User utility function as Quality of Experience(QoE)

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***Abstract*—The realization of a *user-centric* paradigm will revolutionize future wireless networks. For this innovative concept to materialize, a paradigm shift is required from a *long-term contractual* based service delivery to a *short-term contractual* and *dynamic service delivery* concept. However this necessitates that translation of user satisfaction into network technical indices, commonly termed as *Quality of Experience (QoE)*. In this paper we propose the *user utility function* to capture her satisfaction for different services. We validate the proposed utility function by extensively carrying out the simulations for VoIP, video streaming and FTP applications using OPNET simulator. We also suggest three different types of users on the basis of *user preference*.**

***Keywords*-utility function; user satisfaction; QoE;**

I. Introduction and Motivation

The business models of telecommunication operators have traditionally been based on the concept of the so-called closed garden: they operate strictly in closed infrastructures and base their revenue-generating models on their capacity to retain a set of customers and effectively establish technological and economical barriers to prevent or discourage users from being able to utilize services and resources offered by other operators. After the initial monopoly-like era, an increasing number of (real and virtual) network operators have been observed on the market in most countries. Users benefit from the resulting competition by having a much wider spectrum of choices for more competitive prices.

In its most generic sense, the user-centric view in telecommunications considers that the users are free from subscription to any one network operator and can instead dynamically choose the most suitable transport infrastructure from the available network providers for their terminal and application requirements [7]. One envisions that in future telecommunication paradigm, the decision of interface selection will totally be delegated to the mobile terminal enabling end users to exploit the best available characteristics of different network technologies and network providers, with the objective of increased satisfaction. The generic term satisfaction can be interpreted in different ways, where a natural interpretation would be obtaining a high quality of service (QoS) for the lowest price. In order to more accurately express the user experience in telecommunications, the term QoS has been extended to include more subjective and also application-specific measures beyond traditional technical parameters, giving rise to the quality of experience (QoE) concept. QoE reflects the collective effect of service performances that determines the degree of satisfaction of the end user, e.g., what the user really perceives in terms of usability, accessibility, retainability, and integrity of the service. Until now, seamless communications is mostly based on technical network QoS parameters, but a true end-user view of QoS is needed to link between QoS and QoE. While existing 3GPP or IETF specifications describe procedures for QoS negotiation, signaling, and resource reservation for multimedia applications, such as audio/video communication and multimedia messaging, support for more advanced services, involving interactive applications with diverse and interdependent media components, is not specifically addressed. Such innovative applications, likely to be offered by third-party application providers and not the operators, include collaborative virtual environments, smart home applications, and networked games. Perceived quality problems in future internet might lead to acceptance problems, especially if money is involved. For this reason, the subjective quality perceived by the user has to be linked to the objective, measurable quality, which is expressed in application and network performance parameters resulting in QoE. Technical Report 126 of the DSL Forum (Digital Subscriber Line Forum) is a good source of information on QoE for three basic services composing the so-called triple play services. One way to achieve QoE assessment is to perform subjective tests with panel of humans, which is not an attractive solution in online optimization, another approach may be to use objective testing to predict the MOS value of a service. However such approaches require original signals (for real-time applications e.g., ITU-T objective measurement standards like PESQ, E-model etc.) and are computationally complex. In addition these approaches do not capture user-satisfaction (user-preferences) based on the non-technical or economical parameters specifically pricing, reputation of operators etc.

Several research contributions on meeting the user QoS and bandwidth requirements are present in the literature, most of them mainly focus on homogeneous service demands, fairness etc. by suggesting *radio resource management* schemes and *scheduling algorithms* [17].[6] compares several scheduling algorithms for real-time and non-real time applications, similarly [14] suggests the QoS aware packet scheduling for real time multi-media traffic, and a simple priority order queue mechanism for non-real time applications. [9], [15] discuss the problem of utility based throughput allocation and load balancing, where the earlier reference restricts the utility to linear behavior, and the later formulates the objective as network wide utility function balancing network throughput and load distribution. User-centric network selection approaches based on various approaches including *policy based*, *fuzzy logic based* etc. are discussed in [2], [3], [10], [4]. However most of the research literature either formulate the network selection problem as a static optimization problem or theoretically assume that user satisfaction function for any application follows a function, and these assumptions are not supported by the validation that represents the realistic user satisfaction.

To address these issues we propose users utility function, that captures user satisfaction for *real-time* and *non-real-time* applications with respect to both *technical* and *non-technical*

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attributes. The estimated MOS outcome from utility function can then be used for network selection decision making. *The user network selection decision making can be based on maximizing the utility function and user utility also drives the operator strategies*. We in this work validate the proposed utility function by comparing MOS value curves attained from the proposed utility function to the ones obtained from objective measurement techniques and study the relationships between them.

II. PROPOSED UTILITY FUNCTION

We capture the user satisfaction using utility function, the term *utility* comes from the field of Economics. Utility is an abstract concept and is derived largely from Von Neumann and Morgenstern [11]. It is designed to measure the user satisfaction. A *utility function* measures users relative preference for different levels of decision metric attribute values. Thus preference relation can be defined by the function, say $U : X \longrightarrow \Re$, that represents the preference for all x and $y \in X$, if and only if $U(x) \geq U(y)$. Basically a utility function should satisfy non-station and risk aversion properties [11].

Let $U_i(b_{k,c}, Q_{k,c}, \pi_{k,c})$ represents the utility function of user i, then: $U_i(b_{k,c}, S_{k,c}, \pi_{k,c}) :=$

$$v_i(b_{k,c}) \prod_{l \in L} u_{il}(t_{c,k})^{w_l} . \pi_{c,k} + \sum_{j \in J} w_j u_{ij}(Q_{c,k}) \qquad (1)$$

User utility function is the function of offered bandwidth $b_{k,c}$, offered associated satisfaction attributes S_k^c (where $S_k^c = \{Q_{c,k}, t_{c,k}\}$), and the service price $\pi_{k,c}$. Here $k \in \Theta$ represents the finite set of user types (We consider three types of users namely Excellent, Good, and Fair). $w_j \in J$, $w_l \in L$ represents the weights of parameter j and l, these attributes are detailed in later section.

We decompose the user utility function into four components, namely i) bandwidth dependent utility component, ii) associated dependent attributes utility component, iii) associated independent attributes utility component, and iv) price dependent utility component.

A. Bandwidth dependent utility - Availability of bandwidth / transmission data rate plays a key role in evaluating the user QoE, therefore most of the literature work focuses on throughput optimization. However amount of bandwidth is strictly driven by the application types and user preferences. Application specific bandwidth requirements are well studied in the literature and standards documentation, however user preferences over the bandwidth requirements is a subjective quantity and depends on the type and context of users. In this connection, we characterize the proposed user types as; i) Excellent users - the users who prefer quality more than the service price, ii) Good users - the users who stand mid-way between the quality and price, and iii) Fair User - the users who values the service cost more than the service quality, indexed by k. The bandwidth dependent utility component explicitly captures user satisfaction for offered bandwidth values to different user and service types and is given by:

$$u_i(b_k^c) = \begin{cases} 0 & \text{if } b_k^c < \underline{b}_c^k \\ \mu_{k,c} \frac{1-e^{-\beta_c (b_k^c - \underline{b}_k^c)}}{1-e^{-\beta_c (\overline{b}_k^c - \underline{b}_k^c)}} & \text{if } \underline{b}_c^k < b_k^c < \overline{b}_k^c \\ \mu_{k,c} & \text{if } b_k^c \geq \overline{b}_k^c \end{cases} \qquad (2)$$

where $\mu_{k,c}$ is the maximum achievable MOS for the service class c, and user k. β_c represents the sensitivity of application c towards the amount of bandwidth, i.e. $\beta_{real-time-application} > \beta_{non-real-time-applications}$. The value of β is scaled between the value range $[0, 1]$, and for different k type users, $\overline{b}_{excellent} > \overline{b}_{good} > \overline{b}_{fair}$ and $\mu_{excellent} > \mu_{good} > \mu_{fair}$.

B. Associated dependent attributes utility - The term *dependent* here refers to the dependency on the bandwidth. This component of the user utility function is the function of *delay* and *packet loss* QoS metric parameters. Since both the mentioned parameters can be normalized into the *the lower the better* expectancy, therefore we capture the user satisfaction for these parameters as:

$$\prod_{l \in L} u_{il}(t_{c,k})^{w_l} := \left(\begin{cases} 1 & \text{if } l_{k,c} < \bar{l}_{k,c} \\ e^{l_{k,c} \zeta_{k,c}(l)} & \text{if } l_{k,c} \geq \bar{l}_{k,c} \end{cases} \right)^{w_{l,c}} \qquad (3)$$

where $t_{k,c}$ represents the finite set containing $l \in L$ dependent variables namely *delay* and *packet loss*. $\zeta_{k,c}(l)$ represents the sensitivity of user satisfaction towards the increasing values of l. $w_{l,c}$ (driven by the application type) represents the weighted contribution of utility degradation introduced by attribute l. $\bar{l}_{k,c}$ is the ideal attribute values for which the user k for any class of service c has the maximum achievable utility.

C. Associated Independent attributes utility - This component of user utility is the function of various attributes like *reputation of operator*, *security*, *battery life* etc. These attributes are of diverse scope and can be normalized on expectencies of *the lower the better*, *the higher the better*, or *the nominal the better*. User satisfaction for this component is purely attribute dependent i.e., the decision of using linear, exponential, lograthmic functions and control parameters depend on the attribute under consideration e.g., for security parameter, a function like bandwidth dependent utility may be used.

D. Price based utility - In addition to technical, user satisfaction is also influenced by the economical parameters. One can not neglect the importance of this parameter in decision making for network selection, when it comes to cost-sensitive user types (e.g., fair users). We capture the satisfaction of different user types with respect to service prices as: $u_k(\pi_{k,c}) = \tilde{\mu}_{k,c} - \frac{\tilde{\mu}_{k,c}}{1-e^{\tilde{\pi}^{c,k}}} e^{-\pi\epsilon}$, where $\tilde{\mu}_{k,c}$ represents the maximum satisfaction level of user type k, and $\tilde{\pi}^{c,k}$ is the private valuation of service by user, and ϵ represents the price sensitivity of user.

III. EXPERIMENTATION AND UTILITY VALIDATION FOR DIFFERENT SERVICES

A. Real-time VoIP applications

Streaming and conversational traffic classes can be combined in real-time applications, which are commonly termed as *inelastic* or *rigid* applications. Generally real-time applications are constrained by minimum amount of bandwidth i.e., application is admitted only when the demand for minimum required bandwidth is met. Such stringent requirement on bandwidth are represented by step like function, which results in a very narrow transition region between the two states (fully satisfied, unsatisfied). Such transition region is captured by the value of β of user utility function given in equation-2. This transition region represents

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very narrow required bandwidth range for different real-time applications e.g., audio broadcasting demands $60-80Kbs$, video broadcasting demands $1.2Mbs - 1.5Mbs$ with MPEG1 coding standard.

A.1. VoIP objective measurement - In order to capture user satisfaction using simulation measurement methodology, we set up a simulation scenario with heterogeneous wireless technologies, and run a lengthy rounds of simulations to analyze the user satisfaction for different values of delay and packet losses, when she is associated to different codecs using ITU-T PESQ and modified E-models standard models. It should be noted service affecting factors [1] in addition to delay and packet loss are out of the scope of the objective measurements.

A.1.1. OPNET simulation setup - The components involved in the simulation setup include; i) impairment entity - we develop an impairment entity that introduces specified packet delay, packet loss and is also able to limit bandwidth available to a voice communication by performing bandwidth shaping using token bucket algorithm. ii) LTE radio access network, iii) WLAN radio access network, and iv) transport network. The simulation is setup such that the impairment entity resides between the caller and the callee, and introduces various delays and packet losses during the life of a VoIP call.

Note - The packet delay values in the simulation include only codec delay and transport network delay excluding fixed delay components e.g., equipment related delays, compression decompression delays and other internetwork codec related delays etc.

A.1.2. Simulation Results - We analyze the results for three different codecs namely i) G.711, ii) GSM EFR, and iii) G.729, which are characterized by their data rates. Each codec in a lossless (lossless is an ideal scenaio, where packetloss and delay values are idealy zero.) condition achieves the maximum MOS, $\overline{MOS}_c$, such that $\overline{MOS}_c \neq \overline{MOS}_{\tilde{c}}$. This characteristic of codec dictates that a user, when associated with a codec c, will have lossless MOS equal to $\overline{MOS}_c$ unless he is switchedover to the codec $\tilde{c}$. Codec switchover results in step-function like $\overline{MOS}$ value of user in a lossless scenario. This is depicted in Figure-1, which clearly shows that even in a lossless scenario, codec switchover introduces a marginal gain or loss in the MOS value. We now discuss a more realistic scenario, where user satisfaction is influenced by the packet loss and delay values, i.e. user associated with a specific codec c, with the MOS, $\overline{MOS}_c$, experiences delay and packet loss in the communication system. The consequence of system impairments is a degraded service, which in turn has negative impact on user satisfaction. In simulations, we use impairment entity to introduce customized delays and packet losses in the system and study the impact of parameter values on user satisfaction. Figure-2 shows the impact of delay and packet loss values on user satisfaction for $G.711$, $G.729$, and GSM EFR codec, As can be seen that all the codecs lead to different MOS values for different values of packet loss and delays.

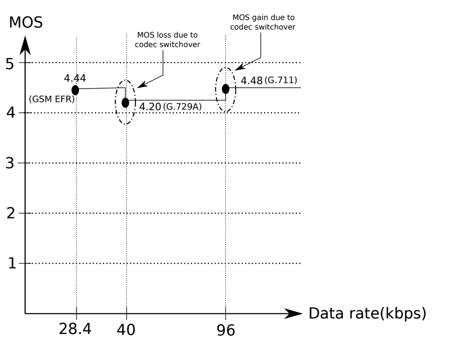

Figure 1. VoIP MOS for Loss-less Scenario with different codecs

Table I
UTILITY CONTROL PARAMETER VALUES FOR VOIP AND FTP APPLICATIONS

Application	Codec/size	$v_i(b_{k,c})$	$\zeta(pl)$	$\zeta(d)$	w_{pl}	w_d
Voice	G.711	4.48	0.03	0.0075	0.75	0.25
	G.729	4.20	0.03	0.0075	0.75	0.25
	GSM EFR	4.44	0.075	0.0033	0.4	0.6
FTP	20Mb	5.0	0.99	0.0429	0.5	0.5
Video	JM	3.98	0.031	0.011	0.7	0.3

A.2. Proposed utility function for VoIP applications - Although the proposed utility function in equation-1 captures user satisfaction for both technical and non-technical aspects, we limit here the scope of utility function to the technical part only so that we can validate it against the results obtained from the objective measurements. Since the objective measurements are carried out for different codecs and different packet loss, and delay values, therefore first two components of the proposed user utility (equation-1) are adequate to capture user satisfaction.

$$U_i(b_{k,c}, S_{k,c}) := v_i(b_{k,c}) \prod_{l \in L} u_{il}(t_{c,k})^{w_l} \tag{4}$$

where $v_i(b_{k,c})$ represent codec data rate, and this utility is given by a step like function. $v_i(b_{k,c})$ is tuned by the $\prod_{l \in L} u_{il}(t_{c,k})^{w_l}$ utility component. The control parameters for this utility component take different values for different codecs, which are given in the Table-I.

A.3. Validation - In this section we validate the proposed utility function for VoIP application by comparing the plots attained from the utility function to the plots we get from objective measurements (simulation results), this is shown in Figure-3. As evident from the figure that most of the points overlap well i.e., few points map exactly, for few MOS values the proposed utility function partially underestimates or overestimates the objective MOS values. This is further elaborated in Figure-4, where the correlation clearly strengthens the claim that proposed utility function for VoIP applications estimates the user satisfaction with appreciable confidence level.

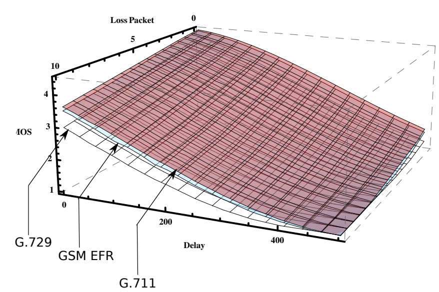

Figure 2. MOS values for different codecs

B. Non-real-time applications

Interactive and Background traffic classes can be combined in *non-real-time* applications, which are commonly termed as *elastic* applications, these applications are further divided into *symmetric* and *asymmetric non-real-time* applications. Generally *non-real-time* applications do not have stringent requirements

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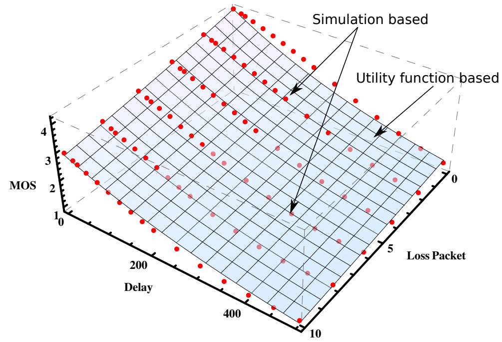

Figure 3. Mapping of objective and utility function measurements for VoIP application

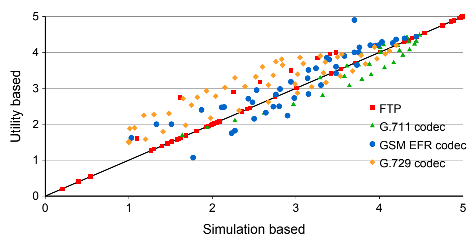

Figure 4. Correlation of objective and utility function for VoIP application

for bandwidth and delay. Such application can run even with minimal amount of available bandwidth, therefore, call admission control may not be needed in this case. TCP based non-real time applications ensure the error free delivery on the cost of waiting time introduced by TCP ARQ mechanism. This necessitates the proper investigation of *how much is the influence of packet loss and packet delay on achievable TCP throughput?* According to literature, TCP throughput is inversely proportional to round trip time of a network. In case of negligible packet loss rate following relation holds i.e., throughput $\leq$ (TCP buffer size) / RTT, where RTT is TCP segment round trip time. But if there are considerable packet losses than following relation holds, i.e. throughput $<$ (MSS/RTT)*(1/sqrt(PLR)), where MSS is the maximum TCP segment size, RTT is the round trip time and PLR is the packet loss rate. However above relations only show the upper bound of achievable TCP throughput. In order to investigate the more concrete throughput values of TCP in the presence of certain packet delay and packet loss rate, extensive simulations runs are provisioned.

User satisfaction metric for Non-real-time application - The performance metric to measure user satisfaction for non-real-time applications include *throughput*, *download response time* [13], and *MOS*. Although different, these performance measuring parameters are correlated. In this paper, we choose the MOS value as a performance metric for FTP applications. The motivation for selecting MOS as the performance metric is to have a generalized and common metric for different services.

B.1. FTP objective measurements - On the similar lines to the VoIP application objective measurmenets, we set up simulation scenario with heterogeneous wireless technologies and run lengthy rounds of simulations to analyze the user satisfaction for different values of delay and packet loss.

B.1.1. Simulation Settings and Methodology - This simulation environment also involves the *impairment entity*, and *LTE* in the similar fashion as discussed in the VoIP simulation settings, however in this case, the caller and the callee are replaced by the FTP server and FTP client. FTP server and client are connected through LTE access network. In our settings, an FTP client downloads a heavy file (of 20MB) through LTE access network. The choice of file size here is dictated by the facts; i) slow start effect of TCP can be ignored, ii) correlation of TCP throughput and distribution of packet losses within a TCP can be reduced. We artificially inject the packet delays by using the impairment entity, packet delays follow *Normal distribution*. A *bandwidth shaping* of 8Mbps is performed at a router in LTE transport network. We use the most widely used TCP flavor *New Reno* with receiver buffer size of 64KB. Moreover *windows scaling* option of TCP is disabled, *window scaling* option allows TCP maximum congestion window size to grow beyond 64KB. Due to deployment of accumulated acknowledgements, TCP is not very sensitive to the loss of few percent of acknowledgement packets in uplink direction, therefore, effect of packet loss is investigated only in downlink direction. It should also be noticed that processing delay and packet losses in network components (other than impairment entity) are negligibly small. Packet losses are injected based on Bernoulli distribution, packet delays are actually RTT values.

B.1.2 Simulation results - We analyze the impact of packet loss and delay values on the *user throughput* as shown in Figure-5. However *user throughput* does not directly show the user satisfaction, in this connection, we need to translate the *user throughput* values into *user satisfaction*. We carried such translation using the *throughput to MOS mapping approach* detailed below.

Throughput to MOS mapping - For such mapping we assume that a user of type k is subscribed to an amount of bandwidth b_k, such that the user remains fully satisfied (has the $MOS = \overline{MOS}$) as long as he receives the bandwidth b_k or $b_k + \epsilon$, and for any bandwidth less than b_k, the user satisfaction degrades and user reaches the *irritated state*, when the received bandwidth is $\underline{b}_k$. We term the bandwidth range $[\overline{b}_k - \underline{b}_k]$ as *feasible bandwidth range* for user k. This further necessitates a function of degradation and scaling the user satisfaction. We scale the user satisfaction for FTP applications on the same lines as in the case of VoIP MOS values i.e., [1-5], whereas the bandwidth dependent component of utility (equation-1) is used as the degradation function between fully satisfied and fully irritated states of user. For mapping the throughput results shown in Figure-5 into MOS scaled results, we set the control parameters of equation-2 to the following values; $\overline{b}_k = 1.265 \times 10^7 kbps$, $\underline{b}_k = 20250.449 kbps$, $\beta = 0.000006$, and $\alpha = 5$. The consequence of such mapping is depicted in Figure-6, which represents the *user satisfaction* in terms of MOS values for different achieveable data-rate values.

B.2. Utility function representation of FTP user satisfaction - We capture the user satisfaction for FTP application using the proposed utility function given in equation-1. From the simulations, what we get is the user throughput and impact of different packet loss and delay on the throughput as shown in Figure-5. The $v_i(b_{k,c}) \prod_{l \in L} u_{il}(t_{c,k})^{w_l}$ utility components capture the user satisfaction that is comparable to measurement results obtained from the objective testings.

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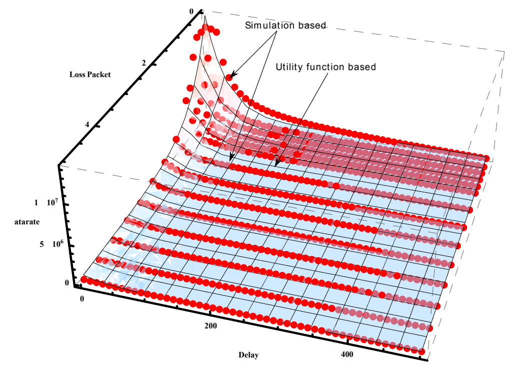

Figure 5. Overlapping of utility-based data rate values over the simulation-based data rate for FTP application

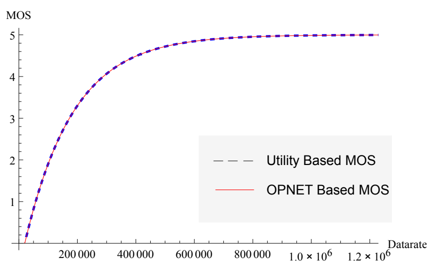

Figure 6. Comparison of Utility Based and Objective Measurement Based MOS values for FTP application

Let us first discuss the case, when we are not mapping the throughput over the MOS values, in this case $v_i(b_{k,c})$ component of the utility function takes constant value that represents the *throughput in lossless conditions* i.e., $\bar{b}_k = 1.265 \times 10^7 kbps$, which is shaped by the $\prod_{l \in L} u_{il}(t_{c,k})^{w_l}$ component of the utility function. The results attained from the utility-based measurements are overlapped over the objective measurement results, and it is observed that these map very well, as can be seen from the Figure-5. We also scaled the throughput by mapping it over the MOS (remember that similar parameter values for mapping in utility function based measurement i.e., $\bar{b}_k = 1.265 \times 10^7 kbps$,$\underline{b}_k = 20250.449 kbps$, $\beta = 0.000006$, and $\alpha = 5$ are used), the scaled result is presented in Figure-6. The results show that the utility function estimates the user satisfaction similar to the objective measurement and hence validate the proposed utility function.

C. Video streaming applications

In video streaming the most commonly used objective evaluations produce PSNR (peak signal to noise ratio) and Ssim (Structural similarity) as output video quality metrics.

PSNR - *PSNR defines the ratio between the maximum possible power of a signal and the power of corrupting noise that affects the fidelity of its representation. When comparing two video files, the signal is the original file and noise is the error which occurs due to compression or during transmission over the network. In the context of video quality evaluation, PSNR is taken as an approximation to human eye perception of image quality. It is measured in decibel units (dB).*

Ssim *- The Structural Similarity (Ssim) index is a novel method for measuring the similarity between two images. It takes the original undistorted image as a reference and provides the quality measure of the compressed/distorted image. Ssim index value ranges from -1 to 1. The higher the Ssim index value the higher the similarity between the two comparing images. For videos Ssim index is computed image by image [16].*

C.1. Methodology and simulation setup - In this work, we use PSNR as video quality metric owing to its widespread use in scientific literature. We calculate MOS value based on PSNR value. There are several parameters which decide the sensitivity of end user video quality to network impairments, such as: i) Type encoding - It is due to the fact that encoding schemes differentiate among frames based on their importance in the decoding process. Hence a loss of more important key frames deteriorates reconstructed video quality much more than the loss of less important non-key frame. ii) Error concealment method, iii) Frames per second, iv) MTU size of transport network, and v) Pre-filtration of codec etc. However a thorough study of the impact of all other above parameters on video file transmitted over a wireless network in the presence of additional IP impairments is beyond the scope of this work. We consider a reference video sequence called *Highway* for this work. The motivation to use this video sequence its repeated reference in a large number studies in video encoding and quality evaluation e.g. *Video Quality Experts Group*[5]. This video sequence has been encoded in $H.264$ format using the JM codec [14] with CIF resolution (352×288) using a target bit rate of $256kbps$. $H.264$ codec has been selected because its widespread use can be seen in future communication devices. The reference video sequence has total 2000 frames and frame rate of $30fps$. Key frame is inserted after every $10th$ frame which provides good error recovery capabilities. An excellent video quality is indicated by $38.9dB$ as an average PSNR value of encoded video sequence. The video file is transmitted over the IP network considering MTU size of 1024 bytes. At the receiving end, video file is reconstructed from received IP packets. The reconstructed video file might have errors due to packet losses and delays in the transport IP network. Results presented in this work have been taken from OPNET simulation setup.

C.1.1 OPNET simulation setup - This simulation setup has two parts, the first part includes implementation of E-UTRAN, EPC network entities of LTE access network, and the second part of the simulation set-up is derived from EvalVid [8]. EvalVid is a framework which can be used for video quality evaluation. It provides both PSNR as well as MOS values of reconstructed video file. The motivation to use Evalvid is its flexibility to be used in conjunction to simulation environments like ns-2 and OPNET. None of the other available video evaluation tools provide such an interface. Target of this task is to get video quality metric for video file which is transmitted over LTE access network. The transport network part of LTE artificially introduces IP impairments to the transmitted video file. Here the IP impairment entity uses *Normal distribution* for packet delays and packet delay variations. This choice is based on empirical study of big IP networks. Moreover packet losses are injected using *Bernoulli distribution*.

Following sequence of action leads to video quality metric for a particular value of mean packet delay and packet loss rate. EvalVid tools are used to generate a file which includes information about packets (e.g. *packet type*, *size*, *count* etc). These are the packet which would carry video frames if video file is transmitted over an IP network in real world scenario.

Packet size and *type* information is used to transmit the same number, type and size of packets over LTE access network using OPNET simulator. IP impairment entity injects specified *packet delays* and *packet loss rate* in the above generated packet stream. Associated information of received packets (e.g. *packet end-to-end delay*, *jitter*, *type* and *sequence number of lost packets*) is used to reconstruct video file. This task is performed using EvalVid tools. *Play-out buffer length* of $250ms$ is used in this step. The reconstructed video file is then compared against the raw formatted reference video file to compute PSNR values frame by frame. It tells about the noise produced by both *encoding* as well as *transmission errors*. Video quality metric is computed by evaluating the difference between quality of $H.264$ encoded video file and reconstructed video file. In MOS of every single frame of the reconstructed video file is compared to the MOS of every single frame of the reference video file. In the end average MOS value of whole video file is output. For PSNR to MOS translation following table is used [12]. The simulation results are shown in Figure-7, depicting the *video user* satisfaction for different packet loss and delay values.

C.3. Utility function representation of video user satisfaction - We capture the user satisfaction for video streaming application using the proposed utility function given in equation-1 on very similar lines to that of VoIP applications(for details, refer to VoIP application section) The first two components of user utility function estimate the user satisfaction for different values of packet loss and delays. For video application, the control parameters of the proposed utility function are listed in Table-I. In order to validate the proposed utility function, we overlap the results obtained from the utility-based measurement over the simulation-based measurements. It is observed that the proposed utility function estimates the *video user* satisfaction very similar to the satisfaction values from experimentation, this is evident from the Figure-7.

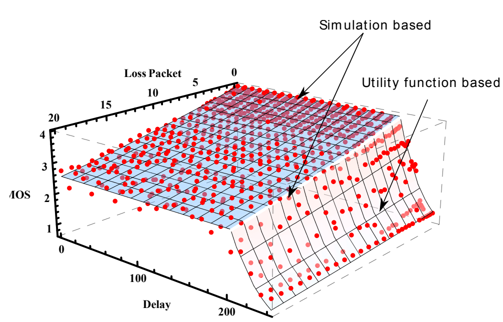

Figure 7. Overlapping of utility-based MOS values over the simulation-based MOS values for video application

Given the validation results in Figures-4,6,5,7, we confidently claim that the proposed utility function estimates the user satisfaction similar to subjective and objective measurements.

IV. Conclusion

In this paper we translated the user satisfaction in utility function. The proposed utility function captures user preferences over both technical and non-technical decision attributes. We carried out the extensive simulations to compute the user satisfaction for different service types including voice, FTP, and video streaming. Factors effecting the user perceived service quality have been discussed in detail for different test scenarios. We proposed the utility function that estimates the user satisfaction for different applications, and also validated the proposed utility function by comparing the utility-based results against the results attained from the objective measurements. We plan to extend this work to model operator utilities and find the local and global optimum solution for resource allocation at the operator level and network selection strategies (using proposed utility function) in different environments at user level.

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