Call Quality Measurement for Telecommunication Network and Proposition of Tariff Rates

Akram Aburas
School of Engineering, Design and Technology, University of Bradford
Bradford, West Yorkshire, United Kingdom

Professor J. G. Gardiner
School of Engineering, Design and Technology, University of Bradford
Bradford, West Yorkshire, United Kingdom

Dr. Zeyad Al Hokail
Electrical Engineering Department, King Saud University
Riyadh, Saudi Arabia

Abstract.
We present our latest and comprehensive research of our ongoing experiment of our work in this paper. The idea of our research is basically the measurement of call quality from the end users perspective and can be used by both end user and operator to benchmark the network. The call quality is measured based on certain call parameters such as Average Signal Strength, the successful call rate, drop rate, handover success rate, handover failure rate and Location Area Code (LAC). The mentioned quality parameters were derived from the active calls and the results have been analyzed and plotted for detail analysis and benchmarking. The critical results would be sent to the relevant personnel for proper action. We have also proposed the tariff plan for different the call quality measures for an operator.
Keywords: *Call Quality Measurement, Mobile Cellular Telecommunication Networks, Signal Strength*

1. Introduction

Traditionally, the speech quality was measured offline using subjective listening tests. The subjective testing is called Mean Opinion Score (MOS), it provides a numerical indication of the perceived quality of received human speech. The MOS would be expressed as a single number in the range 1 to 5, where 1 is lowest perceived quality, and 5 is the highest perceived quality. But subjective estimation at various sites within the coverage area is not practical as it is laborious, expensive and time-consuming. Most of the systems now a days use an automatic objective evaluation system to measure the speech quality.

Objective evaluation of the speech quality in systems is typically achieved by measuring the distortion between the input (transmitted) and output (received) speech signals. Processing steps typically include normalization of signals power, time alignment between input and output records, perceptual modelling and determining a distance value, which is used to estimate the equivalent subjective quality score.

Objective testing algorithms are also called automated quality measurement techniques. Three famous objective tests were Perceptual Speech Quality Measure (PSQM), Perceptual Analysis Measurement System (PAMS) and Perceptual Evaluation of Speech Quality (PESQ).
Currently, the most popular techniques are those based on psychoacoustics models, referred to as perceptual domain measures [5]. In these measures, speech signals are transformed into a perceptually related domain using human auditory models. Most available objective assessment techniques are based on an input-output approach [6]. In input-output objective assessment methods, the speech quality is estimated by measuring the distortion between an “input” or a reference signal and an “output” or received signal.

Currently there are a number of techniques that can be classified as perceptual domain measures. These include the Bark Spectral Distortion (BSD), the Perceptual Speech Quality Measure (PSQM), the Modified BSD (MBSD), the Measuring Normalizing Blocks (MNB), the PSQM+, the Telecommunication Objective Speech Quality Assessment (TOSQA), the Perceptual Analysis

Measurement System (PAMS), and most recently the Perceptual Evaluation of Speech Quality (PESQ) [6], which is specified by ITU-T recommendation P.862 [7], as the international standard for testing networks and codecs. In the case of input-output based speech quality assessment, good correlations were observed, which reaches up to 99% in some cases [8]. Correlation between the objective speech quality measure and the subjective quality measure is mostly used as the system (or method) performance measure.

The field of estimating the speech quality using only received speech without access to the input record is relatively new area. In 1994, Jin Liang and R. Kubichek [9] published the first paper in the field of output-based objective speech quality using perceptually-based parameters as the speech features. Their algorithm gave some good results in special cases achieving 90% correlation. R. Kubichek and Chiyi Jin [10, 11] used the vector quantization method which yields up to 83% correlation.

An output-based speech quality measure which uses only the visual effect of a spectrogram of the received speech signal, reported in [12]. A spectrogram is a two dimensional representation of time dependent frequency analysis, and contains acoustic and phonetic information of the speech signal. Framing the spectrograms into blocks and using digital image processing, the method achieved a reported correlation factor of 0.65 with the subjective score. Most recently, Gray et al [13] reported a novel use of the vocal-tract modelling techniques which enables prediction of the quality of a network degraded speech stream to be made in non-intrusive way.

A novel output-based speech quality evaluation algorithm is proposed in [14]. It is based on characterizing simultaneously the statistical properties of speech spectral density distribution in the temporal and perceptual domains. Results show that the correlations of the proposed algorithm with subjective quality scores attain 0.897 for the training data set and 0.824 for the testing data set, respectively.

Khalid Al-Mashouq and Mohammed Al-Shayee in [15] proposed a time-delay multilayer neural network model that can rate the speech quality after a proper learning stage. The learning set consists of features such as Linear Predictive Coefficients LPC and per-frame energy. A per-frame target is needed to train the neural network. This target is selected to be the Euclidian distance between the features vector of the clean and corrupted speech frames. The best correlation for speaker and text independent case reached to 0.87. However, the published result is not generalized due to limited number of speech files that had been used.

In this Paper we proposed a new scheme which is for measuring speech quality over the network from the subscriber set and analyzing the quality in detail. We obtain the signal strength information of an active call every 5 milliseconds, signal information processing and plotting based on aggregation of successful and un-successful calls and their related scores on the map. The critical scores were sent to concerned authorities and finally the new rating schemes were proposed for different quality measures

2. THE EXPERIMENT

The whole system is divided into four individual processes namely signal strength measurement, signal strength statistics, plotting landmarks and SMS signal information.

The following is basic outline of our research approach in carrying out the experiment in collecting and analyzing the signal quality of an active call. Figure 1 depicts the flow of the basic outline.

1. Upon a call being setup the signal level values are taken every 5 ms.
2. The following criterion in table 1 below is used to decide on the quality of the signal.
3. A score is then given to the sample collected every 5 ms; with 5 being the best on a scale of 5; and 1 indicating very bad signal strength.
4. At the end of the call session a cumulative score is computed and based on the score (ranging from 1 to 5); the speech quality is approximately computed.

The classification of signal strength is as follows:

Signal Level Range (dBm)	Classification
-120 to -95	Extremely Bad
-95.00 to -85.00	Bad
-85.00 to -75.00	Average
-75.00 to -65.00	Good
-65.00 to -55.00	Very Good

Table 1: Perceived quality rating of human speech

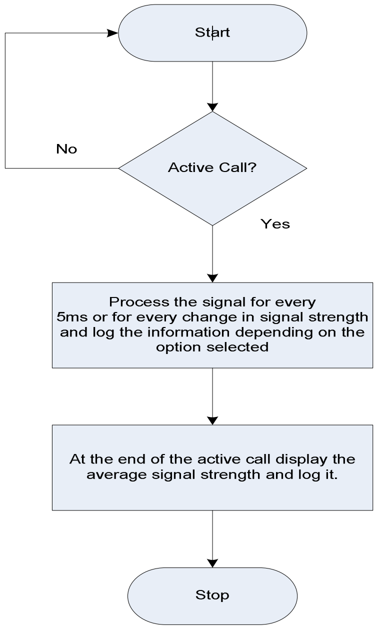

Figure 1: Signal Strength Measure

The system works during the active call, which also records the GPS coordinates during the call. The system has provision of auto-start which helps the user to log and locate the information of all the active calls on the map, once the auto-start is enabled. This reduces the problem of starting the application every time during the active call. This also helps us to analyze the signal quality at any location for a given operator. It supports both internal and external GPS connected through Bluetooth. The captured GPS coordinates were used for plotting the average signal strength on the map. Figure 2 depicts the process flow of plotting landmarks with the measured signal strength information.

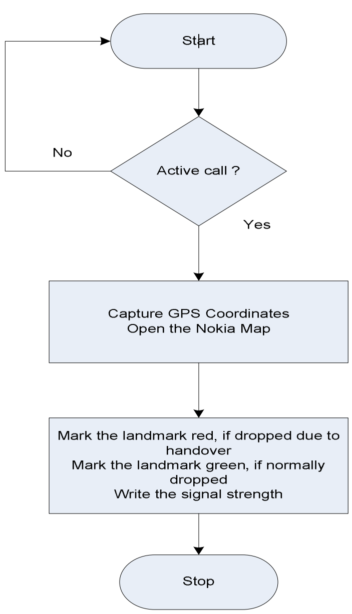

Figure 2: Plotting Landmarks

The landmarks that were marked with red colors are the calls dropped due to handover and the landmarks that were marked with green colors are normally dropped calls. The different colors landmarks help one to easily visualize and analyze the calls.

The system records the number of successful and un-successful call attempts made for every ten call attempts. The successful and un-successful call attempts are classified based on whether the call is successfully connected by the network. The call drop information such as normally dropped from either of the party or dropped due to handover during the cell change is also recorded. The number of normal dropped and handover dropped with there average scores also recorded. Figure 3 below depicts the process flow of signal strength statistics.

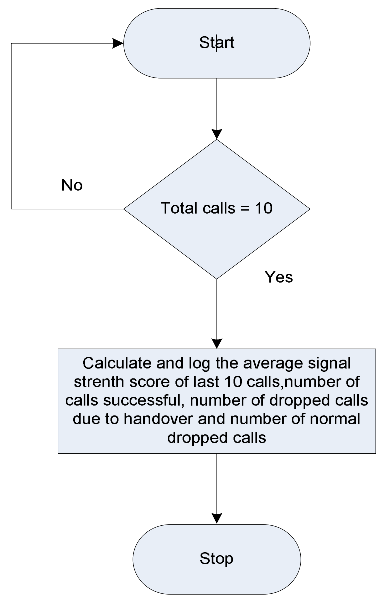

Figure 3: Signal Strength Statistics

The system has the ability to send the signal strength information to the particular number. It has the provision of setting the mobile number, to which the sms would be sent automatically at the end of call. The system has the option of setting to send the sms always, less than bad etc. at the end of 10 calls, the call statistics would also be sent as sms. Figure 4 depicts the process flow of SMS signal information.

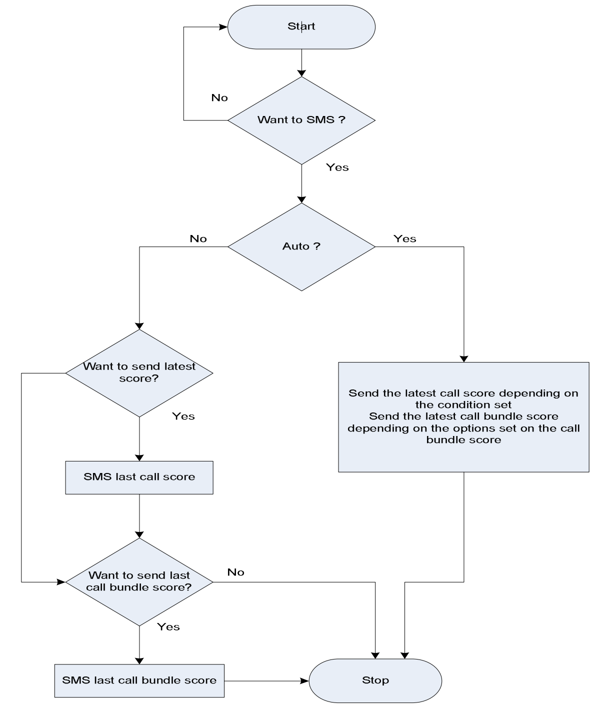

Figure 4: SMS Signal Information

The below Figure 5 illustrates the complete process of signal meter system.

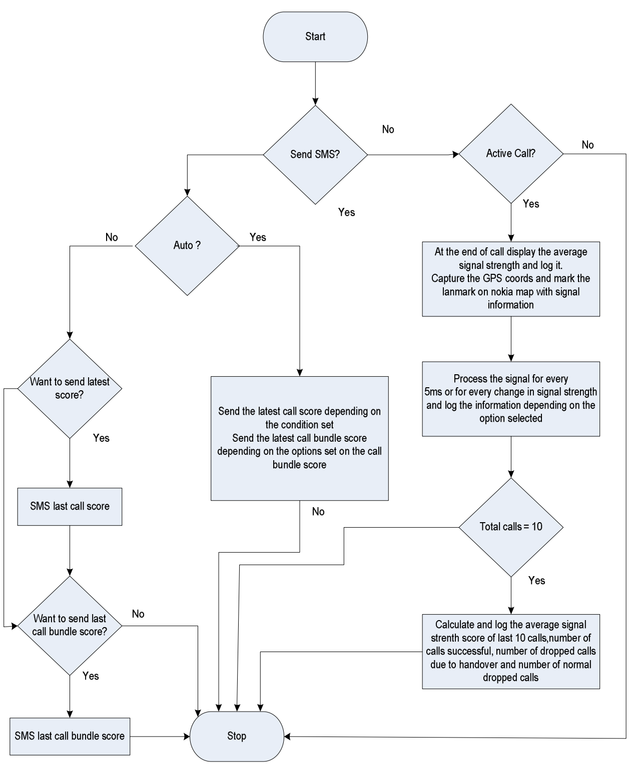

Figure 5: Signal Meter flowchart

3. CHARGING RATES VS QUALITY

The charging rate proposed Table 2 is based on perceived signal quality on the customer equipment. The charging rate can be made more complex by including the call statistics as well. The charging rates proposed in table 2were based on our assumption that if the operator is charging the customer of SR 2.00 per minute for the highest rating quality five.

Table 2 : Basic Charging Rate

RATING	PERCEIVED QUALITY	SIGNAL STRENGTH IN DECIBELS	CHARGING RATE IN SAR
5	Very Good	- 65.00 to - 55.00	2.00
4	Good	-75.00 to - 65.00	1.50
3	Average	-85.00 to - 75.00	1.00
2	Bad	-95.00 to - 85.00	0.25
1	Extremely bad	120.00 to 95.00	0.1

The below Table 3 is the new charging rate proposed based on calls successful, dropped due to handover. The formula used is charging is:
Successful_call_attemps*charging_rate – number_of_handover_dropped*0.2.
The customer would be charged base on the average call quality of the successful call attempts minus 0.2 halala deducted per each call that is dropped due to handover.

Table 3: Proposed charging rate Vs Successful calls in the bundle

Rating	Charging Rate in SAR	No. of normally dropped calls	No. of calls dropped due to handover	No. of unsuccessful attempts	Charge
5	2.00	8	2	0	16-0.4
4	1.50	7	2	1	7*1.5-2*0.2
3	1.00	8	1	1	8-0.2
2	0.25	10	0	0	10*0.25
1	0.1	6	2	2	6*0.1-2*0.2

Table 4 : Proposed Charging Rate Vs Quality of the bundle of ten calls

Rating	Charging Rate in SAR	No. of normally dropped calls	No. of calls dropped due to handover	No. of unsuccessful attempts	Charge
5	2.00	8	2	0	16-0.4-0
4	1.50	7	2	1	7*1.5-2*0.2-1*0.1
3	1.00	8	1	1	8-0.2-1*0.1
2	0.25	10	0	0	10*0.25-0
1	0.1	6	2	2	6*0.1-2*0.2-2*0.1

Table 4, the penalty is applied even in the case of the unsuccessful call attempt. The unsuccessful call attempt is defined as the network problem or network busy, where the user is unable to connect or call. Every unsuccessful call attempt in a bundle will have 0.1 hala penalty to operator. Table 5 contains the sample data of the proposed charging rate.

4. CONCLUSION

Analysis of speech quality is a major research going on for current wireless networks. This research involves investigation of metrics which are used to detect the call, obtaining the signal strength information and analyzing the signal information from various perspectives. The proposed methodology can be used to benchmark the mobile network by the user and hence it can be used as a base for charging the customer by the operators. The charging rates were proposed in this work based on the signal quality and the call statistics recorded.

5. REFERENCES

[1] J. Anderson, “Methods for measuring perceptual speech quality,” *Agilent Technologies-White Paper*, USA, May 2001.

[2] ITU-T Rec. P.862, "Perceptual Evaluation of Speech Quality (PESQ), An objective method for end to end speech quality assessment of narrowband telephone networks and speech codecs,” 2001.

[3] Aruna Bayya and Marvin Vis. "Objective measure for speech quality assessment in wireless communications," *Acoustics, Speech, and Signal Processing, ICASSP-96*, IEEE International Conference 1996, vol.1, p-p. 495-498.

[4] Jin Liang and Robert Kubichek, “Output-based Objective Speech Quality," *Vehicular Technology Conference,* 1994 IEEE 44th vol.3, p-p. 1719-1723.

[5] Chiyi Jin and Robert Kubichek, "Vector Quantization Techniques for Output-Based Objective Speech Quality," *Acoustics, Speech, and Signal Processing, ICASSP-96*, IEEE International Conference 1996, vol.1, p-p. 491-494.

[6] Chiyi Jin and Robert Kubichek, "Output-Based Objective Speech Quality Using Vector Quantization Techniques," *Signals, Systems and Computers, Conference Record of the 29th Asilomar Conference*, IEEE 1995, vol.2, p-p. 1291-1294.

[7] O.C. Au and K. H. Lam “A Novel Output-Based Objective Speech Quality Measure for Wireless Communication,” *IEEE Proceedings of ICSP ’98*, Vol. 1, p-p. 666-669, Beijing, China, Oct. 1998.

[8] P. Gray, M. P. Hollier and R. E. Massara, “Non-Intrusive Speech-Quality Assessment Using Vocal-Tract Models,” *IEE Proc. – Vis. Image Signal Process.,* Vol. 147, No. 6, p-p. 493-501, 2000.

[9] Chen, G. and Parsa, V. "Output-based speech quality evaluation by measuring perceptual spectral density distribution," *IEE Electronics Letters*, 40, p-p. 783-785, 2004.

[10] D. Picovici and A.E. Mahdi, “Output-based objective speech quality measure using self-organizing map,” *IEEE Proceedings of ICASSP-2003*, vol. 1, p-p. 476–479, 2003.

[11] Khalid A. Al-Mashouq and Mohammed S. Al-Shaye, “Output-Based Speech Quality Assessment with Application to CTIMIT Database,” *17th International Conference on Computers and Their Applications* CATA-2002.

[12] Gaurav Talwar and Robert F. Kubichek, “Output-based Speech Quality Measurement Using Hidden Markov Models,” *GSPx Conference*, 200

[13] L. Rabiner and B. Juang, *Fundamentals of Speech Recognition*. NJ: Prentice-Hall Inc, Englewood Cliffs,1993

[14] L. R. Rabiner, "A tutorial on Hidden Markov Models and Selected Applications in Speech Recognition," *Proceedings of the IEEE*, vol. 77, no. 2, pp. 257-296, 1989

[15] Pietro Paglierani, “Uncertainity Evaluation of Objective Speech Quality Mesurement in VOIP systems,” IEEE Transactions on Instrumention and measurement Vol.58 No.1, Jan 2009