# DESIGN OF DESIRABLE LOUDNESS RATINGS FOR ISDN TELEPHONE

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ABSTRACT This paper describes the method for designing loudness ratings as transmission quality for ISDN telephone connected to fully digital network. To design the desirable loudness ratings for ISDN telephone, the model system of digital speech communication for subjective test is developed and opinion tests for establishing the optimal CODEC input level, the range of overall loudness rating, and sidetone masking rating are performed. As the results, the desirable ranges of loudness ratings are proposed as 6 to 8dB for sending, 0 to 2dB for receiving, and 10 to 14dB for sidetone masking rating.

## 1. INTRODUCTION

ISDN telephone service is expected to be expanded as well as improved by digital transmission and switching. Digital technologies enable network planners to avoid various quality deterioration phenomena in conventional analog telephone networks. However, even in ISDN communication, there is no difference in the fact that speech is of major importance in person-to-person communication. Therefore, telephone user's transmission quality requirements about ISDN telephone services need to be reconsidered.

In the fully terminal-to-terminal digital connection between ISDN telephone customers, because the transmission loss of network is 0dB, most of the transmission impairments in the connection are determined by the terminals, except for the delay and non-stationary noise due to the network. Therefore it is important to design ISDN telephone sets which provide good speech quality in order to achieve a high performance in digital connections.

Thus the transmission quality standard must be an important tool to obtain the high quality, and to develop the techniques for design of telephone networks and telephones. ITU-T has provided useful recommendations concerning transmission impairments and their permissible levels. These recommendations allow customers to communicate with good grade. However various customers' transmission quality preferences, depending on their nationality, mother language, feeling, culture, and social environments, are evaluated using a common measure representing customers' satisfaction and are complied with a performance objective. In other words, the ITU-T recommendations give guide lines for constructing a network.

There is now a trend to judge speech transmission performance in a telephone network from the mean opinion score given by subjects talking part in conversations over simulating connections of test circuit with known physical characteristics. In this study, in order to evaluate the transmission quality of ISDN quantitatively, we developed the digital opinion test system which satisfies ISDN user network interface standardization of ITU-T I series. The correlation between the impairment factors of the digital transmission quality and telephone users' opinions was derived.

This paper describes the design concept of digital opinion test system, the result of transmission quality design of ISDN telephone set based on the user's opinion tests.

## 2. DEVELOPMENT OF THE MODEL SYSTEM

For the design of the LRs of ISDN telephone, the model system of digital speech communication which can be used in the opinion test is developed. The module configuration for the model system (called DTS, Digital opinion Test System) is shown in Figure 1. This DTS is designed according to the ISDN UNI(User-Network Interface) standardization in ITU-T I Rec. The main functions of DTS are to control the degradation factors of speech quality in digital telephone module, to connect and terminate the digital speech path, and to provide the matrix switch for module composition. The software program of DTS is implemented according to the ISDN UNI protocol recommended in ITU-T Rec. I.430, I.440, and I.450. The block diagram of digital speech communication path made by the digital telephone module and network interface module is shown in Figure 2.

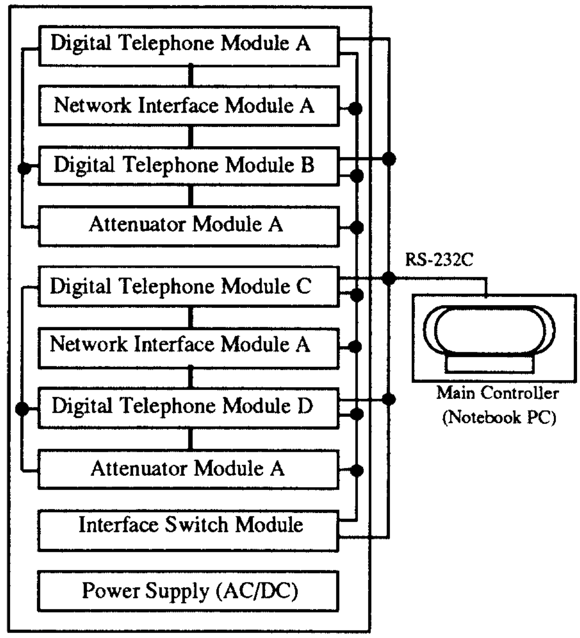


Fig.1 Module configuration of DTS.

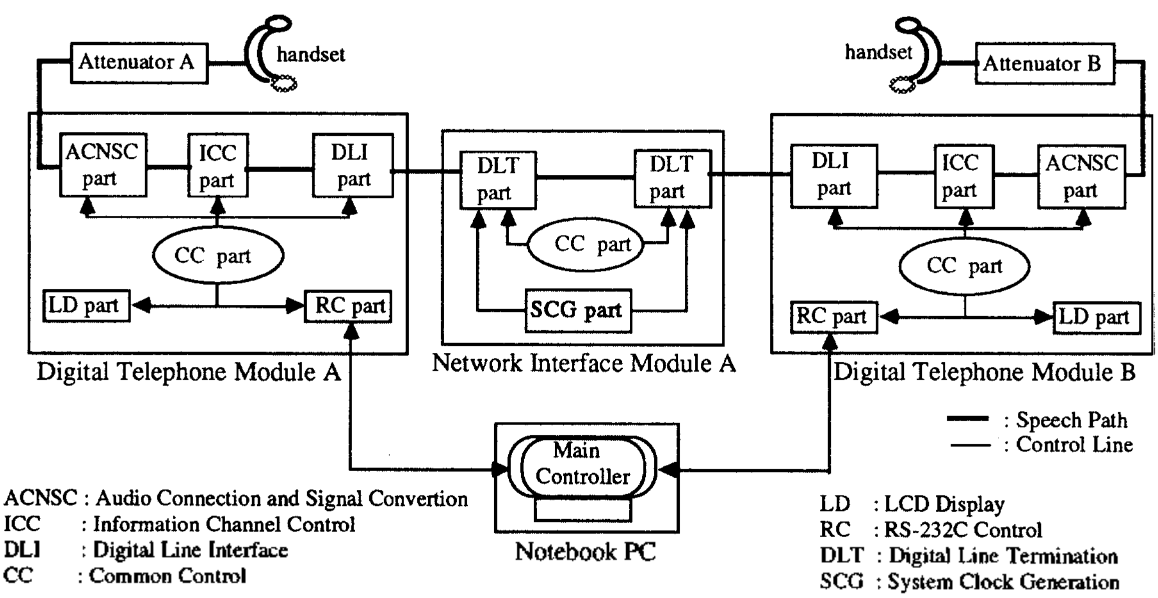


Fig.2 Block diagram of digital speech communication path.

## 3. OPINION TESTS FOR ESTABLISHING LOUDNESS RATINGS

### 3.1 Overview

The speech quality which is assessed by loudness performance among the call connection state uses the LR expressed in decibels to quantify numerically, and then the overall transmission quality expressed in OLR can be allocated to SLR, JLR(Junction LR), and RLR. The loudness rating is related to the loudness loss between a talker's mouth and the listener's ear. The loudness of the received speech signal depends on acoustic pressure provided by the talker and the loudness loss of the acoustic-to-acoustic path from the input to a telephone microphone at one end of the connection to the output of a telephone receiver at the other end of the connection. The effectiveness of speech quality and user satisfaction depend, to a large extent, on the loudness loss which is provided. As the loudness loss is increased from a preferred range, the listening effort is increased and user satisfaction decreases in parallel with decreased intelligibility. On the other hand, if too little loudness loss is provided, user satisfaction is decreased because the received speech is loud. Therefore, the appropriate ranges of the loudness loss for sending and receiving part in ISDN are only necessary, because the JLR is 0dB.

In addition to SLR and RLR, sidetone must be considered as an important factor on speech transmission quality. Sidetone is the result of the acoustic signal originating from the talker's mouth being returned through the telephone set to the talker's ear and there are also various acoustic sidetone paths, i.e., through the air, the telephone handset, and the user's head. Sidetone is characterized by its amplitude and affects telephone transmission quality. Too little sidetone loss causes the returned speech levels to be too loud and this reduces user satisfaction. Another aspect of insufficient sidetone loss is that talkers tend to reduce their speech level and/or move the handset away from the mouth, thus reducing the received levels at the far end of the connection. Handset movement can also reduce the seal at the ear and thus make it easier for room noise to reach the ear through the resulting leakage path, while reducing as well the level of the received signal from the far end of the connection. Sidetone is specified in terms of STMR which takes into account the head conduction and direct acoustic path as a masking threshold.

### 3.2 Opinion test on OLR

For finding out the reasonable range of OLR, user opinion test searching the relationship between OLR and MOS was performed. The test results are shown in Figure 3 as a function of LR. As can be seen in Figure 3, it can be noted that good speech quality above MOS 3.5 is obtained when OLR is 0 to 12dB. Also of interest in Figure 3 is the optimal MOS value which indicated at about 8dB of OLR. In the cumulative percentage of users who response to the opinion scale of speech quality, MOS to which a 50 percent of the subjects respond as above "fair" is 2.7 and MOS to which a 90 percent of the subjects respond as above "fair" or a 50 percent as above "good" is 3.7. Thus, the OLR of 6 to 10dB occurring above MOS 3.7 is selected. This can be the desirable range to provide the comfortable speech transmission quality in ISDN and this

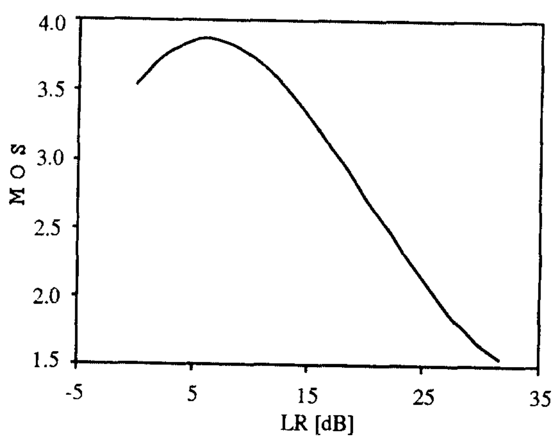

Fig.3 The relationship between MOS and OLR.

range can cover a tolerance of ±2dB from the optimal value.

### 3.3 Selection of input level to the CODEC.

The effect of input level to the CODEC on speech quality must be contemplated to determine the SLR of ISDN telephone. If the input level to the CODEC isn't controlled at the proper range, some problems in the speech quality can be issued. In case of low level, total noise is grown with the amplification of quantization noise due to the amplification of the receiving side. If too high input level to the CODEC is provided, overload distortion generated by limiting the signal amplitude occurs in the CODEC when the amplitude of input signal is over the maximum input level to the CODEC. To ensure acceptable range of input level to the CODEC, the relationship between user opinion and input level to the CODEC must be established. The listening test on which users respond to the speech quality due to the variation of input level is performed. The seven input levels to 0 to -40dB, relative to the overload level(7dBV) of sine wave to the CODEC under test were used and the three listening conditions around the preferred level were utilized not to have an effect on the test results. The relationship between MOS and input level to the CODEC is shown in Figure 4. As illustrated in Figure 4, the desirable input level falls between -12dB and -18dB to reduce quantizing noise and overload distortion occurring in CODEC. The maximum MOS value is indicated at about -15dB of input level. The 70dB SPL of listening level shows the best one in this test. It is noted that listening level has an effect on noise even if the same signal to noise ratio exists.

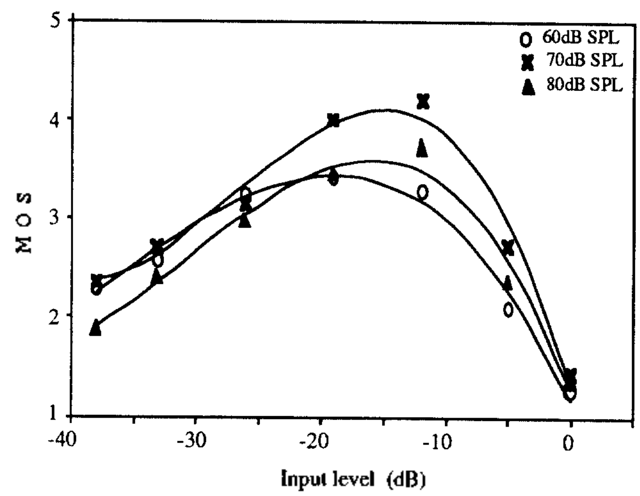

Fig. 4 The relationship between MOS and input level.

### 3.4 Opinion test on STMR

To ensure acceptable quality of STMR, the relationship between user opinion and STMR was established. To control the effect of SLR and RLR on STMR, SLR and RLR are fixed to the value obtained from the design of SLR and RLR. The 30 subjects participated in this test. The variations of STMR with 6 levels, 0 to 25dB, and room noise with the Hoth noise of 45dB(A) and 60dB(A) are used in this test. The relationship between MOS and STMR is shown in Figure 5. It can be seen in Figure 5 that the desirable STMR above MOS 3 exists between 5dB and 18dB and the STMR with maximum MOS is 12dB.

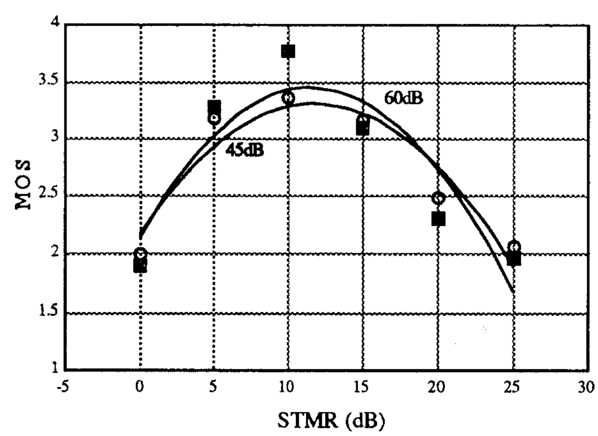

Fig.5 The relationship between MOS and STMR.

## 4. ESTABLISHMENT OF LR

As noted earlier with respect to the opinion test of input level to the CODEC, it is reported that the input level must be reserved below -12dB and the desirable range is between -12dB and -18dB. From this result, let's consider how to establish SLR. First of all, the speaking level of test speech data is controlled to be 85dB SPL at the mouth reference point, because the speaking levels of most telephone users fall in the range of 65 to 87dB SPL and about 98 percent in the below 85dB SPL when the long time average of speaking level at the mouth reference point is measured. Then, the input amplifier gain of the CODEC is adjusted so that speech level may be -5dBV(-12dB of overload level using sine wave) at the CODEC input. The SLR should be calculated from sending sensitivity and frequency characteristic measured in the stated conditions by means of ITU-T Recommendation P.79, and the SLR calculated as an input level of -12dB equals 6dB. This means that the SLR should be not less than 6dB because this value is used as the overload point to the CODEC, and then the value of SLR in the range 6 to 8dB considering a tolerance of +2dB is proposed. The OLR of 8dB as described earlier with respect to opinion test is known as the optimal speech quality of ISDN speech communication. The OLR for an end-to-end connection of ISDN is given by :

$$OLR = SLR + JLR + RLR$$

In this expression, the SLR with a tolerance of +2dB is 6 to 8dB and the JLR is 0dB. The remaining RLR with a tolerance is, therefore, calculated as 0 to 2dB.

From the opinion test of STMR noted earlier, MOS distribution on STMR is obtained as shown in Figure 6. In the cumulative percentage of users who respond to the sidetone amplitude level, if the STMR level which a 50 percent of the subjects respond to as "quieter than preferred" or "much quieter than preferred" is the permissible upper boundary and the STMR which a 50 percent of the subjects respond to as "louder than preferred" or "much louder than preferred" is the permissible lower boundary, the acceptable range lies between STMRs of 6dB and 17.5dB. Using this background and the optimal STMR, the value of STMR in the range 10 to 14dB with a tolerance of ±2dB is proposed.

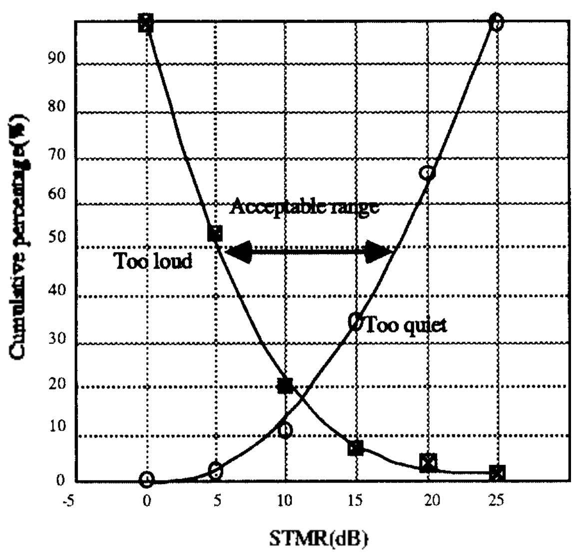

Fig. 6 The acceptable range of STMR.

## 5. CONCLUSION

Up to now, This paper deals with the methods for establishing loudness ratings as transmission quality of ISDN telephone connected to fully digital network. For establishing the SLR, RLR, and STMR of ISDN telephone, opinion tests for analyzing the relationship between MOS and input level of CODEC, and OLR, and sidetone amplitude were performed respectively. The following values are proposed from the test results :

1) Optimal OLR of 8dB in ISDN;
2) SLR in the range 6 to 8dB including a tolerance of 2dB;
3) RLR in the range 0 to 2dB including a tolerance of 2dB by the optimal OLR;
4) STMR in the range 10 to 14dB including a tolerance of ±2dB from the optimal STMR of 12dB,

These values proposed in this paper satisfy ITU-T Recommendation P.31 for LRs of the transmission characteristics for digital telephone, but other values may be used to apply this method to other countries.

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