

DETERMINATION OF OPTIMAL TEST SEQUENCE FOR INTRUSIVE MEASUREMENT OF VTQoS IN MOBILE ENVIRONMENT

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SUMMARY

This paper describes simulations of a test sequence transmission for intrusive measurement of VTQoS in mobile environment. The aim of the simulation was to analyze an influence of multipath signals propagation and propagation attenuation on the test sequence. Evaluation the generated sequences was based on the calculation of mean square measure and correlation coefficient. These measures were used as a criterion for suitable test sequences selection. Reconsideration of an influence of mobile environment on the test sequences, which are composed from simple signals, is the aim of this paper.

Keywords: VTQoS, test sequence, correlation coefficient, mean square measure, intrusive measurement

1. INTRODUCTION

VTQoS (Voice Transmission Quality of Service) is one of the important parts of QoS (Quality of Service). It is very important for providers as well as for users. When communication networks incorporate more and more transmission technologies, an increase in complication and the complexity of networks is seen. Measurement of the voice transmission quality becomes only platform that is available for simultaneous comparison of different transmission technologies and that is the most relevant to the view of the users.

Of course, it is possible to measure and evaluate the transmission parameters of the networks. But only the evaluation of end-to-end quality provides optimal results because of the complexity of network technologies.

Thus, it is the evaluation in the same way as users do. Since voice service is the most wide-spread service, in which a user uses filter and predicative abilities of human brain, it is crucial to optimally evaluate a quality of such service.

Evaluation of a quality of the voice service may be performed using intrusive or non-intrusive methods, objectively or subjectively.

Using non-intrusive method, we only monitor existing dialogue. The drawback of this method is that the evaluation algorithm cannot utilize an original sample of the primary signal. Thus, it is very difficult to detect some types of signal distortion that occur during transmission.

In the intrusive methods, only a test voice sample is transmitted. These methods have been known since the beginning of the telecommunication technologies, when the special sequences of vowels (known as logathoms) were transmitted after the connection had been built-up. A receiver had to recognize these logathoms. This way of subjective evaluation has been used till nowadays (e.g. method MoS).

Today's technical and software facilities provide an objectification of this measurement method by

transmitting the sound sample defined beforehand, its receiving on the destination side, and a comparison of the transmission sample and the original sample using the suitable algorithm that imitates the way of perception and evaluation of the quality transmission opinion by an average listener. It is for example E-model defined in ETR-250, or algorithm PSQM (Perceptual Speech Quality Measurement) defined in P.861 ITU-T also PESQ (Perceptual Evaluation of Speech Quality) defined in P.862 ITU-T. The algorithm PSQM is based on comparison of the power spectrum of the corresponding sections of the original and the received signals. The results of this algorithm more correlate with the results of listening tests, in comparison with E-model. At present, this algorithm is no more used because of a raw time alignment. Instead of it the algorithm PESQ is used. The algorithm PESQ facilitates with very fine time alignment and one single interruption are also taken into account in the calculation of MoS (Mean Opinion Score). It is possible to use PESQ in mobile networks as well as in the networks based on packet transmission. The disadvantages include impossibility to use it for codec with data rate lower than 4 kbps and higher calculation load what is caused by recursions in the algorithm. A choice of the optimal test sequence is very important for all these methods.

The test sequence would consist of non-speech-like (fully artificial) signals. These signals are closer defined in P.501 ITU-T and the recommendation divides them into deterministic and random signals. An advantage of using these signals is simplicity and possibility of a comparison of the results measured in different language areas. The test sequence composed from those signals enables the comparison of networks of individual countries within one corporation (e.g. Deutsche Telecom, Orange, Vodafone) from the point of the view VTQoS.

Nowadays, the VTQoS intrusive measurements are performed by using samples of speech signal but

the comparison is possible only within the single-language area in this case.

New phenomena, which degrades quality of voice transmission e.g. impulse noise, short-term fading, clipping, or non linear distortion (introduced in codecs with loss compression), appears in wireless telecommunication systems. Here we focus to the influence of multipath signals propagation and propagation attenuation for transmission of test sequence in mobile environment.

2. DESCRIPTION OF TEST SEQUENCES

The length of each of the test sequences is set to 90 sec. This period equals to the length of a phone call of average user. The test sequences are composed from the following signals introduced and evaluated in [1]:

- Sinusoidal signal with frequencies 300, 800, 1000, 1700, 2400, 3000 Hz,
- Square bipolar signal with frequencies 300, 400, 500, 600, 635, 670 Hz,
- Gaussian white noise with $\mu = 0$ and $\delta = 0,0001; 0,001; 0,01; 0,1; 0,5; 1$.

Three types of test sequences with different arrangement were formed. The principle of the creation of the final test sequences is based on an arrangement of initial parts of relevant test sequences, which are shown in Figures 1 – 3. The arrangements shown in Figure 1-3 are used six times to form the final test sequences. Thus, each final test sequence consists of six parts. The signals step-by-step have got the values defined above. That means, in the second part of the test sequence (from 15 sec. to 30 sec.), the signals have the following values:

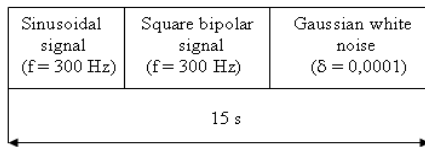


Fig. 1 Initial part of sequence 11 (seq11)

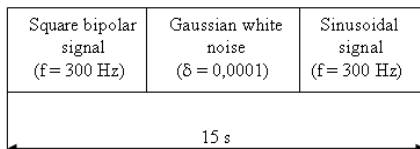


Fig. 2 Initial part of sequence 12 (seq12)

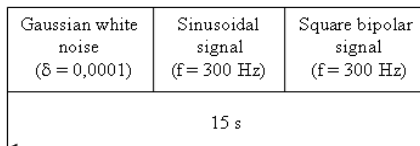


Fig. 3 Initial part of sequence 13 (seq13)

Square bipolar signal $f = 400$ Hz, Gaussian white noise $\delta = 0,001$ and Sinusoidal signal $f = 800$ Hz. The values of the signals in the first parts of the test sequences (from 0 sec. to 15 sec.) are the same as those in Figures 1-3.

3. SIMULATION DESCRIPTION

Test sequences are modelled in Matlab as the sequence of digital samples, which pass through the given type of communication channels (Noise channel AWGN and Rician's channel). The model is to simulate situations in the transmission of the test sequences through the given transmission chain especially from the influence of multipath signals propagation and propagation attenuation on the transmission of the test sequence in mobile environment point of view. Multipath signals propagation is defined by the parameter K for the Rician's channel. K parameter represents the rate of the direct path power P_{DP} to the diffuse power P_D :

$$K = \frac{P_{DP}}{P_D}, \quad (1)$$

Propagation attenuation is defined as follows:

$$PA = 10 \log \frac{P_T}{P_R} = -10 \log \left[\frac{\lambda^2}{(4\pi)^2 x^2} \right], \quad (2)$$

where P_R is the received power, P_T is the transmitted power, λ is the wavelength of the signal and x is the distance from the transmitter. The equation (2) only applies in the case that the both antennas have gain equals to 1. The propagation attenuation in this model is described by SNR parameter for the AWGN channel. The propagation attenuation in mobile environment is modelled by using Gaussian white noise. The Gaussian white noise is introduced into transmission chain by AWGN channel. This solution was chosen because a real receiver could not be modelled in given simulation environment.

The model is made in such way that so many errors as possible may occur during transmission. Therefore no channel coding is used. The maximum number of errors is needed because we want to get the test sequence that is the most sensitive on these errors.

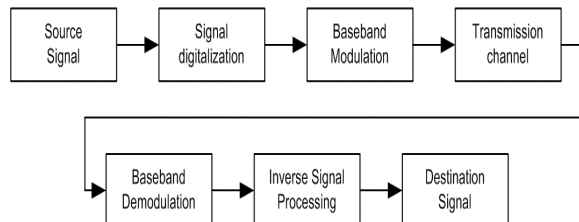


Fig. 4 Principal arrangement of model

3.1. Detailed description of the simulation model

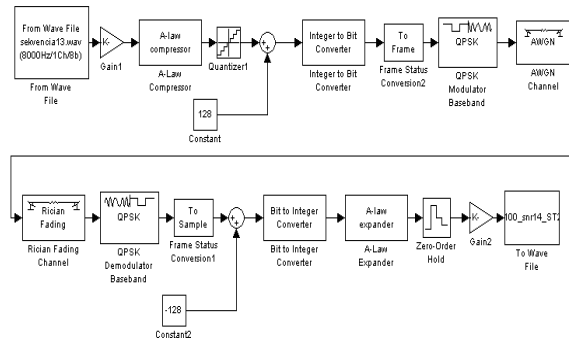


Fig. 5 Simulation model with Rician's channel

This model follows the model described in [1]. The model is divided into the following blocks: "From Wave File", "QPSK Baseband Modulator", "QPSK Baseband Demodulator" and "Rician Fading Channel".

The block "QPSK Baseband Modulator" modulates a signal using the quaternary phase shift keying method. The output is a baseband representation of the modulated signal.

The Block "QPSK Baseband Demodulator" demodulates the signal that was modulated using the quaternary phase shift keying method. The input is a baseband representation of the demodulated signal.

The block "Rician Fading Channel" implements a baseband simulation of a Rician fading propagation. This block is useful for modelling mobile wireless communication systems when the transmitted signal can travel to the receiver along a direct path. Due to the fading the signal spreads and becomes diffuse. The K -factor, which is part of the statistical description of the Rician distribution, represents the ratio between direct-path (unspread) power and diffuse power. The ratio is expressed linearly, not in decibels. While the Gain parameter controls the overall gain through the channel, the K -factor parameter controls the gain's partition into direct and diffuse components. Relative motion between the transmitter and receiver causes Doppler shifts in the signal frequency. The Jakes' PSD (power spectral density) [12] determines the spectrum of the Rician process. More information about Rician's channel is in [13], [14].

The block "AWGN channel" adds Gaussian white noise to real or complex input signal. When the input signal is real, this block adds the real Gaussian noise and produces the real output signal. When the input signal is complex, this block adds the complex Gaussian noise and produces the complex output signal.

The block "From Wave File" reads audio data from the Microsoft Wave (.wav) file and generates a signal of the one of the following data types: double, single, int16, uint8. The audio data has to be in uncompressed PCM (pulse code modulation) format. The block supports 8-, 16-, 24-, and 32-bit Microsoft Wave (.wav) files.

3.2. Principle of simulation

The source sequences are created using Soundforge software. The principle of the creation of source sequences is described in Chapter 2. The destination sequences are created by simulation. The length of the simulation is 90 sec. This length is the same as the length of the source sequence. The length of the destination sequence is also 90 sec. what results from the length of the simulation. The source and the destination sequences are compared after finishing the simulation. The principle of the comparison is described by the following steps:

1. Reading in source and destination test sequences.
2. Segmentation of the test sequences into n intervals, each with 8000 samples. The following parameters are calculated in each interval:
 1. Correlation coefficient r_i ,
 2. Coefficients FFT,
 3. Mean square measure d_i .
3. Calculation of the average values:

$$\bar{r} = \frac{\sum_{i=1}^n r_i}{n}, \quad (3) \quad \text{and} \quad \bar{d} = \frac{\sum_{i=1}^n d_i}{n}, \quad (4)$$

where n is the number of intervals, r_i is the correlation coefficient of the i -th interval, d_i is the mean square measure of the i -th interval.

The segmentation of the test sequences into n intervals and calculation of relevant parameters in these intervals enables to obtain more precise results.

The calculation of the correlation coefficient is realized by the following relation:

$$r_i = \frac{\sum_m \sum_n (A_{mn} - \bar{A})(B_{mn} - \bar{B})}{\sqrt{\left(\sum_m \sum_n (A_{mn} - \bar{A})^2 \right) \left(\sum_m \sum_n (B_{mn} - \bar{B})^2 \right)}}, \quad (5)$$

where $\bar{A} = \text{mean2}(A)$, and $\bar{B} = \text{mean2}(B)$. Matlab function mean2 realize the calculation of the mean value.

The mean square measure is based on the spectral comparison of the test microsegment with the reference microsegment. The most common norm is L_2 -norm, which is defined as:

$$d_i(t, r) = \left[\sum_{j=1}^N (y_{ij} - y_{rj})^2 \right]^{1/2}, \quad (6)$$

where N is the number of FFT points in given microsegment, y_{ij} is the absolute value of the j -th FFT coefficient of the test microsegment, y_{rj} is the absolute value of the j -th FFT coefficient of the reference microsegment.

Nowadays, the comparison is usually done using the correlation coefficient. If we want to discover

differences in the spectrum area, another parameter is added. Using the correlation coefficient and the mean square measure we may find the test sequence that is the most sensitive on noise influences, which arise in communication channels.

4. PRESENTATION OF RESULTS

The simulations of the described test sequences mentioned above were performed on AWGN channel and Rician's channel for A-law compressed characteristic. Transmissions were done simultaneously for both AWGN and Rician's channels. Spreading in mobile environment is modelled using the Rician's channel, and simulations were carried out for different settings of K parameter. This parameter enables to model multipath propagation of signals in mobile environment. The values of K parameter represent particular cases of the transmission in mobile environment. The following values of K parameter were used for the simulations:

- $K=1$, a transmission with a great number of the reflections,
- $K=5$, a transmission with an average number of the reflections,
- $K=10$, a transmission with minimum number of the reflections (line of sight transmission).

The same value of maximum Doppler shift 50 Hz in Rician's channel was adjusted in all simulations.

The propagation attenuation in this model is described by SNR parameter for AWGN channel. The propagation attenuation in mobile environment is modelled by help of Gaussian white noise. The Gaussian white noise is introduced by AWGN channel into transmission chain. This noise represents a noise of the environment. Such solution was chosen because a real receiver could not be modelled in given simulation environment. The QPSK modulation enables simulations in the range of 9-14 dB of SNR. The lower limit is given by handover of base stations for low SNR. The upper

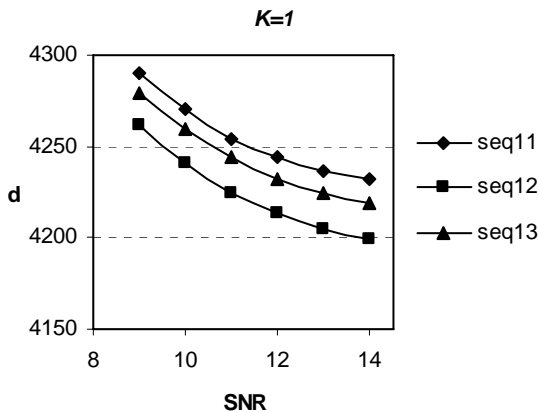


Fig. 6 Graphical presentation of the results of the simulations for mean value of mean square measure ($K=1$)

limit results from the QPSK modulation characteristics, which has error probability $P_e = 10^{-7}$ for SNR 14 dB. More information about the QPSK modulation characteristics is in [9]. Since that, it is no use the go towards to higher values in the simulations, because the sensibility of the demodulator raises rapidly, what can have a negative influence on the results of simulations. The aim of the simulation was to detect an influence of multipath signals propagation and propagation attenuation on the test sequence. The Figures 6-11 show the results obtained by the simulations.

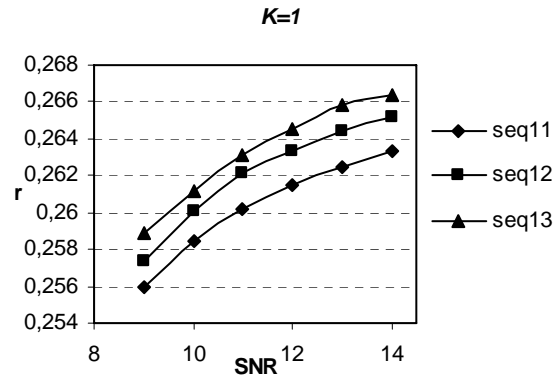


Fig. 7 Graphical presentation of the results of the simulations for mean value of correlation coefficient ($K=1$)

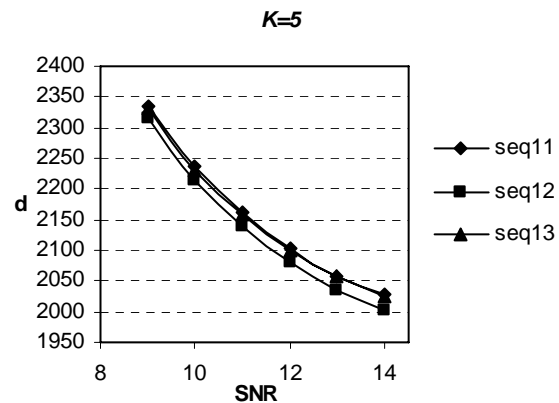


Fig. 8 Graphical presentation of the results of the simulations for mean value of mean square measure ($K=5$)

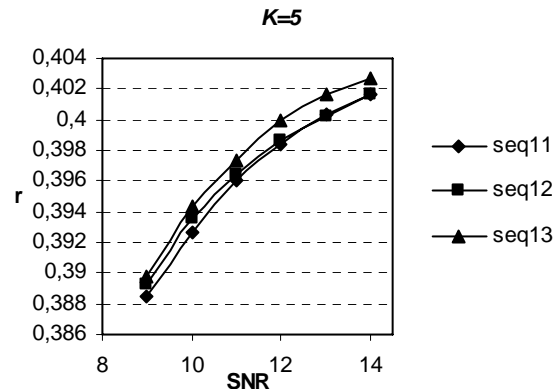


Fig. 9 Graphical presentation of the results of the simulations for mean value of correlation coefficient ($K=5$)

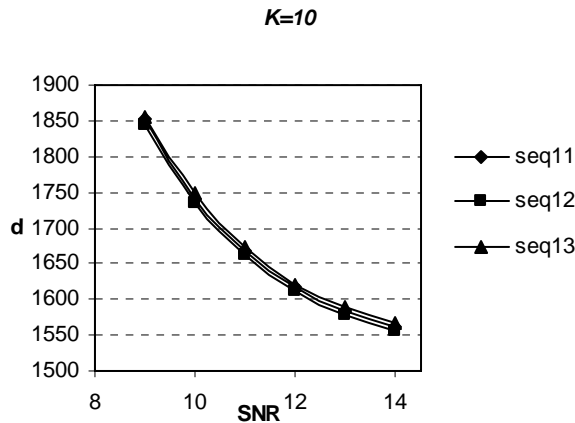


Fig. 10 Graphical presentation of the results of the simulations for mean value of mean square measure ($K=10$)

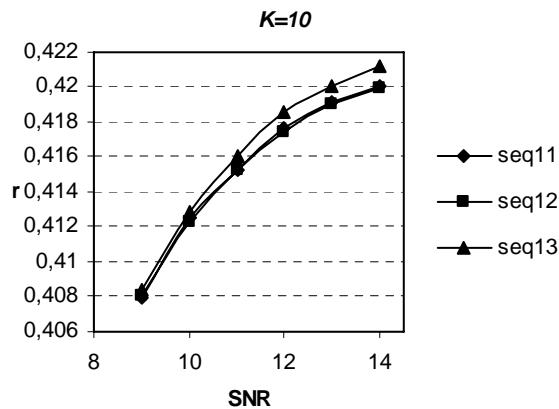


Fig. 11 Graphical presentation of the results of the simulations for mean value of correlation coefficient ($K=10$)

5. CONCLUSION

The principle of test sequences selection for automatic measurement of VTQoS is based on a simple rule. The rule is that such group of test sequences is used for measurement which obtains the maximum difference of values of the relevant boundary coefficients. The following formulas were used for calculations:

$$\delta_r = \bar{r}_{(SNR\ 14)} - \bar{r}_{(SNR\ 9)}, \quad (7)$$

$$\delta_d = \bar{d}_{(SNR\ 9)} - \bar{d}_{(SNR\ 14)}, \quad (8)$$

where δ_r is the difference of average values of boundary parameters of the correlation coefficient, δ_d is the difference of average values of boundary parameters of the mean square measure, $\bar{r}_{(SNR\ 14)}$ is the average value of the correlation coefficient for relevant value of boundary parameter for AWGN channel, $\bar{d}_{(SNR\ 14)}$ is the average value of the mean square measure for relevant value of boundary parameter for AWGN channel.

Boundary parameters are 9 dB SNR and 14 dB SNR in AWGN channel.

This fact was a basis for derivation of this rule because we need the test sequence which is the most sensitive on all influences, which can arise in real networks. We can hypothesize that if the maximum difference is reached for same conditions, the given test sequence is more sensitive on interference influences than the others. Thus the test sequence with such property is more suitable for measuring of VTQoS. The best test sequence in the sense of the mean square measure and the correlation coefficient is chosen from the results.

δ_d	$K=1$	$K=5$	$K=10$
Test sequence11 (seq11)	59	308	290
Test sequence12 (seq12)	63	312	287
Test sequence13 (seq13)	60	307	290

Tab. 1 Difference δ_d for mean square measure

δ_r	$K=1$	$K=5$	$K=10$
Test sequence11 (seq11)	0,0073	0,0132	0,0122
Test sequence12 (seq12)	0,0078	0,0124	0,0119
Test sequence13 (seq13)	0,0075	0,0129	0,0128

Tab. 2 Difference δ_r for correlation coefficient

The higher is value of K parameter for Rician's channel, the higher is value of direct-path power. The higher direct-path power, the less relevant is the influence of multipath signals propagation on transmission signal. It causes increasing the value of the correlation coefficient test sequences by increasing the value of K parameter. And it causes decreasing value of the mean square measure by increasing the value of K parameter in the spectral domain.

We can see from the results that the influence of the mobile environment on these test sequences is approximately the same. Hence we can use any of these test sequences for the measurement in mobile environment. We decided that we used the test sequence for mobile environment that reached the best results of the simulations in the environment of fixed telecommunication network. This decision was influenced by stronger influence of the environment of fixed telecommunication network on these test sequences. Test sequence 12 gave the best results in the environment of fixed telecommunication network in the sense of the mean square measure and the correlation coefficient. The selection of suitable test sequence in environment of the fixed telecommunication network is described in [15]. In

the future, convenience of this test sequence for intrusive measurement of VTQoS will be verified practically by real measurements in convergent network of the University of Žilina.

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BIOGRAPHIES

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Martin Vaculík was born in 1951, in Žilina, Slovakia. He graduated from the University of Žilina as Dipl. Ing in Telecommunications in 1976. He received PhD in 1987. The title of his thesis was some possibilities of dynamic routing control. He worked in Siemens PSE Company, in Optical network department in 2001-2002. Nowadays he works in the Department of the Telecommunications at the University of Žilina as an associate professor. His areas of interest include switching, access networks and convergent networks. He is author of the book with the title Access networks and co-author of 2 books: ISDN and Broadband networks.