QoE Estimation Model for a Secure Real-Time Voice Communication System in the Cloud

Aklilu Daniel Tesfamicael
Science and Engineering Faculty
Queensland University of Technology (QUT)
Brisbane, Queensland, Australia
aki.tesfamicael@hdr.qut.edu.au

Vicky Liu
Science and Engineering Faculty
Queensland University of Technology (QUT)
Brisbane, Queensland, Australia
v.liu@qut.edu.au

Ernest Foo
Science and Engineering Faculty
Queensland University of Technology (QUT)
Brisbane, Queensland, Australia
e.foo@qut.edu.au

Bill Caelli
Science and Engineering Faculty
Queensland University of Technology (QUT)
Brisbane, Queensland, Australia
w.caelli@qut.edu.au

ABSTRACT

As moving towards cloud-based real-time services, we are witnessing the shift from a technology-driven services to service provisioning paradigms, that is, from Quality of Service (QoS) to Quality of Experience (QoE). User experience and satisfaction are placed at the epicenter of the system design. QoE is a measurement of user experience on the provided service by a system. Often QoE is measured by subjective mechanisms, such as user experience surveys and mean opinion scores (MOS) methods, which can be a costly and time-consuming process. Using an adequate QoE model to measure user experience of perceived quality is cost-effective, compared to using time-consuming subjective surveys. Applying an adequate QoE model to assess user experience is advantageous for cloud-based real-time services such as voice and video. This study uses a formula-based QoE estimation model to estimate and predict QoE prior to the deployment or during the planning stage of the system service. This study investigates a real-world scenario of a company that recently moved to its premises-based real-time trading communication system (TCS) to a public cloud. A simulation system using OPNET is also implemented to illustrate the usefulness of the model. Our result shows that the effect of delay on the users experience of the service provided by the cloud-based TCS is minimum comparing to packet loss rate (PLR) and Jitter. However, it has been observed that the overhead of the different security settings of the TCS system had no major negative impact to the user experience. The proposed model can be used as a QoE control mechanism and network optimization for cloud-based TCS services.

CCS CONCEPTS

• Computing methodologies → Model verification and validation
• Networks → Network performance modeling

KEYWORDS

QoE, QoS, TCS, VoIP, Real-time, E-Model

ACM Reference format:


1 Introduction

QoE can be used to measure user perceived experience for a provided voice service of VoIP system. QoE is referred to as user perceived quality of voice service and is one of the key design influence factors and an indicator for impairments affecting the VoIP quality. QoE is basically depends on user satisfaction in terms of usability, accessibility, and integrity of the QoS which reflects network performance. QoE is not limited to the technical performance of the network; there are also non-technical aspects, which influence the user perception and satisfaction of the quality of the service provided by VoIP. Therefore, QOS by itself is incomplete to measure the overall voice quality of the VoIP system without incorporating QoE. QoE and QoS are used to measure the quality of the whole system service.

The ITU-T (Telecommunication Standardization Sector of the International Telecommunications Union) [1] defines Quality of experience (QoE) as “The overall acceptability of an application or service, as perceived subjectively by the end-user”. Namely, QoE can provide an indication of how well the system meets the user’s needs. QoE is the key determining factor for the success of real-time communication services or applications. This study investigates a real-world scenario of a company that recently moved to its premises-based real-time trading communication system (TCS) to a public cloud.

The difference between QoE and QoS is that the former focuses on a service provisioning paradigm based on how the end user feels, whereas the latter is the measurement the overall performance of a service from a technology-driven perspective. The objective of QoS is usually to evaluate using classical network performance
metrics, such as latency, jitter and packet loss [1, 2, 7]. QoE focuses on user perception of application or service, which can be influenced by the network performance or QoS related factors such as delay, jitter and packet loss.

QoS is related to technical aspects of the service quality and normally does not involve in any human perception that is directly related to the end user experience or acceptability of the system service. That is the one of reasons that why considering only QoS aspects is incomplete for rating the overall success of a provided system service. This means different users may have different perceived quality for the same system service. In addition, apart from the technical factors of the network service quality, other factors, such as the context of use (such as surrounding environment, type of device used), the influence of socio-demographic factors (such as age, gender or sex), the delivered content and the pricing of a service, have a significant impact on the finally perceived user experience, which the aspects cannot be captured and measured by the use of QoS alone [9]. However, QoE presents the overall experience on a provided service whereby users experience on the provided service is evaluated. Similarly, QoE can also be incorporated to network design and decision-making process as QoE can measure and predict user satisfaction for upcoming deployments.

QoE measurement techniques of a perceived voice quality can be classified into subjective and objective aspects [12-14]. The Mean opinion score (MOS) recommended by the ITU-T Standard P.800 [16], is one of the most widely used subjective measures on voice quality. The MOS value is usually obtained by interviewing the end users evaluating to grade of the quality of speech on the use of a five-point scale (Excellent, Good, Fair, Poor, and Bad) [16].

In terms of the evaluation of the perceived service quality, QoE primarily evaluate the perceived quality of user experience based on a subjective method. Subjective methods are usually based on controlled actual experiments with human participants who directly evaluate their experience of a service in active or passive way. These methods are empirical in nature. However, in an objective method, quality perceived by end-users is measured or predicted without the user intervention and are statistical in nature.

The cloud-based TCS system is different from the conventional premises-based VoIP system from a QoE perspective. With the case scenario this study investigated, the cloud-based TCS system, the end device is a tablet computer to replace the conventional TCS with intercom handset with keypads that may have an impact to the overall QoS such as user interface, external speaker, open circuit (open lines) etc. QoE estimation based on subjective method is a time-consuming, costly process, is inadequate to meet with real time demands and lacks appropriate reusability. Further, due to the subjective nature of user’s ratings, parametric statistical models cannot be applied for QoE measurement and prediction. Reichl et al. [14] argued that the development of objective methods is necessary to increase in QoE measurement and prediction accuracy. It is also a requirement for cloud-based TCS to set up security mechanisms to maximize the protection of confidential, market-sensitive information. To tackle these problems this study proposes a secure QoE-Efficient model based on an objective method, so that quality perceived by end-users can be measured and/or predicted without user intervention. The contributions from this study is to apply a formula-based QoE model for the prediction and estimation on the perceived quality for a cloud-based real-time TCS service. The contributions of this study are presented as follows:

1. The required service quality is predicted before systems are implemented to meet the QoS requirements.
2. The required service quality is estimated even after systems are implemented to reflect the degradation of the quality experienced by users. Using the proposed QoE modelling, the information can be collected and analyzed to monitor the system for any indication of the quality service degradation experienced by users.
3. In the planning phase to drive a more resource-efficient network operation. This formula can help consider if adding further resource provisioning will improve the QoE for the end-user.
4. To apply the proposed QoE modelling to a real-world scenario case for validating our proposal.

The remainder of this paper is organized as follows. Section 2 presented a literature review and the Quality of Experience (QoE) requirement is presented in Section 3. Analysis of QoE Estimation model is discussed in Section 4 followed by the proposed QoE model and the discussion on the simulated result in Section 5. The conclusions are given in Section 6.

2 Literature Review

Numerous surveys have been conducted to investigate QoE for real-time and non-real-time services. Veria et al. [3] and Charonyktakis et al. [4] investigated the QoE assessment approaches for VoIP services. Their investigations provided a review of recent advances related to the QoE for VoIP, while Wu et al. [12] focused on IPTV QoE assessments. Similarly, authors in [5] provided a guide on how QoE can be standardized and how an actual quality assessment can be conducted mainly for VoIP, online video, video streaming, web browsing, skype, IPTV and file download services. Tsolkas et al. [1] studied speech quality estimation and provided a taxonomy of QoE estimation methods.

Authors in [6] have studied QoE for mobile web services. The authors developed a QoE estimation model for services such as file browsing, file downloading and uploading. This model was built based on a general regression model and verified using numerical and statistical analysis using experimental scenarios developed for each of the services.

A number of studies [5-6, 15] focused on models mapping MOS values to network parameters such as codec, packet loss, mean loss burst size, one-way delay and jitter. These factors may be applied for real-time quality predictions. Some examples of this approach
are the Pseudo-Subjective Quality Assessment (PSQA) method [7] and a modular algorithm for user-centric QoE prediction the (MLQoE) [8]. The MOS model is the most extensively used measurement scale for observations of this kind but it should be incorporated with QoS metrics as well to give an accurate QoE measurement and prediction. In terms of the parametric objective method, most previous studies use the E-model recommended by ITU-T [9-10].

Hoßfeld and Binzenhöfer [22] performed a QoE assessment of Skype calls over the Universal Mobile Telecommunication System (UMTS) that supports VOIP calls operating in a mobile environment. They analyzed the quality of IP-based voice calls using Skype in a subjective and objective ways. Their results were based on the performance analysis of measured QoE in a real UMTS network and a testbed environment. Ding et al. [23] proposed a parametric, non-intrusive speech quality assessment algorithm which combined an Internet protocol analysis and the ITU-T E-model. They measured speech quality from major VoIP impairments including packet loss, temporal clipping and noise.

Tsolkas et al. [24] and Gómez et al. [25] studied the effect of QoE over the resource efficiency of the system. They incorporated a QoE model into the overall network architecture with an aim to achieve more resource-efficient operations. By monitoring and controlling the QoE model can determine if investing extra resource will improve the quality of the offered services as perceived by the users.

As mentioned in section 1 the ITU-T Standard [16] provides two testing methods, subjective and objective methods of testing voice quality. Many research works [27-33] applied the ITU-T standard testing methods to study the performance of the real-time voice quality. In [27] the authors focused their study on the impact of the network impairment of a real-time voice system to the overall perceived quality of the voice. They discovered that packet loss rate and audio bandwidth are the most network characteristics that impact the user’s QoE. Authors in [26] presented a well-structured detailed taxonomy of QoE focusing on the business, technical, and human aspects of the QoE.

What we have observed from our literature review is that a further study is required to propose a simplified QoE estimation model for a secure real-time TCS services, so that perceived quality of secure real-time voice services in the cloud is accurately measured and monitored.

3 QoE Requirement

Real-time communication systems such as TCS should provide network services and quality to a level that meet the needs of user applications such as service reliability, QoS and QoE requirements. The security and QoS requirements of a cloud-based TCS service has been defined and analysed to evaluate the quality of the real-time deployment of TCS and analyzed the impact of security implementation on overall cloud-based TCS performance. This study considers these requirements (as described in Table 2 and 3) with the QoE requirement described in Table 3 and 4 (based on MOS), so that the overall quality of the secure cloud-based TCS is evaluated in both QoE and QoS perspectives. This study also investigates if different levels of security, high, medium and low security mechanisms (as described in Table 4) have a negative impact on the overall perceived quality of the service from the users experience perspective.

MOS is a numerical averaging of the perceived audio quality of speech signals as judged by user experience. MOS is an absolute category rating (ACR) metric and is expressed in number values of 1 – 5, as listed in Table 1.

Table 1. ITU-T TCS audio quality MOS requirement [9]

<table>
<thead>
<tr>
<th>Category</th>
<th>Score Rating</th>
</tr>
</thead>
<tbody>
<tr>
<td>Excellent</td>
<td>5</td>
</tr>
<tr>
<td>Good</td>
<td>4</td>
</tr>
<tr>
<td>Fair</td>
<td>3</td>
</tr>
<tr>
<td>Poor</td>
<td>2</td>
</tr>
<tr>
<td>Bad</td>
<td>1</td>
</tr>
</tbody>
</table>

One of the key user satisfaction factors when evaluating QoE is the R-factor. R-factor is calculated based on the formula we model in Section 4. The user satisfaction rating is listed in Table 2.

Table 2. Relationship between R, MOS and User Satisfaction [9]

<table>
<thead>
<tr>
<th>R-factor Range</th>
<th>MOS</th>
<th>User Satisfaction</th>
<th>Quality Rating</th>
</tr>
</thead>
<tbody>
<tr>
<td>90–100</td>
<td>Best ≥4.34</td>
<td>Very satisfied</td>
<td>Excellent (5)</td>
</tr>
<tr>
<td>80–89</td>
<td>High [4.03, 4.33]</td>
<td>Satisfied</td>
<td>Good (4)</td>
</tr>
<tr>
<td>70–79</td>
<td>Medium [3.6, 4.02]</td>
<td>Some users Dissatisfied</td>
<td>Fair (3)</td>
</tr>
<tr>
<td>60–69</td>
<td>Low [3.1, 3.5]</td>
<td>Many users Dissatisfied</td>
<td>Poor (2)</td>
</tr>
<tr>
<td>50–59</td>
<td>Poor ≤2.8</td>
<td>Nearly all users Dissatisfied</td>
<td>Bad (1)</td>
</tr>
</tbody>
</table>

The provision of QoS is crucial for the deployment of a TCS in the cloud to ensure performance. The ITU-T G.114 [19] recommends that a set of QoS requirements for achieving high quality for a real-time communications system. The QoS requirements are listed in Table 3.

Table 3. QoS Requirement [19]

<table>
<thead>
<tr>
<th>QoS Requirement</th>
<th>Recommendation</th>
</tr>
</thead>
<tbody>
<tr>
<td>Jitter</td>
<td>Less than 30ms</td>
</tr>
<tr>
<td>Latency</td>
<td>150ms or less (one-way end-to-end)</td>
</tr>
<tr>
<td>Packet Loss</td>
<td>Less than 1%</td>
</tr>
</tbody>
</table>

It is deemed good governance for corporations to set up policies and procedures to maximize the protection of confidential, market-sensitive information. Reducing the risk of information leaks promotes and preserves market integrity. The deployment of TCS voice in the cloud also requires the implementation of security to protect the system from potential attacks. If TCS is compromised, trade holds can lead to great financial loss and may cause high risk
of a financial penalty to the business for non-compliance with the financial market rules of operating with agreed service provision. In this research, as per [3], we employed different levels of security, high, medium and low security, to analyze the impact of security implementation on the overall quality of voice of a cloud-based TCS from the users experience perspective. In our simulation, an IPSec-based VPN was used to confidentiality and authentication. The security policy has been configured at the trader and broker host levels (as shows in Figure 3 and 4) and security services are applied to the SIP signaling and media stream as described in Table 3. The OPNET Modeler only supports hash functions SHA1 and SHA256, as such we used SHA1 and SHA256 for our practical implementation and analysis.

<table>
<thead>
<tr>
<th>Security</th>
<th>Authentication</th>
<th>VPN Setting</th>
</tr>
</thead>
<tbody>
<tr>
<td>High (H)</td>
<td>Voice recognition prior to a call setup</td>
<td>SHA256/ AES256</td>
</tr>
<tr>
<td>Medium (M)</td>
<td>A PIN Number Identification</td>
<td>SHA256/ AES192</td>
</tr>
<tr>
<td>Low (L)</td>
<td>No voice recognition</td>
<td>SHA1/ AES128</td>
</tr>
</tbody>
</table>

Table 4. Security Requirement [7]

4 Analysis of QoE Estimation Model

A number of studies use the E-model recommended by ITU-T [9-10]. E-model is a formula based quality of voice perdition model that assesses the combination of effects from parameters on voice quality. The original E-model is based on a complex set of fixed and empirical formulae which is inefficient for real-time quality monitoring or for optimization/control purposes for services like TCS. However, the simplified E-model (4) is improved version of the original E-Model in which the model considers delay and packet loss effects, no echo and loudness is constant. Due to its real-time quality monitor of the TCS our research considers the simplified E-model when we evaluate the perceived voice quality of the cloud-based TCS.

Before we start using the simplified E-model to predict the voice quality it is vital to understand the fundamental relationship between the QoE and an impairment factor corresponding to the QoS. Hösfeld et al. [15] used exponential interdependency of QoE and QoS (IQX) hypothesis to correlate QoS and QoE hypothesis, which is represented as a differential equation. In [15] the authors derived formula (1) to show the performance degradation of the QoE with respect to QoS parameter, like packet loss, jitter and delay assuming a linear dependency of the QoE level.

\[
\frac{\delta QoE}{\delta QoS} = -\beta (QoE - \gamma) \tag{1}
\]

and integrating (1) reproduce the hypothesis to correlate QoS and QoE hypothesis described in (2)

\[
QoE = \alpha e^{-\beta QoS} + \gamma \tag{2}
\]

Where \(\alpha\), \(\beta\) and \(\gamma\) represent the parameters of the model function that are retrieved by means of non-linear regression. The model uses network QoS parameters such as jitter, delay and packet loss when considering the correlation (relationship) between QoE and QoS.

\[
QoE = \alpha e^{-\beta QoS} + \gamma
\]

![Figure 1. QoS and QoE Correlation [19]](image)

For normal parameter settings \(R \in [0;100]\) R can be transformed to an estimation of a mean user judgment on a 5-point quality scale defined in [17], using the fixed S-shaped relationship.

\[
MOS = \begin{cases} 
1 & \text{for } R < 0 \\
1 + 0.035 + R & \text{for } 0 \leq R \leq 100 \\
4.5 & \text{for } R > 100 
\end{cases} \tag{3}
\]

Both the transmission rating factor R and the estimated mean opinion score MOS give an indication of the overall quality of the connection.

The primary output of the E-model is the quality rating value, known as transmission rating factor, R, which can be expressed in terms of the KPIs, such as MOS. The original E-model is based on a complex set of fixed and empirical formula which is inefficient for real-time quality monitoring or for optimization/control purposes of TCS. Moreover, the voice processing is not related significantly to the instantaneous judgment of QoS. Authors in [6] and [21] presented a simplified version of the E-model focusing on important parts and afterwards it was used in a monitoring system [2]. Our model considers the network performance and the signal to noise ratio as the main factors that can affect the voice quality of TCS in the cloud. As such, it considers network performance and signal to noise ratio impact as the main factors to estimate and predict the perceived voice quality of TCS. Authors in [3] recommended the use of G.711 codec for cloud-based voice service such as TCS. As such, in our model, we didn’t consider a codec to be one of the main factors that can severely impact the overall voice quality of TCS as G.711 gives the best call quality for voice on the basis that it uses no compression and the call quality is the best from other codecs available for voice. However, the equipment impairment (i.e codec quality) is incorporated in our model as the model can also be used for conventional voice services that doesn’t have strict requirements as that of TCS. As a result, our simplified E-model of evaluating the R factor can be expressed by equation (4), scaling the value of R from 0 (worst) to 100 (best).

\[
R = Ro - 1d - 1e - 1pl \tag{4}
\]
where Ro represents the basic signal-to-noise ratio, Id represents the impairments caused by delay, Ie represents the impairments caused by low bit rate codecs and Ipl represents the packet loss rate within a particular time. In simple terms, the overall quality (R Factor) is calculated by estimating the signal to noise ratio of a connection (Ro) and subtracting the network impairments (Id, Ipl) and the impairments caused by codec.

Once the R-factor formulation is completed the goal is to correlate its relationship to the overall MOS value so that user satisfaction is rated accordingly. The relationship (interpretation) of the R factor in terms of MOS and user satisfaction is described in Table V.

\[
Id = 0.024d + 0.011 (d - 177.3)H(d - 177.3),
\]

(5)

Where

\[
H(x) = \begin{cases} H(x) = 0 & \text{if } x < 0 \\ H(x) = 1 & \text{if } x \geq 0 \end{cases}
\]

(6)

where d is the one-way delay measurement and Ie is the equipment impairment given by,

\[
Ie = a \ln(1 + bp) + c
\]

(7)

where p is the average packet loss rate and a, b and c are constants. Ie can also be used from simplified E-Model [21] based on the following equation:

\[
Ie = Ie_c + (95 - Ie_c) \frac{Ppl}{Ppl + Bpl}
\]

(8)

where, Ie-c is the equipment impairment (i.e codec quality), Bpl is the packet loss robustness and Ppl is the packet loss rate in percentage. In the case of the packet loss burstiness,

\[
Ie = Ie_c + (95 - Ie_c) \frac{Ppl}{BurstR(BurstR + Bpl)}
\]

(9)

\[
BurstR = \frac{\text{average length of observed bursts in arrival sequence}}{\text{average length of burst expected for the network under "Random" loss}}
\]

(10)

If Packet loss is random,

\[\text{BurstR} = 1\]

If Packet loss is bursty,

\[\text{BurstR} > 1\]

5 The Premise of the Proposed QoE Model

Between the two main QoE quality assessment methodologies, namely subjective and objective assessment, this study focused on the objective approach and supported by the QoE results from our case study of a TCS system deployed in the cloud [3]. An objective approach for measuring voice quality can be intrusive or non-intrusive. The intrusive methods are based on signals, while non-intrusive methods are based on network or application parameters [7].

This study adopted the non-intrusive method using mathematical models to translate the effects of delay, jitter and packet loss rate received QoS to QoE. This method is appropriate for monitoring live traffic. Most non-intrusive methods are based on ITU-T G.107 E-model [8-9], which are used to estimate the relative voice quality when comparing two reference connections. These models are parameter based algorithms, based on parameters related to terminal, environment, network factors and so on and result is calculated to obtain the rating value. According to [8], the E-Model has proven as a useful model for a transmission planning tool. E-model can estimate conventional MOS value using the R-value as per (4).

In Figure 2, an overview on QoE evaluation assessment method for audio is presented.

...QoE Evaluation Method

<table>
<thead>
<tr>
<th>Subjective Evaluation Method</th>
<th>Objective Evaluation Method</th>
</tr>
</thead>
<tbody>
<tr>
<td>Parameter based</td>
<td>Sequential comparison</td>
</tr>
<tr>
<td>E-model</td>
<td>Listening MOS</td>
</tr>
<tr>
<td>R-factor</td>
<td>QoE</td>
</tr>
<tr>
<td>PESQ</td>
<td>MOS</td>
</tr>
<tr>
<td>PEAQ</td>
<td>ODQ</td>
</tr>
</tbody>
</table>

Figure 2. QoE Evaluation Assessment Method

In summary, objective evaluation method characterized as intrusive and non-intrusive approaches. The Intrusive method takes reference signals and degraded signals into account and uses Perceptual Evaluation of Speech Quality (PESQ) and the Perceptual Evaluation of Audio Quality (PEAQ) algorithms. PESQ does not consider the effect of delay impairments, it can be used only to one-way communication. As such, this research takes the non-intrusive method of estimating QoE.

5.1 The Proposed Model

To verify the feasibility of the proposed model, we setup a test bed using the OPNET simulation tool, Riverbed Modeler 18.6. OPNET Modeler [20] is a software tool for computer network modeling and simulation.

OPNET provides modeling, simulation and analysis to design and optimize network-performance. Our test bed is based on a real case scenario as per [3]. This research is based on industry experience, plays a key role in investigating the feasibility of the simplified E-Model in measuring the perceived voice quality of the cloud-based TCS. Our data is based on empirical evidence collected from a real-world scenario. Figures 4 and 5 illustrate the topology of our design with the OPNET simulation tool.
This research considers hosting a public cloud environment with the backend TCS equipment installed at the cloud service provider’s data centre. This includes Internet service to connect between the data centre and the trading floor at the head office, the branch (remote) office, and the teleworkers as shown in Figure 3.

Figure 3. Proposed TCS model

This research considers hosting a public cloud environment with the backend TCS equipment installed at the cloud service provider’s data centre. This includes Internet service to connect between the data centres and the trading floor at the head office, the branch (remote) office, and the teleworkers as shown in Figure 3.

Figure 4. TCS Cloud Modelling (Client site to datacentre)

The research concept is to configure the network with defined parameters, as it is deployed in the real world, and run the simulation. The voice biometric server (also deployed as software) enables secure biometric authentication using a person’s voice.

Biometric authentication is capable of supplementing network access authentication via spoken phrase or by swiping a fingerprint against a reader to gain access. This verifies the identity of a trader on a daily basis during interactions with the brokers. The voice recording server, integrated with the TCS server, captures and archives all trade communications with the ability to access information easily in response to compliance queries in a timely manner. Our modeling is based on OPNET Riverbed Modeler, as shown in Figures 4 and 5.

Figure 5. Global TCS Cloud Modelling

5.2 Performance Evaluation

In order to assess the performance of the proposed model, we consider the effects of delay, jitter and packet loss rate (PLR) that can have an impact on the perceived voice quality of the cloud-based TCS. We also investigate the effect of security overhead (as per Table IV) that can have an impact on user experience of a service provided by a cloud-based TCS. Empirical data was used to evaluate cloud-based TCS with simplified E-Model to characterize the performance of the network impairment factors, latency, packet loss, jitter and efficiency of codec transmitted over the internet. The G.711 codec and different levels of security mechanisms (high, medium and low security) were measured in the experiments. Furthermore, experiments have been performed using the E-model to measure the quality voice of the cloud-based TCS using the R-factor and MOS values.

During the simulation process, MOS values for every cloud-based TCS voice have been computed and analyzed for different values of delay, jitter and PLR. MOS gives us a numerical indication of the perceived quality of the TCS voice traffic received after being transmitted and eventually compressed using G.711 codec. We also calculated the R-factor to correlate its relationship to the overall MOS value so that user satisfaction is rated accordingly.

5.2.1 Effect of Jitter on QoE

As shown from Figure 6, when jitter increases it has a negative impact on MOS. As described in Section 3, an acceptable jitter level for cloud-based TCS voice is less than 30ms. However, our result data shows that with every level of security settings and using the G.711 codec, jitter over 0.1ms is not desirable for the cloud-based TCS as this reaches below the recommended MOS value of cloud-
based TCS. Once jitter reaches nearly to 0.05ms, the MOS value almost reaches the lowest level. This means that jitter higher than 0.1ms has a strong negative impact on users experience of a service provided by a cloud-based TCS. However, what we have concluded from our previous paper [3] that jitter less than 30ms is desirable to maintain the required QoS for the cloud-based TCS service, which is not the case for the QoE.

Though it is not in a big margin what we observe from our result is that a slight effect from the different levels of security shows a negative impact to the QoE for jitter higher than 0.2ms however no negative impact has been observed of the security application for jitter 0.05 and less.

When jitter attains 0% then the MOS becomes 4.5 which indicate that users are very satisfied with the voice quality of the cloud-based TCS. What we can conclude from this result is that the increase of jitter beyond 0.1% have a negative impact to the user experience of the voice service provided by the cloud-based TCS as this draws a MOS value of less than 4 as shown in Figure 6.

5.2.2 Effect of PLR on QoE

Figure 7 shows that the relative received voice quality of a cloud-based TCS transmitted over the internet decreases as the PLR increases. From the results, we observed that a good voice quality of the cloud-based TCS achieved when PLR is 0.5%. Therefore, the results in Figure 7 shows that the good voice quality is achievable between 0% and 0.5% of PLR with the high security applied. Similarly, when PLR rate is 0.5%, the MOS reaches to 4.04. Based on the rating scale from Table 2 the users show satisfaction with the voice quality of the cloud-based TCS. When the PLR attains 0% then the MOS becomes 4.6 which indicate that users are very satisfied with the voice quality of the cloud-based TCS. What we can conclude from this result is that the increase of PLR beyond 0.5% have a negative impact to the user experience of the voice service provided by the cloud-based TCS. We also observed a slight change in the estimated MOS between different set of security application up to PLR rate of 1%.

5.2.3 Effect of Delay on QoE

The experiment we conducted on the impact of transmission delays on QoE is to acquire the threshold level of delay that is acceptable by the users.

Figure 8 illustrates the effect of delay on user perception. The effect of delay on QoE is very minimum comparing to PLR and Jitter. Two-way audio delay up to 300ms shows a good value of MOS and sharply decline beyond 300ms. What we have observed from this result is that as far as the minimum two-way audio delay value is maintained (which is 150ms one-way audio for QoS) it will have no impact to the voice quality of the cloud-based TCS. Similar to the PLR and jitter we have not observed an impact of different levels of security applications to the system on the voice quality of the cloud-based TCS.

5.2.4 Effect of R-factor on QoE

R-Factor is an objective measurement that is based on several factors like signal to noise ratio. R-Factor is calculated by evaluating user perceptions as well as the objective factors that affect the overall quality of a cloud-based TCS. In our experiment the R-factor is calculated using the E-model defined by ITU-T and converted to MOS value using (4). The values of equipment impairment (Ie–c) and Packet-loss robustness factor (Bp) proposed by ITU for G.711 was used when calculating the R-factor.
(for G.711, $1-e^{-c}=0$; $Bpl = 4.3$). Based on our calculation the R-Factor value is converted to MOS as shown in Figure 9. Figure 9 shows that R-Factor value close to 80 or above converts to a MOS value from 4.03, which indicated users are satisfied with the voice quality. We also observed no difference on the impact of the voice quality for different levels of security for R value of 60-85.

![Figure. 9 MOS against R-Factor](image)

**6 CONCLUSION**

There are various approaches for measuring the QoE level of a provided service. One of the available approaches is based on a subjective method whereby users rate a system through interviews. This method is expensive and time consuming. This study investigated a measurement of QoE level through an objective method. Namely, this study proposed a QoE estimation model for a secure real-time cloud-based TCS system based on the simplified E-model.

In order to evaluate the effects of network impairments on voice quality, three main factors are used including delay, jitter and PLR. The simplified E-model is used to measure the voice quality of cloud-based TCS. The MOS values obtained from this model are used to validate cloud-based TCS model. From the test results we observed that network impairments such as jitter and PLR factors have more negative impact on voice quality than the delay factor. It is also interesting to observe applying different levels of security for the cloud-based TCS does not have a significant negative impact on the overall voice quality of the system. As the delays introduced by the security service is consistent latency, unlike jitter is the irregular time delay in the sending of voice traffic over a network.

The proposed model can be used as a prediction model for voice service quality before the system is implemented to meet the QoS requirement. Even after the system is implemented to measure the voice quality to reflect the degradation of the quality experienced by users. The proposed model can also be used as a means of information gathering and analysis to monitor the system for any indication of the quality service degradation experienced by users and for planning purposes to drive a more resource-efficient network operation.

**REFERENCES**


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