL-TO-BINARY CONVERTER

The multilevel-to-binary (M/B) converter utilizes the folder-slicer technique<sup>12</sup> to convert the received baseband signal to 6-bit words as indicated in Fig. 10. The signal is first sampled every  $T$  seconds and the sample held long enough for the conversion to the binary word to take place. Each folder is essentially a full-wave rectifier. A total of six slicing and five folding operations are used to generate a six-bit word. The first four digits represent the quantized amplitude of the sample

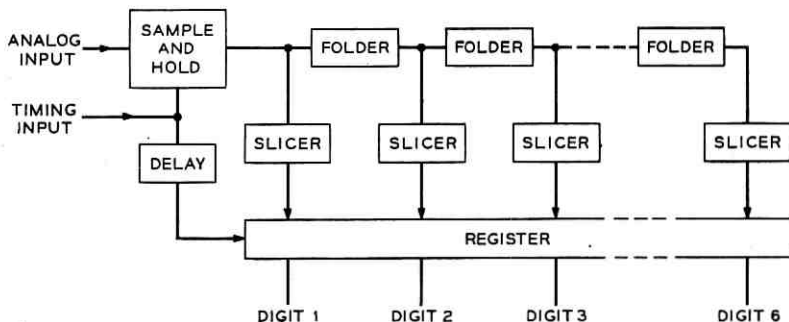


Fig. 10—The multilevel-to-binary converter samples the received 15-level signal at those points in time for which intersymbol interference has been minimized by the adaptive equalizer. Each sample is then represented as a six-bit binary word. The first four bits indicate the quantized amplitude of the sample while the fifth and sixth are used to detect and correct errors in signal level, slicing threshold, and sampling time.

while the fifth is used to detect errors in signal level and slicing threshold. When there are no errors in level or threshold, the output of the fifth slicer will be essentially random. When either one or both parameters are in error, the sense and type of error can be obtained by an appropriate logical combination of the outputs from the first five digits of the M/B converter. The sixth digit provides a measure of distortion at the sampling instants by indicating the percentage of the symbols whose levels deviate from nominal by more than 25 percent. This measure is used as an indication of the quality of transmission and in optimizing adjustments of such parameters as carrier phase and sampling time; it is also used in a control loop to make fine adjustments in the sampling time automatically as described below.

## XII. VIOLATION DETECTOR AND TIMING CONTROL CIRCUIT

The following constraint is derived from equation (2) for the sequence,  $\{B_m\}$ , generated in the transmitter:

$$-A_{m-2} \leq \sum_{j=0}^K B_{m+2j} \leq 7 - A_{m-2} .$$

When errors occur in transmission, the above constraint is violated, and the violation-detector indicates an error.

Figure 11 illustrates the error detector in block form.<sup>6</sup> The circuit alternately operates on the sequence of odd and even numbered received pulses,  $B_m$ . During each pulse interval, the detector supplies an estimate of the received multilevel pulse to the first adder which updates the running sums during alternate pulse intervals. The viola-

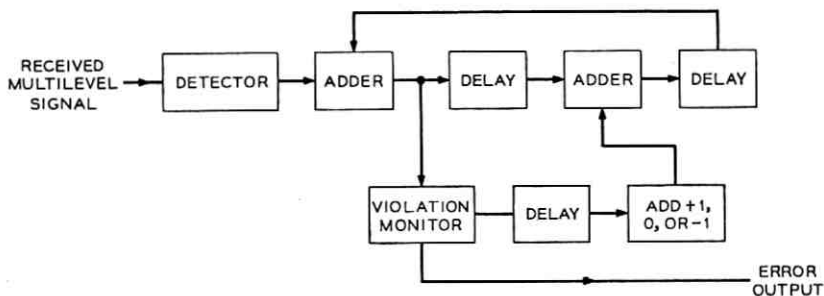


Fig. 11—The violation monitor provides a continuous indication of the error-rate performance of the repeater section.

tion monitor inspects these sums for violations of the correlation constraint. When a violation is noted, an error indication is given, and the running sum is reset. This ensures that a single error does not yield multiple violations.

If the error performance of a repeater section becomes poorer than one error in  $10^4$  bits, the error detector will trigger a major alarm. Since the error performance of each repeater section can be monitored independently, trouble conditions can be easily isolated to the appropriate section.

This violation detector is also used in a control loop to correct for errors in the sampling time. A block diagram is given in Fig. 12. A 2.215-MHz timing pilot is combined with the baseband digital signal at the transmitting terminal and is used in the receiver to synchronize a 13.29-MHz clock oscillator which generates the various timing signals needed to synchronize the M/B converter, digital processor, etc. The timing recovery automatic phase control (APC) loop does not have to maintain absolute control of the timing phase since an additional timing control circuit, in conjunction with the violation detector, is used to set the phase of the timing signal in the neighborhood of the optimum sampling phase. The timing control circuit operates whenever the error detector indicates the error performance has degraded below a specified threshold. In this way, the sampling time is maintained at approximately the correct value. An additional timing control circuit continually adjusts the timing to the optimum point by use of the sixth bit output on the M/B converter. Since the sixth bit is a measure of distortion, the timing can be automatically adjusted to minimize this parameter.

### XIII. SUMMARY

The L-mastergroup digital system will adapt modern coaxial systems for digital transmission on a per-mastergroup basis. Thus, both analog and digital mastergroups will be transmitted on a coaxial system simultaneously. Each LMD channel will carry two 6.312-Mb/s digital signals and, hence, two *Picturephone* signals. These channels are expected to provide a large part of the long-haul transmission capacity required for *Picturephone* service, providing low error rate performance for distances as great as 4000 miles. In the near future, additional equipment will permit the time multiplexing of many slower speed digital signals into high-speed bit streams for transmission over the LMD channels.



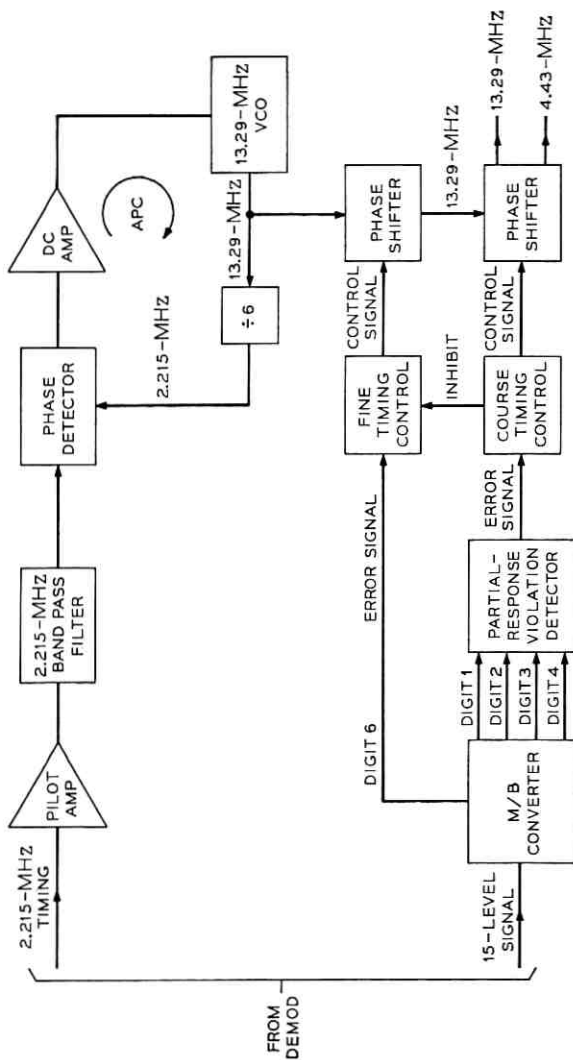


Fig. 12—The timing recovery circuit synchronizes a 13.29-MHz clock oscillator with the 2.215-MHz timing pilot that is transmitted along with the digital signal. The violation detector is used in a control loop to set the phase of the timing signal in the neighborhood of the optimum sampling phase and the sixth bit output of the M/B converter is used in a second control loop to set the timing phase to the optimum position.

## REFERENCES

1. Broderick, C. W., "The Picturephone® System: A Digital Transmission System for TD-2 Radio," B.S.T.J., this issue, pp. 481-499.
2. Klie, R. H., et al., L-4 System, B.S.T.J., 43, No. 4 (April 1969), pp. 819-820.
3. Brown, H. E., "The Picturephone® System: Transmission Plan," B.S.T.J., this issue, pp. 351-394.
4. Gerrish, A. M., and Howson, R. D., "Multilevel Partial-Response Signaling," IEEE Conference on Communications Digest, June 1967.
5. Kretzmer, E. R., "Generalization of a Technique for Binary Data Communication," IEEE Trans. on Communication Technology, COM-14, No. 1 (February 1966), pp. 67-68.
6. Gunn, J. F., and Lombardi, J. A., "Error Detection for Partial Response Systems," IEEE Conference on Communications Digest, June 1969.
7. *Transmission Systems for Communications*, Fourth Edition, Members of the Technical Staff, Bell Telephone Laboratories.
8. Witt, F. J., "An Experimental 224 Mb/s Digital Multiplexer-Demultiplexer Using Pulse Stuffing Synchronization," B.S.T.J., 44, No. 9, (November 1965), pp. 1843-1845.
9. Nyquist, H., "Certain Topics in Telegraph Transmission Theory; Trans. AIEE, 47 (April 1928), pp. 617-644.
10. Shnidman, D. A., "Optimum Phase Equalization of Filters for Digital Signals," B.S.T.J., 49, No. 7 (September 1970), pp. 1531-1554.
11. Lucky, R. W., "Techniques for Adaptive Equalization of Digital Communication Systems," B.S.T.J., 45, No. 2 (February 1966), pp. 255-286.
12. Smith, B. O., Jr., "An Unusual Analog-Digital Conversion Method," IRE Trans., PGIT-S (June 1956), p. 155.

# Switching



# The *Picturephone*<sup>®</sup> System:

## Switching Plan

By J. F. URICH

(Manuscript received October 5, 1970)

*This paper develops the switching plan for Picturephone<sup>®</sup> service from a statement of the service operating procedures. The signaling plan is presented in detail. The switching plan provides the framework for the detailed switching papers which follow.*

### I. INTRODUCTION

A prime objective of *Picturephone* service is to make it a natural extension of present telephone service. The switching plan gives explicit definition to this objective while permitting extensive use of the portion of the switching plant that performs functions common to both services. Where economies dictate that the *Picturephone* network depart from the telephone network, the switching plan provides the means.

The technical aspects of switching and transmitting a video signal are treated in other articles in this issue. This paper deals primarily with the service as seen by the customer, and about those engineering factors which then follow from the service definition.

### II. OPERATING PROCEDURES

The operating procedures for *Picturephone* service form a set of self-consistent rules which define it as a vertical extension of telephone service. From these procedures, requirements are generated for the switching hardware to provide the customer, local, and toll switching for the service. These procedures are as follows:

- (i) *Picturephone* and telephone calls are placed and received on common station equipment.
- (ii) The telephone loop carries *Picturephone* audio.
- (iii) The telephone line is busy when a *Picturephone* call is in progress and vice versa.
- (iv) *Picturephone* and telephone numbers are the same.

- (v) *Picturephone* service will only be provided with *Touch-Tone*<sup>®</sup> calling.
- (vi) *Picturephone* calls are placed by keying a *Picturephone* service indicator followed by the telephone number.
- (vii) The *Picturephone* service indicator is the twelfth *Touch-Tone* combination (# button).
- (viii) *Picturephone* calls placed to non-*Picturephone* numbers will not be allowed to complete on any basis. Such calls will be routed to a suitable announcement.
- (ix) A *Picturephone* call can be originated on any standard *Touch-Tone* telephone set having access to the audio portion of a *Picturephone* loop.
- (x) A *Picturephone* call can be answered on any telephone having access to the audio portion of a *Picturephone* loop, e.g., bridged telephone.
- (xi) A *Picturephone* station can always pick up (with video) a *Picturephone* call previously answered on a telephone.
- (xii) Switchhook supervision on *Picturephone* calls is maintained over the audio connection.
- (xiii) The switched video and audio *Picturephone* service connection is provided at called party answer and throughout the duration of the call.
- (xiv) Charge timing for customer dialed *Picturephone* calls is based upon audio supervision, beginning at called party answer and terminating upon disconnect.
- (xv) A distinctive alerting signal is provided during ringing to distinguish incoming *Picturephone* calls from telephone calls.
- (xvi) The associated switching equipment will provide signaling to energize the video circuits in a *Picturephone* station. On outgoing calls the station is energized in response to the *Picturephone* indicator (#) keyed by the caller; on incoming calls the station is energized when the customer answers at the *Picturephone* station.
- (xvii) *Picturephone* service should not restrict any telephone service available to the associated telephone line.

## 2.1 Discussion

These 17 operating procedures determine the numbering, dialing, and charging plans for the service. Procedures (i) through (iv) say that the customer has one line and one number used in common for both telephone and *Picturephone* calling. For telephone calls only the audio portion of the loop is used; for *Picturephone* calls, both

audio and video portions are used. The normal telephone numbering, based upon a three-digit area code followed by a three-digit office code and four-digit station number, uniquely identifies each customer. Destination routing can be employed within the *Picturephone* hierarchy just as it is used in the telephone hierarchy.

Procedures (v) through (vii) describe the customer dialing plan. Prefixing any telephone call with the # indicator signifies a *Picturephone* call. Item (xvi) further adds that the switching system will respond by turning on the video circuitry in the caller's station as a visual attending function. When the station is energized, a gray raster appears on the screen. With key telephone services a red lamp is lighted as well.

Procedures (ix) through (xii) recognize the boss-secretary relationship in a business communications environment. The boss may have *Picturephone* service while the secretary has only a telephone. She may still place and answer *Picturephone* calls on his line.

Procedures (xii) through (xiv) cover the supervision and charge timing of *Picturephone* calls. The plan calls for "immediate video." That is, the caller declares his intention during the dialing of the call. If the # prefix is dialed, *Picturephone* trunks, composed of audio and video transmission facilities, are selected for routing the call to the terminating office. If the call is answered, be it from a *Picturephone* station or from a bridged telephone-only instrument, *Picturephone* charging commences immediately. Under the "immediate video" discipline, standard telephone signaling and charge recording systems can be employed on the audio portion of the network with little modification. As a safeguard against erroneously dialed *Picturephone* calls resulting in improper charges, item (viii) requires that the switching systems block *Picturephone* calls which cannot, per se, provide the visual capability. In the *Picturephone* network, this screening function will be performed as close to the originating office as the routing plan will allow. In addition, the terminating *Picturephone* office will screen the terminating line for *Picturephone* service capability before ringing. If the call cannot be completed, a recorded audio announcement will inform the caller and suggest that the call be placed again without the # prefix—a subtle benefit of the "one number" procedure.

Procedures (xv) and (xvi) recognize that in integrating *Picturephone* service and telephone service, some additional customer line signaling is required. A single additional signal, Video Supervisory Signal (VSS), is sufficient to perform these functions. This will be described in more detail in Section IV.

Finally, procedure (xvii) recognizes that *Picturephone* is an adjunct to telephone service, not a replacement for it. A single exception has been made to this rule. *Picturephone* service will not be provided on multiparty telephone lines, an exception not deemed to be a serious restriction to the usefulness of the new service.

### III. THE ANALOG SWITCHING HIERARCHY

The *Picturephone* system description discusses the reasons for employing both baseband analog and digitally-encoded video transmission in the network.<sup>1</sup> The allocation of video impairments allows for a five-level switching hierarchy, excluding customer switching systems.<sup>2</sup> The top three levels of this hierarchy will always switch the *Picturephone* signal in its digitally encoded form.

Figure 1 depicts the switching hierarchy from the Class 4 office down to the *Picturephone* station.<sup>3</sup> As a rule, the machines in this portion of the hierarchy switch the video signal in its baseband analog form. Class 5, or local *Picturephone* switching offices are selected telephone central offices in key business wire centers throughout the country. Initially only No. 5 crossbar switching entities can be modified to function as *Picturephone* central offices.<sup>1</sup> The capability to use No. 1

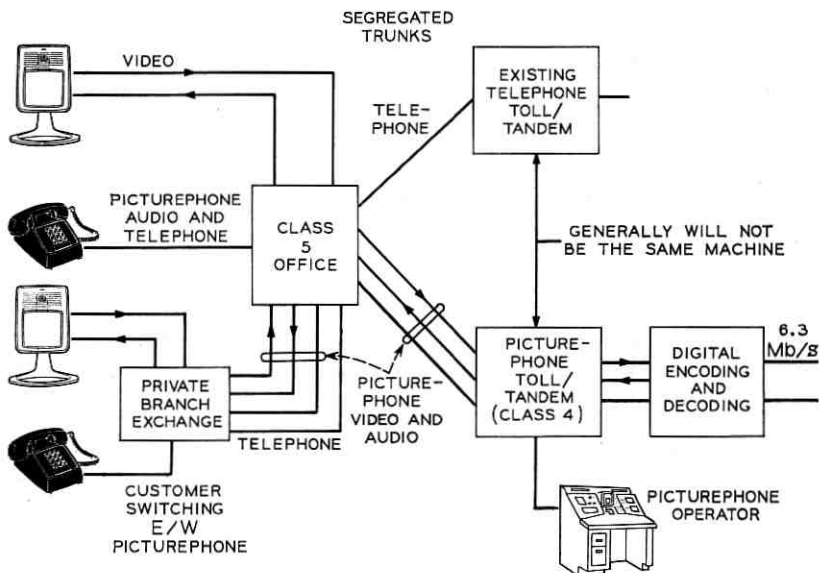


Fig. 1—The lower two levels of the *Picturephone* hierarchy illustrate the switching-trunking plan.



ESS as a local *Picturephone* switching entity will be included as the service develops.

The Class 5 *Picturephone* office terminates the audio and video pairs of individual *Picturephone* lines, *Picturephone* trunks from PBXs, and two-way *Picturephone* trunks to the Class 4 *Picturephone* office. All *Picturephone* lines and trunks require three appearances on the switching machine—one for audio and one for each direction of video. The two-wire audio appearance of a *Picturephone* line is used in common for *Picturephone* and telephone calls from the customer. All *Picturephone* trunks, however, are dedicated to *Picturephone* service only. The audio part of a six-wire *Picturephone* trunk is never used for a telephone call. Thus, for individual line customers, the Class 5 office serves as a point of departure between the telephone and the *Picturephone* networks. To the customer the two services are closely integrated. Beyond the Class 5 office the *Picturephone* transmission-switching network is engineered independent of the telephone hierarchy. *Picturephone* tandem/toll offices are physically located in telephone switching offices, but there is not 1:1 correspondence between the function of the switching machine in the *Picturephone* and telephone hierarchies. Because of the different ratio of switching and transmission costs between the two services, and the large disparity in traffic volume in the early years, it is expected that the *Picturephone* network will employ more final routing and fewer direct trunk groups than the telephone network. In essence, above the Class 5 office the *Picturephone* network is a wideband communication network overlaid on the nationwide telephone network. Routes, right of way, transmission facilities, switching control, and buildings are shared; but network engineering is different.

### 3.1 *Customer Switching Consideration*<sup>4</sup>

Just as the Class 5 *Picturephone* office splits *Picturephone* and telephone traffic into two networks for its customers, the customer switching systems (PBX and Centrex CU) accomplish the same split for their customers. Trunks leaving a *Picturephone* equipped PBX are segregated into a *Picturephone* group and the normal telephone groups. These separate trunk groups can even home on two different Class 5 offices (one being *Picturephone* service equipped of course) without numbering plan ambiguity. In many cases, large business customers with *Picturephone* service will be able to retain their listed telephone directory number even though their local Class 5 office is not equipped for *Picturephone* switching.

### 3.2 Operator Services<sup>5</sup>

*Picturephone* service operator services are directly analogous to operator services provided on telephone. The *Picturephone* service assistance operator, however, must control the video as well as the audio part of the contact. *Picturephone* service customers request assistance by dialing #0. The call is routed using *Picturephone* trunks to *Picturephone* operator trunk circuits which are generally terminated on a Class 4 *Picturephone* switching office. Audio-only facilities link the switching machine to operator positions on standard No. 3 switchboards. On the switchboards, each *Picturephone* answering jack is paired with an outgoing *Picturephone* jack over which the operator can extend the call by dialing back into the class 4 machine. The video portion of the call is not extended from the switching machine to the switchboard although the operator can control it by means of a key on her cord circuit.

In lieu of the customer seeing the assistance operator during the contact, a fixed video image is transmitted to the customer from the switching machine.

The other service and assistance codes applicable to telephone service, e.g., 411 for Directory Assistance and 611 for Repair Service, do not have specific *Picturephone* service counterparts. Calls to obtain these services will follow regular telephone procedures in order to allow calls about *Picturephone* service to be placed from non-*Picturephone* stations. So, for example, customer dialed calls to #411 are blocked and the customer is instructed to dial the call without the # prefix. The intention in blocking such calls is to avoid having the customer expect to receive a video response and then provide him audio only. If, in the future, *Picturephone* counterparts to these services become desirable, they can be provided.

## IV. THE SIGNALING PLAN

### 4.1 General Considerations

As described in the Operating Procedures in Section I, all call control supervision and address signaling for both lines and trunks is carried on the audio portion of the *Picturephone* facilities. Standard telephone supervision and addressing systems can therefore be used for *Picturephone* service without modification. The opportunity was taken, however, to reduce to a minimum the number of different systems employed for signaling in the *Picturephone* system, in order to save development effort. For example, *Touch-Tone* calling is stan-

dard for all *Picturephone* customers. All interoffice trunks employ E&M lead supervision and multifrequency address signaling on the audio path.<sup>6</sup> Because *Picturephone* trunks are never used for non-*Picturephone* calls, there is no necessity to carry the *Picturephone* service indicator past the originating office. The trunk class of service conveys this information.

#### 4.2 *Video Continuity Testing*<sup>7</sup>

Charge recording in the *Picturephone* network is based entirely upon supervisory signals on the audio path. In order to verify that a usable video transmission path is provided on every billed call, an integrity test is made of the entire video connection from station to station on every network call.

This video continuity test is performed link-by-link as the call is set up. Every analog loop and trunk, when it is idle, has its video transmit path connected to its video receive path through a "loop around." When the circuit is to be used in a *Picturephone* call, the switching office selecting or controlling the circuit connects a 12 kilohertz sine wave oscillator to the video transmit pair and a tuned receiver to the receiver pair. The continuity test tone is looped back at the distant end of the circuit, and the receiver checks the returned tone level. Satisfactory completion of this test is required before the call set up procedure can proceed. Failure of the test on a *Picturephone* trunk results in the switching office selecting another trunk and trying again to complete the call. Because all *Picturephone* trunks are two-way—that is, they can be seized from the switching office at either end—loop arounds are required at each end. To prevent singing on idle trunks, 3 dB of attenuation is contained in each loop around circuit. The customer end of video loops do not require attenuation in their loop around circuit, so the test tone is returned at nominally the same level as transmitted.

On *Picturephone* interoffice trunks, the loop around at the outgoing end of the trunk is removed upon seizure of the circuit. At the incoming end the loop around is removed when multifrequency addressing commences on the audio path, which will only happen if the continuity test is satisfied. Thus, no additional control signaling is required for loop-around control on analog *Picturephone* trunks. Loop-around control on *Picturephone* loops does however impose some additional customer line signaling requirements over and above that employed in telephone service.

#### 4.3 Customer Line Signaling for Picturephone Service

Operating Procedures (xv) and (xvi) refer to the requirements that *Picturephone* switching offices provide a means for distinctively alerting a *Picturephone* station to an incoming *Picturephone* call (alerting function) and control the video turn-on of both calling and called *Picturephone* sets (attending function). Section 4.2 described the necessity for loop-around control on *Picturephone* loops. Accomplishing these three functions requires a degree of interaction between the switching office and the loop terminating equipment at the station not necessary in telephone service.<sup>8</sup> One additional customer line signal applied at the switching office is sufficient to provide this interaction. This signal is called Video Supervisory Signal (VSS).

##### 4.3.1 Additional Signaling Constraints

The signaling plan required to accomplish alerting, attending, and loop around control must also satisfy the following operational requirements:

- (i) It should be possible for maintenance personnel at a switching machine location to perform tests on the video portion of a *Picturephone* loop (i.e. through a distant loop around) without alerting the customer.
- (ii) A *Picturephone* customer should never be allowed to inadvertently see himself through a continuity test loop around.
- (iii) First-order failures—that is, a single signal-generator failure in a switching office—should not result in an unsatisfactory call.

##### 4.3.2 The Signaling Logic Design

The new customer line signal required is VSS. This is a video signal composed of vertical and horizontal sync pulses with a fixed brightness level instead of picture information. VSS generators are provided at all switching offices. Notice, however, that a *Picturephone* set transmitting video also generates VSS; e.g., VSS is a subset of the set of all video signals.

At the *Picturephone* loop terminating equipment (a *Picturephone* service unit or a Key Telephone System line circuit) the following binary information is available:

$L$  = Audio loop off hook. Voltage present and current flowing.\*

\*In general the logic circuitry in the *Picturephone* station has metallic access only to the switchhook of the main telephone. The on hook-off hook state of bridged telephone extensions on the audio loop must be deduced from the state of the loop.

$L'$  = Audio loop on hook. Voltage present and no current flowing.  
All telephones are on hook.

$M$  = Main station off hook. Derived from switchhook contacts of the main telephone set.\*

$M'$  = Main station on hook.

$V$  = VSS present on the *incoming* video pair.

$V'$  = VSS not present.

$R$  = 20-Hz ringing present on the audio loop.

$R'$  = 20-Hz ringing not present.

Figure 2 shows the sequential state diagram of the *Picturephone* loop terminating logic package.

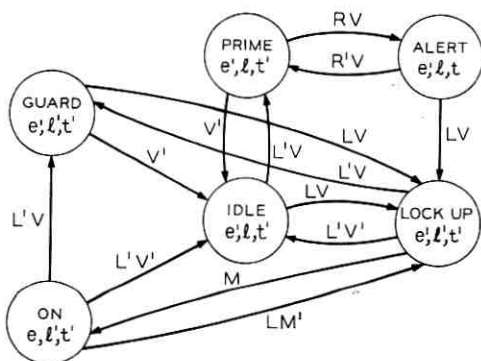


Fig. 2—The sequential state diagram of the *Picturephone* station logic illustrates the switching-station interaction.

Five logic states are defined in order to control three binary outputs. The outputs to be controlled are:

$e$  = video circuitry energized, camera and receiver on;

$e'$  = video circuitry off;

$l$  = video loop around in;

$l'$  = video loop around out;

$t$  = tone ringer sounded; and

$t'$  = tone ringer silent.

In the IDLE state the video circuitry is off ( $e'$ ), the loop around in ( $l$ ), and the tone ringer silent ( $t'$ ). On an originating call the main station or an extension goes off hook, receives dial tone, and dials. The set remains IDLE. The switching equipment performs a con-

\* Note that the  $L$  and  $M$  variables are not independent. State  $L'M$  is impossible.

tinuity test, and then returns a burst of VSS. The combination *LV* causes a transition to LOCK UP. If the call is being originated by the main station, a transition to ON occurs immediately. In LOCK UP and ON the loop around is out, and the set is energized or not energized respectively. Notice that as long as any telephone is off hook, the main station can go on and off hook repeatedly and the video circuitry will turn off and on. The set will not return to IDLE until all stations on loop are on hook. The transition to IDLE takes place directly if *L'V'* occurs, or via GUARD if *L'V'* occurs first. The GUARD state prevents the distant party from seeing himself through a loop around in this station in the case of a called party disconnect.

On an incoming *Picturephone* call, the set is necessarily in the IDLE state. This switching equipment tests video continuity and then applies VSS prior to starting 20-Hz ringing. This causes a transition to PRIME. Note that a test man at the switching machine can perform loop-around transmission tests of frequency versus gain, or can apply and view video test patterns without alerting the customer. The video circuitry in the station cannot be energized, however, unless the station is off hook. When 20-Hz ringing is applied by the switching machine, the tone ringer is sounded (ALERT) keyed on and off by the 20 Hz as long as VSS is present. When the called party answers and VSS is present, the set goes to LOCK UP and then to ON as in the previous example. On terminating *Picturephone* calls, the switching machines apply VSS towards the station prior to ringing and continuously until an answer signal is received over the audio path. Notice the fail-safe feature that even if one VSS generator fails to operate on a call, the calling station (if transmitting video) will turn on the called station or vice versa.

Figure 3 shows the entire sequence of an interoffice *Picturephone* call between two individual stations. This serves to illustrate the entire switching process, and summarize the switching system plan.<sup>9</sup>

#### V. CONCLUSIONS

The switching plan for *Picturephone* service is the definition of a numbering, dialing, charging, and hierarchical plan for the new video-telephone service. It provides a service well integrated with telephone communications from an operational viewpoint, while allowing the incremental use of that sizeable portion of switching systems logical hardware not particularly sensitive to the communications medium. With this plan, switching development effort was minimized, focusing on those aspects of the service which make it different from telephone service.

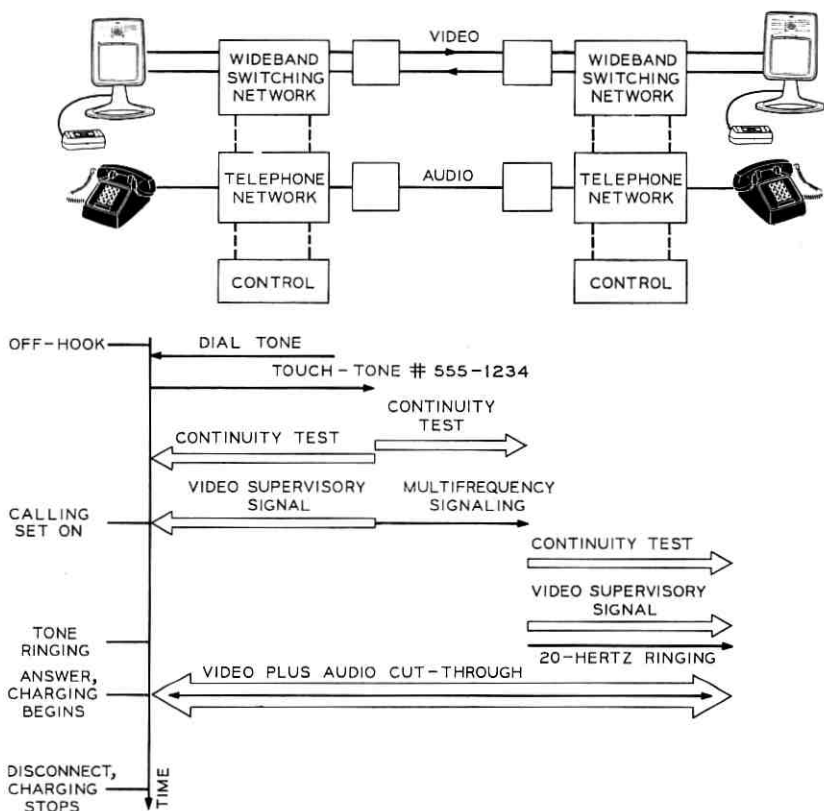


Fig. 3—The complete sequence of events in an interoffice *Picturephone* call.

#### REFERENCES

1. Dorros, Irwin, "The *Picturephone*® System: The Network," B.S.T.J., this issue, pp. 221-233.
2. Brown, H. E., "The *Picturephone*® System: Transmission Plan," B.S.T.J., this issue, pp. 351-394.
3. American Telephone and Telegraph Company, *Notes on Distance Dialing*, 1968.
4. Breen, C., "The *Picturephone*® System: Customer Switching Systems," B.S.T.J., this issue, pp. 553-565.
5. Burgess, P. N., and Stickel, J. E., "The *Picturephone*® System: Central Office Switching," B.S.T.J., this issue, pp. 533-552.
6. Dougherty, H. J., Peterson, E. B., and Schachtman, M. G., "The *Picturephone*® System: Maintenance Plan," B.S.T.J., this issue, pp. 621-644.
7. Breen, C., and Dahlbom, C. A., "Signaling Systems for Control of Telephone Switching," B.S.T.J., 39, No. 6 (November 1960), pp. 1381-1444.
8. Cagle, W. B., Stokes, R. R., Wright, B. A., "The *Picturephone*® System: 2C Video Telephone Station Set," B.S.T.J., this issue, pp. 271-312.
9. Urich, J. F., "The Switching Plan for *Picturephone*® Service," Proceedings of the National Electronics Conference, Chicago: National Electronics Conference, Volume 25, December 1969, pp. 904-908.





# The *Picturephone*<sup>®</sup> System:

## Central Office Switching

By P. N. BURGESS and J. E. STICKEL

(Manuscript received September 30, 1970)

*Central office switching of Picturephone<sup>®</sup> calls required the design of an automated network capable of switching large numbers of wideband signals on an analog basis. We describe in this paper the development of a switching plan that allows for the interconnection of many customers to many trunks and at the same time stays within acceptable transmission limits. We also give an explanation of the transmission constraints on the physical design of the central office network.*

### I. INTRODUCTION

Although devices have been developed to switch megahertz bandwidth signals, they have been limited to a small number of inputs and outputs. Also, the logical controls for these devices have been relatively simple. The development described in this article is the first automated, large-scale switching system designed to switch large numbers of wideband signals on an analog basis. We discuss the network and controls added to the No. 5 crossbar switching system that enable an existing switching system to process calls from *Picturephone* stations as well as telephones.

#### 1.1 *Design Philosophy*

*Picturephone* service is intended to be an extension of basic telephone service rather than a separate communication package. This fact has important implications in all of the Bell System, particularly in the design of switching systems. As such, the wideband switching system developed for *Picturephone* service utilizes the existing telephone plant as much as possible. Clearly this design method is reasonable from an economic standpoint since a large number of existing telephone switching functions—registration of dial digits, charging, call supervision, signaling—are also needed for *Picturephone*

switching. This philosophy also implies a natural extension of the standard numbering plan; therefore, a *Picturephone* customer can receive both *Picturephone* and telephone calls on the same number. The only difference between dialing a *Picturephone* call and a telephone call is that the dialed digits for a *Picturephone* call will be prefixed with the 12th *Touch-Tone*<sup>®</sup> button designated (#).<sup>1</sup>

The No. 5 crossbar system was chosen the switching machine to provide initial service for *Picturephone* customers. That system is widely available throughout most of the major metropolitan areas in the United States. It is organized on a common control basis and, because of its inherent flexibility, can add new services such as *Picturephone* calling. Future developments will provide this capability in electronic switching systems.

### 1.2 Application

The system to be described is capable of serving as a local switching system by handling *Picturephone* calls to and from (i) individual lines, (ii) centrex lines, (iii) PBX switching systems modified for *Picturephone* service, and (iv) Bell System operators. The centrex lines, either terminated directly in the central office (centrex CO) or on an associated PBX (centrex CU), will be able to receive direct-in-dialed (DID) calls, to originate automatically-identified-outward-dialed (AIOD) calls, and to key four- or five-digit intercom calls. Transfer features for centrex CO customers will be provided in a later development.

In addition, the system will serve as a toll and tandem machine by switching calls between *Picturephone* interoffice trunks on an analog basis.

## II. SWITCHING PLAN FOR NO. 5 CROSSBAR

### 2.1 Central Office Switching Network

The switching plan is influenced by the electrical interface with the *Picturephone* set. This interface is presented over three pairs of wires: one for the bi-directional audio signals, and the other two for the video transmit and receive signals. The audio pair is switched by the existing No. 5 crossbar two-wire network, while the video pairs are switched by a new four-wire network, essentially in parallel with the audio network (see Fig. 1). For control purposes, this new switching network, called the wideband link (WBL), is associated with the two-wire network in a definite manner. On the line side of the

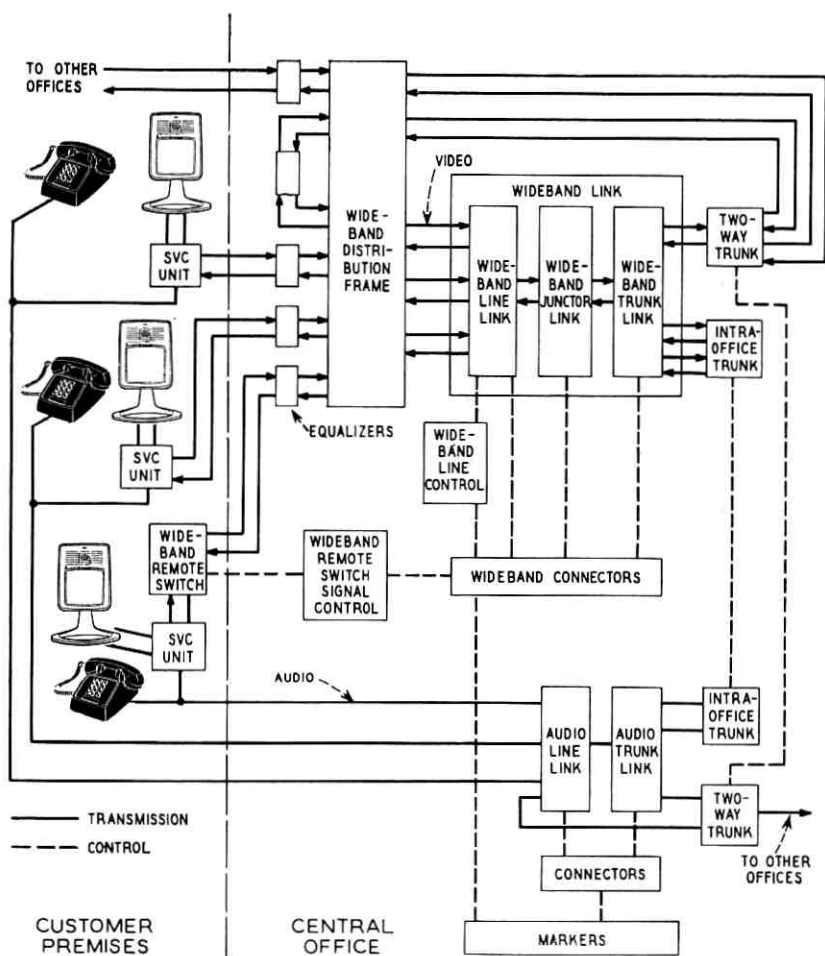


Fig. 1—Switching plan. (WBRSS = wideband remote switch; WBRSSC = wideband remote switch signal control.)

network, the *Picturephone* station's audio appearance on the two-wire network is associated directly on a one-for-one basis with the video appearance on the WBL. Whenever the *Picturephone* station is involved in a *Picturephone* call, one path is switched through the audio network to the audio appearance, and another is switched through the WBL to the associated video appearance. If a *Picturephone* station is involved in a telephone call, only the audio portion is switched through the office.

On the trunk side of the network, the same association principle holds such that every audio trunk with an appearance on the two-wire network has an associated video trunk with an appearance on the WBL. Interoffice *Picturephone* calls are switched via video trunks and their two-wire counterpart. The control is arranged so that these audio trunks are never used for telephone calls.

With this switching plan, the probability of a *Picturephone* call being blocked is greater than or equal to that of a telephone call since a *Picturephone* call is routed through two switching networks rather than one. Although the two networks are not stochastically independent, the dependence is small because the switched channels are selected independently. In particular, if  $P_A$  and  $P_V$  are the probabilities of a particular input being blocked in the audio and video network, respectively, and  $P_P$  is the probability that a *Picturephone* call is blocked through this office, then

$$\begin{aligned} P_P &= 1 - (1 - P_A)(1 - P_V), \\ &= 1 - (1 - P_A - P_V + P_AP_V), \\ &= P_A + P_V(1 - P_A). \end{aligned}$$

Since

$$P_A \ll 1, P_P \approx P_A + P_V.$$

The probability of a *Picturephone* call being blocked should not be significantly above the blocking probability of a telephone call. Hence,  $P_V$  should be significantly smaller than  $P_A$ , and the WBL should be nearly nonblocking.

The above requirements have been satisfied by using a three-stage network which has nearly all the expected blocking in the first stage. Since the second and third stages approximate a Clos nonblocking network<sup>2</sup>, the blocking is nearly zero through these stages. By allowing concentration in the first stage of up to 8:1, the telephone company has the flexibility of controlling the blocking in the network by line reassignment. The network can be provided with a minimum of 100 trunk terminations and is expandable, in 100-trunk termination increments, to the 400-termination size. Market studies have indicated that, for the early years of service, a network with 400 trunk terminals is adequate. For each of the four sizes, a 4:1 or 8:1 line concentration ratio is available. The network topology is shown in Fig. 2, and described below where  $N = 1, 2, 3, \text{ or } 4$ .

The first stage consists of  $10N$  wideband line links, each line link having ten outputs and 40 or 80 inputs for a maximum of 3200 lines.

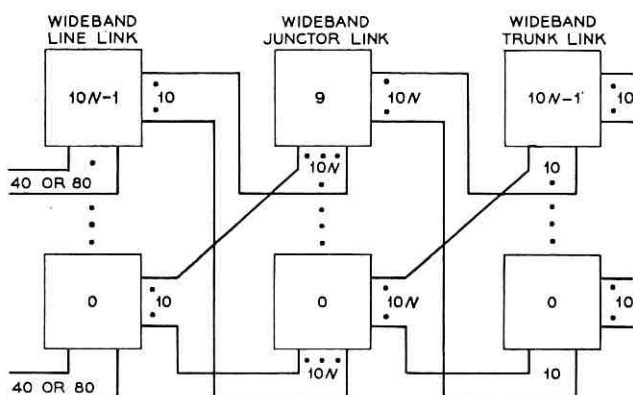


Fig. 2.—Wideband network topology.

The second stage, the wideband junctor link stage, consists of 10 junctor switch groups, each group having  $10N$  inputs and  $10N$  outputs. The third stage, the wideband trunk link stage, consists of  $10N$  groups, each having ten inputs and ten outputs. The packaging of this equipment and its impact on the central office floor plan as it affects the transmission characteristics is explained in Section III.

The ten outputs (horizontals) of a line link are spread over the ten junctor switch groups. The outputs (horizontals) from a junctor switch group are spread uniformly over the wideband trunk groups. This network configuration provides ten paths between any input and any output.

## 2.2 Remote Switching Network

A prime candidate for early *Picturephone* service is the centrex business customer. While he will eventually be making many calls to other customers as the national *Picturephone* network grows, he will initially be making a large percentage of calls within the community of interest on his own company premises. If the WBL were the only switching network provided in the switching plan, every call between two stations within the customer group (i.e., intercom calls) would involve two sets of transmission facilities between the customer and the central office. By providing a switching network at the customer's premises, the wideband portion of intercom calls can be switched entirely at the customer's location, thus reducing the transmission facilities needed by the customer for this type of call. The wideband remote switch (WBRS) was designed to meet this need. It is arranged

to switch just the video signals, and most of the control\* of the WBRS is in the central office. The audio for each station is still switched at the central office implying that no modifications are necessary for the existing audio facilities already provided by centrex CO arrangements. As a consequence of providing the WBRS near the customers, a reduced number of transmission links are provided between the central office and the output of the WBRS. Therefore, the WBRS can also function as a switching concentrator for *Picturephone* centrex customers.

The WBRS is arranged as a single stage crossbar network with 80 inputs and 20 outputs. The inputs accept the video transmit and receive pairs from the *Picturephone* station and the outputs either connect directly to the WBL so as to provide video links to the rest of the *Picturephone* network or connect to intra-remote switch trunks. These trunks complete the switched video connection on *Picturephone* intercom calls.

### III. TRANSMISSION CONSTRAINTS ON PHYSICAL DESIGN

The problems of crosstalk, impulse noise, insertion loss, and echoes in a large scale 1-MHz bandwidth switching network presented an unusual challenge to the switching system designers. In addition to providing a satisfactory transmission path, this network had to be practical from a manufacturing, installation and maintenance standpoint. This section is devoted to a discussion of how these constraints influenced the resulting physical configuration of the network.

Figure 3 shows the wideband network in terms of the actual apparatus utilized in switching the *Picturephone* call. Each line link consists of two or four 20-vertical, ten-level, six-wire crossbar switches, each  $34\frac{1}{2}$  in.  $\times$   $9\frac{1}{2}$  in. The junctor switch groups each have up to eight crossbar switches, like those in the line link, and each trunk switch group has one ten-vertical, ten-level, six-wire crossbar switch,  $20\frac{1}{2}$  in.  $\times$   $9\frac{1}{2}$  in. Regardless of the method of equipment packaging, the distance through the wideband network is an appreciable part of the *Picturephone* signal's wavelength and the transmission characteristics of the switching network must be considered. To control these transmission characteristics, the wideband network was designed within the constraints imposed by an overall transmission *level plan* for analog facilities.

---

\* "Control" is used in the broad sense: it includes registration of dialed digits and translation of information as well as electrical control of the network.

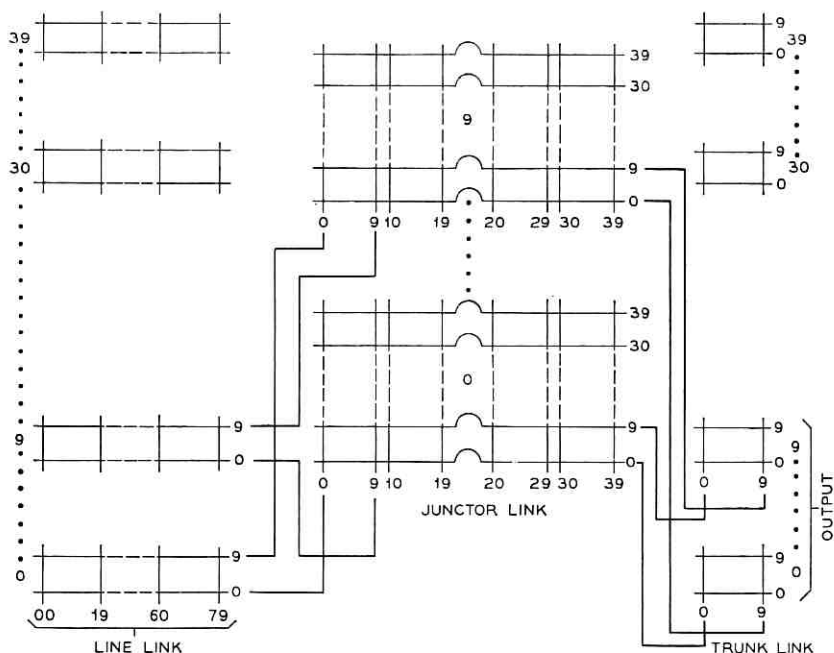


Fig. 3—No. 5 crossbar wideband switching link network.

### 3.1 Level Plan

The overall level plan states that the output of any cable equalizer must be at the same transmission level. This reference point, *Zero Picturephone Transmission Level Point (0 PTLP)* is defined as the output of the central office loop equalizer in the direction of transmission from the station set to the central office. Beginning at the station set equalizer, shown in the overall level plan, Fig. 4, each subsequent equalizer compensates for its preceding transmission path and its output is adjusted to be at 0 PTLP. As long as the signal degradation in this preceding path is reasonably constant and predictable, the equalizers need to have only fixed gain and phase characteristics (fixed equalizer).

The *Picturephone* switching network level plan for No. 5 crossbar, shown in Fig. 5, must satisfy the same transmission requirements as the overall level plan. This plan was particularly difficult to implement within the central office since any of the maximum 3200 *Picturephone* lines could connect to any of the 400 wideband trunks. This meant that the variation in transmission degradation among the paths

through the wideband network had to be within the tolerance allowed in 0 PTLP. To accomplish this, additional transmission constraints were imposed internal to the central office. These constraints still allow fixed equalizers (rather than dynamic equalizers) to be utilized. In addition, traffic balancing of both *Picturephone* lines and trunks is required. The level plan and traffic balancing requirements are satisfied by the use of a wideband distributing frame (WBDF) located a fixed electrical distance between the cable equalizers and the switching equipment. To allow traffic reassignment without equalizer readjustment, the input ( $S1$ ) and output ( $S3$ ) levels at the WBDF are fixed.

### 3.1.1 Transmission Cable Length Restrictions on Wideband Network

Starting at any of the cable equalizers shown in Fig. 5, the effective distance to the WBDF is fixed so that the transmission loss is constant and equal to the loss of 350 feet of cable. This distance, chosen because of the anticipated size of cable equalizers and expected central office space availability, represents the longest cable run from an equalizer to the WBDF. For equalizers closer to the WBDF, the 350-foot path may be a combination of cable buildouts (BO) and actual cable. The level of  $S1$  is therefore equal to 0 PTLP minus the transmission loss of 350 feet of cable. The distance from the WBDF through the switching network and back to the WBDF is 320 feet with an allowable variation of  $\pm 50$  feet. This nominal distance and variation was chosen after a study of trade-offs was made, considering the following:

- (i) Cost and practical limits of gain and phase compensation in the fixed equalizers.
- (ii) Central office space availability for the wideband network.
- (iii) Minimum cable distances possible through a network that has a maximum of 280 crossbar switches.

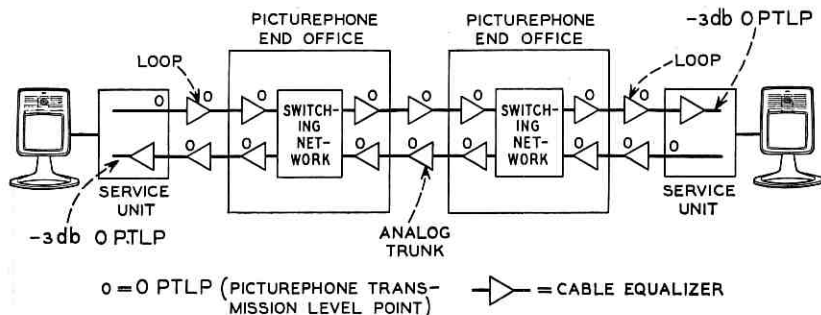


Fig. 4—Overall transmission level plan.





Since the network must switch intraoffice and operator calls, the trunks must be at the transmission midpoint, level  $S_2$ , to satisfy the level plan. Figure 5 summarizes the levels in terms of effective loss of central office cable from 0 PTLP.

The wideband line link (WBLL) appearance of two-way and operator trunks passes through the WBDF and a tandem cable equalizer to reestablish 0 PTLP before it enters the wideband network. These appearances terminate directly on the WBDF to take advantage of the traffic assignment flexibility offered.

### 3.1.2 *Transmission Restrictions Due to Video Impairments*

In addition to the distance restrictions imposed on the central office equipment to satisfy the level plan, a limit on the magnitude of the video impairments—echo, random noise, impulse noise, crosstalk, and power hum—is required to stay within the allowable overall loss allocations.<sup>3</sup> Each of these impairments has a particular effect on the physical design of the wideband network.

For instance, a cable equalizer compensates for some average frequency response and the echo rating impairment of any other path will depend on how it deviates from this average response. Echoes in the switching office, as explained earlier, are caused primarily by the multiplying of cables necessary to interconnect the various junctor switch groups. Figure 6 shows echo rating<sup>3</sup> as a function of the number and length of bridge taps. The shaded region of this figure shows the allowable echo rating of the switching office as a function of only unequalized bridge taps, and assumes perfect equalization of both central office cable and switches. Since the allocation of echo is based on an average path, it is necessary to keep the number of bridge taps along the path constant, as short as possible, and at a fixed distance from the cable equalizers.

Noise impairments result from energy in the paths that control relays and crossbar switches being induced into the video pairs, and from wide electrical current fluctuations in the battery and ground feeders coupling through the framework onto the video paths. To stay within the noise allocations, all video cables are separated by at least two inches from control cables when they run parallel to each other on a frame and are run at  $90^\circ$  to each other where the 2" minimum spacing cannot be maintained. Also, contact protection devices (series RC circuits) are connected between ground and the video switch control leads in order to reduce the effect of high energy transients. Battery and ground feeders for the wideband network are

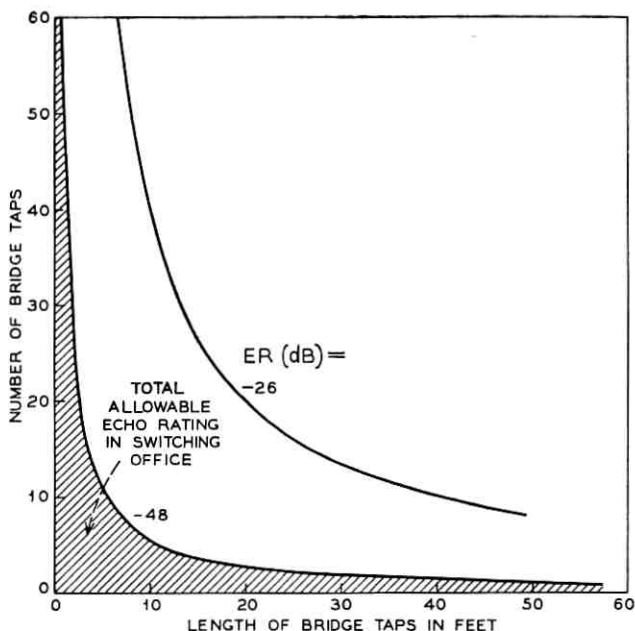


Fig. 6—Echo rating for unequalized bridge taps.

electrically isolated from the audio switching equipment thereby preventing the wide current fluctuations on the audio frames from coupling onto the video frames. Also, all wideband frames in each wideband frame line are interconnected with large copper buses and these buses interconnect at the main aisle to reduce as far as practicable the variation between frame potentials.

Crosstalk is caused by energy in the video transmit pairs coupling into its receive pairs (self crosstalk), or by energy in the transmit and receive pairs of one customer coupling into the transmit and receive pairs of the adjacent customer on the wideband link crossbar switches (worst disturber crosstalk), Fig. 7. While crosstalk cannot be defined exclusively in terms of dB per inch of cable separation, it was found from laboratory measurements that, by applying the following rules to the wideband equipment, the crosstalk allocations can be met.

- (i) Twisted pair wiring with varying twist lengths must be used for all transmission leads.
- (ii) The fourth wire on each level of every crossbar switch must be electrically grounded. This *ground plane* tends to electrically isolate the two directions of transmission on the switch.

- (iii) Transmit and receive leads between switching frames must run in separate cables. This prevents transmission paths of different energy levels from being run adjacent for an appreciable distance.

Power hum impairments are relatively minor in the central office; therefore, the only precautionary step taken is to electrically isolate new appliance outlets from audio frameworks where these outlets are used to supply ac power to video equipment.

### 3.2 Physical Design of the Wideband Network

The wideband network had to satisfy the level plan, stay within the video impairments allotted to it, and grow easily from the 100-trunk termination size to the maximum 400-trunk termination size network. A plan that reserves floor space in the central office for two facing lineups of wideband frames, each approximately 23 feet long, and controls the distance to the surrounding wideband frames satisfied these criteria. This plan with typical frame locations is shown in Fig. 8.

The WBLL frames are designed to have five of the wideband line links shown in Fig. 3 mounted on a basic frame, each with a maximum concentration of 4:1. A supplementary line link frame provides for the additional 4:1 concentration and is interconnected to its associated basic frame by a short cable connected between the switch horizontals. The WBLL frames grow uniformly in the controlled part of the floor plan as the office requirements for line terminations dictate. The con-

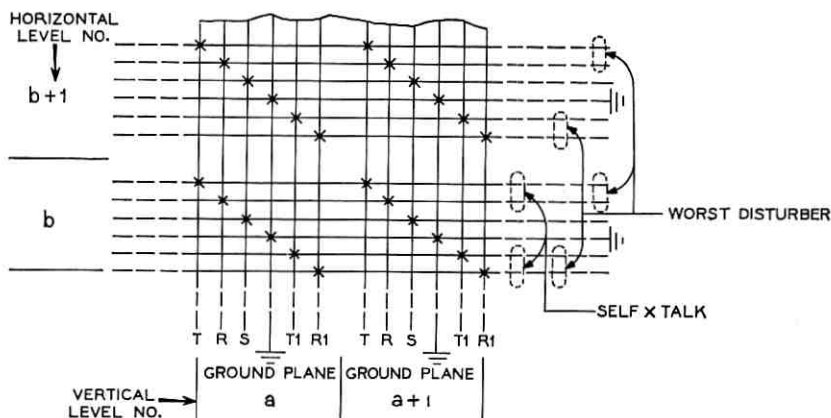


Fig. 7—Wideband crosstalk on crossbar switch.

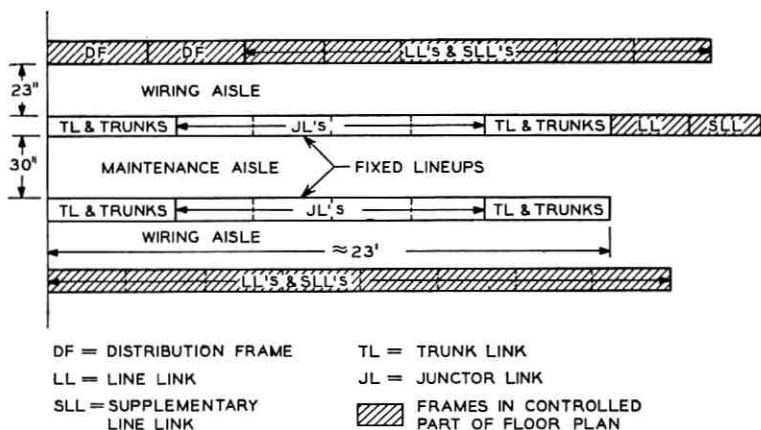


Fig. 8—No. 5 crossbar wideband (WB) switching area typical floor plan.

control exercised in this part of the floor plan is that a WBLL frame can be placed anywhere in the office, provided the sum of the interframe cabling from the WBDF to a WBLL frame and from this WBLL frame to a wideband junctor link frame (WBJL) is equal to 74 feet. This allows the distance from an office equalizer to a WBJL multiple point (star point) to be constant for any path, except for intraframe wiring variations. The 74-foot cable length dimension provides for reasonable office layout flexibility and is within the limits of the level plan.

The WBJL frames are part of the fixed floor plan. This arrangement allows for the rigid control of: the length and number of bridge taps; the distance from the cable equalizers to a star point; and the transmission path distance through the wideband network. A star point is located on each basic WBJL frame and is used to interconnect the verticals of the junctor switches as the office grows from the 100-output to the 400-output size. In the maximum size office the three interframe multiple cables range from 12 to 18 feet. Cable equalizers are adjusted to compensate for the average loss of these cables. Level variations from the average and not the actual loss of the bridge taps must be within the echo rating allocation for the switching network. As the office grows, capacitors that simulate the loss of these multiple cables are removed. Thus the cable equalizers do not have to be adjusted as the office grows.

The connection of the horizontal multiple between the junctor switch groups is made at the wideband trunk link frame. This ensures that

the length of the bridge tap caused by this multiple will have little variation regardless of the path chosen. A more direct connection of this bridge tap could have been made; however, the cable equalizers could not in this case compensate for the variation in loss among the paths.

The wideband trunk link (WBTL) frames are the other frames in the fixed floor plan. The interframe cable distances between the WBJL frames and the WBTL frames are fixed so that compensation for the bridge taps can be accomplished and the length restrictions imposed by the level plan can be satisfied. The WBTL and wideband trunks are mounted on the same double bay frame. The trunk link switches are mounted in one bay and the wideband trunks in the other. Connection between the horizontals of the WBTL switches and the trunks is made through an intraframe cable that terminates on a terminal strip in the middle of the WB trunk bay. This also provides flexibility for the addition of new features. The trunks can be assigned to switch horizontals by loose wire cross-connections made on the front of the bay. This technique of cabling reduces path length variation and is the final step in satisfying the level plan.

By applying the rules listed in Section 3.1.2 and by utilizing the fixed and controlled floor plan, both the video impairment allocations for the central office and the level plan are satisfied.

### 3.3 *Wideband Network Simulation*

A hardware simulation of the overall switching network was developed to determine the realizability of the plan; the simulation was not intended to characterize a typical *Picturephone* switching office. The simulated network was arranged so that initially it represented a best case wideband network from a transmission standpoint without concern for optimizing the cost or flexibility. The network was then modified in steps to minimize cost and maximize physical flexibility.

The test plan for the simulation was to determine if the best case network would satisfy the transmission requirements; if it could not, then testing would stop and a reevaluation of the overall transmission plan would be made. The best case network was satisfactory and the simulated network was modified to incorporate the standard manufacturing and installation techniques that allow the network to be physically practical.

Virtually every modification of the simulation network uncovered a different transmission degradation that was close to the switching

office allocation. As the tests progressed, design changes were made to lower these degradations until the cost trade-offs dictated that no further improvements could be made. As a result, the standard service designs represent a balance of satisfactory transmission with reasonable cost and physical flexibility.

#### IV. CALL PROCESSING

##### 4.1 Control Elements

As indicated earlier, the designs for *Picturephone* service use as much of the telephone plant as possible. This is particularly true with the call processing elements in the No. 5 crossbar system. Nonetheless, additional control functions and circuitry are needed. The following paragraphs describe some of these additional controls.

The network control for the WBL is provided by additional circuitry in completing markers 0 and 1. Only one completing marker out of the possible 12 is needed to handle the expected *Picturephone* traffic; for reliability, a second is modified for *Picturephone* service. The markers gain access to the WBL through new connectors. In order to establish the association between line appearances on the audio and video networks, a wideband line control circuit is provided. This circuit translates the audio line equipment location into a corresponding video line equipment location and provides access by the marker to the corresponding video appearance on the WBL. Wideband trunk circuits provide access by the marker to the video transmission paths and control the state of the video paths based on signals from the associated audio trunk.

In order to transmit and receive control signals to and from the WBRSS, a wideband remote switch signal control (WBRSSC) circuit is provided. To conserve cable pairs between the central office and WBRSS, the WBRSSC circuit encodes much of the information it receives. The signals transmitted from the central office are received by sensitive mercury contact relays and decoded to operate control relays and the select and hold magnets of the crossbar switches. The dc signals are passed over approximately 30 pairs of wires. (While a data link was considered as a way of transmitting these signals, it did not prove economical because of the short distances involved—centrex CO customers are rarely more than a few miles from the central office—and the relatively small number of signals required.)

## 4.2 *Special Signals for Picturephone Call Processing*<sup>1</sup>

### 4.2.1 *Video Supervisory Signal*

Since a *Picturephone* station can originate and receive both telephone and *Picturephone* calls, it is unnecessary (and wasteful of power) to turn the set on every time the station is used. Hence, it would be useful to enable the set on *Picturephone* calls only. Also, it is desirable to trigger the set to give a distinctive ring on incoming *Picturephone* calls. To provide both of these features, a special signal is sent to both calling and called *Picturephone* sets when a *Picturephone* call is being established. The signal, called the video supervisory signal (VSS)<sup>1</sup>, consists of the standard horizontal and vertical sync pulses of the video signal format with a constant "gray level" voltage between these pulses. VSS is coupled from a new signal source to the outgoing video pair via the wideband trunk circuits. The new VSS supply source uses the timing and logic integrated circuit substrate originally designed for the *Picturephone* set.

### 4.2.2 *Video Continuity Signal*

Because charging and supervision for *Picturephone* calls is controlled on the audio path, a test is required on every call to assure the integrity of the video path. To provide this assurance, a video continuity test is made while the marker is processing the call. The test is accomplished by using a 12-kHz sine wave generator and detector per marker and by employing "loop-backs"—switched connections between the video transmit pair and receive pair—at the interoffice wideband trunk circuits and at the *Picturephone* station set. The marker makes the test by applying a tone through the wideband trunk circuit onto the outgoing video pair. The signal is looped back at the distant point (the next trunk or the station set) and is returned on the incoming video pair where it is detected by the marker and checked to determine its amplitude level. The signal level must fall within a specified "window"—8.2 dB wide when checking to the station, 2.6 dB wide when checking to a distant trunk—before proceeding with call setup. Section 4.3 presents a more detailed discussion of how this signal is used.

The chosen frequency and level detection scheme does provide a reasonable check for those problems and equalizer misadjustments that are expected to be encountered most frequently. It could reasonably be argued that a single frequency signal does not provide a thorough check of the entire megahertz bandwidth. However, other signals such as those which are higher in frequency or which have



wide bandwidths are too expensive to generate and detect relative to their usefulness for facility checking.

#### 4.3 *Interaction of Controls and Signals*

As a way of describing how the various control elements and signals interact, a typical call is described that originates from a customer associated with a WBRs and terminates in a distant office.

When the customer at the *Picturephone* station originates a call by going off-hook, a dial tone marker in the No. 5 office connects the audio pair of the station set to a *Touch-Tone* originating register in the usual manner. When the register returns dial tone to the customer, the customer keys the desired number, prefixed with the “#” button if he intends the call to be a *Picturephone* call or without the prefix if the call is a standard telephone call. If the “#” is not keyed, the register bids for a completing marker and the call is handled in the usual fashion. If the prefix is keyed, the register bids only for completing markers 0 and 1, the only markers arranged to complete *Picturephone* calls. (The other markers are made to appear busy to this register on this call.) The register passes the keyed number, a signal to indicate that the “#” was keyed, and other information about the calling line to the marker. With this information, the marker interrogates the wideband line control circuit to determine whether the customer requesting a *Picturephone* call is in fact entitled to *Picturephone* service and whether the video appearance of the line is terminated at the central office or at the remote switch.

Using the called number, the marker tests the busy-idle status of *Picturephone* trunks to the desired destination and then chooses one of the idle trunks. Since this call is from a station with a WBRs appearance, the marker must also select an idle video link between the remote location and central office. This selection process is performed through the WBRSSC circuit via the wideband network connectors. Once the trunk is selected, the marker looks for idle channels through both the audio and video networks.

While in the process of selecting idle channels, the marker performs a continuity test over the trunk video transmission facilities. If the test is unsuccessful, a marker makes a second trial to set the call to a different trunk and then (if again unsuccessful) returns reorder tone over the audio facilities to the calling customer. If the test is successful, all crosspoints of the selected audio and video channels through the networks are closed except for the crosspoints of the WBRs and the audio and video line link stages. (While the end points of the network are logically associated as discussed in Section

2.1, it should be noted that the channels through each network are chosen independently and concurrently. Hence, the WBL is not a slaved network.)

When the crosspoints have closed, the marker applies a "false cross and ground" (FCG) check on both the audio and video channels. The audio test is the same as that now used in regular telephone service. The new video test checks that, within the network, none of the following conditions exists:

- (i) A false dc ground on any one of the four transmission leads.
- (ii) A false dc signal path between the transmit and receive portions of the channel being set up.
- (iii) A false dc signal path between the channel being set up and a channel in use.

Upon completion of the FCG test, the marker operates the remaining crosspoints and applies a continuity test toward the 0-dB loopback at the station set. A failure at this point will result in a reorder tone being applied from the audio trunk to the calling customer.

With a successful test, the marker signals the outgoing trunk to apply a 100-millisecond burst of VSS toward the calling set and then releases from this call. (The VSS signal is applied only as a burst since it is contemplated that future wideband intercept announcements, which inform the calling customer of some problem in setting the call, may include a video scene as well. Hence, the video scene would not be viewed by the calling customer if VSS were applied continuously.) The usual address and supervisory information is passed over the audio trunk to the next office.

At the terminating office, similar actions occur, but with a few differences. For example, the continuity check is made only to the subscriber since the incoming trunk has already been checked by the preceding office. In addition, VSS is applied continuously toward the called station, starting at least 100 milliseconds before ringing begins. The combination of VSS and the ringing signal causes the station set to give a distinctive ring and triggers the set into a state that will turn it on when the customer answers. The video path is cut through when the called party answers, and charging for the call begins.

## V. ADDITIONAL FEATURES

### 5.1 Operator Services

Operator assistance is provided for the *Picturephone* customers on special calls such as person-to-person, collect, and credit card calls. This assistance is provided by operators working at switchboards, called

3CL boards. Trunk circuits connect the No. 5 *Picturephone* office and the 3CL boards to provide the operator assistance. While the central office still switches a parallel audio and video path to these trunks for eventual cut-through to the calling and called *Picturephone* customers, only the audio signals are passed between the central office trunk and the 3CL board. This arrangement allows the operator to be located remotely from the central office without requiring *Picturephone* transmission facilities to the operator.

The operator can assist in two types of calls. The first, a dial zero call, is one in which the customer calls the operator by keying "# + 0" and the operator extends this call to the called customer. A dial zero trunk from the No. 5 office is used for this type of traffic. The second type, a delayed call, is one in which the operator calls both the calling and called customers. The operator uses a delayed call trunk from the No. 5 office.

The primary difference between a customer controlled call and an operator controlled call is video cut-through. In particular, when a customer dials his own toll call, the video path is cut-through when the called party answers. On operator calls, however, the video path is kept under key control by the operator.

## 5.2 Maintenance Features

In addition to the extensive testing arrangements already provided in the maintenance center of No. 5 crossbar, a number of unique arrangements have been provided for additional maintenance capability for *Picturephone* service.<sup>4</sup> A discussion of some of them follows.

### 5.2.1 Video Make-Busy

With the additional complexity of the video transmission facilities, a feature that prevents a *Picturephone* customer from originating or receiving *Picturephone* calls but allows him to have telephone communications is a useful maintenance tool. This feature, a video make-busy capability, is implemented with magnetically latching relays provided on a one per *Picturephone* line basis. With the relay operated, only telephone calls and *Picturephone* test calls can be connected to the line. The state of the relay can be changed by the 15A local test desk<sup>5</sup> or by the master test control (MTC) circuit in the switching maintenance center.

### 5.2.2 WBRs Test Arrangements

If a craftsman is working with a WBRs as part of a trouble-tracing procedure or routine testing, he should not have to call a switchman

at the central office to set up test calls; he should be able to make the tests himself. To give him this capability, a portable circuit was designed to work into a trunk test register circuit at the central office. With this portable set, the craftsman can set a number of switches and keys that indicate the called number, type of test, various test conditions, etc. The information, in the form of up to 30 digits, is outputted to the register using multifrequency (MF) signals. When the information is stored in the register, the register bids for a marker in much the same manner as the MTC circuit. As such, it can simulate a number of controls provided by the MTC.

#### VI. ACKNOWLEDGMENT

The material described in this paper is the result of the system planning and detailed design efforts of many people in the Local Crossbar Switching Laboratory, the Network Planning Center, and the Switching Engineering Center of Bell Telephone Laboratories.

#### REFERENCES

1. Urich, J. F., "The *Picturephone*® System: Switching Plan," B.S.T.J., this issue, pp. 521-531.
2. Clos, C., "A Study of Nonblocking Networks," B.S.T.J., 32, No. 2 (March 1953), pp. 406-424.
3. Brown, H. E., "The *Picturephone*® System: Transmission Plan," B.S.T.J., this issue, pp. 351-394.
4. Dougherty, H. J., Peterson, E. B., and Schachtman, M. G., "The *Picturephone*® System: Maintenance Plan," B.S.T.J., this issue, pp. 621-644.
5. Favin, D. L., and Gilmore, J. F., "The *Picturephone*® System: Line and Trunk Maintenance Arrangements," B.S.T.J., this issue, pp. 645-665.

# Customer Switching



## The Picturephone® System:

# Customer Switching Systems

By CHARLES BREEN

(Manuscript received September 15, 1970)

*Many customers for Picturephone® service will likely be business and industrial enterprises, having PBX, centrex, or key telephone service. This paper discusses switching systems for business customers, and describes how Picturephone service will be added as a natural extension of telephone service. This paper also serves as an introduction to the detailed papers which describe each of the customer switching system Picturephone arrangements.*

### I. INTRODUCTION

Customer switching systems furnish service to Bell System customers in every business category. Such systems range in size from only two or three telephones up to many thousands of extension telephones. Because of the potential value of face-to-face communications to business customers, facilities have been made available so that they can subscribe to *Picturephone* service as part of customer switching system service.

The customer switching category includes several basic types of service. One type, known as Private Branch Exchange (PBX) service, is obtained from switching equipment located on the customer's premises. It permits connections to be set up quickly between the telephone users at the same business location, without the need for connecting to, and switching through, a central office. PBX systems also permit the customer's employees to place and receive calls to and from the central office via switches which provide a form of concentration when fewer central office paths are needed than customer stations. In PBX service, all calls from the central office, and some to the central office, are completed with the aid of the customer's operator, or attendant, who provides many useful services such as accepting calls for the concern and directing them to particular employees, aiding in the placing of outgoing calls, and screening them when

desired by the customer. Figure 1 shows a modern attendant's position arranged to handle *Picturephone* service.

Small customer switching complexes which do not require attendant aids to the extent desired in PBX service are very numerous. Key Telephone Systems (KTS) are used to meet such customer requirements.

A third type of customer switching service, centrex, is a variation on PBX service which has been made available in the past decade. With one form of centrex service, called centrex-CU, stations are served by PBX-type equipment (on the customer's premises), and calls can be directly dialed to any other telephone in the exchange and toll network with the identity of the calling station recorded at the exchange or toll switching office when required for charging purposes. Also, incoming calls can be directly dialed to the station without attendant assistance. However, the attendant is still retained in centrex service to handle calls directed to the main number of the business concern or to render assistance on calls that need to be transferred or require other special attention. An equipment variation, generally known as centrex-CO, accomplishes essentially the same service by switching features included in the central office switching equipment.

## II. COMMON OBJECTIVES AND PRINCIPLES

In adding *Picturephone* service to customer switching systems the objective has been to make the new service appear to the customer as a natural extension of telephone service, i.e., to provide means so that the action required by station users or attendants to initiate, receive, or extend *Picturephone* calls is compatible with the operation performed on ordinary telephone calls. In line with this objective, customer switching systems with *Picturephone* service have been designed to be consistent with general system principles.<sup>1,2</sup> These include the ability to use common station equipment and common telephone numbers for both telephone and *Picturephone* calls, with the *Picturephone* switching function brought into play through the use of the identifying # prefix generated by the 12th button of a *Touch-Tone*® station set. Included also in the *Picturephone* customer switching system designs is a distinctive signal on incoming calls, created by a new tone ringer at PBX, centrex and key telephone stations and also by a distinctive lamp signal on key telephone stations.

As is the case with non-*Picturephone* customer switching systems, answer supervision is returned by the attendant or station, whoever





Fig. 1—A modern PBX or centrex attendant position, arranged to handle *Picturephone* service.

answers first, and this causes charging to begin and persist throughout the call at *Picturephone* rates. Incoming *Picturephone* calls in centrex service are prevented from being directly completed to non-*Picturephone* stations in order to avoid unwanted charges. However a PBX or centrex attendant can complete *Picturephone* calls to a non-*Picturephone* station, if desired. Similarly, customer initiated transfers from a *Picturephone* to a non-*Picturephone* station are possible.

When *Picturephone* service is furnished by centrex switching equipment on the customer's premises, that equipment must be arranged for automatic identified outward dialing (AIOD) and must home on a central office equipped with local automatic message accounting (LAMA) equipment. In a similar manner, when centrex service is derived from central office equipment, the central office must be equipped with LAMA. The use of automatic station identification and LAMA is required since it is not planned at present to route *Picturephone* calls via a centralized automatic message accounting (CAMA) office either for operator identification or automatic charge recording..

## III. SWITCHING SYSTEMS

Since there are a number of standard systems that provide customer switching service, it was necessary first to decide upon those systems that should be arranged for *Picturephone* service, and the general manner of incorporating the video capability in them. Systems were chosen on the basis of having the most potential use for *Picturephone* service, considering both present and future customers. The manner of adapting them—or the question of adapting them at all—became a function of the nature of the system and the economics of accomplishing the functions. In this connection it should be noted that no attempt has been made to provide facilities to serve all stations of large customer installations, since it was judged that these customers would not elect, at least initially, to subscribe to the service for all of their employees.

*Picturephone* service has been adapted to the 701 step-by-step PBX system which presently provides PBX and centrex service to a large number of Bell System customers. To minimize modifications of existing 701 PBX systems, an auxiliary switching arrangement, the 850A PBX<sup>3</sup> has been developed which is automatically called into use to handle both audio and video components of the *Picturephone* traffic generated by, or destined for, stations obtaining regular telephone service from the 701 PBX.

The 757 PBX system represents one of the Bell System's newer packaged crossbar PBX systems which can furnish PBX and centrex service. Since its use has been steadily growing, it too has been made capable of furnishing *Picturephone* service. Designs have been consolidated, and the 850A auxiliary switching arrangement designed for the 701 PBX is also capable of functioning in a similar manner with the 757 PBX.

*Picturephone* service capability has also been added to the No. 101 Electronic Switching System (ESS), presently the Bell System's most versatile customer switching system with respect to range of customer sizes and variety of service features. As described in a companion article, any one of the family of switch units comprising this system can be arranged to provide *Picturephone* service.<sup>4</sup> For this system, the video component of the service is handled through the use of a wideband switch unit controlled, via the audio switch unit, by the stored program information in the control unit.

The No. 5 crossbar system has been arranged so that *Picturephone* service capability can be made available to centrex-CO lines. An optional wideband remote switch controlled from the *Picturephone* central office can be used on the customer's premises.<sup>5</sup> As discussed

in Ref. 5, not all centrex features will be available on *Picturephone* calls until a later development. The No. 1 ESS will also be arranged for the service at a later date.

It was found that *Picturephone* service for KTS customers could be more economically provided with a new system rather than by modifying existing systems. Accordingly the 1P2 system has been developed for use where *Picturephone* key telephone service is desired.<sup>6</sup> The 1P2 system can also be used as a separate entity to furnish *Picturephone* service to PBX and centrex stations when their number does not justify modifying the major system.

Figure 2 illustrates the general method of adding video switching to the systems just discussed.

#### IV. PBX/CENTREX CUSTOMER OPERATING FEATURES

##### 4.1 General

In developing customer operating features for PBX and centrex *Picturephone* service, a basic precept has been that the video features in the different PBX and centrex systems be consistent with each other and coordinate with the "service package" concept currently underlying the PBX and centrex marketing philosophy. The latter means that each of the features in the current telephone service array will have, to the extent possible, a companion video feature, and these features will be optionally available in groups, or individually, to complement the regular telephone service packages or individual options, respectively.

Technologically, the video features are of interest primarily with respect to whether they are station user oriented or attendant oriented (in which case they generally affect the station user also). Accordingly, the features will be discussed from these viewpoints.

##### 4.2 PBX and Centrex Station Features

The basic communication functions available to a station user are: communication with another inside station and communication with someone outside. With *Picturephone* service, video capability will be present on directly dialed inside and outside calls in a straightforward simple manner. PBX and centrex systems are designed so that the digits dialed to identify the station, either on an inside call or when called from outside, can be the same whether the call is made on a telephone basis or on a *Picturephone* basis,\* except that on a

\* If a separate system is used to provide *Picturephone* only service, a common number for *Picturephone* and telephone service is, of course, not possible.

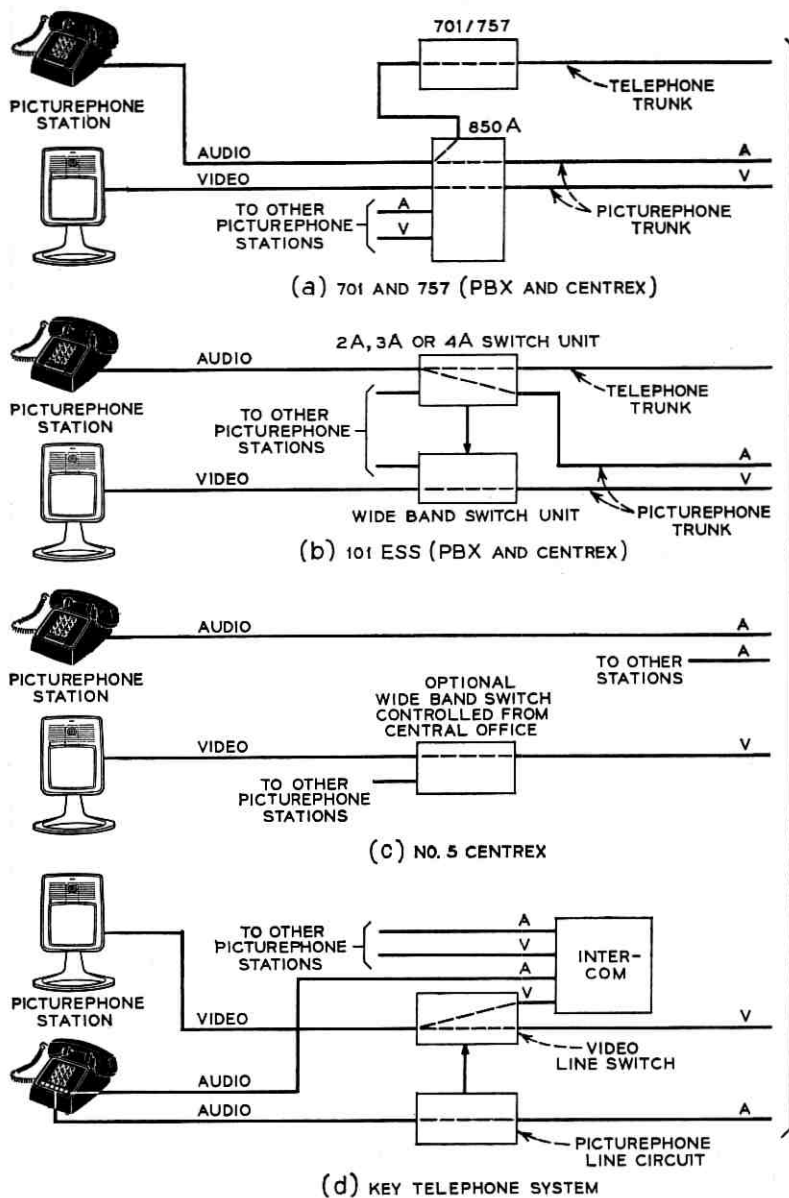


Fig. 2—General method of adding video switching to customer switching systems.

*Picturephone* call the station number must be preceded by the identifying # prefix. If the station number is preceded by the # prefix but the call is for a non-*Picturephone* station, it will be routed to a recorded announcement to avoid unwanted charges. If the *Picturephone* station to which the call is directed is in a hunting group with other *Picturephone* and telephone stations, the *Picturephone* call will hunt over only the *Picturephone* lines in the group. A *Picturephone* line will appear busy to the system for subsequent *Picturephone* and telephone calls if it is in use on either type of call.

PBX and centrex station features involving transfer functions of one kind or another result in some interesting video service features. Present systems can accomplish a number of variations of the transfer function, each variation being somewhat differently named depending upon the point in the call at which the function occurs, and the details of the function. Not all systems are capable of all variations. In general, the transfer function can occur (i) before the call is answered, (ii) at the instant of answer, or (iii) while the call is in process. Table I shows a tabulation of calls in these categories, together with a brief description of the telephone service and the video counterpart.

#### 4.3 Attendant Features

The ability to have an operator, or attendant, assist in the handling of calls is fundamental to PBX and centrex service and perhaps is one of the most important features. An attendant is an employee of the customer and, as such, functions as a "telephone receptionist" portraying her company's image. In the development of *Picturephone* service for PBX and centrex customers, provision has been made so that this concept can be enhanced where desired by the customer through the use of video telephony for the attendant.

It was decided to provide attendant service only with consoles such as the one shown in Fig. 1 since they not only represent the modern trend, but the alternative—cord switchboards—would be unnecessarily cumbersome to implement and operate. However, for those existing installations where cord switchboards prevail, the service can be implemented through the use of an auxiliary console for handling *Picturephone* calls only.

With respect to console operation, the objectives have been (i) that the attendant perform the same relative functions on *Picturephone* calls as on telephone calls handled at present standard consoles, and (ii) that the video operating procedures be identical with, or natural extensions of, the standard telephone operating procedures. The

TABLE I—SOME SERVICES INVOLVING TRANSFER FUNCTIONS

	Function*	Telephone Service	Picturephone Counterpart
Transfer occurs before answer	Call forwarding—busy line	Incoming direct inward dialed calls transferred to attendant if station is busy or does not answer.	Incoming direct inward dialed Picturephone calls transferred to attendant if station is busy or does not answer.
	Call forwarding—don't answer	Station pre-sets system to route incoming calls to another station.	Route Picturephone calls to another or night station if video equipped, otherwise route to recorded announcement.
	Call forwarding	System pre-set to route incoming calls to night station.	
	Night service	A call for a station in a group can be answered by another in the group by dialing special code.	Only video-equipped station can pick-up Picturephone call.
Transfer occurs at answer	Call pick-up	Stations can pick-up incoming trunk calls by dialing special code.	Not activated for Picturephone service.
	Trunk answer from any station	Called station can transfer incoming trunk call to another station.†	Transfer can be made to video or non-video station.
Transfer occurs while call is in process	Call transfer—individual	Outside party can be held while station calls another inside party.†	Same. If other party is video, visual communication can be made.
	Consultation hold	Station can add another inside party to call from outside after the "consultation hold" phase.†	Added station can be video-equipped or not. If video, the station initiating the add-on has visual communication with outside party.
	Add-on		

\* All functions not available with all systems.

† 101 ESS extends service to station-to-station and outgoing calls.

























principal extension of the service occurs when the attendant is equipped with a *Picturephone* set which permits her to see and be seen.

New communication procedures will be recognized to result from this added capability. This occurs because during the establishment of connections, the attendant is "in the middle" of communications between inside and outside parties, a process which involves accepting voice instructions from one of the parties, taking action to advance the call—an action which sometimes involves communication with the other party, returning sometimes to communicate with the originating party and, in some calls, communicating at the same time with both parties. Under these conditions of communication, a question arises as to who should see whom.

Simulation was resorted to in order to demonstrate and develop processes which intuitively appeared logical. With a few exceptions, the general rule evolved was that the video connection should follow the audio connection; that is, the parties that hear each other should also see each other. Where the attendant is involved with both calling and called parties at the same time, the video path to the attendant is not completed. Figure 3 shows "who sees whom" sequences during the process of some typical attendant operations.

It will be noted from the illustrations on Fig. 3 that another new feature has been introduced in the process of providing video service for the attendant. This is the capability to transmit a stationary image to video telephone parties outside the customer group when no live picture would otherwise be transmitted. These stationary images will represent patterns or concepts devised by the customer, and will be changeable by the customer. The images are generated by a video-image generator (VIG) and the capability can be provided at the customer's option.

In order to take care of different situations that can be encountered with various PBX and centrex systems, three modes have been developed for interconnecting the attendant console with the switching system. In the first, *Picturephone* calls can be diverted to a separate console as was previously mentioned for installations with dial auxiliary cord switchboards. In the second mode, *Picturephone* calls appear on specific pick-up keys on the same console that contains keys associated with telephone inputs. These methods are used with the 701/757 PBX approach and with the No. 5 crossbar system. In the third mode, used with the No. 101 ESS, both telephone and *Picturephone* calls are directed to common pick-up keys on the console. In each of

	PICTURE SEEN BY:		
	OUTSIDE PARTY	ATTENDANT	INSIDE STATION
ATTENDANT ANSWERS INCOMING CALL TO LISTED NUMBER. CALLER REQUESTS STATION			
ATTENDANT RINGS STATION. SPLITS CALL TO ANNOUNCE CALLER			
STATION ANSWERS. ATTENDANT ANNOUNCES CALL. STATION REQUESTS MORE INFORMATION			
ATTENDANT RETURNS TO OUTSIDE PARTY FOR INFORMATION			
ATTENDANT RETURNS TO STATION WHO AGREES TO ACCEPT CALL			
ATTENDANT CONNECTS OUTSIDE PARTY AND RELEASES SELF			
STATION RECALLS ATTENDANT REQUESTS INCOMING CALL BE TRANSFERRED TO ANOTHER STATION			
ATTENDANT RELEASES STATION			
ATTENDANT CALLS OTHER STATION			
OTHER STATION ANSWERS AND IS CONNECTED TO OUTSIDE PARTY			

LEGEND:



ATTENDANT



OUTSIDE PARTY



FIRST INSIDE STATION



OTHER INSIDE STATION

Fig. 3—"Who sees whom" during some typical attendant operations.



these modes of operation, uniform call handling methods have been worked out which achieve the basic operating objectives.

## V. KEY TELEPHONE SYSTEM FEATURES

### 5.1 General

The common system principles enumerated in Section II also underlie the *Picturephone* features of KTSs. However, specific service features of these systems differ from PBX and centrex service because of differences in the switching approach.

The KTSs provide service similar to PBX services insofar as they permit (i) controlled switching access between stations and lines to the central office (or to a PBX) and (ii) intercommunication with other stations of the customer group without the need for switching through the central office. Their operating features differ largely because of the difference in line access methods and the use of separate switching entities for the outside and inside functions stated in (i) and (ii).

The 1P2 KTS<sup>6</sup> developed for *Picturephone* service operates with standard key telephone sets and provides the basic communication features of systems used for audio-only service. Modifications and improvements have been made as dictated by the nature of the video technology.

### 5.2 Line Access Features

With conventional audio KTSs, one or more PBX or central office lines appear on keys at a telephone, and the station user obtains access to a line by operating a particular key. The lines may also be accessed by other key telephones, thus making it possible to connect more than one telephone, individually or at the same time, to a line. The station user may operate a hold key at the telephone to hold a call on a particular line and permit picking up another line.

When KTSs are arranged for *Picturephone* service, the same general capabilities for accessing one or more lines at one or more stations are present. However, to avoid video interference when more than one station is bridged to a line, only one video path is enabled at a time. Audio paths can still be bridged.

Line access arrangements have been developed in accordance with the above which contain improved features to enhance the value of the service to the customer. For example, where key telephone service is used by a customer having a secretary, the customer may wish to

have a *Picturephone* set without providing one for his secretary, but at the same time will want her to continue her usual practice of answering and sometimes initiating calls for him. The *Picturephone* KTS line access design provides for this by giving both stations audio access to the line, with video access only to the principal's *Picturephone* set. The secretary can then answer and initiate *Picturephone* calls on an audio basis and turn them over to the principal when desired who can continue on a *Picturephone* basis. To initiate a call, the secretary's telephone must, of course, be equipped for *Touch-Tone* calling in order to key the necessary prefix.

If a *Picturephone* set is provided for the secretary, she may enjoy full *Picturephone* service except that when both telephones are on the line at the same time, only one *Picturephone* set will be connected. The system is arranged to give preference to the principal's station. This type of service can be extended to two or more *Picturephone* lines and to more than two stations.

Another service variation allows a number of principal stations to share one or more lines. As in the previous concept, each principal might have a *Picturephone* set with a telephone set only for a secretary or other subordinate; although as before, the subordinate could also have a *Picturephone* set. With this service, when any one group (i.e., principal and subordinate) is using a line, access is denied to all other groups and the associated line lamp is lighted at the stations of the other groups. A line in use can be transferred from one group to another by placing it on hold and, through some separate means, such as an intercom, directing the other group to pick it up. With this service, incoming calls can be caused to ring and flash at a station user designated to act as attendant who, after answering the call, can place it on hold and, in a manner similar to the group transfer described above, can direct the proper station to pick it up.

### 5.3 Intercom Service

Intercommunication between *Picturephone* stations is obtained by a separate switching entity. Station access to the intercom system is via a separate button on the key telephone set. Two intercom arrangements have been developed. One, accessible to ten stations, allows one *Picturephone* conversation at a time; the other, of larger capacity, provides access for up to 27 stations and up to three conversations can be held at the same time.

#### 5.4 Add-On Service

Another party can be added to a *Picturephone* call that has been set up via the line access or intercom facility, in a manner generally similar to audio-only key telephone service. For example, an intercom call can be added to a central office call, or to a PBX call; or a PBX call can be added to another PBX call or to a central office call. When the calls are conferenced, the audio lines are bridged; but as in PBX service, the question of "who sees whom" arises. Since the station user originating the "add-on" has primary control of switching via the keys on his telephone set, the system has been arranged so that he can view and be viewed by the station associated with the particular access button depressed at his station. If he wishes, he may switch the video path from one to the other of the connected stations by depressing the associated access button.

#### VI. SUMMARY

This article discusses customer switching systems, their services and their features, with emphasis on how *Picturephone* service will be added to provide a useful and natural extension of telephone service. Customers served by video-augmented systems can provide for their employees the opportunity to conduct visual business communications with each other and with outside associates. These opportunities generally will be present on all communications normally provided by PBX, centrex and Key Telephone Systems.

#### VII. ACKNOWLEDGMENTS

The planning of *Picturephone* service for customer switching systems involved the contributions of many people at the American Telephone and Telegraph Company and Bell Telephone Laboratories.

#### REFERENCES

1. Dorros, I., "The *Picturephone*® System: The Network," B.S.T.J., this issue, pp. 221-233.
2. Urich, J. F., "The *Picturephone*® System: Switching Plan," B.S.T.J., this issue, pp. 521-531.
3. Anderson, H. P., "The *Picturephone*® System: The 850A PBX," B.S.T.J., this issue, pp. 585-604.
4. Brown, D. W., Horvath, J. R., and Paxton, T. S., "The *Picturephone*® System: No. 101 ESS," B.S.T.J., this issue, pp. 605-620.
5. Burgess, P. N., and Stickel, J. E., "The *Picturephone*® System: Central Office Switching," B.S.T.J., this issue, pp. 533-552.
6. Daskalakis, A., "The *Picturephone*® System: Key Telephone Systems," B.S.T.J., this issue, pp. 567-584.



## The *Picturephone*<sup>®</sup> System:

# Key Telephone Systems

By A. DASKALAKIS

(Manuscript received October 21, 1970)

*Initially, Picturephone<sup>®</sup> service will consist to a large extent of key telephone systems (KTS) that will permit subscribers to have multiline access to Picturephone Central Office or Private Branch Exchange (PBX) lines as well as to Picturephone intercoms. Add-on-conference capability will also be available. In this paper, Picturephone KTS operation, logic and control, and physical characteristics are described.*

### I. INTRODUCTION

A key telephone system (KTS) is a customer-controlled switching system which permits use of a station set for more than one Central Office, Private Branch Exchange (PBX) or intercom line by switching from one line to another. Generally speaking, customers can signal from one station to another, transfer calls between stations, hold a call on one line while talking on another, and converse between stations without making use of a Central Office (CO) or PBX line.<sup>1</sup> *Picturephone* service adds a new dimension to key systems in that now video as well as audio switching must take place to connect one *Picturephone* station set to a number of *Picturephone* CO or PBX lines or intercoms.

Five key system services will be offered for *Picturephone* service:<sup>2</sup> two Central Office or PBX line services, two intercom services and one add-on-conference service. The line services will permit (i) boss/secretary access to one *Picturephone* CO or PBX line,\* and (ii) shared access of one line by a number of boss/secretary groups. The first of these services is referred to as single-group line service while the second is referred to as multigroup line service. The intercom services will consist of (i) a single-link *Picturephone* intercom that

---

\* Hereafter referred to as "Line."

will provide dial service for up to ten stations on one audio/video link and (ii) a multilink *Picturephone* intercom that will provide service for a maximum of 27 stations on three audio/video links. Add-on-conferencing will be available between two lines or one line and one intercom station.

A *Touch-Tone*<sup>®</sup> telephone set and a speakerphone are an inherent part of the *Picturephone* station<sup>3</sup> and these may be used for normal telephone calling. *Picturephone* service is provided as an adjunct to regular telephone service, the only restriction being the provision for *Touch-Tone* signaling. Key system line services interface with the serving Central Office or PBX over three cable pairs: one pair for audio communications and two pairs for video.

From a subscriber's point of view, calls can be placed either on a video or audio basis in all of the key services mentioned simply by preceding (or not) the called number with the # digit. Incoming *Picturephone* calls are distinguished by a tone ringer (bell for telephone) and a red line lamp (white for telephone). Normal key system lamp flashing rates are used to indicate hold, ringing and busy. A single red and white lamp assembly is used under each line button for visual alerting.

## II. KEY TELEPHONE LINE SERVICES

### 2.1 General

Single Group service is intended to serve boss/secretary arrangements and permits simultaneous audio access to both stations. However, video access is provided on a hierarchical basis and if both boss and secretary are off-hook on the same line, only the boss will have a video connection. Multigroup line service allows a number of boss/secretary groups to share one *Picturephone* line. The first station within a group to seize the line will lock out all other groups.

### 2.2 Single Group Service

The block diagram in Fig. 1 shows a typical key system line installation. A single line picked up by a boss and a secretary is shown; however, additional lines may be connected to a station by adding a line circuit, a video switch and a video cable equalizer<sup>4</sup> per line.

To illustrate the operation of this service, incoming and outgoing video calls will be described. Telephone calls follow standard KTS practice and will therefore not be discussed. On an incoming *Picturephone* call, the CO or PBX applies the Video Supervisory Signal

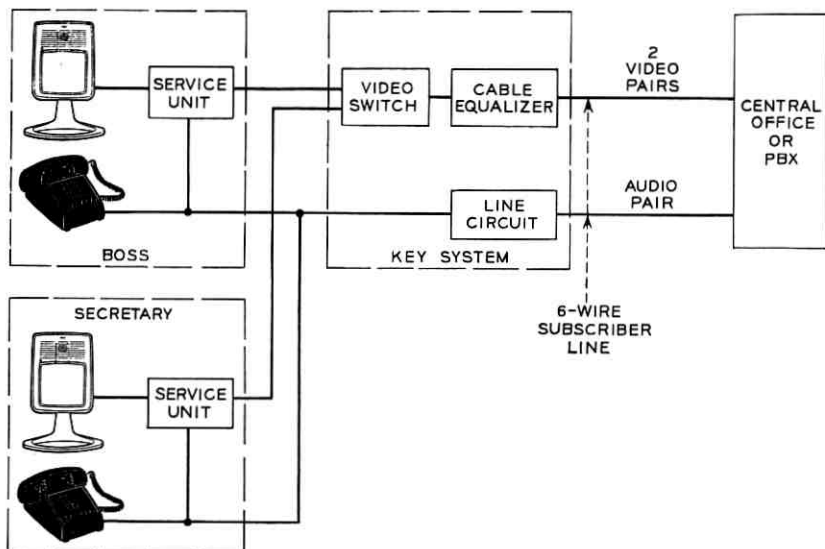


Fig. 1—Single-group *Picturephone* line service.

(VSS) to the incoming video pair at least 100 ms before applying ringing to the audio pair. After detecting VSS and within the 100-ms interval, the cable equalizer switches the line circuit into the video mode. When ringing is detected by the line circuit, a distinctive ring (tone ringer) and a flashing red lamp alert the station user. When the call is answered, ringing is retired, the red lamp goes steady and the video switch extends the video and turn-on-set signal to the called *Picturephone* station. If the call is placed on hold, the line circuit provides a winking red lamp to the telephone set. The video switch is released in the hold state resulting in a blank screen to both parties. Upon re-entering the connection, video as well as audio communications may be continued.

When a station is placed off-hook to originate a *Picturephone* call, the line circuit lights the corresponding white lamp in the set. The # button is depressed followed by the telephone number of the called station. When the CO or PBX detects that a video call has been dialed, a burst of VSS is sent to the video cable equalizer. The equalizer commands the line circuit into the video mode; the line circuit in turn switches the lamp in the telephone set from steady white to steady red. The system goes idle when all stations are on-hook.

A video loopback<sup>5</sup> is provided between the transmit and receive

pairs of the line at the cable equalizer circuit. This loopback is connected when the system is idle and is removed when the line circuit is in the video mode and a station is off-hook. Once removed, the loopback will not be reconnected until the system is idle, i.e., VSS not being received and all stations on-hook. This logic allows a video signal to be transmitted through the customer loop for transmission testing.<sup>6, 7</sup> It also prevents a person from seeing himself if the person to whom he is talking goes on-hook. In the event of a power failure, either at the line circuit, the cable equalizer, or the station set, loopback is removed to prevent the CO or PBX from completing video calls to that line.

### 2.3 Multigroup Service

Multigroup *Picturephone* line service was designed to provide shared-line service to a number of boss/secretary groups. The multigroup system as shown in Fig. 2 utilizes a line circuit, a cable equalizer, and a common control and lockout circuit. It shows boss/secretary groups one through nine sharing one *Picturephone* line. In this illustration, secretaries have video access; however, audio-only service may be provided for secretaries. When any one group (station) is using the line, access is denied to all other groups (stations). Any *Picturephone* group that inadvertently goes off-hook on a line that is in use will receive a busy tone. Possession of the line can be transferred to another group by placing the line on hold and through external signaling means, directing another group to pick up the line. Multigroup equipment arrangements will allow up to nine *Picturephone* groups to share a single line.

If the secretary within a group is equipped with a *Picturephone* station, single-group rules apply. That is, the customer must choose a hierarchical video order for the two *Picturephone* stations but both stations within the group have audio access.

Incoming and outgoing calls, insofar as distinctive audible and visual signaling and call completion are concerned, are handled the same way as in single-group *Picturephone* service with one exception. That is, on incoming calls one group (station) will usually be designated as the attendant group and all incoming calls will ring and flash at the attendant location. During ringing, the lamp associated with each line is lighted steady at all nonattendant stations. If a nonattendant station goes off-hook during ringing, it will receive a busy tone. When the attendant determines the station to which the calling party wishes to be connected, she may place the incoming call on hold and, through external signaling means, advise the proper station to



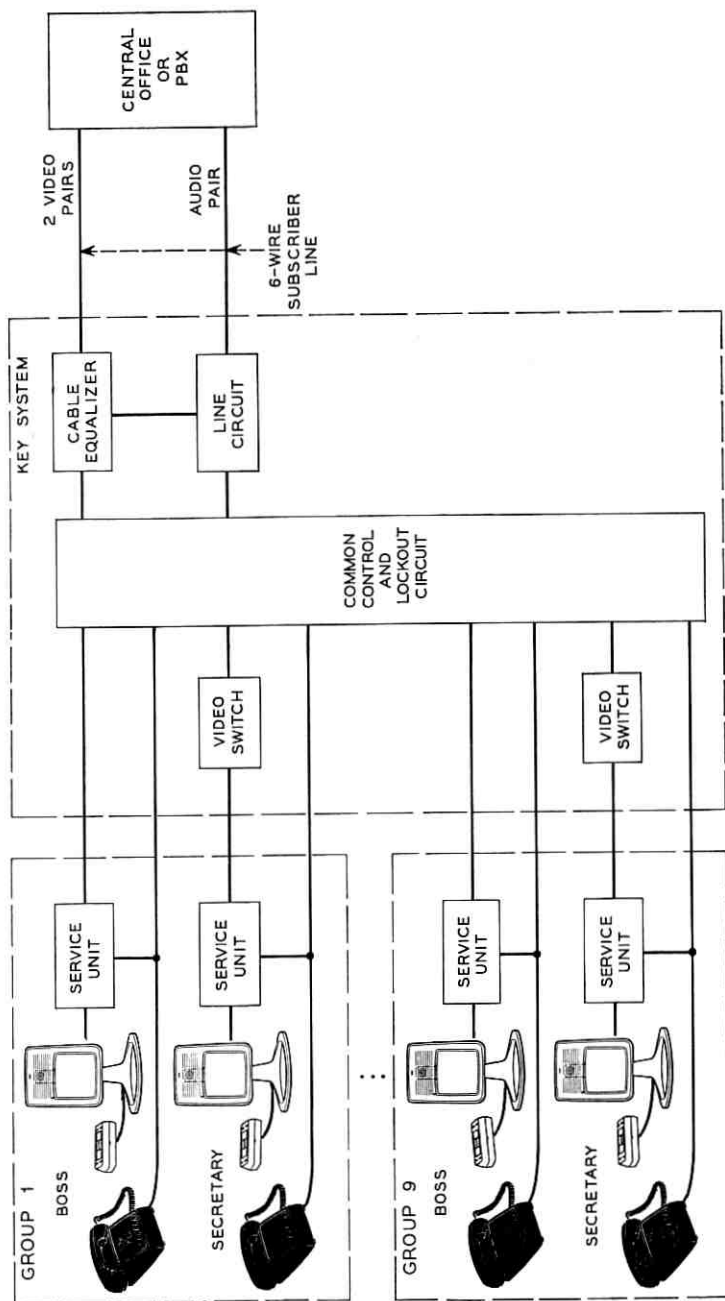


Fig. 2—Multigroup Picturephone line service.

pick up the line. When the alerted station picks up, the call will be transferred to that station and the attendant as well as all other groups will be locked out.

An optional Direct Station Selection (DSS) arrangement can be provided for the attendant to advise a station of an incoming *Picturephone* call. With this arrangement, the attendant operates the proper DSS button which activates a flashing red lamp and audible tone ringer signal associated with the *Picturephone* line to be picked up by that station. As above, when the station picks up, the call will be transferred and the attendant will be locked out.

An application of Multigroup service is given in Fig. 3. It shows three boss/secretary groups sharing two *Picturephone* lines (secretaries are audio only). A button appearance per line is required at each station. If the principal user of group 1 is off-hook on a line, the lamp associated with the line in use will light steady at each station appearance in all three groups. If either a group 2 or group 3 station goes off-hook on the same line, that station will receive a busy tone. If the secretary in group 1 goes off-hook, she will get audio access to that *Picturephone* line.

### III. KEY TELEPHONE INTERCOM SERVICES

#### 3.1 General

Two types of *Picturephone* intercom services will be offered to handle customer switching needs: single link and multilink service. Single link service is intended for the small user and will accommodate a maximum of ten stations. Multilink intercom service is aimed at the larger user and will serve a maximum of 27 stations on a maximum of three links. All basic key telephone features are provided on each of the two systems; i.e., pickup, hold, lamps, audible signals, add-on-conference, etc.

Intercom service is entirely separate from the type of line service chosen. For example, stations having access to the single link intercom can have either single-group or multigroup *Picturephone* line service or telephone-only line service. A button appearance is required at each station that has access to an intercom.

#### 3.2 Single-Link Intercom Service

This service provides private intercommunicating on a # plus one-digit dialing basis among a group of ten or less stations, some or all of which are equipped for *Picturephone* service. Single-link intercom

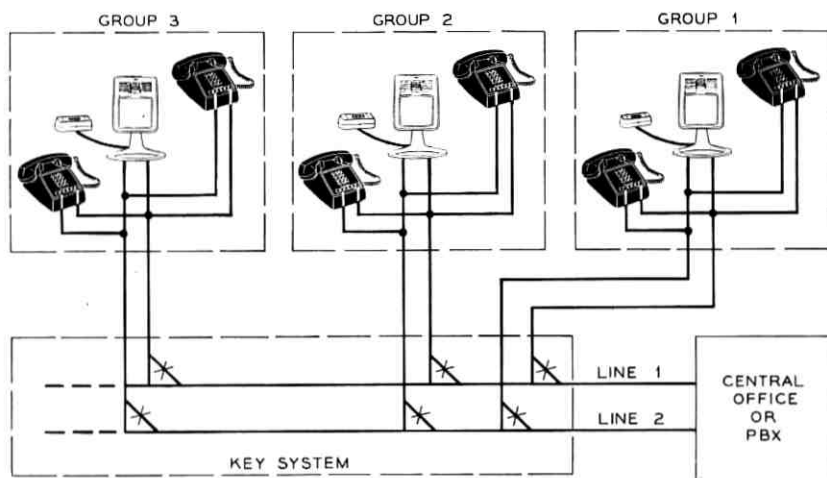


Fig. 3—Multigroup *Picturephone* line service—application.

service is provided by the units shown in Fig. 4, consisting of a station circuit per station, common control to allow access to one station and exclude all others, tone and lamp control and *Touch-Tone* dial receiving equipment.

The origination and termination of calls follow the procedures described for line services insofar as dialing and station alerting are concerned. If a call is in progress, the intercom line lamp will light steady at all stations having intercom access. If a station goes off-hook while a call is in progress, that station will receive busy tone.

The basic core of the single-link *Picturephone* intercom is the 1A2 dial selective intercom (audio only) presently available to the field. Video switching, lockout, and other video features have been added to result in a complete *Picturephone* intercom system. The features of the single-link intercom are privacy, system busy tone, hold, boss/secretary bridging of one intercom line appearance, station hunting from a primary to a secondary when the primary is busy on another line, and add-on conferencing. Rotary dial sets for audio-only stations are permitted.

When a station is placed off-hook to originate a call, the station and common control circuits are seized, preventing other stations from originating a call on the intercom. The station is connected to a battery feed circuit and to the terminated video path. The calling station receives dial tone and all stations receive a steady white lamp signal. If

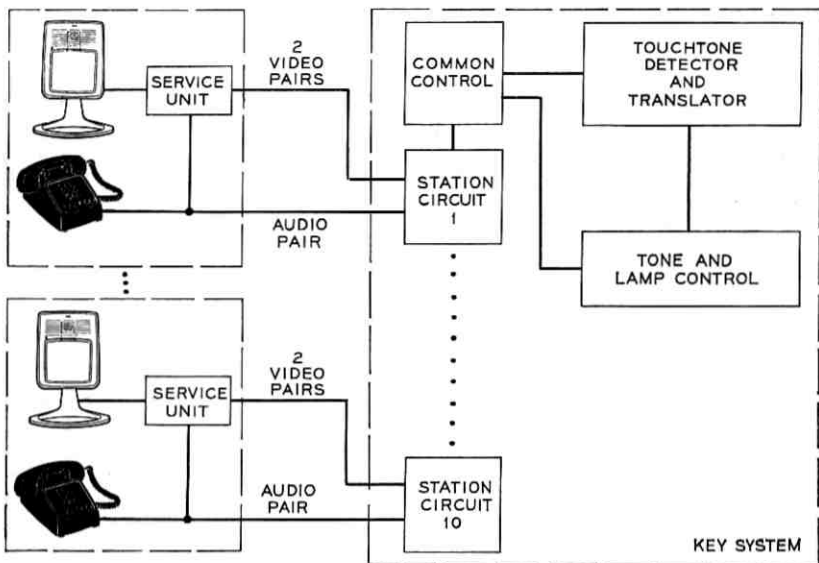


Fig. 4—Single-link *Picturephone* intercom.

the # digit is dialed, indicating a *Picturephone* call is being placed, the lamps change to steady red.

Upon completion of dialing, the common control circuit makes a busy test of the called party. If the called party is busy on another line, busy tone is returned to the calling station; if the called party is idle, ringing is applied. When the called station answers, the called display unit is turned ON, the video paths are connected together in proper transmitter-to-receiver fashion, and the ringer is retired. The system remains in the busy state until the last station is placed on-hook.

On an established intercom call, one of the talking parties may choose to answer an incoming call on a line while the other intercom party remains off-hook. After answering the incoming call, he may then return to the intercom as long as the other party remains off-hook. Therefore, intercom users are provided a means for holding an intercom call.

Boss/secretary bridging of one intercom line appearance and station hunting from a primary to a secondary when the primary is busy on another line are audio key system features and will not be covered here. The optional feature of add-on-conference, which is

available with the single-link *Picturephone* intercom, is discussed in Section IV.

### 3.3 Multilink Intercom Service

This system is intended to fill the intercommunication needs of customers who range in size from those who can be served by the ten-station single-link *Picturephone* intercom to those who require a *Picturephone* PBX but do not require full PBX features.<sup>8</sup> The multi-link intercom is a common control system designed to provide communications to a maximum of 27 stations. It features complete privacy, serves telephone as well as *Picturephone* equipped stations, and can be equipped for either two or three intercommunicating links. Standard key system features such as hold, boss/secretary bridging of one intercom line, and station hunting from a primary to a secondary when the primary is busy on another line are provided on an optional basis. CO or PBX lines can be connected to provide audio add-on and video transfer into the intercom.

#### 3.3.1 System Organization

Figure 5 is a block diagram of the multilink system. It consists of a system control and register, up to three link circuits, a switch matrix, and a station circuit for each station.

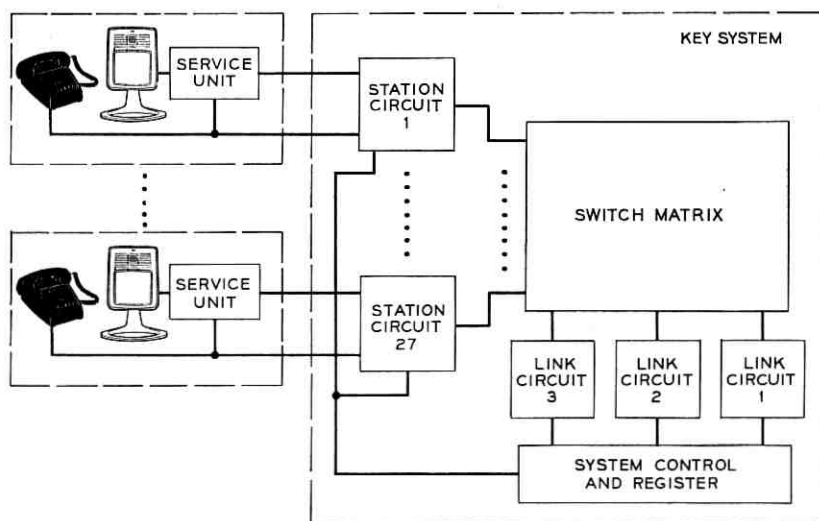


Fig. 5—Multilink *Picturephone* intercom.

The register contains the bulk of the system logic and consists of a *Touch-Tone* receiver, a selector circuit and a logic circuit for controlling the system. The link circuit performs the functions of call supervision and selective signaling. The station circuit connects the stations to the system and controls the functions of video, lamp and ringer switching. The switch matrix connects selected stations to a link circuit and locks out all other stations.

### 3.3.2 System Operation

3.3.2.1 *Telephone Calls.* Referring to Fig. 5, when a station goes off-hook, the register is seized and prevents other stations from originating a call at this time. The register identifies the calling station and also enables an idle link. The enabled link causes the operation of the matrix crosspoint associated with the calling party. At this point, the calling party's tip and ring leads are connected to the enabled link circuit through the matrix circuit. Tones (dial tone, ringback, station busy and permanent signal) are transmitted by the link to the connected station. The link circuit supplies loop current, detects rotary dial pulses and repeats dial pulses to the register. If *Touch-Tone* dialing is involved, as is required to originate *Picturephone* calls, the tones are capacitively coupled to the register.

Upon completion of dialing, the register makes a busy test of the called party. If the called party is busy, busy tone will be returned to the calling party. If the called party is idle, the crosspoint associated with the called station and the enabled link will be operated. The register will be released from the enabled link making it available to handle another call. Ringing and ringback tone will now be applied by the link circuit. After the called station answers, the link will remain in the busy state until the last station is placed on-hook.

3.3.2.2 *Picturephone Calls.* *Picturephone* calls are originated in the same fashion as telephone calls with the additional requirement of preceding the called number with the # digit. The register detects the digit and signals the link which in turn sets the station circuit into the *Picturephone* mode (i.e., red lamp, tone ringer). The viewing path through the switch matrix is controlled by each station circuit in such a way as to connect the transmitters and receivers of the calling and called parties in a transmitter-receiver combination in both directions. When both the calling and called parties have been connected to the switch matrix, the link circuit must prevent additional stations from being added in a *Picturephone* mode to the existing two-party call. Third and fourth parties can be added, audio only, to an existing call

by two methods to be discussed: i.e., DSS and register recall. The intercom call can be placed on hold which results in a distinctive winking red lamp to the user.

3.3.2.3 *Direct Station Selection (DSS)*. The DSS feature may be used to call a *Picturephone* intercom station by going off-hook, dialing the # digit followed by a depression of the DSS key associated with the called party. In addition, after a two-party call has been set up, either party may add stations to the call using the DSS feature (up to four parties total because of audio transmission restrictions). The third and fourth parties will be added as audio-only participants as previously noted. A DSS key is required for each specific station that a user wishes to signal directly.

When a user depresses a DSS key, the link involved is re-enabled thereby connecting it to the register for the called party busy test. If the called party is idle, its crosspoint becomes operated, the station circuit is switched into the ringing mode and ringback tone is transmitted to the parties already connected to the switch matrix. If the party is busy, station busy tone is returned to the calling party(s). Station status tones (busy, ringback, etc.) are retired by a second depression of the DSS key.

3.3.2.4 *Register Recall*. After a call has been established on a given link, either party, if equipped with a register recall button, may add stations to the call by depressing the register recall key (up to four parties total). In much the same way as in DSS, the involved link is re-enabled and connected to the register. In this instance, however, the register returns dial tone to the calling station. Dialing and the resulting status tones then occur as for a normal dialed call. The first two parties will have a video connection; the third and fourth parties will be brought in as audio-only participants. Station status tones (busy, ringback, etc.) may be retired by a second depression of the register recall key.

3.3.2.5 *PBX or CO Line Add-On Transfer*. Each PBX or CO line that is capable of being switched into the intercom may be connected to any one of the three system links. Assume a two-party intercom *Picturephone* call has been established and that one of the intercom stations wishes to add a CO party. The controlling intercom party would first place the intercom call on hold. He would next access a CO line and dial the desired party. When the CO party answers, the controlling station depresses the add-on key associated with the involved line which in turn causes the CO line audio pair to be bridged across the loop of the link in use. The controlling intercom station may then select

the party he wishes to view by depressing either the line or intercom key. The controlling party is seen by the party associated with the depressed key; however, the third party receives a blank screen. All three parties have continuous audio. Upon hang-up of either intercom station, the video from the added-on line is transferred to the remaining party.

**3.3.2.6 Paging.** The purpose of this feature is to provide an intercom user microphone access to a customer-provided public address system. The paging control circuit replaces a station circuit and, therefore, is represented by the station code it replaces. The circuit is the interface between the intercom and a customer-provided loudspeaker paging circuit. The paging control circuit automatically answers upon being dialed and disconnects when the intercom station disconnects (calling party control).

**3.3.2.7 Attendant Operation.** An attendant position may be provided and has the same appearance in the system as a normal station circuit. It may or may not be equipped for *Picturephone* service as desired by the customer. The position would normally be assigned the station code "0" but is not restricted to this code. If a system user does not have access to a PBX or CO line (audio or video) and an attendant position is equipped, the attendant can provide the intercom station access to an outside line.

#### IV. ADD-ON-CONFERENCE SERVICE

The add-on-conference feature is provided for single-group and multi-group line users as well as for single-link intercom subscribers. Conferencing is provided between:

- (i) 2 PBX lines,
- (ii) 1 PBX or CO line and a single-link intercom line, and
- (iii) 1 CO line and 1 PBX line.

The *Picturephone* multilink intercom provides a similar feature called PBX or CO line add-on transfer which has previously been discussed in Section 3.3.2.5.

Assume that a two-party *Picturephone* call has been established on a PBX line and that one of the parties wishes to add an intercom station. The controlling party would first place the PBX call on hold. He would then access the intercom and dial the desired party. When the intercom party answers, the controlling station would depress his add-on-conference key which would in turn cause the intercom audio pair to be bridged across the PBX line in use. The controlling station



would choose the desired video connection by depressing the line button associated with the station to which he wishes to be connected. He would see that station and that station would see him. The controlling station can switch from one line to another by depressing the appropriate line button. The station which is not seeing anyone is in turn not being seen by anyone. At the completion of the conference, the controlling party may continue to talk to either of the parties when the third party hangs up or he may drop both parties by hanging up himself.

#### V. POWER

Seven potentials are used in the circuitry of the *Picturephone* key telephone services:

<i>Potential</i>	<i>Uses</i>
-24 Volts Talk	Talking Battery Cable Equalizers
-24 Volts Signal	Switching Circuitry
+10 Volts	Red <i>Picturephone</i> Lamps
-10 Volts	White Audio Lamps
10 Volt AC	Interrupter Motor
18 Volt AC	Buzzers
117 Volt, 30 Hz	Ringng Supply

The DC potentials are obtained from solid-state diode rectifiers supplied with alternating current from a single core, multiwinding transformer, connected to commercial 60-Hertz single-phase power. Ringing power is provided by a 30-Hertz subharmonic oscillator operating on commercial ac power.

Reserve power arrangements have not been made for the *Picturephone* key telephone services; however, in the event of a power failure, telephone service will be functional. Outgoing telephone calls may be made in the usual way. Incoming telephone calls will cause line ringing to be applied to the main station of each line; however, lamp signals will not be available.

#### VI. EQUIPMENT ARRANGEMENTS

The *Picturephone* key telephone circuits consist of solid-state elements, miniature flat spring relays and assorted components assembled on circuit packs which plug into equipment units that in turn are wired

together into one assembly. The wired and factory-tested units are connected at the time of installation by plug-in cables and quick-connect wall-mounted connector blocks. These plug-in modular design features minimize the time spent on the subscriber's premises for installation and maintenance.

Two types of equipment units will be available to house the *Picturephone* key equipments: a wall-mounted key service unit (KSU) will accommodate the small installations, whereas floor-mounted 7-foot frames will handle the larger installations and the multilink intercom system. Using dedicated connectors, the KSU shown in Fig. 6 may be equipped with the following *Picturephone* services:

- (i) Two single-group *Picturephone* lines, each picked up by a *Picturephone* station, or
- (ii) One single-group line picked up by a *Picturephone* station plus a six-station single-link intercom, or
- (iii) One ten-station single-link intercom.

The KSU shown in Fig. 6 measures 27 inches high by 16 inches wide by 12 inches deep and is equipped with one single-group line and a six-station single-link intercom.

Figure 7 shows the arrangements available in the frame-mounted equipment arrangement. Each of the top three carriers provides for: (i) single-group service for two *Picturephone* lines picked up by one station or one *Picturephone* line picked up by one station, and (ii) add-on conferencing. The fourth carrier provides the same service as the KSU. A total of eight single-group *Picturephone* lines may therefore be accommodated on this frame. Connecting cables are used to complete the connections from the frame to connecting blocks mounted to the wall. The frame shown in Fig. 7 measures 7 feet high by 24 inches wide and has carriers 1 and 4 equipped with circuit packs.

Multigroup *Picturephone* line service is provided by a seven-foot frame similar to the one shown in Fig. 7 except it is arranged to accommodate a total of six *Picturephone* lines. The top two carriers are wired to provide four lines of single-group *Picturephone* service. The bottom two carriers furnish service for one multigroup *Picturephone* line each. Each multigroup carrier is arranged so that shared service can be offered for up to nine *Picturephone* groups (stations).

Basic multilink intercom service will be provided on a seven-foot frame to accommodate a maximum of 27 stations and a maximum of three links.

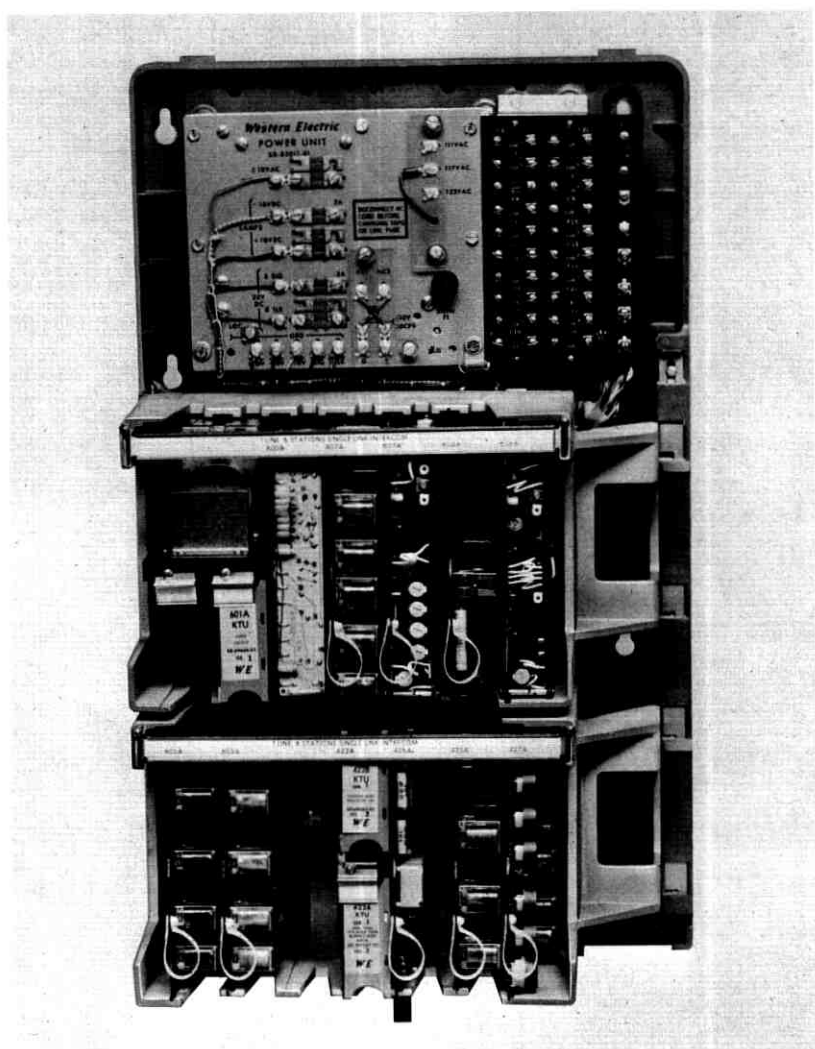


Fig. 6—Key service unit equipped with one *Picturephone* single-group line and a six-station intercom.

#### 6.1 *Plug-in Circuit Pack Physical Characteristics*

Two sizes of circuit packs are used in the *Picturephone* KTS: one is 3.5 inches high by 5.5 inches deep and the other 7.5 inches high by 5.5 inches deep. Figure 8 shows typical examples of the two types of packs.

The double-sided etched epoxy-clad metal boards are gold-plated with either 40 or 80 contacts depending on the size of the circuit pack. The highest component on any of the circuit packs is 1.25 inches with spacing between circuit packs held to 1.5 inches.

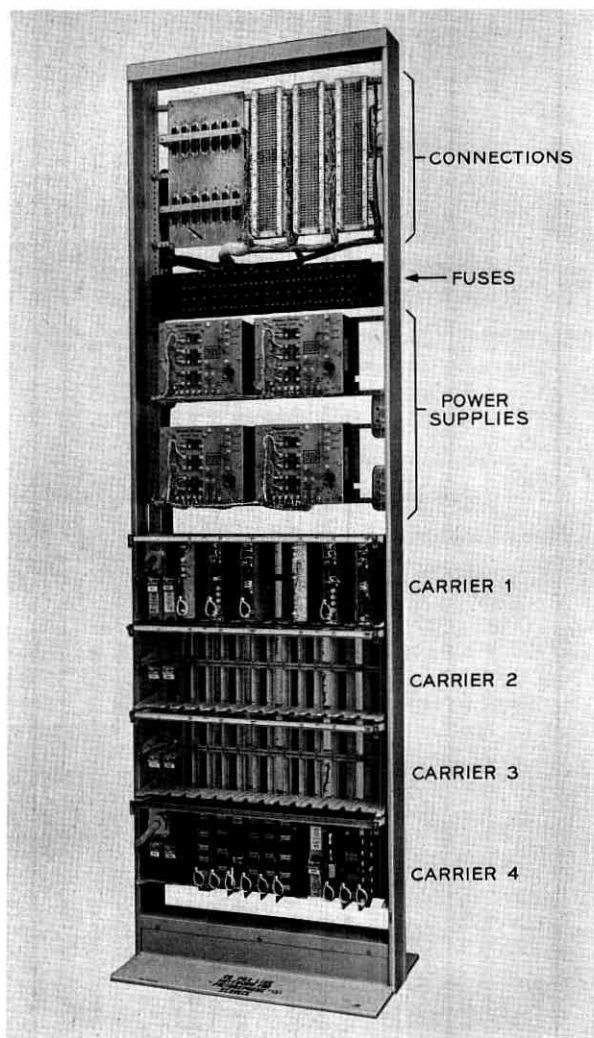


Fig. 7—Frame-mounted equipment arrangement.

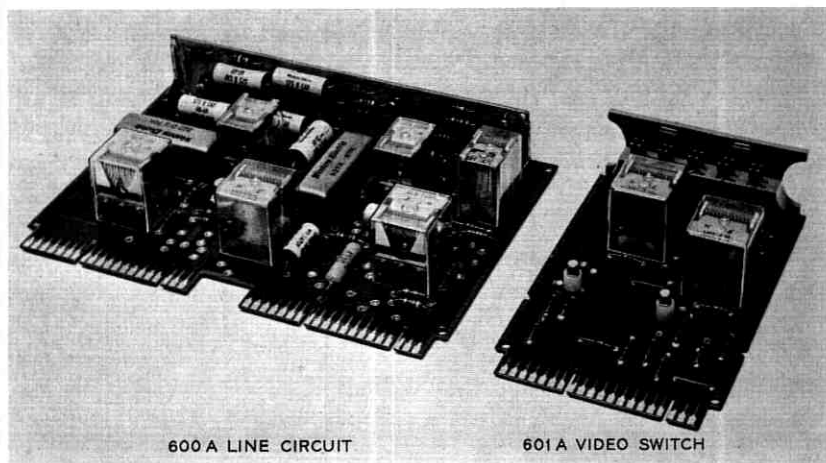


Fig. 8—Circuit packs.

### 6.2 Apparatus Mountings

The circuit packs plug into die-cast aluminum apparatus mountings consisting of trays that are 23 inches wide. The card guide castings on the top and bottom of the tray have slots on 1.5 inch centers to accept the circuit packs. A connector mounts into slots at the rear of the casting to mate with the circuit pack plug. This connector is arranged for up to three levels of wire wrapping.

### 6.3 Wiring and Cabling

The equipment within each apparatus mounting is wired to accept the full complement of circuit packs and is tested before shipment to the field. The various equipment features and options are implemented by inserting or removing a circuit pack or by replacing one circuit pack code by another in dedicated slots.

Only two cable pairs of standard inside wiring cable are required to carry the video signal between a *Picturephone* station set and the *Picturephone* key equipment regardless of the number of *Picturephone* lines or intercoms associated with the *Picturephone* station. This is possible because all the video switching functions take place at the *Picturephone* key equipment location. Audio connections associated with the *Picturephone* lines and intercoms, such as tip and ring, are

made using the standard key telephone system wiring plan. Pair assignment within the cable was made to reduce video self-crosstalk.<sup>7</sup>

The cable length between a *Picturephone* station and the *Picturephone* key equipment is restricted to a maximum of approximately 400 feet of 24-gauge cable. Beyond this distance, video equalizers<sup>4</sup> are required in order to satisfy overall transmission objectives.<sup>7</sup> If the cable length exceeds 1000 feet of 24 gauge, additional circuits are required to extend the control and lamp functions to the *Picturephone* station. The key systems described herein and illustrated in Figs. 1-5 are restricted to cable lengths of up to approximately 400 feet. Circuits and equipment arrangements to extend the cable length to beyond 1000 feet are currently under development.

#### VIII. ACKNOWLEDGMENTS

The design of the *Picturephone* key system described herein has involved a number of persons. In particular, Messrs. G. D. Weed, R. D. Reese, G. E. Brumfield, and M. K. Bunce contributed in a major way. The author wishes to acknowledge other contributors to the circuit design, physical design and systems planning areas too numerous to mention, but without whose help and talents this work could not have been completed.

#### REFERENCES

1. Wood, W. F. B., "Key Telephone System; The Latest Chapter," Bell Laboratories Record, 44, No. 3 (March 1966), pp. 85-88.
2. Breen, C., "The *Picturephone*® System: Customer Switching Systems," B.S.T.J., this issue, pp. 553-565.
3. Cagle, W. B., Stokes, R. R. and Wright, B. A., "The *Picturephone*® System: 2C Video Telephone Station Set," B.S.T.J., this issue, pp. 271-312.
4. Brown, J. M., "The *Picturephone*® System: Baseband Video Transmission on Loops and Short-Haul Trunks," B.S.T.J., this issue, pp. 395-425.
5. Urich, J. F., "The *Picturephone*® System: Switching Plan," B.S.T.J., this issue, pp. 521-531.
6. Dougherty, H. J., Peterson, E. B., and Schachtman, M. G., "The *Picturephone*® System: Maintenance Plan," B.S.T.J., this issue, pp. 621-644.
7. Brown, H. E., "The *Picturephone*® System: Transmission Plan," B.S.T.J., this issue, pp. 351-394.
8. Anderson, H. P., "The *Picturephone*® System: The 850A PBX," B.S.T.J., this issue, pp. 585-604.

# The Picturephone® System:

## The 850A PBX

By H. P. ANDERSON

(Manuscript received September 21, 1970)

*In this paper we describe the system organization, operation, and apparatus of the 850A PBX, a Picturephone® Switching System. The objectives of the 850A PBX in the Bell System Picturephone service plan and the design approach taken are discussed.*

### I. INTRODUCTION

#### 1.1 General

For business people, person-to-person conversation is the most complete means of communicating; therefore, the first major users of *Picturephone* service are likely to be business persons.

Switching systems such as a Private Branch Exchange (PBX), or a key telephone arrangement are an integral part of most business communications set-ups. To obtain full utilization of the existing telephone network, *Picturephone* service is to be added as an integral part of telephone service; therefore, *Picturephone* service must be added to these systems serving the business community.<sup>1</sup> The 850A PBX has been designed to provide *Picturephone* service to customers served by 701B or 757A electromechanical PBXs.

#### 1.2 Objectives

The objective of the 850A PBX is to provide *Picturephone* service on 701B and 757A types of electromechanical PBXs, step-by-step (SXS) and crossbar (X-bar) respectively, with a common machine. In addition, the 850A PBX is to provide the same services on *Picturephone* service that the customer has on telephone; therefore, the 850A provides *Picturephone* service on all the Communications Service Packages, series 100, 200, 300 and centrex (CU) I and II.

Flexibility of the 850A in working with either SXS or X-bar PBXs is achieved by using the 850A in parallel with the existing customer telephone switching system. In the 850A *Picturephone* telephone sys-

tem, normal telephone calls generated at a *Picturephone* station are switched to and carried by the telephone switching equipment; all *Picturephone* traffic (including the audio) is processed by the 850A. Because of this parallel operation, the 850A performs the same switching functions required by a voice PBX, and it has the additional circuits required for video switching and transmission. Since the 850A is a complete PBX, it can be modified to operate as a stand-alone *Picturephone* PBX, i.e., provide only *Picturephone* service to a customer if he so desires.

The 850A PBX is a solid-state machine utilizing a building-block equipment arrangement which allows for easy growth from 16 to 89 *Picturephone* lines. Growable switching networks are obtained by the use of plug-in ferreed packages. Special maintenance features include in-service checks such as guaranteeing a good loop to a *Picturephone* station, before connection and ringing on incoming calls.

The 850A trunk circuits have been designed to operate over PBX loop ranges up to 2000 ohms and are compatible with Unigauge Central Office (CO) lines. Allocation of impairments on the video transmission of the 850A are given in Ref. 2. The 850A has video equalizers on the receive pair from each station and each CO line. A Key Telephone System (KTS) equalizer is used in the attendant video loop, this equalizer is used to provide video loopback independent of the optional attendant display set.

To achieve low manufacturing costs, the logic and circuit pack sizes were chosen to be the same as a high production PBX—the 800A.<sup>3</sup>

The utilization of this system with the major electromechanical PBXs and the similarity of the apparatus with other electronic PBXs allow uniformity of manufacture, installation, and maintenance.

## II. SYSTEM DESCRIPTION

### 2.1 General System Arrangement

A block diagram of the *Picturephone* PBX system is shown in Fig. 1.

*Picturephone* station equipment<sup>4</sup> consists of a 12-button *Touch-Tone*<sup>®</sup> telephone, a display unit, and a control unit. The 12-button *Touch-Tone* telephone is required for *Picturephone* service. The *Picturephone* service user initiates a video call by pressing the 12th button designated #, and then "dials" the regular telephone number of the person he is calling. A normal voice-only call is originated in



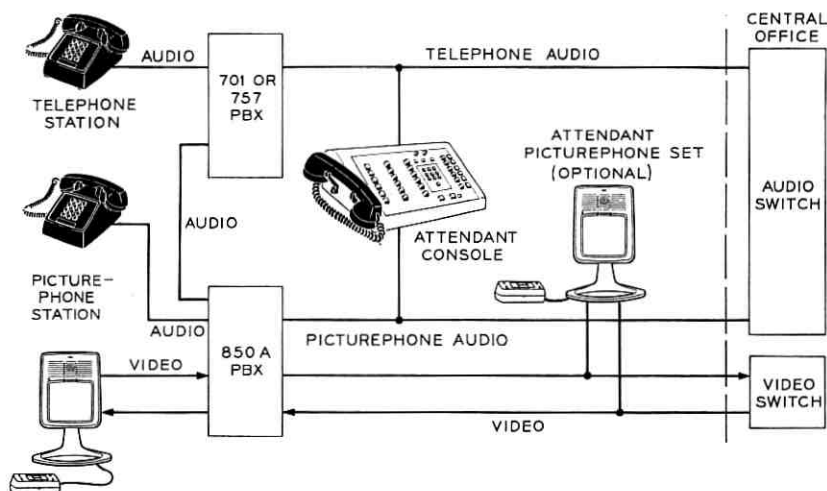


Fig. 1—System interconnection plan.

the usual manner by dialing the number without the # prefix. An incoming *Picturephone* call is characterized by a distinctive ring created by a tone ringer.

In the initial phase of *Picturephone* service, only the No. 5 crossbar central office will be modified to switch video calls; this capability will also be provided by the No. 1 Electronic Switching System (ESS) in the future. *Picturephone* service customers served by other types of telephone central offices such as panel, SXS, etc., will have their *Picturephone* facilities routed to the nearest No. 5 or No. 1 ESS office modified for *Picturephone* service.

No modifications are required on the existing 2-wire voice loops that connect a customer's telephone to the on-premise switching equipment. The ON-OFF switchhook signals, *Touch-Tone* dialing signals, and the voice portion of *Picturephone* calls are transmitted over the voice pair. In addition, two pairs of wires in the standard telephone cables are assigned for the transmission of the video portion of the call, one pair for the transmission in each direction. Video equalizers<sup>5</sup> are inserted at approximately one-mile intervals between the PBX and the central office. A *Picturephone* call to the local central office is established over a dedicated six-wire *Picturephone* trunk. At the local central office, the voice pair is connected to the existing telephone switch in the conventional way. The video quad is con-

nected to a separate four-wire switch which is controlled by the telephone switching system; the audio pair of this six-wire trunk is never used for voice-only traffic.

All the peripheral equipment and systems to the 850A are discussed in detail in other articles in this issue.

### 2.1.1 System Operation

To illustrate the operation of the system shown in Fig. 1, when a call is initiated by an "off-hook," the two-wire audio loop is connected to the register circuit of the 850A and to a line finder or register in the associated PBX, causing dial-tone to be returned to the station user. The detection of the first "dialed" character as an # by the 850A register causes this system to dismiss the regular PBX and to apply a busy indication to the line appearance in the telephone system. Outgoing *Picturephone* calls are carried over the dedicated six-wire trunk and processed by the central office (the central office does not require a # signal in this case since this equipment can only be seized by a *Picturephone* call). Alternatively, if the called number is not prefixed by #, the 850A register releases itself and applies a busy indication on the 850A line appearance. The call will then be handled as a normal voice-only telephone call in the usual manner by the associated PBX. A regular Direct-Outward-Dialed (DOD) telephone call initiated by a *Picturephone* station, after processing by the associated PBX, will be connected via the regular audio trunk to the serving telephone central office.

850A *Picturephone* lines can also be voice-only extensions equipped with either a *Touch-Tone* or a rotary dial to provide such services as secretary's pickup. A video call can be answered on such an extension; however, in this case, the caller will have a blank screen until the call is picked up by a *Picturephone* set. Rotary dial extensions can only initiate telephone calls but can transfer a *Picturephone* call to another *Picturephone* line by calling the attendant who, in turn, will effect the transfer.

Options which are selected by the customer include attendant video capability and a Video-Image-Generator (VIG) which transmits a fixed image to the calling party while the attendant processes his call.

### 2.2 Common Control

The 850A PBX is a solid-state wired-logic, common control switching system, the block diagram of which is shown in Fig. 2.

The common control is an asynchronous circuit with the exception

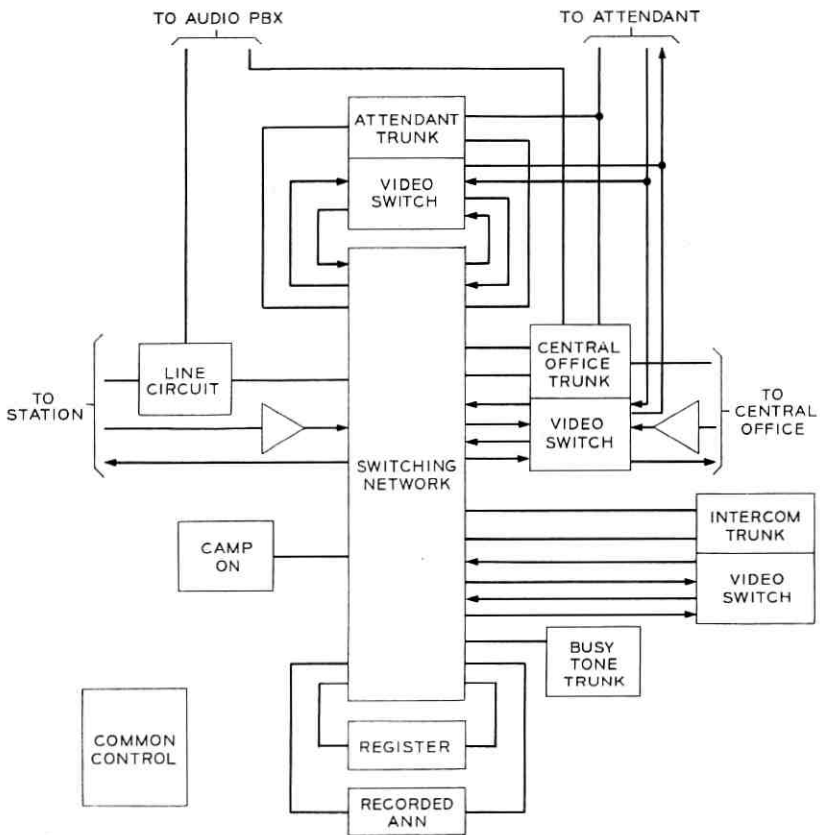


Fig. 2—The 850A PBX block diagram.

of the line-scanner portion which is clocked at a 10 kHz rate. The common control provides the necessary logic for establishing connections between *Picturephone* service circuits through the *Picturephone* switching network.

A "service circuit," such as a line circuit, trunk, register, etc., bids for the common control which determines the mode of operation necessary to provide service to this circuit. After identifying the requesting circuit, the common control locks out all other requests until the initial request has been satisfied. The common control examines the available data from the requesting circuit, marks it on the switching network, and determines what other service circuits, if any, are needed to satisfy the request. An idle circuit of the type requested is selected and also marked on the network. With the two marks on the opposite

sides of the network, the common control enables the network control which establishes the connection. The common control then resets and is ready to receive the next request.

The common control is normally in the idle state. From the idle state, the common control can enter four different functional modes to serve requests. These modes in order of service preference are: read-register (RR), line-dial tone (LDT), camp-on readout (CORO), and attendant-trunk release (ATR). A preference order is assigned to the functional modes in case of simultaneous requests. During the handling of multiple calls, the most important type of request is served first.

The read-register mode is requested by a register when its contents are ready for readout. Readout occurs when the register has the complete number of digits (3 or 4), or a recognizable 1-, 2-, or 3-digit code, or if the register has timed out. The RR mode involves the connecting of the calling circuit to the called circuit.

Line-dial tone mode is requested by circuits that require a register connection. Scanning is required because of multiple circuits requesting service at the same time. These requests include: stations going off-hook, DID trunk seizures by the central office, call-transfer-individual, dial-conference, and attendant trunk seizures. During the LDT mode, requesting circuits are connected to a register through the switching network.

The Camp-On feature is standard with PBX series 200 and 300 and optional with Centrex I and II. Therefore, the camp-on mode is provided only when it is required, and is requested by the camp-on circuit periodically when there is a number stored in one of the three camp-on stores. The CORO mode checks the state of the stored line, and if idle, connects the camped-on trunk to that line.

Attendant-trunk release mode is requested by the attendant trunk when the attendant releases from the trunk and both sides of the trunk are connected to other circuits through the switching network. The function of the ATR mode is to release the attendant trunk from the connection and to establish a new network connection directly between the two circuits previously connected to the attendant trunk.

In addition to providing the logic for the connection of service circuits, the common control performs other functions associated with *Picturephone* calls.

One of these additional functions is related to the identity of a *Picturephone* station. All *Picturephone* stations in the system must be assigned to the same hundreds group of numbers; this hundreds group

is variable from system to system. Any video call to a number not in this hundreds group, e.g., a telephone number or a vacant code, is routed to an intercept-recorded announcement. A listed-directory number (LDN) option is provided which allows the LDN for a Centrex-CU System to be in a different hundreds group.

The common control ascertains the class-of-service of a particular *Picturephone* line. Each *Picturephone* station may be assigned to any one of six classes of service. These classes are: CO nonrestricted, dial-conference nonrestricted; CO nonrestricted, dial-conference restricted; CO toll-diverted, dial-conference nonrestricted; CO toll-diverted, dial-conference restricted; CO restricted, dial-conference nonrestricted; CO restricted, dial-conference restricted. One additional class-of-service is provided for use by *Data-Phone*<sup>®</sup> stations. This data class-of-service is limited to *Data-Phone* stations and is always dial-conference restricted.

Common control also determines when a video-continuity test (VCT) should be made on an established connection. If the video path does not pass this test, the common control will cause the calling party to be routed to an intercept-recorded announcement; this is the same intercept that is used for calls to vacant codes and the wrong hundreds group.

Station hunting is also controlled by the common control. *Picturephone* stations may be arranged for one-way or two-way hunting as required to be consistent with the telephone PBX.

The common control is arranged for automatic reset under certain abnormal conditions. If any operation takes longer than an expected amount of time, the common control resets. If the common control fails to reset, or the network relays fail to return to normal, a major alarm signal is sent to the Fuse, Alarm, and Emergency Transfer Circuit, resulting in the transfer of all *Picturephone* lines not in use at the time to the existing telephone switching system, thus providing at least telephone communications.

### 2.3 Register

Dial Pulse and *Touch-Tone* phone reception is accomplished in the 850A by the register circuit. Connections to the register are made through the switching network. Line-side circuits are connected to trunk-side terminations of registers, and trunk-side circuits are connected to line-side register terminations. The register functions to receive dial pulses or *Touch-Tone* phone signals from either line-side or trunk-side circuits; after receiving the entire number, the register

signals the common control. The information in the register is then used, under the direction of the common control, to complete the required connections within the PBX. In the event that the called station or trunk is busy (including the busy-tone trunk, if provided), the register returns busy tone to the calling party. Upon connection of the register, the tip, ring, sleeve and "P-detect" leads are cut through to the calling line. This connection grounds the sleeve lead which operates the cutoff (CO) relay in the line circuit. The operation of the CO relay connects a tip, ring and sleeve lead to the associated electromechanical PBX which appears to this PBX as an origination. The voice PBX after finding the line returns dial tone to the station. If the first character "dialed" is the *Picturephone* prefix, #, the register detects this signal. Detection of the # causes the register to place a signal on the "P-detect" lead which in turn opens the tip and ring and grounds the sleeve lead to the electromechanical PBX. Grounding of the sleeve lead is done to provide the proper busy indication for incoming telephone calls to this station. The "P-detect" signal also allows the connection of the battery feed and supervision circuits of the 850A to the line; the *Picturephone* call can then proceed. If the register detects that the first character "dialed" is not the #, it places a ground on the sleeve lead of the 850A and then releases from the connection. The call in this latter case will be processed as a telephone call by the associated PBX. In the event the 850A is modified as a stand-alone *Picturephone* System, the register returns dial-tone if the first character "dialed" is not a #. If telephone stations are connected to a stand-alone 850, voice calls must be preceded by #.

The register is arranged to detect the # and to receive a three- or four-digit code for station lines; two- or three-digit code for conference calling; and a one-digit code for CO trunks and attendant trunks. The calling party has approximately 10 seconds to "dial" the first digit after the application of dial tone. If this 10-second interval is exceeded, the register drops the connection and allows the 701(757) PBX to time out and return fast busy tone (or attendant intercept). Subsequent digits must be "dialed" with a time interval no longer than 10 seconds between digits. If this interval is exceeded, after the #, the register times out and engages common control to connect the calling line to an idle attendant trunk on an intercept basis. The "steering" circuit determines when all the digits have been registered; if a valid one-, two-, or three-digit code is received, the register recognizes the code, and engages common control to complete the call.

When digit registration is completed and common control engaged,

the register transmits the called number to common control. Common control then instructs the register to call back the calling line or trunk which is still connected to the register. On intercom calls, the terminating half of the call is established prior to call-back. Call-back identifies the calling circuit and allows common control to set up the remaining portion of the call and to signal the register to release the calling circuit. The register is then available for another call.

In the event that common control encounters a busy connection in attempting to set up a call, it connects the calling party to the busy-tone trunk. If the busy-tone trunk is busy or not provided, the common control reconnects the register to the calling circuit and instructs the register to transmit busy tone. After reconnection to the register, the calling circuit has approximately 10 seconds to abandon the call. If the call is not abandoned in this time interval, the register times out and engages common control to connect the calling line to an idle attendant trunk on an intercept basis.

#### 2.4 Network

Topologically, the 850A PBX network is a three-stage, end-marked, switch for eight wires. The audio and video networks are distinct and separate and are master-slave related. The audio network control tests for idle paths and makes the path selection. When this function is completed identical paths are set up in the audio and video network. The audio network contains the tip and ring, a sleeve lead and a control lead (P-detect); the video network contains the transmit pair and the receive pair for the video portion of the call. Supervision of a *Picturephone* call is always done on the audio network.

There are two network sizes available for the 850A PBX; a 57-line size which has 64-line terminations and 32-trunk terminations, and an 89-line size with 96-line terminations and 48-trunk terminations. The difference of seven between total terminals and usable terminals are those assigned to registers, test line, attendant trunks and the recorded announcement trunk.

The switch apparatus is the 279-type ferreed. This ferreed family is pluggable and constructed on epoxy-clad metal circuit boards in order to meet crosstalk requirements. The ferreeds are available in  $4 \times 4$ ,  $4 \times 8$ , and  $8 \times 8$  sizes which contain four  $4 \times 4$ , two  $4 \times 8$  and one  $8 \times 8$  matrices per ferreed switch respectively. The 850A network uses the  $4 \times 8$  and  $8 \times 8$  switches and for the 89-line size network, a combination of these two is used to form a  $12 \times 8$  switch for the secondary stage. Each crosspoint of the matrices has four dry-reed switch con-

tacts which are closed by pulsing the associated solenoid. The pulse used in the 850A is 8 to 12 amperes and approximately 0.35 ms wide at 25 percent amplitude and 0.45 ms wide at the base. Due to special magnetic properties of the switch, the contacts "remember" their last pulsed position—open or closed. The magnetic structure also insures that any previously operated contacts are released that share either a horizontal or a vertical with the pulsed path.

The ferreed operating characteristics, construction, and their operation in networks have been discussed in detail in many previous publications.

When a path through the network is required, the common control marks the appearances on the line-side and trunk-side of the network. These marks (grounds) operate a unique set of relays that select the associated primary and tertiary switch. The network control also recognizes the request for the network and proceeds to select an idle path. Given a certain line-side appearance and a certain trunk-side appearance, there are four possible paths through the network. When according to a fixed preference an idle path has been tested and the secondary switch selected, the path is established; the common control and trunk-side circuits are signaled that the connection has been made. When the path is pulsed, the appropriate crosspoints are closed and any interfering crosspoints are released; the network operation is then completed. If none of the four paths are idle, the common control is given a no-link-available (NLA) signal.

## 2.5 *Trunk and Attendant Facilities*

Since the trunk circuits of the 850A perform the same functions as in any other PBX system, their operation is discussed in general terms as relating to the 850A PBX. All the trunk circuits use a combination of integrated circuit gates, discrete transistors, and miniature relays.

### 2.5.1 *Intercom Trunks*

Intra-PBX *Picturephone* calls are completed through a two-port intercom trunk which provides all standard audio alerting and supervisory functions as well as a four-wire video connection. A Video Supervisory Signal (VSS) is also applied to activate the calling station's *Picturephone* set and to cause a distinctive alerting signal at the called station.

### 2.5.2 *Central-Office Trunks*

Incoming and outgoing *Picturephone* calls to a *Picturephone*-



equipped central office are completed through the two-way central-office trunk. The trunk is a five-port circuit with one port to the central office, one to the attendant, two to the 850A network, and one to the adjunct telephone system. In addition to providing the required video connections and VSS, the trunk circuit makes a connection to the video continuity test (VCT) equipment at the beginning of all incoming and outgoing calls. Circuit and feature options such as D.I.D. or ground-start type trunk circuits can be provided, Call-Transfer-Individual or Call-Transfer Attendant, and the proper interface circuit with either the 701B step-by-step PBX or 757A crossbar PBX are provided by the proper plug-in boards. The central-office trunk circuit handles all direct-dialed incoming and outgoing central office calls, incoming directory-number calls to the attendant, incoming calls completed by the attendant to either *Picturephone* or telephone stations, and Call-Transfer calls (either direct-dialed or attendant-handled) to either *Picturephone* or telephone stations.

### 2.5.3 Attendant Trunk

The 850A is also equipped with a two-port attendant trunk which is used to complete *Picturephone* station-to-attendant calls or attendant-originated calls. Calls to the attendant may be extended by the attendant to either the *Picturephone* central office or to other *Picturephone* stations, and attendant-originated *Picturephone* calls may be made to either the central office or to *Picturephone* stations. The attendant trunk has no connection to the adjunct telephone system.

In an adjunct installation with either a 701B or 757A telephone PBX, the normal procedure to provide attendant service will be to equip one or two existing telephone attendant consoles for *Picturephone*. Two position applique circuits (one for the 701B and one for the 757A) have been designed to interface the lockout and control signals between the existing attendant position equipment and the 850A trunk circuits. This is necessary since the signals generated by the existing 48-volt electromechanical equipment are not compatible with the 850A trunk circuitry, which consists of 24-volt solid-state logic and miniature relays. In a Stand-Alone system, an 850A position circuit must be added to provide the functions normally provided by the telephone adjunct position equipment.

### 2.5.4 Miscellaneous Trunks

Additional trunks employed by the 850A include the busy tone trunk and the intercept-recording trunk.

## 2.6 Maintenance

Special maintenance features such as, in-service checks, and additional maintenance features beyond those found in similar PBXs have been incorporated in the 850A. Maintenance is performed by the use of two major circuits; the Maintenance Test Circuit and the Fuse, Alarm, and Emergency Transfer Circuit.

### 2.6.1 Maintenance Test Circuit

The functions performed by the Maintenance Test Circuit (MTC) are: (i) monitoring the operational status of the 850A system via status indicating lamps located on the System Status Indicator Panel (SSI); (ii) manual selection of a specific switching path and/or circuit for testing the switching capability of the system; (iii) manual or automatic connection of the selected circuit to the test line; (iv) manual override, reset or reset override maintenance functions for testing the common control circuit; (v) path jack access to all *Picturephone* video equalizers; and (vi) providing maintenance routines such as: (a) inhibit network relay operation, (b) force a register timeout condition, (c) cause transfer of all *Picturephone* stations from *Picturephone* service to audio only service, (d) disable *Picturephone* station line transfer function caused by a major circuit failure, (e) disable all alarm signals generated to local and remote facilities, (f) remove the "loop back" function of calls directed to the attendant to aid in testing and lineup procedures, and (g) reset the video-continuity test (VCT) failure circuit alarms.

In addition to these maintenance functions, the MTC provides a termination for test calls which originate on a No. 23A testboard, No. 15A local test desk, central-office master test frame or local PBX test line.<sup>6,7</sup> The use of this equipment allows the testing of the audio and video connection between the central office and the PBX and within the PBX. Also, lead terminations containing circuit busy/idle or circuit seizure information are grouped together on connectors for use with external traffic measuring equipment.

### 2.6.2 System Status Indicator

In general the circuits, keys, etc., on the MTC are not unique and their operation is self-explanatory by their function. However, the System Status Indicator Panel (SSI) is normally not used in PBXs. The SSI consists of switches and indicator lamps connected to the common control, register, and switching network via Darlington-type transistor amplifiers. Positive signals received from the monitored

circuits reflect the dynamic operation of the circuit as it performs its switching function. Therefore, during normal operation of the system circuit switching operations will produce short duration lamp indications (flashes), indicating which circuit or path or which function was performed during this switching operation. In the event of circuit failure, one or more of these indicators will remain lighted, indicating in which mode the system is locked. Lamp indications are provided for each of the common control modes discussed earlier; additional indicators are provided for register-busy for each of the maximum of three registers, read-out indication of the registers, and a *Picturephone* call "detect" of the registers which indicates that an # signal has been detected by the respective register and that it has initiated a release signal to disconnect the audio PBX from the originating line circuit.

### 2.6.3 Fuse, Alarm and Emergency Transfer

The function of the Fuse, Alarm and Emergency Transfer Circuit (FAET) is the distribution of dc power, ring and tone, and interrupted signals to all 850A circuits associated with the audio PBX system. This circuit also provides the necessary alarm indications and station line transfer functions in the event of power, fuse, or circuit failure within the 850A PBX System.

The dc power for the switching circuits (+24, -24, and -48 volts) is distributed through alarm indication fuses of the Power Distribution and Fuse Circuit. The dc power (-24 V) for the video equalizers is distributed through alarm indication fuses of the equalizer Fuse Alarm Circuit located on the equalizer bays.

PBX *Picturephone* circuit failures are classified as causing major or minor service interruptions. Major circuit failures are those that result in the complete loss of the switching capability of the 850A. Such interruptions, caused by certain fuse failures, loss of dc voltage from the associated power supplies, or failure of the common control will activate related alarms in the 850A and associated audio PBX, and terminate *Picturephone* service by causing all *Picturephone* stations to be transferred to audio telephone service. Minor circuit failures affecting individual *Picturephone* circuits but not affecting the switching capability of the 850A will activate related alarms in the 850A and the associated PBX.

All alarms associated with the video transmission are also indicated on the MTC. In addition, a distinctive *Picturephone* alarm indication (red lamp) is presented at the associated telephone console. This dis-

tinctive *Picturephone* alarm is also transmitted to the central office via alarm extension leads, when provided, and is interrupted at a 60 IPM rate.

### 2.7 *Miscellaneous Circuits*

In addition to the major circuits of the 850A PBX discussed above, there are some smaller, but important, circuits necessary for the proper operation of *Picturephone* service by the 850A.

#### 2.7.1 *Video-Continuity Test (VCT) Circuit*

When a central office trunk is connected to the calling station, a video-continuity test (*VCT*) is made from the trunk circuit through the network to the calling line to insure the proper transmission quality of the video quad. The *VCT* is made only on connections where charging will occur; i.e., intercom calls do not have a *VCT*. When the *VCT* verifies the transmission quality of the loop, the central-office trunk circuit then cuts the station through to the central office. Should the loop fail the *VCT*, an alarm is registered, the connection between the line and trunk released, and the line is marked again. The common control then marks an intercept-recorded announcement trunk circuit on the trunk side of the network. The calling line is then connected to this trunk which returns a recorded message advising the calling party that the call cannot be completed on a video basis.

The *VCT* circuit contains a 12 kHz signal generator and window detector; the generator delivers the precisely controlled level signal to the central office and attendant trunk circuits. The window detector receives the looped-back signal from the trunk circuit; if the received signal lies within the range of +1.8 to -3.0 dB of the transmitted signal, the *VCT* circuit signals "pass." If the received signal is outside this range, then a fail signal is generated causing station reroute to the recorded announcement. An exception to this is in a station-to-attendant call, in this case the call is allowed to complete but the *VCT* fail signal inhibits the attendant from extending the call to the central office.

#### 2.7.2 *Video Supervisory Signal (VSS) Circuit*

Following a successful *VCT*, a video supervisory signal (*VSS*) is applied 100 ms before ringing at the trunk on the video quad. The *VSS* is required to turn on the *Picturephone* display unit, thus prohibiting the use of the set without establishing a valid *Picturephone* call. *VSS*

and the ringing signal are joined to form a composite signal which activates a tone ringer producing a distinctive tone indicating the *Picturephone* status of the incoming call.

One VSS generator circuit is used in the 850A and it is the same as incorporated in the *Picturephone* set.<sup>4</sup> The VSS signal is supplied to the set by the trunk circuits through a distribution amplifier. This VSS circuit is also alarmed in case of failure.

### 2.7.3 Video-Image-Generator

The Video-Image-Generator (VIG) equipment is provided as an option on the 850A PBX; the function of this equipment is to transmit a fixed image to a caller while the attendant processes his call. An incoming central-office call picked up by the attendant will normally transmit the attendant's image if the attendant is equipped with a *Picturephone* set; if the attendant is not equipped, then the caller will have raster but a blank screen. If the 850A customer elects to have a VIG, then when the attendant places the caller on "hold" a fixed image supplied by the customer is transmitted to the calling party until he is connected to his requested party.

The VIG is a combination of a *Picturephone* display unit and a 35-mm projector packaged as a single unit. A regular 35-mm slide can be inserted into the projector to provide the transmitted image. This equipment can be located remotely from the 850A but within the confines of *Picturephone* set transmission limitations.

## III. APPARATUS AND EQUIPMENT

### 3.1 Logic

The basic logic element used in the 850A is the 30A RTL, quad gate, 24-volt, hybrid-integrated circuit (HIC), presently being manufactured by the Western Electric Company.

In the early design stages of the 800A PBX many types of logic were investigated. Because of the relatively low line size (80 lines), the 800A did not require the switching speed of devices used in large electronic systems. In addition, the 800A design called for the use of miniature flat spring relays which operate at a nominal 24 volts. Therefore, a family of RTL logic gates using a 24-volt collector supply was developed. The use of such a logic gate provided for a good marriage between the solid-state circuits and relays, and allowed the use of only one supply voltage.

The family of RTL gates developed were of five basic types, coded

17A, 18A, 19A, 20A, and 21A with additional subgroups B and C on the 17 and 18 types. These different gate codes identified the number of resistor inputs. Each logic gate was constructed of tantalum deposited resistors and a discrete-type 16J transistor mounted on a ceramic chip.

With the advent of integrated circuit technology in Western Electric Company production, a hybrid-integrated circuit (HIC) gate was developed to replace the 17A-21A gates. This new quad gate, type 30A, is a 24-volt HIC and, as shown in Fig. 3, consists of two two-input, one four-input, and one five-input RTL gates. A 30A circuit package is formed by bonding a quad transistor chip to a  $\frac{1}{2}$  by 1 inch ceramic substrate which contains 17 deposited tantalum resistors; two views of the package are shown in Fig. 4. The four base leads are brought out to external terminals so that additional inputs may be added to any gate by means of discrete resistors.

### 3.2 Circuit Packs

The 30A Gates, miniature relays and other miscellaneous discrete components are mounted on 8 by 8 inch epoxy-coated metal printed wire boards which, in turn, plug into 23-inch board carriers. These boards, as shown in Fig. 4, are the same size as used throughout the 800A PBX. The use of large boards reduces system wiring, provides functional modularity, flexibility, ease of growth, and easy replacement for maintenance. "Plated-through" holes, which are easily obtained because of a characteristic of the epoxy, are utilized to provide printed-circuit paths on both sides of the board. The circuit devices mounted on the board are protected by a ventilated plastic cover on the board. Each cover includes a plastic ejection lever molded as part of the cover for easy removal of the circuit pack from its carrier.

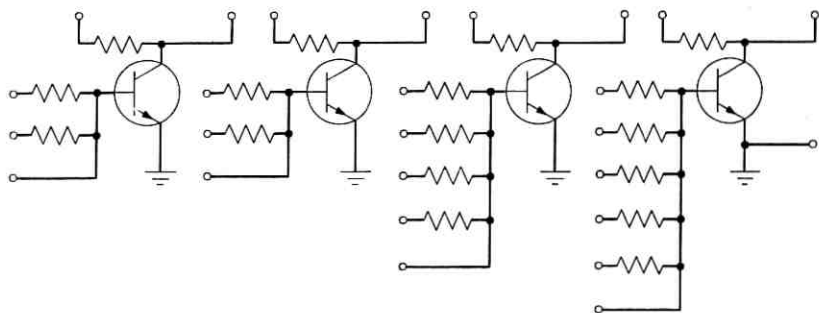


Fig. 3—Schematic representation of the 30 HIC gate.

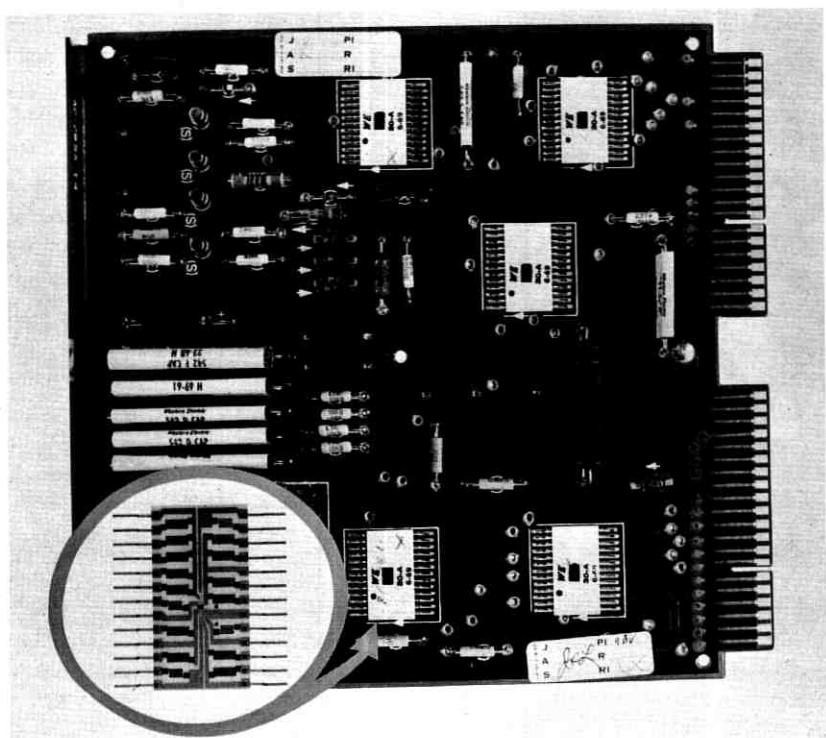


Fig. 4—Typical 850A circuit pack with HIC devices.

The circuit pack carriers (Fig. 5) are self-supporting and bolted to the frames. These carriers are the same as those used in the 800A but are cut to a 23-inch size for frame mounting. The carriers are aluminum die cast with identical top and bottom plates with molded-in slots for the circuit packs, ventilation holes and mounting bosses for the snap-in plastic 100-pin connectors. Each carrier has a circuit pack designation strip that also serves as a gravity-operated locking bar to prevent circuit-packs from vibrating out of their connectors.

### 3.3 Network

The ferreed bottles are also mounted on epoxy-clad plug-in printed wire boards for use in the 850A PBX. The use of the metal board in this application provides capacitive shielding (by grounding the board) which gives the 850A switching network excellent crosstalk protection. The orientation of the printed-circuit paths were subjected

to many tests and then optimized to achieve good characteristics for such transmission characteristics as crosstalk and impulse noise for the one-MHz video transmission bandwidth.<sup>2</sup>

The use of pluggable ferreeds allows the selection of the network of a particular line and trunk size at initial installation but growable at some future date by the addition of ferreed packs. Since, as also stated above, two  $4 \times 8$  matrices are incorporated in one ferreed switch pack, line and trunk terminals are growable in groups of 16 terminals. The secondary stage of the 3-stage network is equipped for either the maximum 57-line or 89-line size network. The 57-line size network can contain a maximum of 16 (8 audio, 8 video) ferreed switches; the 89-line size can be equipped with 32 ferreed switches.

A "patch-field" is associated with the switching network to add further system flexibility. This feature permits adjusting the network traffic distribution in the field by rearranging the plugs connecting each line and trunk to the network on the patch-field.

### 3.4 Equipment Arrangement

A complete 57-line system, including power, is rack-mounted on five standard 701-type 23-inch, 7-foot frames as shown in Fig. 6. The carriers are equipped with the number of circuit packs required for a particular customer order. An 89-line system requires six frames. In order to grow from a 57-line system to an 89-line system, the bay con-

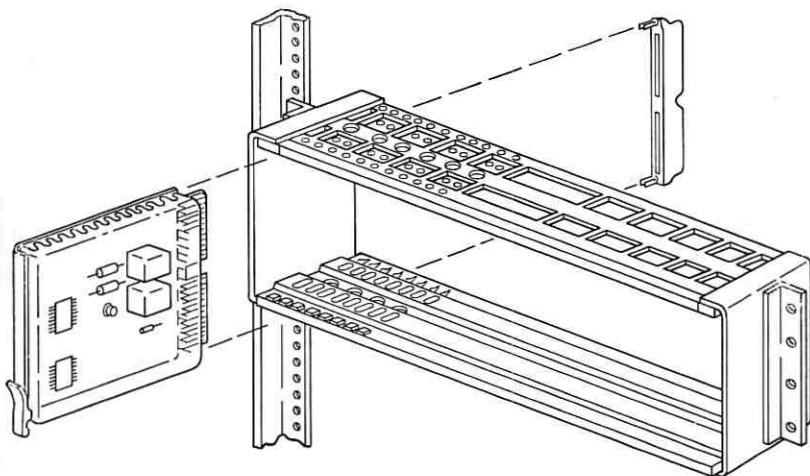


Fig. 5—Die-cast carrier for the circuit packs.



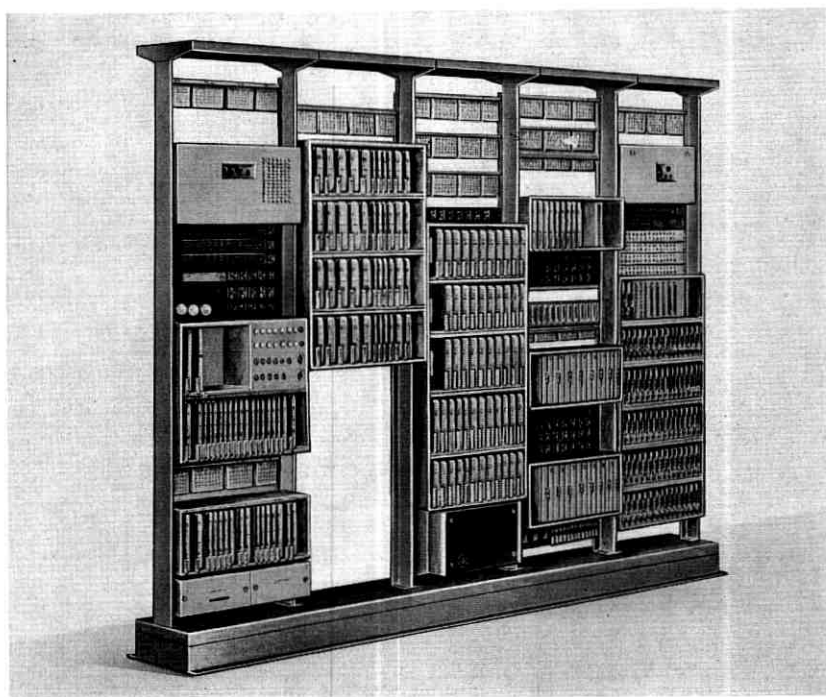


Fig. 6—Equipment arrangement of the 850A PBX.

taining the switching network and video switches is substituted with a bay containing an 89-line network and a bay added to the line up containing the additional video switches required for this line size.

Referring to Fig. 6 and going from left to right, the first bay contains the +24 volt and -48 volt power supply, the SSI, MTC panel, common control and registers. Bay B, the next bay in line, contains the attendant and central-office trunk circuits. Bay C contains the line circuits and intercom trunks. The switching network, VSS, and VCT circuits, and video switches are on Bay D, and Bay E contains the -24 volt supply, fuse and jack panel, and the video-equalizers.

The bays are interconnected by plug-ended cables which allow the test of the complete system at the manufacturing site and fast installation at the customer's location. Connections to the 701- or 757-type PBXs are made to the appropriate wire-wrap terminals located at the top of the bays.

## IV. ACKNOWLEDGMENTS

The author wishes to acknowledge the 850A circuit design contributions of M. A. Hoffman, G. A. Lesser, C. C. Nielson, A. P. Ryan, III, L. E. Saltzman, and G. A. Scharg and the assistance of F. Lukas.

## REFERENCES

1. Breen, C. "The *Picturephone*® System: Customer Switching Systems," B.S.T.J., this issue, pp. 553-565.
2. Brown, H. E., "The *Picturephone*® System: Transmission Plan," B.S.T.J., this issue, pp. 351-394.
3. Baker, D., et al., "Electronic Switching for Small PBXs," Bell Laboratories Record, 45, No. 2 (February 1967), pp. 49-56.
4. Cagle, W. B., Stokes, R. R., and Wright, B. A., "The *Picturephone*® System: 2C Video Telephone Station Set," B.S.T.J., this issue, pp. 271-312.
5. Brown, J. M., "The *Picturephone*® System: Baseband Video Transmission on Loops and Short-Haul Trunks," B.S.T.J., this issue, pp. 395-425.
6. Favin, D. L., and Gilmore, J. F., "The *Picturephone*® System: Line and Trunk Maintenance Arrangements," B.S.T.J., this issue, pp. 645-665.
7. Dougherty, H. J., Peterson, E. B., and Schachtman, M. G., "The *Picturephone*® System: Maintenance Plan," B.S.T.J., this issue, pp. 621-644.

# The *Picturephone*<sup>®</sup> System:

## No. 101 ESS

By D. W. BROWN, J. R. HORVATH, and T. S. PAXTON

(Manuscript received October 2, 1970)

*This paper describes the methods and equipment used in providing Picturephone<sup>®</sup> capability to the No. 101 Electronic Switching System. Included are a discussion of the objectives, design approach, transmission requirements, traffic capacity, and operation of the wideband switch unit (WSU) which provides the video transmission path. Also described are the video trunk circuits, junctor circuits, and the physical characteristics of the WSU.*

### I. INTRODUCTION

#### 1.1 *General*<sup>1-3</sup>

The No. 101 Electronic Switching System (ESS), an electronic private branch exchange (PBX), has been providing customers with Centrex and other specialized services since the first commercial installation was made in Cocoa Beach, Florida, in November 1963.

The system, consisting basically of a control unit (CU) and one or more switch units (SU), effects the actual interconnections between lines, trunks, and service circuits on a time division switched basis as described in the February 1969 issue of the B.S.T.J.<sup>4</sup> The SUs are connected by means of data links to the CU a common, centrally located, special purpose computer which processes calls in the system by means of a stored program technique.<sup>5,6</sup> Although practical for the handling of voiceband frequencies, the time division circuitry in the No. 101 ESS switch units is not capable of handling the higher frequencies required for transmission of video signals. In order to provide video switching capability, a wideband switch unit (WSU) utilizing space division switching has been designed for use in parallel with the 2A, 3A, and 4A switch units.<sup>7-9</sup>

#### 1.2 *Objectives*

The design objectives for the WSU were based upon the expected low demand for *Picturephone* service during the initial years. It was

decided that the WSU must be economical at the 20- to 30-line size and that additional line capacity must be easily administered in the field. The WSU should also be compatible with all No. 101 ESS SUs (2A, 3A, and 4A) and should make as much use as possible of existing SU circuitry.

## II. SYSTEM DESCRIPTION

To handle audio telephone traffic, a SU is located on the customer's premises, as shown on Fig. 1, and is controlled by a CU located at the central office. Communications between the CU and SU is provided by means of data links. To add *Picturephone* capability to this complex, an adjunct WSU consisting of a four-wire space division network and a duplicated wideband switch control (WSC), is also located on the customer's premises and is connected to the SU via connectorized cables. The two video pairs from each *Picturephone* extension and trunk terminate on the WSU, while the audio pair terminates on the SU. A complete *Picturephone* connection is composed of an audio connection in the SU and a parallel video connection in the WSU. All translation needed to associate the video pairs with an audio pair is part of the stored program information in the CU, which eliminates the need for directory number changes when equipping an audio line with *Picturephone* service. The WSU adjunct scheme utilizes existing data links, supervisory and signaling circuits, service circuits,

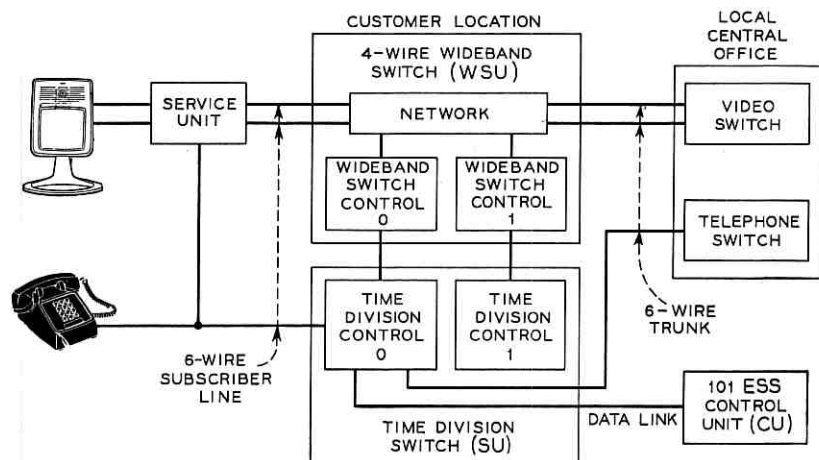


Fig. 1—System plan.

and incoming and outgoing message circuits contained in the SU, as well as all switching functions associated with the audio pair. Messages from the CU to the WSU are transmitted in serial form over an existing data link to the SU and are treated as normal audio messages by the SU until detection of the video address occurs. The message is then converted to a parallel form and gated over a DC bus to the WSU. The circuitry required to detect the video address and gate the video message to the WSU is located in time division controls (TDC) 0 and 1 of the SU. Each wideband switch control in the WSU is directly associated with its respective TDC in the SU. In order for a given WSC to process a video message, the message must be received by the TDC associated with that WSC. When a TDC is disabled by maintenance, the corresponding WSC loses its ability to process video messages.

The WSU communicates with the No. 101 ESS control unit via seven supervisory scan points that have been added to the SU scanner. A change of state on a scan point transmits one bit of maintenance information to the CU. All necessary maintenance information regarding the status of both wideband switch controls and the per call transmission checks made on the video pairs is sent to the CU through these seven scan points.

### III. NETWORK

#### 3.1 *Transmission Path*

The network, which is shown in Fig. 2 at its maximum size of 192 terminals, is basically a folded three-stage network utilizing standard 8 by 8, four-wire ferreed switches.<sup>10</sup> The network is unduplicated and may be controlled by either WSC.

In order to meet the objectives of being economical at the small line size (20 to 30 lines) and of enabling the network to grow efficiently, the network was modularized. Modules 0A and 0B are identical, each containing four first-stage switches, 16 junctor circuits, and two second-stage switches. Modules 1 and 2 are identical and contain eight first-stage switches each. All interconnections between modules as well as to the wideband switch control are connectorized. The network grows on an add-on basis and allows the addition of modules without disturbing existing customers.

With the provision of two or more modules, the network consists of eight concentrators and four second-stage switches. A concentrator contains a maximum of three first-stage switches with switches 00,

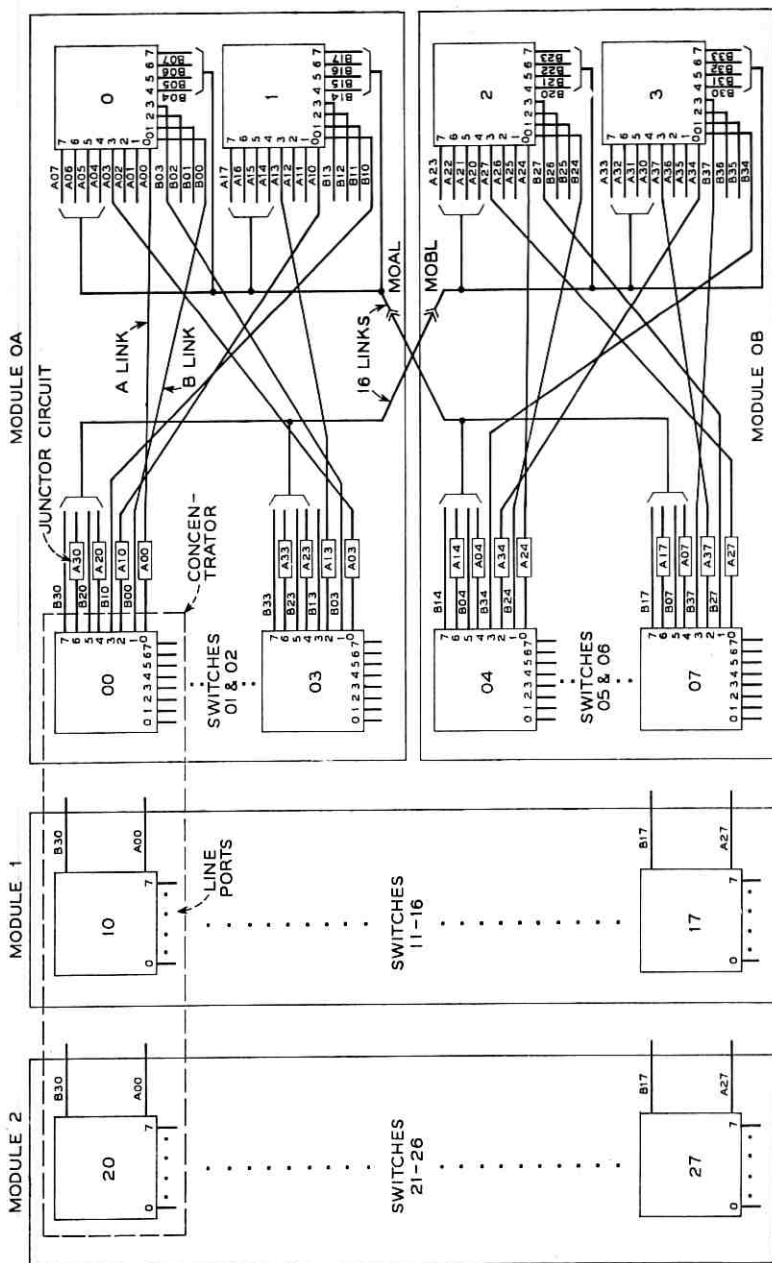


Fig. 2—Fully equipped network.

10, and 20 comprising concentrator 0, etc. It may be seen by looking at concentrator 0 that 24 input terminals have access to eight links resulting in a concentration ratio of 3:1.

At the one module size with only module 0A provided, half of the links (those normally terminated on module 0B) appear to be unterminated. However, the hardware is so arranged that these links may be plugged into the position on module 0A that is normally occupied by links from module 0B. In this way, a full-access network is achieved at the 32 terminal size with no hardware or software modifications. When module 0B is added, the network is full access with a terminal capacity of 64. The addition of module 1 increases the terminal capacity to 128 and the concentration ratio to 2:1. The addition of module 2 increases the network to its maximum of 192 terminals, and the concentration ratio of input terminals to links to 3:1.

Two separate sets of links, termed A and B links, are used to interconnect the first- and second-stage switches. Every connection through the network involves one A link and one B link. The B links provide a straight-through connection between the first- and second-stage ferreed switches. The A links, however, transpose the video pairs via a junctor circuit shown in Fig. 3. The junctor also allows the wideband switch control to access the video pairs for transmission testing, to supply a video supervisory signal to the customer station equipment, and to cut through the video connection after the network path has been established. This latter function assures that the ferreed cross-points are closed and opened with no current flowing in the network path. With the links interconnecting the first- and second-stage

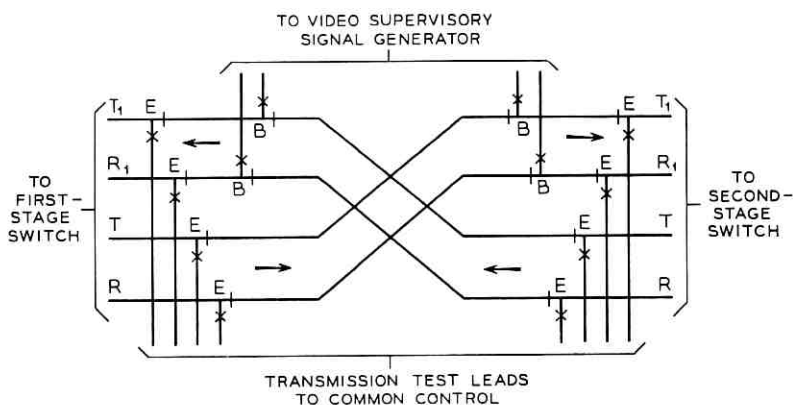


Fig. 3—Junctor circuit.

switches as shown in Fig. 2, there are eight distinct paths between parties located on separate concentrators and four paths between parties involved on an intraconcentrator call.

Lines and trunks are distributed over the switches (as opposed to filling the switches one at a time) in order to balance the traffic load over the links. The first 32 ports of the network, 16 on mod 0A and 16 on mod 0B, are equipped for use as trunk and attendant ports as well as lines. These ports are provided with a loopback circuit and the means of applying the signal for a stationary image to the video transmit pair. The loopback circuit is needed at the WSU on trunk and attendant ports to substitute for the loopback normally provided at the subscriber's station set on line ports. The stationary image is transmitted at appropriate intervals to the distant party on attendant handled calls during the time the call is being processed. Four of the 32 ports are dedicated for attendant use. Since the attendant has the ability to hold up to six calls on the console loops, she must have the capacity to access each party being held. In order to reduce the blocking probability on attendant-handled calls, each console is connected to two ports located on separate concentrators.

### 3.2 *Pulse Path*

Both the first- and second-stage portions of the network utilize standard 8 by 8, four-wire ferreed switches. The control windings within the switch are series connected along the rows and columns. Eight horizontal leads, eight vertical leads, a common horizontal and a common vertical lead are brought out from each switch (see Fig. 4). By pulsing along a given horizontal and a given vertical via the common horizontal and vertical lead, the selected four-wire crosspoint is closed. A half-select, along either a vertical or horizontal, causes the crosspoint to open.

The horizontal and vertical leads are multiplied among the switches associated with a module, so that from any given module only eight vertical leads, eight horizontal leads, one common vertical lead, and one common horizontal lead emerge. In all cases the pulse path exactly parallels the transmission path.

The establishment of a path through the network may be illustrated using the message contents shown in Fig. 5. The network path is selected by a set of wire spring relay trees that operate directly from the party addresses and the second-stage switch information in the message. The party addresses are composed of three elements which specify the module, switch level, and network concentrator of



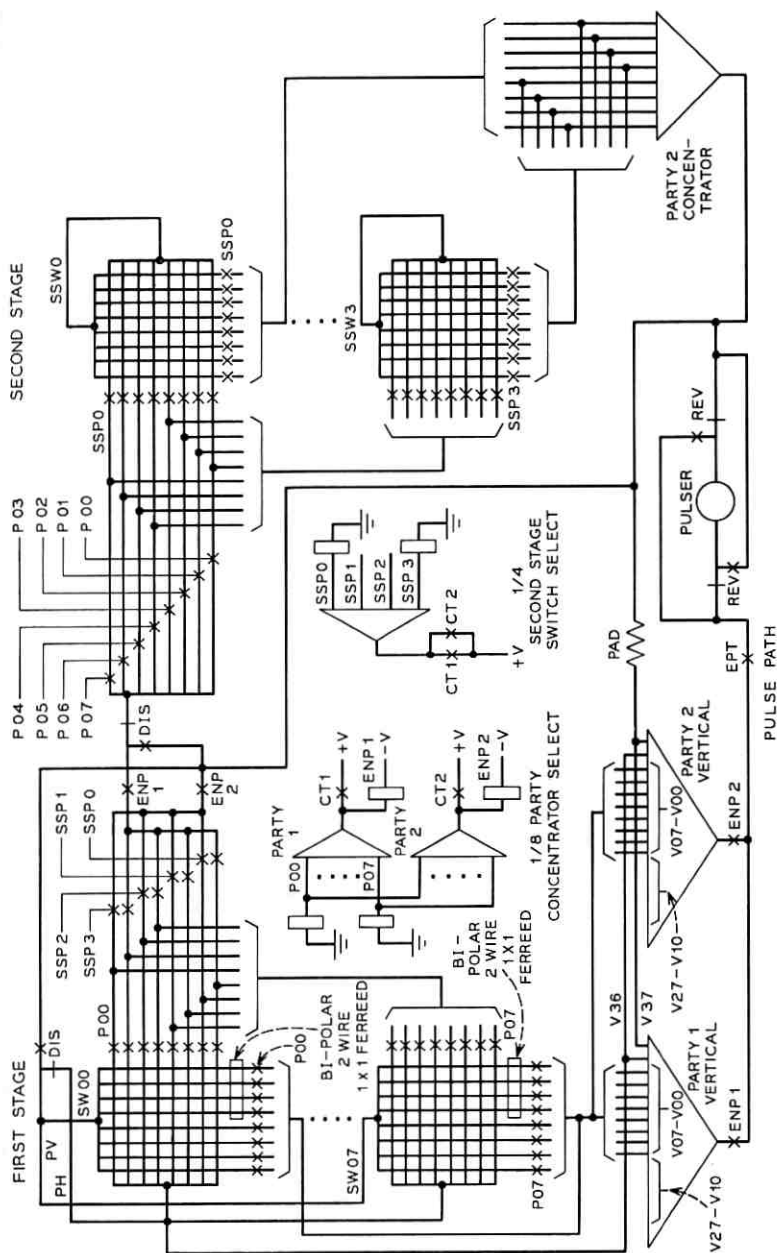


Fig. 4—Pulse path.

the party port. The A and B links to be used in the connection are determined by the second-stage switch bits and the respective concentrator numbers for party 1 and party 2.

The party vertical translators operate from the module and switch level bits to select 1 out of 24 switch vertical leads, and the party concentrator translators operate from the party concentrator bits to select one out of eight network concentrators (Fig. 4). The output of the concentrator translators operates a per switch path select relay for each ferreed switch on the selected concentrator. In a similar manner a path select relay associated with a second-stage switch is operated through the second-stage switch translator.

All translators for both parties operate immediately upon receipt of the message. After the translators have operated, the pulse path is closed across the pulser by relays ENP1 and EPT to fire the pulser. Since a connect order is being performed, the disconnect relay (DIS) will remain unoperated throughout the connect procedure. The first pulser firing simultaneously closes the first-stage switch crosspoint to connect party 1 to the A link, and the second-stage switch crosspoint to connect the A link to the B link. The pulse path runs through the party 1 vertical translator, up the selected vertical lead to the ferreed switch selected by the concentrator translator, around the common vertical (PV) and horizontal (PH) leads, and out on a horizontal level selected by the operated second stage switch path select relay. The pulse then passes through the selected second-stage switch and returns to the pulser. The horizontal and vertical levels on the second-stage switch are respectively selected by the first-stage path select relay and a separate party 2 concentrator translator.

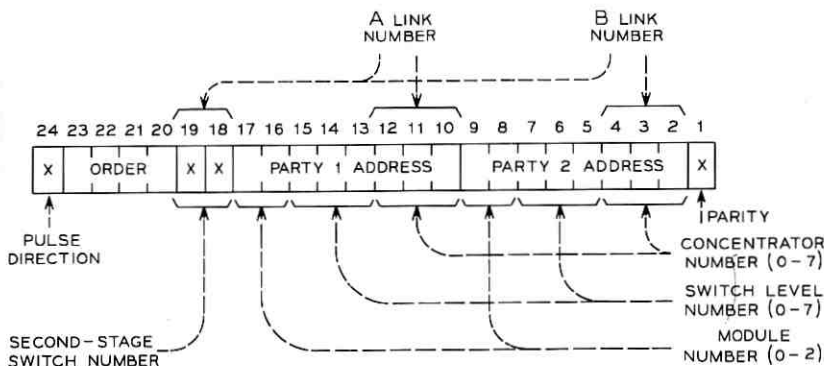


Fig. 5—Message contents.

The second pulser firing connects party 2 to the B link and occurs after relay ENP1 is released and ENP2 is operated. The pulse path is similar to that of the party 1 connect but does not include the second-stage switch.

The REV relay is operated from the pulse polarity bit in the message and controls the direction of the pulser firing. The direction of the pulse is used on trunk and attendant ports to operate or release the bipolar ferreeds for application of the metallic loopback or stationary video-image signal on these ports.

On the termination of each talking connection the link crosspoints are opened to prepare the link for the transmission tests which will be conducted on the next connection. The order is processed in two stages which resemble a connect order except in the following respects. The DIS relay is operated causing a half-select along the verticals of the first-stage switches thereby releasing party 1 and party 2 from their associated links. The second-stage switch crosspoint remains operated, but will be released by the next connection made to either the horizontal or vertical lead associated with it.

For maintenance and initialization purposes, it is desirable to modify the disconnect order so that any connection on a link is released without specifying a port address. For this purpose a disconnect message is formed in which an octal 36 is substituted for the module and level bits in either or both party addresses. This substitution causes the disconnect pulse to be transmitted along the link specified by the second-stage switch and concentrator numbers in the party address.

It is also essential in certain call processing situations to execute an order that affects only one port appearance. An octal 37 substituted for the module and level bits in the party address will cause the pulser to fire into a resistive load leaving the first-stage crosspoints associated with that party unchanged.

### 3.3 *Transmission Tests*

Since the transmission requirements of the video facilities are stringent and charging takes place upon completion of the audio path, it is imperative that the integrity of the video loop be checked before billing takes place. This is accomplished (Fig. 6) in the WSU on a per call basis at the time the initial connection between two parties is being established. On subsequent connections, during call transfers for instance, the transmission checks are omitted from the connect orders. As explained previously, the establishment of a path through the network occurs in two stages. On each stage of an initial connec-

tion a false ground test is made from the junctor circuit on each lead of the video pairs looking toward the party being connected to the link. This test is made in order to detect the presence of stubs attached to the link through stuck network crosspoints. After all network crosspoints have been closed, a test is made on the total video path to check continuity through the network crosspoints and over the video loop to the station apparatus.

The false ground test relies on a  $50\Omega$  ground that is placed on each lead of the video pairs by either the station set equalizer or the equalizer located at the WSU. This ground should not be present on any lead of a link within an idle junctor circuit since each party is disconnected from its link when a talking connection is terminated. Therefore, a stuck crosspoint may be detected by looking for grounds on the video pairs in the junctor circuit before the network crosspoints are closed. All transmission tests employ a 12-kHz tone source and a 12-kHz tuned detector located in the WSC of the WSU. On the first stage of the connection, before party 1 is connected to the A link, the E relay of the junctor circuit is operated to bring the video pairs from both parties into the WSC. The FCG1

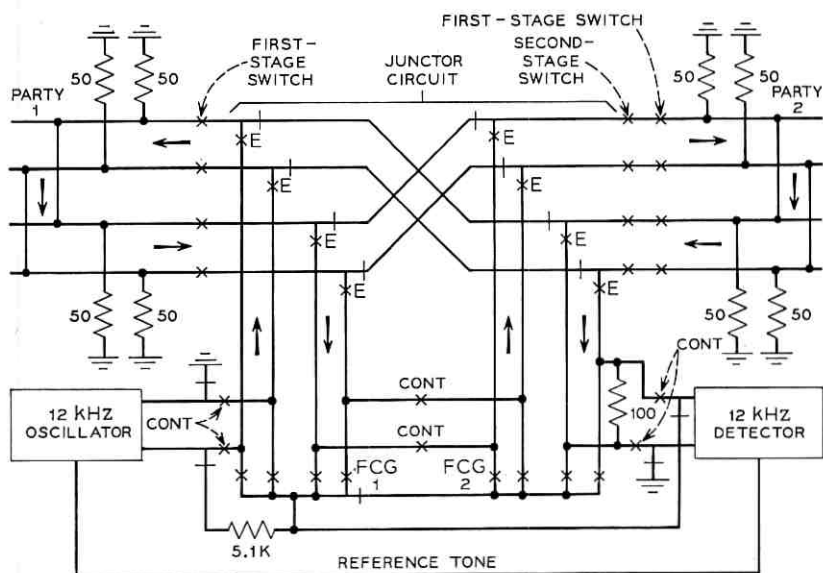


Fig. 6—Transmission tests.

relay is then operated to interconnect the leads of the video pairs for party 1 and to connect them to the 12-kHz tone source. The 12-kHz signal on the video pairs is connected to a tuned detector together with a reference tone from the 12-kHz source. The detector compares the tone level on the video pairs with the reference tone and sends a pass or fail signal to the WSC. If an impedance ground of 3K ohms or less exists on any lead of the party 1 pairs, the signal level will be low enough to cause the tuned detector to transmit a failure signal to the WSC. If no stub is detected on the A link, party 1 is connected to this link and the second-stage switch crosspoint is closed thereby connecting the A link to the B link being used for party 2. On the second half of the connection procedure a similar false ground test is made through the junctor circuit looking toward party 2 through the closed second-stage switch crosspoints. If no stubs are found on the B link, the connection is completed by connecting party 2 to the link.

The continuity test is performed after the network path has been established and utilizes the same 12-kHz oscillator and detector used in the false ground tests. To start the test the CONT relay is operated in the WSC to transmit the 12-kHz tone through the junctor circuit, through the loopbacks at the party 1 and party 2 station sets, and back through the junctor circuit to the tuned detector. If the received tone at the detector is 4 dB less than the reference tone, a failure signal is given to the WSC. A failure on any one of the transmission tests prevents the WSC from completing its sequence of operations for the order and causes a FAIL scan point to be sent to the No. 101 ESS control unit. Upon receipt of the FAIL scan point, the CU will clear the WSU of any effects of the failed order and will attempt to remove possible stubs on the transmission path. The order will then be resent. If the failure persists, the CU will remove suspected faulty equipment from service and cause reorder signal to be returned to the calling party.

#### IV. TRAFFIC HANDLING CAPABILITY

Traffic calculations at three stages of growth were made assuming a blocking probability of  $P(0.01)$  (see Fig. 7). These calculations were made at the equipment break points and assume that all terminals are equipped at a given network size and generate equal traffic. It may be observed that terminal capacity can be traded for increased traffic handling capability in a very high traffic situation.

## V. WIDEBAND SWITCH CONTROL

Each of the duplicated WSCs is fully equipped to handle the maximum network capacity of 192 terminals and is intimately associated with a TDC in the SU (see Fig. 8). The WSC to be used in processing a message is selected by sending the message to its associated TDC. The WSC is an asynchronous circuit which performs the timing, sequencing, and transmission tests necessary for the operation of the switch network. The order repertoire was designed to complement the operation of the SU as closely as possible so that a minimum number of WSU orders are required to complete a connection. This is necessary to prevent excessive loading of the SU data link and to minimize the time required to complete a video connection.

Receipt of a start bit from the SU initiates a sequence of events which leads to the establishment of a four-wire transmission path through the network and leaves the A link junctor circuit in the proper state. The start bit causes the relay tree translators to operate directly from the party and second-stage switch bits in the message. The order is decoded by solid-state translators and, after the relays have settled down, a parity check is made on the 24-bit message. The WSC then operates the E relay in the proper junctor circuit to gain access to the transmission leads for the transmission tests and to open the trans-

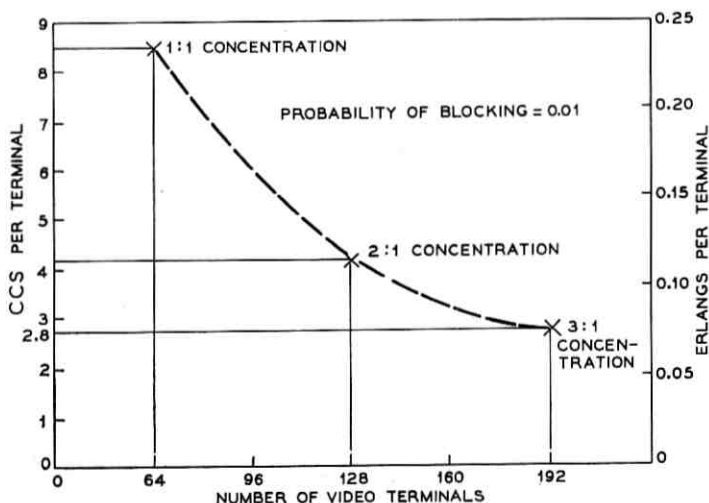


Fig. 7—Traffic estimates. (Notes: 1. X = calculated points. 2. Traffic calculations assume all terminals equipped at a given network size. 3. All terminals generate equal traffic. 4.  $P(0.01)$ ).

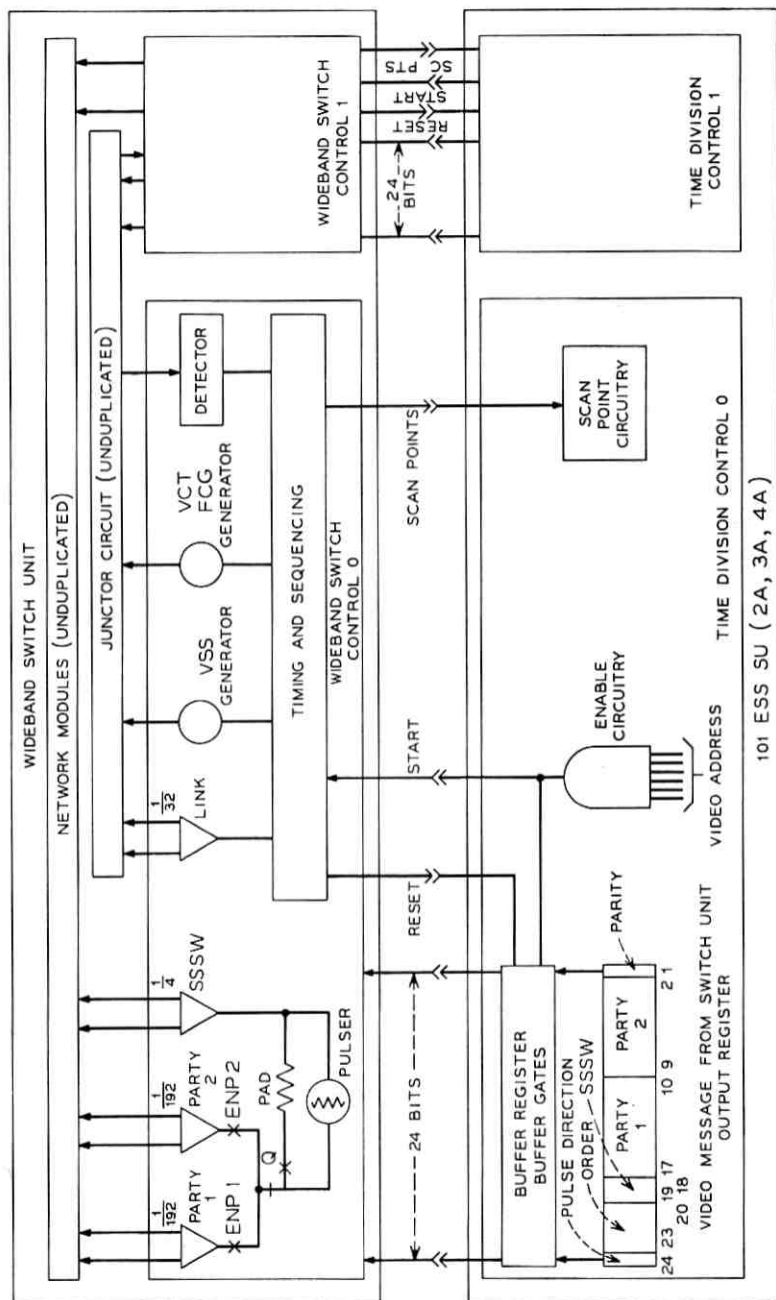


Fig. 8—Common control.

mission leads during the pulser firing. The network path is completed with the required transmission tests as previously described. During the period in which the WSC is setting up the network connection, the junctor circuit is placed in its proper state. Upon completion of the network connection, the junctor relays are checked to verify their operation. If all indications are proper, the buffer register in the SU is reset and an OK scan point is returned to the CU. If the operate time of an order is greater than 85 ms or if any WSC test has failed during processing of the message, a FAIL scan point is returned to the CU.

A faulty WSC may be removed from service or quarantined by the maintenance program. In the quarantined state, the pulser will be made to fire into a dummy load instead of through the network, and the junctor translator will translate to a dummy junctor circuit. If a continuity test or a false cross test is required, the oscillator and detector are interconnected in a manner to allow the test to pass. All other functions of the WSC are performed normally. Responses from the WSC, OK, or FAIL scan points are still received by the CU so that a quarantined WSC may be fully tested without endangering network operation.

## VI. PHYSICAL DESCRIPTION

Figure 9 is an artist's rendition of a fully equipped wideband switch unit plus the equalizers necessary to give *Picturephone* capability to the No. 101 ESS. The equipment shown is housed in existing No. 101 ESS 3A switch unit frames measuring 90 cm wide by 213 cm high by 50.8 cm deep.

The wideband switching frame is used to mount the equipment modules with no electrical connections actually made to the frame. The solid-state circuitry used in the wideband switch controls is mounted on circuit packs 14.2 cm high by 28 cm deep. Nine such packs are needed per wideband switch control.

At the 32-line size the high current pulser unit, wideband switch control unit, and one 32-port network module (0A) are supplied. If it is necessary to grow beyond this terminal capacity, additional network modules are added by mounting them on the frame and making the appropriate connections via connectorized cables.

When the WSU is used in a *Picturephone* application, a cable equalizer is needed for each line or trunk terminal used. For transmission reasons, the equalizers must be physically located near the WSU. The basic equalizer bay provides equalization for the first 64 video ports.



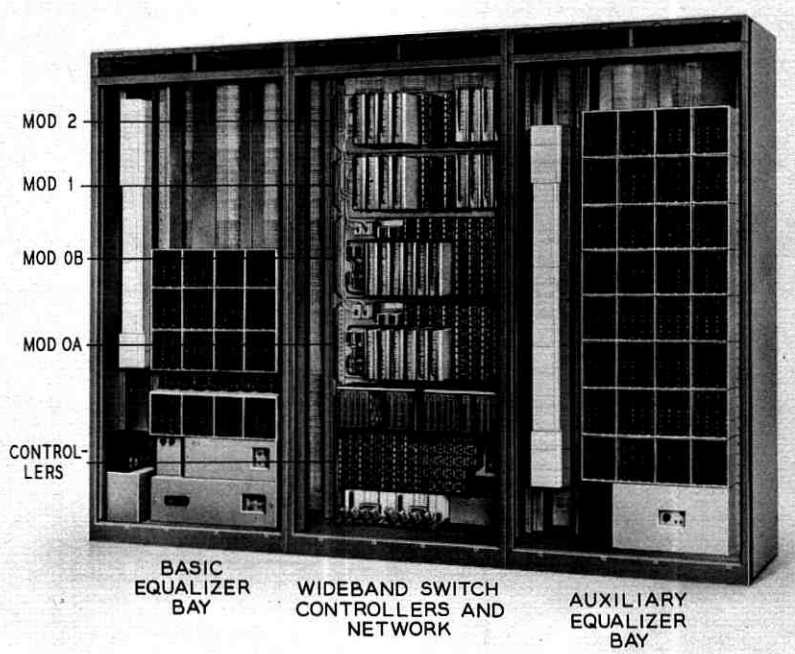


Fig. 9—Wideband switch unit and equalizer bays.

In addition, it houses the power supplies needed by the equalizers and wideband switching frame.

The auxiliary frame is added to the lineup if a terminal capacity greater than 64 is desired. The additional equalizers and their associated power supply are housed in this frame.

## VII. SUMMARY

In order to handle the higher frequencies required to transmit the video signals required for *Picturephone* service, a WSU was developed for No. 101 ESS. System operation of the No. 101 ESS with the WSU is given and the processing of a typical video message is described, indicating the manner in which a network path is established for the video portion of a call.

Laboratory testing of the hardware and software has thus far demonstrated the validity of the overall objectives and philosophy used in the design and operation of the WSU.

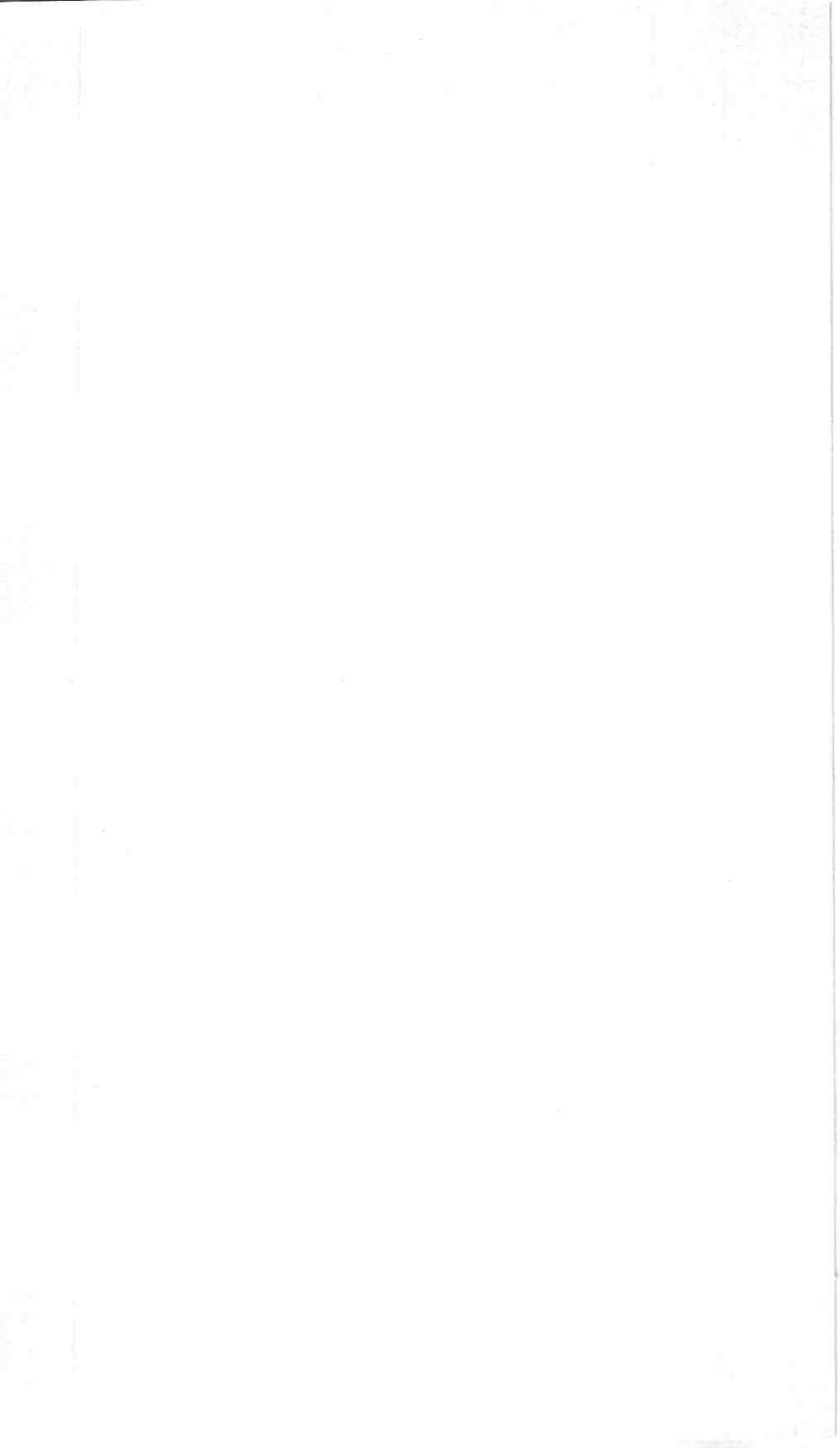
## VIII. ACKNOWLEDGMENTS

A number of persons have been involved in the design of the No. 101 ESS wideband switch unit. The authors wish to acknowledge in particular R. M. Averill and M. L. Larson for their contributions to the WSU circuit design and R. V. Gillingham and R. D. Wiese for the WSU physical design.

## REFERENCES

1. Depp, W. A., and Townsend, M. A., "An Electronic Private Branch Exchange Telephone Switching System," IEEE Winter General Meeting, New York, February 1, 1963, Paper No. CP63-1043.
2. Higgins, W. H. C., "A Survey of Bell System Progress in Electronic Switching," B.S.T.J., 44, No. 6 (July-August 1965), pp. 937-997.
3. Breen, C., "The Picturephone® System: Customer Switching Systems," B.S.T.J., this issue, pp. 553-565.
4. Browne, T. E., Wadsworth, D. J., and York, R. K., "New Time Division Switch Units for No. 101 ESS," B.S.T.J., 48, No. 2, (February 1969), pp. 443-476.
5. Seley, E. L., and Vigilante, F. S., "Common Control for an Electronic Private Branch Exchange," IEEE Trans. Commun. and Elec., 83, No. 73 (July 1964), pp. 321-329.
6. Hughes, E. G., and Taylor, R. G., "The No. 101 ESS Control Unit," Bell Laboratories RECORD, 42, No. 2 (February 1964), pp. 61-66.
7. Priebe, H. F., and Seley, E. L., "No. 101 ESS 4A Switch Unit," B.S.T.J., 48, No. 2 (February 1969), pp. 321-329.
8. Herndon, J. A., and Tendick, F. H., Jr., "A Time Division Switch for an Electronic PBX," IEEE Winter General Meeting, New York, February 27, 1963, Paper No. CP63-577.
9. Averill, R. M., and Stone, R. C., "No. 101 ESS—The Time Division Switch Unit," Bell Laboratories RECORD, 41, No. 11 (December 1963), pp. 425-431.
10. Feiner, A., Lovell, C. A., Lowry, T. N., and Ridinger, P. G., "The Ferreed—a New Switching Device," B.S.T.J., 39, No. 1 (January 1960), pp. 1-30.

## **Maintenance**



# The *Picturephone*<sup>®</sup> System:

## Maintenance Plan

By H. J. DOUGHERTY, E. B. PETERSON and  
M. G. SCHACHTMAN

(Manuscript received September 23, 1970)

*The addition of Picturephone<sup>®</sup> service to the communications services offered by telephone operating companies requires that additional maintenance tasks be performed by the plant forces. The Picturephone plan provides the structure for integrating the new video telephone maintenance functions with existing telephone maintenance functions. Maximum use of built-in testing features in the switching machine and the inclusion of transmission testing facilities that are a logical extension of telephone testing arrangements provide economical and comprehensive tools for accomplishing the complex tasks of installation, circuit acceptance and trouble isolation. The flexibility of the plan permits the assimilation of advanced techniques into the field as well as prompt appraisal of field experience.*

*The plan is described in terms of its major segments, i.e., maintenance for switching, lines and trunks. The interaction of the maintenance hardware arrangements with the Picturephone plant is outlined together with an overview of the work locations and examples of tasks to be accomplished.*

### I. INTRODUCTION

Maintenance planning for *Picturephone* service has proceeded concurrently with the overall planning of equipment and service arrangements. Subscribers will expect *Picturephone* service to be of high quality and to provide nearly trouble-free operating performance. As seen in the other papers in this issue, the complex electronics used to provide *Picturephone* service has the potential to complicate the traditional maintenance functions of trouble detection, isolation, and repair. As in other aspects of the basic concept of *Picturephone* service, the maintenance will be integrated with the current maintenance and administrative operations of the telephone company, providing distinct cost advantages.

Maintenance and Plant Operations in the telephone operating company can be classified into three broad headings:

- (i) Preparing for service.
- (ii) Providing service.
- (iii) Sustaining service.

The first of these categories involves selection of cable pairs and verification that telephone plant is suitable to provide service ordered by a customer on an appropriate service date. Normally, a suitable pool of unassigned equipment is available to provide the service so that the need for holding a service order can be avoided. Next, the implementation of the service order requires that the selected equipment be connected and thereby dedicated to the use of a single customer in the case of individual lines or the aggregate of customers in the case of trunks and central office equipment. Verification that these equipments and interconnections are in fact operable is required before the service may be started. Finally, there will be occasions after the start of service when trouble arises, requiring maintenance action.

Planning for maintenance has also recognized that *Picturephone* service is a new and complex service that initially will be provided on a limited basis. When considering maintenance, the complexity and limited scope of initial *Picturephone* service work at cross purposes. Complexity (and newness) coupled with inexperienced craft people suggest the need for intricate and expensive test equipment designed to make decisions for the craft people (GO-NO-GO testing). A limited initial network, however, can ill afford high-priced hardware since the per-customer costs become excessive. The tradeoff between these factors was decided by the projected growth rate during the initial years. In the absence of a clear indication that a strong growth of service would occur in the initial period, a large expenditure for complex maintenance hardware is not justified. An acceptable alternative is the use of less expensive hardware, operated on a manual basis, coupled with careful training of craft people capable of adapting to a variety of situations. The shortage of skilled manpower complicates this approach, but it is anticipated that the measured growth of the service will not overwhelm existing training capability. During the initial service period, additional information will be gathered to guide the design of the next generation of complex and automatic hardware capable of quality maintenance for a system of increased size and utilizing generally available craftsmen.

## II. MAINTENANCE AND OPERATIONS PLAN

2.1 *Basic Considerations*

The philosophy that has guided the planning of *Picturephone* maintenance operations has been based upon a series of objectives, including:

- (i) Arrangements to insure service integrity and charge accuracy.
- (ii) Features for automatically detecting and identifying troubles.
- (iii) Features for manual trouble verification, location, and repair with centralized testing facilities where feasible.
- (iv) Provision for one-man test operations to the maximum extent possible.
- (v) Capability of adding additional automatic test arrangements as service demands increase.

Primary emphasis was placed on means required to insure that calls originated by customers are completed to the destinations desired with accurate recording of charges. One audio and two video transmission paths must be established between the originating customer and the distant party. Failure to complete any of these paths should always be detectable so as to avoid false charging and to promptly indicate a need to redial. Where possible, trouble detection should result in an alarm or trouble record and removal of faulty equipment from service. Receipt of the trouble indication should cause immediate action by the maintenance forces. Early corrective action is enhanced if one-man test operations are provided. Automation of maintenance functions promises to provide further cost reductions as the scope of *Picturephone* service is expanded. As mentioned in the introduction, close coordination with telephone maintenance was a necessary consideration in realizing all of these objectives. Work toward meeting these objectives has resulted in a System Maintenance Plan, the subject of this article. To grasp the scope of the plan, one must recognize that the designers of component facilities and subsystems specify for their individual subsystems the specialized techniques and tools required for maintenance. In general, these items (installation test sets, trouble detection circuits and alarms and special test sets for trouble isolation and repair) are tailored to the specific hardware item and do not result in system interactions. The System Maintenance Plan deals with the ensemble of subsystems that combine to make up the *Picturephone* service network.

This plan anticipates a *Picturephone* system that is not static, but

is constantly growing and changing. New station arrangements, new subscriber lines, and new trunks between switching offices must undergo *overall installation and acceptance testing*. Additions and rearrangements can adversely affect adjacent equipment through human error and therefore *routine performance testing* is required to detect degrading service situations due to the foregoing as well as normal aging effects. Failures detected by routine tests and failures reported by trouble alarms require that *trouble isolation testing* be instituted on some major portion of the overall system. These three types of testing on the overall system employ similar principles and require much the same craftsman training. In addition, consolidation of training programs and common use of test sets for the three types of testing result in cost savings.

Practical considerations of administration dictated that hardware development and the procedures for the three types of testing continue to be oriented toward the separate areas of responsibility as follows: (i) central office switching, (ii) line and station, and (iii) trunk maintenance. The plan envisions that administrative procedures and the organization of training programs, Bell System Practices, and records be divided according to these three responsibilities.

## 2.2 Work Locations

Figure 1 is a simplified block diagram of a *Picturephone* switching office, the local area served, and the trunking connecting the office with the remainder of the *Picturephone* network. The figure shows the work locations for switching, line and trunk maintenance.

Maintenance operations associated with the switching machine are largely automatic and take advantage of the common control and maintenance features provided for telephone maintenance. Telephone switching machines provide for per-call tests to verify the operation and continuity through the switch. In addition, as an aid for operational and transmission testing, automatic test terminations are normally available to enable craftsmen at distant telephone offices or on customers' premises to check circuits without the need for manual assistance at the switching machine location. Similar automatic test equipments are planned for the wideband switching network when *Picturephone* switching capability is added as shown in Fig. 1.

The manual work location for switching machine maintenance is at the No. 5 crossbar master test frame in the maintenance center. Trouble detected by the automatic equipment is indicated to the



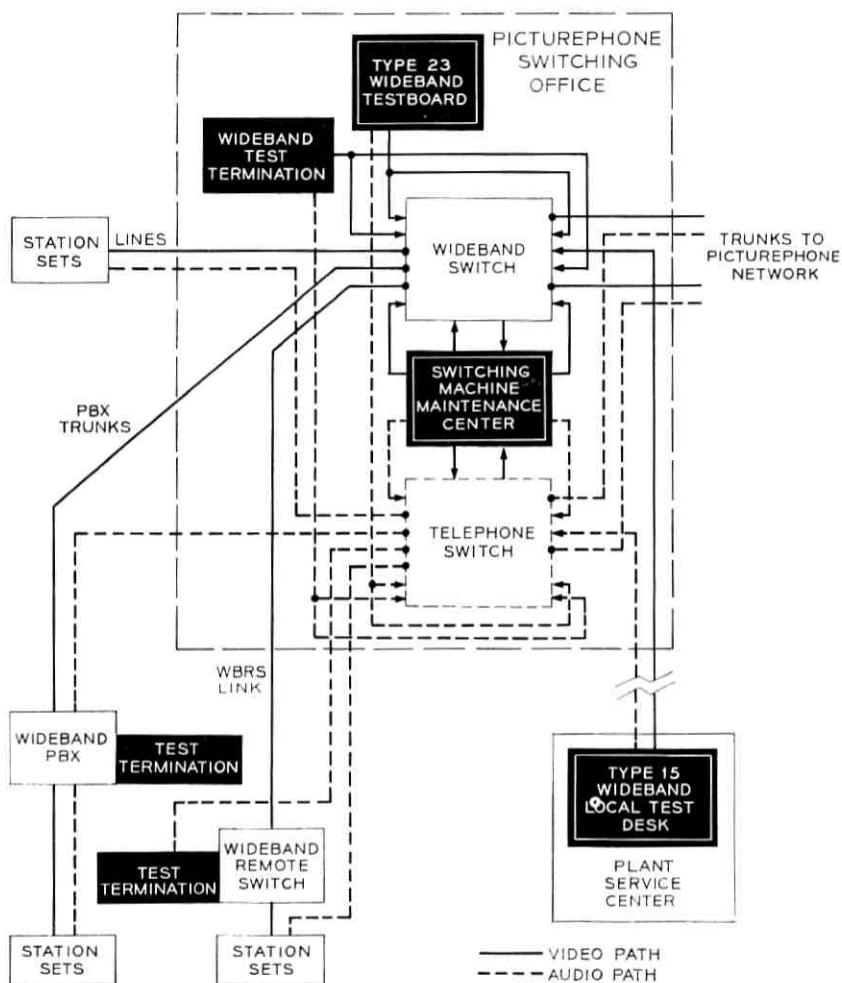


Fig. 1—Picturephone local office configuration with maintenance equipment and work locations (simplified).

craft people located there, and verification tests are conducted using the test access provided at the test frame. Full manual control of the machine (both telephone and wideband portions) is exercised from this work location.

The basic plan for *Picturephone* service provides that, in most cases, the customer's video service will be logically related to his telephone service by the regular telephone number. Telephone line maintenance

is centralized in a plant service center; *Picturephone* line maintenance can be accomplished from the same location. Thereby, the line record cards for both services are coordinated and uniform administrative procedures can be used to handle all customer troubles. Telephone line testing is accomplished from a telephone local test desk<sup>1</sup> and *Picturephone* line testing is accomplished from a wideband local test desk of new design. For both telephone and video, test access from the centralized test location to the thousands of lines served by the office is via the switching machine. Experience has shown that this means of access for telephone testing is quite adequate; for video, the minor variations in measurements due to variable paths through the switch must be considered, although switch performance is a minor source of impairments.<sup>2</sup> Figure 1 shows the work location for video testing and the test access path.

The recent trend in the routine testing of telephone trunks is toward centralized control of the automatic testing, particularly where there are a large number of trunks and trunk groups. In the initial years, the number of *Picturephone* trunks will be small and, in general, not integrated with the telephone network. *Picturephone* interoffice trunks form an independent network of six-wire trunk facilities<sup>2,3</sup> dedicated to *Picturephone* service use. Similarly it is not expected that the *Picturephone* hierarchy of switching offices will coincide with the telephone hierarchy. In light of the relatively small numbers of trunks and offices, it is clear that a complex arrangement for centralized maintenance cannot be justified at this time. In addition, the restrictive performance objectives for trunks<sup>2</sup> require precise measurements, a sufficient technical challenge in itself, and centralization adds the potential of additional measurement error. In contrast to lines, the independence of *Picturephone* trunks from telephone trunks means that there are no overriding administrative or record considerations that make it imperative to test telephone and *Picturephone* trunks from the same location. The work location is at a testboard in the vicinity of the wideband *Picturephone* switch (see Fig. 1). This nearby location is governed by the need to maintain nearly flawless transmission performance in the test access path to accomplish the precise measurements needed for trunk maintenance. Short access trunks, using initial service transmission equipment, achieve the required "transparency" at reasonable cost. Centralization, however, must be anticipated, but it remains for future consideration. To this end, switched access was chosen for all wideband test access (in contrast to some telephone practices where access is on a per-circuit jack basis) because of the potential for future centralization and the technical complications of

providing dedicated per-trunk jack type access for the wideband circuits at standard levels (without introducing intolerable path-length variations).

### 2.3 Additional Considerations

In addition to factors such as maintenance responsibilities and the associated work locations and test access, provision of transmission testing equipment for line and trunk testing has been based on the following additional considerations:

- (i) Initial service analog transmission facilities for lines and trunks are essentially similar.
- (ii) The nature of troubles and their prevalence will not be well defined until service experience is obtained.
- (iii) Training of craftsmen will be facilitated if testing equipment and procedures are analogous to those of existing telephone and carrier systems.
- (iv) Improvement and updating of individual equipment will be facilitated if, initially, test sets are restricted to single categories of tests and are *not* consolidated and if interactions between test instruments are kept minimal.
- (v) Costs and development intervals would be minimized if "on-the-shelf" commercial test sets could be utilized or modified for *Picturephone* service.

The system maintenance plan resulted from a synthesis of all of the aforementioned considerations of organization, technology, and cost. The following sections deal in greater detail with the plan from the standpoint of the switching machine, lines, and trunks.

## III. MAINTENANCE OF CENTRAL OFFICE AND SWITCHING EQUIPMENT

### 3.1 Maintenance and Plant Operation Functions

The use of a wideband switching network similar to the audio network and the inclusion of *Picturephone* features in existing common control circuits permit the *Picturephone* maintenance and plant operation functions to be combined with the present audio functions. Figure 2 illustrates the major blocks of the *Picturephone* system and the maintenance facilities for recording trouble information and testing troubles in a No. 5 crossbar central office.\*

---

\* Initially *Picturephone* service is supplied only from No. 5 crossbar systems equipped with a wideband switching network. The maintenance plan, however, is designed to anticipate provision of *Picturephone* service with other switching machines when suitable wideband networks are developed.



### 3.1.1 *Trouble Detection*

Provision is made to automatically verify the operation of common control circuits and the integrity of the transmission path. Rather than develop a number of special test arrangements that would continually check the video features, troubles are detected on a per-call basis, thereby avoiding false charging when troubles are encountered. The per-call detection tests performed on the video equipment are (i) tests of common control circuits, (ii) checks of the control signals associated with the remote switch, (iii) double connection tests, (iv) false-cross-and-ground tests, and (v) continuity tests.

### 3.1.2 *Verification of Common Control Circuits*

Verification of the operation of common control circuits handling the video call is performed in the same manner as for telephone since most of these circuits also process audio calls (e.g., marker, connectors, etc.). The generators providing the video supervisory signals (VSS),<sup>3</sup> the continuity oscillator and detector circuits, and the control leads connecting to the wideband remote switch units are verified on a continuous or per-call basis to insure that signals are transmitted within specific limits. A continuous monitoring of the video supervisory signal level is made at  $-4.5$  dB with reference to the 0 PTLP.\* A comparison of the continuity peak signal voltage level with a high and low reference voltage is made on a per-call basis.

The coding of the dc signals transmitted via the remote switch signal control circuit (RSSC) are also checked at both the RSSC and wideband remote switch (WBRS) on a per-call basis so that any trouble detected will permit a second attempt to be made to select a path through the remote switch. Replication of common control circuits is provided to permit second attempts to be made using alternate circuits. If second attempts fail, the common control circuits route the call to reorder or announcement thereby protecting accuracy in charging.

### 3.1.3 *Integrity of the Path*

Upon recognizing a request for *Picturephone* service, the common control circuits select idle audio and video paths and verify that they are capable of being used to connect the originating location with the

---

\* Zero *Picturephone* Transmission Level Point, a reference point specifying signal levels that is analogous to specifying elevation relative to sea level. At this reference point, the *Picturephone* signal is the same level as the output of the station set (100-ohm impedance).

terminating location. Double connection and false-cross-and-ground tests similar to those made in the audio path are made on the video path. In the event of a failure of these tests, the common control circuit makes a second attempt to select another path through the switching network. If no failure occurs in the above test, a continuity test of both the originating and terminating video path is performed by inserting a 12-kHz sine wave at a nominal level of  $-20$  dBm\* through the network. This test signal was selected based on economic considerations of the generator and detector arrangements, holding time to make the tests, and probability of faults affecting the megahertz bandwidth of the *Picturephone* signal. In making a continuity test, the 12-kHz test tone is sent over the transmit pair and returned over the receive pair. It is looped around at the station set via a zero loss loopback if customer loop facilities are being tested or at the distant office wideband trunk circuit via a 3-dB pad if trunk facilities are tested. The returned test tone is checked for  $\pm 1$  percent frequency and  $\pm 4.1$ -dB level on loops and  $\pm 1.3$ -dB level on analog trunks.

### 3.1.4 *Trouble Reporting*

When trouble is detected during call processing, the switching equipment determines the degree of severity of the trouble and activates a minor or major alarm.† This is an audio and visual alarm indication and is accompanied by a trouble record card in the No. 5 crossbar system. The circuits associated with the call at the time the trouble is detected are identified in order to facilitate locating the cause of the trouble. In the event that trouble is detected when common control circuits are not attached to the malfunctioning circuit, such as improper MF outpulsing to a distant office, lamp indications are activated at either the maintenance center and/or equipment to identify which circuit is in trouble.

### 3.1.5 *Trouble Location*

Provision is made to test the wideband functions from both the switching maintenance center and the equipment location. After a trouble is detected and reported, a central office craftsman analyzes

---

\* Measured at 0 PTLF.

† Occurrence of major or minor alarms is generally determined by the potential for disrupting many (major) or a few (minor) customers. For example, marker failure is a major failure while a signaling circuit failure will generally be considered minor.

the report and initiates appropriate tests to determine if it still persists. The majority of troubles occurring in the central office are expected to be identified by a single test. Arrangements have been made at the maintenance center to select any video path through the network, any wideband trunk circuit, either of two VSS generators and any WBRS. Additional control functions include the capability to cancel continuity checks, make a circuit busy to service and force trouble records on a test call.

### 3.1.6 Administration and Operation

Standard Bell System Practices have been prepared describing the central office maintenance arrangements required for *Picturephone* service. These will be incorporated into the existing Controlled Maintenance Plan (CMP) and Plant Management Instructions that are used to administer the switching system. The CMP prescribes scheduled routines for preventive maintenance in the office as well as methods for evaluating the equipment performance level of the office. Plant registers are provided to record the peg count of the total number of *Picturephone* calls requested and a total of those calls that fail. The present plant work unit plans that are used in determining the productivity of craftsman will be modified to include the added *Picturephone* functions described above.

Table I is a summary of the central office maintenance functions associated with *Picturephone* service.

### 3.2 Work and Test Access Locations

As noted in Section 3.1.5, the primary test facility in the central office will be the switching maintenance center. In the No. 5 crossbar system this is the master test frame which has a trouble recording facility adjacent to the control equipment. Test access to any central office circuit is gained by providing the marker with priming information from the master test control circuit. Originating test lines and terminating test trunks are connected to the maintenance center for testing the network, WBRS, and trunk circuits. Test access is available at the maintenance center for use with a *Picturephone* set in order to evaluate the qualitative performance of the transmission over the switching circuits tested. This same access is also used for measuring the central office transmission characteristics described in the paper on the transmission plan.<sup>2</sup> Provision was made in the continuity test circuits and VSS circuits to perform marginal tests of the frequency and level of their signals.

TABLE I—MAINTENANCE FUNCTIONS FOR *Picturephone* SERVICE  
IN CENTRAL OFFICES

---

AUTOMATIC TROUBLE DETECTION AND RECORDING

---

Common Control Circuits	Consistent with present test on a per-call basis.
Continuity Test	All transmission paths checked on a per-call basis.
Double Connection Test	Foreign potential and grounds within the switching network on a per-call basis.
Video Supervisory Signal Supply	Continuous trouble detection and automatic transfer to duplicate supply circuit.
Wideband Remote Switch Control	Per-call trouble detection and automatic transfer to duplicate control circuit.
Alarms and Recordings	Visual and audible alarms at central point. Trouble record card with call details.

---

AUTOMATIC ACTION AFTER TROUBLE DETECTION

---

Call Set-Up Trouble	Second trial using alternate equipment or path through switching network. Reorder after second trial failure.
---------------------	---

---

MANUAL TROUBLE LOCATION

---

Equipment Faults	Means for testing common control circuits (markers, connectors, etc.). Means for testing detection circuits (continuity, double connection test, etc.). Means for testing wideband remote switch control leads.
Path Faults	Selection of any path through switch. Selection of any trunk or WBRS link even if "maintenance busy."

---



### 3.3 *Testing Arrangement and Procedures*

The craftsman initiates tests of the central office wideband facilities either on a routine schedule or on a trouble report basis. Using the information he has received (for example a trouble record), he identifies the type of call on which the report was received and the circuits involved in handling the call. Using the controls at the maintenance center, he selects the same circuits and establishes a test call in order to verify the circuit operation. If the test call fails, he uses schematic drawing information, such as the sequence charts, to locate the specific cause of the trouble.

Several significant factors associated with *Picturephone* central office maintenance procedures are: (i) the compatibility with existing procedures (most of the test operations are identical to present testing of audio circuits), (ii) the ability to perform the tests without assistance of additional craftsmen (one man testing), and (iii) the verification and location of the defective unit, in many cases, from the maintenance center. This latter factor minimizes the need for a craftsman to be at the equipment for other than actual repair.

## IV. MAINTENANCE OF LINES

### 4.1 *Maintenance and Plant Operation Functions*

Line testing is required as part of overall installation and acceptance testing (service order procedures), for routine testing, and to isolate troubles referred to the plant service center. All *Picturephone* lines (both individual and key service) can be accessed for testing. Included are remote exchange lines, lines served from wideband remote switch units and PBX lines. Measurements are on a loop basis with signals transmitted through loopback circuits provided in the station set or the Key Telephone System (KTS) equipment. These loopbacks operate to connect the two directions of transmission at equivalent levels so as to permit full wideband round-trip testing.

Measurements are made of echo rating,\* broadband noise, crosstalk, low frequency noise, single frequency interference, impulse noise, and flat gain variation.<sup>4</sup> All of these impairments are allocated as discussed in the article on the transmission plan.<sup>2</sup> In addition, means are provided to interrogate the cable equalizers<sup>5</sup> to locate gross faults in

---

\*At the present time, a measurement algorithm suitable for field measurements of echo rating has not been developed; gain-frequency response measurements are being substituted for direct echo rating measurement as a first approximation.

equalizers or in connecting cable. Audio measurement capability is associated with the video-measuring facilities. Normal techniques presently applied at local test desks (LTD) such as the No. 14 type are used. Any confirmed audio-only troubles, however, requiring extensive measurements can be referred to the telephone (only) LTD so as not to tie up expensive video testing facilities.

No requirements for automation of routine testing (other than the per-call tests mentioned in Section III) were specified for the initial service arrangements. Independent means for measuring each of the impairments considered in the performance objectives<sup>2</sup> is the only practical course until experience with the system can be obtained. However, access arrangements and test terminations have been defined to aid in trouble isolation. Provision is therefore made for switching machine access to any desired line with the ability to manually cancel continuity check and to control ringing. Selection of test access to different office locations is by key control. Connection of test instruments is also by key control, eliminating both the need for cords and possible operator error.

#### 4.2 Work and Test Access Locations

The major test facility for line maintenance is the No. 15 *Picturephone* local test desk shown in Fig. 1 and described more fully in a companion paper.<sup>6</sup> This desk is arranged to normally obtain switched access to lines via a test trunk having an appearance on the trunk link side of the *Picturephone* switching network (No. 5 crossbar). This trunk is designed to be as nearly transparent to *Picturephone* signals as possible and includes a means for checking the transmission integrity of the access path before testing. The test desk also has auxiliary access to lines via the main distributing frame (MDF) which is the interface between the office equipment and the outside plant. This MDF trunk must be patched manually to line appearances permitting measurement of the outside facilities alone or (in conjunction with the switched access trunk) the central office equipment alone.

In addition to the test trunk providing test access from the No. 15 LTD, there are two other test trunks in the central office for line maintenance. One provides access to the video switch at the trunk link frame from the master test frame. This trunk permits testing of lines in small offices not equipped with a No. 15 LTD. The second trunk is a test termination functioning as a station ringer and video test facility. Such terminations are used by station installers and repairmen to verify the proper operation of station sets.

To permit sectionalization of troubles and to provide a known termination and test access point, test circuits are provided at PBX and WBRs locations. These test terminations have a listed directory number (in the case of the WBRs) or a PBX extension number. Test calls can be directed by the deskman over a selected PBX trunk (or WBRs link) to the appropriate test termination to determine that the trunk (link) is satisfactory. This technique is illustrated in the next section below. Note that while the trunk (link) is used for test access to lines served from the PBX (WBRs), its maintenance is a trunk responsibility (see Section V). The test terminations also provide access for testing should (i) a PBX repairman be dispatched to the PBX or (ii) a central office switchman desire to test at a WBRs located in a central office not provided with full *Picturephone* switching.

As stated previously, provision of a wideband loopback circuit at customer locations avoids the need for always dispatching a repairman to the station location for trouble isolation, circuit order or routine tests. A limited need to involve the subscriber during trouble isolation is unavoidable. Once the trouble is isolated to the customer location, then a repairman is dispatched to the customer premises where testing as required occurs using portable test equipment: (i) on a one-man basis to the station-ringer test termination, or (ii) on a two-man basis with the desk man at the No. 15 LTD.

#### 4.3 Testing Arrangements and Procedures

Video measurements from the test desk are accomplished using the following test instruments, described in a companion paper<sup>a</sup>

- (i) Wideband oscillator,
- (ii) Wideband voltmeter,
- (iii) Waveform oscilloscope,
- (iv) Wideband noise measuring set,
- (v) *Picturephone* station set,
- (vi) *Picturephone* test signal generator, and
- (vii) *Picturephone* fault locating test set.\*

The adjustable wideband oscillator together with the wideband voltmeter are used to obtain gain-frequency response measurements. Pre-programmed or switch selection of test frequencies, although desirable operationally, are not provided since measurement pro-

---

\* A portable test set normally used by central office repair forces available for connection to test desk and boards as required on a patch cord basis. Other portable test equipment may also be connected as required.

cedures are expected to change as experience is gained. Therefore, the oscillator must be set manually to any desired frequency in the *Picturephone* band. The wideband voltmeter measures on a broadband basis; the expected noise on *Picturephone* circuits will be such that selective measurements are not required to obtain accurate gain-frequency data. The waveform oscilloscope has provisions for examining the video waveform transmitted either from (i) the subscriber's station or (ii) the test signal generator and/or the *Picturephone* station transmitter in the No. 15 LTD on a round-trip basis via the various loopbacks. This will be a valuable tool for trouble isolation. The oscilloscope has provisions for examining the waveform at line rate (8 kHz) or field rate (60 Hz). The sync pulse can be examined without de-emphasis to determine sync distortion. The video can be examined with de-emphasis to view waveform distortion. The test signal generator produces signals especially useful for these tests.

The wideband noise measuring set is equipped to measure either weighted broadband noise or impulse noise. When used to measure broadband noise, it will measure both random noise and single frequency interference. The requirements are such that a single circuit order or routine measurement will indicate failure due to either source. The measurement is made on a loopback basis by measuring the sum of the near-to-far plus the far-to-near noise. Impulse noise is measured by recording the number of impulses exceeding a variable threshold. Low frequency noise is measured by the wideband voltmeter.

Crosstalk measurements use the test signal generator to transmit a distinctive pattern that is observed on the waveform oscilloscope or *Picturephone* set receiver. In general, only self-crosstalk can be measured; the requirement for self-crosstalk is more severe than worst disturber crosstalk<sup>2,7</sup> due to the assumption that multiple exposures of the same signal source produce a greater interference effect than multiple disturbers where each disturber is from a different signal source.

For initial service, circuit acceptance and routine tests are the same although routine tests have wider limits.\* The data from the initial installations will be closely observed to determine whether the limits or routine intervals should be changed.

---

\* *Picturephone* service is engineered to provide high-quality service under expected conditions of plant variability. The test limits of routine tests serve to trigger corrective action only when performance is degraded. Allocation of margin principally to plant variability (and not installation error) is accomplished through tighter circuit acceptance limits.

Trouble isolation testing will normally involve a repetition of the routine tests (if the test is due to a trouble report) to obtain verification test data for comparison to the installation and routine records. As the expected troubles are quite varied, a flow diagram approach has been adopted to guide the deskman in sectionalizing the trouble. Figure 3 illustrates this procedure for a station trouble where the

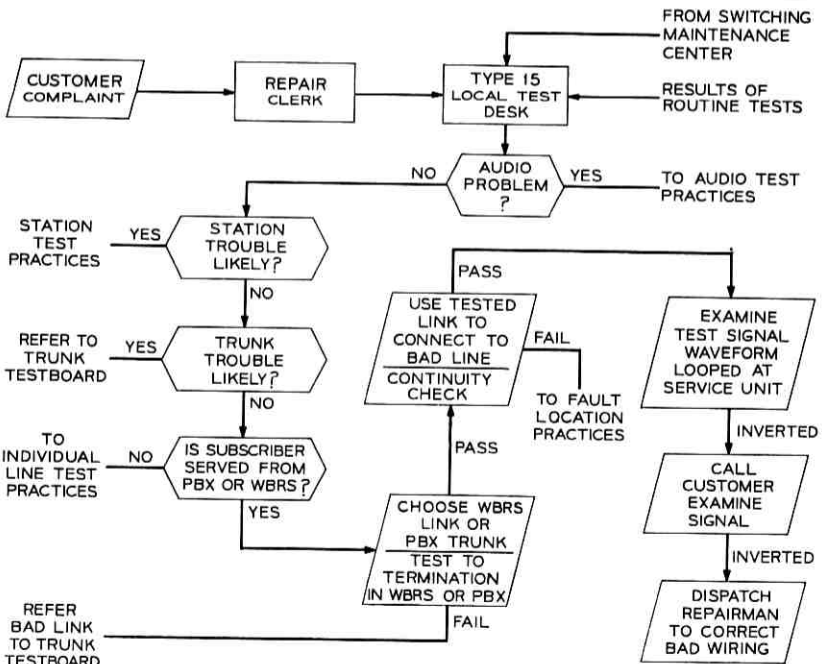
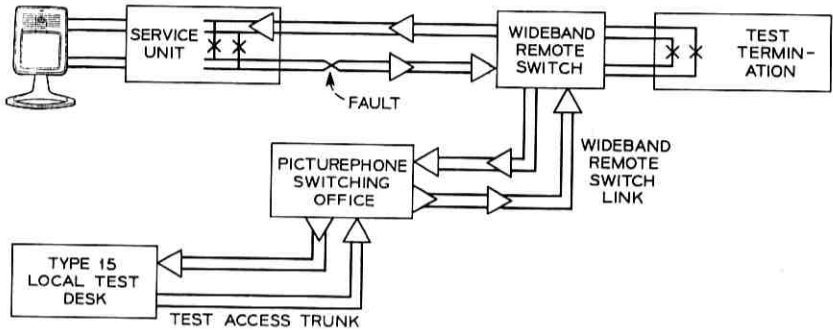


Fig. 3—Example of line fault location.

station is served directly from a WBRs. The trouble example is a reversal of leads in the transmission path from the subscriber to the WBRs, possibly resulting from rearrangements at the customer's premises. The symptom of this trouble is the inability of the called party to see the customer who is in trouble. However, the calling customer can see the called party. Although not shown in the figure, operational practices provide instructions for the tasks indicated in the blocks. The actual procedure flowchart is more detailed and has been simplified for the figure. Note that the flow diagram provides that possible troubles in the access links be referred to the organization having responsibility for trunk testing. The No. 15 LTD has the capability to direct signals over chosen links to the WBRs test termination so as to verify the fitness of the remote switch link used as an access path to the WBRs.

It is expected that detailed transmission measurements can be adequately made from the test desk in most cases. Complex test equipment will not be required in the field. Experience may indicate, however, that testing resolvability is not adequate when the transmission distance between the station and the test desk is great (possible with a long line served from a WBRs that is also a long distance from the *Picturephone* switching office). The maintenance plan provides for access for portable test sets at a jack located in conjunction with the test termination at a PBX or WBRs.

## V. MAINTENANCE OF TRUNKS

### 5.1 *Maintenance and Plant Operation Functions*

In initial service, three types of trunks are provided and the testing procedures differ accordingly:

- (i) PBX trunks or WBRs links—providing transmission between the wideband line link and a PBX on customers premises or a wideband remote switch in a central office or on customers premises. The PBX trunk is a six-wire trunk with the audio and video path dedicated to *Picturephone* calls only. The WBRs link is a four-wire link transmitting only video; the audio is carried on the customer's regular audio pair.
- (ii) Interlocal and toll connecting trunks—dedicated six-wire analog trunks connecting class 5 (end) offices or class 5 to class 4 (toll) offices. In general, these will be of moderate length and confined to a local analog area within a city.
- (iii) Intertoll trunks—dedicated six-wire trunks comprising analog

or combined digital-analog facilities connecting class 4 offices. For initial service, all switching will be analog and therefore all trunks will have analog end sections requiring analog maintenance arrangements.

In telephone service, it is the usual practice to provide fully equipped testboards only at toll offices. Less sophisticated arrangements are normally found at class 5 offices. For *Picturephone* service, testing arrangements are designed to provide testboard functions at all classes of office.\*

The same measurements will be required on *Picturephone* trunks as on lines. Measurements however, will be made on both one-way and looped bases as is appropriate to the measurement and type of trunk. The availability of testboard arrangements at both ends of toll connecting and intertoll trunks assists in accomplishing the tighter requirements that are consistent with trunk objectives.<sup>2</sup> Routine measurements will be more frequent, and test termination arrangements are provided to facilitate single-man routine testing wherever possible. Audio testing ability is included with the video because of the separate dedicated network approach associated with *Picturephone* trunking.

For initial service, the facilities providing trunk transmission are configured using the same hardware as line facilities, although to different engineering rules. Hence the same considerations of flexibility in testing approach apply. Initial requirements for test equipment therefore stressed independent means for measuring the expected impairments, using the same instruments provided for line maintenance in the No. 15 LTD.

### 5.2 Work and Test Access Locations

The testboard arrangement designed for initial *Picturephone* trunk testing is the No. 23 testboard (TB) shown in Fig. 1 and described more fully in a companion paper.<sup>6</sup> This testing arrangement operates in conjunction with a single No. 5 crossbar switching machine. Test connections are established (i) between a test trunk appearance on the trunk link to PBX trunks or WBRs links that appear on the line link, or (ii) between a test trunk tandem appearance on the line link to any trunk on the trunk link. All connections are made on a six-wire basis. Incoming test calls from distant locations are directed to test

---

\* For initial service, the number of trunks may not justify provision of a full testboard; arrangements would then be made to provide the same testing functions at the switching maintenance center.

trunk appearances (i) on the line link in class 5 offices (7-digit telephone number) or (ii) on either the trunk link (101 test termination) or line link (7-digit telephone number) in class 4 offices. Only *Picturephone* calls will complete to these trunks.

The No. 23 TB is designed to establish test connections to particular trunks or links and to override maintenance busy signals that prevent completion by the switch of connections to defective trunks. Means are provided in association with the testboard to test the control and signaling functions of any trunk. The design of the wideband switching network to minimize path length variation and the effect of switch multiples permits measurements of a high degree of accuracy to be made although access is on a switched basis. In addition, a calibration loopback is provided at the video trunk circuit of the test access trunk permitting the craftsman at the testboard to verify the transmission performance of the test trunk before commencing a test. It is expected that the transmission path between the testboard and the switch will be measured at frequent intervals to insure the testboard's accuracy as a measurement *system*.

In addition to the 101 test trunk appearance on both audio and video switches, other coded test trunks provide the video equivalent of test terminations used for maintenance testing on the DDD network. A code 100 termination is used for combined far-to-near loss and noise measurements. A code 102 termination is provided for far-to-near low frequency loss measurements and to provide a loopback for more general two-way measurements on a loop basis. The test lines are described more fully in a companion article.<sup>6</sup>

### 5.3 Testing Arrangements and Procedures

The No. 23 TB includes the same video test equipment provided in the No. 15 LTD (see Section 4.3). In addition a voice frequency gain and noise measuring system is provided that is equivalent to that used for maintenance operations at conventional telephone trunk testboards.

Major responsibilities to be carried out at the No. 23 TB are circuit acceptance and routine tests on all trunks leaving the office. These tests are similar to the tests performed from the No. 15 LTD, although to tighter tolerances reflecting the trunk objectives.<sup>2</sup> Tests on PBX trunks or WBRs links are on a loopback basis to the test termination loopback in the PBX or WBRs. Circuit order tests on trunks are in general conducted on a two-person basis between No. 23 TBs at the ends of each trunk. Routine tests are on a loopback basis to distant



102 test terminations. These routine tests are required more frequently during the period of initial service since temperature regulation is not provided for trunks. Routine tests serve to monitor trunk performance and serve as an indicator of the need for trunk realignment.\*

Trouble isolation is basically a sectionalization procedure utilizing test terminations and the switched access from the No. 23 TB. Figure 4 provides an illustration of this technique. Office A is the control office responsible for maintenance of the trunk. An open circuit at a wideband distributing frame as indicated has resulted in failure of continuity check on a call from A to B. The trouble indication at A is reported to the No. 23 TB at A. The tester at the testboard deduces that the trouble does not involve the tandem equalizers (1) and (2) since those equalizers are not involved in calls from A to B. He then dials the code 102 test termination at B, receives a signal from office B over the far-to-near path, but no return of a round-trip signal when the test line switches to the loop-back mode. The conclusion is that the trouble is in the near-to-far path. The 102 termination is disconnected and the craftsman redials the code 100 test termination at B. This prevents seizure of the trunk from office B and provides a termination of the trunk during fault location. The craft person then interrogates equalizer pairs (3) and (4), (5) and (6), and (7) and (8), normally tested from office A to obtain a response from the fault locating circuitry. These test all right.

The tester then reasons that equalizer pairs (9) and (10) and (11) and (12) should be tested and, because these can be tested only from office B, a test call is directed to the distant No. 23 TB (via a code 101 test termination). The tester at the No. 23 TB at office B checks equalizer pairs (11) and (12), and (9) and (10). Tandem equalizer pair (13) and (14) is not checked since the continuity failure was between the trunk circuits of offices A and B. Equalizer pairs (11) and (12) and (9) and (10) do not provide a return of the fault locating signal indicating a failed equalizer, trunk circuit or interbay wiring. The tester at office B must therefore check his office. Since the tester at office A is no longer required, he releases the connection and the tester at office B establishes a connection to the code 100 termination at office A, providing a resistive termination for both directions of video transmission. He directs that a patch be

---

\* 12-kHz continuity tests performed on a per-call basis (see Section III) serve to detect gross failure or trunks considerably out-of-limits. They cannot serve as a measurement of misalignment due to temperature.

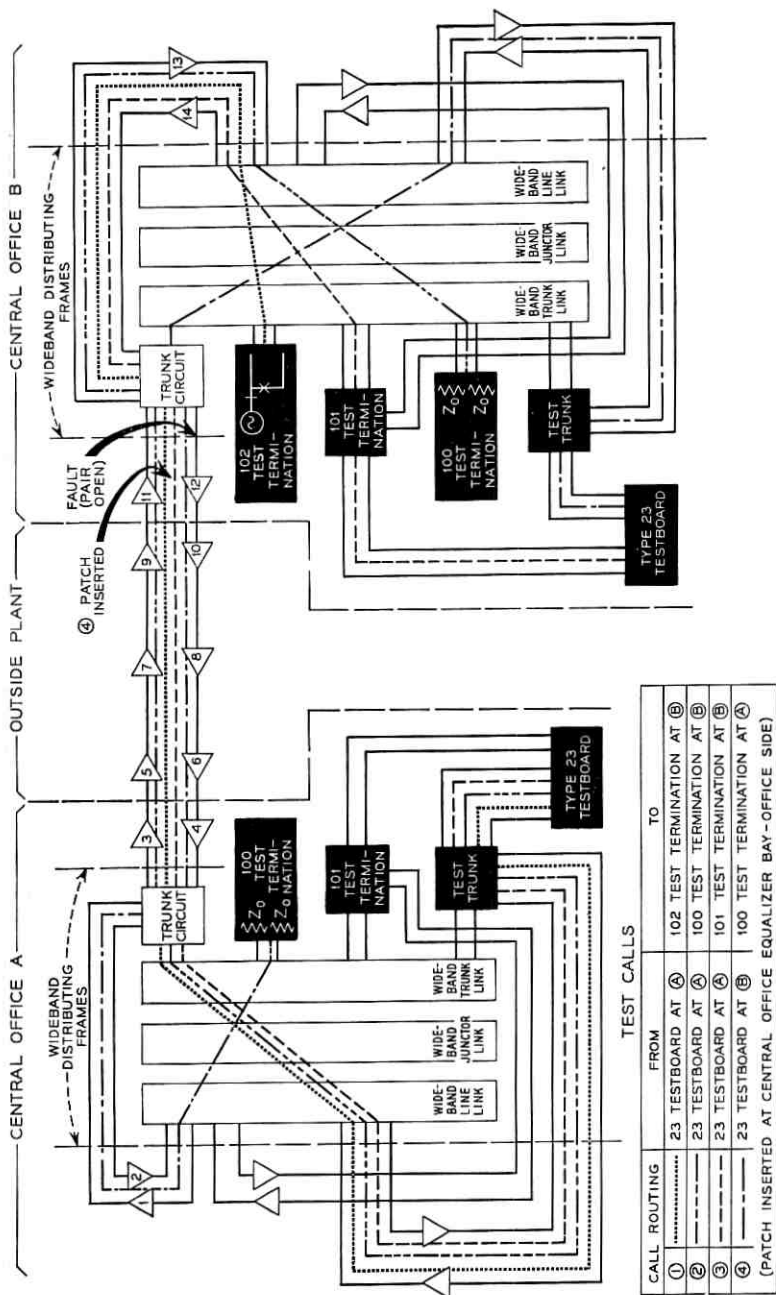


Fig. 4—Trouble isolation on trunks.

inserted at jacks located on the office side of equalizer pair (11) and (12). When he receives no response, he then reasons that the fault is in the cabling or the trunk circuit at office B and refers the trouble to the central office repair forces. They use the test access and control described in Section III to isolate the trouble.

Long *Picturephone* service trunks between central offices consist of both analog and digital portions during initial service. The digital portions utilize facilities that incorporate sophisticated means for determining when transmission is not acceptable. The output of this trouble detection arrangement is coupled to alarm reporting systems insuring that faulty trunks are turned-down from service. The No. 23 TB will be used for overall maintenance of these trunks. However, the non-linear nature of the codec (coder-decoder) that processes the analog video signals for transmission in digital form prevents meaningful quantitative overall measurements from being made on digital trunks. Quantitative measurements are made on the analog end sections through manually operated loopbacks at the analog-digital interface. Overall qualitative measurements are made using visual appraisal of the *Picturephone* test signal generator signal or any suitable *Picturephone* signal transmitted from the distant end.

## VI. CONCLUSIONS

The maintenance plan and hardware developed to accomplish it are expected to result in performance consistent with the premium price of the service. Automatic checks of all components of the video connection at call setup will insure that major failures are promptly detected. Routine measurement of transmission parameters will trigger maintenance action when transmission is out-of-limits. The plan meshes well with existing telephone company organization and procedures and retains flexibility to adapt to unknowns. The associated hardware is within the state-of-the-art and utilizes proven testing approaches.

To improve efficiency and reduce cost, the initial service maintenance performance will be closely monitored. Data will be needed on the man-machine interface and actual performance so that unneeded flexibility now included can be pared and training simplified.

Effort continues to be directed toward planning of improved call-setup testing techniques and off-hours automatic circuit testing to improve the effectiveness of routine testing without heavy manpower requirements.

## VII. ACKNOWLEDGMENT

The authors wish to thank their many colleagues at Bell Telephone Laboratories who have been involved in maintenance planning. In particular, the contributions of E. R. Gallun and C. W. Haas, Jr., have greatly influenced the development of the plan. Suggestions, contributions, and criticism by D. N. Carson have been invaluable in establishing testing methods and instrumentalities.

## REFERENCES

1. Ritchie, D., Jr., "Improvements in Subscriber Line Testing," Bell Laboratories Record, 30, No. 8 (August 1952), pp. 337-340.
2. Brown, H. E., "The Picturephone® System: Transmission Plan," B.S.T.J., this issue, pp. 351-394.
3. Urich, J. F., "The Picturephone® System: Switching Plan," B.S.T.J., this issue, pp. 521-531.
4. Crater, T. V., "The Picturephone® System: Service Standards," B.S.T.J., this issue, pp. 235-269.
5. Brown, J. M., "The Picturephone® System: Baseband Video Transmission on Loops and Short-Haul Trunks," B.S.T.J., this issue, pp. 395-425.
6. Favin, D. L. and Gilmore, J. F., "The Picturephone® System: Line and Trunk Maintenance Arrangements," B.S.T.J., this issue, pp. 645-665.
7. Bunin, B. J., Hirsch, R. B., and Olsen, R. E., "The Picturephone® System: Crosstalk Considerations in the Transmission of Analog Signals on Paired Cable," B.S.T.J., this issue, pp. 427-457.

## The Picturephone<sup>®</sup> System:

# Line and Trunk Maintenance Arrangements

By D. L. FAVIN and J. F. GILMORE

(Manuscript received September 1, 1970)

*New line and trunk maintenance arrangements have been developed for Picturephone<sup>®</sup> service. A switched method of accessing lines and trunks by means of the No. 5 crossbar system has been developed. In addition a new local test desk (LTD No. 15) and testboard (TB No. 23) have been made available. In this article we discuss these arrangements which represent another application of the switched access concept as an integral part of line and trunk maintenance arrangements.*

### I. INTRODUCTION

The wideband nature of the facilities for *Picturephone* service introduces a new degree of complexity in the methods used to maintain lines and trunks. Access methods acceptable in the telephone network are not applicable in this wideband network; in addition, testing capabilities are greatly expanded. The Local Test Desk No. 15 and Testboard No. 23 are designed to provide the required access and test capability. The equipments obtain access by means of the No. 5 crossbar system and it is suggested that the article on that switching system be read first.<sup>1</sup>

### II. BACKGROUND

The objective of integrating the *Picturephone* service into the existing telephone plant requires that the maintenance arrangements should blend into and be absorbed by existing areas of responsibility in the operating companies. Line and trunk maintenance arrangements consequently are treated separately and distinctly, as they currently are, and they are compatible with the existing arrangements.

Figure 1 is a simplified illustration of the existing maintenance arrangements. Lines, in general, are tested on a centralized basis from

local test centers. The test centers are equipped with local test desks (LTD) which have access to a number of offices by the use of interconnecting test access trunks. Trunks, on the other hand, are tested on a decentralized basis with each office having a testboard or test frame for maintenance purposes. The integration of the new *Picture-phone* service maintenance arrangements into the existing testing areas heavily influenced the physical design of the equipment. In the case of line testing a console type of LTD is used, and for trunk testing an upright frame type of testboard was developed.

### III. LINE MAINTENANCE

#### 3.1 Current Arrangements

A line provides the communication channel between a customer and his serving central office. The channel is physically outside the office and troubles in it will generally result in the dispatching of personnel for repair purposes. Consequently, there is no need to locate test desks physically in the office served. Each test center is equipped with a

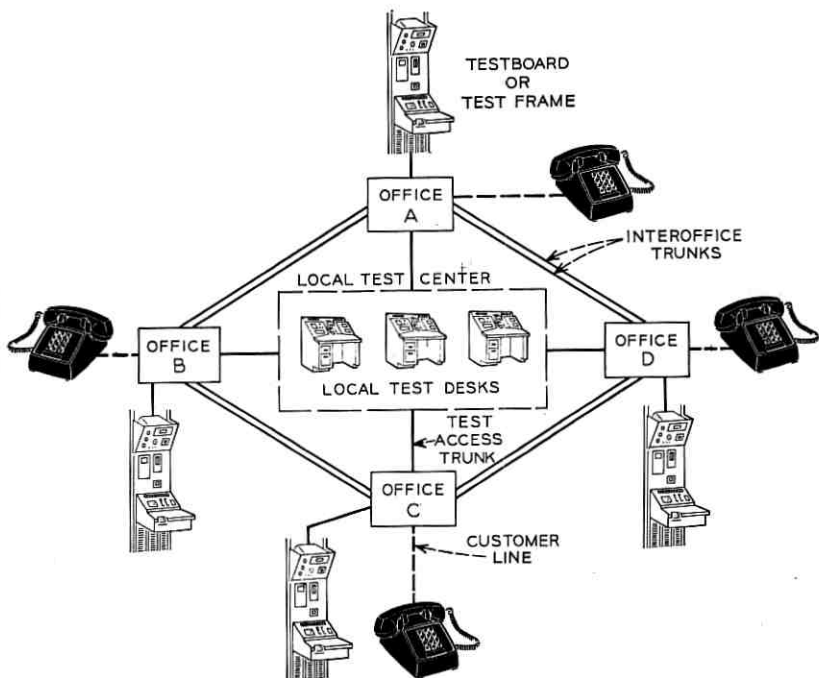


Fig. 1—Line and trunk maintenance arrangements.

number of LTDs as required by the maintenance load and the LTDs provide the necessary access facilities for the offices served. The access to lines in an office is obtained by seizing a trunk to the appropriate office and keying in the desired customer directory (four-digit) number.

For telephone plant, the lines are generally metallic pairs of wires and the tests are of a dc nature, e.g., tests for shorts, opens, grounds, resistance or foreign potentials. The introduction of *Picturephone* service drastically changes the make-up of the customer loop.<sup>2</sup> The signaling and audio information will continue to use a simple wire pair but the video information will be transmitted by two additional pairs, one for transmission in each direction. These pairs require the use of active devices (cable equalizers) to compensate for any losses across the frequency band. What was formerly a simple two-wire line now becomes a six-wire facility with the wideband four-wire portion requiring more sophisticated tests, measurements, and trouble detecting and isolating techniques.<sup>3</sup> The need for expanded test capability and the wideband nature of the facilities all contributed to the need for the development of the new Local Test Desk No. 15.

### 3.2 *Local Test Desk No. 15*

The LTD No. 15 is a cordless console which provides key-up access to circuits and test equipment (see Fig. 2). The test desk is functionally designed so that most of the *Picturephone* test equipment is on the left-hand side and the audio test equipment is on the right-hand side. The center of the console has a recessed opening for the *Picturephone* station set. In addition, keys and a visual readout associated with the access circuits are located in this area. The test equipment is shared between two access circuits. This allows for two different tests to be conducted on two circuits simultaneously. The console is equipped to serve up to five offices, each having varying quantities of six different types of interconnecting test access trunks. Due to transmission impairments that could be incurred in these trunks, it is necessary, initially, to restrict their length to less than 1500 feet. This in turn implies that the offices served must be in close proximity to the local test center. Figure 3 is a simplified illustration of the test-access trunk arrangements.

### 3.3 *Test Access Trunks*

The main functions of the LTD are to isolate line troubles, temporarily remove the line from service, and dispatch repair personnel. The six different types of test-access trunks provided permit test personnel

to perform the first two of these functions. The types of trunks and their function are as follows:

- (i) *Six-wire wideband test trunks.* These trunks provide test personnel with the means for obtaining access and testing the audio and video portions of lines, PBX trunks, and remote switch unit (RSU) links.
- (ii) *Two-wire test trunks.* These trunks allow tests to be performed on the audio portion of a line without requiring the use of a six-wire wideband test trunk.
- (iii) *Two-wire MDF trunks.* These trunks permit test personnel to split or bridge the audio line at the main distributing frame (MDF) in order to determine if an audio trouble is in or out of the central office.
- (iv) *Four-wire MDF trunks.* These trunks permit test personnel to split or bridge the video line at the MDF in order to determine if a video trouble is in or out of the office.



Fig. 2—Local Test Desk No. 15.



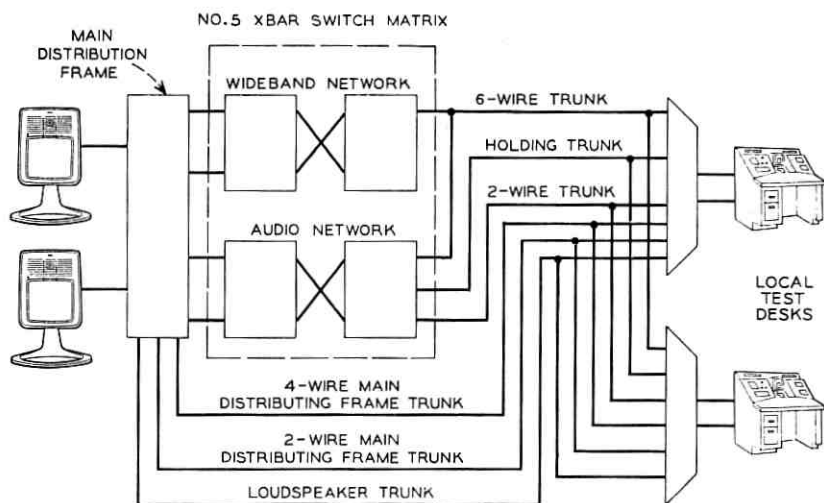


Fig. 3—Test access trunk arrangements, Local Test Desk No. 15.

- (v) *Loudspeaker trunks.* These trunks are used for communication purposes and allow test personnel to call for assistance at the MDF location.
- (vi) *Holding trunks.* These trunks allow test personnel to remove a PBX trunk or RSU link from service if trouble is encountered. The trouble would then normally be referred to personnel at the Testboard No. 23.

### 3.4 Test Trunk Selection

The test trunks to the various offices served by a test center equipped with LTDs No. 15 will be shared by each of the test desks in the test center. This is accomplished by providing each LTD with a small switching network. The trunks have multiple appearances on the office side of the network as shown in Fig. 3 and each LTD "bids" for a trunk by means of one of its two access circuits. The bidding for a test trunk involves the following key selections at the LTD:

- (i) Select one of the two access circuits.
- (ii) Select the office desired.
- (iii) Select the type of trunk desired.

Upon the third selection, control circuits in the LTD test to determine which trunks are busy and return a display indicating the status (busy or idle) of each trunk in the group desired. One of the idle

trunks is then selected by a key operation and the LTD is cut through the network to the test access trunk which connects to an incoming test trunk in the desired office. This test trunk is in a loop-back mode, as far as its video pairs are concerned, and at this point looped measurements can be made from the LTD to check the continuity or quality of the looped test trunk to the designated office.

### 3.5 Line Access

Normally, having selected the test trunk, test personnel would key up the necessary digital information to make a line connection, observe a display of the information, and verify it. To access a line this information needs to be spilled forward to the distant office. This is accomplished by a key operation which results in opening the distant office trunk loop-back and bidding for a trunk-test register to receive the digits. The register is attached and a reversal of battery signal is returned to the test desk. This signal activates a temporary store in the test desk and results in the digits being spilled forward. The register receives them, passes the information to the marker, and a connection is made to the line. Figure 4 is a simplified illustration of the arrangement.

### 3.6 Local Test Desk—Central Office Communication

The digital information required by an office for testing in the telephone network is the directory (four-digit) number. With *Picturephone* service, however, additional digits are used to provide a number of desirable and necessary features. For example, if only the video portion of a customer line has been found in trouble, there is no need to bar service on an audio-only basis. Consequently, provision has been

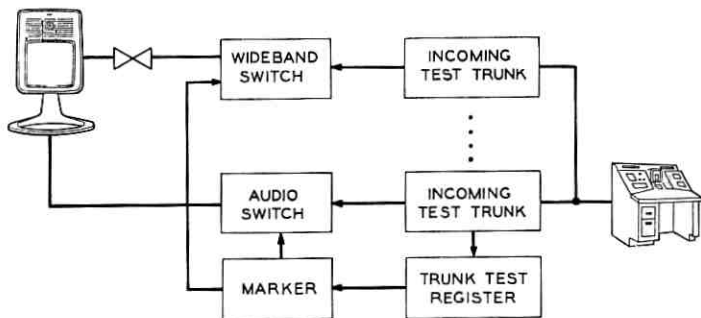


Fig. 4—Picturephone line maintenance arrangements.

made for the common control equipment in the office to recognize this and allow non-*Picturephone* calls to be completed. A line in this state is said to be in trap. The test desk needs the ability to place a line in trap, determine if a line is in trap, make a test call to a line in trap, and finally remove the line from trap following its repair. This requires that more than just directory information be passed between the test desk and the switching machine.

In accessing a group of PBX trunks or RSU links,<sup>1</sup> the normal function of the switching machine is to hunt over the group, select an idle one, and complete a connection. In testing *Picturephone* circuits, however, it is desirable to be able to select a particular trunk or link and this also requires passing more information to the switching machine.

As a result of these considerations and others, the No. 5 crossbar common control requires a maximum of 19 digits for completion of an LTD test call. The first five digits provide class information which indicates to the switching machine the type of call being made. Digits 6 through 8 cover special features such as canceling the continuity check, overriding a make busy or inhibiting hunting. Digits 9 through 12 identify the particular RSU link or PBX trunk to be used in the call and digits 13 through 19 identify the station being called. In cases where particular digits are not required for a call, a skip (SK) digit is sent.

Errors are minimized by a provision whereby the keyed digits are temporarily stored in a register and displayed for verification before being spilled forward. In many cases, only a few of the digits will change from test call to test call and for ease of operation a partial erase feature is provided.

### 3.7 *Trouble Isolation*

The primary function of local test centers is the detection and isolation of troubles in customer lines. Since personnel are ultimately dispatched to clear the trouble, it is essential that troubles be pinpointed as much as possible. The line facilities for *Picturephone* service will generally include active devices or cable equalizers with a loop-back at the key system or station set when it is in the idle mode.<sup>2,3</sup> With this arrangement, faults on the video pairs can be detected by transmitting test signals over the line and observing the returned signal. Should there appear to be a cable equalizer failure, a portable cable equalizer fault locating test set may be used at the LTD to isolate the trouble. With this test set, a major fault can be isolated by interrogating each equalizer with its own preassigned low-frequency signal. If operating,

the cable equalizer will return a signal one half the frequency of the interrogating signal.

Test terminations (Section VII) are also provided at RSUs and PBXs so that the LTD can test the individual links making up a connection. In this use, the test termination would be accessed from an LTD over a particular RSU link or PBX trunk. Upon seizure, the test termination returns a signal which tests the far-to-near condition of this link or trunk. After a timed interval, the test line will then establish the loop-back condition and the test desk can then send a signal over the line, testing the condition of the transmit portion of the link or trunk. These arrangements will enable test personnel at the LTD to analyze and diagnose troubles.

#### IV. TRUNK MAINTENANCE

##### 4.1 *Planning Considerations*

Trunk testing facilities in the existing telephone network have generally used a dedicated jack arrangement whereby the various test-access points in a trunk circuit are wired to a set of jacks and lamps in a testboard. The lamps are used to provide the trunk status information while jacks provide access to particular leads or points for test purposes. Typically, the transmission and control leads (tip, ring, and sleeve) are brought over by a bridged connection from the trunk circuit and signaling leads are looped through testboard jacks to permit signaling tests. Access to the test equipment is generally made through a set of cords which are manually plugged into the jacks.

The jack access arrangement, normally used in testboards, is undesirable for *Picturephone* trunk maintenance. The use of a multiple off of a wideband pair could cause transmission difficulties because, electrically, it is equivalent to a high frequency bridge-tap stub. An effective method of minimizing this problem is to provide switched access for reaching these pairs. This method of access was provided for both the video and audio transmission pairs. In addition to the transmission testing, however, there is still the need to provide access to the control and signaling leads as well as to get trunk status displays. The most economical means of doing this was deemed to be the use of the direct cabled, jack access arrangement. The Testboard No. 23 was developed with these considerations in mind.

##### 4.2 *Testboard No. 23*

The Testboard No. 23 comprises two individual test positions. One, the transmission test position (TTP) is cordless and key-ended; it uses

switched access through the No. 5 crossbar machine to gain access to the transmission pairs of trunks. The second position, the miscellaneous test position (MTP), is a cord type position which provides jack access to the control and signaling leads of the trunk circuits. The number of MTPs depends primarily on the number of trunks in an office while TTPs will be provided as required by the maintenance load.

#### 4.2.1 *Transmission Test Position*

The TTP is a cordless, upright test position which provides key-up access to circuits and test equipments (see Fig. 5). Access to test trunks is obtained by selecting one of two access or position circuits and bidding for the test trunk by a key operation. The access to circuits to be tested is provided by keying-up the necessary digital information, observing a display of the keyed information, verifying, and then spilling the information forward to the switching machine. This is accomplished by a key operation which results in a trunk test register being attached to the incoming test trunk in the office (see Fig. 6). The register returns a reversal of battery signal which activates a temporary store in the testboard and results in the digits being spilled forward. The register receives these, passes the information to the marker, and a connection is set up. All audio and wideband transmission testing is done from this position and various test sets can be switched into the circuit through the use of associated keys. In addition, a jack-ended arrangement is provided to permit the use of miscellaneous, low frequency, portable test equipment at the position.

#### 4.2.2 *Transmission Test Position—Central Office Communication*

The primary function of the testboard is the testing of the interoffice trunks, PBX trunks, and RSU links. To accomplish this, the testboard has access to the two sides (line link and trunk link) of the No. 5 crossbar network and more digital information is required by the switching machine to service test calls than in the case of the LTD. The information required for handling a test call from a testboard consists of a maximum of 27 digits. The first six digits provide class information which indicates to the switching machine the type of call being made. Digits 7 through 10 identify the particular office trunk to be used in the call. Digits 11 through 13 cover special features such as canceling the continuity check or overriding a make busy. Digits 14 through 17 are used to identify the particular RSU link or PBX trunk to be used in the call if the call is to an RSU or PBX. Digits 18 through 27 provide the destination address. In cases where a digit is not required for making a particular test call, a SK

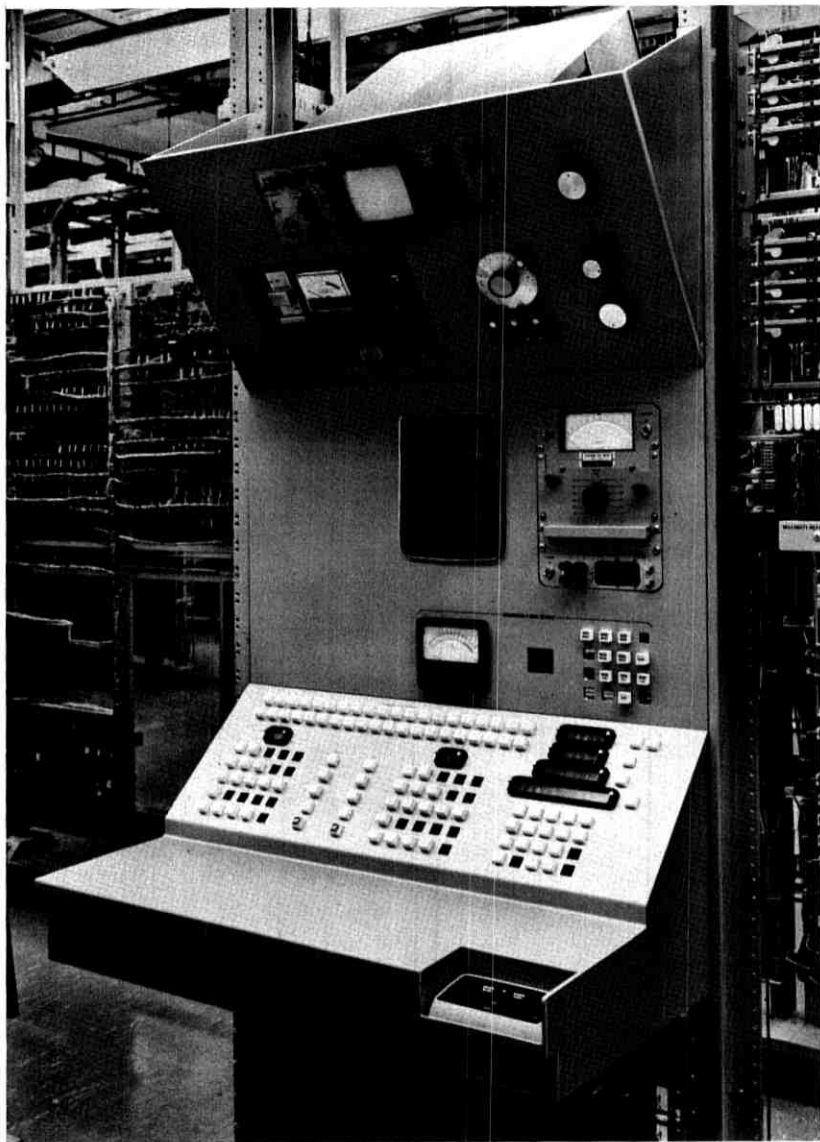


Fig. 5—Testboard No. 23, Transmission Test Position.

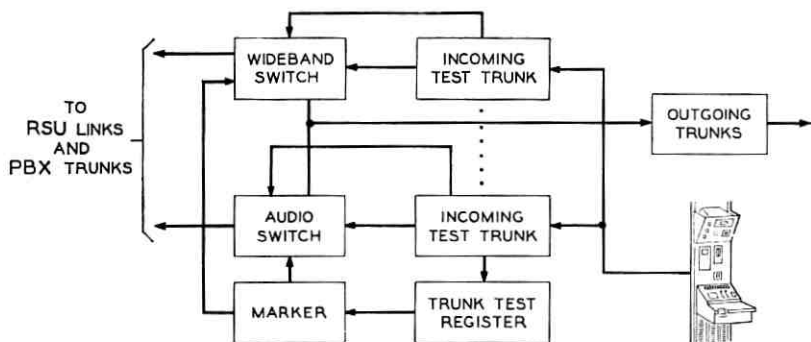


Fig. 6—Picturephone trunk maintenance arrangements.

digit is sent. If, for example, a test call is being made to a distant office, there are no RSU links or PBX trunks involved. In this case, digits 14 through 17 would be SK digits.

Errors are minimized by temporarily storing and displaying the digits before outpulsing. Provision is made to allow partial erasures of groups of digits in the event an error is made. This also reduces the number of digits that needs to be dialed when successive test calls are of a similar nature and where only the address information is changing.

#### 4.2.3 Miscellaneous Test Position

The switched access to transmission pairs in the TTP is possible since these pairs have an appearance on the No. 5 crossbar network. In the case of lines, these pairs are the only access points needed for testing purposes. Trunks, however, are somewhat different for it is necessary to have access to additional points in order to perform signaling tests, remove from service, restore to service, and to obtain individual and group status displays. The access points necessary to provide these functions do not have appearances on the network and consequently another means of providing the access is necessary.

The MTP accomplishes this and provides the required access to the nonswitchable test points necessary for trunk maintenance. Access is provided by directly cabling, on a multiple or loop-through basis, the trunk leads required and by providing a set of dedicated jacks and lamps per trunk. The position is a manual one where access to test points is obtained by plugging cords in the proper jacks. The position is an upright frame type (see Fig. 7) with the same profile as the TTP.

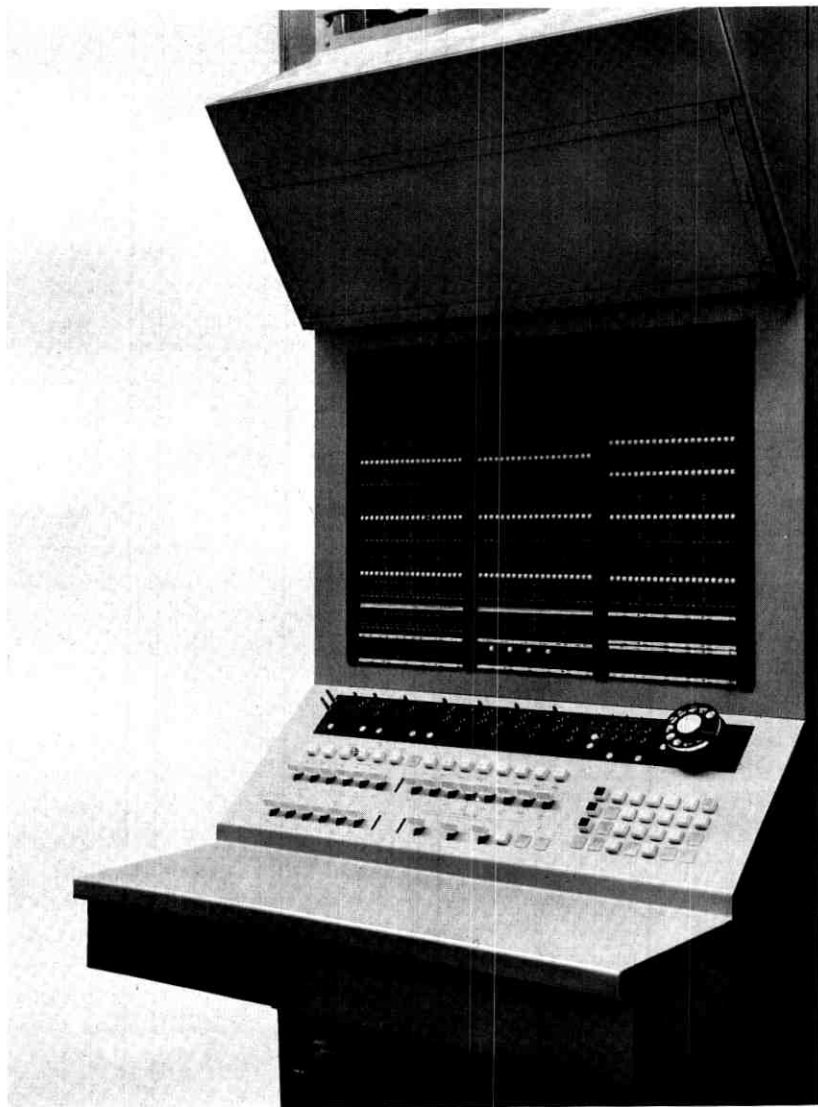


Fig. 7—Testboard No. 23, Miscellaneous Test Position.



## V. WIDEBAND MOBILE TEST BAY

### 5.1 Introduction

In the early years of *Picturephone* service, there will be a need in some offices for an initial "get-started" maintenance arrangement until the standard line and trunk maintenance facilities (LTD No. 15 and TB No. 23) become available. The Wideband Mobile Test Bay (WBMTB) is intended to satisfy this need.

### 5.2 Description

The WBMTB (see Fig. 8) is a roll about type of cabinet containing the same wideband test equipment that is provided in the test desk and testboard. It is designed to provide a convenient, compact housing arrangement for the wideband test sets. The master test frame or other office equipment provides the means for the WBMTB to gain access to lines or trunks.

#### 5.2.1 Wideband Mobile Test Bay Arrangements

The WBMTB will be used in three different arrangements. The first arrangement provides for using the WBMTB in the master test frame (MTF) area. In this case, the WBMTB uses the MTF to gain switched access to a line or trunk. Testing is done by connecting the WBMTB to the circuit by use of a cord. The various test sets are then connected by operating keys associated with the test sets. In some offices this arrangement may be undesirable due to the additional work load or demands on the MTF and also if the maintenance section is an area of high activity.

A second arrangement is provided where the work load is not a problem but where it would be desirable to do the testing at another location. In this arrangement, the MTF still provides the means for accessing the circuits. The circuits, however, are trunked over to an auxiliary test location where the WBMTB is located.

A third arrangement is provided for situations where the additional work load on the MTF is undesirable. In this arrangement the auxiliary test location has access to an incoming test trunk and is equipped with a priming set circuit and the WBMTB. The incoming test trunk is similar to the trunk used with the Testboard No. 23. This trunk is arranged to have access to the trunk test register in the office. The priming set circuit is designed to provide an equivalent means to keying up the digits necessary to access lines and trunks. This is accomplished by providing selector switches rather than a key set and regis-

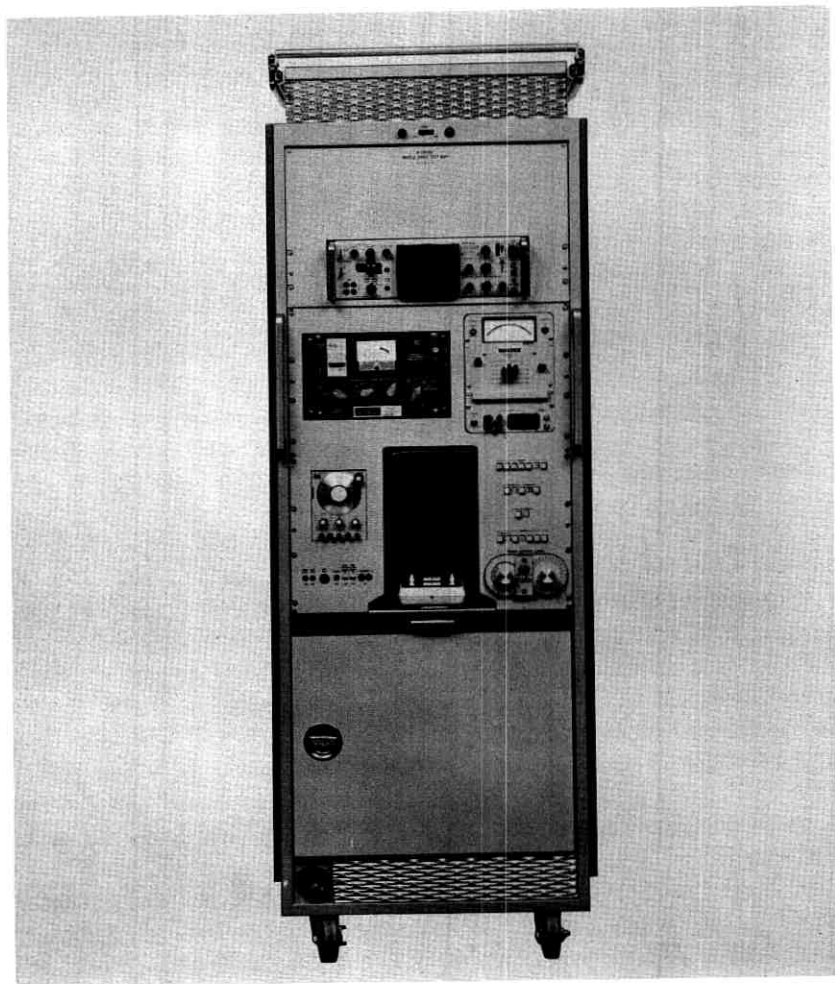


Fig. 8—Wideband mobile testbay

ter. A line or trunk is accessed by setting the priming set circuit switches to the proper digits and operating a start key. The incoming test trunk is then seized and when a register attached indication is returned, the digits are spilled forward on an MF basis and a connection is established. Testing can then proceed.

## VI. TEST EQUIPMENT

### 6.1 Background

It is evident that considerable testing experience will be necessary

before an optimum set of tests or test equipments can be specified. In view of this and the desire to start service as early as possible, general purpose test equipment is specified which is deemed to be sufficient for system maintenance and to obtain the necessary experience. The test sets chosen are:

- (i) A wideband oscillator,
- (ii) A wideband voltmeter,
- (iii) A video waveform oscilloscope,
- (iv) A *Picturephone* test signal generator,
- (v) A *Picturephone* noise measuring set, and
- (vi) A *Picturephone* station set.

## 6.2 Test Sets

### 6.2.1 Wideband Oscillator

The wideband oscillator has a frequency response from 4 Hz to 2 MHz. In the frequency range of 30 Hz to 1 MHz the output is specified to be flat to within  $\pm 0.1$  dB. This permits rapid measurements of transmission characteristics without constant need for readjustment of the output level. The oscillator is capable of delivering +3 dBV into 100 ohms with an output resolution on its level control of less than 0.05 dB. The distortion products in the output signal are less than -40 dB. The frequency accuracy is given as within  $\pm 3$  percent of the dial setting.

### 6.2.2 Wideband Voltmeter

In the main, this instrument is used with the oscillator for gain-frequency measurements. The range of the voltmeter is specified as extending from -60 dBV to +10 dBV. While the signal levels, in general, will not be as low as -60 dBV it may be necessary to measure noise in this range. The upper range has been chosen on the basis of measuring signals on misaligned systems or running overload tests. The level accuracy of the instrument is  $\pm 0.1$  dB in the range from 30 Hz to 600 kHz. Outside of this range the specification is relaxed in accordance with required low end and high end *Picturephone* roll-off characteristic. In addition, a "3 kHz Flat" filter may be selected. The instrument, with this filter, may be used for noise measurements in this frequency range. This will supplement the noise measurements made by the *Picturephone* noise measuring set which has a low frequency cutoff of 4 kHz.

### 6.2.3 Video Waveform Oscilloscope

One can select and display, (with the instrument being used for

waveform observation), either of the interlaced fields. A variable line select control is featured so that any line in a particular field can be observed. A discrete line select position is also available so that the eighth line in a field is displayed. It is intended that this feature be used with a specific signal generated by the *Picturephone* test signal generator (see part (ii) in Section 6.2.4). The amount of tilt observed on this eighth line is a measure of the low-frequency performance of an intervening transmission medium.

The horizontal linearity of the oscilloscope is  $\pm 0.5$  percent of full scale. Multiplication of the horizontal sweep rate (time-scale magnification) by 5 or 25 is also available. In addition, external sync and free-run modes of operation are provided. (The free-run mode is useful for observation of noise or crosstalking signals.)

An internal calibration signal is available for checking the vertical display of this instrument. This calibration signal allows for a stated accuracy of  $\pm 1$  percent of full scale deflection. The required range of input levels is from 0.2 V peak-to-peak to 2.0 V peak-to-peak. An additional 12-dB gain switch is provided. This permits vertical magnification of portions of the input signal. An internal graticule eliminates any parallax error when making measurements on the 8-cm by 10-cm display screen.

Since the video information portion of the *Picturephone* signal is preemphasized, and the sync pulses are not, a response switch has been provided to allow for:

- (i) Viewing sync pulses (with video information still preemphasized).
- (ii) Viewing the deemphasized video (with the sync pulses being distorted because of the deemphasis process).

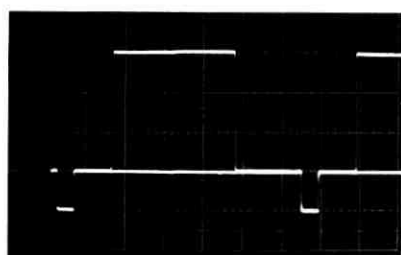
In addition, this switch allows for:

- (iii) Viewing the 512-kHz modulation stripped from a differential gain (modulated stair-step) signal.

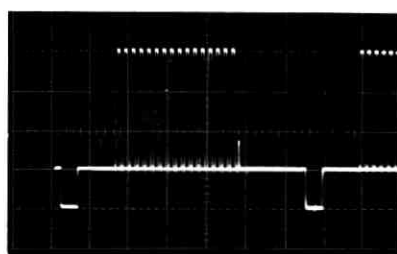
#### 6.2.4 *Picturephone Test Signal Generator*

The *Picturephone* test signal generator has been developed for transmission and CODEC testing. The generator provides seven selectable waveform patterns, some of which are shown in Fig. 9:

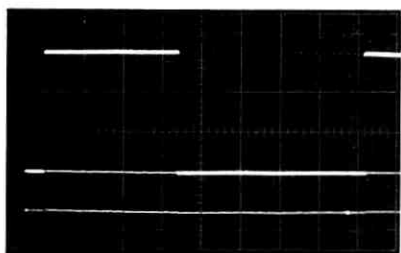
- (i) A white centered window, as it will appear on a station set, is produced by the signal shown in Fig. 9a-1. Such a display makes ringing and overshoots evident. The generator can also modulate this signal with a 256-kHz square wave as indicated by Fig. 9a-2.



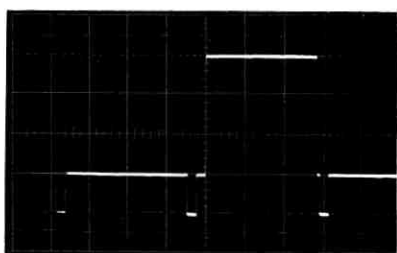
(a-1)



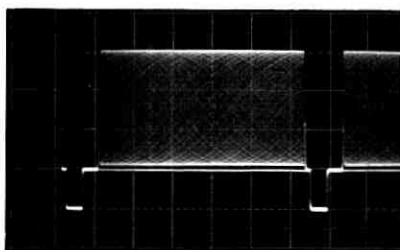
(a-2)



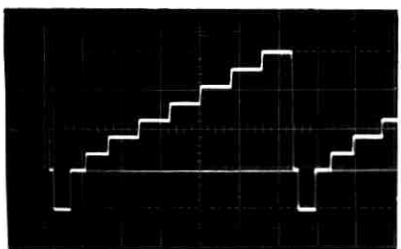
(b)



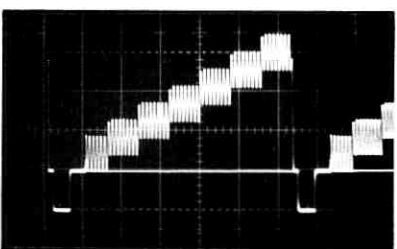
(c)



(d)



(e-1)



(e-2)

Fig. 9—(a-1) One-half white centered. (a-2) One-half white centered with 256-kHz modulation. (b) Half white, half black. (c) Interconnecting Unit test signal. (d) Externally modulated signal. (e-1) Stair step signal. (e-2) Stair step signal with 512-kHz modulation.

- (ii) A field time modulation waveform which will produce a picture on the station set that is half white and half black (horizontal dividing line) is shown in Fig. 9b.
- (iii) A signal which will produce a 4-kHz fundamental component for testing Interconnecting Units. See Fig. 9c.
- (iv) A 256-kHz square wave without sync or blanking is provided for CODEC testing.
- (v) A standard sync signal with the video containing a 1/2 white pedestal. This signal can be modulated by an externally derived signal. Fig. 9d illustrates this pattern with modulation.
- (vi) A seven step-staircase signal with or without a 512-kHz locked sine wave modulation. This signal would provide a grey scale as viewed on a station set. See Figs. 9e-1 and 9e-2.
- (vii) Sync format only.

Any of the above patterns can be transmitted either with or without preemphasis.<sup>4</sup>

One form of the *Picturephone* test signal generator permits it to be operated from a remote position. Such a unit is located on a miscellaneous equipment bay associated with the local test desk and testboard. The selection of patterns and necessary control functions is provided by switches and keys in the test desk and transmission test position.

#### 6.2.5 *Picturephone Noise-Measuring Set*

In another paper<sup>4</sup>, the system requirements are given in terms of percentage time that the noise exceeds a specified level. This can be translated to the number of peaks exceeding a preset threshold by assuming an average effective duration for the noise bursts. This test set counts the number of those noise peaks that exceed a preset threshold level within a preset time interval. It is used also for making background rms noise measurements in the range from 0 to 100 dBrn. The measurement range for impulse noise is from 30 to 100 dBrn in 1-dB steps. Its maximum counting rate is six counts/second on an internally contained mechanical counter. When used with an external electronic counter, it can give indications of 5000 counts/second. It contains a spring-driven timer adjustable from 0 to 15 minutes which shuts off the set at the end of a preset interval. A continuous hold position is also provided. Essentially, this instrument has a bandwidth extending from 4 kHz to 560 kHz. Other shaped bandwidths such as *Picturephone* noise weighting may be selected by means of plug-in filters. The longitudinal balance of this instrument is greater than 70 dB at 25 kHz and greater than 42 dB at 560 kHz. This instrument has an accuracy of  $\pm 0.5$  dB and contains an internal means for calibration.

### 6.2.6 Picturephone *Station Set*

This is a standard 2C video telephone station set used for communication purposes, for confirming trouble conditions, and for observing the effect of transmission impairments on any video test signal.

## VII. TRANSMISSION TEST TERMINATIONS

### 7.1 *Background*

Transmission test terminations are circuits, which upon seizure, assume a particular state, or sequence through a number of states while returning preset signals for each of the states assumed. Test terminations may appear on the line or trunk side of a network. On the line side, seven digits are required to access the termination and they are referred to as test lines. On the trunk side, three digits of a 10X type are generally required to access the termination and they are referred to as test trunks or 10X trunks. The function of a test termination is to provide one-person transmission, signaling, or operational test capability.

### 7.2 *Wideband Transmission Test Terminations*

Three transmission test terminations are being provided for *Picturephone* service:

- (i) The station video test termination,
- (ii) The wideband 100 type test termination, and
- (iii) The wideband 102 type test termination.

#### 7.2.1 *Station Video Test Termination*

The station video test termination permits an installer to check the picture quality and *Picturephone* ringing feature of a station set. The test termination is accessed by dialing a special code from the station set being installed. Upon seizure the test termination transmits both a video and audio signal to the station. These signals are used to check the quality of the loop. A ringing test can be performed by flashing the switch hook and hanging up. Upon answer, the test termination establishes a loop-back condition only on the video pairs. The picture quality can be further observed by the installer viewing himself over this central office loop-back and comparing this with the picture that results when he views himself on a local loop-back in the station set.

#### 7.2.2 *The Wideband 100 Type Test Termination*

The wideband 100 type test termination provides a means for mak-

ing far-to-near noise measurements on the video and audio pairs of trunks. This test termination is accessed by dialing a special code which may be 100 if it is on the trunk side of the network; seven digits if on the line side of the network. In either case, upon seizure the test termination will return a 10-second, 1-kHz,  $-10\text{dBV}$  tone over the video pair. At the end of this time the termination will remove the tone and terminate the audio and video pairs. The test termination will remain in this state while permitting a noise measurement to be made. After an automatic time out, it will return to its idle state with the video pairs looped-back and the audio terminated.

### 7.2.3 *The Wideband 102 Type Test Termination*

The wideband 102 type test termination provides a means for making loss measurements on the video and audio pairs of trunks. This test termination is accessed by dialing a special code which may be 102 if the termination is on the trunk side of the network; seven digits if on the line side of the network. In either case, upon seizure, this test termination will return a 20-second, 1-kHz,  $-10\text{dBV}$  tone over the video pair and a 40-second, 1-kHz, 0 dBm tone over the audio pair. These signals are used for far-to-near loss measurements. At the end of the 20-second interval, the signal on the video pairs is removed and a video loop-back condition is set up for 20 seconds. This permits a loop-back loss measurement to be made on the video pairs. At the end of 40 seconds, the video loop-back is removed along with the tone on the audio pair.

## VIII. SUMMARY

The line and trunk maintenance arrangements for the initial years of *Picturephone* service have been designed to provide a high degree of flexibility. The test equipment is sufficiently general to individually and collectively provide a wide range of test capability. This is particularly important in the early years as experience is gained and improved test methods are evolved.

The switched method of accessing circuits provides a versatile arrangement for testing the audio portion of trunks as well as providing the necessary access to the video portion. Incorporated in this arrangement is the new capability of exercising some control over the switching machine by directing it to cancel or override certain normal machine actions. Overall, the arrangements provide not only wideband test capability but also greater flexibility in the maintenance of lines and trunks.

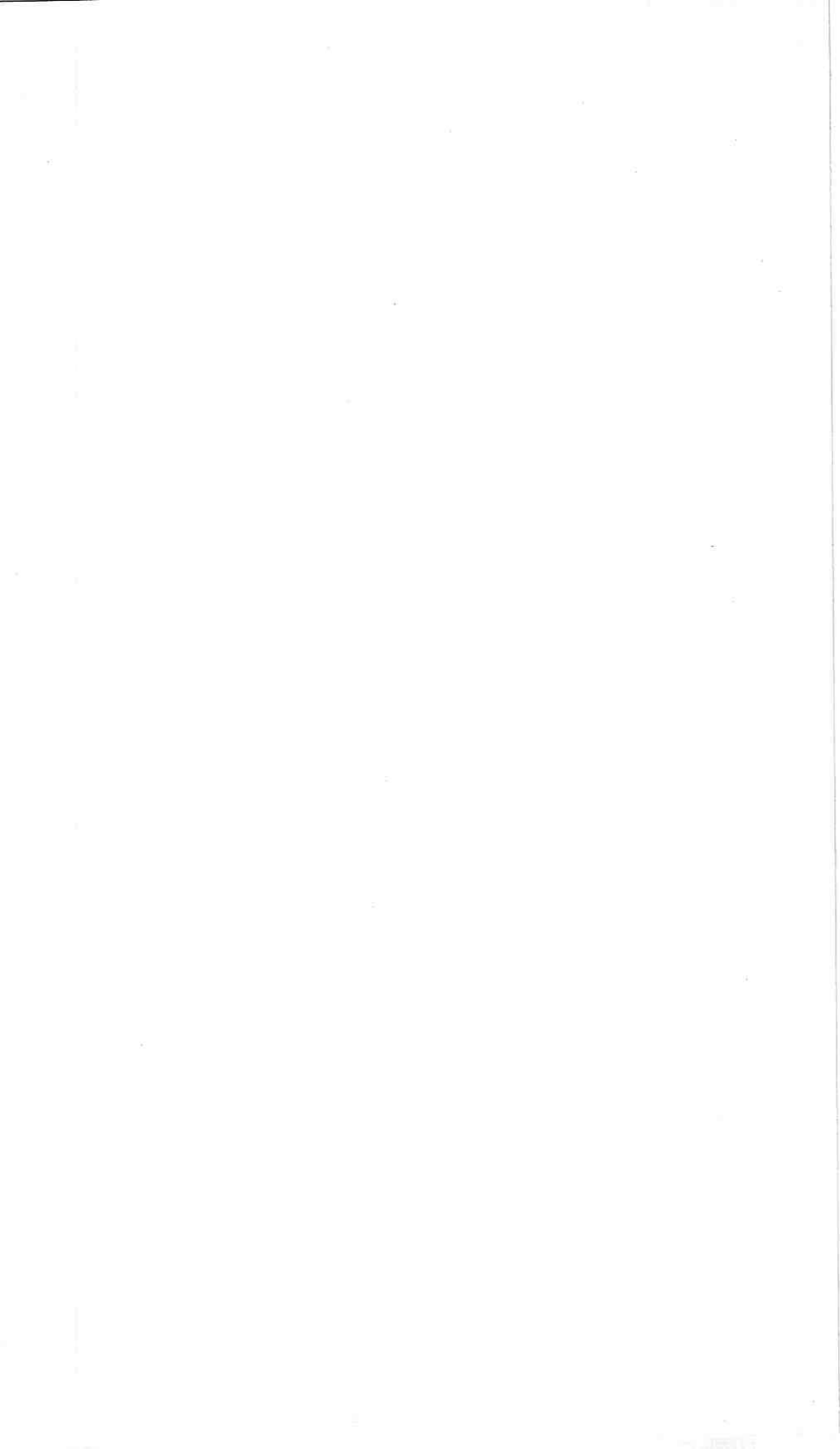


## IX. ACKNOWLEDGMENTS

As in all projects of this magnitude, the final products are the result of the contributions of many people. We would like to acknowledge particularly the efforts of our colleagues at Bell Labs: T. C. Anderson, P. J. Snyder, E. J. Aspell, H. H. Koff, R. E. Yamarone, J. A. Giunta, J. F. Morrongiello, J. E. Kramer, Jr., J. J. Costello and J. C. Willms. In addition, we would like to acknowledge the cooperation and assistance of the Network Planning and Switching Engineering Centers.

## REFERENCES

1. Burgess, P. N., and Stickel, J. E., "The *Picturephone*® System: Central Office Switching," B.S.T.J., this issue, pp. 533-552.
2. Brown, J. M., "The *Picturephone*® System: Baseband Video Transmission on Loops and Short Haul Trunks," B.S.T.J., this issue, pp. 395-425.
3. Dougherty, H. J., Peterson, E. B., and Schachtman, M. G., "The *Picturephone*® System: Maintenance Plan," B.S.T.J., this issue, pp. 621-644.
4. Brown H. E., "The *Picturephone*® System: Transmission Plan," B.S.T.J., this issue, pp. 351-394.



# Data Transmission and Computer Access



# The *Picturephone*<sup>®</sup> System:

## Wideband Data Service

By E. C. BENDER and R. D. HOWSON

(Manuscript received October 9, 1970)

*Use of the evolving Picturephone<sup>®</sup> network to provide common-user, switched wideband data service is another step in the continuing effort to satisfy the growing need for moving information faster, in greater quantities, over longer distances, and at lower cost than is now possible. Full use is made of the planned network with but minor modifications to the basic face-to-face offering. Initial network capabilities permit synchronous data transmission at 460.8 kb/s; it is expected that data service at 1.344 Mb/s will become standard as network improvements are made.*

### I. INTRODUCTION

The rapid growth of data transmission over the past several years has indicated a need for finding more efficient means of transmitting large amounts of data. At present, the majority of data transmission occurs at rates of a few kilobits per second over voicegrade facilities. Modern computer-associated equipment is capable of operating at rates several orders of magnitude greater than this with no significant economic penalty. For data service at such high speeds, wideband private-line facilities are available. Additionally, a common-user switched wideband service, called *Data-Phone 50*, is presently available to provide 50-kb/s transmission capability over a four-city experimental network. Data service using the planned nationwide network of *Picturephone* facilities will provide speed capability an order of magnitude greater than this.

Wideband data service over the *Picturephone* network begins with the offering of a switched 460.8-kb/s capability through use of the initial network of *Picturephone* facilities. Faster service at 1.344 Mb/s is anticipated as network improvements are made within the next few years. It is expected that provision of switched wideband data capability on the *Picturephone* network will be a significant step toward

satisfying the growing need for high-speed data communications. This common-user switched service should be most attractive to users whose community of interest includes many stations spread over a wide geographical area with traffic volume to individual stations not large enough to justify the use of wideband private-line facilities.

## II. INITIAL SERVICE OBJECTIVES

A design objective for initial data service is to make full use of the network with but minor modification to the basic face-to-face service. This must be done while recognizing that although the face-to-face signal occupies a 1-MHz bandwidth, various power, spectral distribution and processing constraints restrict the transmission of arbitrary 1-MHz signals. For data-type signals, these constraints are most severe at the interface between the analog and digital portions of the network.

The equipment used to provide this interface is called a codec.<sup>1</sup> The codec which will be used for initial *Picturephone* service was designed primarily for video service. To accommodate signals the analog interface is adaptively altered, but both data and video signals undergo the same basic encoding process. This process limits wideband data service to rates well below one megabit per second. As the network continues to evolve, an improved codec is expected to handle data rates up to 1.344 Mb/s by adaptively changing the encoding process as well as the analog interface.

Both initial 460.8-kb/s and future 1.344-Mb/s service can be obtained without extensive modification of the planned *Picturephone* network. Many plant items such as analog repeaters, central office switches and digital trunks will be shared between video and data service. The few differences that will be found between the services will be in codec operation, station arrangements, and maintenance procedures.

## III. LIMITATIONS ON DATA RATES

The wideband transmission path in a typical *Picturephone* connection involving digital facilities will consist of an analog loop, switch, and trunk followed by a digital encoder, trunk, and decoder, and completed by the inverse analog trunk, switch, and loop. Several factors limit the bit speeds that can be realized through such a path at reasonable cost. These factors include the capacity of the digital trunks, technical and economic considerations in analog transmission, and the methods used to encode the analog signal into digital form.

### 3.1 *Digital Transmission*

Specifically, the transmission rate of the T2-format bit stream used in the *Picturephone* plant is 6.312 Mb/s.<sup>2</sup> A small percentage of the total bits transmitted must be reserved for control and synchronization of the digital system but the remainder are available to carry message information.

### 3.2 *Analog Transmission*

A fundamental restriction on maximum data rate in analog transmission is limited channel bandwidth. According to theory,<sup>3</sup> the maximum two-level PAM bit rate that can be attained in the 1-MHz video channel without intersymbol interference would be 2 Mb/s. Practically, imperfections in the *Picturephone* loop and trunk transmission characteristics reduce speeds that can be realized without unduly complex instrumentation to the vicinity of 1.5 Mb/s. Simple methods are available to encode data signals at rates up to that speed into the 6-Mb/s digital capacity of T2 bit streams.

Although speed limitations have been discussed above in terms of bit rate, it is basically signaling rate that is limited by restricted bandwidth. Consequently higher bit rates are potentially achievable by transmitting more information per signal pulse; i.e., by using a multilevel signal format. In addition, more efficient use of the available band is possible if partial-response<sup>4</sup> signaling is used.

Unfortunately, both of these techniques for increasing bit rate will also increase the number of signal amplitudes that must be distinguished in order to recover the transmitted data. This, in turn, will reduce the error margin that the signal has against noise and crosstalk interference accumulated during transmission. The margin will be reduced even further by the distortions encountered on the analog facilities.

To overcome the loss of error margin incurred by multilevel signals, it might seem reasonable to increase the transmitted signal power to the point where the error rate is reduced to tolerable levels. However, it is necessary to consider the possibility that high-power data signals will crosstalk into other services. To protect these services (particularly face-to-face video), protective criteria that limit the allowable transmission power for data signals have been formulated.<sup>5</sup>

In Section 4.1, it is shown that the interference, distortions, and power limitations that are expected on *Picturephone* analog loops and trunks will limit economically feasible transmission to binary PAM with a maximum rate near 1.5 Mb/s.

### 3.3 Digital Encoding and Decoding

The objective of maximum alternate use of *Picturephone* facilities leads to the procedure of quantizing, sampling, and pulse-code modulating the analog data signal just as if it were a video signal. It will be shown that this procedure is the principal factor in the choice of 460.8 kb/s for the initial-service bit rate.

## IV. DATA CAPABILITIES OF THE *Picturephone* NETWORK

To determine the exact capabilities for data transmission over the *Picturephone* network, the various components that form the network will be considered separately. In order of discussion, these components are analog loops and trunks, digital facilities, and analog switches.

### 4.1 Analog Loops and Trunks

The loops and trunks have been discussed in detail in a preceding paper;<sup>6</sup> only the pertinent aspects will be reiterated here when needed.

The major source of interference to data signals in the analog plant is impulse noise; both system allocations and actual transmission tests indicate substantial operating margin against thermal noise and crosstalk. Quantitatively, the maximum impulse noise level can be estimated from the objective<sup>7</sup> for face-to-face *Picturephone* service. Measured on the equalized facilities, this objective is that the probability of base-to-peak weighted noise exceeding 18 millivolts be less than or equal to  $1.5 \times 10^{-5}$  [ $P(N > 18 \text{ mV}) \leq 1.5 \times 10^{-5}$ ]. For convenience, the voltage threshold of 18 mV can be translated to a dBV threshold of -35 dBV. Then, noting that the face-to-face objective is a weighted objective, a 6-dB modification to -29 dBV is made since data service is not affected by subjective weighting, and will not use pre- and de-emphasis. Impulse noise distributions found on *Picturephone* facilities are expected to display a slope of 10 to 12 dB per decade change in probability. Applying a 12-dB slope to the -29 dBV objective leads to the upper limit for system noise shown in Fig. 1.

To determine maximum allowable data powers, reference must be made to the protective criteria for interconnection.<sup>8</sup> One criterion concerning power present in narrow bands is met with ample margin by random data signals, but a second criterion for total power in specified wider bands places severe restrictions on signal power. Calculations for the maximum reasonable binary data rate of 1.5 Mb/s indicate allowable transmission levels of 2.5 dBm. In terms of peak voltage across the 100 $\Omega$  line this level is -7.5 dBV.



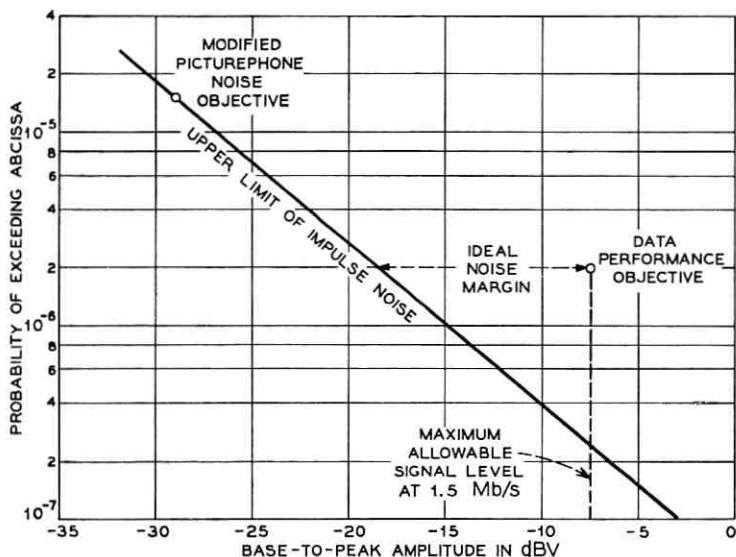


Fig. 1—Ideal data performance relative to impulse noise objective for analog loops and trunks.

The error performance of this signal can now be related to the impulse noise objective discussed previously. To meet the usual wideband data error rate objective of  $10^{-6}$  for a peak signal level of  $-7.5$  dBV, it is required that the probability of noise greater than  $-7.5$  dB be less than or equal to  $2 \times 10^{-6}$  [ $P(N > -7.5 \text{ dBV}) \leq 2 \times 10^{-6}$ ]. The factor of two arises from the 50 percent probability that the noise and signal are of opposite polarity. As shown in Fig. 1, the data performance objective is met with 11 dB of margin.

So far the discussion has been concerned with the effect of noise on ideal signals. In practice, however, even with the in-band gain well equalized, two significant sources of signal distortion remain. The first of these is deviation of the net phase characteristic from an ideal linear characteristic, and the second is deviation of transmission characteristics due to the lack of temperature regulation on the loops and trunks for initial service.

To investigate the effect of distortion on data signals, 1.344-Mb/s data signals were tested on simulated facilities. The results are illustrated by the "eye" patterns\* of Fig. 2. The pattern of Fig. 2a was

\* An "eye" pattern is a bit-synchronized superposition of possible data signals.

taken with the best data set connected back-to-back and is, therefore, the reference pattern. Expected worst-case phase distortion was simulated by 42 kft of 22 AWG cable maintained at constant temperature and gain-equalized by repeaters spaced at 6-kft intervals. The distorted "eye" is shown in Fig. 2b. A Bode Equalizer<sup>8</sup> was used to simulate worst-case distortion due to temperature variations. The equalizer shape was a compromise between the regulation characteristics of 20 kft of 24 and 16 kft of 26 AWG cable; both of these have +8 dB of gain at the 1-MHz band edge corresponding to a 50°F temperature differential. The result is pictured in Fig. 2c. Measurements of the "eye" patterns at the sampling points show an effective loss of margin against noise relative to an ideal undistorted signal of 0.6 dB, 1.5 dB, and 1.5 dB for back-to-back, phase distortion, and temperature distortion respectively. The phase and temperature distortion conditions used in these tests are extreme. It is not unreasonable, therefore, to allot 3 dB of margin for worst-case facility distortions and data set tolerances.

It is anticipated that second-generation cable equalizers will include both temperature regulation and improved phase control, but the extra margin derived therefrom will be allocated for the extension of loop and trunk length. Therefore, no appreciable improvement of analog transmission performance should be expected from second-generation facilities.

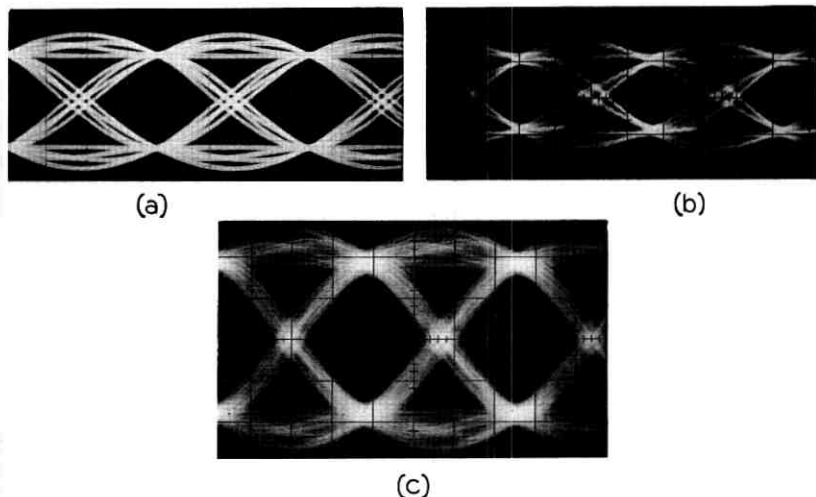


Fig. 2—Eye patterns for a 1.344-Mb/s data signal transmitted: (a) back-to-back, (b) through a phase distorted channel, and (c) through a temperature distorted channel.

The effect of distortion can be related to the performance curve of Fig. 1. The ideal signal has an 11-dB margin against impulse noise. If 3 dB of this is allotted to facility and data set distortions, 8 dB of margin remains to be distributed between gain misalignment, thermal noise, and crosstalk interference.

From the preceding considerations, it can be concluded that data transmission at rates up to 1.5 Mb/s is feasible on the *Picturephone* analog loops and trunks. The transmission levels will cause neither excessive interference into other services nor excessive susceptibility to impulse noise. However, margins are so small that transmission at higher bit rates appears unlikely without appreciable increase in data set complexity.

Before closing this section, two additional points should be considered. First, it will be shown in the following section that the initial-service codec will not support data rates much greater than 500 kb/s and signals encountering digital trunks will be subject to codec jitter and quantizing noise. Furthermore, network protection criteria impose more severe power limitations on signals in this speed range because their energy is concentrated in the region of the transmission band where the risk of crosstalk interference to other services is greatest. Fortunately, cable equalization tends to concentrate interference at the high end of the band; the amplification needed to offset cable rolloff enhances the high-frequency components of noise and crosstalk. This tends to make the signal and interference spectra disjoint; the net result is expected to be performance at least as good as that at 1.5 Mb/s. The expectation has been borne out by preliminary testing on *Picturephone* trial facilities.

The second point concerns low-frequency transmission characteristics. To allow for dc potential differences between repeaters, the transmission path must be ac-coupled. The cutoff is in the vicinity of 1 Hz; the associated time constant is on the order of 160 ms. With the 20-stage scrambler/descrambler normally included in wideband data sets,<sup>9</sup> the probability that more than 20 consecutive symbols of the same polarity will be transmitted is negligible. At a 500 kb/s rate, this corresponds to only 40  $\mu$ s without a signal transition. It is clear, therefore, that low-frequency cutoff will have no appreciable effect on *Picturephone* data transmission.

#### 4.2 Digital Facilities

As mentioned in companion articles,<sup>7, 10</sup> digital facilities will carry the bulk of *Picturephone* toll transmission. The basic building block of the digital facilities is the 6.312 Mb/s differential pulse code modu-

lated (DPCM) signal derived from the A/D operation in the codec. This bit stream may be transmitted by itself over T2 facilities or multiplexed with other bit streams for transmission over radio or coaxial carrier systems.<sup>11, 12</sup>

#### 4.2.1 Codec Operation

The interface between the digital and analog facilities is provided by the codec which uses differential encoding to transform the nominal 1-MHz analog signal to a 6.312 Mb/s PCM signal.<sup>1</sup> Because of basic format differences between video and data signals, it is necessary that the codec be able to operate in two distinct modes. This is achieved by sensing the presence or absence of video sync signals to determine whether the input signal is video or data.

For data signals the codec operates as a simple nonsynchronous differential encoder. The input signal is brought to a nominal level, 0.72 volts peak-to-peak, by a  $\pm 6$  dB AGC which compensates for all but catastrophic gain misalignment in the analog plant. This signal is then sampled at a 2.016 MHz rate with each sample differentially encoded into one of eight signal levels ( $\pm 12$ ,  $\pm 36$ ,  $\pm 84$ ,  $\pm 180$  millivolts relative to the input signal level). The resultant levels are then 3-bit encoded; framing, stuffing and signaling bits are added; and a 6.312-Mb/s PCM signal is delivered to the digital facility. At the far end, the opposite procedure is used to recover a nominal-level analog signal to be sent to the appropriate end section. This is illustrated in Fig. 3 which compares a typical data signal as seen at the input of the near-end codec, Fig. 3a, to the resulting signal at the output of the far-end codec, Fig. 3b.

From Fig. 3 one can observe the distortion caused by three types of impairment introduced by the codec: jitter, slope overload, and quantizing noise. Jitter is introduced into the signal by the nonsynchro-

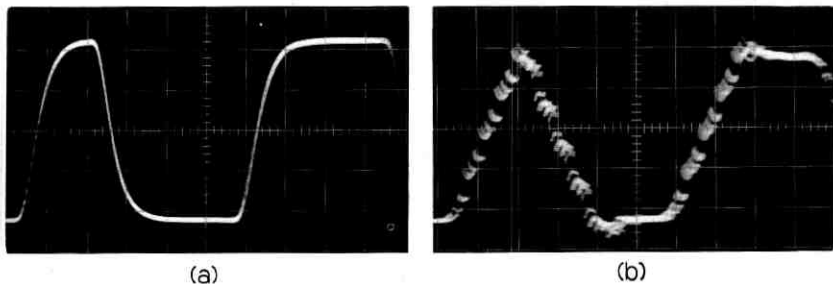


Fig. 3—A typical 460.8-kb/s data signal as found: (a) at the input to a near-end codec, and (b) at the output of a far-end codec.

nous method of sampling used in the A/D conversion. The amount of peak-to-peak jitter, equal to the reciprocal of the sampling rate, is approximately  $0.5 \mu\text{s}$ . It is usually desirable to keep the total range of jitter in a data signal less than 50 percent of a bit period to ensure detection of the signal near its peak. Therefore, codec jitter can be thought of as imposing an upper limit of one half the sampling rate, or 1.008 Mb/s, on binary data transmission.

Slope overload considerations also limit data signals in the following manner. The use of differential encoding with a maximum step size of  $\pm 180$  millivolts means that the codec output signal cannot rise at a rate faster than  $180 \times 2.016$  or about 360 mV per  $\mu\text{s}$ . Thus, for high-level or high-speed signals, the output will not be able to track the input accurately. For a rectangular pulse at the input, the point at which the output signal can no longer rise to the peak of the input signal in one pulse interval is referred to as the overload point. This is illustrated in Fig. 4 where it is seen that the output, bottom trace, cannot rise to the peak value in a single bit interval. However, if the input remains constant over several bit intervals, the output can attain the peak amplitude of the input. Figure 4 then illustrates operation beyond or above the overload point. Operation below the overload point requires the number of samples per bit interval to be greater than the signal voltage divided by 180 mV. For the planned 0.72-volt data signal, four samples are required per bit interval, thus placing an upper limit of 504 kb/s on the data speed.

In addition to jitter and slope overload impairments, the codec, through the process of A/D conversion, introduces distortion known as quantizing noise. For worst-case bit combinations at the input, codec operation is just below the overload point such that maximum quantizing steps are used continuously. Since the signal is simultaneously undergoing jitter, argument can be made that the worst-case quantizing noise found on data signals is 180 mV or 25 percent of the peak data signal. This analysis must, of course, be tempered by far-end loop and data set considerations. The analog loop will add noise to the signal and the data set will contain a low-pass receive filter with the result that the quantization noise will become hidden in other impairments.

From the above discussion, it is apparent that data transmission above one megabit through initial codecs suffers from a large amount of jitter and a low overload point. Since transmission should be satisfactory at rates below 500 kb/s, initial service will be offered at 460.8 kb/s, a rate that is compatible with present private line offerings.

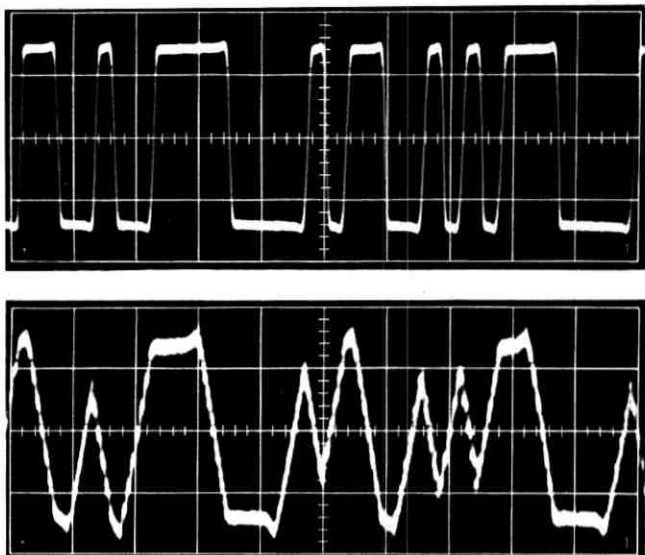


Fig. 4—A 460.8-kb/s data signal operating beyond the overload point of the codec. Top trace is a high-level input which produces the overloaded output seen in the bottom trace.

#### 4.2.2 *Transmission Facilities*

Having been transformed by a codec into a 6.312 Mb/s DPCM bit stream, a data signal is indistinguishable from a similarly encoded video signal. In either case, the DPCM signal can be transmitted to the far-end codec via any one or any combination of digital transmission facilities. The DPCM signal can occupy a complete T2 line or be multiplexed through a M2R onto the digital channels carried on TD-2 radio.<sup>11</sup>

L-Mastergroup Digital terminal equipment (LMD) multiplexes two 6.312-Mb/s bit streams onto mastergroups on L-4 coaxial systems.<sup>12</sup> Future higher capacity digital and analog systems will also include the necessary multiplexers to enable compatibility with the T2 format. Whatever the facility, the digital bit stream is expected to contribute less than 10 percent of the overall data error rate, so that, for most practical cases, the codecs may be thought of as being back-to-back. That is, any distortion introduced into a data signal by a digital facility will in all likelihood arise within the codec and be independent of distance.<sup>13</sup>

#### 4.3 *Analog Switches*

As has been described in a preceding paper,<sup>14</sup> wideband video switch-

ing on the *Picturephone* network will be realized by four-wire video switches slaved to the associated voice-pair No. 5 crossbar switches. Questions on the capability of these switches for handling wideband data services in the megabit range are: (i) what effect in terms of crosstalk would wideband data services have on video-telephone services sharing the same switch, (ii) to what extent are wideband data signals impaired by switch characteristics, e.g., reflections from stub multiples, mismatch, etc., and (iii) what is the effect of office switching noise on data?

To answer the first two equations, data transmission tests were conducted on a laboratory model No. 5 crossbar *Picturephone* switch. Crosstalk investigations included measurement of near-end crosstalk (NEXT), far-end crosstalk (FEXT), and measurements in which both types of coupling paths contributed. The transmission performance of the switch was determined by "eye" pattern observations of a 1.344 Mb/s data signal for one passage through the switch from wideband-line link frame (WBLL) to wideband-trunk link frame (WBTL), as well as for a loopback connection on the WBTL side of the switch. The question of impulse noise could not be answered by direct measurement; the model switch is not in a real central office environment and field installations were not available at the time of the tests.

Based on the measurements made on the laboratory switch model, it appears that dual use of the *Picturephone* switching plant for wideband data will present no difficulties either in terms of data interference into video, or in the quality of transmission through the switch. With crosstalk power weighted by the *Picturephone* noise weighting and de-emphasis curves, the worst-case crosstalk interference from a data disturber was found to be  $-83$  dBV (RMS). This is well below the mean random noise objective of  $-75$  dBV.<sup>7</sup> Observations of the data signal "eye" revealed no significant transmission impairment.

Without suitable test facilities, the effect of impulse noise can be estimated only on the basis of allocation. Since the switching office allocation is included in the overall impulse noise objective, it has already been taken into account in the discussion of analog loops and trunks (Section 4.1).

## V. INITIAL SERVICE

In the preceding section, it has been shown that the initial network of *Picturephone* facilities will not support data speeds greater than 500 kb/s. Therefore, an initial offering of 460.8-kb/s service was

chosen for compatibility with private line services presently being offered through use of T1 carrier facilities. Other possible speeds may arise from customer provided data sets that comply with network protection criteria.

Modifications of the analog video plant are not necessary to provide data service. A loop conditioned for video transmission can be used by either service. This applies even when the customer uses the network for data transmission without subscribing to *Picturephone* face-to-face service.

The terminal equipment that currently provides short-haul 460.8 kb/s service on T1 lines is the Data Station 303 that includes a Data Set 303J25, a T1 Wideband Modem, and a T1 Line Terminating Unit. For initial *Picturephone* service, a new version of this data station has been developed. The changes that have been made include the replacing of the T1-oriented equipment by a 114A Interconnecting Unit and modifying the data set (recoded as the 303J26) as required for interconnection with the 114A. In addition, the telephone handset associated with a video station is replaced by an 804 Data Auxiliary Set (DAS) that provides data-oriented line control functions as well as handset and dial. The complete data station for *Picturephone* network application is pictured in Fig. 5.

Call set-up for data will proceed in the same manner as for video. The call will be originated from the 804 DAS by dialing the desired number preceded by the # prefix to indicate that a wideband channel is required. As the call progresses, maintenance-loopback testing and removal will take place in normal fashion. When the connection is completed, either by manual or automatic answer at the called end, the wideband circuit is ready to accept the data signal.

In addition to the line-continuity testing by the *Picturephone* Maintenance Loopback feature, the data station will have the capability for both remote and local data testing. As in existing data services, this capability will be provided by the line and test unit 806 DAS operating in conjunction with the 804 DAS. The specific tests involved permit isolation of troubles to the transmission facility, the data station, or the customer's equipment by looping the signal at interface points and monitoring transmission quality. The tests are originated and monitored remotely from a Data Test Bay in a serving central office.

## VI. SUMMARY AND CONCLUSIONS

It has been recognized that the system of *Picturephone* facilities



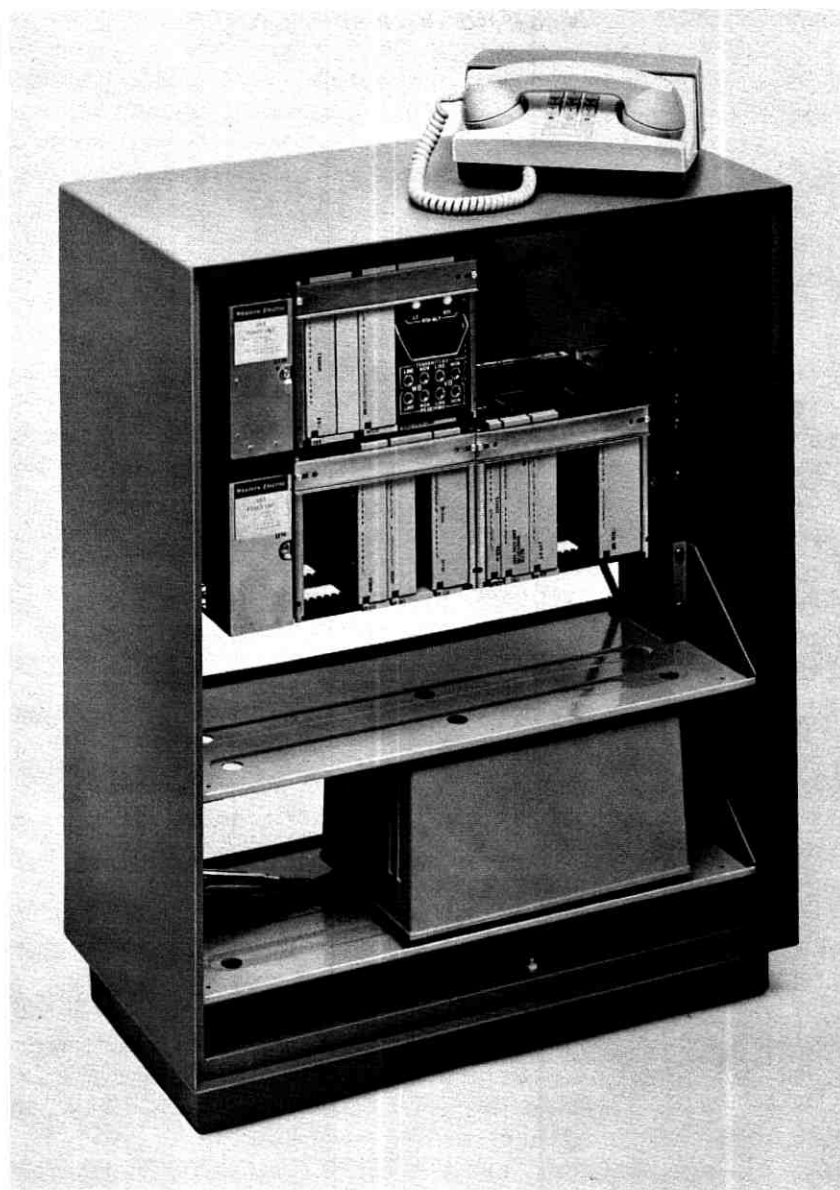


Fig. 5—Typical Data Station 303 used for 460.8-kb/s data transmission on the *Picturephone* network. Lower shelves provide space for the VLTU (shown in position), and optional equipment such as an automatic calling unit.

provides a high-capacity, common-user switched network which can help satisfy the growing demand for wideband data transmission. Because of the objective that network facilities be used without alteration wherever possible, data transmission over the network is now limited to rates below 500 kb/s. A rate of 460.8 kb/s, compatible with service now being provided over T1 carrier facilities, has therefore been chosen for the initial data offering on the *Picturephone* network.

The upper bound on data speed is largely a result of limitations arising from the particular encoding process used to transform the local analog signal to digital form for long-haul transmission. However, as the network evolves, it is anticipated that alternate use codecs will enable data transmission at rates above 1 Mb/s. Since investigation has shown that the analog portion of the network could support rates up to 1.5 Mb/s, it is expected that the next generation of wideband data service over the *Picturephone* network will be at the rate of 1.344 Mb/s consistent with service now being introduced for use on T1 facilities. This expectation is contingent on possible video bandwidth compression techniques that may place additional limitations on attainable data rates.

#### VII. ACKNOWLEDGMENTS

The authors wish to acknowledge the technical assistance and suggestions contributed by many of their colleagues in both the data transmission and *Picturephone* service areas. Particular thanks go to R. S. Libenschek, who played a major role in studies leading to the offering of data service on the *Picturephone* network.

#### REFERENCES

1. Millard, J. B., and Maunsell, H. I. G., "The *Picturephone*® System: Digital Encoding of the Video Signal," B.S.T.J., this issue, pp. 459-479.
2. Davis, J. H., "T2: A 6.3 Mb/s Digital Repeated Line," IEEE International Conference on Communications, Boulder, Colo., pp. 34.9-34.16, June, 1969.
3. Nyquist, H., "Certain Topics in Telegraph Transmission Theory," Trans. AIEE, 47, No. 2 (April 1928), pp. 617-644.
4. Kretzmer, E. R., "Generalization of a Technique for Binary Data Communication," IEEE Trans. on Commun. Technology, COM-14, No. 1 (February 1966), pp. 67-68.
5. Bell System *Picturephone* and Voice Communications Technical Reference, "Picturephone® Connecting Arrangement PVF," American Telephone and Telegraph Company, September, 1970.
6. Brown, J. M., "The *Picturephone*® System: Baseband Video Transmission on Loops and Short-Haul Trunks," B.S.T.J., this issue, pp. 395-425.
7. Brown, H. E., "The *Picturephone*® System: Transmission Plan," B.S.T.J., this issue, pp. 351-394.

8. Bode, H. W., "Variable Equalizers," B.S.T.J., 17, No. 2 (April 1938), pp. 229-244.
9. Fracassi, R. D., and Savage, J. E., "Data Scrambler," U. S. Patent No. 3-515-805, issued June 2, 1970.
10. Dorros, I., "The *Picturephone*® System: The Network," B.S.T.J., this issue, pp. 221-233.
11. Broderick, C. W., "The *Picturephone*® System: A Digital Transmission System for TD-2 Radio," B.S.T.J., this issue, pp. 481-499.
12. Gunn, J. G., Weller, D. C., and Ronne, J. S., "The *Picturephone*® System: Mastergroup Digital Transmission on Modern Coaxial Systems," B.S.T.J., this issue, pp. 501-520.
13. Oliver, B. M., Pierce, J. R., and Shannon, C. E., "The Philosophy of PCM," Proc. I.R.E., 36 (1948), pp. 1324-1331.
14. Urich, J. F., "The *Picturephone*® System: Switching Plan," B.S.T.J., this issue, pp. 521-531.



# The Picturephone<sup>®</sup> System:

## Computer Access

By P. S. WARWICK and G. W. PHIPPS

(Manuscript received September 17, 1970)

*The ability to use a Picturephone<sup>®</sup> station set to display computer generated information will be provided as an additional feature of Picturephone service. The basic element in providing computer-access service is the display data set, which acts as the interface between the computer and the Picturephone station, converting Touch-Tone<sup>®</sup> multifrequency signals from the Picturephone station into digital characters for the computer and converting the computer output into a video signal for the station. The operation of the display data set and the influence of the Picturephone network characteristics on its design are described in this article.*

### I. INTRODUCTION

In addition to face-to-face communication, *Picturephone* stations may be used to access computers. A *Picturephone* user can send information to the computer using his station's *Touch-Tone* dial, and messages from the computer can be displayed on the *Picturephone* screen. Since the computer-access capability is an alternate use of *Picturephone* service, and the objective is that it not interfere with nor increase the cost of face-to-face communication, the additional hardware required has been designed so that no modifications to the station or network are necessary.

### II. COMPUTER-ACCESS SYSTEM

#### 2.1 Description

The basic element of computer-access service is a display data set, which acts as an interface between the *Picturephone* network and a digital computer-communications system as shown in Fig. 1. To the *Picturephone* network, the display data set appears to be a standard *Picturephone* station. To the computer the display data set presents an interface like those used in standard voiceband data services.

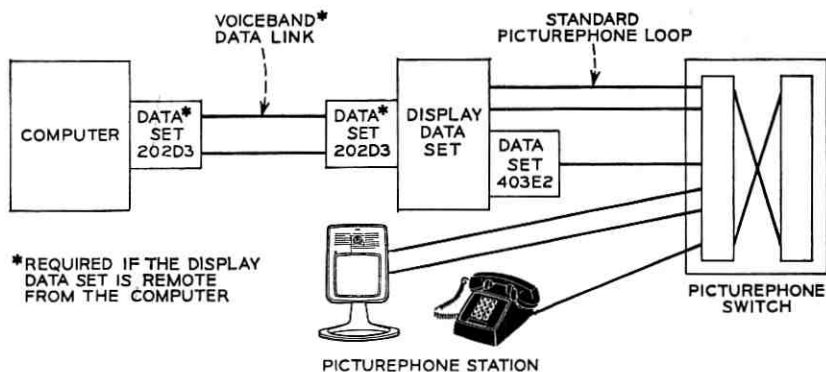


Fig. 1—Picturephone computer-access system.

The essential functions of the display data set are to convert *Touch-Tone* signals received from the station to ASCII\* characters and transmit them to the computer, to store ASCII characters from the computer, and to convert them to the video signals required to refresh the display at the station.

Since the display data set is connected to the *Picturephone* network like a standard station, it must be located in a *Picturephone* service area. However, there are two alternatives for connecting the display data set to the computer. If the display data set is located at the computer site, a direct cable connection is made to the computer. If the display data set is located remote from the computer, a dedicated four-wire voiceband data link is used to connect the display data set and computer; thus, the computer can be located outside of the *Picturephone* service area. The economic decision affecting display data set location is governed by the cost trade-off between switched video facilities and dedicated voiceband facilities. The computer communicates with a distant display data set in ASCII asynchronously at a 1200-baud rate. If the computer and display data set are directly connected, communication is possible at either 1200 or 2400 bauds.

The display data set is connected over a standard *Picturephone* loop to a central office, private branch exchange, or key telephone system. On switched networks, it can be tested remotely from a test desk in the same manner as a *Picturephone* station.<sup>1</sup> A loopback feature in the display data set allows a test deskman to send *Touch-Tone* signals to the set and see them on the screen of his *Picturephone* set; this

\* American National Standard Code for Information Interchange (ANS 3.4-1968).



Figures 3 and 4 show sample displays on a *Picturephone* set. The display data set can generate 60 alphanumeric characters and 16 graph characters; these are shown in Fig. 5. The graticule is displayed whenever any graph characters are shown.

For editing displays, the user has the option of using an alphanumeric keyboard and a frequency shift keying data set (such as the Bell System 113A Data Set) in conjunction with his *Picturephone* station. When this is done, 110-baud serial ASCII signals from the station are received by the Bell System 108A Data Set<sup>3</sup> (See Fig. 2) and speed-converted to 1200 or 2400 bauds before being sent to the computer. With the display data set in the edit mode, an alphanumeric keyboard can be used to modify the contents of the refresher memory. A movable blinking cursor, which marks the location of the next character to be written in the memory, is displayed in the edit mode to aid the user in altering displays. While in the edit mode, characters received from the keyboard are used only to update the display data

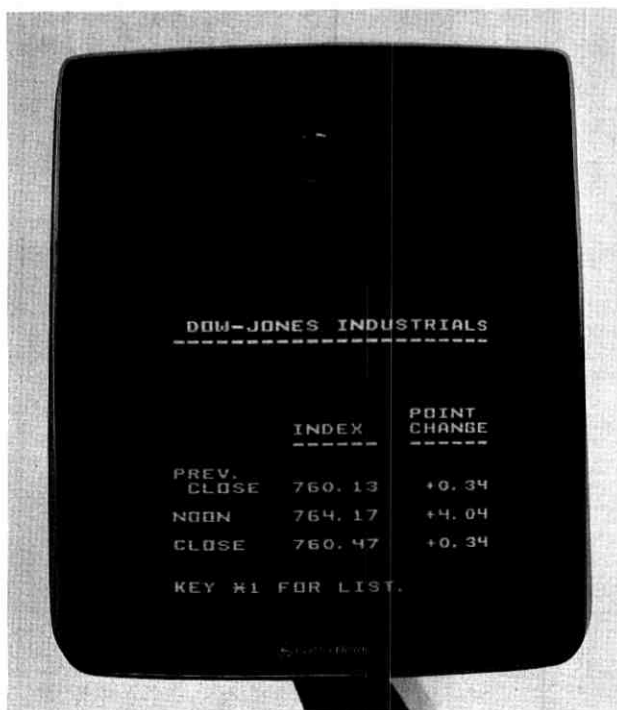


Fig. 3—Typical alphanumeric display.



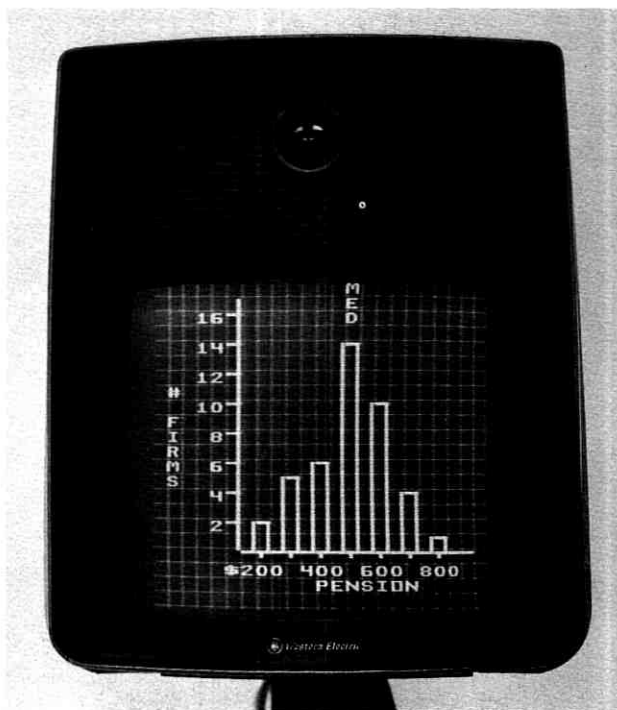


Fig. 4—Typical graphic display.

set memory; they are not sent to the computer. When the user is satisfied with the display, he can request the display data set to transmit the contents of the refresher memory to the computer for storage or processing. This feature allows users to retrieve files from the computer, to modify or update these files, and to send updated files back to the computer.

During the call, the display data set continues to perform its conversion functions, handling inputs and computer responses as required. The information retrieval and input capabilities available to a user of *Picturephone* computer-access service are largely a function of the customer-provided programs. The display data set has been designed not to interfere with or react to most signals sent from the user to the computer, leaving most decisions affecting message formats and interactive procedures to the customer and program designer. Further, the display data set operation is generally under computer control; that is, except when the editing feature is enabled,



5) the maximum number of black-white pairs, including the space between characters, was limited to three per character. Thus, the minimum length of a character space is  $3.75 \mu\text{s}$ . This results in a maximum of 22 characters being displayable, as the visible portion of a horizontal scan line is approximately  $85 \mu\text{s}$ , taking into account worst-case *Picturephone* parameters and allowing space for margins on the sides of the display.

The alphanumeric symbols in the display set are constructed on the basis of  $0.25 \mu\text{s}$  subsegments of a scan line, with each white segment having at least two subsegments and each black segment having at least three subsegments. This allows more flexibility in character design than if a larger subsegment length had been chosen. A character has a 12-subsegment horizontal dimension, and an intercharacter spacing of three subsegments is provided for a total of 15 subsegments in a character space. The resulting character widths are a maximum of 0.165 inches (0.42 cm). Intercharacter spacing is 0.042 inches (0.11 cm) to 0.07 inches (0.17 cm) depending on the particular characters.

The maximum number of lines of characters is determined by the number of scan lines available. Because the *Picturephone* screen is not exactly rectangular, and accounting for worst-case parameters and margins on the top and bottom of the display, only about 207 scan lines will always have the full  $85\text{-}\mu\text{s}$  sweep visible. Based on empirical studies, seven scan lines were chosen as the vertical dimension for the character set. Allowing an interline spacing of three scan lines brought the total to ten scan lines per line of characters. Therefore, 20 lines of text can be displayed. Character heights are thus 0.14 inches (0.36 cm) with an interline spacing of 0.06 inches (0.15 cm).

The sixteen graph characters shown in Fig. 5 may be used to display simple curves and bar charts. In order to provide good point spacing for graphs and continuity when creating bar charts, these characters were chosen to be ten scan lines high and 14 subsegments wide on the horizontal dimension. (The graticule occupies the 15th subsegment to provide horizontal continuity.)

As mentioned above, the character shapes shown in Fig. 5 were determined empirically, although some changes have been made in their designs based on experimental use. Subjective experiments are being run to compare the legibility of the present font with alternatives, for example, a font with larger characters and increased interline spacing, but with fewer possible characters per display. The effects of transmission impairments and digital encoding and decoding on the different fonts are also being evaluated.

### 3.2 Signal Shaping and Spectrum

The minimum white scan line segment of  $0.5 \mu\text{s}$  used to form characters has a fundamental component at 1 MHz. This is at the upper end of the *Picturephone* system bandwidth (station-to-station) which is down 20 dB at 1 MHz. The long horizontal parts of letters, such as the top of the letter E, are  $2.5 \mu\text{s}$  in width with a fundamental component at 200 kHz. This results in the horizontal parts of letters being much brighter than the narrow vertical parts and greatly increases the apparent flicker of displays. To make letters evenly bright, the first and last subsegments of a white interval on a scan line are maintained at full amplitude while the middle white subsegments are reduced some 20 percent in amplitude. Figure 6 shows the effect of this shaping on a signal.

The sharp black-to-white and white-to-black transitions inherent in computer-generated displays and the regularity of these transitions due to character spacing can result in an on-line signal with a frequency spectrum quite unlike a typical face-to-face signal. Figure 7a shows the power spectrum for a display of text. (The sentence "Now is the time for all good men to come to the aid of the party" was repeated to fill 20 lines of 22 characters.) The spectrum of a face-to-face signal is shown in Fig. 7b for comparison. As shown, certain harmonics of the horizontal sync rate are emphasized by the synchronous nature of the alphanumeric display. These harmonics are located at  $1/3.75 \mu\text{s}$ ,  $2/3.75 \mu\text{s}$ , etc., where  $3.75 \mu\text{s}$  is the horizontal character spacing. The amplitude weighting of the  $n/3.75 \mu\text{s}$  spectral lines is different for different displays. For example, if a page full of

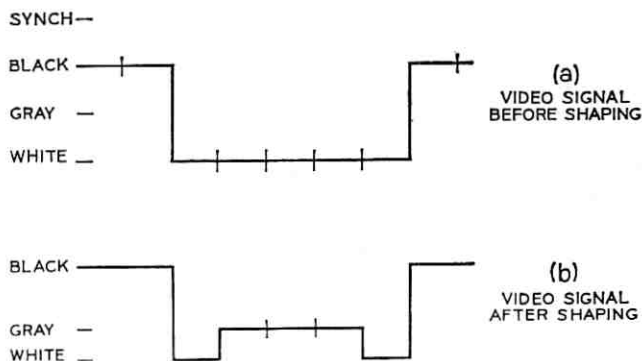


Fig. 6—Video signal before and after shaping.

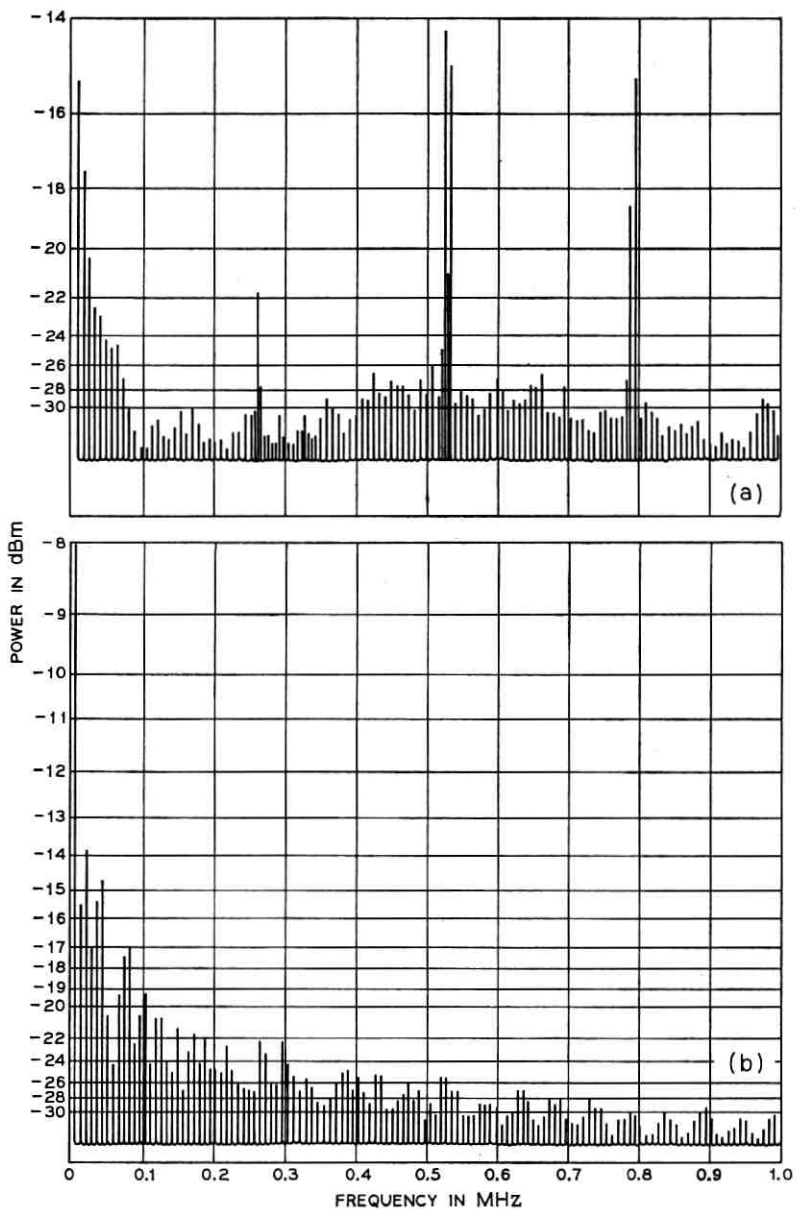


Fig. 7—(a) Example of power spectrum of a display data set signal. (b) Power spectrum of a face-to-face signal.

"I's" were transmitted, the greatest power density would occur at  $1/3.75 \mu\text{s}$ , whereas if a page full of "H's" were transmitted, with two vertical strokes per character, the greatest density would occur at  $2/3.75 \mu\text{s}$ . Similar considerations result in sidebands at multiples of 1625 Hz on these harmonics. The relatively high power density peaks in the display data set signal and possible similar signals generated by *Picturephone* stations in the graphics mode are a factor in the transmission engineering of the network to insure that such signals will not cause excessive interference into other *Picturephone* circuits.<sup>5,6</sup>

#### IV. DETAILED DISPLAY DATA SET CIRCUIT DESCRIPTION

##### 4.1 Logic Circuits and Stores

The bulk of the logic in the display data set is constructed with Western Electric RTL logic chips, coded 1J, 1K and 1L. These are respectively a quad 2-input gate, a dual buffer and a dual type-D flip-flop. The contact closure/ASCII translator and serial output circuit use commercial DTL circuits. The refresher memory is composed of commercial MOS static shift registers. The ASCII/video pattern translator employs a commercial braided transformer read-only memory.

##### 4.2 General System Organization

A block diagram of the display data set is shown in Fig. 8. Serial ASCII characters from the computer (or the customer keyboard when in the edit mode) are sampled and converted to parallel information for the refresher memory. Control characters are also detected.

Figure 9 shows part of a typical line of characters as displayed on a *Picturephone* screen. The *Picturephone* station is refreshed with two fixed-interlaced fields, one every  $1/60$ th of a second.<sup>4</sup> Therefore, the odd scan lines in each line of characters on the screen are displayed on one field and the even scan lines on the next field. Because the character plus interline spacing is ten scan lines high with five lines in each field, the ASCII information corresponding to one line of characters on the *Picturephone* screen must be presented for translation to video signals by the ASCII/video pattern translator (Fig. 8) five times before moving on to the next line of characters. As can be seen in Fig. 8, the refresher store is divided into two parts, one consisting of the entire display, called the refresher store, and the other of the line of characters being scanned onto the screen at the moment, called the line store. These two parts of the refresher memory are synchronized and operated in a start-stop mode, so that the required speed

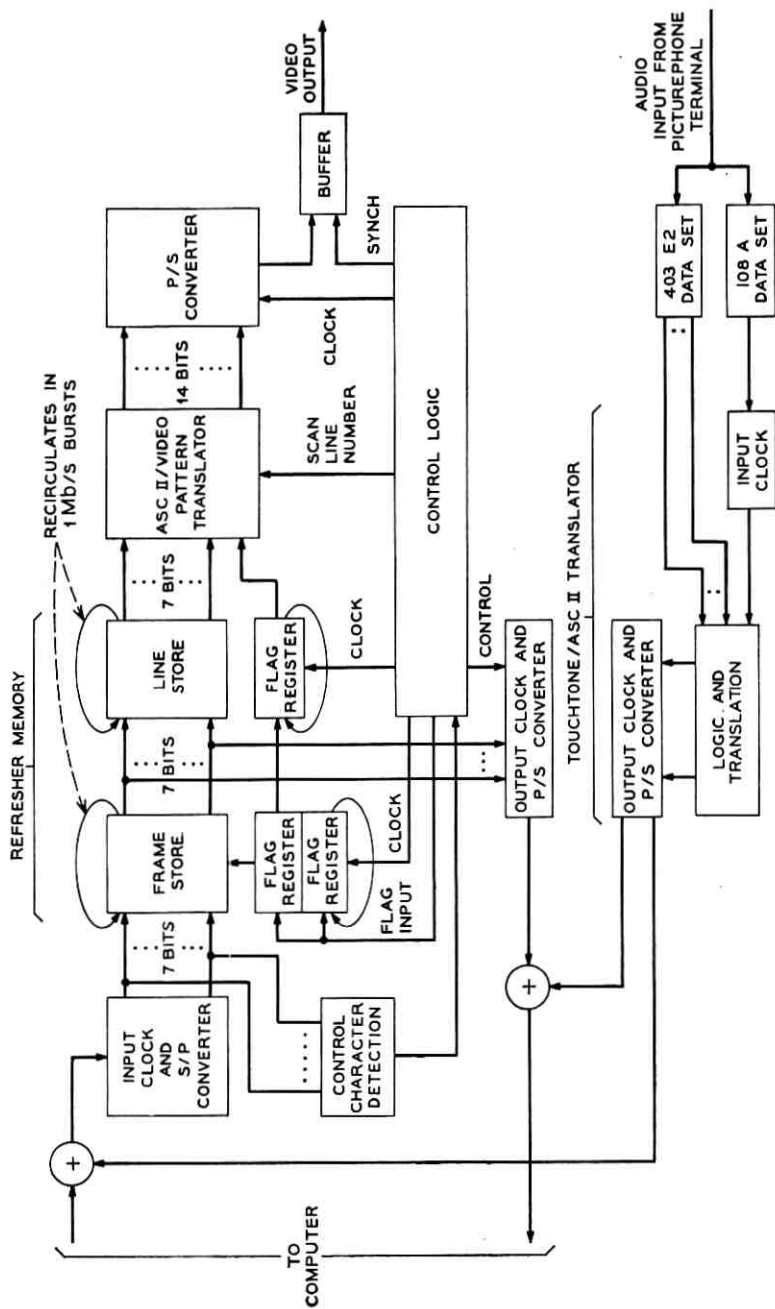


Fig. 8—Display data set block diagram.

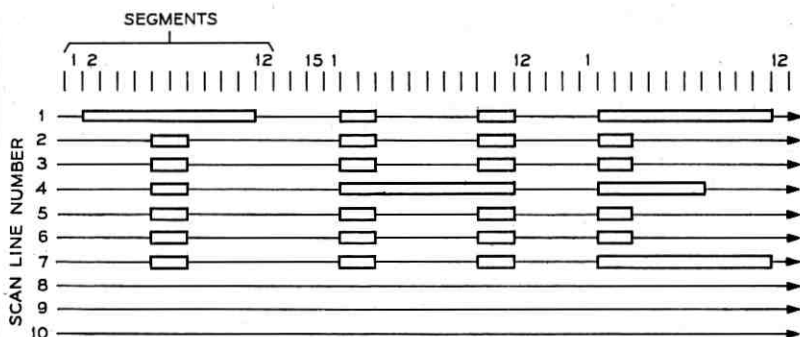


Fig. 9—Part of a typical character line.

of operation is about 1 Mb/s. In addition to lowering the speed requirements on the logic, this allows the use of low cost and compact MOS shift registers for the refresher memory.

The refresher store consists of nine parallel 484-bit shift registers.\* Seven of these store in parallel the ASCII characters making up the display. An additional register stores a flag marking the position of the next character to be written on the screen. This flag may be moved about under the command of control characters from the computer or from the keyboard associated with the *Picturephone* station. Another register contains a flag marking the character position in the top left-hand corner of the display and is used when transmitting the contents of the refresher memory to the computer.

The line store consists of eight parallel 22-bit MOS shift registers. Seven of these store in parallel the line of ASCII characters which is presently being scanned onto the screen. The eighth register contains the flag bit marking the next character position to be written on the screen if that position is on the line of characters being scanned at the moment.

The ASCII/video pattern translator accepts from the line store the seven ASCII bits of an alphanumeric symbol being scanned with the additional information of the one-out-of-ten scan line number from the control logic and generates a 14-bit parallel word. This word is the binary pattern corresponding to the appropriate scan line in the symbol. As mentioned earlier, there are 60 alphanumeric symbols in the display data set repertory, and each of these symbols is up to 12

\*The design described is capable of generating displays of 484 characters (22 rows of 22 characters) even though only 20 lines of 22 characters are guaranteed to be fully visible.



subsegments wide and seven scan lines high, not counting spacing intervals around each character. The patterns for these 60 symbols are generated by a read-only memory organized into 512 twelve-bit words. A nine-bit address formed from six of the seven ASCII bits of the symbol and three bits corresponding to the scan line number accesses one of the twelve-bit words. The 16 graph symbols are formed with combinational logic whose outputs are OR-ed with the memory output and fed into two alternating-feed shift registers, forming a parallel-to-serial converter.

The control logic shifts the contents of the shift registers into a buffer circuit. Here the signal is shaped as mentioned in Section 3.2 and preemphasized. *Picturephone* synchronizing signals generated by the control logic are then added and the composite video signal is matched to a *Picturephone* line with an unbalanced-to-balanced signal converter.

The control logic provides synchronizing clocking for all these circuits and generates appropriately positioned horizontal and vertical synchronizing signals. The memory may be cleared at any time by a character (ASCII-RS) sent from the computer or station keyboard to the display data set. The control logic then clears the refresher store and positions the writing flag in the top left-hand character position on the screen. All subsequent characters are positioned in sequential order following this character unless flag-moving cursor control characters are received. There are five flag-moving characters: Single step right (ASCII-HT), single step left (ASCII-BS), move vertically up one line (ASCII-VT), move vertically down one line (ASCII-FF), and move to the first character position on the next line down (ASCII-LF). The control logic moves the writing flag in the refresher store when these commands are received.

A transmit command (ASCII-DC2) causes the control logic to place the ASCII character marked by the reading flag in the refresher store into a shift register and output clocking circuit for transmission to the computer. After transmission of the character, the reading flag is stepped to the next character position in the refresher store and the process is repeated until the entire contents of the refresher store have been transmitted to the computer.

The *Touch-Tone* signals from the customer are received by the 403E2 Data Set associated with the display data set and converted into two-out-of-seven contact closures. The 403E2 Data Set also performs line supervision functions associated with the customer's call. The contact closures are translated by the contact closure/ASCII

translator logic into 1200- or 2400-baud serial asynchronous ASCII data for the computer. Serial data from an auxiliary alphanumeric keyboard is received via the 108A Data Set and speed-converted before transmission to either the computer or the refresher memory input of the display data set.

#### 4.3 Control Logic Functions

A block diagram of the master clocking section of the control logic is shown in Fig. 10. The basic system clock is a 1.0-MHz crystal clock. Each scan line in the *Picturephone* system is 125  $\mu$ s long. The crystal clock is counted down by a 125-state high-speed counter, which defines each microsecond on a scan line. Counts in this counter define the horizontal synch interval, the start of the visible scan interval, and also drive another 134–135 state high-speed counter, which defines each scan line in the *Picturephone* raster. Counts in this counter define the first and last scan lines on which characters occur and the vertical synch interval in conjunction with counts from the 125-state counter. The precision of a crystal clock and count-down chain is required because each segment in each symbol on the screen must be positioned precisely with respect to the synchronizing pulses and with each other segment in the display to prevent distortion in the shape of characters.

The type of high-speed counters used in the control unit were de-

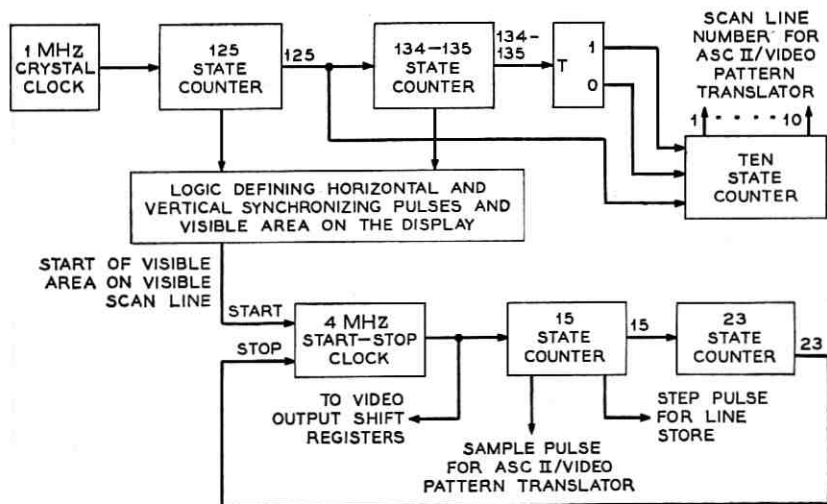


Fig. 10—Master clocking control logic block diagram.

veloped by R. A. Thompson at Bell Laboratories in 1965 and are a form of feedback shift register. The circuit is an  $n$ -stage shift register with the conditioning input to the first stage being a one whenever stages  $n$  or  $(n - 1)$  are one. The counters have a worst-case propagation time of  $2t$  where  $t$  is the delay of one stage, have  $(n - 1)^2 + 1$  states where  $n$  is the number of stages, and any state may be detected with no more than a four-input gate plus sampling.

At the start of each visible scan line in the display, a 4-MHz clock is started, which drives a 15-state counter and a 23-state counter in tandem. These counters define each character interval on the visible scan line, and when the 23-state counter reaches state 23, it turns off the 4-MHz clock. During a character interval, the 15-state counter defines a pulse which samples the output of the ASCII/video pattern translator in order to place its 14-bit output into the parallel-to-serial converter. The 4-MHz clock is used to drive the shift registers in the converter, and another state in the 15-state counter is used to step the line stores so as to present a new ASCII character for translation by the ASCII/video pattern translator.

A ten-state counter, driven in tandem from the 125-state scan line counter, defines which scan line out of ten for each line of characters is being scanned at the moment.

The refresher store and line store are kept synchronized to each other and to the display to keep their shift rates down to 1 Mb/s. This can be achieved by having the 22 ASCII characters, composing the next line of characters to be scanned on the screen, at the head of the refresher store when the scanning of the present line of characters is over. Thus it is only necessary to step the refresher store 22 times while reading the new line of characters into the line store. The refresher store then remains idle, with another new line of 22 characters at its head, while the line store circulates the 22 characters it has just received. The line store must step once after each character has been translated in order to present the next character for translation. It circulates its contents five times as the five scan lines composing one field of a line of characters on the screen are generated. During the horizontal blanking interval following the fifth scan line of this group of characters, the line store accepts a new group of 22 characters from the refresher store. By this means it is never necessary to revolve the line or refresher store to look for any appropriate group of characters, since the required characters are always at the head of the store.

The next requirement is to enter characters into the refresher store

as they are received from the computer or customer's keyboard. The input clocking circuit notifies the refresher store that a character is ready for entry. The refresher store waits until the first nominally idle interval when the line store is scanning a line of characters for translation. This interval is five scan lines or  $625 \mu\text{s}$  in length. The 484-bit refresher store registers are then stepped 484 times at 1 Mb/s and are thus back in place before the line store is ready to accept a new group of characters from the refresher store. While the refresher store is thus being revolved, the control logic looks for the writing flag marking the character position to be written into. When the character position marked by the flag appears at the head of the refresher store, this position is cleared and the new character is entered as the position shifts into the tail of the refresher store. Thus synchronism between the line and refresher stores is maintained while characters are entered into the refresher store. There are a sufficient number of nominally idle intervals in a field so that characters can be entered at a 2400-baud rate.

The next requirement is to be able to shift the position of the writing flag in the refresher store under command of the computer or customer's keyboard. This is achieved by giving the writing flag and reading flag registers outputs at the 483 and 484 stages and adding an extra 485th stage. The 484th stage output is the one normally used when revolving the store. When a step-left flag moving command is given, the refresher store is revolved once during a normally idle interval, but the 483rd stage output is used for reentry on the writing flag register. This register thus appears one stage shorter than the other refresher store registers and the writing flag moves one character position to the left on the screen. To move the flag one step to the right on the screen and when entering a character into the store, the refresher store is revolved once, using the 485th stage output for reentry, so that the writing flag register appears one stage longer than the other refresher store registers. To move the writing flag vertically upward or downward, the step left or step right processes are repeated 22 times. To move the writing flag to the first position on the next character line downward, the step-right process is performed and a check is made to see if the writing flag is in the first character position on a line. This process is repeated until the flag is in this position. It should be noted that stepping right at the end of a line brings the writing flag to the first position on the next line. The writing flag is also stored as an eighth bit associated with each ASCII character in the line store. During the edit mode the character position marked

by the flag is detected by the ASCII/video pattern translator and a blinking underline is displayed on the ninth scan line of the character window in that position.

During all maneuvers of the writing flag and addition of characters to the refresher store, the reading flag in the refresher store remains at the first character position on the screen. When a transmit command is received, the refresher store is revolved once and the ASCII character marked by the reading flag is transferred to an output shift register. During this revolution of the refresher store the 485th stage output on the reading flag register is used so that after the revolution the reading flag has moved one character position to the right on the display. When the ASCII character has been transmitted to the computer, the process is repeated for the next character and this is done 484 times to transmit the entire contents of the refresher store to the computer.

As mentioned earlier, a graticule is displayed on the screen whenever a graph character is present in the display. The graticule is generated by logically OR-ing with an analog weight the ninth scan line in each character position and the first and 15th bit on each scan line in each character position. The analog weights are so chosen that the graticule is equally bright in both horizontal and vertical components and suitably dimmer than the characters making up the graph, so as not to be obtrusive.

## V. EPILOGUE

The technical feasibility of computer access from *Picturephone* stations was first tested at Bell Telephone Laboratories in an experimental system. About 40 *Picturephone* users at Murray Hill and Holmdel, New Jersey, and 40 users at American Telephone and Telegraph Company in New York City have been able to call a computer located at Holmdel, to retrieve simple information (such as personnel statistics), to use the computer's calculating ability, and to use an alphanumeric keyboard to modify or update displays.

The computer support for the experimental system was provided by available facilities at Holmdel. An IBM 360/50 computer operating with the System 360/Operating System under a multiprogrammed environment is used in the system. Two display data sets on the experimental *Picturephone* network are connected to the computer via an IBM 2701 Telecommunications Control Unit equipped with IBM Type III Terminal Adapters.

The *Picturephone* software resides in one partition of the 360/50. A small input/output software module, written in Basic Telecommunication Access Method and Basic Assembly Languages, resides in core at all times. The basic element of this module is a new device-dependent program to support the display data set.

A product trial of the display data set in a customer environment was carried out in cooperation with Westinghouse Electric Corporation. Westinghouse provided the computer support for *Picturephone*/computer access using a Univac 494 computer and their own software. It was not necessary to make any hardware modifications. During the trial, over 900 *Picturephone* calls were made to the computer by Westinghouse executives and those testing the system.

Display data sets are now being produced by Western Electric Company and are being used in initial *Picturephone* service. Computer access capability appears to meet the objectives set for it initially. The service will no doubt be extended and improved as *Picturephone* service evolves.

#### VI. ACKNOWLEDGMENTS

The authors would like to acknowledge the contributions of O. F. Gerkenmeier, M. G. Edelstein and G. E. Larson in the design of the display data set. J. J. Mansell, J. J. Horzepa and R. L. Simms, Jr., were major contributors to the initial systems engineering of the computer access service. E. J. Rodriguez and B. D. Owen have been responsible for the computer support used to demonstrate and test the system.

#### REFERENCES

1. Dougherty, H. J., Peterson, E. B., and Schachtman, M. G., "The *Picturephone*® System: Maintenance Plan," B.S.T.J., this issue, pp. 621-644.
2. Bell System Data Communications Technical Reference—Data Set 403D 403E, Interface Specification, November, 1967, American Telephone and Telegraph Company.
3. Bell System Data Communications Technical Reference—150 Baud Private Line Channels Interface Specification, February, 1968, American Telephone and Telegraph Company.
4. Cagle, W. B., Stokes, R. R., and Wright, B. A., "The *Picturephone*® System: 2C Video Telephone Station Set," B.S.T.J., this issue, pp. 271-312.
5. Brown, H. E., "The *Picturephone*® System: Transmission Plan," B.S.T.J., this issue, pp. 351-394.
6. Bunin, B. J., Hirsch, R. B., and Olsen, R. E., "The *Picturephone*® System: Crosstalk Considerations in the Transmission of Analog Signals on Paired Cable," B.S.T.J., this issue, pp. 427-457.

## Contributors to This Issue

HAROLD P. ANDERSON, B.E.E., 1952, Syracuse University; M.S., 1959, Stevens Institute of Technology; Bell Telephone Laboratories, 1952—. Mr. Anderson was first engaged in circuit design of NIKE and TRADIC computers and later in circuit design and device characterization on exploratory key systems. Since 1962, he has supervised groups on *Touch-Tone*® Calling, PBX-ANI, Magnetic Memories, Exploratory Key System, and the 850A PBX. Senior Member, IEEE; Member, Tau Beta Pi, Eta Kappa Nu, Theta Tau.

JACK A. BAIRD, B.S.E.E., 1943, Texas A. & M. College; M.S., 1950, Stevens Institute of Technology; Ph.D. (Electrical Engineering), 1952, Texas A. & M. University; Bell Telephone Laboratories, 1946—. Mr. Baird has worked on the development of military radar and communications systems, with particular emphasis on the application of transistors and other solid-state devices. He is currently Vice President, Network Planning and Customer Services, of Bell Laboratories. In this position he is responsible for the system planning of the Bell System network and the planning and development of customer services and equipment. Fellow, IEEE.

E. CRAIG BENDER, B.S.E.E., 1964, Syracuse University; M.S.E.E., 1966, University of California at Los Angeles; Bell Telephone Laboratories, 1966—. At Bell Laboratories, Mr. Bender has derived simulation procedures for submarine tracking, automatic equalization, and radio fading. He is currently involved with the study and planning of wide-band data services. Member, Eta Kappa Nu, Tau Beta Pi.

CHARLES BREEN, E.E., 1930, Cornell University; American Telephone and Telegraph Company Long Lines Department, 1930-1938; Bell Telephone Laboratories, 1938-1970. Mr. Breen's early Bell Laboratories work included maintenance studies and formulation of requirements for toll and local switching systems. During World War II he was engaged in the development of military radar and microwave communications. From 1953 until his retirement in August 1970, he headed departments responsible for PBX, Key Telephone Systems, Signaling, and Special Service Projects. This work included responsibility for Systems Engineering work on the complete Bell System line

of modern electronic and electromechanical PBX systems. This work has also included planning of Centrex, *Picturephone*<sup>®</sup>, and other new PBX service features, and special systems for hospitals, motels and telephone answering customers. Member, Eta Kappa Nu.

CHARLES W. BRODERICK, B.S.E.E., 1961, University of Buffalo; M.E.E., 1963, New York University; Bell Telephone Laboratories, 1961—. Mr. Broderick has been engaged in microwave device development for satellite use and improvement of the TD-2 Radio System. He is currently working on digital transmission over microwave radio. Member, IEEE, Tau Beta Pi, Eta Kappa Nu.

DONALD W. BROWN, B.S.E.E., 1962, University of Louisville; M.S.E.E., 1964, New York University; Bell Telephone Laboratories, 1962—. Mr. Brown's initial work at Bell Labs has included circuit and logic design in the development of trunks and peripheral circuits for the No. 101 ESS. He was also engaged in program development for the 50A test position in the No. 2 ESS AUTOVON project. Currently, he is engaged in circuit and program design for the No. 101 ESS wideband switch unit.

H. E. BROWN, B.S. (E.E.), 1956, University of Rhode Island; M.S. (E.E.), 1961, Stevens Institute of Technology; Bell Telephone Laboratories, 1956—. Mr. Brown's initial assignment involved feasibility studies of alternative ways to employ electronic switching offices to serve low-density population areas. He later assumed responsibility for engineering studies on PCM Channel Banks (D1 and D2) and digital transmission lines using paired cable (T1 and T2). These studies included performance standards, maintenance arrangements, field evaluation, and reliability surveys. Following this, his responsibilities shifted to transmission studies related to *Picturephone*<sup>®</sup> service, including planning for the evolution of the network, establishing overall performance objectives and engineering of local video loops and trunks. More recently, he has been engaged in economic and performance studies of new telephone stations, both public and residence sets. He is presently Head of the Station Studies Department.

JAMES M. BROWN, B.S.E.E., 1957, M.S.E.E., 1958, and Ph.D.E.E., 1964, Polytechnic Institute of Brooklyn; Bell Telephone Laboratories, 1958—. Mr. Brown has worked on the design of the lineless telephone, the *Bellboy*<sup>®</sup> personal radio signaling system, the design of a hybrid



network with a dynamically balanced matching impedance, and the subscriber loop multiplexer system. He is presently Supervisor of a group responsible for the design of a subscriber loop video transmission system for *Picturephone*® service. Member, I.E.E.E., Eta Kappa Nu, Sigma Xi.

BARRY J. BUNIN, B.E.E., 1963, Cooper Union; M.S., 1964, University of Pennsylvania; Ph.D., 1970, Polytechnic Institute of Brooklyn; Bell Telephone Laboratories, 1963-1966, 1968—. Mr. Bunin has been concerned with analog transmission problems of the *Picturephone*® visual telephone system. He is presently studying digital encoding and transmission of video signals. Member, IEEE, Eta Kappa Nu.

PAUL N. BURGESS, B.S.E.E., 1963, University of Wisconsin; MScEE, 1965, Ohio State University; Bell Telephone Laboratories, 1963—. Mr. Burgess has been engaged in circuit and system development for the No. 5 crossbar switching system. Since 1968, he has been Supervisor of a group responsible for the circuit design of audio and video switching networks and automatic message accounting (AMA) equipment for the No. 5 system. Member, Eta Kappa Nu.

WILLIAM B. CAGLE, B.S.E.E., 1953, University of Oklahoma; Bell Telephone Laboratories, 1953—. Mr. Cagle was initially involved in logic circuit and control unit design for No. 1 ESS. In 1964, he was appointed Head of a department which was subsequently concerned with the development of mobile telephones, and *Picturephone*® station sets. He is presently Director of the Telephone and *Picturephone* Planning Center, which is responsible for planning features, systems, and station set requirements for new telephone and *Picturephone* services. Member, IEEE, Tau Beta Pi, Sigma Tau, Eta Kappa Nu, Phi Eta Sigma.

THEODORE V. CRATER, B.S.E.E., 1947, Montana State College; M.S.E.E., 1949, California Institute of Technology; Ph.D., 1953, Northwestern University; Bell Telephone Laboratories, 1953—. In his earlier years, Mr. Crater worked primarily on engineering of the T1 Carrier System repeatered line. Since 1961 he has been involved in the planning of systems for *Picturephone*® service. At present he is in charge of a group concerned with digital transmission of the visual telephone signal and with provision of the image in color. Member, IEEE.

ANDREW DASKALAKIS, B.S.E.E., 1957, Drexel University; M.E.E., 1959, New York University; Bell Telephone Laboratories, 1957—. Mr. Daskalakis' initial assignments were in the design of circuits for the 3A Speakerphone System including conferencing arrangements. Later he contributed to the development of the Two-Way Dial MJ Mobile Radio Telephone System for use primarily in automobiles. He has supervised a group that developed a Dial Telephone System for the high-speed "METROLINER" Train and proposed a plan for a high-capacity Mobile Telephone System. Presently he supervises a group responsible for the development of all *Picturephone*® Key Systems. Member, Phi Kappa Phi, Tau Beta Pi, Eta Kappa Nu.

IRWIN DORROS, S.B. and S.M. (Electrical Engineering), 1956, Massachusetts Institute of Technology; Eng. Sc.D. (Electrical Engineering), 1962, Columbia University; Bell Telephone Laboratories, 1956—. Mr. Dorros has worked on electronic switching, data communications and digital transmission development projects and on transmission systems engineering programs. He is currently Director of the Facilities Network Planning Center. Member, IEEE.

H. J. DOUGHERTY, B.S.E.E., 1955, and M.S.E.E., 1956, University of Maine; New England Telephone and Telegraph Company, 1941-1956; Bell Telephone Laboratories, 1956—. Mr. Dougherty was first engaged in the design and development of circuits for the experimental electronic switching system. More recently he supervised a group responsible for electronic switching system operating requirements and maintenance. He presently is Supervisor of a group responsible for trunk maintenance in all kinds of switching systems. Member, IEEE, Tau Beta Pi.

DAVID L. FAVIN, B.S.E.E., 1950, Moore School of the University of Pennsylvania; S.M.E.E., 1952, Massachusetts Institute of Technology; Bell Telephone Laboratories, 1952—. Mr. Favin has been developing field test equipment which include microwave sweepers, envelope delay distortion measuring sets, random data signal generators, noise measuring equipment and echo suppressor measuring sets. He currently supervises a group engaged in designing time domain transmission measuring equipment. He holds sixteen patents in circuit design. Member, IEEE Electro-Acoustic Committee, Eta Kappa Nu, Sigma Xi, Sigma Tau, Tau Beta Pi.

JOHN F. GILMORE, B.S.E.E., 1958, Polytechnic Institute of Brooklyn; M.E.E., 1961, New York University; Bell Telephone Laboratories, 1958—. Mr. Gilmore has worked on the design of traffic measurement circuits, loop and trunk maintenance arrangements, transmission test lines and SMAS No. 1A. He is currently supervising a group engaged in the development of new maintenance access arrangements. Member, Eta Kappa Nu, Sigma Xi, Tau Beta Pi.

ALAN M. GORDON, Scb. (Engineering), 1957, Brown University; M.S.E.E., 1959, New York University; Bell Telephone Laboratories, 1957—. Mr. Gordon has done development work on the *Touch-Tone*<sup>®</sup> dialing system, *Picturephone*<sup>®</sup> station sets, and tantalum thin-film circuits and facilities. He currently is Supervisor of the *Picturephone*<sup>®</sup> Receiver Group. Member IEEE, Tau Beta Pi.

JOHN F. GUNN, B.S.E.E., 1964, Tufts University; M.S.E.E., 1965, California Institute of Technology; Bell Telephone Laboratories, 1965—. Mr. Gunn has worked on the analysis and design of high-speed digital transmission equipment. He is currently Supervisor of the Digital Circuits and Systems Analysis Group in the Carrier Transmission Laboratory. Member, Tau Beta Pi, Eta Kappa Nu.

ROBERT B. HIRSCH, B.E.E., 1963, George Washington University; M.S.E.E., 1965, New York University; Bell Telephone Laboratories, 1963—. Mr. Hirsch initially worked on the SF Submarine Cable system and later was involved in the systems engineering of TASI-B. He has since been concerned with *Picturephone*<sup>®</sup> signal transmission, and is presently responsible for writing the engineering rules for *Picturephone* analog trunks. Member, Tau Beta Pi.

JOHN R. HORVATH, B.E., 1962, and M.S., 1963, Rutgers University; Bell Telephone Laboratories, 1963—. At Bell Labs, Mr. Horvath has been engaged in development work on the No. 101 and No. 2 Electronic Switching Systems. At present he is involved with programming the No. 101 ESS to provide *Picturephone*<sup>®</sup> service. Member, IEEE, Tau Beta Pi, Eta Kappa Nu, Pi Mu Epsilon.

R. D. HOWSON, B.E., 1959, M.E., 1960, and D.E., 1967, Yale University; Bell Telephone Laboratories, 1962—. Mr. Howson has been engaged in the development of wideband data communication sys-

tems with particular emphasis on signal theory and techniques. Member, Tau Beta Pi, Sigma Xi, IEEE.

H. I. MAUNSELL, B.S., 1950, University of the Witwatersrand, Johannesburg, South Africa; Bell Telephone Laboratories, 1957—. For several years Mr. Maunsell worked on the terminal equipment, test sets and protection switching circuits for the TH microwave radio relay system. He later became concerned with communication circuits for the *Telstar*<sup>®</sup> satellite, subsequently supervising the group responsible for the design of the satellite ground station transmitter. From 1965 to 1968, he was involved with studies of radio system performance with regard to digital transmission. He has since been actively associated with codec development.

J. B. MILLARD, B.S. (Engineering Science), 1961, Rockhurst College; B.E.E., 1961, University of Detroit; M.E.E., 1963, and Ph.D., 1966, New York University; Bell Telephone Laboratories, 1960—. At Bell Laboratories, Mr. Millard has worked on the high-speed PCM terminal and on satellite communications. More recently his work has been on the encoding and digital processing of *Picturephone*<sup>®</sup> signals. He is Supervisor in the Digital Transmission Design Department. Member, IEEE, Tau Beta Pi, Eta Kappa Nu.

R. E. OLSEN, S.B. and S.M., 1965, Massachusetts Institute of Technology; Bell Telephone Laboratories, 1965—. In addition to work on *Picturephone*<sup>®</sup> signal transmission, Mr. Olsen has performed field measurements of L-multiplex equipment and economic studies of long-haul transmission facilities. He is currently engaged in system engineering studies of short-haul carrier facilities for use in urban areas. Member, IEEE.

THORNTON S. PAXTON, B.S.E.E., 1963, Brigham Young University; M.S.E.E., 1965, Massachusetts Institute of Technology; Bell Telephone Laboratories, 1963—. Mr. Paxton has been associated with various aspects of No. 101 ESS development. Initially he was engaged in logic design for the 4A SU. He was later concerned with some programming aspects of call processor trouble switching routines and then was associated with peripheral unit maintenance in the No. 2 ESS AUTOVON project. More recently he was jointly responsible for the organization and teaching of the Indian Hill Technology Course for new technical employees and for teaching logic and switch-

ing in the GSP program. He is presently responsible for the attendant call processing program to provide video service in the No. 101 ESS. Member, Tau Beta Pi, Illinois Society of Professional Engineers.

E. B. PETERSON, B.S.M.E., 1948, Cornell University; M.S.I.E., 1954, Stevens Institute of Technology; Bell Telephone Laboratories, 1950—. Mr. Peterson was initially involved in development of electromechanical equipment for NIKE and TERRIER military missile systems. He later served in various personnel and staff positions. In 1960, he rejoined the technical area to supervise a group concerned with reliability studies of military systems. He supervised the Environmental Laboratory operations for *Telstar*® in 1962 and the following year assumed his current responsibilities supervising the local switching systems and special projects maintenance engineering group. Member, Cornell Engineers, IEEE.

G. W. PHIPPS, B.S. (E.E.), 1960, Rose Polytechnic Institute; M.S. (E.E.), 1963, New York University; Bell Telephone Laboratories, 1961—. Mr. Phipps initially worked on data station engineering for both line-switched and store-and-forward data systems. He then became involved in studies of communications for business information systems, primarily in the use of CRT terminals for real-time information retrieval. He is now Supervisor of a group responsible for computer-access planning for *Picturephone*® service. Member, IEEE, Tau Beta Pi.

J. S. RONNE, B.S. (Physics), 1954, and M.A. (Physics), 1955, Oregon State University; Bell Telephone Laboratories, 1955—. Mr. Ronne initially worked on the design and analysis of transistor and magnetic core circuits for analog and digital computers in connection with military defense computer research programs. More recently he has been engaged in the system and circuit design of wideband digital transmission terminals. Member, Sigma Xi, IEEE, Sigma Pi Sigma, Pi Mu Epsilon.

MARSHALL G. SCHACHTMAN, B.S. and M.S. in E.E., 1958, Massachusetts Institute of Technology; Bell Telephone Laboratories, 1955—. After early work on a new low-current telephone subscriber's set, Mr. Schachtman was engaged in designing circuits for use in test equipment for Radio Relay Systems. He later was involved in systems engineering work on TASI and on system studies of the use of lasers in

communications. He continued his systems engineering work on SD submarine cable systems and on studies of methods for improving the performance of HF radio telephone systems. He then began working on the early phases of studies for transmission maintenance of *Picturephone*® service. He has supervised a group working on systems studies of analog video telephone transmission and currently is Supervisor of a group studying advanced transmission maintenance techniques for *Picturephone* service. Member, IEEE, Eta Kappa Nu, Tau Beta Pi.

JOHN B. SINGLETON, B.S. (Physics), 1952, Holy Cross; M.S. (Physics), 1954, University of Rhode Island; Bell Telephone Laboratories, 1954—. Mr. Singleton has worked on the development of numerous semiconductor devices ranging from point contact diodes to integrated circuits. He currently is Head of the Integrated Circuits Department. Senior Member, IEEE; Member, Electromechanical Society, Sigma Xi.

JAMES E. STICKEL, B.E.E., 1963, Georgia Institute of Technology; MScEE, 1965, Ohio State University; Bell Telephone Laboratories, 1963—. Mr. Stickel was primarily concerned with the physical design of the wideband network during the initial phases of the *Picturephone*® development. He now supervises a group responsible for the equipment design of central office trunks for the No. 5 crossbar system. Member, Eta Kappa Nu.

REMBERT R. STOKES, B.S.M.E., 1953, Clemson University; M.S., 1966, Butler University; Bell Telephone Laboratories, 1953—. Mr. Stokes has been involved in the design of card dialers, automatic reporting telephones, public coin telephone apparatus and the laminated coinage for the U. S. Government. Beginning in 1966, he supervised the physical design of the Mod II *Picturephone*® Station Set and the 50A Conference Set. He is presently Head of the Physical Design of Telephones Department and holds 19 patents in this field. Member, ASME, Tau Beta Pi, Phi Kappa Phi.

JOSEPH F. URICH, B.E.E., 1959, Manhattan College; M.S.E.E., 1962, New York University; Bell Telephone Laboratories, 1959—. Mr. Urich has worked on systems engineering of signaling and switching systems. Formerly Supervisor of the *Picturephone*® Switching Studies Group, he now supervises a group performing special studies of local switching systems for data applications.

P. S. WARWICK, B.E.E. 1962, Cornell University; M.E.E., 1963, Cornell University; Bell Telephone Laboratories, 1962—. Mr. Warwick has worked on transfluxor magnetic logic circuits and domain-wall shift-registers and their applications to data terminals. He was the main designer for the 820A Full Duplex Station Controller for the No. 1 ESS-ADF Selective Calling System. Since 1966 he has worked on the design of the Display Data Set providing computer access for *Picturephone*® Service users.

D. C. WELLER, B.A. (Physics), 1947, Kenyon College; M.S. (EE), 1948, University of Illinois; Bell Telephone Laboratories, 1948—. Mr. Weller has worked on the design and development of transmission systems and on the development of magnetic storage systems for computers. Since 1961 he has supervised a group that has developed carrier terminal equipment, particularly, high-speed data transmission terminals. Member, IEEE.

BERNARD A. WRIGHT, B.S.E.E., 1960, University of Michigan; M.S.E.E., 1963, New York University; Bell Telephone Laboratories, 1960—. Since 1962, Mr. Wright has been involved in the design of the Mod I and Mod II *Picturephone*® station sets. He is presently a Supervisor responsible for the transmitter and camera portions of the 2C Video Telephone Station. Member, IEEE, Tau Beta Pi, Eta Kappa Nu, Phi Kappa Phi.

