Field study of intercom systems and loudspeaking telephones

Lundin, F. J.

journal: STL-QPSR
volume: 16
number: 1
year: 1975
pages: 027-054

http://www.speech.kth.se/qpsr
II. SPEECH COMMUNICATION SYSTEMS

A. FIELD STUDY OF INTERCOM SYSTEMS AND LOUDSPEAKING TELEPHONES

Fred J. Lundin*

Abstract

A subjective evaluation of and users' attitudes towards loudspeaking telephone and intercom systems versus handset telephones has been undertaken. Intercom telephone systems are used most commonly for messages shorter than half a minute in a company, but for longer discussions personal visits are preferred to telephone calls. When using a loudspeaking telephone system the voice level is raised approximately 7 dB (A) compared to face-to-face conversation. The main cause of quality degradation in loudspeaking systems is the influence of room acoustics and acoustical noise. The rooms we studied had an average reverberation time of 0.5 seconds. The distance between the speaker and a point where the direct sound is equal to the reflected sound varies from an average of 0.47 m for rooms with low absorption to 0.76 m for rooms with high absorption. The normal distance between the mouth of the speaker and the microphone averages 0.63 m. Average noise levels in the rooms were measured. Loudness and timbre ratings of the various loudspeaking telephone systems were studied as well as speaker identification, intelligibility, naturalness, and the false operation of the automatic voice switching.

* LM Ericsson Telemateriel AB, Tyresö; thesis student at the Department of Speech Communication, KTH.
Introduction

Since the sixties loudspeaking systems have been more commonly available to the general public. By loudspeaking systems we here refer to loudspeaking telephones connected to the public network and loudspeaking intercom telephone sets, mostly used in companies. The systems presently available on the market are not perfect products. There are technical limitations in the matching of systems to the user and vice versa. In order to secure a better understanding of the users' attitude towards available systems and their handling, a study was initiated in the period May-August 1972 in Stockholm. Similar studies have been undertaken on loudspeaking telephones for domestic use (Fletcher 1970) and comparisons with standard handsets (Heberle 1968; Larsson and Johansson 1974).

The study

The main scope of the study was to analyze shortcomings in the sound reproduction and the automatic voice switching performance. Speech levels, perceived quality, room acoustics and noise levels were therefore studied technically and by noting the subjective evaluation of systems, the handling of the equipment and the amount of training previously received.

Rooms of interest were offices with associated spaces where intercom sets were installed, e.g. in store rooms, work shops, canteens etc. The study covered intercom systems from the six most common manufacturers spread in relation to their market shares. Reply forms from 213 intercom users have been processed. The intercom stations were spread on 14 different systems types and had been installed during the past five years. 57 loudspeaking sets were included in the study and were mainly of the type Ericovox (without handset).

A questionnaire was used for data collection. It consisted of a general part for all users with questions on frequencies of usage of different telephone systems. A part for intercom users and a similar part for loudspeaking telephone users dealt with questions concerning the sound transmission. To each questionnaire a measurement part was attached which was filled in by the interviewer.

The methods of the study, the results and the discussions will be presented in the following parts:
Method

An important factor when developing a loudspeaking telephone system is the average speech level that affects the microphone. There exist several studies of speech levels at face-to-face conversation (Knudsen and Harris 1950; Fletcher 1953; Richardson 1953; Peterson and Gross 1963; Gardner 1966). But how do such data relate to speech levels when using a loudspeaking system? Measurements were accordingly made during conversation face-to-face and over a loudspeaking telephone system. In the first case a sound level meter (Brüel and Kjær 2203) was placed on the user's desk in front of him and about one meter from the mouth. The instrument was read off discreetly so as not to affect the speaker in any psychological way. In the second case it was placed close to the telephone set and its microphone.

The distance between the mouth and the microphone, the talking distance, is an important factor and determines the ratio between the direct speech sound and room reflections of the transmitted sound. The distance also affects the sound level at the microphone. This distance was measured when the speaking party had made a call and had a conversation.

Results

At an ordinary conversation "face-to-face" the speech level had an average value of 64.8 dB(A) with a standard deviation of 4.0 dB(A). (These notations are used in this paper to indicate the sound pressure level with A-weighting in relation to 20 µPa. When units not are given, we use SI units.) The distribution is shown in Fig. II-A-1, the dashed plot.

When measuring close to the set the speech level value averaged 73.3 dB(A) with a standard deviation of 4.0 dB(A). The distribution is shown in Fig. II-A-1, the solid line plot. When comparing the two sound levels - "face-to-face" and via a loudspeaking system - there is an average increase
Fig. II-A-1. Distribution of speech levels.
of 8.8 dB(A) for the loudspeaking system. The distribution is shown in Fig. II-A-2, the dashed plot.

To be able to compare the two speech levels mentioned above the speech distance must be equal. The "face-to-face" value was measured at a distance of one meter but the talking distance averaged 0.63 m (standard deviation 0.17 m). Which corrections shall we add to the values obtained when using a loudspeaking system to get the speech levels at one meter distance if we also take into account the influence of the room acoustics? Let us consider some relations from the theory of room acoustics.

The sound pressure level $L_p$ (re 20$\mu$Pa) at the distance $d$ from an omnidirectional point source with the acoustic output power $P_w$, is in free space (and normal temperature and pressure)

$$L_p = 10 \cdot \log \frac{P_w}{P_o} + 10 \cdot \log \frac{1}{4\pi d^2}$$

where $P_o$ is a reference power of $10^{-12}$W. For convenience in further considerations we define the acoustic power level of the sound source $L_w$ (re 1 pW) as

$$L_w = 10 \cdot \log \frac{P_w}{P_o}$$

The "distance law" (eq. 1) gives a decrease of 6 dB of the sound pressure level when the distance to the source is doubled. However, if the sound source is situated in a room with the total reflecting area $S$ and the mean absorption coefficient $\bar{\alpha}$ the sound pressure level is instead given (Brandt 1958; Krokstad 1967; Beranek 1971) by

$$L_p = L_w + 10 \cdot \log \left( \frac{1}{4\pi d^2} + \frac{4}{R} \right)$$

where the room constant (Gardner 1960) for the specific room is

$$R = \frac{\bar{\alpha} S}{1-\bar{\alpha}}$$

The term $1/4\pi d^2$ represents the contribution from the direct sound and the term $4/R$ the contribution from the reflected sound. If the sound pressure level in a room is plotted as a function of the distance to the...
Fig. II-A-2. Distribution of the increases of the speech levels when using a loudspeaking telephone system compared to face-to-face discussion.
source we get the curves in Fig. II-A-3 where the room constant R is the parameter (Beranek 1971).

The received sound level at 1 m in face-to-face conversation is denoted $L_p(1)$. We assume the mouth to be an omnidirectional point source with the acoustic power level $L'_w$. The effect of the specific room acoustics are defined with the room constant R. Thus the sound level at 1 m is

$$L_p(1) = L'_w + 10 \cdot \log\left(\frac{1}{4\pi} + \frac{4}{R}\right) \quad (5)$$

In the second case using a loudspeaking telephone system the acoustic power level is denoted $L'_w$. Here the distance between the mouth and the microphone is $d$ meters and the sound level at the microphone $L_p(d)$.

$$L_p(d) = L'_w + 10 \cdot \log\left(\frac{1}{4\pi d^2} + \frac{4}{R}\right) \quad (6)$$

Now we define the rise in the speech level as the difference

$$\Delta L = L'_w - L'_w', \text{ which is}$$

$$\Delta L = L_p(d) - 10 \cdot \log\left(\frac{1}{4\pi d^2} + \frac{4}{R}\right) - L_p(1) + 10 \cdot \log\left(\frac{1}{4\pi} + \frac{4}{R}\right) \quad (7)$$

or

$$\Delta L = L_p(d) - L_p(1) + 10 \cdot \log\left(\frac{1}{4\pi d^2} + \frac{4}{R}\right) \quad (8)$$

As we have measured $L_p(d)$, $L_p(1)$, $d$, $S$ and estimated $\overline{e}$, we can calculate for everyone of the users and specific room acoustics the difference between the speech levels in the two cases. An average increase of the speech level ($\Delta L$) of 6.6 dB(A) (standard deviation 3.9 dB(A)) at conversation via a loudspeaking system compared to "face-to-face" conversation is then obtained. This distribution is shown in Fig. II-A-2, the solid line plot. For people who make on an average up to two calls a day the average rise of the speech level is 7.4 dB(A). In Fig. II-A-4 the average speech distance is plotted for five groups of users, depending on the number of calls they make a day.

Table II-A-I shows subjective assessments of how frequently the talking party speaks at a distance greater than one arm's length from the set.
DIFFERENCE BETWEEN THE SOUND PRESSURE LEVEL AND THE LEVEL OF THE SOUND SOURCE (L_p - L_w) dB

FIG. 11-A-3. The sound pressure level in a room as a function of the distance to the source. Room constant is the parameter.

DISTANCE (d) FROM SOUND SOURCE (meters)
Fig. II-A-4. Average talking distances for loudspeaking system users depending on the average number of calls they make a day. The vertical lines mark the standard error of the mean.
<table>
<thead>
<tr>
<th>Telephone set</th>
<th>Never</th>
<th>Seldom</th>
<th>Sometimes</th>
<th>Often</th>
<th>Always</th>
<th>Number of users</th>
</tr>
</thead>
<tbody>
<tr>
<td>Loudspeaking telephone</td>
<td>9</td>
<td>39</td>
<td>44</td>
<td>9</td>
<td>0</td>
<td>57</td>
</tr>
<tr>
<td>Intercom</td>
<td>11</td>
<td>25</td>
<td>54</td>
<td>9</td>
<td>1</td>
<td>209</td>
</tr>
<tr>
<td>Total</td>
<td>11</td>
<td>28</td>
<td>51</td>
<td>9</td>
<td>1</td>
<td>266</td>
</tr>
</tbody>
</table>

Table II-A-1. Relative frequency (%) when users speak more than one arm’s length from the set.

Discussion

The speech level adapted by the speaking person depends on several factors. The average value of the sound level at face-to-face conversation agrees with those already documented in the literature in spite of our measurements being made in a reverberant room. Other possible sources of systematic errors in the two sets of measurements are reflections from the desk surface and from the surrounding walls, which affect the sound level meter. However, the difference between positioning the instrument on the desk top or freely in the room should not be very large. If the communication takes place via a telephone system, the line attenuation or gain will influence the speaker and if this attenuation is high, he tends to raise his voice. In a noisy room the voice level may be expected to increase an additional 0.3-0.4 dB for each one dB-increase of the noise level (Gardner 1966).

The increase of the speech level about 7 dB(A) when an intercom or loudspeaking telephone is used compared to face-to-face conversation, depends on psychological factors. When speaking into a microphone which by an amplifying system is connected to a loudspeaker or a registrating system (e.g. tape recorder), the talker concentrates more on the speech with regard to its content, grammar and audibility. This increased concentration results in a raising of the voice in order to make sure that the listening party in an intercom system apprehends the speech. The reproduced sound of the talker’s voice is the only “face” he has in this situation and he wishes to be perceived as clearly as possible. This increase of the speech level may also depend on the fact that subjects talk a few dB louder while speaking into a microphone more distant than the normal handset (CCITT Question 17/XII, Annex 2). For people who make only a few calls a day the rise of the speech level is greater than for the whole
group, which make us suspect that this group of people is badly trained on the system and consequently that people who seldom use loudspeaking systems talk louder than more trained people.

When measuring the speech level at face-to-face conversation the distance between the mouth and the microphone was not measured exactly but estimated to be one meter. The room absorption coefficient was also estimated. Due to the variations in these parameters there is some unreliability in the value of $7\, \text{dB(A)}$ speech level increase. This value is higher than anticipated and further studies are planned to check the validity.

The average talking distance found in this study is applied to the party making a call. The other party may stand several meters from his intercom set when he answers the call if the set is in a direct acceptance mode. The more people are trained on the system and use the loudspeaking telephone or intercom set the greater distance they will allow between mouth and microphone. This range is about $0.1\, \text{m}$ and depends on the fact that the system operates well at this greater talking distance and that this distance is more convenient to use. Untrained people often talk very close to the set until they get instructions that the system operates well on greater talking distance too.

**The listening party**

**Methods**

The data collection was undertaken by means of a questionnaire, which consists of one part for loudspeaking telephone users and a similar part for intercom telephone users. The user was asked to perform the loudness and timbre ratings on a three-level scale. The loudspeaking telephone sets have keybuttons for extra receiving amplification and the frequency of usage of this facility for internal and external calls was noted.

General methods for predicting the intelligibility of speech transmission systems (Beranek 1947; Fletcher 1953; American national standards methods 1969) from their physical characteristics cannot substitute direct evaluations of system quality from the users' point of view. We have selected three parameters of special interest, speaker identification, intelligibility, and naturalness. Five-level scales were used to classify these parameters in grades from 1 to 5 where 5 is the best value of the parameter.
Speaker identification is the ability to recognize a known person by his voice only. Articulation, pauses, and dialect influence this parameter. The user should classify how often he recognizes the speaking party by his voice. Intelligibility is often measured as the relative number of correctly registered nonsense syllables or specially selected words transmitted over a system (Beranek 1949). In this study, on the other hand, the users gave their opinion of how often they had to ask the other party to repeat messages.

In many cases when the intelligibility of a system is measured, it shows values close to 100%. Then it is more realistic to study the sound quality from a subjective point of view and especially the naturalness (Gleiss 1971). The users gave their opinion of the naturalness of the system compared to the naturalness obtained when people speak to each other in the same room.

The loudspeaking telephone sets in this study were equipped with a voice-switched amplifier to prevent self oscillation or "singing" (Busala 1960; Cleary and Cannon 1961; Clemency and Goodale 1961; Copping and Fidler 1967; Galyas 1969; Takeda and Kondoh 1972; Clarke and Gale 1973; Suntop 1974).

The gain of the amplifier was influenced by the speech in such a way that it was normally directed away from the speaking and towards the listening party. As usual this voice-switched amplifier was centrally placed (i.e. in the exchange) in an intercom system and the speech transmission was arranged on separate cable pairs for the microphone and loudspeaker signal (4-wire system). In loudspeaking telephone systems and in some intercom systems the voice-switching function was decentralized and placed in each telephone set. In these systems the speech transmission normally used the same cable pair for the microphone and loudspeaker signal (2-wire system) due to a hybrid network.

A number of conditions apply to voice-switched amplifiers to provide natural voice switching without affecting conversation. In 2-wire systems the voice switched amplifiers in the transmitting end as well as in the receiving end must be set in the right speech direction. The operating time from receiving to transmitting mode should be sufficiently short to prevent the first syllable of the words to be clipped (Barnes 1972). If, on the other hand, the operating time is set too short this results in increased noise sensitivity and a click is heard at each mode transfer. Observations of clipping were noted in a three-level scale.
The acoustic coupling between the loudspeaker and the microphone must be sufficiently low. In rooms with high echo this may cause problems. When A speaks, the speech channel is directed from A to B. The speech reproduced by B’s loudspeaker is reflected by walls etc. and affects B’s microphone a few milliseconds later. The voice switching circuitry may register this as if B had replied and the speech channel is reversed from B to A. The speech from A disappears from B’s loudspeaker. A continues to speak, unconscious of what has happened, the disturbing echo disappears and the speech channel A-B is reopened. This cycle is repeated and B hears only parts of what A is saying (chopped speech). Observations of chopped speech were noted.

If the system is constructed so that an abnormally high speech level is required for B to break in when A is talking, the B-extension will consider the speech reproduction unnatural. Break-in may be required if B wishes to comment on something A has just said. Observations of break-in problems were noted too.

Results

The results of the preferred loudness level for intercom are given in Table II-A-II. 88% of the participants considered the sound level nearest to that of their own intercom sets to be the best.

<table>
<thead>
<tr>
<th>Manufacturer</th>
<th>Too weak</th>
<th>Normal</th>
<th>Too high</th>
<th>Number of replies</th>
</tr>
</thead>
<tbody>
<tr>
<td>A</td>
<td>4</td>
<td>90</td>
<td>7</td>
<td>88</td>
</tr>
<tr>
<td>B</td>
<td>9</td>
<td>88</td>
<td>3</td>
<td>65</td>
</tr>
<tr>
<td>C</td>
<td>12</td>
<td>88</td>
<td>0</td>
<td>16</td>
</tr>
<tr>
<td>D</td>
<td>7</td>
<td>86</td>
<td>7</td>
<td>14</td>
</tr>
<tr>
<td>E</td>
<td>8</td>
<td>92</td>
<td>0</td>
<td>13</td>
</tr>
<tr>
<td>F</td>
<td>9</td>
<td>91</td>
<td>0</td>
<td>11</td>
</tr>
</tbody>
</table>

Total | 7.2 | 88.4 | 4.3 | 207

Table II-A-II. Perceived loudness for intercom. Relative numbers in %.

The loudspeaking telephones had key-buttons for extra amplification of the loudspeaker output. Table II-A-III shows how often the participants of this study used the additional gain feature. Generally speaking they could be classified in three categories: those who never used the additional gain feature, those who always used the additional gain feature and those who did not use it in internal calls but always in external calls.
Table II-A-III. Use of additional gain when a loudspeaking telephone is used. Relative numbers in %.

Table II-A-IV shows how the timbre is perceived in the intercom and loudspeaking set. 88% of the intercom users considered the timbre to be acceptable. The quality was perceived as too much treble in some systems and too much bass in others.

<table>
<thead>
<tr>
<th>Manufacturer</th>
<th>Too much</th>
<th>Normal</th>
<th>Too much</th>
<th>Number of</th>
</tr>
</thead>
<tbody>
<tr>
<td>Intercom</td>
<td>bass</td>
<td></td>
<td>treble</td>
<td>replies</td>
</tr>
<tr>
<td>A</td>
<td>1</td>
<td>92</td>
<td>7</td>
<td>86</td>
</tr>
<tr>
<td>B</td>
<td>3</td>
<td>93</td>
<td>4</td>
<td>68</td>
</tr>
<tr>
<td>C</td>
<td>19</td>
<td>81</td>
<td>0</td>
<td>16</td>
</tr>
<tr>
<td>D</td>
<td>7</td>
<td>64</td>
<td>29</td>
<td>14</td>
</tr>
<tr>
<td>E</td>
<td>0</td>
<td>92</td>
<td>8</td>
<td>13</td>
</tr>
<tr>
<td>F</td>
<td>0</td>
<td>64</td>
<td>36</td>
<td>11</td>
</tr>
<tr>
<td>Total</td>
<td>3.4</td>
<td>88.0</td>
<td>8.7</td>
<td>208</td>
</tr>
<tr>
<td>Loudspeaking telephone</td>
<td>12</td>
<td>75</td>
<td>12</td>
<td>57</td>
</tr>
</tbody>
</table>

The results of speaker identification, intelligibility and naturalness are presented in Table II-A-V. When comparing values of speaker identification and intelligibility one finds that the ranking order between the products of different manufacturers is generally similar (Speaker identification: E-A-B-D, C-F, intelligibility: E, B-A-D, C, F, naturalness: E, D-A-B-F-C). The ranking order of the total of parameter average values (E-B-A-D-C-F) agrees well with that of the intelligibility. The three systems with the lowest total ratings C, D, F deviate from normal timbre as shown in Table II-A-IV.

The relations between the three parameters have been illustrated by calculations of correlation coefficients: speaker identification - intelligibility 0.39, naturalness - intelligibility 0.38, naturalness - speaker identification 0.26. It is apparent that the three parameters are relatively independent of each other. However, the intelligibility is more related to both speaker identification and naturalness than the relation between the latter.
Table II-A-V. Average values of speaker identification, intelligibility and naturalness.

Table II-A-VI shows the frequency of occurrence of problems associated with voice-switched amplifiers which results in losses of intelligibility due to false switching phenomena such as clipping, chopping and break-in. These are graded from 1 to 5 as follows: 5 = no problems, 3 = occasional problems, 1 = frequent problems.

Table II-A-VI. Relative existence (in %) of problems connected with the automatic voice switching in intercom and loudspeaking telephone.

The calculated average values of the replies applying to the different intercom manufacturers are presented in Table II-A-VII.

### Table II-A-V

<table>
<thead>
<tr>
<th>Manufacturer</th>
<th>Speaker identification</th>
<th>Intelligibility</th>
<th>Naturalness</th>
<th>Total</th>
<th>Number of replies</th>
</tr>
</thead>
<tbody>
<tr>
<td>A</td>
<td>4.2</td>
<td>3.5</td>
<td>3.7</td>
<td>11.4</td>
<td>86</td>
</tr>
<tr>
<td>B</td>
<td>4.1</td>
<td>3.8</td>
<td>3.6</td>
<td>11.5</td>
<td>69</td>
</tr>
<tr>
<td>C</td>
<td>3.9</td>
<td>3.4</td>
<td>3.4</td>
<td>10.7</td>
<td>16</td>
</tr>
<tr>
<td>D</td>
<td>3.9</td>
<td>3.4</td>
<td>3.8</td>
<td>11.1</td>
<td>14</td>
</tr>
<tr>
<td>E</td>
<td>4.6</td>
<td>3.8</td>
<td>3.8</td>
<td>12.2</td>
<td>13</td>
</tr>
<tr>
<td>F</td>
<td>3.6</td>
<td>3.4</td>
<td>3.5</td>
<td>10.5</td>
<td>11</td>
</tr>
<tr>
<td><strong>Total</strong></td>
<td><strong>4.1</strong></td>
<td><strong>3.6</strong></td>
<td><strong>3.6</strong></td>
<td><strong>11.3</strong></td>
<td><strong>209</strong></td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Loudspeaking telephone</th>
<th>Speaker identification</th>
<th>Intelligibility</th>
<th>Naturalness</th>
<th>Total</th>
<th>Number of replies</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>4.2</td>
<td>3.7</td>
<td>3.2</td>
<td>11.1</td>
<td>57</td>
</tr>
</tbody>
</table>

Table II-A-VII. Average intercom values.

<table>
<thead>
<tr>
<th>Manufacturer</th>
<th>Clipping</th>
<th>Chopping</th>
<th>Break-in</th>
<th>Total</th>
<th>Number of replies</th>
</tr>
</thead>
<tbody>
<tr>
<td>A</td>
<td>3.1</td>
<td>3.7</td>
<td>4.1</td>
<td>10.9</td>
<td>77</td>
</tr>
<tr>
<td>B</td>
<td>4.0</td>
<td>4.6</td>
<td>4.3</td>
<td>12.9</td>
<td>70</td>
</tr>
<tr>
<td>C</td>
<td>4.0</td>
<td>4.4</td>
<td>3.9</td>
<td>12.3</td>
<td>16</td>
</tr>
<tr>
<td>D</td>
<td>3.0</td>
<td>3.4</td>
<td>3.7</td>
<td>10.1</td>
<td>14</td>
</tr>
<tr>
<td>E</td>
<td>3.3</td>
<td>3.5</td>
<td>4.2</td>
<td>11.0</td>
<td>13</td>
</tr>
<tr>
<td>F</td>
<td>4.5</td>
<td>4.6</td>
<td>4.4</td>
<td>13.5</td>
<td>11</td>
</tr>
<tr>
<td><strong>Average</strong></td>
<td><strong>3.6</strong></td>
<td><strong>4.1</strong></td>
<td><strong>4.1</strong></td>
<td><strong>11.8</strong></td>
<td><strong>201</strong></td>
</tr>
</tbody>
</table>
Discussion

The method of using the questionnaire for classification of perceptual parameters provides subjective results only. The users were asked to class loudness, timbre, naturalness etc. in relation to their own ideas of these parameters. Many of the users had never considered the sound quality from this point of view and had difficulty in giving relevant information. It might have been preferable to grade the quality of the existing telephone set in relation to some reference telephone set. The results should therefore be regarded as relative values only showing how people perceive the sound reproduction guiding us towards a general outline of the users' demands. In order to analyze these outlines of loudness and timbre, 5-digit scales should have been used instead of the 3-digit scales now being employed.

Some types of intercom telephone sets had key buttons for two different listening levels equal to the key button mentioned regarding the loudspeaking telephone. These systems show fewer variations in perceived loudness. However, a continually adjustable volume control was requested on each set to adjust the loudspeaker output to the speaking party's voice level.

It should also be mentioned that when a loudspeaking telephone is used, the party at the other end of the line normally uses a non-loudspeaking telephone set. The speech is then subject to distortion due to the carbon microphone.

Many of the interviewed persons were unaware of the system having automatic voice switching. This indicates poor training in the systems, their functions and use. To make proper use of intercom and loudspeaking telephones presently marketed, it is essential to be informed of the setting-up procedure, the acoustic properties of the system (such as suitable speech level, speech distance, traffic discipline, noise sensitivity) as well as special services provided with the system (e.g. all-call, recall, paging). Due to the lack of knowledge of voice switching when answering the questions, a high degree of accuracy cannot be claimed. However, it is established that 44% of the intercom users had noted disadvantages in voice switching operations. The corresponding figure for loudspeaking telephones was 50%.
Intercom sets, from six different manufacturers have been studied with respect to their various qualities. It should be noted that the sets from the six manufacturers were split up into 14 different system types installed during the past five years. For a technical comparison between intercom sets from different manufacturers from the point of view of quality and performance, equivalent systems should be chosen with consideration to size, age, application etc. The aim of this study was not to establish the best telephone system but to make a survey of the users' attitudes towards, and use of, the systems. However, a manufacturer often has his own criterion of sound reproduction, e.g. loudness and timbre, and on the basis of this fact comparisons may be made. It should be added that three of the manufacturers were represented by just one system each.

The ranking order of the total of parameter average values in Table II-A-V agrees better with intelligibility than speaker identification or naturalness. Moreover the correlation is greater between intelligibility and speaker identification or naturalness, than the correlation between speaker identification and naturalness. Consequently if we want only one of these parameters to specify the quality of a speech transmission channel intelligibility gives the most significant result.

If we compare the ranking order of the sets of the different intercom manufacturers with consideration to the intelligibility as well as the voice switching problems we will notice a relation between good intelligibility and poor voice switching properties and vice versa. The explanation of this can be seen in Tables II-A-II and II-A-IV indicating loudness and timbre. If the loudness is weak and the balance of spectrum chosen so that more treble than bass is transmitted the operation of the voice-switched amplifier will be perfect but the intelligibility will decrease. On the other hand, a high output sound level and reproduction of a wide frequency range results in good intelligibility or naturalness, but due to the acoustic coupling loudspeaker - microphone and due to room reverberation this will result in frequent false-operations of the voice-switched amplifier.

Room reverberation

Method

Room acoustics are a link in the chain: Speaker - transmission system - listener (Huszty 1971). A sound source, e.g. the loudspeaker,
generates a fluctuating sound pattern in the room. We wish to know how the listener perceives the sound from the loudspeaker and also how the microphone is affected by the sound of the speaking party when room reverberation is taken into account.

The reverberation of the room of the speaking party affects the speech intelligibility of the transmitted sound (Lochner and Burger 1961), while reverberation sometimes increases the intelligibility for a listener in the same room (Krookstad 1963). The influence of the room reverberation on the transmitted sound is greater when the distance between the mouth of the speaking party and the microphone increases (Gardner 1960). One reason for using a loudspeaking telephone set is that the user can speak from another part of the room and not necessarily within an arm's length of the set, and the frequency of usage of this situation was noted.

In rooms with minimum dimensions larger than the wave lengths of the lowest frequencies, the reverberation time is the most interesting parameter. A suitable value exists for each particular room and depends on the use of the room. If the reverberation time is too long, the intelligibility will decrease in the room. If the room is well-absorbing the voice will sound unnatural and an increased speech level is required because sound reflecting materials are missing. It therefore takes more effort to speak in a well-absorbing room than in one with longer reverberation time if the noise levels are the same.

In rooms where the wave lengths of the lowfrequency sound are of the same dimensions as the room, the sound pattern cannot be regarded as diffuse. A standing wave may develop between two parallel walls since the incoming sound interferes with the reflected sound. In a room with reflecting surfaces standing waves and, thus, eigentones develop. Because of this sound pressure level in the room will vary when measured from different positions in the room (Bolt and Roop 1950; Doak 1959; Schroeder and Kuttruff 1962).

A main problem in loudspeaking telephony is the influence of room reverberation or echo on the speech at the speaking party's end. The reverberation results in a mixture of direct and reflected sound which is picked up by the microphone and transmitted. If the reflected-to-direct-sound ratio is too high the listening party perceives the speaking party as talking in a bathroom or a barrel (Clemency, Romanow and Rose 1957;
Gardner 1960; Huggler 1961; Clemency and Goodale 1961; Copping and Fidler 1967; Galyas 1969; Reichard and Breeden 1973; Suntop 1974; Berkley and Mitchell 1974). If the echo effect is far worse false switching phenomena appear in the voice switched amplifier (Busala 1960) and these are reported in another part of this study.

An interesting distance is the "reverberation radius" \( r_r \) (Krokstad 1967) which is the distance to the source where the direct sound is equal to the reflected sound. This occurs when the two terms in the log-expression in equation (3), p. 30, are equal.

\[
\frac{1}{4\pi r_r^2} = \frac{4}{R}
\]  

\[
r_r = \frac{1}{4} \sqrt{\frac{S \cdot \alpha}{\pi (1-\alpha)}}
\]

Obviously we have two types of sound fields in the room, the "near field" or the direct sound field between the sound source and up to a distance of the reverberation radius from the source where the distant law is valid, and the "far field" or the reverberant sound field further from the sound source than a distance of the reverberation radius where the reverberant sound dominates. In our application either the sound source is the mouth of the speaking party and we study the distance to the microphone of the set or the sound source is the loudspeaker and we study the distance to the ears of the listening party.

To specify the room acoustics the dimensions (height, width, length) of the room were measured. The rooms were divided into three levels of absorption: well-absorbing (many of absorbers e.g. wall-to-wall carpet, upholstered furniture, curtains, book shelves, acoustic tiles), normal (some of the sound absorbing material mentioned above), and low-absorbing (without good absorbing materials). Since measurements of the reverberation times in all rooms would have increased the extent of the study considerably, more detailed measurements were carried out in ten rooms only.

**Results**

The rooms in the study had an average volume of 68 m\(^3\) (median value 54 m\(^3\)) and the average value of the total room area was 100 m\(^2\) (median value 89 m\(^2\)). The rooms were divided into three absorbing levels and
from the special study of the reverberation times in ten rooms the 
average absorption coefficients were calculated to 0.30 for well absorbing 
rooms, 0.16 for normal and 0.12 for low-absorbing rooms. Using 
these coefficients the average reverberation time for all rooms was cal-
culated to be 0.54 seconds with a standard deviation of 0.20 seconds.

The "reverberation radius" is calculated from the total room area 
and the average absorption coefficient. For well absorbing rooms it is 
found to be 0.76 m, normal rooms 0.55 m and low-absorbing rooms 
0.47 m.

The ratio between the talking distance and the reverberation radius for 
an omnidirectional sound source average 1.16 (standard deviation 0.47) 
and the distribution for all users is plotted in Fig. II-A-5. This ratio we 
compare to the frequency of complaints from the far end party about the 
speech quality and recommendations to use the handset instead of the 
loudspeaking telephone. The result is plotted for the 54 loudspeaking 
telephone users in Fig. II-A-6.

Discussion

The ratio between the talking distance and the reverberation radius is 
approximately equal to one. Thus one half of the sound energy produced 
by the speaking party will reach the microphone directly while the other 
half reaches the microphone via room reflections. For the actual ratio 
1.16 the amount of reverberant sound dominate over the direct sound at 
the microphone when transmitting. Reverse conditions may also apply in-
asmuch as approximately one half of the energy radiated by the loudspeaker 
reaches the listener directly while the other half reaches him via room 
reflections. However, the loudspeaker radiates into a halfspace and 
should not be regarded as an omnidirectional sound source. The rever-
beration radius for such a source is $\sqrt{2}$ times the calculated value for an 
omnidirectional source. If the listening distance is equal to the talking 
distance the ratio between this and the reverberation radius for the loud-
speaker average 0.82. Consequently the amount of direct sound dominate 
over the reverberant sound at the position of the listening party.

The ratio between the talking distance and the reverberation radius 
gives us a score for the quality of the transmitted sound from the micro-
phone to the listening party due to room reverberation. When using a
Fig. II-A-5. Distribution of the ratio between talking distance and reverberation radius for loudspeaking telephone system users.
Fig. II-A-6. The ratio between talking distance and reverberation radius for four groups of complaint frequencies from the far end, when using the loudspeaking telephone. The vertical lines mark the standard error of the mean.
loudspeaking telephone the listening party generally uses a handset telephone. In this case a great amount of transmitted reverberant sound is very disturbing and gives the mentioned "barrel" or "bath-room" effect. This disadvantage is the main reason why the handset user may recommend the loudspeaking telephone user to use his handset if possible. Other reasons are frequent false-switching of the voice-switched amplifier and sound level, which is too low.

Fig. II-A-6 shows that there is a correlation between these two parameters mentioned for the 54 loudspeaking telephone users. The standard error of the mean is also plotted and in group 4 the data is based on only six users, which may explain the fall of the rising curve.

In this study the speaker and the telephone set are regarded as omnidirectional sources to simplify the model, but in the reality the speech level varies about the human head (Dunn and Farnsworth 1939) and the desk surface influences on the sound field about the loudspeaking telephone set. The total reflecting area of the room was calculated as the sum of the walls, the ceiling and the floor areas and the rooms considered as rectangular. Of course there were other reflecting areas in the rooms and not all the rooms had rectangular forms, but these facts do not seriously influence the data in this study.

The special study on reverberation time included only ten rooms, which were classed in the same way as the rooms in the study, and on the bases of these the average absorption coefficients were calculated and used on all the rooms in the study, and consequently there is some unreliability in the data. However, the average absorption coefficients agree well with those in the literature (Beranek 1971).

The lowest frequency transmitted over a loudspeaking telephone system is about 300 Hz and this corresponds to a wavelength of about one meter. Some rooms in the study have a minimum dimension of about two meters and accordingly the standing wave model with eigentones is applicable for the lowest transmitted frequencies. On the other hand, the highest transmitted frequencies are in the region 5-6 kHz for intercom systems and the corresponding wavelength: 6 or 7 cm. In this case the reverberation model is applicable. The absorbing materials in the studied rooms often had higher absorption coefficients for frequencies over
500 Hz than under 500 Hz. This fact elucidates that the dominating reverberation sound in the room is of frequencies under 500 Hz and can be described by the eigentones.

**Room noise**

**Methods**

Acoustical noise affects the sound transmission in different ways. The intelligibility of speech is reduced by masking, when noise is present in the room of the speaking party (Webster 1965). The incidence of such excessive noise levels causing the listening party difficulty in understanding the speaking party was noted. The noise also causes the speaking party to raise his speech level about 0.3-0.4 dB per dB increase of noise level (Gardner 1966).

Acoustical noise at the listening party both masks the sound from the loudspeaker, and results in false switching if it exceeds a fixed limit because the automatic voice-switching circuitry may react to the noise signal as if it were the listening party who began speaking. If noise is present when listening, a higher level is required to obtain an acceptable signal to noise ratio. Different listening levels are shown in Fig. II-A-7 where the effective signal consists of speech and the interfering noise of A-weighted noise (Gardner 1964; Gleiss 1971). The plot number 1 represents suitable or ideal listening level.

Apparently acoustical noise is an important factor affecting loudspeaking telephone conversation. The noise levels were measured with a sound level meter (Brüel and Kjær 2203) with linear scale as well as with A-weighting. Moreover the types of the noise sources were noted.

**Results**

The average level of the environmental noise was 45.4 dB (standard deviation 7.8 dB), weighted in accordance with a standardized A-curve corresponding to the noise level "heard by the ear". Without weighting an average value of 64.1 dB (standard deviation 6.2 dB) was obtained, i.e. the noise level that in reality affects the microphone. The difference between the weighted and the non-weighted values (average 18.7 ± 6.0 dB) depends on the existence of lowfrequency noise below 300 Hz. The distribution of the noise levels is shown in Fig. II-A-8.
1. Suitable or ideal listening level
2. Minimum listening level with acceptable intelligibility
3. The listening level at which one hardly hears the speech
4. Maximum acceptable listening level with the loudspeakers on the chair behind the listener
5. Maximum acceptable listening level when the loudspeakers are mounted in the ceiling above the listener.

Fig. II-A-7. Effect of noise on requested loudspeaker output level for speech.
Fig. II-A-8. Distribution of the noise levels in the studied rooms.
Table II-A-VIII shows the different noise sources and their appearance in the studied rooms. Note that in many rooms more than one source was noted.

<table>
<thead>
<tr>
<th>Source</th>
<th>%</th>
</tr>
</thead>
<tbody>
<tr>
<td>Ventilation systems</td>
<td>47</td>
</tr>
<tr>
<td>Traffic noise</td>
<td>35</td>
</tr>
<tr>
<td>Office machines</td>
<td>25</td>
</tr>
<tr>
<td>Hallway speech and impact sound</td>
<td>24</td>
</tr>
<tr>
<td>Outdoors sounds</td>
<td>13</td>
</tr>
<tr>
<td>Work shops</td>
<td>12</td>
</tr>
<tr>
<td>Radios</td>
<td>4</td>
</tr>
</tbody>
</table>

Table II-A-VIII. Relative observation of noise sources in rooms with loudspeaking telephone systems.

The frequency of speech masking problems due to room noise is presented in Table II-A-IX.

<table>
<thead>
<tr>
<th>Frequency</th>
<th>%</th>
</tr>
</thead>
<tbody>
<tr>
<td>Never</td>
<td>17</td>
</tr>
<tr>
<td>Seldom</td>
<td>36</td>
</tr>
<tr>
<td>Sometimes</td>
<td>44</td>
</tr>
<tr>
<td>Often</td>
<td>12</td>
</tr>
<tr>
<td>Always</td>
<td>1</td>
</tr>
</tbody>
</table>

100 %

Table II-A-IX. The relative frequency of speech masking due to room noise.

Discussion

According to the data in Table II-A-IX room noise is a disturbing factor in many cases, and in 7% of the rooms the level exceeded 60 dB(A). The amount of lowfrequency noise is apparently great and the noise spectra should be measured in octave or third octave bands in a more advanced study. The sound absorption of the lower frequencies in ordinary offices are rather low due to the absorbing materials (carpet, tiles, etc) normally subdue higher frequencies (> 500 Hz) more than lower frequencies. Noise sources also produce more lowfrequency noise than highfrequency noise in many cases and this explains the great amount of lowfrequency noise and the difference between the sound pressure values with linear scale and with A-weighting.

The most frequent noise sources were the ventilation systems. Even though their sound levels were relatively low, and do not affect conversation nor voice-switching they were the major sources in quiet office environments. This category also included noise from the heating and plumbing system.
The traffic noise was the next dominating noise source in the offices. The measurements were carried out during the summer and the windows were opened in many of the rooms. The heavy traffic noise (from buses and lorries) often reached levels making even an ordinary conversation in the room impossible.

Office machines, especially typewriters in the same and adjoining rooms, were noted as serious noise sources and affected the voice switching in many cases. Apart from typewriters, calculating machines, book-keeping machines and telex machines are counted among these groups as well as the sound from telephone exchanges and pneumatic dispatch systems. Noise from hallways and passages mostly consisted of speech from the hallway or adjoining rooms and impact sound from footsteps. Due to easier communication through personal visits, many employees leave their doors open which results in a higher noise level.

Special outdoor sounds consisted of noise from construction sites, loading and unloading, lawn-mowers and children playing. Workshop machines and various noise from the manufacturing part of the company caused serious problems due to false-switching. This group also includes certain office machines, e.g. offset printing machines and computers. In many locations the work is of a character that allows people to listen to their transistor radios. The radio was noted to be a remarkable source of noise, and it was often placed close to the intercom set. Correct voice switching was impossible in these cases.

**Human factors**

**Methods**

Human factors is here referred to as such properties as frequency of calling, conversation times, training and special viewpoints concerning loudspeaking telephone systems. To have the possibility, when evaluating the data, to weight these with respect to the frequency of usage, the call frequencies and conversation times were noted by the user from a subjective point of view. A negative attitude towards a system of this type is associated with a low degree of utilization and a resort to other ways of communication. It is of great interest to know in which ways people prefer to communicate in a company. For two different situations, a short message and a work discussion, the user was asked to give the frequency of usage of intercom, loudspeaking telephone, standard handset telephone and personal visits.
We know from a British study (Maddison 1968) that there are people who find it difficult to dial a call on the public telephone network. But how well are people trained on more complex systems such as a loudspeaking telephone set or an intercom? And how is the training administered? These are important factors to consider.

The users had opportunities to give their viewpoints on the advantages and the disadvantages of a loudspeaking telephone or an intercom versus the standard handset telephone. The users also gave viewpoints on the limitations of the systems of today, which they did not want in future systems and these factors should be considered when new systems are developed.

Results

The conversational frequency was relatively evenly spread among the users from a few to ten calls a day. The loudspeaking and handset telephones were used on an average for more than ten calls a day while the corresponding figure for intercom was approximately six. Subjectively estimated conversation times are shown in Table II-A-X.

<table>
<thead>
<tr>
<th>Conversation time (s)</th>
<th>0-15</th>
<th>15-30</th>
<th>30-60</th>
<th>60-180</th>
<th>&gt;180</th>
<th>Number of replies</th>
</tr>
</thead>
<tbody>
<tr>
<td>Intercom</td>
<td>22</td>
<td>45</td>
<td>26</td>
<td>7</td>
<td>0</td>
<td>204</td>
</tr>
<tr>
<td>Internal calls (in PABX)</td>
<td>3</td>
<td>12</td>
<td>23</td>
<td>49</td>
<td>13</td>
<td>213</td>
</tr>
<tr>
<td>External calls (outside PABX)</td>
<td>0</td>
<td>1</td>
<td>6</td>
<td>36</td>
<td>58</td>
<td>242</td>
</tr>
</tbody>
</table>

Table II-A-X. Relative spread (%) of subjectively estimated conversation times.

The study shows that the intercom is used mainly for short calls, brief inquiries and short messages and is avoided for longer discussions. Personal visits dominate the statistics concerning longer messages.

As loudspeaking telephones and intercom systems are more difficult than ordinary handset telephones to operate, some training should be necessary. However, 7% of all the users had no training or earlier experience of the telephone set. 10% had earlier experience from another intercom or loudspeaking telephone. The most common ways to administer training are by the directions for use (58%) or from a colleague (21%). Some people were trained at special courses (4%).
The advantages of an intercom compared to a handset telephone are:

1. Quick connecting procedure (push-buttons)
2. Possible to make internal inquiries when having a call on the external line
3. Possibility of several persons in the same room speaking and listening to the intercom system
4. Free hands for making notes, sorting out papers and handling binders.

In many cases a handset telephone (PABX) is preferred to an intercom for internal calls. Some of the viewpoints received are:

1. More persons in a company can be reached through the standard telephone (PABX-exchange). The intercom system does not generally include as many employees as the standard telephone system.
2. In a company with a small intercom system the intercom telephone directory is often non-current because it has not been followed-up closely enough. The PABX telephone directory is generally more up to date. The choice of means of communication often depends on whether the PABX telephone number of the intercom extension number is known by heart.
3. The called person may have a visitor. One does not want to disturb. This applies especially when one calls a superior in a hierarchically organized company.
4. The called person may be busy on the phone and an intercom call is then regarded as disturbing. When calling on the PABX telephone, a busy signal will inform that the called party is busy.
5. If the called person is having a conference in his room, all participants will be disturbed by an intercom call.
6. The exchange of information may be of confidential nature. One does not want unauthorized people to overhear the conversation.
7. Some people consider a call via the handset telephone more personal than a call via the intercom. The intercom system is seldom used when making a first contact but only to call people previously known by the caller.
8. Due to room noise, it is easier to speak and listen when using a telephone receiver.
9. Due to the automatic voice switching one does not know how one sounds in the other set. No confirmation is received as to what has passed through the voice-switched channel and one does not know if the speech is too weak or too loud. In a handset the hybrid sidetone provides for a certain reference.

During the study three important cases were brought up where the loudspeaking telephone was especially appreciated.
(1) Several persons can participate in a conference and a person who cannot be present can still follow the conference via telephone. This is one of the most common arguments for the use of a loud-speaking telephone (Clemency, Romanow and Rose 1957; Cleary and Cannon 1961). The drawback is that the microphone sensitivity is so low that the set will have to be placed in front of the speaking party and that the loudspeaker output is often not sufficient enough for a big audience.

(2) When the called person is busy and one wishes to wait until he is free, the loudspeaking telephone can be used for this purpose. Meanwhile other matters may be attended to. When the called party answers, one reverts from loudspeaking to handset telephone.

(3) One can make notes, look in binders and files and still be able to carry on a conversation via the telephone.

A company is often organized so that people in frequent contact are placed close to each other. It is apparently often easier to speak directly to each other rather than to use a telephone.

The interviewed persons submitted their ideas of how the intercom and loudspeaking systems could be improved:

A. Loudspeaking telephone

(1) Better instructions concerning the function and use of the telephone set. Advice as to measures required to obtain the best possible speech connection in prevailing circumstances.

(2) The loudspeaking telephone set should have a handset for non-loud-speaking conversation.

(3) A condition for installation of a loudspeaking telephone should be that the sound absorption of the room is sufficient.

(4) Stepless volume control on each set.

(5) Higher output level especially in conference calls.

(6) Difficult to reverse the speech channel from receiving to transmitting mode when the distance from the mouth to the microphone is long - as in conferences - automatic adjustment of the microphone sensitivity is desirable.

(7) Possibility to connect an external microphone.

(8) Increased frequency range of the speech channel.

(9) Decreased noise sensitivity - especially as regards typewriter and traffic noises.

B. Intercom systems

(1) Stepless volume control is required in each intercom set for adjustment of the loudspeaker output, because the speaking party does not hear his own voice in the set (no "sidetone" as in a telephone with a handset) and consequently he does not know how loud his voice sounds to the listening party (and it is often too loud).
(2) Consideration must be taken to other persons working in the same room - especially during telephone conversation - by adjustment of the sound level.

(3) The microphone should be mounted at such an angle that several persons in a room can use the same intercom set.

(4) Some types of intercoms have such a low microphone sensitivity that conversation is impossible beyond a point some two meters from the microphone.

(5) The influence of the voice switching should be decreased and this was desired to enable both parties to speak at the same time.

(6) "Traffic discipline" should not be a condition for conversation.

(7) Speech communication may become a problem when a handset without sidetone is connected to a voiced switched loudspeaking telephone.

(8) The intercom telephone systems are too sensitive to noise from typewriters and calculating machines. The windows are often opened during summer months and traffic noise prevents a normal conversation with automatic voice switching.

(9) A separate instruction for use should be available when the instrument is used on noisy premises.

(10) The intercom set should be equipped with a receiver and for the future, better intelligibility, lower distortion, increased naturalness and less "plastic sound".

**Discussion**

The frequency of conversation was evenly spread among the participants of this study which contains a representative set of different types of users. A negative attitude towards loudspeaking systems causes a preference for other means of communication.

We have used the same weight for all users on all data, but we could have used the possibility to weight our data so that people who use the loudspeaking telephone or intercom frequently and know the systems and their limitations very well are taken more into account than the "low-frequency" group. This might have given more relevant results concerning e.g. sound quality and the performance of the automatic voice switching. On the other hand the same weight should be used on data concerning e.g. training and talking distances.

The conversation times depend on the type of information which is transmitted and also the position in the company where the intercom set is installed (sales office, workshop, storehouse). These subjectively evaluated conversation times on handset/handsfree telephones (inside
and outside the PABX-exchange) agree with those measured objectively (R. Evers, Germany).

The figures on training show that some people are badly trained on the handling of the set. If they had enough knowledge to set up a call, they can use the loudspeaking telephone or intercom set, but they do not know how to handle the set in highly reverberant and noisy environments or the most suitable speech level and talking distance. The study shows that most of the users get their information on handling the set by the directions for use, delivered with the telephone set. But this will soon disappear if not glued to the set. A new employee does not then have the same possibility to learn the various functions and the use of the telephone set without first being informed by a colleague. In this instance there is a need to standardize the keys and the setting-up procedure in intercom systems as has been done in standard telephone systems.

In spite of the advantages which the loudspeaking system gives, such as handsfree operation and that several people can take part in a call, there are disadvantages which account for a preference of using a handset telephone in many cases. The choice of the handset is governed by the following considerations:

- The accessibility of the telephone system, directory etc.
- The ease of telephone communication in reverberant or noisy environments versus the false-operation of the automatic voice switching.
- Private and psychological factors such as disturbing a busy superior with an intercom call, confidential calls etc.

Possible means of improvements in future loudspeaking telephone and intercom systems fall into two main categories. The first one concerns the sending and receiving sensitivity, e.g. the output level from the loudspeaker should be manually adjustable or an automatic adjustment of the microphone sensitivity could be incorporated. The second one is that the influence of noise and reverberation on the speech transmission should be reduced in future systems.

Conclusion

This study elucidates the advantages and the disadvantages of hands-free telephone systems. The possibility to get in touch with somebody quickly by an intercom set or to reach a group of participants around a
loudspeaking telephone account for the appreciation loudspeaking systems have received in public. Still there remain limitations under conditions of unfavorable room acoustics. Moreover individual spread in users' speech and hearing characteristics cause deviations from optimal system operation.

The average speech level of talkers varies within a range of 15 dB(A), from extreme low to extreme high voices. Generally the voice level is raised when using a loudspeaking telephone system. The range is about 10 dB(A) with an average of 7 dB(A). This increase is an important factor to consider. Also the preferred listening level may vary with the particular hearing of the listening party and his acoustic environment. Consequently many people want a stepless volume control on the telephone sets or an automatic gain control in the system.

The problems connected with room acoustics are of two categories, room reverberation and environmental noise. The influence of reverberation sound transmitted from the talking party's end to the listening party's end can be scored by the ratio between the talking distance and the reverberation radius. The reverberation radius depends on the total absorption in the room and can be calculated when the volume and the reverberation time of the room are known. The most frequently encountered noise is sound from vehicles and from typewriters. The noise both masks the transmitted speech and may cause false-switching in the voice-switched amplifiers.

The loudspeaking systems of today are developed to provide the best quality for average people conversing with average levels in average rooms where the noise levels are sufficiently low and reverberation times short enough. In future systems these limits on conversation will be adjusted to satisfy a wider group of people in a wider group of environments, by more advanced techniques as well as by better training of the users to understand the handling and the function of the systems.

References:


Doak, P. E. (1959): "Fluctuations of the sound pressure level in rooms when the receiver positions is varied", Acustica 2, No. 1, p. 1.


