Voice Quality Estimation in Combined Radio-VoIP Networks for Dispatching Systems

Jiri VODRAZKA, Jan KUCERAK

Department of Telecommunication Engineering, Faculty of Electrical Engineering,
Czech Technical University in Prague, Technicka 2, 166 27 Prague, Czech Republic

vodrazka@fel.cvut.cz, kucerj17@fel.cvut.cz,

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Abstract. *The voice quality modelling assessment and planning field is deeply and widely theoretically and practically mastered for common voice communication systems, especially for the public fixed and mobile telephone networks including Next Generation Networks (NGN - internet protocol based networks). This article seeks to contribute voice quality modelling assessment and planning for dispatching communication systems based on Internet Protocol (IP) and private radio networks. The network plan, correction in E-model calculation and default values for the model are presented and discussed.*

Keywords

Dispatching communication systems, E-model, voice quality.

1. Introduction

In the past, as well as today, the voice communication is a fundamental type of interactive service. The speech communication plays a vital role in dispatching system for the coordination of crews in logistics, transportation, emergency systems, etc.

In cooperation with the company TTC Telekomunikace, the team from the Department of Telecommunication Technology (Faculty of Electrical Engineering, Czech Technical University in Prague) deals with research project called “Universal Radio Gateway for IP Communication in Dispatching Systems”. The project is focused on development and implementation of universal integrated IP Radio Gateway. The gateway is meant to be integrated in a complex dispatching system providing unified voice and data communication including cooperation with internal and external subsystems (communication, security, information, etc.). The project includes a development of advanced system architectures in order to respect the principles and rules of various radio communication networks applied to internal processes and architecture of the access point. The ultimate goal of the project is to create a modular functional sample of universal radio gateway for IP communication in dispatching network to ensure a higher qualitative level of service or services with higher added value in the form of mobility, accessibility and operability of radio equipment.

Firstly, the team in the Department of Telecommunication Technology focused to simulate a Voice over Internet Protocol (VoIP) communication. The framework OMNeT++ is used for signaling network and voice communication modelling [12]. Secondly, the research team focused on voice quality assessment from basic network parameters, presented for example in [9], [10] and [11].

The E-model defined in the Recommendation ITU-T G.107 [5] is one of the objective quality assessment methods. This method can be adopted for voice quality planning in combined VoIP and private radio dispatching communication systems. The E-model is introduced shortly in Section 2. The correction of E-model calculation and default values for the model are presented in next sections and new interactivity classes of the voice communication are discussed.

2. Voice Communication Quality Assessment

For voice communication quality assessment and especially Quality of user Experience (QoE), it is necessary to investigate the end to end communication quality (QoS) as it is experienced by the users. However, the assessment index must also follow an objective and common systematic approach because there are various kinds of factors reflected to the quality [4]. The Mean Opinion Score (MOS) is a commonly used method to determine the quality of speech. With MOS, the quality of a speech is rated on a scale of 1 (bad) to 5 (excellent).

The speech quality estimates and network planning are usually based on the E-model. These models consider the entire Ear-Mouth path and all relevant conditions such as end-to-end level, echo, delay, distortions and frequency characteristics of the various path segments. The E-model uses a computational method that includes factors such as noise, signal level, loudness ratings, delay and echo impairments, packet loss, type of codec, and even network type to derive a quality score. This transmission quality rating is called as the R-factor. With R-factor, the quality of a speech is rated on a scale of 50 (poor) to 100 (best) for the narrow-band voice communication (frequency band to 3.4 kHz). The maximum value of R-factor may be theoretically to 130 for the wide-band voice communication (frequency band to 7 kHz).

For IP based networks, the score assumes ideal conditions outside the IP cloud. Scores are based on the relevant IP impairments such as packet loss, latency, jitter, and even when these impairments occur over the duration of the call. For combined radio-IP networks, the score assumes the relevant impairments in IP network part and radio-communication part.

2.1. Using of E-model

Amongst other, the public telephone network plan of the transmission parameters for the Czech Republic uses the E-model for complex quality view on the network. The guidelines and planning examples of Recommendation G.108 [6] are based on the utilization of the E-model as described in Recommendation G.107 [5]. The intent of this Recommendation is to demonstrate how the E-model can be used in end-to-end transmission planning for a wide range of local, national, multinational and transcontinental networks. The basic equation for the rating from Recommendation G.107 is modified to following equations [1]:

$$R = 100 - I_{tot} + A, \tag{1}$$

where A is advantage factor and I_{tot} is total impairment factor:

$$I_{tot} = I_o + I_q + I_{dte} + I_{dd} + I_{eeff}, \tag{2}$$

where I_o is noise and loudness rating impairment factor, I_q is quantizing distortion impairment factor, I_{dte} is talker echo impairment factor, I_{dd} is impairment caused by too-long absolute delay, I_{eeff} is effective equipment impairment factor.

The Prague Department of Telecommunication Technology produced a web application for E-model transmission parameters calculation for the Public telephone network plan. This application was used for optimization of basic transmission parameters and default constants of the model. The basic tool calculates the R-factor from the input networks parameters. The second tool calculates the R-factor for the parameters of three co-operating networks. The calculation tools were programmed for the MATLAB Server and they are accessible on the web site http://matlab.feld.cvut.cz [2] and [3].

2.2. Relationship Between MOS and R-Factor

On experience with subjective and objective measurements, the R-factor score was mapped to an equivalent Mean Opinion Score to predict the quality of speech path. The scoring includes consideration for the type of subjective test used for scoring. The passive listening or active conversational tests produce slightly different scores. The tests in different voice-bands (narrow or wide) also cause different scores.

According to ITU-T Recommendation G.107 the R-factor can be converted to MOS and vice versa. The quality category limits for active conversation are presented in Tab. 1.

Tab. 1: Relation between R-factor and MOS for narrow-band voice communication.

R-factor lower limit	MOS	Speech quality category	User satisfaction
90	4.3	Best	Very satisfied
80	4	High	Satisfied
70	3.6	Medium	Some users dissatisfied
60	3.1	Low	Many users dissatisfied
50	2.6	Poor	Nearly all users dissatisfied

2.3. Network Plan for the Dispatching Voice Communication Systems

The plan of transmission parameters defines a reference configuration with two interconnected networks A, B and with third possible network C connecting both networks A and B (reference model shown in Fig. 1).

The parameters are defined between acoustic interfaces of end users, eventually between access points

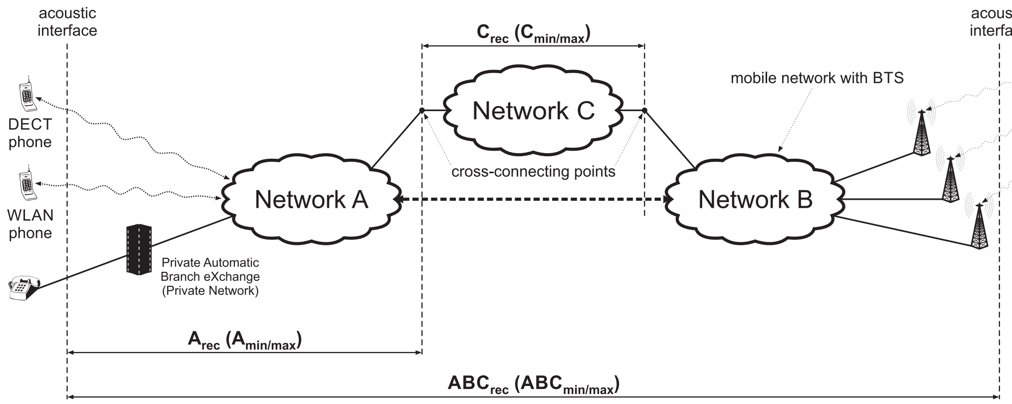

Fig. 1: The reference configuration of dispatching voice communication network plan. Types of parameters - recommended (rec) and limiting (min./max.) for some part of network (A, B, or C) and all network (ABC) are marked.

of these networks. The network "C" is IP based core network for complex dispatching communication systems. The central dispatching point is placed in the network "C". The network "A" is typically external network like public telephone network (fixed or mobile) which is connected via the voice gateway. The network "B" is typically internal private or public radio network like Digital Mobile Radio (DMR or TETRA, MATRA) which is connected via the dispatching gateway.

We primarily assume the half-duplex digital mobile radio terminals connected to dispatching voice gateway. Besides the signaling for voice communication, the information about direction switching must be transmitted.

3. Recommended Values for E-model

The key requirement for voice quality is compliance with total R-factor higher than 50. That corresponds to a total impairment factor equal to 50 (lowered by advantage factor A - the relevant value is from 0 to 20 - as discussed in next subsections).

3.1. Absolute Delay Impairment Calculation

The impairment factor caused by too-long absolute delay I_{dd} was introduced in Eq. (2). The original calculation in the ITU-T Recommendation G.107 (E-model: a computational model for use in transmission planning) used fixed parameters for absolute one-way delay model. In last version from 06/2015, the common values are used. The absolute one-way delay is divided into two intervals:

for $T_a \leq m_T : I_{dd} = 0$,
for $T_a > m_T$:

$$I_{dd} = 25 \left\{ \begin{array}{c} \left(1 + X^{6 \cdot s_T}\right)^{\frac{1}{6 \cdot s_T}} + \dots \\ \dots - 3 \left(1 + \left[\dfrac{X}{3}\right]^{6 \cdot s_T}\right)^{\frac{1}{6 \cdot s_T}} + 2 \end{array} \right\}, \tag{3}$$

$$X = \frac{\log\left(\dfrac{T_a}{m_T}\right)}{\log 2}, \tag{4}$$

where T_a is absolute one-way delay; s_T and m_T are parameters of model. The parameters of the model are presented in Tab. 2 for four interactivity classes. The default class 1 is original case from old version of Recommendation. The class 2 and 3 are new cases for lower levels of interactivity. The class 4 is our newly proposed class to be used for special half-duplex communications in dispatching systems.

Commonly, the level of interactivity is another factor that must also be considered. Usually, an absolute delay below 150 ms one-way provides good interactivity. An absolute one-way delay between 150 ms and 300 ms provides acceptable interactivity (transmission path with the satellite hop). An absolute one-way delay excessing 400 ms should be avoided (in the case of two satellite hops it is about 540 ms). The higher values of absolute delay are possible, but the subscribers must be trained to half-duplex communications and a special signaling system for the voice keying must be used. When there is large delay on the link, the subscriber tends to think that the listener has not heard or paid attention. He will repeat what he said and be interrupted by the delayed response of the remote party. With some operator training it is quite possible to communicate correctly using such service, but the conversation is not natural.

The absolute one-way delay impairment factor dependence is presented on Fig. 2 for different interactivity classes.

Tab. 2: Interactivity classes of voice communication.

Class		Typical delay (ms)	Description	Parameters	
				s_T	m_T
1	Default	0–150	Acceptable for most conversations. Only the most interactive tasks will perceive a substantial degradation.	1	100
2	Low	150–300	Acceptable for communications with low interactivity (for example communication over satellite link and other long links).	0.55	120
3	Very low	300–700	Conversation becomes practically half-duplex.	0.4	150
4	Extra low	Over 700	Conversation impossible without some training to half-duplex communications (for example military communication, special dispatching communication systems).	0.31	200

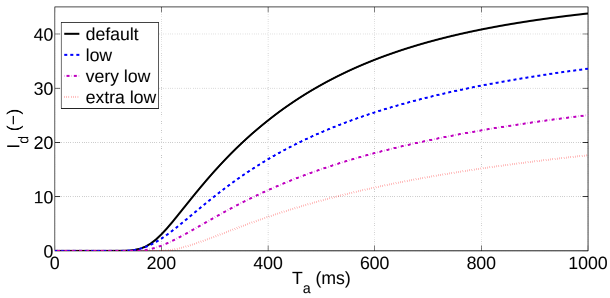

Fig. 2: The absolute one-way delay impairment factor dependence for different interactivity classes.

3.2. Advantage Factor

The Advantage Factor A is specified by some provisional environment. The value $A = 0$ is used for conventional fixed telephone networks. The mobility by cellular network in the building (for example DECT and Wi-Fi cordless phones) is evaluated by advantage factor $A = 5$. The value $A = 10$ can be used for mobile networks in large geographical area or when moving in a vehicle. Access to hard-to-reach location e.g. via multi-hop satellite connections is evaluated by advantage factor $A = 20$. The same value we recommended for special radio networks and voice communication in dispatching systems.

We recommend some selected default parameters in E-model, for example the room noise at the send and the receive side can be set to value between 50 dB and 60 dB for special radio networks and voice communication in dispatching systems.

3.3. Voice Codecs Quality

The quality of voice codecs depends on Effective Impairment Factor, which is calculated with I_e (Equipment impairment factor - basic impairment of codec), B_{pl} (Packet-loss robustness factor) and P_{pl} (Packet-loss Probability) parameters. The major codecs used in dispatching systems for voice communications and their respective equipment impairment factors are listed below in Tab. 3.

The codec ITU-T G.711 uses a classical Pulse Code Modulation (PCM). It captures speech in a range of 3.4 kHz, samples at 8000 samples/second with 8 bits per sample resulting in 64 kbps. The two compress characteristics are defined by the standard, µ-Law (North America & Japan) and A-law (used in Europe and the rest of the world).

The codec ITU-T G.729 operates at a bit rate of 8 kbps with an encoding frame length of 10 ms and 5 ms. The voice encoder uses Conjugate-Structure Algebraic Code Excited Linear Prediction (CS-ACELP). Annex A is a low-complexity version of the G.729 standard. Annex B defines adding functions Voice Activity Detection/Comfort Noise Generator/Discontinuous Transmission (VAD/CNG/DTX).

The GSM-FR is a Full Rate speech coder standardized by the European Telecommunications Standards Institute (ETSI) for GSM digital mobile phone systems. The coder has a bit rate of 13 kbps with an encoding frame length of 20 ms. This coder uses the principle of Regular Pulse Excitation-Long Term Prediction-Linear Predictive coding.

Tab. 3: Quality parameters for selected codecs.

Codec	Nom. MOS	Nom. R factor	I_e
G.711 µ-Law (64 kbps)	4.2	93	0
G.711 A-law (64 kbps)	4.2	93	0
G.729A/G.729AB (8 kbps)	3.91	82	11
GSM-FR	3.57	73	20
GSM HR	3.53	72	21
GSM EFR	4.16	91	2
AMR NB Mode 0 (4.75k)	3.65	75	18
TETRA (ACELP 4.567 kbps)	3.06	59	35
TETRA (AMR 4.75k)	3.65	75	18
MATRA (CELP 4.2 kbps)	3.1	60	34
DMR (AMBE+2 4.8 kbps)	3.82	79	14

The Adaptive Multi-Rate (AMR) is the 3GPP mandatory standard codec for narrowband speech and multimedia messaging services over GSM and evolved GSM (WCDMA, GPRS and EDGE) networks. AMR operates at eight bit rates in the range of 4.75 to 12.2 kbps with an encoding frame length of 20 ms.

AMR uses various techniques, such as ACELP, DTX, VAD and CNG.

The codecs AMBE+ are already used for Digital Mobile Radio (DMR). The codecs Advanced Multiband Excitation (AMBE), AMBE+ and AMBE+2 are based on improved Multi-Band Excitation method. AMBE is a codebook-based vocoder that operates at bit rates of between 2 and 9.6 kbps, and at a sampling rate of 8 kHz in 20 ms frames. The AMBE codec is used by the Inmarsat and Iridium satellite telephony systems.

The quality metrics (R-factor) can be assessed with E-model calculated tool for selected codecs and network configuration used in dispatching voice communication system.

3.4. Example of the Results

Based on the above analysis, the voice communication quality was modeled for some combinations of parameters: factor $A = 20$ corresponds to a special radio networks, the equipment impairment factor is set to double the value of $I_e = 18; 34$ (typically for radio-network TETRA - classically codec ACELP or AMR codec with higher quality; for radio-network MATRA - CELP codec with lower quality). The level of acoustic background noise was set in model firstly to the default value of 35 dB and secondly to the increased level of 60 dB). The dependency of R-factor on the absolute one-way delay is shown on Fig. 3 for two type of codec and two values of acoustic background noise.

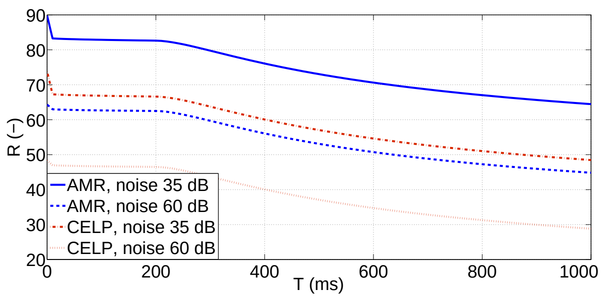

Fig. 3: Dependency of R-factor on the absolute one-way delay for two type of codec and two values of acoustic background noise.

4. Conclusion

The speech communication plays a vital role in dispatching system. The team in the Department of Telecommunication Technology has focused on simulation of VoIP communication, particularly on simulation and comparison of different transport protocols that are used for signaling between different types of radio networks. This article presented our contribution to voice quality modelling assessment and planning for dispatching communication systems based on Internet Protocol (IP) and private radio networks. The network plan, correction in E-model calculation and default values for the model were presented and discussed.

In this work, the new interactivity class “Extra low” was set-up for half-duplex dispatching voice communication with service conversation impossible without some training of communicating parties, Subsection 3.1. The Advantage Factor $A = 20$ was recommended for special radio networks for voice communication in dispatching systems, Subsection 3.1. We recommend some selected default parameters in E-model, for example the room noise at the send and the receive side.

In next step, the voice quality assessment function will be added to network model for complex simulation of voice and signaling communication in the dispatching systems.

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About Authors

Jiri VODRAZKA was born in Prague, Czech Republic in 1966. He joined the Department of Telecommunication Engineering, Faculty of Electrical Engineering, Czech Technical University in Prague in 1996 as a research assistant and received his Ph.D. degree in electrical engineering in 2001. He has been the head of the Transmission Media and Systems scientific group since 2005 and became Associate Professor in 2008. He participates in numerous projects in cooperation with external bodies. Currently he acts also as vice-head of the Department.

Jan KUCERAK was born in Sokolov, Czech Republic. He received his M.Sc. from the Armed Forces Academy of General M. R. Stefanik in 2005. He joined the Department of Telecommunication Engineering, Faculty of Electrical Engineering, Czech Technical University in Prague in 2006 as a research assistant. His research interests include telecommunication protocols and signaling.