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A Laboratory System for Measuring Loudness Loss of Telephone Connections

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Transmission performance of the telephone message network has improved steadily over the years. Coincidental with this improvement has been the evolution of rating plans which provide the basis for transmission planning, design, and evaluation. Loudness of telephone speech was an important consideration in this evolution.

Telephone speech loudness continues to be one of the major factors which need to be taken into account in telephone transmission engineering. This paper covers a laboratory system, called EARS (*Electro-Acoustic* Rating System), *devised to make objective measurements of partial and overall telephone connections (including electro-acoustic transducer efficiencies) in a manner which reflects subjective loudness loss. Topics covered include a historical review of rating plans, computation of speech loudness, evolution of the* EARS, *and description of the system and its capabilities.*

The EARS *essentially comprises a sound source and a meter for measuring either acoustical pressure or electrical voltage. Design of the source and meter are based on telephone speech loudness considerations, and measurements made with the system approximate subjective loudness judgments with an accuracy which is sufficient for telephone engineering purposes.*

The EARS *may be used in implementing* any *telephone transmission rating plan incorporating speech loudness loss as an element. In such application, the system can be used for specifying connection losses, thus eliminating the need for extensive subjective tests of loudness. However, subjective tests will still be required to evaluate effects of other elements, e.g., noise, important in any given plan, and to evaluate the range of loudness losses acceptable for that plan.*

I. INTRODUCTION

A basic Bell System objective—to provide our customers with the best possible telephone message transmission consistent with the state of the art and the economic climate—has remained essentially unchanged since the early days of telephony. However, continuing review of the telephone system in terms of this objective has resulted in steady improvement in transmission performance of the system over the years. Such improvement has been made possible by growth in our technical skills; it has been made necessary by evolving customer needs for improved transmission. Indeed, it has been postulated that as our customers use the telephone, they become accustomed to current performance and come to expect further improvement.[1,2]

What do we mean by telephone message transmission performance? In the broadest sense, this refers to the effect of the system on speech signals when these signals are transmitted over telephone connections. Customers conversing over telephone connections want to hear reasonably faithful, undistorted reproductions of each others' voices with a minimum of effort. Connections for which these conditions pertain can be thought of as providing satisfactory transmission performance. Connections exhibiting severe distortion would thus provide something less than satisfactory performance; customers might be able to converse but only with extreme difficulty.

Speech transmission capabilities of the telephone network are often considered in terms of individual transmission parameters, the combination of which determines overall transmission performance. Some of the more important parameters are loss, amplitude distortion, and unwanted interferences such as noise, crosstalk, and echo. Improvement in performance over the years has been achieved by design to control these parameters, singly and in combination, as dictated by the technology, economics, and needs of the times.

Coincidental with this improvement in transmission performance has been the evolution of telephone transmission rating. The problem of rating, that is evaluation and measurement, has been the subject of much thought and work over many years, and a number of different rating plans evolved to meet the transmission design needs for an improving and expanding telephone message network.

For present purposes, a transmission rating plan comprises (*i*) a rating criterion which represents the basis for rating telephone connections, (*ii*) a reference system (may be a physical simulation of an overall telephone connection, a set of definitions, or both) with some adjustable feature in terms of which connection ratings are established based on the selected criterion, and (*iii*) a rating scale which is essentially represented by the variable feature of the reference system. Thus, a rating plan provides a framework for the design and evaluation of telephone connections.

Historically, the rating criterion was subjective in nature, and extensive subjective testing was required to evaluate telephone connections in terms of the reference. Moreover, application of the various rating plans required subjective testing to evaluate scale values, i.e., determine what scale values (and connections) represented acceptable performance.

Transmission performance of the telephone message network has improved to a point that loudness loss is a major variable which has to be taken into account in transmission planning. This paper describes a laboratory rating system, called the EARS (*E*lectro-*A*coustic *R*ating *S*ystem), which can be used to objectively measure loudness loss of telephone connections in a manner which closely approximates subjective loudness judgments. Thus, the system supplants subjective tests which would ordinarily be required to determine loudness ratings of connections. However, tests will still be needed both to determine subjective reaction to various amounts of loudness loss, and to determine interrelationships between loudness loss and other transmission parameters important in any given rating plan.

The EARS was devised to measure acoustic pressures and electric voltage as required by loudness rating definitions for partial and overall telephone connections.[3] The EARS has been used extensively over the past several years to characterize the loudness performance of telephone sets and connections, and to evaluate design plans such as unigauge design of the customer loop plant.[4] Moreover, loudness loss as measured with the EARS is an important element in current studies of transmission planning based on a multiparameter approach in which other transmission factors, e.g., noise and echo, are included with loudness loss.[5] Also, the EARS concept is one of several candidates for adoption as a standard method of specifying loudness loss.*

The EARS essentially comprises a sound source which is used to energize the talking end of a telephone connection and an indicating meter which is used to measure acoustic pressure or electrical voltage, and provides a simple means of measuring input and output signal levels for partial and overall telephone connections. Loudness losses

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* Methods of measuring the loudness loss of telephone connections are currently being studied by (*i*) Study Group XII of the CCITT (Comite Consultatif International Telegraphique et Telephonique—International Telegraph and Telephone Consultative Committee) as outlined in Question 15/XII of Ref. 6 and (*ii*) the Task Force on Telephone Instrument Testing of the Institute of Electrical and Electronics Engineers.

of connections and connection components, including electro-acoustic transducers, i.e., telephone transmitters and receivers, are then the differences between input and output signal levels expressed in dB-like terms relative to appropriate reference signal levels.

The approach followed in this paper is first to consider the history of transmission rating plans and, second, to discuss the evolution of the EARS. Discussion of rating plans in Section II demonstrates the interrelationships of rating plans, rating systems, and network transmission performance improvements. Concluding remarks outline the relation between loudness rating definitions and the rating system, the EARS, which is used to determine parameter values as required by the rating definitions. Also mentioned are current studies concerned with multiparameter network design and evaluation.

Discussion of the evolution of the EARS involves several steps. We begin in Section III by addressing ourselves to a review of various techniques for computing the loudness of tones, noise, and speech, then discuss in some detail the derivation of a particular speech loudness computation method. This discussion indicates the manner in which frequency response characteristics of partial and overall connections can be measured and shows how, from the measured response for a given connection, we can compute a number which we will call loudness. Also included is a comparison of computed and experimental results.

Section IV covers derivation of the EARS from the speech loudness computation method referred to above and describes the EARS in its present form. Also described is a graphical computation method based on the design concepts leading to the EARS.

In Section V, we discuss the accuracy of the EARS in predicting subjective test results and consider the effects of simplifying assumptions employed in deriving the EARS. The vehicle for this is the graphical method referred to above. Results computed using this method are compared to observed results for the subjective tests used in validating the speech loudness computation method. This approach is necessary since the subjective test systems are no longer available, and hence could not be measured directly using the EARS. Therefore, the comparison of computed and observed results reflects the accuracy of the concepts on which the EARS is based, and not of the EARS itself.

As will become evident, the EARS in its present form is only one of several ways in which the computational method could be implemented as a laboratory measuring system, and is not an exact realization of the computational method. This form was selected because it provides a satisfactory combination of simple equipment arrangements and suitable calibration and measurement procedures.

II. EVOLUTION OF TRANSMISSION RATING PLANS

Historically, transmission rating plans depended on (*i*) the use of reference circuits with which commercial circuits could be compared and (*ii*) systems of units for expressing the relative ratings thus determined. A great deal of subjective testing was necessary to determine ratings of commercial telephone equipments and facilities, and to correlate these ratings with measurable transmission properties. These ratings, in terms of the appropriate prevalent unit for expressing them, were used in design and evaluation of the telephone message network.

Considerable work was also necessary to determine transmission objectives for the network. That is, the reference system and related units provided the means of expressing transmission performance; it was then necessary to determine the rating, or range of ratings, to which the plant should be designed.

2.1 *Transmission Equivalent Plan*

For a time following the turn of the century, telephone circuits in commercial use had quite similar characteristics, and conditions were such that essentially only loudness, or volume, capability was under control of the transmission engineer. The performance of circuits was determined by comparing them, on a loudness basis, with a reference circuit which was adjustable in attenuation, but whose other transmission characteristics were typical of commercial circuits.

The reference circuit in use at that time was known as the Standard Cable Reference System (SCRS).[7–9] It consisted of telephone sets (including transmitters and receivers), cord circuits (for supplying the telephone set carbon transmitters with operating current), and an adjustable artificial line for interconnecting the cord circuits. These connection components were representative of types then used commercially.

The rating of telephone circuits by comparing them to the reference circuit involved a talker speaking alternately over the test circuit and the reference circuit, and a listener switching similarly at the receiving ends. The reference circuit artificial line, calibrated in terms of "miles of standard cable," was adjusted until the listener judged the volume or loudness of the speech sounds reproduced by the two circuits to be equal. The number of miles of artificial line in the reference circuit was then used as the "transmission equivalent" of the circuit under test.[7] The effect of any change in a test circuit on the efficiency of that circuit could then be measured by determining the "transmission equivalent" before and after the change; the number of miles of artificial line required to compensate for the change was used as an index of this effect.

Performance ratings assigned in terms of loss of speech volume or loudness based on the comparison procedure outlined above constituted a practicable and effective means of assessing transmission performance. The adverse effects of sidetone, distortion, and noise on transmission quality were recognized, but no way was known of incorporating them, together with loudness loss, into a single figure of merit for rating transmission performance.[3] Moreover, the similarity between the reference circuit and commercial circuits at that time rendered such incorporation unnecessary.

As the state of the telephone art developed, modifications in the reference circuit became desirable in order for the circuit to more satisfactorily fulfill its purpose. Three factors leading to the modifications were (*i*) improved designs of telephone instruments and circuits, (*ii*) improved measuring techniques and instruments, and (*iii*) the need for a more suitable unit than the "miles of standard cable."

The effect of improved design was to introduce into the plant telephone instruments and circuits which had less distortion than corresponding parts of the Standard Cable Reference System. For this reason, it became desirable to have a new reference system with which transmission over the most perfect telephone circuit, or over circuits less perfect, could be simulated at will.[8,9]

Secondly, the performance of the Standard Cable Reference System was specified by stating the kinds of apparatus and circuits used. The electrical portion of the system could be checked by voltage, current, and impedance measurements. However, for the transmitters and receivers, reliance for constancy of performance was placed primarily upon the careful maintenance and frequent cross comparisons (subjective) of a group of transmitters and receivers which were specially constructed to reduce some of the sources of variation in the regular product instruments.[8,9] Improvements in measuring techniques and instruments made it possible to measure objectively characteristics of electro-acoustic transducers. This, in turn, permitted selection and maintenance of transducers to provide long-term stability.

Finally, a change in rating units became desirable for two reasons. First, circuits were being designed which had less distortion than the artificial cable of the Standard Cable Reference System. Since a mile of cable with this system corresponded not only to a certain volume change, but also to a distortion change, characterizing new circuits in terms of miles of standard cable implied a distortion degradation not necessarily attributable to the circuit under test. Secondly, two different reference systems were in common use, one in the United States, the other in some other countries.[10] These systems used artificial cable with different characteristics. That used in the United States had a loop resistance of 88 ohms and a capacitance of 0.054 microfarad per mile; the other used artificial cable having the same resistance and capacitance but had, in addition, an inductance of 1 millihenry and a conductance of 1 micromho per mile. Thus, a test circuit compared with each of these references would have two different ratings assigned to it. The first of the above reasons suggested the need for a distortionless unit, the second for a common unit.

A new unit, called the Transmission Unit (TU), was devised by the Bell System.[10*] This was a distortionless, logarithmic unit so chosen as to make use of common logarithms convenient in transmission computations. Its magnitude was very nearly the same as the loss of a mile of standard cable and, thus, existing experience learned in terms of miles of standard cable could be transferred to the new system with a minimum of difficulty. The Transmission Unit later became the decibel (dB).[11]

The three factors discussed in preceding paragraphs resulted in design of the Master Reference System (MRS).[8,9] This system utilized transducers with very low distortion, amplifiers to compensate for the lower efficiency of these transducers as compared to commercial instruments, and an artificial line (consisting essentially of a 600-ohm attenuator) for interconnecting the transmitting and receiving elements. The system included provision for inserting distorting networks into the transmitting and receiving elements so that performance of

* An intervening step between the "Mile of Standard Cable" and the "Transmission Unit" was the "800 Cycle Mile." This also was a distortionless logarithmic unit, equal in magnitude to the loss of a mile of standard cable at 800 Hz (actually, the loss at 796 Hz was used). There were thus two "800 Cycle Mile" units because of the two different standard cable specifications. The "800 Cycle Mile" and its successor, the "Transmission Unit," differed in that (*i*) the former represented a current ratio while the latter represented a power ratio and (*ii*) although both were logarithmic, the former was in units of log 1.115 (one 800 Cycle Mile corresponded to a current ratio of 1.115) while the latter was in units of 0.1 log 10 (one Transmission Unit corresponded to a power ratio of $10^{0.1}$).

commercial transducers, characterized by amplitude response curves with pronounced resonances, could be simulated.

Transmission engineering was still done on a loudness basis, and the MRS was used to obtain ratings in the same manner as that previously described for the SCRS. That is, (*i*) speakers talked alternately over the reference and test systems, (*ii*) observers listened over the systems, switching with the talkers, and (*iii*) the line (600-ohm attenuator) of the MRS was adjusted to obtain equal loudness. The average setting of the attenuator (in dB) at balance was then the rating of the system under test.

Adoption of the MRS by the Bell System had important long-term effects on international standardization of telephone reference systems. In 1926, the CCI* invited representatives of the Bell System to meet with a committee appointed by the CCI to consider adoption of a transmission reference system.[9] At the recommendation of this committee, the CCI adopted the MRS. Two of these systems were built by the Bell System. One was retained at Bell Telephone Laboratories in New York, the other was sent to the laboratory of the CCI in Paris in 1928. The CCI System was designated the SFERT.†

In the late 1950s, the SFERT was replaced by a new reference system, the NOSFER.‡ The NOSFER is so constructed and calibrated that it is essentially equivalent to the SFERT. Telephone connection ratings obtained with either the SFERT or the NOSFER are designated as Reference Equivalents (RE) in dB, and are numerically equal to the setting (in dB) of the reference system line attenuator required for loudness balance.

2.2 *Effective Loss Plan*

In the early 1930s, the Bell System adopted a new transmission plan. The change became necessary because of technological advances. Telephone sets incorporating antisidetone circuitry and improved transducers began to be used in quantity in the Bell System. These sets had poorer loudness performance than their predecessors, but

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* CCI = Comité Consultatif International des Communications Telephoniques a Grande Distance (International Consultative Committee for Long Distance Telephone Communication). This organization later became the CCIF and is now the CCITT.

† SFERT = Systeme Foudamental European de Reference pour la Transmission Telephonique (European Fundamental Reference System for Telephone Transmission). See Ref. 12.

‡ NOSFER = Nouveau Systeme Foudamental pour la Determination des Equivalents de Reference (New Fundamental System for the Determination of Reference Equivalents). See Ref. 6.

provided marked improvement in such characteristics as sidetone, amplitude distortion, and nonlinear distortion, all of which have a marked effect on transmission. Evaluation of these telephone sets in terms of the old system tended to emphasize loudness differences at the expense of these other factors. Thus, there was need for a rating plan which properly recognized the improvements and, at the same time, retained applicability for the older telephone set types. The effective loss plan was devised to meet this need.[13,14]

Ratings under the new plan were in terms of dB of "effective" loss to distinguish them from "loudness" or "volume" losses of the old plan. Effective loss represented a figure of merit for evaluating the effectiveness of the transmission over telephone circuits and, as such, was a measure of the ability of telephone listeners to understand as well as to hear transmitted telephone speech.

Effective transmission data were determined in terms of the Working Reference System (WRS).[14] This system consisted of representative customer loops and telephone sets and a variable, distortionless trunk, i.e., an attenuator. Ratings for commercial connections were obtained in effect by comparing these, using live talkers and listeners, with the reference system, adjusting the distortionless trunk until the reference system and test connection were judged to provide equivalent transmission. (Basic data for the plan were obtained from two-way conversation tests, in contrast to the earlier plan for which tests were on a listening-only basis.) The criterion for such equality was repetition rate, that is, the rate of occurrence of repetitions requested by test subjects.[13,14] Thus, balance was achieved when the WRS (with a particular trunk setting) and the test connection provided the same repetition rate. The WRS trunk setting, in dB, was then a measure of the effectiveness of the circuit under test. The reference trunk setting for the Working Reference System was selected to provide numerical equivalence of ratings obtained with the new plan and its predecessor. With reference trunk setting, the Working Reference System had an effective loss of 18 dB which was also its transmission equivalent in terms of the earlier plan.

During the early 1930s when the effective loss plan was under development, in-plant telephone set transducers were predominantly of the resonant type, characterized by large amounts of amplitude and nonlinear distortion, telephone sets were largely of the sidetone variety, and a number of low cutoff cable loading systems were in use. However, improved transducers and antisidetone sets were being introduced into the plant, and additional improvements were under development. (The transitional nature of the telephone set plant at that time and additional planned improvements were important considerations leading to adoption of the effective loss plan.) This resulted in introduction of the 300-type telephone set into the plant in 1937.[15–17] Subsequent study similarly resulted in introduction of the 500-type telephone set into plant in 1950.[18,19] The effective loss plan encouraged these improvements and, also, the adoption of higher cutoff cable loading systems.

By the mid 1950s, telephone sets provided sufficient antisidetone and freedom from amplitude and nonlinear distortion that little could be gained by further improvements. Thus, loudness appeared to be once again a major variable factor in plant design, likely to be the most important consideration in subsequent telephone set design.

The uniformly high level of transmission performance characterizing the plant suggested revision of telephone transmission planning to reflect more emphasis of loudness effects. Impetus for such revision was provided by (*i*) the availability of a speech loudness computation technique and a laboratory measuring system based on this technique and (*ii*) the need for planning in simpler terms than was possible with the effective loss plan which suffered from a number of practical disadvantages.[3]

2.3 *Loudness Rating Plan*[3]

For present purposes, the loudness rating plan comprises definitions for loudness ratings of partial and overall telephone connections.* Loudness is a subjective quantity and, therefore, loudness rating is defined as the ratio of the loudness of the speech into the listener's ear to the loudness of the speech out of the talker's mouth. As used here, however, loudness rating is considered to be determined by objective measurements.

The rating definitions involve both acoustic pressures and electric voltages. The EARS provides a means of measuring these quantities. The EARS consists of a source of acoustic energy, comprising a complex voice-frequency test tone which simulates certain properties of human speech and an artificial mouth, and an indicating meter for measuring voltages or sound pressures in a manner which simulates the loudness perception of an average listener.

Ratings obtained using the EARS do not precisely equal ratings

* Loudness ratings as specified by the definitions are expressed in decibels. Such usage does not strictly conform to the definition of the decibel, but this should have no effect on general use of the term "decibel."

obtained using human beings, in part because the artificial mouth and the 6-cm^3 coupler do not precisely duplicate the characteristics of their human counterparts. However, EARS ratings are sufficiently accurate to be highly useful in telephone transmission engineering although some problems associated with using the EARS to measure nonlinear transducers, e.g., carbon microphones, remain to be solved. Continuing research aimed at characterizing human mouths and ears may result in artificial counterparts the use of which in the EARS will render the differentiation between EARS and subjectively determined loudness ratings unnecessary.

The EARS is thus an instrument used to measure the loudness performance of telephone instruments and facilities in much the same manner that the 3A Noise Measuring Set is used to measure the subjective magnitude of telephone message circuit noise.[20] Loudness performance is expressed as constants and/or families of curves which are used in transmission planning. Examples of such use are the unigauge plan already cited[4] and current studies which are contemplating transmission design based on (*i*) noise and loss and on (*ii*) noise, loss, and echo.[5]

Earlier rating plans utilized physical reference systems which simulated overall telephone connections. Loudness rating as considered here is based on measurement of physical quantities, and does not require specification of a standard simulated connection.

III. COMPUTATION OF LOUDNESS OF SPEECH

A number of tests have been conducted to determine the loudness of telephone speech signals. Study of these test results, and of various methods devised for computing the loudness of speech and tones, resulted in a particular speech loudness computational procedure. This procedure, developed some time ago but not previously reported in the literature, is discussed in some detail in the present section.

We begin with a summary of the essential attributes of the procedure with appropriate reference to functions required in the computation of loudness ratings. Following this, several methods for computing the loudness of continuous spectrum sounds are referred to, and the background for the specific computation method of concern herein is reviewed. We then derive the functions required for the computational procedure. Finally, the procedure is used to compute the loudness performance of several laboratory test systems, and the computed values are compared to experimental results.

The computational method is not based on fundamental properties of speech sounds and of the hearing mechanism and, therefore, is probably not generally applicable in determining the loudness of other types of sounds. Rather, the method is based on a "black box" approach in which a defined "speech" signal is applied at the input, and the signal so processed that the output of the black box correlates with subjective experience of speech loudness. Internal operation of the black box is specified, but such is not intended to reflect exactly the specific manner in which the human being processes speech stimuli to obtain loudness although similarities are noted. This approach is used because, although there are a number of theories covering loudness perception, the specific operation of the hearing mechanism in determining loudness is not known.

3.1 *Description of Speech Loudness Computational Method*

The method, depicted on Fig. 1, is based on performing certain operations on the pressure spectrum delivered by a telephone connection to the ear of a listener. In essence, these operations comprise (*i*) dividing the received speech spectrum into a number of different frequency bands, (*ii*) determining the loudness due to each band, and (*iii*) summing across all bands to obtain the total loudness.

The received pressure spectrum consists of (*i*) a reference speech spectrum applied at the transmitting end of a connection modified by (*ii*) the amplitude transfer characteristic of the connection. The compromise spectrum and system response definitions are given on Figs. 2 and 3 respectively.*

Not all of the received spectrum contributes to loudness, i.e., that portion of the spectrum lower in level than the threshold of hearing does not contribute. Account is taken of this by defining a quantity termed effective spectrum which is the received spectrum minus X, the threshold of audibility for continuous spectrum sounds. The X function is shown on Fig. 4.

The effective spectrum is divided into 50 frequency bands selected such that each contributes equally to the total loudness produced by a flat effective spectrum. The frequency limits for the "2 percent" loudness bands can be derived from the function of Fig. 5.

The effective level in each band is then converted to loudness, in loudness units (LU), using the function of Fig. 6, and the loudness

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* The ordinate of Fig. 2 is in terms of dBt. For purposes of this paper, dBt $=$ dB relative to 2×10^{-5} newton/meter^2; 0 dBt $= 20 \log_{10}(2 \times 10^{-5})$.

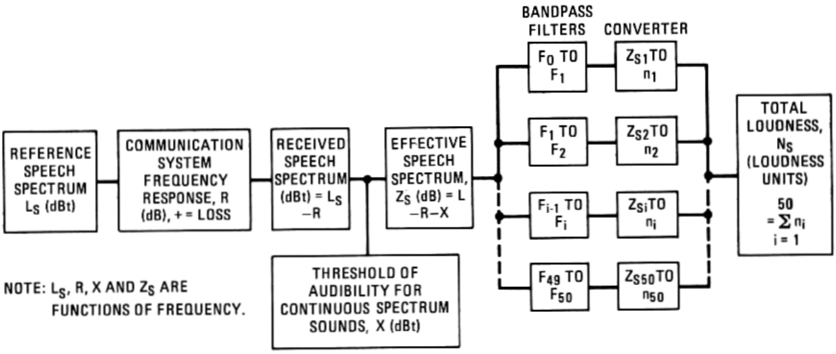


Fig. 1—Computation of speech loudness.

units summed for the 50 bands. The resulting sum, N_s, represents the total loudness contained in the particular received spectrum.

Once the total loudness, N_s, is obtained, the problem of its interpretation arises. One useful way of expressing the loudness is in terms of loudness level.* Functions which may be used to convert N_s to loudness level are shown on Fig. 7.

A more significant way of expressing loudness for speech communication problems is in terms of the level of a reference speech spectrum. The function relating total loudness to the level of the selected reference speech spectrum is given on Fig. 8.

3.2 *Survey of Loudness Studies*

Loudness has been studied extensively as demonstrated by the numerous publications relating to this subject. Studies reported have been largely concerned with single-frequency tones, tone complexes, and continuous spectrum steady sounds. Several different methods for computing the loudness of these different sounds have evolved.† Some of the methods, specifically those covering continuous spectrum

* The loudness level of a sound, in phons, is numerically equal to the median sound pressure level, in dBt, of a free progressive wave of frequency 1000 Hz presented to listeners facing the source, which in a number of trials is judged by the listeners to be equally loud. The unit of loudness is the sone. By definition, a 1000-Hz tone 40 dB above a listener's threshold produces a loudness of 1 sone; the loudness of any sound that is judged by the listener to be n times that of the 1-sone tone is n sones.[21] In this paper, we will use the term loudness units (LU) when considering loudness of speech, and the term sones in discussing other sounds. However, speech of N LU is equal in loudness to any other sound of N sones.

† References 22 through 27 describe some of these computational methods.

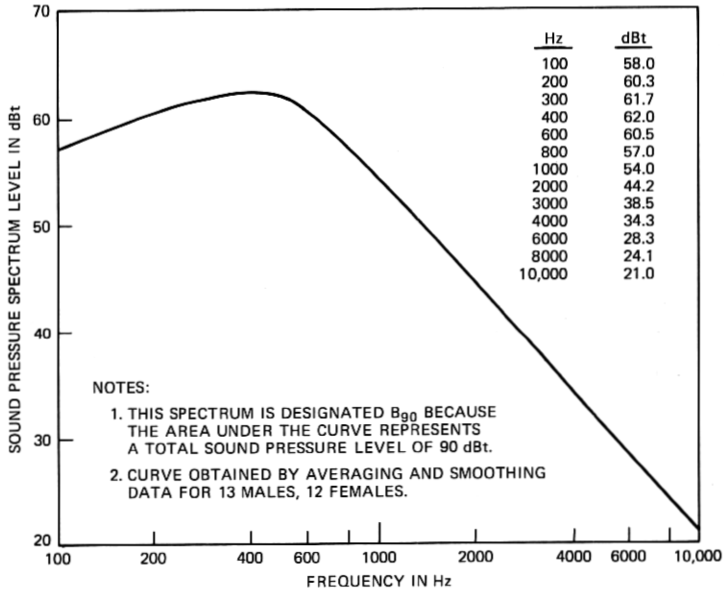


Fig. 2—Sound pressure spectrum of continuous speech at 2 inches from the lips of a talker.

steady sounds, are sufficiently general that with appropriate specification of the speech signal they may be suitable for computing the loudness of speech.* Such suitability has not, to the author's knowledge, been demonstrated and these methods are not considered in this report.

The study of speech loudness has received somewhat less attention than the general subject of loudness. Publications covering the study of speech loudness are limited in number, and of these, only a few consider speech loudness computational methods.†

In 1924, H. Fletcher and J. C. Steinberg devised a method for computing the loudness loss, due to changing the transmission system response, of a sound being transmitted to the ear.[28] Results obtained using the method were in close agreement with experimental results for the specific sounds—speech and a complex test tone—considered in the study.

* See, for example, Ref. 26.

† References 28, 30, 31, 32, and 33 report results of some speech loudness tests. References 28, 29, and 31 describe speech loudness computation methods developed.

In 1925, Steinberg developed a more general method of computing the loudness of any complex sound.[29] (The method referred to in the preceding paragraph was a special case.) Results computed using this method agreed with experimental results then available, including the data reported in the 1924 paper.

In 1938, W. A. Munson developed a method for computing the

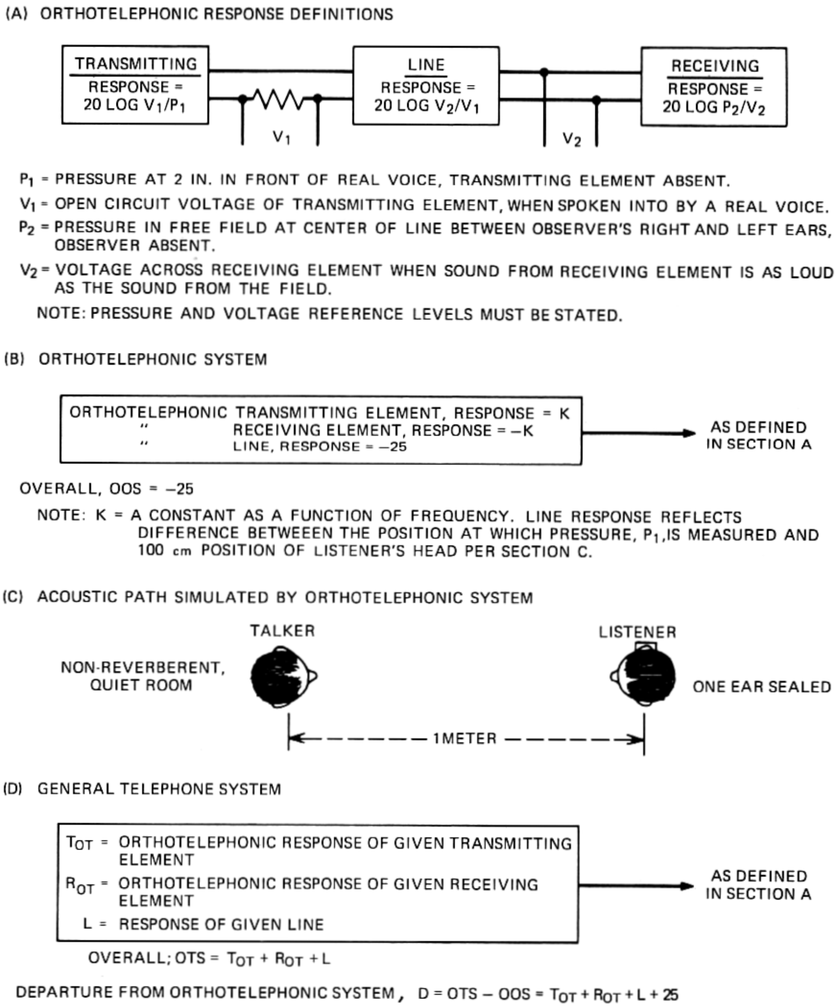
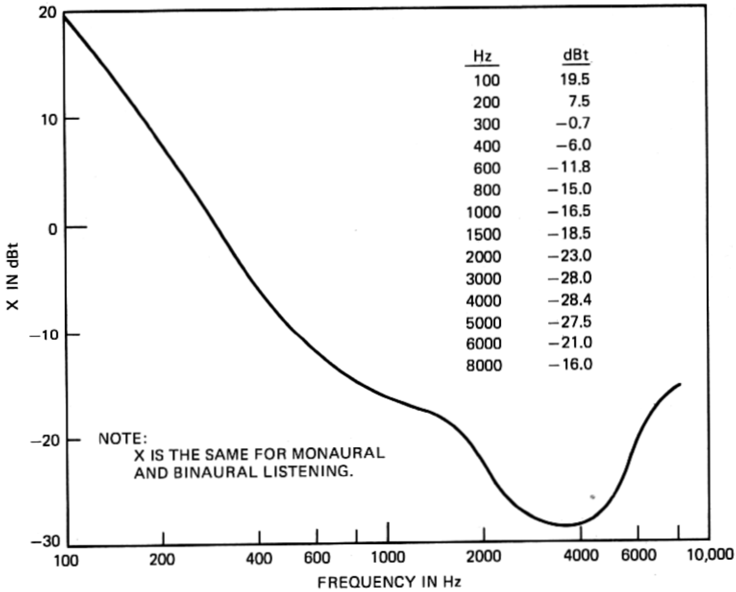
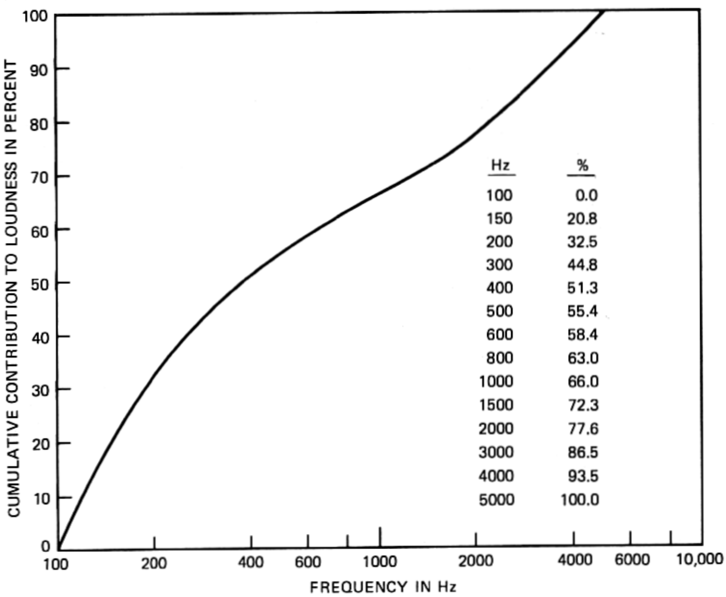


Fig. 3—Orthotelephonic transmission.



Hz	dBt
100	19.5
200	7.5
300	−0.7
400	−6.0
600	−11.8
800	−15.0
1000	−16.5
1500	−18.5
2000	−23.0
3000	−28.0
4000	−28.4
5000	−27.5
6000	−21.0
8000	−16.0

Fig. 4—*X*, the threshold of audibility for continuous spectrum sounds.



Hz	%
100	0.0
150	20.8
200	32.5
300	44.8
400	51.3
500	55.4
600	58.4
800	63.0
1000	66.0
1500	72.3
2000	77.6
3000	86.5
4000	93.5
5000	100.0

Fig. 5—Cumulative contribution to loudness of speech for a flat effective spectrum.

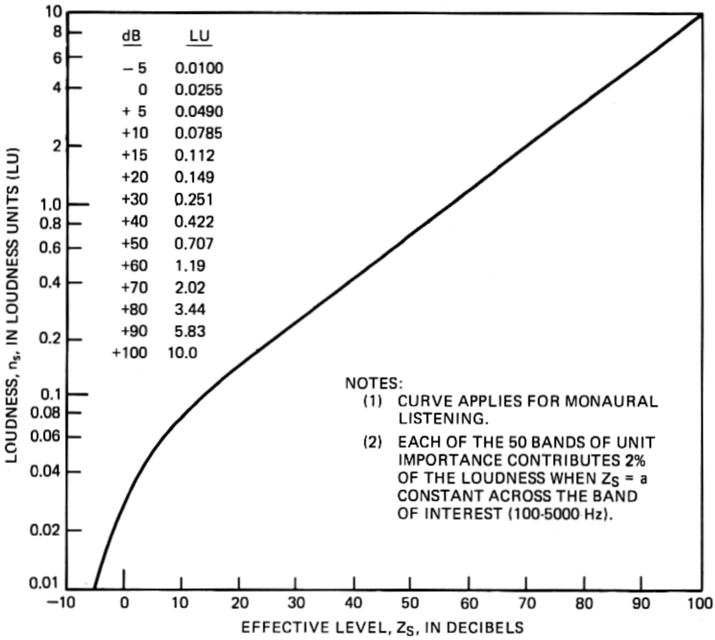


Fig. 6—Loudness (n_s) in a band of unit importance vs effective level (Z_s) in that band.

loudness of speech. The method was based on speech loudness tests conducted in 1935 and 1936.*

Thus, there were available at least two methods, those of Steinberg and Munson, for computing the effects of changes in the response characteristics of telephone circuits on speech loudness. These methods had been found to provide computed results which were reasonably consistent with observed results from experiments on which the methods were based. A review of these experiments revealed substantial inconsistencies between the different sets of data, leading to doubt concerning the generality of the procedures. Moreover, a substantial amount of additional speech loudness data became available in 1939 and 1940. These considerations led to a review of the available speech loudness computation procedures and the various sets of subjective test data for purposes of arriving at a simple procedure, suitable for telephone engineering purposes, which would be reasonably consistent with all of the data.

* Neither the computational procedure nor the tests were reported in the literature.

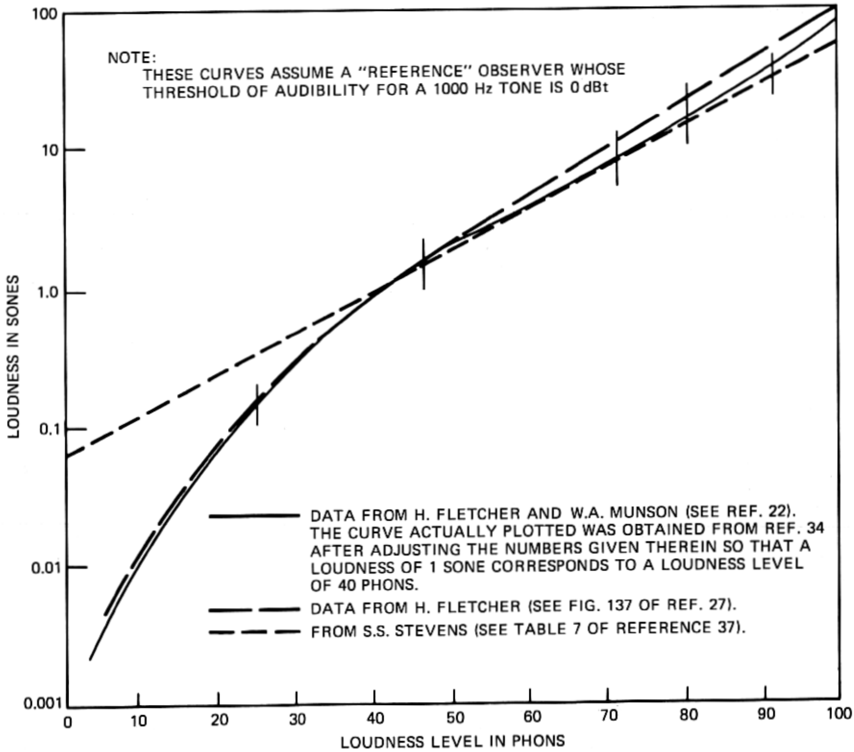


Fig. 7—Relation between loudness and loudness level.

The speech loudness computation procedure resulting from the review referred to above is described in subsequent paragraphs. This procedure, developed in 1941, and the 1939–40 speech loudness data have not been previously reported, primarily for two reasons. First, the group concerned with the speech loudness project was essentially disbanded in late 1941 and assigned other work in support of the war effort. This group was not re-established so that its accomplishments could be reduced to a form suitable for publication in the literature. Secondly, there was no pressing need for the procedure until the loudness rating plan was proposed.[3] Review initiated at that time has verified applicability of the procedure to telephone speech loudness problems.

3.3 *Development of the Speech Loudness Computation Method*

The speech loudness computation method evolved from a specific method devised by Fletcher and Munson for treatment of continuous spectrum sounds, and uses as many of the steady-state definitions as are applicable.[23] Review of the Fletcher-Munson method therefore serves as a convenient introduction.

3.3.1 *Steady-State Sounds—Definitions and Formulae*

Loudness, N, is defined as the intensive attribute of an auditory sensation.[21] Loudness can be expressed in terms of a scale whose units have been found to agree with the common experience of observers estimating the intensive attribute of sounds. Ordinarily, the loudness of a sound is not specified in terms of units of a loudness scale, but, rather, in terms of its loudness level. (See Section 3.1 footnote.)

Results of some experiments to determine the relation between loudness and loudness level are shown on Fig. 7. The solid curve, due to Fletcher and Munson, was first proposed in 1933.[22]

This curve was later adopted in an American Standard.[35] Further study by Fletcher and Munson resulted in a slightly different relation shown by the long dashed line. This curve, first reported in 1937,[23] was later modified slightly.* The short-dashed curve, due to S. S. Stevens, resulted from effort directed at obtaining the best fit for all available data with the simplest possible relation.[26,36,37] This relation is included in a USA Standard.[38]

The loudness, N, for steady-state sounds can be computed from the masking spectrum, M, of the sound.† The loudness equation may be written as

$$N = \int F(M)\, dx. \tag{1}$$

$F(M)$, expressed in loudness units, is a function of the masking, and may be interpreted as expressing the intensity of the nerve stimulation at a particular position, x, along the basilar membrane. The quantity, x, is expressed in terms of the percent of total nerve endings that the maximum stimulation passes over as a stimulating tone is changed from lowest audible frequency to a frequency f, corresponding to x. The product $F(M)$ and dx represents the loudness contributed by a sound within the differential length dx and, consequently, the integration represents the total nerve stimulation, or loudness, over the length of the basilar membrane.

For continuous spectrum sounds, the masking, M,‡ at any fre-

* Figure 137, Ref. 27.

† See Ref. 23 and 39 for more detailed discussion of the concepts considered in this and subsequent paragraphs.

‡ $M = \beta - \beta_0$ where β = the level of a single frequency tone which is just audible in the presence of a specified noise and β_0 = the level of the same tone which is just audible when the noise is absent. (See Ref. 39.)

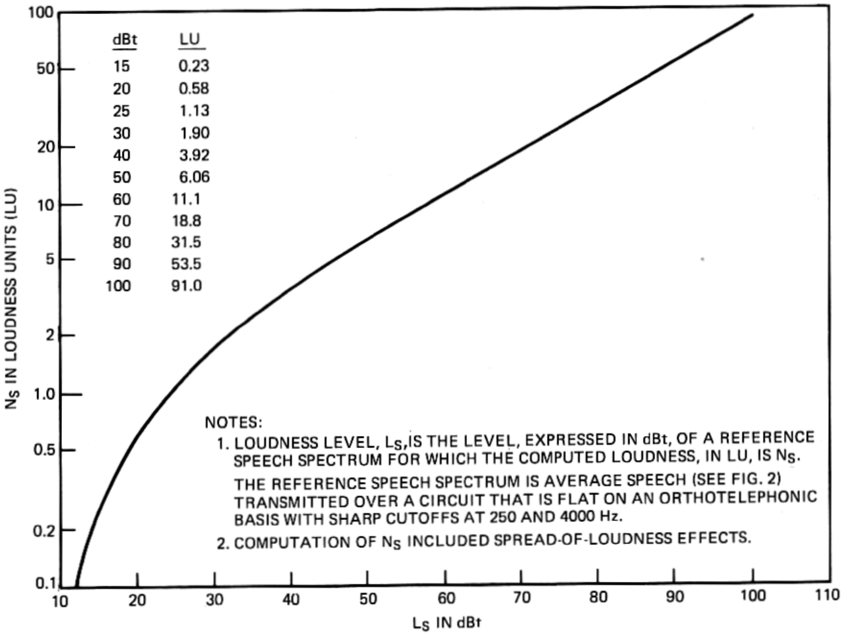


Fig. 8—Loudness of reference speech.

quency is related to the effective level, Z, of the masking sound at the same frequency. Thus, the loudness formula (1) can also be expressed as follows:

$$N = \int Q(Z)\, dx. \tag{2}$$

Interpretation of this formula is the same as that for (1) in that it represents the integrated nerve stimulation over the length of the basilar membrane. However, the intensity of the stimulation at any frequency is determined from the effective level of the continuous spectrum sound, Z, defined as follows:

$$Z = B - X \tag{3}$$

where

Z = effective spectrum level (dB)
B = pressure spectrum level (dBt)
X = a function (dBt), empirically determined from listening tests,

which reflects the fact that not all of the objectively measurable spectrum (B) is effective in producing loudness.

The X function plotted on Fig. 4 may be considered to be the threshold of audibility for continuous spectrum sounds. Although this is not strictly correct, it lends physical interpretation to the function and, as such, facilitates understanding of the loudness computation technique.*

The loudness, N, of a continuous spectrum sound can be computed by (i) dividing the audible frequency range into a number of successive frequency bands conveniently selected for computational purposes, (ii) determining from the effective level and the importance of each band the number of loudness units contributed by the band, and (iii) summing the loudness units over the audible range. This procedure, replacing the integration of (2), may be written as

$$N = \sum^{x} n_x \, \Delta x \tag{4}$$

where

N = total number of loudness units,
n_x = loudness, $Q(Z)$ from equation (2), integrated over a unit length of the basilar membrane, and
Δx = the number of unit lengths along the basilar membrane included in the computation band.†

The quantity, n_x, is a function of the effective level, Z, of that portion of the sound lying in the computation band.

3.3.2 *Speech Sounds—Definitions and Formulae*‡

Development of the speech loudness formula was based on the assumption that a formula of the same general type as (4) could be used, that is, the loudness of speech can be computed by dividing the speech spectrum into a number of successive bands selected for

* Consider the case in which no masking occurs, i.e., $M = 0$. Assuming $M = Z$, then $Z = B - X = 0$, and $X = B$, the pressure spectrum level of a noise which produces zero masking, hence zero loudness. X can thus be defined as the threshold of audibility for continuous spectrum sounds, useful for the purpose indicated, but not strictly correct since it has been found that at low values of M (e.g., $M = 0$), the relationship $M = Z$ is not valid. (See equation (10-5) of Ref. 27.)

† In Ref. 23, the unit length was taken as 1 percent of the basilar membrane length.

‡ Many of the steady-state definitions of Section 3.3.1 apply also for speech. However, since some of the functions differ, the subscript s is used where appropriate to denote those specifically applicable to the speech case.

convenience of computation, determining the loudness contributed by each band, and summing the loudness over the audible range. This can be expressed as follows:

$$N_s = \sum^{s} n_s \, \Delta S \tag{5}$$

where

n_s = total loudness produced by speech in a frequency band of unit importance

ΔS = the number of bands of unit importance included in a computation band.

The quantity, n_s, is a function of Z_s, the effective level of the received speech. Following the definition used in the formulation for continuous spectrum steady sounds, Z_s is defined as follows:

$$Z_s = B_s - X \tag{6}$$

where

B_s = sound pressure spectrum level of received speech (dBt)*

and X is defined in equation (3).

The bands of unit importance for equation (5) are so chosen that no matter where they are located on the basilar membrane, each contributes an equal amount to the total loudness when the effective level of the speech is independent of frequency, i.e., Z_s = k (a constant) across the audible range.† When the speech band is divided into a number of bands suitable for computational purposes, each of these may contain several bands of unit importance. Thus, the loudness, n_s, contributed by a band of unit importance must be multiplied by the number of such bands, ΔS, in the computation band. The value of ΔS used for any particular computation band depends on the width of that band (in Hz) and its location on the frequency scale.

The summation of ΔS along the frequency scale represents the cumulative number of bands of unit importance. The rate of change

---

* Speech varies rapidly in level with time. Accounting for this variability would have entailed development of complex formulae from the limited amount of applicable data available on speech. The approach followed was to develop the simplest possible method of computing speech loudness from the long-term average sound pressure spectrum of speech where long term is understood to comprise a time interval which is long compared to the average syllabic interval.

† Selection of the bands is based on a flat effective spectrum for purposes of simplicity. Any other shape, i.e., $Z_s \neq$ k, would require a different loudness versus Z_s function for each band of unit importance.

of S with frequency may, therefore, be interpreted as showing the relative importance to loudness of equal effective levels of speech in different frequency regions. This interpretation of ΔS and the interpretation of n_s as the loudness of a band of unit importance, based on the similarity of the quantities n_x and Δx in equation (4) and n_s and ΔS in equation (5), are probably not exact because of the radically different nature of speech and steady-state sounds. These interpretations should, therefore, be applied with caution, and regarded primarily as a point of view helpful in systematizing the basic data and in understanding the loudness formulations.

3.3.3 *Received Speech Spectrum*

For any given telephone connection, the sound pressure level of the received speech [B_s of equation (6)] and, hence, the loudness of the speech depends on (*i*) the speech spectrum applied at the talking end of the connection and (*ii*) the loss of the connection as a function of frequency. These factors are taken into account in the speech loudness computation method by assuming a compromise speech spectrum and expressing the connection loss in orthotelephonic terms.

3.3.3.1 *Compromise Speech Spectrum.* The compromise spectrum adopted is shown on Fig. 2 and is designated B_{90} because the area under the curve integrates to a total sound pressure level of 90 dBt.* This spectrum was obtained by averaging and smoothing long average power measurements of continuous speech (including pauses between words) from 13 males and 12 females.† Although some differences in spectral content were found between individual voices, the shapes of the spectrum curves were sufficiently alike to justify averaging across all voices. Moreover, measured spectra from other studies exhibit similar shapes.[41–43]

Several experimenters have reported measurements of the long-term average sound pressure level of speech. Results of these experiments, referred to a point 2 inches in front of a talker's lips, are given in Table I. (Selection of the 2-inch reference point is discussed in the next section.)

The values are in fair agreement excepting the last row. Benson and Hirsh considered the apparent discrepancies at some length, concluding that values reported by Dunn and White and by Rudmose,

* This spectrum is the same as that used in computing articulation index. See Fig. 2 of Ref. 39.

† Spectrum curves reported in Ref. 40 provided some of the data used in deriving this compromise spectrum.

TABLE I—LONG-TERM AVERAGE SOUND PRESSURE LEVEL OF SPEECH AT 2 INCHES FROM TALKERS' LIPS

Source	Males		Females	
	No.	Pressure (dBt)	No.	Pressure (dBt)
Sivian—Ref. 45	5	91.8	3	90.0
Dunn and Farnsworth—Ref. 44	1	90.4	—	—
Dunn and White—Ref. 40	6	90.8*	5	88.0*
Rudmose, et al.—Ref. 42	7	92.6*	—	—
Benson and Hirsh—Ref. 43	5	81.7*	5	79.7*

* Values reported were converted to apply at 2 inches by assuming an equivalent point source 0.6 cm behind the talker's lips. See Ref. 44.

et al., "approximate more nearly monitoring levels (peaks of energy probably corresponding to vowel sounds) than to an actual average."[43]

There is thus some uncertainty regarding the sound pressure level at 2 inches from a real voice. For present purposes, the level is assumed to be 90 dBt, an approximate average for the majority of the data (specifically excludes the Benson-Hirsh data). As will become evident, the exact value assumed for the level at 2 inches does not significantly affect the speech loudness computational procedure as long as the spectrum shape (see Fig. 2) is not changed.

3.3.3.2 *Orthotelephonic Transmission.* The concept of orthotelephonic transmission is based on relating telephone and face-to-face conversation.[16] Specifically, orthotelephonic transmission for present purposes implies reproduction by a telephone system of speech sounds which are indistinguishable from those received with an air transmission system. Conditions applicable for the latter case are negligible noise and acoustic reflections when listening monaurally at a distance 1 meter from the lips of the talker, the listener facing the talker. An orthotelephonic system is thus one that provides the required orthotelephonic reproduction of the speech sounds.

Application of the orthotelephonic concept to telephone connections is detailed in Fig. 3. The definitions are given here as a matter of convenience, and will be considered where appropriate in later discussion. Typical orthotelephonic amplitude responses of overall telephone connections employing specific telephone set types are given in Ref. 16. Such responses vary widely depending on the telephone set and transducer types used.

One point worth noting is that the orthotelephonic transmitting response definition of Fig. 3 is in terms of the sound pressure level at 2 inches from a talker's lips whereby strict conformance with the orthotelephonic definition would require expression in terms of the 1-meter sound pressure level. This change was made for two reasons: (*i*) the 1-meter reference involves a long acoustic path which is ascribed to the transmitter; (*ii*) measurements at 2 inches are more reproducible since they are less likely to be affected by characteristics of the surrounding environment, e.g., reverberation. However, studies show that under free-field conditions the sound pressure spectra at 2 inches (approximately 5 cm) and at greater distances have about the same shape,[44,45] but differ in absolute level by about 25 dB for the 1-meter and 2-inch positions.[44]

3.3.4 *Derivation of n_s and S Functions*

The speech loudness formula [equation (5)] involves the two quantities n_s and S. As noted earlier, n_s is a function of Z_s and is the loudness due to that portion of speech contained in a frequency band of unit importance; S is a function of frequency which shows the relative importance to total loudness of equal effective levels of speech in different frequency regions. These functions, though somewhat interrelated, may be thought of as providing level weighting and frequency weighting respectively.

Data from three tests were used in deriving these functions. The n_s function was based on tests by Munson the results of which show how loudness varies as a function of received speech level. Data obtained by Steinberg and by Van Wynen were used to derive the S function. These data show the manner in which loudness depends on frequency content of the received speech signal. Other data were available, but were not used because (*i*) there were gross inconsistencies between the data and data from other tests and/or (*ii*) the test system and conditions were not reported in sufficient detail to permit utilization of the test results.

An important factor in deriving the n_s and S functions is Z_s, the effective level of the received speech. [See equation (6).] Effective levels for the test results used in the derivation are shown on Fig. 9. The reference effective spectrum curve, consisting of the compromise speech spectrum, B_{90}, of Fig. 2 minus the threshold function, X, of Fig. 4, is also plotted on Fig. 9. This function, utilized in the speech loudness computation technique later, is assumed to represent the average case for a large number of talker-listener pairs conversing over a telephone system with (*i*) orthotelephonic response and (*ii*) 25

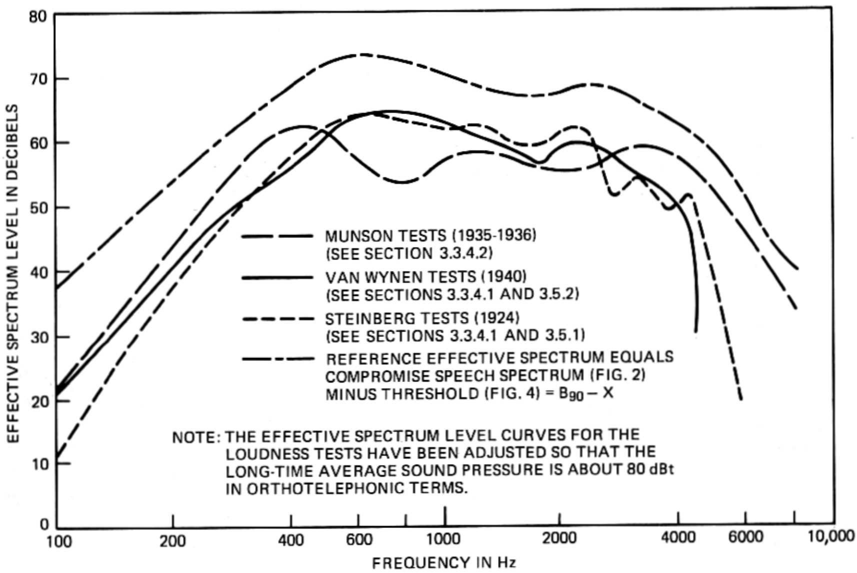


Fig. 9—Effective spectrum level curves.

dB of flat gain relative to a true orthotelephonic system. The function is given here to illustrate the difference between the effective spectrum level curve for a system providing orthotelephonic response and the effective spectrum level curves for the Steinberg and Van Wynen test systems.

Because n_s and S functions were interrelated, their derivations were carried out concurrently by means of a series of successive approximations. For simplicity, the two functions are treated separately in succeeding sections, and the derivation is described in greatly simplified terms. The actual derivation involved obtaining a first estimate of the S function, using the Steinberg and the Van Wynen data (including the respective loudness law exponents—see later discussion), for correcting the actual Z_s to a constant Z_s. This S function was then applied to the Munson data in order to obtain a curve of total loudness versus speech sound pressure level for a 100–5000 Hz band, the band of interest in the speech loudness computation method. The ordinate and abscissa of this curve were then corrected to reflect respectively the loudness in a band of unit importance and the Z_s in a band of unit importance.

These first estimates of the S and n_s functions were used to compute the loudness for a number of conditions. Computed results were compared to experimental results, and the functions modified as indicated by this comparison. This process was continued until computed values reflected adequate accuracy for engineering purposes.

3.3.4.1 *The S Function.* The S function was derived from results of tests by Steinberg (see Section 3.5.1) and by Van Wynen* (see Section 3.5.2). In these tests, speech received through high- and low-pass filters was balanced against undistorted speech. Test results important in the derivation are given in Fig. 10. The intersection of smooth curves drawn through the test results provides two items of information: (*i*) f_{50}, the frequency above and below which the loudness contributions are equal;† (*ii*) LL_{50}, the number of dB by which undistorted speech must be reduced for a 50-percent reduction in loudness.

By assuming a logarithmic relationship between loudness and effective level (see Section 3.3.4.2), we can determine the percentage of loudness contribution as a function of frequency. Thus,

$$Z_{s2} - Z_{s1} = \mathrm{k} \log \frac{N_2}{N_1} \tag{7}$$

where Z_{s1} and Z_{s2} are effective levels before and after a flat change in undistorted speech level, N_1 and N_2 are corresponding loudness numerics, and k is a constant. For the Steinberg tests, we find, substituting -9.1 dB for $Z_{s2} - Z_{s1}$ and 1/2 for N_2/N_1 (50-percent loudness reduction), that

$$\mathrm{k} = \frac{Z_{s2} - Z_{s1}}{\log \dfrac{N_2}{N_1}} = \frac{-9.1}{\log \frac{1}{2}} = 30.2\ dB$$

while for the Van Wynen tests, k = 41.2 dB.

We can express the relationship of equation (7) in the power function form frequently encountered in current literature[33,36] by letting P represent the stimulus magnitude and setting it equal to Antilog $Z/20$. Then

$$N = P^n \tag{8}$$

* Steinberg reports results for sensation levels = 100 dB and 39 dB.[29] (Sensation level is the level above threshold. See Ref. 21.) Only the 100-dB results were used in the derivation. Results for the 39-dB level were near the knee of the loudness curve (see Fig. 8 for example) and their inclusion would have substantially complicated the derivation.

† This assumes ideal filters. The filters used in the tests appear, for speech signals, to be sufficiently close to ideal to warrant making this assumption.

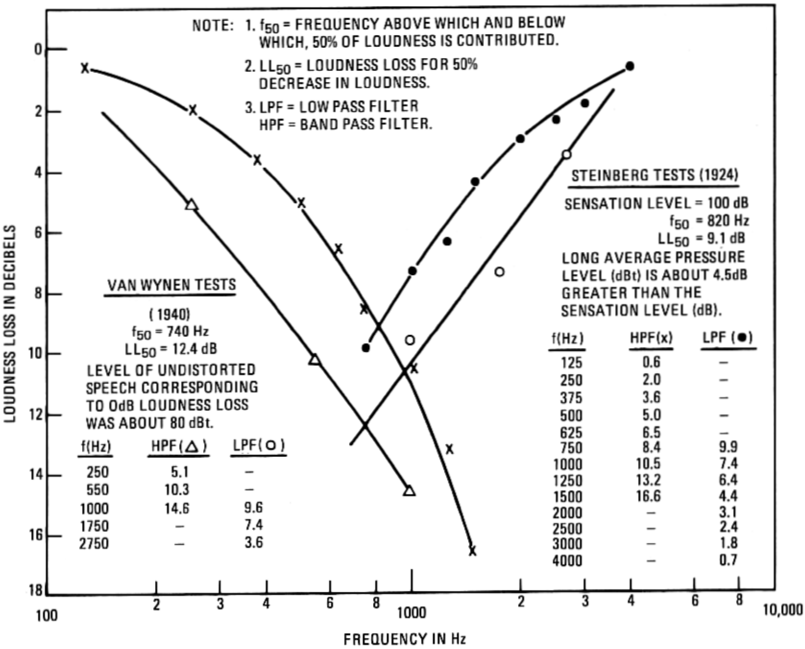


Fig. 10—Filter test results.

where N and P are as defined above and $n = 20/k$. The values of n are then 0.662 and 0.486 for the Steinberg and Van Wynen tests respectively.

By applying equation (7) to the curves of Fig. 10, we can determine the loudness reduction corresponding to various values of loudness loss. For example, the Van Wynen tests show that for a high-pass filter with cutoff frequency = 550 Hz, $Z_{s2} - Z_{s1} = -10.3$ dB. Using the equation, we find that the 550-Hz high-pass filter passes 56.2 percent of the loudness, suppressing 43.8 percent. Proceeding in a similar manner for all of the data, the points shown on Fig. 11 were obtained. It is of interest to note that there appear to be no systematic differences between the Steinberg and Van Wynen test results even though the exponents referred to in the preceding paragraph are somewhat different.

Because the effective spectrum level curves for tests were almost identical (see Fig. 9), the smooth curve of Fig. 11 was drawn to represent all of the data points. This curve shows the percent of total loudness contributed by frequencies below the frequency of the abscissa for the effective spectrum level used in these tests.

The S function required, showing the cumulative contribution to loudness when the effective spectrum level is flat with frequency, is plotted on Fig. 5. The function was obtained from the curve of Fig. 11 by successive approximations as outlined in the introductory paragraphs of Section 3.3.4.

3.3.4.2 *The n_s Function.* The n_s function, relating loudness and effective speech level in a band of unit importance, was derived using test results obtained by Munson in 1935–1936. The test system and general methodology are described in Ref. 46. However, the test results have not previously been reported in the literature.

In the Munson tests, undistorted speech was loudness balanced against a 1000-Hz test tone. Also, thresholds were determined for the 1000-Hz and speech signals. Detailed results of the tests are given in Tables II and III. Average results are plotted on Fig. 12.

The ordinate of Fig. 12 was converted to a loudness scale using the solid curve of Fig. 7. The resulting relation between loudness and

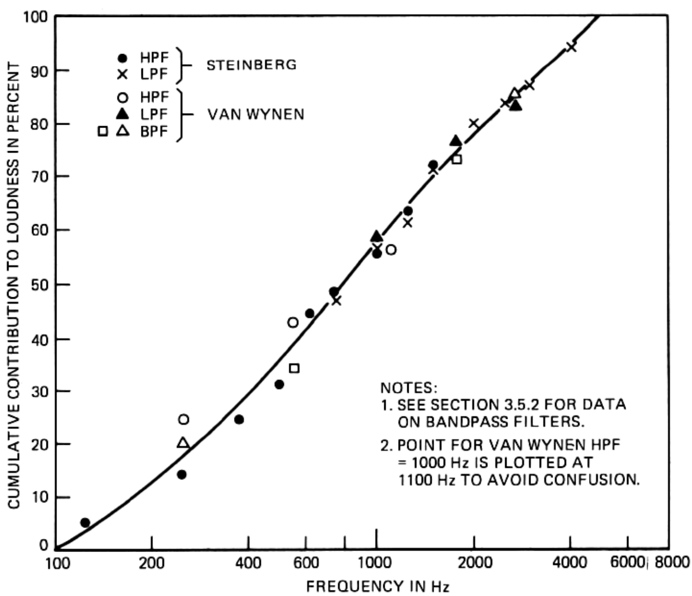


Fig. 11—Cumulative contribution to loudness of speech for the Steinberg and Van Wynen filter tests.

TABLE II—RESULTS OF MUNSON (1935–1936) SPEECH LOUDNESS TESTS

Speech Pressure Level-dBt	Parameter*	Subject Designation											Group†	
		S_1	S_2	S_3	S_4	S_5	S_6	S_7	S_8	S_9	S_{10}	S_{11}	$\bar{\bar{X}}$	S
20.2	$\bar{X}$	29.5	39.1	20.6	29.5	29.1	19.2	24.2	20	20.9	29.6	18.7	25.5	6.4
	σ	4.2	9.2	4.8	12.8	3.4	2.8	8.5	3.3	3.5	5.6	5.1		
	N	6	4	6	6	3	5	4	6	6	4	5		
30.2	$\bar{X}$	54.2	56.3	50.4	52.7	57.1	49.0	41.4	35.9	39.5	43.0	36.9	46.9	7.8
	σ	5.8	5.2	11.4	9.1	9.2	11.8	11.1	3.1	6.7	8.0	8.0		
	N	8	6	8	8	5	7	4	7	8	6	7		
45.2	$\bar{X}$	78.6	76.3	76.5	76.1	76.3	72.3	66.6	61.2	62.9	56.0	60.7	69.4	7.9
	σ	5.8	3.8	5.1	6.8	5.0	6.2	8.7	3.9	3.2	9.0	7.0		
	N	13	12	13	13	7	11	5	12	13	7	11		
54.3	$\bar{X}$	91.5	84.0	86.6	76.5	82.5	86.5		76.5	71.5	55.5	74.5	78.6	10.3
	σ													
	N	1	1	1	1	1	1		1	1	1	1		
60.2	$\bar{X}$	91.5	88.5	90.9	84.6	83.7	85.1	78.8	77.0	75.6	66.2	71.9	81.3	8.1
	σ	3.6	3.8	5.7	4.9	4.0	3.8	5.5	3.7	3.5	9.5	5.6		
	N	10	9	10	9	6	8	5	9	10	8	8		
80	$\bar{X}$	98.2	98.5	103.6	95.9	93.2	100.2	93.3	94.0	91.5	76.9	82.7	93.5	7.7
	σ	4.7	3.9	5.4	3.0	3.2	3.5	1.8	3.8	7.0	6.8	5.9		
	N	8	7	8	7	5	6	5	8	8	6	6		
90.2	$\bar{X}$	107.2	105.2	108.5	99.6	101.5	107.7	100.3	105.0	97.2	78.3	91.9	100.2	9.7
	σ	4.7	3.4	6.5	4.2	9.0	4.5	0.2	2.5	4.0	10.0	4.8		
	N	5	3	5	4	4	5	2	5	5	3	5		

* Each observer participated in N tests, providing $X_1, X_2, \cdots, X_N$ estimates of the 1000-Hz tone level (in dBt) judged to be equal in loudness to the designated speech pressure level.

$$\bar{X} = \frac{\sum^N X_i}{N} \qquad \sigma = \sqrt{\frac{\sum^N (\bar{X} - X_i)^2}{N}}$$

$$\bar{\bar{X}} = \frac{\sum^S \bar{X}_i}{11} \qquad S = \sqrt{\frac{\sum^S (\bar{\bar{X}} - \bar{X}_i)^2}{10}}$$

TABLE III—RESULTS OF MUNSON (1935–1936) THRESHOLD TESTS

Subject Designation	Speech			1000-Hz Tone		
	$\bar{X}$-dBt	σ-dB	N	$\bar{X}$-dBt	σ-dB	N
S_1	12.7	0.8	4	2.2	0	1
S_2	10.3	1.9	4	7.9	0	1
S_3	16.0	3.1	4	5.6	0	1
S_4	16.9	2.0	4	5.9	0	1
S_5	12.4	0.6	3	—	—	—
S_6	12.7	1.2	3	4.3	0	1
S_7	10.4	1.3	2	4.5	0	1
S_8	8.0	2.5	4	2.6	0	1
S_9	12.7	0.3	4	5.7	0	1
S_{10}	11.9	3.6	4	1.2	0	1
S_{11}	11.9	0.9	3	1.6	0	1
	$\bar{\bar{X}}$-dBt 12.4	S-dB 2.5		$\bar{\bar{X}}$-dBt 5.1	S-dB 3.2	

speech sound pressure level is shown on Fig. 13. (The ordinate of Fig. 13 is in terms of Loudness Units—LU rather than sones. See footnote of Section 3.1.)

In terms of equation (8), the curve of Fig. 13 at higher sound pressure levels represents an exponent of 0.455. This is in close agreement with the Van Wynen test results with exponent = 0.486 (see

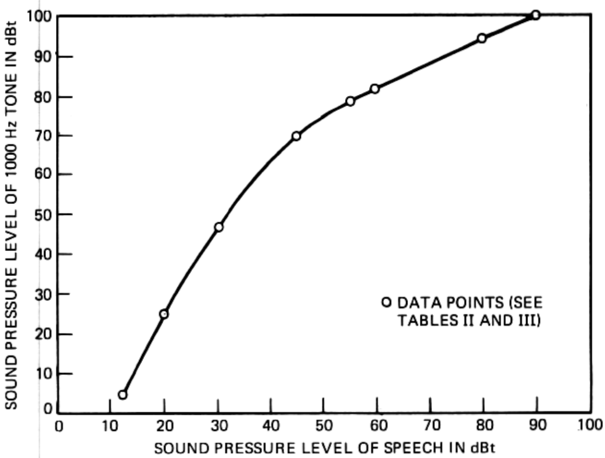


Fig. 12—Loudness balance test results for a 1000-Hz tone and speech (Munson, 1935–1936).

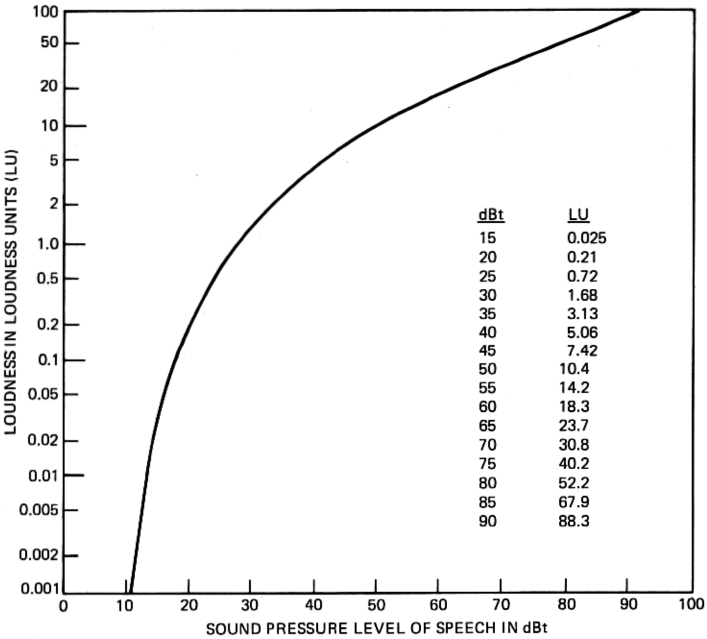


Fig. 13—Loudness (N_s) vs speech sound pressure level for Munson (1935–1936) tests.

previous section), with the "Derived" (from balancing speech against a 1-kHz tone) and "Direct" (from adjusting a speech signal to sound "half as loud" and "twice as loud" as a standard speech signal) from Ref. 32 with approximate exponents of 0.50 and 0.46 respectively, and with the derived curve of Fig. 52, Ref. 31, with an exponent of about 0.51. The exponent does not agree with that from the Steinberg tests which was 0.662 (see previous section), with the exponent of about 0.65 determined from the monaural versus binaural loudness matches of Ref. 32, with the exponent of 0.7 determined from the single phoneme magnitude estimation study of Ref. 33, and with the exponent of 0.6 determined from the limited magnitude estimation work reported in Ref. 47.

The relation of Fig. 13, applicable for wideband speech having the effective spectrum shown by the long-dashed curve of Fig. 9, was then modified to apply for narrow bands of speech, (bands of unit importance) and a flat effective spectrum. To do this, effective spectra were plotted for several of the loudness balance tests, e.g., see Fig. 9. For each of these, the frequency scale was divided into various numbers of bands of equal importance as determined from Fig. 5. It was then assumed that Z_s was constant across each band of unit importance, and was equal in magnitude to the Z_s at the center frequency of the unit band. This resulted in several "staircase" approximations to the smooth curve, each exhibiting a granularity depending on the number of bands selected. Study of the various spectra indicated that using 50 bands of unit importance, referred to as 2-percent loudness bands, represented a reasonable compromise between the conflicting requirements of (*i*) accuracy, improved by increasing the number of bands and thus obtaining a closer match between the "staircase" approximations and the smooth curves, and (*ii*) simplicity, obtained by reducing the number of bands.

The required function of n_s versus Z_s, shown on Fig. 6, was obtained from Fig. 13 by a series of successive approximations as outlined in the introductory paragraphs of Section 3.3.4. Specific steps involved included (*i*) adjusting the ordinate and abscissa scales of Fig. 13 to represent a band of 100–5000 Hz (required making allowance for the fact that the effective level of the speech spectrum used in the Munson tests was not flat with frequency) and (*ii*) dividing the ordinate by 50 (to reflect contribution by each 2-percent band) and by 2 (since the Munson test results and the solid curve of Fig. 7 are both in terms of binaural listening while the case of usual interest in telephony is monaural listening).

3.4 *Computational Procedure*

Mechanics of the computation for any given telephone connection consists of two parts, (*i*) the computation of the loudness, N_s, of the received speech, and (*ii*) interpretation of the computed loudness numbers. The former is carried out utilizing a computational form based on Figs. 5 and 9.

The computational method to be described applies specifically to the monaural case, of primary interest in telephone problems. If speech is received binaurally, we assume that the "average" listener is symmetrical and compute the loudness separately for each ear, adding the resulting loudness numbers to obtain the total loudness. [See equation (5).] Thus, in cases where the speech signals delivered to the two ears are identical, the resulting loudness is twice that obtained with one ear alone.

3.4.1 *Computation of Speech Loudness, N_s*

The procedure followed in computing speech loudness is most readily described in terms of the computation form shown in Fig. 14. The computation involves four steps:

(1) Calculation of the received speech signal effective level, Z_s, at a number of selected frequencies;
(2) Determination of n_s, the loudness in a band of unit importance, from Z_s;
(3) Determination of the loudness in a computation band, the product of n_s and ΔS;
(4) Summation of the loudness numerics across the band of interest.

Column 1 of Fig. 14 lists the frequencies at which the computations are made and columns 2 through 7 are used for determining Z_s. The values entered in column 2, obtained from Fig. 9 (reference effective spectrum), apply for an acoustic talking level of 90 dBt.

Columns 3, 4, and 6 are respectively: T_{ot}, the orthotelephonic response of the transmitting element; R_{ot}, the orthotelephonic response of the receiving element; and L, the response of the circuit interconnecting the transmitting and receiving elements. (Refer to Fig. 3 for appropriate orthotelephonic definitions.) In many cases of interest in telephony, the same transmitting and receiving elements are likely to be used in many computations. Column 5 of the form provides for combining responses of these elements with $B_{90} - X$.

The effective level of the received speech spectrum, Z_s, resulting from combining entries of columns 5 and 6, is entered in column 7. The effective level in each computation band is converted to a loudness number for a band of unit importance using the curve of Fig. 6. The loudness numbers, entered in column 8, apply for frequency bands having 2-percent importance. The bands of Fig. 14 with midfrequencies of 600 Hz and above have been selected to give 2-percent importance. Below this frequency, however, the 2-percent bands become very narrow (see Fig. 5), and circuit measurements at the required frequencies provide a fineness in the response curves not required for telephone rating purposes. Thus, wider bands are used and the greater importance of these bands allowed for by means of column 9 which shows the number of 2-percent bands in each computation band. The loudness for a computation band is thus the product of column 8 and column 9 entries, and is entered in column 10. The total loudness, N_s, is obtained by summing column 10.

In some cases, the computation frequency is not the midfrequency of the computation band according to Fig. 5. The midfrequencies

COMPUTATION NO.

DATE

COMPUTED BY

TRANSMITTER

RECEIVER

CIRCUIT

(1) f	(2) B_{90}-X *	(3) T_{OT}	(4) R_{OT}	(5) 2+3+4	(6) L	(7) Z_S (5) + (6)	(8) n_s	(9) Δs	(10) $n_s\Delta s$
150	46.5							13	
200	53							7	
300	62.5							4	
400	68							3	
500	71.5							2	
600	73							1	
700	72.5							1	
800	72							1	
900	71.5							1	
1000	70							1	
1200	69							1	
1400	67.5							1	
1600	66.5							1	
1800	67							1	
2000	67.5							1	
2200	68							1	
2400	68							1	
2600	68							1	
2800	67							1	
3000	67							1	
3300	66							1	
3600	64.5							1	
3900	63.5							1	
4200	62							1	
4500	60.5							1	
4800	59							1	

$N_S = \Sigma n_s \Delta s =$ ______

* APPLIES ONLY FOR A LONG-TERM SPEECH SOUND PRESSURE LEVEL OF 90 dBt AT 2" FROM THE TALKER'S LIPS. FOR OTHER TALKING LEVELS, CHANGE ENTRIES OF COLUMN 2 BY THE DIFFERENCE BETWEEN 90 dBt AND THE TALKING LEVEL OF INTEREST.

Fig. 14—Speech loudness computation form.

were rounded off as shown for purposes of convenience. For the lowest computation band, this rounding off overemphasizes that band's contribution to the loudness, but this generally affects the total loudness number, N_s, only slightly.

The computation bands and the computation frequencies of Fig. 14 were selected for convenience. These can be changed, e.g., to consider 1/3-octave bands, by first locating the band limits on Fig. 5, then determining (*i*) the center frequency for, and (*ii*) the number of bands of unit importance contained in, the selected computation band. Appropriate values of $B_{90} - X$ can be obtained from Fig. 9.

The procedure described above applies for cases where the loudness of a band of speech is determined by the effective level of the speech within that band. Masking experiments have shown that when the effective level of a continuous spectrum sound changes abruptly with frequency, e.g., a sharp cutoff filter applied to thermal noise, there may be masking in the suppressed frequency band due to energy in the passed frequency band. Since masking has been identified with loudness, it is reasonable to assume that the loudness of a band-limited speech signal will depend not only on the passed region but the suppressed region as well. This effect, termed spread-of-loudness, is considered in the computation by assuming that Z_s does not decrease by more than 10 dB between adjacent 2-percent loudness bands.* If Z_s (column 7 of Fig. 14) shows a more rapid change, a new Z_s is entered which is exactly 10 dB below the Z_s of the preceding or succeeding 2-percent band.

3.4.2 *Interpretation of Total Loudness, N_s*

Loudness in loudness units (LU), although a ratio scale reflecting observers assessment of sound magnitude, has little physical significance. It is thus desirable to express loudness in such a way as to convey a physical interpretation. This can be done by expressing the loudness of a sound in terms of the level of a reference sound which has the same number of loudness units. Thus, speech loudness can be expressed in terms of loudness level, i.e., the level of a 1000-Hz tone having the same loudness. Loudness level is an appropriate way to express speech loudness when comparing computed values with results of tests which involved loudness balancing speech and a 1000-Hz tone. Speech loudness (N_s) can be computed using the form of Fig. 14, then with $N_s = N_{1000}$, the loudness level found from Fig. 7.

Often speech through a test circuit is balanced against speech through some relatively distortionless reference circuit. In these cases, the loudness of the speech transmitted over the test circuit should probably be expressed in terms of the setting of the reference circuit.

* The spread-of-loudness effect was determined from masking data obtained by Fletcher and Munson (Figs. 11 and 12 of Ref. 23 or Figs. 141 and 142 of Ref. 27) using narrow bands of noise. The masking data were plotted in terms of number of 2-percent bands below and above the nominal bandlimits of the noise. A straight line with a slope of 10 dB/band fitted the data with reasonable accuracy, thus permitting use of a simple rule for taking spread-of-loudness effects into account.

Thus, speech loudness can be expressed in terms of several different reference sounds. A single reference would be desirable to enable comparison of results from different tests. Loudness level would be suitable for such purposes. However, considering the different nature of speech and a 1000-Hz tone, a speech signal of specified characteristics might be a more suitable reference, one that would be more intuitively satisfying.

The reference selected for the speech loudness computation procedure is that due to transmitting average speech (Fig. 2) having the reference effective spectrum (Fig. 9) over a circuit that is flat on an orthotelephonic basis (Fig. 3) except for sharp cutoffs at 250 and 4000 Hz. (These frequency limits, selected somewhat arbitrarily, were wide enough to include existing and planned commercial telephone circuits.) The loudness of speech from any telephone circuit can then be specified by the level of the reference that gives an equivalent N_s; the level, in turn, can be conveniently specified by its integrated rms pressure in dBt. The level of the reference speech spectrum, designated as L_s, is related to its loudness, N_s, as shown on Fig. 8. This curve was derived using the form of Fig. 14, successively changing entries per column 2 thereon, and computing N_s for each such change. It should be noted that (*i*) L_s for the entries of column 2 is 90 dBt for the band 100–5000 Hz, 89.4 dBt for the band 250–4000 Hz and (*ii*) spread-of-loudness was taken into account in the computation of N_s.

3.5 *Comparison of Computed and Observed Results*

Observed results from six different subjective tests were compared to computed results obtained using the computation form of Fig. 14 and the frequency response characteristics of the subjective test systems. (Observed results from two of these tests were used in developing the computational method.) Observed results comprised reference circuit attenuator settings or trunk settings obtained by observers when they loudness-balanced reference circuits and test circuits.

Computed reference circuit settings were determined by first computing the loudness (N_s) for each test condition and its corresponding reference condition (a reference circuit with its attenuator or trunk setting at the value to which observers adjusted to obtain loudness balance) using the test and reference circuit frequency response characteristics.* The computed loudness values (N_s) were then converted to levels of the reference spectrum (L_s). By comparing the L_s of each

* Frequency response characteristics used for the loudness computations discussed in this paper are not reported.

test circuit with the L_s of corresponding reference circuit at observed balance, computed settings were obtained.

Computed and observed results are discussed and tabulated in Sections 3.5.1 thru 3.5.6 and are plotted on Fig. 15. (For the latter, the L_s for the reference circuit at balance is considered to be the observed value, the L_s for the test circuit is the computed value.) Comparisons of computed and observed results are summarized in Table IV in terms of error distributions. These, together with Fig. 15 and tabulated values of ensuing sections, indicate that the computational method provides a high degree of accuracy.

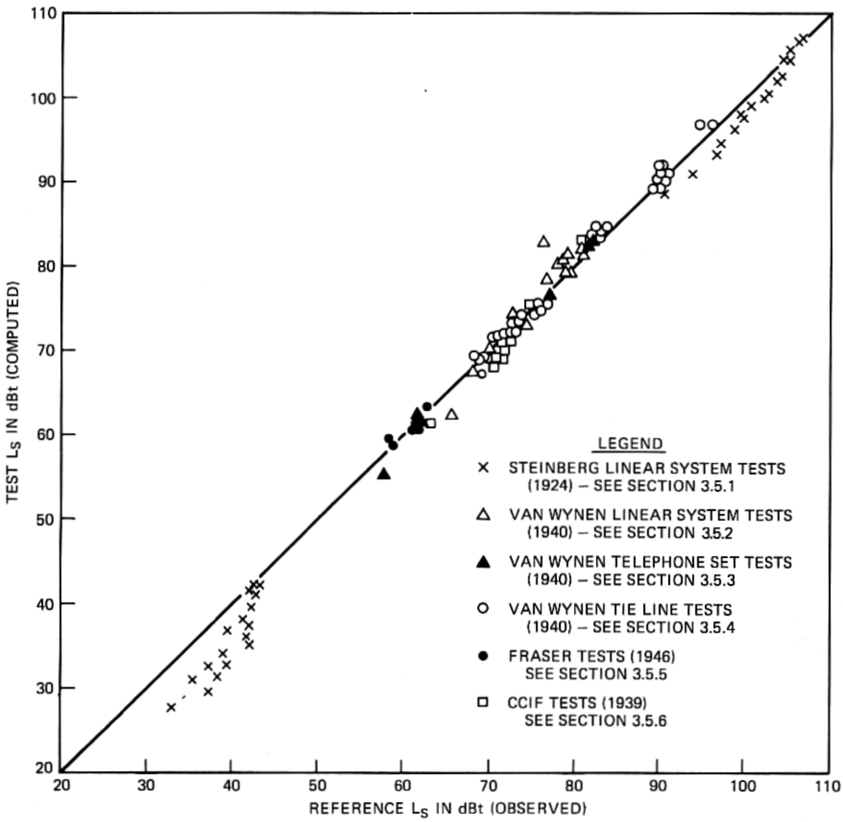


Fig. 15—Comparison of reference (observed) and test (computed) speech loudness.

TABLE IV—SUMMARY OF COMPARISON OF COMPUTED AND OBSERVED RESULTS

Test Designation	Detailed Results In	Computed Minus Observed	
		Average-dB	σ-dB
Steinberg tests (1924)			
Sensation level = 100 dB	3.5.1	1.4	1.1
Sensation level = 39 dB	3.5.1	4.0	2.2
Van Wynen linear system tests (1940)	3.5.2	−0.8	2.1
Van Wynen telephone set tests (1940)	3.5.3	0.3	1.1
Van Wynen tie line tests (1940)	3.5.4	−0.4	0.9
Fraser tests (1940)	3.5.5	0.2	0.9
CCIF tests (1939)	3.5.6	0.8	1.1

3.5.1 *Steinberg Linear System Tests* (1924)

In 1924, J. C. Steinberg conducted loudness balance tests with a system which utilized linear transducers interconnected by either of two channels, a test channel into which various filters were inserted and a reference channel which provided essentially distortionless transmission.[28,29] Observers adjusted an attenuator in the reference channel to obtain loudness balance for each of a number of the test networks.

Computed and observed loudness losses are given in Table V. For purposes of the table, loudness loss is equal to the amount of flat loss by which the reference circuit was varied from its reference setting (for the particular sensation level) so that speech through the reference channel had the same loudness as speech through the test channel. Observed values given for the 100-dB sensation level were used in deriving the S function. (See Section 3.3.4.1.)

For the reference channel, L_s with the 100-dB sensation level reference setting was 107.1 dBt; with the 39-dB sensation level reference setting, L_s = 42.6 dBt. Observed and computed loudness losses per the table may be converted to L_s values for the reference channel by subtracting entries in the "observed" column from the above reference values; L_s for the test channel may be obtained by subtracting entries in the "computed" column, for the appropriate test network, from the reference values given above. Thus, the reference channel L_s = 102.1 dBt and the test channel L_s = 100.1 dBt for a 500-Hz high-pass filter and a 100-dB sensation level.

Agreement between computed and observed values is reasonably good for the most part. In general, the agreement becomes poorer with increasing distortion and is poorer at the lower sensation levels. Neither of these affects is particularly restrictive from a telephone engineering standpoint since they represent extremes which a well-engineered telephone plant will seldom approach.

3.5.2 *Van Wynen Linear System Tests (1940)*

In 1940, K. G. Van Wynen conducted loudness balance tests using a system equipped with linear transducers. (These tests were not

TABLE V—COMPUTED AND OBSERVED LOUDNESS LOSS VALUES FOR THE STEINBERG LINEAR SYSTEM TESTS (1924)

Test Network*	Sensation Level = 100 dB			Sensation Level = 39 dB		
	Loudness Loss (dB)		C = B − A (dB)	Loudness Loss (dB)		F = E − D (dB)
	A Observed	B Computed		D Observed	E Computed	
125 Hz HPF	0.6	0.2	−0.4	0	0.2	0.2
250	2.0	2.7	0.7	0.4	1.0	0.6
375	3.6	5.0	1.4	−0.2	1.5	1.7
500	5.0	7.0	2.0	1.0	4.8	3.8
625	7.6	9.0	1.4	3.0	5.8	2.8
750	8.5	10.8	2.3	0.6	7.4	6.8
1000	10.5	13.8	3.3	3.2	9.8	6.6
1250	13.2	16.2	3.0	4.5	11.3	6.8
1500	16.6	18.5	1.9	5.3	13.0	7.7
		Avg. =	1.7		Avg. =	4.1
		σ =	1.1		σ =	2.8
750 Hz LPF	9.9	12.6	2.7	9.7	14.4	4.7
1000	7.4	9.4	2.0	7.3	11.6	4.3
1250	6.4	8.0	1.6	5.3	10.0	4.7
1500	4.4	6.5	2.1	3.7	8.6	4.9
2000	3.0	4.3	1.3	0.8	6.6	5.8
2500	2.4	2.5	0.1	1.4	4.3	2.9
3000	1.9	1.5	−0.4	0.2	3.0	2.8
4000	0.7	0.4	−0.3	−0.5	0.4	0.9
		Avg. =	1.1		Avg. =	3.9
		σ =	1.1		σ =	1.5

* HPF = high-pass filter; LPF = low-pass filter. Entries designate nominal cutoff frequencies of the filters. See Ref. 48 for description of the filters which were probably used in the Steinberg tests.

TABLE VI—COMPUTED AND OBSERVED RESULTS FOR THE VAN WYNEN LINEAR SYSTEM TESTS (1940)

Distorting Network*	Observed Reference Circuit Attenuator Setting-dB	L_s — dBt		Computed Reference Circuit Attenuator Setting-dB	Computed Minus Observed-dB
		Test Circuit	Reference Circuit (at Balance)		
2700 Hz LPF	24.2	81.4	80.8	23.6	−0.6
1750 Hz LPF	28.3	78.3	76.6	26.6	−1.7
1000 Hz LPF	31.3	74.1	73.4	30.6	−0.7
250 Hz HPF	25.6	79.1	79.2	25.7	0.1
550 Hz HPF	30.8	73.1	74.0	31.7	0.9
1000 Hz HPF	36.4	67.6	68.1	36.9	0.5
250–2700 Hz BPF	29.0	75.0	76.0	30.0	1.0
550–1750 Hz BPF	38.9	62.5	65.6	42.0	3.1
1000 Hz Peak					
15 dB	26.8	80.2	78.1	24.7	−2.1
30	24.2	81.9	80.8	23.1	−1.1
56	34.5	70.3	70.1	34.3	−0.2
700 Hz Peak					
30 dB	25.8	81.4	79.0	23.4	−2.4
2000 Hz Peak					
30 dB	28.7	82.8	76.2	22.1	−6.6
Falling Loss	25.6	79.2	79.2	25.6	0
Rising Loss	26.4	80.6	78.5	24.3	−2.1
				Avg. =	−0.8
				σ =	2.1

* LPF = low-pass filter, HPF = high-pass filter, BPF = bandpass filter; numbers denote nominal filter cutoff frequencies. "Peak" denotes a resonant circuit defined by the resonant frequency (minimum loss of circuit occurred at resonance) and a dB value indicative of the Q, e.g., 56 dB indicates a high Q, 15 dB indicates a low Q. "Falling Loss" designates a network whose loss decreased monotonically with increasing frequency while for the "Rising Loss" network, the loss increases monotonically with frequency.

reported in the literature.) The transducers were interconnected via one of two channels, the test channel into which various test networks were introduced and a reference channel which was essentially distortionless. Switching between channels was controlled by the observers who, for each network tested, adjusted an attenuator in the reference channel to produce equal loudness. Twelve talker-listener pairs participated in the tests.

Computed and observed attenuator settings are given in Table VI. The computed setting was obtained by noting the difference in L_s for the test circuit and the reference circuit at balance, and modifying the observed setting by this amount.*

*The high- and low-pass filter results, expressed in different terms, were used in deriving the speech loudness computational method. See Section 3.3.4.1 and Fig. 10.

The agreement between observed and computed reference circuit settings is quite good, the error being less than 2 dB in ten of the fifteen cases. Those cases showing the larger errors represent severe distortion conditions which would be approached rarely, if ever, in a well-engineered telephone plant.

3.5.3 *Van Wynen Telephone Set Tests (1940)*

In 1940, K. G. Van Wynen conducted tests in which different types of telephone sets were compared on a loudness basis. (These tests were not reported in the literature.) The laboratory system used consisted essentially of two separate acoustic-to-acoustic channels. One of these was equipped with reference transmitting and receiving telephone sets interconnected by a test network which included an adjustable attenuator. (The telephone sets were of the type described in Ref. 16.) The other channel used each of two different types of telephone sets, designated Test Telephone Sets A and B, with fixed connecting circuitry. (Test Telephone Sets A were of the type described in Ref. 15; Test Telephone Sets B were of a design similar to the telephone sets used with the Working Reference System described in Section 2.2 of this paper.) These sets differed from each other and from the reference telephone set in that they had transducers exhibiting different frequency response characteristics.

Results of the tests are expressed in terms of trunk loss, variable and controlled by the observer, required in the reference channel to deliver speech equal in loudness to that from the test channel with a fixed trunk loss. (Twelve talker-listener pairs participated in the tests.) Computed and observed trunk losses shown in Table VII are in close agreement. The computed trunk loss was obtained by noting the difference in L_s for the test channel and the reference channel, and modifying the observed trunk loss by this difference.

3.5.4 *Van Wynen Tie Line Tests (1940)*

In 1940, K. G. Van Wynen conducted loudness balance tests with laboratory systems which simulated selected circuit conditions characteristic of telephone connections between Bell Telephone Laboratories and the American Telephone and Telegraph Company. These connections involved circuits which were called "tie lines," and thus the designation "tie line tests."

TABLE VII—COMPUTED AND OBSERVED LOUDNESS LEVELS FOR THE VAN WYNEN TELEPHONE SET TESTS (1940)

Test Telephone Set	Network*	Observed Trunk Loss at Loudness Balance-dB†		L_s — dB†		Computed Reference Trunk Loss-dB	Computed Minus Observed Reference Trunk Loss-dB
		Test Channel	Reference Channel	Test Channel	Reference Channel		
A	4000 Hz LPF	5	16.8	76.3	76.8	17.3	0.5
	4000 Hz LPF	25	35.2	55.3	57.7	37.6	2.4
B	4000 Hz LPF	5	11.4	82.7	82.2	10.9	−0.5
	4000 Hz LPF	25	31.5	62.2	61.7	31.0	−0.5
	Rising Loss	5	11.7	82.5	81.8	11.0	−0.7
	Rising Loss	25	31.2	61.5	62.2	31.9	0.7
							Avg. = 0.3
							σ = 1.1

* 4000 Hz LPF = low-pass filter with a nominal cutoff frequency of 4000 Hz; Rising Loss = increasing loss with increasing frequency. The network was switched (by the observer) between reference and test channels and loudness balance achieved by adjusting an attenuator in the reference channel connecting circuit.

† Values are the 1-kHz losses (900-ohm source and termination) of the reference and test channel connecting circuits.

Two test systems, designated "overall" and "sidetone," were used.* Each of these permitted comparison between test networks and either of two reference networks. (The transmitter and receiver used in these tests were of a design described in Ref. 17.) The observers' task was to adjust an attenuator in the reference channel so that speech via this channel and the test channel were equally loud. Twelve talker-listener pairs participated in the tests.

Results of the tests expressed in terms of reference circuit settings are shown in Table VIII. Corresponding values of L_s (dBt) are given as parenthetical entries; those in the "Observed" columns apply for the reference channel while those in the "Computed" columns apply for the test channel. Computed settings were obtained by noting the difference between L_s for the test channel and L_s for the reference channel, and modifying the observed circuit setting by this difference. Computed and observed values are in close agreement.

3.5.5 *Fraser Tests (1946)*

In 1946, J. M. Fraser conducted tests in which two different acoustic-to-acoustic channels were compared on a loudness basis. (These tests were not reported in the literature.) The test channel comprised test telephone sets of the type which were the Bell System Standard at that time,[16,17] and connecting circuitry which simulated selected telephone connections. These simulations, arbitrarily designated in column 1 of Table IX, exhibited different amounts of increasing loss with increasing frequency. The reference channel was the Master Reference System discussed in Section 2.1 of this paper.

Results of the tests are expressed in terms of the Master Reference System trunk (600-ohm attenuator) setting, variable and controlled by the observer, required to achieve loudness balance between the test channel and the reference channel. Twelve talker-listener pairs participated in the tests. Computed and observed reference trunk settings shown in Table IX are in close agreement.

3.5.6 *CCIF Tests (1939)*

In 1939, the CCIF† conducted loudness balance tests on selected simulated telephone connections sent them by the American Telephone

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* These designations reflect the frequency response characteristics simulated and not testing conditions. Specifically, those with the "sidetone" system were on a listening only basis, and did not involve a talker listening to himself via the system. Sidetone is discussed in Refs. 33, 49, and 50.

† See Section 2.1 of this paper.

TABLE VIII—COMPUTED AND OBSERVED LOUDNESS LOSSES FOR THE VAN WYNEN TIE LINE TESTS (1940)

Test Network*	DL Reference			Peaked Reference		
	Circuit Setting-dB		Computed Minus Observed-dB	Circuit Setting-dB		Computed Minus Observed-dB
	Observed	Computed		Observed	Computed	
Overall						
20 DL	—	—	—	6.3(76.0)	7.3(75.0)	1.0
20 DL	—	—	—	6.8(75.5)	7.3(75.0)	0.5
20–3000 LPF	21.2(73.8)	20.8(74.2)	−0.4	6.9(75.4)	8.1(74.2)	1.2
20–2400 LPF	22.5(72.5)	22.7(72.3)	0.2	9.6(72.7)	10.0(72.3)	0.4
20–1900 LPF	24.7(70.3)	25.7(69.3)	1.0	13.8(68.5)	13.0(69.3)	−0.8
20–1900 LPF	26.0(69.0)	25.7(69.3)	−0.3	12.9(69.4)	13.0(69.3)	0.1
20–RL1	22.0(75.0)	21.6(73.4)	−0.4	8.7(73.6)	8.9(73.4)	0.2
20–RL2	23.3(71.7)	22.9(72.1)	−0.4	10.4(71.9)	10.2(72.1)	−0.2
20–RL3	24.4(70.6)	23.3(71.7)	−1.1	11.3(71.0)	10.6(71.7)	−0.7
			$\bar{x}$ = −0.2			$\bar{x}$ = 0.1
			σ = 0.6			σ = 0.6
Sidetone						
Ref. DL	0.7(94.8)	−1.3(96.8)	−2.0	−13.7(96.2)	−14.3(96.8)	−0.6
−10 DL	11.5(83.0)	10.7(83.8)	−0.8	−1.6(83.3)	−2.1(83.8)	−0.5
−20 DL	19.2(75.3)	19.0(75.5)	−0.2	5.4(77.0)	6.9(75.5)	1.5
R900	5.4(89.8)	6.2(89.0)	0.8	−6.9(89.1)	−7.0(89.2)	−0.1
R1300	5.4(90.0)	4.4(91.0)	−1.0	−8.6(90.9)	−8.7(91.0)	−0.1
R1800	5.2(89.9)	3.1(92.0)	−2.1	−8.2(89.7)	−10.5(92.0)	−2.3
R3200-A	12.2(82.4)	11.1(83.5)	−1.1	−0.8(82.7)	−1.6(83.5)	−0.8
R3200-B	12.2(82.5)	10.1(84.6)	−2.1	−2.1(83.9)	−2.8(84.6)	−0.7
RL4	5.7(89.6)	5.0(90.3)	−0.7	−8.2(90.7)	−7.8(90.3)	+0.4
			$\bar{x}$ = 1			$\bar{x}$ = −0.4
			σ = 0.9			σ = 1.0
			Overall: $\bar{x}$ = −0.4			
			σ = 0.9			

* The number preceding the overall network designation defines the value of an attenuator associated with that network. For the DL network, this number (20) was the setting of an attenuator associated with the network when it was balanced against the peaked reference. The DL reference and the "20 DL" overall were the same circuit (incorporating in essence a variable attenuator) and differed from the DL circuits for the sidetone tests. The response of all DL networks showed decreasing loss with increasing frequency. LPF = low-pass filters with a nominal cutoff frequency (in Hz) noted. The RL networks had differing amounts of increasing loss with increasing frequency. Networks designated R were resonant circuits with high values of loss at the indicated resonant frequencies. Networks R3200-A and R3200-B differed in that the former had a slightly greater Q.

TABLE IX—COMPUTED AND OBSERVED LOUDNESS LEVELS FOR THE TEST AND REFERENCE CHANNELS OF THE FRASER TESTS (1946)

Test Network	Observed Reference Trunk Setting (dB)	L_s − dBt Test System	L_s − dBt Reference System	Computed Reference Trunk Setting (dB)	Computed Minus Observed (dB)
A	33.3	67.5	69.2	35	1.7
B	39.9	63.3	62.8	39.4	−0.5
C	41.5	60.8	61.2	41.9	0.4
D	43.0	58.8	58.9	43.1	0.1
E	40.3	61.6	61.9	40.6	0.3
F	43.8	59.7	58.4	42.5	−1.3
G	40.6	60.9	61.6	41.3	0.7
					Avg. = 0.2
					σ = 0.9

and Telegraph Company. (These tests were not reported in the literature.) These simulations were a subset of those discussed in Section 3.5.4 of this paper.

The CCIF tests involved two steps. First, the simulated connection designated "DL Reference" with 20 dB of added loss and the SFERT* were loudness balanced. Twenty talker-listener pairs, formed from a trained test crew of five persons, each provided seven balances. Results comprised settings (in dB) of the SFERT trunk (600-ohm attenuator) required to achieve balance; the average value was 30.9 dB.†

The L_s for the DL Reference (with 20 dB added loss) was computed using frequency response characteristics for the DL Reference of Section 3.5.4 while the L_s for the SFERT (with an attenuator setting of 30.9 dB) was computed from response characteristics for the reference system of Section 3.5.5. The respective values were 74 dBt and 71.6 dBt. That is, the CCIF loudness balance tests showed these circuits to be equal while the speech loudness computational method indicates that they differ by 2.4 dB. Probably this difference is due to the system response characteristics (primarily transducers) used in the computations, and not to the computational method itself being in error. In the computations, corrections reflecting differences in specific transducers of the same type were applied where known,

* See Section 2.1 of this paper.
† As noted in Section 2.1, the SFERT line attenuator setting (in dB) required for loudness balance with a given system under test is, by definition, the Reference Equivalent (in dB) of that system. Thus, the Reference Equivalent of the 20 DL Reference connection is 30.9 dB.

but these were not known in all cases, and average responses were necessarily used in these cases. This argument is supported to some extent by the internal consistency for the tests of Section 3.5.4, the tests of Section 3.5.5, and the CCIF tests.

The DL Reference was then used as a secondary reference against which ten simulated telephone connections were loudness balanced. Computed and observed results given in Table X are in close agreement.

3.5.7 *Comparison of Computed Loudness Losses and Reference Equivalents*

Reference Equivalents are, by definition, loudness ratings of telephone connections obtained using the SFERT (and, by inference, the MRS and NOSFER). The speech loudness computational method provides loudness ratings in orthotelephonic terms. Since both are based on speech loudness, information from the preceding sections can be used to derive a correction factor which would permit translating ratings between the two frames of reference. (Such translation depends on bandwidth characteristics of telephone connections and therefore will depend on, among other things, the particular transducer types involved.) Tables XI and XII summarize the appropriate information.

The average conversion factor from Table XI is 12.7 dB while that from Table XII is 15.0 dB, a difference of 2.3 dB. Some possible reasons for this difference were discussed in Section 3.5.6. In addition, the observer groups differed for the Fraser and CCIF tests, and there were probably some minor differences in test procedure.

IV. LABORATORY MEASURING SYSTEM

The speech loudness computational procedure could be realized as a laboratory measuring system in any one of several forms. The "ideal" system would implement exactly the various functions of the computational procedure. The system sound source would energize the transmit end of a telephone connection with an acoustic signal having the amplitude characteristic of Fig. 2 and a time rate-of-change based on Fig. 5. The signal received at the other end of the connection would be applied to a measuring subsystem containing a network with a response according to Fig. 4. System operation so far corresponds to computing the effective spectrum, Z_s. (See column 7 of the computation form on Fig. 14.)

Table X—Loudness Balance Results with DL Reference Connection as Reference

Test System		Observed DL Reference Trunk Setting (dB)	L_s − dBt		Computed DL Reference Trunk Setting (dB)	Computed Minus Observed (dB)
Type	Designation*		Test System	DL Reference System		
Overall	20–1900 LPF	23.5	68.3	70.5	25.7	2.2
	20–2400 LPF	21.7	71.3	72.3	22.7	1.0
	20–3000 LPF	20.8	73.2	73.2	20.8	0.0
	20–RL2	22.3	71.1	71.7	22.9	0.6
	20–RL3	22.2	70.7	71.8	23.3	1.1
						Avg. = 1.0
						σ = 0.7
Sidetone	Ref DL	13.5	82.8	81	11.7	−1.8
	−10 DL	23.7	69.8	70.8	24.7	1.0
	−20 DL	31.8	61.5	62.7	33.0	1.2
	R900	19.0	75.2	75.5	19.3	0.3
	R3200-A	22.9	69.5	71.6	25.0	2.1
						Avg. = 0.6
						σ = 1.3
Overall: $\bar{x}$ = 0.8, σ = 1.1						

* See footnote of Table VIII.

TABLE XI—RELATION BETWEEN REFERENCE EQUIVALENTS (RE) AND LOUDNESS LOSS (LL) FROM THE FRASER TESTS (SECTION 3.5.5)

Test System	RE* (dB)	L_s* (dBt)	LL† (dB)	RE Minus LL (dB)
A	33.3	67.5	21.9	11.4
B	39.9	63.3	26.1	13.8
C	41.5	60.8	28.6	12.7
D	43.0	58.8	30.6	12.4
E	40.3	61.6	27.8	12.5
F	43.8	59.7	29.7	14.1
G	40.6	60.9	28.5	12.1
				Avg. = 12.7
				σ = 0.9

* Columns 2 and 3 of Table IX.
† 89.4 minus entries of column 3. See last paragraph of Section 3.4.2.

The remainder of the measuring subsystem would consist of a device to reflect the spread-of-loudness effect, a device having the law of Fig. 6, an integrating circuit, and, finally, a display mechanism reflecting the law of Fig. 8.

The "ideal" system could be modified, without changing its essen-

TABLE XII—RELATION BETWEEN REFERENCE EQUIVALENTS (RE) AND LOUDNESS LOSS (LL) FROM CCIF TESTS (SECTION 3.5.6)

Test System	RE* (dB)	L_s† (dBt)	LL‡ (dB)	RE Minus LL (dB)
20-DL REF	30.9	74	15.4	15.5
20-1900 LPF	34.4	68.3	21.1	13.3
20-2400 LPF	32.6	71.3	18.1	14.5
20-3000 LPF	31.7	73.2	16.2	15.5
20-RL2	33.2	71.1	18.3	14.9
20-RL3	33.1	70.7	18.7	14.4
REF DL	24.4	82.8	6.6	17.8
−10 DL	34.6	69.8	19.6	15.0
−20 DL	42.7	61.5	27.9	14.8
R900	29.9	75.2	14.2	15.7
R3200-A	33.8	69.5	19.9	13.9
				Avg. = 15.0
				σ = 1.1

* Derived from column 3 entries of Table X and relation that DL Reference with 20-dB trunk setting had a reference equivalent of 30.9 dB.
† From column 4 of Table X and text of Section 3.5.6.
‡ 89.4 minus entries of column 3. See last paragraph of Section 3.4.2.

tial attributes, by combining the "X" network response with that of the signal source. The source output would then have an amplitude characteristic as shown by the dash-dot curve of Fig. 9. (See column 2 of the computation form on Fig. 14.)

Realization of the system design outlined above is limited by a number of practical considerations. Exact implementation would require using an artificial mouth and an artificial ear which duplicated their human counterparts within the context of definitions given in Fig. 3. Such are not available. However, simpler devices are available which are probably adequate for purposes of many telephone engineering problems.

Secondly, the source signal does not incorporate the dynamic properties of speech. This is probably not important as long as the telephone connection under test is composed of linear elements. However, measurements on connections incorporating nonlinear elements, particularly carbon transmitters, may be in error. Carbon transmitters in real use are activated by real speech; testing such with an applied signal of the type referred to above might well result in different device operating characteristics. This matter is under study.

In this paper, we consider a system which is somewhat simpler than the "ideal" and which probably reflects adequate accuracy for telephone rating purposes. This system, derived from the computational procedure, is designated the EARS (for *E*lectro-*A*coustic *R*ating *S*ystem), and is dealt with in ensuing sections.* A graphical computation procedure is also described. Finally, a comparison is made between ratings based on the EARS and ratings obtained from subjective tests.

The EARS utilizes an artificial voice and a 6-cm^3 coupler while the computational procedure utilizes the concept of orthotelephonic transmission. Response definitions appropriate to artificial mouth and 6-cm^3 coupler measurements are shown on Fig. 16 in terms similar to the orthotelephonic response definitions of Fig. 3.

Typical artificial voice/6-cm^3 coupler amplitude responses of connections employing specific telephone set types are given in Ref. 18. As with orthotelephonic responses, artificial voice/6-cm^3 coupler responses vary widely depending on telephone set and transducer types used. Conversion curves for translating between orthotelephonic and artificial voice/6-cm^3 coupler responses are not as variable, but are still highly dependent on telephone set and transducer type.

* Reference 51 describes derivation of a measuring system which is similar to the EARS.

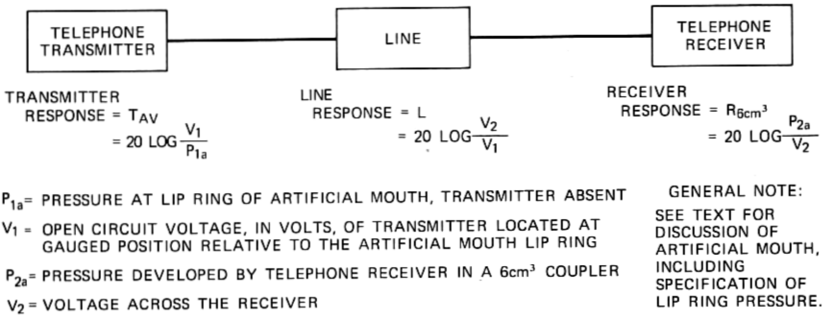

Fig. 16—Telephone circuit responses in artificial voice — **6-cm³** coupler terms.

4.1 *Derivation of EARS*

We begin by considering the computation form of Fig. 14. For the reference effective spectrum (column 2), we compute the cumulative contribution to loudness as a function of computation band, i.e., we assume an orthotelephonically flat transmission system with 0 dB of gain for the sum of columns 3, 4, and 6, and compute the entries for column 10. Cumulating these and dividing each cumulative entry by the total loudness (N_s) results in a relation between cumulative percentage contribution to loudness and frequency. This relation is shown by the small circles plotted on Fig. 17; the ordinate is cumulative and the abscissa is upper frequency limit of the computation band, obtained from Fig. 5. The relation of Fig. 17 differs from that of Fig. 5 because the latter is for a flat effective spectrum (Z_s = a constant) whereas the former is for the effective spectrum of speech ($Z_s = B_{90} - X$) shown by the dot-dash curve of Fig. 9.

The points plotted on Fig. 17 can be reasonably approximated by a straight line drawn from 100 Hz to 5000 Hz on the logarithmic frequency scale. Thus, 2-percent loudness bands, or any bands of equal loudness derived from this curve, have a logarithmic relation to frequency and may be found by laying off equal lengths along the logarithmic frequency scale.

In terms of a measuring system, the above suggests using a source signal which has a flat amplitude versus frequency characteristic but which sweeps across the band at a logarithmic rate. Such a signal would cover equal loudness bands of the effective spectrum ($B_{90} - X$) in equal time divisions, corresponding to equal length divisions along the abscissa of Fig. 17. Integration of the sweep on a time basis

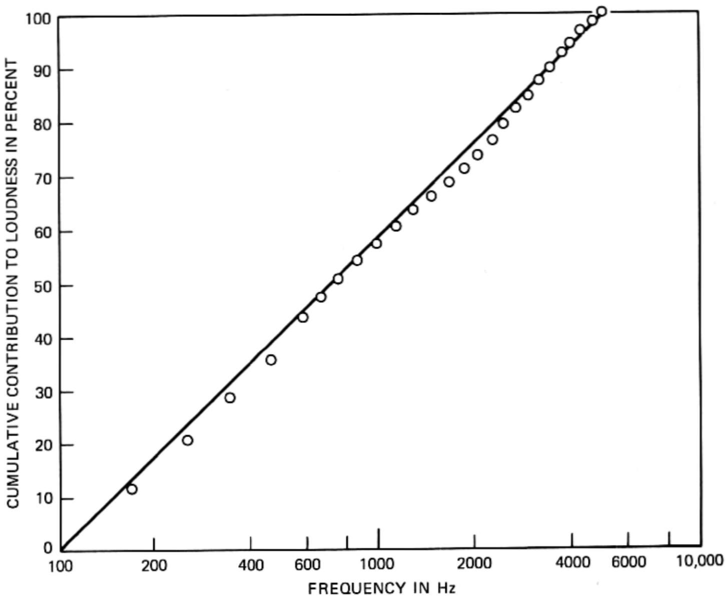

Fig. 17—Cumulative contribution to loudness of speech for the reference effective spectrum (see Figs. 9 and 14).

would result in a value proportional to the total loudness of the effective spectrum since the signal amplitude is constant with frequency.

In essence, the operation described above translates the amplitude weighting of $Z_s = B_{90} - X$ to frequency weighting. The computation form of Fig. 14 could be modified to reflect this by changing columns 1 and 9 to conform with the straight line of Fig. 17 and entering a constant value of Z_s in column 2. The value of Z_s can be determined by dividing the computed N_s for $B_{90} - X$ by the number of computation bands (e.g., 50) to find n_s (the loudness per band of unit importance), then entering the curve of Fig. 6 at that value of n_s.

The source signal described above, in combination with the acoustic-to-acoustic response of a telephone connection, provides the effective spectrum of the received speech for that connection. The next step, then, is to apply the loudness scale of Fig. 6 in order to convert to loudness units.

The loudness scale of Fig. 6 is linear on logarithmic coordinates, i.e., $Z_s = \text{k} \log n_s$, above the knee of the curve, and the number of loudness units increases tenfold for a 44-dB change in effective level, or doubles for a 13.2-dB change in effective level. (This is reasonably checked by subjective loudness tests with filters as plotted on Fig. 10. In the Steinberg tests, a 9.1-dB reduction in effective level corresponded to a 50-percent reduction in loudness, while in Van Wynen tests the level reduction required to produce half loudness was 12.4 dB). This relation is assumed to hold over the entire range of effective levels. The effect of this assumption on accuracy will be considered in a later section.

At the receiving end of a telephone connection activated by the logarithmic signal source at the transmitting end, the received acoustic signal is applied to a measuring circuit and ultimately appears as a voltage across a resistance. At any instant, this voltage is proportional to $10^{C/20}$ where C is the response, in dB, of the telephone connection at a particular frequency. However, because of the sweep rate of the signal source, voltage elements appearing across the resistance in time sequence are proportional to $10^{Z_s/20}$, where Z_s is the effective level of the received speech spectrum. We have already postulated that loudness doubles for a 13.2-dB change in Z_s; the voltage elements we would like to see across the resistance should therefore be proportional to $10^{Z_s/44}$.* Thus, the desired voltage elements, V_1, are proportional to the 2.2 root of the voltage elements, V_{1x}, which actually appear, i.e.,

$$V_1 \propto 2.2 \sqrt{V_{1x}} \tag{9}$$

or

$$(V_1)^{2.2} \propto V_{1x}\,. \tag{10}$$

The above relationship suggests that voltage elements proportional to $10^{Z_s/20}$ be applied to the input of a 2.2-to-1 compressor. With such a device, changes at the input proportional to

$$10^{(Z_{s1}-Z_{s2}/20)}$$

appear at the output proportional to

$$10^{(Z_{s1}-Z_{s2}/44)},$$

* If loudness N_2 is twice as large as loudness N_1,

$$\frac{N_2}{N_1} = 10^{(Z_{s2}-Z_{s1}/x)} = 2,$$

$$Z_{s2} - Z_{s1} = x \log_{10} 2 = 13.2,$$

and

$$x = 44.$$

Thus

$$N \propto 10^{Z_s/44}.$$

which is the 2.2 root of the input change. Thus, the action of the compressor approximates the loudness scale of Fig. 6 in converting elements of the received effective spectrum to loudness units.

Integration of the voltage with respect to time may be either on a square-law or linear basis. Without the compressor, square-law integration adds voltage elements proportional to $10^{Z_s/10}$ while linear integration adds voltage elements proportional to $10^{Z_s/20}$. By including the compressor with linear integration, elements proportional to $10^{Z_s/44}$ are added, these in turn being proportional to loudness units. The combination of the two gives, in effect, 2.2 root law addition.

System operation to this point corresponds to use of the computation form of Fig. 14 in that the integrated voltage, V_s, is proportional to N_s. A means of expressing this voltage in dB is needed for the same reasons that the curve of Fig. 13 was derived to permit interpretation of N_s. This can be accomplished using an indicating meter with a dB scale. By making the meter circuit an "averaging" one with a time constant long compared to the sweep time of the source signal, deflections of the needle will be approximately proportional to the total number of loudness units. A dB scale corresponding to the conversion specified by the curve of Fig. 13 could be provided. This might be difficult since the constant of proportionality of the needle deflections to total number of loudness units is not readily obtainable. Instead, advantage was taken of another approximation.

Comparison of the curves of Figs. 6 and 13 shows that above their respective knees the shapes of these curves are the same. If we assume a straight-line relationship between L_s and N_s as was done earlier for the relation between n_s and Z_s, the addition of distortionless attenuation in any speech spectrum will produce a dB loudness level change equal to the change in attenuation. Thus, the meter scale can be so calibrated that it obeys the same law of addition as that provided by the compressor and linear integrator in converting elements of effective level to loudness, namely a 13.2-dB change in effective level would correspond to doubling the total number of loudness units. The scale on the meter would show a difference of 13.2 dB between half-scale and full-scale deflection of the needle, or between any two deflection points, the greater of which is twice the smaller. Consequently, the meter reading would reflect a change in distortionless loss in the telephone connection under test, dB for dB.

4.2 *Description of EARS*

A block diagram of EARS is shown on Fig. 18. The system consists of two parts: a signal source and a measuring subsystem. The latter includes provision for measuring both voltage and pressure.[3]

4.2.1 *Signal Source*

The logarithmic oscillator provides at its output terminals a signal which sweeps logarithmically with time from 300 Hz to 3300 Hz to 300 Hz and has a flat amplitude versus frequency characteristic. The reason for limiting the measuring band of the EARS to 300–3300 Hz is a practical one. The use of partial connection ratings as an engineering tool implicitly requires that, for any given connection, the sum of the partial ratings should approximately equal the overall rating. Thus, the bandwidth used to obtain these ratings should approximate the bandwidth of the most restrictive element(s) in the connection in order to avoid cumulating bandwidth penalties when summing partial ratings. The specific limits of 300 Hz and 3300 Hz were selected by reviewing bandwidth characteristics of various telephone equipments and facilities.

The oscillator sweep rate is 6 times per second where a sweep is defined in terms of the 300–3300-Hz band. Criteria leading to selec-

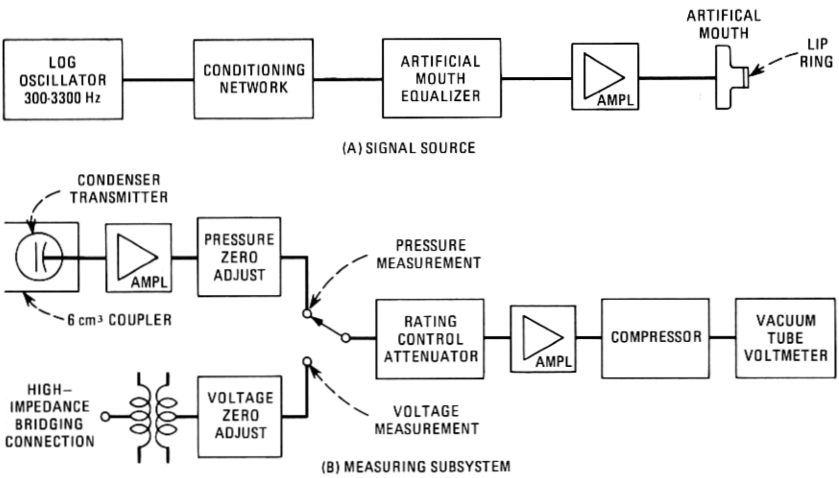

Fig. 18—Block diagram of the EARS.

tion of this sweep rate were (*i*) the sweep interval (1/sweep rate) should be small as compared to an easily realizable integrating time constant for the indicating meter and (*ii*) the rate should approximate the syllabic rate of speech. The latter characteristic appears desirable when measuring carbon transmitters in order to ensure that these operated at an efficiency comparable to that obtained under actual use conditions, i.e., with real speech applied. (Recent measurements indicate that the sweep rate can be changed over the range 2 to 10 sweeps per second without significantly changing the ratings of many telephone sets of modern design.)

The conditioning network (a 6-dB attenuator) is momentarily switched out of the source circuit prior to measuring carbon transmitters thus increasing the source signal level by 6 dB. The higher level is intended to condition the transmitters to operate at the proper level. Conditioning is not required when measuring linear transmitters.

The artificial mouth equalizer comprises a passive network whose frequency response is the inverse of the electric-to-acoustic response of the artificial mouth. The loss of this network is compensated for by the amplifier.

The artificial mouth used is a permanent magnet, moving coil loudspeaking unit, and is, for all practical purposes, the equivalent of an earlier proposal.[52] The mouth includes, as an integral part, a lip ring which is used as a reference for obtaining the proper spatial relationship between the artificial mouth and telephone instruments under test. The location of the lip ring has been empirically determined so as to correspond approximately to the plane of the lips of a human mouth.[52]

Ideally, the source arrangement should, with an oscillator output which is flat with frequency over the band of interest, provide an output pressure at the artificial mouth lip ring which is also flat with frequency.* Practically, control of the overall response to within ± 1 dB of the 1000-Hz value provides acceptable operation, introducing less than about 0.2 dB error in ratings.

4.2.2 *Measuring Subsystem*

The measuring subsystem is arranged to permit both voltage and pressure measurements. For voltage measurements, the input is a

* The pressure is measured with a Type L microphone.[53] The microphone is located in a carefully gauged position so selected that the pressure measured at that point corresponds closely to the pressure at the center of the lip ring opening.[54]

high-impedance transformer, and is bridged across the selected impedance (usually a 900-ohm resistor) terminating the telephone connection at the point where voltage measurement is desired. The transformer is connected to an attenuator, used in system calibration to compensate for gain drift of amplifiers, thence to a switch node.

The pressure measuring circuit consists of a 6-cm^3 test coupler equipped with a Type L pressure microphone.* The microphone is connected to a condenser microphone amplifier which provides bias voltage to the microphone and which produces at its output a voltage proportional to the pressure developed in the 6-cm^3 cavity by the telephone receiver under test. The amplifier is connected through a calibrating attenuator to a switch node.

The measurement mode is selected by operating the switch to "pressure measurement" or "voltage measurement." The switch swinger connects to an attenuator (designated Rating Control), amplifier, compressor, and vacuum tube voltmeter. The compressor design employed provides a 2.2-to-1 characteristic over a limited range. Operation is confined to this range by holding the compressor output voltage at a constant value, indicated on the vacuum tube voltmeter as the "reference" or "zero" point, and adjusting for rating changes using the Rating Control. This takes advantage of the fact that, as previously noted, flat attenuation changes ahead of the compressor equate, on a dB-for-dB basis, to loudness level changes.

4.2.3 *System Calibration*

System calibration consists of first removing the condenser microphone from the coupler and locating it at gauged position relative to the artificial mouth lip ring, then adjusting the source to deliver reference test pressure at that point. This adjustment is based on the condenser microphone calibration, and does not involve the EARS measuring subsystem. When the proper test pressure has been obtained, the EARS measuring system is switched to the pressure measuring mode, Rating Control is set at the "0" dB, and the "Pressure Adjust Attenuator" set to obtain reference reading on the vacuum tube voltmeter. Since the pressure spectrum being measured is flat with frequency, pressure level equates to loudness pressure level.

Calibration of the voltage measuring mode proceeds in a similar

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* The simple 6-cm^3 test coupler used, conforming in design to present standards (Fig. 3 of Ref. 55), has been found suitable for measurements of the type considered in this report. However, handsets with ear caps of unusual shape may require a different coupler configuration.

manner. A voltage signal, derived from the log oscillator, is applied to a 900-ohm resistor. The voltage developed is first read directly with a voltmeter. Then the high-impedance bridging connection of the EARS measuring subsystem is connected across the resistor, the system switched to the voltage measuring mode, and the Rating Control set to read the voltage measured with the voltmeter above, i.e., if the latter was −2 dB relative to 1 volt, then the rating control is set to read −2 dB relative to the "reference" or "zero" setting. The "Voltage Zero Adjust" is then set to obtain "reference" or "zero" reading on the measuring subsystem vacuum tube voltmeter. Since the voltage spectrum being measured is flat with frequency, voltage level equates to loudness voltage level.

4.2.4 *Rating Measurements*

Ratings of partial or overall telephone connections are established by (*i*) the reading of the Rating Control and (*ii*) the reference pressure and/or voltage levels employed. Examples of rating measurements, including signal levels employed with the present EARS, are given on Figs. 19, 20, and 21. The transmitting and receiving loops of Figs. 19 and 20 respectively are the same as those for the overall connection of Fig. 21. Sum of the component ratings indicates a loudness loss of 17.3 dB (= −18.3 + 25.6 + 10) while the overall rating from Fig. 16 is 17.1, a discrepancy of 0.2 dB. Had the measurement band been increased from 300–3300 Hz to, for example, 100–5000 Hz, the discrepancy would have been somewhat larger because of the bandpass response of both the transmitter and receiver.

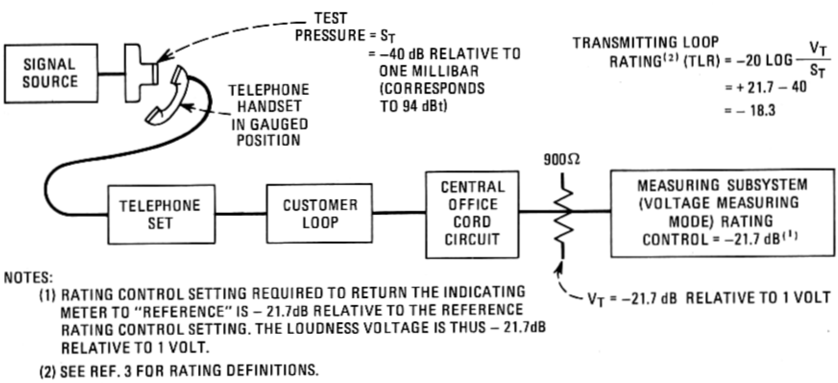
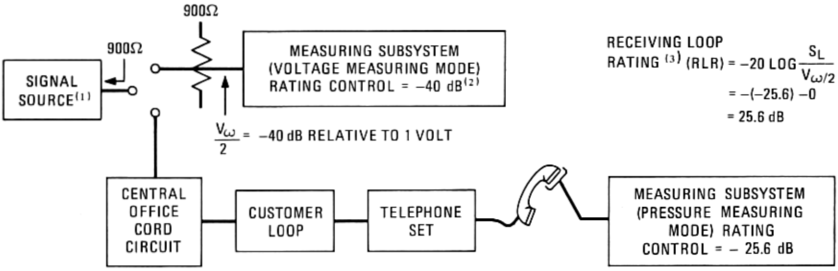


Fig. 19—Transmitting loop rating.



NOTES:

(1) SIGNAL SOURCE PROVIDES AN OUTPUT SIGNAL WHICH CONSISTS OF THE TEST SPECTRUM (FLAT WITH FREQUENCY) MODIFIED BY THE FREQUENCY RESPONSE OF COMPONENTS WHICH NORMALLY PRECEDE THE RECEIVING COMPONENT IN AN OVERALL CONNECTION. THIS IS NECESSARY IN ORDER TO PRESERVE ADDITIVITY OF COMPONENT RATINGS.

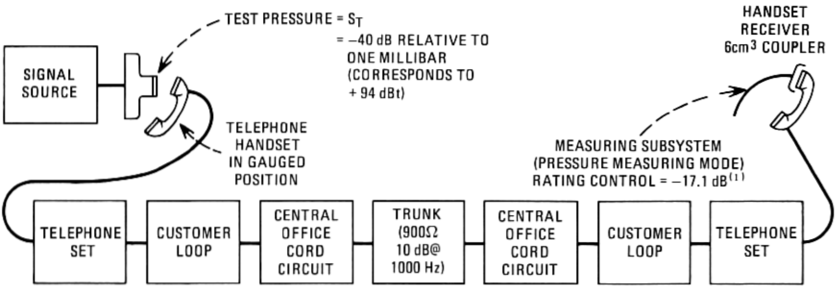
(2) RATING CONTROL SETTING RELATIVE TO REFERENCE SETTING IN ORDER THAT THE INDICATING METER NEEDLE IS AT REFERENCE FOR THE SELECTED TEST VOLTAGE. THE LOUDNESS TEST VOLTAGE IS THUS −40 dB RELATIVE TO 1 VOLT.

(3) SEE REF. 3 FOR RATING DEFINITIONS. THE TEST REFERENCE VOLTAGE IS SO SELECTED THAT IT IS NUMERICALLY EQUAL, IN dBre 1 VOLT, TO THE SELECTED TEST PRESSURE LEVEL IN dBt (SEE FIG. 19.) FOR ANY OTHER TEST VOLTAGE, THE EQUATION FOR RLR MUST CONTAIN A CONSTANT.

Fig. 20—Receiving loop rating.

4.3 *Graphical Computation of Loudness*

A form for graphical determination of loudness ratings from the amplitude response characteristics of telephone connections is shown on Fig. 22. This form is based on the same considerations as those leading to the EARS, and is discussed in this paper because it is the



NOTES:

(1) RATING CONTROL SETTING REQUIRED TO RETURN THE INDICATING METER TO "REFERENCE" IS 17.1 dB BELOW THE REFERENCE SETTING OF THE CONTROL. THUS, THE RECEIVED LOUDNESS PRESSURE IS 17.1 dB BELOW THE TEST PRESSURE LEVEL APPLIED AT THE TRANSMITTING END OF THE CONNECTION.

(2) SEE REF. 3 FOR RATING DEFINITIONS. NOTE THAT FOR OVERALL MEASUREMENTS, A POSITIVE RATING REPRESENTS A LOUDNESS LOSS.

OVERALL RATING (2) (OR) $= -20 \text{ LOG} \frac{S_2}{S_T}$
$= -(-17.1) + 0$
$= 17.1$ dB

Fig. 21—Overall rating.

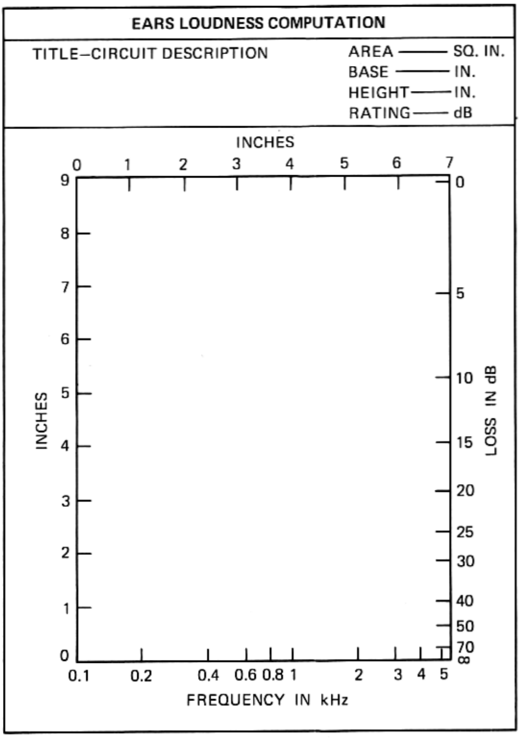


Fig. 22—Graph paper for computing loudness ratings.

vehicle used to study the effects of differences between the loudness computation method and the EARS.

The lower abscissa scale is frequency in Hz, corresponding on a logarithmic basis to the upper abscissa scale in inches. (Use of inches is arbitrary; any length units could be used.) Thus, equal increments on the upper scale correspond to equal increments of log f_2/f_1 where $f_2 > f_1$. (This reflects the straight-line approximation shown on Fig. 17). The equal distance increments closely approximate bands of the $B_{90} - X$ spectrum which are interpreted to be of equal loudness when listened to by the human ear.

The right-hand ordinate scale is proportional to the 2.2 root of a voltage, current, or pressure. This scale and grid are constructed according to the equation

$$44 \log 2X = 55 - L \tag{11}$$

where

X = inches (left-hand ordinate scale)
L = loss of circuit in dB (right-hand ordinate scale).

(Equation (11) assumes that the reference input to the connection is flat with frequency.) The ordinate is measured in inches from the bottom line of the graph which corresponds to zero output voltage, current, or pressure. Correspondence between ordinate scales is demonstrated in Table XIII. Boxes at the top of the sheet are provided for recording graph measurements.

In order to use the graph paper, the loss (in dB) of the partial or overall telephone connection must be known over the band of interest. This loss data may be in terms of

$$\frac{\text{input pressure}}{\text{output pressure}}, \qquad \frac{\text{input voltage}}{\text{output voltage}}, \qquad \frac{\text{input pressure (millibars)}}{\text{output voltage (volts)}},$$

or

$$\frac{\text{input voltage (volts)}}{\text{output pressure (millibars)}}.^*$$

The loss frequency characteristic is plotted on the graph paper. The right-hand ordinate scale may be adjusted by a constant for negative losses, i.e., gains. Where large losses are encountered, greater accuracy can be obtained by similarly applying an adjustment constant. To illustrate, if the lowest loss across the band of interest is −15 dB, the values along the right-hand ordinate scale should have 15 subtracted from them. If, on the other hand, the lowest loss is +15 dB, the right-hand ordinate values should be increased by 15.

The average height of the response as plotted on the graph paper is determined by measuring the area (in square inches) under the curve and dividing by the base width (in inches). The average height is then located on the left-hand scale and the corresponding dB value read from the right-hand scale. This dB value is the rating based on reference input level.

For cases where the input spectrum is not flat with frequency, the graphical computation of a circuit rating involves determining two areas. The first of these is obtained from a plot of the actual input spectrum on the graph paper, the second from a plot of the input

* See Ref. 3.

TABLE XIII— RELATION BETWEEN ORDINATE DISTANCE AND LOSS SCALES

Left-Hand Ordinate Scale-Distance (Inches)	Input/Output	Right-Hand Ordinate Scale-Loss (dB)
0	∞	∞
0.5	561	55
5	3.55	11
8.89	1	0

spectrum modified by the circuit response. The loudness rating is then the dB difference between the losses computed for these areas.

V. COMPARISON OF GRAPHICALLY DETERMINED RATINGS AND OBSERVED RESULTS

The speech loudness computation method accurately predicts loudness performance of telephone connections (see Section 3.5). Laboratory realization of this method, the EARS, involved a number of simplifying assumptions. Thus, validation of the EARS approach requires considering (*i*) the accuracy of the EARS in predicting loudness performance and (*ii*) the effects of the various simplifying assumptions. We do this by comparing computed ratings and observed results for the subjective tests of Section 3.5. (Numerous other tests have been performed which support the EARS concept, but these are not considered because either the tests were limited, i.e., incomplete test designs, small observer groups, etc., or the test system amplitude response characteristics are not known.)

TABLE XIV—FEATURES OF THE SPEECH LOUDNESS COMPUTATION METHOD AND THE EARS METHOD

Feature	Computation Method	EARS Method
Loudness Law	Figs. 6 and 8	Linear Portions of Figs. 6 and 8
Spread-of-Loudness Correction	Yes	No
Analysis Bandwidth	100–5000 Hz	300–3300 Hz
Reference Bandwidth	250–4000 Hz	300–3300 Hz
Rating Definition	Orthotelephonic Terms (Fig. 3)	Artificial Voice/6-cm^3 Coupler Terms (Fig. 16)

Validation of the EARS utilized the graphical computation method described in Section 4.3. This approach was necessary because the systems used in the tests were not available for direct measurement using the EARS. However, the frequency response characteristics of these systems were known and, thus, their loudness ratings could be computed.

In theory, the graphical form (Fig. 22) and the computation form (Fig. 14) should provide about the same results. The reference of the former (the horizontal line at 0 dB of loss) corresponds to that of the latter (entries of column 2) because of the relation between frequency weightings of the two references. There is, however, one important difference between the two forms. For the graphical form, the loudness versus effective level relation has a constant slope (straight-line portion of Fig. 6) while for the computation form, this relation has a pronounced change in slope at low effective levels (see Fig. 6). The effects of this difference should be most apparent at low received speech levels where the graphical method would indicate a higher received speech loudness than would the computational method.

With the above in mind, we can now consider the ratings for the various tests under a number of different conditions. These conditions, listed in Table XIV, reflect the differences in features of the speech loudness computation method and the EARS.

Computed and observed ratings are given in Tables XV through XIX in terms of loudness loss (positive entries) or loudness gain (negative entries). The arrows and associated numbers below the tabular entries refer to distributions of differences between the various conditions considered.

We will consider Table XV in some detail, noting that since the other tables have an identical arrangement, such discussion will in general similarly apply to these tables. Referring to Table XV, column 1 designates the network tested, and is repeated from Table V. Column 2 designates the setting of the reference network for which the test observers judged the test and reference networks to provide equally loud speech. Columns 3 through 9 each contain 3 subcolumns; "a" and "b" are the loudness ratings of the test and reference channels respectively and "c" represents the error, equal to "a" entries minus "b" entries.

Column 3 entries are essentially a repeat of the information contained in Table V, obtained by converting the Table V entries to L_s values, then subtracting these from 89.4 dBt, the level of the reference speech spectrum of Fig. 2 transmitted over a system that is flat

TABLE XV—COMPUTED LOUDNESS LOSSES (dB) FOR STEINBERG LINEAR SYSTEM TESTS (1924) (See also Table V)

1	2	3			4			5			6			7			8			9		
		a	b	c	a	b	c	a	b	c	a	b	c	a	b	c	a	b	c	a	b	c
					GRAPHICAL COMPUTATION (Fig. 22)																	
					Orthotelephonic Responses												Artificial Voice/6-cm³ Coupler Responses					
					Same as Computation Form Except																	
	Observed Ref. Ckt. Setting (dB)	Computation Form (Fig. 14)			Linear Loudness Law			Linear Loudness Law, No Spread-of-Loudness			Analysis Band = Reference Band = 100–5000 Hz			Analysis Band = Reference Band = 300–3300 Hz			Analysis Band = Reference Band = 100–5000 Hz			Analysis Band = Reference Band = 300–3300 Hz		
Test System		Test	Ref.	Δ	Test	Ref.	Δ	Test	Ref.	Δ	Test	Ref.	Δ	Test	Ref.	Δ	Test	Ref.	Δ	Test	Ref.	Δ
Sensation Level = 100 dB																						
125 Hz HPF	−22.0	−17.5	−17.1	−0.4	−17.3	−17.7	0.4	−18.9	−19.3	0.4	−12.3	−12.8	0.5	−15.4	−14.8	−0.6	−4.0	−4.1	0.1	−4.5	−3.9	−0.6
250 Hz HPF	−20.6	−15.0	−15.7	0.7	−14.5	−16.3	1.8	−15.9	−17.9	−2.0	−9.3	−11.4	2.1	−15.3	−13.4	−1.9	0.6	−2.7	3.3	−4.4	−2.5	−1.9
375 Hz HPF	−19.0	−12.7	−14.1	1.4	−12.5	−14.7	2.2	−13.7	−16.3	2.6	−6.7	−9.8	3.1	−13.4	−13.0	−0.4	2.6	−1.1	3.7	−2.7	−0.9	−1.8
500 Hz HPF	−17.6	−10.7	−12.7	2.0	−10.5	−13.3	2.8	−11.2	−14.9	3.7	−4.6	−8.4	3.8	−10.5	−10.6	0.1	4.8	0.3	4.5	−0.1	0.5	−0.6
625 Hz HPF	−15.0	−8.7	−10.1	1.4	−8.8	−10.7	1.9	−9.4	−12.3	2.9	−2.6	−5.8	3.2	−8.3	−8.0	−0.3	6.1	2.9	3.2	1.6	3.1	−1.5
750 Hz HPF	−14.1	−6.9	−9.2	2.3	−6.8	−9.8	3.0	−7.4	−11.4	4.0	−0.8	−4.9	4.1	−5.8	−7.1	1.3	7.1	3.8	3.3	3.0	4.0	−1.0
1000 Hz HPF	−12.1	−3.9	−7.2	3.3	−4.7	−7.8	3.1	−4.0	−9.4	5.4	2.6	−2.9	5.5	−1.5	−5.1	3.6	10.4	5.8	4.6	7.4	6.0	1.4
1250 Hz HPF	−9.4	−1.5	−4.5	3.0	−1.5	−5.1	3.6	−0.3	−6.7	6.4	6.2	−0.2	6.4	3.5	−2.4	5.9	13.2	8.5	4.7	11.2	8.7	2.5
1500 Hz HPF	−6.0	0.8	−1.1	1.9	1.7	−1.7	3.4	3.2	−3.3	6.5	9.8	3.2	6.6	8.8	1.0	7.8	16.1	11.9	4.2	15.0	12.1	2.9
750 Hz LPF	−12.7	−5.1	−7.8	2.7	−4.8	−8.4	3.6	−3.6	−10.0	6.4	3.0	−3.5	6.5	5.0	−5.7	10.7	11.2	5.2	6.0	18.7	5.4	13.3
1000 Hz LPF	−15.2	−8.3	−10.3	2.0	−7.3	−10.9	3.6	−7.1	−12.5	5.4	−0.3	−6.0	5.7	−0.6	−8.2	7.6	8.7	2.7	6.0	12.1	2.9	9.2
1250 Hz LPF	−16.2	−9.7	−11.3	1.6	−9.5	−11.9	2.4	−9.5	−13.5	4.0	−2.8	−7.0	4.2	−4.5	−9.2	4.7	7.7	1.7	6.0	11.0	1.9	9.1
1500 Hz LPF	−18.2	−11.2	−13.3	2.1	−11.1	−13.9	2.8	−11.3	−15.5	4.2	−4.7	−9.0	4.3	−7.2	−11.2	4.0	4.7	−0.3	5.0	5.7	−0.1	5.8
2000 Hz LPF	−19.6	−13.4	−14.7	1.3	−13.1	−15.3	2.2	−13.9	−16.9	3.0	−7.3	−10.4	3.1	−10.8	−12.6	1.8	2.6	−1.7	4.3	2.0	−1.5	3.5
2500 Hz LPF	−20.2	−15.2	−15.3	0.1	−14.3	−15.9	1.6	−15.7	−17.5	1.8	−9.1	−11.0	1.9	−13.3	−13.2	−0.1	0.2	−2.3	2.5	−1.5	−2.1	0.6
3000 Hz LPF	−20.7	−16.2	−15.8	−0.4	−15.3	−16.4	1.1	−16.4	−18.0	1.6	−9.9	−11.5	1.6	−14.3	−13.7	−0.6	−0.8	−2.8	2.0	−3.0	−2.6	−0.4
4000 Hz LPF	−21.9	−17.3	−17.0	−0.3	−16.8	−17.6	0.8	−18.1	−19.2	1.1	−11.5	−12.7	1.2	−15.5	−14.9	−0.6	−3.8	−4.0	0.2	−4.2	−3.8	−0.4
Error: Average				1.4			2.4			3.6			3.7			2.5			3.8			2.4
σ				1.1			1.0			1.9			1.9			3.6			1.7			4.3
Difference Distributions					0.2, 0.5			−0.5, 1.0			6.6			−3.3, 1.8						−1.0, 3.1		
															8.7, 1.0							
																	+11.0, 2.1					
Sensation Level = 39 dB																						
125 Hz HPF	38.4	47.0	46.8	0.2	43.7	42.7	1.0	42.1	41.1	1.0	48.7	47.6	1.1	45.6	45.4	0.2	57.0	56.3	0.7	56.5	56.5	0
250 Hz HPF	38.8	47.8	47.2	0.6	45.5	43.1	2.4	45.1	41.5	3.6	51.7	48.0	3.7	45.7	45.8	−0.1	61.6	56.7	4.9	56.6	56.9	−0.3
375 Hz HPF	38.2	48.3	46.6	1.7	48.5	42.5	6.0	47.3	40.9	6.4	54.3	47.4	6.9	47.6	45.2	2.4	63.6	56.1	7.5	58.3	56.3	2.0
500 Hz HPF	39.4	51.6	47.8	3.8	50.5	43.7	6.8	49.8	42.1	7.7	56.4	48.6	7.8	50.5	46.4	4.1	65.8	57.3	8.5	60.4	57.5	2.9
625 Hz HPF	41.4	52.6	49.8	2.8	52.2	45.7	6.5	51.6	44.1	7.5	58.4	50.6	7.8	52.7	48.4	4.3	67.1	59.3	7.8	62.6	59.5	3.1
750 Hz HPF	39.0	54.2	47.4	6.8	54.2	43.3	10.9	53.6	41.7	11.9	60.2	48.2	12.0	55.2	46.0	9.2	68.1	56.9	11.2	64.0	57.1	6.9
1000 Hz HPF	41.6	56.6	50.0	6.6	56.3	45.9	10.4	57.0	44.3	12.7	63.6	50.8	12.8	59.5	48.6	10.9	71.4	59.5	11.9	68.4	59.7	8.7
1250 Hz HPF	42.9	58.1	51.3	6.8	59.5	47.2	12.3	60.7	45.6	15.1	67.2	52.1	15.1	64.5	49.9	14.6	74.2	60.8	13.4	72.2	61.0	11.2
1500 Hz HPF	43.7	59.8	52.1	7.7	62.7	48.0	14.7	64.2	46.4	17.8	70.8	52.9	17.9	69.8	50.7	19.1	77.1	61.6	15.5	76.0	61.8	14.2
750 Hz LPF	48.1	61.2	56.5	4.7	56.2	52.4	3.8	57.4	50.8	6.6	64.0	57.3	6.7	66.0	55.1	10.9	72.2	66.0	6.2	79.7	66.2	13.5
1000 Hz LPF	45.7	58.4	54.1	4.3	53.7	50.0	3.7	53.9	48.4	5.5	60.7	54.9	5.8	60.4	52.7	7.7	69.7	63.6	6.1	73.1	63.8	9.3
1250 Hz LPF	43.7	56.8	52.1	4.7	51.5	48.0	3.5	51.5	46.4	5.1	58.2	52.9	5.3	56.5	50.7	5.8	68.7	61.6	7.1	72.0	61.8	10.2
1500 Hz LPF	42.1	55.4	50.5	4.9	49.9	46.4	3.5	49.7	44.8	4.9	56.3	51.3	5.0	53.8	49.1	4.7	65.7	60.0	5.7	66.7	60.2	6.5
2000 Hz LPF	39.2	53.4	47.6	5.8	47.9	43.5	4.4	47.1	41.9	5.2	53.7	48.4	5.3	50.2	46.2	4.0	63.6	57.1	6.5	63.0	57.3	5.7
2500 Hz LPF	39.8	51.1	48.2	2.9	46.7	44.1	2.6	45.3	42.5	2.8	51.9	49.0	2.9	47.7	46.8	0.9	61.2	57.7	3.5	59.5	57.9	1.6
3000 Hz LPF	38.6	49.8	47.0	2.8	45.7	42.9	2.8	44.6	41.3	3.3	51.1	47.8	3.3	46.7	45.6	1.1	60.2	56.5	3.7	58.0	56.7	1.3
4000 Hz LPF	37.9	47.2	46.3	0.9	44.2	42.2	2.0	42.9	40.6	2.3	49.5	47.1	2.4	45.5	44.9	0.6	57.2	55.8	1.4	56.8	56.0	0.8
Error: Average				4.0			5.7			7.0			7.8			6.0			7.1			5.7
σ				2.2			3.9			4.6			4.6			5.3			3.9			4.6
Difference Distributions					−2.5, 2.5			−0.3, 1.0			6.6			−3.4, 2.2						−1.1, 3.3		
															8.7, 1.0							
																	11, 2.1					

TABLE XVI—COMPUTED LOUDNESS LOSSES (dB) FOR VAN WYNEN LINEAR SYSTEM TESTS (See also Table VI)

1	2	3			4			5			6			7			8			9		
		a	b	c	a	b	c	a	b	c	a	b	c	a	b	c	a	b	c	a	b	c
											GRAPHICAL COMPUTATION (Fig. 22)											
					Orthotelephonic Responses												Artificial Voice/6-cm³ Coupler Responses					
					Same as Computation Form Except																	
	Observed Ref. Ckt. Setting (dB)	Computation Form (Fig. 14)			Linear Loudness Law			Linear Loudness Law, No Spread-of-Loudness			Analysis Band = Reference Band = 100–5000 Hz			Analysis Band = Reference Band = 300–3300 Hz			Analysis Band = Reference Band = 100–5000 Hz			Analysis Band = Reference Band = 300–3300 Hz		
Test System		Test	Ref.	Δ	Test	Ref.	Δ	Test	Ref.	Δ	Test	Ref.	Δ	Test	Ref.	Δ	Test	Ref.	Δ	Test	Ref.	Δ
2700 Hz LPF	24.2	8.0	8.6	−0.6	7.9	8.9	−1.0	6.9	7.6	−0.7	13.4	14.3	−0.9	10.6	12.5	−1.9	21.1	21.6	−0.5	20.2	21.7	−1.5
1750 Hz LPF	28.3	11.1	12.8	−1.7	10.9	13.0	−2.1	9.9	11.7	−1.8	16.5	18.4	−1.9	15.0	16.6	−1.6	24.6	25.7	−1.1	25.9	25.8	0.1
1000 Hz LPF	31.3	15.3	16.0	−0.7	16.3	16.0	0.3	15.9	14.7	1.2	22.5	21.4	1.1	23.8	19.6	4.2	29.7	28.7	1.0	34.2	28.8	5.4
250 Hz HPF	25.6	10.3	10.2	0.1	10.3	10.3	0	9.1	9.0	0.1	15.7	15.7	0	9.3	13.9	−4.6	25.5	23.0	2.5	19.4	23.1	−3.7
550 Hz HPF	30.8	16.3	15.4	0.9	14.9	15.5	−0.6	14.0	14.2	−0.2	20.7	20.9	−0.2	14.1	19.1	−5.0	29.2	28.2	1.0	22.8	28.3	−5.5
1000 Hz HPF	36.4	21.8	21.3	0.5	21.2	21.1	0.1	22.0	19.8	2.2	28.5	26.5	2.0	23.3	24.7	−1.4	36.0	33.8	2.2	31.2	33.9	−2.7
250–2700 Hz BPF	29.0	14.4	13.4	1.0	14.5	13.7	0.8	13.6	12.4	1.2	20.4	19.1	1.3	11.9	17.3	−5.4	30.0	26.4	3.6	21.4	26.5	−5.1
550–1750 Hz BPF	38.9	26.9	23.8	3.1	26.8	23.6	3.2	20.5	22.3	1.8	33.1	29.0	4.1	23.8	27.2	−3.4	43.0	36.3	6.7	33.6	36.4	−2.8
1000 Hz Peak																						
15 dB	26.8	9.2	11.3	−2.1	9.7	11.5	−1.8	8.0	10.2	−2.2	14.7	16.9	−2.2	11.0	15.1	−4.1	22.5	24.2	−1.7	20.3	24.3	−4.0
30 dB	24.2	7.5	8.6	−1.1	7.5	8.9	−1.4	5.6	7.6	−2.0	12.4	14.3	−1.9	7.0	12.5	−5.5	20.8	21.6	−0.8	16.2	21.7	−5.5
56 dB	34.5	19.1	19.3	−0.2	19.0	19.2	−0.2	17.2	17.9	−0.7	24.0	24.6	−0.6	17.0	22.8	−5.8	30.5	31.9	−1.4	26.3	32.0	−5.7
700 Hz Peak																						
30 dB	25.8	8.0	10.4	−2.4	8.4	10.5	−2.1	6.8	9.2	−2.4	13.4	15.9	−2.5	8.5	14.1	−5.6	21.7	23.2	−1.5	17.7	23.3	−5.6
2000 Hz Peak																						
30 dB	28.7	6.6	13.2	−6.6	7.3	13.4	−6.1	5.8	12.1	−6.3	12.3	18.8	−6.5	7.2	17.0	−9.8	19.8	26.1	−6.3	15.4	26.2	−10.8
Falling Loss	25.6	10.2	10.2	0	10.8	10.3	0.5	9.4	9.0	0.4	16.0	15.7	0.3	12.5	13.9	−1.4	23.8	23.0	0.8	21.5	23.1	−1.6
Rising Loss	26.4	8.8	10.9	−2.1	9.3	11.1	−1.8	7.5	9.8	−2.3	14.1	16.5	−2.4	14.8	14.7	0.1	21.4	23.8	−2.4	25.1	23.9	1.2
Error: Average				−0.8			−0.8			−0.6			−0.6			−3.5			0.1			−3.3
σ				2.1			2.0			2.6			2.4			3.1			2.9			3.6

Difference Distributions

3 ⟷ 4: 0,0.5
4 ⟷ 5: −1.1,0.7
5 ⟷ 6: 6.6
6 ⟷ 7: −4.4,3.0
6 ⟷ 8: 8.1,0.9
7 ⟷ 9: 9.4,0.8
8 ⟷ 9: −3.0,3.9

TABLE XVII—COMPUTED LOUDNESS LOSSES (dB) FOR VAN WYNEN TELEPHONE SET TESTS (1940)
(See also Table VII)

1		2	3			4			5			6			7			8			9		
			a	b	c	a	b	c	a	b	c	a	b	c	a	b	c	a	b	c	a	b	c
						GRAPHICAL COMPUTATION (Fig. 22)																	
						Orthotelephonic Responses												Artificial Voice/6-cm³ Coupler Responses					
Test System						Same as Computation Form Except																	
Test Tele. Set	Test Network	Observed Ref. Ckt. Setting (dB)	Computation Form (Fig. 14)			Linear Loudness Law			Linear Loudness Law, No Spread-of-Loudness			Analysis Band = Reference Band = 100–5000 Hz			Analysis Band = Reference Band = 300–3300 Hz			Analysis Band = Reference Band = 100–5000 Hz			Analysis Band = Reference Band = 300–3300 Hz		
			Test	Ref.	Δ	Test	Ref.	Δ	Test	Ref.	Δ	Test	Ref.	Δ	Test	Ref.	Δ	Test	Ref.	Δ	Test	Ref.	Δ
A	5dB-4000 Hz LPF	16.8	13.1	12.6	0.5	12.6	12.5	0.1	12.4	10.5	1.9	18.5	17.1	1.4	10.5	10.5	0	26.2	23.3	2.9	18.7	18.0	0.7
	25dB-4000 Hz LPF	35.2	34.1	31.7	2.4	32.6	30.9	1.7	32.4	28.9	3.5	38.5	35.5	3.0	30.5	28.9	1.6	46.2	41.7	4.5	38.7	36.4	2.3
B	5dB-4000 Hz LPF	11.4	6.7	7.2	−0.5	6.7	7.1	−0.4	7.7	5.1	2.6	14.7	11.7	3.0	7.2	5.1	2.1	23.4	17.9	5.5	16.3	12.6	3.7
	25dB-4000 Hz LPF	31.5	27.2	27.7	−0.5	26.7	27.2	−0.5	27.7	25.2	2.5	34.7	31.8	2.9	27.2	25.2	2.0	43.4	38.0	5.4	36.3	32.7	3.6
	5dB-Rising Loss	11.7	6.9	7.6	−0.7	6.5	6.9	−0.4	7.5	4.9	2.6	14.5	11.5	3.0	8.1	6.4	1.7	23.5	17.4	6.1	17.3	15.2	2.1
	25dB-Rising Loss	31.2	27.9	27.2	0.7	26.5	26.4	0.1	27.5	24.4	3.1	34.5	31.0	3.5	28.1	25.9	2.2	43.5	36.9	6.6	37.3	34.7	2.6
Error: Average					0.3			0.1			2.7			2.8			1.6			5.2			2.5
σ					1.1			0.8			0.5			0.7			0.7			1.2			1.0

Difference Distributions

	3 ↔ 4	4 ↔ 5	5 ↔ 6	6 ↔ 7	6 ↔ 8	7 ↔ 9	8 ↔ 9
Test Telephone Set A	−1.0,0.5	−0.2	6.1	−8.0	7.7	8.2	−7.5
Test Telephone Set B	−0.6,0.5	+1.0	7.0	−7.0,0.5	8.8	9.1	−6.6,0.5
Reference Telephone Set	−0.5,0.3	−2.0	6.6	−6.1,0.7	6.1	7.9,0.6	−4.3,1.5

Table XVIII—Computed Loudness Losses (dB) for Van Wynen Tie Line System Tests (See also Table VIII)

1	2	3			4			5			6			7			8			9		
		a	b	c	a	b	c	a	b	c	a	b	c	a	b	c	a	b	c	a	b	c
											GRAPHICAL COMPUTATION (Fig. 22)											
					Orthotelephonic Responses												Artificial Voice/6-cm³ Coupler Responses					
					Same as Computation Form Except																	
	Observed Ref. Ckt. Setting (dB)	Computation Form (Fig. 14)			Linear Loudness Law			Linear Loudness Law, No Spread-of-Loudness			Analysis Band = Reference Band = 100–5000 Hz			Analysis Band = Reference Band = 300–3300 Hz			Analysis Band = Reference Band = 100–5000 Hz			Analysis Band = Reference Band = 300–3300 Hz		
Test System		Test	Ref.	Δ	Test	Ref.	Δ	Test	Ref.	Δ	Test	Ref.	Δ	Test	Ref.	Δ	Test	Ref.	Δ	Test	Ref.	Δ
Overall-DL Reference																						
20-3000 LPF	21.2	15.2	15.6	−0.4	14.3	14.6	−0.3	13.4	13.4	0	19.9	20.0	−0.1	11.6	12.5	−0.9	25.9	26.5	−0.6	18.9	19.5	−0.6
20-2400 LPF	22.5	17.1	16.9	0.2	15.7	15.9	−0.2	15.1	14.7	0.4	21.7	21.3	0.4	13.4	13.8	−0.4	28.4	27.8	0.6	21.6	20.8	0.8
20-1900 LPF	24.7	20.1	19.1	1.0	17.6	18.1	−0.5	18.1	16.9	1.2	24.6	23.5	1.1	16.7	16.0	0.7	32.5	30.0	2.5	25.6	23.0	2.6
	26.0		20.4	−0.3		19.4	−1.8		18.2	−0.1		24.8	−0.2		17.3	−0.6		31.3	1.2		24.3	1.3
20-RL1	22.0	16.0	16.4	−0.4	15.5	15.4	0.1	13.9	14.2	−0.3	20.4	20.8	−0.4	12.9	13.3	−0.4	26.8	27.3	−0.5	20.2	20.3	−0.1
20-RL2	23.3	17.3	17.7	−0.4	17.0	16.7	0.3	15.1	15.5	−0.4	21.6	22.1	−0.5	14.3	14.6	−0.3	28.7	28.6	0.1	22.5	21.6	0.9
20-RL3	24.4	17.7	18.8	−1.1	17.3	17.8	−0.5	15.3	16.6	−1.3	21.9	23.2	−1.3	14.9	15.7	−0.8	28.9	29.7	−0.8	23.2	22.7	0.5
Overall-Peaked Reference																						
20-DL	6.3	14.4	13.4	1.0	13.4	12.1	1.3	12.2	10.7	1.5	18.8	17.2	1.6	11.3	9.0	2.3	25.3	24.4	0.9	18.3	16.5	1.8
	6.8		13.9	0.5		12.6	0.8		11.2	1.0		17.7	1.1		9.5	1.8		24.9	0.4		17.0	1.3
20-3000 LPF	6.9	15.2	14.0	1.2	14.3	12.7	1.6	13.4	11.3	2.1	19.9	17.8	2.1	11.6	9.6	2.0	25.9	25.0	0.9	18.9	17.1	1.8
20-2400 LPF	9.6	17.1	16.7	0.4	15.7	15.4	0.3	15.1	14.0	1.1	21.7	20.5	1.2	13.4	12.3	1.1	28.4	27.7	0.7	21.6	19.8	1.8
20-1900 LPF	13.8	20.1	20.9	−0.8	17.6	19.6	−2.0	18.1	18.2	−0.1	24.6	24.7	−0.1	16.7	16.5	0.2	32.5	31.9	0.6	25.6	24.0	1.6
	12.9		20.0	0.1		18.7	−1.1		17.3	0.8		23.8	0.8		15.6	1.1		31.0	1.5		23.1	2.5
20-RL1	8.7	16.0	15.8	0.2	15.5	14.5	1.0	13.9	13.1	0.8	20.4	19.6	0.8	12.9	11.4	1.5	26.8	26.8	0	20.2	18.9	1.3
20-RL2	10.4	17.3	17.5	−0.2	17.0	16.2	0.8	15.1	14.8	0.3	21.6	21.3	0.3	14.3	13.1	1.2	28.7	28.5	0.2	22.5	20.6	1.9
20-RL3	11.3	17.7	18.4	−0.7	17.3	17.1	0.2	15.3	15.7	−0.4	21.9	22.2	−0.3	14.9	14.0	0.9	28.9	29.4	−0.5	23.2	21.5	1.7
Sidetone-DL Reference																						
Ref. DL	0.7	−7.4	−5.4	−2.0	−7.3	−5.5	−1.8	−8.0	−7.1	−0.9	−2.2	−0.5	−1.7	−9.4	−8.2	−1.2	4.1	5.8	−1.7	−2.9	−1.1	−1.8
−10 DL	11.5	5.6	6.4	−0.8	5.1	5.7	−0.6	3.7	4.3	−0.6	10.2	10.8	−0.6	2.5	3.8	−1.3	16.6	17.1	−0.5	9.7	10.2	−0.5
−20 DL	19.2	13.9	14.1	−0.2	12.2	13.3	−1.1	10.8	11.8	−1.0	17.3	18.5	−0.8	9.9	10.9	−1.0	24.9	24.8	−0.1	17.8	18.0	−0.2
R 900	5.4	0.2	−0.6	0.8	0	−1.1	1.1	−1.4	−2.4	1.0	4.3	4.2	0.1	−1.7	−3.3	−1.6	11.3	10.7	0.6	5.1	3.8	1.3
R 1300	5.4	−1.6	−0.6	−1.0	−1.0	−0.9	−0.1	−2.4	−2.3	−0.1	4.2	4.3	−0.1	−2.9	−3.2	0.3	10.5	10.6	−0.1	4.0	3.9	0.1
R 1800	5.2	−2.6	−0.5	−2.1	−2.4	−1.0	−1.4	−3.7	−2.4	−1.3	2.8	4.3	−1.5	−4.5	−3.3	−1.2	9.6	10.5	−0.9	2.8	3.9	−1.1
R 3200-A	12.2	5.9	7.0	−1.1	4.8	6.3	−1.5	3.3	4.9	−1.6	10.0	11.5	−1.5	1.7	4.2	−2.5	17.4	18.1	−0.7	9.6	11.2	−1.6
R 3200-B	12.2	4.8	6.9	−2.1	4.2	6.3	−2.1	2.8	4.9	−2.1	9.3	11.5	−2.2	2.9	4.2	−1.3	16.4	17.8	−1.4	10.2	11.0	−0.8
RL4	5.7	−0.9	−0.2	−0.7	−1.6	−0.8	−0.8	−3.1	−2.3	−0.8	3.5	4.3	−0.8	−4.0	−3.2	−0.8	9.8	9.8	0	2.8	3.0	−0.2
Sidetone Peaked Reference																						
Ref. DL	−13.7	−7.4	−6.8	−0.6	−7.3	−7.2	−0.1	−8.0	−9.6	1.6	−2.2	−2.2	0	−9.4	−11.2	1.8	4.1	4.3	−0.2	−2.9	−3.8	0.9
−10 DL	−1.6	5.6	6.1	−0.5	5.1	4.7	+0.4	3.7	3.3	0.4	10.2	9.8	0.4	2.5	1.6	0.9	16.1	17.0	−0.9	9.7	9.2	0.5
−20 DL	5.4	13.9	12.4	1.5	12.2	11.0	1.2	10.8	9.7	1.1	17.3	16.3	1.0	9.9	7.9	2.0	24.9	23.3	1.6	17.8	15.4	2.4
R 900	−6.9	0.2	0.3	−0.1	0	−1.1	1.1	−1.4	−2.5	1.1	4.3	4.1	0.2	−1.7	−4.2	2.5	11.3	11.3	0	5.1	3.4	1.7
R 1300	−8.6	−1.6	−1.5	−0.1	−1.0	−3.0	2.0	−2.4	−4.4	2.0	4.2	2.1	2.1	−2.9	−6.0	3.1	10.5	9.6	0.9	4.0	1.7	2.3
R 1800	−8.2	−2.6	−0.3	−2.3	−2.4	−2.3	−0.1	−3.7	−3.7	0	2.8	2.9	−0.1	−4.5	−5.4	0.9	9.6	10.5	−0.9	2.8	2.2	0.6
R 3200-A	−0.8	5.9	6.7	−0.8	4.8	5.3	−0.5	3.3	3.8	−0.5	10.0	10.4	−0.4	1.7	2.2	−0.5	17.4	17.8	−0.4	9.6	9.9	−0.3
R 3200-B	−2.1	4.8	5.5	−0.7	4.2	4.1	0.1	2.8	2.7	0.1	9.3	9.3	0	2.9	1.1	1.8	16.4	16.7	−0.3	10.2	8.8	1.4
RL4	−8.2	−0.9	−1.3	0.4	−1.6	−2.7	1.1	−3.1	−4.0	0.9	3.5	2.5	1.0	−4.0	−5.5	1.5	9.8	9.8	0	2.8	1.8	1.0
Error: Average				−0.4			−0.1			0.2			0			0.3			0.1			0.8
σ				0.9			1.1			1.0			1.0			1.4			0.9			1.2

Difference Distributions

−0.8,0.6

−1.4,0.5

6.5

−7.7,0.6

6.6,0.5

7.4,0.5

−7.1,0.7

TABLE XIX—COMPUTED LOUDNESS LOSSES (dB) FOR FRASER TESTS (1946) (See also Table IX)

1	2	3			4			5			6			7			8			9		
		a	b	c	a	b	c	a	b	c	a	b	c	a	b	c	a	b	c	a	b	c
					GRAPHICAL COMPUTATION (Fig. 22)																	
					Orthotelephonic Responses												Artificial Voice/6-cm³ Coupler Responses					
					Same as Computation Form Except																	
	Observed Ref. Ckt. Setting (dB)	Computation Form (Fig. 14)			Linear Loudness Law			Linear Loudness Law, No Spread-of-Loudness			Analysis Band = Reference Band = 100–5000 Hz			Analysis Band = Reference Band = 300–3300 Hz			Analysis Band = Reference Band = 100–5000 Hz			Analysis Band = Reference Band = 300–3300 Hz		
Test System		Test	Ref.	Δ	Test	Ref.	Δ	Test	Ref.	Δ	Test	Ref.	Δ	Test	Ref.	Δ	Test	Ref.	Δ	Test	Ref.	Δ
A	33.3	21.9	20.2	1.7	21.6	20.1	1.5	20.3	18.4	1.9	26.9	25.0	1.9	21.0	22.3	−1.3	33.1	33.9	0.8	28.6	33.7	−5.1
B	39.9	26.1	26.6	−0.5	26.2	26.7	−0.5	25.0	25.0	0	31.5	31.6	−0.1	25.9	28.9	−3.0	37.1	40.5	−3.4	32.9	40.3	−7.4
C	41.5	28.6	28.2	0.4	28.1	28.3	−0.2	26.9	26.6	0.3	33.5	33.2	0.3	27.6	30.5	−2.9	39.6	42.1	−2.5	35.3	41.9	−6.6
D	43.0	30.6	30.5	0.1	29.7	29.8	−0.1	28.4	28.1	0.3	35.0	34.7	0.3	29.1	32.0	−2.9	41.8	43.6	−1.8	36.7	43.4	−6.7
E	40.3	27.8	27.5	0.3	27.3	27.1	0.2	26.1	25.4	0.7	32.7	32.0	0.7	26.8	29.3	−2.5	38.6	40.9	−2.3	34.3	40.7	−6.4
F	43.8	29.7	31.0	−1.3	29.7	30.6	−0.9	28.5	28.9	−0.4	34.9	35.5	−0.6	28.2	32.8	−4.6	40.8	44.4	−3.6	35.7	44.2	−8.5
G	40.6	28.5	27.8	0.7	29.2	27.4	1.8	28.0	25.7	2.3	34.6	32.3	2.3	28.8	29.6	−0.8	40.8	41.2	−0.4	36.5	41.0	−4.5
Error: Average				0.2			0.3			0.7			0.7			−2.6			−2.1			−6.5
σ				0.9			0.9			0.9			1.0			1.2			1.1			1.2
Test System Difference Distributions				−0.2, 0.5			−1.2			6.6			−5.8, 0.3						−4.5, 0.4			
															6.1, 0.3							
																	7.5, 0.2					
Reference System Difference Distributions				−0.2, 0.3			−1.7			6.6			−2.7						−0.2			
															8.9							
																	11.4					

on an orthotelephonic basis, is sharply bandlimited to 250–4000 Hz, and has 0 dB of loss within this band. For example, test network 125 Hz HPF (sensation level = 100 dB) of Table V has an $L_s = 107.1 - 0.2 = 106.9$ dBt. The loudness loss is then $89.4 - 106.9 = -17.5$ dB.

Entries in columns 4 through 9 were computed using the graphical form of Fig. 22. Column 4 entries are essentially a repeat of the column 3 entries, the difference being, as noted earlier, that the former were computed using a linear loudness law while the latter were computed using the loudness law of Figs. 6 and 8. Column 5 entries were computed without applying spread-of-loudness corrections.

The reference spectrum for computing the values given in columns 3, 4, and 5 was bandlimited to 250–3000 Hz. For example, entries of column 5 were obtained by plotting each of the responses on the form of Fig. 22, then measuring the area (square inches) enclosed by this response curve, the base line (designated 0 inches), and the left-hand and right-hand boundaries (designated 100 Hz and 5000 Hz respectively). This area was then divided by the base (inches) corresponding to the 250–4000-Hz bandwidth, and the equivalent height (inches) converted to loudness rating (dB) using the left-hand and right-hand ordinate scales. Entries of column 6 were obtained in the same way as those of column 5 except that the area was divided by the base (inches) corresponding to a 100–5000-Hz bandwidth. Thus, the change between columns 5 and 6 was simply one of reference spectrum bandwidth.

Entries of column 7 were obtained in the same manner as those of column 6 except that the area measured was restricted to the 300–3300-Hz bandwidth characteristic of EARS, and the reference spectrum was limited to the 300–3300-Hz band. Entries of columns 8 and 9 were obtained in the same manner as were entries of columns 6 and 7 respectively except that artificial voice/6-cm^3 coupler responses were used for the former, orthotelephonic responses for the latter.

The entries of columns 9a and 9b were computed in a manner which reflects the essential features of the EARS and, therefore, these entries closely approximate (within the bounds of computational and measurement error) what would be measured with the EARS on those test and reference systems utilizing linear transducers. However, column 9a and 9b entries of Tables XVII and XVIII and column 9a entries of Table XIX do not necessarily represent what would be measured with the EARS. The reason for this is that the systems for these cases utilized carbon transmitters which are nonlinear devices. Transmitter responses used in the computations, based on measurements made with real speech, would probably differ from the responses pertaining during an EARS measurement because of the highly different nature of speech and the EARS acoustic test signal. This matter is now under study.

The error distributions corresponding to each of the seven computation methods exemplified by the column 3 through 9 entries are summarized in Table XX. These distributions reflect the accuracy of the computed ratings in predicting reference channel setting for equal speech loudness. The entries of column 1 are repeated from Table IV. These show that the computational method provides an accurate means of predicting subjective loudness balances for all of the tests except the Steinberg linear system tests at a sensation level = 39 dB. As noted earlier, received speech levels in a well-engineered communications system will seldom be this low, and if they do occur, are likely to represent trouble conditions.

Comparing the entries of column 2 to those of column 1, we note that there is little change in the error distributions, indicating that the graphical method is a close approximation of the computational method. Exceptions to this are the results for the Steinberg tests, particularly at the lower sensation level where the average error and error standard deviation are somewhat greater than when using the computation form. The reason for this is the change from the loudness law of Figs. 6 and 8 to a linear loudness law.

Column 3 entries indicate a further reduction in accuracy due to neglecting spread-of-loudness. The greatest change occurs for the Steinberg tests which involved numerous filter conditions and the Van Wynen telephone set tests which involved transducers with pronounced resonances. Both of these represent instances where spread-of-loudness effects would be important. Somewhat less change occurs for the Van Wynen linear system tests which also involved filter conditions.

The remaining columns show errors which might be expected using the EARS as presently defined (column 7), and the EARS modified to incorporate a wider measuring band (column 6). For the most part, extending the band improves the accuracy appreciably with the exception of the Steinberg tests and the Van Wynen telephone set tests. As regards the former, examination of Table XV indicates that extend-

TABLE XX—ERROR DISTRIBUTIONS

Test Series Designation	(1)		(2)		(3)		(4)		(5)		(6)		(7)	
			GRAPHICAL COMPUTATION (Fig. 22)											
			Orthotelephonic Responses								Artificial Voice/6-cm³ Coupler Responses			
			Same as Computation Form Except											
	Computation Form (Fig. 14)		Linear Loudness Law		Linear Loudness Law, No Spread-of-Loudness		Analysis Band = Reference Band = 100–5000 Hz		Analysis Band = Reference Band = 300–3300 Hz		Analysis Band = Reference Band = 100–5000 Hz		Analysis Band = Reference Band = 300–3300 Hz	
	Avg.	σ	Avg.	σ	Avg.	σ	Avg.	σ	Avg.	σ	Avg.	σ	Avg.	σ
Steinberg Linear System Tests–Sensation Level = 100 dB	1.4	1.1	2.4	1.0	3.6	1.9	3.7	1.9	2.5	3.6	3.8	1.7	2.4	4.3
Sensation Level = 39 dB	4.0	2.2	5.7	3.9	7.0	4.6	7.8	4.6	6.0	5.3	7.1	3.9	5.7	4.6
Van Wynen Linear System Tests	−0.8	2.1	−0.8	2.0	−0.6	2.6	−0.6	2.4	−3.5	3.1	0.1	2.9	−3.3	3.6
Van Wynen Telephone Set Tests	0.3	1.1	0.1	0.8	2.7	0.5	2.8	0.7	1.6	0.7	5.2	1.2	2.5	1.0
Van Wynen Tie Line Tests	−0.4	0.9	−0.1	1.1	0.2	1.0	0	1.0	0.3	1.4	0.1	0.9	0.8	1.2
Fraser Tests	0.2	0.9	0.3	0.9	0.7	0.9	0.7	1.0	−2.6	1.2	−2.1	1.1	−6.5	1.2

ing the band substantially improves the accuracy for the low-pass filter conditions but decreases the accuracy for the high-pass filter conditions.

The entries of columns 4 and 5, in a sense counterparts of columns 6 and 7 respectively, indicate what might happen if an EARS were built to utilize an artificial voice and an artificial ear which were accurate simulations of the human voice and ear within the context of the orthotelephonic definitions (see Fig. 3), and the system were calibrated in conformance with these definitions. Comparing columns 4 and 6, and 5 and 7, we see that the accuracy improves for the Van Wynen telephone set tests and the Fraser tests, remaining about the same for the other tests.

Let us now direct our attention to the difference distributions of Tables XV through XIX. These are given at the bottom of the tables together with arrows showing the computational methods compared. To obtain the difference distributions, differences were obtained between numerical entries, condition by condition. The first number associated with an arrow represents the average difference, a positive number signifying that ratings entered in columns at the right-hand tip of the arrow are numerically larger than those at the left-hand tip of the arrow. The second number, if given, represents the standard deviation of the difference distribution. In many cases, a standard deviation is not given because it was found to be of the order of 0.1 dB, insignificant for present purposes.

The difference distributions referred to above are summarized in Table XXI. Note that there is a set of entries for each different transmitter-receiver combination excepting the Steinberg tests for which entries are given for each of the two sensation levels.

Column 1 of Table XXI indicates that the graphical method results and computation method results differ very little except, as was noted in discussion of Table XX, in the case of the Steinberg tests with the sensation level = 39 dB. Differences in graphical ratings with and without spread-of-loudness included are also seen to be relatively small from entries of column 2. Entries of column 3 show that the change in ratings due to changing the reference spectrum bandwidth from 250–4000 Hz to 100–5000 Hz is essentially a constant, although some slight variation from test to test is apparent.

The remaining columns, 4 through 7, show that (*i*) measuring system bandwidth differences and (*ii*) orthotelephonic versus artificial voice/6-cm^3 coupler differences are highly dependent on transmitter-receiver

TABLE XXI—DIFFERENCE DISTRIBUTIONS

Test Series Designation	(1) Graphical Loudness Rating Minus Computation Form Loudness Rating		(2) Linear Loudness Law Without Spread-of-Loudness Minus with Spread-of-Loudness		(3) Reference Spectrum 100–5000 Hz Minus Reference Spectrum 250–4000 Hz		(4) 300–3300 Hz Minus 100–5000 Hz: Ortho-telephonic Responses		(5) 300–3300 Hz Minus 100–5000 Hz: Artificial Voice/6-cm³ Coupler Responses		(6) Artificial Voice 6-cm³ Coupler Responses Minus Orthotelephonic Responses: 100–5000 Hz		(7) Artificial Voice 6-cm³ Coupler Responses Minus Orthotelephonic Responses: 300–3300 Hz	
	Avg.	σ	Avg.	σ	Avg.	σ	Avg.	σ	Avg.	σ	Avg.	σ	Avg.	σ
Steinberg Linear System Tests														
Sensation Level = 100 dB	0.2	0.5	−0.5	1.0	6.6	—	−3.3	1.8	−1.0	3.1	8.7	1.0	11.0	2.1
Sensation Level = 39 dB	−2.5	2.5	−0.3	1.0	6.6	—	−3.4	2.2	−1.1	3.3	8.7	1.0	11.0	2.1
Van Wynen Linear System Tests	0	0.5	−1.1	0.7	6.6	—	−4.4	3.0	−3.0	3.9	8.1	0.9	9.4	0.8
Van Wynen Telephone Set Tests														
Test Telephone Set A	−1.0	0.5	−0.2	—	6.1	—	−8.0	—	−7.5	—	7.7	—	8.2	—
Test Telephone Set B	−0.6	0.5	1.0	—	7.0	—	−7.0	0.5	−6.6	0.5	8.8	—	9.1	—
Reference Telephone Set	−0.5	0.3	−2.0	—	6.6	—	−6.1	0.7	−4.3	1.5	6.1	—	7.9	0.6
Van Wynen Tie Line Tests	−0.8	0.6	−1.4	0.5	6.5	—	−7.7	0.6	−7.1	0.7	6.6	0.5	7.4	0.5
Fraser Tests														
Test System	−0.2	0.5	−1.2	—	6.6	—	−5.8	0.3	−4.5	0.4	6.1	0.3	7.5	0.2
Reference System	−0.2	0.3	−1.7	—	6.6	—	−2.7	—	−0.2	—	8.9	—	11.4	—

combinations. These differences appear to fall into two categories: (*i*) the Steinberg and Van Wynen linear system tests and the reference system of the Fraser tests for all of which linear, but different type, transducers were used; (*ii*) the Van Wynen telephone set and tie line tests and the test system of the Fraser tests for all of which nonlinear transducers were used. The transmitter and receiver types were identical (although the specific transducers were different) for the reference telephone set of the Van Wynen telephone set tests, the Van Wynen tie line tests, and the test system from the Fraser tests.

That the difference distribution standard deviations are relatively small and there seems to be a strong dependence of average difference on transmitter-receiver combination, suggests the possibility of applying correction factors to ratings computed by one method, e.g., the EARS method, to obtain ratings for some other method, e.g., the computational method. The average differences summarized in Table XXI represent these correction factors.

Returning to Table XX, we note rather large errors in some cases with the EARS method (see column 7) while the errors for the computation method are rather small (see column 1). Obviously, application of correction factors to the Steinberg, Van Wynen linear system, and Van Wynen tie line tests will not improve the EARS method error distributions since in each of the tests, the particular transmitter-receiver combination was common to the test and reference channels.

Different transducers were used in the test and reference channels of the Van Wynen telephone set tests and the Fraser tests. Let us examine the effect of application of correction factors to column 9 entries of Tables XVII and XIX on error distributions. The correction factors can be obtained from Table XXI. When the appropriate entries of column 9 of Tables XVII and XIX are so converted, the error distribution for the Van Wynen telephone set tests is changed from one with average = 2.5 dB, σ = 1.0 dB to average = 0.3 dB, σ = 1.5 dB comparing favorably to the column 3 values (Table XVII). Similar values for the Fraser tests are average = −6.5 dB, σ = 1.2 dB before correction, average = 0 dB, σ = 1.3 dB after correction, comparing favorably to the column 3 values from Table XIX for which average = 0.2 dB, σ = 0.9 dB.

Review of preceding discussion and the information contained in Tables XV through XXI indicates that the EARS provides a simple and reasonably accurate method of measuring telephone connection loudness loss. For a telephone plant utilizing telephone sets of a single design, or of several similar designs, loudness ratings determined following the EARS procedure can provide an effective tool in telephone transmission engineering. In situations where the plant utilizes somewhat different telephone set designs, it appears that reasonably good design can result from using the EARS method to determine performance with the individual set designs. Comparisons between designs may then be made by application of correction factors. Determination of these correction factors can be a time consuming and difficult task as is evident from consideration of the orthotelephonic and artificial voice/6-cm^3 coupler response definitions of Figs. 3 and 16 respectively, but it is worth noting that such correction factors need be determined only once for each transmitter-receiver combination mounted in a particular handset configuration.

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