Transmission

The *Picturephone*® System:

Transmission Plan

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An important step in preparing for a nationwide, switched, video telephone 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[1,2] 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,[1,3] which will closely resemble that of the telephone network. The ultimate number of switching levels in the hierarchy 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

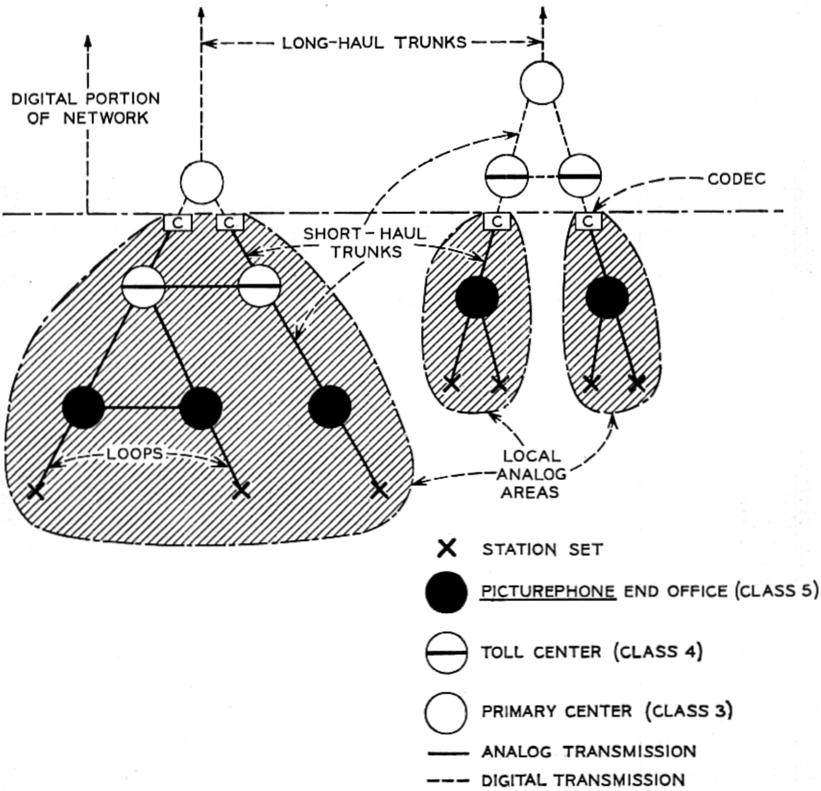

Fig. 1—*Picturephone*® network.

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:[4] (*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.[5] 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.[4] (*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.[6,7]

The interface between the analog and digital portions of the network is through an A/D converter called a codec (for coder-decoder).[5] 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 repeatered line.[4,8]

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.[9] 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.[10] 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,*[11] 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.

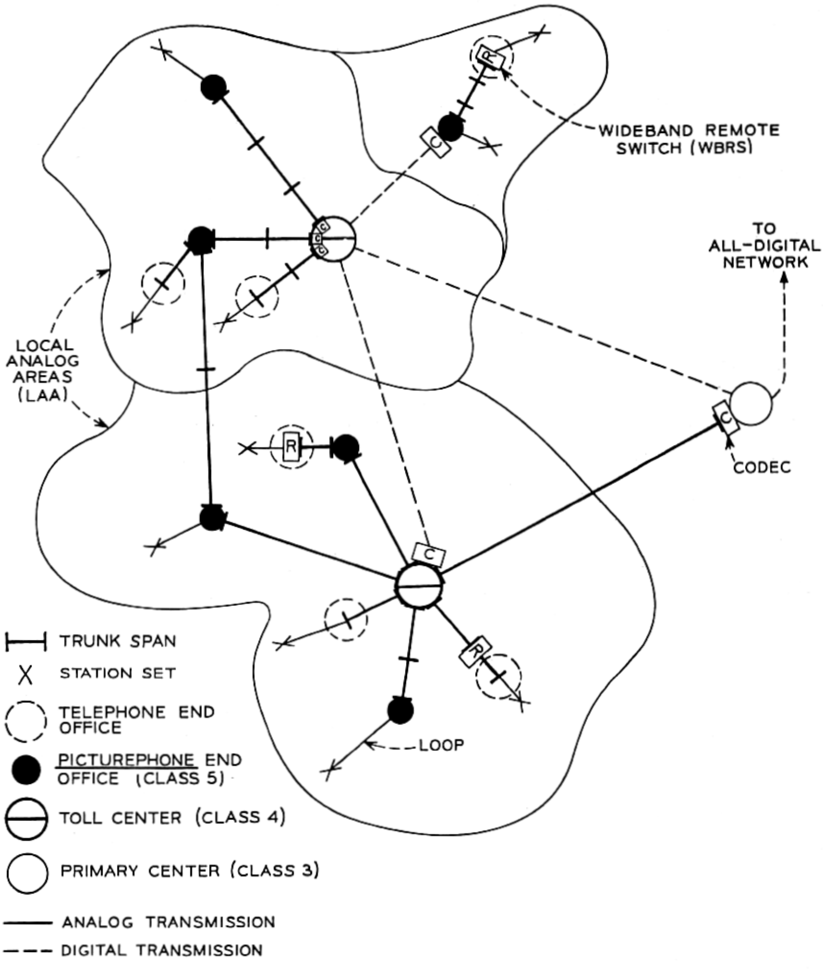

Fig. 2—Illustrative local area configurations.

is used for providing telephone service, with equalizers placed at regular intervals along the cable.[12]

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,[12] 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.[2] 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 information in particular pulse slots. Timing jitter occurs when the time between adjacent pulse positions is altered due to imperfect timing recovery in the regenerators.[13]

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,[2] 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	2.6×10^{-7}	95
Short-haul facilities	500	0.4×10^{-7}	95

Switching Machines†,‡	
Error Rate	% of Connections
3×10^{-9}	95
3×10^{-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.

† Each machine.

‡ 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),[14] behind a private branch exchange (PBX)[15,16] or a WBRS.

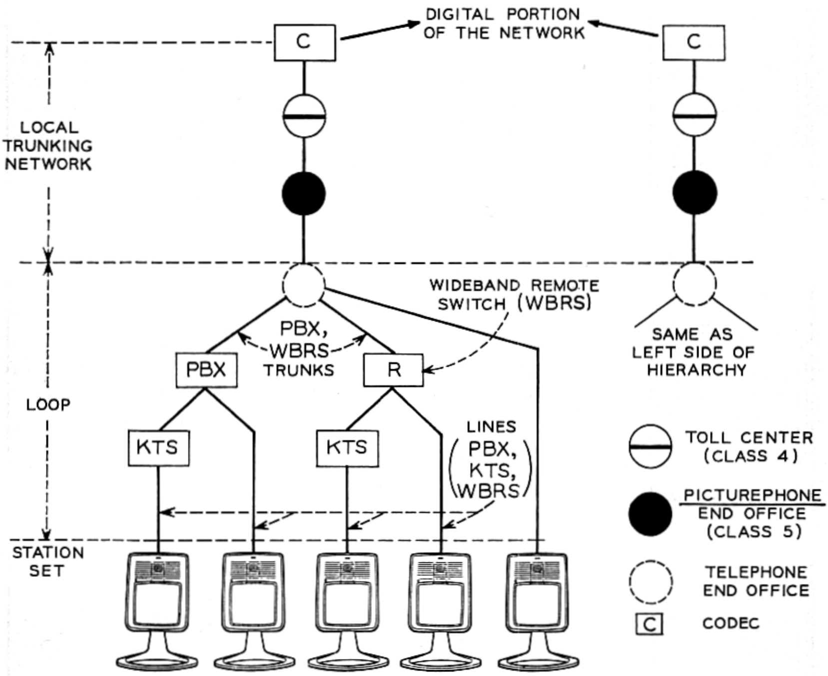

Fig. 3—Local analog area.

Echo rating,[2,17,18] 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 Number in a Full Network‡	Echo Rating (dB)			Random Noise* (dBm**)				Self Crosstalk (loss in dB at 150 kHz**)			Worst Disturber Crosstalk (loss in dB at 150 kHz**)	Impulse Noise* (probability of noise voltage exceeding plus or minus 18 millivolts**)	Flat Gain Variation (dB)			Low Frequency Roll-off (tilt)	Power Hum (dBm**)		
		95%	Mean	σ	95%	Mean	σ	MeanΔ	95%	Mean	σ			95%	Mean	σ		60 Hz	120 Hz	180 Hz
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	0.8%	−39	−45	−48
LOOP†	2 (2)	−31 (−37)	−37.6	4.0	−55	−61.6	4.0		52	66.5	8.8	45	3.0×10^{-6}	$\left\{ \begin{matrix} +2.2 \\ -3.2 \end{matrix} \right\}$	$\left\{ \begin{matrix} +1.0 \\ -2.0 \end{matrix} \right\}$	0.6	1.5%	−28	−34	−37
SWITCHING OFFICE	6	−48	−54.6	4.0	−62	−65.3	2.0		65	69.9	3.0	45	1.0×10^{-6}	±0.2	0	0.1	0.1%	−38	−44	−47
ANALOG TRUNK SPAN	6 (2)	−37 (−43)	−40.3	2.0	−58	−64.6	4.0		57	68.0	6.7	45	0.5×10^{-6}	±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	0.2%	N/A	N/A	N/A
OVERALL SYSTEM OBJECTIVES			−26§				−44§			45§		45	1.5×10^{-5}		$\left\{ \begin{matrix} +5.0 \\ -7.0 \end{matrix} \right\}$§		10%	−22	−28	−31

* Weighted Noise—Use weighting curve, Fig. 5.
** Referenced to 0 PTLP (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.

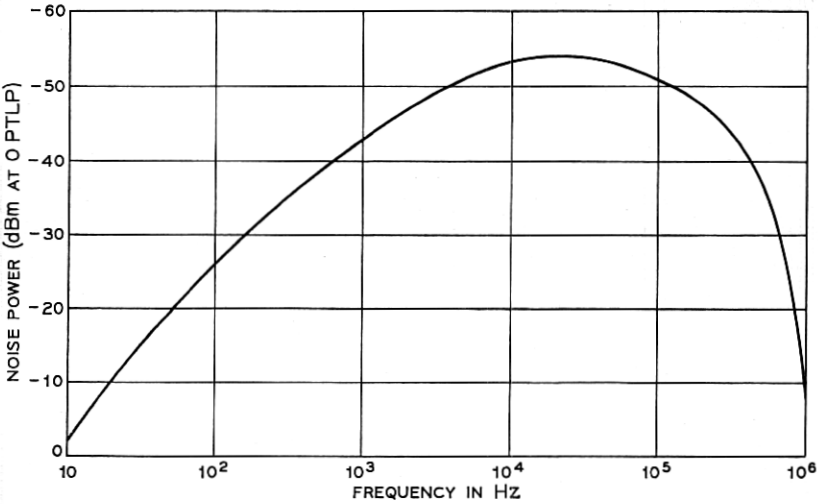

Fig. 4—Single frequency interference objective.

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

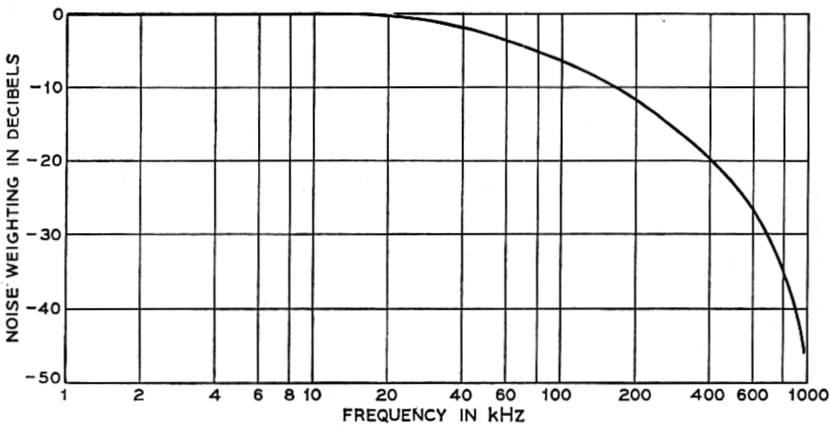

Fig. 5—Noise weighting curve (including de-emphasis, roll-off and eye weighting).

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

	Echo Rating (dB)			Random Noise* (dBm**)			Self Crosstalk (loss in dB at 150 kHz**)			Worst Disturber Crosstalk (loss in dB at 150 kHz**)	Impulse Noise* (probability of noise voltage exceeding plus or minus 18 millivolts**)	Flat Gain Variation (dB)			Low Frequency Roll-off (tilt)	Power Hum (dBm**)		
	95%	Mean	σ	95%	Mean	σ	95%	Mean	σ			95%	Mean	σ		60 Hz	120 Hz	180 Hz
KTS	−50.0	−52.5	1.5	−60.0	−62.5	1.5	58.0	59.6	1.0	45	0.5×10^{-6}	$\left\{ \begin{matrix} 0 \\ -0.4 \end{matrix} \right\}$	−0.2	0.1	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	1.0×10^{-6}	$\left\{ \begin{matrix} -0.1 \\ -0.5 \end{matrix} \right\}$	−0.3	0.1	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	1.5×10^{-6}	±2.5	±1.5	0.5	1.5%	−29	−35	−38
OVERALL LOOP	−31§			−55§			52§			45	3×10^{-6}	$\left\{ \begin{matrix} +2.2 \\ -3.2 \end{matrix} \right\}$ §			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[2] 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

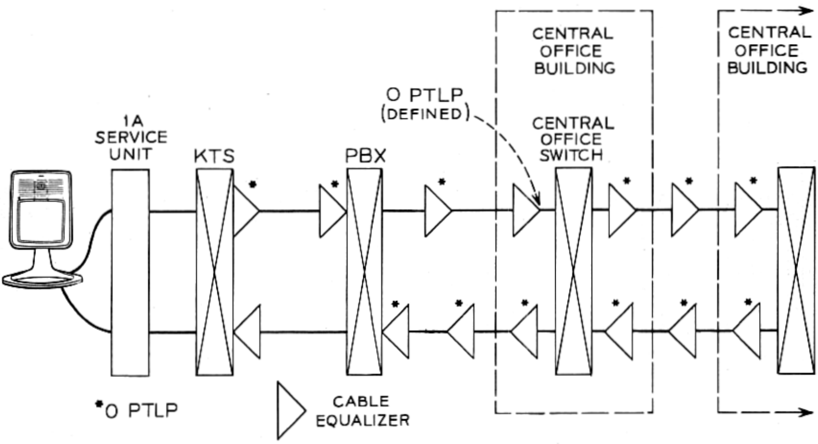

Fig. 6—Transmission level plan (initial service).

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

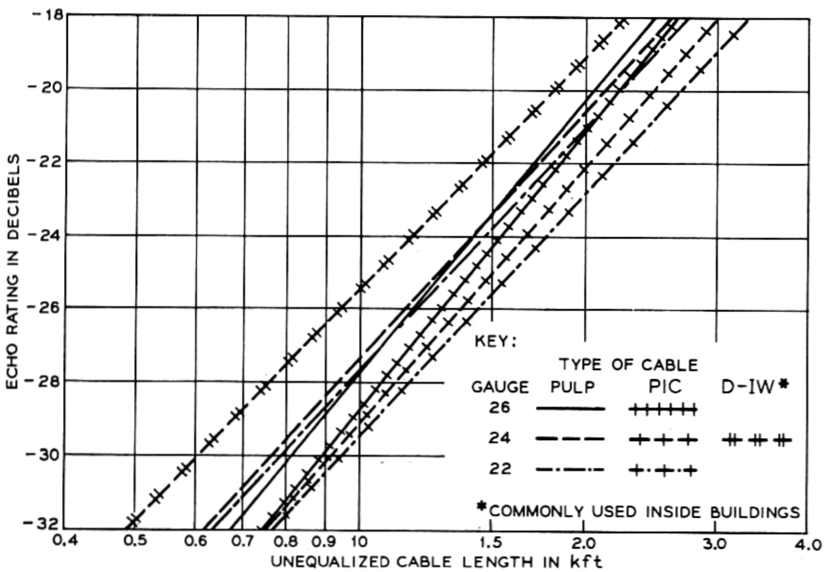

Fig. 7—Echo rating versus unequalized cable length.

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 ±25°F from nominal. This results in an equalized length times unequalized temperature change of 18 × 25 or 450 kft 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

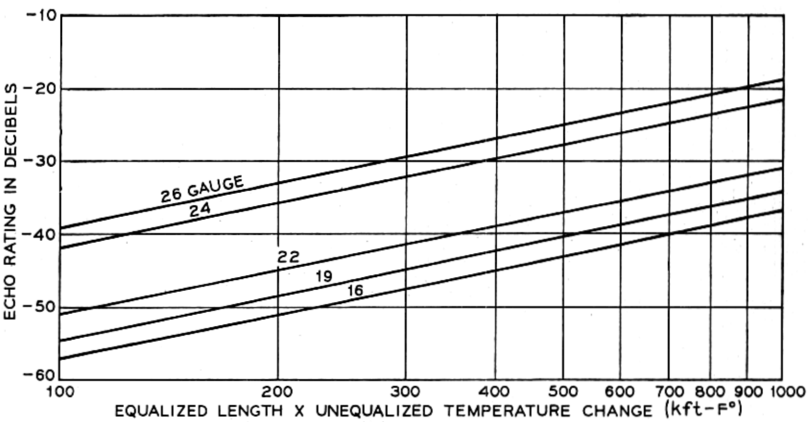

Fig. 8—Effect of change in cable temperature on echo rating.

* 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.[2]

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

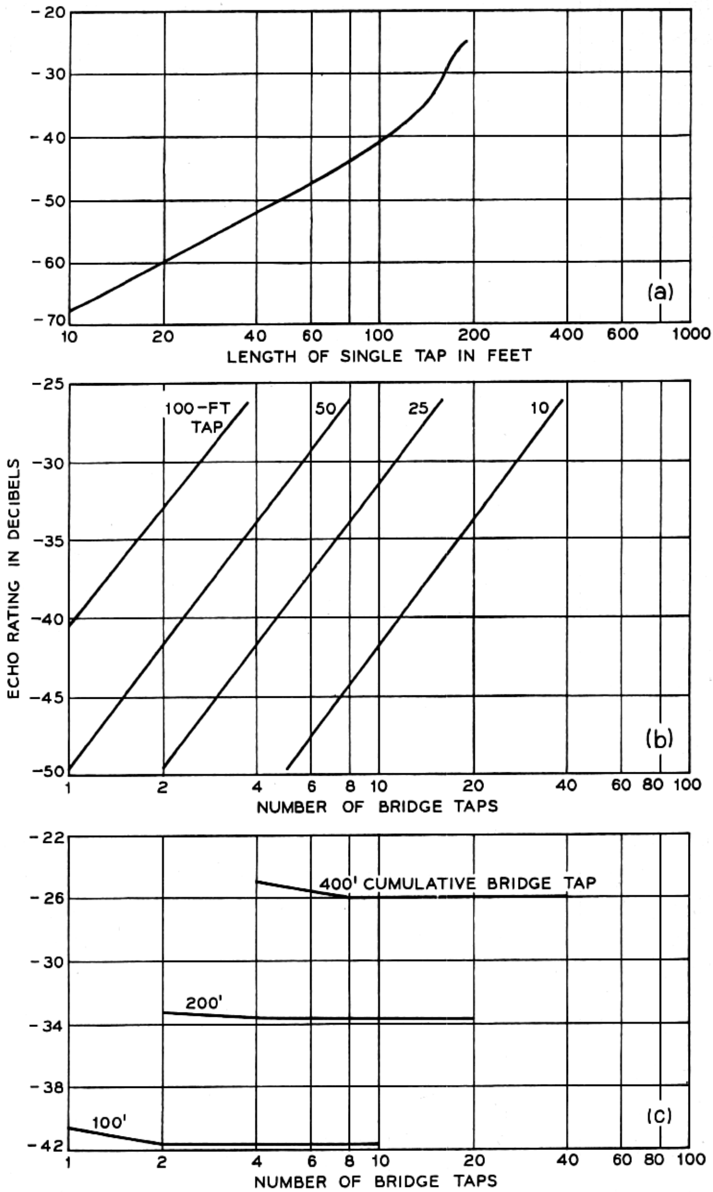

Fig. 9—(a) Echo rating of a 26-gauge equalized cable pair with a single bridge tap (unequalized). (b) Echo rating of a 26-gauge equalized cable pair with multiple bridge taps (unequalized). (c) Echo rating of a 26-gauge equalized cable pair with cumulative bridge taps (unequalized).

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.[20]

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.[5] 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.[19] 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.[21] 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[19] corrects the signal level to meet the end-to-end objectives of ± 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 information, 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:[22] 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	σ
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 dBrnc0	3.0 dB
Noise (VF trunks—0 to 8 miles)	10 dBrnc0	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.[23]

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

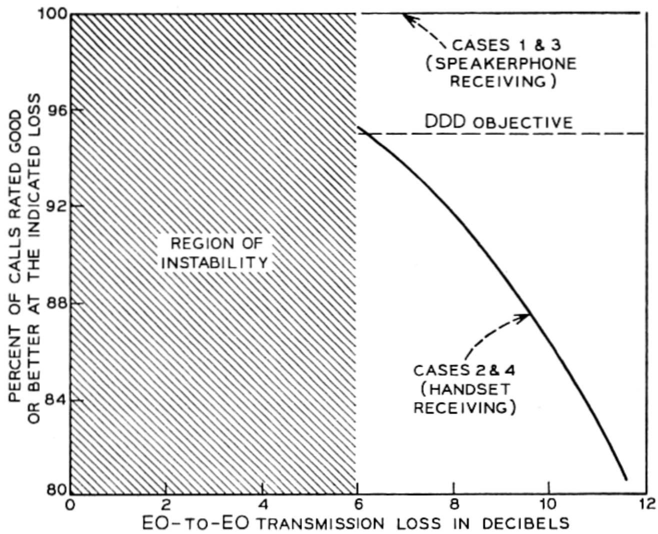

Fig. 10—Received volume performance as a function of the transmission loss.

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[23,24] 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[25] (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.

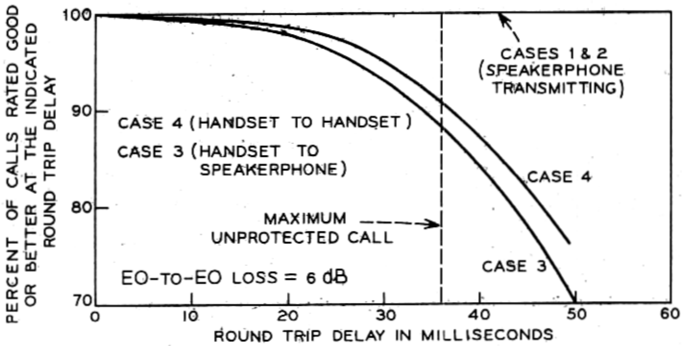

Fig. 11—Echo performance as a function of the round-trip delay.

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.[28] 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 Connections	% 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%*,†
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.

† 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.

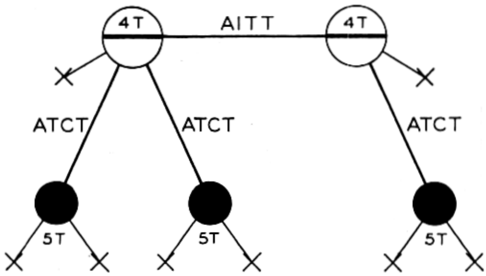

Fig. 12—Audio transmission in the analog-only video network. (Notes: X, *Picturephone* station; ———, analog transmission facility; ATCT, analog toll-connecting trunk; AITT, analog intertoll trunk; 4T,5T, telephone switch at *Picturephone* Class 4,5 offices.)

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-

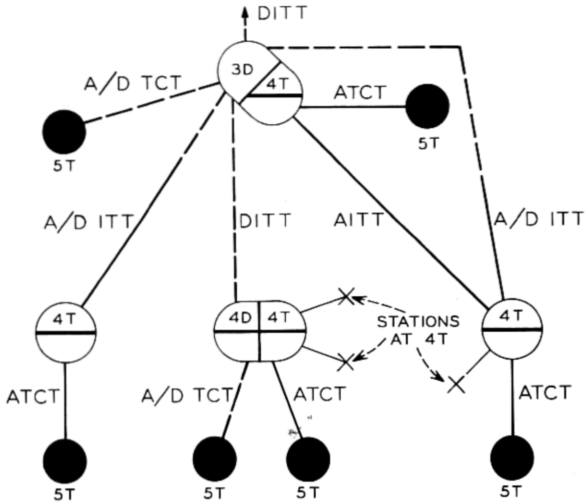

Fig. 13—Audio transmission in the combined analog and digital *Picturephone* hierarchy. (Notes: ———, analog transmission facility; - - - - -, digital transmission facility; ATCT, analog toll-connecting trunk; A/D TCT, toll-connecting trunk, terminating analog at 5T, digital other end; AITT, analog intertoll 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:

(*a*) 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.

(*b*) 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.

(*ii*) As noted in Section 4.2, the loss for any A/D TCT − A/D TCT connection is 6 dB.

(*iii*) 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.

(*iv*) 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

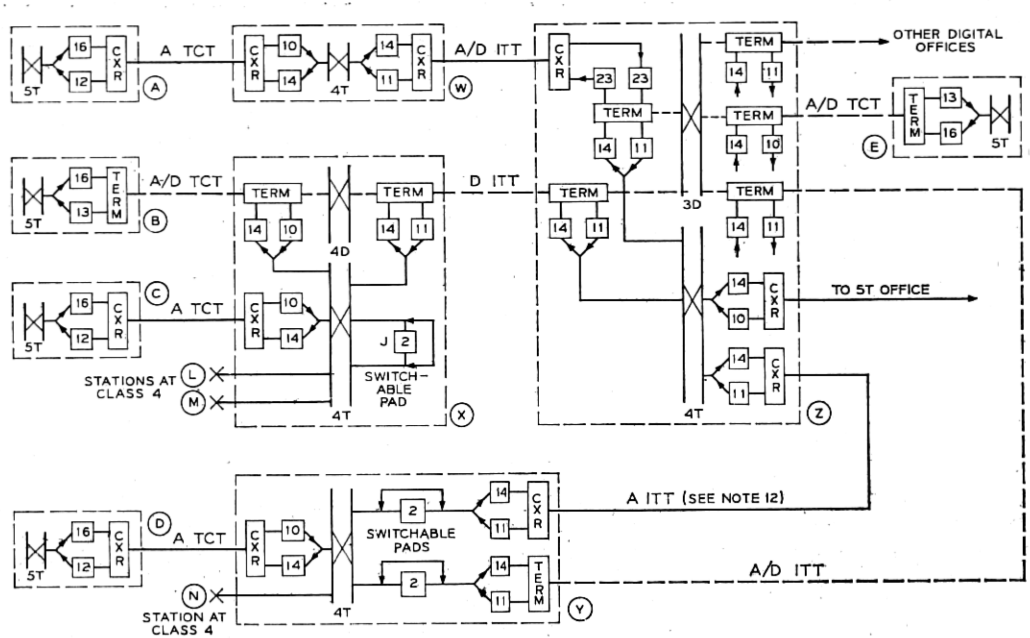

Fig. 14—Audio loss plan. (Notes: 1. ———, analog transmission. 2. - - - - -, digital transmission. 3. 3D,4D, denote bitstream switches which switch the audio in digital form multiplexed with video. 4. 4T,5T, denote audio (POTS) switching networks to which the associated wideband analog and/or bitstream 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. $\boxed{16}$ 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	ATCT	A/D TCT	Station at 4T
ATCT	8	7	7
A/D TCT	7	6	6
Station At 4T	7	6	6

Connection Path (refer to Fig. 14)

	ATCT	A/D TCT	Station at 4T
ATCT	C−XX*−Z−Y−D	D−Y−Z−X−B	D−Y−Z−X$\binom{\text{2dB}}{\text{pad}}^{\dagger}$X*−L
A/D TCT		B−X−Z−E	B−X−Z−$\binom{\text{2dB}}{\text{pad}}^{\dagger}$Y−N
Station at 4T			L−X$\binom{\text{2dB}}{\text{pad}}^{\dagger}$X*−Z−$\binom{\text{2dB}}{\text{pad}}^{\dagger}$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$\binom{\text{2dB}}{\text{pad}}$*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.

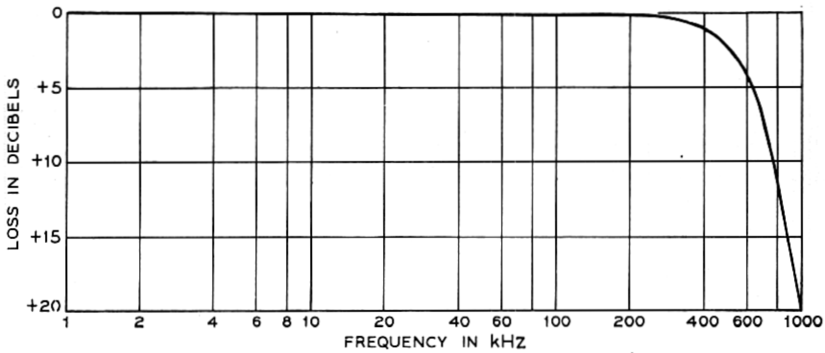

Fig. 15—Station set receiver rolloff—without de-emphasis.

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[2] 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.

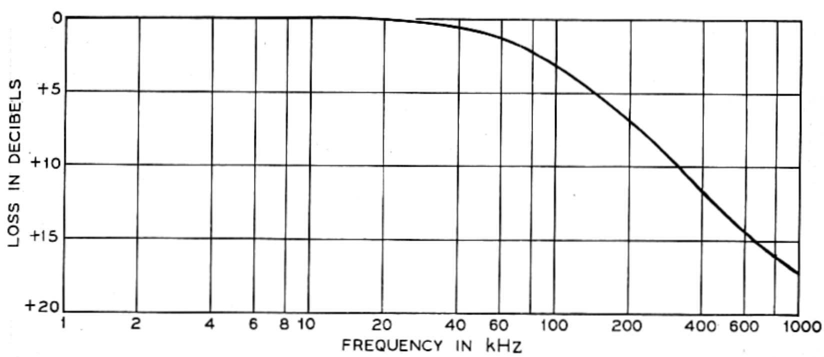

Fig. 16—De-emphasis characteristic.

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

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

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

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

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.[12] 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-

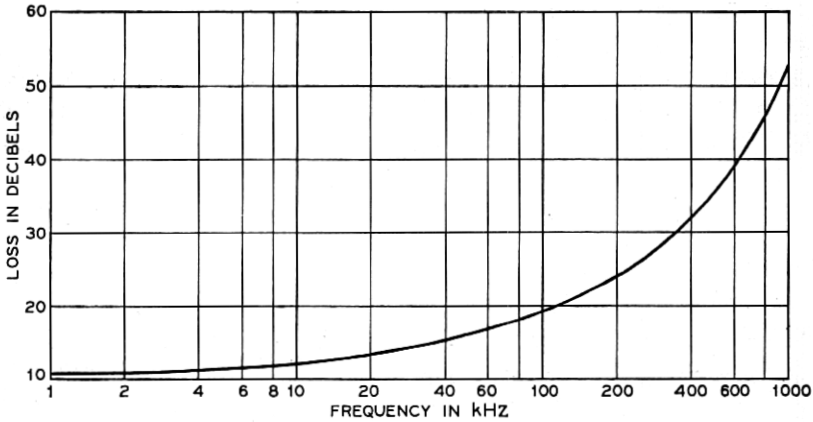

Fig. 17—Cable loss versus frequency—6000 feet of 26-gauge pulp.

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.[19] 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[19] 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.

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