A Methodology for Measuring Voice Quality Using PESQ and Interactive Voice Response in the GSM Channel Designed by OpenBTS

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Abstract. *This article discusses a methodology for rating the quality of mobile calls. Majority telecommunications service from the perspective of the whole world is using mobile telephony networks. One of the problems affecting this service and its quality are landscape barriers, which prevent the spread signal. Price and complex construction of classic BTS does not allow their dense distribution. In such cases, one solution is to use OpenBTS technology. Design of OpenBTS is more available, so it can be applied to much more places and more complex points. Purpose of this measurement is a model for effective stations deployment, due to shape and distribution of local barriers that reduce signal power, and thus the quality of speech. GSM access point for our mobile terminals is OpenBTS USRP N210 station. The PESQ method for evaluating of speech quality is compared with the subjective evaluation, which provides Asterisk PBX with IVR call back. Measurement method was taken into account the call quality depending on terminal position. The measured results and its processing bring knowledge to use this technology for more complicated locations with degraded signal level and increases the quality of voice services in telecommunications.*

Keywords

Asterisk, openBTS, PESQ, SIP, speech quality.

1. Introduction

Testing QoS (Quality of Service) is one of the key challenges in modern telecommunications networks and the importance of these tests increases with increasing complexity of telecommunication networks, where the telecommunication chain involves more transmission technologies (called convergence networks).

To evaluate the quality of speech transmission over a telecommunications network, respectively after codecs processing. There are two basic evaluations, objective and subjective. This parameter is becoming one of the few measurable in general, to compare different transmission equipment, which is essentially the closest in terms of the end-users.

Speech coding becomes a key parameter in modern communication systems with limited bandwidth. The encoded data is sent via radio frequencies and are exposed to sensitive transmission lines which are susceptible to errors. Difficult, almost impossible is reconstruction of signal caused by these errors. One reason is the narrower bandwidth, which does not allow increasing redundancy.

Adaptive speech coding and transmission errors in mobile transmission systems may operate very distracting. Interference is quite different compared to traditional analog interference, and therefore this effect cannot be described by conventional measurement.

Typical errors encountered by mobile transfers are: impulse noise, short interruptions, trimming and nonlinear signal distortion using a loss-codecs. Another reason of distortion channel of communication in GSM networks are landscape barriers and constructions. This deficiency could be removed denser deployment of BTS, but it is financially and legislative demanding. Problem with barriers and complexity of land surface is handled by projects OpenBTS.

This project provides a smaller station, which are inexpensive and can be easier implemented [1], [2].

2. OpenBTS

OpenBTS (Open Base Transceiver Station) is a term for software that will also allow the implementation of GSM access point. OpenBTS software itself is written in the programming language C++ and allows you to connect calls between registered stations to the created network and between networks of different providers too. For proper functioning of a mobile network is needed not only software for creating mobile network (OpenBTS), but also the hardware part - the transmitter. Figure 1 shows a product of Ettus ResearchTM, A National Instruments Company.

Fig. 1: OpenBTS USRP N210 from Ettus Research.

Asterisk PBX is necessary for making a call between registered endpoints. Asterisk PBX and OpenBTS software can be installed on the same workstation. This arises implementation of a fully functional mobile network operated by a single device. Mobile station that is successfully registered on the network created by the OpenBTS software is identified by "IMSIxxxxxxxxxxxxxx" shape in the VoIP network. Fourteen or fifteen digit unique identifier is the IMSI number of the SIM card present in the mobile station. For the IP address of the SIP user is using the same IP address, which has the BTS station. OpenBTS itself isn't visible in the VoIP network.

The OpenBTS system is utilized implementation Um radio interface, the same interface, which is used for communication with the normal BTS. Um provides radio interface for the GSM standard, which consists of three lowest layer of reference ISO/OSI model. Physical layer consists from three sublayers:

- The radio modem: radio transmitters support GMSK modulation with 13/48 MHz modulation rate and 200 kHz channel distance. Providing neighboring channels in the same cell isn't recommended because overlapping. OpenBTS supports four of the most commonly used GSM bands: GSM850, PGSM900, DCS1800 and PCS 1900.

- Multiplexing and coding: Each physical channel is divided into multiple logical channel using time division multiplex. GSM timing is controlled via the BTS with SCH and FCCH channels.

- FEC Encoding: Allows protection bits, thus provides error detection and correction of isolated bits.

The link layer uses the LAPDm protocol, which is a mobile version of the network communication protocols used in ISDN - LAPD. The protocol ensures that messages are sent without error and executed in the correct order. In this layer are also generated logical channels. Third layer of Um radio interface (network layer) provides:

- Management of network resources: This sublayer handles the assignment and release of the logical channels of the radio link.

- Mobility Management: sublayer provides authentication of the user and then monitors its movement between cells. The results are processed by OpenBTS using Asterisk SIP registry.

- Communication Management: Enables connecting telephone calls. Operations in OpenBTS are translated into the corresponding SIP operations and executed by Asterisk [3], [4], [10].

3. Perceptual Evaluation of Speech Quality

PESQ is an algorithm for the measurement and evaluation of speech quality in telecommunication systems. 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. 2.

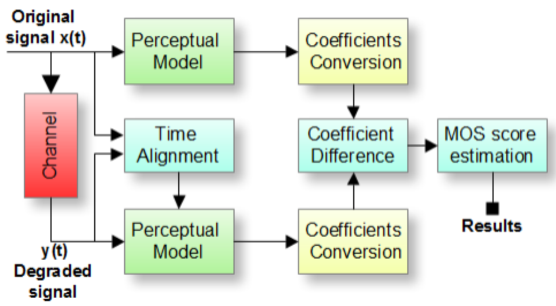

Fig. 2: MOS score estimated by PESQ algorithm.

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 Eg. (1), [5], [6]:

$$y = 0,999 + \frac{4,999 - 0,999}{1 + e^{-1,495x+4,6607}}, \qquad (1)$$

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 Eg. (2):

$$x = \frac{4,6607 - ln\frac{4,999-y}{y-4,999}}{1,4945}. \qquad (2)$$

4. Interactive Voice Response

IVR service (Interactive Voice Response) allows us to use voice or DTMF tone dialing for operating with an automated system and choose from several options in the preset menu. IVR can be used for collecting data from the calling customer, such as account numbers, passwords or personal information, but also can make a call back or put the caller in queue. This service is expanding more and more, it helps to effectively solve problems instead of operators who may not always be available.

Figure 3 shows a hierarchy of IVR tree. IVR was used to compare the results of PESQ method in this case. CALL BACK service was activated after the call. The user was attended to evaluate the quality of call speech via DTMF feedback.

Questionnaire should contain as little as possible questions, because it might get bored caller. This follows from the statistical research about similar interviews. The end user evaluates various quality parameters with 1-5 values, via DTMF signaling. These data are logged in a database [7], [8].

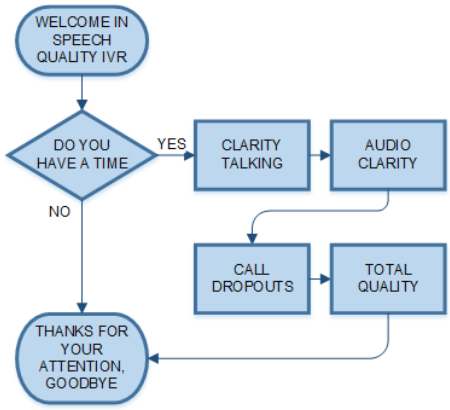

Fig. 3: IVR flowchart for speech quality measurements. Each question is evaluated from 1 to 5. One is the best quality.

5. Measurement

Measurement was performed in a laboratory at Department of Telecommunications, VŠB Technical University of Ostrava. Transmitting side of measuring workplace was compiled using PBX Asterisk and OpenBTS USRP N210 with antenna. Mobile phones with support for 2G serving the end users. Logic diagram is shown in Fig. 4.

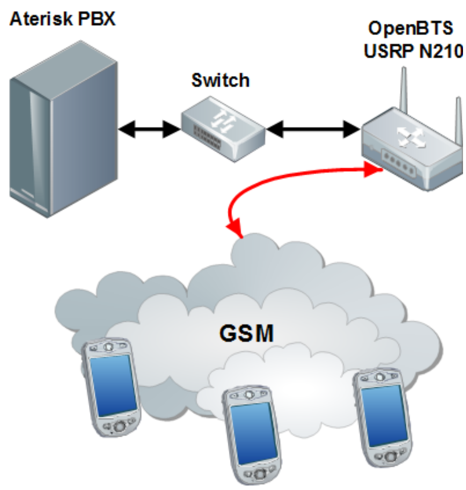

Fig. 4: Measurement schema for quality testing with components.

The N Building on the Department of Telecommunications, was measuring environment for testing voice quality transmitted through GSM created by OpenBTS.

Brick walls, doors and other building components formed barriers for propagation of the GSM network signal set up for this area. Each measuring point in the building has been strategically chosen to consider the barriers and the distance from the station. Measuring stations, five and their distribution show a plan of the ground floor of the building. The positions of the measuring points and station as shown Fig. 5.

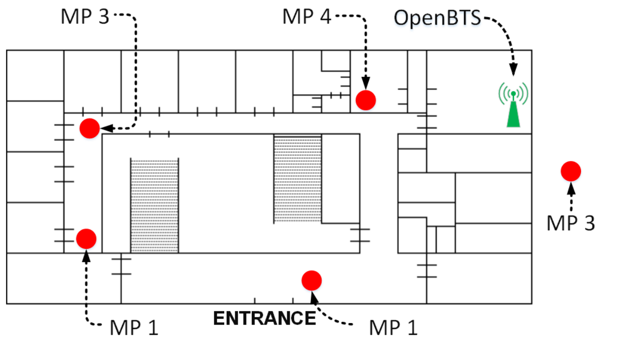

Fig. 5: First floor plan of the building N, where are marked measuring points (MB) and station (OpenBTS). Located in labN211.

6. Results

Measurement was performed simultaneously on two points that BTS had to manage at least two calls. Nokia 5530 and Nokia E52 were used for terminal equipment. At these devices was recorded degraded part of the test call, the end user evaluates the quality via DTMF signaling too. After the call was established, the devices started with recording. Asterisk PBX provide a “Call Back” with IVR questions. Answers were interpreted by DTMF values from 1 to 5 as mentioned earlier.

6.1. First Measurement Point – Entrance

First measurement place was selected into building near the entrance, as shown Fig. 5. The end user had not a direct view to the station, but all doors from station to point 1 were opened. Figure 6 represents results of PESQ and subjective IVR method.

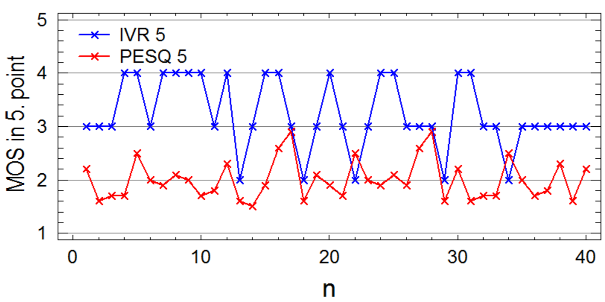

Fig. 6: Mean Opinion Score from PESQ and IVR methods for 1st measurement point.

6.2. Second Measurement Point – Hallway

The end user with device provides calls from hallway. Distance was about 35 meters from OpenBTS station. Device antenna has a direct view to the station, because doors on view were opened. Results from this point are shown in Fig. 7.

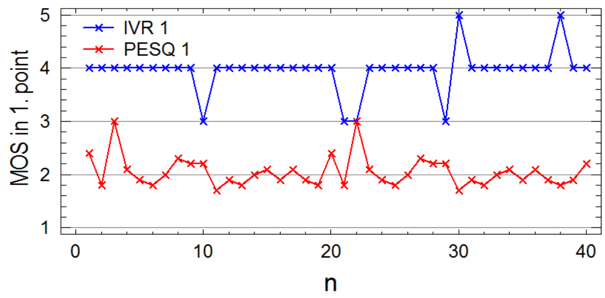

Fig. 7: Mean Opinion Score from PESQ and IVR methods for 2nd measurement point.

6.3. Third Measurement Point - Outside

The aim of this measurement was to compare the results of signal quality when call is from outside the building. Antennas of devices have not a direct view with each other. Moreover, external windows of labN211 have the windows aluminum blinds. Quality of calls is shown in results in Fig. 8.

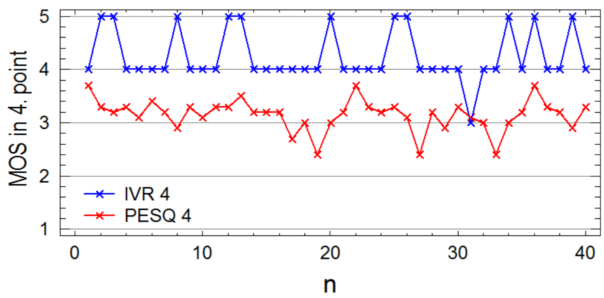

Fig. 8: Mean Opinion Score from PESQ and IVR methods for 3rd measurement point.

6.4. Fourth Measurement Point – Office

This measurement point was selected in office near the station. Closed doors prevent direct view. Although, the results were the best of all measurement points. The end user submitted very well values through IVR method. Course of evaluation is shown in Fig. 9.

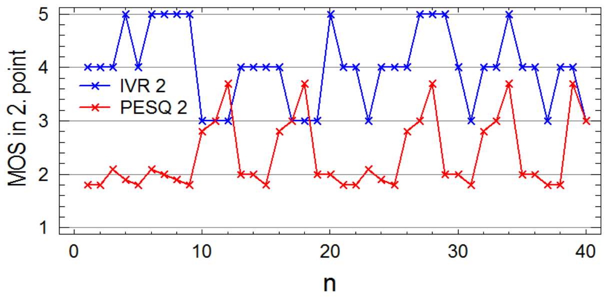

Fig. 9: Mean Opinion Score from PESQ and IVR methods for 4th measurement point.

6.5. Fifth Measurement Point – the End of Hallway

The fifth measurement point is located at the end of the hallway of the first floor. The end user is the largest distance from the transmitter overall. Therefore, the quality of signal has gone down, as shown in Fig. 10.

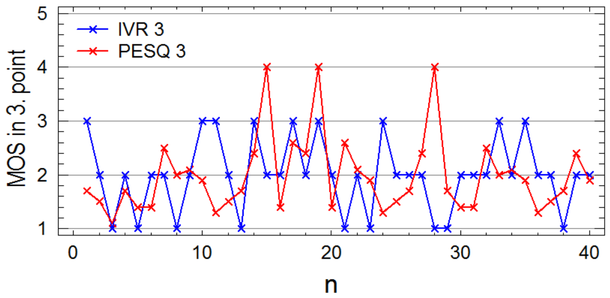

Fig. 10: Mean Opinion Score from PESQ and IVR methods for 5th measurement point.

7. Conclusion

This article shows how to use two methods to measure the quality of a call for GSM networks. It also evaluates the performance of the project OpenBTS in real operation. The idea that the OpenBTS used as an access point for complex areas is real.

To measure the quality of a call, that provided Asterisk and OpenBTS were two methods used. The first method which is used PESQ that using original and degraded, recording evaluate the quality of the connection. Subjective evaluation using IVR was chosen for the second method. Figure 11 shows a box-plot of all measuring points for the PESQ method.

Figure 12 shows results from DTMF signalling, which is results of IVR method.

Comparison Fig. 11 and Fig. 12 says that human evaluates the quality of the transmitted call differently than PESQ method. One reason is the imperfection of human hearing, which cannot detect small changes in

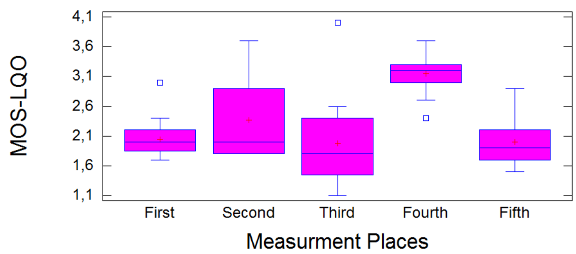

Fig. 11: PESQ method results for all measurements point. The Y axis explains mean opinion score from 1 to 5.

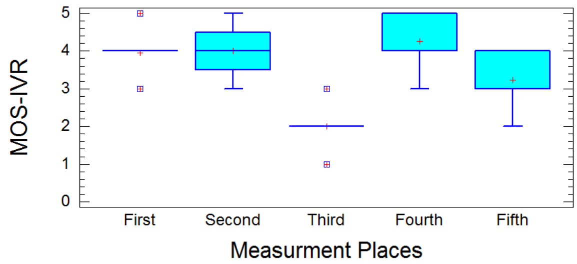

Fig. 12: The IVR method results for all five measure points. The Y axis explain a mean opinion score, same like previous picture.

quality. This error will be decrease with numbers of measurements probably. End device record call from start of dialing, therefore it was necessary to cut off the top of degraded recordings to pick up. This can be a second reason of result differences [9].

On the other hand, the similarity in the results is due to the individual measuring points. Human evaluates various points as PESQ method if we do not take into account the scale. This means that a human value is about 2 MOS score higher than PESQ calculating in the majority of measurements. Finally also been measured signal level in the building on the ground floor, as shown in Fig. 13.

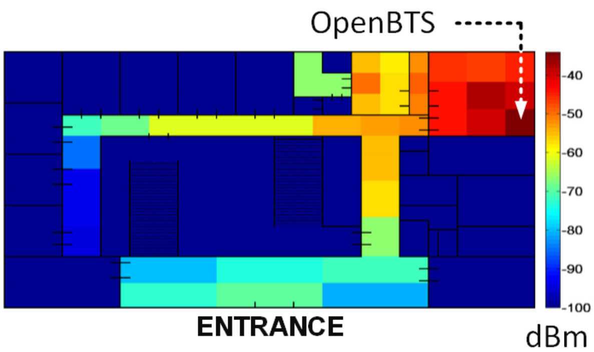

Fig. 13: Power of signal in the ground floor.

This graph shows a lack of performance when using one transmitting station. Logically, the strongest signal is close to the station. The signal level decreases with distance and buildings barriers.

This article presents a combination of two methods of evaluation the quality of voice services, and their possible implementation in practice. At the same time, it can be said that the project OpenBTS is an option for extending GSM networks in difficult areas, possibly to build a separate LAN GSM network.

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