Estimation of Equipment Impairing Factor of GSM Codec in Smartphone to Determine VOIP Quality Transported by 3G/4G Service

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Abstract

Business activities need excellent voice communication quality to avoid misunderstandings between the parties. Currently voice communications is embedded in data communication and related to Internet connections. End devices include IP based telephones set and the ubiquitous smartphones that are now Internet connected as well. Smartphones are cellular based, but technological evolution and customer demand leads to Internet data services. Internet enables the change of communications from voice centric to data centric communications. Traditional analog voice communication is superseded by digital technique that resulted in digital voice. Internet enables the ease of connectivity of digital data. As digital voice is now data, Voice over IP (VOIP) can be part of data service in Internet. Voice can now be embedded in Internet based services. Businesses currently do not need separate voice and data network. The acceptance of technology depends on the quality of service perceived by the customers. In voice communications ITU issued has recommendation for telephony quality known MOS (Mean Opinion Service). It is interesting to have a simple method to measure MOS of voice communications. The MOS should provide an indication if the service is acceptable or inadequate for business use. The MOS indicator will be useful for business entities especially small and medium enterprise that are operating from premises with only cellular network as their Internet access choice. The ITU recommendation provides detailed variables to measure the MOS. This research determines equipment impairment factor that affects R factor that determine MOS based on data obtained from measurements.

Keywords
VOIP, MOS, Data Service, Impairment, voice centric

1. Introduction
TCP/IP based networks currently carrying digitized voice traffic. The digitized voice become the load of IP packets. As IP packets are carrying voice, the voice service is known as Voice over IP (VOIP). Internet it created to function as data network. Internet then evolved to carry whatever information that are digitized and packetized as IP packets. Real time voice in the meantime can be digitized and packetized as IP packets. Voice over IP (VOIP) emerge as solution converge voice communication network and data communication network. Unifying the two network increases the efficiency of business operations. Internet itself as data network was not developed to handle in real-time information. Parameters that typically determine real time quality such as delay, packet loss and jitter are not of particular importance. There must be additional effort in order that these parameters can be controlled in order to provide the required quality of real time data. VoIP is one of real time application that are running on IP. It is important that voice communication should have certain acceptable quality for intelligible conversation. In voice communications the quality of intelligible is better known as Quality-of-Service (QOS). It largely depends on the infrastructure of the communication network. Before the proliferation of cellular network, the communication infrastructure is based on copper cable. Not only communication networks but also data network including Internet is copper based. The versatility of Internet demands higher transmission data transmission speed and naturally bandwidth. The demand is fulfilled by choosing optical cable. Copper and optical cable are known as fixed media and users depend on a fixed geographical location ta access the global Internet network. If there is no infrastructure is available to access the Internet, the network has is no global interconnection. It can only have intranet.

At almost concurrently cellular networks that enable user mobility rise exponentially. The number of users are almost limitless. For practical purposes it can be assumed that everybody have access to cellular devices known as smart phones. Consequently, business entities must provide always accessible service. Start-up businesses are thriving due to innovation and creativity respond to market demands and operational efficiency. Start-up and small businesses are often mobile to efficiently approach their market. High speed 3G and 4G service can provide solution to the demand of the market. 3G and its successor 4G provide ubiquitous communication service due to nationwide coverage. Internet service provided by 3G and increasingly 4G enables global reach. Even though 3G and in particular 4G is emphasizing data services that are demanded by most of their user applications, real time data such as voice and video services are increasingly are part of the user applications. The global pandemic accelerates the digital transformation where real time voice and video services the primary demand as shown by applications such as Zoom, Webex and others.

With this research is thus interesting to find out the adequacy of mobile communications for business purposes in particular micro and small businesses and the effect of devices on the quality. The identified problem in economic voice mobile communications is the degradation of voice quality if voice is part of Internet data communications such as popular smartphone applications. Cellular communications quality is well controlled and maintained by the operators, but voice communications embedded onto applications is beyond the realm of operators. Operators only maintain the quality of Internet access infrastructure. Knowing the effect of device impairment impact, the concerned parties would be able to use the suitable smart devices for their business activities purpose.

1.1 Objectives

In this research, 3G service is the infrastructure to access the Internet to the Cloud where the above mentioned application resides. There were several studies concerning the quality of service of VoIP compared to the traditional analog voice communications. In the voice communication ITU-T provides its recommendation on assessing Quality of Service of voice communications. It is known as MOS (Mean Opinion Score) as ITU-T P.800.1 (ITU-T 2016). In analog communications it is measured subjectively by means of survey in a certain set-up condition in ITU-T P.800 (ITU-T 1996). In packetized or digital voice communication the MOS is measured objectively (ITU-T 2015). The objective MOS can then be mapped to subjective MOS (ITU-T 2015) in order to have an idea that is familiar to the users.

To measure objectively MOS of IP packetized digital voice, this research use a model as specified in ITU-T Recommendation G.107 (ITU-T 2015). The IP packetized digital voice is running on 3G infrastructure. Majority of researches monitor (Manousos et al. 2005), (Assem et al. 2013) voice quality of digitized voice calls that are carried by cable based infrastructure (Markopoulou et al. 2002). The method of the ITU based E-model (Walker 2002), (Daengsi and Wuttidittachotti 2015) to evaluate of voice quality running on the network are its measurable parameters. The obtained model will provide a simple method to arrive to an objective MOS on a 3G data network service. The VoIP in this research is not embedded as part of application but VOIP to VOIP communication handled by SIP Protocol.

2. Literature Review

Voice communication service call quality is rated with Mean Opinion Score or MOS. MOS of conventional voice communication or telephone call is not an objective measurement. The Figure 1 is obtained from the opinion of telephone users samples following the procedure as recommended by ITU. ITU-T Recommendation P.800 (ITU-T 2016) defines how the subjective measurement of analog telephone calls should be performed. ITU-T Recommendation P.800 formalized the voice quality of analogue voice communications. P.800 provides a parameter defined as Mean Opinion Score or MOS. MOS has five-point scale called Excellent, Good, Fair, Poor, and Bad. The scales are classified according to controlled conditions as defined in ITU-T P.800. MOS is internationally accepted metric. It shows voice quality as perceived by users (Raja et al. 2007).

Voice over IP (VOIP) that is carried on data network and Internet often replaces conventional telephony. VOIP quality are rated using a single metric number and called MOS as well. Both metric numbers should present the same perception of a telephone call quality. VOIP has many contributing factors to the quality of its calls as it is transported on public data network. Many of the contributing factors can be objectively measured. Due to the digital nature (Jelassi et al. 2012) of the contributing factors they can be objectively counted. ITU Recommendation G.107 (ITU-T 2015) recommends a model to evaluate objectively the quality of VoIP calls. The model is called E-model and it provides a metric number known as R-value. R-value depends on packet delays and equipment impairment factors. R-value is then mapped to an estimated conventional MOS value to arrive to the quality perception of the call.

Digital voice communications enables the measurement of parameters relevant for quality of service objectively. The parameters can be obtained and calculated from the underlying network. The parameter can approach the traditional MOS. Objective methods are simple to estimate voice communication quality of a business entity corporate network. Objective methods are carried out by machines without involvement of human listeners. The MOS estimates correlate well with the subjective test. Efforts are focused on objective methods to measure the voice quality of VoIP. Objective methods are based on whether a reference signal is needed or intrusive method or without reference signal called non-intrusive methods (ITU-T 2016), (ITU-T 2001). Hence; the MOS of voice communications can be obtained using either intrusive or nonintrusive measurements.

In intrusive methods a reference signal introduced into the network where the digital voice signal is carried. It is then compared to the carried digital voice signal being measured its MOS. Various methods were proposed to define the difference between these digital voice signals. ITU-T has issued P.862 PESQ (ITU-T 2001) to arrive to acceptable estimates of MOS scores. ITU-T specifies the methods in its recommendation ITU-T P.862. ITU-T P.862 specifies intrusive method to find voice quality of VoIP applications. As ITU-T P.862 is an intrusive method, it needs reference data. The major disadvantage intrusive methods is the requirement of a reference digital voice signal for comparison. In intrusive methods some factors which could affect the conversational voice quality are not reflected these methods. It is possible even measuring a high MOS, but the perception of quality is not as it should.

The non-intrusive methods are based on some key parameters and statistics of the network and received signals. The original signal quality is unknown. The impairment parameters are mapped to MOS. Network parameters could affect the resulting quality and may be used to supplement the analysis. E-model are based on the network parameters alone. In VoIP it could be used obtaining baseline network performance and troubleshooting in case the MOS is degraded. Within non-intrusive methods, there are several principles such as speech layer, packet layer and opinion principles. The speech layer is standardized in ITU-T Rec. P.563 (ITU-T 2004), whereas the opinion principles is defined the E-model.

Non-intrusive techniques evaluate voice quality directly from the network. Parameters are collected and calculated to obtain quantitative metrics. Major parameters are packet loss, delay, and jitter. E-model in ITU-T Recommendation G.107 is a computational model to that can provide an objective voice quality metric based on network and other system parameters. Objective method determines voice quality without considering human perception factor. Research such as reported in (Assem et al. 2013), (Walker 2002), (Cole and Rosenbluth 2001) proposed a simplified E-*model* for determining voice quality. The researches were based on voice packets that were transmitted through guided media. More important is the end to end VOIP quality where access network play a major role. The common protocol of VoIP over access networks is Session Initiation Protocol (SIP). Modified E-model or also known as simplified E-model is sufficient for practical purposes for assessing the quality of VOIP call. ITU–T E-model is adapted for evaluating packetized voice in TCP/IP network. It helps in estimating various impairments to voice quality. The model includes equipment impairments, delay, loss and jitter over packet networks. Simplified E-model produces R-factor which is an objective quantitative measure of voice quality. The Mean Opinion Scores (MOS) is then calculated based on the obtained R-factor. R-factor which ranges from R=100 (best case) R=0 (worst case).

3. Methods

The R-factor is mapped to MOS figure. MOS has 5 scores, where 5 is defined as excellent, 4 is good, 3 is fair, 2 is poor and 1 is bad. Mapping of R factor to MOS is shown in Table 1. Table 1 is created based on (Walker 2002), (Cole and Rosenbluth 2001)

Table 1. MOS VoIP

Quality	Score
Excellent	4.3-50
Good	4.0-4.3
Fair	3.6-4.0

Poor	3.1-3.6
Bad	2.6-3.1
Not Recommended	1.0-2.6

R-factor is obtained from several parameters associated with voice telephone communications. The parameters of R-factor according to in ITU-T G.107 consists of various impairment factors. Mapping R-factor to the MOS is as following, $R < 0$ is mapped to MOS = 1, $R > 100$ to MOS = 4.5, if $0 < R < 100$. ITU-T Rec. G.107 defines

$$R = R_o - I_s - I_d - I_e + A \ldots.. (1)$$

R_o is signal-to-noise ratio (SNR), I_s is impairments with the voice signal; I_d is impairments due to delay and I_e represents impairments due to equipment and in particular codecs; and A is the advantage factor as user in using a given technology. If the codec is well-known to determine I_e, and network delay and loss are captured, R factor will be known. Simplified R-factor as proposed by Cole and Rosenbluth (2001), (ITU-T 2004) is applicable to packet network. The impairments of VOIP digital signal now is simplified to an R expression that has only delay impairment factor I_d and equipment impairment I_e as its major parameter and a constant factor (Cole and Rosenbluth 2001).

$$R=94.2 - \{I_d\} - \{I_e\}) \qquad (2)$$

The delay impairment factor I_d is the property of data network. In this research the delay are based upon network delay that is measured by Wireshark (Chappel 2012).

4. Data Collection

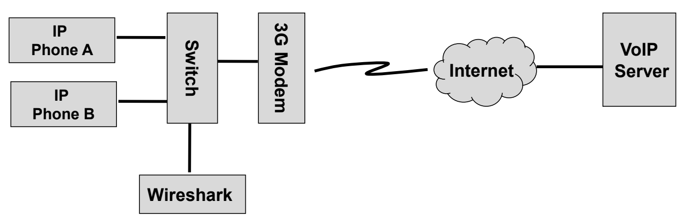

Figure 1. Measurement Diagram

To find equipment impairing factor it is important notice that VOIP is using low bit-rate codecs. The codec has distortions and become the major contributor of equipment impairing factor. Hence, the equipment impairment factor I_{ef} is mainly due low bit-rate codecs. I_e values can be determined according to ITU-T P.833. ITU-T-G.113 lists various I_e values of several codec.

5. Results and Discussion

Simplified E-model gives R-factor that can be mapped to MOS. R-factor itself shows among other equipment impairment factor as a major contributor of the decrease of MOS. The equipment impairment factor depends on the codec being used. There are already results to account several types of codecs except GSM codec. In this research the GSM codec equipment factor is estimated in 3G environment. It would of interest if the equipment impairment factor need significant adjustments for simplified E-model.

5.1 Measurements Results

To find GSM codec equipment impairment factor estimation in 3G environment measurement the diagram as shown in Figure 1 was used. Wireshark collected the parameters of interest that are necessary to find the equipment impairment factor. Parameters of interest are delay time and packet loss. Both parameters are used to calculate the R factor and mapped to MOS. The results is shown in Table 2 and Figure 2 for IP phones with GSM codec and Table 3 for IP phones with G711 codec. The samples were taken during working days and per day there were 5 sampling periods. The sampling periods were period 01 between 10:00 -12:00, period 02 between 12:00 -14:00, period 03 between 14:00 - 16:00, period 04 between 16:00 – 18:00 and period 5 between 18:00 – 20:00.

Table 2. Measurement result for GSM codec

Period	Time Delay (msec)	Packet Loss	R Factor	MOS
1	40	1%	68	3.5
2	40	2%	68	3.5
3	44	15%	42	2.3
4	44	13%	46	2.5
5	40	0%	71	3.6

Table 3 shows that the total delay was less than 45 msec, but the total packet loss at certain period were more than 10%. The R factor and its mapping to the more familiar MOS figure classified the voice quality as poor (MOS 3.1-3.6),

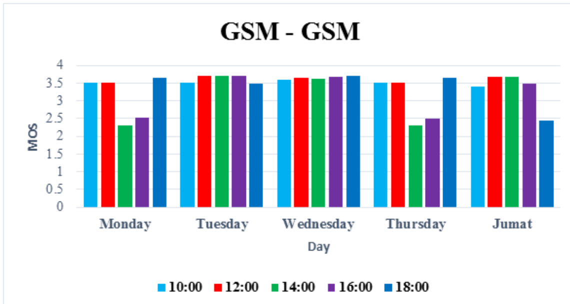

Figure 2. GSM to GSM Measurement Results

Figure 2 shows the graph of measurement result taken for one week. There are periods and days of which the voice quality decreases. It is understandable as the VoIP communication was through public Internet where traffic load was unpredictable.

Table 3. Measurement result for G711 codec

Period	TD (msec)	Packet Loss	R Factor	MOS
1	40	2%	87	4.2
2	39	0%	92	4.4
3	40	0%	92	4.4
4	40	0%	92	4.4
5	41	1%	89	4.3

Table 3 is the measurement of VoIP communication between with G711 codec. It showed that time delay were less than 50 msec and packet loss was basically less than 5%. R factor with these parameters were close to 90. Mapping the R factor to MOS showed that MOS were above 4.2 or the quality is good

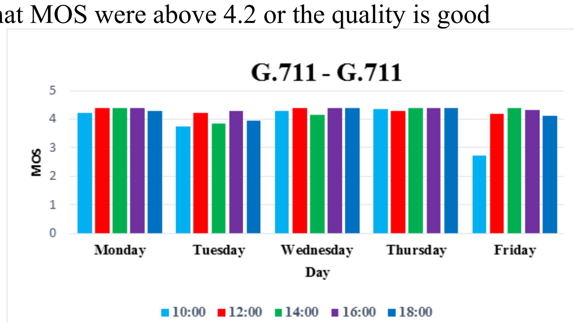

Figure 3. G711 to G711 Measurement Results

Figure 3 shows the graph of measurement result of g711 codec taken for one week. G711 codec provides good MOS for voice communications. It is understandable as G711 does not compress the digital voice, but it need a wide bandwidth which is normally 64 kbps. GSM codec the digital voice and need less bandwidth to 13 Kbps.

5.1 Numerical Results

The packet loss increases I_e and decreases MOS. The formula being used is (ITU-T 2004):

$$I_{ef} = L_1 + L_2 \ln(1 + L_3) \qquad (3)$$

According to the formula

$$I_{ef} = L1 + L2 \ln(1 + L_3e) \qquad (4)$$

Where the known values of L_1, L_2, L_3 for codec G729a are L_1=11, L_2=40, and L_3=10. For codec G711 its values are L_1=0, L_2=30, and L_3=15. In in G.713 the estimated value are L_2=25 and L_3=14. The estimation for GSM is calculated based on its codec rate of 13 kbps with L_2= 30, L_3=15 for G.711 with codec rate of 64 kbps and L_2 = 40, L_3 = 10 for G.729a with codec rate of 8kbps.
With codec rate difference of 56 kbps then the differences in L_2 = -10 and L_3 = 5. For that for 1 kbps difference in codec it is estimated that the rate change for L_3 equal to (15-10)/56. L_3 of GSM is then estimated to become 10 + (13-5) (5/56) = 10.45 or L3 = 10. Meanwhile; the L_2 change in codec rate per kbps is equal to (30-40)/56. L_2 of GSM is estimated to 40 - 0.89 or 39. Hence for GSM codec I_e can be simplified as [13]:

$$I_e = 20 + 39 \ln (1 + 10e) \qquad (5)$$

Estimated GSM Half Rate with C/I (carrier to interference) 10 dB has I_e range between 25 and 32, GSM Full Rate has I_{ef} range between 32 and 39.

6. Conclusion

Simplified E-model gives R-factor that can be mapped to MOS. R-factor itself shows among other equipment impairment factor as a major contributor of the decrease of MOS. The equipment impairment factor depends on the codec being used. There are already results to account several types of codecs except GSM codec. In this research the GSM codec equipment factor is estimated in 3G/4G environment using smartphones. It would of interest if the equipment impairment factor need significant adjustments for simplified E-model.

References

Assem, H., Malone, D., Dunne, J. and O'Sullivan, P., Monitoring VoIP call quality using improved simplified E-model, *2013 International Conference on Computing, Networking and Communication,* pp. 927-31, San Diego, USA, January 28 - 31, 2013

Chappel, L., *Wireshark Network Analysis: The Official Wireshark Certified Network Analyst Study Guide.,* 2nd Edition, Chappell University, 2012

Cole, R.G., Rosenbluth, J.H., Voice over IP performance monitoring, *ACM SIGCOMM Computer Communication Review,* vol.31, no. 2, pp. 9-24, 2001

Daengsi, T. and Wuttidittachotti, P., QoE modelling: A simplified e-model enhancement using subjective MOS estimation model. *2015 Seventh International Conference on Ubiquitous and Future Networks*, pp. 386-390, Sapporo, Japan, July 7-10, 2015

ITU-T, *Mean opinion score (MOS) terminology Recommendation ITU-T P.800.1,* ITU-T Telecommunication Standardization Sector of ITU, Geneva, 2016

ITU-T, *Methods for objective and subjective assessment of quality Recommendation ITU-T P.800,* ITU-T Telecommunication Standardization Sector of ITU, Geneva, 1996

ITU-T, *Objective measuring apparatus Single-ended method for objective speechquality assessment in narrow-band telephony applications. Recommendation ITU-T P.563,* ITU-T Telecommunication Standardization Sector of ITU, Geneva, 2004

ITU-T, *Perceptual evaluation of speech quality (PESQ): An objective method for end-to-end speech quality assessment of narrow-band telephone networks and speech codecs. Recommendation ITU-T P.862,* ITU-T Telecommunication Standardization Sector of ITU, Geneva, 2001

ITU-T, *The E-model: a computational model for use in transmission planning. Recommendation ITU-T G.107,* ITU-T Telecommunication Standardization Sector of ITU, Geneva, 2015

Jelassi, S., Rubino, Melvin, H., Youssef, H. and Pujolle, G., Quality of experience of VoIP service: A survey of assessment approaches and open issues. *IEEE Communications Surveys & Tutorials,* vol. 14, no. 2, pp. 491-513, 2012

Manousos, M,, Apostolacos, S., Grammatikakis, I., Mexis, D., Kagklis, D. and Sykas, E. Voice-quality monitoring and control for VoIP, *IEEE Internet Computing*, vol. 9, no. 4, pp. 35-42, 2005
Markopoulou, A.P., Tobagi, F.A. and Karam, M.J., Assessment of VoIP quality over Internet backbones. *Proceedings of the Twenty-First Annual Joint Conference of the IEEE Computer and Communications Societies,* pp. 150-9, New York, USA, Jun 23-27, 2002
Raja, A., Azad, R.M.A, Flanagan, C. and Ryan, C., Real-time, non-intrusive evaluation of VoIP, *European Conference on Genetic Programming,* pp. 217-228, Valencia, Spain, April 11-13, 2007
Walker, Q., Assessing VoIP call quality using the E-model, *NetIQ Corporation,* 2002

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