How to assess and improve the quality of voice services in telephone communication and alarm systems in mines

The article presents selected objective and subjective methods to assess the quality of voice services in telephone communication. The authors described the impact of the line and acoustic echo phenomena and ambient noise on the functions of telephone communication systems with respect to the speakerphone mode. Additionally, they discussed the possibilities to apply digital technology to improve the quality of voice services in telephone communication and alarm systems.

Keywords: telephone communication, alarm communication, safety, telecommunication, voice transmission, noise reduction

1. INTRODUCTION

Telephone communication and alarm systems in mines are key elements for safe functioning of modern deep mines [15]. They enable to transfer commands, warnings (including alarms) or reports. In order to function properly, these systems should have proper quality of voice transmission in the conditions of underground mines.

The objective of this article is to present the issue how to assess the quality of voice services and the phenomena which impact this quality.

2. VOICE SERVICES QUALITY ASSESSMENT

The assessment of phenomena which we sense (e.g. by hearing or seeing them) is a very complex process. This refers to the assessment of telephone conversation quality too. In order to assess the quality it is not enough to make simple measurements of certain physical quantities, such as attenuation of the connection or frequency characteristics of this attenuation. There are a number of methods to assess the quality of a telephone conversation. The methods can be divided into [12]:

- subjective methods based on listening to the conversation in defined conditions and subjective assessment of the quality of the conversation or the conversation fragments that one can hear,
- objective methods based on the registration of the conversation fragments and their advanced analysis which gives a suitable result of the assessment.

Assessment methods are often used to check the impact of some telephone parameters or telephone circuit parameters on the conversation quality.

The following methods are subjective:

- logatom articulation [17] – listening to logatoms (phonetic elements without any meaning in the native language of people who take part in the measurements) read by a reader; the result is the ratio of the number of properly heard logatoms to the total number of logatoms in the text,
- semantically unpredictable sentences\(^1\) [11] – listening to semantically unpredictable sentences read by the reader; the result is the percentage of properly received sentences or words, depending on the adopted method,

\(^1\) artificially generated sentences which, though consisting of correct words, do not have any logical sense, therefore they are semantically unpredictable (a word which is not understood by the listener cannot be deducted from the sense of the whole sentence). The sentence has proper syntax, words are used in accordance with grammatical rules but the semantics of the sentence is completely disturbed.
- MOS (Mean Opinion Score) [4, 7] referring to:
  - ACR (Absolute Category Rating). MOS for the ACR method is determined as the average value of the assessment of all participants (in the scale from 1 to 5) in 3 categories: absolute rating, listening effort, volume,
  - DCR (Degradation Category Rating). MOS for the DCR method is determined as the average value of the assessment of all participants (in the scale from 1 to 5),
  - CCR (Comparison Category Rating). MOS for the CCR method is determined as the average value of the assessment of all participants (in the scale from -3 to +3).

The following methods are objective:
- comparison methods, such as:
  - PSQM [9] (Psycho-Acoustic Speech Quality Measure), which is based on comparing the input signal (artificial speech acc. to [P.4]) and the output signal after complex transformations. The signal comparison result is presented in the MOS scale,
  - PAMS (Perceptual Analysis Measurement System), which is based on comparing the input signal and the output signal after complex transformations with the use of the so called audibility transform. The signal comparison result is presented in the MOS scale,
  - PESQ [10] (Perceptual Evaluation of Speech Quality), which can be considered an extension of PAMS,
- INMD [5] (In-service Non-intrusive Measurements Device),
- Method according to the P.563 recommendation[6],
- E-model,
- Method for the Assessment of Voice Transmission Quality [1].

3. PARAMETERS WHICH IMPACT THE QUALITY OF VOICE SERVICES

The impact of different parameters on the quality of voice services can be investigated in the situations of listening, speaking, conversation, and ambient noise influence [2].

The quality of a voice service in the situation of listening is affected by the following parameters:
- volume (with respect to a conversation between two persons standing within 1-metre distance from each other),
- sound quality depending on the parameters of a transmission system, such as band width, frequency characteristics, signal-noise ratio, distortions,
- understandability which is particularly important in the presence of noises.

The quality of a voice service in the situation of speaking is affected by the following parameters:
- delay (particularly in VoIP systems) – bigger audibility of the echo,
- both persons speaking simultaneously – switching on echo cancellers, amplification regulations, call signal masking by the echo.

Ambient (background) noise impacts the transmission quality in different phases of the conversation, such as: pause, speaking, listening. A speech signal can be distorted by noise reduction systems while the reception of the signal can be disturbed by ambient noise. The speech signal quality in the presence of noise becomes one of the most essential parameters of the voice service.

4. ECHO PHENOMENON IN TELEPHONE COMMUNICATION SYSTEMS

Voice services in telephone communication systems are characterized by simultaneous two-way signal transmission. Some elements of the telecommunications network can transmit signals in both directions at the same time, which is called a two-wire system (e.g. telecommunications cables). Some elements, in turn, such as amplifiers or switching networks of digital telephone exchanges, are one-way elements and require two separate ways (channels), one for each direction. This is called a four-wire system. The connection of two- and four-wire systems requires that a splitter should be used. A splitter-type system is an element with 4 ports (Fig. 1). In the balanced state the balancer ensures signal transmission between neighbouring connections (e.g. \( a \rightarrow c, a \rightarrow d, b \rightarrow c, b \rightarrow d \)), while signal transmission between opposite connections (e.g. \( a \rightarrow b, c \rightarrow d \)) is not possible. In the case featured in Fig. 1 it is possible to get the balance if a proper dependency between \( Z_a \) and \( Z_b \) impedances is ensured. Then the signal from port \( d \) will not be transmitted to port \( c \). In the one-line scheme in Fig. 1b, we showed the application of a splitter which is to send the signal from port \( d \) to
port $a$, as well as from $b$ with added $Z_{\text{BAL}}$ impedance. When the balancer is balanced, i.e. when the ratio of $Z_L$ input impedance of the circuit connected to port $a$ and $Z_{\text{BAL}}$ impedance is $Z_L/Z_{\text{BAL}}=k$ ($k$ value depends on the structure of the splitter), the signal from port $d$ will not be transmitted to port $c$.

Splitters are applied, for example, in anti-local systems in telephones, SLIC systems in digital telephone exchanges and TBI 2 intrinsically safe barriers [14], and ZSD intrinsically safe barriers [16].

Figure 2 presents the operation of a splitter applied in the JANTAR 2 signalling telephone built with the use of an AS2522 system.

![Fig. 1. Splitter – multi-line symbol (a), one-line symbol (b)](image1)

![Fig. 2. Splitter operation in a telephone with AS2522 system](image2)

A signal from the microphone is sent to the bridge connection. In one arm of the bridge there is a 30Ω resistor and $Z_L$ input impedance of the subscriber circuit. In the other arm of the bridge there is a 300Ω resistor and $Z_{\text{BAL}}$ balancing impedance (circuit balancer). The splitter is balance if the $10Z_L=Z_{\text{BAL}}$ condition is fulfilled in the whole frequency range of the splitter operations.

Figure 3 presents a sample telephone chain in the system of telephone communication for methane mines. The chain consists of two telephones (working in the loudspeaker mode), two intrinsically safe barriers and a digital telephone exchange. This configuration has two kinds of echo phenomena:

- acoustic echo evoked by acoustic coupling between the speaker (receiver) and the microphone in the remote phone,
- line echo caused by incomplete balance of splitters in the intrinsically safe barriers and digital telephone exchange,
- local effect caused by incomplete balance of the splitter in the local phone.
Fig. 3. Sample telephone chain with splitters. PK – switching network, ADC – analogue-digital transformer, DAC – digital-analogue transformer

Sometimes it is possible to have echo evoked by mechanical coupling between the loudspeaker (receiver) and microphone in the phone. In the configuration from Fig. 3 the delays are small (50μs for the 10-km telecommunication cable, a few ms for the digital exchange) and the echo is recognized as a copy of what the user speaks to the microphone. A small delay can happen in the case of the acoustic echo evoked by reverberations from the walls of the room where the remote telephone is placed.

When the telephone works in the speakerphone mode, the balance of the splitter system in the telephone may produce vibrations in this telephone (whistle). This effect is the result of positive feedback in the loop which comprises: the microphone, microphone amplifier, improper path of the splitter system, loudspeaker amplifier, loudspeaker, acoustic coupling of the loudspeaker and the microphone – shown in Fig. 4. This phenomenon can be eliminated by proper enhancement regulation in the microphone and loudspeaker circuit. Such a function is performed by specialized integrated circuits (e.g. AS2522B in the JANTAR 2 signalling telephone [14, 19]).

Fig. 4. Evoking vibrations in a loudspeaker telephone

The attenuation of the splitter on an improper path depends on how the frequency characteristics of the $Z_{BAL}$ impedance of the artificial balancing line are adapted to the frequency characteristics of the $Z_L$ impedance of the cable circuit which is closed by the input impedance of a successive telephone chain element (e.g. intrinsically safe barrier). Most frequently, the $Z_{BAL}$ impedance is adapted, approximately, to the catalogue frequency characteristics of the $Z_f$ wave impedance of the cable circuit. In real installations the input impedance of the cable circuit differs from the wave impedance due to loading the circuit with the impedance different from a wave impedance. This situation causes incompatibility of the splitter system. In order to reduce this effect, the JANTAR 2 signalling telephone has a possibility to control (remotely from the telecommunications server) the $Z_{BAL}$ impedance depending on the length of the cable circuit [14].

The mechanism of echo evoking is slightly different in the case when the VoIP technology is used to provide voice services.

Figure 5 features a sample telephone chain including a VoIP telephone, telephone exchange with a VoIP card and an analogue telephone with an intrinsically safe barrier. In the VoIP telephone and on the VoIP card of the exchange there are codecs with framing (bundling) circuits and buffers for jitter compensation which cause relatively big delays (e.g. 30 ms for framing). In such solutions the users can hear their own echo audibly.
5. IMPACT OF NOISE ON THE QUALITY OF VOICE SERVICES IN TELEPHONE COMMUNICATION SYSTEMS

In industrial conditions (including mines) telephones and alarm signalling devices can be installed in places with high noise levels caused by working machines. Ambient noise has a large impact on the quality of voice services.

Figure 6 shows the impact of noise on the listener’s side on the telephone conversation. The listener hears the speech signal from the loudspeaker (receiver) and the ambient noise. The quality of the telephone conversation is affected by the difference between the level of the desired signal (of the telephone conversation) and the noise level. In addition, the listener’s microphone receives the noise which is transmitted to and heard by the speaker. This also impacts the quality of the voice service.

Figure 7 shows the impact of noise on the speaker’s side on the telephone conversation. The speaker’s microphone receives the speech signal emitted by the speaker and the noise signal. The combination of these two signals is transmitted to the listener’s telephone. At the same time, the noise impacts the speaker’s behaviour and evokes the so called Lombard effect which is the speakers’ tendency to increase their vocal effort, as well as pitch, rate, and duration of syllables [13].
The following methods are used to reduce the impact of ambient noise on the quality of voice services:
− using a handset instead of a loudspeaker wherever possible,
− using a handset and an extra receiver which significantly improves the conditions of listening (the headphones significantly attenuate the noise heard by the user) – see Fig. 8,
− using an extra microphone to compensate the noise or using a differential microphone while ensuring a short distance between the sound source (mouth) and the microphone – see Fig. 9.

In the differential microphone the noise is emitted from the source from a relatively long distance and impacts the microphone membrane in opposite directions, while the signal from the close source (mouth) impacts the membrane only from one direction (such a solution was applied in JANTAR 2 [14]),
− using advance methods of digital processing of voice signals (section 6).

6. USE OF DIGITAL PROCESSING OF SPEECH SIGNAL TO REDUCE ECHO AND NOISE

The basic method to reduce the echo phenomenon is:
− good balancing of splitters for line echo,
− proper location of the microphone and speaker in a loudspeaker phone.

The application of digital technology in telephone communication systems allows further reduction of echo by introducing digital adaptation filters. Figure 10 features a simplified block diagram of a telephone part with an acoustic echo canceller. In the microphone circuit there is a system which deducts the signal coming from the microphone and the signal from the loudspeaker circuit passed through the adaptation filter. The filter is adapted in such a way that the difference between the signal evoked by the acoustic coupling of the loudspeaker and microphone and the output signal of the filter is close to zero. This way the person speaking to the microphone of the remote telephone (not shown in Fig. 10) will not hear the echo evoked by the acoustic coupling.
Digital filters (digital echo cancellers) can be used in analogue devices too. Figure 11 presents a simplified block diagram of a telephone with an integrated circuit CS6422 [20] which includes an acoustic echo canceller and a line echo canceller. The line echo canceller has an adaptation filter which is tuned to reduce, as close to zero as possible, the microphone signal which gets through the improper path of the splitter system (d-c) to the loudspeaker circuit. The acoustic echo canceller works similarly to the situation in Fig. 10. The CS6422 circuit comprises indispensable analogue-digital and digital-analogue transformers.

The use of digital technology allows to improve the quality of voice services in the conditions of noise. If the noise is present near the speaker's telephone (Fig. 12), it is possible to apply a digital filter which reduces the noise by the deduction of noise spectrum from the voice signal spectrum along with noise. The method assumes that the noise spectrum, measured in the pauses of the speech signal, is subject to small changes in short time periods. In addition, it is possible to use an extra microphone (microphones) to measure the noise. If there is a noise near the listener's telephone, it is possible to use the Near End Listening Enhancement (NELE) method [18]. NELE is based on the measurement of the noise spectrum. Then the speech signal spectrum is modified (level is increased) so that to achieve an indispensable difference between the desired signal and the noise (Fig. 13). In the presence of noise the signal from the loudspeaker has a higher level.
7. CONCLUSIONS

Voice services in mines are provided by a number of telecommunications systems, such as telephone alarm and communication systems, VoIP telephone communication systems (stationary and mobile), systems of communication with mobile terminals with the use of different radio communication protocols. The terminals of these systems work in different environments (different acoustic conditions). Currently, it is difficult to assess the quality of voice services in real working conditions of these systems. It is necessary to adapt existing methods for voice services quality assessment to the needs of mining communication systems, with respect to acoustic conditions and structure of these systems (e.g. the presence of intrinsically safe barriers).

Echo and noise evoked by devices working near the telephone are phenomena that impact the quality of voice services. The authors of the paper presented analogue-technique methods employed to reduce the impact of these phenomena on the quality of voice services. Additionally, the possibilities of digital techniques were presented for echo cancellation and noise reduction. This way it is possible to improve the functional parameters of communication systems elements in mines.

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