Voice Quality Assessment of SIP-PBX Softphone Extension in 3G Cellular Service Environment

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Abstract. Voice Communications is still playing important role in business activities. Large companies generally use PBX to handle corporate communications both internally and externally. PBXs have rich features that can fulfil corporate voice communications requirements. Internet changed the corporate communications importance from voice centric to data communications. Intranet becomes the strategic data infrastructure for internal communications and its connection to Internet is an absolute must to have global reach. Voice over IP (VOIP) is the technology that capitalizes the ubiquitous presence of data network in corporates. VOIP provides efficiency in business communications. Naturally, traditional PBX evolves to IP-PBX that use data network which is currently based on TCP/IP for certain connections. The acceptance of SIP (Session Initiation Protocol) as the protocol for VOIP resulted in improvement of IP-PBX to become SIP-PBX. One of the SIP-PBX that is Open Source is called Asterisk. It has rich features found in high end corporate PBX. In this research Asterisk is used as the basic PBX for small corporates in fulfilling corporate voice communications. Sometimes small companies have to operate in locations that have no proper and reliable fixed or cable infrastructure for Internet connection. The only economic viable choice is 3G or 4G cellular service. The problem is whether the service quality is suitable. ITU has recommendation for planning of telephony transmission quality. The parameter is known as MOS. The recommendation shows detailed variables to reach the MOS. Based on the ITU recommendations; this research provides a novel technique to quickly assess the voice quality of internal voice corporate communications using SIP-PBX that is located in the Internet. The data used for evaluating MOS can be directly obtained using Wireshark as the tool. The resulting MOS used only to assess whether the 3G or 4G service in that particular location is suitable for small corporate communications or other solution should be found. This research result showed that based on MOS, 3G access line quality is inadequate for VOIP based voice communication.

1. Introduction
The IP Networks currently carrying all types of data traffic. Voice traffic has evolved to digital data and packetized to IP packets. The applications will collect the packets in the network and reassemble. If the packets payload is packetized voice, the reassembly will result in voice. The technique is known as Voice over IP (VOIP). Unfortunately IP Networks are not intended to support real-time voice communication. It is basically a data network where some parameters are not as critical as in real-time information. Parameters such as delay, packet loss and jitter determine voice quality. To provide
voice communication a network should have acceptable quality in order to have intelligible conversation. A major challenge is how to measure voice quality efficiently [1]. Quality-of-Service will depend on the infrastructure of the network. Traditionally the infrastructure is based on copper cable especially Ethernet complying cables. Due to the demand of bandwidth and speed optical cable is increasingly the media of choice. Both copper and optical cable is known as the fixed media. Users are bound to a fixed location where the facility is available. Users are not free to choose locations. An interconnected network with access to the global network Internet will depend on the physical availability of the copper cable or optical cable that are connected to the Internet access point. If no infrastructure is available, a local or only intranet can be built. The network is without access to Internet and consequently no global interconnection.

The current business state demands ubiquitous communication service. Small businesses are sprouting due to its agility, innovation speed in handling business demands and operational efficiency. Small businesses are often nomadic or mobile due high cost of office space. Cellular communication such as 3G service is a great solution. 3G service and its evolutionary successor 4G can be considered as ubiquitous communication service as its coverage is nationwide. The service is accessible almost anywhere. It is practically economic. 3G or 4G service has still potentials that need further studies for innovation. In this research, 3G service will be used as access line to a SIP-PBX instead of copper cable. In order that SIP-PBX could function according to its configuration the Quality of Service of the communication line must be assessed. The problem is how to assess the connection quality to the SIP-PBX accurately yet economically. The assessment will provide a solution for the go or no-go decision in that particular SIP-PBX location and extended to which service provider will provide the best service [1].

It is interesting to have a model to evaluate the voice quality of a corporate IP infrastructure where voice over IP is running on it. The model should evaluate conversational voice quality service. There are a number of researches to monitor [1][2] or predict voice quality of VoIP calls but mostly on cable based medium. The resulting model should provide an objective and perceptually accurate conversational voice quality on cellular data network specifically 3G services. The method should be efficient and based on the state of the art technology. The method is focused on the evaluation of voice quality directly from network parameters for the ITU based E-model [3][4][5].

2. Previous Studies

In telephony the call quality is rated with a single metric number that is known as Mean Opinion Score or MOS [6][7]. Voice over IP (VoIP) evolved from data network and gradually replace conventional telephony. It is therefore necessary to rate VOIP calls quality using a single metric number as well. Preferably the metric number should have the same significance. Telephone call quality rating or MOS is traditionally a subjective measurement as it is based on the opinion of a sample of telephone users. MOS is defined in the ITU (International Telecommunications Union) recommendation P.800 [6].

VOIP calls are transported in data networks that have many factors contributing to the quality of its performance. Most of the contributing factors are objectively measurable due to their digital nature. ITU Recommendation G.107 [3] recommends a model called E-model to assess objectively the quality of VoIP calls. The output of an E-model is a single metric number defined as R-value. It is derived from delays and equipment impairment factors. R-value can be mapped to an estimated MOS.

There are other ITU standards establishing objective measurement of calls. The essential ones are ITU P.862 PESQ or Perceptual Evaluation of Speech Quality. The standards are not really suitable to assess call quality on a data network as they are based in traditional telephony technology. Data network issues such as delay, jitter, and datagram loss cannot be evaluated or mapped accurately.

Quality of Service (QoS) is measured for different purposes such as monitoring, control to fulfil commercial obligations as stipulated in service level agreements (SLA), and troubleshooting certain disturbance [1]. ITU Recommendation P.800 formalized the voice quality of traditional analogue voice communications using subjective methods and defined a known parameter called Mean OpinionScore or MOS [3]. As voice communications evolve to digital communications, objective
methods are feasible. The resulted parameter can be related to the traditional parameter MOS. MOS is defined as a five-point scale (Excellent, Good, Fair, Poor, and Bad) under controlled conditions according to ITU-T standard P.800 [6][7]. The survey to arrive to a MOS figure could be based on several subjective test such as listening only or one way call, or more common conversational. Conversational test that involves interactivities. The voice quality scores are referred to as conversational voice quality. It is the internationally accepted metric as it provides information about voice quality as perceived by the end user. The major problem in subjective measurement is that it is time consuming, and expensive for an operational network infrastructure. Hence, objective methods are very attractive for voice quality measurement of a business entity communications networks. The objective method is normally performed if the voice communication and infrastructure nature is digital. The data that constitute the quality parameter can be objectively obtained and calculated from the network.

The measurement of voice quality can be intrusive or nonintrusive. Intrusive methods are more accurate because of the need to utilize the network. The ITU-T P.862 Perceptual Evaluation of Speech Quality (PESQ) [8] is the commonly used intrusive method to find voice quality in VoIP applications. It is involves a comparison of a degraded speech signal to a reference speech signal. In this research this method is not suitable as it involves certain reference data that is not readily available from the network. In nonintrusive techniques no reference signal is necessary. It can evaluate the quality of voice directly from the network. Relevant system parameters can be collected and calculated to render quantitative metrics. Parameters of interest among others are packet loss, delay, and jitter. The user perception of degraded voice signal could indicate the reliability of metrics obtained. For planning purposes ITU-T issued Recommendation G.107 [3] with its E-model. It is a computational model that is used to provide voice quality from the network and other system parameters. The use of measured parameters enables a reasonably objective result. QOS objective methods eliminate the human perception factor in deriving the result. ITU-T G.107 provides an objective method for QOS transmission planning. It is commonly known as the E-model.

There were previous research and studies concerning the determination of voice quality in a digital environment especially using IP such as the ubiquitous VoIP. In [2][9][10][11][12] a simplified E-model for determining voice was proposed. The model was derived using RTP packets and various voice codecs. Voice packets are transmitted through normal guided media [13]. Walker [12] described a tool to assess VoIP call using E-model. Another research concerns the VoIP quality over access networks. The voice quality of access lines of interest are VoIP users with access line with cable modem to a router, VoIP users with dial-up modem another router and end to end connection using of users with cable modem to users with dial-up modem. Voice sessions were established between soft phone clients using the Session Initiation Protocol (SIP). The method employed for carrying out the quality assessment was a modified E-model that caters effective equipment impairments due to delay, loss and jitter over packet networks. Codecs under test are G.729a and G.723.1 codecs. The quality of voice calls was evaluated following ITU-T G.107 speech quality E-model. The R-factor and resultant Mean Opinion Scores (MOS) were calculated based on the obtained result and adjustments. The researchers concluded that the quality of the chosen access line are not adequate as shown by the obtained MOS and most likely due to capacity limitation.

Markopoulou, Tobagi, and Karam [14] was interested in assessing the VoIP quality over Internet backbones. As the Internet is currently ubiquitous communication infrastructure for various services including telephony, corporate users expect that the telephone service must have the same voice quality as traditional telephone. They want to find out to what extent the current Internet is meeting this expectation. The assessment is based on delay and loss measurements of the backbone networks. To assess the VoIP quality it is necessary to choose parameters that are relevant to voice traffic. There are several sources of impairments that rendered the deterioration of voice calls,

Even though; performance is usually presented in terms of delay and loss statistics, but ultimately, the judge for the quality of a phone conversation is the user. The most appropriate quality measure
is the user’s opinion, where the commonly used subjective metric is the Mean Opinion Score. MOS has a scale from 1 to 5. The findings show that in general the voice services can be adequately provided, but occasionally a number of paths have poor performance even if the VoIP end-systems are excellent.

2.1. Objective Parameters

ITU –T E-Model is basically a network planning tool that could be used among other for packet-switched networks carrying voice. It provides estimation the effect of impairments to voice quality. It could means to estimate the subjective Mean Opinion Score rating of voice quality. The E-Model is an analytic model of voice quality used for network planning purposes. E-Model provides the calculation of the R-factor. R factor is an objective quantitative measure of voice quality as opposed to MOS (Mean Opinion Score) which subjectively measure voice quality of a telephone call. The range of R is from best case of 100 to a worst case of 0. The R-factor is then used to map to the more familiar MOS figure. MOS itself is the arithmetic average of opinion of telephone call quality score. MOS has 5 scores. Score of 5 is defined as excellent quality, good has a score of 4, fair is defined a score of 3, poor is a score of 2, and finally bad has a score of 1. A mapping of R factor of VoIP quality to the more familiar MOS as shown in Table 1. For practical purposes Table 1 is created based on [5], [9].

| Quality       | Score          |
|---------------|----------------|
| Excellent     | 4.3-5.0        |
| 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       |

The R-factor is defined in terms of several parameters associated a voice phone call passing channels across mixed networks. In this research the networks could be circuit switched, cellular network, and packet switched network. The parameters in the computation of the R-factor as defined in ITU-T G.107 are extensive and covering various impairment factors such as delay, packet loss and others. The R-factor is related to the MOS. R < 0 is defined as MOS = 1, R > 100 is equal to MOS = 4.5, whereas if 0 < R < 100, MOS must be calculated as following [5], [9]:

$$MOS = 1 + 0.035 R + R (R - 60) (100 - R) (7) (10^{-6})$$  \hspace{1cm} (1)

Cole and Rosenbluth [8], [9], provides a reduced expression of R-factor due to the use of default values for the switched circuit impairments and focused to packet network. Evaluating the circuit switch default values resulted in impairments due to digital signal for VoIP resulted in a short form. The R expression has delay impairment factor $I_d$ and equipment impairment $I_e$

$$R = 94.2 - \{I_d\} - \{I_e\}$$  \hspace{1cm} (2)

The significant relevant parameters that are typical properties of data network are delay and network packet loss. The delay component consists of the average of absolute one-way mouth-to-ear delay, the average, one-way delay from the receive side to the point in the end-to-end path where a signal coupling occurs as a source of echo, and the average of round trip delay in the four-wire loop In this research the delays are not separately measured but are based upon network delay that is measured by Wireshark. It provides telephony related statistics that are accessible the Telephony menu. It is considered that all delay components are included as in the delay measured by wireshark. Referring to [5][8] the delay in the packet network can be calculated as

$$I_d = D10.024d + 0.11(d - 177.3) H(d - 177.3)$$  \hspace{1cm} (3)

where $H(x) = 0$ for $x < 0$ and $H(x) = 1$ for $x \geq 0$ and $D1=0.024$/msec and $D2=0.11$/msec [10].
The next component is defined as equipment impairing factor. Modern coding with low bit-rate such as the GSM Standards will contribute with distortions that resulted in a decrease of the perceived speech transmission quality. The impairments due to different types of low bit-rate codecs is called equipment impairment factor $I_{ef}$. It is presumed to cover diverse effects that could be associated with the codec used in the connection beyond what already covered by the E-model such as overall attenuation.

According to [9][11] this factor it can be the result of subjective measurements for each particular codec and certain operating conditions. $I_{ef}$ values can be determined in tests carried out according to a methodology given in ITU-T P.833. The various $I_{ef}$ values for several codec types are listed in ITU-T G.113. The effect of packet loss the increase of $I_{ef}$ and decreasing the call quality (MOS). For GSM Half Rate if C/I (carrier to interference) is 10 dB the $I_{ef}$ range are between 25 and 32, Full Rate has $I_{ef}$range between 32 and 39. G.113 provide the range due to the difficulties in deriving exact impairment factor values for these conditions. The formula being used is:

$$I_{ef} = L_1 + L_2 \ln(1 + L_3)$$  \hspace{1cm} (4)

In [8] the values of $L_1$, $L_2$, $L_3$ are given for certain codecs. For codec G729a $L_1=11$, $L_2=40$, and $L_3=10$, whereas G711 $L_1=0$, $L_2=30$, and $L_3=15$. There are research result for codec used in GSM, but in this research the estimated value $L_2=25$ and $L_3=14$ whereas $L_1=20$ as in G.713. Hence for GSM codec $I_{ef}$ can be simplified as:

$$I_{ef} = 20 + 39 \ln (1 + 10e)$$  \hspace{1cm} (5)

The estimation for GSM is based on its codec rate of 13 kbps. $L_2=30$, $L_3=15$ for G.711 with codec rate of 64 kbps and $L_2=40$, $L_3=10$ for G.729.a with codec rate of 8kpbs. Codec difference of 56 kbps has difference in $L_2 = -10$ and $L_3 = 5$. Estimation that for 1 kbps difference in codec rate will result in change of $L_2$ equal to $(15-10)/56$. $L_3$ of GSM is estimated $10 + (13-5)(5/56) = 10.45$ or $L_3 = 10$. 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.

3. Methodology
The approach toward evaluation of the quality suitability for VOIP service to a SIP-PBX using 3G service as its extension access line is evaluating MOS objectively. To determine if VoIP connections of SIP-PBX extensions to its server using 3G service covering its area have acceptable quality (MOS) as defined in ITU P.800 an E-model is used. ITU-T's model is adapted to accommodate measurable parameters. MOS calculations follow E-model of G107 even though the existing model is applicable to certain voice codecs [10] and network configurations. The model is adjusted following other relevant research that derived the resulting empirical formulae.

This research is focused on the evaluation of voice quality directly from network parameters applicable in the E-model. The measurable quantities are obtained through the use of network protocol analyzer Wireshark. Wireshark is a common multipurpose network analyzer that has features applicable in analyzing SIP (Session Initiation Protocol) the commonly applied VoIP. The parameters are retrieved according to the following configuration:

An open source SIP-PBX that serves as a VOIP sever is selected. This SIP-PBX is commonly known as Asterisk. Asterisk is placed in the Internet accessible by its extensions. It has features commonly available in a traditional PBX. The internal communications and external communications are controlled and handled by Asterisk. The general physical infrastructure is fixed lines such as copper cable and fiber optics and based on TCP/IP. For voice communications common devices are traditional telephone sets with IP adapter (ATA), softphone at workstations or PC and sometimes IP phones. The measurements taken to determine VoIP quality are based on remote softphone connections to Asterisk. A connection is deemed having suitable quality if the features of the SIP-PBX is working properly and the called parties can have proper voice communications.
4. Process

RTP packets are captured during a call. To ensure that the information will have high quality and also a reliable view to a call, samples are taken several weeks. D1, D2, D3, D4, are day of the week or Monday, Wednesday, Thursday and Friday. A number of samples are taken 5 times a day and 4 days per week. RTP packet samples are taken from the caller to server and server to receiver. The time slots are S1 is between 10:00 - 12:00 pm, S2 between 12:00 - 14:00, S3 between 14:00 - 16:00, S4 between 16:00 - 18:00 and S5 between 18:00 - 20:00. The parameters of importance are delay and RTP packet loss. MOS is then calculated from the obtained data. Wireshark provides the duration or length of time of a call and the number of RTP packets send and received by end point. The delay time can be calculated to show the time needed by RTP packets to reach the server. The total delay is the time needed by an RTP packet leaving the sender to arrive at the receiver via the server. Total delay is in msec.

Measurement of calls between Soft IP Phones with GSM codec to another Soft IP Phones with GSM codec is shown at the following tables. Table 2 shows the time delay across day of the week and defined time slots. Daily time delay in average is between 40 msec and 42 msec which is less than 50 msec. The smallest daily delay is 40 msec and maximum delay of 42 msec. Each time slot delay during the week is also less than 50 msec with minimum delay of 40 msec and maximum of 42 msec.

| Slot  | Day 1 | Day 2 | Day 3 | Day 4 | Average |
|-------|-------|-------|-------|-------|---------|
| Slot 1 | 40    | 40    | 40    | 41    | 40      |
| Slot 2 | 40    | 40    | 39    | 40    | 40      |
| Slot 3 | 44    | 40    | 44    | 40    | 42      |
| Slot 4 | 44    | 40    | 44    | 40    | 42      |
| Slot 5 | 40    | 40    | 40    | 45    | 41      |
| Average| 42    | 40    | 42    | 41    |         |

Table 3 shows the corresponding MOS. The smallest daily MOS is 3.1 and maximum MOS is 3.6. Each time slot MOS in average is between 3.1 and 3.6. The GSM to GSM call using 3G service is classified as poor.

| Slot  | Day 1 | Day 2 | Day 3 | Day 4 | Average |
|-------|-------|-------|-------|-------|---------|
| Slot 1 | 3.5   | 3.6   | 3.5   | 3.4   | 3.5     |
| Slot 2 | 3.5   | 3.6   | 3.5   | 3.7   | 3.6     |
| Slot 3 | 2.3   | 3.6   | 2.3   | 3.7   | 3.0     |
| Slot 4 | 2.5   | 3.7   | 2.5   | 3.5   | 3.0     |
| Slot 5 | 3.7   | 3.7   | 3.7   | 2.4   | 3.4     |
| Average| 3.1   | 3.6   | 3.1   | 3.3   |         |
Measurement of calls between Soft IP Phones with G.711 to another Soft IP Phones with GSM codec showed that daily time delay in average is between 40 msec and maximum delay of 48 msec. Each time slot delay during the week is also less than 50 msec. The corresponding MOS is maximum.

The same measurement is performed for Soft IP Phones to another Soft IP Phones and both of them use Codec G.711 or the voice is not compressed. Table 4 shows the time delay measurements results. Daily time delay in average is between 40 msec to 45 msec which is less than 50 msec. Table 5 shows the corresponding MOS. Daily MOS in average is 3.9 and maximum MOS is 4.4. The G711 to G711 call using 3G service is classified as satisfactory. The result is expected as voice is not compressed. Recovery of digitized of voice is can be well performed provided there are no packets loss.

### Table 4. G711-G711 Delay Time

|       | Day 1 | Day 2 | Day 3 | Day 4 | Average |
|-------|-------|-------|-------|-------|---------|
| Slot 1| 40    | 40    | 40    | 62    | 46      |
| Slot 2| 39    | 40    | 40    | 40    | 40      |
| Slot 3| 40    | 41    | 40    | 40    | 41      |
| Slot 4| 40    | 40    | 40    | 40    | 40      |
| Slot 5| 41    | 40    | 40    | 42    | 41      |
| Average| 40    | 40    | 40    | 45    |         |

Table 5 shows the corresponding MOS. Daily MOS in average is 3.9 and maximum MOS is 4.4. The G711 to G711 call using 3G service is classified as satisfactory. The result is expected as voice is not compressed. Recovery of digitized of voice is can be well performed provided there are no packets loss.

### Table 5. MOS G711-G711

|       | Day 1 | Day 2 | Day 3 | Day 4 | Average |
|-------|-------|-------|-------|-------|---------|
| Slot 1| 4.2   | 4.3   | 4.4   | 2.7   | 3.9     |
| Slot 2| 4.4   | 4.4   | 4.3   | 4.2   | 4.3     |
| Slot 3| 4.4   | 4.2   | 4.4   | 4.4   | 4.3     |
| Slot 4| 4.4   | 4.4   | 4.4   | 4.3   | 4.4     |
| Slot 5| 4.3   | 4.4   | 4.4   | 4.1   | 4.3     |
| Average| 4.3   | 4.3   | 4.4   | 3.9   |         |

5. Conclusions

Voice Quality Assessment of VoIP call within a 3G service infrastructure and in particular soft IP phones and SIP PBX as a VoIP server showed that the MOS (Mean Opinion Score) is classified as poor if any soft phone compress voice input. Without compression the quality will rise to good. Main quality measure is MOS for VoIP which is objectively measured using E-model, albeit simplified E-model. Based on MOS figures, 3G is considered inadequate if any of the soft phones compressed voice. The research used GSM compression and G711 as uncompressed voice. Delay time itself is below 50msec which is still accepted as suitable for conversation. It would be beneficial to consider other compression standards and besides of delay time to consider jitter to have complete quality view. The research result is adequate for practical purposes especially if 3G is the only infrastructure choice in certain geographical locations.
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