Speech Quality Measurement of GSM Infrastructure Built on USRP N210 and OpenBTS Project

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Abstract. *The paper deals with the methodology for speech quality measuring in GSM networks using Perceptual Evaluation of Speech Quality (PESQ). The paper brings results of practical measurement of own GSM network build on the Universal Software Radio Peripheral (USRP N210) hardware and OpenBTS software. This OpenBTS station was installed in open terrain, and the speech quality was measured from different distances from the transmitter. The limit parameters of OpenBTS station with USRP N210 were obtained.*

Keywords

Asterisk, GSM, OpenBTS, PESQ, USRP.

1. Introduction

GSM and UMTS technologies have become a common part of almost every aspect of a human activity during its expansion of the last decade. Nowadays people consider using mobile phones for calling, texting and browsing the Internet as natural service like electricity or drinkable water at home. All these services require reliable infra-structure and control mechanism to ensure quality of the provided service. This paper deals with one aspect of control mechanism, measuring and controlling the quality of speech. In GSM/UMTS environment we can take advantage of algorithms that have been invented for IP-based networks. There are two basic evaluations of speech quality, objective and subjective. We have aimed at the objective method of measuring because of the zero influence of human factor and because of the possibility to repeat the same methodology for measuring speech quality in every GSM or UMTS infrastructure. The mobile terminals were connected to GSM infrastructure provided by testing OpenBTS station; build on Universal Software Radio Peripheral. We have aimed at measurement of speech quality in open terrain according to the distance of the mobile terminal from the base transceiver station and according to the number of simultaneous calls.

2. State of the Art

The speech quality evaluation is measured in Mean Opinion Score (MOS), which is five-degree scale developed by ITU-T. The objective methods are trying to be as precise as possible to gain adequate MOS value as it would be obtained by subjective methods with a sufficient number of participants for adequate statistical analysis. The objective methods are divided to intrusive and non-intrusive methods. The intrusive methods use the original sound sample as it has entered the communication channel and compare it with the degraded signal in the output. The most known intrusive methods are:

- PSQM (Perceptual Speech Quality Measurement),
- PAMS (Perceptual Analysis Measurement System),
- PESQ (Perceptual Evaluation of Speech Quality),
- P.OLQA (Perceptual Objective Listening Quality Assessment).

According to the fact, that PESQ algorithm is currently widely used, we have decided to use this algorithm as the keystone of our measurement. The intrusive method is used to determine the quality of the speech, when the test system compares the original signal $x(t)$ with the degraded signal $y(t)$ taken at the other end of the transmission chain. Subsequently, reached values are evaluated in MOS-PESQ scale and then transferred using complementary recommendation P.862.1 on the values of the MOS-LQO scale, as shown in Fig. 1, [1], [2].

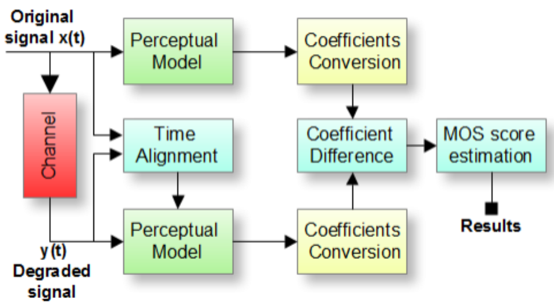

Fig. 1: Basic diagram of PESQ algorithm.

The PESQ method first computes several series of delays between the original and degraded signal, where each of these series corresponds to one interval of signal. The delay for each interval is different due to ensure proper functioning of the PESQ algorithm. For each interval, it is also determined start point and end point of time. Based on these series of delays, PESQ algorithm compares the original and degraded signal using the perceptual model.

The resulting PESQ-MOS score is expressed as a range of values from -0.5 to 4.5. This score has to be converted to more accurate scope, more accurate for human subjective evaluation. Therefore, it is necessary to use complementary ITU-T P.862.1, which will provide scale transfer from MOS-PESQ to MOS-LQO. Scale MOS-LQO provides a range of values from 1 to 5. Conversion from PESQ MOS to MOS LQO is defined by Eq. (1), [3], [4]:

$$y = 0.999 + \frac{4.999 - 0.999}{1 + e^{-1,4945x+4.6607}}, \tag{1}$$

where the variable x represents the value of the MOS-PESQ scale, and the y represent MOS-LQO score. Inverse score (MOS-PESQ) from LQO is shown in Eq. (2):

$$x = \frac{4.6607 - \ln\dfrac{4.999 - y}{y - 4.999}}{1.4945}. \tag{2}$$

3. Testing Platforms

We have designed the measuring platform, which is able to generate GSM calls automatically in regular intervals and analyze the voice sample according to the P.862 specification. The results were logged for future purposes.

As the measured platform, we have used own GSM infrastructure provided by testing OpenBTS station build on Universal Software Radio Peripheral used in our laboratory.

3.1. Measuring Platform Details

The measuring platform was implemented on the low-energy consumption embedded device. According to this fact, we were able to use power supply from car and get high mobility of end mobile devices. The low-cost Huawei K3765 modems were used as end mobile devices for originating and receiving testing calls. These modems were connected directly with embedded device by USB ports and controlled by Asterisk PBX system installed inside the embedded device. As an operating system and core, we have decided to use a special Linux based system developed in our laboratory called BESIP [5].

The calls with original speech sample were generated automatically by Asterisk over SIP (Session Initiation Protocol) and RTP (Real-time Transport Protocol), as shown in Fig. 2. The modems were providing SIP/GSM translation of the outgoing and incoming calls. The Asterisk PBX was recording the incoming voice data to a separate WAV file with adequate timestamp of the measurement. After the end of the call, the PESQ algorithm was applied to the original and recorded degraded signal. The result, speech quality in MOS, was stored in the database for future evaluation.

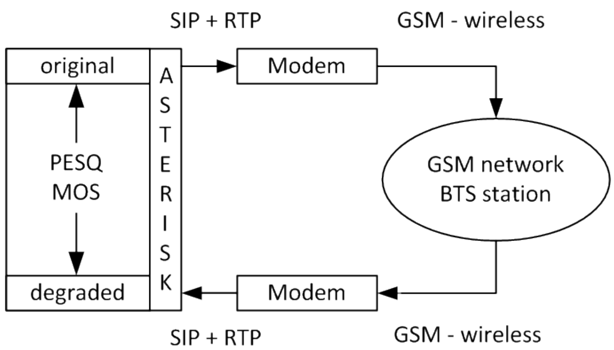

Fig. 2: BESIP measuring tool.

3.2. Measured Platform Details

The tested GSM infrastructure was built on Universal Software Radio Peripheral (USRP) hardware in connection with OpenBTS (Open Base Transceiver Station) software, which is open-source Linux-based application, that provides management of USRP to create wireless GSM interface, as shown in Fig. 3. We have used USRP N210 from Ettus Re-searchTM in combination with Daughterboard WBX [6].

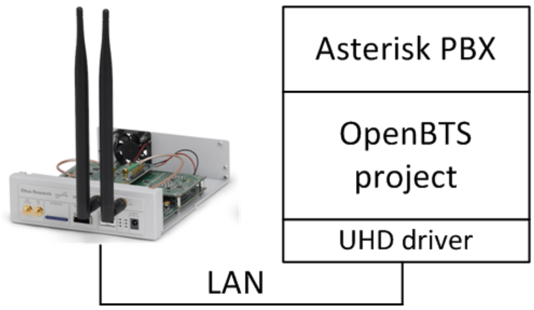

Fig. 3: USRP N210 connected with host OpenBTS server.

The USRP N210 architecture includes Xilinx Spartan 3A-DSP 3400 FPGA kit. In combination with WBX 50–2200 MHz Rx/Tx Daughterboard provides adequate architecture for creating cell phone base station or other applications.

There is UHD (USRP Hardware Driver), which provides API and drivers to the host computer for communication with USRP device. OpenBTS software is written in the C++ programming language and ensures the following functions of Um radio interface, which provides the radio interface for GSM standard [7], [8]:

- GMSK modulation with 13/48 MHz modulation rate and 200 kHz distance – supports GSM850, PGSM900, DCS1800 and PCS1900.
- Multiplexing and coding.
- Management of network resources.

Asterisk PBX is implemented inside the OpenBTS project and provides mobility management, authentication of the user and routing of the calls between registered users. Registered modems are identified by IMSI number of SIM card presented in the modems. The main advantage of this solution is that the calls can't be affected by commercial mobile operator; whole traffic is routed through our low-cost infrastructure.

3.3. Measuring Realization

We have installed USRP N210 with OpenBTS server statically outside the laboratory in an open terrain to ensure minimal influence of surrounding condition on a radio signal spreading.

Figure 4 depicts the measuring platform, represented by embedded BESIP with modems, was installed in the car and placed nearby our OpenBTS station.

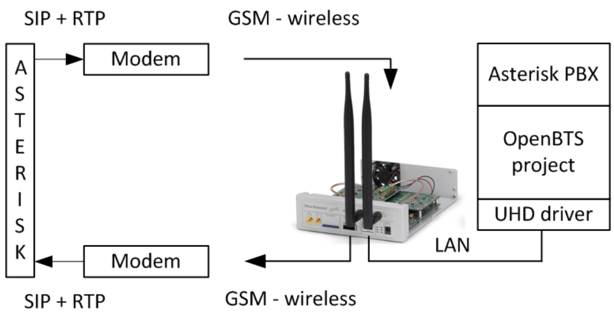

Fig. 4: Scheme of measuring – BESIP + USRP N210 connected with host OpenBTS server.

When the modems were registered to our OpenBTS station, there were 3 calls originated on that place. The reason of the three calls was to get statistically reasonable results of speech quality and strength of the incoming signal to the modems. Then the measurement was repeated with increasing distance from the OpenBTS station.

We were interested in a few measuring parameters:

- The maximum number of simultaneous call, that is our OpenBTS able to provide.
- Dependence of receive signal strength on the distance from our OpenBTS.
- Dependence of a speech quality on the distance from our OpenBTS.
- Dependence of a speech quality on the number of simultaneous calls.
- Time period from originating to answering the call and its dependence on distance from OpenBTS and number of simultaneous calls.

4. Results of Measurement

After many repeated measurements in increasing distance, we have reached a sufficient amount of data to statistically evaluate limit parameters of our OpenBTS system. The results were following.

4.1. Maximum Number of Simultaneous Calls

OpenBTS in a basic non-commercial version uses one logical channel. GSM technologies use time multiplex-

Tab. 1: Success rate of established calls.

Number of simultaneous calls	Number of originated calls	Number of successfully answered calls	Probability of successful answered calls [%]
1	35	31	88.6
2	34	23	67.6
3	31	4	12.9

ing, one channel can be separated to 8 single time channels. We have found out that in our configuration the maximum number of simultaneous calls was 3 simultaneous calls. The probability of successful call establishing was only 12 % (4 successful calls from 31 tries).

4.2. Dependence of Receive Signal Strength on the Distance

There was a received signal strength (RSSI) measured during every active call. Measured and average values are depicted in Fig. 5. As the picture shows, the RSSI decreases -1.95 dB to 1 meter in average.

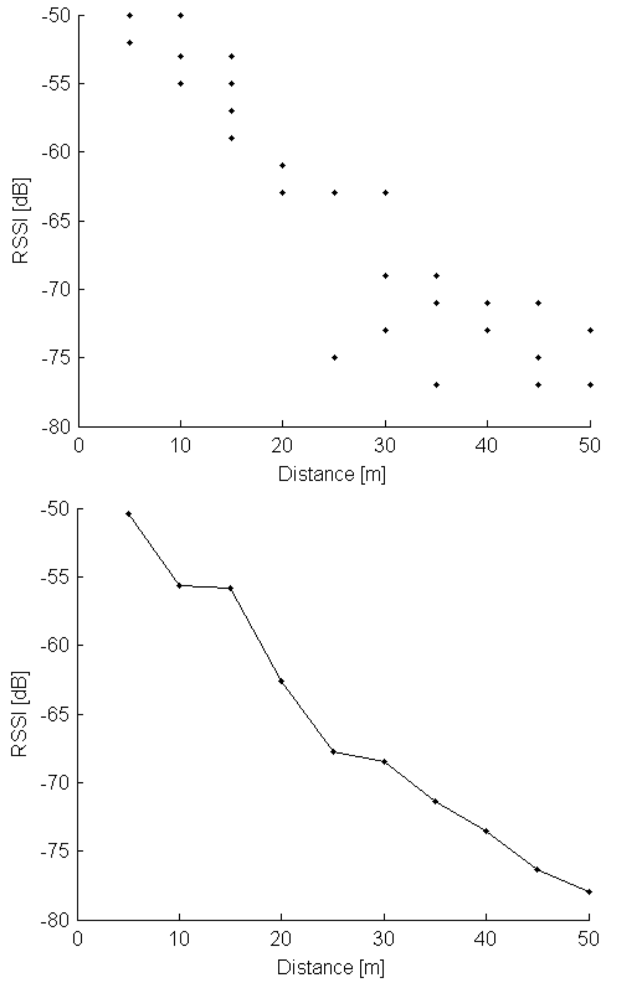

Fig. 5: RSSI of GSM signal in different distance from OpenBTS.

4.3. Dependence of a Speech Quality on the Distance from OpenBTS

The measured MOS values of speech quality are depicted in Fig. 6. We have removed values smaller than 2.2 MOS, because of the poor quality according to the ITU-T P.800 recommendation.

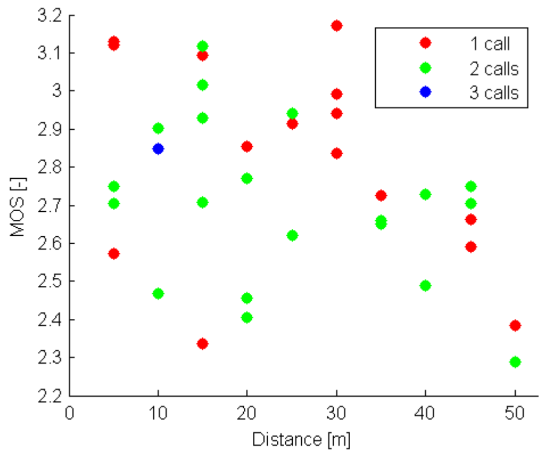

Fig. 6: Speech qualitz in MOS in different distance from OpenBTS.

We have made regression analysis from the measured data. The results are depicted in Fig. 7, Fig. 8, Fig. 9. According to the low count of the successful calls for 3 simultaneous calls, we have added the MOS values < 2.2 to show regression progress for 3 simultaneous calls in Fig. 9.

4.4. Time Period from Originating to Answering the Call

The results in Fig. 10 showed that time period of establishing call do not depend on the distance from the OpenBTS transmitter, but does depend on the count of active simultaneous calls.

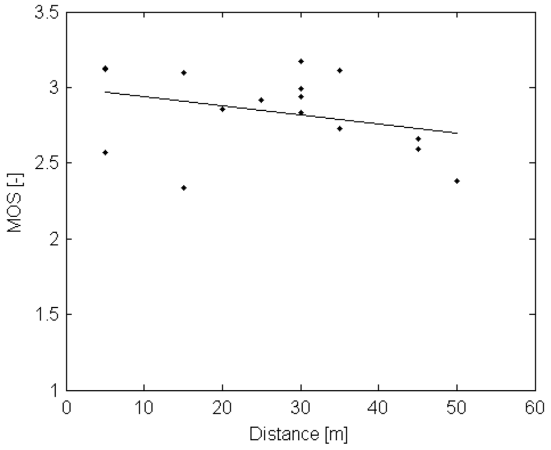

Fig. 7: Linear regression MOS value – 1 call.

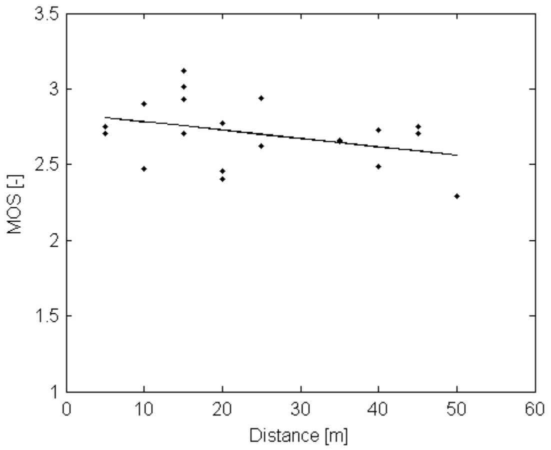

Fig. 8: Linear regression MOS value – 2 call.

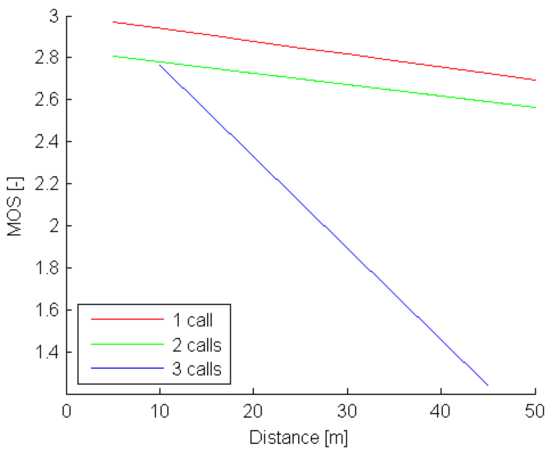

Fig. 9: Linear regression MOS value for all calls.

5. Conclusion

The results of the measurement showed that our OpenBTS station built on the USRP N210 is not suitable for practical application, only for laboratory purposes. The maximum of simultaneous calls were 3 calls, optimal load of the OpenBTS station is 1–2 simultaneous calls from 5 to 50 meters from the transmitter. The poor speech quality and connection failures are detected out of the measured limits. The measurement approved theoretical prerequisite that the speech quality decreases with increasing distance from the OpenBTS station and number of simultaneous calls.

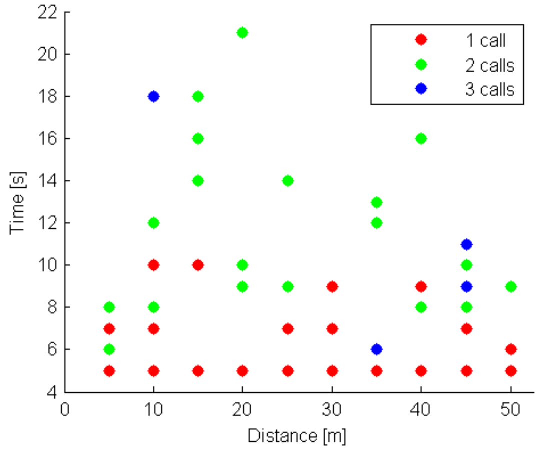

Fig. 10: Time period of establishing calls.

The main benefit of the paper is a methodology for automatic speech quality measurement of a general GSM infrastructure. The methodology uses low-cost embedded hardware, modems and open-source software tools.

Our next step will be to install new OpenBTS station built on more powerful USRP hardware from National Instruments, repeat the same measurement on it and compare the results.

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References

[1] ROZHON, J. and M. VOZNAK. Development of a speech quality monitoring tool based on ITU-T P.862. In: *34th International Conference on Telecommunications and Signal Processing (TSP)*. Budapest: IEEE, 2011, pp. 62–66. ISBN 978-145771411-5. DOI: 10.1109/TSP.2011.6043771.

[2] ITU-T Recommendation G.107. *The E-model: A computation model for use in transmission planning*. Geneva: ITU-T, 2012.

[3] ITU-T Rec. P.862. *Perceptual Evaluation of Speech Quality (PESQ): An Objective Method for End-to-End Speech Quality Assessment of Narrow-Band Telephone Networks and Speech Codecs*. Geneva: ITU-T, 2001.

[4] ZHAO, J.-Y., F. CAO, M.-Y. LIANG and F. LIU YA. Very Low Bit Rate Speech Coding Algorithm and Its Implementation Based on DSP. *Computer Engineering*. 2011, vol. 37, iss. 21, pp. 261–263. ISSN 1000-3428. DOI: 10.3969/j.issn.1000-3428.2011.21.089.

[5] TOMALA, K., L. MACURA, M. VOZNAK and J. VYCHODIL. Monitoring the quality of speech in the communication system BESIP. In: *35th International Conference on Telecommunications and Signal Processing (TSP)*. Prague: IEEE, 2012, pp. 255–258. ISBN 978-146731118-2. DOI: 10.1109/TSP.2012.6256293.

[6] ETTUS RESEARCH. *USRP N200/N210* Avalaible at: `http://www.ettus.com/content/files/07495_\Ettus_N200-210_DS_Flyer_HR_1.pdf`

[7] APVRILLE, A. *OpenBTS for dummies*. 2013. Available at: `http://gnuradio.org/redmine/attachments/\420/fordummies.pdf`

[8] NATALIZIO, E., V. LOSCRI, G. ALOI, N. PAOLI and N. BARBARO. The practical experience of implementing a GSM BTS through open software/hardware. In: *3rd International Symposium on Applied Sciences in Biomedical and Communication Technologies (ISABEL 2010)*. Rome: IEEE, 2010, pp. 1–5. ISBN 978-1-4244-8131-6. DOI: 10.1109/ISABEL.2010.5702870.

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