Design of The Loudness Ratings And Talker Echo For ISDN Telephone

ISDN 전화기의 음량 정격 및 송화자 에코설계

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ABSTRACT

It is the purpose of this paper to describe the methods for establishing loudness ratings and talker echo out of transmission quality of ISDN telephone connected to fully digital network. In order to design the desirable loudness ratings and talker echo for ISDN telephone, the model system of digital speech communication for subjective tests is developed. Using this model system, opinion tests which decide the optimal CODEC input level, the range of overall loudness rating, sidetone masking rating and talker echo are performed. From the results of tests, we decided that the loudness ratings are 6 to 8dB for sending, 0 to 2dB for receiving, and 8 to 12dB for sidetone masking rating. And, the terminal coupling loss of TCLw of at least 40dB is necessary to provide echo-free telephone communications to telophone users when the overall loudness rating of ISDN telephone is normalized to 10dB.

요 약

본 논문은 디지탈 망에 접속된 ISDN 전화기의 전송품질인 음량정격과 송화자 에코를 설정하는 방법을 기술한다. ISDN 전화기의 바람직한 음량정격 및 송화자 에코를 설계하기 위하여 주관평가를 위한 디지탈 음성통신 모델 시스템을 개발하였고, 이 모델 시스템을 이용하여 최적의 코덱 입력레벨, 전체 음량정격의 범위, 그리고 송화자 에코 등을 결정하기 위하여 오피니언 테스트를 수행하였다. 실험결과 송화 음량정격은 6~8dB, 수화음량정격은 0~2dB, 측음마스킹정격은 8~12dB로 설정되었다. 또한, 에코프리의 전화통화를 위한 단말결합손실은 전체음량정격이 10dB 일때 적어도 40dB 이상이어야 한다는 결론을 얻었다.

Ⅰ. INTRODUCTION

Acording to the development of terminal, transmission, and exchange technologies which are fundamental ones in telecommunication, communication network to provide various services are developed to digital-digital network through analog-analog and analog-digital-analog networks

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[1][2]. The introduction of ISDN(integrated services digital network) to realize the end-to-end digital communication is verifying this progress and requires the use of ISDN telephone, service terminal for speech communication. Especially, the speech path configuration using ISDN telephone shows the characteristics that the transmission losses are neglected and that the analog to digital conversion (or digital to analog conversion) is implemented in the ISDN telepone. These variations cause the speech transmission quality of ISDN and ISDN telephone to be newly speecified because of other characteristics than what the existing network has[3].

Speech transmission quality is, generally, specified by transmission characteristics from a talker's mouth to a listener's ear. These transmission characteristics are measured by LRs(loudness ratings) in decibels to evaluate the loudness values of complete call connections. Main purposes for loudnesss rating measurement are that the measured result will be used in network planning for good speech quality and used to evaluate the loudness of telephone sets.

In this regard, speech quality can be measured in terms of MOS(mean opinion score), the mean value on a predefined scale and MOS can be estimated from LRs. In general, the OLR(overall loudness rating) to specify the speech transmission quality of ISDN, the loudness rating from the talker's mouth to the listener's ear, consists only of the LRs of ISDN telephone, because the LR of the transmission line is 0dB. This means that without reasonable design of the LRs, ISDN cann't supply good speech quality for users in spite of several merits of ISDN telephone. Namely, if the LRs of ISDN telephone can be controlled, good speech quality of ISDN can be obtained[4].

On the other hand, talker echo on a telephone connection occurs when a portion of the transmitted signal returns to the talker as echo. If the echo signal returns in less than 3ms it is usually referred to as sidetone and if by more than 3ms it is usually referred to as talker echo. The perceived severity of talker echo depends on a number of parameters and the most significant parameters are the level of echo signal and the echo path delay. Sidetone can be used to provide a masking effect with respect to the echo, but it is known that sidetone is not important when the echo path delay exceeds about 200ms[5].

In ISDN of all digital telephone connections, the talker echo path loss is determined by (i) the sending and receiving loudness ratings of near-end ISDN telephone set and (ii) the TCL(terminal coupling loss) which is the loss/frequency characteristic from digital input to digital output due to the acoustic coupling between the receiver and the transmitter of far-end telephone set.

This paper describes the methods for establishing loudness ratings and talker echo of transmission quality for ISDN telephone connected to fully digital network. Subjective tests to establish the transmission performances of ISDN telephone which consist of SLR(sending LR), RLR(receiving LR), STMR(sidetone masking rating), and talker echo path loss are described, and then their experimental results are summarized.

Ⅱ. DEVELOPMENT OF THE MODEL SYSTEM

For the design of the LRs of ISDN telephone, the model system of digital speech communication for opinion tests is developed. The module configuration for the model system(called DTS, digital opinion test system) is shown in Figure 1. The DTS is designed according to the ISDN UNI (user-network interface) standardization in ITU-T I series, recommendations 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 block diagram of digital speech communication path made by the digital telephone module and network interface module is shown in Figure

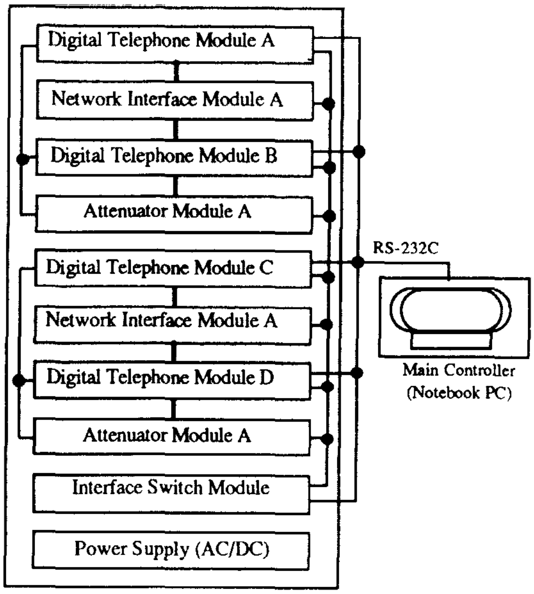

Fig 1. Module confiruration of DTS

2. The following is the functions of each part :

1) ACNSC(audio connection and signal conversion) part ;

Handset interface, A/D and D/A conversion, generation of various signal tone, etc.,

2) ICC(information channel control) part ;

Control and allocation of information channel (B1, B2), MUX/DMUX of speech channel.

3) DLI (digital line interface) part ;

Generation of frame signal, activation of physical line, basic access interface, layer 1 processing, etc.,

4) CC(common control) part ;

ACNSC and ICC control, DLI control, asynchronous communication, system monitoring, software processing, etc.,

5) DLT(digital line terminator) part ;

Termination in 4-wire digital line, recovery of frame signal, activation/deactivation of physical line, basic access interface, layer 1 processing, etc.,

6) SCG(system clock generation) part ;

Frame clock generation, generation of system and bus control clock,

7) RC(RS-232C communication) part ;

Host computer interface, dummy terminal interface, communication of remote control signal,

8) LD(LCD display) part ;

Display of operating status.

The software program of DTS, as shown in Fig 3, is implemented according to the ISDN UNI protocol recommended in ITU-T recommendations I.430, I.440, and I.450.

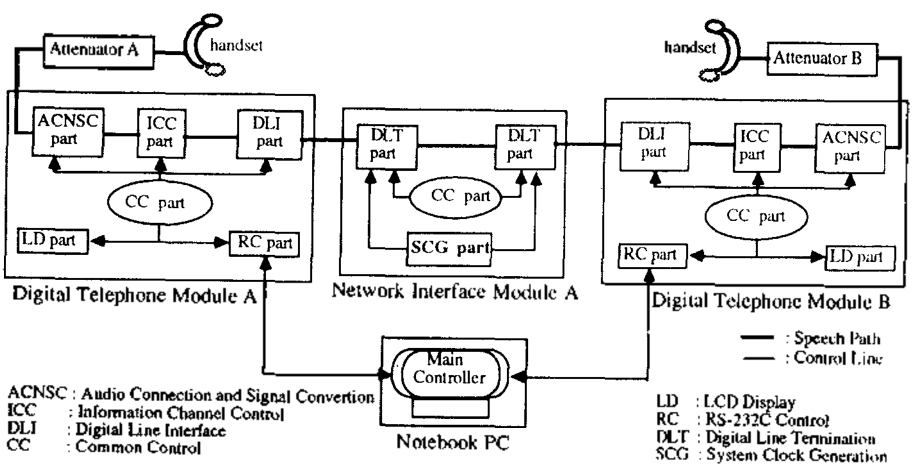

Fig 2. Block diagram of digital speech communication path

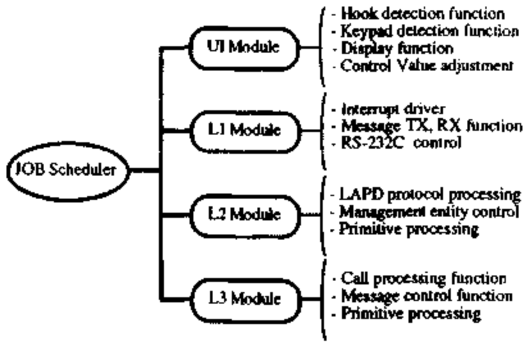

Fig 3. Function and configuration of DTS software

Ⅲ. OPINION TESTS FOR ESTABLISHING LOUDNESS RATINGS AND TALKER ECHO

3.1 Overview

The speech quality of telecommunication depends on sending quality, transmission quality, and receiving quality[6]. In general, the sending and receiving qualities depend entirely upon user's characteristics and they are difficult to quantify. The quality of speech communcation service is, therefore, designed by evaluating the characteristics of transmission quality. The speech quality which is assessed by loudness performance under the state of call connection uses the LR expressed in decibels to quantify numerically, and then the transmission quality can be expressed as a summation of the SLR, JLR(junction loudness rating), and RLR.

The loudness rating is a meaure to characterize the loudness performance from a talker's mouth to a listenr's ear. The loudness of the received speech signal depends on acoustic pressure provided by the talker and the loudess 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[7]. The effectiveness of speech quality and user's satisfaction depend, to a large extent, on the loudess loss. As the loudness rating is increased from a certain value, the listening effort is increased and user satisfaction decreases in parallel with decreased intelligibility. On the other hand, if loudness rating is low, user's satisfaction decreases because the received speech is loud. Therefore, it is important to establish appropriate ranges of the loudness ratings for sending and receiving parts, because the junction LR in the ISDN is 0dB as mentioned before[4].

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 sidetone paths, i.e., through the air, through the telephone handset, and through the user's head [4]. 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's satisfaction. Another aspect of insufficient sidetone loss is that talkers tend to reduce their speech levels 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, 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.

Echo control at the digital terminal is necessary to provide satisfactory echo performance to the customer at the other connection end. Furthermore, as more and more ISDN-based services become available and, later, as networks evolve to B-ISDN, it will become increasingly difficult to provide talker echo control for voice services within the transport network[8][9]. A solution to this problem is to provide an echo path loss at the terminal that is sufficiently high to provide echo-free performance without the use of additional echo control devices in the network. Accordingly, it is necessary to standardize TCL providing echo free telephone communications to telephone users. This standard can be classified into that on weighted TCL(TCLw), which is a weighted integral of loss/frequency function over the band 300 - 3400Hz, and that on stability loss, the least value in the band 0 - 4kHz, in order to express as a characteristic value[10][11].

3.2 Opinion test on OLR

For finding out the optimal range of OLR, opinion test was performed. The test result is shown in Figure 4 indicating the MOS change as a function of OLR.

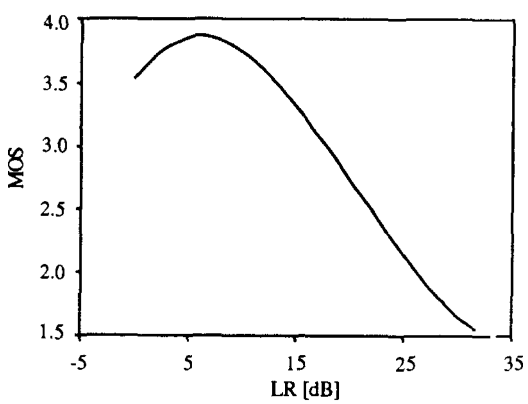

Fig 4. The relation between MOS and OLR

As can be seen in Figure 4, speech quality above MOS 3.5 can be obtained in the range of OLR 0 to 12dB. Also of interest in Figure 4 is the maximum MOS value appeared at about 8dB. In the cumulative percentage of users who respond to the opinion scale of speech quality, MOS to which a 50 percent of the subjects responds above "fair" is 2.7 and MOS to which a 90 percent responds above "fair" or a 50 percent above "good" is 3.7. Thus, the OLR of 6 to 10dB above MOS 3.7 is selected. This can be the desirable range to provide the comfortable speech transmission quality in ISDN and this range can cover a tolerance of ±2dB from the optimal value.

3.3 Opinion test on input level to the CODEC.

The effect of input levels to the CODEC on speech quality must be contemplated to determine the SLR of ISDN telephone[12]. If the input levels to the CODEC are not controlled properly, new transmission impairments can be arised. 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 big 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 quality of input level to the CODEC, the relationship between user opinion and input level to the CODEC was established. The listening test that users respond to the speech quality according to the variation of input level was implemented as the block diagram shown in Figure 5. The seven input levels to the CODEC, 0 to 40dB relative to 7dBV, the overload level of the CODEC using sinewave, were used and three listening levels before and after the preferred level were utilized not to have an effect on test results.

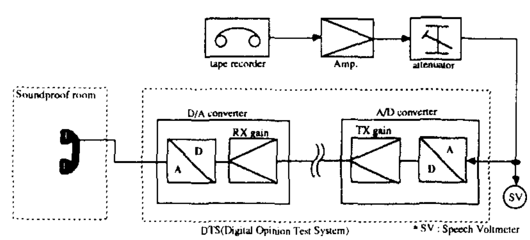

Fig 5. A bolck diagram for opinion test on input level to the CODEC

The relationship between MOS and input level to the CODEC is shown in Figure 6. As illustrated in Figure 6, the desirable input level falls between

-12dB and -18dB to reduce quantizing noise and overload distortion occurring in the CODEC. The maximum MOS value is indicated at about -15dB of input level. The 70dB of listening level shows the best one in this test. It is noted that the effect of quantizing noise on users' perceived quality is different according to the listening levels even if the same signal to noise ratio exists.

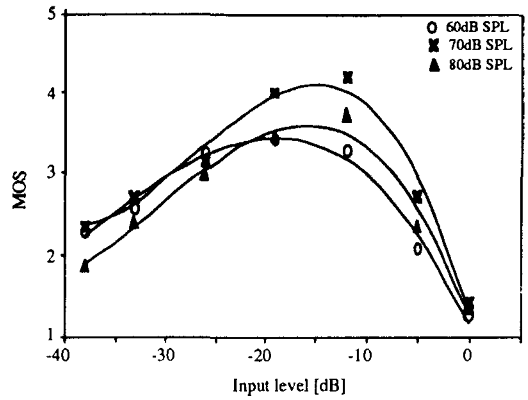

Fig 6. The relation between MOS and input level to the CODEC

3.4 Opinion test on STMR

To ensure acceptable quality of STMR, the relationship between user opinion and STMR was established. The block diagram of test system for measuring STMR is shown in Figure 7. To control the effect of SLR and RLR on STMR, SLR and RLR were fixed to the values obtained from the design of SLR and RLR. The variations of STMR with 6 levels, 0 to 25dB, and room noise with the Hoth noise of 45dB(A) and 60dB(A) were used in this test.

The relationship between MOS and STMR is shown in Fig 8. It shows that the STMR above MOS 3 is between 5dB and 18dB and the optimal STMR is 12dB.

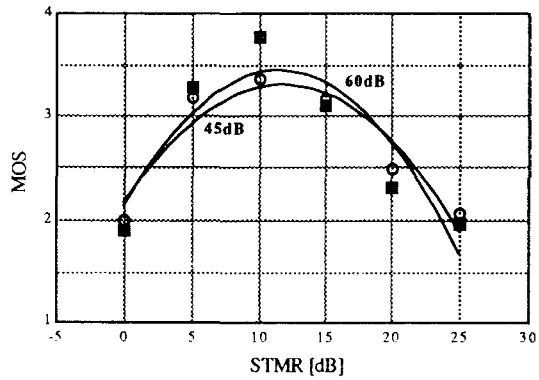

Fig 8. The relation between MOS and STMR

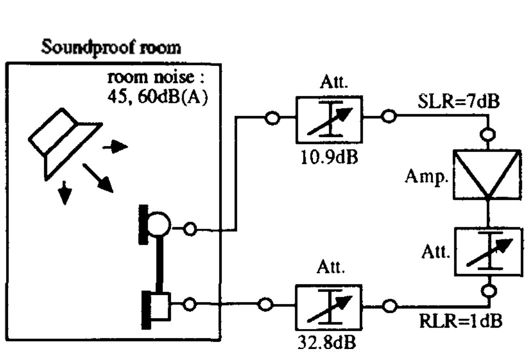

Fig 7. A bolck diagram for opinion opinion test on STMR

3.5 Opinion test on talker echo

The opinion test on talker echo to make a standard on TCLw based on users' perceived quality was performed. From the correlation between talker echo and user opinion on quality, the standard on TCLw can be proposed when overall loudness

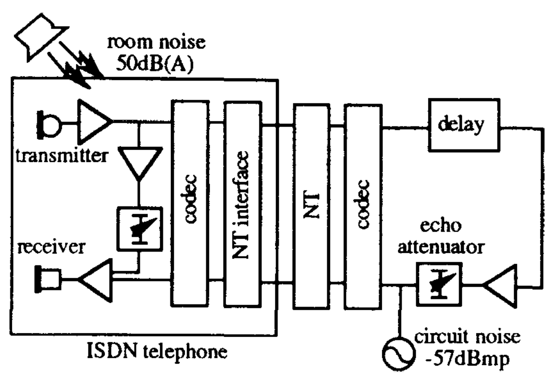

Fig 9. Experiment configuration of opinion test on talker echo

rating of ISDN telephone is normalized to 10dB.

The test configuration is given in Figure 9, using a digital telephone module of the model system of DTS as an ISDN telephone set. The measure of talker echo used was TELR(talker echo loudness rating) which is loudness rating at echo path and the range of TELR was 5~68dB in step of 7dB. The echo path delay covered from 20ms to 600ms and STMRs were 5 and 15dB.

From the opinion test on talker echo, using the parameters of the echo path delay and STMR with the opinion scale on transmission quality, the correlation between TELR and the cumulative percentage of subjects evaluating transmission quality as "bad" when the STMR of 15dB, according to each echo path delay, is given in Figure 10. As can be seen in Figure 10, if the threshold of an echo-free telephone communication is taken as the cumulative percentage of 2.5%[5], the talker echo path loss of at least of 50dB TELR is necessary considering the longest echo path delay of 600ms in the test.

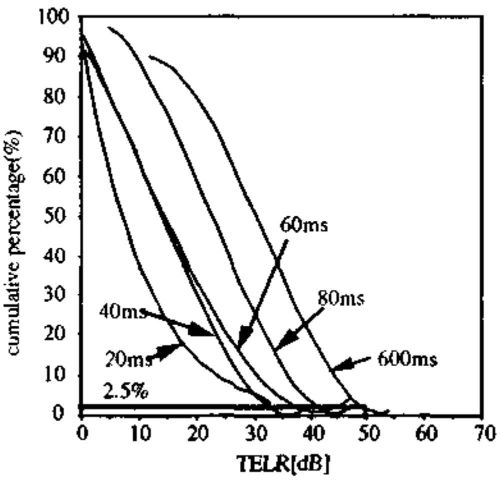

Fig 10. Correlation between TELR and the cumulative percentage of subjects evaluating transmission quality as bad

IV. ESTABLISHMENT OF LR AND TALKER ECHO

4.1 Loudness ratings

As noted earlier from the opinion test on input level to the CODEC, it was reported that input level must be reserved below -12dB and the desirable range was between -12dB and -18dB. From this result, let's consider how to establish SLR. First, the speaking level of test speech data is controlled to be 85dBSPL at the mouth reference point, because the speaking levels of most telephone users fall in the range of 65 to 75dBSPL and about 98 percent in the below 85dBSPL when the long time average of speaking level at the mouth reference point is measured. Then, the input amplifer gain of the CODEC is adjusted speech level to be -5dBV(-12dB relative to the overload level of the CODEC using sinewave) at the CODEC input. The sending sensitivity/frequency characterisitc measured in the stated condition is shown in Figure 11. The SLR should be calculated from the characteristic of Figure 11 by means of ITU-T Recommendation P.79, and the SLR calculated as the input level of -12dB equals to 6dB. This means that the SLR should not be less than 6dB because this value is the limiting point not to generate the overload of the CODEC, and then the value of SLR in the range of 6 to 8dB considering a tolerance of +2dB is proposed.

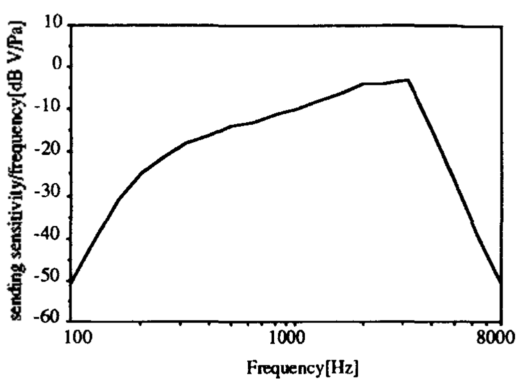

Fig 11. The measured sending sensitivity/frequency characteristic

The OLR of 8dB as described earlier from the opinion test was known as the optimal speech quality of ISDN speech communication and the OLR of 6dB to 10dB considering ±2dB from the optimal OLR is the range covering a tolerance of ±2dB.

The OLR for an end-to-end connection of ISDN is given by

OLR = SLR + JLR + RLR.

In the sum, the SLR containing a tolerance of +2dB is 6 to 8dB and the JLR is 0dB. The RLR containing a tolerance is, therefore, calculated as 0 to 2dB.

From the opinion test on STMR, relation between user's perceived loudness and STMR is also obtained as shown in Figure 12. In the cumulative percentage of users responding to the sidetone amplitude levle, if the STMR level to which a 50 percent of the subjects responds to be "quieter than preferred" or "much quieter than preferred" is taken as permissible high boundary and the STMR to which a 50 percent responds "louder than preferred" or "much louder than preferred" as permissible low boundary, the acceptable range lies between STMRs of 6dB and 17.5dB. Using this background and the optimal STMR, the values of STMR in the range of 8 to 16dB considering a tolerance of ±4dB are proposed.

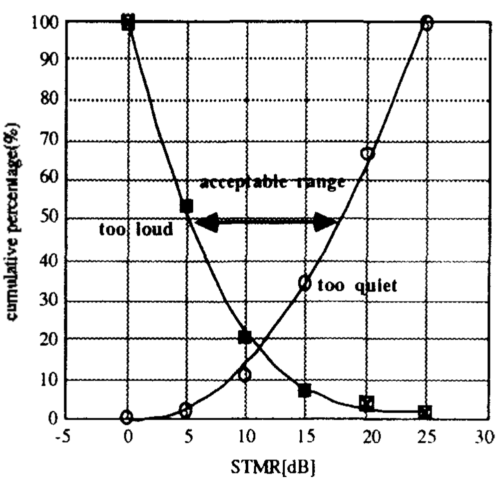

Fig 12. The acceptable range of STMR

4.2 Talker echo

As noted earlier, it was reported that an talker echo path loss of 50dB or greater is necessary to provide adequate talker echo control for all delays in ISDN. From the above section, as the SLR and RLR of ISDN telephone set were designed as in the ranges of 6~8dB and 0~2dB, respectively, the maximum OLR is 10dB.

As the talker echo path loss in ISDN is sum of the OLR of ISDN telephone set and the TCL, the terminal coupling loss of TCLw of at least 40dB is necessary to provide echo-free telephone communications to telephone users when the overall loudness rating of ISDN telephone is normalized to 10dB.

V. CONCLUSION

This paper described the methods for establishing loudness ratings and talker echo of transmission quality of ISDN telephone connected to fully digital network. For establishing the SLR, RLR, STMR, and talker echo of ISDN telephone, opinion tests for analyzing the relations between MOS and input level to CODEC, OLR, sidetone amplitude, and TELR were implemented. The following values are proposed from the test results :

1) nominal values of OLR in the range of 6 to 10dB including a tolerance of ±2dB from the optimal OLR ;

2) nominal values of SLR in the range of 6 to 8dB including a tolerance of +2dB ;

3) nominal values of RLR in the range of 0 to 2dB including a tolerance of +2dB ;

4) nominal values of STMR in the range of 8 to 16dB including a tolerance of ±4dB from the optimal STMR ;

5) TCL, the terminal coupling loss of TCLw of at least 40dB for echo-free when OLR = 10dB.

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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REFERENCES

1. J. Gruber and G. Williams, Transmission performance of evolving telecommunications network, Aretech House. INC., 1992.
2. P. Bocker, ISDN : the integrated services digital network, Springer-Verlag Berine Heidelberg, 1988.
3. CCITT Recommendation, "Methods for evaluating the transmission performance of digital telephone sets," Vol. V, Rec. P.66, Geneva, 1992.
4. H. Nomura, H. Okikawa, and M. Nishinu, "Transmission performance design for digital telephone sets which eliminates talker's echo problems in mixed digital-analogue networks," The Transactions of the IECE of the Japan, Vol.E.69, No.12, 1986.
5. ITU-T SG 12, COM 12-28, "Talker echo on all digital connections including ISDN," Geneva, Mar. 1990.
6. CCITT, Handbook on telephonometry, Geneva, 1987.
7. J. Fennick, Quality measures and the design of telecommunications system," Artech House, INC., 1988.
8. EIA/TIA-579, "Acoustic-to-digital and digital-to-acoustic transmission requirements for ISDN terminals," 1991.
9. ITU-I SG 12, COM 12-59, "Termianl coupling loss measurements of simulated 4-wire analog and digital telephone sets," Geneva, Nov. 1990.
10. ITU-T Recommendation G.122, "Influence of national systems on stability, talker echo and listener echo in interational connections," 1988.
11. K. O. Kang, D. Y. Jang, "Evaluation standard on talker echo of ISDN telephone," *Proc. the Acoustical Society of Korea*, Seoul, Nov. 1993(in Korean).
12. M. Nishino, H. Nomura, and F. Amemiya, "Establishing Transmission Performance for Digital Telephone Sets," E.C.L Tech. Journal, NTT of Japan, Vol.33, No.8, 1984.

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