TASI Quality — Effect of Speech Detectors and Interpolation

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This article describes tests made to select design parameters for the speech detectors in the TASI system. Results of subjective tests carried out to determine maximum permissible loading of TASI circuits during busy hours are also described. Finally, conclusions drawn from observations on a working TASI system are given. These observations indicate that TASI is a more satisfactory method of increasing transatlantic cable capacity than alternate methods, such as the use of 2-kc channel banks.

I. Introduction

TASI (*T*ime *A*ssignment *S*peech *I*nterpolation) is a new component in the telephone system that can approximately double the message capacity of existing long submarine cables. With TASI many calls share the same facilities, each requiring an available channel only when speech is transmitted. In order to recognize that speech is being transmitted by the subscriber, a highly sensitive speech detector is required. To assign the speech to an idle channel and to connect the proper talker and listener at each end requires a rapid switching system. A description of the switching system and other related matters can be found in other sources.[1,2,3] This paper deals with: (a) the work carried out to select the parameters of a speech detector satisfactory for TASI operation and (b) the results of subjective tests made to determine the approximate effect of the type of speech clipping that can occur in a fully loaded TASI during busy traffic periods.

Some speech is lost whenever the number of individuals talking or starting to talk in one direction on TASI circuits exceeds the number of available channels. The amount of lost speech must be kept small so that the transmission quality is not affected appreciably. On the average, less than 0.5 per cent of the total speech is lost due to interpolation, as long as the number of calls in progress is held to no more than twice the number of channels. The effect of this loss on transmission quality is practically negligible.

In addition to the speech lost through interpolation, some speech is lost during the time required to connect a talker and his listener at the other end to the assigned channel. In TASI the switching time (17 milliseconds) has been kept as short as possible, consistent with reliable signaling. During the busy hours a subscriber will be switched about every second talkspurt; however, during occasional periods of peak load, the subscriber may be switched almost every talkspurt. Consequently, 17 milliseconds will be clipped from a large number of talkspurts during the busy hours. In order to minimize the amount of speech lost due to switching time, TASI has been designed so that a subscriber loses his channel only when it is not needed by that subscriber and when it is required by another talker.

A third possible source of lost speech results from the operate time of the speech detector. Whenever a talker has to be reconnected to a channel, the speech detector must recognize that speech is present and initiate the proper action. The interval between the time that the speech starts and the speech detector reacts adds to the amount of lost speech. The operate time of the speech detector can be kept small compared with the clipping caused by interpolation and connect signaling, but it should not be made so fast that the detector operates too often on noise.

In order to determine suitable speech-detector characteristics, subjective laboratory tests and field measurements were made on working transatlantic circuits. Since the demand for transatlantic circuits exceeded the capacity of the existing cable facilities, the schedule for developing TASI was of necessity very short for such a complex system. The tight schedule limited the type and length of test to the minimum needed for reasonable assurance that a satisfactory speech detector could be built. Consequently, a straightforward voltage threshold detector was chosen instead of a more complicated type. The first part of this paper describes the test results that led to the selection of the speech-detector characteristics. The second part of this paper describes the results of subjective tests to determine the impairment in speech quality caused by various amounts of lost speech. This work was needed as a guide to the maximum number of circuits that can be assigned to TASI without affecting speech quality adversely.

II. SPEECH DETECTOR CHARACTERISTICS

The ideal speech detector for TASI should operate only when speech is present and should not operate when noise and extraneous signals are present. A practical detector must represent a compromise between ideal operation on speech signals and ideal rejection of noise signals. In addition, the activity, or percentage of the total time that a detector is operated, must be minimized. The parameters of the detector were chosen to insure that it

i. operates when very low levels of speech are present,

ii. operates a minimum amount of time on line noise, and

iii. minimizes the number of times a subscriber must be switched, consistent with allowing twice as many calls as there are channels available.

The speech-detector parameters are interrelated and one parameter could not be selected without considering the effect on all other parameters. In order to choose the best combination, a series of subjective tests were made in the laboratory to find the speech-detector characteristics that provided good results under simulated plant conditions. Later, field measurements were made during a large number of transatlantic telephone calls to determine the performance of several possible speech detectors under actual plant conditions for a wide variety of talkers. The field tests measured speech activities, talkspurt lengths, and number of talkspurts. The combined results of the laboratory subjective tests and field measurements led to the choice of speech-detector characteristics shown in Table I. Each of these parameters will be treated separately.

When the speech detector is made too sensitive, the detector operates on noise and thereby reduces the possible TASI advantage. Conversely, when the speech detector is not sensitive enough, part of the first syllable is lost before the detector is operated. A sensitivity of the TASI speech detector of −40 dbm at zero transmission level when combined with an adequate speech detector hangover (slow release time) results in satisfactory speech quality with volumes as low as −31 vu and also results in minimum false operations due to noise. As shown on Fig. 1, a −31 vu talker has a lower volume than almost all talkers on transatlantic calls.

Table I — TASI Speech Detector Characteristics

1000-cps sensitivity	−40dbm at zero transmission level point*
Frequency range	500–3000 cps
Operate time	5 milliseconds
Hangover (release time)	240 milliseconds "deferred"
Echo suppression	Maximum 13 db

* The zero transmission level point is a point to which all level points in a toll system can be referred. It is analogous to citing altitude by referring to height above sea level. The zero level point is at the transmitting toll switchboard of the system under consideration.

If the speech detector responded only to speech power within a very narrow frequency band, the problem of noise activity would be greatly reduced. However, for a speech detector with a fixed sensitivity to recognize initial consonants from many talkers, it is desirable to have a reasonably wide frequency band. Tests have shown that a bandwidth of approximately 500 to 3000 cps is suitable for telephone speech. If the bandwidth were extended below 500 cps or above 3000 cps, noise operations would be increased and the initial speech power seen by the detector would not be increased sufficiently to permit an offsetting decrease in sensitivity.

To operate the speech detector the power on the line must remain above −40 dbm for about 5 milliseconds. Laboratory experiments have shown that when the operate time is made as fast as possible, the detector sensitivity for equal quality speech can be decreased to about −37 dbm. However, the faster operate time results in increased operation of the speech detector by noise spikes which increases the activity even with

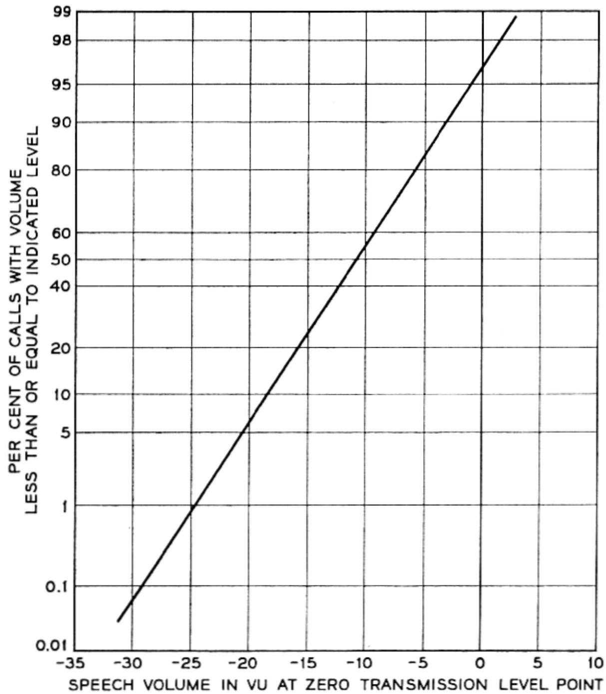

Fig. 1 — Subscriber speech volume distribution on transatlantic telephone circuits.

the lower sensitivity. Alternatively, an increase in operate time from 5 to 10 milliseconds requires an additional 3 db increase in sensitivity (−43 dbm) with no significant reduction in noise activity.

The sensitivity threshold of −40 dbm used for the detector is substantially above the threshold of hearing; hence, noticeable dropouts may occur when the speech power is below the sensitivity threshold. The ear is particularly sensitive to the loss of weak syllables as well as clips within words and closely connected phrases. Consequently, the speech detector should not release until the speech power has remained below the threshold for a period of time that is comparable to the time of one additional syllable. The hangover required for satisfactory transmission of low speech volumes varies with sensitivity and amounts to about 240 milliseconds for a −40 dbm detector. A much shorter hangover results in a higher TASI switching rate which increases the amount of speech lost by connect signaling clipping; a longer hangover results in increased speech activity and higher interpolation speech loss. Listening tests have shown that good transmission quality for weak talkers can be obtained with the combinations of sensitivity and hangover shown in Fig. 2.

Peaks of noise higher than −40 dbm may operate the speech detector and hence add to the circuit activity. Most of the noise peaks last less

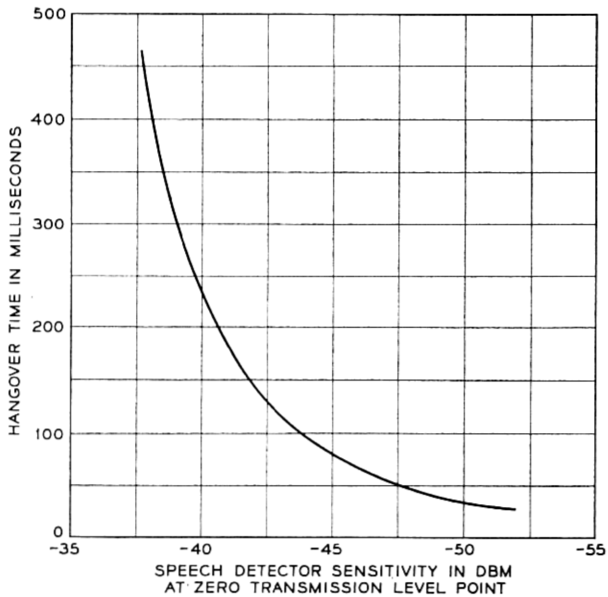

Fig. 2 — Variation of hangover with sensitivity for constant speech quality.

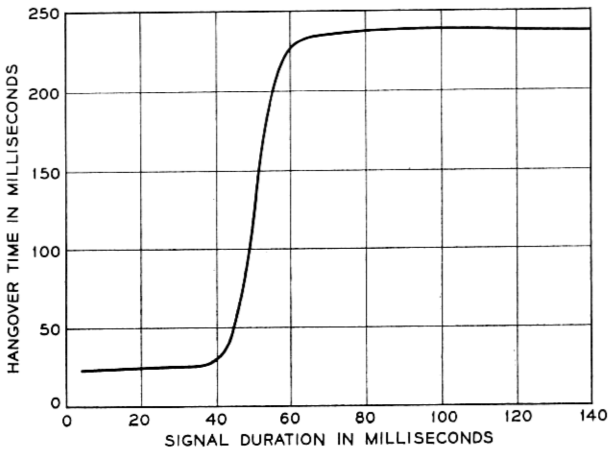

Fig. 3 — Speech-detector hangover vs signal duration.

than 5 milliseconds and are virtually eliminated by the 5-millisecond operate time of the speech detector. Once the speech detector is operated, however, the circuit cannot be released until the hangover time has elapsed. The effects of those noise peaks, which last from 5 to about 50 milliseconds, are minimized by the use of the deferred hangover characteristic shown in Fig. 3. The minimum hangover time is about 25 milliseconds. Noise peaks lasting substantially longer than 50 milliseconds are indistinguishable from speech syllables and operate the circuit to the extent of the full hangover of 240 milliseconds. The combination of 5-millisecond operate time, −40 dbm sensitivity, 240 milliseconds deferred hangover, and 500 to 3000 cps frequency range is about optimum for the expected telephone speech and noise levels.

In the preceding sections only the operation of the speech detector by normal speech and noise incoming to the TASI system has been considered. In the actual telephone plant, another group of unwanted signals, called echoes, can result in false operations of the detector. Transatlantic cable circuits have a one-way delay of about 40 milliseconds and this amount is sufficient to require echo suppressors in the four-wire part of the plant to prevent subscribers from hearing the echo returned from the two-wire part. The echo suppressors have an operate time of about 12 milliseconds to minimize false operation by noise. At times, the slowly operating suppressor will permit small bursts of echo to get past the suppressor at the beginning of talkspurts from the distant terminal. The distant listener is unaware of these short echoes which are therefore

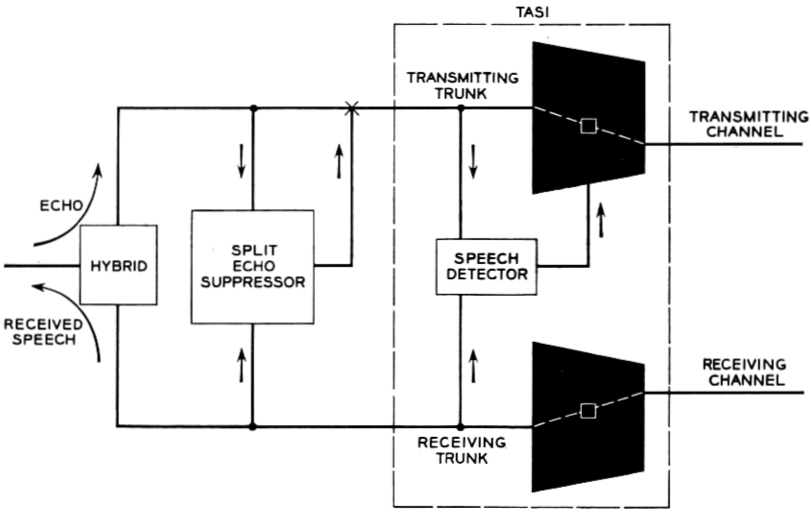

Fig. 4 — Simplified diagram of TASI terminal

of no consequence in non-TASI circuits. However, on TASI-equipped circuits, these short echoes can operate the faster acting (5 milliseconds) speech detector and thereby use valuable channel time. The echo path is shown in Fig. 4 for an individual trunk in which the TASI terminal is represented in simplified form.

To prevent these short echoes from operating the detector, an echo protecting circuit was added to the speech detector and connected to the receiving trunk output. When the distant talker is active, his speech reduces the sensitivity of the near-end speech detector at a uniform rate up to a maximum sensitivity decrease of 13 db as shown in Fig. 5. This value was chosen after consideration of the existing return losses in the plant and of the characteristics of the echo suppressors. The objective was the elimination of nearly all echo operation under all operating conditions. Measurements on many calls have confirmed that echo operation of the speech detector is negligible.

III. LABORATORY TESTS AND FIELD MEASUREMENTS ON SPEECH DETECTORS

From the foregoing description of the interaction of speech-detector parameters, it is apparent that there are several combinations which will result in a speech detector capable of recognizing the presence of speech and acting on this signal with hardly any effect on the speech quality.

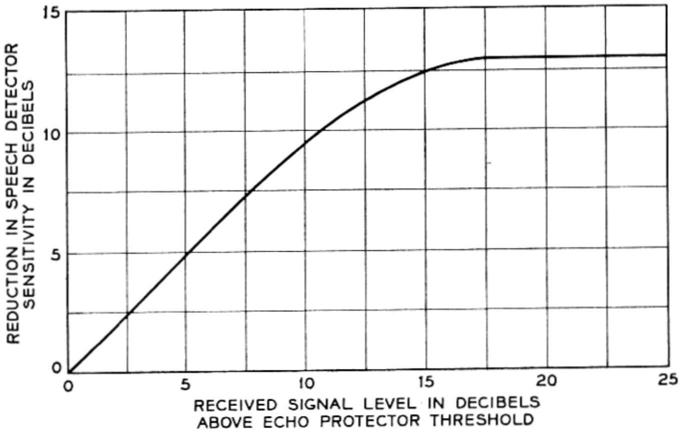

Fig. 5 — Effect of echo protection circuit.

Listening tests in the laboratory were made to determine the speech-detector sensitivity required for each type of detector over a range of speech levels. These tests were made using tape recordings of a number of voices, both male and female.

Speech from the tape recorder operated the speech detector, which in turn operated a gate to connect the observers to the tape recorder. The arrangement was such that observers could hear the recorded speech only when the detector was active. The observers were selected from Bell Telephone Laboratories personnel, both technical and clerical. They were asked to determine the minimum sensitivity required for acceptable speech quality. The rapid deterioration of speech quality when the speech-detector sensitivity was reduced below a certain minimum made this adjustment critical for each observer and resulted in a reproducible relation between speech-detector sensitivity and speech level. While each observer had a well defined tolerance level, the variation of the results among individuals was large. The sensitivity selected as satisfactory for a given speech detector was the value that satisfied 50 per cent of all observers for the minimum speech level of −31 vu at the zero transmission level point.

After determining several combinations of speech-detector characteristics that resulted in equal speech quality, various speech detectors were connected across transatlantic trunks to determine for each call the number of operations and the activity. The average activity of transatlantic subscribers, determined from measurements on many calls, determines to a large extent the possible TASI advantage (ratio of trunks

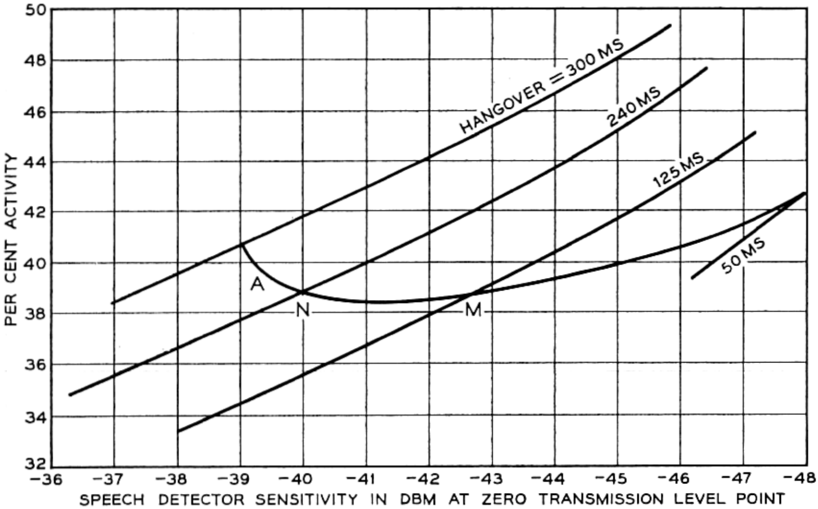

Fig. 6 — Average speech-detector activity during busy hour on transatlantic telephone circuits.

to channels). The number of operations affects how often a subscriber is disconnected from and reconnected to the channels. The results of the field measurements are shown by the four sloping lines on Fig. 6. On the same figure, curve A indicates the locus of points for constant and acceptable quality from Fig. 2. This locus permits the selection of the detector that provides acceptable quality and minimum activity. It will be noted that the point of minimum activity on curve A is not very sharply defined. Two speech detectors, points N (240 milliseconds hangover) and M (125 milliseconds hangover), were selected for further study. The average activity and talkspurt length resulting from each of these detectors were obtained from the measurements on transatlantic calls and are summarized in Table II.

It will be noted in Table II that the activity from the 240-millisecond detector is higher than for the 125-millisecond detector. However, the talkspurt length for the customer using the 240-millisecond detector is almost twice as long as for the 125-millisecond detector. As a guide in the choice between these detectors, computations were made to determine the total amount of speech a subscriber would lose during the busy hour if one or the other detector were used in TASI. The computed speech loss (interpolation plus switching) associated with each detector is shown on Fig. 7. The total speech lost is slightly less for the 240-millisecond hangover detector. Although use of the 240-millisecond de-

Table II — Measured Activities and Talkspurt Lengths for Transatlantic Speech Originating in U.S.

	125-millisecond Detector	240-millisecond Detector
Activity		
Operator	23%	24%
Subscriber	47%	48%
*Average during busy hour	38%	39%
Duration of average detector operation		
Operator	0.4 second	0.7 second
Subscriber	0.8 second	1.3 second
*Average during busy hour	0.6 second	1.1 second

* The results for operators and subscribers are combined in the ratio of 0.375 to 0.625, respectively. This ratio was found to be approximately the division of operator and subscriber time on transatlantic trunks during the busy hour.

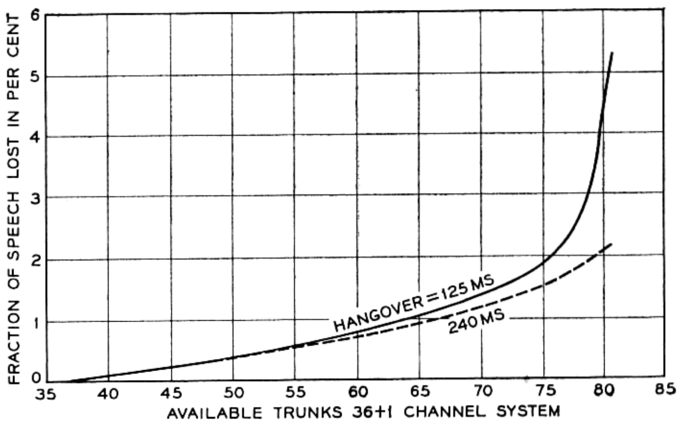

Fig. 7 — Subscriber speech losses caused by TASI signaling and interpolation.

tector results in a higher activity than the 125-millisecond detector and a slight increase in interpolation clipping, these disadvantages are more than offset by the fewer switching operations.

IV. SUBJECTIVE EVALUATION OF CLIPPING

The preceding tests showed that a satisfactory speech detector could be built whose impairment would be negligible. Additional tests were needed to determine how much clipping of all types could be tolerated without a significant impairment in speech quality. These quantitative tests were needed in order to specify the maximum number of circuits that could be used with TASI.

Under normal operation, TASI clipping consists of many segments which are short compared with an average syllable length. During the busiest hours of the day, about one-half of the talkspurts will experience some clipping but in most cases, it will be limited to the 17-millisecond switching clip. Only one talkspurt in 10,000 will be clipped longer than about 0.4 second. Since the listener rarely misses a syllable, the result is a slight abruptness in speech which has a negligible effect on intelligibility.

The initial series of subjective tests to evaluate clipping in TASI was conducted before an actual TASI terminal became available. Preselected clips of a fixed length were introduced at the beginning of every Kth talkspurt, where K was 1, 2, $\cdots$, etc. By eliminating the randomness of TASI, the testing time was shortened and the data could be analyzed to discover whether the most significant factor affecting the impairment of TASI was the total speech lost or the pattern of clip occurrence. In this series of tests, the percentage of total speech time lost by clipping was varied from 0 to 6 per cent, which encompassed the expected range of TASI clipping. A group of observers placed normal business calls over circuits in which the amount of clipping would be controlled and measured. The observers were asked to base their replies solely upon *quality* considerations and to rate the calls as either "good," "fair," or "poor." The testing methods are described more fully in the Appendix.

The results of the clipping tests are shown on Fig. 8. Each point represents the percentage of calls rated "good" out of groups of more than 75 test calls for each condition. The left-hand scale indicates the amount of impairment found to elicit the corresponding percentages of "good" responses in a previous study of other degradations.[4] The curve was an approximate fit with a second-order polynomial using the least squares method.

Impairment in db has the following meaning: if a telephone subscriber is given the choice between the transmission system under test and a reference transmission system, how much loss could be inserted in the reference path for the subscriber to rate the circuits as equivalent? The loss inserted is defined to be the impairment, and is usually expressed in db. This impairment has been determined in the past for such things as noisy systems, restricted bandwidth, etc., but has not been determined directly in this study. The rendering of the present data into db of impairment rests on the assumption that "percentage good" means the same thing in this and the Coolidge-Reier study. This assumption is

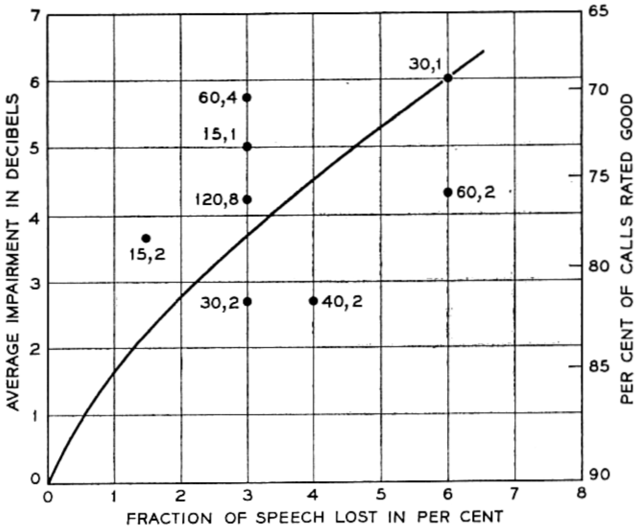

Fig. 8 — Subjective impairment vs fraction of speech lost.
Note: designation 30,2 means a 30 ms clip every second talkspurt.

unlikely to hold with great precision, so the impairment figures given here (and the implied comparability with other impairments) must not be considered precise quantities. This reservation also applies to similar conversions referred to later in this paper including those involving intelligibility figures. The use of such impairments provides a common denominator for comparing the effect of these different forms of speech degradation.

After the initial series of tests had been completed, another series of tests expanded the measurements to still greater amounts of clipping, in order to evaluate the loss in intelligibility that might occur with unusually high TASI loadings. The second series of tests used articulation methods and measured the degree of *successful communication* rather than simply quality as before. By this time a working TASI terminal was available so the occurrence and duration of the clipping were essentially the same as in normal TASI operation. The tests used "phonetically balanced" words and both the articulation in percentage of correct words and the percentage of lost speech were measured. The results of three articulation tests for widely different amounts of lost speech are shown by the dots on Fig. 9. The upper dotted curve is based on the three articulation tests, while the lower dotted curve has been taken from Fig. 8. The combined result is indicated by the solid line. The translation

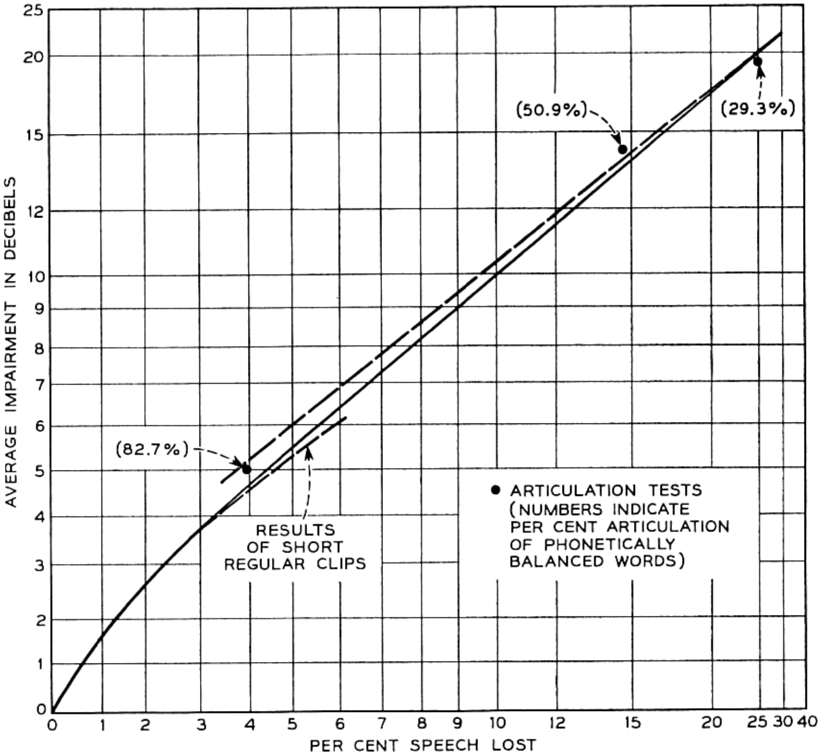

Fig. 9 — Subjective impairment vs fraction of speech lost.

from the percentage of correct words to db impairment made use of previously published results.[5] The testing methods are described more fully in the Appendix.

The foregoing tests indicated that the impairment is determined for the most part by the percentage of speech lost, at least for the case of normal TASI clipping which has certain definite characteristics: the clips are of short duration, they occur only at the beginning of talk-spurts and they are irregularly distributed throughout a call. Hence, these results do not necessarily apply to the case where long clips of many syllables occur very infrequently.

The results on Fig. 9 can be applied to a statistical analysis of TASI clipping to obtain an estimated impairment for an actual TASI system. Fig. 10 shows the probable busy-hour impairment for a typical trans-atlantic TASI system, which uses 36 interpolation channels plus one channel for disconnect and error checking signals. The estimates on Fig. 10 are based on the amount of lost speech which has been computed from appropriate distributions of subscriber activities and talkspurt lengths as a function of the number of trunks available for service.

The median curve indicates that in 50 per cent of the calls, the impairment caused by TASI will be less than 2 db as long as the number of active circuits is no more than 74 trunks on 36 + 1 channels. However, the impairment amounts to about 4 db for one call in a hundred and to about 6 db for one call in 10,000. For comparison purposes, the approximate impairments of 2-kc and 3-kc underseas channel banks are also indicated on Fig. 10. It will be noted that on nearly every call, a 2:1 TASI advantage causes less busy-hour impairment than would have been obtained through the use of 2-kc channel banks. In addition, the TASI impairment decreases rapidly as the number of talkers decreases during non-peak periods, while the impairment caused by narrow channels is independent of the traffic load.

V. DETERMINATION OF THE GRADE OF SERVICE

The foregoing results are presented in terms of db impairment to permit an easy comparison between TASI and alternate methods of

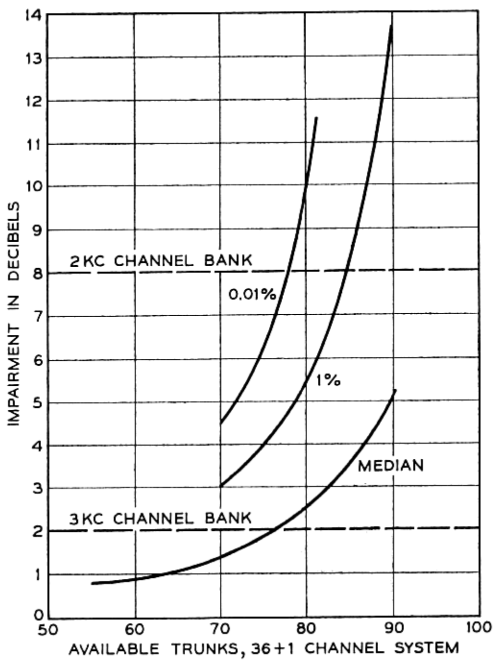

Fig. 10 — Busy-hour subscriber impairment due to TASI.

obtaining additional channels. It is also of interest from an operating standpoint to estimate the over-all grade of service on transatlantic calls when the additional practical factors such as variations in net loss, bandwidth and talker volumes are included. For more accurate results, the effect of each component should be investigated under all possible conditions of the other factors such as noise, volume bandwidth, etc., in order to look for possible interaction effects. However, in order to find the small effects looked for in this study, many samples would be required for each combination, and the amount of work involved would be almost prohibitive. Past experience indicates that a rough estimate of the over-all impairment can be obtained by adding together the individual impairments in db and then expressing the end result in the "good," "fair," and "poor" ratings.

On this basis the estimated grade of service on transatlantic calls with TASI applied to 36 3-kc interpolation channels is shown on Fig. 11 for loading conditions that are equal to or greater than the normal operating condition of 74 trunks on 36 + 1 channels. For comparison the estimated grade of service is given for 3-kc underseas channel banks used without TASI as well as the desired objective for all long-distance circuits. While this objective is not fully met at present, it is expected that the TASI grade of service under normal operation will be improved and will approach the desired objective as the result of the current program to improve local plant transmission. Figures 10 and 11 also indicate that during emergency peak periods, TASI can provide a greater number of higher quality circuits than can be realized with 2-kc channel bank equipment.

VI. SUBSEQUENT OBSERVATIONS OF TASI IN SERVICE

After TASI systems were put in service on transatlantic telephone cables, service observations were made on several TASI circuits as well as on a reference non-TASI circuit. The results indicated that the percentage of calls rated "good" by a qualified service observer was practically the same for TASI and non-TASI trunks.

The grade of service for greater than normal TASI loading was measured when additional trunks were utilized for emergency service at the time of a break in one of the transatlantic cables. One transatlantic system was utilized with TASI to carry traffic normally carried by both transatlantic systems. Table III gives the grade of service for normal loading and for two conditions of greater than normal loading. With the exception of a somewhat greater "poor" rating for 90 trunks, the observations on operating circuits correspond to those predicted in Fig. 11.

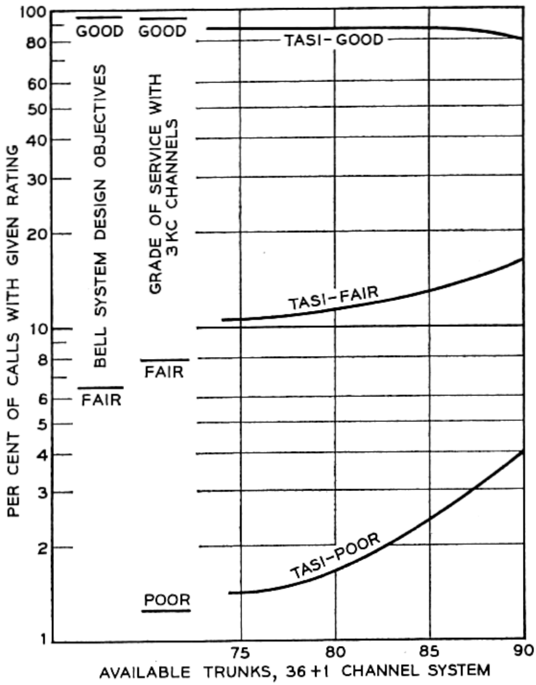

Fig. 11 — Grade of service — TASI plus 3-kc channels.

TABLE III — TASI GRADE OF SERVICE FOR NORMAL AND PEAK PERIOD LOADING AS DETERMINED BY TELEPHONE COMPANY SERVICE OBSERVERS

Trunks on 36 + 1 channels	Per cent Rating "good"	Per cent Rating "fair"	Per cent Rating "poor"
74	90	9	1
84	87	11	2
90	74	16	10

VII. CONCLUSION

The data from these service observations combined with the other results presented in this paper give assurance that most telephone users making calls through a TASI system with normal load will not be aware that their conversation is being interpolated. This holds even during the busy period when any degradation that does exist is at a maximum. During emergency conditions when service is partially disrupted, TASI provides additional circuits with only a moderate decrease in grade of service. This is felt to be preferable to long service delays.

VIII. ACKNOWLEDGMENT

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APPENDIX

Method of Conducting the Tests

A.1 *Initial Series of Conversational Tests*

The random nature of speech and clipping, and the dependence of the subjective ratings on the environment in which the clipping occurs indicated that the tests of clipping should be conducted under the most representative conditions. To implement this requirement, observers at Bell Telephone Laboratories engaged in normal telephone conversations in which the TASI clipping was introduced and the bandwidth and net loss were those of a typical transatlantic circuit. These calls were from the desks of the observers and consisted of normal business traffic that corresponded well with typical traffic on transatlantic calls. Following each call they rated the quality of the circuit as "good," "fair," or "poor." A total of 45 hours of test calls made by 25 talkers was analyzed.

The results of the tests were analyzed and yielded the percentage of all test calls rated "good" in over-all quality of transmission. The rating of per cent "good" for the various clipping conditions was converted to db impairment by the previously mentioned relation between subjective ratings and received volume. This step assumes that the impairment is roughly equivalent to the effect of a decrease in received volume on a transatlantic circuit.

The individual data points are plotted in Fig 8 and the resulting curve is repeated as the lower portion of the curve of Fig. 9. The percentage of speech lost is determined by the length of the clip, the frequency of occurrence (every Kth talkspurt) and the average talkspurt length. For these tests, the 125-millisecond hangover detector was used which results in an average talkspurt length of about 0.5 second. Thus, for 30 milliseconds, $K = 2$, a 30 millisecond clip occurs on every second talkspurt of length 0.5 second, and this condition results in a percentage speech loss of 3 per cent. As indicated in Fig. 8, speech losses of 3 per cent and 6 per cent resulted from more than one test condition. These different conditions were used to determine whether the manner in which a given percentage speech loss occurs results in a significant variation in the impairment. Such a significant variation is not apparent from the results. Rather, the spread that occurred can be attributed more to variability in the testing than to the different ways of producing clipping. While each point represents more than 75 test calls rated by observers, the estimated confidence interval is only ± 2 db because only 3 separate ratings were possible. The solid curve has been used for TASI engineering but any use for other purposes should take into account the testing methods and the variations that are inherent in such subjective tests.

A.2 *Articulation Tests*

Articulation tests were conducted to obtain information about the effect of clipping when the amount of lost speech was great enough to affect comprehension. Single word articulation tests were chosen to provide the severest type of test of TASI clipping.

These tests were conducted with a list of one hundred "phonetically balanced"[6] words, which have been so chosen that all speech sounds are represented according to their frequency of occurrence in normal speech. The list of words was recorded using four different voices which were combined so that twenty-five words from the list were recorded by each voice. Two male and two female voices were used.

The recorded lists of words were played through a TASI terminal which was artificially loaded to insure the desired amount of TASI clipping. The clipped speech was played back in the laboratory through standard telephone handsets. Six technical and nontechnical observers were used for each test. The speech levels at the receivers and the receiver noise were adjusted to simulate an average transatlantic connection. The observers wrote down the words as they heard them and their responses were compared with the original lists to determine the phonetically balanced word articulation in per cent.

The approximate impairment in db corresponding to the per cent PB word articulation can be obtained from previous information that relates phonetically balanced word articulation to syllable articulation and syllable articulation to volume above threshold. By this means the articulation results for the conditions tested are related to the corresponding volumes, and the variation in these volumes from the reference condition yields the impairment in db.

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